ETSI TS 126 131 V10.1.0 (2011-04)

Technical Specification

Universal Mobile Telecommunications System (UMTS); LTE; Terminal acoustic characteristics for telephony; Requirements (3GPP TS 26.131 version 10.1.0 Release 10)

Reference
RTS/TSGS-0426131va10

Keywords
LTE, UMTS

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (http://webapp.etsi.org/IPR/home.asp).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

This Technical Specification (TS) has been produced by ETSI 3rd Generation Partnership Project (3GPP).

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.

The cross reference between GSM, UMTS, 3GPP and ETSI identities can be found under http://webapp.etsi.org/key/queryform.asp.

Contents

Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project (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 the present document, 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;

3 or greater indicates TSG approved document under change control.

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 document.

Introduction

The present document specifies minimum performance requirements for the acoustic characteristics of 3G terminals when used to provide narrow-band or wideband telephony.

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 the main body of the text; the test methods and considerations are described in TS 26.132.

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 minimum performance requirements for the acoustic characteristics of 3G terminals when used to provide narrow-band or wideband telephony.

The set of minimum performance requirements enables a guaranteed level of speech quality while taking possible physical limits of the terminal design into account. Some performance objectives are also defined, if such design limits can be overcome. Care must be taken in applying performance objectives in isolation, not to degrade overall end-user speech quality.

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 26.132: "Narrow-band speech telephony terminal acoustic characteristics - test methods".

[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. 380 (2003): "Electro-acoustic measurements on headsets".

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

[11] ITU-T Recommendation P.79 (1999) with Annex G (2001): "Calculation of loudness ratings for telephone sets".

[12] ITU-T Recommendation G.223: "Assumptions for the calculation of noise on hypothetical reference circuits for telephony".

[13] ITU-T Recommendations P.340: "Transmission characteristics of hands-free telephones".

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

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

[16] 3GPP TS 06.77 R99 Minimum Performance Requirements for Noise Suppresser Application to the AMR Speech Encoder.

3 Definitions, symbols and abbreviations

3.1 Definitions

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

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 (0dBPa is equivalent to 94dB 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
DAC	Digital to Analogue Converter
DAI	Digital Audio Interface
DRP	Eardrum Reference Point
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
UPCMI	13-bit Uniform PCM Interface

4 Interfaces

The interfaces required to define terminal acoustic characteristics are shown in figure 1. These are the air interface and the point of interconnect (POI).

The Air Interface is specified by the 3G 25 series specifications and is required to achieve user equipment (UE) transportability. Analogue 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. Analogue 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;
- Desktop-mounted hands-free UE including softphone UE with external loudspeaker(s) used in handsfree mode;
- Vehicle-mounted hands-free UE including softphone UE mounted in a vehicle;
- 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 requirements and performance objectives for a softphone UE shall be derived according to the following rules:

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

5 Narrow-band telephony transmission performance

5.1 Applicability

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

5.2 Overall loss/loudness ratings

5.2.1 General

An international connection involving a 3G network and the PSTN should meet the overall loudness rating (OLR) limits in ITU-T Recommendation G.111. The national parts of the connection should therefore meet the send and receive loudness rating (SLR, RLR) limits in ITU-T Recommendation G.121.

For the case where digital routings are used to connect the 3G network to the international chain of circuits, the SLR and RLR of the national extension will be largely determined by the SLR and RLR of the 3G network. The limits given below are consistent with the national extension limits and long term objectives in ITU-T Recommendation G.121.

The SLR and RLR values for the 3G network 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 3G network 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. However, in some cases loss adjustment may be needed for interworking situations in individual countries.

5.2.2 Connections with handset UE

The nominal values of SLR/RLR to the POI shall be:

SLR = 8 +/- 3 dB;

RLR = 2 +/- 3 dB.

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

With the volume control set to the minimum position the RLR shall not be ≥ (equal or quieter than) 18 dB.

Compliance shall be checked by the relevant tests described in TS 26.132.

NOTE: The mechanical design of some UE may make it impossible to seal the ear-piece to the knife edge of the ITU-T artificial ear. Minimal additional methods may be used to provide the seal provided that they do not affect the mounting position of the UE with respect to the Mouth Reference Point and the Ear Reference Point.

5.2.3 Connections with Desktop and Vehicle-mounted hands-free UE

The nominal values of SLR/RLR to/from the POI shall be:

SLR = 13 +/- 4 dB;

RLR = 2 +/- 4 dB (for vehicle-mounted hands-free UE)

RLR = 5 +/- 4 dB (for desktop hands-free UE)

1. In case of a vehicle-mounted hands-free :
   Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control. It is recommended that a volume control giving at least 15 dB increase from the nominal RLR (louder) is provided for hands-free units intended to work in the vehicle environment. This is to allow for the increased acoustic noise level in a moving vehicle.
   RLR at maximum volume control setting should be ≤ (equal or louder than) -2 dB

2. In case of desktop hands-free:
   Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control. It is recommended that a volume control giving at least 15 dB increase from the nominal RLR (louder) is provided for hands-free units. This is to allow for the increased acoustic noise level in the environment.
   RLR at maximum volume control setting should ≤ (equal or louder than) 1 dB

Compliance shall be checked by the relevant tests described in TS 26.132.

NOTE: The target value for nominal RLR, as recommended in ITU-T G.111 Annex B – Table B.1, lies between 1 and 3dB. The higher RLR requirement of 5dB for desktop hands-free is appreciative of the limitations in transducer output with current typical form factors.

5.2.4 Connections with handheld hands-free UE

The nominal values of SLR/RLR to/from the POI shall be:

SLR = 13 +/- 4 dB;

RLR = 9 +9 / - 7 dB.

As performance objective it is recommended that the RLR at maximum volume control is ≤ (equal or louder than) 2 dB.

Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control. It is recommended that a volume control of at least 15 dB of control range. Compliance shall be checked by the relevant tests described in TS 26.132.

NOTE: The target value for nominal RLR, as recommended in ITU-T G.111 Annex B – Table B.1, lies between 1 and 3dB. The higher RLR requirement of 9dB for handheld hands-free is appreciative of the limitations in transducer output with typical form factors.

5.2.5 Connections with headset UE

The nominal values of SLR/RLR to/from the POI shall be:

SLR = 8 +/- 3 dB;

RLR = 2 +/- 3 dB;

RLR (binaural headset) = 8 +/- 3 dB for each earphone.

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

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

Compliance shall be checked by the relevant tests described in 3GPP TS 26.132.

5.3 Idle channel noise (handset and headset UE)

5.3.1 Sending

The maximum noise level produced by the apparatus at the output of the SS under silent conditions in the sending direction shall be ≤ -64 dBm0p.

NOTE 1: This level includes the eventual noise contribution of an acoustic echo canceller under the condition that no signal is received.

NOTE 2: This figure applies to the total noise level with psophometric weighting. It is recommended that the level of single frequency disturbances should be ≤ -74 dBm0p in the frequency range from 300 Hz to 3.4 kHz.

Compliance shall be checked by the relevant test described in TS 26.132.

5.3.2 Receiving

The maximum (acoustic) A-weighted noise level at the handset and headset UE when no signal is applied to the input of the SS shall be as follows:

If no user-controlled receiving volume control is provided, or, if it is provided, at the setting of the user-controlled receiving volume control at which the RLR is equal to the nominal value, the noise measured at the DRP with diffuse-field correction contributed by the receiving equipment alone shall be ≤ -57 dB Pa(A).

Where a volume control is provided, the measured noise shall also be ≤ -54 dB Pa(A) at the maximum setting of the volume control.

For the nominal volume control setting, the level of single frequency disturbances should be ≤ -60 dBPa(A) in the frequency range from 100 Hz to 10 kHz. As a performance objective it is recommended that the level should be ≤ -64 dBPa(A).

NOTE: In a connection with the PSTN, noise conditions as described in ITU-T Recommendation G.103 can be expected at the input (POI) of the 3G network. The characteristics of this noise may be influenced by the speech transcoding process (for further study).Compliance shall be checked by the relevant test described in 3GPP TS 26.132.

