Effective Audio Compression using Enhanced Encoding Method

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Abstract

The rapid growth of big data and the emergence of new technologies have posed significant challenges in data compression, particularly in the domain of audio compression. Audio compression has gained widespread applications in advanced audio coding, MP3 encoding, web radio, and lossless audio coding techniques. In this paper, a newly proposed technique is present aimed at improving audio compression. The proposed technique consists of two main stages: preprocessing and the application of compression techniques. The preprocessing stage involves several steps. Firstly, the raw audio data is subjected to set normalization to ensure consistent scaling. Next, the data is divided into smaller segments to facilitate the application of Discrete Cosine Transform (DCT) and quantization on each segment. The quantization parameter is determined based on the bit per sample of the audio test file and the number of channels. To implement the Modified MLZW compression technique, the data is converted into byte values, and a word reference is constructed based on these bytes. To evaluate the performance of the proposed technique, multiple audio files have been utilized in experimental tests. The results demonstrate that the proposed technique achieves smaller MSE of $2.05*10^{-10}$ and significant improvements in terms of compression efficiency with reduction in processing time to 200 ms. The proposed methods works better and ensure that the audio quality is good and the system can be used for transmitting audio over different media that depend on lossy audio compression.

Keywords- Audio data, Audio signal processing, DCT8, Audio sample, Compression, Audio coding.

I. INTRODUCTION

The term “Data Compression” implies compressing the information to reduce its size. The need for compression continuously to transmit information effectively with less noise. In recent years, the urge for compression has risen in storage and communication. Many approaches were developed in order to maintain the best compression ratio. The compression in several aspects depends on the quality and type of data, and coding design [1], [2]. Information compression is intriguing in transmission information preparation, both because of the cost savings it offers and the expansive volume of information controlled in numerous transmission applications [3]–[6]. The sorts of the nearby excess shown in transmission information records incorporate runs of zeros in numeric areas and groupings of spaces in alphanumeric areas which are displayed in a few records and invalid in others. Data structure-based change is one of the later change coding strategies which has been utilized effectively within state-of-art information decorrelation applications [7], [8]. In this paper, a data structured based Transform for audio compression is proposed. Thus, we present a legitimate map structure for audio. It appears that the result of the proposed technique beats the ordinary transformation techniques in speed and compression ratio like Discrete Cosine Transform (DCT) and Walsh-Hadamard Transform (WHT) in decorrelation of the audio signals [9]. The motivation of our work is rooted in addressing the formidable obstacles posed by the exponential surge in big data and the constant evolution of technological landscapes, particularly in the domain of audio compression.
Compression could be a procedure utilized in data handling. It is of monstrous significance since colossal sums of information are more often than not exchanged through communication channels. Information compression is required for distinctive sorts of data specifically audio, video, pictures, and content. When audio (voice) signals are compressed, transmission and gathering get to be much simpler, in this way progressing communication. It includes lessening the number of bits contained in audio data [1], [10]. Compression calculations can be classified as lossy or lossless Information Compression [11], [12].Several researchers have made significant contributions to the field of audio compression, exploring various techniques and algorithms to achieve efficient compression with minimal loss of audio quality. The following are some notable works in the area of audio compression: In[13], the

