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Optimized Approach for Authentication In Audio Signal By Wavelet Packets Transformation Technique

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Abstract

In our proposed framework we present a novel watermarking procedure to embed for copyright protection and authentication into digital audio by directly changing the audio samples then after modifying the audio signals. The modified audio signals are divided into no. of samples each sample is decomposed adaptively by the method of novel Empirical Mode Decomposition (EMD).those decomposed samples after decomposition called as a Intrinsic Mode Functions (IMFs),In this Intrinsic Mode Function the low frequency mode table under different attacks is presented and then after audio perceptual quality of the original audio signal is preserved. Finally in our proposed algorithm we show the robustness efficiency of the hidden watermark for additive noise re-quantization, MP3 compression,, filtering, cropping and re-sampling. A comparison analysis shows that our proposed framework has high end performance than the other watermarking schemes proposed recently in the literature.

KEYWORDS: *Audio watermarking, SC, Authentication, IMF*

1. INTRODUCTION

In digital media the embedding of watermarking in audio is for copyrights protection and authentication .digital media by embedding a watermark in the original audio signal. Main requirements of digital audio watermarking are imperceptibility, robustness and data capacity. Digital watermarking has been proposed as a means to identify the owner or distributor of digital data. Watermarking is the process of encoding hidden copyright information in digital data by making small medications to the data samples. Unlike encryption, watermarking does not restrict access to the data. Once encrypted data is decrypted, the media is no longer protected. A watermark is designed to permanently reside in the host data. When the ownership of a digital work is in question, the information can be extracted to completely characterize the owner.

An effective audio watermarking scheme must satisfy the following basic requirements:

A. Imperceptibility: The quality of the audio should be retained after adding the watermark. Imperceptibility can be evaluated using both objective and subjective measures.

B. Security: In Watermarking audio signals should not reveal any clues about the watermarks in them. Also, the security of the watermarking procedure must depend on secret keys, but not on the secrecy of the watermarking algorithm.

C. Robustness: After watermarking extraction Ability to extract a watermark from a watermarked audio signal after various signals processing attacks.

D. Payload: The amount of data that can be embedded into the original audio signal without losing imperceptibility. For audio signals, data payload refers to the number of watermark data bits that may be reliably embedded within a original signal per unit of time, usually the extracted information can be calculated by the BER (Bit Error Rate)

Previously different methods have been proposed for audio water marking but some problems are arises like robustness, Imperceptibility and data capacity .Now we are proposed a new algorithm in audio watermarking for the copyright protection. That is Empirical Mode Decomposition (EMD). EMD - based time-frequency analysis, called Hilbert-Huang Transform (HHT), this is only one of many applications made possible by EMD. The final result and ideas in time domain applications using EMD apply to two-dimensional signals, such as images, as well as audio. EMD decomposes the spatial frequency components into a set of IMFs (Intrinsic Mode Functions) where the highest spatial frequency component of each spatial position is in the first IMF and the second highest spatial frequency component of each spatial position is in the second IMF, etc. An IMF is defined as a function in which the number of extrema points and the number of zero crossings are the same or differ by one [2]. In the two-dimensional case this demand is relaxed. The upper and lower envelope of the IMF are symmetric with respect to the local mean, which is used to define the IMF instead of the number of extrema points and zero crossings. In two dimensions there are many possibilities to define extrema, each one yielding a different decomposition. In this work we simply extract the extrema points by comparing the candidate data point with its nearest 8-connected neighbors. Approaches, Empirical Mode Decomposition are totally data-driven method that recursively breaks down any signal into a reduced number of zero-mean with symmetric envelopes “Intrinsic Mode Functions” (IMFs).The starting of decomposition from finer scales to coarser ones. Any signal $x(t)$ is expanded by EMD as follows:

$$x(t) = \sum_{i=1}^{z} IMF_i\,(t) + r_z(t)\ldots\ldots\ldots\ldots \text{(1)}$$

Where z is the number of IMFs and r_z(t) denotes the final residual. The IMFs are orthogonal to each other, and total IMFs are near to zero means. The No. of Extrema is decreased when ever mode is going from one to next, and the whole decomposition is guaranteed to be completed with a finite number of modes. The IMFs are fully described by their local extrema and thus can be recovered using these extrema [7], [8]. Low frequency components which higher order IMFs are signal dominated [9] and then their alteration modes can lead to degradation of the signal. As result, these modes can be taken as to be good locations for watermark placement for better robustness. Some results have been visually in recently [10], [11] showing the interest of EMD for audio watermarking. The EMD algorithm is combined with Pulse Code Modulation (PCM) and the watermark is inserted in the sub-bands of an audio which is in transform domain.

