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ESTIMATING THE LOW COST PROBABILITY ERROR IN QAM USING THE SRRC FILTER WITH OTHER FILTER

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Abstract—: Nowadays mobile communication is a part of our life and it is essential for us. In the transmission and receiver side using QAM (Quadrature Amplitude Modulation) with SRRC (Square Root Raise Cosine) filter using transmission and receiver side. In transmission system four signal pass through at a time in multiplication and addition system in one bandwidth, the incoming signal in receiver side which reduce in cost per time, less error rate BER (Bit Error Rate), SNR (Signal to Noise Ratio) with low power. In receiver side we pass through the Multistage filter, Nyquist filter and SRRC filter. As a result, best filter which is QAM with SRRC filter shown this paper using “ESTIMATING THE LOW COST PROBABILITY ERROR IN QAM USING THE SRRC FILTER WITH OTHER FILTER” simulated MATLAB 2019a.

Keywords: — SRRC, QAM, BER, RoF, ISI.

I. INTRODUCTION

As this is the era of modern communication. Our technology becomes so significant that it must progress in parallel with the growth of mankind and modern society[3]. The process of "sending or transmitting" and "receiving" a message from a source point to a destination point across a communication channel is known as communication. Only when the receiver can decode the exact message sent by the transmitter is communication said to be conclusive{[1]}. It is now critical to adopt the required procedures for each system in order to construct multipath communication networks. During data transmission, the phase and frequency are reassembled using

QAM (Quadrature amplitude modulation) to reduce bandwidth and then pass through the AWGN channel check comparator, Error count, and calculate BER (Bit error rate) with SNR in the received signal (signal to noise ratio)[1],[3].

A. QAM Process

The signal must travel a long distance and must reach a large number of destinations, modulation is preferred. The method of altering the carrier signal properties in accordance with the modulating signal is known as modulation. In QAM, there are Four carrier signals separated by $\pm 45^\circ$ degrees (00, 01, 10, & 11). As a result, two phases " I" and quadrature or " Q" are created. These are normally located within the baseband area supplied. The four follow-on signals are added together, and the result is used to forward the required RF signal. Frequency amplification is required to convert this frequency to the final frequency. Demodulation is the reversal of the QAM process. The discriminator receives the QAM output here. It separates QAM output into four quadrants and feeds them into a mixer. A local oscillator signal is received by the mixer and the local carrier, one in phase and the other with phase shift. The ' I' output and the " $\pm O$ " output are the mixer's four outputs. When a $\pm 45^\circ$ phase-shifted carrier is applied to the mixer, the " $\pm O$ " output is produced; otherwise, the ' $\pm I$ ' output is produced. 256- QAM contains eight " I" and eight " Q" values, implying high proficiency and production [2],[3],[4]

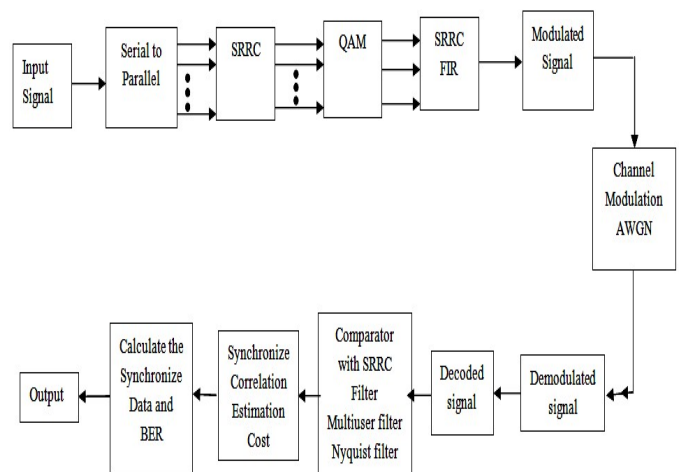


Figure 1:-Block diagram of QAM with SRRC filter

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Generally a baseband channel, modulator, and demodulator are used to process a binary data stream via a communications network. A stem plot to depict a part of the random data, constellation diagrams to show the broadcast and received signals, and a bit error rate calculator (BER).

In this model Process the input signal convert to serial to Parallel then pass through SRRC filter then pass through to QAM processor then pass through SRRC with FIR filter and modulate the signal for broadcasting. In receiver side the incoming signal those are pass through the Channel Modulating with AWGN and Demodulate the signal. After demodulation system signal are decoded and comparator check with SRRC filter, Multiuser filter and Nyquist filter. All the process synchronize correlation estimation cost those are adder and multiply the signal and calculated it. Then it was synchronize data and BER calculate and signal goes to output data.

