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Security system for sending voice signals hiding information using steganography using MATLAB

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Abstract*—The information security plays a key role in sending information that is made through the network; this has developed several studies of algorithms, protocols and systems used to protect information and provide secure communications and entities that communicate, so that even if the information could be intercepted, it cannot be interpreted without knowing the key with which the message has been hidden, ie ensure that information is accessible only to authorized personnel. In this paper a simple and effective method to protect audio information is exposed by various techniques and is subsequently applied to an instant messaging system consisting of text and audio encoding.

I. INTRODUCTION

In an analog signal can vary three properties: the amplitude, frequency and phase. In the audio to be an analog signal, you can modify any of these properties by applying an external signal, which will alter the behavior of the original signal, and knowing this behavior can be eliminated by aggregate for the original signal. The modulation is intended to make a representation of the signal to be transmitted. Mode representation of the modulated signal varies, and this is based on how the signal carrying the information is retrieved

Amplitude modulation (AM) is the simplest methods to understand and simpler to implement. In this method, the wave has a given frequency (carrier), which is allowing be transmitted by the channel of a desired mode. The modulating signal acts as monitoring the envelope of the carrier signal. The envelope shown in Figure 1 (a), where the modulated signal has a dotted line. As the simplest method of obtaining the original signal amplitude modulated signal, it is by detecting the signal envelope.

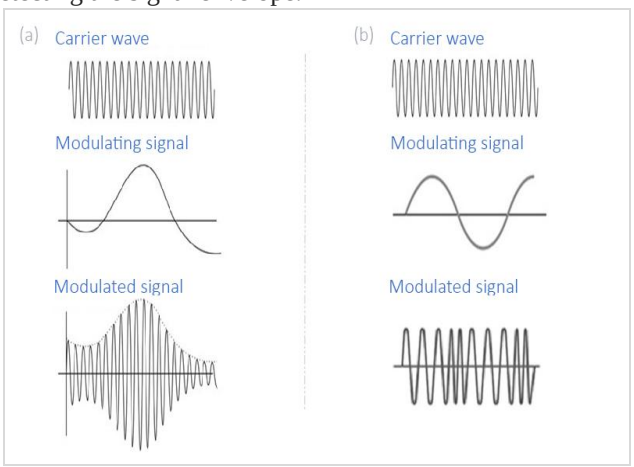

Fig 1.Modulation (a) amplitude and (b) frequency

Frequency modulation (FM) is to vary the frequency of the carrier in proportion to the frequency of the modulating wave (transmitted information) remaining constant amplitude. Unlike AM, the main consequence of the modulation frequency is higher playback quality as a result of their near immunity to interference in the signal, as is its attenuation

II. WORK DEVELOPMENT

By implementing a security system for sending auditory information, you can hide the original signal using frequency modulation, amplitude or phase. Frequency modulation being the optimum for ease and accuracy, when you want to recover the original signal. The implementation was done with Matlab to visualize the waveforms of the modulator, carrier and modulated signal. Methods of audio coding are endless, as it can be a random, sinusoidal, exponential, etc. signal so that only the sender and receiver know the pattern of this signal, ie, the type of the signal which you are coding.

A. Frequency Modulation Coding

This method is to generate vector describing the operation of a mathematical signal in this case is a cosine signal, which encrypts the vector original audio, so if the receiver file modulated audio does not possess the characteristics of this function you cannot get the information contained in the audio file.

In Figure 2 the flow diagram describing encryption processing using frequency modulation shown. an audio file in this example will be recorded from the console in Matlab is obtained. Subsequently, a vector is generated from a high frequency cosine signal; it will be multiplied by the vector of original audio, this process is called encoding, since the file generated, is imperceptible to play the original message

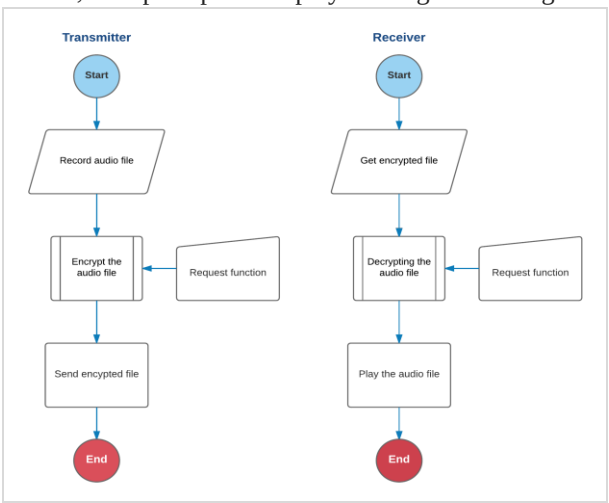

Fig 2. Flowchart encryption modulation.

