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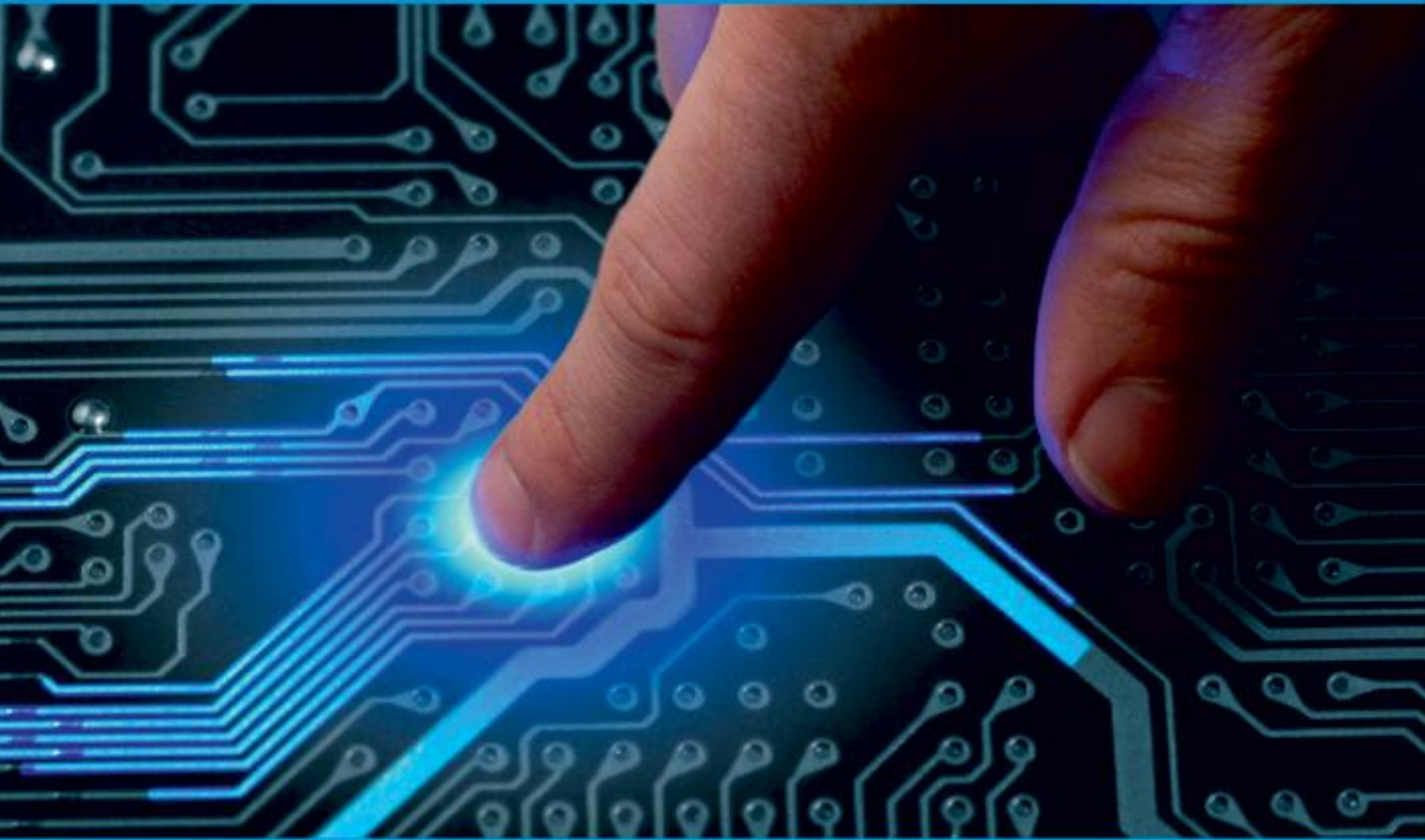

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Audio Compression Using Wavelet Transform

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ABSTRACT: Audio Compression is one of the basic technologies of the modern telecommunication age. Compression is the technique to convert high input data stream into smaller size. Audio coding gives us the digital form of audio with as few bits as possible and also maintains the quality. the reduction in bit rates conserve bandwidth .Audio coding is used in various applications such as digital broadcasting ,high quality audio for satellite transmission , internet audio or music database where the high quality audio signals bit rate is reduced without compromising the quality of the signal.

