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EUROPEAN STANDARD

**Digital Enhanced Cordless Telecommunications (DECT);
Common Interface (CI);
Part 8: Speech and audio coding and transmission**

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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 is part 8 of a multi-part deliverable ([1] to [7]). Full details of the entire series can be found in part 1 [1].

Further details of the DECT system may be found in ETSI TR 101 178 [i.6] and ETSI ETR 043 [i.7].

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
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Modal verbs terminology

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1 Scope

The present document is one of the parts of the specification of the Digital Enhanced Cordless Telecommunications (DECT) Common Interface (CI).

This part of the DECT CI specifies the speech and audio coding and transmission requirements.

In order to ensure satisfactory interworking of different portable and fixed units, it is necessary to specify the transmission performance of the analog information over the digital link. This requires not only use of a common speech algorithm, but also standardization of frequency responses, reference speech levels (or loudness) at the air interface and various other parameters.

The present document applies to DECT equipment which includes all the necessary functions to provide real-time two-way speech conversation and stereo audio transmission. Several speech services are defined in the present document, including conventional 3,1 kHz telephony, wideband 7 kHz voice transmission, super-wideband 14 kHz and fullband 20 kHz service. DECT Fixed part providing such services may be connected to the public circuit switched (PSTN/ISDN) network, to private networks or to the Voice over Internet Protocol (VoIP) network.

Tethered fixed point local loop applications are not required to comply with the requirements of the present document.

For the DECT systems which connect to the Public Switched Telephone Network (PSTN) via an analog interface, the additional requirements, which are implemented in the FP, have as much as possible been aligned with ETSI TBR 038 [29].

A summary of the control and the use of the DECT echo control functions, to guide on need for options to manufacturers and installers, is found in annex A.

Information concerning test methods can be found in ETSI EN 300 176-1 [9] and ETSI EN 300 176-2 [10] (previously covered by ETSI TBR 010 [i.5]). The test methods take into account that DECT is a digital system.

The present document includes New Generation DECT, a further development of the DECT standard introducing wideband speech, improved data services, new slot types and other technical enhancements.

In addition, the present document includes DECT Evolution, providing SWB and FB speech and audio capabilities and a new speech coding algorithm for NB and WB allowing to increase the audio quality of the NB and WB speech service and improve bandwidth efficiency.

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.

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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]	Void.

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

[10]	ETSI EN 300 176-2: "Digital Enhanced Cordless Telecommunications (DECT); Test specification; Part 2: Audio and speech".

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

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

[13]	Recommendation ITU-T G.711 (1988): "Pulse Code Modulation (PCM) of voice frequencies".

[14]	Recommendation ITU-T G.722 (2012): "7 kHz audio-coding within 64 kbit/s".

[15]	Void.

[16]	Void.

[17]	Recommendation ITU-T G.729.1 (2006): "G.729-based Embedded Variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729".

[18]	Void.

[19]	ISO/IEC 14496-3:2009: "Information technology -- Coding of audio-visual objects -- Part 3: Audio" (ISO/IEC JTC1/SC29/WG11 (MPEG)).

[20]	Recommendation ITU-T P.311 (2011): "Transmission characteristics for wideband digital handset and headset telephones".

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

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

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

[24]	Recommendation ITU-T G.111: "Loudness Ratings (LRs) in an international connection".

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

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

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

[28]	Recommendation ITU-T P.51: "Artificial mouth".

[29]   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".

[30]   Recommendation ITU-T G.131: "Talker echo and its control".

[31]   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)".

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

[33]   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".

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

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

[36]   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".

[37]   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".

[38]   Void.

[39]   Void.

[40]   Void.

[41]   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".

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

[43]   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)".

[44]   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)".

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

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

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

[i.2]		IETF RFC 768 (STD 6): "User Datagram Protocol".

[i.3]		IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications".

[i.4]		Void.

[i.5]		ETSI TBR 010: "Digital Enhanced Cordless Telecommunications (DECT); General Terminal Attachment Requirements; Telephony Applications".

[i.6]		ETSI TR 101 178: "Digital Enhanced Cordless Telecommunications (DECT); A High Level Guide to the DECT Standardization".

[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 (2007): "Transmission impairments due to speech processing".

[i.14]		Recommendation ITU-T G.107 (2009): "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]		Recommendation ITU-T G.729: "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)".

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

[i.19]		Recommendation ITU-T G.164 (1988): "Echo suppressors".

[i.20]		Recommendation ITU-T G.165 (1993): "Echo cancellers".

[i.21]		Recommendation ITU-T G.168 (2009): "Digital network echo cancellers".

[i.22]		IEEE 802.11™: "IEEE Standard for 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.23]		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.24]		Recommendation ITU-T Y.1541: "Network performance objectives for IP-based services".

[i.25]　　　　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.26]　　　　Recommendation ITU-T P.310: " Transmission characteristics for narrow-band digital handset and headset telephones".

[i.27]　　　　TIA-920.130-A: "Telecommunications Telephone Terminal Equipment Transmission Requirements for Wideband Digital Wireline Telephones with Headset".

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.

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

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

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

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 methodology.

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 to 14 000 Hz, usually understood to be 20 to 20 000 Hz (adapted from [21])

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

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 [34]

NOTE: LC3plus operates at sample rates 8 kHz, 16 kHz, 24 kHz, 32 kHz and 48 kHz and provides an audio bandwidth of up to 16 kHz in super-wideband mode and an audio bandwidth of up to 24 kHz in fullband mode. The codec also contains error protection capabilities and a very performant packet loss algorithm.

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

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

super-wideband speech: voice service with enhanced quality compared to PCM G.711 [13] and allowing the transmission of a maximum vocal frequency of at least 14 kHz

talker's echo loudness rating: loss of the speaker's voice sound reaching his ear as a delayed echo

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 1: It is used for definition of reference points.

NOTE 2: It does not necessarily mean 8 kHz x 8 bit.

wideband speech: voice service with enhanced quality compared to PCM G.711 [13] and allowing the transmission of a vocal frequency range of at least 150 Hz to 7 kHz

Wi-Fi[TM]: 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 family of standards [i.22]

3.2   Symbols

For the purposes of the present document, the symbols given in Recommendations ITU-T P.10 [21], G.701 [11], ETSI EN 300 175-1 [1] and the following apply:

[.]	Enclosed value is provisional
D	D-value of terminal
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)
dBm0p	weighted dBm0

NOTE: According to Recommendation ITU-T O.41 [32].

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)
ep$<x>$	error protection classes, x = 1…4
I_e	equipment Impairment factor
kB	Kilobyte
$L_{S,min}$	minimum activation level (Sending Direction)
Tr,S,min	built-up time (Sending Direction)

3.3   Abbreviations

For the purposes of the present document, the abbreviations given in Recommendations ITU-T P.10 [21], G.701 [11], ETSI EN 300 175-1 [1] and the following apply:

$<TC>$	IE Terminal Capability
A/D	Analog/Digital
AAC	Advanced Audio Coding (MPEG)
AAC-LD	Advanced Audio Coding-Low Delay
ADPCM	Adaptive Differential Pulse Code Modulation
ADSL	Asymmetric Digital Subscriber Line
AH,R,dt	Attenuation Range in receiving direction during Double Talk
AH,S,dt	Attenuation Range in sending direction during Double Talk
AMR-NB	Adaptive Multi-Rate codec (NB)
AMR-WB	Adaptive Multi-Rate codec (WB)
ARL	Acoustic Reference Level
ATM	Asynchronous Transfer Mode
BRA	ISDN Basic Rate Access
BSS	Base Station Sub-system
CH	Change History
CI	Common Interface
CLRR	Circuit Loudness Rating, Receiving
CLRS	Circuit Loudness Rating, Sending
CRFP	Cordless Radio Fixed Part
CSS	Composite Source Signal
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
ER	Error Resilient (MPEG)
ERP	Ear Reference Point
ES	End System
EUT	Equipment Under Test
EVS	Enhanced Voice Service
FB	Fullband
FES	Far-End Signal
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
HFT	Hands-Free Terminal
IC	Integrated Circuit
IO	Input-Output
IP	Internet Protocol
ISDN	Integrated Services Digital Network
LAN	Local Area Network
LC3	Low Complexity Communication Codec
LD	Low Delay (MPEG)

LL_e	Local echo Loss
LOG	LOGarithmic
LRGP	Loudness Rating Guard-ring Position
LSTR	Listener SideTone Rating
MDCT	Modified Discrete Cosine Transform
MIPS	Million Instructions Per Second
MOS	Mean Opinion Score
MPEG	Moving Picture Expert Group
MRP	Mouth Reference Point
MSB	Most Significant Bit
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
PLC	Packet Loss Concealment
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
PTCL	Portable part TCL
RAM	Random Access Memory
ReFP	Reference Fixed Part
REP	REpeater Part
RePP	Reference Portable Part
RF	Radio Frequency
RFP	Radio Fixed Part
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 [12].

SLR	Sending Loudness Rating
SLR_H	Sending Loudness Rating of the Handset
STMR	SideTone Masking Rating
SWB	Super-wideband
T	Delay
TA	Terminal Adaptors
TCL	Terminal Coupling Loss
TCLw	weighted Terminal Coupling Loss
TCP	Transmission Control Protocol
TDBWE	Time Domain BandWidth Extension
TDM(A)	Time Division Multiple (Access)
TELR	Talker's Echo Loudness Rating
UDP	User Datagram Protocol
USB	Universal Serial Bus
VAD	Voice Activity Detection
VoIP	Voice over Internet Protocol
WB	Wideband
WMOPS	Weighted Millions of Operations Per Second
WRS	Wireless Relay Station

4   Configurations

4.1   Reference configuration

4.1.1   Basic configuration

The basic reference configuration for voice transmission over DECT is shown in figure 4.1.
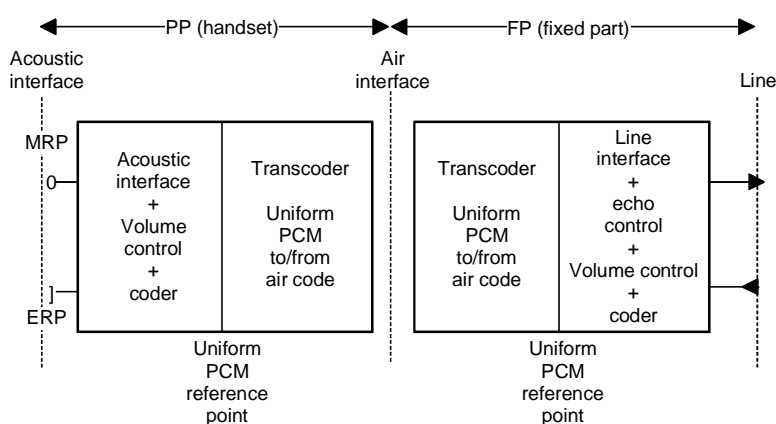
Figure 4.1: Basic reference configuration

The PP and the FP are delimited by physical interfaces:

- Acoustic: from the PP towards the speaker-listener, with the Ear and Mouth Reference Points (ERP and MRP).

- Air(RF):   the interface between PP and FP.

- Line: the interface from the FP towards the network.

Uniform PCM reference points shall be introduced in both the PP and the FP to materialize the limits of the transmission segment which is affected by the air interface coding scheme. This allows the transmission requirements to be specified independently from the coding at the air interface. The various transcoding algorithms are level-transparent, i.e. with an encoder and decoder connected in tandem, the "levels" of the digital signals at the uniform PCM input of the encoder and output from the decoder are identical.

NOTE 1:   The uniform PCM reference points are elements of the descriptive model and may physically exist in a particular implementation or not. For example A/D conversion, transcoding or signal processing without PCM conversion may be used or the reference points may be embedded within an IC, and thus will not be physically accessible.

NOTE 2:   Uniform PCM in the context of the present document means linear PCM with the necessary bit rate and resolution (number of bits) to handle the audio signal according to the codec bandwidth and the audio requirements specified in the present document. It does not mean 8 kHz x 8 bit (Recommendation ITU-T G.711 [13]) PCM. Note that the present document provides specification for narrowband, wideband, super-wideband and fullband services.

Each of the three sections thus delimited deals with distinct functions:

a)   the PP electro-acoustic section (between acoustic interface and PP uniform PCM reference point) which includes the acoustic interface, the volume control and the analog to digital conversion;

b)   the air-code section (between PP and FP uniform PCM reference points) considers the transcoding between uniform PCM and the air-code, as well as the transmission over the air interface;

c)   the FP line interface section (between FP uniform PCM reference point and line interface) comprises the interface to the line, digital or analog, and the FP voice processing, e.g. echo control.

In the case of analog interface, the FP performs the digital/analog conversion and the adaptation to the 2-wire analog line interface according to the requirements described in clauses 7.3.4 and 7.6.3 of the present document.

## 4.1.2		Portable Part (PP)

### 4.1.2.1		Functional organization

Figure 4.2 represents the basic functional organization of a PP from the voice transmission point of view.
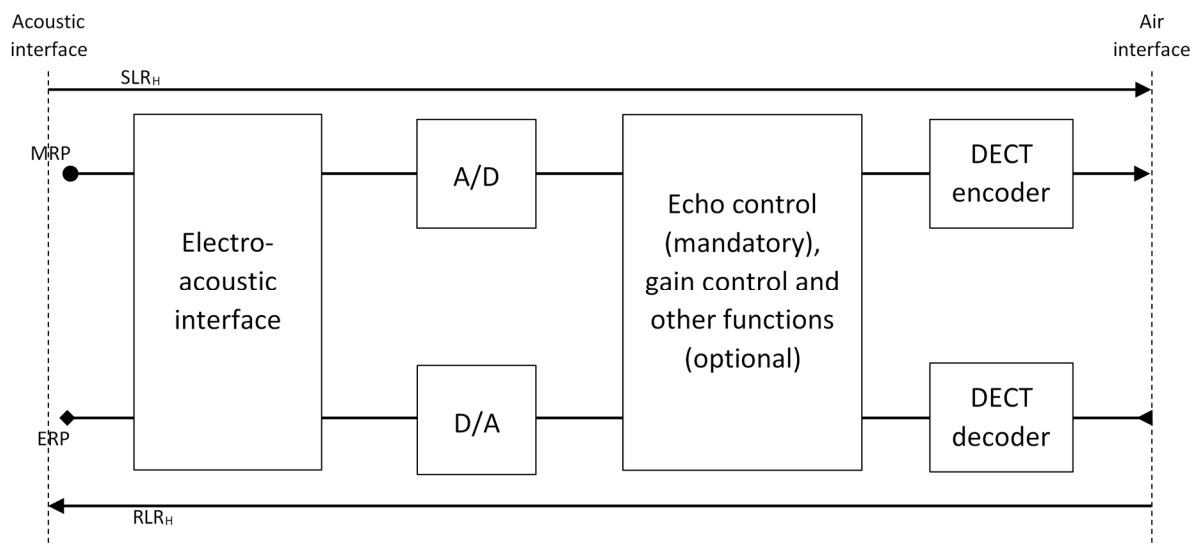
Figure 4.2: PP functional organization

The various functions represented in figure 4.2 are:

a)	the electro-acoustic function, implementing the interface to the transducers;

b)	the analog to digital and digital to analog converters;

c)	echo control (mandatory), gain control (optional) and other functions (optional);

d)	DECT encoder/decoder for conversion between air-code and uniform PCM.

NOTE 1:	Echo control is mandatory as the DECT system could be connected to a VoIP network.

NOTE 2:	See also notes 1 and 2 of clause 4.1.1.

### 4.1.2.2		Volume control

The volume control facility is optional, and can be implemented in a number of different ways:

1)	in the simplest implementation, it affects only the receiving level and is under the manual control of the PP user;

2)	the volume adjustment may also be adaptive, e.g. taking into account the average level of ambient noise at the MRP;

3)	the volume control may affect both receiving and sending levels in opposite directions. This has the advantage of reducing the influence on the terminal coupling loss when the receive volume is increased. Such an implementation would fit a noisy environment, as the need for an increased receiving volume is likely to come from a loud ambient noise, and it is then combined with a loud speech level.

4.1.3   Fixed Part (FP)

4.1.3.1   Digital interface

Figure 4.3 shows the functional model of a FP with a digital interface.
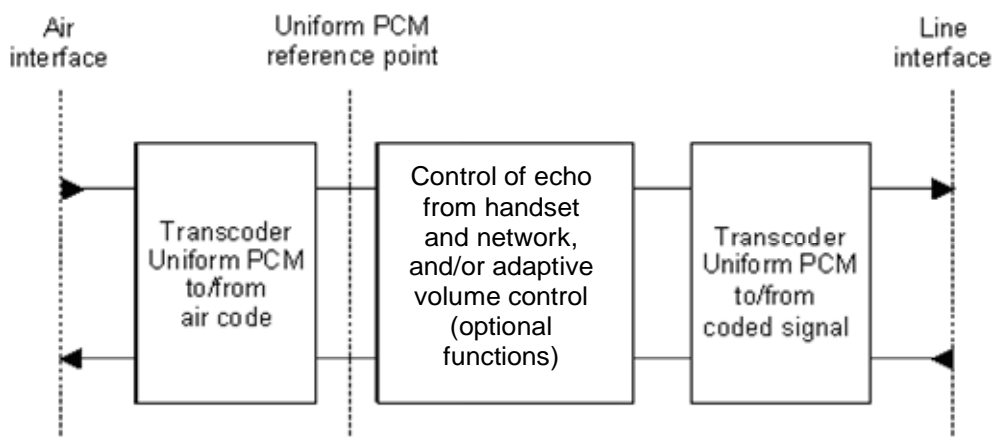
Figure 4.3: FP functional model including optional functions
(echo control and/or adaptive volume control)

The various functions represented in figure 4.3 are:

   a)   transcoder between air code and uniform PCM;

   b)   optional echo control functions which process echoes coming back from both the handset and/or from the network;

   c)   optional adaptive volume control function and other optional needed functions;

   d)   transcoder from uniform PCM to the coded signal used in the line interface.

   NOTE 1: See also notes 1 and 2 of clause 4.1.1.

Figure 4.4 shows a simplified functional model of a FP when:

   1)   no echo control or adaptive volume control are used; and

   2)   there is no need to refer to the internal PCM reference point.
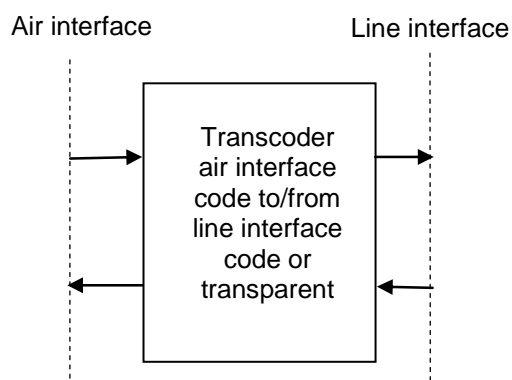
Figure 4.4: FP simplified functional model without optional functions

The various functions represented in figure 4.4 are:

    a)   transcoder between air-code and uniform PCM;

    b)   transcoder from uniform PCM to coded signal towards the line interface.

    NOTE 2: See also notes 1 and 2 of clause 4.1.1.

    NOTE 3: In the particular case when air interface and line interface codes are identical, no transcoding is necessary, and this simplified model becomes transparent.

Echo Control is mandatory if PP type 1a or 2a is connected (see clause 7.3.2.3.3), else optional, but if no echo control is implemented PP types 1a, 2a (TCLw $\leq$ 55dB) shall be rejected.

4.1.3.2   Analog interface

The implementation of a FP with an analog interface is shown in figure 4.5.
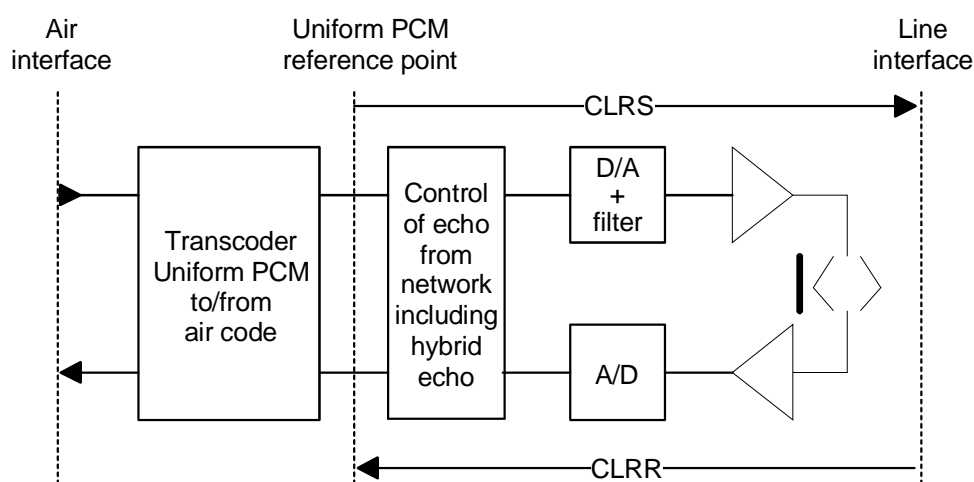
Figure 4.5: FP with an analog interface

The various functions represented in figure 4.5 are:

    a)   the line interface comprises the digital to analog coder-decoder and the adaptation to the line, that can be either 2- or 4-wire. The Circuit Loudness Ratings (CLRR and CLRS) are defined between uniform PCM reference point and the line interface;

    b)   in the case of a 2-wire line, the network echo controller deals both with the local hybrid echo and the far end network echo.

4.1.3.3   FP adaptive volume control

An adaptive dual volume adjustment, as described in clause 4.1.2.2, may optionally be implemented into the FP, for base stations where the associated PPs may be used in a noisy environment (e.g. public access).

4.1.3.4   Voice over IP (VoIP) interface

The implementation of an FP with VoIP interface is shown in figure 4.6.
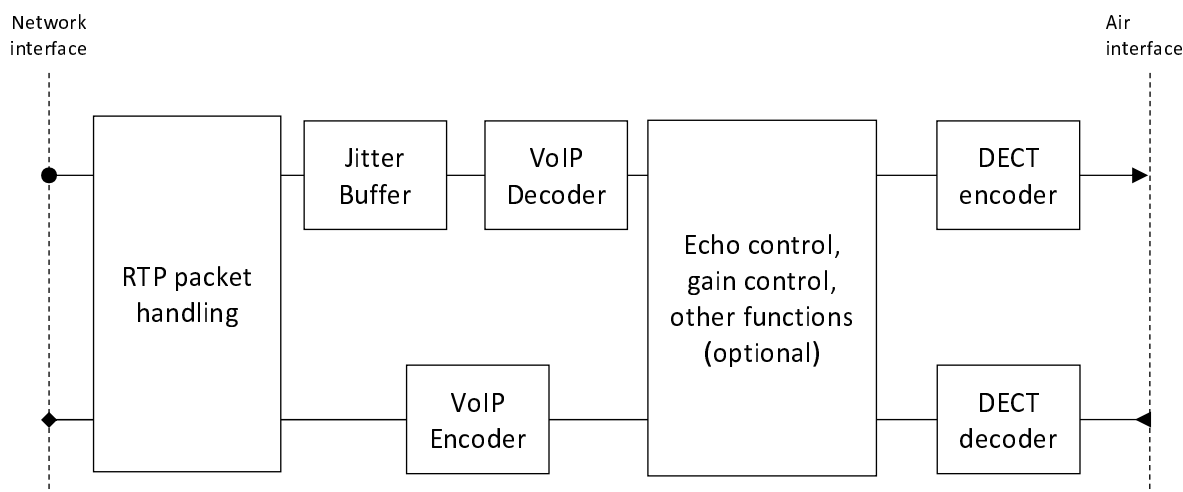
Figure 4.6: FP with VoIP interface

The various functions represented in figure 4.6 are:

a) RTP packet handling: In decoder direction (from network interface) this functional entity may (depending on the actual implementation of the Jitter Buffer/decoder interface) separate the RTP payload from the RTP metadata (e.g. sequence number, timestamp), may dissect the payload in case the RTP packetization time is different from the decoder's intrinsic frame size and may perform basic sanity checks for conformance to RTP standards (e.g. verify RTP version number etc.). In encoder direction (towards network interface) this entity may have to collect encoder output data until the RTP packetization time is reached. Furthermore it has to keep track of and add the RTP metadata (e.g. RTP timestamp and sequence number) to the payload received from the encoder.

b) A Jitter Buffer is required 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. Furthermore its task is to compensate for adverse network effects (e.g. handling of delay variations and packet reordering, triggering packet loss concealment in the VoIP decoder) in order to maintain acceptable speech quality.

c) The VoIP Decoder/Encoder blocks implement the conversion between the compressed speech format and the Uniform PCM format.

d) optional echo control, gain control and other functions.

e) DECT encoder/decoder for conversion between air-code and uniform PCM.

Echo Control is mandatory if PP type 1a or 2a is connected (see clause 7.3.2.3.3), else optional, but if no echo control is implemented PP types 1a, 2a (TCLw ≤ 55 dB) shall be rejected.

4.2   Test configurations

A complete DECT system comprises a FP and a PP. It is a requirement that either the complete system or each of the two separate parts shall be capable of being tested (see ETSI EN 300 176-2 [10]).

5   Encoding

5.0   General

This clause describes the different speech and audio codecs supported by DECT.

# 5.1	32 kbit/s full term (ADPCM) G.726

## 5.1.1	Algorithm

The speech coding algorithm shall conform to Recommendation ITU-T G.726 [12] for 32 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM).

