Minimum Technical Requirements for use of the HD Voice Logo with LTE issued by GSMA (Annex F)

Version 3.0

10 August 2016

This is a Non-binding Permanent Reference Document of the GSMA

1 Introduction

This document holds ANNEX F 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 F: Minimum Requirements for Mobile Networks and Terminals for the usage of the 'HD Voice' logo with LTE

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

Terms:

HD Voice (High Definition Voice) comprises either:

- the AMR-WB Codec or
- the EVS Codec operated in WB primary modes with support for the AMR-WB Codec

along with the enhancements to mobile terminals and networks according to the requirements defined in this document.

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

EVS is the Codec defined in 3GPP TS 26.441. Unless wider single audio bandwidths of operation ('swb', 'fb') are negotiated the EVS codec must be negotiated with the SDP attributes 'bw=wb', 'bw-send=wb' and 'bw-recv = bw' to guarantee wideband operation. In the following, the EVS codec with this audio bandwidth setting is referred to as EVS-WB. The EVS Codec also includes the AMR-WB IO mode of operation which may also be considered part of EVS-WB. The EVS codec may also be operated in SWB or FB although these bandwidths are beyond the scope of this document.

When the EVS codec is supported in a mobile terminal, EVS AMR-WB IO may serve as an alternative implementation of the AMR-WB codec. In this case, a compliant EVS codec implementation will be considered to have also satisfied the codec requirements of AMR-WB and the requirements and recommendations defined in sections F1 and F2 of this Annex. The AMR-WB codec in terminals also apply to, and are satisfied by, EVS AMR-WB IO.

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 F1: Minimum Requirements to be fulfilled by mobile network operators in order to use the 'HD voice' logo for the marketing of the AMR-WB/EVS functionality in LTE networks.
2. ANNEX F2: Minimum Requirements to be fulfilled by mobile device vendors in order to use the 'HD voice' logo for the devices supporting AMR-WB/EVS in LTE mode.

Annex F1: Minimum Network Requirements for HD Voice with LTE

F1.0 HD voice enabled LTE mobile networks

To support HD Voice in the LTE network, the operator shall support either:

- the AMR-WB Codec or
- the EVS Codec operated in WB primary modes with support for the AMR-WB Codec

as described in the rest of Annex N3. AMR-WB is the codec defined in 3GPP TS 26.171 (Speech codec speech processing functions: Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; General description) and other related specifications. The EVS Codec is defined in 3GPP TS 26.441 (Codec for Enhanced Voice Services (EVS); General Description) and other related specifications. The EVS codec may also be operated in SWB or FB although these bandwidths are beyond the scope of this document.

The HD voice service on the LTE network shall be compliant with 3GPP specifications for voice over IP Multimedia Subsystem (IMS) as specified in TS 26.114 (IP Multimedia Subsystem; Multimedia Telephony; Media handling and interaction) and all other related specifications. The HD voice service shall also be compliant with GSMA IR.92 (IMS Profile for Voice and SMS).

AMR-WB may also be supported in the GSM and/or UMTS network; however, this is not a mandatory requirement.

F1.1 Codec rates

At a minimum for the AMR-WB codec, the rate set 0 [6.6, 8.85, 12.65 kbps] shall be supported as described in IR.92 section 3.2.1.

If activated, mode rate adaptation for speech shall be as described in 3GPP TS 26.114 section 7.3 and section 7.5.2.1.2.

For the EVS Codec, the EVS bitrates to be supported are FFS.

F1.2 Codec selection

When end-to-end WB coding is possible, it shall be selected with higher priority than narrowband coding by the LTE network. If WB coding 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.

F1.3 TrFO (Transcoder-Free Operation)

WB coded audio should be transparently transmitted between both ends. TrFO must be used within an operator's LTE network. TrFO should also be used between operators networks according to the network architecture (IP interfaces and transport). Where one operator’s network supports AMR-WB and the other additionally supports the Primary modes of EVS-WB, either AMR-WB or the AMR-WB IO mode of the EVS Codec should be negotiated.

F1.3.1 Support of TrFO of 3rd party equipment

If there is 3rd party equipment in the end-to-end chain, e.g., a Voice Quality Enhancement system, this system must be transparent to TrFO signaling and preserve wideband voice.

