Recommendation

ITU-T P.383 (03/2023)

SERIES P: Telephone transmission quality, telephone installations, local line networks

Voice terminal characteristics

Technical requirements and test methods for digital headsets or headphones and corresponding interfaces of terminals

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Recommendation ITU-T P.383

Technical requirements and test methods for digital headsets or headphones and corresponding interfaces of terminals

Summary

Recommendation ITU-T P.383 specifies requirements and provides corresponding test methods for headsets and headphones, as well as terminals when tested separately. Headsets and headphones equipped with wired or wireless digital interfaces have been widely used in digital mobile terminals in recent years. The consumer is free to choose either the headset or the headphone originally provided with the terminal or other headsets or headphones that are offered separately. However, the quality of service or quality of experience perceived by users is influenced by both the electrical performance of the interface and the compatibility between the terminal and the headset or headphone.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
1.0	ITU-T P.383	2021-06-13	12	11.1002/1000/14691
2.0	ITU-T P.383	2023-03-01	12	11.1002/1000/15466

Keywords

Digital headset interface, headphones, headsets, terminals, wireless digital interface.

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Technical requirements and test methods for digital headsets or headphones and corresponding interfaces of terminals

1 Scope

This Recommendation specifies technical requirements and test methods for wired or wireless headsets and corresponding digital interfaces. See Figure 1.

This Recommendation aims to ensure adequate user experience when wired or wireless digital headsets are connected over the digital headset interface with digital terminals.

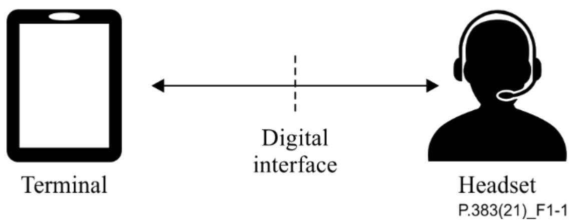


Figure 1 – This Recommendation specifies performance parameters separately for terminals and headsets

Terminals in the context of this Recommendation may be mobile terminals, portable players, information and communication technology terminals, Internet of things devices and augmented reality or /virtual reality devices for communication and multi-media application.

Specifying the performance requirements of the digital interface separately for the headset or headphone on one hand and the terminal on the other enables the consumer to combine commercially available hardware of both types without losing adequate user experience.

This Recommendation applies to digital audio output or input interfaces that wish to support wired or wireless headsets.

This Recommendation does not apply to terminals designed solely for analogue headset or headphone usage.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

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

[ITU-T O.41] Recommendation ITU-T O.41 (1994), *Psophometer for use on telephone-type circuits*.

[ITU-T P.56] Recommendation ITU-T P.56 (2011), *Objective measurement of active speech level*.

[ITU-T P.57] Recommendation ITU-T P.57 (2021), *Artificial ears*.

[ITU-T P.79]	Recommendation ITU-T P.79 (2007), *Calculation of loudness ratings for telephone sets*.
[ITU-T P.340]	Recommendation ITU-T P.340 (2000), *Transmission characteristics and speech quality parameters of hands-free terminals*.
[ITU-T P.381]	Recommendation ITU-T P.381 (2023), *Technical requirements and test methods for analogue wired headsets or headphones and corresponding universal interface of terminals*.
[ITU-T P.382]	Recommendation ITU-T P.382 (2023), *Technical requirements and test methods for analogue wired multi-microphone headsets or headphones and corresponding universal interface of terminals*.
[ITU-T P.501]	Recommendation ITU-T P.501 (2020), *Test signals for use in telephony and other speech-based applications*.
[ITU-T P.502]	Recommendation ITU-T P.502 (2000), *Objective test methods for speech communication systems using complex test signals*.
[ITU-T P.700]	Recommendation ITU-T P.700 (2021), *Calculation of loudness for speech communication*.
[ITU-T P.863]	Recommendation ITU-T P.863 (2018), *Perceptual objective listening quality prediction*.
[EN 50332-1]	European Standard EN 50332-1:2013, *Sound system equipment: Headphones and earphones associated with personal music players – Maximum sound pressure level measurement methodology – Part 1: General method for "one package equipment"*.
[ETSI TS 103 106]	Technical Specification ETSI TS 103 106 V1.6.1 (2021), *Speech and multimedia transmission quality (STQ); Speech quality performance in the presence of background noise: Background noise transmission for mobile terminals – Objective test methods*.
[ETSI TS 103 224]	Technical Specification ETSI TS 103 224 V1.6.1 (2022), *Speech and multimedia transmission quality (STQ); A sound field reproduction method for terminal testing including a background noise database*.
[ETSI TS 103 281]	Technical Specification ETSI TS 103 281 V1.3.1 (2019), *Speech and multimedia transmission quality (STQ); Speech quality in the presence of background noise: Objective test methods for super-wideband and fullband terminals*.
[IEC 61260-1]	International Standard IEC 61260-1:2014, *Electroacoustics – Octave-band and fractional-octave-band filters – Part 1: Specifications*.
[IEC 61672-1]	International Standard IEC 61672-1:2013, *Electroacoustics – Sound level meters – Part 1: Specifications*.

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following terms defined elsewhere:

3.1.1 artificial ear [ITU-T P.381]: A device that incorporates an acoustic coupler and a calibrated microphone for measuring sound pressure, and which has an overall acoustic impedance similar to that of the average adult ear over a given frequency band.

NOTE – Definition based on that in [b-ITU-T P.10].

3.1.2 codec [ITU-T P.381]: Combination of an analogue-to-digital encoder and a digital-to-analogue decoder operating in opposite directions of transmission in the same equipment.

3.1.3 composite source signal (CSS) [b-ITU-T P.10]: A signal composed in time by various signal elements.

3.1.4 eardrum reference point (DRP) [b-ITU-T P.10]: A point located at the end of the ear canal, corresponding to the eardrum position.

3.1.5 earphone [b-IEC 60268-7]: Electroacoustic transducer by which acoustic oscillations are obtained from electric signals and intended to be closely coupled acoustically to the ear.

3.1.6 head and torso simulator (HATS) for telephonometry [ITU-T P.381]: A manikin that extends downwards from the top of the head to the waist. It is designed to simulate the sound pick-up characteristics and the acoustic diffraction produced by the average adult, and to reproduce the acoustic field generated by the human mouth

NOTE – Definition based on that in [b-ITU-T P.10].

3.1.7 headphone (HP) [ITU-T P.381]: An object based on the assembly of one or two earphones on a headband or chinband, the use of which may be optional (e.g., with intra-concha earphones).

NOTE – Definition based on that in [b-IEC 60268-7].

3.1.8 headset [b-ITU-T P.10]: A device which includes a telephone receiver and transmitter which is typically secured to the head or the ear of the wearer.

3.1.9 mean opinion score – listening-only quality objective (MOS-LQO) [ITU-T P.381]: A score calculated by means of an objective model that aims to predict the quality for a listening-only test situation.

NOTE – Objective measurements made using the model given in [ITU-T P.863] give results in terms of MOS-LQO.

3.1.10 mouth reference point (MRP) [b-ITU-T P.10]: Point 25 mm in front of and on the axis of the lip plane of the artificial mouth or a typical human mouth (see Figure A.1 of [b-ITU-T P.64]).

3.1.11 Receive [ITU-T P.381]: The receiving direction of the signal transmission, usually from the measurement system to the device under test.

3.1.12 Send [ITU-T P.381]: The sending direction of the signal transmission, usually from the device under test to the measurement system.

3.2 Terms defined in this Recommendation

None.

4 Abbreviations, acronyms and symbols

4.1 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

AGC Automatic Gain Control

CSS Composite Source Signal

DRP eardrum Reference Point

EC Echo Canceller

EL Echo Loss

FB Fullband

FFT	Fast Fourier Transform
HATS	Head and Torso Simulator
HTCL	Headset Terminal Coupling Loss
JLR	Junction Loudness Rating
JLRr	Junction Loudness Rating in the receive direction
JLRs	Junction Loudness Rating in the send direction
L	Left
LQO	Listening-only Quality Objective
LQOf	Listening-only Quality Objective, fullband
LQOn	Listening-only Quality Objective, narrowband
LQOw	Listening-only Quality Objective, wideband
MOS	Mean Opinion Score
MRP	Mouth Reference Point
NB	Narrowband transmission
PN	Pseudo-Noise
POI	Point of Interconnection
R	Right
RLL	Receive Loudness Level
RLR	Receive Loudness Rating
SLL	Send Loudness Level
SLR	Send Loudness Rating
SRW	Short-Range Wireless
STMR	Sidetone Masking Rating
SWB	Super-Wideband
TCL	Terminal Coupling Loss
TCLw	weighted Terminal Coupling Loss
TELRdt	Talker Echo Loudness Rating under double talk conditions
UE	User Equipment
WB	Wideband

4.2 Symbols

$A_{\mathrm{H,R,DT}}$	attenuation range in the receive direction during double talk
$A_{\mathrm{H,S,dt}}$	attenuation range in the send direction during double talk
dBFS	decibels relative to full scale
dBm0	decibel milliwatts measured at a zero transmission level point
dBPa	decibel sound pressure level relative to 1 Pa
dBV	decibels relative to 1 V

dBVp	decibels relative to 1 V, psophometric noise weighting
f_s	frequency of sine-wave signal
L_{meST}	sidetone path loss
T_r	delay in the receive direction
$T_{r,S,min}$	minimum built-up time in Send
T_{rtd}	round trip delay
T_s	delay in the send direction
$T_{wireless,R}$	Receive delay of terminal with short-range wireless interface
$T_{wireless,rtd}$	round trip delay of terminal with short-range wireless interface
$T_{wireless,S}$	Send delay of terminal with short-range wireless interface

5 Conventions

None.

6 General description

[ITU-T P.381] and [ITU-T P.382] specify performance requirements and corresponding test methods for standardized analogue headset or headphone interfaces used in digital mobile terminals. These Recommendations do not cover digital headset interfaces. Mobile terminals and other devices provided only with digital headset interfaces are becoming available. These digital headset interfaces may provide multiple functionalities including:

– personal hearing preference adjustment;

– application control, motion detection.

From the communication and multi-media perspectives, specifying performance requirements and standardizing a digital interface increases the possibilities of providing adequate user experience when different headsets are used with different devices, which consequently helps in reducing e-waste in the future.

7 Options of possible physical interfaces

This Recommendation does not specify the specific physical form or dimensions of digital headset interfaces; only the electro-acoustic performance at the digital interface and the digital headsets are specified, so that adequate user experience can be achieved.

Terminals providing digital headset interface and digital headsets using any digital audio input/output interfaces or utilizing various wireless access technologies as input/output interfaces to transmit audio or speech, are recommended to meet the requirements specified in clauses 8 and 9.

Two generic configurations are considered in this Recommendation:

1) Digital headsets and terminals, where the speech signal processing such as equalization, automatic gain control (AGC), noise cancellation and echo cancellation is performed in the headset. Test methods for this case are described in clause 8 (digital interface) and in clause 9 (digital headsets). These types of headsets are often equipped with higher signal processing capabilities, while terminals providing such headset interfaces are used for audio gateway functionality e.g., in vehicle hands-free systems. These connection types are typically wireless.

2) Digital headsets and terminals, where the speech signal processing such as equalization, AGC, noise cancellation and echo cancellation is performed in the terminal. Test methods

for this case are described in clause 8 (digital interface) and in clause 9 (digital headsets). These types of headsets are typically equipped with very low or no signal processing capabilities, the terminal connected via this headset interface is required to perform it. These connection types are typically wired but can also be wireless.

8 Terminal digital interface specification (speech signal processing in headset)

8.1 Communication mode

The digital headset interface shall meet corresponding requirements listed for the supported speech bandwidth including narrowband (NB), wideband (WB), super-wideband (SWB), and fullband (FB), depending on the speech communication modes the terminal supports.

Wireless digital headsets perform their own signal processing. Whatever wireless interface is used, a signalling mechanism shall be provided, ensuring unambiguously the deactivation of any signal processing in the terminal. Only access technology-related signal processing such as jitter buffer and packet loss concealment of packet-switched transmission networks shall be performed in the terminal. Specific protocols used for the digital headset interface are not covered by this Recommendation.

NOTE – Any signal processing in the terminal such as noise reduction and echo cancellation when connected to a digital wireless headset that has already performed signal processing should be avoided. Cascaded signal processing may degrade speech and conversation quality significantly.

8.1.1 Test set-up

8.1.1.1 Test configuration and test system

When testing wireless digital headset interfaces in terminals, a wireless reference interface as shown in Figure 8-1 is used to establish the audio transmission between the test system and the headset interface. It shall be able to simulate the essential functionalities of a wireless digital headset including necessary protocol handling to set up an audio link between the reference interface and the terminal. The wireless digital headset reference interface shall not introduce any amplification or attenuation in the audio stream from or to the headset interface. No additional signal processing except the audio or speech coding, or transcoding shall be active. The reference interface shall provide a signalling mechanism that is intended to deactivate any signal processing in the terminal.

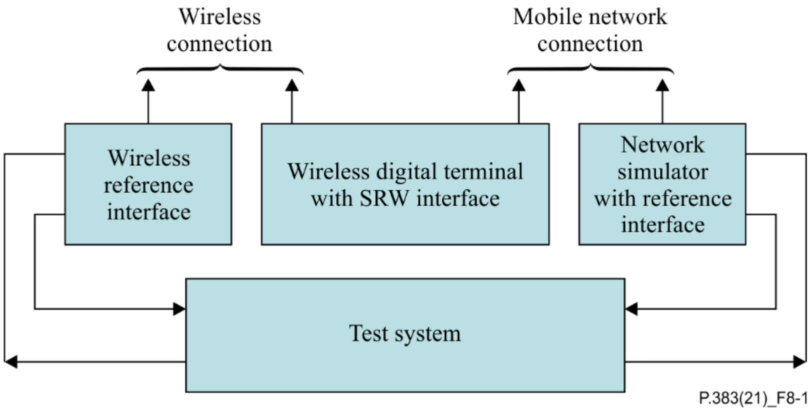


Figure 8-1 – Test set-up for testing the digital headset interface

For testing echo and double talk scenarios, an artificial feedback of the received signal into the Send path as shown in Figure 8-2 shall be used. This echo path shall be realized as part of the test system. The received and decoded signal from the user equipment (UE) is fed back into the sending direction, in advance to the encoding/protocol/hardware layer. For measurements without artificial echo loss (EL), the feedback path is disabled.

