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Audio Watermarking by Reducing Noise Using Fast Fourier Transformation

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Abstract: The motivation for this research work includes the provision of protection of intellectual property rights, an indication of content manipulation, and a means of protecting digital media from tampering and all this without affecting the quality of data. A new scheme is proposed for wave files which reduce the noise by applying Fast Fourier Transformation (FFT) and taking random carrier to embed the watermark into the audio bit stream instead of fixed or low frequency carrier. The new scheme has implemented and its audio quality parameters are compared to the other best known audio watermarking technique. Comparison has been done on the basis of Peak Signal to Noise Ratio, Signal to Noise ratio and Bit Error Rate. The comparison proves that the proposed technique give better results. Comparison is done with best know An Adaptive watermarking Algorithm for Mp3 compressed Audio Signal techniques proves that the proposed and implemented algorithm for audio watermarking gives better results**.**

Keywords: Audio Watermarking; Fast Fourier Transformation; Peak Signal to Noise Ratio; Signal to Noise Ratio; Bit Error Rate

I. INTRODUCTION

The rapid development of the Internet and the digital information revolution caused significant changes in the global society, ranging from the influence on the world economy to the way people nowadays communicate. Broadband communication networks and multimedia data available in a digital format (images, audio, video) opened many challenges and opportunities for innovation. Versatile and simple-to-use software and decreasing prices of digital devices (e.g. digital photo cameras, camcorders, portable CD and mp3 players, DVD players, CD and DVD recorders, laptops, PDAs) have made it possible for consumers from all over the world to create, edit and exchange multimedia data. Broadband Internet connections and almost an errorless transmission of data facilitate people to distribute large multimedia files and make identical digital copies of them. Digital media files do not suffer from any quality loss due to multiple copying processes, such as analogue audio and VHS tapes. Furthermore, recording medium and distribution networks for analogue multimedia are more expensive. These first-view advantages of digital media over the analogue ones transform to disadvantages with respect to the intellectual rights management because a possibility for unlimited copying without a loss of fidelity cause a considerable financial loss for copyright holders. The ease of content modification and a perfect reproduction in digital domain have promote the protection of intellectual ownership and the prevention of the unauthorized tampering of multimedia data to become an important technological and research issue**.**

Digital watermarking has been proposed as a new, alternative method to enforce the intellectual property rights and protect digital media from tampering. In this method a pattern of bits inserted into a digital image, audio or video file that identifies the file's copyright information (author, rights, etc.). Moreover, the actual bits representing the watermark must be scattered throughout the file in such a way that they cannot be identified and manipulated. And finally, the digital watermark must be robust enough so that it can withstand normal changes to the file, such as reductions from lossy compression algorithms.

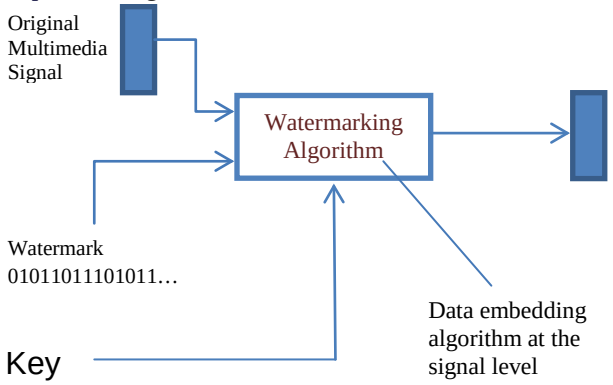

Figure 1: Watermark Embedding Process

Audio watermarking is of two types i.e. blind and non blind audio watermarking. If the detection of the digital watermark can be done without the original data, such techniques are called blind. Here, the source document is scanned and the watermark information is extracted. On the other hand, non blind techniques use the original source to extract the watermark by simple comparison and correlation procedures. However, it turns out that blind techniques are more insecure than non blind methods.

In the audio industry today there is much interest in copyright management and protection. Embedding some form of "hidden signal" or watermark in the audio stream is seen as a potential method for managing the use of the material. To ensure that only those with the right to access it can do so, methods have been proposed to include gatekeepers in audio equipment. That way, unauthorized reproduction, and especially unauthorized copying, could be prevented. However, there are many serious problems with this concept. it is not at all clear that they can be overcome sufficiently to provide a reliable and effective control system.

