

A Feasible Solution for both DAB and DAB+ Audio Decoding

Shebil .P.C¹, S. Selvarasu², S. Saravanan³, T.Kowsalya⁴

PG Scholar, Department of ECE, Muthayammal Engineering College, Rasipuram, Namakkal, Tamilnadu, India

Assistant Professor, Department of ECE, Muthayammal Engineering College, Rasipuram, Namakkal, Tamilnadu, India

Professor, Department of ECE, Muthayammal Engineering College, Rasipuram, Namakkal, Tamilnadu, India

Professor Head, Department of ECE, Muthayammal Engineering College, Rasipuram, Namakkal, Tamilnadu, India

ABSTRACT: From 1990 onwards Digital Audio Broadcasting (DAB) is started and numerous emerging techniques have to be occurred till now. DAB+ is an advanced version of DAB. Both of these techniques coexist in many countries, so receivers are required to be compatible with both standards. A new progress to intend a DAB and DAB+ audio decoder is introduced to build up the quality of audio at the receiver end. Integrate an MPEG-1 Layer II (MP2) decoder and sophisticated Advanced Audio Coding Low Complexity (AAC LC) decoder provides basic audio decoding for DAB and DAB + in FPGA. It also utilizes simple method to improve high frequencies and stereo quality instead of complicated Spectrum Band Replication (SBR) and Parametric Stereo (PS). The audio frames data are produce from DAB and DAB+ channel decoder are store in RAM. The bit stream de multiplexer parses the quantized spectrum data in the audio frame and provisions them in to the acoustic RAM. The converse quantization block reads the quantized spectra from the audio RAM, performs the inverse quantization computation, and writes reverse the consequence to the audio RAM. In order to decode both MP2 and AAC LC Decoder share the channel decoder but cannot share the audio decoder. The proposed decoder consumes low power.

KEYWORDS: HEAAC LC, MPEG Standards, Digital Audio Broadcasting, IMDCT-Inverse Modified Discrete Cosine Transform, Huffman decoding, Parametric Stereo (PS), DAB+

I. INTRODUCTION

The rapid development of digital communication during the past decade has opened numerous opportunities for new multimedia services, some of them operating under severe restrictions regarding the available bit rate for representing audio content. These include terrestrial and Satellite-based digital audio broadcasting and wireless music downloads to cellular phones. Since These applications cannot be served well using regular AAC, an enhanced audio format is desired That can deliver high audio quality and full audio bandwidth even at very low data rates, for Instance at rates of 24 kb/s and below per audio channel. To enable audio and music delivery for very low bit rate applications, a substantial increase of coding efficiency is required compared to the performance offered by regular AAC at such rates.

As a common limitation, previous general audio coders typically have to reduce the transmitted audio bandwidth when operating at low bitrates in order to avoid excessive coding artefacts from being introduced in the transmitted low frequency region. HE-AAC technology was designed to overcome this obstacle by reproducing a wide audio Bandwidth independently of the coding bit rate by using audio bandwidth extension. An enhanced version of the coder (HE-AAC v2) is designed to additionally exploit models of human spatial perception to achieve a further boost in coding efficiency. In both cases, a further objective was to achieve this goal by means of simple extensions to the existing AAC architecture that come at a limited increase in computational complexity

In 1986 DAB consortium formed in Europe and members are France, UK, German, & Netherlands. This formation lead to the development of DAB by Euroca-147 project in 1995 selected MP2 as its audio coding standard when first published. In 1995 first trial broadcast conducted in UK and selected as network broadcast in UK. In 1997 a world DAB forum formed.