B. Generate random binary data stream

A vector or matrix is the most common way to represent a signal in MATLAB. A binary data stream which is converted to radiant function generates a column vector. By default set binary data stream's length to 240,000.

To make the sample repeatable, rearrange the radiant function into its default state or any static seed value. After that, the radiant function can be used to create a random binary data stream[5]. To depict the first 40 bits[5] into binary values from the random binary data stream, use a stem plot. Use the colon (:) operator in the stem function call to select a part of the binary vector.

C. Convert Integer from Signal binary signal to integer value and us 16 QAM, 64 QAM, 128 QAM, & 256 QAM

By default, the QAM modulation function expects integer-valued data for modulation input symbols. Before using the QAM modulation function, the binary input stream is preprocessed into integer values. The binary to decimal function converts each 4-tuple into an integer in the range [0, (M-1)]. The modulation order, M, is 16, 64, 128, 256.

Transform the data into binary k-tuples, where k equals the amount of bits per symbol ($k = \log_2 M$). After that, each 4-tuple is converted to an integer number using the binary function.

The modulation process produces complex column vectors with values that are 16 QAM, 64 QAM, 128 QAM, 256 QAM signal constellation elements. The natural and Gray binary symbol mapping is shown later in this example constellation diagram.

More information on modulation functions can be found at Digital Modulation. Gray coding using phase-shift keying (PSK) modulation is a form of Gray coding.

D. Converted integer_valued signal to binary signal

The data symbols from the QAM demodulator are converted. Data Symbols Out, to a binary matrix, data Out Matrix, with "d" dimensions of sym by Nbits/sym, use the decimal to binary function.

Nbits/sym means the number of bits per symbol and the

total number of QAM symbols in the matrix is Nsym. Nbits/sym = 4 in 256-QAM, converting the matrix to a column vector with a length equal to the number of input bits, which is 240,000. As a result the process for the Gray-encoded data symbols, data Symbols Out G.

E. Compute BER system

Bit error statistics are generated by the bit error function using the original binary data stream data In and the receiving data streams data Out and data Out G. The BER lowers considerably when grey coding is utilized. Error statistics can be computed using the error rate function[5].

II. MATH

1) RAISED COSINE FILTER

In broadcasting system, Individual pulses are typically used to convey the bits. In order to keep power emissions below the legal limit, it is necessary to regulate them. A significant lobe exists in the signal. The minor lobe is prone to showing up in the adjacent ISI spectral region. The minor lobe has a proclivity for appearing in adjacent spectral bands, ISI. To avoid ISI implementation in the transmission system, and stopband filter use for the high attenuated stop-band filter. In the case of time domain pulse shaping, RCF is the optimum option. The RRC is given by (5) and (6).

$$h_{rc}(f) = \begin{cases} 1 & \text{for } |f| < 2W_0 - W \\ \cos^2 \left[\frac{\pi}{4} \frac{|f| + W - 2W_0}{W - W_0} \right] & \text{for } 2W_0 - W < |f| < W \\ 0 & \text{for } |f| > W \end{cases} \quad (1)$$

Where $W - W_0$ is surplus bandwidth and form this signal Raised Cosine falls above bandwidth W_0 . As a result, "Nyquist" pulse filters are pulses that have no ISI when sampled. The Nyquist pulse-shaping metric is satisfied with zero ISI[1]. The following relation

$$x(kT_c) = \delta(k) \quad (2)$$

Where

$$\delta(k) = \begin{cases} 1 & k = 0 \\ 0 & k \neq 0 \end{cases} \quad (3)$$

It was simplified as

$$\sum_{m=-\infty}^{\infty} X \left(f + \frac{m}{T_c} \right) = T_c \quad (4)$$

Where $X(f)$ is representing the Fourier transform signal

$X(t)$ and T_c the time period of the pulse[1]

$$h_{rc}(t) = 2W_0 \operatorname{sinc}(2W_0 t) \frac{\cos \left[\frac{2\pi(W - W_0)t}{T_c} \right]}{1 - \left[\frac{4(W - W_0)t}{T_c} \right]^2} \quad (5)$$

Another α is the ROF, it is determines the sharpness of the frequency response [1].

