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**Speech and multimedia Transmission Quality (STQ);
Transmission requirements for Superwideband/Fullband headset terminals from a QoS perspective as perceived by the user**

Reference
DTS/STQ-152-1

Keywords
QoS, terminal

ETSI

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ).

Introduction

Speech terminals are currently implementing narrowband and wideband bandwidth. Terminal equipment may offer wider bandwidth, due to features already available in these terminals. Such equipment may implement conversational features that may benefit of the electroacoustic equipment already available in the terminal and may provide wider quality for the end users.

The present document is intended to provide initial requirements and test methods for such type of equipment.

1 Scope

The present document provides speech & audio transmission performance requirements and measurement methods for headset functions of superwideband/fullband terminals. The present document provides requirements in order to optimize the end to end quality perceived by users.

Users become more sensitive to voice and music quality (for music used in conversational services) when using ICT/terminal equipment and so are more demanding for further enhancement especially further extension of the audio coded bandwidth.

For instance, this is the case for high quality conferencing services with music on hold, better background environment rendering and longer duration than normal point to point calls.

Standardized superwideband and fullband coders are now available, some being also compatible with wideband coders.

The present document will consider only conversational services (that may be mixed with other services) and does not cover the streaming-only services.

Such applications include:

- Speech and audio communication including conferencing.
- Bandwidth extension which may allow usage for some mixed content.
- Superwideband enhancement coupled with stereo/dichotic.

The send path it can be characterized in two ways:

- The signal picked up by microphone may combine speech, music and every type of environmental signal.
- Direct insertion of any type of signal.

For receive path, signal may be combine two types:

- Communication signals such as described for send path.
- Signal coming from distributed applications (e.g. advertisement, music on hold, etc.).

Handset terminals will not be within the scope of the present document.

2 References

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

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

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

2.1 Normative references

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

[1] Recommendation ITU-T P.501 Amendment 1: "Test signals for use in telephonometry".

[2] Recommendation ITU-T P.10/G.100: "Vocabulary for performance and quality of service".

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

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

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

[6] Recommendation ITU-T G 711-1 (annex D): "Wideband embedded extension for G.711 pulse code modulation".

[7] Recommendation ITU-T G.722.1 (annex C): "Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss".

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

[9] Recommendation ITU-T G.718 (annex B)": "Frame error robust narrow-band and wideband embedded variable bit-rate coding of speech and audio from 8-32 kbit/s".

[10] Recommendation ITU-T G.719: "Low-complexity, full-band audio coding for high-quality, conversational applications".

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

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

[13] ETSI TS 103 739: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband wireless terminals (handset and headset) from a QoS perspective as perceived by the user".

[14] Recommendation ITU-T P.863: "Perceptual objective listening quality assessment".

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

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

[17] Recommendation ITU-T P.800: "Methods for subjective determination of transmission quality".

[18] Recommendation ITU-T P.830: "Subjective performance assessment of telephone-band and wideband digital codecs".

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

[20] ISO 3: 1973: "Preferred numbers -- Series of preferred numbers".

[21] Recommendation ITU-T G.711.1 (annex F): "Wideband embedded extension for G.711 pulse code modulation".

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

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

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

2.2 Informative references

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

[i.1] ISO 532: "Acoustics -- Method for calculating loudness level".

3 Definitions and abbreviations

3.1 Definitions

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

binaural listening (definition found on Internet): both ears are involved for the perception of sound

dichotic (definition found on Internet): relating to or involving the presentation of a stimulus to one ear that differs in some respect (as pitch, loudness, frequency, or energy) from a stimulus presented to the other ear

diotic (definition found on Internet): pertaining to or affecting both ears (same signal in both ears)

dual channel mode: audio mode, in which two audio channels with independent programme contents (e.g. bilingual) are encoded within one audio bit stream

fullband bandwidth: 20 Hz - 20 kHz

stereo mode: audio mode in which two channels forming a stereo pair (left and right) are encoded within one bit stream and for which the coding process is the same as for the Dual channel mode

superwideband: covers at least mono and stereo capabilities

superwideband bandwidth: transmission of speech with a nominal pass-band wider than 100 - 7 000 Hz, usually understood to be 50 - 14 000 Hz (definition from Recommendation ITU-T P.10 /G.100 [2])

3.2 Abbreviations

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

ACR	Absolute Category Rating
DRP	ear Drum Reference Point
ERP	Ear reference Point
EVS	Enhanced Voice Services
FB	Fullband
GAT	Group Audio Terminal
HATS	Head and Torso Simulator
MCU	Multiplexing Control Unit
MRP	Mouth Reference Point
MS	Mid-sized Stereo
POI	Point of Interconnection
SLR	Send Loudness Rating
SWB	Superwideband
TCL	Terminal Echo Loss

4 Applications and coder considerations

4.1 Applications

The following applications are within the scope of the present document:

- Speech and audio communication including conferencing using high quality hands free systems, for which superwideband/fullband coding can better reproduce the audio environment and provide improved quality and audio immersion. These applications cover also GATs (Group Audio Terminals) and teleconference systems such as "Telepresence".

- Bandwidth extension which may allow usage for some mixed content applications where wider bandwidth could bring a significant added value for the customer (support of 14 kHz and 20 kHz bandwidth and stereo/multichannel capability).

- Superwideband enhancement coupled with stereo/multichannel to maximize the quality enhancement for the customer when the terminal device can support this capability.

The send path can be characterized in two ways:

- The signal picked up by microphone(s) may combine speech, music and every type of environmental signal.

NOTE: For some applications (e.g. journalist reporting) the user should have the possibility to cancel the noise environment or to transmit it without degradation.

- Direct insertion of any type of signal.

For receive path, signal may combine the two following types:

- Communication signal such as described for send path.

- Signal coming from distributed applications (e.g. advertisement, music on hold, etc.).

4.2 Coder considerations

As indicated in the scope only coders supporting conversational SWB and FB services are applicable to the present document.

4.2.1 Superwideband (SWB)

Table 4.1: Use cases for coders

Coder Reference	Speech	Other signals	Stereo	Remark
Recommendation ITU-T G.722.1 [7] annex C	X	X Music		For low frame loss
Recommendation ITU-T G.729.1 [8] annex E (extension SWB)	X	X background noise (X) Music		
Recommendation ITU-T G.718 [9] annex B	X	X Music		
Recommendation ITU-T G.711.1 annexes D [6] and F [21]	X	X	X (annex F)	
Recommendation ITU-T G.722 [19] annexes B and D	X	X	X (annex D)	
NOTE: G 722.1 [7] is intended to be used for hand-free application. It is referenced here considering that a terminal using this coder may implement a headset function.				

When X is in brackets, it means that the coder is not optimized for this application.

The following coders are recommended for Superwideband:

- Recommendation ITU-T G.722.1 [7] Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss. Annex C 14 kHz mode at 24, 32 and 48 kbit/s.
  The algorithm is recommended for use in hands-free applications such as conferencing where there is a low probability of frame loss. It may be used with speech or music inputs. The bit rate may be changed at any 20-ms frame boundary. New annex C contains the description of a low-complexity extension mode to G.722.1, which doubles the algorithm to permit 14-kHz audio bandwidth using a 32-kHz audio sample rate, at 24, 32, and 48 kbit/s.
  Annex C: this annex provides a description of the 14-kHz mode at 24, 32 and 48 kbit/s for this Recommendation.

- Recommendation ITU-T G.729.1 [8] annex E (extension SWB for G.729.1 [8]).
  This annex provides the high-level description of the higher bit-rate extension of G.729 designed to accommodate a wide range of input signals, such as speech, with background noise and even music.

