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Delivering Multimedia Content on Next-Generation Networks

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Mobile network operators (MNO) everywhere are looking to maximize their profitability by offering new services. In developed economies where mature markets are reaching the saturation point for subscribers, operators are looking to a new generation of multimedia services to generate additional revenues from their existing customer base. In developing markets, rapid growth is creating not only vast pools of new subscribers but also the possibility of a new paradigm for mobile communications where users on the go and without a PC can access Internet services any time, from any place over their mobile phones. Nobody knows which services will become the killer apps in next-generation networks, but what's certain is that user expectations are rising for both the types of services offered and the quality of those services. Operators need to find the right mix of services and deliver them as efficiently as possible.

This brief looks at next-generation mobile multimedia services in terms of their content and network resource requirements and proposes the Extended Adaptive Multi-Rate Wideband audio codec (AMR-WB+) as a key enabler for delivering these services with great quality and efficiency. Recently standardized by the 3rd Generation Partnership Project international standardization body (3GPP), AMR WB+ is the state of the art in high-fidelity audio compression technology, providing top-quality mono and stereo sound for the full range of multimedia services. Deploying AMR-WB+ in their networks will help operators deliver new multimedia services with unmatchable quality and reliability while using network resources with maximum efficiency.

Rising expectations

The 3GPP has defined and standardized mobile multimedia services like Packet-Switched Streaming (PSS), Multimedia Broadcast/Multicast Service (MBMS), and Multimedia Messaging Service (MMS), which are rapidly gaining popularity around the world. By combining text, images, speech, audio and video content, these services transform mobile phones into access points for media-rich information, packaged entertainment and communication between users on the go. In the early days of mobile telephony, users accepted sometimes poor sound quality or compromised network reliability for the convenience of mobility, but with today's mobile phones becoming terminals for rich media, users expect a new level of sound quality approaching what they hear on their MP3 players. However, two characteristics of mobile multimedia content delivery impose constraints that don't exist on standalone music players. First, the prevailing audio content types across multimedia services are more than just music, and second, network capacity constraints limit the bandwidth available for delivering this mixed content.

Mobile multimedia content mix

Table 1 illustrates the mixed audio content of the new multimedia services, which usually combine speech and other audio. In fact, it highlights that this content is often predominantly speech. Only pure music applications, such as music downloads or streaming, are important exceptions where audio content can be expected to be restricted mainly to music alone.

Table 1. Mobile Multimedia Services Audio Content Mix

Examples \ 3GPP Service	PSS	MMS	MBMS	Download Services
Information – News, sports, traffic, stock quotes, etc.	Speech Dominant, Mixed	Speech Dominant, Mixed	Speech Dominant, Mixed	Speech Dominant, Mixed
Travel guides	Speech Dominant, Mixed	Speech Dominant, Mixed	N/A	N/A
Online shopping, commercials	Speech Dominant, Mixed	Speech Dominant, Mixed	Speech Dominant, Mixed	N/A
Edutainment – training, how-to, corporate presentations	Speech Dominant, Mixed	Speech Dominant, Mixed	Speech Dominant, Mixed	Speech Dominant, Mixed
TV, Movies	Speech, Music, Mixed	Speech, Music, Mixed	Speech, Music, Mixed	Speech, Music, Mixed
Person-to-person MMS	N/A	Speech Dominant, Mixed	N/A	N/A
Audio Content Distribution – Audio books	Speech Dominant, Mixed	Speech Dominant, Mixed	Speech Dominant, Mixed	Speech Dominant, Mixed
Audio Content Distribution – Music	Music	Music	Music	Music

Given the mixed content of these services, it's important to use technology that gives outstanding results for all types of audio, including speech, music and other types of sounds. Traditional compression technologies are specialized for either speech or music, but AMR-WB+ was developed specifically for mixed audio content. Independent testing as well as selection and recommendation by international standards bodies has proven AMR-WB+ to be the best technology for this content mix. Moreover, the importance of audio quality in mixed media should not be underestimated. For example, experiments with videoconferencing in the 1990s showed that higher quality sound (wideband rather than narrowband) had such a positive effect on user satisfaction that trading off some video quality in order to provide wideband audio led to greater user satisfaction. The advanced compression technology of AMR-WB+ delivers the full spectrum of human audio perception in mono or stereo while maintaining very efficient bit rates.

Operator challenges

As many operators compete for customers in saturated markets, they can't depend on growing their customer base to increase revenues. Intense competition among them has seen average monthly revenue per user (ARPU) eroding as well.

