Technical Specification Group Services and System Aspects
Meeting #11, Palm Springs, USA, 19-22 March 2001

TSGS#11(01)0106

Source: **TSG-SA WG4**

Title: **CRs to TS 26.131 on Harmonisation of acoustic requirements between 3GPP and GSM and WB acoustic requirements (R99 and Release 4)**

Document for: **Approval**

Agenda Item: **7.4.3**

The following CRs were agreed at the TSG-SA WG4 meetings #16 and are presented to TSG SA #11 for approval.

Spec	CR	Rev	Phase	Subject	Cat	Ver	WG	Meeting	S4 doc
26.131	005	1	R99	Harmonisation of narrow-band acoustic requirements between 3GPP and GSM	F	3.1.0	S4	TSG-SA WG4#16	S4-010236
26.131	006	3	Rel-4	Wideband acoustic requirements	B	3.1.0	S4	TSG-SA WG4#16	S4-010271

3GPP TSG-SA4 Meeting #16 ***Tdoc S4-010236***
S4#16, Sophia Antipolis, France, Feb. 26-March 2, 2001

CR-Form-v3

CHANGE REQUEST

⌘	**26.131**	**CR**	**005**	⌘	rev	**1**	⌘	Current version:	**3.1.0**	⌘

For ***HELP*** *on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

Proposed change affects: ⌘ (U)SIM [ ] ME/UE [X] Radio Access Network [ ] Core Network [ ]

Title:	⌘	Harmonisation of narrow-band acoustic requirements between 3GPP and GSM
Source:	⌘	TSG-SA WG4
Work item code:	⌘	TEI
Date:	⌘	2001-2-21
Category:	⌘	**F**
Release:	⌘	R99

Use one of the following categories:
- ***F*** *(essential correction)*
- ***A*** *(corresponds to a correction in an earlier release)*
- ***B*** *(Addition of feature),*
- ***C*** *(Functional modification of feature)*
- ***D*** *(Editorial modification)*

Detailed explanations of the above categories can be found in 3GPP TR 21.900.

Use one of the following releases:
- *2* *(GSM Phase 2)*
- *R96* *(Release 1996)*
- *R97* *(Release 1997)*
- *R98* *(Release 1998)*
- *R99* *(Release 1999)*
- *REL-4* *(Release 4)*
- *REL-5* *(Release 5)*

Reason for change:	⌘	Harmonisation of narrow-band acoustic requirements between 3GPP and GSM, as required in Tdoc 147 Appendix_A.
Summary of change:	⌘	Pending approval of Tdoc 147, most of GSM 03.50 R4 will refer to TS 26.131. Various sections of TS 26.131 v3.1.0 are maked '*for further study.*' This change request updates these sections with actual requirements so that the GSM specifications may refer to TS 26.131. This will permit handling of only one document for GSM and 3GPP acoustic requirements. References to Wideband acoustic requirements are deleted. This is because this change request applies only to Release 99. Release 99 does not specify wideband speech services. A spereate CR will be introduced to Release 4 of TS 26.131 to introduce wideband acoustic requirements.
Consequences if not approved:	⌘	Various parts of TS 26.131 will remain incomplete. Furthremore, there will be differences between GSM and 3GPP specifications, preventing harmonisation between GSM and 3GPP acoustic requirements.

Clauses affected:	⌘	2, 4, and 5
Other specs affected:	⌘	[ ] Other core specifications ⌘ [ ] Test specifications [ ] O&M Specifications
Other comments:	⌘	Appendix_A of Tdoc 147 specifies the changes raised in this CR.

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.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

[1] 3GPP Technical Specification 3G 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.38 (1993): "Transmission characteristics of operator telephone systems (OTS)".

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

[11] ITU-T Recommendation P.79 (1999): “Calculation of loudness ratings for telephone sets.”