5.4 Sensitivity/frequency characteristics

5.4.1 Handset and headset UE sending

The sensitivity/frequency characteristics shall be as follows:

The sending sensitivity frequency response, measured either from the mouth reference point (MRP) to digital interface or from the MRP to the SS audio output (digital output of the reference speech decoder of the SS), shall be within a mask, which can be drawn between the points given in table 1. The mask is drawn with straight lines between the breaking points in table 1 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 1: Sending sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	-12	-
200	0	-
300	0	-12
1 000	0	-6
2 000	4	-6
3 000	4	-6
3 400	4	-9
4 000	0	

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

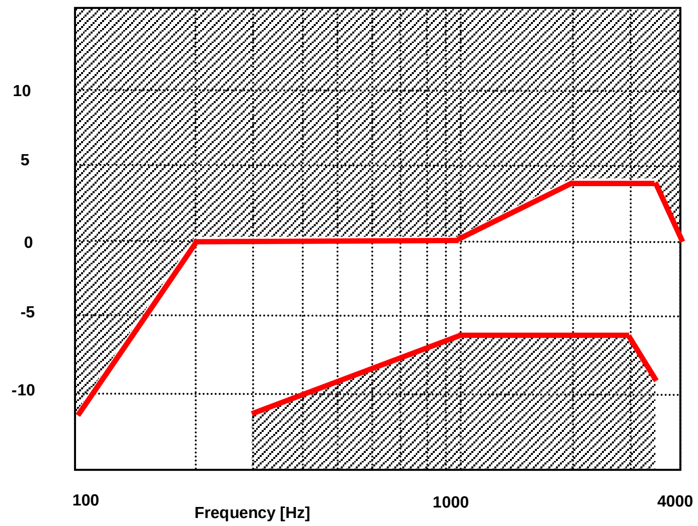


Figure 1: Sending sensitivity/frequency mask

Compliance shall be checked by the relevant test described in TS 26.132.

5.4.2 Handset and headset UE receiving

The sensitivity/frequency characteristics shall be as follows:

The receiving sensitivity frequency response, measured either from the digital interface to the DRP with diffuse-field correction or from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the DRP with diffuse-field correction shall be within a mask, which can be drawn with straight lines between the breaking points in table 2 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 2: Normative values Receiving sensitivity/frequencymask for 8N application force

Frequency	Upper Limit 8N	Lower Limit 8N
100 Hz	6	
300 Hz	6	-6
3 400 Hz	6	-6
4 000 Hz	6	

NOTE 1: The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale.
All sensitivity values are expressed in dB on an arbitrary scale

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

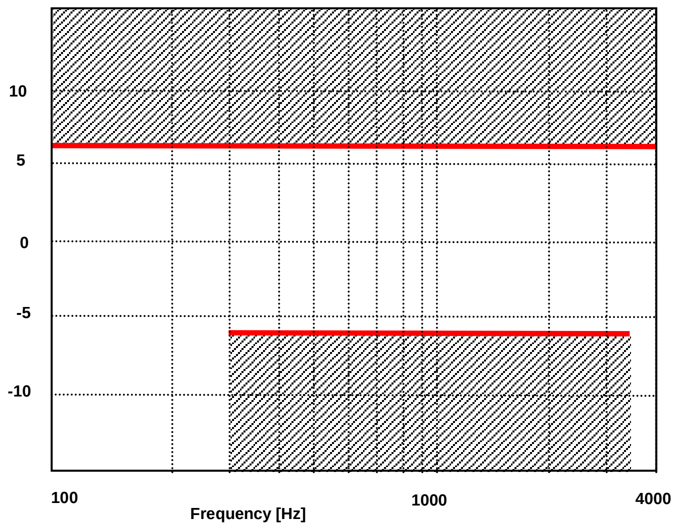


Figure 2: Normative values Receiving sensitivity/frequency mask for 8N application force

Compliance shall be checked by the relevant test described in TS 26.132.

5.4.3 Desktop and Vehicle-mounted hands-free UE sending

The sending sensitivity frequency response from the MRP to the SS audio output (digital output of the reference speech decoder of the SS) shall be as follows:

The sending sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 3 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 3: Sending sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	-12	-
200	0	-
300	0	-12
1 000	0	-6
2 000	4	-6
3 000	4	-6
3 400	4	-9
4 000	0	
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

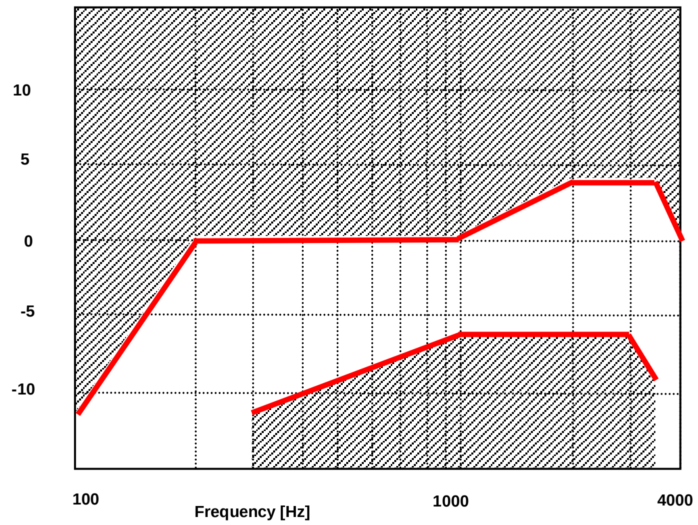


Figure 3: Sending sensitivity/frequency mask

Compliance shall be checked by the relevant test described in TS 26.132.

5.4.4 Desktop and Vehicle-mounted hands-free UE receiving

The receiving sensitivity frequency response from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the free-field shall be as follows:

The receiving sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 4 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 4: Receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
200	6	
250	6	
315	6	-9
400	6	-6
3 100	6	-6
4 000	6	
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

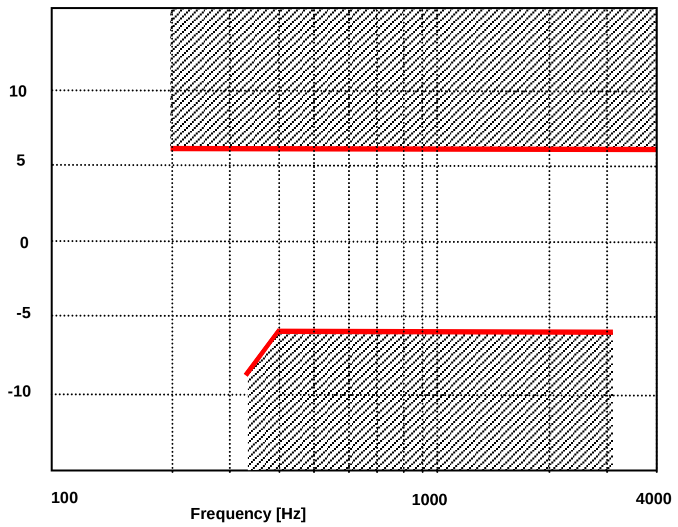


Figure 4: Receiving sensitivity/frequency mask

Compliance shall be checked by the relevant test described in TS 26.132.