NOTE – Evaluation boards from digital headset chipset vendors may be used for the implementation of the digital headset reference interface.

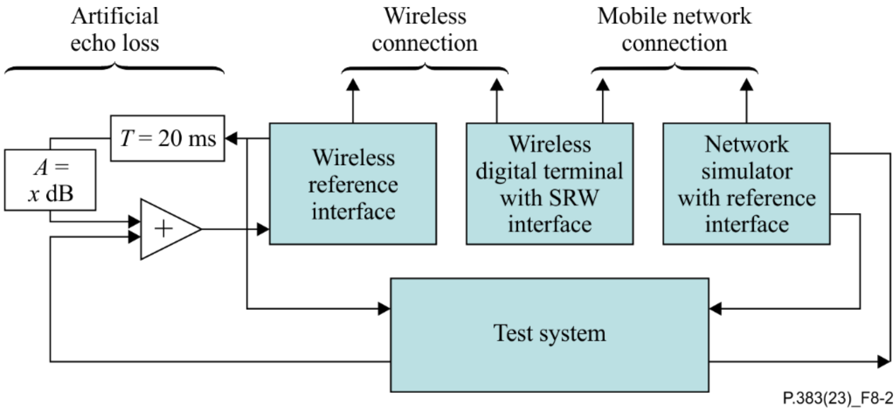


Figure 8-2 – Test set-up with artificial echo loss for echo and double talk testing

8.1.1.2 Test signals and test signal levels

Unless otherwise specified, FB real speech signals, which can be found in [ITU-T P.501], are used for measurements. Detailed information about the test signal used can be found in the corresponding clause of [ITU-T P.501]. For test cases where composite source signals (CSSs) are specified, the speech spectrum-shaped CSSs specified in [ITU-T P.501] shall be used.

All test signals used in the Receive tests must be band limited using a bandpass filter with a roll-off of more than 24 dB/octave. In NB mode, the signal shall be band limited between 100 Hz and 4 kHz. In WB mode, the signal shall be band limited between 100 Hz and 8 kHz. In SWB mode, the frequency range shall be band limited between 50 Hz and 16 kHz. In FB mode, the signal shall be band limited between 20 Hz and 20 kHz.

In Send, the test signals are used without band limitation, other than what is inherent in the digital interface.

For real speech, the test signal levels are referred to the [ITU-T P.56] active speech level of the (band-limited in receiving direction) test signal, calculated over the complete test sequence, unless described otherwise. For other test signals, the test signal levels are referred to the average level of the (band limited in the receiving direction) test signals, averaged over the complete test sequence length.

Unless stated otherwise, the nominal average signal level for the measurements is −16 dBm0 at the electrical interfaces (includes digital headset reference interface and the system simulator), for Send and Receive.

Some tests require the exact synchronization of test signals in the time domain. Therefore, consideration is required of the delays of the terminals. When analysing signals, any delay introduced by the test system, codecs and terminals must be taken into account accordingly.

8.1.2 Delay for the communication mode (terminal)

8.1.2.1 Requirements

The delay in the send direction is measured from the wireless reference interface to the point of interconnection (POI; speech codec of the system simulator output), and in the receive direction from the POI to the wireless reference interface and it is corrected for test equipment delays. See Figure 8-3.

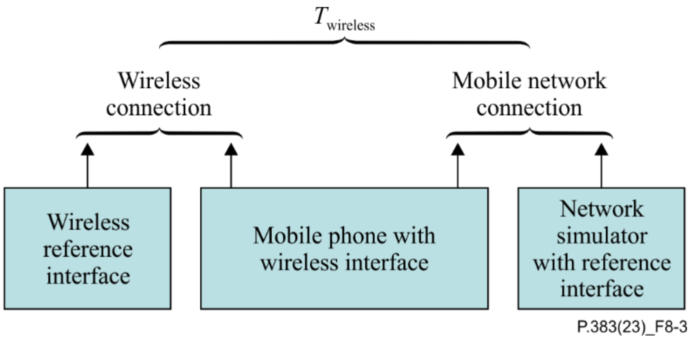


Figure 8-3 – Delay in wireless headset connections

Definitions:

– The mobile phone delay in the send (uplink) direction $T_{\text{wireless,S}}$ is the delay between the first bit of a wireless speech frame at the wireless antenna to the last bit of the corresponding speech frame at the mobile network antenna.

– The mobile phone delay in the receive (downlink) direction with short-range wireless (SRW) interface $T_{\text{wireless,R}}$ is the delay between the first bit of a speech frame at the mobile network antenna and the first bit of a wireless speech frame at the wireless antenna corresponding to that speech frame.

According to these definitions and for error-free radio conditions, the round trip delay of terminal with SRW interface,

$$T_{\text{wireless,rtd}} = T_{\text{wireless,S}} + T_{\text{wireless,R}}$$

where

$T_{\text{wireless,S}}$ is the Send delay of terminal with SRW interface;

$T_{\text{wireless,R}}$ is the Receive delay of terminal with SRW interface,

shall be less than 150 ms.

NOTE 1 – The delay $T_{\text{wireless,rtd}}$ should be minimized.

NOTE 2 – The system delay T_{sys} depends on the transmission method used and the network simulator. The delay T_{sys} must be known.

NOTE 3 – For 3GPP UMTS circuit-switched speech and 3GPP LTE MTSI-based speech, definitions, test methods, performance objectives, and requirements are found in [b-3GPP TS 26.131] and [b-3GPP TS 26.132].

8.1.2.2 Test

8.1.2.2.1 Test of overall delay in Send

1) The test signal to be used for the measurements is a CSS, as described in [ITU-T P.501]. The test signal consists of the voiced part followed by a pseudo-random noise sequence with a minimum periodicity of 500 ms.

2) The test signal level is −16 dBm0, measured at the Send input of the digital headset reference interface.

3) The delay is calculated using the cross-correlation function between the signal at the Send input of digital headset reference interface and the signal at the system simulator output.

4) The measurement is corrected by the delay in the sending direction introduced by the test equipment ($T_{\text{S,sys}}$). The sending delay ($T_{\text{S,wdt}}$), expressed in milliseconds, is determined from the maximum of the cross-correlation function.

8.1.2.2.2 Test of overall delay in Receive

1) The test signal to be used for the measurements is a CSS, as described in [ITU-T P.501]. The test signal consists of the voiced part followed by a pseudo-random noise sequence with a minimum periodicity of 500 ms.

2) The test signal level is −16 dBm0, measured at the output of the system simulator.

3) The delay is calculated using the cross-correlation function between the signal at the system simulator and the signal at the Receive output of the digital headset reference interface.

4) The measurement is corrected by the delay in the receive direction introduced by the test equipment ($T_{R,sys}$). The Receive delay ($T_{R,wdt}$), expressed in milliseconds, is determined from the maximum of the cross-correlation function.

8.1.3 Junction loudness rating in Send for the communication mode (terminal)

8.1.3.1 Requirements

The nominal values of the junction loudness rating in the send direction (JLRs) should be:

$$\text{JLRs} = 0 \pm 2 \text{ dB}.$$

8.1.3.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.

2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 8.1.1.2. The power density spectrum at the sending input of the headset reference interface is used as the reference power density spectrum for determining the sending sensitivity.

3) For NB, the Send sensitivity is calculated from each band of the frequencies given in Table 1 of [ITU-T P.79], bands 4 to 17. For WB, SWB and FB, the Send sensitivity is calculated from each band of the frequencies given in Table A.2 of [ITU-T P.79], bands 1 to 20. For the calculation, the average measured level at the output of the system simulator for each frequency band is referred to the reference signal.

4) The sensitivity is expressed in decibels. For NB, the junction loudness rating (JLR) is calculated according to Equation A-23d of [ITU-T P.79], bands 4 to 17, $m = 0.175$ and the weighting factors (W_J) for JLR according to Table A.2 of [ITU-T P.79]. For WB, SWB and FB, the JLR is calculated according to Equation A-23d of [ITU-T P.79], bands 1 to 20.

8.1.4 Junction loudness rating in Receive for the communication mode (terminal)

8.1.4.1 Requirements

The nominal values of the junction loudness rating in the receive direction (JLRr) should be:

$$\text{JLRr} = 0 \pm 2 \text{ dB}.$$

It is recommended that the terminal with digital headset interfaces transmit the audio signals in the sending and receiving directions in a transparent and neutral way when connecting digital headsets. The digital headset interfaces should not add gains or attenuate the transmitted signal regardless of the volume setting of the terminal.

8.1.4.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.

2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 7.1.1.2. The measured power

density spectrum at the receiving input of the system simulator is used as the reference power density spectrum for determining the Receive sensitivity.

3) For NB, the Send sensitivity is calculated from each band of the frequencies given in Table 1 of [ITU-T P.79], bands 4 to 17. For WB, SWB and FB, the Send sensitivity is calculated from each band of the frequencies given in Table A.2 of [ITU-T P.79], bands 1 to 20. For the calculation, the average measured level at the receiving output of the digital headset reference interface for each frequency band is referred to the reference signal.

4) The sensitivity is expressed in decibels. For NB, the JLR is calculated according to Equation A-23d of [ITU-T P.79], bands 4 to 17, $m = 0.175$ and the weighting factors (W_J) for JLR according to Table A.2 of [ITU-T P.79]. For WB, SWB and FB, the JLR is calculated according to Equation A-23d of [ITU-T P.79], bands 1 to 20.

8.1.5 Linearity in Send for the communication mode (terminal)

8.1.5.1 Requirements

For acoustical signal level variation in the range −30 dB to 0 dB from the nominal signal level, the measured JLRs shall not deviate by more than ±2 dB from the JLRs measured with the nominal signal level.

8.1.5.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.

2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level shall be the active speech level according to [ITU-T P.56]. The test signals are in the range −30 dB to 0 dB in steps of 10 dB relative to the nominal signal level –16 dBm0, measured at the sending input of the digital headset reference interface.

3) JLRs is calculated according to clause 8.1.3.1 at each test signal level.

8.1.6 Linearity in Receive for the communication mode (terminal)

8.1.6.1 Requirements

For acoustical signal level variation in the range–30 dB to 0 dB from the nominal signal level, the measured JLRr shall not deviate by more than ±2 dB from the JLRr measured with the nominal signal level.

8.1.6.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.

2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level shall be the active speech level according to [ITU-T P.56]. The test signals are in the range −30 dB to 0 dB in steps of 10 dB relative to the nominal signal level –16 dBm0, measured at the input of the system simulator.

3) JLRr is calculated according to clause 8.1.4.1 at each test signal level.

8.1.7 Send sensitivity/frequency response for the communication mode (terminal)

8.1.7.1 Requirements

The Send sensitivity frequency response is measured from the sending input of digital headset reference interface to the output of the system simulator.

The tolerance mask for the Send sensitivity frequency response is shown in Tables 8-1, 8-2, 8-3 and 8-4.

Table 8-1 – Tolerance mask for the narrowband sending frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**	**Target**
100	0	–∞	0
200	0	–∞	0
300	0	–3	0
1 000	0	–3	0
3 100	0	–4	0
3 400	0	–4	0
4 000	0	–∞	0
NOTE 1 – All sensitivity values are expressed in db on an arbitrary scale. NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (db) scale.			

Table 8-2 – Tolerance mask for the wideband sending frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**	**Target**
100	0	–3	0
200	0	–3	0
1 000	0	–3	0
3 000	0	–4	0
5 000	0	–4	0
6 300	0	–4	0
8 000	0	–∞	0
NOTE 1 – All sensitivity values are expressed in db on an arbitrary scale. NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.			

Table 8-3 – Tolerance mask for the super-wideband sending frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**	**Target**
100	0	–3	0
200	0	–3	0
1 000	0	–3	0
3 000	0	–4	0
5 000	0	–4	0
8 000	0	–4	0
12 500	0	–4	0
16 000	0	–4	0
NOTE 1 – All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (db) scale.			

Table 8-4 – Tolerance mask for the fullband sending frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**	**Target**
100	0	–3	0
200	0	–3	0
1 000	0	–3	0
3 000	0	–4	0
5 000	0	–4	0
8 000	0	–4	0
12 500	0	–4	0
20 000	0	–4	0
NOTE 1 – All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.			

8.1.7.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.
2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 7.1.1.2. The power density spectrum at the sending input of the headset reference interface is used as the reference power density spectrum for determining the sending sensitivity.
3) In FB, the sending sensitivity is determined in one-12th octave bands, as given by [IEC 61260-1] for frequencies from 100 Hz to 20 kHz inclusive, measured at the output of the system simulator. In SWB, it is determined for frequencies from 100 Hz to 16 kHz. In WB, it is determined for frequencies from 100 Hz to 8 kHz. In NB, it is determined for frequencies from 100 Hz to 4 kHz. In each one-third octave band, the level of the measured signal is referred to the level of the reference signal.
4) The sensitivity is determined in decibels.

8.1.8 Receive sensitivity and frequency response for the communication mode (terminal)

8.1.8.1 Requirements

The Receive sensitivity and frequency response is measured from the input of the system simulators to the receiving output of the digital headset reference interface.

The tolerance mask for the Send sensitivity frequency response is shown in Tables 8-5, 8-6, 8-7, and 8-8.

Table 8-5 – Tolerance mask for the narrowband receiving frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**	**Target**
100	0	$-\infty$	0
200	0	$-\infty$	0
300	0	–3	0
1 000	0	–3	0
3 100	0	-4	0
3 400	0	–4	0

Table 8-5 – Tolerance mask for the narrowband receiving frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**	**Target**
4 000	0	–∞	0
NOTE 1 – All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.			

