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Securing Audio Watermarking System using Discrete Fourier Transform for Copyright Protection

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***Abstract**:-* The recent growth in pc networks, and a lot of specifically, the planet Wide internet, copyright protection of digital audio becomes a lot of and a lot of necessary. Digital audio watermarking has drawn in depth attention for copyright protection of audio information. A digital audio watermarking may be a method of embedding watermarks into audio signal to point out genuineness and possession. Our technique supported the embedding watermark into audio signal and extraction of watermark sequence. We tend to propose a brand new watermarking system victimization separate Fourier remodel (DFT) for audio copyright protection. The watermarks area unit embedded into the best outstanding peak of the magnitude spectrum of every non-overlapping frame. This watermarking system can provides robust lustiness against many styles of attacks like noise addition, cropping, re-sampling, re-quantization, and MP3 compression and achieves similarity values starting from thirteen sound unit to twenty sound unit. Additionally, planned systems attempting to realize SNR (signal-to-noise ratio) values starting from twenty sound unit to twenty-eight sound unit.

***Keywords**- Audio Segment, Discrete Fourier Transform (DFT), Signal-to-noise-ratio (SNR)*

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1. INTRODUCTION

The recent growth in laptop networks, and a lot of specifically, the globe Wide internet, copyright protection of digital audio becomes a lot of and a lot of vital. Digital audio watermarking has drawn in depth attention for copyright protection of audio knowledge. A digital audio watermarking could be a method of embedding watermarks into audio signal to indicate credibility and possession. Audio watermarking ought to meet the subsequent requirements:

(a)Imperceptibility: The digital watermark shouldn't have an effect on the standard of original audio signal once it's watermarked;

(b) Robustness: The embedded watermark knowledge shouldn't be removed or eliminated by unauthorized distributors mistreatment common signal process operations and attacks;

(c)Capacity: Capacity refers to the numbers of bits which will be embedded into the audio signal at intervals a unit of time;

(d)Security: Security implies that the watermark will solely be detectable by the approved person. Of these needs are usually contradictory with one another. However, it ought to satisfy the vital properties like physical property and hardiness.

During this paper, we tend to propose a brand new watermarking system exploitation separate Fourier rework (DFT) for audio copyright protection. The watermarks are embedded into the best outstanding peak of the magnitude spectrum of every non-overlapping frame. Experimental results indicate that the projected watermarking system provides robust hardiness against many sorts of attacks like noise addition, cropping, re-sampling, re-quantization, and MP3 compression and achieves similarity values starting from thirteen to twenty. Additionally, our projected system achieves SNR (signal-to-noise ratio) values starting from twenty decibel to twenty-eight decibel.

2. LITERATURE SURVEY

A significant range of watermarking techniques is reportable in recent years so as to form strong and in cognoscible audio watermarks. Lie proposes a way of embedding watermarks into audio signals within the time domain. The projected formula exploits differential average-of-absolute-amplitude relations inside every cluster of audio samples to represent one-bit info. It additionally utilizes the low-frequency amplitude modification technique to scale the amplitudes in elect sections of samples in order that the time domain wave envelope is virtually preserved. Authors propose a blind audio watermarking system that embeds watermarks into audio signal in time domain. The strength of the audio signal modifications is proscribed by the need to supply a signal for watermark detection. The watermark signal is generated employing a key, and watermark insertion depends on the amplitude and frequency of audio signal that minimizes the perceptibility of the watermarked signal. Ling frequency points of embedding watermark are elect by the key. Zengdescribe a blind watermarking system that embeds watermarks into DCT coefficients by utilizing quantization index Modulation technique. The authors propose a watermarking system that embeds synchronization signals in time domain to resist against many attacks. Pooyan introduce AN audio watermarking system that embeds watermarks in moving ridge domain. The watermarked knowledge is then encrypted and combined with a synchronization code and embedded into low frequency coefficients of the sound in moving ridge domain. The magnitude of quantization step and embedding strength is adaptively determined consistent with the characteristics of human sensory system.

Audio watermarking started consider later as engaging space that have viable applications and area for development (Zhang et al., 2010a, b; Abdul fetah et al., 2010a, b). Within the past few years, many techniques for information hidden in audio sequences are conferred. All of the developed techniques take good thing about the sensory activity properties of the human sensory system (HAS).

