Lossless Audio Coding based on Burrows Wheeler Transform and Run Length Encoding Algorithm

Pratibha Warkade[1], Agya Mishra[2]
M.E. Scholar, Dept. of Electronics and Telecommunication Engineering, Jabalpur Engineering College, Jabalpur, Madhya Pradesh, India[1]
Assistant Professor, Dept. of Electronics and Telecommunication Engineering, Jabalpur Engineering College, Jabalpur, Madhya Pradesh, India[2]

Abstract

In this paper we present a new lossless audio coding algorithm using Burrows-Wheeler Transform (BWT) and Run Length Encoding (RLE).Audio signals used are assumed to be of floating point values. The BWT is applied to the audio signals to get the transformed coefficients and then these resulting coefficients are better compressed using Run Length Encoding. Two entropy coding are used which are Run Length Encoding and Huffman coding. Proposed compression algorithm is experimented and analyzed for two different stereo type audio signals. Compression ratio and Bit rate for audio coding has been used as a comparison parameter for proposed audio coding algorithm. Experimental result shows that the lossless audio coding algorithm outperforms other lossless audio coding methods; using combined Burrows Wheeler Transform & Move to front coding method ,using combined Burrows Wheeler Transform and Huffman coding method, and using Burrows Wheeler Transform ,Move to front coding method & Run Length Encoding method.

Keywords: *Audio Coding, Burrows-Wheeler Transform (BWT), Bit rate, Compression ratio, Huffman Coding(HC), Move to front coding(MTF) and Run Length Encoding(RLE).*

I. INTRODUCTION

With the rapid growing of necessary data and increased number of applications, devising new approach for efficient compression and encryption methods are playing a vital role in performance. There has been an unprecedented increase in the amount of digital data transmitted via networks especially through the internet and mobile cellular networks, over the last decade. Data compression offers an attractive approach to reducing communication cost by using available bandwidth effectively. Digital data represent text, images, audio, video, sound etc. With this trend expected to continue, it makes sense to pursue research on developing algorithms that can be most effectively use available network bandwidth by maximally compressing data. Many methods in conjunction with BWT is discussed to achieve this. It has been observed that a pre-processing of the audio signal prior to conventional compression method will improve the compression efficiency much better.
Lossless audio coding enables the compression of digital audio data without any loss in quality due to a perfect reconstruction of the original signal. But in other case, modern perceptual audio coding standards are always lossy, since they never fully preserve the original audio data.

In this paper, we proposed lossless audio coding methods using the BWT of the input audio signal to convert the resulting coefficients to a form that can be better compressed using Run Length Coding than the resulting coefficients that are compressed using a combination of the BWT & Huffman coding method [4]and only BWT method [3] and BWT,MTF & RLE method[1].

In this paper a new audio coding algorithm is proposed which include the advantages of both BWT and entropy coding in [1][3][4] and combine them into single method as BWT & RLE algorithm. This paper is organized as follows. Section II deals with compression techniques using BWT-RLE. In Section III,the proposed algorithm using BWT-RLE has been discussed in detail. In Section IV,in experimental results, the effectiveness of our proposed audio coding algorithm is checked by couple of case studies using the Compression Ratio and Bit Rate values. Concluding remarks are given in Section V.

II. COMPRESSION TECHNIQUES USING BWT & RLE

In this section ,BWT & RLE has been used as compression techniques and explained how effectively it facilitate the process of audio signal compression with the help of compression ratio and bit rate of compressed audio signal.

A. Burrows Wheeler Transform

First level of compression is BWT; the Burrows-Wheeler transform is a block-sorting, lossless data compression algorithm that works by applying a reversible transformation to a block of input data. The transform does not perform any compression but modifies the data in a way to make it easy to compress with other algorithm such as "move-to-front" coding and then RLE, Huffman and Arithmetic coding. If the file size is large then BWT algorithm does not process data in one time slot, but takes blocks of data at a time unit, and process it sequentially.

The transform operates on a block of symbols, X, of length N to produce a permuted data sequence, Y, of the same length and a single integer u, $1 \leq u \leq N$. The forward transform BWT is given by {Y,u}=BWT{X}..(i)

For example, the BWT of the samples [3];
X = {1, 2, 4, 6, -1, 3, 2, -4, 6, -7} is given in Table I [3].Considering last column and the fourth row of Table I (b),Then BWT {1, 2, 4, 6, -1, 3, 2, -4, 6, -7} = ({ 6, 2, 6, -7, 3, 1, - 1, 2,-4, 4}, 4).

And the inverse Burrows Wheeler Transform is given by {X} = BWT^{-1}{Y,u}...........(ii)

An example for this inverse BWT is found in appendix [1]. Then the IBWT ({ 6, 2, 6, -7, 3, 1, - 1, 2, -4, 4}, 4) = {1, 2, 4, 6, -1,3, 2, -4, 6, -7}.

The dependence of BWT on input data encoding is a relatively unique characteristic for general lossless data compression algorithms.

