TRANSMIT AND RECEIVE OF SSB AND DSB-AM SIGNALS USING SOFTROCK SDR AND MATLAB

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TRANSMIT AND RECEIVE OF SSB AND DSB-AM SIGNALS USING SOFTROCK SDR AND MATLAB

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I certify that in my opinion the thesis submitted by Salama Ghaith. S. ALGHIRYANI titled "TRANSMIT AND RECEIVE OF SSB AND DSB-AM 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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This thesis is accepted by the examining committee with a unanimous vote in the Department of Electrical and Electronical Engineering as a master thesis. December 26, 2016

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The degree of Master of Science by the thesis submitted is approved by the Administrative Board of the Graduate School of Natural and Applied Sciences, Karabük University.

Prof. Dr. Nevin AYTEMİZ ....

Head of Graduate School of Natural and Applied Sciences

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Salama Ghaith. S. ALGHIRYANI

ABSTRACT

M. Sc. Thesis

TRANSMIT AND RECEIVE OF SSB AND DSB-AM SIGNALS USING SOFTROCK SDR AND MATLAB

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In this study, the DSB-SC modulated signals as well as DSB-WC signals are generated by multiplying the message with a carrier, while the SSB ones are generated by filtering out the lower sidebands of the DSB-SC modulated signals. At the receiver, the audio signal is retrieved from the modulated wave in a process called demodulation. All the modulation and demodulation takes place in software using Matlab scripts written with SDR techniques.

Therefore, this project discusses and compares the modulation and demodulation of SSB and DSB-SC AM signals using the Softrock SDR and MATLAB programs. Results revealed that the DSB-SC modulating system is the most efficient one, in which there is no lost power in the carrier signal. Conversely, the SSB modulating system allows the transmission of more information over the same channel based on hopping off the duplicate sideband.

**Key Words : ** DSB-SC, AM, SSB, SDR, Matlab, Softrock.
Science Code : 905.1.067

ÖZET

Yüksek Lisans Tezi

SOFTROCK SDR VE MATLAB'I KULLANARAK SSB VE DSB-AM SİNYALLERİNİN ALIMI VE YAYIMI

Salama Ghaith. S. ALGHIRYANI

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Bu çalışmada DSB-SC ve DSB-WC AM sinyalleri bir taşıyıcıyla çarparak elde edilmiştir. SSB sinyalleri ise DSB-SC modülasyonlu sinyallerin alt veya üst yanbantlarını filtrelemelekle oluşturulmuştur. Alıcıda ses sinyali modüle edilmiş dalgadan demodülasyon denen bir işlemle elde edilir. Bütün bu modülasyon ve demodülasyon işlemleri SDR teknikleri kullanılarak yazılmış Matlab kodları ile yazılımda gerçekleşir. Bu nedenle, bu proje SSB ve DSB-SC AM sinyallerinin Softrock SDR ve MATLAB programları kullanarak modülasyonu ve demodülasyonunu kıyaslar ve tartışır.

Sonuçlar göstermiştir ki, taşıyıcı sinyalde hiçbir güç kaybı olmayan DSB-SC modülasyon sistemi en verimli olanidır. SSB modülasyon sistemi ise tekrar eden yanbantın kesilmesi sayesinde aynı kanal üzerinden daha fazla bilgi akışının sağlanmasına izin verir.

Anahtar Kelimeler : DSB-SC, AM, SSB, SDR, Matlab, Softrock**.**
Bilim Kodu **:** 905.1.067

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ABBREVIATIONS

AM	: Amplitude Modulation
FM	: Frequency Modulation
PM	: Phase Modulation
QPSK	: Quadrature Phase-Shift Keying
SDR	: Software Defined Radio
RF	: Radio Frequency
IF	: Intermediate Frequency
ODMA	: Opportunity Driven Multiple Access
CW	: Continuous Wave
USB	: Upper Side Band
LSB	: Lower Side Band
DSB	: Double Side Band
HSDR	: High Definition Software Defined Radio
BT	: British Telecommunications
DUC	: Dynamic Update Client
UHF	: Ultra High Frequency
RTL	: Radio Television Luxembourg
RX	: Receiver
TX	: Transmitter

CHAPTER 1

INTRODUCTION

1.1. SIDEBAND SIGNALS

Generally, the Amplitude Modulation (AM) is a technique used to modulate a wave based on changing its amplitude based on changes in both the frequency and amplitude of the related modulating signal with keeping the wave frequency constant.

The change in the wave amplitude is directly related to the change in the modulating signal amplitude. Both the negative and positive peaks of the wave change according to changes in the modulating signal, in which the decrease or increase in the modulating signal amplitude causes a related decrease or increase in the wave peaks amplitude [1].

The main two types of the AM technique are the Single Sideband (SSB) and Double Sideband (DSB). The SSB is a kind of AM with one sideband only and without the carrier wave suppressed. On the other hand, the DSB is an AM with two sidebands; upper and lower ones and with the wave carrier suppressed. In practice, the DSB is consistent with the SSB receivers, in which the receiver only rejects the redundant or unwanted sideband. On the other hand, the Independent Sideband (SSB) represents the deployment of both sidebands to carry two independent information channels [2].

DSB signals are generated based initially on suppressing the carrier, which results in upper and lower sideband. In this way, there is no waste in power.

A DSB signal represents the summation of two sinusoidal sidebands and generated based on modulating a carrier via the information signal of a single-tone sine wave. Thus, the carrier is then suppressed, while the amplitude of the DSB sine wave signal changes at the carrier frequency. The main characteristic of the DSB signal is the phase transitions, which happen at the wave lower amplitude portions. Particularly, balanced modulator circuit is mainly used to generate DSB carrier signals based on generating the summation and difference among frequencies and to balance or cancel the carrier. Regardless of this characteristic and both the simple design and low costs of generating DSB signals, they are rarely deployed due to the difficulty in demodulating signals at the receiver. One of the main applications of the DSB signals is in transmitting the information in a television signal [1].

Therefore, SSB signals are used widely since the occupied spectrum space in these signals is half the DSB signals. This ensures conserving the spectrum space and permits transmitting more signals in the same range of frequency. Furthermore, SSB signals allow channelling all devoted power to the carrier and the other sideband to generate stronger and more reliable signals received at longer distances. Another benefit is the smaller size and lower weight of SSB transmitters when compared to those of DSB transmitters due to the use of less power and circuitry. Furthermore, a narrower bandwidth and small noise are occupied in SSB signals. On the other hand, SSB signals have some unusual characteristics, such as the inability to send Radio Frequency (RF) signals when there is no modulating signal or information signal, unlike in DSB signals, where the carrier is sent regardless of its modulation. However, since there is no transmitted carrier in SSB systems, there are no signals when the information signal is zero. In addition, sidebands are produced in the modulation process only [1].

1.2. DIGITAL MODULATION TECHNIQUE

When the variation in the carrier signal is discrete, there are many types of methods using this technique, which are:

1. Phase-shift keying (PSK): is a digital communication method in which the phase of a transmitted signal is changing to convey information.
2. Binary phase-shift keying (BPSK): is a digital communication method, It uses two different signal phases (0 and 180 degrees).
3. Amplitude-shift keying (ASK): is a digital communication method in which the amplitude of a transmitted signal is generated when the input signal in 1 and when it zero there is no transmitted signal.
4. Frequency-shift keying (FSK): is a digital communication method that have two binary states, logic 1 (high) and 0 (low), each one is represented by different analog waveform. The high state is represented by a wave at a specific frequency, and the low state is represented by a wave at a different frequency.
5. Audio frequency-shift keying (AFSK): is a digital communication method by which digital data is represented by changes in the frequency of an audio tone, to have an encoded signal suitable for transmission using telephone or radio.
6. Multi-frequency shift keying (M-ary FSK or MFSK): is a method of digital communication by which discrete audio tone bursts of various frequencies convey digital data.
7. Dual-tone multi-frequency (DTMF): DTMF tones are the tones generated when a key on a standard telephone or mobile keypad is pressed. When any key is pressed, two frequencies are generated, low frequency and high frequency, which are mixed together to form what is called "dual tone".
8. Quadrature amplitude modulation (QAM): is a combination of PSK and ASK.

Among the listed methods, ASK, PSK, and FSK are basic modulations, and the other are advanced schemes. The advanced schemes are variations and combinations of the basic schemes.

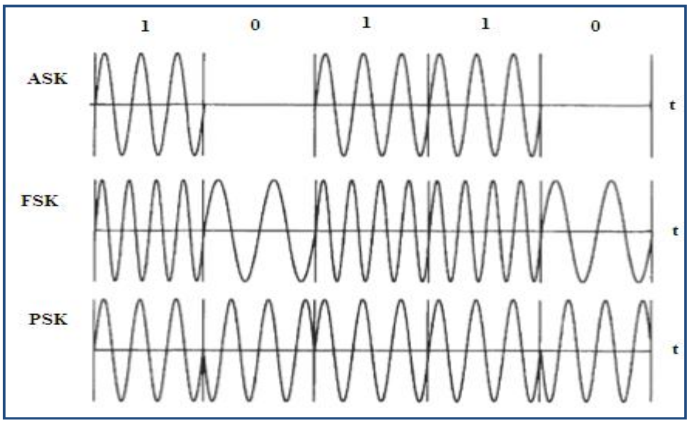


Figure 1.1. Digital modulation methods (ASK, FSK, PSK).

CHAPTER 2

LITERATURE REVIEW

2.1. OVERVIEW AND HISTORICAL REVIEW

The process of Amplitude Modulation saw the start of general public broadcasting almost at start of this century's second decade. As of present, the modulation systems for transmission depended heavily on the economic and practical aspect to receive technology. There is clear evidence that the detection of aptitude was by then the known practical technic for demodulation of signals when the radio communication ideas began to come into formulation by late 1800's all through to early 1900' [3].

From some of backdate technical writing on the radio communication, it is apparent that general angular modulation mathematical knowledge i.e. phase and frequency modulation were not in existence, this was the case until 1920's. These transmission modes and receiving technology necessary were could not be practically proven, until the standard of broadcast technological arts and radio communication were facilitated by amplitude modulation [4].

In broadcasting today, amplitude modulation refers more accurately to DSB-FC-AM, this acronym stands for Double Sideband-Full Carrier-Amplitude Modulation, 'AM' abbreviation has however has become generally accepted in describing the transmission n mode currently that is used all over the world in the Amplitude Modulation Standard broadcast band with between 535 to 1605 kHz in the north America and has a wider band in the rest of the world [4] supports that AM was known to be not the most efficient transmission mode from the very beginning of technology of broadcast, either in transmission of intelligence or spectrum usage. AM broadcast receiver and transmitter pioneer engineering technology had a sense of responsibility that was deeper towards the development of the "marvellous medium" in enhancing the general public lives with education, communication as well as entertainment programs. Some of the early technologies of communication pioneers in the US and other parts of the world have a vision which was greater for both programming and technical aspect of public broadcasting.

With progress in technology and time, there was a need to address the transmission efficiency problem by engineers who could no longer ignore it. In 1930's, when telephone industry had major switch plans from the DSB-FC-AM to favour the (SSB-SC) Single Sideband-Supressed as well as the wireless services, due to its channel capacity and high efficiency in transmission, the broadcast industry on the other had was committee to conventional AM continuance due to the desire for capability of economic public receiver hence, the attention of broadcast engineers naturally turned to transmission standards improvement such as efficiency, fidelity of transmissions, interference of co- and adjacent channels, and transmission reliability [1].

Of the above, the ones that have seen the greatest improvement are transmission reliability and efficiency, and have continued to see the greatest improvement in users and design of transmitters. In the late 1940's as indicated in [5] the fidelity of transmission which is at all times critical reached a plateau, this has significantly been improved every day where the latest design of transmission has been reached through the processing philosophy and equipment of pre-transmitter pro-gram, this influences the reception fidelity perceived and has continually changed.

In early 1980's, there was a widespread reality of the AM band stereo transmissions on the continent of North America. The transmission mode has now been known and referred to AM Stereo. Some of the broadcasters who used AM stereo saw it important means of broadcasting that would compete with FM-stereo that as over time has increasingly gained public acceptance for total programming types [5]. Approval of AM-stereo was done in the 1981 by the FCC as the general basis however, unlike colour television or FM-stereo, the transmission standards were not adopted by the FCC for this potential development in terms of revolution in the standard broadcasting band [1].

Instead, it was allowed by the FCC that any AM-Stereo system of transmission commercially developed company that will past the approval requirements by the FCC regarding the interference and mono-capacity, to compete for consumers favour and the broadcasters in establishing de-facto standards for transmissions of AM-Stereo. Some of the receiver manufacturers have made a decision to establish and build a mass production receiver, which is for only one of the different processed systems that are based on their private and public study of the market potential and system merits. Mass production receivers have also been built by other manufacturers which will manually and/or automatically decode any of some of the systems processed [6]. Which marketing philosophy of receiver will prevail and what the most efficient and favourable long term interest of AM public listening are unknown yet. Some of the AM-Stereo systems have been presented in the following chapter.

2.2. MODULATED SIGNALS REPRESENTATION

Any un-modulated Radio Frequency (RF) signal is expressed as a cosine waveform oscillation at a specific RF, which defined based on its frequency and phase as $f_{RF} = \omega_{RF} / 2\pi$, using the following formula with initial phase φ_0 and peak amplitude $\hat{a}$: [7]

$$S_{RF}(t) = \hat{a}\cos\left(\omega_{RF}t + \varphi_0\right) \tag{2.1}$$

Such waveform can be modulated in amplitude and phase separately, where this results in: [7]

$$S_{RF}(t) = \hat{a}(t)\cos[\varphi(t)] \tag{2.2}$$

Another efficient representation, which resulted from extending the signal with an imaginary part that represents its Hilbert Transform [8, 9], offers an analytic signal:

$$\underline{S}_{RF}(t) = \hat{a}(t)\{\cos[\varphi(t)] + j\sin[\varphi(t)]\} = \hat{a}(t)e^{j\varphi(t)} \tag{2.3}$$

Both the amplitude and phase of analog modulation modes are illustrated in table 2.1. where S_{LF} represents the modulation signal with low frequency, while $\underline{S}_{LF}^{+}$ and $\underline{S}_{LF}^{-}$ are the upper and lower analytic representation SBs of the modulated signal, which are expressed as follows: [7]

$$\underline{S}_{LF}^{+}(t) = \underline{S}_{RF}(t) = S_{LF}(t) + j\mathcal{H}\{S_{LF}(t)\} \tag{2.4}$$

$$\underline{S}_{LF}^{-}(t) = \underline{S}_{LF}^{*}(t) = S_{LF}(t) - j\mathcal{H}\{S_{LF}(t)\} \tag{2.5}$$

Table 2.1. Modulation modes [7].

