CHAPTER 1
INTRODUCTION

1.1 Background

Downloading and uploading of multimedia files has increased dramatically in recent years due to rapid Internet network improvement. People can easily obtain a large amount of multimedia data through the Internet, especially audio files like music. Listeners spend 18.4 hours each week listening to their favorite music, according to the International Federation of the Phonographic Industry (IFPI). This is an increase from 18 hours in 2019, when the survey was previously conducted [1].

With the increase of music listeners and 'free' and 'illegal' were nearly synonymous in the music industry., multiple ways to free download unauthorized copyright music remains the issue. Downloaded music files is estimated 95% to be illegal file-sharing [2]. The consumption of illegal digital music audio or digital audio piracy can result in declining revenue and possibly lower music-making bids. Digital piracy in audio means unauthorized use, copying, and selling of digital audio [3]. The security of digital resources requires the use of digital technologies. At the same time, the work's copyright owner can utilize system technology to check the work's way, time, period, and region more quickly and easily [4].

Digital watermarking is a promising technology to counter digital piracy. Digital watermarking can be used to establish ownership rights, ensure permitted access, avoid illegal reapplications, and make content authentication easier, among other things [5]. Digital watermarking is the process of hiding information by inserting it into the host signals, including video, audio, and images, this thesis simply focusing on audio data. In audio watermarking, information for copyright protection can be imperceptibly embedded into an audio file. For example, copyright information for music files such as publisher ID, user ID, and file transaction details, or maybe in the bio-medical field the hospital can embed their watermark into heartbeat signals. Without resorting to the original audio file, an authorized user or institution could extract the watermark information from the watermarked audio file using a secret key. When the watermark information is extracted, the source of any illegal actions can be traced. Audio watermarking performance is observed at least by three important aspects such as imperceptibility, robustness, and embedding capacity [6]. The majority of digital audio publishers aspire to integrate digital watermarks in their works for copyright protection and integrity certification without sacrificing audio quality [7].

Several different audio watermarking methods have been proposed and researchers have been quite interested in Spread Spectrum (SS) [8,9]. The concept of SS-based audio watermarking is the watermark bit of information is spread throughout the host audio signal spectrum by the use of reference patterns obtained from generators of pseudo random (PN) sequences. The embedding and extraction structures of SS-based audio watermarking are simple and capable of performing extraordinarily in terms of robustness against conventional attacks while maintaining the imperceptibility and embedding capacity [10].

In the previous study [11], the audio watermarking embedding process use SS-based algorithm in the Discrete Cosine Transform (DCT) domain. Xiang *et al.* compares among other SS-based audio watermarking with PN sequence and the proposed method has much higher embedding capacity, because multiple watermark bits is represented by one PN sequence. On the result, the value of ODG is -0.7 with embedding rate of 84 bps. The proposed method is not robust against desynchronization attack and did not compromising perceptual quality, but it ensures that host signal interference problem during the watermark extraction process does not happen and ensuring that it could achieve high robustness and high embedding capacity.

Audio watermarking in the previous study [12], Luo *et al.* proposes a novel dual-domain audio watermarking approach based on flexible segmentation and adaptive embedding. Firstly, the audio beats are detected, and then the audio is segmented using the beats. Then, the embedding points are chosen, and the adaptive quantization steps for each domain are determined. Finally, the DC-DM technique embeds the watermark into both domains at the same time. The proposed approach conducts the same processing on the audio signal during the extraction step as it does during the embedding stage. The experimental data in [12] shows SDG values are nearly equivalent to 0, SNR values range from 26 to 33 dB, and BER values are modest in the face of signal processing attacks. Approaches in paper [12] shows a strong robustness to signal processing attacks and guarantee good imperceptibility.

The previous study [13], Kaur *et al.* proposes a localized audio feature based adaptive audio watermarking technique in the wavelet domain using detailed coefficients of wavelet transform (DWT). In this paper, in order to ensure strong robustness against signal processing attacks, an audio signal is segmented into a number of frames that is equal to the number of watermarking bits, with the goal of allowing one bit to be embedded in each frame. Utilizing the adaptive payload estimate approach suggested in this work, the number of watermarking bits is calculated. The wavelet domain is applied to audio frames, and the detailed wavelets coefficients at the fifth level are chosen for the watermarking bit embedding. Using the high frequency band of fifth level decomposition where human auditory system is not very much sensitive to changes, high embedding capacity has been achieved within perceptual constraints. The experimental results in [13] shows the proposed adaptive algorithm has good imperceptibility with good robustness against signal processing attacks at adjustable payload for different types of audio signals.