Mobile multimedia audio content is more than just music

Voice and data usage, however, are climbing, and in fact, an Analysys study reported in May 2005 forecasts that mobile service revenues will grow at 9% per year between 2004 and 2007 even in nearly saturated Europe. A challenge for operators is to win a large portion of these revenues. Innovators are counting on multimedia services to do this.

Many MNOs today are transitioning to next-generation networks with increased data capacity. Maximizing the return on their current and future capital investments in network infrastructure is a major challenge, and they are experimenting in how to offer the new multimedia services over their valuable and limited licensed spectrum. For example, some offer MMS but limit the size of multimedia messages for fear of overwhelming their system resources.

Opportunities to increase revenues exist by delivering the most compelling, best quality, easiest-to-use services in the market at a fair and reasonable price. Using quality and price to differentiate themselves from their competitors, MNOs can increase their customer base, improve their subscriber satisfaction and loyalty levels and reduce churn.

Using AMR-WB+, MNOs can reduce their application bandwidth requirements and by doing so service more customers and even increase network coverage with the same resources, as well as increase the number of innovative concurrent applications in their portfolios.

What consumers want

While MNOs are looking for ways to increase ARPU, cost is a significant consideration in most consumer purchase and subscription decisions. Users want next-generation services including high-quality sound, ease of use and personalization options. However, moderately priced handsets, long battery life and predictable, reasonable service costs are also key to their satisfaction, so it is to the MNOs' advantage to be able to deliver advanced high-quality multimedia services on low- to mid-range moderately priced handsets. They can then depend on the "network effect" to grow the market – the more people use and can be reached by a service, the more people will want and need to use it, and the more profitable it will become for the operator. When a critical mass of users has a service, it will seem indispensable. The low complexity of the AMR-WB+ decoder allows it to be implemented on lower-tier phones, helping to make multimedia services affordable for consumers. As well, the low service bandwidth requirements of AMR-WB+ mean less costly service implementations for service providers, who can pass their savings along to consumers through lower service pricing.

Optimizing network resource usage

In contrast to the 9.6-kbps data-carrying capacity of GSM, GPRS provides an IP-based data infrastructure with a threshold capacity of more than 100 kbps, with multiple users sharing aggregate channels. However, this capacity is subject to channel availability, and realistically, as shown in Table 2, GPRS actually supports a maximum data rate of approximately 24 kbps for audio streaming in PSS and MBMS. In audiovisual applications, where the bit rates available for audio are further reduced by the bandwidth required for video, less than around 10 kbps is typically available for audio streaming. Although the required video bit rate is highly dependent on the content, for best possible audiovisual quality, it is reasonable to assume that video requires about 75% of the available bit rate, leaving just the remaining 25% for audio. In fact, when forward error correction (FEC) robustness mechanisms are used, the available net bit rates may be even lower.

UMTS wideband-CDMA technology supports data rates of up to 2 Mbps, but when this capacity is shared by multiple users and overhead is taken into account, typically only 48 kbps are available for audio streaming and just 12-16 kbps for audio in audiovisual PSS and MBMS streaming.

Table 2. Available Bandwidth for Mobile Multimedia Streaming

		Audio Content		Audiovisual Content	
Transport / Service	Radio access technology	Channel bandwidth (total available rate)	Audio (net rate)	Channel bandwidth (total available rate)	Audio (net rate)
PSS	GPRS (2.5G)	36 kbps	24 kbps	36 kbps	< ~ 10 kbps
	UMTS (3G)	64 kbps	48 kbps	64 kbps	~ 14 kbps
MMS streaming	GPRS (2.5G)	36 kbps	24 kbps	36 kbps	< ~ 10 kbps
	UMTS (3G)	64 kbps	48 kbps	64 kbps	12–16 kbps

Table 3 shows the amount of playing time provided by files of sizes typically used in MMS and MBMS based on the bit rates used to encode them. For example, if an MMS file of 100 KB of audio-only content is encoded at 14 kbps, its playing time will be almost one minute. If the same-sized file is encoded at 24 kbps, it will play for only around half a minute. (Taken further, if the same 100-KB file is encoded at 64 kbps, it will play for less than 13 seconds!) If the 100-KB file had audiovisual content and only 25 KB were available for audio, the audio content would have to be encoded at 10 kbps to have an audio playing time of around 20 seconds. Ratios are similar for an MBMS file of 300 KB. As these examples illustrate, lower bit rates deliver proportionately more content in files of any given size.

