International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

P.383
(06/2021)

SERIES P: TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

Voice terminal characteristics

Technical requirements and test methods for digital wired or wireless headset interfaces

Recommendation ITU-T P.383

ITU-T P-SERIES RECOMMENDATIONS

TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

Vocabulary and effects of transmission parameters on customer opinion of transmission quality	P.10–P.19
Voice terminal characteristics	P.30–P.39
Reference systems	P.40–P.49
Objective measuring apparatus	P.50–P.59
Objective electro-acoustical measurements	P.60–P.69
Measurements related to speech loudness	P.70–P.79
Methods for objective and subjective assessment of speech quality	P.80–P.89
Voice terminal characteristics	**P.300–P.399**
Objective measuring apparatus	P.500–P.599
Measurements related to speech loudness	P.700–P.709
Methods for objective and subjective assessment of speech and video quality	P.800–P.899
Audiovisual quality in multimedia services	P.900–P.999
Transmission performance and QoS aspects of IP end-points	P.1000–P.1099
Communications involving vehicles	P.1100–P.1199
Models and tools for quality assessment of streamed media	P.1200–P.1299
Telemeeting assessment	P.1300–P.1399
Statistical analysis, evaluation and reporting guidelines of quality measurements	P.1400–P.1499
Methods for objective and subjective assessment of quality of services other than speech and video	P.1500–P.1599

For further details, please refer to the list of ITU-T Recommendations.

Recommendation ITU-T P.383

Technical requirements and test methods for digital wired or wireless headset interfaces

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 and quality of experience (QoS/QoE) 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

Keywords

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

* To access the Recommendation, type the URL http://handle.itu.int/ in the address field of your web browser, followed by the Recommendation's unique ID. For example, http://handle.itu.int/11.1002/1000/11830-en.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications, information and communication technologies (ICTs). The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure, e.g., interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents/software copyrights, which may be required to implement this Recommendation. However, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the appropriate ITU-T databases available via the ITU-T website at http://www.itu.int/ITU-T/ipr/.

Table of Contents

Recommendation ITU-T P.383

Technical requirements and test methods for digital wired or wireless headset interfaces

1 Scope

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

The aim of this Recommendation is to ensure adequate user experience when wired or wireless digital headsets are connected over the digital headset interface with digital terminals:

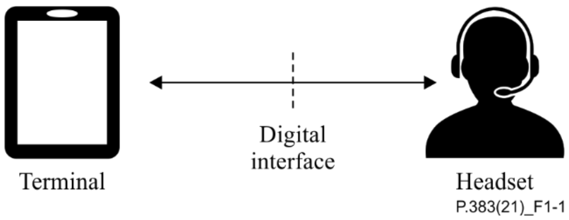

Figure 1 – The present Recommendation specifies performance parameters separately for terminals and headsets

Terminals in the context of this Recommendation may be mobile terminals, portable players, information and communication technology (ICT) terminals, internet of things (IoT) devices, or augmented reality (AR) / virtual reality (VR) devices for communication and multi-media application.

Specifying performance requirements of digital headset/headphone interface separately for headsets and terminals enable users to freely choose their favourite type of headset or headphone in the market without losing adequate user experience.

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

This Recommendation is not applicable for terminals designed solely for analogue headset/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 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.58] Recommendation ITU-T P.58 (2021), *Head and torso simulator for telephonometry*.

[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.380] Recommendation ITU-T P.380 (2003), *Electro-acoustic measurements on headsets*.

[ITU-T P.381] Recommendation ITU-T P.381 (2020), *Technical requirements and test methods for the universal wired headset or headphone interface of digital mobile terminals.*

[ITU-T P.382] Recommendation ITU-T P.382 (2020), *Technical requirements and test methods for multi-microphone wired headset or headphone interfaces of digital wireless 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.581] Recommendation ITU-T P.581 (2014), *Use of head and torso simulator for hands-free and handset terminal testing*.

[ITU-T P.700] Recommendation ITU-T P.700 (2019), *Calculation of loudness for speech communication*.

[ITU-T P.863] Recommendation ITU-T P.863 (2018*), Perceptual objective listening quality prediction*.

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

[EN 50332-1] 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] ETSI TS 103 106 V1.5.1 (2018), *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] ETSI TS 103 224 V1.5.1 (2020), *Speech and multimedia Transmission Quality (STQ); A sound field reproduction method for terminal testing including a background noise database*.

[ETSI TS 103 281] 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] IEC 61260-1:2014, *Electroacoustics – Octave-band and fractional-octave-band filters – Part 1: Specifications.*

[IEC 61672-1] 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 composite source signal (CSS) [b-ITU-T P.10]: Signal composed in time by various signal elements.

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

3.1.3 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.

3.1.4 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.

3.1.5 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 1 of [b-ITU-T P.64]).

3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1 artificial ear: A device which 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 (based on the definition in [b-ITU-T P.10]).

3.2.2 codec: Combination of an analogue-to-digital encoder and a digital-to-analogue decoder operating in opposite directions of transmission in the same equipment.

3.2.3 head and torso simulator (HATS) for telephonometry: 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 (based on the definition in [b-ITU-T P.10]).

3.2.4 headphone: 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) (based on the definition in [b-IEC 60268-7]).

3.2.5 mean opinion score – listening-only quality objective (MOS-LQO): The score is calculated by means of an objective model which aims at predicting the quality for a listening-only test situation. Objective measurements made using the model given in [ITU-T P.863] give results in terms of MOS-LQO.

3.2.6 receive: The receiving direction of the signal transmission, usually from the measurement system to the device under test (DUT).

3.2.7 send: The sending direction of the signal transmission, usually from the device under test to the measurement system.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

AGC Automatic Gain Control

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

$A_{H,S,dt}$ Attenuation range in the send direction during double talk

AR Augmented reality

dBFS	decibels relative to Full Scale.
dBV	decibels relative to 1 Volt
dBVp	decibels relative to 1 Volt, psophometrically weighted
DRP	Drum Reference Position
DUT	Device Under Test
EC	Echo Canceller
FFT	Fast Fourier Transform
HATS	Head and Torso Simulator
$HTCL_w$	Headset Weighted Terminal Coupling Loss
ICT	Information and Communication Technology
IoT	Internet of Things
L	Left audio channel
LQO	Listening-only Quality Objective
LQO_F	Listening-only Quality Objective, fullband
LQOn	Listening-only Quality Objective, narrowband
LQO_w	Listening-only Quality Objective, wideband
MOS	Mean Opinion Score
MRP	Mouth Reference Point
POI	Point of Interconnection
R	Right audio channel
RLL	Receive Loudness Level
SLL	Send Loudness Level
SRW	Short Range Wireless
STMR	Sidetone Masking Rating
TCL	Terminal Coupling Loss
TCLw	weighted Terminal Coupling Loss
TELRdt	Talker Echo Loudness Rating under double talk conditions
T_r	delay in the receiving direction
Tr,S,min	Minimum built-up time in Send
T_{rtd}	round trip delay
T_s	delay in the sending direction
$T_{Wireless}$	delay of terminal with SRW interface
$T_{Wirelessr}$	receive delay of terminal with SRW interface
$T_{Wirelessrtd}$	round trip delay of terminal with SRW interface
$T_{Wirelesss}$	send delay of terminal with SRW interface
UE	User Equipment
VR	Virtual reality

5 Conventions

None.

6 General description

[ITU-T P.381] and [ITU-T P.382] define performance requirements and corresponding test methods for standardized analogue headset/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, etc.

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 define specific physical form or dimensions of the digital headset interfaces, only the electro-acoustic performance at the digital interface and the digital headsets are specified, so that, adequate user experience could 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/speech, are recommended to meet the requirements specified in the following clauses.

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 in, e.g., 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 9 (digital interface) and in clause 10 (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, wideband, super-wideband, and fullband, depending on the speech communication modes the terminal supports.

