Republic of Iraq
Ministry of Higher Education
and Scientific Research
Al-Nahrain University
College of Science

Audio-Hiding System Using Wavelet and DCT Transforms

A Thesis Submitted to the College of Science, Al-Nahrain University in Partial Fulfillment of the Requirements for

The Degree of Master of Science in Computer Science

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October 2008 **Shwal 1429**

بسم الله الرحمن الرحيم

اقْرَأْ وَرَبُّكَ الْأَكْرَمُ ۝٣ الَّذِي عَلَّمَ بِالْقَلَمِ ۝٤ عَلَّمَ

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صدق الله العظيم

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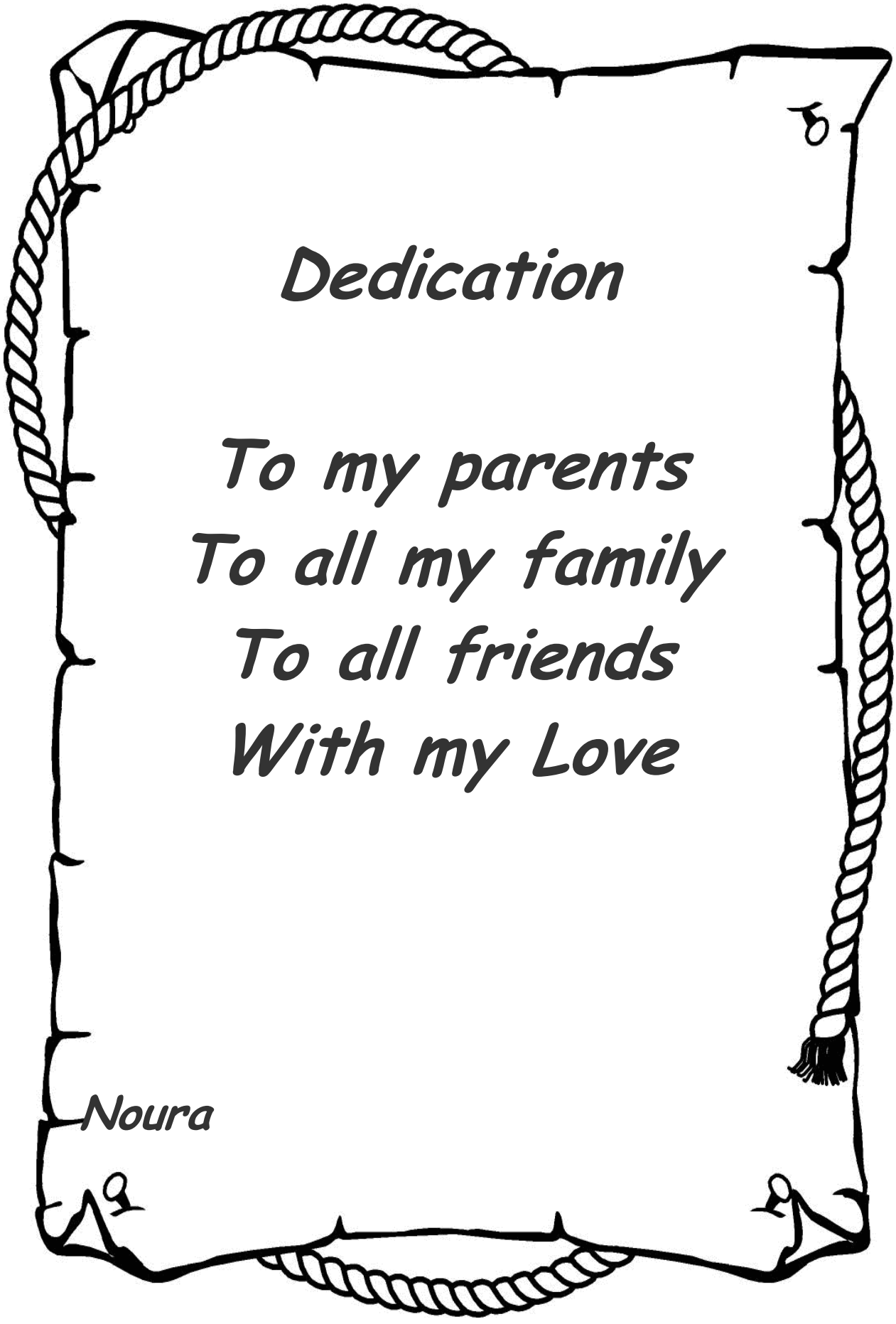

Dedication

To my parents
To all my family
To all friends
With my Love

Noura

Acknowledgment

First of all great thanks are due to Allah who helped me and gave me the ability to achieve this research from first to last step.

I would like to express my sincere appreciation to my supervisor, ***Dr. Loay A. George****, for giving me the major steps to go on to explore the subject, shearing with me the idea in my research "Adaptive Audio-audio Hiding System Using Wavelet and DCT Transform" And perform the points that I felt important.*

Also grateful thanks for the Head of Department of Computer Science ***Dr. Taha S. Bashaga.***

Deep gratitude and special thanks to my family: my Mother, Father, Sister and Brother for their encouragements and supporting to succeed in doing this work.

Deep thanks for all my friends and every one support me, encourage me and giving me advises.

Abstract

Steganography is considered as one of the widely used methods for hiding information; it hides secret data in digital cover data without clear suspicion.

This work focus on using two transform based methods for hiding secret data. First, the data will converted to binary form, then two predefine threshold values are used to categorize the audio blocks. The first threshold is used to decide whether the block is voiced/unvoiced, and the second threshold is used to decide which transform will be used on the voiced blocks only (i.e., DCT or Wavelet).

Second, the audio signal will be transformed by the chosen one of transforms; DCT transform will applied on voiced blocks that have energy less than second threshold value, and wavelet transform will be applied on voiced blocks that have energy grater than or equal to the second threshold value. In the third stage the embedding on secret data on voiced blocks is applied.

The performance of the proposed hybrid system was tested by using some fidelity measures (MSE, MAE, and PSNR) to measure the rate of error, the ratio of corrected retrieved bits and the hiding rate, the later is computed to assess the embedding power of system. Also, the effects of some control parameters on the system performance were investigated to assist user to correctly choose the proper values of the system parameters, some of these parameter is the threshold value which classify the blocks to voice/unvoiced, and our goal is to hide in voiced blocks, this will provide high hiding rate for the system.

List of Abbreviations

A/D	Analog to Digital converter
D/A	Digital to Analog converter
dB	Decibel
DCT	Discreet Cosine Transform
DFT	Discreet Fourier Transform
DHWT	Discreet Haar Wavelet Transform
HAS	Human Audio System
HCB	Hiding in Checked Blocks
HVB	Hiding in Voiced Blocks
HWT	Haar Wavelet Transform
Hz	Hertz
FFT	Fast Fourier Transform
IDCT	Inverse Discreet Cosine Transform
IHWT	Inverse Haar Wavelet Transform
kB	Kilo Byte
KHz	Kilo Hertz
LBE	Low Bit Encoding
LSB	Least Significant Bits
MB	Mega Byte
MAE	Mean Absolute Error
MSE	Mean Square Error
PCM	Pulse Code Modulation
PSNR	Peak Signal to Noise Ratio
RIFF	Resource Interchange File Format
SNR	Signal to Noise Ratio
WAV	Windows Audio Visual

List of Symbols

Symbol	**Description**
BlkLen	Block Length
C	Transform Coefficient
ExSec	Vector contain the extracted bit of secret message
ER	Error of corrective Retrieved bits
HR	Hiding Rate
i	Counter
N	Length of voiced block
NoSamp	total number of audio samples
Q	Quantization Step
qWv	vector contain data after Quantization
rWv	vector contain the reconstructed Wave file
S	value of embedded secret bit
Scr	vector contain Secret Data
Stg	vector contain Stego data
T	Power Threshold value
T'	Transform Threshold value
tWv	vector contain the transformed data
v	pointer
Wv	vector contain audio wave file data
δ	Modulation Step

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Chapter One

General Introduction

Chapter One

General Introduction

1.1 Overview

With the development of information technology, people have paid more and more attention on the information security today because internet provides the facilities to exchange text, image, audio, and video between users. Internet access can reach sensitive locations (like, martial location, ministry of defense, etc) for each government in the world, because it becomes the most public way to link with large companies and banks in the world.

All these facts encouraged some persons (or even companies) to develop ways to some software tools to make unauthorized access to closed locations [Kaw06]. So for this reason information hiding becomes the focus of the information security research because it is the more suitable clue, than encryption, to conceal information.

Information hiding technology can embed secret information into a digital media source without impairing the perceptual quality of that source, such that other people can't feel this secret information. So the secret key, the signature or the private data can be exchanged securely through Internet [Xiu07].

1.2 Information Hiding

Information hiding techniques have received very much less attention from the research community and from industry than cryptography. However, this situation was changed rapidly since the first academic conference on this topic, which was organized in 1996. The main driving force was concentrated on protecting the copyright of audio, video and other works which become available in digital form, the ease with which perfect copies can be made may lead to large-scale unauthorized copying, and this is of great concern to the music, film, book and software publishing

industries. At the same time, the movements that conducted by various governments to restrict the availability of encryption services have motivated people to study methods by which private messages can be embedded in seemingly innocuous cover messages [Fab99]**.** So the hidden data may have no relationship with (or may provide important information about) the cover-object, in which it is embedded [Cac00]**.** The classification of data hiding techniques is presented in Figure (1.1)

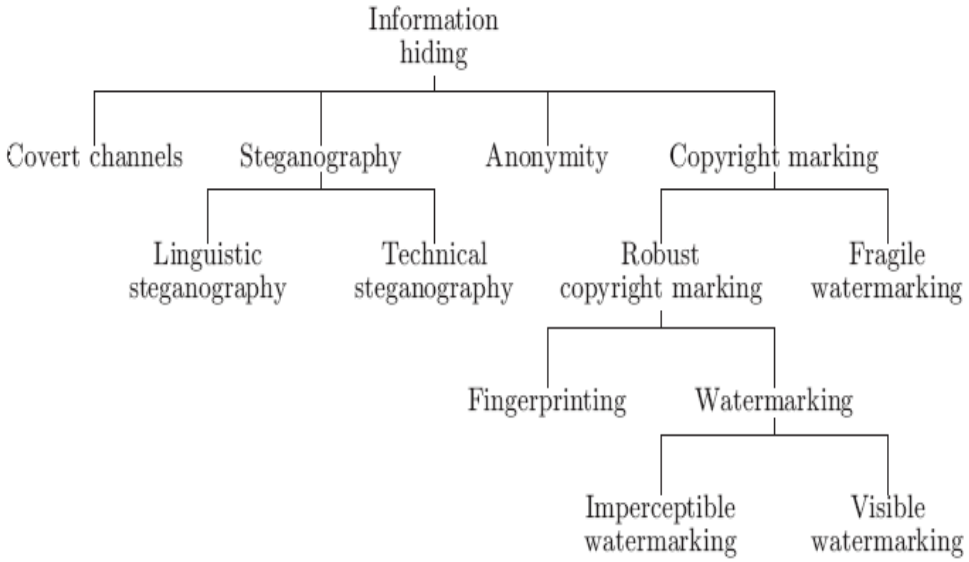

Fig (1.1) A classification of information hiding techniques [Fab99]

Data hiding techniques should be capable of embedding data in a host signal with the following restrictions and features [Ben96]:

1. The host signal should be non objectionally degraded, and the embedded data should be minimally perceptible. This means that the observer should not notice the presence of the data, even if it is perceptible.

2. The embedded data should be directly encoded into the media, rather than into a header, so that the data remain intact across varying data file formats.

3. The embedded data should be immune to modifications ranging from intentional and intelligent removal attempts, to anticipated manipulations (e.g., channel noise, resampling, encoding, lossy compression, digital-to-analog (D/A) conversion, etc).

4. The embedded data should be self-clocking or arbitrarily reentrant. This ensures that the embedded data can be recovered when only fragments of the host signal are available.

1.3 Steganography

Steganography is the art and science of writing hidden message in such a way that no one apart from the intended recipient knows of the existing of the message; this is in contrast to cryptography, where the existence of the message itself is not disguised, but the content is obscured [Cum04]. The word "Steganography" is hard to be found in any dictionary, it comes from the Greek word "Steganos" (means *Covered*) and "graphy" (means *writing*). So steganography literally means "Covered Writing" [Kaw06].

Generally, a steganography message will appear to be something else: a picture, an article, a shopping list, or some other type of messages. Classically, it may be hidden by using invisible ink between the visible lines of innocuous documents, or even written onto clothing, thus forming a message. It may even be a few words written under a postage stamp, the stamp then being the covertext [Kes01].

The advantage of steganography over cryptography alone is that messages do not attract attention to themselves, to messengers, or to recipients. An unhidden coded message, no matter how unbreakable it is, will arouse suspicion and may in itself be incriminating, as in some countries encryption is illegal [Cum04]. So the goal of steganography is to avoid drawing attention to the transmission of a hidden message. If suspicion is raised, then this goal is defeated [Sib04].

1.3.1 Steganography versus Watermark [Moh04]

The purpose of both watermarking and steganography is to hide a message in a carrier signal, but the differences between them are summarized in table (1.1).

Table (1.1) A comparison between watermark and steganography [Moh04]

Watermark	Steganography
The information hidden by a watermarking system is always related to the original object (to be protected) or to its owner.	Steganographic system just hides any information.
Communications are usually one-to-many (between sender and many receivers).	Communications are usually point-to-point (between sender and receiver).
The cover is the transmitted data, and the hidden information is just for authentication purposes.	The hidden information is the transmitted data, and the cover is just to hide it.
Small amount of data need to be hidden.	Any amount of data needs to be hidden.
The hidden data need more robustness because watermarking concerns with potential removal by a pirate.	The hidden data need less robustness, because steganography is mainly concern with the problem of detecting a message in the stego object.