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Note that the required function can be of any type; the only condition is that the function should be the same for encoding and decoding. File receiver must know the function that was codified, thus can obtain the original audio. Using the command Audio recorder can record audio from the console later Matlab save with the extension ".wav". a sampling rate of 44100 Hz and a duration of five seconds for the audio recording is proposed.

```
Fs = 44100;                                %Frecuencia de muestreo.
Tiempo = 5;                                %Duración de la grabación.

recObj = audiorecorder(Fs, 16, 1);         %Configuración de grabación.
disp('Presiona una tecla para comenzar a grabar')
pause;
disp('Comienza grabación...')
recordblocking(recObj, Tiempo);
disp('Fin de grabación.');

myRecording = getaudiodata(recObj);        %Obtención de audio.

filename = 'audioData.wav';                %Nombre del archivo a guardar.
audiowrite(filename,myRecording,Fs);       %Guardar archivo.
disp('Archivo guardado.')
```

If you want to record the audio file with an external program, audio read function allows obtaining the vector amplitude (y) and the sampling frequency (Fs) of the audio track

```
[y,Fs] = audioread(filename);              %Obtener el vector de amplitud y la Fs.
```

This vector can be viewed using the plot command. In Figure 3 the graph of the recorded audio is shown; on the right side you can play this file.

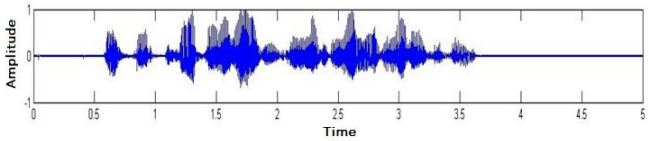

Fig 3.Graphic original audio.

The carrier signal is proposed as a cosine signal with a frequency of 20 kHz and amplitude of a unit as shown below. The linspace function creates a time vector of the same magnitude and the audio recorded with the same number of samples.

```
Amp = 1;                                   %Amplitud de la portadora.
fc = 20000;                                %Frecuencia de la portadora.
t = linspace(0, length(y)/Fs, length(y));  %Vector de tiempo.
Portador = Amp * cos(2*pi*fc*t);           %Señal portadora.
```

In Figure 4 the carrier signal generated is shown with a frequency of 20 kHz. To mount the original audio signal is needed to make a point to point multiplication (. *) Audio vector (y) and vector carrier

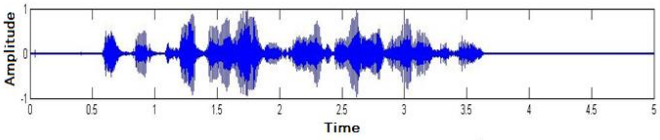

Fig 4.- Graph of the carrier signal of 20 kHz.

To perform the multiplication point to point vector and vector original amplitude of the carrier, both are required to have the same dimensions vector (220500x1 for this example); Applying the command transpose rows and columns is changed by the carrier signal vice versa declared above.

Is assigned to the variable AudioModulado the multiplication of such vectors and subsequently reproduced to verify that the original audio signal is modulated, therefore the content is unrecognizable

```
Portadora = transpose(Portador);           %Invertir filas por columnas.
AudioModulado = Portadora.*y;              %Multiplicación punto a punto.
sound(AudioModulado, Fs);                  %Reproducción del audio modulado.
```

The frequency modulated signal is shown in Figure 5; to play the file becomes imperceptible the message said in the audio

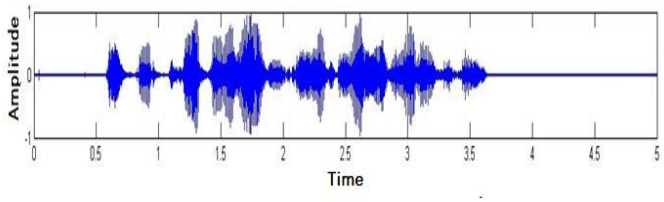

Fig 5. Graph of frequency modulated audio.