B. Run Length Encoding

Second level of compression is Run Length Encoding (RLE) explained as follows:

RLE method is implemented, in order to encode transformed coefficients obtained after applying BWT. Run-length encoding is a popular data compression algorithm in which Runs of data (that is, sequences in which same data values occur in many consecutive data elements) are stored as a single data value and count, rather storing original repeated runs. This type of encoding (RLE) is very useful in data which contain many repeated runs. RLE [7] is lossless data compression technique in which runs of data is stored as single data value and followed by count as shown in Figure1 of [7]. RLE works by reducing the physical size of a repeating string of characters. This repeating string, called a 'run', is typically encoded into two bytes. The first byte represents the number of characters in the run and is called the 'run count'. The second byte is the value of the character in the run, which is in the range of 0 to 255, and is called the 'run value'. For example: A file with '0' as repeating character. Two characters in the compressed file replace each run of zeros. For the first 3 repeating '0's in original file, the first encoded stream in compressed file is showing that '0' was repeating '3' times , '1' time, '5' times and then '2' times as shown below in the equations (iii) & (iv):

Original data stream:17 8 54 0 0 0 97 5 16 0 45 23 0 0 0 0 0 3 67 0 0 8................................(iii)

Run length encoded data:17 8 54 0 3 97 5 16 0 1 45 23 0 5 3 67 0 2 8...................................(iv)

Hence, both the compression techniques has been successfully implemented in our proposed work.

III. PROPOSED ALGORITHM BWT & RLE

Figure1. describes the flow-chart of proposed coding algorithm:

From the given flowchart it can be concluded that input audio signal (.wav file) is taken and sampled at different sampling frequency in order to get sampled input signal to which BWT is applied as a result we get transformed coefficient values of signal to get compressed audio signal & hence compression ratio and bit rate is calculated to measure its performance, whole process is implemented in MATLAB.

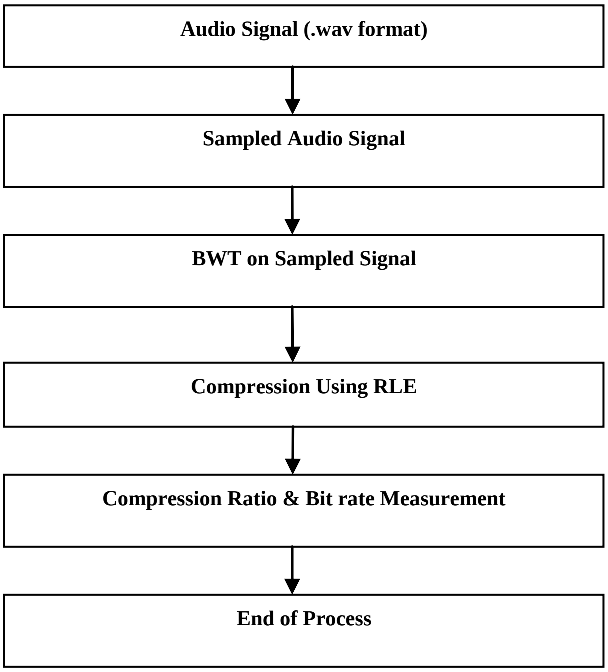

Fig.1.Flowchart of the Proposed Audio Coding Algorithm

A. Measuring Parameters

In present work, Compression ratio & Bit rate has been chosen as the measuring parameters for the performance evaluation of our proposed work. The Compression ratio and Bit rate is calculated as follows:

1) ***Compression Ratio*** is the ratio between the size of the compressed file and the size of the input audio file.

$$\textbf{C. R.} = \frac{\text{size of the compressed file}}{\text{size of the input audio file}} * 100$$

2) ***Bit Rate (bits/sample)*** is the file size in bytes time 8 bits per byte divided by number of channels and divide by number of samples.

$$\textbf{Bit Rate} = \frac{\text{file size in bytes *8 bits/bytes}}{\text{No. of Channels * No. of Samples}}$$

IV. EXPERIMENTAL RESULTS

In this section, we evaluate the proposed Burrows-Wheeler Transform (BWT) and Run Length Encoding (RLE) algorithm by couple of case studies.

Two audio signals of stereo type are considered in case studies and results are discussed here.

A. Case Study 1: In this case audio signal 1 of stereo type is taken at three different sampling rate and sampled at different frequency to compute compression ratio and bit rate of different audio compression methods in four sections 1,2,3 and 4 as follows:

- In this case we have considered audio signal 1 at 48 KHz sampling rate & sampled at 8KHz to compute compression ratio and bit rate of different audio compression methods as shown in Table I.As shown below Figure 2 shows input signal & Figure 3 shows input sampled sequence at 48 KHz sampling rate of audio signal 1.

