TRANSMIT AND RECEIVE OF FM SIGNALS USING SOFTROCK SDR AND MATLAB

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I certify that in my opinion the thesis submitted by Ahmeda. G Ahmeda GAREANE titled "TRANSMIT AND RECEIVE OF FM SIGNALS USING SOFTROCK SDR AND MATLAB" is fully adequate in scope and in quality as a thesis for the degree of Master of Science.

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"I declare that all the information within this thesis has been gathered and presented in accordance with academic regulations and ethical principles and I have according to the requirements of these regulations and principles cited all those which do not originate in this work as well."

Ahmeada G. Ahmeda GAREANE

ABSTRACT

M. Sc. Thesis

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In this study, the transmission and reception of FM signals using the Softrock Ensemble RXTX SDR transceiver are discussed. FM signals are generated and plotted using the MATLAB, with using two input frequencies and a modulation index as inputs for the developed system. The system includes two stages; FM modulation and FM demodulation. In the FM modulation stage, the carrier signal frequency is changed based on the signal intensity to carry a recorded audio file, in which the resultant modulated wave is then transmitted to the receiver. In the FM demodulation stage, the audio signal is recovered from the modulated wave using a low pass filter to remove the carrier frequency and permit the audio signal to reach the speaker. The Audacity is then deployed to listen to and inspect such signal. Software routines in Matlab are used for modulation and demodulation.

**Key Word : ** Frequency Modulation and demodulation, SDR, Matlab, Softrock SDR hardware.

**Science Code : ** 905.1.067

ÖZET

Yüksek Lisans Tezi

SOFTROCK SDR VE MATLAB KULLANARAK FM SİNYALLERİNİN İLETİMİ VE ALIMI

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Bu çalışmada, Softrock Ensemble RXTX SDR telsizi kullanılarak FM sinyallerinin iletimi ve alımı tartışılmıştır. FM sinyalleri, geliştirilen sistem için girişler olarak iki girdi frekansı ve bir modülasyon indeksi kullanmak kaydıyla MATLAB kullanılarak oluşturulur ve çizilir. Sistem iki aşamadan oluşur; FM modülasyonu ve FM demodülasyonu. FM modülasyon aşamasında, taşıyıcı sinyal frekansı, kaydedilen bir ses dosyasını taşımak için sinyal yoğunluğuna bağlı olarak değiştirilir ve sonuçta elde edilen modüle edilmiş dalga daha sonra alıcıya iletilir. FM demodülasyonu, taşıyıcı frekansını modüle edilmiş dalgadan ayirmak ve ses sinyalinin hoparlöre erişmesine izin vermek için alçak geçiren bir filtreden alici sinyalini geçirerek gerçeklestirilir. Audacity programi, daha sonra bu sinyali dinlemek ve denetlemek için kullanilir.

Anahtar Sözcükler : Frekans Modülasyonu ve demodülasyonu, SDR, Matlab, Softrock SDR donanımı.

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CONTENTS

Page

APPROVAL ii
ABSTRACT iv
ÖZET vi
ACKNOWLEDGMENT viii
CONTENTS ix
LIST OF FIGURES xii
LIST OF TABLES xiv

CHAPTER ONE 1
INTRODUCTION 1
1.1. INTRODUCTION OF FREQUENCY MODULATION (FM) 1
1.2. SOFTWARE PROGRAMS 2
1.2.1. Software Defined Radio (SDR) 2
1.2.1.1. SDR Framework 2
1.2.1.2. Applications of SDR 3
1.2.1.3. Employment Opportunities 3
1.2.1.4. SDR Sharp 4
1.2.1.5. SDR RTLSDR Scanner 4
1.2.2. HDSDR Software 4
1.2.2.1. Definition and Features of HDSDR 4
1.2.2.2. Use of HDSDR 5
1.2.3. Softrock Ensemble RXTX 7
1.2.3.1. Background of Softrock 7
1.2.3.2. Advantages of the Softrock 7
1.2.3.3. Band Coverage of Softrock 8
1.2.3.4. Types of Softrock Ensemble 9
1.2.4. Audacity Software 10
1.2.4.1. Introduction to Audacity 10

Page

1.2.4.2. Processing Recordings Using the Audacity 10

CHAPTER TWO 12
OVERVIEW 12
2.1. OVERVIEW... 12
2.2. MODULATION AND DEMODULATION OF ANALOG SIGNALS 13
2.2.1. AM Signals 13
2.2.2. FM Signals... 16
2.2.2.1. Overview and Theory of FM Signals... 16
2.2.2.2. Benefits of FM Signals over AM Signals 17
2.2.2.3. Modulation and Demodulation of FM Signals 17
2.2.3. Comparison Between FM and AM Signals 18
2.3. MODULATION AND DEMODULATION OF DIGITAL SIGNALS......... 19
2.3.1. Modulation and Demodulation of BPSK Signals 19
2.3.2. Modulation and Demodulation of QPSK Signals................................ 21
2.3.3. Modulation and Demodulation of FSK Signals................................... 22
2.4. FM DEMODULATORS 23
2.4.1. DSP Based FM Demodulators... 23
2.4.2. FM Demodulation Methods Summarized 24
2.4.2.1. Discrimination of FM Differentiation.. 24
2.4.2.2. Quadrature FM Demodulation... 25
2.4.2.3. Arctangent FM Demodulation 26
2.4.2.4. Phase Locked Loop.. 27
2.4.2.5. Zero Crossing Detectors 29

CHAPTER 3 30
RXTX.. 30
3.1. RXTX 30
3.2. AUDACITY 34
3.2.1. Introduction.. 34
3.2.2. Getting Started 34
3.2.2.1. Recording Your Audio.. 35

Page

3.2.2.2. Listening to Your Recording 37
3.2.2.3. Adjusting the Recording Level for Optimum Quality 37
3.2.2.4. Saving Your Work 38
3.2.2.5. Editing Your Audio.. 39
3.2.2.6. Adding Other Audio 42
3.2.2.7. Adjusting the Volume of Different Audio Rracks 43
3.2.2.8. Fading in and Out... 43
3.3. ARCHIVING YOUR WORK AND EXPORTING YOUR PODCAST 44
3.3.1. Archiving Your Work as an Uncompressed WAV File 44
3.3.2. Exporting Your Audio as an MP3 File 45
3.3.3. A Note on File Size and Compression... 45
3.3.4. Setting the Bit-Rate in Audacity.. 46

CHAPTER 4 48
FM MODULATION 48
4.1. FM MODULATION 48
4.2. FM DEMODULATION.. 50

CHAPTER 5 53
CONCLUSION.. 53
REFERENCES.. 55

CURRICULUM VITAE 58

LIST OF FIGURES

Page

Figure 1.1. SDR framework projects ... 3

Figure 1.2. HDSDR Control Panel ... 6

Figure 1.3. Representation of receiving FM radio broadcast signal using the HDSDR ... 6

Figure 2.1. Defined voltage in the AM process ... 13

Figure 2.2. Coherent demodulation approach ... 15

Figure 2.3. Non-coherent demodulation approach ... 15

Figure 2.4. Illustration of FM signal ... 16

Figure 2.5. Modulation process of BPSK signal ... 19

Figure 2.6. Bemodulation process of BPSK signal ... 20

Figure 2.7. Carrier recovery process ... 21

Figure 2.8. QPSK modulation process ... 21

Figure 2.9. QPSK demodulation process ... 22

Figure 2.10. FSK signal production ... 22

Figure 2.11. FSK demodulation ... 23

Figure 2.12. Differential FM detection. ... 24

Figure 2.13. Quadrature FM detection. ... 26

Figure 2.14. Arctangent FM detection. ... 27

Figure 3.1. Block diagram. ... 31

Figure 3.2. The main Audacity program screen. ... 35

Figure 3.3. The control toolbar. ... 36

Figure 3.4. Recorded audio shown as an Audio Waveform in an Audio Track. ... 36

Figure 3.5. The mixer toolbar. ... 37

Figure 3.6. Waveforms showing recording levels that are correct, too low and too high. ... 38

Figure 3.7. Selecting part of the waveform for deleting and/or editing. ... 39

Figure 3.8. Overlapping original and newly recorded audio tracks. ... 40

Figure 3.9. Highlight the waveform from the point the second track starts to the end. ... 41

Page

Figure 3.10. Remove the highlighted section of the original track and add it to a new track. 41

Figure 3.11. Audio tracks re-aligned with time shift tool. 42

Figure 3.12. A stereo wave form imported into Audacity. 43

Figure 3.13. Warning message indicating that all separate tracks will be combined on export. 45

Figure 3.14. The Audacity preferences screen. 47

LIST OF TABLES

Page

Table 2.1. Comparison among FM and AM signals 18

CHAPTER ONE

INTRODUCTION

1.1. INTRODUCTION OF FREQUENCY MODULATION (FM)

In practice, Frequency Modulation (FM) is a type of analog modulation, where carrier wave frequency is varied by the information or message signal that carries the baseband information. The most common FM signals are the transmitted audio signals via FM radio communications. On the other hand, Radio Broadcast Data Systems (RBDSs), which are digital data with low bandwidth digital information, can be also transmitted by the FM radio [1].

The operation range of FM signals' frequency is 88-108 MHz, where this makes these signals less vulnerable to both the orientation and presence of humans and objects with small sizes [2,3]. In addition, FM signals are mainly stronger than Wi-Fi signals due to their ability to simply cover large areas with offering efficient indoors penetration. Thus, the use of these signals eliminates the need to using custom infrastructures. In addition, the majority of mobile devices composed of FM radio receivers, which are cheaper and consume less power than those of Wi-Fi signals [4].

Particularly, the simplest way to generate FM signals is the application of the message signal to the control voltage of the Voltage-Controlled Oscillator (VCO) directly. It then offers a carrier sinusoidal wave that has constant amplitude and a control voltage based frequency. Thus, the carrier wave is a function of the message signal; it is the same as its center frequency when there is no message signal, while the output signal instantaneous frequency varies below and above this frequency when there is a message signal [1].

Based on the FM output signal, two observations can be defined. The FM signal amplitude is constant despite of the message signal, where this results in constant envelope property. In addition, the frequency-modulated output is nonlinearly based on the message signal. Thus, the FM signal properties cannot be easily analyzed. On the other hand, the FM signal bandwidth can be estimated using a tone message signal based on signifying the number of its efficient sidebands. A message signal can be retrieved from the FM signal based on performing a frequency demodulation. It includes a frequency discriminator, where it is a differentiator with a specific envelope detector [1].

1.2. SOFTWARE PROGRAMS

1.2.1. Software Defined Radio (SDR)

Even though the hardware components of the SDR device is necessary, explanation of how it operates is well explained by it software components. This section offers the main SDR manipulation of the signals.

1.2.1.1. SDR Framework

Efficient functioning of the SDR device in a computer or any other FPGA digital signals involves acquiring SDR software to enhance the communication. However before softwareis developed, there is a need to identify the framework first to offer interface functions of low level 4 The SDR framework can be described as for time length as demonstrated in the Figure 1.1. beneath.

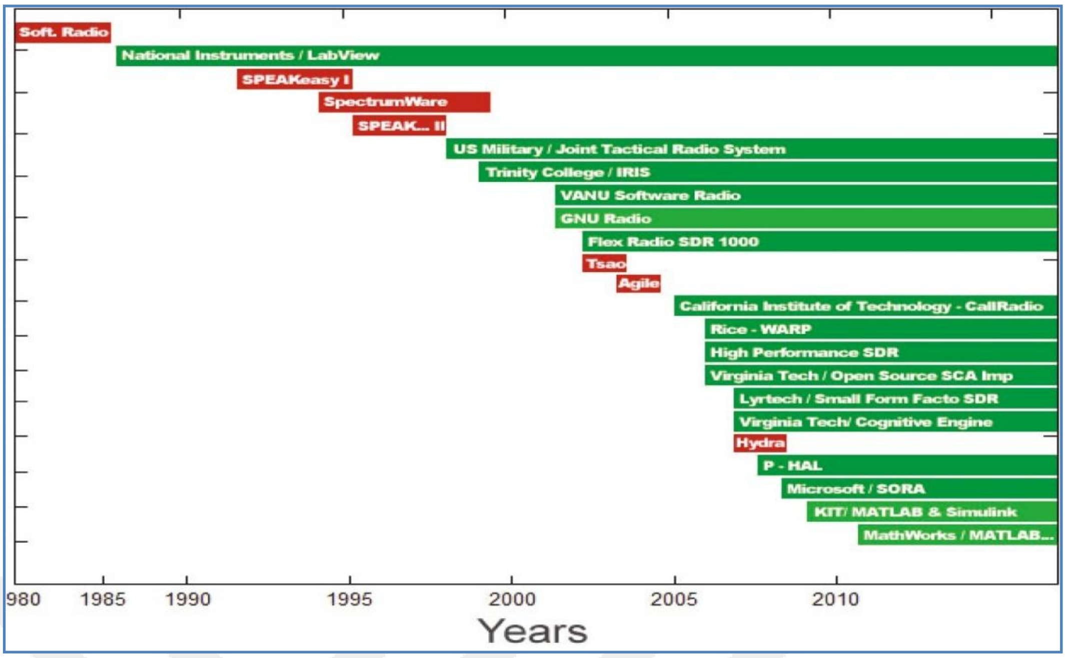

Figure 1.1. SDR framework projects [5].

1.2.1.2. Applications of SDR

After a communication link between the SDR and the computer is complete, various uses of SDR technology can be identified. The standarduse of the technology is its ability to correct errors in real time. However, other applications are connectedto the device that includes: Cost minimization and Spectrum Control, Opportunity Driven Multiple Access as well as Dynamic Spectrum Positioning. Research shows SDR implementations are less expensive that the traditional analogue devices [6]. The SDR technology has created great concern especially in telecommunications such as GPS receiving of signals, HF propagation and Interpretation of Cellular Technology Emissions.