- Recommendation ITU-T G.718 [9] annex B Superwideband scalable (extension for Recommendation ITU-T G.718 [9]). This annex describes a scalable superwideband (SWB, 50-14 000 Hz) speech and audio coding algorithm operating from 36 to 48 kbit/s and interoperable with Recommendation ITU-T G.718 [9].

- Recommendation ITU-T G.711.1 [6] annex D defines the superwideband extension
  Annex F defines the Stereo embedded extension for Recommendation ITU-T G.711.1 [6].
  "Annex F is intended as a stereo extension to the G.711.1 [6] wideband coding algorithm and its superwideband annex D. Compared to discrete two-channel (dual-mono) audio transmission, this stereo extension G.711.1 [6] annex F saves valuable bandwidth for stereo transmission. It is specified to offer the stereo capability while providing backward compatibility with the monaural core in an embedded scalable way. The annex provides very good quality for stereo speech contents (clean speech and noisy speech with various stereo sound pickup systems: binaural, MS, etc.), and for most of the conditions it provides significantly higher quality than low bitrate dual-mono. For some music contents, e.g. highly reverberated and/or with diffuse sound, the algorithm may have some performance limitations and may not perform as good as dual-mono codecs, however it achieves the quality of state-of-the-art parametric stereo codecs".

- Recommendation ITU-T G.722 [19] annex B defines the superwideband extension
  and annex D defines the Stereo embedded extension for Recommendation ITU-T G.722 [19].
  "Annex B describes a scalable superwideband (SWB, 50-14 000 Hz) speech and audio coding algorithm operating at 64, 80 and 96 kbit/s. The Recommendation ITU-T G.722 [19] superwideband extension codec is interoperable with Recommendation ITU-T G.722 [19]. The output of the Recommendation ITU-T G.722 [19] SWB coder has a bandwidth of 50-14000 Hz".
  "Annex D describes a stereo extension of the wideband codec G.722 and its superwideband extension, G.722 annex B. It is optimized for the transmission of stereo signals with limited additional bitrate, while keeping full compatibility with both codecs. Annex D operates from 64 to 128 kbit/s with four superwideband stereo bitrates at 80, 96, 112 and 128 kbit/s and two wideband stereo bitrates at 64 and 80 kbit/s".

NOTE: The potential future mobile coder EVS (Enhanced Voice Services) should be also considered when available. It will be relevant to reconsider the contents of the present document to consider the implications of the EVS coder implementation in terminals within the scope of the present document. EVS is designed for packet-switched networks/Mobile VoIP and VoLTE is a key target application. The key features of EVS are Superwideband speech (32 kHz sampling) with improved speech quality and improved music performance.

A future version of the present document will take into account this coder when available.

4.2.2 Fullband (FB)

The following coder is recommended for fullband:

- Recommendation ITU-T G.719 [10] Low-complexity, fullband audio coding for high-quality, conversational applications.
  "Recommendation ITU-T G.719 [10] describes the G.719 [10] coding algorithm for low-complexity fullband conversational speech and audio, operating from 32 kbit/s up to 128 kbit/s".

The encoder input and decoder output are sampled at 48 kHz. The codec enables fullbandwidth, from 20 Hz to 20 kHz, encoding of speech, music and general audio content. The codec operates on 20-ms frames and has an algorithmic delay of 40 ms.

NOTE: Amendment 1 adds new annex A that specifies the use of the ISO base media file format as container for the G.719 [10] bitstream addresses non-conversational use cases of the codec (e.g. call waiting music playback and recording of teleconferencing sessions, voice mail messages, online "jam"-sessions).

5 Test considerations

The terminals within the scope of the present document are not only dedicated to speech communication but are also mixing speech and audio contents and may implement stereo and multichannel transmissions. As a consequence there is a need to define new parameters, such as:

- **Loudness:** Loudness Rating is determined only for speech or speech-like signals. Loudness may be calculated over any types of signals (audio sequences, speech sequences and mix of these sequences). Moreover it is not intended to define Loudness Rating algorithms for Superwideband and fullband speech. To be consistent with transmission planning, the loudness rating shall be determined using wideband calculation and loudness shall be measurement for all the bandwidths. Clause 5.4.1.2 details the measurement principles.

- **Binaural listening:** The most of the test assessment methods and requirements for speech terminals are based on monaural listening, Even if some of them (e.g. for Handsfree Loudness rating) are intended to take into account binaural listening, the basic methods and requirements are only taking into account correction factors. The plan is to adapt test methods to effective binaural listening.

As a consequence, the present document takes into account test arrangements that are defined for speech terminals or for audio equipment.

Recommendation ITU-T P 58 [3] give informations about use of HATS only from 100 Hz to 10 kHz, but new designs offer wider bandwidths.

For send the HATS can be used between 50 Hz and 16 kHz. Until the development of new systems with larger bandwith, send measurement will be limited to those frequencies.

NOTE 1: With some measurement equipment the use of such of bandwidth is not possible and should be limited to 100 Hz to 14 kHz.

For receive, a correction factor (given, in annex B) allows measurement at DRP until 16 kHz.

NOTE 2: It is not the intention of the present document to define new requirements to adapt HATS for superwideband and fullband. However when terminals implement Superwideband or Fullband within terminals support also WideBand and/or NarrowBand speech, it is intended to use as far as possible test methods defined for wideband terminals and consequently to use HATS for parameters measured in wideband bandwidth.

5.1 Test Setup

5.1.1 Setup for terminals

The preferred acoustical access to terminals is the most realistic simulation of the "average" subscriber. This can be made by using Head And Torso Simulator (HATS) with appropriate ear simulation and appropriate means to fix handset and headset terminals in a realistic and reproducible way to the HATS. HATS is described in Recommendation ITU-T P.58 [3], appropriate ears are described in Recommendation ITU-T P.57 [22] (type 3.3 and type 3.4 ear), a proper positioning of handsets under realistic conditions is to be found in Recommendation ITU-T P.64 [23].

The preferred way of testing a terminal is to connect it to a network simulator with exact defined settings and access points. The test sequences are fed in either electrically, using a reference codec or using the direct signal processing approach or acoustically using ITU-T specified devices.