5.4.5 Handheld hands-free UE sending

The sending sensitivity frequency response from the MRP to the SS audio output (digital output of the reference speech decoder of the SS) shall be as follows:

The sending sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 5 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 5: Sending sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	-12	-
200	0	-
300	0	-12
1 000	0	-6
2 000	4	-6
3 000	4	-6
3 400	4	-9
4 000	0	
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

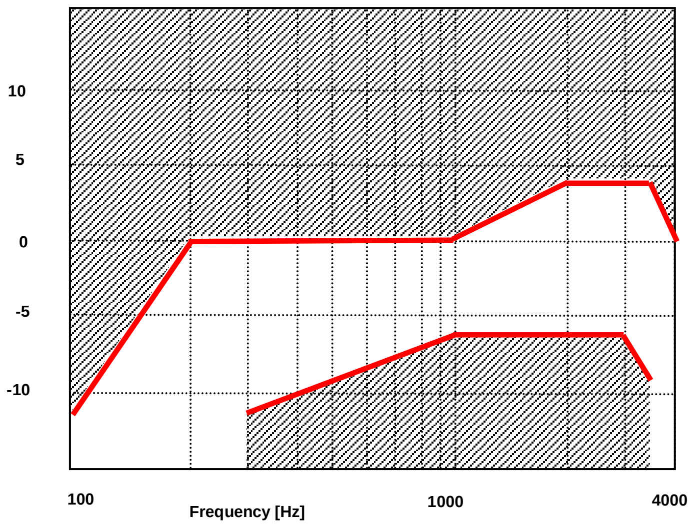


Figure 5: Sending sensitivity/frequency mask

Compliance shall be checked by the relevant test described in TS 26.132.

5.4.6 Handheld hands-free UE receiving

The receiving sensitivity frequency response from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the free-field shall be as follows:

The receiving sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 6 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 6: Hands-free receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
200	6	
500	6	-9 (Note 2)
630	6	-6 (Note 2)
800	6	-6
3 100	6	-6
4 000	6	-
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale.		
NOTE 2: The values stated in the Table 6 for 500 and 630 Hz are listed for performance objective purposes. (not mandatory)		

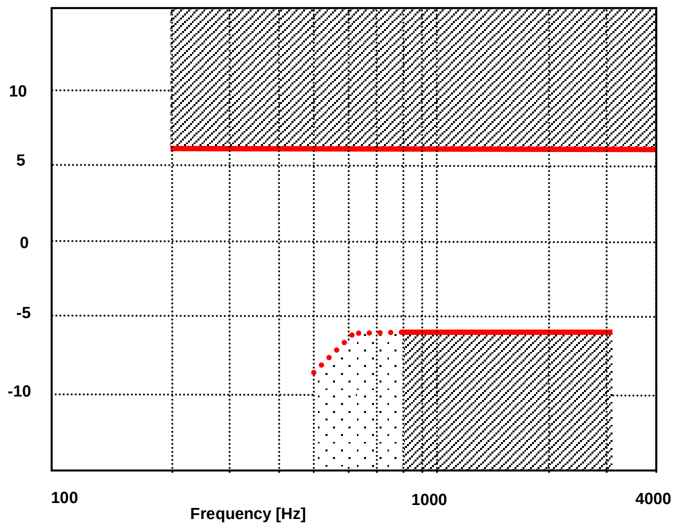


Figure 6: Hands-free receiving sensitivity/frequency mask

Compliance shall be checked by the relevant test described in TS 26.132.

5.5 Sidetone characteristics (handset and headset UE)

5.5.1 Sidetone loss

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

Compliance shall be checked by the relevant test described in 3GPP TS 26.132. The bandwidth for the sidetone path provided by the UE may in some terminals not be restricted to the narrow-band range. In case the sidetone path operates in a mode other than narrowband (to be declared by the manufacturer), compliance shall be checked using the test described for 'Wideband telephony transmission performance'.

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

NOTE 2: It is in general recommended to provide a terminal sidetone path for handset and headset UEs.

NOTE 3: In case the human air conducted sidetone paths are obstructed (one example being some binaural insert type headset UEs), it is important to provide a terminal sidetone path.

5.6 Stability loss

The stability loss presented to the PSTN by the 3G network at the POI should meet the principles of the requirements in clauses 2 and 3 of ITU-T Recommendation G.122. These requirements will be met if the attenuation between the digital input and digital output at the POI is at least 6 dB at all frequencies in the range 200 Hz to 4 kHz under the worst case acoustic conditions at the UE (any acoustic echo control should be enabled). For the normal case of digital connection between the Air Interface and the POI, the stability requirement can be applied at the Air Interface.

The worst case acoustic conditions will be as follows (with any volume control set to maximum):

Handset UE: the handset lying on, and the transducers facing, a hard surface with the ear-piece uncapped.

Headset UE: for further study

Handsfree UE: no requirement other than echo loss.

NOTE: The test procedure must take into account the switching effects of echo control and discontinuous transmission (DTX).

5.7 Acoustic echo control

5.7.1 General

The echo loss (EL) presented by the 3G network at the POI should be sufficient during single talk takeing into account the fact that UE is likely to be used in connections with high transmission delay and a wide range of noise environments.
See ITU-T Recommendation G.131 for general guidance.

The use of acoustic echo control is not mandated for 3G networks and the connection between the UE and the POI is zero loss. Therefore the acoustic echo control provided in UE should provide a sufficient TCLw at the POI over the likely range of acoustic end delays.

If acoustic echo control is provided by voice switching, comfort noise should be injected. This comfort noise shall operate in the same way to that used in DTX.

5.7.2 Acoustic echo control in a Desktop and Vehicle-mounted hands-free UE

The TCLw for the desktop and vehicle-mounted hands-free UE shall be ≥ 40 dB for any setting of the volume control.

The TCLw for the desktop hands-free and vehicle-mounted hands-free UE shall be ≥ 46 dB when measured under free-field conditions at nominal setting of volume control.

NOTE: A TCLw for the desktop hands-free and vehicle-mounted hands-free UE of ≥ 55 dB is a performance objective when measured under free-field conditions at nominal setting of volume control.

The echo canceller should be designed to cope with the expected reverberation and dispersion. In the case of the hands-free UE, this reverberation and dispersion may be time variant. Compliance with this requirement shall be checked by the relevant test described in TS 26.132.

5.7.3 Acoustic echo control in a handheld hands-free UE

The TCLw for handheld hands-free UE shall be ≥ 40 dB for any setting of the volume control.

The TCLw for handheld hands-free UE shall be ≥ 46 dB at nominal setting of volume control.

NOTE: A TCLw for the handheld hands-free UE of ≥ 55 dB is a performance objective when measured under free-field conditions at nominal setting of volume control.

The echo canceller should be designed to cope with the expected reverberation and dispersion. In the case of the hands-free UE, this reverberation and dispersion may be time variant. Compliance with this requirement shall be checked by the relevant test described in TS 26.132.

5.7.4 Acoustic echo control in a handset UE

The TCLw for handset UE shall be ≥ 46 dB for any setting of the volume control.

The TCLw for handset UE shall be ≥ 55 dB at nominal setting of volume control.

NOTE: It is recommended that the volume control should be set back to nominal after each call unless TCLw ≥ 55 dB can be maintained also with maximum volume setting.

The echo canceller should be capable of dealing with the variations in handset positions when in normal use. The implications of this are under study. Compliance with this requirement shall be checked by the relevant test described in TS 26.132.

5.7.5 Acoustic echo control in a headset UE

The TCLw for headset UE shall be ≥ 46 dB for any setting of the volume control.

The TCLw for headset UE shall be ≥ 55 dB at nominal setting of volume control.

NOTE: It is recommended that the volume control should be set back to nominal after each call unless TCLw ≥ 55 dB can be maintained also with maximum volume setting.

The echo canceller should be designed to cope with the expected reverberation and dispersion. Compliance with this requirement shall be checked by the relevant test described in TS 26.132.

5.8 Distortion

5.8.1 Sending Distortion

The sending part shall meet the following distortion requirements:

NOTE 1: Digital signal processing other than the transcoder itself is included in this requirement (e.g. echo cancelling).

Distortion shall be measured between MRP and the SS audio output (output of the reference speech decoder of the SS). The ratio of signal-to-total distortion power measured with the proper noise weighting (see table 4 of ITU-T Recommendation G.223) shall be above the limits given in table 7.

Table 7: Limits for signal-to-total distortion ratio

Sending level (dBPa at the MRP)	Sending Ratio (dB)
+5	30
0	35
-4,7	35
-10	33
-15	30
-20	27

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) - linear (dB ratio) scale.