For wireless digital headsets, signal processing is performed in the headset. 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

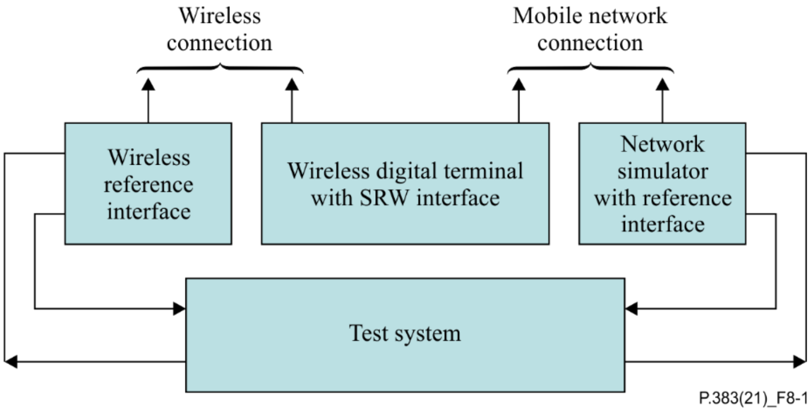

Figure 8-1 – Test set-up for testing the digital headset interface

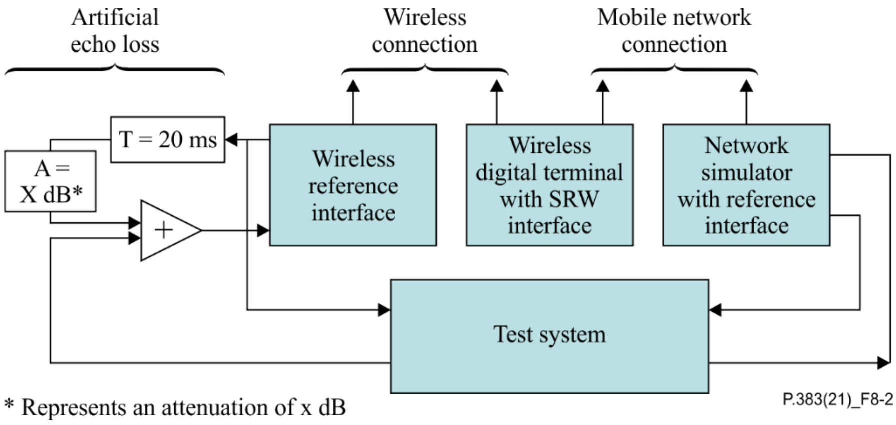

Figure 8-2 – Test set-up with artificial echo loss for echo and double talk testing

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/to the headset interface. No additional signal processing except the audio/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.

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, 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.

8.1.1.2 Test signals and test signal levels

Unless otherwise specified, fullband 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 are specified, the speech-spectrum-shaped CSS signals 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 narrowband mode, the signal shall be band-limited between 100 Hz and 4 kHz. In wideband mode, the signal shall be band-limited between 100 Hz and 8 kHz. In super-wideband mode, the frequency range shall be band-limited between 50 Hz and 16 kHz. In fullband 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 itself.

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 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, it is required to consider 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 send direction is measured from the wireless reference interface to the point of interconnection (speech codec of the system simulator output), and in receive direction from the point of interconnection (POI) to the wireless reference interface and it is corrected for the test equipment delays.

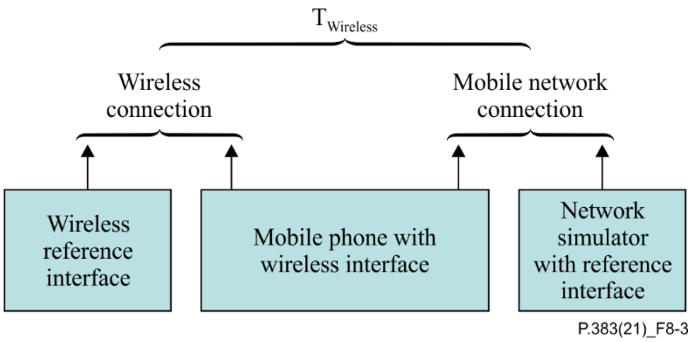

Figure 8-3 – Delay in wireless headset connections

Definitions:

– The mobile phone delay in the send (uplink) direction $T_{Wirelesss}$ 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 $T_{Wirelessr}$ 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 sum of send and receive delays i.e., round trip delay of terminal with SRW interface ($T_{Wirelessrtd}$) = send delay of terminal with SRW interface ($T_{Wirelesss}$) + receive delay of terminal with SRW interface ($T_{Wirelessr}$) shall be less than 150 ms.

NOTE 1 – The delay $T_{Wirelessrtd}$ should be minimized.

NOTE 2 – The system delay T_{system} depends on the transmission method used and the network simulator. The delay T_{system} 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 composite source signal (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_s, sys). The sending delay (T_s, wdt) is expressed in milliseconds, 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 composite source signal (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 digital headset reference interface.
4) The measurement is corrected by the delay in the receiving direction introduced by the test equipment (T_r, sys). The receiving delay (Tr, wdt) is expressed in milliseconds, 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 send junction loudness rating (JLR) should be:

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

8.1.3.2 Test

1) The test arrangement 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 narrowband, the send sensitivity is calculated from each band of the frequencies given in Table 1 of [ITU-T P.79], bands 4-17. For wideband, super-wideband and fullband, the send sensitivity is calculated from each band of the frequencies given in Table A.2 of [ITU-T P.79], bands 1-20. 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 dB. For narrowband, the junction loudness rating (JLR) is calculated according to equation A-23d of [ITU-T P.79], bands 4-17, m = 0.175 and the weighting factors (W_J) for JLR according to Table A.2 of [ITU-T P.79]. For wideband, super-wideband and fullband, the junction loudness rating (JLR) is calculated according to equation A-23d of [ITU-T P.79], bands 1-20.

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

8.1.4.1 Requirements

The nominal values of receive junction loudness rating (JLR) should be:

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

It is recommended that the terminal with digital headset interfaces should transmit the audio signals in sending and receiving direction 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 arrangement 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 narrowband, the send sensitivity is calculated from each band of the frequencies given in Table 1 of [ITU-T P.79], bands 4-17. For wideband, super-wideband and fullband, the send sensitivity is calculated from each band of the frequencies given in Table A.2 of [ITU-T P.79], bands 1-20. For the calculation, the average measured level at the receiving output of digital headset reference interface for each frequency band is referred to the reference signal.

4) The sensitivity is expressed in dB. For narrowband, the junction loudness rating (JLR) is calculated according to equation A-23d of [ITU-T P.79], bands 4-17, m = 0.175 and the weighting factors (W_J) for JLR according to Table A.2 of [ITU-T P.79]. For wideband, super-wideband and fullband, the junction loudness rating (JLR) is calculated according to equation A-23d of [ITU-T P.79], bands 1–20.

8.1.5 Linearity in send for the communication mode (terminal)

8.1.5.1 Requirements

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

8.1.5.2 Test

1) The test arrangement 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 of −30 dB to 0 dB in steps of 10 dB relative to the nominal signal level –16 dBm0, measured at the sending input of digital headset reference interface.

3) The JLRsnd 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 of –30 dB/0 dB from the nominal signal level, the measured JLRrcv shall not deviate more than ±2 dB from the JLRrcv measured with the nominal signal level.

8.1.6.2 Test

1) The test arrangement 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 of −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) The JLRrcv 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, the mask is drawn by straight lines between the breaking points on a logarithmic (frequency) – linear (dB sensitivity) scale.

