DESIGN OF WIRELESS COMMUNICATION SYSTEM ON SDR USING M-QAM MODULATION

Submitted in partial fulfilment of
The requirements for the award of the degree of

Master of Technology
In
Microwave and Optical Communication Engineering

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2K16/MOC/08
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July 2018

DELHI TECHNOLOGICAL UNIVERSITY

Established by Govt. of Delhi vide Act 6 of 2009

(*Formerly Delhi College of Engineering)*

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CERTIFICATE

This is to certify that the work which is being presented in the dissertation entitled "**Design of wireless communication system on SDR using M-QAM modulation**" is the authentic work of **Mohit Jha** under my guidance and supervision in the partial fulfilment of requirement towards the degree of Master of Technology in Microwave and Optical Communication Engineering jointly run by Department of Applied Physics and Department of Electronics and Communication in Delhi Technological University during the 2016-18.

As per the candidate declaration this work has not been submitted elsewhere for the award of any other degree.

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Declaration

I hereby declare that all the information in this document has been obtained and presented in accordance with academic rules and ethical conduct. This report is my own, unaided work. I have fully cited and referenced all material and results that are not original to this work. It is being submitted for the degree of Master of Technology in Engineering at the Delhi Technological University. It has not been submitted before for any degree or examination in any other university.

Abstract

Basic idea of this project is to design a communication channel using software defined radio.The main purpose of Software Defined Radio was to design a device which should be competent of working with many radios working at different parameters. Software Defined Radio provide an rational and economical solution for building multi-mode, multi-range and multifunctional wireless devices that can be revamped by means of software furtherance. In a PC significant signal processing can be done via SDR. In SDR, signal worked in digital sphere rather in analog sphere as in the orthodox radio. The analog signal can be changed to the digital domain with the aid of Analog to Digital converter.

In the digital QAM, a fixed number of two phases and minimum two amplitudes are used. PSK modulation scheme are often made with the QAM theory, but are not deemed as QAM as the amplitude of the modulated carrier signal is same. We are using the QAM because of the issue that a amplitude modulated signal, i.e. double sideband with a suppressed carrier occupies two times the bandwidth of the modulating signal. Assumingly large spectral efficiencies can be get with QAM by adjusting a acceptable constellation range, fixed only by the noise stages and linearity of the communications channel. **Quadrature amplitude modulation** (**QAM**) is combination of analog and a digital modulation system. It fetchs two analog message signals, or two digital bit streams, by amending (modulating) the amplitudes of two carrier waves, using the amplitude-shift keying (ASK) digital modulation scheme or amplitude modulation (AM) analog modulation system. The carrier signals of the like frequencies, mostly sinusoids, are out of phase with one other by 90° and are called quadrature carriers or quadrature parts — so the name of the system. The modulated waves are added, and the end waveform is a merger of both phase-shift keying (PSK) and amplitude-shift keying (ASK), or, in the analog system, of phase modulation (PM) and amplitude modulation.

Acknowledgements

I would like to express my sincere gratitude to my project supervisor, Asst. Professor Dr. Priyanka Jain, for her supervision, invaluable guidance, motivation and support throughout the extent of the project. I have benefitted immensely from her wealth of knowledge.

My gratitude is extended to my colleagues and friends who have not been mentioned here personally in making this project a success.

Last but not least, I want to thank my parents, for inculcating good ethos, as a result of which I am able to do my post-graduation from such an esteemed institution. I would thank my friends for believing in my abilities and for always showering their invaluable support and constant encouragement.

Mohit Jha

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Introduction

1.1 Introduction

Quadrature amplitude modulation, QAM when used for digital broadcast for wireless communications applications is proficient to deliver advanced data rates than standard amplitude modulated arrangements and phase modified schemes. For instance , with phase shift keying, the extent of points at which the signal can break, i.e. the amount of points on the constellation is showed in the modulation set-up explanation, e.g. 16 QAM customs a 16 point constellation.When by means of this scheme, the bit points are typically organized in a square lattice with identical vertical and horizontal positioning and as a outcome the most wide methods of QAM use a constellation with the quantity of points equivalent to a power of 2 i.e. 4, 16, 64 . . . .By expending upper order modulation formats, i.e. added points on the constellation, it is probable to communicate additional bits per symbol. Nevertheless the points are nearer organized and they are consequently further liable to noise and data errors. Typically a QAM constellation is four-sided and thus the utmost conjoint arrangements of QAM 16QAM, 64QAM and 256QAM. The gain of moving to the higher order formats is that there are supplementary points within the constellation and hence it is possible to convey more bits per symbol. The shortcoming is that the constellation points are nearer organized and so the link is more inclined to noise. As a outcome, advanced order varieties of QAM are only used when there is a appropriately high signal to noise ratio.To deliver an example of in what way QAM functions, the constellation diagram beneath shows the values related with the diverse situations for a 16QAM signal. From this it can be understood that a unbroken bit stream may be assembled into fours and embodied as a arrangement.

1.2 Project goals

The purpose of this project revolves around the need of high end communication system. The data limit with the limit on data transfer rate has constantly kept the communication evolving. In this, SDR is used with Laboratory Virtual Instrument Engineering Work-bench (LabVIEW) as software module for programming and Universal Software Radio Peripheral (USRP) as hardware implementation platform, both of them are from National Instrument (NI). The implementation of M-QAM modulation scheme is done in this project. The study of constellation diagram along with modulation error ratio, Error Vector Modulation (EVM), Magnitude Error and phase error. The design of communication channel i.e the transmitter and receiver that will be compatible with M-QAM modulation scheme is done. Major goal includes the study of these parameters and then compare it for future purpose with the other digital modulation techniques.

1.2.1 Implementation

To implement the system on the software defined radio, we needed an interface between the hardware and the computer. The system model was designed in LabVIEW with the help of Software defined radio and the USRP driver software was installed for the interfacing with the hardware After the simulation on Labview, I inserted the blocks of USRP in the Labview block diagram, so that they could be used to transmit the signals from transmitter to the receiver through a physical channel.

1.2.2 Parameters Analysis

To interpret the selves of M array quadrature amplitude modulation, some parameters requisite to be accurately understood such as modulation error ratio (MER), magnitude error, phase error, error vector modulation and transmission range constraints. Affiliations among each of these and its resultant execution has to be seen and proposals on the perfections can be planned.

Chapter 2

Software Defined Radio Platform and Basics

This chapter discusses the project method and theory on the subject of softwar defined radio and its principle.

2.1 Evolution Of Radio Systems

Pre-industrial age, wireless broadcast/reception was complete done line-of-sight distances (later prolonged using contracts) using smoke signals, torch gesturing, sporadic glasses, sparkles and semaphore garlands. Complex messages were generally relayed using an rich set of the above rudimentary signals. The growth of what we presently ponder wireless communications shoots from the mechanism of Oersted, Faraday, Gauss, Maxwell, and Hertz. In 1820, Oersted discovered that a current transport probe yields an electromagnetic field. Michael Faraday discovered electromagnetic induction in 1831 when he showed that a time varying magnetic field produces an electric current across the conductor. In 1864, James Maxwell framed the conventional theory of electromagnetic radiation, fetching collected for the first time electricity, magnetism and light as indexes of the identical occurrence. Maxwell was capable to display in notional and calculated arrangement that electromagnetic waves could spread over free space. Between 1886 and 1889 Heinrich Hertz steered a sequences of tests that verified convincingly Maxwell's theory of electromagnetic radiation. This finding ultimately led to the expansion of commercial Hertzian wave built wireless telegraphy, radio and television. Italian engineer Guglielmo Marconi is frequently attributed as the discoverer of radio; having paramount transferred and received a veiled dispatch at a space of 1.75 miles near his household in Italy. The Indian physicist Jagadish Chandra Bose also individually created and sensed wireless signals of 6mm wavelength[30]. These two triumphs ushered in the era of fresh wireless communications. Shortly commercial radio and television broadcasting locations were setup in the US and the repose of the world. However, it was the WWII years that adage swift and forced increases in nearly all zones of engineering and technology comprising transportations.

The twentieth century saw other significant advancements in the field of communication but none greater than the works of Claude Shannon. In a groundbreaking weekly printed at Bell Test center in 1948, Shannon demarcated in scientific expressions what data is, in what way it can be communicated over a strident network and what are the capacity limits of a channel. Today all modes communication rely on Shannon's contribution and it is for this reason he is called "the father of information theory".

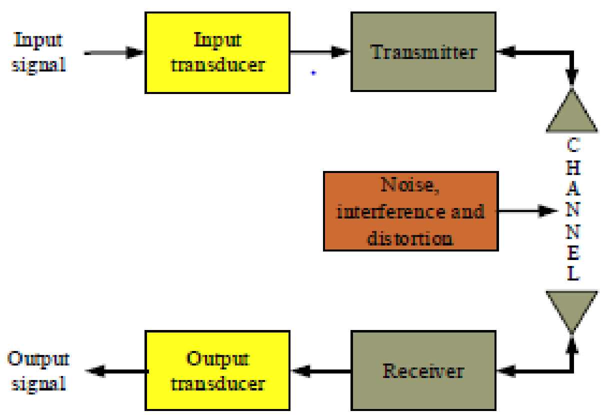


Figure 2.1 Functional layout of a communication

The basic functionality of any communication system is to convey information from a source to one or more destinations. At its heart, the architecture of a communication system consists of three basic parts; transmitter, channel and the receiver as shown in Figure 2.1. The information from an input source, either voice, picture, or plain text, is applied to an input transducer that converts it into electrical signals suitable for transmission. At the transmitter the electrical signals are converted to a form suitable for transmission at the appropriate frequency, by a process called modulation. Modulation normally involves using the message (modulating) signal to alter the characteristics of a sinusoidal carrier signal[30]. There are two different types of communication; analog and digital, depending on the type of the modulating signal Amplitude, frequency, phase modulation (AM, FM, PM) are examples of analog modulation and amplitude, frequency, phase shift keying (ASK, FSK, PSK) are examples of digital modulation.

2.1.1 Digital modulation techniques

In the previous two eras, digital communication systems (DCS) execution constantly added complex digital signal processing actions have developed gradually striking because of the remarkable evolution of the cellular telephone arcade and wireless amenities. These structures bid handling selections and litheness not accessible in solitary channel, narrowband analog broadcasts. The unique representative of a digital communication system is that in a limited break, it guides a waveform from a fixed regular of probable waveforms, in disparity to an analog system that directs one feasibly unremittingly fluctuating waveform from an vast set of waveforms with inestimable tenacity in the existence of noise with alike appearances.

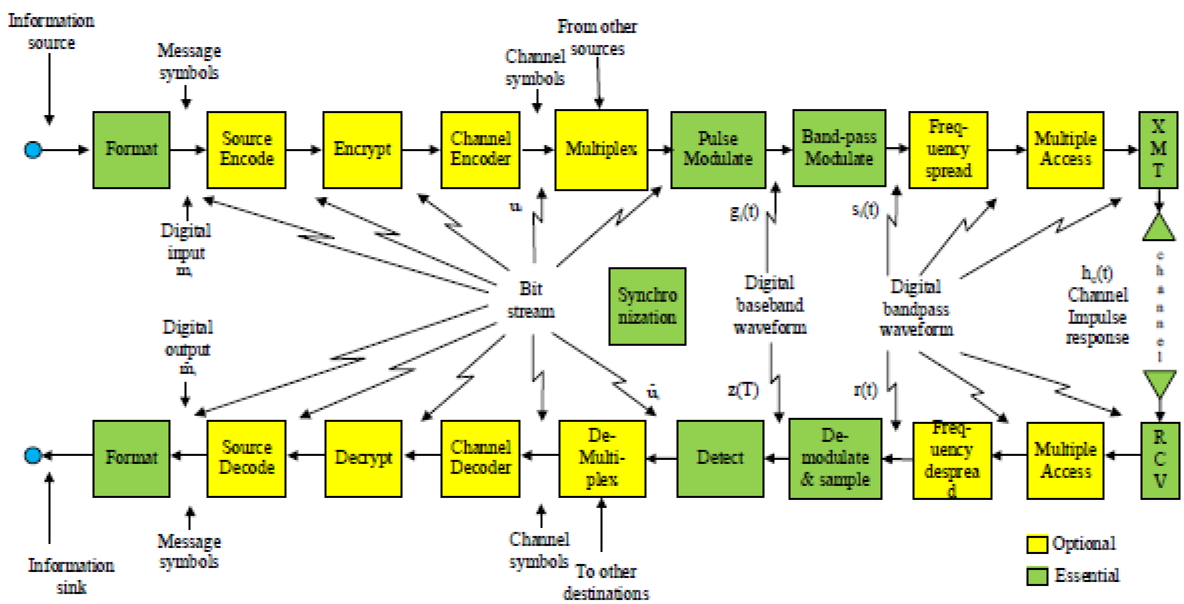


Figure 2.2 Typical digital broadcast system

Fig 2.2 illuminates the plan and signal flow of an advanced digital broadcast system. The details basis can be any an analog signal, such as acoustic or audiovisual signal, or a numeral signal, such as bits from a workstation. This input is then converted to binary digits (bits); the bits are then congregated to form message symbols (m_i, $where$ $i = 1,2,\ldots, M$). If $M = 2$, the message symbol m_i is binary.

Even though definition of M-ary includes binary, it generally refers to those cases where > 2. Although in the simplest form, source encoding is a direct conversion of the message to binary digits, the process may involve additional steps to perform this conversion efficiently. The resolution of the network encoder is to announce certain severance in the binary evidence arrangement so as to permit the mouthpiece to lessen the possessions of clatter and intrusion hosted by the network. The primary purpose of the pulse modulation block (also called digital modulator) is to transform each symbol from a binary representation to a baseband waveform. Suppose veiled data is to be conveyed single symbol at a period, the pulse modulator block may merely plot the dual bit 0 into a waveform $F_0(t)$ and the bit 1 into $F_1(t)$.

2.2 Software defined radio

The word 'Software Radio' was devised in the primary nineties by Joseph Mitola III in a paper on radio constructions at the National Telesystems Meeting, NewYork, in May 1992.
There is no accord on the meaning, possibility and necessities of a software defined radio system but it can be deliberated as one in which all or part of the physical deposit apparatuses are executed on a programmable or reconfigurable platform.

2.2.1 Definition

Starting with a noteworthy quote from GNU Radio originator Eric Blossom, Software-Defined Radio can be demarcated as:

"***Software radio is the practice of receiving code as near to the antenna as conceivable. It fits radio hardware glitches into software glitches*.**"

SDR is a unite of machinery and computer code mechanisms to style the arrangement malleable for wireless broadcast[1]. SDR is to diminish the hardware essential for signal processing exertion and its alteration into software which should be accomplished through a broad – purpose computer. Software radio is the practice of accomplishment encryption as adjacent to the antenna as possible. It turns radio hardware complications into software complications.

2.2.2 Working principle of software defined radio

In standard, a universal applicable hardware functions as boundary amid the baseband and the RF. The waveform of a communicated signal is entirely created from software, as well as a expected signal is wholly administered and demodulated inside software set of rules[2]. The signal should be produced in digital form and operated as much as possible within the PC (the functions performed like modulation, filtering, passing through FFT blocks and even amplification), until the signal is set to be sent. The data in the form of digital samples is then transformed to analog signal by the hardware block. In the last stage, this transformed signal is given to the antenna for transmission.

