Chapter1: Introduction

1-1: Overview:

The term *Software Defined Radio* ***(SDR)*** refers to reconfigurable or reprogrammable radios that can show different functionality with the same hardware. Because the functionality is defined in software, a new technology can easily be implemented in a software radio with a software upgrade. This thesis explores the design and implementation of a software defined radio.

Although most current SDR development was started in the military with initiatives like the US government's Joint Tactical Radio System (JTRS), the advantages of the approach extend far beyond military use, and the technology is now center stage for consideration in many commercial embedded applications as well. SDRs are becoming more common because of the capabilities of reconfigurable digital signal processing technologies such as field programmable gate arrays and digital signal processors, which place radio functions in firmware and software that were traditionally performed with analog hardware components. Software-defined radio can currently be used to implement simple radio modem technologies. In the long run, software-defined radio is expected by its proponents to become the dominant technology in radio communications.

The hardware of a software-defined radio typically consists of a super heterodyne RF front end which converts RF signals from (and to) analog IF signals, and analog to digital converter and digital to analog converters which are used to convert a digitized IF signal from and to analog form, respectively.

1-2: Objective:

This project objective is the design and implementation of software defined radio transceiver.

1-3: Methodology:

We used the **Matlab v7.5 to** to implement the software part in our deigned transceiver.

1-4: Thesis Layout:

- **Chapter1:** gives a general review about the software defined radio and the methodology used to implement it.
- **Chapter2:** this chapter provides description for the design of radio transceivers, generally. This description contains the transceiver type's common parts and design of receiver considerations.
- **Chapter3:** contains a brief of history, benefits and a detailed description about the SDR architecture.
- **Chapter4:** contains the SDR designed model and description for the receiver and transmitter components. Also it contains a demonstration of the matlab codes used to implement the software part of the modeled SDR transceiver.
- **Chapter5:** this chapter contains the results and the analysis of these results.

Chapter2: Design of radio transceivers

2-1: Introduction:

The RF Transceiver should initially be designed for implantable applications. Any communication system, in the simplest form, consists of a transmitter, a signal path, and a receiver. The performance of such systems depends heavily on each of the building blocks and the impact of the given communication link on the signal. Although the impact of the path is fixed by the frequency of the RF signal and the properties of the physical medium in which the signal propagates, the behavior of the transmitter and receiver can be flexible. The electrical performances of the transmitter and receiver determine the impact of these blocks on the signal and limit the quality and range of the communication link.

2-2: The receiver part:

2-2-1: Introduction:

The function of the receiver is to recover the audio signal that was modulated on the radio frequency carrier at the radio station, and apply it to the speaker. AM receiver can be categorized according to the method used to demodulate the signal. In coherent demodulation; an exact copy of the carrier signal is required. The carrier is then demodulated with the input RF signal which can then be low pass filtered to audio frequency (AF) signal. The disadvantage of coherent demodulation is that the frequency and phase of the carrier must be obtained almost exactly. This is almost impossible for the practical use in radios, so this method is almost never used. Incoherent detectors do not require a reproduction of the carrier signal, considerably simplifying the design. A large carrier term is required however, to ensure the envelope of the RF signal never crosses the x-axis. Incoherent receivers use an **envelope detector** to isolate the AF signal. This is usually performed in most practical circuits by a diode detector.

2-2-2: Common receiver components:

Many signals will be included in a receiver's antenna from the radio broadcast stations to unwanted noise. An important function of all receivers is to isolate the desired signal so that the information it contains can be used. This is done by passing the input signal through a band pass filter at a later stage in the receiver. After filtering, there of signal is in some cases amplified prior to demodulation. The demodulation process in most practical receivers is performed by an envelope detector. The output from the envelop detector however, is insufficient to drive a speaker and needs to be amplified.

2-2-3: Basic receiver types:

There are several kinds of receivers but they are of two basic types: straight receivers and super heterodyne receivers. In straight receivers the received signal is amplified at signal frequency, demodulated and then amplified again at audio frequency. Straight receivers can be either direct-conversion where the received signal is mixed with a locally generated carrier of the same frequency as the signal or super-regenerative where the demodulator is an oscillating mixer in which the oscillations are interrupted at a moderate frequency. In super heterodyne receivers, the incoming signal is first converted to a fixed intermediate frequency (IF) using a local oscillator. Then the signal is amplified at IF frequency, demodulated and the audio signal is further amplified. Super heterodyne receivers can provide single or multiple conversions. Single conversion takes place when a single IF is used and multiple when two or more IF’s are used.

Each receiver type has its advantages and disadvantage. Direct conversion receivers can provide any desired degree of selectivity by means of audio filters but they cannot provide high gain due to linear detection and high gain AF amplification is essential. On the other hand, super regenerative receivers provide high gain but they have poor performance in terms of selectivity and interference between receivers.

2-2-4: Receiver characteristics:

2-2-4-1:Sensetivity:

Receiver sensitivity is the lowest power level at which the receiver can detect an RF signal and demodulate data. Sensitivity is purely a receiver specification and is independent of the transmitter. As the signal propagates away from the transmitter, the power density of the signal decreases, making it more difficult for a receiver to detect the signal as the distance increases. Improving the sensitivity on the receiver (making it more negative) will allow the radio to detect weaker signals, and can obviously increase the transmission range. Sensitivity is vitally important in the decision making process since even slight differences in sensitivity can account for large variations in the range.

Receiver sensitivity is one of the key specifications of any radio. The two main requirements of any radio receiver are that it should be able to separate one station from another, i.e. selectivity, and signals should be selected and amplified so that they can be brought to a sufficient level to be heard. As a result receiver designers battle with many elements to make sure that these requirements are fulfilled.

A number of methods of measuring and specifying the sensitivity performance of radio receivers are used. Figures including signal to noise ratio, SINAD(Signal Noise And Distortion), noise factor and noise figure are used. These all use the fact that the limiting factor of the sensitivity of a radio receiver is not the level of amplification available, but the levels of noise that are present, whether they are generated within the radio receiver or outside it.

2-2-4-2: Selectivity:

Selectivity is one of the major specifications of any receiver. The selectivity ability to separate two closely spaced signals is a receiver's selectivity. The characteristics of the filter in the IF amplifier determine the frequency response of the IF stages and the selectivity. The narrower the filter pass-band, the "higher" the selectivity. The receiver

pass-band should be tailored to the characteristics of the incoming signal. Too wide a pass-band and unwanted noise is received which detracts from the reception of the wanted signal. We use bandwidth to measure selectivity. This is how wide a range of frequencies you hear with the receiver tuned to a set frequency. Receiver selectivity is a very important feature of your radio. Radios with poor selectivity may be subject to the effects of receiver desense and inter modulation.

2-2-4-3: Image rejection:

In reception using heterodyning in the tuning process, the **image rejection ratio**, or **image frequency rejection ratio**, is the ratio of the intermediate-frequency (IF) signal level produced by the desired input frequency to that produced by the image frequency. The image rejection ratio is usually expressed in dB. When the image rejection ratio is measured, the input signal levels of the desired and image frequencies must be equal for the measurement to be meaningful.

2-2-4-4: Tracking:

Tracking is the ability of the local oscillator to always remain the distance of the IF away from the antenna signal across the entire tuning range. Careful design of antenna and oscillator LC circuit becomes essential.

2-2-4-5: Dynamic range:

Receiver dynamic range is the measure of a receiver’s ability to handle a range of signal strengths, from the weakest to the strongest, i.e. the human senses of sight and hearing have a very high dynamic range. A person is capable of hearing (and usefully discerning) anything from a quiet murmur in a soundproofed room to the sound of the loudest rock concert. A good quality audio reproduction system should be able to reproduce accurately both the quiet sounds and the loud; and a good quality visual display system should be able to show both shadow details in nighttime scenes and the full brightness of sunny scenes.

In practice it is difficult to achieve the full dynamic range seen by human beings using electronic equipment, since most electronic reproduction equipment is essentially linear rather than logarithmic like human perception. Electronically reproduced audio and video often uses some trickery to fit original material with a wide dynamic range into a narrower recorded dynamic range that can more easily be reproduced.

In digital audio systems the dynamic range is limited by quantization error. The maximum achievable dynamic range for a digital audio system is approximately (6 * bit-depth) dB. For CD audio, which is 16-bits, the corresponding dynamic range is 96.33 dB.

2-2-4-6: Gain control:

It is unacceptable for the reproduction to vary widely at the output of the speaker with the varying input signal. Also there is no need to continue amplifying the desired signal if it is already a strong signal at the antenna, and hence need for automatic gain control.

Automatic gain control (AGC) is an adaptive system found in many electronic devices. The average output signal level is fed back to adjust the gain to an appropriate level for a range of input signal levels. For example, without AGC the sound emitted from an AM radio receiver would vary to an extreme extent from a weak to a strong signal; the AGC effectively reduces the volume if the signal is strong and raises it when it is weaker.

2-3: The transmitter:

2-3-1: Introduction:

The equipments used for generating and amplifying a radio-frequency carrier signal, modulating the carrier signal with the audio signal and feeding the modulated carrier to an antenna for radiation into space as electromagnetic waves.

2-3-2: main components:

Mainly, the transmitter has a power supply, an oscillator, a modulator, and amplifiers for audio frequency (AF) and radio frequency (RF). The RF power amplifier output must be connected to a correctly matched antenna (the “Load”) to work properly.

2-3-2: Transmitter design consideration:

2-3-2-1: Power output:

In broadcasting, and telecommunication, the part which contains the oscillator, modulator, and sometimes audio processor, is called the exciter. Confusingly, the high-power amplifier which the exciter then feeds into is often called the "transmitter" by broadcast engineers. The final output is given as transmitter power output (TPO), although this is not what most stations are rated by.

Effective radiated power (ERP) is used when calculating station coverage, even for most non-broadcast stations. It is the TPO, minus any attenuation or radiated loss in the line to the antenna, multiplied by the gain (magnification) which the antenna provides toward the horizon. This is important, because the electric utility bill for the transmitter would be enormous otherwise, as would the cost of a transmitter. For most large stations in the VHF- and UHF-range, the transmitter power is no more than 20% of the ERP.

2-3-2-2: Power supply:

Transmitters are sometimes fed from a higher voltage level of the power supply grid than necessary in order to improve security of supply.

2-3-2-3: Cooling of final stages:

Low-power transmitters do not require special cooling equipment. Modern transmitters can be incredibly efficient, with efficiencies exceeding 98 percent. However, a broadcast transmitter with a megawatt power stage transferring 98% of that into the antenna can also be viewed as a 20 kilowatt electric heater. For medium-power transmitters, up to a few hundred watts, air cooling with fans is used. At power levels over a few kilowatts, the output stage is cooled by a forced liquid cooling system analogous to an automobile cooling system.

2-3-2-4: Protection equipment:

The high voltages used in high power transmitters (up to 40 kV) require extensive protection equipment. Also, transmitters are exposed to damage from lightning. Transmitters may be damaged if operated without an antenna, so protection circuits must detect the loss of the antenna and switch off the transmitter immediately.

2-3-2-5: Building:

A commercial transmitter site will usually have a control building to shelter the transmitter components and control devices. This is usually a purely functional building, which may contain apparatus for both radio and television transmitters.

2-3-2-6: Planning:

As in any costly project, the planning of a high power transmitter site requires great care. This begins with the location. A minimum distance, which depends on the transmitter frequency, transmitter power, and the design of the transmitting antennas, is required to protect people from the radio frequency energy. Antenna towers are often very tall and therefore flight paths must be evaluated. Sufficient electric power must be available for high power transmitters. Transmitters for long and medium wave require good grounding and soil of high electrical conductivity. Locations at the sea or in river valleys are ideal, but the flood danger must be considered. Transmitters for UHF are best on high mountains to improve the range. The antenna pattern must be considered because it is costly to change the pattern of a long-wave or medium-wave antenna.

Chapter3: About SDR:

3-1: Why SDR? :

The high pace of technological advancement enables the realization of ever more advanced mobile communications standards with more functionality than simple voice communications. The hardware that is used to implement the radio sections of these systems generally require long design cycles, much longer than the design cycles of the other components of a communications system. Another problem is that, once new communications standards are introduced, the current hardware platforms used in the terminal equipment becomes obsolete because they can generally not be used with the new standards. This has serious cost implications for both the service provider and the consumer, because both parties have to acquire new equipment to be able to use the new standards.

An elegant solution to the above issues is to use software-defined radio sections to replace the hardware radio components. New communications standards can then be supported by simply loading new software onto the equipment, provided the maximum processing capacity of the processor(s) that the software runs on can accommodate the bandwidth requirements of that specific standard.

This thesis investigates the ideas behind software defined radio and also describes the design and implementation of a software defined radio transceiver on general-purpose platforms such as personal computers.

3-2: A Brief History of SDR:

The US Air Force's Integrated Communications Navigation and Identification Avionics (ICNIA) system offers an example of an early SDR developed in the late 1970's. The system used a reprogrammable digital signal processor (DSP) to operate multi-function multi-band airborne radios in the 30 MHz to 1,600 MHz spectrum. The ICNIA technology provided the foundation for many military-radio programs.

In the 1990s, a US government program specified the first fully programmable SDR system, Speakeasy; the so-called “PC of the communications world.” Speakeasy used open hardware and software architectures to support a family of voice, multimedia, and networking waveforms in the 2 MHz to 2 GHz frequency range.

Another government program, the Joint Tactical Radio System (JTRS), uses Speakeasy technology in a family of interoperable, multi-band, networked SDRs. The JTRS program aims to replace existing US-military radios with equipment that vendors can upgrade by downloading new software. This program relies on the Software Communications Architecture (SCA) open standard.