Modulation Mode	$\hat{a}(t)$	$\varphi(t)$
Amplitude Modulation (AM)	$a_0 \frac{1}{2}[1 + m s_{LF}(t)]$	$\omega_{RF} t + \phi_0$
Upper Sideband Modulation (USB)	$a_0\ m\ \underline{s}_{LF}^{+}(t)$	$\omega_{RF} t + \phi_0$
Lower Sideband Modulation (LSB)	$a_0\ m\ \underline{s}_{LF}^{-}(t)$	$\omega_{RF} t + \phi_0$
Phase Modulation (PM)	a_0	$\omega_{RF} t + \Delta\phi s_{LF}(t) + \phi_0$
Frequency Modulation (FM)	a_0	$[\omega_{RF} \Delta\omega s_{LF}(t)]\, t + \phi_0$

The peak RF amplitude a_0 can be deployed also to characterize the modulation modes. other efficient parameters are the modulation index (m), the peak frequency variation $\Delta\omega$ and the peak phase variation $\Delta\emptyset$. In practice, the AM mode affects on the amplitude only, both the FM and PM affects on the phase, while both the LSB and USB affects on both components [7].

2.3. OVERVIEW OF MODULATION

2.3.1. Modulation Definition and Advantages

The Modulation process represents encoding specific message information in an appropriate way for transmission. Such process involves converting a baseband message signal into a pass-band one. This baseband signal is called the modulating signal, while the pass-band one is called the modulated one. The modulation process can be achieved based on changing specific characteristics of the carrier waves based on the message signal [10].

The main advantages of the modulation process are:

- Simplifying multiple access: this is based on converting the signals baseband spectrum obtained from several users to various frequency bands, in which multiple users can involve in the electromagnetic spectrum band
- Enhancing the communication range: it solves the attenuation problem of low frequency baseband signals, which prevents their transmission over long distances. Therefore, it is converted into a higher frequency band to ensure long distance transmission
- Reducing the antenna size: there is an inverse relation among both the height and aperture of the antenna and the radiated signal frequency. This in turn results in high frequency signal radiation for small antennas.

2.3.2. Modulation Types

Generally, there are three types of modulation; amplitude modulation, SSB modulation and phase and frequency modulation. These types are introduced in the following subsections. The modulation is divided also into analog and digital types, which also introduced below.

2.3.2.1. Amplitude Modulation (AM)

Overview of Amplitude Modulation (AM)

Practically, when the high frequency carrier wave amplitude varies as a function of the signal intensity, this is known as the amplitude modulation. The AM principle is shown in Figure 2.1. It can be noticed that the amplitudes of the negative and positive carrier wave half-cycles vary in relation with the signal. In other words, the increase in the positive sense results in an increase in the carrier wave amplitude, while the opposite occurs for the negative half-cycle. The AM process is mainly performed using an electronic circuit that known as the modulator [11].

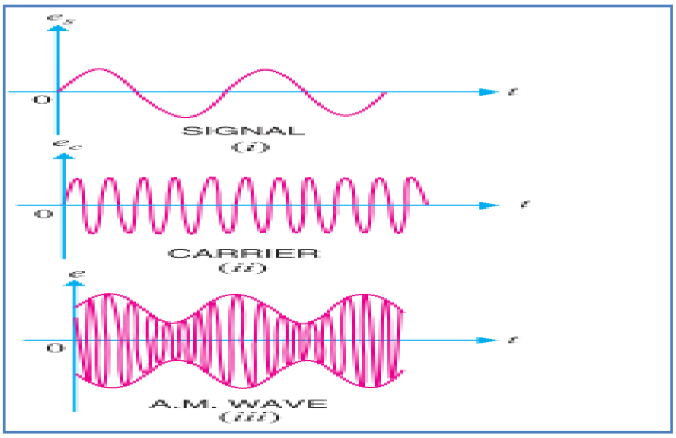


Figure 2.1. Representation of the AM principle [11].

One of the essential considerations in the AM process is the modulation factor, which represents the modulation depth or the change in the carrier amplitude. In other words, it is the ratio among the change in the carrier amplitude and the normal carrier wave amplitude. Such factor determines both the quality and strength of the transmitted signal. For any AM wave, the modulation of the carrier to a small degree results in a small carrier amplitude change. Thus, the transmitted audio signal is not very strong. In other words, the greater the modulation degree, the clearer and stronger the audio signal will be [11].

Presence of Sideband (SSB) Frequencies in AM Wave

Practically, the presence of sideband frequencies in an AM wave is an attractive topic, since the signal frequency is included in the sideband frequencies as shown in Figure 2.2. In practical radio transmissions, the carrier frequency is much larger than the signal frequency. Therefore, sideband frequencies are mainly close to the carrier one [11].

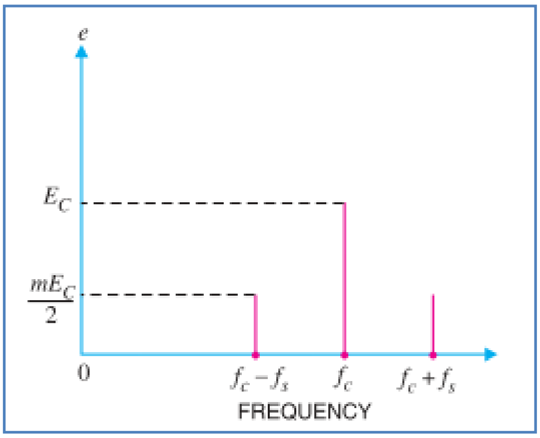


Figure 2.1. Presence of sideband frequencies in an AM wave [11].

Structure of Amplitude Modulator

The general amplitude modulator structure is shown in Figure 2.3. in which the related intermediate signals spectra are shown in Figure 2.4.

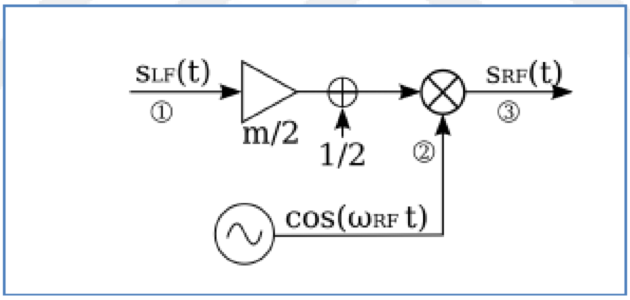


Figure 2.3. General amplitude modulator structure [7].

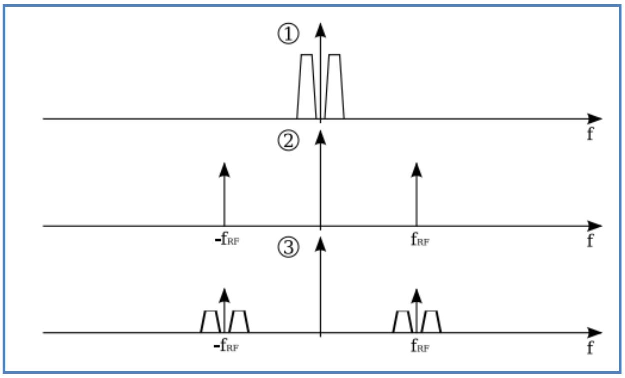


Figure 2.4. Related intermediate signals spectra [7].

Such intermediate signals are computed for a modulated signal as follows:

$$S_{LF}(t) = \cos(\omega_{LF} t + \emptyset_{LF}) \tag{2.6}$$

$$S_{RF}(t) = \cos(\omega_{RF} t)\left[\frac{1}{2} + \frac{m}{2}\cos(\omega_{LF} t + \emptyset_{LF})\right] \tag{2.7}$$

AM Limitations

Regardless of the high effectiveness offered by the AM process, it has various limitations, such as: [11]

- Noisy reception: the signal in the AM wave is in the amplitude changes of the carrier signal. In practice, the whole manmade and natural noises composed of electrical amplitude disturbances. The reception is mainly noisy since the radio receiver could not distinguish among amplitude changes including the desired signal or those producing noise.
- Low efficiency: in the AM, efficient power is in the SBs since they include the signal. However, the AM wave has low SB power. Therefore, its effectiveness is low
- Small Operation range: the AM low efficiency let transmitters using such process having small operation range. Therefore, messages cannot be sent over large distances
- Insufficient audio quality: the high-fidelity reception depends on reproducing the whole audio frequencies up to 15 kHz. This in turn requires a bandwidth of 30 kHz because both SBs should be reproduced. However, the highest modulating frequency of AM broadcasting stations is 5 kHz, where this hardly enough for the reproduction.

2.3.2.2. Single Sideband (SSB) Modulation

The realization of the SSB modulation is based on finding the Hilbert Transform of the modulated signal and them performing multiplying it with the complex RF generator signal. Figure 2.5 illustrates the structure of such modulation. The related intermediate signals spectra are shown in Figure 2.6.

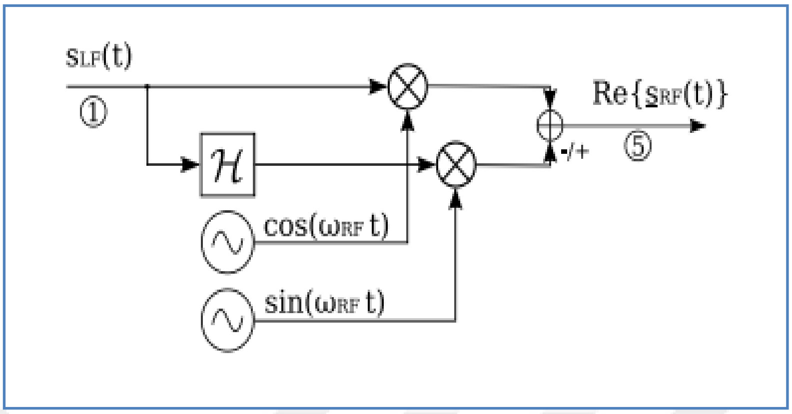


Figure 2.5. General structure of SSB modulation [7].

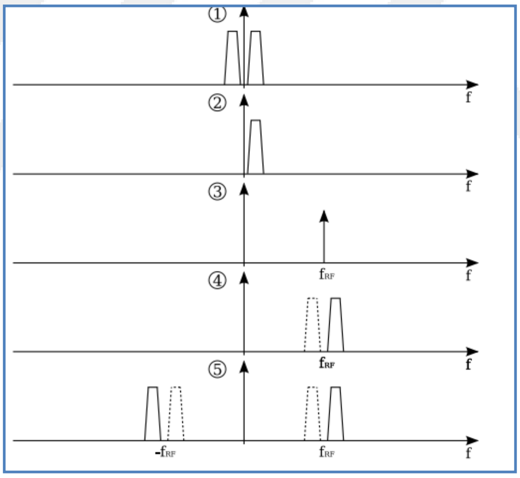


Figure 2.6. Related intermediate signals spectra [7].

Such intermediate signals are computed using the following expression:

$$S_{LF}(t) = \cos(\omega_{LF} t + \emptyset_{LF}) \tag{2.8}$$

The related Hilbert transform of the input signal is represented as follows:

$$\mathcal{H}\{S_{LF}(t)\} = \sin(\omega_{LF} t + \emptyset_{LF}) \tag{2.9}$$

The related expressions of the USB and LSB signals are:

$$\underline{S}_{LF}^{+}(t) = \cos\big((\omega_{\mathrm{RF}} + \omega_{\mathrm{LF}})\mathrm{t} + \emptyset_{\mathrm{LF}}\big) \quad (2.10)$$

$$\underline{S}_{LF}^{-}(t) = \cos\big((\omega_{\mathrm{RF}} - \omega_{\mathrm{LF}})\mathrm{t} - \emptyset_{\mathrm{LF}}\big) \quad (2.11)$$

2.3.2.3. Phase and Frequency Modulation (PM) and (FM)

Practically, the phase and frequency modulation represents a manipulation concerning the RF waveform argument. Therefore, such modulation should be directly realized in the oscillator during the RF signal production [12].

2.3.2.4. Linear and Digital Modulation

There two basic forms that the aptitude modulation of signals of radio occurs, this is Linear (analogue) and coded (digital. The coded aptitude module has capacity to process for long communication distance involved on/off keying of a carrier wave of a radio. The code or "pattern" of off/on keying process is responsible for determining the information content being transmitted [4]. Quasi-linear or linear undulations in the carrier wave amplitude, but, are usually utilised in transmitting the analogue information that exists in music and speech. The signals of radio carrier waves into which the amplitude variations that are analogue can be shown as [4]

$$e(t) = AE_c Cos(w_c t) \quad (2.12)$$

Where: *e(t)* is the carrier wave instantaneous amplitude as a time (t) function, *A* is the carrier wave amplitude modulation factor, $w_c$iscarrier wave angular frequency (second/radians) and *Ec* is the carrier wave peak amplitude.

Assuming that *A* is constant, the carrier wave peak aptitude is also constant and there is no existence of modulation. Carrier wave periodical modulation exists only of *A* magnitude is caused to waver with respect to time, for example, a sinusoidal wave can be as [3]

$$A = 1 + \left(\frac{Em}{Ec}\right) Cos(\text{ɯ}ct) \tag{2.13}$$

This can be viewed as; *Em/Ec* is the modulation and carrier amplitude ratio

Hence;

$$e(t) = Ec\left(1 + \left(\frac{Em}{Ec}\right) Cos(\text{ɯ}ct)\right) Cos(\text{ɯ}ct) \tag{2.14}$$

Which is a periodic amplitude well known basic equation, and when all simple trigonometric identify and multiplications have been done, the result becomes' [1]

$$e(t) = EcCos(\text{ɯ}ct) + \frac{M}{2}\left(Cos(\text{ɯ}ct) + \ (\text{ɯ}mt)\right) + \ \frac{M}{2} Cos(\text{ɯ}ct - \text{ɯ}mt\ ) \tag{2.15}$$

Where: *M* is the factor *Em/EC* amplitude modulation.