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 arrangement 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 fullband, the sending sensitivity is determined in 1/12-octave intervals, as given by [IEC 61260-1] for frequencies from 100 Hz to 20 kHz inclusive, measured at the output of the system simulator. In super-wideband, it is determined for frequencies from 100 Hz to 16 kHz. In wideband, it is determined for frequencies from 100 Hz to 8 kHz. In narrowband, it is determined for frequencies from 100 Hz to 4 kHz. In each 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 (dB).

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

8.1.8.1 Requirements

The receive sensitivity frequency response is measured from the input of the system simulators to the receiving output of 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, the mask is drawn by straight lines between the breaking points on a logarithmic (frequency) – linear (dB sensitivity) scale.

Table 8-5 – Tolerance mask for the narrowband receiving 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 8-6 – 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

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

Frequency (Hz)	Upper limit	Lower limit	Target
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
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 arrangement 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 fullband, the receiving sensitivity is determined in third octave intervals, as given by [IEC 61260-1] for frequencies from 100 Hz to 20 kHz inclusive, measured at the receiving output of digital headset reference interface. In super-wideband, it is determined for frequencies from 100 Hz to 16 kHz. In wideband, it is determined for frequencies from 100 Hz to 8 kHz. In narrowband, it is determined for frequencies from 100 Hz to 4 kHz. In each 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 arrangement is according to clause 8.1.1, Figure 8-1.

2) For the noise measurement, no test signal is used. However, all sources which potentially contribute to noise should be considered. Interference from radio frequencies is not accurately covered by an interface specification as the complete terminal/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 narrowband, between 100 Hz and 8 kHz for wideband, between 100 Hz and 16 kHz for super-wideband, and between 100 Hz and 20 kHz for fullband. 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 k samples/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 k samples/48 kHz sampling rate with Hanning window or equivalent). A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(–1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum from 100 Hz to 3.4 kHz in narrowband, from 100 Hz to 6.3 kHz in wideband, from 100 Hz to 13 kHz in super-wideband and from 100 Hz to 18 kHz in fullband.

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 arrangement is according to clause 8.1.1, Figure 8-1.

2) For the noise measurement, no test signal is used. However, all sources which potentially contribute to noise should be considered. Interference from radio frequencies is not accurately covered by an interface specification as the complete terminal/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 digital headset reference interface in the frequency range between 100 Hz and 4 kHz for narrowband, between 100 Hz and 8 kHz for wideband, between 100 Hz and 16 kHz for super-wideband and between 100 Hz and 20 kHz for fullband. 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 k samples/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 k samples/48 kHz sampling rate with Hanning window or equivalent). A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(–1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum from 100 Hz to 3.4 kHz in narrowband, from 100 Hz to 6.3 kHz in wideband, from 100 Hz to 13 kHz in super-wideband and from 100 Hz to 18 kHz in fullband.

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.

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 arrangement 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 narrowband, up to 7 kHz for wideband, up to 14 kHz for super-wideband and up to 20 kHz for fullband.

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.

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 arrangement 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 narrowband and wideband and up to 20 kHz for super-wideband and fullband.
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 or not when 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 arrangement 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, the level is the test signal is the nominal level as defined 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 or not 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 echo loss measured shall be 20 dB ±2 dB.

8.1.14.2 Test

1) The test arrangement 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 1/3-octave band intervals, as given by [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 narrowband, the used frequency range is from 300 Hz to 3 400 Hz. For wideband, super-wideband, and fullband, the used frequency range is from 300 Hz to 6 700 Hz.
6) Terminal coupling loss (TCL) is calculated as unweighted echo loss according to [ITU-T G.122], Annex B, clause B.4 (trapezoidal rule).
7) The measured TCL is corrected by the measured JLRsnd and JLRrcv.

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 narrowband, wideband, super-wideband and fullband, 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, a complete signal processing that degrades conversation quality can be avoided.

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

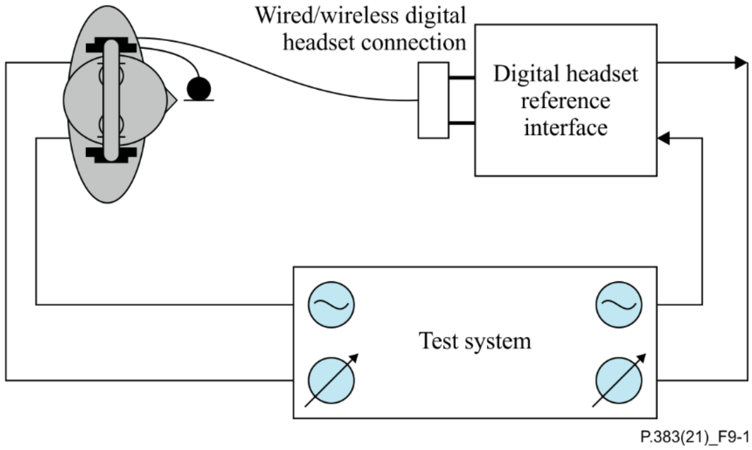

Figure 9-1 – Test arrangement for testing the headset

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 speech communication. The digital headset reference interface shall not introduce any amplification or attenuation in the audio stream from/to the headset. No additional signal processing except the audio/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, fullband 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 are specified, the speech-spectrum-shaped CSS signals 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 narrowband mode, the signal shall be band-limited between 100 Hz and 4 kHz. In wideband mode, the signal shall be band-limited between 100 Hz and 8 kHz. In super-wideband mode, the signal shall be band-limited between 50 Hz and 16 kHz. In fullband 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, it is required to consider the delays of the terminals. 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

Recommendations for the set up and positioning of headsets are given in [ITU-T P.380]. Unless stated otherwise, headsets shall be placed in their recommended wearing position.

Some insert earphones may not fit properly in Type 3.3 or type 4.3 artificial ears. The HATS should be equipped with two artificial ears as specified in [ITU-T P.57]. For binaural headsets, two artificial ears are required.

When a headset is placed on a HATS, the results may vary from trial to trial due to slight variations in positioning. Relatively accurate and repeatable results can be obtained by making several measurements and averaging the results according to [ITU-T P.380]. Unless stated otherwise, the measurements in receive should be repeated five times and averaged. The headset shall be completely removed from the artificial ear and re-mounted for each trial. The averaged result is used.

9.1.1.4 Calibration of HATS

The HATS shall be equipped with Type 3.3 or 3.4 artificial ear specified in [ITU-T P.57]. For measurements of binaural headsets, the HATS shall be equipped with both left and right artificial ears. The pinnae shall be positioned on HATS according to [ITU-T P.58].

The calibration and equalization procedures as well as how to combine the two ears' signals can be found in [ITU-T P.581]. Unless stated otherwise, the measurement from HATS at drum reference position (DRP) shall be corrected to diffuse field. For 1/3-octave band measurements, the inverse of the nominal diffuse field curve in [ITU-T P.58] Table 3 shall be used. For 1/12-octave band measurements, the inverse of the nominal diffuse field curve in [ITU-T P.58] Annex A shall be used. For measurements requiring diffuse field correction values for frequencies other than those used in the [ITU-T P.58] tables, linear interpolation on a log frequency scale from the [ITU-T P.58] Annex A values shall be used.

The HATS shall be equipped with an equalized artificial mouth simulator according to [ITU-T P.58]. The spectrum of an acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP.

9.1.1.5 Test setup for quality in the presence of ambient noise measurements

The setup for simulating realistic ambient noises and the positioning of the HATS in a lab-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 setup for a loudspeaker arrangement suitable to simulate an ambient noise field in a lab-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 setup are given in detail in [ETSI TS 103 224].

The microphone array setup 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:

– 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 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 arrangement is according to clause 9.1.1, Figure 9-1.
2) The test signal to be used for the measurements is a composite source signal (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) is expressed in milliseconds, determined from the maximum of the cross-correlation function.