authors proposed a method to encrypt and compress audio signal by breaking audio into frames of 1024 sample and apply DCT on each frame then multiplied by mathematical matrix generated by chaotic scheme where final results shows it satisfy the compression criteria. Researchers In [14] proposed an algorithm that used a hybrid transformation to analyze the speech data frequencies the applied filter to remove the noise from the speech to scramble the results by mixing the speech segments using fuzzy and c-means then changing their location the final steps scramble and scatter the values of blocks using sudoku puzzle and quadratic map finally it shows highly efficient compression algorithm based on approved statistical measures. While authors in [8]proposed a new method using sensing based system that compress audio signal by divide the data into segments (small sample matrix) multiplied by non square sensing matrix. Those segments reconstructed by solving a linear system based on moore-penrose. The compressed data shows a good results based on MSE and PSNR. These notable works highlight the diverse range of techniques and algorithms employed in audio compression research. They have significantly contributed to the development of efficient and high-quality audio compression methods, paving the way for further advancements in the field. This paper proposes a new approach to Audio Compression that consolidates lossless content compression calculation. The reason for Audio Compression is to decrease the sum of information required to speak to the advanced audio by evacuating excess information. Within the display work, Discrete Cosine Change and lossless content compression strategy (Lempel-Ziv-Welch strategy) based Audio Compression calculation have been proposed which too incorporates audio normalization, scalar quantization, and encoding. The MLZW lexicon gotten after compression is utilized for transmission and capacity. The execution of the proposed calculation is analyzed utilizing different audio examples of unmistakable measures with unmistakable audio flag parameters. The compression execution is surveyed utilizing Top Flag to Commotion Proportion (PSNR) and Compression Proportion (CR). The audio example results demonstrate the auspicious execution of the proposed calculation. The compression proportion can be upgraded by changing different parameters of the framework [15]. MPEG-H 3D Audio is the current standard for the compression of higher-order ambisonics data. It uses singular value decomposition (SVD) to spatially decorrelate higher-order ambisonics data, followed by the modified discrete cosine transform to exploit temporal decorrelation. Prominent and ambient sound components are then separately encoded (e.g., using the standard core audio codec) and sent to the decoder [16]. In this paper, a preprocessing steps applied on the raw data to be processed by the mathematical methods to reach the highest compression ratio and stability. the proposed system used wavelet, DCT8, Quantization and modified LZW and these were affect the system behavior based on the initial parameter of block size, number of channel, sample resolution and wavelet iteration.

II. PROPODED AUDIO COMPERESSION

The improvement of Audio information compression has altogether made our lives less demanding. The mechanical advance has encouraged the method of recording and exchanging audio on distinctive media. In later a long time, much has been accomplished within the field of audio and discourse compression. Numerous benchmarks have been built up in arrange to compare strategies of audio information compression Figure 1 shows the block diagram of the proposed system.

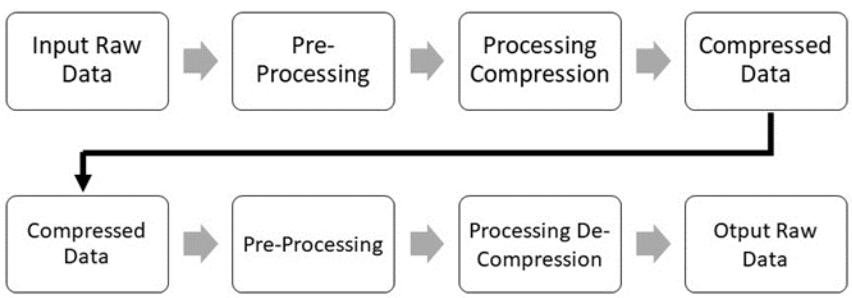

FIGURE 1. PROPOSED SYSTEM BLOCK DIAGRAM

A. Pre-processing

Pre-processing steps are fundamental for the structure of the audio information and make afterward steps of the system work smoothly. Figure 2 shows data pre-processing model applied in our proposed system

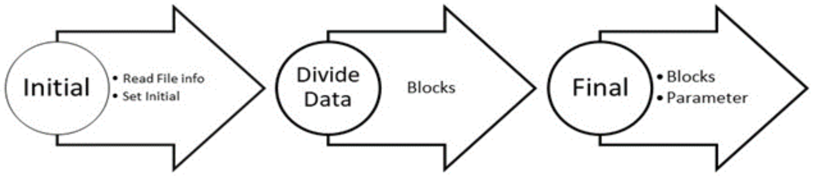

Figure 2. Data pre-processing Model

As can be seen from Figure2, preprocessing model includes parameter initialization (read file and set initial values), Divide Audio file into blocks, and finally, set the number of processing iterations to read the blocks

1. *Initialize Parameter*

The first preprocessing step is to set initial values of the parameters of the system, and that is done by the process of altering the parameters of a model in order to improve its performance. To put it another way, it's the process of determining the optimum parameters in a predetermined hypothesis space in order to achieve the best potential results. As a result, parameter initialization is crucial for accelerating convergence and lowering error rates. The equation (1 - 11) are used to calculate the parameters of the proposed system [23].

