

Adaptive Digital Audio 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 introduced. The audio signal is divided into frames and each one is decomposed adaptively, by EMD, into intrinsic oscillatory components called Intrinsic Mode Functions (IMFs). The watermark and the synchronization codes are embedded into the extremes 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 recompiling. The comparison analysis shows that our method has better performance than watermarking schemes reported recently.

Keywords – Audio Watermarking, Empirical Mode Decomposition, Intrinsic Mode Function, Quantization Index Modulation, Synchronization Code.

I. INTRODUCTION

Digital audio watermarking has received a great deal of attention in the literature to provide efficient solutions for copyright protection of digital media by embedding a watermark in the original audio signal [1]–[5]. Main requirements of digital audio watermarking are imperceptibility, robustness and data capacity. More precisely, the watermark 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 [5]. Different watermarking techniques of varying complexities have been proposed [2]–[5]. In [5] a robust watermarking scheme to different attacks is proposed but with a limited transmission bit rate. To improve the bit rate, watermarked schemes performed in the wavelets domain have been proposed [3], [4]. A limit of wavelet approach is that the basic 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 [6]. 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). One of the most secure techniques of audio watermarking is spread spectrum audio watermarking (ssw). Spread spectrum is a general technique for embedding watermarks that can be implemented in any transform domain or in the time domain. In ssw, a narrow-band signal is transmitted over a much larger bandwidth such that the signal energy presented in any signal frequency is undetectable. Thus, the watermark is spread over many frequency bins so that the energy in one bin is undetectable. An interesting feature of this watermarking technique is that destroying it requires noise of high amplitude to be added to all frequency bins. This type of watermarking is robust since to be confident of eliminating a watermark, the attack must attack all possible frequency bins with modifications of considerable strength. This will create visible defects in the data.

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 [7], [8]. Low frequency components such as higher order IMFs are signal dominated [9] 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 [10], [11] showing the interest of EMD for audio watermarking. In [10], 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 [11] 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 [12], [13]. 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 [14]. Compared to [10], [15], to simultaneously have better resistance against attacks and imperceptibility, we embed the watermark in the extrema of the last IMF. Further, unlike the schemes introduced in [10], [11], 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 [16]. 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 [17] to facilitate its location. 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 [18] 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 compared to audio watermarking approaches reported recently in the literature. Digital watermarking has been proposed as a means to identify the owner or distributor of digital data. Watermarking is the process of encoding hidden copyright information in digital data by making small modifications to the data samples. Unlike encryption, watermarking does not restrict access to the data. Once encrypted data is decrypted, the media is no longer protected.

A watermark is designed to permanently reside in the host data. When the ownership of a digital work is in question, the information can be extracted to completely characterize the owner.

II. PROPOSED METHODOLOGY

A. 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 [4]. 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 [3].

$$E_m = F^{-1} \{ \log(|F(x)|^2) \} \quad (1)$$

where F represents the Fourier Transform and F^{-1} the inverse Fourier Transform can be examined at two locations, corresponding to the delays of the "one" and "zero" kernel, respectively.

B. Watermark Embedding

Before embedding SCs are combined with watermark bits to form a binary sequence denoted by $m_i \in \{0, 1\}$, it bit of watermark basics of watermark embedding are

follows Split original audio signal into frames and decompose each frame into IMFs. Embedded p times the binary sequence $\{m_i\}$ into extrema of the last IMF by QIM [13]. The Watermark Embedding is shown in Figure.1.

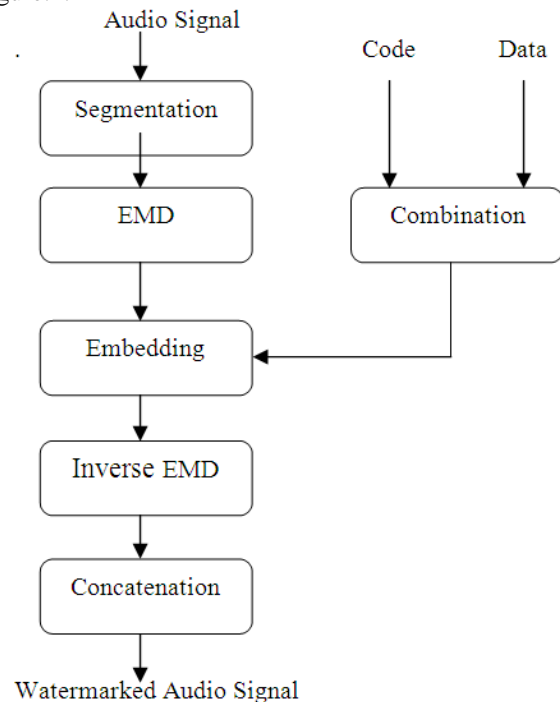


Fig.1. Watermark Embedding

Reconstruct the frame using modified and concatenate the watermarked frames to retrieve the watermarked signal.

