An Efficient Method to Improve the Audio Quality Using AAC Low Complexity Decoder

D. Kalyanasundaram,
PG Scholar,
Nandha Engineering College,
Erode, Tamilnadu, India.
E-Mail-kalyanduraisamy@gmail.com

K. Santhakumar,
ASP/ECE Department,
Nandha Engineering College,
Erode, Tamilnadu, India.
E-Mail-ecebank@gmail.com

Abstract— This paper presents a new approach to design a Digital Audio Broadcast (DAB) audio decoder is introduced to improve the superiority of audio. Countries all over the world use DAB broadcasting systems more prominently, in Europe. DAB+ is the upgraded version of digital audio broadcasting. DAB and DAB+ coexist in many countries, so receivers are essential to be compatible with both standards. DAB+ is approximately twice as efficient as DAB due to the adoption of the AAC+ audio codec, and DAB+ can provide high quality audio with bit rates as low as 64 kbit/s. Integrating an MPEG-1 Layer II (MP2) decoder and Advanced Audio Coding Low Complexity (AAC LC) decoder provides a fundamental audio decoding for DAB and DAB+. The generated audio frames data from the DAB channel decoders are stored in RAM. The bit stream demultiplexer parses the quantized spectrum data in the audio. The inverse quantization performs the inverse quantization computation and synthesis filter generates the time domain Pulse Code Modulation (PCM) samples, all the above operation results writes them back to the audio RAM. The existing system of this project uses HE AAC V2 decoder, that system consists has SBR and PS technologies. This two technologies are used to improve the sound quality in low bit rate program. The proposed scheme is uses AAC LC and MP2 decoder it improve the sound quality in high bit rate. The simulation of this project is carried out by using MATLAB R2011a and Xilinx ISE 9.2i.

Keywords— *AAC LC, DAB, DAB+, MPEG, PCM, RAM.*

*****

1. INTRODUCTION

In a modern electronic system, audio compression is necessary to store up and transport the high quality audio information in resourceful manner. A lot of audio compression standards have been industrial and innovations are still going on. The majority famous and broadly used audio coding standards are the Moving Pictures Experts Group (MPEG) standards. The first MPEG audio standard is MPEG1-LayerI (MP1), which was adopted in the VCD scheme. MP1 is now considered mostly outdated and has been replaced by MPEG1-LayerII (MP2) and MPEG1-Layer III (MP3). MP2 is able to realize high audio quality at bit rates in the range of 64 to 192 kb/s per channel. Now become the mainly broadly used audio format. In MPEG2, a new audio coding standard, namely advanced audio coding (AAC), was introduced to additional improve the audio compression rate. In MPEG4, high-efficiency AAC was introduced by brushing the AAC low-complexity (LC) profile. The AAC audio code 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. Sample rates supported range from 8 kHz to 96 kHz. The LC profile achieves practically 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. DIGITAL AUDIO BROADCASTING

DAB stands for Digital Audio Broadcast, a technology developed in 1980's as a solution for the exhausted bandwidth in FM and AM frequency ranges. AM and FM, which are analog methods of broadcasting, is replaced by digital broadcasting method DAB and its newer standard DAB+ released in 2006. Countries all over the world use DAB broadcasting systems more prominently, in Europe.

Digital Audio Broadcasting (DAB) is a digital radio technology for broadcasting radio stations, used in several countries particularly in Europe. As of 2006 approximately 1002 stations worldwide broadcast in the DAB format. The DAB customary was initiated as a European scientific research within the Nineteen Eighties. The Norwegian Broadcasting Corporation launched the terribly initial DAB channel within the world on 1995. DAB receivers DAB could provide a lot of radio programmes over a selected spectrum than analogue FM radio. DAB is a lot of strong with respect to noise and multipath attenuation for the mobile listening since DAB reception quality initial degrades speedily once the signal strength falls below a crucial threshold, whereas FM reception quality degrades gradually with the decreasing signal. Audio quality varies reckoning on the bitrate used and audio material. Most stations use slightly rate of 128 kbit/s or less with the MP2 audio codec, which needs a 160 kbit/s to realize perceived FM quality.

ISSN: 2321-8169
1315 - 1319

An upgraded version of the system was discharged in 2007,that is named DAB+. DAB isn't forward compatible with DAB+, which implies that DAB-only receivers don't seem to be able to receive DAB+ broadcasts. However, broadcasters will combine DAB and DAB+ programs within constant transmission and then create a progressive transition to DAB+. DAB+ is roughly double as economical as DAB owing to the adoption of the AAC+ audio codec, and DAB+ will offer top quality audio with bit rates as low as 64kbit/s. Reception quality is additionally additional strong on DAB+ than on DAB owing to the addition of Reed-Solomon error correction coding.

