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Technical Specification

3rd Generation Partnership Project;
TSG-SA Codec Working Group;
Terminal Acoustic Characteristics for Telephony – Test
(3G TS 26.132 version 0.0.3)

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Contents

Foreword

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Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

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3 Indicates TSG approved document under change control.

y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the specification;

Introduction

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

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 in TS26.131; the test methods and considerations are specified in the main body of the text.

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

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
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

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. The test methods needed to assess the minimum performance requirements for wideband telephony are for further study.

5 Test configurations

This section describes the test setups for terminals, networks and their various combinations. Since the document describes the generals aspects of end to end speech quality testing, specific test setups and configuration description are made only in general. In case any specific description of terminal or network setups is needed (e.g. buffer sizes, type of codecs, packet loss simulations) these descriptions need to be found in the relevant standards of such transmission systems.

5.1 Test setup for terminals

The general access to terminals is described in Fig. 3. The preferred acoustical access to terminal is the most realistic simulation of the “average” subscriber. This can be made by using HATS (head and torso simulator) with appropriate ear simulation and appropriate means to fix handset, headset or hands-free terminals in a realistic by reproducible way to the HATS. HATS is decribed in ITU-T Recommendation P.58 appropriate ears are described in ITU-T Recommendation P.57 (type 3.3 and type 3.4 ear), a proper positioning of handsets in realistic conditions is found in

ITU-T Recommendation P.64 the test setups for various types of hands-free terminals can be found in ITU-T Recommendation P.581 /xx/.

The preferred way of testing is the connecting a terminal to the network is either a network simulator with exact defined settings and access points or –in case of end to end scenarios- the “typical” network the terminal is used in.. The test sequences are fed in either electrically, using a reference codec or using the direct signal processing approach or acoustically using ITU-T specified devices.

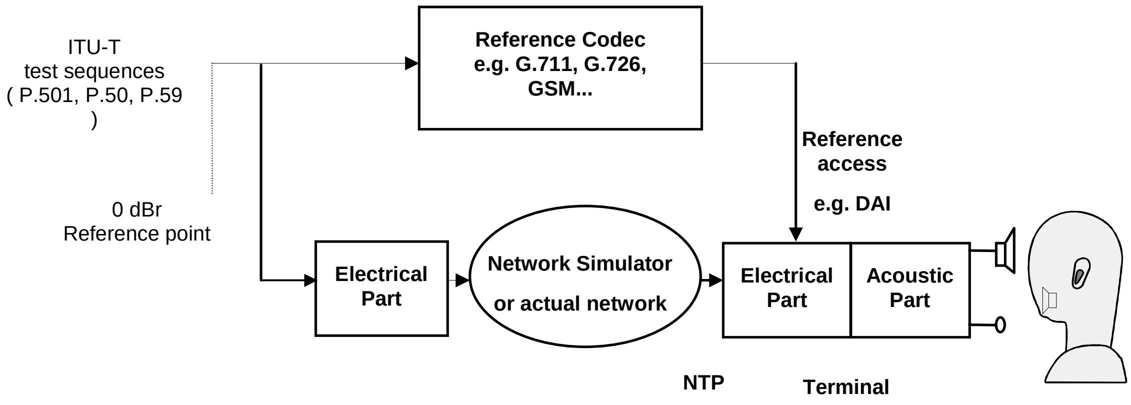


Fig. 3: Test setup for terminals, electrical access using a “reference” access or a network simulator

5.1.1 Setup for handset terminals

When using a handset telephone the handset is placed in the HATS position as described in ITU-T Recommendation P.64 /. The artificial mouth shall conform with P.58 when HATS is used. The artificial ear shall conform with Rec. P.57 type 3.3 or type 3.4 ears shall be used.

Type 1+LRGP

5.1.2 Setup for headset terminals

To be added

5.1.3 Setup for hands-free type terminals and loudspeaking terminals

General definition of hands-free terminals from P.340 All types of terminals, which cannot be fit to the LRGP-position or the HATS-position -except headsets- need to be considered as hands-free type terminals.

P.340&P.581

5.1.4 Position and calibration of HATS

All the sending and receiving characteristics shall be tested with the HATS and it shall be indicated in the test report that HATS is used, it shall be indicated what type of ear was used at what pressure force. In case of hands-free measurements the HATSHFRP(s) shall be used for the calibration(s), the reference point chosen for the HATSHFRP shall be indicated.

The horizontal positioning of the HATS reference plane shall be guaranteed within ±2°.

The HATS shall be equipped with two Type 3.3 or 3.4 Artificial Ears. For hands-free measurements the HATS shall always be equipped with two artificial pinnas. The pinnas are specified in Recommendation P.57 for Types 3.3 and 3.4 artificial ears. The pinna shall be positioned on HATS according to ITU-T Recommendation P.58 .

The exact calibration and equalization procedures as well as the combination of the two ear signals for the purpose of measurements can be found in ITU-T Recommendation P.581 /xx/. This Recommendation also describes the positioning for the various types of hands-free terminals.

5.2 Setup of the electrical interfaces

to be added

all should be mentioned: NTP and any other (soundcard)

5.3 Test signals

Due to the extense coding of the speech signals, standard test signals are not applicable for the tests, appropriate test signals (general description) are defined in ITU-T Recommendation P.50 and P.501. More information can be found in the test procedures described below.

