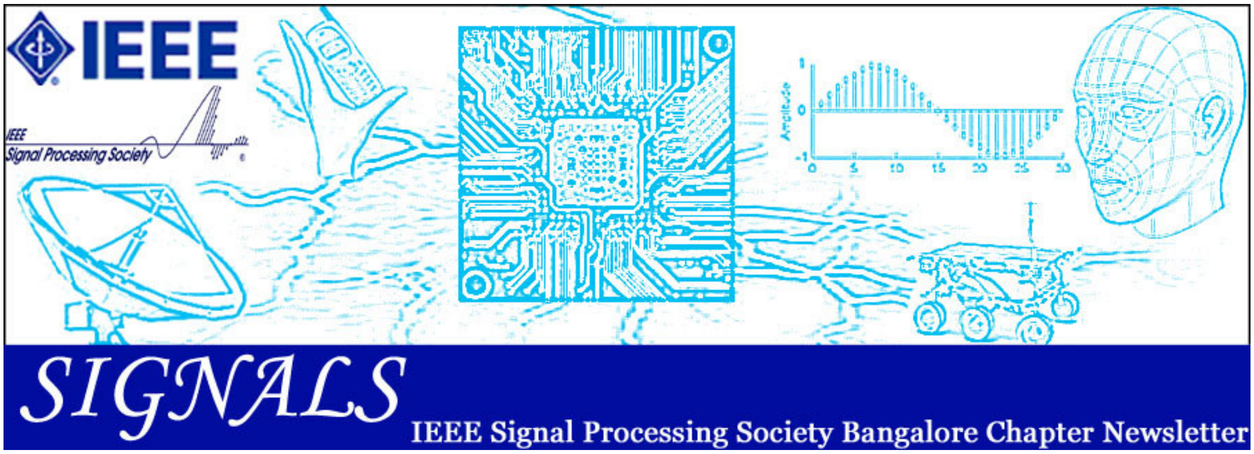


ISSUE #01, Sept 2005

FROM THE EDITORIAL TEAM

Remember the first time you saw that elusive 'text book' sine wave on the CRO, when you connected your very own hand made oscillator to the probes? Remember the thrill and excitement? Well, that more or less sums up what we feel like, bringing out the first edition of SIGNALS - the IEEE SP Bangalore chapter quarterly newsletter. This newsletter aims to be an online forum for sharing the latest information about Signal Processing and related events, and for highlighting the work that is being done in various institutions (both industry and academia) in and around Bangalore.

The SP Chapter has been active in many ways, sponsoring and organizing events related to Signal Processing, and providing a forum for the active Signal Processing community in and around Bangalore. From its start in the year 2000, we have come a long way in our reach and purpose.

Prof. V.U. Reddy, who was the first Chair of the IEEE SP Society, Bangalore chapter, and the current Microsoft Chair

Professor at IIIT Hyderabad, reminds us of the initial years of the SP Society this way –

"*In the 1990's Bangalore became the home for many industries with focus in digital signal processing (DSP). There were already Defence Research and Development Organization (DRDO) and Indian Space Research Organization (ISRO) laboratories engaged in DSP related R &D. Further boost to these activities came from DSP faculty of Electrical Sciences at Indian Institute of Science. To bring the DSP Scientists and Engineers working in different establishments onto one platform, IEEE Signal Processing Society, Bangalore Chapter was formed in November 2000, which was formally inaugurated by Prof. Alan V. Oppenheim of MIT who is regarded as father of DSP, on December 1, 2000.*"

Some of our ongoing activities have been:

- Conducting lectures, tutorials, and courses related to latest SP topics – Recent lectures on H.264, GPS, Speech processing, and courses on DSP have been well received by the community.
- Conducting SPCOM – A Biennial event, focused on SP and Communications. The most recent one was held in 2004 under the aegis of the IEEE.
- Organising the IEEE Workshop on Multimedia Compression – Scheduled to be held in October 2005.

We have very active institutions in Bangalore that work in the field of SP, and we invite everyone working in this field to join the bandwagon. You can contribute by providing your ideas, your time, and by participating in organizing workshops and conferences. Climb aboard and participate in the increasing industrial-academic partnership happening right here in Bangalore!

Looking forward to your active participation,

The Editorial team

ANNOUNCEMENTS

We need your feedback !

