

## A Scheme for Digital Audio Watermarking Using Empirical Mode Decomposition with IMF

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**Abstract:** *In internet, the illegal copying of data has become a grave and thus it made the digital watermarking schemes enters as an important role to handle and deal such a critical issue. In this paper we present a digital audio watermarking scheme with the use of Empirical Mode Decomposition (EMD). A binary referential image, which is obtained by mapping from standard normal distributed pseudo random numbers, is embedded as a covert information into the significant coefficients of the highest energetic IMF that are greater than a specified adaptive threshold. Thus watermarked IMF is less sensitive to common signal processing attacks like such as additive noise, LPF, re-sampling, re-quantization, MP3 compression, and sound processing effects etc . As a result the proposed method increases more robustness of resultant images. All this mean that, our methodology is applied to any image and any audio signals. .Experimentation has ensured the mark physical property, the ability of detection of the mark and therefore the hardness against completely different varieties of attacks.*

**Keywords:** *digital watermark, EMD, HT, IMF, re-quantization.*

### 1. INTRODUCTION

The increased sharing of digital data among multiple users, distributed over the network calls for the protection of the data against unauthorized copying. Digital watermarking has gained reputation due to its significance in content authentication and copyright protection for digital multimedia data. It hides information in the data in such a way that the basic appearance of the data is not destroyed.

In a sturdy watermarking theme to totally different attacks is planned however with a restricted transmission bit rate. To boost the bit rate, watermarked schemes performed within the wavelets domain are planned. A limit of riffle approach is that the essential functions are fastened, and therefore they are doing not essentially match all real signals. To overcome this limitation, recently, a brand new signal decomposition methodology noted as Empirical Mode De-composition (EMD) has been introduced for analyzing non-stationary signals derived or not from linear systems in altogether reconciling means a significant advantage of EMD depends on no a priori selection of filters or basis functions. Compared to classical kernel based mostly approaches, EMD is totally data-driven methodology that recursively breaks down any signal into a reduced variety of zero-mean with bilaterally symmetrical envelopes AM-FM parts referred to as Intrinsic Mode Functions (IMFs). The decomposition starts from finer scales to coarser scales ones. Any signal is expanded by EMD as follows:

$$X(t) = \sum_{j=1}^C \text{IMF}_j(t) + r_c(t) \quad (1)$$

Where C is the number of IMFs and  $r_c(t)$  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. The IMFs are nearly orthogonal to every different, and everyone have nearly zero means that the number of extrema is decreased when going from one mode to succeeding, and therefore the whole decomposition is bound to be completed with a finite number of modes.

The IMFs are totally represented by their local extrema and so are often recovered exploitation these extrema. Low frequency elements like higher order IMFs are signal dominated and so their alteration will result in degradation of the signal. As result, these modes will be thought of to be smart locations for watermark placement. In the EMD is combined with Pulse Code Modulation (PCM) and also the watermark is inserted within the final residual of the sub-bands within the rework domain. We elect in our methodology a watermarking technique with in the class of Quantization Index Modulation (QIM) because of its good hardiness and blind nature. Parameters of QIM are chosen to ensure that the embedded watermark within the last IMF is inaudible. The watermark is related to a synchronization code to facilitate its location. An advantage to use the time domain approach, supported EMD, is that the low value in searching synchronization codes. Audio signal is initially segmented into frames wherever everyone is rotten adaptively into IMFs. Bits are inserted into the extrema of the last IMF specified the watermarked signal in audibility is bonded. Experimental results demonstrate that the hidden information is strong against attacks such as additive noise, MP3 compression, re-quantization, crop-ping and filtering. Our method has high data payload and performance against MP3 compression compared to audio watermarking approaches reported recently in the literature.

## 2. WATER MARKING METHODS

### 2.1 Echo-Hiding

This method 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.

