



**Minimum Technical Requirements
for use of the HD Voice Logo with GSM/UMTS/LTE issued by GSMA
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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/LTE¹

This Annex defines the minimum requirements for the usage of the 'HD Voice' logo for GSM/UMTS/LTE 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:

- 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, UMTS or/and LTE networks.
- 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, UMTS or/and LTE mode.

ANNEX C1: Minimum Network Requirements for HD Voice with GSM/UMTS/LTE

C1.0 HD voice enabled mobile networks

To support HD Voice, the operator shall support AMR-WB in either GSM, and/or UMTS and/or LTE 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

¹ Note, requirements for VoLTE are not addressed in this version of 1.1 but has been noted to be included for future release

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/LTE 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.

As far as possible these requirements are based on the 3GPP Technical Specifications TS 26.131 and TS 26.132 for the required measurement methods. Release 10.2 of TS 26.131 and Release 10 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.

Where a topic is not correctly covered by the 3GPP TS an alternative solution is used based on other standards.

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 3dB.

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, with the addition that TCLw shall be ≥ 55 db for maximum volume control.

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 Headset mode - Acoustical Performance

In 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

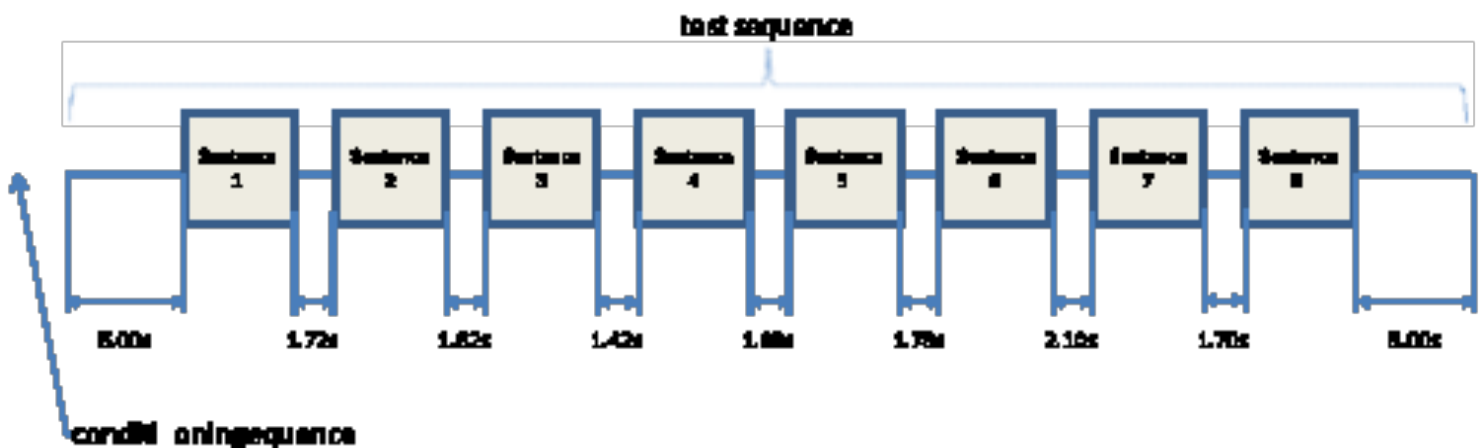
For sending, in handset mode, the HD Voice Device shall reduce the ambient noise picked up by the microphone without degrading the quality of the speech signal.

The Noise Reduction performance shall be measured in wide band mode. It can be tested through objective or subjective methodologies, as described in sections C2.10.1 and C2.10.2, respectively. The test methodology used shall be reported.

