MPEG-4 HE-AAC v2

— audio coding for today's digital media world

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Delivering broadcast-quality content to consumers is one of the most challenging tasks in the new world of digital broadcasting. One of the most critical aspects is the highly efficient use of the available transmission spectrum. Consequently, a careful choice of compression schemes for media content is essential – for both the technical and the economical feasibility of modern digital broadcasting systems.

In the case of audio content, the MPEG-4 High Efficiency AAC v2 profile (HE-AAC v2) has proven, in several independent tests, to be the most efficient audio compression scheme available worldwide. It has recently been selected within DVB as part of its overall codec toolbox.

HE-AAC v2 comprises a fully-featured tool set for the coding of audio signals in mono, stereo and multichannel modes (up to 48 channels) – at high quality levels using a wide range of bitrates.

HE-AAC v2 has proven in several independent tests to be the most efficient audio compression scheme available worldwide. The codec's core components are already in widespread use in a variety of systems and applications where bandwidth limitations are a crucial issue, amongst them:

- ***XM Satellite Radio*** – the digital satellite broadcasting service in the USA;
- ***HD Radio*** – the terrestrial digital broadcasting system of iBiquity Digital in the USA;
- ***Digital Radio Mondiale*** – the international standard for broadcasting in the long-, medium- and short-wave bands.

In Asia, HE-AAC v2 is the mandatory audio codec for the Korean Satellite ***Digital Multimedia Broadcasting*** (S-DMB) technology and is optional for Japan's terrestrial ***Integrated Services Digital Broadcasting*** system (ISDB). HE-AAC v2 is also a central element of the ***3GPP*** (3rd Generation Partnership Project) and ***3GPP2*** specifications and is applied in multiple music download services over ***2.5*** and ***3G*** mobile communication networks.

This article gives an overview of the standardization, its technical components and compression efficiency, and provides an outlook on the future potential of ongoing development work.

International standardization

MPEG-4 HE-AAC v2 (also known as "aacPlus v2") is the combination of three technologies:

- Advanced Audio Coding (AAC);
- Spectral Band Replication (SBR);
- Parametric Stereo (PS).

All three technologies are currently being specified in ISO/IEC 14496-3 and combined in the ***HE-AAC v2*** profile, which is referred to in ISO/IEC 14496-3:2001/Amd.4.

The combination of AAC and SBR is called “HE-AAC” (also known as “aacPlus v1”) and is specified in ISO/IEC 14496-3:2001/Amd.1.

The European Telecommunications Standards Institute (ETSI) has standardized HE-AAC v2 in its Technical Specifications TS 102005 “**Technical Specification for the use of video and audio coding in DVB services directly delivered over IP**“ and TS 101 154 “**Implementation guidelines for the use of video and audio coding in broadcasting applications based on the MPEG-2 transport stream**”.

Based on these standardization efforts, HE-AAC v2 is available for integration into all kinds of DVB services.

Architecture of HE-AAC v2

The underlying core codec of HE-AAC v2 is the well-known MPEG AAC codec. ***AAC*** is considered state-of-the-art for transparent audio quality at a typical bitrate of 128 kbit/s. Below this rate, the audio quality of AAC would start to degrade, which can be compensated to a maximum degree with the enhancement techniques SBR and PS.

SBR is a bandwidth extension technique that enables audio codecs to deliver the same listening experience at approximately half the bitrate that the core codec would require, if operated on its own.

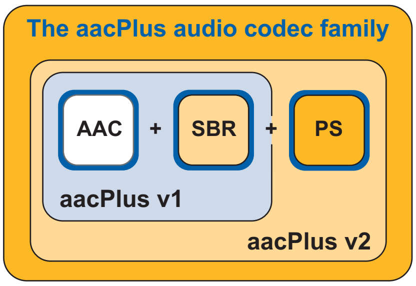

Figure 1
The HE-AAC v2 (aacPlus v2) family of audio codecs

Parametric Stereo increases the coding efficiency a second time by exploiting a parametric representation of the stereo image of a given input signal. Thus, HE-AAC v2 is a superset rather than a substitute for the AAC core codec and extends the reach of high-quality MPEG-4 audio to much lower bitrates. Given this superset architecture, HE-AAC v2 decoders are also capable of decoding plain AAC bit-streams, as well as bit-streams incorporating AAC and SBR data components, i.e. HE-AAC bit-streams. Hence, HE-AAC v2 is also a superset of HE-AAC, providing the highest level of flexibility for broadcasters as it contains all the technical components necessary for audio compression over a high bitrate range (see the section called *Audio quality evaluation* on page 7).

Another important feature of the HE-AAC and HE-AAC v2 architecture is the extremely flexible transport of ***metadata***. Metadata can be embedded as auxiliary data in a way that only compatible decoders take notice of their existence. Non-compatible decoders simply ignore the metadata. A high flexibility is provided in terms of type, amount and usage of the data. Metadata plays an important role in digital broadcasting, e.g. as content description data such as the name of an artist or song, or as system-related data such as control information for a given decoder. In broadcasting especially, metadata – such as DRC (Dynamic Range Control), DN (Dialog normalization), or down-mixing from multichannel to stereo – is widely used to achieve adequate reproduction of the original programme material in particular listening environments.

