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INVESTIGATION OF AUDIO SIGNALS WATERMARKING USING EMPIRICAL MODE DECOMPOSITION

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ABSTRACT

In this paper a new adaptive audio watermarking algorithm based on Empirical Mode Decomposition (EMD) is proposed. The audio signal is separated into frames and each one is decomposed adaptively, by EMD, into intrinsic oscillatory components called Intrinsic Mode Functiions (IMFs). The watermark and the synchronization codes are embedded into the extrema of the last IMF, a low frequency mode stable under different attacks and preserving audio perceptual quality of the host signal. The data embedding rate of the proposed algorithm is 46.9–50.3 b/s. Relying on exhaustive simulations, we show the robustness of the hidden watermark for additive noise, MP3 compression, re-quantization, filtering, cropping and resampling. The comparison analysis elucidates that our method has better performance than watermarking schemes reported recently.

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KEY WORDS

Empirical Mode Decomposition, Intrinsic Mode Functions, Synchronization Codes.

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INTRODUCTION

Digital audio watermarking has received a great deal of attention in the literature to afford efficient solutions for copyright protection of digital media by embedding a watermark in the original audio signal. Main necessities of digital audio watermarking are imperceptibility, robustness and data capacity. More precisely, the water- mark must be inaudible within the host audio data to maintain audio quality and robust to signal distortions applied to the host data. Finally, the watermark must be easy to extract to prove ownership. To achieve these requirements, seeking new watermarking schemes is a very challenging problem. Different watermarking techniques of varying complexities have been proposed in a robust watermarking scheme to different attacks is proposed but with a limited transmission bit rate. To advance the bit rate, watermarked schemes performed in the wavelets domain have been proposed [1]. A limit of wavelet approach is that the basis functions are fixed, and thus they do not necessarily match all real signals. To overcome this limitation, recently, a new signal decomposition method referred to as Empirical Mode Decomposition (EMD) has been introduced for analyzing non-stationary signals derived or not from linear systems in totally adaptive way [2-4]. A major advantage of EMD relies on no a priori choice of filters or basis functions. Compared to classical kernel based approaches, EMD is fully data-driven method that recursively breaks down any signal into 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 ones. Any signal is expanded by EMD as follows:

$$x(t) = \sum_{i} IMF j(t) + r(t)$$

where C is the number of IMFs and xc(t) denotes the final residual. The IMFs are nearly orthogonal to each other, and all have nearly zero means [5]. The number of extrema is decreased when going from one mode to the 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. Low frequency components such as higher order IMFs are signal dominated and thus their alteration can lead to degradation of the signal. As result, these modes can be considered to be good locations for watermark placement. Some preliminary results have appeared recently in showing the interest of EMD for audio watermarking. In [5], the EMD is combined with Pulse Code



Modulation (PCM) and the watermark is inserted in the final residual of the subbands in the transform domain. This method supposes that mean value of PCM audio signal may no longer be zero. As stated by the authors, the method is not robust to attacks such as band-pass filtering and cropping, and no comparison to watermarking schemes reported recently in literature is presented. Another strategy is presented in [6-8] where the EMD is associated with Hilbert transform and the watermark is embedded into the IMF containing highest energy. However, why the IMF carrying the highest amount of energy is the best candidate mode to hide the watermark has not been addressed. Further, in practice an IMF with highest energy can be a high frequency mode and thus it is not robust to attacks.

Watermarks inserted into lower order IMFs (high frequency) are most vulnerable to attacks. It has been argued that for watermarking robustness, the watermark bits are usually embedded in the perceptually components, mostly, the low frequency components of the host signal.