### 2.4 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 depends on the length of the frame also. If  $N_1$  and  $N_2$  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  $2N_1+N_2$ . Thus, these  $2N_1+N_2$  bits are spread out on several last-IMFs or extrema of

the consecutive frames, which having zero mean. Further, this sequence of  $2N_1+N_2$  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.

### 2.5 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  $T$ . Fig.1 Data structure (mi)

Sync-code	Watermark bits	Sync-code
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Figure 1. data structure (mi)

### 2.6 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 and figure 2 indicates the process of embedding of audio signal and binary image.

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.

$$E = \lfloor e_i/s \rfloor \cdot S + \text{sgn}(3s/4) ; \text{ if } m_i=1 \tag{2}$$

$$E = \lfloor e_i/s \rfloor \cdot S + \text{sgn}(s/4) ; \text{ if } m_i=0 \tag{3}$$

where  $e_i$  and  $E$  are the extrema of the host audio signal and the watermarked audio respectively  $\text{sgn}$  function is equal to “+” if is a maxima, and “\_” if it is a minimal.  $\lfloor \cdot \rfloor$  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.

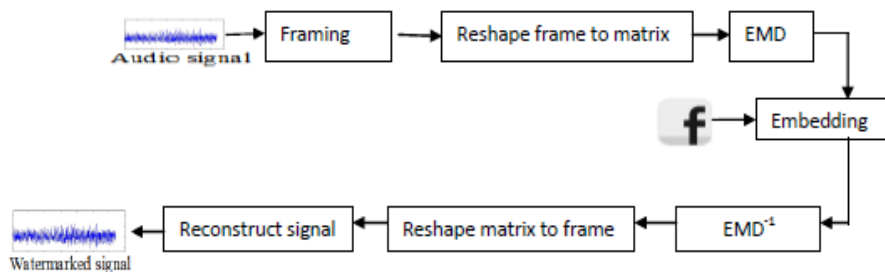


Figure 2. watermark embedding

### 2.7 Watermark Extraction

In the watermark extraction process, host signal is fragmented into frames and EMD is performed on each frame. For extracting the binary data, search the SCs to the decomposed frame. This procedure is repeated by shifting or increasing the count in the algorithm. Examine 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  $y = \{M_i\}$

denote the binary data to be extracted. For locating the embedded watermark search the SCs in the sequence  $\{M_i\}$  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 IMFs.

Step 4: Extract using the following equation

$$M_i = \{ 1 \text{ if } E - |E/S| \cdot S \geq \text{sng}(S/2) \} \tag{4}$$

$$M_i = \{ 0 \text{ if } E - |E/S| < \text{sng}(S/2) \} \tag{5}$$

Step 5: Set the count of the extracted data,  $y$ , to  $I=1$  and  $\text{Select} = N_1$  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  $\geq \tau$ ,  $V$  is 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  $L = N_1$  samples and repeat step 6.

Step 8: Similarity between the second extracted segments

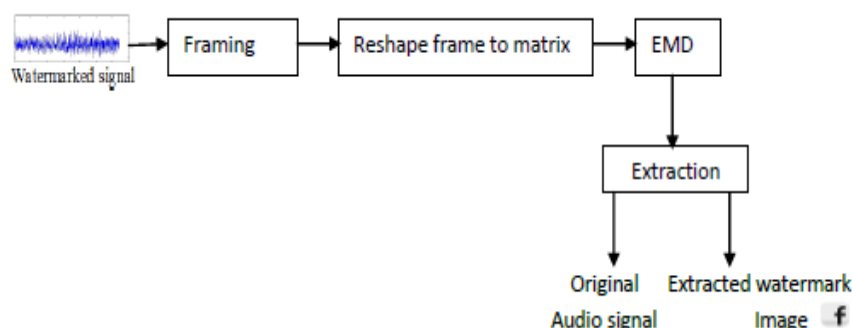
$$V' = y(I + N_1 + N_2 : I + 2N_1 + N_2) \tag{6}$$

and  $U$  may evaluate bit by bit

Step 9:  $I \leftarrow I + N_1 + N_2$ , 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.

