Random Sample Audio Watermarking Algorithm for Compressed Wave Files

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Summary
In this research work, various audio watermarking techniques have been studied and investigated. A novel audio watermarking method is proposed in which watermark is added transparently after ADPCM (Adaptive Differential Pulse Code Modulation). In the proposed scheme we have used random sample instead of fixed or low frequency carriers to embed the watermark into the audio bit stream. Proposed technique has been 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 new scheme is suitable for retaining first generation archived files of high quality. A tool has been used for measuring the audio quality factors. Further, these quality parameters have been used to generate graphical outputs and tabular values for comparison with the best known audio watermarking technique. As the audio signal quality metrics i.e. higher Peak Signal to Noise Ratio (PSNR) and lower Bit Error Rate (BER) and higher Signal to Noise Ratio (SNR) itself are proving that the proposed technique for audio Watermarking is good one. The comparison with the best known Watermarking technique known as An Adaptive Watermarking Algorithm for MP3 Compressed Audio Signal techniques proves that the newly proposed and implemented algorithm has given much better results for audio watermarking. Higher values of PSNR of audio signal also prove the robustness of the proposed algorithm.

Key words:
Digital audio watermarking, ADPCM, Inaudibility, Robustness Peak Signal to Noise Ratio, Signal to Noise ratio and Bit Error Rate.

1. Introduction

In the past few years, a need has arisen for protecting copyright ownership of electronic media. Powerful and low cost computers allow people to easily create and copy multimedia content, and the Internet has made it possible to distribute this information at very low cost. However, these enabling technologies also make it easy to illegally copy, modify, and redistribute multimedia data without regard for copyright ownership. A recent example of this problem is the controversy regarding piracy of high-quality music across the Internet in MPEG Layer III (MP3) format

Digital watermarking is seen as a partial solution to the problem of protecting digital media, for it allows content creators to embed sideband data into a host signal, such as author or copyright information. Many techniques have been proposed for watermarking audio, image, and video.

Audio watermarking is a method to enforce the intellectual property rights and to protect the audio from tampering. It is of two types-Blind and Non-blind audio watermarking. If the detection of the digital watermark can be done without original data such techniques are called blind, On the other hand Non-blind techniques use the original source to extract the watermark by simple comparison and correlation procedures. In the past few years a large number of algorithms for secure and robust embedding and extraction of watermarks in audio files have been developed. A broad range of embedding algorithms goes from simple Least Significant Bit (LSB) methods to various Spread Spectrum schemes. Spread Spectrum schemes have gained a lot of popularity because of their innate robustness to intelligent attacks like dual watermarking. However, Spread Spectrum schemes fail to give a good performance against simple signal processing attacks such as mp3 compression and re-sampling.

Audio watermarking algorithm must satisfy at least two constraints: inaudibility and robustness. Embedded audio watermarks should be almost inaudible. Also, the algorithm should be robust enough to withstand attempts such as removal or alteration of inserted watermarks. These two constraints may seem to be contradictory. However, they must be satisfied.

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.

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In this paper, a new random sample audio watermarking algorithm for compressed audio signals is introduced.

1.1 Basic Watermarking Process

In general any watermarking scheme consists of three parts-the watermark, Encoder and the Decoder [1, 2, 3].

Here the watermark is a binary data sequence inserted into the host signal. For a digital watermark to be effective and practical, it should exhibit the following characteristics.

1) *Imperceptibility.* Embedding this extra data must not degrade human perception about the object. Namely, the watermark should be "invisible" in a watermarked image/video or "inaudible" in watermarked digital music. Evaluation of imperceptibility is usually based on an objective measure of quality, called peak signal-to-noise ratio (PSNR), or a subjective test with specified procedures.

2) *Security.* The watermarking procedure should rely on secret keys, not the algorithm's secrecy, to ensure security, so that pirates cannot detect or remove watermarks by statistical analysis from a set of images. The algorithm should be published and an unauthorized user, who may even know the exact watermarking algorithm, cannot detect the presence of hidden data, unless he/she has access to the secret keys that control this data-embedding procedure.

3) *Robustness.* The embedded watermarks should not be removed or eliminated by unauthorized distributors using common processing techniques, including lossy compression, linear or nonlinear filtering, cropping, and others.

4) *Adjustability.* The algorithm should be tunable to various degrees of robustness, quality, or embedding capacities to be suitable for diverse applications.

5) *Real-time processing.* Watermarks should be rapidly embedded into the host signals without much delay, so that integrated streaming/watermarking functionality in the delivery of audio over a network can be enabled. Also, a web crawler should support fast watermark extraction/ detection to authenticate multimedia presentations

Watermark encoder has two inputs-the watermark and the host signal. The output of watermark encoder is watermarked signal. The watermark is embedded with the use of a key by imposing imperceptible changes to the original host multimedia signal. The principal design challenge is in embedding the watermark so that it reliably fulfills its intended task. For copy protection applications, the watermark must be recoverable even when the signal undergoes a reasonable level of distortion, and for tamper assessment applications, the watermark must effectively characterize the signal distortions.

And the watermark decoder determines whether the watermark is present in the tested multimedia signal and what it is. The key is used to extract the watermark from the possibly distorted watermark signal. Unlike standard cipher systems used for encryption and authentication, digital watermarking does not restrict access to the information to prevent illicit acts. Instead, it provides evidence of a wrong-doing after it has taken place.

