Investigation of Steganography Methods in Audio Standard Coders: LPC, CELP, MELP

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ABSTRACT:
Information security is currently one of the most important issues that have been considered by many researchers. The purpose of Steganography is to hide hidden messages in a non-secret file. In general, information Steganography is a method of secure communication that aims to hide data so that no data appears to be hidden. The principle of Steganography is to use spaces from the information carrier that do not harm the identity of the carrier. By Steganography information from unauthorized recipients, the information is hidden and hidden inside it without harming the signals. This information may be transmitted around us and wherever any file is sent. It may contain very dangerous content for the security of the space in which we live. Audio signals are very used for steganography because Digital audio signals have higher redundancy and higher data transfer speeds, making them suitable for use as a cover. The LPC10, CELP, and MELP audio standards are widely used in audio and speech processing and are powerful high-quality speech coding methods that provide highly accurate estimates of audio parameters and are widely used in communications. Therefore, since considering that these audio standards are used in commercial and military telecommunication systems, they can be considered a suitable platform for sending the following message of audio content. We try to carefully examine these standards and the audio Steganography done in these standards.

KEYWORDS: Steganography, Audio Signal, Coding, LSB.

1. INTRODUCTION

Information hiding methods are used to invisibly hide a message within other characters. Audio Steganography has a lot of potential for encryption because there is a lot of bandwidth to transmit. LPC, CELP and MELP audio Coders standards, which are official NATO standards that are widely used in voice and speech processing. Due to the high bandwidth and the existence of many capacities to hide information, Steganography is very common in these standards, which can provide a very suitable space for use as a cover media for the concealer.

. LPC, CELP and MELP audio Coders standards are widely used in voice and speech processing and are powerful high-quality speech coding methods that provide very accurate estimates of audio parameters and are widely used in communication. Due to the fact that these audio encoders are used in commercial and military telecommunication systems, they can be considered as a suitable platform for sending the following message of audio content.

2. LINEAR PREDICTIVE

The method of operation of voice coders based on the prediction line is as shown in Fig. 1.

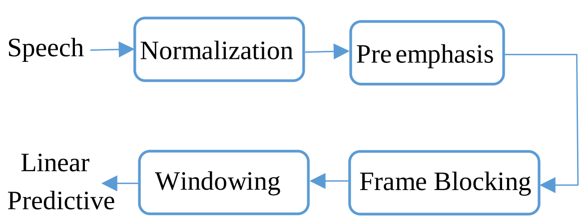

Fig. 1. General structure of Linear Predictive coders.

According to Fig. 1, the signal is entered into the

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system and:

- Normalization normalizes it to (+1 to -1).
- Pre emphasis removes destructive zeros and poles by inserting the $\frac{AZ}{AZ/\gamma}$filter.
- Frame Blocking Specifies the Stationary value of each frame.
- Windowing inserts a Haming or Black man window to reduce signal breaks in the spectrum between frames.

The main purpose of Voice Coders (unlike Wavelet Coders, which aim to obtain a speech signal as similar as possible to the original signal with the maximum signal-to-noise ratio) is to make artificial speech that has the same quality as the original signal. The general structure of these encoders is based on Fig. 2:

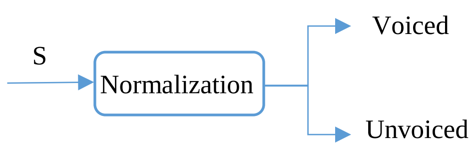

Fig. 2. Separation of voice and unvoiced in audio encoder

According to Fig. 2, the LPC coefficients are extracted from the input speech signal and the sound is divided into audio and non-audio parts.

In general, speech signal has two types of correlations:

Short-term correlation: The correlation between adjacent samples of a speech signal.

Long-term correlation: The correlation between specimens that are one pitch apart.

The audio chamber filter represents the short-term correlation of the speech signal, in other words, this filter minimizes the estimation error between adjacent samples. If we denote the nth estimate with $\tilde{S}(n)$then we have:

$$\tilde{S}(n) = \sum_{k=1}^{p} a_k - s(n-k) \tag{1}$$

And the estimation error is calculated as follows:

$$e(n) - \tilde{S}(n) = s(n) - \sum_{k=1}^{p} a_k - s(n-k) \tag{2}$$

Fig. 2 shows the block diagram of the speech signal encoder estimator.