$$X_k = X_{N-k}^* X_k \quad (2)$$

Where $*$ denotes complex conjugation and the subscripts are interpreted modulo N . It means that we can just modify the first half of complex DFT spectrum and the second half must conform to this symmetry. Therefore, only half of spectrum is involved in the process of watermarking. The whole procedure is depicted as follows [19]. Cepstrum refers to the spectrum in cepstral domain, like frequency spectrum in frequency domain. The complex cepstrum $x(n)$ for a sequence $x(n)$, is calculated by finding the complex natural logarithm of Fourier Transform (FT [.] of $x(n)$, and then taking inverse Fourier transform (FT-1[.] of the resulting sequence, as described.

Fourier Transform:

$$X(e^{j\omega}) = \sum_{n=-\infty}^{\infty} x(n)e^{-j\omega n} = |X(e^{j\omega})| \bullet e^{j \arg[X(e^{j\omega})]} \quad (3)$$

Complex Natural Logarithm:

$$\ln(X(e^{j\omega})) = \ln|X(e^{j\omega})| + j \arg[X(e^{j\omega})] \quad (4)$$

Inverse Fourier Transform:

$$FT^{-1}[\ln(X(e^{j\omega}))] = \frac{1}{2\pi} \int_{-\pi}^{\pi} \ln(X(e^{j\omega})) \bullet e^{j\omega n} d\omega \quad (5)$$

Therefore, the complex cepstrum is given by

$$\hat{x}(n) = FT^{-1}[\ln(FT[x(n)])] = \frac{1}{2\pi} \int_{-\pi}^{\pi} \ln(X(e^{j\omega})) \bullet e^{j\omega n} d\omega \quad (6)$$

In fact, the real part of complex cepstrum is often taken as ‘cepstrum’, hereunder what we use in most applications. It should be distinguished from “real cepstrum”, $r_{\text{real}}(n)$, which is defined as the inverse Fourier transform of the natural logarithm of magnitude of Fourier transform of $x(n)$, equation (7)

$$\hat{x}_{\text{real}}(n) = FT^{-1}[\ln(|FT[x(n)]|)] = \frac{1}{2\pi} \int_{-\pi}^{\pi} \ln|X(e^{j\omega})| \bullet e^{j\omega n} d\omega \quad (7)$$

Real cepstrum is based only on the magnitude of the Fourier transform for the sequence and phase information is excluded, so there is no inverse real cepstrum transformation to complete reconstruction. Currently, [20] cepstrum analysis has been extensively used in speech/speaker recognition and homomorphic filtering, whereas it is a comparatively new area for audio watermarking. cepstrum coefficients experience much less disturbance after most common signal processing attacks than original samples in time domain, propose to embed watermark by manipulating the statistical mean (SMM) of a subset of cepstrum coefficients.

C. 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, using rule given by (3). We then search for SCs in the extracted data. This procedure is repeated by shifting the selected segment (window) one sample at time until a SC is found (6). The Watermark Extraction is shown in Figure.2.

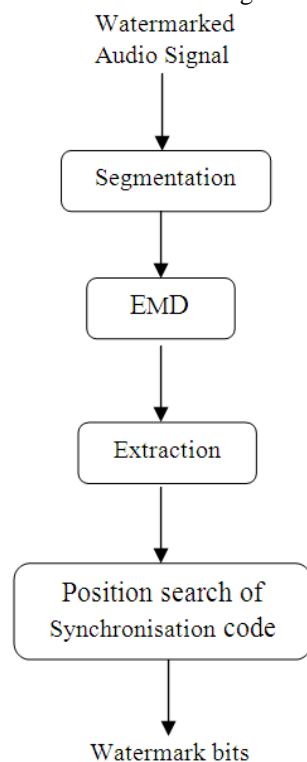


Fig.2. Watermark extraction

With the position of SC determined, we can then extract the hidden information bit, which follows the SC. Let $y = \{m_i^*\}$ 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 $\{m_i^*\}$ bit by bit. The extraction is performed without using the original audio signal. Extract the watermarks and make comparison bit by bit between these marks, for correction, and finally extract the desired watermark. Echo hiding embeds watermark bits into the host signal by introducing two echo kernels with different offsets. As discussed before, if both offsets are below the audible threshold of human ear, the watermarking would not generate noticeable noise. Correspondingly, watermark extraction depends on echo detection, where cepstrum analysis plays an important role. Such minute space is hard to discern in time domain. The equation 8 and 9 and expressed echo kernel formulated.