Together modulation and demodulation is implemented in software and hence the radio is capable to upkeep a expansive array of frequencies and tasks, either parallel or built on distinctive software downloads. The notion of employing radio functions in software rather than hardware is not rather different. Figure 2.3 is shown below :

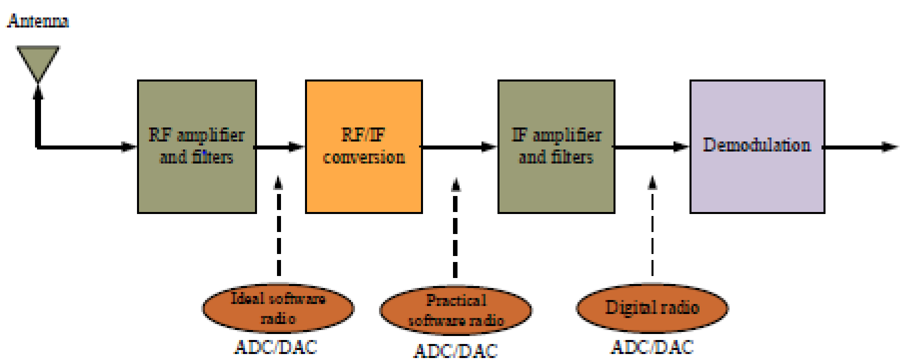


Figure 2.3 Simple software defined radio block diagram

The prohibitive cost of high-end processing solutions also prevented software defined radios from being viewed as commercially viable. However more recently, interest in SDR has been driven by three main factors; soaring cost in the development of chips used for radio systems, diversified market demand for different or even multiple wireless standards, and a faster development cycle due to the rapid evolution of wireless standards.

Technological advancements have also supported interest; including, increased density and lower power consumption of CPUs and DSP devices, high volume demand for WiFi and cellular telephony devices supporting IoT, increased clock rates and bit precision of ADC/DAC, and advancements in hybrid, mixed signal devices. All these factors combine to make SDR solution very attractive.

2.2.3 SDR Architecture

The functional blocks of a software defined radio system are identical to the ones found in any digital communication system[3]. Though, SDR seats different necessities on these modules in order to back numerous frequency bands, numerous services and hastily reconfigurable action required for a backup altered wireless standards. A model for an advanced software radio is shown in the Figure 2.4

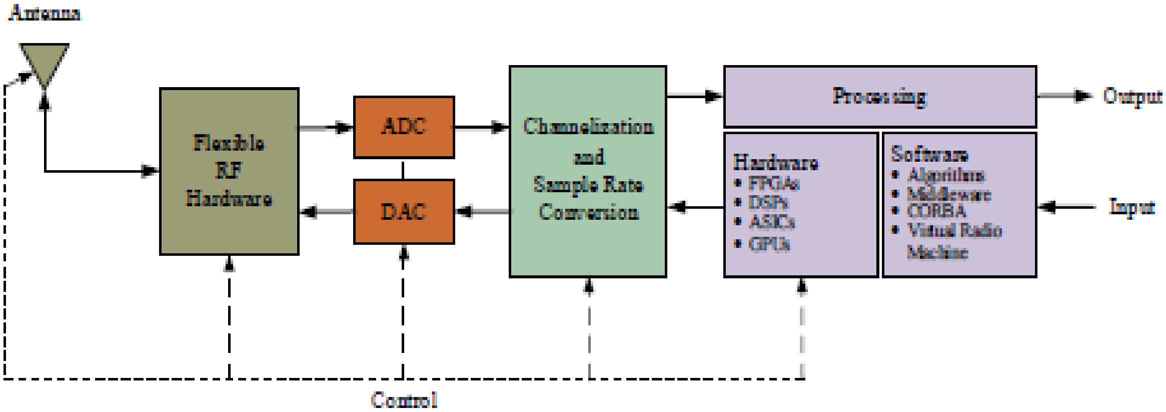


Figure 2.4 Model of a software radio

In current SDR architectures digitization is often performed in the intermediate frequency (IF) domain. RF hardware contains low noise amplifiers (LNA) and local oscillators to mix the desired signal to IF through either direct-conversion or superheterodyne principles[4]. The sort of RF mixer and the total of phases rests on the kind of applications being employed. The multi-frequency band, multi-mode task of an SDR upsets the proposal of the RF hardware and the choice of Analog to digital bits and vice versa. The forward-facing terminal is modifiable for diverse rates and transmission range essential by the distinct criterions that the SDR maintenances. Temporarily, the ADC's and DAC's necessity have appropriate sampling rates to support the maximum preferred frequency bandwidth[7]. Since the sampling requirements varies according to the type of standards supported, software radios often sample at very high rates, greater than the required bandwidth, and use a digital front end to further filter and lower the sampling rate to the desired values. The digital front end consists of digital filters, mixers, decimators and interpolators implemented either on FPGA's, ASIC's, or DSP's.

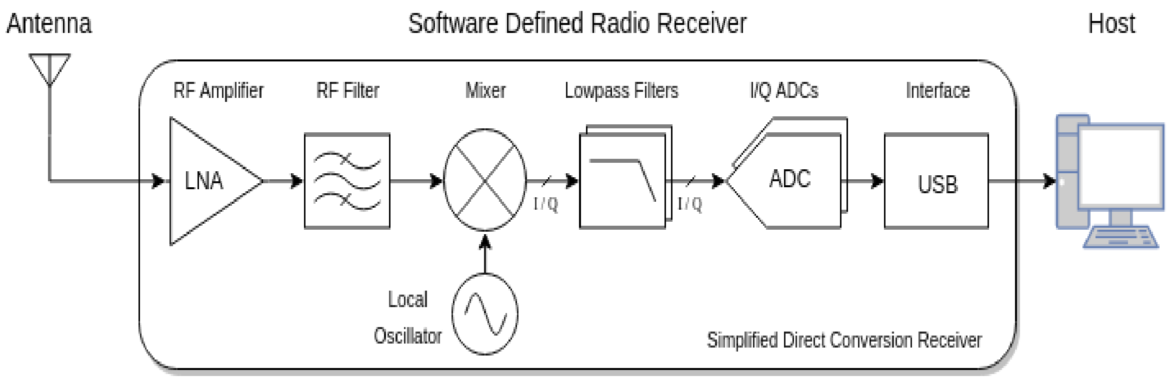


Figure 2.5 SDR receiver

The core digital signal processing hardware can be performed through DSP's, FPGA's, multi-core CPU's, ASIC's and/or GPU's. The choice of core computational platform depends on the required signal processing algorithms and their computational and throughput requirements. In practice, software radios repeatedly serve a grouping of fundamental calculating elements, dividing the signal processing assignment between the several elements. Communication leaning DSP algorithms rather frequently effort top on a certain processor architecture, so many core elements deliver the most supple and competent scheme of fulfilling some standard. Both algorithm processing and control routines may use a variety of software methodologies, such as high level programming languages (MATLAB, C/C++, CUDA C), middleware, or dedicated software applications like GNURadio, LabVIEW or OpenBTS.

2.2.4 SDR Signal Processing Stages

As termed, the SDR has precise structures and traits obligatory for operation. These include the radio frequency front end, the ADC and DAC, the realtime digital handling by keen or configurable hardware mechanisms, software programmable digital signal treating, and general persistence software processing.

Broadcast Frequency Front panel

An idyllic software radio is solitary that would have marginal analog RF front panel, residing of an ADC at or close the antenna. However, any practical SDR implementation still needs an RF front end whose design challenges are only exacerbated in software radio. The key roles of radio frequency front end are up/down alteration, nosiness refusal, straining, pre-amplification and to diminish bias while attaining prerequisite vigorous series. On the source flank, the RF front end receipts the digital image of the analog signal from the DAC, increases the signal to the anticipated RF center frequency, magnifies the strength to the appropriate extent, avoids transmitting harmonics or interference by limiting the signal bandwidth with filtering and inject the signal to the probe. The operation of the receiver is more complex than the transmitter as the desired signal must be separated from the noisy background RF environment. At the receiver, the signal from the antenna is fed through an RF filter, amplified by a low noise amplifier (LNA), down-converted to a frequency that is compatible with the ADC, and then converted into digital samples at the ADC. These digital samples may undergo further filtering, translation and processing in the digital front end stages. Some of the most shared RF front end constructions is the renowned superheterodyne receiver. It was originally aimed by Edwin Armstrong in 1918 as a way to overawed scarcities in initial vacuum tube triodes used as amplifiers in radio track discovery apparatus. In this scheme, the expected signal is changed or moved in frequency by a mixer to a static transitional frequency (IF) is lesser to the focus frequency of the preferred RF signal but advanced than the data transmission width of the wanted signal.

The list below shows some of the parameters that influence the design and selection of the RF front end:

- ***Sensitivity*** is the weakest signal level that the receiver can detect
- ***Selectivity*** denotes the capability of the receiver to perceive the anticipated signal while rebuffing all others.
- ***Dynamic Range*** is the control variance between the feeblest and strongest signal that the receiver can sense.
- ***Stability*** is the lack of change in the gain and frequency response of the receiver with changes in temperature, time, etc.

In Superheterodyne receiver design, the received signal is converted or shifted in frequency by a mixer to a stationary intermediary frequency (IF) which is subordinate to center frequency of the preferred source but superior than the bandwidth of the looked-for signal. The conversion is usually done in two stages to take advantage of the lower filtering requirements and availability of narrowband RF and IF components. The architecture of a superheterodyne receiver employing I/Q modulation at the second stage is shown in Figure 2.6. The received signal is first filter by the pre-selection band pass filter, then amplified by the low-noise amplifier and filtered by another bandpass filter to remove the image frequency effects before being mixed at the first stage. After this, the signal is again filtered, I/Q demodulated to baseband and then fed to the input of the anti-aliasing filter and then the ADC. The superheterodyne receiver has good selectivity and sensitivity but the bandwidth and the center frequency of the filters are narrow, fixed and not flexible.

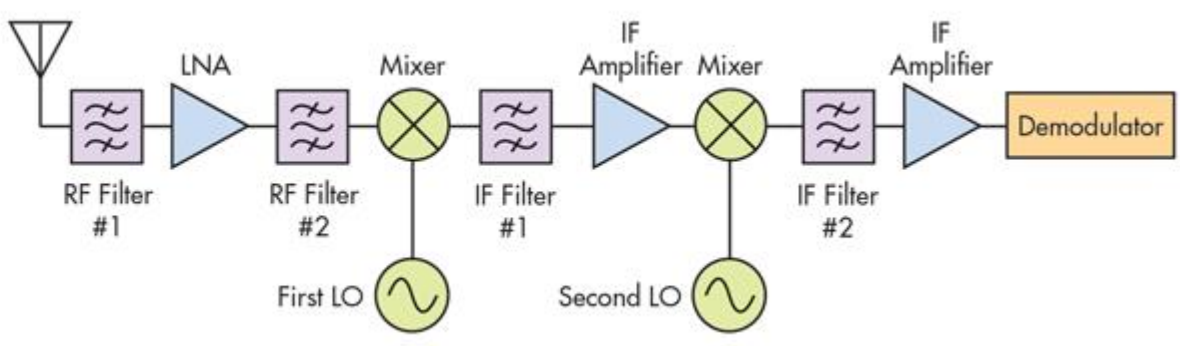


Figure 2.6 Superheterodyne receiver

This makes the superheterodyne architecture unsuitable for wideband or simultaneous narrowband signal SDR applications. Multiple front ends or adjustable filter components mitigation these problems but add to complexity, cost and weight. Another option is to process the IF signal digitally.

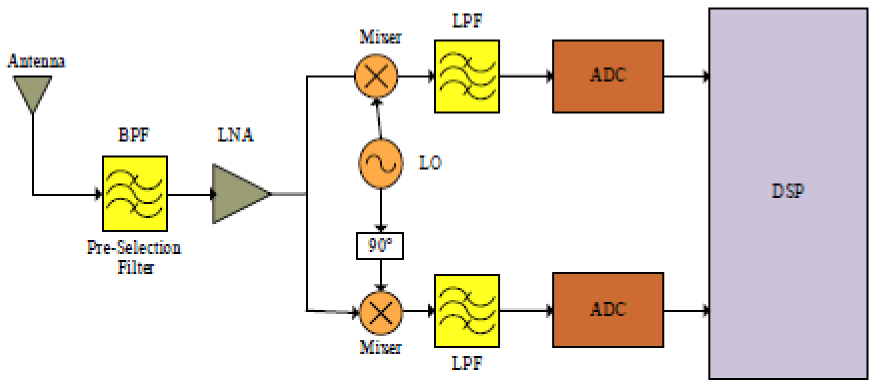


Figure 2.7 Direct conversion or zero-IF receiver

In a straight translation receiver design also identified as the homodyne receiver or the zero-IF receiver, the expected RF signal is openly down altered to baseband by a single mixing step. The received signal is first bandpass filtered, amplified by the LNA and then mixed using an I/Q demodulator. I/Q demodulators are required when detecting phase or frequency modulated signals. As before the down-converted signal is filtered by the anti-aliasing filter before being fed to the ADC. Figure 2.7 illustrates direct conversion receivers are conceptually attractive due to their simplicity and ability to switch between specific bands and modes relatively easily. However there are some air boundary parameters which are unsuitable for direct conversion receivers. Moreover due to the local oscillator is at the desired signal frequencies, this architecture requires an extremely stable local oscillator and mixers with no local oscillator feed through to avoid unauthorized emissions and interferences. Several problems can however be rewarded with digital signal processing which has prompted some sources to advise that the direct conversion architecture is the utmost favorable visible panel architecture for cognitive radios.

As an alternative to a zero-IF design, this architecture may also be used to provide complex signal mixing to an IF frequency where digital signal processing may correct analog I/Q signal offsets and thereby eliminate signal images. It is similar to an IF sampling quadrature superheterodyne technique but can support both zero-IF and complex-IF processing

2.2.5 Challenges of software defined radio implementation

In our application the aim is to implement a communications system based on QAM, which is a kind of wide band application. A number of challenges are prohibiting the use of SDR to a limited number of applications.

- Power consumption of a SDR device

  It has been observed that powerful computers are required to run an application. The power consumption of SDR devices is not same as the power needed bythe hardware radios.

- Larger Size of Hardware Needed

  The size of hardware required for the processing of the signal is much large than the traditional radio s dedicated hardware.

Due to the above stated reasons, the system of having a portable device based on SDR has not developed so much which can be used as many different radios, and the use of SDR has been limited to research or applications in the base station, instead of the mobile terminal, where the power needs are not of much consideration.

Chapter 3

USRP and Data Transfer Framework

3.1 Universal Software Radio Peripheral (USRP)

USRP was first designed and used by Matt Ettus, to which radio front end, DAC and ADC were combined via Universal Serial Bus 2.0 (USB 2.0). USRP had been designed to make the SDR reconfigurable and adjustable according to the given conditions. USRP uses a diverse FPGA, quicker ADCs and DACs with a advanced vigorous range and a Gbit-Ethernet linking. All USRP daughterboards can be used additionally. USRP block diagram is shown in figure.