This newsletter will be published once every three months. We need your contributions to achieve the purpose of this newsletter – to keep all members aware of what's happening on the Signal Processing Frontier, and to encourage active participation in these happenings. Send us your ideas. If you have just cracked up a new theorem related to SP, share it with us. If you are organizing some SP related event, let us know. Paper presentations, commentaries, short articles or announcements, we will take them all. The point is that we need your active participation and newsletter contributions. Because if you don't, we will run out of ideas. So unless you want to read compositions on 'What I did last summer' or 'My dog', send us your contributions!! Send your feedback and contributions to ieeesp@dsplab.ece.iisc.ernet.in

MAIN COURSE

Fast Forwarding Audio Coding

-Mihir Mody, Texas Instruments

The demand for internet media services such as Apple's iTunes, Real's Rhapsody and Microsoft's window media as well as portable audio players / Juke boxes e.g. Apple's iPod, Creative's zen etc are ever-increasing. The music industry is on verse of big take off in consumer space. All these events have generated a lot of interest in the subject of audio coding. Audio coding (also referred to as compression) is the art and science of reducing audio signals for efficient storage (small file size) and high quality streaming (low bandwidth). In this article, we briefly cover the principles of audio coding and overview of the popular MPEG audio codecs (MP3, AAC) as well as a few proprietary standards. The main focus of this article is the recent breakthrough in audio coding technology and on-going work in the MPEG audio committee. Specifically, we take a look at a number of recently discovered techniques such as spectral band replication (SBR), integer MDCT (intMDCT), parametric audio coding and Binaural cue coding (BCC).

1. Basic of audio compression

A "raw" multi-channel digital audio signal consists of sequences (one per channel) of 16-bit samples. Typical sampling frequencies in high quality audio are 44.1 or 48 KHz, although lower sampling frequencies are used for sub-woofer channels. Raw audio needs very large space for storage, or equivalently bandwidth for streaming. As an example, one minute of DVD-quality audio data requires almost 30 Megabytes of space or 3.5 Mbps of bandwidth for real-time streaming. Audio coding can reduce this data rate by a factor of 20 with negligible impact on perceived audio quality.

Audio coding relies on an in-depth understanding of the human hearing system. As an example, when we are listening to a strong tone of a particular frequency, we tend to be insensitive to the presence or absence of weaker sounds in the nearby frequencies. This is known as frequency masking and is illustrated in figure 1. The tone at frequency f_1 is "masked" (and is hence inaudible) by the loud tone at frequency f_0. The tone at frequency f_2 is however audible since it lies outside the "masking region" of frequency f_0.

The frequency masking phenomenon is extensively exploited in audio coding. Each frame of audio is first transformed to the frequency domain, thereby decomposing it into a collection of tones at various frequencies. The signal is then analyzed to determine which tones would be irrelevant because of masking by nearby louder tones. By coding only the tones that are audible to the human ear, tremendous compression ratios can be achieved.

Masking is a complex phenomenon and what we just described is only a simplified model. In practical audio codecs, a comprehensive "psycho-acoustic" model, which mimics the properties of the human ear, captures the relative relevance of the audible frequency tones, while deeming other tones as inaudible. The psycho-acoustic model finds out masking noise curve for given input spectrum taking account of various frequencies as shown in figure 2. Any noise below this curve is inaudible to human being as explained earlier. The quantization process ensures that quantization noise is this and this enables perceptually transparent audio coding. To estimation of making curve from input spectrum is art as well as science in itself.

