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Enhancing Emd Watermarking Scheme For Copyright Protection Of Digital Audio Data

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ABSTRACT

In modern communications, network security is becoming a vitally important task for exchanging data on social sites. The audio data that are uploaded in social sites can be edited by unauthorised person. So we need an authorised protection to give our signature to that data. By avoiding this unauthorised scenario, we propose a digital audio watermarking algorithm for the protection of our audio file from different editing attacks. Two issues are mainly obtained on the basis of attacks, first is to resist the repeated attacks to the similar regions and other is the selection of more appropriate region for the embedding of watermark. These two issues are intelligently avoiding this algorithm, which based on the empirical mode decomposition. Watermark embedding and its extraction is the main steps involved in this algorithm. The performance of this algorithm is evaluated with different types of audio signals. The obtained results permit to assess the achieved efficiency in terms of accuracy in detecting watermark from watermarked audio signal(bit error rate (BER)).

General Terms

Embedding, Extraction, Windowing, Digitalization

Keywords

Quantization, Synchronization code(SC), Empirical mode decomposition(EMD)

1. INTRODUCTION

Digital audio watermarking provides a great deal of attention for copyright protection of digital media by embedding a watermark in the original audio data. Main advantages of digital audio watermarking are imperceptibility; robustness and data capacity. The watermark embedding within the host audio data maintain its audio quality more precisely. Finally, the watermark must be easy to extract to prove its signature. To achieve these qualities, seeking new watermarking schemes is a very challenging problem. Different watermarking techniques have been proposed, which provide high complexities.

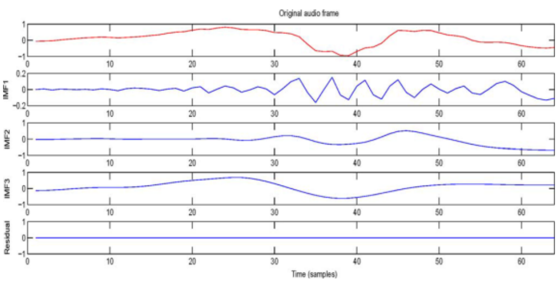

Fig 1. Decomposition of an audio frame by EMD

A limited transmission bit rate is the main problem for some cases. To improve this, wavelets domain have been proposed. It also have a limitation that is, its basis functions are fixed, and thus they cannot match all real signals. To overcome this limitation, a signal decomposition method referred to as Empirical Mode Decomposition (EMD) is introduced. This is mainly used for analyzing non-stationary signals derived or not from linear systems in totally adaptive way. Not a priori choice of filters or basic functions is the main advantage of this system. EMD is fully data-driven method that recursively breaks down any signal in to a reduced number of zero-mean with symmetric envelopes AM-FM components called Intrinsic Mode Functions (IMFs). The decomposition starts from finer scales to coarser scales ones. Any signal is expanded by EMD as follows:

$$X(t) = \Sigma \, IMFj(t)+rc(t)$$

Where C is the number of IMFs and rc(t) is the final residual value .The IMFs are nearly orthogonal and nearly zero mean values. The number of extrema is decreased and whole decomposition is guaranteed to be completed with a finite number of modes. Higher order IMFs are signal dominated and their alteration leads to the degradation of the signal. So these modes can be considered to be good locations for watermark embed. The EMD is combined with Pulse Code Modulation(PCM) form a system, but this combination provide a failure because PCM audio signal may no longer be zero. The method is not robust to attacks such as band-pass filtering and cropping, Another strategy is EMD is associated with Hilbert transform and the watermark is embedded into the low frequency IMF or highest energy values. IMF with highest energy can be a high frequency mode and thus it is not robust to attacks .Watermarks embed into lower order IMFs (high frequency) are not good to attacks. The watermark bits are usually embedded in the low frequency components of the host signal.

