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EUROPEAN STANDARD

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

Reference

REN/DECT-00360

Keywords

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ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - APE 7112B
Association à but non lucratif enregistrée à la
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Foreword

This draft European Standard (EN) has been produced by ETSI Technical Committee Digital Enhanced Cordless Telecommunications (DECT), and is now submitted for the combined Public Enquiry and Vote phase of the ETSI standards EN Approval Procedure.

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

Proposed national transposition dates	
Date of latest announcement of this EN (doa):	3 months after ETSI publication
Date of latest publication of new National Standard or endorsement of this EN (dop/e):	6 months after doa
Date of withdrawal of any conflicting National Standard (dow):	6 months after doa

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

"**must**" and "**must not**" are **NOT** allowed in ETSI deliverables except when used in direct citation.

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 ETSI 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 ETSI EN 300 175-1 [1] to ETSI EN 300 175-8 [8] which are relevant to these aims.

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

- part 1 [9] 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.
  The present document 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

2.1 Normative 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 referenced document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at https://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.

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

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

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

[18] Void.

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

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

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

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

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

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

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

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

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

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

[30] Void.

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

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

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

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

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

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

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

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

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

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

[41] Recommendation ITU-T P.501: "Test signals for use in telephony and other speech-based applications".

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

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

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

[45] IEC 61260-1: "Electroacoustics -- Octave-band and fractional-octave-band filters Part 1: Specifications".

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

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

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

[56] Recommendation ITU-T 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] Recommendation ITU-T 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".

[58] ETSI TS 103 634: "Digital Enhanced Cordless Telecommunications (DECT); Low Complexity Communication Codec plus (LC3plus)".

[59] ETSI TS 103 224: "Speech and multimedia Transmission Quality (STQ); A sound field reproduction method for terminal testing including a background noise database".

[60] ETSI TS 102 924: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for Super-Wideband / Fullband handset and headset terminals from a QoS perspective as perceived by the user".

[61] ETSI TS 102 925: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for Super-Wideband / Fullband handsfree and conferencing terminals from a QoS perspective as perceived by the user".

[62] Void.

[63] Void.

[64] ETSI TS 103 281: "Speech and multimedia Transmission Quality (STQ); Speech quality in the presence of background noise: Objective test methods for super-wideband and fullband terminals".

[65] Void.

[66] Recommendation ITU-T P.10: "Vocabulary for performance, quality of service and quality of experience".

[67] ISO 3745: "Acoustics -- Determination of sound power levels and sound energy levels of noise sources using sound pressure -- Precision methods for anechoic rooms and hemi-anechoic rooms".

[68] Void.

[69] ETSI TS 126 071: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Mandatory speech CODEC speech processing functions; AMR speech Codec; General description (3GPP TS 26.071)".

[70] ETSI TS 126 171: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; General description (3GPP TS 26.171)".

[71] ETSI TS 126 441: "Universal Mobile Telecommunications System (UMTS); LTE; Codec for Enhanced Voice Services (EVS); General overview (3GPP TS 26.441)".

[72] IETF RFC 6716: "Definition of the Opus Audio Codec".

[73] Recommendation ITU-T G.701: "Vocabulary of digital transmission and multiplexing, and pulse code modulation (PCM) terms".

[74] Recommendation ITU-T P.700: "Calculation of loudness for speech communication".

[75] Recommendation ITU-T P.341: "Transmission characteristics for wideband digital loudspeaking and hands-free telephony terminals".

[76] Recommendation ITU-T P.342: "Transmission characteristics for narrow-band digital loudspeaking and hands-free telephony terminals".

[77] ETSI TS 103 706: "Digital Enhanced Cordless Telecommunications (DECT); Advanced Audio Profile".

2.2 Informative 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 referenced document (including any amendments) applies.

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

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

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

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

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

[i.17] Void.

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

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

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

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

[i.22] Recommendation ITU-T 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] Recommendation ITU-R 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 https://tech.ebu.ch/publications/sqamcd.

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

[i.31] Recommendation ITU-T 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] Void.

[i.34] ETSI EG 202 396-3: "Speech and multimedia Transmission Quality (STQ); Speech quality performance in the presence of background noise; Part 3: Background noise transmission - Objective test methods".

[i.35] Recommendation ITU-T P.310: "Transmission characteristics for narrow-band digital handset and headset telephones".

[i.36] TIA-920.130-A: "Telecommunications Telephone Terminal Equipment Transmission Requirements for Wideband Digital Wireline Telephones with Headset".

[i.37] Void.

3 Definition of terms, symbols and abbreviations

3.1 Terms

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

2-wire interface: telephony analog interface over 2-wires used in the local loop

4-wire interface: 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

NOTE: 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

codec gross bit rate: total number of bits including codec's source and channel coding

codec net bit rate: total number of bits per second for codec's source coding only

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 Recommendation ITU-T 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: telecommunications device used to reduce the echo (sometimes "acoustic echo suppressor")

NOTE: 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 ETSI EN 300 175-1 [1] to ETSI 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 centre of the artificial head with no artificial head present)

fullband speech: transmission of speech with a nominal pass-band wider than 50 Hz to 14 000 Hz, usually understood to be 20 Hz to 20 000 Hz (adapted from Recommendation ITU-T P.10 [66])

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

high-resolution mode: LC3plus operation mode for higher bit rates, higher precision and signal bandwidth up to the Nyquist frequency

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 Recommendation ITU-T 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

Low Complexity Communication Codec plus (LC3plus): standard for narrowband to fullband low delay audio communication designed for very high quality communication application including all kind of audio signals, e.g. speech and music, as defined by ETSI TS 103 634 [58]

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): point 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 ETSI EN 300 175-1 [1] to ETSI 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: transmission of speech with a nominal pass-band wider than 100 Hz to 7 000 Hz, usually understood to be 50 Hz to 14 000 Hz

NOTE: Adopted from Recommendation ITU-T P.10 [66].

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

ultra-band: speech or audio sampled at 96 kHz

NOTE: According to ETSI TS 103 634 [58].

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: transmission of speech with a nominal pass-band wider than 300 Hz to 3 400 Hz, usually understood to be 100 Hz to 7 000 Hz

NOTE: According to Recommendation ITU-T P.10 [66].

Wi-Fi®: family of radio technologies that is commonly used for the Wireless Local Area Networking (WLAN) of devices which is based around the IEEE 802.11 [i.27] family of standards

3.2 Symbols

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

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 Recommendation ITU-T 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
ep<x>	error protection classes, x = 1…4
I_e	equipment Impairment factor
$L_{S,min}$	minimum activation level (Sending Direction)
Ssi(diff)	the difference of the send sensitivities between diffuse and direct sound
Ssi(direct)	the sending sensitivities for the direct sound
$T_{r,S,min}$	built-up time (Sending Direction)
Z_R	terminating impedance of a transmission line

3.3 Abbreviations

For the purposes of the present document, the abbreviations given in Recommendations ITU-T P.10 [66], G.701 [73], ETSI EN 300 175-1 [1] and the following 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
AMR-NB	Adaptive Multi-Rate codec (NB)
AMR-WB	Adaptive Multi-Rate codec (WB)
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.c.	direct current
D/A	Digital/Analog
DECT	Digital Enhanced Cordless Telecommunications

DLC	Data Link Control
DRP	ear Drum Reference Point
DTX	Discontinuous Transmission
e.m.f	electromotive force
EP, ep	Error Protection
ER	Error Resilient (MPEG)
ERP	Ear Reference Point
ES	End System
EUT	Equipment Under Test
EVS	Enhanced Voice Service
FB	Fullband
FBHR	Fullband in High Resolution mode of LC3plus
FBLFE	Fullband in LFE mode of LC3plus
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
IP	Internet Protocol
IRT	Institut für Rundfunktechnik
ISDN	Integrated Services Digital Network
LAN	Local Area Network
LC3	Low Complexity Communication Codec
LD	Low Delay (MPEG)
LFE	Low Frequency Effects
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
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
NB	Narrowband
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
ppm	parts per million
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

NOTE: See Recommendation ITU-T G.726 [24].

SLR	Sending Loudness Rating
SLR_H	Sending Loudness Rating of the Handset
SQAM	Sound Quality Assessment Material
SR	Reconstructed Signal

NOTE: See Recommendation ITU-T G.726 [24].

STMR	SideTone Masking Rating
SWB	Super-wideband
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
UB	Ultra-Band
UBHR	Ultra-Band in High Resolution mode of LC3plus
UDP	User Datagram Protocol
USB	Universal Serial Bus
VoIP	Voice over IP
WB	Wideband
WRS	Wireless Relay Stations

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 6.2.4.

5 General test requirements

5.0 General

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), ETSI EN 300 700 [50], tested according to the DECT test specification (see ETSI 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 ETSI 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 ETSI 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" defining the overall audio behaviour between testable reference points (see ETSI 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 ETSI EN 300 175-1 [1].

In this case, a test equipment capable of emulating a PT or FT that conforms to ETSI EN 300 175-1 [1] to ETSI 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.

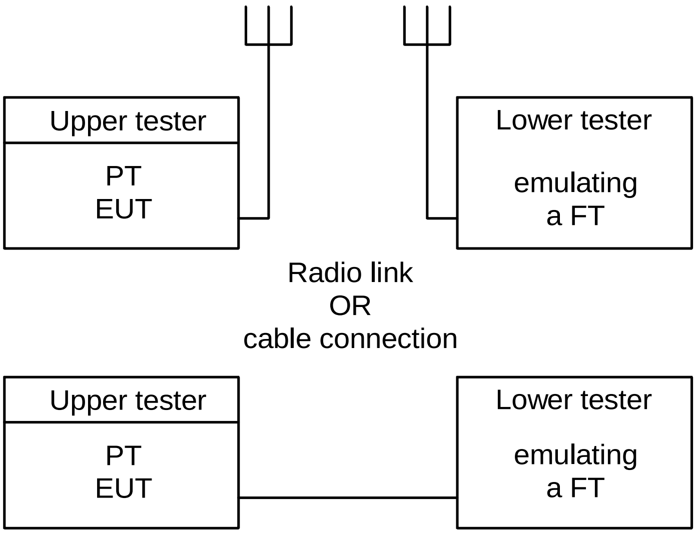


Figure 5.1: The EUT is a PT

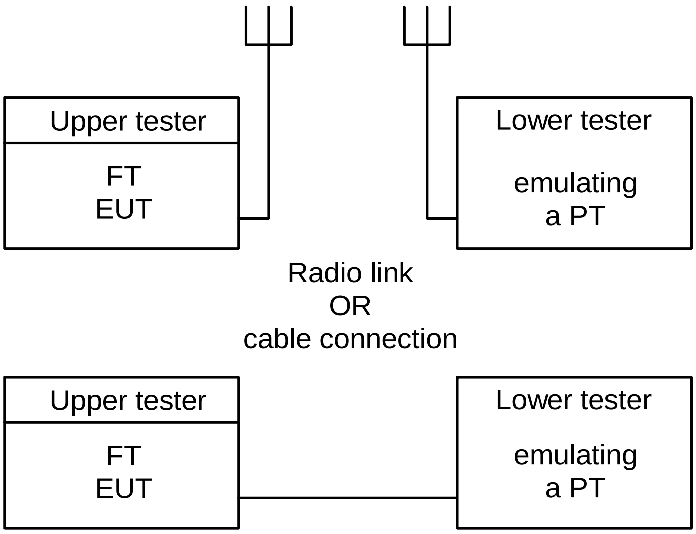


Figure 5.2: The EUT is an FT

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 6.2. 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 ETSI 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 ETSI 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 ETSI 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 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 Acoustical environment

In general, different acoustical environments have to be taken into account: either room noise and background noise are an inherent part of the test environment or room noise and background noise shall be eliminated to such an extent that their influence on the test results can be neglected.

Unless stated otherwise, measurements shall be conducted under quiet and "anechoic" conditions. Considering this, the test laboratory, in the case where its test room does not conform to anechoic conditions as given in Recommendations ITU-T P.310 [i.35] for NB handset/headset, ITU-T P.311 [38] for WB handset/headset, ITU-T P.342 [76] for NB handsfree/loudspeaking, ITU-T P.341 [75] for WB handsfree/loudspeaking, respectively, has to present difference in results for measurements due to its test room. In case where an anechoic room is not available the test room has to be an acoustically treated room with few reflections and a low noise level.

Depending on the distance of the transducers from mouth to ear a quiet office room may be sufficient e.g. for handsets where artificial mouth and artificial ear are located close to the acoustical transducers.

However, for some headsets or handset terminals with smaller dimension an anechoic room will be required.

In cases where real or simulated background noise is used as part of the testing environment, the original background noise should not be noticeably influenced by the acoustical properties of the room.

In all cases where the performance of acoustic echo cancellers shall be tested a realistic room which represents the typical user environment for the terminal shall be used.

Unless specified otherwise, the background noise level shall be less than -64 dBPa(A) in conjunction with NC30 (ISO 3745 [67]).

For specified tests, it is desirable to have a background noise level of less than -74 dBPa(A) in conjunction with NC20, but the background noise level of -64 dBPa(A) in conjunction with NC30 shall never be exceeded.

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 Accuracy of measurements and test signal generation

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

Table 6.0: Measurement accuracy

Item	Accuracy
Electrical signal level	±0,2 dB for levels ≥ -50 dBV ±0,4 dB for levels < -50 dBV
Sound pressure	±0,7 dB
Frequency	±0,2 %
Time	±0,2 %
Application force	±2 N
Measured maximum frequency	20 kHz

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

Table 6.0a: Accuracy of test signal generation

Quantity	Accuracy
Sound pressure level at Mouth Reference Point (MRP)	±3 dB for frequencies from 100 Hz to 200 Hz ±1 dB for frequencies from 200 Hz to 4 000 Hz ±3 dB for frequencies from 4 000 Hz to 14 000 Hz
Electrical excitation levels	±0,4 dB across the whole frequency range
Frequency generation	±2 % (see note)
Time	±0,2 %
NOTE: This tolerance may be used to avoid measurements at critical frequencies, e.g. those due to sampling operations within the terminal under test.	

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 an 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 an 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 an FP. The general test methods described in clause 7 are applicable except that the ReFP and the RePP are replaced by an 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 ETSI EN 300 175-5 [5] then test of an 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], 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 Recommendation ITU-T G.711 [19] and Recommendation ITU-T 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 Recommendation ITU-T 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).

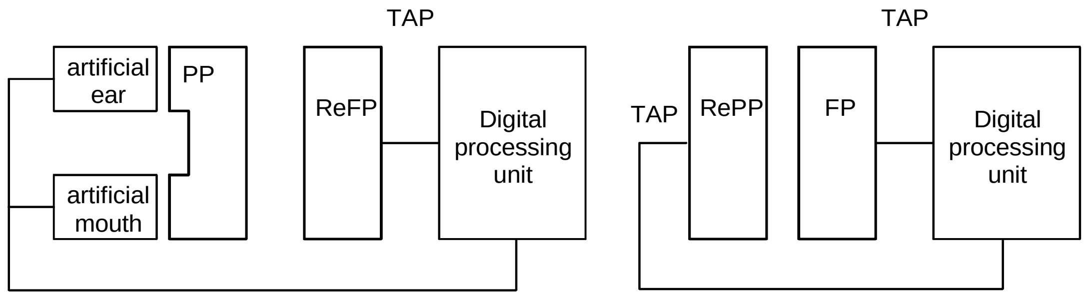


Figure 6.1: The direct digital processing approach test configurations for a separate PP and a separate FP

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).

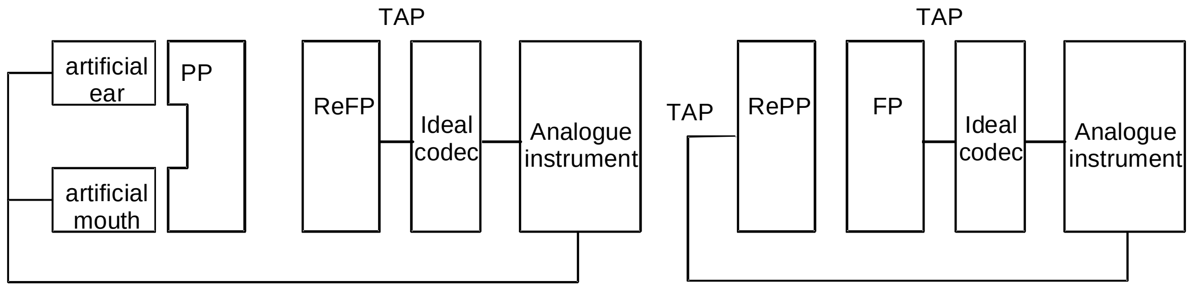


Figure 6.2: The ideal codec approach (Test configuration for a separate PP and a separate FP)

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-wire 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.

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 2 (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 PP audio features as follows:
  - 1d for narrow band test
  - 2c for wideband test
  - 5a for super-wideband test
  - 7a for fullband test
  - 8a for ultra-band test

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 Recommendation ITU-T G.726 [24] at 32 kbit/s.

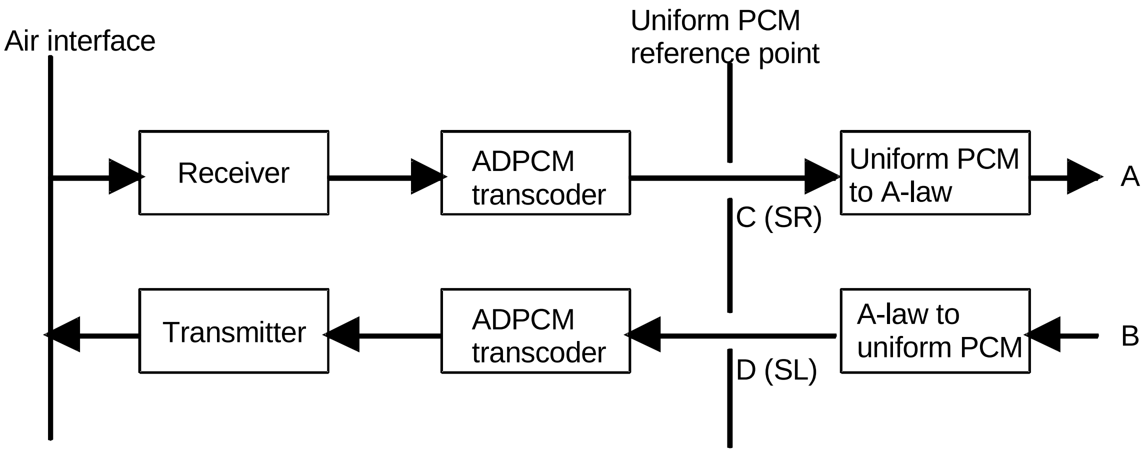


Figure 6.3: Implementation of Reference FP and PP for narrowband service with codec G.726

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 (Recommendation ITU-T G.711 [19]).

The relationship between the PCM encoding law and the audio signal level is defined in Recommendation ITU-T G.711 [19]. The theoretical load capacity of Recommendation ITU-T 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 Recommendation ITU-T 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 5.1.7).

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 Recommendation ITU-T 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.

NOTE: Recommendation ITU-T 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, MPEG-AAC and LC3plus)

6.7.0 General

The ideal codec approach uses a codec to convert the 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, G.711 and LC3plus in NB mode

This clause is applicable when the following codecs are used:

- Recommendation ITU-T G.726 [24] narrowband codec operating at 32 kbit/s (see ETSI EN 300 175-8 [8], clause 5.1);
- Recommendation ITU-T G.711 [19] narrowband codec operating at 64 kbit/s (see ETSI EN 300 175-8 [8], clause 5.2);
- ETSI TS 103 634 [58] LC3plus codec operating in NB mode at 32 kbit/s (see ETSI EN 300 175-8 [8], clause 5.6.2).

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 Recommendation ITU-T 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.

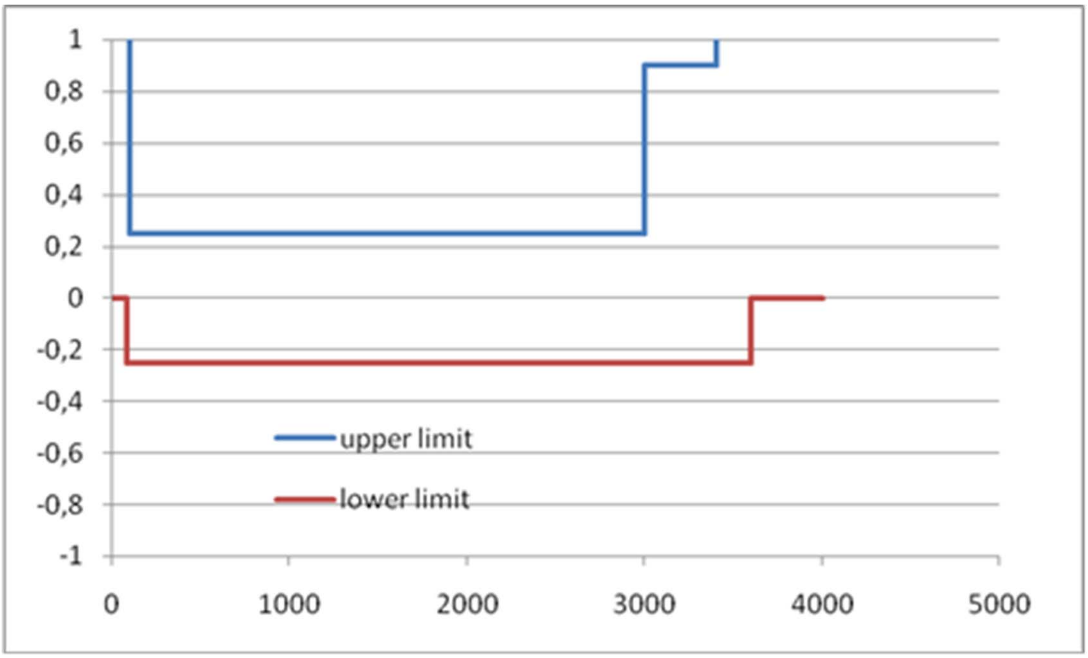


Figure 6.4: Frequency/sensitivity response of an ideal narrowband codec

6.7.2 Ideal codec for codecs G.722, G.729.1, MPEG-4 and LC3plus (wideband mode)

This clause applies when the following codecs are used:

- Recommendation ITU-T G.722 [21] wideband codec operating at 64 kbit/s (see ETSI EN 300 175-8 [8], clause 5.3);
- Recommendation ITU-T G.729.1 [25] wideband codec operating at 32 kbit/s (see ETSI EN 300 175-8 [8], clause 5.4);
- MPEG-4 ER AAC-LD [48] wideband codec operating at 32 kbit/s (see ETSI EN 300 175-8 [8], clause 5.5.3);
- ETSI TS 103 634 [58] LC3plus codec operating in WB mode at 32 kbit/s (see ETSI EN 300 175-8 [8], clause 5.6.3).

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.

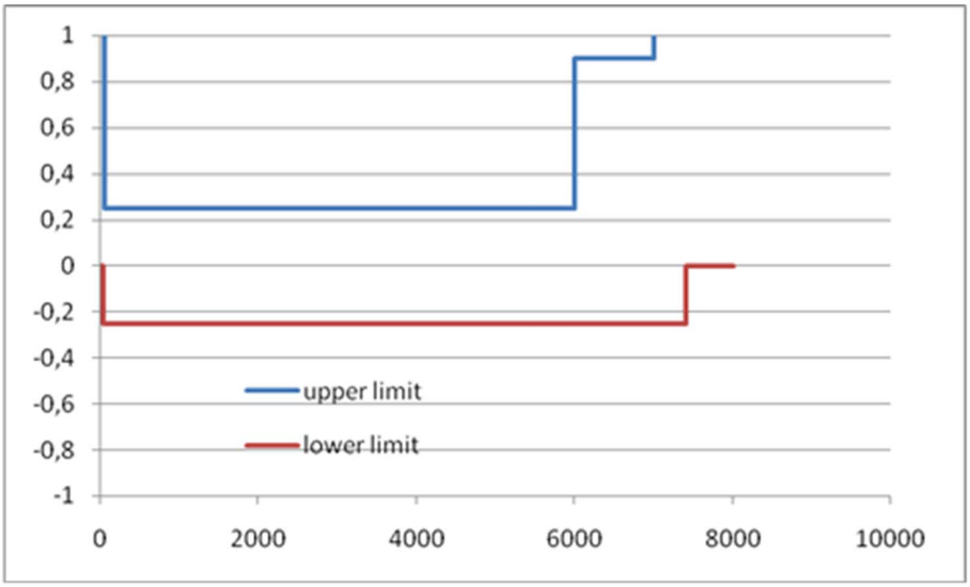


Figure 6.5: Frequency/sensitivity response of an ideal wideband codec

6.7.3 Ideal codec for MPEG-4 or LC3plus (super-wideband mode)

The ideal codec for MPEG-4 ER AAC-LD [48] super-wideband codec (see ETSI EN 300 175-8 [8], clause 5.5.2) and LC3plus [58] codec in SWB mode (see ETSI EN 300 175-8 [8], clause 5.6.4), each operating at 64 kbit/s, is for further study.

6.7.4 Ideal codec for LC3plus in fullband, FBHR, FBLFE or ultra-band mode

The ideal codec for LC3plus [58] codec in FB, FBHR, FBLFE or UBHR mode (see ETSI EN 300 175-8 [8]) is for further study.

6.8 Electro-acoustical equipment

6.8.0 General

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

6.8.1 Artificial mouth and artificial ear

The artificial mouth shall conform to Recommendation ITU-T P.51 [32]. The artificial ear shall conform to Recommendation ITU-T 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 Recommendation ITU-T P.58 [35].

This equipment will be used for test of PP type 1c, 1d, 2b, 2c, 3a, 3b, 4a, 4b, 5a, 5b, 7a, 7b, 7c, 7d, 7e and 7f.

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 Recommendation ITU-T 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 Recommendation ITU-T P.64 [36].

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

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

For making handset receiving measurements, a Type 3 artificial ear shall be used, as specified in Recommendation ITU-T P.57 [34]. Sound pressure levels could be referred to ERP using the correction factors given in tables 2a and 2b of Recommendation ITU-T 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

6.10.3.0 Super-wideband and fullband applications

Recommendation ITU-T P.58 [35] indicates:

"The artificial ears ... support narrowband, wideband, super-wideband, as well as full-band applications."

"The artificial mouth … supports narrowband, wideband and super-wideband applications. However, it should be noted that the directionality of the artificial mouth is limited in its ability to simulate the human mouth in the super-wideband frequency range."

For terminals supporting SWB or FB a HATS (Head And Torso Simulator) should be used. For terminals supporting SWB or FB in combination with Narrowband/Wideband functions a HATS (Head And Torso Simulator) shall be used for parameters defined for limited bandwidth such as RLR and SLR.

For send path the HATS shall be used for super-wideband. Until the development of new systems with larger bandwidth, send path measurements will be limited to super-wideband.

NOTE 1: Some HATS may provide a higher bandwidth. If a lab wants to apply the HATS for fullband testing, the lab should check if the HATS used for the tests has been developed and calibrated over the full bandwidth.

NOTE 2: The applicability for FBHR, FBLFE and UBHR is for further study.

6.10.3.1 Positioning handset or headset

When using a handset telephone the handset is placed in the HATS position as described in Recommendation ITU-T P.64 [36]. The artificial mouth shall be in conformance with Recommendation ITU-T P.58 [35]. The artificial ear shall be in conformance with Recommendation ITU-T P.57 [34], either type 3.3 or type 3.4 ears shall be used. Unless stated otherwise, the handset is mounted in the standard position of the HATS. In case of testing a flat handset with artificial ear of:

- Type 3.4, the *flat handset position* according to annex D.3 of Recommendation ITU-T P.64 [36] shall be used (A=0°, B=5° and C=0°).
- Type 3.3, the *alternative handset position* according to annex E.2 of Recommendation ITU-T P.64 [36] shall be used with the definition A=0°, B=5° and C=0°. This aligns measurements using artificial ears of type 3.3 and 3.4, where the flat handset position is explicitly specified (annex D.3 of Recommendation ITU-T P.64 [36]).

Recommendations for positioning headsets are given in Recommendation ITU-T 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 Recommendation ITU-T 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. Unless stated otherwise the application force of 8 N is used for handset testing. No application force is used for headsets.

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.

The various steps for calibration of the artificial mouth are described in Recommendation ITU-T P.581 [43]. The spectrum of acoustic signal produced by the artificial mouth is equalized under freefield conditions at the MRP. Unless specified otherwise, the test signal level shall be -4,7 dBPa at the MRP.

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

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

The exact calibration and equalization can be found in Recommendation ITU-T P.581 [43].For super-wideband / fullband measurements equalization has to be limited between 50 Hz and 16 kHz for calibration of mouth. With some measurement equipment the use of such of bandwidth is not possible and shall be limited to 100 Hz to 14 kHz.

If not stated otherwise, the HATS shall be diffuse-field equalized. The inverse nominal diffuse field curve as found in table 3 of Recommendation ITU-T P.58 [35] shall be used.

NOTE: The inverse average diffuse field response characteristics of HATS as found in Recommendation ITU-T P.58 [35] is used and not the specific one corresponding to the HATS used. Instead of using the individual diffuse field correction, the average correction function is used because, for handset and headset measurements, mostly the artificial ear, ear canal and ear impedance simulations are effective. The individual diffuse-field correction function of HATS includes all diffraction and reflection effects of the complete individual HATS which are not effective in the measurement and potentially would lead to bigger measurement uncertainties than using the average correction.

Unless specified otherwise, the applied test signal level at the digital input shall be -16 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

6.10.4 Set-up for hands-free measurements

6.10.4.0 Super-wideband and fullband applications

Recommendation ITU-T P.58 [35] indicates:

"The artificial ears ... support narrowband, wideband, super-wideband, as well as full-band applications."

"The artificial mouth … supports narrowband, wideband and super-wideband applications. However, it should be noted that the directionality of the artificial mouth is limited in its ability to simulate the human mouth in the super-wideband frequency range."

For terminals supporting SWB or FB a HATS (Head And Torso Simulator) should be used. For terminals supporting SWB or FB in combination with Narrowband/Wideband functions a HATS (Head And Torso Simulator) shall be used for parameters defined for limited bandwidth such as RLR and SLR.

For send path the HATS shall be used for super-wideband. Until the development of new systems with larger bandwidth, send path measurements will be limited to super-wideband.

NOTE 1: Some HATS may provide a higher bandwidth. If a lab wants to apply the HATS for fullband testing, the lab should check if the HATS used for the tests has been developed and calibrated over the full bandwidth.

For handsfree and conferencing terminals an alternative to HATS is the use of a combination including a free field microphone (for receive measurements) and a loudspeaker (for send measurements). The frequency response of these equipment should cover the bandwidth of the terminal under test (at least from 50 Hz to 14 kHz for SWB and from 20 Hz to 20 kHz for FB). The characteristics of the free-field microphone and the loudspeaker shall be recorded in the test report.

The "lip ring" as defined for the artificial mouth of HATS will be defined as the centre of the front face of the loudspeaker and the acoustic centre of the free field microphone.

NOTE 2: The "centre" of the loudspeaker and the "equivalent lip ring" should be defined in more detail.

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

NOTE 3: The applicability for FBHR, FB_LFE and UBHR is for further study.

6.10.4.1 Positioning handsfree

The ear used for measurement shall be indicated in the test report.

Desktop operated loudspeaker terminal

For HATS test equipment, definition of loudspeaker terminal and setups for loudspeaker terminal can be found in Recommendation ITU-T P.581 [43]. The terminal positioning for HATS is shown in figures 6.6 and 6.7.

When HATS is not used, it is replaced by a free-field microphone for receive measurements and loudspeaker (called "artificial mouth" in figure 6.7a) for send measurements, the arrangement defined in Recommendation ITU-T P.340 [39] applies (see figure 6.7a).

When using a free-field microphone instead of the artificial ears of HATS the centre of the microphone is placed at the point "C" in figure 6.7a.

When using a loudspeaker instead of the artificial mouth of HATS the centre of the front plane is placed at the point "C" in figure 6.7a.

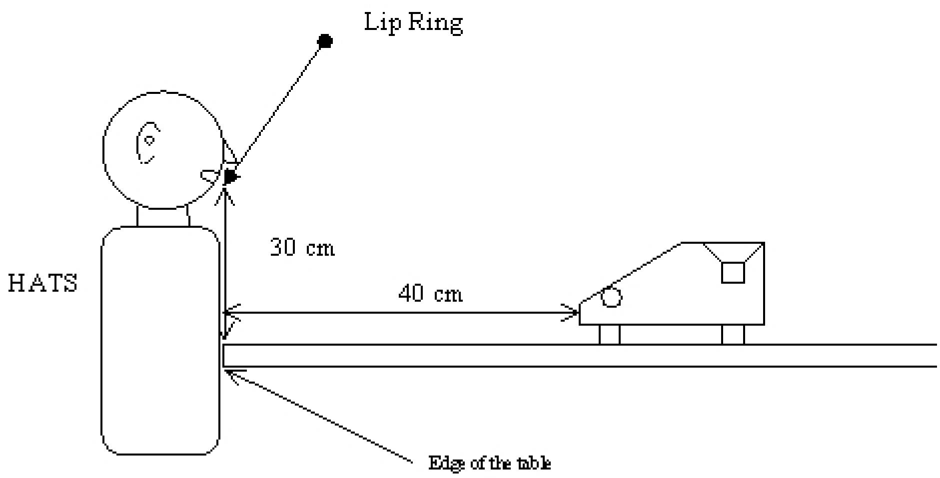


Figure 6.6: Position for test of desktop hands-free terminal side view

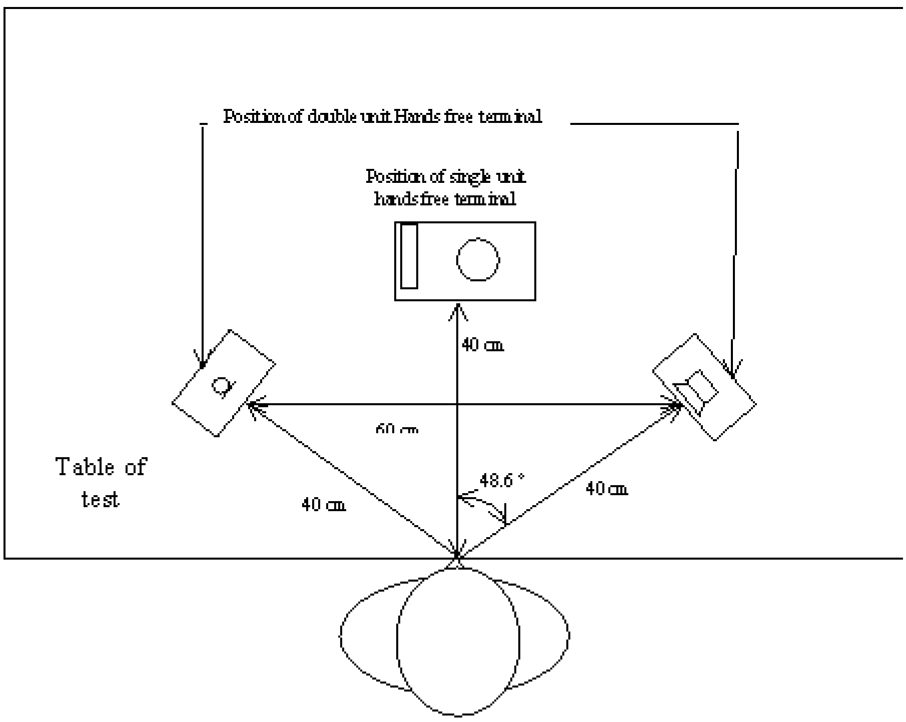


Figure 6.7: Position for test of desktop hands-free terminal top sight

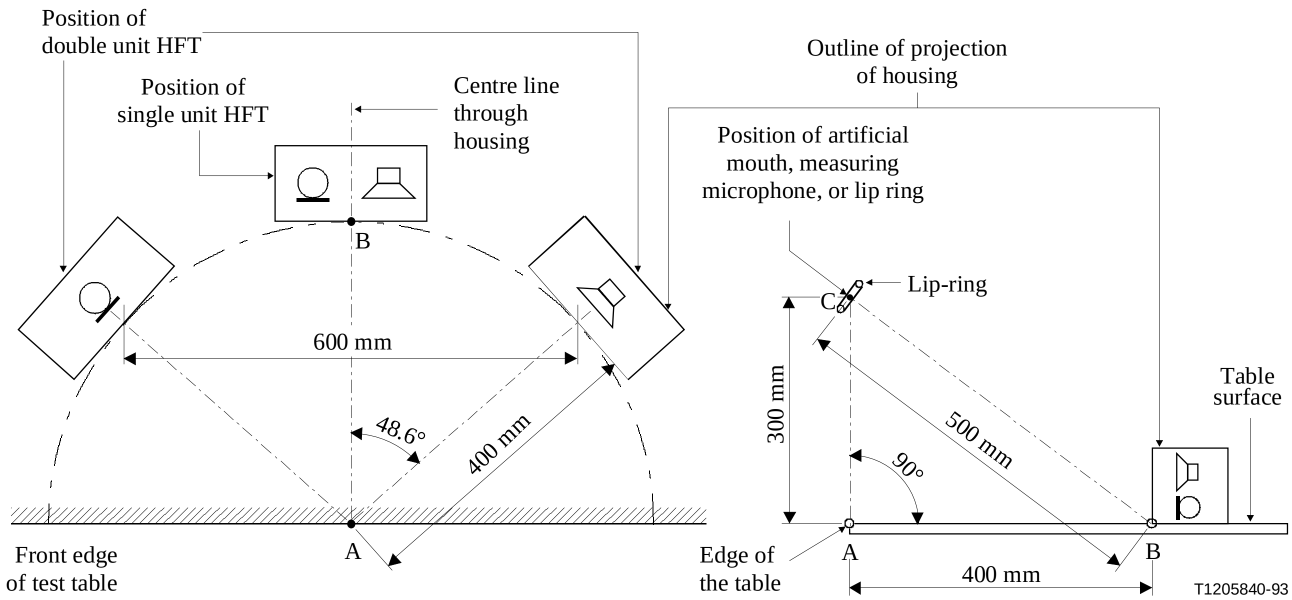


Figure 6.7a: Position for test of desktop handsfree terminal with free-field microphone or with reference loudspeaker (from Recommendation ITU-T P.340 [39]), top and side views

Handheld loudspeaker terminal

HATS measurement equipment shall be configured to the Handheld hands-free UE 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. In case it is not specified the distance d_{HF} shall be 42cm and θ_{HF} shall be 0°.

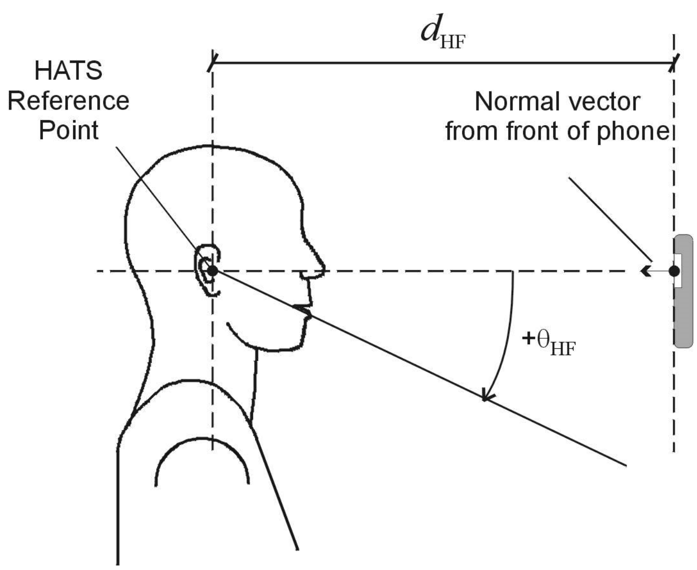


Figure 6.8: Configuration of Hand-Held loudspeaker relative to the HATS side view

NOTE 1: The nominal distance of 42 cm corresponds to lip plane-HATS reference point distance (12 cm) with an additional 30 cm giving a realistic figure as a reference usage of handheld terminals.

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).

For Type 1 the configurations (side and top views) are defined in figures 6.9 and 6.10 when using HATS. When using a free-field microphone instead of the artificial ears of HATS the centre of the microphone is placed at the point "lip ring" in figure 6.9.

When using a loudspeaker instead of the artificial mouth of HATS the centre of the front plane is placed at the point "lip ring" in figure 6.9.

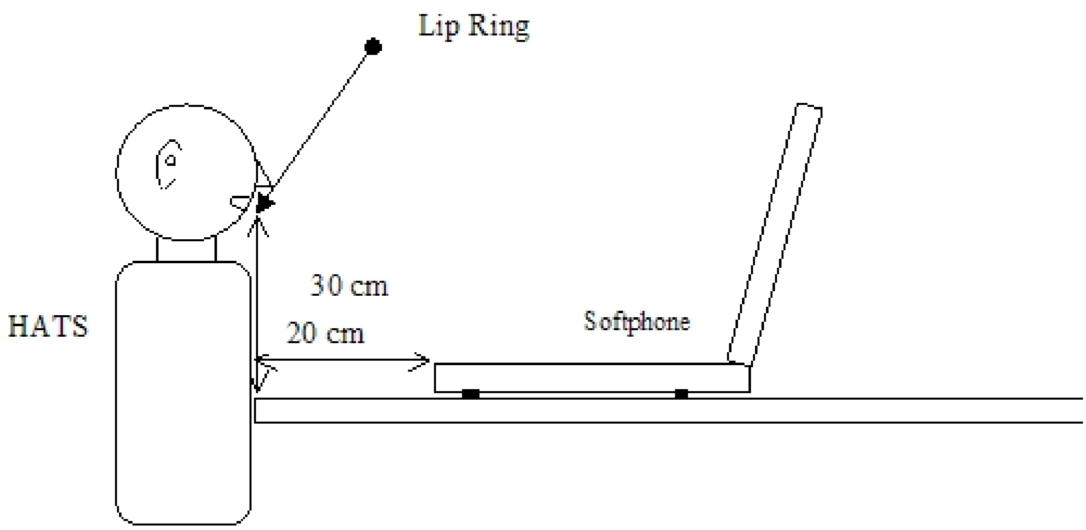


Figure 6.9: Configuration of softphone relative to the HATS side view

When a free-field microphone or reference loudspeaker is used instead of HATS, the microphone centre or the centre of the loudspeaker plane are positioned at the point defined as the lip ring position.

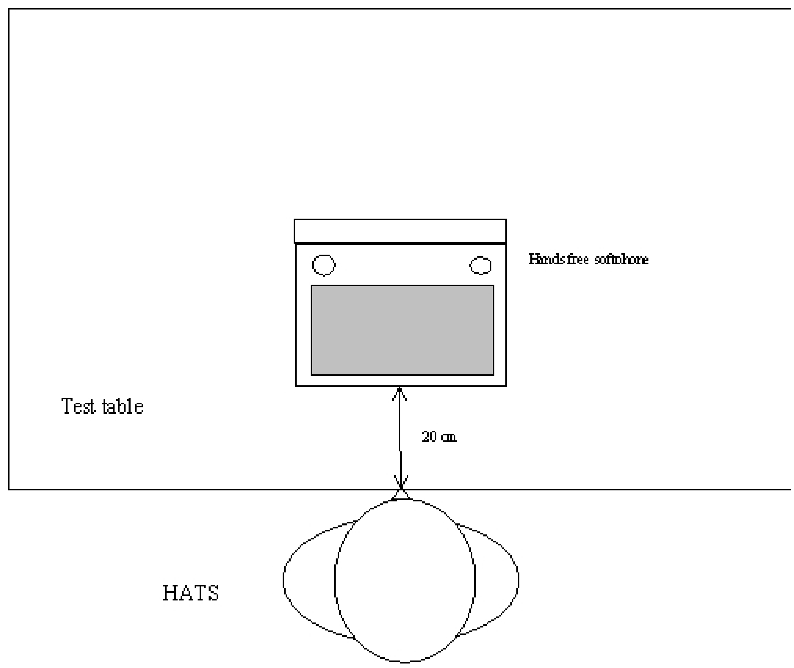


Figure 6.10: Configuration of softphone relative to the HATS top sight

When a free-field microphone or reference loudspeaker is used instead of HATS, the microphone centre or the centre of the loudspeaker plane are positioned at the point defined as the lip ring position.

Softphone with separate speakers

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

When using a free-field microphone instead of the artificial ears of HATS the centre of the microphone is placed at the point "lip ring" in figure 6.11.

When using a loudspeaker instead of the artificial mouth of HATS the centre of the front plane is placed at the point "lip ring" in figure 6.11.

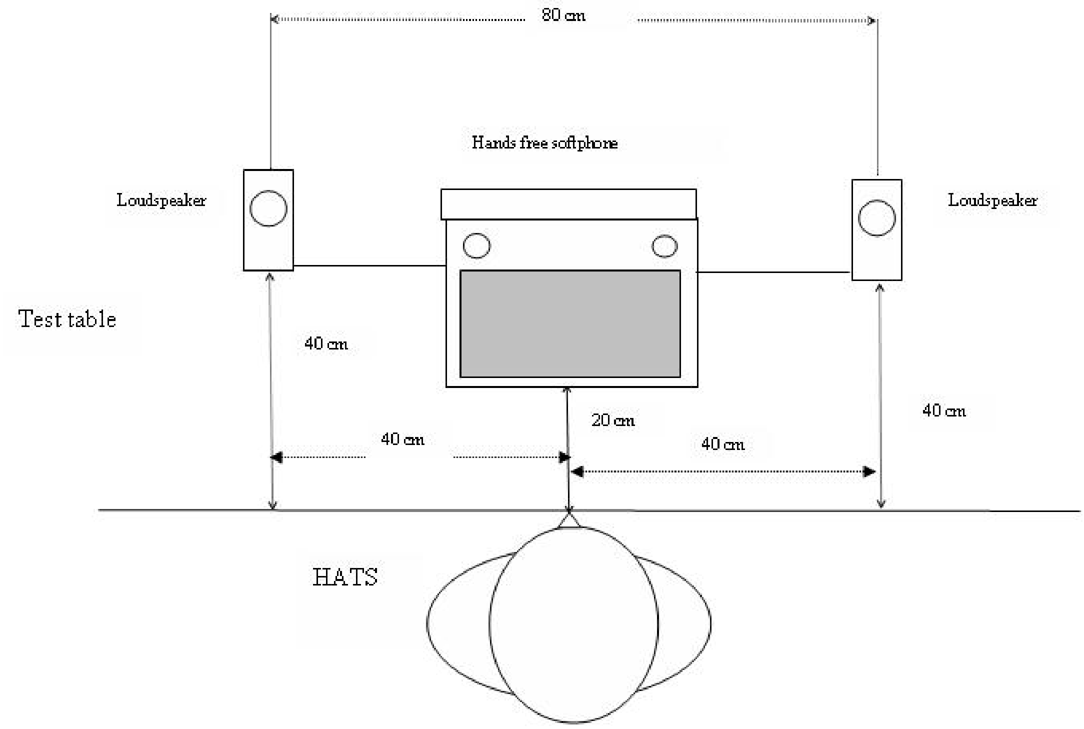


Figure 6.11: Configuration of softphone using external speakers relative to the HATS top sight

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

When using a free-field microphone instead of the artificial ears of HATS the centre of the microphone is placed at the point "lip ring" in figure 6.12.

When using a loudspeaker instead of the artificial mouth of HATS the centre of the front plane is placed at the point "lip ring" in figure 6.12.

NOTE 2: For some specific applications (e.g. sound pick-up, journalist reporting), the terminal may be used with an external microphone (monaural or stereo). The test setup as defined in figure 6.12 applies.

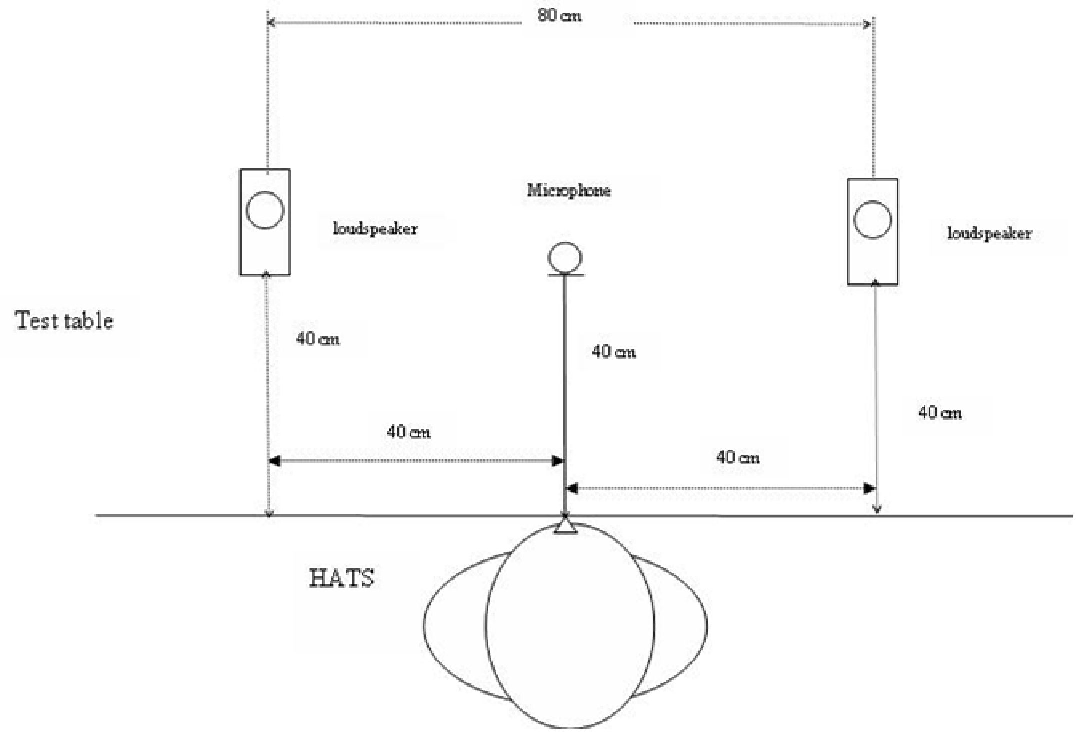


Figure 6.12: Configuration of softphone using external speakers and microphone relative to the HATS top sight

When a free-field microphone or reference loudspeaker is used instead of HATS, the microphone centre or the centre of the loudspeaker plane are positioned at the point defined as the lip ring position.

Group audio 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.

When a super-wideband/fullband Group Audio terminal also implements wideband coders, some parameters may be tested using a HATS test equipment.

Other parameters should be tested using free-field microphone and a reference loudspeaker for super-wideband / fullband measurements.

Figures 6.13 and 6.14 define the test positions to be used when using HATS.

When using a free-field microphone instead of the artificial ears of HATS, the centre of the microphone is placed at the point "lip ring" on figures 6.13 and 6.14.

When using a loudspeaker instead of the artificial mouth of HATS, the centre of the front plane is placed at the point "lip ring" on figures 6.13 and 6.14.

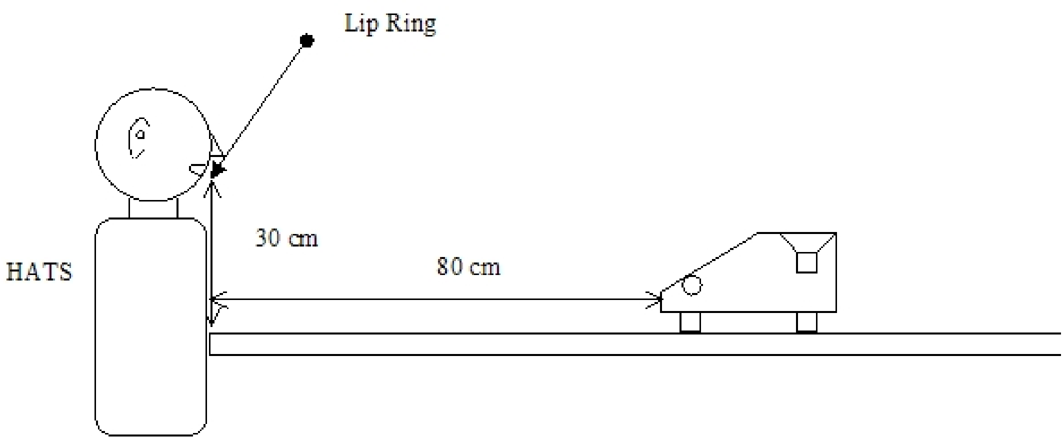


Figure 6.13: Configuration of group terminal relative to the HATS side view

When a free-field microphone or reference loudspeaker is used instead of HATS, the microphone centre or the centre of the loudspeaker plane are positioned at the point defined as the lip ring position.

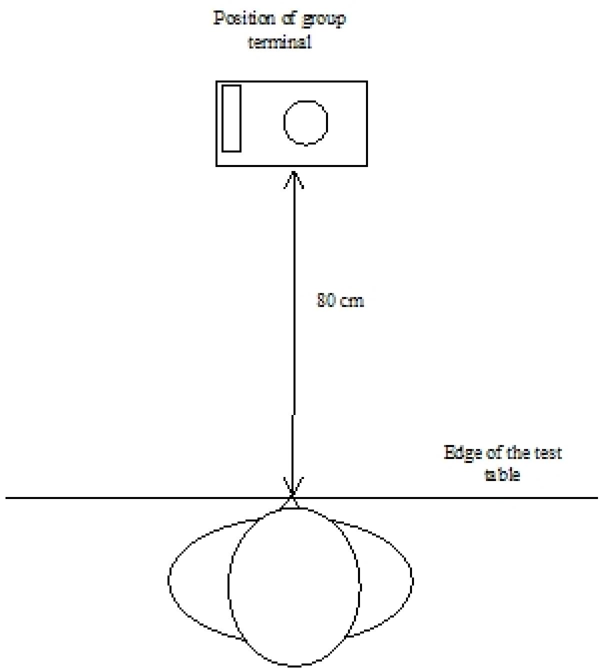


Figure 6.14: Configuration of group terminal relative to the HATS top sight

When a free-field microphone or reference loudspeaker is used instead of HATS, the microphone centre or the centre of the loudspeaker plane are positioned at the point defined as the lip ring position.

NOTE 3: 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.

NOTE 4: Experiences show that it should be ensured that the quality is not substantially affected when the speaker moves in front of the group audio terminal or if he turns his head. Specific arrangements should be defined to check these practical conditions.

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 various steps for calibration of the artificial mouth of the HATS are described in Recommendation ITU-T P.581 [43]. The spectrum of acoustic signal produced by the artificial mouth is equalized under freefield conditions at the MRP.

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 shall be calculated as follows:

$$S_{\mathrm{mJ}} = 20\ \mathrm{log}\mathrm{V}_s - 20\ \mathrm{log}\mathrm{P}_{\mathrm{MRP}} + \mathrm{Corr} - \mathrm{Dcorr}$$

where:

- $\mathbf{V_s}$ is the measured voltage across the appropriate termination (unless stated otherwise, a 600 Ω termination).
- $\mathbf{P_{MRP}}$ is the applied sound pressure at the MRP.
- Corr is 20 log (P_{MRP}/P_{HFRP}) of the used artificial mouth.
- The value of Corr is the value required to calibrate the artificial mouth to the exact value of Dcorr (e.g. 24,0 dB for 50 cm distance).
- Dcorr is the correction to achieve the target sound pressure level at the intended distance (see below).

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.

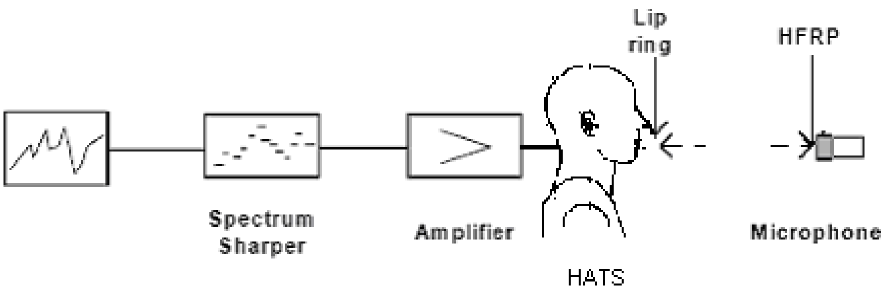


Figure 6.15: Calibration at HFRP (with d_{HFS} = 50 cm)

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

Desktop terminal: 50 cm and level to adjust - 28,7 dBPa, Dcorr = 24 dB.

Handheld terminal: 30 cm with - 24,3 dBPa, Dcorr = 19,6 dB.

Softphone: 36 cm with - 25,8 dBPa, Dcorr = 21,1 dB.

Group terminal: 85 cm with - 33,3 dBPa, Dcorr = 28,6 dB.

6.10.4.2.2 Receiving

Unless specified otherwise, the applied test signal level at the digital input shall be -16 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

All measurement values produced by HATS are intended to be free-field equalized according Recommendation ITU-T P.581 [43].

6.10.5 Set-up for measurements in loudspeaking mode

For those measurements HATS will be used.

The EUT will be positioned as defined in clause 6.10.4.1 (unless 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 the test report. For the handset 8N application force shall be used.

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 ETSI TS 103 224 [59].

Figure 6.16: Void

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

The following noises of ETSI TS 103 224 [59] shall be used per application case:

Table 6.2a: Background noises for handset/headset testing

Name	Description	Length	Handset Levels
Full-size car 130 km/h (FullSizeCar_130)	HATS and microphone array at co-drivers position	30 s	1: 68,5 dB 2: 68,3 dB 3: 68,8 dB 4: 69,5 dB 5: 69,9 dB 6: 70,5 dB 7: 70,8 dB 8: 71,9 dB
Cafeteria (Cafeteria)	HATS and microphone array inside a cafeteria	30 s	1: 70,0 dB 2: 70,0 dB 3: 70,1 dB 4: 70,7 dB 5: 70,5 dB 6: 70,8 dB 7: 70,6 dB 8: 71,0 dB
Roadnoise (Roadnoise)	HATS and microphone array standing outside near a road	30 s	1: 72,8 dB 2: 71,6 dB 3: 72,0 dB 4: 72,9 dB 5: 72,2 dB 6: 73,1 dB 7: 73,0 dB 8: 73,8 dB
Pub Noise (Pub)	HATS and microphone array in a Pub	30 s	1: 77,2 dB 2: 76,6 dB 3: 75,7 dB 4: 76,0 dB 5: 76,0 dB 6: 76,3 dB 7: 76,0 dB 8: 76,4 dB
Airport departure	HATS and microphone array in an airport gate area	30 s	1: 77,5 dB 2: 78,3 dB 3: 78,7 dB 4: 78,7 dB 5: 78,4 dB 6: 78,8 dB 7: 78,1 dB 8: 78,1 dB

Table 6.2b: Background noises for narrowband and wideband loudspeaking/handsfree testing

Name	Description	Length	Handsfree Levels
Full-size car 130 km/h (FullSizeCar_130)	HATS and microphone array at co-drivers position	30 s	1: 69,5 dB 2: 68,6 dB 3: 68,6 dB 4: 68,7 dB 5: 68,8 dB 6: 68,8 dB 7: 69,2 dB 8: 69,7 dB
Cafeteria (Cafeteria)	HATS and microphone array inside a cafeteria	30 s	1: 69,0 dB 2: 69,7 dB 3: 69,6 dB 4: 69,8 dB 5: 69,5 dB 6: 69,5 dB 7: 69,7 dB 8: 70,0 dB
Roadnoise (Roadnoise)	HATS and microphone array standing outside near a road	30 s	1: 69,9 dB 2: 70,7 dB 3: 70,9 dB 4: 71,0 dB 5: 70,8 dB 6: 70,8 dB 7: 70,9 dB 8: 71,0 dB

Name	Description	Length	Handsfree Levels
Pub Noise (Pub)	HATS and microphone array in a Pub	30 s	1: 75,2 dB 2: 75,1 dB 3: 74,9 dB 4: 75,1 dB 5: 74,8 dB 6: 74,8 dB 7: 74,8 dB 8: 75,0 dB
Airport departure	HATS and microphone array in an airport gate area	30 s	1: 77,2 dB 2: 77,4 dB 3: 77,6 dB 4: 77,7 dB 5: 78,1 dB 6: 77,9 dB 7: 77,8 dB 8: 77,9 dB

Table 6.2c: Background noises for super-wideband and fullband loudspeaking/handsfree testing

Name	Description	Length	Handsfree Levels
Pub Noise (Pub)	HATS and microphone array in a pub	30 s	1: 75,2 dB 2: 75,1 dB 3: 74,9 dB 4: 75,1 dB 5: 74,8 dB 6: 74,8 dB 7: 74,8 dB 8: 75,0 dB
Full-size car 130 km/h (FullSizeCar_130)	HATS and microphone array at co-drivers position	30 s	1: 69,5 dB 2: 68,6 dB 3: 68,6 dB 4: 68,7 dB 5: 68,8 dB 6: 68,8 dB 7: 69,2 dB 8: 69,7 dB
Cafeteria (Cafeteria)	HATS and microphone array inside a cafeteria	30 s	1: 69,0 dB 2: 69,7 dB 3: 69,6 dB 4: 69,8 dB 5: 69,5 dB 6: 69,5 dB 7: 69,7 dB 8: 70,0 dB
Roadnoise (*Roadnoise*)	HATS and microphone array standing outside near a road	30 s	1: 69,9 dB 2: 70,7 dB 3: 70,9 dB 4: 71,0 dB 5: 70,8 dB 6: 70,8 dB 7: 70,9 dB 8: 71,0 dB
Recording in airport hallway (*Airport*)	HATS and microphone array in an airport hallway with overhead public address announcement	30 s	1: 77,2 dB 2: 77,4 dB 3: 77,6 dB 4: 77,7 dB 5: 78,1 dB 6: 77,9 dB 7: 77,8 dB 8: 77,9 dB

6.10.7 Setup of variable echo path

The handset is positioned d = 3 cm above a horizontal hard surface, facing the surface with speaker and microphone. The surface shall be at least 35 × 35 cm The handset is fixed like a pendulum with a non-elastic cord 3 cm above the centre of the horizontal surface, see figure 6.17. The pivot is 55 ± 1 cm above the hard plate.