3-3: Analog Vs Digital receivers:

3-3-1: Introduction:

Digital receivers have revolutionized electronic systems for a variety of applications including communications, data acquisition, and signal processing. This series shows how digital receivers, the fundamental building block for software radio, can replace conventional analog receiver designs, offering significant benefits in performance and cost.

In order to fully appreciate the benefits of digital receivers, a conventional analog receiver system will be compared to its digital receiver counterpart, highlighting similarities and differences. The inner workings of the digital receiver will be explored with an in-depth description of the internal structure and the devices used.

3-3-2: Analog Receiver:

The conventional heterodyne radio receiver, as seen in figure (3-1), has been in use for nearly a century. Let's review the structure of the analog receiver so comparison to the digital receiver becomes apparent. First the RF signal from the antenna is amplified, typically with a tuned RF stage, that amplifies a region of the frequency band of interest.

This amplified RF signal is then fed into a mixer stage. The other input to the mixer comes from the local oscillator (LO) whose frequency is controlled by the tuning knob on the radio.

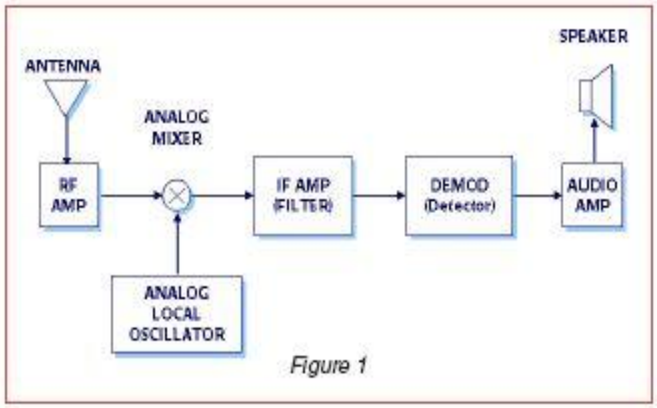


Figure (3-1): *Typical analog receiver block diagram.*

The mixer translates the desired input signal to the intermediate frequency (IF). (See figure (3-2)). The IF stage is a band pass amplifier that only lets one signal or radio station through. Common center frequencies for IF stages are 455 kHz and 10.7 MHz for commercial AM and FM broadcasts. The demodulator recovers the original modulating signal from the IF output using one of several different schemes.

For example, AM uses an envelope detector and FM uses a frequency discriminator. In a typical home radio, the demodulated output is fed to an audio amplifier which drives a speaker.

The mixer performs an analog multiplication of the two inputs and generates a difference frequency signal. The frequency of the LO is set so that the difference between the LO frequency and desired input signal (the radio station you want to receive) equals the IF.

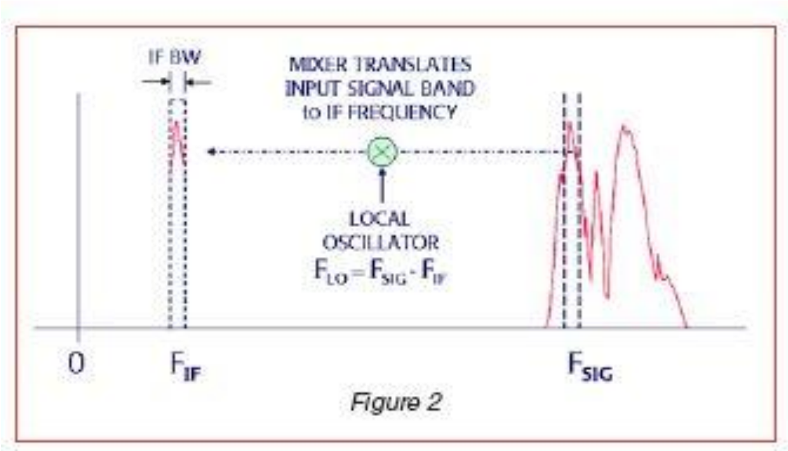


Figure (3-2): The mixer translates the desired input signal to the intermediate frequency.

For example, if you wanted to receive an FM station at 100.7 MHz and the IF is 10.7 MHz, you would tune the local oscillator to: 100.7 - 10.7 = 90 MHz . This is called "down conversion" or "translation" because a signal at a high frequency is shifted down to a lower frequency by the mixer. The IF stage acts as a narrowband filter which only passes a "slice" of the translated RF input. The bandwidth of the IF stage is equal to the bandwidth of the signal (or the "radio station") that you are trying to receive. For commercial FM, the bandwidth is about 100 kHz and for AM it is about 5 kHz. This is consistent with channel spacing of 200 kHz and 10 kHz, respectively.

3-3-3: Digital Receiver:

The digital receiver block diagram shown in Figure (3-3). We note the strong similarity to the analog receiver diagram; all of the basic principles of analog receivers still apply. Right after the RF amplifier and an optional RF translator stage, we use an A/D (analog-to-digital) converter to digitize the RF input into digital samples for the subsequent mixing, filtering and demodulation that are performed using digital signal processing elements.

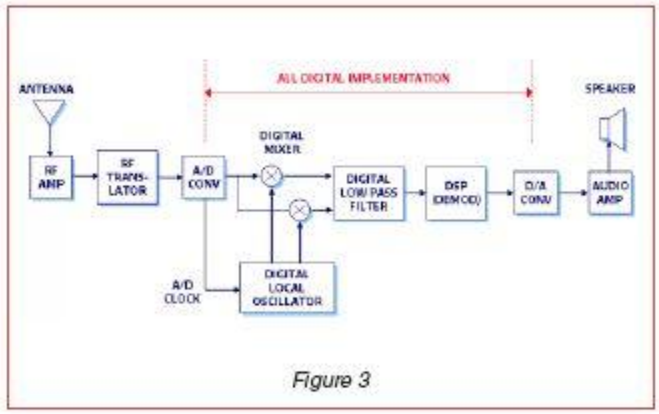


Figure (3-3): *Typical digital receiver block diagram.*

3-4:What is the software defined radio?

Software Defined Radio (SDR) is a collection of hardware and software technologies that enable reconfigurable system architectures for wireless networks and user terminals. SDR provides an efficient and comparatively inexpensive solution to the problem of building multi-mode, multiband, multi-functional wireless devices that can be enhanced using software upgrades. As such, SDR can really be considered an enabling technology that is applicable across a wide range of areas within the wireless industry.

SDR technology facilitates implementation of some of the functional modules in a radio system such as modulation/demodulation, signal generation, coding and link-layer protocols in software. This helps in building reconfigurable software radio systems where dynamic selection of parameters for each of the above-mentioned functional modules is possible. A complete hardware based radio system has limited utility since parameters for each of the functional modules are fixed. A radio system built using SDR technology extends the utility of the system for a wide range of applications that use different link-layer protocols and modulation/demodulation techniques.

A software defined radio is a radio in which the receive digitization is performed at some stage downstream from the antenna. This is typically after wideband filtering, low noise amplification, and down conversion to a lower frequency in subsequent stages, with a reverse process occurring for the transmit digitization. Created in an open-architecture software and hardware platform, SDR simply and effectively integrates wireless applications to operate over any air and mode interface by allowing both software and hardware to be adapted to most effectively handle any given task. This type of dynamic adaptability greatly improves battery life and system performance at significant cost savings. In addition, an SDR allows for quick and easy deployment of new value-added services since they can be instantly delivered over the air to both handsets and base stations. In summary, SDR technology effectively integrates all applications and systems to work over any air interface and protocol.

For network operators and service providers, SDR can have substantial impacts. First, it reduces costs by providing a generic hardware platform and common product line, which can then be customized for any unique application. SDR infrastructures allow for global purchasing and property and geographical risk management by adding flexibility throughout the network. SDR also eliminates concerns about air interfaces and frequency bands for roaming. This allows for simple and quick service differentiation through the ability to offer instant services on demand to one or all subscribers, technology independence from partnerships and market consolidation, and the use of new communications distribution channels by making it practical to embed SDR into computers and automobiles.

3-5: SDR Benefits:

The ‘Software Radio’ provides a flexible radio architecture that allows changing the radios personality, possibly in real-time, and in the process guarantees a desired Quality of Service (QoS). This is motivated by the numerous advantages of software radios.

In summary, the factors influencing the wider acceptance of software radios are:-

Ease of Design : The time required to develop a marketable product is a key consideration in modern engineering design, and software radio implementation reduces the design cycles for new products, freeing the designers from the hard-work associated with analog hardware designs.

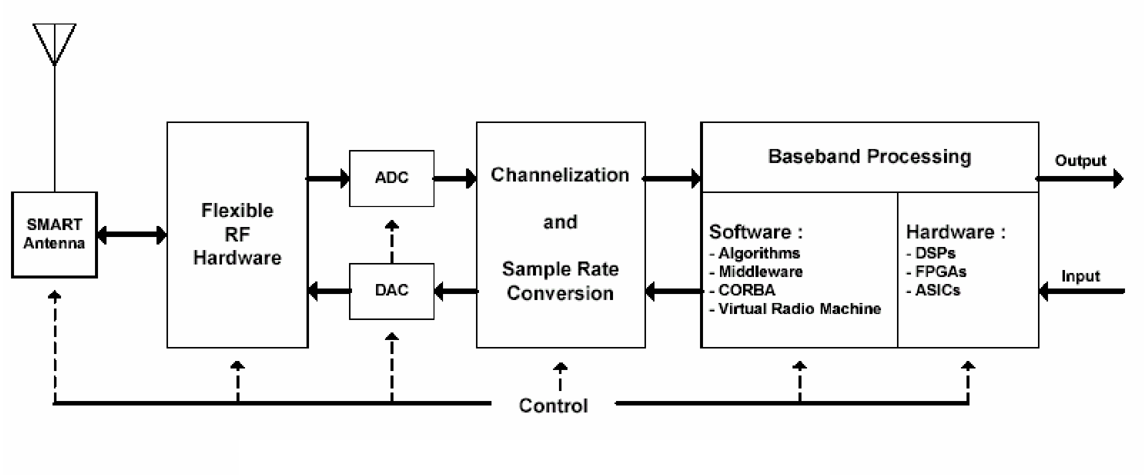
Ease of Manufacture: RF components are notoriously hard to standardize, and may have varying performance characteristics. Optimization of the components in terms of performance may take a significant amount of time and thereby delay product introduction. In general, digitization of the signal early in the receiver chain can result in a design that incorporates few discrete parts, resulting in a reduced inventory for the manufacturer.

Ease of upgrades : In the course of deployment, current services may need to be updated or new services may have to be introduced. Such enhancements have to be made without disrupting the operations of the current system. A flexible architecture like the SDR allows for improvements and additional functionality without the expense of replacing all the old units.

3-6: Architecture:

This section explains the architecture of a ‘Software Defined Radio’ and how the SDR technology can be used to implement complex radio functions in software. A generic model of a SDR system is shown in figure(3-4).

The SDR system consists of three main functional blocks: RF section, IF section, and the baseband section. The RF sections consists of analog hardware modules, while the IF and baseband sections contain digital hardware modules. The receiver begins with a SMART antenna that inhibits certain characteristics to minimize interference, multi-path, and noise. The SMART antenna provides similar benefits to the transmitter. Most software radios digitize the signal as early as possible using ADCs in the receiver chain, while keeping the signal in the digital domain and converting to the analog domain as late as possible for the transmitter using DAC. Often, the received signals are digitized in the Intermediate Frequency (IF) band.



Figure(3-4): *Block diagram of SDR transceiver*

Software Defined Radios (SDR) usually employ a super heterodyne receiver, in which the RF signal is picked up by the antenna along with spurious signals, filtered, amplified with a low noise amplifier (LNA), and mixed with a local oscillator (LO) to IF. Depending on the required application, the number of stages can be varied. On the transmitter, the IF signals are mixed with a local oscillator (LO) to be up-converted to RF. The signal is then passed through a power amplifier (PA) before it is sent to the antenna.

On the receiver, the IF signals are then mixed exactly to baseband. Digitizing the signal with an ADC in the IF range eliminates the last stage in the conventional model in which problems like carrier offset and images/replicas are encountered. Digital filtering (Channelization) and sample rate conversion are needed to interface the output of the ADC to the baseband processor used to implement the receiver. Likewise, digital filtering and sample rate conversion are often necessary to interface the digital hardware that creates the modulated signals to the DACs. Baseband Processing is performed in software using Digital Signal Processors (DSP), Field Programmable Gate Arrays (FPGA), or Application Specified Integrated Circuits (ASIC). The baseband sections performs complex mathematical operations /algorithms like timing synchronization, equalization, channel estimation, cross-polarization interference cancellation, maximum likelihood algorithms, modulation, demodulation, encryption, decryptions, error control coding / correction, source coding, interleaving, correlation, etc..

If the programmability is further extended to the RF section by replacing the discrete components, an ideal Software Defined Radio (SDR) that supports programmable RF bands can be implemented. However, the current generation ADC or DAC devices cannot support operations in the frequencies required to implement this system in a commercially viable manner.

Selecting DSP ,FPGA OR ASIC:

several general guidelines can be developed for the partitioning of a software defined radio design between ASIC, FPGA, and DSP devices. These guidelines can be summarized as follows:

- ASICs typically offer the best solution for a given function, if they provide an acceptable level of "programmability" and integration.
- FPGAs provide the best programmable solution for high-speed signal processing functions that are highly involve linear processing.
- DSPs provide the best programmable solution for functions that involve complex analysis or decision-making.

