Fraunhofer Institut
Integrierte Schaltungen

Audio Coding with Ultra Low Encoding/ Decoding Delay

ULD application example:
Wireless head sets

Fraunhofer Institute for Integrated Circuits IIS

Director
Prof. Dr.-Ing. Heinz Gerhäuser

Am Wolfsmantel 33
91058 Erlangen
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/amm

Contact
Dipl.-Ing. Manfred Lutzky
manfred.lutzky@iis.fraunhofer.de

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Introduction

The use of digital audio coding in professional audio productions is nowadays common practice. For some productions very low encoding and decoding delay have become an essential prerequisite. In live productions with wireless microphones and simultaneous monitoring or in distributed productions where artists perform simultaneously in different studios the tolerable total delay time is less than ten milliseconds. Such a threshold can hardly be surpassed by means of standard audio coding schemes like MPEG-1 Layer 3 (MP3), MPEG-2 Advanced Audio Coding (AAC) and MPEG-4 Low Delay of which the delays range from 20 milliseconds at 48 kHz up to several hundreds of milliseconds which would be too large for the delay-critical applications mentioned above. The Fraunhofer Ultra Low Delay Coder (ULD) meets this challenging requirement posed by delay-critical applications while achieving a compression ratio comparable to those of the standards mentioned above. There is an ULD Core Design Kit (CDK) software version available which can be directly compiled for 16/32 bit fixed point architectures. This allows applications with reduced power consumption like portable devices. An optimized implementation on ADI BF 53x is also available.

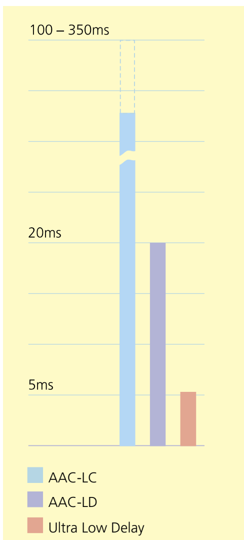
Technological Background

The ULD codec achieves this very low encoding/decoding delay by utilizing predictive coding rather than transform coding. The advantage of predictive coding is that there is no trade-off between delay and compression ratio.

Earlier approaches of predictive coding encountered severe problems when being applied along with psychoacoustic models. The Fraunhofer researchers provide their solution to these problems by applying a psychoacoustically controlled pre-filter to the input signal to reduce irrelevant information within the audio signal. The pre-filtered signal is then processed by the prediction section to remove redundancy. In a further step the resulting difference signal is entropy coded.
With its delay of about six milliseconds at a sample rate ranging from 32 kHz to 48 kHz the ULD codec offers high perceptual quality for delay-critical audio coding.

Key Features

- Ultra low delay of about 6 ms
- Comparable compression to state-of-the-art standard coders like mp3
- High audio quality at 70 kbps
- Large range of feasible bitrates down to 32 kbps
- Delay reduced version: below 3ms
- Complexity reduced version



Comparison of algorithmic delays of several coding schemes