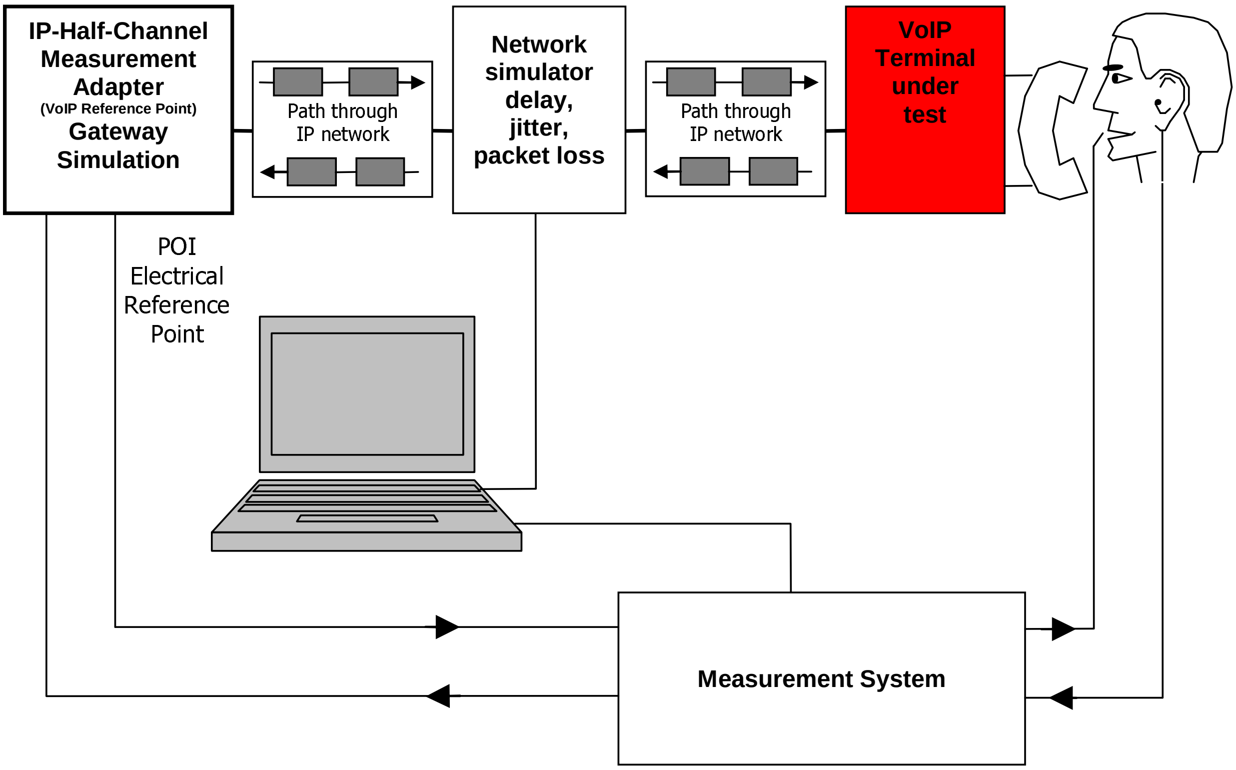


Figure 5.1: Half channel terminal measurement

5.1.2 Setup for headsets

The artificial ear shall be conform with Recommendation ITU-T P.57 [22], type 3.3 or type 3.4 ears shall be used.

Recommendations for positioning headsets are given in Recommendation ITU-T P.380 [15]. If not stated otherwise headsets shall be placed in their recommended wearing position. Further information about setup and the use of HATS can be found in Recommendation ITU-T P.380 [15].

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

5.1.3 Position and calibration of HATS

All the send and receive characteristics shall be tested with the HATS, it shall be indicated what type of ear was used.

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

The HATS shall be equipped with two type 3.3 or type 3.4 artificial ears. For binaural headsets two artificial ears are required. The type 3.3 or type 3.4 artificial ears as specified in Recommendation ITU-T P.57 [22] shall be used. The artificial ear shall be positioned on HATS according to Recommendation ITU-T P.58 [3].

The exact calibration and equalization can be found in Recommendation ITU-T P.581 [4].

For calibration of mouth, equalization has to be limited between 50 Hz and 16 kHz.

NOTE: With some measurement equipment the use of such of bandwidth is not possible and shall be limitated to 100 Hz - 14 kHz.

For receive if not stated otherwise, the HATS shall be corrected using the correction factor given in annex A.

5.1.4 Test signal

The test signals are defined according to Recommendation ITU-T P.501 [1].

As the bandwidth of the speech signals defined in Recommendation ITU-T P.501 [1] is fullband, these test signals shall be used in the present document:

- The test signal to be used for measurements such as Frequency response, Loudness Rating, shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [1].
- The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [1], shall be used as activation signal for measurements such as distortion, send noise.
- The compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 Amendment 1 [1], shall be used for measurements such as TCLw.

For double-talk performance:

- A "double-talk" sequence representing typical double talk scenarios in real conversations is shown in figure 63.4. This uses the single-talk sequence described in section 7.3.1 of Recommendation ITU-T P.501 [1], shown in the lower pane, as the main speech and an additional competing speaker sequence, shown in the upper pane.

5.1.5 Test signal levels

The level dependency should be considered and consequently tests should also be done with signal levels lower and higher than the reference level defined in clauses 5.1.5.1 and 5.1.5.2.

5.1.5.1 Send

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

5.1.5.2 Receive

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

5.1.6 Setup of background noise simulation

A setup for simulating realistic background noises in a lab-type environment is described in ES 202 396-1 [11].

The signals attached to ES 202 396-1 [11] are fullband signals and should be used for background noise simulation.

5.1.7 Acoustic environment

NOTE: The acoustic environment may influence more significantly the results in low and high frequencies. It should be adapted to the terminal bandwidth.

In general two possible approaches need to be taken into account: either room noise and background noise are an inherent part of the test environment or room noise and background noise shall be eliminated to such an extent that their influence on the test results can be neglected.

Unless stated otherwise measurements shall be conducted under quiet and "anechoic" conditions.

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

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

5.1.8 Influence of terminal delay issue for measurements

As delay is introduced by the terminal, care shall be taken for all measurement using an activation signal. It shall be checked that the test is performed on the test signal and not on the activation signal.

5.2 Environmental conditions for tests

The following conditions shall apply for the testing environment:

a) Ambient temperature: 15 °C to 35 °C (inclusive);

b) Relative humidity: 5 % to 85 %;

c) Air pressure: 86 kPa to 106 kPa (860 mbar to 1 060 mbar).

d) Unless specified otherwise, the background noise level shall be less than -64 dBPa(A) in conjunction with NC30 (ISO 3745 [24]).
For specified tests, it is desirable to have a background noise level of less than -74 dBPa(A) in conjunction with NC20, but the background noise level of -64 dBPa(A) in conjunction with NC30 shall never be exceeded.

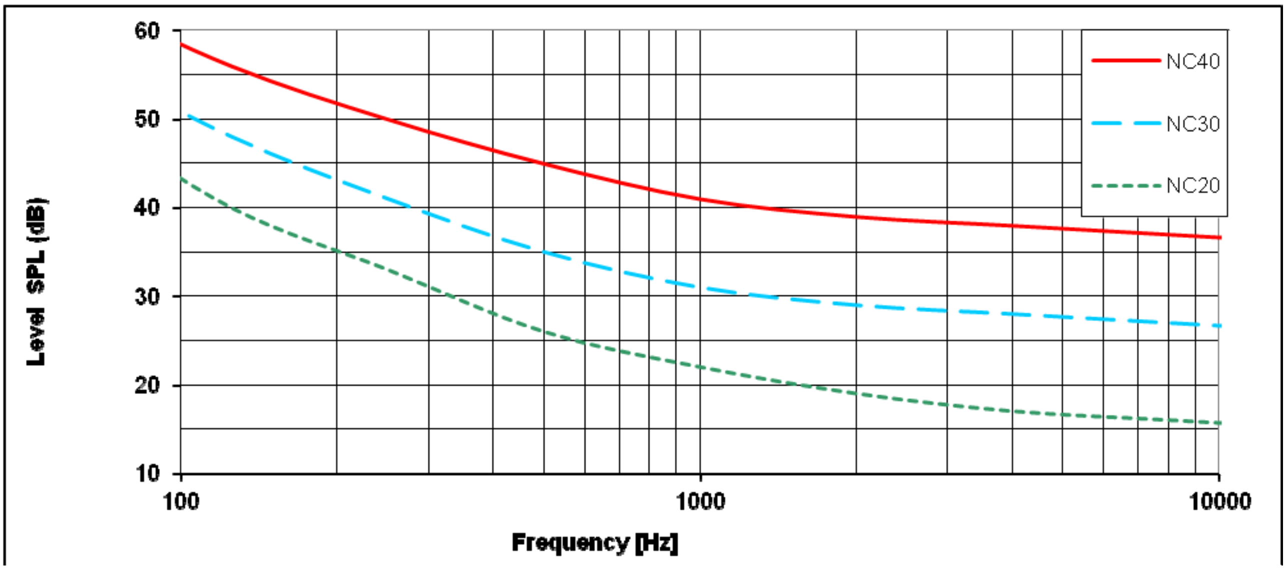


Figure 5.2: NC-criteria for test environment

5.3 Accuracy of measurements and test signal generation

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

Table 5.1: Measurement Accuracy

Item	Accuracy
Electrical signal level	±0,2 dB for levels ≥ -50 dBV ±0,4 dB for levels < -50 dBV
Sound pressure	±0,7 dB
Frequency	±0,2 %
Time	±0,2 %

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

Table 5.2: Accuracy of test signal generation

Quantity	Accuracy
Sound pressure level	+0/-6 dB for frequencies from 50 Hz 100 Hz ±1 dB for frequencies from 100 Hz to 8 000 Hz ±3 dB for frequencies from 8 000 Hz to 16 000 Hz
Electrical excitation levels	±0,4 dB across the whole frequency range
Frequency generation	±2 %
Time	±0,2 %
Specified component values	±1 %
NOTE: This tolerance may be used to avoid measurements at critical frequencies, e.g. those due to sampling operations within the terminal under test.	

