Performance Analysis of Digital Communication Systems in Presence of AWGN Implementing Filter Technique

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ABSTRACT
In digital communication system, data must encounter in a band limited environment. The fact is that constraining the bandwidth of the transmitted signal necessarily increases the probability of BER at the receiver. The system performance in terms of BER is evaluated with as well as without filter developing digital communication model. The likelihood of BER and SER is compared. The results show that Raised Cosine filter exhibits almost equal performance in comparison with rest of the filters. The worst case scenario is obtained while introducing no filtering technique despite applying the modulation scheme QAM. Roll off factor 0.22 has taken into account as Raised cosine filter act as FIR filter at that factor. In this paper inter symbol interference (ISI) is examined and it is seen that ISI plays an important role in digital communication systems. It is also observed that overall transmission errors are less while increasing the ratio of bit energy per symbol (E_b/N_0). MATLAB communication environment has been used for simulation.

General Terms
BER, SER, Raised Cosine filter, Equiripple filter, Hamming Window filter, QAM

Keywords
ISI, Likelihood, Roll off factor, Band limited

1. INTRODUCTION
In modern digital communication system, digital modulation affords more capacity of information, high data security, better quality communications and efficient bandwidth utilization.

Although digital communication is much better than the analog communication, still it has certain issues that need to be addressed. Especially when it comes to wireless communication, one of the major research considerations becomes the effect of multipath propagation. A thorough analysis is necessary for strategic planning of any system designed by doing comparative study of different modulation techniques with and without filter via different multipath communication channels. In this paper AWGN channel is considered. To study and draw the graph in terms BER versus Eb/No in presence of AWGN channel for various QAM modulation schemes with and without filter is tested using computer simulation. Therefore, understand the system could go whether for suitable modulation technique with filter or merely suitable modulation technique to suit the channel quality can suggest better performance [1] [2].

In this work, a digital communication model using QAM i.e. M= 16, 32, 64 with digital filtering and without filtering is proposed. The design criteria of filters at transmitter (Tx) and receiver (Rx) is carefully designed using MATLAB environment with the requirements of reducing inter-symbol interference (ISI). Nyquist's condition of reducing ISI is considered introducing Raised Cosine filter at transmitter side for band limited transmitting signal. On the other hand same Raised Cosine filter, Equiripple and Hamming window filter are used at the receiver side for detecting noisy signal.

The remaining paper is organized as follows. In Section II, Modulation Techniques are described. In Section III, Additive White Gaussian Noise (AWGN) is explained. In Section IV Digital FIR filters i.e. Raised Cosine, Equiripple, Hamming Window filter are described and simulation results are given in Section V. This paper is concluded in Section VI.

2. MODULATION TECHNIQUES
One way to communicate a message signal whose frequency spectrum does not fall within that fixed frequency range, or one that is unsuitable for the channel, is to change a transmittable signal according to the information in the message signal. This alteration is called modulation, and it is the modulated signal that is transmitted. The receiver then recovers the original signal through a process called demodulation.

2.1 Digital Modulation
Digital modulation schemes transform digital signals into waveform that are compatible with the nature of the communications channel. One category uses a constant amplitude carrier and the other carries the information in phase or frequency variations (FSK, PSK). A major transition is from the simple Amplitude Modulation (AM) and Frequency Modulation (FM) to digital techniques such as Quadrature Phase Shift Keying (QPSK), Frequency Shift Keying (FSK), Minimum Shift Keying (MSK) and Quadrate Amplitude Modulation (QAM) [4].

2.2 M-Quadrature Amplitude Modulation
QAM is used in applications including microwave digital radio, DVB-C (Digital Video Broadcasting Cable), and so on.

In 16-QAM, Quadrature Amplitude Modulation used in this paper has four I values and four Q values. This results in a total of 16 possible states for the signal. It can transition from any state to any other state at every symbol time. Since 16 = 24, four bits per symbol can be sent which is as shown in Fig 1(a). This consists of two bits for I and two bits for Q. The symbol rate is one fourth of the bit rate. So this modulation format produces a more spectrally efficient transmission. It is more efficient than BPSK, QPSK, or 8PSK. For example, another variation is 32QAM which is used in this paper along with 16 and 64 QAM.In this case there are six I values and six Q values causing a total of 36 possible states (6x6=36). This is too many states for a power of two (the closest power of two is 32). So the four corner symbol states, which take the most power to transmit, are omitted. This reduces the amount of peak power the transmitter has to generate. Since 25 = 32, there are five bits per symbol and the symbol rate is one fifth of the bit rate. However, the symbols are very close together and are thus more prone to errors due to noise and distortion which is shown in simulation result. The 32 QAM is as shown in Fig. 1(b).

