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Ideal Sampling Rate to reduce distortion in Audio Steganography

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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.

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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;

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

3.2. Digital Representation and Audio Sampling

In this section, conversion of analog sound into a series of bits is discussed. Device used for this process is Analogue to Digital Converter (ADC). The ADC captures snapshots (samples) of the electric voltage of thousands of times per second in audio signals using devices like microphone. It is represented in terms of bit series that can used for further processing.

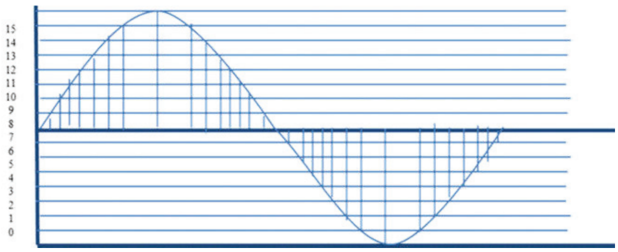

Fig.1 converting a sound from Analogue to Digital

Figure 1 shows audio wave which has been sampled using a 4 bit factors. Audio sampling affects the quality of digital waveform in terms of sample rate and sample size. The rate at which the samples are captured and played back is termed as sample rate measured in Hertz (Hz). Ex, audio CD uses a sample rate of 44,100 Hz because human hearing range is roughly between 20- 20,000 Hz (Olson, 1967). Sample size (Sample format) is the number of digits in each sample. Ex, an audio CD has a precision of 16 bits allowing 65536 levels in total.

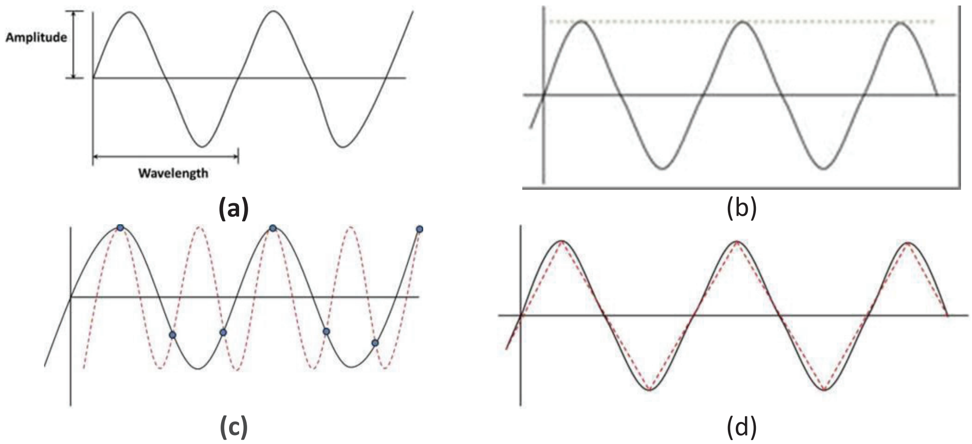

Fig.2 (a) A Sine Wave; (b) Sample at 1 time per cycle; (c) Sampling at 1.5 times per cycle; (d) Sampling at 2 times per cycle

A detailed illustration is given on how sampling of waveforms for made at different sampling rate. Analog wave is represented as in Figure.2 (a). When the sample rate is one time per cycle, the output signal takes a straight line. When the sampling rate is 1.5 times per cycle there is a chance of loss of data. Whereas by analysing Figure.2 (d) we can conclude that the sampling rate given by Nyquist forms a perfect sampling rate.

3.3. Audio quality

Sound can be classified as Stereophonic sound (Stereo) and Monophonic sound (mono). Stereo are with more number of independent audio channels which creates the impression of hearing from various directions. And Mono has single channel.

4. System Design

4. 1. Generating Audio Samples

The sampling rate is the number of times an Analog signal is measured per second. Higher sampling rate results higher sound quality because the analog waveform is approximated closely. Sample rate choosing depends on the application, capability of audio interfaces and the destination of audio.

