DSSS implementation using the Multi-rate Filter Banks

DISSERTATION- II

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By

Vibhooti Sharma

Under the guidance of

Mr. Amanjot Singh

Department of Electronics and Communication,

LOVELY PROFESSIONAL UNIVERSITY

Transforming Education Transforming India

Lovely Professional University, Phagwara

April, 2015

CERTIFICATE

This is to certify that the Thesis titled "**DSSS implementation using the Multi-rate Filter Banks**" that is being submitted by " Vibhooti Sharma" is in partial fulfillment of the requirements for the award of MASTER OF TECHNOLOGY DEGREE, is a record of bonafide work done under my /our guidance. The contents of this Thesis, in full or in parts, have neither been taken from any other source nor have been submitted to any other Institute or University for award of any degree or diploma and the same is certified.

Date: Signature of Advisor

Name: Mr. Amanjot Singh

UID: 15848

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Ex ami n e r I **Ex ami n e r I I**

DECLARATION

I hereby declare that work embodied in this pre-dissertation was carried out by me under the direct supervision of Mr. Amanjot Singh, Professor. This work has not been submitted in part or in full in any other university for any degree or diploma.

Vibhooti Sharma

Place: Jalandhar

Reg. No. 11004276

Date: 4th april, 2015

Department of Electronics and Communication

Lovely Professional University

Abstract

Digital signal processing can be defined as the mathematical interpretation of the data to change in other good way. The data or the information signal can be characterized by the representation of discrete frequency and discrete time. The aim of DSP is to compress the continuous real-world analog signals. As this process is carried by converting the signal from analog to digital by the sampling process and then by using ADC digitizing the data which changes the analog signal into a stream of discrete digital values. At the receiver side DAC is used to get the data in analog form. DSP is used in the communication system because they are having the applications in audio signal processing, video compression, speech processing, digital communication, radar, sonar and in audio compression. Efficient and successful communication of messages via imperfect channel becomes the major triumphs of information technology today. As the demand of communication channels is increasing the importance of proper use of bandwidth becomes necessary. Communication channels can be wireless or wire line channels or the combination of both. If the distortion or noise components introduced in the system the transmission of information with high rate and reliability under such unfavorable conditions has been possible because of information theory, signal processing, linear system theory and mathematics. Multi-rate systems include more than one sample rates. In some of the applications it is necessary to separate a signal into the set of sub band signals occupying usually the non-overlapping portions of the original frequency band but on the other hand it's also necessary to combine many sub band signals into a single composite signal occupying the whole Nyquist range. To achieve these conditions digital filter banks are required.

Here DSSS is designed by the use of interpolation and decimation as they secure the data by adding the signal at the transmitter end and removing those signals at the receiver end. After all this process the BER graph is drawn. Filters are also used to see the effects in BER curve. As the BER curve shows that how many bits are in error. It can be stated as the number of bits in error to the total number of transferred bits. In a noisy channel, the BER is often expressed as a function of the normalized carrier-to-noise ratio measure denoted Eb/N0, (energy per bit to noise power spectral density ratio), or Es/N0 (energy per modulation symbol to noise spectral density). The work here is concentrated over decreasing the BER as much as possible.

Acknowledgement

I would like to offer my sincerest gratitude to my guide Mr. Amanjot Singh for providing me the opportunity to work for my pre-dissertation he has the attitude and substance of a genius: he continuously and convincingly conveyed a spirit of adventure in regard to my research. Without his guidance and persistent help this dissertation would not have been possible.

Gratitude cannot be seen or expressed. It can only be felt in heart and is beyond description. Often words are inadequate to serve as a model of expression of one's feeling, specially the sense of indebtedness and gratitude to all those who help us in our duty.

I would also like to thank him for providing me technical knowledge in order to complete my pre-dissertation work successfully. I also extend my sincere thanks to all those who in their slightest manner helped me in direction of completing my project successfully

And last but not the least, we find no words to acknowledge the financial assistance & moral support rendered by our parents in making the effort a success. All this has become reality because of their blessings and above all by the grace of god. I am beholden to my family and friends for their blessings and encouragement.

Vibhooti Sharma

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CHAPTER - 1

INTRODUCTION

In communication system various blocks are used to transmit the information. The transmitted signal has to be passed through the source coding, channel coding, modulator block and then the signal is passed by the channel where the noise get added in signal after this at the receiver end source encoder, channel encoder, demodulator is used these are used to recover back the information signal that has been transmitted by the transmitter side. In communication system the only requirement is the secure communication that means the data send through the transmitter will transmit through security at the receiver side.[28]

1.1 Transactional Model of Communication

The transactional model of communication can be defined as the graphic representation of collaborative and on-going message transfer among persons, or an individual and a group of individuals, to get back information that has been sent to each other. A transmitter encodes the data or signal then send this message with the help of a channel to the other communicators end that is receiver end which then decode the message. This can be simply understand as arranging the thoughts into the words and then transfer these words through speaking, texting, email and then at the receiver side these words are taken and meanings of these words are applied. The message may get interrupted due to noise due to this message can't be received or fully understood as the sender intended at receiver side.

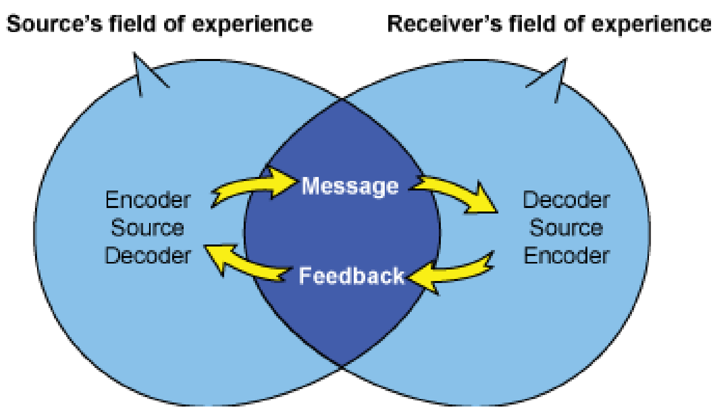

Fig 1.1: Transaction model for communication

The data can be stated as the information that has been processed, organized and stored. The data that can be used is numeric, symbolic, alphabetic and consist of one or more combination of binary-code alpha/numeric symbols, control codes, user address, progress data or data base information between two or more terminals. During the transmission of data the data can be in digital or in analog form, at the both ends that is source and destination data are in digital form. From this the data communication is defined as the process of transferring digital information between two or more points.

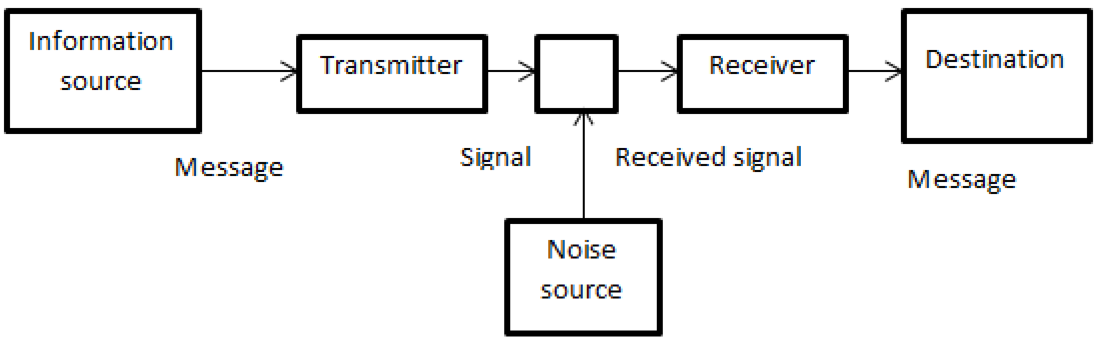

Fig 1.2 Communication model

1.2 Application of data communication-

The application of data communication can be seen in

I. Electronic mail
II. Videotext
III. Automated banking machines
IV. Teleconferencing

1.3 Need for filter in communication:-

Filters play an important role in communication channel

I. Effective in removing spectral leakage
II. Decreasing the channel bandwidth
III. Eliminating/reduce interference

1.3.1 Pulse Shaping Fundamentals

In communications systems, wireless communications channel require pulse shaping filters. The two important requirements of a wireless communication channel are

1) Generating band limited channels

2) From multi-path signal reflections to decreasing/reduce the inter symbol interference (ISI).

These achivements can be achieved with the use of pulse shaping filter. Pulse shaping filter can be functional to every symbol. By using sinc pulse we can generate the band limited channel as well as reduce the ISI from the multi-path signal because they efficiently use the frequency domain to use only a small part of the frequency domain, and due to the windowing effect this can be seen on every symbol period of the modulated signal. Sinc pulse in the symbol sideways with FFT spectrum of signal represented as:

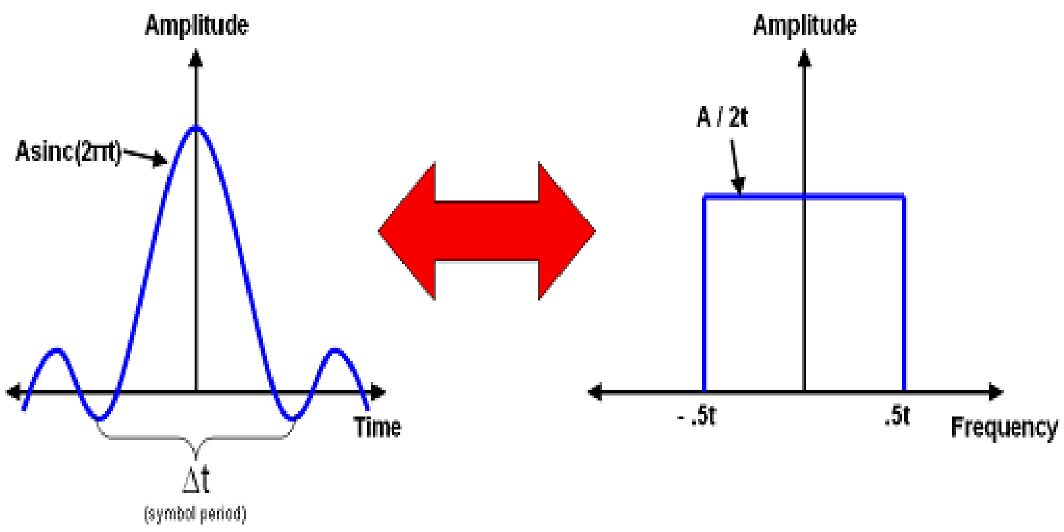

Figure 1.3: Time v Frequency Domain intended for Sinc Pulse

Figure shows, sinc pulse is periodic in nature also having extreme amplitude in center of symbol time. In the frequency domain it seems by means of a square wave and this be able to limit the communications channel to a frequency range that is specific.

1.3.2 Reducing Channel Bandwidth

When the modulation process of carrier signal is done then phase and amplitude of the signal shows the constant transitions. If we take a signal that is having time domain of a carrier signal using a symbol rate which is half of carrier. Phase and amplitude changes or transitions in the signal are seen at every two periods of the carrier. When filtering is not applied then sharp transitions occur and this can be clearly seen in the diagram.

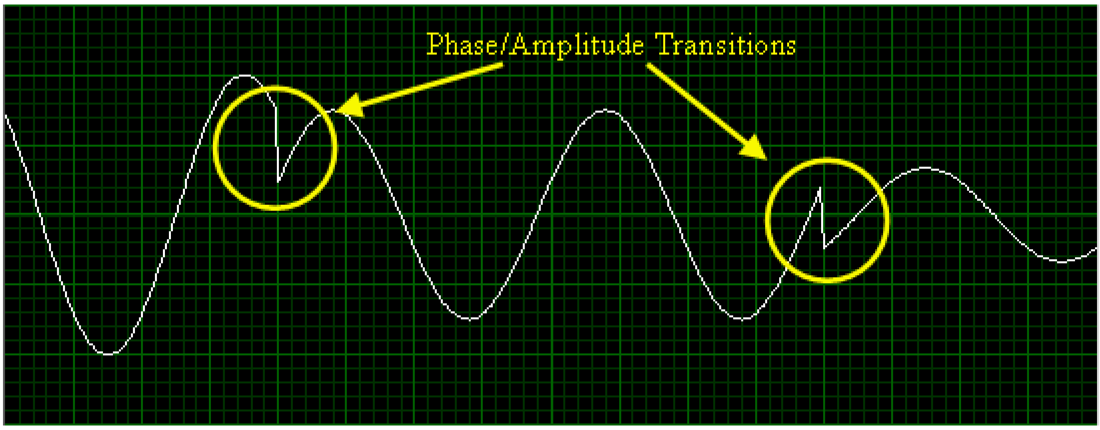

Figure 1.4: Phase in addition to Amplitude Transitions in Unfiltered Modulated Signal

Frequent or sharp transitions in several signal give rise to the frequency domains high-frequency component. At the right side of the graph, FFT is defined and this FFT is of the unfiltered signal. In a multi-channel communication system, if the power of a modulated carrier is limited to the carrier bandwidth then it becomes important for many reasons.

i. When signal having more concentrated frequency range the transmission power get reduced.
ii. Restraining a channel to a exact frequency band leads to the removal of adjacent channel interference due to other channels.

By using a pulse-shaping filter to modulated signal, high-pitched changes give smooth output to very large extent and this signal is restricted up-to a precise frequency band. Let us take the time-domain modulated signal.

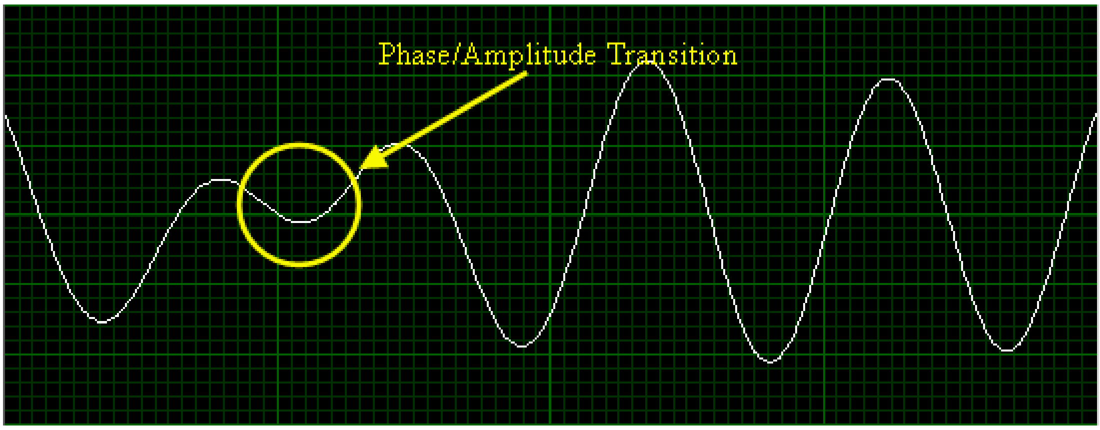

Figure 1.5: Smooth out Phase in addition to Amplitude Conversions in a Filtered Modulated Signal

The phase and amplitude conversions occur frequently after filtering is applied. Due to this the frequency data of the signal turn out to be extra focused up-to a definite frequency band.