Compliance of the sending distortion shall be checked by the test described in TS 26.132.

NOTE 2: It should be ensured that the test signal is treated by speech processing algorithms as a speech-like signal, and not a noise-like signal. Test signals with a time-stationary envelope may be treated by certain algorithms, e.g. noise suppression algorithms defined in 3GPP TS 06.77 R99[16], as a noise-like signal. If speech processing algorithms, including but not limited to noise suppression algorithms, are shown to treat the test signal as a noise-like signal, even where an activation signal has been utilized, then the test should be repeated with said speech processing algorithms disabled. The results of both sets of tests and the state of the processing algorithms should be documented in the test report.

5.8.2 Receiving

The receiving part between the SS audio input (input of the reference speech encoder of the SS) and DRP with diffuse-field correction shall meet the requirements at the nominal setting of the volume control:

The ratio of signal-to-total distortion power measured with the proper noise weighting (see table 4 of ITU-T Recommendation G.223) shall be above the limits given in table 8 when the sound pressure at DRP with diffuse-field correction is up to +10 dBPa. For sound pressures exceeding +10 dBPa at the DRP with diffuse-field correction there is no distortion requirement.

Table 8: Limits for signal-to-total distortion ratio

Receiving level at the digital interface (dBm0)	Receiving Ratio (dB)
0	25,5
-3	31,2
-10	33,5
-20	33,0
-30	30,5
-40 (*)	22,5 (*)
-45 (*)	17,5 (*)
NOTE : (*) For levels -40 and -45 dBm0, the stated limits are recommendations; hence a lower signal-to-distortion ratio shall not be regarded as a failing result. However, the obtained results shall be reported.	

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) - linear (dB ratio) scale.

Compliance of the receiving distortion shall be checked by the appropriate test method in TS 26.132.

NOTE 1: It should be ensured that the test signal is treated by speech processing algorithms as a speech-like signal, and not a noise-like signal. Test signals with a time-stationary envelope may be treated by certain algorithms, e.g. noise suppression algorithms defined in 3GPP TS 06.77 R99[16], as a noise-like signal. If speech processing algorithms, including but not limited to noise suppression algorithms, are shown to treat the test signal as a noise-like signal, even where an activation signal has been utilized, then the test should be repeated with said speech processing algorithms disabled. The results of both sets of tests and the state of the processing algorithms should be documented in the test report.

5.9 Ambient Noise Rejection

Handset and Headset UE:

The nature of mobile telephony is such that the UE will typically be operated in high ambient acoustic noise. Due to the adverse interaction of noise signals with speech codecs operating at lower rates, for example 8kbit/s or less, a minimum noise rejection specification is required.

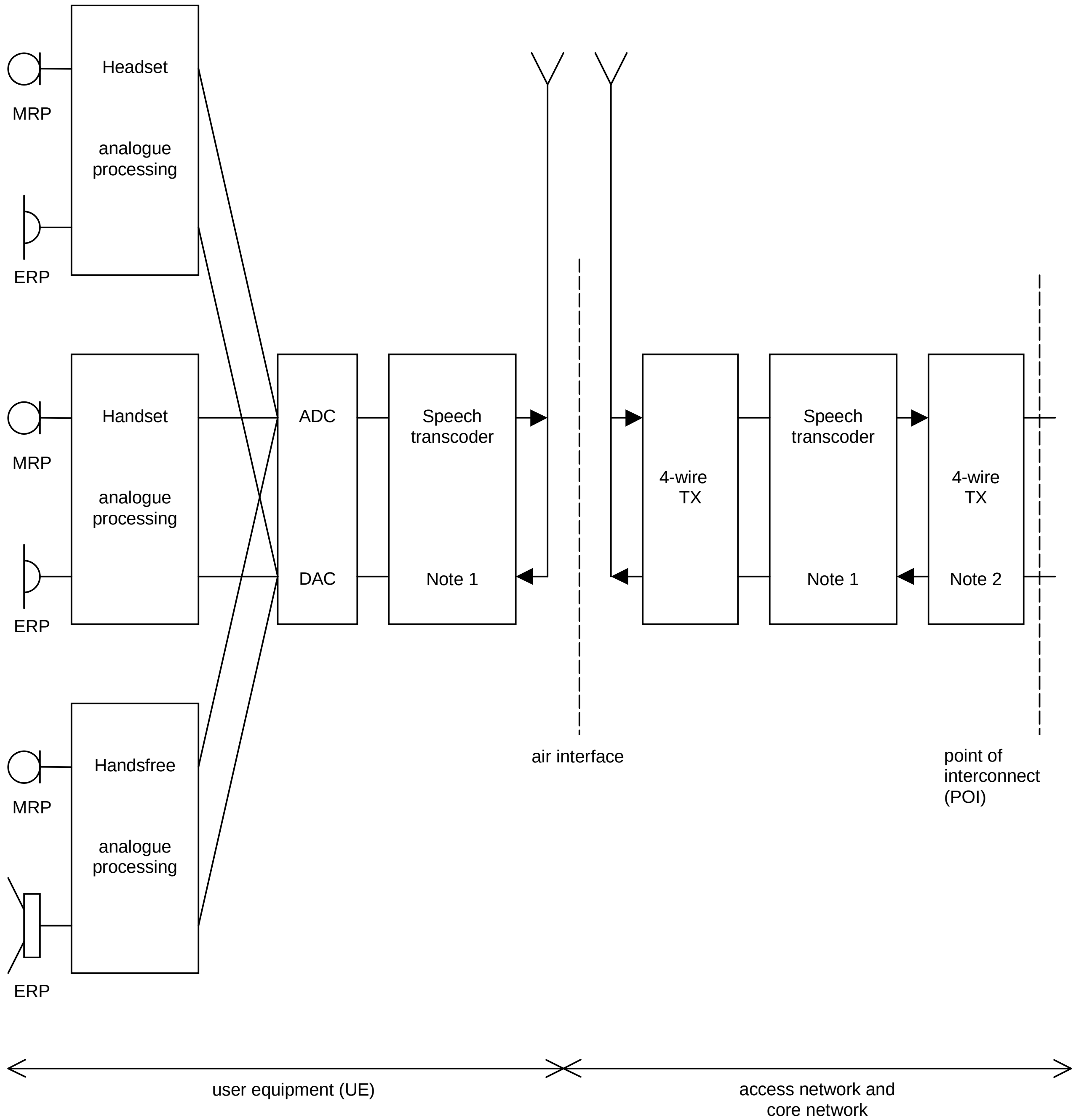
The UE ambient noise rejection ANR, calculated as a Single Figure DELSM (SFDELSM) shall be greater than or equal to the nominal value of 0dB. Due to the uncertainty inherent in the measurement method for ANR, a 3dB tolerance is allowed on the nominal value.

For good performance, it is recommended that a figure of +3 dB should be achieved.

Compliance shall be checked by the relevant test described in 3GPP TS 26.132.

Hands-free UE (all categories):

No requirement in hands-free operations.



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

Figure 1: 3G Interfaces for specification and testing of terminal narrow-band acoustic characteristics

5.10 Information on other Parameters (not normative)

Information about additional parameters relevant to speech quality e.g. for terminals where signal processing is used can be found in ITU-T Recommendations P.340, P.501 and P.502.

6 Wideband telephony transmission performance

6.1 Applicability

The performance requirements in this sub-clause shall apply when UE is used to provide wideband telephony, either as a stand-alone service, or as part of a multimedia service. The requirements in the clause apply only when the far-end terminal is also providing wideband, and not narrow-band telephony. When a wideband-enabled terminal is providing narrow-band telephony, the requirements in clause 5, "narrow-band telephony transmission performance" shall apply.

6.2 Overall loss/loudness ratings

6.2.1 General

An international connection involving a 3G network and the PSTN should meet the overall loudness rating (OLR) limits in ITU-T Recommendation G.111. The national parts of the connection should therefore meet the send and receive loudness rating (SLR, RLR) limits in ITU-T Recommendation G.121.