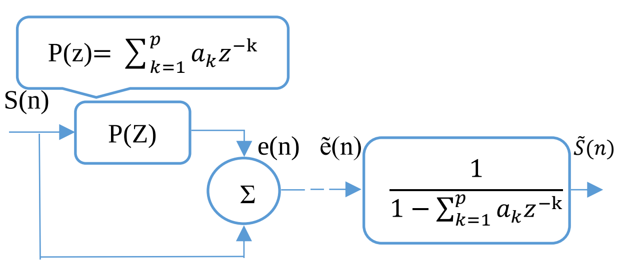

Fig. 3. General structure of the speech signal encoder estimator block diagram.

It should be noted that:

$$\frac{\partial E_n}{\partial a_k} = 0 \tag{9}$$

- The variance of e (n) is much less than the variance of s (n).
- In the receiver, the desired filter is made of IIR.

So the coefficients a_k_k must be chosen so that the filter is stable. So we will have:

$$S(w) = \mathrm{E(w)} * \frac{1}{1-\sum_{k=1}^{p} a_k \mathrm{e}^{-\mathrm{jkw}}} \tag{3}$$

According to Equation 6, the value of E (w) indicates the airflow filter passing through the larynx $1-\sum_{k=1}^{p} a_k \mathrm{e}^{-\mathrm{jkw}}$ indicates the sound chamber filter. We know that speech signal coverage is equal to:

$$H(z) = \frac{1}{1-\sum_{k=1}^{p} a_k z^{-\mathrm{k}}} \tag{4}$$

Indicates that the IIR audio chamber filter is fully polar and stable.

To find the above filter, we need to get the coefficients a_k .

Optimal a_k coefficients are calculated to minimize the amount of error energy $\sum \mathrm{e}^2$ (n). Because the speech signal is a non-static process, the a_k coefficients are variable with time, so the audio encoder filter is a time-varying filter.

Speech signal has the property of being assumed to be in short periods of time. Therefore, the coefficients a_k are calculated not for the whole speech signal but for each individual frame.

If we assume we have a frame with 2M + 1 point. [n-M, n + M] and in this case the estimation error for each frame is calculated as follows.

$$e_n(m) = S_n(m) - \sum_{k=1}^{p} a_k S_n(m-K) \tag{5}$$

In this relation e(n) is the error of estimating the symbol m for the nth frame and we have:

$$E_n(m) = \sum \mathrm{e}^2 \text{ (n} \tag{6}$$

To calculate a_k must be the value E_n minimum. So we will have:

$$E_n = \sum_{m=n-M}^{n+M} [S_n(k) - \sum_{k=1}^{p} a_k S_n(m-K)]^2 \tag{7}$$

Since:

$$\Rightarrow \sum 2\left(S_n(m) - \sum_{k=1}^{p} a_k S_n(m-K) * \left(S_n(m-i)\right)\right) = 0$$

$$\Rightarrow f(x) =$$

$$\begin{cases} \Sigma(S_n(m) S_n(m-i) = \Sigma_k a_k \, \Sigma_m S_n(m-K)\, S_n(m-i) \\ \qquad\qquad\qquad x < i \\ 1 \leq i \leq p \quad , \; 1 \leq k \leq p \end{cases} \tag{8}$$