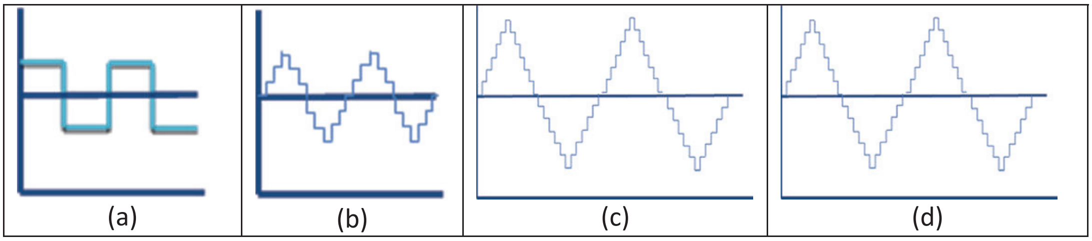

Fig.3- sampling rate variations of ADC

In figure (3) a visual classification is given for single bit (a), two bit sample rate (b), and four bit sample rate (c) and in 16-bit sampling rate in (d). For improved signal quality applied as a cover signal the sampling rate should take at least 16-bit.

4. 2. Selection of audio samples for embedding data

When text message is embedded within the audio file, the audio files and text file are converted into bit streams. The first bit of the text will be inserted into the first sample of least significant bit of cover audio. The process will run until the entire text message is inserted into audio file. When the text message is greater than the audio file, error message is displayed. An interactive based program analysis was developed using Matlab tool functions like wavread (), wavwrite (), wavrecord (), wavplay (), wavinfo ().

4.3. Embedding Scheme

LSB embedding scheme is used to hide of text data in audio files. Encoding and Decoding channels are discussed. Here, bit level manipulation is performed to encode the message. The text data used for embedding process is 1.35KB size. The steps for embedding process is described in figure (4) are as follow: 1) In Encoding channel, audio file is received, 2) Convert it into binary pattern by samplings. 3) Receives the text message. 4) Convert each character into binary pattern. 5) Apply a Cipher key. 6) In decoding Channel, the message is extracted by receiving a Signal of Wave format (.wav). 7) Identify weather a given signal is Steganography Signal. 8) Leave the un-encrypted Signal to play ordinarily. 9) Else the message retrieval process started by using the cipher key. 10) The binary values are converted back into ASCII values and reconstructed back to the original message.

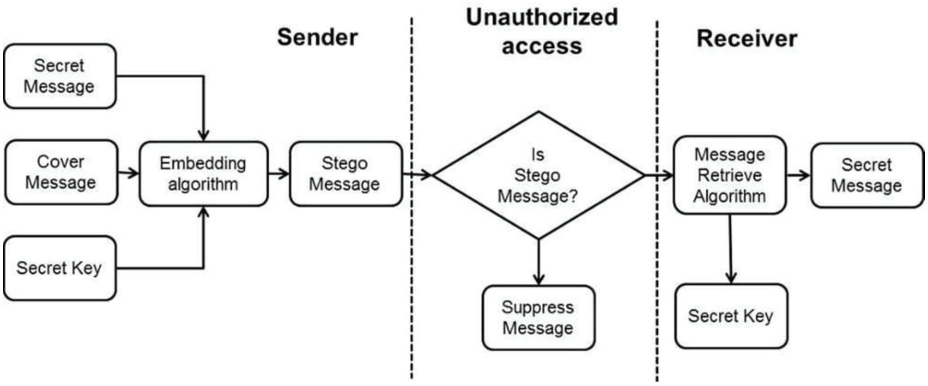

Fig.4-Embedding data in audio samples

4. 4. Evaluation of the scheme

Evaluation of the different sampling techniques is carried out using Signal to noise Ratio (SNR). SNR is the Ratio of the powers of signal to the noise (distortion). It is a measure of quality of the output signal after embedding process. It is measured in decibels (dB).Higher the SNR value cleaner the output signals with less added background noise. With

$$SNR = 10\,\log_{10}\frac{\sum_n A_n^2}{\sum_n (A_n - A'_n)^2} \tag{2}$$

Where; A_n is the original audio signal, $A_n{}'$ is the distorted audio signal

5. Results and discussions

The implementation of proposed system in Section 4 takes three steps:
1) Select cover audio and sample it at 44.1 KHz, 22.05 KHz, 11.05 KHz and Nyquist-sampling rate (Nq)
2) Sampled audio is treated into audio steganography embedding process using LSB
3) SNR value is calculated between the cover audio and the sampled audio after steganography process.
4) Execution Time of audio steganography process is calculated for both mono and stereo audios that are sampled at range of sampling rate. Audio sampled at Nyquist-sampling rate was also included in the test specially. Results are tabulated in table 1.