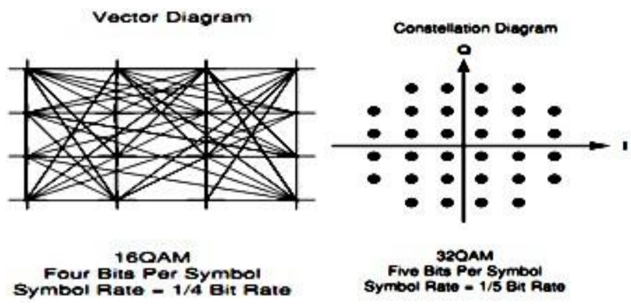

Fig 1: a) Vector diagram for M=16, and b) Constellation diagram for M=32

3. AVERAGE ENERGY OF AN M-QAM CONSTELLATION

In a general M-QAM constellation where M= 2b and b the number of bits in each constellation is even, the alphabets used as Eq. (1).

$$\alpha_{MQAM} = \{\pm(2m-1) \pm (2m-1)j\} \text{where } m \leftarrow \{1, \ldots, -\frac{\sqrt{M}}{2}\} \quad (1)$$

For computing the average energy of the M-QAM constellation, let us proceed as follows:

Find the sum of energy of the individual alphabets:

$$E_\alpha = \sum_{m=1}^{\frac{\sqrt{M}}{2}} |(2m-1) + j(2m-1)|^2$$

$$= \frac{\sqrt{M}}{3}(M-1) \quad (2)$$

(b) Each alphabet is used $2\sqrt{M}$M times in the M-QAM constellation.

So, to find the average energy from M constellation symbols, divide the product of (a) and (b) by M. Plugging in the number

for 16-QAM:

$$E_{16QAM} = 2/3(16-1) = 10$$

64-QAM:

$$E_{64QAM} = 2/3(64-1) = 42$$

From the above explanations, it is reasonably intuitive to guess that the scaling factor of $1/\sqrt{E_\alpha}$Eα which is seen along with M-QAM constellations respectively is for normalizing the average transmits power to unity.

The modulation schema is as shown in Table I.

Table 1. Modulation Schemes

Parameters	Values
Type QAM(M)	16, 32, 64
Symbol Rate (f_o)	1 Hz
Bit Rate (R)	K bit/second
x-symbols	k=$Log_2^{(M)}$
Data source	30k bits

4. BIT ERROR RATE (BER)

The BER, or quality of the digital link, is calculated from the number of bits received in error divided by the number of bits transmitted.

$$BER = (\text{Bits in Error}) / (\text{Total bits received}) \quad (3)$$

In digital transmission, the number of bit errors is the number of received bits of a data stream over a communication channel that has been altered due to noise, interference, distortion or bit synchronization errors. The BER is the number of bit errors divided by the total number of transferred bits during a particular time interval. BER is a unit less performance measure, often expressed as a percentage [10].

BER can also be defined in terms of the probability of error (POE) [13] and represented by Eq. (4).

$$POE = \frac{1}{2}(1 - erf)\sqrt{\frac{E_b}{N_0}} \quad (4)$$

Here, 'erf ' is the error function, Eb is the energy in one bit and N0 is the noise power spectral density (noise power in a 1Hz bandwidth). The error function is different for the each of the various modulation methods. The POE is a proportional to Eb/ N0, which is a form of signal-to-noise ratio. The energy per bit, Eb, can be determined by dividing the carrier power by the bit rate. As an energy measure, Eb has the unit of joules. N0 is in power that is joules per second, so, Eb/ N0 is a dimensionless term, or is a numerical ratio.