4 Interfaces

~~4.1 Narrow-band telephony~~

The interfaces required to define terminal acoustic characteristics ~~for narrow-band telephony~~ are shown in figure 1. These are the air interface~~,~~ and the point of interconnect (POI)~~, and a 13-bit uniform PCM interface (UPCMI)~~.

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.

~~The UPCMI is introduced for design purposes in order to separate the speech transcoder impairments from the basic audio impairments of the UE. The UPCMI interface is also referred to as the digital audio interface (DAI).~~

~~Four~~ Five classes of acoustic interface are considered in this specification:

~~handset~~ Handset UE;
~~headset~~ Headset UE;
~~UE operated with external handsfree functionality;~~Desktop-mounted hands-free UE
Vehicle-mounted hands-free UE
~~UE operated with integrated handsfree functionality.~~Handheld hands-free UE
~~The classification of handsfree UE is for further study.~~

~~4.2 Wideband telephony~~

~~The interfaces used to define terminal acoustic characteristics for wideband telephony are for further study.~~

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 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 ~~UPCMI~~ 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 ~~(0-level) is received from the speech transcoder~~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 dBPa(A)~~A when driven by a PCM signal corresponding to the decoder output value number 1.~~

Where a volume control is provided, the measured noise shall also not exceed -54 dBPa~~A~~(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.

5.5 Sidetone characteristics (handset and headset UE)

5.5.1 Sidetone loss

The talker sidetone masking rating (STMR) shall be 18 ±5 dB.

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

~~The requirements for listener sidetone (LSTR) and talker sidetone (STMR) are for further study.~~

~~5.5.2 Sidetone distortion~~

~~The requirements for sidetone distortion are for further study.~~

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)

~~5.8 Out-of-band signals~~

~~5.8.1 Discrimination against out-of-band input signals~~

~~5.8.1.1 Handset and headset UE~~

~~When out-of-band signals are applied at the MRP, a range of frequencies will be transmitted to the UPCMI. For these signals, the following requirements shall apply.~~

~~With any sine-wave signal above 4.6 kHz and up to 8 kHz applied at the MRP at a level of -4,7 dBPa, the level of any image frequency produced at the digital interface shall be below a reference level obtained at 1 kHz (-4,7 dBPa at the MRP) by at least the amount (in dB) specified in table 9.~~

~~Table 9: Discrimination levels~~

~~Applied sine-wave frequency~~	~~Limit (minimum) (see note)~~
~~4,6 kHz~~	~~30 dB~~
~~8 kHz~~	~~40 dB~~

~~NOTE: 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.~~

~~5.8.1.2 External handsfree UE~~

~~When out-of-band signals are applied at the MRP, a range of frequencies will be transmitted to the SS and input to the speech encoder. For the signals at the output of the speech encoder, the following requirements shall apply.~~

~~With a white Gaussian noise signal bandlimited to 4,6 kHz up to 8 kHz applied at the MRP at a level of -4,7 dBPa, the total power in the frequency band 300 Hz to 3,4 kHz measured after decoding the output of the speech encoder shall be below the reference level by at least 40 dB. This reference level is obtained by applying an ITU-T P.50 artificial speech signal bandlimited to 300 Hz and 3,4 kHz at a level of -4,7 dBPa at the MRP and measuring the average level of the signal at the speech encoder output after decoding it.~~

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

~~5.8.1.3Integrated handsfree UE~~

~~For further study.~~

~~5.8.2 Spurious out-of-band receiving signals~~

~~5.8.2.1 Handset and headset UE~~

~~The level of out of band signals at the ERP shall meet the following requirements when the relevant input signals are simulated at the UPCMI.~~

~~With a digitally-simulated sine-wave signal in the frequency range of 300 Hz to 3,4 Hz and at a level of 0 dBm0 applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4,6 to 8 kHz measured selectively at the ERP shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in table 10.~~

