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Audio Wavelet Compression and Audio Steganography using LSB Technique

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Abstract— *Audio Compression is a type of data compression designed to minimize the need for the transmission bandwidth of digital audio streams and the size of the audio. An effective Discrete Wavelet Transform (DWT) is used at various scales for the decomposition of the original signal into wavelet coefficients. To secure the processed audio signal, Audio Steganography is used. Steganography is the art of secret hiding and its study. The origin plain text can be obscured in one way or other. In this paper, a perceptual audio compressor is developed and it is secured using LSB Technique.*

Keywords—*Phychoacoustic Model, Steganography, Wavelet,Wavelet Transform.*

I. Introduction

Audio is a tool for receiving knowledge from one individual to other bandwidth restricted to 4Khz.Growing Demands for digital contact, efficient usage of the Bandwidth, Storage space and power. To overcome the consequences, we need to compress data and get a novice look in this paper the voice signal is processed and compressed. PCM used to be the simple audio technique used in the earlier times for encoding but has no mechanism in removing redundancy. Researchers in the last few have proposed multiple powerful algorithms based on the Fourier Transform (FT), Wavelet transform (WT), and Discrete Cosine Transform (DCT). Fast Fourier transformation (FFT) became popular. In the earlier days compression algorithm is based on assumption that the signal is stationary. Most of the signals present in the world are non-stationary. For this approach, the use of the wavelets has been revolutionized in signal processing which is used to process the linear and non-stationary signals. Audio is a random non-stationary process due to varying nature of time and sudden change in frequencies of human speech. Audio Compression, which is very high necessary for the present world as a result of advancement in Multimedia Technology as bandwidth is limited.

Rapid increase in the transmission of digital data over internet made researchers for betterment in the security system. There is various alternatives are presented like watermarking, cryptography. But Steganography has gained much important in this context. Steganography is the art of secret hiding and its study. The origin plain text can be obscured in one way or other. It adds two layer protections against cryptography because cryptography only changes the message form but its existence is not hidden, while steganography even hides its presence. The outline of the paper is as follows: section II covers why wavelets?, Section III covers about the wavelets. Section IV covers audio compressing using DWT. Section V covers quality parameters. Section VI covers steganography. Section VII covers LSB encoding. Section VIII includes algorithm. Section XI covers results. Section X covers conclusion. Section XI covers references.

II. Why Wavelets?

Real World data or signals frequently exhibit slowly changing trends or oscillations punctuated with transients. On the other hand audio signals have sharp edges and abrupt changes. These abrupt changes are often the most interesting parts of the data. Both perceptually and the information they provide. The Fourier transform is a powerful tool for data analysis. However it doesn't represent abrupt changes efficiently. The reason for that is that the Fourier transform represents data as a sum of sine waves which are well localized in time or space. These sine waves oscillate forever. Therefore, to accurately analyse signals that have abrupt changes, we need to use a new class of functions that are well localized in time and frequency, which is nothing but the Wavelets.

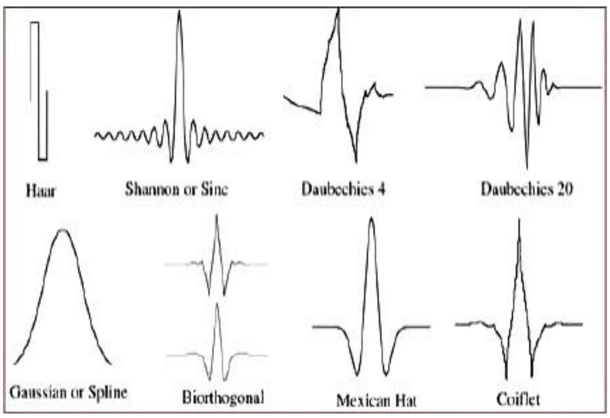

Fig. (III.A). Wavelet family

III. Wavelets

A wavelet is a rapidly decaying wave like oscillation that has zero mean. Unlike sinusoids which extend to infinity a

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wavelet exists for a finite duration. Wavelets come under difference size and shape. The different types of wavelets are: -Daubechies, Morlet, Coiflets, Haar, Biorthogonal etc. The availability of a wide range of wavelets is a key strength of wavelet analysis. To choose the right wavelet you will need to consider the application you will use it for.Wavelet analysis is breaking up of an original signal into different scaled and shifted versions called the mother wavelet.