Sharp transitions in the signal develop high frequency components in frequency field. If filter has been used in carrier signal, then these high frequency components of signal can be detached. So most of the channel power is currently restricted by an exact definite bandwidth. The mandatory bandwidth for a channel having direct relation with symbol rate in addition is arranged next to the carrier frequency.[31][34]

1.3.3 Decreasing Inter-Symbol Interference (ISI)

In band restricted channels, inter symbol interference (ISI) arise due to multi-path fading because signals are communicated above extended spaces in addition to different methods. Due to this some of the symbols spread outside their specified or specified time interval. These symbols can obstruct with the subsequent or previous transmitted symbols. This problem can be solved by using the presentation of the pulse shaping filter. Through these filter take every symbol that has been produced, we can decrease channel bandwidth though decreasing ISI.

In accumulation, match filter on receiver side are used in the direction of reducing these distresses. Therefore, ISI is decreased via providing pseudo-guard interval that diminishes signals from multi-path replications.[29][34]

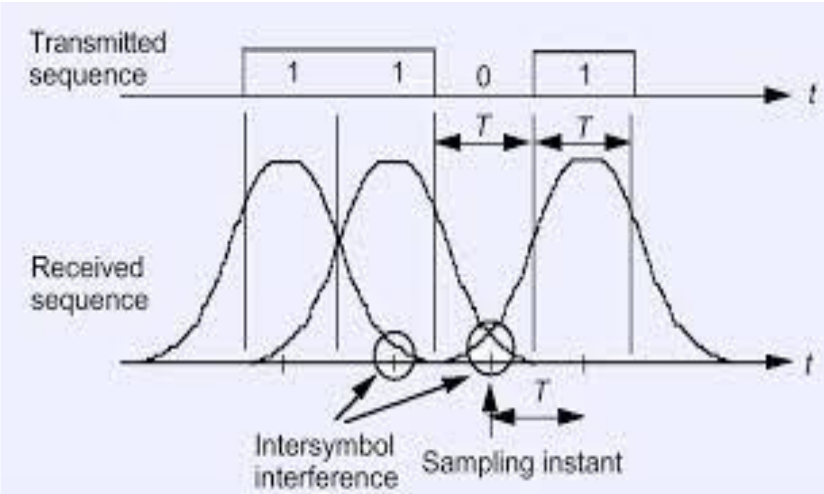

Fig 1.6:Effect of ISI

1.3.4 Direct Sequence Spread Spectrum:-

Spread signal can be alienated into two categories: direct sequence spread spectrum and frequency hopping spread spectrum. Sometimes it becomes difficult to detect and demodulate the communication signal even if they are detected. The detection of signal simply means to find out the presence of the signal or to discover the presence of the signal. In this case it is required that the signal should have low probability of intercept.

In the spread spectrum thousands of the carriers are used to produce the DSBSC signals which are attained after the similar message signal. The carriers are spread above the wide bandwidth and in the similar way the resulting DSBSC signal is spread over similar bandwidth. If the power that has been transmitted is same as that of the single DSBSC then individual control of the DSBSC in spread spectrum is thousand periods fewer. To get message from the transmitted signal the receiver only requires thousands of local carriers at the similar phase as well as frequency, as those at the transmitter side. These carriers are known by the pseudo random binary sequence producer.

The pseudo random binary producer is a generator that generates the unique binary sequence for each transmitted signal and this code is different for each transmitted signal. This code is generated using the ex-or operation. The first bit and the last bit is ex-ored and then a sequence is generated.

The spectrum of a pseudo random binary sequence producer is found to be the good source of carriers. The key of the successful message recovery is awareness of PN sequence at transmitter side.

A DSSS generator requires:

- A modulated signal in the RF band
- A PN sequence to spread it

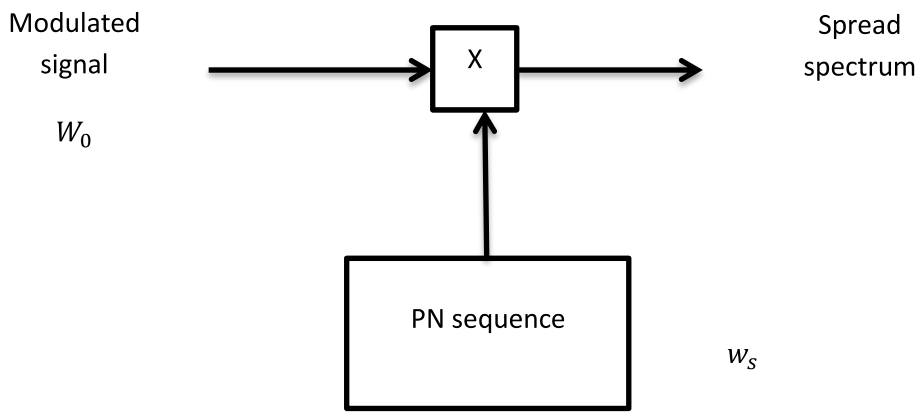

Fig.1.7 DSSS model

The two bandwidths are involved here: modulated signal and spreading sequence. The output spread spectrum signal will be spread at the side of original RF transporter through a quantity equal to PN sequences bandwidth. Here w_s is arrangement clock. As length of the sequence increases the existence of more spectral components will be there. The necessary condition is that w_0 >> w_s. The modulated signal may be of any form but mostly it is in binary form. [30][31][32]

CHAPTER-2

LITERATURE REVIEW

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1) In this paper the BPSK demodulation is described. As for the communication between base station and the satellite, analog communication systems were used. But it was observed that using analog systems for satellite communication is a disadvantage. The analog systems for communication occupy large area on satellite and this in turn consumes more fuel. The digital systems are used in place of analog systems due to their advantages, as they are more compatible, flexible, easy to manipulate, less expensive, transmission without degradation and integral network. Different digital modulation techniques are used like ASK, PSK, FSK, MSK, QPSK. But mostly BPSK is preferred the reason behind is that communication between base station and satellite is small the commands are used in which the data rate is around 4Kbps. PSK is used in military as well as in commercial communication system, as for the telemetry applications, PSK is important form of data modulation because it gives the less probability of error for the received signal level when it is measured over one symbol. In analog systems it becomes difficult to predict that how each and every component will behave when they are placed into the space orbits. BPSK demodulation is done using costas loop and the costas loop requires a PLL. PLL consists of phase detector, LPF and a VCO. To lock the low frequency disturbances that occur in satellite PLL is important. The BPSK shift the phase of the sinusoidal carrier 0 or 180 degree with the uni-polar binary signal. This is also referred to as PM signaling with a digital waveform and is equals to modulating a DSB-SC signal with a polar digital waveform. If the phase is shifted by 180 degrees then its equivalent to change the sin wave or multiplying it by -1. The performance of various schemes takes into consideration the bandwidth, filtering techniques that are used to create the band-pass signal. To improve the bandwidth of various schemes the multilevel PSK technique is used. [1]

2) In this paper the modulation techniques are discussed and BPSK systems performance is compared with respect to Bit Error Rate. As per the requirement of the communication system the main goal is to get the transmitted data accurately at the receiver end. BPSK modulation technique is used in many areas one of the main reasons to use this is due to its simple design as compared to other modulation techniques and its low BER. It is observed that there are transitions from analog to digital communication are happening. The reason to use digital communication system is that it is extra consistent than analog technique. The modulation techniques used in analog and digital communication are different. Baseband signal is converted into band pass technique by the modulation process. In the digital communication the binary data is modulated by means of an analog carrier but in the analog communication an analog is occupied and modulated with an analog carrier. The modulation is done by the modulator and demodulation is done by the demodulator. In BPSK, the carriers phase is modified according to the message signals input that is given. If the demodulator has a reference signal to equate the received signal phase then the modulated signal's phase conveys some information. This is referred to the coherent detection. BPSK uses two phases that is 0 degree and 180 degree. The BER shows that how many bits are in fault when a fixed amount of bits are transferred. The signal that is to be modulated is passed over the channel, in that channel noise is more. The receiver contains matched filter the received signal is reproduced by the carrier which is used in transmitter side.[2]

3) In this paper the techniques to calculate the BER is discussed. BER can also be expressed as the detection techniques performance and it can be calculated by three options. In the first method mathematical formulation of the processes is done. In the second method the simulation is carried out on the computer and the third most method contains the action of measuring BER. Code multiplexing is the principle on which CDMA work, its advance version is W-CDMA. In CDMA multiple users are allowed to share the common channel that is why it is used in multiple communication systems. It has applications in mobile radio, wireless LAN system and satellite. It has a lot of features like asynchronous access, graceful degradation, efficient bandwidth utilization and the main thing is that it provides the secure communication between the users. In case of DS-CDMA the each user is allocated a signature sequence for spreading and documentation process. The main function of the signature codes is to show cross correlation and autocorrelation properties. The conventional CDMA's correlation receiver's performance deteriorates with increased

number of users. This is due to the multiuser interference. In this paper from the simple model of BPSK the CDMA multiuser detection process in MATLAB environment is described.[3]

4) In this paper the DSSS is implemented on Matlab-simulink, one of the main reason to use spread spectrum is it is easy to use and implement and its simplicity, in it the modulation of carrier is done by the code sequence. The performance was tested by simulating the design to achieve the desired received data or information, the data received is compared with the transmitted data and from this the effect of additive white gaussian noise and bit error rate was calculated. In DSS to modulate the carrier an encoding signal is used. DSSS gets its spreading capability by modulating a data signal of narrow bandwidth with the signal of wide bandwidth. When the code comes out to be 1(one) the carrier is transmitted and when code comes out to be 0(zero) a 180 degree phase shift will be there. When the signals get amplified the received signal is multiplied by the reference with same code and we suppose that the transmitter and receiver's code that are generated are synchronous and received signal can be represented by the multiplication of data sequence and PN spreading sequence and carrier frequency, carrier phase angle, data symbol duration.[4]

5) In this paper the main goal is to design trans-multiplexers of TDM-FDM digital multiplexers and similar FDM-TDM digital de-multiplexers. These are having applications in terrestrial communication to satellite and broadband wireless communication in addition it is the replacement for OFDM systems. In this paper basically the use of SSB and VSB is considered. The proposal of VSB has been lead to a well-organized use of spectrum during balancing the orthogonality of channel and this can be considered as the important condition or property in digital filter banks for multiband wireless communication. When the poly phase filters were introduced which shortens clock, control signal generation, error discovery and reduce the no. of arithmetic operations. The factors used in relating the presentation of trans-multiplexer approaches are:-
   a) control in addition to computational complexity
   b) intelligible crosstalk

c) equivalent frequency alteration
d) obtainable band signaling and pilots
e) modularity
f) absolute value of group delay

The perfect reconstruction of the filter bank is important because due to this the whole bandwidth of the channel can be used in an efficient way also the crosstalk can be irrecoverable however the magnitude and frequency response overlay. In trans-multiplexers input-output associations are imitative in terms of the input signal in addition to synthesis and separation, filter transfer functions, together with special effects of up-sampling and down sampling. Impulse replies of the synthesis and analysis filters are given through the center frequency, phase factor as well as delay. Central frequency is considered to be similarly spread out as well as lie between 0 and pi. Two kinds of trans-multiplexers are assumed, in one sort the center frequency are different, which lead to the VSB design. In other kind, prominent to QAM strategies, two signals in quadrature are directed at every center frequencies, expect for those at 0 and pi. Frequency responses of sets of filters are not band limited as well as in every situation the whole frequency range 0 to pi is covered with spectral intersection between filters by means of adjacent center frequencies but orthogonality of the signal transmitted in every FDM channels result in input-output transfer functions for every pair of equivalent terminuses being similar and the cross talk being abolished. Set 1 contains 0 and pi and sets of two frequencies which stay identical and integer multiples of 2π /N among 0 and π. [5]

6) In this paper requirement of multi-media communication is discussed. Sampling is the basic need of any digital signal processing application. The video and audio signals are sampled at different sampling rates. The sampling rate conversions are mostly completed by transmitting the multimedia signal via digital to analog converter and then output analog signal is re-sampled by requisite rate. Due to this re-sampling process familiarizes misrepresentation to signal because of the quantization effects which are in-built in the analog to digital transformation procedure. So in order to overwhelmed the restrictions of analog to digital conversion method single-rate digital filters are taken. These filters accomplish the sampling rate conversion process. These filters use many filtering taps due to this their processing time is slow. Multi rate filters were industrialized to provide the low sampling rate causing less sifting knocks. These filters are working in systems

which require real-time presentation. This paper basically describes the parallel and high speed, single-chip implementation of essential multi rate filter banks. Hardware application platform is founded on Virtex field programmable gate arrays (FPGAs). Field programmable gate array (FPGA) can be defined as an integrated circuit which comprises of many matching logic cells that are interrelated through matrix of wires and programmable switch. Analysis and synthesis filter banks are multi rate filter banks.[6]

7) In this paper the chaos-based DS-CDMA systems are used. Chaotic signals provide sturdiness in multipath environment in addition provide resistance to congestion. The chaotic signals are non-periodic in nature besides they are more hard to forecast and reconstruct, wide band. So, to spread the information over them is difficult to decode and intercept. RAKE receiver is used on behalf of conventional DS-CDMA system below multipath channel. This receiver is having bank of programmable co-relators which correlate every L received replicas of same transmitted signal by the consistent locally produced code. The code that is used is useful in spreading and dispreading the data signals. The replicas are multiplied by matching estimated complex valued conjugate path gain after process of dispreading. Chaotic sequence of each user is produced by means of similar chaotic generator by means of dissimilar original condition. Probability distribution function of chaotic energy is delivered. At receiver side the channel coefficients and delay estimation is recognized. Prefect synchronization is expected in this case. Multipath propagation, correlation properties of spreading codes, and RAKE receiver are considered to derive BER expression, that is focused on asynchronous environment.[7]

