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Advanced Audio Compression For Lossless Audio Coding Using Ieee 1857.2

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Abstract: ***Developed by IEEE-SA (Standard Association) and released in August 2013, IEEE 1857.2 is the latest standard for lossless audio compression. This recent standard for advanced audio coding (AAC) is an efficient lossless audio codec (coding decoding) technique, in improving audio quality for compression and decompression, optimizing bandwidth during transmission, saving storage space, it also speeds up the video and audio streaming, and others video and audio data. It defines a set of tools to support specific audio coding functions, including general audio coding and lossless coding. This project presents the algorithms for lossless audio coding communication, the higher efficiency of this standard and other assets such as compression ratio, encoding and decoding speed of this standard. Different types of audio files have compressed and the results show a good compression ratio (CR) which depends on audio file type, and good encoding and decoding speed.***

Index Terms: codec, Advanced audio coding, IEEE 1857.2, audio compression, lossless coding, LPC.

1. Introduction

During two last decades, the optimization of communication resource and of data recording medium has become an attractive area for many researchers. As well as for the preservation of the quality or originality of data during coding and decoding (codec). IEEE-SA (Standard Association) is a professional and renowned association, which is working in multiple domains of engineering (electrical, electronic, telecommunication, computer engineering, and so on), for providing solutions in such domains. IEEE-SA is a relevant part which, has contributed solution to the compression and decompression of audio and speech signal by, developing a new standard for advanced audio coding (AAC) in August 2013, named IEEE 1857.2. This new standard has been developed for lossless audio codec, which enables the compression and decompression of digital audio and speech data, without any loss in quality due to a perfect reconstruction of the original signal.

Compression is the mechanism or method of compacting 'raw' or uncompressed data into a smaller number of bits. Audio compression (Audio coding) is the act of converting digital audio into a format suitable for transmission or storage requiring least bandwidth consumption, whilst typically reducing the number of bits. Uncompressed digital audio coming directly from microphone typically requires a large bitrate. Therefore, if the decoded video sequence is identical to the original, then the coding process is lossless, such as FLAC, monkey's Audio, WMA, MPEG-4 SLS [5], MPEG-4 ALS, WavPack etc. If the decoded sequence differs from the original, the process is lossy, such as MPEG-1 layer-3 (MP3), AAC, WMA lossy, Musepack etc. For speech coding is a critical technology for videoconferencing systems, digital cellular communications, and voice over Internet protocol (VoIP) applications, while audio coding is essential for portable audio players, audio streaming, and the storage, lower bandwidth consumption, higher resolution and sampling rates, and playback of movies.

In lossless audio coding, the encoding is generally organized by, linear prediction coding (LPC) block followed by pre-processing, and the entropy coding block. The first block of this part avoids redundancy coming from input or 'raw 'data, and predicts the next sample from the previous samples. While the last block known as entropy encoder, encodes the residue received from LPC before storage or, sending the bit-stream through channel. At the decoder side the inverse of the phenomenon is performed.

2. Lossless Audio Codec Using IEEE 1857.2

The new IEEE standard of audio coding named IEEE 1857.2 is the latest technique or algorithm for audio coding [2]. The general block diagram of IEEE 1857.2 lossless audio compression system is shown in figure 1 [1], in which top part denotes the encoder, and the bottom part the decoder.

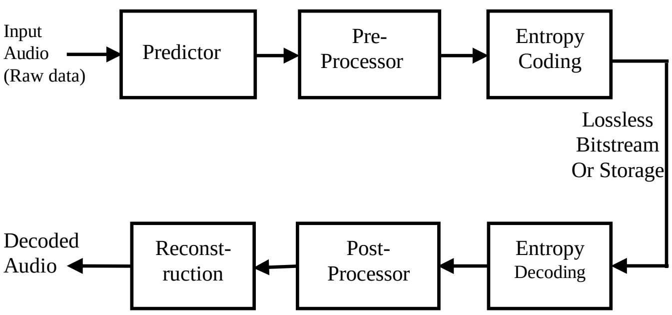

Figure 1: General lossless audio coding: encoder (top) and decoder (bottom) [1].