$$\alpha = \frac{W - W_0}{W_0} \quad (6)$$

The ROF shows how much power is used in RC emitters across a certain bandwidth W_0 . Therefore $0 \leq \alpha \leq 1$. When we use an RC pulse in conjunction with an equalization at the receiver, we can determine the sampling time.

$$h_{RC}(f) = h_{TX}(f) \cdot h_t(f) \cdot h_{RX}(f) \cdot h_E(f) \quad (7)$$

Where the transmission filter $h_{TX}(f)$, is the channel

frequency response is $h_t(f)$, filter is $h_{RX}(f)$, the equalizer is $h_E(f)$. Before they can be used, the transmitter and receiver filters need to be constructed.

$$h_{RC}(f) = h_{TX}(f) \cdot h_{RX}(f) \quad (8)$$

and

$$h_E(f) = \frac{1}{h_t(f)} \quad (9)$$

2) Square Root Raised Cosine Filter:

The SRRC (12) filter, a more advanced RCF technique, is used here. The frequency response of the SRRC filter is unity gain at low frequencies and full gain at high frequencies. It's very frequent in communication networks. Whereas the SRRC channel is applied first, and then a matching filter is applied in the receiver portion [1].

The SRRC is given by (7) and (8)

$$h_{TX}(f) = h_{RX}(f) = \sqrt{h_{RC}(f)} = h_{SRRC}(f) \quad (10)$$

3) Additive White Gaussian Noise (AWGN)

AWGN is created by thermally dissipative electrical components emitting electrons. It can be represented as a Gaussian, zero-mean process. And a random signal is a combination of random noise variables and a signal that is distinct from the current one. as show in (9)(6), (7) and (14).

$$Z = \alpha + v \quad (11)$$

The noise is Gaussian in nature. Has the following probability distribution function:

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

The noise model is Gaussian, with a PSD $G_n(f)$ that is flat for all frequencies denoted as;

$$G_n(f) = \frac{N_0}{2} \quad (13)$$

SRRC pulses, according to (9) are extremely similar to Nyquist pulses, which have a finite bandwidth with PSD.

$$G_{SRRC}(\alpha, f) = \begin{cases} T_c & \|f\| \leq \frac{1-\alpha}{2T_c} \\ \frac{T_c}{2} \left\{ 1 + \cos\left[\frac{\pi T_c}{\alpha} (\|f\| - \frac{1-\alpha}{2T_c})\right] \right\} & \frac{1-\alpha}{T_c} \leq \|f\| \leq \frac{1-\alpha}{2T_c} \\ 0 & \|f\| > \frac{1-\alpha}{2T_c} \end{cases} \quad (14)$$

$$\int_{-\infty}^{\infty} G_{SRRC}(\alpha, f) df = 1 \quad (15)$$

The problem suggests that the PSD may be a bilateral spectrum. Such version noise is common in all digital communication systems, and it is the primary source of noise in many systems with an AWGN physical appearance. AWGN is utilized in practically every communication system to create a noise model, which we have simulated here to attain a suitable performance [1].

$$g(t) = \frac{4\alpha}{\sqrt{T_c}} \frac{\cos\left[\frac{(1+\alpha)}{T_c} \pi t\right] + \frac{T}{4\alpha t} \sin\left[\frac{(1+\alpha)}{T_c} \pi t\right]}{1 - \left(\frac{4\alpha t}{T}\right)^2} \quad (16)$$

4) Multistage filter.

Filters with a cut-off frequency of [5] one quarter of the [5] sampling frequency f_s are low pass FIR filters. and odd

symmetry around $\frac{f_s}{2}$ are known as half-band filters. As it

happens, over half of the coefficients are zero. These filters are quite useful. As it happens, over half of the coefficients are zero. These filters are quite useful. Almost half of the coefficients are zero, as it occurs. Because the pass band and stop band bandwidths are equivalent, these filters are suitable for decimation-by-2 and interpolation-by-2. These filters are common in signal systems because their zero coefficients make them computationally efficient.

The window method can be used to compute half-band coefficients. While the window technique may not always

offer the smallest number of taps for a given performance, it is beneficial in understanding half-band filters. The MATLAB 2019a function FIR half-band can be used to create efficient equiripple half-band filters.

III. RESULT AND DISCUSSION

A. Error Rate Calculation:

The help of this block to compute the error rate by making a comparison between receiver data with the cost of multi-filter, Nyquist filter, and SRRC filter[1]. Which is after filtering and scatted signal arrange one place and error-free data in this system.