2. Proposed Scheme

In the proposed scheme audio file is partitioned into frames which are 90 milliseconds in duration. This frame size is chosen so that embedded watermark does not introduce any audible distortion into the file. Then ADPCM will be performed after ADPCM random sample is chosen so that it will be difficult to detect the watermark and finally the watermark is embedded in the selected sample.

ADPCM 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.

2.1 Watermark Embedding

Adding Watermark in Wave samples is very similar to hiding it in the pixels of a bitmap. Wave file has single WAVE chunk which consists of two sub chunks- fmt chunk and data chunk. The watermark is added into the data chunk of the wave file. We use a key stream to skip a number of samples, grab one sample, put one bit of the message into the lowest bit of the sample, and write the changed unit to the destination stream. When the entire message has been added like that, we copy the rest of the samples.

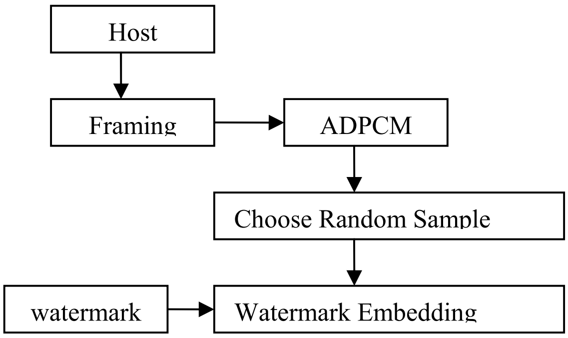

Fig. 1 Proposed watermark embedding algorithm.

2.1.1 Watermark Embedding Steps

The following steps give the detailed watermark embedding process:

1. Audio file is partitioned into frames of 90 ms duration.
2. ADPCM is performed on the wave audio file, which achieves 4:1 compression ratio with less loss as the sampling frequency increases.
3. Then a random sample is selected using the key and watermark is added into the lowest bit of the selected sample by the following procedure:

   a) Read one byte of the message stream.
   b) For each bit in (message)
      i) Read a byte from the key
      ii) Skip a couple of samples.
      iii) Read one sample from the wave Stream.
      iv) Get the next bit from the current message byte.
      v) Place it in the last bit of the sample.

2.2 Watermark Extraction

Again, we use the key stream to locate the right samples, just as we did while adding the Watermark. Then we read the last bit of the sample and shift it into the current byte of the message. When the byte is complete, we write it into the message stream and continue with the next one. [5, 6]

2.2.1 Watermark Extraction Steps

The following steps give the detailed watermark extraction process:

1. Firstly key is used to locate the sample in which watermark is added.
   a) Read a byte from the key.
   b) Skip a couple of samples.
   c) Read one sample from the wave stream.
2. Then the watermark bit is extracted from the lowest bit of the sample and put in watermark array.
3. After the watermark is extracted inverse ADPCM is performed.

3. Experimental Results and Evaluation

In order to evaluate the performance of the proposed watermarking algorithm, the following experiments were considered. The audio PSNR tool is used and the original code is modified to add the proposed watermarking algorithm in. The watermark was embedded into 5 different types of audio signals. Music 1 and Music2 are classic music, Music 3 and Music 4 are pop music and Music 5 is a speech signal.

TABLE 1: Showing SNR, PSNR and BER

Audio	SNR	PSNR	BER
Music1	32.11	61.75	0
Music2	32.08	61.70	0
Music3	31.90	61.35	0
Music4	31.89	61.33	0
Music5	31.87	61.30	0

3.1 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. Here all the BER values are 0.

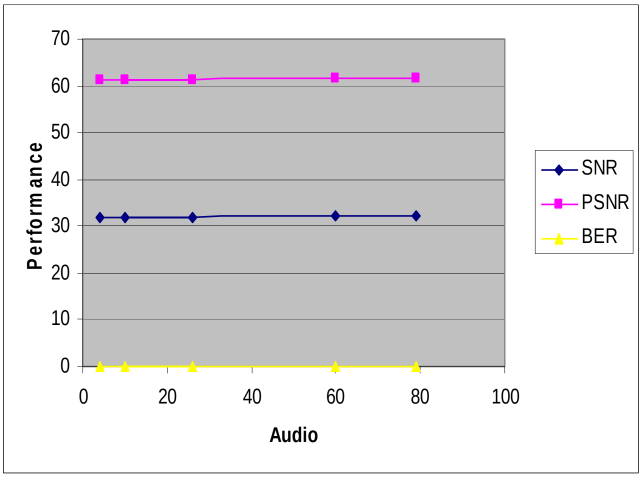

Fig 2: Performance for different audio signals

The Figure 2 summarizes the results of this experimental test. It shows the watermarking algorithm's performance is stable for different types of audio signals. Note BER curve is not visible in the figure, because all the values are zero.

4. Conclusion

In this paper, a random sample audio watermarking for compressed wave files has been proposed. Watermark is embedded in the wave file after compressing the wave file. The experimental results show that this watermarking algorithm gives better results with bit error rate being zero under normal watermark embedding and extraction conditions as shown in Table 1. Moreover the Random Sample audio watermarking based method allows a perfect watermark recovery without much degradation to the original audio signal as indicated by high PSNR values.

5. Future Scope

The study may be carried on in future with following direction: -

1. In this technique we can embedded only text into the audio signal.
2. This technique can be extended to embed an image watermark into an audio signal instead of text.
3. It allows a perfect watermark recovery without much degradation to the original audio signal as indicated by high PSNR values.
4. Its results are improved and efficient.
5. The experimental results show that this watermarking algorithm gives better results with bit error rate being zero under normal watermark embedding and extraction conditions.

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