NOTE: With some measurement equipment the use of such of bandwidth is not possible and should be limited to 100 Hz- 14 kHz.

For terminal equipment which is directly powered from the mains supply, all tests shall be carried out within ±5 % of the rated voltage of that supply. If the equipment is powered by other means and those means are not supplied as part of the apparatus, all tests shall be carried out within the power supply limit declared by the supplier. If the power supply is a.c. the test shall be conducted within ±4 % of the rated frequency.

5.4 Specific test considerations

Even if the present document is dedicated to conversational services, the signals that are transmitted may combine speech and audio.

5.4.1 Loudness rating and Loudness

5.4.1.1 Loudness Rating

Loudness Rating, as defined in Recommendation ITU-T P.79 [5], applies for narrowband and wideband and is specific to telecommunications transmission systems. So, when terminals implement wideband speech or are intended to communicate with wideband or narrowband terminals, the terminals shall be calibrated for SLR and RLR values.

Due to the current bandwidth limitation of loudness rating's calculation it is not possible to calculate superwidband of fullband loudness ratings.

NOTE: For RLR and SLR, values are similar or derived from those defined in ES 202 739 [12] and TS 103 739 [13].

5.4.1.2 Loudness

Loudness is quantifies the level perceived by the user and should be more relevant when the signal combines speech and audio sequences. ISO 532 [i.1] method B defines a standardized way to determine the loudness of a steady-state complex signal.

This assessment method takes into account the level, the spectrum of the signals and takes into account binaural listening. Loudness may be calculated for any type of signal (speech, music, noise) and mixed signals.

Standardized audio and speech signals (possibly based on combination of sequences defined in Recommendation ITU-T P.501 [1] and in ES 202 396-1 [11].

When the terminal provides superwideband or fullband in addition with wideband or narrowband, the reference loudness value (expressed in phones) shall be determined for Narrowband or Wideband transmission.

The loudness measured in superwideband or fullband should be equal and preferably higher than the loudness value measured for narrowband or wideband.

If the superwideband and fullband terminal does not support wideband transmission, standardized loudness levels have to be defined (for further study).

5.4.2 Binaural listening

The scope of the present document includes terminals that may have two earpieces, distant sound pick-up using two or more microphones.

The terminal may also provide stereo listening or binaural rendering built from MCU.

So it should be relevant to consider binaural listening (for further study).

6 Requirements considerations and associated Measurement Methodologies

When possible, parameter requirements will be derived from requirements defined for the wideband terminals. The recommended test method is also provided in the same clause as requirements.

6.1 Send parameters

6.1.1 Send Frequency response

Requirement

The send frequency response of the headset shall be within a mask as defined in table 6.1 for SWB and table 6.2 for FB, and shown in figure 6.1 for SWB and figure 6.2 for FB. This mask shall be applicable for all types of headsets.

Table 6.1: Superwideband send frequency response limits

Frequency	Upper Limit	Lower Limit
100 Hz	5 dB	-5 dB
12 500 Hz	5 dB	-5 dB
14 000 Hz	5 dB	-10 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale. The requirement is based on 1/12th octave measurement.		

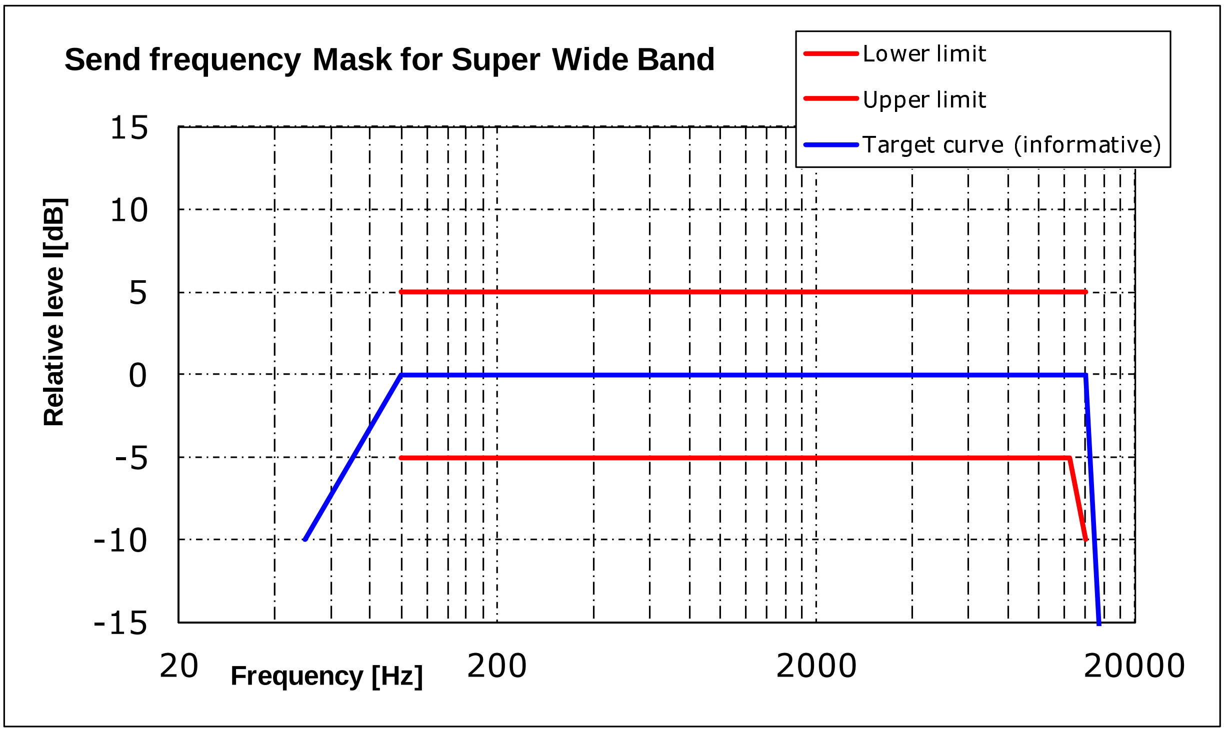


Figure 6.1: Send frequency response mask for superwideband

Table 6.2: Fullband send frequency response limits

Frequency	Upper Limit	Lower Limit
100 Hz	5 dB	-5 dB
12 500 Hz	5 dB	-5 dB
14 000 Hz	5 dB	-5 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale. The requirement is based on 1/12th octave measurement.		