The A-law companding and synchronous tandem adjustment may be omitted in FPs with an analog or digital VoIP line interface and in PPs.

NOTE:	Recommendation ITU-T G.726 [12] 32 kbit/s ADPCM 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 normally limited to 9,6 kbit/s.

## 5.1.2	Bit sequence

The ADPCM words comprised in each burst shall be transmitted in chronological order, and with the most significant bit transmitted first within each word.

## 5.1.3	Characteristics of G.726 ADPCM codec

The characteristics of G.726 [12] ADPCM codec are summarized in table 5.1.

Table 5.1: Recommendation ITU-T G.726 [12] ADPCM Speech codec

Standard	G.726 [12]
	ADPCM
Date	1990
Bandwidth	300 Hz to 3,4 kHz
Sampling rate	8 kHz
Bit rate (kbit/s)	32
Embedded Scalability	No
Type	ADPCM
Frame size	0,125 ms
Algorithmic Delay	0,125 ms
Complexity	≈12 MIPS
RAM (kB)	≈0,3

# 5.2	64 kbit/s PCM G.711

## 5.2.1	Algorithm

Recommendation ITU-T G.711 [13] for 64 kbit/s Pulse Code Modulation (PCM) Narrow Band codec may be optionally implemented in DECT devices in order to improve quality of narrowband communications: slightly higher intrinsic voice quality and no transcoding for PSTN calls.

The two variants of G.711 [13], A-law and μ-law may be supported.

NOTE:	G.711 [13] 64 kbit/s PCM codec support the use of the voice channel for any telefax or modem service, including 56 kbit/s modems.

## 5.2.2	Bit sequence

The same rules for bitstream ordering as for G.726 (see clause 5.1.2) apply: chronological order and Most Significant Bit (MSB) first.

5.2.3   Characteristics of G.711 PCM codec

The characteristics of G.711 PCM codec are summarized in table 5.2.

Table 5.2: Recommendation ITU-T G.711 [13] PCM Speech codec

Standard	G.711 [13]
	PCM
Date	1972
Bandwidth	300 Hz to 3,4 kHz
Sampling rate	8 kHz
Bit rate (kbit/s)	64
Embedded Scalability	No
Type	LOG PCM
Frame size	0,125 ms
Algorithmic Delay	0,125 ms
Complexity	≈0,01 MIPS
RAM (kB)	≈0

5.2.4   Automatic detection of FAX/modem tone and switch to G.711

Systems implementing G.711 [13] and other codecs, could implement the detection of the 1 100 Hz (fax Calling Tone), 1 300 Hz (modem) and 2 100 Hz (fax/modem answer) audio tones and the in-call switching between the other codec and G.711.

5.3   Wideband speech codec G.722 at 64 kbit/s

5.3.0   General

Recommendation ITU-T G.722 codec [14] is chosen as mandatory wideband codec for New Generation DECT in order to greatly increase the voice quality by extending the bandwidth from narrowband to wideband. G.722 provides a high wideband quality at bit rate of 64 kbit/s with low complexity and very low delay.

5.3.1   Algorithm

DECT devices supporting wideband voice (16 kHz frequency sampling) shall comply with Recommendation ITU-T G.722 [14] speech coding recommendation at 64 kbit/s.

Recommendation ITU-T G.722 [14] supports a frequency range of 50 Hz to 7 kHz. The extension of the bandwidth from narrowband 300 Hz to 3,4 kHz to wideband 50 Hz to 7 kHz greatly increases the voice quality.

5.3.2   Bit sequence

The same rules for bitstream ordering as for G.726 (see clause 5.1.2) apply: chronological order and Most Significant Bit (MSB) first.

5.3.3   Characteristics of G.722 wideband codec

G.722 [14] provides a high wideband quality at bit rate of 64 kbit/s with limited complexity and very low delay. The characteristics of G.722 wideband codec are summarized in table 5.3.

Table 5.3: Recommendation ITU-T G.722 [14] Wideband Speech codec

Standard	G.722 [14]
	SB-ADPCM
Date	1988
Bandwidth	50 Hz to 7 kHz
Sampling rate	16 kHz
Bit rate (kbit/s)	64, 56, 48 (see note)
Embedded Scalability	Yes
Type	Sub-Band ADPCM
Frame size	0,125 ms
Algorithmic Delay	1,625 ms
Complexity	≈10 MIPS
RAM (kB)	1
NOTE: Bit rate of 64 kbit/s shall be used for NG-DECT.	

5.3.4 Optional Packet Loss Concealment algorithm (PLC)

Appendix III and Appendix IV of Recommendation ITU-T G.722 [14] are optional packet loss concealment (PLC) solutions extending G.722 decoder [14]. They may be implemented to improve voice quality in degraded transmission conditions where packets/frames may be lost (in IP network or on DECT air interface).

Appendices III and IV meet the same quality requirements but with two different quality/complexity trade-offs:

- Appendix III of Recommendation ITU-T G.722 [14] aims at maximizing the robustness at a price of additional complexity.

- Appendix IV of Recommendation ITU-T G.722 [14] offers an optimized complexity/quality trade off with almost no additional complexity compared with G.722 normal decoding (+0,07 WMOPS).

ITU-T formal subjective tests have demonstrated that both Appendices provide an important quality improvement compared with a basic PLC reference solution (frame repeat based): in 1 %, 3 % and 6 % Frame Error Rate conditions (bursty or random), quality is increased by 0,5 MOS to 0,7 MOS.

Since G.722 does not incorporate any mechanism to cope with lost frames/packets, use of a PLC algorithm is strongly recommended to avoid annoying effects in case of packet/frame losses.

5.4 Wideband speech codec G.729.1 up to 32 kbit/s

5.4.1 Algorithm

Recommendation ITU-T G.729.1 [17] ("G.729EV") codec operating up to 32 kbit/s bit rate, is supported as an optional codec for wideband speech in order to provide similar wideband quality at half the bit rate of G.722 [14]. This allows a better transport efficiency on network side and DECT air interface (fits in one DECT full slot). Further bandwidth saving is possible since the codec is scalable and can operate at much lower bit rates by simple truncation of the bitstream, however with reduction of quality. In addition, G.729.1 [17] allows seamless interoperability with largely deployed VoIP networks and terminals implementing Recommendation ITU-T G.729 [i.17] narrowband codec.

On the other hand, this codec requires more processing power and adds more delay than 64 kbit/s G.722.

5.4.2 Bit sequence

The bitstream ordering shall comply with the order defined in Recommendation ITU-T G.729.1 [17]. MSB is transmitted first.

5.4.3 Characteristics of G.729.1 codec

G.729.1 [17] encodes signals in frames of 20 ms at bitrates of 8 kbit/s, and from 12 kbit/s to 32 kbit/s per steps of 2 kbit/s, both in narrowband or in wideband from 14 kbit/s. The supported bit rate in DECT is 30 kbit/s in wideband mode.

The codec is scalable and delivers an embedded bitstream that can be truncated at any point of the transmission path to dynamically reduce the bit rate and further optimize the bandwidth usage (down to 8 kbit/s for narrowband and 14 kbit/s for wideband with 12 possible intermediate bit rates).

At the bit rate of 8 kbit/s G.729.1 [17] is fully bitstream compliant with Recommendation ITU-T G.729 [i.17]. As a consequence it can fully interoperate with largely deployed VoIP networks and terminals implementing G.729/G.729AB narrowband codecs [17].

G.729.1 [17] can either operate in wideband (16 kHz frequency sampling input/output signals) or in narrowband (8 kHz frequency sampling input/output signals).

Table 5.4: Recommendation ITU-T G.729.1 [17] Wideband Speech codec

Standard	G.729.1 [17]
Date	2006
Bandwidth	50 Hz to 4 kHz 50 Hz to 7 kHz (bit rates ≥ 14 kbit/s)
Sampling rate	8 kHz / 16 kHz
Bit rate (kbit/s)	8, 12, 14, 16, 18, 20, 22, 24, 26, 28, 30, 32
Embedded Scalability	Yes (interoperable at 8 kbit/s with G.729)
Type	EV-CELP Time Domain Bandwidth Extension (TDBWE) Transform Coding (MDCT)
Frame size	20 ms
Algorithmic Delay	48,9375 ms
Complexity	35,8 WMOPS based on new STL2005 (34,7 WMOPS based on STL2000)
RAM (kB)	17,4

5.4.4   Packet Loss Concealment algorithm (PLC)

Recommendation ITU-T G.729.1 [17] incorporates a high efficiency packet loss concealment mechanism.

5.4.5   Supported bit rate in DECT

The nominal supported bit rate of codec G.729.1 [17] over DECT is 30 kbit/s when transported used in combination with DLC service LU12, encapsulated as clause E.1 (see ETSI EN 300 175-4 [4]). The resulting LU12 frames are transported over 32 kbit/s DECT channels.

Under some circumstances the codec may produce less bit rate than the 30 kbit/s. This is also supported by the LU12 encapsulation mechanism (see ETSI EN 300 175-4 [4], clause E.1).

Other bitrates and transports are for further study.

5.5   Super-wideband MPEG-4 speech and audio coding

5.5.1   Algorithm

MPEG-4 ER AAC-LD operating at sampling rates of 48 kHz or 24 kHz is supported as optional codec for wideband and super-wideband speech. MPEG-4 ER AAC-LD is designed for high quality communication application including all kind of audio signals e.g. speech and music. It provides an audio bandwidth around 14 kHz at a bit rate of 64 kbit/s.

On the other hand, this codec requires more processing power and adds more delay than G.722 [14].

MPEG 4 ER AAC-LD (Error Resilient, Low Delay AAC profile) is standardized as an audio profile of MPEG-4 (ISO/IEC 14496-3 [19]).

DECT will support two modes: 64 kbit/s and 32 kbit/s with characteristics defined in clauses 5.5.2 and 5.5.3. The DECT configuration shall follow the limitations defined in [19].

5.5.2   64 kbit/s, MPEG-4 ER AAC-LD codec

MPEG-4 ER AAC-LD at 64 kbit/s is designed for high quality communication applications including all kind of audio signals e.g. speech and music. It provides an audio bandwidth of about 14 kHz at this bit rate of 64 kbit/s and a subjective quality close to FM radio. The packet size shall be 10 ms and algorithmic delay is 20 ms.

Table 5.5: 64 kbit/s, 14 kHz MPEG-4 ER AAC-LD Audio codec

Standard	MPEG-4 ER AAC-LD 64 kbit/s
Date	2000/2006
Recommended Bandwidth	14 kHz
Sampling rate	48 kHz
Bit rate (kbit/s)	64
Embedded Scalability	no
Type	perceptual audio codec
Frame size	10 ms (480 samples)
Algorithmic Delay	20 ms
Example Complexity	~25 MIPS (encoder) ~10 MIPS (decoder)
Example RAM (kB)	~28 kB (encoder) ~13 kB (decoder) IO Buffer not included

5.5.3   32 kbit/s, MPEG-4 ER AAC-LD codec

MPEG-4 ER AAC-LD may also be optionally used in 32 kbit/s mode. It provides a bandwidth of about 11,5 kHz. The packet size shall be 20 ms and the algorithmic delay is 40 ms.

Table 5.6: 32 kbit/s, 11,5 kHz MPEG-4 ER AAC-LD Audio codec

Standard	MPEG-4 ER AAC- LD 32 kbit/s
Date	2000/2006
Recommended Bandwidth	11,5 kHz
Sampling rate	24 kHz
Bit rate (kbit/s)	32
Embedded Scalability	No
Type	Perceptual audio codec
Frame size	20 ms (480 samples)
Algorithmic Delay	40 ms
Example Complexity	~13 MIPS (encoder) ~5 MIPS (decoder)
Example RAM (kB)	~28 kB (encoder) ~13 kB (decoder) IO Buffer not included

5.6 Narrowband, wideband and super-wideband/fullband LC3plus speech and audio coding

5.6.1 Algorithm

LC3plus operating at sample rates 8 kHz, 16 kHz, 24 kHz, 32 kHz and 48 kHz is supported as optional codec for narrowband, wideband and supported as mandatory codec for super-wideband and fullband. LC3plus is designed for very high quality communication application including all kind of audio signals, e.g. speech and music. It provides an audio bandwidth of up to 16 kHz in super-wideband mode and an audio bandwidth of up to 20 kHz in fullband mode, both at a gross bit rate of 64 kbit/s. In wideband mode an audio bandwidth of 8 kHz is provided at a gross bit rate of 32 kbit/s. In narrowband mode the audio bandwidth is up to 4 kHz at a gross bit rate of 32 kbit/s. Additionally, LC3plus can operate in high resolution mode allowing sample rates up to 96 kHz while providing an audio bandwidth up to Nyquist frequency.

LC3plus is standardized as ETSI TS 103 634: "Low Complexity Communication Codec plus (LC3plus); Standard for narrowband to fullband low delay audio communication" [34].

The specification contains forward error correction (FEC) capabilities (channel coder as described in A.1 of [34]) which shall be used for LC3plus transmission over DECT. At constant gross bit rate, this error protection (EP) comes with a reduction of the net bit rate available for source coding. The specification supports four EP classes, where "1" stands for lowest and "4" for highest protection capability. The LC3plus also includes a very performant packet loss algorithm which shall be used.

Due to the limitation by the DECT slot format, only a subset of the possible gross bit rates is actually supported for DECT application: NB at 32 kbit/s, WB at 32 kbit/s, SWB at 64 kbit/s and FB at 64 kbit/s, 96 kbit/s or 128 kbit/s with characteristics as defined in clauses 5.6.2 to 5.6.7.

General characteristics of the LC3plus codec are listed in table 5.7, mode dependent characteristics are listed in the respective clause.

NOTE 1: For ep2 to ep4 the source coding's complexity reduction for lower net bitrates can overcompensate the increasing complexity of the channel coding. Hence the overall complexity can decrease with increasing protection class for ep2 to ep4. All complexity figures involve packet loss concealment for a packet loss rate of 10 %.

NOTE 2: Rules for ep class selection (channel coding) are for further study.

NOTE 3: Though the characteristics presented in this clause only apply to 10 ms frame size, DECT slot formats for 2,5 ms and 5 ms can be supported with LC3plus.

NOTE 4: The characteristics presented in this clause do not include the high resolution mode.

Table 5.7: LC3plus [34] codec, general characteristics

Standard	LC3plus [34]
Date	2018
Bandwidth	0 kHz to 20 kHz (FB)
Sampling rate	8, 16, 24, 32 and 48 kHz
Gross bit rate	16 kbit/s to 320 kbit/s
Type	transform based perceptual audio codec
Frame size	10 ms, 5 ms, 2,5 ms
Algorithmic delay	2,5 ms
Rate adaptation	supported

5.6.2 NB mode at 32 kbit/s gross bit rate

This mode is used for NB speech coding at 32 kbit/s gross bit rate.

Table 5.8: LC3plus [34] codec, NB mode

Standard	LC3plus [34]
Bandwidth	0 kHz to 4 kHz
Sampling rate	8 kHz
Gross bit rate	32 kbit/s
Complexity (for error protection classes 1…4)	ep1: 7,7 WMOPS ep2: 7,9 WMOPS ep3: 7,8 WMOPS ep4: 7,7 WMOPS (see NOTE 1)
RAM	6,8 kB (data); 153,7 kB (code)

5.6.3   WB mode at 32 kbit/s gross bit rate

This mode is used for WB speech coding at 32 kbit/s gross bit rate.

Table 5.9: LC3plus [34] codec, WB mode

Standard	LC3plus [34]
Bandwidth	0 kHz to 8 kHz
Sampling rate	16 kHz
Gross bit rate	32 kbit/s
Complexity (for error protection classes 1…4)	ep1: 11,3 WMOPS ep2: 11,5 WMOPS ep3: 11,4 WMOPS ep4: 11,2 WMOPS (see NOTE 1)
RAM	7,7 kB (data); 154,6kB (code)

5.6.4   SWB mode at 64 kbit/s gross bit rate

This mode is used for SWB speech coding at 64 kbit/s gross bit rate.

Table 5.10: LC3plus [34] codec, SWB mode

Standard	LC3plus [34]
Bandwidth	0 kHz to 16 kHz
Sampling rate	32 kHz
Gross bit rate	64 kbit/s
Complexity (for error protection classes 1…4)	ep1: 20,3 WMOPS ep2: 20,0 WMOPS ep3: 20,2 WMOPS ep4: 19,9 WMOPS (see NOTE 1)
RAM	12,1 kB (data); 159,1 kB (code)

5.6.5   FB mode at 64 kbit/s gross bit rate

This mode is used for FB coding at 64 kbit/s gross bit rate.

Table 5.11: LC3plus [34] codec, FB mode at 64 kbit/s

Standard	LC3plus [34]
Bandwidth	0 kHz to 20 kHz
Sampling rate	48 kHz
Gross bit rate	64 kbit/s
Complexity (for error protection classes 1…4)	ep1: 26,4 WMOPS ep2: 27,9 WMOPS ep3: 28,0 WMOPS ep4: 27,6 WMOPS (see NOTE 1)
RAM	16,8 kB (data); 163,7 kB (code)

5.6.6   FB mode at 96 kbit/s gross bit rate

This mode is used for FB coding at 96 kbit/s gross bit rate.

Table 5.12: LC3plus [34] codec, FB mode at 96 kbit/s

Standard	LC3plus [34]
Bandwidth	0 kHz to 20 kHz
Sampling rate	48 kHz
Gross bit rate	96 kbit/s
Complexity (for error protection classes 1…4)	ep1: 25,3 WMOPS ep2: 29,0 WMOPS ep3: 28,6 WMOPS ep4: 28,6 WMOPS (see NOTE 1)
RAM	16,8 kB (data); 163,7 kB (code)

5.6.7   FB mode at 128 kbit/s gross bit rate

This mode is used for FB coding at 128 kbit/s gross bit rate.

Table 5.13: LC3plus [34] codec, FB mode at 128 kbit/s

Standard	LC3plus [34]
Bandwidth	0 kHz to 20 kHz
Sampling rate	48 kHz
Gross bit rate	128 kbit/s
Complexity (for error protection classes 1…4)	ep1: 26,6 WMOPS ep2: 28,7 WMOPS ep3: 29,1 WMOPS ep4: 29,1 WMOPS (see NOTE 1)
RAM	16,8 kB (data); 163,7 kB (code)

6 Transmission aspects

6.1 Relative level

The digital line interface is a 0 dBr point according to 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.

6.2 Acoustic reference level

The acoustic reference level is defined as the acoustic level which corresponds to a level of -10 dBm0 at the digital interface.

6.3 Volume control

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

7 Audio specifications

7.1 Overall description

7.1.1 Introduction to DECT audio specifications

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

- Homogeneous end-user perception regarding acoustic levels and equalization between devices of different manufacturers.

- A minimum level of acoustic quality is guaranteed for all devices compliant to a DECT specification.

- Complete interoperability between DECT devices of different manufacturers.

- Control of the acoustic quality and behaviour in the conformance test process.

7.1.2 Introduction to the audio types

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

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

The current audio types are listed in clause 7.1.3.

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

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

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

- DECT handsets (PPs), with acoustic interface.

- DECT handsets (PPs) with acoustic interface operating in handsfree mode.

- DECT PPs with external electrical audio interface.

- Speaker handsfree or headset accessories connected to a PP.

- DECT RFPs, with any network interface.

- DECT RFPs operating an internal call or conference between two or more DECT PPs.

- DECT repeaters (REP).

The Audio types for devices with acoustic interface (handsets, handsfree and headsets) are acoustic specifications with an acoustic air interface at the user side. These are defined and tested using the classic artificial head methodology (see Recommendation ITU-T P.57 [26]) or the HATS methodology (see Recommendation ITU-T P.58 [23]).

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 ETSI EN 300 176-2 [10].

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

7.1.3   List of Audio types

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

Table 7.1: List of Audio types

Applicable to:	Type nr.	Type name	Clause	Remarks
PP	0	Reference PP (RePP) narrowband	7.2.2	For test purposes
	1a	"Classic" GAP handset narrowband	7.2.3	This type could produce echo issues in combination with VoIP or "long delay" networks
	1b	"Improved" GAP handset narrowband	7.2.4	
	1c	HATS tested, "standard" narrowband handset	7.2.5	
	1d	HATS tested, "improved" narrowband handset	7.2.6	
	3a	HATS tested, "standard" narrowband loudspeaking and handsfree feature	7.2.7	
	3b	HATS tested, "improved" narrowband loudspeaking and handsfree feature	7.2.8	
	2a	Recommendation ITU-T P.311 [20] tested, wideband handset or headset	7.2.9	Type 2a could produce echo issues in combination with VoIP or long delay networks
	2b	HATS tested, "standard" wideband handset or headset	7.2.10	
	2c	HATS tested, "improved" wideband handset or headset	7.2.11	
	4a	HATS tested, "standard" wideband loudspeaking and handsfree feature	7.2.12	
	4b	HATS tested, "improved" wideband loudspeaking and handsfree feature	7.2.13	
	5a	Super-wideband 14 kHz handset or headset	7.2.14	
	5b	Super-wideband 14 kHz handsfree	7.2.15	
	6	PPs with external 2 wire, 3,1 kHz telephony interface	7.2.16	See also annex B

Applicable to:	Type nr.	Type name	Clause	Remarks
	7a	Fullband handset or headset	7.2.17	
	7b	Fullband handsfree	7.2.18	
	7c,7d,7e	Fullband stereo device	7.2.19	
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 or fullband Fixed Part	7.3.7	
	6a	FP handling an Internal call inside a DECT FP (any service)	7.3.8	Internal call
	6b	FP handling an n-party conference inside a DECT FP (any service)	7.3.9	Internal conference bridge
REP	7	DECT Repeater part (REP)	7.3.10	

7.1.4 Audio types for Portable Parts

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

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

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

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

Type 2a introduces wideband (7 kHz) voice using Recommendation ITU-T P.311 [20] for requirements and testing.

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

Types 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 handset or headset (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 and 7e concerning respectively a device for 20 kHz fullband stereo audio service at

- 7c: 128 kbit/s gross bit rate (64 kbit/s per stereo channel; two DECT long slots)

- 7d: 192 kbit/s gross bit rate (96 kbit/s per stereo channel; three DECT long slots)

- 7e: 256 kbit/s gross bit rate (128 kbit/s per stereo channel; four DECT long slots)

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

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

7.1.5 Audio types for Fixed Parts

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

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

Type 2 audio type is for FPs with PSTN interface providing general purpose 3,1 kHz telephony.

Type 3 audio type is for FPs with VoIP interface providing 3,1 kHz telephony.

Type 4 audio type is for FPs with ISDN interface providing wideband 7 kHz communication.

Type 5 audio type is for FPs with VoIP interface providing wideband, super-wideband or fullband communication.

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

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

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

7.1.6 Complete DECT system

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

7.1.7 Structure of the specification of the audio types

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

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

- **Audio type name:** in the header of each section in the table;

- **CH (Change History):** this column is included for traceability reasons. An "X" indicates that the requirement is new or that there have been any modification compared with the previous audio specification (the present document);

- **Clause number:** indicates the clause that contains the detailed specification of the requirement;

- **Requirement:** it is the requirement name. It matches with the title of the clause describing it;

- **M/O** (mandatory/optional) column**:** indicates the status of the requirement:

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

O:	indicates that this requirement is an optional requirement that may be included with the type (compatible) or not;

- **S/I:** indicates that there is a difference between standard and improved types (applies only to HATS defined PP types);

- **Comments:** for additional comments when needed.

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

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

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

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

Note that there are four audio services: narrowband (3,1 kHz), wideband (7 kHz), super-wideband (14 kHz) and fullband (20 kHz).

