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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 0.0.1)

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Contents

Foreword

This Technical Specification has been produced by the 3GPP.

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where:

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- 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

Access to terminals for acoustic testing is always made via the acoustic or air interfaces. The Air Interface is specified by the 3G 25 series specifications and is required to achieve user equipment (UE) transportability. 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. Measurements may be made at this point using a standard send and receive side, as defined in ITU-T Recommendations.

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 terminal acoustic testing

5.1 Test setup for terminals

The general access to terminals is described in Fig. 3. The preferred acoustic access to 3G terminals is the most realistic simulation of the “average” subscriber. This can shall be made by using HATS (head and torso simulator) with appropriate ear simulation and appropriate mountings for handset and headset terminals as well as arrangements for hands-free terminals in a realistic but reproducible, way to the HATS. HATS is described 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.

The preferred way of testing is the connection of a terminal to the system simulator with exact defined settings and access points. The test sequences are fed ineither, electrically using a reference codec or using hedirect signal processing approach or acoustically using ITU-T specified devices.

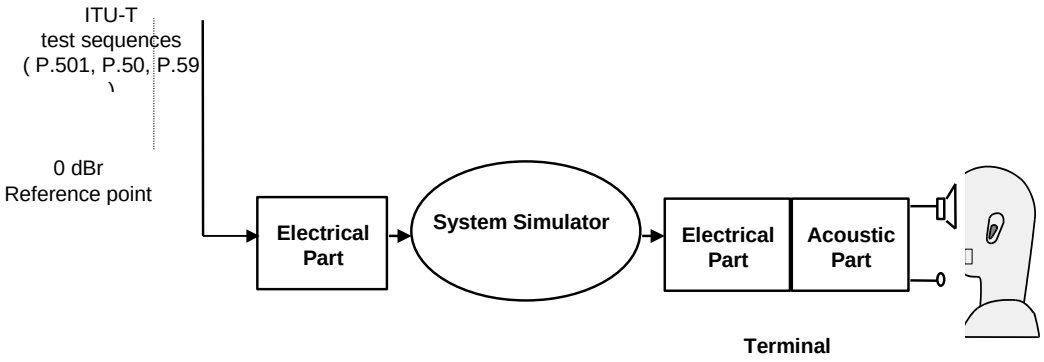

Fig. 1: Test setup for terminals, electrical access using a system simulator

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.

5.1.2 Setup for headset terminals

To be added

5.1.3 Setup for hands-free terminals

General definition of hands-free terminals and setup for desktop hands-free terminals can be found in ITU-T Recommendation P.340. Setup recommendations for other types of hands-free telephones can be found in ITU-T Recommendation P. 581. All types of terminals, which cannot be fixed the HATS-position -except headsets- need to be considered as hands-free type terminals.

5.1.4 Position and calibration of HATS

All the sending and receiving characteristics shall be tested with theThe horizontal positioning of the HATS reference plane shall be guaranteed within $\pm 2°$.

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.

5.2 Setup of the electrical interfaces

5.2.1 Codec approach and specification

Codec approach: In this approach, a codec is used to convert the companded digital input/output bit-stream of the system simulator to the equivalent analogue values. With this approach a system simulator, simulating the radio link to the terminal under controlled and error free conditions is required. The system simulator has to be equipped with a high-quality codec whose characteristics are as close as possible to ideal. For the purposes of 3G acoustic testing, the system simulator shall use the default speech codec, the AMR speech codec as defined in TS26.XXX. The transcoding from the output of the AMR speech coding in the system simulator to analogue signals shall be carried out using an ITU-T G.711 codec performing to ITU-T G.712 (4-wire analogue).

Definition of 0 dBr point:

D/A converter - a Digital Test Sequence (DTS) representing the codec equivalent of an analogue sinusoidal signal whose rms value is 3.14 dB below the maximum full-load capacity of the codec shall generate 0 dBm across a 600 ohm load;

A/D converter - a 0 dBm signal generated from a 600 ohm source shall give the digital test sequence (DTS) representing the codec equivalent of an analogue sinusoidal signal whose RMS value is 3.14 dB below the maximum full-load capacity of the codec.