Sync-code	water mark bits	sync-

Figure 1: data stretcher

Thus the method supposes that mean value of Pulse Code Modulation audio signal may no longer be zero. As well as stated by the authors, the method is not robust to attacks such as filtering ,cropping, and no comparison to watermarking schemes reported recently . Our proposed watermarking is only based on EMD method and without domain transforms, we choosing method a watermarking technique in the category of Quantization Index Modulation (QIM) for the reason to its good robustness and blind nature. The Parameters of QIM are selected to guarantee that the embedded watermark is in the last IMF is inaudible. Finally the watermark is associated with a synchronization code to facilitate its location.

2. BACKGROUND

Audio watermarking is a technique that embeds information with specific meaning into the host media without perceptible interference to the original quality. As a complement to conventional encryption techniques, watermarking provides powerful tools for protecting the copyright of audio works and has become an active research area in recent years. Besides being imperceptible to human ears, the watermark must also be able to withstand audio signal processing and time-domain synchronization attacks. Synchronization attacks are a serious problem to any watermarking schemes.

Audio processing such as random cropping and time-scale modification (TSM) causes displacement between embedding and detection in the time domain and is hence difficult for watermark to survive. Generally speaking, synchronization problem can be alleviated by the following methods: exhaustive search , synchronization pattern , invariant watermark , and implicit synchronization . TSM is a serious attack to audio watermarking, very few algorithms can effectively resist this kind of synchronization attack. According to the Secure Digital Music Initiative (SDMI) Phase-II robustness test requirement.

The most important requirements of a data embedding system are transparency and robustness. Transparency means that there is no perceptual difference between the original and the modified host media. Data embedding techniques usually exploit irrelevancies in digital representation to assure transparency. For audio data embedding, the masking phenomenon is usually exploited to assure that the distortion due to data embedding is imperceptible. Robustness refers to the property that the embedded data should remain in the host media regardless of the signal processing operations that the signal may undergo.

2.1 Time-Scale Invariant Audio Data Embedding

In this work, we propose a novel algorithm for embedding data in audio by changing the interval length of certain segments of the audio signal. The algorithm is invariant after TSM, time shift, and time cropping. We proposed a set of encoding and decoding techniques to survive the common mp3 compression. The embedding rate of the algorithm is above 20 bps.However, as discussed for practical reasons, repetition coding is used and the effective embedding rate is 4–8 bps. The quality of the output is very high and it is indistinguishable from the original signal. The proposed technique is suitable for applications like broadcast monitoring, where the embedded data are information relevant to host signal and used for several purposes, for example, tracking the use of the signal, providing statistical data collection, and analyzing the broadcast content. The required smooth behavior does not occur often in audio signals except for a set of single-instrument audio like a Piano and a flute. For other composite audio signals, this requirement is hardly fulfilled. This greatly reduces the embedding rate if the original signal is used directly in embedding. Moreover, even if such a behavior exists, it is very vulnerable to distortion after compression.

Hence, the direct audio signal is not a good candidate for data embedding. In our implementation, we used a hybrid of orthogonal and nonorthogonal decompositions. The orthogonal decomposition is an exact (nonredundant) representation of the signal. It involves sub sampling after each decomposition stage. Hence, the approximation signal is not smoother than the original because of the frequency spread after sub sampling. If a modification is done in the transform domain, then it is preserved after the inverse and the forward transform because it is nonredundant. On the other hand, nonorthogonal wavelet decomposition does not involve sub sampling after filtering at each scale. For our particular purpose, this decomposition has a two-fold advantage. First, the lengths are preserved so that the lengths at any scale are in one-to-one correspondence with the lengths at the finest scale. The second advantage is that the approximation signal at coarser scales is smoother than the signal at a finer scale.