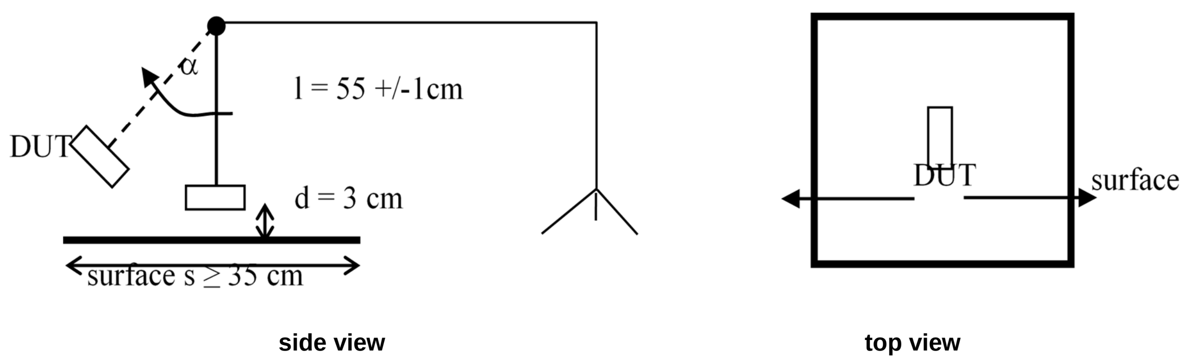


Figure 6.17: Positioning of handset under test

The "handset-pendulum" is displaced at least to the edge of the hard surface. The test signal playback shall start with the release of the displaced handset under test.

For headsets the same measurement arrangement is used as described above. However, it has to be assured that the echo path (audio path between speaker output and microphone input) changes significantly. If the pendular motion across the base surface is not producing a sufficient change in echo path, another hard surface perpendicular to the base surface can be added. The dimension and position of the additional surface should be chosen such that it is positioned within the echo path when crossed by the pendulous headset but not within the echo path when the headset reaches the turning point of the pendulous motion. At the lowest point of pendular motion, the headset speaker and microphone should not exceed a distance of 3 cm from either of the surfaces.

NOTE 1: Depending on the geometry of the headset (monaural / binaural, microphone integrated into earpiece/earplug with microphone on short arm / microphone on long arm) a stable pendular motion has to be established. This may require two cords fixed with respect to the headset's balance point in order to avoid tumbling motion. Alternatively, the headset may be attached to a fixed radial arm to achieve a stable pendular motion.

Figure 6.18 shows an exemplary setup for a binaural headset with long microphone arm and vertical surface to increase echo path variation by changing the coupling between speaker and microphone during pendular motion. During one pendular period, the DUT is exposed to four sudden changes in echo path when passing the vertical surface.

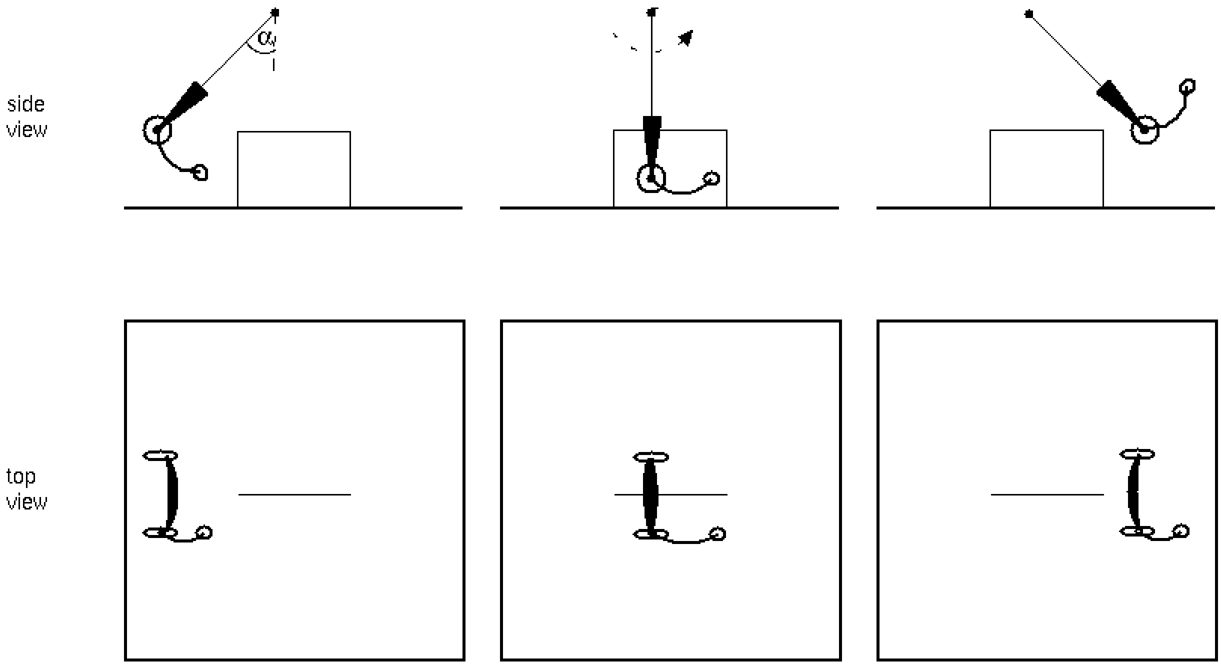


Figure 6.18: Example for positioning of a headset under test

Test setup for desktop hands-free terminals: A notebook is positioned at least 20 cm in front of the device (or devices) with the transducers, as shown in figure 6.19. The notebook lid is moved during the measurement.

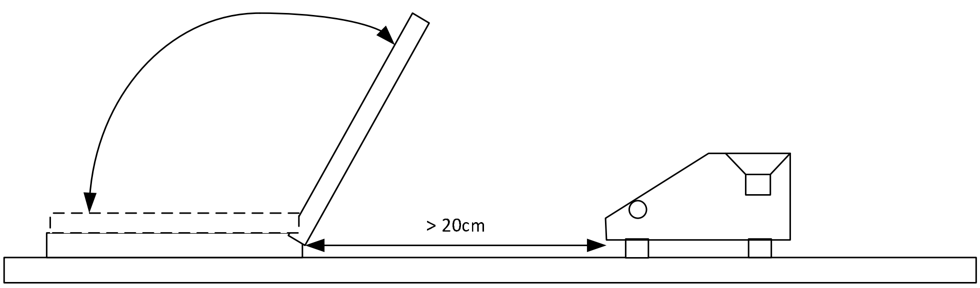


Figure 6.19: Positioning of DUT

Test setup for softphone: The test setup is described in clause 6.10. The notebook lid is moved during the measurement, as shown in figure 6.20. This setup is valid for all combinations of notebook with or without external speakers or microphone.

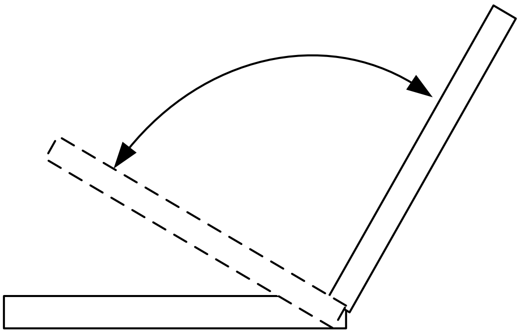


Figure 6.20: Positioning of DUT

Test setup for other handsfree devices is for further study.

NOTE 2: Care should be taken to not generate noise during the movement of the notebook lid. Because of this, this measurement is not applicable for a softphone without external microphone.

6.10.8 Setup for testing positional robustness of handsets

In order to investigate the robustness of certain measurements against non-default positions as described in clause 6.10.3, three modified positions are defined for the sending and receiving side. Tables 6.3 and 6.4 provide a description of these positions, which are derived from typical user behaviour. Figure 6.21 illustrates the different axes and coordinate system. More detailed explanations are provided in Recommendation ITU-T P.64 [36]. All measurements regarding positioning are only applicable for handset testing.

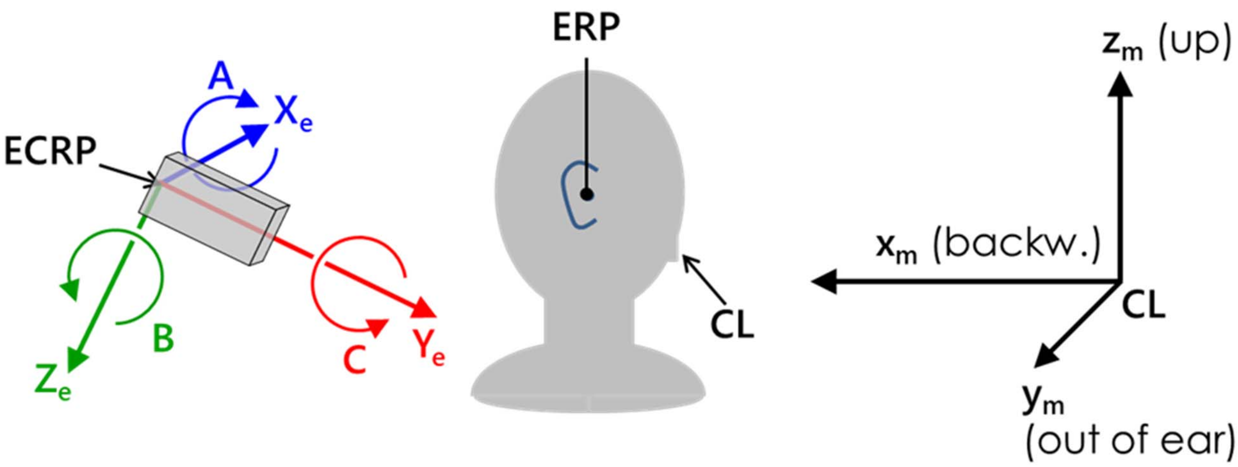


Figure 6.21: Schematic overview over positioning coordinate system

Table 6.3 provides the different angles for the positions in sending direction. With these shifts, distance and direction between MRP and microphone input of the DUT is varied.

Table 6.3: Modified test positions for sending direction

Position name	A [°] (rotation along X_e)	ΔB [°] (rotation along Z_e)	C [°] (rotation along Y_e)	Comment
STD	0	0	0	Standard position at ECRP
UP	-14	+5	0	Terminal elevated
DOWN	+30	0	0	Terminal lowered
AWAY	0	+18	0	Larger distance to MRP

NOTE: The standard position at ECRP is given by A = B = C = 0°. As specified in clause 6.10.3, the positioning angle for "flat handsets" is set to B = 5°. Thus, only the difference to the angle of B is provided here, i.e. angles for A and C are absolute values.

Table 6.4 provides the different angles for the positions in receiving direction. With these shifts, the position of the loudspeaker relative to the ECRP is varied.

Table 6.4: Modified test positions for receiving direction

Position name	Y_e [mm]	Z_e [mm]	Comment
STD	0	0	Standard position at ECRP
Ye_{-5} Ze_{-5}	-5	-5	Above ECRP
Ye_{0} Ze_{+5}	0	+5	Right-below ECRP
Ye_{+5} Ze_{-5}	+5	-5	Right to ECRP

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/headsets (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 Recommendation ITU-T P.57 [34]) or the HATS methodology (see Recommendation ITU-T P.58 [35]).

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

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 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	Recommendation ITU-T P.311 [38] tested, wideband handset or headset	7.2.9	
	2b	HATS tested, "standard" wideband handset or headset	7.2.10	

Applicable to:	Type nr.	Type name	Clause	Remarks
	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 ETSI EN 300 175-8 [8], annex B
	7a	Fullband handset or headset	7.2.17	
	7b	Fullband handsfree	7.2.18	
	7c,7d,7e,7f	Fullband stereo device	7.2.19	
	7g	FBHR (24 kHz) headset device	7.2.20	
	7h	FBHR (24 kHz) loudspeaking device	7.2.21	
	7i	FBLFE device	7.2.22	
	7j	Fullband low-latency microphone	7.2.23	
	8a	Ultra-band headset device	7.2.24	
	8b	Ultra-band loudspeaking device	7.2.25	
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, super-wideband, fullband, FBHR or ultra-band 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 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.

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

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

Types 4a and 4b (respectively standard and improved) concerning loudspeaking and handsfree function for 7 kHz (wideband) telephony service.

Types 5a and 5b concerning respectively handsets or headsets (5a) and loudspeaking handsfree function (5b) for 14 kHz super-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.

Types 7a and 7b concerning respectively handset or headset (7a) and loudspeaking handsfree function (7b) for 20 kHz fullband audio service.

Types 7c, 7d, 7e and 7f concerning respectively a device for 20 kHz fullband stereo audio service at:

- 7c: 128 kbit/s gross bit rate for 10 ms frame size
- 7d: 192 kbit/s gross bit rate for 10 ms frame size
- 7e: 256 kbit/s gross bit rate for 10 ms frame size
- 7f: 256 kbit/s for 5 ms frame size, 320 kbit/s for 2,5 ms frame size and at 512 kbit/s for 2,5 ms frame size

Types 7g and 7h concerning respectively headset (7g) and loudspeaking function (7h) for 24 kHz FBHR service.

Type 7i introduces a FBLFE (250 Hz) loudspeaking device.

Type 7j introduces a fullband (20 kHz) low-latency microphone device.

Types 8a and 8b concerning headset (8a) and loudspeaking function (8b) for 48 kHz ultra-band service.

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

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 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, super-wideband, fullband, FBHR or ultra-band 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 three- 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 of a top-level description and a 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 modifications regarding requirements or measurement methods compared with the previous test 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: However, there can be some performance restrictions in some combinations.

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		

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part					
	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					
	7.2.4.3	General specification	M		
	7.5.1	All specs of type 1a also apply	M/O		See type 1a
	7.5.2.1	Terminal coupling loss	M		
	7.5.2.2	Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$	M		
	7.5.2.3	Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$	M		
	7.5.2.4	Activation in Sending Direction	M		
	7.5.2.5	Activation in Receiving Direction	M		
PP types 1c and 1d: HATS-tested narrowband handset					
	7.2.5.3/7.2.6.3	General specification	M	X	
	7.5.3.1.1.1/ 7.5.3.1.2.1	Sending and receiving frequency responses - nominal position	M	X	
X	7.5.3.1.1.2/ 7.5.3.1.2.2	Frequency responses - positional robustness	M		
X	7.5.3.2.1.1/ 7.5.3.2.1.2	Sending and receiving loudness ratings - nominal position	M		
X	7.5.3.2.5.1/ 7.5.3.2.5.2	Sending and receiving loudness ratings - positional robustness	M		
X	7.5.3.2.4	Microphone Mute	M		
X	7.5.3.2.6	Send Loudness Level	M		
X	7.5.3.2.7	Receive Loudness Level	M		
	7.5.3.3.1	Sidetone masking rating (STMR)	M		
	7.5.3.3.3	Sidetone delay	M		
X	7.5.3.4.1	TCLw	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		
X	7.5.3.11	Double Talk Performance	O		Strongly recommended for improved class
X	7.5.3.12	Switching characteristics	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					
	7.2.7.3/7.2.8.3	General specification	M	X	
	7.5.4.1	Sending sensitivity/frequency response	M		
	7.5.4.2	Receive sensitivity/frequency response	M		
X	7.5.4.3	Send loudness rating	M		

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part					
X	7.5.4.4	Receive loudness rating	M	X	
	7.5.4.5	Sending distortion	M		
	7.5.4.6	Receiving distortion	M		
	7.5.4.7	Out-of-band signals in sending direction	M		
	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 weighted	M	X	
	7.5.4.12	Stability Loss	M		
	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	7.5.4.13.2	Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$	M		
	7.5.4.13.3	Detection of Echo Components during Double Talk	O		
	7.5.4.13.4	Minimum activation level and sensitivity of double talk detection	O		
	7.5.4.14	**Switching characteristics**			
	7.5.4.14.1	Activation in Sending Direction	M		
	7.5.4.14.2	Activation in Receiving Direction	O		
	7.5.4.14.3	Silence Suppression and Comfort Noise Generation	O		
	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		
	7.5.4.15	**Quality of echo cancellation**			
	7.5.4.15.1	Temporal echo effects	O		
	7.5.4.15.2	Spectral Echo Attenuation	O		
X	7.5.4.15.3	Variable echo path	M		
	7.5.4.16	Microphone mute	M		
X	7.5.4.17	Delay	M		
X	7.5.4.18	Send Loudness Level	M		
X	7.5.4.19	Receive Loudness Level	M	X	
PP type 2a: Recommendation ITU-T 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		
	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		
	7.5.5.3.2	Sidetone distortion	M		
X	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					
	7.2.10.3/ 7.2.11.3	General specification	M	X	
	7.5.6.1.1.1/ 7.5.6.1.2.1	Sending and receiving frequency responses - nominal position	M		

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part					
X	7.5.6.1.1.2/ 7.5.6.1.2.2	Sending and receiving frequency responses - positional robustness	M		
X	7.5.6.2.1	Send and receive loudness ratings - nominal values	M		
	7.5.6.2.4	Microphone mute	M		
X	7.5.6.2.5	Send and receive loudness ratings - positional robustness	M		
X	7.5.6.2.6	Send Loudness Level	M		
X	7.5.6.2.7	Receive Loudness Level	M		
	7.5.6.3.1	Sidetone masking rating (STMR)	M		
	7.5.6.3.3	Sidetone delay	M		
X	7.5.6.4.1	Terminal Coupling Loss (TCL)	M		
X	7.5.6.4.2	Stability loss	M		
X	7.5.6.5	Distortion	M		
X	7.5.6.6	Noise	M		
	7.5.6.7	Acoustic shock	M		
	7.5.6.8	Delay	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
X	7.5.6.13	Out-of-band signals	M		
PP type 4a and 4b: wideband loudspeaking and handsfree device					
	7.2.12.3/ 7.2.13.3	General specification	M	X	
	7.5.7.1	Sending sensitivity/frequency response	M		
	7.5.7.2	Receive sensitivity/frequency response	M		
X	7.5.7.3	Send loudness rating	M		
X	7.5.7.4	Receive loudness rating	M	X	
X	7.5.7.5	Sending distortion	M		
	7.5.7.6	Receiving distortion	M	X	
	7.5.7.7	Out-of-band signals in sending direction	M		
	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	M	X	
	7.5.7.12	Stability Loss	M		
	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 $A_{H,R,dt}$	M	X	
	7.5.7.13.3	Detection of Echo Components during Double Talk	O	X	
	7.5.7.13.4	Minimum activation level and sensitivity of double talk detection	O		
	7.5.7.14	**Switching characteristics**			
	7.5.7.14.1	Activation in Sending Direction	M		
	7.5.7.14.2	Activation in Receiving Direction	O		
	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		

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part					
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		
	7.5.7.15	**Quality of echo cancellation**			
	7.5.7.15.1	Temporal echo effects	O		
	7.5.7.15.2	Spectral Echo Attenuation	O		
X	7.5.7.15.3	Variable echo path	M		
X	7.5.7.16	Microphone mute	M		
	7.5.7.17	Delay	M		
X	7.5.7.18	Send Loudness Level	M		
X	7.5.7.19	Receive Loudness Level	M	X	
PP type 5a: super-wideband 14 kHz handset or headset					
	7.2.14.3	General specification	M		
	7.5.8.1.1.1/ 7.5.8.1.2.1	Send and receive frequency responses - nominal position	M		
X	7.5.8.1.1.2/ 7.5.8.1.2.2	Send and receive frequency responses - positional robustness	M		
	7.5.8.2.1.1/ 7.5.8.2.2.1	Send and receive loudness rating - nominal values	M		
X	7.5.8.2.1.2/ 7.5.8.2.2.2	Send and receive loudness rating - positional robustness	M		
	7.5.8.2.1.3	Microphone mute	M		
X	7.5.8.2.3	Send Loudness Level	M		
X	7.5.8.2.4	Receive Loudness Level	M		
X	7.5.8.3.1	Sidetone Masking Rating (STMR)	M		
	7.5.8.3.2	Sidetone delay	M		
	7.5.8.4.1	Unweighted Terminal Coupling Loss	M		
	7.5.8.4.2	Stability loss	M		
	7.5.8.5	Distortion	M		
X	7.5.8.6	Noise	M		
	7.5.8.7	Acoustic shock	M		
	7.5.8.8	Delay	M		
X	7.5.8.9	Double Talk Performance	M		
X	7.5.8.10	Switching characteristics	M		
	7.5.8.11	Quality of echo cancellation	M		
PP type 5b: super-wideband 14 kHz handsfree device					
	7.2.15.3	General specification	M		
X	7.5.9.1	Sending sensitivity/frequency response	M		
X	7.5.9.2	Receive sensitivity/frequency response	M		
X	7.5.9.3.1	Sending loudness rating - nominal value	M		
X	7.5.9.3.2	Microphone mute	M		
X	7.5.9.4	Receive loudness rating	M		
	7.5.9.5	Sending distortion	M		
	7.5.9.6	Receiving distortion	M		
X	7.5.9.7	Sending noise	M		
X	7.5.9.8	Receiving noise	M		
X	7.5.9.9.1	Terminal Coupling Loss	M		
	7.5.9.9.2	Stability Loss	M		
	7.5.9.10	**Double Talk Performance**			
	7.5.9.10.1	Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$	M		
X	7.5.9.10.2	Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$	M		
	7.5.9.10.3	Detection of Echo Components during Double Talk	M		

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part					
	7.5.9.10.4	Minimum activation level and sensitivity of double talk detection	M		
	7.5.9.11	**Switching characteristics**			
	7.5.9.11.1	Activation in Sending Direction	M		
	7.5.9.11.2	Silence Suppression and Comfort Noise Generation	O		
	7.5.9.11.3	Performance in sending direction in the presence of background noise	M		
X	7.5.9.11.4	Speech Quality in the Presence of Background Noise	M		
X	7.5.9.11.5	Quality of Background Noise Transmission (with Far End Speech)	M		
	7.5.9.12	**Quality of echo cancellation**			
	7.5.9.12.1	Temporal echo effects	M		
	7.5.9.12.2	Spectral Echo Attenuation	M		
X	7.5.9.12.3	Variable echo path	M		
	7.5.9.13	Delay	M		
X	7.5.9.14	Send Loudness Level	M		
X	7.5.9.15	Receive Loudness Level	M		
PP type 6: PPs with external 2 wire, 3,1 kHz telephony interface					
	ETSI EN 300 175-8 [8], annex B	2-wire PP end system (informative)	M		Detailed specification informative only
PP type 7a: fullband 20 kHz handset or headset					
	7.2.17.3	General specification	M		
	7.5.10.1.1.1/ 7.5.10.1.2.1	Send and receive frequency responses - nominal position	M		
X	7.5.10.1.1.2/ 7.5.10.1.2.2	Send and receive frequency responses - positional robustness	M		
	7.5.10.2.1.1/ 7.5.10.2.2.1	Send and receive loudness rating - nominal position	M		
X	7.5.10.2.1.2/ 7.5.10.2.2.2	Send and receive loudness rating - positional robustness	M		
	7.5.10.2.1.3	Microphone mute	M		
X	7.5.10.2.3	Send Loudness Level	M		
X	7.5.10.2.4	Receive Loudness Level	M		
X	7.5.10.3.1	Sidetone Masking Rating	M		
	7.5.10.3.2	Sidetone delay	M		
	7.5.10.4.1	Unweighted Terminal Coupling Loss	M		
	7.5.10.4.2	Stability loss	M		
	7.5.10.5	Distortion	M		
X	7.5.10.6	Noise	M		
	7.5.10.7	Acoustic shock	M		
	7.5.10.8	Delay	M		
X	7.5.10.9	Double Talk Performance	M		
X	7.5.10.10	Switching characteristics	M		
	7.5.10.11	Quality of echo cancellation	M		
PP type 7b: fullband 20 kHz loudspeaking and handsfree device					
	7.2.18.3	General specification	M		
X	7.5.11.1	Sending sensitivity/frequency response	M		
X	7.5.11.2	Receive sensitivity/frequency response	M		
X	7.5.11.3	Sending loudness rating	M		
X	7.5.11.4	Receive loudness rating	M		
	7.5.11.5	Sending distortion	M		
	7.5.11.6	Receiving distortion	M		
X	7.5.11.7	Sending noise	M		
X	7.5.11.8	Receiving noise	M		
X	7.5.11.9	Terminal Coupling Loss	M		

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part					
	7.5.11.10	**Double Talk Performance**			
	7.5.11.10.1	Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$	M		
X	7.5.11.10.2	Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$	M		
	7.5.11.10.3	Detection of Echo Components during Double Talk	M		
	7.5.11.10.4	Minimum activation level and sensitivity of double talk detection	M		
	7.5.11.11	**Switching characteristics**			
	7.5.11.11.1	Activation in Sending Direction	M		
	7.5.11.11.2	Silence Suppression and Comfort Noise Generation	O		
X	7.5.11.11.3	Performance in sending direction in the presence of background noise	M		
X	7.5.11.11.4	Speech Quality in the Presence of Background Noise	M		
X	7.5.11.11.5	Quality of Background Noise Transmission (with Far End Speech)	M		
	7.5.11.12	**Quality of echo cancellation**			
	7.5.11.12.1	Temporal echo effects	M		
	7.5.11.12.2	Spectral Echo Attenuation	M		
X	7.5.11.12.3	Variable echo path	M		
	7.5.11.13	Delay	M		
X	7.5.11.14	Send Loudness Level	M		
X	7.5.11.15	Receive Loudness Level	M		
PP types 7c, d, e, f: fullband 20 kHz stereo audio device					
	7.2.19.3	General specification	M		
PP type 7g: FBHR 24 kHz headset device					
X	7.2.20.3	General specification	M		
PP type 7h: FBHR 24 kHz loudspeaking device					
X	7.2.21.3	General specification	M		
PP type 7i: FBLFE 250 Hz loudspeaking device					
X	7.2.22.3	General specification	M		
PP type 7j: fullband 20 kHz low-latency microphone device					
X	7.2.23.3	General specification	M		
PP type 8a: ultra-band 48 kHz headset device					
X	7.2.24.3	General specification	M		
PP type 8b: ultra-band 48 kHz loudspeaking device					
X	7.2.25.3	General specification	M		

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

CH	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		
	7.3.2.3.3	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		

CH	Clause Number	Requirement	M/O	S/I	Comments (see also table E.1)
Fixed Part					
FP type 1b: "new" Fixed Part with ISDN interface, narrowband service					
	7.3.3.3.1	Transcoding and equalization	M		
	7.3.3.3.2	PP type detection	O		
	7.3.3.3.3	Activation of audio processing functions	O		
	7.6.2.1	FP Network echo control	O		
	7.4.2	Echo canceller for PP	O		
	7.4.3	Echo suppressor for PP	O		
	7.6.2.2	FP adaptive volume control	O		
	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					
	7.3.5.3.1	Transcoding and equalization	M		
	7.3.5.3.2	PP type detection	O		
	7.3.5.3.3	Activation of audio processing functions	O		
	7.6.4.3	Adaptive volume control	O		
	7.6.4.4	Clock accuracy	M		
	7.6.4.5	Send jitter	M		
X	7.6.4.6	Send and receive delay - roundtrip delay	M		
	7.4.2	Echo canceller for PP	O		
	7.4.3	Echo suppressor for PP	O		
FP type 4: Fixed Part with ISDN interface, wideband service					
	7.3.6.3.1	Transcoding and equalization	M		
	7.3.6.3.2	PP type detection	O		
	7.3.6.3.3	Activation of audio processing functions	O		
	7.4.2	Echo canceller for PP	O		
	7.4.3	Echo suppressor for PP	O		
	7.6.5.1	FP adaptive volume control	O		
	7.6.5.2	FP Delay	M		
FP type 5: Fixed Part with VoIP interface, wideband, super-wideband, fullband, FBHR or ultra-band service					
	7.3.7.3.1	Transcoding and equalization	M		
	7.3.7.3.2	PP type detection	O		
	7.3.7.3.3	Activation of audio processing functions	O		
	7.6.6.3	FP adaptive volume control	O		
	7.6.6.4	Clock accuracy	M		
	7.6.6.5	Send jitter	M		
X	7.6.6.6	Send and receive delay - roundtrip delay	M		
	7.4.2	Echo canceller for PP	O		
	7.4.3	Echo suppressor for PP	O		
FP type 6a: Internal call inside a DECT FP (any service)					
	7.3.8.3	Specification (transparent)	M		
FP type 6b: n-party conference inside a DECT FP (any service)					
	7.3.9.3	Specification (informative)	M		
FP type 7: DECT Repeater part (REP)					
	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 (TCL(w)). 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 codec. When there is no specific mention the provided figure should be understood as applicable for codecs G.726 [24] (narrowband), G.722 [21] (wideband) and LC3plus [58] (narrowband, wideband, super-wideband, fullband, FBHR, FBLFE and ultra-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

7.2.0 General

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 the following audio services: 3,1 kHz narrowband telephony, 7 kHz wideband telephony, 14 kHz super-wideband audio, 20 kHz fullband audio, 24 kHz FBHR, 250 Hz FBLFE and ultra-band signals, and two specification methodologies: artificial ear, according to Recommendation ITU-T P.57 [34], used in types 1a, 1b and 2a and HATS according to Recommendation ITU-T P.58 [35], used in types 1c, 1d, 2b, 2c, 3a, 3b, 4a, 4b, 5a, 5b, 7a, 7b, 7c, 7d, 7e and 7f. Any new development is strongly recommended to be based on type definitions 1d for narrowband handset/headset, 2c for wideband handset/headset, 3b for narrowband loudspeaking/handsfree and 4b for wideband loudspeaking/handsfree.

7.2.1 Performance levels of DECT Portable Parts (handsets)

ETSI standards for VoIP terminals (ETSI ES 202 737 [i.8], ETSI ES 202 738 [i.9], ETSI ES 202 739 [i.10] and ETSI 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 ETSI ES 202 737 [i.8] and ETSI ES 202 738 [i.9] standard requirements for VoIP terminals, corresponding to devices with enhanced capabilities, with performances tested with HATS methodology. Any new development is strongly recommended to be based on PP type 1d; PP types 1a, 1b and 1c are no longer recommended.

For wideband (7 kHz service):

- PP Type 2a: P.311-tested wideband handset introducing wideband with performances tested according Recommendation ITU-T 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 ETSI ES 202 739 [i.10] and ETSI ES 202 740 [i.11] standard requirements for VoIP terminals, corresponding to devices with enhanced capabilities, with performances tested with HATS methodology. Any new development is strongly recommended to be based on PP type 2c; PP types 2a and 2b are no longer recommended.

For super-wideband (14 kHz service):

- PP Type 5a: ETSI TS 102 924 [60] super-wideband handset or headset introducing super-wideband with performances tested according to ETSI TS 102 924 [60].
- PP Type 5b: ETSI TS 102 925 [61] super-wideband headset introducing super-wideband loudspeaking handsfree function with performances tested according to ETSI TS 102 925 [61].

For fullband (20 kHz service):

- PP Type 7a: ETSI TS 102 924 [60] fullband handset or headset introducing fullband with performances tested according to ETSI TS 102 924 [60].
- PP Type 7b: ETSI TS 102 925 [61] fullband handset or headset introducing fullband loudspeaking handsfree function with performances tested according to ETSI TS 102 925 [61].
- PP Type 7c, d, e, f: ETSI TS 102 924 [60] fullband device introducing fullband stereo audio streaming function with performances tested according to ETSI TS 102 924 [60] / ETSI TS 102 925 [61].
- PP Type 7j: Low-latency microphone device.

For FBHR (24 kHz service):

- PP Type 7g: FBHR handset or headset.
- PP Type 7h: FBHR loudspeaking device.

For FBLFE (250 Hz/LFE service):

- PP Type 7i: FBLFE loudspeaking device.

For ultra-band (48 kHz service):

- PP Type 8a: Ultra-band handset or headset.
- PP Type 8b: Ultra-band loudspeaking device.

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.

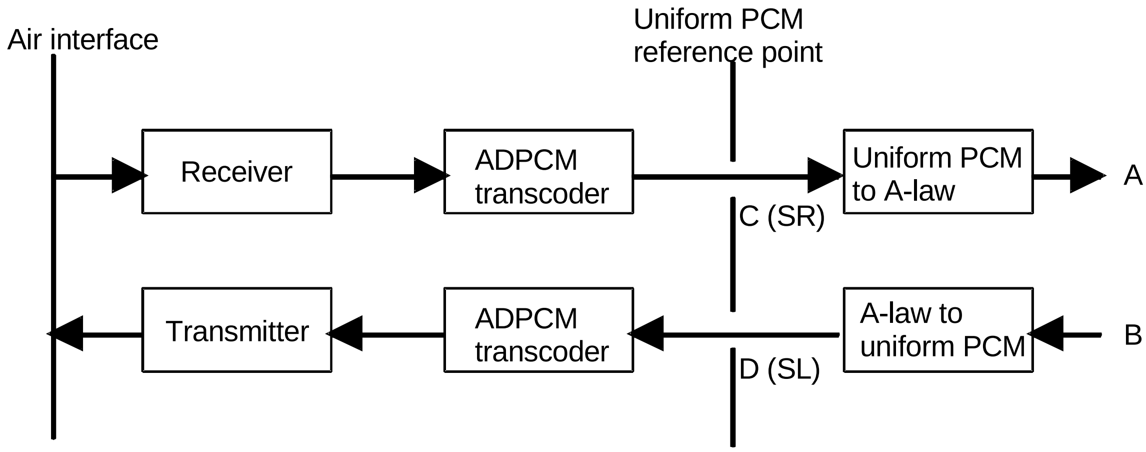


Figure 7.1: Reference PP (narrowband)

The RePP incorporates the specified transcoder algorithm as described in Recommendation ITU-T 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 TCLw value 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 Recommendation ITU-T 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, type 1d is 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], G.711 [19] and LC3plus [58] in NB mode.

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

The PP shall set the flags "echo parameters" in the IE <Terminal capability> (octet 3b) according to its TCLw capabilities (see ETSI EN 300 175-5 [5], clause 7.7.41). See table 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], G.711 [19] and LC3plus [58] in NB mode.

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 Recommendation ITU-T P.51 [32]. The artificial ear shall conform to Recommendation ITU-T 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 ETSI 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], G.711 [19] and LC3plus [58] in NB mode.

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 ETSI 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 the commonly used 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], G.711 [19] and LC3plus [58] in NB mode.

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 ETSI 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], G.711 [19] and LC3plus [58] in NB mode.

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], G.711 [19] and LC3plus [58] in NB mode.

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 Recommendation ITU-T 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], MPEG-4 ER AAC-LD [48] operating at 32 kbit/s and LC3plus [58] in WB mode.

7.2.9.3 Specification

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

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 ETSI 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.

7.2.10 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 than the type 2a.

Type 2b includes strong echo suppression (TCL) 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], MPEG-4 ER AAC-LD [48] operating at 32 kbit/s and LC3plus [58] in WB mode.

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 TCL 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", (TCL > 55 dB, see ETSI 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 TCL 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], MPEG-4 ER AAC-LD [48] operating at 32 kbit/s and LC3plus [58] in WB mode.

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 TCL 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", (TCL > 55 dB, see ETSI 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 TCL 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 2b, 2c, 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], MPEG-4 ER AAC-LD [48] operating at 32 kbit/s and LC3plus [58] in WB mode.

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], MPEG-4 ER AAC-LD [48] operating at 32 kbit/s and LC3plus [58] in WB mode.

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 or headset

7.2.14.1 Introduction

The type 5a super-wideband 14 kHz handset or headset specification is based on HATS methodology.

It provides 14 kHz frequency range and applies to handset and headset devices.

7.2.14.2 Compatible services and codecs

Compatible codecs are MPEG-4 ER AAC-LD [48] and LC3plus [58] in SWB mode, each operating at 64 kbit/s.

7.2.14.3 Specification

The complete specification of PP type 5a is described in clause 7.5.8.

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

7.2.15.1 Introduction

The type 5b super-wideband 14 kHz loudspeaking specification is based on HATS methodology. It provides 14 kHz frequency range.

7.2.15.2 Compatible services and codecs

Compatible codecs are MPEG-4 ER AAC-LD [48] and LC3plus [58] in SWB mode, each operating at 64 kbit/s.

7.2.15.3 Specification

The complete specification of handsfree type 5b is described in clause 7.5.9.

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 ETSI EN 300 175-8 [8], annex B.

7.2.16.2 Compatible services and codecs

Type 6 provides telephony 3,1 kHz service and is compatible with codecs G.726 [24], G.711 [19] and LC3plus [58] in NB mode.

7.2.16.3 Specification

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

7.2.17 PP Type 7a: fullband 20 kHz handset or headset

7.2.17.1 Introduction

The type 7a fullband 20 kHz handset or headset specification is based on HATS methodology.

It provides 20 kHz frequency range and applies to handset and headset devices.

7.2.17.2 Compatible services and codecs

It is compatible with LC3plus [58] in fullband mode operating at 64 kbit/s, 96 kbit/s, 128 kbit/s, 160 kbit/s, 192 kbit/s or 256 kbit/s.

7.2.17.3 Specification

The complete specification of PP type 7a is described in clause 7.5.10.

7.2.18 PP Type 7b: fullband 20 kHz loudspeaking handsfree

7.2.18.1 Introduction

The type 7b fullband 20 kHz loudspeaking specification is based on HATS methodology. It provides 20 kHz frequency range.

7.2.18.2 Compatible services and codecs

It is compatible with LC3plus [58] in fullband mode operating at 64 kbit/s, 96 kbit/s, 128 kbit/s, 160 kbit/s, 192 kbit/s or 256 kbit/s.

7.2.18.3 Specification

The complete specification of handsfree type 7b is described in clause 7.5.11.

7.2.19 PP Types 7c, d, e, f: fullband 20 kHz stereo audio device

7.2.19.1 Introduction

The types 7c, 7d, 7e and 7f provide specifications of fullband 20 kHz frequency range stereo streaming devices. The device could be a handset, headset or stereo loudspeaker.

7.2.19.2 Compatible services and codecs

Type 7c is compatible with LC3plus [58] operating in FB stereomode at 128 kbit/s for left and for right channel combined and 10 ms frame size.

Type 7d is compatible with LC3plus [58] operating in FB stereomode at 192 kbit/s for left and for right channel combined and 10 ms frame size.

Type 7e is compatible with LC3plus [58] operating in FB stereomode at 256 kbit/s for left and for right channel combined and 10 ms frame size.

Type 7f is compatible with LC3plus [58] operating in FB stereo mode at 256 kbit/s for 5 ms frame size, 320 kbit/s for 2,5 ms frame size and at 512 kbit/s for 2,5 ms frame size.

7.2.19.3 Specification

The complete specification of PP types 7c, d, e, f is described in clause 7.5.12.

Only the receive direction is specified. The sending direction should be muted or fulfil specification of PP type 7a or 7b, respectively, depending if the nature of the device is more of type handset/headset (7a) or handsfree/loudspeaking/group audio terminal (7b).

7.2.20 PP Type 7g: FBHR 24 kHz headset device

7.2.20.1 Introduction

Type 7g FBHR headset device specification provides 24 kHz frequency range at bit rates 128 kbit/s, 160 kbit/s and 192 kbit/s for 10 ms frame size.

7.2.20.2 Compatible services and codecs

Type 7g is compatible with LC3plus [58], clause 5.8, operating in FBHR mode at 128 kbit/s, 160 kbit/s and 192 kbit/s.

7.2.20.3 Specification

The complete specification of PP type 7g is described in clause 7.5.13.

7.2.21 PP Type 7h: FBHR 24 kHz loudspeaking device

7.2.21.1 Introduction

Type 7h FBHR loudspeaking device specification provides 24 kHz frequency range at bit rates 128 kbit/s, 160 kbit/s and 192 kbit/s for 10 ms frame size.

7.2.21.2 Compatible services and codecs

Type 7h is compatible with LC3plus [58], clause 5.8, operating in FBHR mode at 128 kbit/s, 160 kbit/s and 192 kbit/s.

7.2.21.3 Specification

The complete specification of PP type 7h is described in clause 7.5.14.

7.2.22 PP Type 7i: FBLFE 250 Hz loudspeaking device

7.2.22.1 Introduction

Type 7i FBLFE loudspeaking device specification provides a frequency range from 0 Hz to 250 Hz at bit rate 32 kbit/s for 10 ms frame size and bit rate 64 kbit/s for 5 ms frame size.

7.2.22.2 Compatible services and codecs

Type 7i is compatible with LC3plus [58], clause 5.3.15, operating in FBLFE mode at 32 kbit/s for 10 ms frame size and at 64 kbit/s for 5 ms frame size.

7.2.22.3 Specification

The complete specification of PP type 7i is described in clause 7.5.15.

7.2.23 PP Type 7j: fullband 20 kHz low-latency microphone device

7.2.23.1 Introduction

The type 7j fullband 20 kHz low-latency microphone device specification provides 20 kHz frequency range and applies to microphone devices operating at a frame size of 2,5 ms.

7.2.23.2 Compatible services and codecs

Type 7j is compatible with LC3plus [58] operating in FB mode at 128 kbit/s or 256 kbit/s for 2,5 ms frame size.

7.2.23.3 Specification

The complete specification of PP type 7j is described in clause 7.5.16. Apart from the delay requirement, the specification of type 7j corresponds to the specification of type 7a and is described in clause 7.5.10. Only the specifications for send direction are applicable for type 7j.

7.2.24 PP Type 8a: ultra-band 48 kHz headset device

7.2.24.1 Introduction

Type 8a ultra-band 48 kHz headset device specification provides a frequency range from 0 kHz to 48 kHz at bit rates 160 kbit/s, 192 kbit/s, 256 kbit/s and 320 kbit/s for 10 ms frame size.

7.2.24.2 Compatible services and codecs

Type 8a is compatible with LC3plus [58], clause 5.8, operating in UBHR mode at 160 kbit/s, 192 kbit/s, 256 kbit/s and 320 kbit/s for 10 ms frame size.

7.2.24.3 Specification

The complete specification of PP type 8a is described in clause 7.5.17.

7.2.25 PP Type 8b: ultra-band 48 kHz loudspeaking device

7.2.25.1 Introduction

Type 8b ultra-band 48 kHz loudspeaking device specification provides a frequency range from 0 kHz to 48 kHz at bit rates 160 kbit/s, 192 kbit/s, 256 kbit/s and 320 kbit/s for 10 ms frame size.

7.2.25.2 Compatible services and codecs

Type 8b is compatible with LC3plus [58], clause 5.8, operating in UBHR mode at 160 kbit/s, 192 kbit/s, 256 kbit/s and 320 kbit/s for 10 ms frame size.

7.2.25.3 Specification

The complete specification of PP type 8b is described in clause 7.5.18.

7.3 Audio transmission types applicable to Fixed Parts

7.3.0 General

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

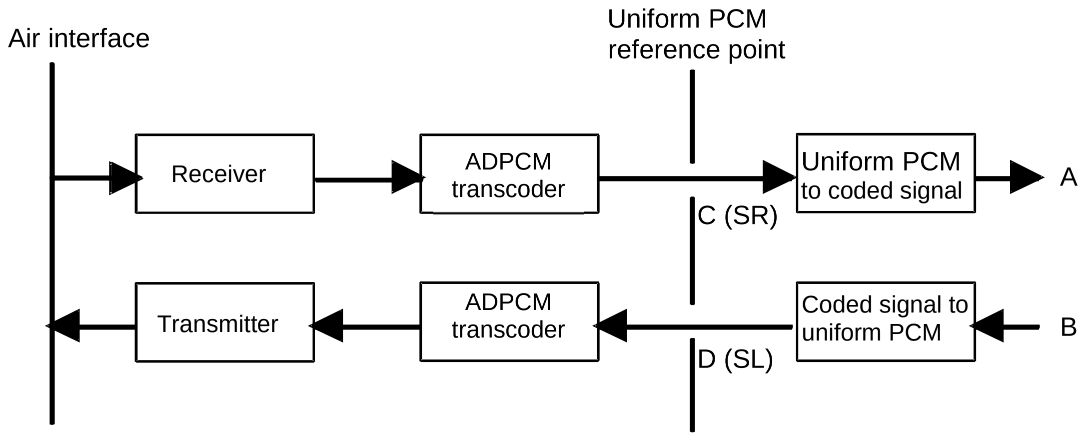


Figure 7.2: Reference FP

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

7.3.2.1 Introduction

The FP type 1a is an 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], G.711 [19] and LC3plus [58] in NB mode 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

7.3.2.3.1.0 General

FP shall perform transparent transcoding to/from ADPCM G.726 [24] or LC3plus [58] in NB mode 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 from clause 7.6.1.3 are not used.

FP shall be transparent regarding audio levels unless the features in clause 7.6.1.3 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 Recommendation ITU-T 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 Recommendations ITU-T 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 ETSI 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: FP 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 FP adaptive volume control.

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

7.3.3 FP Type 1b: Fixed Part for ISDN Network

7.3.3.1 Introduction

The increasing use of internet and VoIP technologies in the networks forced 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 of 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.

7.3.3.2 Compatible services and codecs

Type 1b provides telephony 3,1 kHz service and is compatible with codecs G.726 [24], G.711 [19] and LC3plus [58] in NB mode 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

7.3.3.3.1.0 General

FP shall perform transparent transcoding to/from ADPCM G.726 [24] or LC3plus [58] in NB mode 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 Recommendation ITU-T 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 Recommendations ITU-T 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 ETSI 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 optional echo 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 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 7.6.2.2.

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 FP adaptive volume control.
- 7.4.2 Echo canceller in Fixed Part.
- 7.4.3 Echo suppressor in Fixed Part.

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 ETSI 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], G.711 [19] and LC3plus [58] in NB mode over air interface.

7.3.4.3 Specification

7.3.4.3.1 Transcoding, equalization and conversion

7.3.4.3.1.0 General

FP shall perform conversion to analog 2-wire telephone interface from ADPCM G.726 [24], PCM G.711 [19] or LC3plus [58] in NB mode 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 Recommendation ITU-T 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 Recommendations ITU-T G.107 [i.14], G.108 [i.15] and G.109 [i.16].

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 FP 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], G.711 [19] and LC3plus [58] in NB mode 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] or LC3plus [58] in NB mode are also allowed.

An example of a protocol stack for media is RTP [i.4] over UDP [i.3] over IP [i.2]. Other stacks may be allowed. Any transport below IP is allowed.

Typical physical interfaces are:

- IEEE 802.3 [i.28].

- ADSL/VDSL over a phone line.
- Wi-Fi® 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 media stacks are the following.

Table 7.4: Examples of media stacks for VoIP narrowband interface

Example 1	Example 2	Example 3
Voice G.711 [19], AMR-NB [69], LC3plus [58] in NB mode	Voice G.711 [19], AMR-NB [69], LC3plus [58] in NB mode	Voice G.711 [19], AMR-NB [69], LC3plus [58] in NB mode
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

7.3.5.3.1.0 General

FP shall perform transparent transcoding to/from ADPCM G.726 [24] or LC3plus [58] in NB mode from/to PCM G.711 [19] A-law or μ-law.

In general, transcoding or self-tandeming is needed in FPs. FP shall provide a jitter buffer to convert the packet oriented VoIP network to the Time Division Multiplex (TDM) system of the DECT air interface and to adapt the clock difference between sender and receiver. The air interface codec may but is not required to be equal to VoIP line interface codec. However, the transcoding quality shall be taken care for. Only if it can be assured that there is no clock difference between sender and receiver and the packet loss is near to zero no transcoding/self-tandeming is required.

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 Recommendation ITU-T 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 Recommendations ITU-T 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 ETSI 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 (see 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 34 dB < TCLw < 46 dB, THEN the FP SHALL activate the echo cancellation (see clause 7.4.2) or suppression facility (see 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:

- 7.6.4.4 Clock accuracy.
- 7.6.4.5 Send Jitter.
- 7.6.4.6 Send and receive delay - round-trip delay.

The FP may implement the following feature:

- 7.6.4.3 Adaptive volume control.
- 7.4.2 Echo canceller in Fixed Part.
- 7.4.3 Echo suppressor in Fixed Part.

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 super-wideband (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], MPEG-4 ER AAC-LD [48] and LC3plus [58] 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 ETSI 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 < TCL < 46 dB.
- PP with TCL > 46 dB (Full TCL).
- PP with TCL > 55 dB (TCL compatible with VoIP).

If the FP does not implement any optional echo 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 optional echo 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 TCL > 55 dB, THEN the FP SHALL NOT activate any echo cancellation or suppression facility for echo coming from the PP (see clause 7.4.2 or 7.4.3).
- If the PP has 46 dB < TCL < 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 42 dB < TCL < 46 dB, THEN the FP SHALL activate the echo cancellation (see clause 7.4.2) or suppression facility (see 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 described in the following clauses:

- 7.6.5.1 Adaptive volume control.
- 7.4.2 Echo canceller in Fixed Part.

- 7.4.3 Echo suppressor in Fixed Part.

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

7.3.7 FP Type 5: VoIP wideband, super-wideband, fullband, FBHR or ultra-band 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), 14 kHz (super-wideband), 20 kHz (fullband), 24 kHz (FBHR) or 48 kHz (ultra-band) services.

The FP type 5 is identical to type 3, but with any wideband, super-wideband, fullband, FBHR or ultraband codec on top of the VoIP interface.

7.3.7.2 Compatible services, physical interfaces and codecs

G.722 [21], G.729.1 [25], MPEG-4 ER AAC-LD [48] and LC3plus [58] 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 a protocol stack for media is RTP [i.4] over UDP [i.3] over IP [i.2]. Other stacks may be allowed (e.g. with TCP). Any transport below IP is allowed.

Typical physical interfaces are:

- IEEE 802.3 [i.28].
- ADSL/VDSL over a phone line.
- Wi-F® 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 media stacks are the following.

Table 7.5: Examples of media stacks for VoIP wideband interface

Example 1	Example 2	Example 3
G.722 [21], AMR-WB [70], EVS [71], OPUS [72] or LC3plus [58] in WB mode	G.722 [21], AMR-WB [70], EVS [71], OPUS [72] or LC3plus [58] in WB mode	G.722 [21], AMR-WB [70], EVS [71], OPUS [72] or LC3plus [58] in WB mode
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, transcoding or self-tandeming is needed in FPs. FP shall provide a jitter buffer to convert the packet oriented VoIP network to the Time Division Multiplex (TDM) system of the DECT air interface and to adapt the clock difference between sender and receiver. The air interface codec may but is not required to be equal to VoIP line interface codec. However, the transcoding quality shall be taken care for. Only if it can be assured that there is no clock difference between sender and receiver and the packet loss is near to zero no transcoding/self-tandeming is required.

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 ETSI 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 < TCL < 46 dB.
- PP with TCL > 46 dB (Full TCL).
- PP with TCL > 55 dB (TCL compatible with VoIP).

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

If the FP does implement any optional echo 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 TCL > 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 < TCL < 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 42 dB < TCL < 46 dB, THEN the FP SHALL activate the echo cancellation (see clause 7.4.2) or suppression facility (see clause 7.4.3).

7.3.7.3.4 Transmission specification

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

- 7.6.6.4 Clock accuracy.
- 7.6.6.5 Send Jitter.
- 7.6.6.6 Send and receive delay - round trip delay.

The FP may implement the following features in the following clauses:

- 7.6.6.3 FP adaptive volume control.
- 7.4.2 Echo canceller in Fixed Part.
- 7.4.3 Echo suppressor in Fixed Part.

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

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 a three- 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 Part (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 centres 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 TCL(w) > 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 TCL(w) > 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 has 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.

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 an echo signal and subtracting such estimation (with the proper phase), from the signal coming from the end which echo is to be cancelled (see Recommendations ITU-T 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).

NOTE: The feature should never be activated for other PP handset types, since they all have TCL(w) > 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 Recommendation ITU-T 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 than 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 is 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 echo suppressor should be implemented taking into account the constraints described in clause 7.4.1.

NOTE: The feature should never be activated for other PP handset types, since they all have TCL(w) > 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 Recommendation ITU-T 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 Recommendation ITU-T P.64 [36], using an artificial mouth conforming to Recommendation ITU-T 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 an 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 Recommendation ITU-T 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 shall be:

- 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 Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], formula 2.1, over bands 4 to 17, using m = 0,175 and the sending weighting factors from Recommendation ITU-T 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 Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], formula 2.1, over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of Recommendation ITU-T P.79 [37].

The artificial ear sensitivity shall be corrected using the real ear correction of table 2 of Recommendation ITU-T 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 ETSI 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 ETSI 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 ETSI 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 ETSI 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 Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], using m = 0,225 and the weighting factors in table 3 of Recommendation TU-T P.79 [37].

7.5.1.3.2 Listener sidetone

Requirement

There are no 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 should be measured and should 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 ETSI 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 3: The pressure intensity index as defined in ISO 9614 [46] may prove to be a suitable method for assessing the diffuse field.

NOTE 4: 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 5: 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 Recommendation ITU-T 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 Recommendation ITU-T 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 7.5.1.1.1, 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 Recommendation ITU-T P.79 [37], formulas E-3 and E-2 and the coefficients Ki in table E.1 in Recommendation ITU-T 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 ETSI 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 ETSI 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 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 Recommendation ITU-T 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 Recommendation ITU-T 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 6.4.3 clarifies 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 ETSI 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 ETSI 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 Recommendation ITU-T P.64 [36], using an artificial mouth conforming to Recommendation ITU-T 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 an 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 Recommendation ITU-T G.712 [20] and Recommendation ITU-T 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 an 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 in Recommendations ITU-T 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 an 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 an 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 an FP.

The noise level at the PCM interface point A shall be measured using psophometric weighting as described in Recommendation ITU-T G.223 [17], table 4.

7.5.1.7.2 Band-limited noise

Requirement

The narrowband 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 an 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 an 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 an 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.

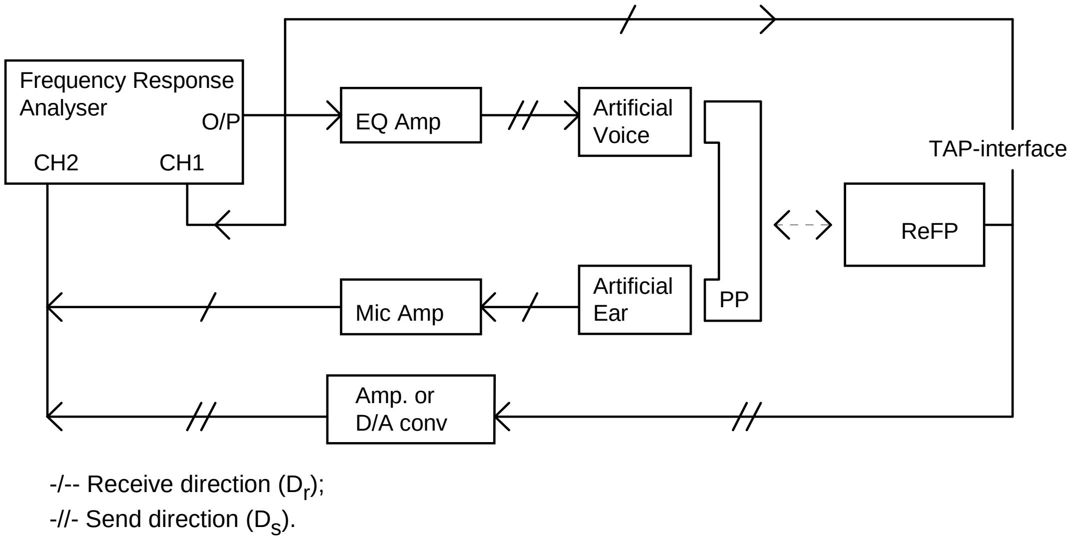


Figure 7.2a: PP delay test configuration

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 Recommendation ITU-T 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)

7.5.2.0 General

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

The measurement method is:

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 Recommendation ITU-T 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 $1/12^{th}$ 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 Recommendation ITU-T 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 shall be classified according to table 7.12.

Table 7.12: Category regarding "duplex capability" depending on $A_{H,S,dt}$

Category (according to Recommendation ITU-T P.340 [39])	**1**	**2a**	**2b**	**2c**	**3**
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
AH,S,dt [dB]	≤ 3	≤ 6	≤ 9	≤ 12	> 12

The category regarding duplex capability in send direction shall be at least 2b.

In general table 7.12 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.

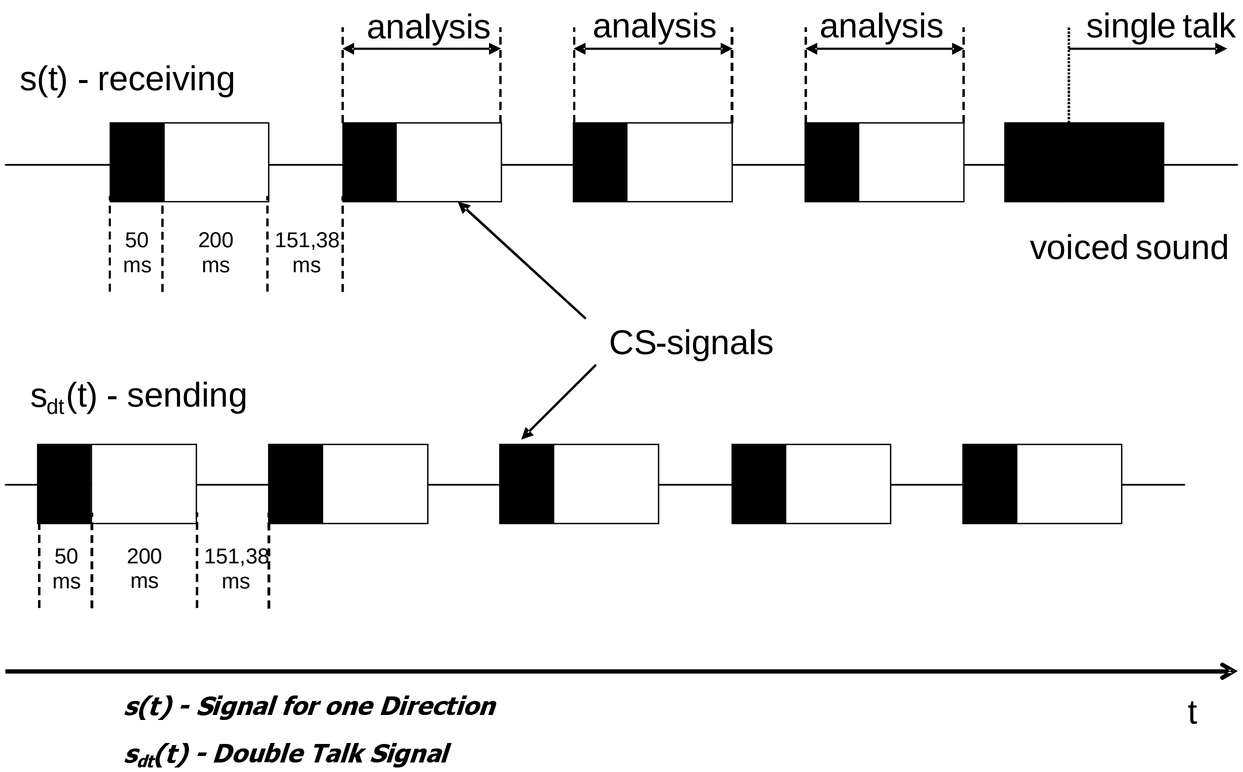


Figure 7.2b: Double Talk Test Sequence with overlapping CS signals in sending and receiving direction

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 shall be classified according to table 7.13.

Table 7.13: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to Recommendation ITU-T 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

The category regarding duplex capability in receive direction shall be at least 2b.

In general table 7.13 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) shall 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 Recommendation ITU-T P.501 [41] with increasing level for each CSS burst.

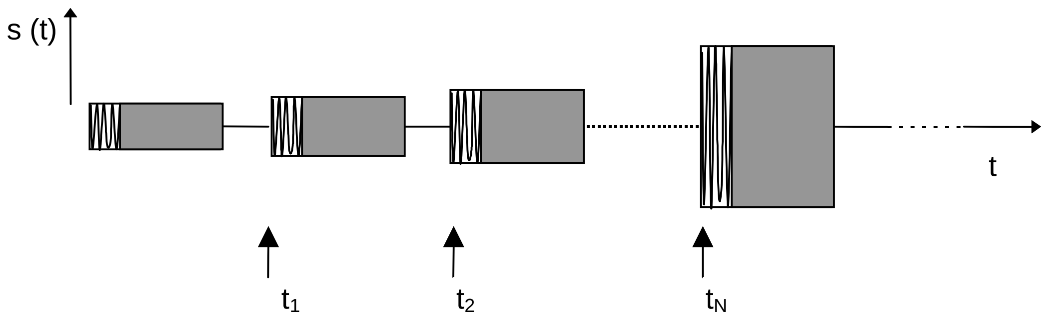


Figure 7.2c: Test Signal to Determine the Minimum Activation Level and the Built-up Time

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 Recommendation ITU-T 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 Recommendation ITU-T P.501 [41] with increasing level for each CSS burst.

