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Contents

Foreword

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

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Introduction

The present document specifies minimum performance requirements for the acoustic characteristics of 3G and LTE terminals when used to provide narrowband or wideband telephony.

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

The performance requirements are specified in 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 narrowband 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 and LTE terminals when used to provide narrowband 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: "Speech and video telephony terminal acoustic test specification".

[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 and talker echo in international connections".

[7] Void

[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 Recommendation P.340: "Transmission characteristics and speech quality parameters of hands-free terminals".

[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".

[17] 3GPP TS 26.114: "IP Multimedia Subsystem (IMS); Multimedia Telephony; Media handling and interaction ".

[18] 3GPP TS 23.203: "Policy and charging control architecture".

3 Definitions, symbols and abbreviations

3.1 Definitions

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

For the purposes of the present document, the terms dB, dBr, dBm0, dBm0p and dBA, shall be interpreted as defined in ITU-T Recommendation B.12 [2]; the term dBPa shall be interpreted as the sound pressure level relative to 1 pascal expressed in dB (0 dBPa is equivalent to 94 dB SPL).

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

3.2 Abbreviations

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

ADC	Analogue to Digital Converter
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 TS 26.132. These are the air interface and the point of interconnect (POI).

The Air Interfaces for GSM, 3G and LTE are specified by GSM 05, 3GPP 45, 3GPP 25 and 3GPP 36 series specifications. MTSI speech aspects are specified by TS 26.114 [17].

Measurements can be made using the system simulator (SS) described in TS 26.132.

The POI with the public switched telephone network (PSTN) is considered to have a relative level of 0 dBr.

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 hands-free mode;
- Vehicle-mounted hands-free UE including softphone UE mounted in a vehicle;
- Hand-held hands-free UE including softphone UE with internal loudspeaker(s) used in hands-free 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 hands-free mode:
    - When using internal loudspeaker(s), requirements and performance objectives shall correspond to hand-held hands-free.
    - When using external loudspeaker(s), requirements and performance objectives shall correspond to desktop-mounted hands-free.

5 Narrowband telephony transmission performance

5.1 Applicability

The performance requirements in this sub-clause shall apply when UE is used to provide narrowband 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 or LTE network and the PSTN should meet the overall loudness rating (OLR) limits in ITU-T Recommendation G.111 [4]. The national parts of the connection should therefore meet the send and receive loudness rating (SLR, RLR) limits in ITU-T Recommendation G.121 [5].

For the case where digital routings are used to connect the 3G or LTE 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 or LTE network. The limits given below are consistent with the national extension limits and long term objectives in ITU-T Recommendation G.121 [5].

The SLR and RLR values for the 3G or LTE 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 or LTE 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 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.

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. For a vehicle-mounted hands-free UE:
   Where a user-controlled volume control is provided, the RLR shall meet the nominal value for at least 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 the maximum volume control setting should be ≤ (equal or louder than) -2 dB.

2. For a desktop hands-free UE:
   Where a user-controlled volume control is provided, the RLR shall meet the nominal value for at least 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 usage environment.
   RLR at the maximum volume control setting should be ≤ (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 [4], lies between 1 and 3 dB. The higher RLR requirement of 5 dB for desktop hands-free is appreciative of the limitations in transducer output with current typical form factors.

5.2.4 Connections with hand-held 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 a performance objective it is recommended that the RLR at the maximum volume control setting is ≤ (equal or louder than) 2 dB.

Where a user-controlled volume control is provided, the RLR shall meet the nominal value for at least one setting of the control. It is recommended that a volume control range ≥ 15 dB be provided. 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 [4], lies between 1 and 3 dB. The higher RLR requirement of 9 dB for hand-held 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 nominal value for at least one setting of the control. When the control is set to maximum, the RLR shall not be ≤ (equal or louder than) -13 dB.

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

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 dBPa(A).

Where a volume control is provided, the measured noise shall be ≤ -54 dBPa(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 [3] can be expected at the input (POI) of the 3G or LTE 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 the 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: Handset and headset 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.		

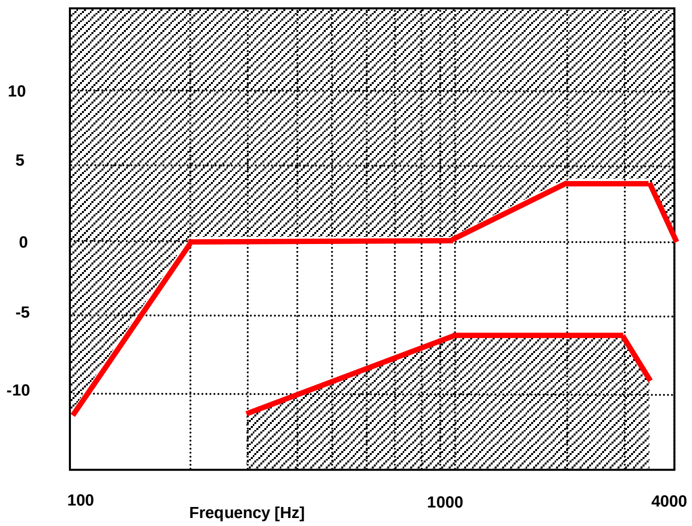

Figure 1: Handset and headset sending sensitivity/frequency mask

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: Handset and headset receiving sensitivity/frequency mask for 8N application force

Frequency (Hz)	Upper limit 8 ± 2 N	Lower limit 8 ± 2 N
100	6	
300	6	-6
3 400	6	-6
4 000	6	

NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale

NOTE 2: The basis for the target frequency responses in send and receive is the orthotelephonic reference response measured between 2 subjects 1 m apart under free-field conditions and assumes an ideal receive characteristic. Under these conditions the overall frequency response shows a rising slope. 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.

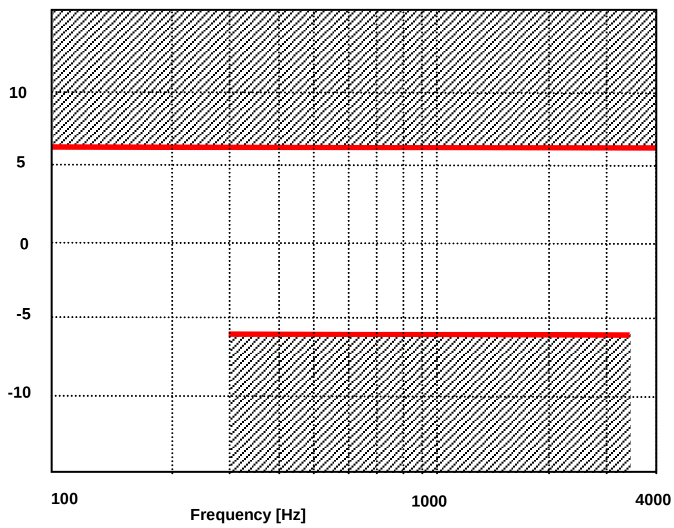

Figure 2: Handset and headset receiving sensitivity/frequency mask for 8N application force

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: Desktop and vehicle-mounted hands-free 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.		

