



Minimum Technical Requirements for use of the HD Voice Logo with GSM/UMTS issued by GSMA (Annex C)

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1 Introduction

This document holds ANNEX C to the GSMA HD Voice Logo Licence Agreement.

The GSMA HD Voice Logo Licence Agreement and further relevant information and contact details can be found on <http://www.gsma.com/hd-voice>.

Annex C: Minimum Requirements for Mobile Networks and Terminals for the usage of the 'HD Voice' logo with GSM/UMTS

This Annex defines the minimum requirements for the usage of the 'HD Voice' logo for GSM/UMTS network operators and device vendors.

Terms:

HD Voice (High Definition Voice) for mobile terminals and networks comprises of AMR-WB codec and the enhancements to terminals and networks according to the requirements defined in this document.

AMR-WB is the Codec defined in 3GPP TS 26.171

Further information and technical details of AMR-WB are described in GSMA PRD IR36 "Adaptive Multirate Wide Band".

This Annex is split into two sections:

1. ANNEX C1: Minimum Requirements to be fulfilled by mobile network operators in order to use the 'HD voice' logo for the marketing of the AMR-WB functionality in GSM and UMTS networks.
2. ANNEX C2: Minimum Requirements to be fulfilled by mobile device vendors in order to use the 'HD voice' logo for the devices supporting AMR-WB in GSM and UMTS mode.

Annex C1: Minimum Network Requirements for HD Voice with and/or GSM/UMTS

C1.0 HD Voice enabled mobile networks

To support HD Voice, the operator shall support AMR-WB in either GSM, and/or UMTS network as described in the rest of Annex C1.

The voice service shall be compliant with 3GPP specifications related to 2G/3G Circuit Switched Telephony (TS 26.103, TS 26.201 and all other related specifications) or packet switched Multimedia Telephony over IMS (TS 26.114 and all other related specifications).

It should be noted that the operator does not need to enable AMR-WB on all networks.

C1.1 AMR-WB codec rates

At minimum the AMR-WB codec set 0 [6.6, 8.85, 12.65 kbps] must be supported.

Mode rate adaptation must be as described in 3GPP TS 45.009 for GERAN and 3GPP TS 25.415 for UTRAN.

C1.2 Codec selection

When end-to-end WB-AMR is possible, it shall be selected with the highest priority by the network(s). If AMR-WB is supported only in a part of the voice path, but wideband voice is feasible end-to-end by transcoding, then this configuration must be preferred over a narrowband voice configuration.

C1.2.1 Usage of half rate codec in GSM network

If due to any reason like loaded cell a half rate codec has to be used in the network, the network might select the normal AMR half rate codec (or the GSM_HR Codec). This can be done e.g. at call setup phase or if necessary, during the call.

C1.3 TFO / TrFO

WB-AMR should be transparently transmitted between both ends. TFO and/or TrFO must be used within an operator's network and should be used between operators networks according to the network architecture (TDM, ATM or IP interfaces and transport).

C1.3.1 Support of TFO / TrFO of 3rd party equipment

If there is a 3rd party equipment in the end-to-end chain, e.g. a Voice Quality Enhancement system, this system must be transparent to TFO/TrFO signalling.

C1.4 Transcoding

If transcoding is necessary between two systems providing a wideband voice codec (e.g. G722 on the one side and G.722.2 (AMR-WB) on the other side) then the transcoding shall keep the extended frequency range, i.e. no fallback to G.711 or another narrowband codec must happen within the transcoding processes.

If Transcoding is necessary the speech level must not be changed more than +/- 3 dB to avoid big loudness differences. This is especially important in case of (undesirable, but sometimes unavoidable) handover from WB to NB.

C1.5 Impact on KPI values

The introduction of AMR-WB in the network shall have no negative impact on any user related KPI values, i.e. Call Setup Success Rate, HO failure rate, Call drop rate, etc. shall be at least as good as before the activation of AMR-WB.

