A Novel Audio Watermarking Technique Using Data Dependent Decomposition and Quantization

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DECLARATION

I, Vinay Sharma, hereby declare that the work which is being presented in the dissertation entitled, "**A Novel Audio Watermarking Technique Using Data Dependent Decomposition and Quantization**" in partial fulfilment of the requirement for the award of degree of Master of Engineering in Electronics and Communication submitted in Electronics and Communication Engineering Department of Thapar University, Patiala, is an authentic record of my own work carried out under the supervision of Dr. Kulbir Singh, Associate Professor, ECED and refers other researchers work which are duly listed in the reference section

The matter presented in this dissertation has not been submitted in any University/Institute for the award of degree.

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It is certified that the above statement made by the student is correct to the best of my knowledge and belief.

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ABSTRACT

In this age of internet and with the advent of digital systems, audio, video and other documents are available in digital format. Internet has become a highway for digital data transportation. Many software's and digital signal processing tools have been developed and distributed through the World Wide Web, thus providing tools to unauthorized personnel who make perfect copies of digital material. This may lead to large-scale unauthorized copying which might harm the software, film, book, and music industries. These threats over protecting copyright have lead to significant research in finding methods to hide copyright messages and serial numbers into digital media. These can help to identify copyright violators, and help to prosecute them. Watermarking can provide security of data depending on the type of application. For example in applications where full security is not needed watermark can be inserted to provide content monitoring and in-band signalling. Some characteristics of watermarking are robustness, Imperceptibility and capacity. User can work to improve any characteristic or all of three depending on the application of watermark.

In the proposed algorithm, robustness, imperceptibility and capacity are targeted. A new method called empirical mode decomposition is used to decompose the audio signal. Quantization index modulation is used to embed the watermark. This method is a hybrid technique which utilizes both time and frequency information. EMD is adaptive or fully data driven unlike some other fixed kernel based methods. Quantization index modulation utilizes an array of quantizers to embed data in the audio samples. In this thesis QIM is varied by making it data dependent or adaptive. Along with literature survey some concepts on non stationary signal analysis have been introduced. Because audio signal is a non stationary signal. Also some techniques to quantify and analyse non stationary signals have been introduced thus providing a base for research work. Proposed algorithm has been simulated in MATLAB. The results are given with discussions. The proposed algorithm is also compared with existing methods and numerical values are compared. From the results it has been found that the proposed method has high capacity and capacity is compared with existing methods in the results section. Also the proposed is fairly robust to most of the attacks utilized during simulation process.

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Abbreviation

IMF	Intrinsic Mode Function
EMD	Empirical Mode Decomposition
QIM	Quantization Index Modulation
AWGN	Additive White Gaussian Noise
STFT	Short Time Fourier Transform
DWT	Discreet Wavelet Transform
DCT	Discreet Cosine Transform
SVD	Singular Value Decomposition
PCA	Principal Component Analysis
LSB	Least Significant Bit
PN	Pseudo Random Noise
DFT	Discreet Fourier Transform
LWT	Lifting Wavelet Transform
FFT	Fast Fourier Transform
PPG	Point To Point Graph
PAM	Psycho Acoustic Model
ICA	Independent Component Analysis
SNR	Signal to Noise Ratio
SWR	Signal to Watermark Ratio
ODG	Objective Difference Grade

CHAPTER 1
INTRODUCTION

This chapter introduces the need and motivation behind watermarking. It introduces the concept of watermarking and history of watermarking. An overview of the terms used in the literature will be given. All subsequent definitions will be presented for further elaborations in coming chapters.

1.1 NEED OF WATERMARKING

With the dawn of digital age, data has been digitized be it audio, video and text. With the boom of internet people have found new ways to communicate and distribute data. But with advantages some serious issues have emerged and the biggest is the copyright infringement. With latest tools unauthorised people can distribute large amount of data .This has led to business losses for music, text and entertainment industry. Also communication over internet is not secure. To deal with these problems cryptography was looked upon with great interest. Here data is first encrypted and sent by digital mediums. At end user terminal this encrypted data is decrypted and used. But the problem remains the same .Because after decryption data still can be distributed and tempered with. To provide remedy to this problem watermarking has been introduced. Watermarking is a subtype of stegnography where a key or information is concealed inside the original data without anyone knowing.

1.2 BRIEF HISTORY OF WATERMARKING

The first audio watermark can be traced to a patent filed by Emil Hembrooke in1954 which was titled “Identification of sound and like signals” [1] where an identification code was inserted into the music. Later many systems and algorithms were designed not just to identify but control devices. More interest in watermarking surged from copyright protection .watermark was used with cryptography to enhance security. Muzak Corporation [1] pioneered the owner identification by using a notch filter and Morse code. Many corporations have used watermarking for in band signalling like Lynch Carrier Systems Inc in 1962 [2] to control telephony systems. And since then many new techniques have been introduced, which will be introduced in chapter two.

1.3 ILLUSTRATION OF WATERMARKING ALGORITHM

In this section we will give definitions, characteristics and tradeoffs of watermarking algorithm.

1.3.1 WATERMARKING DEFINITION

Audio watermarking is the process of incorporating watermark in the host file. Watermark can be an image, audio or some code. Watermarking process is divided in two parts. Embedding at transmitter side and extraction at receiver. During embedding we take an audio file and find some characteristics of that audio which can be changed or modified without affecting the signal quality in all sense .Then we take up the watermark and embed it into those characteristics of audio data. The watermark should be inaudible in the host at receiver side audio is intercepted and watermark is extracted from the audio.

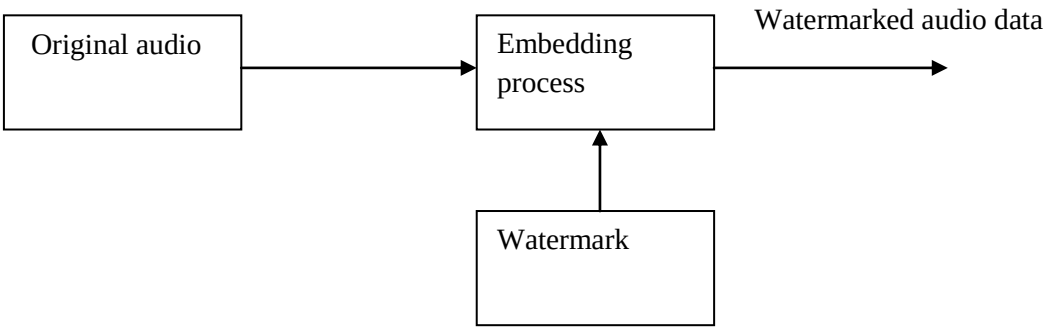

Figure- 1.1: Embedding process

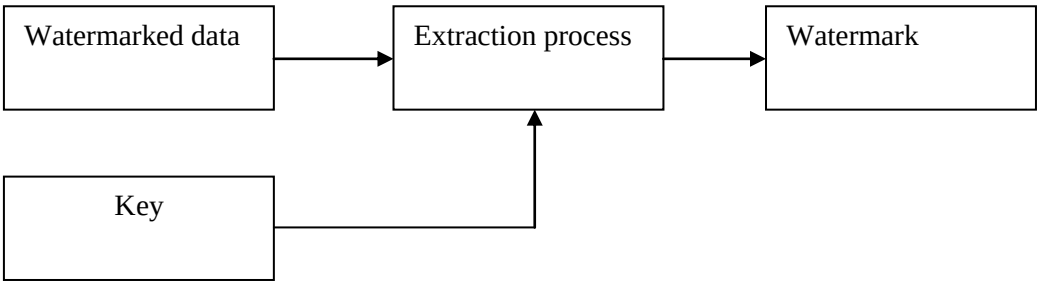

Figure- 1.2: Extraction process

1.3.2 WATERMARKING CHARACTERISTICS AND CLASSIFICATIONS

For a watermark to be considered as an effective tool in audio media it must satisfy some norms such as imperceptibility, robust against attacks and security. These days capacity constraint is also an area of focus [3].

- **Imperceptibility** Watermarking must not degrade the perceptual quality of cover audio file, it should be inaudible. Otherwise its usage is meaningless.
- **Robustness** As a tool watermarking should be able to ensure detection accuracy when extracted. In general most of the channels are active so they can distort the audio file, so at the time of extraction the received file is different which incorporates trouble for watermark. Therefore any watermarking scheme must be immune to such modifications to a certain extent
- **Security** It infers that the water mark should be impossible to detect even though the algorithm is known to others. As in line with kerckhoffs principle that a cryptosystem should be secure even if the algorithm is known except the key. Key is unknown to the unauthorized user.
- **Capacity** It describes the quantity of bits to be embedded per time interval. The theoretical capacity of audio watermarking relies on the proportion of silence to the cover signal and embedding algorithm. Due to trade off between capacity and robustness it is quite hard to estimate the real capacity. Also capacity is less important in watermarking schemes

Watermarking can be classified according to the domain and type of key required for extraction. To embed watermark in host signal its characteristics are required. These characteristics of signal can be in any domain. The type of domain stems from different requirements of user and application. Each domain has its pros and cons and depending on the situation we weigh them accordingly for example some application do not need imperceptibility constraint but for some it is an issue. On the basis of domain watermarking can be categorized in three types

- **Time Domain** These techniques embed watermark in time domain. That is host signal is not transformed into any other domain. Most common example of time domain watermarking is Least Significant Bits (LSB) modulation or encoding.

Where LSB of the host is changed according to the value of watermark. It also includes echo hiding, patchwork and many more such techniques. These techniques are simple and have less computational complexity. But with advantage of simplicity they are also susceptible to different range of attacks, and are nearly obsolete. But some new time domain methods are discussed in literature review showing robustness.

- **Transform Domain** In these kinds of techniques original audio is transformed into frequency or other domain. A watermark is inserted in the corresponding domain. A domain is just another representation of the same signal with different prospective. Domain transformation can show new light on a signal revealing its new characteristics and its working. The very basic transform used in these types of techniques is the Fourier transform, where a signal is transformed into frequency domain and the coefficients of the transformed signal are changed according to watermark. But this transform has a lot of drawbacks, due to which new transforms are proposed and used in watermarking for example Short-Time Fourier Transform (STFT), Discrete Wavelet Transform (DWT) and Discrete Cosine Transform (DCT).
- **Hybrid Techniques** Combinations of time and frequency or transform domain techniques have lead to hybrid techniques .Mainly two methods Spread-Spectrum and Empirical Mode Decomposition are known to be hybrid. But sometimes amalgamation of different methods can be called as hybrid where both time and frequency information is needed. These techniques have advantages of both time and frequency. A lot has not been done in this area and it is open for research.

Watermarking techniques can also be classified in three types depending whether we need original watermark at the receiver side for extraction of embedded watermark.

- **Blind** In these types of methods watermark or any key is not required for extraction at receiver side.
- **Non-Blind** In these methods original watermark or key is required for watermark extraction
- **Semi-Blind** In these methods a key is used or a part of watermarking information is required for extraction

1.3.3 ATTRIBUTES OF ROBUSTNESS

Definition of robustness is in complete without discussing the fragility of watermark.

- **Fragile watermarks** Any modification to the watermarked signal of this kind either removes or alters the watermark. So only an authentic and original signal would pass the watermark check algorithm.
- **Semi- fragile** These methods can withstand some basic watermarking attacks. But breaks down at highly sophisticated attacking software and tools
- **Robust** They are highly robust to any kind of attack. And is the ultimate choice for copyright protection.

1.4 APPLICATION OF WATERMARKING

Major interest and motivation in recent times behind watermarking resulted from copyright protection. But watermarking has been used in other field as well here are some of the applications of watermarking.

1.4.1 Copyright protection

To protect intellectual property content owners can embed a watermark having information about the ownership of the audio file or data. This watermark can be verified at the times of need like a court where it can be used as admissible evidence.

1.4.2 Authentication and temper detection

With advancement in digital systems and tools, people these days can tamper with the data and modify it. So watermarks are used which are fragile to find if any tempering has been done. Also watermarks are used to authenticate users where internet application or mobile phones are used.

1.4.3 Tracking and Identification of digital content

Watermarks have been used to track the digital content .so if any misuse or unauthorised distribution comes to light then sources responsible can be traced. Broadcast monitoring is another use where data is tracked and monitored. And transmission verification can be sent using the watermark.

1.4.4 Copy prevention and in band signalling

Sometimes a service provider may require the need to prevent any user or personnel from copying the data. These problems can be solved by watermarking where a certain type of watermark can represent data that cannot be copied and any attempt to copy will lead to some distorted data. In applications where security is not an issue simple watermarking algorithms which are obsolete can be used for inband signalling. This type of signalling was used in earlier telephony equipment. Also in some new applications different watermarks are used to perform different tasks for example an application using internet senses a watermark which initiates the device to find information about the area and location.

1.5 ORGANIZATION OF THESIS

This thesis is organized in five chapters. All the work is distributed and explained in these five chapters.