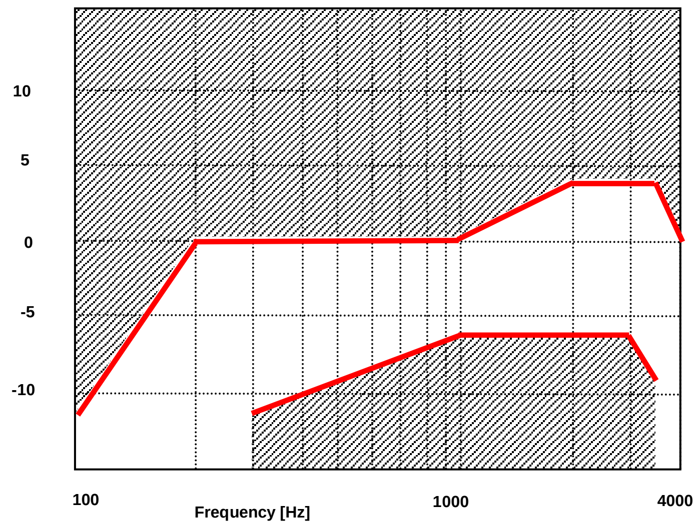

Figure 3: Desktop and vehicle-mounted hands-free sending sensitivity/frequency mask

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: Desktop and vehicle-mounted hands-free receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
200	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.		

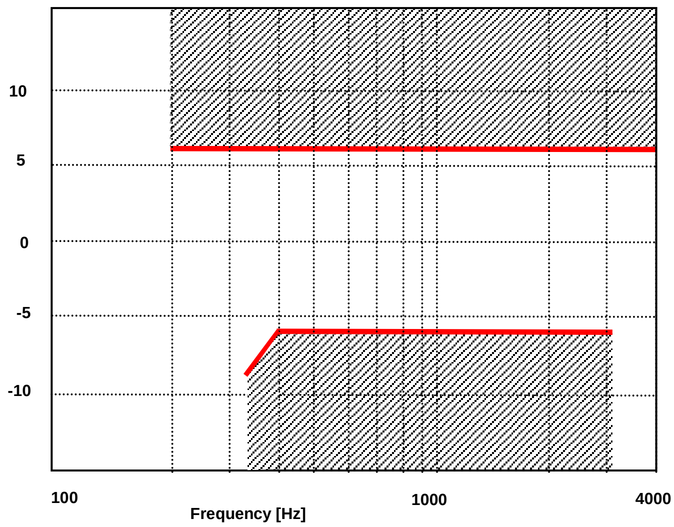

Figure 4: Desktop and vehicle-mounted receiving sensitivity/frequency mask

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

5.4.5 Hand-held 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: Hand-held hands-free 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.		

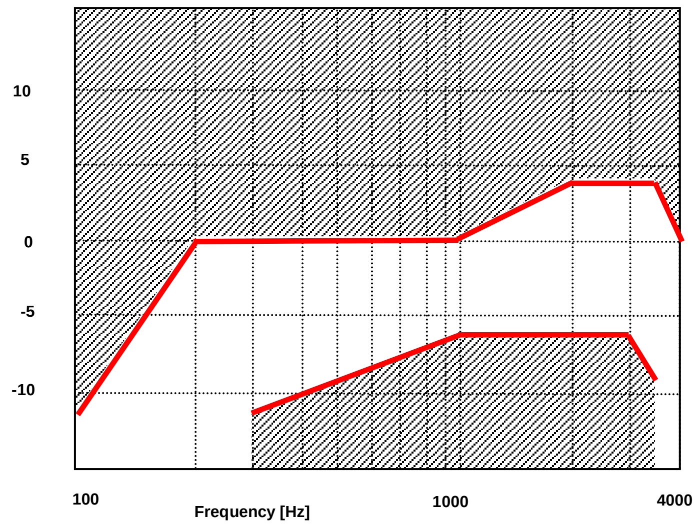

Figure 5: Hand-held hands-free sending sensitivity/frequency mask

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

5.4.6 Hand-held 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: Hand-held 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)		

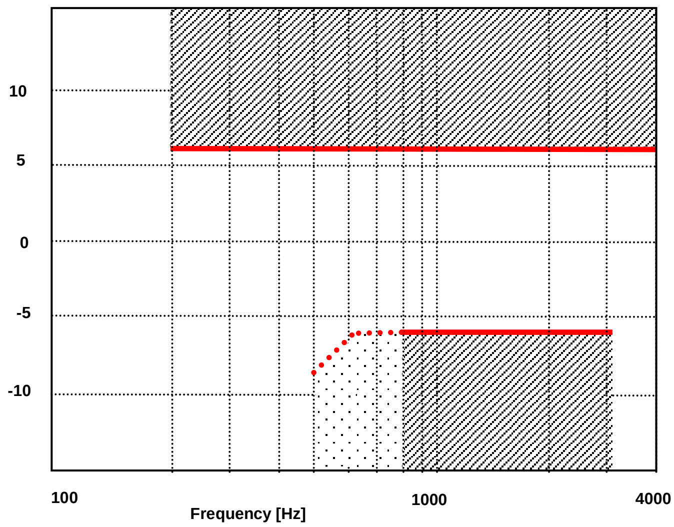

Figure 6: Hand-held hands-free receiving sensitivity/frequency mask

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 the nominal setting of the volume control. For all other positions of the volume control, the STMR shall be ≥ 10 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. The bandwidth for the sidetone path provided by the UE may in some terminals not be restricted to the narrowband 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: 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.

5.5.2 Sidetone delay

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

NOTE: The measured result is only applicable where the level of the electrical sidetone is sufficiently high to be measured. While the STMR value may indicate the presence of sidetone it should be ensured that this is not primarily due to the acoustical or mechanical sidetone path when interpreting sidetone delay results.

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

5.6 Stability loss

The stability loss presented to the PSTN by the 3G or LTE network at the POI should meet the principles of the requirements in clauses 2 and 3 of ITU-T Recommendation G.122 [6]. These requirements will be met if the attenuation between the digital input and digital output at the POI is ≥ 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 volume control set to maximum for each following condition):

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

Headset UE: for further study;

Hands-free 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 or LTE network at the POI should be sufficient during single-talk. This takes into account the fact that the UE is likely to be used in connections with high transmission delay and in 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 or LTE networks and the connection between the UE and the POI is zero loss. Therefore the acoustic echo control provided in the 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 as that used in DTX.

5.7.2 Acoustic echo control in 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 the nominal setting of the volume control.

NOTE: A TCLw for the desktop hands-free and vehicle-mounted hands-free UE of ≥ 55 dB is recommended as a performance objective when measured under free-field conditions at the nominal setting of the volume control. Depending on the UE idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 dB.

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 hand-held hands-free UE

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

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

NOTE: A TCLw for the hand-held hands-free UE of ≥ 55 dB is recommended as a performance objective when measured under free-field conditions at the nominal setting of the volume control. Depending on the UE idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 dB.

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 should be ≥ 55 dB at the nominal setting of the volume control.

NOTE: It is recommended that the volume control should be set back to nominal after each call unless TCLw ≥ 55 dB can also be maintained with the maximum volume setting. Depending on the UE idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 dB.

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 should be ≥ 55 dB at the nominal setting of the volume control.

NOTE: It is recommended that the volume control should be set back to nominal after each call unless TCLw ≥ 55 dB can also be maintained with the maximum volume setting. Depending on the UE idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 dB.