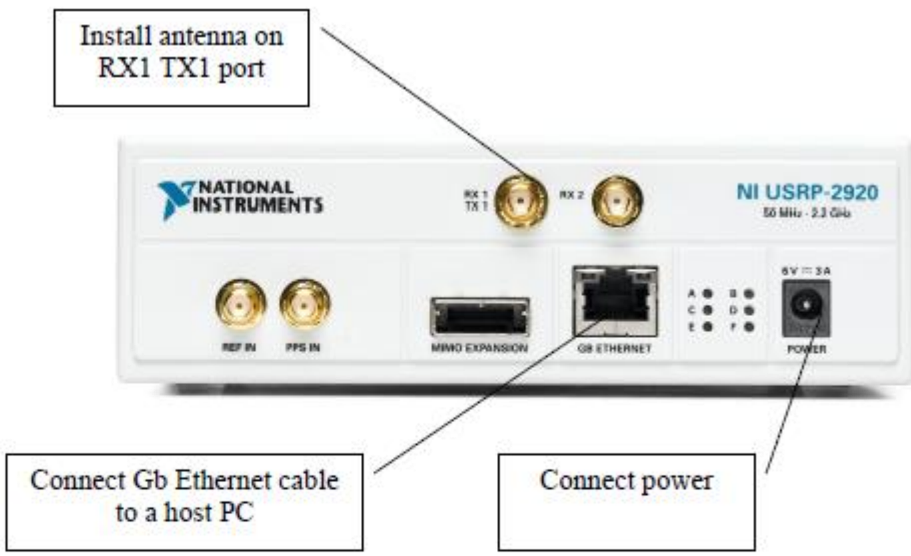


Figure 3.1 USRP Front panel

Inside USRP motherboard, an analog to digital converter (ADC) samples the received signal and adapts it to digital values subject on the ADCs vigorous range of 14 bit. How many times per second the analog signal can be measured is demarcated by the sampling rate of used ADCs – yielding in 100e^6 results per second. The digital sample standards are relocated to the FPGA and administered with digital down converters (DDC) to see precisely the wished output frequency and sample rate. Below, the schematic of a DDC is shown. Digitized samples from ADC are varied down to the chosen IF by being increased with a sine correspondingly cosine function consequential in the I and Q route. The frequency is produced with a numerically-controlled oscillator (NCO) which creates a discrete-time, discrete-amplitude waveform inside the FPGA. Via the used NCO, very rapid frequency hopping is realistic.

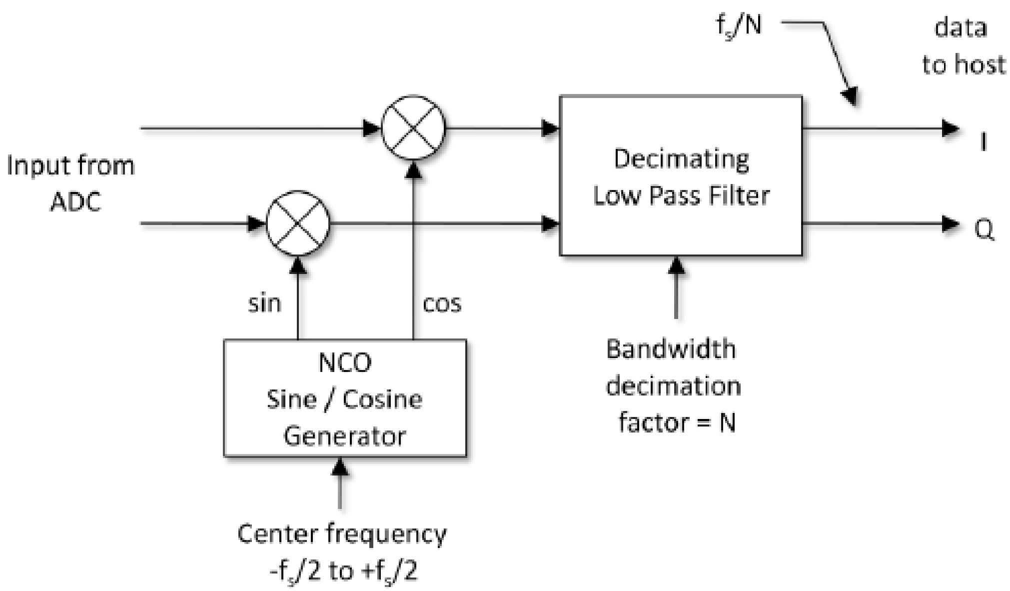


Figure 3.2 Digital Down Converter Block Diagram

After this, the decrease of the sampling rate is done by an subjective decimation factor N. The sampling rate (fs) assigned by N outcomes in the output sample rate, directed to host. In transfer path, the similar process is done vice versa by means of digital up converters (DUC) and digital analog converters (DAC). The FPGA also ropes time reliant on applications which for e.g. use TDMA. A free running inside counter permits received samples to be directed in firmly definable timestamps.

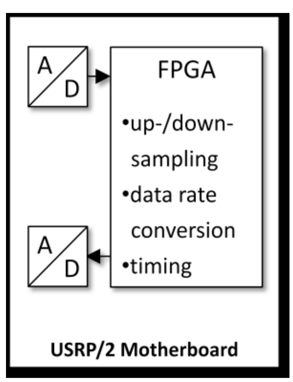


Figure 3.3 USRP block

Near host flank, USRP practices the Gbit-Ethernet linking, letting a noteworthy advanced output than USRP with USB 2.0 attains. The USB linking withstands 32 MB/s in half duplex, so broadcast and reception of samples isn't likely synchronously. Using 4 byte complex samples (16-bit I and 16-bit Q) and valuing the Nyquist criterion hints to a usable (complex) spectral bandwidth of 8 MHz. On behalf of Gbit-Ethernet in USRP, the ideally data rate of 125 MB/s permits for a hypothetical (complex) RF bandwidth of about 30 MHz. A value of 25 MHz working bandwidth, as given in, may aid as a convincing limit.

MODEL	FREQUENCY	GPS	FREQUENCY ACCURACY
NI USRP – 2922	400 MHZ - 4.4GHZ	--	±2500HZ @ 1GHZ
NI USRP – 2930	50 MHZ - 2.2GHZ	YES	±2.5HZ @ 1GHZ
NI USRP – 2932	450 MHZ-4.4GHZ	YES	±2.5HZ @ 1GHZ

Table 1 USRP models and characterstics

3.2 Various units of USRP

We will discuss shortly specific vital parts inside USRP and USRP2 such as ADC,DAC, FPGA and daughter boards .

▪ ADC Section

There are 4 high-speed 12-bit ADC converters. The sampling rate is 64M samples per second. In standard, it could digitize a band as wide as 32MHz. For USRP2 , there are two high-speed 14-bit ADC (of type LTC2284 used at 100 MS/s) . There is two other secondary ADCS(of type AD7922 used at 100 MS/s) for each daughter board connector . Giga Ethernet can withstand 1 gigabit/s . Consequently, this is max 800 MB/s given integer decrease and a 100 MHz DSP clock.

- ## DAC Section

At the transferring path, there are also 4 high-speed 14-bit DA converters. The DAC clock frequency is 128 MS/s, so Nyquist frequency is 64MHz. For USRP , The DAC clock frequency is 400 MS/s, so Nyquist frequency is 200 MHz .USRP 2 has main DAC (Dual of type AD9777 used at 400Ms/s)and two auxiliary DACs (of type AD5623)

- ## FPGA

Inside the FPGA is the command to realize the functionality of every of its structure chunks. Perhaps accepting what goes on the USRP FPGA is the most vital portion for the GNU Radio users. The ADCs and DACs are linked to the FPGA altoghther[23]. This section of FPGA shows a main part in the USRP system. Principally what it does is to achieve high bandwidth math, and to lessen the data rates to rather it can shoot over USB2.0 on USRP respectively .The regular FPGA arrangement comprises digital down converters (DDC) applied with 4 stages cascaded integrator-comb (CIC) filters. Likewise, it contains digital up converters (DUC) realized with 4 stages cascaded integrator-comb (CIC) filters .CIC filters are precise high-performance filters using only improves and delays. The DDC and DUC both contain 2 halfband filters. The high rate one has 7 taps and the low rate one has 31 taps For spectral modelling and out of band signals elimination.

- ## Daughter boards

The mother board has two openings. One of these openings is for TX and the other is for RX . Both daughter board slot has admittance to ADC/DAC . The daughter boards are used to grip the RF receiver edge or tuner and the RF transmitter[16]. Each daughterboard has an I2C EEPROM (24LC024 or 24LC025) onboard which recognizes the board to the system. This permits the host software to repeatedly set up the system suitably created on the installed daughterboard. There are numerous types of daughter boards existing currently such as :

a) Basic TX/RX Daughterboards

Both has two SMA connectors that can be used to link external up/down tuners or signal generators. We can give it as an entry or an departure for the signal devoid of disturbing it. Some form of external RF front end is essential. The ADC inputs and DAC outputs are straight transformer-coupled to SMA connectors (50Ω impedance) with no mixers, filters, or amplifiers. The BasicTX and BasicRX give straight admittance to every signals on the daughterboard edge. All of the Basic TX/RX boards has logic analyzer connecters for the 16 general purpose IOs. These pins can be used to aid debugging FPGA project via on condition that admittance to internal signals.

b) Low Frequency TX/RX Daughterboards

The LFTX and LFRX are precise analogous to the BasicTX and BasicRX, respectively, with 2 key alterations. Because the LFTX and LFRX use differential amplifiers in its place of transformers, their frequency response spreads down to DC. The LFTX and LFRX also consume 30 MHz low pass filters for anti-aliasing.

c) TVRX Daughterboards

This is a receive-only daughter board. It is a thorough VHF and UHF receiver system based on a TV tuner module. The RF frequency ranges from 50MHz to 860MHz, with an IF bandwidth of 6MHz. All tuning and AGC functions can be organized from software. Usual noise figure is 8 dB.

Chapter 4

M-QAM Implementation Using Software Defined Radio

4.1 Digital modulation

Digital modulation is the alteration of the certain parameters in the carrier discrete baseband signal in line with the information signal. The development of the digital communication over the span of time has been shown below:

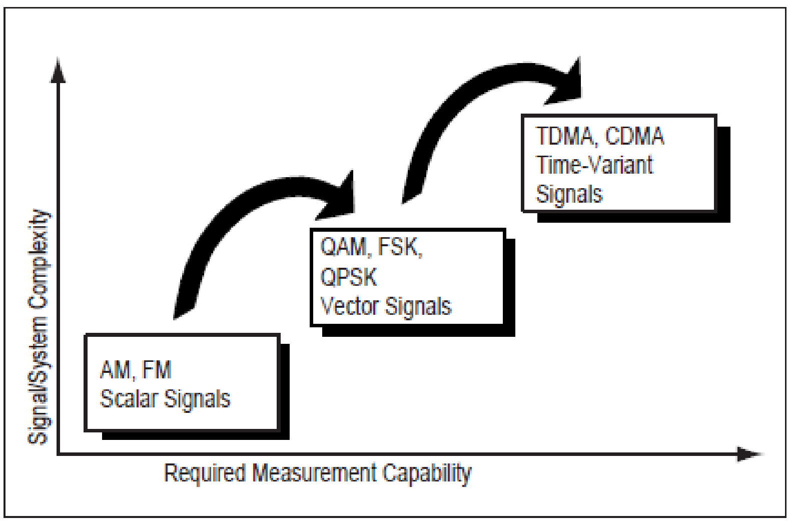


Figure 4.1 Developments in communication

<u>Transmitting information</u>

To spread a signal over the midair, there are three leading stages:

1. A unadulterated carrier is produced at the source.

2. The carrier is modified with the data to be communicated. Any dependable noticeable alteration in signal features can carry material.

3. At the receiver the signal alterations or deviations are sensed and demodulated

Signal individualities that can be improved

There are only three features of a signal that can be reformed over time: amplitude, phase or frequency. But, phase and frequency are unbiased dissimilar methods to opinion or extent the identical signal variation.

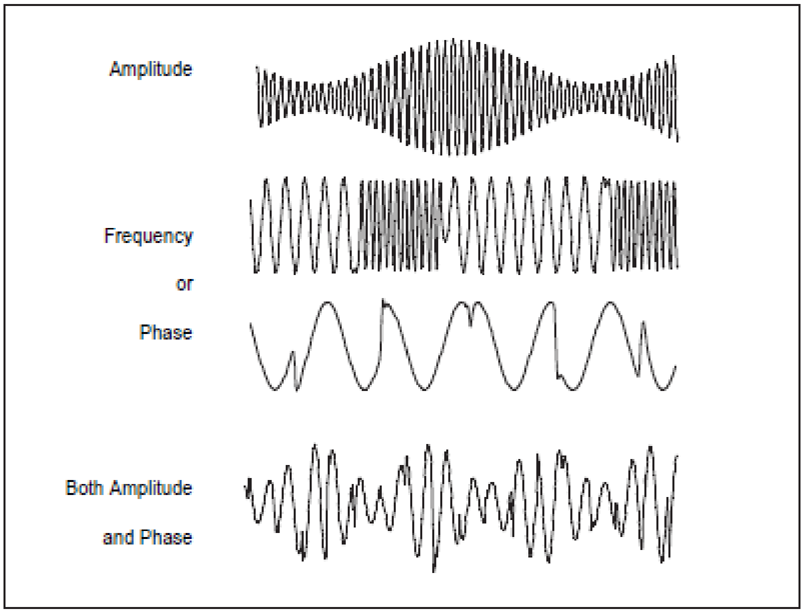


Figure 4.2. Signal Characteristics to Modify

Amplitude and phase can be modulated instantaneously and disjointedly, but this is challenging to produce, and particularly demanding to spot. In its place, in real-world systems the signal is disconnected into alternative series of independent constituents: I (In-phase) and Q (Quadrature). These modules are orthogonal and do not affect with each other

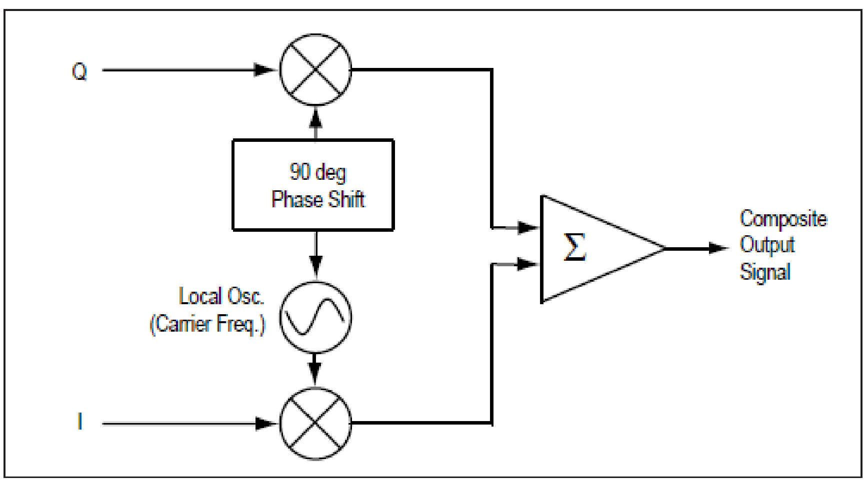


Figure 4.3 I and Q in a Practical Radio Transmitter

1. ASK(Amplitude shift Key)

When the carrier amplitude is wide-ranging in proportion to info signal m(t).

