Wideband Voice Coding – Opportunities and Implementation Challenges

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Introduction

The proliferation of IP telephones and the approaching roll out of 3G wireless handsets have created a tremendous opportunity to provide better than toll-quality audio performance in the “new” packet-based telephone network. This better than toll-quality sound is achieved by using the capabilities of the data network that is not limited to 8-KHZ sampling rate (~4-KHZ pass-band) present in the existing telephone network. The availability of inexpensive DSPs that can compress wide bandwidth signals for transmission over the packet network is also a reason why this capability is becoming feasible today.

The wideband vocoders and the challenges to their implementation is the topic of this paper. We will survey the standardized wideband vocoders and discuss the current state of the art and where they are heading. Also, we will discuss the system implications and some of the roadblocks to widespread use.

Wideband Vocoders

Most vocoders used in Voice over IP (VoIP) systems today are tuned to pass the bandwidth that is allowed in the legacy PSTN, since interoperability with it has been a key requirement. This bandwidth is limited to approximately 200-3,400 HZ because of the 8-KHZ sampling rate used in the PSTN. Wideband vocoders extend the pass band that is useable to 50-7000 HZ and are targeted at applications that do not directly interoperate with the legacy digital PSTN. This gives a much richer sound to the voice conversation, making the speech more intelligible and better speaker recognition possible. Also the transmission of better quality audio signals (music, etc.) becomes possible.

There has been a lot of activity over the years in standards bodies to define wideband vocoders for these applications. The primary use of the early wideband coders was video conferencing, but more recent activity has been focused on VoIP and wireless applications.

The following sections review the various standards for wideband vocoders.

G.722 (SB-ADPCM)

Standardized in 1988, the ITU G.722 was the first standardized wideband speech coding algorithm to be used at a 16-KHZ sampling rate. The codec splits the 16-KHZ spectrum into two sub-bands and uses the ADPCM algorithm to encode in the sub-bands. Its overall delay is around 3 ms, small enough not to cause echo problems in telecommunication networks. In addition, the codec provides acceptable performance for transmission bit error rates up to 10^{-3}. This requirement ensures that performance degrades gently even under the worst transmission conditions that one might encounter in the network.

The G.722 codec takes 16-bit data at 16 kHz (bandwidth from 50 Hz to 7 kHz) and compresses it to 64, 56, and 48 Kbit/s.

G.722.1 (Transform Coder)

The ITU G.722.1 was developed in the late 1990s to provide lower bit rate and more compression than the G.722 codec. The goal was to have the equivalent quality of G.722 at roughly half the bit rate. The codec was developed by Picturetel and uses transform coding method. It takes 16-bit data at 16 KHZ (bandwidth from 50 Hz to 7 kHz) and compresses it to 32 and 24 Kbits/s. Most of its use today is in videoconferencing systems.

G.722.2 (AMR-WB)

AMR-WB is a wideband codec that was originally defined for GSM wireless networks but has recently been expanded to include wireline systems. Since it is very computationally intensive, it takes advantage of the dramatic improvements that have been made in DSP processing capabilities over the years. Based on the CELP vocoder, it has an audio bandwidth of 50-7000 HZ and provides nine speech codings between 6.6 and 23.85 Kbit/s.

AMR-WB was standardized in 2001 for GSM and WCDMA networks. Later in 2002, the ITU decided it would be the wideband standard for wireline applications as well. It is the first codec to have this dual application.

VMR-WB

VMR-WB is the new variable-rate multimode wideband speech codec designed for the wireless CDMA 2000 standard. It is intended for encoding speech in the 50 to 7000 HZ band and is sampled at 16 KHZ. Based on the 3GPP AMR-WB (G722.2) codec, it is interoperable at the 12.65 Kbit per second rate.