International Journal of Innovative Research in Science, Engineering and Technology

(An ISO 3297: 2007 Certified Organization)

Vol. 4, Special Issue 6, May 2015



For many years, DAB has been adopted by many countries because of its higher audio quality and higher spectrum efficiency compared to FM and AM. In order to further improve the spectrum efficiency, the newest audio coding standard, i.e., HE AAC V2, was added to DAB in 2006 as the new audio coding method. DAB systems that use HE AAC V2 to compress audio are called DAB+

A. HE AAC V2

The newest AAC version, HE-AAC V2 (High Efficient Advanced Audio Coding, Version 2), was developed by the International Organization for Standardization/ International Electrotechnical Commission (ISO/IEC) Moving Picture Experts Group (MPEG) by subsequent extension of the established Advanced Audio Coding (AAC) architecture which is composed by the AAC LC (Low Complexity), SBR (Spectrum Band Replication) and PS (Parameter Stereo), is one of the most efficient audio coding methods in the world. The updated version of DAB (Digital Audio Broadcasting), i.e. DAB+, thus adopts HE-AAC as the new audio coding standard to increase the spectrum efficiency and the robustness of the received audio quality.

HE-AAC technology was designed to overcome to avoid excessive coding artifacts from being introduced in the transmitted low-frequency region by reproducing a wide audio bandwidth independent of the coding bit rate by using audio bandwidth extension. The special techniques of SBR and PS are used to improve the sound quality for low bit rate programs

But, AAC LC is the key, not SBR and PS. In other words, if we implement AAC LC, we can decode all DAB+ programs no matter whether these programs use SBR and PS or not. The cost for implementing AAC LC is less than either SBR or PS and AAC LC plays the main role to achieve higher coding efficiency than previous MP2

B. ANALYSIS

DAB and DAB+ programs are often mixed broadcasts from the same transmitter and it can share the same channel decoder but can not share the audio decoder. Many countries have adopted DAB+ as their standard. But some nations like UK, China have no plans to update DAB to DAB+. Thus digital radio receiver sets compatible with both standards need to be developed. These radios must keep the price range, power consumption, volume, and styles of current commercially available DAB products or else they cannot be accepted by current DAB market. These are the big challenges for the implementation of DAB+.

But, it's proved that AAC LC itself can decode all DAB+ programs no matter whether these programs use SBR and PS or not. The cost for implementing AAC LC is less than either SBR or PS and AAC LC plays the main role to achieve higher coding efficiency than previous MP2. So by analysis we come to the point as

- By use of MP2 decoder it is possible to decode traditional DAB signals
- With the help of an AAC LC decoder, DAB+ signal can be successfully decoded.

International Journal of Innovative Research in Science, Engineering and Technology

(An ISO 3297: 2007 Certified Organization)

Vol. 4, Special Issue 6, May 2015

- We can eliminate the decoding complexity of SBR and PS

II. ARCHITECTURE ANALYSIS

A. DAB DECODER

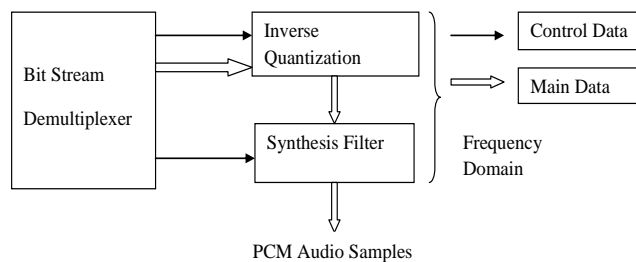


Figure.1. Common MP2 decoding procedure

The DAB decoder mainly consist of IQ and Synthesis filter. In IQ, it reads the quantized spectra from the audio RAM, performs the inverse quantization computations, and writes back the result to the audio RAM. The mathematical representation of IQ as

$$\text{sign}(x_quant) = \begin{cases} 1, & x_quant \geq 0 \\ -1, & x_quant < 0 \end{cases}$$

$$X_{rescal} = \text{sign}(x_quant) * |x_quant|^{\frac{4}{3}} * 2^{0.25(sf-SF_OFFSET)}$$