5. SIGNAL TO NOISE RATIO (SNR)

SNR is the ratio of the received signal strength over the noise strength in the frequency range of the operation. It is an important parameter of the physical layer of Local Area Wireless Network (LAWN). Noise strength, in general, can include the noise in the environment and other unwanted signals (interference). BER is inversely related to SNR, that is high BER causes low SNR. High BER causes increases packet loss, increase in delay and decreases throughput [3]. The exact relation between the SNR and the BER is not easy to determine in the multi channel environment. Signal to noise ratio (SNR) is an indicator commonly used to evaluate the quality of a communication link and measured in decibels and represented by Eq. (5).

$$SNR = 10 \log 10 \text{ (Signal Power / Noise Power) dB} \quad (5)$$

6. EB/N0 (ENERGY PER BIT TO NOISE POWER SPECTRAL DENSITY RATIO)

Eb/N0 is an important parameter in digital communication or data transmission. It is a normalized signal-to- noise ratio (SNR) measure, also known as the "SNR per bit". It is especially useful when comparing the bit error rate (BER) performance of different digital modulation schemes without Taking bandwidth into account. Eb/N0 is equal to the SNR divided by the "gross" link spectral efficiency in (bit/s)/Hz, where the bits in this context are transmitted data bits, inclusive of error correction information and other protocol overhead.

6.1 ADDITIVE WHITE GAUSSIAN NOISE (AWGN)

It is modeled as a zero-mean Gaussian random process where the random signal is the summation of the random noise variable and a direct current signal as shown in Eq. (6) [7], [8] and [15].

$$Z = a + n \quad (6)$$

The probability distribution function for this Gaussian noise can be represented as:

$$p(z) = \frac{1}{\sigma\sqrt{2\pi}} \exp[-\frac{1}{2}(\frac{z-a}{\sigma})^2] \quad (7)$$

The model of this noise assumes a power spectral density Gn(f) which is flat for all the frequencies denoted as;

$$Gn\,(f) = \frac{N_0}{2} \quad (8)$$

The factor 2 indicates that the power spectral density is a two-sided spectrum. This type of noise is present in all communication systems and is the major noise source for most systems with characteristics of additive, white and Gaussian. It is mostly used to model noise in communication systems which are simulated to determine their performance. This noise is normally used to model digital communication systems which can be replaced with other interference schemes.

7. DIGITAL FIR FILTERS

In the digital communication system there is a requirement to design a FIR filter that is not only does well but it is optimal. In each band of interest optimization is the ability to state a maximum error. In this paper three FIR filters are designed, one of them is Raised Cosine filer which is used both in transmitter and receiver side. The others of the filter are Equiripple and Hamming Window filter. These filters are used at the receiver side.

Filter specification for Raised Cosign, Equiripple, and Hamming Window filter is as shown in Table 2. This specification is used for all filters. The length of the all filters is derived from Eq. (9).

$$Filter\ Length\ (L) = 2 * M * D + 1 \quad (9)$$

$$Filter\ Order\ (N) = L - 1 \quad (10)$$

Where, M is oversampling rate which is integer, D is delay of the symbol which is also integer. To avoid half symbol delay one is added to this length equation. In this paper M is equal to 2 and D is equal to 6 are considered. So, the filter length is 25. According to Eq. (10) the filter order is 24, which are shown in Table 2.

Table2. Filter Specifications

Parameters	Values
Filter Type	Raised Cosine, Equiripple, Hamming Window
Design Method	FIR Filter
Filter Length(L)	25
Filter Order (N)	24
Sampling Frequency (F_s)	2Hz
Roll of factor (α)	0.22
Low cutoff frequency(f_p)	(1-α) $\frac{f_0}{2}$
Upper cutoff frequency(f_s)	(1+α) $\frac{f_0}{2}$

7.1 Raised Cosine Filter

In modern data transmission systems, bits or groups of bits (symbols) are typically transmitted in the form of individual pulses of energy. Sometimes Rectangular pulse is probably the most fundamental. It is easy to implement in a real-world system because it can be directly compared to opening and closing a switch, which is synonymous with the concept of binary information. Pulses are sent by the transmitter are detected by the receiver in any data transmission system. At the receiver, the goal is to sample the received signal at an optimal point in the pulse interval by the matched filter to maximize the probability of correct decision. This implies that the fundamental shapes of the pulses be such that they do not interfere with one another at the optimal sampling point. There are two criteria that ensure non-interference.

a. The pulse shape exhibits a zero crossing at the sampling point of all pulse intervals except its own. That is Minimized inter symbol interferences (ISI).

b. The shape of the pulses is such that the amplitude decays rapidly outside of the pulse interval. That is high stop band attenuation.