Table 8-6 – Tolerance mask for the wideband receiving frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**	**Target**
100	0	–3	0
200	0	–3	0
1 000	0	–3	0
3 000	0	–4	0
5 000	0	–4	0
6 300	0	–4	0
8 000	0	–∞	0
NOTE – All sensitivity values are expressed in dB on an arbitrary scale. The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale.			

Table 8-7 – Tolerance mask for the super-wideband receiving frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**	**Target**
100	0	–3	0
200	0	–3	0
1 000	0	–3	0
3 000	0	–4	0
5 000	0	–4	0
8 000	0	–4	0
12 500	0	–4	0
16 000	0	–4	0
NOTE – All sensitivity values are expressed in dB on an arbitrary scale. The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale.			

Table 8-8 – Tolerance mask for the fullband receiving frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**	**Target**
100	0	–3	0
200	0	–3	0
1 000	0	–3	0

Table 8-8 – Tolerance mask for the fullband receiving frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**	**Target**
3 000	0	–4	0
5 000	0	–4	0
8 000	0	–4	0
12 500	0	–4	0
20 000	0	–4	0
NOTE – All sensitivity values are expressed in dB on an arbitrary scale. The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale.			

8.1.8.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.
2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 8.1.1.2. The power density spectrum at the input of the system simulator is used as the reference power density spectrum for determining the receiving sensitivity.
3) In FB, the receiving sensitivity is determined in one-third octave bands, as given in [IEC 61260-1] for frequencies from 100 Hz to 20 kHz inclusive, measured at the receiving output of digital headset reference interface. In SWB, it is determined for frequencies from 100 Hz to 16 kHz. In WB, it is determined for frequencies from 100 Hz to 8 kHz. In NB, it is determined for frequencies from 100 Hz to 4 kHz. In each one-third octave band, the level of the measured signal at the receiving output of the digital headset reference interface is referred to the level of the reference signal.
4) The sensitivity is determined in dB.

8.1.9 Noise in Send for the communication mode (terminal)

8.1.9.1 Requirements

The maximum idle channel noise measured in the send direction shall be less than −64 dBm0(A). Spectral peaks in the frequency domain shall not exceed the averaged spectrum by more than 10 dB.

8.1.9.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.
2) For the noise measurement, no test signal is used. However, all sources that potentially contribute to noise should be considered. Interference from radio frequencies is not accurately covered by an interface specification as the complete terminal and headset system needs to be assessed. Moreover, the necessary test system cabling is likely to introduce further deviations from real-life conditions. Therefore, radio-induced noise is not expected to be accurately covered by the test cases in this Recommendation.
3) The noise is measured at the output of the system simulator in the frequency range between 100 Hz and 4 kHz for NB, between 100 Hz and 8 kHz for WB, between 100 Hz and 16 kHz for SWB, and between 100 Hz and 20 kHz for FB. The length of the time window is 1 s, which is the averaging time for the idle channel noise. The power density spectrum of the noise signal is determined using the fast Fourier transform (FFT) (8 000 samples at 48 kHz sampling rate or equivalent). A Hann window is used.
4) The noise is determined by A-weighting [IEC 61672-1].

5) Spectral peaks are measured in the frequency domain. The frequency spectrum of the idle channel noise is measured by a spectral analysis having a noise bandwidth of 8.79 Hz (determined using FFT 8 000 samples at 48 kHz sampling rate with a Hann window or equivalent). A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) one-third octave wide across the idle noise channel spectrum stated in decibels (linear average in decibels of all FFT bins in the range from $2^{-1/6}f$ to $2^{+1/6}f$). Peaks in the idle channel noise spectrum are compared with a smoothed average idle channel noise spectrum from 100 Hz to 3.4 kHz in NB, from 100 Hz to 6.3 kHz in WB, from 100 Hz to 13 kHz in SWB and from 100 Hz to 18 kHz in FB.

8.1.10 Noise in Receive for the communication mode (terminal)

8.1.10.1 Requirements

The maximum idle channel noise measured in the receive direction shall be less than −64 dBm0(A). Spectral peaks in the frequency domain shall not exceed the averaged spectrum by more than 10 dB.

8.1.10.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.
2) For the noise measurement, no test signal is used. However, all sources that potentially contribute to noise should be considered. Interference from radio frequencies is not accurately covered by an interface specification as the complete terminal and headset system needs to be assessed. Moreover, the necessary test system cabling is likely to introduce further deviations from real-life conditions. Therefore, radio-induced noise is not expected to be accurately covered by the test cases in this Recommendation.
3) The noise is measured at the receiving output of the digital headset reference interface in the frequency range between 100 Hz and 4 kHz for NB, between 100 Hz and 8 kHz for WB, between 100 Hz and 16 kHz for SWB and between 100 Hz and 20 kHz for FB. The length of the time window is 1 s, which is the averaging time for the idle channel noise. The power density spectrum of the noise signal is determined using the FFT (8 000 samples at 48 kHz sampling rate or equivalent). A Hann window is used.
4) The noise is determined by A-weighting [IEC 61672-1].
5) Spectral peaks are measured in the frequency domain. The frequency spectrum of the idle channel noise is measured by a spectral analysis having a noise bandwidth of 8.79 Hz (determined using FFT 8 000 samples at 48 kHz sampling rate with a Hann window or equivalent). A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) one-third octave wide across the idle noise channel spectrum stated in decibels (linear average in decibels of all FFT bins in the range $2^{-1/6}f$ to $2^{+1/6}f$). Peaks in the idle channel noise spectrum are compared with a smoothed average idle channel noise spectrum from 100 Hz to 3.4 kHz in NB, from 100 Hz to 6.3 kHz in WB, from 100 Hz to 13 kHz in SWB and from 100 Hz to 18 kHz in FB.

8.1.11 Send distortion for the communication mode (terminal)

8.1.11.1 Requirements

The distortion in Send is measured from the sending input of the headset reference interface to the output of the system simulator.

The ratio of signal to harmonic distortion shall be above the following mask. See Table 8-9.

Table 8-9 – Limits for the sending signal to harmonic distortion

Frequency (Hz)	**Signal to harmonic distortion ratio limit, Send** (dB)
315	30
400	40
1 000	40
NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale.	

8.1.11.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.
2) For the test, a sinusoidal signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, and 1 000 Hz are used. The duration of the sine wave shall be <1 s. The sinusoidal signal level is the nominal signal level. To ensure a reliable activation, a conditioning sequence is inserted before the actual measurement. The conditioning sequence is according to clause 7.3.7 of [ITU-T P.501]. The short conditioning sequences (either male or female) should be used. The level of the activation signal is the nominal signal level.
3) The signal to harmonic distortion ratio is measured up to 3.5 kHz for NB, up to 7 kHz for WB, up to 14 kHz for SWB and up to 20 kHz for FB.
4) The test is repeated using a signal level 10 dB higher than the nominal signal level. The level of the activation signal is kept at the nominal signal level.

8.1.12 Receive distortion for the communication mode (terminal)

8.1.12.1 Requirements

The distortion in Receive is measured from the system simulator to the receiving output of the digital headset reference interface.

The ratio of signal to harmonic distortion shall be above the following mask. See Table 6-10.

Table 8-10 – Limits for the receiving signal to harmonic distortion

Frequency (Hz)	**Signal to harmonic distortion ratio limit, Receive** (dB)
315	30
400	40
1 000	40
NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale.	

8.1.12.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.
2) For the test, a sinusoidal signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, and 1 000 Hz is used. The duration of the sine wave shall be <1 s. The sinusoidal signal level is the nominal signal level. To ensure a reliable activation, a conditioning sequence is inserted before the actual measurement. The conditioning sequence is according to clause 7.3.7 of [ITU-T P.501]. The short conditioning sequences (either male or female) should be used. The level of the activation signal is the nominal signal level.

3) The signal to harmonic distortion ratio is measured selectively up to 7 kHz for NB and WB and up to 20 kHz for SWB and FB.

4) The test is repeated using a signal level 10 dB higher than the nominal signal level. The level of the activation signal is kept at the nominal signal level.

8.1.13 Presence of noise reduction for the communication mode (terminal)

8.1.13.1 Requirements

The intention of this test is to check whether the noise reduction in the terminal is active when a digital headset interface is used. The noise reduction shall not be active when the digital headset interface is used. When a simulated background noise is inserted at the sending input of the digital headset reference interface, the range of the attenuation of the simulated background noise shall be less than 4 dB.

8.1.13.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-1.

2) For the test, pink noise with 20 s duration is used as the test signal, at the nominal level described in clause 8.1.1.2.

3) The transmitted signal is measured at the output of the system simulator and is referred to the test signal as level versus time analysis in dB. The result represents the attenuation of the pink noise (simulated background noise). The level versus time analysis is calculated from the time domain signals using an integration time of 250 ms.

8.1.14 Presence of echo cancellation for the communication mode (terminal)

8.1.14.1 Requirements

The intention of this test is to check whether the echo cancellation in the terminal is active when a digital headset interface is used.

When an artificial echo path consisting of an attenuation of 20 dB and a delay of 20 ms is inserted at the sending input of the digital headset reference interface. The EL measured shall be 20 dB ±2 dB.

8.1.14.2 Test

1) The test set-up is according to clause 8.1.1, Figure 8-2.

2) The attenuation between the input of the system simulator to the output of the system simulator is measured using a speech-like test signal.

3) The test signal is the compressed real speech signal described in clause 7.3.3 of [ITU-T P.501]. The signal level is −10 dBm0.

4) The first 17.0 s of the test signal (six sentences) are discarded from the analysis to allow for convergence of the acoustic echo canceller (EC). The analysis is performed over the remaining length of the test sequence (last six sentences).

5) The analysis shall be conducted in one-third octave bands, as given in [IEC 61260-1]. For the calculation, the average measured echo level at each frequency band is referred to the average level of the test signal measured in each frequency band. For NB, the frequency range used is from 300 Hz to 3 400 Hz. For WB, SWB, and FB, the frequency range used is from 100 Hz to 8.0k Hz.

6) For NB mode, weighted terminal coupling loss (TCLw) is calculated according to clause B.4 of [ITU-T G.122] (trapezoidal rule). In WB, SWB and FB mode, terminal coupling loss (TCL) is calculated as the EL from 100 Hz to 8.0 kHz. For the calculation, the averaged test signal level at each frequency band is referred to the averaged measured echo signal level in each frequency band. For the measurement, a time window has to be applied that is adapted to the duration of the actual test signal. The EL is calculated by the following equations. The

form of the first is generalized from of the equation specified in clause B.4 of [ITU-T G.122] to calculate EL based on tabulated data, which allows the calculation of EL within any frequency range between f_0 and f_N.

$$L_e = C - 10 \log_{10} \sum_{i=1}^{N} (A_i + A_{i-1})(\log_{10} f_i - \log_{10} f_{i-1}),$$

$$\text{where} \quad C = 10 \log_{10} 2 \,(\log_{10} f_N - \log_{10} f_0),$$

where

A_0 is the output/input power ratio at frequency $f_0 = 100$ Hz;

A_1 is the output/input power ratio at frequency f_i;

A_N is the output/input power ratio at frequency $f_N = 8.0$ kHz

7) The measured TCL is corrected by the measured JLRs and JLRr.

8.2 Multimedia playback mode

For further study.

9 Digital headset specifications (wireless)

9.1 Communication mode

The digital headset should meet corresponding requirements listed for supported speech bandwidth including NB, WB, SWB and FB, depending on the speech communication modes the terminal supports.

In general, the digital headset should communicate with the terminal if the digital headset is capable of and already performs signal processing, the terminal should disable any additional signal processing, such as noise reduction and echo cancellation, so that complete signal processing that degrades conversation quality can be avoided.

The digital headset should at least meet the requirements for NB if the speech communication mode cannot be configured, it is recommended to support wider bandwidth in the communication mode.

9.1.1 Test set-up

9.1.1.1 Test configuration and test system

The test set-up is shown in Figure 9-1. Requirements for a head and torso simulator (HATS) and ear simulators are specified in clause 8.2.1 of [ITU-T P.381].

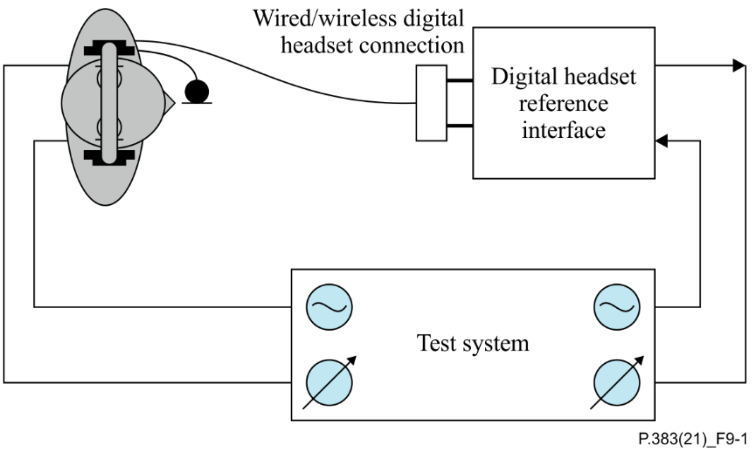


Figure 9-1 – Test set-up for the headset

When testing, a reference interface is used to establish the audio transmission between the test system and digital headsets. It shall be able to simulate the essential functionalities of a digital headset interface of a terminal including necessary protocol handling in order to set up an audio link between the reference interface and the digital headset. It shall be capable of configuring the digital headset into a certain state to support speech communication. The digital headset reference interface shall not introduce any amplification or attenuation in the audio stream from or to the headset. No additional signal processing except the audio or speech coding or transcoding shall be active.

NOTE – Evaluation boards from digital headset chipset vendors may be used for implementation of the digital headset reference interface.

9.1.1.2 Test signals and test signal levels

Unless otherwise specified, FB real speech signals, which can be found in [ITU-T P.501], are used for measurements. Detailed information about the test signal used can be found in the corresponding clause of [ITU-T P.501]. For test cases where CSSs are specified, the speech spectrum-shaped CSSs specified in [ITU-T P.501] shall be used.