For narrow band terminals the test signal used shall be bandfiltered between 200 Hz and 4 kHz with a bandpass filter providing a minimum of 24 dB/Oct. filter steepness, when feeding into the receiving direction.

The test signal levels are referred to the average level of the (band filtered in receiving direction) test signal, averaged over a period of 10 s. Unless specified otherwise, the averaging time for all measurements is 10 s.

6 Test conditions

6.1 Environmental conditions

to be added Note : Need to describe the test chamber

6.2 Network Simulator conditions

to be added i.e. Ideal Radio (No Errors), tests on default speech codec i.e AMR,

5 Narrow-band telephony transmission performance Test Methods

5.1 Applicability

The test methods in this sub-clause shall apply when testing a UE which 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

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.

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).

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.

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.

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.

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.

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.

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.

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.

5.5.2 Sidetone distortion

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

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.

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 . Acoustic echo control in an integrated handsfree UE is For further study.

5.7.4 Acoustic echo control in a handset UE

The TCLw for the handset UE shall be 46 dB.

5.7.5 Acoustic echo control in a headset UE

The TCLw for a headset UE shall be 46 dB.

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.

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.

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.

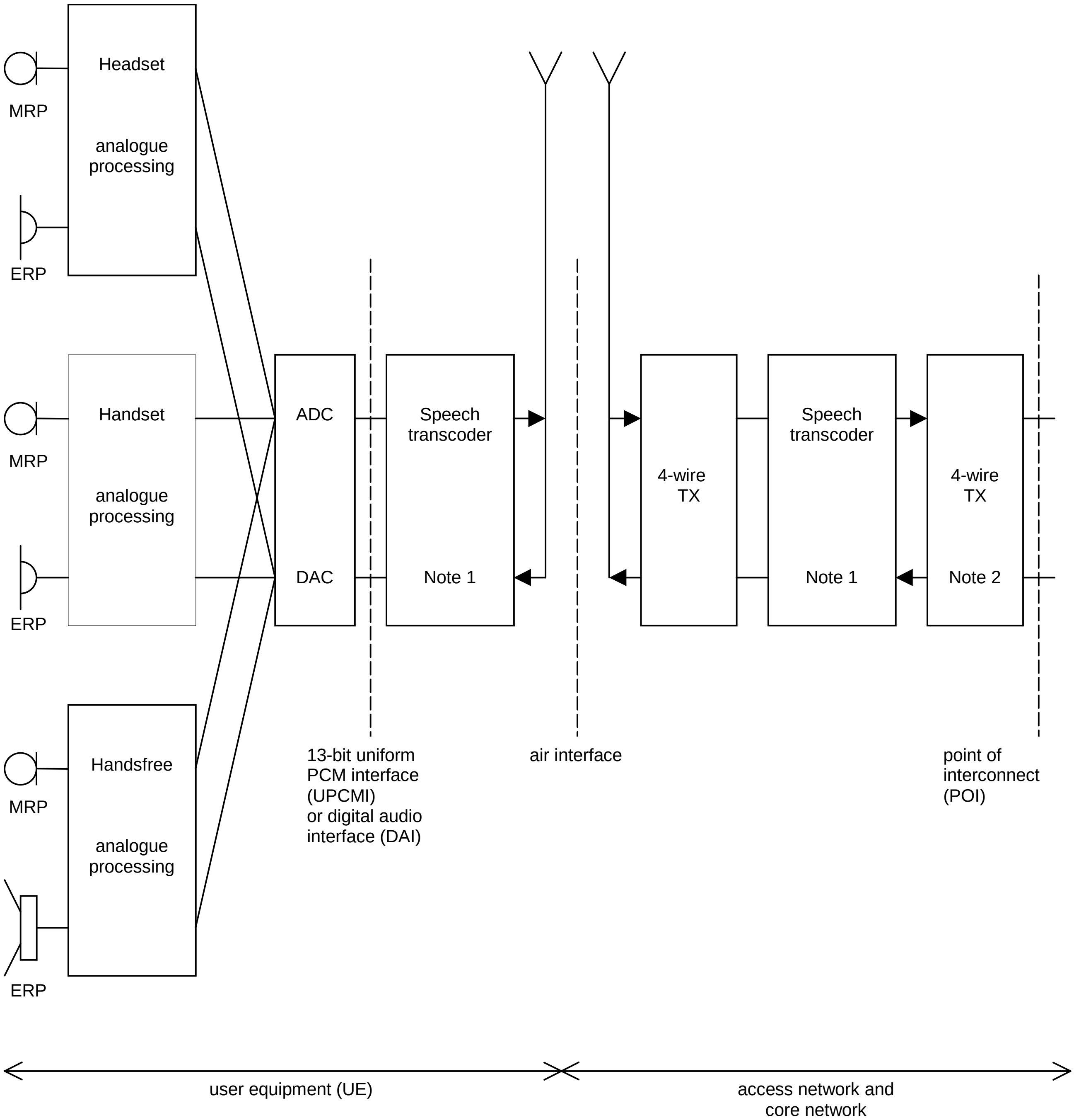
5.8.2.2 Handsfree UE

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.

5.9 Ambient Noise Rejection

The UE ambient noise rejection, calculated as a Single Figure DELSM (SFDELSM) shall be greater than or equal to 0 dB.



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		
0.0.1	January 2000	Initial draft
0.0.3	February 2000	Second draft after comments and scope change from 3GPP SA4.