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Review on High Efficiency MPEG4 AAC Audio Encoder and Decoder

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Abstract: In recent years, a well-known audio coding, MPEG Layer-3 (MP3), is widely used and maintains a good audio quality. However, a new audio standard, MPEG AAC audio coding [1], becomes more and more popular since it has better audio quality than MP3. AAC is the most advanced MPEG standard for digital audio compression. However, based on the characteristics of complex control and irregular data flow, AAC algorithm seems to be difficult in architecture design. AAC has been standardized by ISO and IEC, as part of the MPEG-2 and MPEG-4 specifications. A part of the AAC known as High Efficiency Advanced Audio Coding (HE-AAC) which is part of MPEG-4 Audio is also adopted into digital radio standards like DAB+ and Digital Radio Mondiale, as well as mobile television standards DVB-H and ATSC-M/H. This paper focuses on the mono channel, sampling frequency of 44.1 KHz; bitrates is 128Kb/s, Low Complexity profile implementation of the coder, which represents the configuration that is best suited for consumer electronics applications.

The research focuses on the investigation & implementation of low complexity profile high quality MPEG4 AAC Audio Decoder at a sampling frequency of 44 KHZ on a Field programmable gate array. Through we will try to implement IMDCT filter bank, Noiseless decoder, Inverse quantiser and Scale factor application modules of MPEG- 4 Advanced Audio Coding decoder more efficiently.

Index Terms: MPEG-4, AAC, Architecture

INTRODUCTION

MPEG-4 audio provides the coding and composition of natural and synthetic audio objects at a very wide range of bitrates. In order to achieve the highest audio quality within the full range of bitrates, three types of coding structures are incorporated into the standard, **as** shown in Fig. 1. GA (General Audio) is mainly intended for generic audio coding at high Bitrates. CELP and Parametric are for speech at medium and low bitrates, respectively. [1]

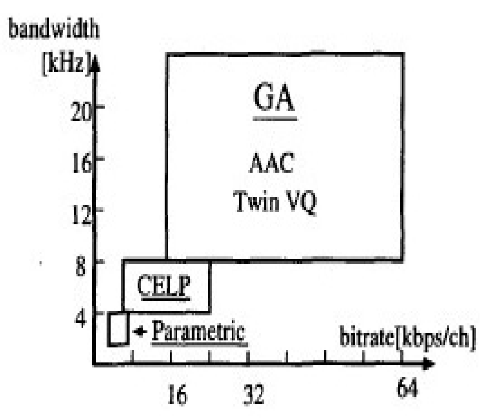

Figure 1: Bitrate and bandwidth of MPEG-4 audio

The AAC audio coding is an international standard first be created in MPEG-2 AAC (ISO/IEC 13818-7) and is the base of MPEG-4 general audio coding. AAC supports up to forty-eight audio channels. Sample rates supported range from 8 kHz to 96 kHz. The LC profile achieves nearly the same audio quality as the Main profile, but with significant savings in memory and processing requirements. With this mode, it is possible to decode the bit stream into a PCM signal having one of a variety of different sample rates [2].

As the high coding efficiency with Low power has become the issue of most importance, a new MPEG standard, called advanced audio coding (MPEG-2 AAC, ISO/IEC 13818-7) has been standardized. This new standard exhibits many advantages over other MPEG audio standards. The MPEG-4 AAC standard provides very high audio quality without compatibility-based restrictions.[3]

LITERATURE REVIEW

An International Standards Organization/Moving Pictures Experts Group (ISO/MPEG) audio coding standard for stereo CD-quality audio was adopted in 1992 after four years of extensive collaborative research by audio coding experts worldwide. ISO 11172-3comprises a flexible hybrid coding technique, which incorporates several methods including subband decomposition, filter bank analysis, transform coding, entropy coding, dynamic bit allocation, nonuniform quantization, adaptive segmentation and psychoacoustic analysis. [3]

In addition the Layer III provides several measures to reduce pre-echo. First, the modifications psychoacoustic model detects the conditions for pre-echo. Second, the bit reservoir reduces quantization noise when pre-echo conditions exist.

Table: AAC Decoder profile-Tools

Finally, the transform coding (MDCT) can switch to smaller MDCT block size to reduce the effective time window. MPEG-1 coders accept 16-bits PCM input data at sample rates of 32, 44.1, and 48 kHz. MPEG-1(1992) offers separate modes for mono,

Tool Name	Required/Optional	Main	LC	SSR
Bit-stream formatter	Required	√	√	√
Noiseless decoding	Required	√	√	√
Inverse quantization	Required	√	√	√
Scale-factors	Required	√	√	√
M/s	Optional	√	√ /x	√ /x
Prediction	Optional	√	x	x
Intensity/Coupling	Optional	√	√ /x	x
TNS	Optional	√	√ /x	√
Filter-bank	Required	√	√	√
Gain control	Optional	x	x	√

stereo, dual independent mono and joint stereo. Available bit rate are 32 - 192 kb/s for mono and 64 - 384 kb/s for stereo.