The rectangular pulse meets first requirement because it is zero at all points outside of the present pulse interval. It cannot cause interference during the sampling time of other pulses. The trouble with the rectangular pulse, however, is that it has significant energy over a fairly large bandwidth as indicated. The unbounded frequency response of the rectangular pulse makes it unsuitable for modern transmission systems. This is where pulse shaping filters come into play. If the rectangular pulse is not the best choice for band-limited data transmission, then what pulse shape will, decay quickly, and provide zero crossings at the pulse sampling times [5]. There are several choices that have but in most systems Raised cosine filter are used to shape the input pulse.

In most cases, the square root raised cosine filter is used in the transmitter and receiver part of the system so that the overall response resembles that of a raised cosine filter. The impulse or time domain response of the raised cosine filter and the square root raised cosine filter [14] are given by Eq. (11), (12), (13), (14).

$$\mathrm{hRC(t)} = \frac{\sin(\frac{\pi T}{T})}{(\frac{\pi T}{T})} \frac{\cos(\frac{\pi\alpha T}{T})}{1-(\frac{\pi T}{T})^2} \quad (11)$$

This expression can be simplified further by introducing the sinc function ($sincx = \frac{\mathrm{Sin\ x}}{\mathrm{x}}$).

$$\mathrm{hRC(t)} = \sin\mathrm{c}\,(\frac{\pi T}{T}) \frac{\cos(\frac{\pi\alpha T}{T})}{1-(\frac{\pi T}{T})^2} \quad (12)$$

The sinc function in the response of the filter ensures that the signal is band-limited. The time domain or impulse response of the square root raised cosine filter is given as;

$$hRRC(t) \; = \frac{\sin[\pi(1-\alpha)t]+4\alpha\left(\frac{t}{T}\right)\cos[\pi(1+\alpha)\frac{t}{T}]}{\left(\frac{\pi T}{T}\right)[1-(\frac{4\alpha t}{T})^2]}$$

(13)

The overall response of the system is given by Eq. (14).

$$hRC(t) \; = \; hRRC(t)\, hRRC(t) \qquad (14)$$

The impulse and magnitude response of Raised Cosine filter are shown in Fig. 2 and 3.