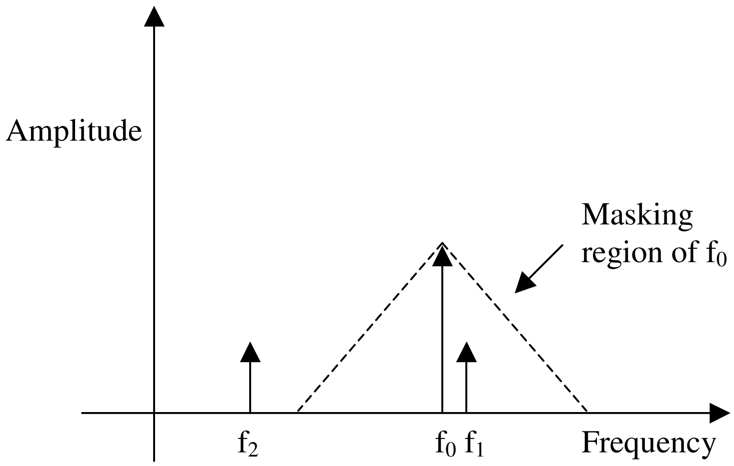


Figure 1: Frequency Masking

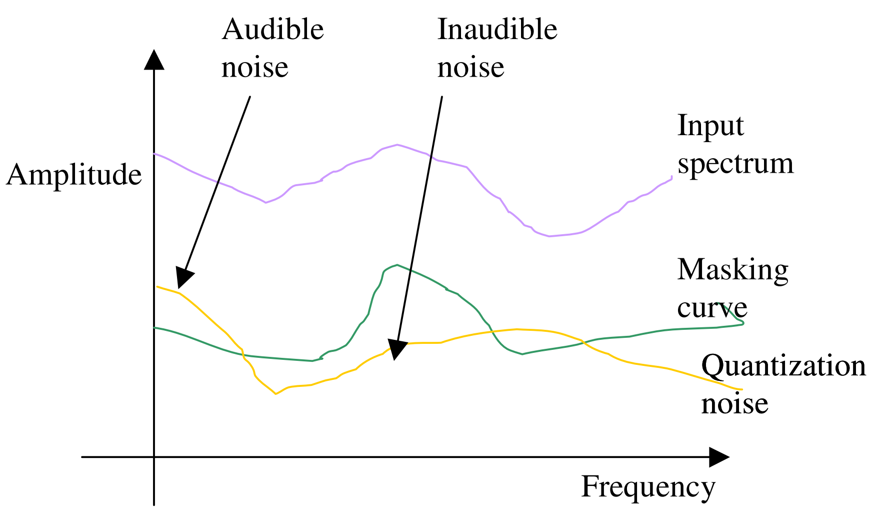


Figure 2: Masking curve

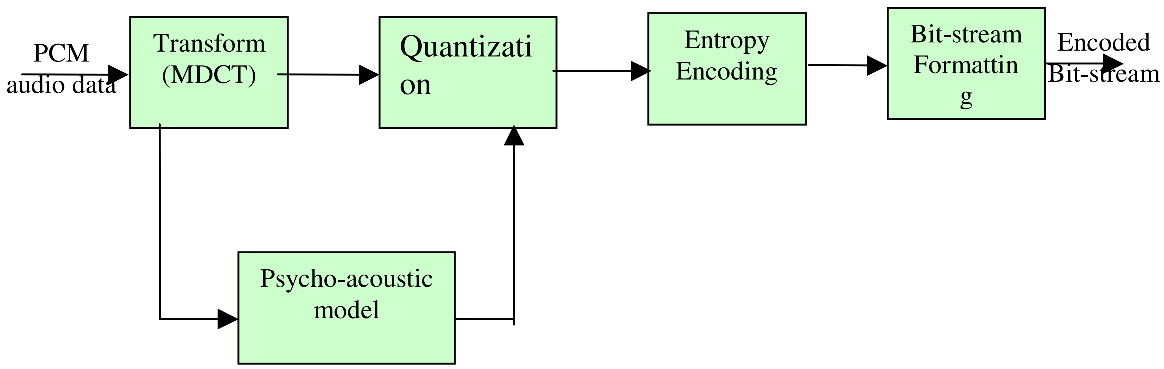


Figure 3: Audio encoder

Figure 3 shows the block-level organization of a typical audio encoder. The input audio signal is first transformed to the frequency domain to allow for analysis. Reversible transforms are best used in this step. In the first generation audio codecs such as Musicam, filters banks were employed. Of late, the Modified Discreet Cosine Transform (MDCT) is most preferred. As mentioned earlier, a psycho-acoustic model is used to determine the relative importance (represented by a "step size") of each transform coefficient. A smaller step size indicates that the corresponding tone is more important. The next step is quantization, where each transform coefficient (equivalently the tone amplitude) is scaled down by the step size and integerized into a "quantization index". Quantization is a lossy process and reduces the amount of information content significantly thereby contributing to a large reduction in bit-rate. Naturally, more important tones suffer lesser quantization error. The quantization indices are next entropy coded. Entropy coding is a lossless technique that uses fewer bits to represent more likely quantization indices. Huffman coding is the most commonly used technique in entropy coding. Finally the resulting data is formatted based on the standard specifications and packetized for storage or streaming.

It is easy to see that the decoder can follow a precise reverse process to re-construct the audio signal. Figure 4 illustrates the typical decoder.

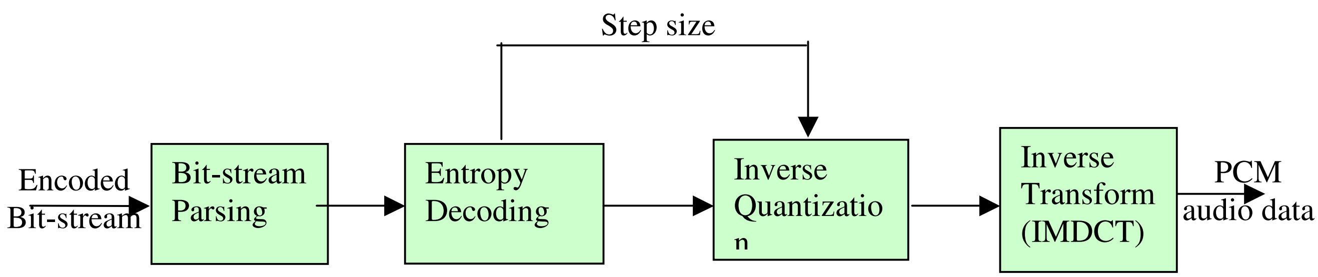


Figure 4: Audio decoder

The basic block diagram of most standard audio codecs mimic closely, the typical structure described above. However, each standard may add some extra blocks (also called tools) to support new features or improve compression performance.

