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Foreword

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The present document may refer to technical specifications or reports using their 3GPP identities, UMTS identities or GSM identities. These should be interpreted as being references to the corresponding ETSI deliverables.

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Contents

Foreword

This Technical Specification has been produced by the 3GPP.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of this TS, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

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y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the specification.

Introduction

The present document specifies test methods to allow the minimum performance requirements for the acoustic characteristics of GSM and 3G terminals when used to provide narrow-band or wideband telephony to be assessed.

The objective for narrow-band services is to reach a quality as close as possible to ITU-T standards for PSTN circuits. However, due to technical and economic factors, there cannot be full compliance with the general characteristics of international telephone connections and circuits recommended by the ITU-T.

The performance requirements are specified in TS 26.131; the test methods and considerations are specified in the main body of the text.

1 Scope

The present document is applicable to any terminal capable of supporting narrow-band or wideband telephony, either as a stand-alone service or as the telephony component of a multimedia service. The present document specifies test methods to allow the minimum performance requirements for the acoustic characteristics of GSM and 3G terminals when used to provide narrow-band or wideband telephony to be assessed.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TS 26.131: "Terminal Acoustic Characteristics for Telephony; Requirements".

[2] ITU-T Recommendation B.12 (1988): "Use of the decibel and the neper in telecommunications".

[3] ITU-T Recommendation G.103 (1998): "Hypothetical reference connections".

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

[5] ITU-T Recommendation G.121 (1993): "Loudness ratings (LRs) of national systems".

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

[7] ITU-T Recommendation G.711 1988): "Pulse code modulation (PCM) of voice frequencies".

[8] ITU-T Recommendation P.11 (1993): "Effect of transmission impairments".

[9] ITU-T Recommendation P.38 (1993): "Transmission characteristics of operator telephone systems (OTS)".

[10] ITU-T Recommendation P.50 (1993): "Artificial voices".

[11] 3GPP TS 03.58 (Release 1997): "Digital Cellular Telecommunications System (Phase 2+) Characterization test methods and quality assessment for hands-free mobiles".

[12] IEC Publication 60651: 'Sound Level Meters'.

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

[14] ITU-T Recommendation P.57 (2005): "Artificial ears".

[15] ITU-T Recommendation P.58 (1996): "Head and torso simulator for telephonometry."

[16] ITU-T Recommendation P.79 (2007) with Annex A: "Calculation of loudness ratings for telephone sets."

[17] 3GPP TS 06.77 R99 Minimum Performance Requirements for Noise Suppresser Application to the AMR Speech Encoder.

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

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

[20] ITU-T Recommendation P.340: "Transmission characteristics of hands-free telepones".

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

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

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

[24] ITU-T Recommendation O.131: "Psophometer for use on telephone-type circuits".

[25] ISO 9614: "Acoustics - Determination of sound power levels of noise sources using sound intensity".

[26] ISO 3745: "Acoustics - Determination of sound power levels of noise sources - Precision methods for anechoic and semi-anechoic rooms".

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

[28] ETSI TS 103 737: "Transmission requirements for narrowband wireless terminals (handset and headset) from a QoS perspective as perceived by the user".

[29] ETSI TS 103 738: "Transmission requirements for narrowband wireless terminals (handsfree) from a QoS perspective as perceived by the user".

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

[31] ETSI TS 103 740: "Transmission requirements for wideband wireless terminals (handsfree) from a QoS perspective as perceived by the user".

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

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document the term *narrow-band* refers to signals sampled at 8 kHz; *wideband* refers to signals sampled at 16 kHz.

For the purposes of the present document, the following terms: dB, dBr, dBm0, dBm0p and dBA, shall be interpreted as defined in ITU-T Recommendation B.12; the term dBPa shall be interpreted as the sound pressure level relative to 1 pascal expressed in dB (0 dBPa is equivalent to 94 dB SPL).

A 3GPP softphone is a telephony system running on a general purpose computer or PDA complying with the 3GPP terminal acoustic requirements (TS 26.131 and 26.132).

3.2 Abbreviations

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

ADC Analogue to Digital Converter
CSS Composite Source Signal
DAC Digital to Analogue Converter
DTX Discontinuous Transmission
EEC Electrical Echo Control
EL Echo Loss

ERP	Ear Reference Point
HATS	Head and Torso Simulator
LSTR	Listener Sidetone Rating
MRP	Mouth Reference Point
OLR	Overall Loudness Rating
PCM	Pulse Code Modulation
PDA	Personal Digital Assistant
POI	Point of Interconnection (with PSTN)
PSTN	Public Switched Telephone Network
RLR	Receive Loudness Rating
SLR	Send Loudness Rating
STMR	Sidetone Masking Rating
SS	System Simulator
TX	Transmission
UE	User Equipment

4 Interfaces

Access to terminals for acoustic testing is always made via the acoustic or air interfaces. The Air Interface is specified by the GSM 05 or 45 and the 3G 25 series specifications and is required to achieve user equipment (UE) transportability. Measurements can be made at this point using a system simulator (SS) comprising the appropriate radio terminal equipment and speech transcoder. The losses and gains introduced by the test speech transcoder will need to be specified.

The POI with the public switched telephone network (PSTN) is considered to have a relative level of 0 dBr, where signals will be represented by 8-bit A-law, according to ITU-T Recommendation G.711. Measurements may be made at this point using a standard send and receive side, as defined in ITU-T Recommendations.

Five classes of acoustic interface are considered in this specification:

- Handset UE including softphone UE used as a handset;
- Headset UE including softphone UE used with headset;
- Vehicle Mounted Hands-free UE including softphone UE mounted in a vehicule;
- Desktop-mounted hands-free UE including softphone UE with external loudspeaker(s) used in handsfree mode;
- Handheld hands-free UE including softphone UE with internal loudspeaker(s) used in handsfree mode.

(See definition of softphone in Clause 3.1)

NOTE: The test setup for a softphone UE shall be derived according to the following rules:

- When using a softphone UE as a handset: the test setup shall correspond to handset mode.
- When using a softphone UE with headset: the test setup shall correspond to headset mode.
- When a softphone UE is mounted in a vehicle: the test setup shall correspond to Vehicle-mounted handsfree mode.
- When using a softphone UE in handsfree mode:
  - When using internal loudspeaker(s), the test setup shall correspond to handheld hands-free.
  - When using external loudspeaker(s), the test setup shall correspond to desktop-mounted hands-free.

5 Test configurations

This section describes the test setups for terminal acoustic testing.

NOTE: If the terminal has several mechanical configurations (e.g. sliding design open or closed), all manufacturer-defined configurations shall be tested.

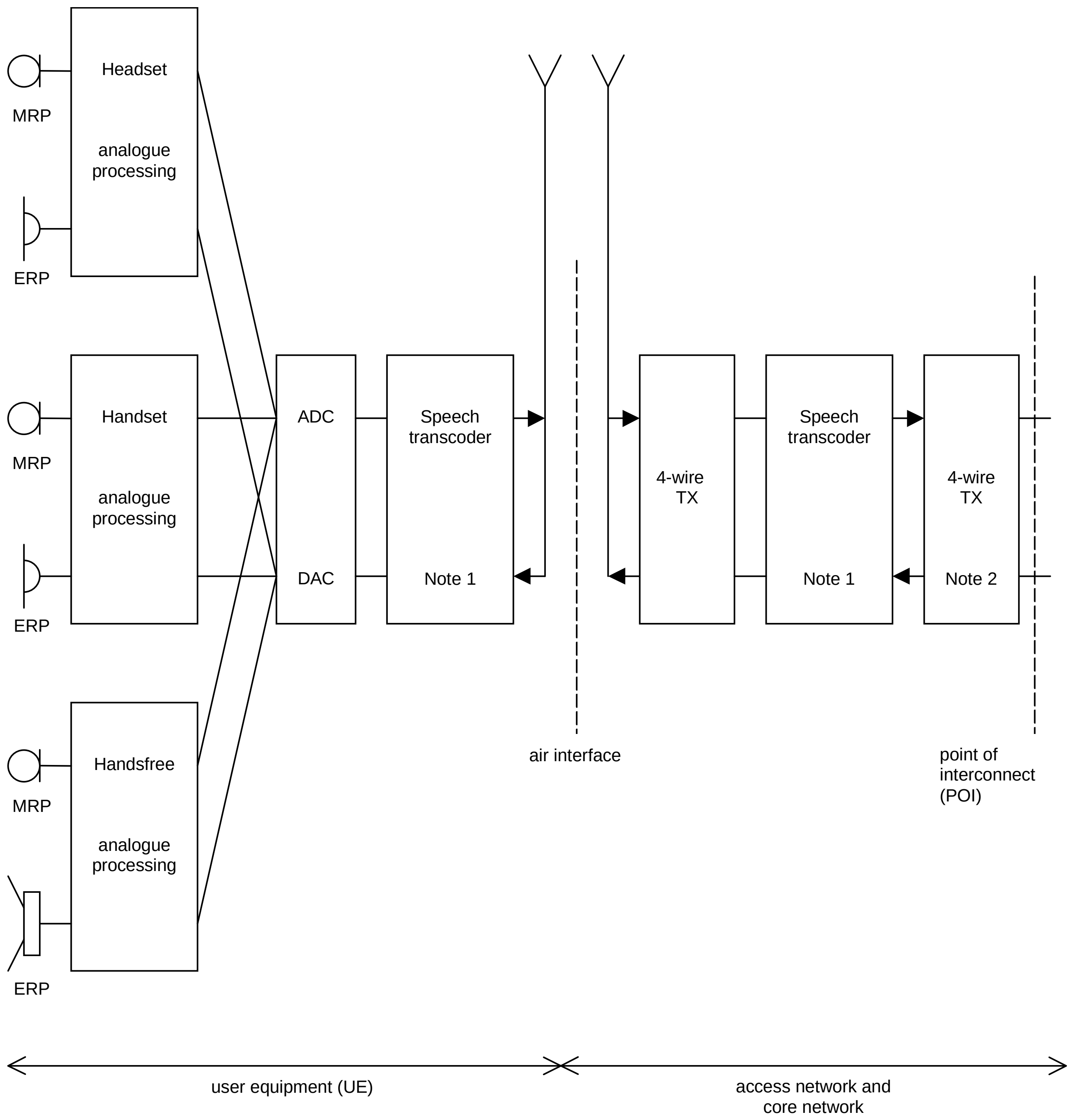
5.1 Setup for terminals

The general access to terminals is described in Figure 1. The preferred acoustic access to GSM and 3G terminals is the most realistic simulation of the 'average' subscriber. This can be made by using HATS (head and torso simulator), with appropriate ear simulation and appropriate mountings for handset terminals, in a realistic but reproducible way to the HATS. Hands-free terminals shall use the HATS or free field microphone techniques in a realistic but reproducible way.

HATS is described in ITU-T Recommendation P.58, appropriate ears are described in ITU-T Recommendation P.57 (Type 3.3 and type 3.4 ear), a proper positioning of handsets in realistic conditions is found in ITU-T Recommendation P.64, the test setups for various types of hands-free terminals can be found in ITU-T Recommendation P.581.

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

The preferred way of testing is the connection of a terminal to the system 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.



NOTE 1: Includes DTX functionality.
NOTE 2: Connection to PSTN should include electrical echo control (EEC).

Figure 1: GSM/3G Interfaces for specification and testing of terminal acoustic characteristics

5.1.1 Setup for handset terminals

When using a handset UE, the handset is placed on HATS as described in ITU-T Recommendation P.64 [18]. The artificial mouth shall conform with ITU-T Recommendation P.58 [15]. The artificial ear shall conform to ITU-T Recommendation P.57 [14]. Type 3.3 or Type 3.4 ear shall be used and positioned on HATS according to ITU-T Recommendation P.58 [15].

Position and calibration of HATS
All the sending and receiving characteristics shall be tested with the HATS. It shall be indicated what type of ear was used at what application force. If not stated otherwise in TS 26.131, 8 +/- 2 N application force shall be used.

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

5.1.2 Setup for headset terminals

Recommendations for setup and positioning headsets are given in ITU-T Recommendation P.380 [32]. If not stated otherwise, headsets shall be placed in their recommended wearing position. Some insert earphones might not fit properly in Type 3.3 or Type 3.4 ear simulators. For such insert type headsets, ITU-T Recommendation P.57 [14] Type 2 ear simulator can be used in conjunction with the HATS mouth simulator. The HATS should be equipped with two artificial ears as specified in ITU-T Recommendation P.57 [14]. For binaural headsets two artificial ears are required.

5.1.3 Setup for hands-free terminals

5.1.3.1 Vehicle mounted hands-free

If not stated otherwise, the artificial head (HATS – head and torso simulator, according to ITU-T Recommendation P.58 [15]) is positioned in the driver's seat for the measurement as shown in figure 3a. The position has to be in line with the average user's position; therefore, all positions and sizes of users have to be taken into account. Typically, all except the tallest 5% and the shortest 5% of the driving population have to be considered. The size of these persons can be derived, e.g., from the 'anthropometric data set' for the corresponding year (e.g., based on data used by the car manufacturers). The position of the HATS (mouth/ears) within the positioning arrangement is given individually by each car manufacturer. The position used has to be reported in detail in the test report. If no requirements for positioning are given, the distance from the microphone to the MRP is defined by the test lab.

By using suitable measures (marks in the car, relative position to A-pillar, B-pillar, height from the floor etc.) the exact reproduction of the artificial head position must be possible at any later time.

NOTE – Different positions of the artificial head may greatly influence the test results. Depending on the application, different positions of the artificial head may be chosen for the tests. It is recommended to check the worst-case position, e.g., those positions where the SNR and/or the speech quality in Send may be worst.

