Adaptive Multirate Wide Band

Version 5.0

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1 Introduction

1.1 Adaptive Multirate Wide Band (AMR-WB) Overview

During the development of mobile voice services, the speech quality has continued to evolve. Significant milestones include the introduction of the enhanced full-rate codec (EFR) and the Adaptive Multirate (AMR) voice codec, which increased voice quality and boosted channel error robustness and capacity. The Adaptive Multirate Narrow-Band (AMR-NB) codec, which supports the bandwidth of traditional telephony, is now widely deployed in Global System for Mobile Communications (GSM) and Universal Mobile Telecommunications System (UMTS) systems. It is also the codec of choice for the forthcoming multimedia telephony service for IP Multimedia Subsystem IMS (MTSI) standard from 3GPP. AMR then evolved to the Adaptive Multirate Wideband (AMR-WB) codec.

The AMR-WB speech codec has been specified in 3GPP Release 5 in a set of publicly available technical specifications [see section 3.4.1]. The goal was to extend the Multirate/Multimode coding principle of AMR-NB to wideband voice while retaining bit rate and essential mechanisms, such as rate control, in-band signalling, and discontinuous transmission (DTX). The ITU-T has released the 3GPP AMR-WB under the code G.722.2.

The main superior feature of AMR-WB is the support of an increased frequency band. While the plain old telephony narrow band services are optimized for a frequency range of 300 Hz to 3400 Hz, the wideband speech services support a frequency range of 50 Hz to 7000 Hz [see section 4.1].

1.2 Scope

This document provides background information on the AMR-WB functionality and the improvements brought by the introduction of the AMR-WB functionality. It also clarifies the technical impact of the introduction of AMR-WB in an already deployed 2G/3G Network. Furthermore, it offers a recommendation for the introduction and usage of the functionality.

AMR-WB has been standardized in 3GPP since 2000 and is ready for the introduction into 3G and for the introduction in 2G (including User Equipment (UE)/MS availability). It can also be assumed that AMR-WB will be part of the voice functionality in Long Term Evolution (LTE) including interworking towards legacy networks and including more advanced voice codecs.

In parallel to the introduction of the AMR-WB functionality in mobile networks, wideband voice is also becoming a more state of the art feature in fixed network Voice over Internet Protocol (VoIP) application and is offered by several operators. Currently, a significant number of Digital Enhanced Cordless Telecommunications (DECT) phones are supporting G.722 as wideband codec (as defined by CAT-iq standard [40]), an interworking proposal between mobile and fixed networks using G.722 will be given.

In detail, this Permanent Reference Document (PRD)

- Presents the voice quality improvements that AMR-WB allows,
- Highlights current existing issues / findings which have been recognised as result of some trials,
- Creates awareness of the necessary network upgrades to allow smooth AMR-WB usage,
- Explains the details of improvements achieved by AMR-WB,
- Covers an impact analysis for supporting AMR-WB in existing deployed technologies (that is 2G and 3G)
- Offers an outlook / guidance on how new 3GPP technologies (for example LTE) could / should fully support AMR-WB from initial rollout.

1.3 Motivation

AMR-NB already offers a reasonable voice quality for mobile networks; however the behaviour of users making phone calls in fixed and mobile networks varies:

- Voice calls in the fixed network are lasting longer in many cases. It is common that mobile users often quote “I’ll give you a call back on the fixed line” when wanting to have a phone call in a relaxed atmosphere.
- Many users still prefer to have a call via the fixed network instead of a mobile network.
- In some cases, the voice quality during mobile use degrades to the point that users may be required to increase the speaking volume, (possibly shout into the phone) or that it is hard to understand the other party.
- Even if in good coverage conditions the voice quality of fixed and mobile networks is comparable and there is a general consensus that the fixed network provides a better voice quality than a mobile network. In order to overcome this perception, the mobile industry must advance the voice codec to a quality which exceeds the voice quality in fixed networks.

In addition to the above considerations, certain VoIP service providers have begun to use wideband speech codecs and are attracting new customers by offering a speech quality far better than the traditional one.

It is shown that customers are not satisfied by the existing voice quality in mobile networks. There is a possible opportunity to increase the voice minutes per customer or to gain new customers (for example fixed line substitution). Furthermore, mobile voice users may in the future use VoIP applications instead of the voice service offered by the mobile industry. A possible counter action is to provide a better and more stable voice quality in parallel with the service offered by 3rd party VoIP service providers. This is achieved either by offering an enhanced Circuit Switched (CS) voice quality or by offering a VoIP solution with an appropriate Quality of Service (QoS) mechanism.

1.4 Definition of Terms

Term	Description
AMR	Adaptive Multi Rate
AMR-NB	Adaptive Multi Rate Narrow Band
AMR-WB	Adaptive Multi Rate Wide Band
CAT-iq	Cordless Advanced Telephony Internet and Quality
EFR	Enhanced Full Rate
HD	High Definition
HD Voice	High Definition Voice
LQO	Listening Quality Objective
LTE	Long Term Evolution
MOS	Mean Opinion Score
MS	Mobile Station
O0BTC	Out-of-Band Transcoder Control
TAS	Telephony Application Server
TFO	Tandem Free Operation
TrFO	Transcoder Free Operation
UE	User Equipment
VoIP	Voice over IP

1.5 Document Cross-References

Ref	Document Number	Title
1	3GPP TS 26.171	Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR WB) speech codec; General description
2	3GPP TS 26.173	AMR Wideband Speech Codec; ANSI C code
3	3GPP TS 26.174	AMR Wideband Speech Codec; Test sequences.
4	3GPP TS 26.190	AMR Wideband Speech Codec; Transcoding functions.
5	3GPP TS 26.191	AMR Wideband Speech Codec; Error Concealment of erroneous or lost Frames
6	3GPP TS 26.192	AMR Wideband Speech Codec; Comfort Noise Aspects
7	3GPP TS 26.193	AMR Wideband Speech Codec; Source Controlled Rate operation
8	3GPP TS 26.194	AMR Wideband Speech Codec; Voice Activity Detector (VAD).
9	3GPP TS 26.201	AMR Wideband Speech Codec; Frame Structure.
10	3GPP TS 26.202	Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Interface to Iu, Uu and Nb.
11	3GPP TS 26.204	Speech codec speech processing functions; Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; ANSI-C code

12	ITU-T Recommendation G.722.2	ITU-T Recommendation G.722.2: “Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)”.
13	3GPP TS 22.003	Circuit Teleservices supported by a Public Land Mobile Network (PLMN)
14	3GPP TS 26.111	Codec for circuit switched multimedia telephony service; Modifications to H.324
15	3GPP TS 26.235	Packet switched conversational multimedia applications; Default codecs
16	3GPP TS 26.140	Multimedia Messaging Service (MMS); Media formats and codes
17	3GPP TS 26.234	Transparent end-to-end Packet-switched Streaming Service (PSS); Protocols and codecs
18	3GPP TS 26.346	Multimedia Broadcast/Multicast Service (MBMS); Protocols and codecs
19	3GPP TS 26.114	“IP Multimedia Subsystem (IMS) Multimedia Telephony; media handling and interaction”
20	3GPP TS 26.141	IP Multimedia System (IMS) Messaging and Presence; Media formats and codecs
21	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, Sjoberg M. Westerlund Ericsson A. Lakaniemi Nokia Q. Xie Motorola April 2007
22	ETSI TS 181 005	“Telecommunications and Internet Converged Services and Protocols for Advanced Networking (TISPAN); Service and Capability Requirements
23	ETSI TR 185 013	“Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Codecs for customer network devices
24	3GPP TR 22.813	Study of Use Cases and Requirements for Enhanced Voice Codecs for the Evolved Packet System (EPS)
25	3GPP TR 29.976	Performance characterization of the Adaptive Multi-Rate Wideband (AMR-WB) speech codec
26	3GPP TR 29.935	Packet Switched (PS) conversational multimedia applications; Performance characterization of default codecs
27	Open IPTV Forum	“Release 2 Specification, Volume 2 – Media Formats”, V2.0, September 2010.
28	I3 FORUM	Technical Interconnection Model for International Voice Services
29	I3 FORUM	White Paper Optimal Codec Selection in International IP based Voice Networks
30		Christina Birkehammar, Stefan Bruhn, Peter Eneroth, Karl Hellwig and Stefan Johansson (2006) New high-quality voice service for mobile networks. Ericsson Review No.3, 2006.
31	Voice Age	Wideband Speech Coding Standards and Applications. White Paper http://www.voiceage.com/media/WidebandSpeech.pdf

32	ITU-T-SG16-TD202(GEN/16),	" LS on audio issues ", Source: Rapporteurs Q7/12, Geneva, 3 - 13 April 2006
33	3GPP TR26.976	" Performance characterization of the Adaptive Multi-Rate Wideband (AMR-WB) speech codec",
34		Anssi Rämö Nokia Research Center, Tampere, Finland, VOICE QUALITY EVALUATION OF VARIOUS CODECS, ICASSP 2010
35	3GPP TS 26 131	Terminal acoustic characteristics for telephony; Requirements
36	3GPP TS 26 132	Speech and video telephony terminal acoustic test specification
37	ITU-T P.835	Subjective test methodology for evaluating speech communication systems that include noise suppression algorithm
38		"Objective evaluation of wideband speech codecs for voice communication over Bluetooth" - Gary Spittle et al – 129th AES Convention , 2010 November 4–7)
39	GSMA PRD IR.92	"IMS Profile for Voice and SMS" 5.0
40	dect cat iq 2.0 ETSI TS 102 527-3	DECT; Extended wideband speech services
41	3GPP TS 26.441	Codec for Enhanced Voice Services (EVS); General Overview
42	3GPP TR 26.952	Codec for Enhanced Voice Services (EVS); Performance Characterization
43	GSMA TS.23	HD Voice Logo Technical Annexes v5.0

2 Improved User Experience with Wideband Telephony

2.1 HD voice promise

High Definition voice (HD voice) offers users an enhanced voice communication experience in line with new communication usage. It relies on high quality wideband speech coding with terminals optimized for wideband sound rendering. As a result:

- The increased naturalness of the reproduced voice helps to better share emotions, to feel closer and to forget the communication tool.
- The enhanced voice clarity provides higher intelligibility and better recognition of the speaker: long calls, in foreign languages or in a noisy environment become more efficient.