C1.6 Use Cases for HD Voice

C1.6.1 Setup of a voice call between two AMR-WB enabled devices

AMR-WB call setup shall be performed

- If both devices are AMR-WB enabled
- If both devices are in a AMR-WB enabled area
- If the HLR entry does allow the user to use AMR-WB (note that an HLR entry is optional).
- If radio and other resources allow HD Voice end to end

C1.6.2 Setup of a voice call between two HD Voice devices, one being in AMR-WB enabled area, the other one is not

This scenario does not allow HD Voice communication between the devices. The used end to end codec shall be selected by the network(s) in an automatic way, i.e. without user interaction.

C1.6.3 Setup of a voice call between a HD Voice and non HD Voice device

This scenario does not allow HD Voice communication between the devices. The used codec shall be selected by the network in an automatic way, i.e. without user interaction.

C1.6.4 Hand Over within AMR-WB coverage

Mobility within AMR-WB coverage shall be supported without any audio defects (for example without higher call drop rates, without extended interruption times compared to AMR-NB). This mobility shall include at least

- Intra- and Inter-BSC mobility
- Intra and Inter RNC mobility
- Intra and Inter MSC mobility
- Inter RAT mobility (BSC ↔ RNC)

C1.6.5 Hand Over between AMR-WB enabled area and non AMR-WB enabled area.

A Hand Over of a HD Voice device during an ongoing WB call to the non AMR-WB enabled area shall be possible without call drop and user interaction. Since AMR-WB can no longer be supported after the Hand Over, the next higher voice codec (i.e. the voice codec providing the highest possible call quality) shall be selected in an automatic way.

C1.7 Access to Services

HD Voice should be supported onto supplementary services “like” Announcements, “Personal Ring-back Tones” Voice Mail, Multi Party calls, and so on. The access to these supplementary services for AMR-WB shall be guaranteed at least in narrow band quality.

Annex C2: Minimum Requirements for GSM/UMTS Mobile HD Voice Devices

A mobile HD Voice device is characterised by:

- Supporting the AMR-WB codec,
- Providing improved wide band and narrow band speech quality acoustical characteristics and speech processing.

The requirements for a mobile terminal carrying the GSMA HD Voice Logo are translated into technical requirements described in the rest of Annex C2.

These requirements are based on the 3GPP Technical Specifications TS 26.131 and TS 26.132 for the required measurement methods. Release 11.0 of TS 26.131 and Release

11.0 of TS 26.132 are referred to in the requirement descriptions below. It is anticipated that in the future they will be replaced by further TS releases provided these do not yield any quality regression.

In all cases where the requirements allow flexibility for positioning the terminal, the position used for the tuning and for the measurement shall be reported. This is valid for all modes (handset, handheld hands-free and headset).

C2.1 AMR-WB support

The mobile HD Voice device shall support the AMR-WB codec as defined in 3GPP Technical Specifications 3GPP TS 26.171, 3GPP TS 26.190, including associated essential mechanisms, such as discontinuous transmission (DTX) as described in 3GPP TS 26.193 and error concealment procedure as described in 3GPP TS 26.191. Mode rate adaptation must be as described in 3GPP TS 45.009 for GERAN and 3GPP TS 25.321 for UTRAN.

C2.2 Wide Band Audio chain

The entire audio chain within the mobile HD Voice device must be wide band compliant. When the call is established with AMR-WB as selected codec, then the complete audio chain of the mobile HD Voice device must operate at 16 kHz sampling rate, or higher.

C2.3 Handset Mode – Frequency Response

C2.3.1 Handset Mode – Frequency Response Sending Side

In handset mode, the HD Voice device frequency response for sending shall be compliant with the mask described in 3GPP TS 26.131 clause 6.4.1 for wide band calls and to the one described in 3GPP TS 26.131 clause 5.4.1 for narrow band calls.

C2.3.2 Handset Mode – Frequency Response Receiving Side

In handset mode, the HD Voice device frequency response for receiving shall be compliant with the mask described in 3GPP TS 26.131 clause 6.4.2 for wide band calls and to the one described in 3GPP TS 26.131 clause 5.4.2 for narrow band calls.