1.3.2 Steganography versus Cryptography

There is a clear distinction between steganography and cryptography, and inspite of the inherent difference between them they are compliment to each other very well. The aim of cryptography is to scramble the message with the help of a key so that the message becomes meaningless to the attacker, but can only be deciphered by either the same key or another key. However steganography does not allow us to change the message itself, rather the message is hided in seemingly innocuous messages like text, picture or other media. So, in this case the attacker does not know if the two parties are even exchanging messages [Bak07].

Generally the purpose of both steganography and cryptography is to provide secret communication, but the differences between them are summarized in table (1.2) [Moh04].

Table (1.2) A comparison between Cryptography and steganography [Bak07]

Cryptography	Steganography
The goal of secure cryptographic method is to prevent an interceptor from gaining any information about the plaintext from the intercepted cipher text.	The goal of secure steganographic methods is to prevent intermediate observer from even obtaining knowledge about the mere presence of the secret data.
The system is broken when the attacker can read the secret message.	Breaking a steganography system has two stages: 1. The attacker can detect the type of steganography has been used. 2. He should be able to read the embedded message.
The cryptographic systems don't use the steganography methods.	In many steganography systems the embedded message is encrypted before hiding.

1.4 Digital Audio [Cyb87]

Over the years, improvements in technology have changed the way music is recorded and the media used. Today, we use computers to record audio and save it to CDs, MP3 players, and other storage devices.

In order to transform sound into a digital format, the sound should be sampled. This process takes place while there is digital recording. The computer takes a snapshot of the sound level at small time intervals. The number of samples taken each second is called the sampling rate. The more samples that are taken, the better sound quality. For instance, audio sampled at 44 kHZ is better than audio sampled at 22 kHz. It also means more storage space for higher quality sound.

Audio files have an extension at the end of the filename that tells users the format type. For example, files with (.snd) or (.aiff) are common on Macintosh computers while .wav is most common on Windows machines. Therefore, it is important to know the file format extensions.

1.5 Related Work

A lot of research works were conduct to develop many Information hiding techniques, the researchers tried to insert new additional features to increase the system robustness and invisibility. Some of the published researches are summarized below:

1. **Areespongsa (2000) [Are00]**, presented a stegosystem using wavelet transform. In this system the data was embedded in the sign of the high frequency coefficients of the cover image in attempt to trade off between the robustness of the embedded data and the invisibility of the stegoimage.
2. **AL- Ta'l (2002) [Ziy02]** developed an algorithm for audio steganography depending on the Human Auditory System (HAS). One of the properties of the HAS is the masking effect which depends on the principle that "*loud sound tends to mask out quiet ones*". Quieting the secret audio message to be masked by loud audio cover could be used for hiding purpose. A reverse procedure is done for extracting the audio message. The work was not for audio copyright purpose, it was used for the purpose of secret transmission without drawing any suspicion that a message is hidden in the cover audio file. The secret message and the cover may be speech, music or any recorded voices. The weak point in this algorithm is that the original audio signal must be sent to the receiver.
3. **Xu, et al. (2003) [Jia03]**, they proposed a system for "text steganography using wavelet transform". This research implies an algorithm to limit errors in lossy transform, and to achieve high capacity text hiding in image files using discrete Haar wavelet transform (DHWT). They have discussed robust text steganography using multiple-level lossless DHWT. The experimental results validated the method for high capacity plain text hiding and demonstrated that lossless recovery of the hidden text from JPEG images with compression rate as high as 67% is possible.
4. **AL Baka'a (2003) [Bak03]**, was concerned with development of invisible watermarking techniques, and had tested their robustness against different attacks

(rotation, sharpening, brightness and JPEG compression). Two approaches for image watermarking were developed, the first one embed the digital watermark data in the spatial domain, while the second approach embeds watermark data in frequency domain using DCT.

5. **Dieab (2003) [Die03]**, presented an audio watermarking system in order to embed a 10 characters digital watermark in the audio signal. The system uses two different techniques; low bit encoding (LBE) in the time domain, and the human auditory characteristics in the frequency domain using Fast Fourier Transform (FFT).
6. **Majeed (2004) [Maj04]**, introduced a steganography system that hides audio in audio using ***Discrete Cosine Transform*** (DCT). He suggested six hiding methods; some of them dedicated to hide audio data in audio data, while others have the ability of hiding any type of secret data in audio data. Also, he applied some additional steps to increase the security immunity of the system using two kinds of encryption: ***Pure key steganography*** and ***private key steganography.***
7. **AL Kawaz (2006) [Kaw06]** had proposed a system for "low rate hiding in audio data using phase domain". This research implies an algorithm for hiding any secret data type in the audio signal. The audio signal is transformed from time domain to frequency domain using a new mechanism for determining Fourier transform, in this introduced mechanism a reduction in the number of mathematical operations is achieved, which consequently leads to significant reduction in computation time. Two hiding method were designed to embed secret data bits in the phase domain coefficients of the audio signal. The first method, called hiding in voiced blocks (HVB), implies the insertion of secret bits in the voiced parts of audio data. While the second method, called hiding in checked blocks (HCB), implies the insertion of secret bits in audio blocks which are successfully passed the retrieved bits integrity tests.

The test results indicated that the HVB method show more embedding capacity than HCB method, and a little amount of the embedded secret bits could incorrectly retrieved when HVB method is utilized.

1.6 Aim of Thesis

The aim of research work is to develop a system for hiding audio signal in another audio signal using both Wavelet and DCT transform. The embedding method is done on the transform coefficients of the cover audio signal that produced after applying either wavelet or DCT transform. Since each transform type have its points, which are different with those belong to other transform, so a hybrid transform coding scheme will be established to overcome the weak aspect may phase the coding task when single type of transform is used. So the behavior of the established hybrid system should be better, but the scheme will be more complicated.

1.7 Thesis Layout

In addition to chapter one, the remaining part of this thesis consists of the following chapters:

- **Chapter Two (Theoretical Background)**

  In this chapter a background to the audio file (format and types) is presented, and some of the audio steganography techniques and transform coding methods are illustrated.
- **Chapter three (System Development)**

  It is dedicated to introduce the design aspects of the proposed system, and the implementation steps to realize the system. Each implementation step is clarified with its related algorithms.
- **Chapter four (Experimental Results)**

This chapter contains the results of the comprehensive tests performed on the proposed system using different test samples.

- **Chapter five (Conclusions and Suggestions)**

  This chapter is dedicated to list the conclusions that derived from the analysis of test results; also some ideas for future work are given in this chapter.

Chapter Two

Theoretical Background

Chapter Two
Theoretical Background

2.1 Introductions

This chapter is concerned with the theoretical concepts of information hiding, which is main objective in this project, and discuss the existing techniques of hiding in different media (such as, hiding in text, image and audio). Then some of the relevant issues to audio steganogrphy techniques are reviewed, including the basics of digital audio and the structure of the wave file that used as cover media. Then, definition of the transform and some of its types are given.

2.2 Steganography

The following formula provides a very generic description of the pieces of the steganographic processes:

cover_medium + hidden_data + stego_key = stego_medium

In this context, the ***cover_medium*** is the file in which the ***hidden_data*** is embedded, which may also be encrypted using the ***stego_key***. The resultant file is the ***stego_medium*** (which will, of course, be the same type of the cover_medium). The cover_medium and, stego_medium are typically image, audio or text files [Gar01]. In this chapter a focus is set toward audio files, and they will be called the ***cover_audio*** or ***stego_audio***.

2.3 Steganography History

Steganography is an old issue, and some of its methods are old, and others are modern techniques.

2.3.1 Old Steganographic Techniques

Steganography had been widely used in historical times, especially before cryptographic systems were developed. Examples of historical usage include:

- Hidden messages in wax tablets: in ancient Greece, people wrote messages on the wood, and then covered it with wax so that it looked like an ordinary, unused tablet.
- Hidden messages on messenger's body: also in ancient Greece. Herodotus tells the story of a message tattooed on a slave's shaved head, hidden by the growth of his hair, and exposed by shaving his head again. The message allegedly carried a warning to Greece about Persian invasion plans.
- Hidden messages on paper written in secret inks under other messages or on the blank parts of other messages.

2.3.2 Modern Steganographic Techniques [Ras02]

Modern steganography entered the world in 1985 with the advent of the Personal Computer, and it is applied to classical steganography problems. The development following that was slow: Some of new steganography techniques are:

- Concealing messages within the lowest bits of noisy images or sound files.
- Concealing data within encrypted data. The data to be concealed is first encrypted before being used to overwrite part of a much larger block of encrypted data. This technique works most effectively where the decrypted version of data being overwritten has no special meaning or use: some cryptosystems, especially those designed for file systems, add random looking padding bytes at the end of a ciphertext so that its size cannot be used to figure out the size of the original plaintext.
- Concealing messages in tampered executable files, exploiting redundancy in the i386 instruction set.
- Embedding pictures in video material (optionally played at slower or faster speed).

2.4 Steganography Requirements [Ras02]

Steganography have to guarantee these requirements:

1. **Robustness:** the embedded information is said to be robust if its presence can be reliably detected after applying modification, to certain extent, on the host (stegocover) object.
2. **Undetectability:** the embedded information is undetectable, if the stego object, with the embedded message, is consistent with the model of the source from which the stego object is drawn.
3. **Perceptual transparency**: it is based on the properties of the human visual system or the human audio system. The embedded information is imperceptible if an average human subject is unable to distinguish between carriers that do contain hidden information and those do not.

The above mentioned requirements are mutually competitive and cannot be clearly full filled at the same time. If we want to hide a large message inside an image, we cannot require at the same time absolute undetectability and large robustness. A reasonable compromise is always a necessity. On the other hand, if robustness to large distortion is an issue, the message that can be reliably hidden cannot be too long. This observation is schematically illustrated in Figure (2.1).

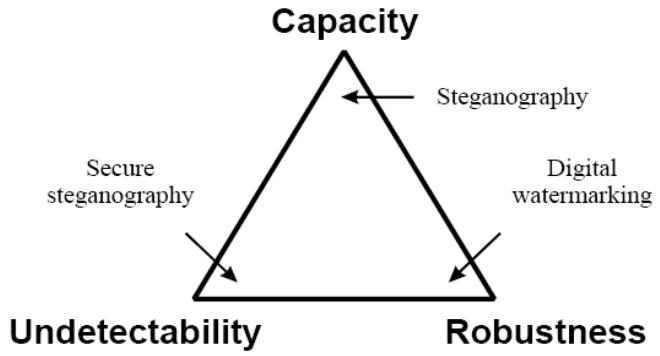

Fig (2.1) Conflicting requirements [Ras02]

2.5 Steganographic Techniques

Steganography techniques can be divided into various categories according to various criteria. The basic and most common used classification criteria are based on the nature of the cover media (i.e, text, audio, video, packets, OS) or on the nature of hiding media (i.e, spatial domain, frequency domain and parametric domain steganography).

Another way for the categorization of steganography methods is based on the condition "whether or not they use the original data for extraction of hiding message" from tested data [Sel99].

2.5.1 Steganography classification based on I/O

According to the nature of inputs and output, there are three types of steganography methods:

a. Pure Steganography [Joh01]

A steganographic system does not require the prior exchange of some secret information (like a stego-key) is called *Pure steganography*. Formally, the embedding process can be described as a mapping $E: C \times M \rightarrow C'$, where C is the set of possible cover, and M is the set of possible messages. The extraction process consists of a mapping $D: C' \rightarrow M$, that is extracting the secret message out of a cover.

In some steganography systems, the extraction process needs the original cover also. Both sender and receiver must have access to the embedding and extraction algorithm, but the algorithms should not be public.

b. Secret Key Steganography [Joh01]

With pure steganography, no information (apart from the functions *E* and *D*) is required to start the communication process; the security of the system thus depends entirely on its secrecy. In practice this is not very secure; one may assume that the attacker knows the algorithm that the sender and receiver use for information transfer. Thus, the security of steganographic system should rely on some secret information traded by the sender and the receiver (i.e. the *stego-key*).

Without knowledge of this key, nobody should be able to extract secret information out of a cover.

A secret key steganography system is similar to a symmetric cipher. (i.e., the sender chooses a cover *C* and embeds the secret message into *C* using a secret key *K*). If the key used in the embedding process is known to the receiver, the receiver can reverse the process and extracts the secret message. Anyone who does not know the secret key should not be able to obtain evidence about the encoded information.

c. Public Key Steganography

In public key steganography, it is not necessary for two people to share a secret key to establish a secure channel. Each one needs to know the other's public key. This suggests a possible approach to steganography in which a secret key does not have to be agreed upon between the sender and the receiver. Some information must still be known a *priori*, where each one must know the other's public key; from a practical perspective this is a much more reasonable requirement [Cra98]. One way to build public key steganography system is the use of public key cryptosystem [Joh01].

2.5.2 Steganography Classification Based on Stego Media

Information hiding techniques are classified according to the media where the information is hidden:

A. Hiding in Text

Methods like line-shift coding, word shift coding, and feature coding are the most commonly used methods to hide data in text. When using a text data as a host media, the embedded data is usually a codeword that is hidden within the text by altering its different textual features. The three methods mentioned above determine what feature is to be changed. To encode the codeword [Sel99], each bit of codeword is embedded using one of the above methods:

1. Line-Shift Coding

Line-shift coding is very easy to perform, and is considered as the most resistant to degradation due to copying. Although this method withstand copying, the human eye and other measurements can easily detected it. It can also be easily defeated through replacing or reformatting of the text [Sel99].

2. Word-Shift Coding

Word-shift coding can also be easily done. The appearance of natural spacing must be maintained in order not arouse suspicion. By determining the location where unnatural spacing has occurred, the encoded bits can be revealed [Bri06].

3. Feature Coding

Feature coding is another way of embedding data into a text file. In feature coding, certain text features are altered depending on the embedded data [Sel99]. In order for this type of feature coding to work, the text must be altered by randomness which will make the text look less suspicious to its readers. This type of feature coding can be easily defeated if the vertical line length is adjusted to a fixed length before the file is opened [Dav00].

B. Hiding in Image

Text files are not the only files that can be used for host data. Images are also another popular source for hidden data. The most popular techniques include least significant bit insertion and the use of algorithms and transformations [Aud03].