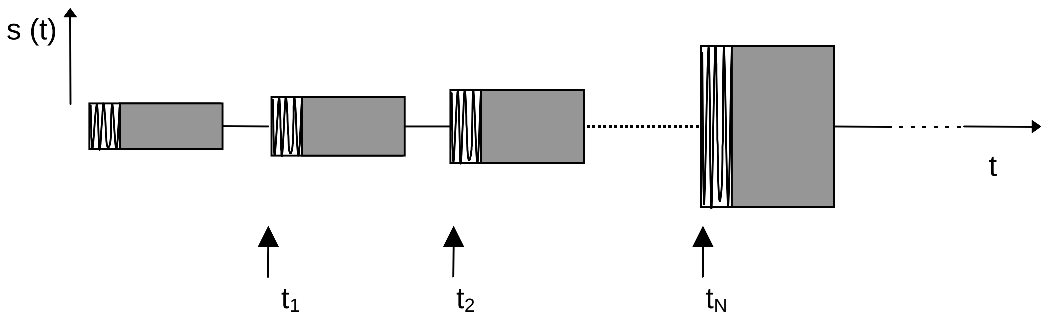


Figure 7.2d: Test Signal to Determine the Minimum Activation Level and the Built-up Time

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 Recommendation ITU-T 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 Frequency responses

7.5.3.1.1 Sending

7.5.3.1.1.1 Send frequency response - nominal position

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	

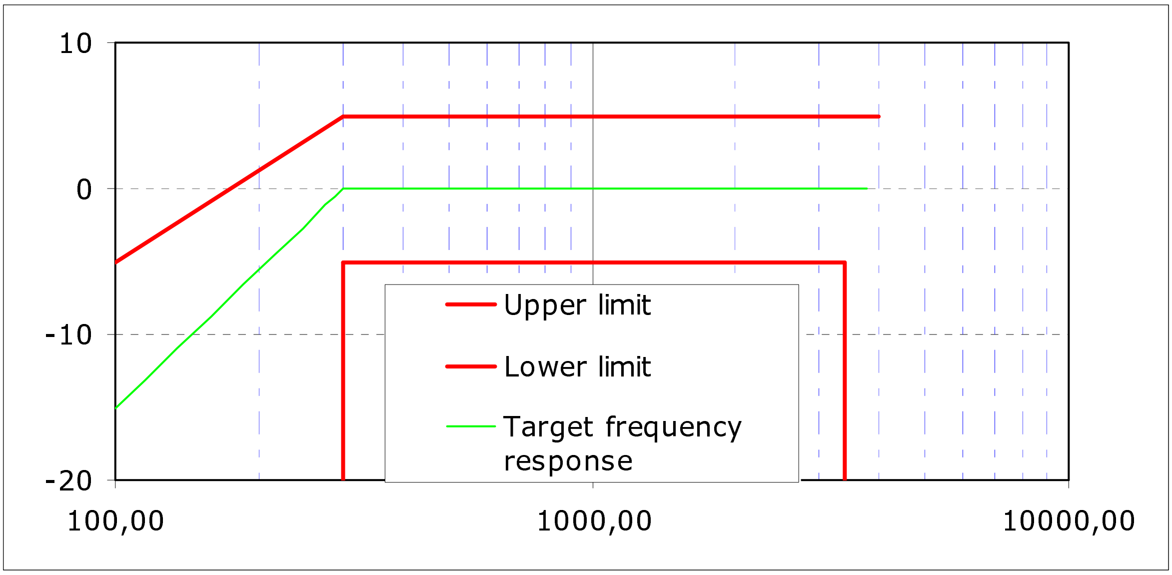


Figure 7.3: Send frequency response mask for PP types 1c and 1d

NOTE: The basis for the target frequency responses in sending and receiving is the orthotelephonic reference response which is measured between two 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. The mask is a floating or "best fit" mask.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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 standard position of the HATS. 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 Recommendation ITU-T P.380 [40] the results are averaged (averaged value in dB, for each frequency).

Measurements shall be made at one twelfth-octave bands as given by the IEC 61260-1 [45] 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.

7.5.3.1.1.2 Send frequency response - positional robustness

Requirement

For each of the modified handset positions, the send frequency response shall be within a given mask. The mask values per frequency are identical to table 7.14, except that an additional tolerance is provided for certain positions. Table 7.14a provides the offset in dB for the lower limit.

Table 7.14a: Tolerance mask offsets for send frequency response

Position	Offset Lower Limit
UP	-1 dB
DOWN	-2 dB
AWAY	-1 dB

Measurement method

The test arrangement and measurement is identical to clause 7.5.3.1.1.1. Instead of the standard handset position, the three modified positions according to table 6.3 for sending direction shall be used. The resulting three frequency responses shall be reported separately for each position.

7.5.3.1.2 Receiving

7.5.3.1.2.1 Receive frequency response - nominal position

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 and 7.5. 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 2N and 13N applicable only to improved	Lower Limit 2N and 13N applicable only to improved
100	4		6	
300	4	-4	6	-6
1 500	4	-4	6	-6
3 000	4	-4	6	-6
3 400	4	-4	6	-6
4 000	4		6	

NOTE 1: The limit curves are 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. The mask is a floating or "best fit" mask.

NOTE 2: The basis for the target frequency responses in sending and receiving is the orthotelephonic reference response which is measured between two 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 3: With current technology it may be difficult or even not possible to achieve the desired frequency response characteristics for handsets with 2 N application force.

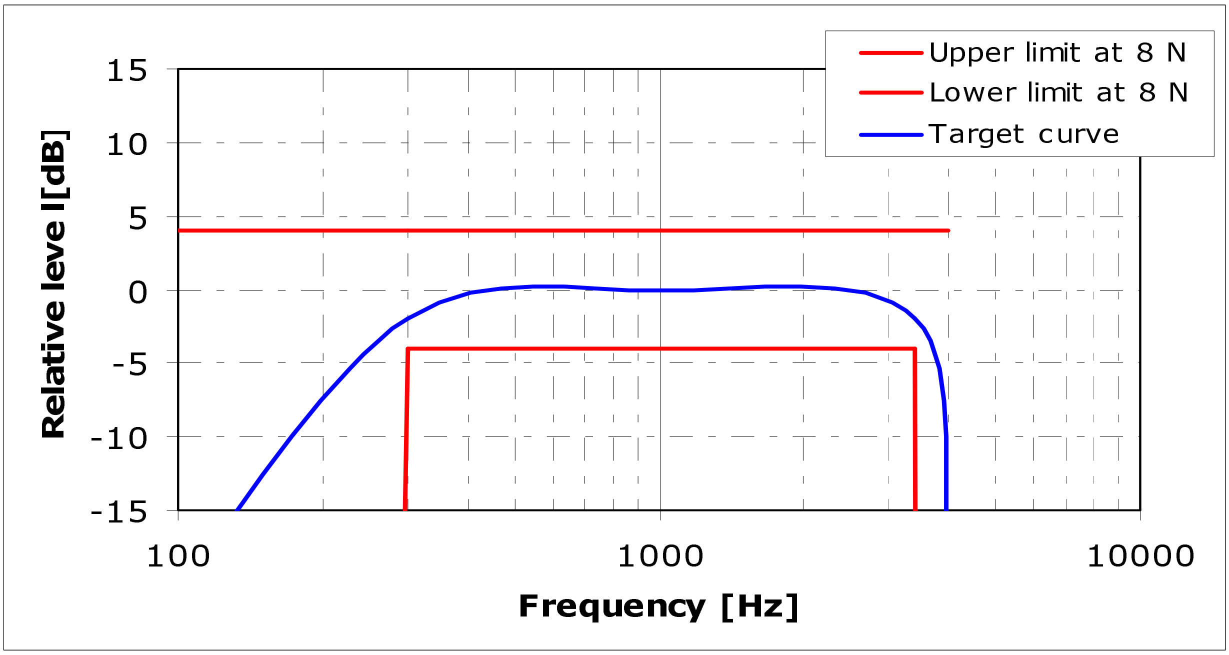


Figure 7.4: Receive frequency response mask for 8N application force

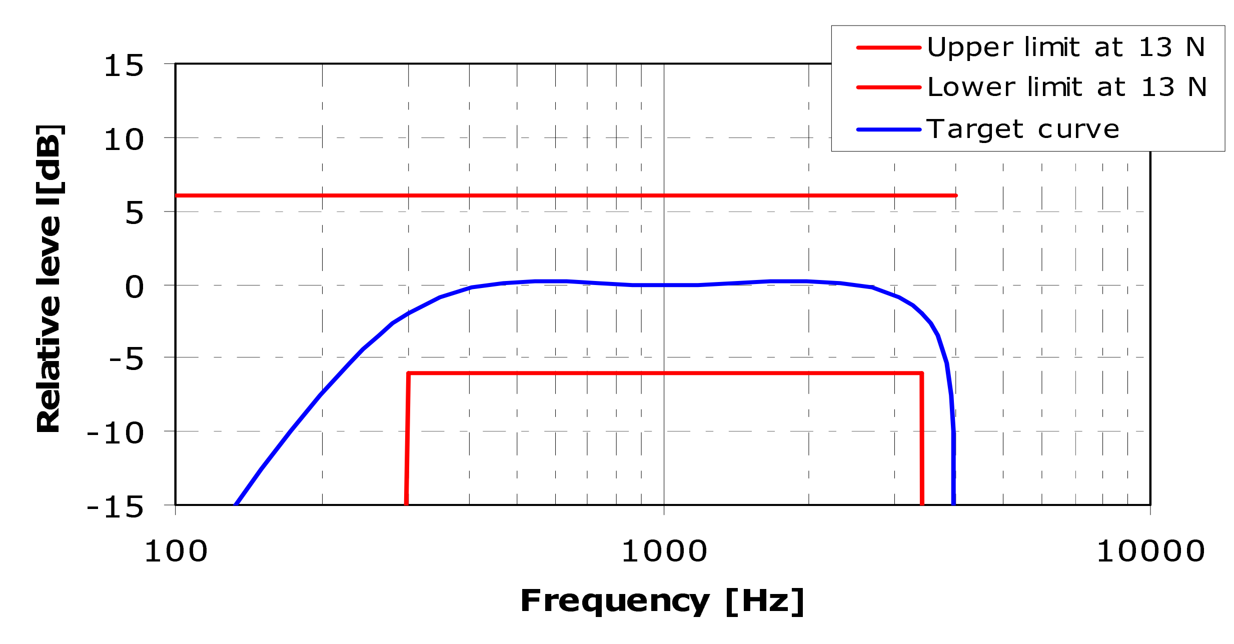


Figure 7.5: Receive frequency response mask for 2N and 13N application force

Figure 7.6: Void

Measurement method

Receive frequency response is the ratio of the measured sound pressure and the input level.
(dB relative Pa/V)

$$S_{Jedf} = 20 \log (pe_{df} / v_{RCV}) \text{ dB rel 1 Pa / V}$$

S_{Jedf}	Receive Sensitivity; Junction to HATS Ear with diffuse field correction.
pe_{df}	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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -16 dBm0, measured according to Recommendation ITU-T 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 standard position of the HATS. The application forces used to apply the handset against the artificial ear is 2 N, 8 N and 13 N.

In case of headset measurements the tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [40] the results are averaged (averaged value in dB, for each frequency).

The HATS is diffuse field equalized as described in Recommendation ITU-T 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 bands as given by the IEC 61260-1 [45] 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.

7.5.3.1.2.2 Receive frequency response - positional robustness

Requirement

For each of the modified handset positions, the receive frequency response shall be within a given mask. The mask values per frequency are identical to table 7.15, except that an additional tolerance is provided for certain positions. Table 7.15a provides the offset in dB for the lower limit.

Table 7.15a: Tolerance mask offsets for receive frequency response

Position	Offset Lower Limit
Ye_{-5} Ze_{-5}	-1 dB
Ye_{0} Ze_{+5}	-1 dB
Ye_{+5} Ze_{-5}	-1 dB

Measurement method

The test arrangement and measurement is identical to clause 7.5.3.1.2.1. Instead of the standard handset position, the three modified positions according to table 6.4 for receiving direction shall be used. The resulting three frequency responses shall be reported separately for each position.

7.5.3.2 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 dB.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS. 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 Recommendation ITU-T P.380 [40] the results are averaged (averaged value in dB, for each frequency).

The send sensitivity shall be calculated for each of the 14 frequency bands given in table 1 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], formula A-23b, over bands 4 to 17, and the send weighting factors from Recommendation ITU-T 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 dB.
- RLR (binaural headset) = 8 dB ± 3 dB for each earphone.

The nominal value of RLR is the RLR closest to the nominal requirement.

The minimum difference between nominal RLR and minimum (loudest, maximum volume setting) RLR shall be higher than 6 dB.

NOTE 1: The mechanical design of some UEs may make it impossible to seal the ear-piece to the knife edge of the ITU-T artificial ear. Minimal additional methods may be used to provide the seal provided that they do not affect the mounting position of the UE with respect to the Mouth Reference Point and the Ear Reference Point.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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 standard position of the HATS. The HATS is **NOT** diffuse field equalized as described in Recommendation ITU-T P.581 [43]. The DRP-ERP correction as defined in Recommendation ITU-T 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, 8 N will be used.

In case of headset measurements the tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

The receive sensitivity shall be calculated for each of the 14 frequency bands given in table 1 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], formula A-23c, over bands 4 to 17, and the receive weighting factors from table 1 of Recommendation ITU-T P.79 [37]. No leakage correction shall be applied for the measurement.

The test shall be repeated for maximum and minimum volume control setting.

NOTE 2: Currently the Loudness Ratings Calculation is still based on the ERP. Therefore no diffuse field correction is applied and still the DRP-ERP correction is used.

7.5.3.2.2 Void

Table 7.16: Void

Table 7.17: Void

7.5.3.2.3 Void

7.5.3.2.4 Microphone mute

Requirement

The SLR (Send Loudness Rating) with microphone mute on shall be at least 50 dB higher than with microphone mute off.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 7.5.3.2.1.1, but its microphone shall be configured to be muted.

Measurement and calculation method are the same as in clause 7.5.3.2.1.1.

7.5.3.2.5 Positional robustness

7.5.3.2.5.1 Send Loudness Rating

Requirement

The difference (in dB) between the SLR measured in each of the three modified handset positions and the one in determined standard position (STD) shall be in the range of -3 to +3 dB.

Measurement method

In addition to the test setup and measurement of clause 7.5.3.2.1.1, each of the three modified handset positions for sending direction according to table 6.3 shall be applied. SLR and delta-SLR values should be calculated and reported for each position.

7.5.3.2.5.2 Receive Loudness Rating

Requirement

The difference (in dB) between the RLR measured in each of the three modified handset positions and the one in standard position (STD) shall be in the range -3 to +3 dB.

Measurement method

In addition to the test setup and measurement of clause 7.5.3.2.1.2, each of the three modified handset positions for receiving direction according to table 6.4 shall be applied. An application force of 8 N is used. RLR and delta-RLR values should be calculated and reported for each position.

7.5.3.2.6 Send Loudness Level

Requirements

The nominal value of Send Loudness Level (SLL) shall be:

$$SLL = 75 \text{ phon} \pm 4 \text{ phon}$$

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The handset or headset terminal is set-up as described in clause 6.10.3.

The loudness (in sone) of the recorded signal is calculated according to ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is calculated according to clause 9 of ITU-T P.700 [74].

7.5.3.2.7 Receive Loudness Level

Requirements

The nominal value of Receive Loudness Level (RLL) for handsets, monaural and binaural/stereo headsets shall be:

$$RLL = 75 \text{ phon} \pm 4 \text{ phon}$$

In case a user controlled receive volume control is provided, for at least one setting of the control the RLL shall meet the nominal value.

When the control is set to maximum, the RLL shall not be louder than 89 phon. With the volume control set to the minimum position the RLL shall not be quieter than 58 phon.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The handset or headset terminal is set-up as described in clause 6.10.3.

The loudness (in sone) of the recorded signal is calculated according to Recommendation ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is determined as follows:

- Handsets, monaural headsets: the loudness level is calculated according clause 8.2 of Recommendation ITU-T P.700 [74] by using the loudness value divided by two (loudness halving for monaural listening).
- Binaural headsets: the loudness level is calculated according clause 8.2 of Recommendation ITU-T P.700 [74] by using directly the loudness value (loudness summation for binaural listening is retained).

7.5.3.3 Sidetone

7.5.3.3.1 Sidetone masking rating (STMR)

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 shall 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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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 standard position of the HATS and the application force shall be 13 N 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 bands as given by the IEC 61260-1 [45] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band (Recommendation ITU-T 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 (LmeST), as expressed in dB, and the SideTone Masking Rating (STMR) (in dB) shall be calculated from the formula 5-1 of Recommendation ITU-T P.79 [37], using m = 0,225 and the weighting factors in table 3 of Recommendation ITU-T P.79 [37].

7.5.3.3.2 Void

7.5.3.3.3 Sidetone delay

Requirement

The maximum sidetone-round-trip delay shall be ≤ 5 ms, measured in an echo-free setup.

Measurement method

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The test signal is a CS-signal complying with Recommendation ITU-T 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 Recommendation ITU-T P.501 [41]. The level of the signal shall be -4,7 dBPa at the MRP.

The cross-correlation function $\Phi_{xy}(\tau)$ 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:

$$\Phi_{xy}(\tau) = \frac{1}{T}\int_{t=\frac{-T}{2}}^{\frac{T}{2}} S_x(t) \cdot S_y(t+\tau) \cdot dt$$

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 $E(\tau)$ of the cross-correlation function $\Phi_{xy}(\tau)$. 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 $E(\tau)$ is calculated by the Hilbert transformation $H\{\Phi_{xy}(\tau)\}$ of the cross-correlation:

$$H\{\Phi_{xy}(\tau)\} = \sum_{u=-\infty}^{\infty} \frac{\Phi_{xy}(u)}{\Pi(\tau-u)}$$

$$E(\tau) = \sqrt{\left[\Phi_{xy}(\tau)\right]^2 + \left[H\{\Phi_{xy}(\tau)\}\right]^2}$$

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 Terminal Coupling Loss weighted (TCLw)

Requirement

The TCLw shall be $\geq$ 55 dB at nominal setting of the volume control.

With the volume control set to maximum TCLw shall be $\geq$ 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 the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS and the application force shall be 2 N on the artificial ear type 3.3 or type 3.4 as specified in Recommendation ITU-T 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 the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The signal level shall be -10 dBm0.

The TCLw is calculated according to Recommendation ITU-T 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 first 17,0 s of the test signal (6 sentences) are discarded from the analysis to allow for convergence of the acoustic echo canceller. The analysis is performed over the remaining length of the test sequence (last 6 sentences).

NOTE: Care should be taken when measuring TCLw: the echo return not to be masked by the residual noise or comfort noise when implemented.

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 the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41] is applied. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with Recommendation ITU-T 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 7.6a.

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;

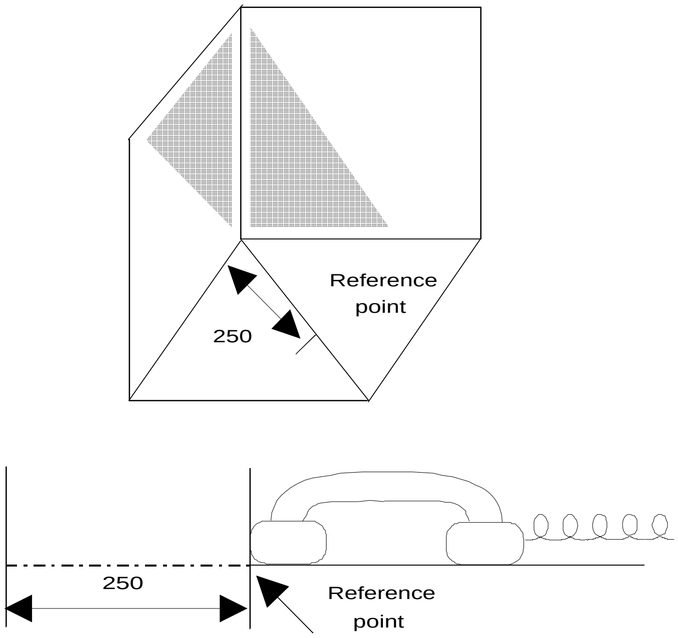
3) the extremity of the handset shall coincide with the normal to the reference point, as shown in figure 7.6a.

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) for monaural headsets the receivers shall be placed centrally at the reference point as shown in figure 7.6a;
for binaural headset, the receivers are placed symmetrically to the diagonal line on both sides of the reference point;

3) the headset microphone is positioned as close as possible to the receiver.



NOTE: All dimensions are 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 setup is as described in clause 6.10.3. The handset is mounted in the standard position of the HATS. After a correct activation of the system, a sinewave signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz is applied. 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.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

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 handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

After a correct activation of the system, a digitally simulated sinewave signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz is applied to the digital interface respectively. The sinewave signal shall be applied to the digital interface at the level of -16 dBm0.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal shall be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 10 kHz.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

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. The handset is mounted in the standard position of the HATS.

For a correct activation of the system, the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T 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-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 -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.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.3.6.2 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 7.21.

Table 7.21: Out of band signal limits, receiving

Frequency	Minimum attenuation
4,6 kHz	35 dB
8 kHz	45 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 handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

After a correct activation of the system, a digitally simulated sinewave signal at frequencies of 500 Hz, 1 000 Hz, 2 000 Hz and 3 150 Hz are 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.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -16 dBm0. The out of band signal shall be measured at DRP of the artificial ear with the diffuse field equalization active.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.3.7 Noise

7.5.3.7.1 Sending

Requirement

The maximum noise level produced by the PP 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 the female speaker of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T 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 or headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS. 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 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 with psophometric weighting according to Recommendation ITU-T O.41 [27] in dBm0p.

Spectral peaks are measured in the frequency domain in the frequency range from 100 Hz to 3,4 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

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 -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.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The A-weighted noise level shall be measured at DRP of the artificial ear with the diffuse field equalization active. The noise level is measured until 10 kHz.

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 female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The activation signal level shall be -16 dBm0.

Spectral peaks are measured in the frequency domain in the frequency range from 100 Hz to 3,4 kHz. The frequency spectrum of the idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE: Care should be taken that only the noise is windowed out by the analysis and the analysis is not impaired by any remaining reverberance or room noise.

7.5.3.8 Acoustic shock

7.5.3.8.0 General

In order to fulfil the acoustic shock requirements, it is recommended to follow the guidelines of Recommendation ITU-T P.360 [57]. If needed, the PP may have to implement some kind of Hardware limiters.

7.5.3.8.1 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

Handset or headset is positioned on HATS (see clause 6.10.3). Signal used and method of measurement are given in ETSI EG 202 518 [i.32].

7.5.3.8.2 Void

7.5.3.9 Delay

Requirement

If connected to an FP type other than type 3, 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 46 ms. This value includes the 5 ms delay of the reference FP looping back the digital signal towards the PP.

NOTE 1: Technically, this PP type can also be connected to an FP of type 3 (Fixed Part with VoIP interface, narrowband service) i.e. a VoIP interface. The roundtrip delay of a VoIP-terminal (PP+FP) is defined as the sum of send and receive delays. For a telecommunication connection, only the roundtrip delay can be experienced. For this reason, also the requirement for VoIP-terminals is given only for the PP+FP roundtrip delay.

NOTE 2: The derivation of the delay value can be found in clause F.3.

If connected to an FP type 3, the requirement given in clause 7.7.1 for the combined FP+PP roundtrip delay applies.

Measurement method

If connected to an FP type other than type 3, 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.

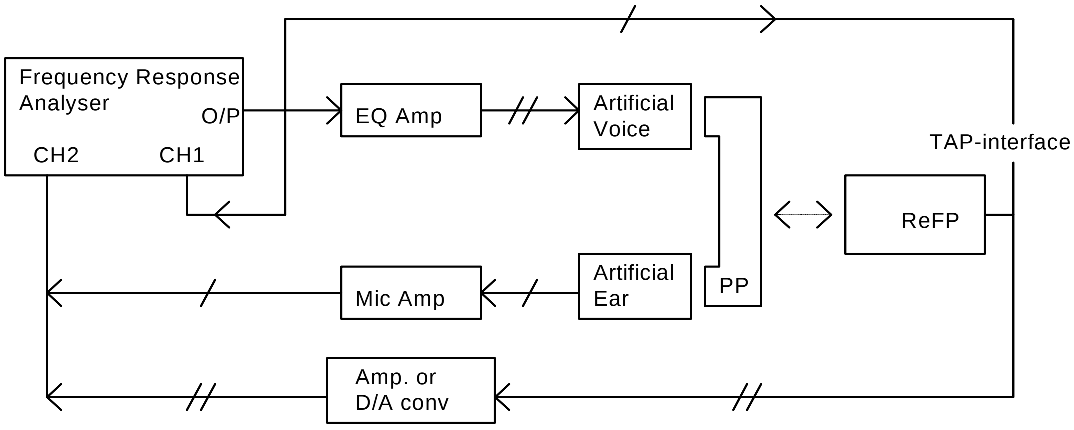


Figure 7.6a1: PP delay test configuration

The delay shall be measured by the cross-correlation method as described in annex C.

If connected to an FP type 3, the measurement method described in clause 7.7.1 applies.

7.5.3.10 Void

Table 7.22: Void

7.5.3.11 Double Talk Performance

7.5.3.11.0 General

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 Recommendations ITU-T 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 shall be classified according to table 7.23.

Table 7.23: Category regarding "duplex capability" depending on $A_{H,S,dt}$

Category (according to Recommendation ITU-T P.340 [39])	**1**	**2a**	**2b**	**2c**	**3**
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
AH,S,dt [dB]	≤ 3	≤ 6	≤ 9	≤ 12	> 12

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

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 handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is the double talk speech sequence as defined in clause 7.3.5 of Recommendation ITU-T P.501 [41] as shown in figure 7.6b. The competing speaker is always inserted as the double talk sequence sdt(t) in send direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

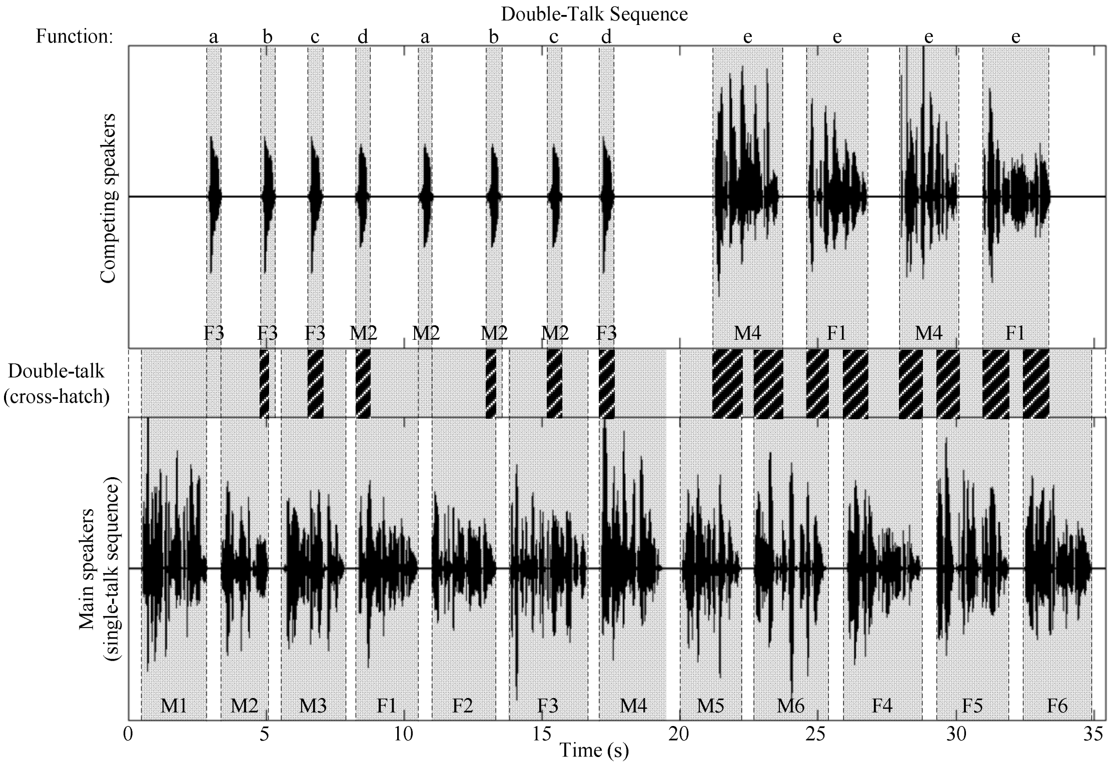


Figure 7.6b: Double Talk Test Sequence with overlapping speech sequences in sending and receiving direction

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequence of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

Table 7.23a: Void

7.5.3.11.2 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 shall be classified according to table 7.24.

Table 7.24: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.24 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 handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is shown in figure 7.6b. The competing speaker is always inserted as the double talk sequence in receive direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

Table 7.24a: Void

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 7.25 are applicable (more information can be found in annex A of Recommendation ITU-T P.340 [39]).

Table 7.25: Category regarding "duplex capability" depending on Echo Loss

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.25 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.

Measurement method

The handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

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. A detailed description can be found in Recommendation ITU-T P.501 [41].

The signals are fed simultaneously in send and receive direction. The level in send direction is -4,7 dBPa at the MRP (nominal level), the level in receive direction is -16 dBm0 at the electrical reference point (nominal level).

Figure 7.6c: Void

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

Send Direction		Receive Direction	
$f_0^{(1)}$ [Hz]	$\pm\Delta f^{(1)}$ [Hz]	$f_0^{(2)}$ [Hz]	$\pm\Delta f^{(2)}$[Hz]
250	±5	270	±5
500	±10	540	±10
750	±15	810	±15
1 000	±20	1 080	±20
1 250	±25	1 350	±25
1 500	±30	1 620	±30
1 750	±35	1 890	±35
2 000	±40	2 160	±35
2 250	±40	2 400	±35
2 500	±40	2 650	±35
2 750	±40	2 900	±35
3 000	±40	3 150	±35
3 250	±40	3 400	±35
3 500	±40	3 650	±35
3 750	±40	3 900	±35
NOTE: Parameters of the Shaping Filter: f ≥ 250 Hz: 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 Recommendation ITU-T 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

7.5.3.12.0 General

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) shall be ≤ 15 ms.

Measurement method

The handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The test signal is the "short words for activation" sequence described in clause 7.3.4 of Recommendation ITU-T P.501 [41] with increasing level for each single word.

Figure 7.6d: Void

The settings of the test signal are as follows.

Table 7.25b: Settings for test signals

	Single word Duration/ Pause Duration	Level of the first single word (at the MRP)	Level Difference between two Periods of the Test Signal
Single word to determine switching characteristic **in send direction**	~600 ms/ ~400 ms	-24 dBPa (see NOTE)	1 dB
NOTE: The signal level is determined for each utterance individually according to Recommendation ITU-T P.56 [33].			

It is assumed that the pause length of about 400 ms is longer than the hang-over time so that the test object is back to idle mode after each single word.

The level of the transmitted signal is measured at the electrical reference point. The test signal is filtered by the transfer function of the test object. The measured signal level is referred to the filtered test signal level and displayed versus time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the single word which indicates the first activation of the test object. The time between the beginning of the single word and the complete activation of the test object is measured.

7.5.3.12.2 Activation in Receiving Direction

For further study.

7.5.3.12.3 Silence Suppression and Comfort Noise Generation

Requirements and measurement methods are for further study.

NOTE: In general it is not recommended to use silence suppression at all.

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 according to Recommendation ITU-T O.41 [27] in dBm0p.

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 (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

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 in the standard position of the HATS.

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 female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] 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.

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

Speech Quality for narrowband systems can be tested based on ETSI EG 202 396-3 [i.34]. The test method is applicable for narrowband (100 Hz to 4 kHz) and wideband (100 Hz to 8 kHz) transmission systems. LQOn is used for narrowband systems.

Requirement

For the background noises defined in clause 6.10.6 the following requirements shall apply:

- N-MOS-LQOn ≥ 3,5.
- S-MOS-LQOn ≥ 3,5.
- G-MOS-LQOn ≥ 3,5.

NOTE: It is recommended to test the terminal performance with other types of background noises if the terminal is likely to be exposed to other noises than specified in clause 6.10.6.

Measurement method

The background noise simulation as described in clause 6.10.6 is used.

The handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The background noise should be applied for at least 5 s in order to adapt noise reduction algorithms in advance of the test.

The near end speech signal consists of 8 sentences of speech (2 male and 2 female talkers, 2 sentences each). Appropriate speech samples can be found in Recommendation ITU-T P.501 [41]. The preferred language is English since the objective method was validated with English language in narrowband. The test signal level is -1,7 dBPa at the MRP.

Three signals are required for the tests:

1) The clean speech signal is used as the undisturbed reference (see ETSI EG 202 396-3 [i.34]).

2) The speech plus undisturbed background noise signal is recorded at the terminal's microphone position using an omnidirectional measurement microphone with a linear frequency response between 50 Hz and 6 kHz.

3) The send signal is recorded at the electrical reference point.

N-MOS-LQOn, S-MOS LQOn and G-MOS LQOn are calculated per sentence as described in ETSI EG 202 396-3 [i.34].

The resulting N-MOS-LQOn-, S-MOS-LQOn- and G-MOS-LQOn-values are averaged over all 8 sentences.

7.5.3.12.6 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out applying a speech signal in the receive direction and by comparing the noise level transmitted in the send direction under reference conditions with no far end speech, to the noise level transmitted in the send direction under test conditions including far end speech. During and after the end of the speech signal the signal level in send direction shall 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 clause 6.10.6.

NOTE: The intention of this measurement is to detect impairments (modulations, switching and others) influencing the background noise transmitted from the terminal under test when a signal from the distant end (receiving side of the terminal under test) is present. Under these test conditions no modulation of the transmitted signal should occur. Modulation, switching or other type of impairments might be caused by an improper behaviour of a nonlinear processor working in conjunction with the echo canceller and erroneously switching or modulating the transmitted background noise.

Measurement method

The test arrangement is according to clause 6.10.3. The handset is mounted in the standard position of the HATS.

The background noises are generated as described in clause 6.10.6.

First the reference 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.

In a second step the same measurement is conducted but with inserting the speech signal at the far end. The exactly identical background noise signal is applied. The background noise signal shall 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 series of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] is applied in receive direction with duration of at least 10 s. The test signal level is -16 dBm0 at the electrical reference point.

For both reference and test conditions, the sending signal is recorded at the electrical reference point and the 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 speech 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 Void

7.5.3.12.8 Positional Robustness of Speech Quality in the Presence of Background Noise

Requirement

The degradation between standard position (STD) and all other modified positions for sending direction shall not exceed the limits for S-MOS and N-MOS according to table 7.26a. The requirements are evaluated on the averaged results over all background noises used in this test.

Table 7.26a: Requirements for allowed degradation

Position	∆ S-MOS	∆ N-MOS
UP	≤ 0,2	≤ 0,2
DOWN	≤ 0,3	≤ 0,5
AWAY	≤ 0,3	≤ 0,4

Measurement method

The test arrangement and measurement is identical to clause 7.5.3.12.5, with the restriction that only the background noises *Roadnoise* and *Pub* are evaluated. The test is conducted with each of the modified handset positions for sending direction according to table 6.3. All S- and N-MOS values as well as the difference to STD shall be reported for all three positions.

7.5.3.13 Quality of echo cancellation

7.5.3.13.0 General

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 shall not decrease by more than 6 dB from the maximum echo attenuation measured.

Measurement method

The test arrangement is according to clause 6.10.3. The handset is mounted in the standard position of the HATS.

The test signal consists of periodically repeated Composite Source Signal according to Recommendation ITU-T 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.

The difference between the maximum attenuation and the minimum attenuation is measured.

NOTE 1: In addition tests with more speech like signals should be made, e.g. Recommendation ITU-T P.501 [41] to see time variant behavior of EC. However, for such tests the simple broadband attenuation based test principle as described above cannot be applied due to the time varying spectral content of the speech like signals.

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 (35 ms) of the level analysis taking into account the exponential character of the integration time in any tolerance scheme.

NOTE 3: Care should be taken not to confuse noise or comfort noise with residual echo. In cases of doubt the measured echo signal should be compared to the residual noise signal measured under the same conditions without inserting the receive signal. If the level vs. time analysis leads to the identical result it can be assumed that no echo but just comfort noise is present.

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 (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

The test arrangement is according to clause 6.10.3. The handset is mounted in the standard position of the HATS.

Before the actual measurement a training sequence consisting of the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is fed. The level of the training sequence is -16 dBm0.

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The measurement is carried out under steady-state conditions. The average test signal level is -16 dBm0, averaged over the complete test signal. 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.

7.5.3.13.3 Variable echo path

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk with dynamically changing echo paths.

Requirement

No level peak shall be more than 10 dB above the minimum noise level during the measurement.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.7.

As test signal the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is used. The signal level shall be -10 dBm0. The terminal volume control is set to nominal RLR. The first 4 sentences of the test signal are used to allow full convergence of the echo canceller. The next 4 sentences (from 10,75 s to 22,5 s) are used for the analysis. The echo signal level is analysed over time. The echo signal level is analysed for 11,75 s, using a time constant of 35 ms.

The measurement result is displayed as echo level versus time.

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	Upper limit	Lower limit
100 Hz	0 dB	
315 Hz	0 dB	-14 dB
400 Hz	0 dB	-13 dB
500 Hz	0 dB	-12 dB
630 Hz	0 dB	-11 dB
800 Hz	0 dB	-10 dB
1 000 Hz	0 dB	-8 dB
1 300 Hz	2 dB	-8 dB
1 600 Hz	3 dB	-8 dB
2 000 Hz	4 dB	-8 dB
3 100 Hz	4 dB	-8 dB
4 000 Hz	0 dB	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.		

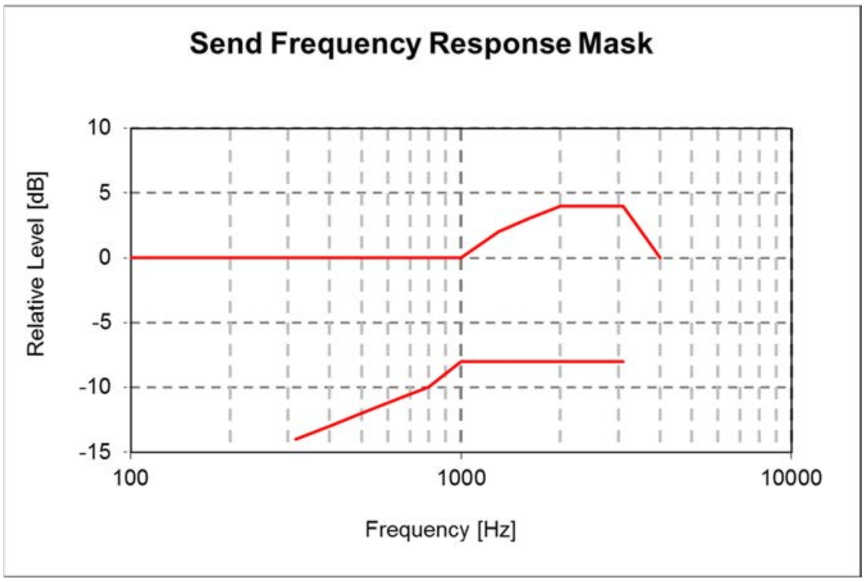


Figure 7.7: Sending sensitivity/frequency mask for HFT

Measurement method

The terminal is set-up as described in clause 6.10.4.

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The spectrum of acoustic signal produced by the artificial mouth is calibrated under freefield conditions at the MRP. The signal level is adjusted according to clause 6.10.4.2.

Measurements shall be made at one third-octave bands as given by the IEC 61260-1 [45] 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 dBV/Pa.

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	Upper limit	Lower limit
100 Hz	6 dB	
315 Hz	6 dB	-9 dB
400 Hz	6 dB	-6 dB
3 150 Hz	6 dB	-6 dB
4 000 Hz	6 dB	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.		

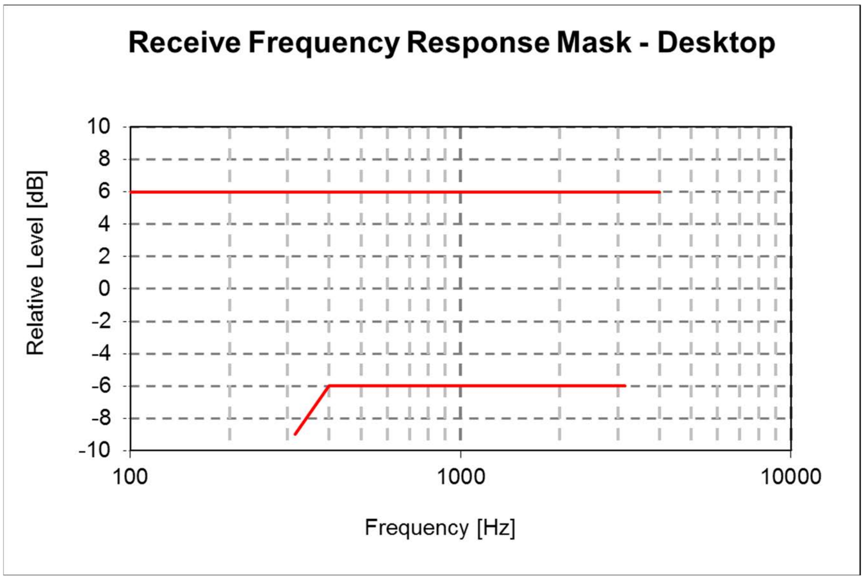


Figure 7.8: Receiving sensitivity/frequency mask for Desktop hands-free PP

- Handheld operated PP:

Table 7.30: Receiving frequency response handheld handsfree PP

Frequency	Upper limit	Lower limit
100 Hz	6 dB	
500 Hz	6 dB	-9 dB
630 Hz	6 dB	-6 dB
3 150 Hz	6 dB	-6 dB
4 000 Hz	6 dB	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.		

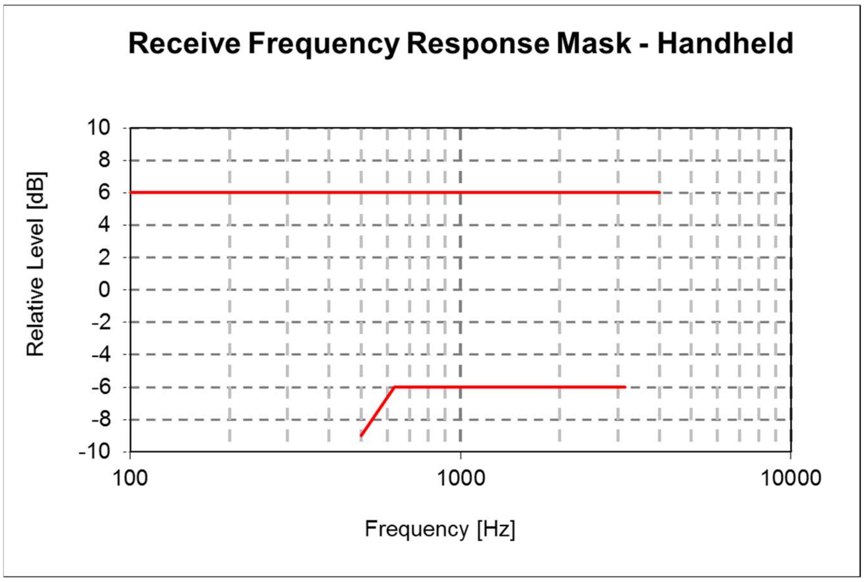


Figure 7.9: Receiving sensitivity/frequency mask for Hand-held PP

Table 7.31: Void

Figure 7.10: Void

- *Softphone (computer-based terminals)*
  - Type 1 or softphone with external speakers: requirement as for desktop terminal.
  - Type 2 requirement as for handheld terminal.
- *Group audio terminal*
  - Same requirement as desktop terminals.

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).

$$S_{Jeff} = 20 \log (pe_{ff} / v_{RCV}) \text{ dB rel 1 Pa / V}$$

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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -20 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

The HATS is free field equalized as described in Recommendation ITU-T P.581 [43]. The equalized output signal is power-averaged on the total time of analysis. The 1/3 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one third-octave bands as given by the IEC 61260-1 [45] 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 Send loudness rating

Requirement

The value of SLR shall be 13 dB ± 3 dB.

This value is derived from Recommendation ITU-T P.310 [i.35]. According to Recommendation ITU-T 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 types of terminals (desktop, handheld, etc.). Difference in efficiency will be given by conditions for measurement.

Measurement method

The test setup is as described in clause 6.10.4.

For a correct activation of the system, the test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The spectrum of acoustic signal produced by the artificial mouth is calibrated under freefield 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.

The send sensitivity shall be calculated for each of the 14 frequency bands given in table 1 of Recommendation ITU-T 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 at the MRP.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to Recommendation ITU-T P.79 [37], formula A-23b, over bands 4 to 17, and the send weighting factors from Recommendation ITU-T P.79 [37], table 1.

7.5.4.4 Receive loudness rating

Requirement

- Desktop operated PP:
  - Nominal value of RLR = 5 ± 3 dB. This value shall be fulfilled for one position of volume range.
  - Value of RLR at upper part of volume range shall be less than (louder) or equal to -2 dB: RLR max ≤ -2 dB.
  - Range of volume control shall be equal to or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.
- Handheld operated PP:
  - Improved class:
    - Nominal value of RLR = 9 ± 3 dB. This value shall be fulfilled for one position of volume range.
    - Value of RLR at upper part of volume range shall be less than (louder) or equal to 5 dB: RLR max ≤ 5 dB.
    - Range of volume control shall be equal to 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 shall be less than (louder) or equal to 8 dB: RLR max ≤ +8 dB Recommended value is RLR max ≤ 6 dB.

- Range of volume control shall be equal to or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.

- Softphone (computer-based terminal):

  - Type 1 or softphone with external speakers: requirement as for desktop terminal.

  - Type 2 requirement as for handheld terminal.

- Group audio terminal:

  - Nominal value of RLR shall be 5 ± 3 dB. This value has to be fulfilled for one position of volume range.

  - Value of RLR at upper part of volume range shall be less than (louder) or equal to -6 dB: RLR max ≤ -6 dB.

  - Range of volume control shall be equal to or exceed 19 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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -20 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

The receiving sensitivity shall be calculated for each of the 14 frequency bands given in table 1 of Recommendation ITU-T 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 shall be calculated according to Recommendation ITU-T P.79 [37], formula A-23c, over bands 4 to 17, using the receiving weighting factors from table 1 of Recommendation ITU-T P.79 [37]; no leakage correction shall be applied.

For binaural measurements, the individual sensitivities for left and right ears are energetically summed up. The hands-free RLR based on this overall sensitivity is then calculated with a correction factor of -8 dB.

The test shall be repeated for maximum and minimum volume control setting.

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 is set-up as described in clause 6.10.4.

After the correct activation of the system, a sinewave signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz is applied. 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.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal shall 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

Desktop and Handheld Terminal:

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, receive for desktop terminal at nominal volume	Signal to distortion ratio limit, receive for handheld terminal at nominal volume	Signal to distortion ratio limit, receive for all terminals at maximum volume
315 Hz	26 dB		
400 Hz	30 dB		
500 Hz	30 dB	20 dB	
800 Hz	30 dB	30 dB	20 dB
1 kHz	30 dB	30 dB	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.			

Softphone (computer-based terminal):

Type 1 or softphone with external speakers: requirement as for desktop terminal.

Type 2 requirement as for handheld terminal.

Group audio terminal:

Same requirement as for desktop terminal.

Measurement method

Test setup is described in clause 6.10.4.

After the correct activation of the system, a sinewave signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz is applied at the digital interface. The duration of the sine-wave shall be less than 1 s. Appropriate signals for activation and signal combinations can be found in Recommendation ITU-T P.501 [41]. The sinusoidal signal level shall be -16 dBm0.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. Level of this activation signal shall be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 10 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 test setup is described in clause 6.10.4.

The female speaker of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal shall 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 -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

The test setup is described in clause 6.10.4.

Measurement is operated at nominal value of volume control.

After a correct activation of the system, a digitally simulated sinewave signal at frequencies of 500 Hz, 1 000 Hz, 2 000 Hz and 3 150 Hz is 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.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. Level of this activation signal shall be -16 dBm0.

7.5.4.9 Sending noise

Requirement

The sending noise level shall not exceed -64 dBm0p.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Requirement as for other tests is identical for all types of terminals.

NOTE: Softphones with cooling devices (fans) can produce a rather high level of noise, furthermore largely dependent of activity of system.

Measurement method

The test setup is described in clause 6.10.4.

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 female speaker of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal shall be -4,7 dBPa at the MRP. The activation signal level is averaged over the complete activation signal sequence.

The psophometric noise level at the output of the test setup is measured. The psophometric filter is described in Recommendation ITU-T O.41 [27]. 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 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.

Spectral peaks are measured in the frequency domain in the frequency range from 100 Hz to 3,4 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

7.5.4.10 Receiving noise

Requirement

The noise level measured until 10 kHz shall not exceed -54 dBPa(A) at **nominal setting of the volume control**.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

NOTE 1: For softphone fan noise should be avoided in order to fulfil this condition.

Measurement method

Test setup is described in clause 6.10.4.

The A-weighted noise level shall be measured at DRP of the artificial ear with the freefield equalization active. The noise level shall be measured until 10 kHz.

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 female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. Level of this activation signal shall be -16 dBm0.

The A-weighted noise level shall be measured at DRP of the artificial ear with the freefield equalization active. The level in any 1/3rd octave band, between 100 Hz and 10 kHz shall be measured at DRP of the artificial ear, including free-field equalization.

The noise shall be measured just after interrupting the activation signal.

Spectral peaks are measured in the frequency domain in the frequency range from 100 Hz to 3,4 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE 2: Care should be taken that only the noise is windowed out by the analysis and the analysis window is not impaired by any remaining reverberance or room noise.

7.5.4.11 Terminal Coupling Loss weighted (TCLw)

Requirement

- Improved class:
  - In order to meet the G.131 [i.19] talker echo objective requirements, the weighted terminal coupling loss during single talk (TCLwst) shall be greater than 55 dB when measured under free field conditions at nominal setting of volume control.

NOTE 1: Depending on the idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 dB.

- 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, the handset is positioned on HATS (right ear).

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The signal level shall be -10 dBm0.

The TCLw is calculated according to Recommendation ITU-T 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 first 17,0 s of the test signal (6 sentences) are discarded from the analysis to allow for convergence of the acoustic echo canceller. The analysis is performed over the remaining length of the test sequence (last 6 sentences).

The ambient noise level shall be < -64 dBPa(A).

NOTE 2: Care should be taken when measuring TCL_W so that the echo return is not to be masked by the residual noise or the comfort noise when implemented.

7.5.4.12 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. The attenuation shall exceed 6 dB for all frequencies and for all settings of volume control.

Measurement method

The setup for terminal for handsfree mode is identical as for TCLw.

For loudspeaking mode handset is placed at 50 cm beside terminal with transducers facing the table as in figure 7.10a.

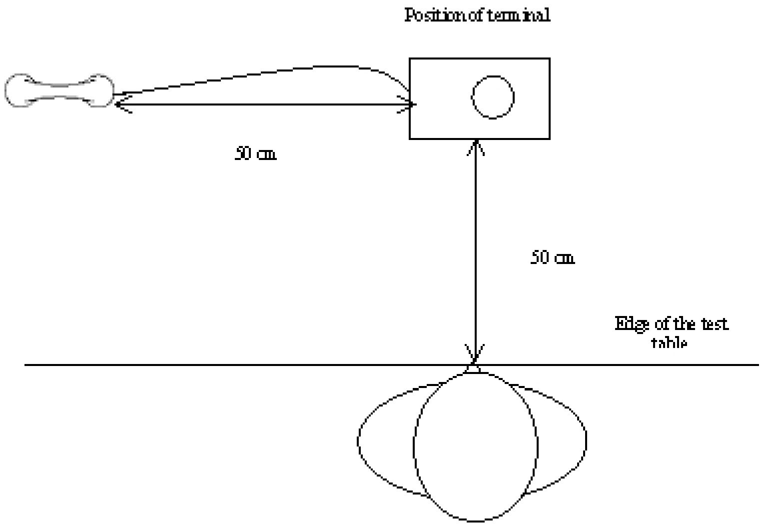


Figure 7.10a: Stability loss position for loudspeaking function

Before the actual test a training sequence consisting of the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41] shall be applied. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with Recommendation ITU-T 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.

7.5.4.13 Double Talk Performance

7.5.4.13.0 General

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 Recommendations ITU-T 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 shall be classified according to table 7.36.

Table 7.36: Category regarding "duplex capability" depending on $A_{H,S,dt}$

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.36 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.

This measurement shall be repeated for desktop type hands-free terminals and softphones with variable echo path.

Measurement method

Test setup is described in clause 6.10.4.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is the double talk speech sequence as defined in clause 7.3.5 of Recommendation ITU-T P.501 [41] as shown in figure 7.10b. The competing speaker is always inserted as the double talk sequence sdt(t) in send direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

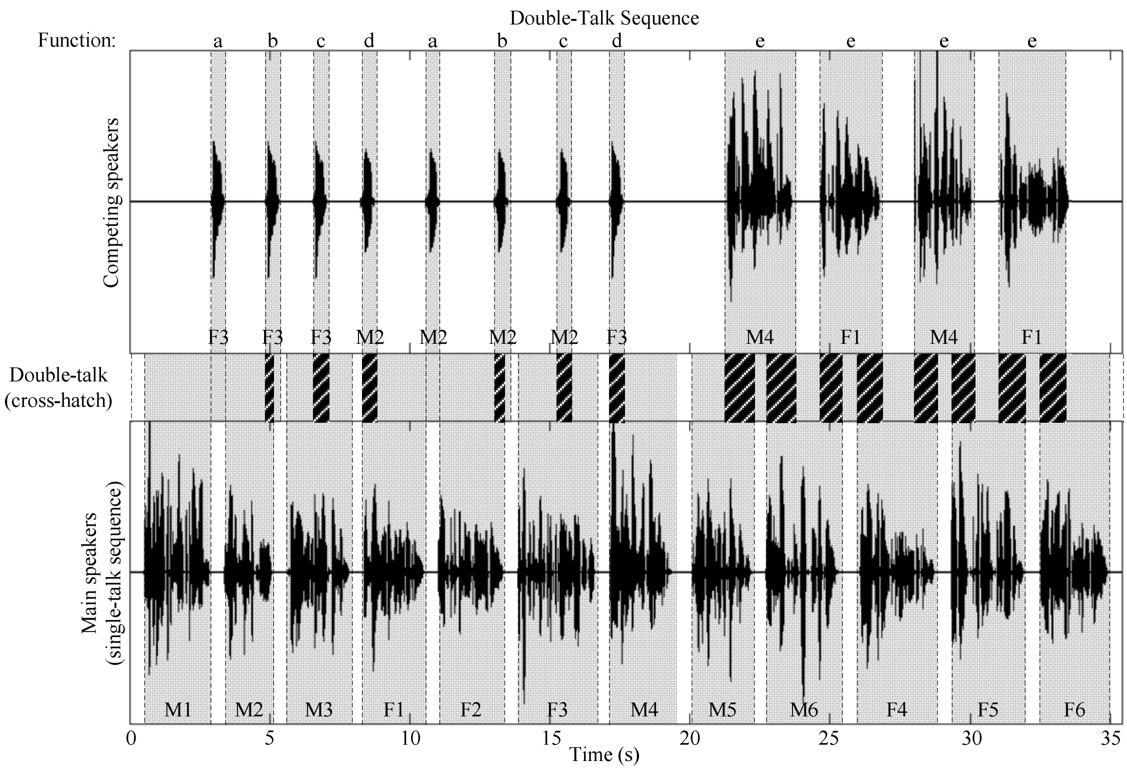


Figure 7.10b: Double Talk Test Sequence with overlapping speech sequences in sending and receiving direction

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

Table 7.36a: Void

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 shall be classified according to table 7.37.

Table 7.37: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.37 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.

Measurement method

Test setup is described in clause 6.10.4.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is shown in figure 7.10b. The competing speaker is always inserted as the double talk sequence in receive direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

Table 7.37a: Void

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 table 7.38 are applicable (more information can be found in annex A of the Recommendation ITU-T P.340 [39]).

Table 7.38: Category regarding "duplex capability" depending on Echo Loss

Category (according to Recommendation ITU-T 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).

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.38 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.

This measurement shall be repeated for desktop type hands-free terminals and softphones with variable echo path.

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. A detailed description can be found in Recommendation ITU-T 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: Void

The settings for the signals are as follows.

Table 7.38a: Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves

Send Direction		Receive Direction	
$f_0^{(1)}$ [Hz]	$\pm\Delta f^{(1)}$ [Hz]	$f_0^{(2)}$ [Hz]	$\pm\Delta f^{(2)}$[Hz]
250	±5	270	±5
500	±10	540	±10
750	±15	810	±15
1 000	±20	1 080	±20
1 250	±25	1 350	±25
1 500	±30	1 620	±30
1 750	±35	1 890	±35
2 000	±40	2 160	±35
2 250	±40	2 400	±35
2 500	±40	2 650	±35
2 750	±40	2 900	±35
3 000	±40	3 150	±35
3 250	±40	3 400	±35
3 500	±40	3 650	±35
3 750	±40	3 900	±35
NOTE: Parameters of the Shaping Filter: f ≥ 250 Hz: 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 Recommendation ITU-T 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

7.5.4.14.0 General

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) shall be ≤ 15 ms.

Measurement method

Test setup is described in clause 6.10.4.

The test signal is the "short words for activation" sequence described in clause 7.3.4 of Recommendation ITU-T P.501 [41] with increasing level for each single word.

Figure 7.10d: Void

The settings of the test signal are as follows.

Table 7.38b: Settings for test signals

	Single word/ Pause Duration	Level of the first single word (active Signal Part at the MRP)	Level Difference between two Periods of the Test Signal
Single word to Determine Switching Characteristic in Send Direction	~600 ms/ ~400 ms	-24 dBPa (see note)	1 dB
NOTE: The signal level is determined for each utterance individually according to Recommendation ITU-T P.56 [33].			

It is assumed that the pause length of about 400 ms is longer than the hang-over time so that the test object is back to idle mode after each single word.

The level of the transmitted signal is measured at the electrical reference point. The test signal is filtered by the transfer function of the test object. The measured signal level is referred to the filtered 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 single word which indicates the first activation of the test object. The time between the beginning of the single word and the complete activation of the test object is measured.

7.5.4.14.2 Activation in Receiving Direction

For further study.

7.5.4.14.3 Silence Suppression and Comfort Noise Generation

Requirements and measurement methods are for further study.

NOTE: In general it is not recommended to use silence suppression at all.

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 according to Recommendation ITU-T O.41 [27] in dBm0p.

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 receive direction consisting of an initial pause of 10 s and a periodical repetition of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] in receive 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.

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

Speech Quality for narrowband systems can be tested based on ETSI EG 202 396-3 [i.34]. The test method is applicable for narrowband (100 Hz to 4 kHz) and wideband (100 Hz to 8 kHz) transmission systems. LQOn is used for narrowband systems.

Requirement

For the background noises defined in clause 6.10.6, the following requirements shall apply:

- N-MOS-LQOn ≥ 3,0.
- S-MOS-LQOn ≥ 3,0.
- G-MOS-LQOn ≥ 3,0.

NOTE: It is recommended to test the terminal performance with other types of background noises if the terminal is likely to be exposed to other noises than specified in clause 6.10.6.

Measurement method

The background noise simulation as described in clause 6.10.6 is used. The terminal is setup as described in clause 6.10.4.

The background noise should be applied for at least 5 s in order to adapt noise reduction algorithms in advance of the test.

The near end speech signal consists of 8 sentences of speech (2 male and 2 female talkers, 2 sentences each). Appropriate speech samples can be found in Recommendation ITU-T P.501 [41]. The preferred language is English since the objective method was validated with English language in narrowband. The test signal level is +1,3 dBPa at the MRP.

Three signals are required for the tests:

1) The clean speech signal is used as the undisturbed reference (see ETSI EG 202 396-3 [i.34]).

2) The speech plus undisturbed background noise signal is recorded at the terminal's microphone position using an omnidirectional measurement microphone with a linear frequency response between 50 Hz and 6 kHz.

3) The send signal is recorded at the electrical reference point.

N-MOS-LQOn, S-MOS LQOn and G-MOS LQOn are calculated per sentence as described in ETSI EG 202 396-3 [i.34].

The resulting N-MOS-LQOn-, S-MOS-LQOn- and G-MOS-LQOn-values are averaged over all 8 sentences.

7.5.4.14.6 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out by applying a speech signal in receive direction and by comparing the noise level transmitted in the send direction under reference conditions with no far end speech, to the noise level transmitted in the send direction under test conditions including far end speech. During and after the end of the speech signal the signal level in Send direction shall 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 clause 6.10.6.

NOTE: The intention of this measurement is to detect impairments (modulations, switching and others) influencing the background noise transmitted from the terminal under test when a signal from the distant end (receiving side of the terminal under test) is present. Under these test conditions no modulation of the transmitted signal should occur. Modulation, switching or other type of impairments might be caused by an improper behaviour of a nonlinear processor working in conjunction with the echo canceller and erroneously switching or modulating the transmitted background noise.

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 reference 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.

In a second step the same measurement is conducted but with inserting the speech signal at the far end. The exactly identical background noise signal is applied. The background noise signal shall 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 series of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] is applied in receive direction with duration of at least 10 s. The test signal level is -16 dBm0 at the electrical reference point.

For both reference and test conditions the send signal is recorded at the electrical reference point and 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 speech 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.

7.5.4.15 Quality of echo cancellation

7.5.4.15.0 General

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 shall not decrease by more than 6 dB from the maximum measured echo attenuation.

Measurement method

Test setup is described in clause 6.10.4.

The test signal consists of periodically repeated Composite Source Signal according to Recommendation ITU-T 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.

The difference between the maximum attenuation and the minimum attenuation is measured.

NOTE 1: In addition tests with more speech like signals should be made, e.g. Recommendation ITU-T P.501 [41] to see time variant behavior of EC. However, for such tests the simple broadband attenuation based test principle as described above cannot be applied due to the time varying spectral content of the speech like signals.

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) taking into account the exponential character of the integration time in any tolerance scheme.

NOTE 3: Care should be taken not to confuse noise or comfort noise with residual echo. In cases of doubt the measured echo signal should be compared to the residual noise signal measured under the same conditions without inserting the receive signal. If the level vs. time analysis leads to the identical result it can be assumed that no echo but just comfort noise is present.