- **Chapter 1** In this chapter the notion of watermarking with its definitions and applications are discussed. Some historical facts on watermarking are also provided. This chapter gives the motivation behind the interest of researcher in this area.
- **Chapter 2** In second chapter literature review is performed. Concept of nonlinear and non stationary signals will be presented. And short coming of Fourier analysis on non stationary signals will discussed. Also some techniques used for non stationary signal analysis are given. Concept of Quantization Index Modulation is discussed. And various analytical tools and tests for quantifying watermarking are discussed.
- **Chapter 3** In this chapter proposed work and methodology of proposed technique are given. This chapter gives the mathematical base along with embedding and extraction steps of proposed method.
- **Chapter 4** In this chapter results are given. And these results are discussed.
- **Chapter 5** In this chapter conclusion of proposed work is given along with future work

CHAPTER 2
LITERATURE REVIEW

The literature review gives all the definitions and key concepts which were required to carry out the research work. Aspects of various techniques have been studied to build a firm foundation for developing the proposed algorithm. It also gives the gaps and motivations behind the work.

2.1 NON STATIONARY SIGNAL ANALYSIS

Audio signal is a form of non stationary signals. To characterise a signal in frequency domain we have to use Fourier analysis which breaks the signal into an infinite number of harmonics. But the signal has to be stationary and linear to hold Fourier analysis. So decomposition of a non stationary signal will result in infinite harmonics, the existence of these harmonics will divert energy to a wide frequency range. To reconstruct the signal the frequency representation has to be truncated .But the constrain of energy conservation will prevent faithful representation of energy spectral density in frequency space and reconstructed signal in time domain. Also Fourier introduces the concept of a negative frequency. So mathematically it may make sense but physically it does not.

The biggest disadvantage of Fourier analysis is that it may give frequency resolution but time resolution is lost. That is we do not know which frequency exists at a particular time instant. To remedy these problems some methods evolved .The methods are given as

2.1.1 SHORT TIME FOURIER TRANSFORM

To decompose the non stationary data Short Time Fourier Transform has been used in literature. Here it is assumed that data is piecewise stationary that is stationary for a certain time period which can be 10 millisecond as used in current literature. It is a linear approach for time –frequency representation [4].

$$STFT(t', u) = \int \left(f(t) \cdot w(t - t') \right) . e^{-j 2\pi u t} \, dt \qquad (2.1)$$

Where $w(t - t')$ is the window and t' is the time parameter and u is the frequency parameter

It decomposes the signal to elementary components called atoms. Atoms are obtained from windowing the Fourier signal, just like passing the signal through a bank of filters. And the filters are band pass filters.

The basic steps are

- Choose a window function
- Place the window on signal at the location t=0
- Truncate the signal using this window and calculate Fourier transform
- Slide the window
- Perform above steps till you reach the end of signal

2.1.2 SPECTROGRAM

By squaring the Short Time Fourier Transform , spectrogram is obtained. It is simple and basic method. Different types of spectrogram come from different windows used. Since it is based on the Fourier analysis the data is assumed to be piecewise stationary. And in case of audio data this assumption cannot always be justified.

2.1.3 WAVELT ANALYSIS

The major issue with Short Time Fourier Transform is the window of fixed size for the whole signal. If the size of window is reduced we may get time resolution but frequency resolution may be lost. For a large window size a good frequency resolution may occur but on the price of time resolution. To eliminate this short coming a new approach called wavelet analysis has been introduced in literature.

The wavelets are like STFT but with adjustable window size with the following [4]

$$W(a,b;X,\varphi) = |a|^{-1/2} \int_{-\infty}^{\infty} X(t)\, \varphi^*\left(\frac{t-b}{a}\right) dt \qquad (2.2)$$

Where $\varphi^*(\cdot)$the basic wavelet function, a is the dilation or scaling parameter and b is the shifting factor or point of translation form origin. The type of wavelet depends on the type of window. Also it provides uniform resolution to all scales used. This is a new method and a lot of interest has been shown in using wavelets. At this time it has been used in images but lately some researchers have used it in audio watermarking problems also.

2.1.4 WIGNER – VILLE DISTRIBUTION

Wigner – Ville distribution can also be called as Heisenberg Wavelet. This distribution is calculated by taking the Fourier transform of the central covariance function. For a given series,$x(t)$ the central variance is

$$C_c(\tau, t) = x\left(t - \frac{1}{2}\tau\right) x^*\left(t + \frac{1}{2}\tau\right) \quad (2.3)$$

The Wigner – Ville distribution is given as [3]

$$V(\omega, t) = \int_{-\infty}^{\infty} C_c(\tau, t) e^{-i\omega\tau}\, d\tau \quad (2.4)$$

The principle issues with this method are the cross terms generated for certain frequency ranges. Because at these frequencies, negative power exists. This problem can be removed by using a window. But then it will have all the short comings of windowed Fourier analysis.

2.1.5 EPIRICAL ORTHOGONAL FUNCTION EXPANSION

The empirical orthogonal expansion [4], also called singular value decomposition or principal component analysis. The concept of EOF is :

A real signal $z(x, t)$ can be decomposed and reduced by

$$z(x, t) = \sum_1^n a_k(t) f_k(x) \quad (2.5)$$

If

$$f_i \cdot f_k = \delta_{jk} \quad (2.6)$$

Here f_k is the orthogonal basis and is the aggregation of Eigen function given as

$$C.f_k = \lambda_k f_k \quad (2.7)$$

Here C is the summation of inner products of variables.

EOF is very different from the other kernel based approaches like Fourier because the basis for expansion is derived from data itself so each expansion is unique. The expanded data is automatically sorted on the base of variance distribution. First component has large variance then second and so on variance gets reduce for every coming component. This is the biggest flaw of EOF; it gives only variance distribution not any idea about frequency or other components. And any component by itself does not give any physical meaning .It has been used mainly in data classification problems.

2.1.6 EMPIRICLA MODE DECOMPOSITION AND HILBERT SPECTRUM

To understand this we have to first understand the basic concept of instantaneous frequency. The definition and concept of instantaneous frequency has been highly controversial. In the Fourier spectral space, frequency is defined for the kernel function spanning the entire signal length. These kernel or basis functions utilized are sin or a cosine function. So a full wave or oscillation is required to specify a local frequency. Due to this the definition of instantaneous frequency for non stationary data will not make any sense except for mono component [4] defined by Cohen (1995). But to get the true sense and meaning of instantaneous frequency restrictive conditions were imposed by Gabor (1946), Bedrosian (1963) and Boashash (1992). That was, for an instantaneous frequency to have meaning, Fourier transform of its real part should have only positive frequency. But still this limitation is global not local to the signal.

From the notion of instantaneous frequency came the idea of intrinsic mode functions which were localized frequencies. This provided a way to physically interpret the data.
The necessary conditions for the existence of instantaneous frequency are

- Signal function should be symmetrical with respect to the local zero mean
- Number of zero crossing and extrema values should be equal

From these definitions comes the Intrinsic Mode Function (IMF) that used for EMD. For a function to be called as Intrinsic Mode Function it should have the following two conditions.

- The number of zero crossings and the number of extremas should be same or utmost vary by one
- The mean of envelope at any point, characterized by local minima and maxima envelops should be zero.

The second condition is a new concept to change global requirements in local requirements. Here local mean comes from local time scale which is not easy to define for a non stationary process. Intrinsic modes are the local oscillations imbibed in the data.
In Empirical Mode Decomposition, we take the following assumptions to decompose the signal

- Local time scale or characteristic time is outlined as the time period between successive extrema values.

- At least two extrema values are present in the signal – one minima and one maxima

The basic steps of decomposition are

- Locate all extrema points of $X(t)$
- Make an envelop a by interpolating the maxima points
- Make an envelop b by interpolating minima points
- Take the mean of a and b
- Subtract the mean from the signal
- Check if result from step five is an IMF

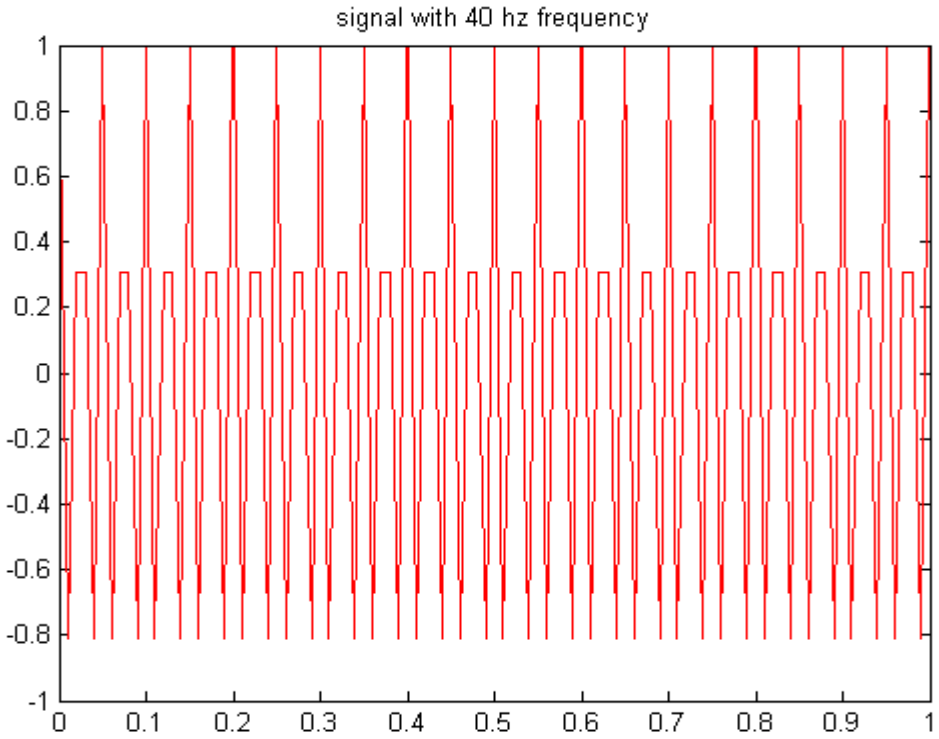

Figure - 2.1: Cosine wave

As seen from the Figure 2.1 a cosine wave with frequency equals to 40 Hz is decomposed using Empirical Mode Decomposition. Envelops are interpolated using maxima and minima points as seen in Figure 2.2. These envelops are used to find an average signal which is depicted in Figure 2.3.

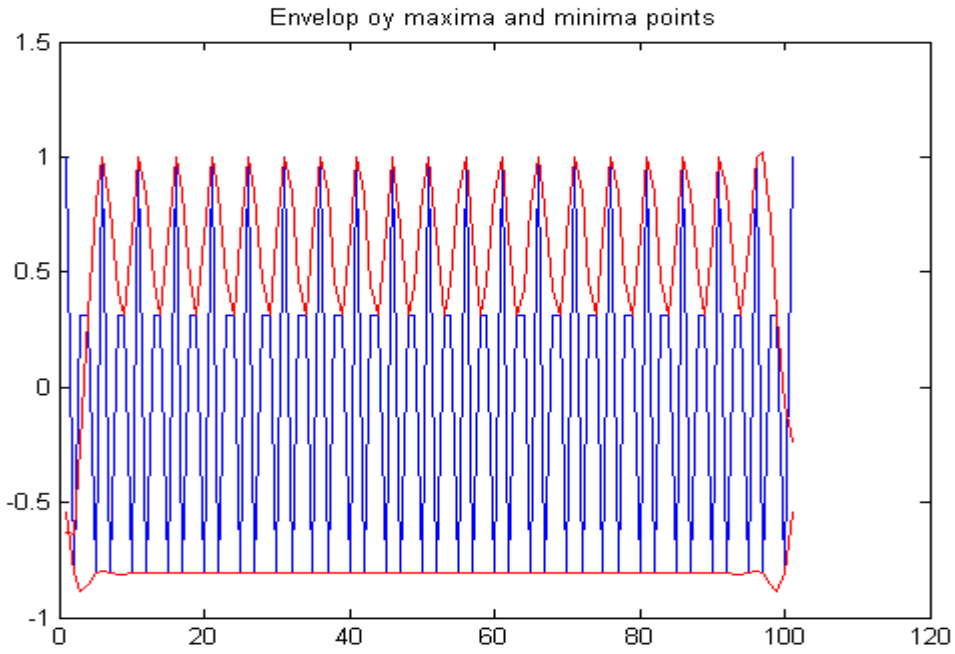

Figure - 2.2: Envelop of maxima and minima points

This average signal is subtracted from the orignal signal .a selection criterion dtermines whether the outcome of subtraction is an IMF as seen in Figure 2.4