Figure 2: void

Figure 3: void

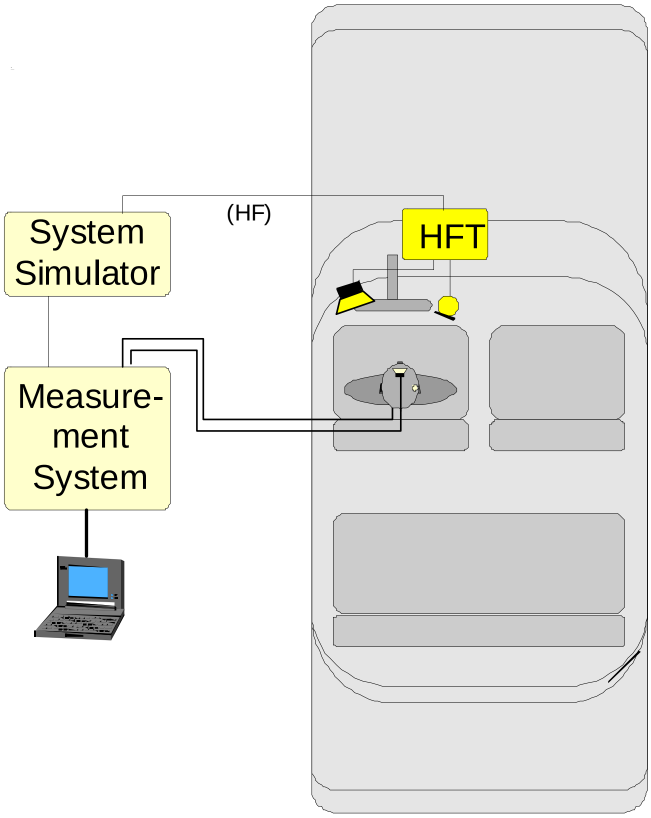


Figure 3a: Test Configuration for Vehicle mounted hands-free, using HATS

5.1.3.2 Desktop mounted hands-free

For HATS test equipment, definition of hands-free terminals and setup for desktop hands-free terminals can be found in ITU-T Recommendation P.581 [19]. Measurement setup using a free field microphone and a discrete P.51 [13] artificial mouth for desktop hands-free terminals can be found in ITU-T Recommendation P.340 [20].The positioning for different types of desktop mounted hands-free terminals is given in ETSI TS 103 738 [29] and ETSI TS 103 740 [31].

5.1.3.3 Handheld hands-free

Either HATS or a free-field microphone with a discrete P.51 [13] artificial mouth may be used to measure Handheld hands-free type UE.

If HATS measurement equipment is used, it shall be configured to the Handheld hands-free UE according to Figure 4. The HATS should be positioned so that the HATS Reference Point is at a distance d_{HF} from the centre point of the visual display of the Mobile Station. The distance d_{HF} is specified by the manufacturer. A vertical angle θ_{HF} may be specified by the manufacturer. In case it is not specified, the nominal distance d_{HF} shall be 42 cm and θ_{HF} shall be 0.

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

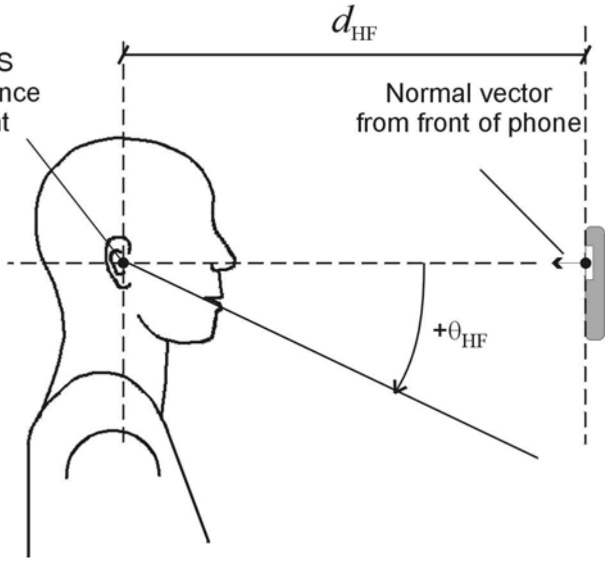


Figure 4: Configuration of Hand-Held Hands-free UE relative to the HATS

If a free-field microphone and a discrete P. 51 [13] mouth are used, they shall be configured to the Handheld hands-free UE as per Figure 5 for receiving measurements and Figure 6 for sending measurements. The microphone should be located at a distance d_{HF} from the centre of the visual display of the UE. The mouth simulator should be located at a distance d_{HF}-12 cm from the centre of the visual display of the UE. The distance d_{HF} is specified by the manufacturer. In case it is not specified the nominal distance d_{HF} shall be 42 cm.

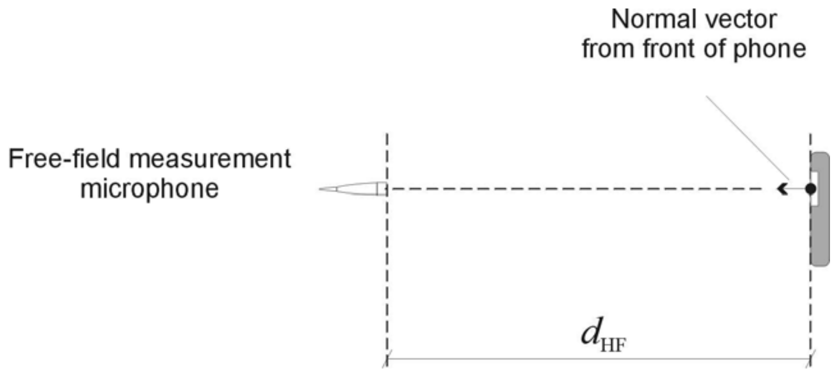


Figure 5: Configuration of Hand-Held Hands-free UE, free-field microphone for receiving measurements

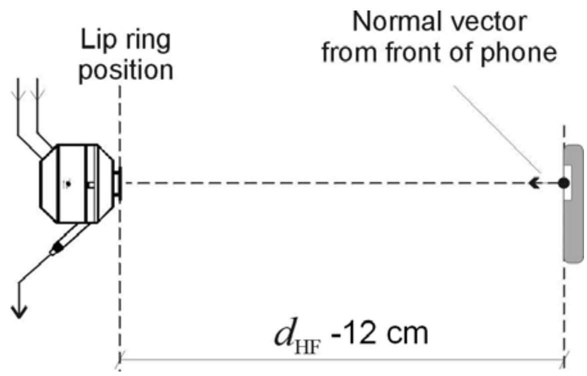


Figure 6: Configuration of Hand-Held Hands-free UE, discrete P. 51 artificial mouth for sending measurements

5.1.3.4 Softphone including speakers and microphone

This test setup is applicable to laptop computers or similar devices as seen in Figures 7 to 11.

When the manufacturer gives conditions of use, they will apply for test. If the manufacturer gives no other requirement, the softphone will be positioned according the following conditions:

Measurement with artificial ear and microphone:

Artificial mouth (for sending tests)

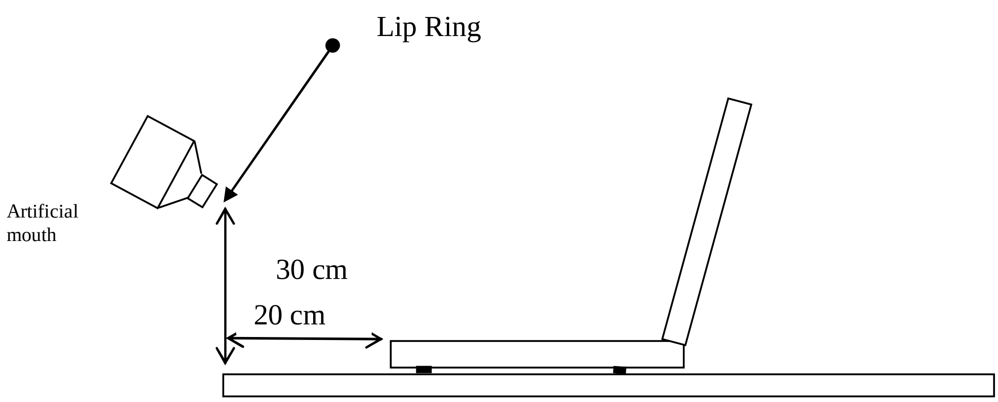


Figure 7: Configuration of softphone relative to the artificial mouth side view

Free field microphone (for receiving):

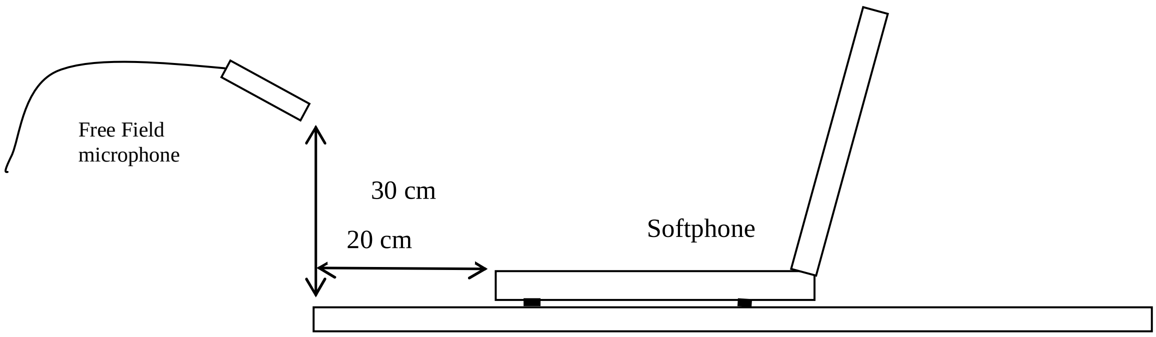


Figure 8: Configuration of softphone relative to the free field microphone side view

Position of softphone on the table:

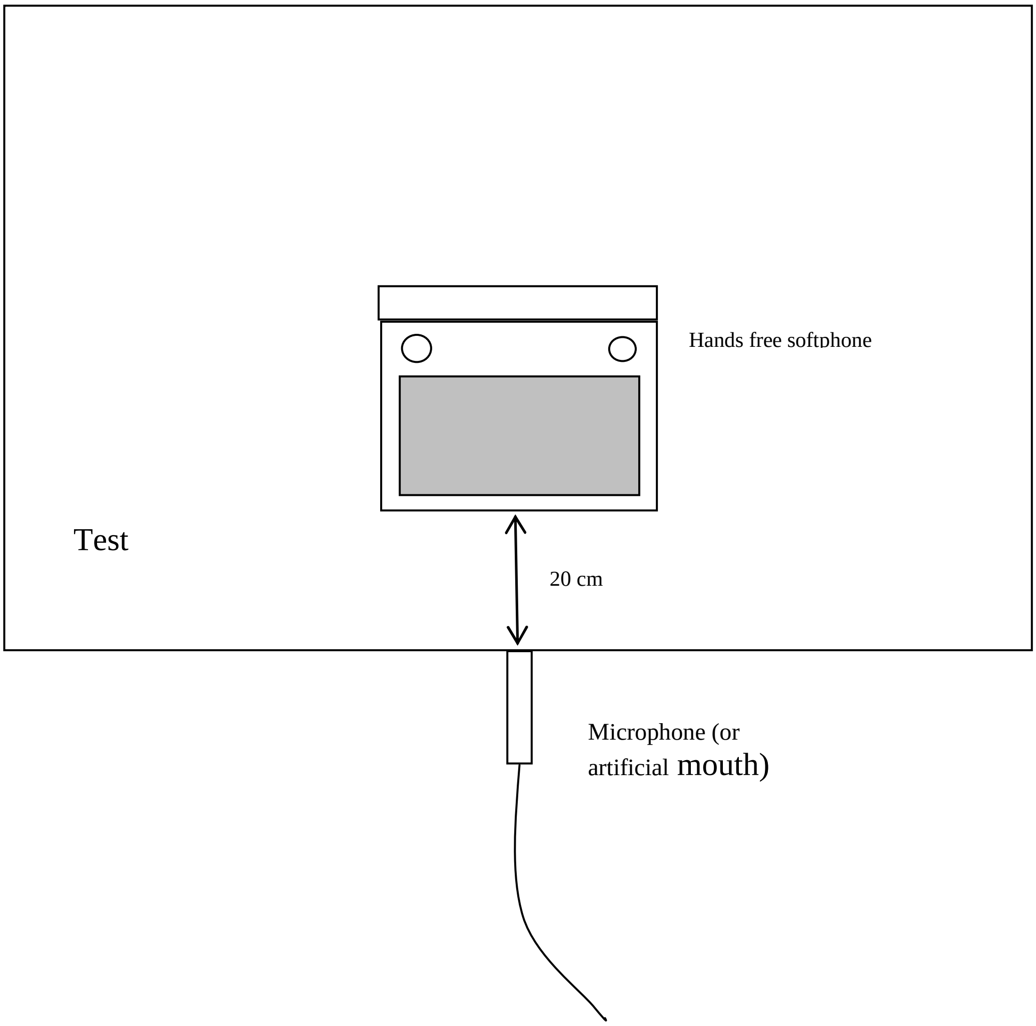


Figure 9: Configuration of softphone relative to the free field microphone or artificial mouth top sight

Measurement with HATS:

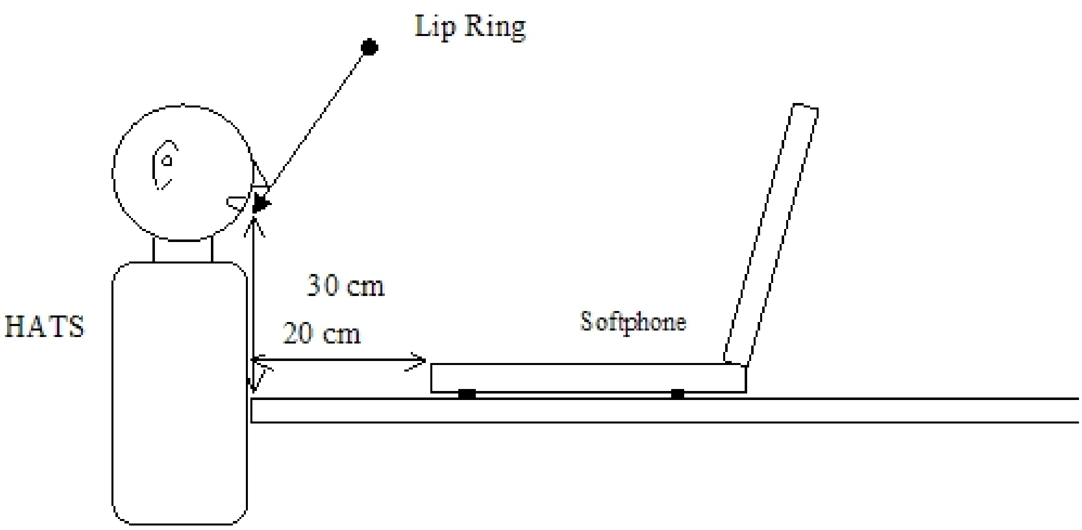


Figure 10: Configuration of softphone relative to the HATS side view

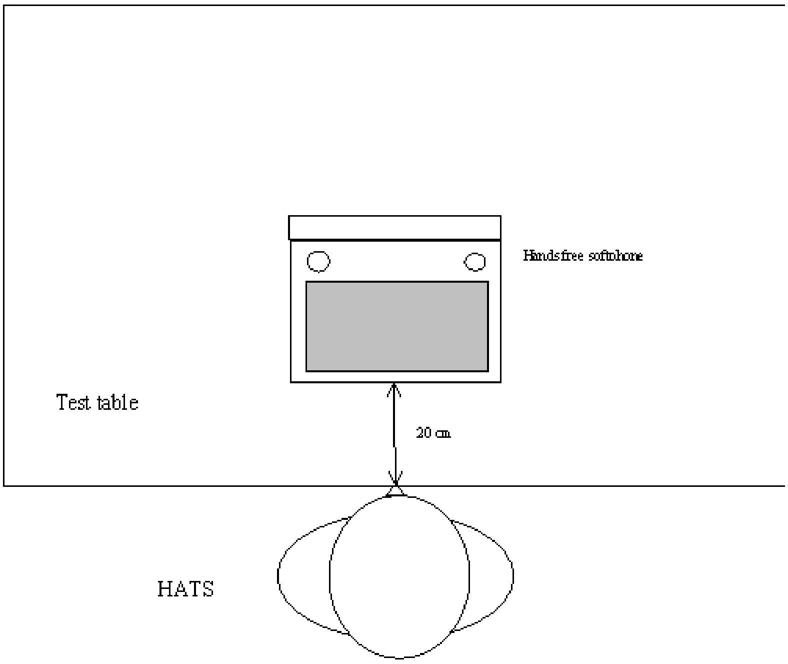


Figure 11: Configuration of softphone relative to the HATS top sight

5.1.3.5 Softphone with separate speakers

This test setup is applicable to laptop computers or similar devices as seen in Figures 12 to 15.