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

CH	Clause Number	Requirement	M/O	S/I	Comments
		Portable Part			
		PP type 1a: "Classic GAP" handset			
	7.2.3.3	General specification	M		
	7.5.1.1.1	Sending frequency response	M		
	7.5.1.1.2	Receiving frequency response	M		
	7.5.1.2.1	Nominal values for loudness ratings	M		
	7.5.1.2.2	User-controlled volume control in PP	M		
	7.5.1.2.3	PP adaptive volume control	O		
	7.5.1.3.1	Talker sidetone	M		
	7.5.1.3.2	Listener sidetone	M		
	7.5.1.4.1	Weighted Terminal Coupling Loss (TCLw)	M		
	7.5.1.4.2	Stability loss	M		
	7.5.1.5.1	Sending distortion	M		
	7.5.1.5.2	Receiving distortion	M		
	7.5.1.6.1	Out of band signals for sending	M		
	7.5.1.6.2	Out of band signals for receiving	M		
	7.5.1.7.1	Sending noise	M		
	7.5.1.7.2	Sending narrowband noise	M		
	7.5.1.7.3	Receiving noise	M		
	7.5.1.7.4	Level of sampling frequency (receiving)	M		
	7.5.1.8.1	Acoustic shock: continuous signal	M		
	7.5.1.8.2	Acoustic shock: peak signal	M		
	7.5.1.9	PP Delay	M		
	7.5.1.10	PP ambient noise rejection	O		
		PP type 1b: Improved GAP handset			
	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 AH,S,dt	M		
	7.5.2.3	Attenuation Range in Receiving Direction during Double Talk AH,R,dt	M		
	7.5.2.4	Activation in Sending Direction	M		
	7.5.2.5	Activation in Receiving Direction	M		

CH	Clause Number	Requirement	M/O	S/I	Comments
**Portable Part**				
**PP type 1c and 1d: HATS-tested narrowband handset**				
	7.2.5.3/ 7.2.6.3	General specification	M	X	
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		
	7.5.3.3.1	Sidetone masking rating (STMR)	M		
	7.5.3.3.3	Sidetone delay	M		
	7.5.3.4.1	TCLw	M		
	7.5.3.4.2	Stability loss	M		
	7.5.3.5	Distortion	M		
X	7.5.3.6	Out of band signals	M		
	7.5.3.7	Noise	M		
X	7.5.3.8	Acoustic shock	M		
X	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	X	
	7.5.4.2	Receive sensitivity/frequency response	M	X	
	7.5.4.3	Send loudness rating	M		
	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		
	7.5.4.9	Sending noise	M		
	7.5.4.10	Receiving noise	M		
	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	
X	7.5.4.13.2	Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$	M	X	
X	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		
X	7.5.4.14.2	Activation in Receiving Direction	O		

CH	Clause Number	Requirement	M/O	S/I	Comments
**Portable Part**				
	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		
X	7.5.4.16	Microphone mute	M		
X	7.5.4.17	Delay	M		
**PP type 2a: Recommendation ITU-T P.311 [20] 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		
	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	X	
X	7.5.6.1.1.2/ 7.5.6.1.2.2	Sending and receiving frequency responses - positional robustness	M		
	7.5.6.2.1	Send and receive loudness ratings - nominal values	M		
X	7.5.6.2.4	Microphone mute	M		
X	7.5.6.2.5	Send and receive loudness ratings - positional robustness	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		
	7.5.6.4.2	Stability loss	M		
	7.5.6.5	Distortion	M		
	7.5.6.6	Noise	M		
X	7.5.6.7	Acoustic shock	M		
	7.5.6.8	Delay	M		
	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

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part				
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		
	7.5.7.4	Receive loudness rating	M	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		
X	7.5.7.8	Out-of-band signals in receiving direction	M		
	7.5.7.9	Sending noise	M		
X	7.5.7.10	Receiving noise	M		
	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	
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		
X	7.5.7.14.2	Activation in Receiving Direction	O		
	7.5.7.14.3	Silence Suppression and Comfort Noise Generation	O		
	7.5.7.14.4	Performance in sending direction in the presence of background noise	O		
X	7.5.7.14.5	Speech Quality in the Presence of Background Noise	O		
X	7.5.7.14.6	Quality of Background Noise Transmission (with Far End Speech)	O		
	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		
X	7.5.7.17	Delay	M		
PP type 5a: super-wideband 14 kHz handset or headset				
X	7.2.14.3	General specification	M		
X	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		
X	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		

CH	Clause Number	Requirement	M/O	S/I	Comments
**Portable Part**				
X	7.5.8.2.1.3	Microphone mute	M		
X	7.5.8.3.1	Sidetone Masking Rating (STMR)	M		
X	7.5.8.3.2	Sidetone delay	M		
X	7.5.8.4.1	Unweighted Terminal Coupling Loss	M		
X	7.5.8.4.2	Stability loss	M		
X	7.5.8.5	Distortion	M		
X	7.5.8.6	Noise	M		
X	7.5.8.7	Acoustic shock	M		
X	7.5.8.8	Delay	M		
X	7.5.8.9	Double Talk Performance	M		
X	7.5.8.10	Switching characteristics	M		
X	7.5.8.11	Quality of echo cancellation	M		
**PP type 5b: super-wideband 14 kHz handsfree device**				
X	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		
X	7.5.9.5	Sending distortion	M		
X	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		
X	7.5.9.9.2	Stability Loss	M		
X	7.5.9.10	**Double Talk Performance**			
X	7.5.9.10.1	Attenuation Range in Sending Direction during Double Talk AH,S,dt	M		
X	7.5.9.10.2	Attenuation Range in Receiving Direction during Double Talk AH,R,dt	M		
X	7.5.9.10.3	Detection of Echo Components during Double Talk	M		
X	7.5.9.10.4	Minimum activation level and sensitivity of double talk detection	M		
X	7.5.9.11	**Switching characteristics**			
X	7.5.9.11.1	Activation in Sending Direction	M		
X	7.5.9.11.2	Silence Suppression and Comfort Noise Generation	O		
X	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		
X	7.5.9.12	**Quality of echo cancellation**			
X	7.5.9.12.1	Temporal echo effects	M		
X	7.5.9.12.2	Spectral Echo Attenuation	M		
X	7.5.9.12.3	Variable echo path	M		
X	7.5.9.13	Delay	M		
**PP type 6: PPs with external 2 wire, 3,1 kHz telephony interface**				
	Annex B	2-wire PP end system (informative)	M		Detailed specification informative only
**PP type 7a: fullband 20 kHz handset or headset**				
X	7.2.17.3	General specification	M		

CH	Clause Number	Requirement	M/O	S/I	Comments
**Portable Part**				
X	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		
X	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		
X	7.5.10.2.1.3	Microphone mute	M		
X	7.5.10.3.1	Sidetone Masking Rating	M		
X	7.5.10.3.2	Sidetone delay	M		
X	7.5.10.4.1	Unweighted Terminal Coupling Loss	M		
X	7.5.10.4.2	Stability loss	M		
X	7.5.10.5	Distortion	M		
X	7.5.10.6	Noise	M		
X	7.5.10.7	Acoustic shock	M		
X	7.5.10.8	Delay	M		
X	7.5.10.9	Double Talk Performance	M		
X	7.5.10.10	Switching characteristics	M		
X	7.5.10.11	Quality of echo cancellation	M		
**PP type 7b: fullband 20 kHz loudspeaking and handsfree device**				
X	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		
X	7.5.11.5	Sending distortion	M		
X	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		
X	7.5.11.10	**Double Talk Performance**			
X	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		
X	7.5.11.10.3	Detection of Echo Components during Double Talk	M		
X	7.5.11.10.4	Minimum activation level and sensitivity of double talk detection	M		
X	7.5.11.11	**Switching characteristics**			
X	7.5.11.11.1	Activation in Sending Direction	M		
X	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		
X	7.5.11.12	**Quality of echo cancellation**			
X	7.5.11.12.1	Temporal echo effects	M		
X	7.5.11.12.2	Spectral Echo Attenuation	M		
X	7.5.11.12.3	Variable echo path	M		
X	7.5.11.13	Delay	M		
**PP type 7c, d, e: fullband 20 kHz stereo audio device**				

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part					
X	7.2.19.3	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 in clause E.1)
Fixed Part					
FP type 1a: "classic" Fixed Part with ISDN interface, narrowband service					
	7.3.2.3.1	Transcoding and equalization	M		
	7.3.2.3.2	PP type detection	M		
	7.3.2.3.4	Activation of audio processing functions	M		
	7.6.1.1	Reduction of echo from PP	M		
	7.6.1.2	FP Network echo control	O		
	7.6.1.3	FP adaptive volume control	O		
	7.6.1.4	FP Delay	M		
FP type 1b: "new" Fixed Part with ISDN interface, narrowband service					
	7.3.3.3.1	Transcoding and equalization	M		
	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		
X	7.6.4.4	Clock accuracy	M		
X	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		
	7.6.2.2	FP Delay	M		
FP type 5: Fixed Part with VoIP interface, wideband, super-wideband or fullband 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		

CH	Clause Number	Requirement	M/O	S/I	Comments (see also table in clause E.1)
		Fixed Part			
	7.6.6.3	FP adaptive volume control	O		
X	7.6.6.4	Clock accuracy	M		
X	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 (TCLw). However, some specific requirements may be influenced by the codec in use. An example is the delay.

When this happens, a table of parameters is provided with the difference values depending on the codec. When there is no specific mention the provided figure should be understood as applicable for codecs G.726 [12] (narrowband), G.722 [14] (wideband) and LC3plus [34] (narrowband, wideband and super-wideband/fullband).

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 four audio services: 3,1 kHz narrowband telephony, 7 kHz wideband telephony, 14 kHz super-wideband audio and 20 kHz fullband audio, and two specification methodologies: artificial ear, according to Recommendation ITU-T P.57 [26], used in types 1a, 1b and 2a and HATS according to Recommendation ITU-T P.58 [23], used in types 1c, 1d, 2b, 2c, 3a, 3b, 4a, 4b, 5a, 5b, 7a, 7b, 7c, 7d and 7e. 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 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], 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 to Recommendation ITU-T P.311 [20].

- 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/fullband (14 kHz and 20 kHz services):

- PP Type 5a: ETSI TS 102 924 super-wideband handset or headset introducing super-wideband with performances tested according to ETSI TS 102 924 [36].

- PP Type 5b: ETSI TS 102 925 super-wideband headset introducing super-wideband loudspeaking handsfree function with performances tested according to ETSI TS 102 925 [37].

- PP Type 7a: ETSI TS 102 924 fullband handset or headset introducing fullband with performances tested according to ETSI TS 102 924 [36].

- PP Type 7b: ETSI TS 102 925 fullband handset or headset introducing fullband loudspeaking handsfree function with performances tested according to ETSI TS 102 925 [37].

- PP Type 7c, d, e: ETSI TS 102 924 fullband device introducing fullband stereo audio streaming function with performances tested according to ETSI TS 102 924 [36] / ETSI TS 102 925 [37].

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.
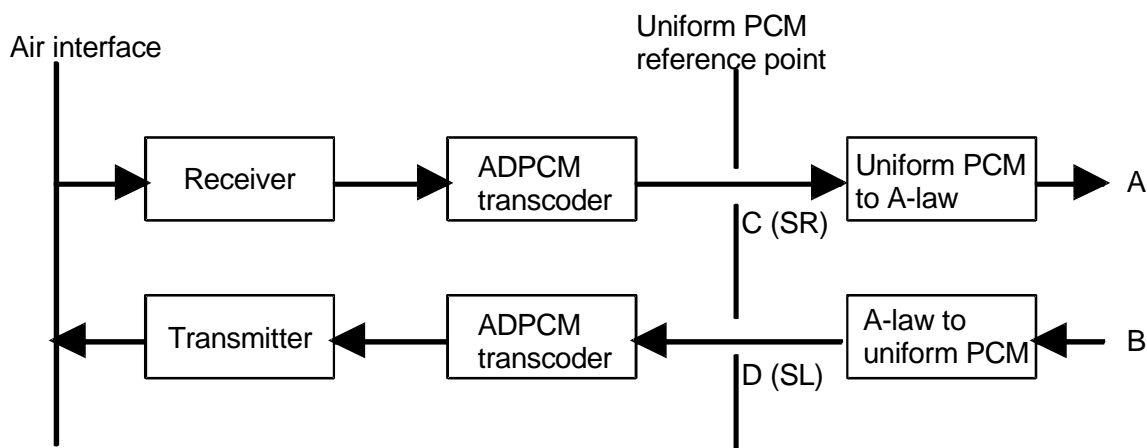
Figure 7.1: Reference PP (narrowband)

The RePP incorporates the specified transcoder algorithm as described in Recommendation ITU-T G.726 [12] at 32 kbit/s.

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

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

The reference PP shall have a value of TCLw of 36 dB $\pm$ 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 [12] 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 [12], G.711 [13] and LC3plus [34] 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 [28]. The artificial ear shall conform to Recommendation ITU-T P.57 [26].

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 available in clause E.1.

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

7.2.4.1   Introduction

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

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

This applies to either:

1) DECT equipment connected to VoIP networks; or

2) VoIP technology used by network operators.

7.2.4.2 Compatible services and codecs

Type 1b provides telephony 3,1 kHz service and is compatible with codecs G.726 [12], G.711 [13] and LC3plus [34] 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 [28]. The artificial ear shall conform to Recommendation ITU-T P.57 [26].

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 [12], G.711 [13] and LC3plus [34] 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 becoming the new standard for audio test methodology.

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

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

This specification is also applicable to headsets.

7.2.6.2   Compatible services and codecs

Type 1d provides telephony 3,1 kHz service and is compatible with codecs G.726 [12], G.711 [13] and LC3plus [34] 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 the present document.

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 [12], G.711 [13] and LC3plus [34] 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 [12], G.711 [13] and LC3plus [34] 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 [20] 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 [14], G.729.1 [17], MPEG-4 ER AAC-LD [19] operating at 32 kbit/s and LC3plus [34] in WB mode.

7.2.9.3   Specification

DECT type 2a PPs shall comply with all requirements of Recommendation ITU-T P.311 [20] as defined and with the exceptions given in clause 7.5.5.

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

    NOTE: The requirement of TCLw defined in Recommendation ITU-T P.311 [20] is 35 dB.

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

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

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

7.2.10.1   Introduction

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

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

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

7.2.10.2   Compatible services and codecs

Type 2b provides wideband telephony 7 kHz service and is compatible with codecs G.722 [14], G.729.1 [17], MPEG-4 ER AAC-LD [19] operating at 32 kbit/s and LC3plus [34] 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 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).

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

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

7.2.11.1   Introduction

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

7.2.11.2   Compatible services and codecs

Type 2c provides wideband telephony 7 kHz service and is compatible with codecs G.722 [14], G.729.1 [17], MPEG-4 ER AAC-LD [19] operating at 32 kbit/s and LC3plus [34] 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 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).

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

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

7.2.12.1 Introduction

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

1) specific PPs designed to operate in handsfree mode;

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

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

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

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

Type 4a device provides wideband 7 kHz (150 Hz to 7 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 [14], G.729.1 [17], MPEG-4 ER AAC-LD [19] operating at 32 kbit/s and LC3plus [34] 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 [14], G.729.1 [17], MPEG-4 ER AAC-LD [19] operating at 32 kbit/s and LC3plus [34] 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 [19] and LC3plus [34] in SWB mode, each operating at 64 kbit/s.

7.2.14.3   Specification

The complete specification of handsfree 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 5a 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 [19] and LC3plus [34] 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 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 [12], G.711 [13] and LC3plus [34] in NB mode.

7.2.16.3   Specification

The detailed specification for PPs type 6 is described in annex B.

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

7.2.17.1   Introduction

The type 7a super-wideband 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 [34] operating at 64 kbit/s.

7.2.17.3 Specification

The complete specification of handsfree 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 7a 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 [34] operating at 64 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 Type 7c, d, e: fullband 20 kHz stereo audio device

7.2.19.1 Introduction

The types 7c, 7d and 7e 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 [34] operating in fullband mode at 64 kbit/s for left and for right channel each.

Type 7d is compatible with LC3plus [34] operating in fullband mode at 96 kbit/s for left and for right channel each.

Type 7e is compatible with LC3plus [34] operating in fullband mode at 128 kbit/s for left and for right channel each.

7.2.19.3 Specification

The complete specification of handsfree types 7c, d, e 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).

7.3 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.22], ADSL, WIFI, USB, etc.). An informative table lists the most usual cases. Most detailed requirements are independent on the physical interface, however some of them (mainly the delay) depend or are influenced by this interface. When this happens, it is noted in the detailed specification text.

7.3.1   FP Type 0: Reference FP (ReFP)

A ReFP is shown in figure 7.2 and they incorporate the specified transcoder algorithm as described in Recommendation ITU-T G.726 [12] 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 [12] at 32 kbit/s.
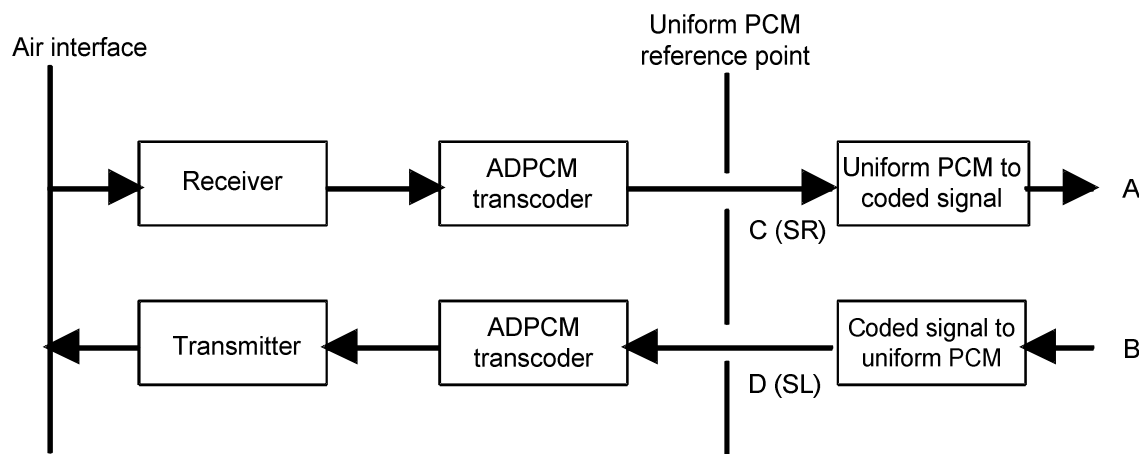
Figure 7.2: Reference FP

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 [12], G.711 [13] and LC3plus [34] in NB mode over air interface. Codec over line is interface is usually G.711 [13] (ISDN interface). In some cases, it could also be G.726 [12].

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 [12] or LC3plus [34] in NB mode from/to PCM G.711 [13] A-law or μ-law.

- Exception 1: no transcoding is needed if the air interface is using G.711 [13] 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 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 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 or 2a.

7.3.3.2   Compatible services and codecs

Type 1b provides telephony 3,1 kHz service and is compatible with codecs G.726 [12], G.711 [13] and LC3plus [34] in NB mode over air interface. Codec over line is interface is usually G.711 [13] (ISDN interface). In some cases, it could also be G.726 [12].

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 [12] or LC3plus [34] in NB mode from/to PCM G.711 [13] A-law or μ-law:

- Exception 1: no transcoding is needed if the air interface is using G.711 [13] 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 in clauses 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 (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 TCLw 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 7.3.3.3.1 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 [29] 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 [12], G.711 [13] and LC3plus [34] 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 [12], PCM G.711 [13] or LC3plus [34] 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 [12], G.711 [13] and LC3plus [34] 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 [13]. G.726 [12] and LC3plus [34] in NB mode are also allowed.

An example of a protocol stack for media is RTP [i.3] over UDP [i.2] over IP [i.1]. Other stacks may be allowed (e.g. with TCP). Any transport below IP is allowed.

Typical physical interfaces are:

- IEEE 802.3 [i.23].

- ADSL/VDSL over a phone line.

- Wifi or WiMAX radio i/f.

- USB.

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

Table 7.4: Examples of media stacks for VoIP narrowband interface

Example 1	Example 2	Example 3
G.711 [13], AMR-NB [43], LC3plus [34] in NB mode	G.711 [13], AMR-NB [43], LC3plus [34] in NB mode	G.711 [13], AMR-NB [43], LC3plus [34] in NB mode
RTP	RTP	RTP
UDP	UDP	UDP
IP	IP	IP
IEEE 802.3 [i.23]	IEEE 802.3 [i.23]	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 [12] or LC3plus [34] in NB mode from/to PCM G.711 [13] 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 TCLw 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 [14], G.729.1 [17], MPEG-4 ER AAC-LD [19] and LC3plus [34] in WB mode 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 (see note) < TCLw < 46 dB.

- PP with TCLw > 46 dB (Full TCLw).

- PP with TCLw > 55 dB (TCLw compatible with VoIP).

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

If the FP does not implement any 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 TCLw > 55 dB, THEN the FP SHALL NOT activate any echo cancellation or suppression facility for echo coming from the PP (see clause 7.4.2 or 7.4.3).

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

- If the PP has TCLw 42 dB < TCLw < 46 dB, THEN the FP SHALL activate the echo cancellation (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:

- 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.

7.3.7   FP Type 5: VoIP wideband, super-wideband or fullband 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) or 20 kHz (fullband) services.

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

7.3.7.2   Compatible services, physical interfaces and codecs

G.722 [14], G.729.1 [17], MPEG-4 ER AAC-LD [19] and LC3plus [34] 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.3] over UDP [i.2] over IP [i.1]. Other stacks may be allowed (e.g. with TCP). Any transport below IP is allowed.

Typical physical interfaces are:

- IEEE 802.3 [i.23].

- ADSL/VDSL over a phone line.

- Wifi or WiMAX radio i/f.

- USB.

There are several possible stacks on top of these physical interfaces. The present document 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 [14], AMR-WB [44], EVS [45], OPUS [46] or LC3plus [34] in WB mode	G.722 [14], AMR-WB [44], EVS [45], OPUS [46] or LC3plus [34] in WB mode	G.722 [14], AMR-WB [44], EVS [45], OPUS [46] or LC3plus [34] in WB mode
RTP	RTP	RTP
UDP	UDP	UDP
IP	IP	IP
IEEE 802.3 [i.23]	IEEE 802.3 [i.23]	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 (see note) < TCLw < 46 dB.

- PP with TCLw > 46 dB (Full TCLw).

- PP with TCLw > 55 dB (TCLw compatible with VoIP).

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

If the FP does not implement any 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 TCLw > 55 dB, THEN the FP SHALL NOT activate any echo cancellation or suppression facility for echo coming from the PP (see clause 7.4.2 or 7.4.3).

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

- If the PP has TCLw 42 dB < TCLw < 46 dB, THEN the FP SHALL activate the echo cancellation (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 - roundtrip delay

The FP may implement the following features:

- 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.

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

7.3.8.1   Introduction

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

This type applies to any service.

7.3.8.2   Compatible services, physical interfaces and codecs

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

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

There is no external interface, by definition.

7.3.8.3   Specification

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

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

7.3.9.1   Introduction

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

This type applies to any service.

7.3.9.2   Compatible services, physical interfaces and codecs

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

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

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

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

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

7.3.9.3   Specification for the conference bridge

This specification is for further study.

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

7.3.10   FP Type 7: DECT Repeater Part (REP)

7.3.10.1   Introduction

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

7.3.10.2   Compatible services, physical interfaces and codecs

Any air interface codec can be used with this feature.

7.3.10.3   Specification

The DECT Repeater 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 TCLw > 46 dB. When inserting the artificial echo loss the network echo cancellers took care of any echo cancellation needed (as for analog terminals), when providing TCLw > 46 dB there was no or only a low risk of echo for mostly all of the connections.

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

- For classic GAP DECT PPs (type 1a), additional echo loss is required since in the worst case the echo loss provided by a classic DECT PP is 34 dB. The only way to provide additional echo loss is to implement additional echo loss in the NG-DECT FP. It is required to provide at least 21 dB echo loss in addition to the echo loss provided by the classic DECT PP. Care has to be taken when implementing additional echo loss in the FP. Either echo suppression or echo cancellation can be used. Echo cancellation is the preferred solution since in principle it provides a better speech quality. The following points have to be considered when providing high quality echo cancellation in the FP:

  - Nonlinear echo path due e.g. to ADPCM Coding and other signal processing in the PP.

  - Low but (depending on delay) annoying echo signals from the classic DECT PP.

  - No additional switching should be detectable.

  - Echo loss during double talk needs to be maintained.

  - Sufficient echo loss needs to be provided in background noise situations.

- PP type 2a (wideband handset, P.311 based specification), may also have TCLw < 55 dB and may also require echo reduction at the FP. However, only 13 dB are required in the worst case, since PPs type 2a always have a TCLw > 42 dB by type definition (see clauses 7.2.9.3 and 7.5.5).

7.4.2   Echo canceller in Fixed Part

In the context of the present document, the term "echo canceller" means an echo reduction feature based on signal processing and placed in the 4-wire portion of the circuit, that works by estimating 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 [i.20] and [i.21]).

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

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

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

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

7.4.3   Echo suppressor in Fixed Part

In the context of the present document, the term "echo suppressor" means an echo reduction feature, also based on signal processing but simpler than the "echo canceller", that works by detecting if there is a voice signal going in one direction on a circuit, and then inserting a great deal of loss in the other direction (see [i.19]).

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

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

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

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

7.5   Transmission characteristics for Portable Parts

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

7.5.1.1   PP frequency responses

7.5.1.1.1   Sending

Requirement

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

Table 7.6: Sending sensitivity-frequency mask

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

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 $\pm 3{,}5$ 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:
See ETSI EN 300 176-2 [10].

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 [29]) 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 [29] 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 [29]).

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.1.3.2 Listener sidetone

Requirement

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

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

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

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

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

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

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.1.4 Terminal coupling loss

7.5.1.4.1 Weighted Terminal Coupling Loss (TCLw)

7.5.1.4.1.0 General

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).

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.1.4.1.1 Test conditions

The PP shall be suspended in free air.

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.

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.

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.1.5.3   Sidetone

Requirement

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

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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.

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.1.8 Acoustic shock

7.5.1.8.0 General

In order to fulfil the acoustic shock requirements, it is recommended to follow the guidelines of Recommendation ITU-T P.360 [33]. If needed the PP may have to implement some kind of hardware limiter.

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 [24]. 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:
See ETSI EN 300 176-2 [10].

### 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 behaviour 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 [22])	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 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:
See ETSI EN 300 176-2 [10].