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 the 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 table 7 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 the DRP with diffuse-field correction shall meet the requirements in this sub-clause 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 the DRP with diffuse-field correction is up to 10 dBPa. For a sound pressure ≥ 10 dBPa at the DRP with diffuse-field correction there is no distortion requirement.

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

Frequency (Hz)	Receiving level at the digital interface (dBm0)	Receiving ratio at nominal volume setting (dB)	Receiving ratio at maximum volume setting (dB)
315	-16	20	
408	-16	28	
510	-16	28	
816	-16	28	
1 020	0	25,5	
	-3	31,2	
	-10	33,5	
	-16	33,5	
	-20	33	
	-30	30,5	
	-40 (*)	22,5 (*)	
	-45 (*)	17,5 (*)	
NOTE: (*)For levels -40 and -45 dBm0 a lower signal-to-total distortion ratio may not be possible, and hence would not be regarded as a failing result. However, the obtained results would 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.

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

5.9 Void

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.

5.11 Sending performance in the presence of ambient noise

5.11.1 General

For sending, in handset mode, the UE shall reduce the ambient noise picked up by the microphone(s) without significantly degrading the quality of the speech signal.

5.11.2 Connections with handset UE

The UE shall comply with the following requirements:

S-MOS-LQOn
- The average of S-MOS-LQOn scores across all test conditions shall be ≥ 3.0
- As a performance objective, the average of the S-MOS-LQOn scores across all test conditions should be ≥ 3.5

N-MOS-LQOn
- The average of the N-MOS-LQOn scores across all test conditions shall be ≥ 2.3
- As a performance objective, the average of N-MOS-LQOn scores across all test conditions should be ≥ 3.0

G-MOS-LQOn
- No requirement.

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

5.12 Delay

5.12.0 UE delay definition

For UMTS circuit-switched operation and MTSI-based speech with LTE access, the UE delays in the send and receive directions are defined as:

- The UE delay in the send (uplink) direction is the delay between the first acoustic event at the MRP to the last bit of the corresponding speech frame at the UE antenna
- The UE delay in the receive (downlink) direction is the delay between the first bit of a speech frame at the UE antenna and the first acoustic event at the DRP corresponding to that speech frame

NOTE: In order to harmonize UMTS and LTE delay definitions, the reference points for UMTS UE delay have been changed in Rel-12. Prior to Rel-12, the UE reference points for UMTS were implicitly defined by the compensation factors declared by system simulator vendors, i.e. half of the air interface delay was attributed to the UE, and the last acoustic event at DRP was used for the receive measurement instead of the first.

Considering 10ms for half of the transmission time in each direction of a UMTS call, a speech frame size of 20ms, a codec look-ahead of 5ms, and a difference between the first and the last acoustic event of 20 ms, the previous reference points took into account a UE implementation independent delay of 2x10ms + 25 ms + 20 ms = 65 ms. The same UE delay remains in the new definition above, which attributes the full air interface delay to the UE but uses the first acoustic event at the DRP, instead of the last (2x20ms + 25 ms + 0 ms = 65 ms). Hence, the UMTS requirements with these new reference points remain the same.

5.12.1 Handset UE

It is in general desirable to minimize UE delays to ensure low enough end-to-end delays and hence a good conversational experience, guidance is found in ITU-T Recommendation G.114.

For UMTS circuit-switched AMR speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall in any case be ≤ 220ms and should be ≤ 185ms.

NOTE: A delay ≤ 185 ms might not be achievable in some cases due to UE implementation trade-offs between delay and other parameters such as speech quality enhancement, performance of noise reduction or UE power consumption optimization, and UE implementation issues such as rebuffering between components.

For MTSI-based speech-only with LTE access in error and jitter free conditions and AMR speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) should be ≤ 150ms. If this performance objective cannot be met, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall in any case be ≤ 190ms.

For MTSI-based speech-only with LTE access in conditions with simulated packet arrival time variations and packet loss and AMR speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall be less than or equal to the delay requirements in Table 8bis, while meeting the speech quality targets defined.

NOTE: The UE delay requirements for MTSI-based speech-only with LTE access is derived from:

- A speech frame buffering and codec look-ahead of 25ms.
- An air interface transmission time of 1ms on receive and 1ms on the send direction.
- A budget allowance for a jitter buffer depth of 40ms for error and jitter free conditions and test conditions 0 and 1 of Table 8bis, and 80ms for test condition 2 of Table 8bis.
- A budget allowance for vendor specific implementation of 83ms corresponding to the performance objective and 123ms corresponding to the required maximum UE send and receive delay.

Table 8bis: UE delay and speech quality requirements for LTE access

Test Condition	Delay and Loss Profile (Note 1)	Performance Objectives for Maximum Delay	Requirements for Maximum Delay	Speech Quality Requirements (Note 2)
0	Error and jitter free condition	$T_S + T_R \leq$ 150ms	$T_S + T_R \leq$ 190ms	No requirement, reference score $MOS\text{-}LQO_{REF}$
1	dly_profile_20msDRX_10pct_BLER_e2e	$T_S + T_R \leq$ 150ms	$T_S + T_R \leq$ 190ms	$MOS\text{-}LQO_{TEST} \geq MOS\text{-}LQO_{REF}$ - 0.3
2	dly_profile_40msDRX_10pct_BLER_e2e	$T_S + T_R \leq$ 190ms	$T_S + T_R \leq$ 230ms	$MOS\text{-}LQO_{TEST} \geq MOS\text{-}LQO_{REF}$ - 0.3
	NOTE 1: The delay profiles for test condition 1 and 2 are theoretically constructed to simulate a semi-persistent scheduling transmission scheme with DRX enabled and target BLER in sending and receiving directions of 10%, with +/- 3ms of EPC jitter. Delay profiles are injected at the IP layer of the test system. Delay profiles are attached electronically to document 3GPP TS 26.132 [1]. The delay profiles in test condition 1 and 2 are static delay variation conditions and do not expose the UE to packet delay variations in the full range of the packet delay budget as defined for QCI1 in 3GPP TS 23.203 [18]. A third test condition that exposes the UE to non-stationary packet delay variations experienced in live operation and packet delay variations in the full range of the packet delay budget for QCI1, and accompanied delay and speech quality requirements, is for further study.			
	NOTE 2: The purpose of this test is to provide a relative comparison of the objective speech quality between the reference and test conditions. This test is not to be construed as a method to evaluate the absolute objective speech quality of the device.			

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

5.12.2 Headset UE

5.12.2.1 Wired headset

It is in general desirable to minimize UE delays to ensure low enough end-to-end delays and hence a good conversational experience, guidance is found in ITU-T Recommendation G.114.

For UMTS circuit-switched AMR speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall in any case be ≤ 220ms and should be ≤ 185ms.

NOTE: A delay ≤ 185 ms might not be achievable in some cases due to UE implementation trade-offs between delay and other parameters such as speech quality enhancement, performance of noise reduction or UE power consumption optimization, and UE implementation issues such as rebuffering between components.

For MTSI-based speech-only with LTE access in error and jitter free conditions and AMR speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) should be ≤ 150ms. If this performance objective cannot be met, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall in any case be ≤ 190ms.

For MTSI-based speech-only with LTE access in conditions with simulated packet arrival time variations and packet loss and AMR speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall be less than or equal to the delay requirements in Table 8ter, while meeting the speech quality targets defined.