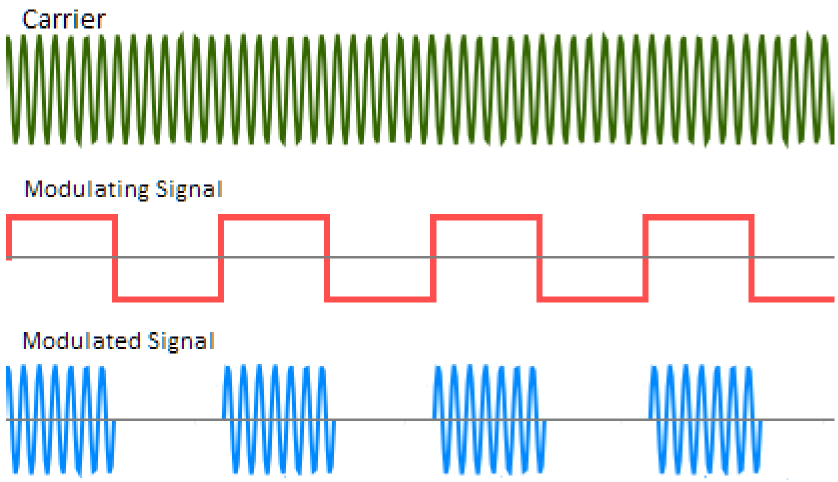


Figure 4.4 Amplitude shift Key

2. FSK (Frequency shift key)

When Data is communicated by changing frequency of the carrier.

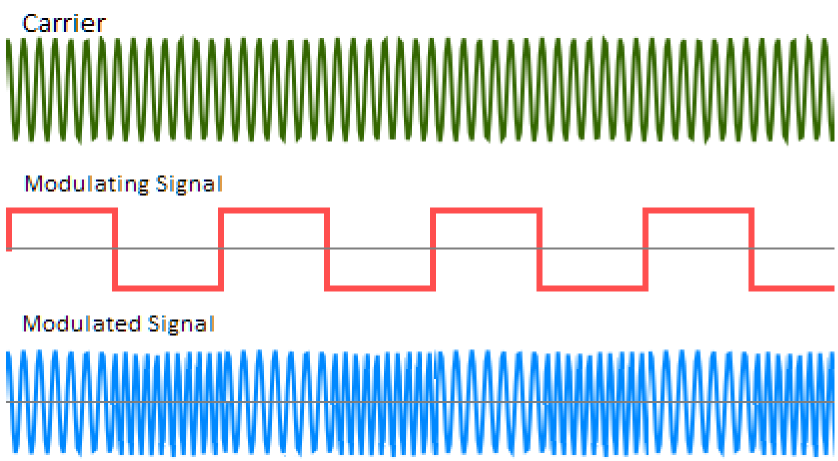


Figure 4.5 Frequency shift Key

Application:

(a) Used in our ultraviolet faraway controls.

(b) Used in fibre visual source.

3. PSK (Phase shift key)

The phase of the carrier is shifted for this modulation. The phase property of the carrier is changed as per the modulating signal.

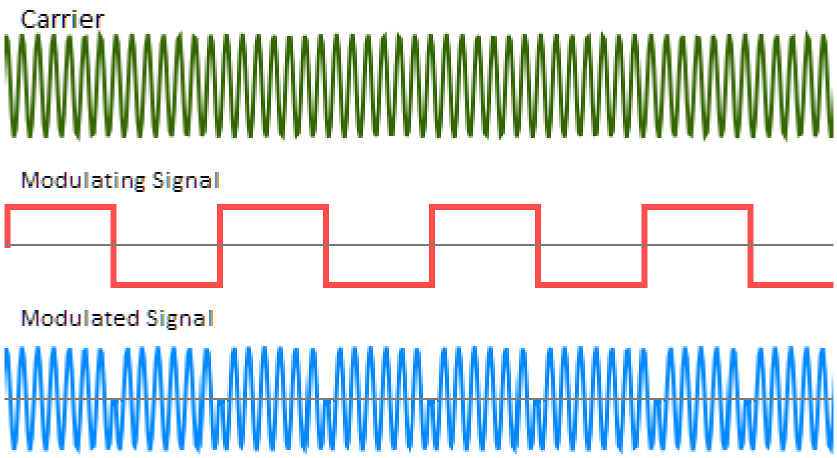


Figure 4.6 Phase shift Key

In this scheme, the numeral inf0rmation is encrypted inside the phase parameter of a carrier signal. PSK uses a limited variety of phases, each alloted a explicit form of binary numbers. In other words, each configuration of bits practices the representation that is quantified by the specific phase. Depending upon the number of bits the Phase shift keying can be altered according to the needs of the communication. For the transmission range, or the data bit rate the higher order of the phase shift keying can be utilized. For the tolerable range of the interference also the higher order of the phase shift keying can be used. As shown in the figure 4.6 , the carrier has the varying phase and the frequency and it is analog in nature , whereas since it is the digital modifying technique the modulating signal is the digital one[18]. It has the high bit value for the 1 and the lower bit value as -1. It is also called as non return to zero carrier signal also. Since the change for the carrier signal takes place at every 180 degree phase shift , that is it changes fromhigher level to low level in this bit duration. Similarly the message signal is also varied in line with the modulating signal. Depending upon the clusters of the bits the phase shift keying can be binary , quadrature ,and higher order depending upon the bits used.

4.2 Applications

This table elaborates the applications for all modulation designs in both wireless communications and video.

Modulation	Application
MSK, GMSK	GSM, CDPD
BPSK	Deep space telemetry, cable modems
QPSK, π/4 DQPSK	Satellite, CDMA, NADC, TETR PHS, PDC, LMDS, DVB-S, cable systm
OQPSK	CDMA, Satellite
FSK, GFSK	DECT, paging, RAM mobile data, AMPS, CT2, ERMES, land public safety
8PSK	Satellite, aircraft, telemetry pilots for monitoring broadband
16 QAM	Microwave digital radio, modems, DVB-C, DVB-T
32 QAM	Terrestrial microwave, DVB-T
64 QAM	DVB-C, modems, broadband set top boxes, MMDS
256 QAM	Modems, DVB-C (Europe), Digital Video (US)

Table 2 Applications of digital modulation

4.3 Quadrature Amplitude Modulation (QAM)

QAM is a modulation order which carries information by modifying the amplitude of two carrier waves. The two waves which are 90 degree lagging to each other are varied in their amplitude. Since these waves are 90 degree apart hence the method is said to be quadrature amplitude modulation[25]. The process shown below gives the idea about the generation of the quadrature amplitude modulated wave. The apparatus given below provides the complete picture from the generation, modulation,reception and demodulation of the system. The two message signals are shown below as $m_1(t)$ and $m_2(t)$.

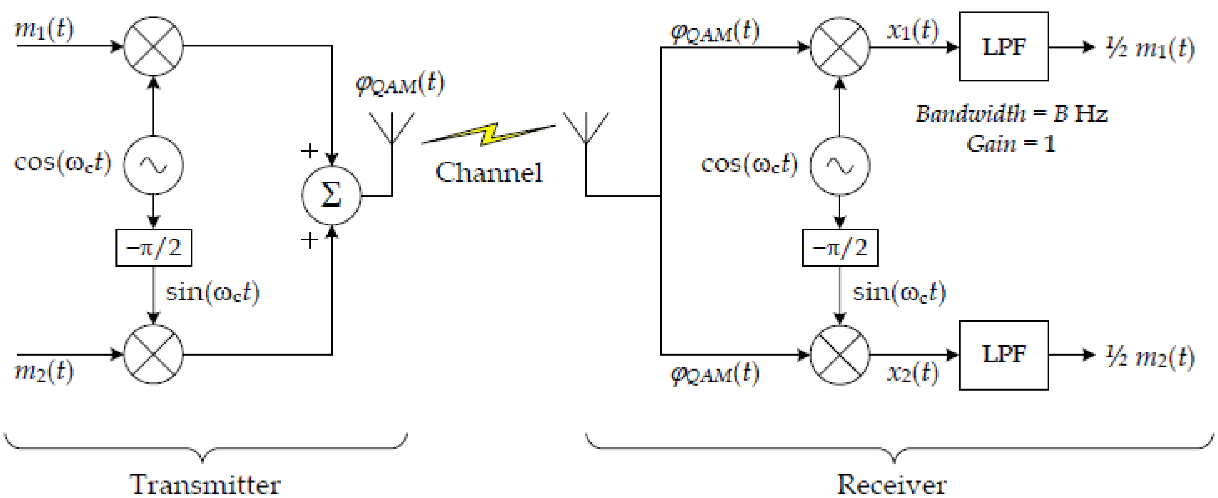


Figure 4.7 Quadrature amplitude modulation

The message signal 1 and message signal 2 is input to the product modulators. The local oscillator generates the cosine wave which is in turn multiplied with the message singals.

QUADRATURE = SINE WAVE + COSINE WAVE

The waves having the higher frequency is curbed by the lowpass filter (LPF). The second message signal is shifted by 90 degrees in order to make the whole modulation process work. Then these two information waves are further sent to the adder[8]. The adder at theend of the transmission terminal add both the signals and produce the required quadrature amplitude wave.

4.4 M-ary QAM

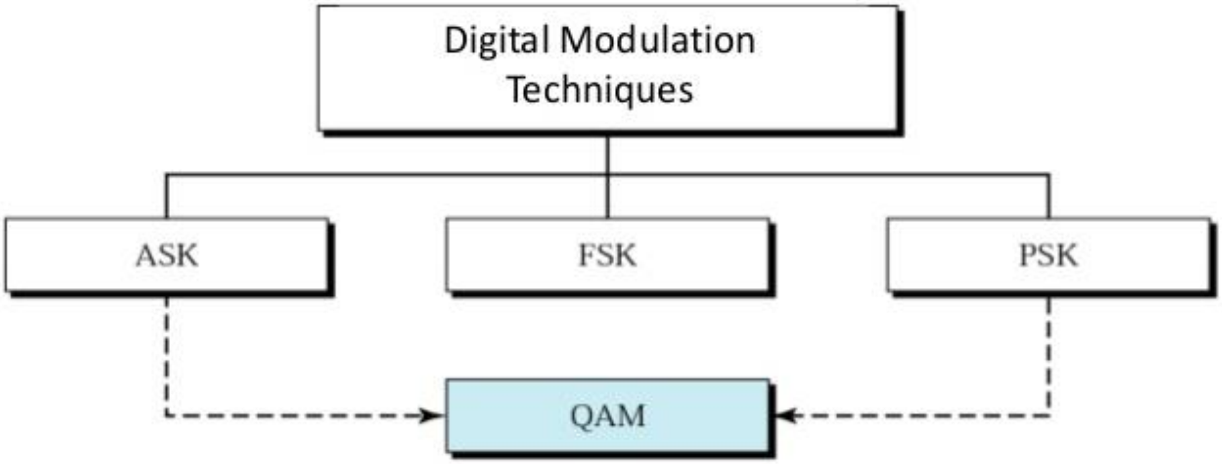


Figure 4.8 QAM as fusion of ASK and PSK

M-QAM can be described on the basis of the constellation diagrams which basically plots the I and Q of the given modulation scheme[19]. Depending on the value of M the QAM can be varied as 4-QAM, 8-QAM, 16-QAM, 32-QAM etc.

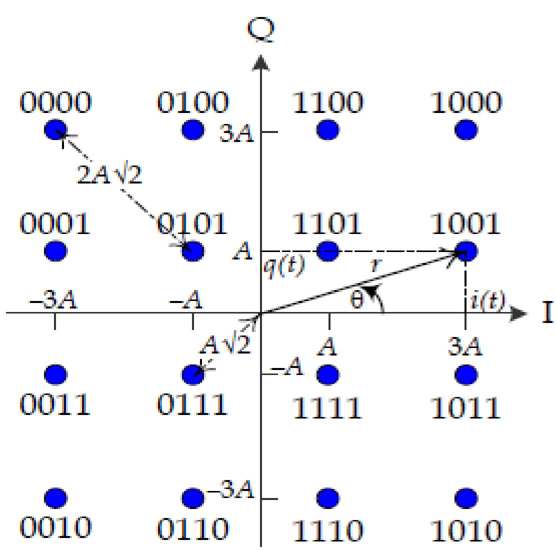


Figure 4.9 16 QAM

For sample, the 16-QAM has been discussed here in detail to provide idea about the process. To guide the bit order 100101110000 by 16-QAM, the bits are selectively parted, according to the phase lock and the amplitude variants[10]. After the bits are parted they are given the phase shift and the amplitude alteration for the time period it is defined for. The two incoming waves are shown here below which will vary according to the phase as the amplitude for the system would remain the same.

$$F_1(t) = \sqrt{\frac{2}{T_b}} \cos(2\pi f_c t) \qquad (4.1)$$

$$F_2(t) = \sqrt{\frac{2}{T_b}} \sin(2\pi f_c t) \qquad (4.2)$$

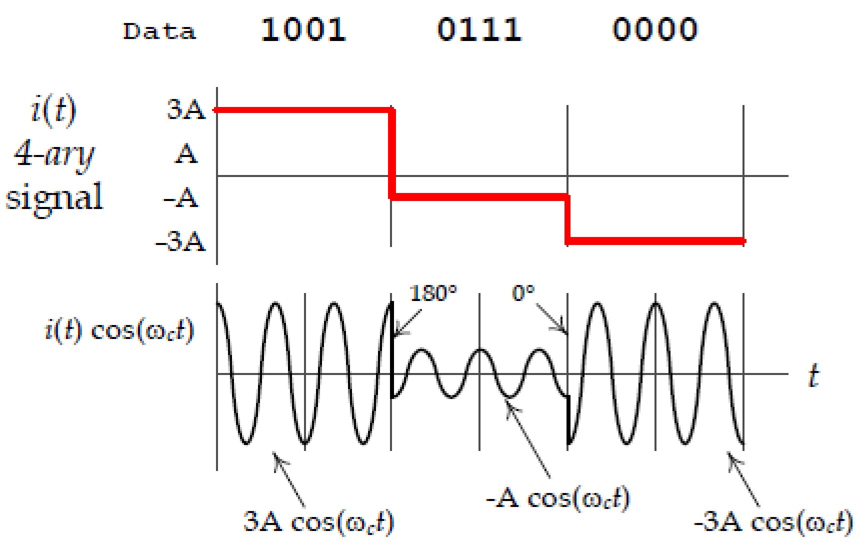


Figure 4.10 16 QAM Bit stream

The time domain analysis of the of the 16 QAM has been shown in figure below. The phase varying of the signals are in accordance with the sort of bits that has been received at the junction of the product modulator[14]. The distance of the two points in the constellation figure gives the precise idea of the happenings. The distance between the two points gives the probability of the two bits to overlap with each other.