Such enhancement is achieved due to the improved rendering (reproduction) of almost the complete speech spectrum as shown in figure 1 below:

- The additional high-frequencies (3400Hz to 7000Hz) result in better fricative differentiation and therefore higher intelligibility, whereas in traditional narrowband speech transmission, the important energy present above 3400 Hz is filtered out, which explains why for instance “s” and “f” cannot be distinguished in traditional telephone calls. Wideband speech transmission also renders all unvoiced sounds related to user emotion (like whispering) and is more transparent to environmental noise.

- The additional low frequencies make the speech sound more natural and increase the effect of presence and closeness.

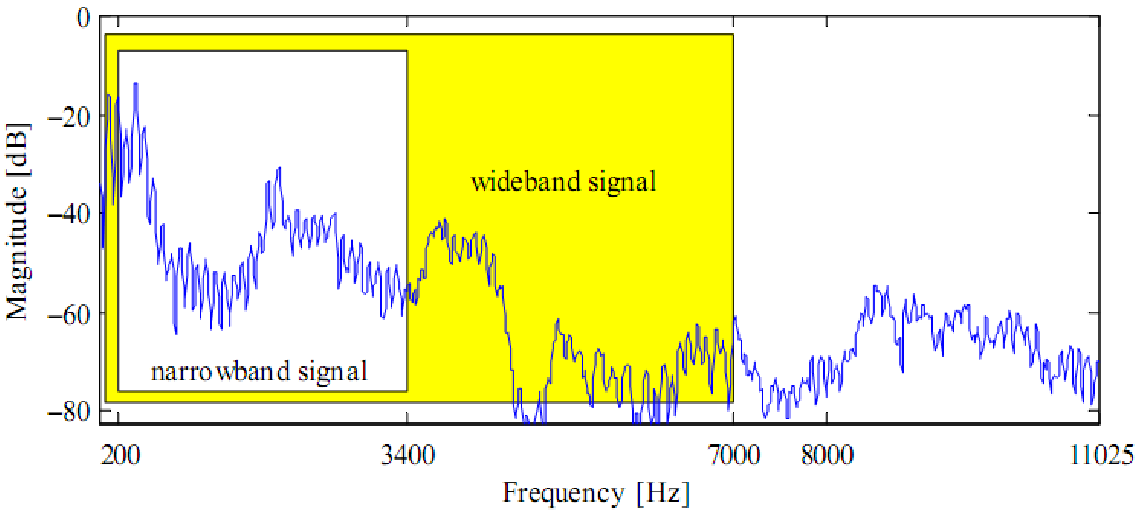


Figure 1: Example of the energy spectrum of some voiced speech customer feedback

Several field trials have been conducted to provide a baseline for the decision to start an AMR-WB introduction project or not. Based on the positive results of these trials several operators have decided to introduce AMR-WB and intensive discussions have begun between operators, network and terminal vendors to ensure an end to end availability of AMR-WB products.

2.1.1 Result from Deutsche Telekom AG trials

One of the first trials was an AMR-WB Consumer Trial conducted by Ericsson and T-Mobile in 2006 in a live network environment in Germany. The objective of this trial was to get an indication of possible user behaviour effects due to the introduction of AMR-WB and to understand subscriber perceptions, motivations and attitudes for AMR-WB. For this test, approximately 150 external subscribers were selected and categorized in three (3) focus groups (Couples and Friends, Family and Business users). A traditional (narrowband) mobile phone was modified to transmit AMR-WB coded speech at 12.65 kbps via the 3G video channel. The video was disabled. Microphone and loudspeaker were not optimized for the wideband spectrum, but taken “as they were”.

The results of these tests were encouraging:

- Approx. 70% of the participants perceived AMR-WB (High voice quality) as an attractive feature.
- 40% were very positive and missed the service after the test ended and definitely felt an added value compared to their own mobile phone (codec).
- Most important values of AMR-WB (High voice quality) are:

Possibility to make calls in situations where it was not achievable previously.

Increased privacy, discretion and comfort because of a more relaxed atmosphere during a mobile phone call.

- Approx. 40% of users believed they would change their calling behaviour (automatically) and they would make longer calls, more calls or both due to AMR-WB (High voice quality).

Figure 2 references an overview of these results. It could also be seen, that there is a very subjective feeling of the voice quality and that the results of the quality perception were more promising among females than among males.

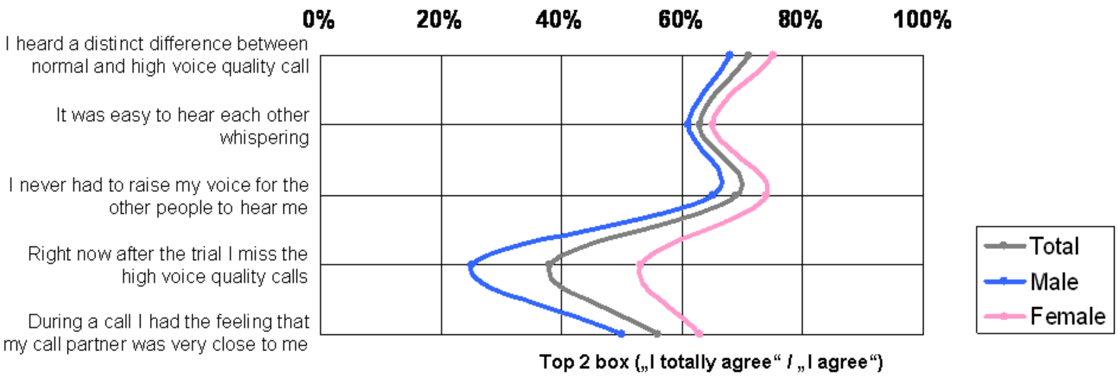


Figure 2: AMR-WB trial results (DTAG/Ericsson 2006)

2.1.2 Result from Orange France "Friendly User Test" over live AMR-WB network

2.1.2.1 Description

Following the deployment of AMR-WB over its 3G network, Orange France conducted tests to identify the usage and the perception of HD voice. For this purpose, 66 testers (the majority of whom were male between 24 and 44 years old) have been selected. They were equipped with two different AMR-WB device models and the experiment lasted approx. two months.

During the tests, AMR-WB codec set 0 was used without load based codec rate adaptation, that is, the tests were conducted with a codec rate of 12.65 kbps. The goal of the trial was to compare a HD device (voice optimized device including AMR-WB, wide band acoustics, noise cancelation, etc.) with traditional phones (non optimized phones).

2.1.2.2 Main results

Almost all participants made HD calls, more than 50% made these calls daily. Compared to their usual voice communications, participants found a very significant quality improvement in particular when calling another HD device. With regards to narrow band calls, more than half appreciated an improved quality. This results from the improved acoustical performance of the device that benefits all communications including narrowband and wideband.

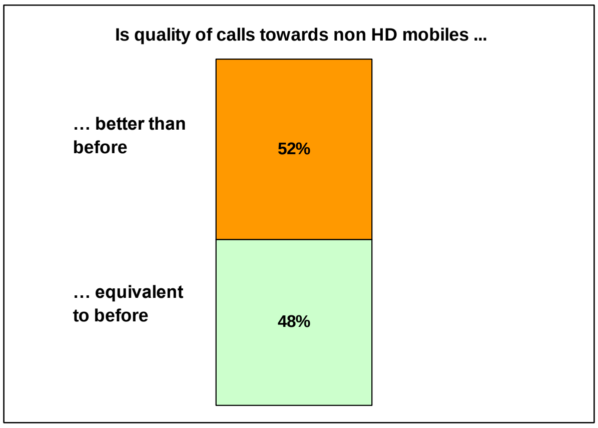


Figure 3: Orange Friendly User Test-Question

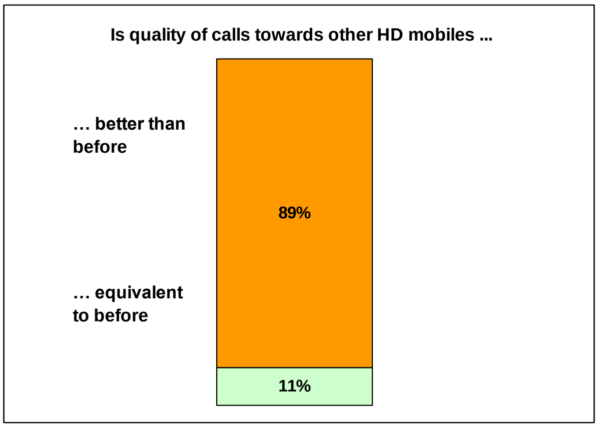


Figure 4: Orange Friendly User Test-Question 2

The satisfaction ratio was particularly high (96%), more than two thirds (68%) of testers were very satisfied. According to the test participants (spontaneous answers), the main strengths of HD Voice were the following:

For 92% of participants, the HD Voice experience met their expectations. Amongst the eight per cent (8%) of the negative answers, some of them resulted from non voice related problems (for example device related). HD Voice is an eagerly anticipated evolution of the mobile network. More than 50% of participants expect their mobile operator to offer it and three quarters (76%) are ready to change their device to benefit from HD Voice.