F1.4 Transcoding

If transcoding is necessary between two systems providing a wideband voice codec (e.g. G.722 on 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 levels shall be aligned to ensure suitable audio levels to the end users. This alignment shall be performed such that the nominal level is preserved (0 dBm0 shall be maintained to 0 dBm0).

F1.5 Impact on KPI values

The introduction of AMR-WB into the LTE 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.

Similarly, the introduction of EVS into the LTE 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 EVS.

F1.6 Use Cases for HD Voice

F1.6.1 Setup of a voice call between two WB enabled devices

WB call setup using AMR-WB or EVS-WB shall be performed

- If both devices are WB enabled
- If both devices are in an AMR-WB/EVS-WB enabled area
- If the HLR-HSS entry does allow the user to use AMR-WB/EVS-WB (note that an HLR-HSS entry is optional)
- If radio and other resources allow HD Voice end to end
- Unless the devices are SWB-enabled or FB-enabled and a wider bandwidth call setup is performed instead.

F1.6.2 Setup of a voice call between two HD Voice devices, one being in an AMR-WB or EVS-WB enabled area, the other one 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.

F1.6.3 Setup of a voice call between a HD Voice and a 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.

F1.6.4 Hand Over within AMR-WB/EVS-WB enabled coverage

Mobility within AMR-WB/EVS-WB enabled coverage shall be supported without higher call drop rates when compared to AMR-NB and with minimal audio defects (for example, extended interruption times, audio distortion or added noise) when compared to AMR-NB.

This holds for mobility inside the same LTE network or to other AMR-WB/EVS-WB enabled access networks. In the case of mobility inside the same LTE network , the MOS score and the e2e delay after the handover shall remain identical to the original ones and the handover coverage shall include at least:

- Intra and Inter eNB (Evolved Node B) mobility
- Intra and Inter MME (Mobility Management Entity) mobility
- Intra and Inter S-GW (Serving Gateway) mobility

F1.6.5 Hand Over between AMR-WB/EVS-WB enabled area and non AMR-WB/EVS-WB enabled area

A Hand Over of a HD Voice device during an ongoing WB call to the non AMR-WB/EVS-WB enabled area shall be possible without call drop and user interaction. Since neither AMR-WB or EVS can 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.

F1.7 Access to Services

The access of HD Voice device to supplementary services like Announcements, “Personal Ring-back Tones”, Voice Mail, Multi Party calls, and so on shall be guaranteed at least in narrow band quality.

HD Voice should be supported onto these supplementary services. In such a case audio prompts shall be stored and reproduced in wide band quality.

Annex F2: Minimum Requirements for LTE Mobile HD Voice Devices

A mobile HD Voice device is characterised by:

- Supporting the AMR-WB codec or the EVS codec and the associated media control and media transport,
- Providing improved wide band and narrow band speech quality acoustical characteristics and speech processing,
- Supporting the mechanisms (jitter buffer management, packet loss concealment …) that minimize the effects induced by the transport channel,
- Ensuring the preservation of voice quality in case of handover or of concurrent data applications.
- SWB or FB may also be provided although these bandwidths are beyond the scope of this document

The requirements for a mobile terminal carrying the GSMA HD Voice Logo are translated into technical requirements described in the rest of Annex F2.

These requirements are based on the GSMA IR.92 specification, which itself refers to 3GPP Technical Specifications TS 26.114, TS 26.131 and TS 26.132 for the required measurement methods. Release 12.0 of 3GPP Technical Specifications 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).

F2.1 AMR-WB and EVS support

As described in IR92 clause 3.2.1, 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 all nine (9) modes, 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. The mobile HD Voice device shall be capable of operating with any subset of these nine (9) codec modes.

The mobile HD Voice device should support the EVS codec as defined in 3GPP Technical Specifications 3GPP TS 26.441, 3GPP TS 26.445, including all operating modes, associated essential mechanisms, such as discontinuous transmission (DTX) as described in 3GPP TS 26.450 and error concealment procedure as described in 3GPP TS 26.447. The mobile HD Voice device shall be capable of operating with any subset of these coding modes. The EVS AMR-WB IO operating modes of EVS will be included and are necessary to permit tandem-free operation with AMR-WB equipped networks and mobile HD Voice devices. (The EVS AMR-WB IO mode may be used to satisfy the requirement above for the mandatory support of AMR-WB and also be used instead of AMR-WB for conformance testing purposes – clauses F2.4 – F2.13).