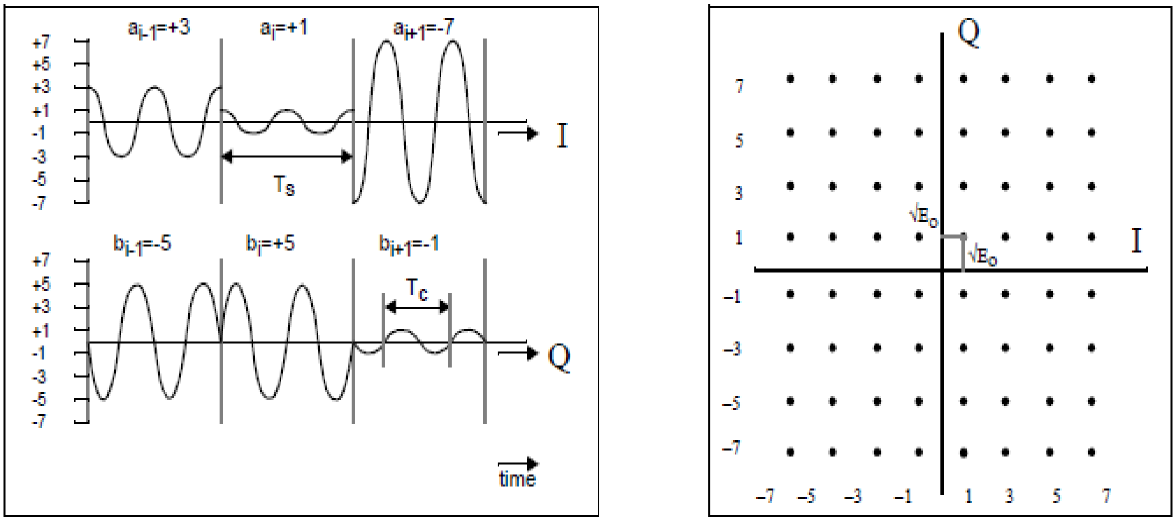


Figure 4.11 16 QAM time domain analysis and 64 QAM

4.5 QAM demodulation

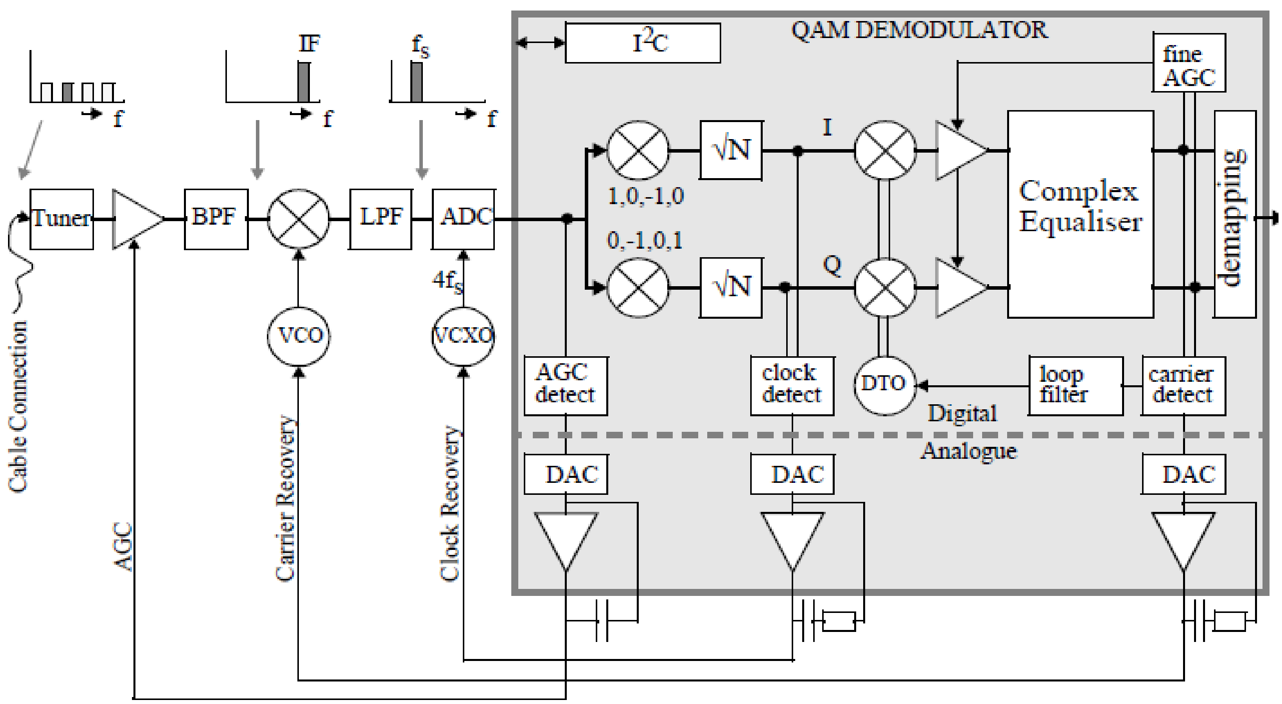


Figure 4.12 QAM demodulation

(Half) Nyquist Filtering

1. Pulse Modelling mandatory to gather ISI=0 in restricted BW
2. ISI is equal to zero when zero overpass happen at Ts=1/fs
3. Merge of Transmitter & Receiver achieve Nyquist Criterion
4. Interruption is in the loops and therefore effects the demodulator construction

Clock Detector

- Energy Expansion procedure
- Afterward Partial Nyquist Filter to reach ISI as zero at indicator response
- Expected regulator has exactness (100 ppm at 7 Msym/s))
- The transmission range of the system me be less
- Delay in nsmission range is tolerable (no instability)

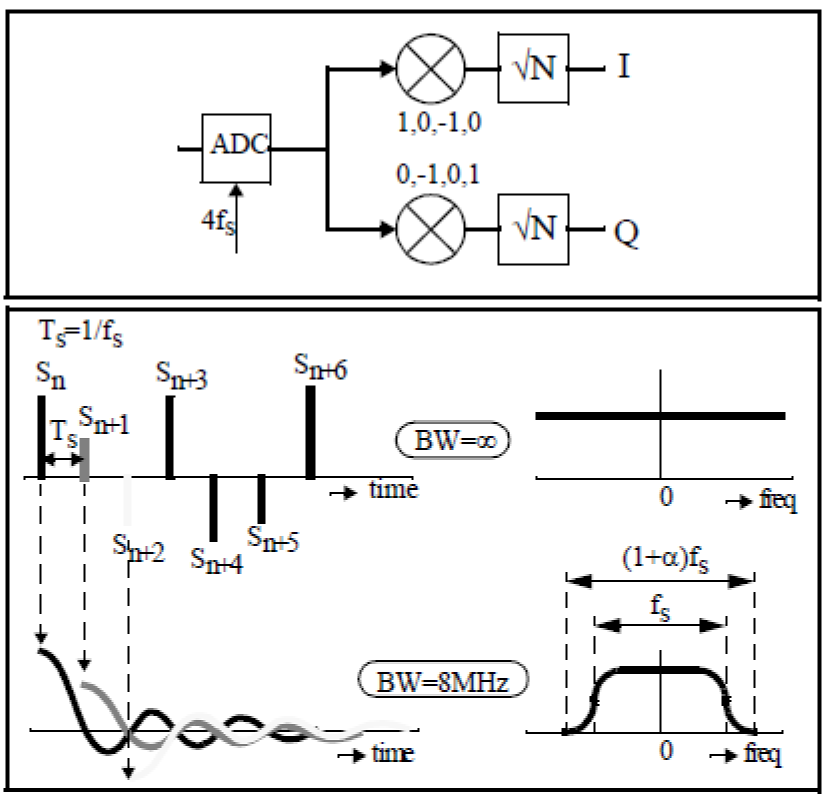
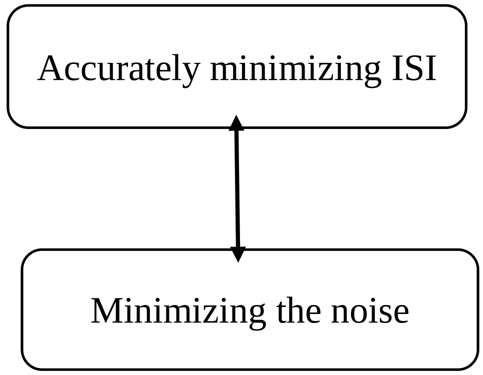


Figure 4.13 Nyquist filtering

Tradeoff occurs between:



4.5.1 Recovery of information

Functions	Results
Automatic Gain Control	Ideal location of constellation diagram in response window
Quadrature down conversion	I & Q base band signals
(Half) Nyquist Filtering	Pulse influencing
Clock Recovery	Sampling orientation for A/D Converter
Carrier Recovery	Carrier frequency location
Adaptive Equaliser	Reimburse for channel alteration
Demapping	Depiction of expected data in bits

Table 3 Recovery of signal

4.6 Measurements

(a) Modulation Error Ratio (MER)

Modulation Error Ratio (MER) is the relation of the communicated symbol vector and alteration between the transmitted and idyllic symbol path. MER can be measured through LabView Software[11]. It is measured in dB. For effective and improved signal receiving, MER should be high. The leading explanations for worsening of MER are i.) Improper signal level ii) Crash of data iii.) low quality modulation.

$$\text{MER (dB)} = \log_{10}(Psignal/Perror) \qquad (1)$$

An error vector is a trajectory in the I-Q plane among the idyllic constellation point and the point received by the receiver. It is the alteration between real expected symbols and idyllic symbols. For the percentage arrangement, root mean square (RMS) average is used.

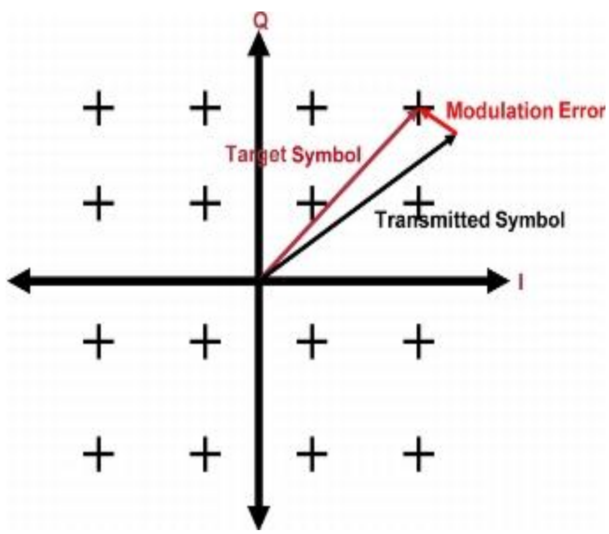


Figure 4.14 Representation of Modulation error

(b) Error Vector Magnitude

EVM is a amount of virtue for modulation precision. It delivers a method to ration and assess multi-level, multi-phase variation procedures like M-QAM and M-PSK. EVM ruminates totally of the potential phase and amplitude alterations as well as blare and offers a solitary inclusive dimension numeral for decisive the excellence of a circuit or product. By means of phasors in the I/Q plane, EVM exemplifies the position of ideal symbol vector position and extent equated to the actual measured vector (Fig 4.12) .

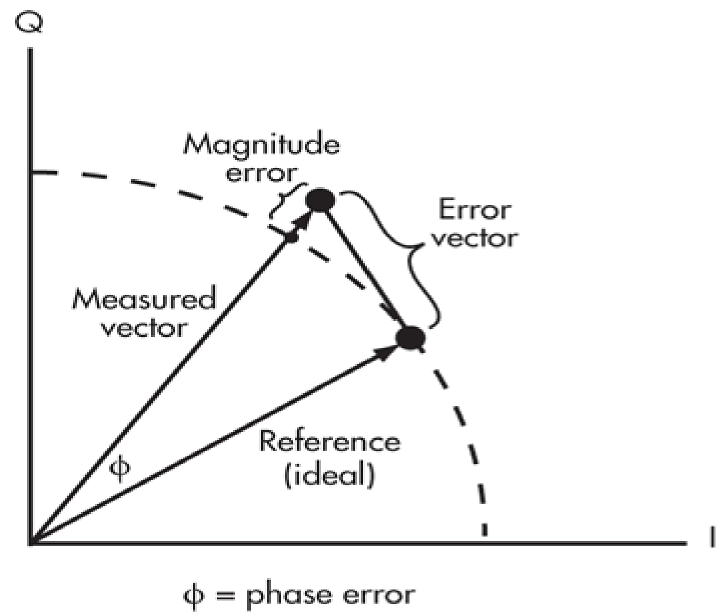


Figure 4.15 Error Vector Magnitude

The variance between the two is the EVM, which can be dignified on source modulator or receiver demodulator trails.

$$EVM\ (dB) = 10 \log (P_{error}/P_{ref}) \qquad (4.3)$$

$$EVM\ (\%) = \sqrt{(P_{error}/P_{ref})} \times 100 \qquad (4.4)$$

Root mean square (RMS) EVM, stated as a positive numeric scalar. It is the square root of the mean of the squares of all the values of the EVM[24]. Peak EVM, resumed as a positive numeric scalar. It is the major single EVM value intended diagonally all input values. For instance, an EVM of –35 dB is superior than one of –15 dB. In relations of percentage, –30 dB converts to 0.8% error while -15 dB translates to 5.9% error.

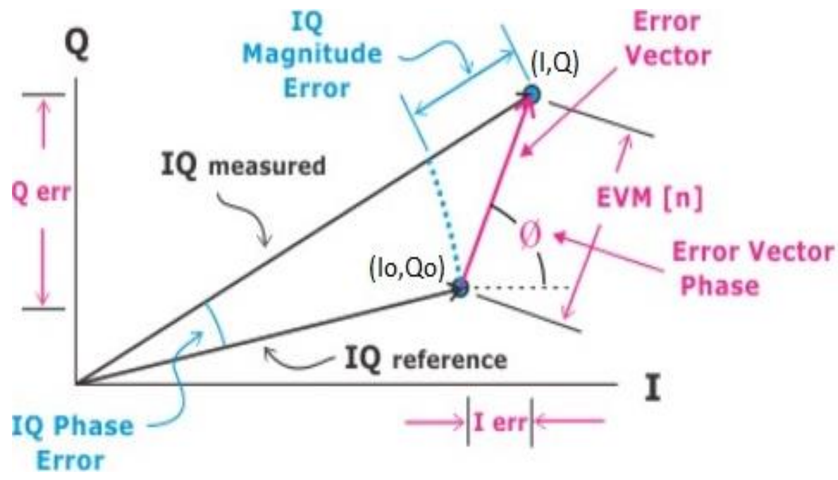


Figure 4.16 vector representation of EVM

Intersymbol interference is the change in the signal due the overlapping of data during telecommunication[28]. One code can hinder with other symbols bring about blare or a less reliable signal. The chief reason of intersymbol interference are multipath broadcast or non-linear frequency in networks. This has the outcome of a fuzziness or blend of signs, which can lessen signal strength. If intersymbol interference arises inside a structure, the receiver production converts flawed at the result platform.

Eye Opening and calculations

An eye diagram is an oscilloscope display that gathers digital signals from the receiver and applies them to vertical inputs, and uses the data rate to function the horizontal sweep. The motive for this is that the design in some sorts of cypher look like a sequence of eyes amongst a duo of signs. This is a apparatus to assess the collective influence of channel noise and the collaboration of codes in the presentation of the baseband pulse broadcast system. This is a synchronous overlay for all possible presentations of signals of notice witnessed during a given retro of time for a signal. Also the decrease in the eye opening is proportional to the increase in the value of M in QAM. Therefore eye opening can also be used as a parameter to judge the system performance based on intersymbol interference.