2. *Audio File*

To retrieve the basic file information, the header of file data is read (i.e., the no. of samples and channels, sampling resolution). Also, when the sample resolution is set to 8 bits per sample, the file (in .wav format) is loaded as a bytes array of unsigned values, and when the audio file resolution is set to 16 bits per sample, it is loaded as bytes array of signed values. The following equations represent the initialization parameter step after reading the audio file information.

$$bps = \frac{bpsample}{8} \tag{1}$$

Where bpsample represent the number of bits per sample, bps is the number of bytes per sample. To be used in reading samples values from raw data.

$$N = \frac{L-44}{bps} \tag{2}$$

Where N is Length of audio data. Which represent the total number of samples. While L is the total length of the file in bytes including the file header.

$$x = \frac{sample\ rate}{10000} * no.channel \tag{3}$$

Where x is the required percentage of normalization that used to normalize the samples values. While the sample rate means the required number of samples within a second to create discrete digital signal and the no. channel means the channels that represent the current audio signal reporodeced by a speakers.

$$Norm = \begin{cases} 1, & x < 1 \\ Norm, & x \geq 1 \end{cases} \tag{4}$$

Where Norm is the normalization value.

$$Data(i) = \frac{raw(i*bps)}{Norm} \tag{5}$$

Where raw is the raw data array of audio file. The output (Data(i)) is prepared to next stage.

$$NT = N\ Mod\ 2^{no.iteration} \tag{6}$$
$$N = N - NT \tag{7}$$

Where NT is the remaining number of the array values for iteration process to keep the length of data the multiplication of 2. 1No. iteration represent the iteration of wavelet process applied.

$$Qv = Round(5 * \left(\frac{samplerate}{8000}\right) * no.channel) \tag{8}$$
$$Qs = Qv * 0.3 \tag{9}$$

Where Qv is the quantization value and Qs is the quantization step that are used in quantization process.

$$Block\ size = chunck\ size * no.channel \tag{10}$$
$$Loop = \frac{L*no.channel}{chunck\ size} \tag{11}$$

Where L is the total number of Samples in audio file and no.channel is the number of source originating the audio and chunk size represent the number of samples are going to be processed in each time. Block size and loop will determine the processing iteration, final block of data process before going to next step.

B. Audio Signal Processing

The data entered into the system in the previous stage is actually pre-processed for interpretation during this stage. The procedure is carried out using mathematical model algorithms, though it may differ slightly depending on the audio file specification and signal information of the data being processed, as well as the intended use of wavelet, DCT, and quantization to boost the MLZW compression ratio. Figure 3 shows the processing stage of our system.

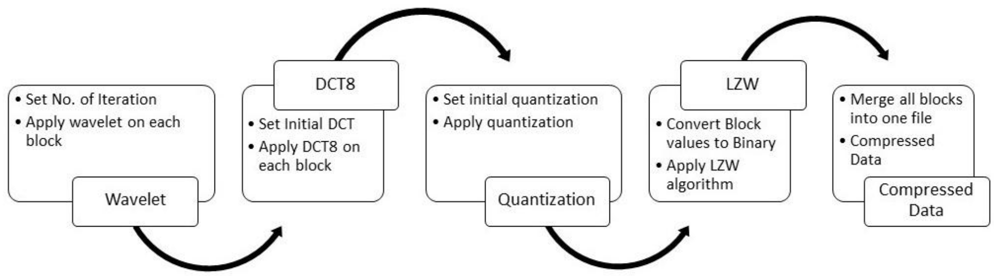

Figure 3. Proposed audio signal processing algorithm

1. *Wavelet Transform*

It is feasible to compress audio using wavelet transform. An audio stream represented in sparse form permits efficient compression without losing important information. The Fourier transform, unlike the wavelet change, speaks to signals locally in recurrence and globally in time. The wavelet transformation refers to a well-supported assumption in both frequency and time. Another key advantage of using wavelet transform which is can be used to extract temporal and spectral info at same time and haar wavelet was the best option a mony different wavelet methods to be used. As shown in Figure 4.