9.1.2.2.2 Test of overall delay in receive

1) The test arrangement is according to clause 9.1.1, Figure 9-1.
2) The test signal to be used for the measurements is a composite source signal (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) is expressed in milliseconds, 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 SLR shall be:

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

9.1.3.2 Test

1) The test arrangement 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 narrowband, the send sensitivity is calculated from each band of the frequencies given in Table 1 of [ITU-T P.79], bands 4-17. For wideband, super-wideband, and fullband, the send sensitivity is calculated from each band of the frequencies given in Table A.2 of [ITU-T P.79], bands 1-20. 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 volt (dBV)/Pa. For narrowband, the send loudness rating (SLR) is calculated according to equation 5-1 of [ITU-T P.79], bands 4-17, m = 0.175 and the weighting factors in the send direction according to Table 1 of

[ITU-T P.79]. For wideband, super-wideband and fullband, the send loudness rating (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 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 ≤ (equal or louder than) −13 dB.

With the volume control set to the minimum position the RLR shall not be ≥ (equal or quieter than) 18 dB and shall not be ≥ (equal or quieter than) 24 dB for a binaural headset.

9.1.4.2 Test

1) The test arrangement 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 narrowband, the send sensitivity is calculated from each band of the frequencies given in Table 1 of [ITU-T P.79], bands 4-17. For wideband, super-wideband and fullband, the send sensitivity is calculated from each band of the frequencies given in Table A.2 of [ITU-T P.79], bands 1-20. 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 dB. For narrowband, the receive loudness rating (RLR) is calculated according to [ITU T P.79], formula (A-23c), over bands 4 to 17, using m = 0,175 and the receiving weighting factors from Table 1 of [ITU-T P.79], without the LE factor. For wideband, super-wideband and fullband, the receive loudness rating (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 send loudness level (SLL) shall be:

SLL = 76 phon ± 4 phon

9.1.5.2 Test

1) The test arrangement 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 send loudness level (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/stereo headsets shall be:

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

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

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

9.1.6.2 Test

1) The test arrangement 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 receive loudness level (RLL) is calculated as described in clause 8 of [ITU-T P.700].

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

9.1.7.1 Requirements

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

For digital headsets only capable of transmitting narrowband speech, the tolerance mask for the sending sensitivity frequency response is shown in Table 9-1. For digital headsets capable of transmitting wideband speech, the tolerance mask shown in Table 9-2 shall be met. For digital headsets capable of transmitting super-wideband and fullband speech, an additional tolerance mask shown in Table 9-3 should be met when measuring in 1/3-octaves, besides meeting the tolerance mask of wideband speech in Table 9-2.

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
3 400	4	–9
4 000	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.

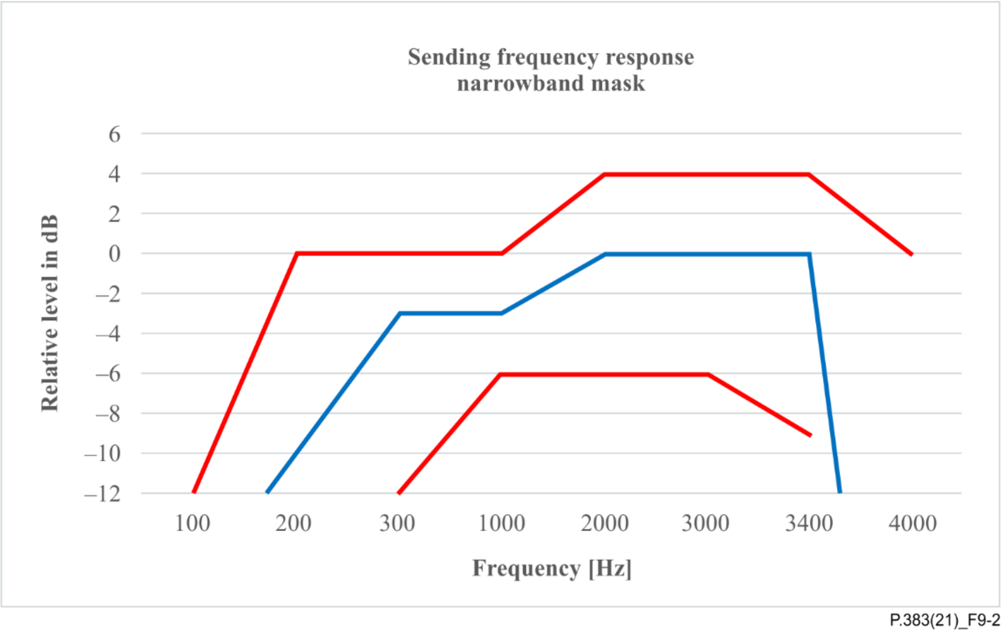

Figure 9-2 – Tolerance mask for narrowband sending frequency response. Upper and lower tolerance mask are shown in red colour. The nominal sending frequency response is shown in blue colour

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.

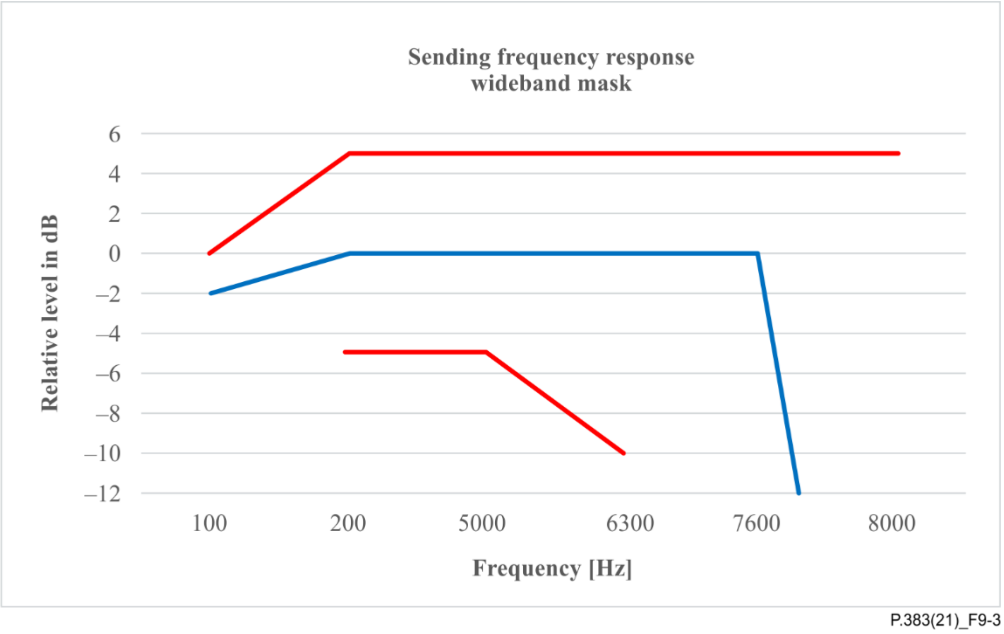

Figure 9-3 – Tolerance mask for wideband sending frequency response. Upper and lower tolerance mask are shown in red colour. The nominal sending frequency response is shown in blue colour

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.		