For the case where digital routings are used to connect the 3G network to the international chain of circuits, the SLR and RLR of the national extension will be largely determined by the SLR and RLR of the 3G network. The limits given below are consistent with the national extension limits and long term objectives in ITU-T Recommendation G.121. The SLR and RLR values for the 3G network 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 3G network 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. However, in some cases loss adjustment may be needed for interworking situations in individual countries.

6.2.2 Connections with handset UE

The nominal values of SLR/RLR to the POI shall be:

SLR = 8 +/- 3 dB;

RLR = 2 +/- 3 dB.

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

With the volume control set to the minimum position the RLR shall not be ≥ (equal or quieter than) 18 dB.

Compliance shall be checked by the relevant tests described in TS 26.132.

6.2.3 Connections with Desktop and Vehicle-mounted hands-free UE

The nominal values of SLR/RLR to/from the POI shall be:

SLR = 13 +/- 4 dB;

RLR = 2 +/- 4 dB (for vehicle-mounted hands-free UE);

RLR = 5 +/- 4 dB (for desktop hands-free UE).

1. In case of a vehicle-mounted hands-free :
   Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control. It is recommended that a volume control giving at least 15 dB increase from the nominal RLR (louder) is provided for hands-free units intended to work in the vehicle environment. This is to allow for the increased acoustic noise level in a moving vehicle.
   RLR at maximum volume control setting should be ≤ (equal or louder than) -2 dB

2. In case of desktop hands-free:
   Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control. It is recommended that a volume control giving at least 15 dB increase from the nominal RLR (louder) is provided for hands-free units. This is to allow for increased acoustic noise level in the environment.
   RLR at maximum volume control setting should ≤ (equal or louder than) 1 dB

Compliance shall be checked by the relevant tests described in TS 26.132.

NOTE: The target value for nominal RLR, as recommended in ITU-T G.111 Annex B – Table B.1, lies between 1 and 3dB. The higher RLR requirement of 5dB for desktop hands-free is appreciative of the limitations in transducer output with current typical form factors.

6.2.4 Connections with handheld hands-free UE

The nominal values of SLR/RLR to/from the POI shall be:

SLR = 13 +/- 4 dB;

RLR = 9 +9/- 7 dB.

Where a user controlled volume control is provided, the RLR shall meet the selected nominal value for at least one setting of the control.

The value of RLR at maximum volume control shall be ≤ (equal or louder than) 12 dB. As performance objective it is recommended that the RLR at maximum volume control is ≤ (equal or louder than) 2 dB.

Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control. It is recommended that a volume control is giving at least 15 dB of control range.

Compliance shall be checked by the relevant tests described in TS 26.132.

6.2.5 Connections with headset UE

The SLR and RLR should be measured and computed using methods given in ITU-T Recommendation P.38. This Recommendation currently gives a measuring technique for supra-aural earphone and insert type receivers. Study is continuing on other types of ear-pieces in ITU-T Study Group 12

The nominal values of SLR/RLR to/from the POI shall be:

SLR = 8 +/- 3 dB;

RLR = 2 +/- 3 dB

RLR (binaural headset) = 8 +/- 3 dB for each earphone

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

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

6.3 Idle channel noise (handset and headset UE)

6.3.1 Sending

The maximum (acoustic) A-weighted noise level at the handset and headset UE when no signal is transmitted to the input of the SS shall be as follows:

NOTE 1: This level includes the eventual noise contribution of an acoustic echo canceller under the condition that no signal is received.

NOTE 2: This figure applies to the total noise level with A-weighting. It is recommended that the level of single frequency disturbances should be ≤ -74 dBm0(A)) in the frequency range from 100 Hz to 10 kHz.

Compliance shall be checked by the relevant test described in TS 26.132.

6.3.2 Receiving

The maximum (acoustic) noise level at the handset and headset UE when no signal is transmitted to the input of the SS shall be as follows:

> If no user-controlled receiving volume control is provided, or, if it is provided, at the setting of the user-controlled receiving volume control at which the RLR is equal to the nominal value, the noise measured at DRP with diffuse field correction contributed by the receiving equipment alone shall not exceed -57 dBPa(A).
>
> Where a volume control is provided, the measured noise shall also not exceed -54 dBPa(A) at the maximum setting of the volume control.

For the nominal volume control setting, the level of single frequency disturbances shall be ≤ -60 dBPa(A) in the frequency range from 100 Hz to 10 kHz. As a performance objective it is recommended that the level should be ≤ -64dBPa(A).

NOTE: In a connection with the PSTN, noise conditions as described in ITU-T Recommendation G.103 can be expected at the input (POI) of the 3G network. The characteristics of this noise may be influenced by the speech transcoding process (for further study).

Compliance shall be checked by the relevant test described in TS 26.132.

6.4 Sensitivity/frequency characteristics

In general it is recommended for all configurations to have a flat sending frequency response.

6.4.1 Handset and headset UE sending

The sensitivity/frequency characteristics shall be as follows:

The sending sensitivity frequency response, measured either from the mouth reference point (MRP) to digital interface or from the MRP to the SS audio output (digital output of the reference speech decoder of the SS), shall be within a mask, which can be drawn between the points given in table 9. The mask is drawn with straight lines between the breaking points in table 1 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 9: Sending sensitivity/frequency mask

Handset-Headset send sensitivity/frequency response Frequency (Hz)	Upper limit	Lower limit
100	0	
200	5	-5
5 000	5	-5
6 300	5	-10
8 000	5	
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

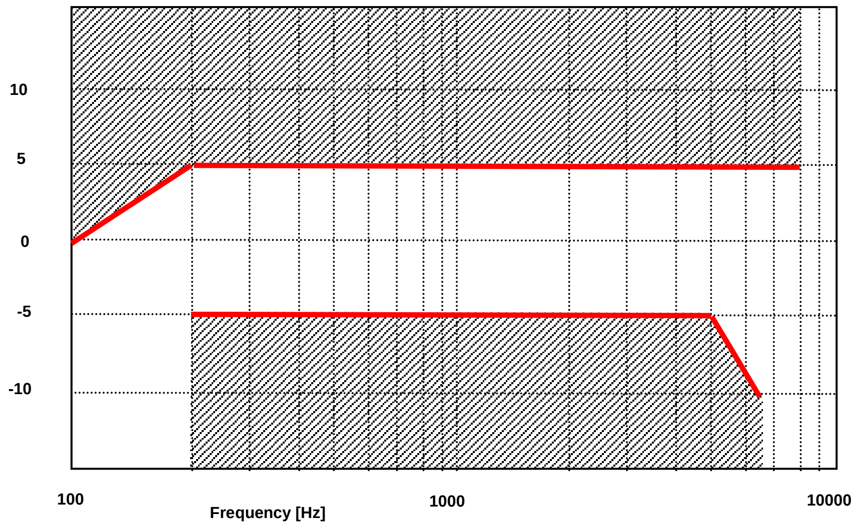


Figure 9: Sending sensitivity/frequency mask

Compliance shall be checked by the relevant test described in TS 26.132.

6.4.2 Handset and headset UE receiving

The sensitivity/frequency characteristics shall be as follows:

The receiving sensitivity frequency response, measured either from the digital interface to the DRP with diffusefield correction or from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the DRP with diffusefield correction, shall be within a mask, which can be drawn with straight lines between the breaking points in table 10 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Frequency (Hz)	Upper limit 8 +/-2 N	Lower limit 8 +/-2 N
100	6	
200	6	-10
300	6	-6
1 000	6	-6
2 000	8	-6
5 000	8	-6
6 300	8	-12
8 000	8	
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.		

NOTE: The limits in the table above are enforced but are under evaluation. The values are expected to be modified taking into account that the change from ERP to DRP with diffuse field correction is reflected in the table.

Compliance shall be checked by the relevant test described in TS 26.132.