B. Transmit pulse shaping in Raised cosine filter

Through transmit filtering using a nyquist pulse, the formation symbol α_m with a symbol period T can be transmitted without inter symbol interference (ISI) by using nyquist pulse [1].

$$g(t) = \frac{\sin(\pi t / T)}{\pi t / T}, \quad t = -\infty \text{ to } \infty \quad (15)$$

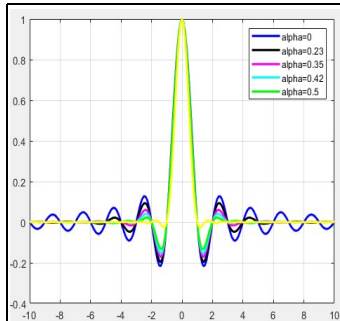


Fig. 2:-Time Domain Pulse shaping filter

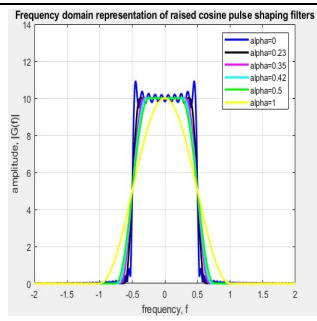


Fig. 3:- Alpha 0.23, 0.35, 0.42 & 0.5 using Frequency Domain

With the help of higher cosine filtering for pulses, we acquire results in a narrow range limited to frequencies between [1]

$$-\frac{1}{2T} Hz \quad \text{and} \quad +\frac{1}{2T} Hz$$

Using Matlab 2019a, I plotted the time domain and frequency domain illustrations of the RCF for various values of, i.e. $\alpha = 0.27, \alpha = 0.35, \alpha = 0.42, \text{ and } = 1$. We found that the filter tail of the RCFr with a value larger than 0 fades off quicker in time domain samples. As can be observed from the frequency response, the filter response is Band limited only until for values larger than 0. When compared to Nyquist bandwidth, we require a bigger bandwidth to transport the wave from. Time domain waveform of RC pulse shaping

shown in fig.(2) and Frequency domain waveform of RC pulse filter shown in fig.(3).

C. Comparison between RRC and SRRC filters Impulse response

A standard RCF and a SRRCF are compared in this case. In a cascade setup, an ideal normal RC pulse-shaping filter is comparable to twice the number of an ideal SRRCF [1]. As a result, when a FIR filter is convolved with itself, the impulse response should be comparable to that of an RCF.

Creating a standard RCF with a Roll of factors 0.23,0.35, 0.42 and 0.5 means this filter spans four symbols and contains three samples for each symbol. It cut short the impulse response outward from the largest after converting the square-root filter with itself.

D. Specification of raised cosine filter

The most significant limitation of an RCF is its RoF which, in a roundabout manner reveals the filters transmission capacity. The number of taps on a perfect RC channel is infinite. Following that all valid RC channels are windowed. The Window Length property of the Channel Span in Symbols property is used to regulate the window length. The window length is expressed in terms of six symbol lengths i.e. the channel ranges six picture terms. A bunch delay of three symbol terms is also included in this sort of channel. RCF is a pulse shaping technique in which the signal is upsampled. As a result we need also provide the upsampling factor. To obtain the desired channel characteristics we use the cosine transmit filter Framework object and specify its attributes. This item is a straightforward depiction of a standard polyphase FIR filter with a unity energy. The Nyquist system samples PerSymbol, or Nyquist system samples PerSymbol+1 taps are arranged on the channel. The Gain property was used to normalize the filter coefficients such that the filtered and unfiltered data matched when compared [1].

E. Roll-off factor with Square Root Raised cosines filter

The filtered output signal received by changing the roll-off factor from input sampled signal 0.5 (blue curve) to 0.23 (red curve[2]), 0.35 (yellow curve), 0.42 (gray curve) .The turning point of 0.5(blue curve) filtered signal has greater than for 0.23(red curve), 0.35(yellow curve) and 0.42(gray curve).

The raised cosine filtering used to divide into the filter for transmitting and receiver signal and this transmitting and receiving signal used to be square root raised cosine filter[6] which output is minimum ISI.

This plotting figure shows the up-sampled and filtered transmitting signal where square-root raised cosine filters are used. And[6] we have not destroyed the filter output of transmitted signal (magenta curve)

The default unit of energy for arranging data in a database to ensure that the gain of a normalized raised cosine filter and the[6] gain of a mixer of transmit and other side raise cosine

filter arrange again and convert to normalized signal from receiver. The receiving filter signal almost similar to the single raised cosine filter signal on which curve is blue signal[6].