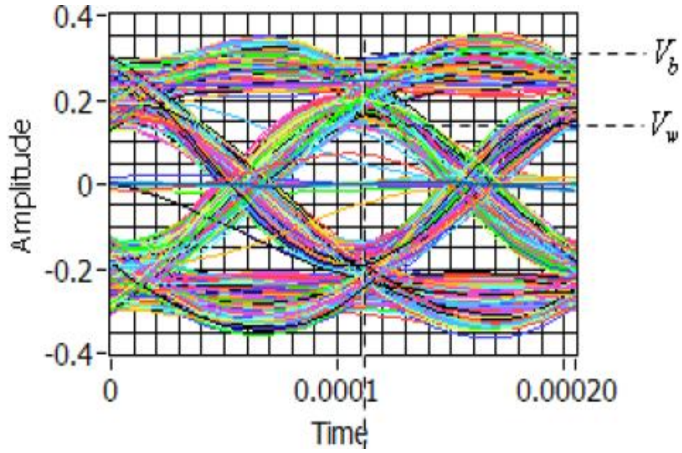


Figure 4.17 best-case and worst-case voltages

The best-case and worst-case voltages are shown as Vb and Vw respectively in the figure

$$\text{Eye opening} = \frac{2Vw}{Vb+Vw} \qquad (4.5)$$

From figure 4.17 , we can find the eye opening using the equation 4.5. Eye opening has been used as parameter in the further calculations. As it is evident from the figure that the eye opening is the eye like opening in the oscilloscope screen representation of the modulation scheme. V_b is known as the best case voltage which is the highest point on the tip of the upper portion of the eye opening, whereas V_w is known as the worst case opening which bottom tip of the upper portion of the eye opening. Eye opening when calculated as percentage should come out less than 100%. Also the decrease in the eye opening is proportional to the increase in the value of M in QAM. Therefore eye opening can also be used as a parameter to judge the system performance based on intersymbol interference. Lesser the eye opening more will be the intersymbol interference which is evident from the number of bits used in the type of M-ary QAM modulation used.

4.7 NI USRP-2922 Front Panel

The front panel of the USRP kit for the demonstration is shown in the following figure. The use of all the ports available is also shown in the diagram itself

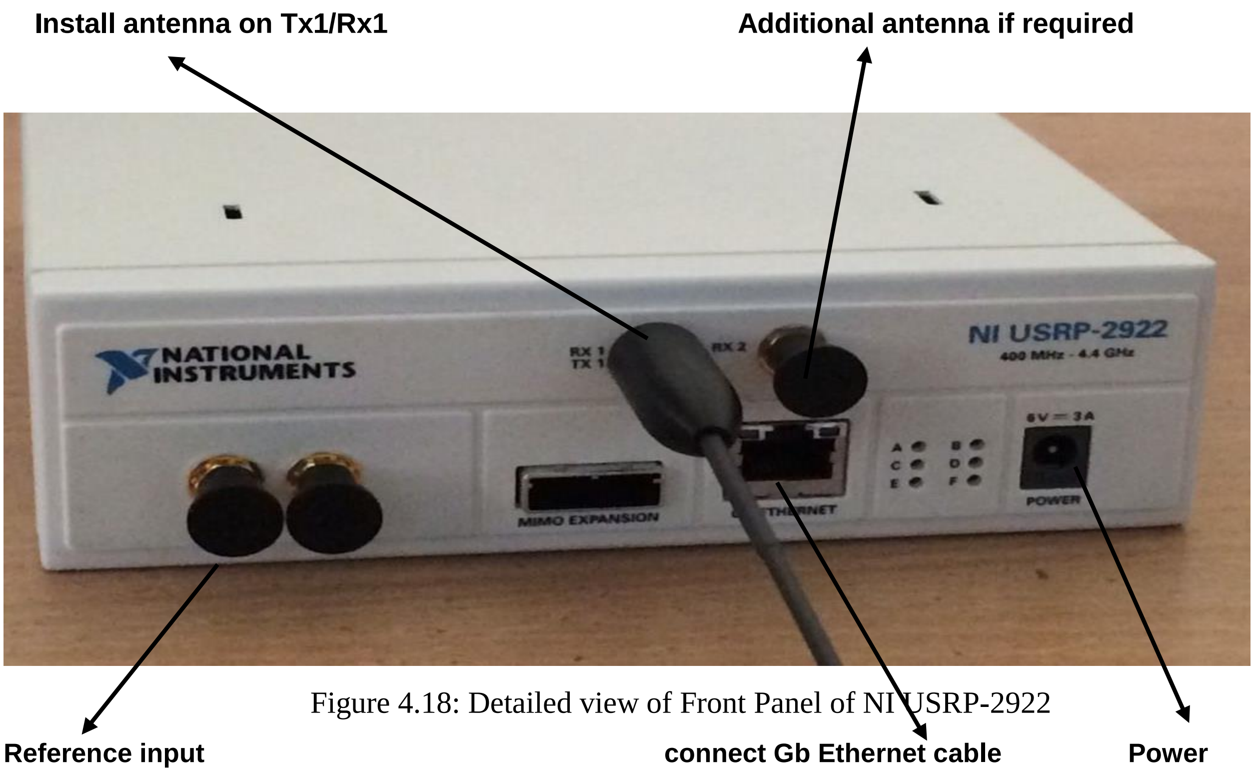


Figure 4.18: Detailed view of Front Panel of NI USRP-2922

4.8 SYSTEM IMPLEMENTATION USING SOFTWARE DEFINED RADIO

There were different methods by which we could implement the system model. Either we could

a) directly write a code in HDL for our system model,

b) use Simulink library,

c) system generator to convert a Matlab code to VHDL code,

d) Use Labview and the USRP kit which is provided by National Instruments Ltd.

These ideas could be implemented with the help of some sort of software and hardware. As the Labview and USRP was available in the research lab, so we decided to use this approach. For this approach the following hardware and Software tools were required

4.8.1. Software Specifications:

1. Processor: Intel (R) Core (TM) i3, CPU 1.80 GHz
2. RAM: 4 GHz
3. System type: 64- bit operating system
4. MATLAB version: 8.3.0.532(R2014a)
5. Labview version 14.0 (64- bit)
6. Driver for NI-USRP 2920 (version 1.4)

4.8.2. Hardware Specifications:

1. National Instruments USRP N-2920
2. spectrum analyzer
3. Scientech technologies Digital-Storage Oscilloscope

Comprehensive Specifications of Hardware Used

1. NI- USRP-2922 Kit (Features and specifications)

The Structure of NI-USRP-2922 is specified-below

I. Transmitter
   a. Frequency-range 50MHz-2.2GHz
   b. Frequency-step <1 kHz
   c. Maximum Output-Power (Pout)
      50MHz to 1.2GHz 50mW to 100mW (17dBm to 20dBm)
      1.2GHz to 2.2GHz 30mW to 70mW (15dBm to 18dBm)
   d. Gain-range 10dB to 31dB
   e. Gain-step 1.0dB
   f. Frequency accuracy 2.5ppm

II. Receiver
   a. Frequency range 50MHz-2.2GHz
   b. Frequency step <1 kHz
   c. Gain range 0dB to 31.5dB
   d. Gain step 0.5 dB
   e. Maximum input-power (Pin) 0dBm
   f. Noise figure 5dB to 7dB

4.9 NI-USRP 2922 Block diagram

The block diagram of NI-USRP 2922 kit is shown in the figure 4.15. The NI USRP kit connects to the PC to serve as a software-defined radio. For transmission, baseband I/Q signal samples are generated by the computer and given to the USRP-2922. The USRP mixes the incoming signal with 400 MS/s with the help of a digital up-convertor (DUC) and then transforms the signal from digital form to analog with a dual-channel, 16-bit DAC. The resulting signal in analog form is then mixed up with some specified carrier frequency.

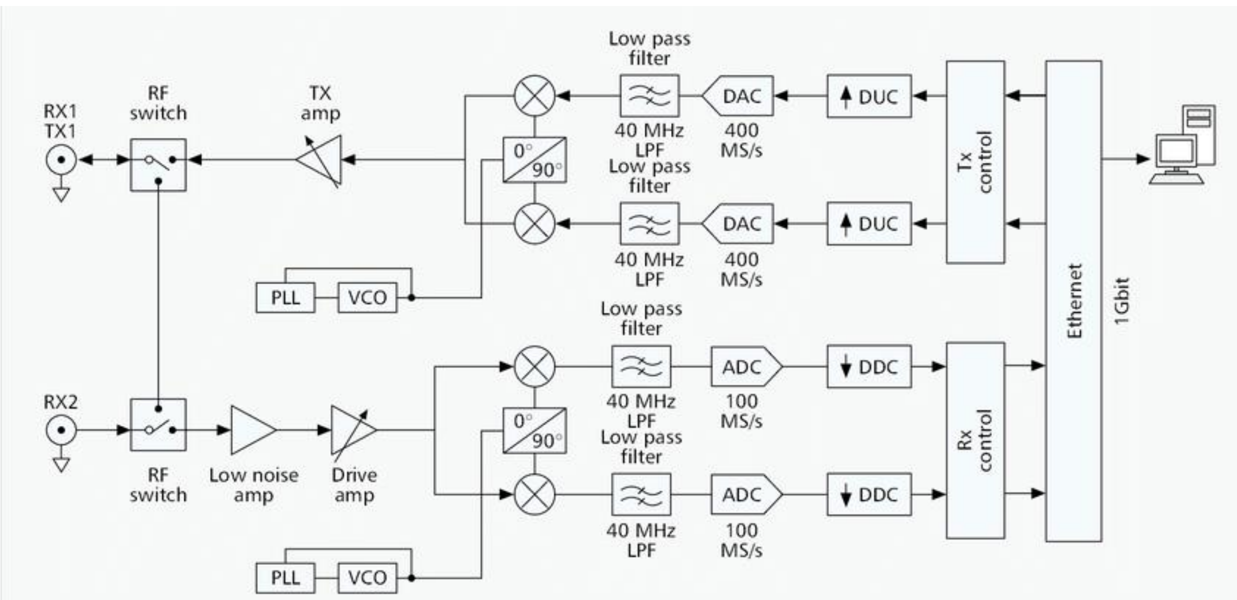


Figure 4.19 Detailed block diagram of USRP

The USRP assemblies the incoming signal with 400 MS/s with the aid of a digital up-convertor (DUC) and then renovates the signal from digital form to analog with a dual-channel, 16-bit DAC. The resultant signal in analog form is then mixed up with some specified carrier frequency. The technology used in the USRP makes it possible to change or alter the type of the modulation based on the modulation tool box which is inbuilt in the software used. Hence, the SDR as a new source of the effective communication has come into picture given its effectiveness being unparallel from the earlier technologies.

4.9.1 Linking of the host Pc with the kit

Before installing the software for the functioning of USRP, we need to install Matlab and LabVIEW. The latest version of LabVIEW available in the market is LabVIEW 2014 which was released for the users in August 2014. After installing the LabVIEW software, follow the following steps

- NI USRP Software Suite DVD should be inserted into the PC and installed.
- There are some optional products which if required may be installed such as LabVIEW Modulation Toolkit, LabVIEW Digital Filter Design Toolkit and LabVIEW Math script RT module.
- Keep the host PC powered on.

  The power cable should be connected to the USRP kit as shown in the following figure

Figure 4.20: Power connections to the USRP kit

Now, attach the cable or antenna to the terminals of the NI USRP-2920 front panel according to the requirement.

Connect the device directly to your computer with the included Ethernet cable as shown in the following figure

Figure 4.21: Ethernet cable and antenna connections to the USRP kit

- After confirming that the connections, are accurate, set up up the linkage proceeds some stretch to begin the message with the USRP unit.
- The IP addresses for the computer and each connected USRP device must be unique.
- To confirm the network connection, we must open the NI-USRP Configuration Utility.
- On this window, go to the Change IP Address tab of the utility. The USRP should come in the designated IP address accessible on the left flank of the tab.
- If the device name is not present in the list, check all the connections and power supply again from start.
- We can also change the IP address of the device by selecting the device from the list. The IP address of the device which we select is displayed in the Selected IP Address textbox.

We can enter the new IP address which for the device in the New IP Address textbox as shown in the figure below:

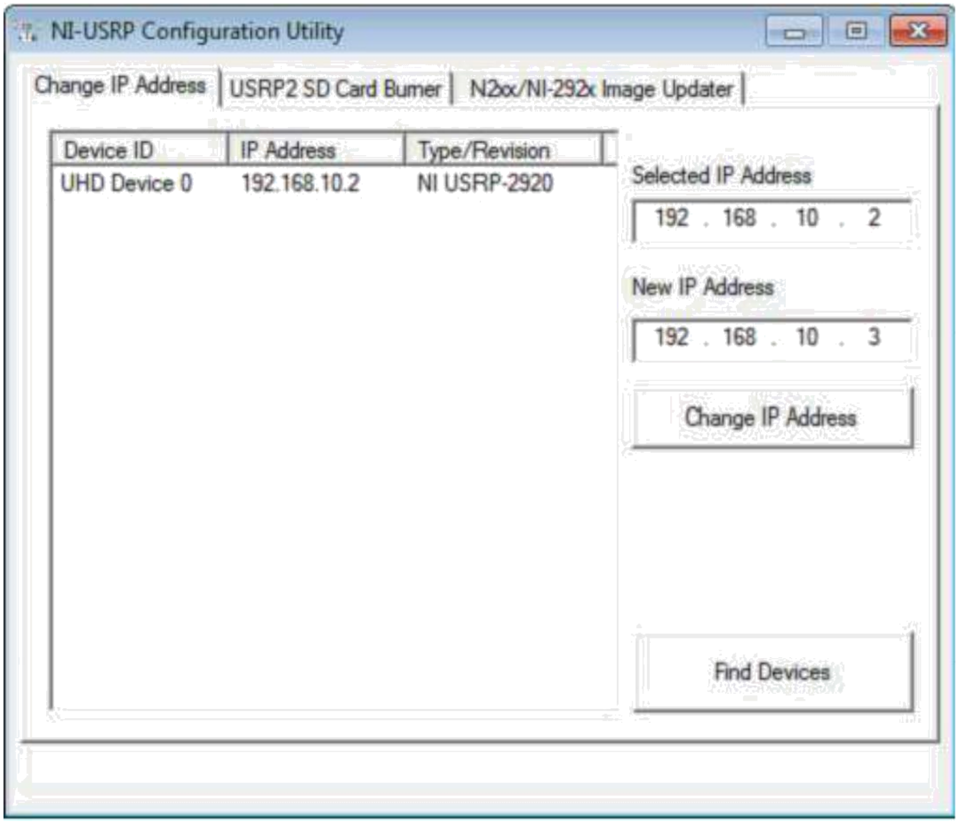


Figure 4.22: Changing the device IP of the device

Figure 4.23 LAB setup of USRP-LABVIEW

4.10 System Implementation On USRP

To implement the system on the software defined radio, we needed an interface between the hardware and the computer. The system model was designed in LabVIEW with the latest version drivers and the USRP driver software was installed for the interfacing with the hardware.