1.2.1.3. Employment Opportunities

Most of the SDR application areas are universities and other large organizations that dedicate massive amounts for research purposes. However through the use of the RTL2831 device in coordination with other devices, low-cost solutions can be achieved.

1.2.1.4. SDR Sharp

This device is used to display the SDR generated readings and consist of four windows. Fist window is where the spectrum is shown and where FM stations can be clearly being viewed. The next window referred to as the waterfall chart indicates the behaviors of the signal time and shows the intense emissions in different colors [5]. The other windows help to plot the bandwidth within the displayed spectrum at the first window. To understand how the SDR-sharp device operates, identification of various applications is crucial.

1.2.1.5. SDR RTLSDR Scanner

Being tested on different pertain systems such as Windows 7, 8 and Ubuntu, RTLSDR Scanner is a necessary software. The software analyses spectrum through the gathering of data, consecutive scans and making comparisons. Other applications of RTLSDR Scanner include identification of repeaters electrical parameter's number through checking of the signal values given by different manufacturers and identifying whether they are used in the organization's equipment7. Also, RTLSDR Scanner is used in noise characterization by use of bands.

1.2.2. HDSDR Software

1.2.2.1. Definition and Features of HDSDR

The HDSDR is a freeware SDR program used for Microsoft Windows. As well, it is an efficient Winfred version. It is mainly deployed for radio astronomy, Non-Directional Radio Beacons (NDBs) hunting, spectrum analysis, ham radio and radio listening. The main features of such software are [7]:

- Dividing large scale spectrum and waterfall display for both input and output signals

- Zooming the Radio Frequency (RF) and Audio Frequency (AF) spectrum and waterfall in order to fit the window width in an independent way of the Fast Fourier Transform (FFT) resolution bandwidth
- Offering effective and flexible usage of screen areas with various dimensions, where this is helpful for the detection of pattern noise and monitoring of short wave conditions
- Offering fundamental transmit (TX) functionality in the AM, FM, SSB and CW modes
- Producing a pair of /Q modulated signal for the TX input signal on the output of the TX
- Offering noise reduction, adaptable band pass filter, noise blanker, automatic notch filter and several manual adaptable notch filters
- Recording and playing back several RF and AF wav files with the use of a recording scheduler
- Storing and loading the HDSDR program options per profile to simplify the use of various receivers
- Offering a frequency manager for user frequency lists, ham bands and radio bands

1.2.2.2. Use of HDSDR

The application of the HDSDR software supports the FM demodulation. After installing the program, several steps must be performed on the HDSDR Control Panel shown in Figure 1.2 below. In the first step, a user clicks on the Soundcard option to choose the RX soundcard to enter the input for I and Q and the speaker output. In the second step, a user clicks on the bandwidth to choose the input bandwidth. In the third, a user clicks on the options to choose the input, which must be softrock one and then select the option of “Swap I and Q Channels for RX Input" [8].

Figure 1.2. HDSDR Control Panel [8].

After loading the HDSDR, the "Line In" jack on the RX must be connected with the "Line In" on the computer soundcard. A 50 ohm antenna must be then connected to the Rig's antenna connector as well as the Rig's USB connector must be connected to a rig's USB port. After that, three controls must be performed; controlling the oscillator output, tuning the frequencies above and below the local oscillator frequency and establishing the center frequency in a frequency memory [8]. A representation of receiving FM radio broadcast signal using the HDSDR is shown in Figure 1.3.

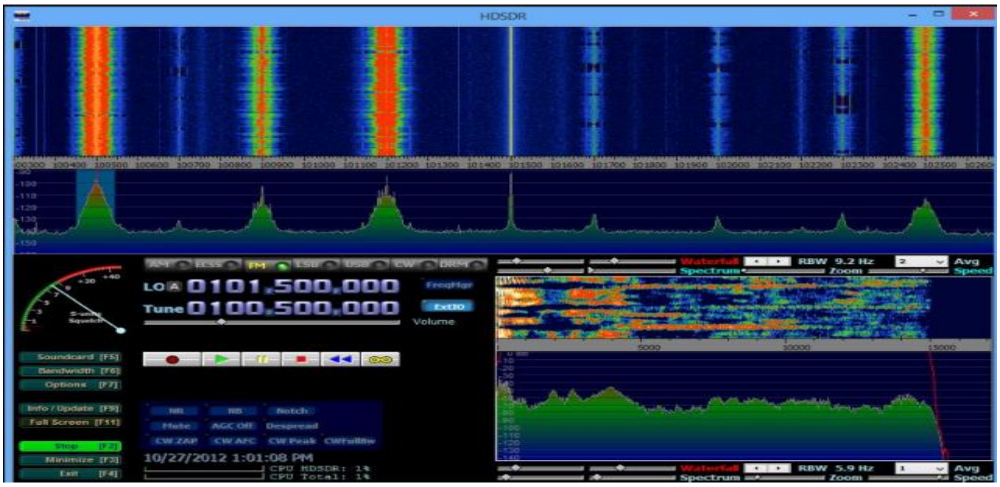

Figure 1.3. Representation of receiving FM radio broadcast signal using the HDSDR [8].

1.2.3. Softrock Ensemble RXTX

1.2.3.1. Background of Softrock

In general, the Softrock is a type of SDR program that composed of three main building blocks [9]:

- SDR hardware, where it is a Softrock kit
- Running SDR software on a computer
- A stereo soundcard including one stereo input for the RX and a stereo output for the TX with another soundcard that delivers the demodulated RX sounds to the speaker

The first production of the softrock Kits was in 2005 depending on the Tony Parks KB9YIG's experience with the SDR. Practically, all of the available softrock rigs offer RX and/or TX capability for signals with constant center frequency. In other words, local oscillator of the SDR offers a consistent frequency, which anchors a portion of radio frequency bandwidth for operation. Softrock permit tuning the local oscillator to various center frequencies, in which the offered bandwidth by such SDRs is based on the soundcard function. The estimated tuning bandwidth for a center frequency (CF) from a local oscillator of the hardware and sampling rate (SR) of the soundcard is in the range from CF - SR/2 to CF + SR/2 [9].

1.2.3.2. Advantages of the Softrock

The main advantages of using the Softrock are [10]:

- Simple design: the recent performed developments in the field of computer technology facilitates dinging analog-to-digital converters (ADC), deploying sophisticated signal processing algorithms and updating the functionality of systems using software upgrades

- Reliability: The deployment of software makes the SDR products more reliable, in which the software damage / corruption chances become less than damages in hardware components
- Consistency: The SDR provides efficient performance since it does not suffer from various problems, such as aging and heating. Hardware components in turn have major impacts on the radio performance and consistency at all levels
- Simple updates: the replacement of hardware component depends on performing several changes and processes, which requires extra costs. Conversely, the whole SDR system can be updates by a simple software upgrade with adding extra functionalities. As well, software updates are free.
- Reusability: deployed software in the SDR system can be simply deployed again on other hardware platforms for various goals, such as producing new receivers
- Simple configuration: the configuration of the SDR product to meet specific requirements of customers concerning adding or improving its functionality is simple and can be performed by some changes in the software
- Offering more functionalities: several functionalities can be included by the SDR system, such as complicated signal decoders/ demodulators and recorders with low costs
- Low costs: the products of the SDR system have low costs since they deploy less hardware components as well as due to the reusability of the software.

1.2.3.3. Band Coverage of Softrock

Both the Softrock RX and TX offer band coverage in kHz on any center frequency side. Some kits have rockbound center frequency, while the center frequency of recent kits is generated from a Si570 programmable oscillator. The band coverage is based mainly on the soundcard sampling rate, where it is +/- 24 kHz for 48 kHz sampling rate. On the other hand, the band coverage is twice such value when the band coverage is 96 kHz [9].

1.2.3.4. Types of Softrock Ensemble

Ensemble RX III Home

It indicates a long softrock SDR receivers' line. It covers High Frequency (HF) ham bands including [10]:

- First band group (0): 160m with coverage range (1.8-4.0 MHz)
- Second band group (1): 80m and 40m with coverage range (4.0 - 8.0 MHz)
- Third band group (2): 30m, 20m, and 17m with coverage range (8.0 to 16 MHz)
- fourth band group (3): 15m, 12m, and 10m with coverage range (16 to 30 MHz)

RX Ensemble II Receiver Kit

It can be mainly deployed to produce a kit for both HF and Low Frequency (LF). Both the HF and LF have a coverage range from180KHZ to 3.0 MHz The circuit board size of the kit is 4.5" x 2 ".

RXTX Ensemble Transceiver Kit

It offers 1 watt SDR transceiver, while its circuit board size is 5" x 2.5", where it can be deployed with several bands; 160m, 80m/40m, 40m/30m/20m, 30m/20m/17m and 15m/12m/10m. Practically, such type comes with the whole important components for constructing an SDR transceiver for all those bands. In this case, a user must only assemble the obtainable components for producing the SDR transceiver for a target band group. The components of such type are included and assembled at the choice of builders. The softrock RXTX ensemble transceiver has full frequency agility in the implemented band pass filter limits.

1.2.4. Audacity Software

1.2.4.1. Introduction to Audacity

The Audacity is a common open source multilingual audio editor and recorder software program, created in 1999. It can work on several operating systems, such as GNU/Linux, Mac and Windows. It mainly deployed for recording tasks, editing audio sounds, mixing stereo tracks, making ringtones, sending records and tapes to computers and segmenting recordings into several tracks. As well, it provides a single-screen dashboard presenting sampling screens including control tools, such as sharpness and volume. Furthermore, it has a multi-track recording property, which allows correlating several tracks and mixing them together to get a refined result. One more property is the ability to add visual tracks to the screen, in which each track can be manipulated for the final results using the mouse controls [11, 12].

1.2.4.2. Processing Recordings Using the Audacity

Recordings obtained by various software programs, such as the HDSDR can be edited using the Audacity, which allows importing several audio file formats, such as MP3 and Wav. The program then displays the imported file waveform, in which the track name is the same as that of the imported audio file. Practically, when the waveform becomes closer to the top and bottom of the track, the audio becomes louder [13].

After displaying the waveform, the DC offset must be removed. It is mainly happen in audio files, where its presence results in a non-centered recorded waveform on the horizontal line at zero amplitude. Such problem resulted from the recording with an imperfect soundcard. A user can then listen to the imported audio track to check it [13].

A 10-second clip must be then generated from the imported audio waveform to edit it based on initially zooming the waveform, choosing the audio range that must be changes and then performing the required edit. This is based on clicking on the point where the 10-second piece must begin, zooming until 10 seconds or more after and before the cursor are shown and clicking click 10 seconds to the right of the cursor with holding the shift key. A user can listen and adjust such piece until achieving the target. After that, the last second must be faded out based on clicking1 second before the end of the waveform to be faded out. The resultant file can be then exported in any file extension. A comparison can be also performed among both the original and resultant files [13].

CHAPTER TWO

OVERVIEW

2.1. OVERVIEW

In practice, there is a need for various hardware components, such as filters, demodulators and detectors in typical radio communication systems, where this in turn requires extra costs. Therefore, the use of the SDR software simplifies the implementation of such systems in a cost effective manner. In comparison with typical radio communication systems, the use of the SDR software neglects the use of hardware components and substitutes them using a pure software program. This in turn enhances the flexibility due to the efficient ability of an SDR receiver to decode all types of signals [14].

In a radio communication system, only high frequency signals can be broadcasted over long distance. On the other hand, there is a relation among the signal frequency and the antenna height, in which the higher the antenna, the lower the signal frequency. Therefore, the transmission of low frequency signals needs using high antennas that cannot be easily made out. Therefore, the transmission of a low frequency signal in a radio communication system depends on modulating it to a high frequency one [14].

Signals used in radio communication systems to carry messages are known as carrier signals, which are high-frequency cosine or sinusoid waveforms. The transmission of such signals can be conducted by the air over long distances. The process in which a carrier signal carries a specific low frequency information signal is known as the modulation. This process in turn can be realized based on changing some of the carrier signal features. When the modulated signal reaches its target receiver, the modulated carrier signal must be processed to extract the carried original information, where this process in turn is known as the demodulation [15].

2.2. MODULATION AND DEMODULATION OF ANALOG SIGNALS

As defined previously, the modulation process represents the change in some of the carrier signal features. The analog modulation in turn represents the modulation of an analog signal with a continuous variation in the carrier signal parameters, namely; angle and amplitude, in which the angle composed of a phase and frequency. When the carrier signal amplitude changes as the modulating signal, the process is called the Amplitude Modulation (AM). On the other hand, the change in the frequency and phase of the angle cause Frequency Modulation (FM) and Phase Modulation (PM), respectively [15]. The main focus of this project is on the FM.

2.2.1. AM Signals

In the AM, the amplitude of the carrier signal changes in relation with the instantaneous modulating signal voltage amplitude [16]. In practice, the AM is linear modulation process due to the superposition principle [15]. Figure 2.1. below demonstrates the defined voltage in the AM process.