1. Least Significant Bit Insertion

Least significant bit insertion, or LSB, is one of the most common techniques used to hide information in images. When working with 24-bit pixel images, three bits can be encoded into each pixel. Because the least significant bits are the ones being altered, the change is difficult to determine by the viewer. However, when working with 8-bit pixel images, this method becomes harder to implement because a change of a bit may result in a change of an entirely different colors. Although this technique is popular due to its simplicity, it is also one of the

easiest methods to accidentally alter [Sel99]. Although it is almost an exact replica of the original image, the bits from the original image cannot be guaranteed.

2. Algorithm and Transformations

Other algorithms and transformations are also used when dealing with images and their usage in hiding data. Some of the most popular methods are the patchwork method, the discrete cosine transform (or DCT), and the Fourier transform. The patchwork method takes the advantage of the fact that the human eye cannot easily detect varying amount of light [Kat00]. The patchwork method gets its name from "using redundant pattern encoding to repeatedly scatter hidden information throughout the cover image" [Sel99]. One advantage of this technique is that it can hide a small message many times throughout an image. Because of this, even when an image is cropped or rotated, the chances of one instance of the encoded message still being intact are very high.

There are many different transforms used to map image data to non-spatial domain, such that the data hiding (embedding) could be done in more robust way. Among these transforms are discrete cosine transform, wavelet transform, the Fourier transform.

The **discrete cosine transform** (DCT) maps the image data into a set of coefficients that allows a small set of cosine functions approximates a portion of the image [Yan01]. For example, the JPEG standard compression schema uses 8x8 blocks of pixels and approximates them with a set of cosine functions, each set can approximate a section of the image. The DCT finds the value of the weight coefficient for each cosine function, so that the weighted sum of the functions added up to recreate the original 8x8 block of pixels [Yan01].

The **wavelet transform** and **Fourier transform** methods use complicated mathematical formulas in order to find the coefficients values required to map a signal into the frequency domain [Kat00].

C. Hiding in Audio

Audio file can also be used to hide information. Steganography is often used to copyright audio file to protect the rights of music artists. Techniques such as least significant bit insertion, phase coding, spread spectrum coding, and echo hiding can be used to protect the content of audio file. The biggest challenge face all these methods are the sensitivity of human auditory system or HAS [Kat00]. Because the HAS is so sensitive, people can often pick up randomly added noise which making it hard to successfully hide data within audio data. To fully understand the different techniques of hiding information in audio data, transmission of audio signals must first be understood. When working in audio the transmission medium must always be considered.

The transmission medium of an audio signal refers to the environment in which a signal might go through to reach its destination. The possible transmission environments can be categorized into the following four groups [Bri06]:

1. Digital end-to-end environment, where the sound files are copied directly from one machine to another.
2. Increased/decreased resampling environment, where the signal is resampled to a higher or lower sampling rate.
3. Analog transmission and resampling, where a signal is converted to analog state, played on a clean analog line, and resample.
4. "Over the air" environment, where the signal is played into the air, pass through a microphone.

By understanding the different mediums in which audio signals may travel, the appropriate technique for embedding data in audio files can be determined. The most commonly used methods for hiding data in audio media are the following methods:

1. Least significant Bit Insertion

Like image file, the least significant bit insertion method can also be used to store data in the least significant bit of audio file. However, like image file the

hidden data, using LSB, can be easily destroyed and detected. Resampling and channel noise may alter the hidden data, while changing the least significant bit may introduce audible noise [Sel99]. Information may also be destroyed through compression, cropping, or A/D, D/A conversion [Yan01].

Although this technique is simple to perform, its lack of dependability makes other methods more appealing.

2. Phase Coding

The phase coding method works by substituting the phase of an initial audio segment with a reference phase that represents the data. The phase of subsequent segment is adjusted in order to preserve the relative phase difference between segments [Yan01].

Phase coding is one of the most effective coding methods in terms of the signal-to-perceived noise ratio. When the phase relation between each frequency component is dramatically changed, noticeable phase dispersion will occur. However, as long as the modification of the phase is sufficiently small (sufficiently small depends on the observer; professionals in broadcast radio can detect modifications that are imperceivable to an average observer), an inaudible coding can be achieved [Ben96].

3. Spread Spectrum Coding

In a normal communication channel, it is often desirable to concentrate the information in as narrow region of the frequency spectrum as possible in order to conserve available bandwidth and to reduce power. The basic spread spectrum technique, on the other hand, is designed to encode a stream of information by spreading the encoded data across as much of the frequency spectrum as possible. This allows the signal reception, even if there is interference on some frequencies [Ben96].

4. Echo Data Hiding

Echo data hiding embeds data into a host audio signal by introducing an echo [Kat00]. The data are hidden by varying three parameter of the echo: initial

amplitude, decay rate, and offset. As the offset (or delay) between the original and the echo decreases the two signals blend. At a certain point, the human ear cannot distinguish between the two signals. This point is hard to determine exactly, it depends on the quality of the original recording, the type of sound being echoed, and the listener. In general, it was found that this fusion occurs around 1/1000 of a second for most sounds and most listeners. The coder uses two delay times, one to represent a binary one (offset) and another to represent a binary zero (offset + delta). Both delay times are below the threshold at which the human ear can resolve the echo [Ben96].

2.6 Audio Steganography

Data hiding in audio signals exploits imperfection of human auditory system known as audio masking. In presence of a loud signal (masker), another weaker signal may be inaudible, depending on spectral and temporal characteristics of both masked signal and masker. So the embedded signal is hidden there, besides transparency of the embedding process, it is important to insure robustness of the embedded signal.

2.6.1 Types of Audio Files

As with other digital media, a number of file types are in use for storage of digital audio data. When selecting a file type, it is important to consider the universality of the file type and thus its readability by a variety of software programs. File types that are proprietary and not likely to be supported in the future should be avoided, some of well knows audio file types are listed below:

1. **WAV**: This file type was developed by Microsoft. It is in widespread use, and is readable by most of audio software programs. The WAV file type has become a standard and is recommended. In addition, the WAV file type is also available in a professional format (i.e, broadcast WAV, BWF), which has the capability to store metadata in the file header. Although not all audio software programs are

currently capable of reading or writing to the metadata header, the BWF format is emerging as the WAV file type of preference for archival audio projects.

2. **MP3**: Is the name of the file extension and also the name of the type of file for MPEG, audio layer 3. Layer 3 is one of three coding schemes (layer 1, layer 2 and layer 3) for the compression of audio signals. Layer 3 uses perceptual audio coding and psychoacoustic compression to remove all superfluous information (more specifically, the redundant and irrelevant parts of a sound signal. The stuff the human ear doesn't hear anyway). It also adds a MDCT (Modified Discrete Cosine Transform) that implements a filter bank, increasing the frequency resolution 18 times higher than that of layer 2. Layer 3 can shrink the original sound data from a CD (with a bit rate of 1411.2 kilobits per one second of stereo music) by a factor of 12 (i.e., down to 112-128kbps) without sacrificing sound quality.
3. **WMA**: Short for Windows Media Audio, WMA is a Microsoft file format for encoding digital audio files similar to MP3 though can compress files at a higher rate than MP3. WMA files, which use the ".wma" file extension, can be of any size compressed to match many different connection speeds, or bandwidths.
4. **Real Audio (.ra .ram .rm)**: Is a proprietary format, and is used for streaming audio that enables you to play digital audio files in real-time. To use this type of file you must have RealPlayer (for Windows or Mac).

2.7 Digital Sound Representation

When developing a data hiding method on sound waves, like speech or music, one of the first considerations is how does sound is represented digitally. Audio refers to the sound within the human hearing range (20Hz to 20 KHz). An audio signal in nature is analog. Analog sound consists of waves detected by human ears. These waves are continues in both time and amplitude. Amplitude represents the height or (volumes), of the sound [Kie98, Dec99]. The analog signal should be converted to

digital form to be stored and processed by computers and transmitted through computer networks.

To do the conversion from analogue to digital from an A/D converter is needed. The A/D conversion process consists of two operations: sampling and quantization.

1. **Sampling**: Sampling involves periodical measurement of the analog signal, and uses these measurements (samples) instead of the original signal. An illustrative sampled wave is shown in figure (2.3)

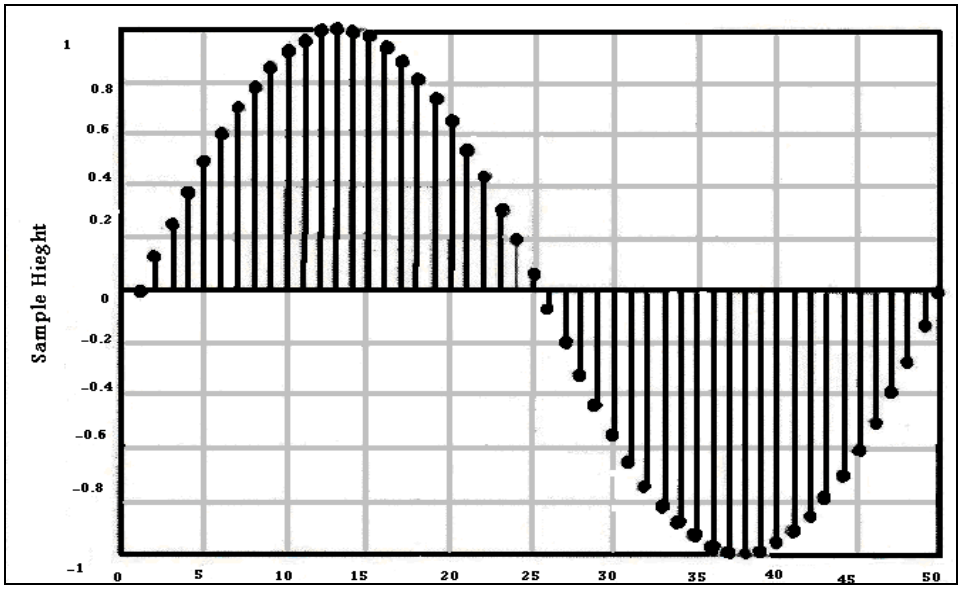

Fig (2.3) Sampled Wave [Dec99]

Usually samples are stored as binary numbers, but they can be stored in other ways. A very well known way is to represent each sample with a series of pulses that represent its binary code, such representation is called Pulse Code Modulation (PCM).

There are various modulation types, but PCM is the widely used in digital audio. For a programmer a various modulation techniques are irrelevant. In a computer's memory, the successive binary values are simply stored as numbers. For most programmers PCM can be thought of as that shown in figure (2.4) [Kie98, Qu96].

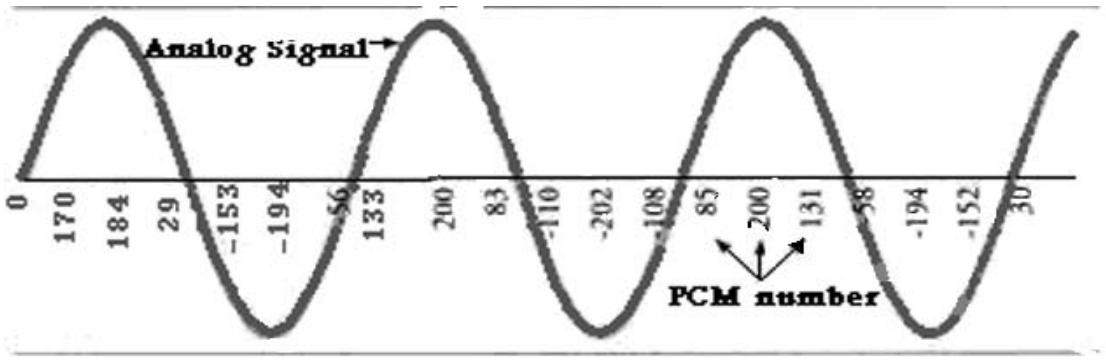

Fig (2.4) PCM for the computer programmer [Qu96]

2. **Quantization:** signal quantization means determining the signal value to some degree of accuracy. Because the finiteness of computer ability, the digital representation is also finite. For example if an 8-bit or 16-bit integers were used, either 256 (2^8) or 65,536 (2^{16}) discrete integer sample value can be obtained, but the original samples are not integers. The process of rounding the exact sample value to less-precise value is referred to as quantization [Aud03].

The quality and resolution of digitized audio is determined by two factors [Aud03]:

1. The number of times per second the amplitude of the wave is measured, and this number is called sampling rate.
2. The range of numbers used to record each measurement, which is called the "bit depth".

The first number, the "sampling rate," is described in kilohertz, or thousands of samples per second. Consumer audio CDs are recorded at a sampling rate of 44.1 kHz. That means that each second of audio is represented as 44,100 separate amplitude measurements, as illustrate in figure (2.5) which represent the wave flows.

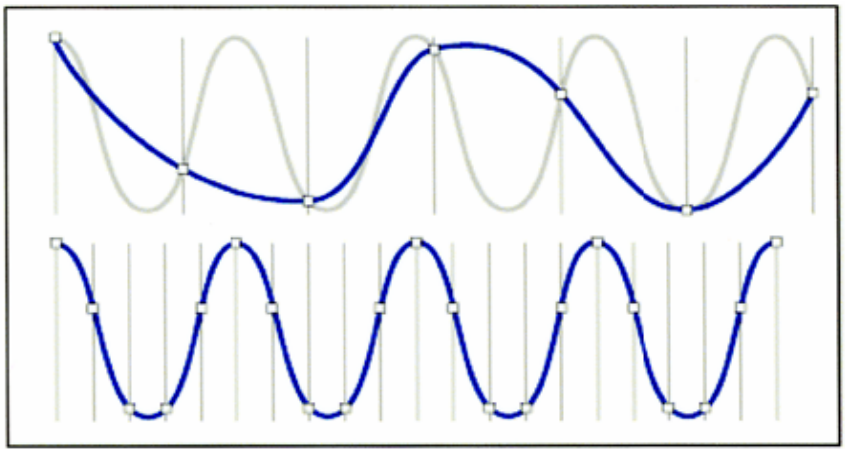

Fig (2.5) Visual representation of two sample rates [Aud03]

The second value, called the "bit depth," describes the range of numbers used to represent each amplitude measurement. For example, if each measurement is represented on a scale of 1 to 10, that would be a rougher measurement than a scale of 1 to 1,000. Sample size is measured in bits, such that:

8-bit numbers range from 0 to 255;

16-bit numbers range from 0 to 65,535;

24-bit numbers range from 0 to 16,777,215.