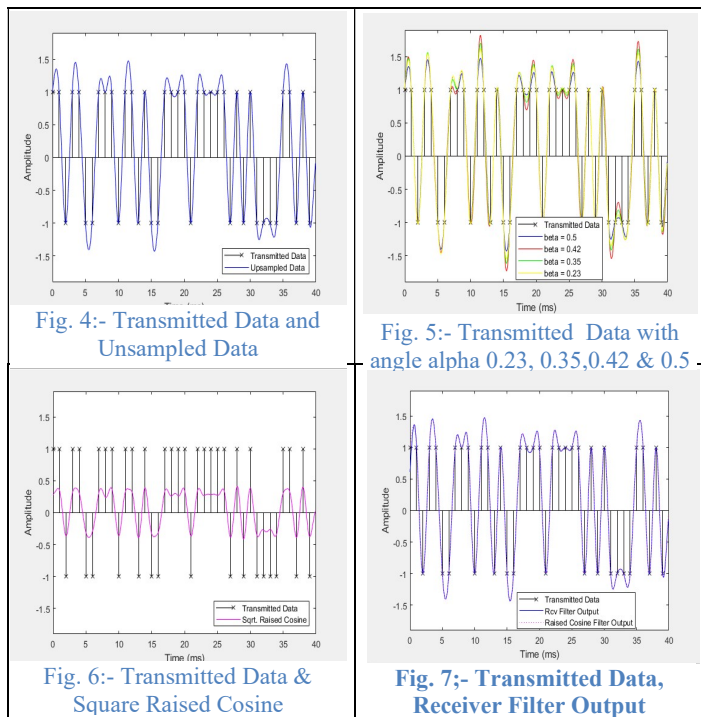


Table-1 Shown Implementation cost comparison between C1, C2, C3, & C4. Where C4 Number of coefficient is 97, Number of State is 96, Multiplication Per 97 and Additions per input. 96.

Cost	Num Coefficients	Num States	Multiplications per	Additions per input
C1	97	8	97	85
C2	97	96	97	96
C3	97	96	97	96
C4	97	96	97	96

F. Pulse shaping with Raised cosine filter

Now bipolar information structure is created. As a result we profile the waveform using the RCF rather than providing ISI. In this graph, the digitised data and the injected signal are compared. The channel's bunch delay (Nyquist symbol/(2R)) delays the channel's top reaction, making comparison of the two signals difficult.

By inserting (Nyquist symbol/2) zeroes after input x we level all the beneficial tests out of the channel. We may compare the RCF group delay by delaying the input signal. At the present

determining how the raised-cosine-filter up-samples and channels the signal at the input test timings is simple. The RCF's ability to band restrict the signal while avoiding ISI is the reason behind this. Shape in normal with RoF 0.23,0.35,0.42 and 0.5 filter span in symbols 6 output sample per symbol 8 and gain is 1[1].

G. Square root cosine filter (SRRCF)

Filtering is separated between the transmitter and receiver portions. the RCF was utilized. We used SRRC channels in both portions. It's possible that the RCF with the lowest ISI will emerge from a combination of transmitter and receiver channels. To partition the filtering across the transmitter and receiver sections, the RCF was used. SRRC channels were used in both portions. It's possible that the mix of transmitter and receiver channels will result in an RCF, which will have the lowest ISI.

To illustrate the SRRC filter, change the form to Square root. At that point, the receiver sorts the sent signal (maroon bend). Standard unit energy normalization guarantees that the transmit and receive channel configurations are detected. Data that has been submitted as well as upsampled data. Data with betas of 0.23, 0.35, 0.42 and 0.5 was supplied. The Square as well as the Transmitted Data Cosine have been increased. Transmitted Data Receiver Filter Output and output of a raised cosine filter Using a normalized RCF pick-up of alpha 39 and alpha 26 data was sent. The sifted acquired flag is represented by the blue bend at the receiver-section shown, which is nearly indistinguishable from the signal sifted with a single RCF[1].

