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Audio Watermarking System Using EMD with Psychoacoustic Model

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Abstract— **A new adaptive audio watermarking algorithm based on empirical mode decomposition (EMD) with human auditory and psychoacoustic model is used in this research. 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 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 EMD algorithm is 46.9–50.3 b/s. Relying on exhaustive simulations, show the robustness of the hidden watermark for additive noise, MP3 compression, re-quantization, filtering, cropping and resampling. Empirical Mode Decomposition (EMD) has been introduced for analyzing non-stationary signals derived or not from linear systems in totally adaptive way. A major advantage of EMD relies on no a priori choice of filters or basis functions. 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.**

The psychoacoustic model is basically based on many studies of human auditory perception. In this research, the watermark in audio can be embedded and extracted with different techniques. The psychoacoustic model can be done before embedding the watermark bit. In the scheme, the additive watermarking technique is used to embed a unique pseudo- random sequence, considered as a watermark, into the transformed domain of audio signal. The watermark strength is properly adjusted based on weighting factors derived from the proposed psychoacoustic models. The results show that at the equivalent quality of the watermarked audio, judged by the human hearing system, the robustness of the embedded watermark was increased to higher percentage, compared to the results obtained from the scheme with non- psychoacoustic model and the psychoacoustic model.

***Index Terms*— Ambisonics, audio watermarking, rotation matrix, spatial masking**

I. INTRODUCTION

An audio watermark is a kind of digital watermark a marker embedded in an audio signal, typically to identify owners of the audio. Watermarking is the process of embedding information into a signal (e.g. audio, video or pictures) in a way that is difficult to remove. If the signal is copied, then the information is also carried in the copy. A signal may carry several different watermarks at the same time. Watermarking has become increasingly important to enable copyright protection and ownership verification.

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.

Spreading spectrum is done by a pseudo noise (PN) sequence. In conventional SSW approaches, the receiver must know the PN sequence used at the transmitter as well as the location of the watermark in watermarked signal for detecting

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hidden information. This is a high security feature, since any unauthorized user who does not have access this information cannot detect any hidden information. Detection of the PN sequence is the key factor for detection of hidden information from SSW.

Although PN sequence detection is possible by using heuristic approaches such as evolutionary algorithms, the high computational cost of this task can make it impractical. Much of the computational complexity involved in the use of evolutionary algorithms as an optimization tool is due to the fitness function evaluation that may either be very difficult to define or be computationally very expensive.

A. *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, 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 obtaining the instantaneous frequency spectrum of nonlinear and nonstationary sequences. These sequences can consequently also be dealt with using the empirical mode decomposition. This is not going to cover the plotting of the instantaneous frequency spectrum using the Hilbert transform, so EMD algorithm is used for obtaining the instantaneous frequency spectrum.

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 extrema (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.

B. *Psychoacoustic Model*

Psychoacoustics is the scientific study of sound perception. More specifically, it is the branch of science studying the psychological and physiological responses associated with sound (including speech and music). It can be further categorized as a branch of psychophysics.

The psychoacoustic model is based on many studies of human perception. These studies have shown that the average human does not hear all frequencies the same. Effects due to different sounds in the environment and limitations of the human sensory system lead to facts that can be used to cut out unnecessary data in an audio signal.

The two main properties of the human auditory system that make up the psychoacoustic model are:

- Absolute threshold of hearing
- Auditory masking.

Each provides a way of determining which portions of a signal are inaudible and indiscernible to the average human, and can thus be removed from a signal.

II. BACKGROUND STUDY

An audio watermarking technology offers means to open digital communication channels within audio content. Given this technology, the question remains as to the nature of the data that is carried by the watermarks. The typically limited data rate of such channel suggests that the transmitted data be the shortest possible. This leads towards rather rigid data structures that will be a direct function of the watermarks purpose. In addition, the choice of a (set of) watermark layers will typically be driven by the applications expected environment.

- Watermarking in the audio signals conventionally limit of wavelet approach which is that the basic functions are fixed, and thus they do not necessarily match all real signals.
- The method in Watermarking methods is not robust to attacks such as band-pass filtering and cropping, and no comparison to watermarking schemes.
- 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.