MPEG ISO/IEC 14496-3 Part 3 (Audio) defines the designated areas for metadata in an MPEG bit-stream. The maximum metadata capacity of a bit-stream depends on a number of different variables such as the sampling rate, and, of course, the bitrate of a given bit-stream. In general, the available amount of metadata to be included does not impose any restrictions on the aforementioned applications.

HE-AAC-encoded audio data can exist in a variety of file formats with different extensions, depending on the implementation and the usage scenario. Most commonly used file formats are the MPEG-4 file formats MP4 and M4A, carrying the respective extensions **.mp4** and **.m4a**. The ".m4a" extension is used to emphasize the fact that a file contains audio only. The 3GP file format supports all HE-AAC features for mono and stereo files up to a 48 kHz sampling rate. Additional file formats, such as MPEG-2 and MPEG-4 ADTS are also available, along with others.

MPEG AAC

Research on perceptual audio codecs started about twenty years ago. Earlier research on the human auditory system had revealed that hearing is mainly based on a short-term spectral analysis of the audio signal. The so-called ***masking effect*** was observed: the human auditory system is not able to perceive distortions that are masked by a stronger signal in the spectral neighbourhood. Thus, when looking at the short-term spectrum, a so-called ***masking threshold*** can be calculated for this spectrum. Distortions below this threshold are inaudible in the ideal case.

The goal is to calculate the masking threshold based on a psychoacoustic model and to process the audio signal in a way that only audible information resides in the signal. Ideally, the distortion introduced is exactly below the masking threshold and thus remains inaudible. *Fig. 2* illustrates the quantization noise produced by an ideal perceptual coding process.

If the compression rate is further increased, the distortion introduced by the codec violates the masking threshold and produces audible artefacts *(Fig. 3)*.

The main method of overcoming this problem in traditional perceptual waveform codecs is to limit the audio bandwidth. As a consequence, more information is available for the remainder of the spectrum, resulting in a clean but dull-sounding signal. Another method, called *intensity stereo*, can only be used for stereo signals. In intensity stereo, only one channel and some panning information is transmitted, instead of a left and a right channel. However, this is only of limited use in increasing the compression efficiency as, in many cases, the stereo image of the audio signal gets destroyed.

At this stage, research on classical perceptual audio coding had reached its limits, as the hitherto known methods did not seem to provide more potential to further increase coding efficiency. Hence, a shift in paradigm was needed, represented by the idea that different elements of an audio signal, such as spectral components or the stereo image, deserve different tools if they are to be coded more efficiently. This idea led to the development of the enhancement tools, Spectral Band Replication and Parametric Stereo.

Spectral Band Replication

In traditional audio coding, a significant amount of information is spent in coding the high frequencies, although the psychoacoustic importance of the last one or two octaves is relatively low. This

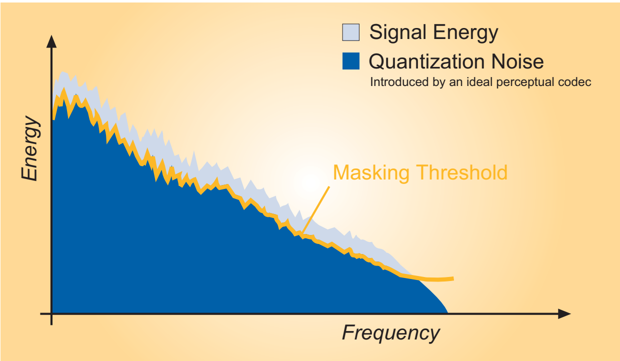

Figure 2
Inaudible quantization noise produced by an ideal perceptual coding process

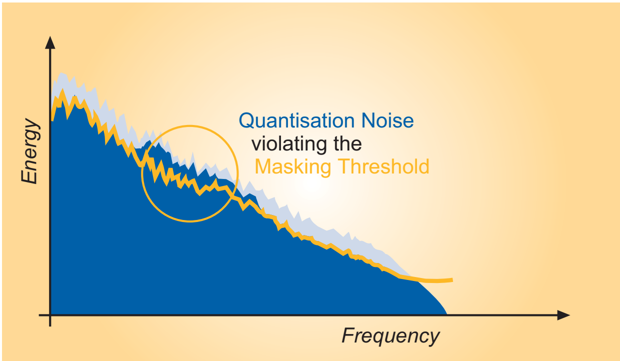

Figure 3
Waveform coding going beyond its limits: audible artefacts appear above the masking threshold