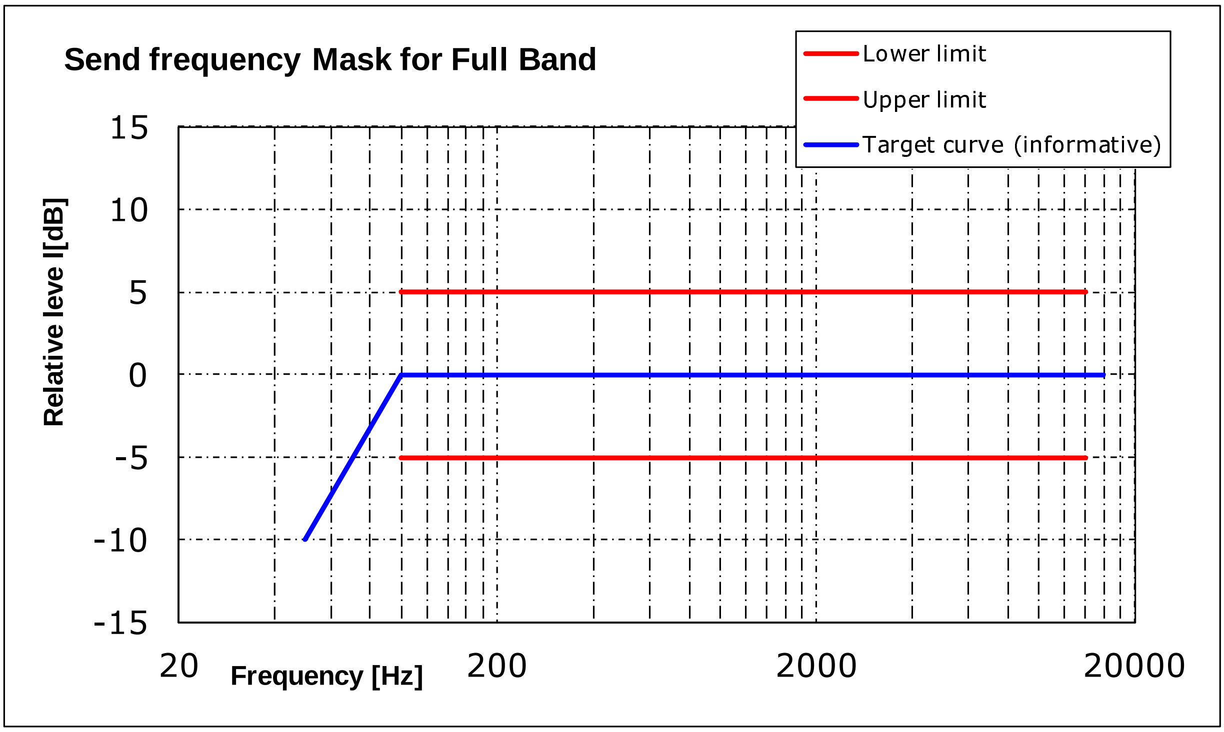


Figure 6.2: Send frequency response mask for Fullband

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

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

Measurement Method

The test signal is defined in clause 5.1.4. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa.
The headset terminal is setup as described in clause 5.1.

The tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [15]. The results are averaged (averaged value in dB, for each frequency).

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [20] for frequencies from 100 Hz to 14 kHz inclusive. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa.

6.1.2 Send Loudness Rating

Requirement:

The nominal value of Send Loudness Rating (SLR) shall be:

- SLR(set) = 8 dB ± 3 dB

Measurement Method:

The test signal to be used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 Amendment 1 [1] shall be used. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The headset terminal is setup as described in clause 5.1.

The tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [15]. The results are averaged (averaged value in dB, for each frequency).

The sending sensitivity shall be calculated from each band of the 20 frequencies given in table 1 of Recommendation ITU-T P.79 [5], bands 1 to 20. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to Recommendation ITU-T P.79 [5] see annex A.

6.1.3 Level dependency for SLR

Requirement:

For further study.

Measurement Method:

The loudness/loudness ratings are tested for different input levels (at least the nominal signal level, a 10 dB lower and a 5 dB higher).

Same method as SLR with adapted level.

6.1.4 Send Distortion

6.1.4.1 Signal to harmonic distortion versus frequency

Requirement:

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

Table 6.3: Send distortion for superwideband

Frequency	Ratio
100 Hz	24 dB
200 Hz	26 dB
400 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
3,15 kHz	30 dB
5 kHz	30 dB
NOTE: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

Table 6.4: Send distortion for fullband

Frequency	Ratio
100 Hz	26 dB
200 Hz	30 dB
400 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
3,15 kHz	30 dB
8 kHz	30 dB
NOTE: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

Measurement method:

The headset terminal is set-up as described in clause 5.1.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 100 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz, 3 150 Hz and 7 000 Hz for superwideband and 100 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz, 3 150 Hz and 8 000 Hz for fullband. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be calibrated to -4,7 dBPa at the MRP.

The signal to harmonic distortion ratio is measured selectively up to 14 kHz for superwideband and 20 kHz for fullband.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [1] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

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

6.1.4.2 Signal to harmonic distortion for higher input level

Requirement:

For the signal defined in the measurement method, the signal to harmonic distortion ratio shall be ≥ 30 dB.

Measurement method:

The headset terminal is set-up as described in clause 5.1.

The signal used is an activation signal followed by a 1 kHz sine wave. The signal to harmonic distortion ratio is measured selectively up to 14 kHz for Superwideband terminals and up to 20 kHz for fullband terminal.

The duration of the sine wave shall be ≤ 1 s. The sinusoidal signal level shall be calibrated to +10 dBPa at the MRP.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [1] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

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

6.1.5 Send Noise

Requirement:

The maximum noise level produced by the VoIP terminal at the POI under silent conditions in the sending direction shall not exceed -68 dBm0 (A).

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

Measurement Method:

For the actual measurement no test signal is used. In order to reliably activate the terminal an activation signal is introduced before the actual measurement.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [1] shall be used for activation. The level of this activation signal is -4,7 dBPa at the MRP.

The headset terminal is set-up as described in clause 5.1.

The send noise is measured at the POI in the frequency range from 50 Hz to 14 kHz for Superwideband and 20 Hz to 20 kHz for Fullband. The analysis window is applied directly after stopping the activation signal but taking into account the influence of all acoustical components (reverberations). The averaging time is 1 s. The test house has to ensure (e.g. by monitoring the time signal) that during the test the terminal remains in activated condition. If the terminal is deactivated during the measurement, the measurement time has to be reduced to the period where the terminal remains in activated condition.

The noise level is measured in dBm0(A).

6.2 Receive parameter

6.2.1 Equalization

This type of terminal may be used for reproduction of signals other than pure speech (e.g. music) for which user's preference may be different in term of sound signature.

So the terminal may implement an equalization function adjusting frequency response according to user's preferences.

When such a function is available it is necessary that receive frequency response be conform to requirement of clause 6.2.2 from the present document, for at least one setting.

For all settings conformance with other parameters of the present document shall be ensured.

6.2.2 Receive Frequency response

Requirement:

The receive frequency response of the headset shall be within a mask as defined in table 6.5 and shown in figure 6.3.

NOTE 1: For the time being, the measurement method defined (with correction factor given in annex B) being only valid until 16 kHz, the requirements for superwideband and for fullband are identical.

Table 6.5: Receive Frequency Response limits

Frequency	Upper Limit	Lower Limit
50 Hz	3 dB	-5 dB
400 Hz	3 dB	-5 dB
1010 Hz	*	-5 dB
1 200 Hz	*	-8 dB
1 500 Hz	*	-8 dB
2 000 Hz	9 dB	-3 dB
3 200 Hz	9 dB	-3 dB
14 000 Hz	9 dB	-13 dB
NOTE: * The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale. It is a floating or "best fit" mask. The requirement is based on 1/12th octave measurement.		