To demodulate the signal, ie removing the carrier to obtain the original audio is multiplied point by point the Audio Modulado and carrier. This is a very simple method and is one of the advantages of working with frequency modulation.

```
AudioDemodulado = Portadora.*AudioModulado;    %Demodulación de audio.
sound(AudioDemodulado, Fs);                    %Reproducción del audio demodulado.
```

In Figure 6 (a) the original audio signal and (b) shows the demodulated signal; you can see at a glance the similarity between the two graphs; which means that the original signal obtained correctly.

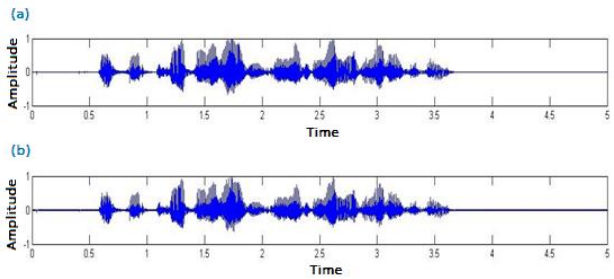

Fig 6. Graph of (a) original signal and (b) frequency demodulated signal.

The disadvantage of using this method is the distortion of the original signal to be recovered; however the message is perceptible. The way to reduce this effect is by applying a low pass filter which allows passing the natural frequency of the original audio and noise frequencies (carrier) are rejected;

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the cost of implementing a digital filter is the use of resources, which when passed to an application on a mobile device, you cannot have many resources, hence, the importance of this efficient and compact method.

III. CODIFICACION THROUGH A PSEUDORANDOM VECTOR

A more effective method is to generate a pseudo-random vector from a seed numbers; said seed shall be known to the sender and receiver of the message. In Figure 7 flow chart describing operation shown. The difference with the previous method is that the function is applied to generate pseudo-random numbers from a seed in common between the transmitter and receiver.

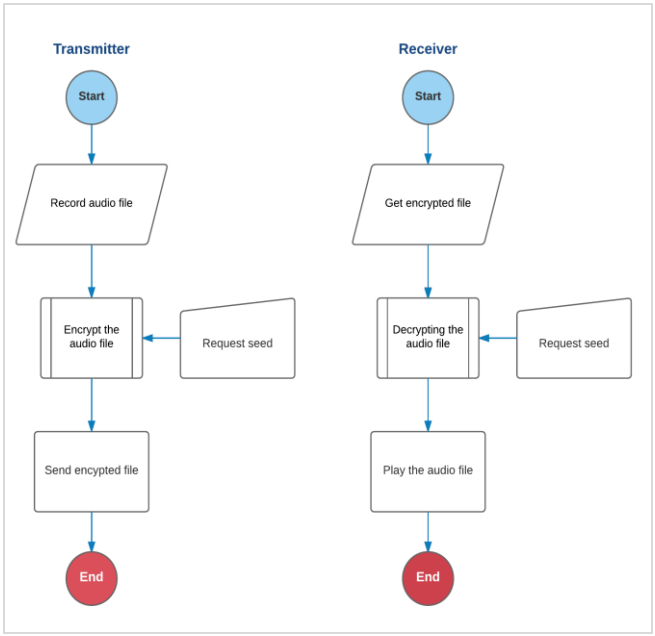

Fig 7.Flowchart encryption pseudo-random vector.

As a seed in common may be codificarar and des-codificarar correctly the original audio without any modification. With a slight difference in the seed, the random vector generated changes and creates an entirely different audio file.

Starting from the audio signal used in the above method, the amplitude vector (y) and its sampling frequency (Fs) is obtained.

```
[y,Fs] = audioread(filename);          %Obtener el vector de amplitud y la Fs.
```

In Figure 8, the graph of the original audio signal is shown.

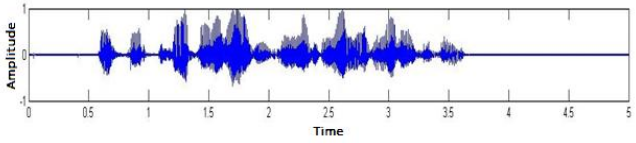

Fig 8. Graph of original audio.