Going forward, DSPs ASICs, and FPGAs will continue to support more functionality on chip, blurring the line between these products even more

The DSP benefits:

Complex functions of protocol-specific algorithms are perfectly suited to programmable DSPs since they combine number-crunching power with intelligence to enable

Multi-tasking. The DSPs math capability allows it to handle all the filtering of massive amounts of data as well as the multiple protocol standards that require error correction.

3-7: APPLICATION:

SDR is being touted as the future of wireless communications. The days of custom Asics and radio hardware may be numbered, giving way to a new era where upgrades and reconfigurations of wireless equipment requires only a new software load. SDR has the potential to open new business opportunities for carriers and cellular providers as well as the entire handset market. However none of these opportunities are expected to happen overnight. Several key development milestones are yet to be reached in order to move SDR technologies into the mainstream market.

Although most current SDR development was started in the military with initiatives like the US government's Joint Tactical Radio System (JTRS), the advantages of the approach extend far beyond military use, and the technology is now center stage for consideration in many commercial embedded applications as well. SDR offers flexibility and field-upgradeability to RF systems that simply cannot be matched by traditional analog-based RF design techniques.

Chapter4: Modeling & Simulation

4-1: The designed model:

The designed model was consist of radio transmitter and receiver shown in figure (4-1) and (4-2) respectively. The transmitter consists of an analog to digital converter, modulator, digital up converter, digital to analog converter, power amplifier and transmit antenna. The receiver consists of band pass filter, low noise amplifier, analog to digital converter, digital down converter, demodulator, digital to analog converter, power amplifier and PC speaker. These parts explained with more details in the following sections in the same order in the block diagrams.

4-2: The transmitter:

4-2-1: The Analog to Digital Converter:

An analog-to-digital converter (abbreviated ADC, A/D or A to D) is an electronic integrated circuit, which converts continuous signals to discrete digital numbers. The reverse operation is performed by a digital-to-analog converter (DAC). The digital output may be using different coding schemes, such as binary, Gray code or two's complement binary. The *sound card* of the PC is used to convert the audio signal to the digital form.

A **sound card** (also known as an audio card) is a computer expansion card that facilitates the input and output of audio signals to/from a computer under control of computer programs. Typical uses of sound cards include providing the audio component for multimedia applications such as music composition, editing video or audio and entertainment (games). Many computers have sound capabilities built in, while others require additional expansion cards to provide for audio capability.

There are three parameters you can adjust on a sound card to tailor your audio quality. The first step is to choose whether you're digitizing in stereo or mono. Capturing in stereo doubles the file size, and increases the demand on your computer. The second parameter is sampling rate. This tells the computer how often to “measure” the analog signal and stores a numerical value. Audio sampling rate has a direct impact on the

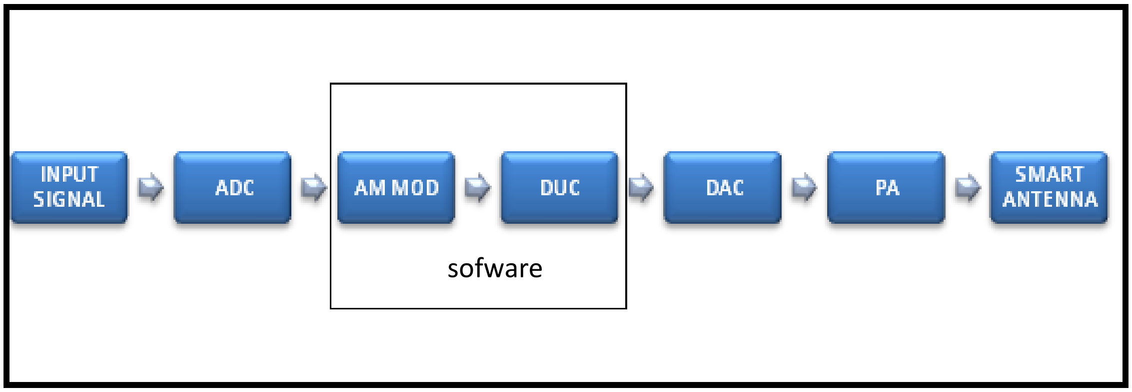


Figure (4-1): *Block diagram of SDR transmitter*

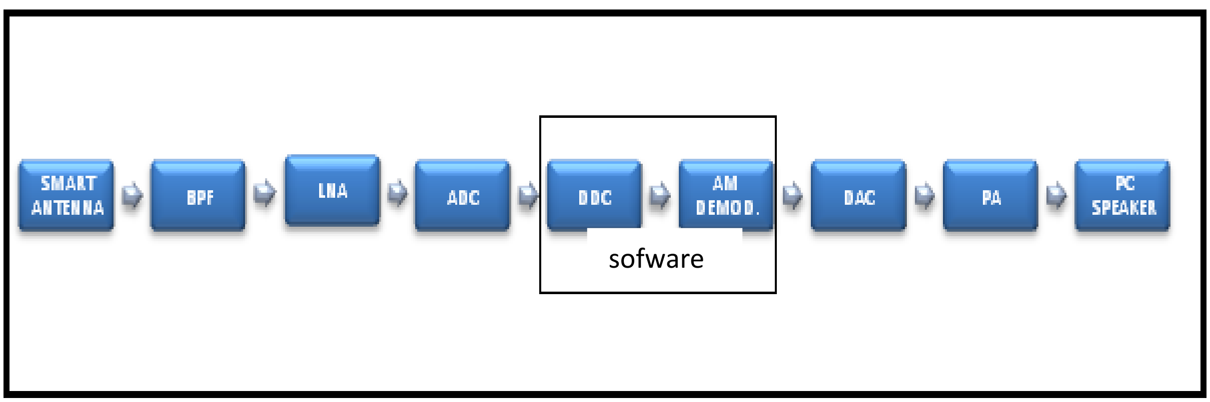


Figure (4-2): *Block diagram of SDR receiver*

frequency response of your recordings; the sampling rate must be double that of the highest frequency you want to capture. An 8 kHz sampling rate, for example, will only capture frequencies up to about 4 kHz. Hence, the lower the sampling rate, the more "dull" your audio sounds. The last parameter is bit depth or "word length." This sets how many bits the computer uses to store each measurement of the audio signal. The more bits used, the more accurate the recording (and the larger the resulting file). Using fewer bits can make for a coarse-sounding, noisy recording.

At the heart of a sound card is the audio digitizer. A digitizer turns an analog waveform- in this case sound- into a series of numbers. The computer can then manipulates these numbers and stores them on a drive. On playback, the sound card converts the numbers back into an audible signal.

4-2-2: The modulator:

There are two main reasons for modulation. The first reason has to do with the laws of electromagnetic propagation, which dictate that the size of the radiating element, the antenna, be a significant fraction of the wavelength of the signal being transmitted. For example, if we want to transmit a 1 kHz signal by a quarter wave antenna, the size of the antenna would need to be 75 km. On the other hand, if the signal is being transmitted on a high frequency carrier, say 630 kHz, the corresponding size of the radiating antenna needs to be only 119 m.

The second reason is for the simultaneous transmission of different signals. As audio signals relevant to humans lie from a few hertz to a few thousand hertz, we could broadcast only one baseband signal at a time. Simultaneous transmission would cause the overlap of signals and we would not be able to separate them. However, through modulation, we can transmit many signals *simultaneously* by shifting their spectra using different carrier frequencies. This is called frequency division multiplexing (FDM).

The audio (sound) or data signal is modulated on to the radio frequency “Carrier” using several types of modulation. The main types are the amplitude modulation (AM) and the frequency modulation (FM). The used modulation is the *amplitude modulation*.

Amplitude modulation is done by adding a dc offset to the baseband signal, m(t) and then multiplying by a sinusoid of frequency f_c .The formula for an amplitude modulated signal S (t) is as follows:

$$S(t) = Ac[1 + k*m(t)]*\cos(2\Pi fct)$$

and the signal has a shape shown in figure (4-3) below.

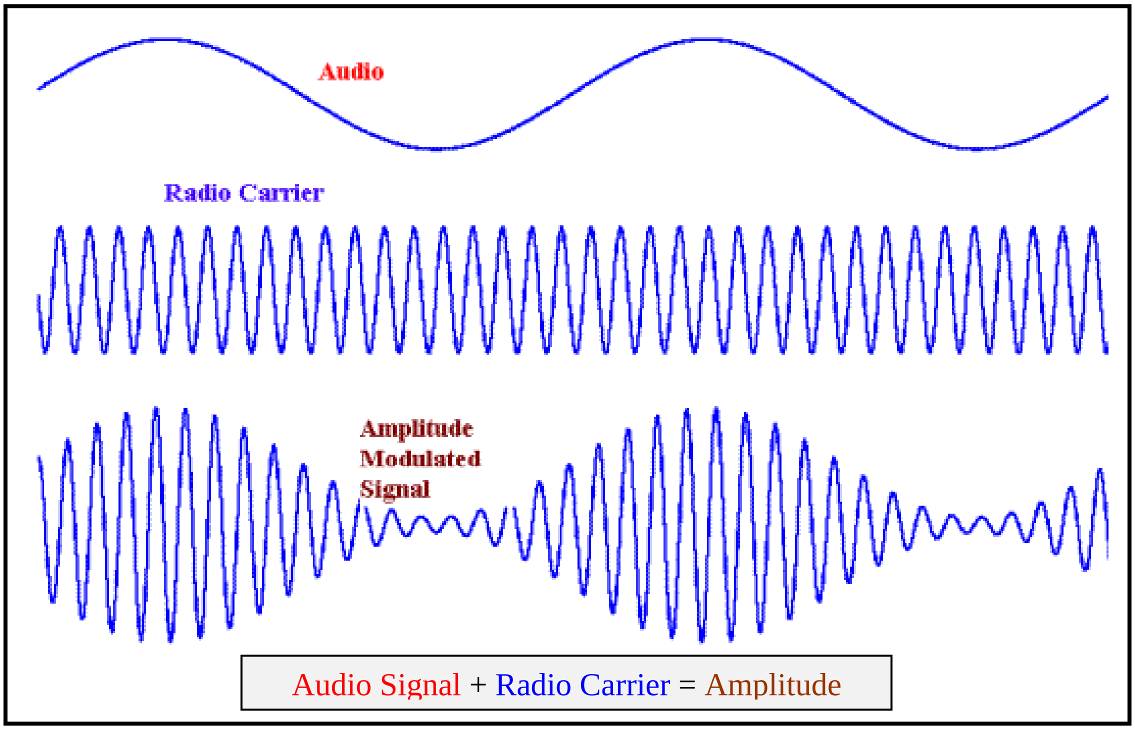


Figure (4-3) : *Amplitude modulation*

4-2-3: The digital up converter (DUC):

A DUC is a digital circuit which converts a digital baseband signal to a pass band signal. The input baseband signal is sampled at a relatively low sampling rate, typically the digital modulation symbol rate. The baseband signal is filtered and converted to a higher sampling rate.

The input signals are passed through three filtering stages. Each stage first filters the signals with a low pass interpolating filter and then performs a sampling rate change. The DUC in this model is a cascade of two FIR Interpolation Filters and one CIC Interpolation Filter. The first FIR Interpolation Filter is a pulse shaping FIR filter that increases the sampling rate by 2. The second FIR Interpolation Filter is a compensation FIR filter that increases the sampling rate by 2 and compensates for the distortion of the following CIC filter. The CIC Interpolation Filter increases the sampling rate by 2. More details about FIR & CIC filters in the following sections.

4-2-3-1: The FIR Filter:

A **finite impulse response (FIR)** filter is a type of a digital filter. The impulse response, the filter's response to a delta input, is 'finite' because it settles to zero in a finite number of sample intervals. This is in contrast to infinite impulse response filters which have internal feedback and may continue to respond indefinitely.

Properties:

An FIR filter has a number of useful properties which sometimes make it preferable to an infinite impulse response filter. FIR filters:

- Are inherently stable. This is due to the fact that all the poles are located at the origin and thus are located within the unit circle.
- Require no feedback. This means that any rounding errors are not compounded by summed iterations. The same relative error occurs in each calculation.
- They can be designed to be linear phase, which means the phase change is proportional to the frequency. This is usually desired for phase-sensitive applications.

4-2-3-2: Cascaded Integrator-Comb (CIC) Filter:

CIC filters are multi-rate filters that are very useful because they can achieve high decimation (or interpolation) rates and are implemented without multipliers. CICs are simply boxcar filters implemented recursively cascaded with an up sampler or down sampler. These characteristic make CICs very useful for digital systems operating at high rates, especially when these systems are to be implemented in ASICs or FPGAs.

Although CICs have desirable characteristics they also have some drawbacks, most notably the fact that they incur attenuation in the pass band region due to their sinc-like response. For that reason CICs often have to be followed by a compensating filter. The compensating filter must have an inverse-sinc response in the pass band region to lift the droop caused by the CIC.

4-2-4: Power Amplifier:

Generally, an amplifier is any device that will convert one signal (often with a very small amount of energy, a few milliwatt) into another signal (often with a larger amount of energy e.g. several hundred watts).

In popular use, the term today usually refers to an electronic amplifier, often as in audio applications. The relationship of the input to the output of an amplifier — usually expressed as a function of the input frequency — is called the transfer function of the amplifier, and the magnitude of the transfer function is termed the gain.

The term *amplifier* is very generic. In general, the purpose of an amplifier is to take an input signal and make it stronger (or in more technically correct terms, increase its *amplitude*). Amplifiers find application in all kinds of electronic devices designed to perform any number of functions. There are many different types of amplifiers, each with a specific purpose in mind. In this case, a radio transmitter uses an RF Amplifier; such an amplifier is designed to amplify a signal so that it may drive an antenna.