8) Amount at which mistakes happen in a transmission system is defined as the bit error rate. That simply means that mistakes that will happen in a sequence of number of bits. The BER can be formulated as: BER is defined as ratio of number of errors to total number of bits sent. If the SNR that is signal to noise ratio is high and the medium among the transmitter in addition to receiver is good then bit error rate will be very lesser in addition to the consequence of bit error rate on the overall system will not have the noticeable effect. But if noise has to be considered then the bit error rate will need to be measured. When we send the information or data over the data link, then errors will introduce into the system and due to the errors in the data the reliability of the system may be

negotiated. The interference level that is present in the system is established by the exterior issues and these can't be transformed by system design as they are external factors. The bandwidth of the system can be set and the interference level in the system can be reduced by decreasing the bandwidth. With reduced bandwidth restrictions the throughput of data can be accomplished. The power of the system can also be increased and by increasing the power level the power bit is increased. The power of the system has to be well-adjusted in contradiction of the interfering stages to former operators and effect of growing power output on extent of power amplifier in addition to complete power consumption and battery life. Inferior order modulation techniques could be utilized, but this is at outlay of information amount. It's essential to stabilize all existing features to get required bit error rate. Bit error rate BER is a constraint that provides a suggestion of presentation of an information link such as radio or fiber optic system. Major parameter in some data link is amount of faults which arise, the bit error rate is a main constraint. If BER increases too high then the system presentation will destroy. It should be in boundaries only then the system will function acceptably.[8]

9) Multi rate filter banks are the filter banks that take a single input signal and by performing filtering and subsampling of an input signal produce the multiple output signals and can also generate the single output by performing the up-sampling and interpolating of the multiple output signals. They are having applications in speech processing and trans-multiplexers for telecommunication. The filter banks are devices which generate single signals by using filtering of M simultaneous filter and this single signal is used to produce M signals. These M signals are subsampled by factor N. It has been found that there is no requirement of using the sampling theorem on channel by channel basis in multi rate filter bank. Finally, a new theory can be applied which checks all the channels in-spite of taking into consideration each one individually. By using QMF filers various things come into consideration that there is relation between trans-multiplexers and sub band coding and the results on complexity of trans-multiplexers was the introduction of Pseudo-QMF filters. The results on aliasing cancellation could be used to cancel the crosstalk in trans-multiplexers. The quadrature mirror filters allows the reconstruction of original signal. The multi-rate filter banks use three basic operations that are linear filtering, up-sampling and subsampling. The multi-rate filter banks comes in the class of linear periodically time varying systems as they consists of time-varying operations and linear filters. The filter

banks which come into consideration are analysis filter bank and synthesis filter banks. The analysis filter banks divide the input signal into M filtered and sub-sampled versions. The synthesis filter banks on the other side generate a single signal from the M up-sampled signals.[9]

10) Multi-rate filters operate on two basic operations, they are decimation and interpolation means increasing or decreasing the sample rate of a signal. Multi-rate filters are used at interface of continuous and sampled data which provides reduction in the cost of the analog signal. The multi-rate systems are used for audio and video dispensation, communication systems and transform analysis. Multi-rate systems having lot of applications which comprise translation of different rate input information to static rate output information in modulator in addition to converting stable rate input data to different rate output information in demodulator, additional presentation includes sampling rate changes of the input signal being treated so that the filtering of the input signal can be accomplished at the Nyquist rate of input signal that is actuality administered. In one presentation, the multi-rate filter is used for level up sampling rate of a sampled data signal preceding to its distribution intended for handing out via digital to analog converter convoluted in transmitting signal among sampled data world and continuous world. In additional main presentation, multi-rate filter is utilized to level down sample rate of a sampled data signal afterward being designed at output of an analog to digital converter convoluted in conveying the signal among continuous world in addition to the sampled data world. [10]

11) Due to increasing demand of mobile systems the use of these devices also increases and this result in the demand for their battery lifetime, the main goal is to make the power efficient. To achieve the battery lifetime for end users, it’s required to use software methods with hardware techniques. Here new system level method intended for the process development is used that practices data regarding process device procedure to make arrangement decisions. The device usage data is associated to a process by means of the device window. This window is used to use the information to schedule processes in a manner so that they

    i. Will increase the duration of idle periods
    ii. Will increase the burstiness of device access.

By assigning the power firmly to procedures and by creating the power a first supply the lifetime of a system can be increased. The throughput or deadlines are not guaranted by increasing the lifetime of batteries. The power can be saved by putting the device to sleep to save the amount of power. The purpose of using low power channels separately is that to determine when to device should be turn off.

$$E_{to} = (P_{idle} - P_{sleep}) * t_{to}$$

Where,

Pidle = average power consumed when system is considered to be idle

Psleep = it’s the sleep power of the device

tto = timeout period

The basic indication of process arrangement created on device window is to vigorously change data about a procedure design of device use based on its earlier performance. To approximate the length of device usage arrangements the device usage information composed at the runtime is used. Process schedule are changed to superior allocated device idle phases depending upon the device usage pattern. To enter into the sleep mode system uses timeouts which decides at what time or whether a device must arrive in the sleep mode. If the timeout decreases then this tends to the increase in energy due to expenditures of frequency switching device states.

12) The multi-rate filter banks provide the high spectral efficiency by elimination of terminated information resulting in the resourceful use of presented spectrum. Multicarrier modulation system subdivides the channel bandwidth in sub-channels. Filter bank based multicarrier system (FBMC) are mostly used in current communication system i.e., cognitive radio due to its essential frequency selectivity and spectral efficiency. It also has improved frequency selectivity and efficient use of available spectral BW.

TMUX is divided into two parts:-

i. Synthesis filter bank (at the transmitter side)
ii. Analysis filter bank (at the receiver side)

These are tuned accordingly to get the desired response from filters.

At the transmitter side, multiplexing is performed by interpolating and filtering. Multiplexing is a method due to which various analog and digital streams of transmission could be treated above shared link. Multiplexing splits high capacity medium addicted to low capacity logical medium which is united by dissimilar streams. When additional sender try to direct information then single medium multiplexer is used and multiplexer splits the physical channel and assigns one to every channel.

At receiver side, composite signal is delivered through structure of separation filter bank and decimators to yield the output signal.

TMUX allow several signals to be transmitted through a single channel e.g. satellite, OBP, switching.

Satellite provides global coverage and if satellite is in orbit the customer required to connect a satellite terminal and subscribe to server.

FMUX and De-Multiplexing

In it digital part of satellite OBP is MIMO system. The amount of input signals differs from no of output and input signal can have different BW and data rates. The main role of OBP is to change entire sub bands to different pre specified output signals and position them in the frequency spectrum. System require to support BW on request i.e., BW of different sub bands may vary with time.

13) In this paper the basic focus is upon Multi-rate systems and filter banks, they play an important role in the source coding and compression for contemporary communication applications. Communications problem are separated among two portions – source coding and channel coding.

Source coding is basically related with developing redundant free information that means representing the information that is redundant free maximally. In communication system distortion is always there, as no system is distortion less, so some degree of distortion is allowed in the representation.

Channel coding also follows source coding and it is developed to reintroduce the information with prescribed level of redundancy back into the source coding stream. The performance of the communication system increases when the redundancy is introduced to some specific characteristics of the channel.

In the communication system multiple user communicate through wireless channels and because of this fading occurs due to time-varying multipath communication. Fading can be described as variation of signal strength at the input of the receiver. To compensate for the effect of fading CDMA is used.[13]

14) In this paper Date Rate Adaptive Channel Algorithm is used because in multi-rate wireless networks high data rate nodes take or consumes less channel resources but the less data rate nodes consumes additional channel resources, this result in the overall low network performance. Data Rate Adaptive Channel Algorithm provides high network efficiency and high throughput for multichannel multi-rate single hop wireless networks. The wireless networks use IEEE802.11b and IEEE802.11a these offer multiple link data rate compatibility at physical layer and this is done by several modulation techniques. In IEEE802.11 MAC standard all the links works at the same data rate in wireless network. The surviving MAC standard gives only throughout in wireless networks. In Data Rate Adaptive Channel Algorithm the main knowledge is to allocate the links that are having the comparable or same data rates on similar channel to yield the benefit of high data rate links than low data rate links. This will reduce wastage of channel resources and this is only possible because of the interference among low data rate links and high data rate links. Intended for this purpose different layer is introduced among the network layer and link layer and this layer is named as Intermediary Multichannel Layer (IML). In IML channel assignment is accomplished. Because of IML no modifications need to be done at the MAC layer and the network layer stack. Each wireless node in a multichannel multi-rate wireless network is having multiple network interface cards (NICs) and these can be transmitted or received on non-overlapping frequencies that are provided by IEEE802.11 standards. In multichannel wireless network IEEE standards are having active and passive links and these links provide the node connectivity. The whole throughput reduces as transmission starts on low data rate links and individual throughput of each link becomes similar. With the help of multiple channels different network topologies of links different networks can be formed and these communicate on dissimilar channels so depletion of channel resources is not there because of low data rate links. To provide security to the network layer and MAC layer the IML achieves the numerous NICs as well as numerous non-overlapping channels by means of DR-CA algorithm. At the intermediary multichannel layer there is no overhead for the data packets in terms of the queuing delay and processing. The data packets are received from network layer and queues it in

interface on the beginning of neighbor to interface table and this table is evaluated by channel assignment algorithm that is applied at the IML. The IML layer exchanges the control packets that are compulsory by rate adaptive channel assignment algorithm.[14]

15) In this paper to reduce the energy consumption it is considered that nodes operate on batteries. As the advancement in technology is doing progress day by day it is supposed that very quickly it will be probable to incorporate the RF transceiver, A/D and D/A converters, baseband processors on a single chip which would be as soon as possible. To reduce the total energy ingestion for energy-constrained wireless network link layer, MAC layer should be jointly designed. The battery energy is finite and node can individually transmit a finite no of bits. The total battery energy separated by compulsory energy per bit is called as maximum no of bits that can be sent. By increasing transmission time for protected packets transmission energy can be reduced. There are three types of multi-modes that are explained in this paper Active mode(when the signal is provided for all the circuits), sleep mode(when no signal is provided in all the circuits), transient mode(when the power is switched on and the circuits goes into the active mode). The batteries that are used are peak-power limited i.e., the maximum available battery power always remain more and the total power consumption either transmitter or receiver not ever go beyond maximum accessible battery power. Circuit can only be turned on when the frequency synthesizer is turned on first and they all are settled down. The main condition to be satisfied is that the leaking power consumption should be less than power consumption in active mode.[15]

16) In this paper basically the performance of BER is evaluated for DSSS. DSSS is direct sequence spread spectrum. The BER performance can’t be obtained in presence of noise and ISI. BER of the system can be minimized using the Rate Transition Techniques. Here it is assumed that the receiver is having the information of the user’s spreading code and is synchronized by means of the amplified matched filter gain. One approx. is to use a S/N in Gaussian approx. which is the standard one. Other approx. check the applicability of central limit theorem. The basic idea here is to calculate the PDF of chaotic bit energy and integrate the BER above whole possible standards of PDF. Chaotic system is non-linear sequence generator which is non-periodic and depends on the initial conditions

assumed for its stochastic quantification. Due to the non-periodicity its autocorrelation doesn't take its peak at t=0.

Central limit theorem says that regardless of dissemination of a population(with mean and variance),if N number of samples are taken from the population then the sample mean follow a normal distribution with a mean of μ besides standard distribution of σ/sqrt(N). As the n increases the normality becomes enhanced. Irrespective of base distribution the probability distribution curve will approach Gaussian or normal distribution as the number of samples grows. CLT states that the sum of self-governing and identically disseminated random variables approaches Normal distribution as (sample size) N->∞.[16]

17) In this paper the discussion is done on the interpolation and decimation, they come in the category of multi-rate filters. The change of sample rate of sampled data signal can be required because of a lot of reasons. As the use of multi-rate filters lead to the cost reduction of analog signal and also the improvement of signal quality. Interpolation is the inverse of decimation it is quite opposite to decimation process. Interpolation can also be termed as the information preserving process where all the x[n] samples are present in the extended signal y[n]. Expansion process is done along with a unique digital low pass filter which is termed as an anti-aliasing filter. Aliasing is not caused by the process of expansion in the interpolated signal. But this process yields undesirable replicas in the frequency spectrum of the signal. Decimation is considered as the discrete time equivalent of the sampling. In the sampling process our main goal is to convert a continuous time signal x(t) into arrangement of samples x[n]. But in the case of decimation process we begin with the discrete time signal x[n] and change it into alternative discrete time signal y[n] and this discrete time signal contains the subsamples of x[n]. An anti-aliasing filter is used before the decimator or down-sampler to prevent aliasing that can occur due to the lower sampling rate.

