

Audio Compression Based on Discrete Cosine Transform, Run Length and High Order Shift Encoding

Zainab T. DRWEESH, Loay E.GEORGE

Abstract— Audio compression addresses the problem of reducing the amount of data required to represent digital audio. It is used for reducing the redundancy by avoiding the unnecessary duplicate data. In the present work a low complexity and efficient coding scheme based on discrete cosine transform (DCT) is proposed. The proposed system consists of audio normalization, followed by DCT transform, scalar quantization, improved run length encoding and a new high order shift coding. To reduce the effect of quantization noise, which is notable at the low energetic audio segments, a post processing filtering stage is introduced as the final stage of decoding process. The system performance is tested using different audio test samples; the test samples have different size and different in audio signal characteristics. The compression performance is evaluated using peak signal to noise (PSNR) ratio and compression ratio (CR). The test results indicated that the compression performance of the system is promising. The compression ratio is increased with the increase of block size. Also the post processing stage improved the fidelity level of reconstructed audio signal.

Index Terms—Audio compression, Discrete Cosine Transform, Scalar Quantization, Lossless Coding.

I. INTRODUCTION

The growth of the computer industry has invariably led to the demand for quality audio data. Compared to most digital data types, the data rates associated with uncompressed digital audio are substantial. For example, if we want send high-quality uncompressed audio data over a modem, it would take each second's worth of audio about 30 seconds to transmit. This means that the data would be gradually received, stored away and the resulting file played at the correct rate to hear the sound. However, if real-time audio is to be sent over a modem link, data compression is needed. Compression can be considered as a key that reduces the amount of data used to convey the same information [1, 2]. The most popular audio coders are based on using two techniques (i.e., sub-band coding and transform coding). Sub-band coding splits signal into a number of sub-bands, using band-pass filter [3]. Transform coding uses a mathematical transformation like FFT and DCT. Jacaba discussed the application of the modified discrete cosine transform (MDCT) to audio compression, specifically the MP3 standard. MDCT plays a very important role in perceptual audio coding [4]. Wang and Vilermo compared the use of Modified Discrete Cosine Transform (MDCT) in audio coding and error concealment with the perspective Fourier frequency analysis [5]. Alvarado and Garcia proposed an

efficient joint implementation of DCT, as a method to obtain a sparse audio signal representation, and the application of the compressive sampling algorithm to this sparse signal [6]. Harmanpreet Kaur and Ramanpreet Kaur proposed a speech compression method using different transform techniques. The signal is compressed by DWT technique afterward this compressed signal is again compressed by DCT and then this compressed signal is decompressed using DWT technique. Harmanpreet, K., Ramanpreet have investigated the use of DWT & DCT as analysis tools for speech signal coding, they used Peak Signal to Noise Ratio and Normalized Root Mean Square Error (NRMSE) to evaluate the effectiveness of different filters of wavelet family [7]. Patil and et al proposed a simple discrete wavelet transform & DCT based audio compression scheme. It was implemented using MATLAB, the experimental results indicated that in general there is improvement in compression factor and signal to noise ratio with DWT based technique [8]. In this paper we introduce a simple DCT transform based coding system to decompose audio signal. DCT is adopted because of its nice de-correlation and energy compaction properties. The resulting transform coefficients are quantized using non uniform scalar quantization. Then, the quantized values are represented using run length encoding (RLE) to prune the long 0's runs, followed by an improved shift coding algorithm.

II. AUDIO COMPRESSION SYSTEM

The most common characteristic of audio signals is the existence of redundant information lay between the adjacent samples. Compression tries to remove this redundancy and make the data de-correlated. Typical audio compression system contains three basic modules to accomplish audio compression. First, an appropriate transform is applied. Second, the produced transform coefficients are quantized to reduce the redundant information; here, the quantized data hold errors but should be insignificant. Third, the quantized values are coded using packed codes; this encoding stage changes the format of quantized coefficients values using one of the suitable variable length coding technique. Figure (1) shows the layout of the proposed system, the modules of our proposed system are described in the following sections.