In order to test performance of the designed system a graph is plotted between the sampling range and corresponding SNR values in figure.5

Table I-Audio data set used for testing

File Name	Bits per sample	Cover audio Size	Audio Quality	Size after sampling(KB)				SNR(dB)			
				11.025	22.05	44.2	Nq	11.025	22.05	44.2	Nq
A	8	866KB	Mono	193	341	530	584	25.8	28.6	50.0	57.8
B	16	1.73MB	Mono	683	730	1095	1167	32.8	39.4	68.1	73.6
C	8	1.34MB	Stereo	230	634	1198	1260	25.0	31.8	44.7	49.8
D	16	1.73MB	Stereo	459	1270	2079	2529	37.0	42.9	67.9	84.6

In figure (5), comparisons of variations are studied in terms of SNR (Muhammad Asad, 2011) when sampling rate is changed. The Mono audios shows less variations compared to stereo audios. In the Nyquist sampling rate the SNR value is more so it forms a best sampling rate of 84.6dB.

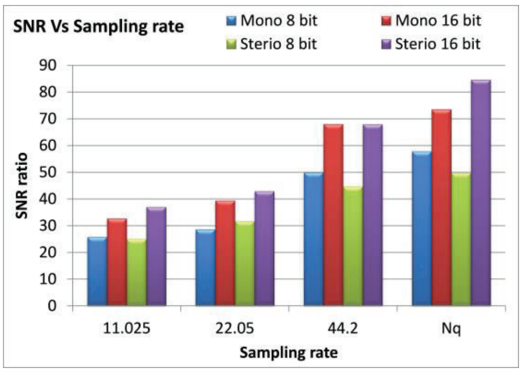

Fig (5) SNR vs. Sampling rate

For both Mono and Stereo audio, for all audio signals generated at different sampling rate was given as input audio to the proposed system and the Time taken for execution was noted as given in table (2). The execution time taken by sampling at Nyquist rate (David Howard, 2013) is nominal compared with other sampling rates.

Table II- Time of execution

Audio Quality	Sampling rate	Time Spent(s)
Mono	11.025	0.5427
	22.05	0.5201
	44.2	0.4875
	Nq	0.4734
Stereo	11.025	0.6845
	22.05	0.5436
	44.2	0.3245
	Nq	0.3435

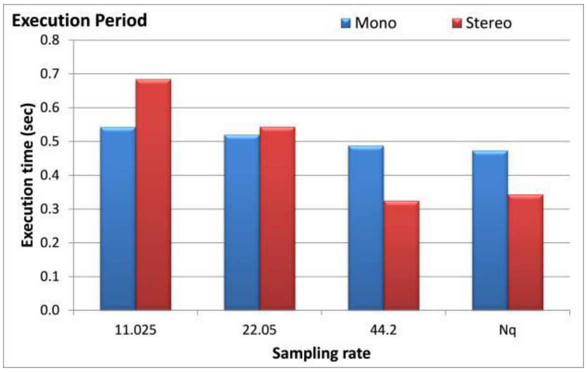

Fig (6) Execution period

6. Limitations

There was drastic increase in SNR value when additive noise like Gaussian noise was added for proposed system. The proposed system is less Robust when compared to other steganography scheme because both Nyquist sampling rate and the LSB method can be easily applied in the steganalysis process. However, when a threshold is added as a key the robustness could get a better. The scheme has a constraint that, when an audio file is treated with lossy compression the hidden message is manipulated as there is a change in whole structure of a file.