It should generate a vector of pseudo-random numbers, ie in which the same initial conditions always produce the same results. In this case by introducing the same seed, the same random values are always generated.

```
seed = input('Semilla: ')          %Recibir semilla. (Tipo int)
rng(seed);                         %Configurar la semilla.
Ruido = 10 * rand(1, length(y));   %Generar vector aleatorio.
```

This random vector must contain the same number of samples the vector amplitude (y) of the original audio with values between zero and ten; in Figure 9 the graph shown Vector.

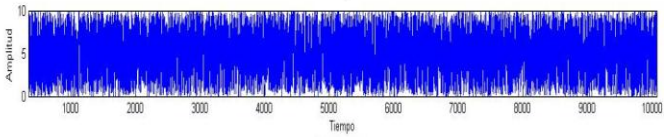

Fig 9.Vector Graphics pseudo-random

Once the vector obtained, simply adding it to the amplitude vector as shown below.

```
AudioModulado = y + ruido;          %Encriptación de audio.
sound(AudioModulado, Fs);
```

In This way the original message is imperceptible to the noise generated. In Figure 10, the audio file is displayed and encoded

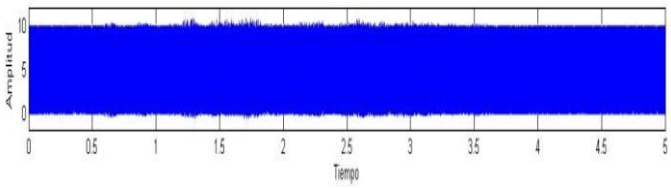

Fig 10. Graph vector encoded audio.

It can be seen that the original signal is imperceptible to a signal of high frequency noise, this is one of the disadvantages of working with noise generated as applying a low pass filter can remove most of the components making perceptible the original message. Knowing the vector of random noise, you can subtract to get the original audio

```
AudioNormalizado = AudioModulado - ruido;      %Des-encriptación.
sound(AudioNormalizado, Fs);
```

In Figure 11 the graph audio vector obtained by removing the noise is shown.

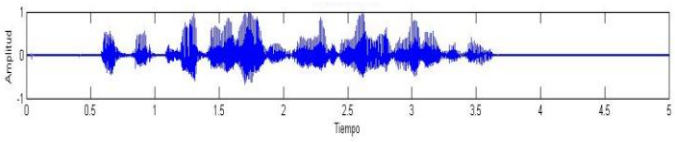

Fig 11. Graph vector des-encrypted audio.

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The audio is exactly the same as the original, this is because the added noise is equal to what is subtracted, so that the original signal however small it is left alone.

IV. IMPLEMENTATION

To achieve the implementation of the method encoded by a vector pseudo-random has been added this utility to an instant messaging application, which also have the option of sending audio incorporates functions messaging cipher text.

Figure 12 shows the main window of the container application of the method for sending encrypted audio.

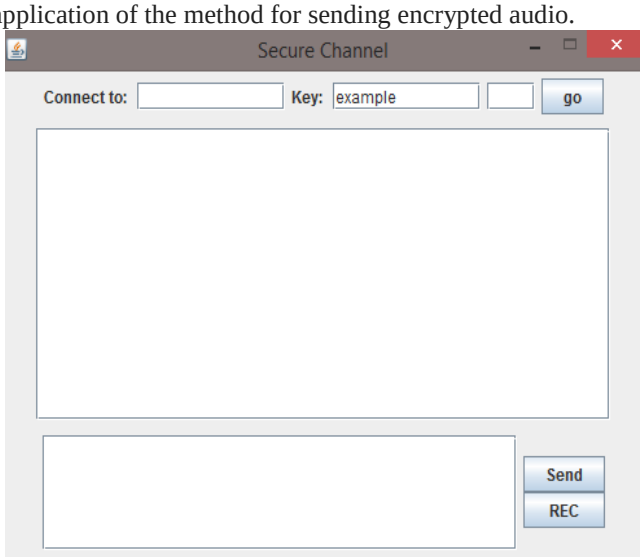


Fig 12.Main window of the application.

The application developed in Java, use the audio library built for handling digital audio sampled. Such an API for controlling audio is identified as javax.sound.sampled and can be found within various utilities to configure the audio features, such as the sampling frequency, the number of bits per sample and channels used.