2.2 Localized Audio Watermarking Technique Robust Against Time-Scale Modification

Synchronization attacks like random cropping and time-scale modification are very challenging problems to audio Watermarking techniques. To combat these attacks, a novel content- dependent localized robust audio watermarking scheme is proposed. The basic idea is to first select steady high-energy local regions that represent music edges like note attacks, transitions or drum sounds by using different methods, then embed the watermark in these regions. Such regions are of great importance to the understanding of music and will not be changed much for maintaining high auditory quality. In this way, the embedded watermark has the potential to escape all kinds of distortions. Experimental results show strong robustness against common audio signal processing, time-domain synchronization attacks, and most distortions introduced in Stir mark for Audio.

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we obtained high robustness against common audio signal processing and synchronization attacks such as random cropping and TSM. The selection of the embedding regions is the most important step in this algorithm as to what extent these regions can be invariant against attacks like TSM directly determines how robust this algorithm is. It should be noted that this method has its inherent limitation. Although it is suitable for most modern music with obvious rhythm, it does not work well on jazz and some classical music without apparent rhythm; in this circumstance, there are not obvious peaks on the original waveform, envelope, or the d3 sub band.

3. LITERATURE SURVEY

Audio Watermarking in Images using Wavelet Transform The rapid growth of digital media and communication network has highlighted the need for Intellectual Property Rights (IRP) protection technology for digital multimedia. Watermarking of multimedia data has become a hotspot for research in recent years. Watermarking can be used to identify the owners, license information, or other information related to the digital object carrying the watermark. Watermarks can provide the mechanism for determining if a particular work has been tampered with or copied illegally. In this thesis, we present a novel algorithm for robust audio watermarking in image using wavelet transform based on image entropy. The motivation of choosing image as a cover is driven by the fact that human visual system is less sensitive than human auditory system thus an image provides better masking effect. The algorithm is based on decomposition of images using Hear wavelet basis. The hidden data can be recovered reliably under certain attacks such as cropping, compression, noise effect, geometrical attacks and contrast enhancement. As a necessary background, a literature survey of the watermarking techniques is presented. The last part of the thesis analyzes the watermarking results of wavelet-based watermarking technique on different images and audio samples, using various quality assessment metrics.

This study presents a novel scheme of watermarking of digital images for copyright protection and authentication. In this study we proposed a method of embedding owner's speech signal. Speech being a biometric data, the watermark signal in this method is expected to be more meaningful and has closer correlation with copyright holder. The main issue of concern here is the capacity because the speech data has large number of samples. Here, speech samples are imperceptibly inserted into one of the frequency band of wavelet transform of image. Applications for such a speech hiding scheme include copy protection, authentication and covert communication. The proposed technique uses the wavelet transformation domain to embed the data so as to exploit the advantages of wavelet transformation being resistant to frequency attacks. The algorithm uses simple substitution

3.1 Motivation

Watermarking an audio file in an image is motivated by several features. First and foremost it allows secret transfer of an audio file and prevents illegal transfer of multimedia content providing copyright protection. Moreover, HAS ismore sensitive than HVS [9], therefore transferring a secret audio file using image file as a medium instead of an audio file exploits the feature of HVS weakness and achieves a better degree of concealment. Robustness and transparency of digital watermarking are two basic and most important requirements of the digital image watermarking technique. In order to make the watermark robust, its strength is to be increased and it is to be placed in the significant parts of the image that can stand low-pass filtering or noise, however, that makes it less transparent and results in the degradation of the host image. Therefore the requirements of robustness and transparency are contradictive to each other. The need of the hour is to strike a balance between both the parameters to achieve maximum of both without trading off either of them.

4. PROPOSED WATERMARKING ALGORITHM

The basic idea of the proposed watermarking system is to hide the data into the original audio signal a watermark (secret data) with a Synchronized Code (SC) in the time domain format .The input signal which is original audio signal is first segmented into samples after that algorithm EMD is conducted on every samples to extract the

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associated IMFs (Fig. 1).Then all the samples are converted into binary data sequence consisted of SCs and informative watermark bits (Fig. 2) is embedded in the extreme of a set of consecutive last-IMFs. All bit (1 or 0) is inserted per extreme.