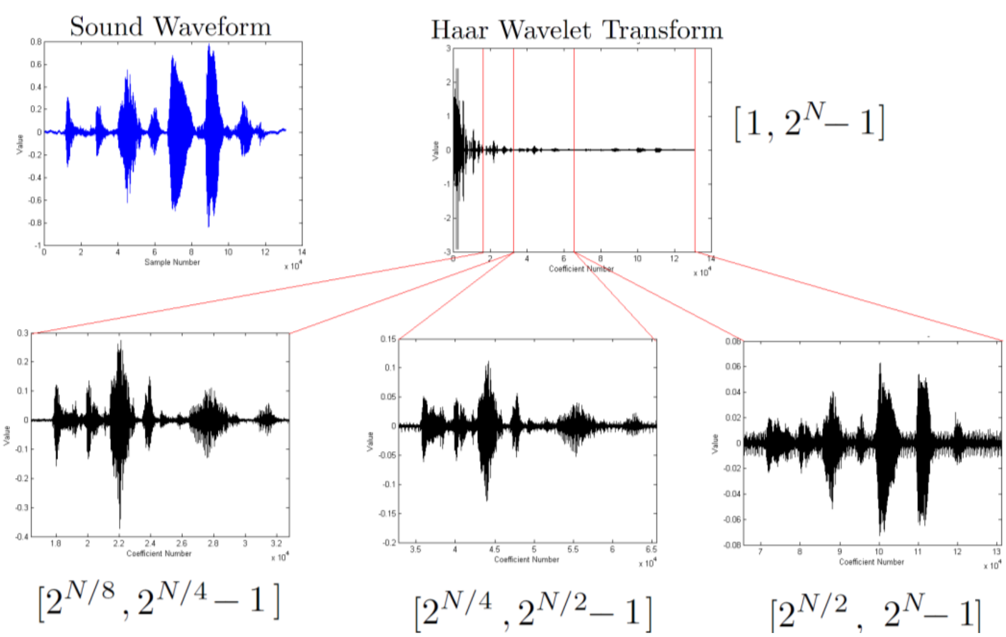

Figure 4. Block of audio signal processed by wavelet transform

2. *Discrete Cosine Transform (DCT-8)*

Algorithmic efficiency on modern hardware is often not primarily dictated by simple arithmetic counts, and optimization necessitates significant engineering work to make the greatest use of available built-in hardware optimization, within its intrinsic bounds. In These papers [11], [17] DCT is discussed and analyzed as a main stage for transforming file data before applying the compression of any file format. The Discrete Cosine Transforms, or DCT, have been used in speech and image processing for the past decade in fields like compression, filtering, and feature extraction. An image can be turned into its simplest form using DCT components, DCT employs an oscillating sum of cosine functions [18]. To express a sequence of finitely. Many symmetrical discrete and genuine data points [19]. This can be represented in [7], which consists of a set of sampled cosine functions as basis vectors. The scaled DCT type II on the specified length-8 array in place was used in the proposed system and described in [20], [21], as shown in Figure (5). In general, DCT applied on block data so breaking audio into small blocks or chunks is suitable for DCT where we used these blocks in the proposed compression process.

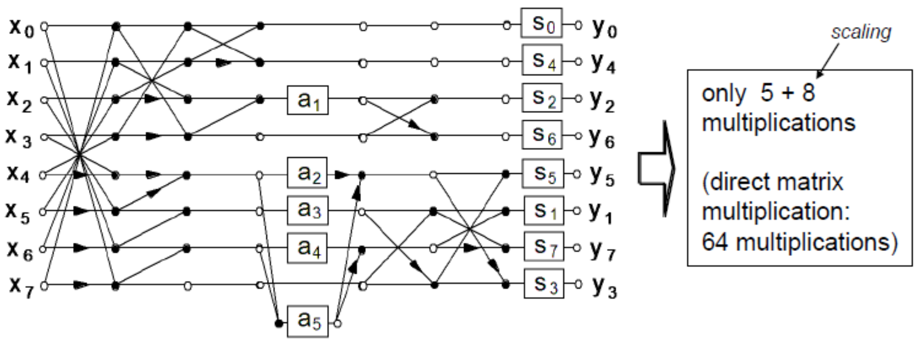

Figure 5.Proposed Scaled Decrite Cosine Transform type II (DCT8)

3. Quantization

Quantization is a critical stage for data compression, it makes inexact mapping of transform coefficient values to numbers values have a limited length of binary representation [22]. Any quantization strategy ought to be based on a rule that decides what information things to quantize and by how much [23]. In change-based coding plot, the lossy encoder must quantize each recurrence component. In case the encoder distributes more bits to each recurrence, less blunder (commotion) is presented, but more space is required to store the result. Then again, fewer bits distributed to each recurrence come about in more clamor, but less space is required to store the result [24]. Quantization step can be defined in equation (12) [23]

$$Q_i = \frac{Round(Val_i)}{Qv} \tag{12}$$

Where Qi is the quantization result and val is the DCT value and Qv is the quantization value.