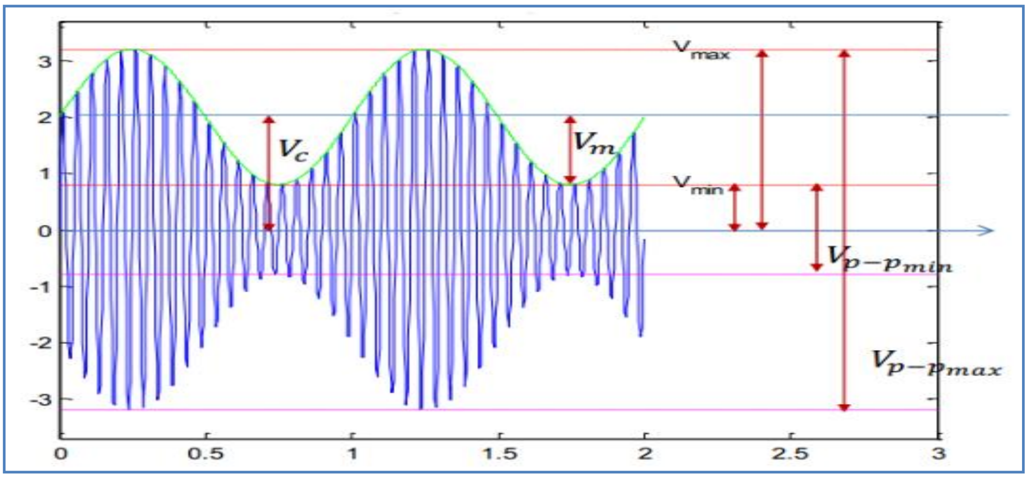

Figure 2.1. Defined voltage in the AM process [16].

Practically, the degree of modulation can be described using the modulation index, where it has an impact on the AM signal shape. When the modulation index value is bigger than 1, the carrier signal will stop. Thus, the modulation index should not exceed 1. On the other hand, the AM signal envelop is an imaginary line, which drawn through the peak value related to each wave cycle, where it can be expressed using the formula below [17]:

$$e_{ENV} = V_c + e_m \tag{2.1}$$

Where: V_c is the voltage of the carrier signal and e_m the instantaneous modulating signal voltage amplitude. On the other hand, when $e_{ENV} = V_m * \sin\omega_m t$, where V_m is represented in the Figure above and ω_m is the signal frequency, the AM envelop is expressed as follows [17]:

$$e_{ENV} = V_c + V_m * \sin\omega_m t \tag{2.2}$$

In the equation above, m stands for the modulation index, where it represents the ratio among V_m and V_c. Thus [17]:

$$e_{ENV} = V_c(1 + m * \sin\omega_m t) \tag{2.3}$$

The AM signal instantaneous voltage of the e_{AM} signal, which is the AM signal that has information within it and must be sent to the receiver, is represented as follows: [17]

$$e_{AM} = V_c(1 + m * \sin\omega_m t) * m * \sin\omega_m t \tag{2.4}$$

The demodulation process in turn is opposite to the modulation one, where this process is mainly conducted using two methods, namely; coherent and non-coherent. When the receiver has information concerning the received signal phase, the coherent demodulation method can be deployed. Such process is shown in Figure 2.2, in which the received AM signal must be multiplied by the carrier one, which generated by the receiver local oscillator. The main procedure of such process is the production of carrier signal by the receiver, in which it has the same frequency and phase of those of the received carrier signal. On the other hand, in the non-coherent demodulation, the carrier signal is not essential, in which the envelop line is deployed, where such process is shown in Figure 2.3 [17].

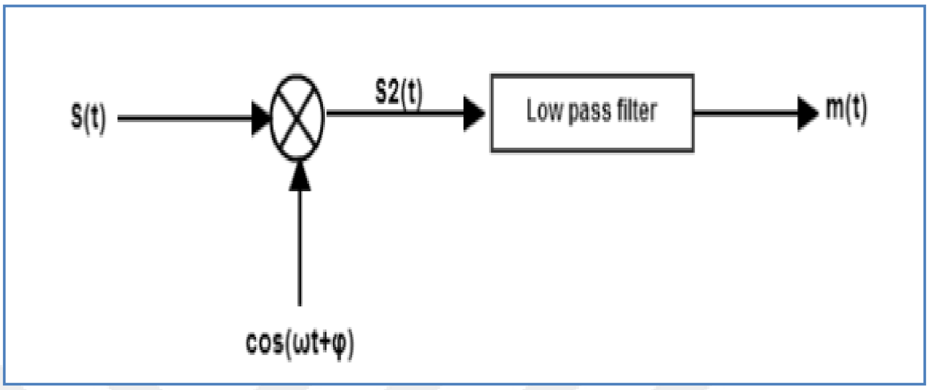

Figure 2.2. Coherent demodulation approach [17].

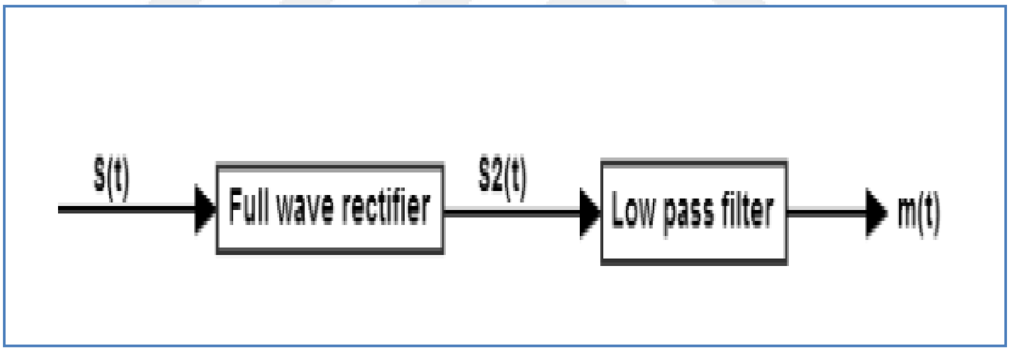

Figure 2.3. Non-coherent demodulation approach [17].

In the non-coherent demodulation, the full wave rectifier is used to transform the signal into a DC one in order to make the whole inputs negatives or positives. On the other hand, the low pass filter is deployed to keep signals within the estimated frequency pass range.

2.2.2. FM Signals

2.2.2.1. Overview and Theory of FM Signals

Practically, FM signals have different properties of AM ones. An FM signal has changed carrier wave frequency in relation with the signal intensity as shown in Figure 2.4 below with constant modulated wave amplitude. As shown in the Figure below, carrier frequency remains constant when the voltage of the signal equals to zero, A, C, E and G points. On the other hand, the carrier frequency increases to its maximum value at its positive peaks; B and F, while it decreases to its minimum value at its negative peaks [18].

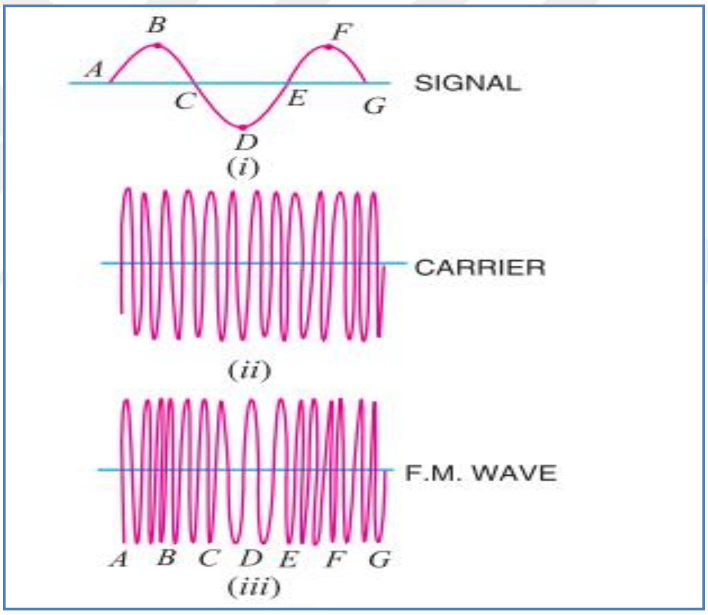

Figure 2.4. Illustration of FM signal [18].

The FM signal has constant carrier amplitude and variable carrier frequency, which varies by the modulating signal. Such variation is at the rate of the signal frequency, which represents that the deviation in the frequency is related to the instantaneous modulating signal amplitude. In practice, the frequency deviation reaches its maximum at the modulating signal peak voltage. The basic properties of FM signals are [18]:

- The FM signal frequency deviation is based on the modulating signal amplitude
- The FM signal center frequency represents the frequency without modulation or the related frequency for zero modulating voltage
- The frequency deviation cannot be determined by the audio frequency

2.2.2.2. Benefits of FM Signals over AM Signals

FM signals have several benefits over AM ones, where some of such benefits are illustrated below: [18]

- FM signals have quite larger operating range.
- FM signals offer high-fidelity reception.
- FM signals have higher transmission efficiency.
- FM signals offer noiseless reception, in which an FM receiver reject noises, which are forms of amplitude variations.

2.2.2.3. Modulation and Demodulation of FM Signals

In the modulation of FM signals, the carrier amplitude is kept constant, while its frequency changes depending on the modulating signal. Particularly, the frequency deviation among the original frequency and instantaneous frequency of the carrier signal after modulation is related to the modulating signal instantaneous amplitude [16]. On the other hand, the modulation index is used to express the change of the frequency of the modulated signal from that of the carrier one. In addition, it stands for the radio of the maximum frequency deviation to the frequency of the modulating signal, where it can be expressed as $m = \frac{\Delta f}{f_m}$ [19].

The modulated signal can be expressed using the following formula with supposing a cosine wave modulating signal $f_m(t)$ and a sine wave carrier signal

$f_c(t) = A_c \sin(2\pi f_c t)$ [17]:

$$y(t) = A_c \sin(2\pi \int_0^t f(\tau)d\tau) \quad (2.5)$$

Where: $f(\tau)$ stands for the in stannous frequency that must be equal to $(f_c + \frac{\Delta f}{f_m} * f_m(t))$. Thus, the previous equation can be expressed as follows: [17]

$$y(t) = A_c \sin(2\pi ft + 2\pi \frac{\Delta f}{f_m} \int_0^t f_m(\tau)d\tau) \quad (2.6)$$

The related modulated signal to $f_m(\tau) = A_m \cos(2\pi f_m \tau)$ and $\int_0^t f_m(\tau)d\tau = A_m \frac{\sin(2\pi f_m t)}{2\pi f_m}$ can be expressed as follows, where this expression represents the basic theory of FM modulation in mathematics: [17]

$$y(t) = A_c \sin\left(2\pi f_c t + \frac{\Delta f}{f_m} * \sin(2\pi f_m t)\right) \quad (2.7)$$

The demodulation of FM signals is similar to that of AM signals, in which both coherent and non-coherent methods can be deployed. In practice, the coherent method is appropriate for FM signals with narrow bands, in which the receiver knows the received signal phase shift. This in turn limits the application of such methods to specific areas only. On the other hand, the non-coherent method is appropriate for both wide and narrow band FM signals [17].

2.2.3. Comparison Between FM and AM Signals

Table 2.1. Below includes a comparison among FM and AM signals.

FM signals	AM signals
The carrier amplitude is constant all over the modulation process	**The carrier amplitude varies all over the modulation process**
The carrier frequency varies all over the modulation process	**The carrier frequency is constant all over the modulation process**
The variation in the carrier frequency is a function of the modulating signal strength	**The variation in the amplitude is a function of the modulating signal strength**
The modulation index value can exceed	**The modulation index value should not**

2.3. MODULATION AND DEMODULATION OF DIGITAL SIGNALS

When the modulating signal is a digital one, the modulation process is considered as a digital one. In comparison with the analog modulation, there are various benefits of using digital modulation, such as the higher security and noise immunity [20]. The common issue among both processes is that the same three features can be modulated on the carrier signal via the digital information. Such features are the Phase-Shift Keying (PSK), Frequency-Shift Keying (FSK) and Amplitude-Shift Keying (ASK). Some examples of such modulation methods are introduced in the following subsections.

2.3.1. Modulation and Demodulation of BPSK Signals

The Binary Phase-Shift Keying (BPSK) is the simplest type of PSK, in which the phase of the carrier signal changes among two values based on the modulating signal bit; 0 or 1. When the carrier signal is $y_1 = \cos(2\pi f_c t)$, its phase is then 0. This signal can be modulated using the formula below: [17]

$$y_2 = \cos(2\pi f_c t + \pi) = -\cos(2\pi f_c t) = -y_1 \tag{2.8}$$

Therefore, when the signal is 1, the BPSK signal is then y_1. Else, it is $-y_1$. Such modulation process is shown in Figure 2.5.

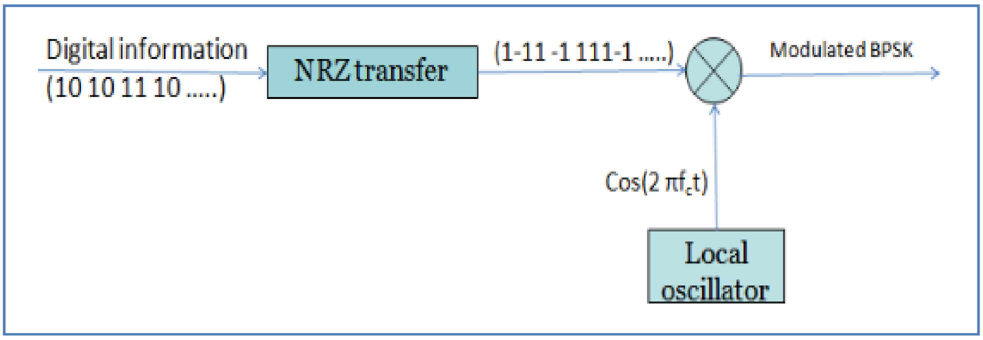

Figure 2.5. Modulation process of BPSK signal [17].

Both the coherent and non-coherent methods can be used in the demodulation of BPSK signals as appeared in Figure 2.6.