Power is not really something that can be "amplified". *Voltage* and *current* can be amplified. The term "power amplifier" although technically incorrect has become understood to mean an amplifier that is intended to drive a load (such as a speaker, a motor, etc).

4-2-5: Smart antnna:

Smart antenna systems can adaptively point the main antenna beam in the direction of a desirable transceiver and point one or more antenna nulls towards interfering signals. Alternatively, smart antennas can be made to resonate at different frequencies, depending upon the need to emulate different radios or use a different part of the spectrum with fewer interferers.

4-3: The receiver:

4-3-1: band pass filter:

A **band-pass filter** is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range. An example of an analogue electronic band-pass filter is an RLC circuit (a resistor–inductor–capacitor circuit). These filters can also be created by combining a low-pass filter with a high-pass filter

An ideal filter would have a completely flat pass band (e.g. with no gain/attenuation throughout) and would completely attenuate all frequencies outside the pass band. Additionally, the transition out of the pass band would be instantaneous in frequency. In practice, no band pass filter is ideal. The filter does not attenuate all frequencies outside the desired frequency range completely; in particular, there is a region just outside the intended pass band where frequencies are attenuated, but not rejected. This is known as the filter roll-off, and it is usually expressed in dB of attenuation per octave or decade of frequency. Generally, the design of a filter seeks to make the roll-off as narrow as possible, thus allowing the filter to perform as close as possible to its intended design. Often, this is achieved at the expense of pass-band or stop-band ripple. The bandwidth of the filter is simply the difference between the upper and lower cutoff frequencies.

4-3-2: The low noise amplifier:

The **low noise amplifier (LNA)** is a special type of electronic amplifier or amplifier used in communication systems to amplify very weak signals captured by an antenna. It is often located very close to the antenna. If the LNA is located close to the antenna, then losses in the feed line become less critical. It is a key component, which is placed at the front-end of a radio receiver circuit. Using an LNA, the noise of all the subsequent stages is reduced by the gain of the LNA and the noise of the LNA is injected directly into the received signal. Thus, it is necessary for an LNA to boost the desired signal power while adding as little noise and distortion as possible so that the retrieval of this signal is possible in the later stages in the system.

4-3-3: The Analog to Digital Converter:

The typical architectural approach to SDR receiver locates a wideband analog-to-digital converter (ADC) very near the antenna of an RF system in order to sample and digitize incoming RF waveforms. The channel modulation scheme is therefore implemented completely in software, so down-conversion and demodulation happen entirely in the digital domain, typically on a digital signal processor (DSP) or other general purpose processor.

4-3-4: The Digital Down Converter:

Digital Down-Converter (DDC) is a key component of digital radio receivers. The DDC performs the frequency translation necessary to convert the high input sample rates found in a digital radio receiver, down to lower sample rates for further and easier processing.

4-3-5: The demodulator:

The used demodulator is an envelope detector. An envelope detector is an electronic circuit that takes a high-frequency signal as input, and provides an output which is the "envelope" of the original signal. The simplest form of envelope detector is the diode detector. A diode detector is simply a diode between the input and output of a circuit, connected to a resistor and capacitor in parallel from the output of the circuit to the ground. If the resistor and capacitor are correctly chosen, the output of this circuit should approximate a voltage-shifted version of the original (baseband) signal. A simple filter can then be applied to filter out the DC component.

An envelope detector can be used to demodulate a previously modulated signal by removing all high frequency components of the signal. The capacitor and resistor form a low-pass filter to filter out the carrier frequency. Such a device is often used to demodulate AM radio signals because the envelope of the modulated signal is equivalent to the baseband signal.

An envelope detector is sometimes referred to as an envelope follower in musical environments. It is still used to detect the amplitude variations of an incoming signal to produce a control signal that reassembles those variations. However, in this case the input signal is made up of audible frequencies. Modern envelope followers can be implemented directly as electronic hardware or completely in software and in this design we use the **software implementation**.

4-3-6: The Power Amplifier:

The purpose of a power amplifier, in very simple terms, is to take a signal from the signal processor and make it suitable for driving a loudspeaker. Ideally, the ONLY thing different between the input signal and the output signal is the *strength* of the signal. In mathematical terms, if the input signal is denoted as **S**, the output of a *perfect* amplifier is **X*S**, where X is a *constant* (a fixed number). Audio power amplifiers are which designed to drive loudspeakers

4-3-7: The PC speaker:

The PC speaker was often used in very innovative ways to create the impression of polyphonic music or sound effects within computer games. Several programs, including MP (Module Player, 1989), ScreamTracker, Fast Tracker, Impulse Tracker, and even a Microsoft Windows device driver, could play pulse-code modulation (PCM) sound through the PC speaker.

In this project the PC speaker was the device used to send audio signals to it.

4-4: Implementation & Simulation

4-4-1: Introduction:

4-4-1-1:Sampling:

Sampling consists of a “sample and hold” circuit followed by conversion a digital code word, and is implemented using a chip called an ADC (analogue to digital converter). For an audio signal, the output of the ADC will be a code word which is a binary word whose number in binary represents the amplitude of the sampled signal. A typical number of bits for an ADC that is used to sample an audio signal will be 16 or 24 bits. This means there will be 2^16 or 2^24 possible quantisation levels respectively. An ADC used for audio would use linear quantisation which means that all the quantisation levels are equally spaced. Because the digital sample values are discrete there will be a small error between the actual analogue voltage and the encoded amplitude. This error is called quantisation noise. Increasing the number of bits of the ADC decreases the voltage difference between adjacent levels (increases the resolution) and therefore the quantisation noise is reduced.

The job of the ADC is to store the amplitude value of the analogue signal as a digital number which is proportional to the amplitude of the analogue signal voltage. This is called pulse code modulation. As these numbers are in binary format the number of bits defines how many discrete levels can be used. For an N-bit PCM codeword there are 2^N different codewords, for example, if an 8-bit codeword was used then there are 2^8 = 256 discrete amplitudes and therefore discrete codewords that can be used.

4-4-1-2: Signal Reconstruction:

To reconstruct the analogue signal that was sampled by the ADC and encoded as a sequence of PCM numbers, a digital to analogue converter (DAC) is used in conjunction with a lowpass filter. The DAC does the opposite to that which the ADC does. That is, it takes a PCM encoded number (for example a 16- or 24-bit number) and outputs an analogue voltage that is proportional to the value that the PCM number signifies. This analogue voltage is then held constant by a sample-and-hold circuit so the analogue waveform consists of a sequence of rectangular pulses all next to each other and with different amplitudes. To finally reconstruct the original signal these rectangular pulses are sent through a low pass filter. This smoothes the waveform to produce exactly the original signal assuming a perfect low pass filter. Theoretically, the sampling of the original signal, followed by reconstruction using a DAC and low pass filter will perfectly reconstruct the original signal. There will in practice be small errors because it is impossible to construct perfect 'brick-wall' filters but it is possible to obtain a reconstructed signal with a very small error.

4-4-1-3: Decimation:

In digital signal processing, decimation is a technique for reducing the number of samples in a discrete-time signal.

Decimation is a two-step process:

1. Low-pass filtering.
2. Down sampling.

4-4-1-4: Interpolation:

In the mathematical subfield of numerical analysis, interpolation is a method of constructing new data points within the range of a discrete set of known data points. In engineering and science one often has a number of data points, as obtained by sampling or experiment, and tries to construct a function which closely fits those data points. This is called curve fitting or regression analysis. Interpolation is a specific case of curve fitting, in which the function must go exactly through the data points. Suppose we know the function but it is too complex to evaluate efficiently. Then we could pick a few known data points from the complicated function, creating a lookup table, and try to interpolate those data points to construct a simpler function. Of course, when using the simple function to calculate new data points we usually do not receive the same result as when using the original function, but depending on the problem domain and the interpolation method used the gain in simplicity might offset the error.

4-4-2: Implementation of the designed model:

First step was the audio signal recording. We record the audio data and then play it back using the Windows Sound Recorder panel. To access this application, select the following: Start > Programs > Accessories > Entertainment > Sound Recorder. The figure below shows how to record and play data.

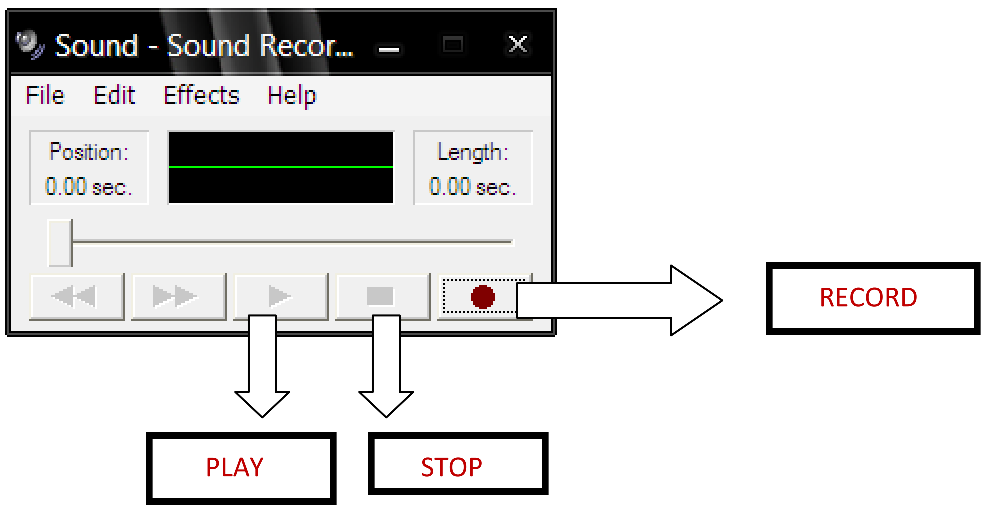


Figure (4-4): *Windows sound recorder*

The recorded audio data saved as wav file in the matlab work file, so that it can be read and processed using the matlab functions.

We used the 'wavread' function to load the WAVE file specified by its name, into variable named x. The variable x was containing the audio samples that had sampling frequency of 44,100 KHz. The 'wavread' supports Pulse-code Modulation (PCM) data format ***only.***

The second step was the modulation of the audio signal. AM modulation was used in this part. The audio signal was multiplied with the carrier signal sample by sample to give the signal shown in figure (5-4).

After that the modulated signal was filtered and converted to a higher sampling rate using digital up converter. The input signals are passed through three filtering stages.

Each stage first filtered the signals with a low pass interpolating filter and then performed the sampling rate change. The DUC in this model is a cascade of two FIR Interpolation Filters and on CIC Interpolation Filter. The first FIR Interpolation Filter is a pulse shaping FIR filter that increases the sampling rate by 2 and performs transmitter Nyquist pulse shaping. The second FIR Interpolation Filter is a compensation FIR filter that increases the sampling rate by 2 and compensates for the distortion of the following CIC filter. The CIC Interpolation Filter increases the sampling rate by 2.

The ‘mfilt.firinterp’ function was used to create the two FIR interpolator filters and the ‘mfilt.cicinterp’ matlab function to create the CIC filter.

In the receiver part the reverse operations was manipulated on the output signal of the DUC. The first operation was the digital down conversion. Also, the DDC was consist of three stages. The signal low pass filtered by a Cascaded Integrator-Comb (CIC) filter followed by two FIR decimating filters to achieve a low sampling rate. Each of three filters gave a decimation by 2.

The ‘mfilt.cicdecim’ function was used to create the CIC filter. The first thing to note is that the CIC filter has a huge pass band gain, which is due to the additions and feedback within the structure. We can normalize the CIC's magnitude response by cascading the CIC with a gain that is the inverse of the gain of the CIC. Normalizing the CIC filter response to have 0 dB gain at DC will make it easier to analyze the overlaid filter response of the next stages filters.

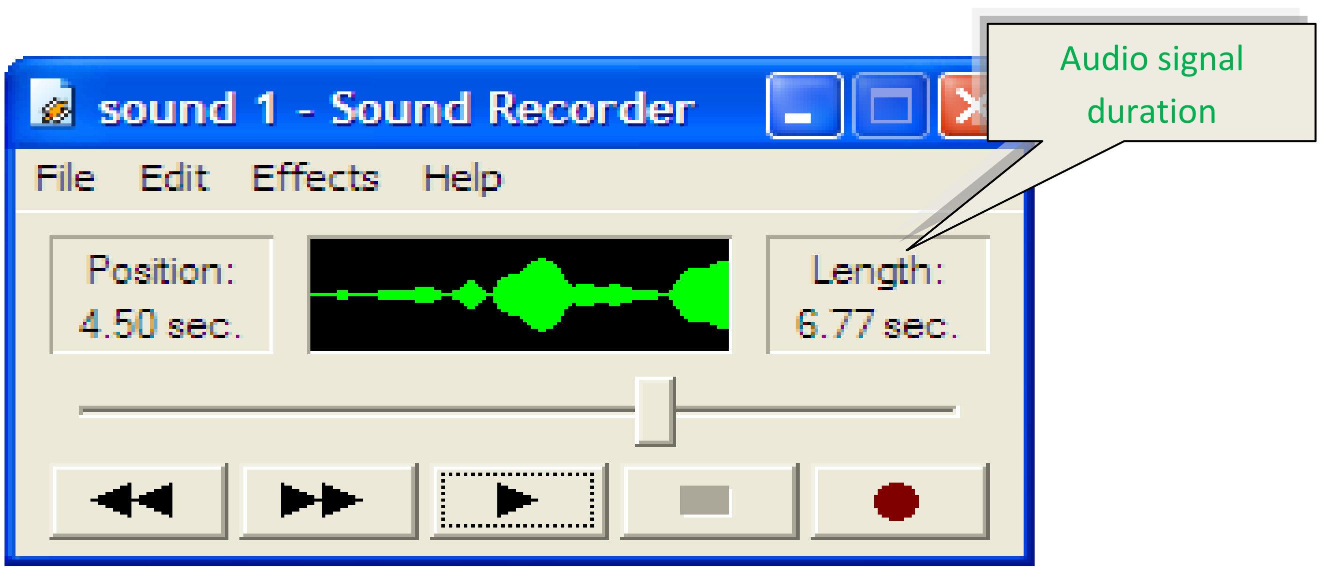
A CIC filter is essentially a cascade of boxcar filters and therefore has a sinc-like response which causes the droop. This droop needs to be compensated by the FIR filter in the next stage. The second stage of our DDC filter chain needs to compensate for the pass band droop caused by the CIC and decimate by 2. Since the CIC has a sinc-like response, we can compensate for the droop with a low pass filter that has an inverse-sinc response in the pass band.