4. Modified LOSSY MLZW AUDIO COMPRESSION

The Lempel-Ziv compression method [25]and the extension of prescient coding [26]form the basis of the contemporary compression technique. A numerical process known as linear expectation might be used to evaluate the present value of a discrete-time flag as a direct result of previous (forward expectation) and/or future (in reverse expectation) checks. Each flag square is encoded by predictive coders as an additional- or interpolation work of a small window of previously and alternatively subsequently encoded bits. Lossy compression situations for audio and video signals frequently employs such coders. Lem–Ziv's LZ77 calculation [27] maintains a history window of the most important information that has recently been seen and compares the information now being encoded with the information in the history window. References to the position inside the history window and the coordinate length are actually placed into the compressed stream. The character itself is essentially encoded into the stream after being declared as demanding in the event that a coordinate cannot be identified. In these compositions, we use variations of this computation and expand the LZ77 calculation to back lossy compression. Every flag part might often be of varying length—is referred to as a simple combination of parts from the history window and an error vector. Pointers pointing to those squares' locations in the history window are used to encode the direct combinations of previous squares. A perceptual display is used to encode the error vector. The goal is to reduce the entropy of the approaching error vector by causing perceptually substandard data destruction. The idea is that the mistake should manifest as low-magnitude noise given the assumption that repeating material, such as audio, exists. The CR and PSNR are the metrics used to measure the efficiency of the compression procedure in this set of tests [28]. The need to use alternative audio measurement of PEAQ is mentioned in [29].

III. RESULTS AND DISCUSSIONS

To evaluate the effectiveness of the proposed audio compression system, eight different sample audio files were used to measure the efficiency of audio compression. The waveform patterns of the audio files are shown in Figure 4 as presented in previous section. Table 1 lists the details of the eight audio recordings that were used. All of these files are in wave file format (.wav) and have a mono and double channel. In this table, sample rate, number of channels, file size, and audio type for each audio file is presented

The impacts of the following system factors have been examined: a) the numbers of wavelet iteration (Niter), b) the quantization steps Qv and Qs, and c) block size (BS). Table 2 shows the set of parameters for the proposed system parameters. The fundamental concern in audio compression is high compression gain, while the main purpose of this research is to achieve a good compression ratio at a high-fidelity level. As shown in table 2 the parameter used in the system is as follows: a) iteration used for wavelet transform, (Qv, Qs) were used for the quantization process, the Block size used for dividing the data into blocks the predefined size and Normalization value used for scaling down the high data values.

TABLE 1. PROPERTIES OF SAMPLES USED IN THE ASSESSMENT.

No.	Sample rate	No. channel	Size	Audio type
1	22050	1	88.286KB	Bell Ring
2	8000	2	1073KB	Music
3	44100	2	5226.722KB	Music
4	44100	2	10406.69KB	Music
5	44100	1	26296,95KB	Music
6	44100	2	400.788KB	Horn
7	16000	1	97.486KB	Music
8	4410	2	2200KB	Music

TABLE 2. SERIES OF PARAMETERS USED FOR THE PROPOSED SYSTEM.

Iteration	Qv	Qs	Block size	Normalization value
1-3	1-35	1-15	8000-128000	128

Table 3 shows the compression result obtained using the proposed system. Each audio file presented in table 1 was processed by the MLZW audio compression, as can be seen, the compressed sizes shows significant file size reduction compared to the original one.

TABLE 3. RESULTS OF THE COMPRESSION SYSTEM FOR THE SELECTED SAMPLES FROM TABLE 1

No.	Size	Compressed Size
1	88.286KB	11.25KB
2	1073KB	125.33KB
3	5226.722KB	496.78KB
4	10406.69KB	901.3KB
5	26296,95KB	213.12KB
6	400.788KB	41.25KB
7	97.486KB	11.64KB
8	2200KB	135.46KB

The compression system shows good results for each file compressed. In table 4 we will discuss the impact of parameters on each process step and how they affect the process time and compression ratio. Table 4 shows the different parameters on the same file produced different results as example sample 1 with sample rate (22500) and a number of channels (1) at the first round of the iteration of wavelet transforms (2) and the quantization value is (14) and quantization step is (4) with chunk size is (32000) produced a small value of compression ratio with MSE (5.13*10-10) and the processing time is (280 msec). Thus, to increase the compression ratio all the parameters were increased to an acceptable ratio where the decompressed audio is restored to the point to match the original audio for the human ear. Based on the results conducted in the proposed system it shows different behavior based on the initial parameters. Figure 6 showing the wavelet iteration is effecting processing time. By increasing the iteration number, the process time reduced accordingly.