All test signals used in Receive must be band limited using a bandpass filter with a roll-off of more than 24 dB/octave. In NB mode, the signal shall be band limited between 100 Hz and 4 kHz. In WB mode, the signal shall be band limited between 100 Hz and 8 kHz. In SWB mode, the signal shall be band limited between 50 Hz and 16 kHz. In FB mode, the signal shall be band limited between 20 Hz and 20 kHz.

In Send, the test signals are used without band-limitation.

For real speech, the test signal levels are referred to the [ITU-T P.56] active speech level of the (band-limited in receiving direction) test signal, calculated over the complete test sequence, unless described otherwise. For other test signals, the test signal levels are referred to the average level of the (band-limited in receiving direction) test signals, averaged over the complete test sequence length.

Unless stated otherwise, the nominal average signal levels for the measurements are −16 dBm0 at electrical interfaces (includes digital headset reference interface) and −4.7 dBPa at the mouth reference point (MRP).

Some tests require exact synchronization of test signals in the time domain. Therefore, consideration of terminal delays is required. When analysing signals, any delay introduced by the test system, codecs and terminals must be taken into account accordingly.

9.1.1.3 Positioning of the headsets

The same guidelines and requirements on headset positioning apply as in clause 8.1.1.3 of [ITU-T P.381].

9.1.1.4 Position and calibration of HATS

The same requirements on position and calibration of a HATS apply as specified in clause 8.1.1.4 of [ITU-T P.381].

9.1.1.5 Test set-up for quality in the presence of ambient noise measurements

The set-up for simulating realistic ambient noises and the positioning of the HATS in a laboratory-type environment is described in [ETSI TS 103 224].

[ETSI TS 103 224] contains a description of the recording arrangement for realistic ambient noises, a description of the set-up for a loudspeaker arrangement suitable to simulate an ambient noise field in a laboratory-type environment, and a database of realistic ambient noises, part of which is used for testing the noise handling performance with a variety of conditions.

The equalization and calibration procedure for the test set-up are given in detail in [ETSI TS 103 224].

The microphone array set-up for handset and headset applications as described in chapter 5.2 of [ETSI TS 103 224] shall be used.

9.1.2 Delay for communication mode (headset)

9.1.2.1 Requirements

For wireless headsets, the delays in the send and receive directions are preferentially defined as follows.

– The headset delay in the send (uplink) direction, T_S, is the delay between the first acoustic event and the MRP to the last bit of the corresponding speech frame at the UE wireless antenna.

– The headset delay in the receive (downlink) direction, T_R, is the delay between the first bit of a speech frame at the UE wireless antenna and the first acoustic event at the eardrum reference point (DRP) corresponding to that speech frame.

The round trip delay: $T_{rtd} = T_S + T_R$ (the delay in the send direction T_S plus the delay in the receive direction T_R) shall be less than 90 ms.

9.1.2.2 Test

9.1.2.2.1 Test of overall delay in Send

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) The test signal to be used for the measurements is a CSS, as described in [ITU-T P.501]. The test signal consists of the voiced part followed by a pseudo-random noise sequence with a minimum periodicity of 500 ms. The test signal level is −4.7 dBPa, measured at MRP.

3) The delay is calculated using the cross-correlation function between the signal at the output of the digital headset reference interface and the signal at MRP.

4) The measurement is corrected by the delay in the sending direction introduced by the test equipment ($T_{S,sys}$). The sending delay ($T_{S,wdt}$), expressed in milliseconds, is determined from the maximum of the cross-correlation function.

9.1.2.2.2 Test of overall delay in receive

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) The test signal to be used for the measurements is a CSS, as described in [ITU-T P.501]. The test signal consists of the voiced part followed by a pseudo-random noise sequence with a minimum periodicity of 500 ms.

2) The test signal level is −16 dBm0, measured at the input of digital headset reference interface.

3) The delay is calculated using the cross-correlation function between the measured signal at the DRP and the signal at the system simulator.

4) The measurement is corrected by the delay in the sending direction introduced by the test equipment ($T_{R,sys}$). The receiving delay ($T_{R,wdt}$), expressed in milliseconds, is determined from the maximum of the cross-correlation function.

9.1.3 Send loudness rating for communication mode (headset)

9.1.3.1 Requirements

The nominal values of Send loudness rating (SLR) shall be:

$$SLR = 8\text{ dB} \pm 3\text{ dB}$$

9.1.3.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 8.1.1.2. The measured power density spectrum at MRP is used as the reference power density spectrum for determining the sending sensitivity.
3) For NB, the Send sensitivity is calculated from each band of the frequencies given in Table 1 of [ITU-T P.79], bands 4 to 17. For WB, SWB, and FB, the Send sensitivity is calculated from each band of the frequencies given in Table A.2 of [ITU-T P.79], bands 1 to 20. For the calculation, the average measured level at the output of digital headset reference interface for each frequency band is referred to the reference signal.
4) The sensitivity is expressed in decibels relative to 1 V (dBV) per pascal. For NB, the SLR is calculated according to Equation 5-1 of [ITU-T P.79], bands 4 to 17, m = 0.175 and the weighting factors in the send direction according to Table 1 of [ITU-T P.79]. For WB, SWB and FB, the SLR is calculated according to Annex A of [ITU-T P.79].

9.1.4 Receive loudness rating for communication mode (headset)

9.1.4.1 Requirements

The nominal values of the Receive loudness rating (RLR) shall be:

RLR (mon) = 2 dB ± 3 dB for monaural digital headsets.

RLR (bin) = 8 dB ± 3 dB for each earphone of binaural digital headsets.

Where a user-controlled receiving volume control is provided, the RLR shall meet the nominal value for at least one setting of the control. When the control is set to maximum, the RLR shall not be ≤−13 dB.

With the volume control set to the minimum position the RLR shall not be ≥18 dB and shall not be ≥24 dB for a binaural headset.

9.1.4.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.
2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 9.1.1.2. The measured power density spectrum at the digital headset reference interface is used as the reference power density spectrum for determining the receiving sensitivity.
3) For NB, the Send sensitivity is calculated from each band of the frequencies given in Table 1 of [ITU-T P.79], bands 4 to 17. For WB, SWB and FB, the Send sensitivity is calculated from each band of the frequencies given in Table A.2 of [ITU-T P.79], bands 1 to 20. For the calculation, the sound pressure is measured separately at the DRP of the right ear and the left ear for binaural headsets respectively and corrected to the ear reference point (ERP) according to [ITU-T P.57]. The average measured level for each frequency band is referred to the reference signal.
4) The sensitivity is expressed in decibels. For NB, the RLR is calculated according to formula (A-23c) of [ITU T P.79], bands 4 to 17, m = 0.175 and the Receive weighting factors from Table 1 of [ITU-T P.79], without the LE factor. For WB, SWB and FB, the RLR is calculated according to Annex A of [ITU-T P.79], without the LE factor.

9.1.5 Send loudness level

9.1.5.1 Requirements

The nominal value of the Send loudness level (SLL) shall be:

$$SLL = 76 \text{ phon} \pm 4 \text{ phon}$$

9.1.5.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.
2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 9.1.1.2.
3) The SLL is calculated as described in clause 9 of [ITU-T P.700].

9.1.6 Receive loudness level

9.1.6.1 Requirements

The nominal value of Receive loudness level (RLL) for handsets, monaural and binaural or stereo headsets shall be:

$$RLL = 73 \text{ phon} \pm 4 \text{ phon}$$

If a user-controlled Receive volume control is provided, the RLL shall meet the nominal value for at least one setting of the control.

When the control is set to maximum, the RLL shall not be less than (louder than) 88 phon. With the volume control set to the minimum position, the RLL shall not be <57 phon.

9.1.6.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.
2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 9.1.1.2.
3) The RLL is calculated as described in clause 8 of [ITU-T P.700].

9.1.7 Send sensitivity and frequency response for communication mode (headset)

9.1.7.1 Requirements

The Send sensitivity and frequency response is measured from the MRP to the output of the digital headset reference interface. In general, a flat sending frequency response is recommended.

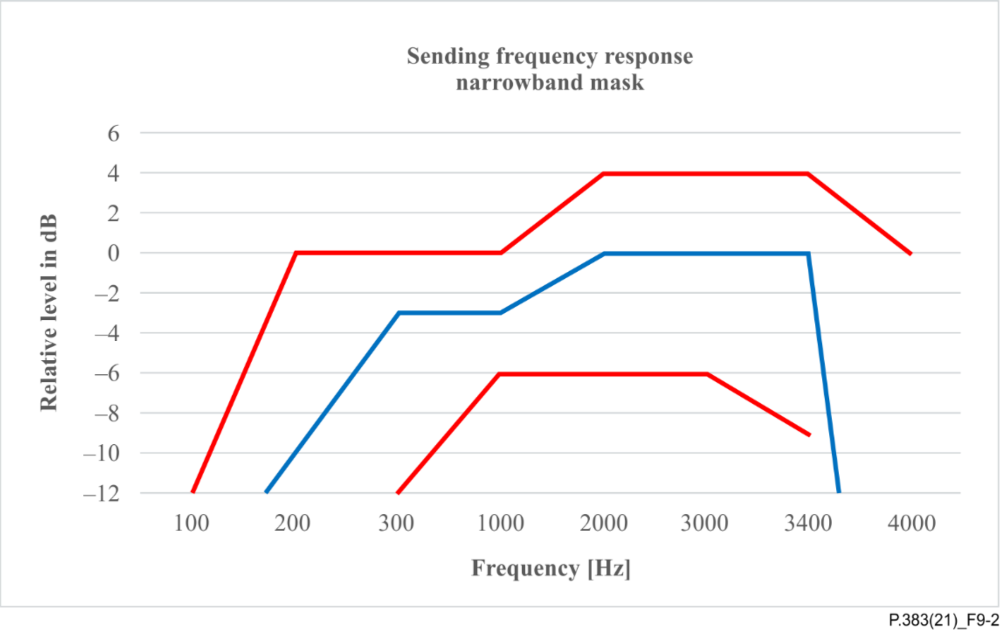
For digital headsets only capable of transmitting NB speech, the tolerance mask for the sending sensitivity frequency response is shown in Table 9-1. See Figure 9-2. For digital headsets capable of transmitting WB speech, the tolerance mask shown in Table 9-2 shall be met. See Figure 9-3. For digital headsets capable of transmitting SWB and FB speech, an additional tolerance mask shown in Table 9-3 should be met when measuring in one-third octave bands, besides meeting the tolerance mask of WB speech in Table 9-2. See Figure 9-4.

Table 9-1 – Tolerance mask for narrowband sending frequency response

Frequency (Hz)	Upper limit	Lower limit
100	–12	
200	0	
300	0	–12
1 000	0	–6
2 000	4	–6
3 000	4	–6

Table 9-1 – Tolerance mask for narrowband sending frequency response

Frequency (Hz)	Upper limit	Lower limit
3 400	4	–9
4 000	0	
NOTE 1 – All sensitivity values are expressed in decibels on an arbitrary scale. NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (decibel) scale.		



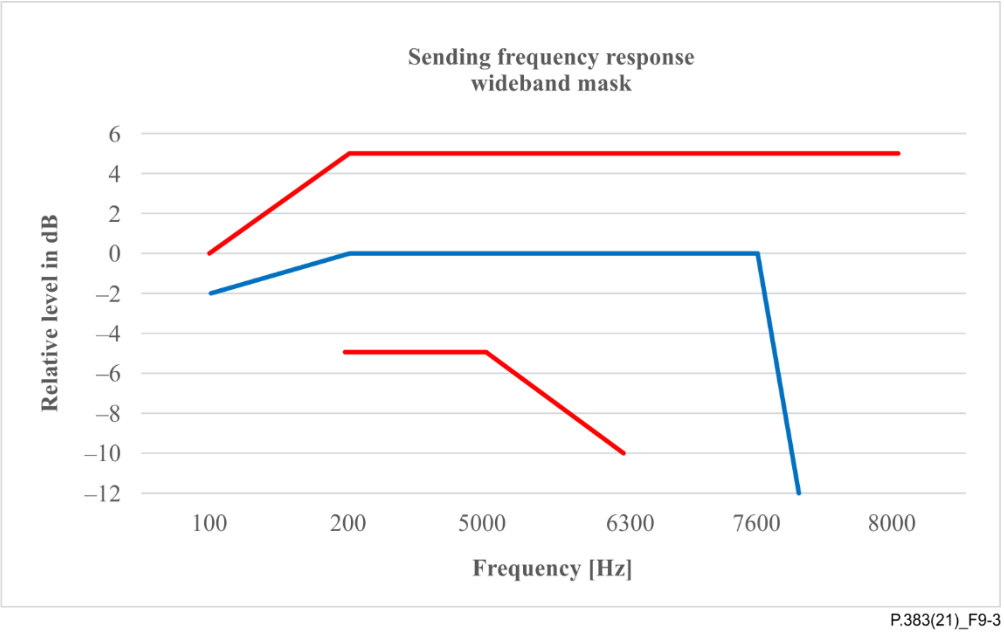
red: upper and lower tolerance masks; blue: nominal sending frequency response

Figure 9-2 – Tolerance mask for narrowband sending frequency response

Table 9-2 – Tolerance mask for wideband sending frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**
100	0	
200	5	–5
5 000	5	–5
6 300	5	–10
8 000	5	

NOTE 1 – All sensitivity values are expressed in dB on an arbitrary scale.
NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.

