On Line Secret Watermark Generation for Audio files

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***Abstract*— This paper presents an on line dynamic robust and efficient approach of audio watermarking for copyright protection. In this paper I am using on line secret watermark generation to generate additional information (watermark) automatically during embedding process. The Direct Sequence Spread Spectrum (DSSS) technique is taken to spread the watermark bits over the entire spectrum of audio signals imperceptibly by creating chip sequences using the bits of watermark. The watermark is shaped in audio signals without the violation of psycho acoustic properties of Human Auditory System (HAS) in frequency domain. The generated watermark is unique for every audio file. The size of watermark depends upon the size and numbers of audio file samples.**

***Index Terms*—DSSS- Direct Sequence Spread Spectrum, HAS – Human Auditory System, HVS – Human Visual System,.**

I. Introduction

With the advances in the field of information technology and its application in different areas, the issue of information security has gained importance. Various methods such as cryptography, staganography, and others have been used to address the issue. One of the major concerns in the area of security of information is unauthorized copying and copyright protection of digital media.

In encryption/decryption system a key and an algorithm are used to encrypt plain text into cipher text. The decryption algorithm with key is used to decrypt the cipher text into plain text. If this technique of security management is applied on the digital media such as audio then the audio signals need to be decrypted before playing [1]. Therefore, the user must be supplied with the decryption key after he has paid for his copy of digital media. But, once the digital file is decrypted, it can be used for creating multiple identical copies and redistribution. This violates the security related with digital media for unauthorized copying and ownership, the owner of the media may not be able to claim for ownership.

The concept of digital watermarking has been discovered for protecting digital media from tampering and enforcing digital copyright protection. Watermarking technology for digital media such as audio, video, image and text started in 1996. Video and image watermarking became very popular due to simplicity for producing illusion of vision to watermark. Audio watermarking was not so popular at that time due to tedious task for producing illusion for dynamic varying human hearing system. Digital audio watermarking is now accepted as industry standardization for the music industry.

In audio watermarking, watermark is special secret information, a key or a message which is being hidden under the audio file on behalf of the owner of the audio [2]. The key serves for the copyright of the owner of the audio and is not required for playing back the audio after watermark insertion. When the additional watermark bits replaces the bits of host audio samples, an attempt is being made that there should not be recognizable distortion in sound quality observed. Further, the size of the audio file does not change after embedding the watermark [3]. There are two types of information an audio file contains after watermarking, original PCM samples and watermarked bits. Watermark does not have any significance to the user of audio but the author can extract this hidden watermark for claims against his/her ownership. Audio watermarking uses the weakness of Human Auditory System (HAS) to hide information in audio and is more challenging than others due to the fact that HAS has more precision than HVS (Human Visual System).

In this paper, I propose a DSSS-Based audio watermarking method. Direct sequence spread spectrum, considered as one of the best technique for audio watermarking attempts to spread the watermark bits over the entire spectrum of audio signals by creating chip sequences using the bits of watermark. Audio watermarking system using DSSS allows embedding large number of watermark bits in the audio and is more robust and secure against adversary attacks and digital signal processing attacks for modification and removal of watermark bits from the audio signal.

II. Related Works

Over the past few years, various watermarking schemes for digital audio have been proposed. In 1998, a watermarking algorithm was produced based on perceptual masking of audio signals and in the same year, another algorithm was produced which was based upon time domain of audio signals. Spread spectrum audio watermarking with perceptual masking of audio signals was produced in 1999 [5]. In 2002, a revised and improved version of algorithm was produced by H.S. Malwar and Dr. Kirovski through Microsoft Corporation [8].

A novel audio watermarking scheme using direct sequence spread spectrum (DSSS) method which can embed a text message as a watermark into an audio signal

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imperceptibly was proposed by Yuksel Tokur and Ergun Ercelebi [4].

Jongwon Seok et al. discussed digital watermark technology as a new method for protecting digital content from unauthorized copying. The proposed watermarking scheme includes a psychoacoustic model of MPEG audio coding to ensure that the watermarking does not affect the quality of the original sound. After embedding the watermark, their scheme extracts copyright information without access to the original signal by using a whitening procedure for linear prediction filtering before correlation [7].

Swanson M et al. presented the transparent data embedding algorithms that embed binary streams in host multimedia signals. The embedded data is perceptually inaudible to maintain the quality of the source data. The embedded data can add features to the host multimedia signal or provide copyright protection [3].

Most of the developed audio watermarking algorithms do not generate the watermark on the basis of the properties of audio such as sampling frequency, size in bytes of audio and number of quantization bits. The most popular system of audio watermarking proposed by H.S Malwar and Dr. Kirovski based on direct sequence spread spectrum along with psycho acoustic model also requires the user to input watermark bits during the embedding process [6]. This watermarking system embeds only 4 bytes of integers as watermark which have to be selected by watermarking people. Most of the systems require the original version of audio signal during watermark extraction for solving the dispute.