Bingwei Chen et al. [1] proposed an adaptive watermarking algorithm for MP3 compressed audio signals, based on the human auditory system. In the proposed algorithm watermark is embedded adaptively and transparently after Modified Discrete Cosine Transformation (MDCT) and before quantization. Gaussian distribution statistic analysis is introduced to make this watermarking algorithm adaptive. In this paper, a new FFT based audio watermarking algorithm is introduced.

II. WATERMARKING PROPERTIES

A watermarking algorithm can be characterized by a number of properties. The relative importance of each property however depends on the demands of the application. The six important properties are as follows [2].

A. Perceptual Transparency:

In all most every application, the watermark-embedding algorithm has to insert watermark data without changing the perceptual quality of the host audio signal. The fidelity of a watermarking algorithm is usually defined as a perceptual similarity between the original and watermarked audio sequence. However, the quality of the watermarked audio may get tainted, either intentionally by an adversary or unintentionally during the transmission process, before a person perceives it. In such a case, it is more sensible to redefine the fidelity of a watermarking algorithm as a perceptual similarity between the watermarked audio and the original host audio at the point at which they are presented to a consumer.

B. Watermark bit Rate:

Bit rate is of an embedded watermark is defined as the number of bits of the watermark embedded in one second of the host audio signal and is given in bits per second (bps). The bps requirement of a watermark depends on the application. For example, some applications, such as copy control, require the insertion of a serial number or author ID, with the average bit rate of 0.5 bps. In some envisioned applications, like hiding speech in audio, algorithms have to be able to embed watermarks with the bit rate that is a significant fraction of the host audio bit rate, i.e. up to 150 kbps.

C. Robustness:

The robustness of a watermarking algorithm is defined as its ability to detect/ extract the watermark after common signal processing manipulations. The set of signal processing modifications to which a watermarking algorithm needs to be robust against is completely application dependent. For example, in radio broadcast monitoring, embedded watermark need only to survive distortions caused by the transmission process, including dynamic compression and low pass filtering, because the watermark detection is done directly from the broadcast signal. On the other hand, in some algorithms robustness is completely undesirable and those algorithms are labeled fragile audio watermarking algorithms.

D. Blind or Informed Watermark Detection:

In some applications, a detection algorithm may use the original host audio to extract watermark from the watermarked audio sequence (informed detection). It often significantly improves the detector performance, in that the original audio can be subtracted from the watermarked copy, resulting in the watermark sequence alone. However, if detection algorithm does not have access to the original audio (blind detection) and this inability substantially decreases the amount of data that can be hidden in the host signal. The complete process of embedding and extracting of the watermark is modeled as a communications channel where watermark is distorted due to the presence of strong interference and channel effects. A strong interference is caused by the presence of the host audio, and channel effects correspond to signal processing operations.

E. Security:

Watermark algorithm must be secure in the sense that an adversary must not be able to detect the presence of embedded data, let alone remove the embedded data. The security of watermark process is interpreted in the same way as the security of encryption techniques and it cannot be broken unless the authorized user has access to a secret key that controls watermark embedding. An unauthorized user should be unable to extract the data in a reasonable amount of time even if he knows that the host signal contains a watermark and is familiar with the exact watermark embedding algorithm. Security requirements vary with application and the most stringent are in cover communications applications, and, in some cases, data is encrypted prior to embedding into host audio.

F. Computational complexity and cost:

The implementation of an audio watermarking system is a tedious task, and it depends on the business application involved. The principal issue from the technical point of view is the computational complexity of embedding and detection algorithms and the number of embedders and detectors used in the system. For example, in broadcast monitoring, embedding and detection must be done in real time, while in copyright protection applications; time is not a crucial factor for a

practical implementation. One of the economic issues is the design of embedders and detectors, which can be implemented as hardware or software plug-ins, is the difference in processing power of different devices (laptop, PDA, mobile phone, etc.).

III. PROPOSED SCHEME

The watermarking method performed is shown in figure 4.1. The audio file is portioned into frames which are 90 milliseconds in duration. This frame size is chosen so that the embedded watermark does not introduce any audible distortion into the file. Then Fast Fourier Transformation (FFT) will be performed. After FFT random sample is chosen so that it will be difficult to detect the watermark. Finally the watermark is embedded in the .wav file with random sample. FFT process is a simple conversion based on the assumption that the changes between samples will not be very large. The first sample value is stored in its entirety, and the each successive value describes the amount +/- 8 levels that the wave will change, which uses only 4 instead of 16 bits.