- Voice clarity: “Sound Quality is excellent!”, “Voice quality is impressive, and I'll miss it when getting back my old phone”.
- Great difference with respect to legacy narrow band voice: “Difference with respects to legacy sound quality is well perceptible”; “Listening quality much more pleasant than with conventional system”.
- Noise was reduced, remote talker voice was reinforced: “Ambient noise disappears and you only hear remote talker voice and nothing else”; “No ambient noise, you have the feeling that only the remote talker and you are on line”.
- Closeness sensation: “Person seems to be nearby”; “You should almost look for the remote talker in front of you”.

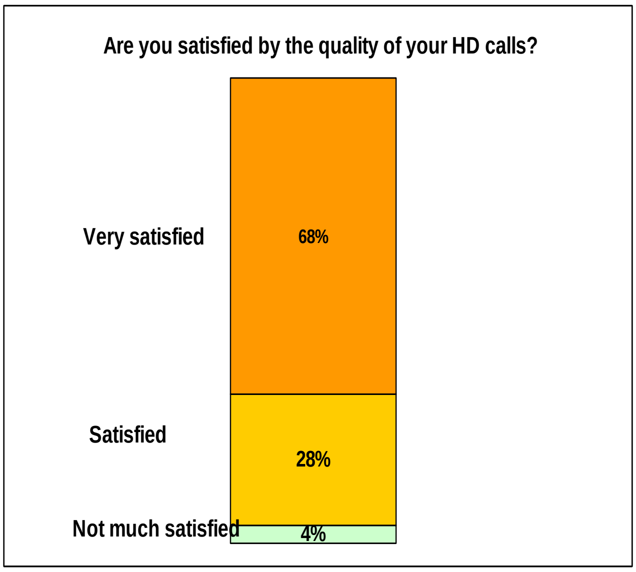


Figure 5: Orange Friendly User Test-Question 3

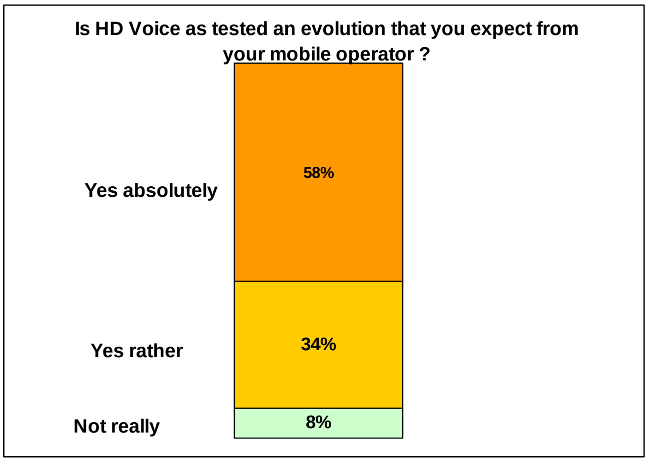


Figure 6: Orange Friendly User Test-Question 4

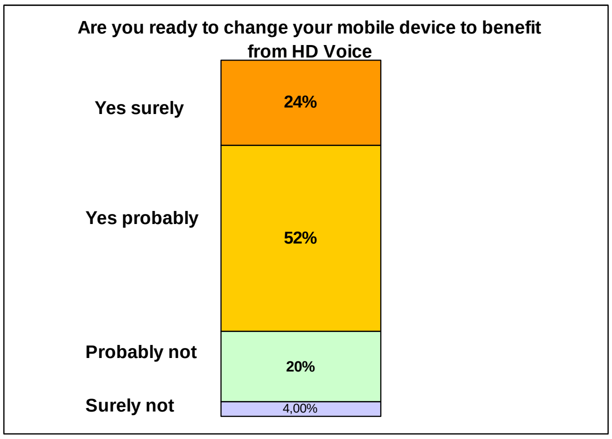


Figure 7: Orange Friendly User Test-Question 5

3 The wideband voice encoding with AMR-WB

3.1 Technology

As with AMR-NB, AMR-WB employs algebraic code-excited linear prediction (ACELP) technology, a sparse pulse train and a periodic signal component excite a linear predictive coding (LPC) synthesis filter. The LPC synthesis filter, in turn, generates the output voice signal. Besides a particular DTX/comfort noise mode, AMR-WB comprises nine coding rates.

AMR-WB requires a higher processing power compared to AMR-NB, however, High Speed Packet Access (HSPA) capable chipsets have sufficient processing power to also process the AMR-WB speech. Therefore, the additional cost for the inclusion of AMR-WB to the chipset is negligible for a dual mode 2G/3G device.

3.2 AMR-WB Operating modes

AMR-WB can operate in a number of modes as shown in the table below (from [3GPP TS 26.171]):

Codec mode	Source codec bit-rate
AMR-WB_23.85	23.85 Kbit/s
AMR-WB_23.05	23.05 Kbit/s
AMR-WB_19.85	19.85 Kbit/s
AMR-WB_18.25	18.25 Kbit/s
AMR-WB_15.85	15.85 Kbit/s
AMR-WB_14.25	14.25 Kbit/s
AMR-WB_12.65	12.65 Kbit/s
AMR-WB_8.85	8.85 Kbit/s
AMR-WB_6.60	6.60 Kbit/s

Table 1: Source bit-rates of the AMR-WB codec

In addition to these nine modes for active speech the AMR-WB codec uses WB-SID frames every 160ms to control Comfort Noise generation during speech pauses, where otherwise nothing is transmitted.

The AMR-WB configurations specified for 2G and 3G are:

- WB-Set 0 = {12.65 8.85 6.60} for all radio access technologies (GSM, EDGE, UTRAN).
- WB-Set 2 = {15.85 12.65 8.85 6.60} for UTRAN and for EDGE.
- WB-Set 4 = {23.85 12.65 8.85 6.60} for UTRAN and for EDGE.

No other combination of the nine AMR-WB modes is allowed for CS voice telephony. Out of the codec modes given in table 1 only five codec rates are used for CS speech and these are 6.6, 8.85, 12.65, 15.85 and 23.85 kbps. The support of these five codec modes is mandatory for the CS-UEs. The other modes of AMR-WB may be used for other applications.

All three AMR-WB configurations are Transcoder-Free Operation (TrFO) compatible. WB-Set 0 is the guaranteed minimum common denominator, and mandatory for all configurations. This Configuration also includes DTX, WB-SID frames and no data transmission during inactive speech.

As shown in the diagram in section 3.3, WB-Set 0 with bit rates limited to 12.65 kbps provides sufficiently high quality improvements. The low bit rates of 6.6 and 8.85 kbps intend providing a higher robustness to transmission errors in poor radio conditions.

It can be noted that AMR-WB has been designed to operate with WB-Set 0, approximately the same bit rate range as NB-Set 1 = {12.2 7.40 5.90 4.75} for AMR-NB. Both Codecs, AMR-WB and AMR-NB, follow very similar network design rules and both provide very similar network capacity and network transport costs.

3.3 AMR-WB quality

3.3.1 "Intrinsic" quality

The MOS (Mean Opinion Score) scale is one of the most commonly used scales to assess the quality of voice codecs in the telecom industry. It is based on a subjective testing methodology standardized in ITU-T P800 recommendation. MOS scores range from one (1) (worse quality) to five (5) (excellent). It is to be noted that MOS scores are suitable for ranking Codecs from within one MOS test, but they should generally not be used for comparison of results between different MOS tests.

To assess the voice quality improvement provided by wideband codecs with respect to narrow band ones, this MOS scale has recently been extended for subjective tests mixing narrow band and wideband conditions within one MOS test. Most of mixed subjective test results published in the literature or delivered by standardization organizations like ITU-T show wideband voice scores 0.5 to 1 MOS greater than narrow band voice scores. The widely experienced "PSTN" narrow band quality (G.711 PCM encoding at 64 kbps) gets within these mixed tests a MOS score not exceeding 3.5 to 3.7, whereas it receives a very good MOS score of 4.4 to 4.5 in narrow band only tests. Customers exposed to wideband speech rate (perfect) narrow band speech as being of lower quality.

The official ITU-T test results in [32] (fig 8) confirm the major improvement of perceived voice quality provided by the bandwidth extension from narrowband to wideband. Results in fig-8 (extract from experiment 1b) show that without taking into account any codec impairment, non-coded wideband voice (original signal) scores 1 MOS point greater than non-coded narrow band voice. It is still 0.5 MOS greater if voice is encoded with ITU-T-G.722 WB codec (at 56kbps).

Results from AMR-WB Characterization tests reported in [33] (fig 9) show that at the much lower bit rate of 12.65 kbps AMR-WB provides the same high wideband quality than G.722 used at its maximum bit rate of 64 kbps.