Previous study [14] Lu *et al.* proposes a Robust Feature Points Scheme (RFPS) or determining the second-order derivative of the original audio signal's greatest response value. As a feature point, the maximum response value will be employed, and audio segments centered on the identified feature points will be taken for both watermark embedding and extraction. In this paper, the watermark embedding is using the SS algorithm with Pseudo-Noise (PN) sequences for spreading the watermark information throughout the audio signal spectrum. Stationary Wavelet Transform (SWT) technique is used to transform the host audio into the frequency domain. The experimental results in [14] show that the proposed method can survive common audio signal processing, well resisted to various attacks without compromising the imperceptibility. Moreover, the proposed method has good robustness against desynchronization attacks.

In this thesis, the audio watermarking methods utilize the signal differential concept the second-order derivation to extract feature points. The detected feature points are used to center the embedded segment and extracted segment. For watermark embedding, this thesis utilize the SS-based algorithm, specifically Multibit SS technique and SWT method in a certain decomposition level and subband to transform the host audio signal from time domain to frequency domain.

Based on the issues in digital music industry mentioned above, digital audio watermarking is an important technology to be applied in the digital music industry. This research is an update from previous research. This research also has a positive impact in the environment, especially for digital music artists. Therefore, research on Adaptive Segmentation on Audio Watermarking using Signal Differential Concept and Multibit Spread Spectrum Technique is important to do.

1.2 Problem Formulation

The SS-based algorithm has natural defects of weak robustness against desynchronization attacks and have relatively low embedding capacity.

1.3 Objectives

The objective of this thesis is to develop an audio watermarking system that has high embedding capacity and has a good robustness against common signal processing attacks and desynchronization attacks, simultaneously.

1.4 Scope of Works

The scope of works in this thesis are namely:

1. In this thesis, the watermarking is focused only to audio data.
2. The inserted watermark into the audio signal is a 41×41 pixels image in *.bmp format.
3. The experimental simulations are performed using MATLAB R2019a in Windows 11 operating system.
4. The audio file format is in *.wav with 44.1 kHz sampling rate, mono channel, 16 bits/sample, and 10 seconds audio duration.
5. The audio watermarking performance are analyzed within three aspects: robustness, imperceptibility, and capacity.
6. Several attacks are performed for performances evaluation, namely: desynchronization attacks and common audio signal processing attacks.
7. The audio watermarking perceptual quality are tested objectively and subjectively.

1.5 Research Method

The research method in the process of completing this thesis consists of several stages that is:

1. Identifying research problem

   The research problem was identified by a literature study. Literature is studied from journals, articles, books, and conferences article.

2. Designing problem solving model

   Problem-solving scheme for the audio watermarking technique is designed by combining two data from the audio host and watermark imagery. Then, the audio watermarking system will produce four watermarked audio signals that are robust to various signal attacks and imperceptible.

3. Problem solving models and research validation

   Problem solving technique is tested objectively and subjectively to determine the watermarked audio quality.

4. Collecting and analyzing data

   The data is retrieved from experimental results and analyzed to evaluate the performance of designed audio watermarking system.

5. Drawing conclusions

   The conclusions is seen based on the results of data analysis and the achievement of objectives.

1.6 Bachelor's Thesis Organization

The rest of this thesis is organized as follows:

- Chapter 2 BASIC CONCEPT
  Chapter 2 encompass basic explanation of audio watermarking, the methods used in the designed system, and type of attacks tested to the system.

- Chapter 3 SYSTEM MODEL AND DESIGN
  Chapter 3 describes the work flow and system design in the software for audio watermarking.

- Chapter 4 PERFORMANCE ANALYSIS
  This chapter presents the system analysis on the results obtained from the design, testing and simulation stages of the system.

- Chapter 5 CONCLUSIONS
  Chapter 5 comprises the conclusions and suggestions of this thesis.