As an example, one possible feature of an audio codec is support for scalability. In scalable audio codecs, the output bit-stream is divided into multiple “layers” of bit-streams. The first bit-stream is called the base layer, while the other bit-streams are called enhancement layers. While the superset of all the bit-streams represents the complete audio codec output, the base layer bit-stream has the property that it can independently represent the audio signal, although at a reduced quality. Scalability is useful for streaming applications, where a client connected via a narrow-band modem may listen to lower-quality music by decoding only the low bit-rate base layer, while a broadband client may consume bits at all layers to enjoy higher quality audio.

We next look at some of the early audio codecs that have been internationally standardized and deployed in popular products and services.

2. Previous MPEG audio standards

The Motion Picture Experts Group (MPEG) audio committee has standardized a series of audio codecs as shown in Figure 5.

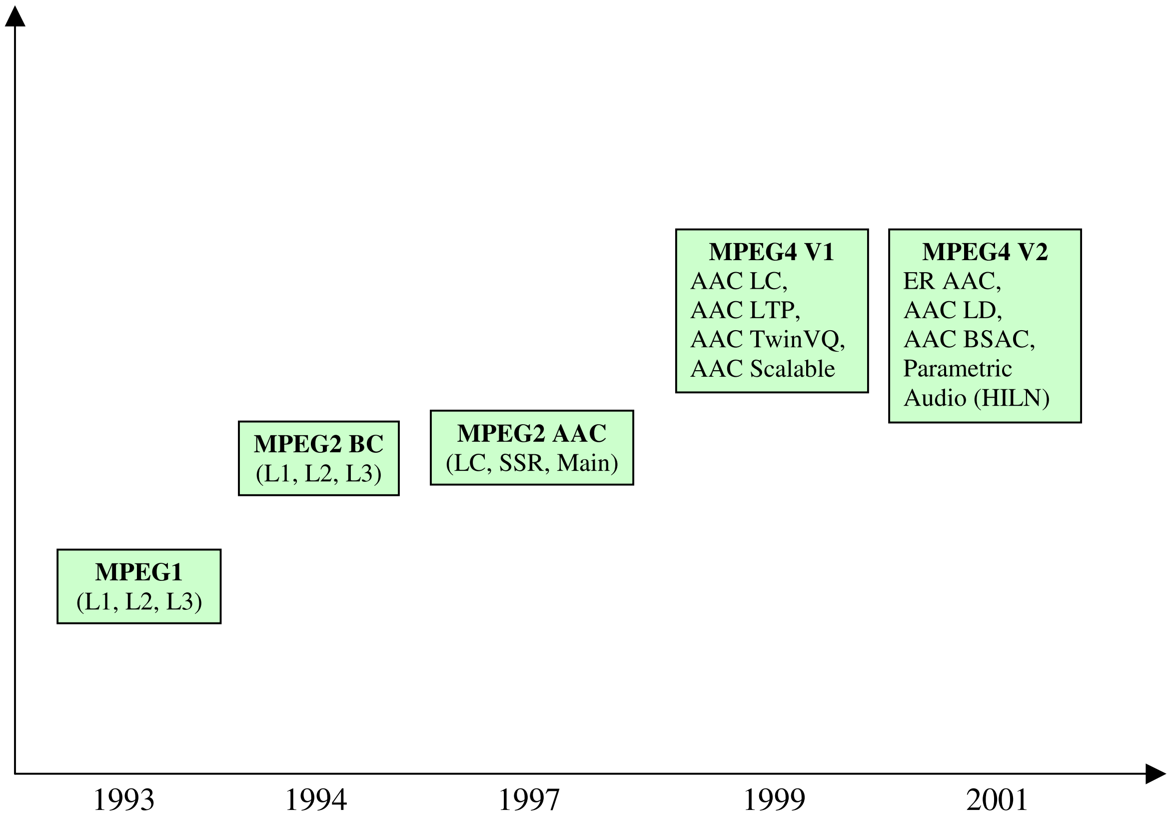


Figure 5: Early MPEG Audio standards

MPEG1 audio was the first generation standard from the MPEG audio committee. Standardized in 1993, MPEG1 supports mono (one) and stereo (two) channels at three sampling frequencies (32, 44.1 and 48 KHz). MPEG1 defines three layers namely L1, L2 (aka Musicam) and L3 (aka MP3), with the layers representing tradeoff points between quality and computational complexity. The L1 layer is the least complex of the three while MP3 provides the highest quality. MP3 has been shown to be perceptually transparent at 192 kbps for stereo songs (i.e. an expert listener cannot distinguish between raw audio and MP3 audio coded at this bit-rate).