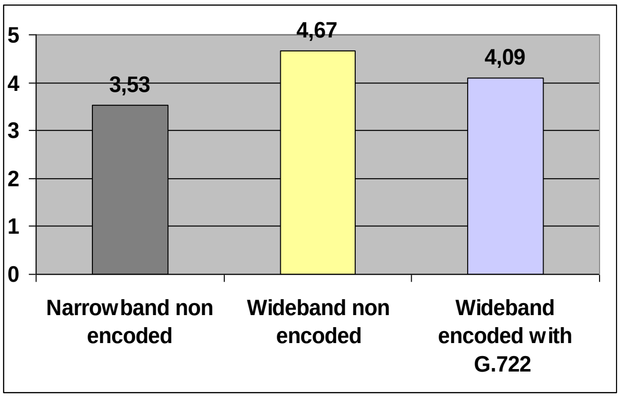


Figure 8Extract from experiment 1b [32]

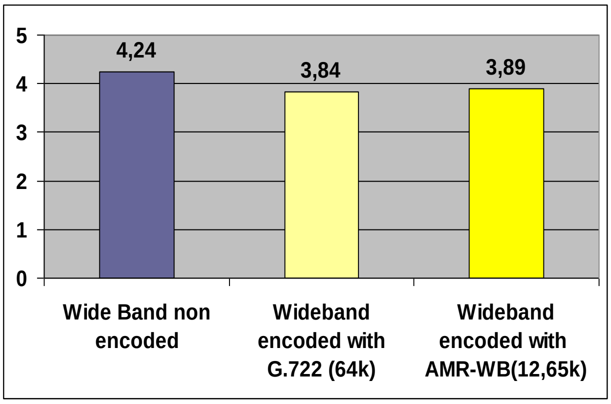


Figure 9: Extract from experiment 1 [33]

The results of subjective tests directly comparing AMR-NB and AMR-WB have also been reported in the article [34] from Nokia at ICASSP 2010. A MOS scale extended to nine categories was used from excellent to very bad. Results in figure 10 show that AMR-WB at 12.65 kbps scores very close to six compared to AMR at below 4.5 and non-coded NB below five.

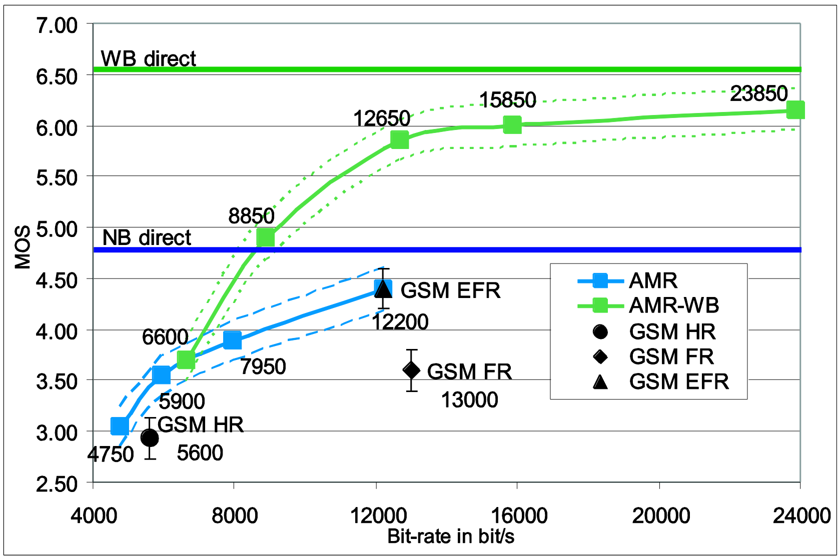


Figure 10: AMR and AMR-WB comparison with older 3GPP codecs

3.3.2 Results over Orange 3G live network

Before deployments of AMR-WB over Orange 3G networks, several tests were performed in controlled environments (lab tests) and over the live deployed networks (field tests). They aimed to verify that the expected quality increase provided by AMR-WB - with respect to AMR - is well preserved once integrated into a complete network infrastructure. For this purpose, live narrow band (AMR codec @ 12.2 kbps with TrFO activated) and AMR-WB calls between two mobile devices were electrically recorded through modified pedestrian kits. These recordings contain all effects of a real life network, except the acoustic output of the device used for recording.

Recordings are used to derive either an "objective MOS score" named MOS-LQO Listening Quality Objective) or a "subjective MOS score" namely MOS-LQS. MOS-LQO score is computed according to ITU-T P.862.2 standard. It is recognised that P.862.2 does not evaluate AMR-WB correctly (underestimation by more than 0.5 MOS with respect to subjective tests) however, it enables a relatively decent evaluation of the quality.

MOS-LQS scores are obtained by off-line subjective tests. In the subsequent sub-sections, these tests are ACR type (reference) with up to 32 listeners.

Whatever the method chosen, clean speech sequences are used. They include different male and female voices (French language).

The evaluated quality depends on the different rates selected by the network during the call. To control these changes during the tests, the Rate Control feature is deactivated and the Maximum Bit Rate is forced at the value to be tested. However, the device can decide to modify the uplink rate according to its estimation of the radio conditions. On the commercial devices used, this feature called autonomous mode cannot be deactivated nor controlled. Incidentally, the actual rate varies between the Maximum Bit Rate and the minimum value (6.6 kbps). Results presented below for a given rate – for instance 12.65 kbps – actually include a part of frames coded at lower rates – for instance 6.6 kbps – in particular when radio conditions are degraded.

3.3.2.1 Controlled environment- Orange Lab tests

Before activation over commercial networks, this phase enables testing of the complete infrastructure (CORE and UTRAN) to be deployed. Different radio conditions were considered thanks to a test bench that reproduced controlled UMTS radio channels including propagation profiles and distance between the UE and the Node B. The following figures correspond to the propagation profile "Pedestrian @ 3kmph" (PA-3), which is one of the most common profiles used to characterize mobile voice applications. Similar results have been obtained for other profiles. The main results of this test are:

In Nominal radio conditions it can be seen from figure 11 that:

- The difference between narrow band and wide band is approx. 0.9 MOS-LQS. This difference is relatively the same for all network provider infrastructures that were tested. This confirms the reliability of AMR-WB specification and implementation.
- The same MOS scores were obtained whatever the direction (Uplink or Downlink).
- By forcing the AMR-WB rate, it was possible over one network infrastructure to observe its influence on the perceived quality. The results were similarly obtained when testing the "codec only". Namely:

On clean speech, the best wideband quality is achieved from 12.65 kbps.

At 6.6 kbps, AMR WB quality is close to narrow band AMR one at 12.2 kbps.

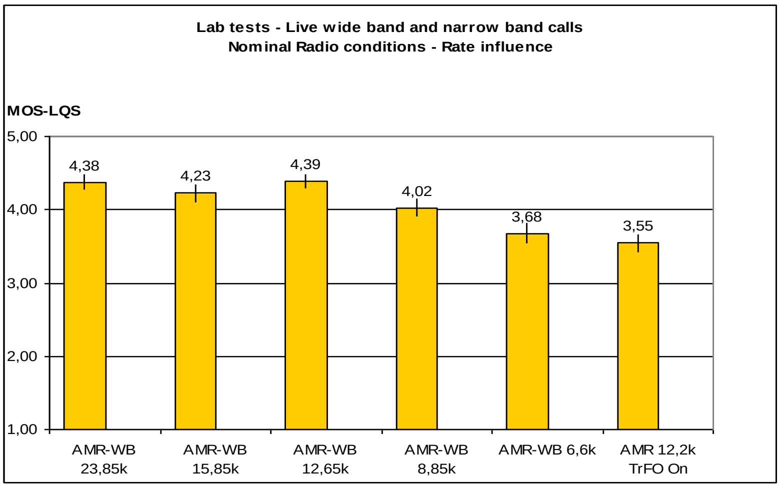


Figure 11: Live WB and NB calls – Rate influence

On degraded radio conditions corresponding to the cell edge (path loss of 142 dB uplink and 120 dB downlink), it can be seen from figure 12 that:

- AMR-WB performance is the same at 12.65 kbps and 8.85 kbps rates.
- At 6.6 kbps, AMR-WB is scored a bit lower than AMR at 12.2 kbps. In terms of perceived quality, 6.6 kbps rate does not show any real interest over 3G Network.

As mentioned above, due to device autonomous mode, recorded speech sequences include frames coded at lower rates.

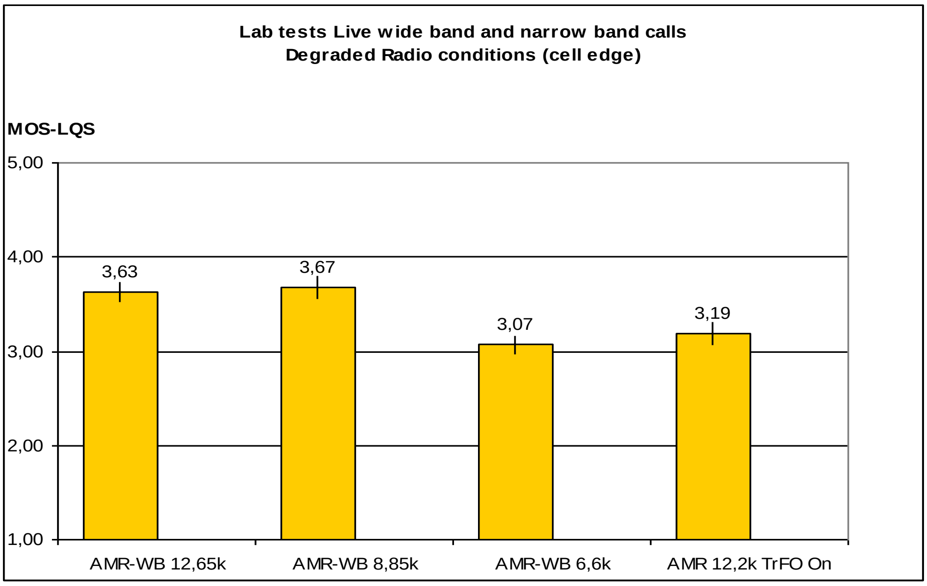


Figure 12: Live WB and NB calls – Degraded Radio Conditions

A complete analysis of radio channel impact was made with regards to objective measurements based onto P.862.2 standards.

