

A Novel Approach for Signal Security and Video Transmission using Lower Bandwidth Technique

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ABSTRACT: Normally mobile phone sends and receives information (voice messages, fax, computer data, etc) by radio communication. Radio frequency signals are transmitted from the phone to the nearest base station and incoming signals (carrying the speech from the person to whom the phone user is listening) are sent from the base station to the phone at a slightly different frequency. In order to avoid this kind of insecurity interface, generating noise at the transmitting end and Filter (noise removal) at the receiving end generation is provided with Steganography technology and also introduce transmit the video signal in lower bandwidth. By converting the video signals into frames and hiding it into audio signal and then transmit through a transmitting medium and receive by the receiver. By extract the video from the audio signal we can get the original video signal. We transmit the video signal like a audio signal. Thus the video, Thus we get the secure TX & RX, but in this method any try to view or try to hack the filter key can able to transmit in lower bandwidth with security.

KEYWORDS: Mobile Communication, Base Station, Noise , Filter, Steganography Keying, Frames, Lower bandwidth.

I. INTRODUCTION

The genius of the cellular system is the division of a city into small cells. This allows extensive frequency reuse across a city, so that millions of people can use cell phones simultaneously. The carrier chops up the city into cells. Each cell is typically sized at about 10 square miles (26 square kilometres). Cells are normally thought of as hexagons on a big hexagonal grid. Because cell phones and base stations use low-power transmitters, the same frequencies can be reused in non-adjacent cells. The two purple cells can reuse the same frequencies. Each cell has a base station that consists of a tower and a small building containing the radio equipment (more on base stations later). A single cell in an analog system uses one-seventh of the available duplex voice channels. That is, each cell (of the seven on a hexagonal grid) is using one-seventh of the available channels so it has a unique set of frequencies and there are no collisions

In other words, in any cell, 56 people can be talking on their cell phone at one time. With digital transmission methods, the number of available channels increases. For example, a TDMA-based digital system can carry three times as many calls as an analog system, so each cell has about 168 channels available (see this page for lots more information on TDMA, CDMA, GSM and other digital cell-phone techniques). Cell phones have low-power transmitters in them. Many cell phones have two signal strengths: 0.6 watts and 3 watts (for comparison, most CB radios transmit at 4 watts). The base station is also transmitting at low power. Low-power transmitters have two advantages: The transmissions of a base station and the phones within its cell do not make it very far outside that cell. Therefore, in the figure above, both of the purple cells can reuse the same 56 frequencies. The same frequencies can be reused extensively across the city. The power consumption of the cell phone, which is normally battery-operated, is relatively low. Low power means small batteries, and this is what has made handheld cellular phones possible. The cellular approach requires a large number of base stations in a city of any size. A typical large city can have hundreds

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of towers.

In this method the voice signal is added with the additive white Gaussian noise and it becomes noise signal. It transmit by the transmitter and receive by the receiver. At the receiver side the added noise remove by the IIR filter. Thus the original signal is reach the receiver with high security.

In this method by applying steganography key in the transmit side and transmit the signal and it will receive the receiver and it ask a secret key to deliver the output. If the receiver know the key then who can listen the output signal. Video transmitting in lower bandwidth, it requires high bandwidth to transmit a video signal in the mobile communication. If we use these much of bandwidth and high frequency for transmit video signal it may affect the human and also the animals. So by applying this methodology we can reduce the frequency and prevent from the harmful affect.

In this method there is video input which is convert into audio and also it convert into frames and also the frames are hide behind the audio signal. The audio signal is transmit through the transmitting medium like a normal mobile communication transmitting.

In the receiver side the transmitted signal received by the receiver which is in the form of audio signal. Then the hidden video frames are extracted from the audio signal. Then the frames are combined to gather to form a video signal. Then combine the audio and video signal to get the original video signal as the input signal