~~Table 10: Discrimination levels~~

~~Image Signal frequency~~	~~Equivalent Input Signal Level (see note)~~
~~4,6 kHz~~	~~-35 dBm0~~
~~8 kHz~~	~~-45 dBm0~~

~~NOTE: 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.~~

~~5.8.2.2 Handsfree UE~~

~~The level of out-of-band signals at the output of the head and torso simulator (HATS) shall meet the following requirements when the relevant input signals are applied in the receive direction.~~

~~With an ITU-T P.50 artificial speech signal in the frequency range of 300 Hz to 3,4 Hz and at a level of -12 dBm0 applied in the receive direction, the level of spurious out-of-band image signals in the frequency range of 4,6 to 8 kHz measured at the ERP shall be below the reference level by at least 45 dB. This reference level is obtained by measuring the in-band acoustic reference level produced by the same input signal.~~

~~Compliance 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: 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 3 of ITU-T Recommendation G.223) shall be above the limits given in table 5 unless the sound pressure at MRP exceeds +10 dBPa.

Table 5: Limits for signal-to-total distortion ratio

Sending level dB relative to ARL	Sending Ratio (dB)
-35	17,5
-30	22,5
-20	30,7
-10	33,3
0	33,7
+7	31,7
+10	25,5

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.

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 CCITT Recommendation G.223) shall be above the limits given in table 7 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 7: Limits for signal-to-total distortion ratio

Receiving level at the digital interface (dBm0)	Receiving Ratio (dB)
-45	17,5
-40	22,5
-30	30,5
-20	33,0
-10	33,5
-3	31,2
0	25,5

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.

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 0 dB. 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 TS 26.132.

Hands-free UE (all categories):

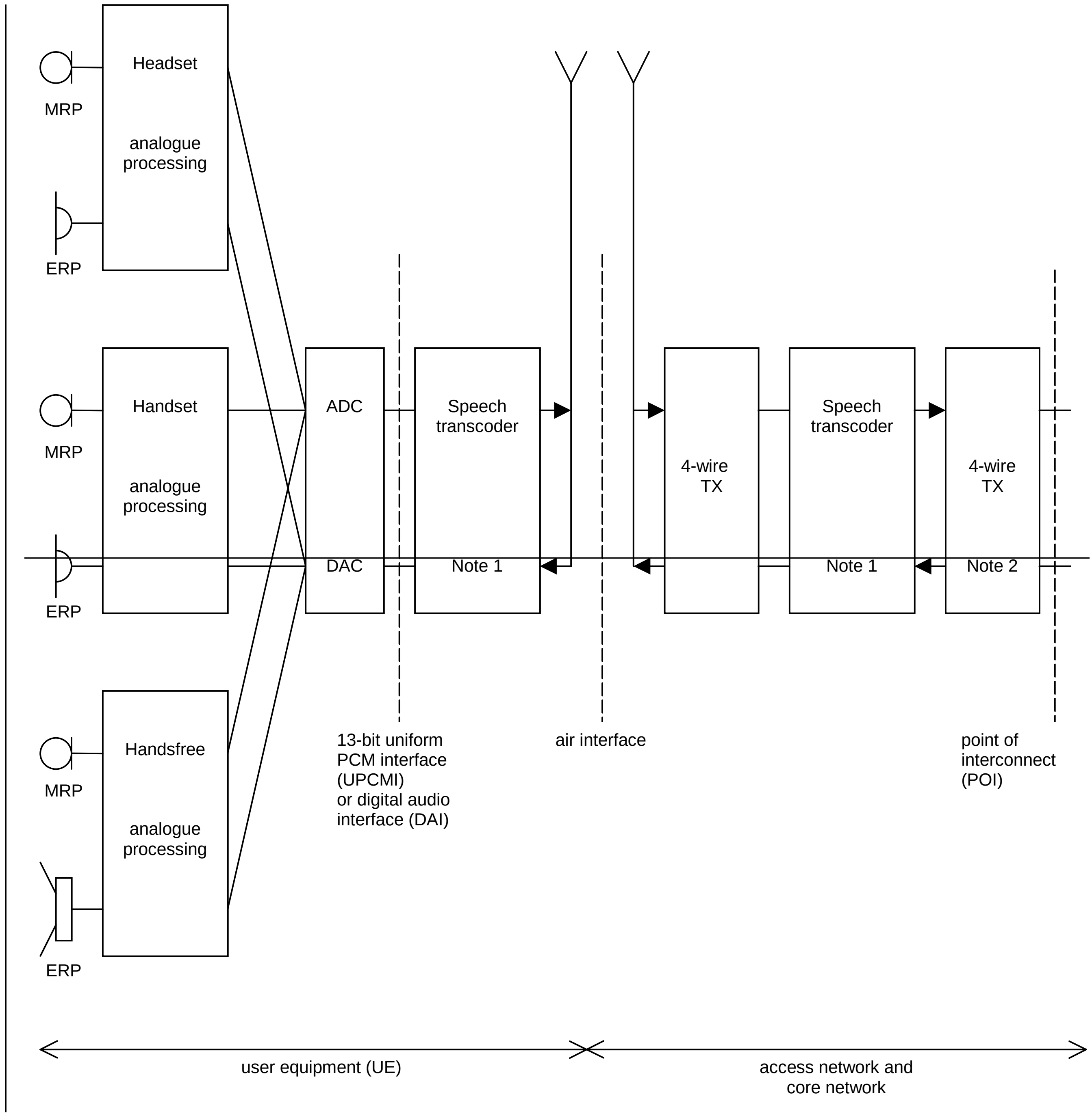
For further study.