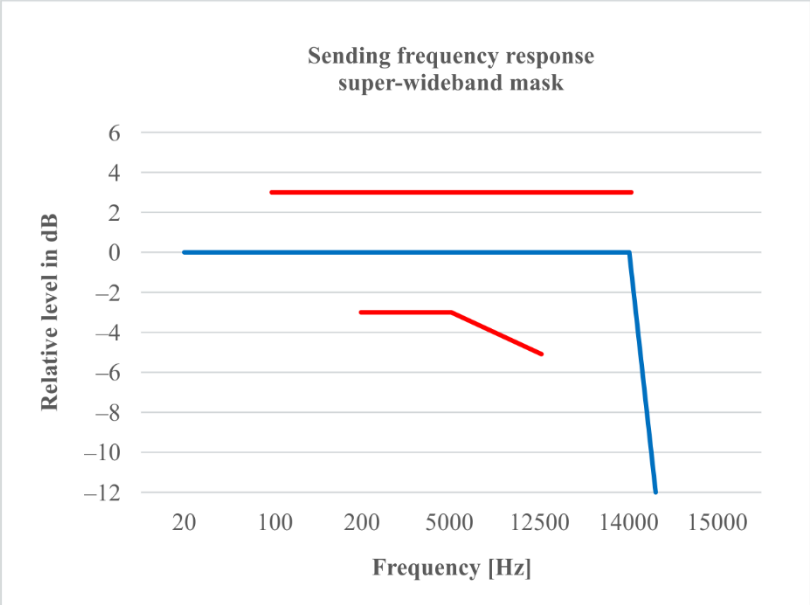

Figure 9-4 – Tolerance mask for super-wideband sending frequency response. Upper and lower tolerance mask are shown in red colour. The nominal sending frequency response is shown in blue colour

9.1.7.2 Test

1) The test arrangement 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 1/12-octave intervals, as given by [IEC 61260-1] for frequencies from 100 Hz to 4 kHz inclusive, measured at the POI. For wideband, measurements shall be made for frequencies from 100 Hz to 8 kHz inclusive for super-wideband and fullband, measurements shall be made at both 1/3-octave and 1/12-octave intervals as given by [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/frequency response for communication mode (headset)

9.1.8.1 Requirements

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

For digital headsets only capable of transmitting narrowband speech, the tolerance mask for the receiving sensitivity frequency response is shown in Table 9-4. For digital headsets capable of transmitting wideband speech, the tolerance mask shown in Table 9-5 shall be met. For digital headsets capable of transmitting super-wideband and fullband speech, an additional tolerance mask shown in Table 9-6 should be met when measuring in 1/3-octaves, besides meeting tolerance mask of wideband speech in Table 9-6.

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.		

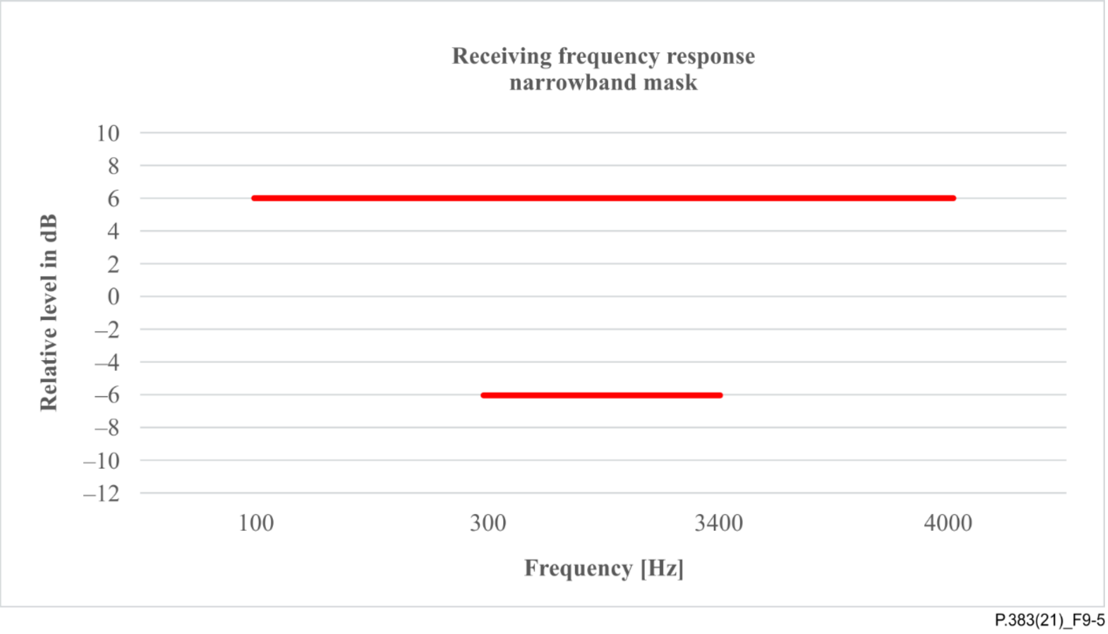

Figure 9-5 – Tolerance mask for narrowband receiving frequency response. Upper and lower tolerance mask are shown in red colour. The nominal receiving frequency response is shown in blue colour

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.		

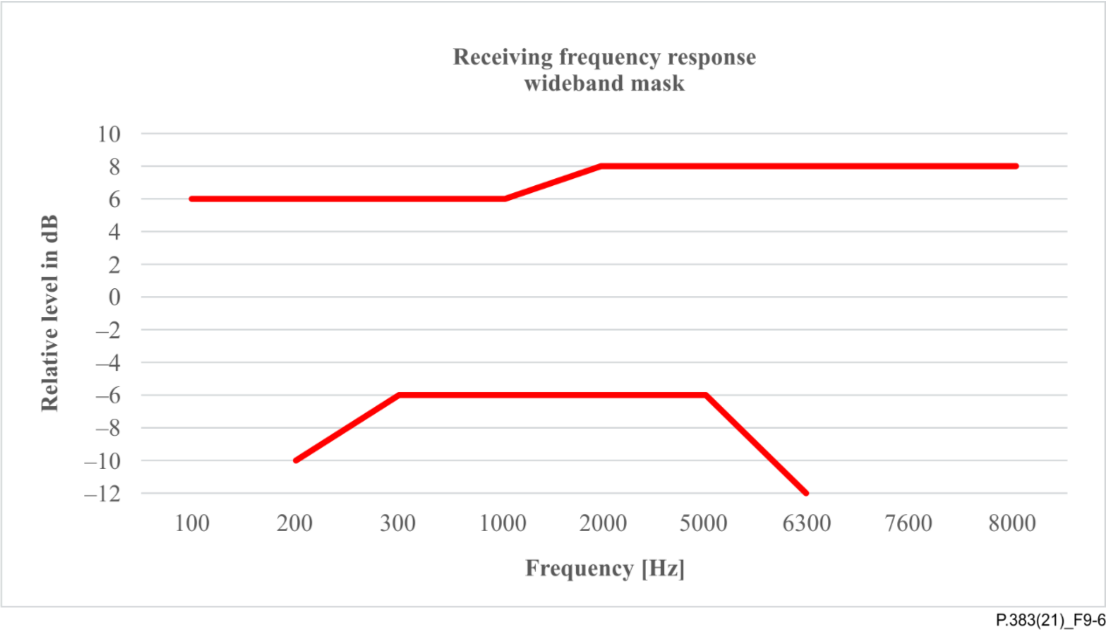

Figure 9-6 – Tolerance mask for wideband receiving frequency response. Upper and lower tolerance mask are shown in red colour. The nominal receiving frequency response is shown in blue colour

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.