The essence of embedding is passing every portion of host signal through one of two possible delay units according to its desired bit, ‘1’ or ‘0’. Two echo kernels are illustrated. Where d_1, d_0 are delay offsets and α_1, α_0 are echo amplitudes.

The echo kernel is formulated as

$$h(n) = \delta(n) + \alpha \delta(n - d) \quad (8)$$

So the echoed signal is expressed as

$$s(n) = x(n) \otimes h(n) = x(n) + \alpha x(n - d) \quad (9)$$

Where \otimes denotes convolution, and $x(n)$ and $s(n)$ represent host signal and its echoed version respectively. To detect the echo embedded, we compare the peak values at two possible locations in cepstrum domain. This is based on the fact that complex cepstrum will show a spike at d_1 or d_0 , which indicates the echo offset. The complex cepstrum of echoed signal is given by equation 10 and 11.

$$\begin{aligned} \tilde{s} &= FT^{-1}[\ln(FT[s(n)])] \\ &= FT^{-1}[\ln(FT[x(n)])] + FT^{-1}[\ln(FT[h(n)])] \\ &= C_x(n) + C_h(n) \end{aligned} \quad (10)$$

$$C_x(n) = FT^{-1}[\ln|X(e^{j\omega})|], C_h(n) = FT[h(n)] = 1 + \alpha e^{-j\omega d} \quad (11)$$

It shows that a series of pulses with exponentially decaying amplitudes repeatedly appear. But the first spike at $n=d$ is dominant and its amplitude is α . Moreover, it is seen that the delay offset should be chosen carefully to ensure not only inaudibility of echo but also high detection possibility. We compare the peak values at two possible locations in cepstrum domain. However, in real experiment, the result of the cepstrum is still unsatisfied. The cepstrum peak is small relative to the host signal and is subject to be buried by surrounding peaks. Therefore, autocorrelation of cepstrum is taken equation 12.

$$FT^{-1}[\ln(FT[s(n)])^2] \quad (12)$$

The embedding watermark in narrow bands of the host signal, it is more difficult to intercept and remove the

watermark carried in SS watermarking scheme. However, perceivable distortion is often introduced into the host signal. As a result, amplitude shaping by masking threshold based on psychoacoustic model is always crucial to keep the noise inaudible to human auditory system. The providing greater frequency resolution and less time resolution for low frequencies. In this respect, DWT exhibits similar time-frequency resolution characteristics to human ear.

D. Empirical Mode Decomposition

The Empirical Mode Decomposition (EMD) was proposed as the fundamental part of the Hilbert–Huang transform (HHT). The Hilbert Huang transform is carried out, so to speak, in 2 stages. First, using the EMD algorithm, we obtain intrinsic mode functions (IMF). Then, at the second stage, the instantaneous frequency spectrum of the initial sequence is obtained by applying the Hilbert transform to the results of the above step. The HHT allows to obtain the instantaneous frequency spectrum of nonlinear and non-stationary sequences. The decomposition starts from finer scales to coarser ones. Any signal $x(t)$ is expanded by EMD as follows[13].

$$x(t) = \sum_{j=1}^C \text{IMF}_j(t) + r_c(t) \quad (13)$$

Where C is the number of IMFs and r_c denotes the final residual. The IMFs are nearly orthogonal to each other, and all have nearly zero means.

These sequences can consequently also be dealt with using the empirical mode decomposition. However, this article is not going to cover the plotting of the instantaneous frequency spectrum using the Hilbert transform. We will focus only on the EMD algorithm. In contrast to the previously mentioned Fourier transform and wavelet transform, the EMD decomposes any given data into intrinsic mode functions (IMF) that are not set analytically and are instead determined by an analyzed sequence alone. The basic functions are in this case derived adaptively directly from input data. An IMF resulting from the EMD shall satisfy only the following requirements:

1. The number of IMF extreme (the sum of the maxima and minima) and the number of zero-crossings must either be equal or differ at most by one;
2. At any point of an IMF, the mean value of the envelope defined by the local maxima and the envelope defined by the local minima shall be zero.