DAB AND AM/FM COMPARED

Traditionally radio programmes were broadcast on completely different frequencies via AM and FM, and therefore the radio had to be tuned into every frequency, as needed. This ran down a relatively great amount of spectrum for a comparatively little range of stations, limiting listening alternative. DAB may be a digital radio broadcasting system that through the applying of multiplexing and compression combines multiple audio streams onto a comparatively slender band targeted on one broadcast frequency known as a DAB ensemble. at intervals associate overall target bit rate for the DAB ensemble, individual stations is allotted completely different bit rates. the amount of channels at intervals a DAB ensemble is exaggerated by lowering average bit rates, however at the expense of the standard of streams. Error correction below the DAB customary makes the signal additional strong however reduces the whole bit rate obtainable for streams.

3. DAB AUDIO DECODER

AAC LC DECODER FLOW ALGORITHAM

There are several difficult algorithms used in the AAC decoding stream which make it hard to implement as resourceful design. There are three different outline defined in AAC. They are the main profile, the low-complexity (LC) profile, and the scalable sampling-rate profile. It allows tradeoffs in audio quality and encoding/decoding difficulty for different applications. AAC is the majority advanced MPEG standard for digital audio compression. The design flow of the AAC LC profile is shown in Figure 1.

The comparing three profiles, the LC profile can present high audio quality as the main profile but with reduction in memory and processing requirements. The AAC audio coding, there are switch to two type of audio formats. One is an Audio Data Interchange Format (ADIF) and another one is Audio Data Transport Stream (ADTS). The ADIF format it used to all data scheming the decoder into a single header preceding the actual audio stream. It is useful for file replace but does not permit for break-in or start of decoding at any position in time. The ADTS format, it packs AAC data into frames with headers and allows decoding to begin in the middle of an audio bitstream. There are several tools used in AAC decoding, which include bit stream parser, Huffman decoding, pulse data decoding, Inverse Quantizer (IQ), rescale, Middle/Side (M/S) and perceptual noise substitution (PNS), Intensity Stereo, Temporal Noise Shaping (TNS), and Filter bank.

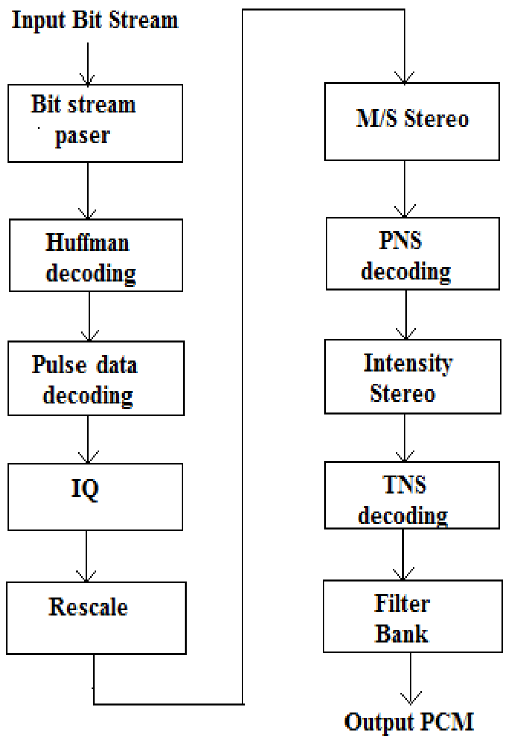

Figure 1: Design Flow of the AAC LC Profile

The first block is bit stream parser, which is used to extracts the audio frame signal and information about decoding that are used in the following decoding tools. The Huffman decoding have 12 Huffman code books. The one code book used for scale-factor coding and remaining eleven code books used for spectrum coding. The spectrum Huffman decoding are two stages. The one stage is the Huffman decoding which unpack the Huffman code index. The another one stage is regrouping of 4 or 2 tuples of unsigned or signed code word into quantized spectrum coefficients. The stage 2 of regrouping processing is performed using algorithmic approach. The 11 spectral Huffman codebooks, the 11th is a special one. It allows the encoding of quantized spectral coefficients when the largest absolute value (LAV) is greater than 15. If the value is equal to 16, an escape flag is used. There are eleven Huffman codebook for the spectral data and one differential scale factor codebook. Additionally there is a single zero codebook indicating that neither scale factor nor quantized data are transmitted. The quantized values are inversely quantized by the inverse quantized tool. The IQ value is scaled by rescale tool. The IQ samples are rescale by scale factor gain.