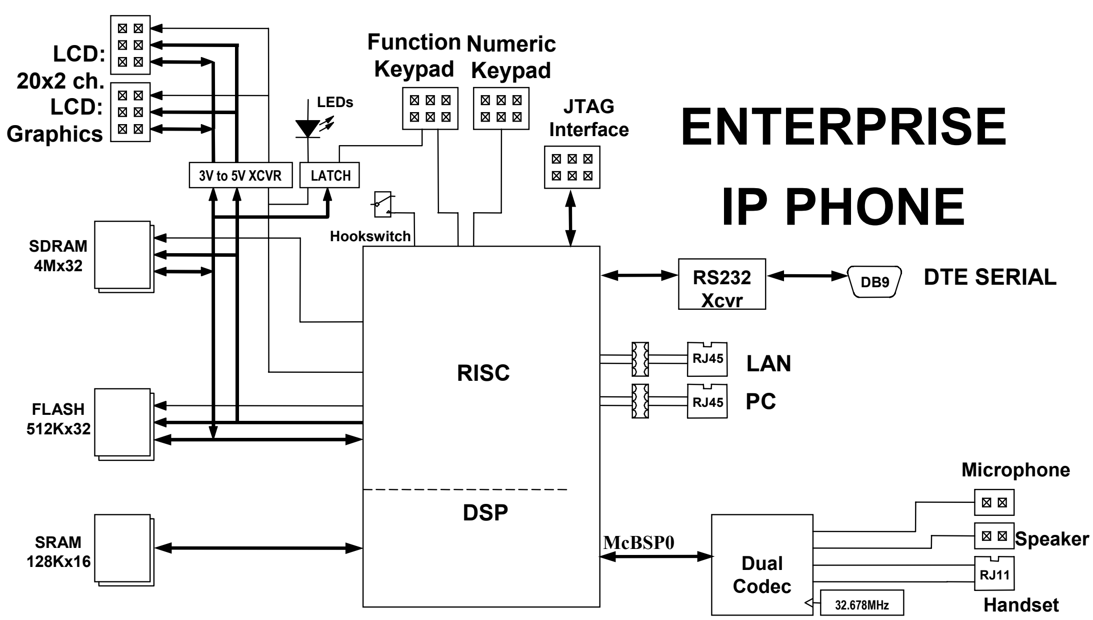
VMR-WB operates in four modes that are chosen based on traffic conditions in the network. The first three modes are specific to the CDMA system, and the fourth mode is the AMR-WB interoperability mode.

The interoperability of the VMR-WB and AMR-WB allows GSM/WCDMA and CDMA2000 systems to communicate without the need for transcoding that adds additional delay and voice quality degradation.

System Implications of Implementing Wideband Codecs

There are two primary areas of the telephone system design that are impacted by the use of wideband codecs: 1) the analog to digital input/output subsystem, and 2) the software processing that is performed in the DSP.

A sample block diagram of an IP phone is shown below with the relationship between the DSP processor and the A/D I/O subsystem at the bottom right of the picture.



Analog to Digital Input/Output Subsystem

The A/D I/O subsystem consists of the speaker, microphone, and hardware codec, which samples the audio signals and digitizes them for transmission to the DSP. In a telephone that is capable of wideband voice coding, it is crucial that all of these components support the band of 50-7000 Hz that is required for the wideband vocoder.