When the manufacturer gives conditions of use, they will apply for test. If the manufacturer gives no other requirement, the softphone will be positioned according to the following conditions:

When separate loudspeakers are used, system will be positioned as in Figure 12 or 13.

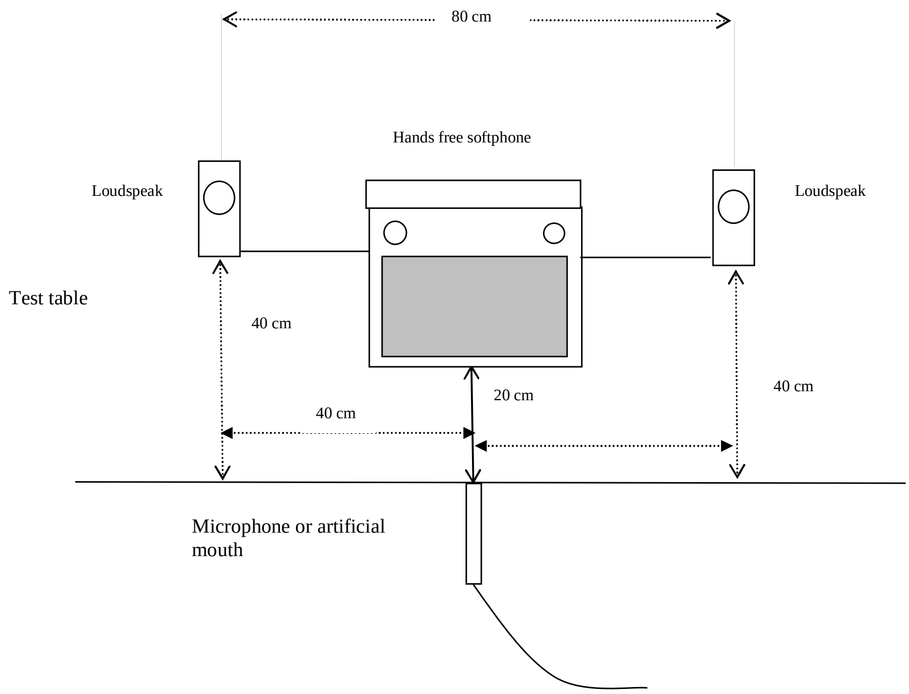


Figure 12: Configuration of softphone using external speakers relative to microphone or artificial mouth top sight

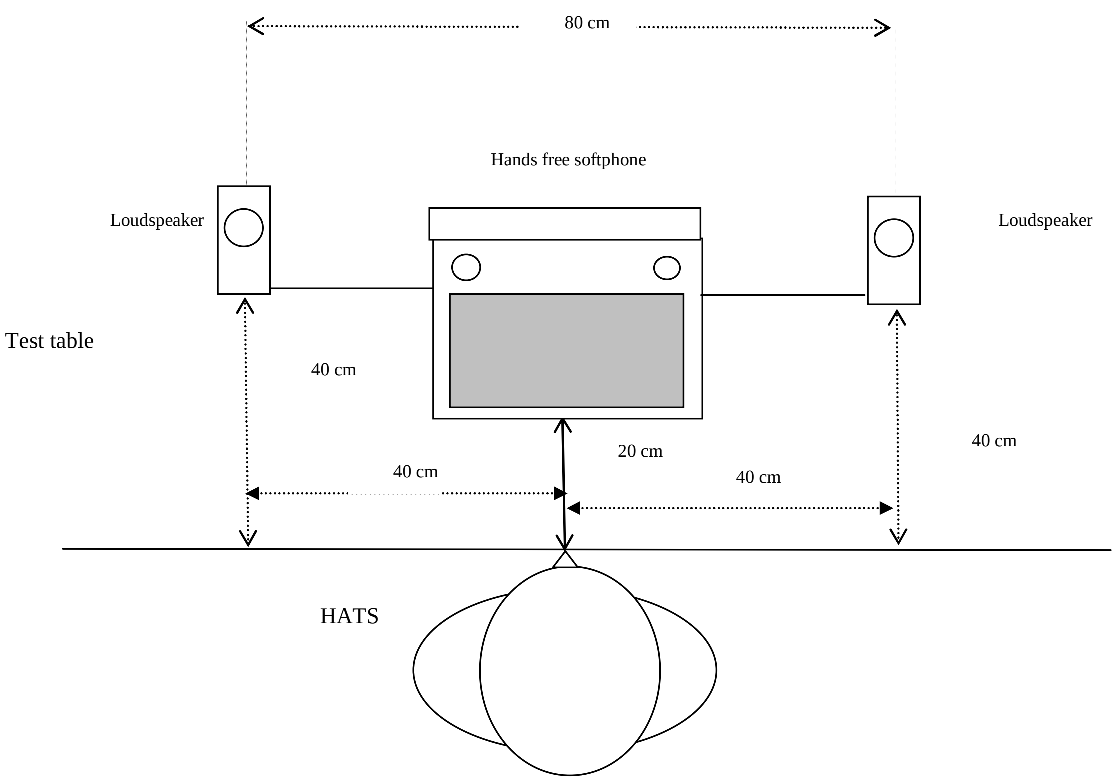


Figure 13: Configuration of softphone using external speakers relative to the HATS top sight

When external microphone and speakers are used, system will be positioned as in Figure 14 or 15.

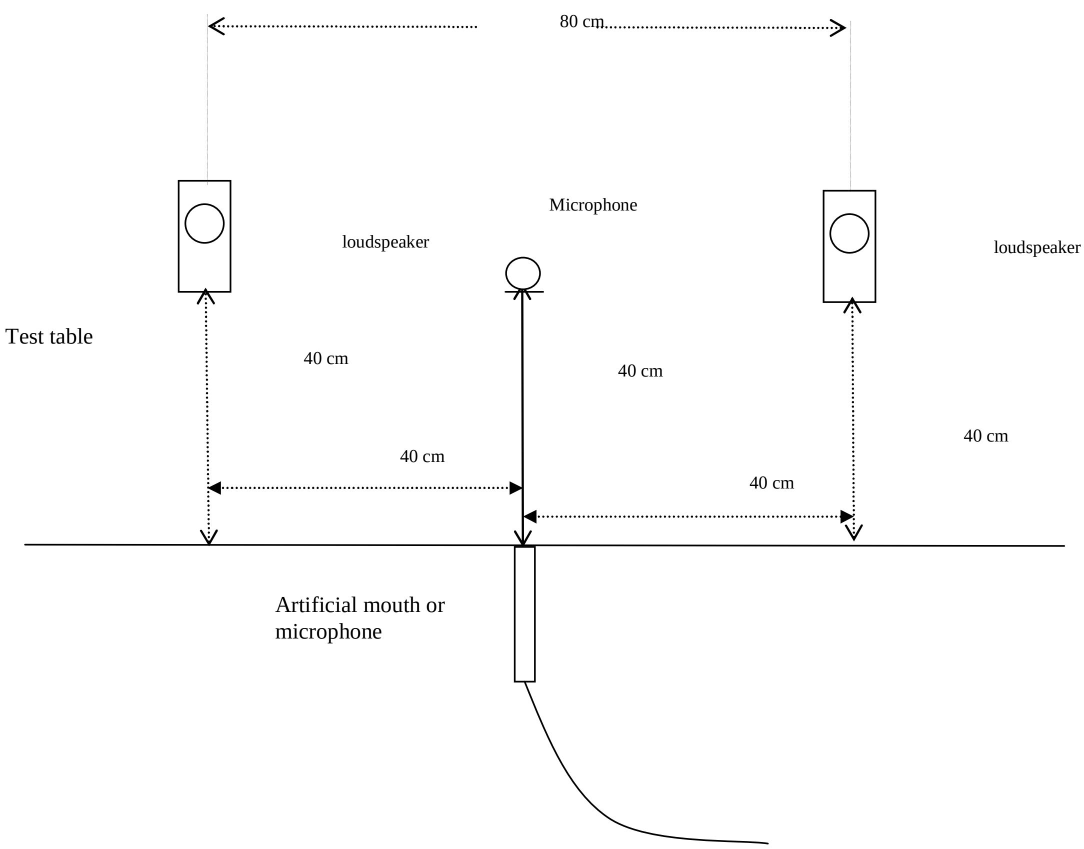


Figure 14: Configuration of softphone using external speakers and microphone relative to microphone or artificial mouth top sight

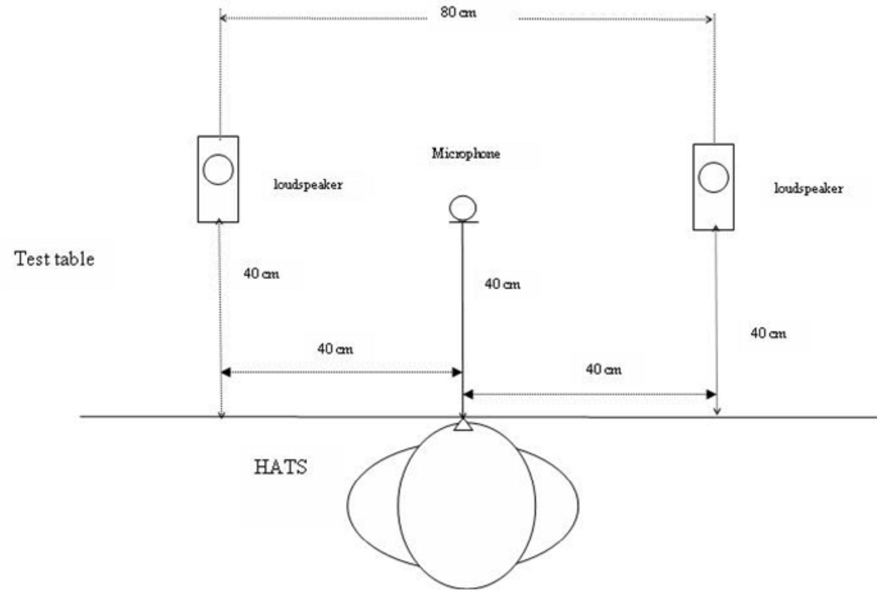


Figure 15: Configuration of softphone using external speakers and microphone relative to the HATS top sight

5.1.4 Position and calibration of HATS

The horizontal positioning of the HATS reference plane shall be guaranteed within ± 2° for testing hands-free equipment.

The HATS shall be equipped with either Type 3.3 or 3.4 Artificial Ear. For hands-free measurements the HATS shall be equipped with two artificial ears. The pinnae are specified in Recommendation P.57 [14] for Types 3.3 and 3.4 artificial ears. The pinnae shall be positioned on HATS according to ITU-T Recommendation P.58 [15].

The exact calibration and equalization procedures as well as the combination of the two ear signals for the purpose of measurements can be found in ITU-T Recommendation P.581. If not stated otherwise, the HATS shall be diffuse-field equalized. The reverse nominal diffuse field curve as found in table 3 of ITU-T Recommendation P.58 [15] shall be used. For measurements requiring diffuse-field correction values for closer frequency spacing than what is available in P.58, an interpolation method should be used.

For Handheld hands-free UE, the set-up corresponding to 'portable hands-free' in P.581 [19] should be used.

5.2 Setup of the electrical interfaces

5.2.1 Codec approach and specification

Codec approach: In this approach, a codec is used to convert the digital input/output bit-stream of the system simulator to the equivalent analogue values. With this approach a system simulator simulating the radio link to the terminal under controlled and error free conditions is required. The system simulator has to be equipped with a high-quality codec whose characteristics are as close as possible to ideal.

Definition of 0 dBr point:

D/A converter - a Digital Test Sequence (DTS) representing the codec equivalent of an analogue sinusoidal signal whose rms value is 3,14 dB below the maximum full-load capacity of the codec shall generate 0 dBm across a 600 ohm load;

A/D converter - a 0 dBm signal generated from a 600 ohm source shall give the digital test sequence (DTS) representing the codec equivalent of an analogue sinusoidal signal whose RMS value is 3,14 dB below the maximum full-load capacity of the codec.

Narrow band telephony testing

For testing a GSM or 3G terminal supporting narrow-band telephony, the system simulator shall use the AMR speech codec as defined in 3GPP TS 26 series specifications, at the source coding bit rate of 12,2kbit/s.

Wide band telephony testing

For testing a GSM or 3G terminal supporting wide-band telephony, the system simulator shall use the AMR-WB speech codec as defined in 3GPP TS26 series specifications, at the source coding bit rate of 12,65 kbit/s.