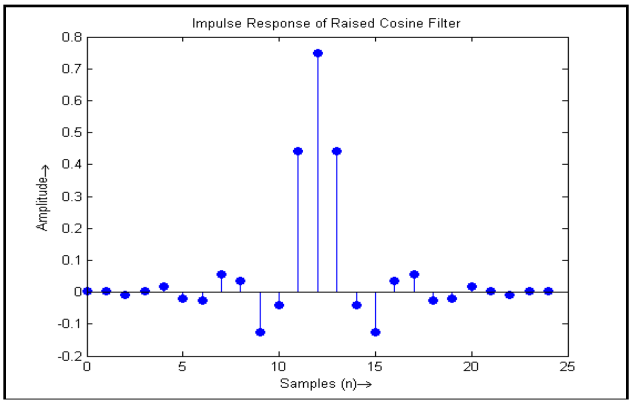

Fig 2: Impulse response of raised cosine filter

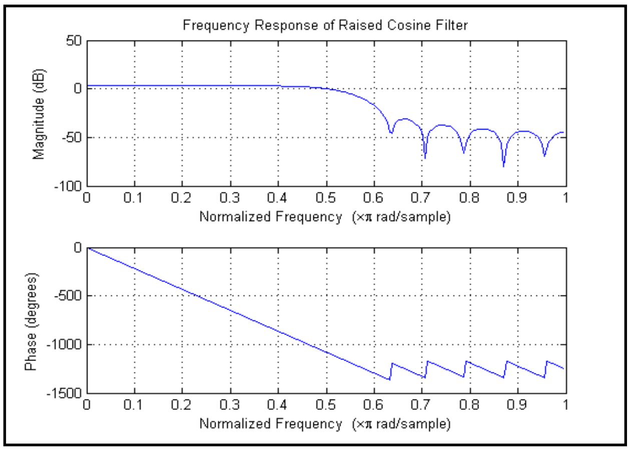

Fig 3: Frequency response of raised cosine filter

7.2 Equiripple Filter

To design a linear phase Equiripple FIR filter, the Parks-McClellan and Remez exchange method is used. For meeting the specific tolerance application, Equiripple filters are mostly suited as it has equal ripple in both pass band and stop band. In general, the modified Fourier series method is used where the pass band frequency response is monotonically decreasing and the maximum error is found at cutoff frequency [9].

A typical filter allows the specification of pass band and stop band frequencies, ideal gains, and deviation (or Equiripple) from the desired transfer function. In terms of ripples, the transition band is most often assumed to be arbitrary. An Equipipple FIR is a special class of FIR filter which is particularly effective in meeting such specifications [13]. The maximal deviations (ripple error) from the ideal transfer function are minimized by an Equiripple design protocol. The Equiripple algorithm applies to a number of FIR design instances.

The Magnitude response of an Equiripple low pass filter is as shown in Fig. 4.

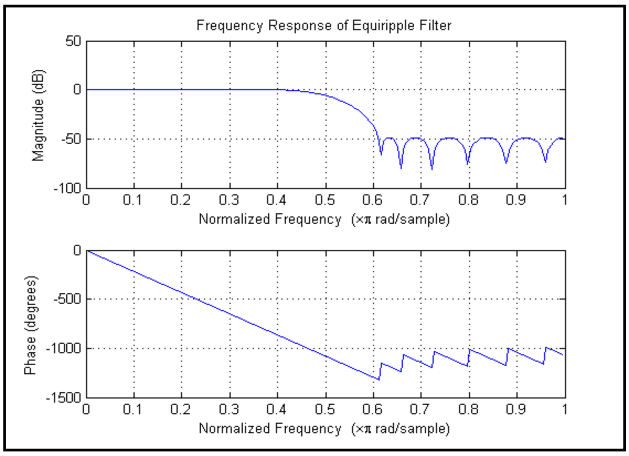

Fig 4: Frequency response of Equiripple filter

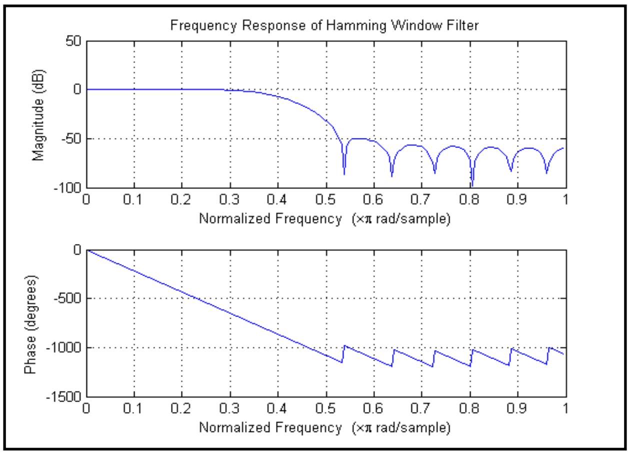

Fig 5: Frequency response of Hamming Window filter

7.3 Hamming Window Filter

The hamming window is like a raised cosine window. The hamming window exhibits similar characteristics to the raised cosine window but further suppress the first side lobe [11].

The hamming window is defined by Eq. (15).

$$W(n) = \begin{cases} \left(0.54-0.46\cos\left(\frac{2\pi n}{N}\right)\right), \text{for } n=0 \text{ to } N \\ 0, \text{else where} \end{cases} \qquad (15)$$

Where, N is the order of the filter which is depends on filter length (L). It can be define with the Eq. (10).The Magnitude response of a hamming Window filter is as shown in Fig. 5.

As the Fig. 2, 3, and 4 are shown the transition band of the filter is between the lower cut off frequency which is 0.39 and upper cut off frequency which is 0.61.In addition, these figures indicate that the pass band attenuation is greater than or equal to 0 dB while the stop band attenuation is greater than or equal to 50 dB. Moreover, Frequency response curves of those figures reveal that all of filter act as linear phase characteristic which is important factor for FIR filter.