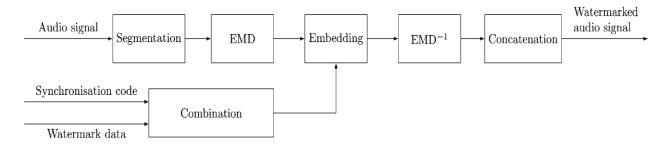


Fig: 1. Watermark embedding

It concurrently has better resistance against attacks and imperceptibility, we embed the watermark in the extrema of the last IMF. Further, unlike the schemes introduced in, the proposed watermarking is only based on EMD and without domain transform. We choose in our method a watermarking technique in the category of Quantization Index Modulation (QIM) due to its good robustness and blind nature. Parameters of QIM are chosen to guarantee that the embedded watermark in the last IMF is inaudible. The watermark is associated with a synchronization code to facilitate its location [9]. An advantage to use the time domain approach, based on EMD, is the low cost in searching synchronization codes. Audio signal is first segmented into frames where each one is decomposed adaptively into IMFs. Bits are inserted into the extrema of the last IMF such that the watermarked signal inaudibility is guaranteed. Experimental results demonstrate that the hidden data are robust against attacks such as additive noise, MP3 compression, requantization, cropping and filtering. Our method has high data payload and performance against MP3 compression.

MATERIALS AND METHODS

PROPOSED WATERMARKING ALGORITHM

The thought of the proposed watermarking technique is to cover up into the first sound flag a watermark together with a Synchronized Code (SC) in the time area. The information sign is initially sectioned into edges and EMD is directed on each casing to extricate the related IMFs [Figure -2]. At that point a paired information arrangement comprised of SCs and enlightening watermark bits [Figure - 3] is installed in the extrema of an arrangement of successive last-IMFs. A bit (0 or 1) is embedded per extrema.

Since the number of IMFs and then their number of extrema depend on the amount of data of each frame, the number of bits to be embedded varies from one frameto the following. Watermark and SCs are not all embedded in extrema of last-IMF of only one frame. In general the number of extrema per last-IMF (one frame) is very small compared to length of the binary sequence to be embedded.

This also depends on the length of the frame. If we design by N1 and N2 the number of bits of SC and watermark respectively, the length of binary sequence to be embedded is equal to 2N1 + N2. Thus , these 2N1+N2 bits are spread out on several last IMFs (extrema) of the consecutive frames. Further, this sequence of 2N1 + N2 bits is embedded P times .Finally, inverse transformation (EMD-1) is applied to the modified extrema to recover the watermarked audio signal by superposition of the IMFs of each frame followed by the concatenation of the frames [Figure -1]. For data extraction, the watermarked audio signal is split into frames and EMD applied to each frame [Figure - 4]. Binary data sequences are extracted from each last IMF by searching for SCs [Figure -5]. We show in [Figure - 6] the last IMF before and after watermarking. This figure shows that there is little



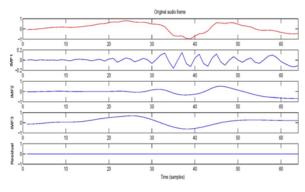


Fig: 2. Decomposition of an audio frame by EMD

Sync-code Watermark bits Sync-code

.....

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Fig:3. Data structure (mi)

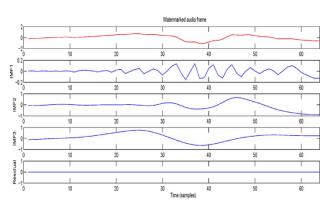


Fig: 4. Decomposition of watermarked audio frame by EMD

difference in terms of amplitudes between the two modes. EMD being fully data adaptive, thus it is important to guarantee that the number of IMFs will be same before and after embedding the watermark [Figure. -2, 4]. In fact, if the numbers of IMFs are different, there is no guarantee that the last IMF always contains the watermark information to be extracted. To overcome this problem, the sifting of the watermarked signal is forced to extract the same number of IMFs as before watermarking. The proposed watermarking scheme is blind, that is, the host signal is not required for watermark extraction. Overview of the proposed method is detailed as follows:

SYNCHRONIZATION CODE

To locate the embedding position of the hidden watermark bits in the host signal a SC is used. This code is unaffected by cropping and shifting attacks.