7. Conclusion

In this work, embedding of digital data in an audio file is discussed using LSB scheme. An ideal sampling rate using Nyquist–Shannon techniques is justified so that distortion is not added even before the embedding process is started. The effects of different sampling rate on SNR value was studied w.r.t sampled audio file size. Based on the test results, the performance of audio steganography was analysed. The range of sampling rate was tested for SNR value and the audio sampled using Nyquist–Shannon techniques gives the best value for both sampling and executions time taken on the audio steganography process.

References

Asad, M., Telecommun. Eng. Dept., U. o., Gilani, J., & Khalid, A. (2011). An enhanced least significant bit modification technique for audio steganography. *Computer Networks and Information Technology (ICCNIT), 2011 International Conference* , 143 - 147.

Cvejic, N., Inf. Process. Lab., O. U., & Seppanen, T. (2004). Reduced distortion bit-modification for LSB audio steganography. *3*, 2318 - 2321.

Cvejic, N., MediaTeam Oulu Group, O. U., & Seppanen, T. (2004). Increasing robustness of LSB audio steganography using a novel embedding method. *Information Technology: Coding and Computing*, 533 - 537.

David Howard, J. A. (2013). *Acoustics and Psychoacoustics.* CRC Press.

Gopalan, K. (2003). AUDIO STEGANOGRAPHY USING BIT MODIFICATION. *IEEE*, 629-632.

Guoshen Yu, J.-J. S. (2009). Audio classification from time-frequency texture. *International Conference on Acoustics, Speech and Signal Processing* (pp. 1677-1680). ACM Digital Library.

Heena Malik, S. S. (2013). Designing and Evaluation of Performance of a Spread Spectrum Technique for Audio Steganography. *International Journal of Advanced Research in Computer Science and Software Engineering, 3*(8), 374-378.

Mazdak Zamani, A. B. (2009). A secure audio steganography approach. *Internet Technology and Secured Transactions*, 9-12.

MAZDAK ZAMANI, A. B. (n.d.). Correlation between PSNR and Bit per Sample Rate in Audio Steganography. *Recent Researches in Communications, Signals and Information Technology*.

Md. Shafakhatullah Khan, M. S. (2011). Steganography, Optimized Method for Concealing Data using Audio. *International Journal of Computer Applications, 33*(4), 25-30.

Mitchell D Swansona, ,. B. (1998). Robust audio watermarking using perceptual masking. *Signal Processing, 66*(3), 337-355.

Ms. Lalita, M. S. (2014). Designing of Algorithm for Hiding of Multimedia Information in to Colored Image Using Spread Spectrum Method. *International Journal of Advanced Research in Computer Science and Software Engineering, 4*(5), 275-281.

Muhammad Asad, J. G. (2011). enhanced least significant bit modification technique for audio steganography. *Computer Networks and Information Technology*, 143 - 147.

Nedeljko Cvejic, T. S. (2002). Increasing the capacity of LSB-based audio steganography. *IEEE*, 336-338.

Nedeljko Cvejic, T. S. (2004). Increasing Robustness of LSB Audio Steganography Using a Novel Embedding Method. *IEEE.*

Nedeljko Cvejic, T. S. (2004). Reduced Disortion bit modification for LSB audio steganography. 2318-2321.

Nugraha, R. M. (2011). Implementation of Direct Sequence Spread Spectrum Steganography on Audio Data. *International Conference on Electrical Engineering and Informatics*, 17-19.

Prasanthi, O. (2012). *New Approach for Steganography.* Lap Lambert Academic Publishing GmbH KG.

Ravali.S.V.K, N. ,. (2014). Implementation of Blowfish Algorithm for Efficient Data Hiding in Audio. *International Journal of Computer Science and Information Technologies, 5*(1), 748-750.

Rupanshi, P. ,. (2014). Audio Steganography by Direct Sequence. *International Journal of Computer Trends and Technology (IJCTT), 13*(2).

Tanmay Bhattacharya, N. D. (2012). A Novel Session Based Dual Steganographic Technique Using DWT and Spread Spectrum. *International Journal of Modern Engineering Research, 1*(1), 157-161.

Zamani, M., Univ. Teknol. Malaysia, S. M., Manaf, A., Ahmad, R., & Jaryani, F. (2009). A secure audio steganography approach. *Internet Technology and Secured Transactions*, 1-6.