In existing research copyright protection of digital media by embedding a watermark in the original audio signal has been used. The watermark is associated with a

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synchronization code to facilitate its location. 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 works against attacks such as additive noise, MP3 compression, requantization, cropping and filtering.

III. METHODOLOGY

A. Input Signal and Segmentation

The input signal is first segmented into frames and EMD is conducted on every frame to extract the associated IMFs

Any signal x(t) is expanded by EMD as follows:

$$x(t)=\sum_{j=1}^{C} IMF_j(t)+r_C(t)$$

where is the number of IMFs and $r_C(t)$ denotes the final residual.

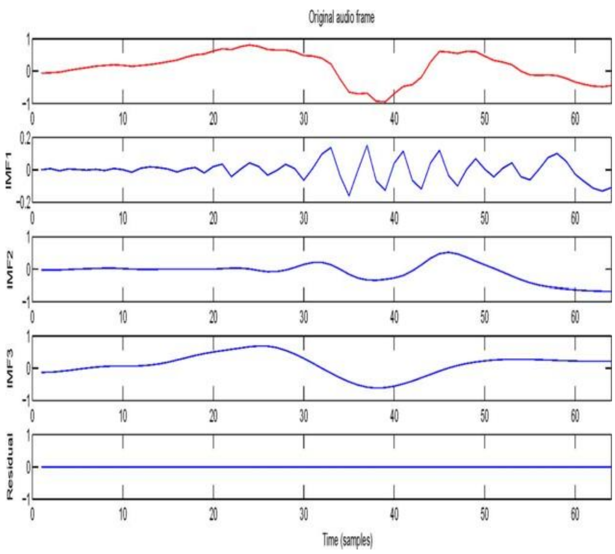

Fig.1 Decomposition of an Audio Frame by EMD

B. Synchronization Code

To locate the embedding position of the hidden watermark bits in the host signal a SC is used. Let U be the original SC and V be the unknown sequence of 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 T.

C. The Psychoacoustic Model

The branch of psychoacoustics examines the concept of auditory masking and its effect on compression. Within each subband where blurring occurs the presence of a strong tonal signal can mask a region of weaker signals. If the noise resulting from approximation of sound data can be kept below the masking threshold for each partition, then the compression result should be indistinguishable from the original audio data.

Bit Allocation

Through an iterative algorithm, the bit allocation uses information from the psychoacoustic model to determine the number of code bits to be allocated to each subband. This process can be described using the following formula:

$$MNR_{dB} = SNR_{dB} - SMR_{dB}$$

where

- MNR_{dB} is the mask-to-noise ratio
- SNR_{dB} is the signal-to-noise ratio, given with the MPEG audio standard in a table
- SMR_{dB} is the signal-to-mask ratio, derived from the psychoacoustic model

Then the subbands are placed in order of lowest to highest mask-to-noise ratio, and the lowest subband is allocated the smallest number of code bits and this process continues until no more code bits can be allocated. Two nested iteration loops called the rate loop and the distortion loop serve to quantize and code in MP3 encoders. The quantized values are coded using Huffman methods, which is lossless.

D. Watermark Embedding

Before embedding, SCs are combined with watermark bits to form a binary sequence denoted by $m_i \in \{0,1\}$,i-th bit of watermark is shown below.

Sync-code	Watermark bits	Sync -code

Fig.2 Data structure (m_i)

Proposed: Watermark embedding steps

Step 1: Split original audio signal into frames.

Step 2: Decompose each frame into IMFs.

Step3: Psychoacoustic model is applied to determine the masking thresholds for each frame $T_p = 3.97\sqrt{2\sigma_{pB}^2}$

Step4: Embed P times the binary sequence into extrema of the last IMF(IMF_C) by QIM (Quantization Index Modulation)

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

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where ei and e_i^* are the extrema of the host audio signal and the watermarked signal respectively. sgn function is equal to "+" if e_i is a maxima, and " "if it is a minima denotes the floor $\lfloor\ \rfloor$ function, and S denotes the embedding strength chosen to maintain the inaudibility constraint.

Step 5: Reconstruct the frame EMD^{-1} using modified and concatenate the watermarked frames to retrieve the watermark signal.

3.5 Watermark Extraction

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 m_i^* from e_i^* using the following rule.