The main challenge in digital audio watermarking is that if the sensory activity transparency parameter is fastened, the planning of a watermark system cannot acquire high strength and a high watermark information rate at a similar time (Cvejic, 2004; principle et al., 2009). To achieve any of knowledge hidden goals, we\'d like to pick out a proper cowl, domain, and take into the account the challenges of knowledge hidden approaches.

Large work has been distributed in audio watermarking using unfold spectrum technology and is given in several key publications like (Bender et al., 1996), (Coxet al., 2002) and (Cvejic, 2004). The primary technique of spread spectrum into watermarking was in (Cox et al., 1997). Xu et al. (1999) planned a multiple echo technique. Instead of embedding one massive echo into the host audio signal, they use multiple echoes with different offsets. Oh et al. (2001) introduced the positive negative echo activity theme. Their echo kernels comprise positive and negative echoes at near locations. Since the frequency response of a negative echo is that the inversed form having similar ripples as that of a positive echo, the frequency response of the positive and negative echoes has the sleek form within the low frequency band. By using positive and negative echoes, one will so enter multiple echoes to permit that the host audio quality isn't apparently deteriorated. Kim associate degreed Choi (2003) given an echo activity theme with backward and forward kernels. The on paper derived results show that the amplitude of the cepstrum coefficient at the echo position from the backward and forward kernels is larger than that from the backward kernel only if the embedded echoes are parallel. Therefore, the backward and forward kernels will improve the hardiness of echo activity theme.

Ko et al. (2005) went additional to propose the time-spread echo kernel. With the utilization of pseudo-noise sequence, an echo is displayed as various very little echoes during a time region. Once the embedded information of watermarked audio signals square measure extracted, the pseudo-noise sequence functions sort of a secret key. While not getting the pseudo noise sequence utilized in the embedding method, extracting the embedded information would be tougher.

In order to feature a watermark into a bunch signal in an exceedingly perceptually clear manner, a good vary of embedding techniques square measure projected going from straightforward least important bits (LSB) theme or Low-bit encryption, Phase writing, unfold spectrum, Patchwork writing, Echo coding and noise gate technique.

Description About the Detect Process:

The work flow of detecting duplicated audio segment mainly includes three steps:

(1) We divide the audio file into several segments with the time span of T. This is often the essential making ready for the analysis of next half. And for this division, choosing a proper time span T is important for US. If the value of T is simply too tiny, it'll increase the computation load and it's no sensible significance. If the value of T is simply too giant, not all duplicate audio inserting parts may be detected, and it'll impact the accuracy of our outcome. Therefore below traditional circumstances, 0.2 s will be chosen as a replica half that isn't any but 0.4 s. so we are able to make certain that we tend to ne'er miss touch a duplicated section and cut back the computation load.

(2) We need to calculate the similarity that is that the standard of the similar degree of 2 components in associate audio file. The step is that the most essential step in our method. Within the half one, we've got introduced the definition and that means of similarity and therefore the quick convolution algorithms to scale back the calculated quantity. Besides, the sample rate can have an excellent impact on the calculation of similarity too. Normally, the sample rate of a WAV format audio is forty four. 1 kHz; if getting used directly, the computation of this program continues to be vast and can lead to a decrease to the computation potency. So computing prices will be reduced and potency will be advanced additional by reducing the sample rate to associate applicable degree. That is, if the sample rate is reduced about D times, the calculation of the entire program will be shriveled by D times. this may improve operative efficiency greatly. At constant time, the reduced multiples will be set every which way consistent with the accuracy being desired. For convenience, we tend to assume D = 1 in this study.

(3) We need to set a threshold to decide which parts are duplicated. If the value of a similarity is larger than this threshold, the relative parts of this similarity are duplicated. From this method, we can know the exact location of duplicate parts too. This threshold can be set based on the precision we want.

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3. Problem Defination

There are many ways to tamper digital audio. Audio forensics is becoming more and more important as digital audio can be used as evidence in court and other special occasions. The authenticity of an audio file has got some researchers attention recently as the authenticity of digital audio becomes more and more important. And inserting duplicated segment into WAV file is a common audio tampering method. There are indeed some technologies to detect the authenticity of digital audio, but little effort has been put to the research on to provide more security. To overcome this, we are going to use watermarking system using Discrete Fourier Transform (DFT) for copyright protection.