There are three familiar ways in which equation [1] can be graphically presented. Figure 2.7 shows the time domain representation, Figure 2.8 shows the frequency domain and Figure 2.9 the relative vectors. The presentations have been done as single tone modulation index (M) of 0.7, that is: 70% peak modulation voltage, of (*Em/Ec* =0.7) peak carrier wave voltage. In Figure 2.8 occupation bandwidth is shown, comprising an AM signal with modulation of a single tone [5].

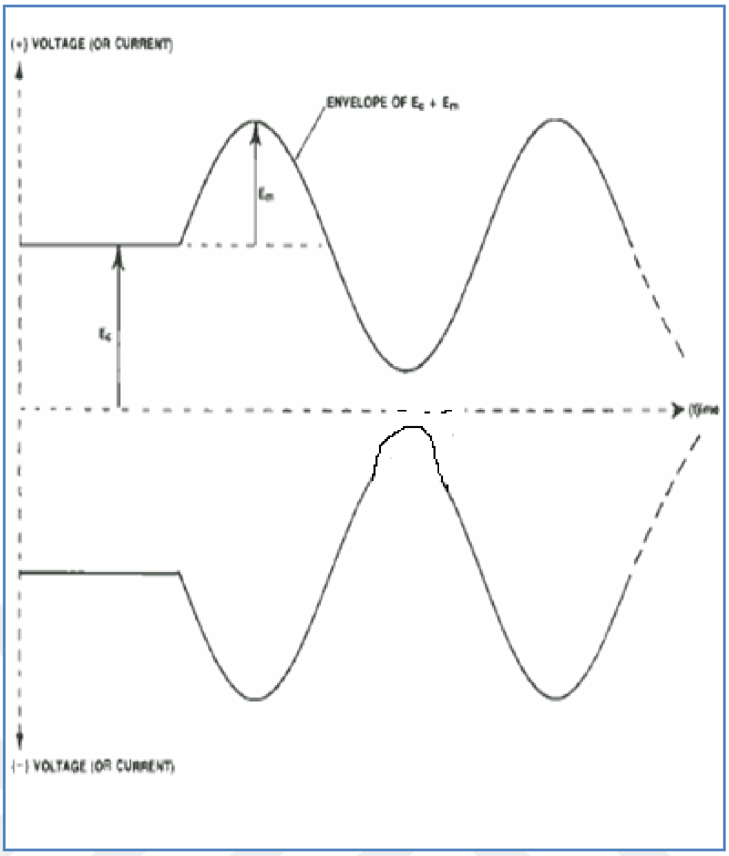


Figure 2.7. Time domain representation of a carrier wave signal amplitude modulated by a sinusoidal audio signal to a peak modulation depth of 70% [1].

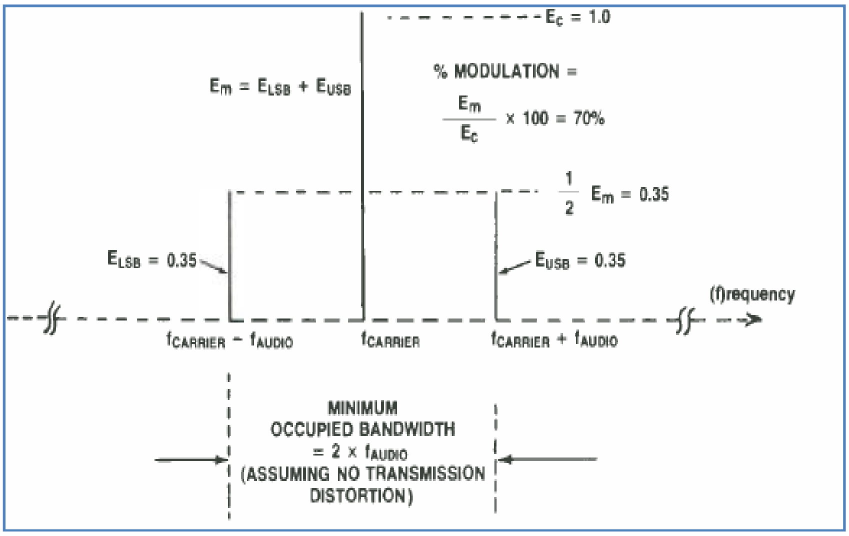


Figure 2.8. Frequency domain representation of an amplitude modulated signal showing the carrier wave signal and two resultant modulation sidebands at 70% modulation [1].

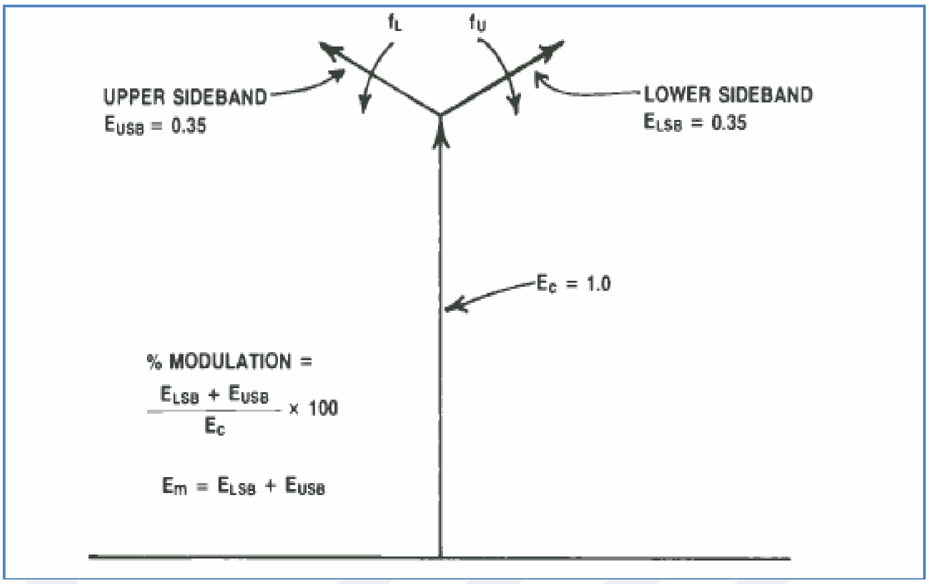


Figure 2.9. Vector representation of an amplitude modulation signal showing the carrier wave signal and two resultant modulation sidebands at 70% modulation [1].

2.3.3. Selection of the Best Modulation Type

There are various factors that effect on the selection of the best modulation type. The appropriate modulation scheme type must offer low bit error rates with low received signal to noise ratios. As well, it must have efficient performance with the presence of multipath and fading conditions. Furthermore, it must be simple, cost effective and occupy a minimum bandwidth. Generally, the modulation scheme performance is assessed as a function of the bandwidth efficiency and power efficiency. The bandwidth efficiency represents the scheme ability to accommodate data with a restricted bandwidth, while the power efficiency represents the scheme ability to preserve the message fidelity at low power levels.

Practically, the signal power must be increased in any communication system in order to enhance its noise immunity. As well, the system capacity is in a direct relation with the scheme bandwidth efficiency. On the other hand, the highest possible bandwidth efficiency is affected by the channel noise. [13, 14].

2.4. OVERVIEW OF DEMODULATION TYPES

The demodulation represents the reciprocal process of the modulation one, where it represents the extraction of the original baseband signal from the modulated pass-band one [10]. The main demodulation types are expressed below.

2.4.1. Amplitude Demodulation

The amplitude modulation can be achieved using both simple and complex product detectors. The simple type can be constructed using both multiplication and low pass filter. The multiplication represents multiplying the real RF signal by a local one, which gives the following: [7]

$$S_{RF}(t). = \left(a_0 \frac{1}{2}[1 + mS_{LF}(t)]\cos(\omega_{RF} t)\right).(\cos(\omega_{RF} t + \emptyset_{RF})) \tag{2.16}$$

$$= a_0 \frac{1}{2}[1 + mS_{LF}(t)].[\cos(\emptyset_{RF}) + \cos(2\omega_{RF} t + \emptyset_{RF})] \tag{2.17}$$

The filtration process represents applying a low pass filter and a DC clock one, where this reconstructs the modulated signal to be:

$$S_{RX}(t) = a_0 \frac{1}{2} mS_{LF}(t)\cos(\emptyset_{RF}) \tag{2.18}$$

When the receiver RF phase is the same as that of the transmitter, an optimal reconstruction is obtained. Else, the demodulated signal is fading with a specific frequency difference$\emptyset_{RF} = \Delta\omega t$. The use of a Phase Locked Loop (PLL) can offer an equal phase [15,16].

The complex product detector is based on using a further quadrature (Q) component, which produced by mixing a sine wave signal, which gives the following:

$$S_{RX,Q}(t) = a_0 \frac{1}{2} mS_{LF}(t)\sin(\emptyset_{RF}) \tag{2.19}$$

This results in forming a complex signal:

$$\underline{S}_{RX} = S_{RX,I}(t) + jS_{RX,Q}(t) \tag{2.20}$$

This signal is computed using a Cartesian to polar coordinate conversion as follows:

$$S_{RX}(t) = \sqrt{S_{RX,I}^2(t) + S_{RX,Q}^2(t)} = a_0 \frac{1}{2} m S_{LF}(t) \tag{2.21}$$

Thus, there is no fading for small frequency differences due to the cancelation in the real (I) component. In practice, the frequency difference should not exceed the cut-off one of the DC filter clock.

2.4.2. SSB Demodulation

The main SSB demodulation stages are:

1. Shifting the input signal frequency based on the modulation with the complex conjugate RF signal
2. Choosing the baseband approximately 0 based on using a low pass filter
3. Choosing the upper or lower SB based on filtering the negative or positive components of the frequency
4. Finding the resultant real part

Figure 2.10 below illustrates the structure of such demodulation process.

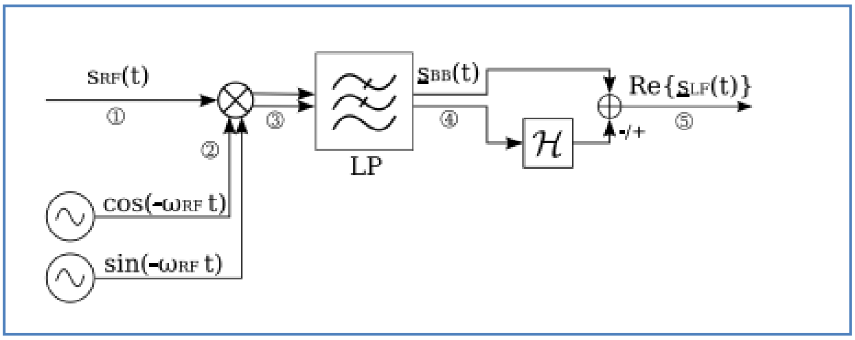


Figure 2.10. Structure of SSB demodulation process [7].

2.4.3. Phase and Frequency Demodulation

This type of modulation is the same as that of the complex product detector in the amplitude modulation, with a difference that the phase information should be assessed rather that the magnitude. Thus, the phase modulated signal can be demodulated with the use of the Cartesian to polar transformation to get: [7]

$$\varphi(t) = \arctan\left(\frac{S_{RX,Q}(t)}{S_{RX,I}(t)}\right) \quad (2.22)$$

Based on using the relation among the angular frequency and phase, the following expression is obtained:

$$\varphi(t) = \int_0^t \omega(t) + C \quad (2.23)$$

This in turn offers a relation for the demodulation of the modulated signal frequency as follows:

$$f(t) = \frac{1}{2\pi}\omega(t) = \frac{1}{2\pi}\frac{d}{dt}\varphi(t) \quad (2.24)$$

Practically, the PLL can be deployed as a sophisticated frequency demodulator. As well it can be deployed as a demodulator output for a frequency error signal inside with the assumption that the PLL is slow enough not to track the related modulated signal [15,16].

CHAPTER 3

SDR AND AUDACITY AND ENSAMBLE RXTX

3.1. SOFTWARE DEFINED RADIO (SDR) PROGRAM

The initial wireless communication was discovered in late 80's and since then there have been various innovations in radio communication technology with an aim of ensuring that the radio users are always connected. Triumphant radio was the first type of transmission in the 1930s and used analog voice communication due to limitations of bandwidth at the time. The era was followed by broadcast communication in the 1950s where analogic television communication was standard but used much bandwidth with excellent customer experience. In the 1960s use of computers became popular, and they were capable of transmitting information over a long distance and could use both wired and wireless connections. Wireless voice communication was later discovered after the emergence of cell phones thus enabling transmission from any point; however, the mobile devices were hard to operate considering that they were not portable [4].

The mobile devices used today are simple to use and carry and can provide both internet and cellular networks and also receives and transmit wireless communication at an extremely high speed. Even though there is a commendable advancement on these devices, the biggest problem with them is that they still rely on hardware radios and protocols. Due to this challenge, if a problem emerges within the hardware, software or firmware of the devices, it was difficult to solve it since the inbuilt resources are hard to remove. This brings the need to have a new approach in which these problems can be solved, and thus we propose the application of SDR.

The device is designed toapproach for wireless communication devices that were discovered by Joseph Mitola in the early 90s. The device aims at identifying, reprograming and reconfiguring radio problems. The SDR receiver device is very different from traditional receiver devices in various ways [3].

Like other technologies, SDR originated from the military environment to civilians. The initial users of the SDR device were the US Navy, but the device could not be used without the hardware. Various advancements have occurred and today both the software and the hardware of this device are available at a considerable cost [5] In the research will show how SDR operates and its employment areas at an affordable price thus offering a permanent solution to local radio environments.