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

Step 5: Set the start index of the extracted data, y, to I=1 and Select L=N_1 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 $\geq \tau$ then V is taken as the SC and go to Step8.Otherwise proceed to the next step.
Step 7: Increase I by 1 and slide the window to the next L=N_1 samples and repeat Step 6.
Step 8: Evaluate the similarity between the second extracted segment V'=y(I+N1+N2:I+2N1+N2) and bit U by bit.
Step 9: I$\leftarrow$I+ N_1+ N_2 , of the new value is equal to sequence length of bits, go to Step 10 else repeat Step 7.
Step 10: Extract the P watermarks and make comparison bit by bit between these marks, for correction, and finally extract the desired watermark.

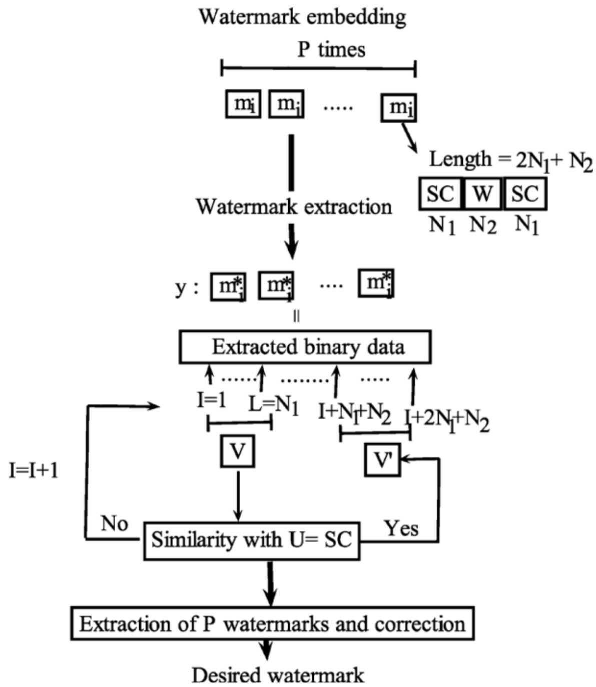

Fig.3 Embedding and Extraction Processes.

E. Performance Analysis

The performance of the proposed 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 and Normalized cross-Correlation(NC).

To evaluate the watermark detection accuracy after attacks, used in this system and it is defined as follows

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

Where $\oplus$ is the XOR operator and MxN are the binary watermark image sizes. W and W is 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 use the measure defined as follows:

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

A large NC indicates the presence of watermark while

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a low value suggests the lack of watermark. Two types of errors may occur while searching the SCs:

1. The False Positive Error (FPE) and
2. 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

$$P_{FPE}=\frac{1}{2^p}\sum_{k=p-T}^{p} C_p^k$$

$$P_{FNE}=\frac{1}{2^p}\sum_{k=p-T}^{p} C_p^k\,(\mathrm{BER})^k(1-\mathrm{BER})^{p-k}$$

IV. EXPERIMENTAL RESULT

Initially the original signal has been taken to embed the watermark. Then audio signal has to be segmented into frames. Each frame can be decomposed into IMF. Watermarking via EMD on signal can be processed and corresponding EMD and inverse EMD signal can be generated in the original signal. Then the watermark can be extracted from the secret image. Then Psychoacoustic model is applied to determine the masking thresholds for each frame. The corresponding EMD and inverse EMD signal can be generated for psychoacoustic model. Then the watermarking bit can be embedded to the original signal. The extraction of watermark bit can be done on the psychoacoustic based watermarked embedded signal. The watermark extracted image of original image shows less distortion of image than with existing system. The performance of the proposed system can be compared with the existing system performance. Finally Performance can be compared for computation time, Peak Signal to Noise Ratio (PSNR) to calculate the distortion rate performance can be obtained respectively.

The proposed system results in increased performance of the watermarking method, it gives robustness against various attacks and no distortion is obtained. Results of the individual watermarking technique have been compared on the basis of PSNR (Peak Signal to Noise Ratio), RMSE (Root Mean Square Error).