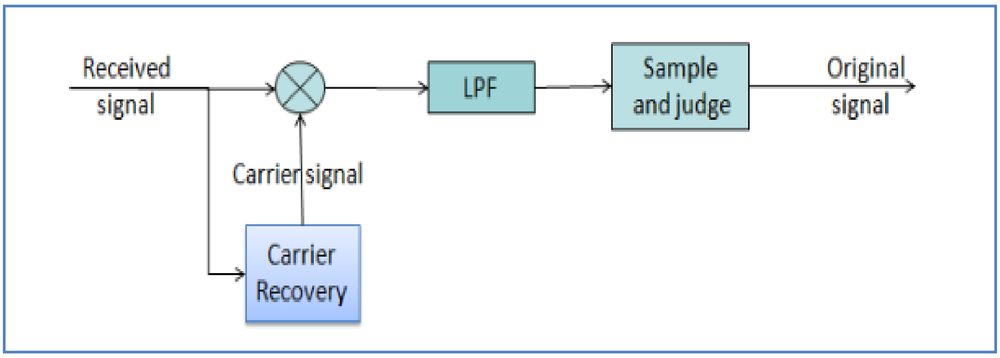

Figure 2.6. Bemodulation process of BPSK signal [17].

In theory, the BPSK signal at the receiver must be in the form below, in which the deployed carrier signal to demodulate the received signal is the same as that of the transmitter one. In this case, the carrier recovery process can be neglected [17].

$$r(t) = Acos(2\pi f_c t + k\pi) \quad K = 0,1 \tag{2.9}$$

However, the phase of the received signal has a phase shift $\emptyset$. Thus, the equation above can be updated to be: [17]

$$r(t) = Acos(2\pi f_c t + k\pi + \emptyset) \quad K = 0,1 \tag{2.10}$$

In practice, the demodulation process cannot be efficient if the receiver deploy the same carrier recovery signal. Thus, the preamble, which is a specific sequence that transmitted always before the actual information, is used to let the receiver knows the phase that must be received. Another method is the use of a carrier recovery circuit, which is mainly deployed to get a carrier signal from the received one as shown in Figure 2.7.

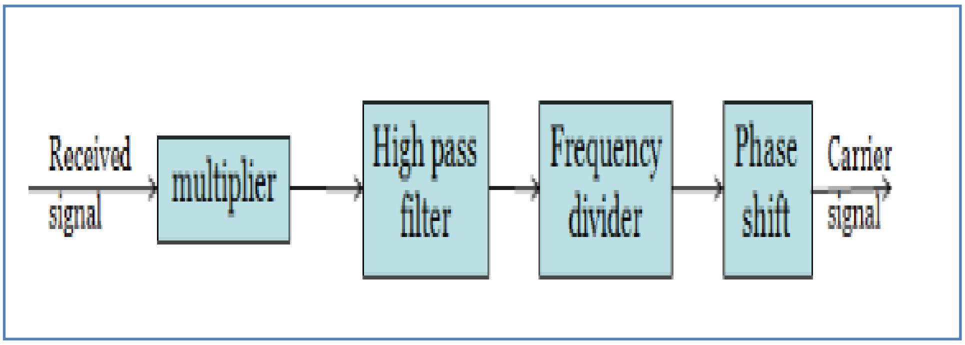

Figure 2.7. Carrier recovery process [17].

2.3.2. Modulation and Demodulation of QPSK Signals

Another PSK modulation method is the Quadrature Phase-shift keying (QPSK) that has twice the BPSK bandwidth efficiency due to the transmission of bits in one modulation symbol [20]. Those two bits lead to four patterns; 00, 01, 10 and 11. Therefore, the phase of the carrier signal changes between four values. In practice, there are two types of phase combinations, namely; π/4, 3π/4, 5π/4 and 7π/4 and 0, π/2, π and 3π/2. The Figure below shows the phase relation, in which circles stand for the first combination, while triangles stand for second combination. In practice, the information signal is sent bit by bit, in which the bits stream is divided into two sequences, where each one has the same modulation process of the BPSK [21].

The QPSK modulation process is shown in Figure 2.8.

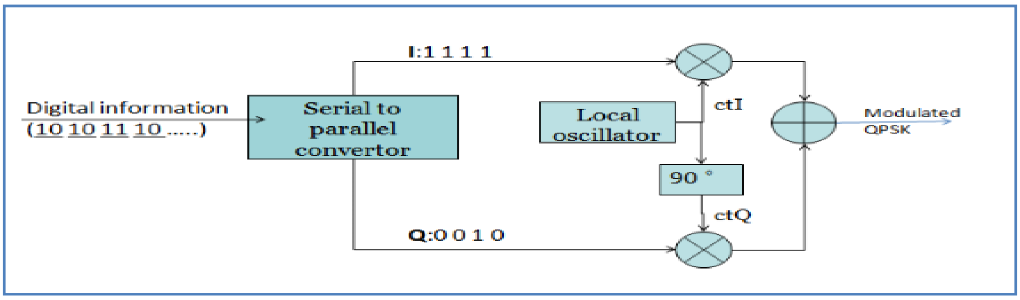

Figure 2.8. QPSK modulation process [17].

As for previous cases, both the coherent and non-coherent methods can be deployed in the QPSK demodulation as shown in Figure2.9, where ctI and ctQ stand for the two carrier signals. In such process, there is a phase shift problem, where it can be solved also using the preamble and carrier recovery circuit.

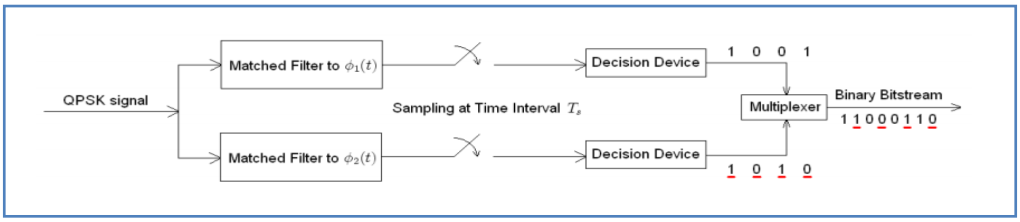

Figure 2.9. QPSK demodulation process [21].

2.3.3. Modulation and Demodulation of FSK Signals

Practically, the FSK is simpler than the PSK, where the simplest FSK type is the binary (BFSK) one. In the BFSK modulation, two frequencies are deployed to present the signal bits 0 and 1 [19]. The frequency of the carrier then changes among two values based on the signal state-binary, where it is mark frequency for binary 1 and space one for binary 0 [22].

Various methods can be used in the BFSK modulation, such as the dependence on the feature that the BFSK signal is formed of two carriers, where the carrier amplitude changes based on the signal bits. Therefore, the ASK can be deployed for each one. Thus, the FSK signal is a sum of two ASK signals. Figure 2.10 below shows the FSK signal production based on the ASK theory.

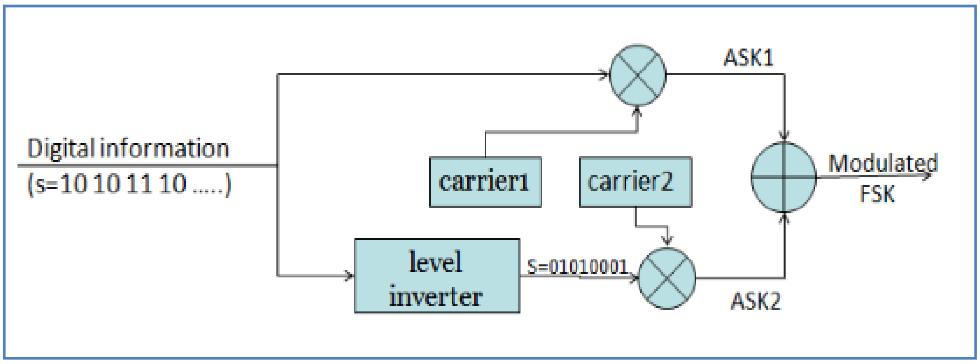

Figure 2.10. FSK signal production [17].

The non-coherent method is deployed in the FSK demodulation as shown in Figure 2.11. In this process, the received signal passes two band pass filters in order to get the related signal. After that the information of two branches are obtained based on the envelop detector. The information is then demodulated based on comparing among the outputs of both branches.

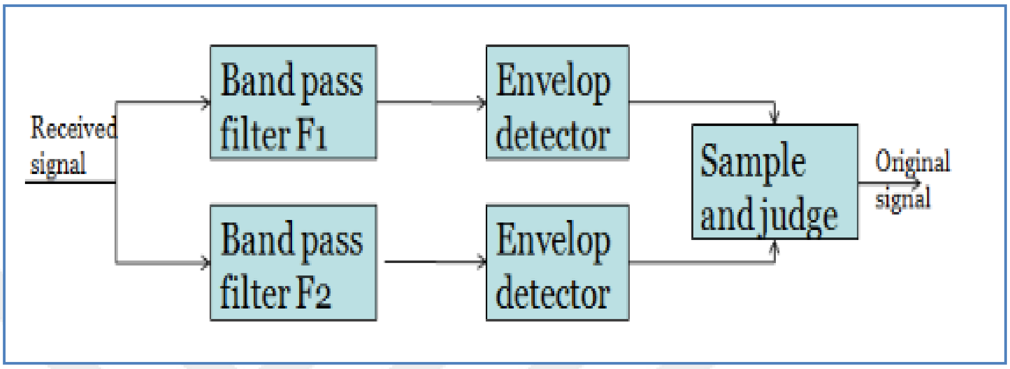

Figure 2.11. FSK demodulation [17].

2.4. FM DEMODULATORS

2.4.1. DSP Based FM Demodulators

Frequency modulation can be explained as a sample traditional modulation method. This chapter concentrates on the application of the analog FM demodulator through DSP schemes.

The DSP algorithms are normally associated with several benefits such as their simplicity during application, their economic nature as well as them being flexibility during usage. In addition, the frequency demodulators have extra benefits such as they offer high-quality robust voice as well as a wide range of SNR exchange. Furthermore, these demodulators also offer effective CDMA DSP processing blocks that are hectic during upgrading. The DSP and FM demodulators discussed here offers room for signal processing [1].

In addition, all the hardware difficulties that are associated with DSP usage are not part of this study. The objective of this research is to obtain a high quality, algorithmic and explanation of different types of DSP FM demodulators.

In the process of evaluating the performance of analog modulation schemes, output to noise signal ratio is uses as the performance measure. Identification of SNR output is not a difficult task since there are several ways in which SNR can be calculated. Basically, there are four different techniques in which SNR can be calculated. Figure 2.12 shows the differential FM detection process.

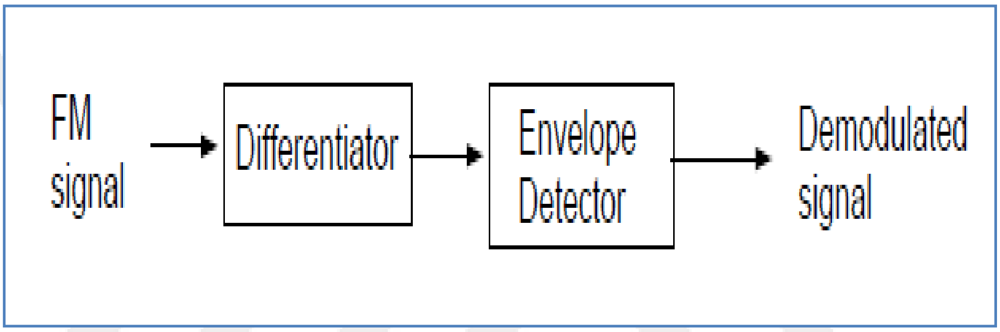

Figure 2.12. Differential FM detection.

2.4.2. FM Demodulation Methods Summarized

In this section various advanced signal processing FM demodulator application. T is a donation of continuous time variance and t_n is a donation of discrete time variance.

2.4.2.1. Discrimination of FM Differentiation

The equation below is an expression of FM signal

$$s(t) = A_c e^{(j2\pi f_c t+\theta(t))} \tag{2.11}$$

$$= A_c e^{(j w_c t+D_f \int_{-\infty}^{t} m(\sigma)\mathrm{d}\sigma)} \tag{2.12}$$

The operation of demodulating the FM signal includes the extraction of the instantaneous frequency originating from all the signals received i.e. $s_r(t) = Re\{s_r(t)\}$.

$$d(s_r(t) = \frac{d}{dt}\left[A_c \cos(w_c t + D_f \int_{-\infty}^{t} \mathrm{m}(\sigma)\mathrm{d}\sigma\right] \tag{2.13}$$

$$A_c\left[w_c + D_f m(t)\right]\left[w_c t + D_f \int_{-\infty}^{t} \mathrm{m}(\sigma)\, \mathrm{d}\sigma\right] \tag{2.14}$$

Frequency and amplitude modulation is included as differentiated as FM signal. The phase of $s(t_n)$ involves the extraction of a DSP implementation of slope discrimination where the $m(t_n)$ message signs are recovered [1]. The IF signal conversion to base signal is one of the various approaches in which this effect can be exempted. This recovery is made through the expression below.

$$m(t_n) = \frac{d}{dt_n}[< \{s(t_n)e^{-w_c}t_n\}] \tag{2.15}$$

It is important to ensure that the phase term is unwrapped initially before the performance of differentiation activities. Hilbert's analytical FM signal transformation is used for the purpose of getting the imaginary section.

$$s(t_n) = s_r(t_n) + js_i(t_n) = s_r(t_n) + js_i(t_n) \tag{2.16}$$

From the equation above $s_r(t_n)$ is a representation of Hilbert signal transformation.