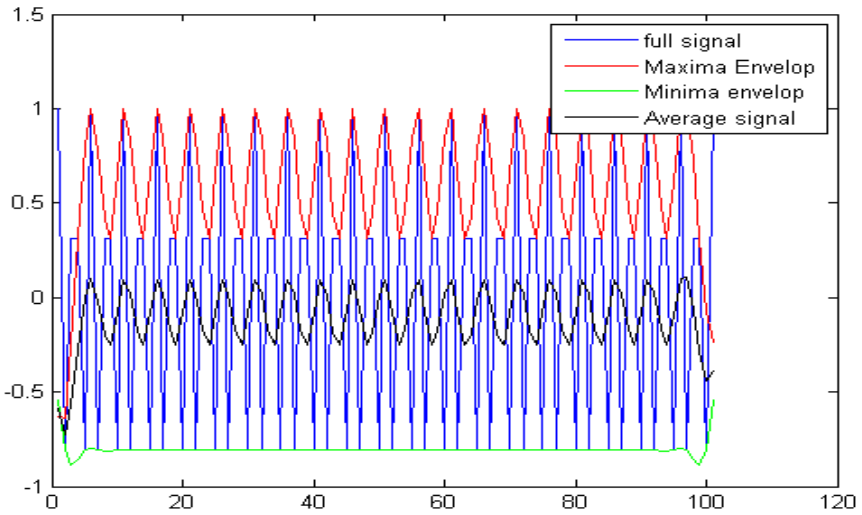

Figure - 2.3: Average signal depicted in black

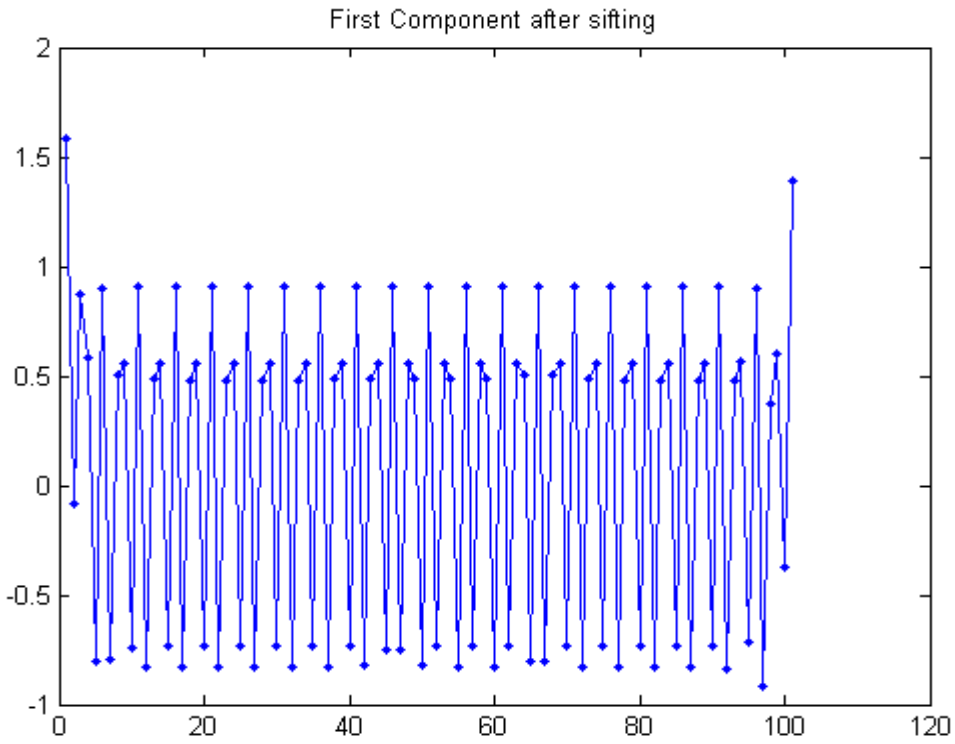

Figure - 2.4: First IMF of signal

If the result is an IMF, stop. But if condition is false keep repeating steps 1, 2,3,4,5. This process of removing or extracting IMFs is called sifting process. A full decomposition algorithm with mathematical analysis will be given in chapter three. Here only the relevant concept will be given. After decomposing a signal into Intrinsic Mode Functions, Hilbert transform for each IMF is computed and data can be expressed as [3]

$$X(t) = \sum_{j=1}^{n} a_j(t)\, exp\left(i \int w_j(t)\, dt\right) \qquad (2.8)$$

The same IMFs can be expanded in Fourier but this will give a constant amplitude and constant frequency in expansion. Here Hilbert transform will provide variable amplitude and variable frequency component, which enables depiction of non stationary data.

2.2 EXISTING AUDIO WATERMARKING ALGORITHMS

Watermarking can be done in mainly three domains. Many researchers have proposed algorithms in the all of these domains. Some basic methods that are used in literature fall in these categories. These methods have been utilized as a base or building block in all the literature and appear with many variants. These techniques are

- **Least Significant Bit (LSB) Coding** substitution of the LSB of each point with bits from the watermark
- **Phase Coding and Phase Modulation** As a fact that human auditory system is insensitive to phase changes watermark is embedded into the phase
- **Echo Hiding** Embedding watermarks by adding delayed echoes at certain time or points.
- **Spread Spectrum Data** is converted into high frequency components through codebook or predefined basis.
- **Patchwork Technique** It uses a pseudorandom process to change a certain data statistics to embed watermark.

2.2.1 TIME DOMAIN

In time domain approaches watermark is inserted on time basis.

In 2003 Aweke Negash Lemma *et al.* [5] proposed the idea of temporal domain audio watermarking. This system was called modified audio signal keying (MASK). In this system a code or watermark was planted by altering the envelop of host signal. This modification was done with a pre processed version of random sequence carrying payload. In this, filter would extract the parts of audio signal suitable for watermark information .Then this signal was multiplied with watermark and added to original signal. The watermark is selected by keeping in mind that multiplying it with data would modify the envelop. At detection side a temporally synchronized detection for signals is used. Here there is no time offset between the detector and embedding module. Another method employed at detector is temporally non synchronized detection, employed when the received signal is a delayed version or a time scaled modified version of transmitted signal.

In year 2009 Nima Khademi Kalantari *et al.* [6] proposed a method which used patchwork method the basic idea of patchwork is that, host signal is divides into two subsets and signal characteristic for these subsets are selected using two secret keys. Single bit is inserted by dividing or multiplying the values of one subset and other subset is not altered. The watermark is extracted on the receiver by comparing the two subsets on the basis of their energy at a given threshold. As the watermark is inserted in some

frames of the original audio signal, therefore the probability of error is zero for attack free case. This technique is implemented in the transform domain which is wavelet domain and subsets are taken from the approximate coefficients. In order to curb the inaudibility constraint due to watermarking, an iterative approach is utilized to change the watermark strength for each frame depending on the frame.

In the year 2012 Yong Xiang *et al.* [7] proposed a dual-channel time-spread echo method for improving imperceptibility and robustness in watermarking. At the embedding stage samples of host are arranged in the form of two sub signals. It was assumed that each sub signal is a different channel. Take two different sets of echo signals here each signal is antipodal to each other. They are referred with the watermark, and are added to the two sub signals. And at the latter stage the watermarked sub signals are summed to produce an integrated watermarked signal. At decoder, the samples of the collective watermarked signal are reshuffled to reconstruct the two watermarked sub signals.

In the year 2012 Ryouichi Nishimura [8] devised a watermarking technique using spatial masking and Ambisonics. Like the echo method that uses the temporally shifted version of cover signal which is also amplitude-scaled is added like a watermark. In this schema a amplitude-scaled and spatially rotated versions of host signal are summed like a watermark. A sound lookalike is anticipated to come out between the cover and rotated version as there is no time lag between the two. Therefore on hearing the watermarked and original there would be very less difference. Watermark is depends on the difference between the cover and the rotated edition. Higher the level difference, the jester the it works. But major drawback of this method is security. It is prone to attacks.

In year 2009 Konrad Hofbaue *et al.* [9] presented a blind watermarking method that implants the watermark in the phase component of speech. This was achieved by displacing the perturbation signal of an AR speech signal representation. The watermark signal was incorporated in sub band of some frequency, this has introduced robustness against channels which have band pass filtering properties. They derived many types of shapes for pulses and these prevent ISI and this allows the pass band watermark to be produced by just filtering. For detection they used recursive least-squares (RLS) equalization-based watermark detector. This counterbalances the vocal tract filtering,

and extracts the watermarks, even if band pass filtering , nonlinear phase, additive white Gaussian noise (AWGN), and amplitude modulation are present. This makes the watermarking scheme highly robust.

2.2.2 TRANSFORM DOMAIN

In transform domain techniques mainly utilize a transform to decompose the given signal in frequency or in some other domain. Transform domain techniques are found to be robust than the time domain techniques. Here are some of the algorithms.

In 2009 Yousof Erfani *et al.* [10] proposed a method in which time spread echo hiding was used. Divide the audio signal into two channels and these channels are convolved using a kernel signal for watermarking. Then the signal is transformed in cepstrum domain. Then we compute a correlation matrix of coefficients in cepstrum domain with PN sequence used to create watermark and some tracking data. This paper proposed a new method for detection in which we take complex cepstrum transformation and take real portion of normalized correlation between cepstrum coefficient and the PN sequences. The original audio signal effect is removed at detection which makes this scheme more efficient.

In year 2013 Hong Peng *et al.* [11] proposed a learning-based watermarking scheme using kernel Fisher discriminant analysis (KFDA). Here we divide the audio signal into frames. And then by a mathematical rule we obtain two sub frames .then we randomly select audio frames for embedding watermark using a secret key. Then we apply two level DWT on selected frames ,which decompose each sub frame into two sub band coefficients .after this we calculate the energy of these coefficients . The watermarks are embedded in selected frames using energy relationship modulation technique .we reconstruct the audio signal by applying inverse DWT. For detection we use kernel Fisher discriminant analysis. Due to its non-linear learning ability it is very powerful.

In year 2013 Shuo-Tsung Chen *et al.* [12] proposed a method using wavelet-based entropy as the central idea .Divide the audio signal in segments and DWT is used on each segment. We take the low frequency coefficients and divide them into groups

Then an adaptive embedding rule is used to embed watermark based on a key. Also it is a blind method so, to detect watermark original audio signal is not needed. To detect p the watermark only values of the WBE are required

In the year 2012 Iynkaran Natgunanathan *et al.* [13] proposed a method based on patchwork for audio watermarking. In this scheme, the cover audio file is divided into sub segments; in this case it is two. The DCT coefficients are calculated for each sub segments. Then DCT coefficients which correspond to high frequency are discarded. Then the left out coefficients are broken into frame pairs. A standard is utilized to pick the frame pairs for embedding watermarks. By using a pseudo noise (PN) sequence, a watermark is implanted into the frame pairs by changing the corresponding DCT coefficients. The embedding process is fashioned in such a manner that the selection formula or benchmark used during embedding can also be used during extraction procedure. So first we locate the frame pairs which contain watermarks. And then we can pull out the watermark by utilizing the pseudorandom code as a key.

In 2011 Ming-Quan Fan *et al.* [14] proposed a method of watermarking which can with stand some audio processing attacks, the main focus on playback speed modification. In this a chaotic mapping procedure generates chaotic sequence and through secret key watermarks is generated from this mapped sequence. The audio signal is broken into frames which are non-overlapping. And DFT if performed on these frames. Then we obtain absolute mean value of these frames. Again each frame is dividing into segments and we take an absolute value of DFT coefficients of each segment .then we find relation among neighboring segments. And according to a pre defined rule the watermark bit is embedded.

In year 2011 Baiying Lei *et al.* [15] proposed a method for audio watermarking using lifting wavelet and SVD. LWT is , variant of wavelet transform. The coefficients of LWT are integers so quantization error is not introduced. At embedding side first we take LWT of audio signal, then we form non overlapping blocks of length dependent on the amount of data and number of LWT decomposition levels. The watermark signal is embedded into low frequency sub bands of these LWT coefficients. Then we take SVD transformation and embed watermark into the obtained values using quantization index modulation.

In year 2011 Pranab Kumar Dhar *et al.* [16] intoduced a watermarking schema rooted on Fast Fourier Transform. For embedding, the original audio signal is divided in the form of non -overlapping frames. Then we apply FFT to calculate the phase and magnitude spectrum of each frame. Forms the magnitude spectrum number of prominent peak are found. This is done using peak detection algorithm. Then we embed watermarks into the prominent peaks found above. We put back these modified peaks into frames. Then we apply IFFT and concatenate all frames to get back watermarked signal audio signal. For detection or extraction of watermark we apply FFT to the watermarked signal. Again we find peaks and extract watermark by performing inverse of embedding process.

In year 2010 S.T Chen *et al.* [17] proposed a method for audio watermarking which embeds watermark by energy-proportion scheme. First host audio Signal is segmented into sections. And then DWT is applied to each section. After that the SC and watermark are organized in the form of PN sequence. They are implanted in coefficients representing the low frequency. Then the characteristic curve of energy-proportion function (CCEP) is obtained. Based on the properties of this function we embed and extract watermark.

In year 2010 S.T Chen *et al.* [18] proposed a method for audio water marking using wavelet – domain audio watermarking scheme. It uses seven levels DWT, here watermark is inserted in coefficients representing low frequency. Due to attacks such as shifting or cropping in watermarked audio they embedded synchronization codes together with the watermark. These codes are used to locate the positions of the watermark in the audio signal. The watermark and code is grouped into a binary sequence B. The DWT coefficients are grouped in vector form. To balance the tradeoff between SNR and BER an optimization-based quantization scheme is used. SNR is written in the form of a watermarking cost function matrix. Then an equation connecting the cost matrix and group amplitude quantization equation is formed. The optimization is done using Lagrange multiplier. The optimum result is used to embed the watermark.