6.4.3 Desktop and Vehicle-mounted hands-free UE sending

The sending sensitivity frequency response from the MRP to the SS audio output (digital output of the reference speech decoder of the SS) shall be as follows:

The sending sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 11 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 11: Desktop and Vehicle-mounted hands-free sending sensitivity/frequency mask

Frequency	Upper limit	Lower limit
100	0	
200	5	-5
5 000	5	-5
6 300	5	-10
8 000	5	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale. All sensitivity values are expressed in dB on an arbitrary scale.		

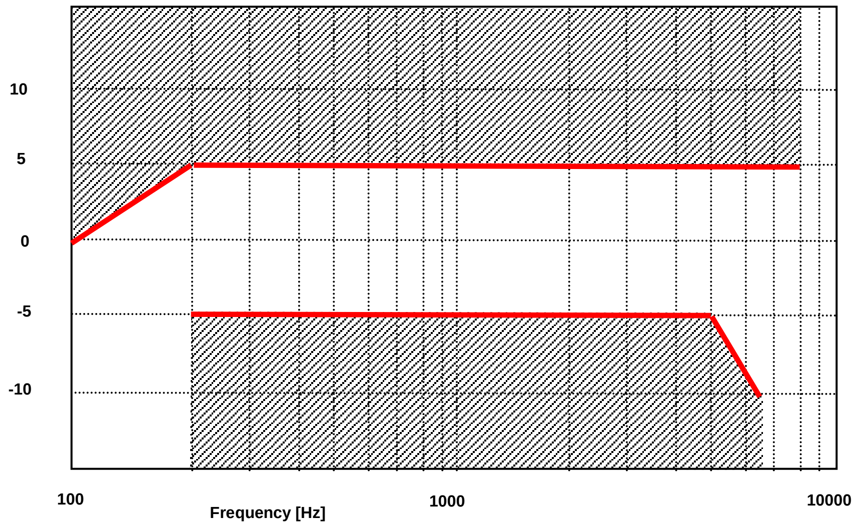


Figure 11: Desktop and Vehicle-mounted hands-free sending sensitivity/frequency mask

Compliance shall be checked by the relevant test described in TS 26.132.

6.4.4 Desktop and Vehicle-mounted hands-free UE receiving

The receiving sensitivity frequency response from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the freefield shall be as follows:

The receiving sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 12 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 12: Hands-free receiving sensitivity/frequency mask

Frequency	Upper limit	Lower limit
125 Hz	8	
200 Hz	8	-12
250 Hz	8	-9
315 Hz	7	-6
400 Hz	6	-6
5 000 Hz	6	-6
6 300 Hz	6	-9
8 000 Hz	6	-∞
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale. All sensitivity values are expressed in dB on an arbitrary scale.		

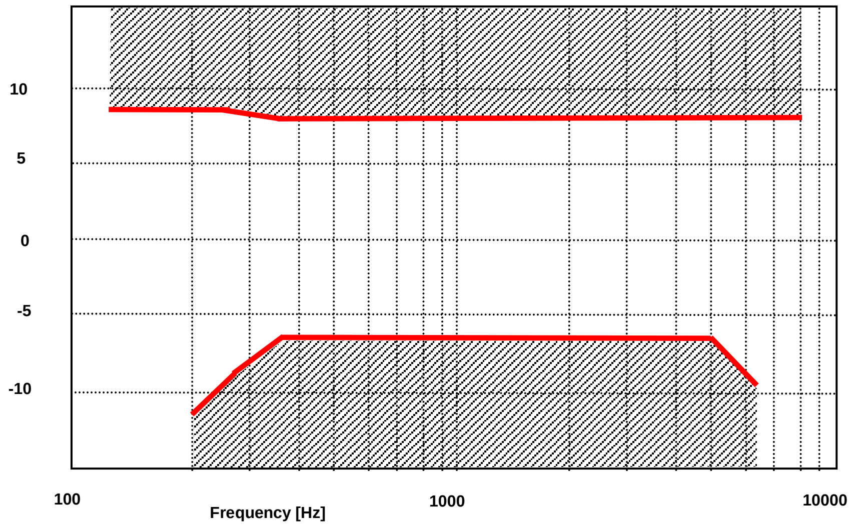


Figure 12: Hands-free receiving sensitivity/frequency mask

It is recommended that the receiving sensitivity frequency response be within the mask which can be drawn with straight lines between the breaking points in table 12.a on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 12.a: Performance objective for hands-free receiving sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
100	0	
200	0	-18
250	0	-15
315	0	-12
400	0	-12
500	0	-12
630	0	-12
800	0	-12
1 000	0	-12
1 300	0	-12
1 600	0	-12
2 000	0	-12
2 500	0	-12
6 300	0	-12
8 000	0	

Compliance shall be checked by the relevant test described in TS 26.132.

6.4.5 Handheld hands-free UE sending

The sending sensitivity frequency response from the MRP to the SS audio output (digital output of the reference speech decoder of the SS) shall be as follows:

The sending sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 13 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 13: Table 13: Handheld hands-free sending sensitivity/frequency mask

Frequency	**Upper limit**	**Lower limit**
Frequency	Upper limit	Lower limit
100	0	
200	5	-5
5 000	5	-5
6 300	5	-10
8 000	5	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale. All sensitivity values are expressed in dB on an arbitrary scale.		

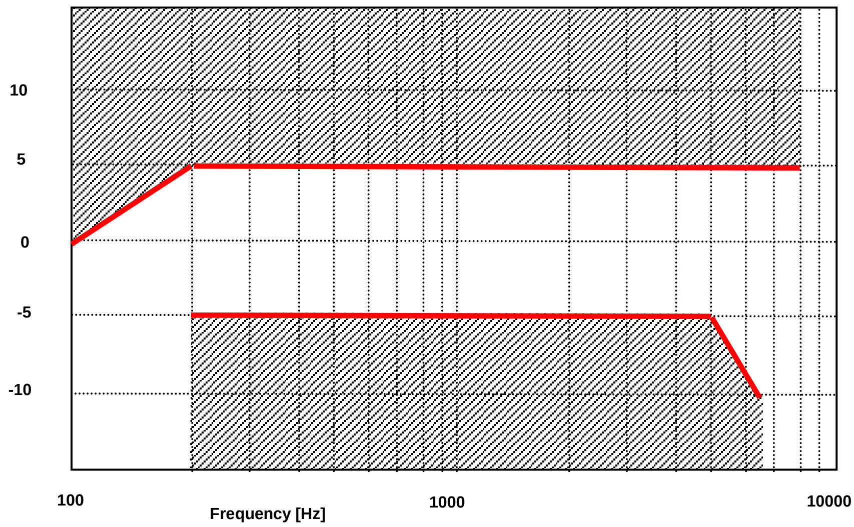


Figure 13: Handheld hands-free sending sensitivity/frequency mask

Compliance shall be checked by the relevant test described in TS 26.132.

6.4.6 Handheld hands-free UE receiving

The receiving sensitivity frequency response from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the freefield shall be as follows:

The receiving sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 14 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 14: Hands-free receiving sensitivity/frequency mask

Frequency	Upper limit	Lower limit
315 Hz	6	
630 Hz	6	-12
800 Hz	6	-6
4 000 Hz	6	-6
6 300 Hz	6	-12
8 000 Hz	6	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale. All sensitivity values are expressed in dB on an arbitrary scale.		