Since human ears are sensitive to the volume of sound, measured in decibels (dB), higher bit depths result in a “smoother” or more realistic representation of the audio source, or greater “dynamic range”, the standard for audio CDs is 16 bits, see figure (2.6).

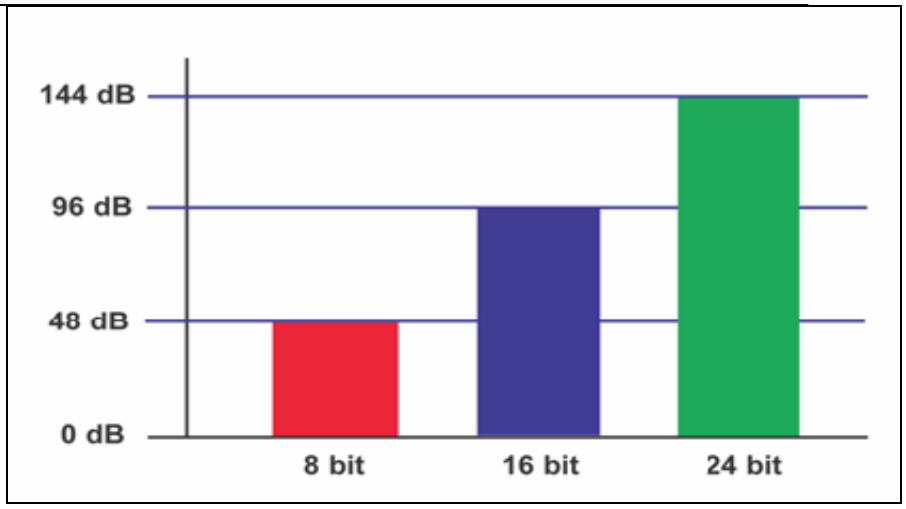

Fig (2.6) Higher bit depths can represent a wider volume range because they possess a greater "dynamic range" [Aud03]

2.8 Transform Domain Embedding Technique

One of the hiding classes is embedding the content of message by the modulation technique, applied on coefficients of the transform domain.

Transform based hiding techniques can offer superior robustness. Transform-based steganography typically offers increased robustness to scaling, rotations, and/or cropping, depending on the invariant properties of the used transform. The three most commonly used discrete transforms are:

1. Fourier Transform
2. Discrete Cosine Transform
3. Wavelet Transform

2.8.1 Fourier Transform

The Fourier transform is the best known, and the most widely used, transform. It was developed by Baptite Joseph Fourier (1768-1830) to explain the distribution of temperature and heat conduction. Since that time Fourier transform had been used in numerous applications including vibration analysis in mechanical engineering, circuit analysis in electrical engineering, and in computer imaging. This transform allows for

the decomposition of an image into a weighted sum of 1-D and 2-D sinusoidal terms [Ahm74].

2.8.2 DCT Transform

In 1974, the discrete cosine transform was invented by Ahmed, Natarajan and Rao. It is the most commonly used transform for image and video coding [Ahm74]. The cosine transform like the Fourier transform uses sinusoidal basis function. The difference is that the cosine transform basis functions are not complex; and they use only cosine functions. The cosine transform yields better performance that the DFT (Discrete Fourier Transform) is coding tasks, because DCT possesses better energy compaction in low frequencies than DFT [Ahm74]. In terms of *energy compaction*, DCT is the best among DFT, DWT, DHT and discrete Haar transform, and this because audio signals usually contain many harmonic components, transforming such signals to the frequency-domain gives a representation where energies of the audio signals are packed to a few large spectral peaks, and the rest part of the spectrum is made of small components .

In DCT also have end-head discontinuities; these discontinuities cause a high frequency distribution in the corresponding DFT. The DCT improves the DFT by eliminating the high frequency components induced by the sharp discontinuities at the boundary between two consecutive periods in the time (or spatial) domain of a periodic signal [Hai07].

In DCT, when the audio signal is transformed the most important DCT coefficients (i.e. high in values) will mostly be at the beginning of the DCT block, so possible to replace the other unimportant coefficients (i.e. low in values) with zero, and the result of applying inverse discrete cosine transform (IDCT) on the new set of coefficients will reconstruct the audio data that have an acceptable quality.

The DCT can be quickly calculated, and the DCT coefficients are all real numbers unlike the Fourier Transform. The Inverse Discrete Cosine Transform (IDCT) can be used to retrieve the audio data from its transform representation [Ben96]. The DCT and IDCT can be described by the following equations:

$$X_{\cos}(K) = \alpha(K)\sum_{n=0}^{N-1} x(n)\cos\left(\frac{\pi(2n+1)K}{2N}\right), \textit{for k=0,1,..., N-1 ,.....(2.1)}$$

$$x(k) = \sum_{n=0}^{N-1} \alpha(n)X_{\cos}(n)\cos\left(\frac{\pi(2k+1)n}{2N}\right), \textit{for k=0, 1,...,N-1 ,.....(2.2)}$$

Where, $\alpha(0) = \sqrt{\frac{1}{N}}$, $\alpha(K) = \sqrt{\frac{2}{N}}$ $\quad$ *for* $1 \le K \le N-1$ *,.....(2.3)*

2.8.3 Wavelet Transform

The term wavelet means a small wave. The smallness refers to the condition that this function is of finite length. The wave refers to the condition that this function is oscillatory. The term mother implies that the function with different region of support that used in the transformation process is derived from one main function, which is called the mother wavelet, The discrete cosine transform is Fourier-based while wavelet transform is not Fourier-based and for this reason its do a better job of handling discontinuities in data [Pol98].

There is a push toward the use of wavelet in signal processing and analysis in place of (or addition to) Discrete Cosine Transform (DCT), recently many algorithms have been proposed to use wavelet for image or audio processing. The techniques that are currently being used with audio can be generalized for use with wavelet transforms. There are numerous applications for wavelets, and the uses of wavelets in signal processing seem to be endless.

Fourier transform is based on spectral analysis; it is the dominant analytical tool for frequency domain analysis. However, Fourier transform cannot provide any information of the signal changes with respect to time. Fourier transform assumes the signal is stationary, but real signals are always non-stationary. To overcome this deficiency, a modified method, called short time Fourier transforms, allows to

represent the signal in both time and frequency domain through time windowing function [Ibr04].

The window length determines a constant time and frequency resolution. Thus, a shorter time windowing is used in order to capture the transient behavior of a signal; in such case there is a sacrifice in the frequency resolution. The nature of the real signals is nonperiodic and transient (such as sound, image and video signals), such signals cannot easily be analyzed by conventional transform. So, an alternative mathematical tool (like, wavelet transform) must be selected to extract the relevant time-amplitude information from a signal [Wah02].

Wavelet transform is capable of providing the time and frequency information simultaneously, hence giving a time-frequency representation of the signal. Signals whose frequency content does not change in time are called *stationary signals*. In other words, the frequency content of stationary signals does not change in time. In this case, one does not need to know at what times frequency components exist, since all frequency components exist at all times. But, basically Wavelet Transform (WT) is needed to analyze non-stationary signals (i.e., whose frequency response varies in time). Because Fourier Transform (FT) is not suitable for non-stationary signals.

Traditional transform techniques have been designed to take the advantage of the statistical redundancy present in most of the audio data. Removing redundancy can only provide limited embedding ratio. Increasing embedding ratio will remove some of the non-redundancy data and produce visual degrade in audio signal quality. To achieve higher embedding ratio and acceptable reconstructed audio file, other transform method such as wavelet transform can be considered [Rob96].

Wavelet transform have proven to be very efficient and effective in analyzing a very wide class of signals and phenomena. The properties that give the effectiveness are [Bur98]:

1. The wavelet expansion allows a more accurate local description and separation of signal characteristics. A Fourier coefficient represents components that last for all

time and, therefore, temporary events must be described by the phase characteristics that allow cancellation and reinforcement over large time periods. Wavelet expansion coefficients represent a component that itself local and easier to interpret. The wavelet expansion may allow a separation of components of a signal that overlaps in both time and frequency.

2. Wavelets are adjustable and adaptable. Because there is not just one wavelet, they can be designed to fit individual systems which can adjust themselves to suit the signal.
3. The generation of wavelet coefficients is well matched to the digital computers. There are no derivatives or integrals, just multiplications and additions operations that are basic to the digital computer.

2.9 Haar Wavelet Transform (HWT)

The oldest and most basic wavelet system had been constructed from the Haar basis function. The equations for forward Haar Wavelet transform and inverse Haar Wavelet transform are shown in the following sub-sections:

1. Forward Haar Wavelet Transform (FHWT) [Jia03]

Given an input sequence (X_i) i=0… N-1, its FHWT consists of L(i) and H(i) components, {i=0, 1…$\frac{N}{2}$-1}, they determined using the following transform equations:

(a) If N is even

$$L(i) = \frac{x(2i) + x(2i+1)}{\sqrt{2}} \text{ , For } i = 0,1\ldots(n-1)/2 \text{ ,}\ldots\ldots\ldots \textit{(2.4)}$$

$$H(i) = \frac{x(2i) - x(2i+1)}{\sqrt{2}} \text{ For } i = 0,1\ldots(n-1)/2,\ldots\ldots\ldots\ldots \textit{(2.5)}$$

(b) If N is odd

$$L(i) = \frac{x(2i) + x(2i+1)}{\sqrt{2}}$$

$$H(i) = \frac{x(2i) - x(2i+1)}{\sqrt{2}}$$

For i= 0,1…(n-1)/2 ,…..……*(2.6)*

$$L(\frac{N+1}{2}) = x(N-1)\sqrt{2}$$

$$H(\frac{N+1}{2}) = 0$$

,……..……...*(2.7)*

2. Inverse Haar Wavelet Transform (IHWT) [Jia03]

The inverse Haar Wavelet Transform equation is simply the inverse to those applied in the FHW;

(a) If N is even

$$x(2i) = \frac{L(i) + H(i)}{\sqrt{2}}$$

$$x(2i+1) = \frac{L(i) - H(i)}{\sqrt{2}}$$

,…….*(2.8)*

(b) If N is odd

$$x(2i) = \frac{L(i) + H(i)}{\sqrt{2}}$$

$$x(2i+1) = \frac{L(i) - H(i)}{\sqrt{2}}$$

For i=0, 1… (n-1)/2,…….. *(2.9)*

$$x(N-1) = L\left(\frac{N+1}{2}\right)\sqrt{2}$$

,…….. *(2.10)*

Where,

N is the number of data samples.

L is the low frequencies subband.

H is the high frequencies subband.

2.10 Fidelity Measure

The audio steganography methods are capable to hide information in audio files without changing their size. Accurately assessing for the quality of routine which requires a scheme for measuring the amount by which a loaded file differs from the output file. We use schemes, namely the mean-squared error (MSE) and the signal-to-noise ratio (SNR) of a loaded file to perform the measurement.

2.10.1 Mean Squared Error (MSE)

The MSE is a measure, commonly used in statistics. It is used to estimate the expected value of the error between corresponding values of two populations. MSE measures the average of the square of the "error." The error is the amount by which the estimator differs from the quantity to be estimated.

If x_1, $x_{2,}$. . . x_n and y_1, y_2, . . . , y_n represent samples of two signals X and Y over time, then the MSE between them is defined as:

$$MSE = \frac{1}{N}\sum_{i=1}^{n}(x_i - y_i)^2 \qquad \text{,..... (2.11)}$$

This is able to give us a sense of how much the loaded file differs from the original cover file on a sample-by-sample basis. The main drawback of the MSE is that it gives no indication of how the error is related to the original signal. For example, two signals could have the same MSE, but the effect of this noise would be much greater in the quieter signal than the louder signal [Geo99].

2.10.2 Peak signal-to-noise ratio (PSNR)

PSNR is a term for the ratio between the maximum possible power of an original signal and the power of loaded cover signal that affects the fidelity of its representation. Although, this measure is totally objective and often it does not model properly the “human auditory perception”. It is used as a transparency function in

Audio environment. Also, it is used as a measure of quality of reconstruction in audio.

Because many signals have a very wide dynamic range, PSNR is usually expressed in terms of the logarithmic scale. It is most easily defined via the mean squared error (MSE):

$$PSNR = 10Log_{10}\left(\frac{Max_i^2}{MSE}\right) \quad \text{,..... (2.12)}$$

Where the Max is the maximum possible value of the audio samples [Jor07].

2.10.3 Signal to Noise Ratio (SNR)

Signal to Noise Ratio often abbreviated **SNR** or **S/N**, it provides a measure of noise which takes into account the strength of the original signal. If X is the cover file and Y is the stego file, we define X to be the signal and X − Y to be the noise. Now the SNR is defined by

$$SNR = \frac{p_{signal}}{p_{error}} \quad \text{,..... (2.13)}$$

Where P_{signal} and P_{error} represent the power of the signal (X) and the noise (X − Y), respectively.

The power of a signal is defined as the average squared amplitude of the signal over all time. If the signal is represented as discrete samples

$x_1, x_2, \ldots, x_n$, the analogous definition of power is

$$p = k\sum_{i=1}^{n} |x_i|^2 \quad \text{,..... (2.14)}$$

Where k is a constant. This gives us an approximation of the SNR as

$$SNR = \frac{|x_1|^2 + |x_2|^2 + \ldots + |x_n|^2}{|x_1 - y_1|^2 + |x_2 - y_2|^2 + \ldots + |x_n - y_n|^2} \quad \text{,..... (2.15)}$$

As we mentioned earlier, this has the advantage of giving a sense of how prominent the error is when listening to the loaded file. The SNR is also well studied

in acoustics, and the systems which degrade audio quality often rate their performance in terms of a signal-to-noise ratio [Jor07].