REC function button in the application window is as follows:

First.- Once the button is pressed, it generates an audio input channel for the case received microphone system. The digital signal which is received through said channel is subsequently sampled at the specified frequency. While the sampled received bytes are stored in a vector for further processing.

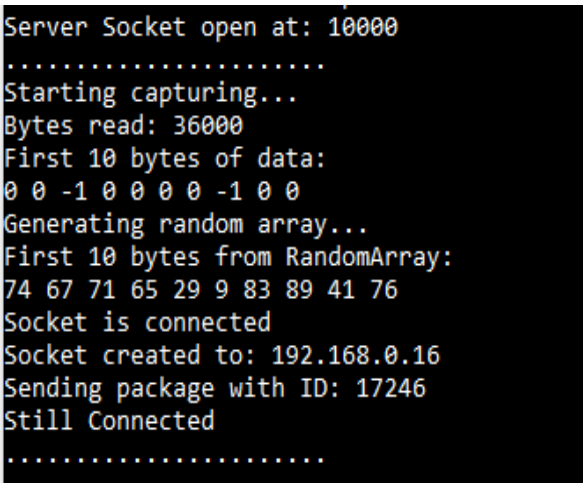
Second.- Releasing the button by the user, the program closes the inlet channel and works only two data vectors: The vector generated above, which will be referred to as original data vector and a new vector pseudorandom. The latter will be the same size as the original.

To obtain the pseudorandom vector Random class library included in the Java java.util.Random is used. This class allows the generation of numbers that vary according to an initial value; this value is called the seed. For the case of an instant messaging application, the seed can be used as key coding and decoding, which is known in the area of information security as a pre-shared key (Pre-Shared Key - PSK).

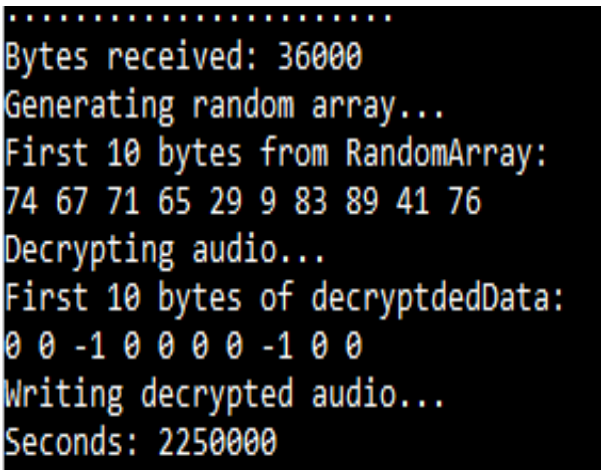
Once the two vectors are ready, the sum of vectors, byte by byte and the resulting vector corresponding to the coded signal is performed. This vector is passed to other methods within the application that are responsible for sending through the communication channels for the network.

Figure 13 shows a debug A.- execution processes audio sending, the number of bytes read and vectors, random and detailed output.

Figure 13 B shows the opposite side of communication, where the information received by the node for the specified port is described.



a)



b)

Fig 12. Debug the process of sending and digital audio capture.

V. CONCLUSIONS

In this paper three methods to hide audio signals are occupied; coding using a pseudo-random vector is one of the most simple and practical as it offers quite acceptable results and consumes few resources if you want to implement in a mobile application.

Another of the most optimal methods is the implementation of a low-pass filter, so that only permits a original audio frequency and high frequency carrier is suppressed. This requires to design a filter and its difference

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equation to implement a microcontroller, an application in C, Java, etc., where resources are more limited. More complex methods, which can generate a table of random numbers corresponding to a seed in common between the message sender and receiver, ie both participants can generate the same vector so that it can be subtracted to obtain the original audio.

Implementing digital audio coding using a vector pseudorandom number is an easy way to secure communications. It is a very safe method, since unlike other methods in which a small change in the pre-shared key generates only a small deformation in the original message, while a more minimal change it in the seed of a pseudo-random number will the total strain vector original audio. This work leaves us very satisfied because it provides security with limited resources.

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Bárbara Emma Sánchez Rinza Bachelor in Physics, Master Degree in Optics, Doctor's Degree in Optics. She has written 45 chapters of books, 44 national and international articles, 12 memoirs. She has participated in 105 conferences in different forums. She has directed 27 Bachelor Thesis and 6 Master Thesis.