8. SIMULATION AND RESULTS

The model for digital communication system with filter is as shown in Fig. 6. The work has been simulated by using communication toolbox of MATLAB [16].

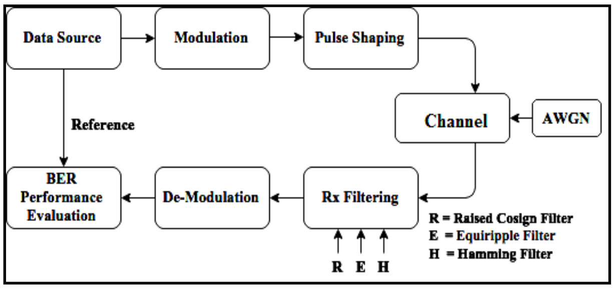

Fig 6: Model of digital communication system with filter

In this simulation, k=30*103 information bits have been generated using ‘randtint’ function to create a column vector that lists successive values of binary data stream. This binary data stream can be equivalently described by an analog signal, band limited to ½ times 1 sample per pulse. That is, if the bit rate is one bit per second and sampled it at 1 sample per second, the necessary band width is ½ Hz. That is why a Raised Cosine filter is used which acts as low pass filter with roll of factor 0.22. In this case square root Raise Cosine filter is equivalent to FIR filter.

The simulation study has been done for digital communication system with and without filter at the different QAM (M=16, 32, 64) transmission rates in presence of AWGN.

The same filter specification is used for all three filters. The impulse and frequency response of Equiripple and Hamming Window filter are almost same compared to Raised Cosine filter.

8.1 Simulation and discussions

Fig. 7, 8 and 9 show the result of SNR vs. BER for M=16, 32 and 64 with all specified filters and without filter respectively. In simulation, Raised Cosine filter is used both in transmitter and receiver side where as Equiripple and Hamming Window filters are only used at the receiver side.