### 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 behaviour 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 [22])	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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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

Frequency	Upper Limit	Lower Limit
100 Hz	-5	
300 Hz	5	- ∞
300 Hz	5	-5
3 400 Hz	5	-5
3 400 Hz	5	- ∞
3 758 Hz	5	
4 000 Hz	5	
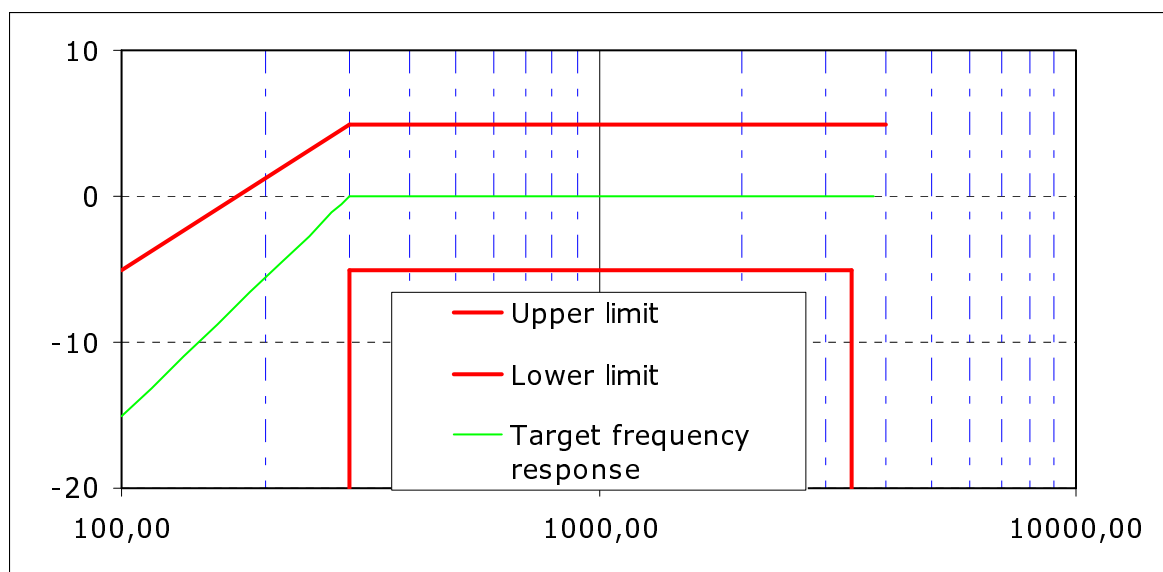
Figure 7.3: Send frequency response mask for PP types 1c and 1d

NOTE: The basis for the target frequency responses in sending and receiving is the orthotelephonic reference response which is measured between 2 subjects in 1 m distance under free field conditions and is assuming an ideal receive characteristic. Under these conditions the overall frequency response shows a rising slope. In opposite to other standards the present document no longer uses the ERP as the reference point for receiving but the diffuse-field. With the concept of diffuse-field based receive measurements a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a diffuse field based receiving frequency response.

The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale. The mask is a floating or "best fit" mask.

Measurement method:

See ETSI EN 300 176-2 [10].

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

See ETSI EN 300 176-2 [10].

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 8N 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 2 subjects in 1 m distance under free field conditions and is assuming an ideal receive characteristic. Under these conditions the overall frequency response shows a rising slope. In opposite to other standards the present document no longer uses the ERP as the reference point for receiving but the diffuse-field. With the concept of diffuse-field based receive measurements a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a diffuse field based receiving frequency response.

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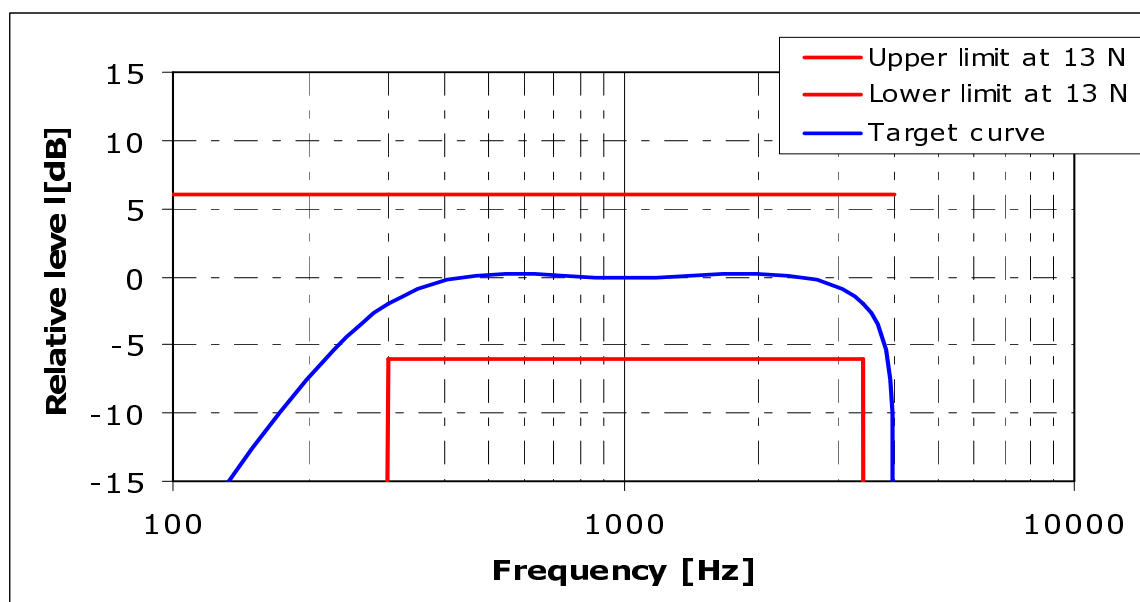
Figure 7.5: Receive frequency response mask for 2N and 13N application force

Figure 7.6: Void

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.3.1.2.2   Receive 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 7.15, 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:
See ETSI EN 300 176-2 [10].

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 $\pm$ 3 dB.

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.3.2.1.2   Receive Loudness Rating

Requirement:
The nominal value of Receive Loudness Rating (RLR) shall be:

RLR(set) = 2 dB $\pm$ 3 dB.

RLR (binaural headset) = 8 dB $\pm$ 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.3.4　　Terminal coupling loss

7.5.3.4.1　　Terminal Coupling Loss weighted (TCLw)

Requirement:
The TCLw shall be ≥ 55 dB.

With the volume control set to maximum TCLw shall be ≥ 46 dB.

It is recommended to set back the volume control to nominal level at the establishment of each new call, if TCLw does not reach 55 dB at the selected volume control.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 (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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 [33]. If needed the PP may have to implement some kind of hardware limiter.

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:
See ETSI EN 300 176-2 [10].

7.5.3.8.2 Void

7.5.3.9 Delay

The sum of the delays from the MRP to the air interface and from the air interface to the ERP (round-trip delay) shall not exceed 46 ms. This value includes the 5 ms delay of the reference FP looping back the digital signal towards the PP.

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

Measurement method:
See ETSI EN 300 176-2 [10].

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 (TELR) 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 [22] and P.502 [27]):

- 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 behaviour 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 [22])	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 7.23 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:
See ETSI EN 300 176-2 [10].

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 behaviour 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 [22])	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:
See ETSI EN 300 176-2 [10].

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 (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 7.25 are applicable (more information can be found in annex A of Recommendation ITU-T P.340 [22]).

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

Category (according to Recommendation ITU-T P.340 [22])	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:
See ETSI EN 300 176-2 [10].

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 $\leq$ -20 dBPa.

The built-up time $T_{r,S,min}$ (measured with minimum activation level) shall be $\leq$ 15 ms.

Measurement method:
See ETSI EN 300 176-2 [10].

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.

> NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

> NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.26.

Table 7.26: Mask for requirements for Spectral Adjustment of Comfort Noise

Frequency (Hz)	Upper Limit	Lower Limit
200	12	-12
800	12	-12
800	10	-10
2 000	10	-10
2 000	6	-6
4 000	6	-6
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

Measurement method:
See ETSI EN 300 176-2 [10].

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.25]. 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 of ETSI EN 300 176-2 [10] the following requirements shall apply:

- N-MOS-LQOn $\geq$ 3,5.

- S-MOS-LQOn $\geq$ 3,5.

- G-MOS-LQOn $\geq$ 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 of ETSI EN 300 176-2 [10].

Measurement method

See ETSI EN 300 176-2 [10].

### 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 receive direction. 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 of ETSI EN 300 176-2 [10].

> 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:

See ETSI EN 300 176-2 [10].

### 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	Δ S-MOS	Δ N-MOS
UP	≤ 0,2	≤ 0,2
DOWN	≤ 0,3	≤ [0,5]
AWAY	≤ [0,3]	≤ 0,4

Measurement method

See ETSI EN 300 176-2 [10].

### 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:

See ETSI EN 300 176-2 [10].

### 7.5.3.13.2	Spectral Echo Attenuation

Requirement:

The echo attenuation vs. frequency shall be below the tolerance mask given in table 7.27.

Table 7.27: Mask for echo attenuation vs. frequency

Frequency (Hz)	Upper Limit
100	-20
200	-30
300	-38
800	-34
1 500	-33
2 600	-24
4 000	-24
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale.	
NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.	

During the measurement it should be ensured that the measured signal is really the echo signal and not the comfort noise which possibly may be inserted in sending direction in order to mask the echo signal.

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.3.13.3   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. 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

See ETSI EN 300 176-2 [10].

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.		
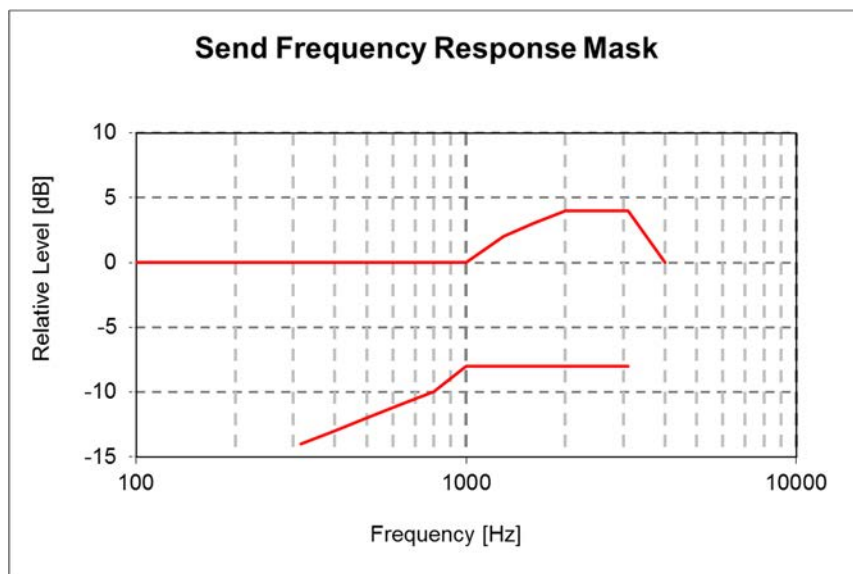
Figure 7.7: Sending sensitivity/frequency mask for HFT

Measurement method:
See ETSI EN 300 176-2 [10].

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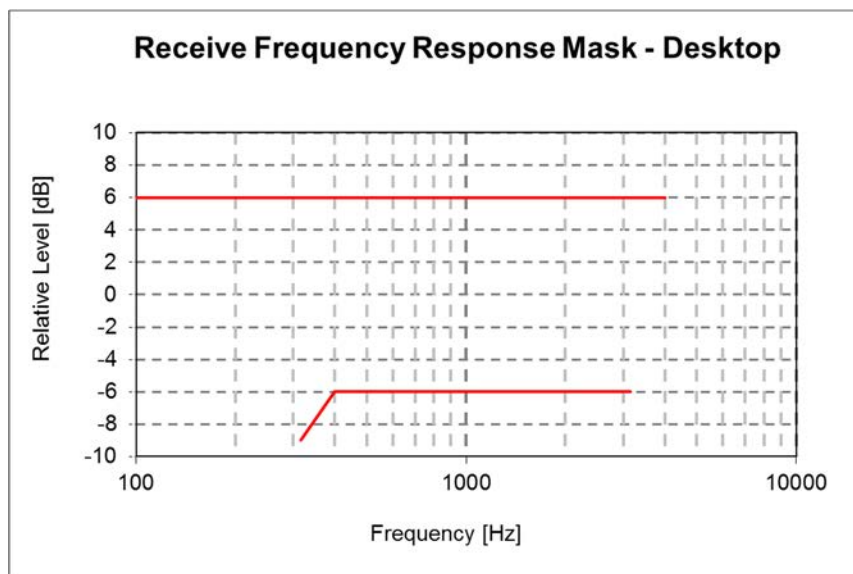
Figure 7.8: Receiving sensitivity/frequency mask for Desktop hands-free PP

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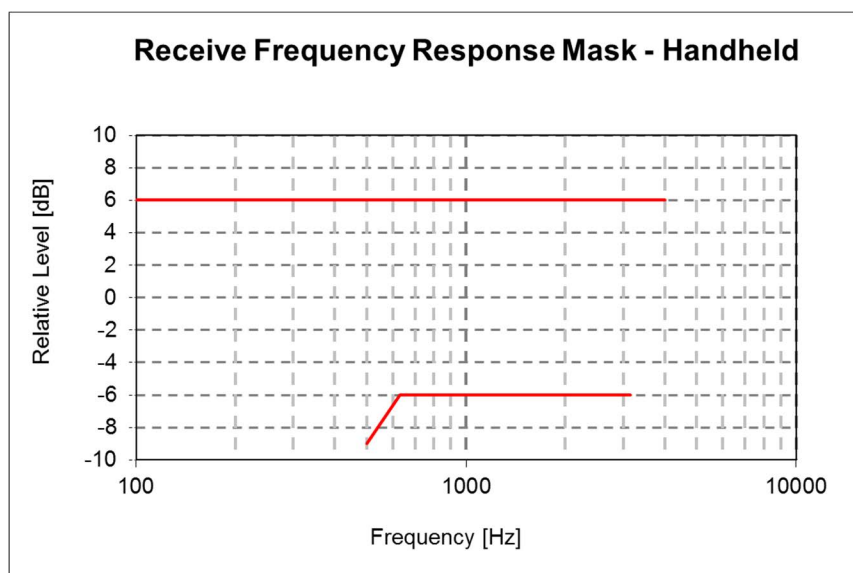
Figure 7.9: Receiving sensitivity/frequency mask for Hand-held PP

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:
See ETSI EN 300 176-2 [10].

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.26]. According to Recommendation ITU-T P.340 [22] 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:
See ETSI EN 300 176-2 [10].

7.5.4.4   Receive loudness rating

Requirement:

Desktop operated PP

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

Value of RLR at upper part of volume range 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: (RLR min - RLRmax) ≥ 15 dB.

Handheld operated PP

Improved class
Nominal value of RLR = 9 ± 3 dB. This value has to be fulfilled for one position of volume range.

Value of RLR at upper part of volume range 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: (RLR min - RLR max) ≥ 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: (RLR min - RLR max) ≥ 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.4.9 Sending noise

Requirement:
The limit for the maximum sending noise level shall be -64 dBm0p.

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

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:
See ETSI EN 300 176-2 [10].

7.5.4.10 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**.

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

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

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.4.11   Terminal Coupling Loss weighted (TCLw)

Requirement:

Improved class
In order to meet the G.131 [30] talker echo objective requirements, the recommended 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: Depending on the idle channel noise in the sending direction, it may not always be possible to measure an echo loss $\geq$ 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 [22] and P.502 [27]):

   •   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 behaviour 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 [22])	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 done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

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 behaviour 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 [22])	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.

This measurement shall be done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

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 [22]).

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

Category (according to Recommendation ITU-T P.340 [22])	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 done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

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:

See ETSI EN 300 176-2 [10].

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.

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	Lower Limit
200	12	-12
800	12	-12
800	10	-10
2 000	10	-10
2 000	6	-6
4 000	6	-6
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

Measurement method:

See ETSI EN 300 176-2 [10].

7.5.4.14.5   Speech Quality in the Presence of Background Noise

Speech Quality for wideband systems can be tested based on ETSI EG 202 396-3 [i.25]. 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 of ETSI EN 300 176-2 [10], 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 of ETSI EN 300 176-2 [10].

Measurement method:
See ETSI EN 300 176-2 [10].

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

Requirement:
The test is carried out applying a speech signal in receive direction. 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 of ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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. 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.4.17 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 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. As long as the measured PP+FP roundtrip delay fulfils the requirements given in clause 7.6.4, PP send or receive delays may be above the theoretical requirements.

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

See ETSI EN 300 176-2 [10].

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 [20]. They complete or replace the corresponding parameters of P 311 Recommendation when necessary.

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

7.5.5.1 Sending characteristics

7.5.5.1.1 Loudness rating

Requirement:
See Recommendation ITU-T P.311 [20], clause 4.1.

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

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.4.1.

7.5.5.1.2   Sensitivity/frequency characteristics

Requirement:
See Recommendation ITU-T P.311 [20], clause 4.2.

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.4.2.

7.5.5.1.3   Noise

Requirement:
See Recommendation ITU-T P.311 [20], clause 4.3.

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

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.4.3.

7.5.5.1.4   Distortion

Requirement:
See Recommendation ITU-T P.311 [20], clause 4.4.

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 [20], clause A.4.4.

7.5.5.1.5   Discrimination against out-of-band input signals

See Recommendation ITU-T P.311 [20], clause 4.5.

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.4.5.

7.5.5.2   Receiving characteristics

7.5.5.2.1   Loudness rating

Requirement:
See Recommendation ITU-T P.311 [20], clause 5.1.

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

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.5.1.

7.5.5.2.2   Sensitivity/frequency characteristics

Requirement:
See Recommendation ITU-T P.311 [20], clause 5.2.

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.5.2.

7.5.5.2.3   Noise

See Recommendation ITU-T P.311 [20], clause 5.3.

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.5.3.

7.5.5.2.4   Distortion

Requirement:
See Recommendation ITU-T P.311 [20], clause 5.4.

The measurements shall be done in the level range from +5 dBm0 to -20 dBm0 and the limits shall be:

Table 7.41: Mask for signal to distortion ratio

Receiving Level at the digital interface (dBm0)	Signal-to-distortion ratio limit (dB)		
	200 Hz	1 kHz	6 kHz
+5	0	35,0	29,0
+0 to -10	29,0	35,0	29,0
-20	27,0	27	0

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 [20], clause A.5.4.

7.5.5.2.5   Spurious out-of-band receiving signals

Requirement:
See Recommendation ITU-T P.311 [20], clause 5.5.

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.5.5.

7.5.5.3   Sidetone characteristics

7.5.5.3.1   Talker sidetone

Requirement:
See Recommendation ITU-T P.311 [20], clause 6.1.

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.6.1.

7.5.5.3.2   Sidetone distortion

Requirement:
See Recommendation ITU-T P.311 [20], clause 6.2.

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.6.2.

7.5.5.4   Echo path loss characteristics

7.5.5.4.1   Weighted terminal coupling loss

Requirement:
The requirements of Recommendation ITU-T P.311 [20], clause 7.1 shall apply with the following difference:

- The limit for TCLw shall be at least 42 dB.

This value differs from P 311 requirement (35 dB) in order to ensure better compatibility with long delay networks.

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.7.1.

7.5.5.4.2 Stability loss

Requirement:
See Recommendation ITU-T P.311 [20], clause 7.2.

Measurement method:
See Recommendation ITU-T P.311 [20], clause A.7.2.

7.5.6 Transmission characteristics for PP type 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	
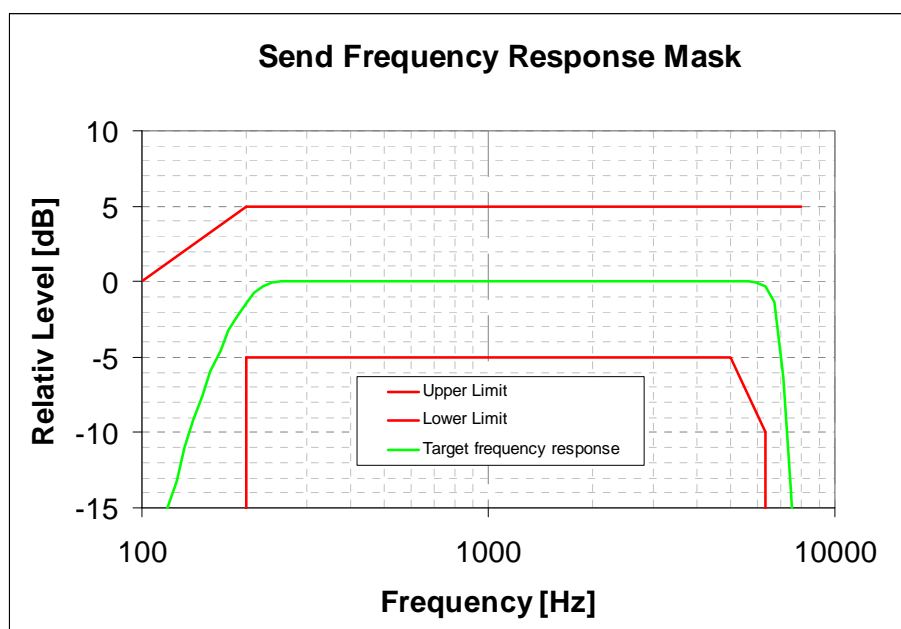
Figure 7.11: Send frequency response mask

NOTE 1: The basis for the target frequency responses in sending and receiving is the orthotelephonic reference response which is measured between 2 subjects in 1 m distance under free field conditions and is assuming an ideal receive characteristic. Under these conditions the overall frequency response shows a rising slope. In opposite to other standards the present document no longer uses the ERP as the reference point for receiving but the diffuse-field. With the concept of diffuse-field based receive measurements a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a diffuse-field based receiving frequency response.

NOTE 2: A "balanced" frequency response is preferable from the perception point of view. If frequency components in the low frequency domain are attenuated in a similar way frequency components in the high frequency domain should be attenuated.

NOTE 3: A gap, defined in annex G of ETSI EN 300 176-2 [10], 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 2N, 8N and 13N. This mask defined for 8N 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	

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
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.27].					

NOTE: A gap, defined in annex G of ETSI EN 300 176-2 [10], for the lower mask limit, is allowed.
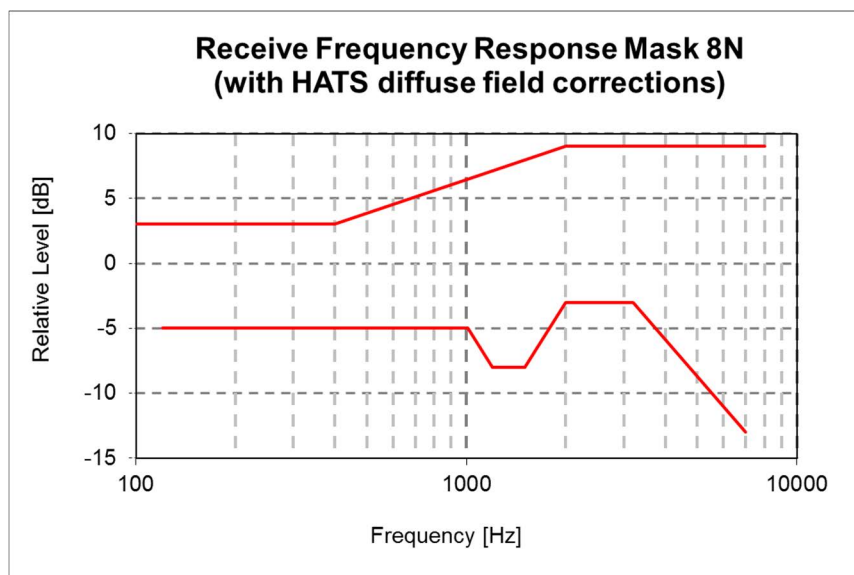
Figure 7.12: Receive frequency response mask for 8N application force

Receive Frequency Response Mask 13N
(with HATS diffuse field corrections)
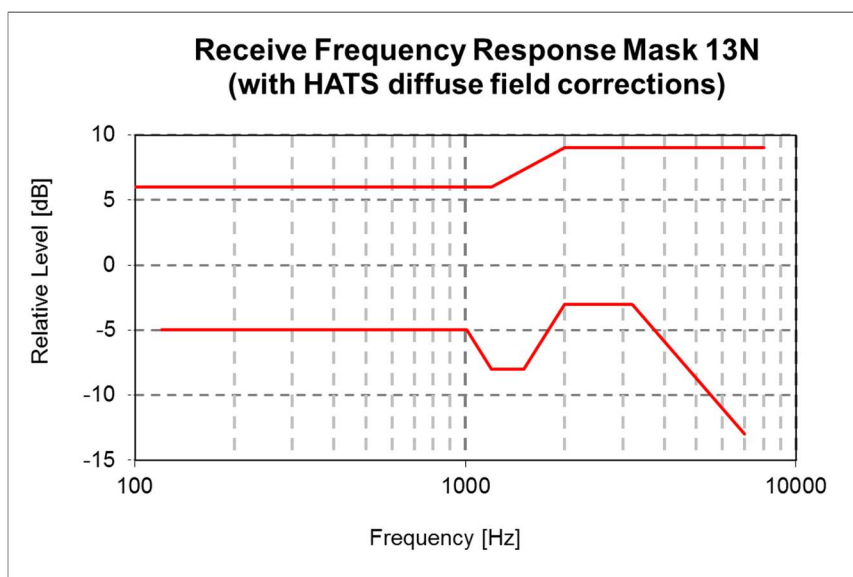
Figure 7.13: Receive frequency response mask for 13N application force

Receive Frequency Response Mask 2N
(with HATS diffuse field corrections)
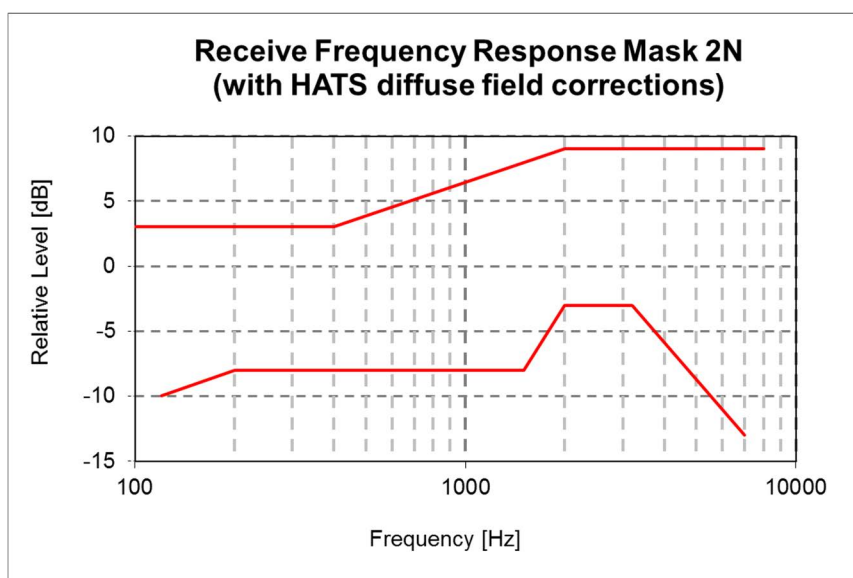
Figure 7.14: Receive frequency response mask for 2N application force

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:
See ETSI EN 300 176-2 [10].