Decomposition results in a family of frequency ordered IMF components. Each successive IMF contains lower frequency oscillations than the preceding one. And although the term "frequency" is not quite correct when used in relation to IMFs, it is probably best suited to define their nature. The thing is that even though an IMF is of oscillatory nature, it can have variable amplitude and frequency along the time axis. It is quite difficult to visualize the EMD algorithm performance results based on the description alone so let us proceed to its software implementation that will give us an opportunity to get to know the algorithm peculiarities. The algorithm as

proposed by Huang is based on producing smooth envelopes defined by local maxima and minima of a sequence and subsequent subtraction of the mean of these envelopes from the initial sequence. This requires the identification of all local extrema that are further connected [21] by cubic spline lines to produce the upper and the lower envelopes.

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 follow [14].

$$\text{BER}(W, \tilde{W}) = \frac{\sum_{i=1}^M \sum_{j=1}^N w(i, j) \oplus \tilde{w}(i, j)}{M \times N} \quad (14)$$

where \oplus is the XOR operator and $M \times N$ are the binary

watermark image sizes. W and \tilde{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 follow[15].

$$\text{NC}(W, \tilde{W}) = \frac{\sum_{i=1}^M \sum_{j=1}^N W(i, j) \tilde{W}(i, j)}{\sqrt{\sum_{i=1}^M \sum_{j=1}^N W^2(i, j)} \sqrt{\sum_{i=1}^M \sum_{j=1}^N \tilde{W}^2(i, j)}} \quad (15)$$

The embedding is based on the notion that rounding a certain decimal place of frequency spectrum coefficients to an even number stands for watermark '1', or to an odd number for watermark '0'. The decimal place depends on the perceptibility and security, probably the fourth or fifth decimal place of Matlab data. Besides, there are two additional features in the scheme. One is employing four times repetitive encoding to ensure reliability. The other is using PN sequence to randomize the quantization scale to enhance further the security, which is regarded as secret key. Here, the fourth decimal place is for '1' and the fifth for '0'. As for the watermark detection, the procedure is exactly the first part of embedding. The watermark bit is determined as '0' or '1' on the basis of whether the specific decimal place is even or odd. The SNR and ODG between original and watermarked audio is shown in Table 1.

Table 1: SNR AND ODG between original and watermarked audio

| Audio file | SNR(dB) | ODG |
|------------|---------|------|
| Classic | 25.67 | -0.5 |
| Jazz | 26.38 | -0.4 |
| Pop | 24.12 | -0.6 |
| Rock | 25.49 | -0.5 |

To show the effectiveness of our 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 $M \times N = 34 \times 48 = 1632$ bits. 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 τ is set to 4. The value is fixed to 0.98.

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 credibility of the watermarking system. The associated probabilities of these errors are given by [14], [15]:

$$P_{FPE} = \frac{1}{2^p} \sum_{k=p-\tau}^p C_k \quad (16)$$

$$P_{FNE} = \frac{1}{2^p} \sum_{k=p+\tau}^p C_k (BER)^k (1-BER)^{p-k} \quad (17)$$

where p is the SC length and τ is the threshold. P_{FPE} is the probability that a SC is detected in false location while P_{FNE} is the probability that a watermarked signal is declared as unwatermarked by the decoder. We 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). Perceptual quality assessment can be performed using subjective listening 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 in Table 1.

III. RESULTS

In the first step, the audio is converted to a general format, often to raw format (16 bits PCM), to mono averaging left and right channels, to a sampling rate of 44.1 KHz. The input audio signal is shown in figure.3.

The system operates on rich set of audio file types such as .wav.amv .mps etc, for experimental purpose, The test signal database consists of audio signals. The audio signal is segmented to obtain series of frames with size of 128 samples. The segmentation makes the job of embedding more easily. When a signal is multiplied by a segmentation function, the product is also zero valued outside the interval. Segmentation shown in figure 4.