In 1994, the **MPEG2** standard was announced. MPEG2 audio extended MPEG1 to support multi-channel audio (greater than 2 channels) and lower sampling frequencies (16, 22.05 and 24 KHz). The MPEG2 extension is backward compatible with MPEG1 audio. As an aside, a non-standard extension of MPEG2 known unofficially as **MPEG 2.5** allows even lower sampling rates (8-12 kHz) at bit rates from 16-32 kbps.

Among all the standards introduced so far, MP3 is by far the most successful one with the widest range of applications. MPEG1/2 layer 2 is also used widely – notably in DVD audio and some broadcast audio standards (e.g. DMB in Korea).

MPEG2 Advanced Audio coding (AAC) in 1998 was next major technological milestone in the audio coding field. While MPEG2 AAC is capable of perceptually transparent quality at 128 kbps for most stereo signals, it is not backward compatible with the earlier standards. AAC defines three profiles namely low complexity (LC) for embedded devices, main profile for high coding gains and Sampling rate scalable (SSR) for bandwidth scalability. The MPEG2 AAC LC has become the most popular of these profiles.

MPEG4 audio: Version 1 (in 1999) extends MPEG2 AAC by adding a perceptual noise substitution (PNS) tool to improve quality at intermediate bit-rates (around 32 kbps). MPEG4-AAC is one of the most popular AAC coding formats in use currently. Popular services such as iTunes from Apple Inc. and multimedia messaging (MMS) in 2.5G cellular systems use MPEG4 AAC.

In order to mitigate the complexity of the MPEG2 AAC main profile, an alternate tool called Long Term prediction (LTP) has been proposed in MPEG4. LTP promises quality comparable to AAC main profile, but with less computation and memory costs. The AAC-LTP combination is one of the optional choices in the MMS specifications defined by Third Generation Partnership Projects (3GPP). Today, however, there is very little content available in the AAC-LTP format.

MPEG4 also standardized two other codecs that are much less popular in the content marketplace: For low bit-rates like 6 kbps to 16 kbps, a so-called TwinVQ codec that uses a vector quantization scheme was defined. Also defined was the AAC scalable profile that offers scalability in bandwidth as well as quality.

MPEG4 audio Version 2 (in 2001) standardized four new profiles in addition to those in MPEG4 version 1 audio.

To support error-prone environments like wireless channels, it defined the error resilience (ER*)* profile for AAC. AAC-ER uses new tools like reversible variable length codes (RVLC), virtual codebooks, and codeword re-ordering for robustness against error during transmission. The Digital Radio Mondiale (DRM) service uses the AAC-ER profile for audio coding.

For real time conversation applications such as conferencing, MPEG4 v2 standardized the AAC low delay (LD) profile. AAC-LD has an algorithmic delay of 20 ms, which is much lower than the 130ms delay of standard AAC. However there are very few applications that use the AAC-LD profile.

To support fine grain scalability, MPEG4 v2 defined the Bit slice arithmetic coding (BSAC) profile. The AAC-BSAC codec is used in digital media broadcast (DMB) applications in Korea.

Finally, MPEG4 v2 introduced a first version of parametric coding scheme called HILN (Harmonic and Individual Line plus Noise) to support low bit-rates (e.g. 4 kbps to 16 kbps).

3. Recent technological advances in audio coding

The major achievements of the MP3 and AAC era have been followed by some key technical improvements that we highlight in this section. Jointly they have provided a big leap to the capabilities of audio coding.

Spectral Band replication (SBR)

SBR is a "bandwidth expansion" approach invented by Coding Technologies Inc, which has emerged as one of the important contributors to the field of audio coding. SBR addresses a typical drawback of transform coding – *i.e.* the bandwidth of the reproduced audio signal must, generally speaking, reduce as bit rate is reduced. The SBR approach overcomes this – it retains the full bandwidth of the reconstructed audio. To make up for the shortage of bits to represent the full signal, SBR exploits the correlation that exists between the energy of the audio signal at high and low frequencies as shown in Figure 6. It uses a well-guided transposition approach to predict the energy at higher frequencies, and uses a few additional bits to encode the prediction error. The perceived audio quality at low bit-rates improves tremendously as a result.