In year 2010 Vivekananda Bhat K *et al.* [19] proposed a method utilizing singular value decomposition and discreet wavelet domain. In this method audio host signal is dividing into two parts with watermark and synchronization code embedded into both part

respectively. The code was inserted by taking one part and breaking it into segments. Then mean value of each segment is computed. And code is embedded through a rule.
The watermark is embedded in the other part by taking the Harr wavelet filter. The low frequency coefficients are converted into 2- d matrix blocks. Then we compute SVD of each block. Then we compute norm, standard deviation and mean value of this block. These values are used to create quantization step interval and then each bit of watermark is embedded based on some pre defined rule. At last mw apply Inverse SVD. And after combining all blocks we perform inverse DWT. At extraction stage we use synchronization codes to locate the position of watermarks and perform DWT and take the low frequency coefficients, calculate SVD and by some pre defined rule we extract the watermark.

In 2009 Mohammad A. Akhaee *et al.* [20] devised two blind audio watermarking method using the quantization in transform domain. The transform used is called PPG (point-to-point graph). This transform increases the dimensions of a given signal. It uses the correlation of magnitude between the samples . For the first technique, hard quantization codeword's are chosen as two circles in a PPG domain. To increase robustness a guard margin is placed near the quantization levels. In next technique soft quantization codeword's are utilized. This technique is based on making patternss in the radii of points according to the watermark bit. At extraction side a histogram based extraction method is used for both methods. In this PPG points near zeros or a single quantization level are counted and histogram is compared around these levels to detect the data .

2.2.3 HYBRID OR TIME-FREQUENCY DOMAIN

Hybrid methods utilize both time and frequency domain techniques. This is a new area for work.

In the year 2003 Darko Kirovski *et al.* [21] proposed a method of watermarking using spread spectrum .They devised a number of techniques for improving the effectiveness of embedding and detection of watermark in audio. Such as psycho-acoustic frequency masking (PAFM)(this introduces an imbalance in the number of negative and positive watermark chips in the part of the Spread spectrum utilized for Watermark correlation

detection and that relates frequency spectrum which is audible.), cepstrum filtering ,block repetition coding for prevention against de-synchronization attacks and chess watermarks. In this watermarking method, the signal is made of magnitudes of many frames of a modulated complex lapped transform (MCLT) in a decibel (dB) scale. The MCLT is oversampled filters bank, providing reconstruction. After addition of the watermark, the time-domain marked audio signal is generated by combining the signal with the original phase and sending these altered frames to the inverse MCLT .

In year 2008 H.Malik *et al.* [22] proposed a scheme using spread spectrum. Unlike other SS-watermarking techniques which use entire audible frequency range it employs frequency-selective spread spectrum in which a small band of audible frequencies is used for watermarking. The suggested method utilizes a secret key to select sub bands of original signal for embedding watermark. The number of sub bands for selection is dependent on required capacity. The embedding of watermark is done in frequency domain using DFT. It uses correlation and ICA based estimator for detection and extraction of watermark.

In year 2011 Xiangui Kang *et al.* [23] devised a technique using a multi-bit spread-spectrum watermarking algorithm using geometric invariant log coordinate mapping (LCM) feature. The LCM properties are very robust to active channels. The watermark is inserted in the LCM space, but in reality it incorporated in the Fourier coefficients .These coefficients are inserted to the feature through LCM, therefore the incorporation is performed in the discreet Fourier transform domain not using interpolation, which totally eliminates the distortion arising from the mapping and non-uniform interpolation. The proposed method aims to solve geometric distortions simultaneously, such as pitch shifting, pitch-invariant time scale modification (TSM), random cropping, resample TSM, and jittering.

In year 2013 Kais Khalid *et al.* [24] proposed a method of audio watermarking utilizing Empirical Mode Decomposition (EMD). EMD decomposes audio signal in frames adaptively and transforms them into oscillatory components called as intrinsic mode function. Intrinsic mode functions are orthogonal and have zero mean. Data bits are embedded into the extreme of the last intrinsic mode function (also the low frequency components) which makes the watermarked signal inaudibility .Along with watermark,

synchronization codes are also embedded. These code give the location of the hidden watermark bits .Any kind of attack cannot harm these codes. All of this is done in time domain.

2.3 GENERAL INFORMATION EMBEDDING MODEL

A general embedding model in given in the Figure 2.5. Here v is the host signal vector given as $v \in R^N$ and information is m . These vectors are given to a function s which is the embedding function. This function maps the cover signal v to complex signal $s \in R^N$ subjected to noise and degradation in the channel.

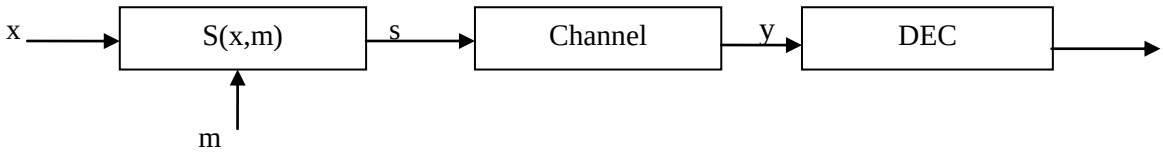

Figure - 2.5: General embedding model

At receiver a decoder extracts the data that is it estimates output by utilizing y that is channel output. In general all the embedding techniques can be classified or divided into two types.

- **Host- Interference Nonrejecting Methods** These methods hold the assumption that the host signal is a source of noise in the system. So the systems or the encoder do not exploit fully the characteristics of host signal. These methods are best utilized when watermark extraction is not blind.
- **Host–Interference Rejecting Methods** These methods consider host characteristics as an inert part of encoding and decoding process. And knowledge of host is fully utilized. Most blind extraction algorithms come under this definition. The biggest example is the Quantization Index Modulation discussed in the section below.

2.3.1 QUANTIZATION INDEX MODULATION

In the year 2001 Brian Chen *et al.* [25] proposed a new method for information embedding in digital watermarking. It is called quantization index modulation and it has many different variants. This method is a host-interference rejecting method. To

understand the concept of QIM we look at the embedding function $s(x;m)$ which is an ensemble of functions with dependent variable x the host signal and m , data. In this method a characteristic of host is chosen and modulated according to the data. The modulation process is just selection of embedding function according to data bits. The output is approximately equal to the input or distortion introduced due to watermark in host is small only if the embedding functions are identity functions. These identity functions are a set of quantizers which quantize the host signal

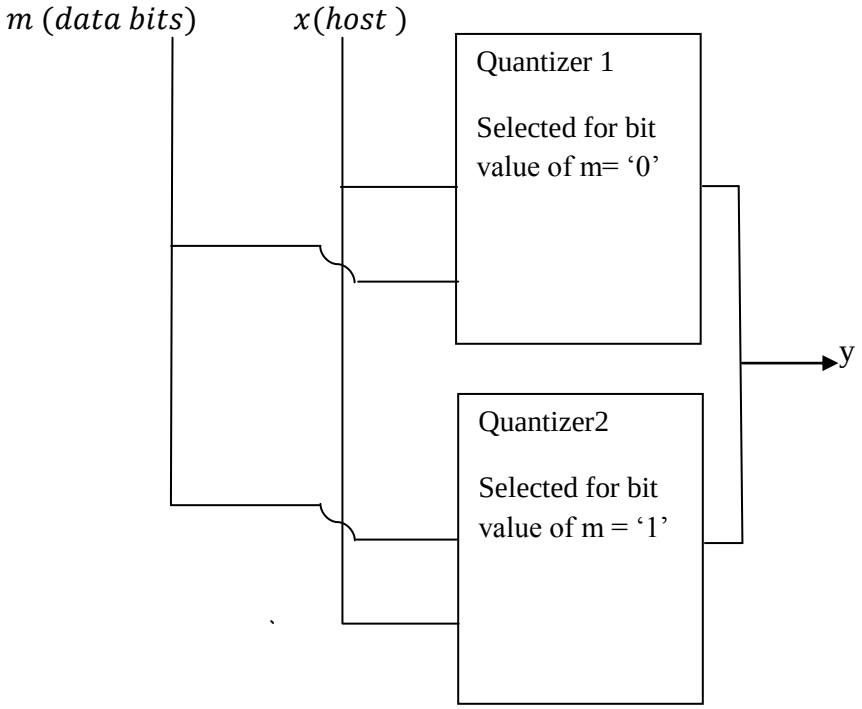

Figure - 2.6: Quantization Index Modulation process

To clear the concept let's take an example suppose host is an audio signal and the indexes to be quantized is the peak values of signal every frame (positive or negative). The data is binary and one bit is embedded at one peak. Therefore there are two quantizers from which one is selected on the basis of the incoming data bit i.e. '0' bit selects quantizer number one and '1' bit selects quantizer number two. After selecting a quantizer the quantization is done with respect to some host index or characteristic. So each quantizer has a sub section .Where quantization is done on the basis of host signal for example positive peaks have a different quantization formula than the negative peaks. From the above example two quantizers are used for two bits. The reconstruction points of two quantizers are different as seen in Figure 2.7. To represent n data bits , 2^n quantizers are

required. Information embedding rate depends on the number of quantizers used. The properties of quantizers will affect the performance parameters of watermarking.

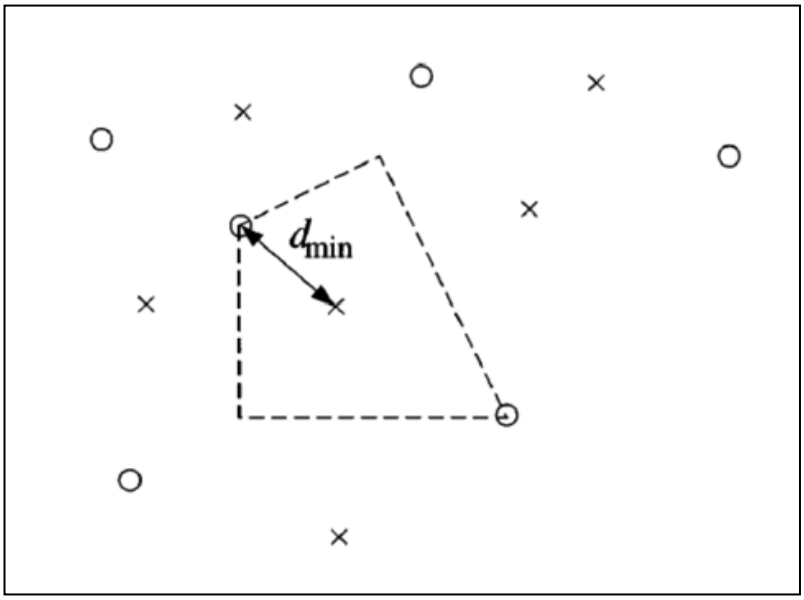

Figure - 2.7: Reconstruction points of quantizer one and two. Where o denotes quantizer one and x denotes quantizer two [25].

2.3.2 SNR USABILITY

The visual system or eyes are less sensitive to distortion than the human auditory systems. So to make a watermarking algorithm the factors such as noise and distortion has to be kept in mind. Embedding a watermark can be seen as adding noise to a signal. This noise is inaudible and is related to the embedding formula and embedding strength or factor. So SNR can be utilized to calculate embedding factor or scaling factor.

2.4 PERFORMANCE PARAMETERS

All systems are analysed on the basis of some predefined parameters and benchmarks. To evaluate performance of watermarking some basic parameters are discussed like robustness, imperceptibility and capacity. So here are some norms to quantify these parameters

2.5. IMPERCEPTIBILITY TEST

Imperceptibility tests are both subjective and objective. And it depends on the researcher to use which type of test.

2.5.1 OBJECTIVE EVALUATION

A new term Signal –to-Watermark ratio has been introduced [26]. This is given as

$$SWR \cong 10log_{10}\left(\frac{\sigma_x^2}{\sigma_w^2}\right) \quad (2.9)$$

Where σ_w^2 is the variance of the watermarked audio and σ_x^2 is the variance of host audio. It measures the strength of the watermark to the strength of host signal. This is an indicator of imperceptibility. For high imperceptibility the value of SWR must be high. SWR does not fully measure imperceptibility

2.5.2 SUBJECTIVE LISTENING TEST

Subjective test are made by humans or a set of individuals in a set environment. In this test five participants are given two audio clips. One original and other watermarked. After listening to the clips each participant has to rate the dissimilarity between the two as five point subjective grade (SG). An average of their scores is taken to get final answer. Here is a table for SG and ODG [26]

TABLE I

SUBJECTIVE AND OBJECTIVE DIFFERENCE GRADE

SG	ODG	Impairment description	Quality
1	-4.0	Very annoying	Bad
2	-3.0	Annoying	Poor
3	-2.0	Slightly annoying	Fair
4	-1.0	Perceptible, but not annoying	Good
5	0.0	Imperceptible	Excellent

2.5.3 OBJECTIVE LISTENING TEST

Objective listening test simulate the outcome of listening tests by modelling human listening response. So that subjective listening test can be substituted by it. However outcome of objective listening test the ODG (Objective Difference Grade) are not similar to subjective listening tests. Final quality judgement of audio is based in the criterion of subjective listening tests.