5.2.2 Direct digital processing approach

In this approach, the digital input/output bit-stream of the terminal connected through the radio link to the system simulator is operated upon directly. For the purposes of GSM/3G acoustic testing, the direct digital processing shall use the default speech codec, the AMR speech codec as defined in 3GTS26 series specifications, at it"s highest source coding bit rate of 12,2kbit/s.

Narrow band telephony testing

For testing a GSM or 3G terminal supporting narrow-band telephony, the system simulator shall use the AMR speech codec as defined in 3GPP TS 26 series specifications, at the source coding bit rate of 12,2kbit/s.

Wide band telephony testing

For testing a GSM or 3G terminal supporting wide-band telephony, the system simulator shall use the AMR-WB speech codec as defined in 3GPP TS 26 series specifications, at the source coding bit rate of 12,65 kbit/s.

5.3 Accuracy of test equipment

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

Item	Accuracy
Electrical Signal Power	±0,2 dB for levels ≥ -50 dBm
Electrical Signal Power	±0,4 dB for levels < -50 dBm
Sound pressure	±0,7 dB
Time	±5 %
Frequency	±0,2 %

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

Quantity	Accuracy
Sound pressure level at MRP	**±1 dB for 200 Hz to 4 kHz** **±3 dB for 100 Hz to 200 Hz** **and 4 kHz to 8 kHz**
Electrical excitation levels	**±0,4 dB (see note 1)**
Frequency generation	**±2 % (see note 2)**
NOTE 1: Across the whole frequency range. NOTE 2: When measuring sampled systems, it is advisable to avoid measuring at sub-multiples of the sampling frequency. There is a tolerance of ±2 % on the generated frequencies, which may be used to avoid this problem, except for 4 kHz where only the -2 % tolerance may be used.	

The measurements results shall be corrected for the measured deviations from the nominal level.
The sound level measurement equipment shall conform to IEC 60651 Type 1.

5.4 Test signals

Unless stated otherwise, appropriate test signals for GSM/3G acoustic tests (general description) are defined in ITU-T Recommendation P.50 and P.501. More information can be found in the test procedures described below.

NOTE: As stated in section 5.2, the source coding bit rate shall be 12,2kbit/s for narrow-band and 12,65kbit/s for wide-band for all measurements. Tests at lower bit rates are not covered by 3GPP TS 26.132. If measurements are still performed at lower bit rates the use of multisine signal is not recommended, because the results depend on the selected bit rate.

For testing the narrow-band telephony service provided by the UE, the test signal used shall be band limited between 100 Hz and 4 kHz with a bandpass filter providing a minimum of 24 dB/Oct. filter roll off, when feeding into the receiving direction.

For testing the wide-band telephony service provided by the UE, the test signal used shall be band limited between 100 Hz and 8 kHz with a bandpass filter providing a minimum of 24 dB/Oct. filter roll off, when feeding into the receiving direction.

The test signal levels are referred to the average level of the (band limited in receiving direction) test signal, averaged over the complete test sequence, unless specified otherwise.

5.5 Void

5.5.1 Void

5.5.2 Void

6 Test conditions

6.1 Environmental conditions

6.1.1 Handset and headset terminals

The environmental conditions for testing handset and headset UE is specified in §6.1.1 TS 26.132, as follows:

For handset and headset measurements the test room shall be practically free-field down to a lowest frequency of 275 Hz, the handset or the headset including the HATS shall lie totally within this free-field volume. This shall be met if deviations of the ideal free-field conditions are less than +/- 1 dB. Qualification of the test room may be performed using the method described in either ISO 3745 Annex A, or ITU-T P. 340 §5.4.

Alternatively, a test room may be used which meets the following two criteria:

1. The relationship between the pressure at the mouth opening and that at 5,0, 7,5 and 10 cm in front of the centre of the lip ring is within ±0.5dB of that which exists in a known acoustic free-field.
2. The relationship between the pressure at the mouth opening and at the Ear canal Entrance Point (EEP) at both the left and right ears of the HATS does not differ by more than ±1dB from that which exists in a known free-field.

The ambient noise level shall be less than -30 dBPa(A), for idle channel noise measurements the ambient noise level shall be less than –64dBPa(A).

Echo measurements shall be conducted in realistic rooms with an ambient noise level less then -64 dBPa(A).

6.1.2 Hands-free terminals

Hands-free terminals generally should be tested in their typical environment of application. Care must be taken, that e.g. noise levels are sufficiently low in order not to interfere with the measurements.

For Desk-Top hands-free terminals the appropriate requirements shall be taken from ITU-Recommendation P.340.

The broadband noise level shall not exceed –70 dBPa(A). The octave band noise level shall not exceed the values specified in Table 2.

TABLE 2: P.340 Noise level

Center frequency (Hz)	Octave band pressure level (dBPa)
63	–45
125	–60
250	–65
500	–65
1 k	–65
2 k	–65
4 k	–65
8 k	–65

Echo measurements shall be conducted in realistic rooms with an ambient noise level less then -70 dBPa(A).

6.2 System Simulator conditions

The system simulator should provide an error free radio connection to the UE under test. The default speech codec in narrowband, the AMR speech codec, shall be used at it"s highest bit rate of 12,2 kbit/s. The default speech codec, in wideband, AMR-WB, shall be used at 12,65 kbit/s. Discontinuous Transmission, DTX, (silence suppression) shall be disabled for the purposes of GSM/3G acoustic testing.

7 Narrow-band telephony transmission performance test methods

7.1 Applicability

The test methods in this sub-clause shall apply when testing a UE that is used to provide narrow-band or wideband telephony, either as a stand-alone service, or as part of a multimedia service.

7.2 Overall loss/loudness ratings

7.2.1 General

The SLR and RLR values for the GSM or 3G networks apply up to the POI. However, the main determining factors are the characteristics of the UE, including the analogue to digital conversion (ADC) and digital to analogue conversion (DAC). In practice, it is convenient to specify loudness ratings to the Air Interface. For the normal case, where the GSM or 3G networks introduces no additional loss between the Air Interface and the POI, the loudness ratings to the PSTN boundary (POI) will be the same as the loudness ratings measured at the Air Interface.

7.2.2 Connections with handset UE

7.2.2.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The handset terminal is setup as described in subclause 5.

   The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula (A-23b), over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79, table 1.

7.2.2.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The handset terminal is setup as described in subclause 5. The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17. For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [16], formula (A-23c), over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79 [16].

d) DRP-ERP correction is used. No leakage correction shall be applied.

7.2.3 Connections with Vehicle Mounted & Desktop hands-free UE

Vehicle mounted hands-free should be tested in the vehicle (for the totally integrated vehicle hands-free systems) or in a vehicle simulator, ref ETSI 0358 601 (TR101110) Digital Cellular Telecommunications System (Phase 2+) Characterization test methods and quality assessment for hands-free mobiles.

Free-field measurements for vehicle mounted hands-free are for further study.

7.2.3.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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 broadband signal level is then adjusted to –28,7 dBPa at the HFRP or the HATS HFRP (as defined in P.581) and the spectrum is not altered.

   The spectrum at the MRP and the actual level at the MRP (measured in third octaves) are used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5. The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17. For the calculation, the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula (A-23b), over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79, table 1.

7.2.3.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively, a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The hands-free terminal is setup as described in subclause 5. If HATS is used, then it is free-field equalized as described in ITU-T Recommendation P.581. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the 'right ' and 'left' signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17.

   For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [16], formula (A-23c), over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79.

d) No leakage correction shall be applied. The hands-free correction as described in P.340 shall be applied. To compute Receiving loudness rating (RLR) for hands-free terminal (see also ITU-T Recommendation P.340), when using the combination of left and right ear signals from HATS, the HFL_E has to be 8 dB instead of 14 dB. For further information see ITU-T Recommendation P.581.

7.2.4 Connections with Handheld hands-free UE

7.2.4.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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 broadband signal level is then adjusted to –28,7 dBPa at the HFRP or the HATS HFRP (as defined in P.581) and the spectrum is not altered.

   The spectrum at the MRP and the actual level at the MRP (measured in third octaves) are used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17. For the calculation, the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula (A-23b), over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79, table 1.

7.2.4.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. If HATS is used, then it is free-field equalized as described in ITU-T Recommendation P.581. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the 'right ' and 'left' signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17.

   For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [16], formula (A-23c), over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79.

d) No leakage correction shall be applied. The hands-free correction as described in P.340 shall be applied. To compute the Receiving loudness rating (RLR) for hands-free terminals (see also ITU-T Recommendation P.340), when using the combination of left and right ear signals from HATS, the HFL_E has to be 8 dB instead of 14 dB. For further information see ITU-T Recommendation P.581.

7.2.5 Connections with headset UE

Same as for handset.

7.3 Idle channel noise (handset and headset UE)

For idle noise measurements in sending and receiving directions, care should be taken that only the noise is windowed out by the analysis and the result is not impaired by any remaining reverberation or by noise and/or interference from various sources. Some examples are air-conducted or vibration-conducted noise from sources inside or outside the test chamber, disturbances from lights and regulators, mains supply induced noise including grounding issues, test system and system simulator inherent noise as well as radio interference from the UE to test equipment such as ear simulators, microphone amplifiers etc.

7.3.1 Sending

The terminal should be configured to the test equipment as described in subclause 5.1.

The environment shall comply with the conditions described in subclause 6.1.

The noise level at the output of the SS is measured with psophometric weighting.. The psophometric weighting filter is described in ITU-T Recommendation O.41.

A test signal may have to be intermittently applied to prevent "silent mode" operation of the MS. This is for further study.

The measured part of the noise shall be 170.667 ms (equals 8192 samples in a 48 kHz sample rate test system). The spectral distribution of the noise is analyzed with an 8k FFT using a window, which has less than 0.1 dB leakage for non bin-centered signals. This can be achieved with a window function commonly known as 'flat top window'. Within the specified frequency range, the FFT bin that has the highest level is searched. The level of this bin is the maximum level of a single frequency disturbance.

To increase the repeatability, the test sequence (optional activation + noise level measurement) may be repeated n times.

The thus obtained total noise powers shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

The thus obtained single frequency maximum powers shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

7.3.2 Receiving

The terminal should be configured to the test equipment as described in subclause 5.1.

The environment shall comply with the conditions described in subclause 6.1.

A test signal may have to be intermittently applied to prevent "silent mode" operation of the MS. This is for further study.

The noise level shall be measured with A-weighting at the DRP with diffuse-field correction. The A-weighting filter is described in IEC 60651 [12].

The measured part of the noise shall be 170.667 ms (equals 8192 samples in a 48 kHz sample rate test system). The spectral distribution of the noise is analyzed with an 8k FFT using a window, which has less than 0.1 dB leakage for non bin-centered signals. This can be achieved with a window function commonly known as 'flat top window'. Within the specified frequency range, the FFT bin that has the highest level is searched. The level of this bin is the maximum level of a single frequency disturbance.

To increase the repeatability, considering the test sequence (optional activation + noise level measurement) may be repeated n times.

The thus obtained total noise powers shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

The thus obtained single frequency maximum powers shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

7.4 Sensitivity/frequency characteristics

7.4.1 Handset and headset UE sending

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The handset terminal is setup as described in subclause 5.

   Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 4 kHz inclusive. For the calculation, the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa.

7.4.2 Handset and headset UE receiving

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The handset terminal is setup as described in subclause 5.

   Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 4 kHz inclusive. For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The HATS is diffuse-field equalized. The sensitivity is expressed in terms of dBPa/V. Information about correction factors is available in ITU-T Recommendation P.57.

Optionally the measurements might be repeated with 2N and 13N application force. In these cases no normative values apply.

7.4.3 Vehicle Mounted & Desktop hands-free UE sending

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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 broadband signal level is then adjusted to –28,7 dBPa at the HFRP or the HATS HFRP (as defined in P.581) and the spectrum is not altered.

   The spectrum at the MRP and the actual level at the MRP (measured in third octaves) are used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 4 kHz inclusive. For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBV/Pa.

7.4.4 Vehicle Mounted & Desktop hands-free UE receiving

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The hands-free terminal is setup as described in subclause 5. If the HATS is used, then it is free-field equalized as described in ITU-T Recommendation P.581. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the 'right ' and 'left' signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V.

7.4.5 Handheld hands-free UE sending

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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 broadband signal level then is adjusted to –28,7 dBPa at the HFRP or the HATS HFRP (as defined in P.581) and the spectrum is not altered.

   The spectrum at the MRP and the actual level at the MRP (measured in third octaves) are used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 4 kHz inclusive. For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBV/Pa.

7.4.6 Handheld hands-free UE receiving

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. If the HATS is used, then it is free-field equalized as described in ITU-T Recommendation P.581. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the 'right ' and 'left' signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 4 kHz inclusive. For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V.

7.5 Sidetone characteristics

7.5.1 Connections with Handset UE

The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be –4,7 dBPa measured at the MRP. The test signal level is averaged over the complete test signal sequence.

7.5.1.1 void

7.5.1.2 Connections with Handset UE – HATS method

The handset UE is setup as described in subclause 5. The application force shall be 13N on the artificial ear Type 3.3 or Type 3.4.

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

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 8 kHz inclusive. For the calculation, the averaged measured level at each frequency band (ITU-T Recommendation P.79, table 4, bands 4 to 17) 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 ITU-T Recommendation P.79, using m = 0.225 and the weighting factors in table B.2, unsealed condition, of ITU-T Recommendation P.79. No leakage correction (L_E) shall be applied. DRP-ERP correction is used.

7.5.2 Headset UE

The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be –4,7 dBPa measured at the MRP. The test signal level is averaged over the complete test signal sequence.

Measurements shall be made at one twelfth-octave intervals as given by the R.10 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 8 kHz inclusive. For the calculation, the averaged measured level at each frequency band (ITU-T Recommendation P.79, table 4, bands 4 to 17) is referred to the averaged test signal level measured in each frequency band. The sidetone path loss L_{meST} as expressed in dB shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17. The STMR (in dB) shall be calculated from the formula B-4 of ITU-T Recommendation P.79 [16], using m = 0.225 and the weighting factors in Table B.2, unsealed condition, of ITU-T Recommendation P.79 [16]. No leakage correction (L_E) shall be applied. DRP-ERP correction is used.