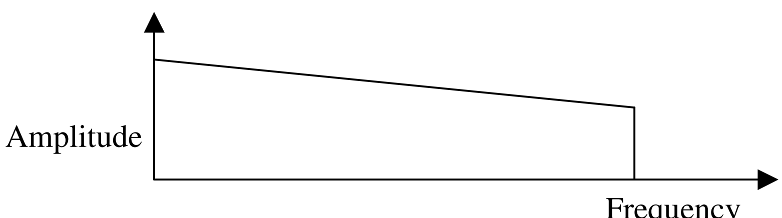


Figure 6a: Original Signal

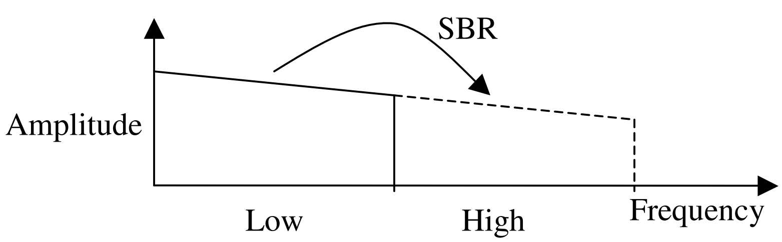


Figure 6b: SBR Reconstruction

SBR is seen to be capable of high quality stereo at bit-rates as low as 48 kbps. The technique can "extend" most standard algorithms. It has already been deployed to extend MP3, AAC etc to generate new codecs called MP3Pro & AACplus respectively.

Integer MDCT

Integer MDCT (IntMDCT) is a recent invention that addresses a key demand of applications such as high-resolution audio & professional audio system - lossless audio coding. Lossless coding of audio is analogous to applying one of the popular "zip" type utilities to audio signals. As opposed to the lossy approaches prescribed by other standards, lossless coding results in a reconstructed audio signal that is identical to the input digital audio. Traditional audio algorithms are hard-pressed to support lossless coding mainly because they use the MDCT that is based on floating point arithmetic. IntMDCT was developed by researchers at Fraunhofer IIS (known as FhG) another pioneering company in audio coding. IntMDCT is an integer approximation to the MDCT transform. It is derived from the MDCT using a lifting scheme. It is reversible, based on integer arithmetic and inherits most of the attractive properties of the MDCT. Advanced Audio Zip (AAZ) from Institute of Infocom Research (I2R) uses IntMDCT to achieve low bit-rate lossless coding. Integer MDCT is a big boon to audiophiles – enabling high compression with the unmatched fidelity of lossless coding. Applications that mimic WinZip ("AudioZip") are likely to become popular in the near future. There are a number of companies including Musicmatch, that have started download services for lossless audio, which proclaims arrival of loss-less audio coding

Parametric audio coding

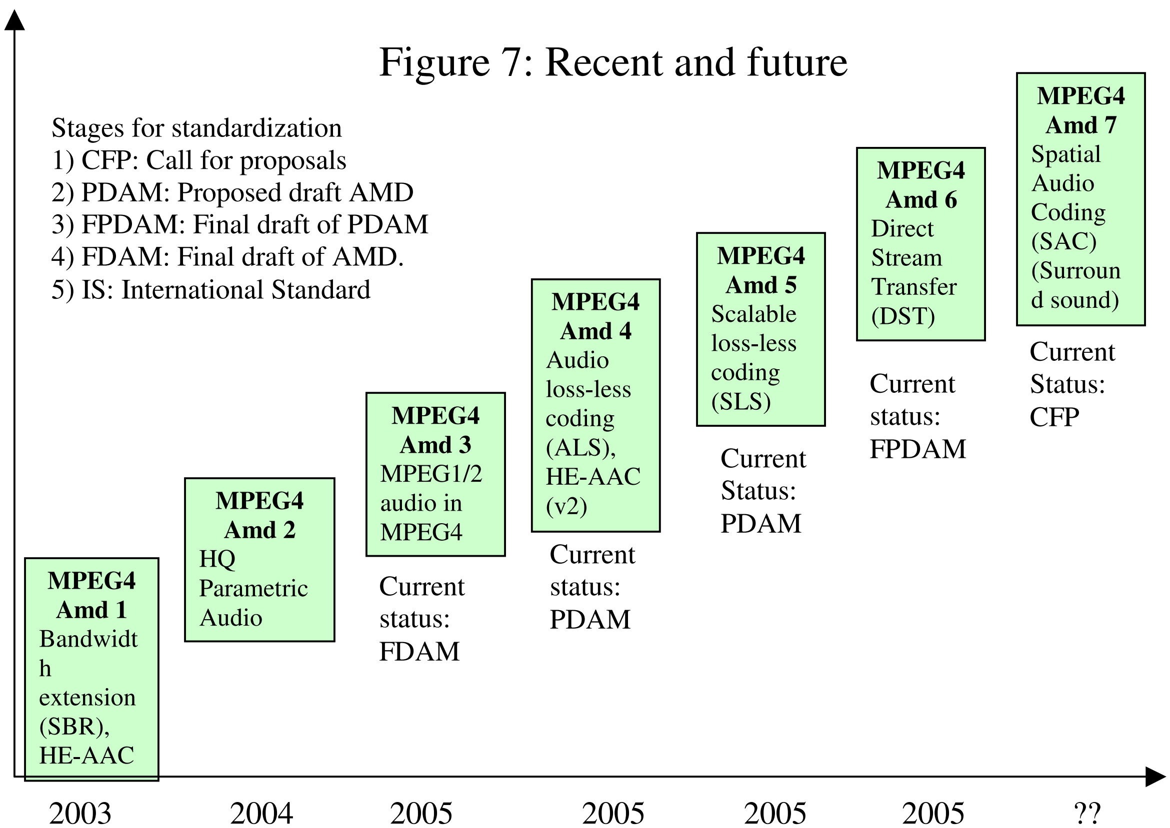
Parametric audio coding is a technique that was first successfully used in speech coding, but is now applied to audio coding as well. In the audio coding area, this technology is superior to standard transform-based codecs especially at very low bit-rates (< 24 kbps). In the parametric approach, the audio signal is separated into its transient, sinusoid and noise components. Next, each component is re-represented via parameters that drive a model for the signal, rather than the standard approach of coding the actual signal itself. The MPEG4v2-HILN standard is based on parametric coding. Two new enhancements to this model have been subsequently proposed to improve the overall performance of HILN. Another advantage of parametric coding is that audio manipulation e.g. time scale modification, pitch shifting etc. can easily be incorporated in a parametric representation.