~~6Wideband 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.~~

~~Performance requirements for the acoustic characteristics of 3G terminals supporting wideband telephony are for further study.~~


Headset
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Speech
transcoder
Note 1
4-wire
TX
Speech
transcoder
Note 1
4-wire
TX
Note 2
13-bit uniform
PCM interface
(UPCMI)
or digital audio
interface (DAI)
air interface
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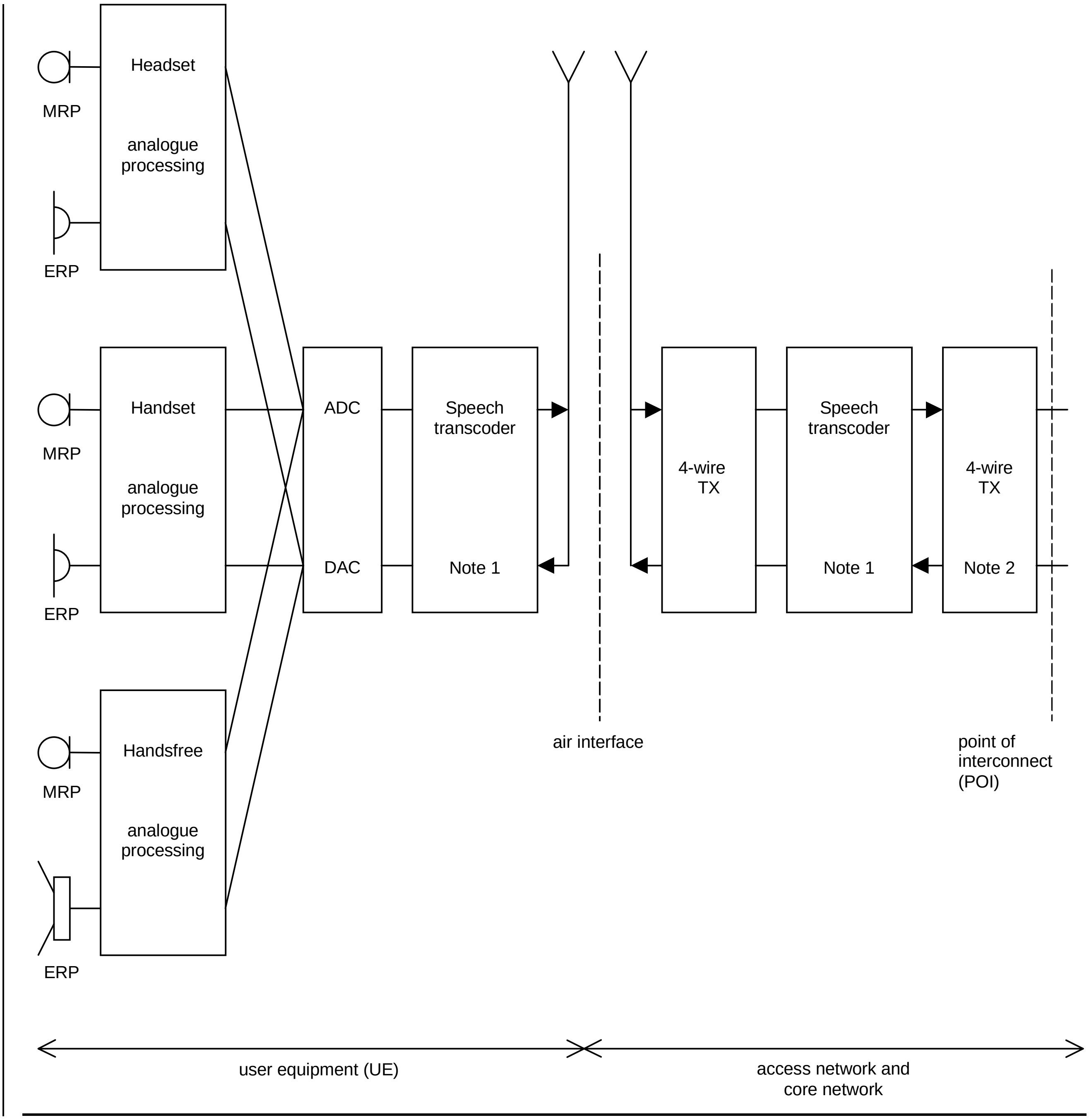

NOTE 1: Includes DTX functionality.
NOTE 2: Connection to PSTN should include electrical echo control (EEC).

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

3GPP TSG-SA4 Meeting #16 ***Tdoc S4-010271***
S4#16, Sophia Antipolis, France, Feb. 26-March 2, 2001

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CHANGE REQUEST

⌘ **26.131**	**CR** **006**	⌘	rev **3**	⌘	Current version: **3.1.0**	⌘

For ***HELP*** *on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

Proposed change affects: ⌘	(U)SIM [ ]	ME/UE [X]	Radio Access Network [ ]	Core Network [ ]

Title:	⌘	Wideband acoustic requirements, inclusion to TS 26.131		
Source:	⌘	TSG-SA WG4		
Work item code:	⌘	TEI (WB)	***Date:*** ⌘	2001-2-21
Category:	⌘	**B**	***Release:*** ⌘	REL-4
		Use *<u>one</u>* *of the following categories:* ***F*** *(essential correction)* ***A*** *(corresponds to a correction in an earlier release)* ***B*** *(Addition of feature),* ***C*** *(Functional modification of feature)* ***D*** *(Editorial modification)* Detailed explanations of the above categories can be found in 3GPP TR 21.900.		*Use* *<u>one</u>* *of the following releases:* *2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *REL-4 (Release 4)* *REL-5 (Release 5)*

Reason for change:	⌘	To include specify acoustic requirements for the frequency range over which wideband telephony operates.
Summary of change:	⌘	Requirements for wideband telephony transmission performance are proposed. The proposed requirements are generally taken from ITU-T Recommendation P. 311 and P. 341. For compatibility with previously accepted narrow-band requirements, there are some differences from the ITU recommendations and the requirements proposed in this document.
Consequences if not approved:	⌘	There will be no requirements for wideband telephony terminals