1. Discrete Wavelet Transform

DWT is the crucial tool for the analysis of audio signal which is non-stationary. DWT consists of coefficients of expansion of main signal with reference to basis ψ j, k(t) of which is a dilated version of mother wavelet.

The expression for DWT is:-

$$\psi_{j,k}(t) = \frac{1}{s_o^j}\psi\left(\frac{t-k\tau_o s_o^j}{s_o^j}\right), j, k \in Z$$

It is the decomposition of signal into approximation and detailed coefficients using series of filter banks consisting of low pass and high pass filters .The low pass output gives approximation coefficients and high pass gives detailed coefficients. This decomposition method is called efficient algorithm method.

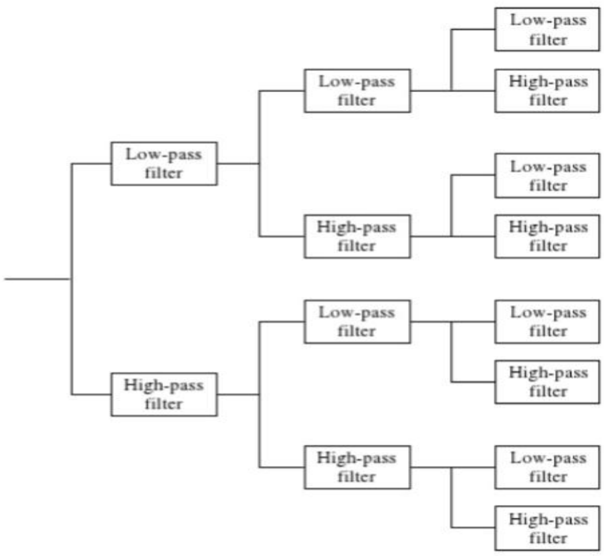

Fig. (III.1.A). Bank of Filters

2. Wavelet Transform Concepts

The two main properties of wavelet transform are:-

1. Scaling
2. Shifting

Scaling refers to the process of stretching or shrinking the signal in time which can be expressed by

Scaling: -$\Psi(\frac{t}{s})$ s>0

A stretched wavelet helps in capturing the slowly varying changes in the signal, while the compressed helps in capturing the abrupt changes.

Shifting means delaying or advancing the onset along the length of the signal.

Shifting: -ϕ(t-s)

IV. Audio Compression Using Discrete Wavelet Transform

1. Choosing the Type of Wavelets

The primary importance in the application is to choose the correct mother wavelet. It has to be well localized in time and frequency and has more number of vanishing moments for the compression. The key goal of the paper is to minimize the recovered error rate and optimize SNR

We know that Daubechies wavelets concentrate more than 90% of signal energy in Level 1 Approximation.

2. Wavelet Decomposition

Wavelet Decomposition is decomposition of signal into different frequency bands and it is performed using appropriate wavelet function with level of decomposition.

Wavelets decompose using suitable bank of filters the output of low pass filters is “cA” which is Approximation coefficient and the output of high pass is “cD” which is detailed coefficient. The decomposition level depends on the type of signal used. Usually level 5 are recommended.

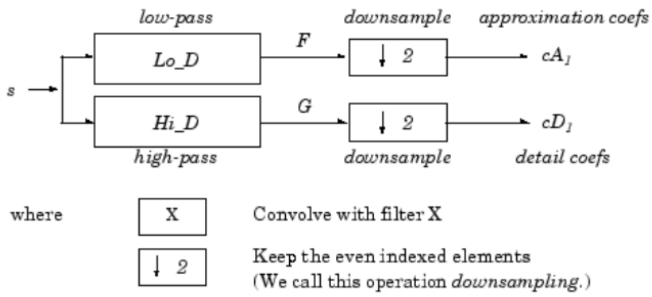

Fig. (IV.2.A). Wavelet Decomposition

3. Filtering

Frequency is the key parameter to characterize the signal.