And by defining the following two parameters:

$$\varphi_n(\mathrm{i,k}) \triangleq \sum S_n(m-K) S_n(m-i) \tag{10}$$

Which we conclude from the relations 8 and 10:

$$*\,{}_9\# \Rightarrow \sum_{k=1}^{p} a_k \varphi_n(\mathrm{i,k}) = \varphi_n(\mathrm{i,0}) 1 \leq i \leq p$$

$$a_1 \varphi_n(1{,}1) + a_n \varphi_n(1{,}2) + \cdots a_p \varphi_n(1,\mathrm{p}) = \varphi_n(1{,}0)$$

$$p_1 \varphi_n(1{,}1) + a_2 \varphi_n(\mathrm{p},2) + \cdots a_n \varphi_n(\mathrm{p,p}) = \varphi_n(\mathrm{p},0)$$

$$\begin{bmatrix} \varphi_n(1{,}1) & \cdots & \varphi_n(1,\mathrm{p}) \\ \vdots & \ddots & \vdots \\ \varphi_n(\mathrm{p},1) & \cdots & \varphi_n(\mathrm{p,p}) \end{bmatrix} \underbrace{\begin{bmatrix} a_1 \\ \cdots \\ a_p \end{bmatrix}} = \begin{bmatrix} \varphi_n(1{,}0) \\ \cdots \\ \varphi_n(\mathrm{p},0) \end{bmatrix}$$

A **B** **C**

At the end we have:

$$A = B^{-1} C \tag{11}$$

2.1 LPC

The LPC method was introduced in 1966 and was completed in 1978. LPC is one of the most common methods of encoding audio, which represents a compressed digital signal using a linear estimation model. This method is widely used in audio and speech processing and is one of the most powerful methods of high quality audio encoder analysis that provides very accurate estimates of audio parameters. Fig. 4 shows the block diagram of the LPC encoder. [2]

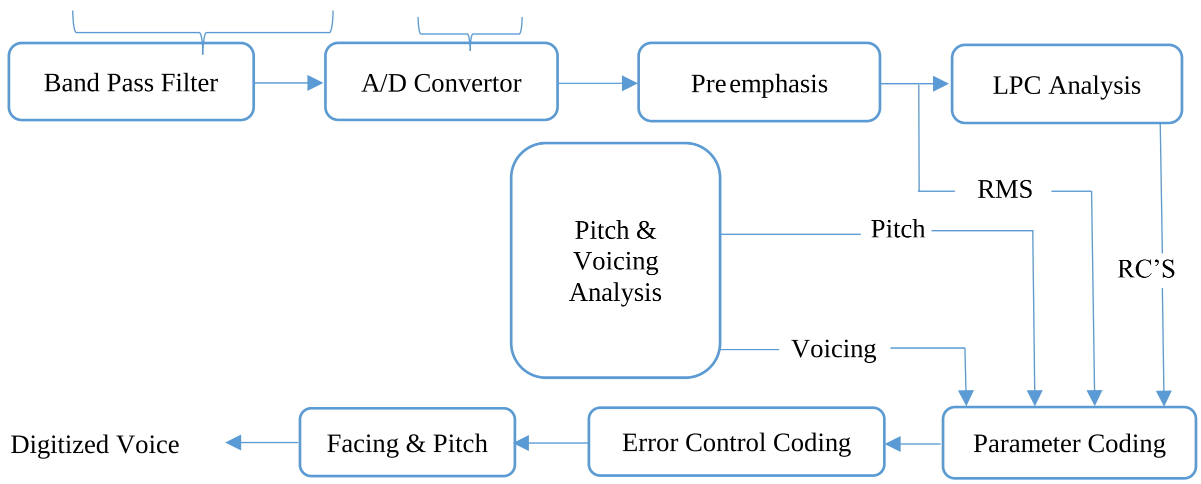

Fig. 4. Block diagram of LPC. [2]

Which is filtered according to Fig. 4 before converting analog to digital. After converting analog to digital, it is emphasized that at this stage, by inserting the $\frac{AZ}{AZ/\gamma}$ filter, zero and destructive poles are removed, and then linear estimation coefficient analysis is used for Determine Reflection Coefficients. It becomes. The Root Mean Square domain is also calculated. In addition, after converting analog to digital, pitch and audio analysis is performed to determine whether to model the LPC encoded frame as audio or non-audio. If there is sound, the pitch is set, and if there is no sound, it is determined whether the frame should be considered without sound or in transition. Next, the RMS, Pitch, and audio domains are encoded. Error control coding is added if necessary, followed by a synchronization bit. And a 54-bit frame is produced.