5.2.2 Direct digital processing approach

In this approach, the companded digital input/output bit-stream of the terminal connected through the radio link to the system simulator is operated upon directly.

5.3 Accuracy of test equipment

Unless specified otherwise, the accuracy of measurements made by test equipment shall be better than:

Item	Accuracy
Electrical Signal Power	±0,2 dB for levels ≥ -50 dBm
Electrical Signal Power	±0,4 dB for levels < -50 dBm
Sound pressure	±0,7 dB
Time	±5 %
Frequency	±0,2 %

Unless specified otherwise, the accuracy of the signals generated by the test equipment shall be better than:

Quantity	Accuracy
Sound pressure level at MRP	±1 dB for 200 Hz to 4 kHz ±3 dB for 100 Hz to 200 Hz and 4 kHz to 8 kHz
Electrical excitation levels	±0,4 dB (see note 1)
Frequency generation	±2 % (see note 2)
NOTE 1: Across the whole frequency range. NOTE 2: When measuring sampled systems, it is advisable to avoid measuring at sub-multiples of the sampling frequency. There is a tolerance of ±2 % on the generated frequencies, which may be used to avoid this problem, except for 4 kHz where only the -2 % tolerance may be used.	

The measurements results shall be corrected for the measured deviations from the nominal level.

The sound level measurement equipment shall conform to IEC 651 Type 1.

5.4 Test signals

Due to the coding of the speech signals, standard sinusoidal test signals are not applicable for 3G acoustic 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 band limited between 200 Hz and 4 kHz with a bandpass filter providing a minimum of 24 dB/Oct. filter roll off, when feeding into the receiving direction.

The test signal levels are referred to the average level of the (band limited in receiving direction) test signal, averaged over the complete test sequence . unless specified otherwise.

6 Test conditions

6.1 Environmental conditions

6.1.1 Handset and headset terminals

For handset and headset measurements the test room shall be practically free-field down to a lowest frequency of 275 Hz, the handset or the headset including the HATS shall lie totally within this free-field volume. This shall be met if deviations of the ideal free-field conditions are less than +/- 1 dB.

The ambient noise level shall be less than -30 dBPa(A).

Echo measurements shall be conducted in realistic rooms with an ambient noise level less then -64 dBPa(A).

6.1.2 Hands-free terminals

Hans-free terminals generally should be tested in their typical environment of application. Care must be taken, that e.g. noise levels are sufficiently low in order not to interfere with the measurements.

For office type hands-free terminals the appropriate requirements shall be taken from ITU-Recommendation P.340.

The broadband noise level shall not exceed –70 dBPa(A). The octave band noise level shall not exceed the values specified in Table 2:

TABLE 2/P.340

Noise level

Centre frequency (Hz)	Octave band pressure level (dBPa)
63	–45
125	–60
250	–65
500	–65
1 k	–65
2 k	–65
4 k	–65
8 k	–65

Echo measurements shall be conducted in realistic rooms with an ambient noise level less then -70 dBPa(A).

6.2 System Simulator conditions

to be added i.e. Ideal Radio (No Errors), tests on default speech codec i.e AMR, which modes ? 12.2kbit/s ?,

7 Narrow-band telephony transmission performance Test Methods

7.1 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.

7.2 Overall loss/loudness ratings

7.2.1 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.

7.2.2 Connections with handset UE

7.2.2.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be –4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is setup as described in subclause 5. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The pressure force used to apply the handset against the artificial ear is noted in the test report.

   The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula 2.1, over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79, table 1.

7.2.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The test signal level shall be –16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is setup as described in subclause 5. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The pressure force used to apply the handset against the artificial ear is noted in the test report. The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [18], formula 2.1, over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79 [18].

d) No leakage correction shall be applied when using HATS for the measurement.

7.2.3 Connections with external handsfree UE

7.2.3.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be –4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to –28.7 dBPa at the HFRP and the spectrum is not altered.

   The spectrum at the MRP and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5. The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 , bands 4 to 17. For the calculation the averaged measured level at the electrical reference point foreach frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula 2.1, over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79, table 1.

7.2.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The test signal level shall be –16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is setup as described in subclause 5. The HATS is freefield equalized as described in ITU-T Recommendation P.581. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the “right ” and “left” signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 , bands 4 to 17.