7.5.6.1.2.2			Receive 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 7.43, 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:
See ETSI EN 300 176-2 [10].

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 dB $\pm$ 3 dB.

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.6.2.1.2   Receive Loudness Rating

Requirement:
The nominal value of Receive Loudness Rating (RLR) shall be:

- RLR(set) = 2 dB $\pm$ 3 dB.

- RLR (binaural headset) = 8 dB $\pm$ 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

### 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:
See ETSI EN 300 176-2 [10].

#### 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:
See ETSI EN 300 176-2 [10].

### 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:
See ETSI EN 300 176-2 [10].

#### 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:
See ETSI EN 300 176-2 [10].

### 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.

With the volume control set to maximum TCLw shall be ≥ 46 dB.

It is recommended to set back the volume control to nominal level at the establishment of each new call, if TCLw does not reach 55 dB at the selected volume control.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.6.5.2   Receiving Distortion

Requirement:
The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.47: Ratio of signal to harmonic distortion (receiving)

Frequency	Signal to distortion ratio limit, receiving
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 [33]. 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:
See ETSI EN 300 176-2 [10].

7.5.6.8   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 46 ms. This value includes the 5 ms delay of the reference FP looping back the digital signal towards the PP.

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

Measurement method:
See ETSI EN 300 176-2 [10].

### 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 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 [22] and P.502 [27]):

- 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 behaviour 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 [22])	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:
See ETSI EN 300 176-2 [10].

#### 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 behaviour 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 [22])	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:
See ETSI EN 300 176-2 [10].

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 (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 7.51 are applicable (more information can be found in annex A of Recommendation ITU-T P.340 [22]).

Table 7.51: Category regarding "duplex capability" depending on Echo Loss

Category (according to Recommendation ITU-T P.340 [22])	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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 psophometric weighting.

NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.52.

Table 7.52: Requirements for Spectral Adjustment of Comfort Noise (Mask)

Frequency (Hz)	Upper Limit	Lower Limit
200	12	-12
800	12	-12
800	10	-10
2 000	10	-10
2 000	6	-6
4 000	6	-6
8 000	6	-6
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

Measurement method:
See ETSI EN 300 176-2 [10].

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.25]. 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 of ETSI EN 300 176-2 [10] the following requirements shall apply:

- N-MOS-LQOw $\geq$ 3,5.

- S-MOS-LQOw $\geq$ 3,5.

- G-MOS-LQOw $\geq$ 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 of ETSI EN 300 176-2 [10].

Measurement method:
See ETSI EN 300 176-2 [10].

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. 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 of ETSI EN 300 176-2 [10].

    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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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. 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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.		
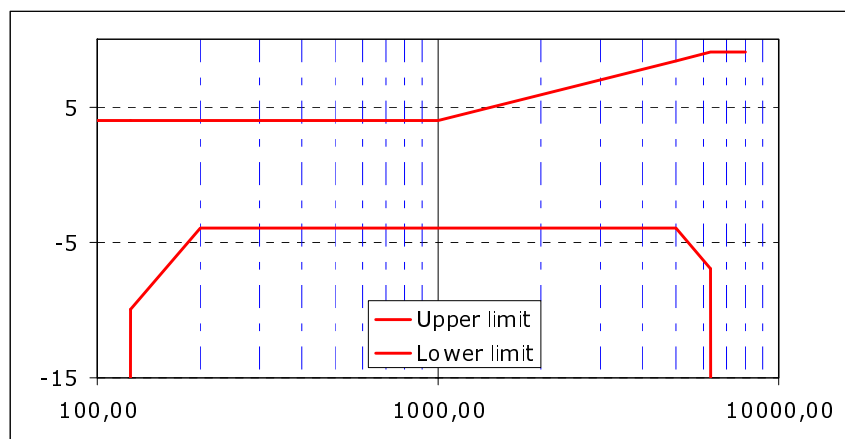
Figure 7.15: Sending sensitivity/frequency mask for HFT

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 of ETSI EN 300 176-2 [10], for the lower mask limit, is allowed.

Measurement method:
See ETSI EN 300 176-2 [10].

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 of ETSI EN 300 176-2 [10], for the lower mask limit, is allowed.
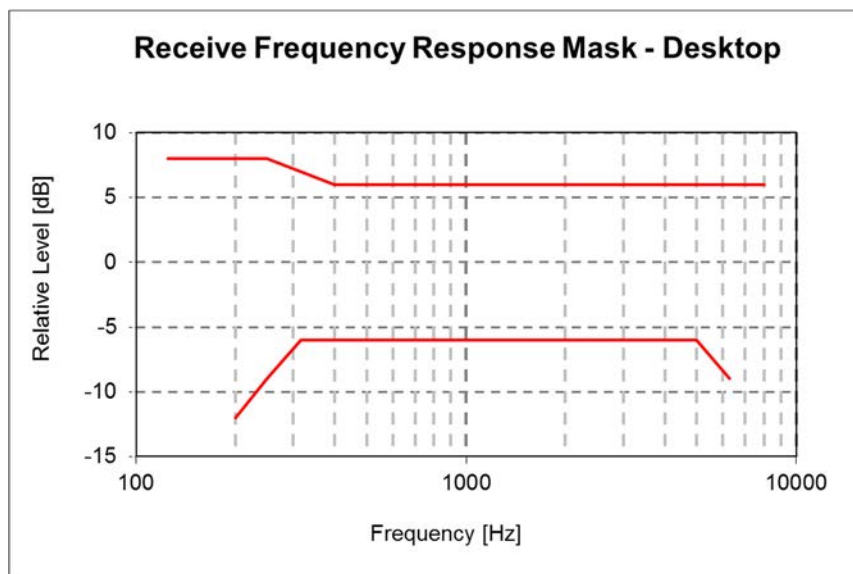
Figure 7.16: Receiving sensitivity/frequency mask for Desktop handsfree PP

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.		
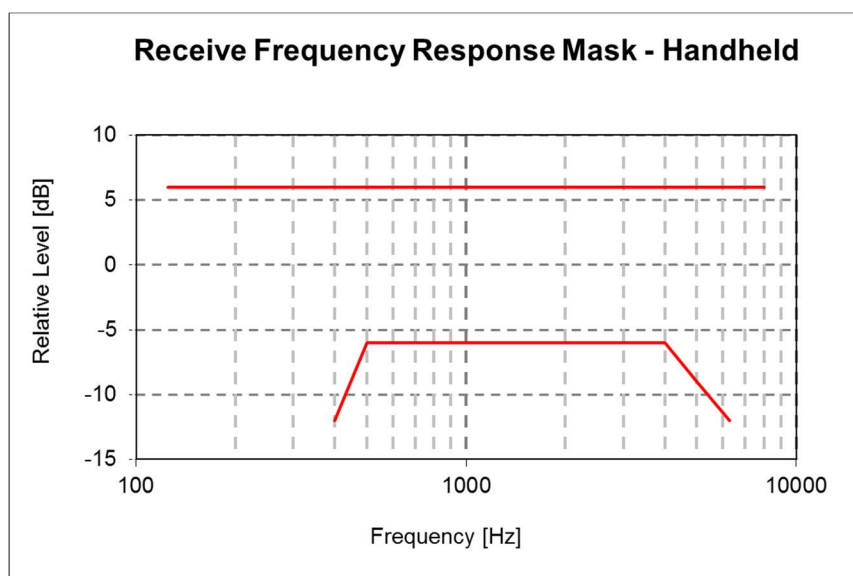
Figure 7.17: Receiving sensitivity/frequency mask for improved handheld PP

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.

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:
See ETSI EN 300 176-2 [10].

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.26]. According to Recommendation ITU-T P.340 [22] 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 [42] the loudness rating calculation as described in annex A is used.

Measurement method:
See ETSI EN 300 176-2 [10].

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: (RLR min - 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: (RLR min - 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: (RLR min - 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.7.6   Receiving distortion

Requirement:

Desktop and Handheld terminals

The ratio of signal to harmonic distortion shall be above the following mask for all types of terminals.

Table 7.59: Ratio of signal to harmonic distortion (receiving)

Frequency	Signal to distortion ratio limit, receiving for desktop terminal (dB)	Signal to distortion ratio limit, receiving for improved handheld terminal (dB)	Signal to distortion ratio limit, receiving for standard handheld terminal (dB)	Signal to distortion ratio limit, receiving for all terminals at maximum volume (dB)
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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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**.

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

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

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.7.11   Terminal Coupling Loss

Requirement:

Improved class

In order to meet the G.131 [30] talker echo objective requirements, the recommended weighted terminal coupling loss during single talk (TCLwst) 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 TCLwst shall be not less than 40 dB for the higher gain settings above the nominal setting of the volume control.

Standard class

TCLw shall be greater than 40 dB when measured under free field conditions at nominal setting of volume control.

TCLwst shall be not less than 34 dB for the higher gain settings above the nominal setting of the volume control.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 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 [22] and P.502 [27]):

- 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 behaviour 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 [22])	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 done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

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 behaviour 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 [22])	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.

This measurement shall be done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

### 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 table 7.64 are applicable (more information can be found in annex A of the Recommendation ITU-T P.340 [22].

Table 7.64: Category regarding "duplex capability" depending on Echo Loss

Category (according to Recommendation ITU-T P.340 [22])	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 done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

### 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:
See ETSI EN 300 176-2 [10].

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 psophometric weighting.

   NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

   NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.65.

Table 7.65: Mask for requirements for Spectral Adjustment of Comfort Noise

Frequency (Hz)	Upper Limit	Lower Limit
200	12	-12
800	12	-12
800	10	-10
2 000	10	-10
2 000	6	-6
4 000	6	-6
8 000	6	-6
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

Measurement method:
See ETSI EN 300 176-2 [10].

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.25]. 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 of ETSI EN 300 176-2 [10] the following requirements shall apply:

   •   N-MOS-LQOw ≥ 3,0;

- S-MOS-LQOw ≥ 3,0;

- G-MOS-LQOw ≥ 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 of ETSI EN 300 176-2 [10].

Measurement method:
See ETSI EN 300 176-2 [10].

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. 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 of ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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. 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.7.17   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 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 or fullband 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. As long as the measured PP+FP roundtrip delay fulfils the requirements given in clause 7.6.6, PP send or receive delays may be above the theoretical requirements.

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

Measurement method:
See ETSI EN 300 176-2 [10].

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 7.18a. 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.		
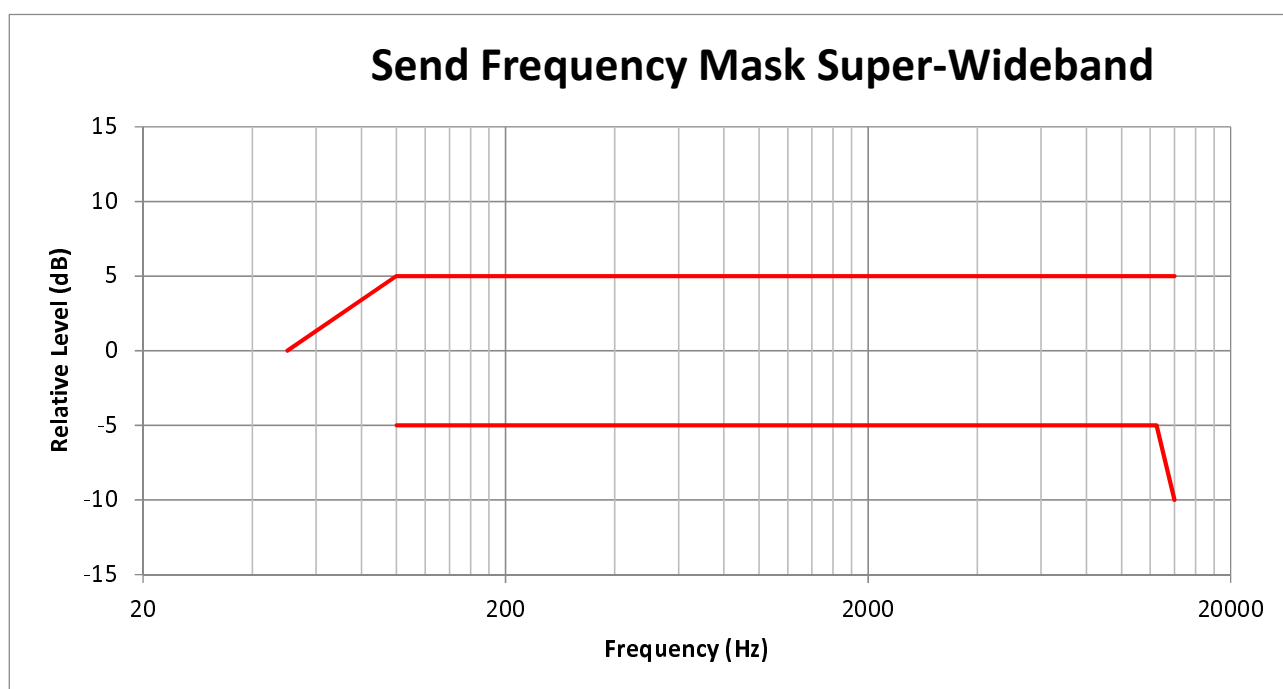
Figure 7.18a: Send frequency response mask for super-wideband

NOTE 1: The basis for the target frequency responses in sending and receiving is the orthotelefonic reference response which is measured between 2 subjects in 1 m distance under free field conditions and is assuming an ideal receive characteristic. Under these conditions the overall frequency response shows a rising slope. In opposite to other standards the present document no longer uses the ERP as the reference point for receiving but the 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.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 7.18b.

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.		
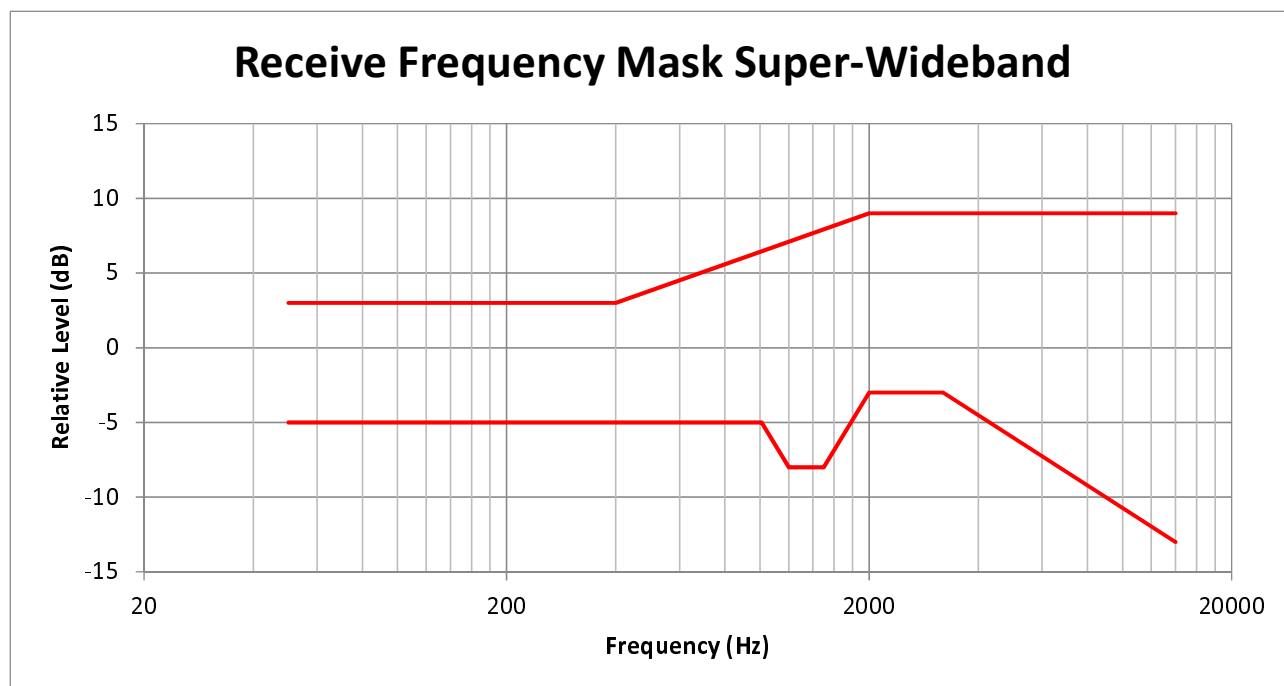
Figure 7.18b: Receive frequency response mask for super-wideband

NOTE: 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 [36] should be checked regularly for updates on this topic.

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.8.1.2.2   Receive 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 7.66b, 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:
See ETSI EN 300 176-2 [10].

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 dB ± 3 dB

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 [42] is not defined above 8 kHz, but sidetone signal is not supposed to have such limitation.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 $\geq$ 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 $\geq$ 55 dB.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 $\geq$ 30 dB.

Measurement method:
See ETSI EN 300 176-2 [10].

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:

See ETSI EN 300 176-2 [10].

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:

See ETSI EN 300 176-2 [10].

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:

See ETSI EN 300 176-2 [10].

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 [33]. 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 EN 300 176-2 [10].

7.5.8.8　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 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 only be connected to an FP of type 5 (Fixed Part with VoIP interface, wideband, super-wideband or fullband 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. As long as the measured PP+FP roundtrip delay fulfils the requirements given in clause 7.6.6, PP send or receive delays may be above the theoretical requirements.

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

Measurement method:

See ETSI EN 300 176-2 [10].

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 [22] and P.502 [27]):

- 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 Rec. ITU-T P.340 [22])	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:
See ETSI EN 300 176-2 [10].

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 Rec. ITU-T P.340 [22])	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:
See ETSI EN 300 176-2 [10].

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 [22]).

Table 7.66g

Category (according to Rec. ITU-T P.340 [22])	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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 psophometric weighting.

NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.66h.

Table 7.66h: 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:
See ETSI EN 300 176-2 [10].

7.5.8.10.4　　Speech quality in the presence of background noise

Requirement:
For the background noises defined in clause 6.10.6 of ETSI EN 300 176-2 [10] 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 of ETSI EN 300 176-2 [10].

Measurement method:
See ETSI EN 300 176-2 [10].

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 7.66i. The requirements are evaluated on the averaged results over all background noises used in this test.

Table 7.66i: 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:
See ETSI EN 300 176-2 [10].

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. 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 of ETSI EN 300 176-2 [10].

> 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.8.11.2 Spectral echo attenuation

Requirement:
The echo attenuation versus frequency shall be below the tolerance mask given in table 7.66j.

Table 7.66j: 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 7.66k and figure 7.18c.

Table 7.66k: 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.		
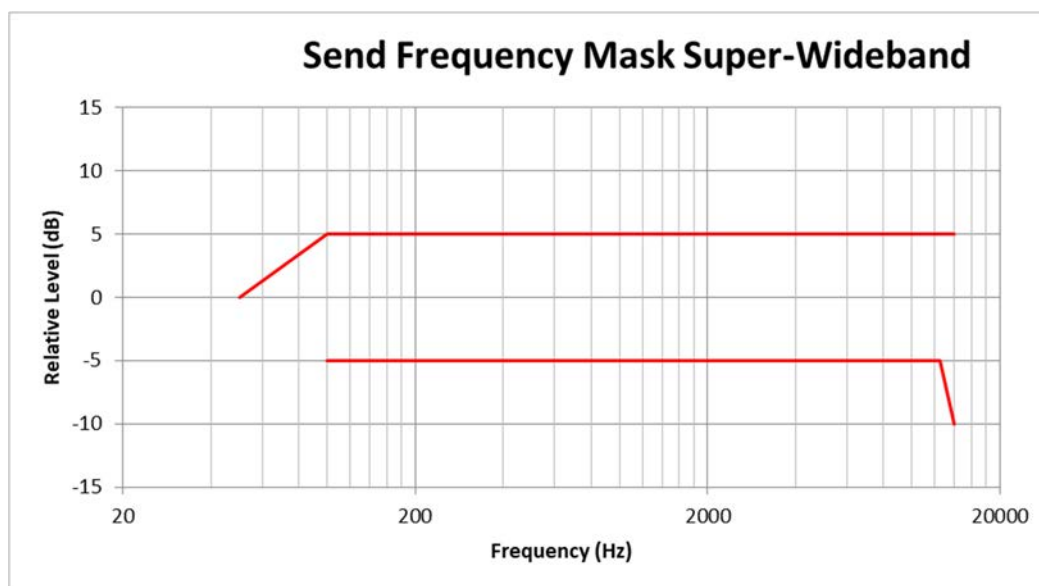
Figure 7.18c: Send frequency response mask for super-wideband

NOTE: 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.

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:
See ETSI EN 300 176-2 [10].

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 7.66l and figure 7.18d.

Table 7.66l: 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.		
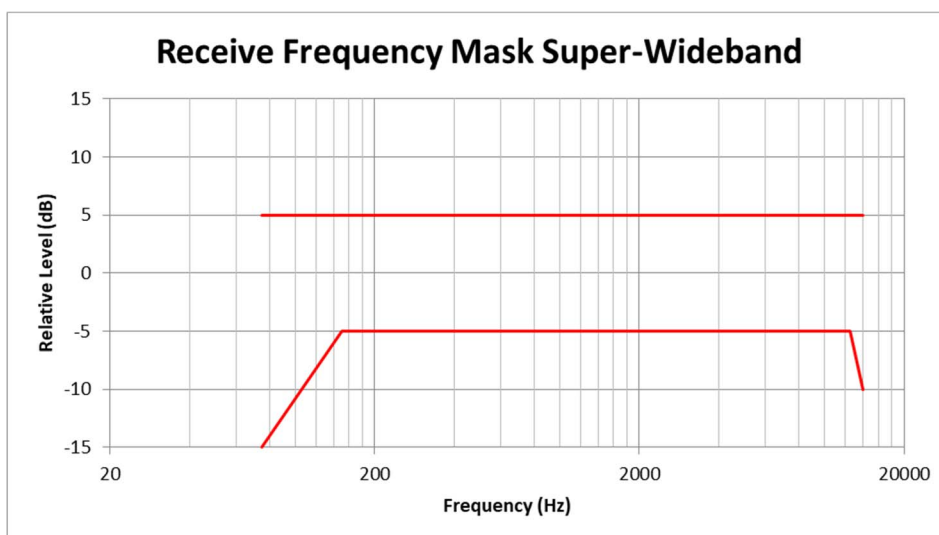
Figure 7.18d: Frequency mask for super-wideband handheld terminals - Receive

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.9.2.2   Desktop terminal

Requirements:
The frequency response shall fulfill the mask as defined in table 7.66m and figure 7.18e.

Table 7.66m: 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.		

Receive Frequency Mask Super-Wideband
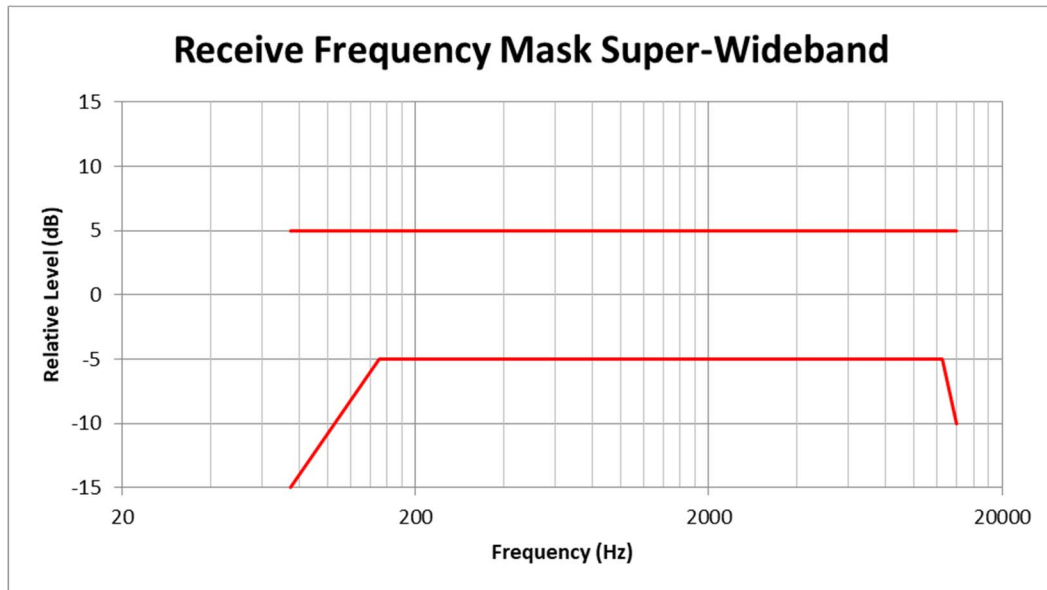
Figure 7.18e: Frequency mask for super-wideband desktop terminals - Receive

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.9.2.3   Terminals intended to be used simultaneously by several users

Requirement:
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

Requirement:
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 SWB bandwidth.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 [42] 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:

See ETSI EN 300 176-2 [10].