Chapter Three

System Devlopment

Chapter Three

System Development

3.1 Introduction

The purpose of this project is to develop a hybrid system using two type of transform for hiding binary data in the audio. In the design phase of the proposed system the following two basic requirements have been taken into consideration: the first one is the perceptual transparency, and the second is the high embedding capacity. This first requirement is important to make cover object and stego cover object perceptually indiscernible, while the second is to insure high hiding gain.

In established transform based hiding system both wavelet and DCT transforms have been utilized, in a hybrid way, to get better hiding rate with low perceptual changes in cover media.

The above mentioned idea of using transformation in the proposed system is due to the results of previous published works which indicated that hiding in frequency domain is more effective than hiding in time domain, due to the compactness attributes of some transforms.

3.2 Audio Files

Today, many types of audio file are available, such as Windows Audio Visual (WAV), Windows Media Audio (WMA), and MPEG (MP3). The type used in this study is WAV file format of type PCM, because it is uncompressed audio format, which gives more flexibility for data hiding.

A stego object (WAV file) with high sampling rate and sampling resolution may draw suspicion, because of its large size, especially if its subjective quality is not high. Usually, it is easy to hide more secret data in the high quality audio data (for example, the use of least significant bit encoding to embed one bit in each sample,

consist of 16 bits, sample has less effect on the stego object than adding one bit in a sample consist of 8 bits). In the proposed system the wave files, with 8-bit samples resolution, are used as cover media for hosting the secret data.

3.3 The Overall System Module

The proposed system consists of two modules: Hiding module and Extraction module. The hiding module is used to hide the secret massage in cover audio file, and the extraction module is used to retrieve the secret message from stego-cover audio file.

3.3.1 Hiding Module

The block diagram of hiding module is shown in figure (3.1); it consists of the following seven main parts:

- Load audio file (cover and secret)
- Voiced /Unvoiced segmentation
- Transformation
- Quantization/ Dequantization
- Secret data embedding
- Inverse transform
- Reconstruction of stego cover file.

Each part of this module is illustrated, separately in the following subsections.

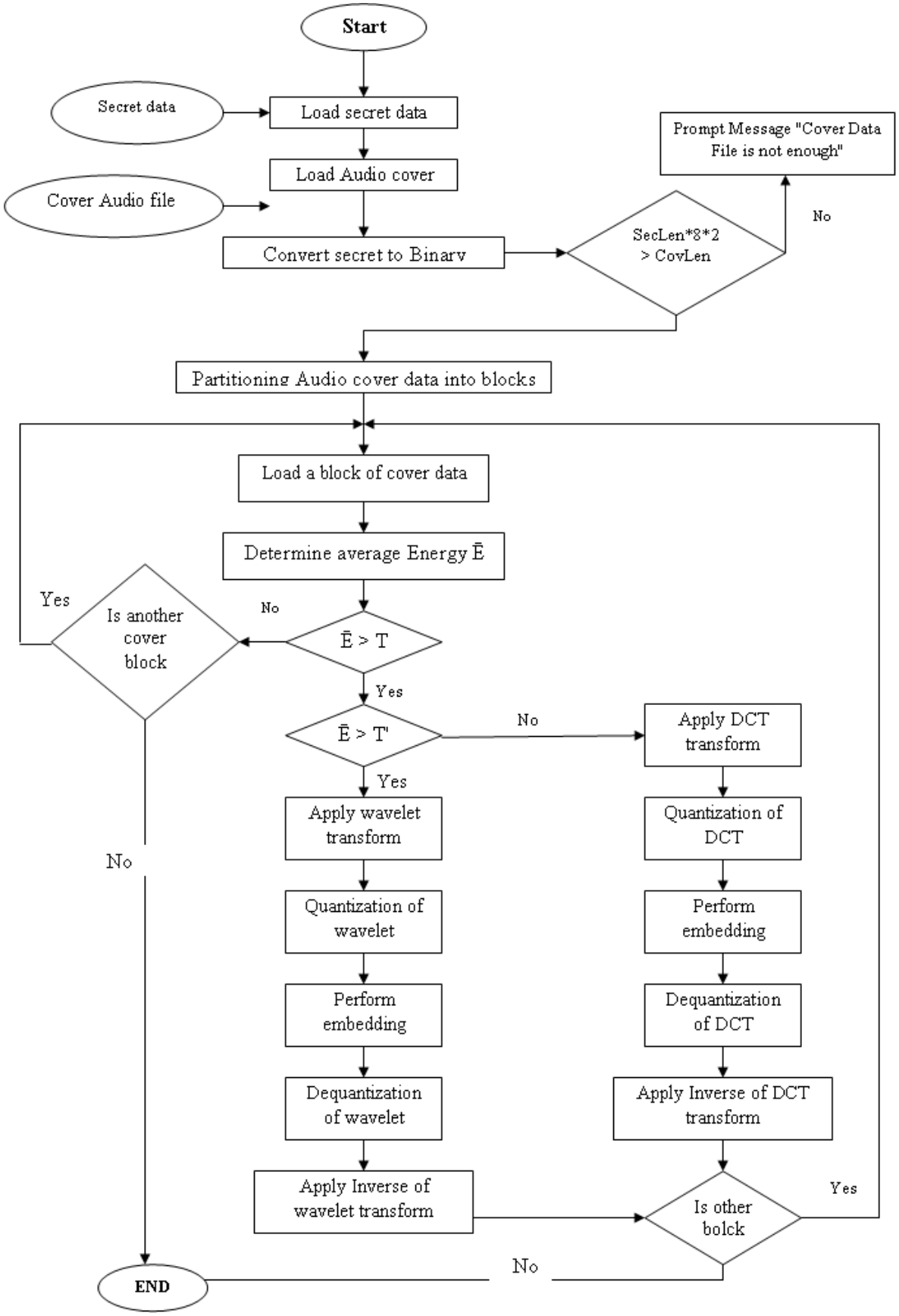

Fig (3.1) Flow chart of hiding module

3.3.1.1 Input Audio File

The cover audio file used in the proposed system is of type WAV (Windows Audio Visual) with PCM format. Opening the audio (.wav) file need a good knowledge about WAV header format. The WAV file starts out with a header contain ***main chunk*** (called RIFF chunk), followed by a sequence of data chunks. A WAV file often contains single "WAVE" chunk, which consists of two subchunks, a format (fmt), subchunk and the (data) subchunk. The first subchunk holds the file format information, while the second subchunk holds the audio_data samples. A detailed description of the WAV file format is presented in appendix (A). The wave data is stored as one channel (called mono), or as two channels (called stereo).

The audio samples resolution (i.e, number of bits per sample) is either 8 bits (ranging from 0 to 255), or 16 bits (ranging from -32768 to 32768). This may cause confusion when dealing with both kinds of sample resolution, because to handle both cases two structures for the system must be established, the first one to handle the 8-bits samples, and the second to handle the 16-bits samples. To avoid this confusion and to unify the way of handling both cases the range of the 16-bits sample was remapped to be between 0 and 255 (i.e., 8-bits resolution). Therefore, the value of each sample was represented by 8-bits (whatever its original sample resolution is 8 or 16 bits), in this case, an array of byte type was originated to represent the audio cover data. Figure (3.2) shows the block diagram of the process of loading a cover WAV file.

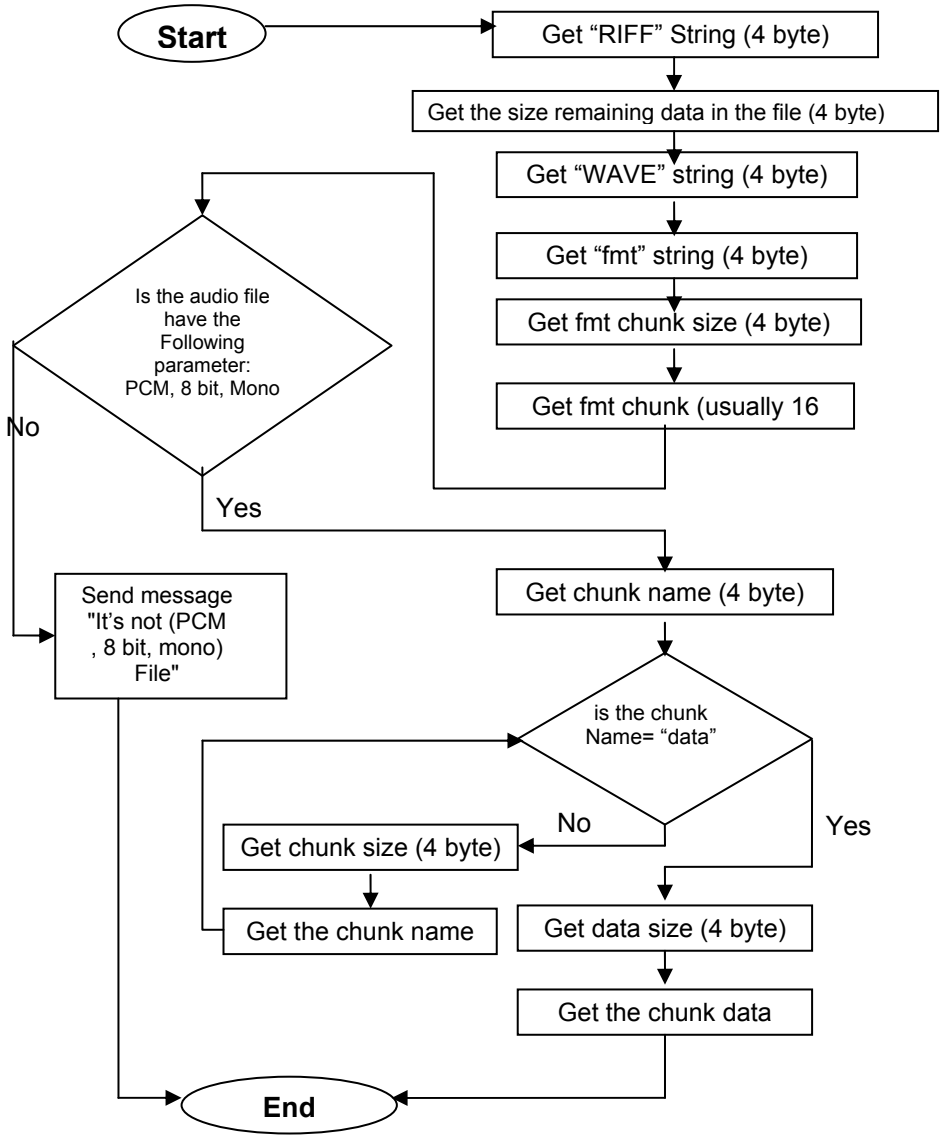

Fig (3.2) The flowchart of "Loading wave file" stage

3.3.1.2 Voiced / Unvoiced Segmentation

Sound consists of pressure waves which have a wide variety of waveforms, these waveforms can be broadly categorized into voiced and unvoiced sound.

The embedding mechanism in the proposed system is hiding secret data in the voice segments, only, of the cover. So, the audio cover data is partitioned into non-overlapped blocks, and the hiding process will be applied on the voiced blocks only. To recognize the voiced from unvoiced block the average power of each block is compared with a predefine threshold value (T), to decide if the block is voiced or unvoiced.

The adopted way to measure the power of the voice is based on calculating the average of the square of samples values of each block and then this average is compared with a threshold value (T); if it is below threshold value then the block is considered unvoiced block, while if it is equal or above threshold then the block is considered as voiced block. Algorithm (3.1) lists the steps of the voiced/unvoiced segmentation process.

Also, in this stage the power of voiced block where tested to be sure its value is not very close to threshold value (T), because if it is close to (T) then the sample values of the block should be adjusted to make its power value is far enough from the value (T). This additional step is taken as precaution step to avoid the occurrence of the case "the power of the voiced block after embedding secret data may lowered and its new value become less than T". The occurrence of such case will cause a problem in the extraction phase, because the extraction module will consider such hosting blocks as unvoiced blocks, and consequently a part of secret binary data will be lost.

Algorithm (3.1) Voiced/ Unvoiced segmentation

Goal: *determine if each block is Voiced or Unvoiced.*
Input:
Wv (0 to No. of Samples -1) //array contains audio file data
PowThr1 // first threshold value
Block Length // the size of each block
Output:
Each block of Wv vector is categorized either as Voiced or Unvoiced

Step1: *"Define the Eps1, Eps2 value*
Set *Eps1 ← PowThr1 * 0.1*
Set *Eps2 ← PowThr2 * 0.02*
Step2:" *Compute the Power of each block and compare it with PowThr1 value*
No.Block ← No.Samples / Block Length
For *i* ***Do*** *{where 0 ≤ i < No.Blocks}*
*Offset ← i * Block Length*
Pow ← 0
For *j* ***Do*** *{ where 0 ≤ j < Block Length}*
Pow ← Pow + Wv (Offset+j)^2
End For { j }
Pow ← Pow / Block Length
If *Pow ≤ Pow Thr1* ***then***
" Block is considered as unvoiced

```
        Else
            " Block is considered as voiced
            "Check if it near threshold value then adjusts the block samples
            If Pow – PowThr1 ≤ EPS1 then
                For j Do { 0 ≤ j < Block Length}
                    Wv(Offset + j) ← Wv(Offset + j) +1.1
                EndFor { j }
            End If
        End If
    End For { i }
```

3.3.1.3 Transformation

The basic step in the proposed system is the transformation of audio data from time domain to frequency (or scale domain) using either DCT or Wavelet transform. The input to transformation stage will be the voiced blocks, exclusively, of the cover audio data. To decide whether cosine or wavelet transform is applied on the voiced blocks, a second thresholding criteria is used; the threshold value (T') of this criteria should be higher than the first threshold (T). According to this thresholding criteria the block is send to cosine transform if its average power is less than the second threshold value, otherwise it is send to wavelet transform. The usage of this criteria is based on the fact that "at the signal parts which shows high variations the performance of wavelet transform is better than DCT, and vice versa.

