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An Improved Audio Steganography Based On DWT and Direct Sequence Spread Spectrum

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ABSTRACT

In the proposed work we have presented an enhancement of the Audio Steganographic system using Wavelet approach to provide a means of secure communication. A Discrete Wavelet Transform has been applied to the Audio file during embedment of the Direct Sequence message into the Audio file. In our proposed approach, the message bits are embedded adaptively into the Audio instead of sequentially. The technique embeds the hidden information in the transformation domain of the Audio and uses simple arithmetic equations. Besides, the embedded confidential information can be extracted from stego-Audios without the assistance of original Audio data. The information to be embedded must first modulated using the pseudo-noise. This paper discusses implementation of the method in audio data to hide text message. Spread Spectrum method is known to be very robust, but as a consequence the cost is very large, the implementation is relatively complex, and the information capacity is very limited.

INTRODUCTION

The objective of steganography is to hide a secret message within a cover-media in such a way that others cannot discern the presence of the hidden message [1].Technically in simple words "steganography means hiding one piece of data within another". Modern steganography uses the opportunity of hiding information into digital multimedia files and also at the network packet level. Hiding information into a media requires following elements. The cover media(C) that will hold the hidden data

- The secret message (M), may be plain text, cipher text or any type of data
- The stego function (Fe) and its inverse (Fe-1)
- An optional stego-key (K) or password may be used to hide and unhide the message [3].

Steganography is the art or study of hiding information by inserting secret messages in other messages. Medium where information is inserted can be anything. This medium is called the cover object. Steganography that is applied to hide information on the cover of digital objects is called Digital Steganography [3]. Cover objects that are used in digital steganography can vary, for example in the image archive. Steganography algorithms in the image archive have been widely developed. Meanwhile, steganography algorithms in audio archive are relatively few. This paper discusses the application of digital steganography on audio archives using the method of Direct-sequence Spread Spectrum [1]. and Discrete Wavelet Transform techniques. Steganography in the audio archive is not as easy as in the image archive. Unlike the archives of raw images, raw sound files are usually larger. In comparison, the raw image file type and resolution of 1280x800 24 bit color (standard resolution of desktop screen) has a size of about 3 MB of data. While the raw audio files with 44.1 kHz sampling frequency, 16 bit stereo channels with 4 minutes duration (the standard duration of song) has a size of about 40 MB of data. The difference is quite large, resulting in the implementation of steganography in audio data becomes more difficult [4]. As an illustration, suppose we use the Discrete Fourier Transform to convert the data domain, then the audio archives clearly require substantial cost because

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the number of samples that must be transformed is much greater. Moreover, suppose we use the LSB method [2] and [3] the noise generated at the sound archive is greater. This is because the range of the sound signal is lower than the pixel signal. Pixel is encoded by 24 bits, while sound signal is encoded by 15 bits (because there are positive and negative sound signal). In addition, the use of raw audio files (WAV) is less frequent than the raw image files (BMP), because the size in audio files is too large. Therefore, we need such a scheme that enables us to preserve the hidden messages [5], even if the audio files are compressed. In the next section, the author will explain some basic theories that need to be known in advance. In our proposed algorithm we are using Discrete Wavelet transform to convert the audio data domain in to frequency domain of four sub bands (LL, LH.HL and HH). In this four sub bands LL referred as Approximation Coefficients where as remaining three sub bands referred as detail coefficients [5]. In the Section II we will discuss Audio Steganography Scheme Using Direct Sequence Spread Spectrum. The proposed algorithm Discrete Wavelet transforms will be discussed in Section III. Experimental results and Conclusion will be discussed in section IV and V respectively.

EXISTING METHOD

In this section, we discussed Fast Fourier Transform (FFT) based steganography scheme proposed in [6]. And explained the reasons why the author has used FFT instead of Fourier and short time Fourier transforms.

A.Fourier Transform

The signal can be analyzed more effectively in frequency domain than the time domain, because the characteristics of a signal will be more in frequency domain. One possible way to convert or transform the signal from time to frequency domain is Fourier transform (FT). FT is an approach which breaks down the signal into different frequencies of sinusoids and it is defined as a mathematical approach for transforming the signal from time domain to frequency domain.

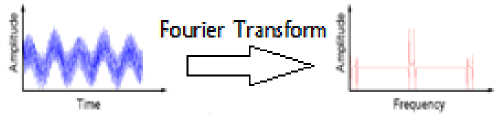

Fig.1. Analysis of FT with an example

FT has a drawback that it will work out for only stationary signals, which will not vary with the time period. Because, the FT applied for the entire signal but not segments of a signal, if we consider non-stationary signal the signal will vary with the time period, which could not be transformed by FT. and one more drawback that we have with the FT is we cannot say that at what time the particular event will has occurred.