Below figure 3 shows the extracting of original image from watermarked audio signal.



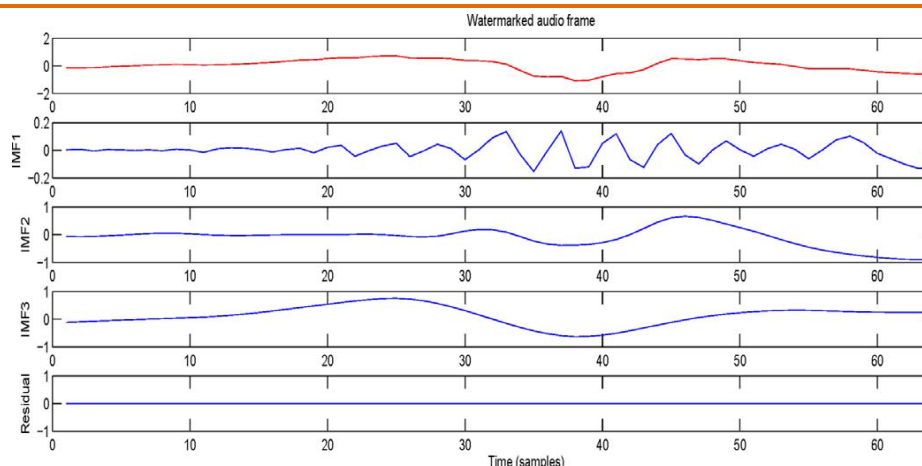
**Figure 3.** watermark Extraction

### 3. EXPECTED RESULT ANALYSIS

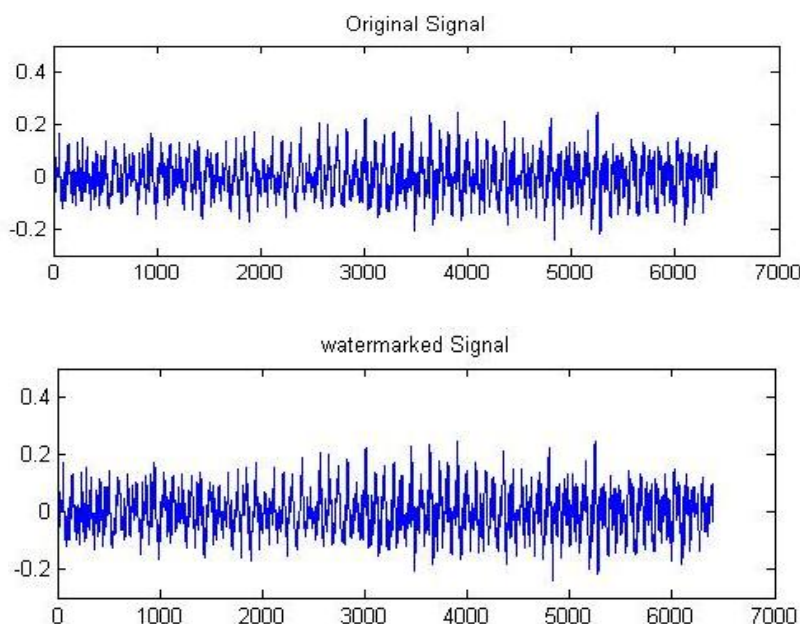
Simulations are performed on audio signals including Rhythm divine, pop, jazz, and classic sampled at 42.1 kHz. The embedded watermark,  $W$ , is a binary logo image of size bits which is in figure 4. We convert this 2D binary image into 1D sequence in order to embed it into the audio signal. Each audio signal is divided into frames of size 64 samples and the threshold is set to 4 and the 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. For data extraction, the watermarked audio signal is split into frames and EMD applied to each frame showed in figure 5. And figure 6 shows a portion of the rhythm divine signal and its watermarked version. This figure shows that the watermarked signal is visually indistinguishable from the origin alone.