Synthesis filter read the inverse s

B. DAB+ DECODER

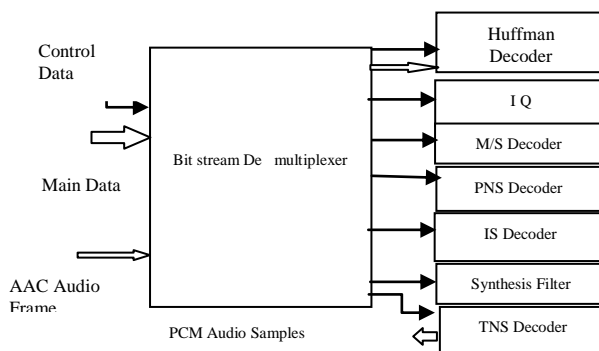


Figure.2. Common AAC LC decoding procedure

The DAB+ decoder mainly consist of Huffman, M/S, PNS, IS, and TNS decoders. In Huffman decoding, there are 12 Huffman codebooks. Eleven codebooks are used for spectrum coding and one for scale-factor coding. There are two stages in spectral Huffman decoding. In stage1 is the Huffman decoding which unpacks the Huffman code index. Stage 2 is the regrouping of 2 or 4-tuples of signed or unsigned code words into quantized spectral coefficients. The M/S decoder recover and add the redundancies removed from the encoder to the frequency domain. In M/S decoding, the left and right channel signals are dematrixed by either the identity matrix or the inverse M/S matrix. In IS, The inverse quantized samples are rescaled by a scale-factor gain. PNS decoder will Removing the noise due to the inverse

International Journal of Innovative Research in Science, Engineering and Technology

(An ISO 3297: 2007 Certified Organization)

Vol. 4, Special Issue 6, May 2015

quantization. The TNS tool controls the temporal shape of the quantization noise in the frequency domain. It can be applied to the entire spectrum or only a part of the spectrum. In particular, it can use several filters on distinct coefficient regions. Synthesis filter will convert the frequency domain signal into time domain signal.

The Fig1 and 2 show that, general decoding procedures of the MP2 and AAC LC decoders, respectively. It can be seen that both audio decoders have the following blocks: bit stream demultiplexer, inverse quantization block, and synthesis filter. These blocks take up most of the computation for the two audio decoders. This means most of the circuitry for the two audio decoders can be shared, to save silicon area and power consumption. So the AAC LC decoder is then designed as shown in Fig.4, to make it a similar structure to the proposed MP2 decoder.

III. PROPOSED SYSTEM

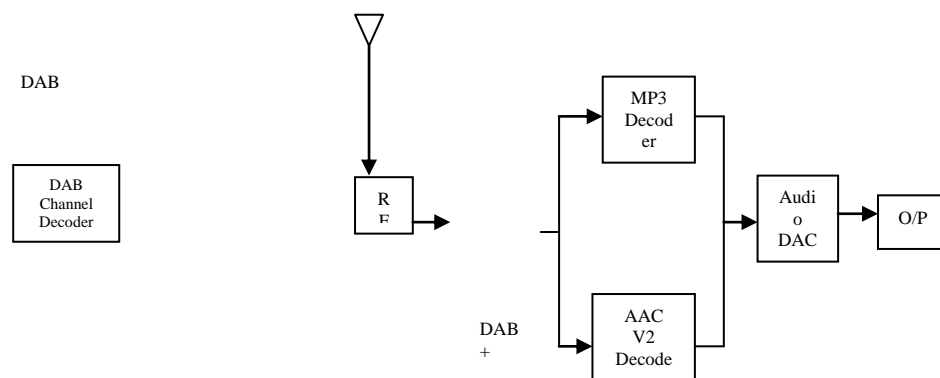


Figure.3. Structure diagram of the DAB/DAB+ compatible receiver

The audio frame data are generated from the DAB channel decoder and stored in RAM2 (Fig. 4). A “bit_payload” module conducts the task of reading the desired bits from RAM2 and transferring them to the MP2 decoder. The bit stream demultiplexer parses the quantized spectrum data in the audio frame and stores them into the “audio_ram” via the “audio_ram_interface.” The inverse quantization block reads the quantized spectra from the “audio_ram,” performs the inverse quantization computations, and writes back the result to the “audio_ram.” The synthesis filter reads the inverse quantized 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 are then read out by the digital audio interface (DAI) to drive an audio digital-to-analog converter (DAC) to play the sound. The coefficients and constants needed for each block are stored in the “audio_rom.”