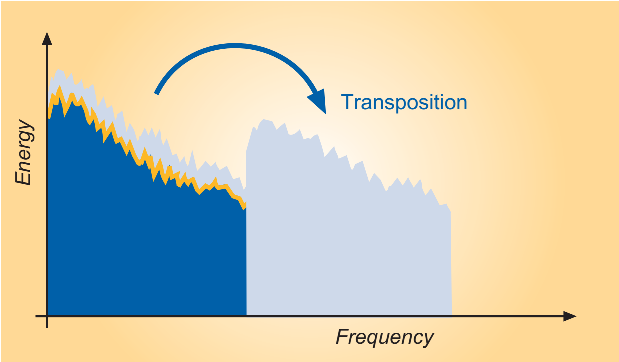

Figure 4
Creation of high frequencies by transposition

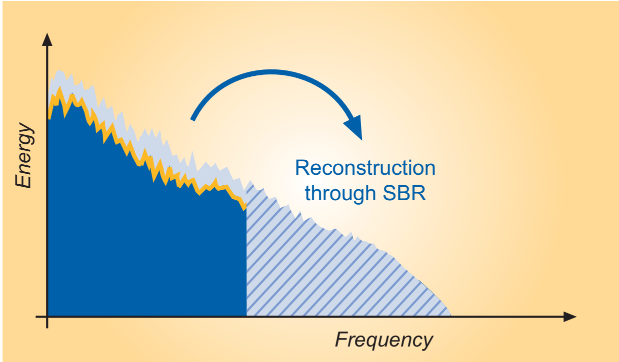

Figure 5
Envelope Adjustment of the high band

triggered the basic idea behind SBR. Based on the cognition of a strong correlation between the high- and the low-frequency range of an audio signal (hereafter referred to as the “high band” and the “low band” respectively), a good approximation of the original input signal high band can be achieved by a transposition from the low band *(Fig. 4)*.

Besides pure transposition, the reconstruction of the high band *(Fig. 5)* is conducted by transmitting guiding information such as the spectral envelope of the original input signal or additional information to compensate for potentially missing high-frequency components. This guiding information is referred to as SBR data. Also, efficient packaging of the SBR data is important to achieve a low data-rate overhead.

At the encoder side, the original input signal is analysed, the high band spectral envelope and its characteristics in relation to the low band are encoded and the resulting SBR data is multiplexed with the core coder bit-stream. At the decoder side, firstly the SBR data is de-multiplexed, then the core decoder is used on its own. Finally, the SBR decoder operates on its output signal, using the decoded SBR data to guide the Spectral Band Replication process. A full bandwidth output signal is obtained. Non-SBR decoders would still be able to decode the backwards compatible part of the core decoder ... but resulting in a band-limited output signal only.

Whereas the basic approach seems to be simple, making it work reasonably well in practice is not. Obviously it is a non-trivial task to code the guiding information such that all the following criteria are met:

- Good spectral resolution is required;
- Sufficient time resolution on transients is needed to avoid pre-echoes;
- Cases with non-highly-correlated low band and high band need to be taken care of carefully, since transposition and envelope adjustment alone could sound artificial here;
- A low overhead data-rate is required in order to achieve a significant coding gain.

The crossover frequency between the low band and the high band is chosen on the basis of different factors such as target bitrate and input sampling frequency. Generally the low band needs to cover the frequency range from DC up to around 4 to 12 kHz, depending on the target bitrate. The higher the crossover frequency between AAC and SBR, the higher the bitrate needed to fulfil the psycho-acoustic masking threshold of the AAC encoder.

The limited frequency range that is covered by the AAC coder allows the use of a low sampling frequency of ≤ 24 kHz which improves the coding efficiency significantly compared with using a higher sampling frequency of 48 or 44.1 kHz. Thus, aacPlus is designed as a dual-rate system, where AAC operates at half the sampling rate of SBR. Typical configurations are 16/32 kHz, 22.05/44.1 kHz or 24/48 kHz sampling rates, while 8/16 or even 48/96 kHz are also possible. The resulting audio bandwidth can be configured flexibly and may also be dependent on the application or audio content type.

The following table shows typical examples of the crossover frequency between AAC and SBR, as well as the audio bandwidth at a number of bitrates, using sample rates of 24/48 kHz in stereo, given a proper configuration of the HE-AAC encoder. The bitrate of the SBR data varies depending on the encoder tuning but, in general, it is in the region of 1 - 3 kbit/s per audio channel. This is far lower than the bitrate that would be required to code the high band with any conventional waveform coding algorithm.

Typical HE-AAC crossover frequency and audio bandwidth at different bitrates

Stereo bitrate (bit/s)	AAC frequency range (Hz)	SBR frequency range (Hz)
20 000	0 - 4 500	4 500 - 15 400
32 000	0 - 6 800	6 800 - 16 900
48 000	0 - 8 300	8 300 - 16 900

When combining AAC with SBR, the resulting codec is called HE-AAC or aacPlus v1. It has proven to fulfil all the criteria above and was standardized within MPEG-4 in 2003.

Parametric Stereo (PS)

Whereas SBR exploits the possibilities of a parameterised representation of the high band, the basic idea behind PS is to parameterise the stereo image of an audio signal such as "panorama", "ambience", or "time/phase differences" of the stereo channels to enhance the coding efficiency of the codec.

In the encoder, only a monaural downmix of the original stereo signal is coded after extraction of the Parametric Stereo data. Just like SBR data, these parameters are then embedded as PS side information in the ancillary part of the bit-stream.

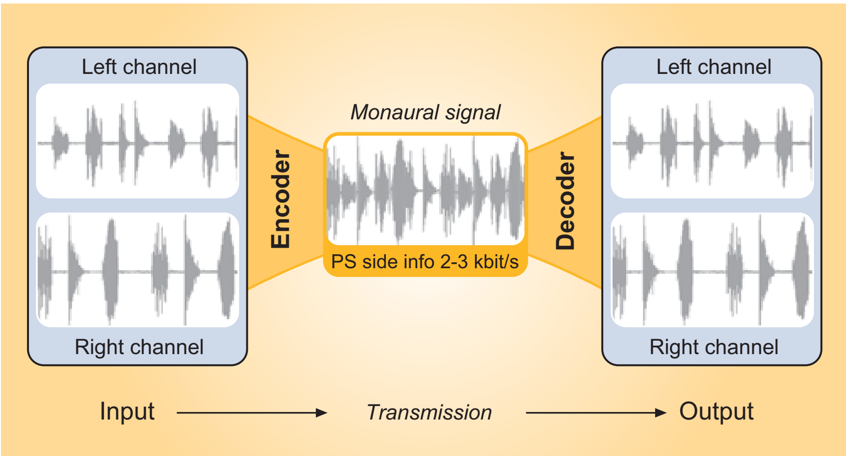

Figure 6
Basic principle of the parametric stereo coding process

In the decoder, the monaural signal is decoded first. After that, the stereo signal is reconstructed, based on the stereo parameters embedded by the encoder. *Fig. 6* shows the basic principle of the parametric stereo coding process.