2.4.2.2. Quadrature FM Demodulation

During the process of FM, demodulation implementation is one of the most popular Quadrature detection scheme. The detector has a shift network phase that changes the FM received signal in relation to the instantaneous frequency and applies a product detector in order to identify the real differential phase between output shift network and the signal received [2]. Generally, there is an introduction of 90^o shift of the phase network.

The phase term differentiation involves the indirect simulation of the quadrature model performance. Figure 2.13 shows the Quadrature FM detection scheme.

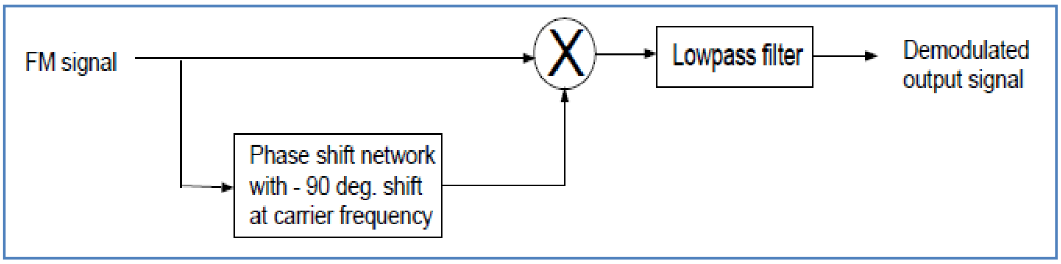

Figure 2.13. Quadrature FM detection.

The technique of quadrature demodulation effect cannot be felt on the phase wrapping due to the fact that there is a difference in version of the extracted phase.

2.4.2.3. Arctangent FM Demodulation

This section describes the arctangent FM demodulation scheme that can be understood through a consideration of the analytical FM signal section as indicated in the equation below.

$$s_r(t_n) = Re\{A_c e^{(\mathrm{j}2\pi f_e \mathrm{t}+\theta(\mathrm{t}))}\} \tag{2.17}$$

$$= A_c \cos[\theta(\mathrm{t})] \cos(w_c t) - \sin[\theta(\mathrm{t})]\sin(w_c t) \tag{2.18}$$

$$= (\mathrm{t}) \cos(w_c t) - y(\mathrm{t})\sin(w_c t) \tag{2.19}$$

From the expression above $x(t)$ *and* $y(t)$ are both imaginary and real sections of the complex baseband FM envelope. From$\frac{\mathrm{y(t)}}{\mathrm{x(t)}} = \frac{\sin[\theta(\mathrm{t})]}{\cos[\theta(\mathrm{t})]} = \tan[\theta(\mathrm{t})]$, message signal can be easily obtained by obtaining the arctangent ratio of the time derivative to real sections of the difficult FM signal envelope.

$$\mathrm{m(t)} = \frac{\mathrm{d}}{\mathrm{it}}\theta(\mathrm{t}) \tag{2.20}$$

$$= \frac{\mathrm{d}}{\mathrm{it}} tan^{-1}\left[\frac{\mathrm{y(t)}}{\mathrm{x(t)}}\right] \tag{2.21}$$

$$= \frac{1}{1+\left[\frac{y(t)}{x(t)}\right]^2} \frac{d}{dt}\left[\frac{y(t)}{x(t)}\right] \qquad (2.22)$$

Figure 2.14 shows the arctangent FM detection scheme.

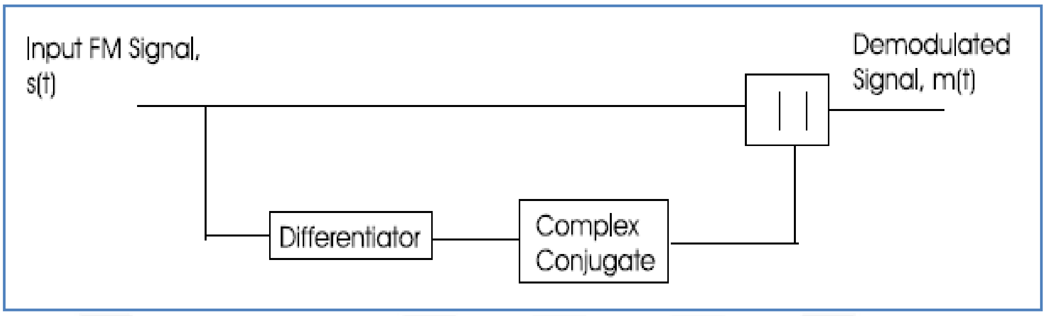

Figure 2.14. Arctangent FM detection.

Differentiation of the message signal is normally the last step together with the activities of the conjugation on the FM analytic sign. Due to the fact that obtaining of the phase is less important, the phase wrapping effect and phase detection of the numerical impression are not common. The sensitivity of this technique depends on the DSP differentiation scheme [2].

2.4.2.4. Phase Locked Loop

After frequency modulated carrier signal has been received, it is extracted using the phase locked loop (PLL) system. Voltage controlled oscillators and a phase comparator are the main components of a PLL system [3].

The importance of this system is that it has a high performance thus ensuring that the dynamic range of the system is extended.

Analog PLL as FM Demodulator

The PLL sinusoidal signs of the input and output can be explained by the following expressions

$$s_i(t) = A_i \sin(w_i t + \mathrm{i(t)}) \tag{2.23}$$

$$s_r(t) = A_r \sin(w_i t + \theta\mathrm{o(t)}) \tag{2.24}$$

At this point, the low pass filter excretes all components with high frequency thus making the phase detector output to act as the sinusoidal differentiating factor between the PLL input and the VCO output.

Where frequency modulated input signals are involved, the PLL's functions include acquiring and tracking of the phase input signal. At times, the PLL is normally locked on the input signal phase making the phase detector output to a time variance signal. The reasons as to why the VCO output frequency differs in the same way as that of the input signal frequency is because the phase output frequency modulator is always available. In order to ensure that the PLL remains locked to the input signal, several conditions need to be met. The conditions can only be understood by having a comparison between the analog PLL operation and the mechanical device [4].

In case there is no recorded weight on the selected area, the pendulum has to remain standing vertically. In such conditions, as light decrease in the input signal frequency results to light increase in the rope weight thus resulting in pendulum deflection. On the other hand, if there is an increase in the rope weight to an extent of causing a more than 90^{o} deflection, the pendulum will always rotate around the shaft.

There are several factors that are associated PLL that includes Lock range and pulls in range. The highest frequency changes that the PLL gains at the input phase is referred to as the pull-in range while the highest frequency change obtained by the PLL in a single cycle both at the output and input PLL frequency difference is the lock range. In this case, the lock range is normally small in size than the pulling range [5].

Daring the designing of the FM demodulator that has PLL characteristics, the frequency deviations should always be smaller than the hold range.

Simulation Models of the Analog PLL

For the purpose of having a valid small phase linear designs the input signal to noise ratio should be set at a higher range than the FM demodulation point. The transfer function of the loop filter from the first low pass IIR filter is as follows

$$F(s) = \frac{st_1 + 1}{st_2} \tag{2.25}$$

From the above expression helps to obtain the closed-loop transfer function of the linear PLL model.

$$H(s) = \frac{K_0 K_d F_{(s)}}{S + K_0 K_d F_{(s)}} \tag{2.26}$$

$$= \frac{K_0 K_d (st_2 + 1)/(t_1 +}{S^2 + \frac{1 + K_0 K_d t_2}{t_1 + t_2} s + \frac{K_0 K_d}{t_1 + t_2}} \; t_2 \tag{2.27}$$

The processes of DSP simulation of a traditional PLL modulation need high-frequency sampling. When the sampling rate is increased, the delay effect, on the other hand, is reduced thus obtaining a higher value from a sampling rate for the purpose of nonlinear PLL simulations.

2.4.2.5. Zero Crossing Detectors

Through measurement of the FM frequency in an instantaneous way, the message signal can be easily obtained. The function of the hand-limiter is to convert the input signals into a square wave through the use of a single stable multi-vibrator that is set at the positive edge. A low-pass filter is then used for the purpose of obtaining the average activity of moon shot conversion to obtain a different signal equal to the intended message signal [6]. The improper mono-shot models, as well as the low pass filter, were the reasons behind the failure of the DSP algorithmic implementation.

CHAPTER 3

RXTX

3.1. RXTX

This pack is a SDR Transceiver that follows in the outstandingly viable line of RXTX Softrock, the most recent being the RXTX V6.3 multi-band handset. The Outfit unit is essentially the RXTX V6.3, yet with its band-specific sections modified and destined (as opposed to the module young lady sheets for these limits from the RXTX V6.3). The unit, in like manner, is bad with the Mobo course of action of extra things, since they depend on the connection and connection blueprints of the RXTX V6.3. The unit comes in five structures, contrasting with five "super-bunches":

1. 160m – covering the 160m band.
2. 80m , 40m – covering the 40m and 80m bands.
3. 40m, 30m, 20m - covering the 40 through 20 meter bands (added 7/12/2010).
4. 30m, 20m, 17m - covering the 30, 20 and 17 meter bands.
5. 15m, 12m, 10m - covering the 15, 12, and 10 meter bands.

When you have chosen your band decision, you can choose the band by tapping on the "Groups" tab at the highest point of any page on the Troupe RXTX site. This will alter these notes (and their segment values) for the chose band choice. The band choice will hold for the term of your "session" on the site keeping in mind in the Group RXTX partition. Get in the propensity for checking the header on every page to make certain your band choice is still basically.

In every variant, the radio has finish recurrence readiness inside the "super-band" (because of the dependable Si570 programmable oscillator), restricted just by the altered/introduced band-particular segments. This implies, for instance, with the 30/20/17m adaptation, the client can work all modes on every one of the three of those ham groups, anyplace in those groups, subject just to the constraints of their permit and the SDR programming being utilized. Truth be told, the unit can likewise get any HF motions inside the introduced "super-band".

The unit gives an Atmel AVR Microcontroller on-board, customized to go about as a USB gadget and introduced in a galvanic partner detached segment of the board. A transform from the prior handsets is the expansion of jumpers on board to switch the ring and tip assignments of RX and TX I and Q signals.

Another much needed development from the before models is that all associations with the outside world are taken care of through on-board jacks (furnished with the pack).

The pack is offered with the greater part of the parts important to fabricate it for any of the five conceivable forms. Thusly, the developer will dependably have a few sections "left-over" toward the end of the construct. These archives contain band-particular Bill of Materials postings for every variant (notwithstanding the Bill of Materials for the parts of the radio that are not band-particular.

<u>Block Diagram</u>

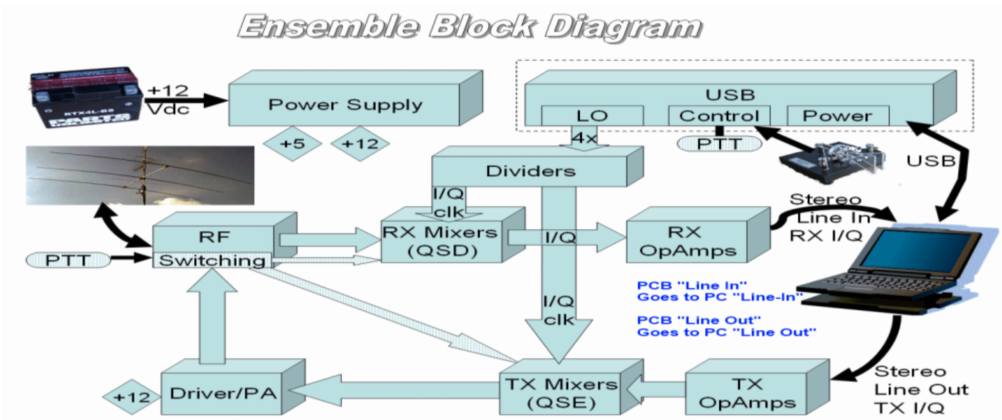

Figure 3.1. Block diagram.

RX Circuitry

The transceiver has a typical reception antenna terminal and RF way which is exchanged between the RX Band pass channels (default) and the TX low-pass channels, by means of hardware in the RF I/O and Switching Stage. The exchanging is performed because of the/PTT motion from the microcontroller, as chose by the SDR program on the PC. In the RX chain, the approaching RF is band-pass separated in T5/L4/C39, with the RF yield at T5's optional in insect stage.

The ant phase RF signs of T5 are coupled into the RX Mixer Stage by means of R53 and R54. The blender chip (really a commutating switch, timed by the two QSD Clock signals from the Dividers Stage) yields the item and distinction signs of the approaching RF against the QSD clock. The impact is to down-change over the approaching RF into its quadrature analogs at frequencies extending from 0 to around 100 kHz.

These quadrature sets from the RX Mixer, indistinguishable in all regards with the exception of stage, are encouraged to the RX Op Amp Stage for enhancement and sifting into the sound and infra-sound range and conveyed to the RX yield stereo sound jack, J4, to be contribution to the PC's STEREO info (line-in or Mic).

The PC's sound card plays out a transformation of the two simple signs ("I and Q") to an advanced representation, which is then worked upon by modules of the SDR program to play out the numerous "radio" capacities, for example, demodulation, separating, AGC, and so forth., that are anticipated from a fine beneficiary.

To do this, it significant that the soundcard being utilized backings STEREO info. The testing rate, quality and specs of the soundcard will Figure out if and how well the PC can function flags whose recurrence is either side of the inside recurrence. Normal sound card examining rates for this data transfer capacity are 48 kHz, 96 kHz, and 192 kHz. These each relate to the capacity to bolster SDR handling of "lumps" of transfer speed of 48 kHz, 96 kHz, and 192 kHz, every piece fixated on the middle recurrence (CF), with "wings" on either side of the CF that are one-a large portion of the testing rate in width.