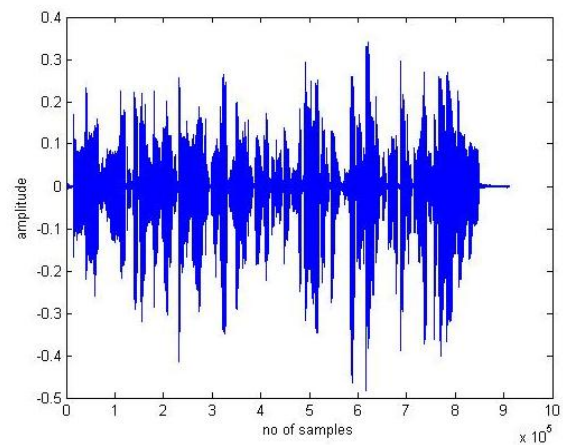


Fig.3. Input audio signal

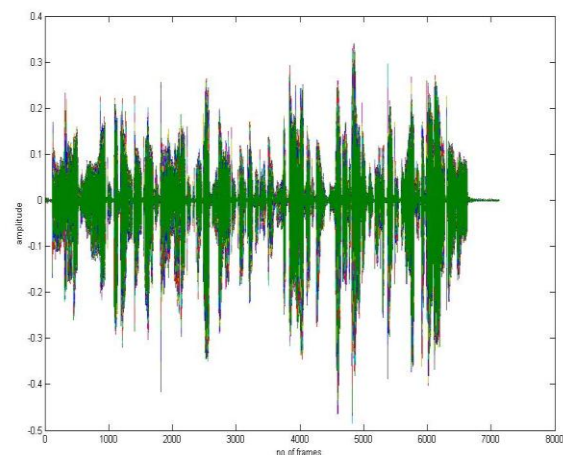


Fig.4. Segmentation

The watermark text is typed in a text file and then later imported in the mat lab editor for embedding. The Embedding of watermark should be completely lossless because watermark signal is the proof of ownership. The watermark signal is a low amplitude signal that is generated by a noninvertible function that takes as input a set of parameters (watermark key). The watermarking procedure should rely on a key to ensure security, not on algorithms secrecy. The watermark Embedded Signal shown in figure.5.

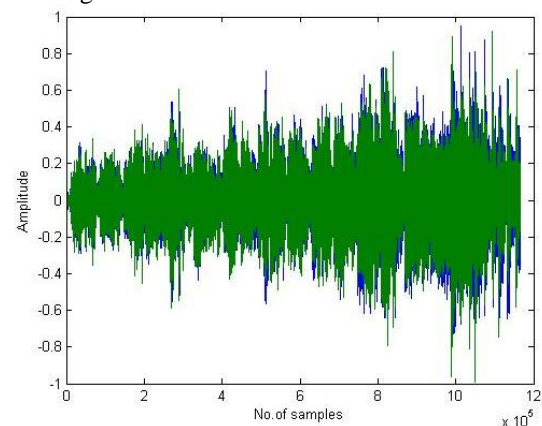


Fig.5. Watermark Embedded Signal

The form a binary sequence by $m_i \in \{0,1\}$, the embedded watermark, W , is a binary logo image of size $M \times N = 34 \times 48 = 1632$ bits figure.5. We convert this 2D binary image into 1D sequence in order to embed it into the audio signal. The synchronization code used is a 16 bit Barker sequence 1111100110101110. Each audio signal is divided into frames of size 64 samples and the threshold is set to 4. 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. Fig.5 shows a portion of the pop signal and its watermarked version. This figure shows that the watermarked signal is visually indistinguishable from the original one.

The retrieved audio signal is representation of audio frequency range of roughly 20 to 20,000 Hz. Audio signals may be synthesized directly, or may originate at a transducer. The signal flow is the path an audio signal will take from source or recording devices. Audio signals may be characterized by parameters such as their bandwidth, power level in decibels and voltage level. For watermark extraction, host signal is spitted into frames and EMD is performed on each one as in embedding. We then search for SCs in the extracted data. 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 follow the synchronization code. Split the watermarked signal into frames. Decompose each frame into IMFs. Extract the extreme $\{e_i^*\}$ of IMF_c. Set the start index of the extracted data, y , to $I=1$ and select $L = N_1$ samples (sliding window size). Extract the P watermarks and make comparison bit by bit between these marks, for correction, and finally extract the desired watermark.

IV. CONCLUSION

In this project, The Segmented input audio signal into frames. EMD conducted on every frame to extract the associated IMFs. Combination of Watermark bits and synchronization code is embedded with set of consecutive IMFS. EMD⁻¹ is applied to the modified to recover the watermarked audio signal. Thus, analysis of watermarking scheme based on empirical mode decomposition shows better performance than other watermarking schemes in terms of security. The idea of the proposed watermarking method is to hide data and watermark together into the original audio signal with a Synchronized Code (SC) in the time domain. The input signal is first segmented into frames and EMD is conducted on every frame to extract the associated IMFs. For data extraction, the watermarked audio signal is split into frames and inverse EMD applied to each frame. Binary data sequences are extracted from each last IMF by searching for SCs.

ACKNOWLEDGMENT

It is a great pleasure to have the opportunity to express my gratitude to all those who gave me the possibility to complete this project. First and foremost, we thank the LORD ALMIGHTY for his abundant grace and countless blessings for making this work a great success. It is a pleasure to pay tribute to my parents, brother, friends and well-wishers for their prayers, unbounded love, support and endless patience during this endeavour.

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