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Audio Watermarking with EMD Technique

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ABSTRACT: A new audio watermarking technique, Empirical Mode Decomposition (EMD) is introduced in this paper. The audio signal is splitted first into frames and each frame is then decomposed by EMD to get its basic intrinsic periodic elements known as Intrinsic Mode Functions (IMFs). These collective combinations of watermark and synchronization code are embedded into the last IMF. Last IMFs are most stable frequency component under various attacks and so protects quality of the host signal. So this methodology is applied to any image and audio signals. Experimentation has given the physical marking and its detection to protect audio on different attacks.

KEYWORDS: Empirical mode decomposition, quantization index modulation, frequency modulation Intrinsic Mode Function synchronization code, amplitude modulation, signal to noise ratio.

I. INTRODUCTION

Digital audio watermarking has achieved a noble attention to offer efficient solutions for copyright protection of digital media. Main objectives of digital audio watermarking are robustness and imperceptibility. The watermark should be imperceptible inside host audio so that its quality is preserved but it should be robust to the attacks or signal alterations given to the host data. At last, watermark should be easy to extract and to evidence the ownership. To achieve above requirements, pursuing new schemes for watermarking is a very challenging. Different watermarking techniques with different complexities have been given. To protect the dataagainst various attacks, a robust scheme of watermarking is given but it fails in bit rate transmission. To eliminate bit rate problem, watermarked schemes in the domain of wavelet has also been proposed. Watermarking in wavelet domain has fixed basis functions, which do not guarantee to match all real time signals. To overcome these drawbacks, Empirical Mode Decomposition (EMD), this new has been introduced which works on both stationary as well as non-stationary signals. EMD does not require a primary choice of filters or any basis functions. This scheme breaks down any signal in zero-mean symmetrical envelopes AM-FM segments known as Intrinsic Mode Functions (IMFs). Any signal is expanded by EMD as follows:

$$x(t) = \sum_{j=1}^{c} IMF_j(t) + r_c(t) \qquad (1)$$

where C denotes IMF count and $r_c(t)$ denotes final residual .
The IMFs are orthogonal to each other and have zero means. The extent of extrema is reduced when it from one to the next mode, and the whole scheme of decomposition is

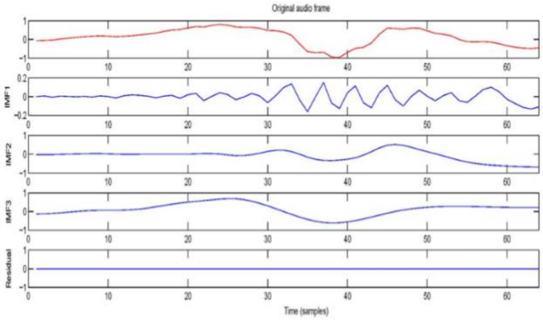

Fig. 1. Decomposition of an audio frame by EMD

concluded in a finite number of modes. The IMFs are totally labeled by their local extrema and so can be recovered back using it. Low frequency components that is higher order IMFs are signal controlled hence their modification can end up in withdrawal of the signal. So, these positions are the best for placing watermark. In some initial results, the EMD is united with Pulse Code Modulation and the watermark is implanted in the final residual of the sub bands in the transform domain. This method provides the mean value of PCM audio signal may no longer be zero. The method is not robust to various attacks like band-pass filtering and cropping. Likewise in other scheme, EMD with Hilbert transform the watermark is embedded into the IMF having highest energy. Though, how the IMF carrying the maximum amount of energy is the best mode to hide the watermark has not been given. In practice an IMF with highest energy is highest frequency mode which is not robust to attacks. Watermarks inserted into lower order IMFs or high frequency components are most prone to attacks. It has been claimed that for robustness, the watermark bits are generally embedded in the perceptually components of the host signal to have better disapproval against attacks and imperceptibility. Thus in this project, the watermark are implanted in the extrema of the last IMF. The watermark is linked with a synchronization code to support its location. The advantage to use the time domain method in EMD, is the low budget in probing synchronization codes. Audio signal is first segmented into frames and each one is decomposed into IMFs. Bits are added into the extrema of the last IMF such that the watermarked signal inaudibility is established. Experimental results verify that the hidden data are robust for attacks such as noise additions, compressions, requantization, cropping etc. This process is easy and robust as related to audio watermarking methods informed recently