3.1.1. SDR Hardware

The section provides a theoretical overview of differences in hardware compatibility of current SDR and the traditional device. Also discussion of the performance of the software, thetransmission will be included as the applications of the SDR technology.

3.1.2. Traditional Receiver

Apart from demodulation, the conventional receiver had other functions such as it contained the frequency tuning for the purpose of selecting the best signal, had a filter that separated transmissions received from other sources and amplified transmission to avoid wastage. For more than a century, the traditional receivers used the conventional heterodyne designs in remitting [1].

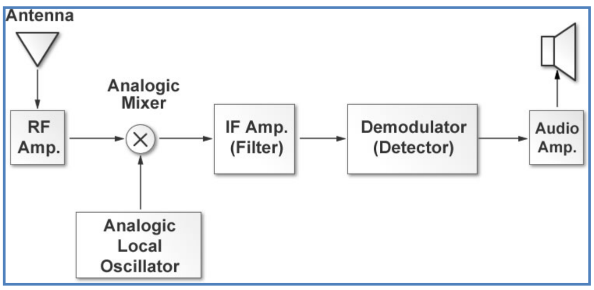


Figure 3.1. Internal blocks of a regular super heterodyne receiver [1].

In this scheme, signals enter through the antenna and then sent to RF for amplification. After amplification, the signal is then forwarded to the mixer that has the capacity of receiving local oscillators. The setting of the oscillator's frequency involves turning the radio control. Also, the mixer is also responsible for translating the signal to the IF (intermediate frequency) The converted signal arrives at the demodulator, where the original modulating signal is worked on through application of various options [5].

3.1.3. SDR Receiver

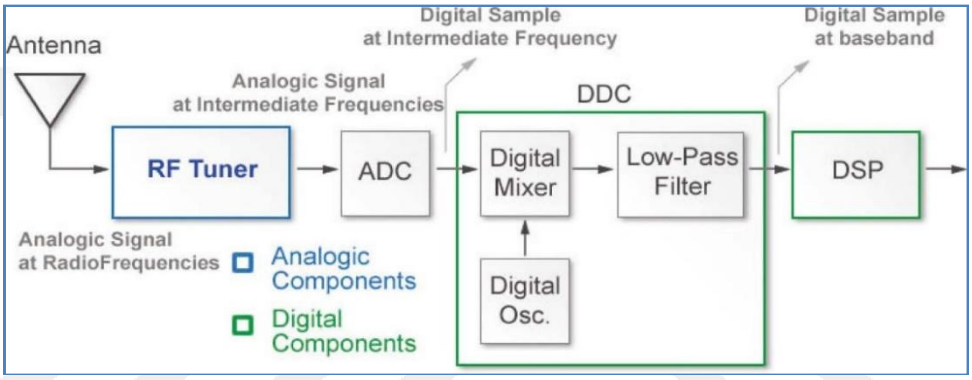


Figure 3.2. SDR receiver [3].

The operation of SDR receiver and the traditional receiver is similar in that the digital mixer and the local oscillator changes the IF digital signals to baseband while the FIR low enables passage of filters.

3.1.4. SDR Transmitter

Receivers are identified as the most known SDR devices but within the technology, there are the transmission strategies. The antenna gets signals in input form through DSP generation. The digital upconverter moves the received signal to IF. The DAC then translates the signals into analogue form and then the RF transfers the signals near higher frequencies [6]. When the signal is amplified and sent to the antenna. Inside the DUC is an interpolation filter that raises the baseband signals to be equal to the operating frequency. Later the digital mixer and the oscillator moves the sample signals to IF under the control of the local oscillator.

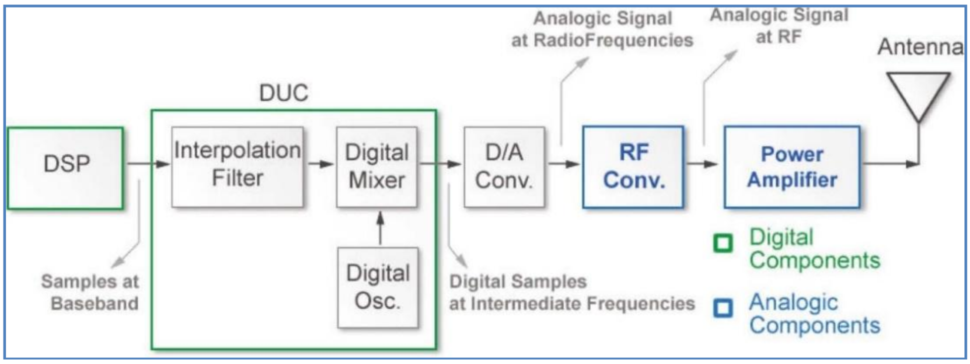


Figure 3.3. SDR transmitter [5].

3.2. HIGH DEFINITION SOFTWARE DEFINED RADIO (HDSDR)

3.2.1. Definition of HDSDR

The HDSDR is a free type of SDR program used for windows, XP and Vista, with more compatibility with Windows 8. It is an updated Winrad version. This program has real-time pan windows for both the complete spectrum and the pass-band one. As well, it can be easily installed and set up with including adjustable resolution bandwidth, adjustable radio frequency scaling, minimums, speed and averaging for both pan and waterfall and adjustable proportioning among pan and waterfall. In addition, it has the same adjustments for the pass-band window. The HDSDR program permits a user to enter global and mode offsets in order to sync correctly the pitch among the radio and SDR audio. As well, it has very flexible setup options of the sound card with constant mage rejection null adjustments. [17,18].

3.2.2. Setup Up and Use of HDSDR

The majority of settings are in the soundcard, bandwidth and some other options in the program screen. On the other hand, various settings are kept at their default values. The first setting is the soundcard, which allows the user to select the supported soundcard by his/her device. The input is the signal, which feeds the display of the pan, while the output is the card speaker output, in which a user then can hear a demodulated audio from the HDSDR. The bandwidth also must be chosen, in which the highest supported sampling rate value by the soundcard must be selected for the input, while any value can be used for the output, where the optimal one to demodulate the output is 12000 or 6 kHz audio bandwidth. The soundcard selection stages are shown below [18].

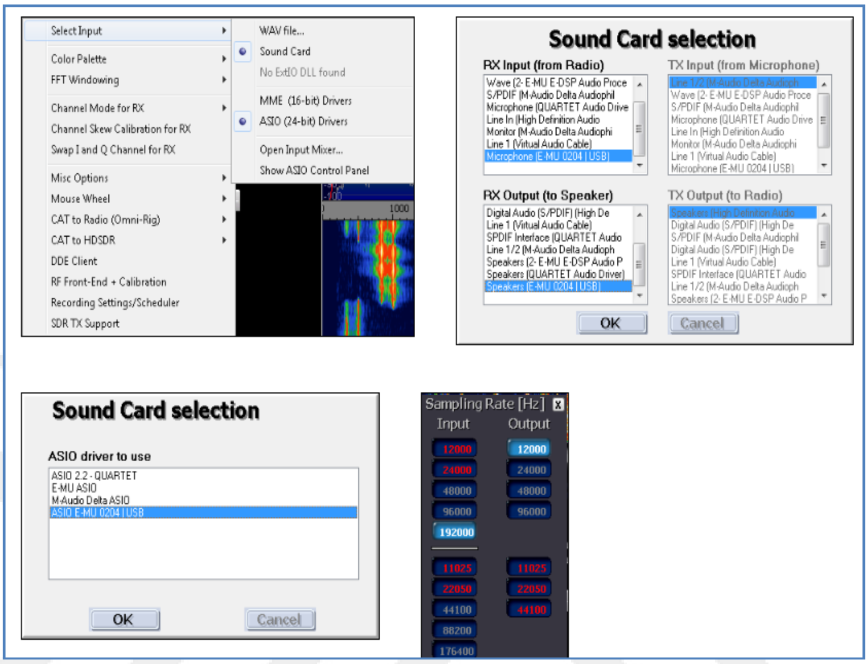


Figure 3.4. Soundcard selection stages [18].

The CAT type must be then defined as shown in Figure 3.5. As well, the Rig1 selection must be performed using the Omni-Rig Setup choice as in Figure 3.6.

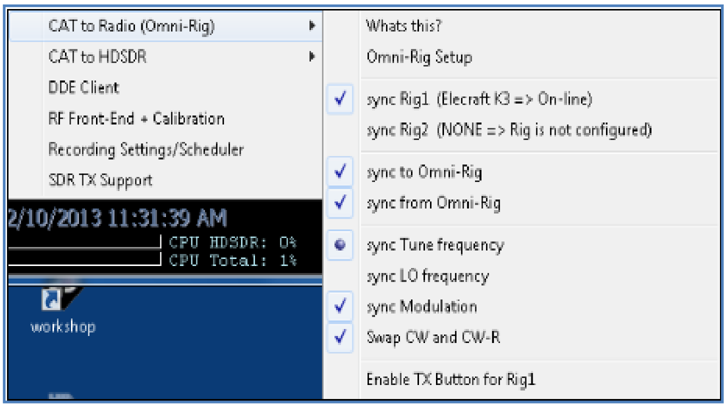


Figure 3.5. CAT type setup [18].

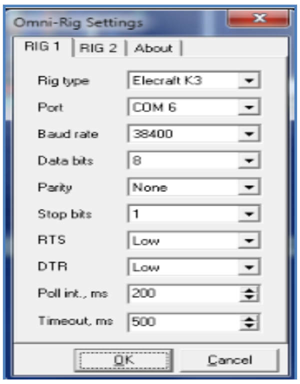


Figure 3.6. Omni-Rig setup [18].

Both the global and mode offsets must be set up them as shown below. Generally, the Lower Sideband (LSB) must be positive, while the Upper Sideband (USB) must be negative, in which their values are separated by 2700 to 3000 Hz

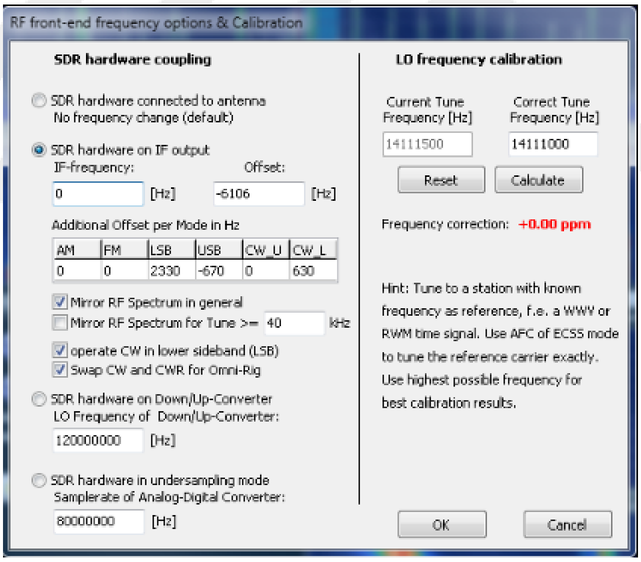


Figure 3.7. Global and mode offsets setup [18].

The image rejection must be adjusted then based on setting the zoom to the maximum on the pan display. As well, a tuning must be performed in a strong steady carrier. The image can be null then with the use of the raw phase and amplitude controls and the fine controls. 50-60dB image rejection must be achieved across most of the when for edge to edge tuning.

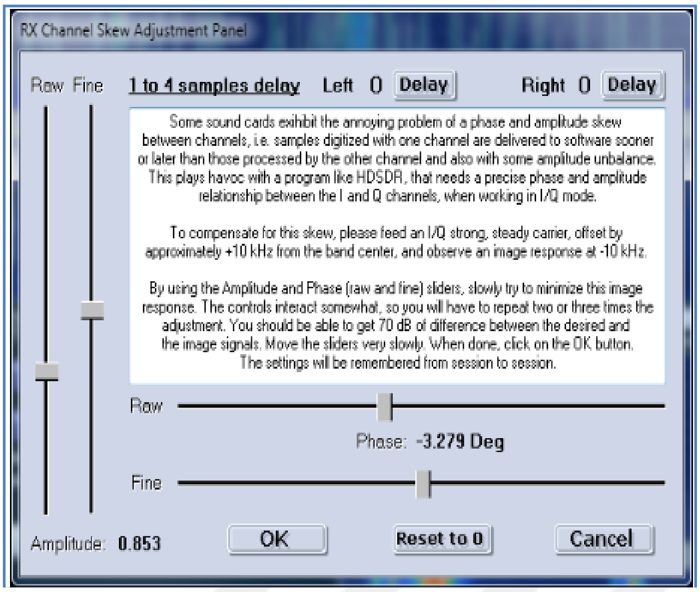


Figure 3.8. Image rejection adjustment [18].

3.3. SOFTROCK ENSEMBLE RXTX

3.3.1. General Structure of SoftRock Ensemble RXTX

In general, the SoftRock ensemble RXTX transceiver gives 1 watt SDR transceiver, which can be generated for five band groups; 160m, 80m/40m, 40m/30m/20m, 30m/20m/17m and 15m/12m/10m. Its main components are a soundcard, stereo line-in connector for the receiver, stereo line-out connector for the transmitter and SDR software. All these components can be assembled at the choice of builders [19, 20]. The ensemble block diagram is shown in Figure 3.9.