The number of IMFs and their number of extrema depend on the amount of data of each sample; the number of bits is to be embedded varies from last-IMF of one frame to the following. Watermark and Synchronized Code (SC) are not all embedded in extrema of last IMF of only one samples. In general the number of extrema per last-IMF (one sample) is very small compared to length of the binary sequence to be embedded in audio signal.

This also depends on the length of the sample. If we design by the M_1 and M_2 the numbers of bits of SC and watermark respectively, the length of binary sequence to be embedded is equal to $2M_1 + M_2$ Then, these bits are spread out on several last-IMFs in extrema of the consecutive samples. This sequence of bits is embedded times in farther.

Finally, inverse transformation function is applied to the modified extrema to recover the watermarked audio signal by superposition of the IMFs of each sample followed by the concatenation of the sample (Fig. 3). For data extraction presses, the watermarked audio signal is split into the no. of samples and EMD applied to each sample (Fig.1).after that covert into the Binary data sequences are extracted from each last-IMF by searching for SCs (Fig. 5).because we are embedding the data into last-IMF .Fig. 6 shows that the last IMF before and after watermarking. This figure shows that there is small difference in terms of amplitudes between the two modes. The EMD being full data adaptive, thus it is very necessary to guarantee that the number of IMFs will be same as before embedding the data and after embedding the watermark data (Fig. 1),(Fig. 4) respectively However , if the numbers of IMFs are totally different form original signal, there is no guarantee that the last IMF always contains the watermark(secret data) information to be extracted. To minimization of the problem, the sifting of the watermarked signal is forced to extract the same number of IMFs as before watermarking. The proposed EMD watermarking scheme is blind, that is, the original signal(host signal) is not required for watermark extraction. These are the basically 3 steps those are

1. Synchronization of code
2. Watermark embedding
3. Watermark extraction

Synchronization of code:

For the hidden the secret data in the original audio signal synchronization code is used. This code is unaffected (non-degradation) by shifting attacks and cropping [4]. Let P be the original SC and Q be an unknown sequence of the same length.

If only the number of different bits between P and Q, when compared bit by bit, is less or equal than to a predefined threshold τ.When Sequence Q is considered as a SC

BLOCK DIAGRAM

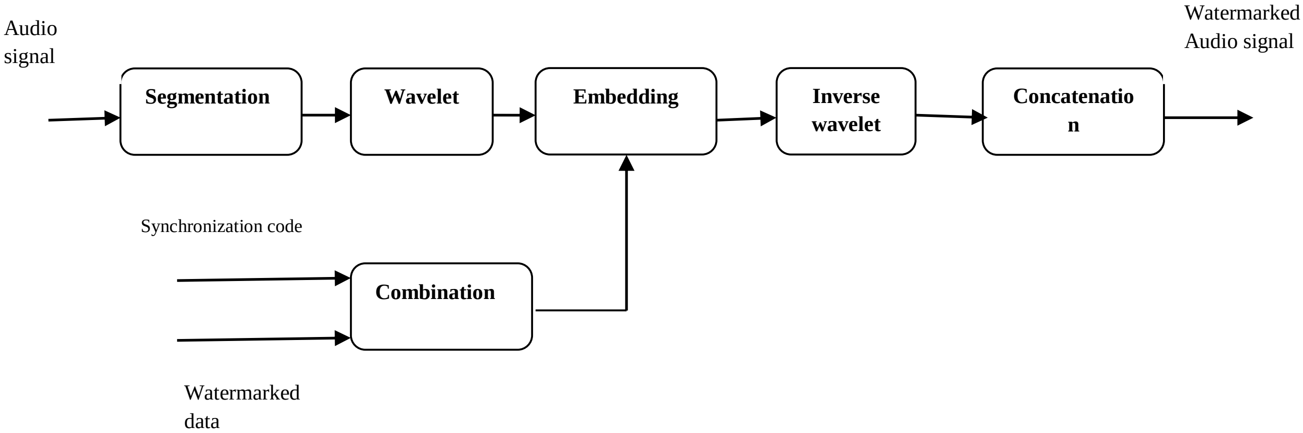

Figure 2: watermark embedded presses

AUDIO SIGNALS EMBEBEDED:

In the watermark embedding process Synchronization of code are efficiently combined with watermark bits from a obtained binary sequence Before embedding, then after it is denoted by $n_j \in \{0\ 1\}$ bit of watermark (Fig. 2). Basics of our watermark embedding are shown in Fig. 3 and detailed as follows:

Steps to be embedded:

1. In step Divide the original host signals to the no. of samples
2. Each sample is decomposed into IMFs (intrinsic mode function)
3. Embed T times the binary sequence $\{n_i\}$ into extrema of the last IMF (IMF_z) by QIM

$$e_j^* = \begin{cases} \left\lfloor \frac{e_j}{H} \right\rfloor * H + sgn\left(\frac{3H}{4}\right) & if \quad n_j = 1 \\ \left\lfloor \frac{e_j}{H} \right\rfloor * H + sgn\left(\frac{H}{4}\right) & if \quad n_j = 0 \end{cases}$$

Where e_j and e_j^* are the extrema of IMF_z of the original host audio signal and the watermarked signal respectively. If "Sgn" function is equal to "[+] " then it is a maxima, and equal to "[-] " then it is a minimal. Denotes the floor function, and S denotes the embedding strength chosen to maintain the inaudibility constraint

4. Reconstruct the samples by using inverse EMD modified IMF_z and concatenate the watermarked frames to retrieve the watermarked signal

STEPS TO WATER MARK EXTRACTION

Host signal is splitted into samples and EMD is performed on each one as in embedding for the watermark extraction .extract binary data using rule given by (3).then find out the SCs in the extracted data. This procedure is continuously repeated by shifting the selected segment one sample at time until a SC is found. With the position of SC determined, after that we can extract the hidden data i.e. information bits, which given as

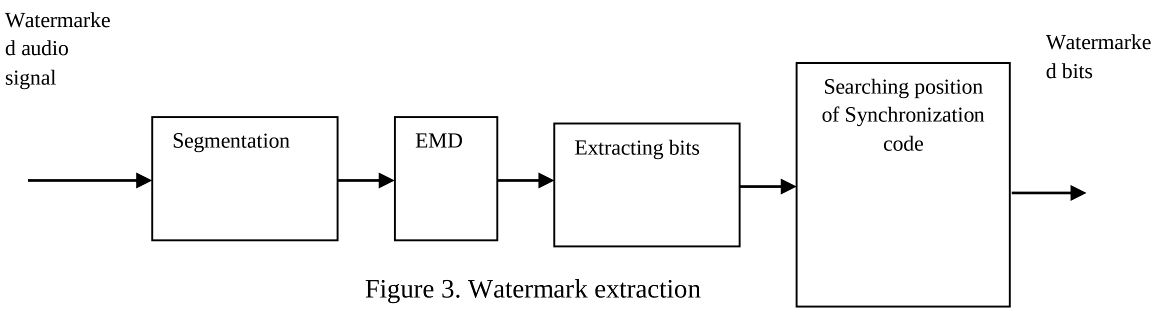

Figure 3. Watermark extraction

1. Divided the watermarked signal into no. of samples.
2. Decompose each and every sample into IMFs.
3. Extract the extrema e_j^* of IMF_z
4. Extract n_j^* from e_j^* using the following rule [3]

$$e_j^* = \begin{cases} 1 \ \ if \ e_j^* - \left\lfloor e_j^*/H \right\rfloor . \mathrm{H} \geq \mathrm{sgn}(\mathrm{H}/2) \\ 0 \ \ if \ e_j^* - \left\lfloor e_j^*/H \right\rfloor . \mathrm{H} \leq \mathrm{sgn}(\mathrm{H}/2) \end{cases}$$

5. Set the start index of the extracted data, y, to I=1 and select samples L=N1 (sliding window size).
6. Evaluate the similarity between the extracted segment V= y(I:L) and U bit by bit. If the similarity value is$\geq \tau$, then is taken as the SC and go to Step 8. Otherwise proceed to the next step.
7. Increase by 1 and slide the window to the next samples and repeat Step 6.
8. Evaluate the similarity between the second extracted segments $V' = y(I + N_1 + N_2 : I + 2N_1 + N_2)$ and bit by bit.
9. $I \leftarrow I + N_1 + N_2$ of the new value is equal to sequence length of bits, go to Step 10 else repeat Step 7.