A. Discrete Cosine Transform (DCT)

This transform had been originated by [Ahmed et al. 74]. Since that time it was studied extensively and commonly used in many applications [9]. At present, DCT is widely used transforms in image and video compression algorithms. Its

popularity is due mainly to the fact that it achieves a good data compaction; because it concentrates the information content in a relatively few transform coefficients [10]. Its basic operation is to take the input audio data and transforms it from one type of representation to another, in our case the signal is a block of audio samples. The concept of this transformation is to transform a set of points from the spatial domain into an identical representation in frequency domain. It identifies pieces of information that can be effectively thrown away without seriously reducing the audio's quality.

All other transform coefficients are called the AC coefficients [6].

B. Quantization

Quantization is an important stage for data compression, it makes approximate mapping of transform coefficient values to integer values have finite length of binary representation [11]. Any quantization method should be based on a principle that determines what data items to quantize and by how much [9]. In transform based coding scheme, the lossy encoder must quantize each frequency component. If the encoder allocates more bits to each frequency, less error (noise) is introduced, but more space is required to store the result. Conversely, fewer bits allocated to each frequency results in more noise, but less space is required to store the result [12].

C. Entropy Encoding

Entropy encoding is used to further compresses the quantized values losslessly to give better overall compression. Different encoding methods can be used (e.g. Huffman, Run Length, Arithmetic, LZW, and Shift Coding). Statistical based encoding methods are used to remove data that are repetitively occurring. Some encoding methods can, also, reduce the number of coefficients by removing the redundant data. In our proposed system the run length encoding method is applied first to prune the long runs of quantized coefficients, then an improved progressive shift coding method is used to firstly prune the existing second order statistical redundancy and, lastly, to encode the produced coefficients individually using variable encoding that based on shift-key mechanism.

D. Run Length Encoding (RLE)

Run length encoding is the simplest form of redundancy removal. It removes redundancy by relying on the fact that any string of symbol may contain repeated sequences or "runs" of the same symbol. The runs of the same symbol are encoded using two entities: (i) a count suggesting the number of repeated symbols and (ii) the symbol itself [2].

E. Shift Coding

Shift coding method is used to reduce the size (in-bits). In this method a list of code words is generated to encode the sequence of numbers. The lengths (in terms of bits) of code words representing the high redundant symbols are less than the length required to represent the maximum symbol value when it is coded using fixed length coding. The size (in bits) required to encode this sequence when using fixed length coding can be calculated as follows:

$$Size = n \lceil \log_2(Max) \rceil \quad (4)$$

Where, Max is the highest symbol value found in the input sequence, n is the number of elements, and $\lceil \cdot \rceil$ is a function returns lowest integer value that higher or equal to the input argument. The performance of shift coding is better when the sequence of numbers has a histogram whose shape is highly peaked. The main advantage of shift coding is its required

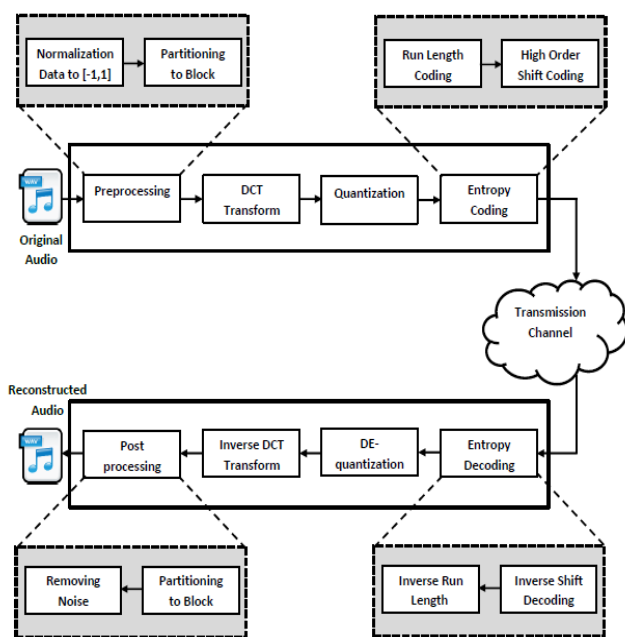


Fig. 1: The Proposed System Layout

The mathematical representation for one-dimensional DCT is:

(1) The forward 1D DCT equation is [9]:

$$C(u) = \alpha(u) \sum_{t=0}^{N-1} \cos \left[\frac{\pi u}{2N} (2t + 1) \right] f(t) \quad (1)$$

Where, $f(0...N-1)$ is the discrete sequence of signal f ; N is the number of $f()$ elements, $C(0...N-1)$ is the cosine transform coefficients, and

$$\alpha(u) = \begin{cases} \sqrt{\frac{1}{N}} & \text{for } u = 0 \\ \sqrt{\frac{2}{N}} & \text{otherwise} \end{cases} \quad (2)$$

(2) The inverse 1D DCT equation is:

$$f(t) = \sum_{u=0}^{N-1} \alpha(u) \cos \left[\frac{\pi u}{2N} (2t + 1) \right] C(u) \quad (3)$$

The first transform coefficient $C(0)$ is an indicator for the average value of the sample sequence; in the literature, this coefficient is called the DC coefficient of the signal.

overhead information is short. The disadvantages of the traditional shift coding method are: (i) it deals with each symbol value individually (i.e., use a code word to represent each symbol), (ii) it spends more bits if the sequence histogram has long tails, this leads the encoder became less efficient. The consequences of these disadvantages appearing clearly when we using noisy wave file with 16 sample resolution. In order to handle the above encoder weakness, we have introduced additional steps to shift coding algorithm, such that the efficiency of this new encoder variant is more stable and robust. The introduced steps are the following:

1. Mapping symbol values to be always positive.
2. Making the dynamic range of input symbols values bounded to be always lower than certain value.
3. Detect the most redundant pairs of symbols and represent each pair by a single symbol.
4. Apply a shift encoding optimizer to find the best lengths of the short and long code words.

After applying the above sequence of coding steps the traditional shift coding is applied on the produced sequence of symbols. Due to the above added steps the required overhead information is increased, but this increase is minor when compared with the length of the input sequence of generated symbols due to encoding audio data.

III. PROPOSED SYSTEM WORKFLOW

The introduced system consists of two major units; the first is the *Encoding unit* and the second one is the *Decoding unit*.

A. Encoding Unit

This unit involves the following stages:

1. Loading the waveform audio file (with WAVE) format. This process includes reading the header data to get the basic file and signal specification information (i.e., number of samples, number of channels, sampling rate, and sampling resolution). Then the audio signal data is loaded as an array of unsigned bytes when the sample resolution is (8 bit/sample) and as an array of signed integers if the sample resolution is 16 bit/sample.
2. Normalizing the loaded audio data values to the range [-1,1] using the following equations:
 - a. For 8 bit sampling resolution:

$$W_n(i) = \frac{W(i) - 128}{128} \quad (5a)$$

- b. For 16 bit sampling resolution:

$$W_n(i) = \frac{W(i)}{32768} \quad (5b)$$

Where, $W(i)$ is the i^{th} element value of the loaded audio data.

3. Partition the array of data into a number of non-overlapping blocks with same size, except the last partition which may be of smaller size. Signal partitioning into blocks is necessary because the characteristics of the sound changes

with the time; if DCT is applied once on the whole signal samples and doing the rest steps of the encoding unit then the encoding process outcome become less packed (i.e., less compression gain is attained). The number of partitions is computed as follows:

$$N_{blocks} = \frac{WavSize}{S_{block}} \quad (6)$$

$$LastBlockSize = WavSize \bmod B_{size} \quad (7)$$

Where, N_{blocks} is the number of the partitions generated from original wave, $WavSize$ is the number of samples of the loaded audio data, S_{block} is the predefined size for each partition (block), and $LastBlockSize$ is the size of the last partition of the original wave.