ISSN: 2321-8169
1315 - 1319

The M/S joint channel decoding mechanism reconstructs the spectral coefficients or the left and right channel. M/S stereo coding is used to control the imaging of coding noise to remove imaging artifacts. The switching state (M/S coding on or off) is transmitted to the decoder as an array of signaling hits. Note that, when the intensity coding is on, no m/s decoding is performed. In this AAC decoder, intensity stereo decoding and coupling channel mechanism are implemented in the LC profile. The use of intensity stereo decoding is signaled by the pseudo codebook intensity, IICB (in-phase signal) and intensity IICB2 (out-of-phase signal) in the right channel. The directional information for the Intensity decoding is represented by an "intensity stereo Position" value, which indicates the relation between left and right channel scaling. These positions arc coded by the differential coding with two differences. 'The same codebook is used for coding intensity stereo positions and scale factors.

The PNS is a lossy compression technique that is based on the statement that all white noise sounds parallel to the human being. The technique is supported on detection the “white-noise” like frequencies of a given audio signal and coding their power and frequency ranges instead of coding the original data. In fact, what PNS does is that it checks noisy bands in the signal. The TNS tool controls the temporal shape form of the division noise within the frequency domain. It is applied to the whole spectrum or solely an area of the spectrum. Especially it will use many filters on distinct constant regions. Therefore, an all-pole filter is required to select groups of coefficients in the spectral domain. In the filters bank block, the frequency domain signals are changed into time domain and generate the output audio signals. In contrast to the hybrid filter bank of MP3, AAC uses a plain Modified Discrete Cosine Transform (MDCT). The AAC filter bank out performs the filter banks of previous coding methods. The AAC filter bank out performs the filter banks of previous coding strategies. AAC decoder filter bank consists of AN Inverse MDCT(IMDCT) and windowing and overlap add functions. Since the windowing operate encompasses a important result on the filter bank frequency response, the filter bank has been designed to permit the switch in window form to adapt to input-signal conditions. AAC decoder wants the method that extracts frequency parts from a time domain signal. Most of the digital audio decoders use IMDCT the frequency domain signal convert into the time domain signal. The MP2 decoder is then designed to form it an analogous structure to the AAC LC. It consists of the precise blocks of the AAC LC Decoder and different blocks are shared with the MP2 decoder. for every block it perpetually reads information from the audio RAM and writes the results back to a special space of an equivalent RAM, even as the AAC LC decoder.

ARCHITECTURAL DESCRIPTION

MP2 and AAC LC are mutually based on the psychoacoustic model for audio compression via a time to frequency conversion. So still if two decoders have different audio decoding algorithms.

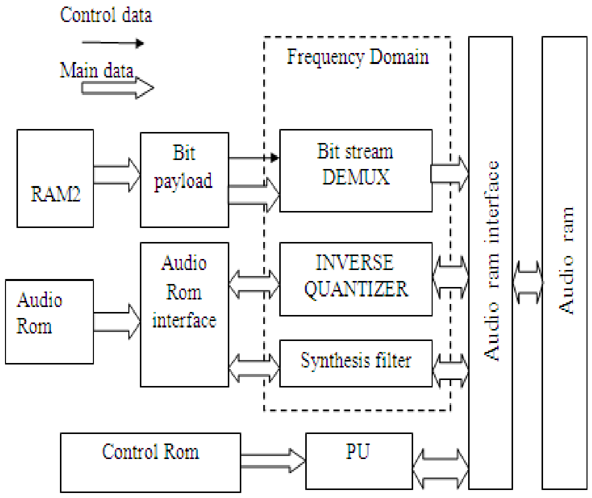

Figure 2: Block Diagram of MP2 Decoder

The Figure 2 and Figure 3 illustrate the general decoding connections of the MP2 and AAC LC decoder. It can be seen that mutually audio decoders have the subsequent blocks are bit stream demultiplexer, inverse quantization and synthesis filter. These blocks take up the majority of the computation for the two audio decoders and guide to low power utilization.

