ETSI TS 126 131 V9.5.0 (2014-01)

Technical Specification

Universal Mobile Telecommunications System (UMTS);
LTE;
Terminal acoustic characteristics for telephony;
Requirements
(3GPP TS 26.131 version 9.5.0 Release 9)

A GLOBAL INITIATIVE

Reference

RTS/TSGS-0426131v950

Keywords

LTE,UMTS

ETSI

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Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

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Foreword

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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
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 less than (louder than) -13 dB.

With the volume control set to the minimum position the RLR shall not be greater than (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.

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

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 noise volume in a moving vehicle.

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 = 6 +12 / - 4 dB.

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

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

5.2.5 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 with any volume control set to mid position.

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 less than (louder than) -13 dB.

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

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 not exceed -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 wideband noise signal. It is recommended that the level of single frequency disturbances should be 10 dB lower (ITU-T Recommendation P.11).

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

5.3.2 Receiving

The maximum (acoustic) 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 ear reference point (ERP) contributed by the receiving equipment alone shall not exceed -57 dB Pa(A).

Where a volume control is provided, the measured noise shall also not exceed -54 dB Pa(A) at the maximum setting of the volume control.

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.

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 ERP or from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the ERP, 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. The values in table 2 are provisional and are for further study.

Table 2: Receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	6	-
200	6	-
300	6	-6
1000	6	-6
2000	8	-6
3400	8	-6
4 000	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.

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: Hands-free sending sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
200	0	
250	0	
315	0	-14
400	0	-13
500	0	-12
630	0	-11
800	0	-10
1 000	0	-8
1 300	2	-8
1 600	3	-8
2 000	4	-8
2 500	4	-8
3 100	4	-8
4 000	0	

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

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 ERP 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: Hands-free receiving sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
200	0	
250	0	
315	0	-15
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
3 100	0	-12
4 000	0	

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

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: Hands-free sending sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
200	0	
250	0	
315	0	-14
400	0	-13
500	0	-12
630	0	-11
800	0	-10
1 000	0	-8
1 300	2	-8
1 600	3	-8
2 000	4	-8
2 500	4	-8
3 100	4	-8
4 000	0	

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

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 ERP 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 response

Frequency (Hz)	Upper limit	Lower limit
200	0	
250	0	
315	0	
400	0	
500	0	
630	0	
800	0	-12
1 000	0	-12
1 300	0	-12
1 600	0	-12
2 000	0	-12
2 500	0	-12
3 100	0	-12
4 000	0	

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

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

If the HATS method is used as described in 3GPP TS 26.132, then the talker sidetone masking rating (STMR) shall be 18 dB ± 5 dB for nominal setting of the volume control. For all other positions of the volume control, the STMR must not be below 8 dB.

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

Compliance shall be checked by the relevant test described in 3GPP 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: In general, it is 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.

NOTE 4: The STMR calculation algorithm being used was developed for quantifying the audibility of the electrical sidetone path using a sealed coupler. The air-conducted path was not intended to be included in the test setup. A lower STMR limit was specified to avoid annoying effects (e.g. howling, increase of ambient noise level in the ear) of an excessive electrical sidetone. In HATS-based measurements, the air-conducted path cannot be avoided in the test setup. With some UE form factors the air-conducted path can be substantial resulting in low STMR figures also when there are no annoying effects from any excessive electrical sidetone. See ITU-T Recommendation P.76 for definitions of sidetone paths.

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 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.

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 TCLw of at least 46 dB 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 an Desktop and Vehicle-mounted hands-free UE

The TCLw for the handsfree UE shall be at least 40 dB at the nominal setting of the volume control in quiet background conditions and at least 33 dB at the maximum user selectable volume control setting. If acoustic echo control is provided using some form of echo cancellation technique, the cancellation algorithm 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 an handheld hands-free UE

The TCLw for the hands-free UE shall be at least 40 dB at the nominal setting of the volume control in quiet background conditions and at least 33 dB at the maximum user selectable volume control setting. If acoustic echo control is provided using some form of echo cancellation technique, the cancellation algorithm 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 the handset UE shall be at least 46 dB. Careful acoustic design of the handset body and selection of the mouth and ear piece transducers may facilitate the required acoustic echo loss without the need for active echo control techniques. However, should echo cancellation be employed 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 a headset UE shall be 46 dB. Due to the obstacle effect of the head in this type of terminal, careful design might mean that no active echo control is necessary.

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 ERP 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 ERP is up to +10 dBPa. For sound pressures exceeding +10 dBPa at the ERP 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.