II. RELATED WORK

Arnold, et al, has specified a imperceptible watermarking technique in audio watermarking is not as developed, yet as related to the researches which has done in image and video watermarking. The audio signals are described by a few samples per time interval shows that the quantity of information bits which can be embedded powerfully and noiselessly in audio files is much lesser than the amount of information bits that can be inserted in visual files. Gordy et al, has stated an algorithm independent framework for equating digital watermarking algorithm with respect to perceptual quality, bit rate, robustness and complexity to signal processing. on the basis of four properties perceptual quality, bitrate, complexity and robustness the algorithms for public watermarks were equaled. Frequency masking embedding was the costly operation as of the necessity to compute a complex, time-varying perceptual masking analysis to be passed out on the host signal. Tom et al, in watermark security has stated the incapability by unapproved users to have access to the watermarked channels. The security of watermarking trusts in the secrecy of the keys and only the information of both the algorithm and the keys can break the algorithm. Martins, et al, has given frequency domain approach, is expensive to appliance than time domain techniques. As they have a higher bit rate and more robust watermark elimination under signal processing.The assumption for the PCM watermarking was that zero cross withdrawals change and removal of all watermarks while resampling terminates the watermark completely. Christine, et al, has given a general explanation of digital watermarking algorithm and their applications which was on general watermarking problems and to recognize the fundamental properties and the restrictions of watermarking system.Martin, et al, in his paper dedicated on the design of a content subtle audio watermarking scheme to allow many post production growths and on the design of an invertible watermarking pattern cooperative with digital signatures for high security. Anastasias, et al, has declared in his overview of watermarking schemes based on chaotic producers and correlation sensor. High pass chaotic watermarks are better than low pass watermark which have worst performance when no distortion is exposed on the watermarked signal. Xian young, et al, has given the contrast of watermarks given by chaotic functions, logistic map and Bernoulli maps in the presence of practical attacks. He appealed the watermark to be imperceptible and robust to some typical signal treating operations. Ali, et al, has proposed audio watermarking created on discrete wavelet transform. He has given singular value decomposition (SVD) technique. Ke-Xin Yin, et al, has given a sightless robust digital watermarking scheme by means of chaotic encryption and visual system of human in the DWT domain. The watermark is encrypted by means of logistic map. K.Khaldi, et al, has proposed a comparative study of diverse decomposition methods used to de-noise an ECG signal. The difficulties of FT method can be cracked out by Wavelet Transform scheme. As Wavelet Transform is a non-adaptive, it is not suitable to remove the high frequency noise from signal. The problems of WT method can be narrowed by Empirical Mode of Decomposition scheme. A in depth study is mandatory to find out the most advantageous decomposition method for an ECG signal. A latest accommodative and adaptive audio watermarking technique supportive Empirical Mode Decomposition (EMD) was revealed.

III. **PROPOSED METHODOLOGY**

The idea of the projected watermarking method is to hide watermarked data in the original audio signal itself. The input signal is first fragmented into frames and EMD is carried on every frame to remove the associated IMFs. Then a binary data sequence contained of SCs and revealing watermark bits are embedded in the last of a set of consecutive last-IMFs. A bit (0 or 1) is implanted per extrema. As the number of IMFs and their number of extreme rest on on the amount of data of each frame, the amount of bits to be embedded varies from last-IMF of one frame to the next. Watermark and SCs are not altogether embedded in extrema of former IMF of only one frame. In general the amount of extrema per last-IMF of one frame is very slight compared to length of the binary sequence to be inserted. This totally depends on the length of the frame.