The applications of multi-rate filters involves alteration of adaptable rate input data to static rate output data in a modulator and reverse is done in case of demodulator i.e., changing static rate input data to adjustable rate output data. Other application also takes the sample rate changes due to which filtering process could be done at Nyquist rate of signal that is to be processed. Multi-rate filter is used to rise sample rate of a sampled data signal preceding to its transmission for treating by digital to analog converter involving transfer of signal between sampled data world and continuous world. Multi-rate filter is used for reduction of sample rate of a sampled data signal afterward being generated at output of an analog to digital converter which takes into account transporting the signal among the continuous world and the sampled data world. [17][30]

18) In this explanation of digital filters is discussed. The general purposes of using digital filters are, they can be used for the separation of those signals that are combined and also to restore the signal that is distorted. This shows that they are having advantages of signal separation and signal restoration. Whenever a signal has been affected by noise and interference or by other signals then signal separation is needed. But when a signal gets distorted then signal restoration is used. Their applications can be seen in EKG (electrocardiogram), de-blurring of an image, shaky camera. The performance of digital filters is much better than the anlaog filters. Digital filters show thousands times improved presentation than the analog filters. Filter's input and output signals are in time domain in digital signal processing. This is due to the reason that when signals are created then sampling is done at regular intervals of time. Sampling can also be done at equal intervals in space. Every linear filter comprises of a step response, impulse response and frequency response. All these replies consist of data about filter but in dissimilar form. These replies describe that how filters react under the different situations. The digital filters are implemented by convolving the input signal with digital filter's impulse response. The information could be characterized in time domain as well in frequency domain. The information that is represented in time domain explains at what time anything happens and it shows that what is the amplitude of the existence. The indirect way of representing the information can be seen in the frequency domain. Step response represents that how information can be characterised in time domain can be changed by the system. The frequency response describes that the data in the frequency domain can be modified. [18]

19) In this paper the role of DSP in communication system is discussed. As per to get the signal that has been transmitted with the higher speed and the requirement of higher bandwidth many new technologies have been introduced. The main aim of the communication system to make efficient use of limited bandwidth and deal with the difficult channel atmosphere. The multi-rate dispensation techniques are used in image

processing, digital audio and in multi-media system. These systems can reduce the computational density. Multi-rate systems are used in digital signal processing because they can change the motion or speed of the discrete-time signals and this can be known by adding or removing the signal. Multi-rate also used to merge the pulse shaping in interpolation method. Cyclic prefix is used sometimes at the receiver side to get the signal back that is transmitted. The signal is converted in digital for processing. These conversions are made to get the discretization in the amplitude of the signal but in the practice the amplitude can be real or composite number. The multi-rate DSP systems are designed by three basic blocks that are linear time invariant strain, decimator and expander. At the output of an expander the rate of indication is M times the rate at its contribution. Poly segment representation of signal is mostly used in the multi-rate indication processing. For the wireless systems the programmable and reconfigurable DSP system are implemented. [19][29][33]

CHAPTER 3
PRESENT WORK

3.1 Scope and Objectives of the study

Multi-rate signal processing technique is mostly used in many parts such as communications, image processing, digital audio and multimedia. The key benefit of a multi-rate system is to reduce the computational complication and due to this the cost reduces. The computational effectiveness of multi-rate algorithms is based on the capability to use different sampling rates simultaneously at the dissimilar part of the system. Sampling rate alterations generate the unwanted effects through the system: spectral aliasing in the sampling rate decrease and spectral images in the sampling rate increase. The main role of multi-rate filtering is to permit the sampling rate alteration of the digital signal deprived of abolishing the signal components. The multi-rate filtering makes the general perception of multi-rate signal processing applicable in practice. The basic multi-rate signal processing building blocks is the important to many signal compression and communication applications such as-

- Digital Audio Systems
- Sub-band Coding
- Analog Voice Privacy Systems
- Speech coding
- Image compression
- Adaptive equalization
- Echo cancellation
- Adaptive beam forming

Central objective of this study is to make the communication more secure and reliable as the demand for communication increasing day by day so in order to provide the users required bandwidth some steps should be taken. As the filter banks are considered to be an array of band pass filters which separate data i.e., input signal into the multiple components and these multiple components carry a single frequency sub band of original signal. The filter banks perform various operations like the process of disintegration achieved by filter banks known to be analysis and the reconstruction process means the reconstitution of a complete signal resulting from the process of filtering is called synthesis. When we send data or the information signal over some frequency and some of the frequencies are important than others then filter banks plays a vital role in signal compression. The important frequencies can be coded after decomposition. [22][24]

3.2 Research Methodology

The key perseverance of this report is to make the communication system more efficient and secure. To achieve this multi-rate signals are used because they provide an efficient way for communication by reducing cost. Multi-rate signals can also be used in communication system to get the better sampling rate and these having advantages in DSP also, so it is good to use this technique in the communication. It's a very simple and straight forward method of varying the sampling rate of a digital signal. Here reconstruction of continuous-time signal from original set of samples as well as resample signal at different rate is done by presumptuous that no additional anti-aliasing filtering is compulsory. Ideal actions are mandatory to recreate the continuous- time signals from the original samples but because of restrictions of practical ADC and DAC devices, the resulting signal will surely have noise and distortion in it. Due to this decimation and interpolation are used. In the decimation our goal is to decrease the sampling rate by factor of M, where M is an integer. In the interpolation our aim is to maximize sampling rate by an amount of L, where L is an integer.
In multi-rate digital signal processing sampling rate is altered in order to increase or decrease the effectiveness of signal. Decimation or down sampling decreases the sampling rate on the other side up sampling increases the sampling rate.

Sampling can be defined as a method of dividing a continuous signal into discrete signal. Sampling means that we can take the series of samples of a continuous varying signal and utilize these values of samples to describe the whole signal without any loss of the available information (data, signal). These samples can be used to reconstruct the original signal. As, sampling theorem states that we can gather whole of the information in a signal by performing sampling at a rate 2B and B can be defined as the bandwidth. By this, we can find out or reconstruct the original shape of the continuous signal at any instant in between the sampled instants. Reconstruction of a signal is a real or true reconstruction it's not the estimate of the original sequence of signal.[27][28]

3.2.1 Decimation

Decimation is treated as the discrete time complement of the sampling. In the sampling process our main goal is to convert a continuous time signal x(t) into arrangement of samples x[n]. But in the case of decimation process we begin through the discrete time signal x[n] and change it into an additional discrete time signal y[n] and this discrete time signal contains the subsamples of x[n]. An anti-aliasing filter is used before the decimator or down-sampler to prevent aliasing that can occur due to the lower sampling rate.

The decrement in the sampling rate by an amount of M is attained through removal of each M-1 samples or consistently maintaining each M^{th}sample. When removing M-1 of each M input samples decreases original sample rate by an amount of M, it similarly makes input frequencies overhead one-half decimated sample rate to be aliased into the frequency band from DC to the decimated Nyquist frequency. On the way to diminish this result, input signal need to be low-pass filtered to take away frequency components from portions of output spectrum that are essential to be alias free in succeeding signal processing phases. A advantage of the decimation method is that the low-pass filter may be considered to function at the decimated sample rate, rather than the nearer input sample rate by using an FIR filter arrangement, as well as by observing that output samples linked with M-1 unwanted sample need not to be calculated.

Let x(m) is input signal, h(k), 0 <= k <= K be constants of an assumed low-pass filter and z(m) be output signal before decimating by an amount of M, then:

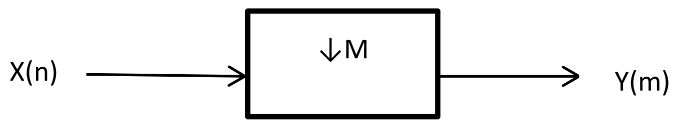
$$Z(m) = \sum_{k=0}^{K} h(k)x(m-k)$$

Currently, if we assume that output signal afterward decimator is y(r) = z(rM) where sampling rate is decreased with an amount of M. Then, y(r) = z(rM) if e output signal is decimated with an amount of M.

$$y(r) = \sum_{k=0}^{K} h(k)x(rM-k)$$

This equation shows that the filter is in consequence by means of the down-sampled signal. Consequently processes of down-sampling and low-pass filter have been added in such a way that low-pass filter function at concentrated data rate and average number of calculations to produce one output sample is concentrated by factor M. Every output sample needs K multiply/accumulate cycles but only 1 out of M samples required to be considered. If down-sampling of a signal is functional afterward the anti-aliasing low-pass filter, the amount of multiply is K*M for every Min-put samples. By adding down-sampling into low-pass filter, number of multiply cycles are decreased to K for each M input samples. Average number of multiply each input sample is K/M.

Let us consider here, v(n) is the output of low pass filter.

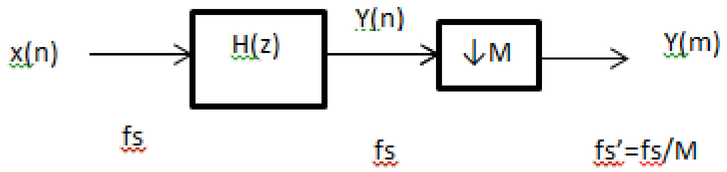


3.1 Decimation by factor of M

Output of desired signal is:

$$y(m)=v(Mm)$$

The decimated signal y does not contain all information about the original signal x. thus the decimation is applied in filter bank and preceded by filters which extract the relevant frequency bands. In order to analyze the frequency domain characteristics of a multi-rate processing system with decimation, we need to study the relation between Fourier transform or z-transform of signals x and y.

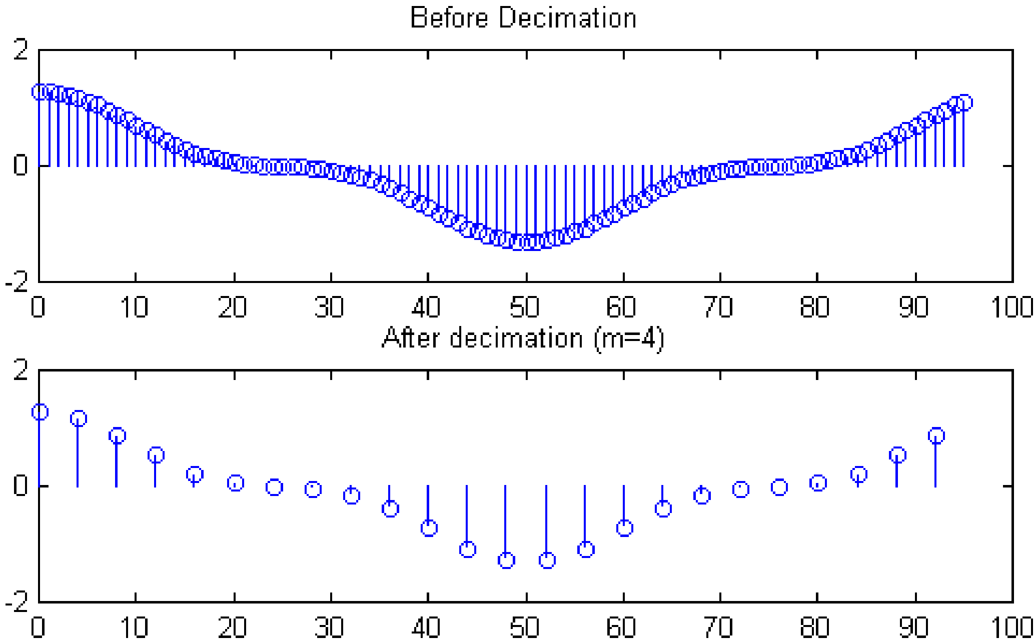


3.2 Diagram of decimator

We require low pass filter H(z) because this will filter the input signal x(n) in such a way thus the energy directly overhead frequency that is w=π/M can be neglected. This low pass filter acts as an anti- aliasing filter. If frequency response of this filter becomes equals to or approximate the ideal response of filter then

$$Y(e^{jw})=\frac{1}{M}X(e^{jw/M}) , |w|<\pi$$

If we take the example of decimator and decimate the sample or signal by 4 then the signal will be:-



3.3 Signal after or before decimation

3.2.2 <u>Interpolation</u>

Interpolation is the inverse of decimation, the situation is quite opposite to decimation process. Interpolation can also be termed as the information preserving process where all the x[n] samples are exist in the expanded signal y[n]. The expansion process is done along with a unique digital low pass filter which is termed as an anti-aliasing filter. Aliasing is not caused by the process of expansion in the interpolated signal. But this process yields undesirable replicas in the frequency spectrum of the signal.

An escalation in sample rate (interpolation) by an amount of L is gained through adding L-1 consistently spaced, zero value samples among every input sample. During the insertion of L-1 new samples among every input sample escalates sample rate by an amount of L, it also familiarises output of the input signal into interpolated output signal at frequencies among original Nyquist frequency and the higher interpolated Nyquist frequency. To remove consequence, interpolated signal need to be low-pass filtered to neglect some output frequencies that interrupt succeeding signal processing stages. An advantage of interpolation procedure is that low-pass filter might be considered as function at input sample rate, rather than faster output sample rate by by means of FIR filter structure as well as by note that input related with L- 1 introduced values have zero value.

Let x(n) be original input signal and v(n) is sequence having L-1 zeros introduced, y(n) is output sequence of low-pass filter and let h(0), ..., h(k-1) be constants of low-pass filter, then:

$$y(n) = \sum_{k=0}^{K} h(k)v(n-k)$$

However, v(n-k) = 0 except n-k is a multiple of L, interpolation factor. Due to this L-1 zeros were added in sequence x(n) to acquire v(n). Let x(n) be input signals, and h(k) be filter constants. Then output signal y(r) can be represented as:

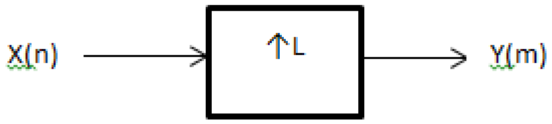
$$y(r) = \sum_{n=0}^{K/L} h(r-Ln)x(n)$$

On behalf of a particular input sample, L output samples are produced. If the anti-imaging filter is not added in the interpolation method, then number of multiply cycles for L output samples will be L*K. Though, by captivating benefit of the information that L- 1 zeros were added into output sequence, anti-imaging filter has merely K/L nonzero values. Average of multiply/accumulate cycles per output is K/L.