Considering the Uplink System (from device to network) and varying the Path Loss (PL) by means of the radio test bench, it can be seen from figure 13 that:

- Objective MOS are logically positioned according to the codec rate, and remain stable up to a certain distance from the Node B, at which point they decrease quite rapidly.
- This fall is not only due to the radio degradation. As explained above, because of UE's autonomous mode, some frames are degraded due to coding at lower rates: Therefore, the curve marked as “12.65” is in reality the resulting envelope of 12.65 (with good conditions) plus 8.85 at medium conditions plus 6.60 at worst radio conditions. This explains why the four curves converge in the last point when the radio channel is strongly deteriorated, since most of the frames are then encoded at 6.6 kbps.

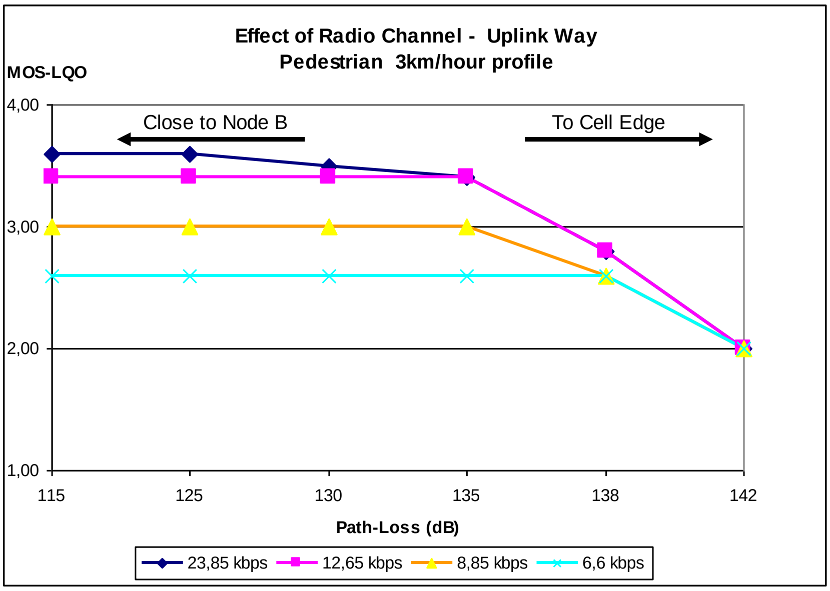


Figure 13: Effect of Radio Channel – Uplink Way – Pedestrian 3km/h profile

Considering the Downlink System (from network to device) it can be seen from figure 14 that:

- Objective MOS decrease is more regular, when moving away from the serving Node B. This different behaviour with respects to Uplink results from a smaller range of the power control.
- Objective MOS are ordered according to the rates of the codec modes, but MOS for 23.85 kbps mode decreases faster, and its advantage over 12.65 kbps disappears quite rapidly.
- Quality of 12.65 kbps mode remains always higher than 8.85 kbps one. The same holds for 8.85 kbps mode versus 6.6 kbps.

Since Downlink is not considered by the device autonomous mode, the displayed MOS scores correspond entirely to the indicated rate.

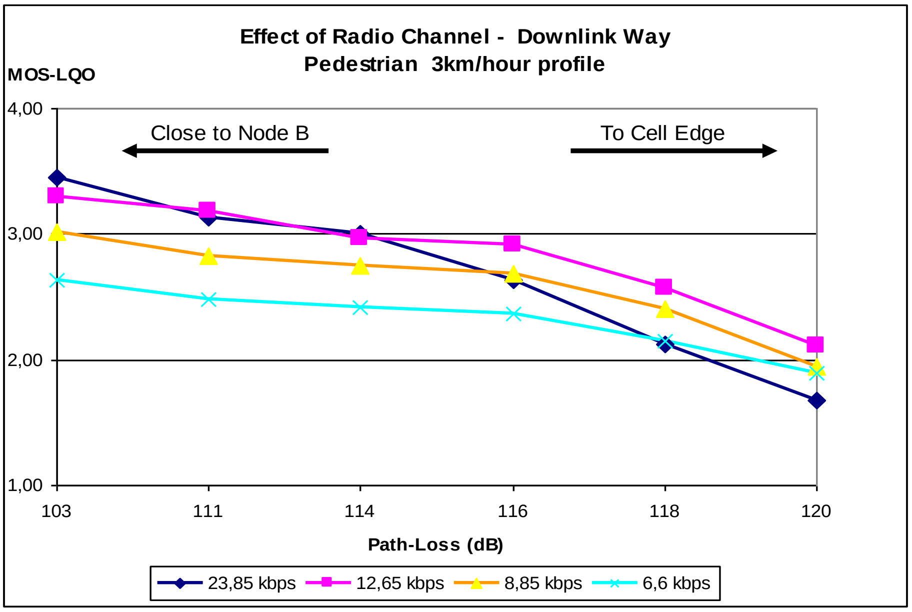


Figure 14: Effect of Radio Channel – Downlink Way – Pedestrian 3km/h profile

From these live tests, it can be confirmed that:

- Expected quality improvement with respect to AMR is verified (about 1.0 MOS).
- Rates higher than 12.65 kbps do not show any interest in terms of perceived voice quality.
- Over degraded radio channels, low rates (8.85 kbps and 6.6 kbps) do not yield higher MOS scores than 12.65 kbps. Therefore Rate Control for 3G does not seem to have a real interest. When UE drives out of this coverage, with path loss degradations exceeding the defined values for cell edge, the UE performs normally a Hand Over to GSM therefore there is no further requirement to study path loss degradations.
- The 12.65 kbps rate provides the best quality / capacity compromise.

3.3.2.2 Field test – Customer configuration

During the first AMR-WB deployment over 3G in Moldova (Orange, September 2009), several tests and measurements were undertaken on the live network. This included drive tests corresponding to different propagation conditions (pedestrian, vehicular – static, mobility – rural, urban) and to different call cases (inter or intra MSC, Handovers, etc.). Voice quality was evaluated either through off-line subjective tests (based onto recordings as explained above) or by objective measurements P.862.2 based. A similar behaviour as for the lab tests reported in section 3.3.2.1 could be observed. More precisely:

- The perceived difference between narrow band and wide band calls is close to 1.0 MOS.
- The perceived quality is the same whatever the types of environment and user mobility profiles are.
- In mobility conditions, the time variation of the MOS-LQO scores is quite small which means that quality is not physically affected by mobility.

- Also the analysis of recordings shows that handovers have the same effect as for narrow band AMR calls (short silence period lower than 100 ms).

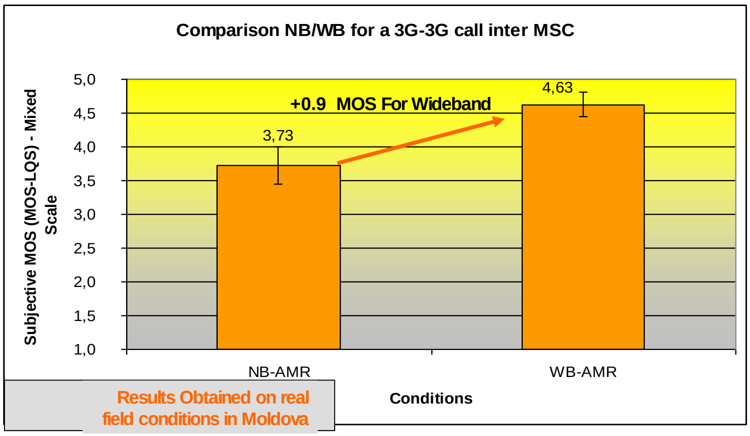
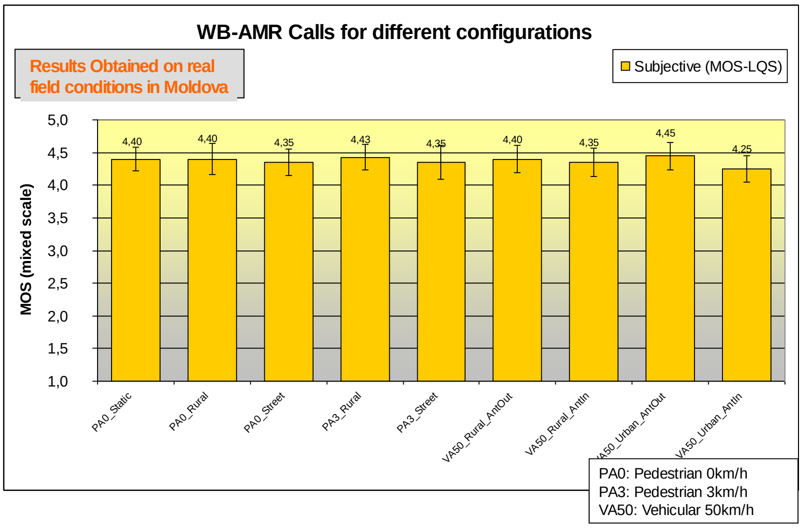


Table 2: Real field conditions – comparison NB/WB for a 3G-3G call inter MSC



Field tests confirmed good performance of AMR-WB over 3G network**.**

Table 3: Real field conditions – WB-AMR calls for different configurations

3.4 AMR-WB in standard recommendations

The AMR-WB codec has been standardized by 3GPP (SA4 as main body) and it consists of a full set of technical specifications covering all codec features (see reference [1]). Its usage has also been standardized by 3GPP and ETSI for a wide range of voice and multimedia services [2]. AMR-WB has also been standardized by ITU-T (SG16) as "G.722.2" for any usage including fixed networks and services, which means AMR-WB is the only voice codec standardized both by 3GPP and ITU-T.