Let U be the original SC and V be an unknown sequence of the same length. Sequence V is considered as a SC if only the number of different bits between U and V, when compared bit by bit, is less or equal than to a predefined threshold.

WATERMARK EMBEDDING

Before embedding, SCs are combined with watermark bits to form a binary sequence denoted by mi ϵ {0. 1}. i-th bit of watermark[Figure -3]. Basic of our watermark embedding are shown in [Figure -1]and detailed as follows:



- Step 1: Split original audio signal into frames.
- Step 2: Decompose each frame into IMFs.
- Step 3: Embed P times the binary sequence (mi) into extrema of the last IMF (IMFo) by QIM:
- Step 4: Reconstruct the frame (EMD-1) using modified IMFc and concatenate the watermarked frames to retrieve the watermarked signal.

WATERMARK EXTRACTION

For watermark extraction, host signal is splitted into frames and EMD is performed on each one as in embedding. We extract binary data . We then search for SCs in the extracted data



Fig: 5. Watermark extraction

This procedure is repeated by shifting the selected segment (window) one sample at time until a SC is found. With the position of SC determined, we can then extract the hidden information bits, which follows the SC. Let $y = \{m_i / x^*\}$ denote denote the binary data to be extracted and U denote the original SC. To locate the embedded watermark we search the SCs in the sequence $\{m_i / x^*\}$ bit by bit. The extraction is performed without using the original audio signal. Basic steps involved in the watermarking extraction, shown in **[Figure -5]**, are given as follows:

- Step 1: Split the watermarked signal into frames.
- Step 2: Decompose each frame into IMFs.
- Step 3: Extract the extrema {e_i^*} of IMFc.

Step 4: Extract the P watermark and make comparison bit by bit between these marks, for correction, and finally extract the desired watermark.

We evaluate the performance of our method in terms of data payload, error probability of SC, Signal to Noise Ratio (SNR) between original and the watermarked audio signals, Bit Error Rate (BER) and Normalized cross-Correlation (NC). According to International Federation of the Photographic Industry (IFPI) recommendations, a watermark audio signal should maintain more than 20 dB SNR. To evaluate the watermark detection accuracy after attacks, we used the BER and the NC defined as follows:

$$BER(w, \hat{w}) = \frac{\sum_{i=1}^{M} \sum_{j=1}^{N} W(i, j) \oplus W(i, j)}{M \times N}$$

Where \oplus is the XOR operator and M × N are the binary watermark images sizes. w and \hat{w} are the original and the recovered watermark respectively. BER is used to evaluate the watermark detection accuracy after signal processing operations. To evaluate the similarity between the original watermark and the extracted one we use the NC measure defined as follows:

$$NC(w, \hat{w}) = \frac{\sum_{i=1}^{M} \sum_{j=1}^{N} w(i, j) \hat{w}(i, j)}{\sqrt{\sum_{i=1}^{M} \sum_{j=1}^{N} w^{2}(i, j)} \sqrt{\sum_{i=1}^{M} \sum_{j=1}^{N} \hat{w}^{2}(i, j)}}$$

A large NC indicates the presence of watermark while a low value suggests the lack of watermark. Two types of errors may occur while searching the SCs: the False Positive Error (FPE) and the False Negative Error (FNE). These errors are very harmful because they impair the credibilty of the watermarking system.

RESULTS

OUTPUT

To show the effectiveness of the scheme , simulations are performed on audio signals including pop, jazz , rock, and classic sampled at 44.1 kHz. The embedded watermark, W , is a binary logo image of size $M \times N = 34 \times 48$ =1632bits . We convert this 2D binary image into 1D sequence in order to embed it into the audio signal. The SC used is a 16 bit Barker sequence 1111100110101110. Each audio signal is divided into frames of size 64 samples and the threshold T is set to 4. The S value is fixed to 0.98. . These parameters have been chosen to have a good compromise between imperceptibility of the watermarked signal, payload and robustness. [Figure -9]shows a

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portion of the pop signal and its watermarked version. This figure shows that the watermarked signal is visually indistinguishable from the original one.