4. Apply DCT (equation 1) on each audio block to produce the set of transform coefficients $C()$.
5. Apply progressive scalar quantization to quantize the DCT coefficients of each block. This step will reduce the number of bits needed to represent approximately the transform coefficients, and to prepare them to the entropy encoding step. The applied equations of quantization process are the following:
 - a. For DC Coefficient:

$$C_q(0) = \text{round} \left(\frac{C(0)}{Q_0} \right) \quad (8)$$

Where, $C(0)$ is the DC-coefficient, $C_q(0)$ is the quantized DC coefficient.

- b. For DC Coefficient:

$$C_q(u) = \text{round} \left(\frac{C(u)}{Q_1(1 + \beta(u - 1))} \right) \quad (9)$$

Where, Q_1 is the quantization step value that used only to quantize the second DCT coefficient (i.e., 1st AC Coefficient) of each block. The coefficient index u takes the values ($u=1, 2 \dots S_{block}$). The parameter β is the progressive rate of increase of the quantization step for the AC-coefficients.

6. In our proposed system, the run length encoding (RLE) step is used because of the redundant occurrence of many sequences consist of long runs of zeros. Shift coding could not efficiently handle these long runs because it prunes only the short runs by pairing step. RLE will represent the long repeated sequences of symbols as records each consist of (Run Value, Length).
7. Shift coding is applied after RLE, the stream coefficients are shift coded, usually the incoming stream, after the previous steps (i.e., DCT, quantization, and RLE), has highly peaked histogram which makes shift coding a good entropy coding based choice for further compression. The introduced enhanced shift coding consists of the following steps:
 - a. Mapping to positive: Convert all samples of the stream to positive numbers; this step is useful to simply the next coding steps. The following mapping equation had been used to convert the signed samples into positive samples:

$$C'(i) = \begin{cases} 2C(i) & \text{if } C \geq 0 \\ 2|C(i)| + 1 & \text{if } C < 0 \end{cases} \quad (10)$$

Where, C(i) the i^{th} element value of the quantized signed samples value.

- b. Determine the optimal range value whose length is less than the highest value found in input sequence. The following equations have been applied to calculate the optimal range value:

$$P' = MAD * Alpha + Mean \quad (11)$$

$$P = \begin{cases} \sqrt{Max} & \text{if } P'^2 > Max \\ P' + 1 & \text{if } P' \leq Max \end{cases} \quad (12)$$

$$Mean = \frac{1}{LenM} \sum_{i=0}^{LenM-1} S(i) \quad (13)$$

$$MAD = \frac{1}{LenM} \sum_{i=0}^{LenM-1} |S(i) - Mean| \quad (14)$$

Where, P is the optimal range value, Max is the highest sequence elements value, Alpha is a user defined value (in this work its value set 2), Mean is the mean value of the sequence elements, MAD is the corresponding Mean Absolute Deviation, S() is the sequence array, LenM is number of the coefficients.

- c. Do range clipping by scanning the sequence elements, during this scanning any element has a value larger or equal to P it is coded using a set consist of three numbers (P, S₁, S₂), their values are determine by using the following equations:

$$S_1 = C'(i) \text{ div } P \quad (15)$$

$$S_2 = C'(i) \text{ mod } P \quad (16)$$

- d. Before starting the pairing process, set the value of Pp=P, this parameter is used to count the dynamic range after applying the Pairing stage.
- e. Do pairing step by finding the most redundant pair and replace it by single value (Pp+1); this step should repeated for a number of times (use the stopping condition: if the reduction ratio of sequence length after the pairing stage is less than 0.01 then stop the pairing process; otherwise repeat the process). The pair replacement step is as follows: Let (OptL1, OptL2) are the elements of the highest redundant pair, then replace the pairs (OptL1, OptL2) found in the sequence by a symbol value (P+1). After each pairing round increment Pp by 1.
- f. Establish the histogram of the produced sequence after pairing stage, then establish the necessary index table for inserting the histogram values of the new symbols elements (whose values lay within the range [P, Pp]) added in the pairing stage within the histogram sequence whose index is [0, P]. Use this index table to change the elements values lay within [P+1, Pp] by index values, and all values of the histogram elements [0, P] are shifted up properly to host newly indexed symbols.