2.1. Predictor

Predictor is a tool used mostly in audio signal processing and speech processing for representing the spectral envelope of a digital signal of speech in compressed form, using the information of a linear predictive model [2]. It is one of the most powerful speech analysis techniques, and one of the most useful methods for encoding good quality speech at a low bit rate and provides extremely accurate estimates of speech parameters. In this part the coefficients of audio are finding, named PARCOR coefficient, they are computed using the Levinson-Durbin algorithm [2] [6], given by the equations (1) and (2).

$$b_k = \begin{cases} \left\lfloor 64\left(\ln\left(\frac{2}{3}+\frac{5}{2}\sqrt{\frac{1+p_1}{2}}\right)/\ln\left(\frac{3}{2}\right)\right)\right\rfloor & k=1 \\ \left\lfloor 64\left(\ln\left(\frac{2}{3}+\frac{5}{2}\sqrt{\frac{1-p_2}{2}}\right)/\ln\left(\frac{3}{2}\right)\right)\right\rfloor & k=2 \\ \lfloor 64 p_k \rfloor & k=3,\ldots\ldots,lpc\_order \end{cases} \quad (1)$$

Where b_k denotes the resulting quantized values and P_k is the PARCOR coefficients (P_k, $k = 1,\ldots.,lpc\_order$).

$$par\lfloor k \rfloor = \begin{cases} \Gamma(b_k) & k=1 \\ -\Gamma(b_k) & k=2 \\ b_k 2^{14} + 2^{13} & k=3,\ldots,lpc\_order \end{cases} \quad (2)$$

The next samples are predicted using Levinson-Durbin algorithm, and the plot is given by the figure 2.

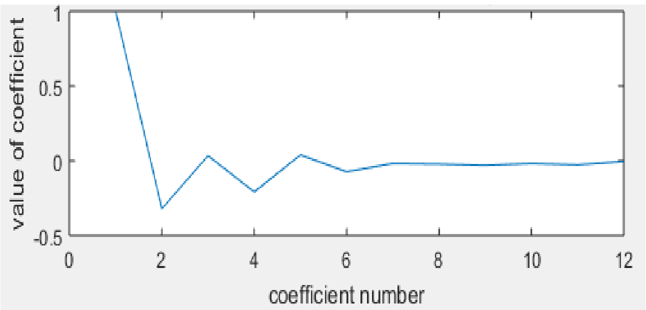

Fig 2: plot of Levinson Durbin in LPC encoder

2.2. Pre-processing

The method of transient decision in this standard is based on a two-level decision on the temporal stability. It uses the energy variation of the signal in the temporal domain and the unpredictability in the frequency domain to decide whether the signal is transient or stationary [1]. The result of this decision indicates whether the input frames need to use the multi-resolution analysis [2].

2.3. Entropy coding

Basically, the entropy coding is an algorithm used in lossless data coding, to compress digital data by representing frequently occurring patterns with few bits, and rarely occurring patterns with many bits [1]. Huffman [5] and arithmetic [7] coding are examples of entropy coding.

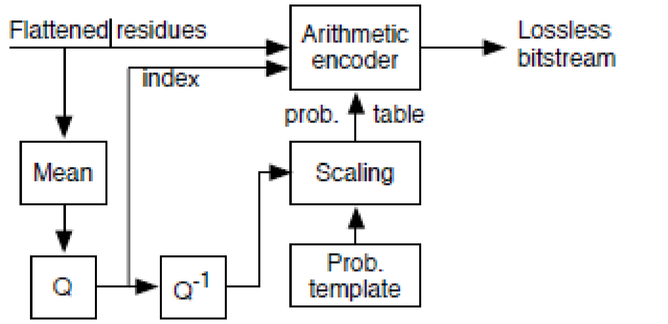

Fig 3: entropy coding block diagram [2].

2.4. Entropy decoding

Similar to entropy coding, entropy decoding also can be considered as 2-step process. The first step converts the input bit stream into the intermediate symbols. The second step converts the intermediate symbols into the quantized MDCT coefficients. In fact, the output of the second step is the DC difference, the output of DPCM, and the AC coefficients after zig-zag scan. The arithmetic coding is implemented is this block of decoding part, [8].

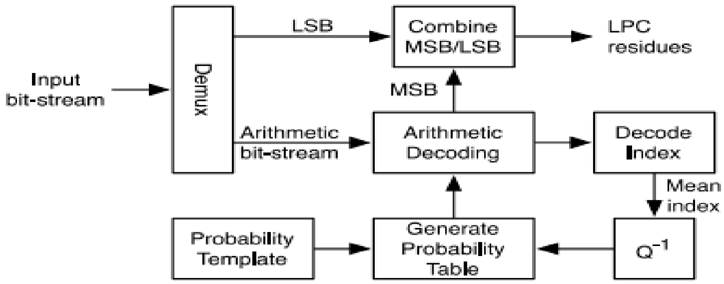

Fig4. Block diagram of the arithmetic decoder [2].

2.5. Post processing

This part of audio decoding is very significant. Indeed, during the transmission or storage procedures, some noises may be introduced due to the imperfections of channel and storage devices equipment [2]. The post-processing block can be useful by removing noise applying the post-filtering process.