Let us deliberate a discrete-time signal x(n) and the sampling rate of this signal is increased via an amount of L which is an integer value. By this L-1 new samples can be interpolated among every pair of samples of input signal x(n) and this gives

u(m)={x(m/L), m=0,±L,±2L

u(m)=0, otherwise

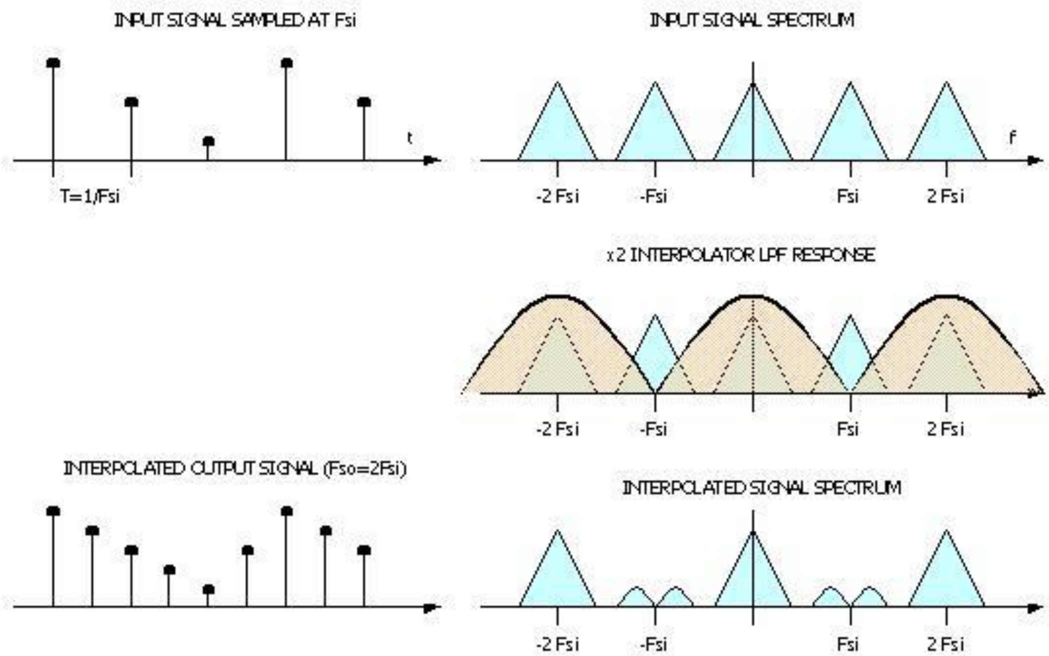


3.4 Expansion by factor of L

By the process of interpolation zero samples are inserted between input signal samples due to this unwanted spectral image components will be there. That is why filteration of signal u(m) with low pass filter is done to decrease the effect of unwanted components.

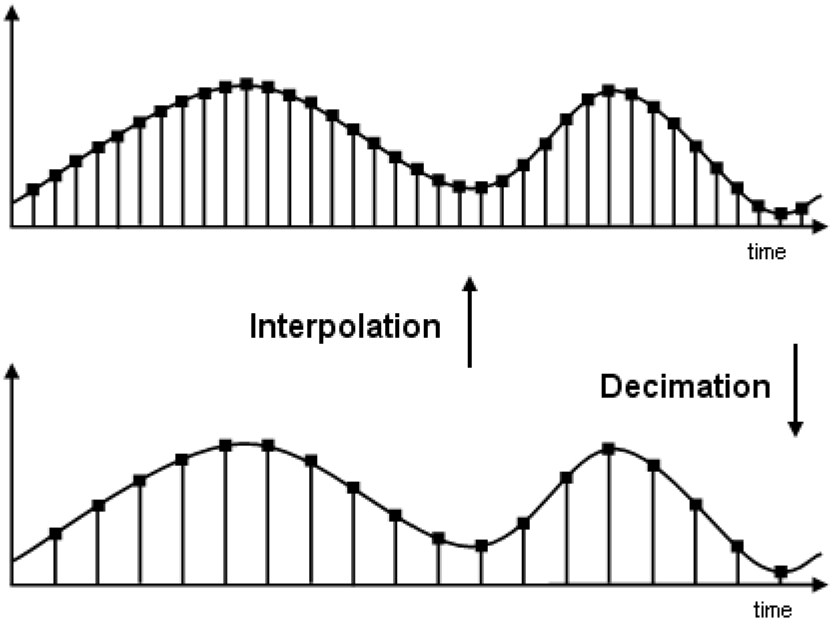
$$F(e^{jw})=L,\ |w|<\pi/L$$
$$F(e^{jw})=0, \text{otherwise}$$

The implementation of multi-rate signals need a software (Matlab) on which the signal is to be considered and the output can be easily detected. In multi-rate digital signal processing the sampling rate is transformed in order to increase or decrease the effectiveness of signal. Decimation or down sampling decreases the sampling rate on the other side up sampling increases the sampling rate.[10]



3.5 Interpolated signal

The basic difference between the interpolation and decimation is that in decimation the values of signal



3.6 Diagram to show basic difference between interpolation and decimation

Interpolation and decimation comes in the category of multi-rate filters. The change of sample rate of sampled data signal can be required because of a lot of reasons. As the use of multi-rate filters lead to the cost reduction of analog signal and also the improvement of signal quality. [22][23][28][29]

3.2.3 Advantages and disadvantages of Interpolation and decimation

Decimation:

- The sample rate of discrete time signal decreases.
- Storage of the information reduces as the sampling rate decreases.
- Computation requirement also reduces. [28]

Interpolation:

- The sampling rate of discrete-time signal increases.
- The fidelity of the signal also becomes more due to high sampling rate.
- Aliasing reduces. [28]

3.2.4 Chebyshev Filter:

Chebyshev filter is used in RF and many more applications. This filter gives us a steeper roll-off as compared to other filters. This additional roll-off arises due overhead of ripple and it makes it inappropriate for a number of applications. Chebyshev filter is commonly used in RF applications where ripple is not a big issue. The steep roll-off can be used to eliminate band specious emissions such as harmonics or else intermodulation.

3.2.4.1 Chebyshev Filter Basics

Chebyshev filter can be shortened as:

Roll-off: Chebyshev filter has a steep roll-off. This will influences its eventual roll-off earlier than other procedures of filter. It is utilized in RF applications in which a steep transition among pass-band and stop-band is needed to eliminate undesirable yields like intermodulation of harmonics.

Ripple: Chebyshev filter delivers a steep roll-off, which is at rate of ripple. The in-band ripple of one type of Chebyshev filter avoids this arrangement of filter is utilized in other applications in which flat in-band response is desirable.

Cut-off frequency: Cut-off frequency is a point where the response falls to -3 dB does not hold for Chebyshev filters in opinion of the way the filter rolls of faster than other types of filters.

Chebyshev filter name: Chebyshev filter name comes from the point that format and calculations for filter are founded on Chebyshev polynomials.

3.2.4.2 Chebyshev filter types

Chebyshev filter are of two types:

Chebyshev type I filter: it's the furthermost common Chebyshev filters. It's having the steepest roll-off but displays in-band ripple.

Chebyshev type II filter: The type 2 Chebyshev filter also called as the reverse Chebyshev. It is not as much used than the Type 1 filter because it does not roll off as fast, and involves additional components. Though, its benefit is that it has no ripple in the pass-band, but having equi-ripple in the stopband.

3.2.4.3 Applications:

- They are used in signal monitoring due to importance of phase.
- They can reduce the amplitude of signal below or above a particular frequency.
- They are optimized to provide a steeper roll-off.
- The magnitude response of chebyshev filter show ripples.
- They have sharper frequency cut-off.
- They provide better attenuation beyond pass band.[25][21]

3.2.5 Butterworth Filter:

Butterwoorth filter is a form of RF filter and its applications are seen in radio frequency. It has been seen that the response of butterworth filter is flat as compared to other with in its pass band and satisfactory roll-off.

The cut off frequency of the butterworth filter is defined as the point where the power drops to half which means the voltage drops to 71%. The minimum loss for butterworth filter occurs at 0Hz.

3.2.5.1 Applications

- These filters are used in high quality audio applications due to their flat response in pass band and stop band.
- To filter the noise from a system low pass filter is used. As noise is high frequency signal but when it is passed by a low pass filter most of the noise is rejected and a clear sound is created.
- The magnitude response decreases with increase in frequency.
- They have a wide transition band.
- They has better pulse response.[23][21]

3.2.5 BER Testing:-

When we use the pseudo-random data source it is compulsory to simulate the transmission path when we use. BER testing can be done with the transmitter and receiver but the condition is that they should be close to each other. It is important to set up a channel that can represent the definite path of the data transmission that is to be used to simulate the transmission path. In the radio transmission, noise and propagation fading are included.

Noise: Noise can be present in the system from many sources in the radio path. Noise can be produced externally to the electronics system itself and it can also be the received noise, or it may be produced internally. The receiver noise represents that whether the system is in the real environment or in the simulated environment. In the receiver side the rest of the noise can be replicated and presented by means of a noise diode generator.

Fading characteristics for radio communications systems: For the real life features of the transmission path in the accurate way is of very importance. With signals which are constantly changing as a consequence of numerous issues it is compulsory to simulate the fading characteristics. It is important to use a fading simulator which enhances Rayleigh fading features of the signal. A good fading simulator may also use numerous channels with different time delays to simulate the varying path conditions.

One of the main thing that is to be taken into account in the laboratory when BER testing is done then it should be taken into account that transmitted signal don’t drips straight into the receiver and avoids passing through the fading simulator. It is tough to provide correct levels of transmission and few analysis's may be invalid if the transmitter power is high, Great care must be used to confirm that all signals travel through the fading simulator. Significant points of transmission may be mandatory. In some cases separated rooms have been used.[23][27][30]

CHAPTER-4

EXPECTED OUTCOMES

A transmitter, a receiver and a channel is required for BER testing. Firstly, a extended sequence of random bits is given as a input to transmitter. The work of the transmitter is to modifies these bits into particular form of digital signaling, that are transmitted through a simulated channel. The channel is simulated through addition of a measured amount of noise to the transmitted signal. The input to the receiver will be noisy channel. The receiver demodulates the signal, by creating the long arrangement of recovered bits. These bits are compared with the transmitted bits and find out faults. Bit-error-rate presentation is frequently portrayed on a two dimensional graph. The standardized signal-to-noise ratio (SNR) is articulated as Eb /N0: the energy- per-bit to one-sided power spectral density of the noise, which is expressed in decibels (dB). The bit-error-rate is a dimensionless quantity, frequently conveyed in powers of ten.[19]

Simulation Procedure

4.1 Run Transmitter

The transmitter is treated as the first step in the simulation. The transmitter is used to generate digitally modulated signal after a arrangement of pseudo-random bits. When the signal, $x(n)$ is created it becomes easy to measure the other things.

4.2 Establish SNR

Signal-to-noise-ratio (SNR) is represented as (Eb /N0). The SNR is expressed in decibels. If SNR is used then db's should be changed into an ordinary ratio to get the results. Let us take an example to understand the conversion:

SNR = m dB

$Eb/N0 = 10m/10$. $Eb/N0$ is a dimensionless quantity.

4.3 Determine *Eb*

Energy-per-bit can be stated as total energy of signal to number of bits which a signal contains. Energy-per-bit can also be defined as average signal power multiplied by period of one bit.

$$Eb = 1/(N.fbit)\sum_{n=1}^{N} x\text{^}2(n)$$

N= Number of illustrations a signal contains.

fbit=bit rate in bits per second.

4.4 Calculate *N*0

*N*0 can be calculated from the values of SNR and energy per bit values. To find out the *N0* divide *Eb* by the value of SNR. The value of SNR is converted from decibels to a ratio. The unit of power spectral density of the noise is W/Hz.

4.5 Calculate σ_n

The value of *N0* represents that how much noise is present in a particular band of signal. The noise bandwidth should be known to find out the values of average power of the noise and variance.

Let us take a signal, *x*(*n*), that is sampled on *fs* Hz frequency. In this case the bandwidth of noise is half of sampling rate. Average power of noise can be calculated by multiplying power spectral density of noise with noise bandwidth as :

$$\sigma_{n} = \frac{N_{0}.f_{s}}{2}$$

Wherever, σ_n is noise variance in W

*N*0 is one-sided power spectral density of noise.

The units of *N0* is W/Hz. and units of average noise power is W.

4.6 Generate Noise

The noise is having zero mean so it's having identical power and its variance. The length of the noise vector should be same as that of our signal vector *x(n)*.The unit of noise vector is volts.

4.7 Add Noise

Noisy signal can be created by addition of noise vector to signal vector. By adding the signal vector 'x' to noise vector 'n' noisy signal vector 'y' can be calculated by:

$$y = x+n$$

where, x= signal vector and n= noisy signal

4.8 Run Receiver

If noisy signal vector is created then one can practise the receiver to demodulate this signal. The sequence of demodulated bits is produced by the receiver. The sequence of demodulated bits is compared with the transmitted bits. This process will lead to know that exactly how many bits are in fault.

4.9 Determine Offset

Offset will be there among received bits and transmitted bits due to the delay inducing operations and the filtering process. The off-set should be determined before comparing the two bit sequences to check for errors. This can be done by associating the two sequences and then finding out the correlation peak.

Let us assume that the transmitted bits are kept in the vector 'tx', and received bits are kept in vector 'rx'. The number of bits in the received vector should be greater than transmitted vector. Receiver will give output whereas filters are filling and flushing. If length of transmitted bit vector is taken as *lt x* , and length of the received vector is *lrx* , range of possible offsets is between zero and $lrx - lt\ x - 1$. The offset is calculated with the help of fractional cross-correlation among the two vectors.

4.10 Create Error Vector

Bit errors can be calculated if the offset among the transmitted and received bit vectors is known. On behalf of bit values of zero and one difference will tell the bit error. Everywhere the bit error is encountered, the dissimilarity among the bits will be ±1, and where there is not a bit error, an alteration will be zero.

4.11 Count Bit Errors

The error vector consists of the non-zero features in state. The comparison should be done in order to find the number of non-zero elements.

4.12 Calculate Bit-Error-Rate

Whenever the simulation of the bit error rate is done a static number of bits are transmitted and received. It becomes necessary to know that in what manner the received bits are in mistake, find bit-error-rate as number of bit errors to total number of bits in the transmitted signal.[19][21]

When the processing gain calculations are made and by analyzing the outcomes it has been realized that by increasing the sample time processing gain becomes unchanged and processing gain and bandwidth procedure calculations are done deprived of using multi-rate filters. Development in the performance could be attained by use of spread spectrum, which can be defined as the processing gain of spread spectrum system. The BER calculations are studied devoid of with the matched filter and also by using matched filters.