For most of the signals the low frequency content is the most important part which contains the identity of the signal, whereas high frequency gives details of the signal.

The original signal consider S passes the bank of filters emerges as two signals say A and D. If the original signal has 2500 samples the output emerges into 5000 samples. To overcome this problem we perform down sampling to produce “cA and “cD”. The cD contains high frequency noise and cA contains less noise than the original signal S.

In the Matlab we use “wavedec” function for the wavelet Decomposition.

4. Thresholding

After performing the wavelet transform, compression involves thresholding. From the experiments we came to know that most of the energy is concentrated in high valued coefficients which are limited. So we can truncate the low valued coefficient. The different methods of thresholding are:-

Global Thresholding: -It involves only including the high valued coefficients of the signal and it can be done in the Matlab manually by setting the global threshold.

Level Thresholding:-This includes the level of decomposition by setting the level on each decomposition.

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5. Encoding

Encoding the zero valued coefficients. Encoding is method to remove data that are repetitively occurring. It removes the redundant data.

6. Psychoacoustic Model

Human hearing and voice Frequency range is about 20 Hz to 20 kHz, most sensitive at 1 to 5 KHz. Dynamic range (quietest to loudest) is about 96 dB .Normal voice range is about 500 Hz to 2 kHz ï Low frequencies are vowels and bass High frequencies are consonants.

The two major psychoacoustic techniques are the absolute threshold of hearing effect and the masking effect. Absolute hearing level characterizes how much energy required in a pure tone, so that the listener can sense it in a noiseless medium. Absolute threshold is expressed in sound level db. (dB SPL). Auditory masking relies on both the masking signal and the masked signal times and frequencies. Here masking is presumed to be additive, so that the total masked power is measured at any frequency.

V. Quality Parameters

There are some parameters which evaluate the quality of compressed signal:

A. Compression Ratio:-Compression Ratio is the ratio of length of original signal to the length of the compressed signal. It is denoted as CR.

$$\mathrm{CR} = \frac{\text{length of x(n)}}{\text{length of y(n)}}$$

B. Mean Square Error (MSE):-It is the cummulative squared error between original and compressed.

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

The lower the value of MSE, the lower the error.

C. Peak Signal to Noise Ratio (PSNR):-It is the ratio of maximum energy of the signal to the MSE. It is represented in logarithmic.

$$\mathrm{PSNR}=10*\log_{10}\left(\frac{R^2}{MSE}\right)$$

Where R is maximum possible energy.

VI. Steganography

Steganography is the art of hiding the text, image or audio file. In general, the hidden messages appear to be something else like article or cover message .For example, the hidden message can be invisible ink between visible lines of a private letter. The benefit of steganography over cryptography is that the expected hidden message as subject of inspection does not draw attention to itself. Hidden message is encoded in digitized audio signal in audio steganography which results in a slight alteration of binary sequence of the corresponding audio signal. This is easily achieved using LSB Technique.

VII. LSB Encoding

One of the earliest techniques studied in hiding information about digital audio (and other types of media) is the least significant bit coding technique. In this Binary LSB Technique sequence of each digitized audio file sample shall be replaced by binary Secret Equivalent note. LSB concealment is simple and fast method to insert information into an audio signal.

The length of the secret message should be smaller than that of total number of samples of audio file. It is the easiest way to embed information into audio file. It can conceal large amount of data with sampling rates nearly from 8kbps to 44kbps including all samples and including low computational complexity. The obvious disadvantage is that the robustness is considerably low, due to the fact that simple random changes to the LSBs destroy the coded watermark.

VIII. Algorithm

The LSB encoding is explained:-

Encoding:-

1. Consider the audio file in two channels named ch1 and ch2 using “wavwrite” function in Matlab.
2. Now, both the channels must be ANDed with 240
3. Replace the first four bits of the text with the LSB of the new ch1 (after AND operation) and the next four bits with the LSB of the new channel2
4. Perform the bit ANDing of the secret text which must be embedded with the powers of 2, if the result is equal to the ANDing value then OR operation with the LSB of power of 2.