III. PROPOSED SYSTEM

Watermarking system for digital audio includes two processes, watermark embedding and watermark extraction. The proposed system embeds watermark in the form of a key of at least 32 bits or more bits into the host audio signal depending upon the size of audio file. The key is directly generated during the embedding process using specific properties of the audio such as audio size, sampling frequency and number of bits used for resolution per sample and does not require the user to provide the key details. The purpose of embedding system is to insert the additional pseudo random key bits in the host audio signal. The watermark embedding system is represented in fig. 1.

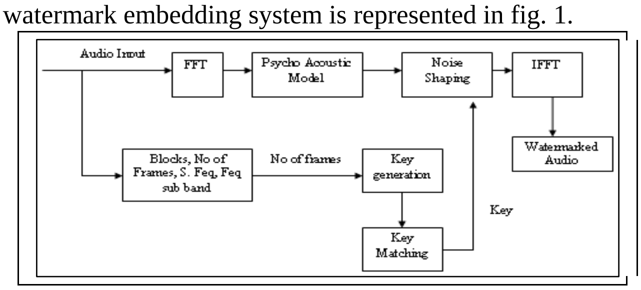


Fig. 1. Watermark Embedding System.

In embedding process of watermark audio file is taken as input with watermark embedder. The watermark embedder generates the watermark according to size, resolution and sampling frequency of the audio files. A pseudo random number generator is implemented with watermark embedder to generate integers for watermark.. The generated watermark is compared in database of watermark information. if the generated watermark is present in database then the currently generated watermark is discarded and regeneration is taken again and again.

The audio samples of 1024 bits are taken

The steps of watermark embedding system are as follows:

1. Read the audio samples from the host audio and read the sampling frequency, number of resolution bits, and samples into the local array variables.
2. Generate frames from audio samples and assign sequence number for each.
3. Generate the key according to the number of samples in the given audio file by pseudo random number generator algorithm.
   - If the key is less than 4 bytes in size then it is adjusted by adding decimal ‘1000’.
   - Repeat the following steps till a unique key is generated
     - Check the key in the database if already assigned to some audio file
     - If found in database then the key is discarded and a new key is generated using the same parameters.
4. Store the key in a local variable.
5. Calculate masking threshold of current analysis frame using psycho acoustic model with 1024 bits (8 X 128 samples of audio) by 128 point FFT.
6. Using the masking threshold, shape the watermark signal to be imperceptible in frequency domain.
7. Compute the inverse FFT of the shaped watermark audio frame.
8. Assign output sequence number to current frame.
9. Write output frame in wav audio file.

IV. WATERMARK EXTRACTION PROCESS

The watermark extraction system extracts/ decodes the watermark from the watermarked audio signal and does not require the original host audio for the purpose. The proposed audio watermarking system manages the database of watermarked audio for the specified owner with respective watermark bits. The watermark extraction system is represented in fig. 2.

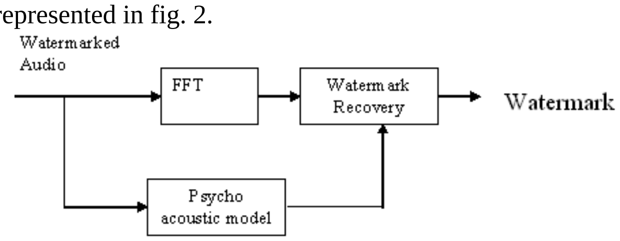


Fig. 2. Watermark Extraction System.

The steps of watermark extraction system are as follows:

1. Read the audio samples from the watermarked audio and store the sampling frequency, number of resolution bits, and samples into the local array variables.
2. Generate frames of audio samples with 1024 bits (8 bits X 128 samples).
3. Calculate masking threshold value with 1024 bits (8 X 128 samples of audio) by 128 point FFT and select the bit which lies onto the threshold value in the audio samples.

4. Read the bit value in the local variable and repeat steps 3-4 till end of the frames.
5. Calculate the key value from the extracted bits.

V. Experimental Results

The proposed algorithm of audio watermarking systems is tested with the 20 to 30 seconds audio clips by embedding 32 bits key as watermark in the host audio signal. No noise is observed while playing the watermarked audio and there was no effect seen on the size of the audio file. The SNR of both host and watermarked audio is not the same. The SNR of watermarked audio decreases due to the insertion of watermark bits, which are treated as noise in the audio samples. Fig. 3 (a), (b), (c) indicates the frequency map of host audio, watermark and watermarked audio respectively. Table 1 represents the signal to noise ratio of host and watermarked audio.