2.2 CELP

In 1985, the CELP method was introduced. This method is a linear speech coding programming algorithm. This method is one of the most widely used speech coding algorithms due to its high quality. It is also used in the MPEG-4 audio speech encoder. Fig. 4 shows the block diagram of the CELP encoder. [3]

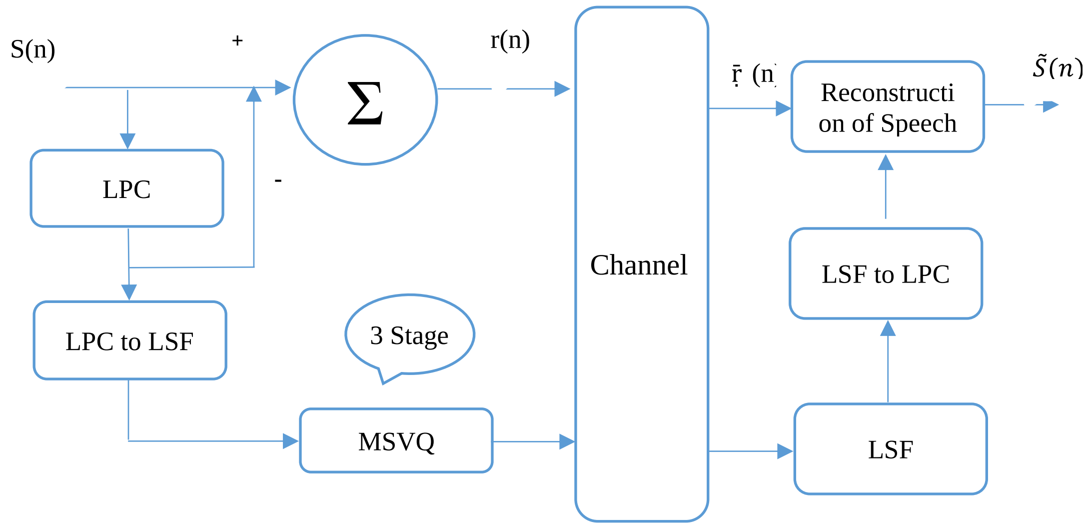

Fig. 5. CELP encoder block diagram. [3]

The signal S (n) is the input signal as shown in Fig. 5lo:

• First, LPC coefficients are extracted from this signal. (Here the extracted coefficients are compared with the original sound and the residual value of r (n) is obtained by comparing it with the original signal.)

• The LPC then converts to LSF. (The advantage of LSF over LPC is that firstly, when we do quantization, the correlation does not disappear and secondly, there is no instability in quantization.)

• The signal then enters the three-level MSVQ quantizer.