The next generation communication codec EVS [41 for still enhanced voice services over LTE/EPS includes AMR-WB as an integral part. This ensures full backward interoperability with legacy AMR-WB environment (see [3]).

3.4.1 AMR-WB codec standard recommendations

AMR-WB is a fully specified voice coding and decoding scheme with a level of specification that ensures a precise implementation and a guaranteed level of encoding/decoding quality. In addition to the high level [1] and detailed [4] descriptions of the codec algorithm, the Software (C Code) simulating the implementations of both encoder and decoder on fixed and floating point Digital Signal Processors is provided within the standard [2, 11] together with the bit stream format [9]. A set of test sequences [3] is also provided to check the conformance (bit exactness) of the AMR-WB implementations.

Performance requirements have been evaluated during the 3GPP codec selection process and the quality of the resulting AMR-WB standard has been fully and extensively characterized in several subjective test labs for a wide range of conditions and in several languages as reported in [25] and [26].

Other related features to extend and optimize AMR-WB usage are specified. This includes for instance, the capability to operate with discontinuous transmission (DTX) [7]: the Voice Activity Detector [8] can be activated to detect non speech periods and the bit rate can be saved by suspending the transmission of encoded speech parameters and generating comfort noise at the decoder instead [6]. An error concealment algorithm [5] to limit the quality degradation in case of lost frames or packets (reconstruction of the lost parts of the signal at the decoding side) is also specified (informative).

Similar recommendations can be found in ITU-T G.722.2 standard and related Annexes [12]

3.4.2 Recommendation of AMR-WB format in network and service standards

AMR-WB is specified by 3GPP as the mandatory codec to be supported by terminals for a wide range of services when wideband speech sampled at 16 kHz is used. This includes circuit switched telephony [13] and video telephony services (3G.324M) [14], packet-switched conversational multimedia applications [15], Multimedia Messaging Service (MMS) [16], Packet-switched Streaming Service (PSS) [17], Multimedia Broadcast/Multicast Service (MBMS) [18], IMS Multimedia Telephony (MMTel) [19], Messaging , Presence, and Push-to-talk over Cellular (PoC)[20].

Specifically, usage of AMR-WB for Voice Over IP/IMS Networks and inter working between networks is covered by the standards in 3GPP TS 26.114 on Multimedia Telephony over IP/IMS Networks [19] in relation with the RTP payload format specified in IETF RFC 4867 [21]. IMS profile for Voice and SMS defined by GSMA in IR.92 [39] also specifies AMR-WB as the codec for wideband voice services.

For fixed IP/NGN Networks, AMR-WB has been specified by ETSI/TISPAN as the mandatory wideband codec for wideband telephony services in terminals originating and terminating end to end IP media flows in NGN [22] and for customer network devices [23]. In addition, it is recommended that network equipment supporting wideband audio should provide AMR-WB/G.722.2 to support 3GPP inter-working [22]. AMR-WB is also specified by I3Forum (International Interconnection Forum for services over IP) to be supported between IP Carriers for IP interconnected international voice services [28]. A related White Paper on "Optimal Codec Selection in International IP based Voice Networks" [29] gives more details and guidance on voice codec issues including wideband voice quality benefits and measurement. The use of AMR-WB is also specified by OIPF as audio format for IPTV and voice and video telephony services [27].

3.4.3 Beyond AMR-WB: improved HD voice and new HD voice+ with EVS

3GPP/TSG-SA4 has standardized in Release 12 the EVS (Enhanced Voice Service) with an even higher quality of user experience than is currently possible with the AMR-WB codec [24]. EVS codec provides extended features like jitter buffer management and improves quality for speech and non-pure voice signals (for example mixed with music or in a noisy environment). It extends audio bandwidth modes from narrowband and wideband, to super-wideband or full band. EVS codec also utilizes a significant improved error robustness [42].

A general overview of EVS codec is given in 3GPP TS 26.441 [41] which includes references to all other EVS related specifications. Especially, detailed information on the performance of EVS with respect to AMR and AMR-WB can be found in 3GPP TR 26.952 [42] that reports extensive results from the subjective quality tests conducted during EVS standardization process.

HD Voice for LTE mobile terminals and networks has been extended to comprise of the EVS codec in GSMA TS.23 Annex F of HD Voice Logo Technical Annexes [43].

HD Voice+ (High Definition Voice+) for LTE mobile terminals and networks comprises of the EVS codec operated in Super-Wideband (SWB) or Fullband (FB) modes and the enhancements to terminals and networks according to the requirements defined in GSMA TS.23 Annex G of HD Voice Logo Technical Annexes [43].

While the service enhancements enabled with the EVS codec is highly desirable, 3GPP has also recognized the necessity of interoperability to legacy AMR-WB-based wideband telephony systems and the advantages of solutions that avoid transcoding to the largest possible extent. An improved but still bitstream interoperable AMR-WB mode (EVS AMR-WB IO) is therefore an integral part of the EVS codec, such that the EVS codec can natively interoperate with the existing AMR-WB codec without any transcoding. This ensures interoperability and will also greatly facilitate the future deployment of the EVS codec. If the EVS codec is supported, then the EVS AMR-WB IO mode of operation may also be used as an alternative implementation of AMR-WB codec. Equipment supporting AMR-WB today are future-proof in that they already support a subset of the EVS codec and vice-versa the EVS codec will (at any time) be able to interoperate with legacy AMR-WB equipment. The additional features provided with the EVS codec support the use of AMR-WB and they may be used if both communication end points support them. Otherwise, AMR-WB will be used as the fallback.

4 System aspects

This section investigates, in detail the functions necessary to support the introduction of AMR-WB codec.

4.1 Network impacts

Sections 4.1.2 and 4.1.3 respectively focus on the Session Establishment for VoLTE and for 2G/3G Circuit Switched Networks.

4.1.1 Impact on core network: Transcoding free operation modes (TFO, TrFO)

Ordinarily, the voice payload for transport in the core network is PCM-coded at 64kbps according to ITU-T recommendation G.711. AMR-NB must thus be transcoded to and from PCM, which costs in terms of voice quality (degradation) and signal processing (greater complexity). A corresponding PCM-based transport cannot be used with AMR-WB telephony, as G.711 applies only to narrowband voice. For AMR-WB telephony two complementary solutions for this problem were specified in the 3GPP standards: Tandem-Free Operation (TFO) and Transcoder-Free Operation (TrFO).

TFO is an in-band signalling protocol that allows voice codec parameters to pass unmodified through the PCM links in traditional core networks. The TFO protocol provides transcoding free operation and therefore preserves voice quality; however TFO does not reduce the transport bit rate inside the core network. TrFO is a combination of out-of-band control signalling (OoBTC) and enhanced transport technology (via ATM or IP) for the voice payload. At call setup, OoBTC negotiates the optimum codec types for the radio and core networks and allocates the necessary processing and transport resources. The objective of both, TFO and TrFO, is transcoding-free operation, end-to-end. Meaning the voice signal is encoded in the transmitting mobile terminal and transported without modifications to the receiving mobile terminal. A wideband voice connection cannot be set up for calls placed to a traditional PSTN phone. Instead, OoBTC negotiation of TrFO ensures that AMR-NB coding with G.711 transcoding is employed at the edge to the PSTN.

While TFO is typically used in 2G systems, TrFO is used in 3G and might also be used for 2G for inter Media Gateways transmissions.

The latest standardisation for 2G resulted in “AoIP” (speak: “A-Interface User Plane transmission over the Internet Protocol”). AoIP allows TrFO also for 2G calls and TFO is then no longer necessary. The resulting voice quality is identical, the equipment and operational costs are, however, substantially reduced.

The AoIP standard was completed in 2008 and products (2G BSS and MGW) are available on the market.

4.1.2 Session Establishment for VoLTE

4.1.2.1 General

Section 4.1 investigates the functions necessary in the core network by looking at information flows for the following use cases:

1. VoLTE to VoLTE call,
2. PSTN/UMTS to VoLTE Call and
3. VoLTE to PSTN/UMTS Call

It is assumed that VoLTE UE is IR.92 compliant.

4.1.2.2 Scenario 1: VoLTE - VoLTE

Table 2 shows the Network Impacts for this scenario.