- g. Apply shift coding optimizer to find the optimal two code word sizes needed to represent the small and large sequence elements values. The applied optimization criteria is:

$$T(\text{in Bits}) = b_1 \sum_{i=0}^{Pp} His(i) + b_2 \sum_{i=R}^{Pp} His(i) \quad (17)$$

The optimizer should find the best values for b_1 & b_2 leads to lowest possible value of T. The values of b_1 & b_2 represent the length of short code words and long code words, respectively. The range of b_1 and b_2 lay within the range $[1, \lceil \log_2(Pp) \rceil - 1]$. The array His() represent the histogram array. The symbol R is equal to $2^{b_1} - 2$, which represent the highest value of the short codeword.

- h. Do the traditional shift coding operation, and then save the output code words into the binary output file.

B. Decoding Unit

The decoding unit consists of the inverse operations to those applied in the encoding process; also these operations are applied in reverse order. The operations are: (i) shift decoding, (ii) long runs expansion, (iii) de-quantization, (iv) inverse DCT, (v) mapping to the byte range [0,255] in case of 8-bit sample resolution, and to the integer range [-32768, 32767] in case of 16-bit sample resolution. In our proposed system a post processing stage is added as last stage of the decoding module. This stage is a selective based low pass filter used to reduce the subjective effect of the produced noise whose effect is sensible at the sound segments have low power. Also, the conducted tests indicated that the error level due to lossy compression is reduced when the proposed post-processing is applied.

The proposed low pass filter could describe mathematically as follows:

$$Wav_{new}(i) = \sum_{j=-2}^2 c(j) Wav(i+j) \quad (18)$$

Where,

$$c(0) = \exp(-1.2 \times (1 - Wav(i))) \quad (19a)$$

Or

$$c(0) = \exp(-1.2 \times (1 - Wav(i))^2) \quad (19b)$$

$$c(1) = c(-1) = 2B, c(2) = c(-2) = B \quad (20)$$

Where

$$B = (1 - c(0))/6 \quad (21)$$

IV. TEST RESULT

This section is dedicated to present the results of the tests conducted to investigate the effects of some involved coding parameters on the performance of the proposed DCT based audio coding schema. The performance of any lossy audio compression method is defined in terms of compression gain (either compression ratio, CR, or bitrate, BR) and fidelity criteria (either root mean square error or PSNR) [9]:

$$CR = \frac{\text{Original audio file size}}{\text{Compressed audio file size}} \quad (22)$$

$$BR = \frac{\text{Compressed audio}}{\text{Original audio}} \quad (23)$$

$$MSE = \frac{1}{n} \sum_{i=0}^{n-1} (x(i) - y(i))^2 \quad (24)$$

$$PSNR = 10 \log_{10} \left[\frac{R^2}{MSE} \right] \quad (25)$$

Where, $x(i)$ is the i^{th} element value of the original audio data, $y(i)$ is the value of the reconstructed audio file, n is the number of the audio samples, and R is the dynamic range of audio samples values (i.e., for 8-bits sample resolution $R=255$, and for 16-bits sample resolution $R=65535$). In our work, a set of tests was conducted to evaluate the performance of the proposed system. The importance of our work lies in the possibility of making good reduction in audio data size while the audio quality remains above the acceptable. Several parameters were taken into consideration to study the performance of the suggested audio compression system. The proposed audio system was implemented using C# programming language, also all additional programs for testing purpose have been developed using same programming language. The effects of the following control parameters have been investigated: (i) quantization step (Q_0) for DC coefficient, (ii) initial quantization step (Q_1) for AC coefficient, (iii) progressive rate parameter (β), (iv) block size (S_{block}), (v) sampling rate, (vi) sampling resolution, and (vii) post filtering. Table (1) shows the characteristics of the audio file, which was used in the tests. Figure (2) present the waveform patterns of these samples. As performance indicators the compression ratio (CR), and PSNR have been calculated. Table (2) presents the adopted default values of the considered control parameters, these values are selected after making a comprehensive tests and choosing the best setup of parameters. The effects of each parameter are explored by varying its value while setting other parameters fixed at their default values.