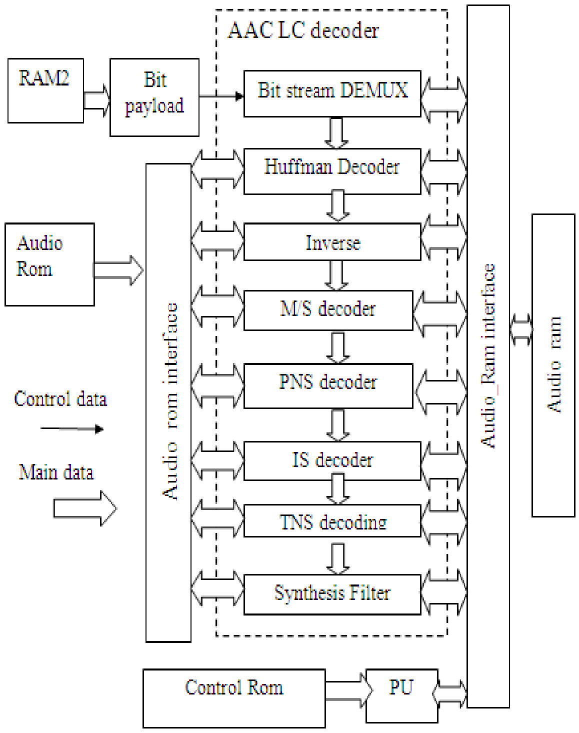

Figure 3: Block Diagram of AAC LC Decoder

The audio frame information generated from the DAB channel decoder and keep in audio RAM. To a small

ISSN: 2321-8169
1315 - 1319

degree load module conducts the task of reading the specified bits from RAM and transferring them to the audio decoder. The bit stream demultiplexer parses the amount spectrum information within the audio frame and stores them into the audio RAM via the audio RAM interface. The inverse division block reads the amount spectra from the audio RAM performs the inverse division computations and writes back the result to the audio RAM. The synthesis filter reads the inverse amount spectra from the audio RAM generates the time domain Pulse Code Modulation (PCM) samples and writes them back to the audio RAM. The PCM samples area unit, then scan out by the Digital Audio Interface (DAI) to drive Associate in Nursing audio digital-analog converter (DAC) to play the sound. The coefficients and constants required for every block area unit keep within the audio read-only storage. The process Unit (PU) may be a terribly compact and totally made-to-order circuit that consists of adders, D-type flip-flops, and multipliers and is intended particularly for MP2 and AAC cryptography.

The PU and every block of the audio decoder exchange information via the audio RAM Interface. The results from every block ar written to the audio RAM, so browse out by Pu for process. when the Pu has finished process, the results ar written back to the audio RAM and wait to be browse out by another block. With this style design, the computation circuits of all the blocks ar shared with only one PU and therefore the chip space are often greatly reduced. As a result of the computations of every block ar fastened, the management logic for the PU is additionally fastened and might be enforced in store to additional scale back the chip space.

SUMMARY OF AAC LC DECODER TOOLS

The summary of the AAC LC decoder tools and it operations are given below in Table 1.

Table-1 summary of AAC LC decoder tools

Tool Name	Required/Optional	Main	LC	SSR
Bitstream Formatter	Required	✓	✓	✓
Noiseless Decoding	Required	✓	✓	✓
Inverse Quantiser	Required	✓	✓	✓
Scalefactors	Required	✓	✓	✓
M/S	Optional	✓	✓/x	✓/x
Prediction	Optional	✓	x	x
Intensity/Coupling	Optional	✓	✓/x	x
TNS	Optional	✓	✓	✓
Filter bank	Required	✓	✓	✓
Gain Control	Optional	x	x	✓

4. RESULT AND DISCUSSION

The AAC LC decoder is developed using MATLAB R2010a software and run the simulation by using verilog code in XILINX ISE 9.2i and MODELSIM 6.3g software.

The video file has been given as an input to the MATLAB code and processed, to produce the encoded digital output. This encoded output is given as input to the verilog code by using Xilinx ISE 9.2i software. It produce a decoded data as a simulation result. The snapshots of the simulation results are shown below.

Figure 4. MP2 decoder Simulation Result

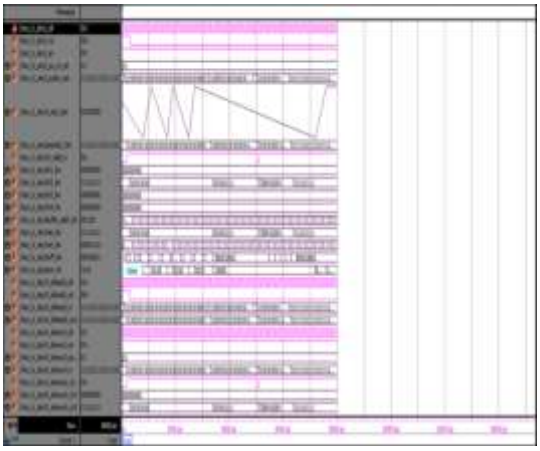

Figure 5. AAC LC decoder Simulation Result

Figure-4 and figure-5 an encoded output from MATLAB is given as input to verilog simulation and set clock, Reset, enable, channel selection, audio ram and get the output as decoded data from modelsim software.