Entity	Impacts
IMS Entities	
P-CSCF	- A check whether or not to allow AMR-WB - A function to indicate PCRF the Codec capability of the UE
S-CSCF	- A check whether or not to allow AMR-WB media
I-CSCF	- A check whether or not to allow AMR-WB media
SLF	- None
TAS	- None
EPS Entities	
HSS	- None
PGW	- None
SGW	- None
PCRF	- A function to decide the use of AMR-WB and derive QoS/Bitrate values for AMR-WB
MME	- None
eNB	- None

Table 4: Impacts for VoLTE-VoLTE Call

4.1.2.3 Scenario 2: VoLTE - PSTN/UMTS

Scenario 2-1: VoLTE - PSTN/ G.711 UMTS:

Entity	Impacts
IMS Entities	
P-CSCF	- Same as described in Table 2
S-CSCF	
SLF	
TAS	
MGCF	- Setting Codec Capabilities to Initial SDP Offer - Selecting Codec based on the ones included in SDP Answer - A capability to exchange AMR-WB related information with MGW using H.248 protocol.
MGW	- A capability to exchange AMR-WB related information with MGCF using H.248 protocol. - Transcoding capability between AMR-WB and G.711
PSTN/UMTS Entities	
PSTN/UMTS	- None

Table 5: Impacts for VoLTE - G.711 PSTN/UMTS Call

If the G.711 codec is used in PSTN/UMTS interworking, then the overall voice quality will be limited to that of G.711, even if AMR-WB is used in the VoLTE network. If this is not desirable, for example because more bandwidth is used despite the lower voice quality, a measure must be taken in the Media Gateway Controller Function (MGCF)/MGW to ensure the use of AMR-NB. For example, depending on the supported codec in the terminating network, MGCF does not indicate AMR-WB in the SDP Answer to UE.

Scenario 2-2: VoLTE - BICC/SIP-I UMTS:

In this scenario, the codec negotiation between the VoLTE Network and UTMS takes place. MGCF and UMTS must then negotiate codec capability by using OoBTC via BICC or via SIP-I.

If both, VoLTE Network and UMTS, support AMR-WB, then AMR-WB can be used end-to-end, that is, transcoding will not take place in MGWs and wideband voice quality is achieved. UMTS includes both, 3G access via CS-UTRAN and 2G access via TFO or AoIP. Rate Control is specified and operational end to end between VoLTE and UMTS.

See section 4.1.3 for the required network impacts to support AMR-WB in BICC/SIP-I UMTS.

4.1.2.4 Scenario 3: PSTN - VoLTE

The specified Codec negotiation via OoBTC in UMTS and SIP in VoLTE is fully symmetrical with respect to the call setup direction.

A call from VoLTE to PSTN results in the identical Codec Selection as a call from PSTN to VoLTE. In both cases PCM is used at PSTN side and therefore a narrowband Codec is the best choice in VoLTE.

Similarly a call from VoLTE to UMTS achieves the same results as a call from UMTS to VoLTE.
If AMR-WB is supported on all involved networks links and accesses, then AMR-WB achieves the best voice quality.

4.1.2.5 Scenario 4: Early Media for AMR-WB

If AMR-WB is used for Early Media (for example Pre-call announcements and Customised Alerting Tone), then in addition to the impacts specified above, there are the following network impacts:

- Audio files to be played need to be encoded by the AMR-WB codec
  Note: it is not strictly necessary that the audio files are stored in wideband quality. For a transient time is may be permissible to use the existing narrowband material, but encode and transmit it in AMR-WB format to the UE.
- MRF and AS (for example CAT-AS) need to be able to indicate the use of AMR-WB during the SDP exchange between the UE for the Early Media session.

4.1.3 Session Establishment for 2G/3G Circuit Switched NW

For the 2G/3G CS domain network to provide AMR-WB without any transcoding at MGW, the network and the UE must be compliant to 3GPP Rel-4 or later specifications.

Pre Rel-4 network and UE do not provide a means to exchange the codec capability. In this instance, the MSC (or MGW) must provide the transcoding with G.711.

One option to provide AMR-WB over 2G/3G is that the CS domain core network supports the Bearer Independent Core Network, as specified in 3GPP TS23.205. The other option is that the CS domain core network supports the SIP-I based circuit-switched core network as specified in 3GPP TS23.231.

The network is also required to support the Transcoder Free Operation as specified in TS23.153. All MSC Servers in the network must support the codec negotiation mechanism to use AMR-WB. All RNCs and MSC Servers in the network also require the capability to exchange AMR-WB related information in the Iu-UP initialisation procedures.

4.1.4 Summary of network impacts

The necessary functions to support AMR-WB in VoLTE networks can be summarised as follows:

Entity	Impacts
IMS Entities	
P-CSCF	- A check whether or not to allow AMR-WB - A function to indicate PCRF the Codec capability of the UE
S-CSCF	- A check whether or not to allow AMR-WB media
I-CSCF	- A check whether or not to allow AMR-WB media
SLF	- None
TAS	For Early Media, Audio files to be played need to be encoded by AMR-WB codec Capability to indicate the use of AMR-WB during the SDP exchange between the UE for the Early Media session.
MRF	Capability to indicate the use of AMR-WB during the SDP exchange between the UE for the Early Media session.
MGCF	- Setting Codec Capabilities to Initial SDP Offer - Selecting Codec based on the ones included in SDP Answer - A capability to exchange AMR-WB related information with MGW using H.248 protocol.
MGW	- A capability to exchange AMR-WB related information with MGCF using H.248 protocol. - Transcoding capability between AMR-WB and G.711
EPS Entities	
HSS	- None
PGW	- None
SGW	- None
PCRF	- A function to decide the use of AMR-WB and derive QoS/Bitrate values for AMR-WB
MME	- None
eNB	- None

Table 6: Summary of VoLTE Network Impacts

The necessary functions to support AMR-WB in 2G/3G CS networks can be summarised as follows:

Entity	Impacts
HSS	- None
MSC Server	- All MSC Servers in the NW must be compliant to 3GPP Rel-4 or later - All MSC Servers in the NW must support the OoBTC protocol, either via BICC or SIP-I - All MSC Servers in the network have to support the codec negotiation mechanism to use AMR-WB.
MGW	- All MGWs in the NW must be compliant to 3GPP Rel-4 or later - AMR-WB transcoding capability
RNC	- All RNCs must be compliant to 3GPP Rel-4 or later - The capability to exchange AMR-WB related information in the Iu-UP initialisation procedures.
BSS	- All BSS must be compliant to 3GPP Rel-4 or later - The capability to exchange AMR-WB related information in the A-Interface initialisation procedures.

Table 7: Summary of 2G/3G CS Network Impacts

4.2 Interoperability aspects

4.2.1 Interoperability between 3GPP and 3GPP2

AMR-WB enables interoperability between 3GPP and 3GPP2 (CDMA2000) mobile systems. VMR-WB as the first 3GPP2/TIA wideband codec that was standardized as 3GPP2 C.S0052-0 v1.0 and, respectively, TIA-1016, provides one mode of operation (mode 3) that is interoperable with AMR-WB (at 12.65 kbps and below). The actual wideband speech service interoperability between 3GPP and 3GPP2 systems depends on the degree of deployment of these codecs in the respective mobile systems.

4.2.2 Interworking between AMR-WB in mobile networks and wideband voice codecs in DECT systems

As defined at DECT Forum by CAT-iq standard [40] the typical wideband speech codec in DECT is the G.722 codec. G.722 is a fixed rate codec of 64 kbps and consumes much more transmission bandwidth than AMR-WB. An interworking between these codecs is possible by the implementation of a transcoder at the network border which provides a direct transcoding between the AMR-WB (G722.2) and the G722 codec.

4.3 Terminal aspects

Wide band terminals must support the AMR-WB codec according to the standard recommendation as described in section 3.4 and including DTX and Packet loss concealment. They must also support all features and signalling capabilities for call establishment and call control related to system aspects as described in sections 4.1 (such as mechanisms associated with TrFO, Setting Codec Capabilities to Initial SDP Offer, selecting Codec based on the ones included in SDP Answer, TFCi / RFCi integration or to Rate Control and autonomous mode, etc.).

However, these features directly related to AMR-WB codec are not sufficient to provide a real end-to-end wide band experience to customers. Quality and performance of terminal acoustic and related speech processing functions have a key influence on the end-to-end voice quality experienced by the end user. This puts additional requirements on wideband terminals that are described below.

4.3.1 Acoustics and Speech processing

In wideband voice terminals, the entire chain must be wideband compliant. This includes acoustical elements, electronic components (AD/DA, etc.), voice enhancement processing modules for example, noise reduction and Acoustic Echo Cancellation. Digital processing modules have to function at 16 kHz sampling rate and must also comply with performance requirements suited for wideband voice quality.

The acoustical components (earpiece, microphone) play a determining role. Their qualities directly influence the customer perceived quality for all their calls (narrow band and wide band ones). For example, some distortion at the receiver side or a highly non-flat frequency response would produce a severe quality degradation that could not be corrected by signal processing. A well balanced global frequency response is consequently mandatory for preserving the voice fidelity.

Requirements for acoustical parts are defined by 3GPP in recommendation 3GPP TS 26.131 [35]. The related recommendation 3GPP TS 26.132 [36] specifies the test methodologies for evaluating those requirements both for narrowband and wideband (from Release 5) and for the different use modes (handset, headset, hands free, etc.). These recommendations have been recently updated in 3GPP releases to enforce the guarantee of a good acoustical quality in a consistent way for wideband and narrowband voice and further evolutions are planned for Release 10.

Furthermore, mobile devices generally embed some additional signal processing for example, noise reduction, acoustic echo cancellation and other technologies that aim at improving the speech quality either for the remote talker or for the local one. These technologies are generally known as VQE (Voice Quality Enhancement). Although not directly related to AMR WB, VQE issues are impacted by wideband speech. For example, at high frequencies, the Signal to Noise Ratio can be lower than in the narrowband part of the spectrum which results in a lower speech energy.