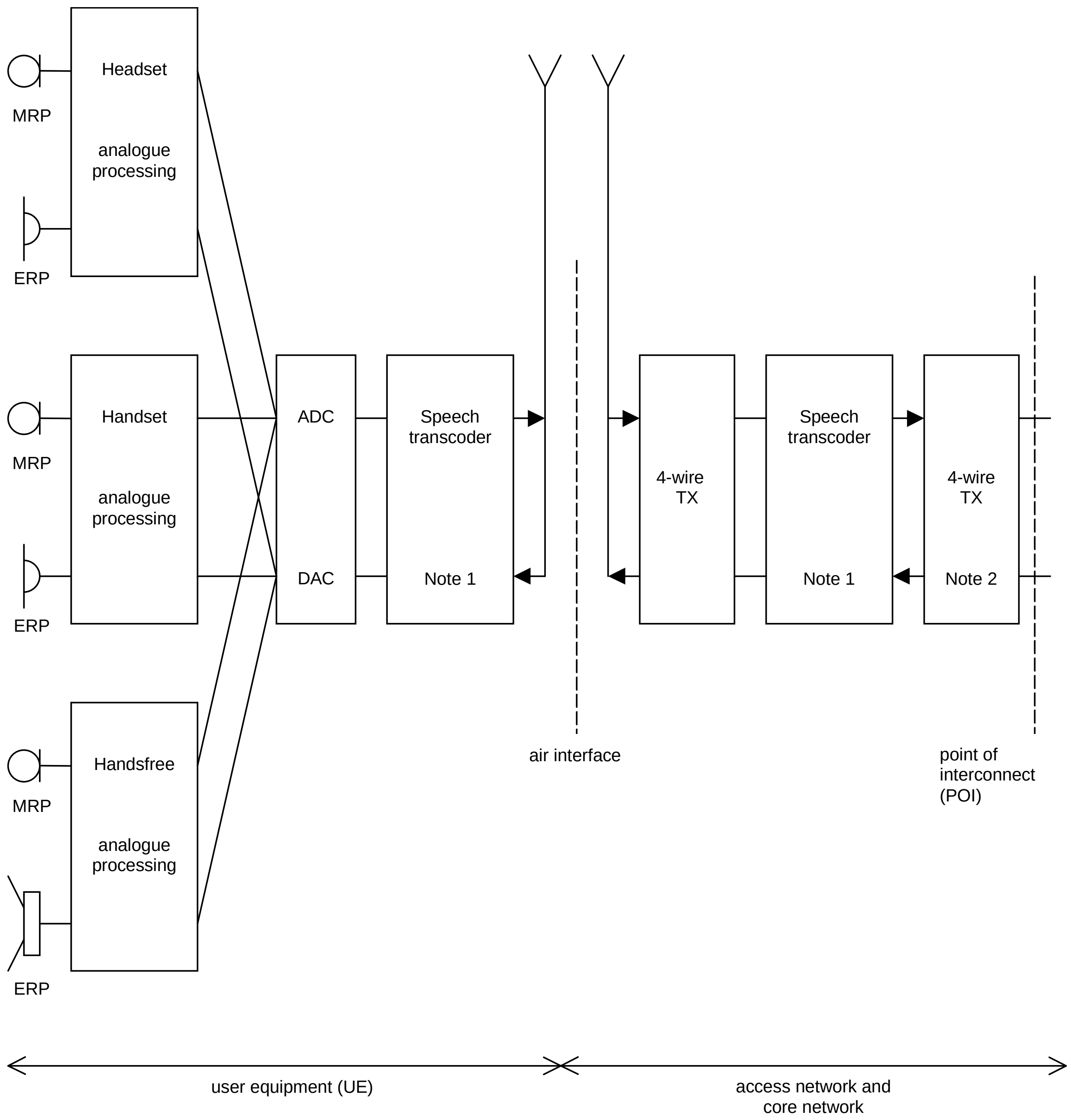
The UE ambient noise rejection ANR, calculated as a Single Figure DELSM (SFDELSM) shall be greater than or equal to the nominal value of 0dB. Due to the uncertainty inherent in the measurement method for ANR, a 3dB tolerance is allowed on the nominal value.

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.

Requirements for wideband telephony are based on ITU-T Recommendations P. 311, for handset user-equipment, and ITU-T Recommendation P. 341 for hands-free user-equipment.

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 less than (louder than) -13 dB.

With the volume control set to the minimum position the RLR shall not be greater than (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.

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 noise volume in a moving vehicle.

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

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 = 15 +/- 3 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 less than (louder) or equal to 12 dB. As performance objective it is recommended that the RLR at maximum volume control is less than (louder) or equal to 6 dB.

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 less than (louder than) -13 dB. With the volume control set to the minimum position the RLR shall not be greater than (quieter than) 24 dB.

6.3 Idle channel noise (handset and headset UE)

6.3.1 Sending

The maximum noise level produced by the apparatus at the output of the SS under silent conditions in the sending direction shall not exceed -64 dBm0(A).

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 wideband noise signal. It is recommended that the level of single frequency disturbances should be 10 dB lower (ITU-T Recommendation P.11).

