Available online at www.sciencedirect.com

ScienceDirect

Procedia Computer Science 85 (2016) 418 – 424

Procedia Computer Science

International Conference on Computational Modeling and Security (CMS 2016)

Ideal Sampling Rate to reduce distortion in Audio Steganography

Ramya Devi R[a] *, D Pugazhenthi[b]

Quaid-E-Millath Government College for Women (A), Chennai -600002, Tamil Nadu, India

Abstract

This report presents a method to embed and extract digital data in an audio file using LSB embedding technique. The intended use of this system regards for reducing noise level that is added during audio steganography process. The aim was to study the effects of sampling rate on audio cover during the process of digitization.

The motivation came from our human auditory system (HAS) which is sensitive towards distortions added during steganography process and making the process suspicious. We first show how to embed the text data into audio file by LSB embedding techniques by applying cipher key. Basic audio sampling is discussed. We further segment audio by using Nyquist–Shannon sampling theorem. The report concludes that our system successfully preforms audio steganography with decreased noise level in terms of Signal to Noise Ratio (SNR) when sampled at the rate proposed in Nyquist–Shannon techniques.

Keywords: Audio Steganography; Sampling-Rate; SNR; HAS; LSB; Nyquist–Shannon techniques;

1. Introduction

Steganography is the art of hiding data over a medium and make it undetectable. In this report basic steganography processes like encryption of text data into audio signal (.wav) and decryption was carried out using Least Significant Bit (LSB) encoding technique along with cipher key authentication process. A clear understanding of the noise that gets added in the audio signal when the text is embedded using steganography process is discussed. This paper presents the effect of SNR value on different sampling rate.

In this work, uses LSB encoding technique which is simple and easy way of embedding and better imperceptibility. At the receiving end the embedded data is extracted without knowledge of the original audio. Proposed system is implemented and tested for different sampling sizes and performance is evaluated using SNR values comparison. Our main contribution is to have a perfect sampling rate for different audio type.

* Corresponding author. Tel.: +0-000-000-0000 ; fax: +0-000-000-0000 .
E-mail address: :ramyadeviresearch@gmail.com

Peer-review under responsibility of the Organizing Committee of CMS 2016
doi:10.1016/j.procs.2016.05.185

In the following, section 2 summarizes related work on LSB based audio steganography for noise detection and reduction processes. Basic nature of sound and its digital representation of sampling are discussed in section 3. Section 4 provides generation of audio samples and Nyquist–Shannon based sampling techniques and embedding scheme. Results of proposed scheme are discussed in section 5. And the contribution of the work is concluded in section 6.

2. Related Work

In literature many audio steganography scheme are provided. Here, some of the similar works are discussed (Nedeljko Cvejic T. S., 2002) (Gopalan, 2003) (Nedeljko Cvejic, Tapio Seppänen, 2004) (Mazdak Zamani, 2009) (Muhammad Asad, 2011). These citations are primarily based on LSB embedding scheme. Substitution methods either use time domain or temporal domain of samples generated over sampling techniques. In the above works the sampling rate is mentioned as the major criteria that affect the embedding process.

The author in (Nedeljko Cvejic T. S., 2002) do modification to standard LSB algorithm and was given that embed data at the rate of 4 bits per sample that improves capacity of data hiding by 33%. SNR value of 3-bit embedding and 4-bit embedding takes approximately the similar value.

Comprehensive overview was given by author (Gopalan, 2003) on Audio steganography and bit modification was discussed. Here, the message which is embedded is kept in the compressed form and the special key is used in the embedding process. The results are based on the selected noisy cover audio. The results were compared by the cover audio spectrogram and bit 10 of steganographed audio in each frame was embedded with data.

(Muhammad Asad, 2011) As well as (Nedeljko Cvejic, Tapio Seppänen, 2004) used an embedding process which decreases the distortion in the output audio in different stages of samples using LSB scheme. Especially (Muhammad Asad, 2011) generated a random position of embedding the data which is generated using the cipher key. And also he used the sampling rate of 8000 each of 8-bit samples. They both tested the perceptual quality of data.

(Mazdak Zamani, 2009) Deal with embedding secrete message bit into various types of file including digital images, audio and video. The proposed system was tested with imperceptibility (SNR), payload and robustness. A correlation between the SNR and sample rate was given.

3. Basic Theory

Before we can start to manipulate sound we need to study how the character of sound is and its digitalization process. In this section audio fundamentals and its digital representation are discussed.

3.1. Nature and Characteristics of a Sound wave

At a fundamental physical level, sound is simply a disturbance of molecules within a substance. (David Howard, 2013).The three main characterizes of sound waves which is used to describe or reconstruct a signal are:

1) Amplitude: A measure of the degree of change in atmospheric pressure caused by sound waves. This amplitude is directly related to loudness. (Rumsey & McCormick, 2006),

2) Frequency (f): The frequency of sound is the rate of cycle formed per second and is represented by unit Hertz Hz. The frequency is directly proportional to pitch of a sound.

3) Wavelength: The wavelength (λ) is distance between two adjacent peaks of crest or trough. It depend on velocity of sound (v) represented by unit (m/s) and its frequency (f). We can therefore define in equation (1) the wavelength as;

$$\lambda = \frac{v}{f} \tag{1}$$