   For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [18], formula 2.1, over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79.

d) No leakage correction shall be applied. The hands-free correction as described in P.340 shall be applied. To compute Receiving loudness rating (RLR) for handsfree terminal (see also ITU-T Recommendation P.340), when using the combination of left and right ear signals from HATS the HFL_E has to be 8 dB, instead of 14 dB. For further information see ITU-T Recommendation P.581.

7.2.4 Connections with integrated handsfree UE

! Why should not the same requirements apply (also in TS 26.131 as for external handsfree?).

7.2.5 Connections with headset UE

! to be added: ,text similar to handset UE but positioning not yet clear, see STQ work item.

7.3 Idle channel noise (handset and headset UE)

7.3.1 Sending

To be added.7.3.2 Receiving

To be added.7.4 Sensitivity/frequency characteristics

7.4.1 Handset and headset UE sending

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test

signal level shall be –4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is setup as described in subclause 5. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The pressure force used to apply the handset against the artificial ear is noted in the test report.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [17] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa.

7.4.2 Handset and headset UE receiving

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The test signal level shall be –16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is setup as described in subclause 5. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The pressure force used to apply the handset against the artificial ear is noted in the test report.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [17] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V.

7.4.3 External hands-free UE sending

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501.

The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be –4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to –28.7 dBPa at the HFRP and the spectrum is not altered.

The spectrum at the MRP and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is setup as described in subclause 5. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [17] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBV/Pa.

7.4.4 External hands-free UE receiving

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The test signal level shall be –16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is setup as described in subclause 5. The HATS is freefield equalized as described in ITU-T Recommendation P.581 [xx]. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the “right ” and “left” signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [17] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V.

7.4.5 Integrated hands-free UE sending

For further study. See note for loudness rating: Why different to external hands-free?

7.4.6 Integrated hands-free UE receiving

For further study. See note for loudness rating: Why different to external hands-free?

Sidetone characteristics (handset and headset UE)

!!! Note: Neither LSTR nor STMR can be measured realistically or easy when using modern “leaky” designed handsets or headsets and ITU-T type 3..x ears. Instead of these values the D-value should be measured!! This needs changes in TS 26.131 as well..

7.5 Measurement and calculation of the value of the D-factor

7.5.1 Connections with handset and headset UE

a) Sound field calibration: The diffuse sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within ±3 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands according to IEC 225 [19] from 100 Hz to 8 kHz (bands 1 to 20).

NOTE 1: The pressure intensity index, as defined in ISO 9614, may prove to be a suitable method for assessing the diffuse field.

NOTE 2: Where more than one loudspeaker is used to produce the desired sound field, the loudspeakers may require to be fed with non-coherent electrical signals to eliminate standing waves and other interference effects..

b) Where adaptive techniques or voice switching circuits are not used (need to be declared by the supplier of the UE) the spectrum shall be band limited (50 Hz to 10 kHz) "pink noise" (see ITU-T Recommendation P.64 [xx], annex B) to within ±3 dB and the level shall be adjusted to 70 dB(A) (-24 dBPa(A)). The tolerance for this level is ±1 dB.

In other cases the level shall be adjusted to 50 dB(A) (-44 dBPa(A)). The tolerance for this level is ±1 dB.

c) The handset or the headset UE is mounted as described in subclause 5. Measurements are made on one-third octave bands according to IEC 225 [19] for the 14 bands centred at 200 Hz to 4 kHz

(bands 4 to 17). For each band the diffuse sound sensitivity S_{si}(diff) is measured. The sensitivity shall be expressed in terms of dBV/Pa.

d) The direct sound sensitivity shall be measured using the test set-up specified in subclause 5.1 and a spech like test signal as defined in ITU-T Recommendation P.50 or P.501. The type of test signal used shall be stated in the test report. The direct sound sensitivity is measured in one-third octave bands according to IEC 225 [19] for the 14 bands centred at 200 Hz to 4 kHz (bands 4 to 17). For each band the direct sound sensitivity S_{si}(direct) is measured. The sensitivity shall be expressed in terms of dBV/Pa.

e) The value of the D-factor shall be calculated according to ITU-T Recommendation P.79 [18], annex E, formulas E2 and E3, over the bands from 4 to 17, using the coefficients K_i from table E1 of ITU-T Recommendation P.79 [xx].