7.5.3 Hands-free UE (all categories)

No requirement for other than echo control.

7.6 Stability loss

Where a user controlled volume control is provided it is set to maximum.

Handset UE: The handset is placed on a hard plane surface with the earpiece facing the surface.

Headset UE: The requirement applies for the closest possible position between microphone and headset receiver within the intended wearing position.

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

Hands-free UE (all categories): no requirement other than echo loss.

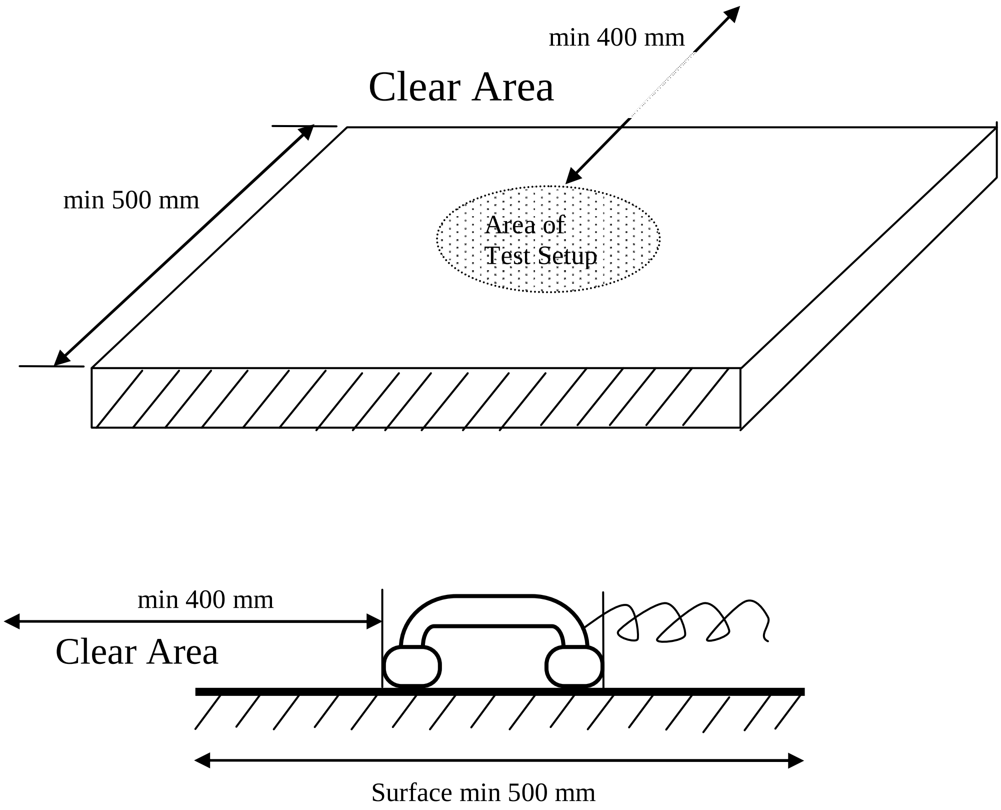
Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is applied. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB instead of 11 dB. The duration of the test signal is 250 ms. With an input signal of -3 dBm0, the attenuation from input to output of the system simulator shall be measured under the following conditions:

a) the handset or the headset, with the transmission circuit fully active, shall be positioned on a hard plane surface with at least 400 millimetre free space in all directions; the earpiece shall face towards the surface as shown in figure 15c;

b) the headset microphone is positioned as close as possible to the receiver(s) within the intended wearing position;

c) for binaural headset, the receivers are placed symmetrically around the microphone.



NOTE: All dimensions in mm.

Figure 15c

The attenuation from input to output of the system simulator shall be measured in the frequency range from 200 Hz to 4 kHz. The spectral distribution of the output signal is analysed with a 4k FFT (for the 48 kHz sampling rate), thus the measured part of the output signal is 85.333 ms. To avoid leakage effects, the frequency resolution of the FFT must be the same as the frequency spacing of the PN sequence.

7.7 Acoustic echo control

7.7.1 General

The echo loss (EL) presented by the GSM or 3G networks at the POI should be at least 46 dB during single talk. This value takes into account the fact that UE is likely to be used in a wide range of noise environments.

7.7.2 Acoustic echo control in a Hands-free UE

TCLw:

The hands-free is setup in a room with acoustic properties similar to a typical 'office-type' room; a vehicle-mounted hands-free UE should be tested in a vehicle or vehicle simulator, as specified by the UE manufacturer. [For reference on a suitable vehicle simulator see ETSI 0358 601 (TR101110) Digital Cellular Telecommunications System (Phase 2+).] The ambient noise level shall be less than -70 dBPa(A). The attenuation from reference point input to reference point output shall be measured using a speech like test signal.

Before the actual test, a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is applied.

Either a logarithmically spaced multi-sine or PN-sequence test signal shall be used.

When using a logarithmically spaced multi-sine test signal, it is defined as:

$$s(t) = \sum_i \left[\left[A + \mu_{AM} \cos(2\pi * f_{AM})\right] * \cos(2\pi * f_{0i})\right]$$

with A = 0,5

f_{AM} = 4 Hz, μ_{AM} = 0,5

$f_{0\,i}$= 250Hz * $2^{(i/3)}$;i=1..11

CF= 14dB ±1 dB (10 dB + 4,26 dB due to 100% AM modulation)

CF = Crest Factor = Peak to RMS ratio

The PN sequence is complying with ITU-T Recommendation P.501 with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB instead of 11 dB. The duration of the test signal is 250 ms. The test signal level is -3 dBm0 (from 100 Hz to 3.5 kHz).

The training sequence level shall be –16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1,0 s).

Note:

Full scale of coder input signal corresponds to +3,14 dBm0 with sinusoidal signal, CF= 3dB. A test signal with a CF of maximum 15 dB can thus have a level of up to –8,86 dBm0 without overloading the codec. In order to get best dynamic range the signal amplitude should be as high as possible.

The training sequence level shall be –16 dBm0 in order not to overload the codec. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Care should be taken that the terminal under test considers the test signal as a speech-like signal.

7.7.3 Acoustic echo control in a handset UE

The handset is set up according to clause 5. The ambient noise level shall be less than -64 dBPa(A). The attenuation from reference point input to reference point output shall be measured using the speech like test signal defined below.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is applied.

Either a logarithmically spaced multi-sine or a PN-sequence test signal shall be used.

When using a logarithmically spaced multi-sine test signal, it is defined as:

$$s(t) = \sum_i \left[\left[A + \mu_{AM} \cos(2\pi * f_{AM})\right] * \cos(2\pi * f_{0i})\right]$$

with A = 0,5

f_{AM} = 4 Hz, μ_{AM} = 0,5

$f_{0\,i}$= 250Hz * $2^{(i/3)}$;i=1..11

CF= 14dB ±1 dB (10 dB + 4,26 dB due to 100% AM modulation)

CF = Crest Factor = Peak to RMS ratio

The PN sequence is complying with ITU-T Recommendation P.501 with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB instead of 11 dB. The duration of the test signal is 250 ms. The test signal level is -3 dBm0 (from 100 Hz to 3.5 kHz).

The training sequence level shall be –16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1,0 s).

Note:

Full scale of coder input signal corresponds to +3,14 dBm0 with sinusoidal signal, CF= 3dB. A test signal with a CF of maximum 15 dB can thus have a level of up to –8,86 dBm0 without overloading the codec. In order to get best dynamic range the signal amplitude should be as high as possible.

The training sequence level shall be –16 dBm0 in order not to overload the codec. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Care should be taken that the terminal under test considers the test signal as a speech-like signal.

7.7.4 Acoustic echo control in a headset UE

The headset is set up according to clause 5. The ambient noise level shall be less than -64 dBPa(A). The attenuation from reference point input to reference point output shall be measured using the speech like test signal defined below.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is applied.

Either a logarithmically spaced multi-sine or a PN-sequence test signal shall be used.

When using a logarithmically spaced multi-sine test signal, it is defined as:

$$s(t) = \sum_i \left[\left[A + \mu_{AM} \cos(2\pi * f_{AM})\right] * \cos(2\pi * f_{0i})\right]$$

with $A = 0,5$

$f_{AM} = 4$ Hz, $\mu_{AM} = 0,5$

$f_{0\,i} = 250\text{Hz} * 2^{(i/3)}$;i=1..11

CF= 14dB ±1 dB (10 dB + 4,26 dB due to 100% AM modulation)

CF = Crest Factor = Peak to RMS ratio

The PN sequence is complying with ITU-T Recommendation P.501 with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB instead of 11 dB. The duration of the test signal is 250 ms. The test signal level is -3 dBm0 (from 100 Hz to 3.5 kHz).

The training sequence level shall be –16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1,0 s).

Note:

Full scale of coder input signal corresponds to +3,14 dBm0 with sinusoidal signal, CF= 3dB. A test signal with a CF of maximum 15 dB can thus have a level of up to –8,86 dBm0 without overloading the codec. In order to get best dynamic range the signal amplitude should be as high as possible.

The training sequence level shall be –16 dBm0 in order not to overload the codec. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Care should be taken that the terminal under test considers the test signal as a speech-like signal.

7.8 Distortion

7.8.1 Sending Distortion

The handset, headset, or hands-free UE is setup as described in clause 5.

The signal used is a sine-wave signal with a frequency of 1020 Hz. The sine-wave signal level shall be calibrated to the following RMS levels at the MRP: +5, 0, -4.7, -10, -15, -20 dBPa. The test signals have to be applied in this sequence, i.e. from high levels down to low levels.

The duration of the sine-wave signal is recommended to be 360 ms. The manufacturer shall be allowed to request tone lengths up to 1 second. The measured part of the signal shall be 170.667 ms (equals 2*4096 samples in a 48 kHz sample rate test system). The times are selected to be relatively short in order to reduce the risk that the test tone is treated as a stationary signal.

It is recommended that an optional activation signal be presented immediately preceding each test signal to ensure that the UE is in a typical state during measurement (see Note 1.). An appropriate speech or speech-like activation signal shall be chosen from ITU-T Recommendations P.501 or P.50. A recommendation for the use of an activation signal as part of the measurement is defined in Figure 16. The RMS level of the active parts of this activation signal is recommended to be equal to the subsequent test tone RMS level. In practice, certain types of processing may be impacted due to the introduction of the activation signal. The manufacturer shall be allowed to specify disabling of the activation signal. It shall be reported whether an activation signal was used or not, along with the characteristics of the activation signal, as specified by the manufacturer.

The ratio of the signal to total distortion power of the signal output of the SS shall be measured with the psophometric noise weighting (see ITU-T Recommendations G.712, O.41 and 0.132). 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).

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 * f_S, and an upper passband starting at 1.4142 * f_S with f_S being the frequency of the sine-wave signal. The passband ripple of the filter shall be less than 0.2 dB. The attenuation of the band stop filter at the sine-wave frequency shall be at least 60 dB. Alternatively the described characteristics can be implemented by an appropriate weighting on the spectrum obtained from an FFT ('Fast Fourier Transform'). The total distortion component is defined as the measured signal within the frequency range 200-4000 Hz, after applying psophometric and stop filters (hence no correction for the lost power due to the stop filter, known as 'bandwidth correction', shall be applied).

To increase the repeatability, considering the variability introduced by speech coding and voice processing, the test sequence (activation + test signal) may be repeated n times. The thus obtained single signal-to-total-distortion power ratios shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

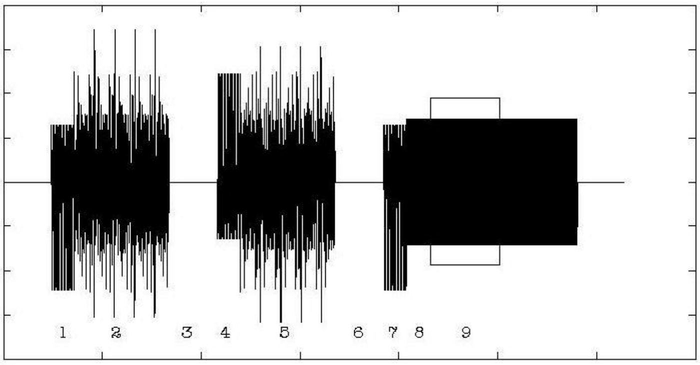


Figure 16: Recommended activation sequence and test signal.

The activation signal consists of a P.501 'Bandlimited composite source signal with speech-like power density spectrum' signal with 48.62 ms voiced part (1), 200 ms unvoiced part (2) and 101.38 ms pause (3), followed by the same signal but polarity inverted (4, 5, 6), followed by the voiced part only (7). The pure test tone is applied and after 50 ms settling time (8), the analysis is made over the following 170.667 ms (9).

NOTE 1: Void.

NOTE 2: In order to ensure that the correct part of the signal is analyzed, the total delay of the terminal and SS may have to be determined prior to the measurement.

NOTE 3: For handsfree terminals tested in environments defined in clause 6.1.2, care should be taken that the reverberation in the test room, caused by the activation signal, does not affect the test results to an unacceptable degree, refer to clause 5.3.

7.8.2 Receiving

The handset, headset, or hands-free UE is setup as described in clause 5.

The signal used is a sine-wave signal with frequency of 1020 Hz..

The signal shall be applied at the signal input of the SS at the following levels: 0, -3, -10, -16, -20, -30, -40, -45 dBm0. The test signals have to be applied in this sequence, i.e. from high levels down to low levels.

The duration of the sine-wave signal is recommended to be 360 ms. The manufacturer shall be allowed to request tone lengths up to 1 second. The measured part of the signal shall be 170.667 ms (equals 2*4096 samples in a 48 kHz sample rate test system). The times are selected to be relatively short in order to reduce the risk that the test tone is treated as a stationary signal.

It is recommended that an optional activation signal be presented immediately preceding each test signal to ensure that the UE is in a typical state during measurement (see Note 1.). An appropriate speech or speech-like activation signal shall be chosen from ITU-T Recommendations P.501 or P.50. A recommendation for the use of an activation signal as part of the measurement is defined in Figure 17. The RMS level of the active parts of this activation signal is for low and medium test levels recommended to be equal to the subsequent test tone RMS level. To avoid saturation of the SS speech encoder, it is for high test levels recommended that the activation signal level is adjusted so that its peak level equals the peak level of the test tone. In practice, certain types of processing may be impacted due to the introduction of the activation signal. The manufacturer shall be allowed to specify disabling of the activation signal. It shall be reported whether an activation signal was used or not, along with the characteristics of the activation signal, as specified by the manufacturer.