NOTE: The UE delay requirements for MTSI-based speech-only with LTE access is derived from:

- A speech frame buffering and codec look-ahead of 25ms.
- An air interface transmission time of 1ms on receive and 1ms on the send direction.
- A budget allowance for a jitter buffer depth of 40ms for error and jitter free conditions and test conditions 0 and 1 of Table 8ter, and 80ms for test condition 2 of Table 8ter.
- A budget allowance for vendor specific implementation of 83ms corresponding to the performance objective and 123ms corresponding to the required maximum UE send and receive delay.

Table 8ter: UE delay and speech quality requirements for LTE access

Test Condition	Delay and Loss Profile (Note 1)	Performance Objectives for Maximum Delay	Requirements for Maximum Delay	Speech Quality Requirements (Note 2)
0	Error and jitter free condition	$T_S + T_R \leq$ 150ms	$T_S + T_R \leq$ 190ms	No requirement, reference score $MOS\text{-}LQO_{REF}$
1	dly_profile_20msDRX_10pct_BLER_e2e	$T_S + T_R \leq$ 150ms	$T_S + T_R \leq$ 190ms	$MOS\text{-}LQO_{TEST} \geq MOS\text{-}LQO_{REF}$ - 0.3
2	dly_profile_40msDRX_10pct_BLER_e2e	$T_S + T_R \leq$ 190ms	$T_S + T_R \leq$ 230ms	$MOS\text{-}LQO_{TEST} \geq MOS\text{-}LQO_{REF}$ - 0.3
	NOTE 1: The delay profiles for test condition 1 and 2 are theoretically constructed to simulate a semi-persistent scheduling transmission scheme with DRX enabled and target BLER in sending and receiving directions of 10%, with +/- 3ms of EPC jitter. Delay profiles are injected at the IP layer of the test system. Delay profiles are attached electronically to document 3GPP TS 26.132 [1]. The delay profiles in test condition 1 and 2 are static delay variation conditions and do not expose the UE to packet delay variations in the full range of the packet delay budget as defined for QCI1 in 3GPP TS 23.203 [18]. A third test condition that exposes the UE to non-stationary packet delay variations experienced in live operation and packet delay variations in the full range of the packet delay budget for QCI1, and accompanied delay and speech quality requirements, is for further study.			
	NOTE 2: The purpose of this test is to provide a relative comparison of the objective speech quality between the reference and test conditions. This test is not to be construed as a method to evaluate the absolute objective speech quality of the device.			

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

5.12.2.2 Wireless headset

For further study.

5.13 Echo control characteristics

Echo cancellation is commonly deployed in the UE to fulfil the Acoustic echo control requirements. Echo cancellers are complex devices of which the subjective performance is affected by several attributes. The main attribute is its ability to suppress echo. The process of suppressing the echo may introduce impairments to the near-end speech signal, mainly manifested as distortion or clipping of the near-end signal during simultaneous speech from both the far and near-end ('double-talk').

To characterise the echo control performance, the activity (in % of total time) and averaged level difference (in dB) of the duration of any level difference according to Figure 6a and Table 8a between the clean near-end signal and the send-signal shall be reported for 'double-talk' as well as the far-end single talk periods adjacent to the 'double-talk'.

NOTE: The limits for specifying the categories in Figure 6a and Table 8a are provisional pending further analysis and validation.

NOTE: The categories in Figure 6a and Table 8a are labelled in a functional order and the subjective impression of the respective categories is for further study.

All percentage values and averaged level differences described in the relevant test of 3GPP TS 26.132 shall be reported.

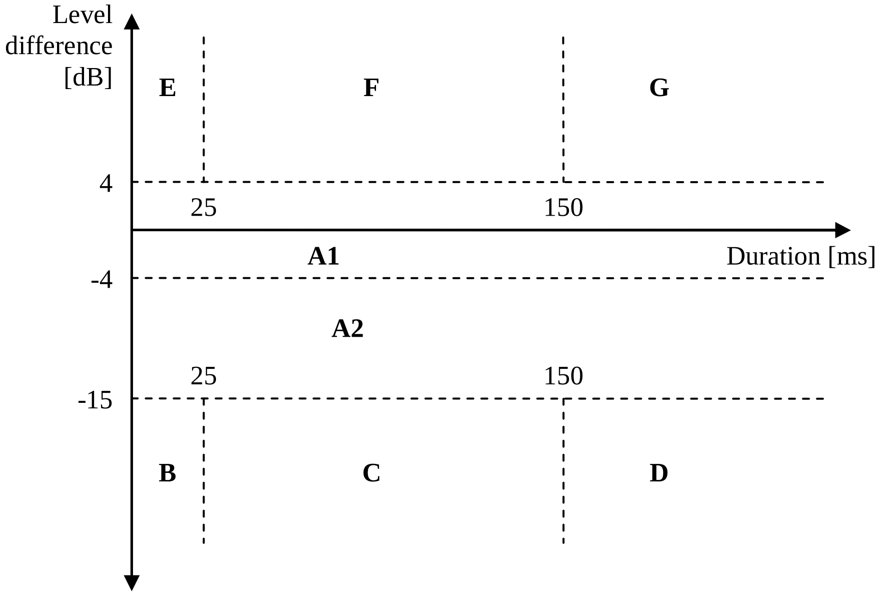

Figure 6a: Classification of echo canceller performance

Table 8a: Categories for echo canceller performance classification

Category	Description
A1	Full-duplex and full transparency
A2	Full-duplex with level loss in Tx
B	Very short clipping
C	Short clipping resulting in loss of syllables
D	Clipping resulting in loss of words
E	Very short residual echo
F	Echo bursts
G	Continuous echo

5.13.1 Handset

Requirements are for further study.

5.13.2 Headset

Requirements are for further study.

5.13.3 Handheld hands-free

Requirements are for further study.

5.13.4 Desktop and vehicle mounted hands-free

Requirements are for further study.

6 Wideband telephony transmission performance

6.1 Applicability

The performance requirements in this 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 narrowband telephony. When a wideband-enabled terminal is providing narrowband telephony, the requirements in clause 5, "narrowband telephony transmission performance" shall apply.

6.2 Overall loss/loudness ratings

6.2.1 General

An international connection involving a 3G or LTE network and the PSTN should meet the overall loudness rating (OLR) limits in ITU-T Recommendation G.111 [4]. The national parts of the connection should therefore meet the send and receive loudness rating (SLR, RLR) limits in ITU-T Recommendation G.121 [5].

For the case where digital routings are used to connect the 3G or LTE 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 or LTE network. The limits given below are consistent with the national extension limits and long term objectives in ITU-T Recommendation G.121 [5].

The SLR and RLR values for the 3G or LTE 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 or LTE 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 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 and shall not be ≥ (equal or quieter than) -3 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. For a vehicle-mounted hands-free UE:
   Where a user-controlled volume control is provided, the RLR shall meet the nominal value for at least 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 the maximum volume control setting should be ≤ (equal or louder than) -2 dB.

2. For a desktop hands-free UE:
Where a user-controlled volume control is provided, the RLR shall meet the nominal value for at least 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 usage environment.
RLR at the 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 [4], lies between 1 and 3 dB. The higher RLR requirement of 5 dB for desktop hands-free is appreciative of the limitations in transducer output with current typical form factors.

6.2.4 Connections with hand-held 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 nominal value for at least one setting of the control.