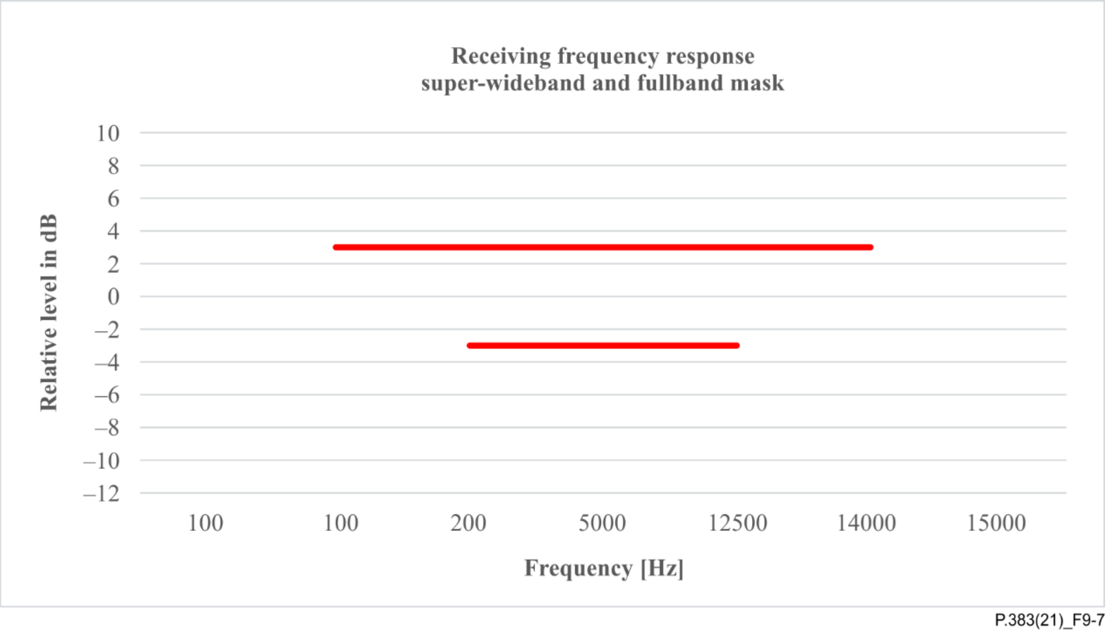

Figure 9-7 – Tolerance mask for super-wideband receiving frequency response. Upper and lower tolerance mask are shown in red colour. The nominal receiving frequency response is shown in blue colour

9.1.8.2 Test

1) The test arrangement 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 1/12-octave intervals, as given by [IEC 61260-1] for frequencies from 100 Hz to 4 kHz inclusive, measured at the POI. For wideband, measurements shall be made for frequencies from 100 Hz to 8 kHz inclusive. For super-wideband and fullband, measurements shall be made at both 1/3-octave and 1/12-octave intervals as given by [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 is referred to 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 is independent of the volume control setting.

NOTE 2 – In case the human air-conducted sidetone paths are obstructed (for example binaural insert type headsets), it is important to provide a sidetone path.

NOTE 3 – In case 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 arrangement 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 1/12-octave intervals as given by [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 (LmeST), as expressed in dB, is calculated from each 1/3rd-octave band according to [ITU-T P.79], Table B.1, bands 1 to 20. The sidetone masking rating (STMR), expressed in dB, 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) In case 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. In case 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 is normally disabling 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 setup, 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 acoustical or mechanical sidetone path when interpreting sidetone delay results.

9.1.10.2 Test

1) The test arrangement is according to clause 9.1.1, Figure 9-1.
2) The test signal is a CS-signal complying with [ITU-T P.501] using a pn sequence with a length of 4 096 points (for the 48 kHz sampling rate) which equals to the period *T*. The duration of the complete test signal is as specified in [ITU-T 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 DRP is calculated in the time domain:

$$\Phi_{xy}(\tau) = \frac{1}{T} \int_{t=\frac{-T}{2}}^{\frac{T}{2}} S_x(t) \cdot S_y(t+\tau)dt \tag{9-1}$$

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 one occurs with a possible delayed

sidetone signal. The difference between the two maxima corresponds to the sidetone delay. The envelope $E(\tau)$ is calculated by the Hilbert transformation $H\{xy(\tau)\}$ of the cross-correlation:

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

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

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 arrangement 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 narrowband, between 100 Hz and 8 kHz for wideband, between 100 Hz and 16 kHz for super-wideband and between 100 Hz and 20 kHz for fullband. 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 k samples/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 k samples/48 kHz sampling rate with Hanning window or equivalent). A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(–1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum from 100 Hz to 3.4 kHz in narrowband, from 100 Hz to 6.3 kHz in wideband, from 100 Hz to 13 kHz in super-wideband and from 100 Hz to 18 kHz in fullband.

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 arrangement 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 narrowband and wideband, between 100 Hz and 16 kHz for super-wideband and fullband. 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 k samples/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 k samples/48 kHz sampling rate with Hanning window or equivalent). A smoothed average idle channel noise spectrum is calculated by a moving average (arithmetic mean) 1/3rd octave wide across the idle noise channel spectrum stated in dB (linear average in dB of all FFT bins in the range from 2^(–1/6)f to 2^(+1/6)f). Peaks in the idle channel noise spectrum are compared against a smoothed average idle channel noise spectrum from 100 Hz to 6.3 kHz in narrowband and wideband, from 100 Hz to 13 kHz in super-wideband and fullband.

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
1020	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 arrangement 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 1020 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 composite source signals, 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 * 4096 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 volt (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 0.7071 * fs, and an upper passband starting at 1.4142 * fs, where fs 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 200Hz to 4kHz for narrowband and 100 Hz to 6 kHz for wideband, super-wideband, and fullband, after applying psophometric and stop filters.

9.1.14 Receive distortion for communication mode (headset)

9.1.14.1 Requirements

The distortion in 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
1020	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 (dB) scale.		

9.1.14.2 Test

1) The test arrangement 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 1020 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 levels down to low levels.

3) In order to guarantee a reliable activated state of the digital headset, a sequence of four composite source signals, 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 ∗ 4096 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 0.7071 ∗ fs, and an upper passband starting at 1.4142 * fs, where fs 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 narrowband and 100 Hz to 6 kHz for wideband, super-wideband and fullband, 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 narrowband digital headsets:

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

For wideband digital headsets:

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

For super-wideband and fullband digital headsets:

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 arrangement is according to clause 9.1.1, Figure 9-1.

2) Before starting the measurements, a proper conditioning sequence shall be used. For narrowband and wideband, 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 super-wideband and fullband, 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 narrowband and wideband, 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:

   – 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 super-wideband and fullband, 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:

   – 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 narrowband and wideband, 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 super-wideband and fullband, 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	Handset levels
Pub Noise (*Pub*)	HATS and microphone array in a pub	30 s	1: 77,2 dB 2: 76,6 dB 3: 75,7 dB 4: 76,0 dB 5: 76,0 dB 6: 76,3 dB 7: 76,0 dB 8: 76,4 dB
Roadnoise (Roadnoise)	HATS and microphone array standing outside near a road	30 s	1: 72,8 dB 2: 71,6 dB 3: 72,0 dB 4: 72,9 dB

Table 9-9 – Noises used for background noise simulation

Name	Description	Length	Handset levels
			5: 72,2 dB 6: 73,1 dB 7: 73,0 dB 8: 73,8 dB
Departure platform (TrainStation)	HATS and microphone array on the departure platform of a train station	30 s	1: 78,9 dB 2: 78,8 dB 3: 79,1 dB 4: 80,0 dB 5: 79,4 dB 6: 79,6 dB 7: 78,8 dB 8: 80,1 dB
Full-size car 130 km/h (FullSizeCar_130)	HATS and microphone array at co-drivers' position	30 s	1: 68,5 dB 2: 68,3 dB 3: 68,8 dB 4: 69,5 dB 5: 69,9 dB 6: 70,5 dB 7: 70,8 dB 8: 71,9 dB
Sales counter (SalesCounter)	HATS and microphone array in a supermarket	30 s	1: 66,6 dB 2: 66,1 dB 3: 65,7 dB 4: 66,5 dB 5: 66,3 dB 6: 66,8 dB 7: 66,6 dB 8: 67,1 dB
Cafeteria (Cafeteria)	HATS and microphone array inside a cafeteria	30 s	1: 70,0 dB 2: 70,0 dB 3: 70,1 dB 4: 70,7 dB 5: 70,5 dB 6: 70,8 dB 7: 70,6 dB 8: 71,0 dB
Callcenter 2 (Callcenter)	HATS and microphone array in business office	30 s	1: 60,2 dB 2: 60,0 dB 3: 60,1 dB 4: 60,8 dB 5: 60,2 dB 6: 60,6 dB 7: 60,2 dB 8: 60,7 dB

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 weighted terminal coupling loss (HTCLw) is measured from the input of the digital headset reference interface to the output of the digital headset reference interface.