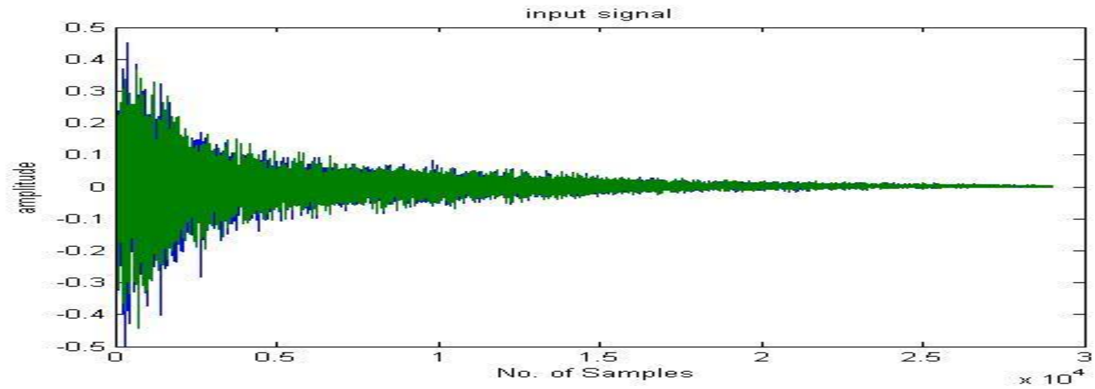

Fig.2. No. of samples v/s amplitude

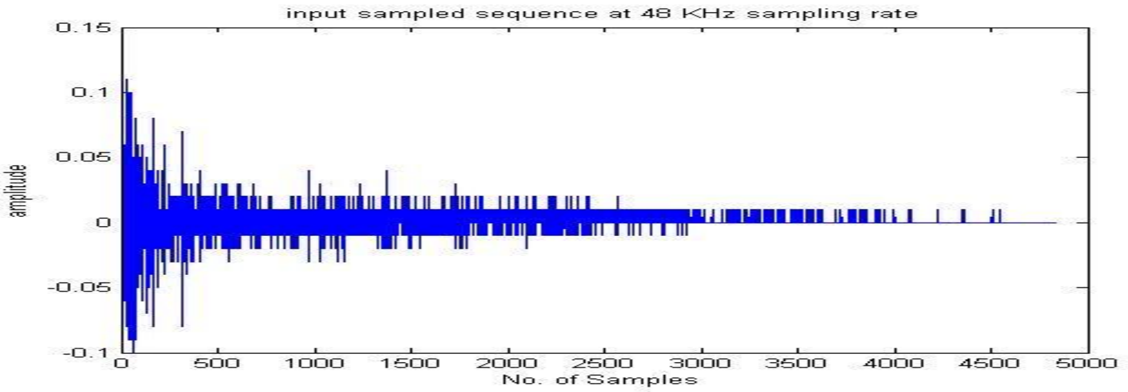

Fig.3. No. of samples v/s amplitude at 48 KHz

Table I :Calculation of compression ratio and bit rate for audio signal 1

Input Signal	Sampled Input Signal(a)	Compressed Signal	Compression Ratio	Bit Rate(bits/Sample)
BWT method	75.56 KB	37.78 KB	50%	5.20 bps
BWT & Huffman Method	75.56 KB	66.72 KB	88.30%	9.19 bps
BWT & RLE Method	75.56 KB	31.62 KB	41.84%	4.35 bps

- In this case we have considered audio signal 1 at 44.1 KHz sampling rate & sampled at 8KHz to compute compression ratio and bit rate of different audio compression methods as shown in Table II.As shown below Figure 4 shows input signal & Figure 5 shows input sampled sequence at 44.1 KHz sampling rate of audio signal 1.

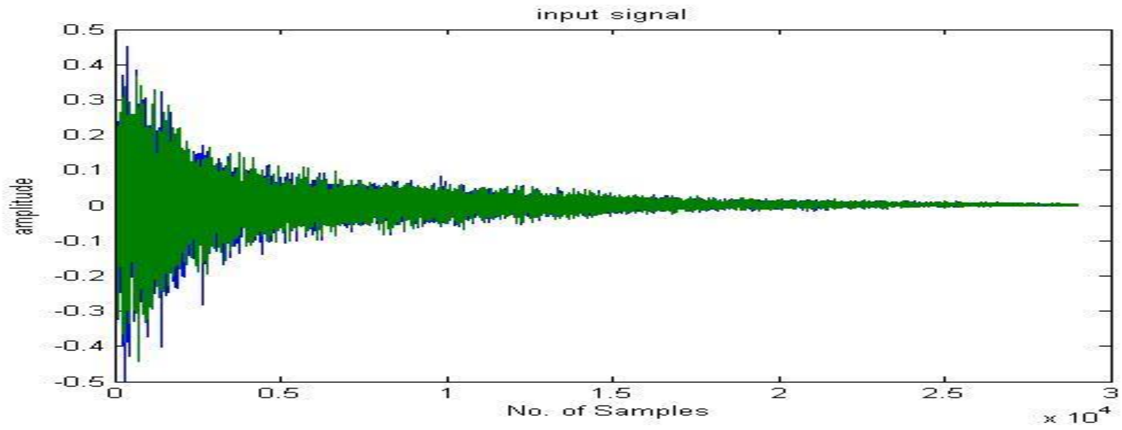

Fig.4. No. of samples v/s amplitude

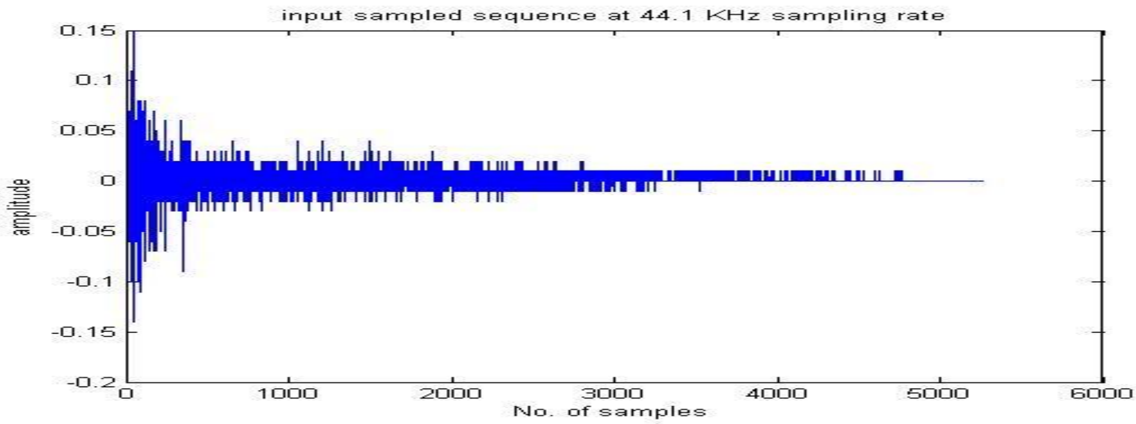

Fig 5. No. of samples v/s amplitude at 44.1 KHz

Table II: Calculation of compression ratio and bit rate for audio signal 1

Input Signal	Sampled Input Signal(a)	Compressed Signal	Compression Ratio	Bit Rate(bits/Sample)
BWT method	82.25 KB	41.13 KB	50%	5.66 bps
BWT & Huffman Method	82.25 KB	76.61 KB	93.14 %	10.56 bps
BWT & RLE Method	82.25 KB	36.65 KB	44.55%	5.05 bps

- In this case we have considered audio signal 1 at 32 KHz sampling rate & sampled at 4KHz to compute compression ratio and bit rate of different audio compression methods as shown in Table III .As shown below Figure 6 shows input signal & Figure 7 shows input sampled sequence at 32KHz sampling rate of audio signal 1.