C2.4 Handset Mode - Loudness

In handset mode, for narrow band calls, loudness rating for sending (SLR) and receiving (RLR) shall be compliant with 3GPP TS 26.131 clause 5.2.2. For wide band calls, loudness rating for sending (SLR) and receiving (RLR) side shall be compliant with 3GPP TS 26.131 clause 6.2.2.

To avoid strong level variations in case of handover from AMR-WB enabled area to a non-AMR-WB-enabled one or back, all values for narrow band and wide band cases should be as close as possible for the same volume control setting. Their difference should not exceed 3 dB.

When the control is set to its maximum, the RLR value (RLR_MAX) shall not be lower than or equal to -13 dB and shall not be higher than or equal to -3 dB.

C2.5 Handset Mode - Echo Loss

In handset mode, for narrow band calls, the echo loss shall be compliant with 3GPP TS 26.131 clause 5.7.4.

In handset mode, for wide band calls, the echo loss shall be compliant with 3GPP TS 26.131 clause 6.7.4.

C2.6 Handset Mode - Distortion

In handset mode, for narrow band calls, the distortion levels for sending and for receiving shall be compliant respectively with 3GPP TS 26.131 clause 5.8.1 and clause 5.8.2.

In handset mode, for wide band calls, the distortion levels for sending and for receiving shall be compliant respectively with 3GPP TS 26.131 clause 6.8.1 and clause 6.8.2.

C2.7 Handset Mode - Idle Noise

In handset mode, for narrow band calls, the idle noise for sending and for receiving shall be compliant respectively with 3GPP TS 26.131 clause 5.3.1 and clause 5.3.2.

In handset mode, for wide band calls, the idle noise for sending and for receiving shall be compliant respectively with 3GPP TS 26.131 clause 6.3.1 and clause 6.3.2.

C2.8 Handheld hands-free mode - Acoustical Performance

In handheld hands-free mode, the HD Voice device frequency response, loudness and echo loss shall comply with 3GPP specification.

For narrow band calls:

- The nominal loudness ratings shall be compliant with 3GPP TS 26.131 clause 5.2.4.
- The frequency response shall be compliant with 3GPP TS 26.131 clause 5.4.5 for sending and clause 5.4.6 for receiving.
- The echo loss shall be compliant with 3GPP TS 26.131 clause 5.7.3.

For wide band calls:

- The nominal loudness ratings shall be compliant with 3GPP TS 26.131 clause 6.2.4.
- The frequency response shall be compliant with 3GPP TS 26.131 clause 6.4.5 for sending and clause 6.4.6 for receiving.
- The echo loss shall be compliant with 3GPP TS 26.131 clause 6.7.3.

C2.9 Wired Headset mode - Acoustical Performance

In wired headset mode, the HD Voice device shall be compliant with 3GPP headset related specifications.

For narrow band calls:

- The loudness ratings shall be compliant with 3GPP TS 26.131 clause 5.2.5.
- The frequency response shall be compliant with 3GPP TS 26.131 clause 5.4.1 for sending and clause 5.4.2 for receiving.
- The idle noise for sending and for receiving shall be compliant respectively to 3GPP TS 26.131 clause 5.3.1 and clause 5.3.2.

For wide band calls:

- The loudness ratings shall be compliant with 3GPP TS 26.131 clause 6.2.5.
- The frequency response shall be compliant with 3GPP TS 26.131 clause 6.4.1 for sending and clause 6.4.2 for receiving.
- The idle noise for sending and for receiving shall be compliant respectively to 3GPP TS 26.131 clause 6.3.1 and clause 6.3.2.

The headset shall be tested with identified HD Voice devices. The list of HD Voice devices a headset is compliant with will be made available.

C2.10 - Noise Reduction – General Requirement

For sending, in handset mode, the HD Voice device shall reduce the ambient noise picked up by the microphone without degrading the quality of the speech signal.

The noise reduction performance shall be measured in wide band mode. It shall be tested through the objective method as described in section C2.10.1. In case of doubt on the results of the objective method, the subjective methodology, as described in section C2.10.2 can be used.