The proposed watermarking is only based on EMD and without domain transform. We choose a watermarking technique in the category of Quantization Index Modulation (QIM) due to its good robustness and blind nature. Parameters of QIM guarantee that the embedded watermark in the last IMF is more appropriate. The synchronization code is to locate the watermark bit in the embedded audio data. The low cost in searching synchronization codes is the main advantage of this algorithm. Audio signal is segmented into frames and them is decomposed into IMFs. Watermarked bits are inserted into the extrema of the last IMF which guaranteed its quality. Process like MP3 compression, quantization, cropping and filtering results various data loss due to various attacks. This method has data payload and high performance against MP3 compression.

2. RELATED WORKS

2.1 Echo-Hiding

This consists of a technique for embedding the watermark into the host or cover audio by taking advantage of the human auditory system's inability to detect certain very short echoes in a sound. As a simple explanation, a signal might be broken into non-overlapping frames of user-defined length before the encoder adds a delayed version of a candidate frame (or even just some components from the frame), delayed by, say 0.005s to represent a '0' and 0.008s to represent a '1' bit. In theory, even neglecting to add a delay for a '0' bit would be potentially useful but more likely to increase incorrect detection as there will obviously be times when an echo is either present or absent in the original audio and this might be confused with deliberately controlled echo.

2.2 Phase Coding Techniques

In this, substituting the phase of one piece of audio by the phase of another, or simply by altering the phase of the cover audio to represent some binary value. Embedding a watermark by altering the phase of components within the cover signal can be troublesome as, while the human auditory system is generally not able to detect absolute phase, any sharp alteration of phase from one frame to the another results in audible phase discontinuities.

2.3 Amplitude Masking

A process of embedding the watermark into the host audio in the form of an additional audio signal at very weak power. This technique utilizes the known masking effect of sounds on other sounds. Masking of one sound to another is dependent on various parameters including the frequency, distance between the components, the amplitude or magnitude difference between the two components and the individual magnitude of the components themselves. Low-powered components may simply be too quiet to mask another component. Conversely components may be below the threshold of hearing and therefore the presence of other components is irrelevant.

3. SYSTEM OVERVIEW

3.1 Proposed method

The proposed watermarking algorithm performs an idea of hiding a watermark bit together with a synchronization code to the host audio signal in the time domain manner. The first steps involve segmentation of audio file then empirical mode decomposition is applied on every frame to extract the associated IMFs. Embedded the digitalized watermarked bit together with SC to the last extrema of the IMFs. A bit 0 or 1 is inserted per extrema. Since the number of IMFs and their number of extrema depends on the data of each frame. The number of bits to be embedded in the extrema varies from last-IMF of one frame to the following. Digitalized watermark bit and following SCs are not all embedded in extrema of last IMF or zero mean of only one frame.

The binary sequence to be embedded is larger than the number of extrema per last IMF. This a depends on the length of the frame also. If N1 and N2 are the numbers of bits of Synchronization Code and watermark bit respectively, the bit length of binary sequence to be embedded to the host signal is equal to 2N1+N2 .Thus, these 2N1+N2 bits are spread out on several last-IMFs or extrema of the consecutive frames, which having zero mean. Further, this sequence of 2N1+N2 bits is embedded P times. Then take the inverse transform (EMD inverse) of the modified extrema of the watermarked audio signal by superposition of the IMFs of each frame, and concatenate each frame.

Next process is the extraction of watermark bit. Initially the watermarked audio signal is split into frames and EMD applied to each frame. Search the SCs using algorithm and which is extracted from each last IMF. The last IMF before and after watermarking produce little difference in its amplitudes. EMD being fully data driven, thus the number of IMFs will be same before and after embedding the watermark. In fact, if the numbers of IMFs are different, we cannot get any guarantee that the last IMF always contains the hiding watermarking data .The proposed watermarking algorithm is blind, that is, the host signal is not used for watermark extraction. Overview of the proposed watermarking algorithm is detailed as follows:

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3.2 Synchronization Code

A synchronization code is used to locate the embedding position of the hidden watermark bits in the host signal. This code is unaffected by any cropping and shifting attacks during signal transmission. Let U be the original SC and V be an unknown sequence of data of the same length. Sequence V is considered as a SC, then compared bit by bit to the length of U which is less or equal than to a predefined threshold Ŧ.