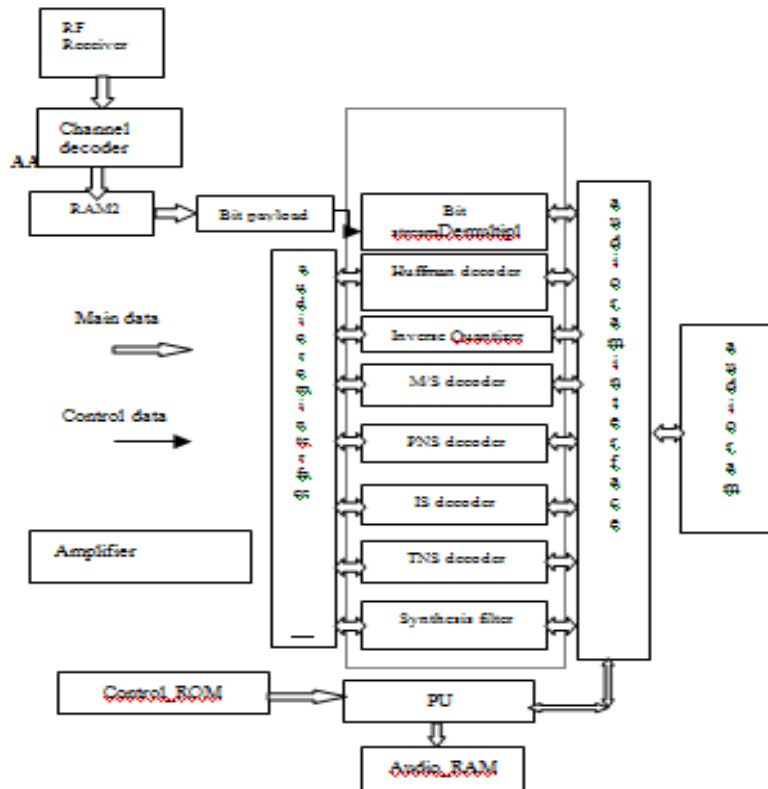


Figure4. Proposed DAB/DAB+ Decoder

The PU is a very compact and fully customized circuit which is composed of adders, D-type flipflops, and multipliers, and is designed specifically for MP2 and AAC decoding. The PU and each block of the audio decoder exchange data via the “audio_ram_interface.” The results from each block are written to the “audio_ram,” and then read out by PU for processing.

The Bit payload reads the data from the RAM. The bit stream DE multiplexer extracts the audio frame signals and decoding information. IQ will read the quantized spectra from the audio RAM, performs the inverse quantization computations, and writes back the result to the audio RAM. Synthesis filter reads the inverse quantized 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 are then read out by the Digital Audio Interface (DAI) to drive an audio Digital-to-Analog Converter (DAC) to play the sound. The coefficients and constants needed for each block are stored in the audio ROM. The M/S and intensity stereo tools recover and add the redundancies removed from the encoder to the frequency domain. The Temporal Noise Shaping (TNS) approach is based on two main considerations, namely

1. Consideration of the time/frequency duality between spectral envelope and (squared) Hilbert envelope
2. Shaping of quantization noise spectra by means of open-loop predictive coding.

The TNS predictive encoding/decoding process over frequency can easily be realized by adding one block to the standard structure of a generic perceptual encoder and decoder. An additional block, “TNS Filtering” is inserted after the analysis filter bank performing an in-place filtering operation on the spectral values. The TNS tool controls the

International Journal of Innovative Research in Science, Engineering and Technology

(An ISO 3297: 2007 Certified Organization)

Vol. 4, Special Issue 6, May 2015

temporal shape of the quantization noise in the frequency domain. It can be applied to the entire spectrum or only a part of the spectrum.

A. Power optimization method:

Two main methods are employed to reduce power consumption. The first method is algorithm design, together with circuit optimizations of each block of the MP2 and AAC decoders, as described above. Reducing the number of computations means the maximum clock frequency of the decoder can be lowered, and, consequently, the power consumption reduces. The second method is the use of gated clocks. Because the MP2 and AAC decoders will never work at the same time, only one decoder will be in the active state and the other one can be completely shut down, to save the power.