C2.10.1 - Noise Reduction - Objective evaluation Test method

The objective test method is according to ETSI TS 103 739 clause 6.10.2, with the following observations:

- The near end speech signal consists of 8 sentences of speech (2 male and 2 female talkers, 2 sentences each). Appropriate speech samples shall be taken from ITU T Recommendation P.501 and P.50. The language shall be a mixture of American and British English. The test signal level is -1.7 dBPa at the MRP. The exact test sequence should be as follows:



The following sentences shall be used:

Sentence 1 & 2: Male1, BE (P.50, Appendix I, EN2M06)

1 He could not remember his name.

2 I never can leave you two alone.

Sentence 3 & 4: Male2, AE (P.501, AE_Male1)

3 The shelves were bare of both jam or crackers.

4 A joy to every child is the swan boat.

Sentence 5 & 6: Female1, BE (P.501 Version 2004, BE_Female1)

5 You must go and do it at once.

6 There were several small outhouses.

Sentence 7 & 8: Female2, AE (P.501, AE_Female2)

- 7 The stems of the tall glasses cracked and broke.
- 8 The wall phone rang loud and often.

A proper conditioning sequence should be used in advance of the measurement.

- In addition to the minimum test conditions described in ETSI TS 103 739, clause 5.5, five other noise types from ETSI EG 202 396-1 shall be used as described in Table C1.
- The measurement over the 8 noise types shall be made in the same unique and dedicated call and not in the same as for example the one established for acoustic measurement. The noise types shall be presented according to the order specified in Table C1.
- NOTE: The use of additional test conditions is according to ETSI TS 103 739 Clause 6.10.2 note recommendation.

Description	File name	Duration	Level	Type
Recording in pub	Pub_Noise_binaural_V2	30 s	L: 75,0 dB(A) R: 73,0 dB(A)	binaural
Recording at pavement	Outside_Traffic_Road_binaural	30 s	L: 74,9 dB(A) R: 73,9 dB(A)	Binaural
Recording at pavement	Outside_Traffic_Crossroads_binaural	20 s	L: 69,1 dB(A) R: 69,6 dB(A)	Binaural
Recording at departure platform	Train_Station_binaural	30 s	L: 68,2 dB(A) R: 69,8 dB(A)	Binaural
Recording at the drivers position	Fullsize_Car1_130Kmh_binaural	30 s	L: 69,1 dB(A) R: 68,1 dB(A)	Binaural
Recording at sales counter	Cafeteria_Noise_binaural	30 s	L: 68,4 dB(A) R: 67,3 dB(A)	Binaural
Recording in a cafeteria	Mensa_binaural	22 s	L: 63,4 dB(A) R: 61,9 dB(A)	Binaural
Recording in business office	Work_Noise_Office_Callcenter_binaural	30 s	L: 56,6 dB(A) R: 57,8 dB(A)	Binaural

Table C1: Noises used for background noise simulation

Requirements

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

- N-MOS-LQOW: Transmission quality of the background noise
 - The average of the N-MOS-LQOW scores across all 8 different test conditions from Table C1 shall be $\geq 3,0$
- S-MOS-LQOW: Transmission quality of the speech
 - The average of the S-MOS-LQOW scores across all 8 different test conditions from Table C1 shall be: $\geq 3,0$

Individual scores as well as the average across all 8 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 (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 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 -26dBov. 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 Sequenc

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 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 $\geq 3,0$
- 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: $\geq 3,0$

