



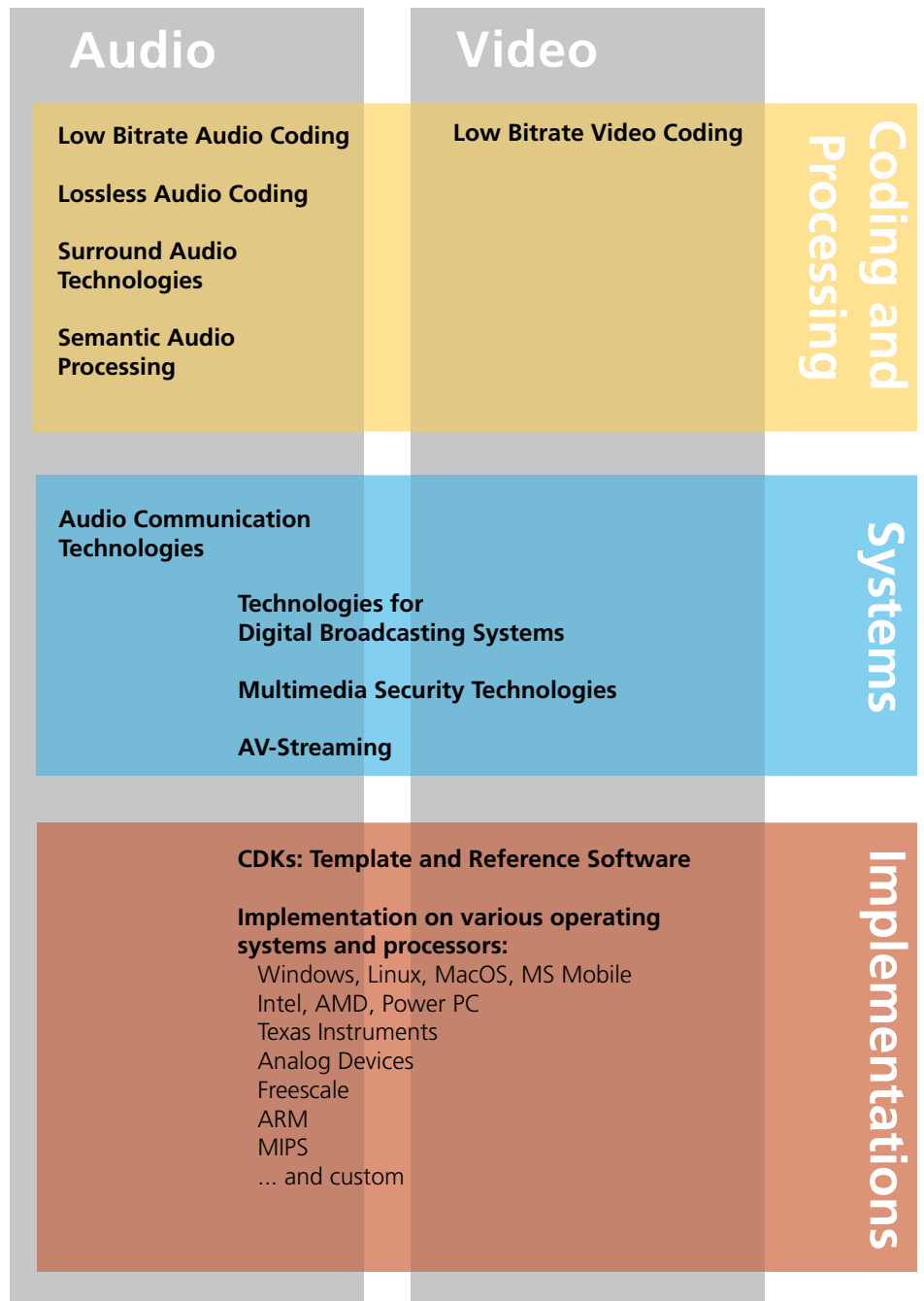
Audio and Multimedia



Fraunhofer Institut
Integrierte Schaltungen

Audio and Multimedia

Fraunhofer IIS is the leading international research lab in the field of high-quality low bit rate audio coding. In 20 years of creating innovative solutions for audio-visual applications on multimedia devices and desktop PCs, Fraunhofer IIS has established an extensive knowledge base. In the business field "Audio and Multimedia" of Fraunhofer IIS, a team of over 100 engineers focuses on the development, enhancement and implementation of codecs and multimedia systems.