Sync-code	Watermark bits	Sync-code

Fig.2 Data structure $m(i)$

Watermark and SCs are not all implanted in extrema of last IMF of one frame. In overall the number of extrema per last-IMF is very small related to length of the binary sequence to be embedded. This also depends on the length of the frame. If we draw byN_1andN_2 the numbers of bits of SC and watermark respectively, the length of binary sequence to be embedded is equivalent to $2N_1$ +N_2.Hence, these $2N_1$ +N_2bits are range out on several last-IMFs (extrema) of the consecutive frames. Again this sequence of$2N_1$ +N_2bits is embedded Ptimes. Lastly, inverse transformation$(EMD)^{-1}$ is applied to the improved extrema to recover the watermarked audio signal by superposition of the IMFs of each frame trailed by the concatenation of the frames (Fig. 1). For data extraction, the watermarked audio signals aresplitedinto frames and EMD applied to each frame (Fig. 3). Binary data arrangements are removed from each last IMF by probing for SCs (Fig. 4). We classify the last IMF before and after watermarking. This figure shows that there is little alteration in terms of amplitudes amongst the two modes. EMD being fully data adaptive, thus it is important to guarantee that the number of IMFs will be unchanged before and after embedding the watermark (Figs. 3,5). If the numbers of IMFs are different, there can be no guarantee that the last IMF always comprises the watermark information to be removed. To come overthis difficulties, the sifting of the watermarked signal is enforced to extract the same number of IMFs as beforehand watermarking. The projected watermarking system is blind, that is, the host signal is not mandatory for watermark extraction. Overview of the proposed scheme is detailed as follows:

A. SYNCHRONIZATION CODE

To locate the embedding location of the hidden watermark bits in the host signal a SC is used. This code is not disturbed by cropping and shifting attacks. Let be the original SC and be an unidentified sequence of the same length. Order V is considered as a SC if only the number of unlike bits between when equated bit per bit, is less or equal to the predefined threshold. This aids to locate the embedding location of the hidden watermark bits in the host signal a SC which is used. This code is not affected by cropping and shifting attacks.

B. WATERMARK EMBEDDING

Beforehand embedding, SCs are joined with watermark bits to form a binary sequence represented by $m_i \epsilon \{0,1\}, i$ -th bit of watermark (Fig.2). Fundamentals of our watermark embedding are shown in Fig. 1 and detailed as follows:
Step 1: Split original audio signal to frames.
Step 2: Decompose every frame into IMFs.
Step 3: Embed P times the binary arrangement $\{m_i\}$ into extrema of the last IMF $(IMF)_c$by QIM.

$$e_i{}^* = \begin{cases} \left\lfloor \frac{e_i{}^*}{S} \right\rfloor . S + \operatorname{sgn}\left(\frac{3S}{4}\right) & if\ m_i = 1 \\ \left\lfloor \frac{e_i{}^*}{S} \right\rfloor . S + \operatorname{sgn}\left(\frac{S}{4}\right) & if\ m_i = 0 \end{cases} \qquad (2)$$

Where $e_i$and e_i^* are the extrema of IMF_C of the host audio signal and the watermarked signal correspondingly. sine function is equal to “+ ” (maxima) and “ -” (minima). $\lfloor \ \rfloor$ indicates the floor function, and S denotes the embedding strength selected to preserve the inaudibility constraint.

Step 4: Rebuild the frame $(EMD)^{-1}$ using improved IMF_C and concatenate the watermarked frames to recover the watermarked signal.

C. WATERMARK EXTRACTION

For extraction of watermark, host signal is splitted into frames and EMD is achieved on each one as in embedding. We then hunt for SCs in the extracted data. This process is repetitive by shifting the selected segment one sample at a time until a SC is discovered. After determining SC, we can then extract the hidden information bits which trail the SC. Let $y = \{m_i^*\}$indicate the binary data to be extracted $U$and denote the original SC. To trace the embedded watermark, hunt the SCs in the arrangement $\{m_i^*\}$bit by bit. The extraction is performed devoid of using the original audio signal. Simple steps tangled in the watermarking extraction, shown in Fig. 4, are given as follows:

Step 1: Split watermarked signal into individual frames.

Step 2: Decompose each and every frame into IMFs.

