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EMPIRICAL MODE DECOMPOSITION IN AUDIO WATERMARKING BY USING WAVELET METHOD

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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.

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 is expanded by EMD as follows:

………… (1)

Where z is the number of IMFs and (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.

Wavelet transform

Wavelet transform [3] offers effective time-frequency representation of signals. All basis functions are formed by shifting and scaling of "mother" wavelet function $\psi(t) \in L^2(R)$:

$$\psi_{m,n}(t) = 2^{-\frac{m}{2}} \psi\left(2^{-m} t - n\right) \quad m, n \in Z$$

Signal $f(t) \in L^2(R)$ can be then represented as

$$f(t) = \sum_m \sum_n d_{m,n} \psi_{m,n}(t)$$

where $d_{m,n}$ are spectral wavelet coefficients

$$d_{m,n} = \left\langle f(t), \psi_{m,n}(t) \right\rangle$$

For discrete signals $f(k) \in L^2(Z)$ hold similar results and corresponding transform is called Discrete Wavelet Transform (DWT).

In this article we use orthogonal Haar wavelet transform, where:

$$\psi_{Haar}(t) = \begin{cases} 1 & for \ 0 < t < 0.5 \\ -1 & for \, 0.5 < t < -1 \\ 0 & otherwise \end{cases}$$

A limit of wavelet approach is that the basis functions are fixed, and thus they do not necessarily match all real signals.

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 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 and the numbers of bits of SC and watermark respectively, the length of binary sequence to be embedded is equal to 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 .When Sequence Q is considered as a SC.]

WAVELET PACKETS:

Originally known as Optimal Subband Tree Structuring (SB-TS) also called Wavelet Packet Decomposition (WPD) (sometimes known as just Wavelet Packets or Subband Tree) is a wavelet transform where the discrete-time (sampled) signal is passed through more filters than the discrete wavelet transform(DWT).

In the DWT, each level is calculated by passing only the previous wavelet approximation coefficients (cA_j) through discrete-time low and high pass quadrature mirror filters.[1] However in the WPD, both the detail (cD_j (in the 1-D case), cH_j, cV_j, cD_j (in the 2-D case)) and approximation coefficients are decomposed to create the full binary tree.

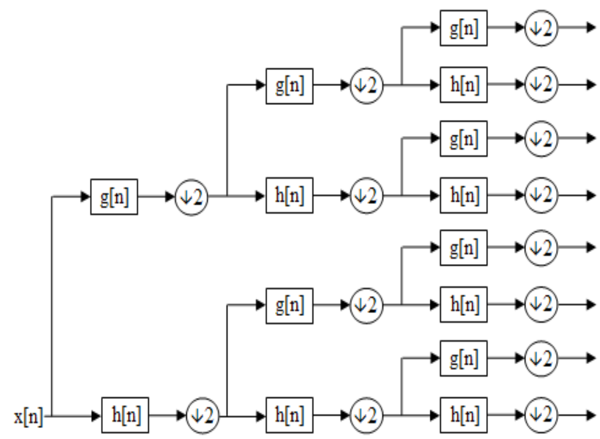

Fig: Wavelet Packet decomposition over 3 levels.

For n levels of decomposition the WPD produces 2^n different sets of coefficients (or nodes) as opposed to (3n + 1) sets for the DWT. However, due to the down sampling process the overall number of coefficients is still the same and there is no redundancy.

From the point of view of compression, the standard wavelet transform may not produce the best result, since it is limited to wavelet bases that increase by a power of two towards the low frequencies. It could be that another combination of bases produce a more desirable representation for a particular signal. The best basis algorithm by Coifman and Wicker hauser finds a set of bases that provide the most desirable representation of the data relative to a particular cost function (e.g.entropy).

There were relevant studies in signal processing and communications fields to address the selection of subband trees (orthogonal basis) of various kinds, e.g. regular, dyadic, and irregular, with respect to performance metrics of interest including energy compaction (entropy), subband correlations and others.Discrete wavelet transform theory (continuous in the variable(s)) offers an approximation to transform discrete (sampled) signals. In contrast, the discrete subband transform theory provides a perfect representation of discrete signals.