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10. Extract the P watermarks and make comparison bit by bit between these marks, for correction, and finally extract the desired watermark. Watermarking embedding and extraction processes are summarized.

5. SIMULATION RESULTS

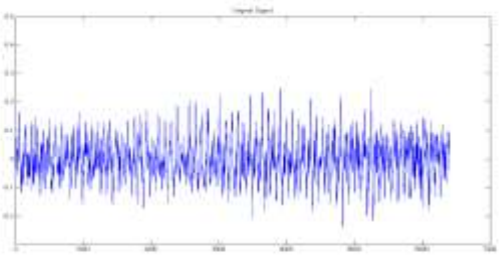

Figure 4: Original Signal

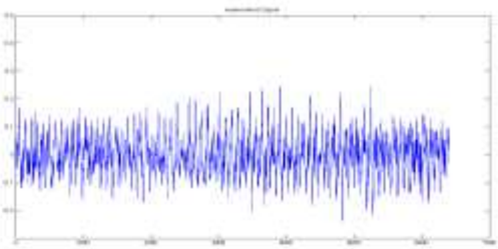

Figure 5: Watermarked Signal

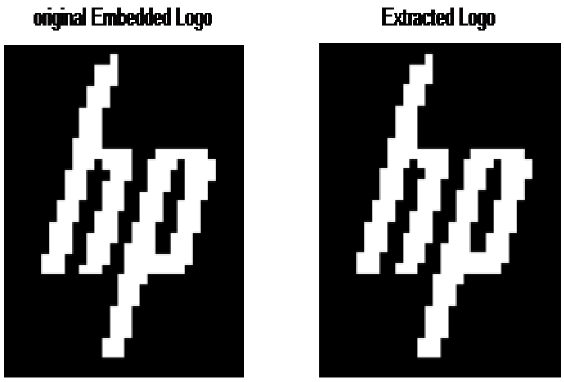

Figure 6: Watermarked image and Extracted image

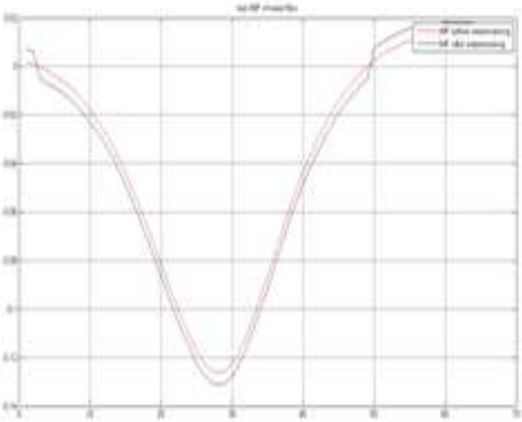

Figure 7: IMF of wave files

EXTENSION

The proposed algorithm is for audio watermarking based on IMF (Intrinsic mode function) of Empirical mode decomposition. The proposed method shows good performance and robustness against attacks such as audio compression, quantization, attacks etc but it takes lot of time to complete the task. In extension we present a work which is on advance wavelet packet decomposition which uses recursion process to complete the task with in target time and the extension work shows the better performance.

6. CONCLUSION

In our proposed framework Audio signals which are used for watermarking technique are divided into number of samples each sample is further decomposed adaptively by the method of new Empirical Mode Decomposition (EMD).those decomposed samples are called as a Intrinsic Mode Functions (IMFs), In this Intrinsic Mode Function the low frequency table under different attacks is presented and then after audio perceptual quality of the original audio signal is preserved. Finally in our proposed algorithm we show the robustness efficiency of the hidden watermark for additive noise re-quantization, MP3 compression,, filtering, cropping and re-sampling. . A comparison analysis shows that our proposed framework has high end performance than the other watermarking schemes proposed recently in the literature.

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