TX Circuitry

The transmit usefulness is basically the turnaround of the RX usefulness. In the PC, instead of demodulating information I and Q motions as in the RX, the PC tweaks the advanced signs (from the mouthpiece or a keyed module. for instance) into simple I and Q (infra) sound STEREO yields, normally yield to the line-out jack on the soundcard. The I and Q signals at the line-out are in quadrature (indistinguishable in all regards spare stage) and show up at stereo jack J3. There are bolstered to the TX Op Amp Stage. This unitary pick up stage interprets the I and Q signals into four equivalent signs, at 0, 90, 180, and 270 degrees of stage.

These four signs from the TX Op amp Stage are coupled by means of R25-R28 to the TX Mixer Stage. Pretty much as the RX Mixers "blended" approaching RF with the QSD clock (focus recurrence) signs to create (infra)audio signals, the TX Mixer does the switch, "blending" balanced (infra) sound signs with the QSE RF clock signs to deliver up-changed over RF yields that are analogs of the TX I and Q inputs. This regulated RF yield of the TX Mixer (otherwise known as "Quadrature Sampling Exciter", or "QSE") is coupled by means of T2, C20, L1, and C21 to the Driver/PA arrange.

The Driver/PA organize shapes and enhances the adjusted RF yield from the TX Mixer stage and encourages the outcome to the reception apparatus way as exchanged by the PTT exchanging hardware. The exchanging hardware actuates the Driver/PA stage and powers a S12 line to high (roughly 12 V dc), to allow exchanging an outside intensifier. This stage will convey around one watt of yield into a 50 ohm stack.

3.2. AUDACITY

3.2.1. Introduction

Audacity is a simple to utilize yet capable sound recording and altering bundle. It is additionally allowed to download and utilize (see Appendix A – Downloading and Installing Audacity and the LAME MP3 encoder toward the end of this report for data on the most proficient method).

Daringness empowers you to record your voice, alter your recording to adjust any slip-ups you may make, and to join sound recordings from different sources, for example, such as interviews, music, or other sound recordings you may have. Daringness additionally empowers you to trade your recording as a MP3 document, and as a result of this it is perfect for producing podcasts.

3.2.2. Getting Started

To Start Audacity:

1. Click on the Start button from Windows.
2. Choose Audacity at the Programs menu.

After you reach the point when Audacity begins, we will see the accompanying screen As in Figure 3.2.

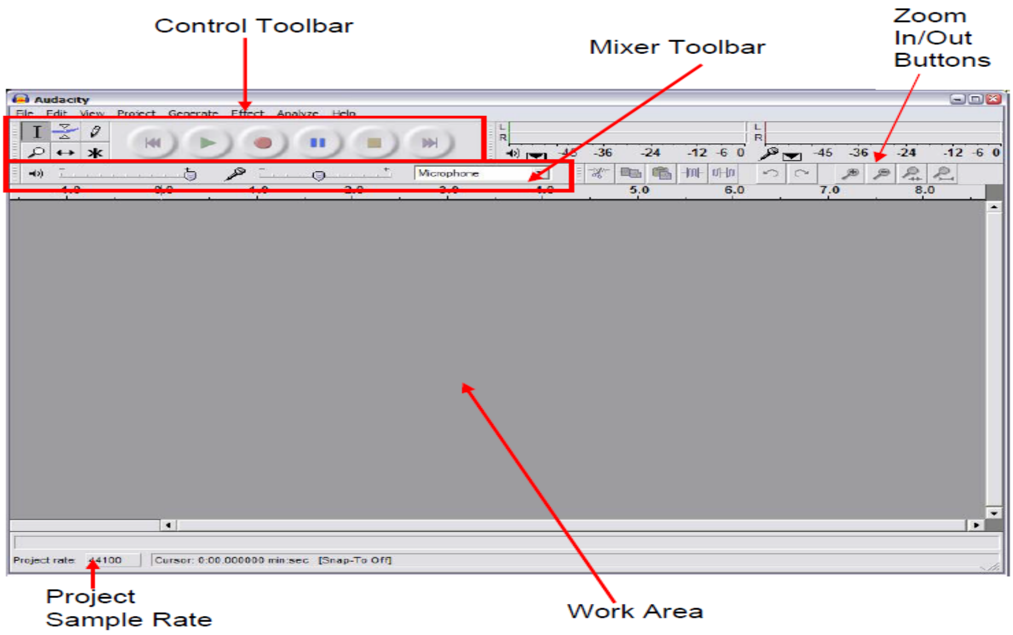

Figure 3.2. The main Audacity program screen.

1. Recording, Saving and Editing Your Audio.

3.2.2.1. Recording Your Audio

When we start to record we sound, we should do the do the following :

1. Ensure your mouthpiece and speaker (or headset if utilizing one) are accurately associated with PC - check your amplifier or PC direction manual on the off chance that we are uncertain how to do this.
2. Must Prepare something to say.
3. Check that the Project Sample Rate is set at 44100, As in Figure 1.
4. If we are using loudspeaker, you may need to turn these down to prevent feedback.

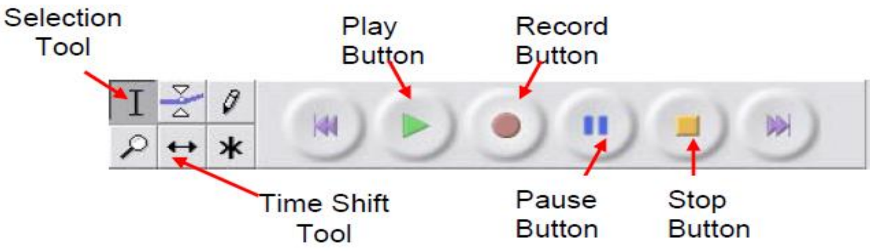

Figure 3.3. The control toolbar.

To start recording:

1. Tap on the Record Button from the Control Toolbar, As in Figure 3.3.
2. Begin saying what we wish to state.
3. Press the Pause Button from the Control Toolbar on the off chance that we wish to delay recording anytime – squeeze it again to beginning register.
4. Press the Stop Button from the Control Toolbar when we have completed your recording.

When you have completed recording, we ought to see the voice spoke to as a "waveform", As shown in Fig. 3.4 underneath.

Waveform will be visible as an "audio track". Audacity allows we've got multiple recordings in multiple audio tracks.

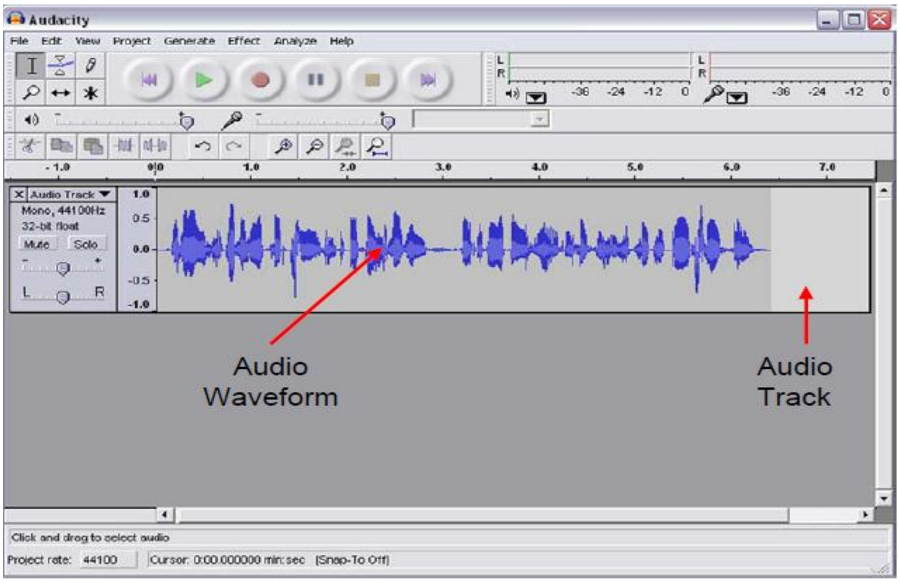

Figure 3.4. Recorded audio shown as an Audio Waveform in an Audio Track.

3.2.2.2. Listening to Your Recording

We will in all likelihood need to listen back to recording to watch that we are content with the nature of the recording, and with execution. we may wish re-record it

To listen to recording:

1. Click on the Play Button in the Control Toolbar As in Figure 3.3.
2. Click on the Pause Button or stop Button from the Control Toolbar to stop or pause the playback.
3. we can adjust the volume at which we listen to the recording using the Playback Volume slider from the Mixer Toolbar, As in Figure 3.5.

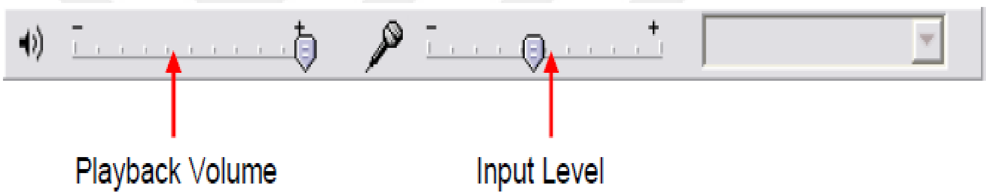

Figure 3.5. The mixer toolbar.

3.2.2.3. Adjusting the Recording Level for Optimum Quality

To ensure that you record your sound at the right recording level to get the best quality – a recording level causes mutilation, while too low a recording level presents an excessive amount of foundation clamor and murmur. Nor are attractive and make the recording hard to tune in to.

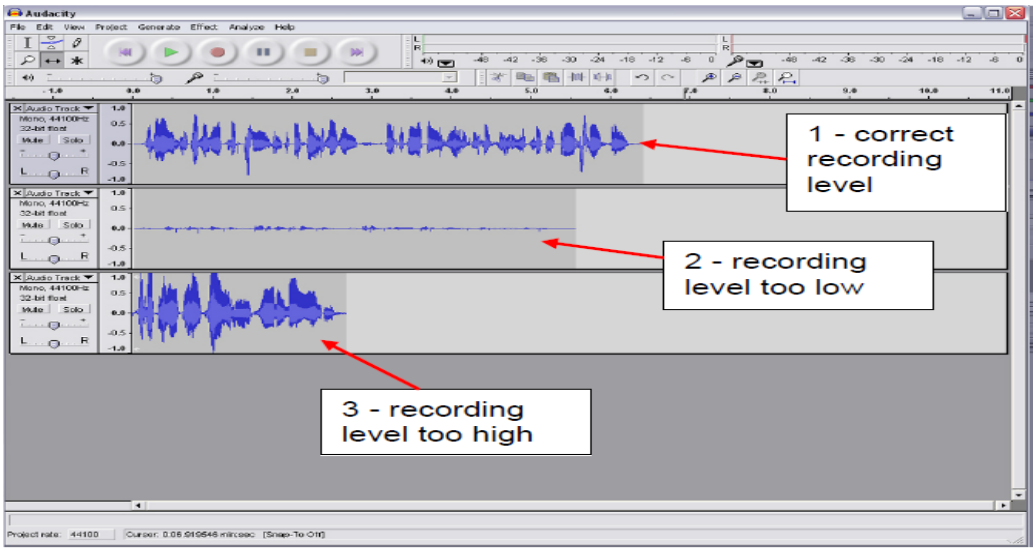

Figure 3.6. Waveforms showing recording levels that are correct, too low and too high.

If recording level is too low:

1. Increase the recording level by adjusting the Input Level slider from the Mixer Toolbar, As in Figure 3.5.
2. Sit closer to the microphone.

If recording level is too high:

1. Decrease the recording level by adjusting the Input Level slider from the Mixer Toolbar, As in Figure 3.5.
2. Sit away from the microphone.

3.2.2.4. Saving Your Work

Now, we have to spare the venture interestingly. It is vital that you spare your work consistently to abstain from losing any work in the event that any problem arises.

1. Pick File, Save Project in the fundamental menu.
2. Spare the Audacity Project in a fitting area.

3.2.2.5. Editing Your Audio

We will perpetually wish to alter the sound somehow after we have recorded it. This might be to evacuate undesirable segments, record extra material, or fuse sound from different sources, for example, mood melodies, interviews and so on. Daringness gives a far reaching scope of altering offices, which are quickly presented here.

To remove an unwanted part of the recording:

1. Click on the Selection Tool in the Control Toolbar as in Figure 3.3.
2. Click and drag the part of the audio waveform we want to delete from the audio track.
3. The selection should appear as in Figure 3.7.