2.5.4 MEAN SQUARE ERROR

The mean square error is mean of squared differences. Where difference is measured between the original and watermarked audio signal

$$MSE = \sum_{i=1}^{total\ \ points} \frac{(original - watermarked\ audio)^2}{no\ of\ audio\ samples} \qquad (2.10)$$

A low MSE indicates high imperceptibility

2.5.5 SIGNAL TO NOISE RATIO

The signal to noise ratio is given by formula

$$SNR = 10log_{10}\left(\frac{\sum_{i=1}^{audio\ \ oints} x(i)^2}{\sum_{i=1}^{audio\ \ oints}(y(i)^2) - x(i)^2}\right) \qquad (2.11)$$

Here y is the watermarked audio and x is the host signal

2.6 ROBUSTNESS TEST

Robustness test check the endurance of a given algorithm to many different kind of attacks. For testing purpose first watermarked audio is attacked and then some parameters are checked to calculate robustness.

2.6.1 BIT ERROR RATE

Bit error rate gives the measure of changed or erroneous bits from original watermark to extracted watermark. The watermark is extracted after a certain attack is applied. The formula is given by [19]

$$BER(W, W^*) = \frac{\sum_{i=1}^{M} \sum_{j=1}^{N} W(i,j) \oplus W^*(i,j)}{M \times N} \quad (2.12)$$

Here W^* extracted watermark, W original watermark and $M \times N$ is the size of watermark.

2.6.2 NORMALIZED CORRELATION

This parameter evaluates the similarity or closeness between the watermark used at transmitter and watermark extracted at reciver. The formula is given as [19]

$$NC(W, W^*) = \frac{\sum_{i=i}^{M} \sum_{j=1}^{N} W(i,j)\, W^*(i,j)}{\sqrt{\sum_{i=1}^{M} \sum_{j=1}^{M} W^2(i,j)}\ \sqrt{\sum_{i=1}^{M} \sum_{j=1}^{M} W^{*2}(i,j)}} \quad (2.13)$$

Presence of watermark is indicated by the large value of NC. A small value of NC means absence of watermark.

2.7 GAPS

Every method and algorithm has some constraints and limitations which form the base of gaps. Gaps are a quintessential part of research. After a tedious literature survey some gaps have cropped up.

- Most of the techniques for signal decomposition have fixed kernel and do not utilize data characteristics for decomposition.
- The parameters for embedding in some techniques have to be constant and optimized after rigours testing.
- No adaptive approach has been utilized fully.
- Psycho acoustic model has not been properly utilized for watermarking
- Resistance to desynchronization attacks is big problem.

2.8 OBJECTIVES

Principle gaps comprehended from the literature have been used to form objectives. The objectives from the gaps are given below

- To decompose audio signal using data dependent decomposition method. The decomposition will have mathematical rigour and physical sense as well.
- To embed data on a set of samples using an array of quantizers. The embedding strength will be data dependent

2.9 METHODOLOGY

Both objectives have been considered to formulate the research problem. So in proposed method EMD will be utilized to decompose the signal. But EMD has issues during reconstruction. Envelops are interpolated from maxima and minima points. The method of interpolation utilized is cubic spline which has the problem of swings at the end points of the envelop. This lead to reconstruction error, although negligible. Also embedding should not distort the signal so much that it will introduce a different number of IMFs at extraction side. To embed the watermark QIM is utilized. But the embedding strength is an issue because it has to be calculated after rigours simulations. This problem is dealt in the proposed method.

2.10 SUMMARY

In this chapter concepts related to non stationary signal analysis have been discussed. Also various different methods and tools used for non stationary signal analysis are also discussed. Main aspects of watermarking have been addressed. Literature review gives a clear picture of different watermarking techniques proposed and utilized in current literature. A model of non interference rejecting watermarking method has been discussed. Concept of quantization index modulation is introduced. Methods for benchmarking and quantative analysis of the watermark are given. Finally the objectives and the research problem have been formulated on the basis of gaps from literature review. In chapter three proposed work is introduced. Mathematics of proposed work is given step by step.

CHAPTER 3

PROPOSED TECHNIQUE AND METHODOLOGY

There have been several uses of EMD in literature like in [27], [28] and [29]. In [27] a hybrid technique using EMD and ISA are applied. Also in [28] the IMF with highest energy is used to embed the watermark. In [29] EMD is used in audio encoding process. The low frequency signals or the high mode IMF are prone to degradation if any tempering is introduced. So it is the best position to hide a watermark. Also low frequency components have high energy which makes the watermark imperceptible.

Issues introduced by EMD and QIM have been studied and steps to improve them have been applied. But the problems due to cubic spline and signal distortion due to embedding is still a problem.

3.1 BASIC IDEA

In the proposed technique the signal is segmented into frames and these frames are decomposed by Empirical Mode Decomposition. The watermark is pre-processed by scrambling. And converted into an 1D array. The last IMF is used to embed the data. This data is embedded in the extreme points using a adaptive Quantization Index Modulation and then the inverse EMD is applied. At extraction side the received signal is segmented and decomposed using EMD. And watermark is extracted from the last IMF.

3.2 MATHEMATICAL FORMULATION OF ALGORITHM

This section provides the mathematical basis for the algorithm. The values used are

$X(i)$ = input audio signal

$Y_l(i)$ = Watermarked audio signal of l^{th} frame

α_l = Embedding strength of l^{th} frame

k = Running variable for extrema values in each frame

W = Original $M \times N$ watermark

W^* = Extracted $M \times N$ watermark

3.2.1 EMBEDDING PROCEDURE

The embedding block is shown in Figure 3.2. One of the watermarks shown in Figure 3.1 is a $M \times N$ binary image. This binary image is a 2-D matrix which is converted into a 1-D array of length N_1 . Here a synchronization code of length N_2 is used.

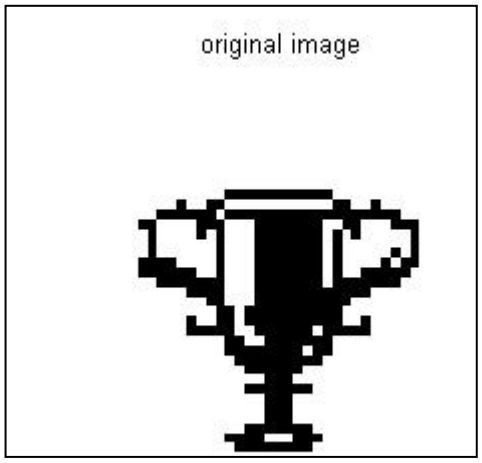

Figure - 3.1: Cup

The steps for embedding procedure are given as

Step 1 The first step of the algorithm is segmentation. The audio is segmented into l non overlapping frames. Here each frame has i number of sample points.

Step 2 The l frames are decompose using Empirical Mode Decomposition into IMF given as

$$EMD_l = \sum_{i=1}^{no\ of\ imfs} IMF_i + residue \qquad (3.1)$$

Where for every l^{th} frame the decomposition is given by EMD_l , IMF_i is the i^{th} IMF.

Step 3 scramble the original watermark using Arnolds Transform and convert the 2-D image into an 1-D array. Insert synchronization codes according to the Figure-ure 3.3.

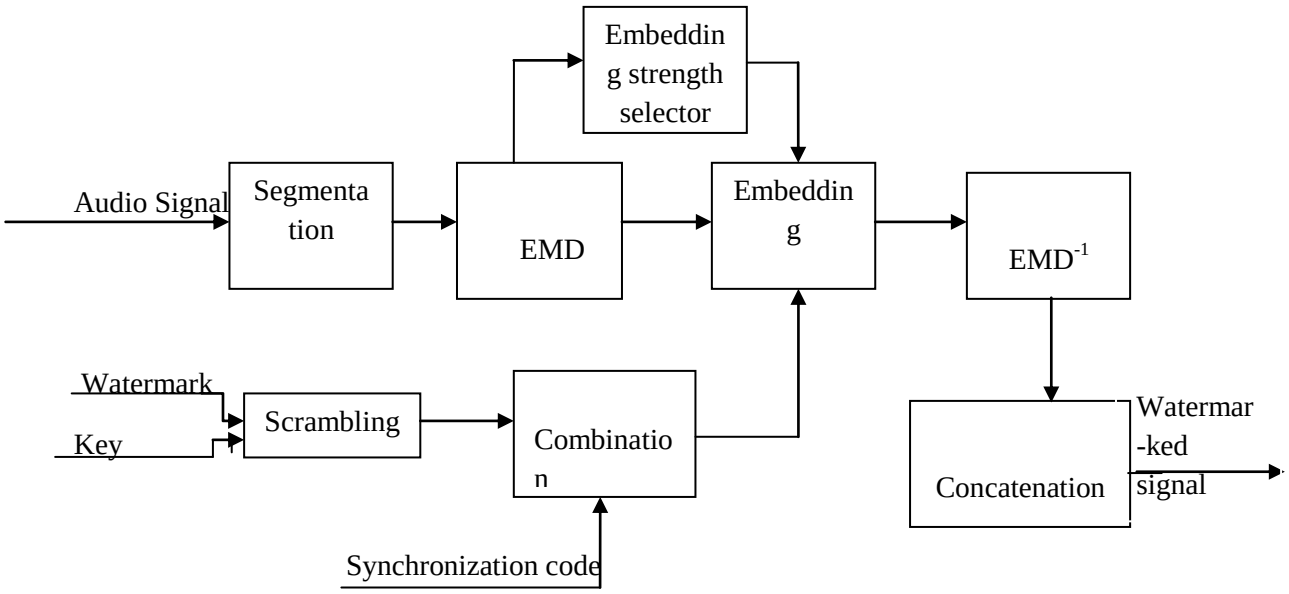

Figure - 3.2: Watermark embedding process

Sync – code	Watermark - bits	Sync - code

Figure - 3.3: Arranged binary data (b_i)

Step 4 Embed the data a binary sequence into the extrema of low frequency components by an adaptive form of QIM [4]

$$ex_i^* = \begin{cases} \lfloor ex_i \ / \alpha_l \rfloor \cdot \alpha_l + sgn\left(\frac{3\alpha_l}{4}\right) & if \ b_i = 1 \\ \lfloor ex_i \ / \alpha_l \rfloor \cdot \alpha_l + sgn\left(\frac{\alpha_l}{4}\right) & if \ b_i = 0 \end{cases} \quad (3.2)$$

Here ex_i^* is the i^{th} watermarked extrema point and ex_i is the original i^{th} extrema point of the last IMF.sgn is "+" for maxima and "– "for minima. $\lfloor \ \rfloor$ denotes the floor function and α is the embedding strength which depends on the SNR value of the IMF.

The human auditory system is sensitive to distortion Embedding watermark in audio signal can be compared to adding noise to signal, so SNR is used to determine the embedding strength. S_{snr} is calculated from [30]

$$S_{snr} = 10 \log \frac{\sum X^2(i)}{\sum_i [X(i) - Y(i)]^2} \quad (3.3)$$

$$Y_l(i) = \left\{ \left\lfloor \frac{X(k)}{\alpha_l} \right\rfloor \cdot \alpha_l + d(k) \cdot e(k) \quad for\ i = k \right. \quad (3.4)$$

$Y_l(i) =$ Watermaked audio of l^{th} frame

$d(k) = k^{th}$ Data bit where its value is

$$d(k) = \begin{cases} \left(\frac{3\alpha_l}{4}\right) & if\ \ bit\ k = 1 \\ \left(\frac{\alpha_l}{4}\right) & if\ bit\ k = 0 \end{cases} \quad (3.5)$$

$e(k) =$ sign for equation, for k^{th}maxima it is '+' and k^{th}minima '-'.

Form equation 7

$$\frac{-Ssnr}{10} = log \left[\frac{\sum_i \left(X(i) - Y(i)\right)^2}{\sum_i \left(X(i)\right)^2} \right] \quad (3.6)$$

$$\sum_i \left(X(i) - Y(i)\right)^2 = \sum_i \left(X(i)\right)^2 \ 10^{\frac{-Ssnr}{10}} \quad (3.7)$$

Taking some assumptions for calculation

- The strength of watermark is independent of extrema value i.e. if it is minima or minima.
- The strength of watermark is independent of the data bit i.e. 1 or 0.
- The IMFs are symmetric with zero local mean.

$$\alpha = \sqrt{(\sum_i X^2(i))\ 10^{\frac{-S_{snr}}{10}}} \quad (3.8)$$

Here $X(i)$ is the i^{th} last IMF of several consecutive frames. The value of S_{snr} can be assumed for our method this value in turn can be thought of as a key. For a different S_{snr} different results are produced.