B.Short-Time Fourier Analysis

To correct the deficiency in FT, Dennis Gabor in 1946 introduced a new technique called windowing, which can be applied to the signal to analyze a small section of a signal. This adaptation has been called as the Short-Time Fourier Transform (STFT), in which the signal will be mapped into time and frequency information.

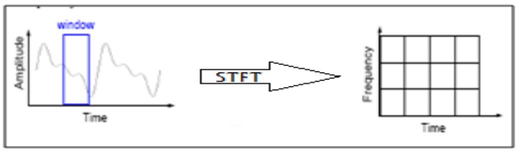

Fig.2. STFT analysis of a signal

In STFT, the window is fixed. So, we this window will not change with the time period of the signal i.e., for both narrow resolution and wide resolution. And we cannot predict the frequency content at each time interval section.

C. Fast Fourier Transform

A Fast Fourier Transform (FFT) refers to a special class of algorithm to compute the Discrete Fourier Transform (DFT) and inverse DFT of a sequence. Fourier analysis converts a signal from its original domain (often time or space) to the frequency domain and vice versa. FFT algorithm is an important use for this DSSS implementation. Fast Fourier transforms are widely used for many applications in engineering, science, and mathematics. The basic ideas were

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popularized in 1965, but some algorithms had been derived as early as 1805.In 1994 Gilbert Strang described the Fast Fourier transform as "the most important numerical algorithm of our lifetime" and it was included in Top 10 Algorithms of 20th Century by the IEEE journal Computing in Science & Engineering.

PROPOSED SCHEME

3.1. Wavelet Analysis

To overcome the drawbacks of STFT, a wavelet technique has been introduced with variable window size. Wavelet analysis allows the use of long time intervals where we want more precise low-frequency information, and shorter regions where we want high-frequency information.

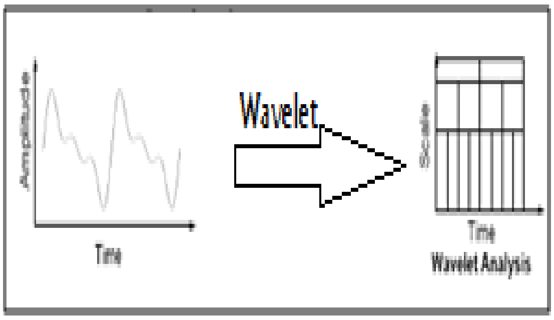

Fig.3. Wavelet analysis with an example

In fig.4 it is shown that the comparison of FT, STFT and wavelet transform by considering an example input signal and how the analysis of transformation techniques will apply to get the frequency information of input signal. We can observe that in wavelet analysis the graphical representation shows that the wavelet has more number of features than the FT and STFT. Wavelet is also called as multi resolution analysis (MRA). Here's what this looks like in contrast with the time-based, frequency-based, and STFT views of a signal:

In this section, the steganography scheme will be explained. Cover object used is the raw audio files like WAV files. Suppose we have a byte-sequence information that will be inserted into the cover object, the byte-sequence will be converted into a bit-sequence information.

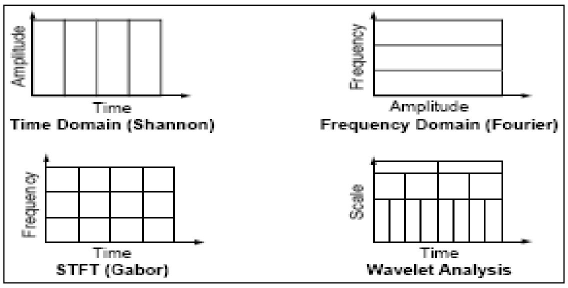

Fig.4. Comparison of FT, STFT and Wavelet analysis of a signal

Then we represent these bits into such signal so that if the bit is 1 then the amplitude of the signal is 1, whereas if the bit is 0 then the amplitude of the signal is -1. As shown as follows:

$$A = \{a_i | a_i \in \{-1,1\}\} \qquad (1)$$

Next, open the WAV files and obtain the signal's amplitude data. The amplitude is represented as 16 bit signed integer value with a range of 2^{15}-1 to -2^{15}+1. So, for divide this amplitude with a value of 215-1 in order to make the range of values obtained between 1 to -1. Then, transform the data into frequency domain using FFT. Create a long and random PN sequence with a value of 1 or -1. If the PN sequence has a chip rate cr, and if the information signal has a total of n signals, then the PN sequences that must be generated is cr x n. We call the PN sequence P, then