The ‘mfilt.firdecim’ was used to implement the decimation in the next two stages. The signal is decimated in the two stages with a factor of two.

The last operation was the demodulation of the DDC output signal. The demodulation was done by a simulation of an envelope detector. The filtered signal was first converted to an exponential signal using a Hilbert transform. The magnitudes of the resulted signal were the amplitudes of the original signal with the addition to a DC offset equal to mean value of gotten signal.

Chapter5: results& results analysis

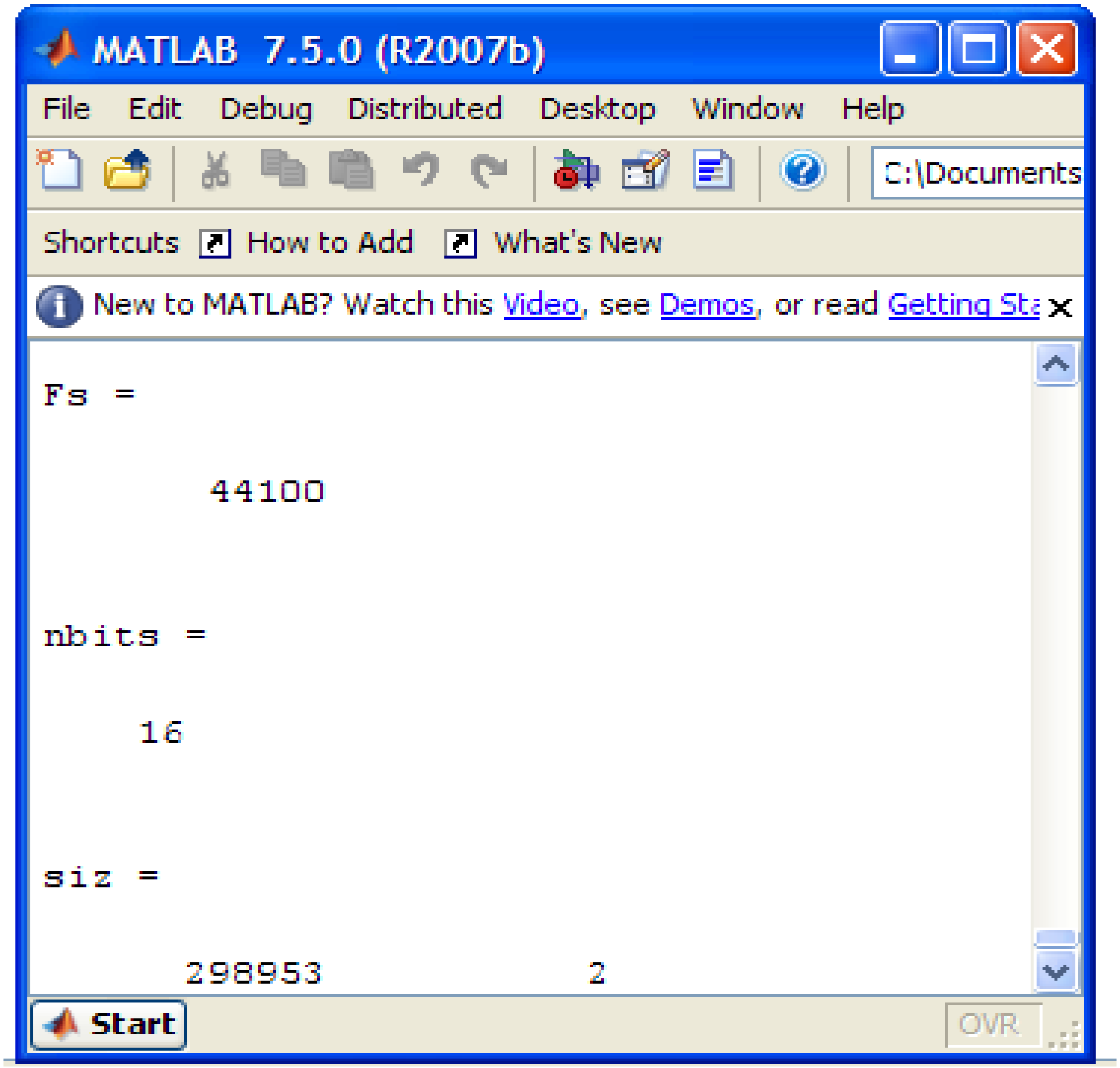
This chapter will have a discussion of the results. The first step was the audio signal recording. We used the windows sound recorder and the file saved in the matlab work as WAVE file, so that it can be loaded to a defined variable and then processed. Figure (5-1) below show the duration of the recorded signal.



Figure(5-1): *The windows sound recorder*

The recorded signal has a sampling rate of 44100 samples per second. This sampling rate should be specified when the signal is to be sent to the PC speaker.

The result of the 'size' function which used to get the audio signal size is shown in figure (5-2) below.



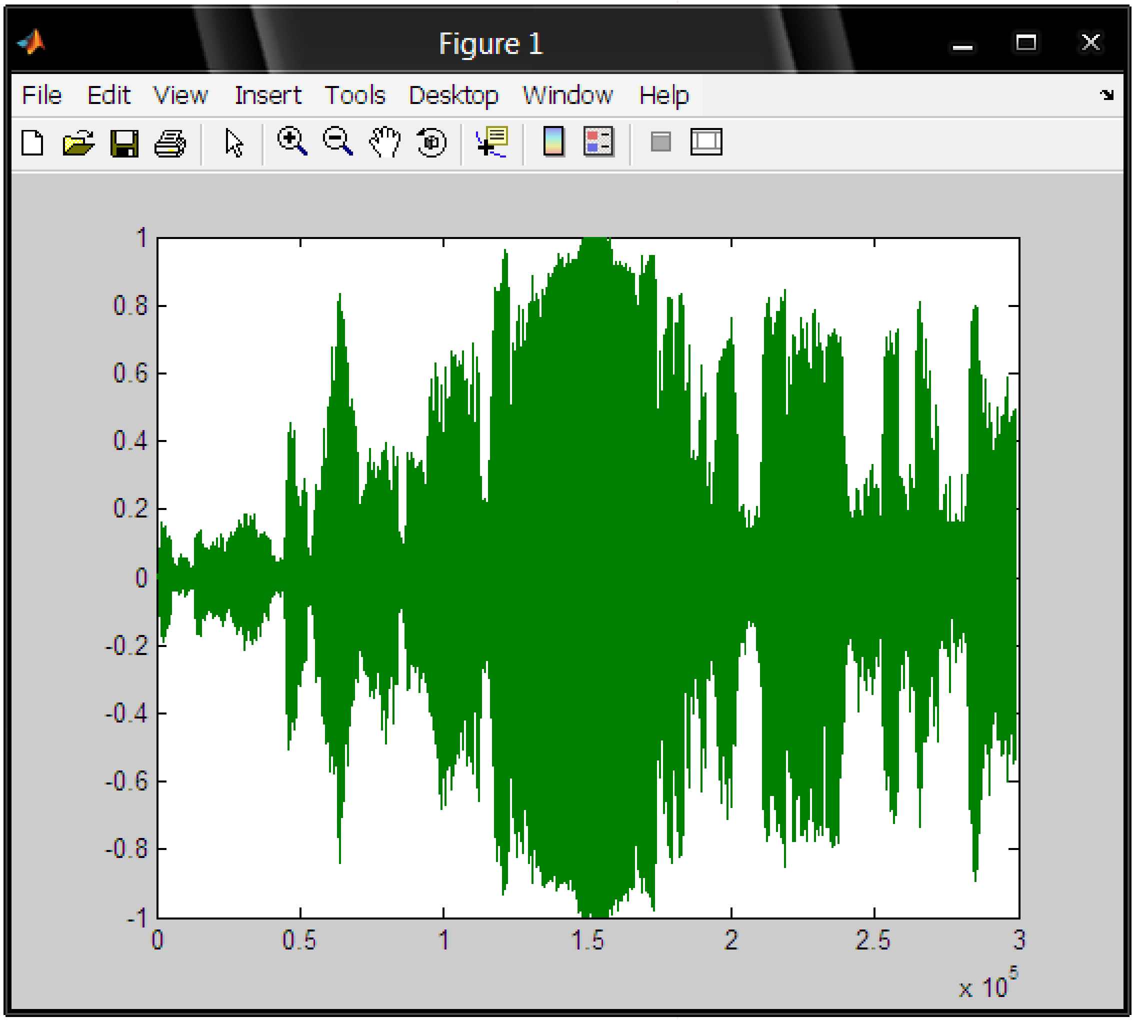
Figure(5-2): *The recorded signal sampling rate and size*

Fs : Sampling rate

nbits : number of bits per second.

siz : signal size (number of signal samples)

Figure (5-3) below show the plot of the recorded audio signal. The figure shows that the audio signal has amplitude variations according to voice intensity. These variations has a limited range of values between the -1 & +1. This is because the audio wave stored in the wave file must have values in this range only.



Figure(5-3): *The recorded audio signal*

The size of the signal was gotten to be 298953 samples. If we divide this number of samples by the signal sampling rate the answer will be 6.77898 which is the same duration of the signal shown in figure (5-1).

Another thing to observe was that the plotted signal seems to be continuous. This is because of the large number of samples. When zooming this signal as shown in figure (5-4) we can observe that this signal is consist of large number of discrete samples.

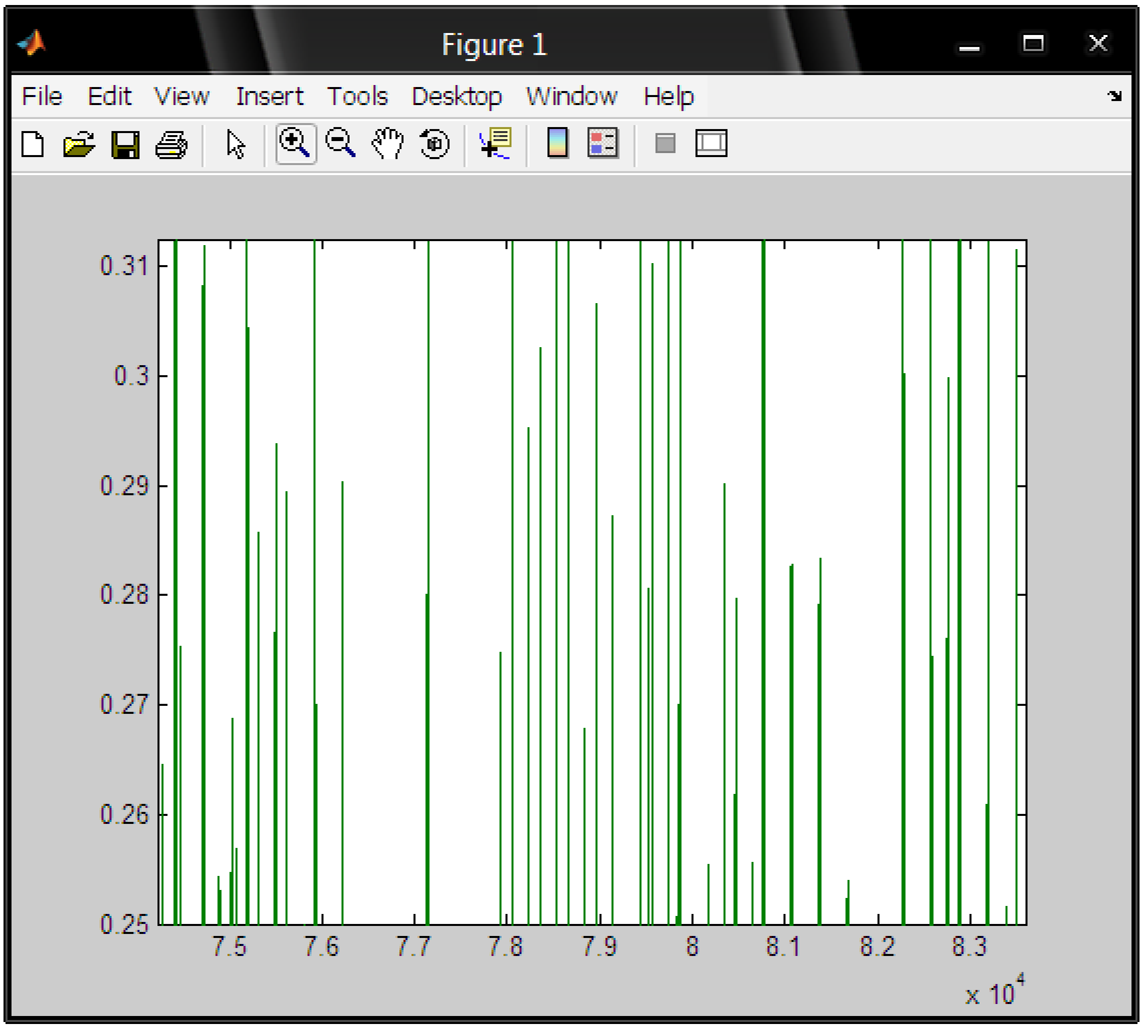


Figure (5-4): *Number of samples when zooming the audio signal*

The audio signal was then modulated and processed using FIR & CIC filters. The resultant signals plots are shown in the following figures.