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 the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The level of the training sequence shall be -16 dBm0.

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The measurement is carried out under steady-state conditions. The average test signal level shall be -16 dBm0, averaged over the complete test signal. 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.

7.5.4.15.3 Variable echo path

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk with dynamic changing echo paths. No level peak shall be more than 10 dB above the minimum noise level during the measurement.

Measurement method

Test setup is described in clause 6.10.7.

As test signal the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is used. The signal level shall be -10 dBm0. The terminal volume control is set to nominal RLR. The first 4 sentences of the test signal are used to allow full convergence of the echo canceller. The next 4 sentences (from 10,75 s to 22,5 s) are used for the analysis. The echo signal level is analysed over time. The echo signal level is analysed for 11,75 s, using a time constant of 35 ms.

The measurement result is displayed as echo level versus time.

7.5.4.16 Microphone mute

Requirement

The SLR (Send Loudness Rating) with microphone mute on shall be at least 50 dB higher than with microphone mute off.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 7.5.4.3, but its microphone shall be configured to be muted.

Measurement and calculation method are the same as in clause 7.5.4.3.

7.5.4.17 Delay

Requirement:

If connected to an FP type other than type 3, 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 52 ms. This value includes the 5 ms delay of the reference FP looping back the digital signal towards the PP.

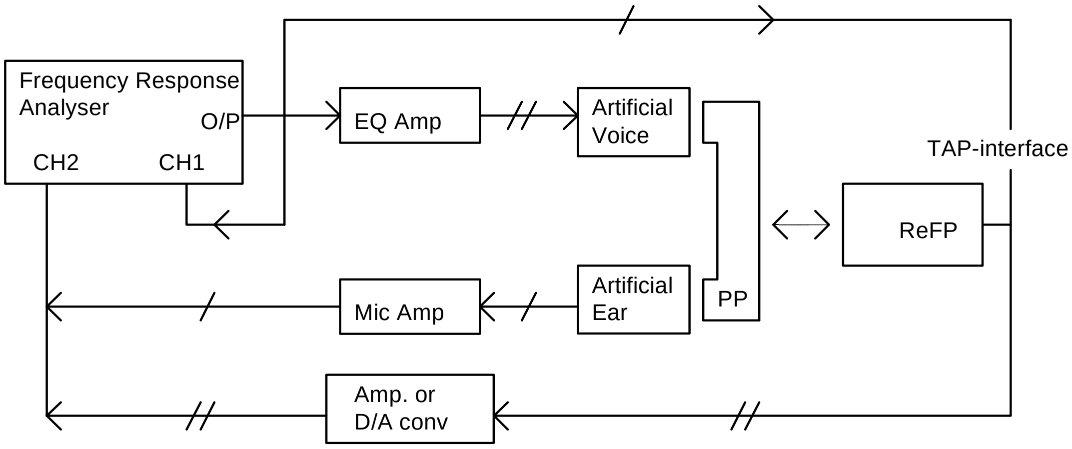
NOTE 1: Technically, this PP type can also be connected to an FP of type 3 (Fixed Part with VoIP interface, narrowband service) i.e. a VoIP interface. The roundtrip delay of a VoIP-terminal (PP+FP) is defined as the sum of send and receive delays. For a telecommunication connection, only the roundtrip delay can be experienced. For this reason, also the requirement for VoIP-terminals is given only for the PP+FP roundtrip delay.

NOTE 2: The derivation of the delay value can be found in clause F.3.

If connected to an FP type 3, the requirement given in clause 7.7.1 for the combined FP+PP roundtrip delay applies.

Measurement method

If connected to an FP type other than type 3, 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.



-/-- Receive direction (T_r);
-//- Send direction (T_s).

Figure 7.10e: PP delay test configuration

The delay shall be measured by the cross-correlation method as described in annex C.

If connected to an FP type 3, the measurement method described in clause 7.7.1 applies.

7.5.4.18 Send Loudness Level

Requirements:

The nominal value of Send Loudness Level (SLL) shall be:

SLL = 71 phon ± 4 phon

Measurement method:

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The terminal is set-up as described in clause 6.10.4 and calibration is realized as explained in clause 6.10.4.

The loudness (in sone) of the recorded signal is calculated according to Recommendation ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is calculated according to clause 9 of Recommendation ITU-T P.700 [74].

7.5.4.19 Receive Loudness Level

Requirements:

- *Desktop operated PP:*
  - The nominal value of Receive Loudness Level (RLL) shall be 71 ± 4 phon. This value shall be fulfilled for at least one volume setting.
  - The value of RLL at maximum volume setting shall be louder than or equal to 78 phon.

- The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 15 phon.

- Handheld operated PP:
  - Improved class:
    - The nominal value of Receive Loudness Level (RLL) shall be 67 ± 4 phon. This value has to be fulfilled for at least one volume setting.
    - The value of RLL at maximum volume setting shall be louder than or equal to 71 phon.
    - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 15 phon.
  - Standard class:
    - The nominal value of Receive Loudness Level (RLL) shall be 67 ± 4 phon. This value has to be fulfilled for at least one volume setting.
    - The value of RLL at maximum volume setting shall be louder than or equal to 68 phon.
    - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 15 phon.

- *Softphone (computer-based terminal):*
  - Type 1 or softphone with external speakers: requirement as for desktop terminal.
  - Type 2: requirement as for handheld terminal.

- Group audio terminal*:*
  - The nominal value of Receive Loudness Level (RLL) shall be 71 ± 4 phon. This value has to be fulfilled for at least one volume setting.
  - The value of RLL at maximum volume setting shall be louder than or equal to 82 phon.
  - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 19 phon.

Measurement method:

The terminal is set-up as described in clause 6.10.4.

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

For each recorded artificial ear signal, the loudness (in sone) of the recorded signal is calculated according to Recommendation ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is determined as follows for binaural measurements: The resulting loudnesses for left and right ears are first halved (individual loudness per ear). Both loudnesses are added (assuming perfect loudness summation). With this overall loudness, the overall loudness level is finally determined according to clause 8.2 of Recommendation ITU-T P.700 [74].

7.5.5 Transmission characteristics for PP type 2a (P.311 tested, wideband handset)

7.5.5.0 General

The requirements defined in this clause are based on Recommendation ITU-T 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

Requirement

See Recommendation ITU-T P.311 [38].

The tolerance for SLR shall be ±3,5 dB.

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.1.2 Sensitivity/frequency characteristics

Requirement

See Recommendation ITU-T P.311 [38].

NOTE: A gap, defined in annex G, for the lower mask limit, is allowed.

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.1.3 Noise

Requirement

See Recommendation ITU-T P.311 [38].

The limit for sending noise shall be -64 dBm0(A).

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.1.4 Distortion

Requirement

See Recommendation ITU-T P.311 [38].

The measurement shall be done in the level range from -10 dB to +5 dB re ARL.

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.1.5 Discrimination against out-of-band input signals

Requirement

See Recommendation ITU-T P.311 [38].

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.2 Receiving characteristics

7.5.5.2.1 Loudness rating

Requirement

See Recommendation ITU-T P.311 [38].

The tolerance for RLR shall be ±3,5 dB.

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.2.2 Sensitivity/frequency characteristics

Requirement

See Recommendation ITU-T P.311 [38].

NOTE: A gap, defined in annex G for the lower mask limit, is allowed.

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.2.3 Noise

Requirement

See Recommendation ITU-T P.311 [38].

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.2.4 Distortion

Requirement

See Recommendation ITU-T P.311 [38].

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	27,0 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 Recommendation ITU-T P.311 [38].

7.5.5.2.5 Spurious out-of-band receiving signals

Requirement

See Recommendation ITU-T P.311 [38].

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.3 Sidetone characteristics

7.5.5.3.1 Talker sidetone

Requirement

See Recommendation ITU-T P.311 [38].

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.3.2 Sidetone distortion

Requirement

See Recommendation ITU-T P.311 [38].

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.4 Echo path loss characteristics

7.5.5.4.1 Weighted terminal coupling loss

Requirement

See Recommendation ITU-T P.311 [38].

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.5.4.2 Stability loss

Requirement

See Recommendation ITU-T P.311 [38].

Measurement method

See Recommendation ITU-T P.311 [38].

7.5.6 Transmission characteristics for PP types 2b and 2c (HATS tested wideband handsets)

7.5.6.1 Frequency responses

7.5.6.1.1 Sending

7.5.6.1.1.1 Send frequency response - nominal position

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	

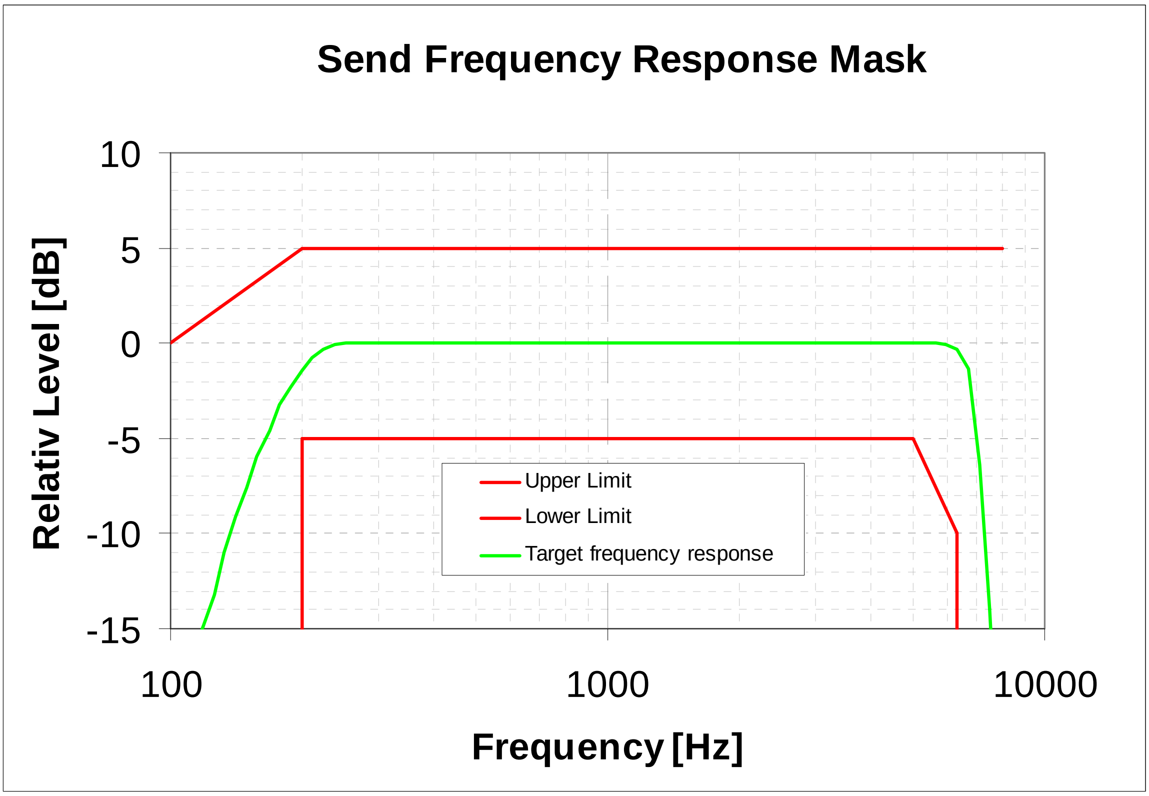


Figure 7.11: Send frequency response mask

NOTE 1: The basis for the target frequency responses in sending and receiving is the orthotelephonic reference response which is measured between two 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. The mask is a floating or "best fit" mask.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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 terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS. 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 Recommendation ITU-T P.380 [40] the results are averaged (averaged value in dB, for each frequency).

Measurements shall be made at one twelfth-octave bands as given by the IEC 61260-1 [45] 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.

7.5.6.1.1.2 Send frequency response - positional robustness

Requirement

For each of the modified handset positions, the send frequency response shall be within a given mask. The mask values per frequency are identical to table 7.42, except that an additional tolerance is provided for certain positions. Table 7.42a provides the offset in dB for the lower limit.

Table 7.42a: Tolerance mask offsets for send frequency response

Position	Offset Lower Limit
UP	-1 dB
DOWN	-2 dB
AWAY	-1 dB

Measurement method

The test arrangement and measurement is identical to clause 7.5.6.1.1.1. Instead of the standard handset position, the three modified positions according to table 6.3 for sending direction shall be used. The resulting three frequency responses shall be reported separately for each position.

7.5.6.1.2 Receiving

7.5.6.1.2.1 Receive frequency response - nominal position

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, 7.13 and 7.14. The application force for handsets is 2 N, 8 N and 13 N. This mask defined for 8 N application force shall be applicable for all types of headsets.

Table 7.43: Receive Frequency Response Mask

Frequency	Upper limit 8 N	Lower limit 8 N	Upper limit 2 N	Lower limit 2 N	Upper limit 13 N	Lower limit 13 N
100 Hz	3 dB		3 dB		6 dB	
120 Hz	3 dB	-5 dB	3 dB	-10 dB	6 dB	-5 dB
200 Hz	3 dB	-5 dB	3 dB	-8 dB	6 dB	-5 dB
400 Hz	3 dB	-5 dB	3 dB	-8 dB	6 dB	-5 dB
1 010 Hz	See note 1	-5 dB	See note 1	-8 dB	6 dB	-5 dB
1 200 Hz	See note 1	-8 dB	See note 1	-8 dB	6 dB	-8 dB
1 500 Hz	See note 1	-8 dB	See note 1	-8 dB	See note 1	-8 dB
2 000 Hz	9 dB	-3 dB	9 dB	-3 dB	9 dB	-3 dB
3 200 Hz	9 dB	-3 dB	9 dB	-3 dB	9 dB	-3 dB
7 000 Hz	9 dB	-13 dB	9 dB	-13 dB	9 dB	-13 dB
8 000 Hz	9 dB		9 dB		9 dB	

NOTE 1: 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. The mask is a floating or "best fit" mask.

NOTE 2: The basis for the target frequency responses in send and receive is the orthotelefonic reference response which is measured between 2 subjects in 1 m distance under free field conditions and is assuming an ideal receive characteristic. This flat response characteristics is shown as the target curve. 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 receive 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 send and a diffuse field based receive frequency response.

NOTE 3: With current technology it may be difficult or even not possible to achieve the desired frequency response characteristics for handsets with 2 N application force.

NOTE 4: With current technology it may be difficult or even not possible to achieve the desired frequency response characteristics for headsets below 250 Hz.

NOTE 5: The basis for the frequency response mask requirements is a subjective experiment which is described in annex B of ETSI ES 202 739 [i.10]. It may be difficult to be compliant with both this frequency response mask and the current frequency response mask as defined in TIA-920.130-A [i.36].

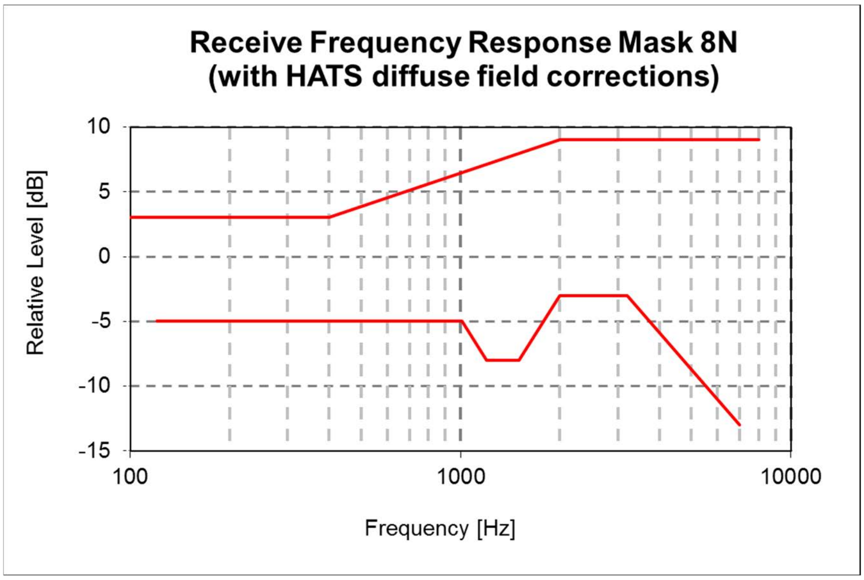


Figure 7.12: Receive frequency response mask for 8N application force

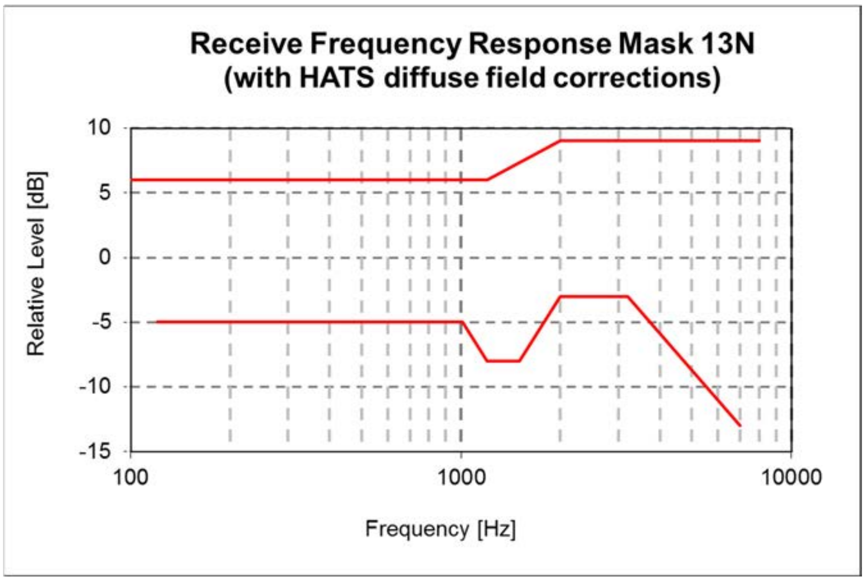


Figure 7.13: Receive frequency response mask for 13N application force

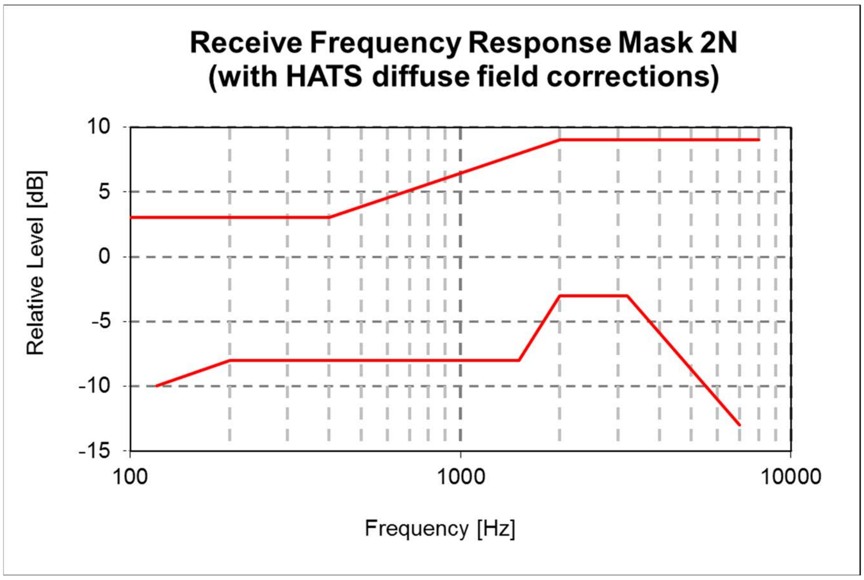


Figure 7.14: Receive frequency response mask for 2 N application force

NOTE 1: A gap, defined in annex G, for the lower mask limit, is allowed.

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.

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. The mask 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).

$$S_{Jedf} = 20 \log (pe_{df} / v_{RCV}) \text{ dB rel 1 Pa / V}$$

S_{Jedf} Receive Sensitivity; Junction to HATS Ear with diffuse field correction.

pe_{df} 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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -16 dBm0, measured according to Recommendation ITU-T 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 standard position of the HATS. The application forces used to apply the handset against the artificial ear is 2 N, 8 N and 13 N.

In case of headset measurements the tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [40] the results are averaged (averaged value in dB, for each frequency).

The HATS is diffuse field equalized as described in Recommendation ITU-T P.581 [43]. The diffuse field correction as defined in Recommendation ITU-T P.58 [35] is applied. 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 bands as given in IEC 61260-1 [45] 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.

7.5.6.1.2.2 Receive frequency response - positional robustness

Requirement

For each of the modified handset positions, the receive frequency response shall be within a given mask. The mask values per frequency are identical to table 7.43, except that an additional tolerance is provided for certain positions. Table 7.43a provides the offset in dB for the lower limit.

Table 7.43a: Tolerance mask offsets for receive frequency response

Position	Offset Lower Limit
Ye_{-5} Ze_{-5}	-1 dB
Ye_{0} Ze_{+5}	-1 dB
Ye_{+5} Ze_{-5}	-1 dB

Measurement method

The test arrangement and measurement is identical to clause 7.5.6.1.2.1. Instead of the standard handset position, the three modified positions according to table 6.4 for receiving direction shall be used. The resulting three frequency responses shall be reported separately for each position.

7.5.6.2 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:

$$SLR(set) = 8 \text{ dB} \pm 3 \text{ dB}.$$

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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 set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS. 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 Recommendation ITU-T P.380 [40] the results are averaged (averaged value in dB, for each frequency).

The send sensitivity shall be calculated from each of the 20 frequency bands given in table A.2 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], formula A-23b, over bands 1 to 20 and the send weighting factors from Recommendation ITU-T P.79 [37], Annex A, table A.2.

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.

The nominal value of RLR is the RLR closest to the nominal requirement.

The minimum difference between nominal RLR and minimum (loudest, maximum volume setting) RLR shall be higher than 6 dB.

NOTE 1: The mechanical design of some UEs may make it impossible to seal the ear-piece to the knife edge of the ITU-T artificial ear. Minimal additional methods may be used to provide the seal provided that they do not affect the mounting position of the UE with respect to the Mouth Reference Point and the Ear Reference Point.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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 standard position of the HATS. The HATS is *NOT* diffuse field equalized as described in Recommendation ITU-T P.581 [43]. The DRP-ERP correction as defined in Recommendation ITU-T 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, 8 N will be used.

In case of headset measurements the tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [40] the results are averaged (averaged value in dB, for each frequency).

The receive sensitivity shall be calculated for each of the 20 frequency bands given in table A.2 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], formula A-23c, over bands 1 to 20 and the receive weighting factors from table A.2 of Recommendation ITU-T P.79 [37]. No leakage correction shall be applied for the measurement.

NOTE 2: Currently the Loudness Ratings Calculation is still based on the ERP. Therefore no diffuse field correction is applied and still the DRP-ERP correction is used.

7.5.6.2.2 Void

Table 7.44: Void

Table 7.45: Void

7.5.6.2.3 Void

7.5.6.2.4 Microphone mute

Requirement

The SLR (Send Loudness Rating) with microphone mute on shall be at least 50 dB higher than with microphone mute off.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 7.5.6.2.1.1, but its microphone shall be configured to be muted.

Measurement and calculation method are the same as in clause 7.5.6.2.1.1.

7.5.6.2.5 Positional robustness

7.5.6.2.5.1 Send Loudness Rating

Requirement

The difference (in dB) between the SLR measured in each of the three modified handset positions and the one in determined standard position (STD) shall be in the range of -3 to +3 dB.

Measurement method

In addition to the test setup and measurement of clause 7.5.6.2.1.1, each of the three modified handset positions for sending direction according to table 6.3 shall be applied. SLR and delta-SLR values should be calculated and reported for each position.

7.5.6.2.5.2 Receive Loudness Rating

Requirement

The difference (in dB) between the RLR measured in each of the three modified handset positions and the one in standard position (STD) shall be in the range -3 to +3 dB.

Measurement method

In addition to the test setup and measurement of clause 7.5.6.2.1.2, each of the three modified handset positions for receiving direction according to table 6.4 shall be applied. An application force of 8 N is used. RLR and delta-RLR values should be calculated and reported for each position.

7.5.6.2.6 Send Loudness Level

Requirements

The nominal value of Send Loudness Level (SLL) shall be:

$$SLL = 75 \text{ phon} \pm 4 \text{ phon}$$

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The handset or headset terminal is set-up as described in clause 6.10.3.

The loudness (in sone) of the recorded signal is calculated according to Recommendation ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is calculated according to clause 9 of Recommendation ITU-T P.700 [74].

7.5.6.2.7 Receive Loudness Level

Requirements

The nominal value of Receive Loudness Level (RLL) for handsets, monaural and binaural/stereo headsets shall be:

RLL = 75 phon ± 4 phon

In case a user controlled receive volume control is provided, for at least one setting of the control the RLL shall meet the nominal value.

When the control is set to maximum, the RLL shall not be louder than 89 phon. With the volume control set to the minimum position the RLL shall not be quieter than 58 phon.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The handset or headset terminal is set-up as described in clause 6.10.3.

The loudness (in sone) of the recorded signal is calculated according to Recommendation ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is determined as follows:

- Handsets, monaural headsets: The loudness level is calculated according clause 8.2 of Recommendation ITU-T P.700 [74]by using the loudness value divided by two (loudness halving for monaural listening).
- Binaural headsets: The loudness level is calculated according clause 8.2 of Recommendation ITU-T P.700 [74] by using directly the loudness value (loudness summation for binaural listening is retained).

7.5.6.3 Sidetone

7.5.6.3.1 SideTone Masking Rating (STMR)

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 shall 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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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 standard position of the HATS and the application force shall be 13 N 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 bands as given in IEC 61260-1 [45] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band (Recommendation ITU-T 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 (LmeST), as expressed in dB, and the SideTone Masking Rating (STMR) (in dB) shall be calculated from the formula 5-1 of Recommendation ITU-T P.79 [37], using m = 0,225 and the weighting factors in table 3 of Recommendation ITU-T P.79 [37].

7.5.6.3.2 Void

7.5.6.3.3 Sidetone delay

Requirement

The maximum sidetone-round-trip delay shall be ≤ 5 ms, measured in an echo-free setup.

Measurement method

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The test signal is a CS-signal complying with Recommendation ITU-T 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 Recommendation ITU-T P.501 [41]. The level of the signal shall be -4,7 dBPa at the MRP.

The cross-correlation function $\Phi_{xy}(\tau)$ 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:

$$\Phi_{xy}(\tau) = \frac{1}{T}\int_{t=\frac{-T}{2}}^{\frac{T}{2}} S_x(t)\cdot S_y(t+\tau)\cdot dt$$

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 $\Phi_{xy}(\tau)$. 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 $\{\Phi_{xy}(\tau)\}$ of the cross-correlation:

$$H\{\Phi_{xy}(\tau)\} = \sum_{u=-\infty}^{\infty}\frac{\Phi_{xy}(u)}{\Pi(\tau-u)}$$

$$E(\tau) = \sqrt{\left[\Phi_{xy}(\tau)\right]^2 + \left[H\{\Phi_{xy}(\tau)\}\right]^2}$$

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 Terminal Coupling Loss (TCL)

Requirement

The TCL measured as unweighted Echo Loss shall be ≥ 55 dB at nominal setting of the volume control.

With the volume control set to maximum TCL shall be ≥ 46 dB.

It is recommended to set back the volume control to nominal level at the establishment of each new call, if TCL 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 standard position of the HATS and the application force shall be 2 N on the artificial ear type 3.3 or type 3.4 as specified in Recommendation ITU-T 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 the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The signal level shall be -10 dBm0.

TCL is calculated as difference between the averaged test signal level and the averaged echo level in the frequency range from 100 Hz to 8 000 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. The first 17,0 s of the test signal (6 sentences) are discarded from the analysis to allow for convergence of the acoustic echo canceller. The analysis is performed over the remaining length of the test sequence (last 6 sentences).

For the measurement, a time window has to be applied which is adapted to the duration of the actual test signal. The echo loss is calculated by the equations:

$$L_e = C - 10\log_{10}\sum_{i=1}^{N}(A_i + A_{i-1})(\log_{10} f_i - \log_{10} f_{i-1}) \qquad (7.1)$$

and

$$C = 10\log_{10}(2\,(\log_{10} f_N - \log_{10} f_0)) \qquad (7.2)$$

where:

- A_0 is the output/input power ratio at frequency f_0 = 100 Hz;
- A_i the ratio at frequency f_i; and
- A_N the ratio at frequency f_N = 8 000 Hz.

Equation (7.1) is a generalized form of the equation defined in Recommendation ITU-T G.122 [16], clause B.4, for calculating echo loss based on tabulated data, which allows the calculation of echo loss within any frequency range between f_0 and f_N.

NOTE 1: The extension of the frequency range is for further study.

NOTE 2: Care should be taken when measuring TCL: the echo return not to be masked by the residual noise or the comfort noise when implemented.

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 the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41] is applied. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with Recommendation ITU-T 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;

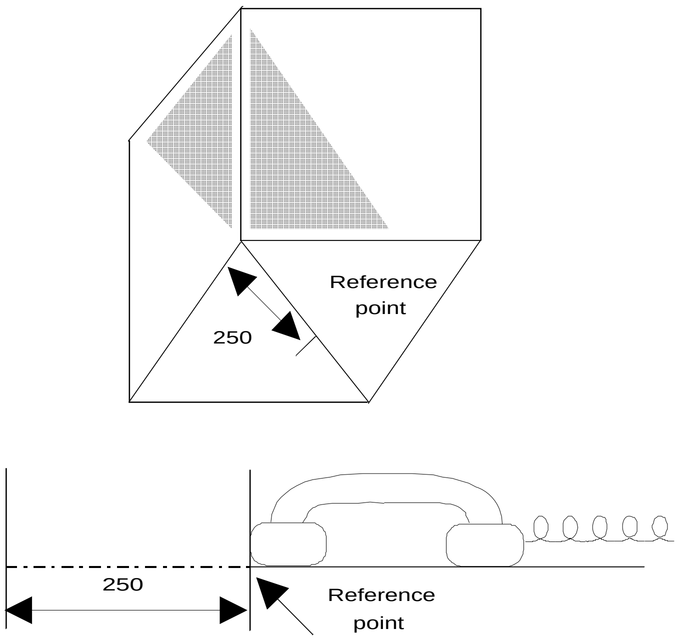
3) the extremity of the handset shall coincide with the normal to the reference point, as shown in figure 7.14a.

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) for monaural the headset receiver shall be placed centrally at the reference point as shown in figure 7.14a; for binaural headset, the receivers are placed symmetrically to the diagonal line on both sides of the reference point;

3) the headset microphone is positioned as close as possible to the receiver.



NOTE: All dimensions in mm.

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 handset terminal or headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

After the correct activation of the system, a sinewave signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000 Hz is applied. 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 7 kHz.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -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
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 handset terminal or the headset terminal is positioned as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

After a correct activation of the system, a digitally simulated sinewave signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz and 3 000 Hz shall be applied to the digital interface at the level of -16 dBm0.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal shall be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 10 kHz.

The ratio of signal to harmonic distortion shall be measured at the DRP of the artificial ear with the diffuse 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 the female speaker of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T 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 or headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS. 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).

Spectral peaks are measured in the frequency domain in the frequency range from 100 Hz to 6,3 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

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.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

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 female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall 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 diffuse field equalization active. The noise level is measured until 10 kHz.

Spectral peaks are measured in the frequency domain in the frequency range from 100 Hz to 6,3 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE: Care should be taken that only the noise is windowed out by the analysis and the analysis is not impaired by any remaining reverberance or room noise.

7.5.6.7 Acoustic shock

7.5.6.7.0 General

In order to fulfil the acoustic shock requirements, it is recommended to follow the guidelines of Recommendation ITU-T P.360 [57]. If needed, the PP may have to implement some kind of hardware limiter.

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 ETSI EG 202 518 [i.32].

7.5.6.8 Delay

Requirement

If connected to an FP other than type 5, 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 46 ms. This value includes the 5 ms delay of the reference FP looping back the digital signal towards the PP.

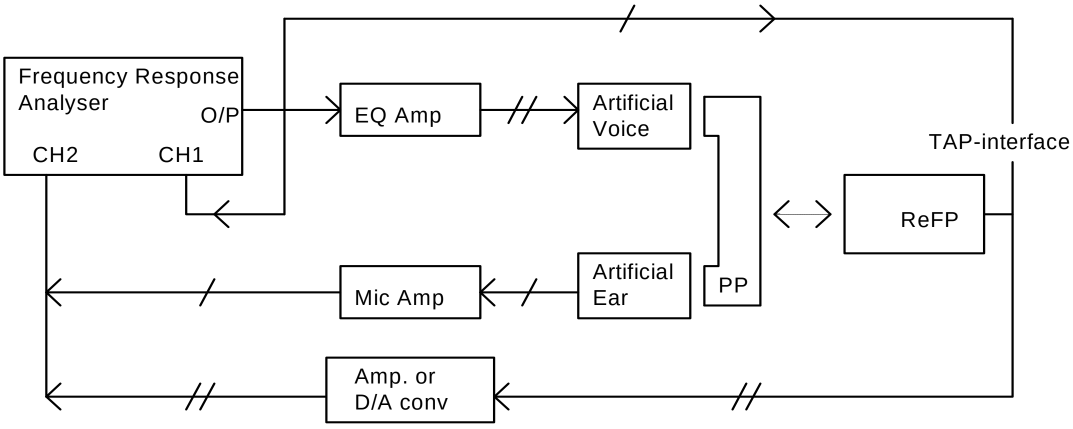
NOTE 1: Technically, this PP type can also be connected to an FP of type 5 (Fixed Part with VoIP interface, wideband, super-wideband, fullband, FBHR or ultra-band service) i.e. a VoIP interface. The roundtrip delay of a VoIP-terminal (PP+FP) is defined as the sum of send and receive delays. For a telecommunication connection, only the roundtrip delay can be experienced

NOTE 2: The derivation of the delay value can be found in clause F.3.

If connected to an FP type 5, the requirement given in clause 7.7.1 for the combined FP+PP roundtrip delay applies.

Measurement method

If connected to an FP other than type 5, 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.



-/-- Receive direction (T_r);
-//- Send direction (T_S).

Figure 7.14b: PP delay test configuration

The delay shall be measured by the cross-correlation method as described in annex C.

If connected to an FP type 5, the measurement method described in clause 7.7.1 applies.

7.5.6.9 Void

Table 7.48: Void

7.5.6.10 Double talk Performance

7.5.6.10.0 General

NOTE: Those parameters are optional, but are strongly recommended for improved class.

During double talk the speech is mainly determined by two 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 Recommendations ITU-T 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 shall be classified according to table 7.49.

Table 7.49: Category regarding "duplex capability" depending on $A_{H,S,dt}$

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.49 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 handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is the double talk speech sequence as defined in clause 7.3.5 of Recommendation ITU-T P.501 [41] as shown in figure 7.14c. The competing speaker is always inserted as the double talk sequence in send direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

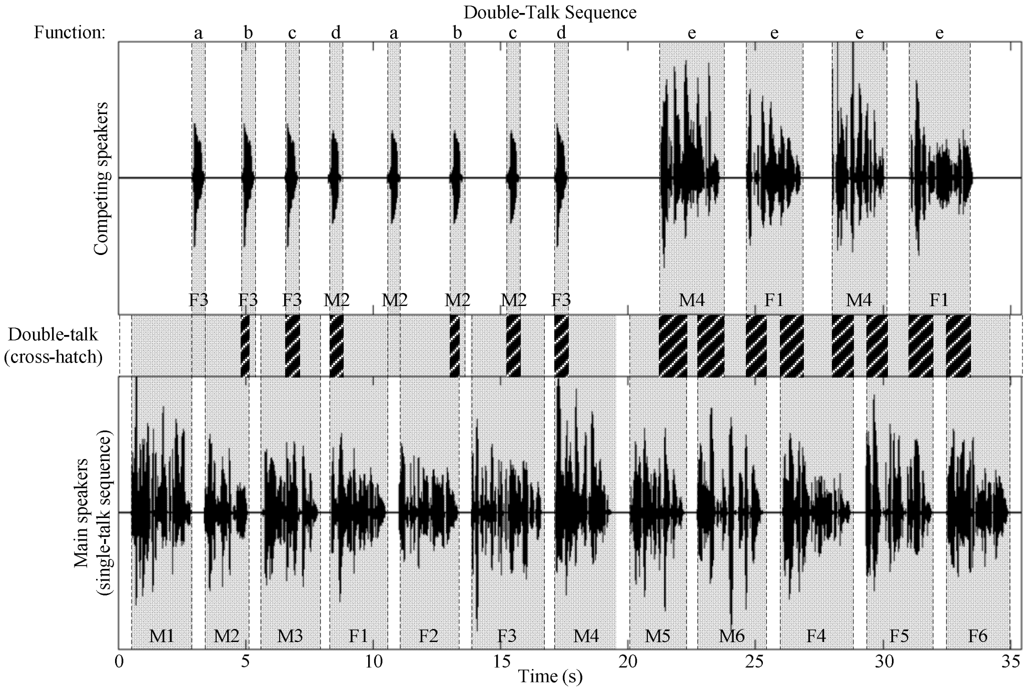


Figure 7.14c: Double Talk Test Sequence with overlapping speech sequences in sending and receiving direction

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

Table 7.49a: Void

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 shall be classified according to table 7.50.

Table 7.50: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

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 handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is shown in figure 7.14c. The competing speaker is always inserted as the double talk sequence in receive direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

Table 7.50a: Void

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 table 7.51 are applicable (more information can be found in annex A of Recommendation ITU-T P.340 [39]).

Table 7.51: Category regarding "duplex capability" depending on Echo Loss

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.51 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.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

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. A detailed description can be found in Recommendation ITU-T P.501 [41].

The signals are fed simultaneously in send and receive direction. The level in send direction is -4,7 dBPa at the MRP (nominal level), the level in receive direction is -16 dBm0 at the electrical reference point (nominal level).

Figure 7.14d: Void

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

Send Direction		Receive Direction	
$f_0^{(1)}$ [Hz]	$\pm\Delta f^{(1)}$ [Hz]	$f_0^{(2)}$ [Hz]	$\pm\Delta f^{(2)}$[Hz]
125	±2,5	180	±2,5
250	±5	270	±5
500	±10	540	±10
750	±15	810	±15
1 000	±20	1 080	±20
1 250	±25	1 350	±25
1 500	±30	1 620	±30
1 750	±35	1 890	±35
2 000	±40	2 160	±35
2 250	±40	2 400	±35
2 500	±40	2 650	±35
2 750	±40	2 900	±35
3 000	±40	3 150	±35
3 250	±40	3 400	±35
3 500	±40	3 650	±35
3 750	±40	3 900	±35
4 000	±40	4 150	±35
4 250	±40	4 400	±35
4 500	±40	4 650	±35
4 750	±40	4 900	±35
5 000	±40	5 150	±35
5 250	±40	5 400	±35
5 500	±40	5 650	±35
5 750	±40	5 900	±35
6 000	±40	6 150	±35
6 250	±40	6 400	±35
6 500	±40	6 650	±35
6 750	±40	6 900	±35
7 000	±40		
NOTE: Parameters of the Shaping Filter: f ≥ 250 Hz: 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 Recommendation ITU-T 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

7.5.6.11.0 General

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) shall be ≤ 15 ms.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The test signal is the "short words for activation" sequence described in clause 7.3.4 of Recommendation ITU-T P.501 [41] with increasing level for each single word.

Figure 7.14e: Void

The settings of the test signal are as follows.

Table 7.51b: Settings for test signals

	Single word duration / pause duration	Level of the first single word (at the MRP)	Level difference between two periods of the test signal
single word to determine switching characteristic in send direction	~600 ms / ~400 ms	-24 dBPa (see NOTE)	1 dB
NOTE: The signal level is determined for each utterance individually according to Recommendation ITU-T P.56 [33].			

It is assumed that the pause length of about 400 ms is longer than the hang-over time so that the test object is back to idle mode after each single word.

The level of the transmitted signal is measured at the electrical reference point. The test signal is filtered by the transfer function of the test object. The measured signal level is referred to the filtered test signal level and displayed versus time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the single word which indicates the first activation of the test object. The time between the beginning of the single word and the complete activation of the test object is measured.

7.5.6.11.2 Activation in Receiving Direction

For further study.

7.5.6.11.3 Silence Suppression and Comfort Noise Generation

Requirements and measurement methods are for further study.

NOTE: In general it is not recommended to use silence suppression at all.

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 A-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 (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
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 mounted in the standard position of the HATS.

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 female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] 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.

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

Speech Quality for wideband systems can be tested based on ETSI EG 202 396-3 [i.34]. The test method is applicable for narrowband (100 Hz to 4 kHz) and wideband (100 Hz to 8 kHz) transmission systems. LQOw is used for wideband systems.

Requirement

For the background noises defined in clause 6.10.6 the following requirements shall apply:

- N-MOS-LQOw ≥ 3,5.
- S-MOS-LQOw ≥ 3,5.
- G-MOS-LQOw ≥ 3,5.

NOTE: It is recommended to test the terminal performance with other types of background noises if the terminal is likely to be exposed to other noises than specified in clause 6.10.6.

Measurement method

The background noise simulation as described in clause 6.10.6 is used.

The handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The background noise should be applied for at least 5 s in order to adapt noise reduction algorithms in advance of the test.

The near end speech signal consists of 8 sentences of speech (2 male and 2 female talkers, 2 sentences each). Appropriate speech samples can be found in Recommendation ITU-T P.501 [41]. The preferred language is French since the objective method was validated with French language. The test signal level is -1,7 dBPa at the MRP.

Three signals are required for the tests:

1) The clean speech signal is used as the undisturbed reference (see ETSI EG 202 396-3 [i.34]).

2) The speech plus undisturbed background noise signal is recorded at the terminal's microphone position using an omnidirectional measurement microphone with a linear frequency response between 50 Hz and 12 kHz.

3) The send signal is recorded at the electrical reference point.

N-MOS-LQOw, S-MOS LQOw and G-MOS LQOw are calculated per sentence as described in ETSI EG 202 396-3 [i.34].

The resulting N-MOS-LQOw-, S-MOS-LQOw- and G-MOS-LQOw-values are averaged over all 8 sentences.

7.5.6.11.6 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out applying a speech signal in receive direction and comparing the noise level transmitted in the send direction under reference conditions with no far end speech, to the noise level transmitted in the send direction under test conditions including far end speech. During and after the end of the speech signal the signal level in send direction shall 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 clause 6.10.6.

NOTE: The intention of this measurement is to detect impairments (modulations, switching and others) influencing the background noise transmitted from the terminal under test when a signal from the distant end (receiving side of the terminal under test) is present. Under these test conditions no modulation of the transmitted signal should occur. Modulation, switching or other type of impairments might be caused by an improper behaviour of a nonlinear processor working in conjunction with the echo canceller and erroneously switching or modulating the transmitted background noise.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The background noises are generated as described in clause 6.10.6.

First the reference measurement is conducted without inserting the signal at the far end. At least 10 s of noise is analysed. The transmitted 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 speech signal at the far end. The exactly identical background noise signal is applied. The background noise signal shall start at the same point in time as was used for the reference measurement without far end signal. The background noise shall be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms. After at least 5 s a series of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] is applied in receive direction with a duration of at least 10 s. The test signal level in the receive direction is -16 dBm0 at the electrical reference point.

For both reference and test conditions, 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 speech 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 Void

7.5.6.11.8 Positional Robustness of Speech Quality in the Presence of Background Noise

Requirement

The degradation between standard position (STD) and all other modified positions for sending direction shall not exceed the limits for S-MOS and N-MOS according to table 7.52a. The requirements are evaluated on the averaged results over all background noises used in this test.

Table 7.52a: Requirements for allowed degradation

Position	Δ S-MOS	Δ N-MOS
UP	≤ 0,2	≤ 0,2
DOWN	≤ 0,3	≤ 0,5
AWAY	≤ 0,3	≤ 0,4

Measurement method

The test arrangement and measurement is identical to clause 7.5.6.11.5 with the restriction that only the background noises *Roadnoise* and *Pub* are evaluated. The test is conducted with each of the modified handset positions for sending direction according to table 6.3. All S- and N-MOS values as well as the difference to STD shall be reported for all three positions.

7.5.6.12 Quality of echo cancellation

7.5.6.12.0 General

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 shall not decrease by more than 6 dB from the maximum echo attenuation measured.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The test signal consists of periodically repeated Composite Source Signal according to Recommendation ITU-T 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.

The difference between the maximum attenuation and the minimum attenuation is measured.

NOTE 1: In addition, it is recommended to also conduct tests with more speech like signals, e.g. Recommendation ITU-T P.501 [41] in order to investigate time variant behaviour of EC. However, for such tests the simple broadband attenuation based test principle as described above cannot be applied due to the time varying spectral content of the speech like signals.

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) taking into account the exponential character of the integration time in any tolerance scheme.

NOTE 3: Care should be taken not to confuse noise or comfort noise with residual echo. In cases of doubt the measured echo signal should be compared to the residual noise signal measured under the same conditions without inserting the receive signal. If the level vs. time analysis leads to the identical result it can be assumed that no echo but just comfort noise is present.

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: Echo attenuation limits

Frequency	Limit
100 Hz	-41 dB
1 300 Hz	-41 dB
3 450 Hz	-46 dB
5 200 Hz	-46 dB
7 500 Hz	-37 dB
8 000 Hz	-37 dB
NOTE: 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 handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

Before the actual measurement a training sequence consisting of the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is fed in. The level of the training sequence shall be -16 dBm0.

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The measurement is carried out under steady-state conditions. The average test signal level shall be -16 dBm0, averaged over the complete test signal. 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.

7.5.6.12.3 Variable echo path

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk with dynamically changing echo paths. No level peak shall be more than 10 dB above the minimum noise level during the measurement.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.7.

As test signal the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is used. The signal level shall be -10 dBm0. The terminal volume control is set to nominal RLR. The first 4 sentences of the test signal are used to allow full convergence of the echo canceller. The next 4 sentences (from 10,75 s to 22,5 s) are used for the analysis. The echo signal level is analysed over time. The echo signal level is analysed for 11,75 s, using a time constant of 35 ms.

The measurement result is displayed as echo level versus time.

7.5.6.13 Out of band signals

7.5.6.13.1 Out-of-band signals in sending direction

Requirement

The level of any in-band image frequencies resulting from application of input signals at 8 kHz and above shall be attenuated by at least 25 dB compared to the output level of a 1 kHz input signal.

Measurement method

The terminal will be positioned as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal shall 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 and 10 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.

NOTE 1: The frequency range of artificial mouth according to Recommendation ITU-T P.58 [35] is specified up to 8 kHz. The production of out-of-band frequencies up to 10 kHz however is possible. So the out-of-band test is limited up to 10 kHz.

NOTE 2: Depending on the type of codec the test signal used may need to be adapted.

7.5.6.13.2 Out-of-band signals in receiving direction

Requirement

Any spurious out-of-band image signals in the frequency range from 9 kHz to 12 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 7.53a.

Table 7.53a

Frequency	Signal limit
9 kHz	50 dB
10 kHz	52 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 handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

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 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 10 kHz is measured selectively at measurement point.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -16 dBm0.

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.		

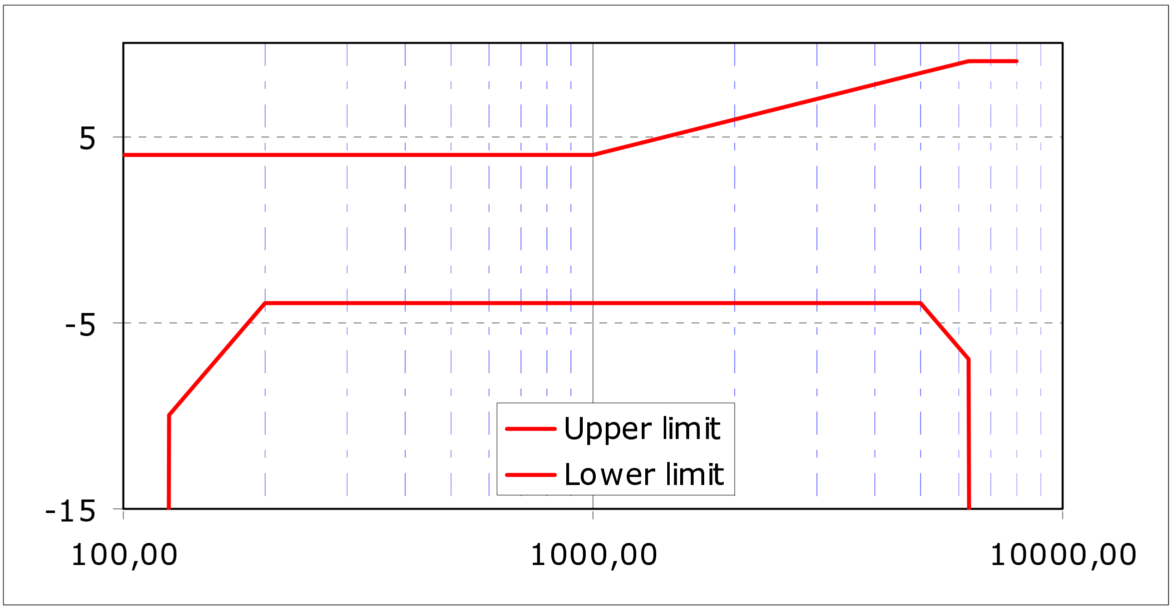


Figure 7.15: Sending sensitivity/frequency mask for HFT

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 test setup is as described in clause 6.10.4.

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The spectrum of acoustic signal produced by the artificial mouth is calibrated under freefield conditions at the MRP. The signal level is adjusted according to clause 6.10.4.

Measurements shall be made at one third-octave bands as given by the IEC 61260-1 [45] 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 dBV/Pa.

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	Upper limit	Lower limit
125 Hz	8 dB	
200 Hz	8 dB	-12 dB
250 Hz	8 dB	-9 dB
315 Hz	7 dB	-6 dB
400 Hz	6 dB	-6 dB
5 000 Hz	6 dB	-6 dB
6 300 Hz	6 dB	-9 dB
8 000 Hz	6 dB	

NOTE 1: Referring to ETSI 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 cases to introduce "bass boost" for amplification.

NOTE 2: A gap, defined in annex G, for the lower mask limit, is allowed.

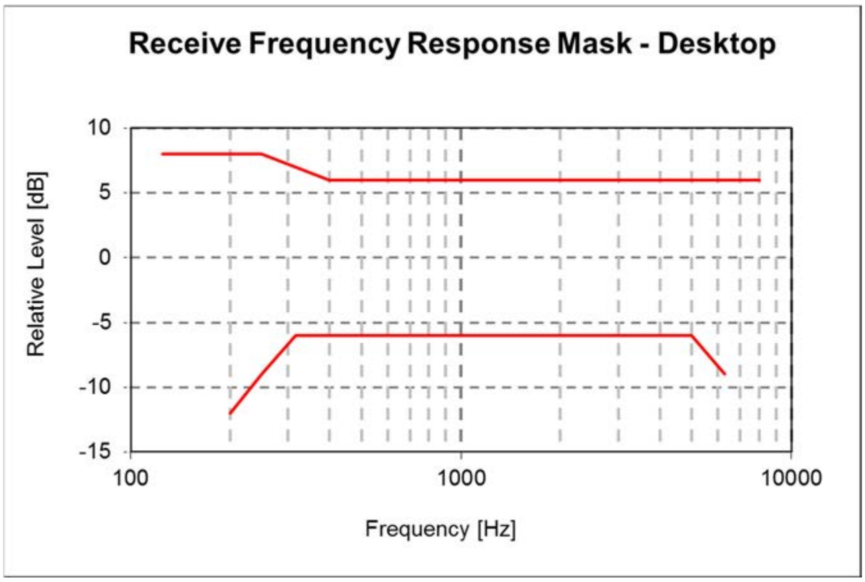


Figure 7.16: Receiving sensitivity/frequency mask for Desktop handsfree PP

- Handheld operated PP:

Table 7.56: Receiving frequency response handheld handsfree PP

Frequency	Upper limit	Lower limit
125 Hz	6 dB	
400 Hz	6 dB	-12 dB
500 Hz	6 dB	-6 dB
4 000 Hz	6 dB	-6 dB
5 000 Hz	6 dB	-9 dB
6 300 Hz	6 dB	-12 dB
8 000 Hz	6 dB	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale.		

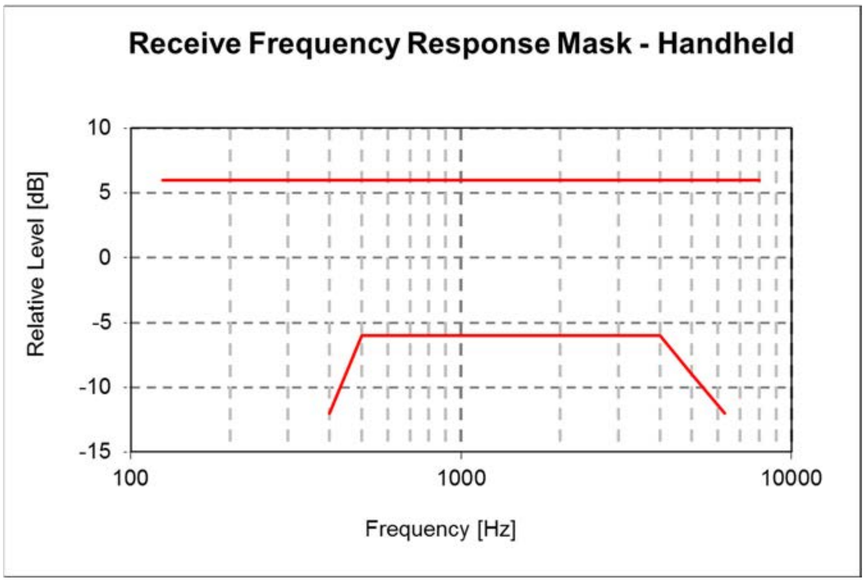


Figure 7.17: Receiving sensitivity/frequency mask for improved handheld PP

NOTE 3: At high frequencies, lower limit is relaxed. It is necessary to take into account that in most cases measurement will be made facing to the opposite side of output of loudspeaker.

NOTE 4: A gap, defined in annex G, for the lower mask limit, is allowed.

Table 7.57: Void

Figure 7.18: Void

- *Softphone (computer-based terminals):*
  - Type 1 or softphone with external speakers: requirement as for desktop terminal.
  - Type 2 requirement as for handheld terminal.
- *Group audio terminal:*
  - Same requirement as desktop terminals.

Measurement method

The test set-up 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)

$$S_{Jeff} = 20 \log (pe_{ff} / v_{RCV}) \text{ dB rel 1 Pa / V}$$

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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -20 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

The HATS is free field equalized as described in Recommendation ITU-T P.581 [43]. The equalized output signal is power-averaged on the total time of analysis. The 1/3 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one third-octave bands as given by the IEC 61260-1 [45] 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.

7.5.7.3 Send loudness rating

Requirement

The value of SLR shall be 13 dB ± 3 dB.

This value is derived from Recommendation ITU-T P.310 [i.35]. According to Recommendation ITU-T 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.

NOTE: Due to the lack of experience in the application of wideband loudness rating calculation as defined in annex G of Recommendation ITU-T P.79 [37] the loudness rating calculation as described in annex A is used.

Measurement method

The test setup is as described in clause 6.10.4.

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under freefield 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 for each of the 20 frequency bands given in table A.2 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], formula A-23b, over bands 1 to 20 and the send weighting factors from Recommendation ITU-T P.79 [37], Annex A, table A.2.

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 shall be less than (louder) or equal to -2 dB: RLR max ≤ -2 dB.
  - Range of volume control shall be equal to or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.
- Handheld operated PP:
  - Improved class:
    - Nominal value of RLR = 9 ± 3 dB. This value shall be fulfilled for one position of volume range.

    - Value of RLR at upper part of volume range shall be less than (louder) or equal to 5 dB: RLR max ≤ 5 dB.
    - Range of volume control shall be equal to or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.
  - Standard class:
    - Nominal value of RLR = 9 ± 3 dB. This value shall be fulfilled for one position of volume range.
    - Value of RLR at upper part of volume range shall be less than (louder) or equal to 8 dB: RLR max ≤ 8 dB.
    - Recommended value is RLR max ≤ 6 dB.
    - Range of volume control shall be equal to or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.

- Softphone (computer-based terminal):
  - Type 1 or softphone with external speakers: requirement as for desktop terminal.
  - Type 2 requirement as for handheld terminal.
- Group audio terminal:
  - Nominal value of RLR shall be 5 ± 3 dB. This value shall be fulfilled for one position of volume range.
  - Value of RLR at upper part of volume range shall be less than (louder) or equal to -6 dB: RLR max ≤ -6 dB.
  - Range of volume control shall be equal to or exceed 19 dB.

NOTE: Due to the lack of experience in the application of wideband loudness rating calculation as defined in annex G of Recommendation ITU-T P.79 [37] the loudness rating calculation as described in annex A is used.

Measurement method

The test setup is described in clause 6.10.4.

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -20 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

The receiving sensitivity shall be calculated for each of the 20 frequency bands given in table A.2 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], formula (A-23c), over bands 1 to 20 and the receive weighting factors from table A.2 of Recommendation ITU-T P.79 [37]. No leakage correction shall be applied.

The test shall be repeated for maximum and minimum volume control setting.

For binaural measurements, the individual sensitivities for left and right ears are energetically summed up. The hands-free RLR based on this overall sensitivity is then calculated with a correction factor of -8 dB.

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
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 PP will be positioned as described in clause 6.10.4.

After the correct activation of the system, a sinewave signal at frequencies of 200 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 kHz and 2 kHz is applied. 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 7 kHz.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal shall 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: Limits for harmonic distortion ratio (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	20 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	

  - Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.
- Softphone (computer-based terminal):
  - Type 1 or softphone with external speakers: requirement as for desktop terminal.
  - Type 2 requirement as for handheld terminal.
- Group audio terminal:
  - Same requirement as for desktop terminal.

Measurement method

The test setup is described in clause 6.10.4.

After a correct activation of the system, a sinewave signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 kHz, 2 kHz, 3 kHz is applied at the digital interface. The duration of the sine-wave shall be of less than 1 s. The sinusoidal signal level shall be -16 dBm0.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. Level of this activation signal shall be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 10 kHz.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.7.7 Out-of-band signals in sending direction

Requirement

The level of any in-band image frequencies resulting from application of input signals at 8 kHz and above shall be attenuated by at least 25 dB compared to the output level of a 1 kHz input signal.

Table 7.60: Void

Measurement method

The PP will be positioned as described in clause 6.10.4.

The female speaker of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal shall 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, 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.

NOTE: The frequency range of artificial mouth according to Recommendation ITU-T P.58 [35] is specified up to 8 kHz. The production of out-of-band frequencies up to 10 kHz however is possible. So the out-of-band test is limited up to 10 kHz.

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 12 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
10	52
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 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 10 kHz is measured selectively at measurement point.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. Level of this activation signal shall 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.

NOTE: Softphones with cooling devices (fans) can produce a rather high level of noise, furthermore largely dependent of activity of system.

Measurement method

The test setup is as described in clause 6.10.4.

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 the female speaker of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41]. The level of this activation signal shall be -4,7 dBPa, measured at the MRP. The activation signal level is averaged over the complete activation signal sequence.

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).

Spectral peaks are measured in the frequency domain in the frequency range from 100 Hz to 6,3 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

7.5.7.10 Receiving noise

Requirement

The noise level measured until 10 kHz shall not exceed -54 dBPa(A) at **nominal setting of the volume control**.

The level in any 1/3-octave band, between 100 Hz and 10 kHz shall not exceed a value of -64 dBPa.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

NOTE 1: For softphone fan noise should be avoided in order to fulfil this condition.

Measurement method

The test setup is described in clause 6.10.4.

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 the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41]. The level of this activation signal shall be -16 dBm0.

The A-weighted noise level shall be measured at DRP of the artificial ear with the freefield equalization active. The noise level is measured until 10 kHz.

The noise shall be measured just after interrupting the activation signal.

Spectral peaks are measured in the frequency domain in the frequency range from 100 Hz to 6,3 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3 octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE 2: Care should be taken that only the noise is windowed out by the analysis and the analysis window is not impaired by any remaining reverberance or room noise.

7.5.7.11 Terminal Coupling Loss

Requirement

- Improved class:
  - In order to meet the G.131 [i.19] talker echo objective requirements, the terminal coupling loss during single talk (TCLst) shall be greater than 55 dB when measured under free field conditions at **nominal setting of the volume control**.
  - For terminals fitted with a volume control the TCLst shall be not less than 40 dB for the higher gain settings above the nominal setting of the volume control.
- Standard class:
  - TCL shall be greater than 40 dB when measured under free field conditions at nominal setting of volume control.
  - TCLst 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).

The attenuation from electrical reference point input to electrical reference point output shall be measured using the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The signal level shall be -10 dBm0.

TCL is calculated as the difference between the averaged test signal level and the averaged echo level in the frequency range from 100 Hz to 8 kHz. 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 first 17,0 s of the test signal (6 sentences) are discarded from the analysis to allow for convergence of the acoustic echo canceller. The analysis is performed over the remaining length of the test sequence (last 6 sentences).

For the measurement, a time window has to be applied which is adapted to the duration of the actual test signal. The echo loss is calculated by the equations:

$$L_e = C - 10\log_{10}\sum_{i=1}^{N}(A_i + A_{i-1})(\log_{10} f_i - \log_{10} f_{i-1}) \qquad (7.3)$$

and

$$C = 10\log_{10}(2\,(\log_{10} f_N - \log_{10} f_0)) \qquad (7.4)$$

where:

- A_0 is the output/input power ratio at frequency f_0 = 100 Hz;
- A_i the ratio at frequency f_i; and
- A_N the ratio at frequency f_N = 8 000 Hz.

Equation (7.3) is a generalized form of the equation defined in Recommendation ITU-T G.122 [16], clause B.4, for calculating echo loss based on tabulated data, which allows the calculation of echo loss within any frequency range between f_0 and f_N.