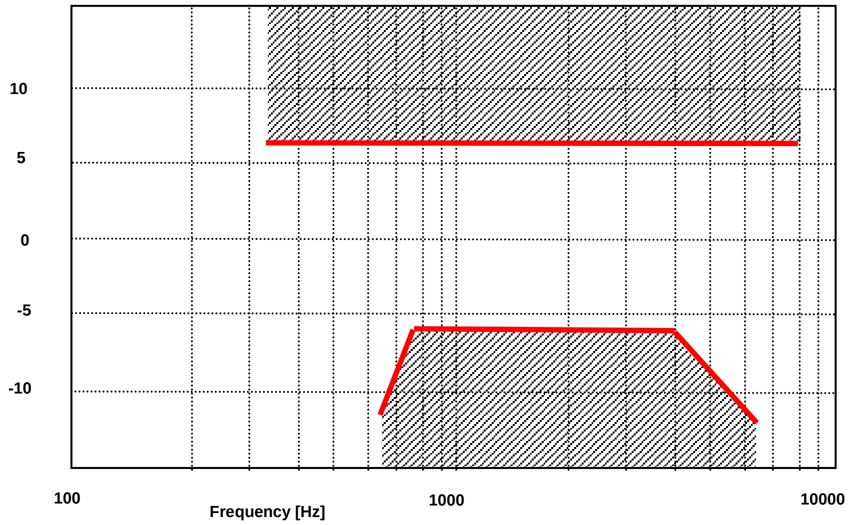


Figure 14: Hands-free receiving sensitivity/frequency mask

It is recommended that the receiving sensitivity frequency response be within the mask which can be drawn with straight lines between the breaking points in table 14.a on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 14.a: Performance objective for hands-free receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
315 Hz	6	
400 Hz	6	-12
500 Hz	6	-6
630 Hz	6	-6
800 Hz	6	-6
4 000 Hz	6	-6
6 300 Hz	6	-12
7 000 Hz	6	
8 000 Hz	6	
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale. All sensitivity values are expressed in dB on an arbitrary scale.		

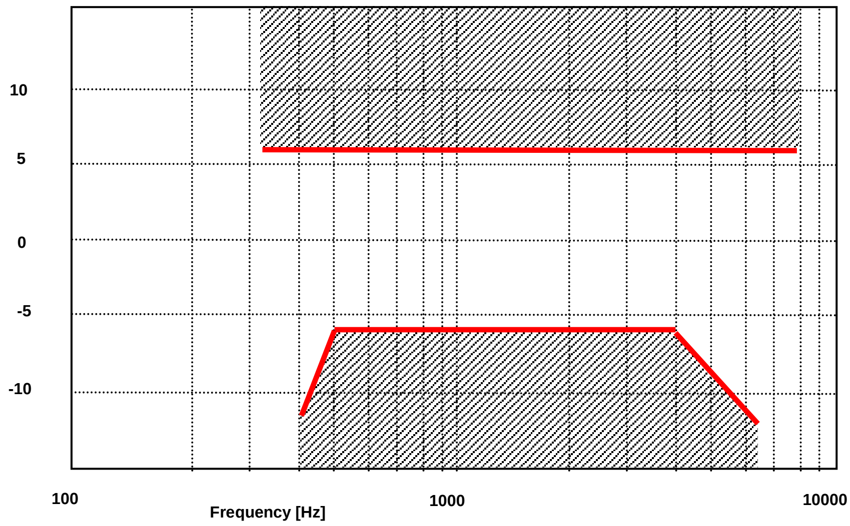


Figure 14.a: Performance objective for hands-free receiving sensitivity/frequency mask

Compliance shall be checked by the relevant test described in TS 26.132.

6.5 Sidetone characteristics (handset and headset UE)

6.5.1 Sidetone loss

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

Compliance shall be checked by the relevant test described in TS 26.132.

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

NOTE 2: It is in general recommended to provide a terminal sidetone path for handset and headset UEs.

NOTE 3: In case the human air conducted sidetone paths are obstructed (one example being some binaural insert type headset UEs), it is important to provide a terminal sidetone path.

6.5.2 Sidetone delay

It is recommended that the maximum sidetone delay be less than 5 ms, measured in an echo-free setup.

Compliance shall be checked by the relevant test described in TS 26.132.

6.6 Stability loss

The stability loss presented to the PSTN by the 3G network at the POI should meet the principles of the requirements in clauses 2 and 3 of ITU-T Recommendation G.122. These requirements will be met if the attenuation between the digital input and digital output at the POI is at least 6 dB at all frequencies in the range 100 Hz to 8 kHz under the worst case acoustic conditions at the UE (any acoustic echo control should be enabled). For the normal case of digital connection between the Air Interface and the POI, the stability requirement can be applied at the Air Interface.

The worst case acoustic conditions will be as follows (with any volume control set to maximum):

Handset UE: the handset lying on, and the transducers facing, a hard surface with the ear-piece uncapped.

Headset UE: for further study

Handsfree UE: no requirement other than echo loss.

NOTE: The test procedure must take into account the switching effects of echo control and discontinuous transmission (DTX).

6.7 Acoustic echo control

6.7.1 General

The echo loss (EL) presented by the 3G network at the POI should be sufficient during single talk. This takes into account the fact that UE is likely to be used connections with high transmission delay and in a wide range of noise environments.

The use of acoustic echo control is not mandated for 3G networks and the connection between the UE and the POI is zero loss. Therefore the acoustic echo control provided in UE should provide a sufficient TCLw at the POI over the likely range of acoustic end delays.

If acoustic echo control is provided by voice switching, comfort noise should be injected. This comfort noise shall operate in the same way to that used in DTX.

6.7.2 Acoustic echo control in a Desktop and Vehicle-mounted hands-free UE

The TCLw for the desktop and vehicle-mounted hands-free UE shall be ≥ 40 dB. for any setting of the volume control..

The TCLw for the desktop hands-free and vehicle-mounted hands-free UE shall be ≥ 46 dB when measured under freefield conditions at nominal setting of volume control.

NOTE: A TCLw for the desktop hands-free and vehicle-mounted hands-free UE of ≥ 55 dB is a performance objective when measured under freefield conditions at nominal setting of volume control.

The echo canceller should be designed to cope with the expected reverberation and dispersion. In the case of the hands-free UE, this reverberation and dispersion may be time variant. Compliance with this requirement shall be checked by the relevant test described in TS 26.132.

6.7.3 Acoustic echo control in a handheld hands-free UE

The TCLw for handheld hands-free UE shall be ≥ 40 dB for any setting of the volume control.

The TCLw for handheld hands-free UE shall be ≥ 46 dB at nominal setting of volume control.

NOTE: A TCLw for the handheld hands-free UE of ≥ 55 dB is a performance objective when measured under freefield conditions at nominal setting of volume control.

The echo canceller should be designed to cope with the expected reverberation and dispersion. In the case of the hands-free UE, this reverberation and dispersion may be time variant. Compliance with this requirement shall be checked by the relevant test described in TS 26.132.

6.7.4 Acoustic echo control in a handset UE

The TCLw for handset UE shall be ≥ 46 dB for any setting of the volume control.

The TCLw for handset UE shall be ≥ 55 dB at nominal setting of volume control.

With the volume control set to maximum TCLw shall be ≥55 dB. The volume control shall be set back to nominal after each call unless TCLw ≥ 55 dB can be maintained also with maximum volume setting.

The echo canceller should be capable of dealing with the variations in handset positions when in normal use. The implications of this are under study. Compliance with this requirement shall be checked by the relevant test described in TS 26.132.

6.7.5 Acoustic echo control in a headset UE

The TCLw for headset UE shall be ≥ 46 dB for any setting of the volume control.

The TCLw for headset UE shall be ≥ 55 dB at nominal setting of volume control.

The volume control shall be set back to nominal after each call unless TCLw ≥ 55 dB can be maintained also with maximum volume setting.

Due to the obstacle effect of the head in this type of terminal, careful design might mean that no active echo control is necessary.

The echo cancellation algorithm should be designed to cope with the expected reverberation and dispersion. Compliance with this requirement shall be checked by the relevant test described in TS 26.132.

6.8 Distortion

6.8.1 Sending Distortion

The sending part shall meet the following distortion requirements:

NOTE 1: Digital signal processing other than the transcoder itself is included in this requirement (e.g. echo cancelling).

Distortion shall be measured between MRP and the SS audio output (output of the reference speech decoder of the SS). The ratio of signal-to-total distortion power measured with the proper noise weighting (see table 4 of ITU-T Recommendation G.223) shall be above the limits given in table 15.

NOTE 2: Frequencies from 315 Hz to 816 Hz do not apply to the hands-free UE case, but only to handset and headset UE.