Three types of parameters can be employed in a Parametric Stereo system to describe the stereo image.

- ***Inter-channel Intensity Difference*** (IID), describing the intensity difference between the channels.
- ***Inter-channel Cross-Correlation*** (ICC), describing the cross correlation or coherence between the channels. The coherence is measured as the maximum of the cross-correlation as a function of time or phase.
- ***Inter-channel Phase Difference*** (IPD), describing the phase difference between the channels. This can be augmented by an additional ***Overall Phase Difference*** (OPD) parameter, describing how the phase difference is distributed between the channels. The ***Inter-channel Time Difference*** (ITD) can be considered as an alternative to IPD.

Functionality of HE-AAC v2

The described technologies AAC, SBR and PS are the building blocks of the MPEG-4 HE-AAC v2 profile. The AAC codec is used to encode the low band, SBR encodes the high band, and PS encodes the stereo image in a parameterised form. In a typical aacPlus encoder implementation, the audio input signal at an input sampling rate of f_S is fed into a 64-band Quadrature Mirror Filter bank and transformed into the QMF domain.

If the Parametric Stereo tool is used (i.e. for stereo encoding at bitrates below ~36 kbit/s), the PS encoder extracts parametric stereo information based on the QMF samples. Furthermore, a stereo-to-mono downmix is applied. With a 32-band QMF synthesis, the mono QMF representation is then transformed back into the time domain at half the sample rate of the audio signal, $f_S/2$. This signal is then fed into the AAC encoder.

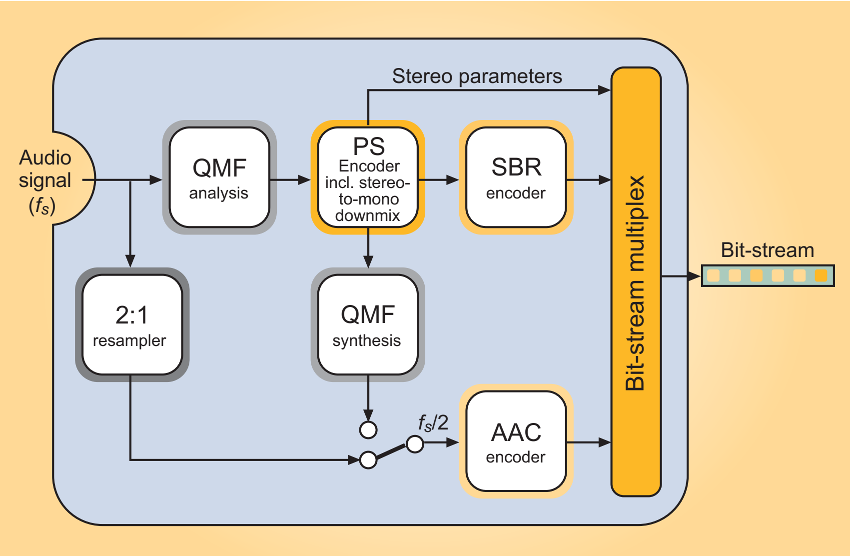

Figure 7
Block diagram of an HE-AAC v2 encoder

If the Parametric Stereo tool is not used, the audio signal is fed into a 2:1 resampler and, again, the downsampled audio signal is fed into the AAC encoder. The SBR encoder also works in the QMF domain; it extracts the spectral envelope and additional helper information to guide the replication process in the decoder. All encoded data is then multiplexed into a single bit-stream for transmission or storage.

Fig. 7 shows the block diagram of a complete HE-AAC v2 encoder.

In the HE-AAC v2 decoder, the bit-stream is first split into the AAC, SBR and PS data portions. The AAC decoder outputs a time domain low-band signal at a sample rate of $f_S/2$. The signal is then transformed into the QMF domain for further processing. The SBR processing results in a reconstructed high band in the QMF domain. The low and high bands are then merged into a full-band QMF representation.

If the Parametric Stereo tool is used, the PS tool generates a stereo representation in the QMF domain. Finally, the signal is synthesized by a 64-band QMF synthezis filter bank. The result is a time domain output signal at the full sampling rate f_S.

Fig. 8 shows the block diagram of a complete HE-AAC v2 decoder.