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

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

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.380 [9]. 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 nominal value for at least one setting of the control. When the control is set to maximum, the RLR shall not be ≤ (equal or louder than) -13 dB.

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

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

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 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 8 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 the 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 be ≤ -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 ≤ -64 dBPa(A).

NOTE: In a connection with the PSTN, noise conditions as described in ITU-T Recommendation G.103 [3] can be expected at the input (POI) of the 3G or LTE 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 the 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: Handset and headset sending sensitivity/frequency mask

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.		

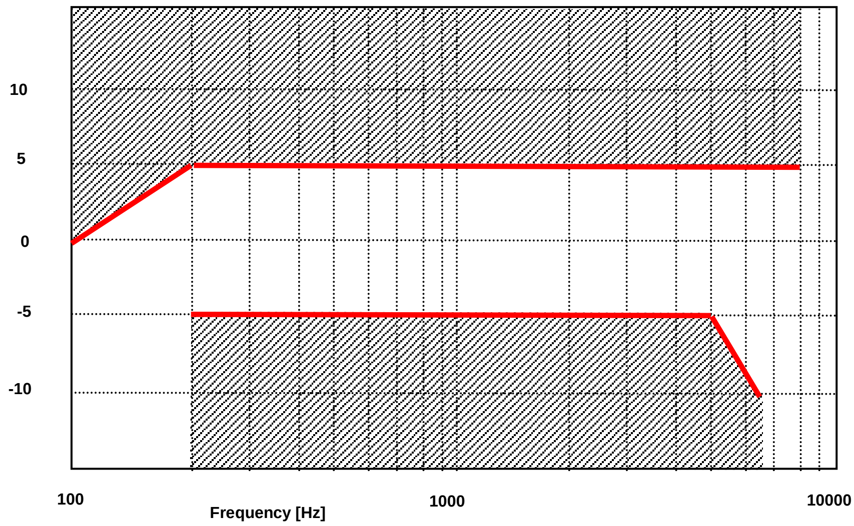

Figure 9: Handset and headset sending sensitivity/frequency mask

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 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 10 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 10: Handset and headset 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: All sensitivity values are expressed in dB on an arbitrary 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 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 (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.		

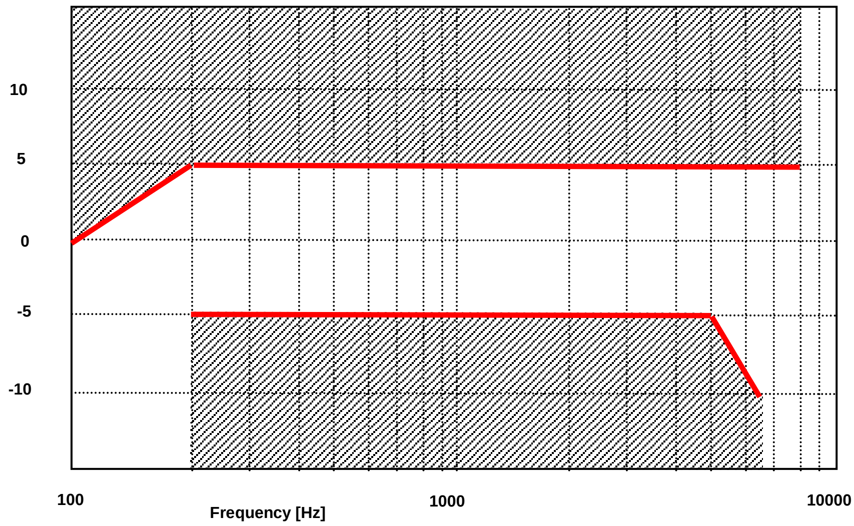

Figure 11: Desktop and vehicle-mounted hands-free sending sensitivity/frequency mask

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 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 12 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 12: Desktop and vehicle-mounted 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.		

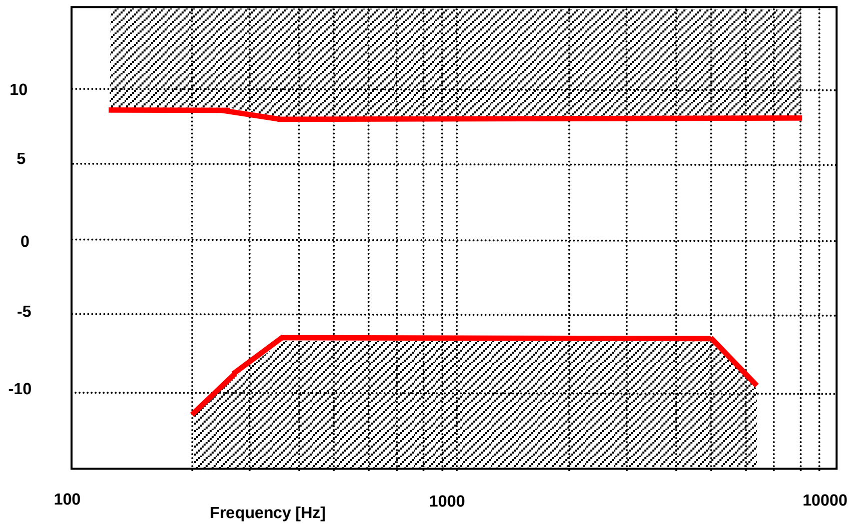

Figure 12: Desktop and vehicle-mounted hands-free receiving sensitivity/frequency mask

It is recommended as a performance objective 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 12a: Performance objective for desktop and vehicle-mounted hands-free receiving sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
100	0	
200	0	-18
250	0	-15
315	0	-12
6 300	0	-12
8 000	0	

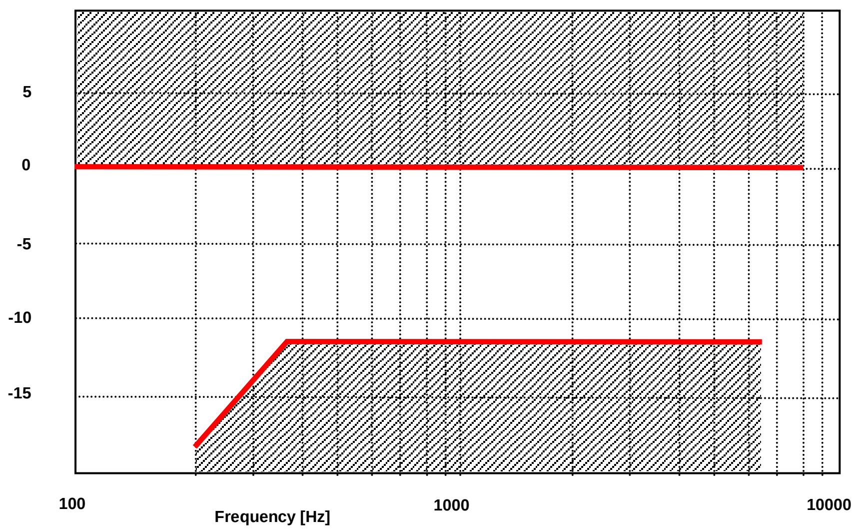

Figure 12a: Performance objective for desktop and vehicle-mounted hands-free receiving sensitivity/frequency response

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

6.4.5 Hand-held 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: Hand-held hands-free sending sensitivity/frequency mask

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.		