Step 4 Reconstruct the signal by summation of IMFs in each frame and then concatenate all these frames. This can be termed as inverse EMD and the concatenated signal as the watermarked signal

3.2.2 EXTRACTION PROCEDURE

The watermark is extracted by splitting watermarked audio signal into frames and EMD is applied on these frames. Extract the watermark from the equation provided. The block diagram is shown in Figure 3.4. The extraction process is executed without the original signal. The basic steps of watermarking are as follows:

Step 1 Break the watermark signal into l non overlapping frames each of length .

Step 2 Decompose each frame using EMD in IMFs.

Step 3 Locate the extrema points $\{ex_i^*\}$ fro IMF and apply the following rule

$$d_i^* = \begin{cases} 1 & if \quad ex_i^* - \lfloor ex_i^*/\alpha \rfloor \cdot \alpha \geq sgn(\alpha/2) \\ 0 & if \quad ex_i^* - \lfloor ex_i^*/\alpha \rfloor \cdot \alpha < sgn(\alpha/2) \end{cases} \quad (3.9)$$

Step 4 Initialize the starting point of d_i^* , J = 1 and L = N_2. Compare the data $d_i^*(J:L)$ With SC bit by bit. If the similarity value is $\geq \tau$, then that particular segment is taken as SC tan we go to step 6, else proceed to next step

Step 5 Shift the window by one sample or change J by 1 and slid it to next samples and repeat step 4.

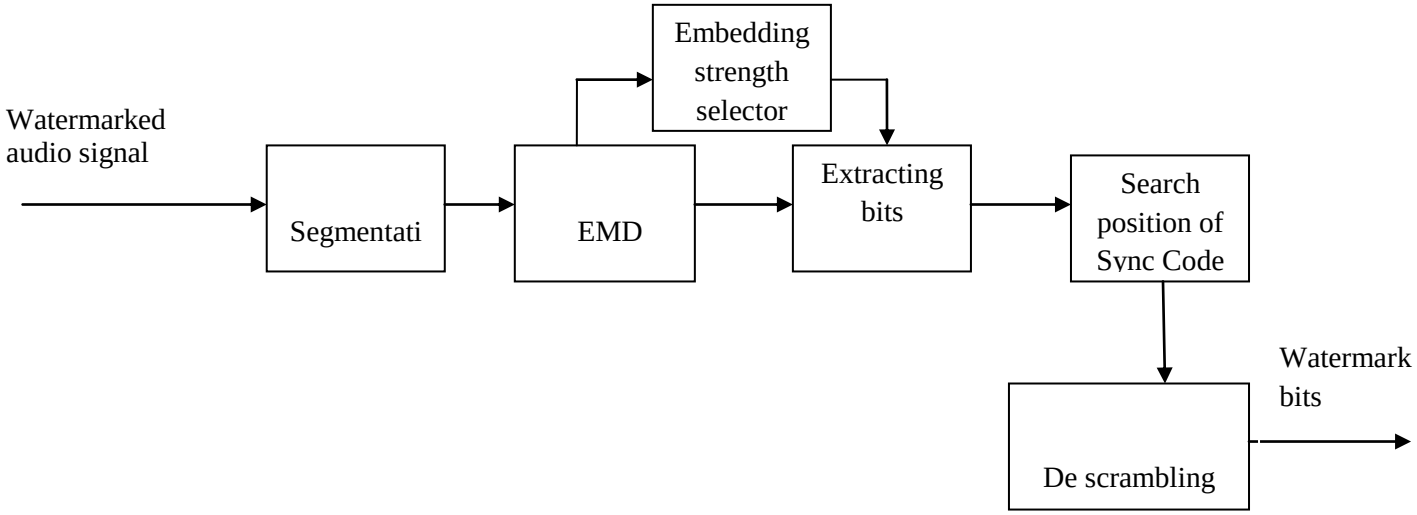

Figure - 3.4: Watermark Extraction Process

Step 6 Second segment is $d_i^*(J + N_1 + N_2 : J + N_1 + 2N_2)$ evaluated for similarity with SC bit by bit

Step 7 Go on repeating this procedure till value of J is equal to the length of payload or data

After converting 1-D array into 2-D matrix, inverse scrambling is applied on the data to get the watermark.

3.3 WATERMARKS USED

Five watermarks are used during the simulation.

Figure - 3.5: Cup

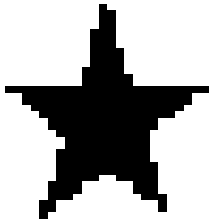

Figure - 3.6: Star

Figure - 3.7: Christmas Tree

Figure - 3.8: Rose

Figure - 3.9: Thapar Logo

CHAPTER 4

RESULTS AND DISCUSSIONS

The research started with an extensive literature survey on audio watermarking. The basis of this survey has lead to a research problem. Core point of this problem is to make a robust watermarking technique. The schemes and algorithms dealt in the literature survey have shown remarkable characteristic. For realization of objectives a new method is proposed and simulated. The technique is adaptive and data driven. It utilizes an adaptive method for decomposing and embedding watermark on a non-linear and non-stationary signal, which is audio. The method of embedding leads to higher capacity or embedding rate of data in host signal.

4.1 SIMULATION SPECIFICATIONS

The proposed watermarking scheme is simulated in the system given below

Software : Matlab version 7.110.584 (R2010b).

Processor : Intel® i3 @2.40 GHz

RAM : 4.00 GB

System type: 64-bit

Database : SQAM (Sound Quality Assessment Material) [31].

4.2 RESULTS OF ALGORITHM WITHOUT ATTACK

Here results generated from each step of the algorithm on a test signal are shown in Figure. The signal is a speech signal from the SQAM database

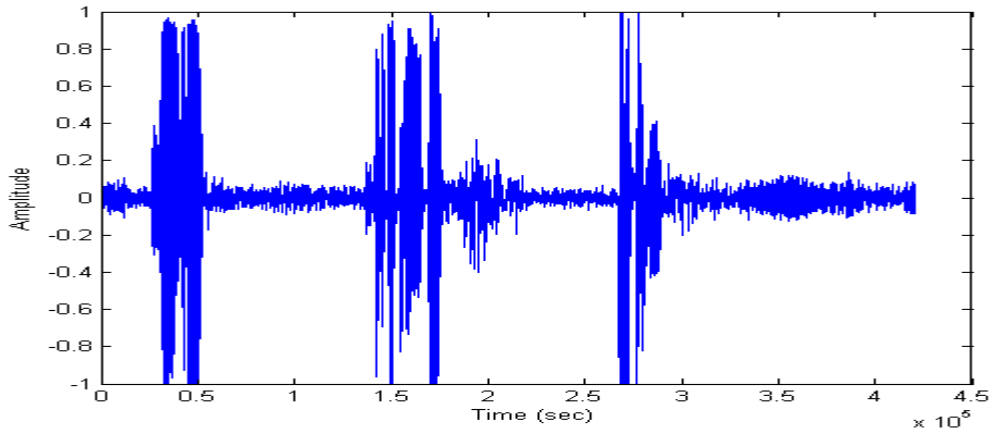

Figure - 4.1: Host audio signal

The host audio clip is a speech portion that contains 3 words. This clip is in .wav format. This signal is divided in frames each of length 64 samples. Figure 4.2 is the first frame of host signal.

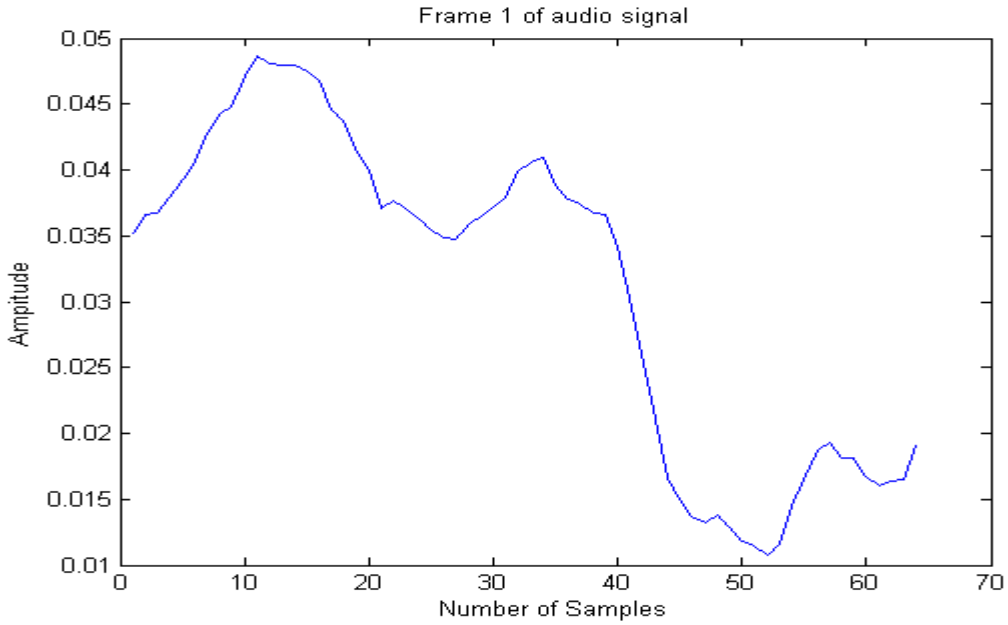

Figure - 4.2: Frame 1 of host signal

We decompose all frames into IMF functions. Figure 4.3, 4.4, 4.5, showcase the three IMF of frame one. Length of these IMFs is 64 samples.

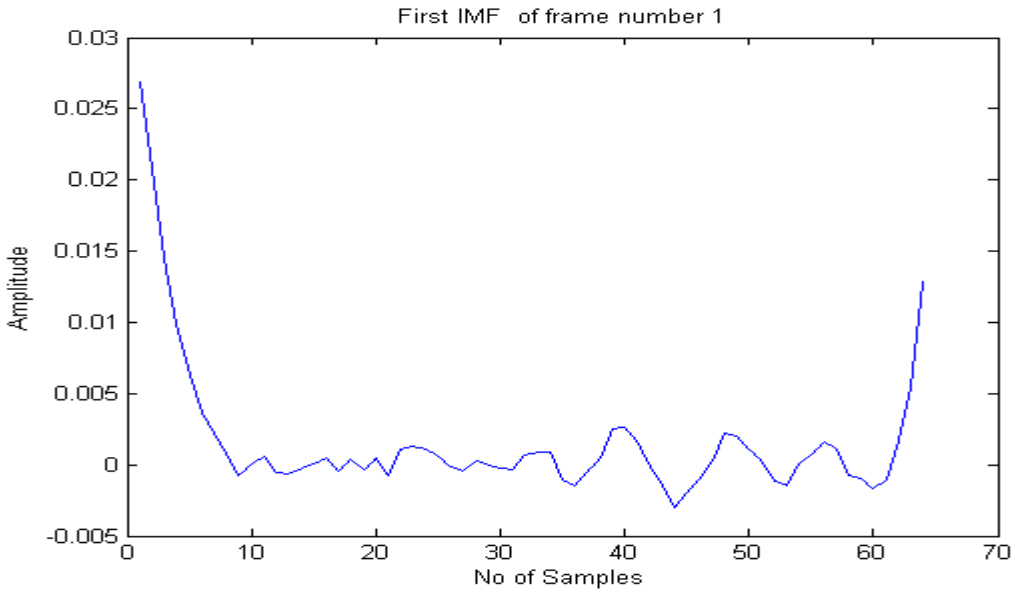

Figure - 4.3: First IMF of Frame Number 1

We embed data in the last IMF of each fame . The last IMF is the low frequency componet.As most of signal energy is cocentrated in low frequency regions and any change in them will cause drastic change in data , so if an malacious attacker tries to remove the watermark , data will be lost.