C2.10.1 - Noise Reduction - Objective evaluation

In handset mode, for wide band calls, the S-MOS-LQO and N-MOS-LQO scores shall be compliant with 3GPP TS 26.131 clause 6.10.2 performance objective. Namely:

- The average of the S-MOS-LQOw scores across all test conditions shall be ≥ 3.5 .
The average of the N-MOS-LQOw scores across all test conditions shall be ≥ 3.0 .
- Individual scores as well as the average across all test conditions shall be reported.

C2.10.2 - Noise Reduction – Subjective evaluation

The subjective evaluation may be applied as an optional procedure, only in cases where there is evidence that the objective method significantly underestimates one of the scores. In such cases the subjective result will supersede the objective result for each of the individual and averaged scores.

The objective measurement results should be made available with any subjective results. Note: The results of the objective method and the subjective method generally are not directly comparable. Due to the limited types of impairments covered in the subjective test and the variability which might be seen in different test labs, for different languages and also in different cultures, different scores may be obtained.

Test method

The subjective test method is according to ITU-T P.835 and the ITU-T Handbook of subjective testing practical procedures, with the following observations:

- The speech material (near end signal) shall consist of 32 sentences of speech (2 male and 2 female talkers, 8 sentences each). The speech database shall conform to the guidelines specified in ITU-T handbook of subjective testing practical procedures, section 5, and section B.3 of ITU-T P.501. Each sentence shall be normalized to an active speech level of -26dBov.

The background noise shall be setup and equalized according to ETSI EG 202 396-1. Noise types shall be reproduced at their realistic levels according to EG 202 396-1 clause 8. The test conditions are specified in Table C2.

Reference Conditions			
File	MNRU.	SNR	Noise Type
i01	Source	No Noise	-

	(original)		
i02	Source (original)	0dB	Fullsize_Car1_130Kmh_binaural
i03	Source (original)	12dB	Fullsize_Car1_130Kmh_binaural
i04	Source (original)	24dB	Fullsize_Car1_130Kmh_binaural
i05	Source (original)	36dB	Fullsize_Car1_130Kmh_binaural
i06	0dB	No Noise	-
i07	12dB	No Noise	-
i08	24dB	No Noise	-
i09	36dB	No Noise	-
i10	24dB	24dB	Fullsize_Car1_130Kmh_binaural
i11	12dB	12dB	Fullsize_Car1_130Kmh_binaural
i12	0dB	0dB	Fullsize_Car1_130Kmh_binaural
Test Conditions			
File	Speech level @ MRP	Noise level @ HATS ear simulators	Noise Type
i13	-1.7dBPa	L: 75,0 dB(A) R: 73,0 dB(A)	Pub_Noise_binaural_V2
i14	-1.7dBPa	L: 74,9 dB(A) R: 73,9 dB(A)	Outside_Traffic_Road_binaural
i15	-1.7dBPa	L: 69,1 dB(A) R: 69,6 dB(A)	Outside_Traffic_Crossroads_binaural
i16	-1.7dBPa	L: 68,2 dB(A) R: 69,8 dB(A)	Train_Station_binaural
i17	-1.7dBPa	L: 69,1 dB(A) R: 68,1 dB(A)	Fullsize_Car1_130Kmh_binaural
i18	-1.7dBPa	L: 68,4 dB(A) R: 67,3 dB(A)	Cafeteria_Noise_binaural
i19	-1.7dBPa	L: 63,4 dB(A) R: 61,9 dB(A)	Mensa_binaural
i20	-1.7dBPa	L: 56,6 dB(A) R: 57,8 dB(A)	Work_Noise_Office_Callcenter_binaural

Table C2: Test and Reference conditions for subjective evaluation of Noise Reduction

- The handset terminal shall be set-up on HATS and the handset mounting position documented as described in 3GPP TS 26.132 clause 6.1.1.