Reducing the number of computations means the maximum clock frequency of the decoder can be lowered, and, consequently, the power consumption reduces. The MP2 and AAC decoders will never work at the same time, only one decoder will be in the active state and the other one can be completely shut down, to save the power. Also, for each audio decoder, not all the blocks work at the same time. So when some blocks of the audio decoder are active, other blocks that are not active can be shut down to reduce the power consumption when the inverse quantization block of the AAC decoder is active, the clocks of the temporal noise shaping decoder, perceptual noise substitution decoder, intensity stereo decoder, mid/side stereo decoder, and synthesis filter can all be shut down to save power.

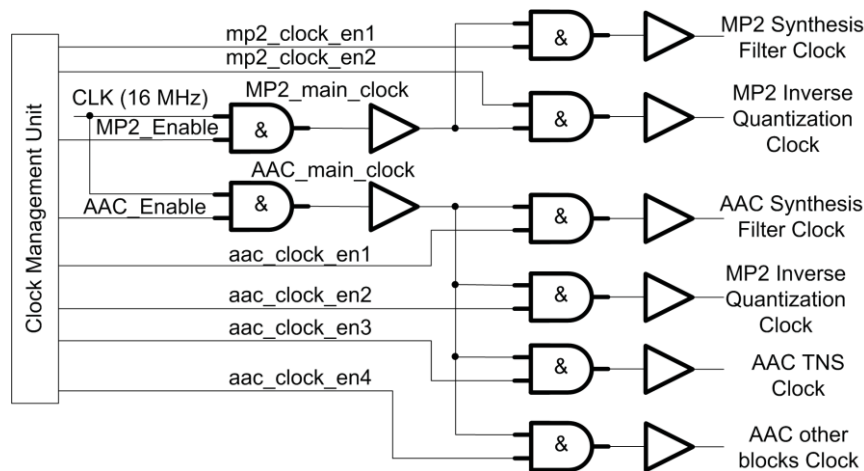


Figure5. Clock network for the proposed audio decoding

B:IMDCT

The modified discrete cosine transform/inverse modified discrete cosine transform (MDCT/IMDCT) algorithm using a matrix representation. THE modified discrete cosine transform/inverse modified discrete cosine transform (MDCT/IMDCT) is used to realizethe analysis/synthesis filter bank of time-domain aliasingcancellation scheme for sub band coding. This transform,because of its efficiency and flexibility in window-sizeswitching, has been adopted in several international standards and commercial products, such as MPEG-1, MPEG-2,and AC-3 for audio coding.

AAC decoder filter bank consists of an Inverse MDCT (IMDCT) and windowing and overlap add functions. Since the windowing function has a significant effect on the filter bank frequency response, the filter bank has been designed to allow the switch in window shape to adapt to input-signal conditions. AAC decoder needs the process that extracts frequency components from a time domain signal. Most of the digital audio decoders use IMDCT to convert the frequency domain signal to the time domain signal.

The IMDCT output samples are obtained by shuffling the outputs of the IFFT in the post-processing step. To store the real and imaginary coefficients of each step we used RAM buffers. All three stages were coded to be able to process either long (1024 samples) or short windows (128 samples) on the same module.

International Journal of Innovative Research in Science, Engineering and Technology

(An ISO 3297: 2007 Certified Organization)

Vol. 4, Special Issue 6, May 2015

$$\bar{x}(n) = \sum_{k=0}^{N/2-1} X(k) \cos \left[\frac{\pi}{2N} \left(2n+1 + \frac{N}{2} \right) (2k+1) \right], \quad n=0,1,\dots,N-1$$

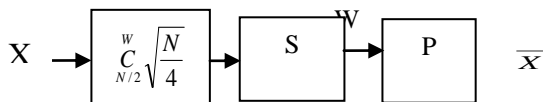


Figure6. IMDCT Block diagram

Advanced Audio Coding (AAC) is an audio coding standard for Lossy digital audio compression. Designed to be the successor of the MP3 format, AAC generally achieves better sound quality than MP3 at similar bit rates. AAC supports inclusion of 48 full-bandwidth (up to 96 kHz) audio channels in one stream plus 16 low frequency effects (LFE, limited to 120 Hz) channels, up to 16 "coupling" or dialog channels, and up to 16 data streams. The quality for stereo is satisfactory to modest requirements at 96 Kbit/s in joint stereo mode; however, hi-fi transparency demands data rates of at least 128 Kbit/s (VBR). The MPEG-2 audio tests showed that AAC meets the requirements referred to as "transparent" for the ITU at 128 Kbit/s for stereo and 320 Kbit/s for 5.1 audio. AAC also continues to enjoy increasing adoption by manufacturers of in-dash car audio systems.

