New high-quality voice service for mobile networks

Christina Birkehammar, Stefan Bruhn, Peter Eneroth, Karl Hellwig and Stefan Johansson

The ongoing evolution of wireless communication systems and mobile phones has given rise to a variety of compelling mobile applications (music player, camera, game console) and services (mobile internet, mobile TV, and so on). Likewise, many services have evolved significantly in order to satisfy user demands. In contrast, from a user perspective, voice telephony has not changed noticeably since mobile telephony was still very young. Notwithstanding, voice service has continued to evolve. Significant milestones include the introduction of the enhanced full-rate codec (EFR) and, later, the adaptive multirate (AMR) voice codec, which increased voice quality and boosted channel error robustness and capacity. The narrowband AMR (AMR-NB) codec, which supports the bandwidth of traditional telephony, is now widely deployed in GSM/EDGE and UMTS systems. It is also the codec of choice for the forthcoming multimedia telephony service for IMS (MTSI) standard from 3GPP.

As one would expect, AMR also continues to evolve. The new wideband AMR (AMR-WB) codec, whose voice frequency band is twice that of AMR-NB, enables telephony services with true, natural voice quality, clearly outperforming other existing mass-market telephony services, including those used for wireline telephony.

User and operator benefits of improved voice quality

Results of laboratory listening tests

A MUSHRA listening test carried out by Ericsson Research showed that users clearly prefer AMR-WB voice over uncoded narrowband voice. (MUSHRA: multi-stimulus test with hidden reference and anchor, ITU-R recommendation BS.1534.) In particular, AMR-WB is superior to the 64kbps pulse-code-modulation (PCM) coding of the public switched telephone network (PSTN).

On error-free channels, AMR-WB (even in its lowest mode at 6.6 kbps) outperformed every AMR-NB mode up to 12.2kbps (Figures 1-2). And on error-prone GSM/EDGE and UTRAN channels under normal operating conditions, AMR-WB was favored over AMR-NB.

It should be noted, however, that these results are only indicative of an artificial lab environment and testing with short voice samples. The results might thus be inconclusive when comparing voice service based on AMR-WB with that based on AMR-NB. As a consequence, to compile more evidence, Ericsson and T-Mobile International also conducted a joint consumer trial in Germany.

Consumer trial in live network

The joint consumer trial ran for four weeks during April and May 2006. Ericsson and T-Mobile equipped 150 end users with special AMR-WB mobile phones in order to study the effects of improved voice quality on end-user perception and behavior. The results of the trial are very encouraging (Figure 3). More than 70% of the participants perceived a distinct improvement in voice quality – they found that they could more easily place and complete calls in noisy environments, and reported that the improved voice quality created a greater sense of privacy, discretion and comfort. Accordingly, Ericsson anticipates that AMR-WB will lead to positive changes in calling patterns, generating substantially more mobile traffic, both in terms of number and duration of calls.

Initially, while AMR-WB terminal penetration is low, operators can boost income and strengthen their brands by providing special AMR-WB services with high voice quality, such as conference calls and announcements. They can also drive information services, ring-back signals, enhanced automatic voice recognition, and improved voice mail. In addition, they might offer AMR-WB telephony to limited user groups or as an enterprise solution. During the consumer trial, for instance, business users indicated that they have high expectations regarding voice quality, in particular because improved voice quality will help them to communicate important figures, reduce expenses, and leave a positive impression.

AMR-WB can also help operators to cut costs, for example, by reducing the cost of acquiring subscribers. A more satisfying voice service should also help reduce helpdesk costs.

The AMR-WB phone and consumer trial typifies the planned activities of Ericsson's Network Solution Area Characteristics unit (Business Networks), whose aim is to continuously enhance the end-to-end performance of Ericsson services and applications.