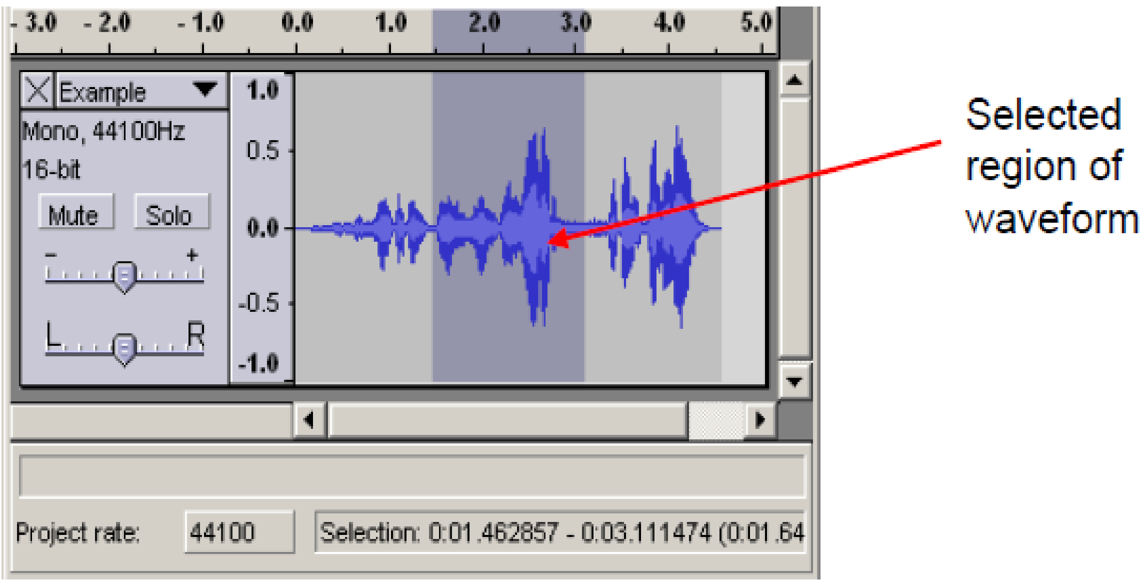

Figure 3.7. Selecting part of the waveform for deleting and/or editing.

1. Press the Delete key in the keyboard, For deletion from the main menu.
2. The unwanted audio is removed, and the audio either side of.

<u>Inserting audio into your recording</u>

We might need to include additional sound amidst the unique recording – this must be done in a few steps:

1. Utilizing the Selection Tool (or "cursor"), click in the waveform at the point from where we need to include the new sound.
2. Tap on the Record Button from the Control Toolbar to begin the new recording.
3. Dauntlessness consequently makes another track and begins to record from the position we tapped the cursor.
4. Press the Stop Button when we have wrapped up the new piece of sound.
5. We ought to see the new recording in another soundtrack, as in Figure 3.8 underneath

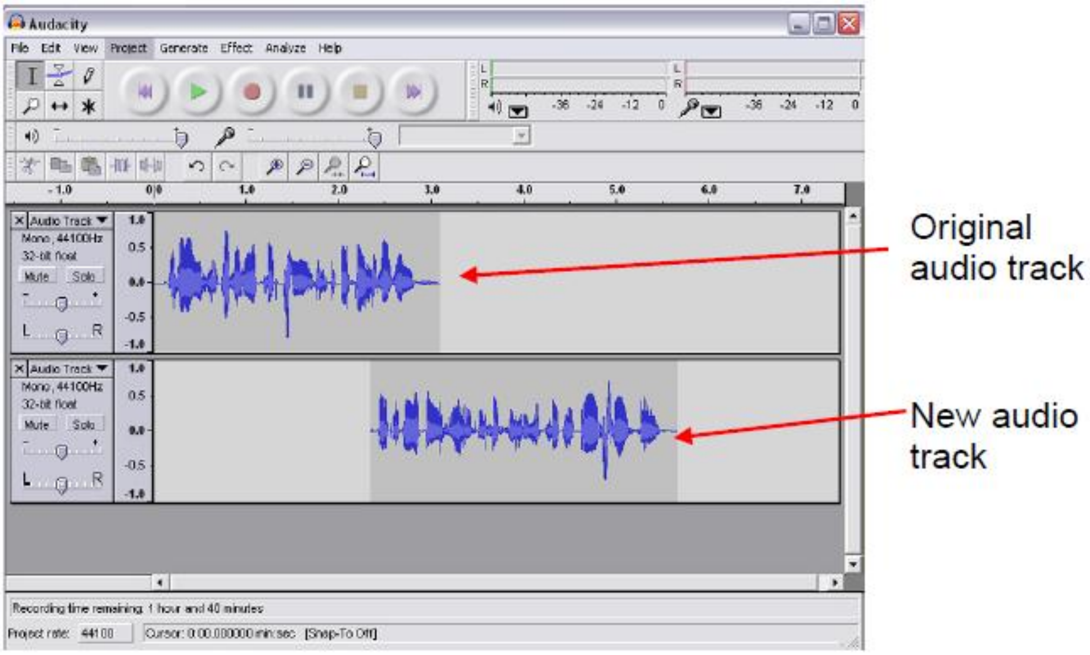

Figure 3.8. Overlapping original and newly recorded audio tracks.

1. At the moment the two tracks overlap and if we press play, the two recordings mix together in the middle. To complete the insertion of the new audio, We want to move the second half of the original recording.
2. Go to the original track, and highlight the waveform from the point the second track starts to the end as in Figure 3.9.
3. Choose Edit, Split from the main menu. This will remove the highlighted section of the original track and add it to a new track in the bottom from the work area, As in Figure 3.10.

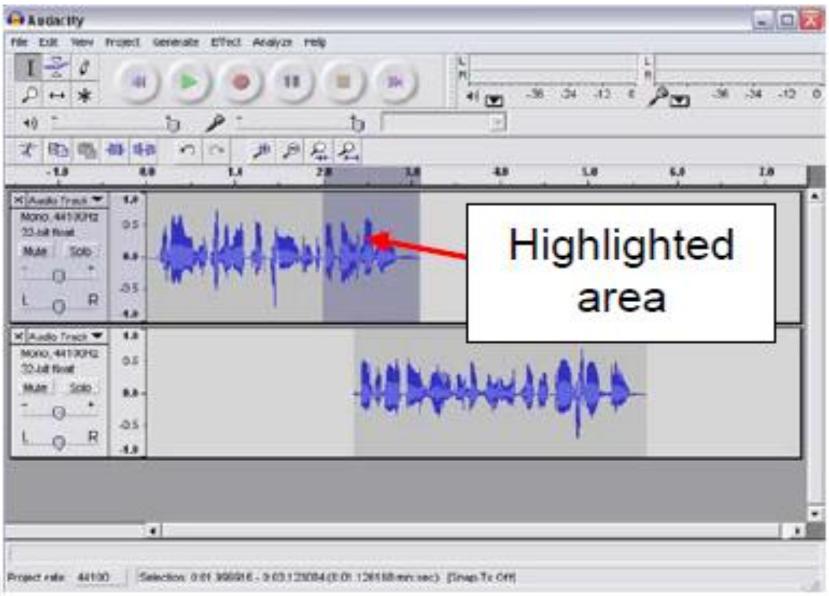

Figure 3.9. Highlight the waveform from the point the second track starts to the end.

Figure 3.10. Remove the highlighted section of the original track and add it to a new track.

1. Presently select the Time Shift Tool in the Control Toolbar and snap and drag the waveform on the new track at the base so it lines up with the end of the waveform on the track above, as in Figure 3.11 beneath.

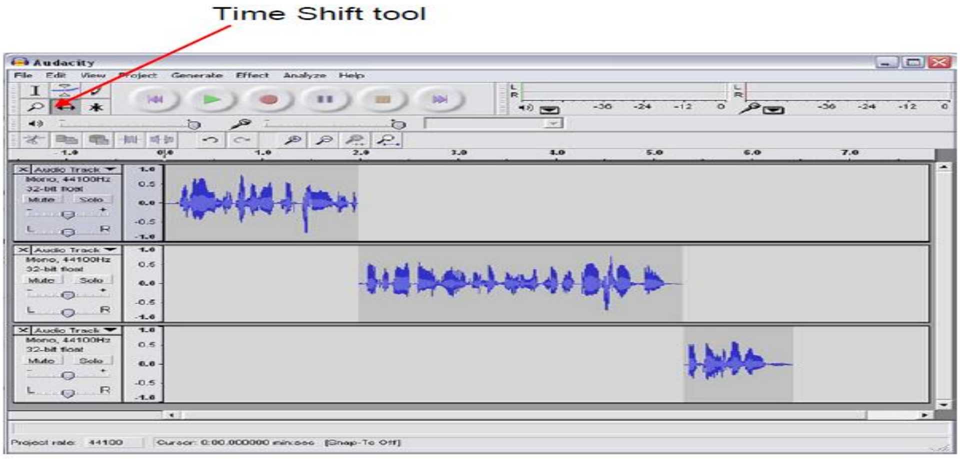

Figure 3.11. Audio tracks re-aligned with time shift tool.

3.2.2.6. Adding Other Audio

On the off chance that we need to include sound that we have not recorded straightforwardly with the amplifier, for example, music, or material that we have recorded somewhere else, for example, a meeting, we should import a sound document. A few of points to note on importing audio

To add an audio file to your Audacity project:

1. Pick Project, Import Audio in the principle menu.
2. Select the document we wish to import and Audacity include it as another track at the base of the work area.
3. In the event that the track you have imported is from stereo, as will regularly be the situation with music, it look unique in relation to the track we have recorded with the mouthpiece, having two waveforms - one for stereo left and one for stereo ideal as in Figure 3.12 below.

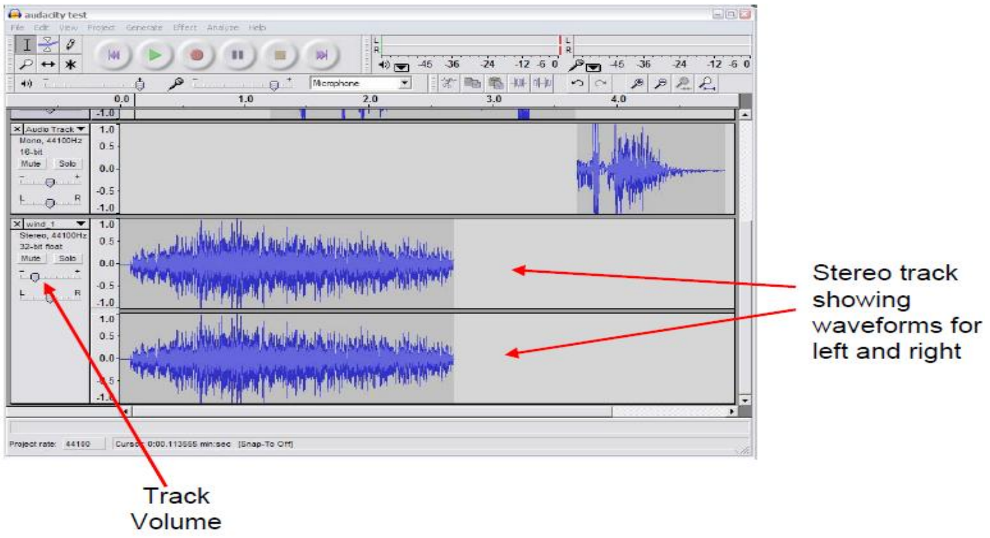

Figure 3.12. A stereo wave form imported into Audacity.

3.2.2.7. Adjusting the Volume of Different Audio Rracks

In the event that we have consolidated sound in various diverse sources, we may find that we have to change their relative volumes to have the capacity to hear them all legitimately .

To adjust the volume of an audio track:

1. Snap and drag the Track Volume slider for the track we need to alter, As in Figure 3.12 above.

3.2.2.8. Fading in and Out

We may need to truncate this new solid since it is longer than we require. Instead of simply slicing of the end which doesn't sound awesome, we can without a doubt make this track obscure away to nothing, which is known as a “fade out”.

To create a fade out:

1. Highlight a region of waveform toward the end of the track where we wish the sound to become diminish.
2. Choose Effect, Fade Out on the principle menu.

3. We will see the waveform steadily get smaller toward, and on the off chance that we hear it out, we will hear it blur away to nothing.
4. We can create the opposite effect by choosing Effect, Fade In on the main menu, to create a gradual “fade in” at the beginning of an audio track if we wish.

3.3. ARCHIVING YOUR WORK AND EXPORTING YOUR PODCAST

In the midst of work, you should have been reliably saving the wander using Save Project from the File menu. This is indispensable to keep all your work set up, however this can't be played by some other programming than Audacity, and the wander reports can be enormous. It is to a great degree important to keep these wander reports if you have to come back to or re-changes your podcast. Likewise, as with whatever other basic work you do on a PC, it is exceedingly reasonable that you make a support of this work in the conventional way (by keeping in touch with a CD, or replicating to a mutual system drive for instance).

3.3.1. Archiving Your Work as an Uncompressed WAV File

We furthermore earnestly suggest that you make an 'account variation' of recording by conveying it as a WAV archive. This is uncompressed, astonishing sound, that can be opened, played and adjusted by a broad assortment of sound ventures.
To make an uncompressed chronicle rendition of recording:

1. Pick File, Trade As WAV on the central menu.
2. In the event that we have made a few sound tracks over the span of your recording, you will see the accompanying message, as appeared in Figure 3.13 below.

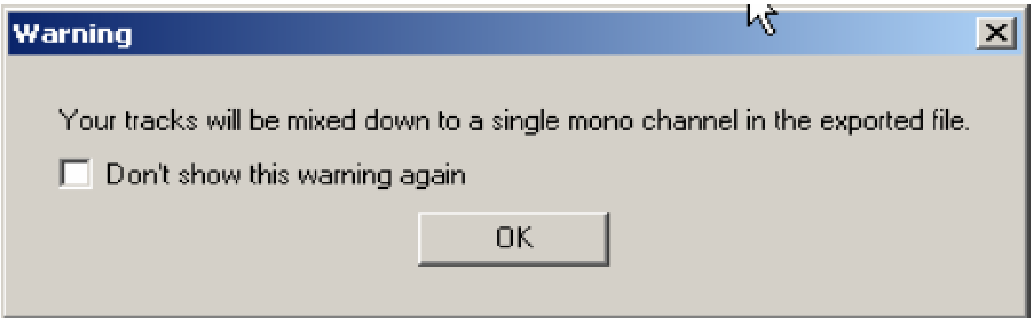

Figure 3.13. Warning message indicating that all separate tracks will be combined on export.