The ambient noise level shall be < -64 dBPa(A).

NOTE 1: It should be relevant to perform the test in a real room instead of an anechoic room.

NOTE 2: The extension of the frequency range is for further study.

NOTE 3: Care should be taken when measuring TCL: the echo return not to be masked by the residual noise or the comfort noise when implemented.

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. The attenuation shall exceed 6 dB for all frequencies and for all settings of volume control.

Measurement method

For handsfree mode test setup is identical as for TCL.

For loudspeaking mode handset is placed at 50 cm beside terminal with transducers facing the table (see figure 7.18a).

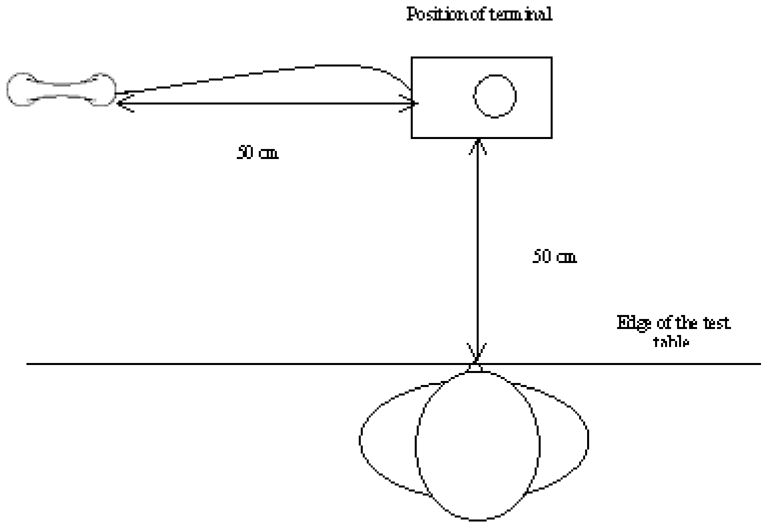


Figure 7.18a: Stability loss position for loudspeaking function

Before the actual test a training sequence consisting of the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41] shall be applied. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with Recommendation ITU-T 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.

7.5.7.13 Double Talk Performance

7.5.7.13.0 General

NOTE: When those parameters are optional, they are strongly recommended for improved class.

During double talk the speech is mainly determined by two 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 Recommendations ITU-T 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 shall be classified according to table 7.62.

Table 7.62: Category regarding "duplex capability" depending on $A_{H,S,dt}$

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.62 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.

This measurement shall be repeated for the desktop type terminals and softphones with variable echo path.

Measurement method

The test setup is described in clause 6.10.4.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is the double talk speech sequence as defined in clause 7.3.5 of Recommendation ITU-T P.501 [41] as shown in figure 7.18b. The competing speaker is always inserted as the double talk sequence sdt(t) in send direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

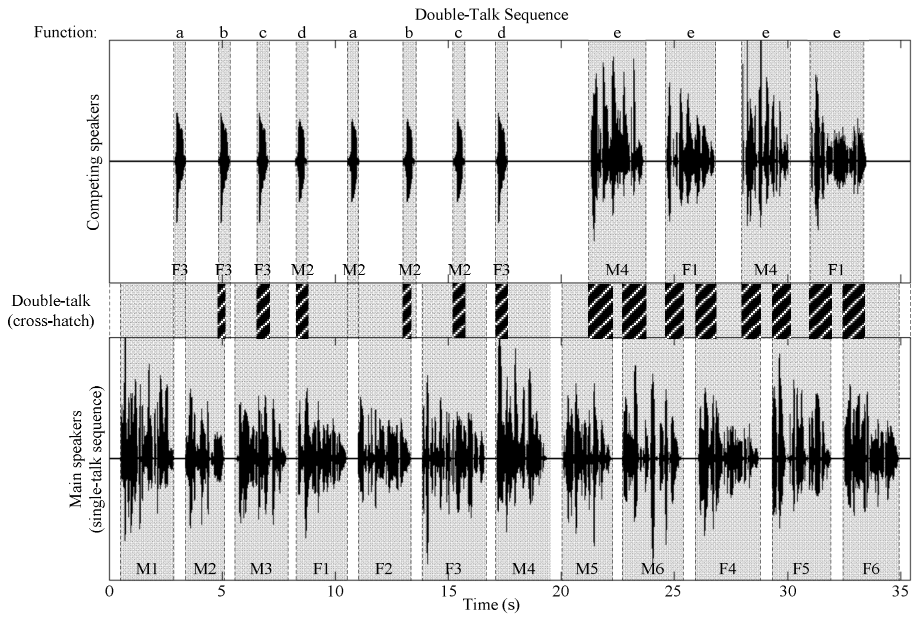


Figure 7.18b: Double Talk Test Sequence with overlapping speech sequences in sending and receiving direction

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

Table 7.62a: Void

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 shall be classified according to table 7.63.

Table 7.63: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.63 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.

Measurement method

The test setup is described in clause 6.10.4.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is shown in figure 7.18b. The competing speaker is always inserted as the double talk sequence in receive direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

Table 7.63a: Void

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 Recommendation ITU-T P.340 [39]).

Table 7.64: Category regarding "duplex capability" depending on Echo Loss

Category (according to Recommendation ITU-T 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).

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.64 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.

This measurement shall be repeated for the desktop type terminals and softphones with variable echo path.

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. A detailed description can be found in Recommendation ITU-T 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: Void

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

Send Direction		Receive Direction	
$f_0^{(1)}$ [Hz]	$\pm\Delta f^{(1)}$ [Hz]	$f_0^{(2)}$ [Hz]	$\pm\Delta f^{(2)}$[Hz]
125	±2,5	180	±2,5
250	±5	270	±5
500	±10	540	±10
750	±15	810	±15
1 000	±20	1 080	±20
1 250	±25	1 350	±25
1 500	±30	1 620	±30
1 750	±35	1 890	±35
2 000	±40	2 160	±35
2 250	±40	2 400	±35
2 500	±40	2 650	±35
2 750	±40	2 900	±35
3 000	±40	3 150	±35
3 250	±40	3 400	±35
3 500	±40	3 650	±35
3 750	±40	3 900	±35
4 000	±40	4 150	±35
4 250	±40	4 400	±35
4 500	±40	4 650	±35
4 750	±40	4 900	±35
5 000	±40	5 150	±35
5 250	±40	5 400	±35
5 500	±40	5 650	±35
5 750	±40	5 900	±35
6 000	±40	6 150	±35
6 250	±40	6 400	±35
6 500	±40	6 650	±35
6 750	±40	6 900	±35
7 000	±40		
NOTE: Parameters of the Shaping Filter: f ≥ 250 Hz: 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 Recommendation ITU-T 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 7.64. 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

7.5.7.14.0 General

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) shall be ≤ 15 ms.

Measurement method

The test setup is described in clause 6.10.4.

The test signal is the "short words for activation" sequence described in clause 7.3.4 of Recommendation ITU-T P.501 [41] with increasing level for each single word.

Figure 7.18d: Void

The settings of the test signal are as follows.

Table 7.64b: Settings for test signals

	Single word/ Pause Duration	Level of the first single word (active Signal Part at the MRP)	Level Difference between two Periods of the Test Signal
Single word to Determine Switching Characteristic in Send Direction	~600 ms/ ~400 ms	-24 dBPa (see NOTE)	1 dB
NOTE: The signal level is determined for each utterance individually according to Recommendation ITU-T P.56 [33].			

It is assumed that the pause length of about 400 ms is longer than the hang-over time so that the test object is back to idle mode after each single word.

The level of the transmitted signal is measured at the electrical reference point. The test signal is filtered by the transfer function of the test object. The measured signal level is referred to the filtered 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 single word which indicates the first activation of the test object. The time between the beginning of the single word and the complete activation of the test object is measured.

7.5.7.14.2 Activation in Receiving Direction

For further study.

7.5.7.14.3 Silence Suppression and Comfort Noise Generation

Requirements and measurement methods are for further study.

NOTE: In general it is not recommended to use silence suppression at all.

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 A-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 (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
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 receive direction consisting of an initial pause of 10 s and a periodical repetition of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] in receive 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.

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.

7.5.7.14.5 Speech Quality in the Presence of Background Noise

Speech Quality for wideband systems can be tested based on ETSI EG 202 396-3 [i.34]. The test method is applicable for narrowband (100 Hz to 4 kHz) and wideband (100 Hz to 8 kHz) transmission systems. LQOw is used for wideband systems.

Requirement

For the background noises defined in clause 6.10.6 the following requirements shall apply:

- N-MOS-LQOw $\geq$ 3,0;
- S-MOS-LQOw $\geq$ 3,0;
- G-MOS-LQOw $\geq$ 3,0.

NOTE: It is recommended to test the terminal performance with other types of background noises if the terminal is likely to be exposed to other noises than specified in clause 6.10.6.

Measurement method

The background noise simulation as described in clause 6.10.6 is used. The terminal is set-up as described in clause 6.10.4.

The background noise shall be applied for at least 5 s in order to adapt noise reduction algorithms in advance of the test.

The near end speech signal consists of 8 sentences of speech (2 male and 2 female talkers, 2 sentences each). Appropriate speech samples can be found in Recommendation ITU-T P.501 [41]. The preferred language is French since the objective method was validated with French language. The test signal level is +1,3 dBPa at the MRP.

Three signals are required for the tests:

1) The clean speech signal is used as the undisturbed reference (see ETSI EG 202 396-3 [i.34]).

2) The speech plus undisturbed background noise signal is recorded at the terminal's microphone position using an omnidirectional measurement microphone with a linear frequency response between 50 Hz and 12 kHz.

3) The send signal is recorded at the electrical reference point.

N-MOS-LQOw, S-MOS LQOw and G-MOS LQOw are calculated per sentence as described in ETSI EG 202 396-3 [i.34].

The resulting N-MOS-LQOw-, S-MOS-LQOw- and G-MOS-LQOw-values are averaged over all 8 sentences.

7.5.7.14.6 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out applying a speech signal in receive direction and comparing the noise level transmitted in the send direction under reference conditions with no far end speech to the noise level transmitted in the send direction under test conditions including far end speech. During and after the end of the speech signal the signal level in send direction shall 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 clause 6.10.6.

NOTE: The intention of this measurement is to detect impairments (modulations, switching and others) influencing the background noise transmitted from the terminal under test when a signal from the distant end (receiving side of the terminal under test) is present. Under these test conditions no modulation of the transmitted signal should occur. Modulation, switching or other type of impairments might be caused by an improper behaviour of a nonlinear processor working in conjunction with the echo canceller and erroneously switching or modulating the transmitted background noise.

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 reference 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 speech signal at the far end. The exactly identical background noise signal is applied. The background noise signal shall start at the same point in time as was used for the reference measurement without far end signal. The background noise shall be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms. After at least 5 s a series of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] is applied in receive direction with duration of at least 10 s. The test signal level in receive direction is -16 dBm0 at the electrical reference point.

For both reference and test conditions 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 speech 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.

7.5.7.15 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 shall not decrease by more than 6 dB from the maximum measured echo attenuation.

Measurement method

The test setup is described in clause 6.10.4.

The test signal consists of periodically repeated Composite Source Signal according to Recommendation ITU-T 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.

The difference between the maximum attenuation and the minimum attenuation is measured.

NOTE 1: In addition tests with more speech like signals should be made, e.g. Recommendation ITU-T P.501 [41] to see time variant behaviour of EC. However, for such tests the simple broadband attenuation based test principle as described above cannot be applied due to the time varying spectral content of the speech like signals.

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) taking into account the exponential character of the integration time in any tolerance scheme.

NOTE 3: Care should be taken not to confuse noise or comfort noise with residual echo. In cases of doubt the measured echo signal should be compared to the residual noise signal measured under the same conditions without inserting the receive signal. If the level vs. time analysis leads to the identical result it can be assumed that no echo but just comfort noise is present.

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	Limit
100 Hz	-41 dB
1 300 Hz	-41 dB
3 450 Hz	-46 dB
5 200 Hz	-46 dB
7 500 Hz	-37 dB
8 000 Hz	-37 dB
NOTE: 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 the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The level of the training sequence shall be -16 dBm0.

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The measurement is carried out under steady-state conditions. The average test signal level shall be -16 dBm0, averaged over the complete test signal. 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.

7.5.7.15.3 Variable echo path

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk with dynamic changing echo paths.

No level peak shall be more than 10 dB above the minimum noise level during the measurement.

Measurement method

Test setup is described in clause 6.10.7.

NOTE: Care should be taken to not generate noise during the movement of the notebook lid. Because of this, this measurement is not applicable for a softphone without external microphone.

As test signal the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is used. The signal level shall be -10 dBm0. The terminal volume control is set to nominal RLR. The first 4 sentences of the test signal are used to allow full convergence of the echo canceller. The next 4 sentences (from 10,75 s to 22,5 s) are used for the analysis. The echo signal level is analysed over time. The echo signal level is analysed for 11,75 s, using a time constant of 35 ms.

The measurement result is displayed as echo level versus time.

7.5.7.16 Microphone mute

Requirement

The SLR (Send Loudness Rating) with microphone mute on shall be at least 50 dB higher than with microphone mute off.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 7.5.7.3, but its microphone shall be configured to be muted.

Measurement and calculation method are the same as in clause 7.5.7.3.

7.5.7.17 Delay

Requirement

If connected to an FP other than type 5, 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 52 ms. This value includes the 5 ms delay of the reference FP looping back the digital signal towards the PP.

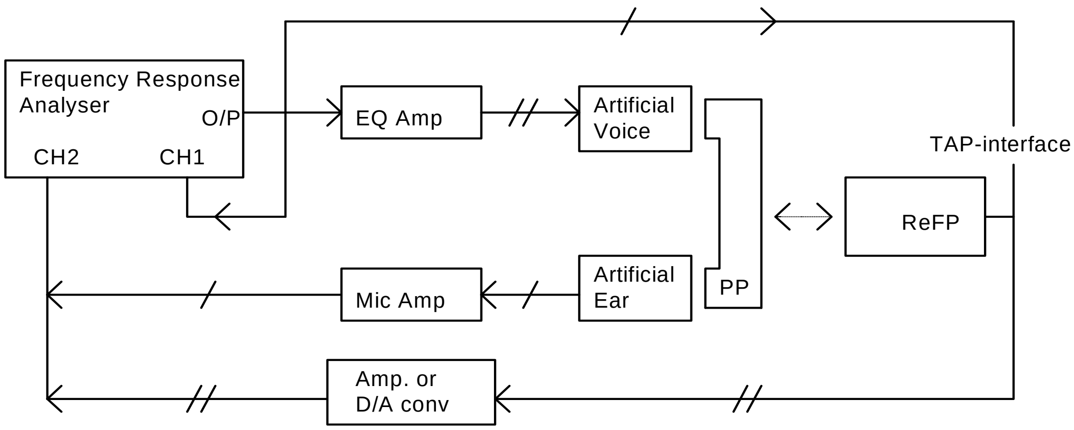
NOTE 1: Technically, this PP type can also be connected to an FP of type 5 (Fixed Part with VoIP interface, wideband, super-wideband, fullband, FBHR or ultra-band service) i.e. a VoIP interface. The roundtrip delay of a VoIP-terminal (PP+FP) is defined as the sum of send and receive delays. For a telecommunication connection, only the roundtrip delay can be experienced. For this reason, also the requirement for VoIP-terminals is given only for the PP+FP roundtrip delay.

NOTE 2: The derivation of the delay value can be found in clause F.3.

If connected to an FP type 5, the requirement given in clause 7.7.1 for the combined FP+PP roundtrip delay applies.

Measurement method

If connected to an FP type 5 other than type 5, 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.



-/-- Receive direction (T_r);
-//- Send direction (T_S).

Figure 7.18d1: PP delay test configuration

The delay shall be measured by the cross-correlation method as described in annex C.

If connected to an FP type 5, the measurement method described in clause 7.7.1 applies.

7.5.7.18 Send Loudness Level

Requirements:

The nominal value of Send Loudness Level (SLL) shall be:

$$SLL = 71 \text{ phon} \pm 4 \text{ phon}$$

Measurement method:

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The terminal is set-up as described in clause 6.10.4 and calibration is realized as explained in clause 6.10.4.

The loudness (in sone) of the recorded signal is calculated according to Recommendation ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is calculated according to clause 9 of Recommendation ITU-T P.700 [74].

7.5.7.19 Receive Loudness Level

Requirements:

- *Desktop operated PP:*
  - The nominal value of Receive Loudness Level (RLL) shall be 71 ± 4 phon. This value has to be fulfilled for at least one volume setting.
  - The value of RLL at maximum volume setting shall be louder than or equal to 78 phon.
  - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 15 phon.
- Handheld operated PP:
  - Improved class:
    - The nominal value of Receive Loudness Level (RLL) shall be 67 ± 4 phon. This value has to be fulfilled for at least one volume setting.
    - The value of RLL at maximum volume setting shall be louder than or equal to 71 phon.
    - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 15 phon.
  - Standard class:
    - The nominal value of Receive Loudness Level (RLL) shall be 67 ± 4 phon. This value has to be fulfilled for at least one volume setting.
    - The value of RLL at maximum volume setting shall be louder than or equal to 68 phon.
    - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 15 phon.
- Softphone (computer-based terminal):
  - Type 1 or softphone with external speakers: requirement as for desktop terminal.
  - Type 2: requirement as for handheld terminal.
- Group audio terminal*:*
  - The nominal value of Receive Loudness Level (RLL) shall be 71 ± 4 phon. This value has to be fulfilled for at least one volume setting.
  - The value of RLL at maximum volume setting shall be louder than or equal to 82 phon.
  - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 19 phon.

Measurement method:

The terminal is set-up as described in clause 6.10.4.

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

For each recorded artificial ear signal, the loudness (in sone) of the recorded signal is calculated according to Recommendation ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is determined as follows for binaural measurements: the resulting loudnesses for left and right ears are first halved (individual loudness per ear). Both loudnesses are added (assuming perfect loudness summation). With this overall loudness, the overall loudness level is finally determined according to clause 8.2 of Recommendation ITU-T P.700 [74].

7.5.8 Transmission characteristics for PP type 5a ("super-wideband 14 kHz handset or headset")

7.5.8.1 Frequency responses

7.5.8.1.1 Sending

7.5.8.1.1.1 Send frequency response - nominal position

Requirement

The send frequency response of the handset or headset shall be within a mask as defined in table 7.66a and shown in figure 7.18e. This mask shall be applicable for all types of handsets and headsets.

Table 7.66a: Super-wideband send frequency response limits

Frequency	Upper Limit	Lower Limit
50 Hz	0 dB	
100 Hz	5 dB	-5 dB
12 500 Hz	5 dB	-5 dB
14 000 Hz	5 dB	-10 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale. The requirement is based on 1/12th octave measurement.		

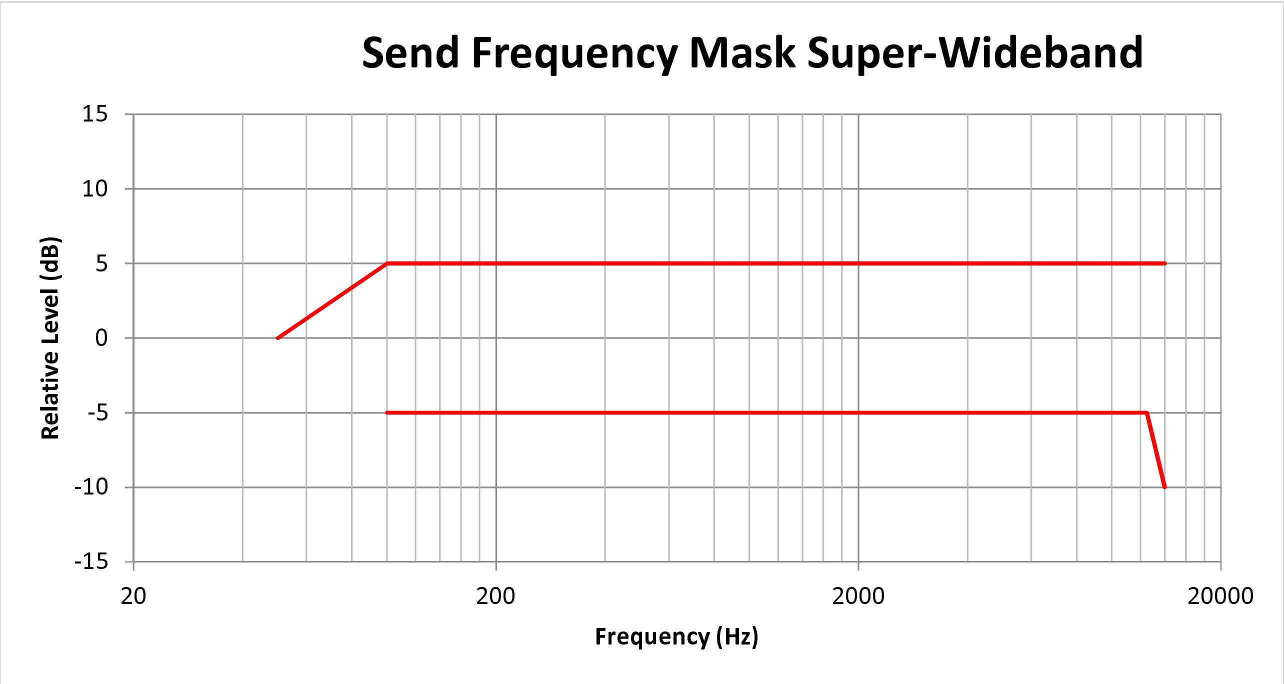


Figure 7.18e: Send frequency response mask for super-wideband

NOTE 1: The basis for the target frequency responses in sending and receiving is the orthotelefonic reference response which is measured between two 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 free-field. With the concept of free-field based receive measurements a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a free 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.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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 standard HATS position. 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 Recommendation ITU-T 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 IEC 61260-1 [45] for frequencies from 50 Hz to 14 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 at the MRP.

The sensitivity is expressed in terms of dBV/Pa.

7.5.8.1.1.2 Send frequency response - positional robustness

Requirement

For each of the modified handset positions, the send frequency response shall be within a given mask. The mask values per frequency are identical to table 7.66a, except that an additional tolerance is provided for certain positions. Table 7.66a1 provides the offset in dB for the lower limit.

Table 7.66a1: Tolerance mask offsets for send frequency response

Position	Offset Lower Limit
UP	-1 dB
DOWN	-2 dB
AWAY	-1 dB

Measurement method

The test arrangement and measurement is identical to clause 7.5.8.1.1.1. Instead of the standard handset position, the three modified positions according to table 6.3 for sending direction shall be used. The resulting three frequency responses shall be reported separately for each position.

7.5.8.1.2 Receiving

7.5.8.1.2.1 Receive frequency response - nominal position

Requirement

The receive frequency response of the handset or the headset shall be within a mask as defined in table 7.66b and shown in figure 7.18f.

Table 7.66b: Super-wideband receive frequency response limits

Frequency	Upper Limit	Lower Limit
50 Hz	3 dB	-5 dB
400 Hz	3 dB	-5 dB
1010 Hz	(see note)	-5 dB
1 200 Hz	(see note)	-8 dB
1 500 Hz	(see note)	-8 dB
2 000 Hz	9 dB	-3 dB
3 200 Hz	9 dB	-3 dB
14 000 Hz	9 dB	-13 dB
NOTE: 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. It is a floating or "best fit" mask. The requirement is based on 1/12th octave measurement.		

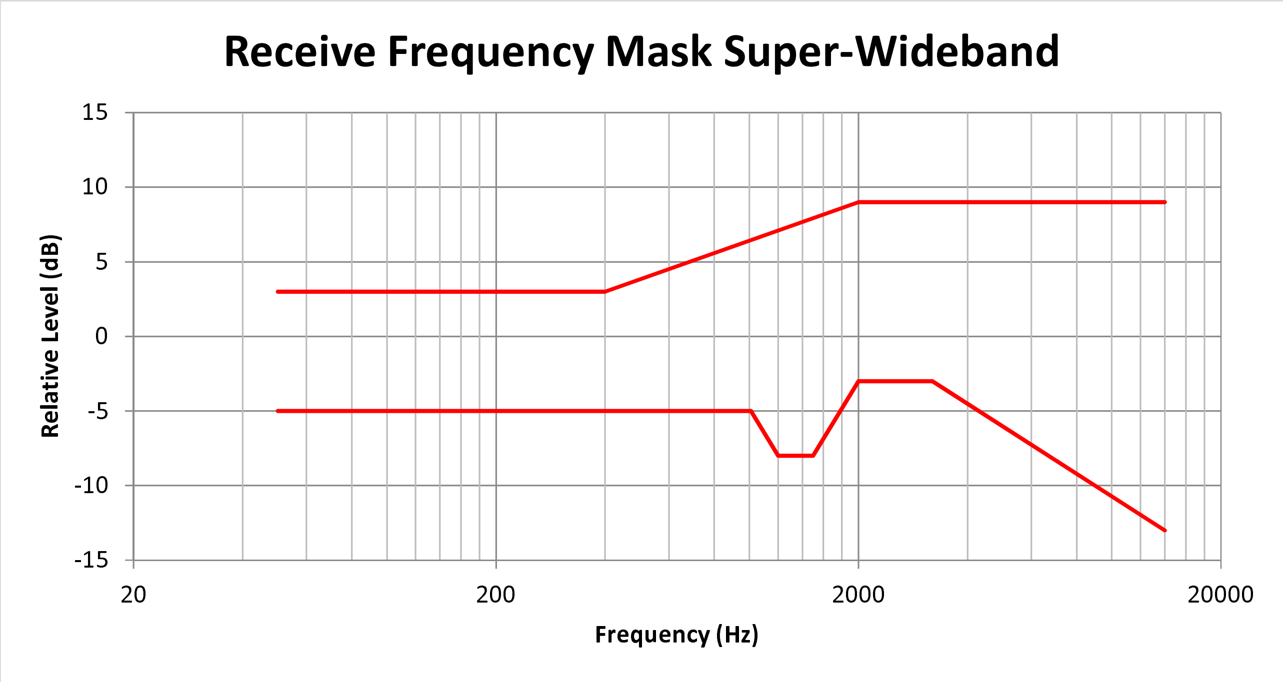


Figure 7.18f: Receive frequency response mask for super-wideband

NOTE 1: This requirement applies to headphones not primarily designed for super-wideband communication but rather for music audition. It is the reason of rather open limits. In the next future, new limits will be discussed to apply when specially designed super-wideband headphones will be available. Hence ETSI TS 102 924 [60] should be checked regularly for updates on this topic.

NOTE 2: A gap, defined in annex G, for the lower mask limit, is allowed.

Measurement method

Receive frequency response is the ratio of the measured sound pressure and the input level.
(dB relative Pa/V).

$$S_{Jedf} = 20 \log (pe_{df} / v_{RCV}) \text{ dB rel 1 Pa / V}$$

S_{Jedf} Receive Sensitivity; Junction to HATS Ear with diffuse-field correction.

pe_{df} 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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -16 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard HATS position. The application forces used to apply the handset against the artificial ear is 2 N, 8 N and 13 N.

The sound pressure level is measured at the DRP of the HATS for each $1/12^{th}$ octave band.

In case of headset measurements the tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

The HATS is diffuse-field equalized as described in Recommendation ITU-T P.581 [43]. The diffuse-field correction as defined in Recommendation ITU-T P.58 [35] is applied.

Measurements shall be made at one twelfth-octave intervals as given by the IEC 61260-1 [45] for frequencies from 50 Hz to 14 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.

7.5.8.1.2.2 Receive frequency response - positional robustness

Requirement

For each of the modified handset positions, the receive frequency response shall be within a given mask. The mask values per frequency are identical to table 7.66b, except that an additional tolerance is provided for certain positions. Table 7.66b1 provides the offset in dB for the lower limit.

Table 7.66b1: Tolerance mask offsets for receive frequency response

Position	Offset Lower Limit
Ye_{-5} Ze_{-5}	-1 dB
Ye_{0} Ze_{+5}	-1 dB
Ye_{+5} Ze_{-5}	-1 dB

Measurement method

The test arrangement and measurement is identical to clause 7.5.8.1.2.1. Instead of the standard handset position, the three modified positions according to table 6.4 for receiving direction shall be used. The resulting three frequency responses shall be reported separately for each position.

7.5.8.2 Send and receive loudness ratings

7.5.8.2.1 Send Loudness Rating

7.5.8.2.1.1 Send loudness rating - nominal position

Requirement

The nominal value of Send Loudness Rating (SLR) shall be:

$$SLR(set) = 8 \text{ dB} \pm 3 \text{ dB}$$

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T 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 standard HATS position. 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 Recommendation ITU-T P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

The sending sensitivity shall be calculated for each of the 20 frequency bands given in table 1 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], annex A.

7.5.8.2.1.2 Send Loudness Rating - positional robustness

Requirement

The difference (in dB) between the SLR measured in each of the three modified handset positions and the one in determined standard position (STD) shall be in the range of -3 to +3 dB.

Measurement method

In addition to the test setup and measurement of clause 7.5.8.2.1.1, each of the three modified handset positions for sending direction according to table 6.3 shall be applied. SLR and delta-SLR values should be calculated and reported for each position.

7.5.8.2.1.3 Microphone mute

Requirement:

The SLR (Send Loudness Rating) with microphone mute on shall be at least 50 dB higher than with microphone mute off.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 7.5.8.2.1.1, but its microphone shall be configured to be muted.

Measurement and calculation method are the same as in clause 7.5.8.2.1.1.

7.5.8.2.2 Receive Loudness Rating

7.5.8.2.2.0 Note

NOTE: Only requirements for monaural reproduction are provided, stereo/dichotic reproduction is for further study.

7.5.8.2.2.1 Receive Loudness Rating - nominal position

Requirement

When terminal implements wideband speech functions or when the super-wideband functions may interact with wideband terminals, the terminal shall fulfil the requirements on RLR as defined in ETSI ES 202 739 [i.10], clause 6.3.13.1:

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.

The nominal value of RLR is the RLR closest to the nominal requirement.

The minimum difference between nominal RLR and minimum (loudest, maximum volume setting) RLR shall be higher than 6 dB.

NOTE: The mechanical design of some UEs may make it impossible to seal the ear-piece to the knife edge of the ITU-T artificial ear. Minimal additional methods may be used to provide the seal provided that they do not affect the mounting position of the UE with respect to the Mouth Reference Point and the Ear Reference Point.

Measurement method

The test signal to be used for the measurements shall be British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T 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 or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard HATS position. 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 Recommendation ITU-T P.581 [43]. The DRP-ERP correction as defined in Recommendation ITU-T 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, 8 N is used.

In case of headset measurements the tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

The receiving sensitivity shall be calculated for each of the 20 frequency bands given in table 1 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], annex A. No leakage correction shall be applied for the measurement.

7.5.8.2.2.2 Receive Loudness Rating - positional robustness

Requirement

The difference (in dB) between the RLR measured in each of the three modified handset positions and the one in standard position (STD) shall be in the range -3 to +3 dB.

Measurement method

In addition to the test setup and measurement of clause 7.5.8.2.2.1, each of the three modified handset positions for receiving direction according to table 6.4 shall be applied. An application force of 8 N is used. RLR and delta-RLR values should be calculated and reported for each position.

7.5.8.2.3 Send Loudness Level

Requirements

The nominal value of Send Loudness Level (SLL) shall be:

$$SLL = 75 \text{ phon} \pm 4 \text{ phon}$$

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The handset or headset terminal is set-up as described in clause 6.10.3.

The loudness (in sone) of the recorded signal is calculated according to ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is calculated according to clause 9 of ITU-T P.700 [74].

7.5.8.2.4 Receive Loudness Level

Requirements

The nominal value of Receive Loudness Level (RLL) for handsets, monaural and binaural/stereo headsets shall be:

RLL = 75 phon ± 4 phon

In case a user controlled receive volume control is provided, for at least one setting of the control the RLL shall meet the nominal value.

When the control is set to maximum, the RLL shall not be louder than 89 phon. With the volume control set to the minimum position the RLL shall not be quieter than 58 phon.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The handset or headset terminal is set-up as described in clause 6.10.3.

The loudness (in sone) of the recorded signal is calculated according to ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is determined as follows:

- Handsets, monaural headsets: the loudness level is calculated according clause 8.2 of ITU-T P.700 [74] by using the loudness value divided by two (loudness halving for monaural listening).
- Binaural headsets: the loudness level is calculated according clause 8.2 of ITU-T P.700 [74] by using directly the loudness value (loudness summation for binaural listening is retained).

7.5.8.3 Sidetone

7.5.8.3.1 Sidetone Masking Rating STMR (Mouth to ear)

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 shall not be below 8 dB.

NOTE 1: It is preferable to have a constant STMR independent of the volume control setting.

NOTE 2: STMR measurement in Recommendation ITU-T P.79 [37] is not defined above 8 kHz, but sidetone signal is not supposed to have such limitation.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -4,7 dBPa, measured at the MRP. The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard HATS position and the application force shall be 13 N 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 IEC 61260-1 [45] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band (Recommendation ITU-T 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 (LmeST), as expressed in dB, and the SideTone Masking Rate (STMR) (in dB) shall be calculated from the formula 5-1 of Recommendation ITU-T P.79 [37], using m = 0,225 and the weighting factors of in table 3 of Recommendation ITU-T P.79 [37].

7.5.8.3.2 Sidetone Delay

Requirement

The maximum sidetone-round-trip delay shall be ≤ 5 ms, measured in an echo-free setup.

Measurement method

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard HATS position.

The test signal is a CS-signal complying with Recommendation ITU-T 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 Recommendation ITU-T P.501 [41].

The level of the signal shall be -4,7 dBPa at the MRP.

The cross-correlation function $\Phi_{xy}(\tau)$ 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:

$$\Phi_{xy}(\tau) = \frac{1}{T}\int_{t=\frac{-T}{2}}^{\frac{T}{2}} S_x(t) \cdot S_y(t+\tau) \cdot dt$$

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 $\Phi_{xy}(\tau)$. 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 $\{\Phi_{xy}(\tau)\}$ of the cross-correlation:

$$H\{\Phi_{xy}(\tau)\} = \sum_{u=-\infty}^{\infty} \frac{\Phi_{xy}(u)}{\Pi(\tau - u)}$$

$$E(\tau) = \sqrt{\left[\Phi_{xy}(\tau)\right]^2 + \left[H\{\Phi_{xy}(\tau)\}\right]^2}$$

It is assumed that the measured sidetone delay is less than T/2.

7.5.8.4 Terminal Coupling Loss

7.5.8.4.1 Unweighted Terminal Coupling Loss

Requirement

The TCL measured as unweighted Echo Loss shall be ≥ 55 dB for all positions of the volume control (if supplied).

NOTE: Depending on the idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 dB.

Measurement method

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard HATS position and the application force shall be 2 N on the artificial ear type 3.3 or type 3.4 as specified in Recommendation ITU-T P.57 [34]. The ambient noise level shall be < -64 dBPa(A). The attenuation from electrical reference point input to electrical reference point output shall be measured using the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41].

TCL is calculated as unweighted echo loss from 100 Hz to 8 kHz. 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 first 17,0 s of the test signal (6 sentences) are discarded from the analysis to allow for convergence of the acoustic echo canceller. The analysis is performed over the remaining length of the test sequence (last 6 sentences).

For the measurement, a time window has to be applied which is adapted to the duration of the actual test signal. The echo loss is calculated by the equations:

$$L_e = C - 10\log_{10}\sum_{i=1}^{N}(A_i + A_{i-1})(\log_{10} f_i - \log_{10} f_{i-1}) \quad (7.5)$$

and

$$C = 10\log_{10}(2\,(\log_{10} f_N - \log_{10} f_0)) \quad (7.6)$$

where:

- A_0 is the output/input power ratio at frequency f_0 = 100 Hz;
- A_1 the ratio at frequency f_i; and
- A_N the ratio at frequency f_N = 8 000 Hz.

Equation (7.5) is a generalized form of the equation defined in Recommendation ITU-T G.122 [16], clause B.4, for calculating echo loss based on tabulated data, which allows the calculation of echo loss within any frequency range between f_0 and f_N.

7.5.8.4.2 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 50 Hz to 16 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 of the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41] is applied. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with Recommendation ITU-T 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 50 Hz to 16 kHz under the following conditions:

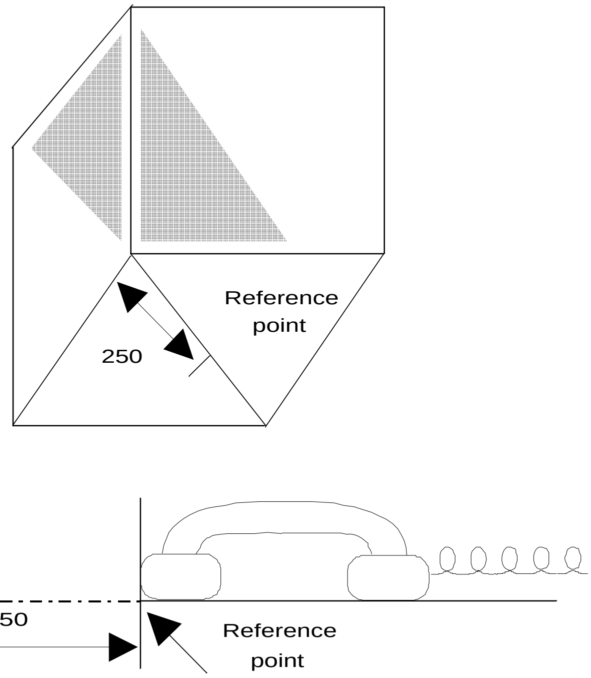
a) the headset, with the transmission circuit fully active, shall be positioned on one inside surface that is of three perpendicular planes, smooth, hard surfaces forming a corner. Each surface shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line, extending from the corner formed by the three surfaces, and a reference position 250 mm from the corner, as shown in figure 7.18g;

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 7.18g;

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 7.18g;
3) the headset microphone is positioned as close as possible to the receiver.



NOTE: All dimensions are in mm.

Figure 7.18g

7.5.8.5 Distortion

7.5.8.5.1 Sending Distortion

7.5.8.5.1.1 Signal to harmonic distortion versus frequency

Requirement

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.66c: Send distortion for super-wideband

Frequency	Ratio
100 Hz	24 dB
200 Hz	26 dB
400 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
3,15 kHz	30 dB
5 kHz	30 dB
NOTE: 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 or headset terminal is set-up as described in clause 6.10.3.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 100 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz, 3 150 Hz and 7 000 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 14 kHz.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.8.5.1.2 Signal to harmonic distortion for higher input level

Requirement

For the signal defined in the measurement method, the signal to harmonic distortion ratio shall be ≥ 30 dB.

Measurement method

The handset or headset terminal is set-up as described in clause 6.10.3.

The signal used is an activation signal followed by a 1 kHz sine wave. The signal to harmonic distortion ratio is measured selectively up to 14 kHz.

The duration of the sine wave shall be ≤ 1 s. The sinusoidal signal level shall be calibrated to +10 dBPa at the MRP.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.8.5.2 Receiving Distortion

Requirement

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.66d: Receive distortion for super-wideband

Frequency	Signal to distortion ratio limit, receiving
100 Hz	24 dB
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
3,15 kHz	30 dB
5 kHz	30 dB
NOTE: 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 or 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 100 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz, 3 150 Hz and 7 000 Hz.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The signal level shall be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 14 kHz.

The ratio of signal to harmonic distortion shall be measured at the DRP of the artificial ear with a correction by the curve of reference microphone.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.8.6 Noise

7.5.8.6.1 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 female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

The handset or headset terminal is set-up as described in clause 6.10.3.

The send noise is measured at the POI in the frequency range from 50 Hz to 14 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).

Spectral peaks are measured in the frequency domain in the frequency range from 100 Hz to 14 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE: In case spectral peaks higher than 10 dB above the average noise floor are produced by the terminal, but which are considered to be inaudible due to the very low noise floor produced by the terminal on average, it is the responsibility of the test lab to demonstrate the desired performance subjectively.

7.5.8.6.2 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.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement method

The handset or 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 diffuse field equalization active. The noise level is measured until 20 kHz.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The activation signal level shall be -16 dBm0.

Spectral peaks are measured in the frequency domain in the frequency range from 50 Hz to 14 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE: In case spectral peaks higher than 10 dB above the average noise floor are produced by the terminal, but which are considered to be inaudible due to the very low noise floor produced by the terminal on average, it is the responsibility of the test lab to demonstrate the desired performance subjectively.

7.5.8.7 Acoustic shock

7.5.8.7.0 General

In order to fulfil the acoustic shock requirements, it is recommended to follow the guidelines of Recommendation ITU-T P.360 [57]. If needed the PP may have to implement some kind of hardware limiter.

7.5.8.7.1 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

See ETSI EG 202 518 [i.32].

7.5.8.8 Delay

Technically, this PP type can only be connected to an FP of type 5 (Fixed Part with VoIP interface, wideband, super-wideband, fullband, FBHR or ultra-band service), i.e. a VoIP interface. Hence the requirements and measurement methods given in clause 7.7.1 for the combined FP+PP roundtrip delay apply.

Figure 7.18h: Void

Figure 7.18i: Void

7.5.8.9 Double talk Performance

7.5.8.9.0 General

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 (ELR) 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 Recommendations ITU-T P.340 [39] and P.502 [42]):

- attenuation range in send direction during double talk $A_{H,S,dt}$;
- attenuation range in receive direction during double talk $A_{H,R,dt}$;
- echo attenuation during double talk.

7.5.8.9.1 Attenuation range in send direction during double talk $A_{H,S,dt}$

Requirement

Based on the level variation in send direction during double talk $A_{H,S,dt}$ the behaviour of the terminal shall be classified according to table 7.66e.

Table 7.66e

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.66e 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 handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is the double talk speech sequence as defined in clause 7.3.5 of Recommendation ITU-T P.501 [41] as shown in figure 7.18j. The competing speaker is always inserted as the double talk sequence sdt(t) in send direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

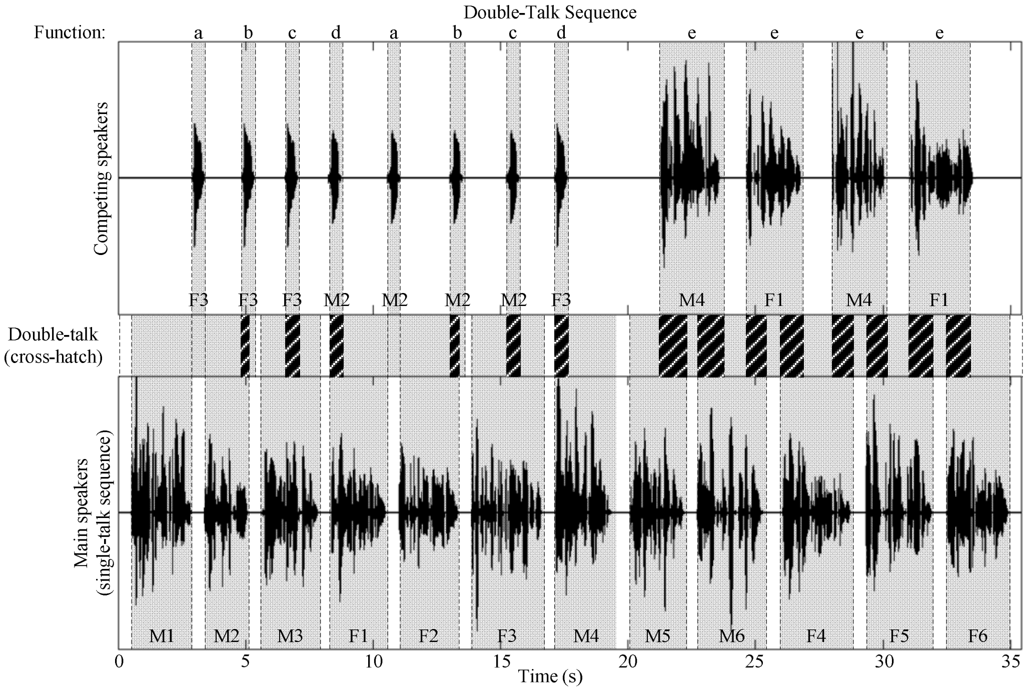


Figure 7.18j: Double talk test sequence with overlapping speech sequences in send and receive direction

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

7.5.8.9.2 Attenuation range in receive direction during double talk $A_{H,R,dt}$

Requirement

Based on the level variation in receive direction during double talk $A_{H,R,dt}$ the behaviour of the terminal shall be classified according to table 7.66f.

Table 7.66f

Category (according to Recommendation ITU-T P.340 [39])	1	2a	2b	2c	3
	Full duplex capability	Partial duplex capability			Full duplex capability
$A_{H,R,dt}$ [dB]	≤ 3	≤ 5	≤ 8	≤ 10	> 10

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.66f 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 handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is shown in figure 7.18k. The competing speaker is always inserted as the double talk sequence in receive direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

7.5.8.9.3 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 (TELRdt). 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 table 7.66g are applicable (more information can be found in annex A of the Recommendation ITU-T P.340 [39]).

Table 7.66g

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

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. A detailed description can be found in Recommendation ITU-T P.501 [41].

The signals are fed simultaneously in send and receive direction. The level in send direction is -4,7 dBPa at the MRP (nominal level), the level in receive direction is -16 dBm0 at the electrical reference point (nominal level).

The settings for the signals are as follows.

Table 7.66h: Parameters of the two test signals for double talk measurement based on AM-FM modulated sine waves

Send Direction		Receive Direction	
$f_0^{(1)}$ [Hz]	$\pm\Delta f^{(1)}$ [Hz]	$f_0^{(2)}$ [Hz]	$\pm\Delta f^{(2)}$[Hz]
125	±2,5	180	±2,5
250	±5	270	±5
500	±10	540	±10
750	±15	810	±15
1 000	±20	1 080	±20
1 250	±25	1 350	±25
1 500	±30	1 620	±30
1 750	±35	1 890	±35
2 000	±40	2 160	±35
2 250	±40	2 400	±35
2 500	±40	2 650	±35
2 750	±40	2 900	±35
3 000	±40	3 150	±35
3 250	±40	3 400	±35
3 500	±40	3 650	±35
3 750	±40	3 900	±35
4 000	±40	4 150	±35
4 250	±40	4 400	±35
4 500	±40	4 650	±35
4 750	±40	4 900	±35
5 000	±40	5 150	±35
5 250	±40	5 400	±35
5 500	±40	5 650	±35
5 750	±40	5 900	±35
6 000	±40	6 150	±35
6 250	±40	6 400	±35
6 500	±40	6 650	±35
6 750	±40	6 900	±35
7 000	±40		
NOTE: Parameters of the Shaping Filter: f ≥ 250 Hz: Low Pass Filter, 5 dB/oct.			

The test signal is measured at the electrical reference point (send 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 receive direction (see Recommendation ITU-T P.501 [41]). The filter will suppress frequency components of the double talk signal.

In each frequency band which is used in receive 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 7.66g. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

7.5.8.10 Switching Characteristics

7.5.8.10.0 Note

NOTE: Additional requirements may be needed in order to further investigate the effect of NLP implementations on the users' perception of speech quality.

7.5.8.10.1 Activation in send direction

The activation in send 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 send 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) shall be ≤ 15 ms.

Measurement method

The handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The test signal is the "short words for activation" sequence described in clause 7.3.4 of Recommendation ITU-T P.501 [41] with increasing level for each single word.

Table 7.66i

	Single word duration / pause duration	Level of the first single word (at the MRP)	Level difference between two periods of the test signal
single word to determine switching characteristic in send direction	~600 ms / ~400 ms	-24 dBPa (see note)	1 dB
NOTE: The signal level is determined for each utterance individually according to Recommendation ITU-T P.56 [33].			

It is assumed that the pause length of about 400 ms is longer than the hang-over time so that the test object is back to idle mode after each single word. The level of the transmitted signal is measured at the electrical reference point. The test signal is filtered by the transfer function of the test object. The measured signal level is referred to the filtered test signal level and displayed versus time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the single word which indicates the first activation of the test object. The time between the beginning of the single word and the complete activation of the test object is measured.

7.5.8.10.2 Silence suppression and comfort noise generation

Requirements and measurement methods are for further study.

NOTE: In general it is not recommended to use silence suppression at all.

7.5.8.10.3 Performance in Sending in the Presence of Background Noise

Requirement

The level of comfort noise shall be within in a range of +2 dB to -5 dB compared to the original (transmitted) background noise. The noise level is calculated with A-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 7.66j.

Table 7.66j: Requirements for spectral adjustment of comfort noise (mask)

Frequency	Upper limit	Lower limit
200 Hz	12 dB	-12 dB
800 Hz	12 dB	-12 dB
800 Hz	10 dB	-10 dB
2 000 Hz	10 dB	-10 dB
2 000 Hz	6 dB	-6 dB
4 000 Hz	6 dB	-6 dB
14 000 Hz	6 dB	-6 dB
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 or headset terminal is set-up as described in clause 6.10.3. The handset is mounted at the standard HATS position.

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 receive direction consisting of an initial pause of 10 s and a periodical repetition of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] in receive 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.

The transmitted signal is recorded in send direction at the POI.

The power density spectra measured in send direction without far end speech simulation averaged between 10 s and 20 s is referred to the power density spectrum measured in send direction determined during the period with far end speech simulation in receive direction averaged between 10 s and 20 s. Level and spectral differences between both power density spectra are analysed and compared to the requirements.

7.5.8.10.4 Speech quality in the presence of background noise

Requirement

For the background noises defined in clause 6.10.6 the following requirements shall apply:

- N-MOS-$LQO_f \geq 3{,}5$.
- S-MOS-$LQO_f \geq 4$.
- G-MOS-$LQO_f \geq 3{,}5$.

NOTE: It is recommended to test the terminal performance with other types of background noises if the terminal is likely to be exposed to other noises than specified in clause 6.10.6.

Measurement method

The background noise simulation as described in clause 6.10.6 is used. The handset or headset terminal is set-up as described in clause 6.10.3. The handset is mounted at the standard HATS position.

The background noise should be applied for at least 5 s in order to adapt noise reduction algorithms in advance the test.

The near end speech signal consists of 16 sentences of speech (4 male and 4 female talkers, 2 sentences each). The American English speech samples from ETSI TS 103 281 [64] are used. The test signal level is -1,7 dBPa at the MRP.

The following signals are required for the tests:

1) The clean speech signal is used as the undisturbed reference (see ETSI TS 103 281 [64]).

2) The send signal is recorded at the electrical reference point.

N-MOS-LQO$_f$, S-MOS LQO$_f$ and G-MOS LQO$_f$ are calculated as described in ETSI TS 103 281 [64]. Either Model A or Model B can be used. The model chosen shall be documented in the test report.

When using model A the following mapping functions apply:

$$\text{S-MOS}'_{\text{LQOf}} = 1{,}418 \cdot \text{S-MOS}_{\text{LQOf}} - 1{,}145$$
$$\text{N-MOS}'_{\text{LQOf}} = 1{,}346 \cdot \text{N-MOS}_{\text{LQOf}} - 1{,}584$$
$$\text{G-MOS}'_{\text{LQOf}} = 1{,}279 \cdot \text{G-MOS}_{\text{LQOf}} - 0{,}7364$$

7.5.8.10.5 Positional Robustness of Speech Quality in the Presence of Background Noise

Requirement

The degradation between standard position (STD) and all other modified positions for sending direction shall not exceed the limits for S-MOS and N-MOS according to table 7.66k. The requirements are evaluated on the averaged results over all background noises used in this test.

Table 7.66k: Requirements for allowed degradation

Position	∆ S-MOS	∆ N-MOS
UP	≤ 0,2	≤ 0,2
DOWN	≤ 0,3	≤ 0,5
AWAY	≤ 0,3	≤ 0,4

Measurement method

The test arrangement and measurement is identical to clause 7.5.8.10.4. The test is conducted with each of the modified handset positions for sending direction according to table 6.3. All S- and N-MOS values as well as the difference to STD shall be reported for all three positions.

7.5.8.10.6 Quality of background noise transmission (with far end speech)

Requirement

The test is carried out applying a speech signal in receive direction and by comparing the noise level transmitted in the send direction under reference conditions with no far end speech, to the noise level transmitted in the send direction under test conditions including far end speech. During and after the end of the speech signal the signal level in send direction shall 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 clause 6.10.6.

NOTE: The intention of this measurement is to detect impairments (modulations, switching and others) influencing the background noise transmitted from the terminal under test when a signal from the distant end (receiving side of the terminal under test) is present. Under these test conditions no modulation of the transmitted signal should occur. Modulation, switching or other type of impairments might be caused by an improper behaviour of a nonlinear processor working in conjunction with the echo canceller and erroneously switching or modulating the transmitted background noise.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3.

The background noises are generated as described in clause 6.10.6.

First the reference measurement is conducted without inserting the signal at the far end. At least 10 s of noise is analysed. The transmitted 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 speech signal at the far end. The exactly identical background noise signal is applied. The background noise signal shall start at the same point in time as was used for the reference measurement without far end signal. The background noise shall be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms. After at least 5 s a series of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] is applied in receive direction with a duration of at least 10 s. The test signal level in the receive direction is -16 dBm0 at the electrical reference point.

For both reference and test conditions, the send 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 send direction is determined during the time interval when the speech signal is applied and after it stops. The level difference is determined from the difference of the recorded signal levels versus time between reference signal and the signal measured with far end signal.

7.5.8.11 Quality of echo cancellation

7.5.8.11.1 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 shall not decrease by more than 6 dB from the maximum echo attenuation measured.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3.

The test signal consists of periodically repeated Composite Source Signal according to Recommendation ITU-T 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 versus time. The exact synchronization between input and output signal has to be guaranteed.

The difference between the maximum attenuation and the minimum attenuation is measured.

NOTE 1: In addition, it is recommended to also conduct tests with more speech like signals, e.g. Recommendation ITU-T P.501 [41] in order to investigate time variant behaviour of EC. However, for such tests the simple broadband attenuation based test principle as described above cannot be applied due to the time varying spectral content of the speech like signals.

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 (35 ms) of the level analysis taking into account the exponential character of the integration time in any tolerance scheme.

NOTE 3: Care should be taken not to confuse noise or comfort noise with residual echo. In cases of doubt the measured echo signal should be compared to the residual noise signal measured under the same conditions without inserting the receive signal. If the level vs. time analysis leads to the identical result it can be assumed that no echo but just comfort noise is present.

7.5.8.11.2 Spectral echo attenuation

Requirement

The echo attenuation versus frequency shall be below the tolerance mask given in table 7.66l.

Table 7.66l: Echo attenuation limits

Frequency	Limit
100 Hz	-41 dB
1 300 Hz	-41 dB
3 450 Hz	-46 dB
5 200 Hz	-46 dB
7 500 Hz	-37 dB
8 000 Hz	-37 dB
NOTE: 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 send direction in order to mask the echo signal.

NOTE: The measurement is performed in wideband since no super-wideband data are available. The extension of the method to super-wideband is for further study.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3.

Before the actual measurement a training sequence consisting of the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is fed in. The level of the training sequence shall be -16 dBm0.

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The measurement is carried out under steady-state conditions. The average test signal level shall be -16 dBm0, averaged over the complete test signal. 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.

7.5.8.11.3 Variable echo path

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk with dynamic changing echo paths. The measured echo level over time during single talk shall not be more than 10 dB above the minimum noise level during the measurement.

Measurement method

The test arrangement is according to clause 6.10.7.

As test signal the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is used. The signal level shall be -10 dBm0. The terminal volume control is set to nominal RLR. The first 4 sentences of the test signal are used to allow full convergence of the echo canceller. The next 4 sentences (from 10,75 s to 22,5 s) are used for the analysis. The echo signal level is analysed over time. The echo signal level is analysed for 11,75 s, using a time constant of 35 ms.

The measurement result is displayed as echo level versus time.

7.5.9 Transmission characteristics for PP type 5b ("super-wideband 14 kHz loudspeaking and handsfree devices")

7.5.9.1 Sending sensitivity/frequency response

Requirements

The objective is to define a flat frequency curve over the whole bandwidth.

The frequency response for super-wideband shall fulfill the mask as defined in table 7.66m and figure 7.18k.

Table 7.66m: Frequency mask for Super-wideband terminals - Send

Frequency	Upper Limit	Lower Limit
50 Hz	0 dB	
100 Hz	5 dB	-5 dB
12 500 Hz	5 dB	-5 dB
14 000 Hz	5 dB	-10 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale. The requirement is based on 1/12th octave measurement.		

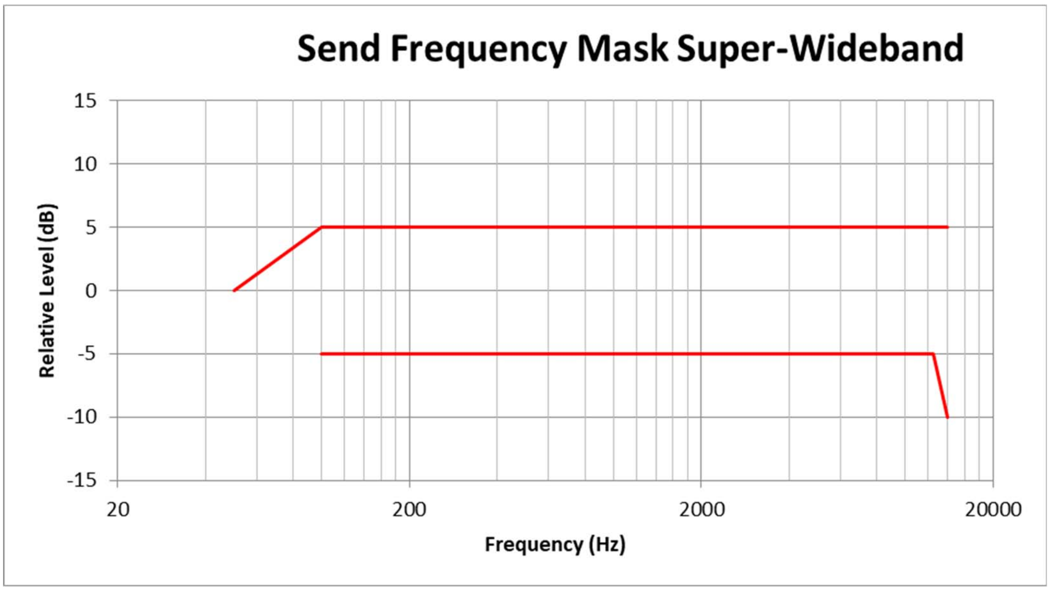


Figure 7.18k: Send frequency response mask for super-wideband

NOTE 1: A "balanced" frequency response is preferable from the perception point of view. If frequency components in the low frequency domain are attenuated, frequency components in the high frequency domain should be attenuated in a similar way.

NOTE 2: A gap, defined in annex G, for the lower mask limit, is allowed.

Additional requirements are for further study when the system is intended to be used by several users, when stereo features are made available or when microphone array(s) are used.

Measurement method

The terminal is set-up according to clause 6.10.4. The test to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level is defined according to clause 6.10.4.2.

Measurements shall be made at one twelfth-octave intervals as given by IEC 61260-1 [45] for frequencies from 50 Hz to 14 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 HFRP.

The sensitivity is expressed in terms of dBV/Pa.

7.5.9.2 Receive sensitivity/frequency response

7.5.9.2.1 Handheld terminal

Requirements

The frequency response shall fulfill the mask as defined in table 7.66n and figure 7.18l.

Table 7.66n: Frequency mask for super-wideband handheld terminals - Receive

Frequency	Upper Limit	Lower Limit
50 Hz	5 dB	
400 Hz	5 dB	-5 dB
12 500 Hz	5 dB	-5 dB
14 000 Hz	5 dB	-10 dB
16 000 Hz	5 dB	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale.		

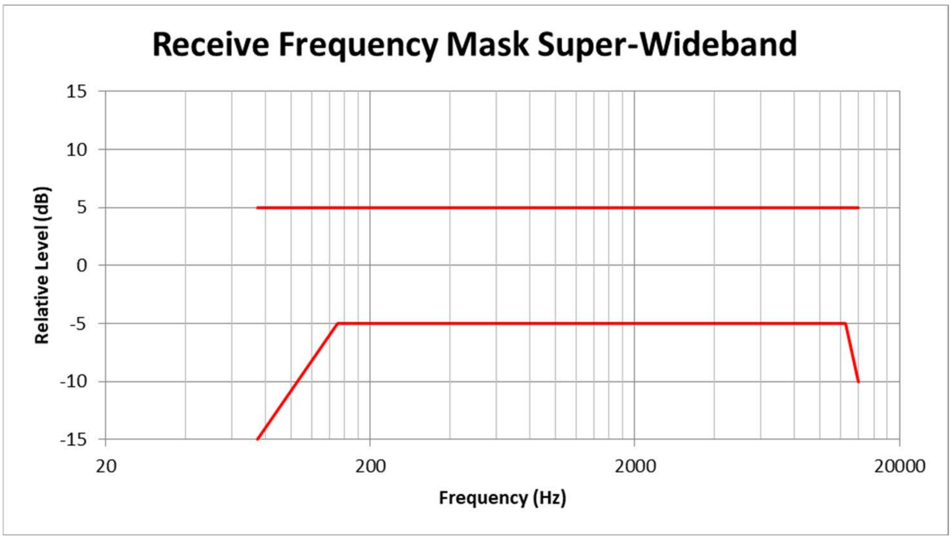


Figure 7.18l: Frequency mask for super-wideband handheld terminals - Receive

NOTE: A gap, defined in annex G, for the lower mask limit, is allowed.

Measurement method

The terminal is set-up according to clause 6.10.4.

The test signal to be used for the measurements shall be British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -16 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

The equalized output signal is power-averaged on the total time of analysis. The 1/3 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3745 [67] for frequencies from 400 Hz to 14 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.

7.5.9.2.2 Desktop terminal

Requirements

The frequency response shall fulfill the mask as defined in table 7.66o and figure 7.18m.