Table 1: The Attributes of the Audio Test Samples

Attribute	Audio Samples		
	Test1.Wav	Test2.Wav	Test3.Wav
Sampling Rate (KHz)	32	44.1	44.1
Sample Resolution (bps)	8	8	16
Size (KB)	848	447	893
Audio Type	Soft Music	Song	Animal Voice

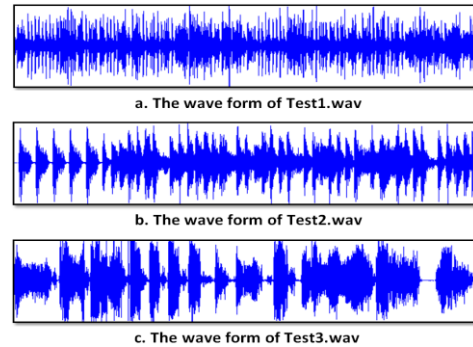


Fig. 2: The Waveform of the Tested Waveform Files

Table 2: The Default Values of the Control Parameters

Parameter	Default Value	Range
S_{block}	36 Sample	Range=[20, 40]
Sampling Rate	30 KHz	{44100, 30000, 22050, 11025}
Sample Resolution	8 bps	{8, 16}
Q_0	0.02	[0.02, 0.026]
Q_1	0.077	[0.045, 0.081]
B	0.43	[0.08, 1.03]

Figure (3) presents the effects of Q_0 on PSNR and compression ratio. While, figure (4) shows the effects of Q_1 , and figure (5) presents the effects of (β) parameter. It is obvious that the increase of these parameters causes increase in the attained compression gain while decrease the fidelity level.

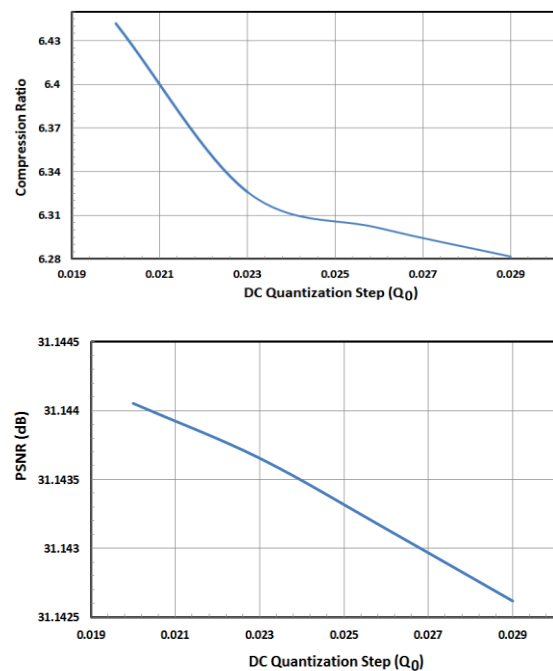


Fig. 3: The Effects of Q_0 (the audio sample is Test1.wav)

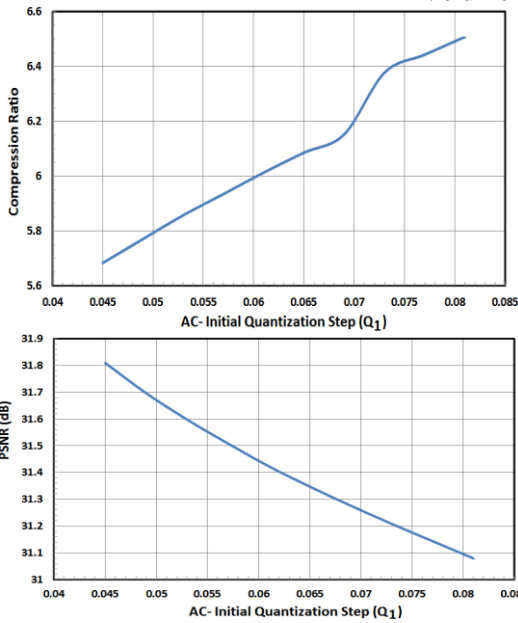


Fig. 4: The Effects of Q1 (the audio sample is Test1.wav)