7.5.1 Sidetone loss

7.5.2 Sidetone distortion

The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The pressure force used to apply the handset against the artificial ear is noted in the test report.

(An instrument capable of measuring the third harmonic distortion of signals with fundamental frequencies in the range of 315 Hz to 1 kHz shall be connected to the artificial ear.

A pure-tone signal of -4,7 dBPa shall be applied at the MRP at frequencies of 315 Hz, 500 Hz, and 1 kHz. For each frequency, the third harmonic distortion shall be measured in the artificial ear.)

!!!!Note: It is very unlikely, that a pure-tone signal is suitable to determine the sidetone distortion. Therefore it is proposed, to use a test-signal which is more “speech-like” e.g. an AM-FM modulated sinewave composed signal having a fundamental frequency similar to speech and the typical speech like harmonics, limit the spectrum to 1 kHz and measure the total distortion in the frequency range from 1 kHz to 3,4 kHz.

The test signal is defined as:

$$s(t) = \sum_i [[A + \mu_{AM} \cos(2\pi t n * f_{AM})] * \cos(2\pi t * f_{0i}) + \mu_{FM} * \sin(2\pi t * f_{FM})]$$

with A = 0,5

f_{AM} = 3 Hz, μ_{AM} = 0.5

f_{FM} = 5 Hz, μ_{FM} =1

$F_{0\,i}$= i*240 Hz ;i=1..4

The spectrum should be shaped using a shaping filter as described in Table3/P.501 which provides a slope of 5 dB/oct.

The test signal level is adjusted to –4,7 dBPa at the MRP.

The total energy of the distortion components is measured in the frequency range from 1 kHz to 4 kHz.

Text to be changed in TS 26.131:

The level of the distortion generated by the terminal shall be 20 dB below the level of the test signal.

7.6 Stability loss

to be added

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).

7.7 Acoustic echo control

7.7.1 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.

7.7.2 Acoustic echo control in an external handsfree UE

!!!Note. It is not sufficient for hands-free UE to measure TCL only. Convergence time, background noise behaviour and prameters like double talke performance should be measured as well. G.168 gives good guidance on such parameters and the measurements and can be adapted to the 3GPP standard easily using the signals and analysis methods in P.501 and P.502.

TCLw:

The hands-free is setup in a room where it is intended to be used, eg. for an office type hands-free UE a typical “office-type” room should be used. The ambient noise level shall be less than -70 dBPa(A). The attenuation from reference point input to reference point output shall be measured using a speech like test signal .

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is altered.

The test signal is defined as:

$$s(t) = \sum_i [[A + \mu_{AM} \cos(2\pi tn * f_{AM})] * \cos(2\pi t * f_{0i}) + \mu_{FM} * \sin(2\pi t * f_{FM})]$$

with A = 0,5

f_{AM} = 3 Hz, μ_{AM} = 0.5

f_{FM} = 5 Hz, μ_{FM} =1

$F_{0\,i}$= i*240 Hz ;i=1..14

The spectrum should be shaped using a shaping filter as described in Table3/P.501 which provides a slope of 5 dB/oct. In case of insufficient signal to noise ratio the filter may be switched off in order to provide more signal energy to the higher frequency range.

The training sequence level shall be –16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to CCITT Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Note:

The calculation shall be conducted only for frequencies where a signal is present.

7.7.4 Acoustic echo control in a handset UE

A more realistic setup may be to apply the handset to the HATS as defined in subclause 5.1.

The handset is suspended in free air in such a way that the inherent mechanical coupling of the handset is not effected. The testing shall be made under real use conditions, a typical “office-type” room should be used. The ambient noise level shall be less than -64 dBPa(A). The attenuation from reference point input to reference point output shall be measured using the speech like test signal defined below.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is altered.