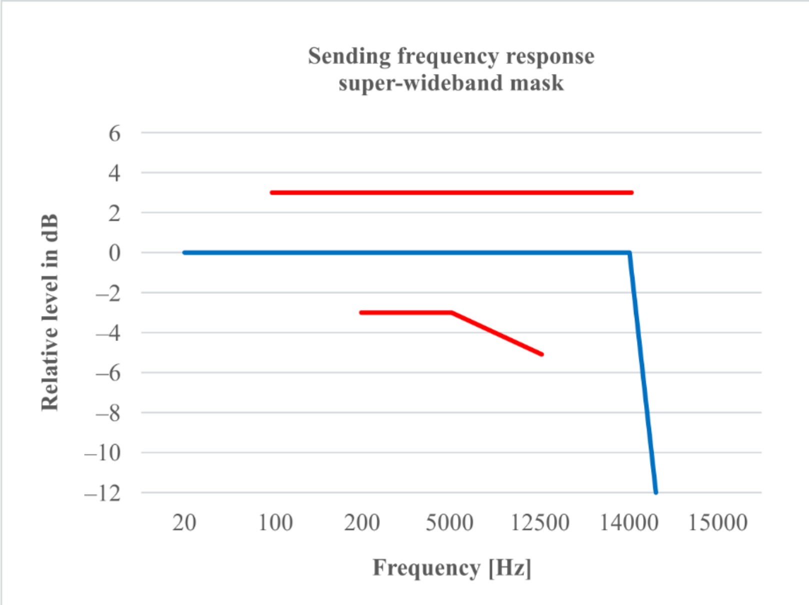


red: upper and lower tolerance masks; blue: nominal sending frequency response

Figure 9-3 – Tolerance mask for wideband sending frequency response

Table 9-3 – Additional tolerance mask for the super-wideband and fullband sending frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**
100	[3]	
200	[3]	[–3]
5000	[3]	[–3]
12500	[3]	[–5]
16000	[3]	
NOTE 1 – All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.		



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red: upper and lower tolerance masks; blue: nominal sending frequency response

Figure 9-4 – Tolerance mask for super-wideband sending frequency response

9.1.7.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.
2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 9.1.1.2. The measured power density spectrum at MRP is used as the reference power density spectrum for determining the sending sensitivity.
3) The sending sensitivity is determined in one-12th octave bands, as given in [IEC 61260-1] for frequencies from 100 Hz to 4 kHz inclusive, measured at the POI. For WB, measurements shall be made for frequencies from 100 Hz to 8 kHz inclusive for SWB and FB, measurements shall be made at both one-third octave and one-12th octave bands as given in [IEC 61260-1] for frequencies from 100 Hz to 16 kHz inclusive. For the calculation, the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.
4) The sensitivity is expressed in dBV/pa.

9.1.8 Receive sensitivity versus frequency response for communication mode (headset)

9.1.8.1 Requirements

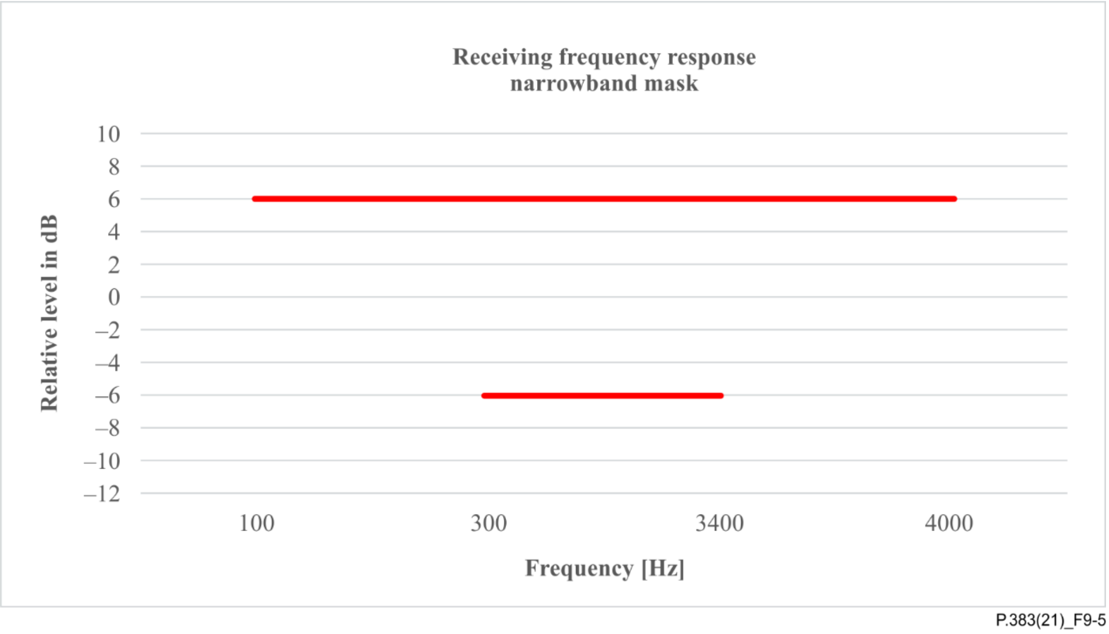
The receive sensitivity versus frequency response is measured from the digital headset reference interface to the DRP with diffuse field correction. In general, a flat receiving frequency response is recommended.

For digital headsets only capable of transmitting NB speech, the tolerance mask for the receiving sensitivity frequency response is shown in Table 9-4. See Figure 9-5. For digital headsets capable of transmitting WB speech, the tolerance mask shown in Table 9-5 shall be met. See Figure 9-6. For digital headsets capable of transmitting SWB and FB speech, an additional tolerance mask shown in Table 9-6 should be met when measuring in one-third octave bands, besides meeting the tolerance mask of WB speech in Table 9-6. See Figure 9-7.

Table 9-4 – Tolerance mask for narrowband receiving frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**
100	6	
300	6	–6
3 400	6	–6
4 000	6	

NOTE 1 – All sensitivity values are expressed in dB on an arbitrary scale.
NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.



red: upper and lower tolerance mask

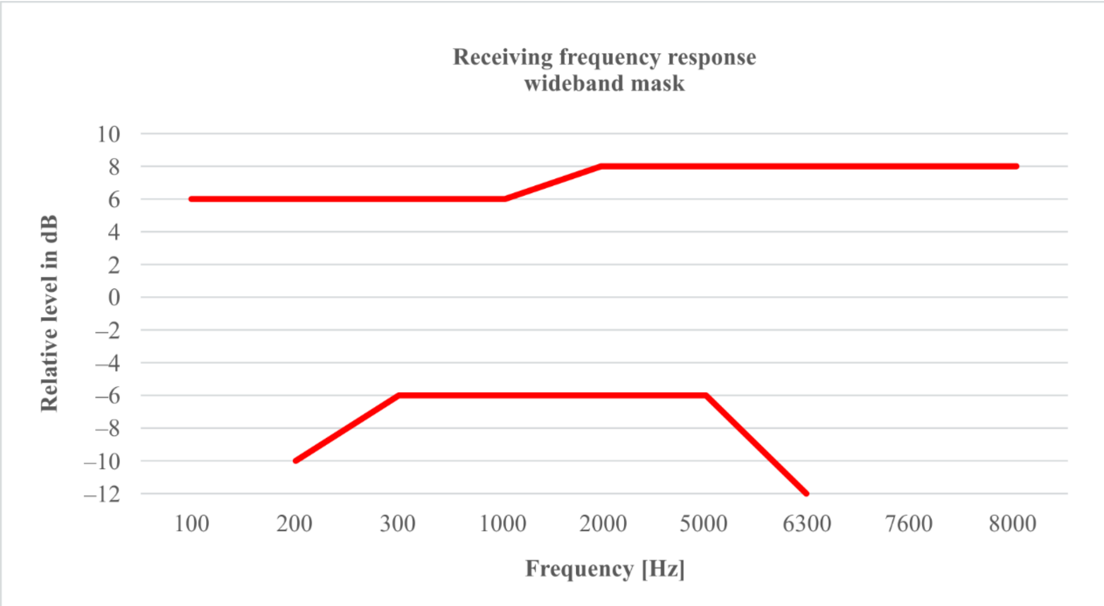
Figure 9-5 – Tolerance mask for narrowband receiving frequency response

Table 9-5 – Tolerance mask for wideband receiving frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**
100	6	
200	6	–10
300	6	–6
1 000	6	–6
2 000	8	–6
5 000	8	–6
6 300	8	–12
8 000	8	

NOTE 1 – All sensitivity values are expressed in dB on an arbitrary scale.

NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.

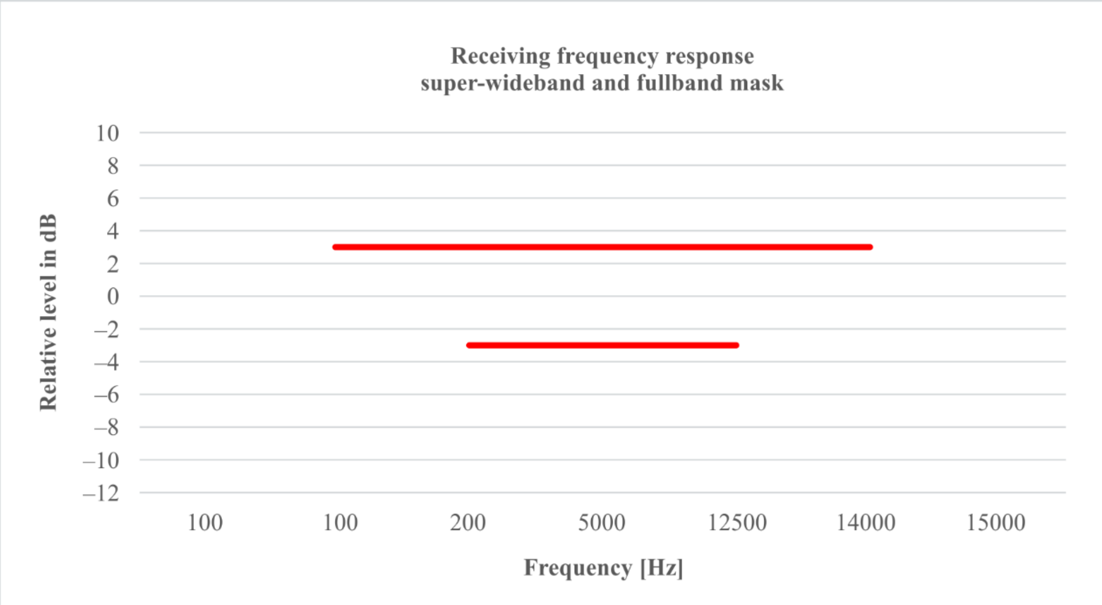


red: upper and lower tolerance mask

Figure 9-6 – Tolerance mask for wideband receiving frequency response

Table 9-6 – Additional tolerance mask for the super-wideband and fullband receiving frequency response

Frequency (Hz)	**Upper limit**	**Lower limit**
100	[3]	
200	[3]	[–3]
5000	[3]	[–3]
12500	[3]	[–5]
16000	[3]	
NOTE 1 – All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2 – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.		



red: upper and lower tolerance mask

Figure 9-7 – Tolerance mask for super-wideband receiving frequency response

9.1.8.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 8.1.1.2. The measured power density spectrum at the sending input of the digital reference headset interface is used as the reference power density spectrum for determining the sending sensitivity.

3) The sending sensitivity is determined in one-12th octave bands, as given in [IEC 61260-1] for frequencies from 100 Hz to 4 kHz inclusive, measured at the POI. For WB, measurements shall be made for frequencies from 100 Hz to 8 kHz inclusive. For SWB and FB, measurements shall be made at both one-third octave and one-12th octave bands as given in [IEC 61260-1] for frequencies from 100 Hz to 16 kHz inclusive. For the calculation, the sound pressure is measured separately at the DRP of the right ear and the left ear for binaural headsets respectively. The averaged measured level at the DRP with diffuse field correction for each frequency band takes as reference the averaged reference test signal level measured in each frequency band.

4) The sensitivity is expressed in dBPa/V.

9.1.9 Sidetone loss for communication mode (headset)

9.1.9.1 Requirements

The talker sidetone masking rating (STMR) shall be ≥15 dB and should be ≤23 dB for the nominal setting of the volume control. For all other positions of the volume control, the STMR shall be ≥10 dB.

NOTE 1 – Where a user-controlled receiving volume control is provided, it is recommended that the sidetone loss be independent of the volume control setting.

NOTE 2 – If the human air-conducted sidetone paths are obstructed (e.g., binaural insert type headsets), it is important to provide a sidetone path.

NOTE 3 – If the STMR is below the lower limit, also when the electrical sidetone path has been disabled, the result shall not be regarded as a failure.

9.1.9.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) The test signal used for the measurements is the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the active speech level according to [ITU-T P.56], adjusted as described in clause 9.1.1.2. The measured power density spectrum at MRP is used as the reference power density spectrum for determining the sending sensitivity.

3) Measurements shall be made at one-12th octave bands as given in [IEC 61260-1] for frequencies from 100 Hz to 8 kHz inclusive. The averaged measured level at DRP of each frequency band is referred to the averaged test signal level measured in each frequency band.

4) The sidetone path loss (L_{meST}), as expressed in decibels, is calculated from each one-third octave band according to Table B.1 of [ITU-T P.79], bands 1 to 20. The STMR, expressed in decibels, is calculated from formula B-4 of [ITU-T P.79], using $m = 0.225$ and the weighting factors in Table B.2 (unsealed condition) of [ITU-T P.79]. No leakage correction (LE) is applied. DRP-ERP correction is used.

5) If the STMR is below the limit, the measurement shall be repeated with the electrical sidetone path disabled and both sets of results shall be reported. If the STMR is below the limit, also with the electrical sidetone path disabled, the result shall not be regarded as a failure. Disconnecting the headset connection normally disables the electrical sidetone.

9.1.10 Sidetone delay for communication mode (headset)

9.1.10.1 Requirements

The maximum sidetone delay shall be ≤5 ms.

NOTE – The sidetone delay is measured in an echo-free set-up, and only applicable where the level of the electrical sidetone is sufficiently high to be measured. While the STMR value may indicate the presence of sidetone it should be ensured that this is not primarily due to the acoustic or mechanical sidetone path when interpreting sidetone delay results.

9.1.10.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) The test signal is a CSS complying with [ITU-T P.501] using a pseudo-noise (PN) sequence with a length of 4 096 points (for the 48 kHz sampling rate) which equals the period T. The duration of the complete test signal is as specified in [ITU-T P.501]. The test signal level is –4.7 dBPa at the MRP.

