Fraunhofer Institut
Integrierte Schaltungen

MPEG AAC Family (Advanced Audio Coding)

The MPEG AAC family is the consequent continuation of the truly successful audio codec MPEG Layer-3, widely known as MP3. With bit rates ranging from 24 kbit/s to 256 kbit/s, AAC (Advanced Audio Coding) combines excellent coding efficiency with highest audio quality and is fully multi-channel capable.
Fraunhofer IIS offers quality and resource-optimized software implementations of the MPEG AAC encoding and decoding algorithms for various platforms.
Fraunhofer IIS is the leading research laboratory in the area of high-quality low bit rate audio coding and is "the Home of MP3". A creative team with more than 15 years of experience in the field of audio coding ensures state-of-the-art solutions tailored to the customer's needs.

Overview

Development of AAC started in 1994 by a collaboration of Fraunhofer IIS, AT&T, Dolby, and Sony; only three years later the new format became part of the MPEG standard as MPEG-2 AAC. Later, in the development of the MPEG-4 Audio standard, AAC was further enhanced and amended, e. g. for delay-critical applications or for scalable encoding of multimedia content. Finally, today's most efficient audio coding formats were standardized in 2003 and 2004 within MPEG-4 Audio: High Efficiency AAC (HE-AAC) and HE-AAC v2.

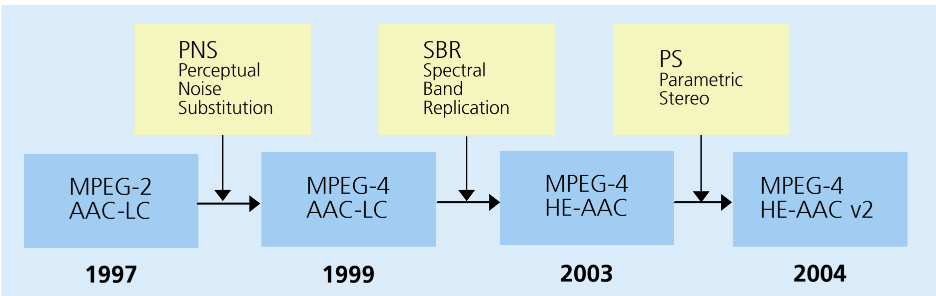
Key Features

»MPEG AAC«

MPEG AAC provides excellent audio quality. Reaching perceptually transparent quality at only 64 kbit/s per channel, it fulfills the requirements for broadcast quality as defined by the European Broadcasting Union. With sampling rates ranging from 8kHz up to 96kHz and above, with bit rates up to 256 kbit/s, and with support for up to 48 channels, MPEG AAC is one of the most flexible audio codecs. Of course, the standard also supports mono, stereo, and all common multichannel configurations (e. g. 5.1 or 7.1). The low computational demands make AAC the ideal codec for any low bit rate high-quality audio application.

»MPEG-4 HE-AAC«

HE-AAC is the low bit rate codec in the AAC family and is a combination of the AAC LC (Advanced Audio Coding Low Complexity) audio coder and the SBR (Spectral Band Replication) bandwidth expansion tool. This combination achieves good stereo quality already at bit rates of 32 to 48 kbit/s. HE-AAC is also known as aacPlus and can be used in multi-channel operations.



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»MPEG-4 HE-AAC v2«

Combined with parametric stereo, the HE-AAC codec provides good audio quality starting at bit rates around 16 to 24 kbit/s for stereo content. HE-AAC v2 is also known as aacPlus v2.

Applications

Due to their coding efficiency, AAC and HE-AAC are prime candidates for any digital audio system with limited storage or transmission capacities. The following examples are just a snap-shot of the increasing number of applications:

The most successful music download shop, Apple iTunes Music Store, uses AAC to deliver its content to the millions of customers around the globe. The portable Apple iPod has got Fraunhofer IIS technology inside as well.

HE-AAC v2 is part of the 3GPP standard for the delivery of audio content to 3G devices.

The Japanese Association of Radio Industries and Businesses (ARIB) selected MPEG-2 AAC as the only audio coding scheme for all of Japan's digital broadcast systems.

In 2001 the satellite based XM Radio started service using HE-AAC.

The DRM Consortium (Digital Radio Mondiale) develops new digital services for the current analog services in long, medium, and shortwave bands. DRM has selected HE-AAC as audio coding scheme.

HE-AAC is a mandatory format for 2nd Session DVD-Audio and is part of the DVD-AR (Recordable DVD-Audio) specification.

”Fraunhofer inside“: The Apple ipod uses Fraunhofer IIS technology

AAC Solutions by Fraunhofer IIS

PC Software

Two different types of encoder software and one decoder are available for PC platforms. The ”professional“ encoder is designed for maximum audio quality, the ”consumer“ encoder is targeted to achieve maximum encoding speed without noticeably compromising audio quality. The PC decoder is an MPEG-4 compliant natural audio decoder.

Core Design Kit Software (CDK)

The Fraunhofer IIS CDKs are bit-precise reference and template codes optimized in terms of memory requirements and processing power. They are written in C and are available in two different versions: directly compilable for 16-bit or 32-bit fixed-point processors (ARM, MIPS, PowerPC, ADI, TI, Motorola and more) or as a template code for DSPs with fractional or integer arithmetic of any word length.

Digital Signal Processor (DSP) Software and Platforms

Highly optimized AAC source code or libraries are available for several DSPs. Fraunhofer IIS offers different levels of support for the integration of the software into target systems.
A library can be delivered together with example code and API documentation. Additional hardware dependent adaptations can be done to deliver a ready-to-run bootable binary executable.