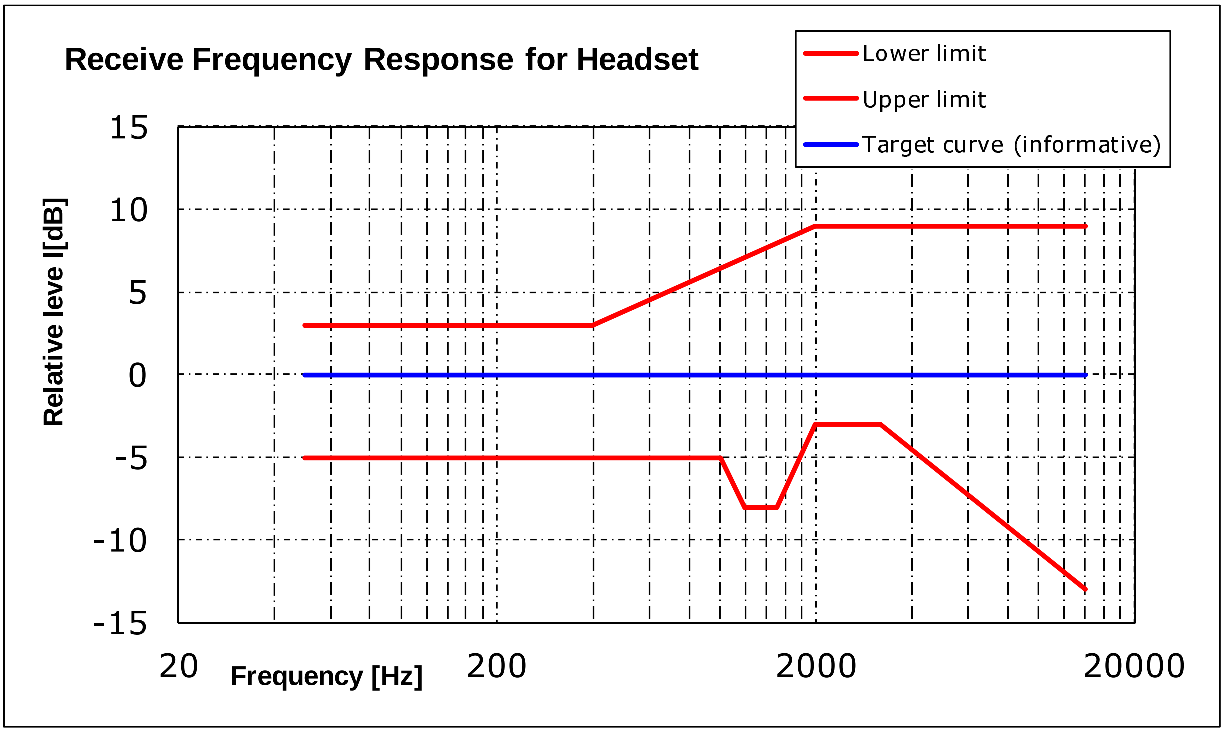


Figure 6.3: Receive frequency response mask for Superwideband

NOTE 2: This requirement applies to headphones not primarily designed for superwideband communication but rather for music audition. It is the reason of rather open limits. In the next future, new limits will be discussed to apply when specially designed for superwideband headphones will be available.

Measurement Method:

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

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

S_{Jeff} Receive Sensitivity; Junction to HATS Ear with free field correction.

pe_{ff} DRP Sound pressure measured by ear simulator Measurement data are converted from the Drum Reference Point to free field.

v_{RCV} Equivalent RMS input voltage.

The test signal to be used for the measurements is defined in clause 5.1.4.

The headset terminal is setup as described in clause 5.1.

The sound pressure level is measured at the DRP of the HATS for each $1/12^{th}$ octave band.

The tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [15]. The results are averaged (averaged value in dB, for each frequency).

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [20] for frequencies from 50 Hz to 16 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The obtained response curve is corrected by the correction factor given in annex B.

The sensitivity is expressed in terms of dBPa/V.

6.2.3 Receive Loudness Rating (monaural reproduction)

Requirement:

When terminal implements Wideband speech functions or when the superwideband/fullband functions may interact with wideband terminals, the headset terminal shall fulfil the requirements on RLR as defined in ES 202 739 [12], clause 7.1.7.

Measurement Method:

The test signal to be used for the measurements shall be British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [1] shall be used. The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

The headset terminal is setup as described in clause 5.1.

The tests are repeated 5 times, in conformance with Recommendation ITU-T P.380 [15]. The results are averaged (averaged value in dB, for each frequency).

The receiving sensitivity shall be calculated from each band of the 20 frequencies given in table 1 of Recommendation ITU-T P.79 [5], bands 1 to 20. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to Recommendation ITU-T P.79 [5], see annex A. No leakage correction shall be applied for the measurement.

6.2.4 RLR for stereo/dichotic

For further study.

6.2.5 Loudness

For further study.

6.2.6 Receive Distortion

Requirement:

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

Table 6.6: Receive distortion for superwideband

Frequency	Signal to distortion ratio limit, receiving
100 Hz	24 dB
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
3,15 kHz	30 dB
5 kHz	30 dB
NOTE: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

Table 6.7: Receive distortion for fullband

Frequency	Signal to distortion ratio limit, receiving
50 Hz	24 dB
100 Hz	26 dB
315 Hz	30 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
3,15 kHz	30 dB
8 kHz	30 dB
NOTE: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

Measurement Method:

The headset terminal is positioned as described in clause 5.1.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 100 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz, 3 150 Hz and 7 000 Hz for superwideband and 50 Hz, 100 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz, 3 150 Hz and 8 000 Hz for fullband.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [1] shall be used for activation. The signal level shall be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 14 kHz for superwideband and 20 kHz for fullband.

The ratio of signal to harmonic distortion shall be measured at the DRP of the artificial ear with a correction by the curve of reference microphone.

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

6.2.7 Minimum activation level and sensitivity in Receive direction

For further study.

6.2.8 Receive Noise

Requirement:

Telephone sets with adjustable receive levels shall be adjusted so that the RLR is as close as possible to the nominal RLR.

The receive noise shall be less than -57 dBPa(A).

Where a volume control is provided, the measured noise shall not be greater than -54 dBPa(A) at the maximum setting of the volume control.

Measurement Method:

The headset terminal is setup as described in clause 5.1.

The female speaker signal of the short conditioning sequence described in clause 7.3.7 of Recommendation ITU-T P.501 [1] shall be used for activation. The activation signal level shall be -16 dBm0.

The measurement shall be A-weighted.

The noise shall be measured at the DRP of the artificial ear with a correction by the curve of reference microphone.

6.2.9 Automatic Gain Control in Receiving

For further study.

6.3 Other parameters

6.3.1 Sidetone Masking Rating STMR (Mouth to ear)

Requirement:

The STMR shall be 16 dB ± 4 dB for nominal setting of the volume control.

For all other positions of the volume control, the STMR shall not be below 8 dB.

NOTE 1: It is preferable to have a constant STMR independent of the volume control setting.

NOTE 2: STMR measurement in P.79 [5] is not defined above 8 kHz, but sidetone signal is not supposed to have such limitation.

Measurement Method:

The test signal is defined in 5.1.4. The test signal level shall be -4,7 dBPa, measured at the MRP. The headset terminal is setup as described in clause 5.1.

Where a user operated volume control is provided, the measurements shall be carried out the nominal setting of the volume control. In addition the measurement is repeated at the maximum volume control setting.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [20] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band (Recommendation ITU-T P.79 [5] table 3, bands 1 to 20) is referred to the averaged test signal level measured in each frequency band.

The Sidetone path loss (LmeST), as expressed in dB, and the SideTone Masking Rate (STMR) (in dB) shall be calculated from the formula 5-1 of Recommendation ITU-T P.79 [5], using m = 0,225 and the weighting factors of in table 3 of Recommendation ITU-T P.79 [5].