Table 3. Download Times for Typical Multimedia File Sizes

		Audio Content		Audiovisual Content	
Transport / Service	Radio access technology	Encoded file size	Playing time @ encoding rate	Encoded file size	Playing time @ encoding rate
MMS	GPRS/UMTS	100 KB	~0.5 min@24 kbps ~1 min@14 kbps	75 KB (video) + 25 KB (audio)	~20 sec@10 kbps
MBMS download	GPRS/UMTS	300 KB	~1.5 min@24 kbps ~3 min@14 kbps	225 KB (video) + 75 KB (audio)	~60 sec@10 kbps

In summary, the target bit rate range for mobile audio applications, whether audio or audiovisual, is from about 10 to 24 kbps. Compressing audio such as a 16-bit stereo PCM signal sampled at 48 kHz to achieve these rates requires a highly efficient codec with compression ratios of 64:1 to more than 150:1. AMR-WB+ achieves these rates and delivers best quality sound across all content types, including speech, music and mixed content.

AMR-WB+ codec overview

AMR-WB+ is the state of the art in speech and audio compression. Standardized and recommended for mobile multimedia applications by the 3GPP, AMR-WB+ provides excellent reproduction of the full audio spectrum (up to 20 kHz) and operates at bit rates ranging from 6-36 kbps for mono and 7-48 kbps for stereo. AMR-WB+ is actually a hybrid of two technologies, ACELP® and TCX, combined in one codec to deliver the best quality for both speech and audio, which, as shown in Table 1, are equally important in multimedia content.

Because of its versatility, AMR-WB+ is adaptable to a broad range of audio services, so it's the only audio codec you need.

AMR-WB+ is built for real-world networks, where bandwidth is limited. In fact, AMR-WB+ is the only codec that can provide an exceptional user experience at very low bit rates. It supports bit rate scalability so it can be adapted to different network conditions, enabling seamless, dynamic link adaptation to network congestion or degradation on a frame-by-frame basis and ensuring consistent high service quality for users of mobile streaming services as they move around. Using high-efficiency packet loss concealment, AMR-WB+ preserves end-user service even in severely degraded network conditions.

The scalability and flexibility of AMR-WB+ address the capabilities of diverse networks and handsets, so content providers can reach customers on several networks without duplicating their hardware investments. This enables consistent service offerings across different network types and eases service transitions from 2.5G to 3G networks.

For its Release 6 multimedia service definitions, 3GPP selected two codecs: AMR-WB+ and Enhanced AACPlus (eAAC+). However, in 3GPP verification testing, AMR-WB+ greatly outperformed eAAC+ for all content types at low bit rates, as shown in Figure 1. At higher bit rates, AMR-WB+ performed equally to the other codec, as shown in Figure 2.

Figure 1. MUSHRA Test Results for All Content Types at Low Bit Rates

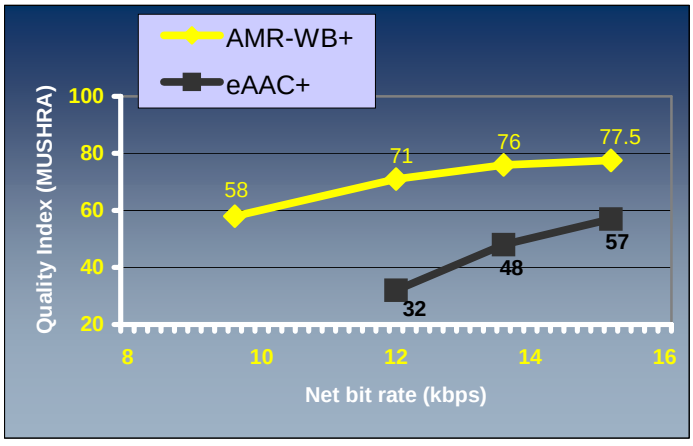
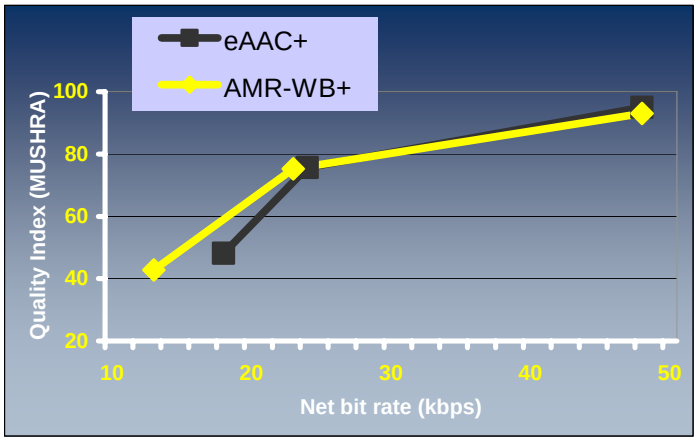