Filter response:-

NBand=4;

N=60; %filter order

R=0.5; %rolloff factor

Pass-band ripple response-

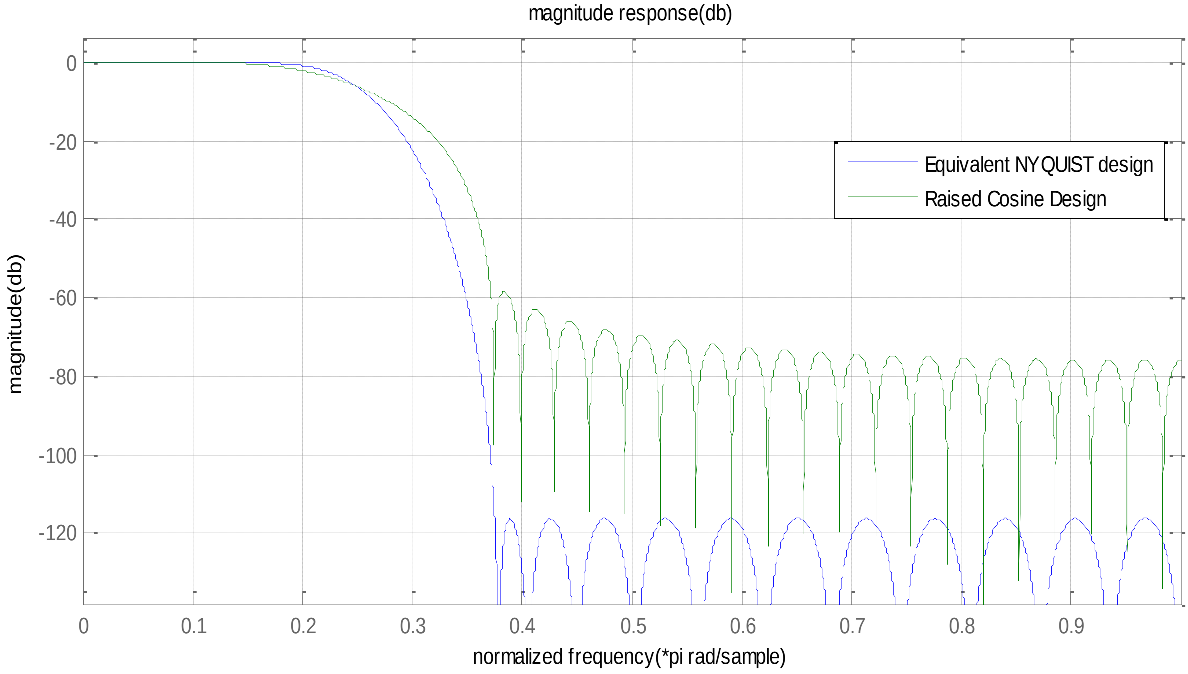


Fig 4.1:- Pass-band ripple response

Minimum phase design graph using filters

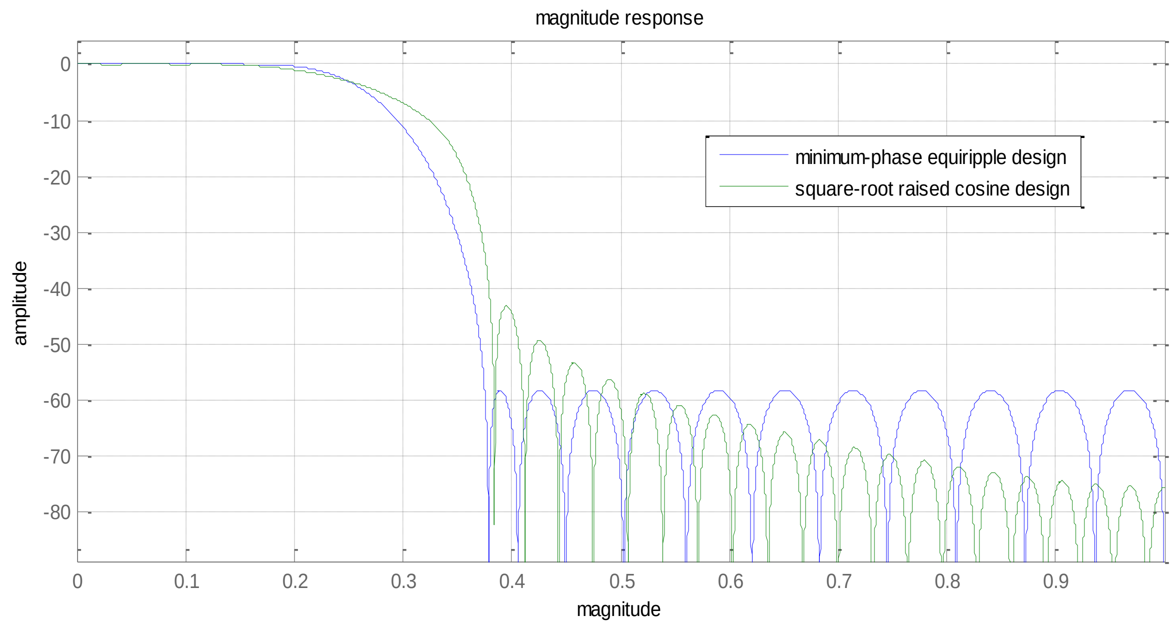


Fig 4.2:- Minimum phase design graph using filters

Magnitude response of the filter-

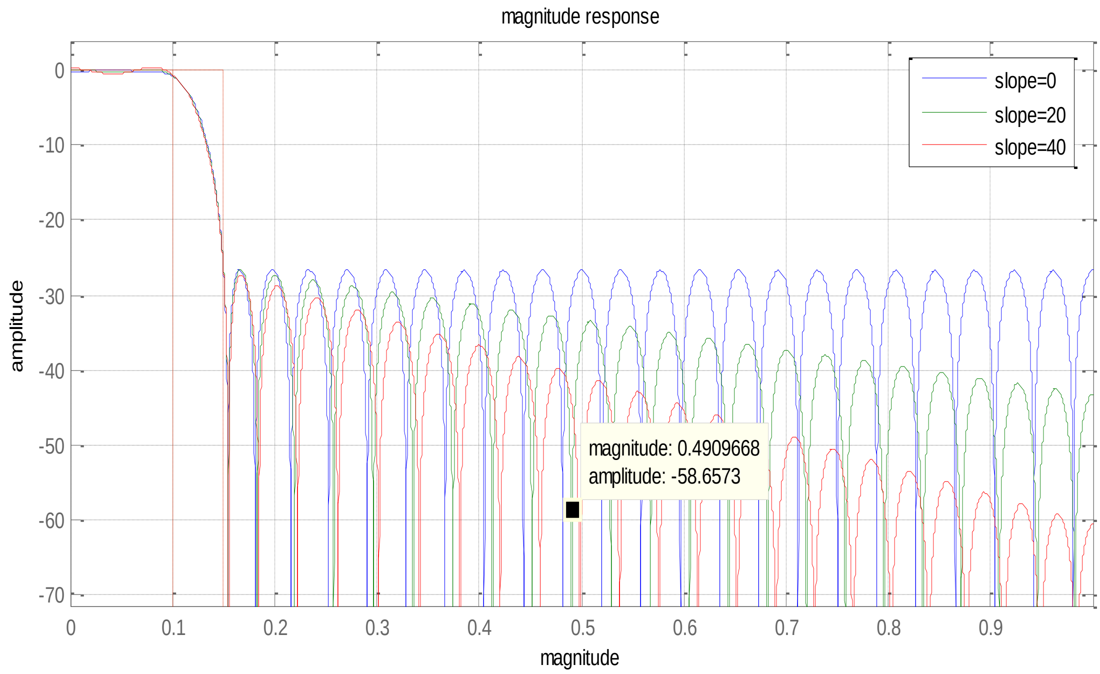


Fig 4.3:-Magnitude response of the filter

Magnitude response comparison graph-

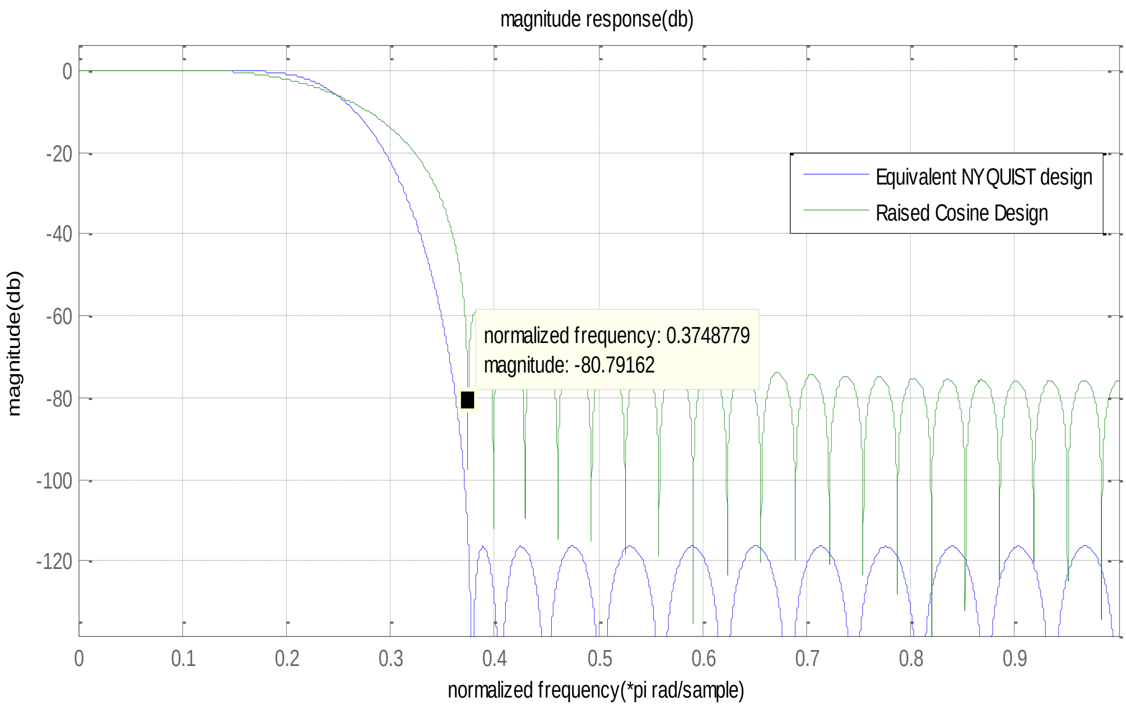


Fig 4.4:-Magnitude response comparison graph

Impulse response graph-

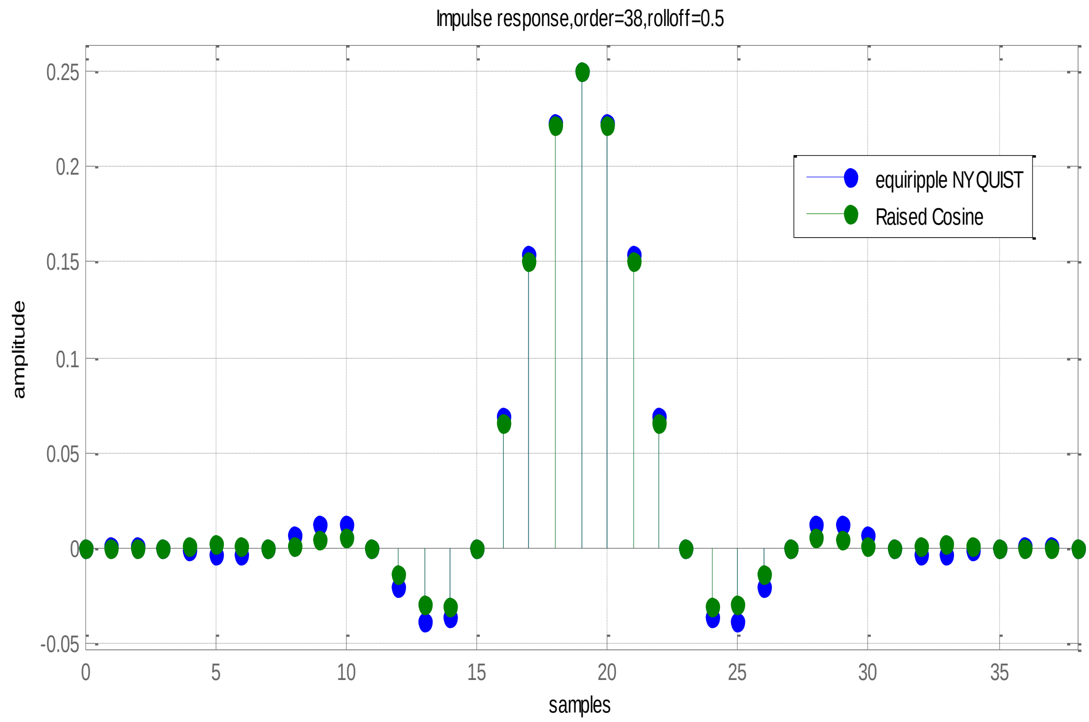
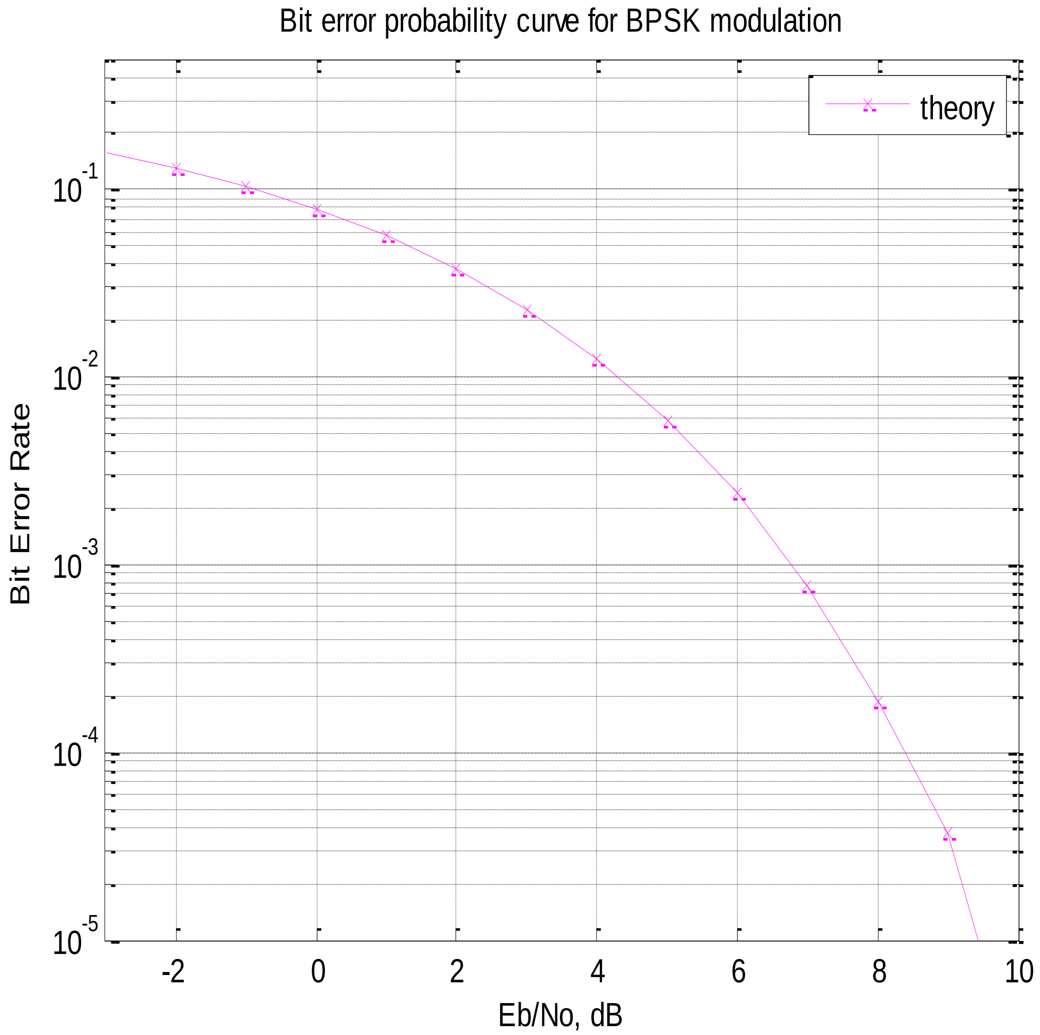