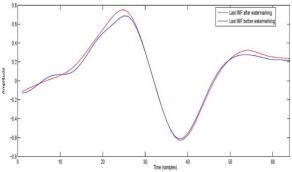


Fig: 6. Last IMF of an audio frame before and after watermarking

Perceptual quality assessment can be performed using subjective lis- tening tests by human acoustic perception or using objective evaluation tests by measuring the SNR and Objective Difference Grade (ODG). In this work we use the second approach. ODG and SNR values of the four watermarked signals are reported. The SNR values are above 20 dB showing the good choice of value and confirming to IFPI standard. All ODG values of the watermarked audio signals are between -1 and 0 which demonstrates their good quality.

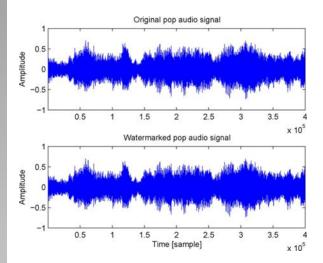


Fig:7. A portion of the pop audio signal and its watermarked version

ROBUSTNESS TEST

To assess the robustness of our approach, different attacks are per-formed:

Noise: White Gaussian Noise (WGN) is added to the watermarked signal until the resulting signal has an SNR of 20 dB.

Filtering: Filter the watermarked audio signal using Wiener filter.

Cropping: Segments of 512 samples are removed from the wa- termarked signal at thirteen positions and subsequently replaced by segments of the watermarked signal contaminated with WGN.

Perceptaints The system of 512 samples are removed from the wa- termarked signal at thirteen positions and subsequently replaced by segments of the watermarked signal contaminated with WGN.

Resampling: The watermarked signal, originally sampled at 44.1 kHz, is re-sampled at 22.05 kHz and restored back by sampling again at 44.1 kHz.

Requantization: The watermarked signal is re-quantized down to 8 bits/sample and then back to 16 bits/sample.



CONCLUSION

In this paper another versatile watermarking plan in view of the EMD is proposed. Watermark is installed in low recurrence mode (last IMF), subsequently accomplishing great execution against different assaults. Watermark is connected with synchronization codes and accordingly the syn-chronized watermark can oppose moving and trimming. Information bits of the synchronized watermark are inserted in the extrema of the last IMF of the sound sign in view of QIM. Broad reenactments over various sound signs show that the proposed watermarking plan has more prominent power against normal assaults than nine re-cently proposed calculations. This plan has higher payload and better execution against MP3 pressure contrasted with these before sound watermarking strategies. In all sound test flags, the watermark introduction duced no discernable mutilation. Tests exhibit that the waterchecked sound signs are vague from unique ones. These exhibitions exploit the self-versatile decay of the sound sign gave by the EMD. The proposed plan accomplishes low false positive and false negative blunder likelihood rates. Our watermarking strategy includes simple estimations and does not utilize the first sound sign. In the directed analyses the inserting quality S is kept consistent for all sound documents. To assist enhance the execution of the strategy, the S parameter ought to be adjusted to the sort and plan of an answer technique for versatile installing issue. Additionally as future examination we plan to incorporate the attributes of the human sound-related and psychoacoustic model in our watermarking plan for a great deal more change of the execution of the watermarking technique. At long last, it ought to be fascinating to research if the proposed strategy underpins different inspecting rates with the same payload and heartiness furthermore if in genuine applications the technique can deal with D/An A/D transformation issues. Additionally, the execution of sound watermarking utilizing EMD is being finished by ARM processor.

CONFLICT OF INTEREST

The authors declare no conflict of interests.

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None

FINANCIAL DISCLOSURE

The authors report no financial interests or potential conflicts of interest.

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