When the EVS codec is not supported, the AMR-WB codec must be listed as the first payload type in the SDP offer. When the EVS codec is supported, the AMR-WB codec must be listed as a payload type in the SDP offer and 3GPP Release 12 of TS 26.114 subclauses 5.2.1.1, 5.2.1.3, 5.2.1.4, 5.2.1.5 and 5.2.1.6 shall apply.

F2.2 Media control and media transport

As described in GSMA IR.92, clause 3.2.2:

The Real Time Protocol (RTP) profile, Audio Video Profile (AVP) IETF RFC 3551 shall be used.

The SDPCapNeg framework shall not be used, but the VoLTE device shall be able to receive and answer to an SDP offer which uses SDPCapNeg. The answer shall indicate the use of the RTP AVP profile. ECN as described in 3GPP TS 26.114 is not requested.

The HD Voice device must use RTP over UDP as described in IETF RFC 3550 and IETF RFC 768, respectively, to transport voice and use symmetric RTP as defined in IETF RFC 4961 (see GSMA IR.92 clause 3.2.3).

In accordance in GSMA IR.92, clause 3.2.5,

The AMR-WB and the AMR payload formats as specified in IETF RFC 4867 must be used. The two modes “bandwidth-efficient” and “octet-aligned” must be supported. TIf the EVS codec is supported then the EVS payload format as specified in 3GPP TS 26.445 Annex A maymust be used.

When originating a session, the HD Voice device must request the use of bandwidth-efficient mode.

The HD Voice device must send the number of speech frames, or fewer, encapsulated in each RTP packet as requested by the other end using the ptime attribute.

The HD Voice device must request to receive one speech frame encapsulated in each RTP packet but must accept any number of frames per RTP packet, up to the maximum limit of 12 speech frames per RTP packet. Consequently, in the SDP negotiation, the ptime attribute must be set to 20 and the maxptime attribute must be set to 240.

The HD Voice device must be able to sort out the received frames based on the RTP Timestamp and must remove duplicated frames if present.

The HD Voice device must support RTCP as described in GSMA IR.92 clause 3.2.4.

F2.3 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 or EVS-WB as the selected codec, then the complete audio chain of the mobile HD Voice device must operate at 16 kHz sampling rate, or higher.

F2.4 Handset Mode – Frequency Response

F2.4.1 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.

F2.4.2 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.

F2.5 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 an AMR-WB enabled or EVS-WB enabled area to a non-AMR-WB enabled or non-EVS-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.

F2.5 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.

F2.6 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.

F2.7 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.

F2.8 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.

F2.9 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.

F2.10 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 F2.10.1. In case of doubt on the results of the objective method, the subjective methodology, as described in section F2.10.2 can be used.

F2.10.1 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.

F2.10.2 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 CF.

Reference Conditions			
File	**MNRU.**	**SNR**	**Noise Type**
i01	Source (original)	No Noise	-
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 F2: 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 or the AMR-WB IO 12.65kbps mode of the EVS Codec when the EVS codec is supported in a mobile terminal.
- 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 F2.
- 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 F3. Each of the eight presentation sequences in Table F3 shall be presented to four of the 32 listeners.

Subjective Test
Presentation Sequen(

Table F3: 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 F2 shall be: ≥ 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 F2 shall be ≥ 3,0