Fig 4.4:- Impulse response graph

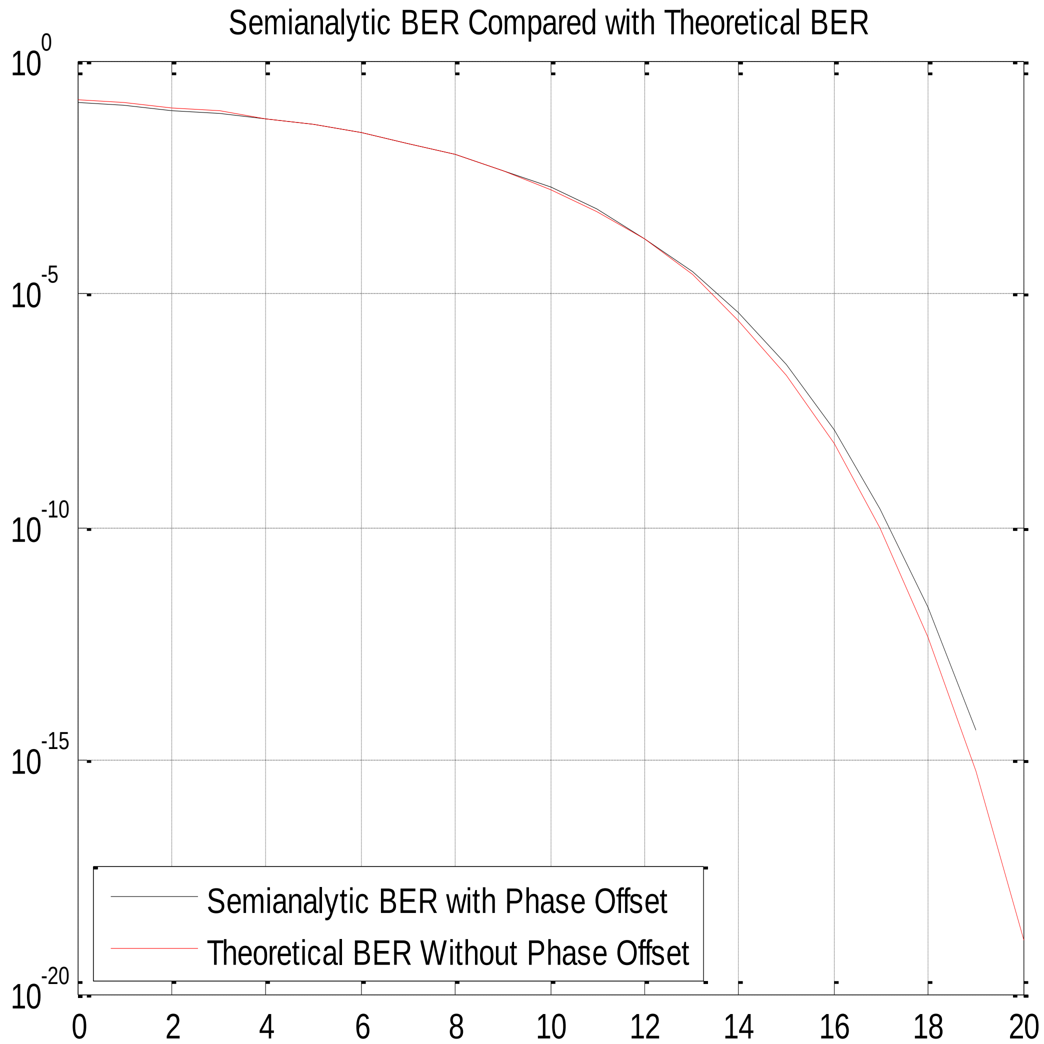
Graph for BPSK modulation:-

The BER plot can also be drawn by using the PN sequence and PN sequence can be created using rate conversion techniques. SNR and Eb/No plots are extra connected with the radio links and radio communication systems. To improve a system to deliver the presentation stages which are mandatory it’s important to manipulate the variables that can be controlled. These changes are done at the design stage of the data communication system thus the presentation restrictions are used at the original phases.

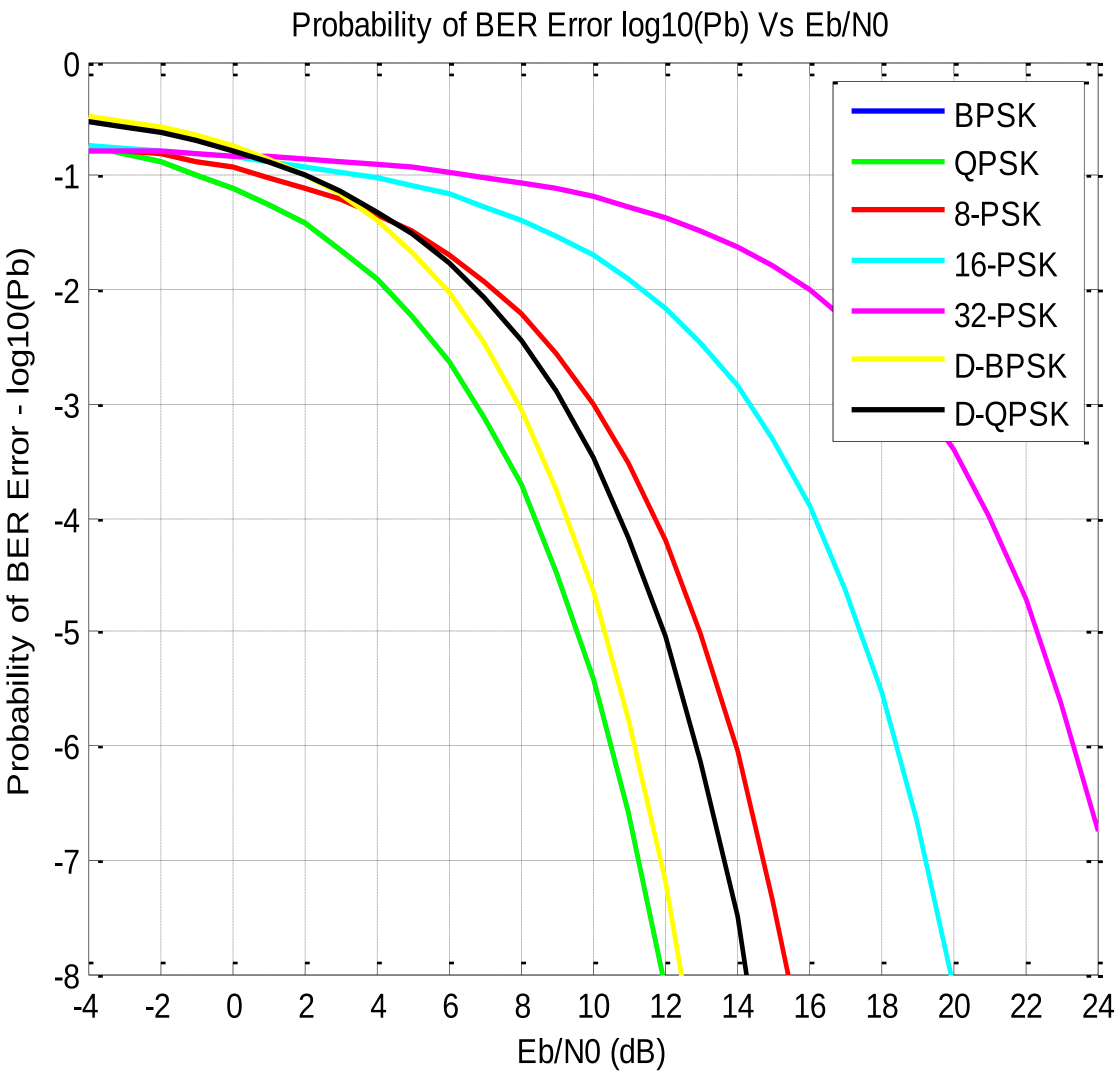


Comparison of theoratical BER with semianalytic BER:-

This graph represents the BPSK modulations Bit error rate in this coding is done to get the required result. The code perfroms following operations that includes the generation of random BPSK modulated symbols that is +1 and -1. After that these symbols are passed through the AWGN channel and then demodulation of the received symbols is done on the basis of the location in the constellation and then number of errors are counted and to get the multiple values of Eb/No this process is repeated again and again.

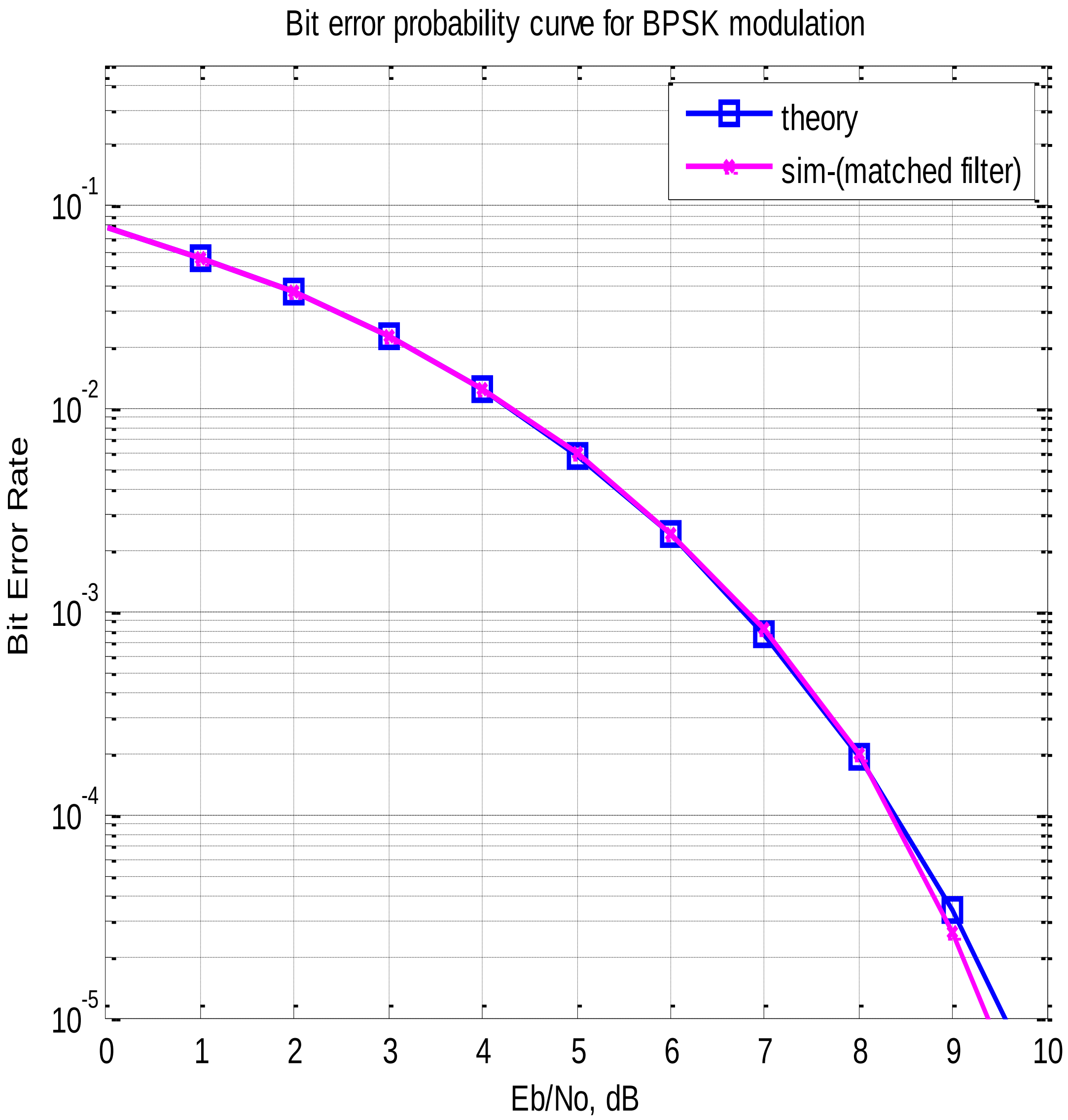

Semianalytic BER Compared with Theoretical BER
10^0
10^-5
10^-10
10^-15
10^-20
0
2
4
6
8
10
12
14
16
18
20
Semianalytic BER with Phase Offset
Theoretical BER Without Phase Offset