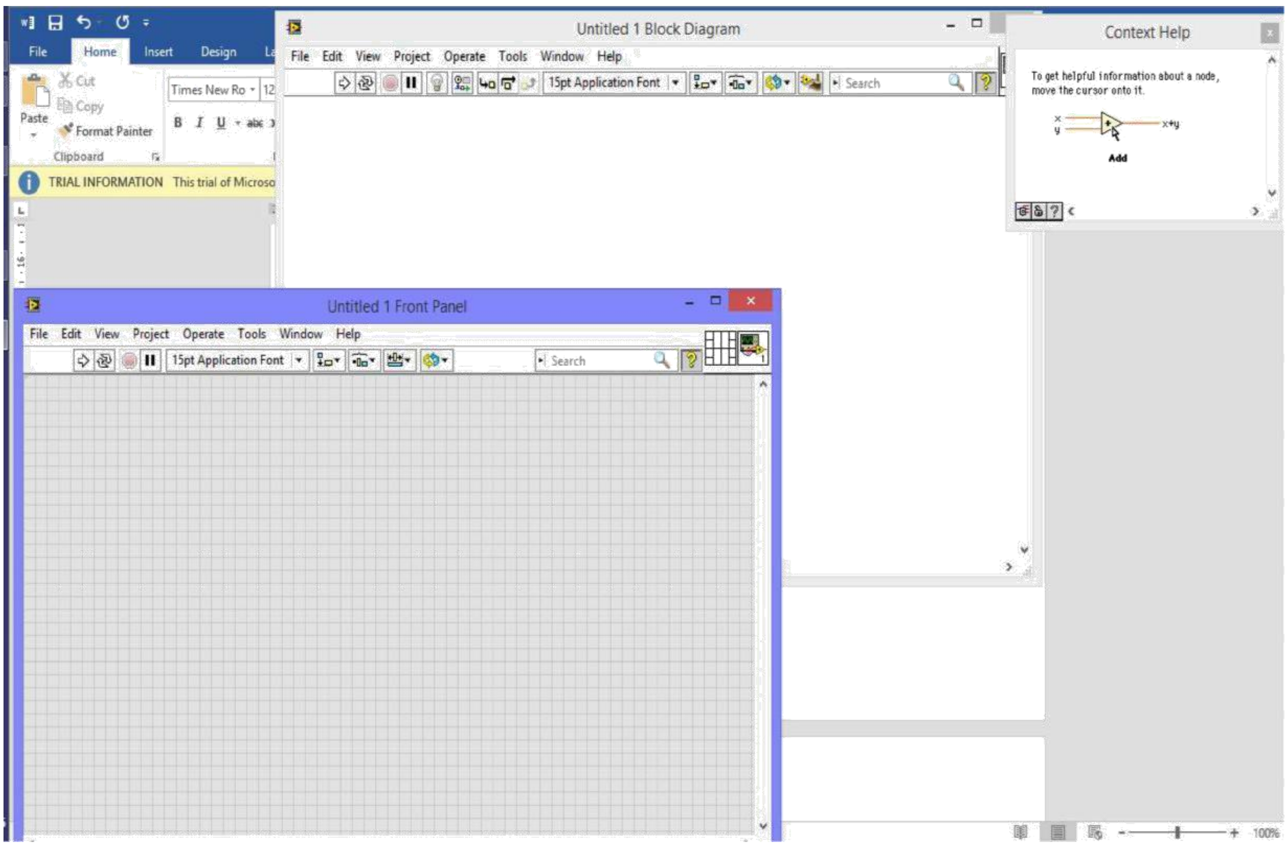


Figure 4.24: Block diagram and Front panel of LabVIEW

Chapter 5

Results and Discussions

This chapter presents the results obtained by simulating and implementing the M-QAM system. In the initial steps, the QAM system was designed with the help of different blocks. Some of these blocks were available for the use. Whereas, other were designed according to the need of the system. And some of them were generated by creating different Virtual instrument in the Labview. The whole system was designed along with the transmitter and receiver. The block diagram of the transmitter is shown in the following figure:

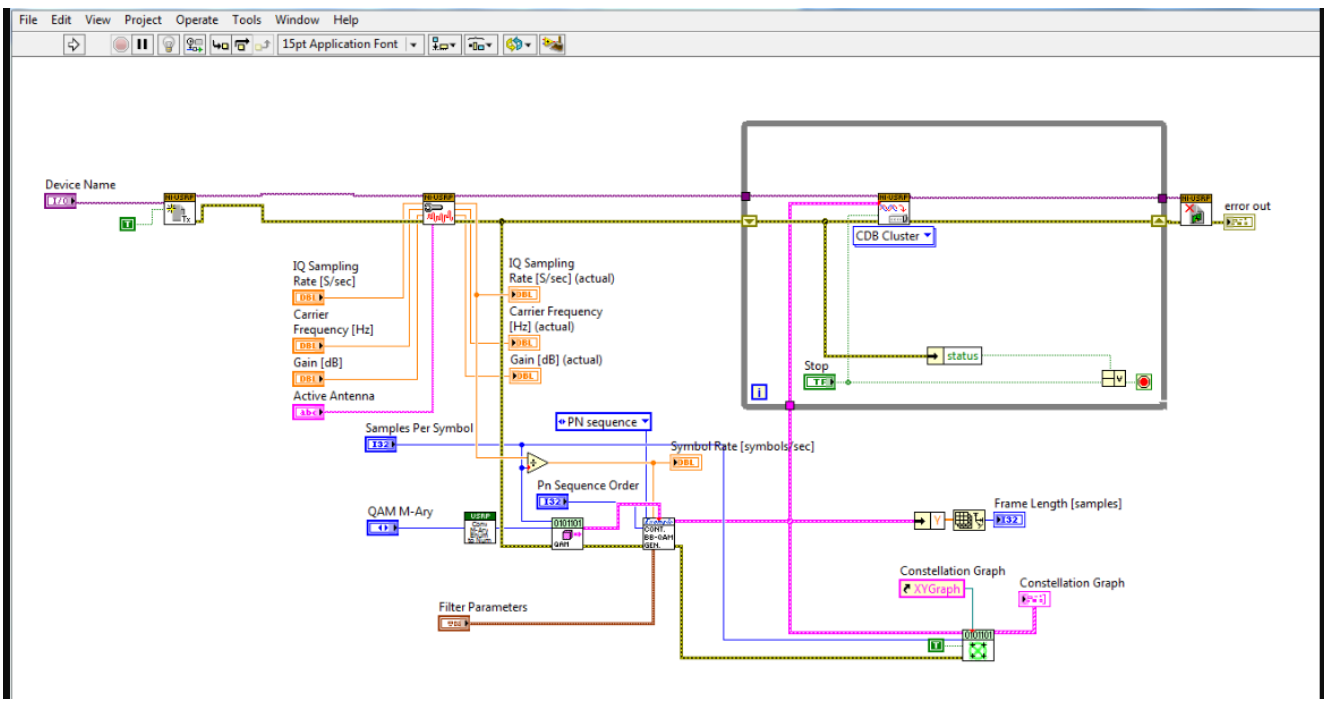


Figure 5.1: M-QAM transmitter design with USRP using various blocks

The front panel of the transmitter is shown in the following figure. The constellation diagram of the 16 QAM is shown in the figure.

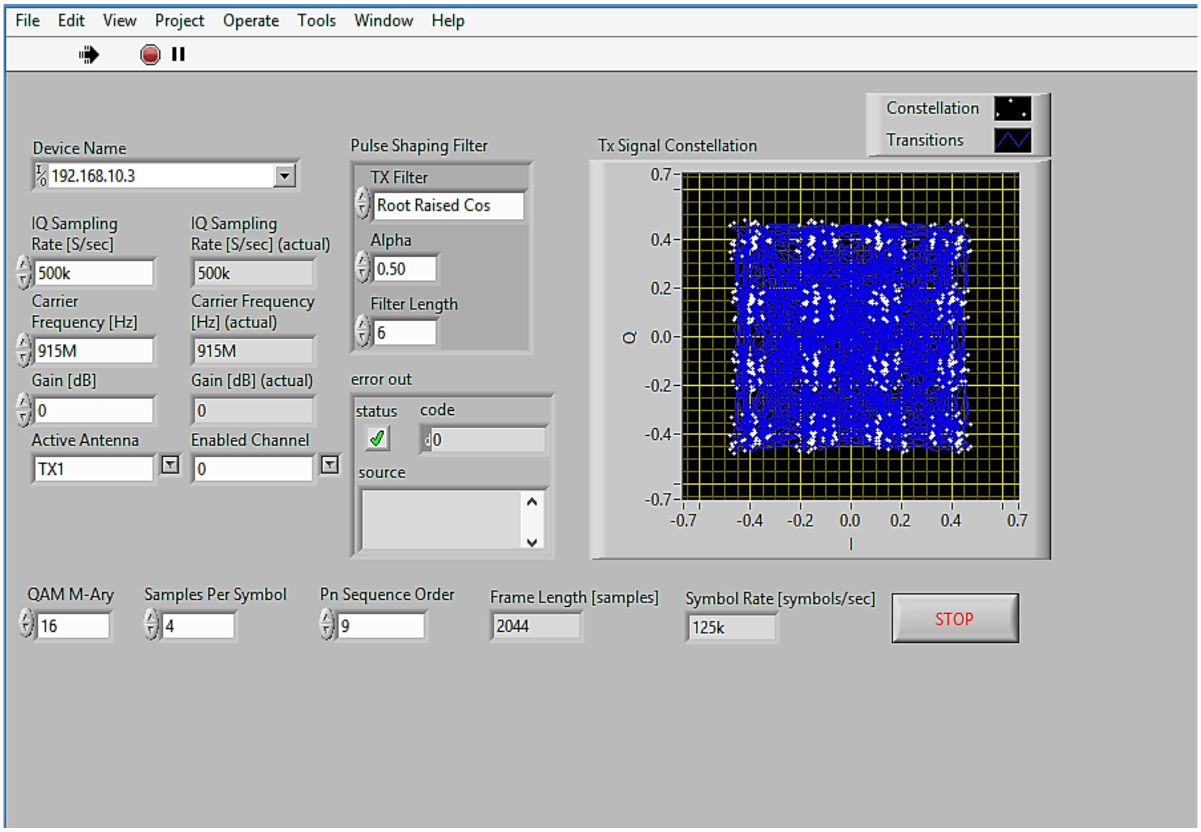
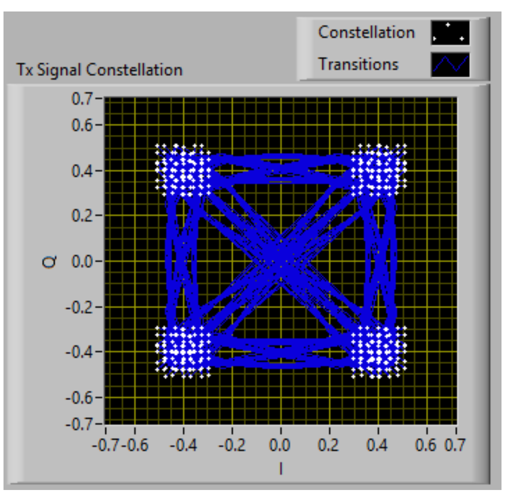
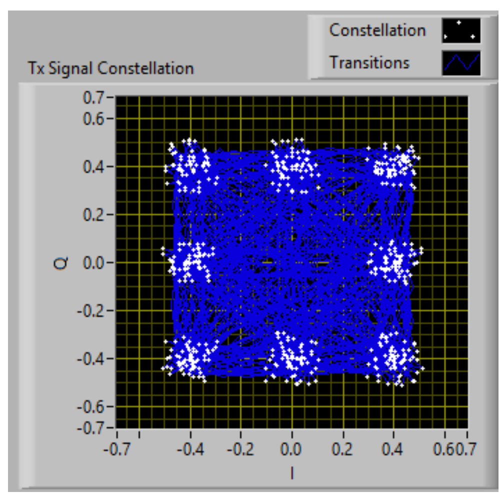


Figure 5.2: Front panel of 16 QAM transmitter with USRP



4 – QAM constellation diagram



8 – QAM constellation diagram

The block diagram of QAM receiver is shown in the following figure:

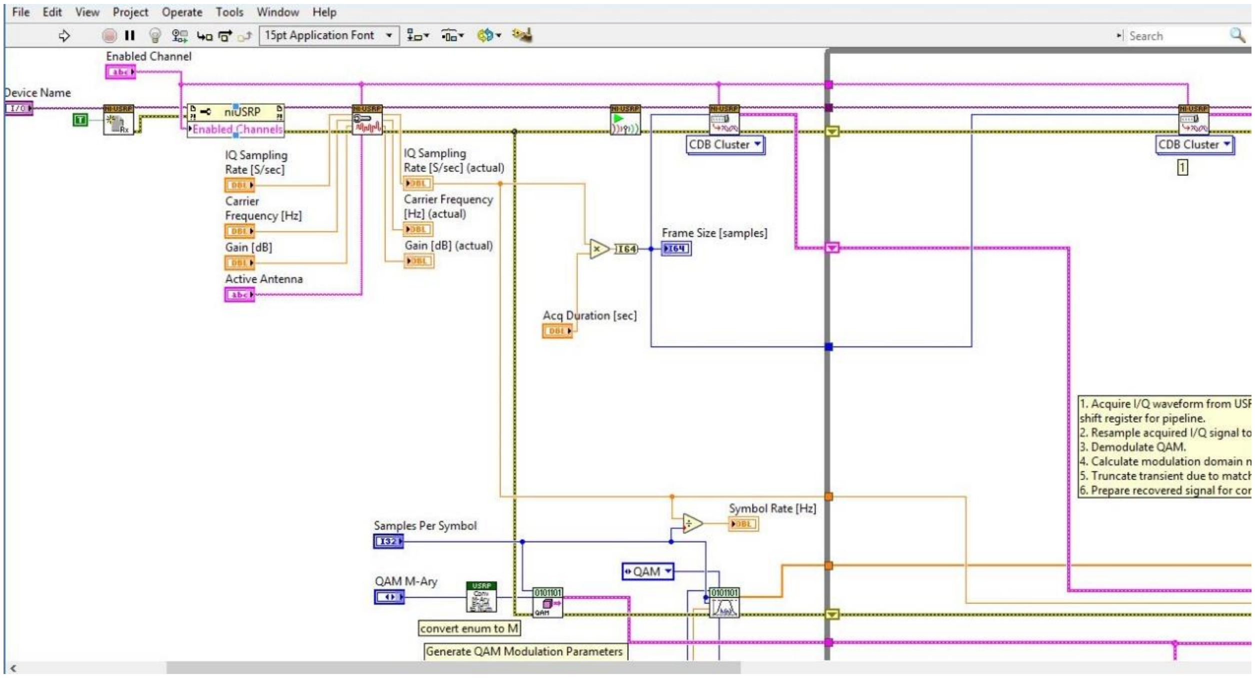


Figure 5.3 Block diagram of QAM receiver 1

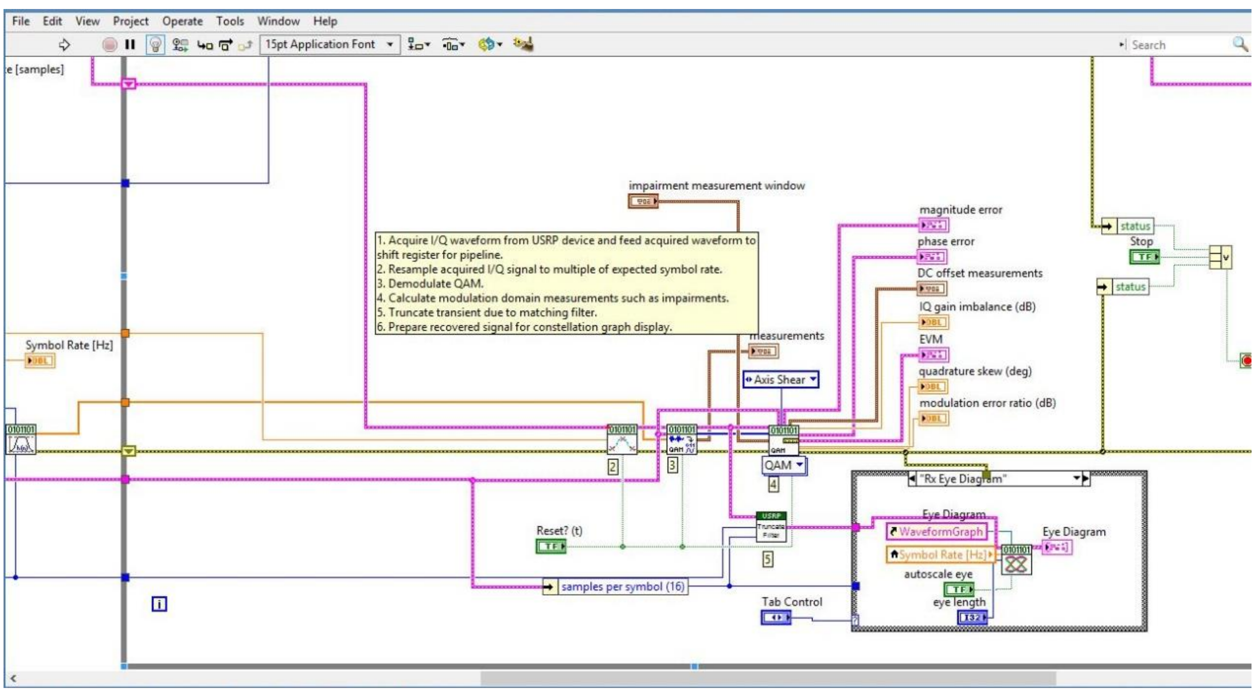


Figure 5.4 Block diagram of QAM receiver 2

The front panel of the receiver is shown in the following figure. The constellation diagram of the 16 QAM is shown in the figure:

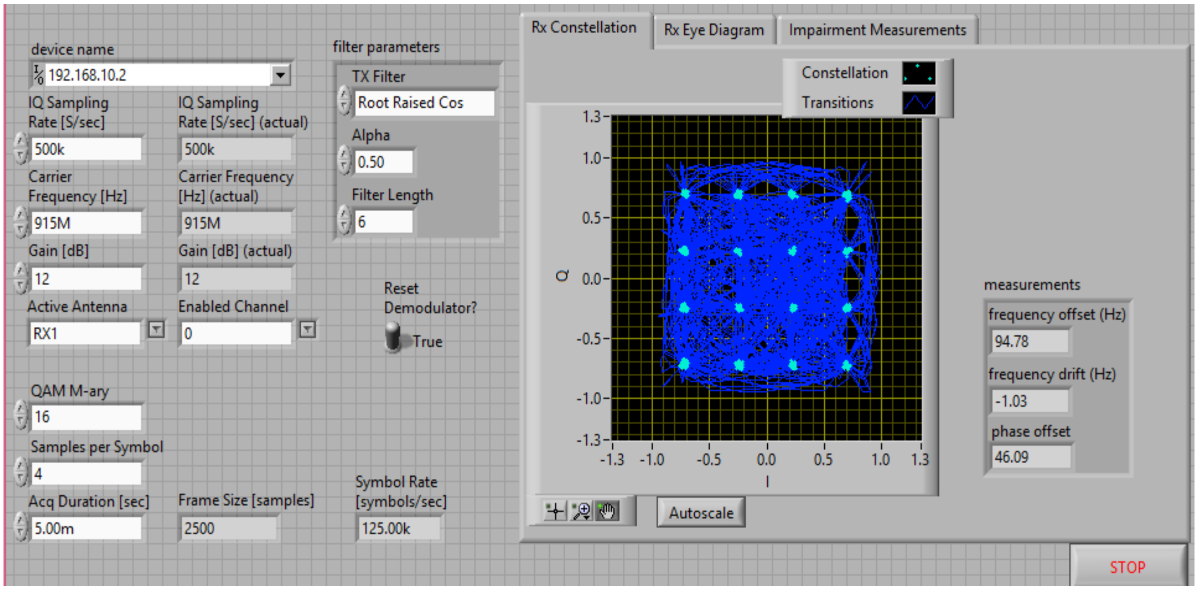
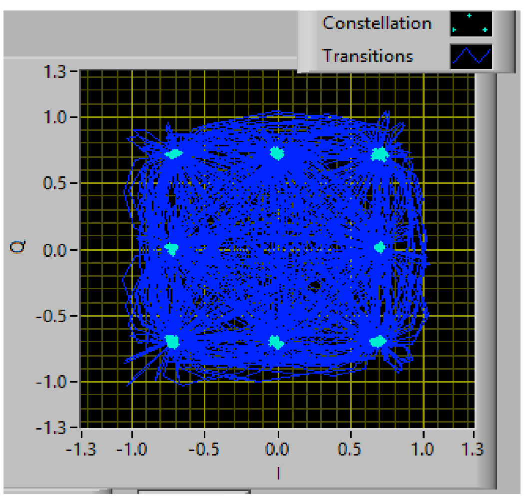
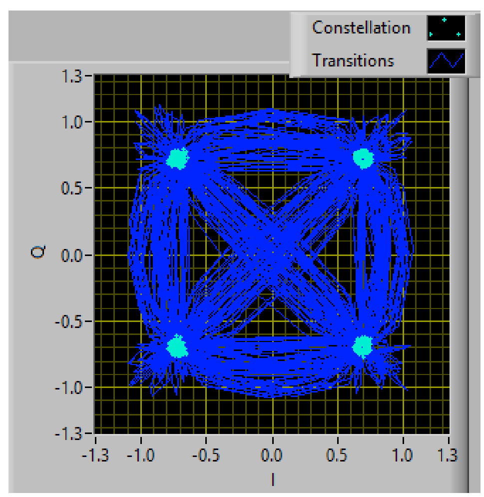


Figure 5.5 The constellation diagram of the 16 QAM receiver



4 – QAM constellation diagram (receiver)



8 – QAM constellation diagram (receiver)

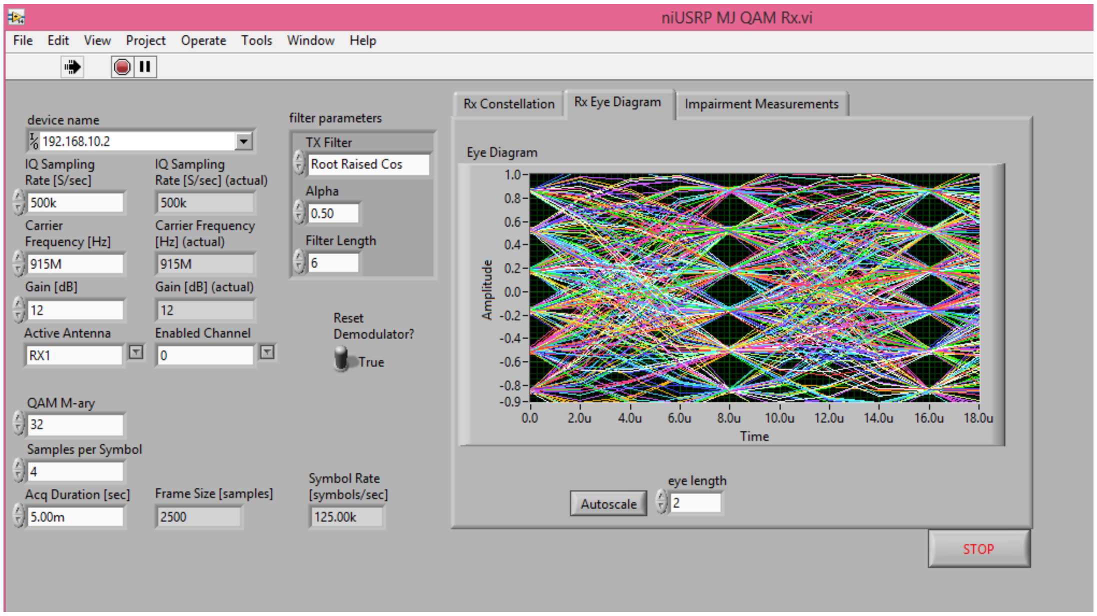
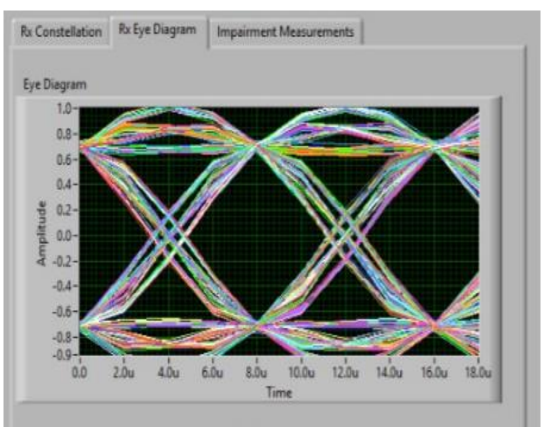
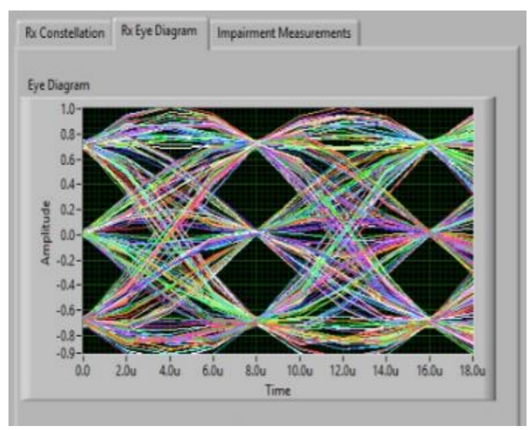


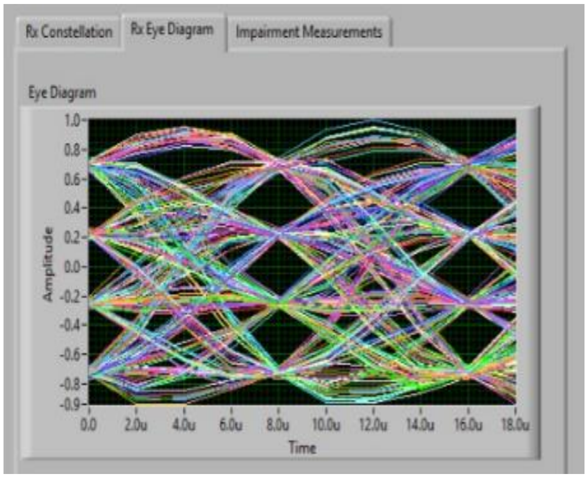
Figure 5.6 32-QAM Eye Diagram



4-QAM Eye Diagram



8-QAM Eye Diagram



16-QAM Eye Diagram

Calculations

Modulation technique	Modulation error ratio (MER) dB	Error vector magnitude(EVM)%	Magnitude error (%)
32-QAM	23.19	5.30	5.50
16-QAM	26.78	4.44	3.78
8-QAM	27.90	3.48	3.10
4-QAM	31.33	2.71	1.89

Table 6 Relative constellation error parameters for 0.3m

Modulation technique	Modulation error ratio(MER)dB	Error vector magnitude(EVM)%	Magnitude error (%)
32-QAM	21.33	6.54	6.21
16-QAM	24.01	4.71	4.97
8-QAM	25.75	3.75	3.96
4-QAM	27.28	3.01	3.26

Table 7 Relative constellation error parameters for 1m

we can calculate the eye opening for various M values for M-QAM modulation. This in turn gives us estimation for intersymbol interference.

Modulation technique	Vb (volts)	Vw (volts)	Eye opening (%)	Intersymbol interference(ISI)
32-QAM	0.8	0.6	85%	Very high
16-QAM	0.75	0.62	90.5%	High
8-QAM	0.78	0.67	92.4%	Medium
4-QAM	0.9	0.85	97%	Low

Table 8 ISI for various M-QAM

It has been concluded from this project that as the number of Bits i.e M value is increased the corresponding value of MER decreased and EVM increases. This concludes that the higher the no. of bits per symbol can be transmitted but at the same time due to closeness of the constellation points , the link is more susceptible to noise higher is the inter symbol interference (ISI). Hence, depending on the need of short distance or long distance communication we can use M-QAM modulation technique. For the long distance communication and tolerant intersymbol interference we can use the higher M bits as it will provide greater bandwidth whereas for short distance we can use lower M bits with higher data speed. Also the decrease in the eye opening is proportional to the increase in the value of M in QAM. Therefore eye opening can also be used as a parameter to judge the system performance based on intersymbol interference. The project aim lies in the fact that we can adjust the modulation technique using the integrated system of the hardware and software without disrupting the communication.

Chapter 6

Conclusion & Future Scope

The gist of the project was to yield the design of the wireless communication system using Software Defined Radio using quadrature amplitude modulation format and to link network connections such as constellation drawings, eye diagram, modulation error measurement, Error Phase, Magnitude error, Error Vector Dimensions in various technical fields. Our complete system (including hardware and simulation simulation) works according to the first specifications and requirements of our project. Scheme Comparison is performed on LabVIEW software while the matching hardware works through the Universal Software Peripheral Radio (USRP) providing an expanded function in advance. The USRP used for our project works in a range of 400MHz-4.4GHz. Statistics and ratings are taken by 915MHz frequency with 12dB antenna profits.

From this project, we concluded that using the integrated system of the hardware and software which is USRP and LABVIEW in our case we can adjust the modulation methodology according to the need of present communication system. The built-in transceiver here is reinstated and because of that unlimited world, it can be used in most 5G techs.

6.1 Future Work

As the testing of 5G and further technologies improved significantly, there is a necessity to make overall communication more resourceful and effective. In the first instance, a Present 4G mobile speeds have the impending to provide 10s of Mbps. 5G bids to take that on gigabits per second in a effective method, and we have to study numerous features of the growth of the face. SDR permitted strategies have a good economic applications, such as multimedia satellites, 5G shortcuts launches, and MIMO major message systems.

Fresh novelties in hardware won't be very beneficial, nonetheless, if the software and gears don't trail. That is the entire idea of SDR. To empower the progress of the chips, as well as the waveforms and submission software administration on them, there will be a prerequisite for improved system-level gears that can be used to project and restore across the analog and digital fields. As SDRs develop used for progressively composite responsibilities, they are being considered with more influential FPGAs deliberated for rigorous DSP. As a consequence, there is an predictable mounting necessity for FPGA gears that can grip rapidly amassed quantities of data,figures,statistics and complexity.

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