



# Motorola's UMTS/HSxPA Wideband Adaptive Multi-Rate Solution

Technical Brief



# Improved Speech and Audio ...

Listen up! .... music to your ears! ..... the mobile phone audio experience is about to get a boost in sound quality thanks to improvements in network technology. Wideband Adaptive Multi-Rate (WB-AMR) introduces new voice coding techniques to deliver improved fidelity audio and in doing so improves the all important end-user experience.

Voice quality and customer satisfaction go hand in hand with each other, especially as speech will always be the primary means of communication. An important consideration for the service provider is that satisfied end-users generally make longer calls - increasing minutes of use (MoU) and Average Revenue per User (ARPU) as a result.

This paper describes the principles and benefits of this exciting and audible development.

## Wideband Adaptive Multi-Rate

The speech CODECs used in today's mobile networks limit the audio bandwidth to between 300Hz and 3.4 kHz (narrow-band). While providing good performance, the perceived speech or audio lacks naturalness of face to face communication, especially in the case of the human voice.

To help overcome this effect, WB-AMR uses speech compression techniques to increase the audio bandwidth to 7 kHz, roughly doubling the speech bandwidth compared to narrow-band AMR without impacting radio or backhaul requirements. As a result, WB-AMR provides speech quality beyond that experienced in narrow-band wire-line telephone and wireless networks.

The WB-AMR speech CODEC has been adopted and standardized by 3GPP<sup>1</sup> and ITU-T<sup>2</sup>. The adoption of WB-AMR by these bodies is significant as it is the first time that the same CODEC has been adopted for both wireless and wire-line telecommunications. This approach has the advantage of eliminating the need for transcoding and therefore eases the implementation and integration of more speech and audio applications and services across the various communication systems and platforms available today

1 3GPP (2001) TS 26.171, "Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; General description"

2 ITU-T (2002) G.722.2, "Wideband coding of speech at around 16 kbps using Adaptive Multi-Rate Wideband (AMR-WB)";



# Wideband AMR effectively doubles the audio bandwidth

<sup>3</sup> Bit rates: 23.85, 23.05, 19.85, 18.25, 15.85, 14.25, 12.65, 8.85 and 6.6 kbps.

The WB-AMR CODEC operates on nine speech coding bit-rates between 23.85 and 6.6 kbps<sup>3</sup> and uses ACELP (Algebraic Code Excitation Linear Prediction) techniques (also used in narrow-band AMR). Linear prediction (LP) is a mathematical operation where future values of a discrete time signal are estimated as a linear function of previous samples.

In WB-AMR a limited set of bits is distributed as excitation to a LP filter to synthesize the high frequency end of the audio envelope using predictions based on parameters from a LP filter associated with the lower frequency envelope; the gain is adjusted relative to the lower frequency envelope based on voicing information.

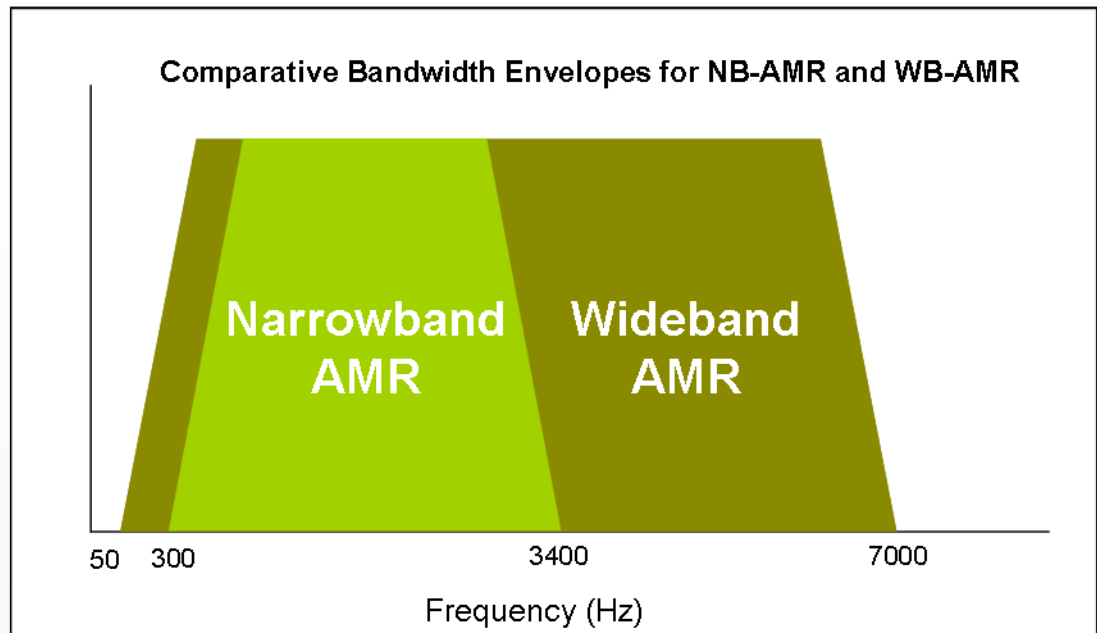
ACELP is based on Analysis by Synthesis (AbS), which means that the encoding (analysis) of the audio is performed by optimizing the decoded (synthesized) signal in a closed loop. Ideally the best possible ACELP stream is produced by trying all possible bit combinations and selecting the one that produces the best-sounding decoded signal. However this is impractical due to the complexity of the hardware required. Instead, taking advantage of the fact that human perception of audio quality is subjective to the listener, the system is able to use a subset of codes to decode the signal.