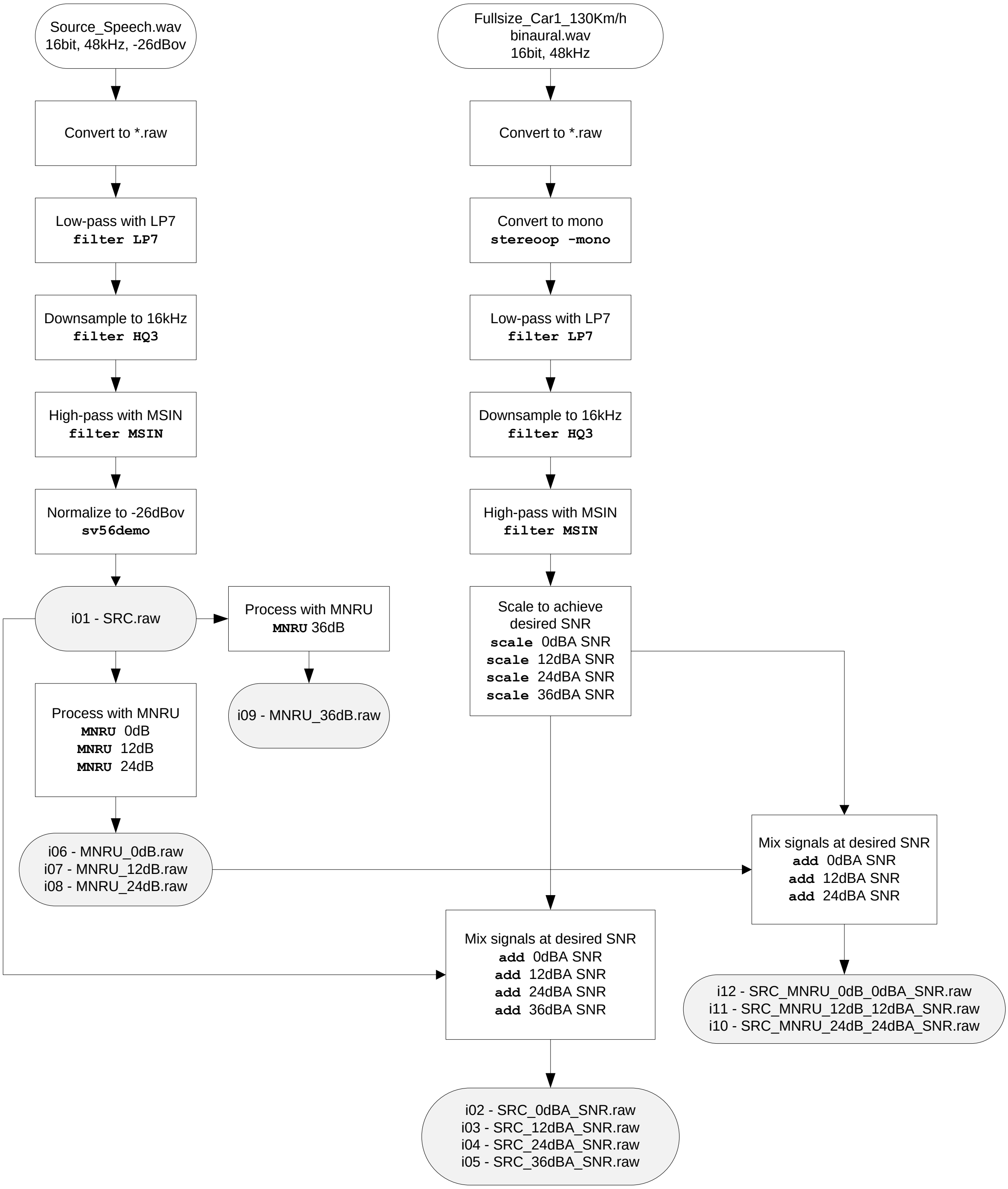

Figure F2.10-1: Processing for generating the reference conditions of the subjective test.

F2.11 Speech path delay of mobile HD Voice devices

F2.11.1 UE delay in jitter and error free conditions

In handset and wired headset mode, for narrow band calls, the UE delay in jitter and error free conditions shall be measured according to the method described in 3GPP TS 26.132 clause 7.10.

In handset and wired headset mode, for wide band calls, the UE delay in jitter and error free conditions shall be measured according to the method described in 3GPP TS 26.132 clause 8.10.

For both narrow band and wide band calls, performance objectives and requirements for maximum delay are given in the table:

Test Condition	Delay and Loss Profile	Performance Objectives for Maximum Delay	Requirements for Maximum Delay	Speech Quality Requirements
0	Error and jitter free condition	$T_S + T_R \leq 150ms$	$T_S + T_R \leq 190ms$	No requirement, reference score $MOS\text{-}LQO_{REF}$

Table F4a - UE delay and speech quality performance objectives and requirements in conditions with packet arrival time variations and packet loss

F2.11.2 UE delay and speech quality in conditions with packet arrival time variations and packet loss

In handset and wired headset mode, for narrow band calls, the UE delay in conditions with packet arrival time variations and packet loss shall be measured according to the method described in 3GPP TS 26.132 v12.3.0 clause 7.10.4.