Step 3: Extract the extrema $\{e_i^*\}$ of IMF_C

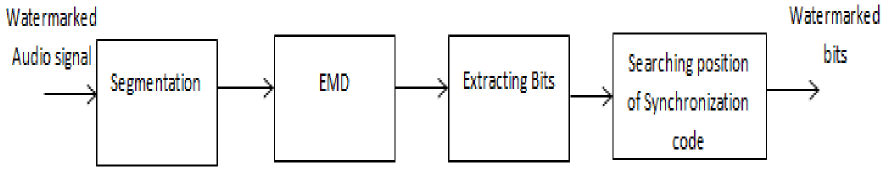

Fig. 3 Watermark extraction.

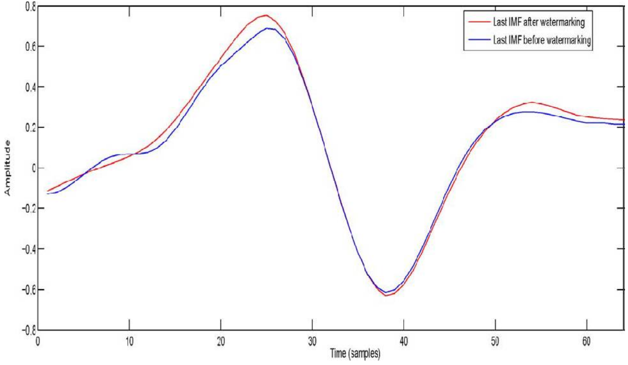

Fig. 4 Last IMF of an audio frame before and after watermarking.

Step 4: Extract m_i^* frome_i^* using the following rule:

$$m_i{}^* = \begin{cases} 1\ if\ e_i{}^* - \left\lfloor \frac{e_i{}^*}{S} \right\rfloor . S \geq \operatorname{sgn}(S/2) \\ 0\ if\ e_i{}^* - \left\lfloor \frac{e_i{}^*}{S} \right\rfloor . S < \operatorname{sgn}(S/2) \end{cases} \qquad (3)$$

Step 5: Set the starting of index of the extracted data y, to I =1 and $L = N_1$ select samples (sliding window size).

Step 6: Calculate the similarity between the extracted segment $V = y(I:L)$and U bit by bit. If the correspondence value is $\geq \boldsymbol{\tau}$, then V is taken as the SC and go to Step 8. Or else proceed to the next step.
Step 7: Increment$I$ by 1 and slide the window to the next$L = N_1$samples and repeat Step 6.
Step 8: Calculate the similarity between the second extracted segment,

$$\boldsymbol{V^1 = y(I + N1 + N2 : I + 2N1 + N2)}$$and$U$ bit by bit.

Step 9: $\boldsymbol{I \leftarrow I + N1 + N2}$, of the new$\boldsymbol{I}$ value is equal to sequence length of bits, follow Step 10 or else repeat Step 7.
Step 10: Extract the $\boldsymbol{P}$ watermarks and make evaluate bit by bit amongst these marks, for correction, and lastly extract the desired watermark. Watermarking embedding and extraction procedures are concise in Fig. 7.

Fig.5. Binary Watermark

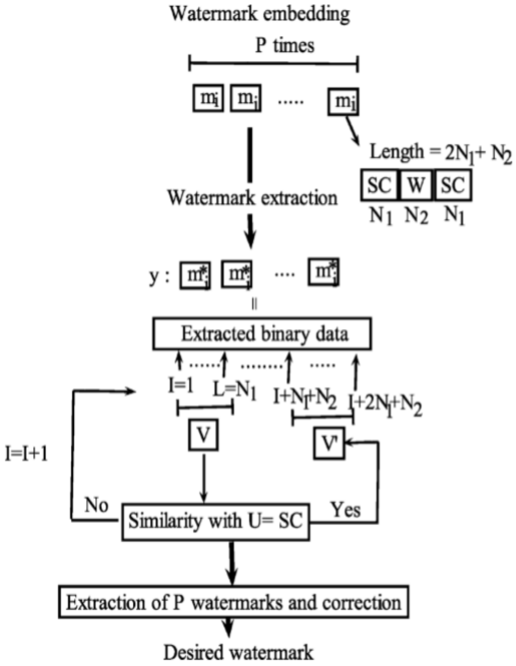

Fig.6. Embedding and extraction of watermark actual process

IV. QUALITY ANALYSIS

To evaluate the performance of proposed technique Bit error rate (BER) and Signal to Noise Ratio (SNR), amongst original and the watermarked audio signals is taken. A watermarked audio signal should preserve SNR more than 18 dB. To estimate the watermark revealing accuracy after attacks ,we used the and the defined as belows:

$$BER(W,W') = \frac{\sum_{i=1}^{M}\sum_{j=1}^{N} W(i,j) \oplus W'(i,j)}{M*N} \qquad (4)$$

where⊕ denotes XOR operator and M × N denotes the sizes of binary watermark image. *W* and *W*' gives original and the recovered watermark separately. Is used to evaluate theaccuracy in watermark detection after signal processing operations.

V. RESULT AND CONCLUSION

The graphical representation of results is given below. The first graph represents the input audio signal or original audio signal. Second graph represents the audio signal after decomposition by Empirical mode decomposition technique. The comparative study of both the graphs shows that there are very negligible changes in both the input and modified input that is decomposed audio signal by EMD.

Table 1 gives the comparative analysis of input and output in terms of the analysing factors that are BER and SNR ratio.

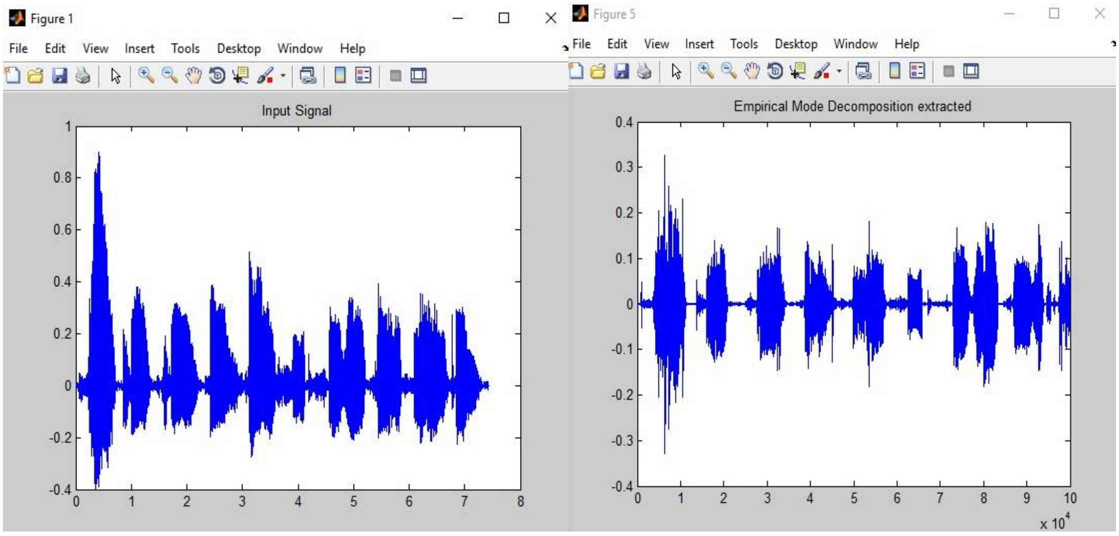

TABLE 1
BER AND SNR OF ORIGINAL AND WATERMARKED AUDIO

Audio Signal	BER	SNR
Signal 1	1.643	19.72
Signal 2	1.810	21.44
Signal 3	1.213	18.71

VI. CONCLUSION

In this paper a new watermarking scheme based on the EMD is projected. Watermark is introduced in very low frequency mode and is associated with synchronization codes, so defends against various attacks. Data bits of the synchronized watermark are inserted in the extrema of the last IMF of the audio signal so does not obstructs the quality of host signal. It works in real time and due to its adaptive nature, its enhanced watermarking technique. Altogether audio test signals, the watermark inserted are imperceptible. Experiments validate that the watermarked audio signals are indistinguishable from their original ones. This watermarking scheme has easy calculations and does not use the original audio signal to recover.

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