IV. RESULT AND CONCLUSION

A real time MP2 and AAC decoding system supporting LC profile for DAB audio decoder was designed. AAC LC increased spectrum efficiency compared to MP2. SBR and PS have effect on low bit rate programs but the subjective feeling of the effect is not consistent, and the improvement is counteracted by other factors such as complexity and higher power consumption for decoding. MP2 and AAC LC decoder are both based on the psychoacoustic model for audio compression using a time-to-frequency transformation. To verify the designed system by using simulation. The result of simulation showed in both Xilinx ISE tool. Then the power consumption and area occupied by the architecture have been analyzed and at the same time maintaining the high audio quality.

Future work of this project is to implementing this designed architecture using FPGA SPARTAN6E, and also implementing delta sigma demodulation technique to improve the audio quality

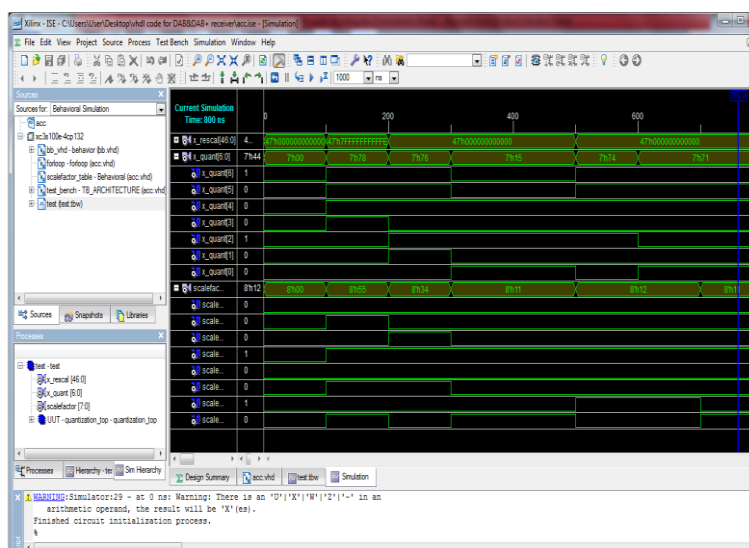


Figure5. Xilinx Simulation Result

International Journal of Innovative Research in Science, Engineering and Technology

(An ISO 3297: 2007 Certified Organization)

Vol. 4, Special Issue 6, May 2015

Every audio file is first converted into wave file, then that wave file has been given as an input to the MATLAB code and processed, to produce the encoded output. This encoded output is given as input to the Verilog code by using Xilinx ISE 12.1 software in ISim. It produces a decoded data as a simulation result. This result can be verified by using MATLAB. The snapshots of the simulation results are shown below.

The power analysis report produce by Xilinx ISim after simulation process is completed. Power consumption of designed architecture is 64mW in FPGA Spartan6.