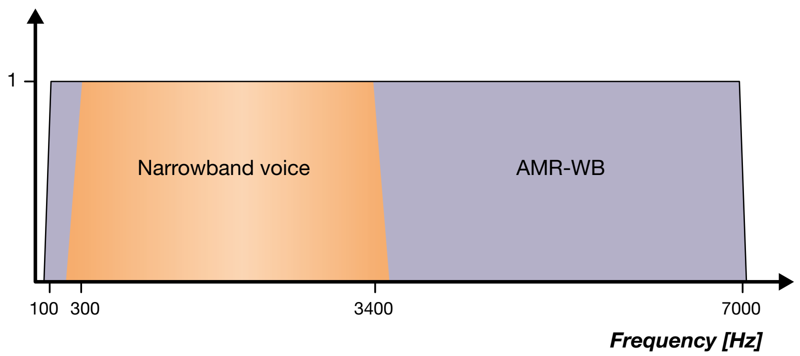


Figure 1
Typical passband characteristics of traditional narrowband voice and AMR-WB coded voice. Only AMR-WB encodes the perceptually relevant portion of the voice spectrum (up to 7kHz).

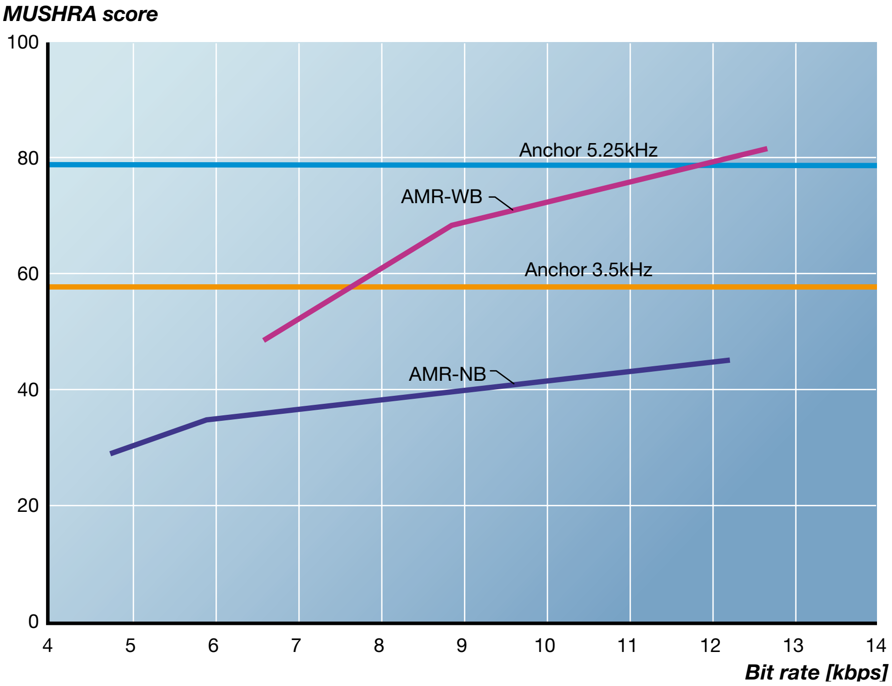


Figure 2
Voice quality on clean channel as a function of bit rate.

Roadmap

AMR-WB, which builds on operator requirements for GSM and UMTS networks, is an important candidate for deployment in Ericsson's upcoming mobile network release. It will be assigned "commercial" status in Ericsson's mobile platform deliveries from 2007Q1 and onward, and will be operational in GSM and UMTS. It is also positioned for inclusion in fixed networks with support for telephony, multimedia and video telephony and has been included in the Enterprise roadmap as part of MX-ONE version 4 (scheduled for launch in 2007Q3 (Table 1).

Technical details

Standardization

The 3GPP standardized the AMR-NB codec for voice telephony in 1998. AMR-WB standardization followed in 2000 and was finalized in 3GPP Rel-5. 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 signaling, and discontinuous transmission (DTX). Later, 3GPP Rel-6 introduced the extended AMR-WB codec, which complements the AMR codec family with a high-quality, low-rate, audio codec for messaging and streaming applications.

Technology

AMR-WB employs algebraic code-excited linear prediction (ACELP) technology, in which 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. The first three rates, 6.60, 8.85 and 12.65kbps, make up the mandatory multirate configuration (set 0) for wideband voice telephony. Two optional configurations with 15.85 or 23.85kbps modes have been defined for use with specific telephony applications, such as multiparty conferencing. These optional configurations are less important, however, because the associated enhancement is small for the price (much higher bit rate).