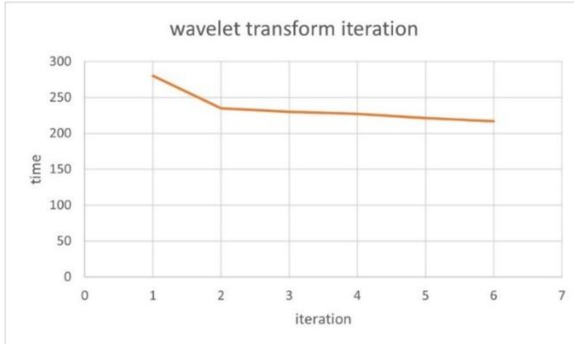

Figure 6 wavelet iteration effect on time

TABLE 4. Different parameters used on a few samples to show the impact of parameters in each process step

File Name No. channel Sample Ratio Size	ITERATION	QV	QS	BS	NORM	MSE	PSNR	TIME msec	CS
T1 1 22050 25681 KB	2	14	4	32000	128	$5.13*10^{-10}$	111.34	280	19.729
	3	14	4	32000	128	$2.05*10^{-10}$	105.32	231	15.726
	3	44	44	32000	128	$2.053*10^{-10}$	105	231	9.874
	3	44	44	64000	128	$2.035*10^{-10}$	105.32	217	8.99
T2 2 44100 392KB	2	55	16	32000	128	$1.615*10^{-10}$	92.891	1197	46.962
	4	55	16	32000	128	$1.055*10^{-10}$	94.741	1079	20.242
	4	80	40	64000	128	$1.055*10^{-10}$	94.741	1410	18.71
	6	80	40	6400	128	0.002	41.515	1393	10.102
B1 2 8000 1049KB	2	10	3	32000	128	0.044	37.385	2195	256.462
	4	20	10	32000	128	0.048	37.032	1929	132.46
B2 1 16000 2069KB	4	20	1	32000	128	0.0058	41.449	47.10	132.958
	4	30	1	64000	128	0.0058	41.449	48.78	87.696
B3 2 44100 5105KB	2	55	16	32000	128	0.00024	51.054	12523	368.474
	2	80	40	128000	128	0.00024	51.054	12720	226.46
	2	80	40	128000	256	6.134	54.064	12070	162.586

Figure 7 shows the number of wavelet iterations vs. compression size. As can be seen the increase of wavelet iteration can lead to increasing compression size by reducing the resulted data.

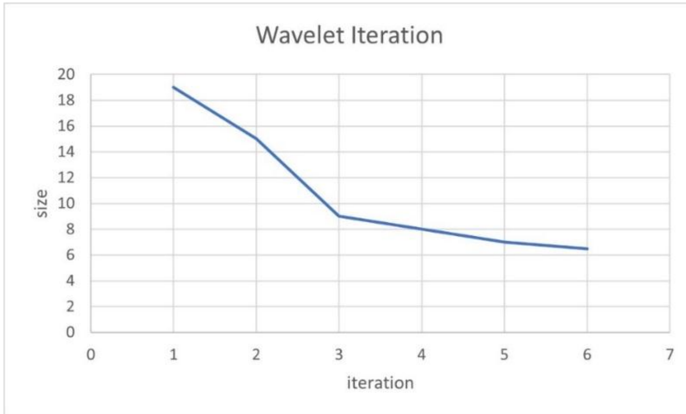

Figure 7 wavelet iteration effect on compression size

In Figure 8, the wavelet iteration effects MSE where, at first two iteration significant reduction in MSE can be observed, and then the effect decreases gradually as the number of iteration increase. Figure 9 show the Quantization values effect on compression ratio when the quantization step increase it give high compression ratio.