Algorithm (3.2) shows the steps taken to decide whether the voiced block is transformed using DCT or wavelet, according to power value of the block. Also, in this algorithm the voiced block whose power value is very close to threshold (T'), are adjusted to ensure that its new power value is not close to (T').

In the following subsections the implemented steps to conduct DCT and Wavelet transform are illustrated.

Algorithm (3.2) Choosing the transform type

Goal: *to decide where the voiced block is DCT or Wavelet transform*
Input:
Wav (0 ... BlockLength-1)// a voiced block
PowThr2// a threshold value

Output:
A decision (DCT or Wavelet)

```
Step1: "Compute the power of block
       Pow ← 0
       For j Do {where 0 ≤ j < BlockLength}
           Pow ← Pow + Wav(j)
       End For { j }
       Pow ← Pow / BlockLength
Step2: "Make a decision
       If Pow < PowThr2 then
           " The block is DCT transformed
           " Adjust its samples if its power is closed to PowThr2
           If PowThr2 – Pow < 0.1* PowThr2 then
               For j Do {where 0 ≤ j < BlockLength}
                   Wav(j) ← Wav (j)* 0.85
               End For { j }
           End If
       Else
           " The block is wavlet transformed
           " Adjust its samples if its power is close to PowThr2
           If Pow- PowThr2 < 0.1 * PowThr2 then
               For j Do {where 0 ≤ j ≤ BlockLength}
                   Wav(j) ← Wav(j)*1.1
               End For { j }
           End If
       End If
```

1. DCT Transform

The DCT is one of the powerful compact transforms. It relocates most of the signal energy into the first transform coefficients, lesser energy or information is relocated into other (i.e., high frequency) coefficients.

DCT and IDCT are described by equations (2.1) and (2.2), respectively: the DCT is applied on the voiced blocks that have power less than the predefined second threshold value (T'). The block size was taken small to avoid the high computational complexity of DCT calculations which makes the system slow. Algorithm (3.3) illustrates the steps taken to map the voiced block using DCT.

Algorithm (3.3) DCT transformation

Goal: *DCT transform the data of a voice block (which belong to audio cover data)*
Input: *wav (0 .. BlockLength-1) a voice blocks (or vector).*

BlkLen // length of block.
Output: *tWv // a vector contain the embedding data.*

Step1: *"Initialization to the parameters of DCT*
Set *limit (Lm) ← blklen-1*
Set *Sm ← 0*
Set *BlkLn4 ← 4 * BlikLen*
For *j* ***Do*** *{where 0 ≤ j ≤ BlkLn4 -1}*
*Cn(j)= cos(π *j /(BlkLen*2))*
End For { j }
Set *Fac1 ← sqr (1/BlkLen)*
Set *Fac2 ← sqr(2/BlkLen)*
Step2:
1. For *I* ***Do*** *{where 0<=i<=Lm}*
Set *Sm ← Sm + wav(I)*
Set *tWv(0) ← Fac1 * Sm*
End For { I }
2. ***For*** *u* ***Do*** *{where 1<=u<= Lm}*
Set *Sm ← 0, Set k ← u, Set M ← (2* u)*
For *I* ***Do*** *{where 0 ≤ I ≤ Lm}*
Set *Sm ← Sm + Wav(I) * Cn(k)*
Set *k ← k + M*
If *k >= BlkLn4* ***Then*** *k = k - BlkLn4*
End For {I}
Set *tWv (u) ← Sm* Fac2*
End For { u }
Step3: *Save the tWv vector*
Step4: *Return the tWv vector*

2. Wavelet Transform

The input data to the wavelet transform are the samples of the cover voiced blocks; these samples are considered as waveform representation in time domain. The output is considered as representation in scale-shift domain. The blocks whose average energy is above the second threshold value is sent to wavelet transform.

In this work, the Haar wavelet is used to transform the data by calculating the sums and differences between two adjacent elements. The equations of forward Haar wavelet transform are described by equations (2.4 – 2.7) mentioned in section (2.9).

Algorithm (3.4) presents the implemented steps of forward Haar wavelet transform.

Algorithm (3.4) of Haar Wavelet Transform
Goal: *Apply Haar wavelet transform on a voice block extracted from Audio cover data.* ***Input:*** *wav (0 ..BlkLen) // a block of voice block.* *BlkLen // length of the block.* ***Output:*** *tWv // a vector contains the transform coefficients.*
Step1: *"Determine some involved parameters.* ***Set*** *Low high ← block length \2* ***Set*** *Low high M ← low high – 1* ***Step2: For*** *I* ***Do*** *{where 0≤ I ≤ Limit high M}* ***Set*** *i1 = i + I* ***Set*** *i2= i1+1* ***Set*** *tWv(I) ← Wav(i1) + Wav(i2)* ***Set*** *tWv (I+ Limit high) ← Wav(i1) – Wav(i2)* ***End For { I }*** ***Step3:*** *Return the tWv vector.*

3.3.1.4 Quantization/ Dequantization

Quantization is a process comes after the transform; the type used here is the uniform scalar quantization, where each sample is treated individually. The gaps made between the quantized values of the transform coefficients should be enough to carry the added values of secret data. The adopted uniform scalar quantization was applied using the following equation:

$$Qwv(i) = Q \times round\left(\frac{tWv(i)}{Q}\right) \qquad \text{,…….. (3.1)}$$

Where:

*t*Wv *(i)* : is the original value of i^{th} transform coefficient.

Q : is the quantization step.

Qwv (i) : is the corresponding i^{th} quantized coefficient.

i= 1, 2, ..., N-1

In both types of transforms (i.e., DCT and wavelet), the left half of their coefficients was chosen as a host media for embedding secret data. For DCT the coefficients of the left half represent the high frequency coefficients, and for wavelet transform the left half coefficients holds the detail coefficients. The reason behind

choose high frequency coefficients is relevant with the fact that the auditory system, normally, shows weak sensitivity in high frequency regions. Since the hosting of secret data will be in upper half part of transform coefficients; then only the coefficients belong to this half will be quantized. Algorithm (3.5) shows the implemented quantization process.

Algorithm (3.5) Quantization of transform Coefficients
Goal: *Quantized the cover file for embedded data* ***Input:*** *tWv // a vector represent transform coefficient* *Q // is the quantization step* *BlkLen // is the number of transform coefficients* ***Output:*** *Qwv // a vector represent data after Quantization.*
Step1: "*Determine the boundary of upper half part* *Lh = BlkLen/2* ***Step2: For*** *i* ***Do*** *{where Lh ≤ i <BlkLen}* ***Set*** *Qwv (i) ← round (tWv (i) /Q (i))* Q* ***End For { i }*** ***Step3:*** *Return Quantized vector (Qwv)*

3.3.1.5 Data Embedding

It is an important stage in hiding module. In this stage, the quantized coefficients are used as stego cover media to hold the secret data. The adopted embedding technique in the proposed system is "modulation", where the value of host coefficients are changed either by subtracting or adding a specific value (δ). The addition or subtraction will depend on the value of secret bit, such that:

$$tstg(i) = \begin{cases} tWv(i) & if\ \ i < N/2 \\ tWv(i) - \delta & if\ \ i \geq N/2\ and\ \ S = 0 \\ tWv(i) + \delta & if\ \ i \geq N/2\ and\ \ S = 1 \end{cases} \quad \ldots\ldots\ldots\ldots (3.2)$$

Where,

tstg(i) is the i^{th} transform coefficient after embedding process.

tWv(i) is the i^{th} quantized value of transform coefficient.

δ is the modulation step.

N is the length of voiced block.

S is the value of the embedded secret bit.

Algorithm (3.6) lists the implemented step of embedding secret bit in on voice cover block.

Algorithm (3.6) Embedding Process

Goal: *embedding the Secret data in cover audio file.*
Input:
Qwv // a vector holds the quantized coefficients of cover voice block
Sec //a vector holds the binary sequence of the secret massage
Blklen //number represent the length of block.
V // pointer refers to the position of secret bits waiting for embedding.
Output:
tWv //a vector represent the cover with secret massage

Step1: *"Initialization for hiding process parameter*
Set *Lh ← block length \ 2*
Set *Delta = (1 / 3) * Quantized step*
Step2:For *i* ***Do*** *{where Lh ≤ i < BlkLen}*
If *Sec (v) = 0* ***Then***
tWv (i) ← Quantized value(i) - Delta
Else
tWv (i) ← Quantized value (i) + Delta
End If
End For { i }
Step3: *Return tWv vector which represent the audio cover holding the embedded secret data.*

It is clear from equation (3.2), and also from algorithm (3.6), the embedding process is based on dealing with secret data as a sequence of binary digits {0, 1}. And, the value of each secret bit (binary digits), is used to decide what kind of modulation (i.e., addition or subtraction) is done on each transform coefficients used as host media.

The secret message is converted from an array of characters (or bytes) to an array of bits. Algorithm (3.7) shows the applied steps to make the conversion of secret message from a string to an array of bits.

Algorithm (3.7) conversion of secret message

***Goal**: To convert the string of secret message to a sequence of bits*
Input:
SecMsg // the secret message
***Output**:*
Sec ( ) // an array of secret bits

Step1: *"Ffind the length of secret message*
Lng = len (SecMsg)
Step2: *"Conversion to binary sequence*
L= 0
***For** i **Do** {where 1 ≤ i ≤ Lng}*
***For** j **Do** { where 0 ≤ j ≤ 7}*
If** SecMsg (i) and (2^ j) =0 **Then
Sec (L) =0
Else
Sec (L) =1
End If
L ← L + 1
End for { j }
End for { i }
Step3: *return the array Sec*

3.3.1.6 Inverse Transform

As a final step the transform coefficients after the embedding stage are inverse transformed (either IDCT or IHWT) to reconstruct the stego object. If the block is DCT transformed then it is, automatically, transformed using IDCT, and same thing occurs with wavelet coefficients the transformed using IHWT.

Algorithm (3.8) shows the implemented steps to conduct IDCT, and algorithm (3.9) list the steps of IHWT.

Algorithm (3.8) Inverse of DCT (IDCT) transform

Goal: *Apply the inverse DCT transform.*
Input: *tWv // vector of stego transform coefficient.*
Blklen // Block length of vector

Output:
Stg ( ) // vector of stego_cover data after applying IDCT

Step1: *"Determine of some involved parameters*
Set *Lm ← block length - 1*
Set *Fac ← sqr (1/ block length)*
Fac C= sqr(2/BlkLen)
*BlkLn4 = 4 * BlkLen*
Cn (0) = Fac C
For *u* ***Do*** *{where 0 ≤ u < BlkLn4}*
*Cn (u)= Fac C * cos (π* 4/ (2*BlkLen))*
End for {u}
Set *m ← 1*
Set *T ← tWv(0) * Fac*
Step2: For *i* ***Do*** *{where 0 ≤ i ≤ Lm}*
Set *Sum ← T*
Set *K ← M*
For *u* ***Do*** *{where 1≤ u ≤ Lm}*
Set *Sum ← Sum+ Cn(k) * tWv (u)*
Increment k by m
If *k ≥ BlkLn4* ***Then*** *decrement k by BlkLn4*
End For {u}
Set *Stg (i) ← Sum*
End For {i}
Step3:*Return the reconstructed vector {rWv}*

Algorithm (3.9) Inverse of Haar Wavelet transform

Goal: *Apply the inverse Haar Wavlet transform*
Input:
tWv // vector of stego transform coefficients
Blklen // block length
Output:
Stg ( ) // vector represent the stego_cover data after applying Inverse of Wavelet transform.

Step1: *"Determination of the involved parameter*
Set *Lh ← block length \ 2*
Step2: For *i* ***Do*** *{where 0 ≤ i < Lh}*
Set *Ij ← 2 * i*
Set *Ip ← I + Lh*
Set *rWv (Ij) = (tWv (i) + tWv (Ip))/2*
Set *rWv (Ij+1)= (tWv (i) – tWv (Ip))/2*
End For
Step3: *Return the reconstructed Wave block {rWv ()}*

3.3.1.7 Reconstruction of Stego Cover File

This step is performed to save the stego-object data into an audio file. The chosen audio format (.wav) format with 8-bit sample resolution, PCM-coding, single channel (mono). At first the header of wave file is constructed and saved in the file, then the stego-audio data are saved.

3.3.2 Extraction Module

This module is used to retrieve the original secret binary message from the stego audio data; it works in reverse way to that of hiding module. The extraction module consists of the following stages:

- Loading stego audio file
- Voiced \Unvoiced segmentation
- Transformation
- Extracting the embedding secret data

Figure (3.3) illustrates the stages of the extraction module. The main stages of this module are described in the following subsections:

3.3.2.1 Load Audio File

The steps of loading the stego audio file are the same like the steps mentioned in section (3.3.1.1), which load the WAV file and extract the header information and the audio stego data.

3.3.2.2 Voiced /Unvoiced Segmentation

Segmentation for voiced and unvoiced is also the same process mentioned in section (3.3.1.2). This stage is important to determine if the block is voiced or unvoiced. Taking into consideration that only the voiced blocks are used as host space for hiding the secret data.

3.3.2.3 Transformation

The steps of transformation stage are similar to those followed in the corresponding stage in hiding module (see section 3.3.1.3).

3.3.2.4 Extraction of the Embedded Data

The extraction process is the inverse of embedding process, in this stage the value of the transform coefficient is compared with its quantized value to decide whether the embedded data is 0 or 1, such that:

If $C > Q * round\left(\frac{C}{Q}\right)$ then S=0 else S=1

Where,

C is the transform coefficient.

Q is the quantization step.

S is the retrieved secret bit.

Algorithm (3.9) shows the applied steps for extraction.