The technology proposed to achieve the design and implementation of audio compression using discrete wavelet transform technique. The efficiency performance of the audio encoding methods has been measured using compression ratio as well as peak signal to noise (PSNR) ratio, SNR.

1. INTRODUCTION

The revolution in computer history had invariably led to the demand of quality audio data. But the data rates associated with the uncompressed audio signal is massive. The product of the sampling rate and number of bits is known as the bit rate. The subtraction of the information rate and bit rate of the signal is known as redundancy. The audio compression technique works to reduce this redundancy without affecting the quality of the audio signal it has found many applications in various areas such as multimedia signal coding, high fidelity audio for radio broadcasting, audio transmission for HDTV, audio data transmission /sharing through internet etc.

Sheetal D. Gunjal, Rajeshee D. Raut, 2015. Traditional Psychoacoustic model and daubechies wavelets for enhanced speech coder performance described the dependencies of Compression ratio, SNR and the decomposition level it shows increase in the compression ratio value limited by the SNR value.[1] Oathman O.Khalifa, Sering Habib Harding and Aisha-Hassan described the method for audio compression the signal is compressed using wavelet and Reconstructed signals are compared using factors like Signal to Noise Ratio (SNR), Peak Signal to Noise Ratio (PSNR), Normalized Root Mean Square Error (NRMSE), Compression Ratio for different levels of wavelet .[6] P. Srinivasan and L. H. Jamieson. "High Quality Audio Compression Using an Adaptive Wavelet Packet Decomposition and Psychoacoustic Modelling described Wavelet packet-based compression scheme suitable for high-quality audio transfers over the internet or storage.[5]

To reduce the coded bit rate there are basically two types of techniques used in the first type some form of digital encoding is applied on the basis of statistical redundancy it is a lossless audio coding in which the original audio signal is remains unharmed and can be totally recovered. In the second type some sort of signal processing is used so that sorting of unwanted and information signal components can be done the recovered audio signal and the original signal is not identical therefore we can say this technique is lossy technique.

1.1 Types of Compression

a) Lossless compression:-
As their name implies, involve no loss of information. if data have been losslessly compressed, the original data can be recovered exactly from the compressed data lossless compression is generally used for applications that cannot tolerate any difference between the original and reconstructed data.

b) Lossy compression:-
Lossy compression techniques involve some loss of information, and data that have been compressed using lossy techniques generally cannot be recovered or reconstructed exactly. In return for accepting this distortion in the reconstruction, we can generally obtain much higher compression ratios than is possible with lossless compression.

II.TECHNIQUES FOR AUDIO COMPRESSION

Audio compression is classified into three methods
A. Waveform coding
B. Parametric coding
C. Transform coding
  i) Fast Fourier Transform (FFT)
  ii) Discrete Cosine Transform (DCT)
  iii) Continuous Wavelet transform(CWT)
  iv) Discrete Wavelet transform (DWT)

2.1 **Introduction to wavelet**:

The fundamental idea behind wavelets is to analyse according to scale. The wavelet analysis procedure is to adopt a wavelet prototype function called an analysing wavelet or mother wavelet. Any signal can then be represented by translated and scaled versions of the mother wavelet. Wavelet analysis is capable of revealing aspects of data that other signal analysis techniques such as Fourier analysis miss, aspects like trends, breakdown points, discontinuities in higher derivatives, and self-similarity. Furthermore, because it affords a different view of data than those presented by traditional techniques, it can compress or de-noise a signal without appreciable degradation. The different types of wavelet families are like Daubechies, haar, symmlet etc.

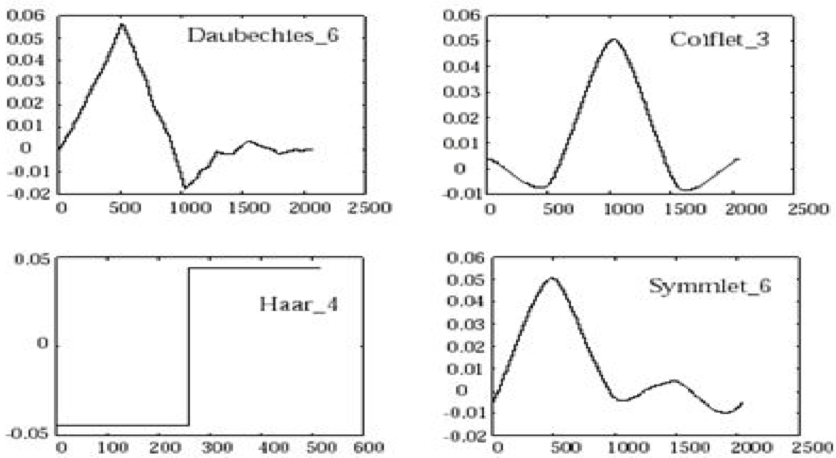

Figure 1: Different wavelet families

2.2 Audio compression using wavelet:

Transform means converting time domain signal into the frequency domain. Transform coding is the type of data compression for natural data like audio signal. In this, the knowledge of the application is used to choose the information to discard, thereby lowering its bandwidth.