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

6.3.2 Receiving

6.3.2.1 Receiving: total noise level

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 the ear reference point (ERP) 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.

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.

6.3.2.2 Receiving: noise level in 1/3-octave band

For the nominal volume control setting, the level in any 1/3rd -octave band, between 100 Hz and 10 kHz shall not exceed a value of -60 dBPa(A). As a performance objective it is recommended that the level in any 1/3rd -octave band, between 100 Hz and 10 kHz does not exceed a value of -64 dBPa(A).

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

Frequency (Hz)	Upper limit	Lower limit
100	-6	
200	2	
300	2	-12
1 000	2	-6
2 000	4	-6
5 000	4	-6
6 300	4	-9
8 000	0	

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

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 ERP or from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the ERP, 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.

Table 10: Receiving sensitivity/frequency mask

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.

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 response

Frequency (Hz)	Upper limit	Lower limit
100	-6	
200	2	
300	2	-12
1 000	2	-6
2 000	4	-6
5000	4	-6
6300	4	-9
8000	0	

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

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 ERP 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 response

Frequency (Hz)	Upper limit	Lower limit
200	0	
315	0	-15
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	

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

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: Handheld hands-free sending sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
100	-6	
200	2	
300	2	-12
1 000	2	-6
2 000	4	-6
5 000	4	-6
6 300	4	-9
8 000	0	

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

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 ERP 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 response

Frequency (Hz)	Upper limit	Lower limit
315 Hz	0	
400 Hz	0	
500 Hz	0	
630 Hz	0	-18
800 Hz	0	-12
4 000 Hz	0	-12
5 000 Hz	0	-15
6 300 Hz	0	-18
7 000 Hz	0	
8 000 Hz	0	

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

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 response

Frequency (Hz)	Upper limit	Lower limit
315 Hz	0	
400 Hz	0	-18
500 Hz	0	-12
630 Hz	0	-12
800 Hz	0	-12
4 000 Hz	0	-12
5 000 Hz	0	-15
6 300 Hz	0	-18
7 000 Hz	0	
8 000 Hz	0	

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 HATS method is used as described in 3GPP TS 26.132. The talker sidetone masking rating (STMR) shall be 18 dB ± 5 dB for nominal setting of the volume control. For all other positions of the volume control, the STMR must not be below 8 dB.

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

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: In general, it is 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.

NOTE 4: The STMR calculation algorithm being used was developed for quantifying the audibility of the electrical sidetone path using a sealed coupler. The air-conducted path was not intended to be included in the test setup. A lower STMR limit was specified to avoid annoying effects (e.g. howling, increase of ambient noise level in the ear) of an excessive electrical sidetone. In HATS-based measurements, the air-conducted path cannot be avoided in the test setup. With some UE form factors the air-conducted path can be substantial resulting in low STMR figures also when there are no annoying effects from any excessive electrical sidetone. See ITU-T Recommendation P.76 for definitions of sidetone paths.

6.5.2 Sidetone delay

It is recommended that the maximum sidetone delay be less than 10 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 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.

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 TCLw of at least 46 dB 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 an Desktop and Vehicle-mounted hands-free UE

The TCLw for the handsfree UE shall be at least 40 dB at the nominal setting of the volume control in quiet background conditions and at least 33 dB at the maximum user selectable volume control setting. If acoustic echo control is provided using some form of echo cancellation technique, the cancellation algorithm 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 an handheld hands-free UE

The TCLw for the hands-free UE shall be at least 40 dB at the nominal setting of the volume control in quiet background conditions and at least 33 dB at the maximum user selectable volume control setting. If acoustic echo control is provided using some form of echo cancellation technique, the cancellation algorithm 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 the handset UE shall be at least 46 dB for all settings of volume control. Careful acoustic design of the handset body and selection of the mouth and ear piece transducers may facilitate the required acoustic echo loss without the need for active echo control techniques. However, should echo cancellation be employed 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 a headset UE shall be at least 46 dB for all settings of volume control. Due to the obstacle effect of the head in this type of terminal, careful design might mean that no active echo control is necessary.

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.

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

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

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

Frequency	Receiving level at the digital interface (dBm0)	Receiving ratio at nominal volume setting (dB)	Receiving ratio at maximum volume setting (dB)
315 Hz	-16	20	
408 Hz	-16	28	
510 Hz	-16	28	
816 Hz	-16	28	
1020 Hz	-16 -20 -30 -40 (*) -45 (*)	33,5 33,0 30,5 22,5 (*) 17,5 (*)	
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.			

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
2011-03	51	SP-110149	0043	3	Correction of WB receive distortion requirements	9.3.0	9.4.0
2013-12	62	SP-130563	0058	2	STMR - adaptation to modern form factors	9.4.0	9.5.0

History

Document history		
V9.2.0	January 2010	Publication
V9.3.0	April 2010	Publication
V9.4.0	April 2011	Publication
V9.5.0	January 2014	Publication