The ratio of the signal-to-total distortion power shall be measured at the applicable acoustic measurement point (ERP for handset mode, freefield for handsfree modes) with the psophometric noise weighting (see ITU-T Recommendations G.712, O.41 and 0.132). 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).

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 * f_S, and an upper passband starting at 1.4142 * f_S with f_S being the frequency of the sine-wave signal. The passband ripple of the filter shall be less than 0.2 dB. The attenuation of the band stop filter at the sine-wave frequency shall be at least 60 dB. Alternatively the described characteristics can be implemented by an appropriate weighting on the spectrum obtained from an FFT ('Fast Fourier Transform'). The total distortion component is defined as the measured signal within the frequency range 200-4000 Hz, after applying psophometric and stop filters (hence no correction for the lost power due to the stop filter, known as 'bandwidth correction', shall be applied).

To increase the repeatability, considering the variability introduced by speech coding and voice processing, the test sequence (activation + test signal) may be repeated n times. The thus obtained single signal-to-total-distortion power ratios shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

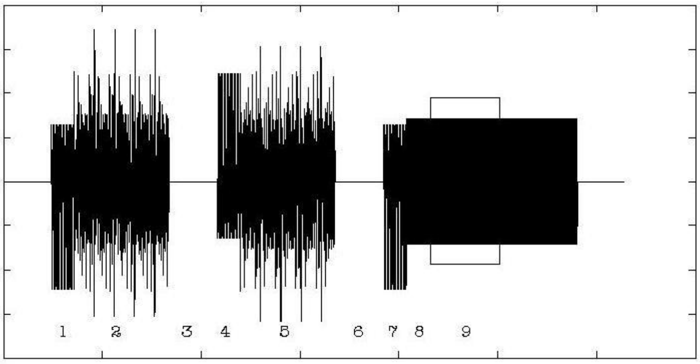


Figure 17: Recommended activation sequence and test signal.

The activation signal consists of a P.501 'Bandlimited composite source signal with speech-like power density spectrum' signal with 48.62 ms voiced part (1), 200 ms unvoiced part (2) and 101.38 ms pause (3), followed by the same signal but polarity inverted (4, 5, 6), followed by the voiced part only (7). The pure test tone is applied and after 50 ms settling time (8), the analysis is made over the following 170.667 ms (9).

NOTE 1: Void.

NOTE 2: In order to ensure that the correct part of the signal is analyzed, the total delay of the terminal and SS may have to be determined prior to the measurement.

NOTE 3: For handsfree terminals tested in environments defined in clause 6.1.2, care should be taken that the reverberation in the test room, caused by the activation signal, does not affect the test results to an unacceptable degree, refer to clause 5.3.

7.9 Ambient Noise Rejection

Handset and Headset UE:

NOTE: This section applies to terminals providing narrow- and wide-band telephony. However, the procedure for measuring ambient noise rejection is defined only over narrow-band frequency range. Thus the test method for ambient noise rejection is the same for either narrow- or wide-band telephony.

a) A 1/2 inch pressure microphone is calibrated using a known sound source and mounted at the MRP, without the HATS present. A frequency analyser is calibrated to enable the sound pressure levels at the microphone to be determined in 1/3rd Octave bands.

b) Flood the room in which the measurement is to be made with a band limited (100 Hz to 8 kHz) pink noise to within ±3 dB. The level at MRP shall be adjusted to 70 dB(A) (-24 dBPa(A)). The tolerance on this level is +/-1 dB. The resulting sound spectrum is P_{rn} dBPa, measured in 1/3rd Octave bands.

   To ensure that the sound field is diffuse enough, the following apply:

   The diffuse sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within +/- 3 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100 Hz to 3,15 kHz.

NOTE 1: The pressure intensity index, as defined in ISO 9614, may prove to be a suitable method for assessing the diffuse field.

NOTE 2: Where more than one loudspeaker is used to produce the desired sound field, the loudspeakers must be fed with non-coherent electrical signals to eliminate standing waves and other interference effects.

c) Position the HATS test head in the correct relative position to the MRP and mount the MS under test, according to clause 5.1.1. Recalibrate the 1/3rd Octave frequency analyser using a known voltage source to facilitate the analysis of the voltage V_{rn}, where V_{rn} is the voltage at the audio output of the SS due to the noise spectrum input.

d) Set up a speech path between the MS and the System Simulator (SS).

e) Determine, as a function of frequency, using the frequency analyser, in 1/3rd Octave bands (index j), the electrical output V_{jrn}, (expressed as dB rel . 1V) at the audio output of the SS for the applied acoustic pressure P_{jrn} (expressed as dB rel 1Pa) at the MRP. Since, the MS sending sensitivity is not defined above 3,4 kHz the measurement shall be cut off at 3,4 kHz. For the bands below 315 Hz, the noise level shall be referenced to the speech level at 315 Hz to yield the DELSM.

   The room noise sensitivity is expressed as:- $Sm_{jrn} = V_{jrn}$ (dBV) - P_{jrn} (dBPa).

   The MS ambient noise send sensitivity has now been determined.

f) The MS speech send sensitivity is now required. The required sensitivity is defined as the electrical output from the MS, measured at the audio output of the SS, as a function of the free field sound pressure at the MRP of the artificial mouth.

   The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P.50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The 1/2 inch pressure microphone is calibrated using a known sound source. The frequency analyser is calibrated to measure in 1/3rd Octave bands. 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 resulting sound spectrum is P_o dBPa, measured in 1/3rd Octave bands. The 1/3rd Octave frequency analyser should be re-calibrated, using a known voltage source, to facilitate the analysis of the voltage V_j.Where V_j is the voltage in each 1/3rd octave band at the audio output of the SS due to the test signal input. Set up a speech path between the MS and the SS. Determine the function of frequency, using the frequency analyser, and in 1/3rd Octave bands, the electrical output, V_j, (expressed as dB rel. 1V), at the audio output of the SS for the applied acoustic pressure, P_{j0}, (expressed as dB rel. 1Pa/V), at the MRP.

   The speech sending sensitivity is expressed as:

   Sm_{js} (dB) = V_j (dBV) - P_{jo} (dBPa) dBrel. 1V/Pa.

g) The difference of the room noise sensitivity and the speech sending sensitivity DELSM (Δ_{jSM}) in each 1/3rd Octave band for the MS is determined as:

   Sm_{jrn} - Sm_{js} (dB) (for j = 1 to 2, $Sm_{js} = Sm_{3s}$).

h) The Ambient noise rejection ANR is calculated as the single figure value according to the following formula, the ANR shall be ≥ 0dB.

$$ANR = -\frac{4}{5}\sum_{i=1}^{13}\Delta_{jSM} \cdot 10^{-0.0175W_{jsi}}$$

j = The index of third octave bands centered at frequencies from 200 Hz to 3 150 Hz inclusive.

Wjsi = The sending weighting factors from ITU-T Recommendation P.79 [16], table 1 for the jth 1/3rd Octave band centre frequency.

Hands-free UE (all categories):

No test method for hands-free operations.

8 Wideband telephony transmission performance test methods

8.1 Applicability

The test methods in this sub-clause shall apply when testing a UE that is used to provide narrow-band or wideband telephony, either as a stand-alone service, or as part of a multimedia service.

The application force used to apply the handset against the artificial ear shall be 8 ± 2 N. For the headset case, the application of the headset shall comply with P.57.

8.2 Overall loss/loudness ratings

8.2.1 General

The SLR and RLR values for the GSM or 3G networks apply up to the POI. However, the main determining factors are the characteristics of the UE, including the analogue to digital conversion (ADC) and digital to analogue conversion (DAC). In practice, it is convenient to specify loudness ratings to the Air Interface. For the normal case, where the GSM or 3G networks introduces no additional loss between the Air Interface and the POI, the loudness ratings to the PSTN boundary (POI) will be the same as the loudness ratings measured at the Air Interface.

8.2.2 Connections with handset UE

8.2.2.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The handset terminal is setup as described in subclause 5.

   The sending sensitivity shall be calculated from each band of the 20 frequencies given in table G.1 of ITU-T Recommendation P.79 Annex A, 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.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula (A-23b), over bands 1 to 20, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79 Annex A, table A2.

8.2.2.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The handset terminal is setup as described in subclause 5. The receiving sensitivity shall be calculated from each band of the 20 frequencies given in table A.2 of ITU-T Recommendation P.79 Annex A, 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.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [16], formula (A-23c), over bands 1 to 20, using m = 0,175 and the receiving weighting factors from table A.2 of ITU-T Recommendation P.79 Annex A [16].

d) DRP-ERP correction is applied. No leakage correction shall be applied.

8.2.3 Connections with Vehicle Mounted & Desktop Mounted hands-free UE

Vehicle mounted hands-free should be tested in the vehicle (for the totally integrated vehicle hands-free systems) or in a vehicle simulator, ref ETSI 0358 601 (TR101110) Digital Cellular Telecommunications System (Phase 2+) Characterization test methods and quality assessment for hands-free mobiles.

Free-field measurements for vehicle mounted hands-free are for further study.

8.2.3.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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 broadband signal level then is adjusted to –28,7 dBPa at the HFRP or the HATS HFRP (as defined in P.581) and the spectrum is not altered. The spectrum at the MRP and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5. The sending sensitivity shall be calculated from each band of the 20 frequencies given in table A.2 of ITU-T Recommendation P.79 Annex A, 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.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula (A-23b), over bands 1 to 20, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79 Annex A, table A.2.

8.2.3.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The hands-free terminal is setup as described in subclause 5. If HATS is used, then it is free-field equalized as described in ITU-T Recommendation P.581. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the 'right ' and 'left' signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. The receiving sensitivity shall be calculated from each band of the 20 frequencies given in table A.2 of ITU-T Recommendation P.79 Annex A, 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.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [16], formula (A-23c), over bands 1 to 20, using m = 0,175 and the receiving weighting factors from table A.2 of ITU-T Recommendation P.79 Annex A.

d) No leakage correction shall be applied. The hands-free correction as described in P.340 shall be applied. To compute Receiving loudness rating (RLR) for hands-free terminal (see also ITU-T Recommendation P.340), when using the combination of left and right ear signals from HATS the HFL_E has to be 8 dB, instead of 14 dB. For further information see ITU-T Recommendation P.581.

8.2.4 Connections with Handheld hands-free UE

8.2.4.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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 broadband signal level then is adjusted to –28,7 dBPa at the HFRP or the HATS HFRP (as defined in P.581) and the spectrum is not altered. The spectrum at the MRP and the actual level at the MRP (measured in third octaves) are used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. The sending sensitivity shall be calculated from each band of the 20 frequencies given in table A.2 of ITU-T Recommendation P.79 Annex A, 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.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula (A-23b), over bands 1 to 20, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79 Annex A, table A.2.

8.2.4.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. If HATS is used, then it is free-field equalized as described in ITU-T Recommendation P.581. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the 'right ' and 'left' signals are voltage-summed for each 1/3 octave frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. The receiving sensitivity shall be calculated from each band of the 20 frequencies given in table A.2 of ITU-T Recommendation P.79 Annex A, 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.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [16], formula (A-23c), over bands 1 to 20, using m = 0,175 and the receiving weighting factors from table A.2 of ITU-T Recommendation P.79 Annex A.

d) No leakage correction shall be applied. The hands-free correction as described in P.340 shall be applied. To compute the Receiving loudness rating (RLR) for hands-free terminals (see also ITU-T Recommendation P.340) when using the combination of left and right ear signals from HATS the HFL_E has to be 8 dB, instead of 14 dB. For further information see ITU-T Recommendation P.581.

8.2.5 Connections with headset UE

Same as for handset.

8.3 Idle channel noise (handset and headset UE)

For idle noise measurements in sending and receiving directions, care should be taken that only the noise is windowed out by the analysis and the result is not impaired by any remaining reverberation or by noise and/or interference from various sources. Some examples are air-conducted or vibration-conducted noise from sources inside or outside the test chamber, disturbances from lights and regulators, mains supply induced noise including grounding issues, test system and system simulator inherent noise as well as radio interference from the UE to test equipment such as ear simulators, microphone amplifiers etc.

8.3.1 Sending

The terminal should be configured to the test equipment as described in subclause 5.1.

The environment shall comply with the conditions described in subclause 6.1.

The noise level at the output of the SS is measured with A-weighting. The A-weighting filter is described in IEC 60651.

A test signal may have to be intermittently applied to prevent "silent mode" operation of the MS. This is for further study.

The measured part of the noise shall be 170.667 ms (equals 8192 samples in a 48 kHz sample rate test system). The spectral distribution of the noise is analyzed with an 8k FFT using a window, which has less than 0.1 dB leakage for non bin-centered signals. This can be achieved with a window function commonly known as 'flat top window'. Within the specified frequency range the FFT bin that has the highest level is searched. The level of this bin is the maximum level of a single frequency disturbance.

To increase the repeatability, the test sequence (optional activation + noise level measurement) may be repeated n times.

The thus obtained total noise powers shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

The thus obtained single frequency maximum powers shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

8.3.2 Receiving

The terminal should be configured to the test equipment as described in subclause 5.1.

The environment shall comply with the conditions described in subclause 6.1.

A test signal may have to be intermittently applied to prevent "silent mode" operation of the MS. This is for further study.

The noise shall be measured with A-weighting at the DRP with diffuse-field correction. The A-weighting filter is described in IEC 60651 [12].

The measured part of the noise shall be 170.667 ms (equals 8192 samples in a 48 kHz sample rate test system). The spectral distribution of the noise is analyzed with an 8k FFT using a window, which has less than 0.1 dB leakage for non bin-centered signals. This can be achieved with a window function commonly known as 'flat top window'. Within the specified frequency range the FFT bin that has the highest level is searched. The level of this bin is the maximum level of a single frequency disturbance.

To increase the repeatability, the test sequence (optional activation + noise level measurement) may be repeated n times.

The thus obtained total noise powers shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

The thus obtained single frequency maximum powers shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

8.4 Sensitivity/frequency characteristics

8.4.1 Handset and headset UE sending

The headset case is similar to the handset one, except for the application force.

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The handset terminal is setup as described in subclause 5.

   Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 8 kHz inclusive. For the calculation, the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa.