Binaural Cue coding (BCC)

Binaural Cue Coding (BCC) enables parametric representation of spatial audio, delivering multi-channel output (with an arbitrary number of channels) from a single audio channel plus some side information. It uses parameters like inter-channel time, level, correlations etc. to support surround sound (e.g. 5.1) at very low bit rate. MP3Surround technology from FhG uses this technique to add surround sound capabilities to MP3 algorithm at the cost of few additional bits. This technique can be used as parametric stereo in case of two channels.

4. Recent standards and ongoing work in MPEG4 audio

To keep up the pace with new technologies, MPEG4 has added amendments (AMD) to existing standards. It should be observed that MPEG4 is currently the last standard planned in the core audio coding area. Future standards (e.g. MPEG7 and MPEG21) will continue to use the audio codecs standardized in MPEG4 but address advanced systems layer requirements such as audio representation, layering etc. Figure 7 summarizes on-going work in the MPEG audio committee.



The first amendment (AMD1) defines the High Efficiency profile (HE-AAC) version 1 by combining SBR technique with AAC. The same combination is also known as AACplus, which is a trademark of Coding Technologies. In the MPEG world, the codec is referred to as HE-AAC. HE-AAC is getting widespread acceptance in many standardization bodies (e.g. 3GPP, DRM, DVD forum & ISMA). A few commercial CD rippers (e.g. Nero) support audio CD ripping in the HE-AAC format.

The second amendment (AMD2) improves the performance of parametric audio coding (HILN) with better noise coding and parametric stereo. This high quality (HQ) parametric audio coding results in reasonable quality audio between 6 - 21 kbps.

The third amendment (AMD3) defines usage of MPEG1 and MPEG2 audio in the MPEG4 system layer. This amendment is expected to become an international standard (IS) in 2005.

AMD4 and AMD5 deal with lossless audio coding.

Audio Lossless Coding (ALS) (AMD 4) is based on linear predictive coding (LPC) followed by noiseless coding. ALS is based on models proposed by NTT, Real Networks, and Berlin University. This amendment is expected to become an international standard by 2005. The other part of the AMD4 defines the use of parametric stereo toolset with HE-AAC. The resulting algorithm is known as *HE-AAC v2* (also called AAC++ or Enhanced AACplus) and gives good quality stereo below 48 kbps. Enhanced AACplus was adopted by 3GPP release 6 as an optional audio codec for MMS and streaming applications.

AMD5 defines Scalable loss-less coding (SLS) based on proposals by FhG, Microsoft and I2R. It uses AAC technology to encode the base layer at 128 kbps. The base layer is lossy but perceptually transparent. This is backward compatible with standard AAC. The enhancement layer uses IntMDCT to achieve lossless compression at 600-700 kbps for two channel audio. AMD5 is expected to become international standard by 2005.

AMD6 supports lossless coding of output/input of sigma delta modulated (SDM) based A/D & D/A converters. SDM technology uses over-sampling of signal to provide linearity over large dynamic range of audio signals (e.g. 24 bits). The AMD6 supports handling of over-sampled signals without unnecessary conversion to PCM and vice a versa. This is known as Direct Stream Transfer (DST). It is based on a proposal submitted by Philips. AMD5 is expected to become an international standard in 2005.