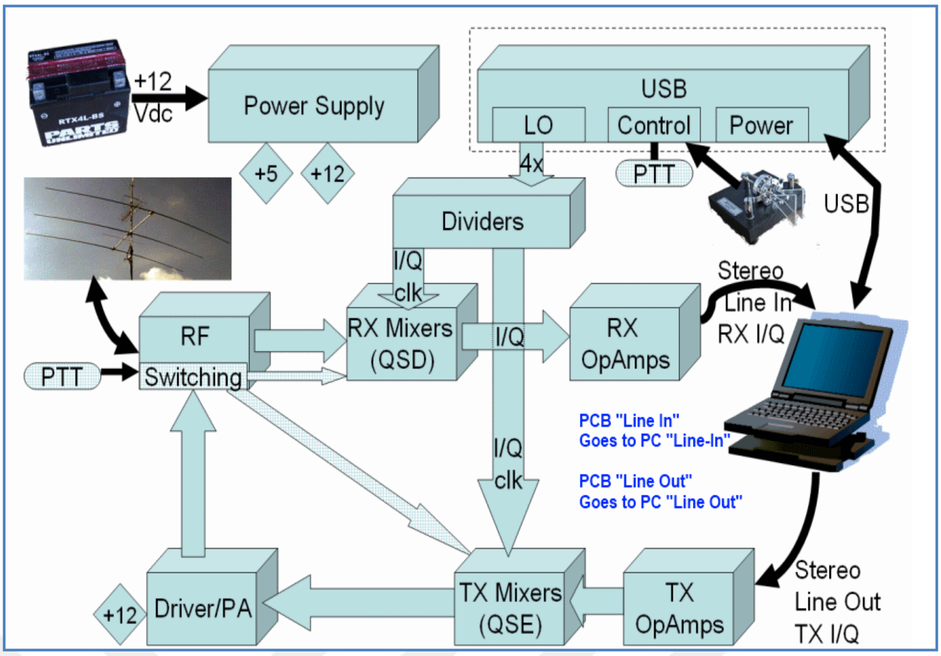


Figure 3.9. Ensemble block diagram [19].

3.3.2. Soundcard Requirements

The SoftRock ensemble RXTX transceiver requires one or two soundcards in order to handle the RX/TX. When this kit is utilized for non-digital modes, such as Frequency Modulation (FM), Amplitude Modulation (AM) and Single Sideband (SSB), a second soundcard must be added for both the microphone input and speaker or headphone output. This additional soundcard can be inexpensive with lesser quality. A careful selection must be carried out about the best band, where it will hold the session duration in the Ensemble RXTX portion [19].

3.3.3. Foundation Circuitry

A programmed Attiny85 microcontroller as a USB device is used to control the transceiver to offer PTT control, control of local oscillator frequency and paddles input. The USB bus provides the power for this section. The local oscillator is used to generate signal having a frequency that is four times the target radio center frequency. This center frequency and the soundcard control the spectrum width and its visible bandwidth that displayed in the RX and TX display. The bandwidth is a number of kHz in the both sides of the center frequency, where this number equals to the half of the sampling rate of the soundcard. This sampling rate is directly related to the soundcard cost [19].

The local oscillator output is the input of the dividers stage, in which the signal is segmented into two identical, 90 degrees out of phase signals at the center frequency, where they clock both the RX and TX mixers.

3.3.4. RX Circuitry

An antenna terminal is included mainly in the transceiver with a switched RF path among the TX low-pass filters and RX Band-pass filters. This switch is related to the PTT signal obtained from the microcontroller as chosen by the SDR program. In the RX circuitry, the input RF is band-pass filtered, while the output RF is in anti-phase. A mixer chip is also included in the RX circuitry to transform the input RF into its quadrature analogues with 0-100 kHz frequency range. The quadrature pairs are the inputs of the RX Op-Amp stage to be amplified and filtered into infra-audio and audio range to then transmit to the RX output stereo audio jack [19].

The soundcard transforms the “I and Q” analog signals into digital representations, which are operated by the SDR program modules to carry out several radio functions, as filtering and demodulation. In practice, the soundcard specs, quality and sampling rate affect on the working of signals within the bandwidth [19].

3.3.5. TX Circuitry

The functionality of transmission is the opposite of the RX functionality. In other words, instead of demodulating the input I and Q signals in the RX, the obtained digital signals from the microphone are modulated into analog I and Q audio outputs. These signals are in quadrature, emerge at the stereo jack and fed to the TX Op-Amp to be translates into four identical signals with0, 90, 180, and 270 phase degrees. The mode is SSB or AM, where this is a main soundcard requirement to handle the input to the computer. The four signals are fed to a TX mixer to generate up-converted RF output audio signals. These signals then are shaped and amplified by the driver stage where results are then fed to the antenna path [19].

3.4. AUDACITY SOFTWARE

3.4.1. Introduction of Audacity

To begin Podcasting we need every one of the apparatuses that make the procedure conceivable. A basic part of that procedure is the recording of Podcast onto a PC. This requires the right sound programming. There are numerous bundles yet the one we'll be utilizing is a bit of programming called Audacity that it is genuinely simple to utilize and free.

The Audacity is an available, free, simple multi-track audio recording and editing software program for Windows, GNU/Linux, Mac OS X and several operating Systems.

Its interface is translated into various languages. The main applications of the Audacity are: [21]

1. Recording and playing back live audio
2. Recording computer playbacks
3. Cutting, copying, splicing and mixing multiple sounds with undo options
4. Transforming records and tapes into digital recordings
5. Editing several sound files, as AIFF, MP2, MP3 and WAV
6. Adding effects, such as changing the record pitch or speed
7. Supporting, importing and exporting WMA, M4A/M4R (AAC), AC3 and several formats with the use of optional libraries
8. Reducing noises depending on minimizing the noise sampling
9. Analysing audio spectra

3.4.2. Getting Hold of a Copy of Audacity

To download audacity-wi-1.3.0b. This file can be stored anywhere on PC, but unless told will usually out itself in the temp folder. When the download is complete an icon will appear on your computer, where the download was directed to.

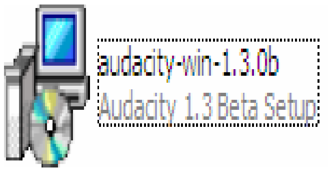


Figure 3.10. Audacity-wi-1.3.0b.

This file is a launch file, which initiates the installation of the full software. This file can then be transferred onto any computer (with the same operating system) to install Audacity.

3.4.3. Getting Started

At this stage we will just require a couple of the accessible elements as demonstrated below.

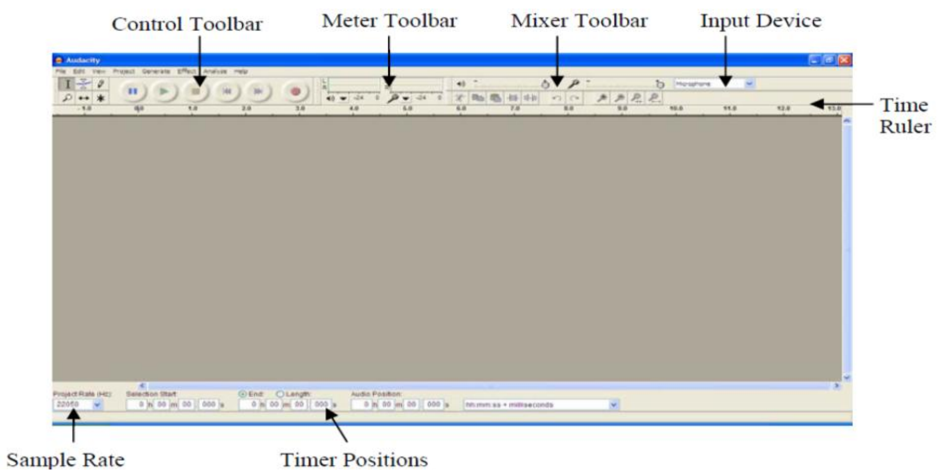


Figure 3.11. Getting started.

When we press stop and the voice Track summary will appear at the left hand side of the voice wave.

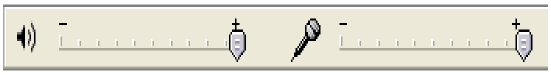
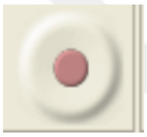


Figure 3.12. Track summary.

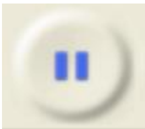
3.4.4. Control Toolbar

This toolbar works especially like a video or tape recorder

Record – starts the recording of a sound document



Delay – incidentally stops the recording or playback. This comes in valuable record one section and afterward to the following segment. To restart the recording press the delay catch once more.

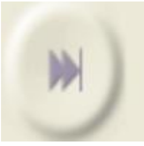


Stop – Ends the recording or playback. This must be done before any further altering is done

Play – plays back a recording

Back Skip - Skips to the start of the recording

Forward Skip – Skips to the end or to a position where the cursor is put



The Cursor – This is an extremely helpful instrument that can be utilized to choose a specific position in a recording or selecting a specimen of the recording to be altered

The only other feature that is important at this stage is the input device drop Located down arrow a list of four options, to select the input using for vocal recording the Microphone.

3.4.5. Considering Recording Conditions

When making a recording need remember the microphone will pick up best of the sound made around it during that time. Almost tips to consider.

1. Foundation clamor, the recording will get different sounds made in the room including other individuals talking, machine commotion and phone rings to maintain a strategic distance from this it is best to record in a very and segregated room.
2. Microphone position, Microphone must not to be to near your mouth, as it tends to get undesirable clamors like breathing, in addition to the recording has a tendency to be noisy if the amplifier is to close.

3. When making a recording attempt a for the survival the microphone At a fixed distance from the mouth on the grounds that fluctuating the separation changes the volume of the recording. This is aided by the utilization of a headset.
4. It is better to all the recording of a Podcast on an indistinguishable PC from various machines have distinctive encompassing sounds which will fluctuate through the recording if not done on a similar machine.

3.4.6. Starting Recording

Before we Beginning Register just do are a few last minute final checks, (i) Check the microphone is plugged in and (ii) that the input is set to microphone and finally (iii) the sample rate is set a 44100Hz.

Then press register and start talk. We are talking, you will know it is working because a voice wave trace of you voice is produced. The other thing you may notice is meter toolbar will move to the voice of Our voice, and finally the Timer Positions at the bottom of the screen will be counting, some these features We can see it in Figure 3.13.

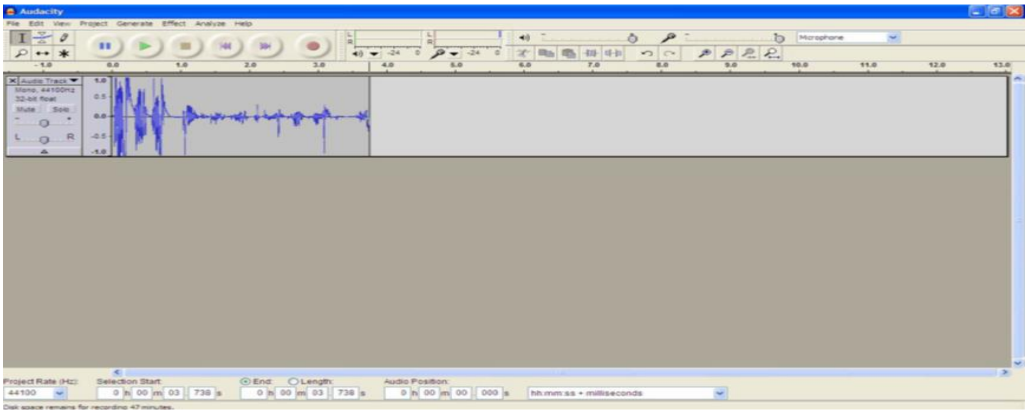


Figure 3.13. Timer positions.

When we press stop and the voice Track summary will appear at the left hand side of the voice wave.

3.4.7. If an Error Occurs During the Recording

At the point when making a record it can disappoint on the off chance that you say the wrong thing or commit an error, particularly on the off chance that you are doing a long recording, I can prescribe a straightforward well ordered to overcome it sketched out in Fig 3.14.

1. Try not to Panic.
2. Must silence for around 3 seconds, this delivers a level line in the sound that can be utilized to recognize the error later.
3. Press the stop button to temporarily stop the recording. Then press the stop button again to continue the recording.
4. Then continue by repeating the past leading up to the error and carrying on the correct interlocutor.

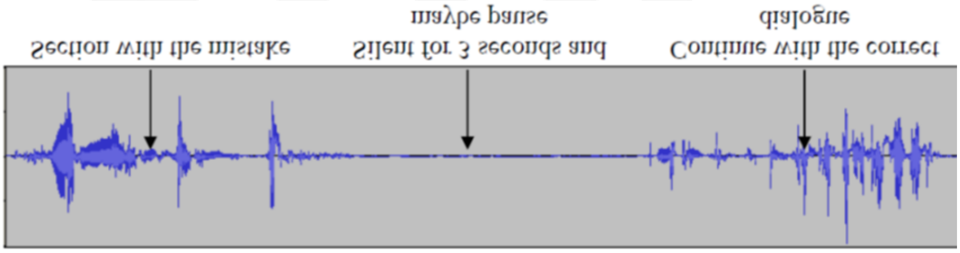


Figure 3.14. An error occurs during the recording.

To remove the error, after expiration of the recording, once Pressure stop. Listen to the section the error to decide must remove. This may take a few the attempts When deletion of recording, using the Indicator highlight the section .This is done my left pressure the mouse, then whilst holding down the left mouse button move the Indicator along the require section as in Figure 3.15.

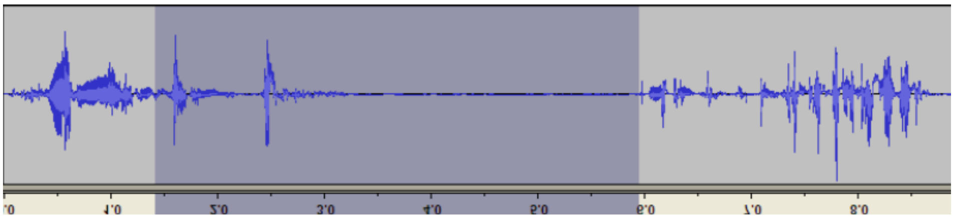


Figure 3.15. Indicator holding.

At that point to evacuate this area of the recording press the Delete key on the console and the sound wave will realign itself as appeared in (Figure 3.16) shortening the length of the recording by the measure of time expelled

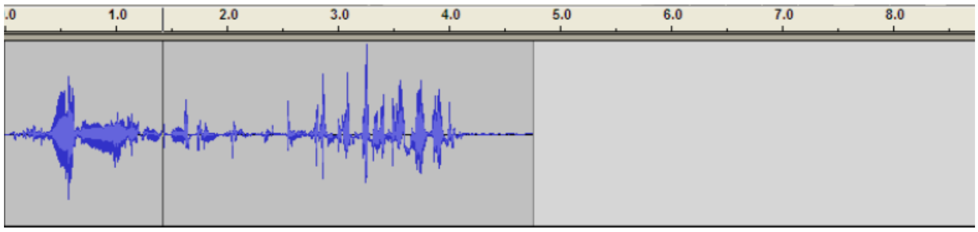


Figure 3.16. Delete a specific area.