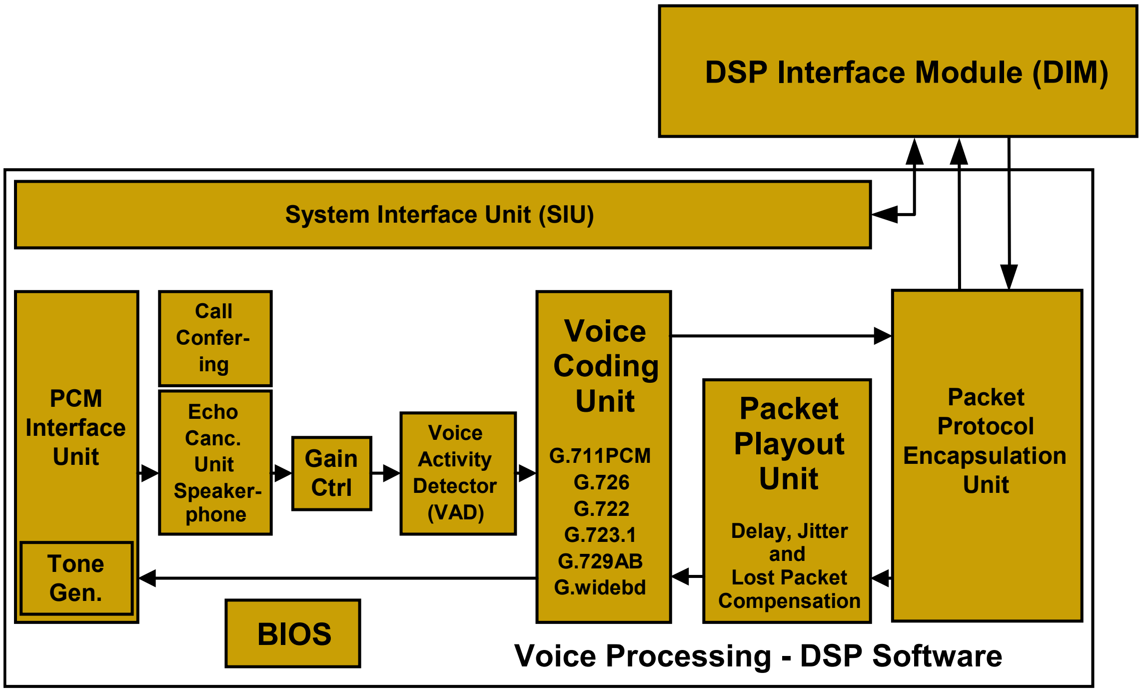
This means that the speaker and microphone that are typically used in standard PSTN capable telephones are not suitable and will limit the bandwidth based on their design. These must be replaced with wideband capable speakers and microphones.

Also, the filters in a standard hardware codec used in a conventional PSTN application will have anti-aliasing filters that limit the bandwidth to 200-3400 Hz. Therefore, another important consideration in the design of a wideband capable phone is the use of a hardware codec that supports 50-7000-Hz bandwidth and can sample the signal at 16 Khz.

Software

One of the main considerations with respect to software in a wideband capable telephone is the requirement that the processing components between the serial input port and the wideband vocoder must run at 16 KHz. This requirement doubles the amount of data that must be processed per unit of time and typically results in twice the processing requirements. The block diagram on the next page shows the components that are impacted beginning at the left with the PCM interface unit. The module that impacts the processing requirements the most is the echo canceller, which must run at twice the rate of the conventional IP phone. This can be significant, since an acoustic echo canceller uses a lot of computing resources.

The second area of impact is the processing requirements needed by the wideband vocoder itself. The table on the next page shows how the processing requirements for the AMR-WB codec have gone up relative to the original G.722 ADPCM based coder. This 25-MIPS requirement for AMR-WB is also a significant jump over the narrowband AMR codec that is in the range of 12 MIPS.



Telogy Software® TMS320C55x™ CODEC (Wideband) Modules

CODEC Module	Interface	C55x MIPS	C55x Memory Allocation Prog Mem	Table	Data Mem/ch	Scratch
G.722	Xdias	5.61	1726	340	164	438
G.722.1	Xdias	NA	NA	NA	NA	NA
AMR-WB	Xdias	25.85	22436	13073	2537	4435

Summary

This paper has presented the opportunities and challenges that arise when implementing wideband voice coders in a VoIP system. The requirements for new acoustic input/output devices and processing requirements enabled by more powerful DSPs allow the user to realize the benefits of improved audio quality through wideband audio coding.

To learn more about how Texas Instruments has addressed these challenges, go to: www.ti.com/voip

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