The carrier signal was modulated with the recorded audio signal sample by sample and the resulted signal is shown in figure (5-5). This is can be done only if the carrier has a sampling frequency which is same as the audio signal, so that, the sampling rate of the carrier was the same as the recorded signal. We chose that the amplitude of the carrier to be 1. From figure (5-5) we observe that the maximum value for the modulated signal is 2 which is equal to the maximum value of the audio signal added to the maximum carrier amplitude. The resulted modulated signal has the same sampling frequency. After that the sampling rate of the signal had to be increased using digital up conversion.

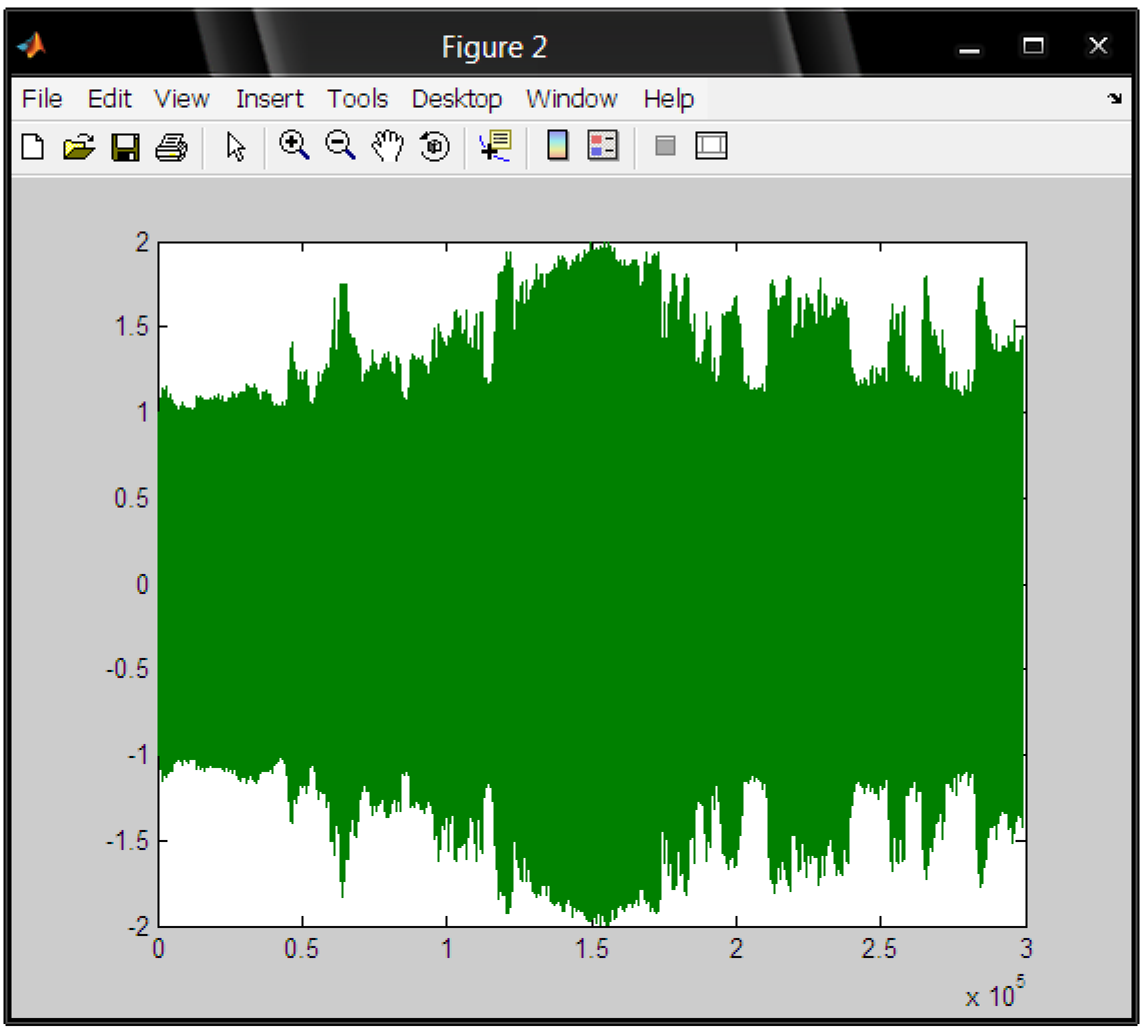
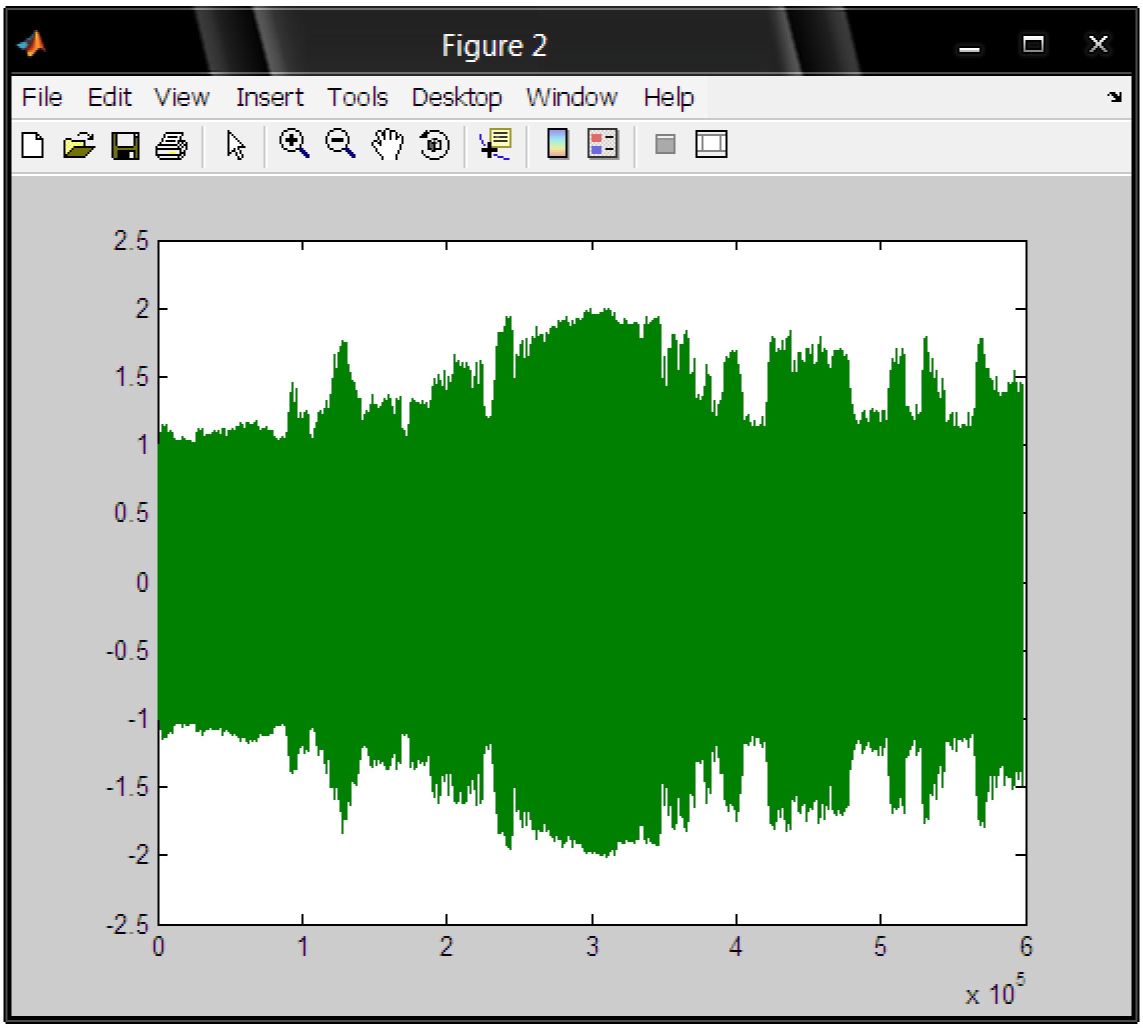


Figure (5-5): ***The modulated signal***

The digital up conversion was obtained by using three interpolation filters. The output of the first FIR filter is shown n figure (5-6). We observe that the signal has the same shape as modulated signal. The only difference is the length of the signal (total number of samples). The size of this signal is 597906 samples, which is equal to the size of the modulated signal multiplied by two.

The output of this filter seems to be the same as its input signal. This is because the interpolation (sampling rate increment) is done by curve fitting. The curve fitting fit the new samples between the signal samples to get the desired higher sampling rate.



Figure(5-6): *The output of the first FIR filter*

The second interpolation filter was also an FIR filter. The output of this filter is shown in figure (5-7). Second FIR also has an interpolation factor equals two. This filter is used to compensate the distortion caused by the following CIC filter. The sampling rate of this signal is doubled and the shape of the signal is also like the previous signal (interpolating the signal is also done by curve fitting). The only difference is in the size or total number of samples.

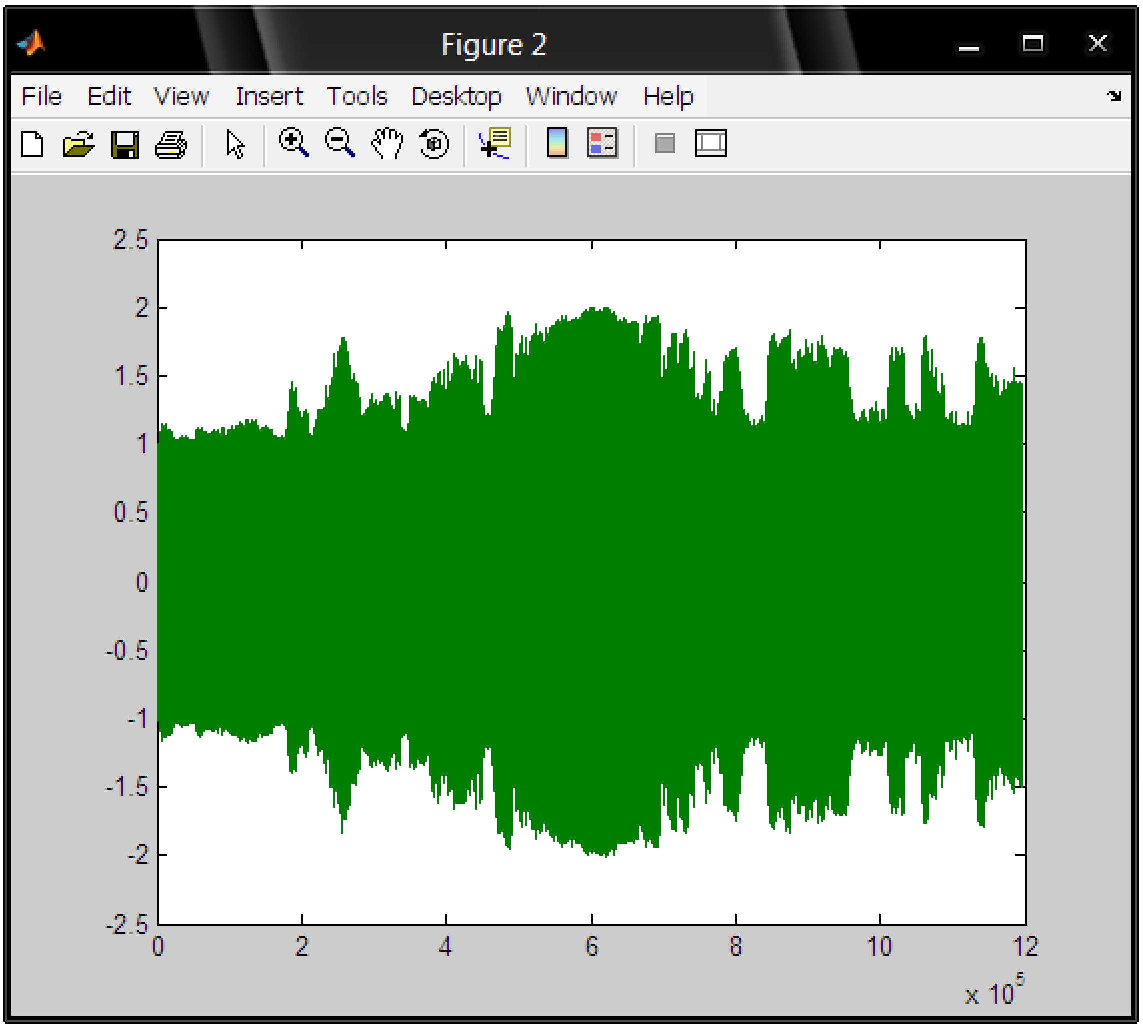


Figure (5-7): *The output of the second FIR filter*

The last stage in the digital up conversion is a CIC filter. The output of this filter is shown in figure (5-8) below. This CIC filter has interpolation factor of 2.

The CIC filter has a huge pass band gain. This gain had been normalized by cascading the CIC with gain that is the inverse of the CIC gain. The output of this filter is the signal which would be entered to the receiver software defined part. It has a sampling rate of 352800 samples per second, which is equal to the sampling rate of the recorded signal multiplied by eight.