Graph showing comparison of various modulation techniques:-

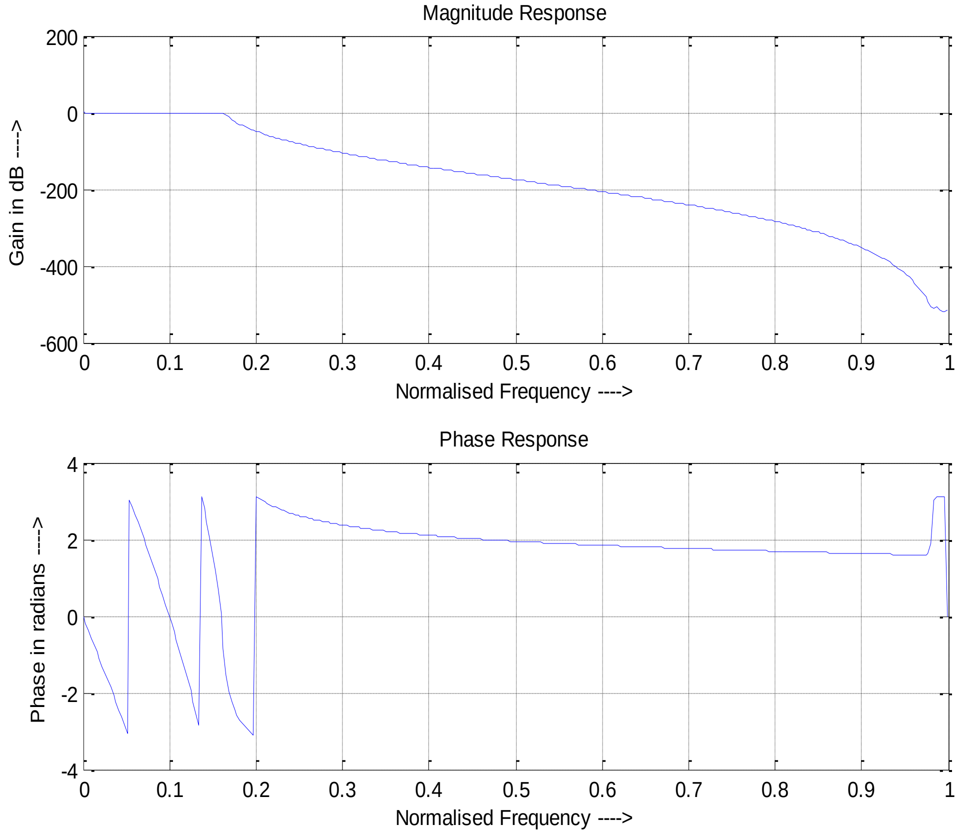


The comparison of various modulation graph illustrates that the comparison of various modulation techniques the various modulation techniques are considered. Here we have taken BPSK, QPSK, 8-PSK, 16-PSK, 32-PSK and differential QPSK. In these techniques the data input that is provided is digital or in binary form and the output is modulated analog spectrum. In the BPSK technique if input provided is 1 then output comes out to be 1 and if the input is 0 then output will be -1. In QPSK the bits are represented using complex carrier symbols each having the 90 degrees shift with one another. In QPSK it becomes easy for the receiver to get the original bits. While the BPSK technique is considered to be robust modulation scheme.

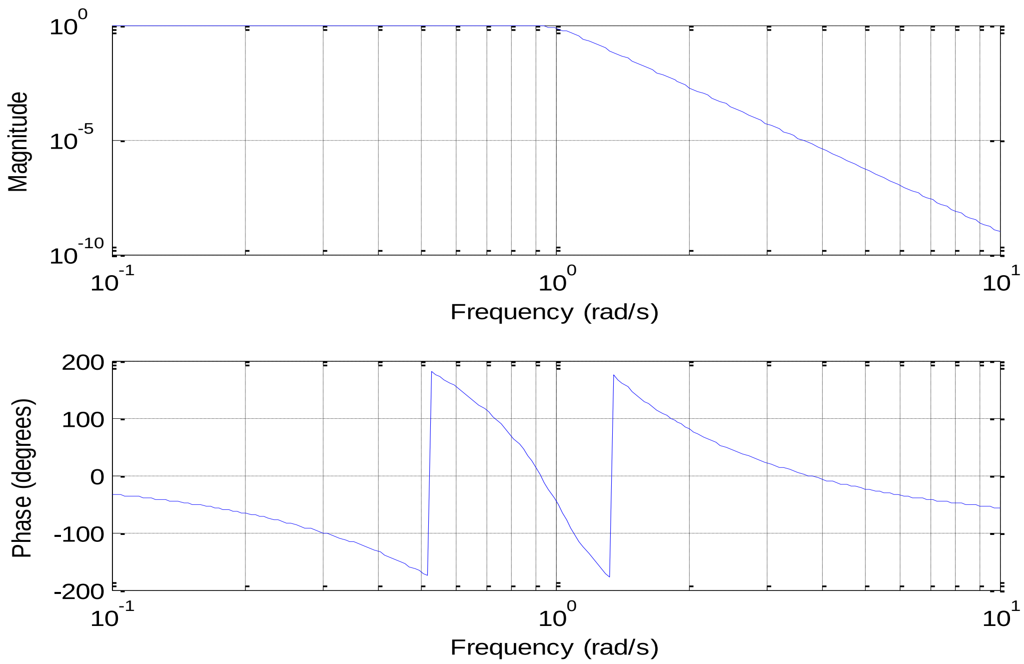
Graph for bit error probability curve for BPSK modulation:-



Chebyshew filter output:

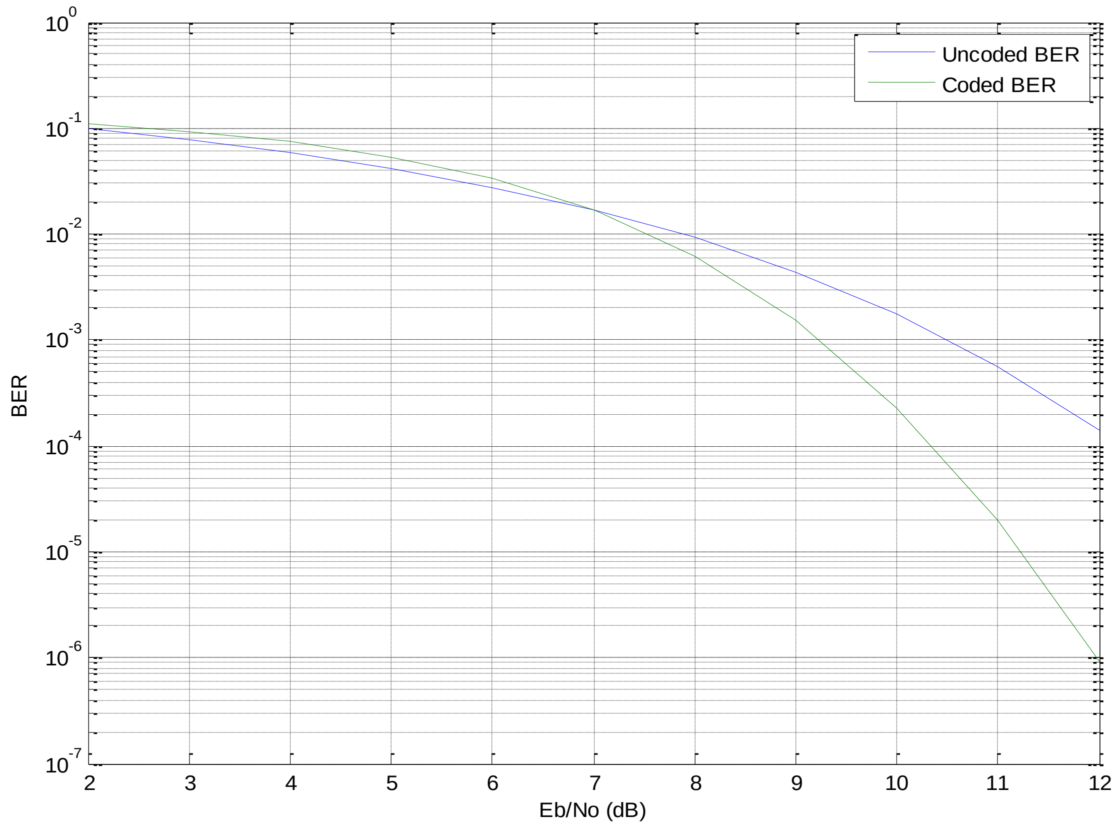


Response of Butterworth Filter:



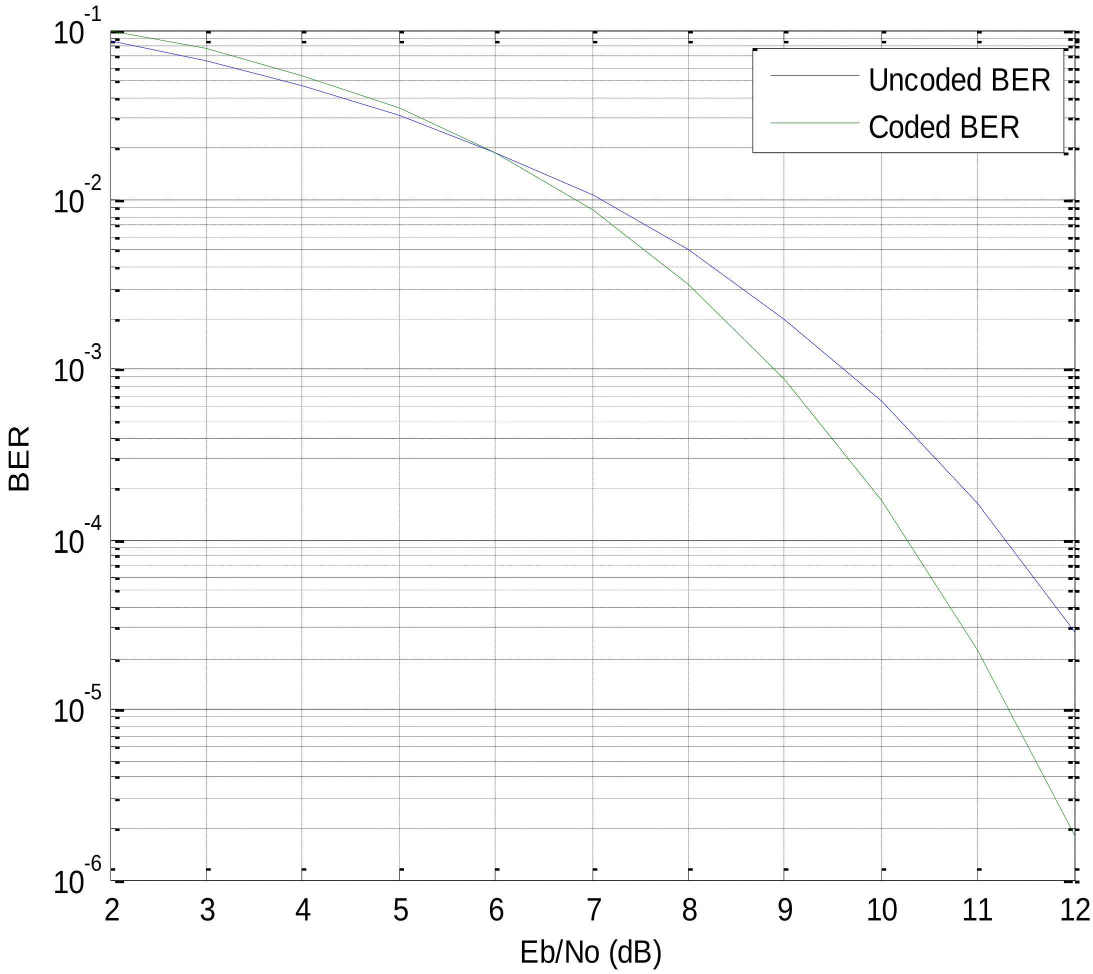
The chebyshev filter graph here shows us the magnitude and phase response it shows that how the magnitude and phase changes with respect to the frequency. The magnitude response remains constant for some time after that it starts to decrease as the frequency increases. But in the phase response the graph shows some fluctuations after that it shows a linear response.

Graph of BER by using the Cehbyshev Filter:



This graph shows that by using the filter according to the noise added we can decrease the BER of the system. Here Cehbyshev Filter is used at the receiver side which due to which errors introduced in the system and when these errors are calculated then the corresponding BER graph can be seen. Which shows the sharp transitions and the corresponding BER reduces.

Graph of BER by using the Butterworth Filter:



This graph shows that by using the filter according to the noise added BER of the system can be minimized. Here butterworth filter is used at the receiver side which due to which errors introduced in the system and when these errors are calculated then the corresponding BER graph can be seen. Which shows the sharp transitions and the corresponding BER reduces. The BER of the system decreases when we use the Butterworth Filter as in the graph the transitions can be seen.

CHAPTER 5

SUMMARY AND CONCLUSION

This study contributes in the part of multi-rate signal processing and also describes the convergence in between multi-rate signal processing techniques and multicarrier communication systems. Several developments also took place during this course of study, which were also included and considered in drawing the conclusions, as reported in this chapter. After introducing the area of multi-rate signal processing previously, the literature review was carried out and reported in next chapter. The literature review helps in identifying potential areas where improvisation can be made successfully.The basic of communication model are studied like BPSK model and many more. Here our main aim is to decrease the BER. For end to end testing of digital transmission systems BER testing is an influential technique. The BER test offers a quantifiable and beneficial suggestion for the presentation of the system that can be directly linked to its operative presentation. The system performance will noticeably degrade if the BER rises too high. If it is within restrictions then the system will work adequately. The BER can be controlled by some factors, the interference level existing in a system is generally set by exterior features and these can't be changed by the system design. By dropping bandwidth level of interference can be decreased. Though decreasing bandwidth restricts data throughput that can be attained. It is also likely to increase power level of system so that power per bit is increased. At the outflow of data throughput lower order modulation arrangements can be used. To reduce the bit error rate another technique is to reduce the bandwidth. The signal to noise ratio will improve when the lower levels of noise will be established. Due to this the reduction of the data throughput can be achieved. To attain a acceptable bit error rate it is necessary to balance every available factors. Bit error rate BER is a parameter that provides a brilliant suggestion of the presentation of a data link such as radio or fiber optic system. As one of main constraints of interest in any data link is number of errors that arise, bit error rate is an important constraint. In all the circumstances a close estimate of BER has been detected among the experimental and theoretical results. Here the experimental outcomes are the results which are computed from the model (communication model that has been taken under consideration) and the theoretical results are the results which are obtained from the available analytic formulation. By using the rate transmission technique the bit error rate can be minimized by enhancement in the processing gain that develops problematic for the interceptor to get the information and then sustaining the privacy of the information.[25][26][28][32]

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