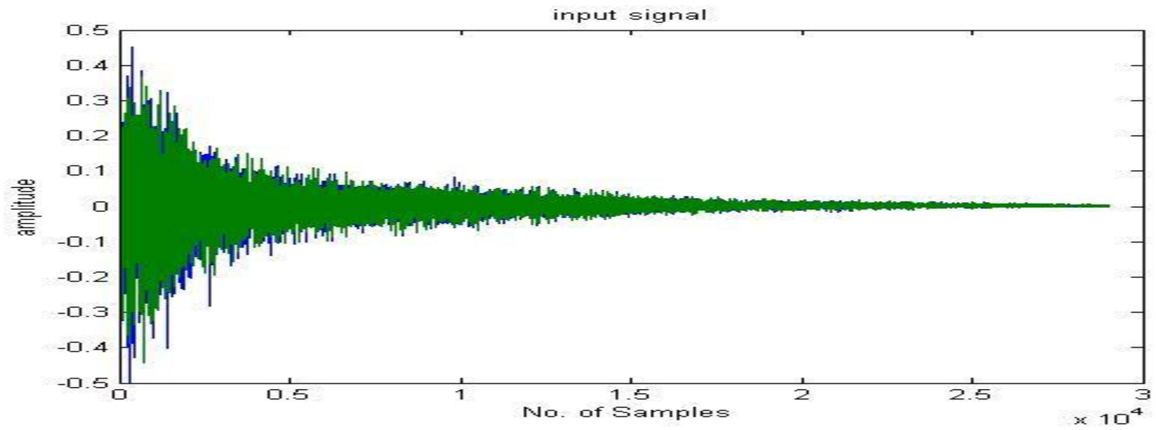

Fig. 6. No. of samples v/s amplitude

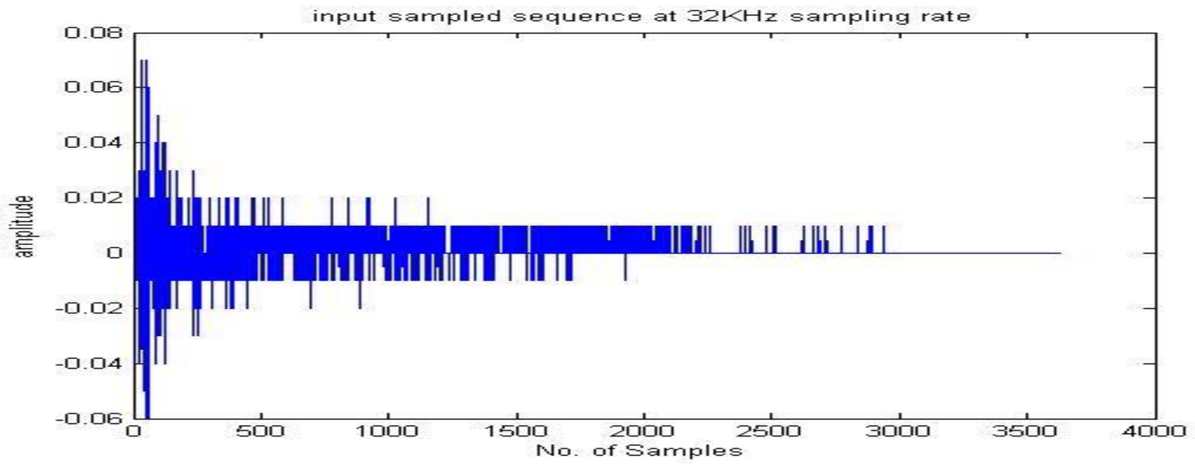

Fig.7. No. of samples v/s amplitude at 32 KHz

Table III: Calculation of compression ratio and bit rate for audio signal 1

Input Signal	Sampled Input Signal(a)	Compressed Signal	Compression Ratio	Bit Rate(bits/Sample)
BWT method	56.67KB	28.34KB	50%	3.90 bps
BWT& Huffman Method	56.67KB	40.45KB	71.37 %	5.57 bps
BWT & RLE Method	56.67KB	17.56KB	30.98%	2.42 bps

- ***Performance Analysis:***

Figure 8 shows that the bit rate for three different sampling rates 32,44.1, 48 KHz using proposed method. From this Figure, we show that the bit rate depends on the sampling rate; the bit rate for the 32 KHz is less than the bit rate of 44.1 and 48 KHz, hence it is better than 44.1and 48 KHz sampling rates.