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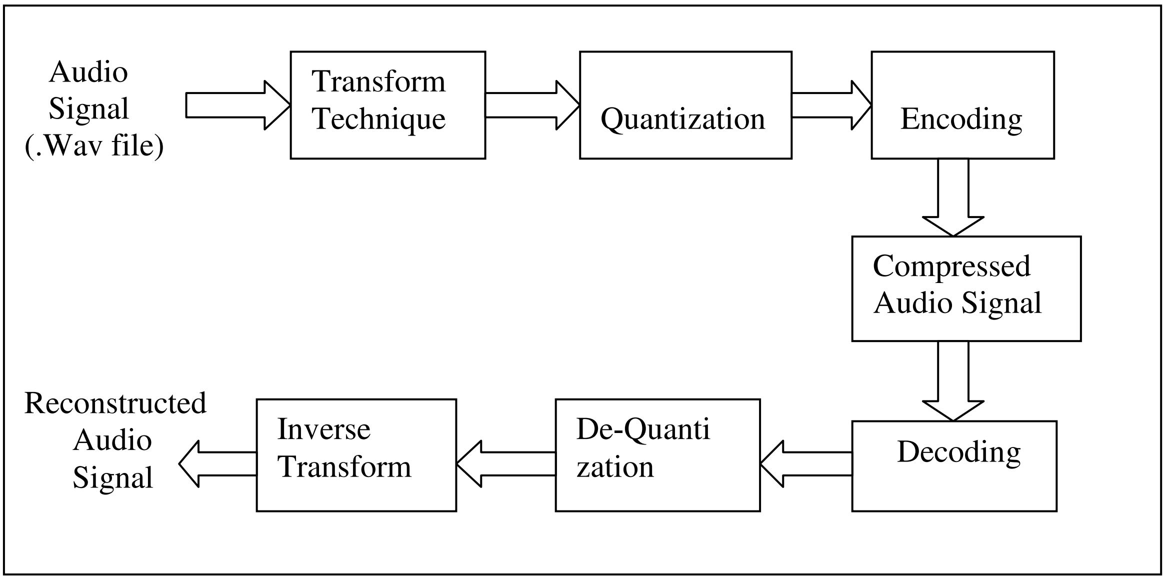

A) Transform Technique :
Discrete wavelet transform is used for the audio signal compression. Wavelet transform is very suitable for audio compression. DWT uses multiresolution technique to analyze different frequencies.

B) Quantization and coding of transform coefficients :
If the amount of information conveyed by each coefficient is different, it makes sense to assign differing numbers of bits to the different coefficients. The Process of quantization is to convert the discrete time continuous amplitude into discrete time discrete amplitude. This is done by rounding off each sample to the nearest quantization level. Each discrete time, discrete amplitude is further represented by finite number of digits using a coder.

C) Encoding :
In Encoding techniques reduce the number of coefficients by removing the redundant data. The compressed audio signal can be reconstructed to form the original signal by
Decoding followed by de-quantization and then performing inverse transform Methods.

III. RESULTS

3.1 Compression Ratio:
The compression ratio (CR) is defined as the ratio of the length of original signal to the length of the compressed signal.

$$\text{Compression Ratio} = \frac{\text{Length of original signal}}{\text{Length of compressed signal}}$$

3.2 PSNR (Packet Signal to Noise Ratio):

It is Defined as the ratio between the maximum possible power of a signal and the power of corrupting noise that affects the fidelity of its representation. Because many signals have a very wide dynamic range, PSNR is usually expressed in terms of the logarithmic decibel scale.

$$PSNR = 10 \log_{10} \frac{NX^2}{\| x - x' \|^2}$$

3.3 SNR (Signal to Noise Ratio):

$$SNR = 10\log\left[\frac{\sigma x^2}{\sigma e^2}\right]^2$$

σx^2 is the mean square of the speech signal and σe^2 is the mean square difference between the original and reconstructed signals.