- For reproduction of the near-end signal, a HATS conforming to ITU-T P.58 is used. The mouth simulator shall be equalized according to 3GPP TS 26.132 guidelines and the gain adjusted to produce an active speech level of -1.7 dBPa at the MRP.
- The send signal is recorded at the electrical reference point of a network simulator to generate the processed (noise suppressed) speech materials for the subjective test. The network simulator shall be set to an UMTS call with AMR-WB 12.65kbps speech codec.
- The recordings of processed speech materials and reference conditions shall be normalized for use in the subjective test.
- For the test conditions, the normalization gain is the gain necessary to obtain an active speech level of -26dBov with a clean speech condition (no noise applied in the room). This normalization gain shall then be applied to all other test conditions (noise suppressed speech signals).
- For the reference conditions, the clean speech and noise signals shall be filtered with the MSIN and LP7 filters available from ITU-T G.191. LP7 filter will be used in combination with the HQ3 up-sampling (1:3) and down-sampling (3:1) as defined in G.191 as well. Prior to mixing, the speech shall be normalized to an active speech level of -26 dBov. The mixing shall be performed to obtain the SNRs described in Table C2. The SNR is defined as the ratio between active speech levels to unweighted noise level.
- The headphones used are calibrated and equalized using a HATS conforming to ITU-T Recommendation P.58 and an artificial ear type 3.3 according to ITU-T Recommendation P.57. The HATS is diffuse field equalized. The resulting frequency response characteristic of the headphones used in the subjective experiments shall be within the mask given in TS 26.131, clause 6.4.2.
- The presentation of the test and reference conditions to listeners shall be diotic, and the system gain adjusted so that a speech segment of -26dBov corresponds to a presentation level of 73 dB SPL measured at the DRP with diffuse-field equalization of the HATS active.
- The experimental design shall include the 12 reference and 8 test conditions described in Table C2.
- The test and reference conditions shall be presented to a total of 32 naive listeners. The listeners shall be native speakers of the language used for the test. The subjective test presentation sequence (i.e. “randomizations”) is provided in Table C3. Each of the eight presentation sequences in Table C3 shall be presented to four of the 32 listeners.



Subjective Test
Presentation Sequence

Table C3: Subjective test presentation sequence

Requirements

When testing through the subjective methodology, the HD Voice device shall comply with the following requirements:

- P.835 SIGw: Transmission quality of the speech
 - The average of P.835 SIGw scores across all 8 different ambient noise conditions from Table C2 shall be: $\geq 3,5$
- P.835 BAKw: Transmission quality of the background noise
 - The average of P.835 BAKw scores across all 8 different ambient noise conditions from Table C2 shall be $\geq 3,0$