In handset and wired headset mode, for wide band calls, the UE delay in conditions with packet arrival time variations and packet loss shall be measured according to the method described in 3GPP TS 26.132 v12.3.0 clause 7.10.4.

For both narrow band and wide band calls, performance objectives and requirements for maximum delay and speech quality degradation are according to table

Test Condition	Delay and Loss Profile	Performance Objectives for Maximum Delay	Requirements for Maximum Delay	Speech Quality Requirements (Note 2)
1	dly_profile_20msDRX_10pct_BLER_e2e	$T_S + T_R \leq 150ms$	$T_S + T_R \leq 190ms$	$MOS\text{-}LQO_{TEST} \geq MOS\text{-}LQO_{REF} - 0.3$
2	dly_profile_40msDRX_10pct_BLER_e2e	$T_S + T_R \leq 190ms$	$T_S + T_R \leq 230ms$	$MOS\text{-}LQO_{TEST} \geq MOS\text{-}LQO_{REF} - 0.3$

Table F4b - UE delay and speech quality performance objectives and requirements in conditions with packet arrival time variations and packet loss

F2.12 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.

F2.13 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_AnnexF.xlsx)

GSMA_HDVoice_Aco
ustic_Performance_T

Table F5:HD Acoustic Performance

Reference Documents

Tag	Title	Reference	Available at:
3GPP TS 26.131	Terminal acoustic characteristics for telephony; Requirements	3GPP TS 26.131 Rel.12 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.12 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-	3GPP TS 26.191 Rel.11 or later	http://www.3gpp.org/ftp/Specs/html-info/26191.htm

Tag	Title	Reference	Available at:
	WB) speech codec; Error concealment of erroneous or lost frames		
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.12 or later	http://www.3gpp.org/ftp/Specs/html-info/26114.htm
3GPP TS 26.441	Codec for Enhanced Voice Services (EVS); General Overview	3GPP TS 26.441 Rel.12 or later	http://www.3gpp.org/ftp/Specs/html-info/26441.htm
3GPP TS 26.445	Codec for Enhanced Voice Services (EVS); Detailed Algorithmic Description	3GPP TS 26.445 Rel.12 or later	http://www.3gpp.org/ftp/Specs/html-info/26445.htm
3GPP TS 26.447	Codec for Enhanced Voice Services (EVS); Error Concealment of Lost Packets	3GPP TS 26.447 Rel.12 or later	http://www.3gpp.org/ftp/Specs/html-info/26447.htm
3GPP TS 26.450	Codec for Enhanced Voice Services (EVS); Discontinuous Transmission (DTX)	3GPP TS 26.450 Rel.12 or later	http://www.3gpp.org/ftp/Specs/html-info/26450.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
GSMA IR.92	IMS Profile for Voice and SMS	IR.92 - Version 7.0	http://www.gsma.com/newsroom/wp-content/uploads/2013/04/IR.92-v7.0.pdf
IETF RFC 768	User Datagram Protocol		http://www.ietf.org/rfc/rfc768.txt
IETF RFC 3550	RTP: A Transport Protocol for Real-Time Applications		http://www.ietf.org/rfc/rfc3550.txt
IETF RFC 3551	RTP Profile for Audio and Video Conferences with Minimal Control		http://www.ietf.org/rfc/rfc3551.txt

Tag	Title	Reference	Available at:
IETF RFC 4867	RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs		http://tools.ietf.org/search/rfc4867
IETF RFC 4961	Symmetric RTP / RTP Control Protocol (RTCP)		http://tools.ietf.org/html/rfc4961
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 telephonometry	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 telephonometry	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
1.0	4 April 2014	Inclusion of minimum network & terminal requirements for the use of HD Voice Logo with LTE	IREG, TSG PSMC	Andre Schevciw, (Qualcomm, Incorporated)
2.0	14 August 2015	Extension of the LTE HD Voice Logo Requirements to include the EVS Codec Wideband (Primary and AMR-WB IO) Modes of Operation	TSG	Jon Gibbs (Huawei)
3.0	10 August 2016	Updated section F2.11 with inclusion of UE Delay Requirements for LTE terminals aligning with 3GPP TS 26.131 (CR 1004).	TSG	Andre Schevciw

Other Information

Type	Description
Document Owner	GSMA Terminal Steering Group; GSMA IREG Group
Editor / Company	Andre Schevciw (Qualcomm)

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