Although the noise perception is enhanced by wideband voice and AMR WB, which provides a better user experience in a noisy environment, it is preferable to provide an improved voice quality experience that uses devices equipped with VQE. Field tests (as described in section 4.2) show customer expectation for noise reduction who would not necessarily understand that wide band devices do not embed such VQE. As mobile calls can occur in noisy environments, the interest for an efficient noise suppressor is very high, not only for the customer but also for operators. The noise reduction challenge is to reduce as much as possible, the noise level while preserving the speech quality. This can be achieved by different technologies including the ones based around bi-microphone recordings which receive sound equally from both the front and back of the element. Unlike AMR WB codec, these technologies are not standardized which yields to a great variety of performance. Noise reduction quality from a user perspective can be evaluated through subjective tests as defined by ITU-T P.835 recommendation [37]. Such subjective tests are however time consuming and may not be compliant with mobile industry device delivery process and how the industry currently achieves the definition of objective and automatic measurements methods. This is currently being addressed by the standardisation bodies such as 3GPP SA4 and ITU-T SG12.

4.3.2 Wide band quality onto peripherals

Peripherals such as headsets or car kits are commonly used with mobile devices and are generally connected via a wireless Bluetooth link. Providing wideband speech over such peripherals requires those systems to be capable of transporting wideband speech (over the air interface if wireless) and have acoustical properties compliant with wideband requirements:

- Acoustical properties compliant with wideband requirements: handset mode, requirements and the associated measurement methods are defined in technical specifications 3GPP TS 26.131 [35] and 3GPP TS 26.132 [36]. For headset mode, the closeness of the ear makes it easier to render the whole spectrum (reduced leaks) and wide band rendering on headsets can provide a reasonable wideband quality experience.
- Transport of wideband speech using over the air interface (Bluetooth) and associated protocol to handle the wide band codec support: Bluetooth Sig, the Consortium in charge of Bluetooth standardization, has selected the "modified SBC wideband codec" to encode and transport wideband voice over Bluetooth air interface. The SBC codec is currently used for audio streaming via Bluetooth. The modified version deals with wide band speech (16 kHz sampling rate) and offers a lower latency. It is up to the terminal to perform the transcoding between AMR-WB and the modified SBC codec.

5 Executive Summary

High Definition Voice (HD Voice) offers users a greatly enhanced voice communication experience. It has been measured and proven that HD voice sounds more natural and intelligible, thus increasing the effect of presence and closeness. HD voice relies on the coding of an extended frequency range (wideband) of the speech signal and on "HD" Terminals supporting adequate acoustic performance and related speech processing functions.

The AMR-WB (or ITU-T G722.2) is the high quality wideband codec standardized by 3GPP (from Release 5) to offer HD voice services over 3GPP Mobile Networks. It is optimized for usage in a mobile environment with adaptable operating bitrates from 6.6 kbps to 23.85 kbps. For speech signals, a maximum bit rate of 12.65 kbps is recommended to achieve the best quality/capacity trade off. At such rate there is no requirement for increasing the network capacity required for traditional narrowband voice while fully benefiting from the wideband quality.

AMR-WB is an end to end solution: both UEs and the network need to enable it. In this case, the audio stream is passed transparently through the network thanks to TFO/TrFO. If end to end transmission of wideband voice is not possible, an automatic fallback to the next higher codec (for example AMR-NB) occurs. As such codec negotiation happens at call setup, it is fully transparent to the end user. Existing UMTS networks can be easily upgraded to support

AMR-WB by introducing the appropriate software release. For GSM networks, depending on the age of the equipment, systems can be upgraded at a reasonable cost.

HD voice with AMR-WB has been previously trialled by various operators and is currently launched in commercial networks in 3G mode. Commercial UEs with AMR-WB enabled by default and compliant to acoustics and audio requirements have entered the market since 2010.

A similar development for the mobile industry can be observed for fixed networks. Specifically, new generation wideband capable VoIP DECT terminals compliant with CAT-iq standard offer HD voice encoded with ITU-T G.722. To ensure an end-to-end HD voice interoperability between fixed and mobile, an interworking function must be added to transcode between the G.722 and AMR-WB formats.

HD voice mobile services are currently emerging and gaining momentum however future success will rely on three key factors:

1. The capability of telcos to ensure an end-to-end HD Voice interoperability, in particular between their fixed and mobile networks,
2. A wider choice of HD Voice terminals available on the market from all manufacturers,
3. An efficient adaptation of AMR-WB to new "LTE" Mobile Networks especially with the future EVS codec, which still extends the encoded voice bandwidth while preserving full backward interoperability with AMR-WB for wideband speech.

Annex A Annex A Information flows

To allow a detailed view on the function in the core network and to underline the description in section 4.1.2, three different information flows are attached.

A.1 Information Flow #1 - VoLTE Call

Originating Side: VoLTE, Terminating Side: VoLTE

H:\
Kopie_IREG_SIGNAL\

A.2 Information Flow #2

Originating Side: G.711 ISUP PSTN/UMTS, Terminating Side: VoLTE

H:\
Kopie_IREG_SIGNAL\

A.3 Information Flow #3

Originating PLMN: PSTN/UMTS, Terminating PLMN: VoLTE

H:\
Kopie_IREG_SIGNAL\

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

B.1 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 B .

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

B.2 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

B.3 Codec selection

When end-to-end AMR-WB is possible and end-to-end EVS is not possible, then either AMR-WB shall be selected with the highest priority by the network(s) , or alternatively, AMR-WB on one end with TrFO-interworking to EVS on the other end. 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.

B.3.1 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.

B.4 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 operator’s networks according to the network architecture (TDM, ATM or IP interfaces and transport).

B.4.1 Support of TFO / TrFO of 3rd party equipmentSupport 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.

B.5 Trancoding

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.

B.6 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 rate, etc. shall be at least as good as before the activation of AMR-WB.

B.7 Use Cases for HD Vioce

B.7.1 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

B.7.2 Setup of a voice call between two HD Voice devices, one being in AMR-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..

B.7.3 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.

B.7.4 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)

B.7.5 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 cannot 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.

B.8 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 C Minimum Network Requirements for HD Voice with CDMA2000

C.1 HD voice enabled CDMA 2k mobile networks

To support HD Voice, the operator shall support EVRC-NW in CDMA2000 networks as described in the rest of Annex C.

The voice service shall be compliant with 3GPP2 specifications related to 2G/3G Circuit Switched Telephony or packet switched telephony (C.S0050, C.S0055-0v1.0, C.S0085 and all other related specifications). .

It should be noted that the operator does not need to enable EVRC-NW on all its networks

C.2 EVRC-NW codec service

For CDMA2000 networks, at a minimum, the EVRC-NW codec service option 73 with capacity operating point 0 “COP 0” must be supported.

C.3 Codec selection

When end-to-end EVRC-NW is possible, it shall be selected with the highest priority by the network(s). If EVRC-NW 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.

C.3.1 Usage of lower bit rate codecs in CDMA2000 networks

If due to any reason like loaded cell a lower capacity operating point has to be used in the network, then the network might select EVRC-NW COP 1 to 7. This can be done preferably at the call setup phase or if necessary, during the call.

C.4 TFO / TrFO

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

C.4.1 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.

C.5 Transcoding

If transcoding is necessary between two systems providing a wideband voice codec (e.g. G.722 on the one side and EVRC-NW 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.

C.6 Impact on KPI values

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

C.7 Use Cases for HD Voice

C.7.1 Setup of a voice call between two EVRC-NW enabled devices

EVRC-NW call setup shall be performed

- If both devices are EVRC-NW COP 0 enabled
- If both devices are in a EVRC-NW COP 0 enabled area
- If the HLR entry does allow the user to use EVRC-NW COP 0 (note that an HLR entry is optional).
- If radio and other resources allow HD Voice end to end

C.7.2 Setup of a voice call between two HD Voice devices, one being in EVRC-NW 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..

C.7.3 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.

C.7.4 Hand Over within EVRC-NW coverage

Mobility within EVRC-NW coverage shall be supported without any audio defects (for example without higher. call drop rates, without extended interruption times compared to EVRC, EVRC-B). 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)

C.7.5 Hand Over between EVRC-NW enabled area and non EVRC-NW enabled area.

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

C.8 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 EVRC-NW shall be guaranteed at least in narrow band quality.

Annex D Minimum Network Requirements for HD Voice with LTE

Minimum network requirements for HD Voice with LTE are specified in GSMA TS.23 Annex F1 of HD Voice Logo Technical Annexes [43]

Annex E Minimum Network Requirements for HD Voice+ with LTE

Minimum network requirements for HD Voice+ with LTE are specified in GSMA TS.23 Annex G1 of HD Voice Logo Technical Annexes [43]

Annex F Document Management

F.1 Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
1.0	28.07.2011	Version for approval	IREG Signal	Vincent Danno Orange
1.0	December 2011	Submitted to DAG & EMC for approval, final approval date 28 December 2011	EMC	Vincent Danno Orange
2.0	29.10.2012	Include mCR Signal Doc 59_004rev1 Include MCR DAG Doc 99_014	EMC	Vincent Danno Orange
3.0	30.05.2014	CR IR36 T1 & CR1001 incorporated	IREG & PSMC	Vincent Danno Orange
4.0	06.02.2015	CR IR36 CR1002 incorporated	IREG & PSMC	Vincent Danno Orange
5.0	12/06/2020	Implementation of IR.36 CR1003	NG	Merieme El Orch / Orange

Other Information

Type	Description
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It is our intention to provide a quality product for your use. If you find any errors or omissions, please contact us with your comments. You may notify us at prd@gsma.com

Your comments or suggestions & questions are always welcome.