Sync-code	Watermark bits	Sync-code

Fig.2 Data structure (mi)

3.3 Watermark Embedding

Before embedding to the host audio , SCs are combined with watermark bits to form a binary sequence denoted form a £ {0, 1}, it bit of watermark. Basics of watermark embedding are shown below

Step 1: Fragmented the original audio signal into frames.

Step 2: Decompose each frame into IMFs.

Step 3: Embed times the binary sequence into extrema of the last IMF by QIM.

$E=|e_i/s|.S+sgn(3s/4)$; if mi=1

$E=|e_i/s|.S+sgn(S/4)$;if mi=0

where ei and E are the extrema of the host audio signal and the watermarked audio respectively sgn function is equal to "+" if is a maxima, and "_ " if it is a minimal.'|| denotes the floor function, and S denotes the embedding strength.

Step 4: Reconstruct the frame (EMD inverse) using modified and concatenate the watermarked frames to retrieve the watermarked signal.

3.4 Watermark Extraction

In the watermark extraction process, host signal is fragmented into frames and EMD is performed on each frame. For extracting thet binary data, search the SCs to the decomposed frame. This procedure is repeated by shifting or increasing the count in the algorithm. Examin the selected segment (window) one sample at a time until a SC is found. However the position of SC is determined, and then extract the hidden information bits or watermarked data, which follow the SC. Let y= {Mi} denote the binary data to be extracted. For locating the embedded watermark search the SCs in the sequence {Mi} bit by bit.

The extraction is performed without the original audio signal. Basic steps involved in the watermarking using extraction, are given as follows:

Step 1:Segmented the watermarked signal into frames.

Step 2: Decompose each frame into IMFs.

Step 3: Extract the extrema{E} of IMFc.

Step 4: Extract using the following equation

Mi= {1 if $E-|E/S|.S \geq sng(S/2)$

Mi= {0 if $E- |E/S| < sng(S/2)$

Step 5: Set the count of the extracted data, y, to I=1and Select= N1 samples (sliding window size).

Step 6: Evaluate the similarity between the extracted segments V= y (I: L) and U bit by bit. If the similarity value is ≥ Ŧ,Vis taken as the SC and go to Step 8. Otherwise proceed to the next step.

Step 7: Increase the count, I by 1 and slide the window to the next L=N1 samples and repeat step 6.

Step 8:Similarity between the second extracted segments V'=y(I+N1+N2:I+2N1+N2) and U may evaluate bit by bit

Step 9: I←I+N1+N2, of the new I value is equal to sequence length of bits, go to Step 10 else repeat Step 7.

Step 10:Watermarks is extracted P times and make comparison in bit by bit between these watermark, for correction, and finally extract the desired watermark.