5. CONCULSIONS AND FUTURE WORK

Thus the MP2 and AAC LC decoding system supporting LC profile for DAB and DAB+ audio decoder was designed. The existing system of this project uses HE AAC V2 decoder, that system consists has SBR and PS technologies. This two technologies are used to improve the sound quality in low bit rate program. AAC LC increased spectrum efficiency compared to MP2. A DAB/DAB+ compatible audio decoding solution that integrates MP2 and AAC LC in a chip fabricated in 0.18-μm CMOS technology for both audio decoders. The proposed scheme is uses AAC LC and MP2 decoder it improve the sound quality in high bit rate. The receivers were built, which worked well for both DAB and DAB+ receiving. All DAB+ programs could be decoded correctly using AAC LC. Excellent audio quality was achieved for high bit rate DAB+ programs without any frequency compensation. The output shall be processed by many blocks before hear the audio services. To verify the designed system by using a simulation. The simulation of this project will be carried out by using Modelsim 6.3g and Xilinx ISE 9.2i. Future work of this project is to implementing this designed architecture using FPGA SPARTAN6. And also measure the audio quality of AAC LC decoder by using PSNR.

REFERENCES

[1] Guoyu Wang, Hongsheng Zhang, Mingying Lu, Chao Zhang, Tao Jiang and Guangyu Guo, "Low-cost Low-power ASICN solution for both DAB+ and DAB audio decoding" IEEE Transactions on VLSI systems, pp. 913-921, 2014.

[2] M. Lu, H. Zhang, G. Wang, T. Jiang, and G. Guo,"Optimization and hardware implementation of the AAC inverse quantization algorithm forportable DAB+ receivers," Int. J. Digit. Content Technol. Appl., vol. 6,no. 6, pp. 125–132, Apr. 2012.

[3] T.-H. Tsai and C.-N. Liu, "Low-power system design for MPEG-2/4AAC audio decoder using pure ASIC approach," IEEE Trans. CircuitsSyst. I, Reg. Papers, vol. 56, no. 1, pp. 144–155, Jan. 2009.

[4] R.Amutha, PR.Bhuvaneswari, "Low power FPGA solution for DAB audio decoder by IJRET", International Journal of Research in Engineering and Technology eISSN: 2319-1163 | pISSN: 2321-7308 vol.3 pp.428-431 Apr-2014.

[5] J.C. Hsu, C. Y. Chou, S. W. Chang, and A. Tseng,"A low-cost and low-power single- chip DAB+/DAB/FM receiver,"in Proc. ESSCIRC, Sep. 2010, pp. 470–473.

[6] R.C.Sampaio, P. Azevedo Berge, and R. P. Jacobi, "Hardware and software co-design for the AAC audio decoder," in Proc. 25th SBCCI, 2012, pp. 1–6.

[7] Tsung-HanTsai, Chun-Nan Liu, Hsueh-Yi Lin, Hsing-Chuang Liu and Chia-Ying Wu, "A 1.4 MHz 0.21 mW MPEG-2/4 AAC single chip decoder" IEEE Custom Integrated Circuits Conference (CICC), pp.653–656,2009.

[8] Chao-Tang Yu, Kai-Sheng Yang and Yu-Pin Chang,"DAB Channel Decoder Implementation Using FPGA and Its Testing Platform Buildup", IEEE Transaction on VLSI system, pp. 424-429,2005.

[9] Hsu.J.C, Chou.C.Y, Chang.S.W, and Tseng.A, "A low-cost and low-power single- chip +/DAB/FM receiver," in Proc. ESSCIRC, pp. 470–473,2010.

[10] Tsai.T-H, and Liu.C-N, "Low-power system design for MPEG-2/4AAC audio decoder using pure ASIC approach," IEEE Trans. CircuitsSyst. I, Reg. Papers, Vol. 56, No. 1, pp. 144–155,2009.

[11] Watson.M.A, and Buettner.P, "Design and implementation of AAC decoders," IEEE Trans. Consum. Electron., Vol. 46, No. 3, pp. 819–824,2000.

[12] Bang.K.H, Kim.J.S, Jeong.N.H, Park.Y.C, and Youn.D.H, "Design optimization of MPEG-2 AAC decoder", IEEE Trans. Consum. Electron., Vol. 47, No. 4, pp. 895–903,2001.

[13] Chen.J and Tai.H.M, "MPEG-2 AAC decoder on a fixed-point DSP," IEEE Trans. Consum. Electron., Vol. 45, No. 4, pp. 1200–1205,1999.