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Requirements
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Contents

Foreword

This Technical Specification has been produced by the 3GPP.

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

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Introduction

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

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

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

1 Scope

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

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".

3 Definitions, symbols and abbreviations

3.1 Definitions

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

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

3.2 Abbreviations

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

ADC	Analogue to Digital Converter
DAC	Digital to Analogue Converter
DAI	Digital Audio Interface
DTX	Discontinuous Transmission
EEC	Electrical Echo Control
EL	Echo Loss
ERP	Ear Reference Point
HATS	Head and Torso Simulator
LSTR	Listener Sidetone Rating
LRGP	Loudness Rating Guardring Position
MRP	Mouth Reference Point
OLR	Overall Loudness Rating
PCM	Pulse Code Modulation
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

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.

5 Narrow-band telephony transmission performance

5.1 Applicability

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

5.2 Overall loss/loudness ratings

5.2.1 General

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

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

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

5.2.2 Connections with handset UE

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

SLR = 8 +/- 3 dB;

RLR = 2 +/- 3 dB.

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

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

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

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

5.2.3 Connections with external handsfree UE

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

SLR = 13 +/- 4 dB;

RLR = 2 +/- 4 dB.

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

Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control. It is recommended that a volume control giving at least 15 dB increase from the nominal RLR (louder) is provided for handsfree units intended to work in the vehicle environment. This is to allow for the increased noise volume in a moving vehicle.

5.2.4 Connections with integrated handsfree UE

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.

5.3 Idle channel noise (handset and headset UE)

5.3.1 Sending

The maximum noise level produced by the apparatus at the UPCMI 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 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 dBA 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 dBA 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.

5.4 Sensitivity/frequency characteristics

5.4.1 Handset and headset UE sending

The sensitivity/frequency characteristics shall be as follows:

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

Table 1: Sending sensitivity/frequency mask

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

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

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

5.4.2 Handset and headset UE receiving

The sensitivity/frequency characteristics shall be as follows:

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

Table 2: Receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	[-12]	[-]
200	[0]	[-]
300	[2]	[-7]
500	(see note 2)	[-5]
1 000	[0]	[-5]
3 000	[2]	[-5]
3 400	[2]	[-10]
4 000	[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.

5.4.3 External handsfree 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: Handsfree sending sensitivity/frequency response

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

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

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

5.4.4 External handsfree 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: Handsfree receiving sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
200	0	
250	0	
315	0	-15
400	0	-12
500	0	-12
630	0	-12
800	0	-12
1 000	0	-12
1 300	0	-12
1 600	0	-12
2 000	0	-12
2 500	0	-12
3 100	0	-12
4 000	0	

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

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

5.4.5 Integrated handsfree UE sending

For further study.

5.4.6 Integrated handsfree UE receiving

For further study.

5.5 Sidetone characteristics (handset and headset UE)

5.5.1 Sidetone loss

A sidetone requirement is appropriate for UE using handsets and headsets. There are separate requirements for listener sidetone (LSTR) and talker sidetone (STMR). The listener sidetone performance is considered as the major parameter affecting the user perception of the system, although talker sidetone is important to give the user some comfort in using the equipment.

The value of the listener sidetone rating (LSTR) shall not be less than 15 dB. Where a user-controlled receiving volume control is provided, the LSTR shall meet the requirement given above at the setting where the RLR is equal to the nominal value.

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

The nominal value of the sidetone masking rating (STMR) shall be 13 dB +/- 5 dB. Where a user-controlled receiving volume control is provided, the STMR shall meet the requirement given above at the setting where the RLR is equal to the nominal value.

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

It is recommended that the STMR is independent of the volume control.

5.5.2 Sidetone distortion

The third harmonic distortion generated by the terminal equipment shall not be greater than 10 %.

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 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.

Handsfree UE: no requirement other than echo loss.

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

5.7 Acoustic echo control

5.7.1 General

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

The use of acoustic echo control is not mandated for 3G networks and the connection between the UE and the POI is zero loss. Therefore the acoustic echo control provided in UE should provide a TCLw of at least 46 dB at the POI over the likely range of acoustic end delays.

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

5.7.2 Acoustic echo control in an external handsfree 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 handsfree UE, this reverberation and dispersion may be time variant. Compliance with this requirement shall be checked by the relevant test described in TS 26.132.

5.7.3 Acoustic echo control in an integrated handsfree UE

For further study.