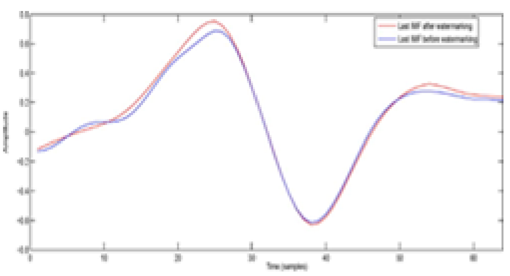

Fig 3 Last IMF of an audio frame before and after watermarking

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Figure 4 Binary watermark

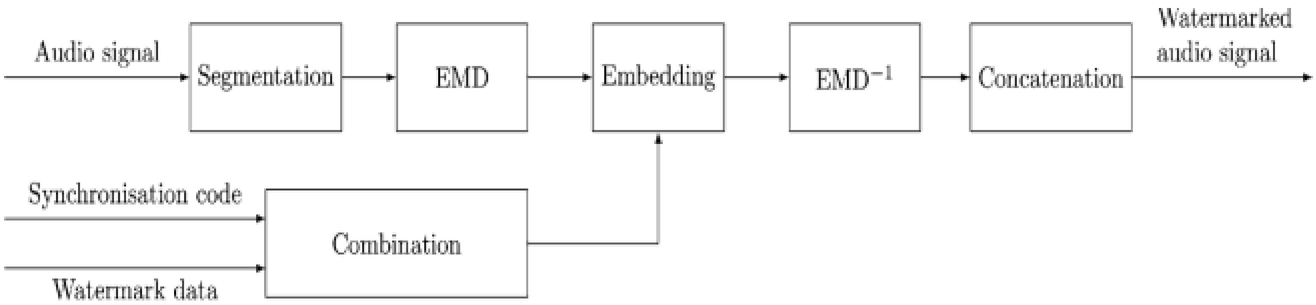

Fig 5.Watermark embedding

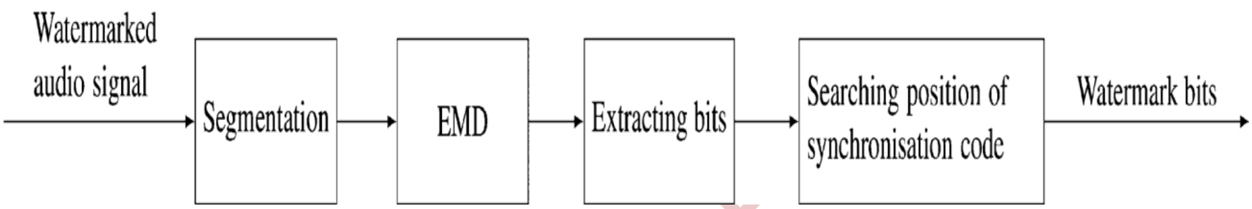

Fig 6 Watermark extraction

4 PERFORMANCE ANALYSIS

A large NC(Normalized Cross Correlation) indicates the presence of watermark while a low value suggests the lack of watermark. Errors may occur while searching the SCs are: the False Positive Error (FPE) and the False Negative Error (FNE). These errors are very harmful because they impair the credibility of the watermarking system. Where p is the SC length and the threshold is the probability that a SC is detected in false location while ' Pep' is the probability that a watermarked signal is declared as unwatermarked by the decoder. Also use as performance measure the payload which quantifies the amount of information to be hidden. More precisely, the data payload refers to the number of bits that are embedded into that audio signal within a unit of time and is measured in unit of bits per second (b/s).NC are defined below

NC(W,w)=

$$\sum_N^M \sum_M^N W(i,j)\, w(i,j)]/$$
$$[\sqrt{\sum_{i=1}^M \sum_{j=1}^N w2(i,j)}\ \sqrt{\sum_{i=1}^M \sum_{j=1}^N W2(I,J)}]$$

Where W and w are original and recovered watermark respectively. Based on this equation above comparison table is created.

Audio signal	Attack type	BER %	NC
classic	No attack	0	1
	AWGN	0	1
	Filtering	0	1
	Cropping	0	1
	Resampling	2	0.9986
	MP3 (64 kb/s)	0	1
	MP3 (32 kb/s)	0	1
	Requantization	0	1
jazz	No attack	0	1
	AWGN	0	1
	Filtering	3	0.9964
	Cropping	0	1
	Resampling	2	0.9983
	MP3 (64 kb/s)	0	1
	MP3 (32 kb/s)	1	0.9973
	Requantization	0	1
rock	No attack	0	1
	AWGN	0	1
	Filtering	0	1
	Cropping	0	1
	Resampling	1	0.9989
	MP3 (64 kb/s)	0	1
	MP3 (32 kb/s)	0	1
	Requantization	0	1

Table 1 Comparison of BER and NC in Extracted Watermark for Different Audio Signals

5 CONCLUSIONS

An adaptive watermarking scheme based on the EMD is proposed in this paper. For achieving good performance watermark is added to the higher energy region of the last IMF. Addition of synchronization code helps to resist the data loss during shifting and cropping. The proposed algorithm has greater robustness against common attacks than nine recently proposed algorithms. In MP3 compression, it produce a better performance compered to earlier audio watermarking method. The watermark used for performance produce no audible distortion. We cannot distinguish watermarked audio and its original version. Using of EMD in this algorithm results some better performances that's are self-adaptive decomposition of the audio signal , low false positive and false negative error probability rates and easy calculations.

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