Table 7.66o: Frequency mask for super-wideband desktop terminals - Receive

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
75	5	-15
150	5	-5
12 500	5	-5
14 000	5	-10
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

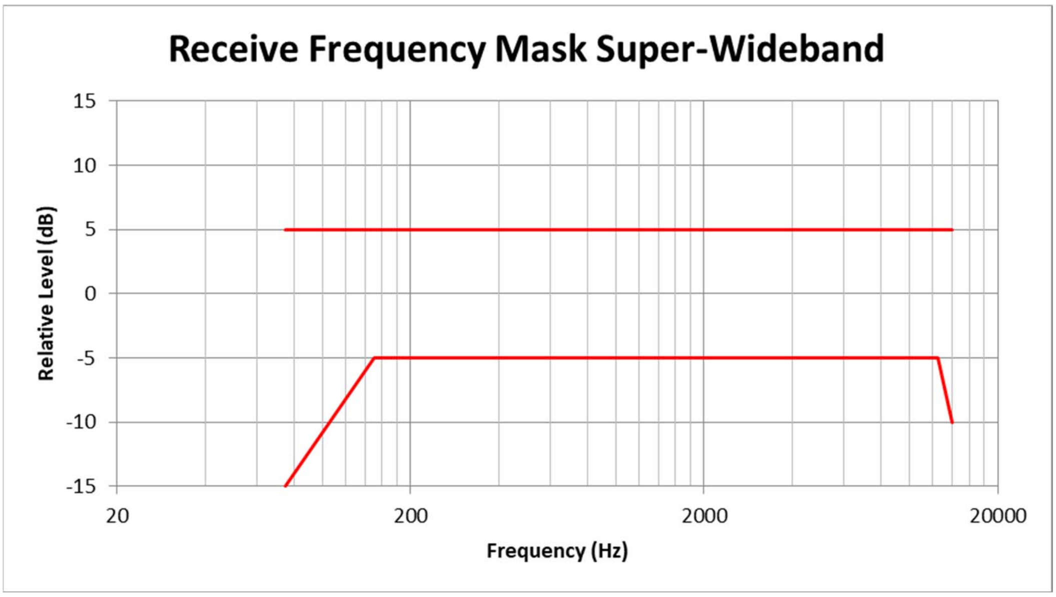


Figure 7.18m: Frequency mask for super-wideband desktop terminals - Receive

NOTE: A gap, defined in annex G, for the lower mask limit, is allowed.

Measurement method

The terminal is set-up according to clause 6.10.4.

Receive frequency response is the ratio of the measured sound pressure and the input level.
(dB relative Pa/V)

$$S_{Jeff} = 20 \log (p_{eff} / v_{RCV}) \text{ dB rel 1 Pa / V}$$

S_{Jeff} Receive Sensitivity; Junction to HATS Ear with free field correction.

p_{eff} 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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -16 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

The equalized output signal is power-averaged on the total time of analysis. The 1/3 octave band data are considered as the input signal to be used for calculations or measurements.

The HATS is free-field equalized as described in Recommendation ITU-T P.581 [43]. The free-field correction as defined in Recommendation ITU-T P.58 [35] is applied.

Measurements shall be made at one third-octave intervals as given by IEC 61260-1 [45] for frequencies from 75 Hz to 14 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.

7.5.9.2.3 Terminals intended to be used simultaneously by several users

Requirements

Additional requirements to be defined:

- when the terminal is intended to be used by several users;
- when stereo features are made available.

For all the testing positions the frequency curve shall fulfil the requirements defined for desktop terminals.

7.5.9.3 Sending loudness rating

7.5.9.3.1 Nominal Value

Requirements

To ensure the compatibility with other terminals or systems a reference SLR shall be defined.

The requirements refer to wideband handsfree terminals, ETSI ES 202 740 [i.11].

The nominal value shall be 13dB ± 3 dB.

There is no specific requirement for SWB bandwidth.

Measurement method

The terminal is positioned as described in clause 6.10.4.

For a correct activation of the system, the test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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.

Calibration is realized as explained in clause 6.10.4.2.

The send sensitivity shall be calculated for each of the 20 frequency bands given in table A.2 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], Annex A.

7.5.9.3.2 Microphone mute

Requirement

The SLR (Send Loudness Rating) with microphone mute on shall be at least 50 dB higher than with microphone mute off.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 7.5.9.3.1, but its microphone shall be configured to be muted.

Measurement and calculation method are the same as in clause 7.5.9.3.1.

7.5.9.4 Receive loudness rating

Requirements

When terminal implements wideband speech functions or when the super-wideband functions may interact with wideband terminals, the handsfree terminal shall fulfill the requirements on RLR as defined in ETSI ES 202 740 [i.11], clause 6.3.10:

- Desktop terminal:
  - Nominal value of RLR shall be 5 dB ± 3 dB. This value has to be fulfilled for one position of volume range.
  - The value of RLR at the upper part of the volume range shall be less than (louder) or equal to -2 dB: RLR ≤ -2 dB.
  - The range of volume control shall be ≥ 15 dB.
- Handheld terminal:
  - Nominal value of RLR shall be 9 dB ± 3 dB. This value has to be fulfilled for one position of volume range.
  - Value of RLR at upper part of volume range shall be less than (louder) or equal to 5 dB: RLR ≤ 5 dB.
  - Range of volume control shall be equal to or exceed 15 dB.
- Softphone (computer-based terminal):
  - Type 1 or softphone with external speakers: requirement as for desktop terminal.
  - Type 2 requirement as for handheld terminal.
- Group audio terminal:
  - Nominal value of RLR shall be 5 dB ± 3 dB. This value has to be fulfilled for one position of volume range.
  - Value of RLR at upper part of volume range shall be less than (louder) or equal to -6 dB: RLR ≤ -6 dB.
  - Range of volume control shall be ≥ 19 dB.

NOTE 1: Due to the lack of experience in the application of wide band loudness rating calculation as defined in Annex G of Recommendation ITU-T P.79 [37] the loudness rating calculation as described in Annex A is used.

NOTE 2: Loudness Rating measurement corresponding to level with speech signal, it can be considered that a measurement in wideband may be sufficient. Indeed, energy of speech beyond bandwidth of wideband is rather small.

NOTE 3: Receive Loudness Rating for stereo/dichotic is for further study.

Measurement method

The test setup is described in clause 6.10.4.

The measurement is conducted at nominal volume control setting.

The test signal to be used for the measurements shall be British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -16 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

The HATS is free field equalized as described in Recommendation ITU-T P.581 [43]. The equalized output signal is power-averaged on the total time of analysis. The 1/3 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one third-octave intervals as given by IEC 61260-1 [45] for frequencies from 100 Hz to 8 kHz inclusive. The receiving sensitivity shall be calculated for each of the 20 frequency bands given in table A.2 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], formula (A-23c), over bands 1 to 20 and the receive weighting factors from table A.2 of Recommendation ITU-T P.79 [37]. No leakage correction shall be applied.

For binaural measurements, the individual sensitivities for left and right ears are energetically summed up. The hands-free RLR based on this overall sensitivity is then calculated with a correction factor of -8 dB.

The test shall be repeated for maximum and minimum volume control setting.

7.5.9.5 Sending distortion

7.5.9.5.1 Signal to harmonic distortion versus frequency

Requirements

The ratio of signal to harmonic distortion shall be above the following masks.

The following draft requirements are defined for all the terminals within the scope of the present clause, as it is needed to ensure that any terminal intended to be used in super-wideband sends good quality signals. Care should be taken on the distortion of the HATS or of the loudspeaker used to test the send distortion of the terminal.

Table 7.66p

Frequency	Ratio
100 Hz	25 dB
200 Hz	30 dB
400 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
3,15 kHz	30 dB
5 kHz	30 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale.	

The signal to harmonic distortion ratio is measured selectively up to 16 kHz.

Measurement method

The terminal is set according to clause 6.10.4.

The signal used is an activation signal followed by a series sine wave signal with a frequency at 100 Hz, 200 Hz, 400 Hz, 1 kHz, 2 kHz, 3,15 kHz and 5 kHz. The signal to harmonic distortion ratio is measured selectively up to 14 kHz.

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.

For a correct activation of the system, the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

NOTE 1: Depending on the type of codec or signal processing the test signal used may need to be adapted.

NOTE 2: When using HATS, due to the distortion limit of the artificial mouth at 100 Hz, as defined in table 10 of Recommendation ITU-T P.58 [35], the measurements with a frequency of 100 Hz and possibly 200 Hz have to take into account the actual distortion of the artificial mouth.

7.5.9.5.2 Signal to harmonic distortion for higher input level

Requirement

For the signal defined in the measurement method, the signal to harmonic distortion ratio shall be ≥ 30 dB.

Measurement method

The terminal is set according to clause 6.10.4.

The signal used is an activation signal followed by a series sine wave signal with a frequency at 1 kHz. The signal to harmonic distortion ratio is measured selectively up to 14 kHz.

The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be calibrated to +10 dBPa at the MRP.

For a correct activation of the system, the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

NOTE: Depending on the type of codec or signal processing the test signal used may need to be adapted.

7.5.9.6 Receiving distortion

Requirements

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.66q: Receive distortion for super-wideband

Frequency	Signal to distortion ratio limit, receive for desktop terminal
200 Hz	20 dB
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
5 kHz	30 dB
8 kHz	30 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale.	

In low frequencies, GAT and telemeeting terminals should have higher signal to noise ratio than desktop terminals, to guarantee a better use of these terminals.

The requirements defined above apply to speech transmission only. Higher values for signal to distortion ratio are for terminals, also intended to transmit audio signals, e.g. music. This is for further study.

Measurement method

The terminal is set according to clause 6.10.4.

The signal used is an activation signal followed by a sine wave signal with a frequency at 100 Hz, 200 Hz, 315 Hz, 400 Hz, 500 Hz, 800 Hz, 1 kHz, 2 kHz, 5 kHz and 8 kHz. The duration of the sine wave shall be of less than 1 s. Appropriate signals for activation and signal combinations can be found in Recommendation ITU-T P.501 [41]. The sinusoidal signal level shall be -16 dBm0.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. Level of this activation signal shall be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 14 kHz.

The ratio of signal to harmonic distortion shall be measured at the DRP of the artificial ear with a correction by the curve of reference microphone.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.9.7 Sending noise

Requirement

The limit for the send noise shall be the following:

- send noise level maximum -64 dBm0(A).

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

NOTE 1: Softphones with cooling devices (fans) can produce a rather high level of noise; largely dependent of the activity of system.

Measurement method

The terminal is set-up according to clause 6.10.4.

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 the female speaker of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41]. The level of this activation signal is -4,7 dBPa, measured at the MRP. The activation signal level is averaged over the complete activation signal sequence.

The send noise is measured at the POI in the frequency range from 50 Hz to 14 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).

Spectral peaks are measured in the frequency domain in the frequency range from 100 Hz to 14 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE 2: In case spectral peaks higher than 10 dB above the average noise floor are produced by the terminal, but which are considered to be inaudible due to the very low noise floor produced by the terminal on average, it is the responsibility of the test lab to demonstrate the desired performance subjectively.

7.5.9.8 Receiving noise

Requirement

A-weighted

- The noise level measured until 10 kHz shall not exceed -54 dBPa(A) at nominal setting of the volume control.

Third-octave band spectrum.

- The level in any 1/3-octave band, between 50 Hz and 12,5 kHz shall not exceed a value of -64 dBPa.

- No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

NOTE 1: For softphone, fan noise should be avoided in order to fulfill this condition.

Measurement method

The terminal is set-up according to clause 6.10.4.

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 the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41]. The level of this activation signal will be -16 dBm0.

For the A-weighted noise level measurement the noise level shall be measured at DRP of the artificial ear until 14 kHz. Freefield equalization shall be used.

For the 1/3rd octave band spectrum the level is measured in all the 1/3-octave bands between 50 Hz and 12,5 kHz.

The noise shall be measured just after interrupting the activation signal.

Spectral peaks are measured in the frequency domain in the frequency range from 50 Hz to 14 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3 octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE 2: In case spectral peaks higher than 10 dB above the average noise floor are produced by the terminal, but which are considered to be inaudible due to the very low noise floor produced by the terminal on average, it is the responsibility of the test lab to demonstrate the desired performance subjectively.

NOTE 3: Care should be taken that only the noise is windowed out by the analysis and the analysis window is not impaired by any remaining reverberance or room noise.

7.5.9.9 Terminal Coupling Loss

7.5.9.9.1 Unweighted Terminal Coupling Loss

Requirement

The TCL measured as unweighted Echo Loss shall be ≥ 55 dB for all positions of the volume control (if supplied).

Measurement method

The handsfree terminal is set-up as 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).

The ambient noise level shall be < -64 dBPa(A). The attenuation from electrical reference point input to electrical reference point output shall be measured using the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41].

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41].

TCL is calculated as the difference between the averaged test signal level and the averaged echo level in the frequency range from 100 Hz to 8 kHz. 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 first 17,0 s of the test signal (6 sentences) are discarded from the analysis to allow for convergence of the acoustic echo canceller. The analysis is performed over the remaining length of the test sequence (last 6 sentences).

For the measurement, a time window has to be applied which is adapted to the duration of the actual test signal. The echo loss is calculated by the equations:

$$L_e = C - 10\log_{10}\sum_{i=1}^{N}(A_i + A_{i-1})(\log_{10} f_i - \log_{10} f_{i-1}) \quad (7.7)$$

and

$$C = 10\log_{10}(2\,(\log_{10} f_N - \log_{10} f_0)) \quad (7.8)$$

where:

- A_0 is the output/input power ratio at frequency f_0 = 100 Hz;
- A_1 the ratio at frequency f_i; and
- A_N the ratio at frequency f_N = 8 000 Hz.

Equation (7.7) is a generalized form of the equation defined in Recommendation ITU-T G.122 [16], clause B.4, for calculating echo loss based on tabulated data, which allows the calculation of echo loss within any frequency range between f_0 and f_N.

7.5.9.9.2 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 shall 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 TCL.

For loudspeaking mode handset is placed at 50 cm beside terminal with transducers facing the table (see figure 7.18n).

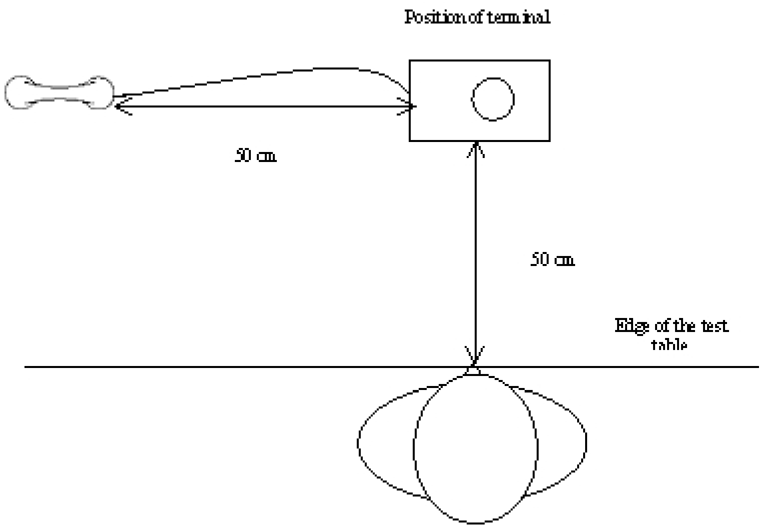


Figure 7.18n: Stability loss position for loudspeaking function

7.5.9.10 Double Talk Performance

7.5.9.10.0 General

During double talk the speech is mainly determined by two 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 Recommendations ITU-T P.340 [39] and P.502 [42]):

- Attenuation range in send direction during double talk $A_{H,S,dt}$.
- Attenuation range in receive direction during double talk $A_{H,R,dt}$.
- Echo attenuation during double talk.

7.5.9.10.1 Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$

Requirement

Based on the level variation in send direction during double talk $A_{H,S,dt}$ the behaviour of the terminal shall be classified according to table 7.66r.

Table 7.66r

Category (according to Recommendation Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.66s 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.

This measurement shall be repeated for the desktop type terminals and softphones with variable echo path.

Measurement method

The test setup is described in clause 6.10.4.

To assess double talk performance, the signals to be used are defined in Recommendation ITU-T P.501 [41]: The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. A "double-talk" sequence representing typical double talk scenarios in real conversations is shown in figure 7.18o. This uses the single-talk sequence described in clause 7.3.1 of Recommendation ITU-T P.501 [41], shown in the lower pane, as the main speech and an additional competing speaker sequence, shown in the upper pane.

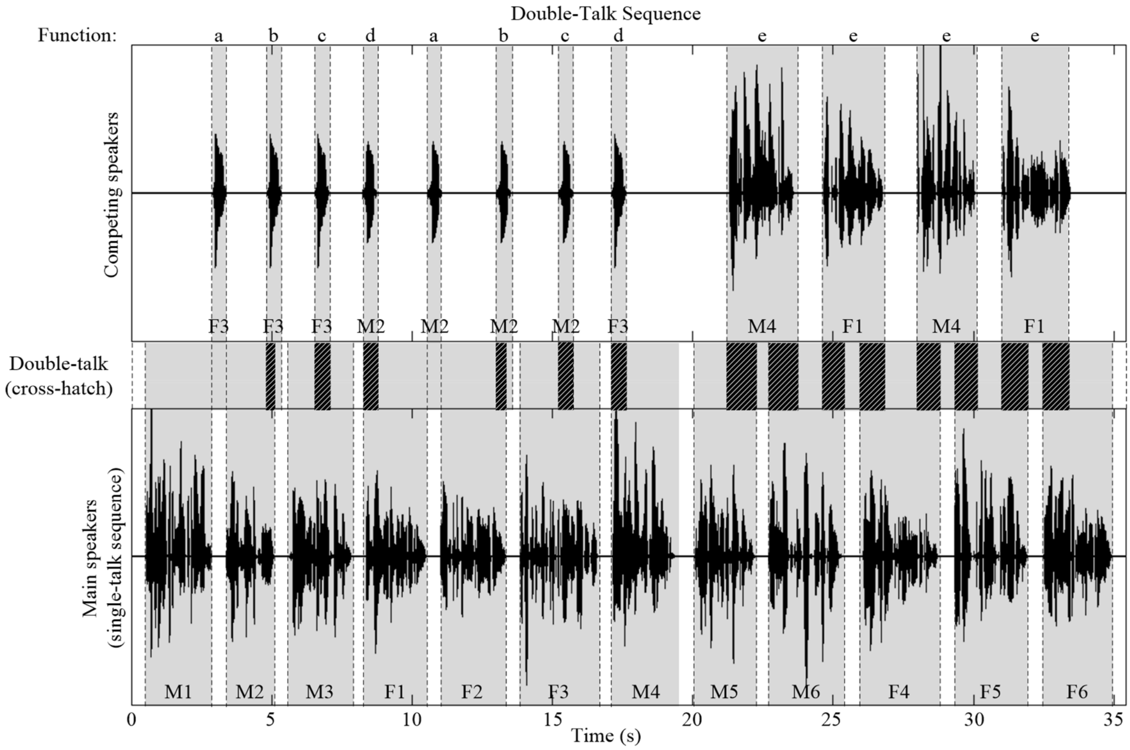


Figure 7.18o: Double-talk test sequence using the single-talk sequence and competing speech serving different functions (a - e)

NOTE: Cross-hatched areas between the upper and lower panes show periods of double talk.

The competing-speaker sequence includes single words (the word "five") spoken by speakers F3 and M2 during the first half of the sequence followed by full sentences by speakers F1 and M4 during the second half of the sequence. No speaker is competing with themselves during the sequence.

The competing samples serve different double-talk functions, defined as functions "a" to "e" above the upper pane of figure 7.18o. The functions are:

a) Competing word within a speech pause.

b) Competing word partially masked.

c) Competing word fully masked within a sentence.

d) Competing word fully masked coincident with the start of a sentence.

e) Sentence masking another sentence.

These are meant to represent possible double-talk situations in normal conversation. The area between the upper and lower pane of figure 7.18o shows the periods during which double-talk happens as cross-hatched patches. The competing sequence can be used either as a send signal or a receive signal in testing.

7.5.9.10.2 Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$

Requirement

Based on the level variation in receive direction during double talk $A_{H,R,dt}$ the behaviour of the terminal shall be classified according to table 7.66s.

Table 7.66s

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.66s 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 setup is described in clause 6.10.4.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is shown in clause 7.5.9.10.1. The competing speaker is always inserted as the double talk sequence in receive direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequence of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

7.5.9.10.3 Detection of Echo Components during Double Talk

Requirement

"Echo Loss" (EL) is the echo suppression provided by the terminal measured at the electrical reference point. Under these conditions the requirements given in table 7.66t are applicable (more information can be found in Annex A of Recommendation ITU-T P.340 [39].

Table 7.66t

Category (according to Recommendation ITU-T 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).

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

This measurement shall be repeated for desktop type terminals and softphones with variable echo path.

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. A detailed description can be found in Recommendation ITU-T P.501 [41].

The signals are fed simultaneously in send and receive direction. The level in send direction is -4,7 dBPa at the MRP (nominal level), the level in receive direction is -16 dBm0 at the electrical reference point (nominal level).

The settings for the signals are as follows.

Table 7.66u: Parameters of the two test signals for double talk measurement based on AM-FM modulated sine waves

Send Direction		Receive Direction	
$f_0^{(1)}$ [Hz]	$\pm\Delta f^{(1)}$ [Hz]	$f_0^{(2)}$ [Hz]	$\pm\Delta f^{(2)}$[Hz]
125	±2,5	180	±2,5
250	±5	270	±5
500	±10	540	±10
750	±15	810	±15
1 000	±20	1 080	±20
1 250	±25	1 350	±25
1 500	±30	1 620	±30
1 750	±35	1 890	±35
2 000	±40	2 160	±35
2 250	±40	2 400	±35
2 500	±40	2 650	±35
2 750	±40	2 900	±35
3 000	±40	3 150	±35
3 250	±40	3 400	±35
3 500	±40	3 650	±35
3 750	±40	3 900	±35
4 000	±40	4 150	±35
4 250	±40	4 400	±35
4 500	±40	4 650	±35
4 750	±40	4 900	±35
5 000	±40	5 150	±35
5 250	±40	5 400	±35
5 500	±40	5 650	±35
5 750	±40	5 900	±35
6 000	±40	6 150	±35
6 250	±40	6 400	±35
6 500	±40	6 650	±35
6 750	±40	6 900	±35
7 000	±40		
NOTE: Parameters of the Shaping Filter: f ≥ 250 Hz: Low Pass Filter, 5 dB/oct.			

The test signal is measured at the electrical reference point (send 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 receive direction (see Recommendation ITU-T P.501 [41]). The filter will suppress frequency components of the double talk signal.

In each frequency band which is used in receive 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 7.66u. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

7.5.9.10.4 Minimum activation level and sensitivity of double talk detection

For further study.

7.5.9.11 Switching characteristics

7.5.9.11.0 Note

Additional requirements may be needed in order to further investigate the effect of NLP implementations on the users' perception of speech quality.

7.5.9.11.1 Activation in Sending Direction

The activation in send 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 send 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) shall be ≤ 15 ms.

Measurement method

The test setup is described in clause 6.10.4.

The test signal is the activation of the short conditioning sequence described in clause 7.3.4 of Recommendation ITU-T P.501 [41] with increasing level for each single word.

The settings of the test signal are as follows.

Table 7.66v: Test file settings

	Single word/ Pause Duration	Level of the first single word (active Signal Part at the MRP)	Level Difference between two Periods of the Test Signal
Single word to Determine Switching Characteristic **in Send Direction**	~600 ms / ~400 ms	-24 dBPa (see note)	1 dB
NOTE: The signal level is determined for each utterance individually according to Recommendation ITU-T P.56 [33].			

It is assumed that the pause length of about 400 ms is longer than the hang-over time so that the test object is back to idle mode after each single word.

The level of the transmitted signal is measured at the electrical reference point. The test signal is filtered by the transfer function of the test object. The measured signal level is referred to the filtered 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 single word which indicates the first activation of the test object. The time between the beginning of the single word and the complete activation of the test object is measured.

7.5.9.11.2 Silence Suppression and Comfort Noise Generation

Requirements and measurement methods are for further study.

NOTE: In general it is not recommended to use silence suppression at all.

7.5.9.11.3 Performance in sending direction in the presence of background noise

Requirement

The level of comfort noise shall be within in a range of +2 dB to -5 dB compared to the original (transmitted) background noise. The noise level is calculated with A-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 7.66w.

Table 7.66w: Requirements for spectral adjustment of comfort noise (mask)

Frequency	Upper limit	Lower limit
200 Hz	12 dB	-12 dB
800 Hz	12 dB	-12 dB
800 Hz	10 dB	-10 dB
2 000 Hz	10 dB	-10 dB
2 000 Hz	6 dB	-6 dB
4 000 Hz	6 dB	-6 dB
14 000 Hz	6 dB	-6 dB
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 handsfree is set-up as described in clause 6.10.4.

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 receive direction consisting of an initial pause of 10 s and a periodical repetition of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] in receive 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.

The transmitted signal is recorded in send direction at the POI.

The power density spectra measured in send direction without far end speech simulation averaged between 10 s and 20 s is referred to the power density spectrum measured in send direction determined during the period with far end speech simulation in receive direction averaged between 10 s and 20 s. Level and spectral differences between both power density spectra are analysed and compared to the requirements.

7.5.9.11.4 Speech Quality in the Presence of Background Noise

Requirement

For the background noises defined in clause 6.10.6 the following requirements shall apply:

- N-MOS-$LQO_f \geq 2,7$.
- S-MOS-$LQO_f \geq 3,3$.
- G-MOS-$LQO_f \geq 2,7$.

NOTE: It is recommended to test the terminal performance with other types of background noises if the terminal is likely to be exposed to other noises than specified in clause 6.10.6.

Measurement method

The background noise simulation as described in clause 6.10.6 is used. The handsfree terminal is set-up as described in clause 6.10.4.

The background noise shall be applied for at least 5 s in order to adapt noise reduction algorithms in advance the test.

The near end speech signal consists of 16 sentences of speech (4 male and 4 female talkers, 2 sentences each). The American English speech samples from ETSI TS 103 281 [64] are used. The test signal level is -1,7 dBPa at the MRP.

The following signals are required for the tests:

1) The clean speech signal is used as the undisturbed reference (see ETSI TS 103 281 [64]).

2) The send signal is recorded at the electrical reference point.

N-MOS-LQO_F, S-MOS LQO_F and G-MOS LQO_F are calculated as described in ETSI TS 103 281 [64]. Either Model A or Model B can be used. The model chosen shall be documented in the test report.

When using model A the following mapping functions apply:

$$\text{S-MOS}'_{\text{LQOf}} = 1{,}418 \cdot \text{S-MOS}_{\text{LQOf}} - 1{,}145$$

$$\text{N-MOS}'_{\text{LQOf}} = 1{,}346 \cdot \text{N-MOS}_{\text{LQOf}} - 1{,}584$$

$$\text{G-MOS}'_{\text{LQOf}} = 1{,}279 \cdot \text{G-MOS}_{\text{LQOf}} - 0{,}7364$$

7.5.9.11.5 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out applying a speech signal in receive direction and by comparing the noise level transmitted in the send direction under reference conditions with no far end speech to the noise level transmitted in the send direction under test conditions including far end speech. During and after the end of the speech signal the signal level in send direction shall 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 clause 6.10.6.

NOTE: The intention of this measurement is to detect impairments (modulations, switching and others) influencing the background noise transmitted from the terminal under test when a signal from the distant end (receiving side of the terminal under test) is present. Under these test conditions no modulation of the transmitted signal should occur. Modulation, switching or other type of impairments might be caused by an improper behaviour of a nonlinear processor working in conjunction with the echo canceller and erroneously switching or modulating the transmitted background noise.

Measurement method

The test arrangement is according to 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 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 speech signal at the far end. The exactly identical background noise signal is applied. The background noise signal shall start at the same point in time as was used for the reference measurement without far end signal. The background noise shall be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms. After at least 5 s a series of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] is applied in receive direction with duration of at least 10 s. The test signal level in receive direction is -16 dBm0 at the electrical reference point.

For both reference and test conditions the send 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 send direction is determined during the time interval when the speech signal is applied and after it stops. The level difference is determined from the difference of the recorded signal levels versus time between reference signal and the signal measured with far end signal.

7.5.9.12 Quality of echo cancellation

7.5.9.12.1 Temporal echo effects

Requirements

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk. The measured echo attenuation during single talk shall not decrease by more than 6 dB from the maximum measured echo attenuation.

Measurement method

The test setup is described in clause 6.10.4.

The test signal consists of periodically repeated Composite Source Signal according to Recommendation ITU-T 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 seconds which represents 8 periods of the CSS 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.

The difference between the maximum attenuation and the minimum attenuation is measured.

NOTE 1: In addition tests with more speech like signals should be made, e.g. Recommendation ITU-T P.501 [41] to see time variant behaviour of EC. However for such tests the simple broadband attenuation based test principle as described above cannot be applied due to the time varying spectral content of the speech like signals.

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 (35 ms) of the level analysis taking into account the exponential character of the integration time in any tolerance scheme.

NOTE 3: Care should be taken not to confuse noise or comfort noise with residual echo. In cases of doubt the measured echo signal should be compared to the residual noise signal measured under the same conditions without inserting the receive signal. If the level vs. time analysis leads to the identical result it can be assumed that no echo but just comfort noise is present.

7.5.9.12.2 Spectral Echo Attenuation

Requirements

The echo attenuation vs. frequency shall be below the tolerance mask given in table 7.66x.

Table 7.66x: Spectral echo loss limits

Frequency	Limit
100 Hz	-41 dB
1 300 Hz	-41 dB
3 450 Hz	-46 dB
5 200 Hz	-46 dB
7 500 Hz	-37 dB
8 000 Hz	-37 dB
NOTE: 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 send 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 consisting of the compressed real speech signal is described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The level of the training sequence shall be -16 dBm0.

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The measurement is carried out under steady-state conditions. The average test signal level shall be -16 dBm0, averaged over the complete test signal. 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.

7.5.9.12.3 Variable echo path

Requirement:

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk with dynamic changing echo paths.

No level peak shall be more than 10 dB above the minimum noise level during the measurement.

Measurement method:

The test setup is described in clause 6.10.7.

NOTE: Care should be taken to not generate noise during the movement of the notebook lid. Because of this, this measurement is not applicable for a softphone without external microphone.

As test signal the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is used. The signal level shall be -10 dBm0. The terminal volume control is set to nominal RLR. The first 4 sentences of the test signal are used to allow full convergence of the echo canceller. The next 4 sentences (from 10,75 s to 22,5 s) are used for the analysis. The echo signal level is analysed over time. The echo signal level is analysed for 11,75 s, using a time constant of 35 ms.

The measurement result is displayed as echo level versus time.

7.5.9.13 Delay

Technically, this PP type can only be connected to an FP of type 5 (Fixed Part with VoIP interface, wideband, super-wideband, fullband, FBHR or ultra-band service), i.e. a VoIP interface. Hence the requirements and measurement methods given in clause 7.7.1 for the combined FP+PP roundtrip delay apply.

Figure 7.18o1: Void

Figure 7.18o2: Void

7.5.9.14 Send Loudness Level

Requirements:

The nominal value of Send Loudness Level (SLL) shall be:

SLL = 71 phon ± 4 phon

Measurement method:

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The terminal is set-up as described in clause 6.10.4 and calibration is realized as explained in clause 6.10.4.

The loudness (in sone) of the recorded signal is calculated according to ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is calculated according to clause 9 of ITU-T P.700 [74].

7.5.9.15 Receive Loudness Level

Requirements:

- *Desktop operated PP:*
  - The nominal value of Receive Loudness Level (RLL) shall be 71 ± 4 phon. This value has to be fulfilled for at least one volume setting.
  - The value of RLL at maximum volume setting shall be louder than or equal to 78 phon.
  - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 15 phon.
- *Handheld operated* PP*:*
  - The nominal value of Receive Loudness Level (RLL) shall be 67 ± 4 phon. This value has to be fulfilled for at least one volume setting.
  - The value of RLL at maximum volume setting shall be louder than or equal to 71 phon.
  - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 15 phon.
- Softphone (computer-based terminal):
  - Type 1 or softphone with external speakers: requirement as for desktop terminal.
  - Type 2: requirement as for handheld terminal.
- Group audio terminal*:*
  - The nominal value of Receive Loudness Level (RLL) shall be 71 ± 4 phon. This value has to be fulfilled for at least one volume setting.
  - The value of RLL at maximum volume setting shall be louder than or equal to 82 phon.
  - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 19 phon.

Measurement method:

The terminal is set-up as described in clause 6.10.4.

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

For each recorded artificial ear signal, the loudness (in sone) of the recorded signal is calculated according to ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is determined as follows for binaural measurements: the resulting loudnesses for left and right ears are first halved (individual loudness per ear). Both loudnesses are added (assuming perfect loudness summation). With this overall loudness, the overall loudness level is finally determined according to clause 8.2 of ITU-T P.700 [74].

7.5.10 Transmission characteristics for PP type 7a ("fullband 20 kHz handset or headset")

7.5.10.1 Frequency responses

7.5.10.1.1 Sending

7.5.10.1.1.1 Send frequency response - nominal position

Requirements

The send frequency response of the handset or headset shall be within a mask as defined in table 7.66y and shown in figure 7.18p. This mask shall be applicable for all types of handsets and headsets.

Table 7.66y: Fullband send frequency response limits

Frequency	Upper Limit	Lower Limit
20 Hz	0 dB	
50 Hz	0 dB	-10 dB
100 Hz	5 dB	-5 dB
12 500 Hz	5 dB	-5 dB
16 000 Hz	5 dB	-5 dB
20 000Hz	5 dB	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale. The requirement is based on 1/12th octave measurement.		

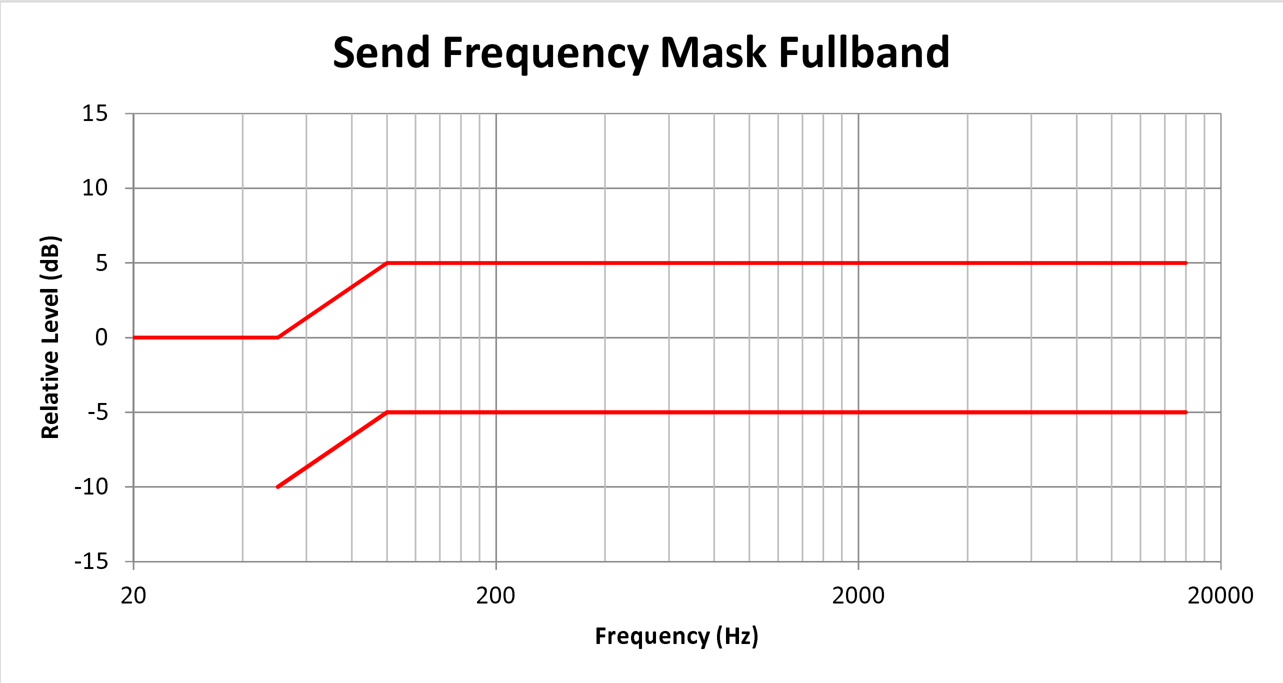


Figure 7.18p: Send frequency response mask for Fullband

NOTE 1: The basis for the target frequency responses in sending and receiving is the orthotelefonic reference response which is measured between two 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 free-field. With the concept of free-field based receive measurements a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a free 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.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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 standard position of the HATS. 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 Recommendation ITU-T 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 IEC 61260-1 [45] for frequencies from 20 Hz to 20 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 at the MRP.

The sensitivity is expressed in terms of dBV/Pa.

7.5.10.1.1.2 Send frequency response - positional robustness

Requirement

For each of the modified handset positions, the send frequency response shall be within a given mask. The mask values per frequency are identical to table 7.66y, except that an additional tolerance is provided for certain positions. Table 7.66z provides the offset in dB for the lower limit.

Table 7.66z: Tolerance mask offsets for send frequency response

Position	Offset Lower Limit
UP	-1 dB
DOWN	-2 dB
AWAY	-1 dB

Measurement method

The test arrangement and measurement is identical to clause 7.5.10.1.1.1. Instead of the standard handset position, the three modified positions according to table 6.3 for sending direction shall be used. The resulting three frequency responses shall be reported separately for each position.

7.5.10.1.2 Receiving

7.5.10.1.2.1 Receive frequency response - nominal position

Requirements

The receive frequency response of the handset or the headset shall be within a mask as defined in table 7.66aa and shown in figure 7.18q.

Table 7.66aa: Fullband receive frequency response limits

Frequency	Upper Limit	Lower Limit
20 Hz	3 dB	-15 dB
50 Hz	3 dB	-5 dB
400 Hz	3 dB	-5 dB
1 010 Hz	(see note)	-5 dB
1 200 Hz	(see note)	-8 dB
1 500 Hz	(see note)	-8 dB
2 000 Hz	9 dB	-3 dB
3 200 Hz	9 dB	-3 dB
20 000 Hz	9 dB	-15 dB
NOTE: 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. It is a floating or "best fit" mask. The requirement is based on 1/12th octave measurement.		

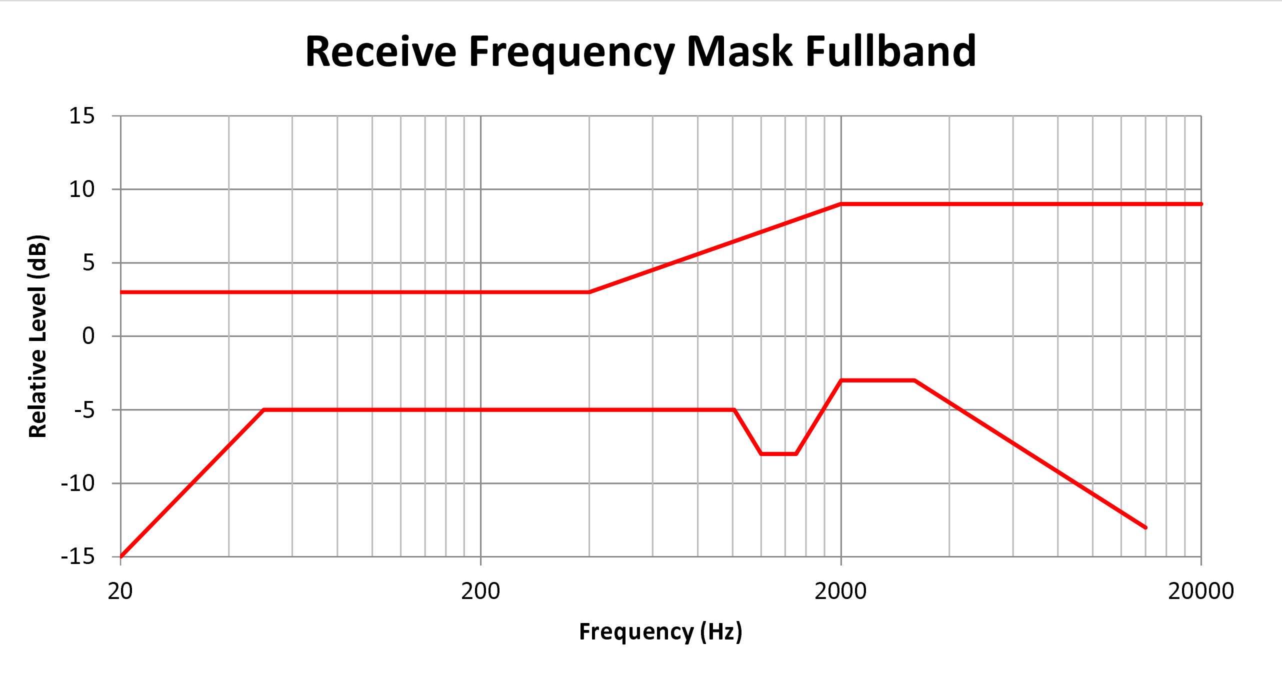


Figure 7.18q: Receive frequency response mask for Fullband

NOTE 1: This requirement applies to headphones not primarily designed for fullband communication but rather for music audition. It is the reason of rather open limits. In the next future, new limits will be discussed to apply when specially designed for fullband headphones will be available. Hence ETSI TS 102 924 [60] should be checked regularly for updates on this topic.

NOTE 2: A gap, defined in annex G, for the lower mask limit, is allowed.

Measurement method

Receive frequency response is the ratio of the measured sound pressure and the input level.
(dB relative Pa/V).

$$S_{Jedf} = 20 \log (pe_{df} / v_{RCV}) \text{ dB rel 1 Pa / V}$$

S_{Jedf} Receive Sensitivity; Junction to HATS Ear with diffuse-field correction.

pe_{df} 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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -16 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard HATS position. The application forces used to apply the handset against the artificial ear is 2 N, 8 N and 13 N.

The sound pressure level is measured at the DRP of the HATS for each 1/12th octave band.

In case of headset measurements the tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

The HATS is diffuse-field equalized as described in Recommendation ITU-T P.581 [43]. The diffuse-field correction as defined in Recommendation ITU-T P.58 [35] is applied.

Measurements shall be made at one twelfth-octave intervals as given by the IEC 61260-1 [45] for frequencies from 20 Hz to 20 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.

7.5.10.1.2.2 Receive frequency response - positional robustness

Requirement

For each of the modified handset positions, the receive frequency response shall be within a given mask. The mask values per frequency are identical to table 7.66aa, except that an additional tolerance is provided for certain positions. Table 7.66ab provides the offset in dB for the lower limit.

Table 7.66ab: Tolerance mask offsets for receive frequency response

Position	Offset Lower Limit
Ye_{-5} Ze_{-5}	-1 dB
Ye_{0} Ze_{+5}	-1 dB
Ye_{+5} Ze_{-5}	-1 dB

Measurement method

The test arrangement and measurement is identical to clause 7.5.10.1.2.1. Instead of the standard handset position, the three modified positions according to table 6.4 for receiving direction shall be used. The resulting three frequency responses shall be reported separately for each position.

7.5.10.2 Send and receive loudness ratings

7.5.10.2.1 Send Loudness Rating

7.5.10.2.1.1 Send Loudness Rating - nominal position

Requirements

The nominal value of Send Loudness Rating (SLR) shall be:

SLR(set) = 8 dB ± 3 dB

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T 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 standard HATS position. 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 Recommendation ITU-T P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

The sending sensitivity shall be calculated for each of the 20 frequency bands given in table 1 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37] see annex A.

7.5.10.2.1.2 Send Loudness Rating - positional robustness

Requirement

The difference (in dB) between the SLR measured in each of the three modified handset positions and the one in determined standard position (STD) shall be in the range of -3 to +3 dB.

Measurement method

In addition to the test setup and measurement of clause 7.5.10.2.1.1, each of the three modified handset positions for sending direction according to table 6.3 shall be applied. SLR and delta-SLR values should be calculated and reported for each position.

7.5.10.2.1.3 Microphone mute

Requirement

The SLR (Send Loudness Rating) with microphone mute on shall be at least 50 dB higher than with microphone mute off.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 7.5.10.2.1.1, but its microphone shall be configured to be muted.

Measurement and calculation method are the same as in clause 7.5.10.2.1.1.

7.5.10.2.2 Receive Loudness Rating

7.5.10.2.2.0 Note

NOTE: Only requirements for monaural reproduction are provided, stereo/dichotic reproduction is for further study.

7.5.10.2.2.1 Receive Loudness Rating (RLR) - nominal position

Requirements

When terminal implements wideband speech functions or when the fullband functions may interact with wideband terminals, the terminal shall fulfil the requirements on RLR as defined in ETSI ES 202 739 [i.10], clause 6.3.13.1.

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.

The nominal value of RLR is the RLR closest to the nominal requirement.

The minimum difference between nominal RLR and minimum (loudest, maximum volume setting) RLR shall be higher than 6 dB.

Measurement method

The test signal to be used for the measurements shall be British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T 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 or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS. 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 Recommendation ITU-T P.581 [43]. The DRP-ERP correction as defined in Recommendation ITU-T 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, 8 N is used.

In case of headset measurements the tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

The receiving sensitivity shall be calculated for each of the 20 frequency bands given in table 1 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], see annex A. No leakage correction shall be applied for the measurement.

7.5.10.2.2.2 Receive Loudness Rating (RLR) - positional robustness

Requirements

The difference (in dB) between the RLR measured in each of the three modified handset positions and the one in standard position (STD) shall be in the range -3 to +3 dB.

Measurement method

In addition to the test setup and measurement of clause 7.5.10.2.2.1, each of the three modified handset positions for receiving direction according to table 6.4 shall be applied. An application force of 8 N is used. RLR and delta-RLR values should be calculated and reported for each position.

7.5.10.2.3 Send Loudness Level

Requirements:

The nominal value of Send Loudness Level (SLL) shall be:

SLL = 75 phon ± 4 phon

Measurement method:

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The handset or headset terminal is set-up as described in clause 6.10.3.

The loudness (in sone) of the recorded signal is calculated according to ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is calculated according to clause 9 of ITU-T P.700 [74].

7.5.10.2.4 Receive Loudness Level

Requirements:

The nominal value of Receive Loudness Level (RLL) for handsets, monaural and binaural/stereo headsets shall be:

RLL = 75 phon ± 4 phon

In case a user controlled receive volume control is provided, for at least one setting of the control the RLL shall meet the nominal value.

When the control is set to maximum, the RLL shall not be louder than 89 phon. With the volume control set to the minimum position the RLL shall not be quieter than 58 phon.

Measurement method:

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The handset or headset terminal is set-up as described in clause 6.10.3.

The loudness (in sone) of the recorded signal is calculated according to ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is determined as follows:

- Handsets, monaural headsets: the loudness level is calculated according clause 8.2 of ITU-T P.700 [74] by using the loudness value divided by two (loudness halving for monaural listening).
- Binaural headsets: the loudness level is calculated according clause 8.2 of ITU-T P.700 [74] by using directly the loudness value (loudness summation for binaural listening is retained).

7.5.10.3 Sidetone

7.5.10.3.1 Sidetone Masking Rating STMR (Mouth to ear)

Requirements

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 shall not be below 8 dB.

NOTE 1: It is preferable to have a constant STMR independent of the volume control setting.

NOTE 2: STMR measurement in Recommendation ITU-T P.79 [37] is not defined above 8 kHz, but sidetone signal is not supposed to have such limitation.

Measurement method

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -4,7 dBPa, measured at the MRP. The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard HATS position and the application force shall be 13 N 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 IEC 61260-1 [45] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band (Recommendation ITU-T 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 (LmeST), as expressed in dB, and the SideTone Masking Rate (STMR) (in dB) shall be calculated from the formula 5-1 of Recommendation ITU-T P.79 [37], using m = 0,225 and the weighting factors in table 3 of Recommendation ITU-T P.79 [37].

7.5.10.3.2 Sidetone Delay

Requirements

The maximum sidetone-round-trip delay shall be ≤ 5 ms, measured in an echo-free setup.

Measurement method

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard HATS position.

The test signal is a CS-signal complying with Recommendation ITU-T 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 Recommendation ITU-T P.501 [41].

The level of the signal shall be -4,7 dBPa at the MRP.

The cross-correlation function $\Phi_{xy}(\tau)$ 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:

$$\Phi_{xy}(\tau) = \frac{1}{T}\int_{t=\frac{-T}{2}}^{\frac{T}{2}} S_x(t) \cdot S_y(t+\tau) \cdot dt$$

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 $\Phi_{xy}(\tau)$. 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 $\{\Phi_{xy}(\tau)\}$ of the cross-correlation:

$$H\{\Phi_{xy}(\tau)\} = \sum_{u=-\infty}^{\infty} \frac{\Phi_{xy}(u)}{\Pi(\tau - u)}$$

$$E(\tau) = \sqrt{\left[\Phi_{xy}(\tau)\right]^2 + \left[H\{\Phi_{xy}(\tau)\}\right]^2}$$

It is assumed that the measured sidetone delay is less than T/2.

7.5.10.4 Terminal Coupling Loss

7.5.10.4.1 Unweighted Terminal Coupling Loss

Requirements

The TCL measured as unweighted Echo Loss shall be ≥ 55 dB for all positions of the volume control (if supplied).

NOTE: Depending on the idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 dB.

Measurement method

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard HATS position and the application force shall be 2 N on the artificial ear type 3.3 or type 3.4 as specified in Recommendation ITU-T P.57 [34]. The ambient noise level shall be < -64 dBPa(A). The attenuation from electrical reference point input to electrical reference point output shall be measured using the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41].

TCL is calculated as unweighted echo loss from 100 Hz to 8 kHz. 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 first 17,0 s of the test signal (6 sentences) are discarded from the analysis to allow for convergence of the acoustic echo canceller. The analysis is performed over the remaining length of the test sequence (last 6 sentences).

For the measurement, a time window has to be applied which is adapted to the duration of the actual test signal. The echo loss is calculated by the equations:

$$L_e = C - 10\log_{10}\sum_{i=1}^{N}(A_i + A_{i-1})(\log_{10} f_i - \log_{10} f_{i-1}) \qquad (7.9)$$

and

$$C = 10\log_{10}(2\,(\log_{10} f_N - \log_{10} f_0)) \qquad (7.10)$$

where:

- A_0 is the output/input power ratio at frequency f_0 = 100 Hz;
- A_1 the ratio at frequency f_i; and
- A_N the ratio at frequency f_N = 8 000 Hz.

Equation (7.9) is a generalized form of the equation defined in Recommendation ITU-T G.122 [16], clause B.4, for calculating echo loss based on tabulated data, which allows the calculation of echo loss within any frequency range between f_0 and f_N.

7.5.10.4.2 Stability Loss

Requirements

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 20 Hz to 20 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 of the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41] is applied. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with Recommendation ITU-T 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 20 Hz to 20 kHz under the following conditions:

a) the headset, with the transmission circuit fully active, shall be positioned on one inside surface that is of three perpendicular planes, 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 7.18r;

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;

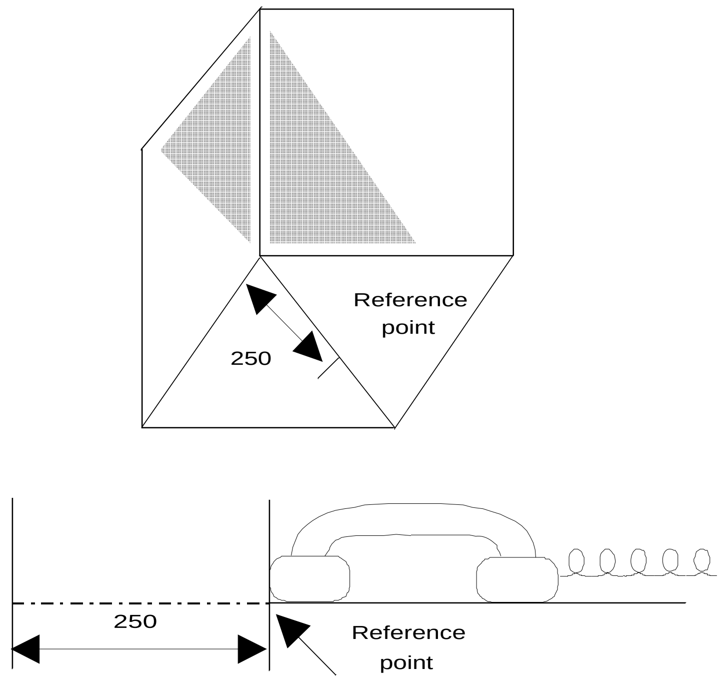
3) the extremity of the handset shall coincide with the normal to the reference point, as shown in figure 7.18r;

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 7.18r;

3) the headset microphone is positioned as close as possible to the receiver.



NOTE: All dimensions in mm.

Figure 7.18r

7.5.10.5 Distortion

7.5.10.5.1 Sending Distortion

7.5.10.5.1.1 Signal to harmonic distortion versus frequency

Requirements

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.66ac: Send distortion for fullband

Frequency	Ratio
100 Hz	26 dB
200 Hz	30 dB
400 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
3,15 kHz	30 dB
8 kHz	30 dB
NOTE: 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 or headset terminal is set-up as described in clause 6.10.3.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 100 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz, 3 150 Hz and 8 000 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 20 kHz.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.10.5.1.2 Signal to harmonic distortion for higher input level

Requirements

For the signal defined in the measurement method, the signal to harmonic distortion ratio shall be ≥ 30 dB.

Measurement method

The handset or headset terminal is set-up as described in clause 6.10.3.

The signal used is an activation signal followed by a 1 kHz sine wave. The signal to harmonic distortion ratio is measured selectively up to 20 kHz.

The duration of the sine wave shall be ≤ 1 s. The sinusoidal signal level shall be calibrated to +10 dBPa at the MRP.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.10.5.2 Receiving Distortion

Requirements

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.66ad: Receive distortion for fullband

Frequency	Signal to distortion ratio limit, receiving
50 Hz	24 dB
100 Hz	26 dB
315 Hz	30 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
3,15 kHz	30 dB
8 kHz	30 dB
NOTE: 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 or 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 50 Hz, 100 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz, 3 150 Hz and 8 000 Hz.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The signal level shall be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 20 kHz.

The ratio of signal to harmonic distortion shall be measured at the DRP of the artificial ear with a correction by the curve of reference microphone.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.10.6 Noise

7.5.10.6.1 Sending

Requirements

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 female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

The handset or headset terminal is set-up as described in clause 6.10.3.

The send noise is measured at the POI in the frequency range from 20 Hz to 20 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).

Spectral peaks are measured in the frequency domain in the frequency range from 50 Hz to 16 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE: In case spectral peaks higher than 10 dB above the average noise floor are produced by the terminal, but which are considered to be inaudible due to the very low noise floor produced by the terminal on average, it is the responsibility of the test lab to demonstrate the desired performance subjectively.

7.5.10.6.2 Receiving

Requirements

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.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement method

The handset or 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 diffuse field equalization active. The noise level is measured until 20 kHz.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The activation signal level shall be -16 dBm0.

Spectral peaks are measured in the frequency domain in the frequency range from 50 Hz to 18 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE: In case spectral peaks higher than 10 dB above the average noise floor are produced by the terminal, but which are considered to be inaudible due to the very low noise floor produced by the terminal on average, it is the responsibility of the test lab to demonstrate the desired performance subjectively.

7.5.10.7 Acoustic shock

7.5.10.7.0 General

In order to fulfil the acoustic shock requirements, it is recommended to follow the guidelines of Recommendation ITU-T P.360 [57]. If needed the PP may have to implement some kind of hardware limiter.

7.5.10.7.1 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

See ETSI EG 202 518 [i.32].

7.5.10.8 Delay

Technically, this PP type can only be connected to an FP of type 5 (Fixed Part with VoIP interface, wideband, super-wideband, fullband, FBHR or ultra-band service), i.e. a VoIP interface. Hence the requirements and measurement methods given in clause 7.7.1 for the combined FP+PP roundtrip delay apply.

Figure 7.18s: Void

Figure 7.18t: Void

7.5.10.9 Double talk Performance

7.5.10.9.0 General

During double talk the speech is mainly determined by two 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 (ELR) 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 Recommendations ITU-T P.340 [39] and P.502 [42]):

- attenuation range in send direction during double talk $A_{H,S,dt}$;
- attenuation range in receive direction during double talk $A_{H,R,dt}$;
- echo attenuation during double talk.

7.5.10.9.1 Attenuation range in send direction during double talk $A_{H,S,dt}$

Requirements

Based on the level variation in send direction during double talk $A_{H,S,dt}$ the behaviour of the terminal shall be classified according to table 7.66ae.

Table 7.66ae

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.66ae 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 handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is the double talk speech sequence as defined in clause 7.3.5 of Recommendation ITU-T P.501 [41] as shown in figure 7.18u. The competing speaker is always inserted as the double talk sequence sdt(t) in send direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

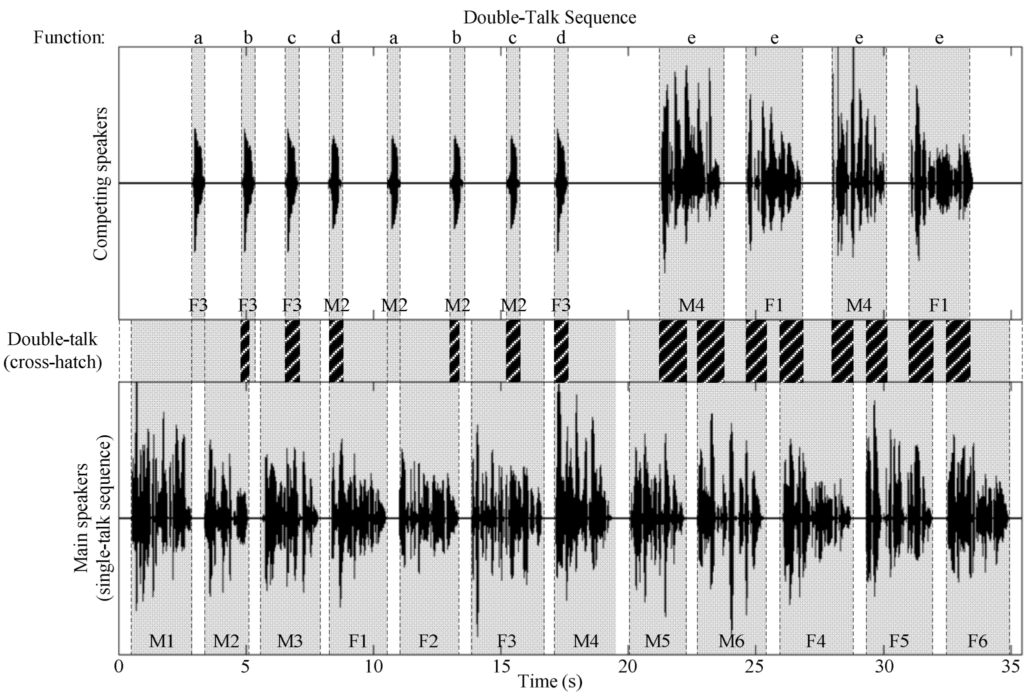


Figure 7.18u: Double talk test sequence with overlapping speech sequences in send and receive direction

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

7.5.10.9.2 Attenuation range in receive direction during double talk $A_{H,R,dt}$

Requirements

Based on the level variation in receive direction during double talk $A_{H,R,dt}$ the behaviour of the terminal shall be classified according to table 7.66af.

Table 7.66af

Category (according to Recommendation ITU-T P.340 [39])	1	2a	2b	2c	3
	Full duplex capability	Partial duplex capability			Full duplex capability
$A_{H,R,dt}$ [dB]	≤ 3	≤ 5	≤ 8	≤ 10	> 10

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.66af 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 handset terminal or the headset terminal is set-up as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is shown in figure 7.18w. The competing speaker is always inserted as the double talk sequence in receive direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequences of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

7.5.10.9.3 Detection of echo components during double talk

Requirements

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 (TELRdt). 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 table 7.66ag are applicable (more information can be found in annex A of the Recommendation ITU-T P.340 [39]).

Table 7.66ag

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

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. A detailed description can be found in Recommendation ITU-T P.501 [41].

The signals are fed simultaneously in send and receive direction. The level in send direction is -4,7 dBPa at the MRP (nominal level), the level in receive direction is -16 dBm0 at the electrical reference point (nominal level).

The settings for the signals are as follows.

Table 7.66ah: Parameters of the two test signals for double talk measurement based on AM-FM modulated sine waves

Send Direction		Receive Direction	
$f_0^{(1)}$ [Hz]	$\pm\Delta f^{(1)}$ [Hz]	$f_0^{(2)}$ [Hz]	$\pm\Delta f^{(2)}$[Hz]
125	±2,5	180	±2,5
250	±5	270	±5
500	±10	540	±10
750	±15	810	±15
1 000	±20	1 080	±20
1 250	±25	1 350	±25
1 500	±30	1 620	±30
1 750	±35	1 890	±35
2 000	±40	2 160	±35
2 250	±40	2 400	±35
2 500	±40	2 650	±35
2 750	±40	2 900	±35
3 000	±40	3 150	±35
3 250	±40	3 400	±35
3 500	±40	3 650	±35
3 750	±40	3 900	±35
4 000	±40	4 150	±35
4 250	±40	4 400	±35
4 500	±40	4 650	±35
4 750	±40	4 900	±35
5 000	±40	5 150	±35
5 250	±40	5 400	±35
5 500	±40	5 650	±35
5 750	±40	5 900	±35
6 000	±40	6 150	±35
6 250	±40	6 400	±35
6 500	±40	6 650	±35
6 750	±40	6 900	±35
7 000	±40		
NOTE: Parameters of the Shaping Filter: f ≥ 250 Hz: Low Pass Filter, 5 dB/oct.			