TABLE 1, ROADMAP OF PRODUCTS THAT SUPPORT AMR-WB

	Status	Release
Mobile Core	Candidate	MSS R5.1
UTRAN	Candidate	P6.0
GERAN	Candidate	BSS08A
EMP	Commercial	2007Q1, GSM/UMTS platform
MGW (fixed)	Candidate	MGW 2.0
Enterprise	Candidate	MX-ONE version 4

System aspects

Core network

The mobile core network (Figure 4) handles call setup, routing, and a variety of supplementary services. Ordinarily, unless the TFO or TrFO standards have been used, 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).

Analog PCM-based transport cannot be used with AMR-WB telephony because G.711 only applies to narrowband voice. Consequently, AMR-WB telephony must be based on one of two complementary 3GPP standards: tandem-free operation (TFO) or transcoder-free operation (TrFO).

TFO is an in-band signaling protocol that allows voice codec parameters to pass unmodified through the PCM links in traditional core networks. The protocol preserves voice quality, but does not reduce the transport bit rate inside the core network.

TrFO is a combination of out-of-band control signaling (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 is transcoding-free operation, end-to-end. In other words, the voice signal is encoded in the transmitting mobile terminal and transported without modification to the receiving mobile terminal. A wideband voice connection cannot be set up for calls placed to a PSTN phone. Instead, OoBTC negotiation of TrFO ensures that AMR-NB coding with G.711 transcoding is employed at the edge to the PSTN. A similar mechanism exists in the TFO protocol.

The introduction of AMR-WB into GSM systems requires TFO, which is part of GERAN. The introduction does not require substantial modification of the core network; it affects only minor parts of the transport plane. AMR-WB and TFO can also be introduced into UMTS. A better option, however, is to use the recommended TrFO. Initially, the introduction of TrFO is more expensive, but in the long term this added cost is more than compensated for because TrFO makes excellent use of the revolutionary layered UMTS core network architecture in which a few highly concentrated mobile switching centers (MSC) control a variety of locally distributed media gateways (MGW).

The OoBTC of TrFO guarantees a higher success rate for AMR-WB calls. This is an important aspect, especially during market introduction, when the penetration of AMR-WB-capable equipment is low. OoBTC also simplifies the control of supplementary services and transcoder resources.

The combination of TFO and TrFO enables AMR-WB calls between all types of 3GPP mobile terminals (GSM/EDGE and UMTS). In cases where TrFO is used, TFO support in the media gateway enables interworking with TFO in GERAN.

GERAN

The GERAN air interface is characterized by time-division multiple access (TDMA) radio technology and a relatively small channel bandwidth of 200kHz per carrier. GERAN does not provide a fast link-level means of combating non-frequency-selective fading – caused, for example, by shadowing. AMR link adaptation, which in principle is identical for AMR-NB and AMR-WB, addresses this issue, drawing on the unique multirate feature of the codecs and adaptively selecting the most suitable rate for current channel conditions. This results in the best possible voice quality at any given moment and much enhanced robustness against poor channel conditions. The gain in robustness can be used to enhance voice quality, radio network capacity, or both.

The implementation of the AMR-WB codec and its radio channel in GERAN is similar to that of AMR-NB. The base transceiver station (BTS) and base station controller (BSC) require only minor software upgrades provided single or double TRUs are used. This is because AMR-NB and AMR-WB apply the same principles for channel coding, adaptation, and in-band signaling. However, additional support must be provided in the transcoder unit (TRAU) because the AMR-WB voice codec is about twice as complex as AMR-NB. TFO must also be supported.

UTRAN

UTRAN, which offers a large channel bandwidth of 5MHz per carrier, effectively controls non-frequency-selective fading, such as shadowing, by means of fast power control. Therefore, the multirate feature of the AMR codec is not needed to control voice quality but can instead be used to optimize capacity.

To allow interconnections with GERAN by means of TFO and TrFO, UTRAN maps the GERAN multimode configurations into multimode radio bearer (RB) configurations. The coding rate can thus also change dynamically without reconfiguration or interruption in the voice path. As a means of controlling capacity, operators will be able to limit the maximum AMR-WB coding rate separately in the uplink and downlink on a per-cell basis. Seamless handover of AMR-WB voice service between UTRAN and GERAN is supported in both directions. Every change needed for introducing AMR-WB in UTRAN can be made through software updates.