3) The cross-correlation function $\Phi xy(\tau)$ between the input signal $S_x(t)$ generated by the test system in Send and the output signal $S_y(t)$ measured at the DRP is calculated in the time domain:

$$\Phi_{xy}(\tau) = \frac{1}{T}\int_{-T/2}^{T/2} S_x(t) \cdot S_y(t+\tau)\mathrm{d}t \qquad (9\text{-}1)$$

The measurement window T shall be identical to the time period T of the test signal, the measurement window is positioned to the PN sequence of the test signal.

4) The sidetone delay is calculated from the envelope $E(\tau)$ of the cross-correlation function $\Phi xy(\tau)$. The first maximum of the envelope function occurs in correspondence with the direct sound produced by the artificial mouth, the second occurs with a possible delayed sidetone signal. The difference between the two maxima corresponds to the sidetone delay. The envelope $E(\tau)$ is calculated by the Hilbert transformation $H\{xy(\tau)\}$ of the cross-correlation:

$$H[xy(\tau)] = \int_{-\infty}^{+\infty} \frac{\Phi_{xy}(u)}{\pi(\tau-u)} \mathrm{d}u \tag{9-2}$$

$$E(\tau) = \sqrt{[\Phi_{xy}(\tau)]^2 + \{H[xy(\tau)]\}^2} \tag{9-3}$$

It is assumed that the measured sidetone delay is < T/2.

9.1.11 Noise in Send for communication mode (headset)

9.1.11.1 Requirements

The maximum idle channel noise measured in the sending direction shall be less than −64 dBm0(A).

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

9.1.11.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.
2) The noise is measured at the output of the digital headset reference interface in the frequency range between 100 Hz and 4 kHz for NB, between 100 Hz and 8 kHz for WB, between 100 Hz and 16 kHz for SWB and between 100 Hz and 20 kHz for FB. The length of the time window is 1 s, which is the averaging time for the idle channel noise. The power density spectrum of the noise signal is determined using the FFT (8 000 samples at 48 kHz sampling rate or equivalent). A Hann window is used.
3) The noise is determined with A-weighting.
4) Spectral peaks are measured in the frequency domain. The frequency spectrum of the idle channel noise is measured by a spectral analysis having a noise bandwidth of 8.79 Hz (determined using FFT 8 000 samples at 48 kHz sampling rate with a Hann window or equivalent). A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) one-third octave wide across the idle noise channel spectrum stated in decibels (linear average in decibels of all FFT bins in the range from $2^{-1/6}f$ to $2^{+1/6}f$). Peaks in the idle channel noise spectrum are compared with a smoothed average idle channel noise spectrum from 100 Hz to 3.4 kHz in NB, from 100 Hz to 6.3 kHz in WB, from 100 Hz to 13 kHz in SWB and from 100 Hz to 18 kHz in FB.

9.1.12 Noise in Receive for communication mode (headset)

9.1.12.1 Requirements

The maximum idle channel noise measured in the receive direction shall be less than −54 dBPa(A) for monaural headsets and less than −60 dBPa (A) for binaural headsets.

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

9.1.12.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.
2) The noise is measured at the DRP with diffuse field correction between 100 Hz and 10 kHz for NB and WB, between 100 Hz and 16 kHz for SWB and FB. The length of the time window is 1 s, which is the averaging time for the idle channel noise. The power density spectrum of the noise signal is determined using the FFT (8 000 samples at 48 kHz sampling rate or equivalent). A Hann window is used.
3) The noise is A-weighted.
4) Spectral peaks are measured in the frequency domain. The frequency spectrum of the idle channel noise is measured by a spectral analysis having a noise bandwidth of 8.79 Hz (determined using FFT 8 000 samples at 48 kHz sampling rate with a Hann window or equivalent). A smoothed average idle channel noise spectrum is calculated by a moving

average (arithmetic mean) one-third octave wide across the idle noise channel spectrum stated in decibels (linear average in decibels of all FFT bins in the range from $2^{-1/6}f$ to $2^{+1/6}f$). Peaks in the idle channel noise spectrum are compared with a smoothed average idle channel noise spectrum from 100 Hz to 6.3 kHz in NB and WB, from 100 Hz to 13 kHz in SWB and FB.

9.1.13 Send distortion for communication mode (headset)

9.1.13.1 Requirements

The distortion in sending direction shall be measured between the MRP and the output of the digital headset reference interface. The ratio of signal to total distortion power measured shall be above the limits given in Table 9-7.

Table 9-7 –Limits for signal to total distortion ratio in sending

Frequency (Hz)	**Sending level** (dBPa at the MRP)	**Sending ratio** (dB)
315	–4,7	28
408	–4,7	32
510	–4,7	32
816	–4,7	32
1 020	5	30
	0	35
	–4.7	35
	–10	33
	–15	30
	–20	27
NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.		

9.1.13.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) The test signal used is a sine-wave signal with frequencies specified in Table 8-7. The sine-wave signal level is calibrated to –4.7 dBPa at the MRP for all frequencies, except for the sine-wave with a frequency 1 020 Hz which is applied at the following levels at the MRP: 5; 0; –4.7; –10; –15; –20 dBPa. The test signals must be applied in this sequence, i.e., from high levels down to low levels.

3) To guarantee a reliable activated state of the digital headset, a sequence of four CSSs, according to [ITU-T P.501], is sent to the headset before the actual test signal. The activation signal level is –4.7 dBPa, measured at the MRP. The activation signal level is averaged over the total length of the activation signal. The test signal is inserted immediately after the activation sequence, after the voiced sound of the last CSS burst (instead of the PN sequence). The duration of the sine-wave signal is recommended to be 360 ms, and the manufacturer shall be allowed to request tone lengths up to 1 s. The measured part of the signal is 170.667 ms (which equals 2 × 4 096 samples in a 48 kHz sample rate test system). The times are selected to be relatively short to reduce the risk of the test tone being treated as a stationary signal.

3) The ratio of the signal to total distortion power at the signal output of the digital headset reference interface is measured with the decibels relative to 1 V psophometric (dBVp) noise

weighting. The psophometric filter shall be normalized (0 dB gain) at 800 Hz as specified in [ITU-T O.41]. The weighting function shall be applied to the total distortion component only (not to the signal component).

4) For measurement of the total distortion component an octave-wide band-stop filter shall be applied to the signal to suppress the sine-wave signal and associated coding artefacts. The filter shall have a lower passband ending at 0.707 1 × f_s, and an upper passband starting at 1.414 2 × f_s, where f_s is the frequency of the sine-wave signal. The passband ripple of the filter shall be ≤2 dB. The attenuation of the band-stop filters at the sine-wave frequency shall be ≥60 dB. Alternatively, the described characteristics can be implemented by an appropriate weighting on the spectrum obtained from an FFT. The total distortion component is defined as the measured signal within the frequency range 200 Hz to 4 kHz for NB and 100 Hz to 6 kHz for WB, SWB, and FB, after applying psophometric and stop filters.

9.1.14 Receive distortion for communication mode (headset)

9.1.14.1 Requirements

The distortion in the receiving direction shall be measured between the input of the digital headset reference interface and DRP. The ratio of signal to total distortion power measured shall be above the limits given in Table 9-8.

Table 9-8 –Limits for signal to total distortion ratio in receiving

Frequency (Hz)	**Receiving level at the reference interface** (dBm0)	**Receiving ratio at nominal volume setting** (dB)
315	–16	20
408	–16	28
510	–16	28
816	–16	28
1 020	0	25,5
	–3	31,2
	–10	33,5
	–16	33,5
	–20	33
	–30	30,5
NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (decibel) scale.		

9.1.14.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) The test signal used is a sine-wave signal with frequencies specified in Table 9-8. The sine-wave signal level is calibrated to –16 dBm0 at the input of digital headset reference interface for all frequencies, except for the sine-wave with a frequency 1 020 Hz, which shall be applied at the following levels: 0; 3; 10; 16; 20; 30; 40; 45 dBm0. The test signals must be applied in this sequence, i.e., from high down to low levels.

3) In order to guarantee a reliable activated state of the digital headset, a sequence of four CSSs, according to [ITU-T P.501], is sent to the headset before the actual test signal. The activation signal level is −16 dBm0. The activation signal level is averaged over the total length of the activation signal. The test signal is inserted immediately after the activation sequence, after

the voiced sound of the last CSS burst (instead of the PN sequence). The duration of the sine-wave signal is recommended to be 360 ms, and the manufacturer shall be allowed to request tone lengths up to 1 s. The measured part of the signal shall be 170.667 ms (which equals 2 × 4 096 samples in a 48 kHz sample rate test system). The times are selected to be relatively short to reduce the risk of the test tone being treated as a stationary signal.

3) The ratio of the signal to total distortion power at the DRP with diffuse field correction shall be measured with the (dBVp) psophometric noise weighting. The psophometric filter shall be normalized (0 dB gain) at 800 Hz as specified in [ITU-T O.41]. The weighting function shall be applied to the total distortion component only (not to the signal component).

4) For measurement of the total distortion component, an octave-wide band-stop filter shall be applied to the signal to suppress the sine-wave signal and associated coding artefacts. The filter shall have a lower passband ending at 0.707 1 × f_s, and an upper passband starting at 1.414 2 × f_s. The passband ripple of the filter shall be ≤2 dB. The attenuation of the band-stop filters at the sine-wave frequency shall be ≥60 dB. Alternatively, the described characteristics can be implemented by an appropriate weighting on the spectrum obtained from an FFT. The total distortion component is defined as the measured signal within the frequency range 200 Hz to 4 kHz for NB and 100 Hz to 6 kHz for WB, SWB and FB, after applying psophometric and stop filters.

9.1.15 Noise cancellation test in Send for communication mode (headset)

9.1.15.1 Requirements

The objective of this test is to check the performance of the noise cancellation in Send.

When testing through the objective methodology, the digital headset shall comply with the following requirements.

For NB digital headsets, where LQOn is listening-only quality objective, narrowband; N is noise; S is speech; and G is global:

N-MOS-LQOn	Average N-MOS-LQOn ≥ 3.0
S-MOS-LQOn	Average S-MOS-LQOn ≥ 3.5
G-MOS-LQOn	No requirement.

For WB digital headsets, where LQOw is listening-only quality objective, wideband:

N-MOS-LQOw	Average N-MOS-LQOw ≥ 3.0
S-MOS-LQOw	Average S-MOS-LQOw ≥ 3.5
G-MOS-LQOw	No requirement.

For SWB and FB digital headsets, where LQOf is listening quality objective, fullband:

N-MOS-LQOf	Average N-MOS-LQOf ≥ 3.3
S-MOS-LQOf	Average S-MOS-LQOf ≥ 3.9
G-MOS-LQOf	No requirement.

9.1.15.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) Before starting the measurements, a proper conditioning sequence shall be used. For NB and WB, the conditioning sequence shall be comprised of the four additional sentences 1-4 described in [ETSI TS 103 106], applied to the beginning of the 16-sentence test sequence. For SWB and FB, the conditioning sequence shall be comprised of the four additional sentences 1-4 described in [ETSI TS 103 281], applied to the beginning of the 16-sentence test sequence.

3) For NB and WB, the Send speech signal consists of the 16 sentences of speech as described in [ETSI TS 103 106]. The test signal level is –1.7 dBPa at the MRP, measured as the active speech level according to [ITU-T P.56]. Three signals are required for the tests as follows.

- The clean speech signal is used as the undisturbed reference (see [ETSI TS 103 106], [b-ETSI EG 202 396-3]).
- The speech plus undisturbed background noise signal is recorded at the headset's microphone position using an omnidirectional measurement microphone with a linear frequency response between 50 Hz and 12 kHz.
- The Send signal is recorded at the output of digital headset reference interface.

  For SWB and FB, the Send speech signal consists of the 16 sentences of speech as described in [ETSI TS 103 281]. The test signal level is –1.7 dBPa at the MRP, measured as active speech level per [ITU-T P.56]. Two signals are required for the tests as follows:

  - The clean speech signal is used as the undisturbed reference (see [ETSI TS 103 281];
  - The Send signal is recorded at the output of digital headset reference interface.

4) For NB and WB, the mean opinion score N-MOS-LQOn, S-MOS-LQOn and G-MOS-LQOn are calculated as described in [ETSI TS 103 106] on a per sentence basis and averaged over all 16 sentences. The results shall be reported as average and standard deviation.

   For SWB and FB, N-MOS-LQOfb, S-MOS-LQOfb, and G-MOS-LQOfb are calculated for one of the two objective predictor models described in [ETSI TS 103 281] on a per sentence basis and averaged over all 16 sentences. Model A is used.

   N-MOS-LQOfb, S-MOS-LQOfb, and G-MOS-LQOfb are calculated according to model A on a per sentence basis and averaged over all 16 sentences. The final results are derived as follows:

   - S-MOS-LQOfb = S-MOS-LQOfb_modelA
   - N-MOS-LQOfb = 1.438*N-MOS-LQOfb_modelA – 1.959
   - G-MOS-LQOfb = G-MOS-LQOfb_modelA

5) The measurement is repeated for each ambient noise condition described in Table 9-9. The average of the results derived from all ambient noise types is calculated.

Table 9-9 – Noises used for background noise simulation

Name	**Description**	**Length** (s)	**Handset levels** (dB)
Pub noise (Pub)	HATS and microphone array in a pub	30	1: 77,2 2: 76,6 3: 75,7 4: 76,0 5: 76,0 6: 76,3 7: 76,0 8: 76,4
Roadnoise (Roadnoise)	HATS and microphone array standing outside near a road	30	1: 72,8 2: 71,6 3: 72,0 4: 72,9 5: 72,2 6: 73,1 7: 73,0 8: 73,8
Departure platform (TrainStation)	HATS and microphone array on the departure platform of a train station	30	1: 78,9 2: 78,8 3: 79,1 4: 80,0 5: 79,4 6: 79,6 7: 78,8 8: 80,1
Full-size car 130 km/h (FullSizeCar_130)	HATS and microphone array at co-drivers' position	30	1: 68,5 2: 68,3 3: 68,8 4: 69,5 5: 69,9 6: 70,5

Table 9-9 – Noises used for background noise simulation

Name	**Description**	**Length** (s)	**Handset levels** (dB)
			7: 70,8 8: 71,9
Sales counter (SalesCounter)	HATS and microphone array in a supermarket	30	1: 66,6 2: 66,1 3: 65,7 4: 66,5 5: 66,3 6: 66,8 7: 66,6 8: 67,1
Cafeteria (Cafeteria)	HATS and microphone array inside a cafeteria	30	1: 70,0 2: 70,0 3: 70,1 4: 70,7 5: 70,5 6: 70,8 7: 70,6 8: 71,0
Callcentre 2 (Callcenter)	HATS and microphone array in business office	30	1: 60,2 2: 60,0 3: 60,1 4: 60,8 5: 60,2 6: 60,6 7: 60,2 8: 60,7

9.1.16 One-way speech quality in Send for communication mode (headset)

For further study.