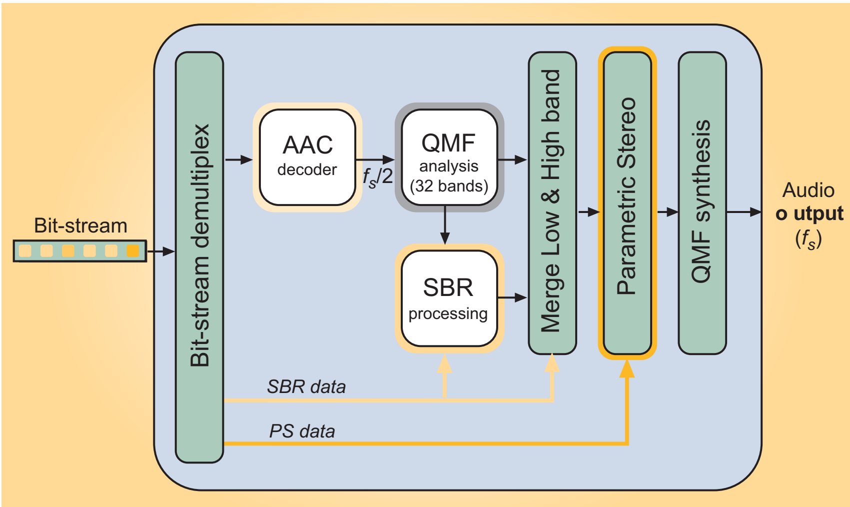

Figure 8
Block diagram of an HE-AAC v2 decoder

Audio quality evaluation

The audio quality of HE-AAC and HE-AAC v2 has been evaluated in multiple double-blind listening tests conducted by independent entities such as the European Broadcasting Union (EBU), the Moving Pictures Expert Group (MPEG), the 3rd Generation Partnership Project (3GPP), and the Institut für Rundfunktechnik (IRT).

EBU subjective listening test on low-bitrate audio codecs

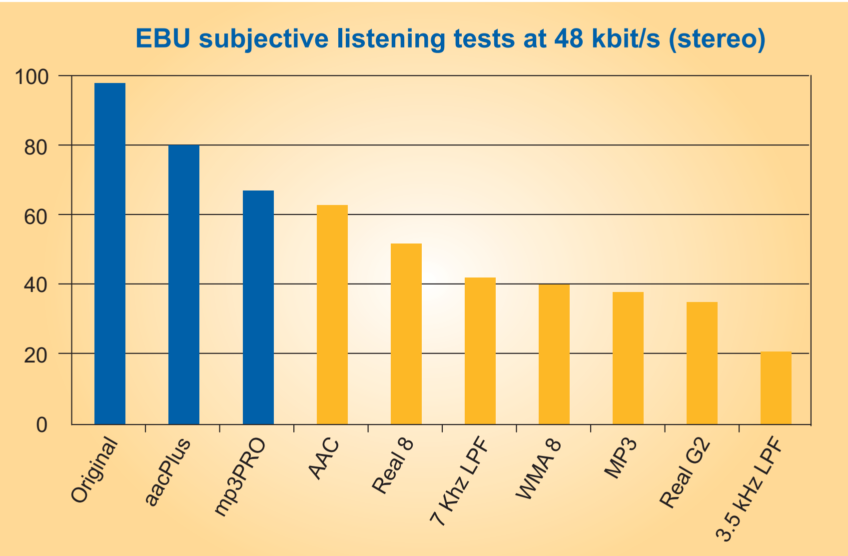

Figure 9
Results of the EBU subjective listening tests

In 2003, the EBU conducted a comprehensive test, evaluating a variety of open standard and proprietary audio codecs including HE-AAC (AAC+SBR), AAC, Windows Media Audio, and others, at a bitrate of 48 kbit/s [1].

The tests were conducted according to the MUSHRA test method (*MU*ltiple *S*timulus test with *H*idden *R*eference and *A*nchors). The results clearly show the superior compression efficiency of HE-AAC. Remarkably, the second best codec in the tests was mp3PRO – the combination of MPEG Layer-3 (mp3) and SBR.

MPEG and 3GPP listening tests

Prior to standardization of HE-AAC v2, MPEG carried out listening tests to verify the efficiency improvement of HE-AAC v2 (incorporating Parametric Stereo) over HE-AAC v1. The MUSHRA test method was also used for this evaluation. According to the scope of the listening tests, the bitrates used included 24 kbit/s for HE-AAC, and 32 and 24 kbit/s for HE-AAC v2.

The results of these tests showed a clear performance gain introduced by Parametric Stereo. At 24 kbit/s, HE-AAC v2