6.3.2 Sidetone delay

Requirement:

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

Measurement Method:

The headset terminal is setup as described in clause 5.1.

The test signal is a CS-signal complying with Recommendation ITU-T P.501 [1] using a PN sequence with a length of 4 096 points (for the 48 kHz sampling rate) which equals to the period T. The duration of the complete test signal is as specified in Recommendation ITU-T P.501 [1].

The level of the signal shall be -4,7 dBPa at the MRP.

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

$$\Phi_{xy}(\tau) = \lim_{T\to\infty} \sum_{t=-T/2}^{T/2} S_x(t) S_y(t+\tau) \qquad (1)$$

The measurement window T shall be exactly identical with the time period T of the test signal, the measurement window is positioned to the pn-sequence of the test signal.

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

$$H\{xy(\tau)\} = \sum_{-\infty}^{\infty} \frac{\Phi_{xy}(u)}{\Pi(\tau - u)} \quad (2)$$

$$E() = \sqrt{[\Phi xy(\tau)]^2 + \{H[\Phi \ xy(\tau)]\}^2} \quad (3)$$

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

6.3.3 Stability loss

Requirement

With the headset lying on and the transducers facing a hard surface, the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range of 50 Hz to 16 kHz for superwideband and 20 to 20 kHz for Fullband. The requirement applies for the closest possible position between microphone and headset receiver.

NOTE: Depending on the type of headset it may be necessary to repeat the measurement in different positions.

Measurement method:

Before the actual test a training sequence of the British-English single talk sequence described in clause 7.3.2 of Recommendation ITU-T P.501 [1] is applied. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with Recommendation ITU-T P.501 [1] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 250 ms. With an input signal of -3 dBm0, the attenuation from digital input to digital output shall be measured for frequencies from 100 Hz to 8 kHz under the following conditions:

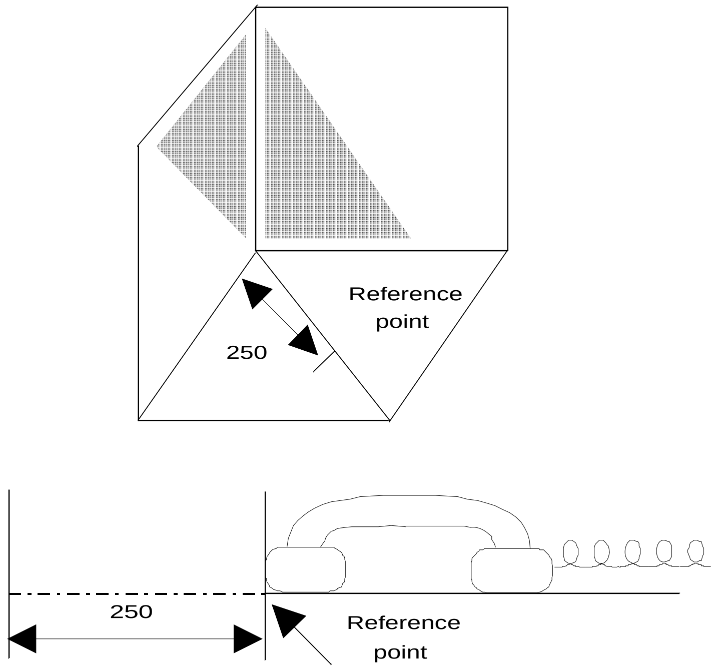
a) the headset, with the transmission circuit fully active, shall be positioned on one inside surface that is of three perpendicular planes, smooth, hard surfaces forming a corner. Each surface shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line, extending from the corner formed by the three surfaces, and a reference position 250 mm from the corner, as shown in figure 4;

b) the headset, with the transmission circuit fully active, shall be positioned on the defined surface as follows:

   1) the microphone and the receiver shall face towards the surface;

   2) the headset receiver shall be placed centrally at the reference point as shown in figure 6.4;

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



NOTE: All dimensions in mm.

Figure 6.4

6.3.4 Round-trip Delay

The round trip delay includes send delay plus receive delay. A maximum acceptable value has to be defined in the next release of the present document.

Codec delay is not included.

For conversational services, the delay shall be kept as small as possible to ensure a good quality and in particular a high level of interactivity between the users.

6.3.5 Terminal Echo Loss

Requirement:

The TCL measured as unweighted Echo Loss shall be ≥ 46 dB for all positions of the volume control (if supplied).

Measurement method:

The headset terminal is setup as described in clause 5.1. The ambient noise level shall be < -64 dBPa(A). The attenuation from electrical reference point input to electrical reference point output shall be measured using the compressed real speech signal described in clause 7.3.3 of Recommendation ITU-T P.501 Amendment 1 [1].

TCL is calculated as unweighted echo loss from 100 Hz to 8 kHz. For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The first 17,0 s of the test signal (6 sentences) are discarded from the analysis to allow for convergence of the acoustic echo canceller. The analysis is performed over the remaining length of the test sequence (last 6 sentences).

6.3.6 Objective listening quality

For further study"Recommendation ITU-T P.863 [14] describes an objective method for predicting overall listening speech quality from narrowband (300 Hz to 3 400 Hz) to superwideband (50 Hz to 14 000 Hz) telecommunication scenarios as perceived by the user in an Recommendation ITU-T P.800 [17] or Recommendation ITU-T P.830 [18] ACR listening only test".

NOTE: Particular attention has to be taken for the choice of appropriate test sequence.

6.3.7 Double talk performance

To assess double talk performance, the signals to be used are defined in Recommendation ITU-T P.501 [1]:
A "double-talk" sequence representing typical double talk scenarios in real conversations is shown in figure 63.4. This uses the single-talk sequence described in section 7.3.1 of Recommendation ITU-T P.501 [1], shown in the lower pane, as the main speech and an additional competing speaker sequence, shown in the upper pane.

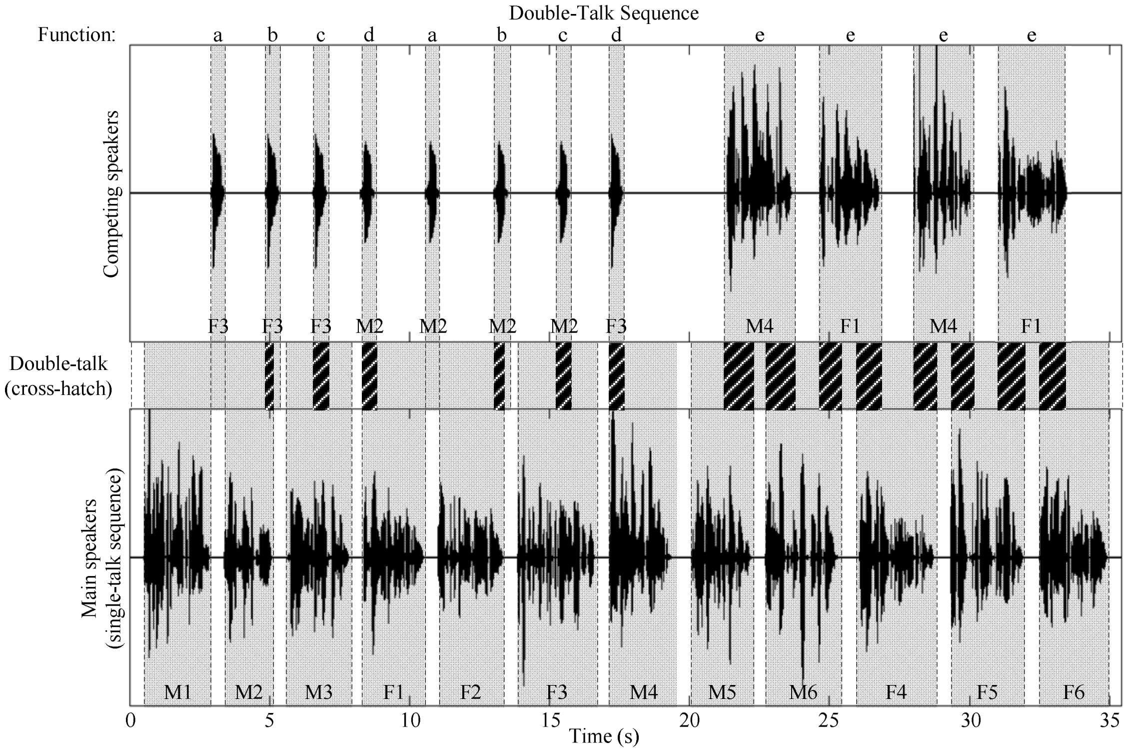


Figure 6.5: Double-talk test sequence using the single-talk sequence and competing speech serving different functions (a - e). Cross-hatched areas between the upper and lower panes show periods of double talk.