MPEG-2 extends the capabilities offered by MPEG-1 to support the so called 3/2 channel format with left, right, center, and left and right surround channels. The first MPEG-2 standard was backward compatible with MPEG-1 in the sense that 3/2 channel information transmitted by an MPEG-2 standard-2 encoder can be correctly decoded for two-channel presentation by an MPEG-1 receiver. The second MPEG-2 standard sacrificed backward MPEG-1 compatibility to eliminate quantization noise unmasking artifacts which are potentially introduced by the forced backward compatibility. The MPEG-2 coder is extended downward to include 16, 22.05, and 24 kHz, of PCM input data of sample rate. [4]

The MPEG-1 and MPEG-2 architecture contains three layers of increasing complexity, delay, and output quality. Each layer represents a family of coding algorithms. These layers are denoted in Roman figures (Layer I, Layer II, and Layer III). Different layers have been defined and they all have their own advantages.

The MPEG-2 AAC standard employs high-resolution filter banks, prediction techniques, and noiseless coding. It is based on recent evaluations and definitions of tools, each having been selected from a number of proposals. The self-contained tools include an optional pre-processing, a filter bank a perceptual coding, prediction, M/S stereo coding, quantization, noiseless coding, and a bit-stream multiplexer. The filter bank is a 1024-line modified discrete cosine transform and the perceptual model is taken from MPEG-1(mode 2). The temporal noise shaping tool controls the time dependence of the quantization noise, intensity, and M/S coding and the second-order backward-adaptive predictor improves coding efficiency. The predictor reduces the bit rate for coding subsequent subband samples in a given subband, and it bases its prediction on the quantized spectrum of the previous block, which is also available in the decoder. [6]

Finally, for quantization and noiseless coding, an iterative method is employed so as to keep the quantization noise in all critical bands below the global masking threshold. In order to serve different needs, the standard provides three profiles: (i) the main profile offers highest quality (ii) the low-complexity profile works without prediction (iii) the sampling-rate-scalable profile offers the lowest complexity.

MPEG-4 audio encompasses a great deal more functionality than just perceptual coding. It comprises an integrated family of algorithms with wide-ranging provisions for scalable, object-based speech and audio coding at bit rates from as low as the 200 b/s up to 64 kb/s at per channel. The distinguishing features of MPEG-4 relative to its predecessors are extensive scalability, object-based representations, user interactivity/object manipulation, and a comprehensive set of coding tools available to accommodate almost any desired trade-off among bit rate, complexity, and quality. [7]

AAC MAIN PROFILE

MPEG-4 AAC main profile can work at a bitrate of 64kbpslch or .less and high sampling rate of 44.1/48kHz. Each AAC decoding process is executed frame by frame, which is composed of 1,024 samples. Table.1 shows a decoding process of this AAC main profile.

Main: The Main profile is used when memory cost is not significant, and when there is substantial processing power available.

Low Complexity

The Low Complexity profile is used when RAM usage, processing power, and compression requirements are all present. In the low complexity profile, prediction, and gain control tool are not permitted and TNS order is limited.

Scalable Sampling Rate

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In the Scalable Sampling Rate profile, the gain-control tool is required. Prediction and coupling channels are not permitted, and TNS order and bandwidth are limited.

ARCHITECTURE:

The architecture of the system is shown in Fig.2.The overall system consists of a dedicated processing core and two hardwired logic modules. The hardwired logic modules are the Huffman decoding module and the prediction module. Due to the high computational loads of Huffman decoding tool, its architecture design must be focused on the fast operation. Since prediction tool needs high accuracy operations, high computational power, and a large amount of memory requirement, it must be designed to contain floating-point arithmetic unit and external memory modules.

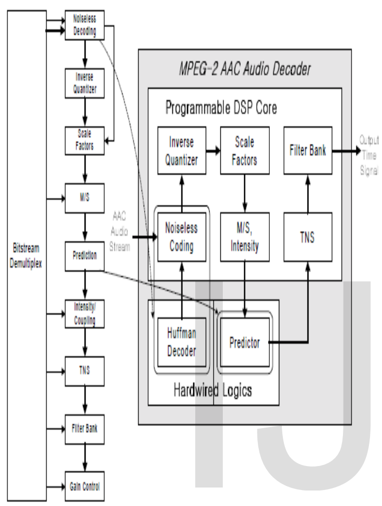

Figure2.: The overall architecture of the AAC decoding system

The developed system will be coded in VHDL. The processing core is a fixed-point programmable processor suitable for audio signal processing. It performs noiseless coding, inverse quantizer, scale factors, M/S intensity stereo, TNS, and filter bank tool.

The designed system has hybrid architecture of a fixed-point processing core for the software implementation and two hardwired logic modules: Huffman decoder module and predictor module.The system will be designed using VHDL and tested on the FPGA.

CONCLUSION

This paper has proposed review on an architecture of MPEG-4 audio decoder, which can realize high quality sound for a portable music player. MPEG4 AAC Audio Decoder at a sampling frequency of 44 KHZ on a Field programmable gate array. At the operation frequency of 4MHz, this decoder can play 2 channel sound at 44kHz sampling rate, dissipating low-power of 4.54mW with SNR of 51.3. Development is continuing on devising whole MPEG-4 codec architecture.

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