3.4.8. If We Want to Add Some More Dialogue to My Registration?

If we’ve made a recording, but once we’ve listened to it feel we’ve missed something out, and would like to add extra dialogue, follow these steps:

1. Registration the extra dialogue place the cursor ultimately of The voice track, and press record to record the extra dialogue, which will appear as a second voice track This is shown in the form of Figure 3.17.

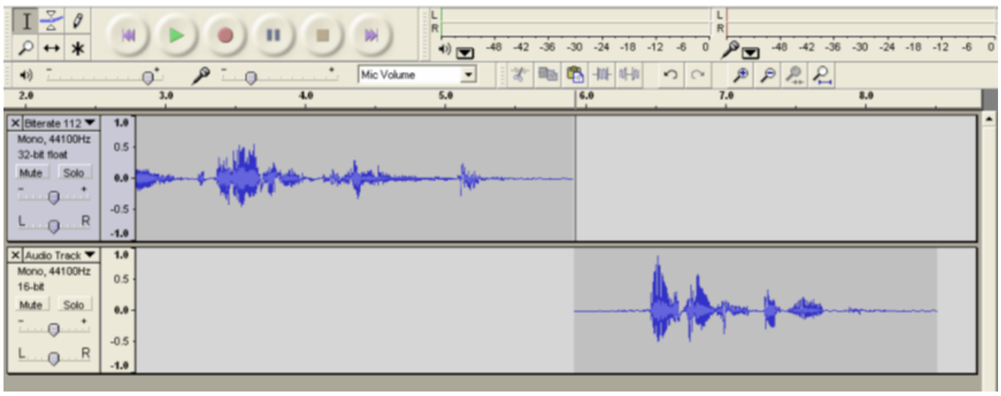


Figure 3.17. Registration.

2. To position the new dialogue in the correct position in the he voice recorded, place the cursor where the new dialogue will start, then highlight the voice track from that position to the end (on the right) the following Figure shows that

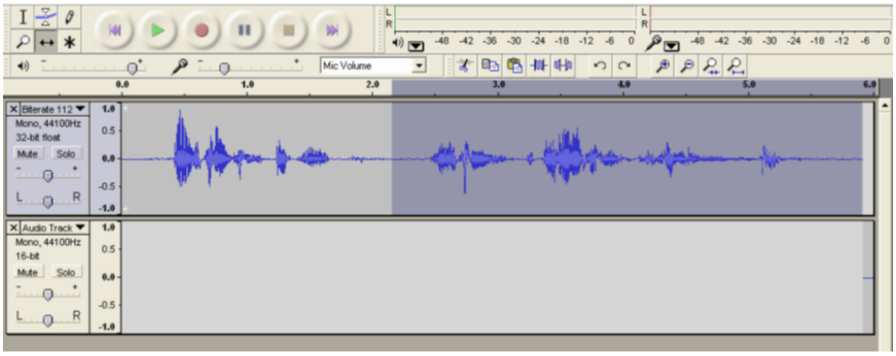


Figure 3.18. Position of the new dialogue.

At that point go to the Edit Menu on the menu bar and select the Split alternative. This will drop the highlighted area of the sound track into another sound track as appeared in Figure 3.19.

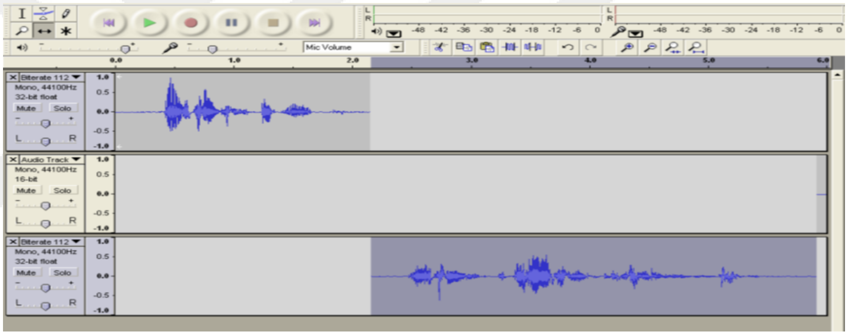


Figure 3.19. Split alternative.

4. To order the dialogue correctly use the Time Shift tool found in the Control Toolbar. For moving the voice tracks the right place the mouse cursor over the voice track and transformation them left or right as required to achieve a seamless. As in Figure 3.20.

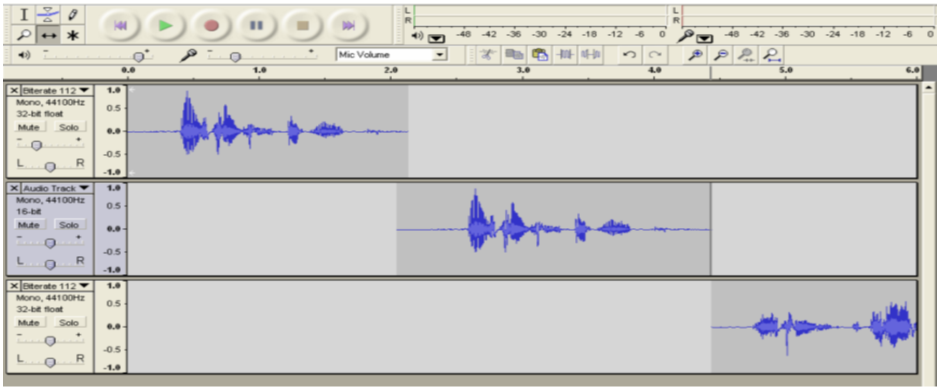


Figure 3.20. Time shift tool.

If we carry out this process when the recording is converted to MP3 the message shown in Fig (3.21) will show. This simply means if we were to open the file again using Audacity it will appear as one continuous audio track, instead of the split structure.

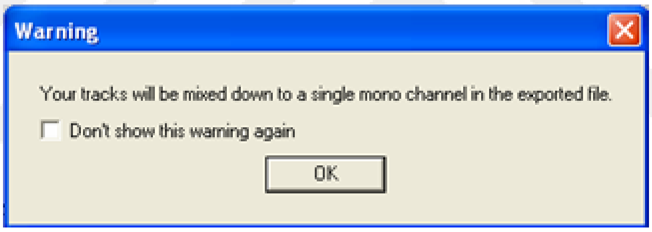


Figure 3.21. Conversion into MP3.

3.4.9. Save Recording

We can spare we item by heading off to the File Menu and selecting Save Project As… This alternative spares the recording as a 'Dauntlessness Project' which is an arrangement called .aup. The organization can then be listened to again however just on Audacity programming. Tragically this arrangement cannot be played on commonplace advanced sound gadgets. In this way the record must be changed over into a mp3 arrange, which is a great deal more available, which should be possible by selecting the Export As mp3 alternative on the File Menu.

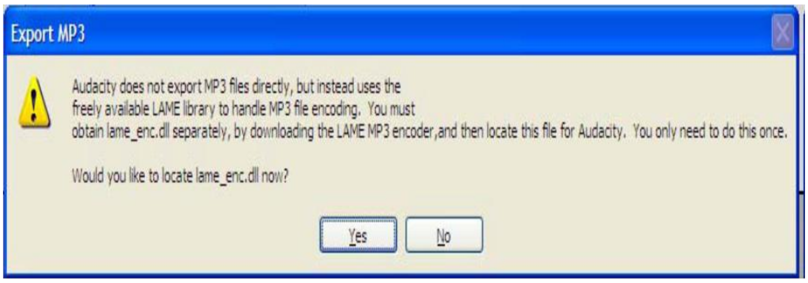


Figure 3.22. Save recordings.

3.4.10. Considering File Size

At the point when recording a Podcast considering the file size is very, in light of the fact that toward the end of the venture the record will be made accessible on the web, and while getting documents from the web the bigger they are the more they take to download. Hence they need to be made as little as could be expected under the circumstances. To accomplish this with a sound document there are three factors that can be adjusted.

1. The example rate, this is a measure of how frequently a moment the PC makes a recording of your voice. Dauntlessness has default settings somewhere around 11025Hz and 96000Hz. Hertz (Hz) is a recurrence measure of every second. In this manner, bring down recurrence implies less recordings every second. Obviously lower recurrence additionally implies bring down quality.

2. The example rate, this is a measure of how frequently a moment the PC makes a recording of your voice. Dauntlessness has default settings somewhere around 11025Hz and 96000Hz. Hertz (Hz) is a recurrence measure of every second. Along these lines, bring down recurrence implies less recordings every second. Obviously lower recurrence additionally implies bring down quality.

Both of these setting can be modified in the Edit Menu in the Preferences segment this opens a window called Audacity Preferences, once in this window select the Quality Tab as appeared in Figure 3.23.

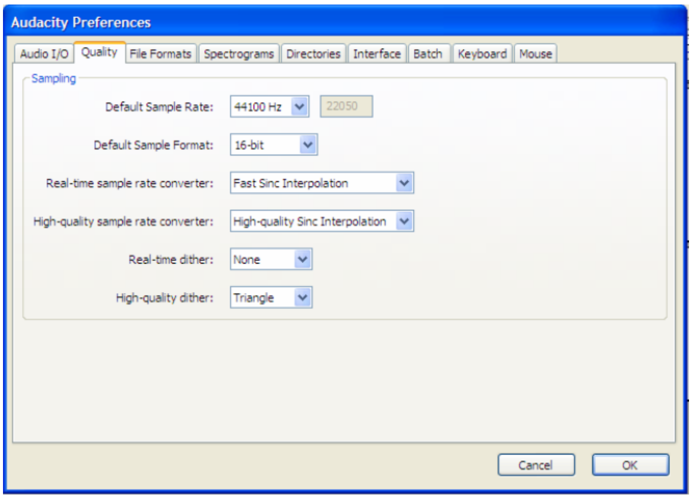


Figure 3.23. Audacity preferences.

The two variables discussed above can be changed in the top two drop down boxes. The settings show of a Sample Rate of 44100Hz and a Format of 16-bit. These are also the settings used by the BBC to produce their Podcasts and CDs

3. The last factor that can be adjusted is changing the Bit-rate of the mp3 made an mp3 is a PC sound document and the Bit-rate is what number of bits, or the amount of that record is played back in one moment.

This can be modified in the Edit Menu in the Preferences segment, this time in the File Format tab, indicated is Figure 3.24.

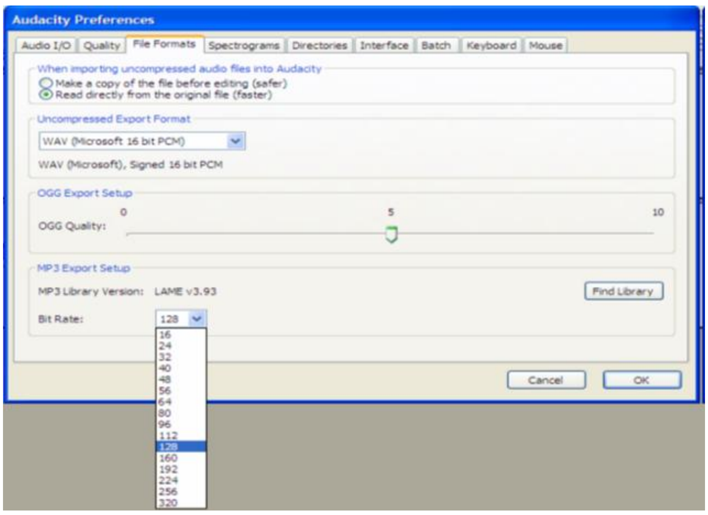


Figure 3.24. File format.

Audacity has a default setting of 128 and this is quite a typical rate for an mp3 file, this quality is only required for music files for a voice recording

CHAPTER 4

SSB AND DSB AM MODULATION DEMODULATION AND MATLAB

4.1. SSB AND DSB-SC MODULATION

The DSB-SC modulated signal can be generated by multiplying the message with a carrier. The SSB modulated signal can be generated by filtering out the lower sidebands of the DSB-SC signal must provide a plot of the message signal after and anti-aliasing. Provide a plot of its magnitude spectrum as Also, must provide plots of the modulated signals along with the magnitude spectrum. These should be provided for each of the four simulations.The channel introduces a noise signal, which can be modelled as zero mean additive white Gaussian noise. For noise-free simulations, the variance of the noise is equal to zero; for noisy simulations, the variance is equal to1/400. The signal seen at the receiver is simply the modulated signal plus the noise.

Table 4.1. Modulation modes.