5.7.4 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.

5.7.5 Acoustic echo control in a headset UE

The TCLw for a headset UE shall be 46 dB. Due to the obstacle effect of the head in this type of terminal, careful design might mean that no active echo control is necessary.

5.8 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.

5.8.1.3 Integrated 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.

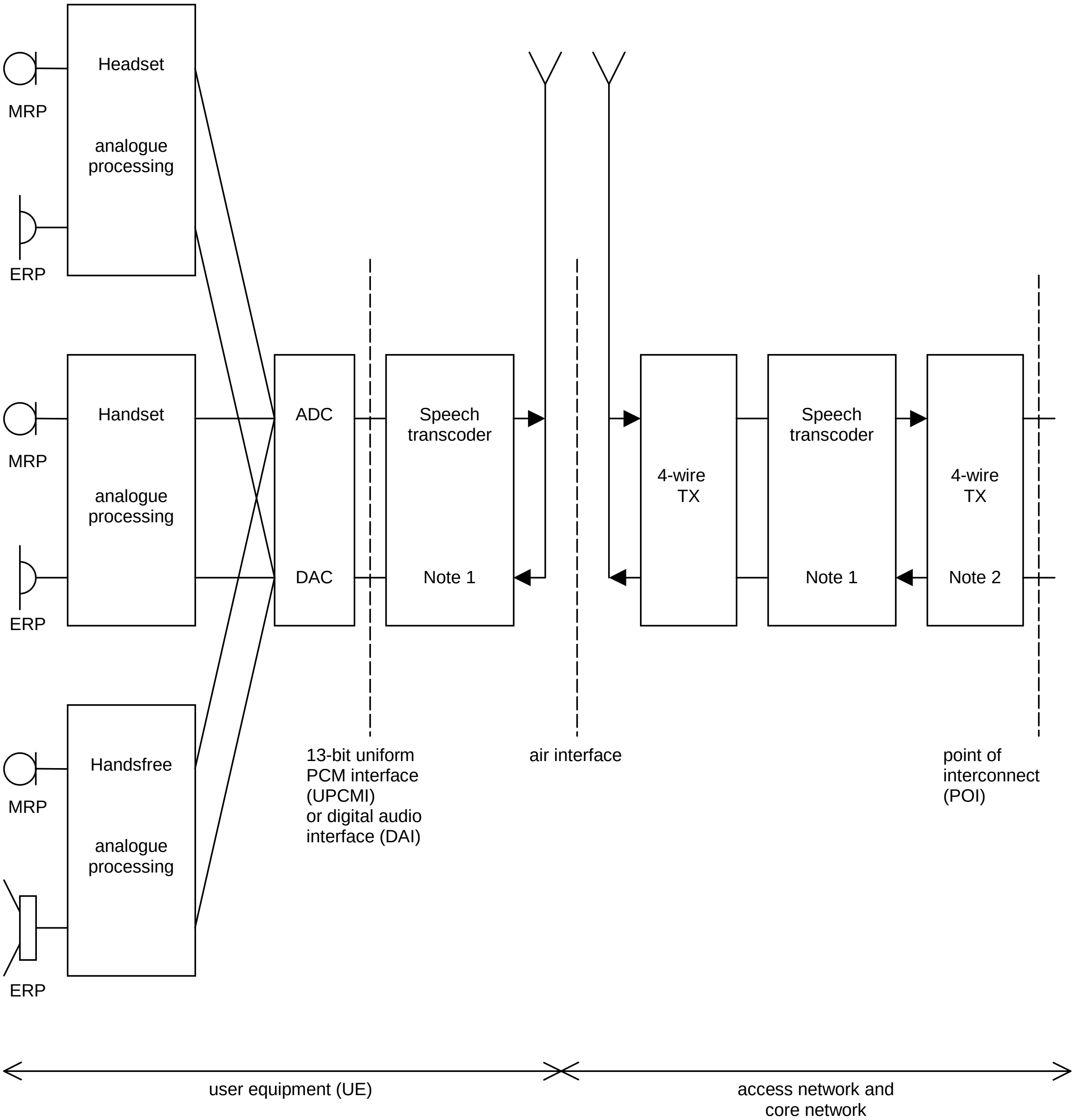
Compliance shall be checked by the relevant test described in TS 26.132.

5.9 Ambient Noise Rejection

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, 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.



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

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.

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

History

Document history		
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