Working fields of Fraunhofer IIS Audio and Multimedia

Audio and Multimedia

Low Bitrate Audio Coding

mp3

The very popular mp3 audio coding format was mainly developed at Fraunhofer IIS in the late 1980s and early 1990s. In 1992, it was standardized as MPEG-1 Audio Layer 3 and, in 1994, extended towards MPEG-1/2 Audio Layer 3. Today, mp3 is the most popular audio codec: Virtually any consumer music playback device supports mp3. Licensing of Fraunhofer mp3 software and patents is handled by Thomson. For more information please visit www.mp3licensing.com.

MPEG-2/4 AAC

MPEG AAC features high coding efficiency, flexibility and multi-channel capability. With MPEG AAC, perceptually transparent quality is reached at only 64 kbps per channel. Bit rates of more than 256 kbps per channel and sampling rates up to 192 kHz are possible granting an audio quality vastly superior to CD Audio. MPEG-2 AAC, standardized in 1997, today is widely used, e.g. in portable music players as well as in ISDB,

the Japanese digital broadcasting system for radio and television. MPEG-4 AAC provides enhanced versatility by adding a number of coding tools to MPEG-2 AAC technology. The coding performance increases, and extensions for robustness in error-prone environments as well as support for scalability are introduced. With MPEG-4 software players such as QuickTime and with the enormous success of Apple iTunes and iPod, AAC has attained a broad public awareness.

MPEG-4 HE-AAC (v2)

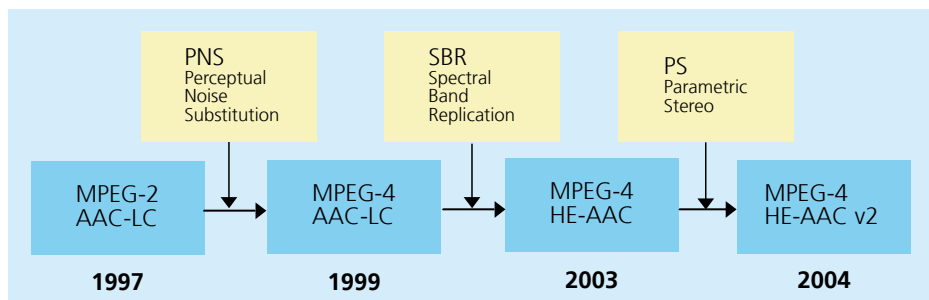
MPEG-4 High Efficiency AAC is the lowest bit rate codec within MPEG. HE-AAC is a combination of MPEG-4 AAC with SBR (Spectral Band Replication). In HE-AAC v2 the Parametric Stereo tool was added to even enhance the coding efficiency. Thus, good stereo quality can be achieved at bit rates of 16 to 48 kbps. Due to its outstanding efficiency and audio quality, HE-AAC has already been employed in a large number of applications such as Digital Radio Mondiale DRM, DAB+, XM Satellite Radio, 3GPP, mobile TV systems (e.g. DVB-H, DMB), and mobile music download services.



Lossless Audio Coding

HD-AAC

Lossless audio coding schemes offer audio data compression, with the decoded audio data being bit by bit identical to the original audio data. Based on the MPEG-4 Scalable to Lossless Coding (SLS) technology, HD-AAC extends proven AAC encoding to a lossless and near-lossless compression. The flexible scalability of the extension makes this combination an ideal solution for production environments, where the result is to be transmitted to several recipients through channels of differing bandwidth (for example HD or SD broadcasting). For private and professional use in music archives, HD-AAC makes it possible storing only one file that includes both, the lossless version of a song for playback on e.g. home theatre systems and the highly compressed portion which can be transferred e.g. to portable AAC playback device without time-consuming transcoding operations.



Evolution of MPEG AAC

Audio Communication Technologies

MPEG-4 AAC-LD

AAC-LD is the low delay audio codec within MPEG-4 and is the de facto standard for high quality video conferencing. It features an algorithmic delay of only 20 ms while offering good compression ratio and high sound quality for all kinds of audio signals. It is the best choice codec for high-quality audio and video conferencing systems, bi-directional communication, or Audio over IP applications, preserving the acoustic ambiance.