Table 15: Limits for signal-to-total distortion ratio

Frequency	Sending level (dBPa at the MRP)	Sending Ratio (dB)
315 Hz	-4,7	28
408 Hz	-4,7	32
510 Hz	-4,7	32
816 Hz	-4,7	32
1020 Hz	+5 0 -4,7 -10 -15 -20	30 35 35 33 30 27

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) - linear (dB ratio) scale.

Compliance of the sending distortion shall be checked by the test described in TS 26.132.

NOTE 3: It should be ensured that the test signal is treated by speech processing algorithms as a speech-like signal, and not a noise-like signal. Test signals with a time-stationary envelope may be treated by certain algorithms, e.g. noise suppression algorithms defined in 3GPP TS 06.77 R99[16], as a noise-like signal. If speech processing algorithms, including but not limited to noise suppression algorithms, are shown to treat the test signal as a noise-like signal, even where an activation signal has been utilized, then the test should be repeated with said speech processing algorithms disabled. The results of both sets of tests and the state of the processing algorithms should be documented in the test report.

6.8.2 Receiving

The receiving part between the SS audio input (input of the reference speech encoder of the SS) and ERP shall meet the requirements in this clause at the nominal setting of the volume control (except when another volume is specified):

The ratio of signal-to-total distortion power measured with the proper noise weighting (see table 4 of ITU-T Recommendation G.223) shall be above the limits given in table 16 when the sound pressure at ERP is up to +10 dBPa. For sound pressures exceeding +10 dBPa at the ERP there is no distortion requirement.

NOTE 1: Frequencies from 315 Hz to 816 Hz do not apply to the hands-free UE case, but only to handset and headset UE.

Table 16: Limits for signal-to-total distortion ratio

<table>
<tr><th>Frequency</th><th>Receiving level at the digital interface (dBm0)</th><th>Receiving ratio at nominal volume setting (dB)</th><th>Receiving ratio at maximum volume setting (dB)</th></tr>
<tr><td>315 Hz</td><td>-16</td><td>tbd</td><td></td></tr>
<tr><td>408 Hz</td><td>-16</td><td>tbd</td><td></td></tr>
<tr><td>510 Hz</td><td>-16</td><td>tbd</td><td></td></tr>
<tr><td>816 Hz</td><td>-16</td><td>tbd</td><td></td></tr>
<tr><td>1020 Hz</td><td>0
-3
-10
-16
-20
-30
-40
-45</td><td>25,5
31,5
33,5
tbd
tbd
tbd
tbd
tbd</td><td>tbd</td></tr>
</table>

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) - linear (dB ratio) scale.

Compliance of the receiving distortion shall be checked by the appropriate method in TS 26.132.

NOTE 2: It should be ensured that the test signal is treated by speech processing algorithms as a speech-like signal, and not a noise-like signal. Test signals with a time-stationary envelope may be treated by certain algorithms, e.g. noise suppression algorithms defined in 3GPP TS 06.77 R99[16], as a noise-like signal.

NOTE 3: The informative values for limits for signal to total distortion ratio shown in Table 16a have been under consideration but are not in force.

Table 16a (INFORMATIVE): Informative values for limits for signal-to-total distortion ratio

<table>
<tr><th>Frequency</th><th>Receiving level at the digital interface (dBm0)</th><th>Receiving ratio at nominal volume setting (dB)</th><th>Receiving ratio at maximum volume setting (dB)</th></tr>
<tr><td>315 Hz</td><td>-16</td><td>20</td><td></td></tr>
<tr><td>408 Hz</td><td>-16</td><td>28</td><td></td></tr>
<tr><td>510 Hz</td><td>-16</td><td>28</td><td></td></tr>
<tr><td>816 Hz</td><td>-16</td><td>28</td><td></td></tr>
<tr><td>1020 Hz</td><td>-16
-20
-30
-40 (*)
-45 (*)</td><td>33,5
33,0
30,5
22,5 (*)
17,5 (*)</td><td></td></tr>
<tr><td colspan="4">NOTE: (*)For levels -40 and -45 dBm0 a lower signal-to-distortion ratio may not be possible, and hence would not be regarded as a failing result. However, the obtained results would be reported.</td></tr>
</table>

6.9 Ambient Noise rejection

Handset and Headset UE:

The nature of mobile telephony is such that the UE will typically be operated in high ambient acoustic noise. Due to the adverse interaction of noise signals with speech codecs operating at lower rates, for example 8kbit/s or less, a minimum noise rejection specification is required.

The UE ambient noise rejection ANR, calculated as a Single Figure DELSM (SFDELSM) shall be greater than or equal to 0 dB. For good performance, it is recommended that a figure of +3 dB should be achieved. Due to the uncertainty inherent in the measurement method for ANR, a 3dB tolerance is allowed on the nominal value.

Compliance shall be checked by the relevant test described in TS 26.132.

Hands-free UE (all categories):

For further study.

Annex A (informative): Change history

3.0.0	December 1999	Approved at TSG-SA#6 Plenary

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New
2000-06	8	SP-000264	001	2	CR on Addition of a chapter pointing to ITU-T Recommendations for extended parameters	3.0.0	3.1.0
2000-06	8	SP-000264	002		CR on Listener side tone (LSTR) and talker side tone (STMR) requirements	3.0.0	3.1.0
2000-06	8	SP-000264	003	1	CR on Change of Handset and headset UE receiving sensitivity/frequency characteristic mask	3.0.0	3.1.0
2000-06	8	SP-000264	004	1	CR on Acoustic requirements for Handheld-type hands-free user equipment	3.0.0	3.1.0
2001-03	11	SP-010106	005	1	Harmonisation of narrow-band acoustic requirements between 3GPP and GSM	3.1.0	3.2.0
2001-03	11				Release 4		4.0.0
2001-03	11	SP-010106	006	3	Wideband acoustic requirements	4.0.0	5.0.0
2001-09	13	SP-010453	009		Introduction of ANR tolerance of 3 dB	5.0.0	5.1.0
2002-09	17	SP-020435	014		Correction on the ANR requirement for hands-free Ues	5.1.0	5.2.0
2004-09	25	SP-040649	022		Change of sending distortion requirement	5.2.0	6.0.0
2007-03	35	SP-070026	0023	1	Minimum echo loss requirements	6.0.0	6.1.0
2007-03	35	SP-070026	0024	1	Correcting wrong reference to ITU-T G.223	6.0.0	6.1.0
2007-03	35	SP-070026	0025	1	Update of reference [11] to P.79-2001 Annex G	6.0.0	6.1.0
2007-03	35	SP-070026	0027	1	Sending distortion requirements for WB-AMR	6.0.0	6.1.0
2007-06	36				Version for Release 7	6.1.0	7.0.0
2007-12	38	SP-070759	0028	2	Creating a sidetone requirement for the case where HATS method is used	7.0.0	7.1.0
2008-12	42	SP-080682	0030	1	Receiving characteristics harmonization	7.1.0	8.0.0
2008-12	42	SP-080682	0031	1	Updated requirements and performance objectives for wideband terminal acoustics	7.1.0	8.0.0
2009-03	43	SP-090017	0029	2	Terminal acoustic characteristics for telephony	8.0.0	9.0.0
2009-06	44	SP-090257	0033		Receiving sensitivity/frequency mask correction	9.0.0	9.1.0
2009-09	45	SP-090568	0035	1	Correction of STMR calculation	9.1.0	9.2.0
2010-03	47	SP-100021	0036	1	Correction of distortion measurements	9.2.0	9.3.0
2010-09	49	SP-100470	0039	4	Enhancement of STMR requirements	9.3.0	10.0.0
2011-03	51	SP-110042	0041	3	Alignment of 3GPP Audio Test Requirements	10.0.0	10.1.0
2011-03	51	SP-110149	0044	3	Correction of WB receive distortion requirements	10.0.0	10.1.0

History

Document history		
V10.1.0	April 2011	Publication