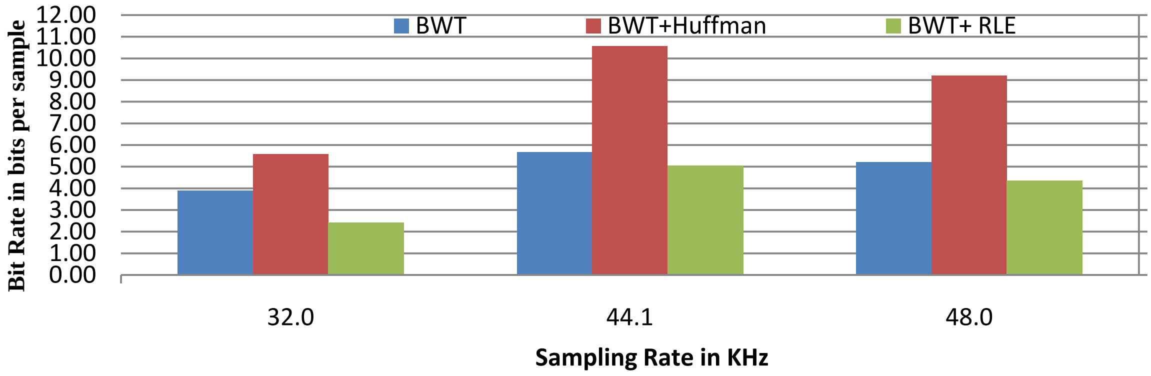

Fig.8.Sampling Rate v/s Bit Rate of audio signal 1

B. Case Study 2: In this case audio signal 2 of stereo type is taken at three different sampling rate and sampled at different frequency to compute compression ratio and bit rate of different audio compression methods in four sections 1,2,3 and 4 as follows:

- In this case we have considered audio signal 2 at 48 KHz sampling rate & sampled at 4KHz to compute compression ratio and bit rate of different audio compression methods as shown in Table IV.As shown below Figure 9 shows input signal & Figure 10 shows input sampled sequence at 48KHz sampling rate of audio signal 2.

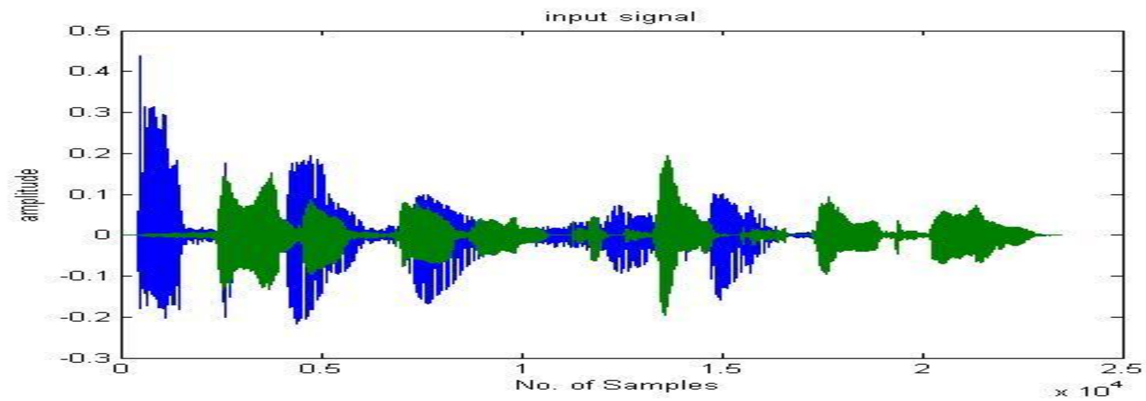

Fig.9. No. of samples v/s amplitude

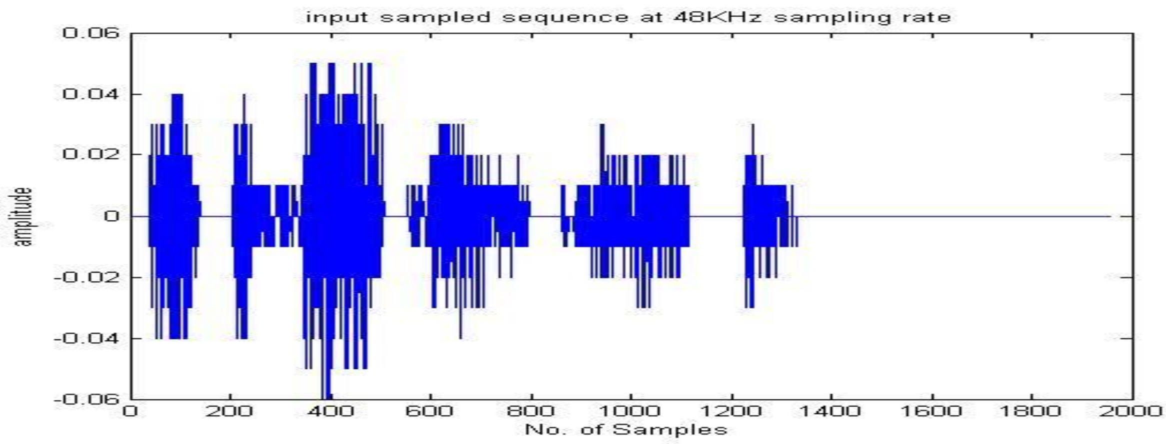

Fig.10. No. of samples v/s amplitude at 48 KHz

Table IV: Calculation of compression ratio & bit rate for audio signal 2

Input Signal	Sampled Input Signal(a)	Compressed Signal	Compression Ratio	Bit Rate (bits/Sample)

BWT method	30.59 KB	15.30 KB	50%	2.60 bps
BWT & Huffman Method	30.59 KB	29.67 KB	96.99%	5.05 bps
BWT & RLE Method	30.59 KB	10.26 KB	33.54 %	1.74 bps

- In this case we have considered audio signal 2 at 44.1 KHz sampling rate & sampled at 3KHz to compute compression ratio and bit rate of different audio compression methods as shown in Table V .As shown below Figure 11 shows input signal & Figure 12 shows input sampled sequence at 44.1 KHz sampling rate of audio signal 2.