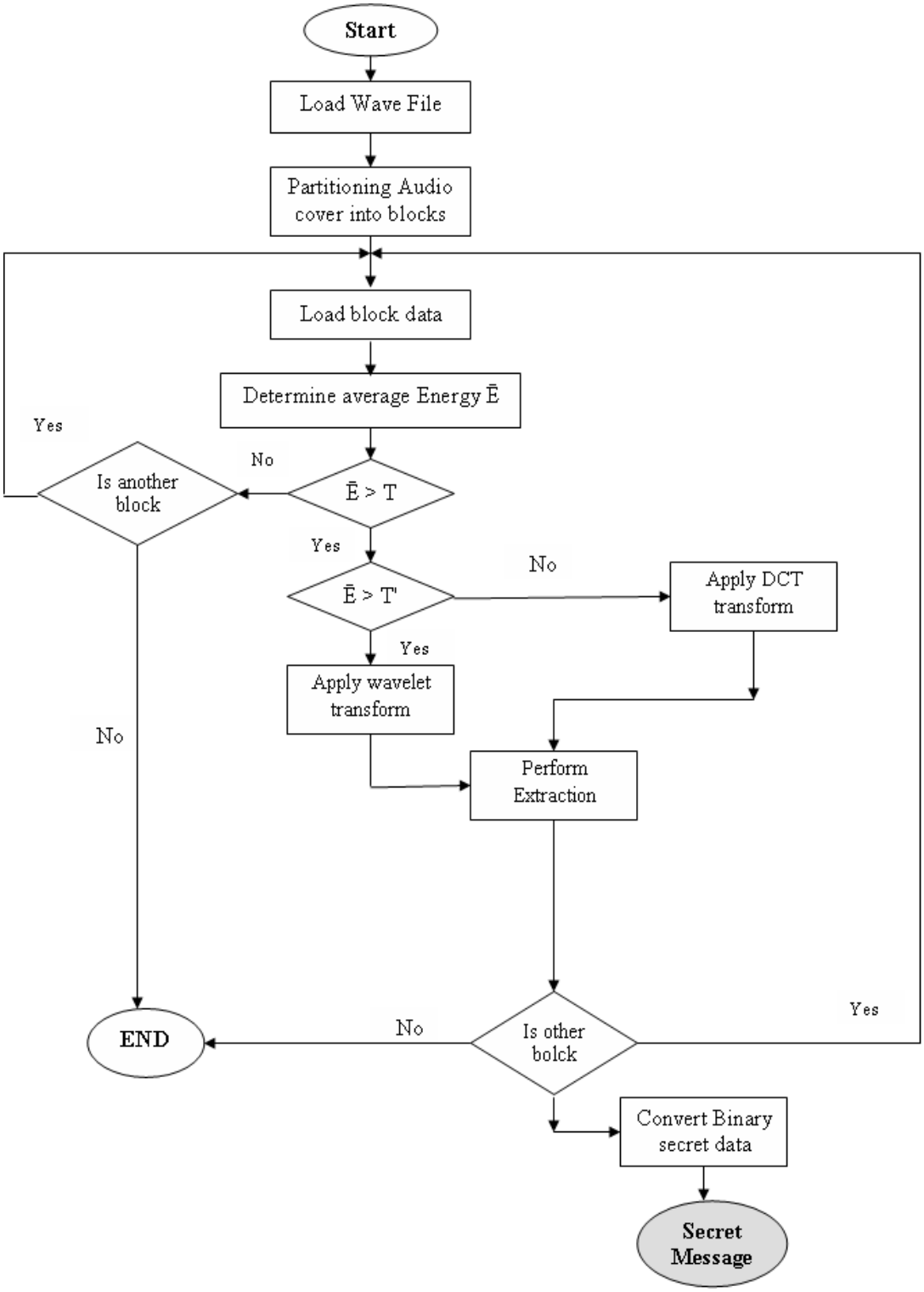

Fig (3.3) The extraction module

Algorithm (3.9) for Extraction Module

Goal: *retrieve the embedding data from cover file.*

Input:

tWv // vector represent the quantized transform coefficient of stego block
BlkLen // number represent the block length.
V // a pointer to the location of retrived secret bit in the array of secret binary array.
Q // the quantization step

Output:

ExSec // vector represent the orginal embedded data (secret message).

Step1: *"Determine the start index of the upper half part of the block*
Lh= BlkLen /2

Step2: For *i* ***Do*** *{ where Lh ≤ i < BlkLen}*
*Qwv ← Q * round (tWv (i) / Q)*
If *qwv > tWv (i)* ***Then*** *ExSec(v)=1 Else ExSec (v)=0*
Increment v by one
End for{i}

Step3: *Return the Secret Message (ExSec) from the cover file.*

Chapter Four

Experimental Results

Chapter Four

Experimental Results

4.1 Introduction

In this chapter, the results of some conducted tests are presented. These tests are performed to evaluate the performance of the proposed audio hiding methods. Also, in this chapter the effects of the involved system parameters on its performance behavior are explored.

4.2 Test Measures

The measures used in this research work to assess the performance of the hiding method are the followings:

A. Fidelity Measures

The adopted fidelity measures are:

- Mean Absolute Error (MAE),
- Mean Square Error (MSE),
- Peak Signal to Noise Ratio (PSNR).

The definition of the above fidelity measures are given in chapter two, see equations *(2.11), (2.12), (2.13)*. PSNR are often measured in logarithmic scale and the unit of this measure is called decibel (dB). These measures have been used to represent the overall error caused in stego-cover due to embedding of secret data.

B. Hiding Rate (HR)

This parameter is determined to assess the capability of the tested hiding method to embed secret information in the cover object. Mathematically, hiding rate is defined as equation (4.1):

$$HR = \frac{S}{8 \times M} \times 100\% \qquad ,\ldots\ldots(4.1)$$

Where, S is the maximum possible number of inserted bits in the cover media.

M is the size of cover media (in bytes).

C. Ratio of Correctly Retrieved Secret Bits (ER)

This parameter is determined to describe the capability of the tested hiding method to preserve the integrity of the embedded secret data. It is defined as the ratio of the number of correct secret bits (S_c) extracted from the stego object relative to the total number (S) of embedded secret bits:

$$ER = \frac{S_c}{S} \times 100\% \qquad ,\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(4.2)$$

4.3 Test Samples

Some audio files have been selected as cover media to investigate the hiding performance. Table (4.1) shows a list of the selected cover files, while table (4.2) shows a list of files whose content used as test material for embedding.

Table (4.1) The list of Wave files used as cover media

File Name	*File Size*	*Sampling Rate (KHz)*
Spch1	28.2 KB	8
Sng1	46.2 KB	11
Sng2	46.5 KB	8
Music1	88 KB	11.10
Spch2	148 KB	11
Sng3	156 KB	8
Sng4	183 KB	8
Spch3	213 KB	8
Sng5	626 KB	8

All the above sound files are of WAV type, PCM format, Mono and Sample resolution=8 bits.

Table (4.2) The list of files whose contents used as secret material

File Name	*File Size (KByte)*	*File Type*
SngSec	148	Wav
SpcSec	28	Wav
DocSec	55.5	DOC
ImgSec	21.1	JPG

4.4 Test Results and Discussion

The performance of the proposed hiding methods was based on studying the effects of changing the involved parameters on hiding performance measures. The results of the various conducted tests have indicated that the type of cover audio file has significant effect on the performance parameters (like, hiding rate, correct retrieval bits and quality of stego cover file). Therefore, the choice of audio cover file should be handled carefully, for example the low quality audio cover is recommended more than the high quality audio cover because hiding in the first type of cover data will not cause noticeable extra noise. It was noticed that the result of the tests vary from cover file to another.

The control parameters that have significant effects on the performance parameters are the two threshold values, DCT quantization step, Wavelet quantization step, and the block length. In the following sub sections the effects of the above control parameters on the system performance are classified.

4.4.1 Threshold Value (Thr)

The classification decision of voiced/unvoiced blocks depends mainly on the value of power threshold, if the energy of the audio block is higher than the first threshold (T) then block is classified as voiced block. The second classification is applied only on the voiced block; this classification is for deciding the type of transform (DCT or Wavelet) should be applied on the voiced block.

Table (4.3) illustrates the effect of first threshold (T) on the performance parameters, when T is increased the hiding rate of secret data is decreased. during this

test the values of other coding parameters were set to have their default values: T'= 8→12 (must be greater than first threshold value); QStp=3 (for both DCT and Wavelet); Block Length=12.

Table (4.3) The effect of the first threshold value (T) on the performance of hiding method

T	*T''*	*MAE*	*MSE*	*PSNR*	*MAE(Sec)*	*Ratio of correct Retrieval bits*	*Hiding Rate*
1	8	0.456	1.684	45.87	0.077	92.260	5.077
2	8	0.425	1.645	45.97	0.786	92.135	4.753
3	8	0.406	1.621	46.03	0.080	91.965	4.559
4	8	0.387	1.598	46.10	0.081	91.881	4.373
5	9	0.373	1.584	46.13	0.822	91.777	4.181
6	9	0.355	1.561	46.20	0.084	91.584	3.974
7	9	0.334	1.534	46.27	0.086	91.387	3.776
8	10	0.316	1.515	46.33	0.086	91.320	3.571
9	11	0.302	1.504	46.36	0.086	91.345	3.388
10	12	0.289	1.491	46.40	0.084	91.526	3.223

Table (4.4) illustrates the effect second threshold (T') of the performance parameters, when T' is increased the correct retrieval bits of secret data is increased. During this test the values of other coding parameters were set to have their default values: T=2; QStp=3 (for both DCT and Wavelet); Block Length=12.

Table (4.4) The effect of the second threshold value (T') on the performance of hiding method

T''	*MAE*	*MSE*	*PSNR*	*MAE(Sec)*	*Ratio of correct Retrieval bits*	*Hiding Rate*
3	0.352	2.475	44.20	0.128	% 87.182	% 4.290
4	0.357	2.485	44.18	0.122	%87.760	% 4.290
5	0.363	2.496	44.16	0.118	%88.185	% 4.290
6	0.369	2.509	44.14	0.113	%88.676	% 4.290
8	0.376	2.524	44.11	0.106	%89.356	% 4.290

9	0.380	2.532	44.10	0.103	%89.607	% 4.290
11	0.387	2.547	44.07	0.099	%90.028	% 4.290
12	0.390	2.553	44.06	0.097	%90.199	% 4.290
14	0.399	2.575	44.02	0.094	%90.541	% 4.290
16	0.407	2.594	43.99	0.090	%90.978	% 4.290

4.4.2 Quantization Step (Q)

Quantization process provides a suitable space for hiding secret data. The main parameter used to mange the amount of space is the quantization step (*Q*). In our system there are two quantization parameters: one for Wavelet coefficients (*WaveletQnt*),and the second for DCT Coefficients (*DCTQnt*). Any change in one of these two parameters affects the hiding space in some voiced blocks. If the values of Quantization steps are increased then the ratio of correct retrieval bit is increased and the error in stego cover signal increased too. Table (4.5) presents the effect of changing the *wavelet QStp* on the hiding performance parameters. The values of other system parameters were fixed to have the following values: T=2; DCT QStp=3; T'=9; Block Length=12.

Table (4.5) The effect of wavelet quantization step on the performance of hiding method

Wavelet QStp	*MAE*	*MSE*	*PSNR*	*MAE(Sec)*	*Ratio of correct Retrieval bits*	*Hiding Rate*
2	0.250	2.401	44.33	0.192	% 80.745	% 4.290
3	0.380	2.532	44.10	0.103	% 89.607	% 4.290
4	0.515	2.799	43.66	0.104	% 89.519	% 4.290
5	0.554	2.925	43.47	0.008	% 99.133	% 4.290
6	0.699	3.299	42.95	0.008	% 99.137	% 4.290
7	0.803	3.717	42.43	0.008	% 99.133	% 4.290
8	0.850	3.871	42.25	0.008	% 99.137	% 4.290

Table (4.6) presents the effect of changing the value *DCT QStp* on the hiding method. The values of other system parameters were taken as follows: T=2; T'=9; Wavelet QStp=3; Block Length=12.

Table (4.6) The effect of DCT quantization step on the performance of hiding method

DCT QStp	*MAE*	*MSE*	*PSNR*	*MAE(Sec)*	*Ratio of correct Retrieval bits*	*Hiding Rate*
2	0.348	2.476	44.19	0.124	% 87.555	% 4.290
3	0.380	2.532	44.10	0.103	% 89.607	% 4.290
4	0.411	2.607	43.97	0.097	% 90.199	% 4.290
5	0.443	2.705	43.81	0.095	% 90.424	% 4.290
6	0.472	2.813	43.64	0.095	% 90.465	% 4.290
7	0.500	2.933	43.46	0.095	% 90.465	% 4.290
8	0.529	3.075	43.25	0.095	% 90.465	% 4.290

4.4.3 Block Length

The effect of block length on the hiding performance is illustrated in table (4.7). The number of inserted bits is increased when block length become large, but at same time the error level is increase because number of changed samples in cover data is increased due to embedding. The values of other coding parameters were fixed to have the following values: T=2; T'=9; Wavelet QStp=3; DCT QStp=3.

Table (4.7) The effect of block length parameter on hiding method

Block Length	*MAE*	*MSE*	*PSNR*	*Secret*	*no. of Hidden bits*	*Correct Retrieval*	*Hiding Rate*
8	0.375	1.681	45.87	0.102	26020	% 89.727	%4.242
12	0.380	2.532	44.10	0.103	26316	% 89.607	%4.290
16	0.378	1.681	45.87	0.106	26560	% 89.363	%4.330
24	0.405	5.096	41.06	0.106	26904	% 89.325	%4.386
32	0.379	1.676	45.89	0.108	27056	% 90.000	%4.411

4.5 Performance Comparison

In this section, the effects of using diffrent. Table (4.8) lists some of the result obtained by using different covers and secret files. Also, the values of the involved control parameter was taken according the pervious testes which indicates the most suitable value for each parameter, The values of other constant parameters were taken as follows: PowThr=2; TrnsThr=9; Wavelet QStp=3; DCT QStp=3; Block Length=12. Table (4.8) illustrates the effect of using different covers.