H. Use pulse shaping of on various QAM signal

On a Wide Range of QAM Signals, Pulse Shaping Several variants of QAM are utilized in the execution of pulse shaping with the help of the matched filtering technique and a combination of SRRC filters. In addition to the forward error correction (FEC) method to the communication line, the increase cosine function provides filters, and the results may be regarded as an improvement in BER performance. Adding FEC to the communication connection to the pulse-shape filtering in various kinds of QAM. That is a 16 QAM, 64QAM, 128QAM AND 256 QAM [1].

A binary data stream is used in a communication link that includes a baseband modulator, channel demodulator, pulse shaping, and match filters. This exhibition shows the transmission of random data in a specific stem plot. In received side constellation diagrams, incoming signals are formed with a bit error rate (BER) [1]. Applying the RRC filter with 256 QAM in 0 dB power after upsampling the signal by the oversampling factor to clean the filter, do the following: the up fir function adds zeros to the end of the upsampled signal. The channel is then applied by the function. We transformed the ratio of energy per bit to noise EbNo to an SNR value using the number of k and the sample space system[1],[4].

In this situation, we use filtered down sampling from the receiving signal to upsample the broadcast signal. Now that the SRRC filter is in use, half of the symbol's filter length is added to the delay. The filter length is equal to the receiving signal. 16_QAM =39 dB, 64_QAM =39 dB, 128_QAM =39 dB, and 256_QAM =39 dB, impulse response

I. Eye pattern the filter effect

We first reduced the Eb /N0 s value, then used 16-QAM and 256-QAM filter effects to reconstruct the received data in an eye pattern. In a cascade, an RCF is employed as a same pair of filters, such as 256-QAM and 512-QAM. Due to a high SNR of the signal and no kind of multipath effects, the ISI reduction output data in a matched pair of pulse shaping RRC filters[1] is not zero –ISI until the second RRC filter is attached in this system. The noisy eye pattern signal reveals ISI as a narrowing after matched receive filtering and pulse shaping.

When filtering is used, the constellation diagram of the received signal is shown, as well as when filtering is not used. Multiply the received signal by the square root of the number of samples per symbol to equalize the transmit and receive power levels[1]. Here we show the scatter plotting diagram of the data mode modulated signal and also show the receiving noisy signal after the channel.

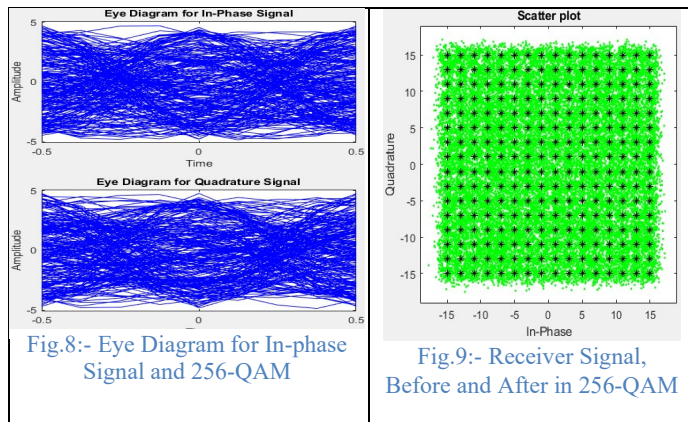


Fig.8:- Eye Diagram for In-phase Signal and 256-QAM

Fig.9:- Receiver Signal, Before and After in 256-QAM

The filter effects on the eye pattern We initially reduced the Eb /N0 value setting and regenerated the received data in an eye pattern fig.(8) the filter effects of 256-QAM and cascade as a matched pair of 256-QAM. ISI reduction output data in matched pair of pulse shaping RRC filters is not zero –ISI until the second RRC filter is connected in this system due to a high SNR of the signal without any form of multipath effects. After matched receive filtering and pulse shaping, the noise eye pattern signal shows ISI as a narrowing[7]. Again rearranging the scattered signal as shown in fig.(9).

The result shown of different QAM in receiver side before and after filtering 8-QAM in fig.(10), 16-QAM in fig.(11), 128-QAM in fig.(12) and 512-QAM in fig.(13).