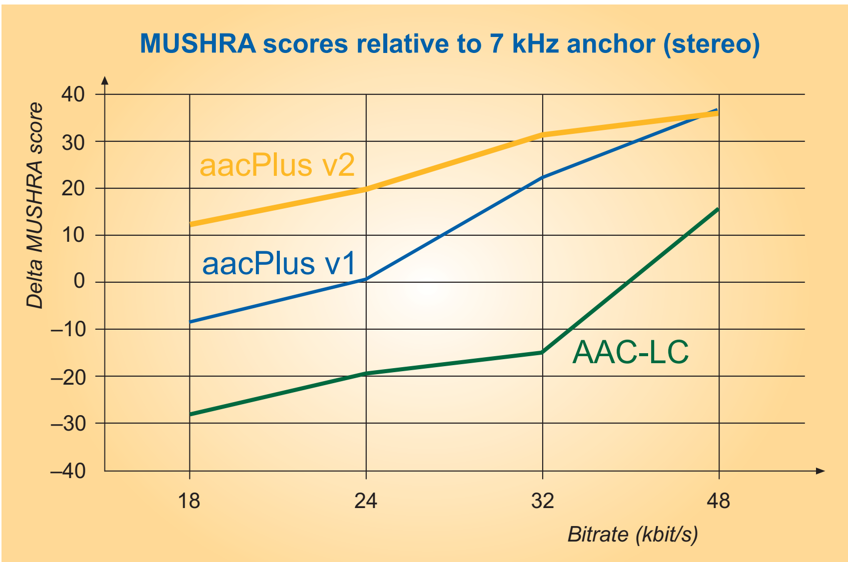

Figure 10
Performance of HE-AAC (aacPlus v1), HE-AAC v2 (aacPlus v2), and AAC

1. EBU Tech doc. 3296: **EBU subjective listening test on low bit rate audio codecs**
   http://www.ebu.ch/CMSimages/en/tec_doc_t3296_tcm6-10497.pdf

was seen to perform significantly better than HE-AAC and equal to or better than HE-AAC at 32 kbit/s. A second test conducted by 3GPP predictably displayed similar results, also at additional bitrates.

Multichannel listening tests of the Institut für Rundfunktechnik (IRT)

In 2004, the IRT conducted listening tests comprising a number of audio codecs for multichannel applications, amongst them HE-AAC, Dolby AC-3 and Windows Media. The results showed a clear advantage for HE-AAC which demonstrated significantly higher audio quality at 160 kbit/s compared to Dolby AC-3 operating at 384 kbit/s and Windows Media at 192 kbit/s.

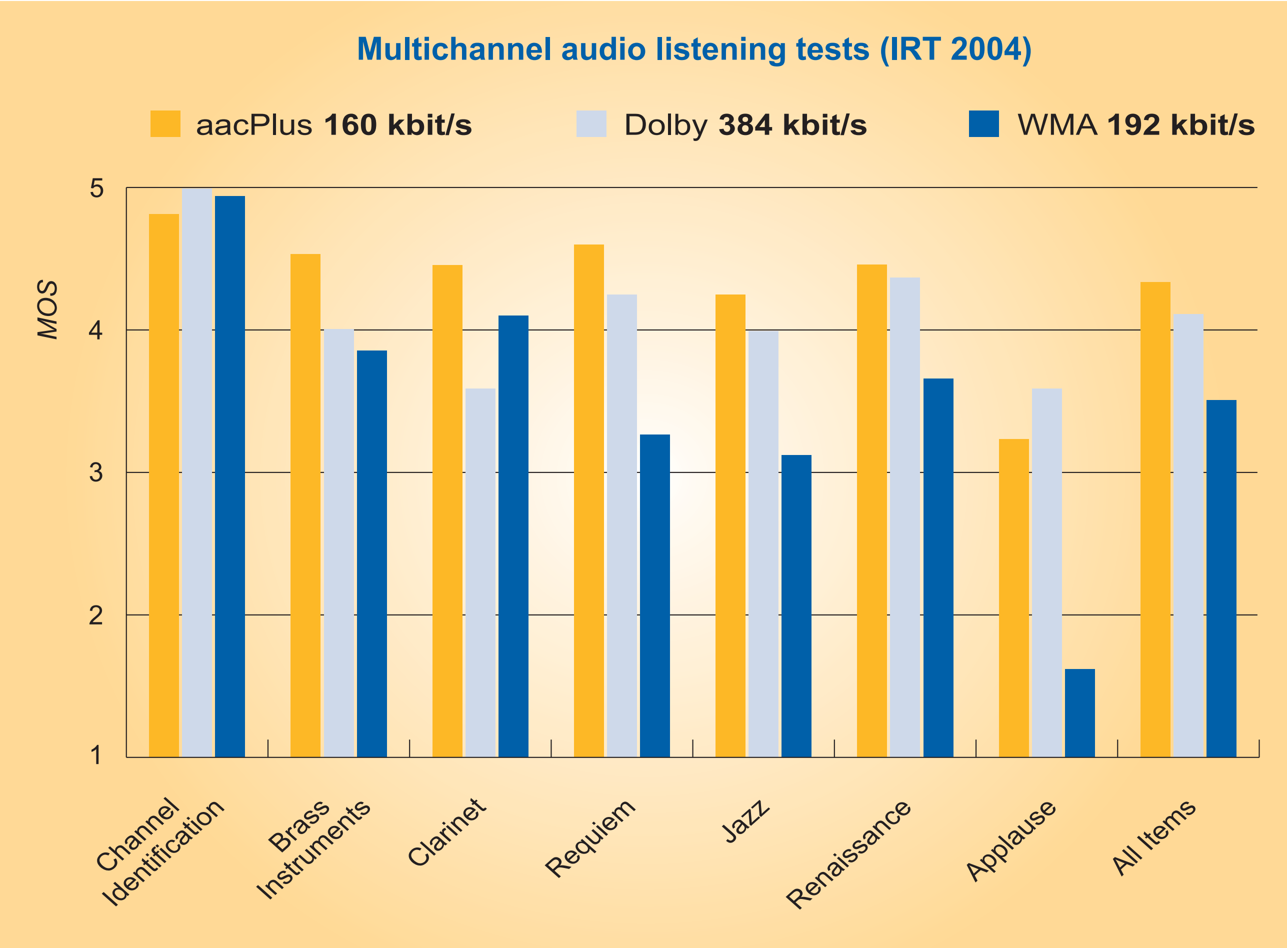

Figure 11
Results of the multichannel audio codec tests performed by the IRT

Fig. 11 shows the Mean Opinion Scores (MOS) of the coded audio signals from the original. Although HE-AAC operated at the lowest bitrate of all codecs, it outperformed all competitors in terms of audio quality. Calculating the average MOS of all the audio items under test, it can be stated that HE-AAC provides a better quality at half the bitrate compared with WMA or Dolby AC-3.