Modulation Mode	$\hat{a}(t)$	$\varphi(t)$
Amplitude Modulation (AM)	$a_0\frac{1}{2}\left[1+m s_{LF}(t)\right]$	$\omega_{RF}t+\phi_0$
Upper Sideband Modulation (USB)	$a_0\ m\ \underline{s}^{+}_{LF}(t)$	$\omega_{RF}t+\phi_0$
Lower Sideband Modulation (LSB)	$a_0\ m\ \underline{s}^{-}_{LF}(t)$	$\omega_{RF}t+\phi_0$
Phase Modulation (PM)	a_0	$\omega_{RF}t+\Delta\phi s_{LF}(t)+\phi_0$
Frequency Modulation (FM)	a_0	$\left[\omega_{RF}\Delta\omega s_{LF}(t)\right]t+\phi_0$

4.2. DSB SC CODES

```
% LSB Mod. with MUSIC by B. ERKAL 2016
clear all;
% sound file 1 loading (4Khz mono (8KSps)) [iff1 , afs]=audioread('a1.wav');
[y1,~]=size(iff1);
% upsample x6 (8x6=48Khz) yu1=upsample(iff1,6);
```

% Baseband signal is filtered and normalized yu1=filter(fir1(128,4e3/24e3),1,yu1); yu1=yu1./(1.01*max(abs(yu1)));

% baseband result is written to wav fileaudiowrite('as1.wav', yu1, 48e3);

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

% carrier parameters: amplitude frequency and phase C=1;fct=12e+3;tetac=0*(pi/180);

% carrier signal ct1=(C*cos(2*pi*fct*t+tetac))'; ct2=(C*sin(2*pi*fct*t+tetac))';
% LSB signal
% Analytic signal: ath=at+j*at_tilde
% Real part is the signal itself while imaginer part is its Hilbert transform
ath=(hilbert(yu1));
% LSB Carrier modulation IF at 12KHz m=real(ath).*ct1+imag(ath).*ct2;

% bandpass filter AM signal for transmission
% center freq=f=12KHz, BW=5KHz, start=7KHz stop=12KHz taps=128;f=12e3; transition=[0.57,1.01];W=f*transition/(fs/2); BP=fir1(taps,Wn,'bandpass'); m=filter(BP,1,m);
%}

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

The Result By Hdsdr

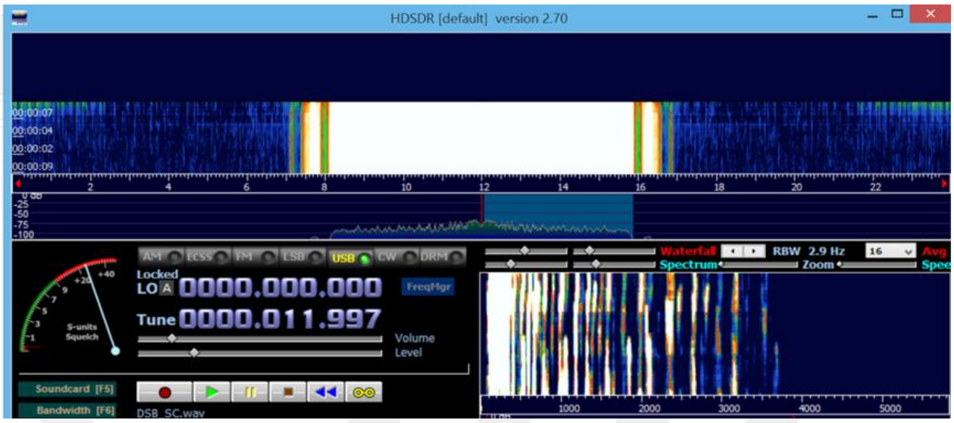


Figure 4.1. 36HDSD.

4.3. DSB-SC AND SSB DEMODULATION

The received signal must be up sampled by a factor R which must be at least a factor of two to prevent aliasing prior to demodulation. Perform synchronous demodulation of the received signal and down sample by a factor of U*R. Finally, remove the zero-padding so that the final signal is the same length as the original.

must provide magnitude spectrum plots for the received signal, the demodulated and received signal before low-pass filtering, and the demodulated and received signal after low-pass filtering. Also, provide a plot of the final message signal as a function of time in the same Figure as the original. Finally, write the final message signal to a file (using the “audiowrite” command in MATLAB). There should be four simulation runs (DSB-SC with noise, DSB-SC without noise, SSB with noise, and SSB without noise). Save all four of the output audio files.

4.4. DSB SC CODES

```
% DSB-SC demod by B. ERKAL 2016
clear all;
% AM IF file loaded
```

```
[iff , afs]=audioread('rec_DSB_SC.wav'); [y,~]=size(iff);
% complex shift frequency entered here w=2*pi*(-12e+3-24.3);

% 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);

% bandpass filter for received signal
% center freq=f=12KHz, BW=10KHz, start=7KHz stop=17KHz taps=128;f=12e3;
transition=[0.57,1.42];       Wn=f*transition/(afs/2);       BP=fir1(taps,Wn,'bandpass');
iffc=filter(BP,1,iffc);
%}% 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 5KHz yu=filter(fir1(128,5e3/afs),1,iffc);

% zero IF normalizedyu=yu./(1.05*max(abs(yu)));
% zero IF written to wav file yl=real(yu);
yr=imag(yu);
audiowrite('as1.wav', [yl,yr], afs);
% DSB-SC coherent dedection
%low performance demodulator
%yout=real(yu)+imag(yu);

%high performance demodulator yout=abs(yu).*sign(100*(yl+yr));

% message signal normalized and written to wav file
yout=yout./(1.05*max(abs(yout)));audiowrite('as2.wav', yout, afs);
```

The Result By Hdsdr And Audacity

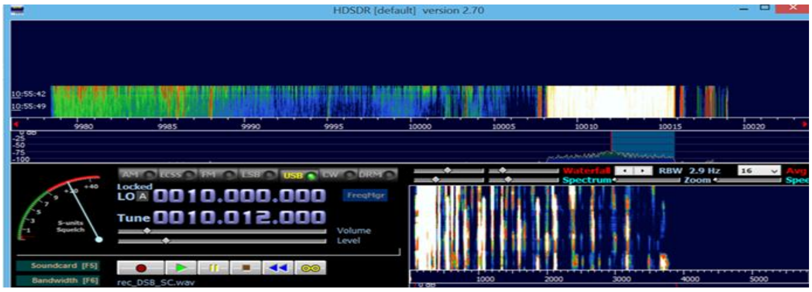


Figure 4.2. HDSDR.

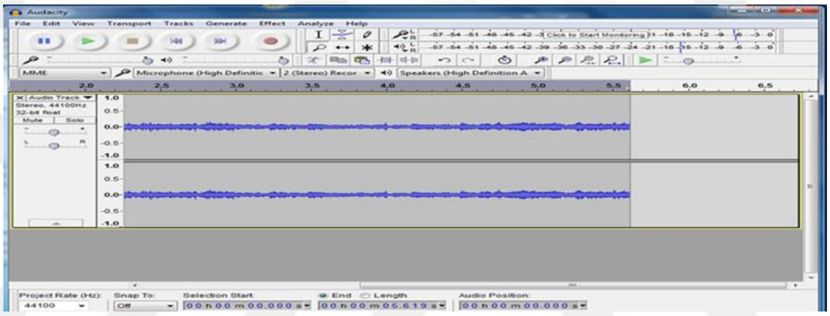


Figure 4.3. Audacity.

4.5. SINGLE SIDEBAND MODULATION

As Table 4.1 shows, the single sideband modulation can be realized by taking the Hilbert Transform of the modulation signal followed by a complex multiplication with the complex RF generator signal. This structure is shown in Figure 3 The upper sideband (USB) can be selected by a plus sign and the lower sideband (LSB) by the minus sign.

The spectra of the intermediate signals of Figure 4.3 are shown in Figure 4.4. The output is a complex signal only interested in the real part of that signal. This simplifies the complex multiplication as the real part of the multiplication result as the complex multiplication as the real part of the multiplication result.

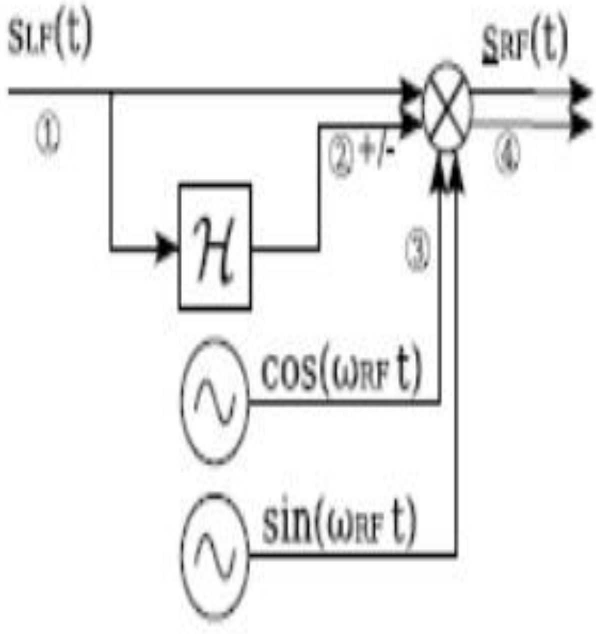


Figure 4.4. Spectra of the intermediate signals.

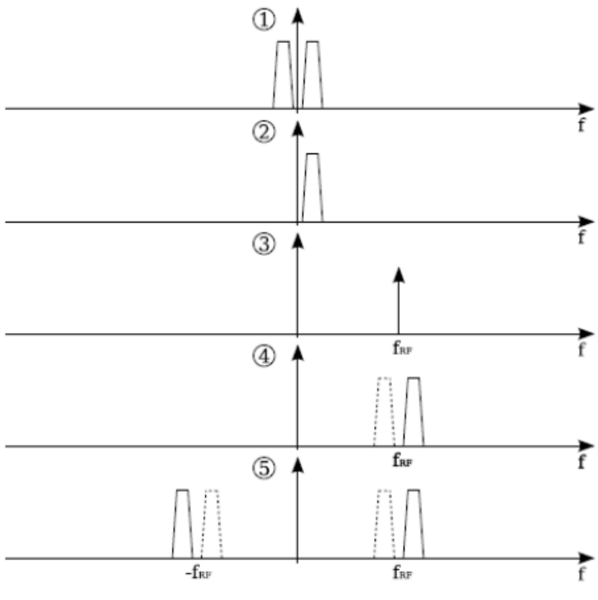


Figure 4.5. Spectra of intermediate signal of modulation (solid line USB dashed line LSB).

4.6. LSB CODES

```
% LSB Mod. with MUSIC by B. ERKAL 2016
clear all;
```

```
% sound file 1 loading (4Khz mono (8KSps)) [iff1 , afs]=audioread('a1.wav');
[y1,~]=size(iff1);
% upsample x6 (8x6=48Khz) yu1=upsample(iff1,6);
% Baseband signal is filtered and normalized yu1=filter(fir1(128,4e3/24e3),1,yu1);
yu1=yu1./(1.01*max(abs(yu1)));

% baseband result is written to wav file audiowrite('as1.wav', yu1, 48e3);

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

% carrier parameters: amplitude frequency and phase
C=1;fct=12e+3;tetac=0*(pi/180);

% carrier signalct1=(C*cos(2*pi*fct*t+tetac))'; ct2=(C*sin(2*pi*fct*t+tetac))';

% LSB signal
% Analytic signal: ath=at+j*at_tilde
% Real part is the signal itself while imaginer part is its Hilbert transform
ath=(hilbert(yu1));

% LSB Carrier modulation IF at 12KHz m=real(ath).*ct1+imag(ath).*ct2;


% bandpass filter AM signal for transmission
% center freq=f=12KHz, BW=5KHz, start=7KHz stop=12KHz taps=128;f=12e3;
transition=[0.57,1.01]; Wn=f*transition/(fs/2); BP=fir1(taps,Wn,'bandpass');
m=filter(BP,1,m);
%}


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

The Result By Hdsdr

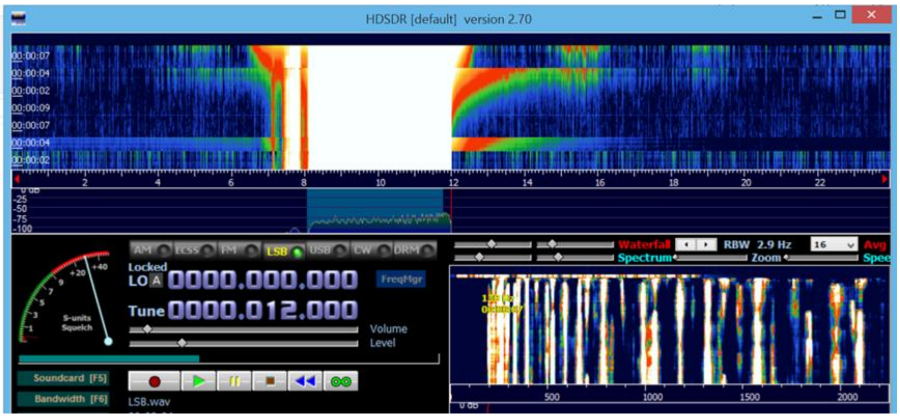


Figure 4.6. HDSDR.

4.7. USB CODES

% USB Mod. with MUSIC by B. ERKAL 2016

clear all;

% sound file 1 loading (4Khz mono (8KSps)) [iff1 , afs]=audioread('a1.wav');

[y1,~]=size(iff1);

% upsample x6 (8x6=48Khz) yu1=upsample(iff1,6);

% Baseband signal is filtered and normalized yu1=filter(fir1(100,4e3/24e3),1,yu1);

yu1=yu1./(1.01*max(abs(yu1)));

% baseband result is written to wav file audiowrite('as1.wav', yu1, 48e3);

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

% carrier parameters: amplitude frequency and phase C=1;fct=12e+3;tetac=0*(pi/180);

% carrier signal ct1=(C*cos(2*pi*fct*t+tetac))'; ct2=(C*sin(2*pi*fct*t+tetac))';

```
% USB signal
% Analytic signal: ath=at+j*at_tilde
% Real part is the signal itself while imaginer part is its Hilbert transform
ath=(hilbert(yu1))

% USB Carrier modulation IF at 12KHz m=real(ath).*ct1-imag(ath).*ct2;