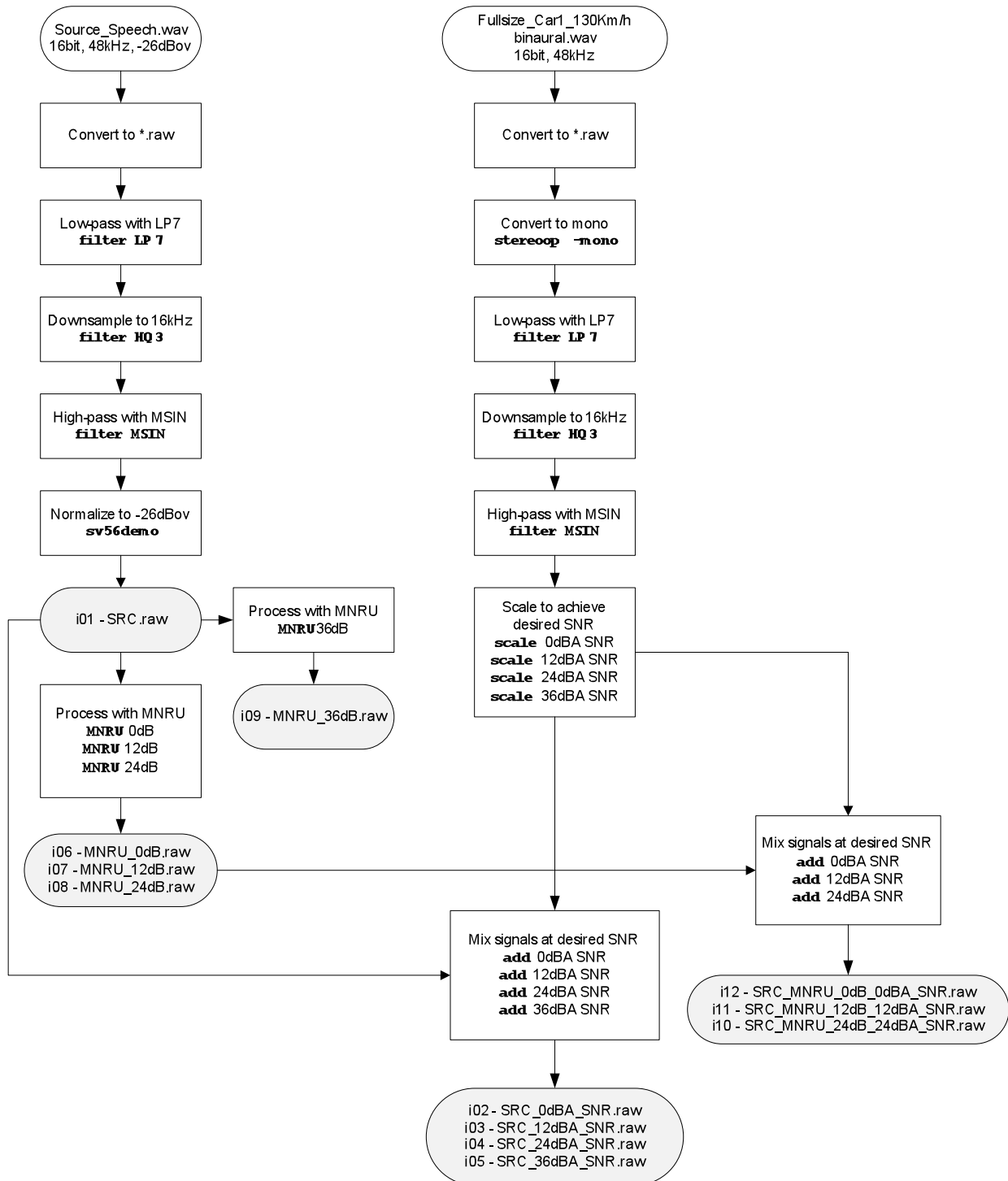


Figure C2.10-1: Processing for generating the reference conditions of the subjective test.

C2.11 Speech path Delay of mobile HD Voice devices

An excessive end to end delay reduces communication interactivity and naturalness. The 2 ways terminal delay (send + receive) T_s+r introduced by the HD Voice device shall remain reasonable; it should not exceed 200ms and in any case shall not exceed 220ms. This requirement applies to UMTS mode and both to handset and wired headset mode.

Terminal delay requirement for GSM and LTE mode is for further study.

Send (uplink) and receive (downlink) delays of the HD Voice devices are tested separately but the requirement is defined for the addition of send and receive delays (T_s+r).

The delay shall be estimated by the measurement method as defined in ETSI TS 103 739 (V1.1.2), Clause 6.12 with the following observations:

- The measurement shall be made in handset mode or in wired headset mode.
- The measurement shall be made in 3G mode at the rate of 12.65 kbps.
- The mobile HD Voice device shall be in voice only mode.
- The delay requirement in case of concurrent data applications is for further study.
- In contrast to TS 103 739 (V1.1.2) the measured delay covers the complete mobile phone including speech coding and the radio link as shown in Figs. C2.11-1 and C2.11-2. Only the systems simulator delay is then deducted from the measured result.²