$$P = \{p_i | p_i \in \{-1,1\}\} \qquad (2)$$

Modulate each information signal with the PN sequence until cr times, by multiplying the value. It will produce a signal B which is the distributed signal of A and of course with length cr times its original length. Initially, spread the information in A to B as follows:

$$B = \{b_i | b_i = a_i, j \cdot cr \ \leq i < \ (j+i) \cdot cr\} \qquad (3)$$

Next, modulate B and P and multiply it by a factor α. This message will be inserted into the cover-object. Suppose w is a message that will be inserted, v is the cover-object and v' is a cover-object containing the message. So this process can be formulated as follows:

$$w_i = \alpha \cdot \ b_i \cdot p_i \qquad (4)$$

$$v'_i = \ v_i + w_i \qquad (5)$$

This scheme will generate noise. If the factor of the amplifier is too large, the noise is also large and may damage the cover-object. So the strength factor and

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chip-rate must be carefully chosen. Extraction scheme will be described next. Because of the effect of PN sequence generated previously, the signal added will be random. In order for the information to be retrieved, the receiver must generate the same PN sequence. Multiply the PN sequence signal corresponding with each cover object signal, the relationship can be shown as follows:

$$\sum_{i=j\cdot cr}^{(j+1)\cdot cr-1} p_i v_i' = \sum_{i=j\cdot cr}^{(j+1)\cdot cr-1} p_i v_i + \sum_{i=j\cdot cr}^{(j+1)\cdot cr-1} \alpha b_i p_i^2$$

If we look at the following terms:

$$\sum_{i=j\cdot cr}^{(j+1)\cdot cr-1} p_i v_i$$

The value of these terms will be close to 0 for a large number of samples (large chip rate). This is because the random value of PN sequence causes the sum of the signal approaching 0 or a certain threshold value.

While the second term:

$$\sum_{i=j\cdot cr}^{(j+1)\cdot cr-1} \alpha b_i p_i^2$$

The second term has interesting properties. Because the PN sequence has value 1 or -1, then the result of p_i^2 is 1.

Thus, the term can be simplified into:

$$\sum_{i=j\cdot cr}^{(j+1)\cdot cr-1} \alpha b_i$$

Because we have defined Bi has a value 1 or -1, then we simply conclude that if the term exceed the value of zero, we assume that the information retrieved is 1 and if the value is less than zero, we assume that the information retrieved is 0. This is the reason we choose the domain of B and P. From the previous explanation, we can conclude that the value of $\propto b_i$ must exceed a certain threshold value in order for a clear information retrieval.

3.2 Discrete Wavelet Transform (DWT)

Although the discretized continuous wavelet transform enables the computation of the continuous wavelet transform by computers, it is not a true discrete transform. As a matter of fact, the wavelet series is simply a sampled version of the CWT, and the information it provides is highly redundant as far as the reconstruction of the signal is concerned. This redundancy, on the other hand, requires a significant amount of computation time and resources. The discrete wavelet transform (DWT), on the other hand, provides sufficient information both for analysis and synthesis of the original signal, with a significant reduction in the computation time. The DWT is considerably easier to implement when compared to the CWT. The basic concepts of the DWT will be introduced in this section along with its properties and the algorithms used to compute it. As in the previous chapters, examples are provided to aid in the interpretation of the DWT.

We now look how the DWT is actually computed: The DWT analyzes the signal at different frequency bands with different resolutions by decomposing the signal into a coarse approximation and detail information. DWT employs two sets of functions, called scaling functions and wavelet functions, which are associated with low pass and high pass filters, respectively. The decomposition of the signal into different frequency bands is simply obtained by successive high pass and low pass filtering of the time domain signal. The original signal x[n] is first passed through a half band high pass filter g[n] and a low pass filter h[n]. After the filtering, half of the samples can be eliminated according to the Nyquist's rule, since the signal now has a highest frequency of π /2 radians instead of π. The signal can therefore be sub sampled by 2, simply by discarding every other sample. This decomposition halves the time resolution since only half the number of samples now characterizes the entire signal. However, this operation doubles the frequency resolution, since the frequency band of the signal now spans only half the previous frequency band, effectively reducing the uncertainty in the frequency by half. The above procedure, which is also known as the sub and coding, can be repeated for further decomposition. At every level, the filtering and sub sampling will result in half the number of samples (and hence half the time resolution) and half the frequency

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band spanned (and hence doubles the frequency resolution). Figure 4.1 illustrates this procedure, where x[n] is the original signal to be decomposed, h[n] and g[n] is low pass and high pass filters, respectively. The bandwidth of the signal at every level is marked on the figure as "f".