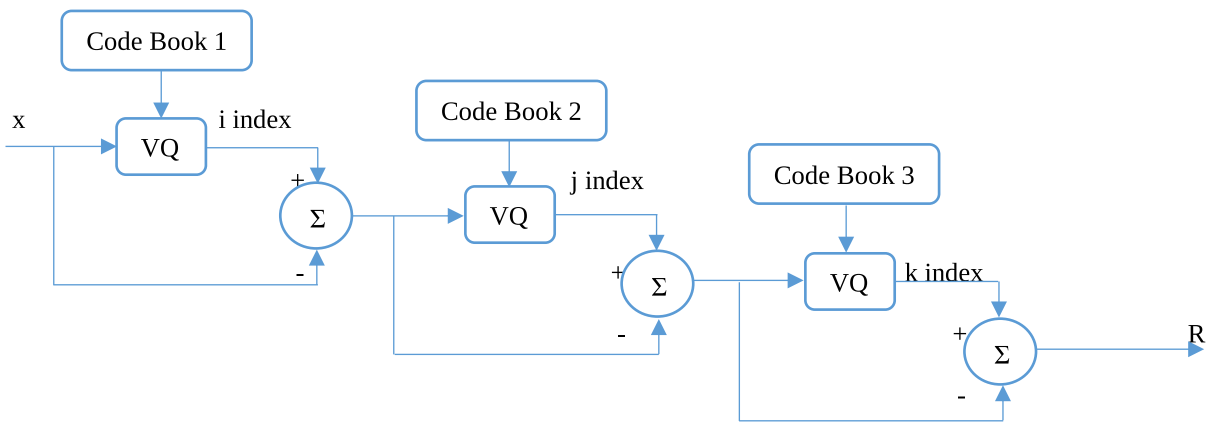

Fig. 6. MSVQ three-level quantizer

Consecutive codebooks (such as MSVQ) are used to solve the problem of increasing the binary code rate of bookkeeping. At the input, LSF coefficients such as x are quantized using the code book 1 and extracted by comparing to the original value of index i, the output is quantized and extracted by comparing to the original value of index j, and the output is quantized again by Comparison with the original value of index k is extracted. Finally, indices i, j and k are added together and the quantized signal is extracted. The value of R remains, which is so small that it does not need to be made and is similar to noise.

• The signal leaves the transmitter and enters the channel.

• LSF coefficients are first generated in the receiver.

• LPC coefficients are then constructed from LSF coefficients. These coefficients are compared with the remainder and a sound estimate is obtained.

• The audio signal is reconstructed.

2.3 MELP

In 1995, the MELP method was registered based on LPC. The original speech coder was standardized in 1997 and became known as the MIL-STD-3005. [4]

This encoder is a standard of the United States Department of Defense speech coder used primarily in military and satellite communications programs, secure voice transmission, and radio communications security. Its subsequent standardization and development was produced and supported by the US National Security Agency and NATO. Fig. 6 shows the block diagram of the MELP encoder.

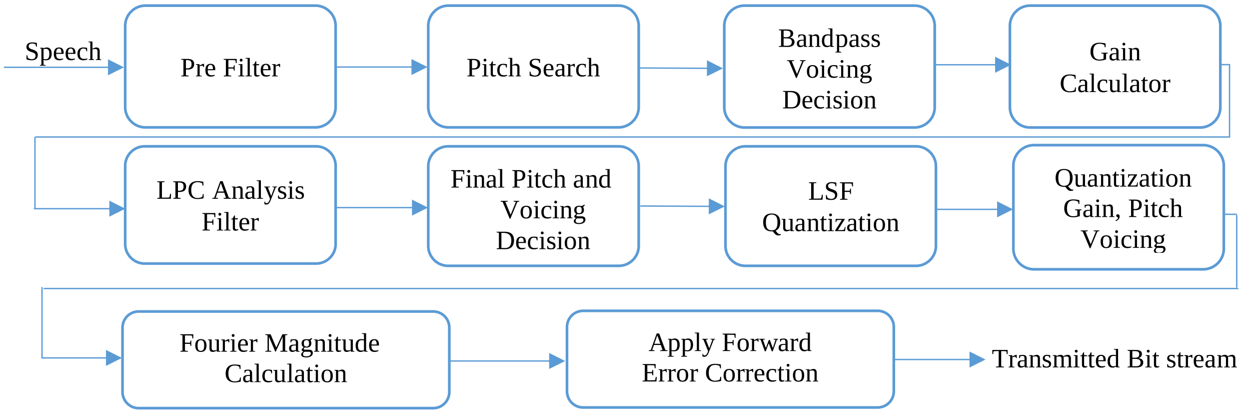

Fig. 7. MELP encoder block diagram. [4]

Fig.7 shows the signal input to the MELP encoder, which we will describe in the following:

• Pre filter: This section detects the nature of the input signal. (This is the task of detecting the speech of the input signal).

• Pitch Search: Initially performs a full search and Pitch detects the sound.

• Band pass Voicing Decision In this part, according to Fig. 8, the speech spectrum is filtered or sampled into 4 parts with different bandwidth in a non-uniform manner.