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 7.66n

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:
See ETSI EN 300 176-2 [10].

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 $\geq$ 30 dB.

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.9.6   Receiving distortion

Requirement:
The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.66o: Receive distortion for super-wideband

Frequency	Signal to distortion ratio limit, receive for desktop terminal
100 Hz	
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:
See ETSI EN 300 176-2 [10].

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

Measurement method:
See ETSI EN 300 176-2 [10].

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: For softphone, fan noise should be avoided in order to fulfill this condition.

Measurement method:
See ETSI EN 300 176-2 [10].

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 $\geq$ 55 dB for all positions of the volume control (if supplied).

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.9.10   Double Talk Performance

7.5.9.10.0   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 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 [22] and P.502 [27]):

- 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 7.66p.

Table 7.66p

Category (according to Recommendation ITU-T P.340 [22])	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 7.66p 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 done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

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 7.66q.

Table 7.66q

Category (according to Recommendation ITU-T P.340 [22])	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 7.66q 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 done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

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 7.66r are applicable (more information can be found in Annex A of Recommendation ITU-T P.340 [22]).

Table 7.66r

Category (according to Recommendation ITU-T P.340 [22])	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 done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

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).

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:
See ETSI EN 300 176-2 [10].

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 psophometric weighting.

> NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

> NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.66s.

Table 7.66s: 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:
See ETSI EN 300 176-2 [10].

7.5.9.11.4   Speech Quality in the Presence of Background Noise

Requirement:
For the background noises defined in clause 6.10.6 of ETSI EN 300 176-2 [10] the following requirements shall apply:

- N-MOS-LQO$_f$ ≥ 2,7.

- S-MOS-LQO$_f$ ≥ 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 of ETSI EN 300 176-2 [10].

Measurement method:
See ETSI EN 300 176-2 [10].

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. 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 of ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.9.12.2 Spectral Echo Attenuation

Requirements:
The echo attenuation vs. frequency shall be below the tolerance mask given in table 7.66t.

Table 7.66t: 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:
See ETSI EN 300 176-2 [10].

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. 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:
See ETSI EN 300 176-2 [10].

7.5.9.13   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 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 only be connected to an FP of type 5 (Fixed Part with VoIP interface, wideband, super-wideband or fullband 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. As long as the measured PP+FP roundtrip delay fulfils the requirements given in clause 7.6.6, PP send or receive delays may be above the theoretical requirements.

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

Measurement method:
See ETSI EN 300 176-2 [10].

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 7.66u and shown in figure 7.18f. This mask shall be applicable for all types of handsets and headsets.

Table 7.66u: 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 $1/12^{th}$ octave measurement.	
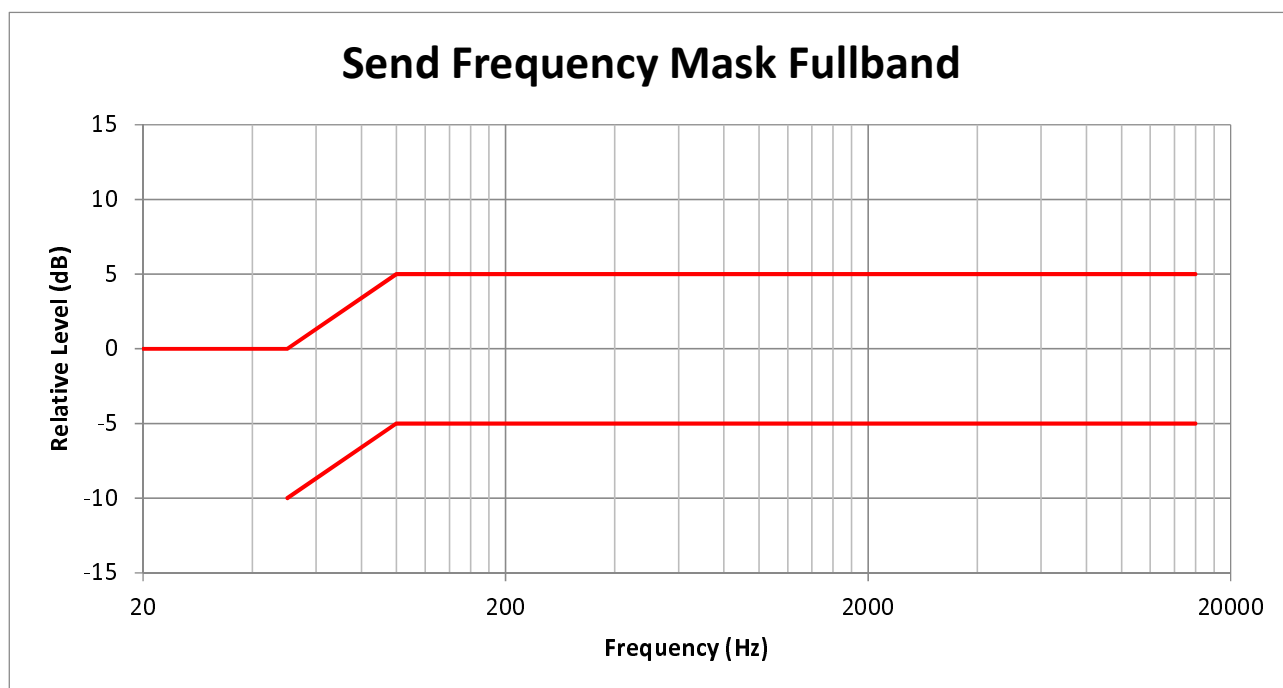
Figure 7.18f: Send frequency response mask for Fullband

NOTE 1: The basis for the target frequency responses in sending and receiving is the orthotelefonic reference response which is measured between 2 subjects in 1 m distance under free field conditions and is assuming an ideal receive characteristic. Under these conditions the overall frequency response shows a rising slope. In opposite to other standards the present document no longer uses the ERP as the reference point for receiving but the 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.

Measurement method:
See ETSI EN 300 176-2 [10].

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 7.66u, except that an additional tolerance is provided for certain positions. Table 7.66u1 provides the offset in dB for the lower limit.

Table 7.66u1: Tolerance mask offsets for send frequency response

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

Measurement method:

See ETSI EN 300 176-2 [10].

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 7.66v and shown in figure 7.18g.

Table 7.66v: 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
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
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.		

Receive Frequency Mask Fullband
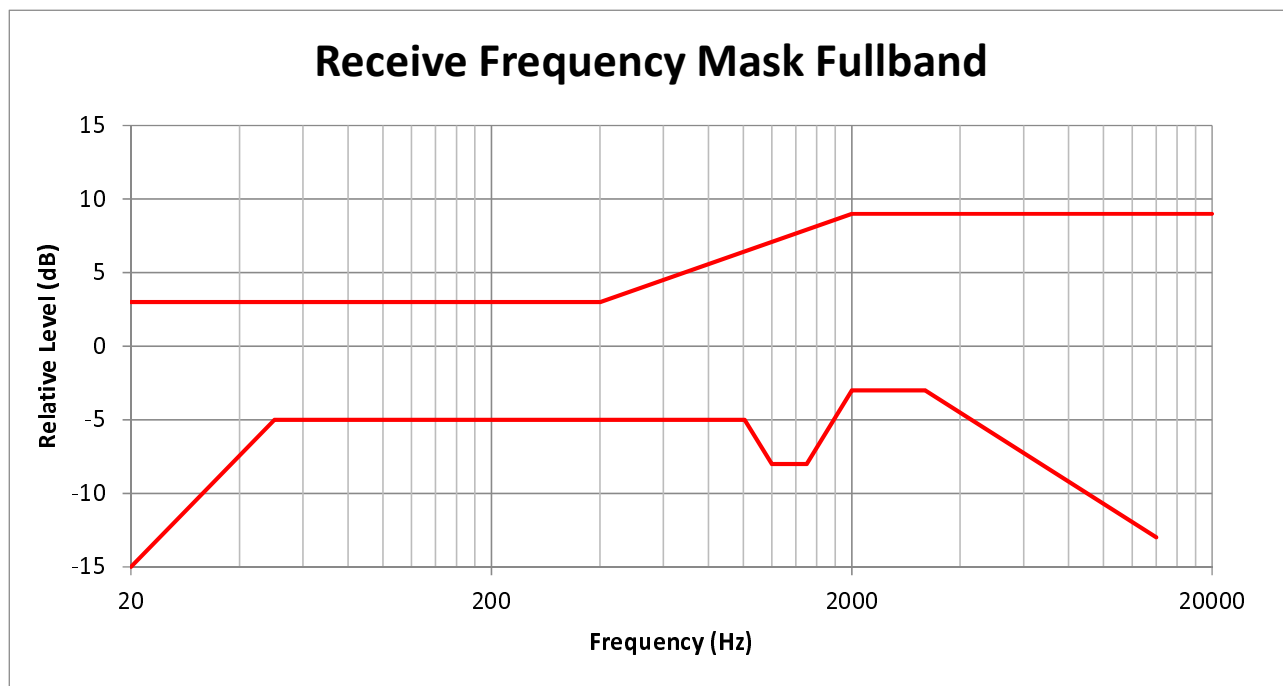
Figure 7.18g: Receive frequency response mask for Fullband

NOTE: 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 [36] should be checked regularly for updates on this topic.

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.10.1.2.2   Receive 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 7.66v, except that an additional tolerance is provided for certain positions. Table 7.66v1 provides the offset in dB for the lower limit.

Table 7.66v1: 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 [42] is not defined above 8 kHz, but sidetone signal is not supposed to have such limitation.

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.10.3.2   Sidetone Delay

Requirements:
The maximum sidetone-round-trip delay shall be $\leq$ 5 ms, measured in an echo-free setup.

Measurement method:
See ETSI EN 300 176-2 [10].

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 $\geq$ 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 $\geq$ 55 dB.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 7.66w: 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.10.5.2   Receiving Distortion

Requirements:
The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.66x: 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

### 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:
See ETSI EN 300 176-2 [10].

### 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 [33]. 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 EN 300 176-2 [10].

### 7.5.10.8	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 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 only be connected to an FP of type 5 (Fixed Part with VoIP interface, wideband, super-wideband or fullband 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. As long as the measured PP+FP roundtrip delay fulfils the requirements given in clause 7.6.6, PP send or receive delays may be above the theoretical requirements.

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

Measurement method:
See ETSI EN 300 176-2 [10].

### 7.5.10.9	Double talk Performance

### 7.5.10.9.0	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 [22] and P.502 [27]):

- 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 7.66y.

Table 7.66y

Category (according to Rec. ITU-T P.340 [22])	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 7.66y 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:
See ETSI EN 300 176-2 [10].

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 7.66z.

Table 7.66z

Category (according to Rec. ITU-T P.340 [22])	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 7.66z 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:
See ETSI EN 300 176-2 [10].

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 7.66aa are applicable (more information can be found in annex A of the Recommendation ITU-T P.340 [22]).

Table 7.66aa

Category (according to Rec. ITU-T P.340 [22])	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:
See ETSI EN 300 176-2 [10].

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 $\leq$ -20 dBPa.

The built-up time $T_{r,S,min}$ (measured with minimum activation level) shall be $\leq$ 15 ms.

Measurement method:

See ETSI EN 300 176-2 [10].

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 psophometric weighting.

NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.66ab.

Table 7.66ab: 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:

See ETSI EN 300 176-2 [10].

7.5.10.10.4   Speech quality in the presence of background noise

Speech Quality for fullband systems can be tested based on ETSI TS 103 281 [41].

Requirement:

For the background noises defined in clause 6.10.6 of ETSI EN 300 176-2 [10] 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 of ETSI EN 300 176-2 [10].

Measurement method:

See ETSI EN 300 176-2 [10].

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 7.66ac. The requirements are evaluated on the averaged results over all background noises used in this test.

Table 7.66ac: 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

See ETSI EN 300 176-2 [10].

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. 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 of ETSI EN 300 176-2 [10].

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:

See ETSI EN 300 176-2 [10].

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:

See ETSI EN 300 176-2 [10].

7.5.10.11.2   Spectral echo attenuation

Requirements:

The echo attenuation versus frequency shall be below the tolerance mask given in table 7.66ad.

Table 7.66ad: 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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 7.66ae and figure 7.18h.

Table 7.66ae: 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.		
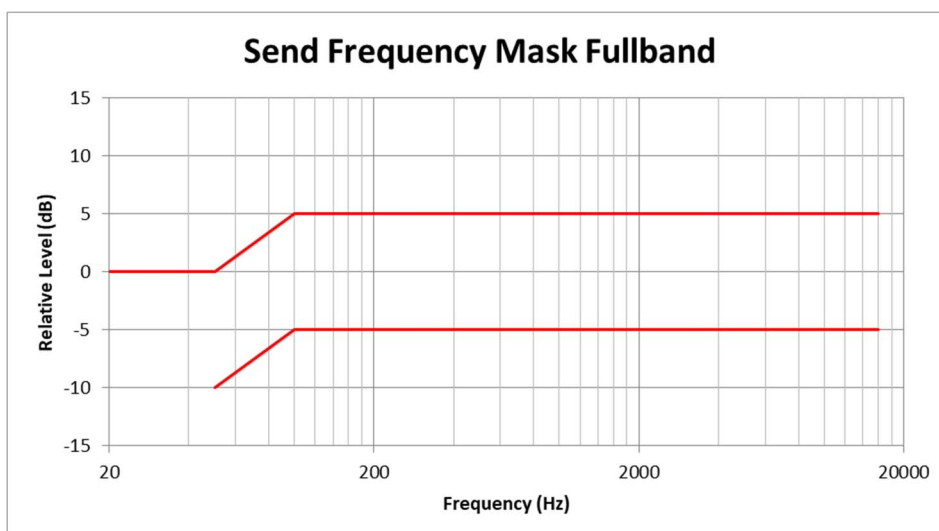
Figure 7.18h: Send frequency mask for fullband

NOTE: 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.

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:

See ETSI EN 300 176-2 [10].

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 on ETSI EN 300 176-2 [10]) HATS shall be equalized according to Recommendation ITU-T P.581 [35].

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 7.66af and figure 7.18i.

Table 7.66af: 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.		
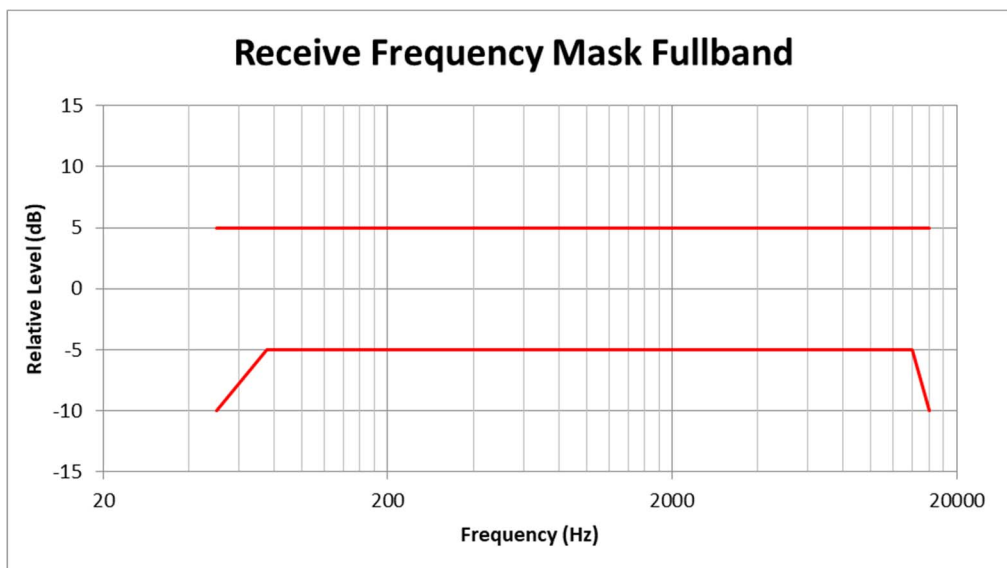
Figure 7.18i: Frequency mask for fullband handheld terminals - Receive

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.11.2.2   Desktop terminal

Requirements:
The frequency response shall fulfill the mask as defined in table 7.66ag and figure 7.18j.

Table 7.66ag: 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.		
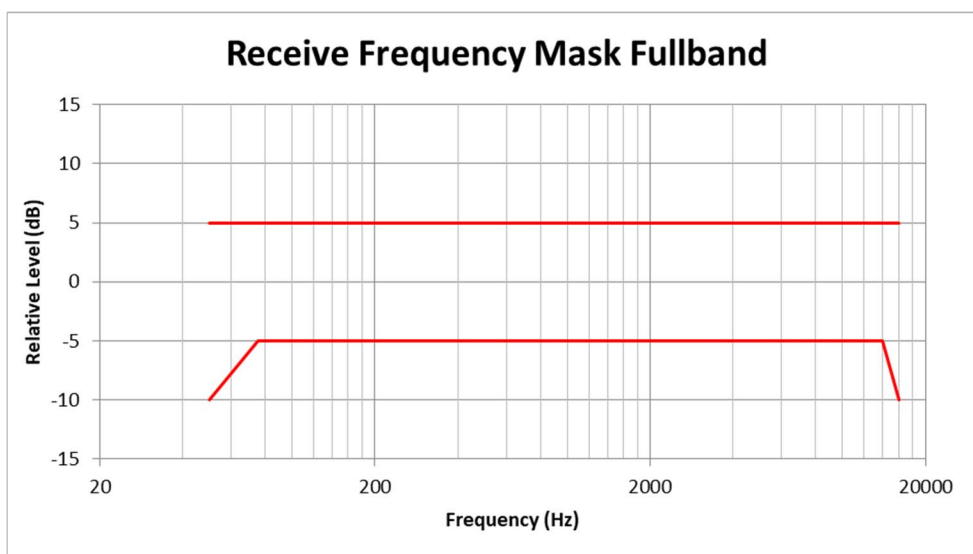
Figure 7.18j: Frequency mask for fullband desktop terminals - Receive

Measurement method:
See ETSI EN 300 176-2 [10].

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 13dB ± 3 dB.

There is no specific requirement for FB bandwidth.

Measurement method:
See ETSI EN 300 176-2 [10].

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
See ETSI EN 300 176-2 [10].

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 [42] 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:
See ETSI EN 300 176-2 [10].

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 7.66ah

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.11.6   Receiving distortion

Requirements:
The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.66ai: 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:
See ETSI EN 300 176-2 [10].

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

Measurement method:
See ETSI EN 300 176-2 [10].

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: 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:
See ETSI EN 300 176-2 [10].

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 $\geq$ 55 dB for all positions of the volume control (if supplied).

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.11.10   Double Talk Performance

7.5.11.10.0   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 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 [22] and P.502 [27]):

- 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 7.66aj.

Table 7.66aj

Category (according to Recommendation ITU-T P.340 [22])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,S,dt}$ [dB]	$\leq$ 3	$\leq$ 6	$\leq$ 9	$\leq$ 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 7.66aj 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 done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

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 7.66ak.

Table 7.66ak

Category (according to Recommendation ITU-T P.340 [22])	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 7.66ak 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 done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

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 7.66al are applicable (more information can be found in Annex A of Recommendation ITU-T P.340 [22]).

Table 7.66al

Category (according to Recommendation ITU-T P.340 [22])	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 done for the desktop hands free terminals and softphones also with variable echo path.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 psophometric weighting.

NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.66am.

Table 7.66am: 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:
See ETSI EN 300 176-2 [10].

7.5.11.11.4   Speech Quality in the Presence of Background Noise

Requirement:
For the background noises defined in clause 6.10.6 of ETSI EN 300 176-2 [10] the following requirements shall apply:

- N-MOS-LQO$_f$ ≥ 2,7.

- S-MOS-LQO$_f$ ≥ 3,3.

- G-MOS-LQO$_f$ ≥ 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 of ETSI EN 300 176-2 [10].

Measurement method:
See ETSI EN 300 176-2 [10].

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. 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 of ETSI EN 300 176-2 [10].

  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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.5.11.12.2 Spectral Echo Attenuation

Requirements:
The echo attenuation vs. frequency shall be below the tolerance mask given in table 7.66an.

Table 7.66an: 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:
See ETSI EN 300 176-2 [10].

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. 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:
See ETSI EN 300 176-2 [10].

7.5.11.13 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 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 only be connected to an FP of type 5 (Fixed Part with VoIP interface, wideband, super-wideband or fullband 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. As long as the measured PP+FP roundtrip delay fulfils the requirements given in clause 7.6.6, PP send or receive delays may be above the theoretical requirements.

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

Measurement method:
See ETSI EN 300 176-2 [10].

7.5.12 Transmission characteristics for PP type 7c, d, e ("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.

7.6   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 $\geq$ 46 dB.

A FP with a 4-wire interface (analog or digital) shall meet at least one of the two following requirements:

a)   artificial echo loss:

- an artificial echo path shall be implemented into the FP between the line input and the line output, as shown in figure 7.19. The loss of that echo path shall be 24 dB $\pm$ 2 dB.
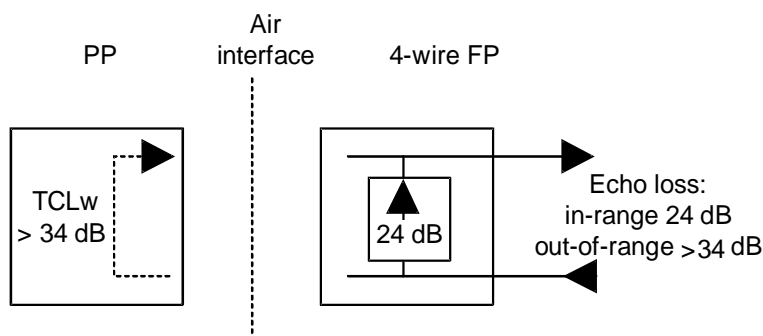
NOTE: The artificial echo simulates the echo from a very good analog 2-wire telephone. When a public network operator uses an echo canceller in the network (for a connection with long delay e.g. for a satellite link), the artificial echo loss path provides an in-range echo to ensure that the echo canceller and its Non-Linear Processor (NLP) is active. The NLP cancels the 34 dB DECT handset echo. See clause A.2.1.
In some countries the echo cancellers in the public network do not depend on the artificial echo path to activate the NLP. Installations in such countries could have the artificial echo path permanently disabled.

Figure 7.19: Artificial echo path in a 4-wire FP

b)   echo control device:

- an echo control device shall be implemented into the FP. The weighted Terminal Coupling Loss (TCLw) of the DECT system, defined from the FP line input to the FP line output, shall be at least 46 dB.

NOTE 1: Option a) is the option normally used for connections to the PSTN/ISDN. Option b) is needed, e.g. for tandem connection with GSM, clauses 8.2.1 and 8.3.2. Clause A.1.2 provides, for guidance and illustration, the description of a NLP implementation of option b). The control range for an echo canceller and hangover time for an NLP should be greater than or equal to 40 ms. A soft suppressor implementation is not recommended.

NOTE 2: The connect/disable function for options a) and b) is required for approval testing under ETSI EN 300 176-1 [9] and ETSI EN 300 176-2 [10]. Messages from the PP with control information are defined in ETSI EN 300 175-5 [5].

Measurement method:
See ETSI EN 300 176-2 [10].

### 7.6.1.2	FP Network echo control

In the most of the cases, there is no need for such an implementation (e.g. connections within a PABX, or a 4-wire connection via the PSTN/ISDN to an ISDN terminal).

However in some exceptional case FP Network echo control is implemented. In such a case the following requirements shall apply:

Requirement:
The network echo shall be controlled by inserting into the receiving speech path of the FP an echo loss meeting an extra echo loss ≥ 9 dB.

> NOTE:	The connect/disable function is required for approval testing (see ETSI EN 300 176-2 [10]). Messages from the PP with control information are defined in ETSI EN 300 175-5 [5], clause 7.7.16.

Measurement method:
See ETSI EN 300 176-2 [10].

### 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 annex D for further information.

Measurement method:
See ETSI EN 300 176-2 [10].

### 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:
See ETSI EN 300 176-2 [10].

## 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 the most of the cases, there is no need for such an implementation (e.g. connections within a PABX, or a 4-wire connection via the PSTN/ISDN to an ISDN terminal).

However in some exceptional cases FP Network echo control is implemented. In such a case the following requirements shall apply.

Requirement:
The network echo shall be controlled by inserting into the receiving speech path of the FP an echo control device inserting an extra echo loss ≥ 9 dB.

NOTE: The connect/disable function is required for approval testing (see ETSI EN 300 176-1 [9] and ETSI EN 300 176-2 [10]). Messages from the PP with control information are defined in ETSI EN 300 175-5 [5], clause 7.7.16.

Measurement method:
See ETSI EN 300 176-2 [10].

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 annex D for further information.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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 annex D for further information.

Measurement method:
See ETSI EN 300 176-2 [10].