8.4.2 Handset and headset UE receiving

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The handset terminal is setup as described in subclause 5.

   Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 8 kHz inclusive. For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The HATS is diffuse-field equalized. The sensitivity is expressed in terms of dBPa/V. Information about correction factors is available in ITU-T Recommendation P.57.

Optionally, the measurements might be repeated with 2N and 13N application force. In these cases no normative values apply.

8.4.3 Vehicle Mounted & Desktop hands-free UE sending

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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 broadband signal level is then adjusted to –28,7 dBPa at the HFRP or the HATSvHFRP (as defined in P.581) and the spectrum is not altered. The spectrum at the MRP and the actual level at the MRP (measured in third octaves) are used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBV/Pa.

8.4.4 Vehicle Mounted & Desktop hands-free UE receiving

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The hands-free terminal is setup as described in subclause 5. If the HATS is used, then it is free-field equalized as described in ITU-T Recommendation P.581. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the 'right ' and 'left' signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 8 kHz inclusive. For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V.

8.4.5 Handheld hands-free UE sending

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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 broadband signal level is then adjusted to –24,3 dBPa at the HFRP or the HATS HFRP (as defined in subclause 8.2.3.1) and the spectrum is not altered. The spectrum at the MRP and the actual level at the MRP (measured in third octaves) are used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 8 kHz inclusive. For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBV/Pa.

8.4.6 Handheld hands-free UE receiving

a) The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. 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.

b) The hands-free terminal is setup as described in subclause 5.1.3.3. If the HATS is used, then it is free-field equalized as described in ITU-T Recommendation P.581 . The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the 'right ' and 'left' signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 8 kHz inclusive. For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V.

8.5 Sidetone characteristics

8.5.1 Connections with Handset UE

The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. The spectrum of the acoustic signal shall be produced by the HATS. The test signal level shall be -4,7 dBPa measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset UE is set up as described in clause 5. The application force shall be 13N on the artificial ear Type 3.3 or Type 3.4.

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

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 8 kHz inclusive. For the calculation, the averaged measured level at each frequency band (ITU-T Recommendation P.79, table 4, 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 ITU-T Recommendation P.79, using m = 0.225 and the weighting factors in table B2, unsealed condition, of ITU-T Recommendation P.79. No leakage correction (L_E) shall be applied. DRP-ERP correction is used.

8.5.2 Headset UE

The test signal to be used for the normative measurements shall be the artificial voice according to ITU-Recommendation P. 50. Alternatively a speech like test signal as described in ITU-T Recommendation P.501 may be used. The type of test signal used shall be stated in the test report. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be –4,7 dBPa measured at the MRP. The test signal level is averaged over the complete test signal sequence.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 for frequencies from 100 Hz to 8 kHz inclusive. For the calculation, the averaged measured level at each frequency band (ITU-T Recommendation P.79, table 4, bands 1 to 20) is referred to the averaged test signal level measured in each frequency band.

The sidetone path loss L_{meST} as expressed in dB shall be calculated from each band of the 20 frequencies given in table G.1 of ITU-T Recommendation P.79 Annex A, bands 1 to 20. The STMR (in dB) shall be calculated from the formula B-4 of ITU-T Recommendation P.79 [16], using m = 0.225 and the weighting factors in Table B.2, unsealed condition, of ITU-T Recommendation P.79 [16]. No leakage correction (L_E) shall be applied. DRP-ERP correction is used.

8.5.3 Hands-free UE (all categories)

No requirement for other than echo control.

8.5.4 Sidetone delay for handset or headset

The handset or the headset terminal is setup as described in clause 5.

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

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

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

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:

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

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

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

8.6 Stability loss

Where a user controlled volume control is provided it is set to maximum.

Handset UE: The handset is placed on a hard plane surface with the earpiece facing the surface.

Headset UE: The requirement applies for the closest possible position between microphone and headset receiver within the intended wearing position.

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

Hands-free UE (all categories): no requirement other than echo loss.

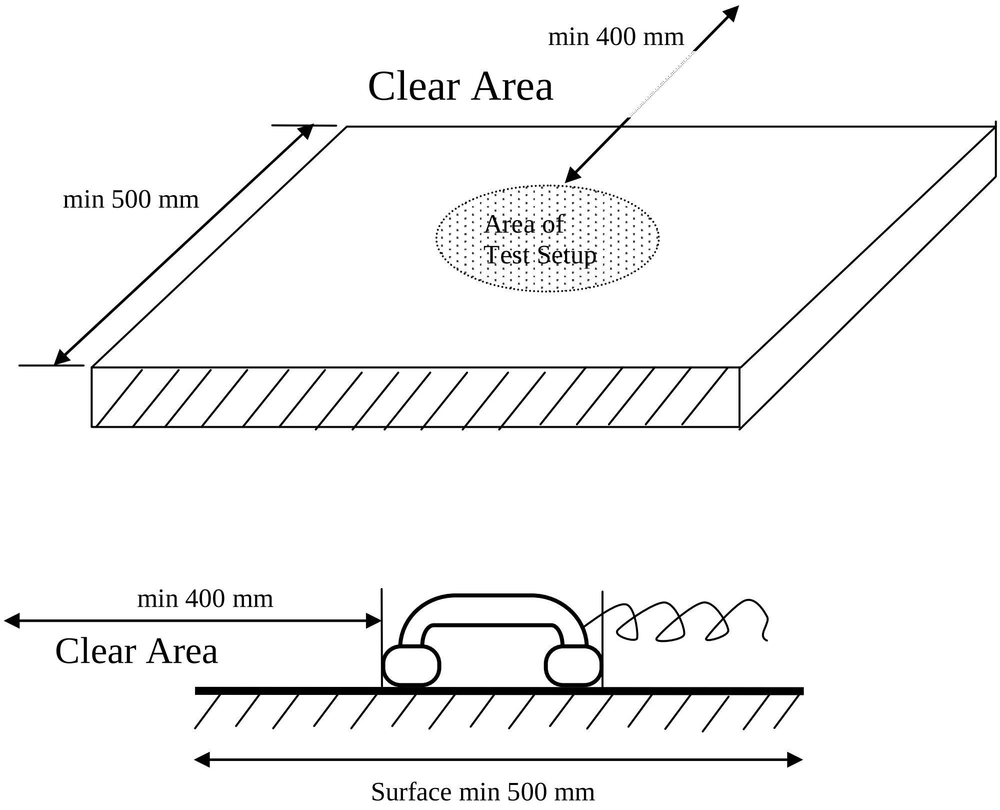
Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is applied. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB instead of 11 dB. The duration of the test signal is 250 ms. With an input signal of -3 dBm0, the attenuation from input to output of the system simulator shall be measured under the following conditions:

a) the handset or the headset, with the transmission circuit fully active, shall be positioned on a hard plane surface with at least 400 millimetre free space in all directions. The earpiece shall face towards the surface as shown in figure 17c;

b) the headset microphone is positioned as close as possible to the receiver(s) within the intended wearing position;

c) for binaural headset, the receivers are placed symmetrically around the microphone.



NOTE: All dimensions in mm.

Figure 17c

The attenuation from input to output shall be measured in the frequency range from 100 Hz to 8 kHz. The spectral distribution of the output signal is analysed with a 4k FFT (for the 48 kHz sampling rate), thus the measured part of the output signal is 85.333 ms. To avoid leakage effects the frequency resolution of the FFT must be the same as the frequency spacing of the PN sequence.

8.7 Acoustic echo control

8.7.1 General

The echo loss (EL) presented by the GSM or 3G networks at the POI should be at least 46 dB during single talk. This value takes into account the fact that UE is likely to be used in a wide range of noise environments.

8.7.2 Acoustic echo control in a hands-free UE

TCLw:

The hands-free is setup in a room with acoustic properties similar to a typical 'office-type' room; a vehicle-mounted hands-free UE should be tested in a vehicle or vehicle simulator, as specified by the UE manufacturer. [For reference on a suitable vehicle simulator see ETSI 0358 601 (TR101110) Digital Cellular Telecommunications System (Phase 2+).] The ambient noise level shall be less than -70 dBPa(A). The attenuation from reference point input to reference point output shall be measured using a speech like test signal.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is altered.

Either a logarithmically spaced multi-sine or a PN-sequence test signal shall be used.

When using a logarithmically spaced multi-sine test signal, it is defined as:

$$s(t) = \sum_i \left[\left[A + \mu_{AM} \cos\left(2\pi t * f_{AM}\right)\right] * \cos\left(2\pi t * f_{0i}\right)\right]$$

with A = 0,5

f_{AM} = 4 Hz, μ_{AM} = 0,5

$f_{0\,i}$= 250Hz * $2^{(i/3)}$;i=1..11

CF= 14dB ±1 dB (10 dB + 4,26 dB due to 100% AM modulation)

CF = Crest Factor = Peak to RMS ratio

The PN sequence is complying with ITU-T Recommendation P.501 with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB instead of 11 dB. The duration of the test signal is 250 ms. The test signal level is -3 dBm0 (from 100 Hz to 3.5 kHz).

The training sequence level shall be –16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1,0 s).

Note:

Full scale of coder input signal corresponds to +3,14 dBm0 with sinusoidal signal, CF= 3dB. A test signal with a CF of maximum 15 dB can thus have a level of up to –8,86 dBm0 without overloading the codec. In order to get best dynamic range the signal amplitude should be as high as possible.

The training sequence level shall be –16 dBm0 in order not to overload the codec. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Care should be taken that the terminal under test considers the test signal as a speech-like signal.

8.7.3 Acoustic echo control in a handset UE

The handset is set up according to clause 5. The ambient noise level shall be less than -64 dBPa(A). The attenuation from reference point input to reference point output shall be measured using the speech like test signal defined below.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is altered.

Either a logarithmically spaced multi-sine or a PN-sequence test signal shall be used.

When using a logarithmically spaced multi-sine test signal, it is defined as:

$$s(t) = \sum_i \left[\left[A + \mu_{AM} \cos\left(2\pi t * f_{AM}\right)\right] * \cos\left(2\pi t * f_{0i}\right)\right]$$

with A = 0,5

f_{AM} = 4 Hz, μ_{AM} = 0,5

$f_{0\,i}$= 250Hz * $2^{(i/3)}$;i=1..11

CF= 14dB ±1 dB (10 dB + 4,26 dB due to 100% AM modulation)

CF = Crest Factor = Peak to RMS ratio

The PN-sequence is complying with ITU-T Recommendation P.501 with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB instead of 11 dB. The length of the complete test signal composed of at least four sequences of CSS shall be at least one second (1,0 s). The test signal level is -3 dBm0 (from 50 Hz to 7 kHz).

The training sequence level shall be –16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1,0 s).

Note:

Full scale of coder input signal corresponds to +3,14 dBm0 with sinusoidal signal, CF= 3dB. A test signal with a CF of maximum 15 dB can thus have a level of up to –8,86 dBm0 without overloading the codec. In order to get best dynamic range the signal amplitude should be as high as possible.

The training sequence level shall be –16 dBm0 in order not to overload the codec. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Care should be taken that the terminal under test considers the test signal as a speech-like signal.

8.7.4 Acoustic echo control in a headset UE

The headset is set up according to clause 5. The ambient noise level shall be less than -64 dBPa(A). The attenuation from reference point input to reference point output shall be measured using the speech like test signal defined below.

Before the actual test, a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is altered.

Either a logarithmically spaced multi-sine or a PN-sequence test signal shall be used.

When using a logarithmically spaced multi-sine test signal, it is defined as:

$$s(t) = \sum_i \left[\left[A + \mu_{AM} \cos(2\pi t * f_{AM})\right] * \cos(2\pi t * f_{0i})\right]$$

with A = 0,5

f_{AM} = 4 Hz, μ_{AM} = 0,5

$f_{0\,i}$= 250Hz * $2^{(i/3)}$;i=1..11

CF= 14dB ±1 dB (10 dB + 4,26 dB due to 100% AM modulation)

CF = Crest Factor = Peak to RMS ratio

The PN-sequence is complying with ITU-T Recommendation P.501 with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB instead of 11 dB. The length of the complete test signal composed of at least four sequences of CSS shall be at least one second (1,0 s). The test signal level is -3 dBm0 (from 50 Hz to 7 kHz).

The training sequence level shall be –16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1,0 s).

Note:

Full scale of coder input signal corresponds to +3,14 dBm0 with sinusoidal signal, CF= 3dB. A test signal with a CF of maximum 15 dB can thus have a level of up to –8,86 dBm0 without overloading the codec. In order to get best dynamic range the signal amplitude should be as high as possible.

The training sequence level shall be –16 dBm0 in order not to overload the codec. The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation, the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Care should be taken that the terminal under test considers the test signal as a speech-like signal.

8.8 Distortion

8.8.1 Sending Distortion

The handset, headset, or hands-free UE is setup as described in clause 5.

The signal used is a sine-wave signal with frequencies specified in clause 6.8 of TS 26.131. The sine wave signal level shall be calibrated to -4,7 dBPa at the MRP for all frequencies, except for the sine wave with frequency 1020 Hz which shall be applied at the following levels at the MRP: +5, 0, -4.7, -10, -15, -20 dBPa. The test signals have to be applied in this sequence, i.e. from high levels down to low levels.

The duration of the sine-wave signal is recommended to be 360 ms. The manufacturer shall be allowed to request tone lengths up to 1 second. The measured part of the signal shall be 170.667 ms (equals 2*4096 samples in a 48 kHz sample rate test system). The times are selected to be relatively short in order to reduce the risk that the test tone is treated as a stationary signal.

It is recommended that an optional activation signal be presented immediately preceding each test signal to ensure that the UE is in a typical state during measurement (see Note 1.). An appropriate speech or speech-like activation signal shall be chosen from ITU-T Recommendations P.501 or P.50. A recommendation for the use of an activation signal as part of the measurement is defined in Figure 18. The RMS level of the active parts of this activation signal is recommended to be equal to the subsequent test tone RMS level. In practice, certain types of processing may be impacted due to the introduction of the activation signal. The manufacturer shall be allowed to specify disabling of the activation signal. It shall be reported whether an activation signal was used or not, along with the characteristics of the activation signal, as specified by the manufacturer.

The ratio of the signal to total distortion power of the signal output of the SS shall be measured with the psophometric noise weighting (see ITU-T Recommendations G.712, O.41 and O.132). 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).