4. Proposed Method

This project consists of following two methods:
1. Embedding
2. Extraction
The descriptions of proposed methods are as follows:

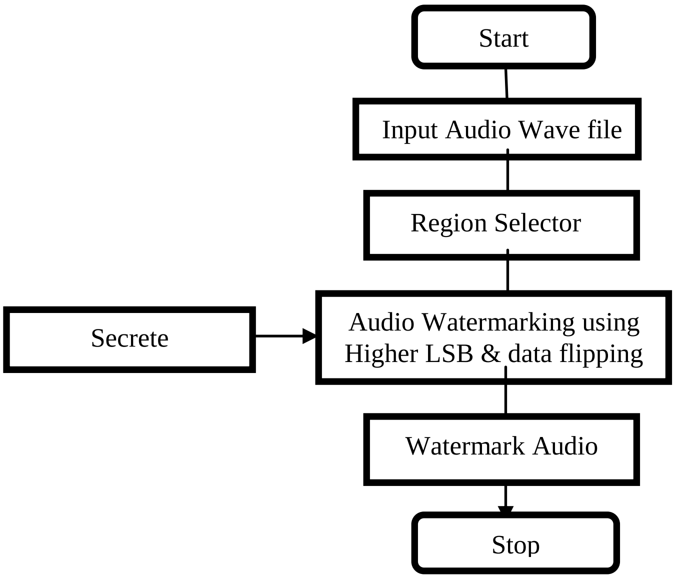
4.1. Embedding

•The original audio is initial segmental into non-overlapping frames.
•Calculate the magnitude and section spectrum of every frame victimization separate Fourier rework (DFT).
•Find the foremost outstanding peak V from magnitude spectrum employing a peak detection algorithmic rule.
•Place watermarks into the best outstanding peak of the magnitude spectrum of every frame to get watermarked peak V'.
•Insert back the changed peak into the magnitude spectrum of every non-overlapping frame.
•Take associate degree inverse DFT of the complicated spectrum to calculate the watermarked frame.
•Finally watermarked audio signal is computed by non-overlap-adding (NOLA) the watermarked frames within the time domain.

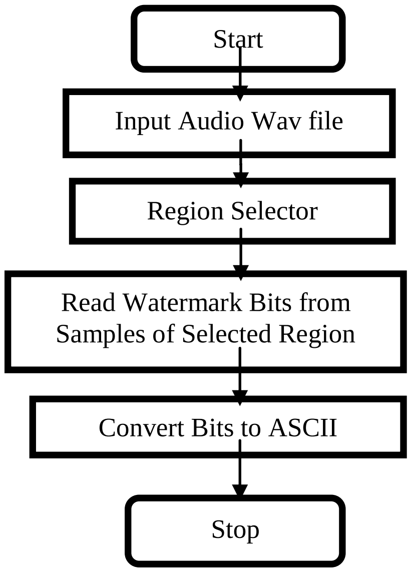
4.2. Extraction

•Calculate the DFT of the attacked watermark audio frame.
•Extract the very best outstanding peak from the magnitude spectrum that is found at an equivalent position within the embedding method higher than.
•The watermark sequence extracted

4.1.1 Data Flow Diagram of Embedding:



4.2.1 Data Flow Diagram of Extracting



5. Experimental Results

In this section, we will show the effect of checking the distortion between original and watermark embedded audio files by using the method mentioned above. We will design three experiments to check the effect of this method. The first experiment is using a WAV format file to check the distortion between original and watermark embedded file, while the second experiment is time required for embedding and extraction of watermark on same size of audio files with different number of samples selected i.e. region. The last experiment is the time required for embedding and extraction of watermark on different size of audio files.

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5.1 Outcome of distortion between files

	Audio 1	Audio 2	Audio 3
Total Samples	179660	40690	9749
Number of Selected Samples	14990	19980	180
MSE	0	24.0650	0
PSNR	Infinity	281	Infinity
Accuracy Rate: 90%			

Table 5.1 overall Proposed System Result

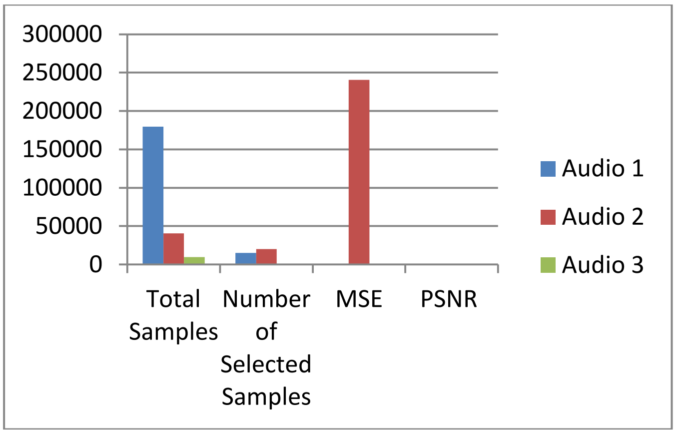

Fig 5.1 Input Audio Vs Parameter

5.2 Outcome of Time required for watermark embedding and extraction on Audio File with same size

Table 5.2 describes the time for embedding and extraction of watermark into audio files of same size. In that we are hiding watermark into same number of samples. So from this we decided accuracy 100%. Its graphical representation is shown in fig 5.2