Figure 2. MUSHRA Test Results for All Content Types at Higher Bit Rates

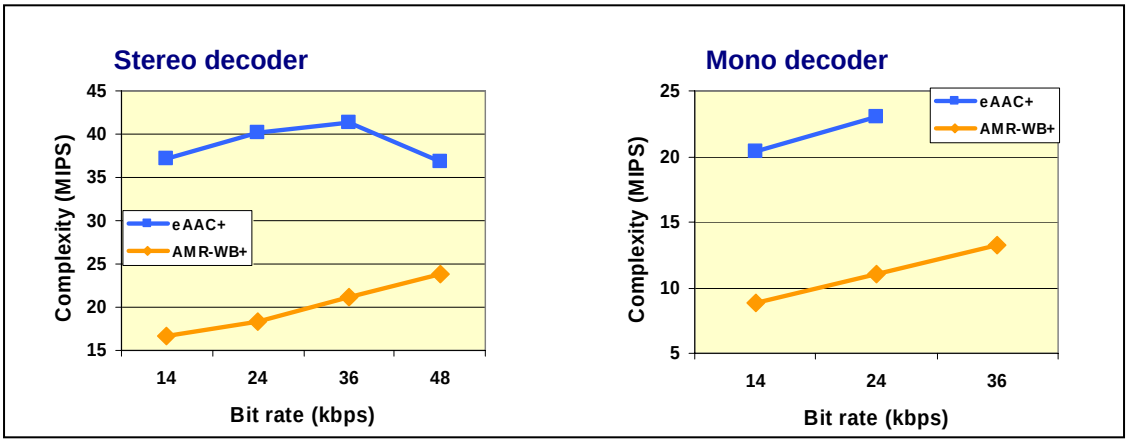


Testing for U.S. National Public Radio also proved the superior quality of AMR-WB+ for all audio content types, especially at low and very low bit rates. It also found that listeners enjoyed audio content encoded/decoded by AMR-WB+ for longer listening periods without tiring compared to the other codecs they tested. (The full report of the study, which was carried out by independent testers, is available online at the following URL: http://www.euonline.org/pub/iboc/low_bit_rate_coder_report.pdf.)

Supports longer battery life

Low complexity is crucial in the decoder to conserve battery power and support simultaneous video and audio decoding on mobile devices with limited computational resources. Here also, AMR-WB+ has a significant advantage over eAAC+ – typically having half the complexity of eAAC+, as shown in the following graphs – while delivering great-sounding audio for all content types.

Figure 3. 3GPP Complexity Comparison



Robust in real-world operating conditions

Error resilience is important because of the high likelihood of packet losses over a wireless link. AMR WB+ uses high-efficiency packet loss concealment, preserving service to end-users even in severely degraded conditions. Furthermore, AMR-WB+ was developed for robustness in mobile use and has excellent performance under various speech background noise conditions, whether in the car, in the office, or on the street.

Conclusion

By optimizing bandwidth usage as they transition to next-generation networks, MNOs can deliver top-quality multimedia services while containing costs for themselves and their customers. Highly efficient AMR-WB+ audio compression enables operators to deliver more services to more users on a given set of resources. Only AMR-WB+ can deliver excellent quality in both speech and other audio, especially at low bit rates, where no other codec can compare. The excellent sound quality of AMR WB+ meets or exceeds customers' expectations for all types of audio, even at low bit rates, while low complexity in the decoder helps prolong the intervals between recharging the handset. AMR-WB+ is a key enabler for MNOs who want to create the future and reap the rewards of being early to market with top-quality next-generation multimedia services.

Abbreviations

3GPP	Third Generation Partnership Project
ACELP	Algebraic Code-Excited Linear Prediction
AMR-WB+	Extended Adaptive Multi-Rate Wideband codec
ARPU	Average monthly revenue per user
eAAC+	Enhanced Advanced Audio Coding Plus
CDMA	Code Division Multiple Access
FEC	Forward error correction
GPRS	General Packet Radio Service
M-commerce	Mobile commerce
MBMS	Multimedia Broadcast/Multicast Service
MMS	Multimedia Messaging Service
MNO	Mobile network operator
MUSHRA	MUlti Stimulus test with Hidden Reference and Anchors
PCM	Pulse Code Modulation
PSS	Packet-switched Streaming Service
TCX	Transform-Coded Excitation
UMTS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Radio Access Network

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