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

9.1.18.2 Test

1) The test arrangement 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) HTCLw is calculated according to weighted terminal coupling loss (TCLw) in clause B.4 of [ITU-T G.122] (trapezoidal rule). For narrowband, the used frequency range is from 300 Hz

to 3 400 Hz. For wideband, super-wideband and fullband, the used frequency range is from 300 Hz to 6 700 Hz.

Terminal coupling loss (TCL) is calculated as unweighted echo loss from 100 Hz to 8 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 must be applied which is then adapted to the duration of the actual test signal. The echo loss is calculated by the equations

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

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 ratio at frequency f_i; and

A_N: the ratio at frequency $f_N = 8000$ Hz.

The above equation is a generalized form of the equation defined in clause B.4 of [ITU-T G.122] for calculating echo loss based on tabulated data, which allows the calculation of echo loss within any frequency range between f_0 and f_N.

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 echo canceller. 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 CS-signal 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 seconds 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 arrangement is according to clause 9.1.1, Figure 9-1.

2) The test signal consists of periodically repeated composite source signal 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 CS-signal. The integration time for the level analysis shall be 35 ms, the analysis is referred to the level analysis of the reference signal. 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 CS-signal 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 CSS, the analysis is conducted only during the active signal part, the pauses between the composite source signals are not analysed. The analysis time is reduced by the integration time of the level analysis (35 ms).

9.1.20 Double talk performance for communication mode (headset)

NOTE – Before starting the double talk tests, the test lab should ensure that the echo canceller 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 which is realized by a high attenuation range in this case.

The most important parameters determining the speech quality during double talk are (see [ITU-T 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: $A_{H,S,dt}$

9.1.20.1.1 Requirements

Based on the variation of the level in 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 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 will decrease.

In general, Table 8-10 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 defined 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.

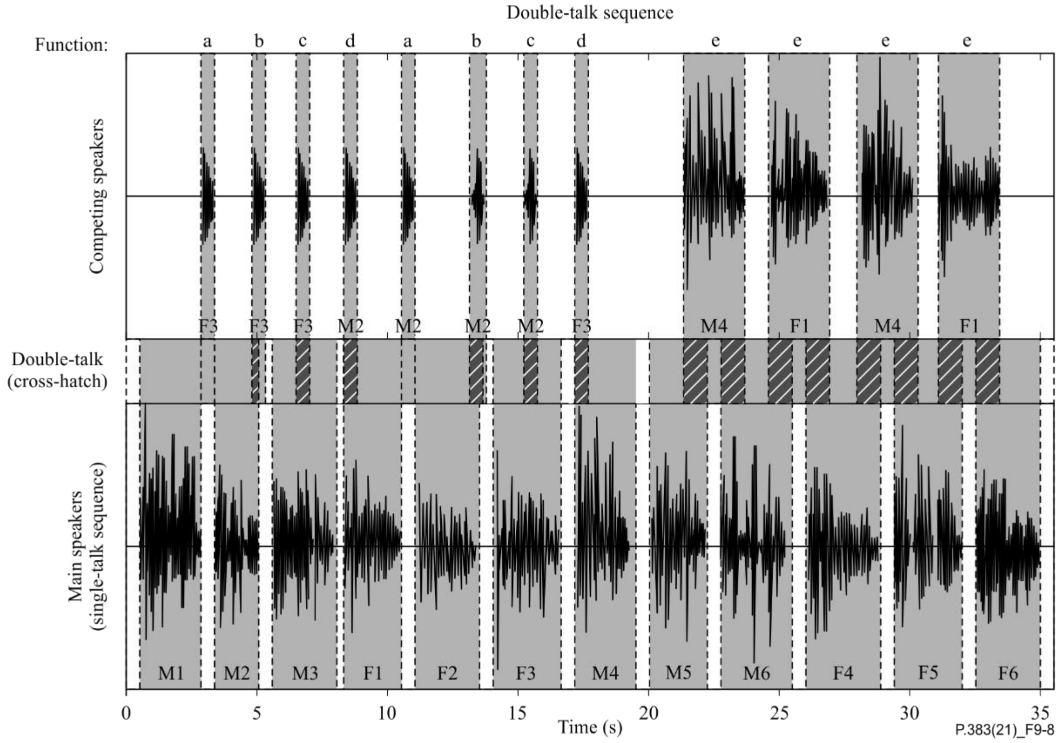

Figure 9-8 – Double talk test sequence with overlapping speech sequences in send and receive directions

The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement 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 arrangement is according to clause 9.1.1, Figure 9-1. Before the actual test, a training sequence for the echo canceller 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 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 defined 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*
$A_{H,R,dt}$ [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 defined 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 arrangement shall be constant during the measurement.

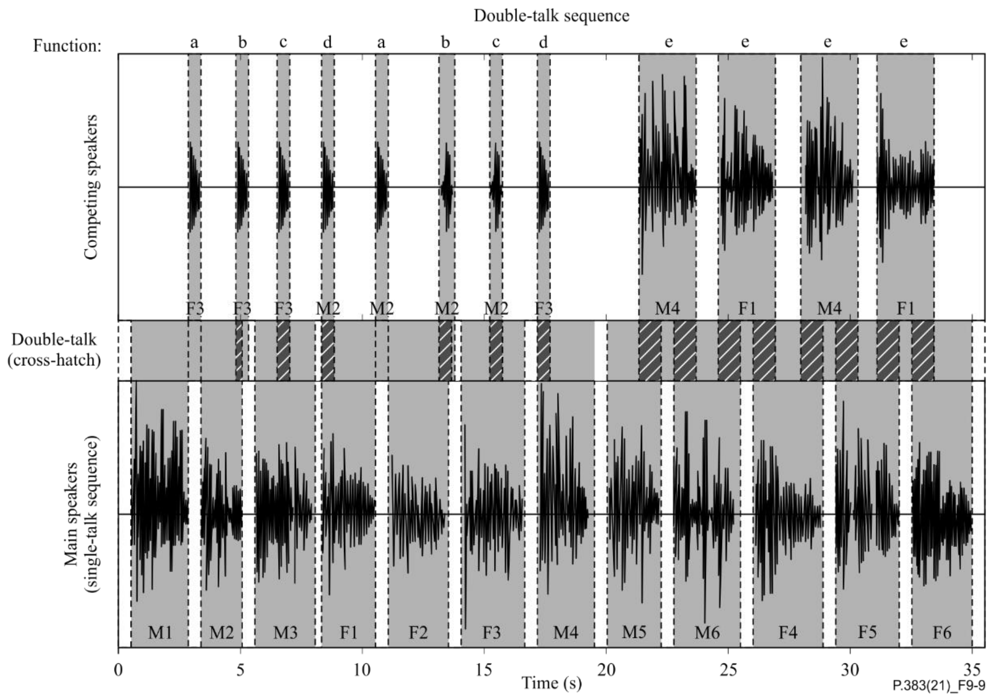

Figure 9-9 – Double talk test sequence with overlapping speech sequences in receive and send directions

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 maximum volume control setting in receive direction.

1) The test arrangement 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 defined 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 ($TELR_{dt}$). It is assumed that the terminal at the opposite end of the connection provides a nominal loudness rating (SLR + RLR = 10 dB). "Echo Loss" 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 arrangement is according to clause 9.1.1, Figure 9-1.
2) The double talk signal consists of a sequence of orthogonal signals which 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 send and receive directions. The level in send direction is –4.7 dBPa at the MRP (nominal level), the level in receive direction is –16 dBm0 at the electrical reference point (nominal level).