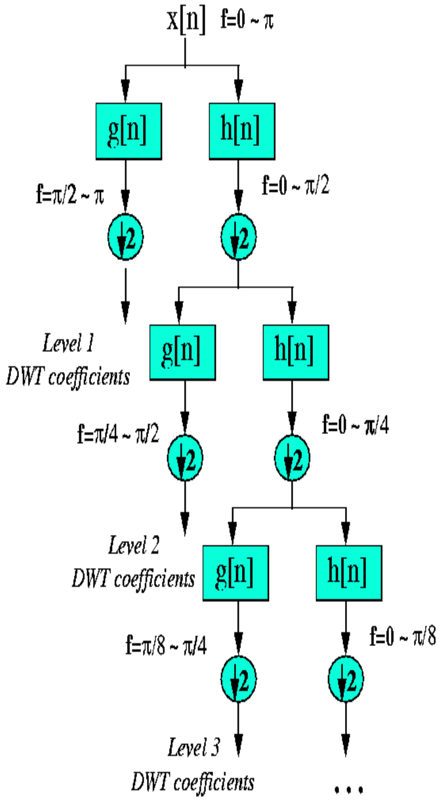

Fig.5 Wavelet decomposition tree

EXPERIMENTAL RESULTS

Simulation results have been done in MATLAB 2011a. We tested the proposed and existing methods for various audio samples with different cr values. In fig.6 FFT based audio steganography has been shown, in which the stego audio is different from the original audio i.e., the unauthorized party can observe the difference in original and stego audio which in results insecure system. We can found that both original and stego audio looks like same in fig. 7, which has achieved by our proposed DWT approach. Also compared the both existing and proposed schemes in below figures

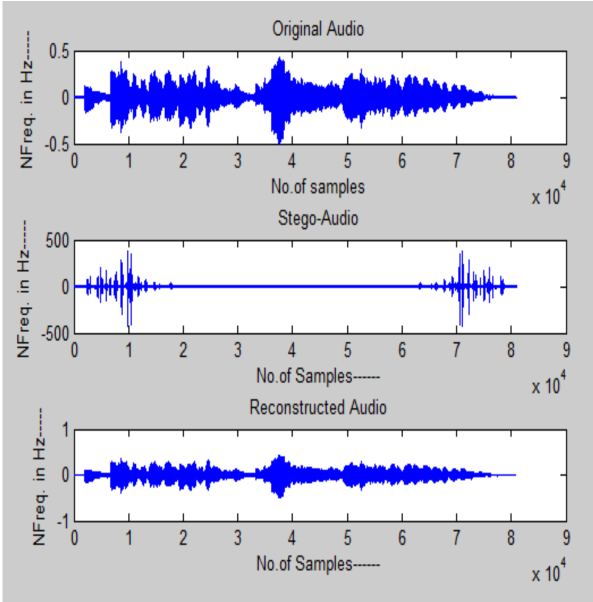

Fig.6 Output of FFT based audio steganography scheme

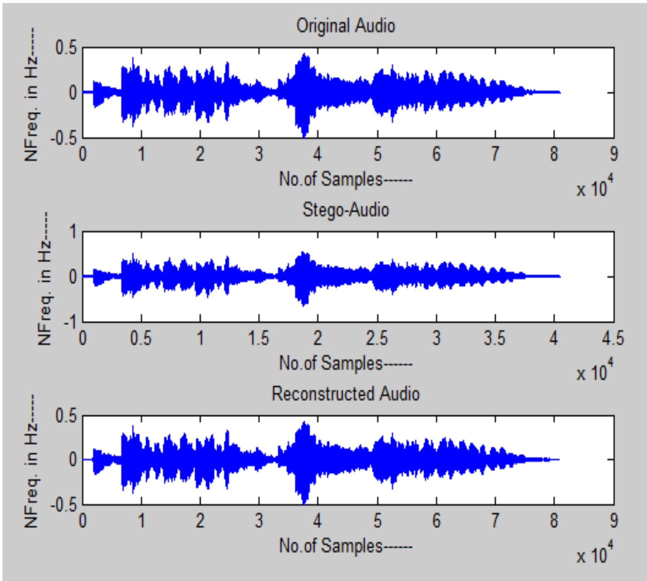

Fig.7 Output of proposed audio steganography scheme

CONCLUSION

Based on the test results, we conclude that the proposed scheme of Audio Steganography shows the best results than the FFT algorithm. Direct-sequence Spread Spectrum steganography on audio cover object is possible and practical to use, at least for the duration of the first 25 seconds of data. This method proved very robust against audio manipulation and very safe with the resulting noise is quite small.

Accuracy is more for DWT as compared to FFT.

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