Technical Specification Group Services and System Aspects **TSGS#11(01)0107**
Meeting #11, Palm Springs, USA, 19-22 March 2001

Source: **TSG-SA WG4**

Title: **CRs to TS 26.132 on Harmonisation of test methods for acoustics between 3GPP and GSM and Compatibility with testing wideband telephony transmission performance (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
26.132	002	1	R99	Harmonisation of test methods for acoustics between 3GPP and GSM	F	3.1.0	S4	TSG-SA WG4#16	S4-010237
26.132	003	1	Rel-4	Compatibility with testing wideband telephony transmission performance	B	3.1.0	S4	TSG-SA WG4#16	S4-010156R

3GPP TSG-SA4 Meeting #16
S4#16, Sophia Antipolis, France, Feb. 26-March 2, 2001
Tdoc S4-010237

CR-Form-v3

CHANGE REQUEST

⌘ **26.132 CR 002** ⌘ rev **1** ⌘ Current version: **3.1.0** ⌘

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

Title:	⌘	Harmonisation of test methods for acoustics 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 test methods for acoustics between 3GPP and GSM
Summary of change:	⌘	Pending approval of Tdoc 147, most of GSM 03.50 will refer to TS 26.132. Various sections of TS 26.132 were maked 'for further study.' This change request updates some of these sections with actual specifications. Additionally, future versions of GSM acoustic requirements will refer to 3GPP specifications. Various sections of TS 26.132 v3.1.0 need to be updated for compatibility with GSM requirements.
Consequences if not approved:	⌘	Various parts of TS 26.132 will remain incomplete. Furthremore, there will be differences between GSM and 3GPP specifications, which complicates testing of dual mode GSM-3GPP terminals.

Clauses affected:	⌘	2, 5, 7
Other specs affected:	⌘	[ ] Other core specifications ⌘ [ ] Test specifications [ ] O&M Specifications
Other comments:	⌘	Appendix_A of Tdoc 147 specifies the changes 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~~131: "Terminal Acoustic Characteristics for Telephony; Requirements ~~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] ETSI 0358 601 (TR101110) Digital Cellular Telecommunications System (Phase 2+) Characterisation test methods and quality assessment for hands-free mobiles.

[12] IEC Publication 60651 “Sound Level Meters”

[13] ITU-T Recommendation P.51 (1996): "Artificial mouth".

[14] ITU-T Recommendation P.57 (1996): "Artificial ears".

[15] ITU-T Recommendation P.58 (1996): "Head and torso simulator for telephonometry."

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

[17] Minimum Performance Requirements for Noise Suppresser Application to the AMR Speech Encoder (3GPP TS 06.77 R99)

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 3GTS26 series specifications, at it's highest source coding bit rate of 12.2kbit/s. 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.

Narrow band telephony testing

For testing a 3G terminal supporting narrow-band telephony, the system simulator shall use the AMR speech codec as defined in 3GPP TS 26 series specifications, at the source coding bit rate of 12.2kbit/s. 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).

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. For the purposes of 3G acoustic testing, the direct digital processing shall use the default speech codec, the AMR speech codec as defined in 3GTS26 series specifications, at it's highest source coding bit rate of 12.2kbit/s.

Narrow band telephony testing

For testing a 3G terminal supporting narrow-band telephony, the system simulator shall use the AMR speech codec as defined in 3GPP TS 26 series specifications, at the source coding bit rate of 12.2kbit/s.

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~~ testing the narrow-band telephony service provided by a terminal~~s~~ 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.

7.3 Idle channel noise (handset and headset UE)

7.3.1 Sending

~~To be added.~~ The terminal should be configured to the test equipment as described in subclause 5.1.

The environment shall comply with the conditions described in subclause 6.1 for idle channel noise measurement.

The Psophometric noise level at the output of the SS is measured. The psophometric filter is described in ITU-T Recommendations~~s G. 714 and~~ O.~~-~~41~~32~~.

A test signal may have to be intermittently applied to prevent 'silent mode' operation of the MS. This is for further study.

7.3.2 Receiving

~~To be added.~~ The terminal should be configured to the test equipment as described in subclause 5.1.

The environment shall comply with the conditions described in subclause 6.1.

A test signal may have to be intermittently applied to prevent 'silent mode' operation of the MS. This is for further study.

The A-weighted level of the noise shall be measured at the ERP. The A-weighting filter is descried IEC 60651 [12].