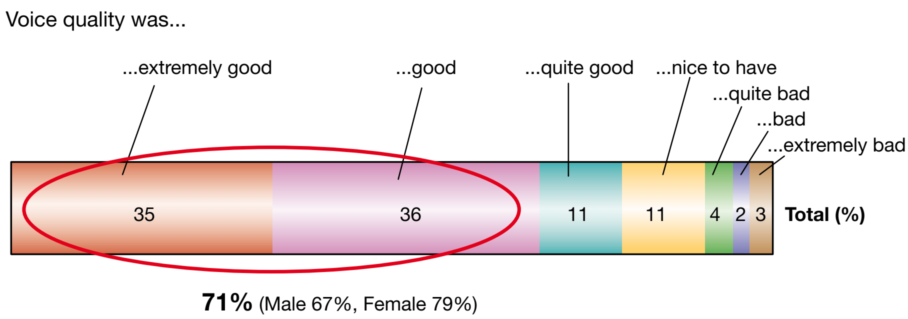
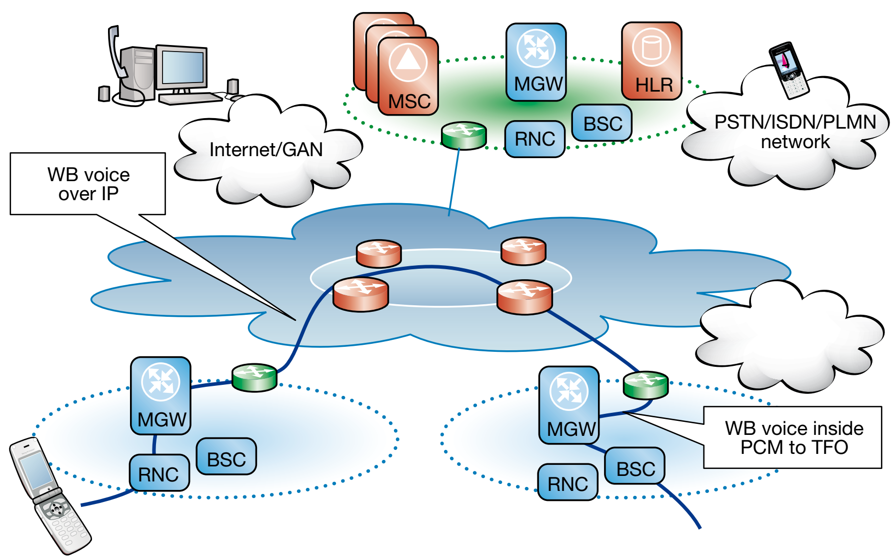


Figure 3
User perception of voice quality using AMR-WB mobile phones.

Figure 4
Modern layered architecture with centralized MSC servers and distributed media gateways connected via an IP backbone. The traditional interface to the BSC in GERAN is realized via PCM.



Terminal aspects

To obtain AMR-WB voice quality in an end-to-end solution, terminal designers must

- update audio-processing algorithms (acoustic echo canceller, noise reduction or level control) for operation at 16kHz sampling frequency;
- plan for increased demands on DSP power – compared with AMR-NB, AMR-WB puts roughly twice the demand on DSP power for processing audio;
- optimize the terminal's acoustic design for an audio bandwidth of approximately 100Hz to 7kHz; and
- increase linearity in the microphone and loudspeaker to match the greater dynamic range of AMR-WB and demands for higher voice quality.

Capacity

Although AMR-WB doubles the audio bandwidth of voice communication, the bit rates are similar to those of AMR-NB, in particular when operating in the mandatory configuration (set 0). Radio network planning for an AMR-NB-based mobile telephony system is thus suitable for AMR-WB. Radio network capacity is also similar. Accordingly, AMR-WB can co-exist with AMR-NB in a system planned for AMR-NB. This is especially important during the introduction of the AMR-WB service.