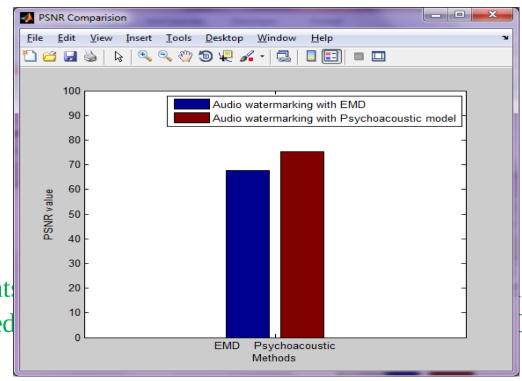

Fig.4 Psychoacoustic Model with EMD comparing the PSNR values.

The above graph shows the performance evaluation for existing and proposed system. PSNR value has been calculated for both existing and proposed system. High PSNR value has been achieved for proposed audio watermarking with psychoacoustic model than with EMD.

PSNR Value for audio watermarking with EMD is 60 decibels and PSNR value for proposed audio watermarking with psychoacoustic model achieves 70-80 decibels. Higher PSNR value shows higher level of accuracy. Thus the proposed method gives out higher level of accuracy than with existing system.

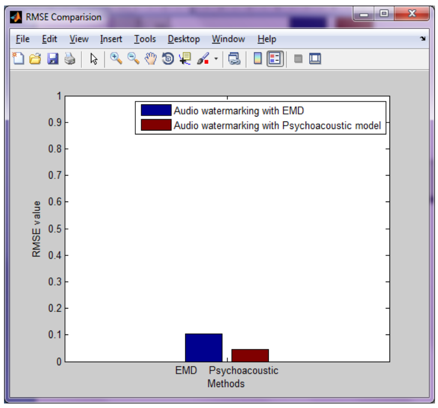

Fig.5 Psychoacoustic Model with EMD comparing the RMSE values

The above graph shows the performance evaluation of existing and proposed system with Root Mean Square Error (RMSE) value. Less RMSE value can be obtained for proposed method than with existing system.

RMSE value for existing method is between 0.1-0.2. This depicts that the error rate after watermarking to original signal is high in rate. RMSE value for proposed system is in the range of 0-0.1. This depicts that the error rate is low after watermarking to the original signal.

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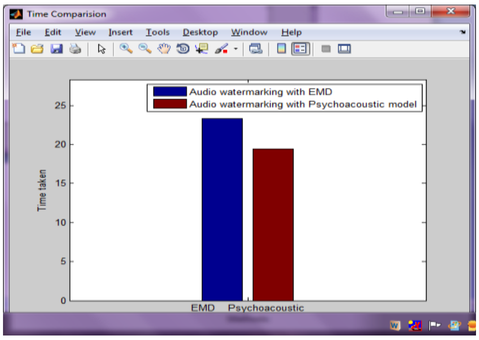

Fig.6 Time Comparison

The computation time can be compared for existing and proposed system. Existing system EMD takes high computation of time. Proposed system psychoacoustic model takes less amount of time for computation. This is because the computation time may vary depending on the resources used. Psychoacoustic model makes use of lesser amount of time to embed watermarking bit and extraction process. Watermarking via EMD uses larger amount of time to do the process of embedding and extracting watermarking bits in the original signal.

V. CONCLUSION

In this research the psychoacoustic models are applied to the EMD based audio digital watermarking is proposed. The characteristics of the proposed watermarking scheme human auditory and psychoacoustic model are used for much more improving the performance of the watermarking method. Psychoacoustics is based heavily on human anatomy, especially the ear's limitations in perceiving sound as outlined preciously. The psychoacoustic model provides for high quality lossy signal compression by describing which parts of a given digital audio signal can be removed (or aggressively compressed) safely that is, without significant losses in the (consciously) perceived quality of the sound. Watermark is embedded in very low frequency mode (last IMF), thus achieving good performance against various attacks. Watermark is associated with synchronization codes and thus the synchronized watermark has the ability to resist shifting and cropping. Data bits of the synchronized watermark are embedded in the extrema of the last IMF of the audio signal based on QIM. The watermark strength is properly adjusted based on weighting factors derived from the proposed psychoacoustic models.

A more sophisticated psychoacoustic model can give a better evaluation of the masking effect, thus gives a larger space for watermarking. Besides, the key field of watermarking scheme includes the original host signal; thus it is a private watermarking scheme in nature. Therefore, the range of application is limited. Thus, the research on public watermarking schemes which is the future work.

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