II. RELATED WORKS

1. All cell phones have special codes associated with them. These codes are used to identify the phone, the phone's owner and the service provider.
2. The control channel is a special frequency that the phone and base station use to talk to one another about things like call set-up and channel changing. If the phone cannot find any control channels to listen to, it knows it is out of range and displays a "no service" message.
3. When it receives the SID, the phone compares it to the SID programmed into the phone. If the SID's match, the phone knows that the cell it is communicating with is part of
4. Along with the SID, the phone also transmits a registration request, and the MTSO keeps track of your phone's location in a database -- this way, the MTSO knows which cell you are in when it wants to ring your phone.
5. The MTSO gets the call, and it tries to find you. It looks in its database to see which cell you are in.
6. The MTSO picks a frequency pair that your phone will use in that cell to take the call.
7. The MTSO communicates with your phone over the control channel to tell it which frequencies to use, and once your phone and the tower switch on those frequencies, the call is connected. You are talking by two-way radio to a friend!
8. As you move toward the edge of your cell, your cell's base station notes that your signal strength is diminishing. Meanwhile, the base station in the cell you are moving toward (which is listening and measuring signal strength on all frequencies, not just its own one-seventh) sees your phone's signal strength increasing. The two base stations coordinate with each other through the MTSO, and at some point, your phone gets a signal on a control channel telling it to change frequencies. This hand off switches your phone to the new cell.

Cellular networks offer a number of desirable features:

1. More capacity than a single large transmitter, since the same frequency can be used for multiple links as long as they are in different cells
2. Mobile devices use less power than with a single transmitter or satellite since the cell towers are closer
3. Larger coverage area than a single terrestrial transmitter, since additional cell towers can be added indefinitely and are not limited by the horizon
4. Major telecommunications providers have deployed voice and data cellular networks over most of the inhabited land area of the Earth. This allows mobile phones and mobile Computing devices to be connected to the public switched telephone network and public Internet. Private cellular networks can be used for research^[1] or for large organizations and fleets, such as dispatch for local public safety agencies or a taxicab company.^[2]

A land area to be supplied with radio service is divided into regular shaped cells, which can be hexagonal, square, circular or some other regular shapes, although hexagonal cells are conventional. Each of these cells is assigned with multiple frequencies ($f_1 - f_6$) which have corresponding radio base stations. The group of frequencies can be reused in

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other cells, provided that the same frequencies are not reused in adjacent neighbouring cells as that would cause co-channel interference.

The increased capacity in a cellular network, compared with a network with a single transmitter, comes from the mobile communication switching system developed by Amos Joel of Bell Labs^[3] that permitted multiple callers in the same area to use the same frequency by switching calls made using the same frequency to the nearest available cellular tower having that frequency available and from the fact that the same radio frequency can be reused in a different area for a completely different transmission. If there is a single plain transmitter, only one transmission can be used on any given frequency. Unfortunately, there is inevitably some level of interference from the signal from the other cells which use the same frequency. This means that, in a standard FDMA system, there must be at least a one cell gap between cells which reuse the same frequency.

III. CELL SIGNAL ENCODING

To distinguish signals from several different transmitters, frequency division multiple access (FDMA) and code division multiple access (CDMA) were developed.

With FDMA, the transmitting and receiving frequencies used in each cell are different from the frequencies used in each neighbouring cell. In a simple taxi system, the taxi driver manually tuned to a frequency of a chosen cell to obtain a strong signal and to avoid interference from signals from other cells.

The principle of CDMA is more complex, but achieves the same result; the distributed transceivers can select one cell and listen to it.

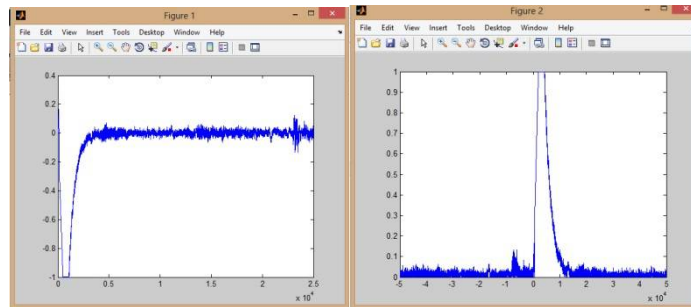
- The term digital filter arises because these filters operate on discrete-time signals
- The term finite impulse response arises because the filter output is computed as a weighted, finite term sum, of past, present, and perhaps future values of the filter input, i.e.,

A key is used to encrypt and decrypt whatever data is being encrypted/decrypted. Modern cryptographic systems include symmetric-key algorithms (such as DES and AES) and public-key algorithms (such as RSA). Symmetric-key algorithms use a single shared key; keeping data secret requires keeping this key secret. Public-key algorithms use a public key and a private key.