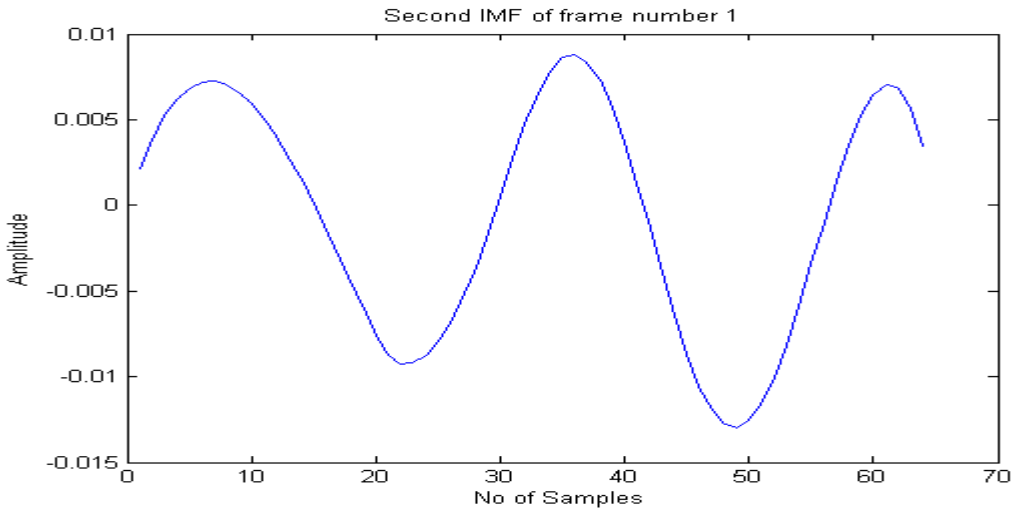

Figure - 4.4: Second IMF of Frame Number 1

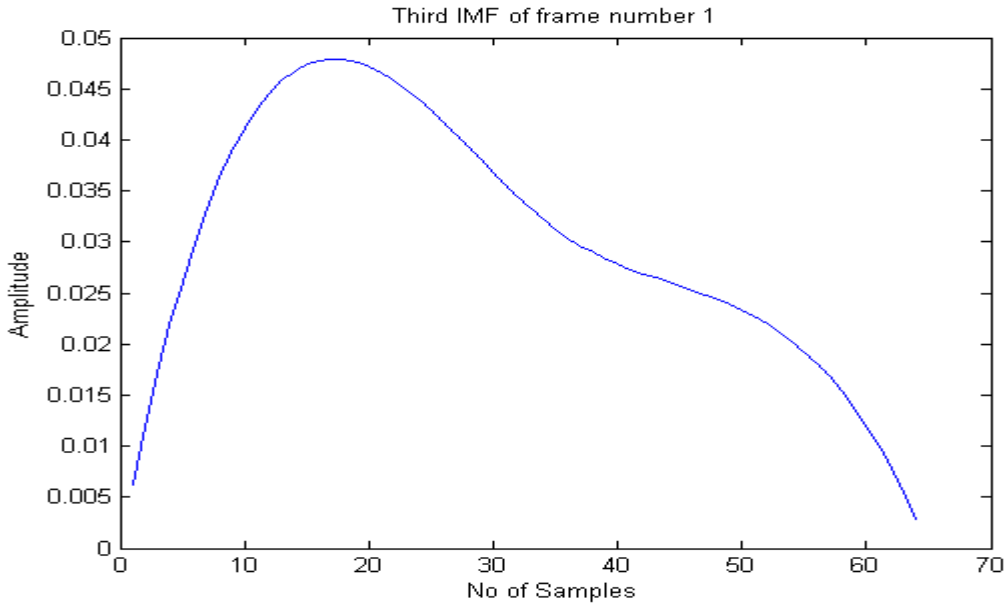

Figure - 4.5: Third IMF of Frame Number 1

Finally all the IMFs are summed to get the final frame. As seen from Figure 4.6 watermarked and original host signal are compared.

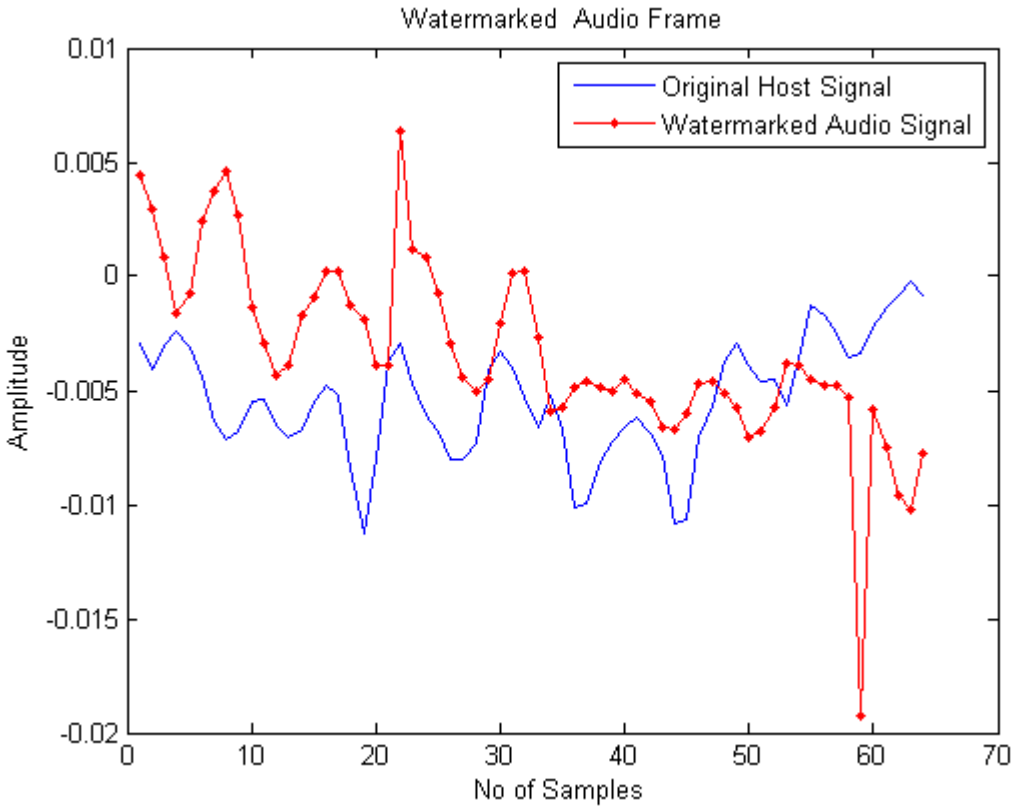

Figure - 4.6: Watermarked and original audio frame.

Table II

Watermark for diffrent values of S_{snr}

S_{snr}	Cup	Star	Rose	Tree	Thapar
1					
2					
3					
4					
5					
6					
7					
8					

9					
10					
11					
12					
13					
14					
15					

4.2.1 RESULTS FOR DIFFRENT VALUE OF S_{snr} VARIABLE

The S_{snr} factor was used to calculate the embedding constant or embedding strength. But not all values S_{snr} can be used to calculate the embedding strength. To find the appropriate range of values, extensive simulation was conducted with different values

starting from 1 and incremented by 1. Finally an upper limit of 15 was found where watermark was still recognizable visually. From equation 3.8 it can be inferred that as the value of S_{snr} increases, the embedding strength decreases. A small embedding strength is not good for watermarking as it quantizes the extrema values, near the original values. Also too small value of S_{snr} leads to a large value of embedding strength, which will lead to erroneous signal reconstruction, but embedding strength is also data dependent so the effects of data thus cannot be quantified properly. A good range will be 4 to 7.

4.3 RESULTS AFTER ATTACKS

The performance is measured on the basis of Bit Error Rate and Normalized Correlation. After applying an attack these values are calculated to quantify the performance index. The signals used for testing are Speech, Solo Instrument, Pop and Orchestra. These signals are mono signals with 44.1 kHz sampling rate. Different attacks used are based on information from [32].

The different attacks are given below

- **Low pass filtering** A filtering attack

  The specification of filtering and mainly low pass filter is

  Low pass filter = cut-off frequency is 6 kHz

  Roll of factor is 12dB/octave

  High pass filter = cut-off frequency is 200Hz

  Roll of factor is 12dB/octave

  Band pass filter = lower and upper cut-off frequency respectively 200Hz and 6 kHz

  Roll of factor is 15dB/octave

  From all the testing it has been established that high pass and band pass filtering distort the watermarked signal fully. And therefore is not suggested to be utilized for future use. All the filtering attacks used in this dissertation are low pass filtering attacks. Also adaptive filters such as Wiener filters can be used in place of conventional filters

- **Resampling** Resampling signifies device features and application protocols. The sampling frequency of original signal is 44.1 kHz. This signal is resampled at 22.05 kHz and again back to 44.1 kHz.
- **Requantization** Here requantization is done by converting 16 bit data into 8 bit representation and again in 16 bit. This process represents data flow from one device to another. Where different devices work on different protocols and bit representation.
- **Cropping** This attack is performed by removing some audio frames at random and inserting frames containing AWGN.
- **AWGN** AWGN insertion represents the channel effects on data. Every medium imbibes certain noise in the signal and act as a noise source, not only channel but devices can be seen as a noise source. AWGN is added till SNR is 20 dB.
- **MP3(32 kb/s)** This attack signifies change due to compression and expansion. Audio files used for watermarking are in .Wave format. Watermarked files are converted into MP3 files and again back to .Wave format.

These attacks have been applied on different types of audio signal which are

- Speech
- Pop
- Solo Instrumental
- Orchestra

Five different watermarks are used during the simulation for better experimentation. The results of simulation are given below

Table III

Result of attacks on speech signal (Female) with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9720	4		.9632	5		.9776	3		.9284	8		.9750	2	
Cropping	1	0		1	0		1	0		1	0		1	0	
Resampling	.9974	0		.9983	.2		1	0		.9808	.2		1	0	
MP3(32 kb/s)	.9922	1.2		.9983	.2		.9832	1		.9735	1.5		.9991	0	
Requantizat-ion	1	0		1	0		1	0		1	0		1	0	

Table IV

Result of attacks on speech signal (Male) with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9895	1		.9837	2		.9926	1		.9675	3		.9836	2	
Cropping	.9944	0		.9930	0		.9980	0		1	0		1	0	
Resampling	.9830	1		.9944	0		.9940	0		1	0		.9938	0	
MP3(32 kb/s)	.9930	0		.9824	2		1	0		.9950	0		.9950	0	
Requantizat-ion	1	0		1	0		1	0		1	0		1	0	

Table V

Result of attacks on speech signal (Three words) with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9896	2		.9842	2		.9645	5		.9932	1		1	0	
Cropping	1	0		1	0		1	0		1	0		1	0	
Resampling	.9912	0		1	0		1	0		1	0		1	0	
MP3(32 kb/s)	.9912	0		.9953	.67		.9960	.1		1	0		.9912	0	
Requantizat-ion	1	0		1	0		1	0		1	0		1	0	

Table VI

Result of attacks on pop signal with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9907	1		.9846	2		.9919	0.5		.9736	2		.9854	2	
Cropping	1	0		.9914	0		.9920	3		.9950	0		1	0	
Resampling	.9978	0		.9996	0		.9981	1		.9771	2		.9973	0	
MP3(32 kb/s)	1	0		.9870	1		.9950	1		1	0		.9890	1	
Requantization	1	0		1	0		1	0		1	0		1	0	

Table VII

Result of attacks on pop signal 2 with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9975	1		1	.5		.9989	.2		.9970	1		1	0	
Croppin g	1	0		.9914	0		.9920	.5		.9950	0		.9930	0	
Resampl ing	1	0		1	0		1	0		1	0		1	0	
MP3(32 kb/s)	1	0		1	0		1	0		1	0		1	0	
Requanti zat-ion	1	0		1	0		1	0		1	0		1	0	

Table VIII

Result of attacks on Solo Instrument (Piano) with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9903	1		.9783	3		.9933	1		.9569	5		.9819	2	
Croppin g	1	0		.9920	0		.9870	1		1	0		1	0	
Resampl ing	.9948	0		.9876	1		.9980	0		.9750	2		.9870	1.5	
MP3(32 kb/s)	1	0		.9880	1.7		.9922	0		.9960	0		.9945	0	
Requanti zat-ion	1	0		1	0		1	0		1	0		1	0	

Table IX

Result of attacks on Solo Instruments (Guitar) signal with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9864	0		.9754	3		.9854	2		.9980	.3		.9989	0	
Cropping	1	0		1	0		1	0		1	0		1	0	
Resampling	.9870	0		1	0		1	0		.9985	0		.9954	0	
MP3(32 kb/s)	1	0		.9845	1.2		.9875	0		.9864	0		.9875	0	
Requantizat-ion	1	0		1	0		1	0		1	0		1	0	

Table X

Result of attacks on Solo Instrument (Violin) with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9886	1		.9774	3		.9930	1		.9556	5		.9865	2.21	
Cropping	.9934	0		1	0		.9970	0		1	0		1	0	
Resampling	.9933	1		.9816	2		.9963	0		.9644	3		.9830	1.3	
MP3(32 kb/s)	.9947	0		.9826	0		.9918	0		.9927	1		1	0	
Requantizat-ion	1	0		1	0		1	0		1	0		1	0	

Table XI

Result of attacks on Orchestra (Mozart) with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9929	0		.9863	1		.9950	0		.9573	5		.9914	1.14	
Cropping	.9830	2		.9976	0		.9990	0		.9980	0		.9950	0	
Resampling	1	0		.9880	0		.9736	2		.9690	3		.9713	4.72	
MP3(32 kb/s)	.9910	0		1	0		.9850	1		.9860	1		1	0	
Requantization	1	0		1	0		1	0		1	0		1	0	

Table XII

Result of attacks on Orchestra (Soprano) with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9877	2		.9615	5		.9779	3.6		.9238	9.1		.9865	2.21	
Cropping	1	0		1	0		1	0		1	0		1	0	
Resampling	1	0		.9872	1.8		.9950	0		.9721	3.4		.9980	0	
MP3(32 kb/s)	.9993	0.12		.9953	0.67		.9870	1.5		.9840	2		.9945	0	
Requantizat-ion	.9840	1.3		1	0		1	0		1	0		1	0	

Table XIII

Result of attacks on Orchestra (Choir) with five different watermarks

Attack	Cup			Christmas tree			Star			Rose			Thapar		
	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark	NC	BER %	Extracted Watermark
No Attack	1	0		1	0		1	0		1	0		1	0	
AWGN	1	0		1	0		1	0		1	0		1	0	
Filtering	.9240	9		.9972	.42		.9891	1.28		.9771	3.8		.9527	7.90	
Cropping	1	0		1	0		1	0		1	0		1	0	
Resampling	.9945	0		1	0		1	0		1	0		1	0	
MP3(32 kb/s)	.9840	2		.9956	0		1	0		1	0		.9930	0	
Requantizat-ion	.9875	1		1	0		1	0		1	0		1	0	

4.4 DISCUSSION

From Table III, effects of proposed method on female speech signal can be seen. From the data given in this table for most of the attacks BER is less than 1%. But with the exception of filtering attacks . Here filtering attack affects watermark the most. Watermarks with curves and less straight lines are affected much more than the geometric shapes with less curved perimeter. It has been seen that resampling attack also affects the watermark, but less severe than the filtering attack. Filtering attacks is most severe in female speech signal. Values in Table IV represent male speech signal and like female it has some similar outcomes except for filtering attacks, where male speech signal has small value of BER than female speech signal. The main difference between male and female speech signal is the pitch and loudness. Here male voice has low pitch than the female voice. Table V parameters represent a speech signal with three words and silence.