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 [29], clause A.2.3. No recommendation is made for any particular implementation of the echo control device. For guidance and illustration, a reference soft suppressor that meets requirement 2 is described in clause A.3.1, and a reference echo canceller that meets requirement 1, is described in clause A.3.2.

Depending on routing information and on type of service, it shall be possible to connect and disable each of the echo control functions which perform the respective requirements 1 and 2.

The echo device implemented to meet requirement 2 may be disabled, or its loss may be reduced, in accordance with optionally available routing information, e.g. connections within a PABX, or a 4-wire connection via the PSTN/ISDN to an ISDN terminal.

NOTE: The connect/disable function for requirements 1 and 2 is required for approval testing (see ETSI EN 300 176-1 [9] and ETSI EN 300 176-2 [10]). Messages from the PP with control information are defined in ETSI EN 300 175-5 [5], clause 7.7.16.

Measurement method:
See ETSI EN 300 176-2 [10].

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 [29] 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 [29] 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 [29] 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 [29] 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 [29] 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 [29] 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 [29] 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 [29] shall apply, except that the test with input e.m.f. of 0 dBV in clause 4.2.4.2 of ETSI TBR 038 [29] 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 [29] shall apply, except in the test of clause 4.2.6.1 of ETSI TBR 038 [29], 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 [29] 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.

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 annex D for further information.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.6.4.6 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. In the following clauses the calculation of the requirements for send and receive delay are explained. 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. As long as the measured FP+PP roundtrip delay fulfils the requirements, send or receive delays of the connected PP may be above the theoretical requirements.

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 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 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.

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.2.

Measurement method
See ETSI EN 300 176-2 [10].

7.6.5 Transmission characteristics for FP type 4 (Fixed Part with ISDN network interface, wideband service)

7.6.5.1 FP adaptive volume control

Requirement:
An adaptive volume control, depending on the level of environmental noise at the PP, may be implemented into the FP. The gain variation shall be symmetrical, i.e. the increase in the receiving direction shall be equal to the decrease in the sending direction.

If the PP adaptive volume control feature is implemented then the FP adaptive volume control feature shall be disabled.

See annex D for further information.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.6.6 Transmission characteristics for FP type 5 (Fixed Part with VoIP interface, wideband, super-wideband or fullband 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 for further information.

Measurement method:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

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:
See ETSI EN 300 176-2 [10].

7.6.6.6   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. In the following clauses the calculation of the requirements for send and receive delay are explained. 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. As long as the measured FP+PP roundtrip delay fulfils the requirements, send or receive delays of the connected PP may be above the theoretical requirements.

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 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 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.

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.2.

Measurement method
See ETSI EN 300 176-2 [10].

8   Additional features

8.1   Loudspeaking hands-free and headset facilities

8.1.1   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 described in clause 7.2.

These audio types are applicable to either:

  1)   specific PPs designed to operate in handsfree mode;

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

  3)   handsfree accessory devices connected to a handset by any wired or wireless technology.

If loudspeaking and/or handsfree telephony is implemented in a handset, this device when operating in normal handset mode shall fulfil any of the audio types for handsets (types 1 or 2 of clause 7.2).

8.1.2   Headset facility

Headset accessory devices connected to a DECT PP by any wired or wireless technology are considered handsets from audio profile point of view. The sub-system composed by the DECT PP (muted), the wire or wireless link and the headset speaker-microphone set, shall fulfil any of the profile types for handsets (types 1 and 2 of clause 7.2).

8.2   Tandem with mobile radio network

8.2.0   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 8.1.1.

8.2.1   Tandem with GSM

8.2.1.0   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 8.1.
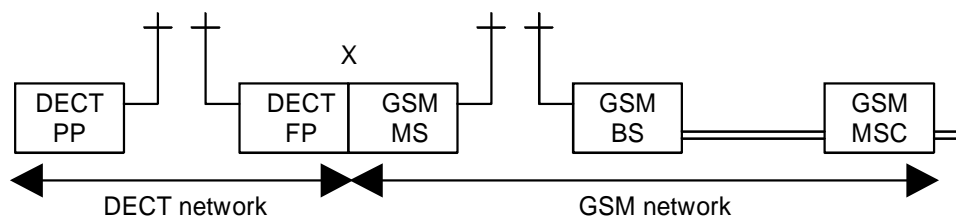
Figure 8.1: DECT in tandem with the GSM mobile network

8.2.1.1 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 [31].

8.2.1.2 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.

8.2.1.3 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.

8.2.1.4 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 for information on GSM, DTX.

8.3 DECT connected to the GSM fixed network

8.3.0 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 8.2.
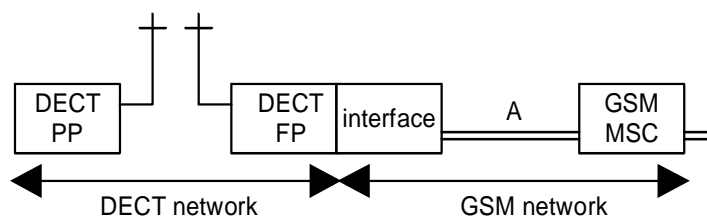
Figure 8.2: DECT in tandem with the GSM fixed network

8.3.1 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 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 [31].

8.3.2 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.

8.4 Wireless Relay Stations (WRS)

8.4.0 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 8.3.
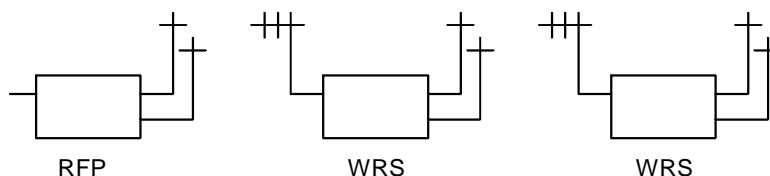
NOTE: A WRS is locked to the closest RFP, or closest WRS when cascaded WRSs are allowed. An RFP and WRS, appear equal to a PP.

Figure 8.3: 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 and 7.6.3.2) may need to be modified depending on the characteristics of the specific network to which the FP is connected.

8.4.1 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 Requirement 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 - 4 ms TELR):	> 27 dB	> 29 dB

Requirement 2 (4 - 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):
Description of "reference" echo control devices

A.0 General

The present annex describes reference echo control devices based upon concepts that are as simple as possible but include a sufficient number of features to give guidance for a wide range of possible implementations. The use of the term "reference" denotes an implementation given for guidance and illustration only. It does not exclude other implementations nor does it imply that the described devices are necessarily the most appropriate realization on any technical, operational or economic grounds. See Recommendation ITU-T G.168 [i.21].

A.1 Handset echo

A.1.0 Applicability

This clause only applies to FP type 1a and FP type 2.

A.1.1 Overview

A.1.1.1 Connection to the PSTN/ISDN

A.1.1.1.0 General

FPs with 4-wire connection should provide the artificial echo specified in clause 7.6. It should be disabled if the PP meets the requirements of clause 7.6 TCLw > 46 dB, and may be disabled on available call routing information.

The combined handset and hybrid echo from analog telephones is considerably higher than the DECT 34 dB TCLw echo. Therefore the international planning rules demand provision of echo control devices in the public network, depending on the one-way network echo delay and the Talker Echo Loudness (TELR) according to Recommendation ITU-T G.131 [30].

A.1.1.1.1 Local and national calls

The 34 dB TCLw for a DECT system will give the far-end user a TELR of about 44 dB. According to the "acceptable" curve of figure 1 in Recommendation ITU-T G.131 [30], a mean one-way transmission time of 70 ms could be accepted. The delay requirements will be met for the majority of national connections with DECT system at one end, at both ends of the connection and also with a DECT tethered local loop replacement in the public network.

A.1.1.1.2 Long distance connections with echo control devices in the PSTN/ISDN, e.g. calls via satellites

Echo control devices are provided in the public network according to the guidelines of Recommendation ITU-T G.131 [30]. It is nowadays recommended only to deploy echo cancellers that conform to Recommendation ITU-T G.168 [i.21]. Older echo canceller equipment conforms to G.165 [i.20]. Older echo suppressor devices according to G.164 [i.19] may also be in use.

The echo path delay capabilities of modern echo cancellers are often 64 ms (or more). These echo cancellers will eliminate echoes with less than 30 ms one-way delay.

However, as long as these devices are active, they will also eliminate the DECT acoustic echo. The 34 dB attenuated echo is low enough not to be detected as break-in speech during single talk condition during speech from the far-end. The DECT echo will therefore be eliminated by an echo suppressor or by the Non-Linear Processor (centre clipper), NLP, of an echo canceller, whichever device is used. Both the suppressor and the NLP have hangover times which will prevent the delayed DECT echo to be heard even at the end of a speech sentence.

The artificial echo, clause 7.6 simulates the echo from a very good analog 2-wire telephone. When an echo canceller is used in the network, the artificial echo loss path provides an echo, that is guaranteed in-range for the linear echo estimation process. This will ensure that the echo canceller and its NLP are active. The artificial echo is principally not needed to keep echo suppressors nor all types of echo cancellers active, but it does not hurt either.

A.1.1.2   Connection to the GSM network

When a DECT system is connected in tandem with a GSM mobile network (see clause 8.2.1) or a GSM fixed network (see clause 8.3) the FP provides the echo control function specified in clause 7.6.1 It should be disabled if the PP meets the requirement TCLw > 46 dB, and for the case of Discontinuous, DTX, specified in clause 8.2.1.4.

Echo control function specified in clause 7.6 should be disabled.

A.1.2   Implementation of the FP echo control function

A.1.2.0   General

The FP echo control function is described in clause 7.6.

The needed extra echo loss can preferably be implemented by using a Non-Linear Processor (NLP) that, when active, clips all sending signals lower than the suppression threshold. See figure A.1.
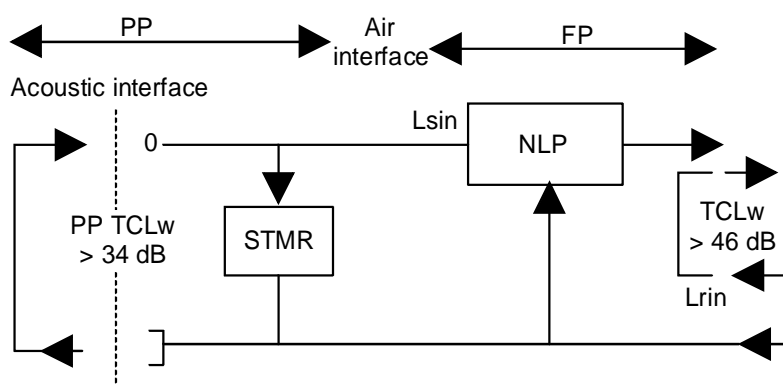
Figure A.1: Implementation of the handset echo control

Such a NLP is described in Recommendation ITU-T G.168 [i.21], annex B. Two variants are included, both based on an analog centre clipper having the transfer function illustrated in figure A.2.
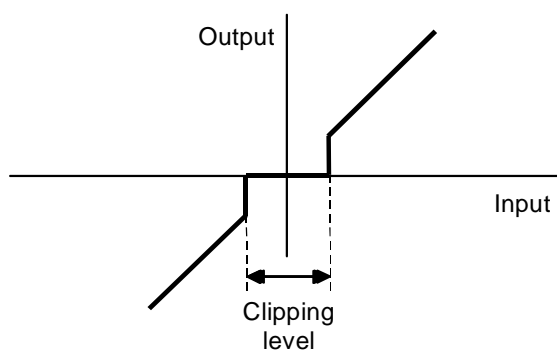
Figure A.2: Transfer function of non-linear processor

In the first case the suppression threshold level is adaptive, adaptation being by reference to Lrin. In the second variant the suppression level is fixed.

A.1.2.1   Suppression threshold

The value of the suppression threshold depends on the handset acoustic TCL, below called PTCL:

    Fixed: Vsup = (- PTCL) dBm0;

    Adaptive: Vsup = (Lrin - PTCL + 6) dBm0.

                     6 dB is the operation margin that prevents the possible ambient noise to pass through the activated NLP. If the noise is high enough to pass through, it will also be high enough to mask the echo.

PTCL is > 34 dB.

If for example PTCL is set to 30 dB then:

    Vsup = -30 dBm0; or

    Vsup = Lrin - 24 dB.

A.1.2.2   Static characteristics of activation control

In annex B of Recommendation ITU-T G.168 [i.21], the activation control is defined by reference to Lrin. The NLP is active when the level of signal in the sending direction Lsin is lower than the activation threshold Vact:

    Fixed: Vact = (Lrin - 6) dBm0;

    Adaptive: Vact = (Lrin - PTCL + 6) dBm0.

As the subject here is acoustic echo in a terminal equipment instead of electric echo in the network, a simplification seems reasonable and Vact can be fixed:

    Vact = -31 dBm0.

A.1.2.3   Dynamic characteristics of activation control

Table A.1: Non-linear processor hangover times

Switch from	Vsup variant	Recommended values
active to inactive	fixed Adaptive	15 ms to 64 ms 2 ms (echo path delay)
inactive to active	fixed Adaptive	20 ms to 120 ms 30 ms to 50 ms

A.2   Network echo

A.2.0   Applicability

This clause only applies to FP types 1a, 1b and 2.

The requirement on network echo control (see clause 7.6) can be met with the following implementation, depending on the type of line interface:

    1)   for digital circuit switched (4-wire) interface (FP types 1a and 1b), a soft suppressor (see figure A.3);

    2)   for analog (2-wire) interface (FP type 2), a soft suppressor and an echo canceller (see figure A.4).
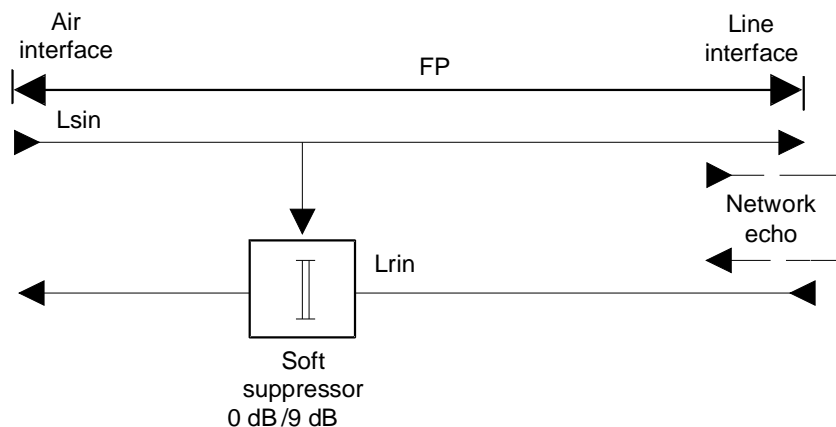
Figure A.3: Implementation of the network echo control 4-wire line interface
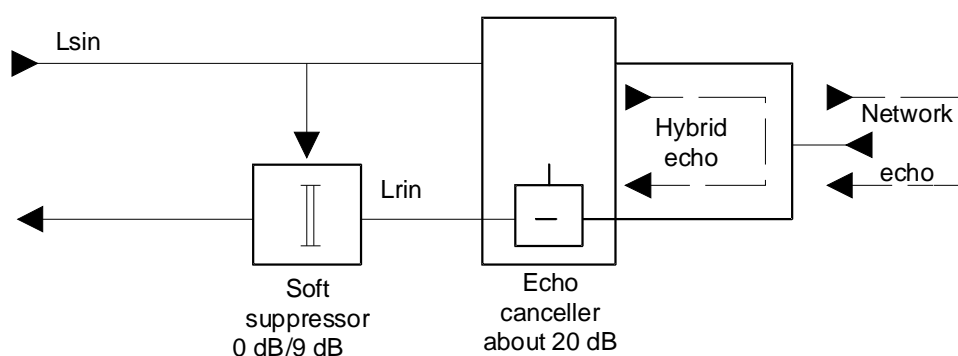
Figure A.4: Implementation of the network echo control 2-wire line interface

A.2.1   Soft suppressor implementation of requirement 2

A.2.1.0   General

A 9 dB soft echo suppressor meets the whole requirement for 4-wire connection and a part of the requirement for the 2-wire connection. The principle of operation is similar to the one described in Recommendation ITU-T G.164 [i.19]. Therefore the same operational parameters can be used.

The 9 dB to 12 dB loss for requirement 2 is intended to cover different foreseen connection cases. It is also a compromise between reduction of echoes from badly matched far-end national hybrids, and low noise contrast requirements.

In most applications the majority of calls will not need any extra echo suppression, and therefore optimization should be made to minimize the noise contrast. Thus the lower 9 dB value is recommended.

The 9 dB to 12 dB loss is required when no routing information is available. When suitable routing information is available, the operator is allowed to disconnect or reduce the loss below 9 dB to optimize it for specific connections or applications.

This parameter of requirement 2 only affects the DECT user and not the general network planning rules.

A.2.1.1   Static characteristics

The echo suppressor is active if the speech level in the sending direction Lsin is greater than the suppression threshold:

Vsup = -31 dBm0.

The suppressor becomes inactive if the speech level falls under the release threshold:

Vrel = Vsup (+0 dB, -3 dB).

A.2.1.2   Dynamic characteristics

The switching characteristics are taken from Recommendation ITU-T P.340 [22], that relates to terminal equipment and seems more appropriate than Recommendation ITU-T G.164 [i.19], that relates to network echo suppressor (Tr ≤ 2 ms and Th = 24 ms to 36 ms).

1)   Build-up time: Tr < 15 ms, preferably < 10 ms.

2)   Hang-over time: Th > 100 ms, preferably between 150 ms and 250 ms.

The soft suppressor may pass a 6 dB state at the switch on and switch off of the 9 dB to 12 dB attenuation. The 6 dB state should be of less than 15 ms duration.

A.2.2   Additional echo control for a 2-wire interface of requirement 1

For the 2-wire case a Talker's Echo Loudness Rating (TELR) of 24 dB is required for the delayed sidetone from the FP's 2-wire interface. This can be performed in some cases by hybrid matching, or more generally by means of an echo canceller.

In order to meet the requirement of TELR = 24 dB for the three terminating impedances according to clause 7.6.3.2, around 20 dB extra loss is estimated to be required for the worst cases of mismatch. This implies that for normal cases, TELR > 30 dB is expected for the delayed sidetone.

TELR = 24 dB equates to LL_e = 14 dB at the PCM reference point of the FP.

NOTE: The solution using an adaptive soft suppressor with up to 20 dB attenuation is not recommended.

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

The echo canceller is operating on the uniform PCM speech signals. It makes a linear estimate Acanc of the echo path Aecho (see figure A.5), and subtracts this estimate at the subtractor. The residual echo level after the subtractor is Lres. Lfes is the level of the far-end speech.

Lres = Lsin - Aecho - Acanc (in power);

Lrout = Lres + Lfes.

Since the worst case Aecho is -1 dB, the required Acanc is about 15 - A dB. Therefore no NLP is needed to further reduce Lrout. A is the attenuation of the soft suppressor for requirement 2. If the soft suppressor can be disabled (e.g. for local PABX calls), A is set to zero during type approval testing (see ETSI EN 300 176-2 [10]).
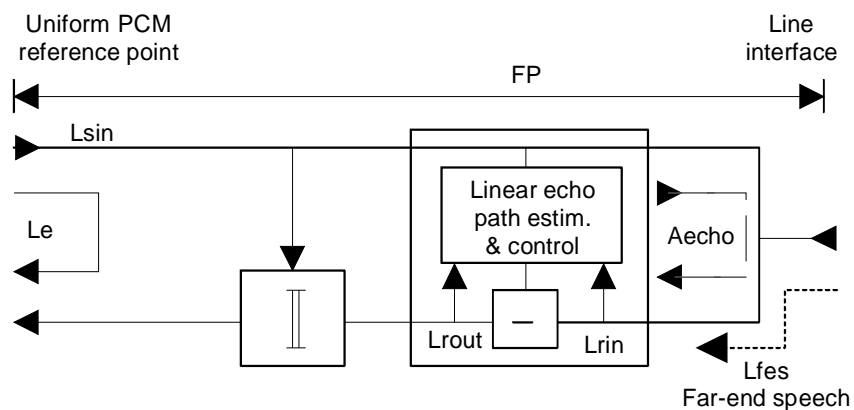
Figure A.5: Echo canceller implementation

Speech detection for e.g. double talk, cannot be made before the subtractor (Lrin), as in Recommendation ITU-T G.168 [i.21], because the level of the hybrid echo can be larger than the level of the far-end signal.

Detection of the far-end speech has to be made after the subtractor (Lrout), and detection is only possible when the near end echo is cancelled to a certain degree.

Thus at the start (adaptation coefficients = 0), the FES detector is switched off. It is switched on once Lrout is below a certain threshold. This will only happen when Lfes is very low and the filter adaptation is satisfying (Lres low).

At detected double talk, the adaptation cannot be completely switched off, but the adaptation speed should be reduced by a factor of e.g. 4. This allows adaptation during conversation if the echo path suddenly is changed.

A.2.3 Echo canceller used for both requirements 1 and 2

An alternative to implementing a soft suppressor, is to use an echo canceller as described in clause A.2.2 with the control range prolonged to 70 ms. This solution may be more expensive, but will give better performance for PP reception on a noisy connection.

Annex B (informative):
Local loop application

B.0　General

The present annex relates to public local loop applications.

A PP is a physical grouping that contains all elements between the user and the DECT air interface. A PP is a generic term that may describe one or several physical pieces.

A PP is logically divided into one PT plus zero or more Terminal Adaptors (TA) plus one or more End Systems (ESs).

The ES can thus be implemented in a handset, or be e.g. a standard ISDN terminal, a standard 2-wire telephone or even a cordless telephone system, tethered to a TA via a standard subscriber's telephone socket.

Below, guidance is given for the case when the ES is connected via a standard 2-wire socket, e.g. in a public local loop evolutionary application.

B.1　DECT tethered local loop replacement with 2-wire PP end system

B.1.0　Configuration

Figure B.1 below considers the case when a public operator uses DECT for a local loop replacement.
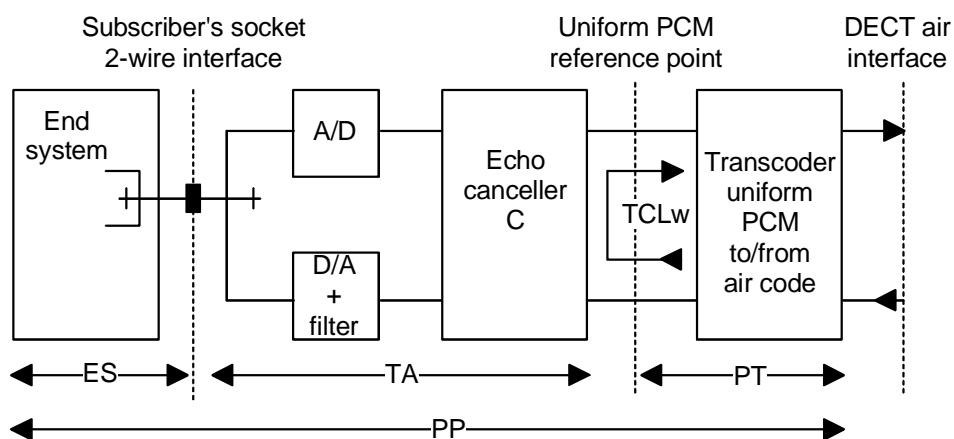
Figure B.1: PP in tethered local loop application

B.1.1　TCLw requirements

If the public operator plans the control range of his echo cancellers at the international gateways to include the hybrid at the subscriber's socket interface and a standard telephone end system, then no extra echo canceller C is needed.

If the hybrid in figure B.1 is outside the control range of the echo cancellers at the international gateways, then the hybrid's echo return loss LL_e has to be > 34 dB. If this is not met, an echo canceller C, with about 20 dB extra echo attenuation and with a 4 ms control range, of the same type as described in clause A.2.2 (without the soft suppressor) is needed in the PP TA.

The echo control device may in principle for some installation cases, alternatively be implemented at the FP with a control range of 20 ms to 28 ms (or more if Wireless Base Stations (WBSs) are implemented). At the FP some trunking gain can be provided on needed number of simultaneous echo control functions.

If the ES has higher loop delay than a standard telephone (2 ms), e.g. a residential DECT system, this system has to meet the normal TCLw requirements of clause 7.5.1.4.1. See clauses A.1.1.1.1 and A.1.1.1.2.

B.1.2   Network echo
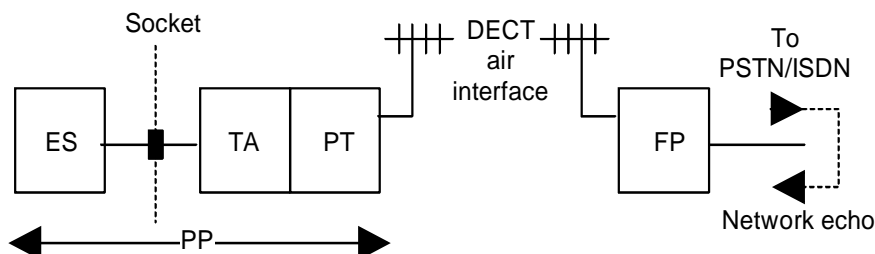
Figure B.2: DECT local loop replacement

The FP of figure B.2 has to meet the normal echo control requirements 1 and 2 of clause 7.6.3.2.

If the ES is a (residential) DECT system, each FP will meet the requirement of clause 7.6.3.2 and take care of its own network echo.

