



MPEG-4 AAC-LD
Low-Delay High-Quality
Audio Coding



High-quality video conferencing

**Fraunhofer Institute for
Integrated Circuits IIS**

Executive Director

Prof. Dr.-Ing. Heinz Gerhäuser

Director

Prof. Dr.-Ing. Günther Elst

Am Wolfsmantel 33
91058 Erlangen, Germany
Phone +49 (0) 91 31/7 76-0
Fax +49 (0) 91 31/7 76-9 99
info@iis.fraunhofer.de
www.iis.fraunhofer.de

Contact

Dipl.- Ing. Manfred Lutzky
amm-info@iis.fraunhofer.de
www.iis.fraunhofer.de/amm

**Fraunhofer USA, Inc.
Digital Media Technologies***

100 Century Court
Suite 504
San Jose, California 95112
www.dmt.fraunhofer.org

Contact

Jan Nordmann
Phone +1 408 573 9900
codecs@dmf.fraunhofer.org

* Fraunhofer USA Digital Media Technologies, a division of Fraunhofer USA, Inc., promotes and supports the products of Fraunhofer IIS in the U. S.

Low Delay Advanced Audio Coding (AAC-LD) is the high-quality low-delay audio coding standard within MPEG-4. It features an algorithmic delay of only 20 ms while offering good compression ratios and high sound quality for all kinds of audio signals including speech, music and atmospheric sounds.

Traditionally, there was no way to accomplish high-quality low bit rate two-way communication: On the one hand, perceptual audio codecs like MPEG Layer 3 (mp3) offer good audio quality for all kinds of signals, including music and speech, at bit rates down to 16 kbit/s. But the algorithmic delay of such encoder/decoder chains easily exceeds 100 ms and thus is not acceptable for two-way communication. On the other hand, the algorithmic delay of speech coding schemes, such as G.722.2, is small enough to make

them suitable for two-way communication. However, such schemes only provide good quality for speech signals and usually fail when used for more complex audio signals like music. Furthermore, the sound quality is usually not satisfactory for high-quality applications, such as realistic conferencing (Figure 1).

MPEG-4 AAC-LD is designed to combine the advantages of perceptual audio coding with the low delay necessary for bi-directional communication. The codec was developed by Fraunhofer IIS and is derived from MPEG-4 Low Complexity Advanced Audio Coding (AAC-LC). AAC-LD features an algorithmic delay of only 20ms, good audio quality for all kinds of signals and multi-channel support. In addition, AAC-LD can be used together with MPEG-4 Error Resilience Tools to alleviate the impact of transmission errors on the audio quality.

Applications

MPEG-4 AAC-LD is the perfect choice for any high-quality low-delay communication application. It has been deployed in many telephone and video-conference systems for example from Tandberg, Cisco and others. Texas Instruments licensed AAC-LD for its VoIP-DSP solutions and Apple has elected it for its PC based VoIP application iChat. The codec is also used in broadcasting chains for example in Zephyr transceiver from Telos. AAC-LD has been standardized in ETSI Dect Group as the only super wideband communication codec for NG DECT. This makes AAC-LD an attractive extension for wireless telephone devices.

Key Features

- Suitable for all kinds of audio signals including speech and music
- Audio quality better than ITU-T G.722/G.722.1-C, G.729.1 at the same bit rate
- Audio quality better than mp3 at the same bit rate
- Algorithmic delay of only 20 ms
- Delay of real-time DSP implementation down to 30 ms
- Multi-channel support
- Large range of bit rates possible
- Computational and storage complexity comparable to MPEG-4 AAC-LC
- Error resilient AAC-LD audio object type

Performance

The perceived audio quality of AAC-LD was compared to G.722.1-C and mp3 using speech items. The results show that AAC-LD at 32 kbit/s offers better quality than G.722.1-C at the same bit rate (Figure 2).

Availability

Codecs developed by Fraunhofer IIS stand for optimum audio quality at any given data rate. Fraunhofer IIS contributes to AAC-LD applications at the following levels:

- Encoder and decoder source or object code for Windows, Linux and Mac OS X
- Encoder and decoder Core Design Kits (CDK) for fixed-point processors of any word length
- Object code libraries and source code for several 16 and 32 bit architectures, including ARM, MIPS, PowerPC, ADI Blackfin, TI TM-S320C64 and others
- Error robust against packet loss up to 30 percent (understandable speech up to 50 percent packet loss rate)
- Low delay streaming system with additional features for echo cancellation, adaptive jitter buffer management, and seamless switching between several MPEG codecs for example AAC-LD and HE-AAC

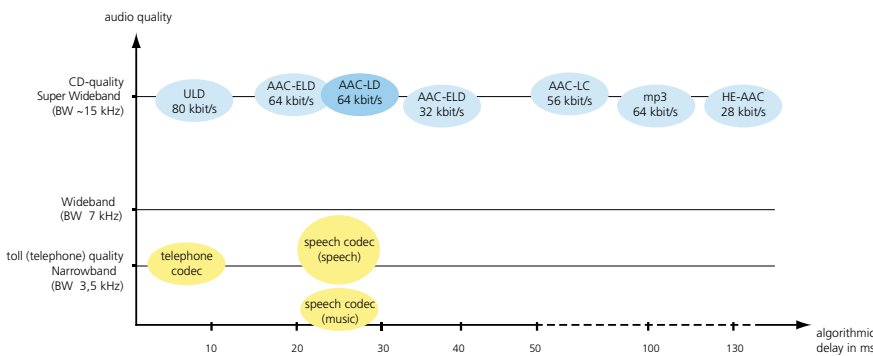


Figure 1: Comparison of audio quality and delay of chosen audio and speech codecs (mono bit rate)

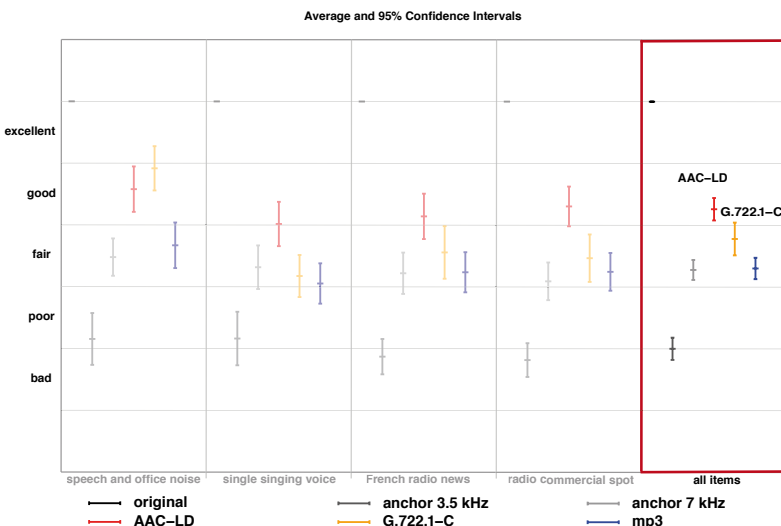


Figure 2: Comparison of AAC-LD, G.722.1-C and mp3 using speech items