The test signal is measured at the electrical reference point (send 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 receive direction (see Recommendation ITU-T P.501 [41]). The filter will suppress frequency components of the double talk signal.

In each frequency band which is used in receive 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 7.66ah. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

7.5.10.10 Switching Characteristics

7.5.10.10.0 Note

NOTE: Additional requirements may be needed in order to further investigate the effect of NLP implementations on the users' perception of speech quality.

7.5.10.10.1 Activation in send direction

The activation in send 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 send 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) shall be ≤ 15 ms.

Measurement method

The handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the standard position of the HATS.

The test signal is the "short words for activation" sequence described in clause 7.3.4 of Recommendation ITU-T P.501 [41] with increasing level for each single word.

Table 7.66ai

	Single word duration / pause duration	Level of the first single word (at the MRP)	Level difference between two periods of the test signal
Single word to determine switching characteristic in send direction	~600 ms / ~400 ms	-24 dBPa (see note)	1 dB
NOTE: The signal level is determined for each utterance individually according to Recommendation ITU-T P.56 [33].			

It is assumed that the pause length of about 400 ms is longer than the hang-over time so that the test object is back to idle mode after each single word.

The level of the transmitted signal is measured at the electrical reference point. The test signal is filtered by the transfer function of the test object. The measured signal level is referred to the filtered test signal level and displayed versus time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the single word which indicates the first activation of the test object. The time between the beginning of the single word and the complete activation of the test object is measured.

7.5.10.10.2 Silence suppression and comfort noise generation

Requirements and measurement methods are for further study.

NOTE: In general it is not recommended to use silence suppression at all.

7.5.10.10.3 Performance in Sending in the Presence of Background Noise

Requirements

The level of comfort noise shall be within in a range of +2 dB to -5 dB compared to the original (transmitted) background noise. The noise level is calculated with A-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 7.66aj.

Table 7.66aj: Requirements for spectral adjustment of comfort noise (mask)

Frequency	Upper limit	Lower limit
200 Hz	12 dB	-12 dB
800 Hz	12 dB	-12 dB
800 Hz	10 dB	-10 dB
2 000 Hz	10 dB	-10 dB
2 000 Hz	6 dB	-6 dB
4 000 Hz	6 dB	-6 dB
14 000 Hz	6 dB	-6 dB
20 000 Hz	6 dB	-6 dB
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 or headset terminal is set-up as described in clause 6.10.3. The handset is mounted at the standard HATS position.

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 receive direction consisting of an initial pause of 10 s and a periodical repetition of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] in receive 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.

The transmitted signal is recorded in send direction at the POI.

The power density spectra measured in send direction without far end speech simulation averaged between 10 s and 20 s is referred to the power density spectrum measured in send direction determined during the period with far end speech simulation in receive direction averaged between 10 s and 20 s. Level and spectral differences between both power density spectra are analysed and compared to the requirements.

7.5.10.10.4 Speech quality in the presence of background noise

Requirements

For the background noises defined in clause 6.10.6 the following requirements shall apply:

- N-MOS-LQO$_f$ ≥ 3,5.
- S-MOS-LQO$_f$ ≥ 4.
- G-MOS-LQO$_f$ ≥ 3,5.

NOTE: It is recommended to test the terminal performance with other types of background noises if the terminal is likely to be exposed to other noises than specified in clause 6.10.6.

Measurement method

The background noise simulation as described in clause 6.10.6 is used. The handset or headset terminal is set-up as described in clause 6.10.3. The handset is mounted at the HATS position (see Recommendation ITU-T P.64 [36]).

The background noise should be applied for at least 5 s in order to adapt noise reduction algorithms in advance the test.

The near end speech signal consists of 16 sentences of speech (4 male and 4 female talkers, 2 sentences each). The American English speech samples from ETSI TS 103 281 [64] are used. The test signal level is -1,7 dBPa at the MRP.

The following signals are required for the tests:

1) The clean speech signal is used as the undisturbed reference (see ETSI TS 103 281 [64]).

2) The send signal is recorded at the electrical reference point.

N-MOS-LQO_f, S-MOS LQO_f and G-MOS LQO_f are calculated as described in ETSI TS 103 281 [64]. Either Model A or Model B can be used. The model chosen shall be documented in the test report.

When using model A the following mapping functions apply:

$$\text{S-MOS}'_{\text{LQOf}} = 1{,}418 \cdot \text{S-MOS}_{\text{LQOf}} - 1{,}145$$
$$\text{N-MOS}'_{\text{LQOf}} = 1{,}346 \cdot \text{N-MOS}_{\text{LQOf}} - 1{,}584$$
$$\text{G-MOS}'_{\text{LQOf}} = 1{,}279 \cdot \text{G-MOS}_{\text{LQOf}} - 0{,}7364$$

7.5.10.10.5 Positional Robustness of Speech Quality in the Presence of Background Noise

Requirement

The degradation between standard position (STD) and all other modified positions for sending direction shall not exceed the limits for S-MOS and N-MOS according to table 7.66ak. The requirements are evaluated on the averaged results over all background noises used in this test.

Table 7.66ak: Requirements for allowed degradation

Position	Δ S-MOS	Δ N-MOS
UP	≤ 0,2	≤ 0,2
DOWN	≤ 0,3	≤ 0,5
AWAY	≤ 0,3	≤ 0,4

Measurement method

The test arrangement and measurement is identical to clause 7.5.10.10.4. The test is conducted with each of the modified handset positions for sending direction according to table 6.3. All S- and N-MOS values as well as the difference to STD shall be reported for all three positions.

7.5.10.10.6 Quality of background noise transmission (with far end speech)

Requirements

The test is carried out applying a speech signal in receive direction and by comparing the noise level transmitted in the send direction under reference conditions with no far end speech, to the noise level transmitted in the send direction under test conditions including far end speech. During and after the end of the speech signal the signal level in send direction shall 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 clause 6.10.6.

NOTE: The intention of this measurement is to detect impairments (modulations, switching and others) influencing the background noise transmitted from the terminal under test when a signal from the distant end (receiving side of the terminal under test) is present. Under these test conditions no modulation of the transmitted signal should occur. Modulation, switching or other type of impairments might be caused by an improper behaviour of a nonlinear processor working in conjunction with the echo canceller and erroneously switching or modulating the transmitted background noise.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3.

The background noises are generated as described in clause 6.10.6.

First the reference measurement is conducted without inserting the signal at the far end. At least 10 s of noise is analysed. The transmitted 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 speech signal at the far end. The exactly identical background noise signal is applied. The background noise signal shall start at the same point in time as was used for the reference measurement without far end signal. The background noise shall be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms. After at least 5 s a series of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] is applied in receive direction with duration of at least 10 s. The test signal level in the receive direction is -16 dBm0 at the electrical reference point.

For both reference and test conditions, the send 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 send direction is determined during the time interval when the speech signal is applied and after it stops. The level difference is determined from the difference of the recorded signal levels versus time between reference signal and the signal measured with far end signal.

7.5.10.11 Quality of echo cancellation

7.5.10.11.1 Temporal echo effects

Requirements

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk. The measured echo attenuation during single talk shall not decrease by more than 6 dB from the maximum echo attenuation measured.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3.

The test signal consists of periodically repeated Composite Source Signal according to Recommendation ITU-T 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 versus time. The exact synchronization between input and output signal has to be guaranteed.

The difference between the maximum attenuation and the minimum attenuation is measured.

NOTE 1: In addition, it is recommended to also conduct tests with more speech like signals, e.g. Recommendation ITU-T P.501 [41] in order to investigate time variant behaviour of EC. However, for such tests the simple broadband attenuation based test principle as described above cannot be applied due to the time varying spectral content of the speech like signals.

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 (35 ms) of the level analysis taking into account the exponential character of the integration time in any tolerance scheme.

NOTE 3: Care should be taken not to confuse noise or comfort noise with residual echo. In cases of doubt the measured echo signal should be compared to the residual noise signal measured under the same conditions without inserting the receive signal. If the level vs. time analysis leads to the identical result it can be assumed that no echo but just comfort noise is present.

7.5.10.11.2 Spectral echo attenuation

Requirements

The echo attenuation versus frequency shall be below the tolerance mask given in table 7.66al.

Table 7.66al: Echo attenuation limits

Frequency	Limit
100 Hz	-41 dB
1 300 Hz	-41 dB
3 450 Hz	-46 dB
5 200 Hz	-46 dB
7 500 Hz	-37 dB
8 000 Hz	-37 dB
NOTE: 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 send direction in order to mask the echo signal.

NOTE: The measurement is performed in wideband since no fullband data are available. The extension of the method to fullband is for further study.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 6.10.3.

Before the actual measurement a training sequence consisting of the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is fed in. The level of the training sequence shall be -16 dBm0.

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The measurement is carried out under steady-state conditions. The average test signal level shall be -16 dBm0, averaged over the complete test signal. 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.

7.5.10.11.3 Variable echo path

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk with dynamic changing echo paths. The measured echo level over time during single talk shall not be more than 10 dB above the minimum noise level during the measurement.

Measurement method

The test arrangement is according to clause 6.10.7.

As test signal the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is used. The signal level shall be -10 dBm0. The terminal volume control is set to nominal RLR. The first 4 sentences of the test signal are used to allow full convergence of the echo canceller. The next 4 sentences (from 10,75 s to 22,5 s) are used for the analysis. The echo signal level is analysed over time. The echo signal level is analysed for 11,75 s, using a time constant of 35 ms.

The measurement result is displayed as echo level versus time.

7.5.11 Transmission characteristics for PP type 7b ("fullband 20 kHz loudspeaking and handsfree devices")

7.5.11.1 Sending sensitivity/frequency response

Requirements

The objective is to define a flat frequency curve over the whole bandwidth.

The frequency response shall fulfill the mask as defined in table 7.66am and figure 7.18v.

Table 7.66am: Frequency mask for fullband terminals - Send

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
20	0	
50	0	-10
100	5	-5
12 500	5	-5
16 000	5	-5
20 000	5	
NOTE: All sensitivity values are expressed in dB on an arbitrary scale. The requirement is based on 1/12th octave measurement.		

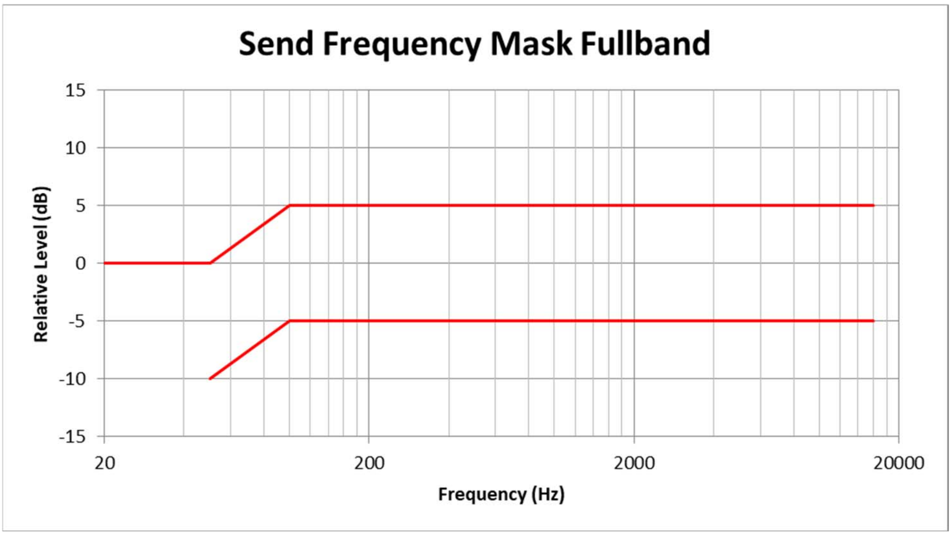


Figure 7.18v: Send frequency mask for fullband

NOTE 1: A "balanced" frequency response is preferable from the perception point of view. If frequency components in the low frequency domain are attenuated, frequency components in the high frequency domain should be attenuated in a similar way.

NOTE 2: A gap, defined in annex G, for the lower mask limit, is allowed.

Additional requirements are for further study when the system is intended to be used by several users, when stereo features are made available or when microphone array(s) are used.

Measurement method

The terminal is set-up according to clause 6.10.4. The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level is defined according to clause 6.10.4.2.

Measurements shall be made at one twelfth-octave intervals as given by IEC 61260-1 [45] for frequencies from 20 Hz to 20 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 HFRP.

The sensitivity is expressed in terms of dBV/Pa.

7.5.11.2 Receive sensitivity/frequency response

7.5.11.2.0 General

When using HATS (with the restrictions defined in clause 6.10.4) it shall be equalized according to Recommendation ITU-T P.581 [43].

However, for fullband terminals it is recommended to use free-field microphones instead of the HATS.

7.5.11.2.1 Handheld terminal

Requirements

The frequency response shall fulfill the mask as defined in table 7.66an and figure 7.18w.

Table 7.66an: Frequency mask for fullband handheld terminals - Receive

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
20	5	
400	5	-10
500	5	-5
12 500	5	-5
14 000	5	-5
16 000	5	-5
20 000	5	
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

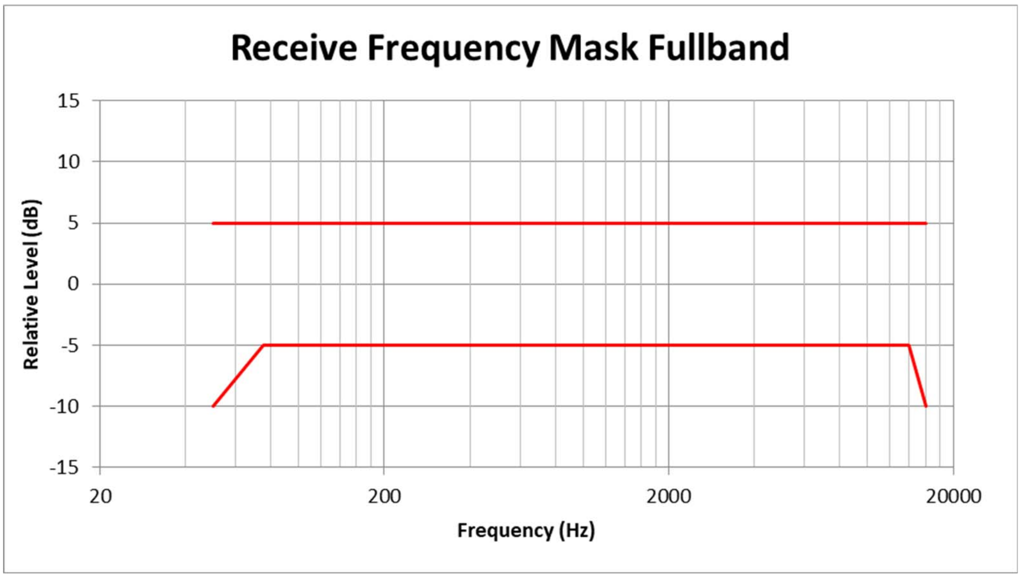


Figure 7.18w: Frequency mask for fullband handheld terminals - Receive

NOTE: A gap, defined in annex G, for the lower mask limit, is allowed.

Measurement method

The terminal is set-up according to clause 6.10.4.

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -16 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

The equalized output signal is power-averaged on the total time of analysis. The 1/3rd octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one third-octave intervals as given in ISO 3745 [67] for frequencies from 400 Hz to 16 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.

7.5.11.2.2 Desktop terminal

Requirements

The frequency response shall fulfill the mask as defined in table 7.66ao and figure 7.18x.

Table 7.66ao: Frequency mask for fullband desktop terminals - Receive

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
50	5	-10
75	5	-5
14 000	5	-5
16 000	5	-10
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

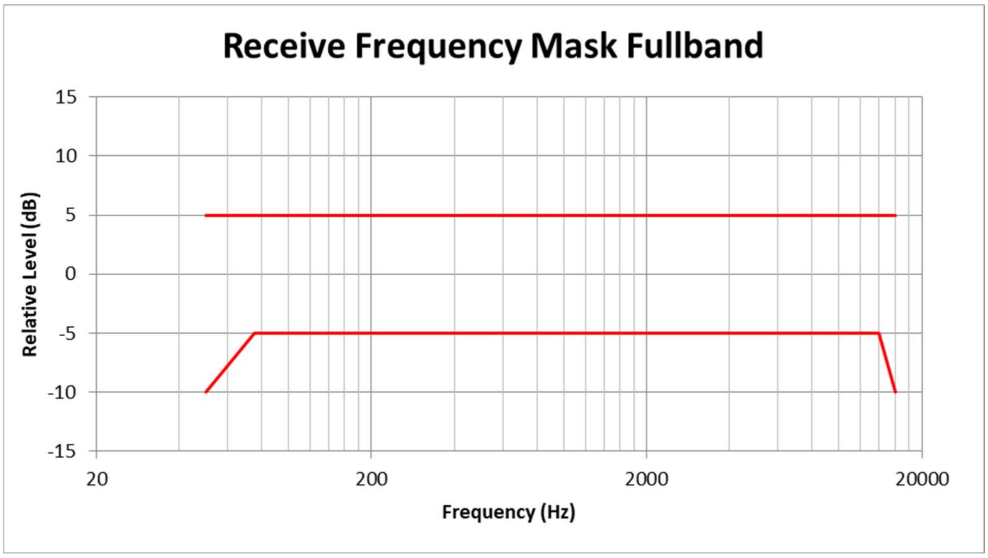


Figure 7.18x: Frequency mask for fullband terminals - Receive

NOTE: A gap, defined in annex G, for the lower mask limit, is allowed.

Measurement method

The terminal is set-up according to clause 6.10.4.

Receive frequency response is the ratio of the measured sound pressure and the input level.
(dB relative Pa/V)

$$S_{Jeff} = 20 \log (p_{eff} / v_{RCV}) \text{ dB rel 1 Pa / V}$$

S_{Jeff} Receive Sensitivity; Junction to HATS Ear with free field correction.

p_{eff} 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 British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -16 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point.

The equalized output signal is power-averaged on the total time of analysis. The 1/3rd octave band data are considered as the input signal to be used for calculations or measurements.

The HATS is free-field equalized as described in Recommendation ITU-T P.581 [43]. The free-field correction as defined in Recommendation ITU-T P.58 [35] is applied.

Measurements shall be made at one third-octave intervals as given by IEC 61260-1 [45] for frequencies from 50 Hz to 20 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.

7.5.11.2.3 Terminals intended to be used simultaneously by several users

Additional requirements to be defined:

- when the terminal is intended to be used by several users;
- when stereo features are made available.

For all the testing positions the frequency curve shall fulfil the requirements defined for desktop terminals.

7.5.11.3 Sending loudness rating

7.5.11.3.1 Nominal Value

Requirements

To ensure the compatibility with other terminals or systems a reference SLR needs to be defined.

The requirements refer to wideband handsfree terminals, ETSI ES 202 740 [i.11].

The nominal value shall be 13 dB ± 3 dB.

There is no specific requirement for FB bandwidth.

Measurement method

The terminal is positioned as described in clause 6.10.4.

For a correct activation of the system, the test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. 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.

Calibration is realized as explained in clause 6.10.4.2.

The send sensitivity shall be calculated for each of the 20 frequency bands given in table A.2 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], Annex A.

7.5.11.3.2 Microphone Mute

Requirement

The SLR (Send Loudness Rating) with microphone mute on shall be at least 50 dB higher than with microphone mute off.

Measurement method

The handset terminal or the headset terminal is set-up as described in clause 7.5.11.3.1, but its microphone shall be configured to be muted.

Measurement and calculation method are the same as in clause 7.5.11.3.1.

7.5.11.4 Receive loudness rating

Requirements

When terminal implements wideband speech functions or when the fullband functions may interact with wideband terminals, the handsfree terminal shall fulfill the requirements on RLR as defined in ETSI ES 202 740 [i.11], clause 6.3.10:

- Desktop terminal:
  - Nominal value of RLR shall be 5 dB ± 3 dB. This value has to be fulfilled for one position of volume range.
  - The value of RLR at the upper part of the volume range shall be less than (louder) or equal to -2 dB: RLR ≤ -2 dB.
  - The range of volume control shall be ≥15 dB.
- Handheld terminal:
  - Nominal value of RLR shall be 9 dB ± 3 dB. This value has to be fulfilled for one position of volume range.
  - Value of RLR at upper part of volume range shall be less than (louder) or equal to 5 dB: RLR ≤ 5 dB.
  - Range of volume control shall be equal to or exceed 15 dB.
- Softphone (computer-based terminal):
  - Type 1 or softphone with external speakers: requirement as for desktop terminal.
  - Type 2 requirement as for handheld terminal.
- Group audio terminal:
  - Nominal value of RLR shall be 5 dB ± 3 dB. This value has to be fulfilled for one position of volume range.
  - Value of RLR at upper part of volume range shall be less than (louder) or equal to -6 dB: RLR ≤ -6 dB.
  - Range of volume control shall be ≥19 dB.

NOTE 1: Due to the lack of experience in the application of wideband loudness rating calculation as defined in Annex G of Recommendation ITU-T P.79 [37] the loudness rating calculation as described in Annex A is used.

NOTE 2: Loudness Rating measurement corresponding to level with speech signal, it can be considered that a measurement in wideband may be sufficient. Indeed, energy of speech beyond bandwidth of wideband is rather small.

NOTE 3: Receive Loudness Rating for stereo/dichotic is for further study.

Measurement method

The test setup is described in clause 6.10.4.

The measurement is conducted at nominal volume control setting.

The test signal to be used for the measurements shall be British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41]. The test signal level shall be -16 dBm0, measured according to Recommendation ITU-T P.56 [33] at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

The HATS is free field equalized as described in Recommendation ITU-T P.581 [43]. The equalized output signal is power-averaged on the total time of analysis. The 1/3 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one third-octave intervals as given by IEC 61260-1 [45] for frequencies from 100 Hz to 8 kHz inclusive. The receiving sensitivity shall be calculated for each of the 20 frequency bands given in table A.2 of Recommendation ITU-T 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 Recommendation ITU-T P.79 [37], formula (A-23c), over bands 1 to 20 and the receive weighting factors from table A.2 of Recommendation ITU-T P.79 [37]. No leakage correction shall be applied.

For binaural measurements, the individual sensitivities for left and right ears are energetically summed up. The hands-free RLR based on this overall sensitivity is then calculated with a correction factor of -8 dB.

The test shall be repeated for maximum and minimum volume control setting.

7.5.11.5 Sending distortion

7.5.11.5.1 Signal to harmonic distortion versus frequency

Requirements

The ratio of signal to harmonic distortion shall be above the following masks.

The following draft requirements are defined for all the terminals within the scope of the present clause, as it is needed to ensure that any terminal intended to be used in fullband sends good quality signals. Care should be taken on the distortion of the HATS or of the loudspeaker used to test the send distortion of the terminal.

Table 7.66ap

Frequency	Ratio
100 Hz	25 dB
200 Hz	30 dB
400 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
3,15 kHz	30 dB
5 kHz	30 dB
8 kHz	30 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale.	

The signal to harmonic distortion ratio is measured selectively up to 20 kHz.

Measurement method

The terminal is set according to clause 6.10.4.

The signal used is an activation signal followed by a series sine wave signal with a frequency at 100 Hz, 200 Hz, 400 Hz, 1 kHz, 2 kHz, 3,15 kHz, 5 kHz and 8 kHz. The signal to harmonic distortion ratio is measured selectively up to 18 kHz.

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.

For a correct activation of the system, the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

NOTE 1: Depending on the type of codec or signal processing the test signal used may need to be adapted.

NOTE 2: When using HATS, due to the distortion limit of the artificial mouth at 100 Hz, as defined in table 10 of Recommendation ITU-T P.58 [35], the measurements with a frequency of 100 Hz and possibly 200 Hz have to take into account the actual distortion of the artificial mouth.

7.5.11.5.2 Signal to harmonic distortion for higher input level

Requirement

For the signal defined in the measurement method, the signal to harmonic distortion ratio shall be ≥ 30 dB.

Measurement method

The terminal is set according to clause 6.10.4.

The signal used is an activation signal followed by a series sine wave signal with a frequency at 1 kHz. The signal to harmonic distortion ratio is measured selectively up to 20 kHz.

The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be calibrated to +10 dBPa at the MRP.

For a correct activation of the system, the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

NOTE: Depending on the type of codec or signal processing the test signal used may need to be adapted.

7.5.11.6 Receiving distortion

Requirements

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.66aq: Receive distortion for fullband

Frequency	Signal to distortion ratio limit, receive for desktop terminal
100 Hz	20 dB
200 Hz	22 dB
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
5 kHz	30 dB
8 kHz	30 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale.	

In low frequencies, GAT and telemeeting terminals should have higher signal to noise ratio than desktop terminals, to guarantee a better use of these terminals.

The requirements defined above apply to speech transmission only. Higher values for signal to distortion ratio are for terminals, also intended to transmit audio signals, e.g. music. This is for further study.

Measurement method

The terminal is set according to clause 6.10.4.

The signal used is an activation signal followed by a sine wave signal with a frequency at 100 Hz, 200 Hz, 315 Hz, 400 Hz, 500 Hz, 800 Hz, 1 kHz, 2 kHz, 5 kHz and 8 kHz. The duration of the sine wave shall be of less than 1 s. Appropriate signals for activation and signal combinations can be found in Recommendation ITU-T P.501 [41]. The sinusoidal signal level shall be -16 dBm0.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for activation. Level of this activation signal shall be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 18 kHz.

The ratio of signal to harmonic distortion shall be measured at the DRP of the artificial ear with a correction by the curve of reference microphone.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.11.7 Sending noise

Requirement

The limit for the send noise shall be the following:

- send noise level maximum -64 dBm0(A).

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

NOTE 1: Softphones with cooling devices (fans) can produce a rather high level of noise; largely dependent of the activity of system.

Measurement method

The terminal is set-up according to clause 6.10.4.

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 the female speaker of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41]. The level of this activation signal is -4,7 dBPa, measured at the MRP. The activation signal level is averaged over the complete activation signal sequence.

The send noise is measured at the POI in the frequency range from 20 Hz to 20 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).

Spectral peaks are measured in the frequency domain in the frequency range from 50 Hz to 16 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE 2: In case spectral peaks higher than 10 dB above the average noise floor are produced by the terminal, but which are considered to be inaudible due to the very low noise floor produced by the terminal on average, it is the responsibility of the test lab to demonstrate the desired performance subjectively.

7.5.11.8 Receiving noise

Requirement

- A-weighted

The noise level measured until 10 kHz shall not exceed -54 dBPa(A) at nominal setting of the volume control.

- Third-octave band spectrum.

The level in any 1/3-octave band, between 50 Hz and 16 kHz shall not exceed a value of -64 dBPa.

NOTE 1: For softphone, fan noise should be avoided in order to fulfill this condition.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement method

The terminal is set-up according to clause 6.10.4.

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 the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41]. The level of this activation signal will be -16 dBm0.

For the A-weighted noise level measurement the noise level shall be measured at DRP of the artificial ear until 18 kHz. Freefield equalization shall be used.

For the 1/3 octave band spectrum the level is measured in all the 1/3-octave bands between 50 Hz and 16 kHz.

The noise shall be measured just after interrupting the activation signal.

Spectral peaks are measured in the frequency domain in the frequency range from 50 Hz to 18 kHz. The frequency spectrum of the A-weighted idle channel noise is measured by a spectral analysis having a noise bandwidth of 8,79 Hz (determined using FFT 8 k samples/48 kHz sampling rate with Hanning window or equivalent). The idle channel noise spectrum is stated in dB. A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3 octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(-1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum.

NOTE 2: In case spectral peaks higher than 10 dB above the average noise floor are produced by the terminal, but which are considered to be inaudible due to the very low noise floor produced by the terminal on average, it is the responsibility of the test lab to demonstrate the desired performance subjectively.

NOTE 3: Care should be taken that only the noise is windowed out by the analysis and the analysis window is not impaired by any remaining reverberance or room noise.

7.5.11.9 Terminal Coupling Loss

7.5.11.9.1 Unweighted Terminal Coupling Loss

Requirement

The TCL measured as unweighted Echo Loss shall be ≥ 55 dB for all positions of the volume control (if supplied).

Measurement method

The handsfree terminal is set-up as 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).

The ambient noise level shall be < -64 dBPa(A). The attenuation from electrical reference point input to electrical reference point output shall be measured using the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41].

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41].

TCL is calculated as the difference between the averaged test signal level and the averaged echo level in the frequency range from 100 Hz to 8 kHz. 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 first 17,0 s of the test signal (6 sentences) are discarded from the analysis to allow for convergence of the acoustic echo canceller. The analysis is performed over the remaining length of the test sequence (last 6 sentences).

For the measurement, a time window has to be applied which is adapted to the duration of the actual test signal. The echo loss is calculated by the equations:

$$L_e = C - 10\log_{10}\sum_{i=1}^{N}(A_i + A_{i-1})(\log_{10} f_i - \log_{10} f_{i-1}) \quad (7.11)$$

and

$$C = 10\log_{10}(2\,(\log_{10} f_N - \log_{10} f_0)) \quad (7.12)$$

where:

- A_0 is the output/input power ratio at frequency f_0 = 100 Hz;
- A_1 the ratio at frequency f_i; and
- A_N the ratio at frequency f_N = 8 000 Hz.

Equation (7.11) is a generalized form of the equation defined in Recommendation ITU-T G.122 [16], clause B.4, for calculating echo loss based on tabulated data, which allows the calculation of echo loss within any frequency range between f_0 and f_N.

7.5.11.9.2 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 shall 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 TCL.

For loudspeaking mode handset is placed at 50 cm beside terminal with transducers facing the table (see figure 7.18y).

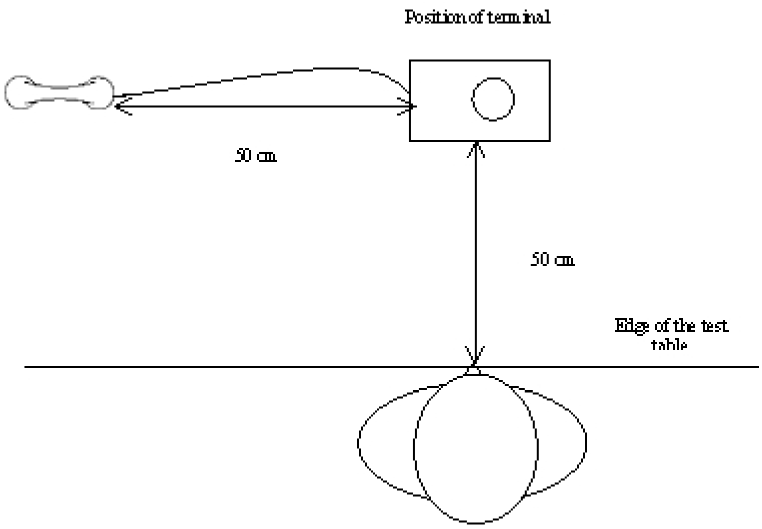


Figure 7.18y: Stability loss position for loudspeaking function

7.5.11.10 Double Talk Performance

7.5.11.10.0 General

During double talk the speech is mainly determined by two 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 Recommendations ITU-T P.340 [39] and P.502 [42]):

- Attenuation range in send direction during double talk $A_{H,S,dt}$.
- Attenuation range in receive direction during double talk $A_{H,R,dt}$.
- Echo attenuation during double talk.

7.5.11.10.1 Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$

Requirements

Based on the level variation in send direction during double talk $A_{H,S,dt}$ the behaviour of the terminal shall be classified according to table 7.66ar.

Table 7.66ar

Category (according to Recommendation Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.66ar 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.

This measurement shall be repeated for desktop type terminals and softphones with variable echo path.

Measurement method

The test setup is described in clause 6.10.4.

To assess double talk performance, the signals to be used are defined in Recommendation ITU-T P.501 [41]: The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. A "double-talk" sequence representing typical double talk scenarios in real conversations is shown in figure 7.18z. This uses the single-talk sequence described in clause 7.3.1 of Recommendation ITU-T P.501 [41], shown in the lower pane, as the main speech and an additional competing speaker sequence, shown in the upper pane.

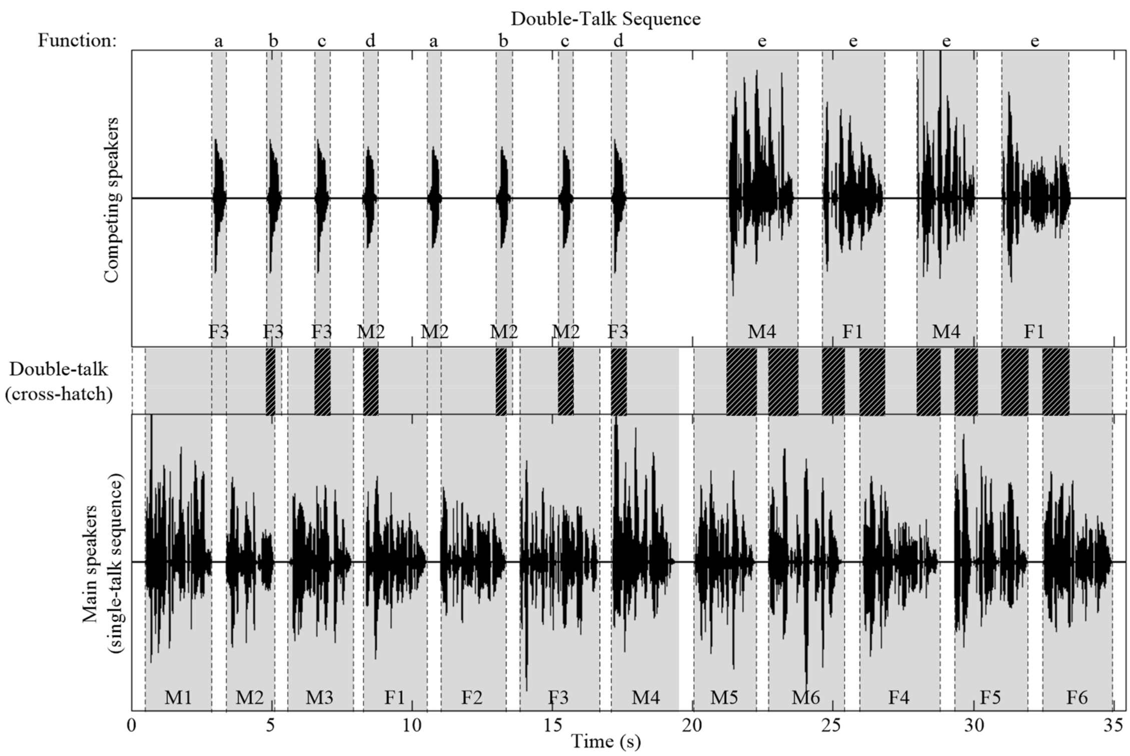


Figure 7.18z: Double-talk test sequence using the single-talk sequence and competing speech serving different functions (a - e)

NOTE: Cross-hatched areas between the upper and lower panes show periods of double talk.

The competing-speaker sequence includes single words (the word "five") spoken by speakers F3 and M2 during the first half of the sequence followed by full sentences by speakers F1 and M4 during the second half of the sequence. No speaker is competing with themselves during the sequence.

The competing samples serve different double-talk functions, defined as functions "a" to "e" above the upper pane of figure 7.18ab. The functions are:

a) Competing word within a speech pause.

b) Competing word partially masked.

c) Competing word fully masked within a sentence.

d) Competing word fully masked coincident with the start of a sentence.

e) Sentence masking another sentence.

These are meant to represent possible double-talk situations in normal conversation. The area between the upper and lower pane of figure 7.18ab shows the periods during which double-talk happens as cross-hatched patches. The competing sequence can be used either as a send signal or a receive signal in testing.

7.5.11.10.2 Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$

Requirements:

Based on the level variation in receive direction during double talk $A_{H,R,dt}$ the behaviour of the terminal shall be classified according to table 7.66as.

Table 7.66as

Category (according to Recommendation ITU-T 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

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

In general table 7.66as 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 setup is described in clause 6.10.4.

The long conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] shall be used for conditioning the terminal, with the female speaker in the receive direction. The test signal to determine the attenuation range during double talk is shown in clause 7.5.11.10.1. The competing speaker is always inserted as the double talk sequence in receive direction and is used for analysis. The test signals are synchronized in time at the acoustical interface.

The attenuation range during double talk is determined as described in Appendix III of Recommendation ITU-T P.502 [42]. The double talk performance is analysed for the sequence of words (first 20 s) and sentences (from 20 s to 35 s) produced by the competing speaker. The requirements have to be met for both sequences.

7.5.11.10.3 Detection of Echo Components during Double Talk

Requirement

"Echo Loss" (EL) is the echo suppression provided by the terminal measured at the electrical reference point. Under these conditions the requirements given in table 7.66at are applicable (more information can be found in Annex A of Recommendation ITU-T P.340 [39]).

Table 7.66at

Category (according to Recommendation ITU-T 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).

The terminal shall have full duplex capability (category 1).

The requirement applies with nominal setting of the volume control for nominal signal levels in send and receive directions as well as for the level combinations +6 dB (re. nominal level) in send/-6 dB (re. nominal level) in receive and +6 dB (re. nominal level) in receive/-6 dB (re. nominal level) in send. The requirement applies also for maximum setting of the volume control with nominal signal levels in send and receive directions.

This measurement shall be repeated for desktop type terminals and softphones with variable echo path.

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. A detailed description can be found in Recommendation ITU-T P.501 [41].

The signals are fed simultaneously in send and receive direction. The level in send direction is -4,7 dBPa at the MRP (nominal level), the level in receive direction is -16 dBm0 at the electrical reference point (nominal level).

The settings for the signals are as follows.

Table 7.66au: Parameters of the two test signals for double talk measurement based on AM-FM modulated sine waves

Send Direction		Receive Direction	
$f_0^{(1)}$ [Hz]	$\pm\Delta f^{(1)}$ [Hz]	$f_0^{(2)}$ [Hz]	$\pm\Delta f^{(2)}$[Hz]
125	±2,5	180	±2,5
250	±5	270	±5
500	±10	540	±10
750	±15	810	±15
1 000	±20	1 080	±20
1 250	±25	1 350	±25
1 500	±30	1 620	±30
1 750	±35	1 890	±35
2 000	±40	2 160	±35
2 250	±40	2 400	±35
2 500	±40	2 650	±35
2 750	±40	2 900	±35
3 000	±40	3 150	±35
3 250	±40	3 400	±35
3 500	±40	3 650	±35
3 750	±40	3 900	±35
4 000	±40	4 150	±35
4 250	±40	4 400	±35
4 500	±40	4 650	±35
4 750	±40	4 900	±35
5 000	±40	5 150	±35
5 250	±40	5 400	±35
5 500	±40	5 650	±35
5 750	±40	5 900	±35
6 000	±40	6 150	±35
6 250	±40	6 400	±35
6 500	±40	6 650	±35
6 750	±40	6 900	±35
7 000	±40		
NOTE: Parameters of the Shaping Filter: f ≥ 250 Hz: Low Pass Filter, 5 dB/oct.			

The test signal is measured at the electrical reference point (send 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 receive direction (see Recommendation ITU-T P.501 [41]). The filter will suppress frequency components of the double talk signal.

In each frequency band which is used in receive 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 7.66at. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

7.5.11.10.4 Minimum activation level and sensitivity of double talk detection

For further study.

7.5.11.11 Switching characteristics

7.5.11.11.0 Note

Additional requirements may be needed in order to further investigate the effect of NLP implementations on the users' perception of speech quality.

7.5.11.11.1 Activation in Sending Direction

The activation in send 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 send 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) shall be ≤ 15 ms.

Measurement method

The test setup is described in clause 6.10.4.

The test signal is the activation of the short conditioning sequence described in clause 7.3.4 of Recommendation ITU-T P.501 [41] with increasing level for each single word.

The settings of the test signal are as follows.

Table 7.66av: Test file settings

	Single word/ Pause Duration	Level of the first single word (active Signal Part at the MRP)	Level Difference between two Periods of the Test Signal
Single word to Determine Switching Characteristic **in Send Direction**	~600 ms / ~400 ms	-24 dBPa (see note)	1 dB
NOTE: The signal level is determined for each utterance individually according to Recommendation ITU-T P.56 [33].			

It is assumed that the pause length of about 400 ms is longer than the hang-over time so that the test object is back to idle mode after each single word.

The level of the transmitted signal is measured at the electrical reference point. The test signal is filtered by the transfer function of the test object. The measured signal level is referred to the filtered 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 single word which indicates the first activation of the test object. The time between the beginning of the single word and the complete activation of the test object is measured.

7.5.11.11.2 Silence Suppression and Comfort Noise Generation

Requirements and measurement methods are for further study.

NOTE: In general it is not recommended to use silence suppression at all.

7.5.11.11.3 Performance in sending direction in the presence of background noise

Requirement

The level of comfort noise shall be within in a range of +2 dB to -5 dB compared to the original (transmitted) background noise. The noise level is calculated with A-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 7.66aw.

Table 7.66aw: Requirements for spectral adjustment of comfort noise (mask)

Frequency	Upper limit	Lower limit
200 Hz	12 dB	-12 dB
800 Hz	12 dB	-12 dB
800 Hz	10 dB	-10 dB
2 000 Hz	10 dB	-10 dB
2 000 Hz	6 dB	-6 dB
4 000 Hz	6 dB	-6 dB
14 000 Hz	6 dB	-6 dB
20 000 Hz	6 dB	-6 dB
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 handsfree is set-up as described in clause 6.10.4.

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 receive direction consisting of an initial pause of 10 s and a periodical repetition of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] in receive 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.

The transmitted signal is recorded in send direction at the POI.

The power density spectra measured in send direction without far end speech simulation averaged between 10 s and 20 s is referred to the power density spectrum measured in send direction determined during the period with far end speech simulation in receive direction averaged between 10 s and 20 s. Level and spectral differences between both power density spectra are analysed and compared to the requirements.

7.5.11.11.4 Speech Quality in the Presence of Background Noise

Requirement

For the background noises defined in clause 6.10.6 the following requirements shall apply:

- N-MOS-LQO$_f \geq 2{,}7$.
- S-MOS-LQO$_f \geq 3{,}3$.
- G-MOS-LQO$_f \geq 2{,}7$.

NOTE: It is recommended to test the terminal performance with other types of background noises if the terminal is likely to be exposed to other noises than specified in clause 6.10.6.

Measurement method

The background noise simulation as described in clause 6.10.6 is used. The handsfree terminal is set-up as described in clause 6.10.4.

The background noise shall be applied for at least 5 s in order to adapt noise reduction algorithms in advance the test.

The near end speech signal consists of 16 sentences of speech (4 male and 4 female talkers, 2 sentences each). The American English speech samples from ETSI TS 103 281 [64] are used. The test signal level is -1,7 dBPa at the MRP.

The following signals are required for the tests:

1) The clean speech signal is used as the undisturbed reference (see ETSI TS 103 281 [64]).

2) The send signal is recorded at the electrical reference point.

N-MOS-LQO_F, S-MOS LQO_F and G-MOS LQO_F are calculated as described in ETSI TS 103 281 [64]. Either Model A or Model B can be used. The model chosen shall be documented in the test report.

When using model A the following mapping functions apply:

$$\text{S-MOS}'_{\text{LQOf}} = 1{,}418 \cdot \text{S-MOS}_{\text{LQOf}} - 1{,}145$$

$$\text{N-MOS}'_{\text{LQOf}} = 1{,}346 \cdot \text{N-MOS}_{\text{LQOf}} - 1{,}584$$

$$\text{G-MOS}'_{\text{LQOf}} = 1{,}279 \cdot \text{G-MOS}_{\text{LQOf}} - 0{,}7364$$

7.5.11.11.5 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out applying a speech signal in receive direction and comparing the noise level transmitted in the send direction under reference conditions with no far end speech to the noise level transmitted in the send direction under test conditions including far end speech. During and after the end of the speech signal the signal level in send direction shall 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 clause 6.10.6.

NOTE: The intention of this measurement is to detect impairments (modulations, switching and others) influencing the background noise transmitted from the terminal under test when a signal from the distant end (receiving side of the terminal under test) is present. Under these test conditions no modulation of the transmitted signal should occur. Modulation, switching or other type of impairments might be caused by an improper behaviour of a nonlinear processor working in conjunction with the echo canceller and erroneously switching or modulating the transmitted background noise.

Measurement method

The test arrangement is according to 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 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 speech signal at the far end. The exactly identical background noise signal is applied. The background noise signal shall start at the same point in time as was used for the reference measurement without far end signal. The background noise shall be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms. After at least 5 s a series of the female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [41] is applied in receive direction with duration of at least 10 s. The test signal level in receive direction is -16 dBm0 at the electrical reference point.

For both reference and test conditions the send 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 send direction is determined during the time interval when the speech signal is applied and after it stops. The level difference is determined from the difference of the recorded signal levels versus time between reference signal and the signal measured with far end signal.

7.5.11.12 Quality of echo cancellation

7.5.11.12.1 Temporal echo effects

Requirements

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk. The measured echo attenuation during single talk shall not decrease by more than 6 dB from the maximum measured echo attenuation.

Measurement method

The test setup is described in clause 6.10.4.

The test signal consists of periodically repeated Composite Source Signal according to Recommendation ITU-T 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 seconds which represents 8 periods of the CSS 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.

The difference between the maximum attenuation and the minimum attenuation is measured.

NOTE 1: In addition tests with more speech like signals should be made, e.g. Recommendation ITU-T P.501 [41] to see time variant behaviour of EC. However for such tests the simple broadband attenuation based test principle as described above cannot be applied due to the time varying spectral content of the speech like signals.

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 (35 ms) of the level analysis taking into account the exponential character of the integration time in any tolerance scheme.

NOTE 3: Care should be taken not to confuse noise or comfort noise with residual echo. In cases of doubt the measured echo signal should be compared to the residual noise signal measured under the same conditions without inserting the receive signal. If the level vs. time analysis leads to the identical result it can be assumed that no echo but just comfort noise is present.

7.5.11.12.2 Spectral Echo Attenuation

Requirements

The echo attenuation vs. frequency shall be below the tolerance mask given in table 7.66ax.

Table 7.66ax: Spectral echo loss limits

Frequency	Limit
100 Hz	-41 dB
1 300 Hz	-41 dB
3 450 Hz	-46 dB
5 200 Hz	-46 dB
7 500 Hz	-37 dB
8 000 Hz	-37 dB
NOTE: 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 send 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 consisting of the compressed real speech signal is described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The level of the training sequence shall be -16 dBm0.

The test signal is the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41]. The measurement is carried out under steady-state conditions. The average test signal level shall be -16 dBm0, averaged over the complete test signal. 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.

7.5.11.12.3 Variable echo path

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk with dynamic changing echo paths.

No level peak shall be more than 10 dB above the minimum noise level during the measurement.

Measurement method

The test setup is described in clause 6.10.7.

NOTE: Care should be taken to not generate noise during the movement of the notebook lid. Because of this, this measurement is not applicable for a softphone without external microphone.

As test signal the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 [41] is used. The signal level shall be -10 dBm0. The terminal volume control is set to nominal RLR. The first 4 sentences of the test signal are used to allow full convergence of the echo canceller. The next 4 sentences (from 10,75 s to 22,5 s) are used for the analysis. The echo signal level is analysed over time. The echo signal level is analysed for 11,75 s, using a time constant of 35 ms.

The measurement result is displayed as echo level versus time.

7.5.11.13 Delay

Technically, this PP type can only be connected to an FP of type 5 (Fixed Part with VoIP interface, wideband, super-wideband, fullband, FBHR or ultra-band service), i.e. a VoIP interface. Hence the requirements and measurement methods given in clause 7.7.1 for the combined FP+PP roundtrip delay apply.

Figure 7.18aa: Void

Figure 7.18ab: Void

7.5.11.14 Send Loudness Level

Requirements:

The nominal value of Send Loudness Level (SLL) shall be:

SLL = 71 phon ± 4 phon

Measurement method:

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

The terminal is set-up as described in clause 6.10.4 and calibration is realized as explained in clause 6.10.4.

The loudness (in sone) of the recorded signal is calculated according to ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is calculated according to clause 9 of ITU-T P.700 [74].

7.5.11.15 Receive Loudness Level

Requirements:

- *Desktop operated PP:*
  - The nominal value of Receive Loudness Level (RLL) shall be 71 ± 4 phon. This value has to be fulfilled for at least one volume setting.
  - The value of RLL at maximum volume setting shall be louder than or equal to 78 phon.
  - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 15 phon.
- *Handheld operated PP:*
  - The nominal value of Receive Loudness Level (RLL) shall be 67 ± 4 phon. This value has to be fulfilled for at least one volume setting.
  - The value of RLL at maximum volume setting shall be louder than or equal to 71 phon.
  - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 15 phon.
- *Softphone (computer-based terminal):*
  - Type 1 or softphone with external speakers: requirement as for desktop terminal.
  - Type 2: requirement as for handheld terminal.
- Group audio terminal*:*
  - The nominal value of Receive Loudness Level (RLL) shall be 71 ± 4 phon. This value has to be fulfilled for at least one volume setting.
  - The value of RLL at maximum volume setting shall be louder than or equal to 82 phon.
  - The range of volume control (RLL measured at minimum and maximum volume setting) shall be larger than or equal to 19 phon.

Measurement method:

The terminal is set-up as described in clause 6.10.4.

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41].

For each recorded artificial ear signal, the loudness (in sone) of the recorded signal is calculated according to Recommendation ITU-T P.700 [74] and noted in the test report. The loudness level (in phon) is determined as follows for binaural measurements: the resulting loudnesses for left and right ears are first halved (individual loudness per ear). Both loudnesses are added (assuming perfect loudness summation). With this overall loudness, the overall loudness level is finally determined according to clause 8.2 of Recommendation ITU-T P.700 [74].

7.5.12 Transmission characteristics for PP types 7c, d, e, f ("fullband 20 kHz stereo audio device")

The specification for the receive direction is for further study. As a guideline the receiving direction should fulfill specification of type 7a for handset/headset-like applications or 7b for loudspeaking/handsfree-like applications.

The sending direction should be muted or fulfill specification of PP type 7a or 7b for FB, 5a or 5b for SWB, 2c or 4b for WB and 1d or 3b for NB application, respectively.

7.5.13 Transmission characteristics for PP type 7g ("FBHR 24 kHz headset device")

The specification for the receive direction is for further study. As a guideline the receive direction should fulfill the specification of type 7a.

In case the device is also used in the send direction it should for this fulfill the specification of PP type 7a or 7b for FB, 5a or 5b for SWB, 2c or 4b for WB and 1d or 3b for NB application, respectively.

7.5.14 Transmission characteristics for PP type 7h ("FBHR 24 kHz loudspeaking device")

The specification for the receive direction is for further study. As a guideline the receive direction should fulfill the specification of type 7b.

In case the device is also used in send direction it should for this fulfill specification of PP type 7a or 7b for FB, 5a or 5b for SWB, 2c or 4b for WB and 1d or 3b for NB application, respectively.

7.5.15 Transmission characteristics for PP type 7i ("FBLFE 250 Hz loudspeaking device")

The specification of type 7i is for further study.

7.5.16 Transmission characteristics for PP type 7j ("fullband 20 kHz low-latency microphone device")

Apart from the delay requirement, the specification of type 7j corresponds to the specification of type 7a and is described in clause 7.5.10. Only the specifications for send direction are applicable for type 7j.
The one-way delay in send direction shall not exceed 12,5 ms.

7.5.17 Transmission characteristics for PP type 8a ("ultra-band 48 kHz headset device")

The specification for the receive direction is for further study. As a guideline the receive direction should fulfill the specification of type 7a.

In case the device is also used in send direction it should for this fulfill specification of PP type 7a or 7b for FB, 5a or 5b for SWB, 2c or 4b for WB and 1d or 3b for NB application, respectively.

7.5.18 Transmission characteristics for PP type 8b ("ultra-band 48 kHz loudspeaking device")

The specification for the receive direction is for further study. As a guideline the receive direction should fulfill the specification of type 7b.

In case the device is also used in send direction it should for this fulfill specification of PP type 7a or 7b for FB, 5a or 5b for SWB, 2c or 4b for WB and 1d or 3b for NB application, respectively.

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)

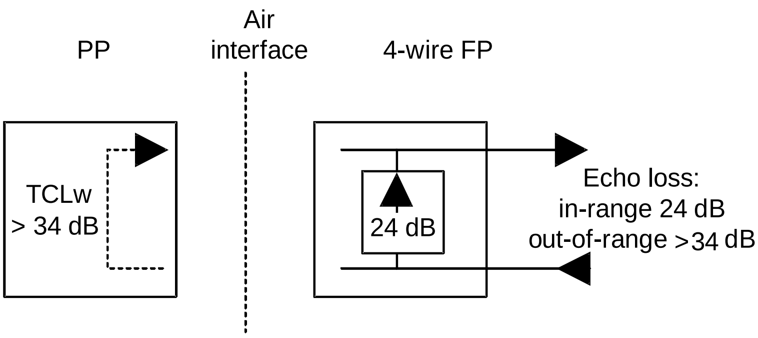
7.6.1.1 Reduction of echo from PP

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.

An 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 of ETSI EN 300 175-8 [8].
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 of ETSI EN 300 175-8 [8]. Clause A.1.2 of ETSI EN 300 175-8 [8] provides, for guidance and illustration, the description of an 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 ETSI EN 300 176-1 [9] and ETSI EN 300 176-2 (the present document). Messages from the PP with control information are defined in ETSI 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 Recommendation ITU-T 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 ETSI 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 Recommendation ITU-T 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 ETSI 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 7.5.3.4.1.

7.6.1.2 FP Network echo control

In 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. Messages from the PP with control information are defined in ETSI 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 Recommendation ITU-T 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 Recommendation ITU-T 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.

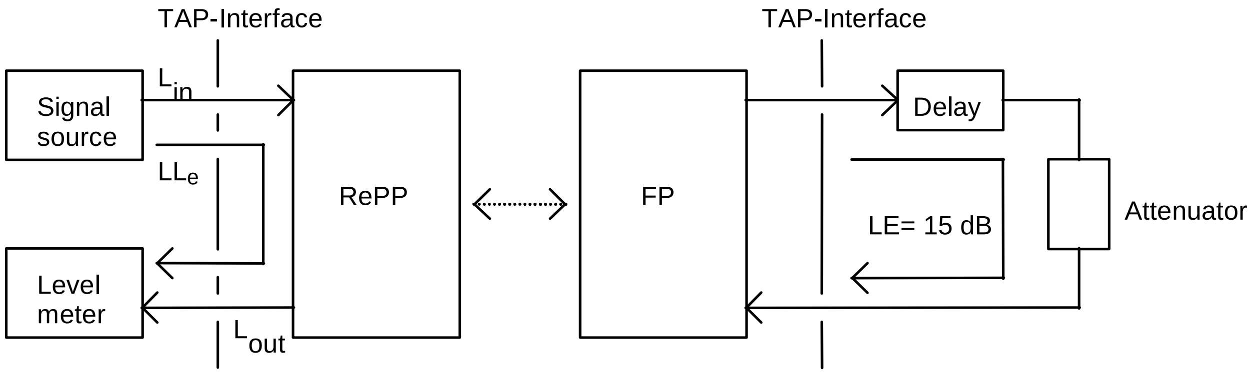


Figure 7.19a: FP with 4-wire interface

With the network echo control enabled, the output level L_{out} shall be measured at the delays 0 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 Recommendation ITU-T 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 ETSI 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.

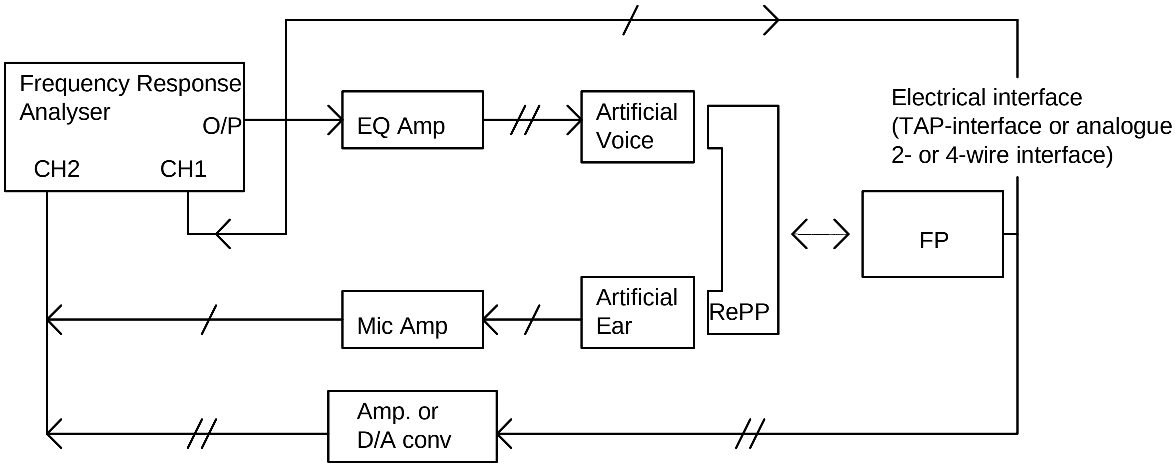
7.6.1.4 FP Delay

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 7.66ay in turn, the delay at each value of f0 shall be derived from the measurements at the corresponding values of f1 and f2.

Table 7.66ay: 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)

7.6.2.0 General

NOTE: This clause is also applicable for FP type 4.

7.6.2.1 FP Network echo control

In 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 (see also ETSI EN 300 176-1 [9]). Messages from the PP with control information are defined in ETSI 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 Recommendation ITU-T 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 Recommendation ITU-T 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.

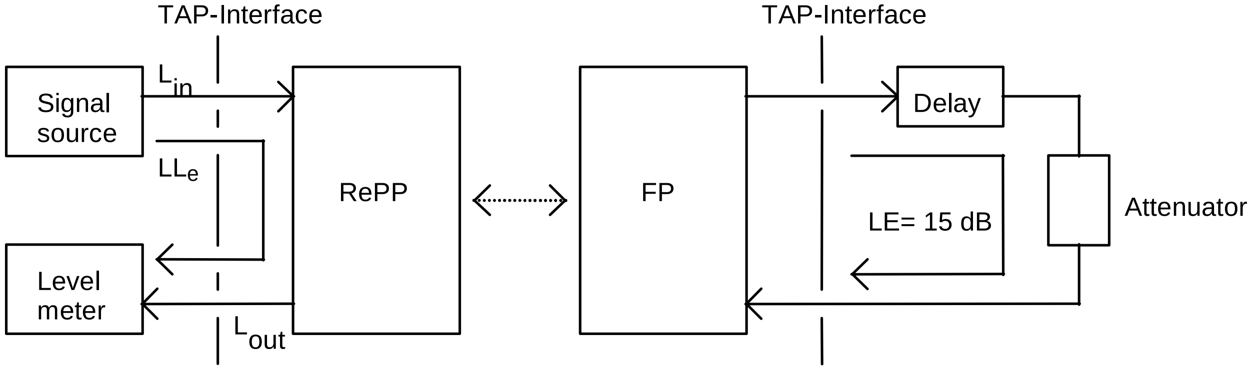


Figure 7.19c: FP with 4-wire interface

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 Recommendation ITU-T 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 ETSI 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.

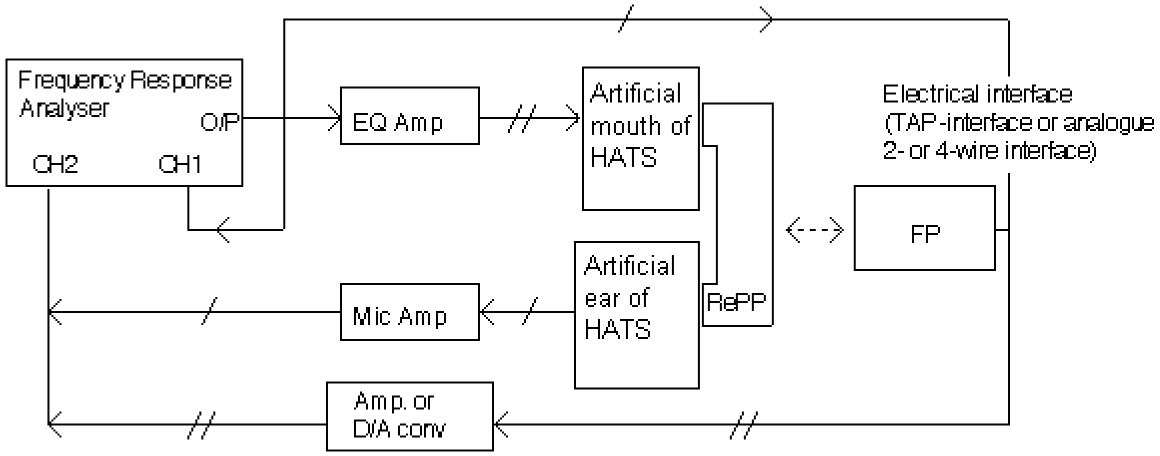
7.6.2.3 FP Delay

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 7.66az in turn, the delay at each value of f0 shall be derived from the measurements at the corresponding values of f1 and f2.

Table 7.66az: 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:

$$D = [-1\,000 \times\acute{} (P1 - P2)] / [360 \times\acute{} (f1 - f2)]$$

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 ETSI EN 300 175-8 [8], annex D for further information.

Measurement method

Measurement of SLR and RLR shall 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 ETSI 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 of ETSI TBR 038 [49], and a reference echo canceller that meets requirement 1, is described in clause A.3.2 of ETSI TBR 038 [49].

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 (see also ETSI EN 300 176-1 [9]). Messages from the PP with control information are defined in ETSI 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 Recommendation ITU-T 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 7.6.3.3 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 Recommendation ITU-T 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 7.67) shall be set to the lowest value that can be selected according to routing information.