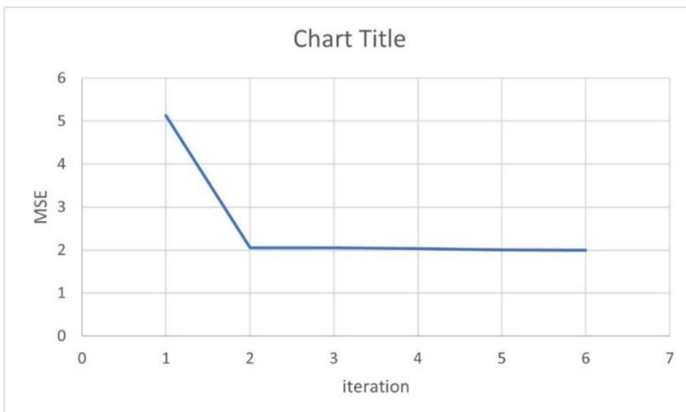

Figure 8 wavelet iteration effect on MSE

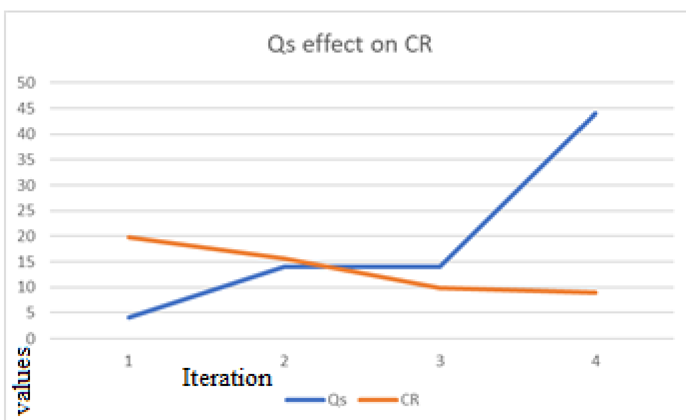

Figure 9 Quantization Step value effect on CR

Figure 10 shows the Quantization step (QS) effect. As can be seen, increasing QS can decrease PSNR drastically. Figure 11 presents the effect of block size on CR and PSNR. The block size parameter effects the relation between two measures i.e. (CR, PSNR).as can be seen, smaller block size have a better result. Also, the sample resolution effect the relation between the (CR, PSNR) and Figure 12 show the higher resolution gives higher compression ratio.

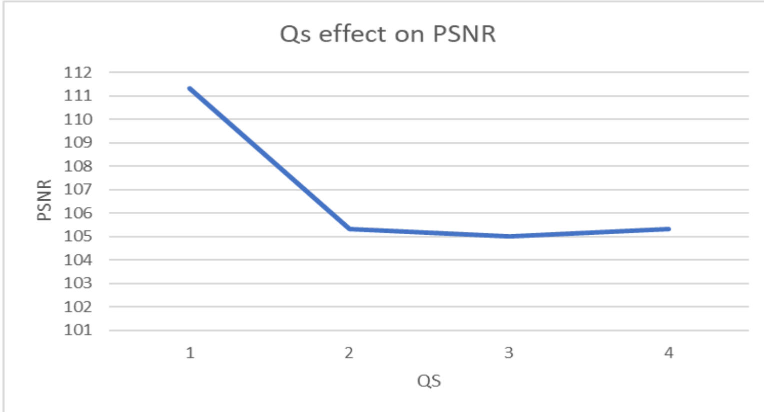

Figure 10 Quantization Step value effect on PSNR

In comparing the existing work with the proposed approach presented in this paper, a comprehensive analysis was conducted to highlight the key differentiating factors. The existing work, as observed from the table 5, primarily focuses on KB size, MSE, and Compressed size of the file to assess its performance.