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 $0.7071 * f_S$, and an upper passband starting at $1.4142 * f_S$ with f_S being the frequency of the sine-wave signal. The passband ripple of the filter shall be less than 0.2 dB. The attenuation of the band stop filter at the sine-wave frequency shall be at least 60 dB. Alternatively the described characteristics can be implemented by an appropriate weighting on the spectrum obtained from an FFT ('Fast Fourier Transform'). The total distortion component is defined as the measured signal within the frequency range 100-6000 Hz, after applying psophometric and stop filters (hence no correction for the lost power due to the stop filter, known as 'bandwidth correction', shall be applied).

To increase the repeatability, considering the variability introduced by speech coding and voice processing, the test sequence (activation + test signal) may be repeated n times. The thus obtained single signal-to-total-distortion power ratios shall be averaged. The total result shall be $10*\log_{10}$ of this average in dB.

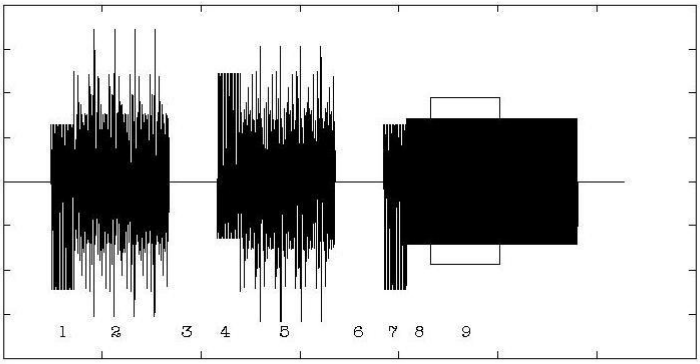


Figure 18: Recommended activation sequence and test signal.

The activation signal consists of a P.501 'Bandlimited composite source signal with speech-like power density spectrum' signal with 48.62 ms voiced part (1), 200 ms unvoiced part (2) and 101.38 ms pause (3), followed by the same signal but polarity inverted (4, 5, 6), followed by the voiced part only (7). The pure test tone is applied and after 50 ms settling time (8), the analysis is made over the following 170.667 ms (9).

NOTE 1: Depending on the type of codec the test signal used may need to be adapted. If a sine wave is not usable, an alternative test signal could be a band limited noise signal centered on the above frequencies.

NOTE 2: Void.

NOTE 3: Void.

NOTE 4: In order to ensure that the correct part of the signal is analyzed, the total delay of the terminal and SS may have to be determined prior to the measurement.

NOTE 5: For handsfree terminals tested in environments defined in clause 6.1.2, care should be taken that the reverberation in the test room, caused by the activation signal, does not affect the test results to an unacceptable degree, refer to clause 5.3.

8.8.2 Receiving

The handset, headset, or hands-free UE is setup as described in clause 5.

The signal used is a sine-wave signal with frequencies specified in clause 6.8 of TS 26.131. The signal level shall be -16 dBm0, except for the sine-wave signal with a frequency 1020 Hz that shall be applied at the signal input of the SS at the following levels: 0, -3, -10, -16, -20, -30, -40, -45 dBm0. The test signals have to be applied in this sequence, i.e. from high levels down to low levels.

The duration of the sine-wave signal is recommended to be 360 ms. The manufacturer shall be allowed to request tone lengths up to 1 second. The measured part of the signal shall be 170.667 ms (equals 2*4096 samples in a 48 kHz sample rate test system). The times are selected to be relatively short in order to reduce the risk that the test tone is treated as a stationary signal.

It is recommended that an optional activation signal be presented immediately preceding each test signal to ensure that the UE is in a typical state during measurement (see Note 1.). An appropriate speech or speech-like activation signal shall be chosen from ITU-T Recommendations P.501 or P.50. A recommendation for the use of an activation signal as part of the measurement is defined in Figure 19. The RMS level of the active parts of this activation signal is for low and medium test levels recommended to be equal to the subsequent test tone RMS level. To avoid saturation of the SS speech encoder, it is for high test levels recommended that the activation signal level is adjusted so that its peak level equals the peak level of the test tone. In practice, certain types of processing may be impacted due to the introduction of the activation signal. The manufacturer shall be allowed to specify disabling of the activation signal. It shall be reported whether an activation signal was used or not, along with the characteristics of the activation signal, as specified by the manufacturer.

The ratio of the signal-to-total distortion power shall be measured at the applicable acoustic measurement point (ERP for handset mode, freefield for handsfree modes) with the psophometric noise weighting (see ITU-T Recommendations G.712, O.41 and O.132). 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).

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 * f_S, and an upper passband starting at 1.4142 * f_S with f_S being the frequency of the sine-wave signal. The passband ripple of the filter shall be less than 0.2 dB. The attenuation of the band stop filter at the sine-wave frequency shall be at least 60 dB. Alternatively the described characteristics can be implemented by an appropriate weighting on the spectrum obtained from an FFT ('Fast Fourier Transform'). The total distortion component is defined as the measured signal within the frequency range 100-6000 Hz, after applying psophometric and stop filters (hence no correction for the lost power due to the stop filter, known as 'bandwidth correction', shall be applied).

To increase the repeatability, considering the variability introduced by speech coding and voice processing, the test sequence (activation + test signal) may be repeated n times. The thus obtained single signal-to-total-distortion power ratios shall be averaged. The total result shall be 10*$\log_{10}$ of this average in dB.

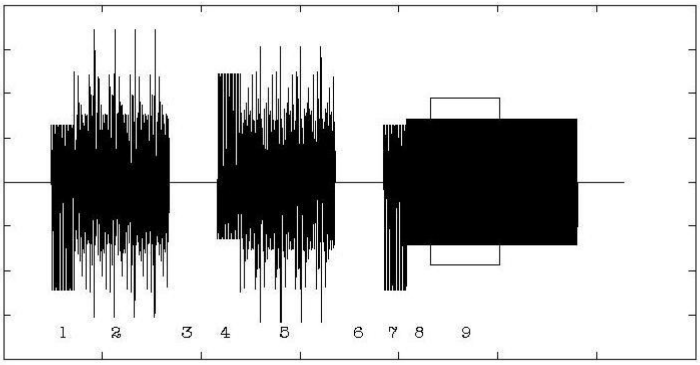


Figure 19: Recommended activation sequence and test signal.

The activation signal consists of a P.501 'Bandlimited composite source signal with speech-like power density spectrum' signal with 48.62 ms voiced part (1), 200 ms unvoiced part (2) and 101.38 ms pause (3), followed by the same signal but polarity inverted (4, 5, 6), followed by the voiced part only (7). The pure test tone is applied and after 50 ms settling time (8), the analysis is made over the following 170.667 ms (9).

NOTE 1: Void.

NOTE 2: Void.

NOTE 3: In order to ensure that the correct part of the signal is analyzed, the total delay of the terminal and SS may have to be determined prior to the measurement.

NOTE 4: For handsfree terminals tested in environments defined in clause 6.1.2, care should be taken that the reverberation in the test room, caused by the activation signal, does not affect the test results to an unacceptable degree, refer to clause 5.3.

8.9 Ambient Noise Rejection

Handset and Headset UE:

NOTE: This clause applies to terminals providing narrow- and wide-band telephony. However, the procedure for measuring ambient noise rejection is defined only over narrow-band frequency range. Thus the test method for ambient noise rejection is the same for either narrow- or wide-band telephony.

a) A 1/2 inch pressure microphone is calibrated using a known sound source and mounted at the MRP, without the HATS present. A frequency analyser is calibrated to enable the sound pressure levels at the microphone to be determined in 1/3rd Octave bands.

b) Flood the room in which the measurement is to be made with a band limited (100 Hz to 8 kHz) pink noise to within ±3 dB. The level at MRP shall be adjusted to 70 dB(A) (-24 dBPa(A)). The tolerance on this level is +/-1 dB. The resulting sound spectrum is P_{rn} dBPa, measured in 1/3rd Octave bands. To ensure that the sound field is diffuse enough, the following apply: the diffuse sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within +/- 3 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100 Hz to 6,3 kHz.

NOTE 1: The pressure intensity index, as defined in ISO 9614, may prove to be a suitable method for assessing the diffuse field.

NOTE 2: Where more than one loudspeaker is used to produce the desired sound field, the loudspeakers must be fed with non-coherent electrical signals to eliminate standing waves and other interference effects.

c) Position the HATS test head in the correct relative position to the MRP and mount the MS under test, according to clause 5.1.1. Recalibrate the 1/3rd Octave frequency analyser using a known voltage source to facilitate the analysis of the voltage V_{rn}, where V_{rn} is the voltage at the audio output of the SS due to the noise spectrum input.

d) Set up a speech path between the MS and the System Simulator (SS).

e) Determine, as a function of frequency, using the frequency analyser, in 1/3rd Octave bands (index j), the electrical output V_{jrn}, (expressed as dB rel . 1V) at the audio output of the SS for the applied acoustic pressure P_{jrn} (expressed as dB rel 1Pa) at the MRP. Since, the MS sending sensitivity is not defined above 6,3 kHz the measurement shall be cut off at 6,3 kHz. For the bands below 315 Hz, the noise level shall be referenced to the speech level at 315 Hz to yield the DELSM. The room noise sensitivity is expressed as: $Sm_{jrn} = V_{jrn}$ (dBV) - P_{jrn} (dBPa). The MS ambient noise send sensitivity has now been determined.

f) The MS speech send sensitivity is now required. The required sensitivity is defined as the electrical output from the MS, measured at the audio output of the SS, as a function of the free field sound pressure at the MRP of the artificial mouth. The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P.50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The 1/2 inch pressure microphone is calibrated using a known sound source. The frequency analyser is calibrated to measure in 1/3rd Octave bands. 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 resulting sound spectrum is P_o dBPa, measured in 1/3rd Octave bands. The 1/3rd Octave frequency analyser should be re-calibrated, using a known voltage source, to facilitate the analysis of the voltage V_j.Where V_j is the voltage in each 1/3rd octave band at the audio output of the SS due to the test signal input. Set up a speech path between the MS and the SS. Determine the function of frequency, using the frequency analyser, and in 1/3rd Octave bands, the electrical output, V_j, (expressed as dB rel. 1V), at the audio output of the SS for the applied acoustic pressure, P_{j0}, (expressed as dB rel. 1Pa/V), at the MRP. The speech sending sensitivity is expressed as:

Sm_{js} (dB) = V_j (dBV) - P_{jo} (dBPa) dBrel. 1V/Pa.

g) The difference of the room noise sensitivity and the speech sending sensitivity DELSM (Δ_{jSM}) in each 1/3rd Octave band for the MS is determined as:

$Sm_{jrn} - Sm_{js}$ (dB) (for j = 1 to 2, $Sm_{js} = Sm_{3s}$).

h) The Ambient noise rejection ANR is calculated as the single figure value according to the following formula, the ANR shall be ≥ 0dB.

$$ANR = -\frac{4}{5}\sum_{i=1}^{16}\Delta_{jSM} \cdot 10^{-0.0175 W_{jsi}}$$

j = The index of third octave bands centered at frequencies from 200 Hz to 6 300 Hz inclusive.

Wjsi = The sending weighting factors from ITU-T Recommendation P.79 Annex G [16], table G.1 for the jth 1/3rd Octave band centre frequency.

Hands-free UE (all categories):

No test method for hands-free operations.

Annex A (informative): Change history

TSG SA#	TSG doc	Spec	CR	Rev	Cat	Vers	New Vers	Subject
08							3.0.0	Approved
09	SP-000397	26.132	001		F	3.0.0	3.1.0	Handheld hands-free Test Setup
11	SP-010107	26.132	002	1	F	3.1.0	3.2.0	Harmonisation of test methods for acoustics between 3GPP and GSM
11		26.132				3.2.0	4.0.0	Release 4
11	SP-010107	26.132	003	1	B	4.0.0	5.0.0	Compatibility with testing wideband telephony transmission performance
13	SP-010454	26.132	004		B	5.0.0	5.1.0	Extended scope of test signals for Ambient Noise Rejection
13	SP-010454	26.132	006		F	5.0.0	5.1.0	Restricted scope of ITU-T P.501 test signals for 3G acoustic tests
13	SP-010454	26.132	008		A	5.0.0	5.1.0	Bandwidth of test signals for acoustic testing
15	SP-020080	26.132	011	1	A	5.1.0	5.2.0	Correction of references and editorial changes (wrong decimal separators)
16	SP-020435	26.132	016		F	5.2.0	5.3.0	Correction on ANR test for hands-free Ues
21	SP-030445	26.132	026		F	5.3.0	5.4.0	Loudness rating measurements at lower bit rates
25	SP-040649	26.132	028		C	5.4.0	6.0.0	Change of sending distortion test case
35	SP-070026	26.132	0030		F	6.0.0	6.1.0	Reference Update for ITU-T Recommendation P.57 'Artificial Ears'
35	SP-070026	26.132	0031	1	F	6.0.0	6.1.0	Update of reference [16] to P.79-2001 Annex G
35	SP-070026	26.132	0032	1	F	6.0.0	6.1.0	Distinction between narrow-band and wideband telephony in the frequency ranges and loudness rating and STMR weights, and in ANR calculation
36		26.132				6.1.0	7.0.0	Version for Release 7
38	SP-070759	26.132	0034	2	F	7.0.0	7.1.0	Changing the sidetone test to allow type 3.3 or 3.4 artificial ears
42	SP-080674	26.132	0035	1	F	7.1.0	7.2.0	Correction to allow wideband testing for GSM terminals
42	SP-080685	26.132	0037	3	C	7.2.0	8.0.0	Updated test methods for wideband terminal acoustics
43	SP-090015	26.132	0038		F	8.0.0	8.1.0	Clarification on Distortion with psophometric filter
43	SP-090018	26.132	0036	2	C	8.1.0	9.0.0	Speech and video telephony terminal acoustic test
45	SP-090568	26.132	0040	2	A	9.0.0	9.1.0	Correction of STMR calculation
45	SP-090573	26.132	0041	1	F	9.0.0	9.1.0	Handling Acoustic Testing with Noise Suppression Algorithms Employed
47	SP-100021	26.132	0042	1	F	9.1.0	9.2.0	Correction of distortion measurements
51	SP-110042	26.132	0043	5		9.2.0	10.0.0	Alignment of 3GPP Audio Test Case Specification

History

Document history		
V10.0.0	April 2011	Publication