LP encoding is performed every 20ms while fixed and adaptive excitation codebooks are searched every 5ms to ensure optimal CODEC parameter values are maintained. The processing itself is carried out at a sampling rate of 16 kHz.

Of the nine available bit rates, the 12.65 kbps and higher coding bit-rates are used to provide high-quality wideband audio, whereas the lower bit-rates of 8.85 and 6.6 kbps are intended for temporary use during adverse radio conditions or periods of cell congestion.

In industry "listener perception of quality" subjective testing, it has been shown that all but the lowest bit rate wideband CODEC (6.6 kbps) produces better results than the best narrow-band CODEC (12.2 kbps). As the telecommunications industry move more and more towards All-IP networks and voice communication is packet switched, so the selection and use of the best suited CODEC becomes even more important.

The introduction of WB-AMR means that service providers will be able to offer improved fidelity speech and audio quality in multimedia content while end-users will benefit from improved clarity and intelligibility. Furthermore, service providers will be able to offer improved existing end-users services such as multi-way conference calls, personalized ring tones, audio broadcasts and as previously noted, new services such as VoIP over HSxPA.



Furthermore the 3GPP standard for speech coding provides the possibility to increase the traffic capacity of the radio network dependent on the prevailing radio conditions by continuously adjusting the speech coding.

WB-AMR not only improves audio quality, it provides distinct service provider benefits. WB-AMR adapts the bit allocation between speech and channel coding, optimizing the audio quality to prevailing radio channel conditions. That is, the bandwidth is reduced (higher compression used) when cell congestion is high; and increased (lower compression) as the cell congestion decreases.

As the compression ratio is altered dependent on cell load/congestion, network efficiencies can be realized and improve network capacity during times of congestion or poor radio conditions while maintaining speech quality that exceeds existing narrow-band standards.

In common with other GSM and UMTS speech CODECs, WB-AMR has a low bit-rate coding (DTX) dependent mode, which improves battery life and reduces RF interference.

#### **Motorola's WB-AMR solution**

Motorola's WB-AMR solution is released as part of UMTS Systems Release 6 (USR6).

With the adoption and availability of UMTS and HSxPA multimedia services increasing rapidly the need for high-quality audio capabilities are essential for end-user satisfaction. Similarly, any feature that improves network capacity without the need for the deployment of additional equipment will secure interest from the service provider community.

Adaptive Multi-Rate (AMR) has already proven its worth in GSM networks. Now the benefits of Wide-band Adaptive Multi-Rate are available as part of Motorola's extensive UMTS/HSxPA network solutions.

WB-AMR will provide the subscriber with speech and audio quality improvements, which will provide a superior experience when compared to today's UMTS/HSxPA networks; as the end-user experience improves, so does customer satisfaction and along with that comes the chances of increased ARPU and customer retention. Apart from a software upgrade, Motorola's WB-AMR solution does not require any hardware upgrade and as such represents a relatively easy way in which to realize another step improvement in that all important customer satisfaction.

For further details on this particular feature and Motorola's market leading HSxPA solutions please contact your sales representative.



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