Audio Files (Length Constant)	Audio 1	Audio 2	Audio 3
Number of Hidden Selected Samples (Constant)	2475	2475	2475
Time for Embedding	1.58	1.69	1.61
Time for Extraction	0.03	0.04	0.04
Accuracy Rate: 100%			

Table 5.2 Time for Embedding and Extraction of Watermark with constant size of Audio File

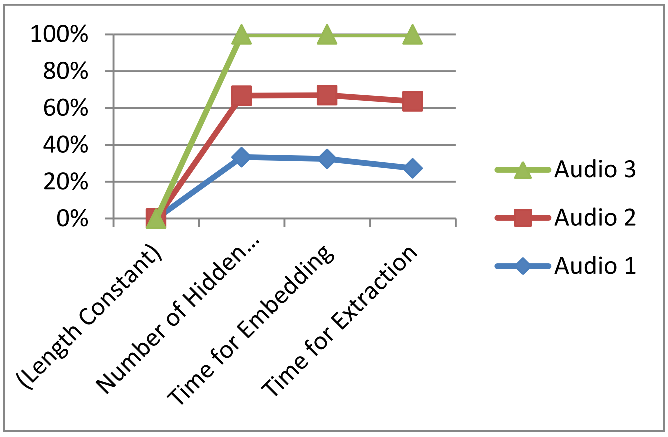

Fig 5.2 Input Audio File with size constant versus parameter

5.3 Outcome of Time required for watermark embedding and extraction on Audio File with different size

Table 5.3 describes the time for embedding and extraction of watermark into audio files of different size. In that we are hiding watermark into same number of samples. So from this we decided accuracy 100%. Its graphical representation is shown in fig 5.3

Audio Files (Length Increase)	Audio 1	Audio 2	Audio 3
Number of Hidden Selected Samples (Constant)	2480	2480	2480
Time for Embedding	1.94	1.56	2.01
Time for Extraction	0.02	0.01	0.01

Table 5.3 Time for Embedding and Extraction of Watermark with different size of Audio File

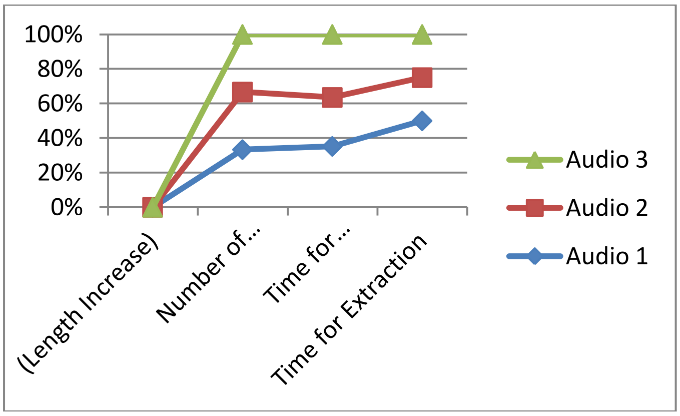

Fig 5.3 Input Audio File with different size of audio files versus parameter

6. Conclusion

The existing system is not sufficient for recognizing the accurate and more precise result. from the results, it was observed that the proposed system is highly robust means embedded watermark data should not be removed or eliminated by unauthorized distributors using common signal processing operations. Lastly, security of watermark is

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investigated that implies that the watermark can only be detectable by authorized person. It was found that the digital watermark should not affect the quality of original audio signal after it is watermarked. The proposed system provides imperceptibility, robustness, capacity and security.

The following conclusions were made:

Audio Watermarking System achieves highly accurate and reliable embedding and extraction of watermark

Audio Watermarking System provides imperceptibility, robustness, capacity and security.

Audio Watermarking System achieves minimum distortion and securely watermark authenticate signature.

Peak Signal to Noise Ratio is minimum and size of the audio file is same after it is being watermarked.

Audio Watermarking System achieves irreversible authentication

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