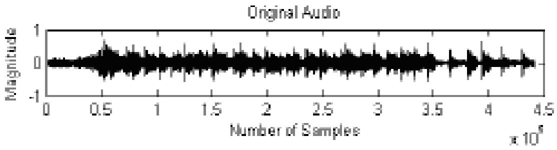


Fig. 3(a). Host Audio

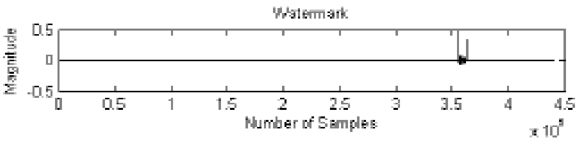


Fig. 3(b). Watermark

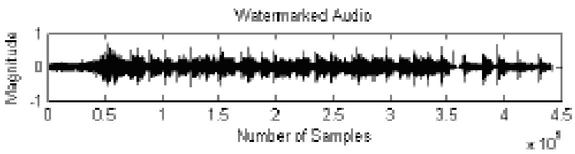
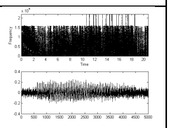
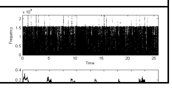


Fig. 3(c). Watermarked Audio

Table 1. SNR comparison

Time	Host Audio	Watermarked Audio	Figure
20 Sec.	Music1.wav	Music1w.wa	
SNR	10.2418	9.621 dB	
25 Sec.	Music2.wav	Music2w.wa	

VI. Robustness Testing For Various Attacks

The audio files used in the experiments are mono audio signals, 10-20 seconds long, sampled at 44.1 KHz with 16 bits resolution. 32 bits key have been embedded in the audio clips as watermark. The robustness tests against various types of attacks are as follows.

Amplitude Compression

The amplitude compression having ratio of 2:1 above -20 dB, and with ratio 1:1 below -20 db does not cause the loss of frequency components of watermarked audio but some change in the pitch was observed during the playback of audio.

Echo Addition

The watermarked audio was tested by echo addition with a delay of 60 ms and decay of 40% respectively. The echo produces the noise but there is no loss of watermarked signals in audio.

MPEG 1 Audio Layer III Compression

The robustness against MPEG 1 audio Layer III compression has been tested by using a compression rate of 96 kbps for the watermarked audio file. The mp3 version of watermarked audio is produced with reduction of size at least 10 times of original wav form watermarked audio. There is no degradation in the sound quality and no addition of noise was observed during playback of the new version of audio.

Resampling

The original audio signals have been sampled with a sampling rate of 44.1 kHz. The sampling rate of the watermarked audio was reduced to 22.050 kHz and then was re-sampled to the original rate of 44.1 kHz. No quality degradation of audio was observed.

Requantization

The 8-bit resolution of audio samples is often used in games and multimedia applications. We therefore tested the process of quantization of a 16-bit watermarked audio signal to 8-bits and then back to 16-bits. This does not increase the incoherent background noise of the audio track due to the rounding errors in the processing [2].

Noise Addition

White noise with a constant level of 50 dB was added to the watermarked audio under the averaged power level of the audio signal. This added white noise produces extra bits and file size is increased. The audio quality is degraded and the watermarked audio performance becomes lower.

Table 2 represents the bit error rate for various types of attacks on watermarked audio [7].

Attack type on watermarked audio	**Bits errors w.r.t embedded bits**		**No. of watermark bits embedded**
	B1(1st WM bit)	B2(2nd WM bit)	
MP3 compression	1/16	0.5/16	32
Resampling	0.1/16	0.3/16	32
Requantization	0.2/16	0.4/16	32
DSP (Filtering)	4/16	3/16	32
Noise addition	1/16	1.5/16	32
Totals	6.3	5.7	160
Bit Error rate	**3.93%**	**3.56%**	**7.5%**

VII. CONCLUSION

In the proposed system of audio watermarking, direct sequence spread spectrum has been used. Psycho acoustic auditory model is taken for calculating masking threshold of audio signal. Masking threshold is used for shaping the watermark for imperceptibility. The embedding algorithm generates the watermark in the form of key and frames the audio to 128 samples each. The extraction system successfully extracts the watermark from the watermarked audio signal. The watermarked audio file was tested against various types of attacks and was found robust against these attacks.

The proposed watermarking system is tested to watermark the pulse code modulated wave format of audio file. The system can be extended for watermarking various other formats of audio files such as .mp3, .wma etc. The performance of the watermark embedding system can be further enhanced by embedding the watermark both in lower and higher frequency components of audio signals simultaneously and thus can accommodate more number of watermark bits.

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