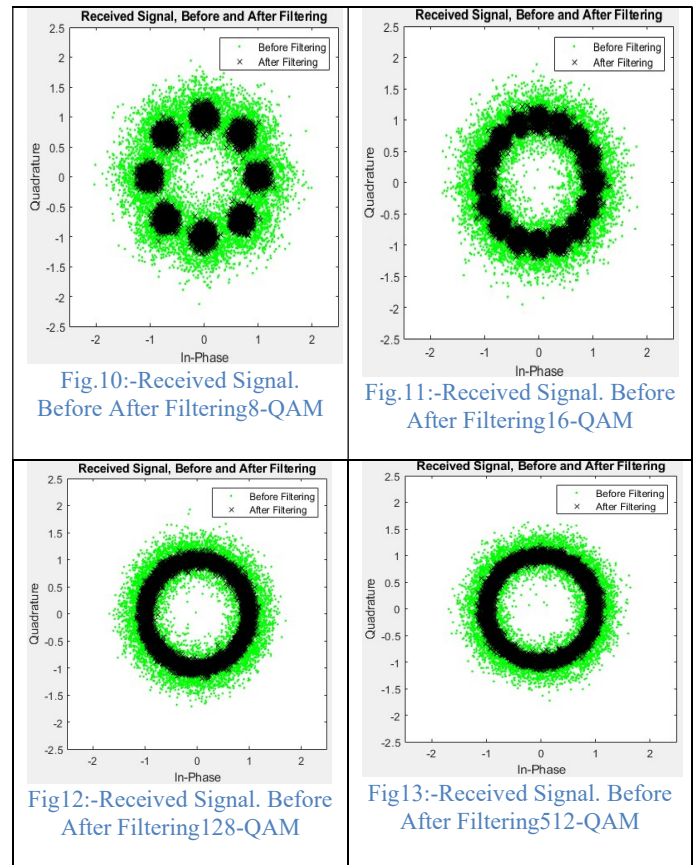


Fig.10:-Received Signal, Before After Filtering8-QAM

Fig.11:-Received Signal, Before After Filtering16-QAM

Fig.12:-Received Signal, Before After Filtering128-QAM

Fig.13:-Received Signal, Before After Filtering512-QAM

In this Table-2 shown different QAM with Roll off factor 0.23, 0.42 which is Error estimation with Binary code, Gray code and Bit error rate. The best result is 256- QAM with Binary 4.03e-02 and error detect 9638.

Table -2							
S	QAM	Roll off factor	Binary Error	Error	Gray Error	Bit Error	Error
1	16 QAM	0.23	4.17e-05	10	2.92e-05	7	2.50e-05
2	16 QAM	0.35	4.17e-05	10	2.92e-05	7	1.67e-05
3	16 QAM	0.42	4.17e-05	10	2.92e-05	7	1.25e-05
4	64 QAM	0.23	7.81e-03	1874	4.84e-03	1162	5.19e-03
5	64 QAM	0.35	7.81e-03	1874	4.84e-03	1162	5.11e-03
6	64 QAM	0.42	7.81e-03	1874	4.84e-03	1162	5.00e-03
7	256 QAM	0.23	6.87e-02	16495	3.95e-02	9487	4.03e-02
8	256 QAM	0.35	6.87e-02	16495	3.95e-02	9487	4.02e-02
9	256 QAM	0.42	6.87e-02	16495	3.95e-02	9487	4.03e-02

J. Pulse shaping different between Multistage, Nyquist and raised cosine filter

Now we used a raised cosine filter, multistage filter and Nyquist filter for shaping the waveform without ISI and generating bipolar data sequences . The plot is difficult to compare three digital signals and interpolated signals because the group delay of the filter delayed the response of peak point.

The sampled input signal is with input sample time and this shows a raised cosine filter to band-limit signal without ISI.

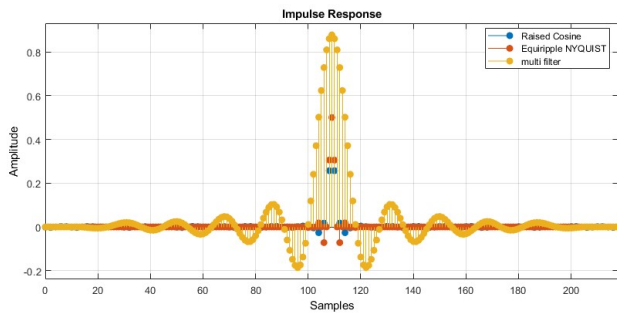


Fig. 14:- Impulse response between Raise Cosine, Nyquist, Multi stage.

K. Using Multistage filter with Magnitude

Using the multistage filter applied in the signal process, which is create low pass, high pass, band pass, in default system, which created with limitation conditions. Generally it is designed for a special purpose which is to specify the number of stags to be used in this design and automatically generate it for filtering the signal. Then it allows the optimal number of stages while minimizing the cost of using the resulting filter. It also specifies an integer for the design algorithm that minimizes the cost for the number of stages we want. It compares all filters with raised cosine and square root raised cosine filters. It is used for digital transmission systems in pulse shaping and also used under interpolation, decimation and filter banks.