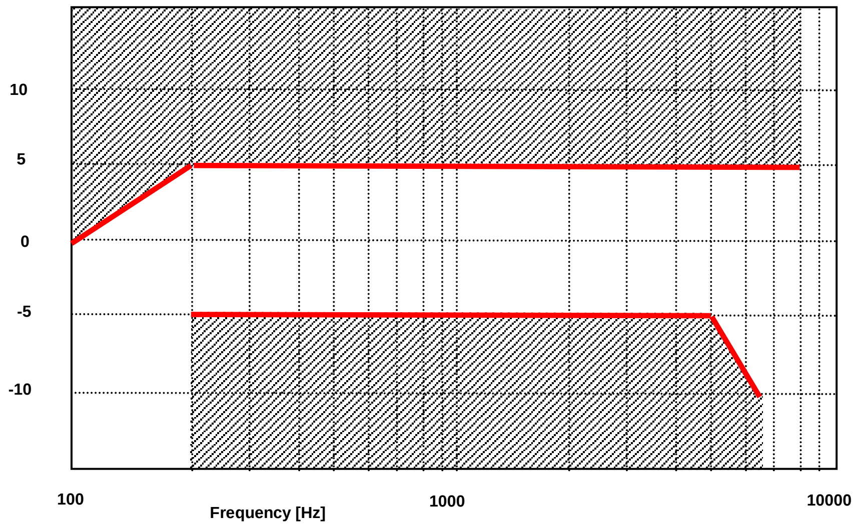

Figure 13: Hand-held hands-free sending sensitivity/frequency mask

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

6.4.6 Hand-held 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 14 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 14: Hand-held hands-free receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
315	6	
630	6	-12
800	6	-6
4 000	6	-6
6 300	6	-12
8 000	6	
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

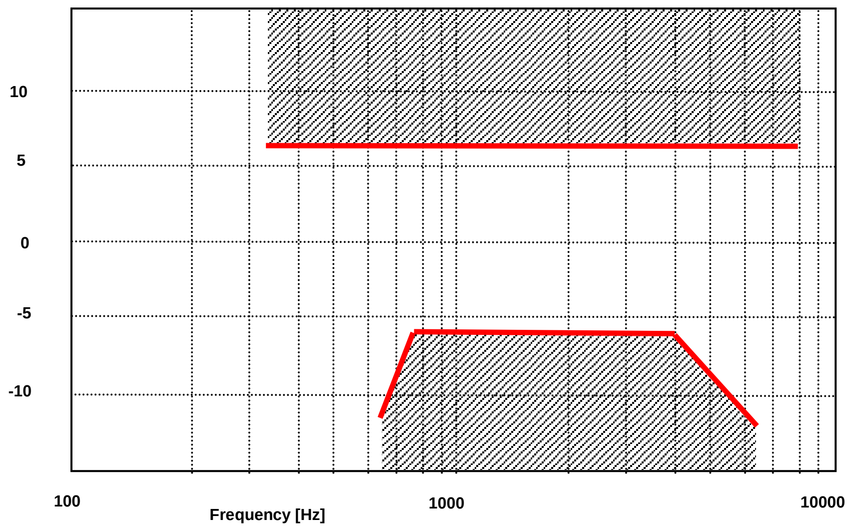

Figure 14: Hand-held hands-free receiving sensitivity/frequency mask

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

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

Frequency (Hz)	Upper limit	Lower limit
315	6	
400	6	-12
500	6	-6
4 000	6	-6
6 300	6	-12
8 000	6	
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

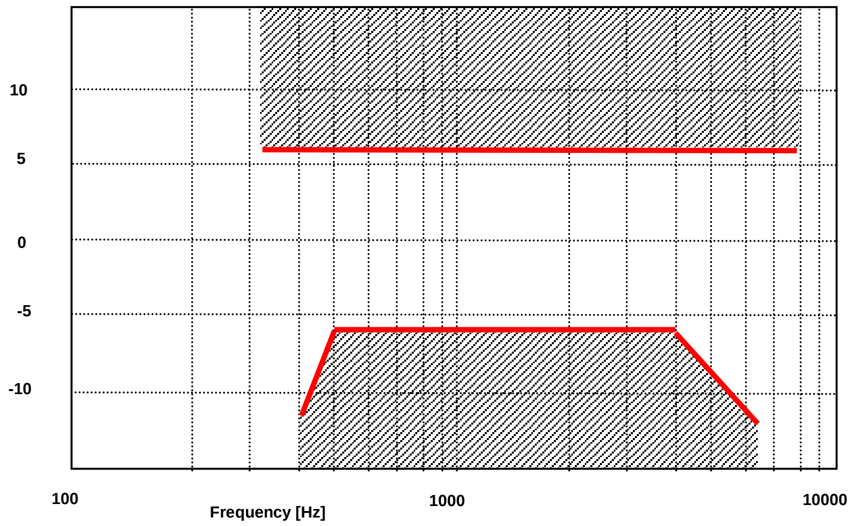

Figure 14.a: Performance objective for hand-held hands-free receiving sensitivity/frequency mask

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 the nominal setting of the volume control. For all other positions of the volume control, the STMR shall be ≥ 10 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

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

NOTE: The measured result is only applicable where the level of the electrical sidetone is sufficiently high to be measured. While the STMR value may indicate the presence of sidetone it should be ensured that this is not primarily due to the acoustical or mechanical sidetone path when interpreting sidetone delay results.

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 or LTE network at the POI should meet the principles of the requirements in clauses 2 and 3 of ITU-T Recommendation G.122 [6]. These requirements will be met if the attenuation between the digital input and digital output at the POI is ≥ 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 volume control set to maximum for each following condition):

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

Headset UE: for further study;

Hands-free 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) if applicable.

6.7 Acoustic echo control

6.7.1 General

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

The use of acoustic echo control is not mandated for 3G or LTE networks and the connection between the UE and the POI is zero loss. Therefore the acoustic echo control provided in the 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 as that used in DTX.

6.7.2 Acoustic echo control in 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 the nominal setting of the volume control.

NOTE: A TCLw for desktop hands-free and vehicle-mounted hands-free UE of ≥ 55 dB is recommended as a performance objective when measured under free-field conditions at the nominal setting of the volume control. Depending on the UE idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 dB.

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 hand-held hands-free UE

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

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

NOTE: A TCLw for the hand-held hands-free UE of ≥ 55 dB is recommended as a performance objective when measured under free-field conditions at the nominal setting of the volume control. Depending on the UE idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 dB.

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 should be ≥ 55 dB at the nominal setting of the volume control.

With the volume control set to maximum TCLw should be ≥55 dB.

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

NOTE. Depending on the UE idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 dB.

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 the nominal setting of the volume control.

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

NOTE: Depending on the UE idle channel noise in the sending direction, it may not always be possible to measure an echo loss ≥ 55 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.

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 the 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 (Hz)	Sending level (dBPa at the MRP)	Sending Ratio (dB)
315	-4,7	28
408	-4,7	32
510	-4,7	32
816	-4,7	32
1 020	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 table 15 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 the DRP with diffuse-field correction shall meet the requirements in this sub-clause at the nominal setting of the volume control (except where another volume setting 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 the DRP with diffuse-field correction is < 10 dBPa. For a sound pressure ≥ 10 dBPa at the DRP with diffuse-field correction there is no distortion requirement.