The main advantage digital IIR filters have over FIR filters is their efficiency in implementation, in order to meet a specification in terms of pass band, stop band, ripple, and/or roll-off. Such a set of specifications can be accomplished with a lower order IIR filter than would be required for an FIR filter meeting the same requirements. If implemented in a signal processor, this implies a correspondingly fewer number of calculations per time step. the computational savings is often of a rather large factor

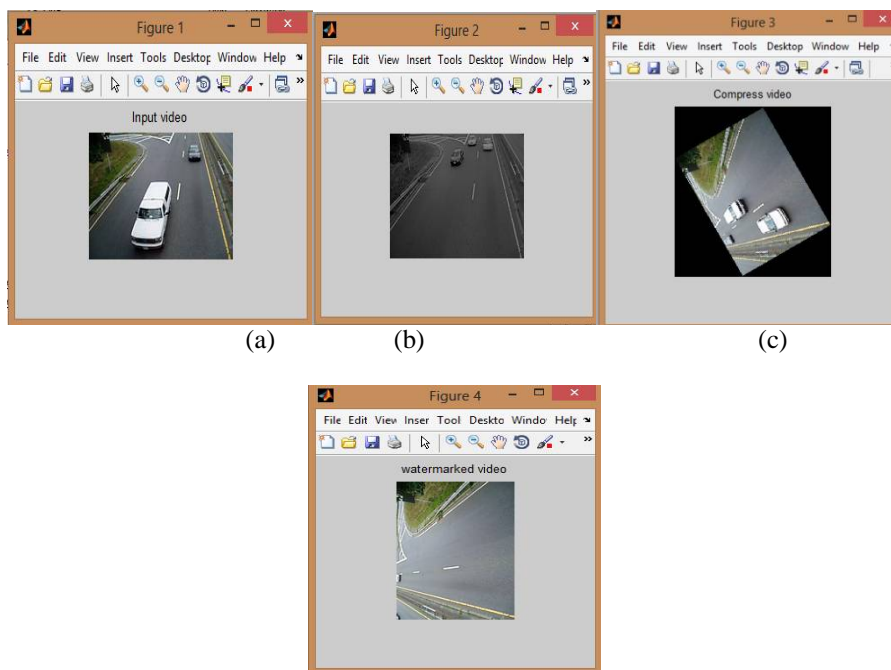
IV. EXPERIMENTAL RESULTS

The expected output is the recorded voice signal is added with noise then the signal is transmitted through the transmitting medium and then reach the receiver. At the receiver side the noise is removed by the IIR filter. And the original voice signal is reach the receiver with high productivity. Figure.1 (a) input audio signal. (b) Audio signal denoised output.



(a) (b)

Figure.2 The video signal is converted into frames and the frames are hidden into audio signal. Output of the voice signal analysis. In the image we are adding with audio in the original signal (a)Input video with signal with audio (b)Video converted into frames (c)frames converted into compressed video (d) The image explain about the removing the noise from the output signal



(d)

The input video is compressed into small in size and video is converted into frames and compressed which is displayed . The image is converted into frames and that frames is hidden into audio that audio is transmitted and received by the receiver and extracted from the receiver end.

V. CONCLUSION

The proposed method firstly, Additive White Gaussian noise(AWGN) added with voice signals and then secondly, using Infinite impulse response (IIR) filters and also signals security achieved by using key generation model. The audio is added with noise in the transmitter side and extracted in the receiver side. In this method the input data in the form of voice signal is protected by the noise and this is implement the high security in communication process. And also by transmitting the video signal in lower bandwidth. In this process video signal is converted into frames and hidden in the audio signal by the steganography method. Using lower bandwidth video can transmit as like a voice signal in the mobile communication. By using this we can able to control the bandwidth and attain the efficiency which is

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equal to the higher bandwidth .by reducing the bandwidth people and animals are prevent from harmful effect.By this methodology we can avoid the cross talk in the mobile communication. The cross talk occur due to mismatching of time duration. If any cross talk occur at the time of transferring the unknown user can listen only the noise signal not able to listen the original signal.

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