The calculation is as follows: $T_s+r = T_s + T_r$

$T_s = T_{m-s} - T_{simul-s}$

$T_r = T_{m-r} - T_{simul-r}$

T_{m-s} : measured delay in send (uplink) direction

T_{m-r} : measured delay in receive (downlink) direction

$T_{simul-s}$ and $T_{simul-r}$ are the delays of the network simulators as provided by simulator manufacturers. They are given in the Table C4:

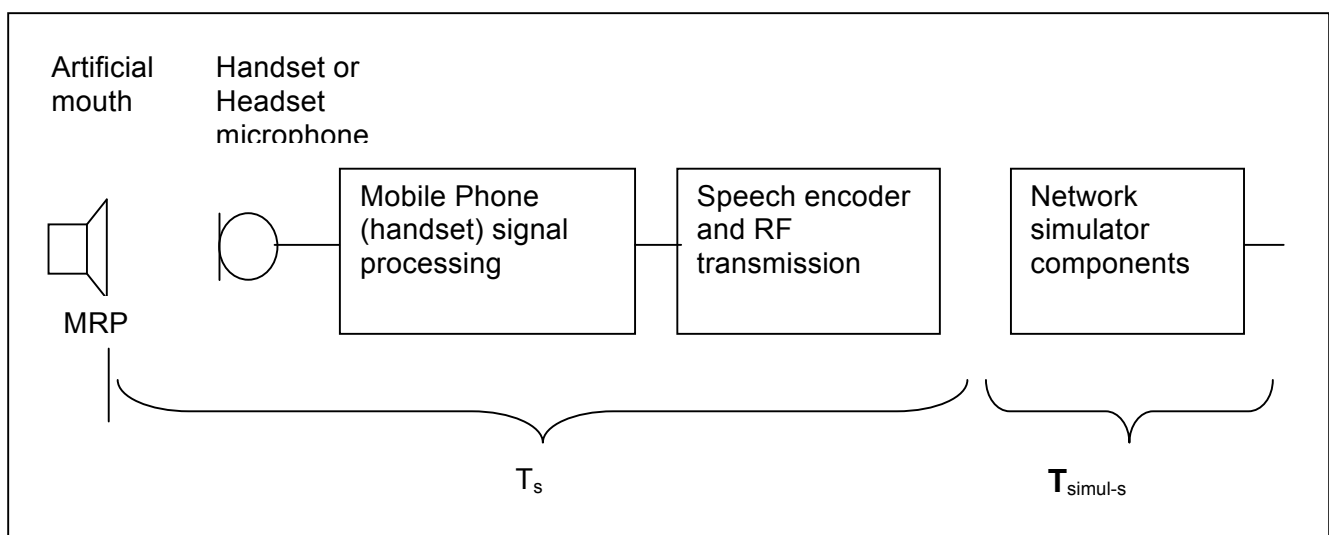


Figure C2.11-1: Different blocks contributing to the delay in uplink direction (handset and wired headset mode)

²The method defined in ETSI TS 103 739 requires that the delay component T_{system} be known. According to the definitions implied in Figures 6.12.1 (Sending) and 6.12.2 (Receiving), T_{system} is comprised of two components, one due to the speech encoder/decoder and RF transmission in the terminal, and one due to the speech decoder/encoder and RF transmission in the network simulator.

As the user is impacted by the complete round-trip delay introduced by the terminal, the radio access and the core network, here the complete delay is measured (with usage of network system simulator) and the delay introduced by the terminal is then calculated.