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

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

Frequency (Hz)	Receiving level at the digital interface (dBm0)	Receiving ratio at nominal volume setting (dB)	Receiving ratio at maximum volume setting (dB)
315	-16	20	
408	-16	28	
510	-16	28	
816	-16	28	
1 020	0	25,5	tbd
	-3	31,5	tbd
	-10	33,5	tbd
	-16	33,5	tbd
	-20	33	tbd
	-30	30,5	tbd

Frequency (Hz)	Receiving level at the digital interface (dBm0)	Receiving ratio at nominal volume setting (dB)	Receiving ratio at maximum volume setting (dB)
	-40	22,5 (*)	tbd
	-45	17,5 (*)	tbd
NOTE: (*)For levels -40 and -45 dBm0 a lower signal-to-total distortion ratio may not be possible, and hence would not be regarded as a failing result. However, the obtained results would 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 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.

6.9 Void

6.10 Sending performance in the presence of ambient noise

6.10.1 General

For sending, in handset mode, the UE shall reduce the ambient noise picked up by the microphone(s) without significantly degrading the quality of the speech signal.

6.10.2 Connections with handset UE

The UE shall comply with the following requirements:

S-MOS-LQOw
- The average of S-MOS-LQOw scores across all test conditions shall be ≥ 3.0
- As a performance objective, the average of the S-MOS-LQOw scores across all test conditions should be ≥ 3.5

N-MOS-LQOw
- The average of the N-MOS-LQOw scores across all test conditions shall be ≥ 2.3
- As a performance objective, the average of N-MOS-LQOw scores across all test conditions should be ≥ 3.0

G-MOS-LQOw
- No requirement.

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

6.11 Delay

6.11.0 UE delay definition

For UMTS circuit-switched operation and MTSI-based speech with LTE access, the UE delays in the send and receive directions are defined as:

- The UE delay in the send (uplink) direction is the delay between the first acoustic event at the MRP to the last bit of the corresponding speech frame at the UE antenna
- The UE delay in the receive (downlink) direction is the delay between the first bit of a speech frame at the UE antenna and the first acoustic event at the DRP corresponding to that speech frame

NOTE: In order to harmonize UMTS and LTE delay definitions, the reference points for UMTS UE delay have been changed in Rel-12. Prior to Rel-12, the UE reference points for UMTS were implicitly defined by the compensation factors declared by system simulator vendors, i.e. half of the air interface delay was attributed to the UE, and the last acoustic event at DRP was used for the receive measurement instead of the first.

Considering 10ms for half of the transmission time in each direction of a UMTS call, a speech frame size of 20ms, a codec look-ahead of 5ms, and a difference between the first and the last acoustic event of 20 ms, the previous reference points took into account a UE implementation independent delay of 2x10ms + 25 ms + 20 ms = 65 ms. The same UE delay remains in the new definition above, which attributes the full air interface delay to the UE but uses the first acoustic event at the DRP, instead of the last (2x20ms + 25 ms + 0 ms = 65 ms). Hence, the UMTS requirements with these new reference points remain the same.

6.11.1 Handset UE

It is in general desirable to minimize UE delays to ensure low enough end-to-end delays and hence a good conversational experience, guidance is found in ITU-T Recommendation G.114.

For UMTS circuit-switched AMR-WB speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall in any case be ≤ 220ms and should be ≤ 185ms.

NOTE: A delay ≤ 185 ms might not be achievable in some cases due to UE implementation trade-offs between delay and other parameters such as speech quality enhancement, performance of noise reduction or UE power consumption optimization, and UE implementation issues such as rebuffering between components.

For MTSI-based speech-only with LTE access in error and jitter free conditions and AMR-WB speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) should be ≤ 150ms. If this performance objective cannot be met, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall in any case be ≤ 190ms.

For MTSI-based speech-only with LTE access in conditions with simulated packet arrival time variations and packet loss and AMR-WB speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall be less than or equal to the delay requirements in Table 16a1, while meeting the speech quality targets defined.

NOTE: The UE delay requirements for MTSI-based speech-only with LTE access is derived from:

- A speech frame buffering and codec look-ahead of 25ms.
- An air interface transmission time of 1ms on receive and 1ms on the send direction.
- A budget allowance for a jitter buffer depth of 40ms for error and jitter free conditions and test conditions 0 and 1 of Table 16a1, and 80ms for test condition 2 of Table 16a1.
- A budget allowance for vendor specific implementation of 83ms corresponding to the performance objective and 123ms corresponding to the required maximum UE send and receive delay.

Table 16a1: UE delay and speech quality requirements for LTE access

Test Condition	Delay and Loss Profile (Note 1)	Performance Objectives for Maximum Delay	Requirements for Maximum Delay	Speech Quality Requirements (Note 2)
0	Error and jitter free condition	$T_S + T_R \leq$ 150ms	$T_S + T_R \leq$ 190ms	No requirement, reference score $MOS\text{-}LQO_{REF}$
1	dly_profile_20msDRX_10pct_BLER_e2e	$T_S + T_R \leq$ 150ms	$T_S + T_R \leq$ 190ms	$MOS\text{-}LQO_{TEST} \geq MOS\text{-}LQO_{REF} - 0.3$
2	dly_profile_40msDRX_10pct_BLER_e2e	$T_S + T_R \leq$ 190ms	$T_S + T_R \leq$ 230ms	$MOS\text{-}LQO_{TEST} \geq MOS\text{-}LQO_{REF} - 0.3$
	NOTE 1: The delay profiles for test condition 1 and 2 are theoretically constructed to simulate a semi-persistent scheduling transmission scheme with DRX enabled and target BLER in sending and receiving directions of 10%, with +/- 3ms of EPC jitter. Delay profiles are injected at the IP layer of the test system. Delay profiles are attached electronically to document 3GPP TS 26.132 [1]. The delay profiles in test condition 1 and 2 are static delay variation conditions and do not expose the UE to packet delay variations in the full range of the packet delay budget as defined for QCI1 in 3GPP TS 23.203 [18]. A third test condition that exposes the UE to non-stationary packet delay variations experienced in live operation and packet delay variations in the full range of the packet delay budget for QCI1, and accompanied delay and speech quality requirements, is for further study.			
	NOTE 2: The purpose of this test is to provide a relative comparison of the objective speech quality between the reference and test conditions. This test is not to be construed as a method to evaluate the absolute objective speech quality of the device.			

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

6.11.2 Headset UE

6.11.2.1 Wired headset

It is in general desirable to minimize UE delays to ensure low enough end-to-end delays and hence a good conversational experience, guidance is found in ITU-T Recommendation G.114.

For UMTS circuit-switched AMR-WB speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall in any case be ≤ 220ms and should be ≤ 185ms.

NOTE: A delay ≤ 185 ms might not be achievable in some cases due to UE implementation trade-offs between delay and other parameters such as speech quality enhancement, performance of noise reduction or UE power consumption optimization, and UE implementation issues such as rebuffering between components.

For MTSI-based speech-only with LTE access in error and jitter free conditions and AMR-WB speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) should be ≤ 150ms. If this performance objective cannot be met, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall in any case be ≤ 190ms.

For MTSI-based speech-only with LTE access in conditions with simulated packet arrival time variations and packet loss and AMR-WB speech codec operation, the sum of the UE delays in sending and receiving directions ($T_S + T_R$) shall be less than or equal to the delay requirements in Table 16a2, while meeting the speech quality targets defined.