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

Send direction		Receive direction	
$f_0^{(1)}$ (Hz)	$\pm\Delta f^{(1)}$ (Hz)	$f_0^{(2)}$(Hz)	$\pm\Delta f^{(2)}$ (Hz)
125	±2.5	180	±2.5
250	±5	270	±5
500	±10	540	±10
750	±15	810	±15
1 000	±20	1 080	±20
1 250	±25	1 350	±25
1 500	±30	1 620	±30
1 750	±35	1 890	±35
2 000	±40	2 160	±35
2 250	±40	2 400	±35
2 500	±40	2 650	±35
2 750	±40	2 900	±35
3 000	±40	3 150	±35
Send direction		**Receive direction**	
$f_0^{(1)}$ (Hz)	$\pm\Delta f^{(1)}$ (Hz)	$f_0^{(2)}$(Hz)	$\pm\Delta f^{(2)}$ (Hz)
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 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 8-14. 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 which 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 DUT is considered compliant with this requirement.

9.1.21 Activation in send for communication mode (headset)

The activation in 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 send direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

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

9.1.21.1 Requirements

The minimum activation level $L_{S,min}$ shall be ≤ –20 dBPa.

The minimum built-up time $T_{r,S,min}$ (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.

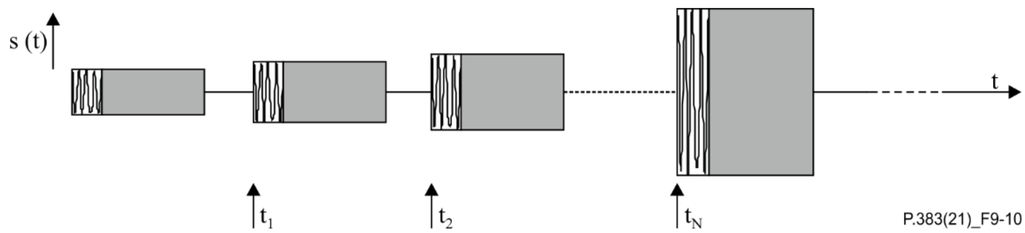

Figure 9-10 – Test signal to determine the minimum activation level and the built-up time

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 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 arrangement 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 vs 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 CS-signal does not allow to clearly identify the minimum activation level, the measurement may be repeated by using a one-syllable word instead of the CS-signal. The word used should be of similar duration, the average level of the word must be adapted to the CS-signal 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.

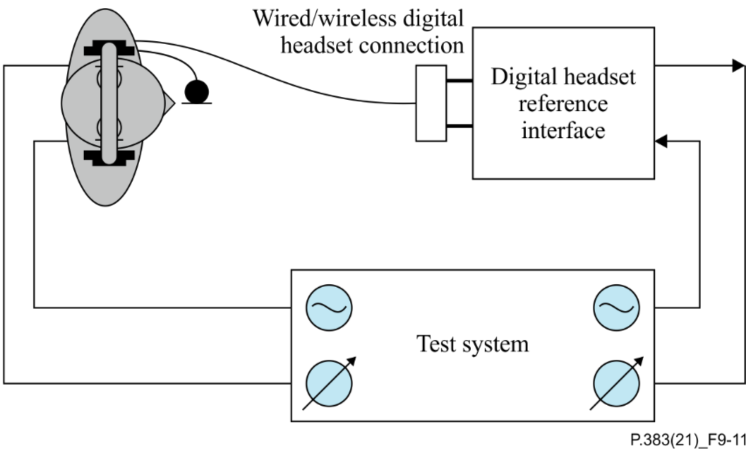

Figure 9-11 – Test arrangement for testing the headset

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/to the digital headset. No additional signal processing except the audio/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 decibels relative to full scale (dBFS), where 0 dBFS is defined as being the maximum RMS 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, it is required to consider the delays of the terminal. 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

Recommendations for the set up and positioning of headsets are given in [ITU-T P.380]. Unless stated otherwise, headsets shall be placed in their recommended wearing position. Some insert earphones may not fit properly in Type 3.3 ear simulators. For such insert type headsets, an [ITU-T P.57] Type 2 ear simulator may be used in conjunction with the HATS mouth simulator. The HATS should be equipped with two artificial ears as specified in [ITU-T P.57]. For binaural headsets two artificial ears are required.

Unless stated otherwise the measurements in receive are repeated five times and averaged. The averaged result is used.

9.2.1.4 Position and calibration of HATS

The HATS shall be equipped with a Type 3.3 or 4.3 artificial ear. For the measurement of binaural headsets, the HATS shall be equipped with two artificial ears. The pinnae are specified in [ITU-T P.57] for Type 3.3 and Type 4.3 artificial ears. The pinnae shall be positioned on HATS according to [ITU-T P.58].

The exact calibration and equalization procedures as well as how to combine the two ear signals for the purpose of measurements can be found in [ITU-T P.581]. Unless stated otherwise, the HATS shall be diffuse field equalized. The DRP to diffuse field correction curve as found in Table 14A and Table 14B of [ITU-T P.58] shall be used for 1/3rd octaves and 1/12th octaves respectively.

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 the following mask as shown 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 arrangement 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 audio channel-right audio channel (L-R) 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.

In the 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. In the 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 crosstalk

level / period	Left audio channel (dBFS)	Right audio channel (dBFS)
0-5 s	−10	–∞
5-10 s	–∞	−10

9.2.6.2 Test

1) The test arrangement 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 8-11.
3) Output the test signal to the headset, the crosstalk is determined by analysing the measured signal at the output of artificial ear. In 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 crosstalk. In 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 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 mouth reference point (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 headsets 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.

Bibliography

[b-ITU-T P.10] Recommendation ITU-T P.10/G.100 (2017), *Vocabulary for performance, quality of service and quality of experience*.

[b-ITU-T P.64] Recommendation ITU-T P.64 (2019), *Determination of sensitivity/frequency characteristics of local telephone systems.*

[b-3GPP TS 26.131] 3GPP TS 26.131 V12.3.0 (2014), *Terminal acoustic characteristics for telephony; Requirements*. http://www.3gpp.org/DynaReport/26131.htm>

[b-3GPP TS 26.132] 3GPP TS 26.132 V10.1.0 (2011), *Speech and video telephony terminal acoustic test specification (Release 10).*

[b-ETSI EG 202 396-3] ETSI EG 202 396-3 V1.7.1 (2018), *Speech and multimedia Transmission Quality (STQ); Speech Quality performance in the presence of background noise; Part 3: Background noise transmission – Objective test methods.*

[b-IEC 60268-7] IEC 60268-7:2010+AMD1:2020 CSV Consolidated version, *Sound system equipment – Part 7: Headphones and earphones*.

SERIES OF ITU-T RECOMMENDATIONS

Series A	Organization of the work of ITU-T
Series D	Tariff and accounting principles and international telecommunication/ICT economic and policy issues
Series E	Overall network operation, telephone service, service operation and human factors
Series F	Non-telephone telecommunication services
Series G	Transmission systems and media, digital systems and networks
Series H	Audiovisual and multimedia systems
Series I	Integrated services digital network
Series J	Cable networks and transmission of television, sound programme and other multimedia signals
Series K	Protection against interference
Series L	Environment and ICTs, climate change, e-waste, energy efficiency; construction, installation and protection of cables and other elements of outside plant
Series M	Telecommunication management, including TMN and network maintenance
Series N	Maintenance: international sound programme and television transmission circuits
Series O	Specifications of measuring equipment
Series P	**Telephone transmission quality, telephone installations, local line networks**
Series Q	Switching and signalling, and associated measurements and tests
Series R	Telegraph transmission
Series S	Telegraph services terminal equipment
Series T	Terminals for telematic services
Series U	Telegraph switching
Series V	Data communication over the telephone network
Series X	Data networks, open system communications and security
Series Y	Global information infrastructure, Internet protocol aspects, next-generation networks, Internet of Things and smart cities
Series Z	Languages and general software aspects for telecommunication systems

Printed in Switzerland
Geneva, 2021