It is however recommended that requirement 2 is realized with an echo canceller, or a soft suppressor with reduced echo loss if the operator has a modern network, see clause A.2.1.

This recommendation is to minimize the noise contrast for the case when the ES is a DECT residential system, also implementing requirement 2.

It is allowed to disconnect or reduce requirement 2, when suitable routing information is available. This may be the case for FPs under control of one operator in a local loop application. For instance, if an operator has a modern network where the far-end hybrids have $LL_e > 20$ dB, a reduction to 6 dB could be suitable if the loss in the FP in figure B.2 is implemented as a soft suppressor and not as an echo canceller.

Annex C (informative):
GSM Discontinuous Transmission (DTX), and Voice Activity Detection (VAD)

Discontinuous Transmission (DTX) and Voice Activity Detection (VAD) are mandatory features in a GSM mobile, but not in a GSM base station. A GSM mobile has to be able to operate in both modes.

Without DTX and VAD in the up-link, TCL 34 dB and NLP in the FP meet the requirements.

If VAD and DTX is implemented the DECT TCL echo path is broken except for double talk. At double talk TCL 34 dB should be enough, since the echo level will be 34 dB below the (annoying) double talk and is masked by the double talk. Furthermore double talk occupies only a small fraction of the speech time.

The 34 dB attenuated echo is not interpreted as a speech break-in signal. When DTX is implemented, the normal background speech or music in a public environment should not eliminate the DTX function. For example, in Recommendations ITU-T G.164 [i.19], G.165 [i.20] and G.168 [i.21] double talk state occurs only if the "sending speech" level is higher than the "receiving speech" level -6 dB.

The VAD is extracting noise parameters in speech pauses in the signal from the microphone (sending speech). The VAD might need to detect speech pauses at -40 dBm0 to -45 dBm0 levels, in order to deliver clean noise parameters to the GSM base station, for noise insertion in DTX pauses. But these low levels need not be related to levels required to prevent background speech or an acoustic echo to eliminate the DTX function. The VAD is specially designed for traffic and motor noise in a car, and not optimized for an environment with background speech and music.

Therefore it is stated in clause 8.2.1.4 that "*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 receiving speech level*". If Recommendations ITU-T G.164 [i.19], G.165 [i.20] and G.168 [i.21] are used as guidance for an upper level, a threshold may be implemented somewhere in the range 6 dB to 30 dB below the level of the speech received from the far-end.

Annex D (informative):
Speech levels in relation to ambient room noise and examples of adaptive volume control settings

When room noise is present a higher received speech signal level (lower RLR_H) is required to give the best possible receiving speech quality and intelligibility. The increase in the receiving level is a function of increasing room noise level.

When room noise is present people raise their voice level (talk louder) and the sending speech level will be higher than the optimum level. By decreasing the sending sensitivity (higher SLR_H) the sending level to the line will be around optimum and at the same time the absolute level of the transmitted noise will decrease.

The increase in receiving sensitivity and the decrease in sending sensitivity is approximately linearly related to increasing room noise level. For every dB increase in room noise level there is a corresponding increase in receiver sensitivity and decrease in sending sensitivity of about 0,5 dB.

Table D.1 presents - for guidance and illustration only - three examples of volume control characteristics.

Table D.1: Examples of FP adaptive volume control

Ambient noise		Volume variation receiving (sending)	
Single threshold	Between 50 dBA and 55 dBA	8 dB gain	(attenuation)
Double threshold	50 dBA	5 dB gain	(attenuation)
	60 dBA	10 dB gain	(attenuation)
Continuous variation	from 30 dBA to 50 dBA	3 dB increase	(decrease)
	From 50 dBA to 70 dBA	8 dB increase	(decrease)

Annex E (informative):
Echo related topics

E.1   Summary table on echo parameters for PPs and FPs

Tables E.1 to E.3 summarize the requirements for PPs and FPs and the consequences for implementations.

NOTE 1:   The value of TCLw given in this table is for nominal setting of volume control.

NOTE 2:   Handsfree or loudspeaking audio feature is not analysed in this table, 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	Clause	TCLw Requirement for type (dB)	TCLw Real Value (dB)	Setting of flags "echo parameter" (bits 5 and 6 in octet 3b) of IE <TC>	Action for FP type 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)

Table for Narrowband PP types									
Applicable to:	**Type nr.**	**Audio type name**	**Clause**	**TCLw Requirement for type (dB)**	**TCLw Real Value (dB)**	**Setting of flags "echo parameter" (bits 5 and 6 in octet 3b) of IE <TC>**	**Action for FP type 1a**	**Action for FP type 1b or 3**	**Action for FP type 2**
	1b	Improved GAP handset narrowband	7.2.4 7.5.2.1	> 55	> 55	11	Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard)	Nothing (transparent)	Nothing (transparent)
	1c	HATS tested, "standard" narrowband handset	7.2.5 7.5.3.4	> 55	> 55	11	Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard)	Nothing (transparent)	Nothing (transparent)
	1d	HATS tested, "improved" narrow band handset	7.2.6 7.5.3.4	> 55	> 55	11	Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard)	Nothing (transparent)	Nothing (transparent)

Table E.2: Summary on echo parameters for wideband PP types

Table for Wideband PP types							
Applicable to:	Type nr.	Audio type name	Clause	TCLw Requirement for type (dB)	TCLw Real Value (dB)	Setting of flags "echo parameter" (bits 5 and 6 in octet 3b) of IE <TC>	Action for FP type 4 or 5
	2a	Recommendation ITU-T P.311 [20] tested, wideband handset	7.2.9 7.5.5.4.1	> 42	> 42	01	Optional insertion of echo canceller (clause 7.4.2) or echo suppressor (clause 7.4.3) if implemented
				> 42	> 46	10	Optional insertion of echo canceller (clause 7.4.2) if implemented. Echo suppressor should not be inserted
				> 42	> 55	11	Nothing (transparent)
	2b	HATS tested, "standard" wideband handset	7.2.10 7.5.6.4.1	> 55	> 55	11	Nothing (transparent)
	2c	HATS tested, "improved" wideband handset	7.2.11 7.5.6.4.1	> 55	> 55	11	Nothing (transparent)

Table E.3: Summary on echo parameters for FPs types

Table for FPs								
Applicable to:	Type nr.	Audio type name	Clause	TCLw Requirement for type (dB)	TCLw Real Value (dB)	Flag "echo parameter" (bits 5 and 6 in octet 3b of Terminal Capability)	Action	Clause (echo handling)
FP	1a	"classic" Fixed Part with ISDN interface, narrowband service	7.3.2			(if PP = 01)	Mandatory insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1)	7.6.1.1, see also A.1
						(if PP = 10)	Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard)	7.6.1.1, see also A.1
						(if PP = 11)	FP type 1a does not need to distinguish PP = 11. Action is as for PP = 10 Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard)	7.6.1.1, see also A.1
	1b	"new" Fixed Part with ISDN interface, narrowband service	7.3.3			(If PP = 01)	Optional insertion of echo canceller (clause 7.4.2) or echo suppressor (clause 7.4.3) if implemented	7.4.2, 7.4.3
						(If PP = 10)	Optional insertion of echo canceller (clause 7.4.2) if implemented. Echo suppressor should not be inserted	7.4.2
						(If PP = 11)	Nothing (transparent). FP will not insert echo canceller or suppressor	7.4
	2	Fixed Part with analog 2-wire interface, narrowband service	7.3.4				Nothing	See A.1

Table for FPs							
Applicable to:	Type nr.	Audio type name	Clause	TCLw Requirement for type (dB)	TCLw Real Value (dB)	Flag "echo parameter" (bits 5 and 6 in octet 3b of Terminal Capability)	Action	Clause (echo handling)
	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
	5	Fixed Part with VoIP interface, wideband, super-wideband or fullband 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.
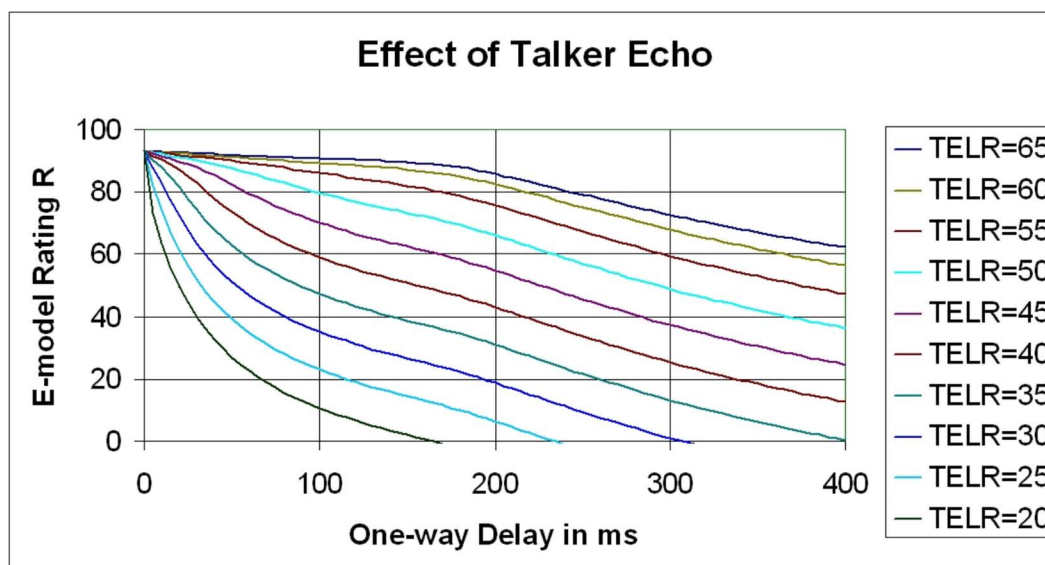

Figure E.1: E-model rating versus delay and TELR

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:

$$TCLw = TELR - 10 \text{ dB}$$
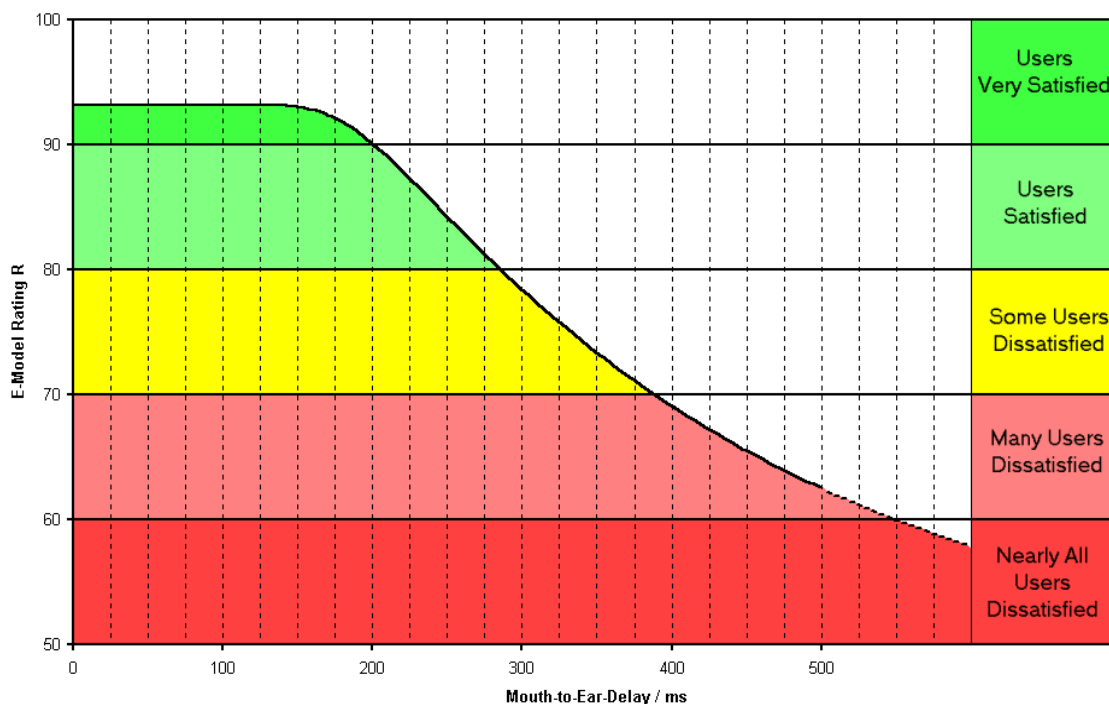
Figure E.2: Subjective rating versus E-model rating and round-trip delay

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 ITU-T Y.1541 [i.24], class 0; the delay of the DECT system has been assumed to be 10 ms each.

> EXAMPLE 1:   **DECT phone -> digital i/f -> NGN -> digital i/f -> DECT phone**
> **T = 170 ms, TCLw = 34.**

This example constitutes a case with an "classic" DECT GAP PP using the low TCLw option.

The resulting quality is **R=60;**

which translates into **"Many Users Dissatisfied/Nearly All Users Dissatisfied".**

> EXAMPLE 2:   **DECT phone -> digital i/f -> NGN -> digital i/f -> DECT phone**
> **T = 170 ms, TCLw = 46.**

This example constitutes a case with an "classic" DECT GAP PP using the high TCLw option.

The resulting quality is **R=80;**

which translates into **"Users Satisfied/Some Users Dissatisfied".**

> EXAMPLE 3:   **DECT phone -> digital i/f -> NGN -> digital i/f -> DECT phone**
> **T = 170 ms, TCLw = 55.**

This example constitutes a case with "improved" DECT GAP PP over an NGN.

The resulting quality is **R=88;**

which translates into **"Users Satisfied".**

> EXAMPLE 4:   **DECT phone -> digital i/f -> no network -> digital i/f -> DECT phone**
> **T = 20 ms, TCLw = 34.**

This example constitutes a case with an "classic" DECT GAP PP using the low TCLw option but no network delay involved.

The resulting quality is **R=89;**

which translates into **"Users Satisfied".**

Annex F (informative):
Guidelines on specific requirements

F.1 Delay 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.23] (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 G.1020 [25], figure 2 and figure A.1 respectively.

The sending delay T(s) is defined as follows:

$$T(s) = T(ps) + T(ead) + T(aif) + T(asp) + T(tra)$$

Where:

    T(ps) = packet size = N 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;

    T(tra) = allowance for transcoding.

The additional delay required for IP packet assembly and presentation to the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time will usually be quite small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.

    NOTE 1: The size of T(aif) is for further study.

Informative values:
In order to use the formula it is needed to use a table such as provided below.

The allowance for signal processing should be T(asp) < T(ps).

    NOTE 2: With the knowledge of the codec specific values for T(fs) 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 [12]	G.711 [13]	80	0,125	10	0	0	10	< 1	< 21
		160	0,125	20	0	0	10	< 1	< 31
G.726 [12]	G.726 [12]	80	0,125	10	0	0	10		< 20
		160	0,125	20	0	0	10		< 30
G.711 [13]	G.711 [13]	80	0,125	10	0	0	10		< 20
		160	0,125	20	0	0	10		< 30
LC3plus [34]	LC3plus [34]	1	10	10	2,5	0	10		< 22,5
		2	10	20	2,5	0	10		< 32,5
LC3plus [34]	G.711 [13]	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 G.1020 [25], figure 3 and figure A.2 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(aif) = air interface framing;

T(dad) = decoder algorithmic delay (filtering, etc.);

T(jb) = jitter buffer size;

T(plc) = PLC buffer size;

T(asp) = allowance for signal processing;

T(tra) = allowance for transcoding.

The additional delay required for IP packet dis-assembly and presentation from the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time will usually be quite small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.

NOTE 1: The size of T(aif) is for further study.

Informative values:
In order to use the formula it is needed to use a table such as provided below.

The allowance for signal processing by decoder should be T(asp) < max (T(plc),T(fs)).

The additional delay introduced by the jitter buffer should be $T(jb) \leq 3 \times T(ps)$.

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 codec 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 [12]	G.711 [13]	80	0,125	0	0	< 30	10	10	< 1	< 51,125
		160	0,125	0	0	< 60	10	10	< 1	< 81,125
G.726 [12]	G.726 [12]	80	0,125	0	0	< 30	10	10		< 50,125
		160	0,125	0	0	< 60	10	10		< 80,125
G.711 [13]	G.711 [13]	80	0,125	0	0	< 30	10	10		< 50,125
		160	0,125	0	0	< 60	10	10		< 80,125
LC3plus [34]	LC3plus [34]	1	10	0	0	< 30	0	10		< 50
		2	10	0	0	< 60	0	10		< 80
LC3plus [34]	G.711 [13]	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 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.1.2 Delay considerations for FP type 5 (Fixed Part with VoIP interface, wideband, super-wideband or fullband service)

F.1.2.0 General

The delay data provided assumes an IEEE 802.3 [i.23] (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 G.1020 [25], figure 2 and figure A.1 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) 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 [14]	G.722 [14]	160	0,0625	10	0	0	10	< 20,0625
		320	0,0625	20	0	0	10	< 30,0625
G.729.1 [17]	G.729.1 [17]	1	20	20	26,97	0	20	< 66,97
MPEG-4 ER AAC-LD [19] 64 kbit/sec	MPEG-4 ER AAC-LD [19] 64 kbit/sec	1	10	10	10	0	10	< 30
MPEG-4 ER AAC-LD [19] 32 kbit/sec	MPEG-4 ER AAC-LD [19] 32 kbit/sec	1	20	20	20	0	20	< 60
LC3plus [34]	LC3plus [34]	1	10	10	2,5	0	10	< 22,5
		2	10	20	2,5	0	10	< 32,5
LC3plus [34]	G.722 [14]	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), the value for delay should be identical.

NOTE 4: There may be some extra delay due to optional features.

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 G.1020 [25], figure 3 and figure A.2 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(aif) = air interface framing;

T(dad) = decoder algorithmic delay (filtering, etc.);

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 [14]	G.722 [14]	160	0,0625	0	0	< 30	10	10	< 50,0625
		320	0,0625	0	0	< 60	10	10	< 80,0625
G.729.1 [17]	G.729.1 [17]	1	20	1,97	0	< 60	0	20	< 101,97
MPEG-4 ER AAC-LD [19] 64 kbit/sec	MPEG-4 ER AAC-LD [19] 64 kbit/sec	1	10	0	0	< 30	0	10	< 50
MPEG-4 ER AAC-LD [19] 32 kbit/sec	MPEG-4 ER AAC-LD [19] 32 kbit/sec	1	20	0	0	< 60	0	20	< 100
LC3plus [34]	LC3plus [34]	1	10	0	0	< 30	0	10	< 50
		2	10	0	0	< 60	0	10	< 80
LC3plus [34]	G.722 [14]	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 Derivation of Delay requirements for PPs and FPs with VoIP interface

Figure F.1 shows an examplary 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 10 ms 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 can not 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 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 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, 7b, 7c, 7d, 7e 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").
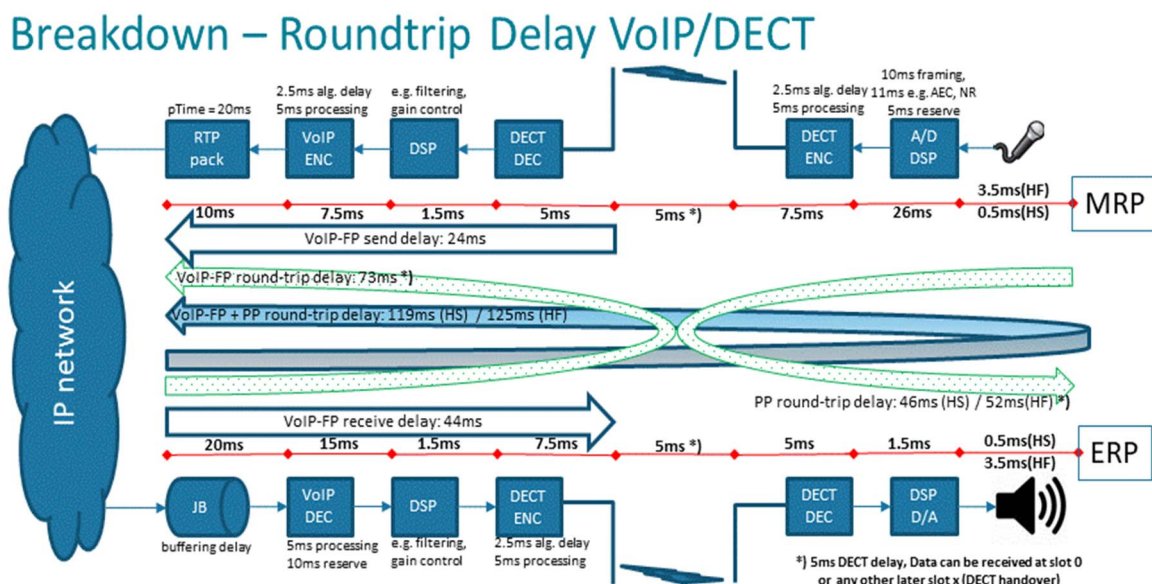

Figure F.1: Delay breakdown - PP and VoIP FP

Annex G (informative):
Bibliography

- ETSI TR 101 310: "Digital Enhanced Cordless Telecommunications (DECT); Traffic capacity and spectrum requirements for multi-system and multi-service DECT applications co-existing in a common frequency band".

- ETSI EN 300 444: "Digital Enhanced Cordless Telecommunications (DECT); Generic Access Profile (GAP)".

- Recommendation ITU-T I.241 (1988): "Teleservices supported by an ISDN".

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

- Recommendation ITU-T G.122: "Influence of national systems on stability and talker echo in international connections".

- ETSI TR 102 648-1: "Speech Processing, Transmission and Quality Aspects (STQ); Test Methodologies for ETSI Test Events and Results; Part 1: VoIP Speech Quality Testing".

- ETSI EG 202 425: "Speech Processing, Transmission and Quality Aspects (STQ); Definition and implementation of VoIP reference point".

- ETSI EG 202 518: "Speech Processing, Transmission and Quality Aspects (STQ); Acoustic Output of Terminal Equipment; Maximum Levels and Test Methodology for Various Applications".

- Recommendation ITU-T P.341 (2005): "Transmission characteristics for wideband (150-7000 Hz) digital hands free telephony terminals".

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

- ETSI TR 103 590: "Integrate Super Wideband Codec in DECT standard for Narrowband, Wideband and Super Wideband audio communication and study option of low delay audio connections (<10ms framing)".

- DECT Forum: "Test specification Audio for CAT-iq 2.0 Devices", V1.19. Doc. No.: DF_CAT-iq T_004_V1.19_2014-02-26.

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

- Recommendation ITU-T P.1010: "Fundamental voice transmission objectives for VoIP terminals and gateways".

Annex H (informative):
Change history

The following table presents main changes from a published version to the next version (published or to be published).

Subject/Comment	Old	New
The enhancement of the DECT base standard to support higher data rates includes the 16 QAM/64 QAM modulation option and the Channel Coding based on the Turbo Code Principle.	1.6.1	1.7.1
No changes - just compatible versions alignment with the other parts.	1.7.1	1.8.1
Aligned with the Recommendations ITU-T to use Equipment Impairment Factor according to the E-Model instead of Quantization distortion and the new more flexible application rules for echo cancelling. Also the sending and receiving loudness rating of the portable part have been aligned with other standards for wireless terminals.	1.8.1	1.9.1
New Generation DECT: A major review of the DECT base standard introducing wideband speech, improved data services, new slot types and other technical enhancements.	1.9.1	2.1.1
Major review of the audio specifications introducing the following improvements: 1) definition of different audio models for different PPs and FPs applications; 2) more detailed technical specification for all types; 3) improvement in TCLw parameter taking into account the increasing use of VoIP in networks; 4) introduction of HATS defined audio specifications; 5) introduction of audio specifications for handsfree (loudspeaking) devices; 6) new echo cancellation strategy in FPs; 7) Informative annex describing the echo scenarios and subjective effects depending on network delays and terminal TCLw. This major review has been performed in a back-compatible way, by keeping existing (V2.1.1) specifications as specific types: type 1a (classic DECT narrowband PP specification) and 2a (wideband PP specification introduced in V2.1.1) for PPs and types 1a (FP with ISDN interface) and 2 (FP with analog 2-wire interface) for FPs.	2.1.1	2.2.1
Addition of informative delay data for MPEG-4 AAC LD codec.	2.2.1	2.3.1
Change in reference to MPEG-4 codec due to revision 2009 of ISO/IEC 14496-3.	2.3.1	2.4.1
No changes.	2.4.1	2.5.1
No changes.	2.5.1	2.6.1
No changes.	2.6.1	2.7.1
New narrowband, wideband, superwideband and fullband audio codec for telephony (LC3plus); alignment with latest versions of standards by ETSI TC STQ (ES 202 737-740; TS 102 924/925).	2.7.1	2.8.1

History

Document history		
Edition 1	October 1992	Publication as ETSI ETS 300 175-8 (Historical)
Edition 2	September 1996	Publication as ETSI ETS 300 175-8 (Historical)
V1.4.2	June 1999	Publication
V1.5.1	February 2001	Publication
V1.6.1	January 2002	Publication
V1.7.1	July 2003	Publication
V1.8.1	November 2004	Publication
V1.9.1	September 2005	Publication
V2.1.1	August 2007	Publication
V2.2.1	February 2009	Publication
V2.3.1	June 2010	Publication
V2.4.1	April 2012	Publication
V2.5.1	August 2013	Publication
V2.6.1	July 2015	Publication
V2.7.1	November 2017	Publication
V2.7.14	August 2019	EN Approval Procedure AP 20191128: 2019-08-30 to 2019-11-28