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RESEARCH ARTICLE

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A NEW ADAPTIVE AUDIO WATERMARKING SCHEME BASED ON EMD

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ABSTRACT

In this paper a new adaptive audio watermarking algorithm primarily based on Empirical Mode Decomposition (EMD) is introduced. The audio sign is divided into frames and each one is decomposed adaptively, via EMD, into intrinsic oscillatory components known as Intrinsic Mode capabilities (IMFs). The watermark and the synchronization codes are embedded into the intense of the last IMF, a low frequency mode stable underneath special attacks and keeping audio perceptual fine of the host signal. The statistics embedding rate of the proposed algorithm is 46.9-50.3 b/s. relying on exhaustive simulations, we display the robustness of the hidden watermark for additive noise, MP3 compression, re-quantization, filtering, cropping and re-sampling. The assessment evaluation suggests that our method has higher overall performance than watermarking schemes pronounced recently.

Key Words—Empirical mode decomposition, intrinsic mode function, audio watermarking, quantization index modulation, synchronization code.

I. INTRODUCTION

Digital audio watermarking has received a good buy of interest inside the literature to produce economical solutions for copyright protection of digital media by using embedding a watermark in the authentic audio signal [1][2]. Main needs of virtual audio watermarking are physically property, hardiness and information capability. Additional precisely, the watermark should be supersonic most of the host audio records to take care of audio great and robust to signal distortions carried out to the host records. in the end, the watermark must be easy to extract to prove possession[3]. Searching for new watermarking schemes may be a very difficult drawn side.

Finally distinctive watermarking techniques of variable complexities are planned. to reinforce the bit rate, watermarked schemes performed within the wavelets domain are planned. A restriction of riffle technique is that the crucial functions are fastened, and consequently they are doing now not essentially fit for all real alerts.

To beat this limitation, currently, signal decomposition method mentioned as Empirical Mode Decomposition (EMD) has been delivered for reading non stationary signals derived or not from linear systems. a giant advantage of EMD depends on no a priori choice of filters or basis features. as compared to classical kernel primarily based often processes[2], EMD is totally statistics driven technique that recursively breaks down any signal into zero mean with bilaterally symmetrical envelopes AM FM elements known as Intrinsic Mode features (IMFs).The IMFs are nearly orthogonal to every exceptional, and everyone a have almost zero mean that the variety of extreme is decreased while going from one mode to succeeding, and therefore the entire decomposition is certain to be completed with a finite variety of modes. The IMFs are definitely represented by way of their nearby extreme and so is frequently recovered exploitation these intense [3][4]. Low frequency elements like higher order IMFs are sign dominated and so their alteration will result in degradation of the signal. As result, those modes might be concept of to be clever locations for watermark placement.

Some preliminary consequences have seemed recently in displaying the interest of EMD for audio watermarking. The EMD is combined with Pulse Code Modulation (PCM) [11] and additionally the watermark is inserted in the very last residual of the sub bands in the transform domain. This system supposes that average of PCM audio signal couldn't be zero.

As specific through the authors, the technique isn't always sturdy to attacks like band pass filtering and cropping, and no comparison to watermarking schemes in accordance currently in literature is given. other strategy is bestowed in which the EMD is associated with Hilbert remodel and additionally the watermark is embedded into the IMF containing highest electricity [5]. but, why the IMF carrying the pleasant amount of energy is that the high-quality candidate mode to cover the watermark has no longer been addressed. Further, in have a look at associate IMF with highest power is often an excessive frequency mode and so it's now not strong to attacks.

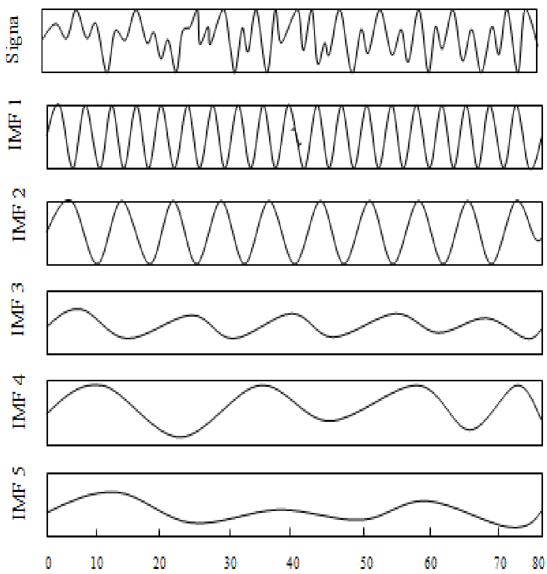

Fig.1. Decomposition of an audio signal by EMD.