AAC-ELD (Enhanced Low Delay AAC)

High audio quality, low coding delay and very low data rates: AAC-ELD is the perfect choice for any delay critical application that demands full audio bandwidth at data rates down to 24 kbps. AAC-ELD combines the strengths of its two main components: MPEG-4 AAC-LD and Spectral Bandwidth Replication (SBR). Whereas MPEG-4 AAC-LD features low encoding/decoding latency, SBR provides high quality audio at very low bit rates. AAC-ELD is currently under standardization in MPEG. The finalization of the standard is expected for the end of 2007.

ULD (Ultra Low Delay Audio Coding)

The Ultra Low Delay audio codec enables high quality low bit rate audio coding with an algorithmic delay down to 1.5 ms. Such a very low encoding and decoding delay is an essential prerequisite, for instance, for wireless speakers, microphones, or headphones and distributed music productions where artists perform simultaneously in different locations.



Audio Communication System

The Fraunhofer IIS Audio Communication System ensures high audio quality even over poor connections. The Audio over IP SDK allows easy integration into existing applications. The SDK includes support of AAC family in mono and stereo (AAC-LC, AAC-LD, AAC-ELD, HE-AAC and HE-AACv2), advanced echo cancellation, instant switching between codec types, sophisticated error concealment providing intelligible speech up to 30% packet error rates, adaptive jitter buffer control optimizing the trade-off between buffering-delay and late-loss, and log files and trace files for remote analysis and debugging.

Surround Audio Technologies

mp3 Surround

mp3 Surround is the backward compatible surround extension to mp3. It achieves high-quality multi-channel sound at a total bit rate of 128-192 kbps for 5.1 channels while maintaining excellent stereo quality on all mp3 players and devices. It is for instance the ideal solution for introducing surround sound to Internet radios. mp3 Surround is available for licensing through Thomson at www.mp3licensing.com. Demo songs and evaluation software are available at www.mp3surround.com and www.all4mp3.com.

MPEG Surround

MPEG Surround is a generic surround extension, which can be associated with perceptual audio codecs or PCM data while remaining fully backward compatible to stereo or even mono. The achieved audio quality has proven to be excellent in independent listening tests despite the fact that the surround image is represented by a very low additional bit rate down to 4 kbps. Combined with MPEG-4 HE-AAC, MPEG Surround provides 5.1 channel surround sound at a total bit rate of 64 kbps or less. The ISO standard MPEG Surround can be used to upgrade existing distribution infrastructures for stereo or mono audio content towards the delivery of multi-channel audio while retaining full compatibility with existing receivers.

Ensonido®

The Ensonido technology renders multi-channel audio material for stereo headphones, modeling the natural reception of surround sound by the human ear. That way, listeners get the full surround experience using ordinary stereo headphones. It can be used in combination with MPEG Surround, mp3 Surround or discrete multi-channel formats such as MPEG-4 HE-AAC. For evaluation software, please visit www.ensonido.com.

Semantic Audio Processing

SX Pro®

SX Pro is a high quality and highly flexible upgrade solution for legacy stereo material: It offers the possibility to up-mix two-channel content into multi-channel surround sound. The SX Pro upmix process is based on a semantic analysis of the input signal that guarantees a high degree of control over the resulting sound image. Thus, SX Pro is the best choice for any type of application where two-channel legacy content is transmitted over or played back in a multi-channel environment.

mp3 SX (mp3 Stereo eXtended)

mp3 SX is an addition to the mp3 Surround technology. It offers the possibility to upgrade two-channel content into the mp3 Surround format. mp3 SX licenses are available as part of the mp3 and mp3 Surround licensing program. For evaluation software, please visit www.mp3surround.com.

SAOC (Spatial Audio Object Coding)

The upcoming standard MPEG SAOC allows highly efficient storage and transport of individual audio objects (e.g. voices, instruments, ambience, etc.) in an audio mix, while preserving the possibility for the listener to adjust the mix based on his personal taste. That includes changing the rendering configuration of the audio scene from stereo over surround to even binaural (see Ensonido®) reproduction. Interactive audio mixing (e.g. Karaoke) or clear and comfortable teleconferencing solutions will be the most prominent applications for this research topic.