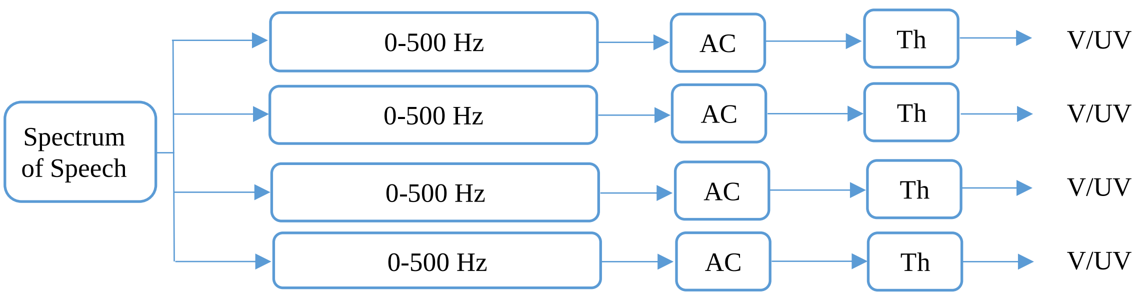

Fig. 8. Examining the spectrum of speech.

As shown in Fig. 8, the first and second sections are sampled at 500 Hz, the third section at 1 kHz, and the fourth section at 2 kHz. The reason for this difference is that the higher the frequency, the lower the value, and the lower the frequency, the higher the value. An Auto Correlation is then taken from them and compared with a certain threshold level, which is recognized as sound if it is higher than that threshold and as non-sound if it is lower than that threshold.

• Gain Calculator: In each interval, Gain calculates and obtains power energy.

• Filter Analysis LPC: Performs LPC analysis and extracts LPC coefficients.

• Final Pitch and voicing Decision: Detects sound or non-sound by obtaining Pitch

LSF Quantization: Calculates LSF from the obtained LPC coefficients and then quantifies it.

• Pitch Search Quantize (Gain, pitch, Voicing, Jitter) Quantifies the obtained values of gain, pitch, sound and jitter. (In some places we have random sounds (such as a lot of noise in the battlefield) that can be said to be neither sound nor non-sound. In this part these special sounds are recognized).

• Fourier Magnitude Calculation: In each subband, the amplitude for the spectrum is determined (how to select this subband is that first a gain is calculated for the whole spectrum, then this gain is subtracted from the whole spectrum, then in each subband, the range is set for the spectrum).

• Apply Forward Error Correction: Channel coding is used to deal with the error.

And at the end the bit string will be sent. This encoder has a very low rate of 2400 bits per second. Of course, it should be noted that the main speech signal files have a bit rate of 128 kbps, which MELP converts to 2.4 kbps. In this method, the frequency band is divided into 4 bands using a filter bank.

3. STEGANOGRAPHY

Steganography is one of the parts of information. And it's the practice of concealing a message within another message or a physical object. In computing/electronic contexts, a computer file, message, image, or video is concealed within another file, message, image, or video.

3.1 PERCEPTUAL CRITERIA OF STEGANOGRAPHY

In order to rationally evaluate the performance of various Steganography methods, it is necessary to determine some acceptable criteria by the majority. The three most common requirements are security, capacity and transparency; Criteria for the performance of Steganography methods are as follows.

Capacity is the maximum amount that can be stored in a carrier without detection.

Resistance to intentional and unintentional attacks is said to indicate the extent to which the extraction algorithm is able to recover the original signal from the received signal after the attack.

Transparency depends on the human perceptual system, which is the protection against perceptual change that occurs as a result of the inclusion of information in the host.

3.2 EVALUATE CRITERIA OF STEGANOGRAPHY

Equation 12 indicates the amount of noise added to the Steganography content by embedding the information in the original content, which is the s carrier signal and sw the post- Steganography signal.[5]

$$SNR = 10\log\frac{s^2}{(s-sw)} \quad (12$$

Equation 13 is a criterion for calculating the mean squared error obtained by subtracting the value of the original content from the value after Steganography n is the number of data in a set y is the original data and 'y is the data after Steganography.

$$MSE = \frac{1}{n}\sum_{i=1}^{n}(Y(i) - Y'(i)^2 \quad (13$$

The relation is 14 criteria for quality assessment, which is obtained by comparing the files of the folder and the treasure and using the MSE criterion. In this relation, Maxi is the maximum possible value of the original data.

$$PSNR = 10\,log(\frac{{MAX_i}^2}{MSE}) \quad (14$$

Equation 15 refers to the error resulting from the embedding of information and as a percentage of incorrect bits extracted in the cover sound.

$$BER = ((\sum_{i=1}^{l}(M(i) - M'(i))\hat{}2)/l\,) * 100 \quad (15$$

4. LSB

The Audio Steganography LSB-based works by moving hidden message bits with low-value bits. [6]

Its benefits include:

- The simplest technique
- High capacity;

But its disadvantages can be:

- Low resistance.