Sync-code	Watermark bits	Sync-code

Fig. 2. Data structure

Watermarks inserted into lower order IMFs (excessive frequency) are maximum vulnerable to assaults. it has been argued that for watermarking hardiness, the watermark bits are typically embedded inside the perceptually components, generally, the low frequency elements of the host signal [5].

In comparison to privies techniques [5] [11] the identical time have better resistance towards assaults and physical belongings [4] [5], we will be inclined to embed the watermark within the extreme of the last IMF.

Further, unlike schemes introduced [4] [5] within the planned watermarking is really primarily based on EMD and even as no longer domain remodel. We choose in our method a watermarking technique within the class of quantization Index Modulation (QIM) [6] because of its desirable hardiness and blind nature. Parameters of QIM are selected to ensure that the embedded watermark inside the remaining IMF is inaudible. The watermark is related to a synchronization code to facilitate its location

An advantage to use the time domine technique, supported EMD [9], is that the low value in looking synchronization codes. Audio signal is preliminary segmented into frames wherever every person is rotten adaptively into IMFs. Bits are inserted into the extreme of the last IMF certain the watermarked signal in audibleness is bonded. Experimental consequences display that the hidden information is robust against attacks together with additive noise, MP3 compression, re quantization, cropping and filtering. Our technique has high facts payload and performance against MP3 compression in comparison to audio watermarking techniques reported lately inside the literature.

II. PROPOSED WATERMARKING ALGORITHM

Proposed algorithm is EMD (Empirical Mode Decomposition). EMD is nothing but decomposition of audio signal in to variable Intrinsic Mode function. The idea of the proposed watermarking method is to hide into the authentic audio sign a watermark together with a Synchronized Code (SC) in the time domine. The input sign is first segmented into frames and EMD is carried out on each frame to extract the related IMFs (figure.1). Then a binary data collection consisted of SCs and informative watermark bits (figure.2) is embedded inside the extrema of a set of consecutive last-IMFs. A bit (0 or 1) is inserted in step with extrema. Because the number of IMFs after which their quantity of extrema depends upon the ammount of data of every frame, the number of bits to be embedded varies from last-IMF of one frame to the subsequent. Watermark and SCs are not all embedded in extrema of last IMF of most effective one frame. In general the quantity of extrema according to ultimate-IMF (one frame) may be very small in comparison to period of the binary collection to be embedded. This additionally depends on the length of the frame.

If we design through N1 and N2 the numbers of bits of SC and watermark respectively, the length of binary series to be embedded is same to 2N1+N2 .as a result, these 2N1+N2 bits are spread out on numerous last-IMFs (extrema) of the consecutive frames.

Further, this collection 2N1+N2 of bits is embedded P times. In the end, inverse transformation (EMD^-1) is applied to the modified extrema to get better the watermarked audio signal with the superposition of the IMFs of every frame. For information extraction, the watermarked audio signal is split into frames and EMD implemented to each frame. Binary statistics sequences are extracted from each last IMF via searching for SCs We show in figure.4. The last IMF before and after watermarking. The figure 6 suggests that there may be little distinction in terms of amplitudes among the two modes.

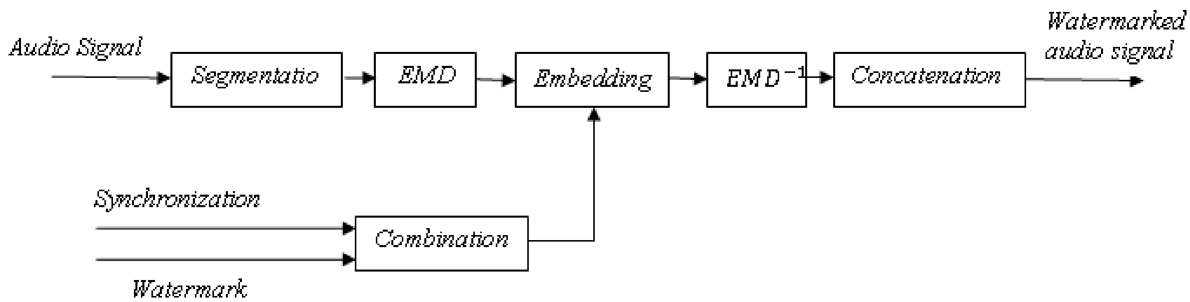

Fig.3.watermarking embedding

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. 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. An Intrinsic Mode Function (IMF) is a function that satisfies the condition that at any time instant, the mean value of the upper envelope as defined by the local maxima and the lower envelope as defined by the local minima is zero.

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 [6]. 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 collection denoted by $\mathrm{mi} \in \{0,1\}$, i-th bit of watermark. Basics of our watermark embedding are proven in figure.3 and precise as follows.

Step 1: break up original audio signal into frames.