Decoding:-

1. Consider the updated values of ch1 and ch2.
2. The reverse process is done, instead of AND we perform OR and vice versa.
3. Finally we get the secret text.

IX. Results

1. Audio Compression

The following are the experimental results of the different sizes of audio files (10Mb, 2Mb) performed in Matlab. The Fig (IX.1.A) and Fig (IX.1.B) shows the GUI with original audio file waveform, and compressed audio file waveform with performance factors. The value of the MSE is low indicating the reduction of the error. The size of the file is reduced, maintaining the quality of the signal. This is achieved by using the Matlab functions.

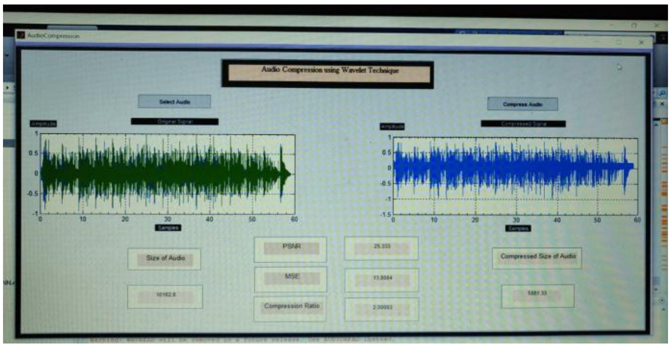

Fig. (IX.1.A). Original and compressed signal of (10MB)

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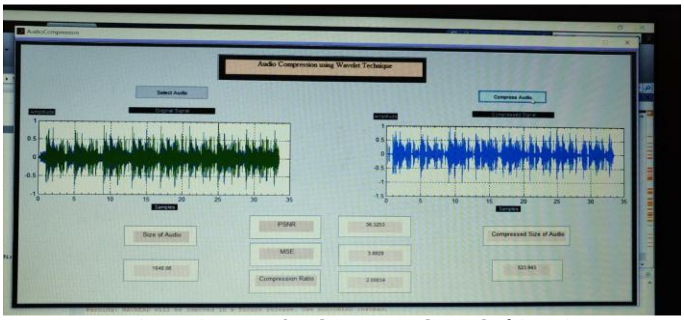

Fig. (IX.1.B). Original and compressed signal of (2MB)

The table represents the size of the audio file before and after compression, PSNR, MSE:

TABLE 1. Results of Audio Compression

Sl. no.	Size		CR	MSE	PSNR
	Before Compression	After Compression			
1	1048.06Kb	523.943Kb	2.003	3.8929	36.325
2.	2095.67Kb	1047.34Kb	2.017	6.7487	31.546
3.	10162.8Kb	5081.33Kb	2.03	13.800	25.353

2. Audio Steganography

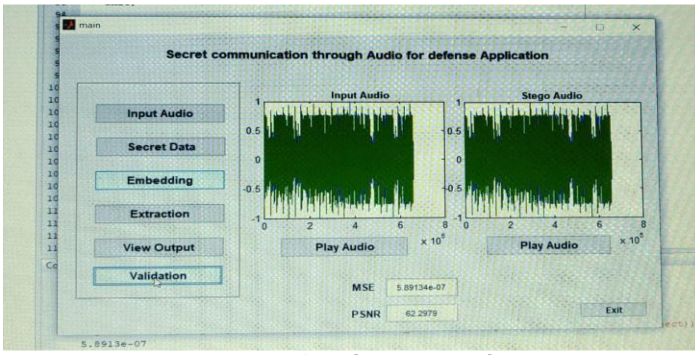

Fig. (IX.2.A). Audio Steganography

The Fig (IX.2.A) represents the GUI of Audio Steganography

X. Conclusion

Audio compression has become significant factor since there is an increase in multimedia communication. DWT performs well in the case of recorded signals and in real time. In the case of wavelet approach the CR can be varied whereas it is fixed in other techniques.

We have presented a framework for steganography using LSB Technique for audio files. The message signal is received with utmost protection and can be retrieved without any loss of transmission by this process.

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