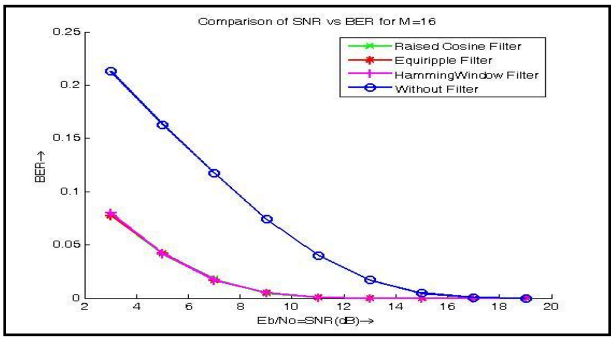

Fig 7: Comparison of SNR vs. BER for M=16

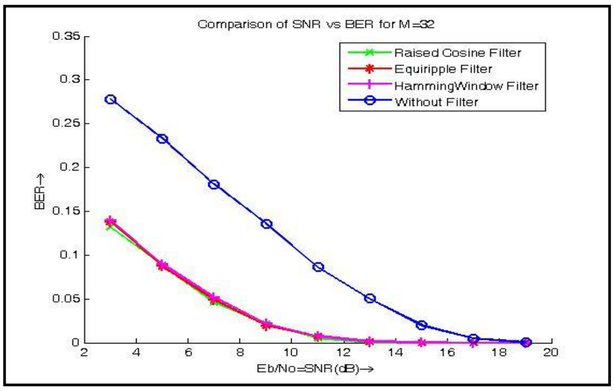

Fig 8: Comparison of SNR vs. BER for M=32

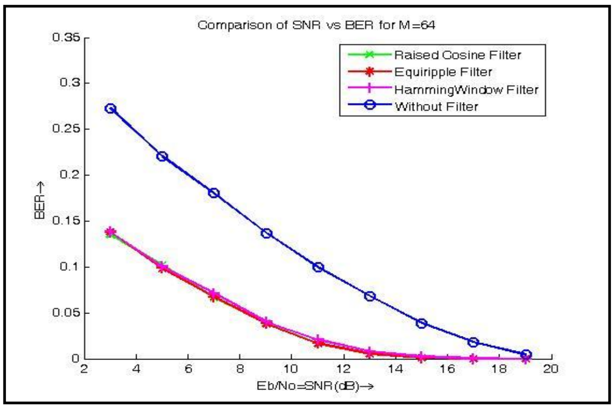

Fig 9: Comparison of SNR vs. BER for M=64

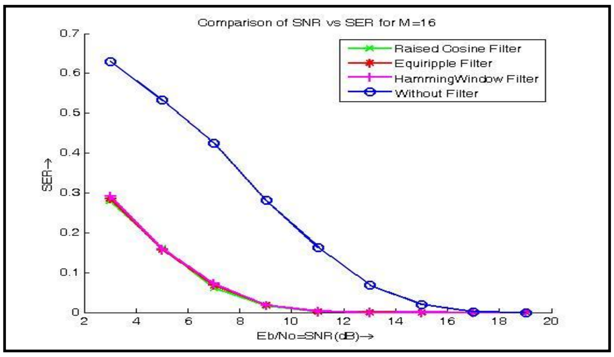

Fig 10: Comparison of SNR vs. SER for M=16

Fig. 10, 11 and 12 show the result of SNR vs. SER for M=16, 32 and 64 with all specified filters and without filter respectively. In these cases, Raised Cosine filter also used both in transmitter and receiver side. Equiripple and Hamming Window filter only used at the receiver side.

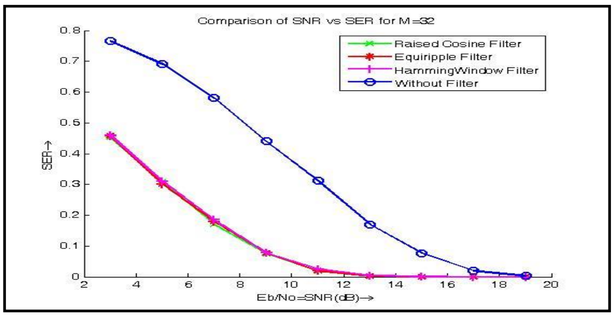

Fig 11: Comparison of SNR vs. SER for M=32

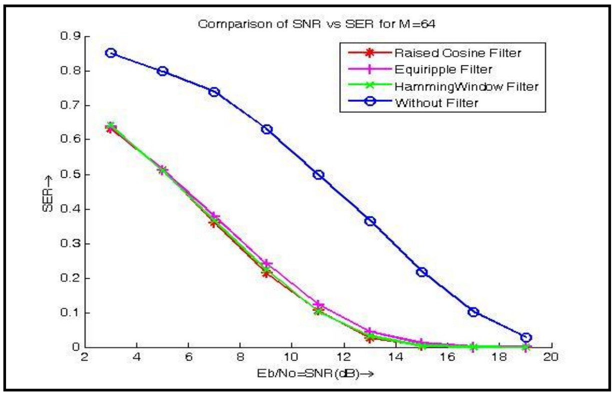

Fig 12: Comparison of SNR vs. SER for M=64

8.2 Discussion of results

The resulting bit error rate (BER) and Symbol error rate (SER) at the output of the receiver with respect to (Eb/No=SNR) are shown in Fig. 7,8,9,10,11,12 for different QAM transmission rates i.e. M=16, 32, 64. These results reveal a comparison between the resulting errors in at the received signal at different noise levels. Since (Eb/No) is defined as the ratio of bit energy per symbol to noise power spectral densities in dB increasing this ratio causes less overall bit error rate and decreasing this ratio causes higher bit errors rate.

The system performance is degraded severely when applying no filter technique. On the other hand, system shows almost equal performance for all receiving filters both for the BER and SER. Among them Raised Cosine filter shows marginally better performance.

9. CONCLUSION

The BER and SER performance have been simulated for digital communication system with and without filter scheme for baseband transmission where the error rate depends on the modulation technique used, the ratio (Eb/No) and channel conditions, inter symbol interference and filter technique. This paper has analyzed the performance of both systems for various QAM over the AWGN channel with and without filter. These results reveal that the model without filter is lesser efficient than that of model with filters. This analysis emphasized that ISI degrades the performance of the system if ISI is not eliminated before transmission of digital signal.

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