In addition to the above amendments, the MPEG4 committee has issued a call for proposals for **Spatial Audio Coding (SAC).** This will be an extension to AAC for multi-channels on similar lines to SBR. FhG/Agere, Coding Technologies/Philips, Dolby and Panasonic have submitted proposals in response to the call. The proposals are based on Binaural Cue Coding (BCC). The final model will be a combination of proposals from FhG/Agere and CT/Philips. The reference software model for this standard will be available in near future.

5. Non-MPEG Audio standards

A number of audio codecs have been developed outside the MPEG standardization framework and some of these codecs have become popular in the market. This section gives an overview of some of those codecs.

As a part of the Bluetooth standard for short-distance wireless applications, Philips developed a "sub-band codec" (SBC) that is optimized for the low delay and complexity demanded by wireless handheld applications. SBC is accepted as a mandatory audio codec in the Bluetooth standard for the audio profile. It is gaining traction at the expense of MPEG4 AAC LD.

Voiceage Inc. developed an extension of AMR-WB (Adaptive Multi Rate, Wideband) known as **AMR-WB+**. This combines the standard GSM AMR-WB speech codec with a technique known as TCX (Transform Coded eXcitation) to ensure good quality audio/speech coding between 14 kbps to 24 Kbps. It has been adopted by 3GPP release 6 as an optional audio codec for MMS and streaming applications. This algorithm will compete in the market with Enhanced AACplus (HE AAC v2) and HQ parametric audio from MPEG portfolio.

China is also developing its own standard for audio-video called **AVS** (Audio Visual standard). This standard has reached the final draft stage at the end of December 2004. The details will be known later after availability of standard in English.

Proprietary audio codecs are prominent especially in two market segments:

- Streaming and portable multimedia - Windows media audio (WMA) from Microsoft Inc and Real Audio (RA) from Real Network inc.
- Professional & high-performance audio - Audio codecs from Dolby Inc & DTS.

Windows Media Audio (WMA) is another popular format for streaming and portable multimedia. In its most recent version, WMA9, Microsoft has added 3 new algorithms to standard WMA - WMAVoice for low bit rate speech, WMAPro for multi-channel audio and WMALossless. MS is also planning newer standard WMAPROPLUS to improve coding efficiency for lower bit-rate.

In the same market segment, Real Networks has introduced the **Real Audio** codec – RA9. Real networks' media player products are also compatible with AAC.

Dolby Inc and DTS dominate the professional audio space. There is widespread usage of Dolby Digital (AC3) with major application being DVD Audio. Dolby has recently introduced new codecs such as Dolby-E, E-AC3 etc.

Sony has announced introduction of **ATRAC3plus** to support low bit-rate (less then 64 kbps) in the past. This is in succession to ATRAC3 (Adaptive TRansform Acoustic Coding) standard and it is widely used in MiniDisc and Ministick storage devices used in Walkman and digital still cameras.

6. Peeping into future

One of important consideration for success of any technology is the ecosystem of Digital rights management, copyrights, distribution methods & rights to alley fear rampant piracy in music industry. Although there are innumerable directions that various researchers are pursuing, there are visible trends that are difficult to hide. The AAC and its variations (e.g. HE-AAC, Surround and parametric extensions) will become the foundations for future improvements and will slowly replace legacy MP3 in usage as well as content. Lossless audio coding and its applications will grab a prominent mindshare thanks to the fidelity advantages that they offer. Proprietary codecs will continue hold market share in certain applications e.g. streaming, portable multimedia and professional audio. Audio coding is a happening space, enabling newer applications to consumers.

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About the Author

The author is a Technical Lead with the Multimedia Codecs group in Texas Instruments, Bangalore India. You can contact him at *Mihir@ti.com*. The views expressed in this article are the personal views of the author and do not reflect the opinion or views of Texas Instruments Inc or its subsidiaries.

UPCOMING EVENTS

September/ October

- Two Day Workshop on "Wireless LAN"
  Organised by M. S. Ramaiah School of Advanced Studies

  Date: 30th September & 1st October 2005
  Link: http://www.msrsas.org/cpd/seminars.htm

- AES 119th Convention, NY.
  Date: 7th –10th October, 2005
  Link: http://www.aes.org/events/119/

- 2005 IEEE WORKSHOP ON MULTIMEDIA COMPRESSION, Bangalore
  Date: 27th, 28th October, 2005
  Link: http://dsplab.ece.iisc.ernet.in/~ieeesp/imcw05/

December

- Second World Congress on Lateral Computing, WCLC 2005, Bangalore, India
  Date: 16th-18th December, 2005
  Link: http://www.lateral-computing.org/wclc/

February

- TI Developer Conference, Bangalore
  Date: 30th November – 1st December, 2005
  Link: http://tii.developerconference.ext.ti.com/

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