~~7.5~~7.5 ~~Measurement and calculation of the value of the D-factor~~Sidetone characteristics

7.5.1 Connections with Handset 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 , 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].~~

~~7.5.2 Sidetone loss (STMR).~~

~~For further study~~

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 the 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 handset shall be positioned in the LRGP. The handset terminal is setup as described in subclause 5, with the following exception: The Type 3.2 Low Leak artificial ear, according to ITU-T Recommendation P. 57 shall be used.

The possible use of Type 3.2 High Leak, Type 3.3, or Type 3.4 artificial ears for measurement of the sidetone loss is for further study.

The sidetone path loss L_{meST} as expressed in dB shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17. The STMR (in dB) shall be calculated from the formula 2.1 of ITU-T Recommendation P.79, using m = 0,225 and the weighting factors in table 3 of ITU-T Recommendation P.79.

7.5.2 Headset UE

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 the 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 artificial ear type is for further study.

The sidetone path loss L_{meST} as expressed in dB shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, bands 4 to 17. The STMR (in dB) shall be calculated from the formula B-4 of ITU-T Recommendation P.79 [16], using m = 0,225 and the weighting factors in Table 3 of ITU-T Recommendation P.79 [16].

7.5.3 Hands-free UE (all categories)

No requirement for other than echo control.

~~7.5.3 Sidetone distortion~~

~~For further study.~~

-7.6 Stability loss

Where a user controlled volume control is provided it is set to maximum.

~~For further study.~~**Handset UE:** The handset is placed on a hard plane surface with the transducers facing the surface.

Headset UE: for further study

Hands-free UE (all categories): no requirement other than echo loss.

A gain equivalent to the minimum stability margin is inserted in the loop between the go and return paths of the reference speech coder in the SS and any acoustic echo control is enabled.

A test signal according to ITU-T O.131 is injected into the loop at the analogue or digital input of the reference speech codec of the SS and the stability is measured. The test signal has a level of -10 dBm0 and a duration of 1 s.

No continuous audible oscillation shall be detected after the test signal is switched off.

7.8 Distortion

7.8.1 Sending Distortion

The handset, headset, or hands-free UE is setup as described in clause 5.

A sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz is applied at the MRP.

The level of this signal is adjusted until the output of the terminal is -10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels: -35, -30, -25, -20, -15, -10, -5, 0, 5, 10 dB relative to ARL.

The ratio of the signal to total distortion power of the signal output of the SS shall be measured with the psophometric noise weighting (see ITU-T Recommendations G.712 and 0.132).

NOTE: It should be ensured that the test signal is treated by speech processing algorithms as a speech-like signal, and not a noise-like signal. Test signals with a time-stationary envelope may be treated by certain algorithms, e.g. noise suppression algorithms defined in 3GPP TS 06.77 R99[17], as a noise-like signal.

7.8.2 Receiving

The handset, headset, or hands-free UE is setup as described in clause 5.

A sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz shall be applied at the signal input of the SS at the following levels: -45, -40, -35, -30, -25, -20, -15, -10, -5, 0 dBm0.

The ratio of the signal-to-total distortion power shall be measured at the ERP with the psophometric noise weighting (see ITU-T Recommendations G.712 and 0.132).

NOTE: It should be ensured that the test signal is treated by speech processing algorithms as a speech-like signal, and not a noise-like signal. Test signals with a time-stationary envelope may be treated by certain algorithms, e.g. noise suppression algorithms defined in 3GPP TS 06.77 R99[17], as a noise-like signal.

7.~~8~~9 Ambient Noise Rejection

Handset and Headset UE:

Note: This section applies to terminals providing narrow- and wide-band telephony. However, the procedure for measuring ambient noise rejection is defined only over narrow-band frequency range. Thus the test method for ambient noise rejection is the same for either narrow- or wide-band telephony.

a) A 1/2 inch pressure microphone is calibrated using a known sound source and mounted at the MRP, without the LRGP or HATS ~~head~~ present. A frequency analyser is calibrated to enable the sound pressure levels at the microphone to be determined in 1/3rd Octave bands.

b) Flood the room in which the measurement is to be made with a band limited (100 Hz to 8 kHz) pink noise to within ±3 dB. The level at MRP shall be adjusted to 70 dB(A) (-24 dBPa(A)). The tolerance on this level is +/-1 dB.~~the selected noise file, and adjust the level such that the noise level at the MRP is 70 dBA. A single noise file of real noise, covering the various noise environments that the MS could be subjected to is used ([Filename TBD]). This file is three minutes long and also commences with a three minute calibration signal. Once this tone has been adjusted to a level of 70 dBA, the average level of the noise will be 70 dBA.~~ The resulting sound spectrum is P_{rn} dBPa, measured in 1/3rd Octave bands.
To ensure that the sound field is diffuse enough, the following apply:
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 from 100 Hz to 3,15 kHz.
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 ~~my require to~~must be fed with non-coherent electrical signals to eliminate standing waves and other interference effects.