Interpretation of the combined results

Considering the fact that the quality of compressed audio signals scales with the bitrate, the following interpretation of the available test results can be made.

Combining AAC with SBR and PS into HE-AAC v2 results in a very efficient audio codec, providing high audio quality over a wide bitrate range, with only moderate gradual reduction of the perceived audio quality towards very low bitrates. *Fig. 12* gives an impression of the anticipated audio quality vs. bitrate for the various codecs of the HE-AAC v2 family.

The diagram shows only a smooth degradation in audio quality of HE-AAC v2 towards low bitrates over a wide range down to 32 kbit/s. Even at bitrates as low as 24 kbit/s, HE-AAC v2 still produces a quality far higher than that of any other audio codec available.

For multichannel 5.1 signals, HE-AAC provides a coding efficiency that is a factor of two higher than Dolby AC-3.

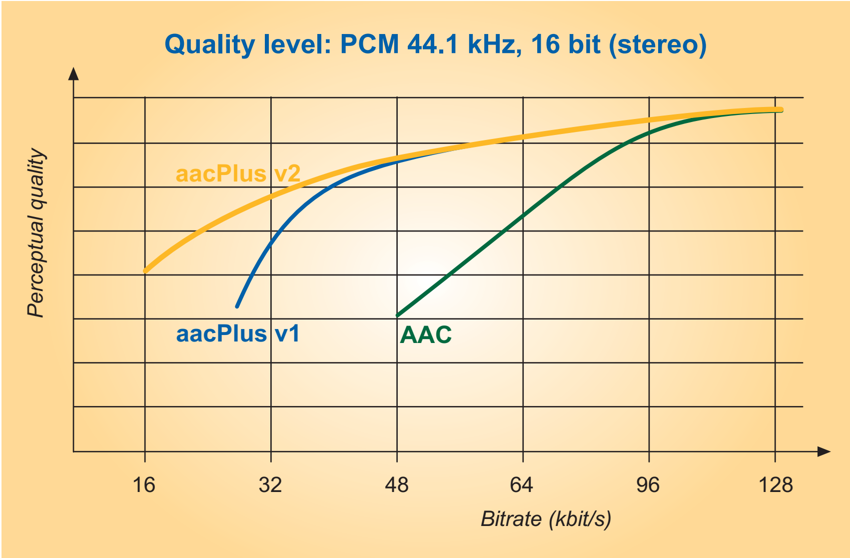

Figure 12
Comparison of audio quality degradation of HE-AAC, HE-AAC v2 and AAC

Availability of the HE-AAC v2 codec family in products and applications

Hardware implementations

After finalisation of the initial MPEG standard, the HE-AAC v2 codec family has quickly established itself as the common denominator codec across multiple media applications. Crucial to this progression was the availability of implementations on commonly-used embedded hardware platforms. Today, the following implementations of the MPEG-4 HE-AAC codec family are available, optimized in terms of memory usage, processing power and error robustness:

Target device	Encoder	Decoder
ARM	X	X
TI c64	X	X
TI c55	X	X
TI c67	X	X
Blackfin 533	X	X
Motorola DSP 56k	X	X
ZSP		X
SH Mobile		X
ARC A4		X
Equator BSP-15		X
CEVA		X
Tensilica Xtensa HiFi 2 Audio Engine	X	X

Abbreviations

3GPP	3rd Generation Partnership Project	**IRT**	*Institut für Rundfunktechnik GmbH* (German broadcast engineering research centre)
AAC	(MPEG) Advanced Audio Coding	**ISDB**	Integrated Services Digital Broadcasting
DC	Direct Current	**ISO**	International Organization for Standardization
DMB	Digital Multimedia Broadcasting	**MOS**	Mean Opinion Score
DN	Dialogue Normalization	**MPEG**	Moving Picture Experts Group
DRC	Dynamic Range Control	**MUSHRA**	(EBU) MUlti Stimulus test with Hidden Reference and Anchors
DTS	Digital Theatre Systems	**PS**	Parametric Stereo
DVB	Digital Video Broadcasting	**S/PDIF**	Sony/Philips Digital InterFace
ETSI	European Telecommunication Standards Institute	**SBR**	Spectral Band Replication
HE-AAC	High Efficiency AAC	**WMA**	(Microsoft) Windows Media Audio
IEC	International Electrotechnical Commission		

All implementations are compliant with relevant standards such as MPEG and 3GPP, and support plain AAC as well as HE-AAC v1 and HE-AAC v2, and include advanced error-concealment strategies. Additionally, fixed-point firmware reference code is available, allowing implementations for proprietary or currently unsupported platforms.

Software applications

Besides the above-mentioned processor-specific implementations, HE-AAC is also available as software development kits for PC platforms running the Windows, Linux and Macintosh operating systems. This has led to the adoption of HE-AAC by popular software applications for streaming and file playback. Amongst others, majors players such as RealPlayer and Winamp include HE-AAC: the latter also providing HE-AAC ripping/encoding from audio CDs. Professional encoding products for real-time encoding/streaming and file encoding are available from companies such as Orban, Mayah, and Cube-Tec. The adoption of HE-AAC has enabled Internet radio services to stream audio content in high quality, even to users with standard modem connections.