Step 2: Decompose every frames into IMFs.

Step 3: Embed p times the binary sequence {mi} into extrema of the last IMF by QIM [6]

e_i^*=[e_i/s].S + sgn(3S/4) if m_i=1

e_i^*=[e_i/s].S + sgn(S/4) if m_i=0

where ei and ei* are the extrema of IMFc of the host audio signal and the watermarked signal respectively. sig function is same to “+” ⍰ if ei is a maxima, and “-“⍰ if it is a minima. [ ] denotes the floor function, and S denotes the embedding strength chosen to keep the inaudibility constraint.

Step 4: Reconstruct the frame (EMD^-1) using modified and concatenates the watermarked frames to retrieve the watermarked signal.

Watermark Extraction.

For watermark extraction, host sign is splitted into frames and EMD is executed on each one as in embedding. We extract binary data the use of rule given. We then search for SCs inside the extracted information. This process is repeated by way of shifting the selected segment (window) one sample at time till a SC is determined. With the placement of SC determined, we will then extract the hidden information bits, which follows the SC.

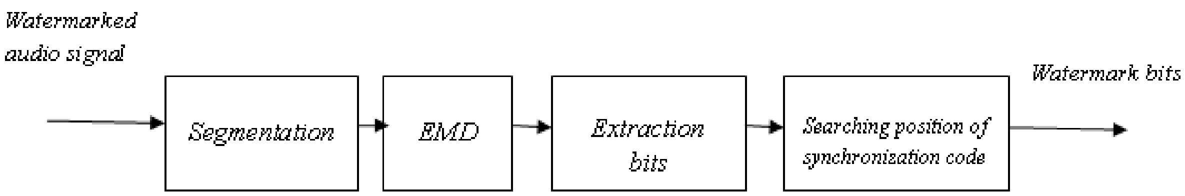

Fig.4.watermarking extraction

let Y={mi*} denote the binary statistics to be extracted and denote the original SC. To locate the embedded watermark we search the SCs in the sequence {mi*} bit by bit. The extraction is carried out without the usage of the unique audio signal(original signal) in the extraction process. simple steps involved within the watermarking extraction, shown in figure.4, are given as follows:

Step 1: Break up the watermarked signal into frames.

Step 2: Decompose every frame into IMFs

Step 3: Extract the extrema ei* of IMFc .

Step 4: Extract e_i^* from m_i^* using the following rule

m_i^*=1 if e_i^*-[e_i^*/s].s>=sgn(S/2)

m_i^*=0 if e_i^*-[e_i^*/s].s<sgn(S/2)

Step 5: Set the start index of the extracted data y to I=1 and L=N1 select samples (sliding window size).

Step 6: Evaluate the similarity between the extracted segment V=y(I:L) and U bit by bit. If the similarity value is>=T , then V is taken as the SC and go to Step8. Otherwise proceed to the next step.

Step 7: Increase by 1 and slide the window to the next samples and repeat Step 6.

Step 8: Evaluate the similarity between the second extracted segment, V'=y(I+N1+N2) And U bit by bit.

Step 9: I ←I+N1+N2, of the new value is equal to sequence length of bits.Go to Step 10 else repeat Step7.

Step 10: Extract the P watermarks and make comparison bit by bit between these marks, for correction, and finally extract the desired watermark.

III. EXPERIMENTAL RESULTS

On this consideration an audio as input and a logo as water mark data. The audio input is converted in to wave (.wav) format and input image that is water mark statistics should be in portable format(.png). Our Audio input that is in wave format (.wav) as follows.

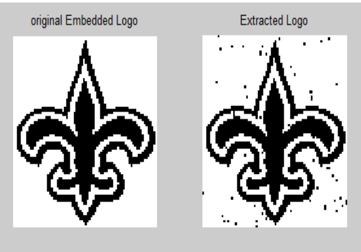

Fig.5: Original embedded Logo and Extracted logo

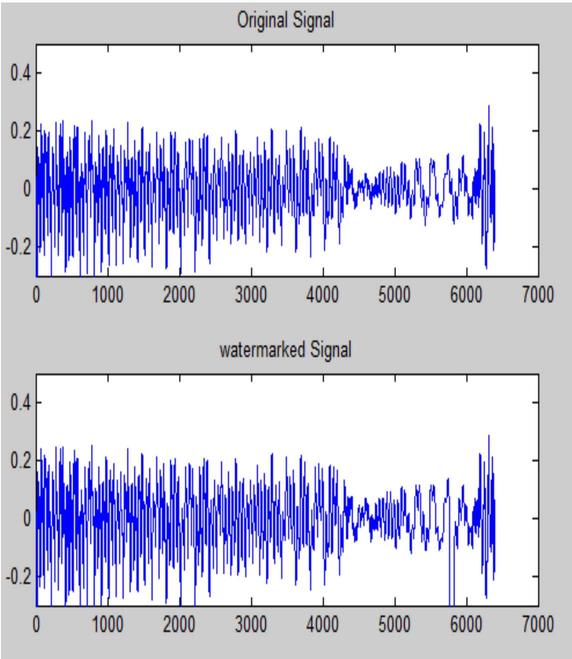

Fig.6. Original audio signal and watermarking Signal

In Figure.6. the last IMF before and after watermarking waveform is shown and This figure shows that there is little difference in terms of amplitudes between the two modes.