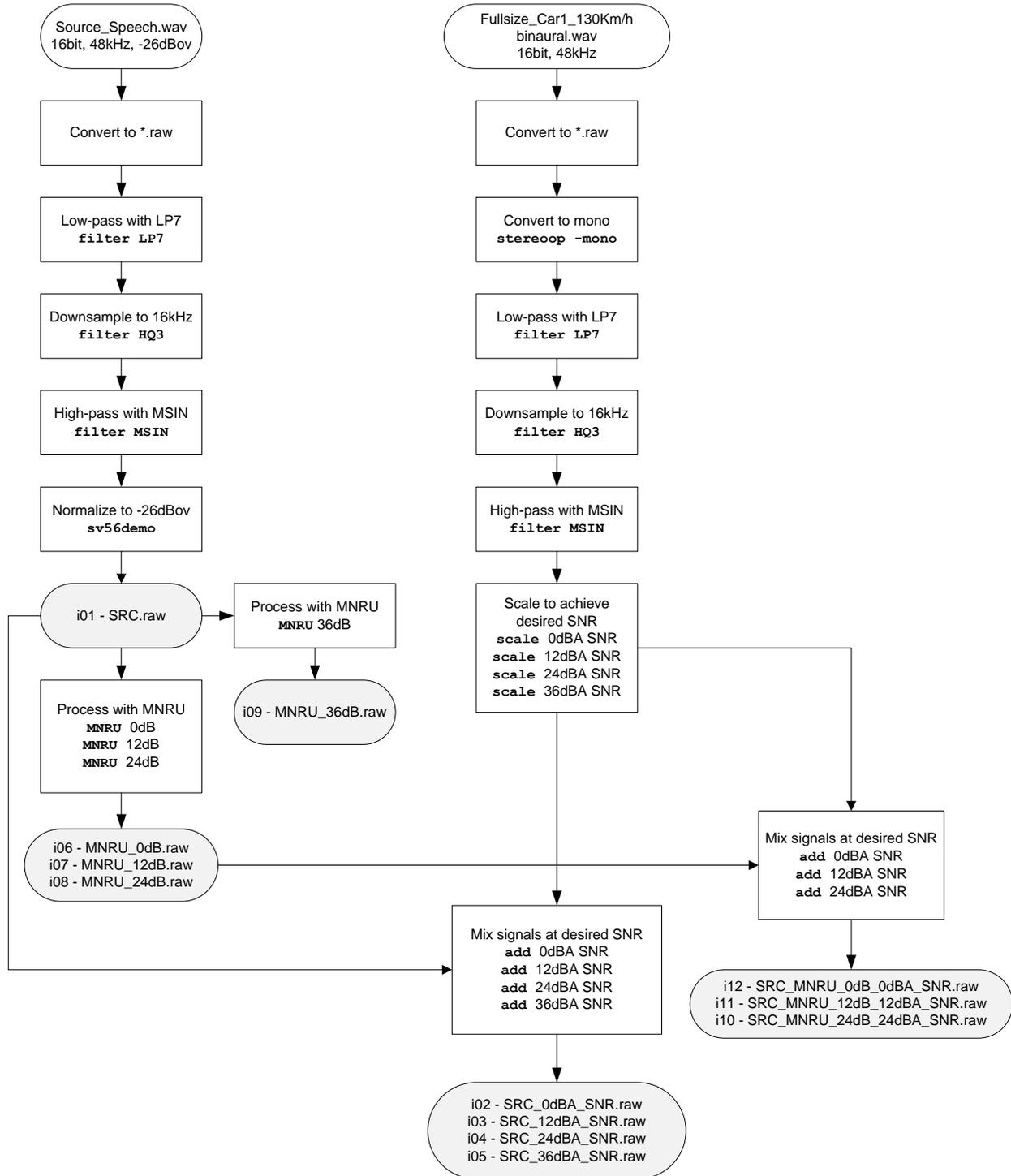


Figure C2.10-1: Processing for generating the reference conditions of the subjective test.

C2.11 Speech path Delay of mobile HD Voice devices

In handset mode, for narrow band calls, the speech path delay shall be compliant with 3GPP TS 26.131 clause 5.12.1.

In handset mode, for wide band calls, the speech path delay shall be compliant with 3GPP TS 26.131 clause 6.11.1

In wired headset mode, for narrow band calls, the speech path delay shall be compliant with 3GPP TS 26.131 clause 5.12.2.1.

In wired headset mode, for wide band calls, the speech path delay shall be compliant with 3GPP TS 26.131 clause 6.12.2.1

T_{TES} and T_{TER} are the delays of the network simulator respectively for the sending and receiving directions. Examples as provided by simulator manufacturers are given in Table C4.

	Rohde & Schwarz CMU 200 with UMTS AMR and AMR-WB all rates Equipment version number: CMU 200 SW 5.20 CMU-B21v14 + CMU-B52v14 for GSM and UMTS	Other manufacturer: data to be provided by GSMA ¹
T_{TES}	85 ms	
T_{TER}	125 ms	

Table C4: System Delay of the network simulators

C2.12 Sidetone characteristics

In handset and wired headset mode, for narrow band calls, the sidetone characteristics shall be compliant with 3GPP TS 26.131 clause 5.5. The maximum sidetone delay shall be ≤ 5 ms.

In handset and wired headset mode, for wide band calls, the sidetone characteristics shall be compliant with 3GPP TS 26.131 clause 6.5.