Compared with today's most common fixed-rate codec configurations (EFR in GERAN, and AMR at 12.2kbps in UTRAN), the AMR-WB configuration allows for greater radio network capacity and improves robustness against channel errors.

When TFO is used, AMR-WB requires the same transport capacity in the core network as PCM or AMR-NB. By contrast, the introduction of TrFO in the core network reduces the demand for transport capacity from 64kbps to the actual voice codec bit rate, which, in the case of AMR-WB, may be 12.65kbps or less.

Beyond 3GPP circuit-switched voice: GAN/IMS

There is growing need for a voice codec that supports wideband voice in converged wireline and wireless services. Seen in the context of mobile phones, voice will likely be transmitted via packet-switched generic access networks (GAN), which might typically include WLAN, or as a single media flow within an IMS multimedia telephony application. For interoperability with 3GPP, the ongoing standardization in ETSI TISPAN recommends the use of the AMR-NB codec. In all likelihood, it will also refer to AMR-WB for interoperability with 3GPP wideband telephony.

AMR-WB was originally designed to withstand typical circuit-switched radio channel errors, but it is also suitable for packet-switched channels with IP transport. Indeed, it is the codec of choice for VoIP applications, in particular because, when operated with adaptive rate and combined with transport redundancy, it maintains outstanding quality even when IP packet loss is severe (IP packet loss rates of 10% to 20% or more).

Conclusion

The new AMR-WB codec enables true, natural voice quality in wireless communication systems. A MUSHRA listening test showed that users clearly prefer the new codec over the leading AMR-NB codec. A four-week network trail jointly conducted by Ericsson and T-Mobile International confirmed these findings.

AMR-WB achieves this gain in quality by doubling the audio bandwidth of voice communication at a coding bit rate which is similar to that of AMR-NB. The radio network planning for AMR-NB is thus suitable for AMR-WB. Radio network capacity is also similar. Therefore, AMR-WB can co-exist with AMR-NB in a network planned for AMR-NB. Compared with today's fixed-rate configurations, the gain in capacity from using AMR-WB is similar to that from using AMR-NB.

The implementation of AMR-WB in GERAN and UTRAN is similar to that of AMR-NB. Every change needed for introducing AMR-WB can be made through software updates provided suitable hardware versions are in place.

The core network must support TFO (GSM/EDGE/UMTS) or TrFO (UMTS). System impact is only moderate and all changes can be made via software upgrades.

New terminals with corresponding acoustic designs and audio-processing functions will be needed to obtain AMR-WB voice quality. Ericsson's terminal platform will give the codec commercial in 2007Q1.

TERMS AND ABBREVIATIONS

3GPP	Third Generation Partnership Project
ACELP	Algebraic code-excited linear prediction
AMR	Adaptive multi-rate
AMR-NB	Narrowband AMR
AMR-WB	Wideband AMR
ATM	Asynchronous transfer mode
BSC	Base station controller
BTS	Base transceiver station
Codec	Coder/decoder
CS	Circuit-switched
DTX	Discontinuous transmission
EDGE	Enhanced data GSM environment
EFR	Enhanced full rate
ETSI	European Telecommunications Standards Institute
GAN	Generic access network
GERAN	GSM/EDGE radio access network
GSM	Global system for mobile communication
IMS	IP multimedia subsystem
IP	Internet protocol
ITU	International Telecommunication Union
ITU-R	ITU radio communication sector
ITU-T	ITU telecommunication standardization sector
LPC	Linear predictive coding
MGW	Media gateway
MSC	Mobile switching center
MTSI	Multimedia telephony service for IMS
MUSHRA	Multi-stimulus test with hidden reference and anchor
OoBTC	Out-of-band Transcoder control
PCM	Pulse code modulation
PS	Packet-switched
PSTN	Public switched telephone network
RAB	Radio access bearer
RB	Radio bearer
RTP	Real-time transport protocol
TDMA	Time division multiple access
TFO	Tandem free operation
TISPAN	Telecoms and internet converged services and protocols for advanced networks
TRAU	Transcoder unit
TrFO	Transcoder free operation
UMTS	Universal mobile telecommunications system
UTRAN	UMTS terrestrial radio access network
VoIP	Voice over IP
WLAN	Wireless LAN