NOTE: The UE delay requirements for MTSI-based speech-only with LTE access is derived from:

- A speech frame buffering and codec look-ahead of 25ms.
- An air interface transmission time of 1ms on receive and 1ms on the send direction.
- A budget allowance for a jitter buffer depth of 40ms for error and jitter free conditions and test conditions 0 and 1 of Table 16a2, and 80ms for test condition 2 of Table 16a2.
- A budget allowance for vendor specific implementation of 83ms corresponding to the performance objective and 123ms corresponding to the required maximum UE send and receive delay.

Table 16a2: UE delay and speech quality requirements for LTE access

Test Condition	Delay and Loss Profile (Note 1)	Performance Objectives for Maximum Delay	Requirements for Maximum Delay	Speech Quality Requirements (Note 2)
0	Error and jitter free condition	$T_S + T_R \leq 150ms$	$T_S + T_R \leq 190ms$	No requirement, reference score MOS-LQO$_{REF}$
1	dly_profile_20msDRX_10pct_BLER_e2e	$T_S + T_R \leq 150ms$	$T_S + T_R \leq 190ms$	MOS-LQO$_{TEST}$ ≥ MOS-LQO$_{REF}$ - 0.3
2	dly_profile_40msDRX_10pct_BLER_e2e	$T_S + T_R \leq 190ms$	$T_S + T_R \leq 230ms$	MOS-LQO$_{TEST}$ ≥ MOS-LQO$_{REF}$ - 0.3
	NOTE 1: The delay profiles for test condition 1 and 2 are theoretically constructed to simulate a semi-persistent scheduling transmission scheme with DRX enabled and target BLER in sending and receiving directions of 10%, with +/- 3ms of EPC jitter. Delay profiles are injected at the IP layer of the test system. Delay profiles are attached electronically to document 3GPP TS 26.132 [1]. The delay profiles in test condition 1 and 2 are static delay variation conditions and do not expose the UE to packet delay variations in the full range of the packet delay budget as defined for QCI1 in 3GPP TS 23.203 [18]. A third test condition that exposes the UE to non-stationary packet delay variations experienced in live operation and packet delay variations in the full range of the packet delay budget for QCI1, and accompanied delay and speech quality requirements, is for further study.			
	NOTE 2: The purpose of this test is to provide a relative comparison of the objective speech quality between the reference and test conditions. This test is not to be construed as a method to evaluate the absolute objective speech quality of the device.			

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

6.11.2.2 Wireless headset

For further study.

6.12 Echo control characteristics

Echo cancellation is commonly deployed in the UE to fulfil the Acoustic echo control requirements. Echo cancellers are complex devices of which the subjective performance is affected by several attributes. The main attribute is its ability to suppress echo. The process of suppressing the echo may introduce impairments to the near-end speech signal, mainly manifested as distortion or clipping of the near-end signal during simultaneous speech from both the far and near-end ('double-talk').

To characterise the echo control performance, the activity (in % of total time) and averaged level difference (in dB) of the duration of any level difference according to Figure 14b and Table 16b between the clean near-end signal and the send-signal shall be reported for 'double-talk' as well as the far-end single talk periods adjacent to the 'double-talk'.

NOTE: The limits for specifying the categories in Figure 14b and Table 16b are provisional pending further analysis and validation.

NOTE: The categories in Figure 14b and Table 16b are labelled in a functional order and the subjective impression of the respective categories is for further study.

All percentage values and averaged level differences described in the relevant test of 3GPP TS 26.132 shall be reported.

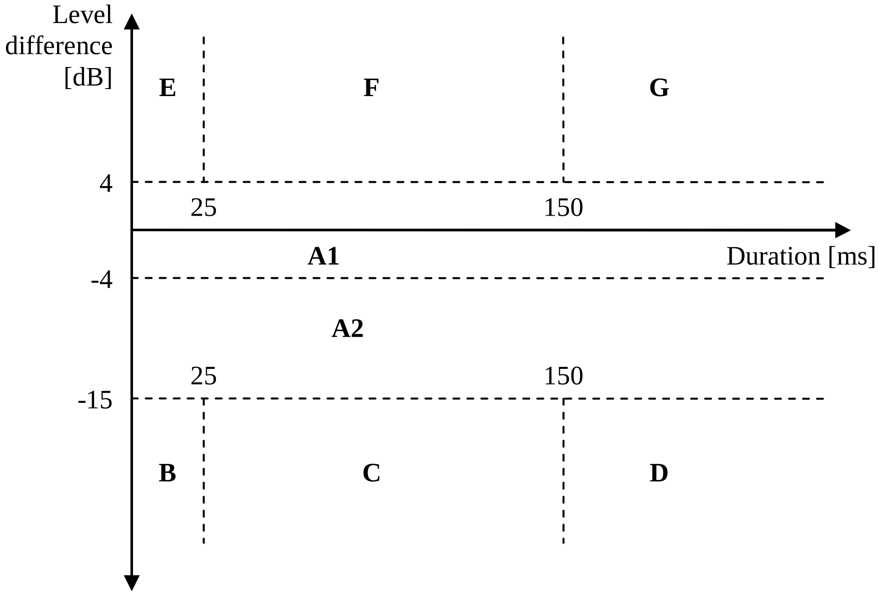

Figure 14b: Classification of echo canceller performance

Table 16b: Categories for echo canceller performance classification

Category	Description
A1	Full-duplex and full transparency
A2	Full-duplex with level loss in Tx
B	Very short clipping
C	Short clipping resulting in loss of syllables
D	Clipping resulting in loss of words
E	Very short residual echo
F	Echo bursts
G	Continuous echo

6.12.1 Handset

Requirements are for further study.

6.12.2 Headset

Requirements are for further study.

6.12.3 Handheld hands-free

Requirements are for further study.

6.12.4 Desktop and vehicle mounted hands-free

Requirements are 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
2011-06	52	SP-110304	0040	3	Remaining modifications to EAAT WI	10.1.0	10.2.0
2011-09	53	SP-110549	0046	1	Note on applicability of WB sidetone delay	10.2.0	10.3.0
2011-11	54	SP-110793	0047		Correction of sending idle channel noise requirement	10.3.0	10.4.0
2011-11	54	SP-110793	0048		Corrections to volume control setting	10.3.0	10.4.0
2012-09	57	SP-120503	0052	3	Addition of UE delay requirement	10.4.0	11.0.0
2012-09	57	SP-120503	0053	1	Extension of Acoustic Test Requirements	10.4.0	11.0.0
2012-12	58	SP-120760	0054	2	Minor clarification of UMTS UE Delay Requirements	11.0.0	11.1.0
2013-03	59	SP-130017	0055	1	Voiding of ambient noise rejection test cases	11.1.0	11.2.0
2013-06	60	SP-130189	0056		Adding receiving distortion tests at frequencies lower than 1020Hz	11.2.0	12.0.0
2013-06	60	SP-130189	0057	1	Update acoustic requirements specification to cover MTSI speech-only services over LTE (narrowband and wideband)	11.2.0	12.0.0
2013-12	62	SP-130563	0061	2	STMR - adaptation to modern form factors	12.0.0	12.1.0
2014-09	65	SP-140469	0062	1	LTE UE delay requirements	12.1.0	12.2.0

History

Document history		
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