% bandpass filter AM signal for transmission
% center freq=f=12KHz, BW=5KHz, start=12KHz stop=17KHz taps=128;f=12e3;
transition=[0.99,1.42];     Wn=f*transition/(fs/2);     BP=fir1(taps,Wn,'bandpass');
m=filter(BP,1,m);
%}
% IF signal is recorded in wav file
% IF normalized m=m./(20*max(m));
audiowrite('USB.wav', [m], 48e+3)
```

The Result By Hdsdr

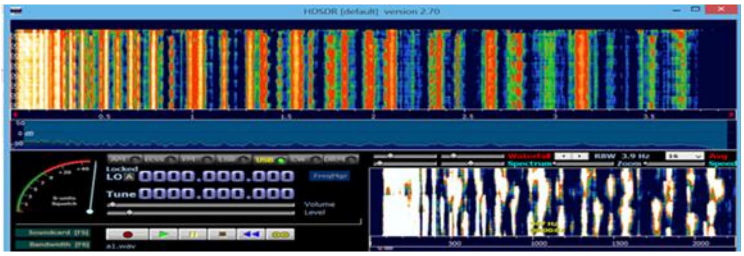


Figure 4.7. HDSDR.

4.8. SINGLE SIDEBAND DEMODULATION

The demodulation can be realized in the following steps:

1. Shifting the frequency of the input signal by modulating with the complex conjugate RF signal.
2. Selecting the baseband around zero by filtering with a low-pass filter
3. Selecting the upper or lower sideband by filtering the negative or positive frequency components.
4. Taking the real part of the result.

A block diagram of this structure is shown in Figure 4.8 (steps 3 and 4 are combined) The corresponding spectra are shown in Figure 4.9. Of course, the low-pass filter can be combined with the Hilbert Transformation filter. The upper sideband (USB) can be selected by a minus sign and the lower sideband (LSB) by the plus sign.

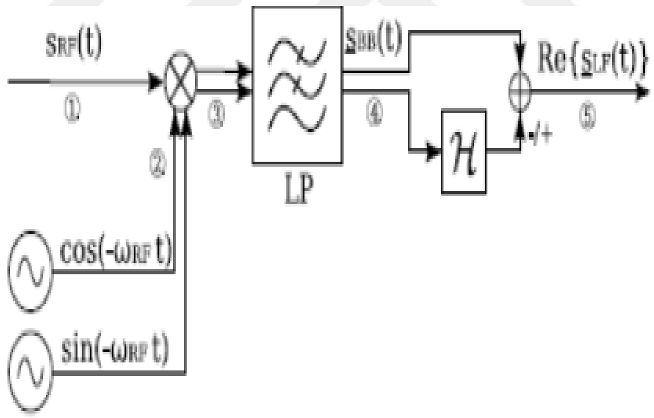


Figure 4.8. SSB demodulation structure with real output signal.

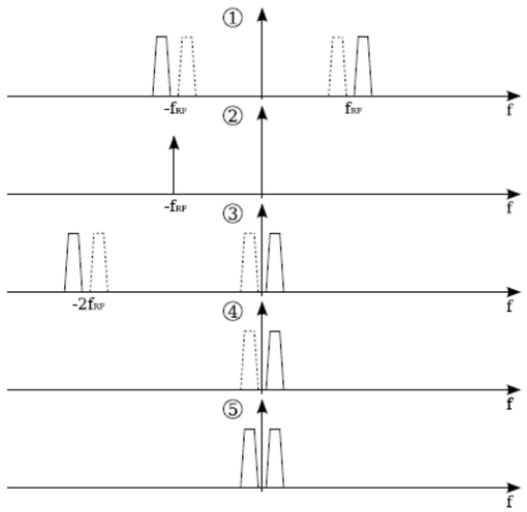


Figure 4.9. Spectra of intermediate signal of SSB demodulation solid line USB.

4.9. LSB CODES

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

% AM IF file loaded
[iff , afs]=audioread('rec_LSB.wav'); [y,~]=size(iff);
% complex shift frequency w=2*pi*(-12e+3-12);

% 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);

% bandpass filter for received signal
% center freq=f=12KHz, BW=5KHz, start=7KHz stop=12KHz
taps=128;f=12e3;    transition=[0.57,1.01];    Wn=f*transition/(afs/2);
BP=fir1(taps,Wn,'bandpass'); iffc=filter(BP,1,iffc);
%}
% time axis created t=0:1/afs:(1/afs)*(y-1);
% complex shift operation iffc=iffc.*exp(1i*w*t');
% zero IF cut at 5KHz yu=filter(fir1(128,5e3/(afs/2)),1,iffc);
% LSB demodulation yout=real(yu)/2+imag(yu)/2;
```

```
% message normalized and written to wav file yout=yout./(1.2*max(yout));
audiowrite('as2.wav', yout, afs);% demodüle mono audio dosyaya yaz

% zero IF normalized and written to wav file yu=yu./(1.01*max(abs(yu)));
yl=real(yu); yr=imag(yu);
audiowrite('as1.wav', [yl,yr],
```

The Result By Hdsdr And Audacity

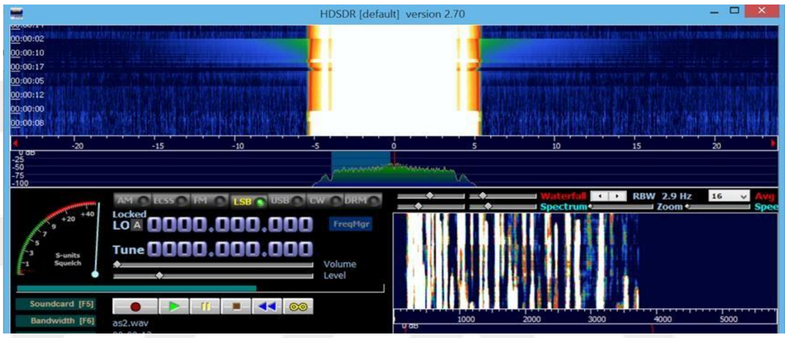


Figure 4.10. HDSDR.

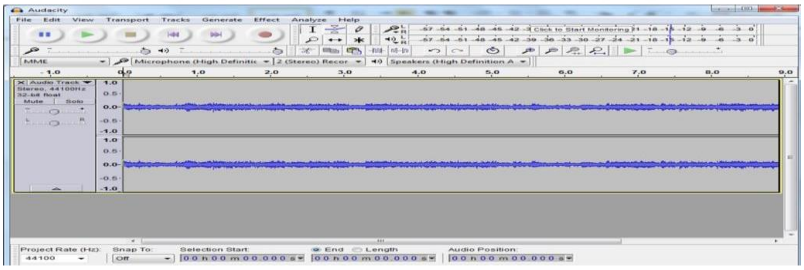


Figure 4.11. Audacity.

4.10. USB CODES

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

% AM IF file loaded
[iff , afs]=audioread('rec_USB.wav'); [y,~]=size(iff);
% complex shift frequency w=2*pi*(-12e+3);
% 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);

% bandpass filter for received signal
% center freq=f=12KHz, BW=5KHz, start=12KHz stop=17KHz taps=128;f=12e3;
transition=[0.99,1.42];     Wn=f*transition/(afs/2);     BP=fir1(taps,Wn,'bandpass');
iffc=filter(BP,1,iffc);
%}
% time axis created t=0:1/afs:(1/afs)*(y-1);
% complex shift operation iffc=iffc.*exp(1i*w*t');
% zero IF cut at 5KHz yu=filter(fir1(128,5e3/(afs/2)),1,iffc);

% USB demodulation yout=real(yu)/2+imag(yu)/2;

% message normalized and written to wav file yout=yout./(1.2*max(yout));
audiowrite('as2.wav', yout, afs);

% zero IF normalized and written to wav file yu=yu./(1.01*max(abs(yu)));
yl=real(yu); yr=imag(yu);
audiowrite('as1.wav', [yl,yr], afs)
```

The Result By Hdsdr And Audacity

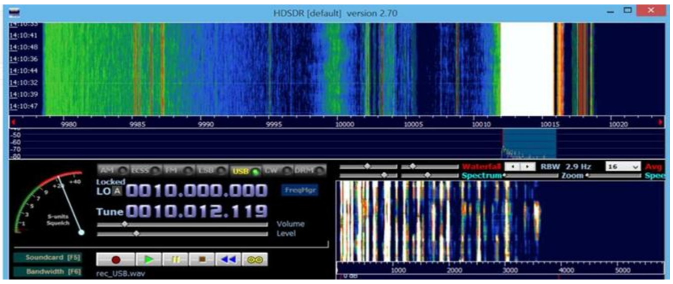


Figure 4.12. HDSDR.

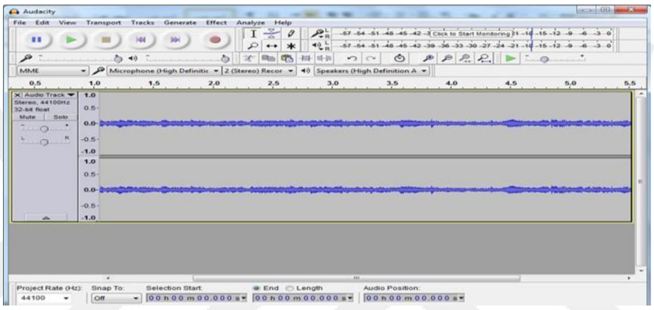


Figure 4.13. Audacity.

CHAPTER 5

CONCLUSION

In this study, investigation and analysis concerning the operation of spectrum analyser, oscilloscope and the function generator to generate and view different waveforms are presented using Matlab scripts. As well, comparisons among the transmission and receiving of SSB and DSB AM signals and among their modulation and demodulation processes are conducted using the Softrock SDR and MATLAB programs. The SoftRock Ensemble RXTX transceiver is deployed in the transmission and receiving of SSB and DSB AM signals. Such transceiver composed of a soundcard, stereo line-in connector for the receiver, stereo line-out connector for the transmitter and SDR software. As well, the Audacity program is used to check the recording modulating and modulated signal waveforms. Written Matlab scripts are used for modulation and demodulation. HDSDR is used to transmit and receive of the IF signals prepared by such Matlab scripts.

In the modulation of the SSB and DSB-SC AM signals, the SSB modulated signals are generated by filtering out the lower sidebands of the DSB-SC modulated signals, which in turn generated by multiplying the message with a carrier. In this stage, a noise signal is introduced by the channel that can be modelled as zero mean additive white Gaussian noise.

In the demodulation of SSB and DSB-SC AM signals, the received signal is sampled by a factor R which must be at least a factor of two to prevent aliasing prior to demodulation. In this stage, magnitude spectrum plots are presented or the received signal, the demodulated and received signal before low-pass filtering, and the demodulated and received signal after low-pass filtering. As well, the final message signal is plotted in terms of the time and then written to a file in the MATLAB. Four simulation runs are conducted in the demodulation stage; DSB-SC, DSB-WC, LSB, and USB in which the four output audio files are then saved.

In this project, the SSB modulation is performed based on finding the Hilbert transform of the modulation signal with multiplying it with a complex RF generator signal. The resultant output is a complex signal, in which the interest is in its real part only. Conversely, the SSB demodulation is performed based on shifting the input signal frequency, choosing both the baseband around zero and the upper or lower sideband and taking the resultant real part.

Based on the performed investigations, it can be concluded that the DSB-SC modulating system is the most efficient one, in which there is no lost power in the carrier signal. On the other hand, the SSB modulating system permits transmitting more information over the same channel based on hopping off the duplicate sideband. For furthermore improvements, several signal processing techniques can be included in the receiver side to get more efficient results.

REFERENCES

1. Dierker, M., “Chapter Three: Amplitude Modulation Fundamentals”, ***Press***, 93-117 (2007).

2. Sheets, W and Graf, R. F., “Modulation Methods SSB and DSB”, ***Press***, (2011).

3. McAbee, C., Tummala, M. and McEachen, J., "GSM to UMTS Network Handover Vulnerability Testing Using Software-Defined Radio," ***48th Hawaii International Conference on System Sciences (HICSS2015),*** 54225431, Jan, 5-8, (2015).

4. MdZahangir, A. and Abdus Sobhan, M., "Design of Future Software Defined Radio (SDR) for All-IP Heterogeneous Network", ***Recent Patents on Signal Processing*** 2 (2010): 12-21 (2010).

5. Weirather, R. D., "Amplitude Bandwidth, Phase Bandwidth, Incidental AM, and Saturation Characteristics of Power Tube Cavity Amplifiers for FM ", ***Harris Corporation,*** Quincy, Illinois, (1982).

6. Ronan, F., Sanchez, M. and Corley, G., "Software-defined radio demonstrators: An example and future trends", ***International Journal of Digital Multimedia Broadcasting***, (2009).

7. Kumm, M., “Modulation and Demodulation of Analog Signals”, ***EduRev,*** March 5 (2009).

8. Kammeyer, K. D. and Ubertragung, N., „3. Au. Number”, ***Teubner,*** Stuttgart, (2004).

9. Kammeyer, K. D., Korschel, K. and Signalverarbeitung, D., “Filtering und Spektralanalyse mit MATLAB-□Ubungen ; mit 33 Tabellen“, ***Teubner***, Wiesbaden, (2006).

10. Mitra, A., “Chapter 6 Transmitter and Receiver Techniques”, ***Press,*** (2009).

11. Mehta, V. K. and Mehta, R., “Chapter 16: Modulation and Demodulation, Principles of Electronics”, ***S. Chand & Company***, Ram Nagar, New Delhi (2014).

12. Kroupa, V. F., “Direct Digital Frequency Synthesizers”, ***IEEE Press***, Piscataway, NJ, (1998).

13. Lathi, B. P. and Ding, Z., “Modern Digital and Analog Communication Systems, 4th ed.” ***NY: Oxford University Press***, (2009).

14. Proakis, J. G. and Salehi, M., “Communication Systems Engineering, 2nd ed”, ***Singapore: Pearson Education, Inc.***, (2002).

15. Gardner, F. M., “Phaselock Techniques, third edition”, ***John Wiley & Sons, Inc.,*** New York, Chinchester, Brisbane, Toronto, (2005).

16. Kroupa, V. F., “Phase Lock Loops”, ***John Wiley & Sons Ltd,*** Chichester, England, (2003).

17. İnternet: HDSDR, “High Definition Software Defined Radio”, **http://www.hdsdr.de/** (2016).

18. İnternet: Telepostinc, “Preliminary, Using HDSDR with LP-PAN”, **www.telepostinc.com/hdsdr.html**, (2013).

19. İnternet: Giga Technology, “SoftRock RXTX Ensemble Transceiver Kit (SDR) (KIT)”, **www.giga.co.za/ocart/index.php?route=product/product&product_id=54,** (2016).

20. İnternet: Socialhams, “Introducing SoftRock Ensemble - A Software Defined Radio”, **https://www.socialhams.net/m/articles/view/Introducing-SoftRock-Ensemble-A-Software-Defined-Radio**, (2016).

21. İnternet: Audacity Team, “About Audacity”, **http://www.audacityteam.org/about/** (2016).

CV

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