REFERENCES

- [1] T.Kowsalya and Dr.S.Palaniswami (2014) 'A Clock Control Strategy Based clustering Method For Peak Power And Rms Current Reduction' in Journal of Theoretical and Applied Information Technology Vol. 63 No.2 © 2005 - 2014 JATIT & LLSISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195 459
- [2] T.Kowsalya and Dr.S.Palaniswami(2012) ' Decoupled SRAM Cell with Bit Line Decoupled Current Mode Sense Amplifier' Published in European journal of Scientific Research in volume 84 issue 2 Aug 2012
- [3] Cheng M.-H and Hsu Y.-H., "Fast IMDCT and MDCT algorithms—A matrix approach," IEEE Trans. Signal Process., vol. 51, no. 1, pp. 221–229, Jan. 2003.
- [4] S.Anu and Mrs.T.Kowsalya "Low Power FSK Modulation and Demodulation using VHDL" International Journal of Advances in Engineering Science and Technology Dec 2014".
- [5] FS1235: Advanced Digital Radio and Audio Baseband Processor, Frontier Silicon Co., London, U.K., 2011.
- [6] Herre . J, "Temporal noise shaping, quantization and coding methods in perceptual audio coding: A tutorial introduction," in Proc. AES 17th Int.Conf. High Qual. Audio Coding, 1999, pp. 312–315.
- [7] Herre. U, Dietz. M., "MPEG-4 high-efficiency AAC coding," IEEE Signal Process. Mag., vol. 25, no. 3, pp. 137–142, May 2008.
- [8] Hsu J. C, Chou C. Y, Chang S. W, and Tseng. A., "A 159.2mW SoC Implementation of T-DMB Receiver is including Stacked Memories," in Proc. ESSCIRC, Sep. 2010, pp. 470–473.
- [9] Information Technology—Coding of Audio-Visual Objects—Part 3: Audio, ISO/IEC Standard 14496-3, Dec. 2005.
- [10] Joohyun. L, Sungdo. K, and Jinkyu. K., "A 159.2 mWSoC implementation of T-DMB receiver including stacked memories," in Proc. IEEE Custom Integr. Circuits Conf., Sep. 2008, pp. 679–682.
- [11] Lee J.S, Jeong J.H, and Chang T.G., "An efficient method of Huffman decoding for MPEG-2 AAC and its performance analysis," IEEE Trans. Speech Audio Process., vol. 13, no. 6, pp. 1206–1209, Nov. 2005.
- [12] Liu. P, Liu. L, Deng. N, Fu. X, Liu. J, Liu. Q, Zhang. G, and He. B, "VLSI implementation for portable application oriented MPEG-4 audio codec," in Proc. IEEE Int. Symp. Circuits Syst., May 2007, pp. 777–780.
- [13] Meltzer. S and Moser. G., "HE-AAC v2—Audio coding for today's digital media world," EBU, Geneva, Switzerland, Tech Rev. 305, Jan. 2006.
- [14] Radio Broadcasting System; Digital Audio Broadcasting (DAB) to Mobile, Portable and Fixed Receivers, ETSI Standard EN 300 401, Jan. 2006.
- [15] Renner. A. and Susin. A. A., "An MPEG-4 AAC decoder FPGA implementation for the Brazilian digital television," in Proc. 8th Southern Conf. Program. Logic, 2012, pp. 1–6.
- [16] Sampaio. R. C, Azevedo Berge. P., and Jacobi. R. P., "Hardware and software co-design for the AAC audio decoder," in Proc. 25th SBCCI, 2012, pp. 1–6.
- [17] Seong-Do. K. and Seung-Hyeub. O., "A CMOS single chip receiver with RF front-end and baseband processor for band-III T-DMB/DAB applications," in Proc. 11th Int. Conf. Adv. Commun. Technol., 2009, pp. 1238–1241.
- [18] Tsai. T. H, Liu. C. N, Lin. H. Y, Liu. H. C, and Wu. C. Y., "A 1.4 MHz 0.21 mW MPEG-2/4 AAC single chip decoder," in Proc. IEEE Custom Integr. Circuits Conf., Sep. 2009, pp. 653 656.
- [19] Lu. M., Zhang. H, Wang. G, Jiang. T, and Guo. G., "Optimization and hardware implementation of the AAC inverse quantization algorithm for portable DAB+ receivers," Int. J. Digit. Content Technol. Appl., vol. 6, no. 6, pp. 125–132, Apr. 2012.
- [20] Zhang. H, Lu. M., and Wang. G., "An ASIC implementation of MPEG audio decoders," in Proc. 7th Int. Conf. ASIC, Oct. 2007, pp. 754–757.
- [21] Zhou. D, Liu. P, Kong. J, Zhang. Y, He. B, and Deng. N., "An SoC based HW/SW co-design architecture for multi-standard audio decoding," in Proc. IEEE Asian Solid-State Circuits Conf., Nov. 2007, pp. 200–203.