The test signal is defined as:

$$s(t) = \sum_i [[A + \mu_{AM} \cos(2\pi tn * f_{AM})] * \cos(2\pi t * f_{0i}) + \mu_{FM} * \sin(2\pi t * f_{FM})]$$

with A = 0,5

f_{AM} = 3 Hz, μ_{AM} = 0.5

f_{FM} = 5 Hz, μ_{FM} =1

$F_{0\,i}$= i*240 Hz ;i=1..14

The spectrum should be shaped using a shaping filter as described in Table3/P.501 which provides a slope of 5 dB/oct. In case of insufficient signal to noise ratio the filter may be switched off in order to provide more signal energy to the higher frequency range.

The training sequence level shall be –16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to CCITT Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Note:

The calculation shall be conducted only for frequencies where a signal is present.

7.7.5 Acoustic echo control in a headset UE

A more realistic setup may be to apply the headset to the HATS as defined in subclause 5.1.

The headset is suspended in free air in such a way that the inherent mechanical coupling of the handset is not effected. The testing shall be made under real use conditions, a typical “office-type” room should

be used. The ambient noise level shall be less than -64 dBPa(A). The attenuation from reference point input to reference point output shall be measured using the speech like test signal defined below.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 is altered.

The test signal is defined as:

$$s(t) = \sum_i [[A + \mu_{AM} \cos(2\pi t n * f_{AM})] * \cos(2\pi t * f_{0i}) + \mu_{FM} * \sin(2\pi t * f_{FM})]$$

with A = 0,5

f_{AM} = 3 Hz, μ_{AM} = 0.5

f_{FM} = 5 Hz, μ_{FM} =1

$F_{0\,i}$= i*240 Hz ;i=1..14

The spectrum should be shaped using a shaping filter as described in Table3/P.501 which provides a slope of 5 dB/oct. In case of insufficient signal to noise ratio the filter may be switched off in order to provide more signal energy to the higher frequency range.

The training sequence level shall be –16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to CCITT Recommendation G.122 [8], annex B, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Note:

The calculation shall be conducted only for frequencies where a signal is present.

5.8 Out-of-band signals

5.8.1 Discrimination against out-of-band input signals

5.8.1.1 Handset and headset UE

a) The handset is mounted in the HATS position as defined in subclause 5.1.

   The headset is mounted as defined in subclause 5.1.

b) For input signals at frequencies of 4,65 kHz, 5 kHz, 6 kHz, 6,5 kHz, 7 kHz and 7,5 kHz at the level specified in TS 26.131.

c) The level of any image frequencies at the digital interface shall be measured selectively.

5.8.1.2 External hands-free UE

!!!! Note: The text in TS 26.131 defining the test signal should be removed, this belongs to TS 26.132

a) The hands-free terminal is setup as described in subclause 5.

b) With a white Gaussian noise signal bandlimited to 4,6 kHz up to 8 kHz applied at the MRP at a level specified in TS 26.131. The test signal level is averaged over the complete test signal sequence.

c) The total power in the frequency band 300 Hz to 3,4 kHz is measured after decoding the output of the speech encoder (electrical reference point). The 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 (at the electrical reference point).7.8.1.3Integrated hands-free UE

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a) The hands-free terminal is setup as described in subclause 5.

b) With a white Gaussian noise signal bandlimited to 4,6 kHz up to 8 kHz applied at the MRP at a level specified in TS 26.131. The test signal level is averaged over the complete test signal sequence.

c) The total power in the frequency band 300 Hz to 3,4 kHz is measured after decoding the output of the speech encoder (electrical reference point). The 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 (at the electrical reference point).

7.8.2 Spurious out-of-band receiving signals

7.8.2.1 Handset and headset UE

a) The handset is mounted in the HATS position as defined in subclause 5.1.

The headset is mounted as defined in subclause 5.1.

b) For input signals at the frequencies 500 Hz, 1 000 Hz, 2 000 Hz, and 3 150 Hz applied at the level specified in TS 26.131.

c) The level of spurious out-of-band image signals at frequencies of up to 8 kHz shall be measured selectively in the artificial ear.