2.6. Reconstruction

This part of the decoder consists of reconstructing the original received in the bit-stream form. Since at encoder side the signal was analyzed using decimator, at decoder side the audio signal will be synthesized using interpolator. [7]

Since at the encoder side the audio signal has been analyzed, at the decoder side the same audio signal or file will be synthetized. And the analysis and synthesis or reconstruction are denoting by the figures (5) et (6)

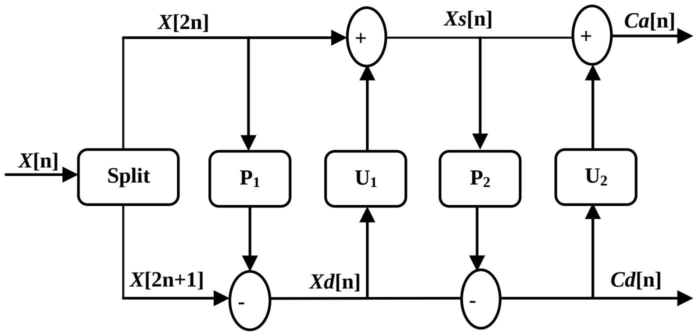

Fig 5: Lifting wavelet transform for synthesis

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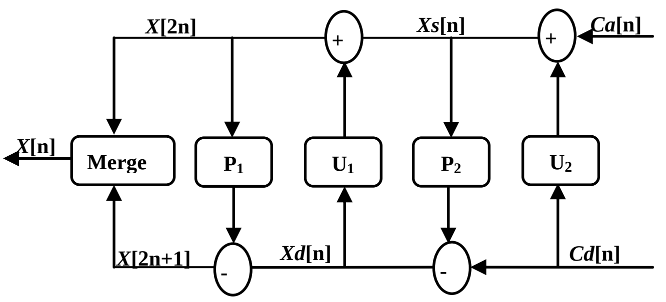

Fig 6: Lossless audio data reconstruction [2]
P denotes Prediction and U update.

3. Results

This section shows the result of our experiment, we coded three different audio files (music audio file and speech audio file) from uncompressed format to compressed format. The uncompressed file can be obtained by recording or by taking an existed file and start the coding process. The experiment has been performed on laptop Intel i5, 1.8 GHz AND 6 GB, 1333 MHz RAM. The waveform the input or raw audio data in represented by the figure 7.

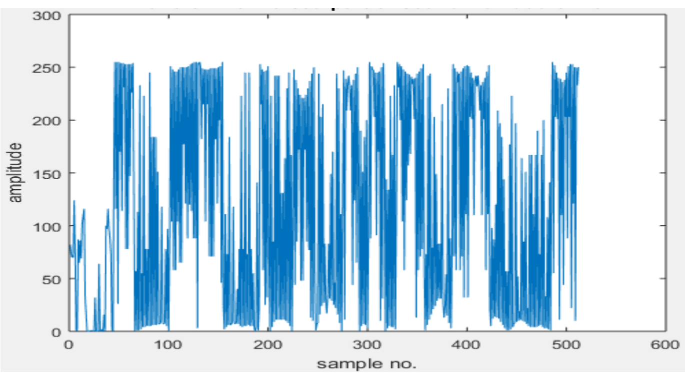

Figure 7: Plot of the waveform of the input audio file.

The compression ratio is computed by dividing the compressed file by original file size, and given by the equation (3)

$$CR = \frac{\text{Compressed file size}}{\text{Original file size}} \times 100\% \qquad (3)$$

The features of the experiments (music and speech audio) files, such as compression ratio, encoding speed (ES), decoding speed (DS) are gathered in the table 1.

Audio format	Music audio file				Speech audio file			
	C S (Kb)	CR (%)	ES (s)	D.S (s)	C S (Kb)	CR (%)	ES (s)	D.S (s)
wav	Uncompressed (6794)	-	-	-	Uncompressed (6891)	-	-	-
IEEE 1857.2	671	10	3	4	646	9.4	3	6
mp4	657	9.7	3	5	646	9.4	5	5
ogg	651	9.6	8	7	832	12.0	9	4
flac	4709	69.3	2	3	3340	48.5	2	4

NB: CS=compressed size, CR= compression ratio, ES= Encoding speed, DS= decoding speed

4. Conclusions

This paper presents the lossless audio compression tools recently approved by IEEE Standard association (SA) called new IEEE standard for Advanced Audio Coding (IEEE 1857.2). The experiment has been performed on two varieties of audio signal such as, music and speech audio file. The performance evaluation shows that, more the compression ratio is less, high is its encoding speed more we save memory space. The encoding and decoding speed prove the efficiency of this recent algorithm in lossless audio coding.

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