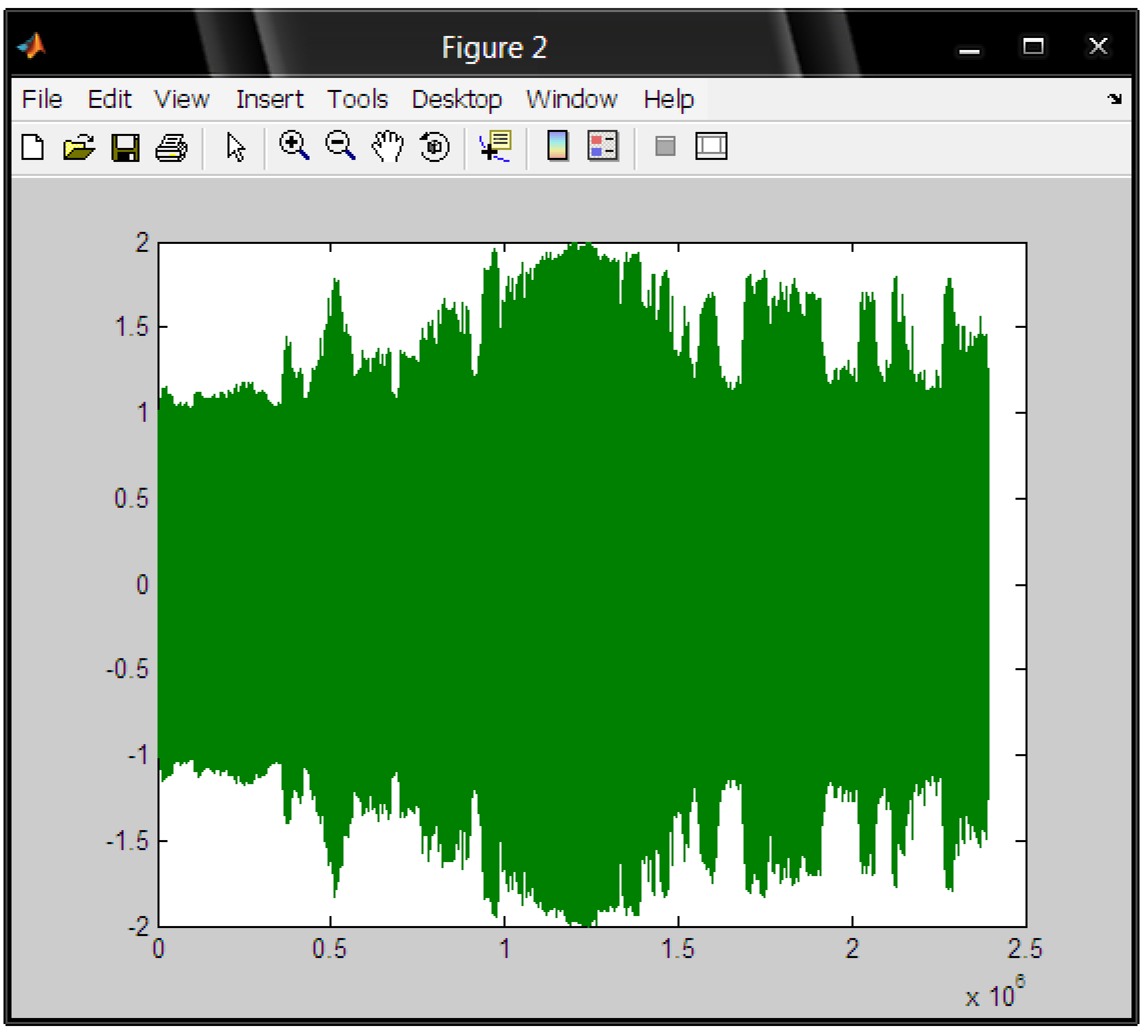


Figure (5-8): *The output of the transmitter CIC filter*

The following operation was the digital down conversion in the receiver part. The digital down conversion had also three stages. These three stages were three decimating filters with the reverse order of the DUC filters stages. So that, the first filter in the DDC is a CIC filter and the output of this filter is shown in figure (5-9). The decimating factor of this filter is 2. We can observe that the output of this filter is the same as the input signal of the CIC in the transmitter part. The CIC filter incur attenuation on the pass band region due its sinc like response. This filter must be followed by compensation filter.

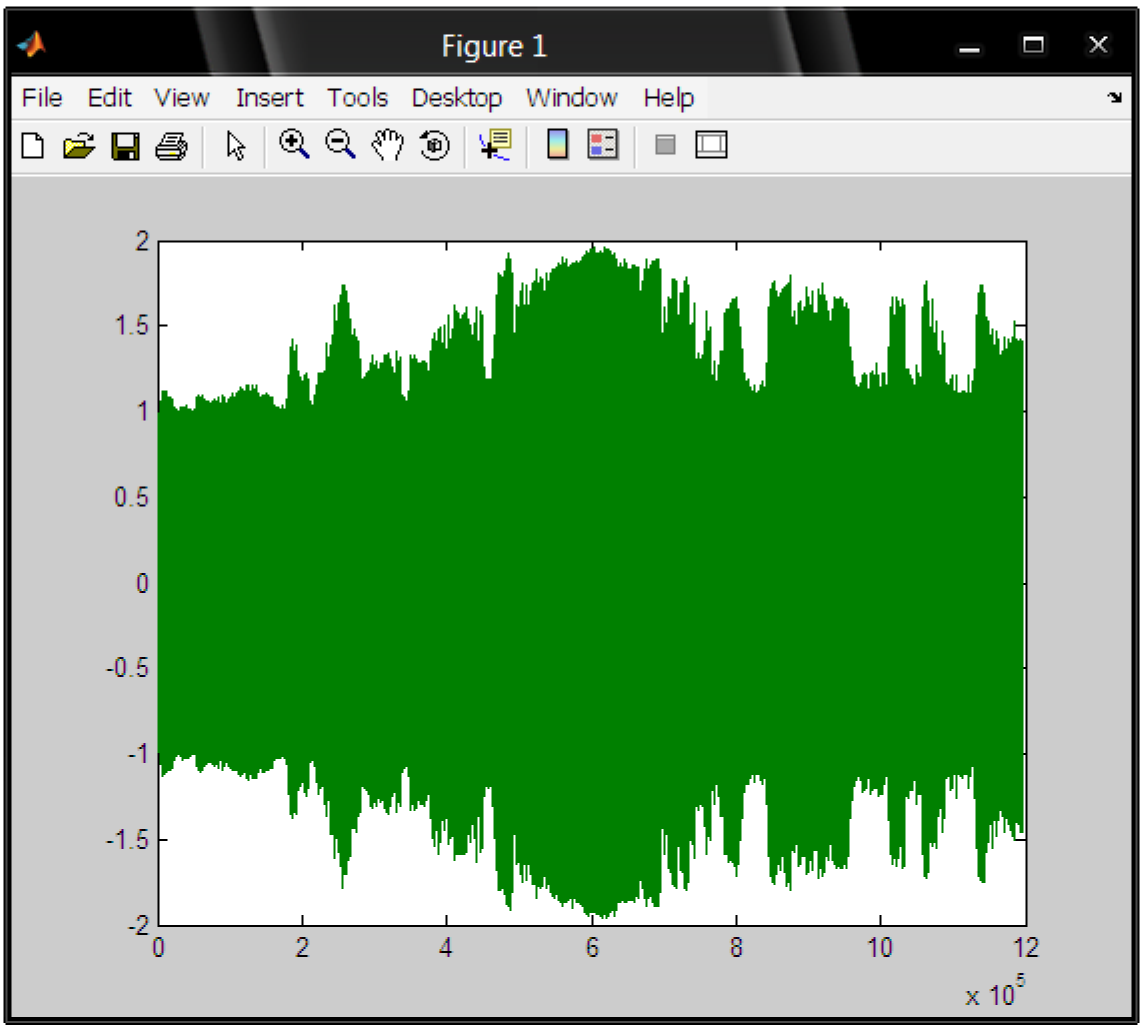


Figure (5-9): ***The output of the receiver CIC filter***

The compensation filter must have an inverse sinc response to lift the droops caused by the CIC filter. This filter was implemented using FIR filter. The output of this filter is shown in figure (5-10) below. This filter has a decimation factor of two. The resulted signal is the same as the previous one, except the number of samples is divided by two.

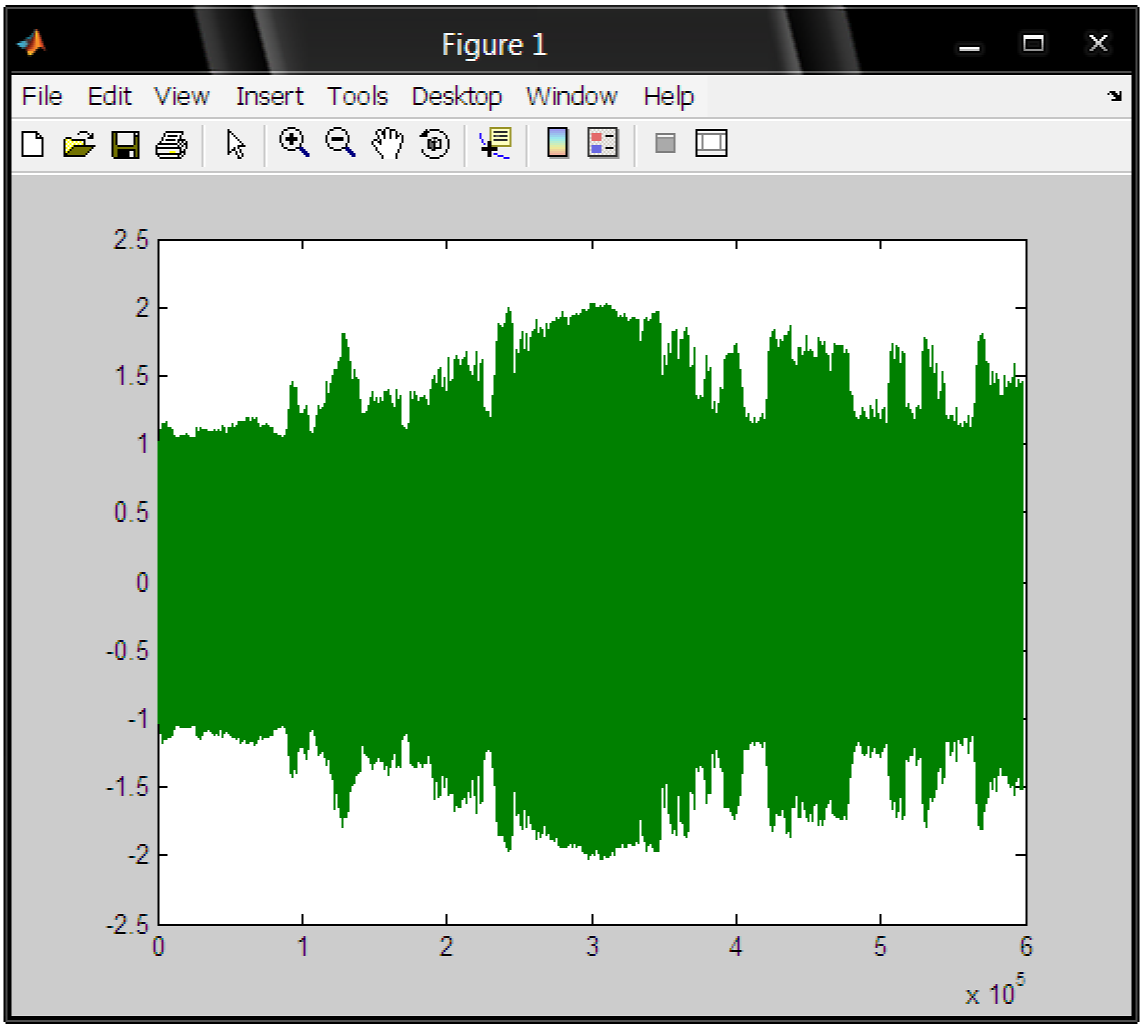


Figure (5-10): *The output of the first FIR*

The last filtering stage was implemented using FIR filter. This filter has a decimating factor of 2. The output of this filtering stage is shown in figure (5-11). The resulted signal is same in shape and number of samples as the modulated signal shown in figure (5-5). This signal would be demodulated in the next stage. We observe that the decimating filters exactly did the inverse process done by the interpolating filters.