Most of the audio files used in watermarking applications are not pure speech signals. But a combination of speech and music signals. Therefore some mixed signals were used for simulations. In Table VI the outcomes represent the audio signal of type pop. This category uses low frequency tones with moderate loudness. Filtering attack affects the watermarked signal. Same for table VII but with better results. Here maximum value of BER is 2% even for filtering attack.

Solo instrumentals are also used for simulation. Table VIII represents piano, Table IX represents guitar and Table X represents violin. It has been noted that these three instrumentals have almost identical outcomes with small difference. From Table VIII, IX and X robustness to different attacks has been quantified. Again the average amount of BER for these signals is 2%, for some cases it is 5 % as well. But throughout thw watermark is detectable visually.

Table XI ,XII and XIII are an orchestra tone. The maximum value of BER which is 9.9 has been obtained from these type of signals. This value is observed in the filtering attack. Filtering attack has affected the watermark. But unlike other signals, orchestra

signal are highly affected by the resampling attacks. But still the watermark is visually detectable .

The information given by tables above implies robustness of the watermarking algorithm. The method has given very good results except for filtering attacks. But the watermark can be comprehended visually. From [33] it was inferred that patch work technique was robust but form the tables above it is inferred that the proposed method is robust to most of the attacks. Also the most affected signal from the filtering attack is orchestra signal. Effects of Cropping, Resampling and Requantization attacks on watermark are small.

4.5 COMPARITIVE ANALYSIS

From all the simulations it has been found that the algorithm solves the problem of embedding strength calculations and gives outcomes with respect to some pre defined SNR values. The proposed method is an improvement from the one introduced by Khalid [24] on the grounds of embedding strength and security due to the added feature of scrambling . But the novelty introduced by the proposed technique is the payload or the capacity which is higher than most of the referred works

TABLE XIV

Comparison of Different Audio Watermarking Method.

Reference	**Robustness to MP3 attacks(kb/s)**	**Payload(b/s)**
Proposed algorithm	32	95.19-98.20
Khalid *et al*. [24]	32	46.9-50.3
Bhat *et al.* [19]	32	45.9
Yeo *et al.* [34]	96	10
Xiang *et al.* [35]	64	2

CHAPTER 5

CONCLUSION AND FUTURE WORK

5.1 CONCLUSION

In this dissertation watermarking method is proposed, utilizing EMD and QIM. Watermark is inserted into the low frequency component of the signal. Spreading of watermark in cover is in time domain. IMFs calculated after applying EMD provides frequency resolution. The biggest advantage of this method lays in the fact that decomposition of audio signal results in a small number of components unlike Fourier which results in infinite or a large number of coefficients. The data is inserted in the extrema points of the low frequency mode achieving robustness and high capacity. A synchronization code is used to locate the watermark and hamper the chance of a desynchronization attack. The watermark is scrambled and then embedded into the signal using an adaptive form of QIM. Here the embedding strength is calculated from the low frequency component or higher order IMF. Experimental results have shown that this method is robust to a range of different attacks. This method has a high embedding rate. But the biggest plus point of proposed work is capacity. It has a high data capacity than most of the methods cited. Also the signals used for watermarking were mono signal. A stereo signal will give double the capacity. The proposed method does not requires original watermark at the extraction point making it semi blind , as a key is required for scrambling . Also without scrambling this technique is blind in nature. This method has easy calculations, does not require complex equation and easy to simulate.

5.2 FUTURE WORK

Future work includes

- Using psychoacoustic models and fuzzy logic in design of a new algorithm.
- Removing the flaws of EMD especially interpolation problems
- To use multiple quantizers to increase the capacity of watermarking algorithm

REFERENCES

[1] E.F Hembrooke, "Identification of sound and like signals". United States Patent, 3 00404 ,1961.

[2] W.E Noller, "Inband signalling systems". United States Patent, 3061783, 1962

[3] Ingermar J. Cox and Matt L. Miller ,"The First 50 Years of Electronic watermarking," *EURASIP Journal on Applied Signal Processing*, vol.2, pp.126-132, 2002

[4] Norden E.Huang, Zheng Shen, Steven R , Long,Manli C.W , Hsing H. Shih,Quanan Zheng, Nai- Chyuan Yen,Chi Chao Tung and Henry H. Liu, " The empiricla mode decomposition and the Hilbert spectrum for nonlinear and non-Stationary time series analysis," *Proc,Royal Society. England*, pp.903-995, 1996.

[5] A.N. Lemma, J. Aprea, W. Oomen, and L. van de Kerkhof "A Temporal Domain Audio Watermarking Technique". *IEEE transaction on signal processing*, vol. 51, pp.1088-10977, 2003.

[6] N. K. Kalantari, M. A. Akhaee, S. M. Ahadi, and H. Amindavar ,"Robust Multiplicative Patchwork Method for Audio Watermarking," *IEEE Transaction on Audio, Speech, and Language Processing*, vol. 17, pp.1133-1141, 2009.

[7] Y. Xiang, I. Natgunanathan, D. Peng, Member,W. Zhou and S. Yu, "A Dual-Channel Time-Spread Echo Method for Audio Watermarking ,"*IEEE Transaction on Information Forensics and Security*,vol. 7, pp.383-392, 2012.

[8] R. Nishimura ,"Audio Watermarking Using Spatial Masking and Ambisonics," *IEEE Transaction on Audio, Speech, and Language Processing*, vol. 20, pp.2461-2465, 2012.

[9] K.Hofbauer, G. Kubin and W. Bastiaan Kleijn ,"Speech Watermarking for Analog Flat-Fading Bandpass Channels," *IEEE Transaction on Audio, Speech, and Language Processing* , vol. 17, pp.1624-1637, 2009.

[10] Y. Erfani and S. Siapousha ," Robust audio watermarking using TS echo hiding," *Journal of signal processing*, vol.19, pp.809-814, 2009.

[11] H . Peng, X. Luo, J. Wang and Z. Zhang, " A learning-based audio watermarking scheme using kernel Fisher discriminant analysis," *Journal of Signal Processing*, vol. 23, pp.382-389, 2013.

[12] S-T Chen, H-N Huang, C-J Chen , K-K Tseng and S-Y Tu, " Adaptive audio watermarking via the optimization point of view on the wavelet-based entropy ," *Journal of Digital Signal Processing*, vol. 23, pp.971-980, 2013.

[13] I. Natgunanathan, Y. Xiang, Y. Rong, , W. Zhou, and S. Guo, "Robust Patchwork-Based Embedding and Decoding Scheme for Digital Audio Watermarking," *IEEE Transaction on Audio, Speech, and Language Processing*, vol. 20, pp.2232-2239, 2012.

[14] M-Q Fan and H-X Wang ," Statistical characteristic-based robust audio watermarking for resolving playback speed modification," *Journal of Digital Signal Processing*, vol. 21, pp.110-117, 2011.

[15] B. Lei, I.Y Soon, F. Zhou, Z. Li and H. Lei, "A robust audio watermarking scheme based on lifting wavelet transform and singular value decomposition," *Journal of Digital Signal Processing*, vol. 92, pp.1985-2001, 2011.

[16] P. K Dhar and J-M Kim ," Digital Watermarking Scheme Based on Fast Fourier Transformation for Audio Copyright Protection," *International Journal of Security and its Application*, vol. 5, pp.33-48, 2011.

[17] S-T Chen, G.D Wu , H-N Huang, C.J Chen, " Energy-proportion based scheme for audio watermarking ," *IET Audio Signal Processing*, vol. 4, pp.576-587,2010.

[18] S-T Chen, G.D Wu and H-N Huang,"Wavelet-domain audio watermarking scheme using optimization-based quantization," *IET Audio Signal Processing*, vol. 4 pp.720-727, 2010.

[19] Vivekanada B. K, I. Sengupta and A. Das ,"An adaptive audio watermarking based on the singular value decomposition in wavelet domain ," *Journal of Digital Signal Processing* , vol. 20, pp.1547-1558, 2010.

[20] Mohammad J . Akhaee, M. Saberian, S. Feizi, and F. Marvasti ,"Robust Audio Data Hiding Using Correlated Quantization With Histogram-Based Detector,". *IEEE Transaction on Multimedia*, vol. 11, pp.834-842, 2009.

[21] D. Kirovski and H. Malvar, "Spread-Spectrum Watermarking of Audio Signals," *IEEE Transaction on Signal Processing*, vol. 51, pp.1020-1033, 2003

[22] H. Malik, R. Ansari and A. Khokhar ,"Robust audio watermarking using frequency-selective spread spectrum," *IET Information Security*, vol. 2, pp.129-150, 2008.

[23] X. Kang, R. Yang and J. Huang, "Geometric Invariant Audio Watermarking Based on an LCM Feature," *IEEE Transaction on Multimedia*, vol. 13, pp.181-190, 2011.

[24] K. Khaldi and A.O. Boudraa, "Audio Watermarking Via EMD,"*IEEE Transaction on Audio, Speech, and Language Processing*, vol. 21, pp.675-680, 2013

[25] Brian Chen and Gregory W,Wornell ,"Quantization Index Modulatio : A Class of Provably Good Methods For Digital Watermarking and Information Embedding," *IEEE Transaction on Information Theory*,vol.47, pp 1423-1443, 2001

[26] Atul Deshpande."Audio Watermarking Algorithms Based on Interpolation ." MSc. dissertation, Indian Institute of Technology Madras, Electrical Engineering Dept, Madras, 2008.

[27] Nawal El Hamdouni, Abdellah Adib, Sonia Djaziri and Monia Turki,"Hybrid Embedding Stratergy For A Blind Audio Watermarking System Using EMD and ISA Techniques." *In Proc. International Symposium on Communications, Control and Signal Processing*, pp.1-4 2010.

[28] A N K Zamana, K.M Ibrahim Khalilullah,Md. Wahedul Islam and Md. Khademul Islam Molla ,"A Robust Digital Audio Watermarking Algorithm Using Empirical Mode Decomposition." *In Proc. Canadian Conference on Electrical and Computer Engineering*, pp .1-4, 2010.

[29] Kais Khalid, Abdel-Ouahab Boudraa, Monia Turki, Thierry Chonavel and Imen Samaali ,"Audio Encoding Based On The Empirical Mode Decomposition." *In Proc.17th European Signal Processing Conference* , pp. 1-6, 2009

[30] Xueyao Li , Min Zhang and Shiliang Sun," Adaptive Watermarking Algorithm Based on SNR in Wavelet Domain ," *In Proc. International conference on Natural Language Processing and Knowledge*, pp. 287-292, 2003.

[31] "Sound Quality Assessment Material," *https://tech.ebu.ch/publications/sqamcd* , Jan. 10, 2015.

[32] Sviatolsav Voloshynovskiy, Shelby Pereira, Thierry Pun, Joachim J.Eggers and Jonathan k, Su , "Attacks on Digital Watermarks: Classification, Estimation-

Based Attacks, and Benchmarks ," *IEEE Communications Magazine*,vol.39, pp 118-126, 2001

[33] E.K Marnani, Z. Karami,E. MolavianJazi," A Comparision of Some Audio Watermarking Methods ," *In Proc. International conference on Electrical Engineering, Computing Science and Automatic Control*, pp. 1-4, 2009.

[34] I.K. Yeo and H. J. Kim ," Modified Patchwork Algorithm: A Novel Audio Watermarking Scheme ," *IEEE Transaction on Signal Processing* , vol.11, pp. 381-386, 2003.

[35] S. Xiang, H. J. Kim and J. Huang," Audio Watermarking Robust Against Time-Scale Modification and MP3 Compression," *Journal of Signal Processing*, vol. 88, pp. 2372-2387, 2008.

PUBLICATIONS

1 "Adaptive Audio watermarking algorithm based on Empirical Mode Decomposition and Quantization Index Modulation" Communicated in *Journal of Signal Processing Systems for Signal, Image, and Video Technology* ,Springer Publication (*SCI Indexed*)

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