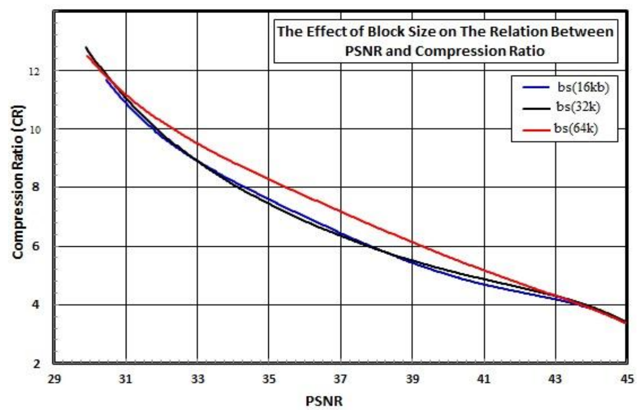

Figure 11 the effect of block size on CR and PSNR

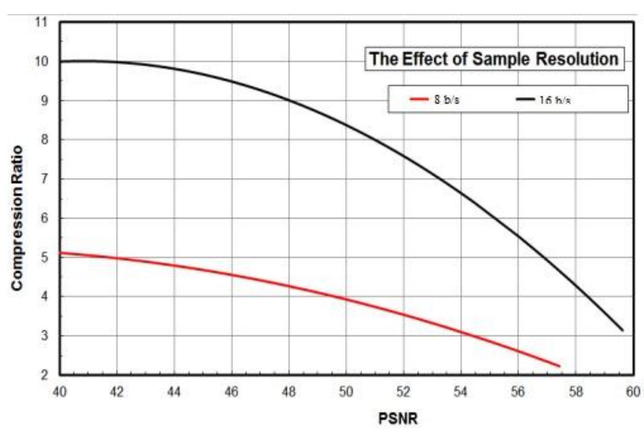

Figure 12 the effect of sample resolution on CR

TABLE 5. COMPARATIVE ANALYSIS OF EXISTING LITERATURE AND THE PROPOSED APPROACH.

Research	KB size	MSE	Compressed Size
Moreno-Alvarado et al. [13]	128	0.28	88
	256	0.11	176

	512	0.026	352
	700	0.006	492
	1024	0.0000004	704
Abduljaleel and Khaleel [14]	128	0.000000729	61.2483
	256	0.000000658	103.0695
	512	0.00000062188	259.0295
	700	0.0000006868	303.7014
	1024	0.00000087965	612.1763
Enas Wahab Abood [8]	13	0.007	18
	50	0.0000008306	28
	147	0.0000008154	81
	505	0.0000007051	254
	1020	0.000004001	501
Proposed	25681	$2.05*10^{-10}$	8900
	392	$1.615*10^{-10}$	10.102
	1049	0.044	132.958
	2069	0.0058	87.696
	5105	0.00024	162.586

While these studies have made notable contributions in their respective domains, they often encounter limitations in terms of KB size, MSE, and Compressed size. In contrast, the proposed approach addresses these limitations by combining lossless content compression techniques, such as Discrete Cosine Transform and Lempel-Ziv-Welch strategy to improve performance. By incorporating aforementioned method, the proposed approach offers distinct advantages, including KB size and compressed size while MSE remained within the same range. Overall, the comparison table showcases the strengths and potential of the proposed approach, indicating its potential to surpass the existing work and make significant contributions to the field.

IV. CONCLUSION

The audio compression process is considered as one of the most important fields in audio processing, which has a direct impact on the transmission of audio over the Internet and digital applications, in addition to the storage process. The ratio of the audio compression process and its stability is one of the most important characteristics of the audio compression process, which is determined by some standards. The proposed method varied dealing with audio files with different characteristics. Those characteristics have a major effect on the proposed method results and it gives different results each time the characteristics changed. A preprocessing steps applied on the raw data to be processed by the mathematical methods to reach the highest compression ratio and stability. the proposed system used wavelet, DCT8, Quantization and modified LZW and these were affect the system behavior based on the initial parameter of block size, number of channel, sample resolution and wavelet iteration. It shows that small block size with higher number of channel and sample resolution gives best result of audio compression and by increasing the wavelet iteration it decreases the process time of compression. Along with MSE, PSNR and CR as a metrics it shows that the proposed methods works better and ensure that the audio quality is good and the system can be used for transmitting audio over different media that depend on lossy audio compression. This novel approach of audio compression offers promising advancements in addressing the challenges posed by the growing demand for efficient audio compression in various applications. For future work, optimization techniques can be used for parameter tuning and DCT-8 adjustment to achieve a higher compression ratio with less processing time.

ACKNOWLEDGMENT

Authors would like to thank the Ministry of Higher Education and Scientific Research, Al-Iraqia University / College of Engineering for their support and engorgement.

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