The competing-speaker sequence includes single words (the word "five") spoken by speakers F3 and M2 during the first half of the sequence followed by full sentences by speakers F1 and M4 during the second half of the sequence. No speaker is competing with themselves during the sequence.

The competing samples serve different double-talk functions, defined as functions "a" to "e" above the upper pane of figure 6.5. The functions are:

a) competing word within a speech pause;

b) competing word partially masked;

c) competing word fully masked within a sentence;

d) competing word fully masked coincident with the start of a sentence;

e) sentence masking another sentence.

These are meant to represent possible double-talk situations in normal conversation. The area between the upper and lower pane of figure 6.5 shows the periods during which double-talk happens as cross-hatched patches. The competing sequence can be used either as a send signal or a receive signal in testing.

6.3.8 Speech and audio quality in presence of noise

For further study.

Annex A (normative): Correction factor used for measuring superwideband headset using HATS

A correction factor has to be added to the acoustic level measured at the DRP of the 3.3 or 3.4 ear.

This correction factor is given for 1/3rd octave measurement and for 1/12th octave measurements.

For 1/12th octave measurements, two tables are given, corresponding to two sets of frequencies.

The first one are the 1/12 octave centre frequencies specified in IEC 61260 [16], the second one are center frequencies corresponding to 1/12th octave intervals as given by the R.40 series of preferred numbers in ISO 3 [20].

A.1 1/3rd octave correction

Table A.1: 1/3rd octave correction

Freq. Hz	Rep. dB	Freq. Hz	Rep. dB	Freq. Hz	Rep. dB	Freq. Hz	Rep. dB
16	0	100	0	630	2	4 000	12,66
20	0	125	0	800	4	5 000	9,94
25	0	160	0	1 000	5	6 300	5,59
31	0	200	0	1 250	6,5	8 000	8,77
40	0	250	0,5	1 600	8	10 000	6,53
50	0	315	0,5	2 000	11,78	12 500	-0,20
63	0	400	1	2 500	14,06	16 000	-0,52
80	0	500	1,5	3 150	13,44		

A.2 1/12th octave correction

Table A.2: 1/12th octave correction using centre frequencies specified in IEC 61260 [16]

Freq. Hz	Rep. dB	Freq. Hz	Rep. dB	Freq. Hz	Rep. dB	Freq. Hz	Rep. dB	Freq. Hz	Rep. dB	Freq. Hz	Rep. dB
19	0	61,3	0	193	0	613	1,94	1 939	11,17	6 131	5,5
20,5	0	64,9	0	205	0,06	649	2,25	2 053	12,06	6 494	4,95
21,8	0	68,8	0	218	0,19	688	2,74	2 175	12,91	6 879	4,9
23	0	72,9	0	230	0,31	729	3,22	2 304	13,61	7 286	5,35
24,4	0	77,2	0	244	0,45	772	3,7	2 441	14,05	7 718	6,57
25,9	0	81,8	0	259	0,5	818	4,1	2 585	14,18	8 175	9,26
27,4	0	86,6	0	274	0,5	866	4,36	2 738	14,07	8 659	11,14
29	0	92	0	290	0,5	917	4,61	2 901	13,79	9 170	9,55
30,7	0	97	0	307	0,5	972	4,87	3 073	13,46	9 720	6,13
32,5	0	103	0	325	0,57	1 029	5,19	3 255	13,18	10 290	3,45
34,5	0	109	0	345	0,69	1 090	5,58	3 447	12,97	10 900	1,75
36,5	0	115	0	365	0,81	1 155	5,97	3 652	12,84	11 550	0,56
38,7	0	122	0	387	0,93	1 223	6,35	3 868	12,73	12 230	-0,46
41	0	130	0	410	1,06	1 296	6,72	4 097	12,55	12 960	-1,3
43,4	0	137	0	434	1,18	1 372	7,07	4 340	12,16	13 720	-1,44
46	0	145	0	460	1,31	1 454	7,42	4 597	11,4	14 540	0,87
48,7	0	154	0	487	1,44	1 540	7,77	4 870	10,31	15 400	2,71
51,6	0	163	0	516	1,57	1 631	8,55	5 158	9,02		
54,6	0	173	0	546	1,69	1 728	9,39	5 464	7,67		
57,9	0	183	0	579	1,82	1 830	10,23	5 788	6,45		

Table A.3: 1/12th octave correction using center frequencies corresponding to 1/12th octave intervals as given by the R.40 series of preferred numbers in ISO 3 [20]

Freq. Hz	Rep. dB	Freq. Hz	Rep. dB	Freq. Hz	Rep. dB	Freq. Hz	Rep. dB	Freq. Hz	Rep. dB	Freq. Hz	Rep. dB
19	0	60	0	190	0	600	1,89	1 900	10,84	6 000	5,86
20	0	63	0	200	0,03	630	2,09	2 000	11,65	6 300	5,24
21,2	0	67	0	212	0,13	670	2,52	2 120	12,54	6 700	4,92
22,4	0	71	0	224	0,25	710	3,00	2 240	13,27	7 100	5,15
23,6	0	75	0	236	0,37	750	3,46	2 360	13,79	7 500	5,96
25	0	80	0	250	0,47	800	3,95	2 500	14,10	8 000	8,25
26,5	0	85	0	265	0,50	850	4,27	2 650	14,14	8 500	10,54
28	0	90	0	280	0,50	900	4,53	2 800	13,96	9 000	10,07
30	0	95	0	300	0,50	950	4,77	3 000	13,60	9 500	7,47
31,5	0	100	0	315	0,53	1 000	5,03	3 150	13,34	10 000	4,79
33,5	0	106	0	335	0,63	1 060	5,39	3 350	13,08	10 600	2,57
35,5	0	112	0	355	0,75	1 120	5,76	3 550	12,90	11 200	1,19
37,5	0	118	0	375	0,86	1 180	6,11	3 750	12,79	11 800	0,18
40	0	125	0	400	1,00	1 250	6,49	4 000	12,63	12 500	-0,77
42,5	0	132	0	425	1,14	1 320	6,83	4 250	12,30	13 200	-1,34
45	0	140	0	450	1,26	1 400	7,19	4 500	11,68	14 000	-0,64
47,5	0	150	0	475	1,39	1 500	7,61	4 750	10,79	15 000	1,87
50	0	160	0	500	1,50	1 600	8,29	5 000	9,72		
53	0	170	0	530	1,63	1 700	9,15	5 300	8,38		
56	0	180	0	560	1,75	1 800	9,9904	5 600	7,15		

History

Document history		
V1.1.1	March 2013	Publication