Clauses affected:	⌘	4, 6		
Other specs affected:	⌘	[ ] Other core specifications [ ] Test specifications [ ] O&M Specifications	⌘	
Other comments:	⌘			

4 Interfaces

~~4.1 Narrow-band telephony~~

The interfaces required to define terminal acoustic characteristics for narrow-band telephony are shown in figure 1. These are the air interface, the point of interconnect (POI), and a 13-bit uniform PCM interface (UPCMI).
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.
The UPCMI is introduced for design purposes in order to separate the speech transcoder impairments from the basic audio impairments of the UE. The UPCMI interface is also referred to as the digital audio interface (DAI).
Four classes of acoustic interface are considered in this specification:

- handset UE;
- headset UE;
- UE operated with external handsfree functionality;
- UE operated with integrated handsfree functionality.

The classification of handsfree UE is for further study.

~~4.2 Wideband telephony~~

~~The interfaces used to define terminal acoustic characteristics for wideband telephony are for further study.~~

~~6~~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.

~~Performance requirements for the acoustic characteristics of 3G terminals supporting wideband telephony are for further study.~~

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 = 5 +/- 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) -10 dB.

The loudness rating is 3dB higher (quieter) than for narrow-band telephony, due to the increased loudness of the wider bandwidth.

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 = 5 +/- 4 dB.

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

The loudness rating is 3dB high (quieter) than for narrow-band telephony, due to the increased loudness of the wider bandwidthWhere 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.

6.2.4 Connections with handheld hands-free UE

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

SLR = 13 +/- 4 dB;

RLR = 9 +12 / - 4 dB.

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

The loudness rating is 3dB higher (quieter) than for narrow-band telephony, due to the increased loudness of the wider bandwidth

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

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 = 5 +/- 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.

The loudness rating is 3dB high (quieter) than for narrow-band telephony, due to the increased loudness of the wider bandwidth

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

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.

6.4 Sensitivity/frequency characteristics

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 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
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.

6.4.2 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.

Table 2: Receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
70	-10	-
200	2	
300	2	-9
500	2	(see note 2)
1 000	2	-7
3 000	2	(see note 2)
3 400	2	(see note 2)
6300	2	-14
8000	2	-

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

Table 3: Desktop and Vehicle-mounted hands-free sending sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
100	-12	
200	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
6 300	4	-8
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.

6.4.4 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	
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.

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

Table 5: Handheld hands-free sending sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
100	-12	
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
6 300	4	-8
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 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
6 300	0	-12
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.

6.5 Sidetone characteristics (handset and headset UE)

6.5.1 Sidetone loss

The talker sidetone masking rating (STMR) shall be 18dB ±5. 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.
Handset 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 40 dB at the nominal setting of the volume control in quiet background conditions and 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 40 dB at the nominal setting of the volume control in quiet background conditions and 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 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.

6.7.5 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.

6.8 Distortion

6.8.1 Sending Distortion

The sending part shall meet the following distortion requirements:
NOTE: 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 3 of ITU-T Recommendation G.223) shall be above the limits given in table 5 unless the sound pressure at MRP exceeds +10 dBPa.

Table 5: Limits for signal-to-total distortion ratio

Sending level dB relative to ARL	Sending Ratio (dB)
-35	17,5
-30	22,5
-20	30,7
-10	33,3
0	33,7
+7	31,7
+10	25,5

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.

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:
The ratio of signal-to-total distortion power measured with the proper noise weighting (see table 4 of CCITT Recommendation G.223) shall be above the limits given in table 7 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 7: Limits for signal-to-total distortion ratio

Receiving level at the digital interface (dBm0)	Receiving Ratio (dB)
-45	17,5
-40	22,5
-30	30,5
-20	33,0
-10	33,5
-3	31,2
0	25,5

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.

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

Hands-free UE (all categories):

For further study.