Therefore, a 4:1 compression ratio is achieved with less loss as the sampling frequency increases. At 44.1 kHz, the compressed signal is an accurate representation of the uncompressed sample that is difficult to discern from the original. This method is used widely today because of its simplicity, wide acceptance, and high level of compression.

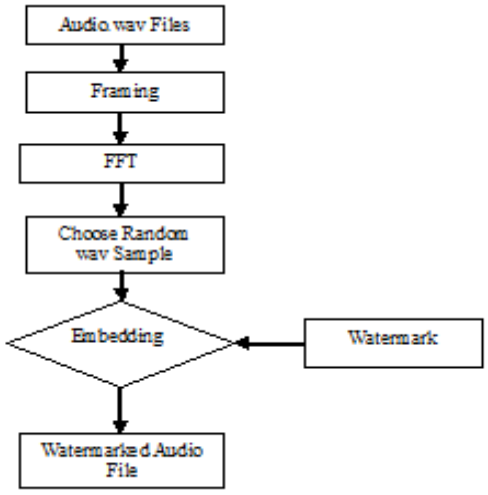

Figure 2. Proposed watermark embedding algorithm

A. *FFT based Watermarking Embedding Steps:*

a. Input the audio file.
b. Convert the audio file into audio samples.
c. Applying FFT to reduce the noise from the audio file.
d. Randomize the sample audio bit and find the starting point of the inserting text into it.
e. Insert the text in the audio sample bits.
f. Repeat step 5 till all the text bits are inserted.
g. Reconstruct and form audio file.

B. *FFT based Watermarking Extraction Steps:*

a. Firstly key is used to locate the sample in which watermark is added.
b. Read a byte from the key and read one sample from the wave stream.
c. Then the watermark bit is extracted from the lowest bit of the sample and put in the watermark array.
d. After the watermark is extracted IFFT is performed.

IV. EXPERIMENTAL RESULTS

In order to evaluate the performance of proposed watermarking algorithm, the following experiments were considered. The experiment was conducted on personnel computer AMD Turion-X2 with 2GB RAM. Music 1 is classic music, Music 2 and 3 are Vocal and Music 4 and 5 are Pop music.

Table 1: Experimental Calculations of SNR, PSNR, BER

Audio	SNR	PSNR	BER
Music 1	45.58	91.17	0
Music 2	47	94.87	0
Music 3	88	177.65	0
Music 4	99.62	199.24	0
Music 5	102.69	205.39	0

A. *Performance on Different Types of Audio:*

Watermark was embedded into the five different types of 16-bit mono audio signals sampled at 44.1 kHz. Table 1 gives the experimental results in terms of SNR (Signal to Noise Ratio), PSNR (Peak Signal to Noise Ratio) and BER (Bit Error Rate). The Table 1 shows PSNR, SNR and BER values for the proposed algorithm, where BER value is 0.

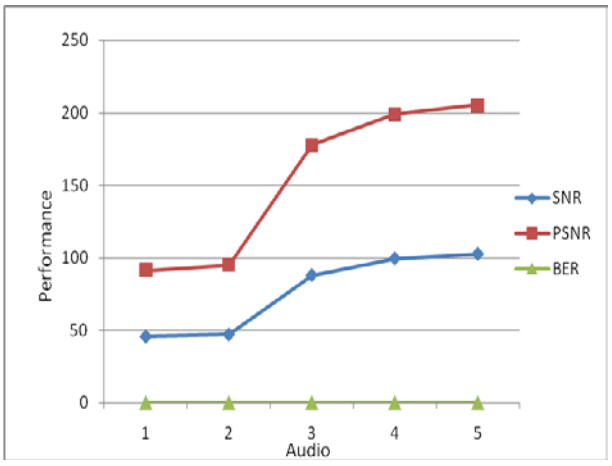

Figure 3. Performance of Different audio signals

Figure 3 shows the results of experiment test. It shows the watermarking algorithm's performance is good according to PSNR, SNR and BER. The Bit Error Rate is Zero in all type of signals.

V. CONCLUSION

In this paper, the FFT based audio watermarking algorithm is proposed for wav files. Watermark is embedded into wav file after reducing the noise using Fast Fourier Transformation. This experimental results shows that this watermarking give better results with high PSNR, SNR values and zero BER

under normal embedding and extracting conditions. Arbitrary sampling allows the perfect recovery of watermark.

VI. REFERENCES

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