DTS and Coding Technologies for digital broadcasting

In recent years, with the advent of the DVD and A/V receivers, multichannel audio systems have become more and more popular in home entertainment environments. This trend presents broadcasters with a readily installed, advanced decoder base through which to deliver multichannel audio content. As stated earlier in the article, HE-AAC delivers high quality, discrete 5.1 multichannel audio at bitrates as low as 160 kbit/s, offering twice the efficiency of other currently used formats. Backwards compatibility with existing A/V receivers can easily be addressed by the combination of HE-AAC and DTS Coherent Acoustics, enabling the adoption of HE-AAC audio coding for digital broadcasting in a backwards compatible way.

Together with DTS, Coding Technologies has created a solution combining an HE-AAC decoder and a DTS encoder, both residing in a consumer set-top-box. The HE-AAC encoded broadcast signal gets decoded in the set-top box, and encoded by the DTS encoder at the maximum possible bitrate (1.5 Mbit/s). The DTS-encoded bit-stream can then be transmitted to the A/V receiver over the S/PDIF interface and played back in high audio surround sound quality.

The aacPlus/DTS solution offers considerable advantages over existing multichannel broadcast systems. For example:

- The bandwidth reduction for aacPlus transmissions compared to competing systems exceeds a factor of 2; thus, for example, two language channels may be broadcast instead of one.
- Feasible bandwidth savings for each audio channel equal a saving of more than $US 100,000 in transponder leasing costs per annum.

- The obvious benefits of an open standard. Any encoder manufacturer is able to implement the aacPlus encoder, and broadcasters and network operators will be able to choose from a plethora of vendors, in accordance with their procurement policies.
- Total cost of ownership savings will accrue from multiple sources; e.g. truly integrated encoder solutions, single vendor support services, training, operations etc.

Conclusions

This article has described the new HE-AAC v2 audio codec, and how existing audio-coding technologies, such as MPEG AAC, can be significantly enhanced by using the novel enhancement techniques – SBR and PS. Preliminary studies show that the compression efficiency of AAC can be increased by a factor of up to four.

HE-AAC v2 – the combination of AAC, SBR and PS – is undoubtedly the most powerful audio codec available today. It is thus the first choice for all application scenarios where bandwidth is limited or very expensive, such as digital broadcasting or mobile applications.

Outlook

The latest MPEG standardization effort relating to compression efficiency is based on the idea of extending the Parametric Stereo approach towards multichannel coding. Under the name ***MPEG Surround***, a new system is currently being specified that is expected to reach final standardization in July 2006. MPEG Surround exploits similarities between the single channels of a multichannel signal, aiming at a parameterised representation of the surround sound information.

On the encoder side, for example, a stereo downmix of a multichannel signal can be created. Parameters of the information residing in the surround channels are then extracted and embedded as parameterised surround information in the auxiliary part of the bit-stream. The decoder then decodes the stereo downmix and recreates the full multichannel signal, based on the parameterised surround information. The MPEG Surround data in the bit-stream are only recognized by decoders capable of decoding MPEG Surround, and ignored by any other decoder. Hence, legacy decoders only decode the stereo downmix generated by the encoder.

This approach provides an easy upgrade path to multichannel audio for existing digital broadcast systems, without affecting legacy stereo-only decoders. Combining e.g. MPEG Layer-2 audio with

Stefan Meltzer studied electrical engineering at the Friedrich-Alexander University in Erlangen Germany. After he received his Dipl.-Ing. degree in 1990, he joined the Fraunhofer Institute for Integrated Circuits (IIS) in Erlangen. After working in the field of IC design for several years, in 1995 he became the project leader for the development of the WorldSpace Satellite Broadcasting system at the Fraunhofer Institute. From 1998 until 2000 he led the development team at the Institute for the development of the XM Satellite Radio broadcasting system.

In 2000, Mr Meltzer joined Coding Technologies in Nuremberg, Germany. He currently occupies the position of Vice President for business development.

Gerald Moser studied communications at the Universities of Essen and Mainz in Germany. After receiving his Master of Arts degree in 1998, he joined the Fraunhofer Institute for Integrated Circuits (IIS) in Erlangen to work in the field of Internet Radio. In 1999, he joined the audio department of the IIS as communications representative. Since 2002, Mr Moser has been working at Coding Technologies as Manager, Communications and Public Relations.

MPEG Surround enables broadcasters to offer multichannel content to consumers with MPEG Surround enabled equipment, whereas any stereo decoder would still play back the stereo signal. Most importantly, the added value of multichannel content can be introduced without the need for simultaneous transmission of stereo and multichannel content, and even at very moderate transmission bandwidth costs of only a few kilobits per second.

Combining the new MPEG Surround technology with HE-AAC promises high audio quality at bitrates in the range of 64 - 96 kbit/s for 5.1 multichannel audio signals, thus introducing another gain in coding efficiency by a factor of 2 - 3 for this configuration. This combination extends the existing family of codecs towards an even more powerful open standard that is suitable for all kinds of application scenarios – from mobile music to home theatre – in a forward and backwards compatible way.