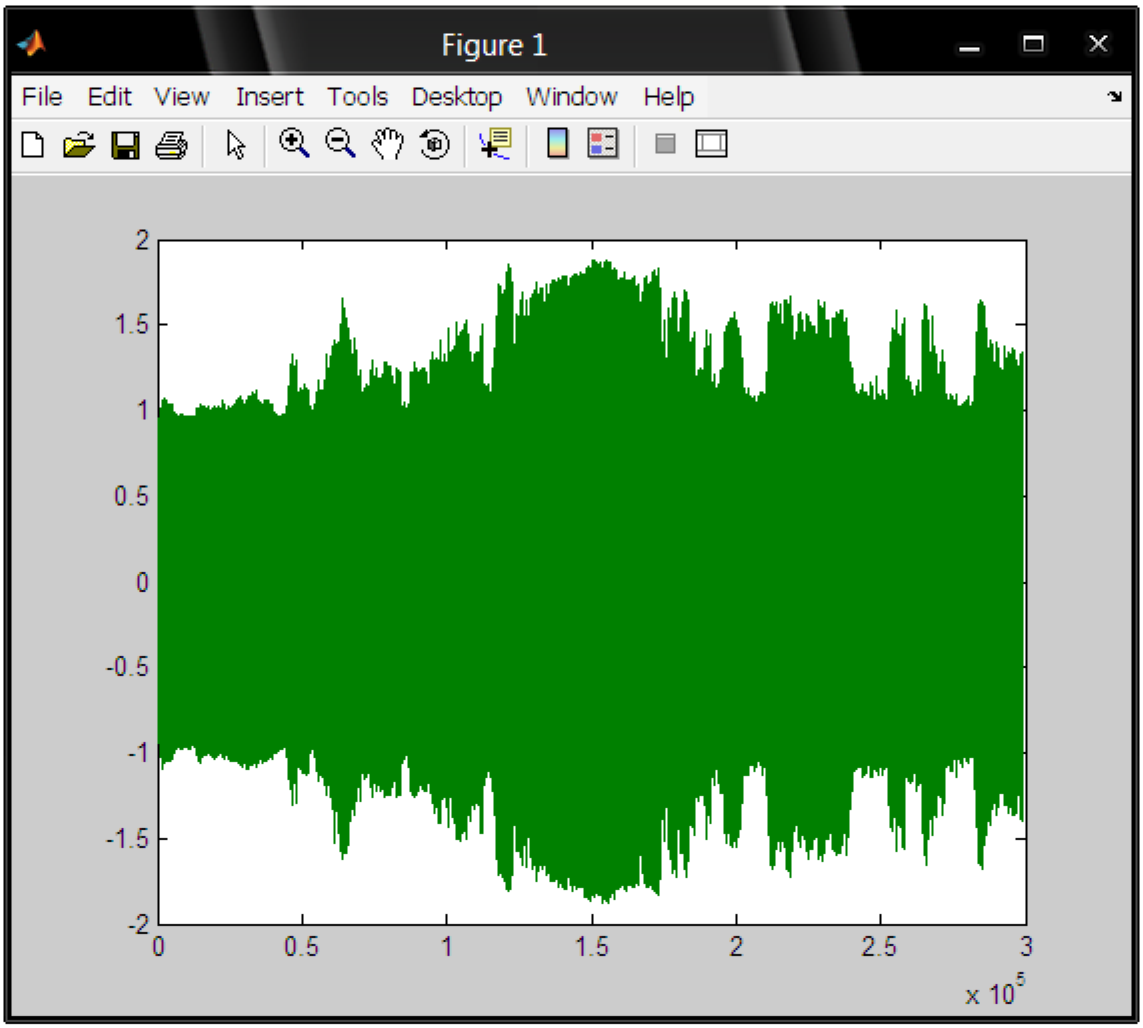
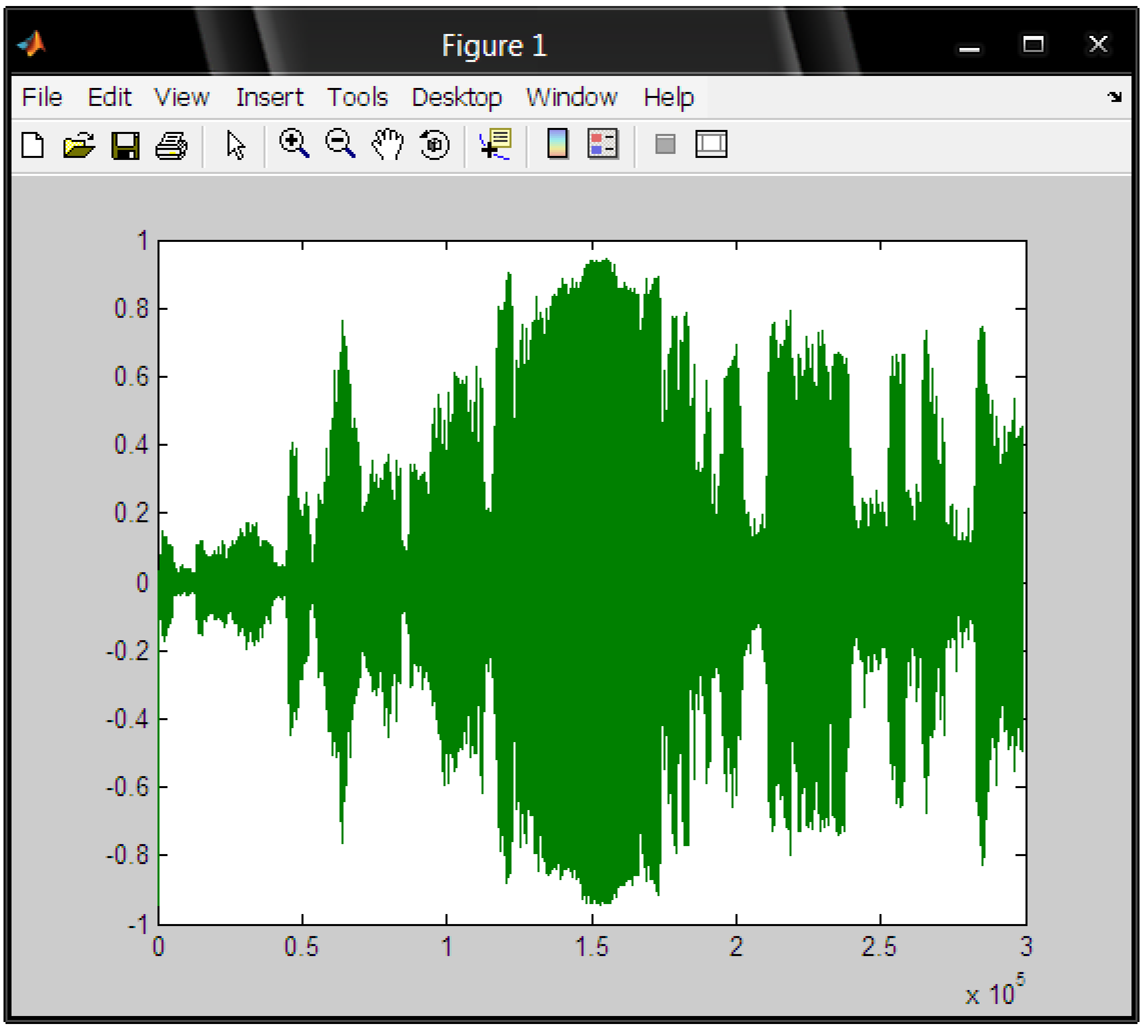


Figure (5-11): *The output of the second FIR filter*

The output of the last decimating filter was demodulated using a simulated envelope detector. The output plot is shown in figure (5-12). This demodulated signal was then sent to PC sound card. The audio signal was heard *clearly* to be same as the recorded signal. The resultant error was 11.1% . This error caused by the used filters. Also audio signals have a redundancy which can increase the error. Although that the audio signal recovered from the processed signal was heard clearly.



Figure(5-12): *The demodulated signal*

Chapter6: Comments and recommendations

6-1: Comments

The software defined radio has benefits in two sides. For the designers it gives the benefits of reprogrammability, ease of manufacture and ease of design. For the user the main benefit is that the user can for example can get a new service by loading the software of this service onto the same equipment. Also, it reduces costs by providing a generic hardware platform and common product line, which can then be customized for any unique application.

The implementation of the design gave satisfying results. We observe that the final demodulated signal was not distorted. The resultant error was quantization error. This another benefit for the SDR that we will get rid of the signal distortion when using analog devices.

6-1: Recommendations

- The audio signal saved by the matlab take large space into the PC RAMs. So, for this project to be applied it will be better to use a computer with a large RAMs.
- The matlab built in functions are explained in the matlab help, so the matlab has the benefit that it can be used to implement and learn more about these functions.
- The DSPs have higher signal processing speeds, and it will be better to implement the SDR using DSPs

References:

- Circuit design for RF transceiver. Edited by Markus Helfenstein And George S. Moschytz.
- Circuits and systems for wireless communication. By DOMINE LEENEARTS, JOHAN VAN DER TANG and CICRO S.VOUCHER
- Modern receivers front ends. By: JOY LASKAR, BABAK MATINPOUR, and SUDIPTO CHAKRABORTY
- Design and development of compact and monolithic direct conversion receivers. By BABAK MATINPOUR.
- The websites:

  www.mathworks.com

  www.RFdesign.com

  www.wikipedia.com
- The matlab help.

Appendix

Appendix: Simulation Matlab Code:

```
[x, Fs, nbits] = wavread('spp.wav');% loading the sampled signal in x,sampling rate & number of bits
siz = wavread('spp.wav','size')%  returns the size of the audio data & no. of channel the vector siz = [samples channels].
sound(x,44100);% sends the signal in vector x (with sample frequency Fs) to the speaker on PC
plot(x);

%%The AM Modulation%%
Fs=44100;% Sampling frequency of the modulated signal
Fc=12000;% Carrier frequency
t = (0:1/Fs:((size(x, 1)-1)/Fs))';
t = t(:, ones(1, size(x, 2)));
y = (x + 1) .* cos(2 * pi * Fc * t);% The modulated signal
plot(y);
sound(y,44100);
```

```
%%Digital Up Convertor%%
pfir = [0.0007    0.0021   -0.0002   -0.0025   -0.0027    0.0013    0.0049
0.0032 ...
      -0.0034   -0.0074   -0.0031    0.0060    0.0099    0.0029   -0.0089   -
0.0129 ...
      -0.0032    0.0124    0.0177    0.0040   -0.0182   -0.0255   -0.0047
0.0287 ...
       0.0390    0.0049   -0.0509   -0.0699   -0.0046    0.1349    0.2776
0.3378 ...
       0.2776    0.1349   -0.0046   -0.0699   -0.0509    0.0049    0.0390
0.0287 ...
      -0.0047   -0.0255   -0.0182    0.0040    0.0177    0.0124   -0.0032   -
0.0129 ...
      -0.0089    0.0029    0.0099    0.0060   -0.0031   -0.0074   -0.0034
0.0032 ...
       0.0049    0.0013   -0.0027   -0.0025   -0.0002    0.0021    0.0007 ];

hpfir = mfilt.firinterp(2, pfir); % pulse shaping FIR filter that increases the
sampling rate by 2
```

Appendix

```
set(hpfir, ...
   'arithmetic', 'double');
f1=filter(hpfir,y);% the output signal frequency interpolated by 2
plot(f1);

cfir = [-0.0007   -0.0009    0.0039    0.0120    0.0063   -0.0267   -0.0592   -
0.0237 ...
        0.1147    0.2895    0.3701    0.2895    0.1147   -0.0237   -0.0592   -
0.0267 ...
        0.0063    0.0120    0.0039   -0.0009   -0.0007];

hcfir = mfilt.firinterp(2, cfir);
set(hcfir, ...
   'arithmetic', 'double');
f2=filter(hcfir,f1);% compensation FIR filter that increases the sampling rate
by 2
plot(f2);
```

```
R    = 2; % Decimation factor
D    = 1;  % Differential delay
Nsecs= 2;  % Number of sections
IWL  = 16; % Input word length
OWL  = 20; % Output word length
hcic = mfilt.cicinterp(R,D,Nsecs,IWL,OWL); %The CIC Interpolation Filter increases the sampling rate by 2.
hgain = dfilt.scalar(1/gain(hcic));
hcicnorm = cascade(hgain,hcic); % Normalizing the CIC filter response to have 0 dB
f3=filter(hcicnorm,f2);
plot(f3);

%%Digital Down Convertor%%
R    = 2; % Decimation factor
D    = 1;  % Differential delay
Nsecs= 2;  % Number of sections
IWL  = 20; % Input word length
OWL  = 20; % Output word length
hcic = mfilt.cicdecim(R,D,Nsecs,IWL,OWL);
hgain = dfilt.scalar(1/gain(hcic)); % Define gain
```

Appendix

```
hcicnorm = cascade(hgain,hcic);
b1=filter(hcicnorm,f3);
plot(b1);


% Filter specifications
Fs   = 1.2e6;      % Sampling frequency 69.333 MHz/64
N    = 20;          % 21 taps
Npow = 5;            % Sinc power
w    = 0.5;         % Sinc frequency factor
Apass  = 5.7565e-4;   % 0.01 dB
Astop  = 0.01;       % 40 dB
Aslope = 60;         % 60 dB slope over half the Nyquist range
Fpass  = 88618.11/(Fs/2); % 80 kHz passband-edge frequency

% Design of filter.
cfir   = firceqrip(N,Fpass,[Apass,Astop],'passedge','slope',Aslope,...
               'invsinc',[w,Npow]);
```

```
hcfir = mfilt.firdecim(2,cfir);
set(hcfir,...
   'Arithmetic',      'double');
b2=filter(hcfir,b1);
plot(b2);

% Filter specifications
N = 62;      % 63 taps
Fs= 600000;  % 541.666 kHz
F = [0 88615.49 110769.367 Fs/2]/(Fs/2);
A = [1 1 0 0];
W = [2 1];   % Weight the passband more than the stopband
pfir = firgr(N,F,A,W);
hpfir = mfilt.firdecim(2,pfir);
set(hpfir,...
   'Arithmetic',      'double');
b3=filter(hcfir,b2);
plot(b3);
```

Appendix

```
%%AM DEMODULATION%%
sAMplus = hilbert(b3); % sAMplus is the original signal added to imaginary
part which is the original signal shifted by 90°
sAMtilde = sAMplus .* exp(-j*2*pi*12000); % Complex envelope
mAMdemod = abs(sAMtilde);               % Envelope detector output
mean(mAMdemod);
mAMdemod = mAMdemod - 0.9449;    % Remove DC from output
plot(mAMdemod); % plotting the demodulated signal
sound(mAMdemod,44100);


%%ERORR%%
erorr = sum((x-mAMdemod).^2)/length(x)
```