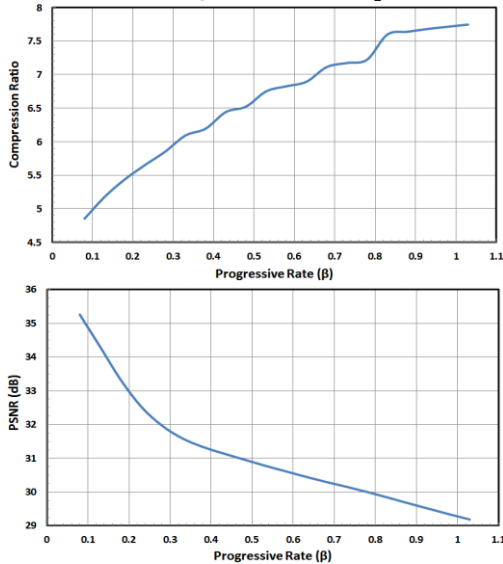


Fig. 5: The Effects of Progressive Rate β (the audio sample is Test1.wav)

Figure (6) shows the effect of block size on the relationship between compression ratio and PSNR, the results indicate that when high compression is the main concern then the selection of large block size is recommended, while for target case "low compression gain and high fidelity" the choice small block size is more suitable.

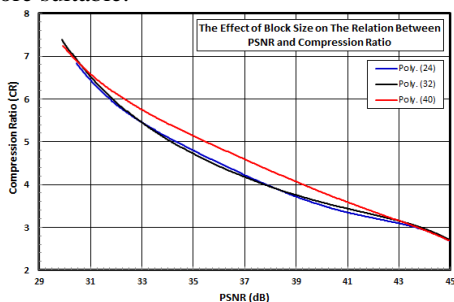


Fig. 6: The Effect of Block Size on the Relationship between Compression Ratio and PSNR (the audio sample is Test1.wav)

Figure (7) shows the effect of sampling rate on compression performance (for the two case of block size 20 & 40). The results indicate that the effect of sampling rate on the compression performance is reversed when the block size becomes high. For small block size the sampling rate doesn't show significant effect on the compression relationship, but at large block size the compression gain is higher at large sampling rate. In this test the test samples "Test2.wav" and the "Test3.wav" are down sampled by (2) twice times to get the audio wave form at sampling rate (11025 KHz).

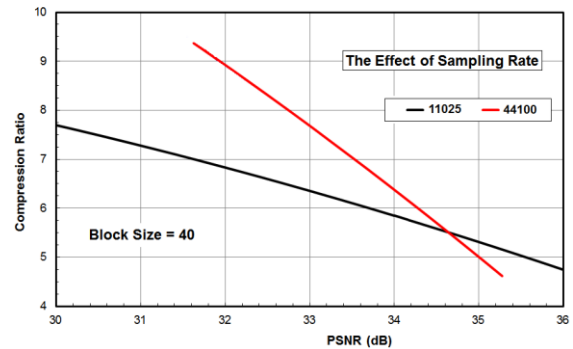
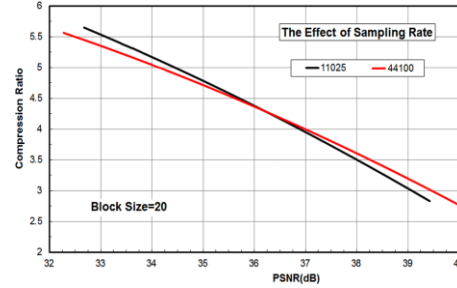