Video Coding

MPEG-4 Part 2

MPEG-4 Part 2 offers a variety of tools for visual coding. These tools are combined in sub-sets called profiles. The most commonly used profiles for video coding are the Simple Profile (SP) and the Advanced Simple Profile (ASP). SP and ASP are the best choice for the traditional rectangular shaped video. The Core and ACE Profile, also offered by Fraunhofer IIS, are used for arbitrarily shaped video.

MPEG-4 AVC/H.264

MPEG-4 AVC, also known as MPEG-4 Part 10 or ITU-T H.264, is the current state-of-the-art video coding standard in terms of coding efficiency. Using more sophisticated coding algorithms, it can save up to 50 percent bit rate compared to MPEG-4 Part 2 or MPEG-2 video at the same subjective visual quality.

AVC is widely adopted in several application areas, from small sized mobile video in 3GPP or mobile TV, to Internet video services up to high definition video for HDTV or HD-DVD/Blu-Ray.

Real-time Implementations

A team specialized on the efficient implementation of multimedia technologies on various PC and DSP platforms as well as operating systems guarantees Fraunhofer IIS clients instant access to turnkey software solutions.

CDKs (Core Design Kits)

For real-time implementation of multimedia codecs and components on fixed point processors, CDKs are available as reference software. They are optimized for devices with low resources in terms of memory and computational power, e.g. mobile players, PDAs or smartphones. The CDKs are written in C or C++ and exist in two different versions. One is directly compileable for 16 bit or 32 bit fixed point processors, the other one is a template code for DSPs with fractional or integer arithmetic of any word length.

Technologies for Digital Broadcasting Systems

Fraunhofer DRM ContentServer™ (for Digital Radio Mondiale)

The professional broadcast system encodes audio in real-time, provides data services like NewsService Journaline, TextMessages and EPG, and supports all DRM features including "Announcement" and "Alternative Frequency Signaling" as well as automatic configuration scheduling.

Fraunhofer DAB/DMB ContentServer™

The Fraunhofer DAB/DMB ContentServer for Eureka 147 Digital Audio and Multimedia Broadcasting is a professional and easy to use encoder system for the DAB broadcast chain. It features real-time audio and video encoding (MPEG Audio Layer 2, MPEG Surround, MPEG-4 HE-AACv2 (DAB+), MPEG-4 AVC) as well as generation and insertion of almost any data service in PAD and NPAD such as NewsService Journaline, Dynamic Labels, and EPG.

NewsService Journaline®

Journaline, the "teletext for radio", provides an intuitive and direct access to contextually produced and hierarchically structured text information that is transmitted over digital broadcasting systems. Integration in DRM, DMB or DAB receivers is easy and cost efficient.

Multimedia Streaming and IPTV

For audiovisual streaming applications Fraunhofer IIS provides key components such as advanced audio and video codecs (MPEG-4 AVC/H.264, MPEG-4 HE-AAC, MPEG Surround), protocols (RTP, RTSP), file formats (MP4, 3GP, DCF), and encryption systems (ISMACryp, HECA). This allows, for example, high-quality audio-visual services in SD resolution with 5.1 surround sound at bit rates below 2 Mbps.

Mobile TV: DVB-H

Fraunhofer IIS offers fully integrated DVB-H Player Modules and SDKs supporting RTP-Streaming and MP4-file playback of MPEG-4 codecs. These player modules are designed to reduce development time for system integrators and are offered as C/C++ source code, SDKs or custom designed libraries for existing DVB-H frameworks.

Multimedia Security Technologies

Digital Rights Management Systems

Fraunhofer IIS develops user-friendly and secure Digital Rights Management (DRM) systems for Internet-based services, cellular phone systems and broadcasting systems. These DRM solutions are based on open standards such as OMA DRM 2.0.

Watermarking

Watermarking is an essential technology for multimedia security. Fraunhofer IIS has years of experience developing high performance audio watermarking solutions and customizing the system for specific applications. Latest developments include technologies for ultra low watermark decoder complexity.

Conditional Access: HECA

HECA (High Efficient Conditional Access) is designed to meet the special requirements of digital broadcasting systems with low bandwidth capacity. The CA related overhead is kept to a minimum while at the same time enabling full functionality in terms of business models. Latest security algorithms are used. Adaptations exist for various broadcasting systems including DAB, Digital Radio Mondiale and TPEG.

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