**Figure 4.** input embedded watermark image

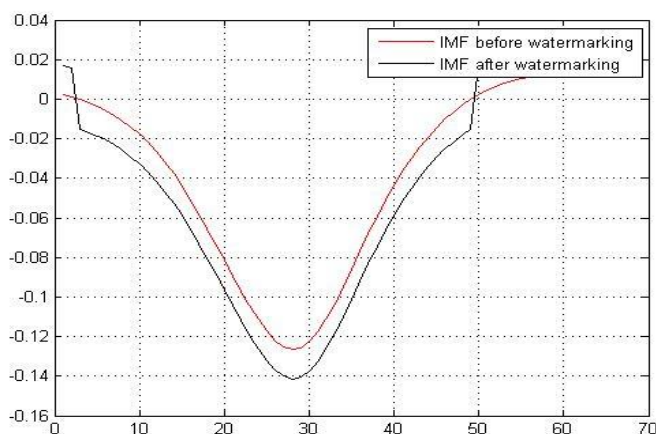


**Figure 5.** Decomposition of the watermarked audio frame by EMD



**Figure 6.** A portion of the rhythm divine audio signal and its watermarked version

The binary data sequences are extracted from each last IMF by searching for SCs showed in extraction process and we show in Figure 7, the last IMF before and after watermarking. 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

Finally, figure 8 shows the extracted binary image from the watermarked audio signal.





**Figure 8.** *Extracted image from audio signal*

#### 4. CONCLUSION

In this paper, for achieving a good performance watermark is added with the last IMF. In addition of synchronization code helps to resist the data loss during shifting and cropping. The proposed algorithm has greater robustness against common attacks than recently proposed algorithms. In MP3 and wave file compression, it produces a better performance compared to existing and old audio watermarking methods. Using of EMD, algorithm results some better performances that's self-adaptive decomposition of the audio signal, low false positive and false negative error probability rates and easy calculations.

#### REFERENCES

- [1] L. Ghouti, A. Bouridane, M. K. Ibrahim, S. Boussakta, "Image Watermarking using Balanced Multiwavelets", IEEE Transactions on Signal Processing, Vol. 54(4), pp. 1519-1536, 2006.
- [2] H Hammami, Z. Tariq, "Audio in image steganography using same weak point in HAS&HVS", International Conference on ICT, Cairo, Egypt, pp. 181-188, Dec (2004).
- [3] K. Yeo and H. J. Kim, "Modified patchwork algorithm: A novel audio watermarking scheme," IEEE Trans. Speech Audio Process., vol. 11, no. 4, pp. 381–386, Jul. 2003.
- [4] K. Khaldi and A. O. Boudraa, "On signals compression by EMD," Electron. Lett., vol. 48, no. 21, pp. 1329–1331, 2012.
- [5] K. Khaldi, M. T.-H. Alouane, and A. O. Boudraa, "Voiced speech enhancement based on adaptive filtering of selected intrinsic mode functions," J. Adv. in Adapt. Data Anal., vol. 2, no. 1, pp. 65–80, 2010.
- [6] N. K. Zaman, K. M. I. Khalilullah, Md. W. Islam, and Md. K. I. Molla, "A robust digital audio watermarking algorithm using empirical mode decomposition," in Proc. IEEE CCECE, 2010, pp. 1–4.
- [7] Shaoquan Wu, Jiwu Huang, Daren Huang, and Yun Q. Shi, "Efficiently Self-Synchronized Audio Watermarking for Assured Audio Data Transmission," IEEE Transactions On Broadcasting, VOL. 51, NO. 1, pp. 69 – 76, 2005
- [8] Mingquan Fan, Hongxia Wang, "Chaos-based discrete fractional Sine transform domain audio watermarking scheme," Computers and Electrical Engineering, Volume 35, Issue 3, May 2009, pp. 506– 516.
- [9] Ko, B.-S., Nishimura, R., and Suzuki, Y. "Time-spread echo method for digital audio watermarking using pn sequences". Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing, vol. 2, 2002, pp. 2001–2004.

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