c) Position ~~an~~ the HATS or LRGP test head in the correct relative position to the MRP and mount the MS under test, according to clause 5.1.1. Recalibrate the 1/3rd Octave frequency analyser using a known voltage source to facilitate the analysis of the voltage V_{rn}, where ~~Vrn~~ V_{rn} is the voltage at the audio output of the SS due to the noise spectrum input.

d) Set up a speech path between the MS and the System Simulator (SS).

e) Determine, as a function of frequency, using the frequency analyser, in 1/3rd Octave bands (index j), the electrical output ~~V_{rn},~~ V_{jrn}, (expressed as dB rel . 1V) at the audio output of the SS for the applied acoustic pressure ~~P_{rn}~~ P_{jrn} (expressed as dB rel 1Pa) at the MRP. Since, the MS sending sensitivity is not defined above 3,4 kHz ~~and below 300 Hz~~, the measurement shall be cut off at 3,4 kHz. ~~and for~~ For the bands below ~~300~~ 315 Hz, ~~the~~ the noise level shall be referenced to the speech level at ~~300~~ 315 Hz to yield the DELSM.

The room noise sensitivity is expressed as:

~~$Smj_{rn} = V_{rn}$ (dBV) - P_{rn} (dBPa).~~ $Sm_{jrn} = V_{jrn}$ (dBV) - P_{jrn} (dBPa)

The MS ambient noise send sensitivity has now been determined.

f) The MS speech send sensitivity is now required. The required sensitivity is defined as the electrical output from the MS, measured at the audio output of the SS, as a function of the free field sound pressure at the MRP of the artificial mouth.

The measurement is made using an artificial speech source at the MRP of the artificial mouth. The 1/2 inch pressure microphone is calibrated using a known sound source. The frequency analyser is calibrated to measure in 1/3rd Octave bands. The artificial mouth output shall be in accordance with the ITU-T P.50 male artificial voice. Whilst maintaining the ITU-T P.50 "male" spectrum, adjust the total signal level to -4,7 dBPa. The resulting sound spectrum is P_o dBPa, measured in 1/3rd Octave bands. The 1/3rd Octave frequency analyser should be re-calibrated, using a known voltage source, to facilitate the analysis of the voltage V_j.Where V_j is the voltage in each 1/3rd octave band at the audio output of the SS due to the speech spectrum input. Set up a speech path between the MS and the SS. Determine the function of frequency, using the frequency analyser, and in 1/3rd Octave bands, the electrical output, V_i~~V_j~~, (expressed as dB rel. 1V), at the audio output of the SS for the applied acoustic pressure, P_{j0} ~~P_o~~, (expressed as dB rel. 1Pa/V), at the MRP.

The speech sending sensitivity is expressed as:

~~S_{mjs} (dB) = V_j (dBV) - P_o (dBPa) dBrel. 1V/Pa.~~ Sm_{js} (dB) = V_j (dBV) - P_{jo} (dBPa) dBrel. 1V/Pa.

g) The difference of the room noise sensitivity and the speech sending sensitivity DELSM (Δ_{jSM}) in each 1/3rd Octave band ~~D_{SM}~~ for the MS is determined as:

~~D~~$\Delta_{\cancel{S}jSM}$ = ~~Smj_{rn} - S_{mjs} (dB)~~Sm_{jrn} - Sm_{js} (dB) (for j = 1 to 2, $Sm_{js} = Sm_{3s}$).

h) The Ambient noise rejection ANR is calculated as the single figure value according to the following formula, the ANR shall be ≥ 0dB.

$$ANR = -\frac{4}{5}\sum_{i=1}^{13}\Delta_{jSM}\cdot 10^{-0.0175W_{jsi}}$$

j = The index of third octave bands centered at frequencies from 200 Hz to 3 150 Hz inclusive.

Wjsi = The sending weighting factors from ITU-T Recommendation P.79 [16], table 1 for the jth 1/3rd Octave band centre frequency.