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 {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:
BLOCK DIAGRAM

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 { } into extrema of the last IMF ( ) by QIM

=

Where and are the extrema of 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 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 given below

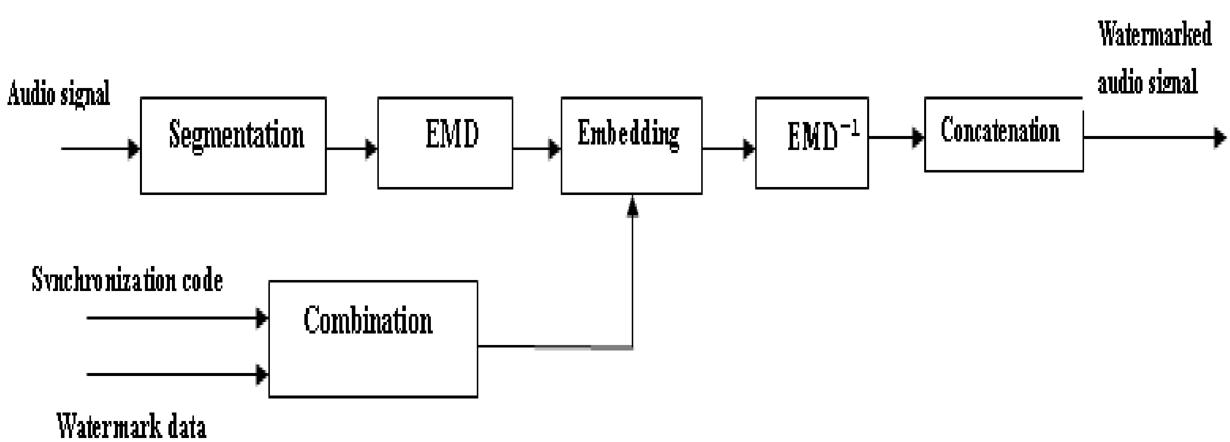

Fig.2 watermark embedded presses

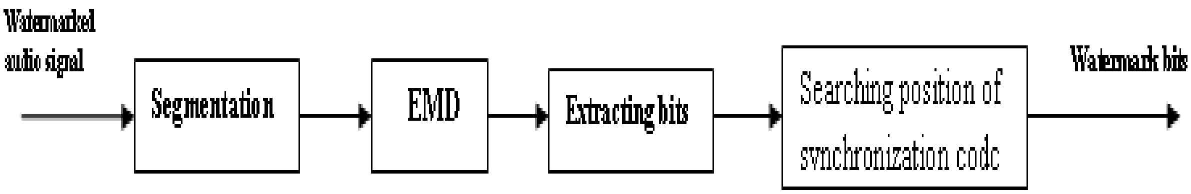

Fig. 3. Watermark extraction

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

=

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 , 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

   and bit by bit.
9. of the new value is equal to sequence length of bits, go to Step 10 else repeat Step 7.
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.

SIMULATION RESULTS

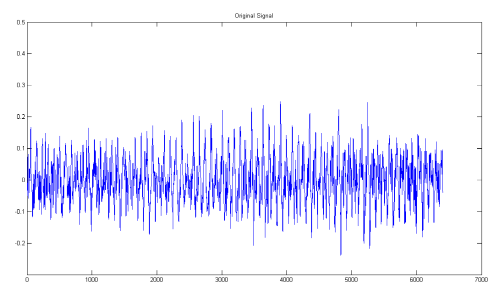

Figure: Original Signal

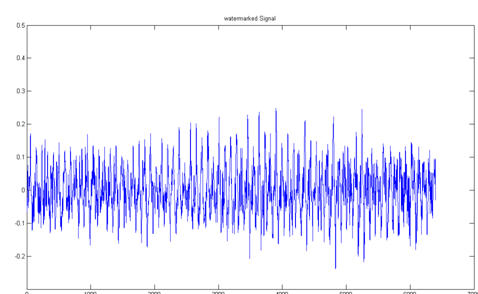

Figure : Watermarked Signal

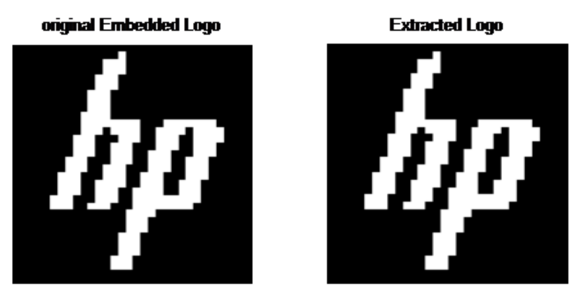

Figure: Watermarked image and Extracted image

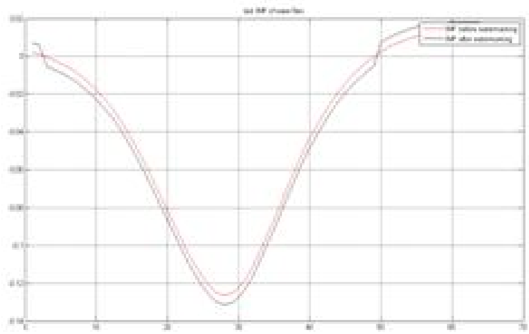

Figure: IMF of wave files

Different attacks	BER(%)	Extracted watermark
No attack	0	
Gaussian attack	0	
Filtering attack	18.5547	
Cropping Attack	0.1953	

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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