9.1.17 One-way speech quality in receive for communication mode (headset)

For further study.

9.1.18 Terminal coupling loss for communication mode (headset)

9.1.18.1 Requirements

The headset terminal coupling loss (HTCL for NB, HTCL for WB, SWB and FB) is measured from the input of the digital headset reference interface to the output of the digital headset reference interface.

The HTCL provided by the headset signal processing shall be ≥55 dB at the nominal volume control setting. The HTCL shall also be ≥46 dB at the maximum setting of volume control.

9.1.18.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) The test signal is the compressed real speech signal described in clause 7.3.3 of [ITU-T P.501]. The test signal level shall be −10 dBm0 and be band limited as described in clause 9.1.1.2.

3) HTCL is calculated according to clause B.4 of [ITU-T G.122] (trapezoidal rule). For NB, the frequency range used is from 300 Hz to 3 400 Hz. For WB, SWB and FB, the frequency range used is from 100 Hz to 8.0 kHz.

   HTCL is calculated as unweighted EL from 100 Hz to 8.0 kHz. For the calculation, the averaged test signal level at each frequency band takes as reference the averaged measured echo signal level in each frequency band. For the measurement, a time window has to be applied that is adapted to the duration of the actual test signal. The EL is calculated from the following equations.

$$L_e = C - 10\log_{10}\sum_{i=1}^{N}(A_i + A_{i-1})(\log_{10} f_i - \log_{10} f_{i-1})$$

and

$$C = 10\log_{10} 2\,(\log_{10} f_N - \log_{10} f_0)$$

where:

A_0: is the output/input power ratio at frequency $f_0 = 100$ Hz

A_1: the output/input power ratio at frequency f_i;

A_N: the output/input power ratio at frequency $f_N = 8.0$ kHz.

The form of the first equation in the preceding entry is generalized from that specified in clause B.4 of [ITU-T G.122] to calculate EL based on tabulated data, which allows the calculation of EL within any frequency range between f_0 and f_N.

4) For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The first 17 s of the test signal (six sentences) are discarded from the analysis to allow for convergence of the acoustic EC. The analysis is performed over the remaining length of the test sequence (last six sentences).

9.1.19 Temporal echo effects for communication mode (headset)

9.1.19.1 Requirements

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk. When measuring using the CSS, the measured echo attenuation during single talk should not decrease by more than 6 dB from the maximum measured during the test. When measuring using the British-English single talk sequence the echo level variation shall be less than 6 dB.

NOTE 1 – The echo path is kept constant during this test, and the test should begin 5 s after the initial application of a reference signal such that a steady state converged condition is achieved.

NOTE 2 – The analysis is conducted only during the active signal part.

9.1.19.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.

2) The test signal consists of a periodically repeated CSS according to [ITU-T P.501], with an average level of –5 dBm0 as well as an average level of –25 dBm0. The echo signal is analysed during a period of at least 2.8 s, which represents eight periods of the CSS. The integration time for the level analysis shall be 35 ms, the analysis is referred to the level analysis of the reference signal. In addition, the test is repeated with the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. One sequence of male and one sequence of female voice is used. The average test signal level is –16 dBm0. The echo signal is analysed during the complete test signal. The integration time for the level analysis shall be 35 ms, the analysis is referred to the level analysis of the reference signal.

3) When using the CSS, the measurement result is displayed as attenuation versus time. The exact synchronization between input and output signal must be guaranteed.

4) When using the speech signal the measurement is displayed as level versus time.

   NOTE – When testing using CSSs, the analysis is conducted only during the active signal part, the pauses between the CSSs are not analysed. The analysis time is reduced by the integration time of the level analysis (35 ms).

9.1.20 Double talk performance for communication mode (headset)

NOTE – Before starting the double talk tests, the test laboratory should ensure that the EC is fully converged. This can be done by an appropriate training sequence.

During double talk, the speech is mainly determined by two parameters: impairment caused by echo during double talk; and level variation between single and double talk (attenuation range).

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 that do not allow double talk in any case should provide a good echo attenuation that is realized by a high attenuation range in this case.

The most important parameters determining the speech quality during double talk are (see [ITU-T P.340] and [ITU-T P.502]):

– attenuation range in the send direction during double talk ($A_{H,S,dt}$);

– attenuation range in the receive direction during double talk ($A_{H,R,dt}$);

– echo attenuation during double talk.

9.1.20.1 Attenuation range in the send direction during double talk

9.1.20.1.1 Requirements

Based on the variation of the level in the send direction during double talk $A_{H,S,dt}$, the behaviour of terminals can be classified according to Table 9-10.

Table 9-10 – Categorization of double talk capability according to [ITU-T P.340]

Category (according to [ITU-T P.340])	**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 requirements apply for nominal and maximum setting of the receive volume control.

The requirements apply for nominal signal levels in the 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. Furthermore, the test is conducted with nominal levels but with maximum setting of the volume control.

NOTE – If the maximum setting of the volume control is chosen such that non-linearities occur in the echo path, the double talk performance decreases.

In general, Table 9-10 provides a quality classification of terminals regarding double talk performance. However, concerning the overall quality, this does not mean that a terminal that is in category 1 based on the double talk performance is of high quality as well.

9.1.20.1.2 Test

The test signal to determine the attenuation range during double talk is shown in Figure 9-8. The test signal to determine the attenuation range during double talk is the double talk speech sequence as specified in clause 7.3.5 of [ITU-T P.501]. The competing speaker is always inserted as the double talk sequence sdt(t) in Send and is used for analysis.

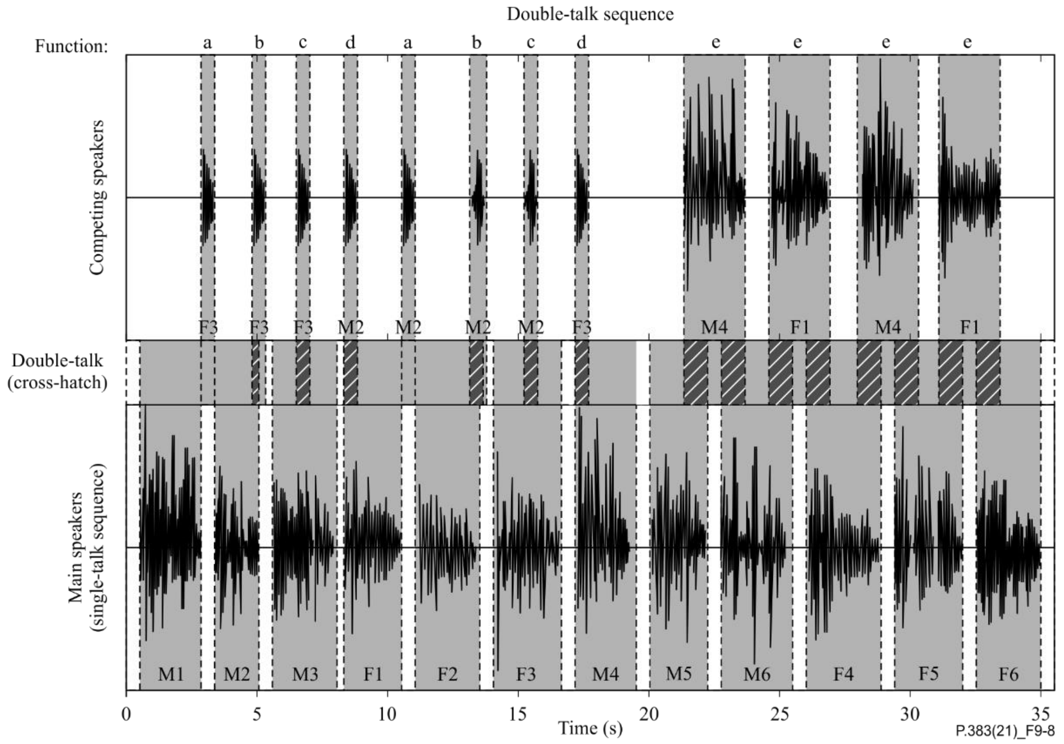


Figure 9-8 – Double talk test sequence with overlapping speech sequences in send and receive directions

The test signals are synchronized in time at the acoustical interface. The delay of the test set-up shall be constant during the measurement.

The settings for the test signals are given in Table 9-11:

Table 9-11 – Signal levels of the double talk sequences

	Receive direction	Send direction
Average signal level	–16 dBm0	–4.7 dBPa

The tests are repeated with a maximum volume control setting in the receive direction.

1) The test set-up is according to clause 9.1.1, Figure 9-1. Before the actual test, a training sequence for the EC consisting of the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501] with a level of –16 dBm0 is applied to the electrical reference point.
2) When determining the attenuation range in the send direction, the signal measured at the electrical reference point is referred to the test signal inserted.
3) The attenuation range during double talk is determined as described in [ITU-T P.502]. The double talk performance is analysed for the sequence of words and the sequence of sentences produced by the competing speaker. The requirement must be met for the sequence of words and the sequence of sentences produced by the competing speaker.
4) The test is repeated for all level combinations as specified in the requirements.

9.1.20.2 Attenuation range in the receive direction during double talk: $A_{H,R,dt}$

9.1.20.2.1 Requirements

Based on the level variation in the receive direction during double talk, $A_{H,R,dt}$, the behaviour of the free terminal can be classified according to Table 9-12.

Table 9-12 – Categorization of double talk capability according to [ITU-T P.340]

Category (according to [ITU-T P.340])	**1**	**2a**	**2b**	**2c**	**3**
	Full duplex capability	*Partial duplex capability*			*No duplex capability*
$A_{\mathrm{H,R,dt}}$ [dB]	≤3	≤5	≤8	≤10	>10

The tests are repeated with maximum volume control setting in the receive direction.

In general, Table 9-12 provides a quality classification of terminals regarding double talk performance. However, concerning the overall quality, this does not mean that a terminal that is in category 1 based on the double talk performance is of high quality as well.

9.1.20.2.2 Test

The test signal to determine the attenuation range during double talk is shown in Figure 9-9. The test signal to determine the attenuation range during double talk is the double talk speech sequence as specified in clause 7.3.5 of [ITU-T P.501]. The competing speaker is always inserted as the double talk sequence sdt(t) in Receive and is used for analysis. The test signals are synchronized in time at the acoustical interface. The delay of the test set-up shall be constant during the measurement.

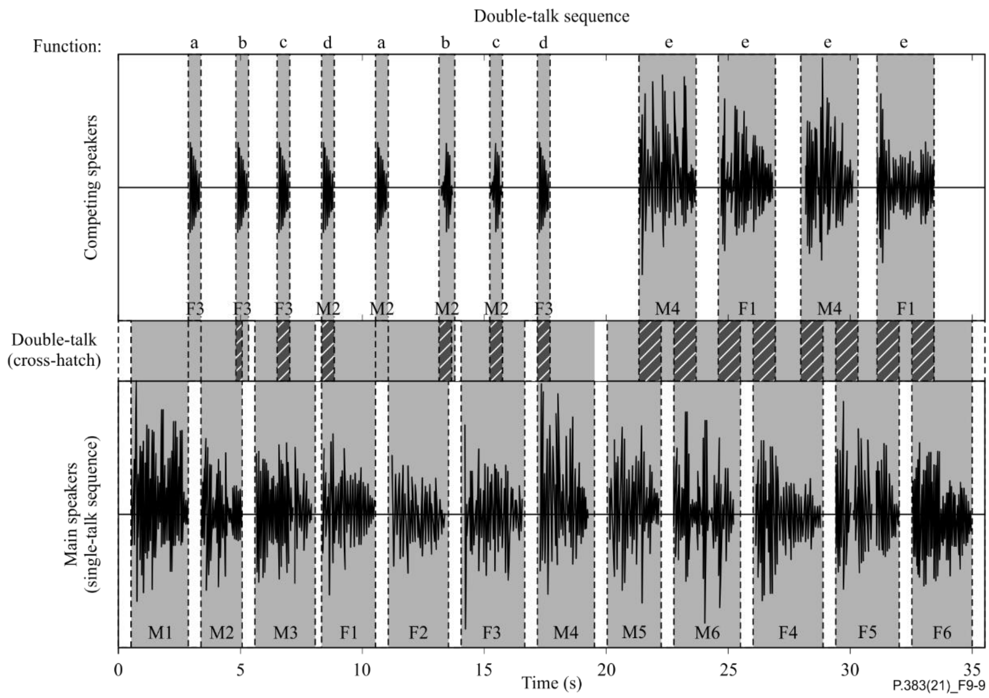


Figure 9-9 – Double talk test sequence with overlapping speech sequences in receive and send directions

The settings for the test signals are given in Table 9-13:

Table 9-13 – Signal levels of the double talk sequences

	Receive direction	**Send direction**
Average signal level	–16 dBm0	–4.7 dBPa

The tests are repeated with the maximum volume control setting in the receive direction.

1) The test set-up is according to clause 9.1.1, Figure 9-1.
2) When determining the attenuation range in the receive direction, the signal measured at the DRP is referred to the test signal inserted.
3) The attenuation range during double talk is determined as described in [ITU-T P.502]. The double talk performance is analysed for the sequence of words and the sequence of sentences produced by the competing speaker. The requirement must be met for the sequence of words and the sequence of sentences produced by the competing speaker.
4) The test is repeated for all level combinations as specified in the requirements.