C2.13 Template for reporting test results

Acoustic performance test results and handset positioning information for the UE may be reported using the template attached (GSMA_HDVoice_Acoustic_Performance_Template_v2_0_AnnexC.xlsx)



GSMA_HDVoice_Acoustic_Performance_Template_v2_0_AnnexC.xlsx

¹ Data will be compiled and given to GSMA to make it available

Table C5:HD Acoustic Performance**Reference Documents**

Tag	Title	Reference	Available at:
3GPP TS 25.321	Medium Access Control (MAC) protocol specification	3GPP TS 25.321 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/25321.htm
3GPP TS 26.103	Speech codec list for GSM and UMTS	3GPP TS 26.103 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/26103.htm
3GPP TS 26.201	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Frame structure	3GPP TS 26.201 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/26201.htm
3GPP TS 26.131	Terminal acoustic characteristics for telephony; Requirements	3GPP TS 26.131 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/26131.htm
3GPP TS 26.132	Speech and video telephony terminal acoustic test specification	3GPP TS 26.132 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/26132.htm
3GPP TS 26.171	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; General description	3GPP TS 26.171 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/26171.htm
3GPP TS 26.190	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Transcoding functions	3GPP TS 26.190 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/26190.htm
3GPP TS 26.191	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Error concealment of erroneous or lost frames	3GPP TS 26.191 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/26191.htm
3GPP TS 26.193	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Source controlled rate operation	3GPP TS 26.193 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/26193.htm
3GPP TS 26.114	IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction	3GPP TS 26.114 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/26114.htm

Tag	Title	Reference	Available at:
3GPP TS 45.009	Radio Access Network; Link adaptation	3GPP TS 45.009 Rel.10 or later	http://www.3gpp.org/ftp/Specs/html-info/45009.html
3GPP TS 25.415	UTRAN Iu interface user plane protocols	3GPP TS 25.415 Rel.10 or later	http://www.3gpp.org/ftp/Specs/html-info/25415.htm
ETSI EG 202 396-1	Speech quality performance in the presence of background noise; Part 1: Background noise simulation technique and background noise database	ETSI EG 202 396-1	http://www.etsi.org/deliver/etsi_eg/202300_202399/20239601/01.02.02_60/eg_20239601v010202p.pdf
GSMA PRD IR36	Adaptive Multirate Wide Band.	IR.36.1.0 - Version 1.0	http://www.gsma.com/documents/ir-36-1-0-adaptive-multirate-wide-band/21877
ITU-T G.191	Software tools for speech and audio coding standardization	ITU-T Recommendation G.191	http://www.itu.int/rec/T-REC-G.191-200509-S
ITU-T P.57	Artificial ears	Recommendation ITU-T P.57	http://www.itu.int/rec/T-REC-P.57-200904-S
ITU-T P.58	Head and torso simulator for telephony	ITU-T Recommendation P.58	http://www.itu.int/rec/T-REC-P.58-201112-P
ITU T P.501	Test signals for use in telephony	ITU-T Recommendation P.501	http://www.itu.int/rec/T-REC-P.501-200912-S
ITU-T P.835	Subjective test methodology for evaluating speech communication systems that include noise suppression algorithm	ITU-T Recommendation P.835	http://www.itu.int/rec/T-REC-P.835-200311-I

Document Management

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
BETA	27 September 2011	Initial set of minimum terminal and network requirements for the use of the HD Voice Logo with GSM, UMTS	IREG, TSG PSMC	Yannick Mahieux (FT/Orange)
1.0	14 June 2012	BETA version updated with editorial updates, précised requirements and test measurements, new requirements for Sidetone characteristics and extensions of requirements to also support wired headsets.	IREG, TSG PSMC	Yannick Mahieux (FT/Orange)
1.0	22 March 2013	First version of Minimum terminal and network requirements for the use of the HD Voice Logo with CDMA2000	IREG, TSG PSMC	Yannick Mahieux (FT/Orange)
2.0	01 August 2013	Second version of Minimum terminal and network requirements for the use of the HD Voice Logo with CDMA2000	IREG, TSG PSMC	Andre Schevciw (Qualcomm, Incorporated)

Other Information

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