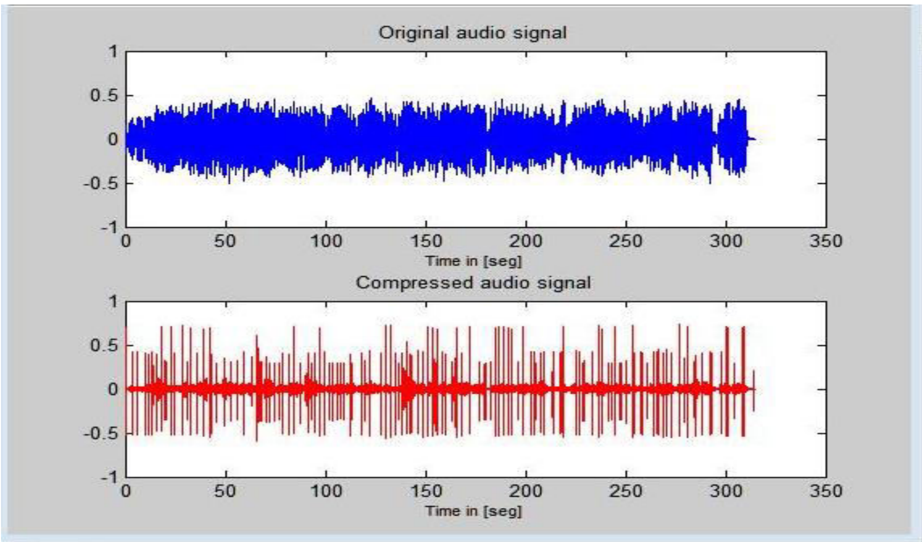

Figure 3: Original & Compressed audio signal

Signal	Compression Ratio	SNR (dB)	PSNR (dB)
Sample1.wav	1.7191	1.0327	177.887

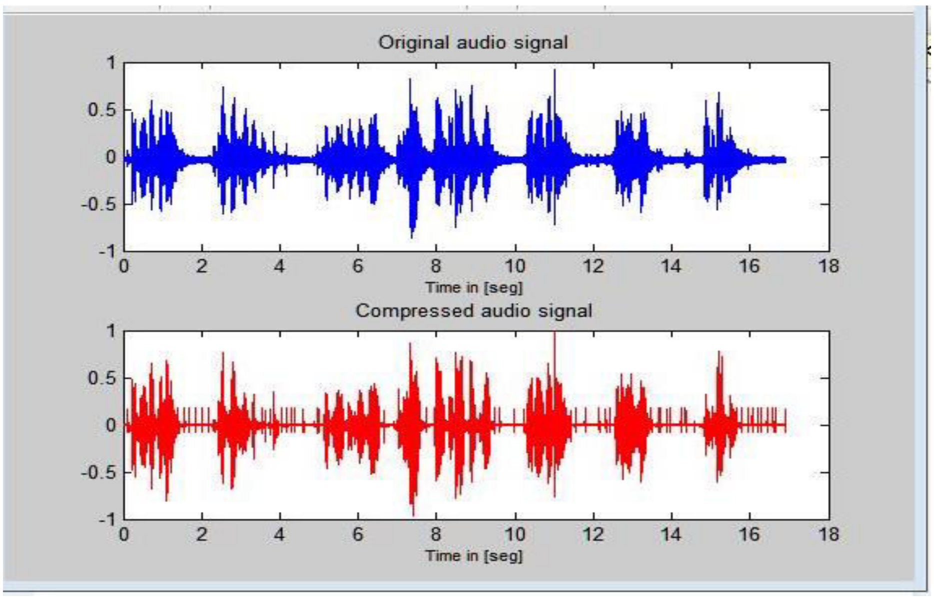

Figure 4: Original & Compressed audio signal

Signal	Compression Ratio	SNR (dB)	PSNR (dB)
Sample2.wav	1.9998	6.1106	122.322

IV. CONCLUSION

Wavelet transform based audio compression scheme presented in this paper. It is implemented using MATLAB. The selection of the daubechies wavelet with DWT yielded comparable improvement in the performance parameters with a good quality reconstruction of audio signal. Results show that in general there is improved in compression factor & signal to noise ratio with DWT based technique. It is also observed that Specific wavelets have varying effects on the Audio signal being represented.

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