1. Tap on OK in the occasion that we see the above notice message.
2. Spare WAV file to a fitting area utilizing the record spare exchange box.

3.3.2. Exporting Your Audio as an MP3 File

To make a podcast that anyone can listen to you, have to exchange sound as a MP3 file.

With the goal for Audacity to send out your recording as a MP3 file, you should have introduced the LAME MP3 file, as portrayed in Appendix An underneath. On the off chance that you have not yet done this, take after the means laid out in Appendix A now.

3.3.3. A Note on File Size and Compression

MP3 file uses what is known as "weight" with a particular true objective to keep their report sizes pretty much nothing, and this is the reason they have ended up being so notable for use as podcasts. By analyzing, MP3 document can be as much as 20 times littler than an equivalent CD soundtrack of a comparative length.

Notwithstanding the way that this ability to pack the measure of MP3 records is outstandingly useful for Internet assignment, it does similarly incorporate some real drawbacks, which is sound quality. There are a couple of parts that effect how much weight that we can use in making a MP3 document, and subsequently the last record size and sound quality. The most basic variable we can control while using Audacity is to change what is known as the bit-rate.

Before you convey your sound as a MP3 document in Audacity, you along these lines need to guarantee you have picked an appropriate piece rate for last recording. If your last podcast will simply contain talk, you can remain to set a lower bit-rate for your yield than you would in case you're recording was chiefly included music (the last of which needs higher sound quality to be seen as satisfactory).

The following bit-rates are strongly recommended as a matter of policy:

Type of Podcast	**Bit-rate**	**File size per minute**
letter	**64kbps**	**500K**
Music	**128kbps**	**1Mb**

3.3.4. Setting the Bit-Rate in Audacity

To set the bit-rate in Audacity before sending out your MP3 record:

1. Choose Edit, Preferences from the primary menu.
2. You ought to see the accompanying screen as appeared in Figure 3.14 underneath.
3. Tap on the File Formats tab.
4. Pick the fitting piece rate beginning from the drop list box.
5. Remember – 64 kbps for discourse, 128 kbps for music.

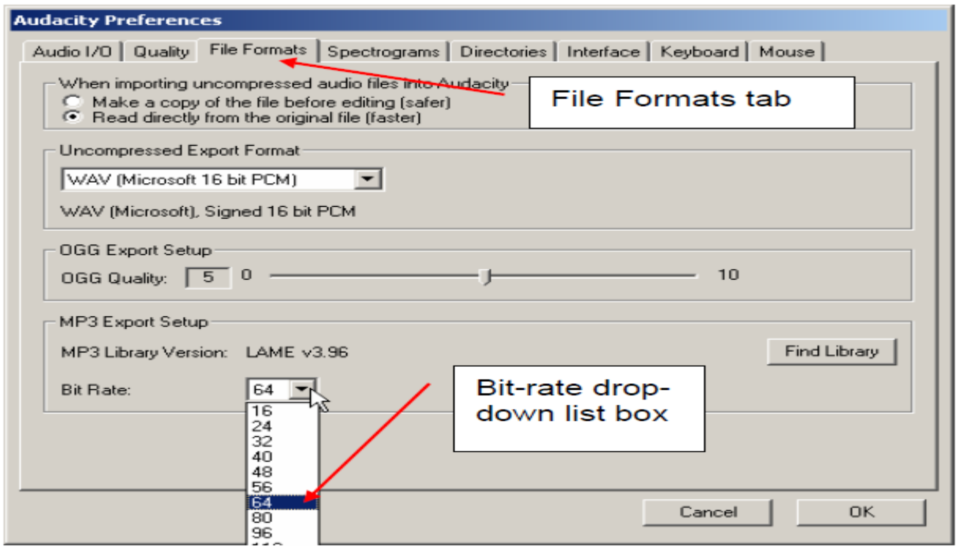

Figure 3.14. The Audacity preferences screen.

CHAPTER 4

FM MODULATION

How to generate frequency modulation (FM) in MATLAB? Prior to that we should recognize what is FM (frequency modulation). In FM, frequency of the carrier signal flag having high frequency is fluctuated as per the quick of the balancing signal having low Frequency . Frequency modulated signals are generally utilized as a part of TV and radio transmission systems. FM signs can be effectively plotted utilizing straightforward MATLAB capacities. The MATLAB code is demonstrated as follows, the utilization of specific code is additionally given as remarked shape (%Comment field). The given MATLAB program is capable of accepting two input frequencies and modulation index from the user.

Mathematical representation of FM Signal :

$$e_m\ (t) = E_m sin(w_m t)$$
$$e_c(t) = E_c sim(w_c t)$$
$$e(t) = E_c sin\big((w_c t) + m\ sin(w_m t)\big)$$

4.1. FM MODULATION

The modulator inputs the audio file as a1.wav and frequency modulates a carrier by phase modulating the carrier using the integral of the message signal. The resulting modulated waveform is recorded as FM.wav. Then this file is used for transmitting. After receiving by the corresponding Softrock the IF file is recorded as rec_FM.wav. The rec_FM.wav file is taken by the demodulator code and demodulated using a differentiator based algorithm which looks the difference between angles for FM modulation. The demodulated signal is recorded as as2.wav. This recording then be listened and inspected using the software Audacity.

Codes

```
% FM mod with MUSIC by B. ERKAL 2016
clear all;

% sound file loading (4Khz mono (8KSps))
[iff1 , afs]=audioread('a1.wav');
[y1,~]=size(iff1);

yu1=upsample(iff1,6);% upsample x6 (8x6=48Khz)
yu1=filter(fir1(100,4e3/24e3),1,yu1);
A=max(abs(yu1));% message signal max. amplitude
yu1=yu1./(1.01*A);

fs=48e+3;% sampling frequency
ts=1/fs;% sampling interval
t=0:ts:10-ts;% time axis

% message signal integral
ati=0;
yu1=0.5*yu1;
[y,~]=size(yu1);
at(1:y,1)=0;
for i=1:1:y
   ati=ati+yu1(i,1);
   at(i,1)=ati;
end;
% Kp=48K and Amax=0.5 result in deltafmax=3.9K and BTFM=18K
% (BTFM=2*(deltaf+BWm)=~2*(4K+5K)=18KHz and Beta=delta/BWm=0.78)

% carrier parameters
C=1;fct=12e+3;tetac=0*(pi/180);
% FM IF signal (12KHZ)
```

```
m=C*cos((2*pi*fct)*t+tetac+at');

% IF signal is recorded in wav file
% IF normalized
m=m./(20*max(m));
audiowrite('FM.wav', [m'], 48e+3);
```

4.2. FM DEMODULATION

The demodulation simulates a balanced frequency discriminator. A simple frequency discriminator. The idea is to design a low pass filter so that the frequency range is within the roll off of the filter centered around its cutoff frequency. This filter attenuates the signal based on its instantaneous frequency. That result is followed by an envelope detector similar to the one used to demodulate AM signals.

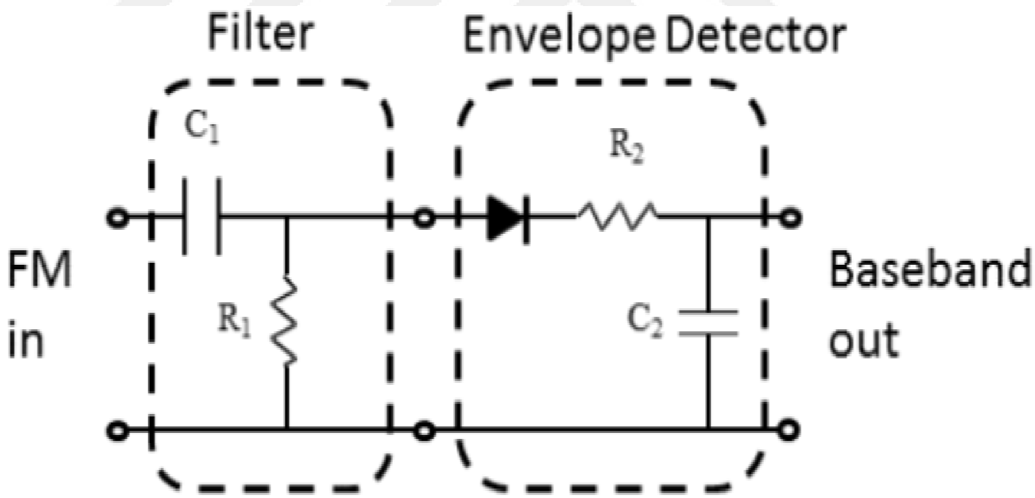

Figure 4.1. Simple frequency discriminator.

In the simulation we use a balanced frequency discriminator with full rectification in the envelope detectors

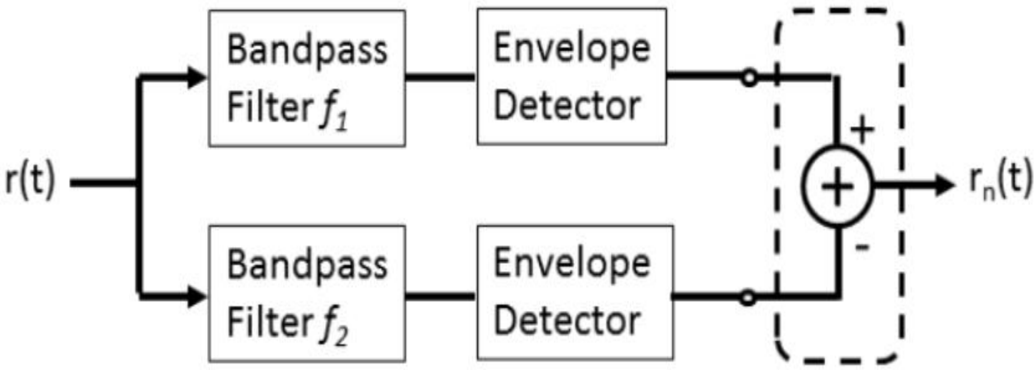

Figure 4.2. Balanced frequency discriminator.

The balanced frequency discriminator uses two band pass filters with different center frequencies. The center frequencies correspond to the minimum and maximum signal frequencies. To get the frequency discrimination, each band pass filter output is envelope detected and then differenced. The mathematics for these is indicated near the question marked lines in the source code below.

Codes

```
% FM demod by B. ERKAL 2016
clear all;

% FM RF file loaded
[iff , afs]=audioread('rec_FM.wav');
[y,~]=size(iff);
% complex shift frequency entered here
w=2*pi*(-12e3);

% samples are normalized
norm=max((iff(1:y,1).^2+iff(1:y,2).^2).^0.5);
iff=iff./norm;

%complex transformation
iffc=iff(:,1)+1i*iff(:,2);
```

```
% time axis created
t=0:1/afs:(1/afs)*(y-1);
% complex shift operation for zero IF
iffc=iffc.*exp(1i*w*t');
% zero IF cut at 6 KHz
yu=filter(fir1(128,6e+3/(48e3/2)),1,iffc);
% zero IF normalized
yu=yu./(20*max(abs(yu)));

% zero IF written to wav file
yl=real(yu);
yr=imag(yu);
audiowrite('as1.wav', [yl,yr], 48e+3);

% FM dedection
% unwrapped angles are calculated
%(unwrap is necessary for angles exceeding pi radians)
tetat=unwrap(atan2(yr,yl));
% derivative of angles give demodulated audio
yout=(tetat(2:y,1)-tetat(1:y-1,1));
% demodulated audio is cut at 5Khz
yout=filter(fir1(128,5e+3/24e+3),1,yout);

% normalize audio and write the result to wav file
yout=yout./(1.1*max(abs(yout)));
audiowrite('as2.wav', yout, 48e+3);
```

CHAPTER 5

CONCLUSION

This project investigates and analyzes the transmission and reception of FM signals using the Softrock Ensemble RXTX SDR transceiver. Such transceiver composed of SDR hardware, Running SDR software and a stereo soundcard. As well, its diagram composed of two main components; TX circuitry and RX circuitry. The TX circuitry is responsible for the production of signal waves for transmission into the space, while in the RX circuitry, the transmitted radio waves are received and amplified, where the signal is then extracted from them in the demodulation process using software routines written in Matlab. Audio amplifiers are used to amplify the signal, which then fed to the speaker to reproduce sound waves. The HDSDR is used to capture radio signals, in which its outputs are processed and edited using the Audacity audio editor and recorder software. Then the FM demodulation takes place using Matlab.

FM signals are generated and plotted using the MATLAB with considering that its carrier signal with high frequency is fluctuated as per the quick of the balancing signal, which has a low frequency. The inputs of the written MATLAB program are two input frequencies and a modulation index from the user. There are two main stages in the developed system; FM modulation and FM demodulation.

The FM modulation represents changing the frequency of the carrier signal based on the signal intensity with keeping its amplitude constant. In the FM modulation stage, the recorded audio file is imported as a1.wav, then the message signal integral is deployed to modulate the carrier signal. The modulated signal is then saved as FM.wav and used for transmission. When the related Softrock receives, the IF file is recorded as rec_FM.wav to be demodulated in the FM demodulation stage with the use of a differentiator based algorithm. Such algorithm is based on finding the difference among the angles for FM modulation. The resultant demodulated signal is then filtered, normalized and saved as as2.wav. The Audacity is then deployed to listen to and inspect such signal.

For further investigations in the future, other modulation methods can be deployed in the developed system to be studied and analyzes. For example an FSK modulation scheme where a preamble can be added in the transmitter signal can be used to communicate digital data. The receiver also can be modified to check such added preamble, in which when the preamble is right, it continues in receiving the signal, else, it rejects the transmission. As well, a checksum can be added to the receiver to investigate the validity of the data.

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