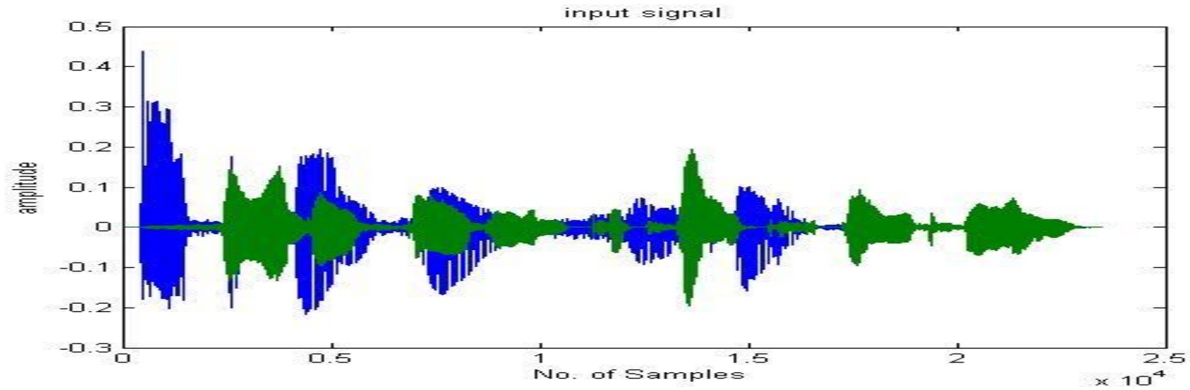

Fig.11. No. of samples v/s amplitude

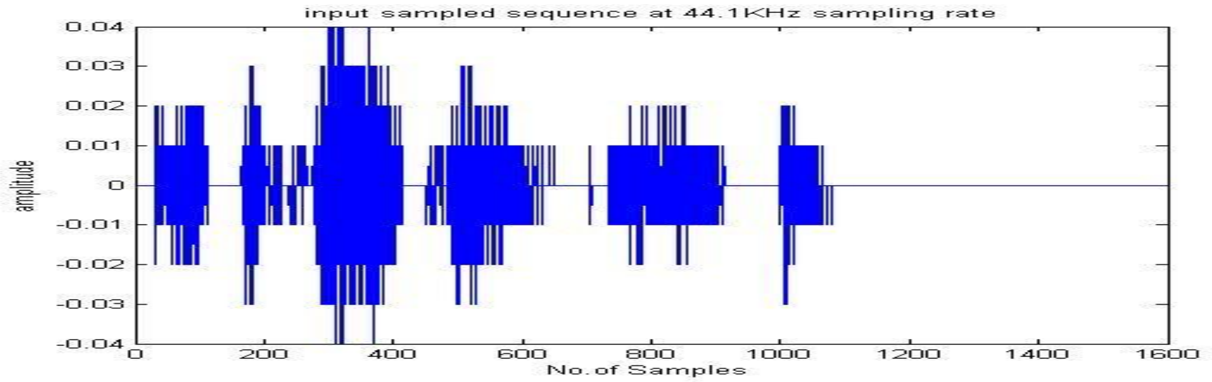

Fig.12. No. of samples v/s amplitude at 44.1 KHz

Table V: Calculation of compression ratio & bit rate for audio signal 2

Input Signal	Sampled Input Signal(a)	Compressed Signal	Compression Ratio	Bit Rate (bits/Sample)
BWT method	24.98KB	12.5KB	50%	2.12 bps
BWT & Huffman Method	24.98KB	22.85KB	91.47%	3.89 bps
BWT & RLE Method	24.98KB	6.81KB	27.26%	1.15bps

- In this case we have considered audio signal 2 at 32 KHz sampling rate & sampled at 2.6 KHz to compute compression ratio and bit rate of different audio compression methods as shown in Table VI.As shown below Figure 13 shows input signal & Figure 14 shows input sampled sequence at 32 KHz sampling rate of audio signal 2.