5. ECHO HIDING

Acoustic steganography based on echo hiding embeds information in the main signal by inserting a short signal into the carrier discrete signal. [7]

The advantage of this method is:

- High data transfer rate.

But its disadvantages are:

- Limited signal size.
- Low capacity
- Low resistance.

6. HIDING IN SILENCE INSERTION

The Audio Steganography based Hiding in Silence Insertion method works by placing the message in silent intervals. [9]

The advantage of this method is:

- High Transparency.

But its disadvantages are:

- Low capacity.
- Sensitivity to compression.
- Incorrect extraction if changed during the silent interval.

7. DISCRETE WAVELET TRANSFORM DOMAIN

Audio Steganography method based on Discrete Wavelet Transform Domain based on moving hidden message bits with low value bits of discrete wavelet coefficients. [10]

Its benefits include:

- High capacity.
- High Transparency.

But its disadvantages can be:

- A lot of waste on data retrieval.

8. DIRECT SEQUENCE SPREAD SPECTRUM

Audio Steganography method based on Direct Sequence Spread Spectrum works by broadcasting hidden information through the frequency spectrum at existing frequencies. [11]

Its benefits include:

- Simple detection of information in the receiver
- High resistance.

But its disadvantages can be:

- Add noise to the original file.
- Low capacity.

9. TONE INSERTION

Audio Steganography method based on Tone Insertion using frequency coverage and using the weakness of the human auditory perceptual system by placing a weak sound after a strong sound.

The advantage of this method is:

- High resistance.

But its disadvantages are:

- Low capacity.
- Phase distortion.

10. PHASE CODING

Audio Steganography method based on phase coding using the replacement of the main and hidden sound phase components and using the weakness of the human auditory perceptual system. [13]

Its benefits include:

- Resistance to bit-cutting attacks and low-pass filter attacks.

- MP3 compression capability.

But its disadvantages can be:

- Low resistance to software attacks noted.
- Low capacity.

Table 1. Comparison of advantages and disadvantages of different audio Steganography techniques.

Steganography method	Advantages	Disadvantages
LSB	The simplest technique. High capacity.	Low resistance.
Echo Hiding	High data transfer rate.	Limited signal size. Low capacity Low resistance.
Hiding in Silence Insertion	High Transparency.	Low capacity. Sensitivity to compression. Incorrect extraction if changed during the silent interval.
Discrete Wavelet Transform Domain	High capacity. High Transparency.	A lot of waste on data retrieval.
Discrete Sequence Spread Spectrum	Simple detection of information in the receiver High resistance.	Add noise to the original file. Low capacity.
Tone Insertion	High resistance.	Low capacity. Phase distortion.
Phase Coding	Resistance to bit-cutting attacks and low-pass filter attacks. MP3 compression capability	Low resistance to software attacks noted. Low capacity.

11. SUPPORT

This work is an excerpt from Saeed Talati's doctoral dissertation entitled "Recognition of Digital Audio Steganography in Standards of Audio Encoder's LPC10, CELP and MELP" at Imam Hossein University.

12. CONCLUSION

In this paper, we examine Steganography methods in the LPC10, CELP, and MELP audio encoder standards in the two domains of amplitude and time. As can be seen, these voice coding techniques are powerful voice coding standards that are widely used in the mobile, commercial and military industries. The results of Steganography using different methods in these voice transition standards are given in Tab1. Since transparency and capacity are the main criteria for Steganography, the user can use any Steganography method he needs to increase communication security, depending on which criterion he needs the most, because between three factors. There must be a compromise, and it is not possible for all three criteria to be high.

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