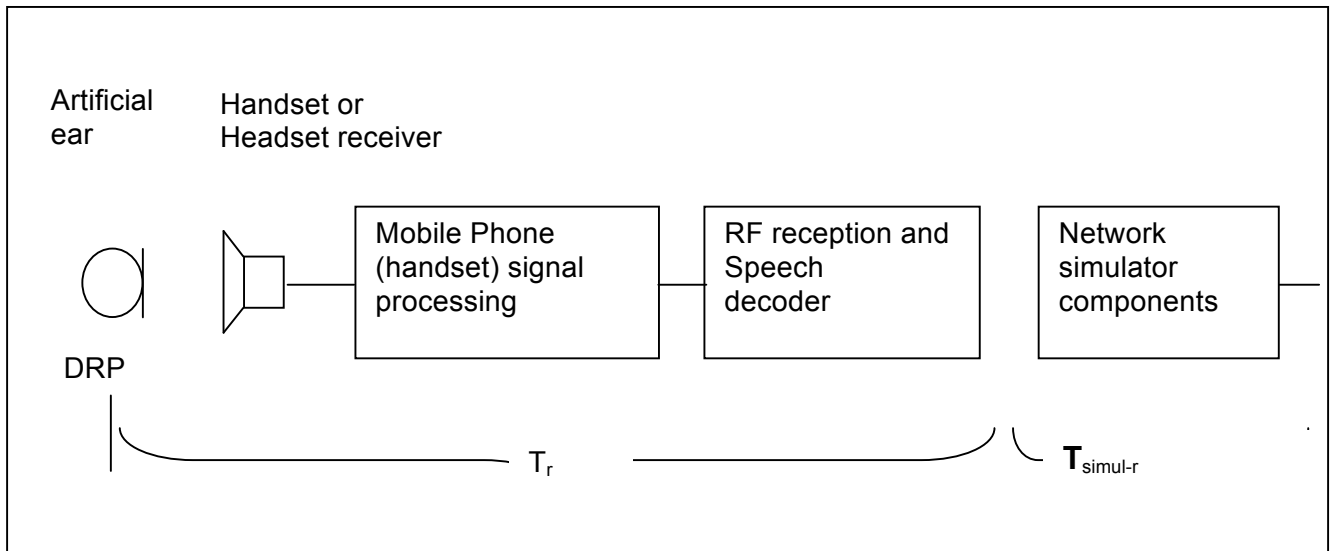


Figure C2.11-2: Different blocks contributing to the delay in downlink direction (handset and wired headset mode)

	Rohde & Schwarz CMU 200 with UMTS 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 ³
$T_{simul-s}$	80 ms	
$T_{simul-r}$	125 ms	

Table C4: System Delay of the network simulators

Illustrative example:

It is known from the network simulator manufacturer that $T_{simul-s} = 80$ ms

The measured value in the Sending (uplink) direction is $T_{m-s} = 200$ ms

The terminal portion of the Sending delay is then given by:
 $T_s = T_{m-s} - T_{simul-s} = 120$ ms.

C2.12 Sidetone characteristics

In handset and 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 headset mode, for wide band calls, the sidetone characteristics shall be compliant with 3GPP TS 26.131 clause 6.5. The maximum sidetone delay shall be ≤ 5 ms.

³ Data will be compiled and given to GSMA to make it available

Reference Documents

Tag	Title	Reference	Available at:
3GPP TS 25.321	Medium Access Control (MAC) protocol specification	3GPP TS 25.321 Rel.10 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.10 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.10 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.10 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.10 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.10 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.10 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.10 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.10 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
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
ETSI TS 103 737	Transmission requirements for narrowband wireless terminals (handset and headset) from a QoS perspective as perceived by the user	ETSI TS 103 737 V1.1.2	http://www.etsi.org/deliver/etsi_ts/103700_103799/103737/01.01.02_60/ts_103737v010102p.pdf

ETSI TS 103 739	Transmission requirements for wideband wireless terminals (handset and headset) from a QoS perspective as perceived by the user	ETSI TS 103 739 V.1.1.2	http://www.etsi.org/deliver/etsi_ts/103700_103799/103739/01.01.01_60/ts_103739v010101p.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.50	Artificial voices	ITU-T Recommendation P.50	http://www.itu.int/rec/T-REC-P.50-199909-I
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

Document History

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)
1.1	22 March 2013	Clarification of requirements for VoLTE to be addressed in a future version	IREG, TSG PSMC	Yannick Mahieux (FT/Orange)

Other Information

Type	Description
Document Owner	GSMA Terminal Steering Group; GSMA IREG Group
Editor / Company	Yannick Mahieux (FT/Orange)

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