Hands-free UE (all categories):

For further study

3GPP TSG-SA4 Meeting #16 ***Tdoc S4-010156R***
S4#16, Sophia Antipolis, France, Feb. 26-March 2, 2001

CR-Form-v3

CHANGE REQUEST

⌘ **26.132 CR 003**	⌘ rev **1**	⌘ Current version:	**3.1.0**	⌘

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

Title:	⌘	Amendments to TS 26.132 for compatibility with testing wideband telephony transmission performance.		
Source:	⌘	TSG-SA WG4		
Work item code:	⌘	TEI (WB)	***Date:*** ⌘	2001-2-21
Category:	⌘	**B**	***Release:*** ⌘	REL-4
		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:	⌘	To describe methods for testing acoustic requirements of terminals supporting wideband telephony.
Summary of change:	⌘	In TS 26.132 v. 3.1.0 test methods for terminal acoustics are specified for narrow-band telephony. TS 26.132 should also specify test methods for wideband telephony. It is Nokia's opinion generally that the same test methods can be used for wideband telephony as are specified for narrow-band telephony. There are a few exceptions, for example regarding which codec should be used, and the mode in which it should be used. The necessary differences between narrow and wideband in test specification are proposed.
Consequences if not approved:	⌘	Methods for testing terminals supporting wideband telephony will be undefined.

Clauses affected:	⌘	Title page, 4, 5.2, 7, 8		
Other specs affected:	⌘	[ ] Other core specifications [ ] Test specifications [ ] O&M Specifications	⌘	
Other comments:	⌘	This is a further revision of Tdoc 29 presented at SA4 #15, Munich		

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
~~Narrow band (3,1 kHz) s~~Speech and video telephony terminal acoustic test specification
(Release 1999)

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.
Five classes of acoustic interface are considered in this specification:

- Handset UE;
- Headset UE;
- Vehicle Mounted Hands-free UE
- Desk Top Operated Hands-free UE .
- Hand-Held Hands-free UE

~~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.~~

5.2 Setup of the electrical interfaces

~~1.1.1.1~~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 3GTS26 series specifications, at it's highest source coding bit rate of 12.2kbit/s. 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.

Narrow band telephony testing

For testing a 3G terminal supporting narrow-band telephony, the system simulator shall use the AMR speech codec as defined in 3GTS26 series specifications, at the source coding bit rate of 12.2kbit/s. 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).

Wide band telephony testing

For testing a 3G terminal supporting wide-band telephony, the system simulator shall use the AMR-WB speech codec as defined in 3GTS26 series specifications, at the source coding bit rate of 19.85kbit/s. The transcoding from the output of the AMR-WB 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).

~~1.1.1.1~~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. For the purposes of 3G acoustic testing, the direct digital processing shall use the default speech codec, the AMR speech codec as defined in 3GTS26 series specifications, at it's highest source coding bit rate of 12.2kbit/s.

Narrow band telephony testing

For testing a 3G terminal supporting narrow-band telephony, the system simulator shall use the AMR speech codec as defined in 3GTS26 series specifications, at the source coding bit rate of 12.2kbit/s.

Wide band telephony testing

For testing a 3G terminal supporting wide-band telephony, the system simulator shall use the AMR-WB speech codec as defined in 3GTS26 series specifications, at the source coding bit rate of 19.85kbit/s.

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~~ testing the narrow-band telephony service provided by a terminal~~s~~ 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.

For testing the wide-band telephony service provided by a terminal the test signal used shall be band limited between 100 Hz and 8 kHz with a bandpass filter providing a minimum of 24 dB/Oct. filter roll off, when feeding into the receiving direction.

7 ~~Narrow-band t~~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 or wideband telephony, either as a stand-alone service, or as part of a multimedia service.

~~1.1.1~~7.3.1 Sending

~~To be added~~ The terminal should be configured to the test equipment as described in subclause 5.1.
The environment shall comply with the conditions described in subclause 6.1 for idle channel noise measurement.

For testing narrow-band functionality, the psophometric noise level at the output of the SS is measured. The psophometric filter is described in ITU-T Recommendations G. 714 and O. 132.
For testing wideband functionality, the A-weighted noise level at the output of the SS is measured. The A-weighting filter is described in IEC 60651.
A test signal may have to be intermittently applied to prevent ‘silent mode’ operation of the MS. This is for further study.~~.~~

~~1.1.2~~7.3.2 Receiving

~~To be added.~~ The terminal should be configured to the test equipment as described in subclause 5.1.
The environment shall comply with the conditions described in subclause 6.1.
A test signal may have to be intermittently applied to prevent ‘silent mode’ operation of the MS. This is for further study.
For testing narrow-band or wideband functionality, the A-weighted level of the noise shall be measured at the ERP. The A-weighting filter is descried IEC 60651 [12].

~~8 Wideband telephony transmission performance~~

~~8.1 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.~~