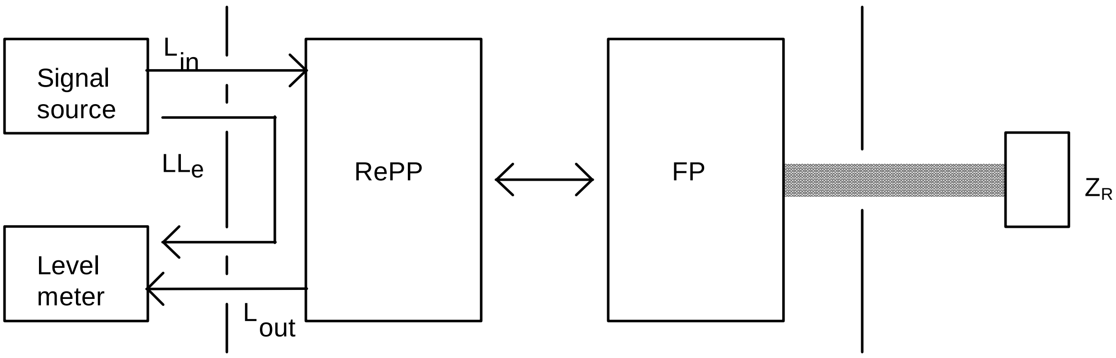


Figure 7.19e: FP with analogue 2-wire interface

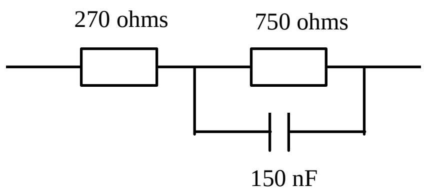


Figure 7.19f: Terminating impedance Z_R

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 Recommendation ITU-T G.122 [16].

The tests shall be repeated with Z_R replaced by each of the two terminating impedances, a and c, defined in ETSI 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 Recommendation ITU-T 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 Recommendation ITU-T G.122 [16].

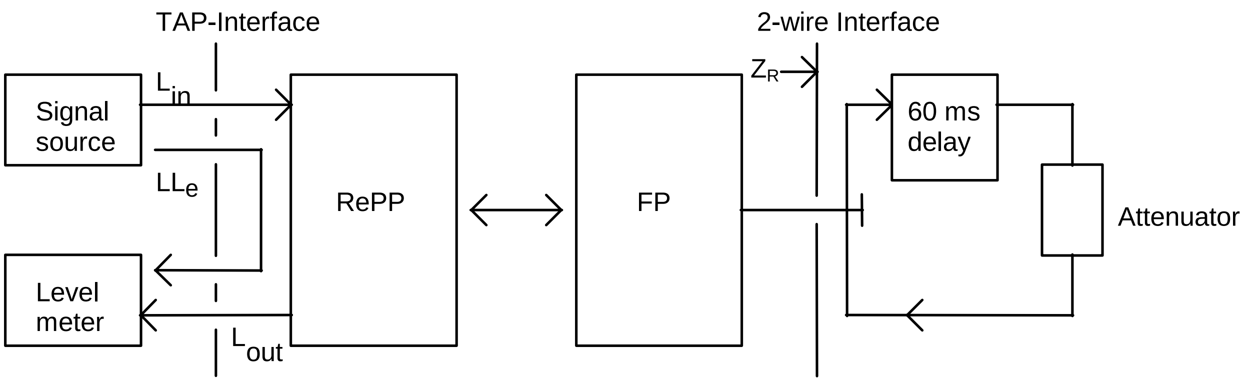


Figure 7.19g: FP with analogue 2-wire interface

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

7.6.3.3.0 Test methods

These additional requirements and test methods are based on ETSI 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 requirements

7.6.3.3.1.1 Polarity independence

The requirements and associated test methods of clause 4.1.1 of ETSI 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 ETSI 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 ETSI TBR 038 [49] shall apply.

7.6.3.3.2 Speech performance characteristics

7.6.3.3.2.0 Test considerations

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 ETSI 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 ETSI 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 ETSI 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 ETSI TBR 038 [49] shall apply, except that the test with input e.m.f. of 0 dBV in clause 4.2.4.2 of ETSI 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 ETSI TBR 038 [49] shall apply, except in the test of clause 4.2.6.1 of ETSI 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 ETSI 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.

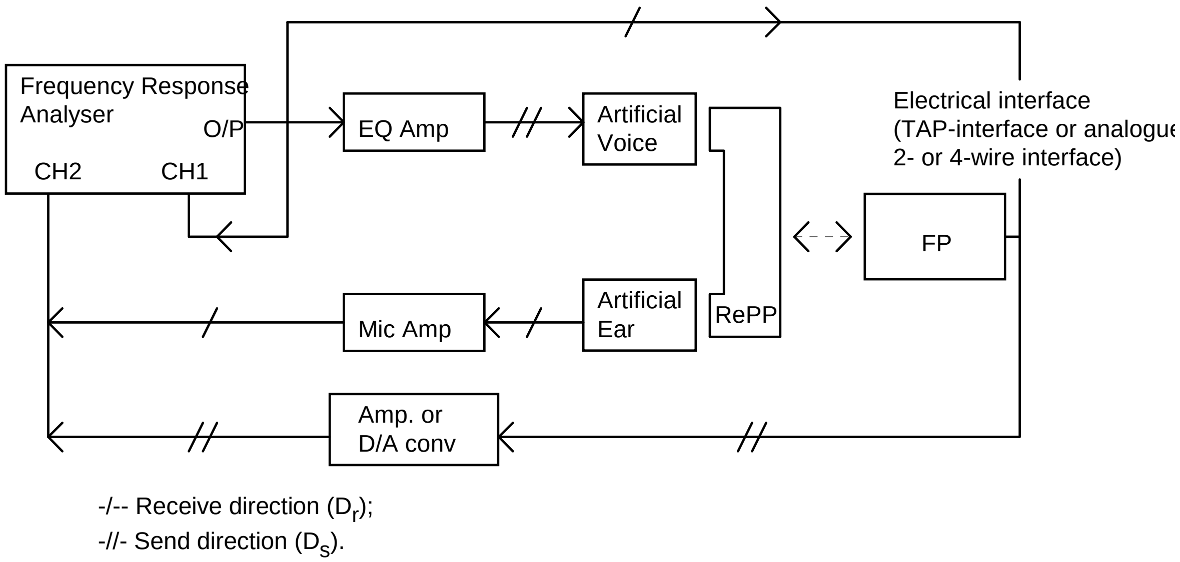


Figure 7.19h: FP delay test configuration

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 Void

7.6.4.2 Void

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 ETSI 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.

7.6.4.4 Clock accuracy

Requirement

The clock drift between the DUT FP and the IP reference interface shall be less than 40 ppm in sending and receiving direction under ideal network conditions.

Measurement method

Send direction

A sequence of CS signals (active signal length = 250 ms) is repeated for 120 s in order to analyse clock accuracy and any other time-variant delay. The pause length between two CS bursts is 100 ms and 1,2 s after every fourth burst in order to simulate a speech pause, which may lead to buffer adjustments. The test signal level shall be -4,7 dBPa at the MRP.

A cross correlation analysis versus time is carried out over the whole 120 s sequence between the received and the original test signal. The duration of the measurement (120 s) is indicated on the x-axis, the result of the cross correlation analysis (delay) is plotted on the y-axis.

The resulting clock accuracy within an analysis time range of at least 60 s is calculated as follows:

$$\text{clock accuracy [ppm]} = \frac{\text{delay change } [s]}{\text{analysis duration } [s]} \times 1 \cdot 10^6$$

Receive direction

A sequence of CS signals (active signal length = 250 ms) is repeated for 120 s in order to analyse clock accuracy and any other time-variant delay. The pause length between two CS bursts is 100 ms and 1,2 s after every fourth burst in order to simulate a speech pause, which may lead to buffer adjustments. The test signal level at the IP reference interface shall be -16 dBm0.

A cross correlation analysis versus time is carried out over the whole 120 s sequence between the received and the original test signal. The duration of the measurement (120 s) is indicated on the x-axis, the result of the cross correlation analysis (delay) is plotted on the y-axis.

The resulting clock accuracy within an analysis time range of at least 60 s is calculated as follows:

$$\text{clock accuracy [ppm]} = \frac{\text{delay change } [s]}{\text{analysis duration } [s]} \times 1 \cdot 10^6$$

7.6.4.5 Send Jitter

Requirement

The measured interarrival jitter in sending direction of the FP under test shall be less than 1 ms.

NOTE: Any jitter introduced in sending direction will lead to potentially increased delay due to increased de-jitter buffer at the far end terminal.

Measurement method

The RTP data stream in sending direction should be monitored with a tap or a switch providing a monitoring port. The monitoring time should be 60 s. A signal like the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41] is played back in send direction using a nominal level of -4,7 dBPa at the MRP. This speech signal is only necessary to make sure, RTP is played out, even in the case VAD is active.

The delay variation for each packet D(i) is evaluated according to IETF RFC 3550 [i.4]:

$$d(i) = \Delta t_{eff(i)} - \Delta t_{exp(i)}$$

$$D(i) = (15 \times D(i-1) + |\, d(i)\, |) / 16$$

With:

- $\Delta t_{exp(i)}$ = the expected time between packet i and packet i-1; and

- $\Delta t_{eff(i)}$ = the effective time between packet i and packet i-1.

Maximum delay variation = MAX(D(i)).

7.6.4.6 Delay

See clause 7.7.1.

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 ETSI 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.

7.6.5.2 FP Delay

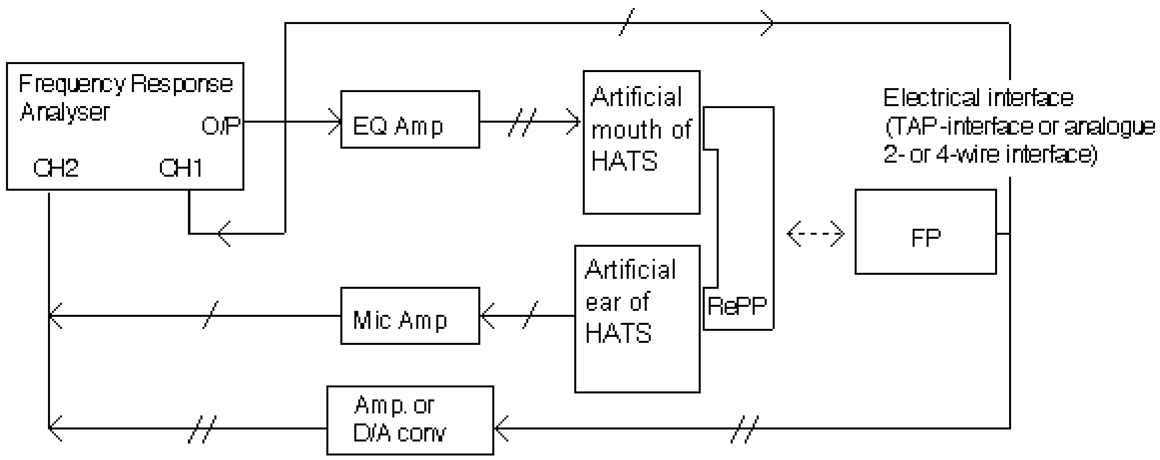
Requirement

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, super-wideband, fullband, FBHR or ultra-band service)

7.6.6.1 Void

7.6.6.2 Void

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 D of ETSI 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 7.5.3.2.

7.6.6.4 Clock accuracy

Requirement

The clock drift between the DUT FP and the IP reference interface shall be less than 40 ppm in sending and receiving direction under ideal network conditions.

Measurement method

Send direction:

A sequence of CS signals (active signal length = 250 ms) is repeated for 120 s in order to analyse clock accuracy and any other time-variant delay. The pause length between two CS bursts is 100 ms and 1,2 s after every fourth burst in order to simulate a speech pause, which may lead to buffer adjustments. The test signal level shall be -4,7 dBPa at the MRP.

A cross correlation analysis versus time is carried out over the whole 120 s sequence between the received and the original test signal. The duration of the measurement (120 s) is indicated on the x-axis, the result of the cross correlation analysis (delay) is plotted on the y-axis.

The resulting clock accuracy within an analysis time range of at least 60 s is calculated as follows:

$$\text{clock accuracy [ppm]} = \frac{\text{delay change } [s]}{\text{analysis duration } [s]} \times 1 \cdot 10^6$$

Receive direction:

A sequence of CS signals (active signal length = 250 ms) is repeated for 120 s in order to analyse clock accuracy and any other time-variant delay. The pause length between two CS bursts is 100 ms and 1,2 s after every fourth burst in order to simulate a speech pause, which may lead to buffer adjustments. The test signal level at the IP reference interface shall be -16 dBm0.

A cross correlation analysis versus time is carried out over the whole 120 s sequence between the received and the original test signal. The duration of the measurement (120 s) is indicated on the x-axis, the result of the cross correlation analysis (delay) is plotted on the y-axis.

The resulting clock accuracy within an analysis time range of at least 60 s is calculated as follows:

$$\text{clock accuracy [ppm]} = \frac{\text{delay change } [s]}{\text{analysis duration } [s]} \times 1 \cdot 10^6$$

7.6.6.5 Send Jitter

Requirement

The measured interarrival jitter in sending direction of the FP under test shall be less than 1 ms.

NOTE: Any jitter introduced in sending direction will lead to potentially increased delay due to increased de-jitter buffer at the far end terminal.

Measurement method

The RTP data stream in sending direction should be monitored with a tap or a switch providing a monitoring port.

The monitoring time should be 60 s. A signal like the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [41] is played back in send direction using a nominal level of -4,7 dBPa at the MRP. This speech signal is only necessary to make sure RTP is played out, even in the case VAD is active.

The delay variation for each packet D(i) is evaluated according to IETF RFC 3550 [i.4]:

$$d(i) = \Delta t_{eff(i)} - \Delta t_{exp(i)}$$

$$D(i) = (15 \times D(i-1) + | d(i) |) / 16$$

With:

- $\Delta t_{exp(i)}$ = the expected time between packet i and packet i-1; and
- $\Delta t_{eff(i)}$ = the effective time between packet i and packet i-1.

Maximum delay variation = MAX(D(i)).

7.6.6.6 Delay

See clause 7.7.1.

7.7 Transmission characteristics for a complete DECT system consisting of PP and FP with VoIP interface

7.7.1 Send and receive delay - round trip delay

The roundtrip delay of a VoIP-terminal is defined as the sum of FP+PP send and receive delays. For a telecommunication connection, only the FP+PP roundtrip delay can be experienced. For this reason, also the requirement for VoIP-terminals is given only for the roundtrip delay.

Requirement:

It is recognized that the end to end delay should be as small as possible in order to ensure high quality of the communication.

If a PP of type handset/headset is connected, the roundtrip delay of the VoIP-terminal T_{rtd} (sum of FP+PP receive and send delay) shall be less than 119 ms.

If a PP of type loudspeaking and handsfree device is connected, the roundtrip delay of the VoIP-terminal T_{rtd} (sum of FP+PP receive and send delay) shall be less than 125 ms.

NOTE 1: The limit for the roundtrip delay T_{rtd} of the VoIP-terminal is derived from the sum of the send and receive delay limits shown in figure F.1.

NOTE 2: This requirement is 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 3: The derivation of the delay values can be found in clause F.3.

Measurement method:

The roundtrip delay is determined as the sum of the individual contributions of send and receive direction. The setup of the handset/headset/handsfree terminal is in correspondence to clauses 6.10.3 and 6.10.4.

- **Send direction**

The delay in send direction is measured from the MRP to POI. The delay measured in send direction is:

$$T_s + t_{system}$$

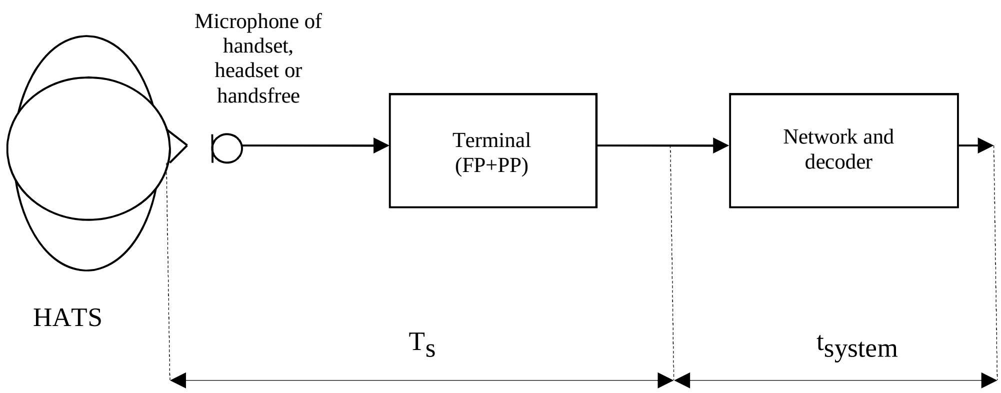


Figure 7.19i1: Different blocks contributing to the delay in send direction

The system delay t_{system} is depending on the transmission method used and the network. The delay t_{system} shall be provided by the manufacturer of the test equipment and considered in the calculation of the delay, which shall be determined as follows:

1) For the measurements a Composite Source Signal (CSS) according to Recommendation ITU-T P.501 [41] is used. The pseudo random noise (pn)-part of the CSS has to be longer than the maximum expected delay. It is recommended to use a pn sequence of 16 k samples (with 48 kHz sampling rate). The test signal level is -4,7 dBPa at the MRP.

2) The reference signal is the original signal (test signal).

3) The delay is determined by cross-correlation analysis between the measured signal at the electrical access point and the original signal. The measurement is corrected by delays which are caused by the test equipment.

4) The delay is measured in ms and the maximum of the cross-correlation function is used for the determination.

- **Receive direction**

The delay in receive direction is measured from POI to the Drum Reference Point (DRP). The delay measured in receive direction is:

$$T_r + t_{system}$$

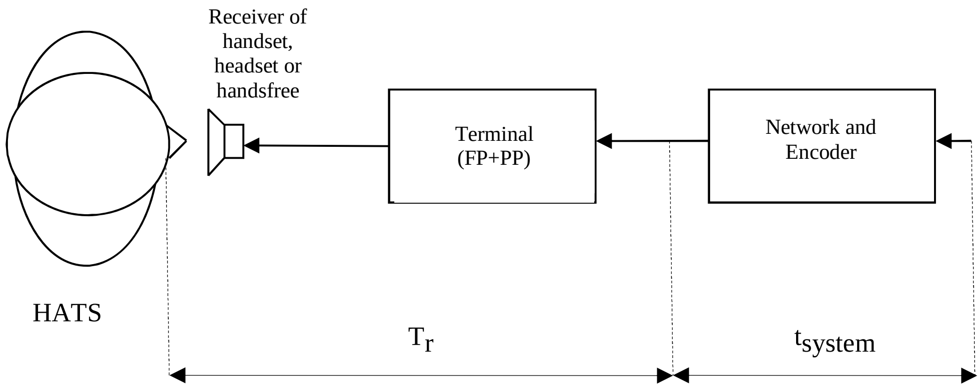


Figure 7.19i2: Different blocks contributing to the delay in receive direction

The system delay t_{system} is depending on the transmission system and the network. The delay t_{system} shall be provided by the manufacturer of the test equipment and considered in the calculation of the delay, which shall be determined as follows:

1) For the measurements a Composite Source Signal (CSS) according to Recommendation ITU-T P.501 [41] is used. The pseudo random noise (pn)-part of the CSS has to be longer than the maximum expected delay. It is recommended to use a pn sequence of 16 k samples (with 48 kHz sampling rate). The test signal level is -16 dBm0 at the electrical interface (POI).

2) The reference signal is the original signal (test signal).

3) The delay is determined by cross-correlation analysis between the measured signal at the DRP and the original signal. The measurement is corrected by delays which are caused by the test equipment.

4) The delay is measured in ms and the maximum of the cross-correlation function is used for the determination.

NOTE 4: It is not necessary to know the delays T_s, T_r and t_{system} per direction. The roundtrip delay of the terminal is the sum of send and receive delays minus the roundtrip delay of the measurement equipment and (if applicable) the network.

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 than 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 Recommendation ITU-T 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 Recommendation ITU-T G.191 [53].

A set of test vectors on which to perform this verification is provided in Recommendation ITU-T 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: Recommendation ITU-T 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 µ-law encoding laws shall conform to Recommendation ITU-T 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 Recommendation ITU-T G.191 [53].

8.2.3 G.722 wideband codec operating at 64 kbit/s

The speech coding algorithm shall conform to Recommendation ITU-T 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 Recommendation ITU-T G.191 [53].

A set of test vectors on which to perform this verification is provided in Recommendation ITU-T G.722 [55], Appendix II.

As stated in Recommendation ITU-T 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: Recommendation ITU-T G.722 [22], Appendix III "A high quality packet loss concealment algorithm for G.722" or Recommendation ITU-T 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 Recommendation ITU-T G.729.1 [25].

Maximum data rate shall be 30 kbit/s as defined in ETSI EN 300 175-8 [8].

NOTE: DECT transport uses 32 kbit/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 Recommendation ITU-T G.729.1 [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 F.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.

8.2.8 LC3plus narrowband, wideband, super-wideband, fullband, FBHR, ultra-band codec operating at 32 to 512 kbit/s

The LC3plus implementation shall comply to the conformance criteria defined in ETSI TS 103 634 [58], clause 7.6. For NG voice services, the conformance test groups listed in table 8.1 shall pass the conformance criteria. For the advanced audio services ETSI TS 103 706 [77], the conformance test groups for the used codec configuration listed in table 8.2 shall pass the conformance criteria. Other codec configurations are considered as optional. An overview of the LC3plus conformance procedure is given in clause 7.1 of ETSI TS 103 634 [58].

Table 8.1: Conformance test configuration for LC3plus DECT voice services

LC3plus application	**Codec configuration**				**Conformance test groups** (see NOTE)		
	Sample rate [Hz]	Frame size [ms]	Channels	Bit rate [bytes per frame]	Core coder	Concealment	Channel Coder
Narrow band LC3plus voice service [NG1.7]	8 000	10	1	40	Enc, Dec, EncDec	Dec	Enc, Dec
Wideband LC3plus voice service [NG1.8]	16 000	10	1	40	Enc, Dec, EncDec	Dec	Enc, Dec
Super-wideband LC3plus voice service [NG1.9]	32 000	10	1	80	Enc, Dec, EncDec	Dec	Enc, Dec
Fullband LC3plus voice service [NG1.10]	48 000	10	1	80	Enc, Dec, EncDec	Dec	Enc, Dec
NOTE: Some conformance tests are conducted for the modules encoder (Enc), decoder (Dec) and codec (EncDec) separately.							

Table 8.2: Conformance test configuration for LC3plus DECT advanced audio profile services ETSI TS 103 706 [77]

LC3plus configuration for DECT Advanced Audio Profile [77]	**Codec configuration**					**Conformance tests group (see clause 7.3.1 of [58])**		
	Bandwidth [Hz]	**Sampling rate [Hz]**	**Frame size [ms]**	**Channels**	**Bit rate [bytes per frame]**	**Core coder**	**Conceal-ment**	**Channel Coder**
AA.AC.1 (see note 1)	0 to 8 000	16 000	10	1	40	Enc, Dec	Dec	Enc, Dec
AA.AC.2 (see note 2)	0 to 16 000	32 000	10	1	80	Enc, Dec	Dec	Enc, Dec
AA.AC.3 (see note 3)	0 to 20 000	48 000	10	1	80	Enc, Dec	Dec	Enc, Dec

LC3plus configuration for DECT Advanced Audio Profile [77]	Codec configuration					Conformance tests group (see clause 7.3.1 of [58])		
	Bandwidth [Hz]	Sampling rate [Hz]	Frame size [ms]	Channels	Bit rate [bytes per frame]	Core coder	Conceal-ment	Channel Coder
AA.AC.4	0 to 20 000	48 000	10	1	120	Enc, Dec	Dec	Enc, Dec
AA.AC.5	0 to 20 000	48 000	10	1	160	Enc, Dec	Dec	Enc, Dec
AA.AC.6	0 to 20 000	48 000	5	1	80	Enc, Dec	Dec	Enc, Dec
AA.AC.7	0 to 20 000	48 000	5	1	100	Enc, Dec	Dec	Enc, Dec
AA.AC.8	0 to 20 000	48 000	5	1	120	Enc, Dec	Dec	Enc, Dec
AA.AC.9	0 to 20 000	48 000	2,5	1	40	Enc, Dec	Dec	Enc, Dec
AA.AC.10	0 to 20 000	48 000	2,5	1	80	Enc, Dec	Dec	Enc, Dec
AA.AC.11	0 to 24 000	48 000	10	1	160	HR (see note 4) Enc, HR (see note 4) Dec	HR (see note 4) Dec	HR (see note 4) Enc, HR (see note 4) Dec
AA.AC.12	0 to 24 000	48 000	10	1	200	HR (see note 4) Enc, HR (see note 4) Dec	HR (see note 4) Dec	HR (see note 4) Enc, HR (see note 4) Dec
AA.AC.13	0 to 24 000	48 000	10	1	240	HR (see note 4) Enc, HR (see note 4) Dec	HR (see note 4) Dec	HR (see note 4) Enc, HR (see note 4) Dec
AA.AC.14	0 to 48 000	96 000	10	1	200	HR (see note 4) Enc, HR (see note 4) Dec	HR (see note 4) Dec	HR (see note 4) Enc, HR (see note 4) Dec
AA.AC.15	0 to 48 000	96 000	10	1	240	HR (see note 4) Enc, HR (see note 4) Dec	HR (see note 4) Dec	HR (see note 4) Enc, HR (see note 4) Dec
AA.AC.16	0 to 48 000	96 000	10	1	320	HR (see note 4) Enc, HR (see note 4) Dec	HR (see note 4) Dec	HR (see note 4) Enc, HR (see note 4) Dec
AA.AC.17	0 to 48 000	96 000	10	1	400	HR (see note 4) Enc, HR (see note 4) Dec	HR (see note 4) Dec	HR (see note 4) Enc, HR(see note 4) Dec
AA.AC.18	0 to 250	48 000	10	1	40	Enc, Dec, Enc LFE (see note 5)	Dec	Enc, Dec
AA.AC.19	0 to 250	48 000	5	1	40	Enc, Dec, Enc LFE (see note 5)	Dec	Enc, Dec
AA.AC.20	0 to 20 000	48 000	10	2[6]	160	Enc, Dec	Dec	Enc, Dec
AA.AC.21	0 to 20 000	48 000	5	2[6]	160	Enc, Dec	Dec	Enc, Dec

LC3plus configuration for DECT Advanced Audio Profile [77]	Codec configuration					Conformance tests group (see clause 7.3.1 of [58])		
	Bandwidth [Hz]	Sampling rate [Hz]	Frame size [ms]	Channels	Bit rate [bytes per frame]	Core coder	Conceal-ment	Channel Coder
AA.AC.22	0 to 20 000	48 000	2,5	2[6]	100	Enc, Dec	Dec	Enc, Dec
AA.AC.23	0 to 20 000	48 000	2,5	2[6]	160	Enc, Dec	Dec	Enc, Dec
NOTE 1: AA.AC.1 identical to NG1.8. NOTE 2: AA.AC.2 identical to NG1.9. NOTE 3: AA.AC.3 identical to NG1.10. NOTE 4: HR denotes the high-resolution mode, therefore clause 7.3.5 of ETSI TS 103 634 [58] applies additionally. NOTE 5: LFE denotes the Low Frequency Enhancement mode, therefore clause 7.3.6 of ETSI TS 103 634 [58] applies additionally. NOTE 6: For stereo channels, the combined channel coder tests clause 7.3.4.4 and 7.3.4.5 of ETSI TS 103 634 [58] need to be tested additionally.								

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, 4b, 5b 7b, 7c, 7d, 7e, 7f, 7h, 7i, 8b 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 for handsets of clause 7.2, 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 described in 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 described in clause 7.2.

9.2 Tandem with mobile radio network

9.2.0 General

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

9.2.1.0 Configuration

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.

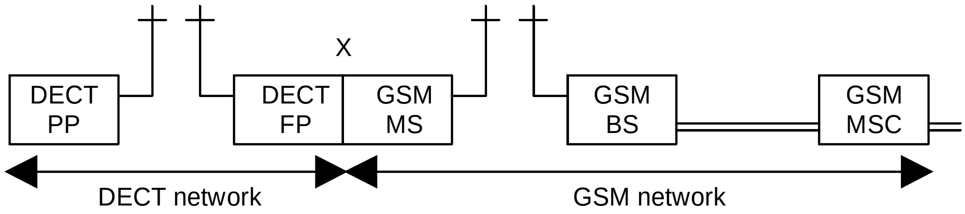


Figure 9.1: DECT in tandem with the GSM mobile network

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 ETSI 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 ETSI EN 300 175-8 [8] for information on GSM, DTX.

9.3 DECT connected to the GSM fixed network

9.3.0 General

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.

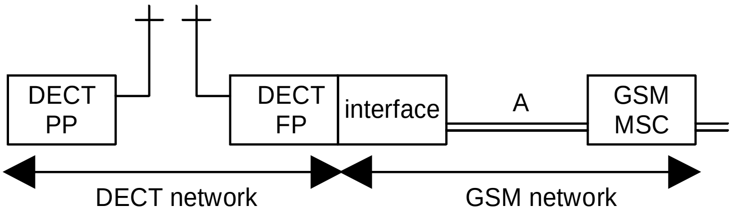


Figure 9.2: DECT in tandem with the GSM fixed network

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 ETSI ETS 300 540 [12].

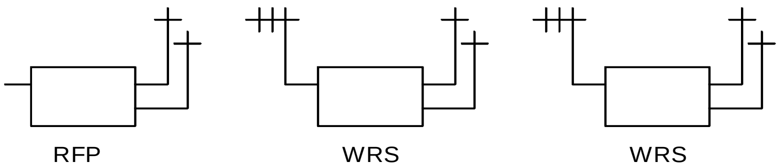
9.3.2 Terminal coupling loss

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)

9.4.0 General

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 clause 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).

Handset and headset:

- SLR.
- RLR.
- STMR.
- Sending response curve.
- Receiving response curve.
- Sending distortion.
- Receiving distortion.
- Receiving noise.
- TCL/TCLw.
- Acoustic shock for headsets.

Loudspeaking and handsfree:

- SLR.
- RLR.
- Sending response curve.
- Receiving response curve.
- Sending distortion.
- Receiving distortion.
- Receiving noise.
- TCL/TCLw.
- Attenuation range in sending direction during double talk.
- Attenuation range in receiving direction during double talk.
- Sending 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 Recommendation ITU-T P.51 [32] and specified in annex B of ETSI 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.

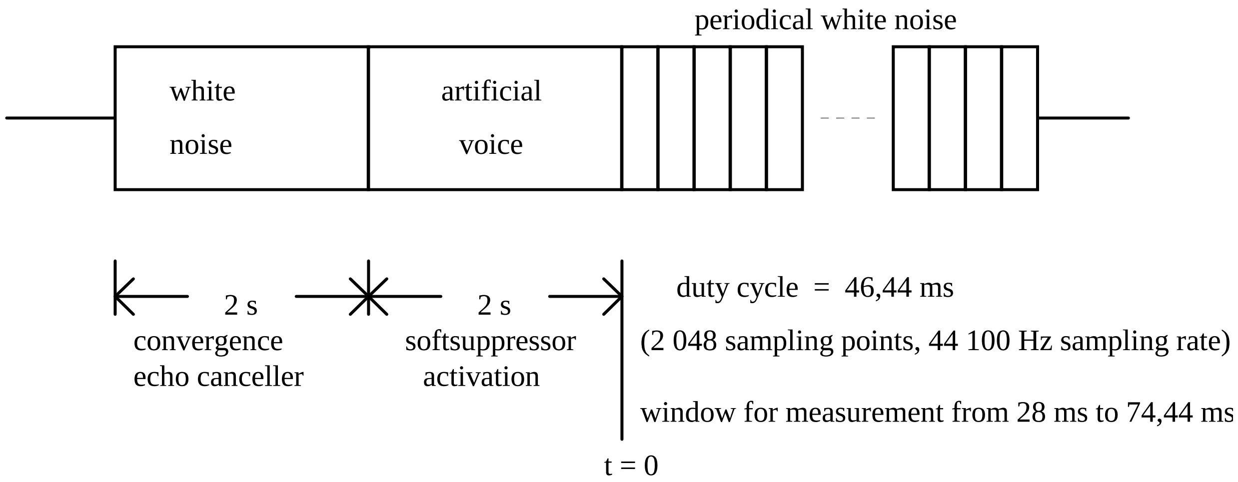


Figure B.1

Artificial voice: according to Recommendation ITU-T P.50 [31] white noise band limited 100 Hz to 10 kHz.

Table B.1

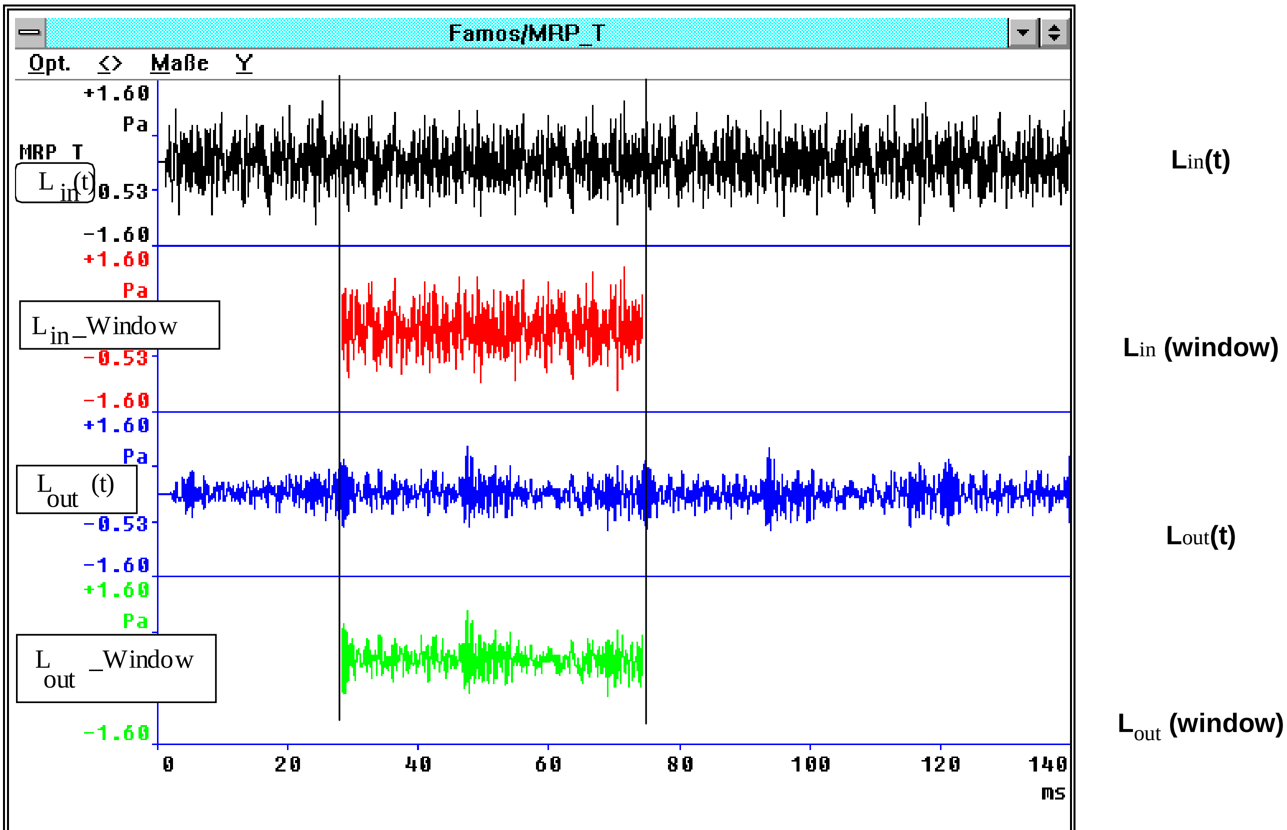
Upper limit		Lower limit	
45 Hz	-36 dB	< 100 Hz	-∞∞ dB
90 Hz	-21 dB	100 Hz	-26,5 dB
		200 Hz	-19,5 dB
		5 000 Hz	-5,5 dB
11 200 Hz	0 dB	10 000 Hz	-6,5 dB
22 400 Hz		> 10 000 Hz	-∞∞ dB

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

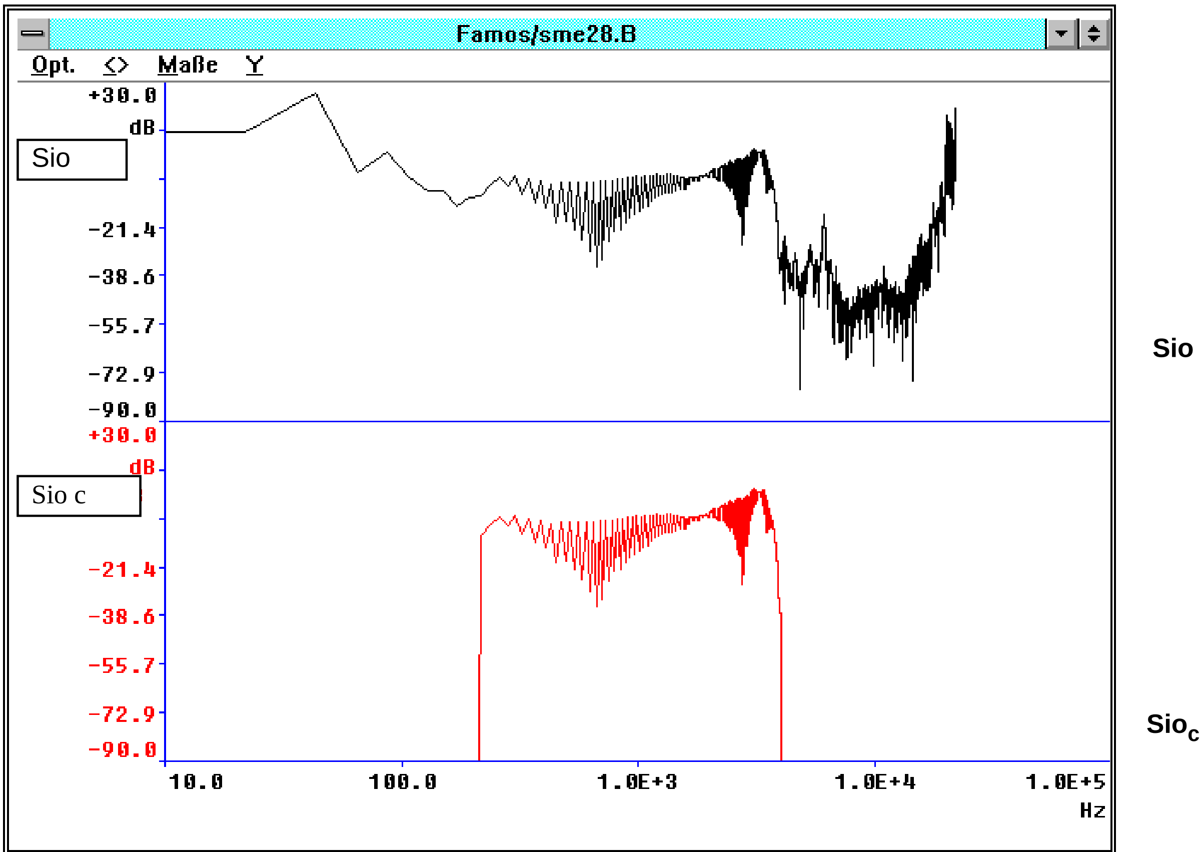
The cut signals will be used for the calculation of the sensitivity Sio by application of the Fast Fourier Transformation (FFT):

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.

$$Sio_{corrected} = Sio \Big|_{200\text{ Hz to }4\,000\text{ Hz, other values equal to }0} \qquad (B.2)$$



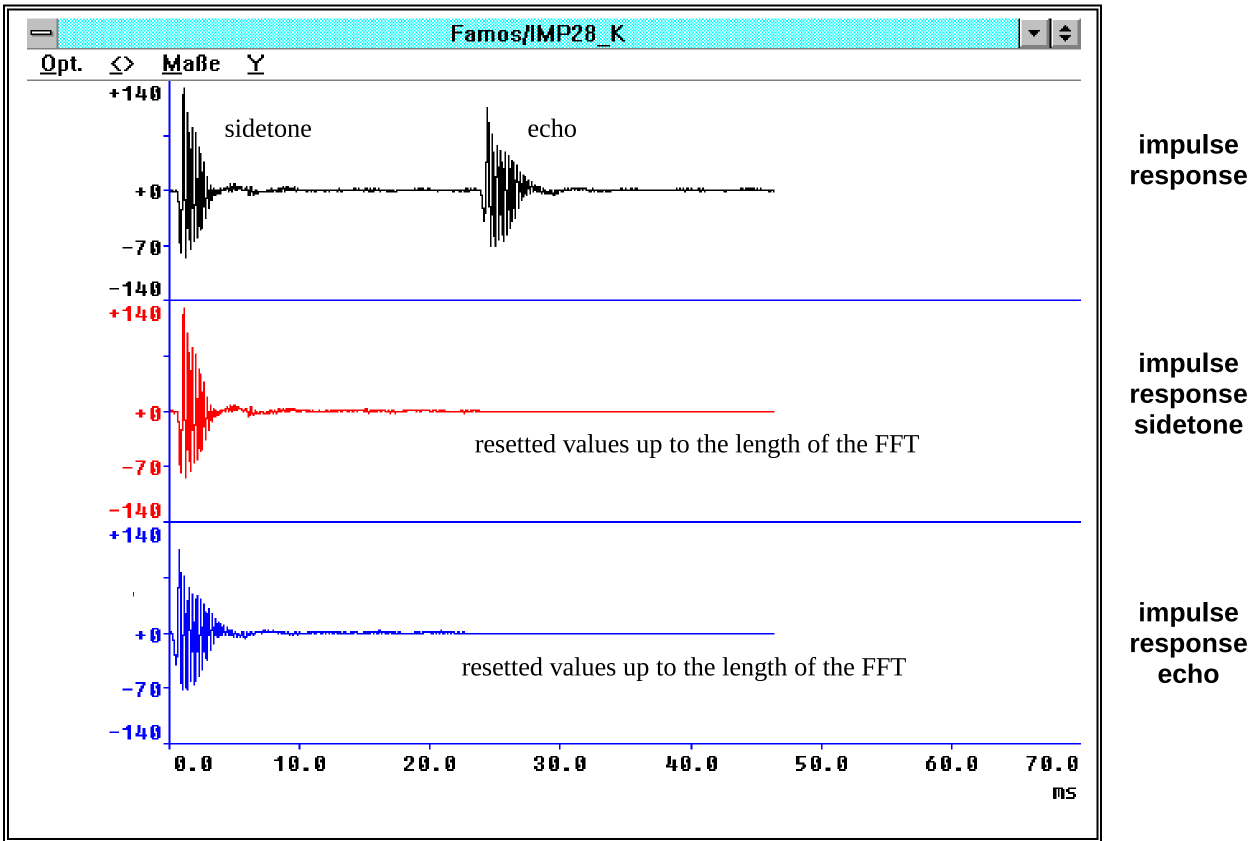
maximum of the delay of the echo path	23 ms
sampling frequency	44 100 Hz
length of the FFT/solution	2 048/21,53 Hz

Figure B.3

The Inverse Fourier Transformation applied to the corrected transfer function Sio_c provides the impulse response at the time domain:

$$Im_{impulse\ response} = iFFT(Sio_c) \qquad (B.3)$$

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.

$$Sio_{echo} = FFT(Im_{\text{impulse response sidetone}}) \Big|_{\text{rectangular window}} \qquad \text{(B.4)}$$

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 ≤≤ 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 $\Phi_{xy}(\tau)$ between the input signal $S_x(t)$ and the output signal $S_y(t)$ is calculated in the time domain:

$$\Phi_{xy}(\tau) = \frac{1}{T}\int_{t=\frac{-T}{2}}^{t=\frac{+T}{2}} S_x(t) \times S_y(t+\tau) \times dt \qquad \text{(C.1)}$$

The measurement window T shall be exactly identical with the time period T of the test signal.

The delay is calculated from the envelope $E(\tau)$ of the cross-correlation function $\Phi_{xy}(\tau)$. The maximum of the envelope function occurs in correspondence to the measured delay. The envelope $E(\tau)$ is calculated by the Hilbert transformation $H\{\Phi_{xy}(\tau)\}$ of the cross-correlation:

$$\mathrm{H}\{\Phi_{xy}(\tau)\} = \frac{1}{\pi}\sum_{u=-\infty}^{+\infty}\frac{\Phi_{xy}(u)}{\tau-u} \qquad \text{(C.2)}$$

$$E(\tau) = \sqrt{\left[\Phi_{xy}(\tau)\right]^2 + \left[\mathrm{H}\{\Phi_{xy}(\tau)\}\right]^2} \qquad \text{(C.3)}$$

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

D.0 General

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 Recommendation ITU-T 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 sent 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	7.2.3 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)

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**
	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)
	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**	**TCL/TCLw Requirement for type (dB)**	**TCL/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	Recommendation ITU-T P.311 [38] tested, wideband handset	7.2.9 7.5.5.4.1	> 55	> 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**	**TCL/TCLw Requirement for type (dB)**	**TCL/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 clause A.1 of ETSI EN 300 175-8 [8]
						(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 of clause A.1 of ETSI EN 300 175-8 [8]

Table for FPs								
Applicable to	**Type nr.**	**Audio type name**	**Clause**	**TCL/TCLw Requirement for type (dB)**	**TCL/TCLw Real Value (dB)**	**Flag "echo parameter" (bits 5 and 6 in octet 3b of Terminal Capability)**	**Action**	**Clause (echo handling)**
						(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 clause A.1 of ETSI EN 300 175-8 [8]
	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 clause A.1 of ETSI EN 300 175-8 [8]

Table for FPs								
Applicable to	**Type nr.**	**Audio type name**	**Clause**	**TCL/TCLw Requirement for type (dB)**	**TCL/TCLw Real Value (dB)**	**Flag "echo parameter" (bits 5 and 6 in octet 3b of Terminal Capability)**	**Action**	**Clause (echo handling)**
	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
	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

Table for FPs								
Applicable to	**Type nr.**	**Audio type name**	**Clause**	**TCL/TCLw Requirement for type (dB)**	**TCL/TCLw Real Value (dB)**	**Flag "echo parameter" (bits 5 and 6 in octet 3b of Terminal Capability)**	**Action**	**Clause (echo handling)**
	5	Fixed Part with VoIP interface, wideband, super-wideband, fullband, FBHR or ultra-band 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 Recommendation ITU-T 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.

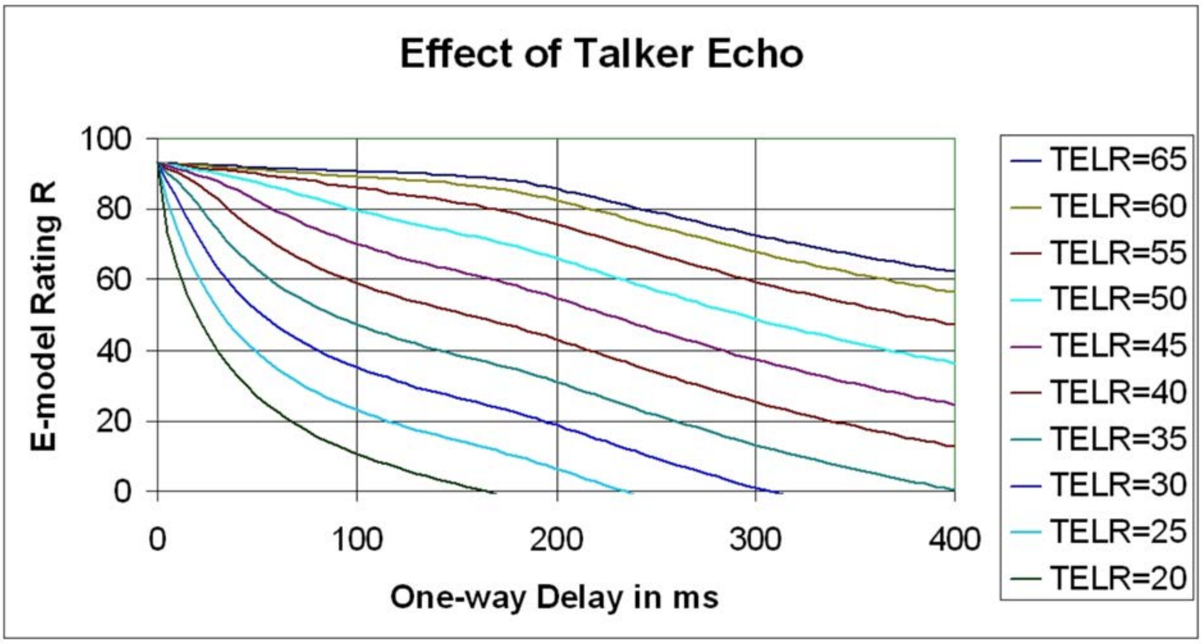


Figure E.1: E-model rating versus delay and TELR

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

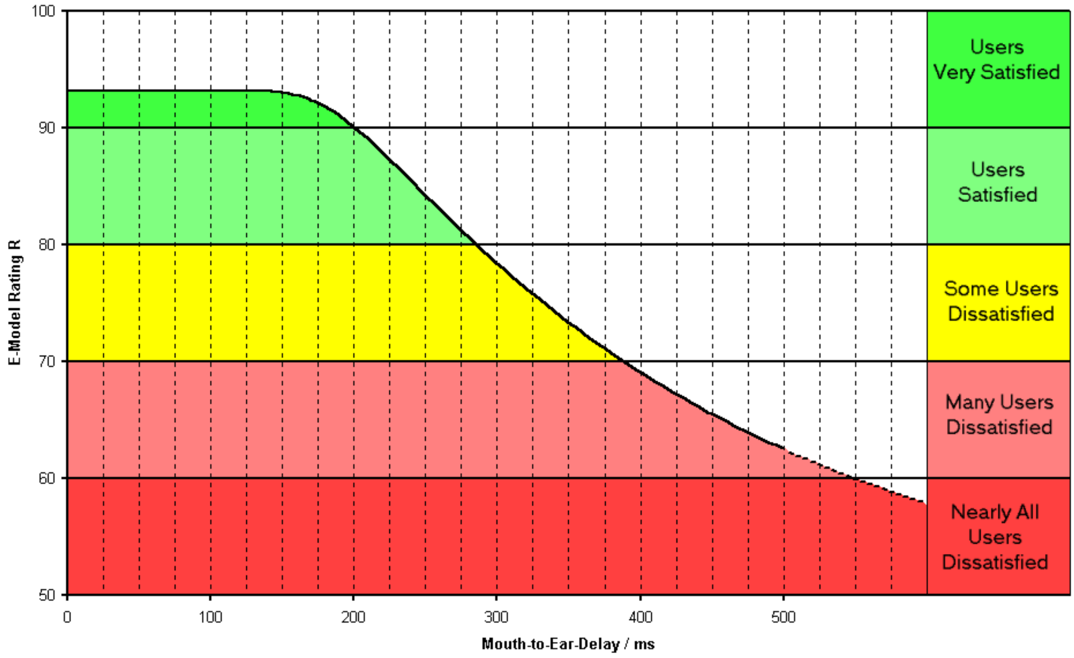


Figure E.2: Subjective rating versus E-model rating and round-trip delay

The result of an E-model calculation is an estimation of average user satisfaction which can according to Recommendation ITU-T 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 Recommendation TU-T 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 a "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 a "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 a "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 considerations for FPs with VoIP interface

F.1.1 Delay considerations for FP type 3 (Fixed Part with VoIP interface, 3,1 kHz service)

F.1.1.0 General

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 Recommendation ITU-T G.1020 [26] and in figure A.1 of Recommendation ITU-T 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:

$$T(ps) = \text{packet size} = N \times T(fs)$$

$$N = \text{number of frames per packet}$$

$$T(fs) = \text{frame size of encoder}$$

$$T(ead) = \text{additional encoder algorithmic delay (look-ahead filtering)}$$

$$T(aif) = \text{air interface framing}$$

$$T(asp) = \text{allowance for signal processing}$$

$$T(tra) = \text{allowance for transcoding}$$

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(ead) 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
LC3plus [58]	LC3plus [58]	1	10	10	2,5	0	10		< 22,5
		2	10	20	2,5	0	10		< 32,5
LC3plus [58]	G.711 [19]	1	10	10	2,5	0	10	< 1	< 23,5
		2	10	20	2,5	0	10	< 1	< 33,5

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 Recommendation ITU-T G.1020 [26] and in figure A.2 of Recommendation ITU-T 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 decoder (=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) ≤ 3 x 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(dad) the values for receive 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
LC3plus [58]	LC3plus [58]	1	10	0	0	< 30	0	10		< 50
		2	10	0	0	< 60	0	10		< 80
LC3plus [58]	G.711 [19]	1	10	0	0	<30	0	10	< 1	< 51
		2	10	0	0	< 60	0	10	< 1	< 81

T(ps) = packet size = N x 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 considerations for FP type 5 (Fixed Part with VoIP interface, wideband, super-wideband, fullband, FBHR or ultra-band service)

F.1.2.0 General

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 Recommendation ITU-T G.1020 [26] and in figure A.1 of Recommendation ITU-T G.1020 [26], respectively.

The sending delay T(s) is defined as follows:

T(s) = T(ps) + T(ead) + T(aif) + T(asp)

where:

T(ps) = packet size = N x 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(ead) 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/s	MPEG-4 ER AAC-LD [48], 64 kbit/sec	1	10	10	10	0	10	< 30
MPEG-4 ER AAC-LD [48], 32 kbit/s	MPEG-4 ER AAC-LD [48], 32 kbit/s	1	20	20	20	0	20	< 60
LC3plus [58]	LC3plus [58]	1	10	10	2,5	0	10	< 22,5
		2	10	20	2,5	0	10	< 32,5
LC3plus [58]	G.722 [21]	1	10	10	2,5	0	10	< 22,5
		2	10	20	2,5	0	10	< 32,5

NOTE 3: In the case of G.729.1 with lower rate (down to 8 kbit/s) 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 Recommendation ITU-T G.1020 [26] and in figure A.2 of Recommendation ITU-T 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 decoder (= 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

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 (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 x T(ps).

For codecs without integrated PLC the additional PLC buffer size should be T(plc) < 10 ms.

For codecs 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) the values for receive 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/s	MPEG-4 ER AAC-LD [48], 64 kb/s	1	10	0	0	<30	0	10	< 50
MPEG-4 ER AAC-LD [48], 32 kb/s	MPEG-4 ER AAC-LD [48], 32 kb/s	1	20	0	0	<60	0	20	< 100
LC3plus [58]	LC3plus [58]	1	10	0	0	< 30	0	10	< 50
		2	10	0	0	< 60	0	10	< 80
LC3plus [58]	G.722 [21]	1	10	0	0	< 30	0	10	< 50
		1	10	0	0	< 60	0	10	< 80

T(ps) = packet size = N x 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 bits.

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_30017602v020305a0.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 ETSI 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	Castanets	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_30017602v020305a0.zip which accompanies the present document.

F.3 Derivation of Delay requirements for PPs and FPs with VoIP interface

Figure F.1 shows an exemplary breakdown of the contributions of the various elements in a typical DECT system consisting of an FP with VoIP interface (FP type 3 or 5) and any PP type.

The following blocks can be identified:

- JB: Jitter Buffer of the VoIP terminal. Although the delay requirements are valid only for perfect network conditions, in practice a jitter buffer will usually keep a reserve in order to cope with any sudden increase of packet interarrival delay variations (jitter) and/or clock skew between sender and receiver: This is calculated with 20 ms.
- VoIP DEC/VoIP ENC: Assuming 10 ms codec frame size, the joint processing time of encoder and decoder is 10ms due to the realtime constraints. Thus VoIP encoder and decoder are calculated with 5 ms processing time each. 10 ms delay have been added to the VoIP decoder as reserve, whereas 2,5 ms algorithmic delay have been added to the VoIP encoder.

- DSP: 1,5 ms delay contribution are added caused by any required signal processing in FP send and receive direction and in PP receive direction. In PP send direction, echo cancellation is mandatory in order to achieve the TCL(w) requirements (>55 dB), 11 ms have been added for this task. Another contribution of 10 ms originates from the framing (accumulation of samples until a block of 10 ms is collected). Finally, a reserve of 5 ms has been assigned such that the DSP block in PP send direction sums up to 26 ms.

- DECT ENC/DEC: Due to realtime constraints the joint delay of DECT encoder and decoder cannot exceed 10 ms, thus each of them is calculated as contributing 5 ms. The encoder is calculated with additional algorithmic delay of 2,5 ms.

- RTP pack: Commonly RTP packets contain data representing 20 ms of audio date (RTP pTime), thus another 10 ms delay have been calculated. This is the amount by which any first encoded frame in an RTP packet has to be delayed in order to form the 20 ms RTP packet combined with any second encoded frame.

- The DECT transmission delay has been calculated with 5 ms in each direction.

- The sound propagation time from the electro-acoustic interface (D/A) to the ERP and from the MRP to the electro-acoustic interface (A/D) have been calculated with 0,5 ms for handset/headset (HS) applications (very short distance) and with 3,5 ms for loudspeaking/handsfree (HF) applications (longer distance).

The round-trip delay values shown in figure F.1 include 5 ms delay for looping back the signal in the respective counterpart.

In case a PP of type 1d, 2c, 3b and 4b is connected to an FP other than type 3 or 5, the round-trip delays of FP and PP can be determined separately. Else, if connected to an FP with VoIP interface (type 3 or 5), the recommendation is to determine and assess the round-trip delay of the complete terminal as shown in figure F.1 ("VoIP-FP + PP round-trip delay"). Hence only requirements for the FP+PP roundtrip delays are given for these FP types.

Technically, PPs of type 5a, 5b, 7a-h, 7j, 8a and 8b can only connect to an FP of type 5. Thus the recommendation is to determine and assess the round-trip delay of the complete terminal as shown in figure F.1 ("VoIP-FP + PP round-trip delay").

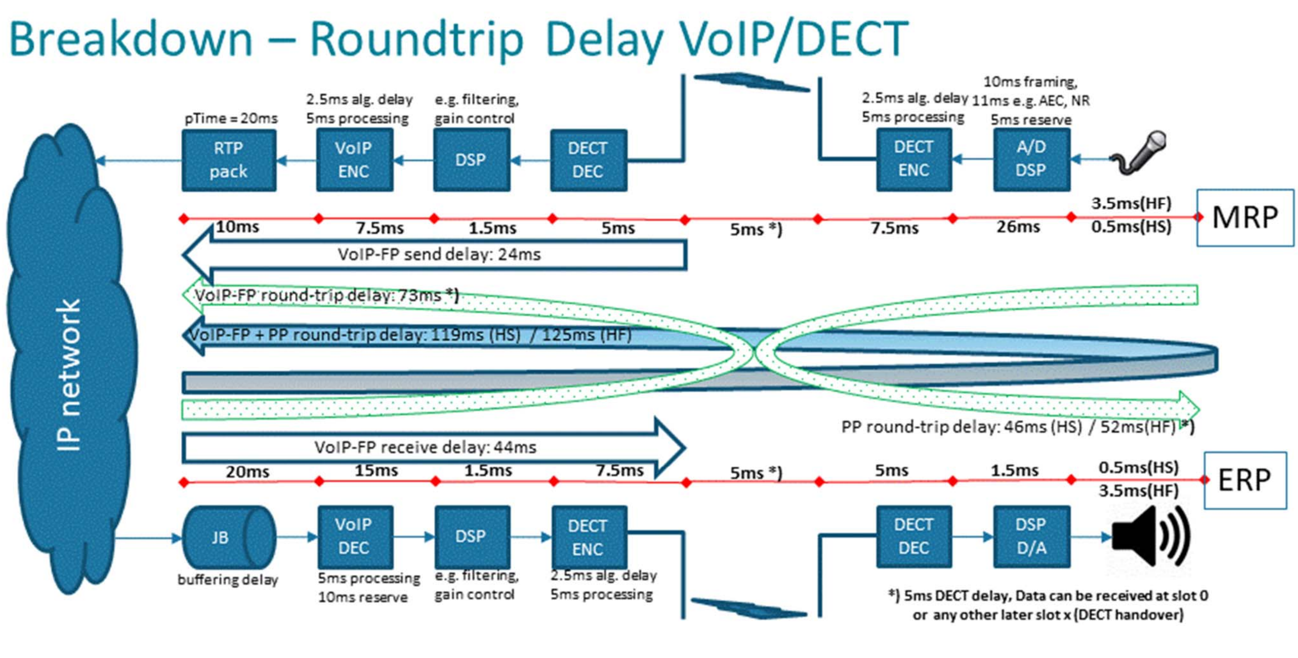


Figure F.1: Delay breakdown - PP and VoIP FP

Annex G (normative): Allowed tolerances

G.1 Description of the allowed frequency gap in the lower limit frequency response mask for wideband, super-wideband and fullband applications

Within the frequency range specified in table G.1, it is allowed that the frequency response of the device under test falls once (for WB, SWB) or twice (for FB) 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 centre frequency of the allowed gap.

Table G.1: Allowed frequency gaps per signal bandwidth

Bandwidth	Number of allowed gaps	Lower frequency limit for gaps	Upper Frequency limit for gaps
Wideband	1	3 400 Hz	7 000 Hz
Super-wideband	1	3 400 Hz	14 000 Hz
Fullband	2	3 400 Hz	20 000 Hz

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".
- ETSI TBR 010: "Digital Enhanced Cordless Telecommunications (DECT);General terminal attachment requirements: Telephony applications".
- Recommendation ITU-R BS.1534: "Method for the subjective assessment of intermediate quality levels of coding systems".
- Recommendation ITU-R BS.1116: "Methods for the subjective assessment of small impairments in audio systems including multichannel sound systems".

History

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