Fig. 7: The Effect of Sampling Rate on the Compression Performance (the audio sample is Test2.wav)

Figure (8) shows the effect of sampling resolution on performance of our compression schema. The results refer that the impact of sampling resolution is significant; the 16 bit sampling resolution gives more impressive result. In this set of test a conversion from 16-bits (2's complement integer representation) to 8-bit (unsigned byte representation) is done to get the corresponding low resolution audio sample. Figure (9) presents the effects of block size on the encoding time; as long as the block size is increased the encoding time is increased. So, a trade off should be taken into consideration for choosing the suitable block size when both high compression ratio low fidelity level and low encoding time.

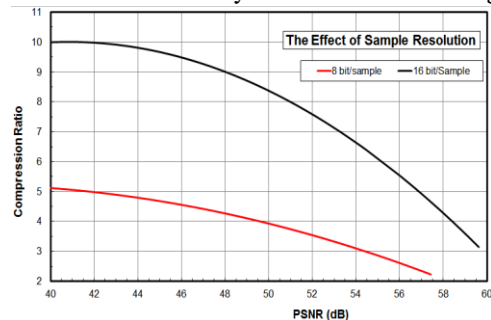


Fig. 8: The effect of Sample Resolution on the Compression Performance (the audio sample is Test3.wav)

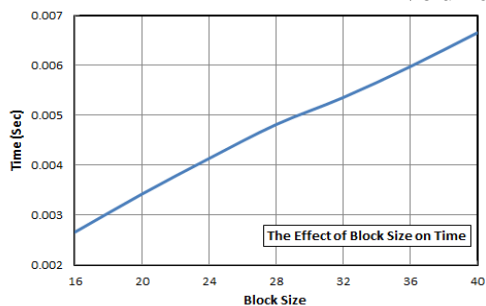


Fig. 9: The effect of Block Size on the Encoding Time

Table (3) presents the results of the test conducted to illustrate the effect of block size on compression ratio and encoding time while preserving the fidelity level (PSNR). The results indicate that the effect of block size on the compression ratio for same PSNR level is lower than its effect on the encoding time.

Table 3: The Effect of Block Size on Compression Ratio at Same PSNR (=32.8 dB)

Block Size (Sample)	Time (sec)	CR
20	2.9014	5.23
24	3.5066	5.28
28	4.0837	5.35
32	4.1477	5.42
36	5.0696	5.59
40	5.6454	5.61

Table (3) shows the effect of post processing step on PSNR and encoding time. The results that the post processing step improves the fidelity level with time expense doesn't exceed (6%) of the total encoding time. Also, with the increase of block size the fidelity level improvement is decreased. All the above mentioned results indicate that the proposed audio compression system can lead to high compression ratio without making significant degradation in quality. Also, post filtering improved the quality of reconstructed audio signal. The block size (40 samples), gives better compression results but it causes an increase in the encoding/coding time.

Table 4: The Effect of Post Processing on PSNR and Encoding Time

Block Size (sample)	Without Post Processing		With Post Processing	
	PSNR (dB)	Time (sec)	PSNR (dB)	Time (sec)
20	32.80	2.9014	33.17	3.0055
24	32.80	3.5066	33.15	3.7214
28	32.80	4.0837	33.12	4.1477
32	32.80	4.1477	33.11	4.2749
36	32.80	5.0696	33.07	5.2925
40	32.80	5.6454	33.05	5.8325

V. CONCLUSION

In this paper, an audio compression scheme using discrete cosine transform (DCT), run length, and high order shift coding had been introduced, the performance test results indicated that the proposed scheme is promising. The new

high order shift coding had successfully improved the compression gain. The new post filtering algorithm enhances the audio quality and improved the fidelity level (in terms of MSE and PSNR). As a future work the developed system can be improved by applying wavelet transform, as first transform step, in order to decompose the audio signal, then each one is passed through the scheme given in this paper to get better compression gain.

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