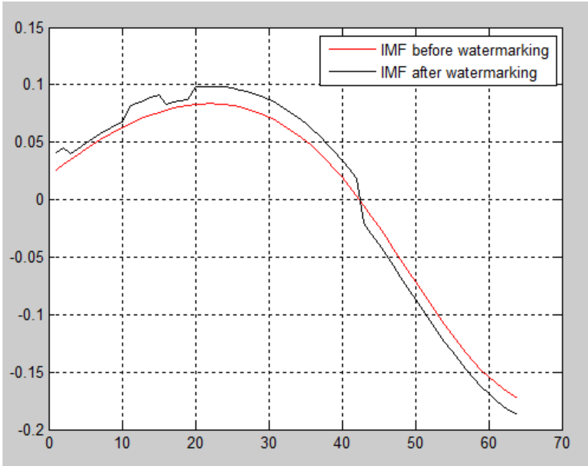

Figure.7. Last IMF of an audio frame before and after watermarking.

IV. PERFORMANCE EVALUATION OF PROPOSED SYSTEM

Table.I. BER and NC Of Extracted Watermark For Different Audio Signals

Audio Signal	Attack Type	BER	NC
	No attack	0	1
	AWGN	0	1
	Filtering	0	1
	Cropping	0	1
Classic	Re Sampling	2.03	0.99
	MP3(64kb/s)	0	1
	MP3(32kb/s) Requantization	0	1
		0	1
	No attack	0	1
	AWGN	0	1
	Filtering	3.02	0.963
	Cropping	0	1
Jazz	Re Sampling	2.08	0.992
	MP3(64kb/s)	0	1
	MP3(32kb/s) Requantization	1	1
		0	0.973

	No attack	0	1
	AWGN	0	1
	Filtering	0	0.963
	Cropping	0	1
Rock	Re Sampling	1	0.992
	MP3(64kb/s)	0	1
	MP3(32kb/s) Requantization	0	1
		0	0.973

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 NC.

Table.II. SNR And ODG B|W Original And Watermarked Audio

Audio files	SNR	ODG
Classic	27.22	-0.766
Jazz	27.272	-0.767
POP	27.065	-0.761
Rock	27.177	-0.764

Table.III. Comparison Of Audio Watermarking Methods, Sorted By Attempted Payload.

References	Payload	Robustness
Proposed	71	44
Bhat and. Sengupta[1]	45.9	32
Lie and Chang[7]	43	80
S. Xiang, H. J. Kim[12]	2	64

In our paper a new adaptive watermarking scheme based on the EMD is proposed. Watermark is embedded in very low frequency mode (last IMF), thus achieving good performance against various attacks. Watermark is associated with synchronization codes. Thus the synchronized watermark has the ability to resist shifting and cropping. Data bits of the synchronized watermark are embedded in the extreme of the last IMF of the audio signal based on QIM. Extensive simulations over different audio signals indicate that the proposed watermarking scheme has greater robustness. Against common attacks than nine recently propose the algorithms.

This scheme has higher payload and better performance againstMP3 compression compared to these earlier audio watermarking methods. In all audio test signals, the watermark introduced no audible distortion. Experiments demonstrate that the watermarked audio signals are indistinguishable from original ones. These performances take advantage of the self-adaptive decomposition of the audio signal provided by the EMD.

V. CONCLUSION

The proposed scheme achieves very low false positive and false negative error probability rates. Our watermarking method involves easy calculations and does not use the original audio signal in extraction of watermark bits. In the conducted experiments the embedding strength S is kept constant for all audio files. To further improve the performance of the method, the parameter should be adapted to the type and magnitudes of the original audio signal. Our future works include the design of a solution method for adaptive embedding problem. Also as future research we plan to include the characteristics of the human auditory and psychoacoustic model in our watermarking scheme for much more improvement of the performance of the watermarking method.

Finally, it should be interesting to investigate if the proposed method supports various sampling rates with the same payload and robustness and also if in real applications the method can handle D/A-A/D conversion problems. The idea of the proposed watermarking method is to hide into the original audio signal a watermark together with a Synchronized Code (SC) in the time domain.

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