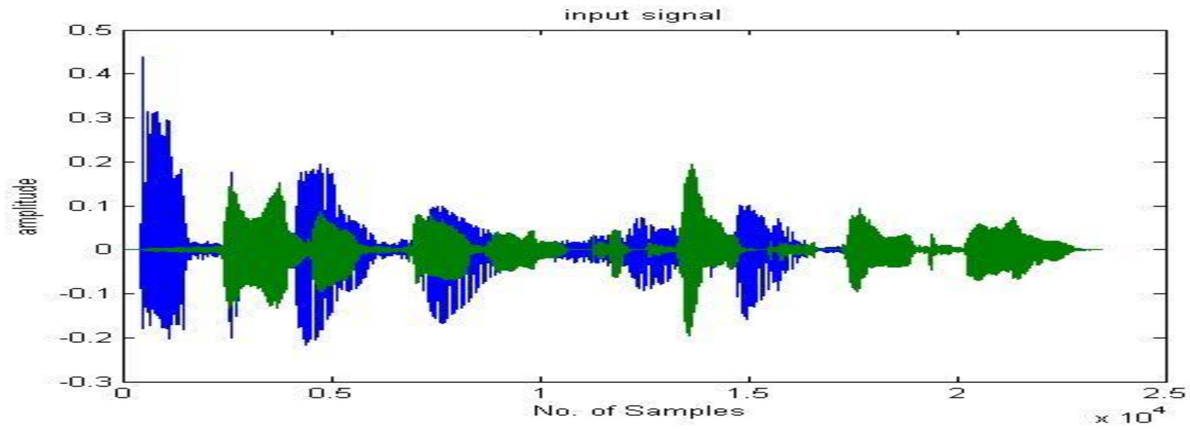

Fig.13. No. of samples v/s amplitude

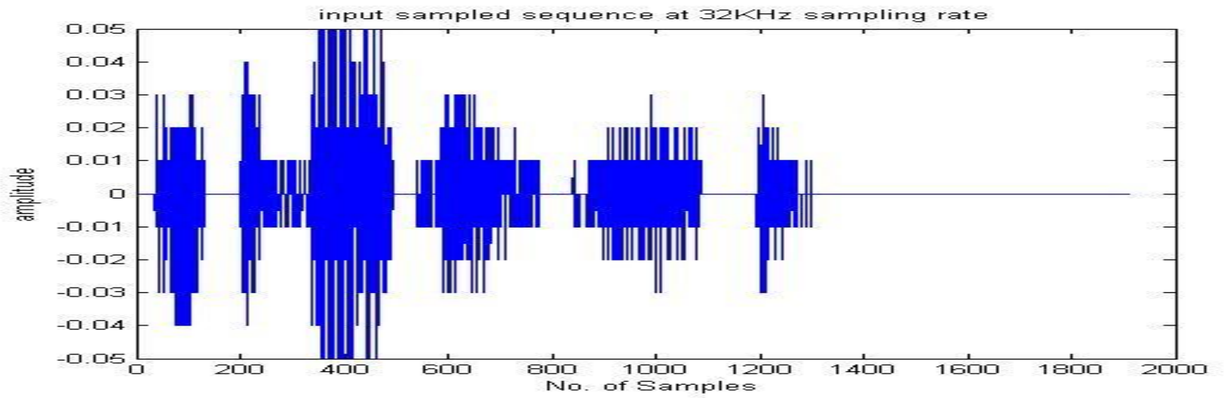

Fig.14. No. of samples v/s amplitude at 32 KHz

Table VI: Calculation of compression ratio & bit rate for audio signal 2

Input Signal	Sampled Input Signal(a)	Compressed Signal	Compression Ratio	Bit Rate (bits/Sample)
BWT method	29.82KB	14.92KB	50%	2.54bps
BWT& Huffman Method	29.82KB	28.67KB	96.14%	4.88bps
BWT & RLE Method	29.82KB	9.01KB	30.21%	1.53bps

- ***Performance Analysis**:* Figure 15 shows that the bit rate for three different sampling rates 32, 44.1, 48 KHz using proposed method. From this figure, we show that the bit rate depends on the sampling rate; the bit rate for the 44.1 KHz is less than the bit rates of 32 and 48 KHz, hence it is better than 32 and 48 KHz sampling rates.

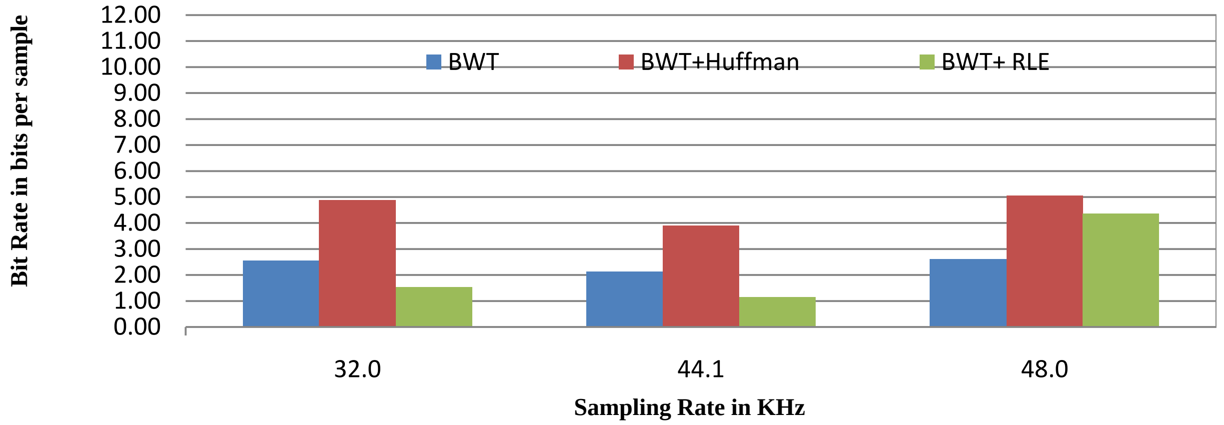

Fig.15.Sampling Rate v/s Bit Rate of audio signal 2

Figure16 shows the average bit rate for both audio signal with three different sampling rates 32, 44.1, and 48 KHz using the proposed method. From this figure, we show that the bit rate depends on the sampling rate; the bit rate of 32 KHz is less than 44.1 and 48 KHz, then it is better than 44.1 and 48KHz sampling rates.

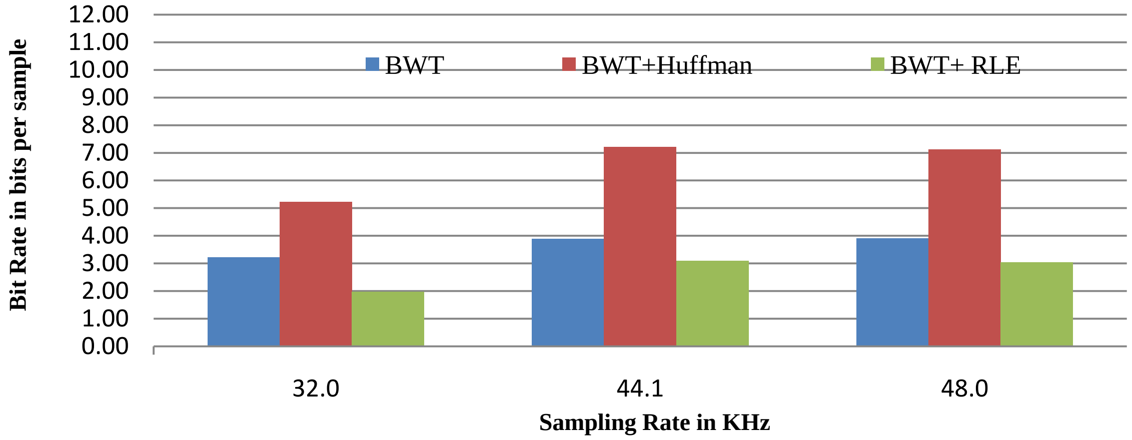

Fig. 16: Average bit rate for the proposed method for both the signal with different sampling rates