9.1.20.3 Detection of echo components during double talk

9.1.20.3.1 Requirements

The echo attenuation during double talk is based on the parameter talker echo loudness rating under double talk conditions (TELRdt). It is assumed that the terminal at the opposite end of the connection provides a nominal loudness rating (SLR + RLR = 10 dB). EL is the echo suppression provided by the headset measured at the electrical reference point. Under these conditions, the requirements given in Table 9-14 are applicable (more information can be found in Annex A of [ITU-T P.340]).

Table 9-14 – Categorization of double talk capability according to [ITU-T P.340]

Category (according to [ITU-T P.340])	**1**	**2a**	**2b**	**2c**	**3**
	Full duplex capability	*Partial duplex capability*			*No duplex capability*
Echo loss [dB]	≥27	≥23	≥17	≥11	<11

9.1.20.3.2 Test

1) The test set-up is according to clause 9.1.1, Figure 9-1.
2) The double talk signal consists of a sequence of orthogonal signals that are realized by voice-like modulated sine waves spectrally shaped that is similar to speech. The measurement signal is described in [ITU-T P.501]. The signal settings used are shown in Table 9-15. A detailed description can be found in [ITU-T P.501].

   The signals are fed simultaneously in the send and receive directions. The level in the send direction is –4.7 dBPa at the MRP (nominal level), the level in the receive direction is –16 dBm0 at the electrical reference point (nominal level).

Table 9-15 – Parameters of the two test signals for double talk measurement based on amplitude- or frequency-modulated sine waves

Send direction		**Receive direction**	
$\boldsymbol{f_0^{(1)}}$ (Hz)	$\pm\Delta \boldsymbol{f}^{(1)}$ (Hz)	$\boldsymbol{f_0^{(2)}}$(Hz)	$\pm\Delta \boldsymbol{f}^{(2)}$ (Hz)
125	±2.5	180	±2.5
250	±5	270	±5
500	±10	540	±10
750	±15	810	±15
1 000	±20	1 080	±20
1 250	±25	1 350	±25
1 500	±30	1 620	±30

Table 9-15 – Parameters of the two test signals for double talk measurement based on amplitude- or frequency-modulated sine waves

Send direction		**Receive direction**	
$f_0^{(1)}$ (Hz)	$\pm\Delta f^{(1)}$ (Hz)	$f_0^{(2)}$(Hz)	$\pm\Delta f^{(2)}$ (Hz)
1 750	±35	1 890	±35
2 000	±40	2 160	±35
2 250	±40	2 400	±35
2 500	±40	2 650	±35
2 750	±40	2 900	±35
3 000	±40	3 150	±35
3 250	±40	3 400	±35
3 500	±40	3 650	±35
3 750	±40	3 900	±35
4 000	±40	4 150	±35
4 250	±40	4 400	±35
4 500	±40	4 650	±35
4 750	±40	4 900	±35
5 000	±40	5 150	±35
5 250	±40	5 400	±35
5 500	±40	5 650	±35
5 750	±40	5 900	±35
6 000	±40	6 150	±35
6 250	±40	6 400	±35
6 500	±40	6 650	±35
6 750	±40	6 900	±35
7 000	±40		

The signal generation is according to [ITU-T P.501].

3) The test signal is measured at the electrical reference point (send direction). The measured signal consists of the double talk signal which was fed in by the artificial mouth and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and bandwidth according to the signal components of the signal in the receive direction (see [ITU-T P.501]). The filter will suppress frequency components of the double talk signal.

4) In each frequency band that is used in the receive direction, the echo attenuation can be measured separately. The requirement for category 1 is fulfilled if, in any frequency band, the echo signal is either below the signal noise or below the required limit. If echo components are detectable, the classification is based on Table 9-14. The echo attenuation is to be achieved for each individual frequency band from 200 Hz to 6 950 Hz according to the different categories.

NOTE – Some headsets may fail this requirement due to perceptually based spectral filters that allow low levels of the double-talk signal to leak into the analysis window used for measuring echo. If it can be demonstrated that failures are not caused by echo, then the device under test is considered compliant with this requirement.

9.1.21 Activation in Send for communication mode (headset)

The activation in the send direction is mainly determined by the minimum built-up time in Send ($T_{r,S,min}$) and the minimum activation level ($L_{S,min}$). The minimum activation level is the level required to remove the inserted attenuation in the send direction during idle mode. The built-up time is determined for the test signal burst that is applied at the minimum activation level.

The activation level described in the following is always referred to the test signal level at the MRP).

9.1.21.1 Requirements

$L_{S,min}$ shall be ≤–20 dBPa.

$T_{r,S,min}$ (measured with minimum activation level) shall be ≤50 ms.

9.1.21.2 Test

The structure of the test signal is shown in Figure 9-10. The test signal consists of CSS components according to [ITU-T P.501] with increasing levels for each CSS burst.

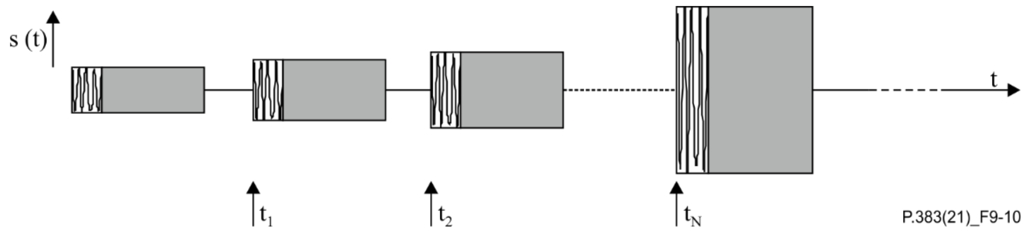


Figure 9-10 – Test signal to determine the minimum activation level and the built-up time

The settings of the test signal are given in Table 9-16 and the text that follows.

Table 9-16 – Settings of the CSS in the send direction

	CSS duration/ Pause duration	Level of the first CS signal (active signal part at the MRP)	Level difference between two periods of the test signal
CSS to determine switching characteristics in send direction	248.62 ms/ 451.38 ms	–23 dBPa (Note 1)	1 dB
NOTE – The level of the active signal part corresponds to an average level of –24.7 dBPa at the MRP for the CSS according to [ITU-T P.501], assuming a pause of 101.38 ms.			

It is assumed that the pause length of 451.38 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

1) The test set-up is described in clause 9.1.1, Figure 9-1.

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

3) The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

NOTE – If the measurement using the CSS does not allow clear identification of the minimum activation level, the measurement may be repeated by using a one-syllable word instead of the CSS. The word used should be of similar duration, the average level of the word must be adapted to the CSS level of the corresponding CSS burst.

9.2 Multimedia playback mode

9.2.1 Test set-up

9.2.1.1 Test configuration and test system

The test set-up is shown in Figure 9-1, which is reproduced below as Figure 9-11. Requirements on HATS and ear simulators are specified in clause 8.2.1 of [ITU-T P.381].

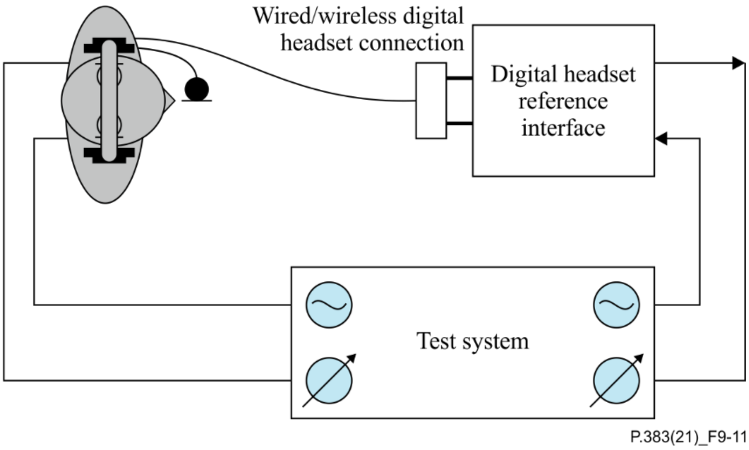


Figure 9-11 – Test set-up for the headset

When testing digital headsets, a digital headset reference interface is used to establish the audio transmission between the test system and the digital headsets. It shall be able to simulate the essential functionalities of digital headset interface of a terminal including necessary protocol handling in order to set up an audio link between the reference interface and the digital headset. It shall be capable of configuring the digital headset into a certain state to support multi-media playback. The digital headset reference interface shall not introduce any amplification or attenuation in the audio stream from or to the digital headset. No additional signal processing except the audio or speech coding, or transcoding shall be active.

NOTE – Evaluation boards from digital headset chipset vendors may be used for implementation of the digital headset reference interface.

9.2.1.2 Test signals and test signal levels

Programme simulation noise is used for the measurements. Detailed information about the test signal used can be found in the corresponding clause of this Recommendation.

Artificial test signals – which are used in Receive – have to be band limited using a bandpass filter providing more than 24 dB/octave roll-off. The band-limitation is achieved by bandpass filtering in the frequency up to 22 kHz using low-pass filter providing more than 24 dB/octave filter roll-off. The programme simulation noise according to [EN 50332-1] is band limited by design and requires no filtering.

All test signal levels are referred to the average level of the test signals, averaged over the complete test sequence length unless described otherwise.

The nominal average signal level for the measurements is −10 dBFS (decibels relative to full scale), where 0 dBFS is specified to be the maximum root mean square amplitude of a sinusoidal signal corresponding to the full scale of the digital interface.

Some tests require exact synchronization of test signals in the time domain. Therefore, consideration of terminal delays is required. When analysing signals, any delay introduced by the test system, codecs and terminals must be taken into account accordingly.

9.2.1.3 Positioning of the headsets

The same guidelines and requirements on headset positioning apply as in clause 8.1.1.3 of [ITU-T P.381].

9.2.1.4 Position and calibration of HATS

The same requirements on position and calibration of HATS apply as specified in clause 8.1.1.4 of [ITU-T P.381].

9.2.2 Output level in multimedia playback mode

For further study.

9.2.3 Frequency response in multimedia playback mode

For further study.

9.2.4 Noise in multimedia playback mode

For further study.

9.2.5 Distortion in multimedia playback mode

9.2.5.1 Requirements

The distortion is measured from the receiving output of the digital headset reference interface to the DRP with diffuse field correction.

The ratio of signal to harmonic distortion shall be above mask as listed in Table 9-17.

Table 9-17 – Limits for the signal to harmonic distortion

Frequency (Hz)	**Signal to harmonic distortion ratio limit** (dB)
100	40
315	50
5 000	50
NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale.	

9.2.5.2 Test

1) The test set-up is according to clause 9.2.1, Figure 9-2.
2) For the test, a sinusoidal signal at frequencies of 100 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz and 5 000 Hz is used. The duration of the sine wave shall be <1 s. The sinusoidal signal level shall be the nominal signal level of −10 dBFS.
3) The signal to harmonic distortion ratio is measured selectively up to 10 kHz.
4) The measurement is repeated for the second channel.

The measurement is only conducted once and *not* repeated five times.

9.2.6 Receiving crosstalk in multimedia playback mode

9.2.6.1 Requirements

The receiving crosstalk is measured as the left (L) audio channel-right (R) audio channel crosstalk and the R-L crosstalk generated by the headset by playing programme simulation noise at the output of the headset interface and measuring the resulting level at the two output channels of the headset interface.

Over a duration of 0–5 s, the attenuation, measured at the right ear and referenced to the level at the left ear, shall be above 20 dB. Over a duration of 5–10 s, the attenuation, measured at the left ear and referenced to the level at the right ear, shall also be above 20 dB.

NOTE – The crosstalk attenuation of ≥20 dB should be kept over the frequency range from 50 Hz to 16 000 Hz.

Table 9-18 – Signal sequence for the L-R and R-L audio channel crosstalk

Period (s)	**Level**	
	Left audio channel (dBFS)	**Right audio channel** (dBFS)
0-5	−10	–∞
5-10	–∞	−10

9.2.6.2 Test

1) The test set-up is according to clause 9.2.1, Figure 9-2.
2) The test signal used for the measurements shall have a programme simulation noise up to 22 kHz. The test signal is the nominal signal level. The level is averaged over the complete test signal. The signal sequence is shown in Table 9-11.
3) When the test signal is output to the headset, the crosstalk is determined by analysing the measured signal at the output of artificial ear. Over a duration of 0–5 s, the measured level at the right ear is referenced to the level at the left ear, and the attenuation is L-R audio channel crosstalk. Over a duration of 5–10 s, the measured level at the left ear is referenced to the level at the right ear, and the attenuation is R-L audio channel crosstalk.
4) The crosstalk is determined in dBPa/Pa.

The measurement is only conducted once and *not* repeated five times.

10 Terminal digital interface specification (speech signal processing in the terminal)

Tests for digital wireless terminals are performed as described in [ITU-T P.381]. Instead of the signal levels stated in [ITU-T P.381], the signal levels for digital wired headsets are as follows.

For terminal communication mode testing according to clause 7.1 of [ITU-T P.381], unless stated otherwise, the nominal average signal levels for the measurements are:

- –16 dBm0 in Receive;
- –16 dBm0 in Send (typical equivalent microphone signal level corresponding to –4.7 dBPa at the MRP.

For headset interfaces that do not provide a control channel for the headset, the Receive volume control is adjusted to the setting that produces the level closest to –16 dBm0. For headset communication mode testing according to clause 8.1 of [ITU-T P.381], unless stated otherwise, the nominal average signal levels for the measurements are:

- –16 dBm0 in Receive;

- –4.7 dBPa at the MRP.

NOTE – It is assumed that the level difference of 6 dB between monaural and binaural presentation is taken into account by the headset receiver sensitivity.

Signal levels stated otherwise are adapted accordingly.

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[b-3GPP TS 26.131] Technical Specification 3GPP TS 26.131 V18.0.0 (2022), *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Terminal acoustic characteristics for telephony; Requirements (Release 18)*.

[b-3GPP TS 26.132] Technical Specification 3GPP TS 26.132 V18.1.0 (2023), *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Speech and video telephony terminal acoustic test specification (Release 18)*.

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Published in Switzerland
Geneva, 2023