7.8.2.2 Hands-free UE

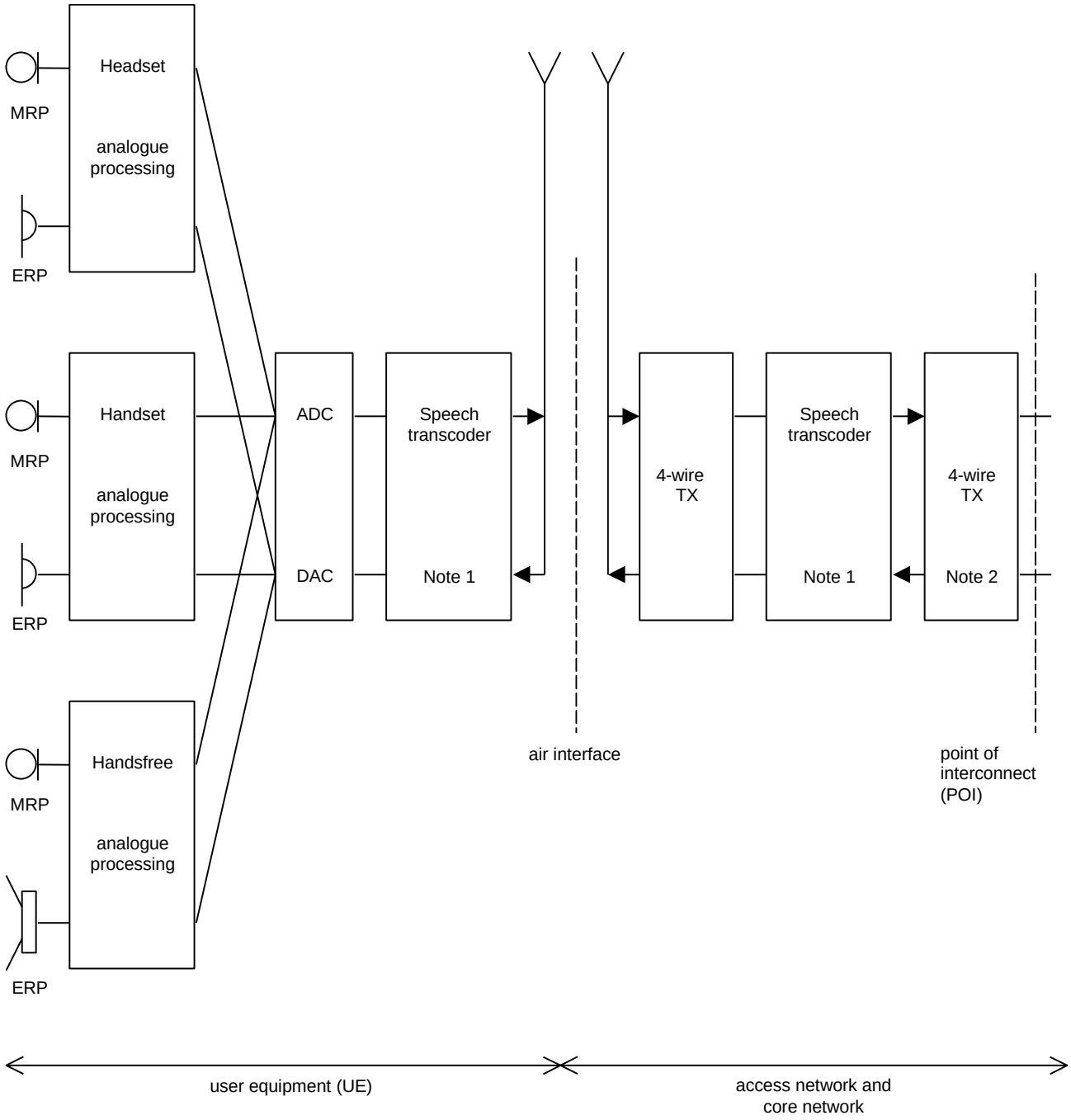
a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The test signal level is specified in TS 26.131, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is setup as described in subclause 5. The HATS is freefield equalized as described in ITU-T Recommendation P.581 [xx]. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the “right ” and “left” signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements.

c) The level of spurious out-of-band image signals is measured in the frequency range of 4,6 to 8 kHz . The reference level is obtained by measuring the in-band acoustic reference level produced by the same input signal.

5.9 Ambient Noise Rejection

See Ch. 7.5



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