C. Comparison of all methods: In this section the proposed BWT & Run Length Encoding algorithm is compared with BWT method [3], BWT & Huffman Coding method[4] & BWT+MTF+RLE[1] method. Table VII compares bit rate values obtained from BWT algorithm[3] of scheme 1 & scheme 2 and BWT+MTF+RLE[1]algorithm with the proposed algorithm in this work.

- ***For Case Study 1:***

Table VII: Bit Rate comparison with BWT, MTF & RLE [1] & BWT+MTF [3]

S. No.	Methods	Sampling Rate	Average Bit Rate (bits per sample)	
			Existing	Proposed
1.	**BWT [3] Scheme1**	44.1 KHz	9.8bps	3.89bps
2.	**BWT [3] Scheme2**	44.1 KHz	9.1bps	7.22bps
3.	**BWT+MTF+RLE coding [1]**	44.1 KHz	9.02bps	3.10bps

Table VII shows that the result obtained from the proposed algorithm is more reasonable as compared to the existing algorithm [1].

- ***For Case Study2:***

Table VIII compares compression ratio obtained from proposed method and BWT+Huffman method[4].In our proposed method, BWT+RLE shows better compression ratio than existing methods[4].

Table VIII: Compression ratio comparison with BWT & Huffman method [4]

S. No.	Methods	Average Compression Ratio	
		Existing	**Proposed**
1	**BWT+HUFFMAN [4]**	40.3%	89.56%
2	**BWT+RLE**	40.3%	34.72%

Table VIII shows that the result obtained from the proposed algorithm is more reasonable as compared to the existing algorithms.

V. CONCLUSION

The goal of this proposed work is to achieve reduced bit rate and improved compression ratio, which is an essential step towards achieving better audio quality and hence algorithm is succesfully implemented. In this proposed work, we presented a lossless audio coding method using Burrows Wheeler Transform and Run length Encoding. And we have showed the dependence of the bit rate of lossless audio coding on the signal type, entropy coding methods using Huffman coding ,Run length encoding and sampling rate. Entire process is implemented in MATLAB and comparison result shows that lossless audio coding using BWT and RLE method outperforms the other lossless audio coding using BWT method [3], and combined BWT & Huffman coding [4] method, BWT+MTF & RLE methods[1] in terms of bit rate in TableVII & in terms of compression ratio in Table VIII.

REFERENCES

[1] Hend. A. Elsayed,"Burrows-Wheeler Transform and Combination of Move-to-Front Coding and Run Length Encoding for Lossless Audio Coding", 9th International Conference on Computer Engineering & Systems (ICCES), IEEE, pp.354 -359, Dec. 2014 .

[2] Chapter 6: A Word Level Context Based Speech Coding Scheme.

[3] H. A. Elsayed and M. Alghoniemy, "Lossless Audio Coding Using Burrows Wheeler Transform and Move-to-Front Coding," International Conference on Computer Engineering & Systems (ICCES), IEEE,pp.209-212 ,Nov. 2007.

[4] M.K.Borahi and V.K.Govindan, "A Simple Lossless Audio Data Compression Based On Burrows Wheeler Transform," International Conference on Electrical Engineering and Computer Science(ICEECS),pp. 68-73 ,April. 2013.

[5] I Made Agus Dwi Suarjaya ,"A New Algorithm for Data Compression Optimization" , International Journal of Advanced Computer Science and Applications (IJACSA), Vol. 3, No.8, pp.14-17,Sep.2012.

[6] M.P.Bhuyan, V.Deka, S.Bordoloi, "Burrows Wheeler Based Data Compression And Secure Transmission",International Journal of Research in Engineering and Technology(IJRET), Vol.2 , No.2 , pp.14-19,Dec.2013 .

[7] Rafeeq Mohammad and M. Vijaya Kumar, "Audio Compression using Multiple Transformation Techniques"International Journal of Computer Applications (IJCA) ,Vol. 86 , No. 13, pp.9-14,Jan. 2014 .

[8] Paul Brittan, "Evaluating Lossless Data Compression Algorithms for Use on Mobile Devices", Literature Synthesis ,2008.

[9] Dengfeng Zhang, Qing Liu+,Yuewan Wu, Yaping Li, Lin Xiao, "Compression and Indexing Based on BWT: A Survey", 10th Web Information System and Application Conference,IEEE,pp.61-64,Nov.2013.

[10] Asral Bahari Jambek and Nor Alina Khairi "Performance Comparison Of Huffman And Lempel-Ziv Welch Data Compression For Wireless Sensor Node Application", American Journal of Applied Sciences(AJAS), pp.119-126, Accepted on Dec.2014.

[11] Hari Narayan Mishra, Agya mishra, "Audio Enhancement Using Remez Exchange Algorithm with DWT",SSRG International Journal of Electronics and Communication Engineering (SSRG-IJECE),Vol.2,No.2,pp.60-65,Feb 2015.