Table (4.8) The effect of using different covers and secret audio data

Cover	*Secret*	*MAE*	*MSE*	*PSNR*	*Correct Retrieval*	*Hiding Rate*
Spch1	SngSec	0.432	2.205	44.70	93.276	4.494
Spch1	SpchSec	0.432	2.205	44.70	93.258	4.494
Spch1	DocSec	0.421	2.203	44.70	91.878	4.494
Spch1	ImgSec	0.436	2.212	44.68	91.415	4.494
Sng1	SngSec	0.908	5.252	4.93	99.962	6.214
Sng1	SpchSec	0.908	5.252	40.93	99.974	6.214
Sng1	DocSec	0.554	4.385	41.71	90.518	6.214
Sng1	ImgSec	0.561	4.387	41.71	89.693	6.214
Sng2	SngSec	0.266	2.671	43.86	91.612	2.928
Sng2	SpchSec	0.266	2.671	43.86	91.639	2.928
Sng2	DocSec	0.266	2.675	43.86	90.098	2.928
Sng2	ImgSec	0.267	2.675	43.86	89.417	2.928
Music1	SngSec	0.380	2.532	44.10	89.607	4.290
Music1	SpchSec	0.384	2.537	44.09	91.187	4.290
Music1	DocSec	0.379	2.535	44.09	90.937	4.290
Music1	ImgSec	0.384	2.536	44.09	89.808	4.290
Spch2	SngSec	0.432	1.660	45.93	92.334	4.753
Spch2	SpchSec	0.430	1.660	45.93	90.843	4.753

Spch2	DocSec	0.425	1.657	45.94	90.840	4.753
Spch2	ImgSec	0.430	1.659	45.93	90.261	4.753
Sng3	SngSec	0.523	0.886	48.66	91.652	5.906
Sng3	SpchSec	0.524	0.890	48.64	90.319	5.906
Sng3	DocSec	0.520	0.886	48.66	90.370	5.906
Sng3	ImgSec	0.521	0.887	48.65	89.807	5.906
Sng4	SngSec	0.551	0.800	49.10	91.561	6.108
Sng4	SpchSec	0.551	0.800	49.10	90.466	6.108
Sng4	DocSec	0.551	0.801	49.09	90.580	6.108
Sng4	ImgSec	0.549	0.799	49.11	90.102	6.108
Spch3	SngSec	0.320	0.364	52.52	92.549	3.563
Spch3	SpchSec	0.322	0.366	52.49	91.268	3.563
Spch3	DocSec	0.318	0.365	52.51	91.064	3.563
Spch3	ImgSec	0.318	0.364	52.52	90.756	3.563
Sng5	SngSec	0.567	0.964	48.29	92.051	5.885
Sng5	SpchSec	0.564	0.959	48.31	91.784	5.885
Sng5	DocSec	0.560	0.959	48.31	92.323	5.885
Sng5	ImgSec	0.565	0.961	48.30	91.403	5.885

Table (4.9) The effect of using different transform type on different audio cover

Cover	*Secret*	*Wavelet (HR)*	*Wavelet (MSE)*	*DCT (HR)*	*DCT (MSE)*
Spch1	SngSec	4.559	2.552	4.559	2.821
Spch1	SpchSec	4.559	2.539	4.559	2.829
Spch1	DocSec	4.559	2.530	4.559	2.818
Spch1	ImgSec	4.559	2.528	4.559	2.10
Sng1	SngSec	4.290	2.470	5.201	2.729
Sng1	SpchSec	4.290	2.471	5.201	2.731
Sng1	DocSec	4.290	2.474	5.201	2.735
Sng1	ImgSec	4.290	2.479	5.201	2.740
Sng2	SngSec	3.885	1.744	2.885	1.711

Sng2	SpchSec	3.885	1.740	2.885	1.709
Sng2	DocSec	3.885	1.737	2.885	1.707
Sng2	ImgSec	3.885	1.734	2.885	1.704

Chapter Five

Conclusions and Suggestions

Chapter Five

Conclusions and Suggestions

5.1 Introduction

In this chapter, a list of conclusion remarks are presented, they derived from the investigation of the test results shown in chapter four. Also, some suggestions for a future work are presented; they may enhance the system efficiency.

5.2 Conclusions

From the test results listed in pervious chapter the following remarks were stimulated:

1. Using two type of transform (DCT and Wavelet) provides a flexibility to utilize the power of more than one transform to perform hiding in audio signal without making subjective distortion. The block may be DCT or Wavelet transformed according to its power relative to a threshold value. The change in threshold value will cause a change in the ratio of voiced blocks transformed by Wavelet or DCT.
2. The use of two predefined threshold value: power threshold (T) and transform threshold (T'), added some power to the steganography process, because it is divided the audio cover signal blocks into different types (voiced and unvoiced) blocks. Large threshold values increase the avoidance of unvoiced blocks and cause an increase in the ratio of correct retrieved bits, but decreases the hiding rate.
3. Hiding in voiced blocks is more suitable to avoid the occurrence of stick noise, which is more probably happen when unvoiced blocks are used as host area.
4. Large quantization step provides more robust space to hide secret bits, while it causes subjective distortion in cover audio.

5. The unvoiced data samples are one of the worst host region in the cover audio data, because the noise can be easily noticed when it reside the silent region, so the selection of the cover file must be done carefully. In other words the selection of aloud audio cover with little unvoiced area in necessary to get good hiding performance result so the use of some music audio as cover media is most suitable for hiding since the distortion due to data embedding doesn't obviously sensible.

5.3 Suggestions

The developed hybrid system needs some enhancements to improve its performance and to override some of its weak aspects. Therefore, a future development is needed, and in the following some suggestions are derived for future relevant work:

1. Using other criteria to partition cover data into voiced or unvoiced regions. Since the insertion of the secret data may reduce the power of block, and convert its identity from voiced to unvoiced block. So, some efforts must spent to prevent the occurrence of such a case, because its occurrence affects the integrity of the extracted secret message.
2. Using other criteria to determine the most suitable transform that a block should mapped to (i.e., which is better for a block to be transformed using DCT or Wavelet).
3. Develop the proposed system to be used for hiding in other audio file formats likes (MP3 and ADPCM wav file).
4. Develop a system for hiding audio in image or audio in video using the same embedding technique that used in the proposed system.

References

References

[Ahm74] Ahmed, N., Nararajan, T., and Rao, K. R., "*Discrete cosine transform*", IEEE Transactions on Computers, pp. 90-93, Jan. 1974.

[Are00] Areepongsa, S., Kaewkamnerd, N., Syed, Y. F., and Rao, K. R., "Exploring on steganography for low bit rate wavelet based coder in image retrieval system", Research in Tencon Unversity, 2000.

[Aud03] Auday, A., "Fractal image compression with fasting approach", Msc. Thesis, Dept. of Computer Science, Al-Nahrain University, Iraq, 2003

[Alf06] Alfaraj, A., "On the limits of steganography", M.Sc. thesis, Information Security, UCL, 2006.

[Ben96] Bender, W., "Techniques for data hiding", IBM System Journal; vol .35, no. 3-4, pp. 1-10, February, 1996.
http://www.research.ibm.com/journal/sj/353/sectiona/bendeaut.html#bender

[Bak03] AL Baka'a , A. T. ,"Image copyright protection using digital watermarking ", M.Sc. Thesis, Computer Science Department, College of Science, AL-Nahrain University, Iraq, 2003.

[Bri06] Brian, J., Yuliya, K., and Andrew, L., *"Audio steganography"*, CDP Digital Audio Working Group, 2006.

[Bak07] Bakshi, B., "Steganography", Reseach , Syracus University, 2007.

[Cyb87] Cyber, B., "Digital Audio Basics ", Paper, Department of Interior, National Park Service, Edison National Historic Site, 1987.

[Cra98] Craver, S., *"On public key steganography"*, Information Hiding: Second International Workshop, Proceedings, vol. 1525 of lecture Notes in Computer Science, Springer , pp 355-368, 1998.

[Cac00] Cacciguerra, S., and Ferretti, S.,"Data hiding: Steganography and copyright marking", Paper, Department of Computer Science, Bologna University, Italy, 2000.

[Cum04] Cummins, J., Diskin, P., Lau, S. and Parlett, R.,"Introduction to steganography", School of Computer Science, University of Birmingham, 2004.

[Dec99] Decamro, L., "New Technologies for audio copy Protection", Electronic Media Article, 1999.

[Dav00] Davern, P., and Scott, M., "Steganography: its history and its application to computer based data files", Research, School of Computer Application, UK, 2000.

[Die03] Yasmeen I. Dieab., "Text in audio watermarking system", M.Sc. Thesis, Computer Science Department, College of Science, Al-Nahrain University, Baghdad-Iraq, 2003.

[Ett98] Ettinger, J. M., *"Steganalysis and game equilibria"*, Information Hiding: Second International workshop, Proceedings, vol. 1525 of Lecture Notes in Computer Science, pp. 319-330, Springer, 1998.

[Fab99] Fabien, A., Petitcolas, P., Ross, J. A., and Markus G. K., "Information hiding-a survey", IEEE special issue on protection of multimedia content, 87(7):1062{1078, July 1999.

[Geo99] George, C., and Lehmann, E.L., "*Theory of point estimation*", Springer, 1999.

[Gar01] Gary C. K., "Steganography: hiding data within data", Paper in Windows & .NET Magazine, September, 2001.

[Hai07] Haibin, H., "Lossless audio coding for MPEG-4", Research, Computer Science and Statistics Dissertation, University of Joensun 2007.

[Joh98] Johnson, N. F., jajodia, S., "*Steganalysis of images created using current steganography software*", information hiding Second International Workshop, Proceedings, vol. 1525 of Lecture Note in computer science, pp 273-289, Springer, 1998.

[Joh01] Johnson, N. F., Duric, Z., and Jajodia, S., "*Information hiding: steganography and watermarking – attacks and countermeasures*", research, Kluwer Academic Publishers, 2001.

[Jia03] Jianyun , X., Andrew H. S., Peipei, S. and Qingzhong, L., "Text steganography using wavelet transform", M.Sc. thesis, Department of Computer Science, New Maxico Tech, Socorro, NM87801, USA, 2003.

[Jor07] Jordi, H., "*Information hiding and steganography",* Conference on Cryptology and Digital Content Security**,** study on Information hiding and Multimedia, Universitat Oberta de Catalunya, May 15, 2007.

[Kie98] Kientzle, T., "Programmers guide to sound", Addison Wesley Developers Press, 1998.

[Kat00] Katezbeisser, S. P., and Fabian, A., "*Information hiding techniques for steganography and digital watermarking*", Artech House Inc., Norwood University, 2000.

[Kes01] Kessler, G. C.," Steganography: hiding data within data", Windows & .net Magazine, September 2001.

[Kaw06] AL Kawaz , H. M**.**, "Low rate hiding in audio data using phase domain ", M.Sc. thesis, Computer Science Department, College of Science, AL-Nahrain University, Baghdad, Iraq, 2006.

[Maj04] Majeed, M. N, "Sound hiding and extraction using DCT", M.Sc. Thesis, Computer Science Department, College of Science, Al-Nahrain University, Baghdad-Iraq, 2004.

[Moh04] Mohammad, N.M., "Sound hiding and extraction using DCT", M.Sc. Thesis, Computer Science Department, College of Science, AL- Nahrain University, Baghdad, Iraq, 2004.

[Rob96] Robi, P., "*The wavelet tutorial*", research, 136 Rowan Hall, Dept. of Electrical and Computer Engineering, Rowan University, 1996..

[Ras02] Rastislav, H., Peter, F., Dušan, L**.,** *"Steganography based on DWT transform"*, Department of Electronics and Multimedia

Telecommunications, paper, Technical University of Košice, Slovak Republic, 2002.

[Sel99] Sellars, D., "Techniques for data hiding", In IBM Systems Journal, Vol. 35, Nos. 3-4, pages 313-336, February 1999.

[Sib04] Si, B., "Introduction to steganography", Athabasca University, COMP607 Project, 25 July 2004.

[Xiu07] Xiuhui G., Renpu J., Hao T., Jiazhen W., **"**Research on information hiding**",** Research, Hebei University of Economics and Business/Ordnance Engineering College, 2007.

[Yan01] Yang, Y., "Digital watermarking technologies", Computer Science, Dalhousie University, 2001.

[Ziy02] AL-Ta'l, Z., "Development of new cover audio cryptographic models", Ph.D Thesis, University of Technology, July 2002.

Appendix A

The WAV file Format

The RIFF Header Chunk

Offset	*Size (Byte)*	*Field name*	*Content Description*
0	4	Chunk ID	Contain the string "RIFF" in ASCII
4	4	Chunk Size	The size of WAV file in bytes minus 8 bytes (4 byte to Chunk ID and 4 byte to Chunk size)
8	4	Format	Contain the string "WAVE"

The fmt Sub-Chunk

Offset	*Size (Byte)*	*Field name*	*Content Description*
12	4	Sub-Chunk1 ID	Contain the string *"fmt"*
16	4	Sub-chunk1 Size	This number describes the rest sub-chunk data: audio format (2 bytes), number of channels (2 bytes), sample rate (4 bytes), bytes/sec (4 bytes), block alignment (2 byte), bits/sample (2bytes). This, usually, has length (16) bytes.
20	2	Audio Format	Type of WAVE format. (1) mean PCM other than 1 indicates some form of compression.
22	2	No. of Channels	Channels: Mono=1, Stereo=2, etc.
24	4	Sample Rate	Sample per second. e.g., 8000,22000,44100, etc.
28	4	Byte Rate	Byte per second = Sample Rate × Channel Numbers × BitsPerSample/8
32	2	Block Align	The number of bytes for one sample including all channels. It is equal to number of channels ×

			BitsPerSample/8
34	2	Bits Per Sample	Value 8= 8 bits, value 16= 16 bits, etc.

The Data Sub-Chunk

Offset	***Size (Byte)***	***Field name***	***Content Description***
36	4	Sub-Chunk2 ID	Contains the string "data"
40	4	Sub-Chunk2 Size	Size in bytes refers to the size of the rest of sub-chunk, which means the actual data sound.
44	*	Data	The sound data.