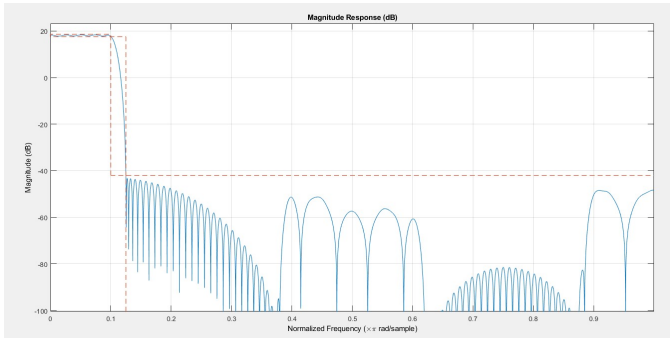


Fig.15:- Magnitude Response in dB Normalize frequency

L. Using Nyquist, Raised cosine filtering & multistage filter.

The output response of the equiripple Nyquist filter, multistage filter and increased cosine filter is shown in the plotting fig.(15) above. An ideal equiripple stopband filter with a bigger stopband attenuation filter and transition width is an equiripple Nyquist filter. The roll off factor for all filter orders is 0.23. This raised-cosine filter is made by cutting off the analytical impulse response at the end.

In this Nyquist filter we can use square-root raised- cosine spectral factor filters with the filtering application. A square-root filter fixed on both transmitter and receiver of the end. The decrease of different roll-off factor (RoF = 0.23, & RoF=0.42)shown in fig.(16) & (17) improves the response of raised-cosine filters and this is caused by containing the

direction of the maximum radiation frequency response rectangular window which is used in cutting off the end of the impulse response.

Now we can design a windowed impulse response method and cut off frequency response with a multi filter .Due to side lobe increase more than multi filter, raised- cosine design and also requires order raised- cosine filter and equiripple to meet the stopband specs.

The polyphase all pass filters have multipliers, which allows for an efficient implementation of that polyphase branch.The input is delivered directly to the interpolation filter but raise cosine filter is more better then multi filter and Nyquist filter shown in fig(16). The capability of rejection system is intersymbol interference and splitting the raised cosine filter between transmitter and receiver with filter system object (comm. Raised Cosine Transmit Filter and comm. Raised Cosine Receive Filter). The roll-off factor of raised cosine filter parameters indirectly specifies bandwidth filter[8]. An ideal raised cosine filter has an infinite number of taps. raised cosine filter used for pulse shaping on which upsample signals are used for specification.The object designs a polyphase FIR filter through the energy order of $N_{sym} * \text{sampsPERSym}$ or $N_{sym} * \text{PerSym} + 1$ taps . In an overlaid situation we can apply the gain property to normalize the coefficient filter and unfiltered data

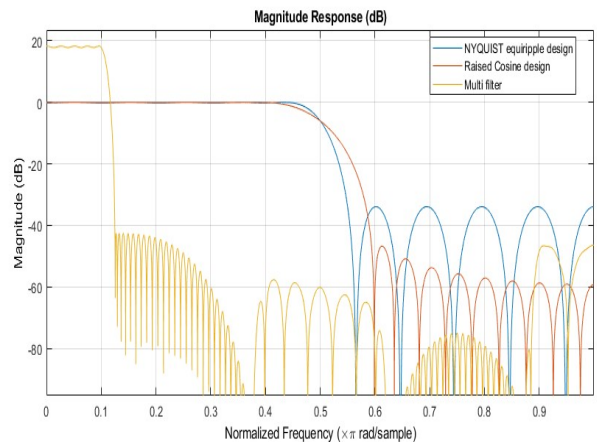


Fig. 16:- Roll off factor 0.42 using Magnitude Response in Normalized frequency-(13 dB) by Nyquist, raised cosine and Multi filter

M. Kaiser window design with Equiripple filter, multi filter & raise cosine filter

Kaiser window design for modified discrete cosine transform which length $N+1$ the formula apply when $2N$ and Constructi” d_n ” satisfies with equiripple filter, multi filter & raise cosine filter as shown in fig.(17) in roll of factor 0.42 and 39 dB power of normalized frequency.

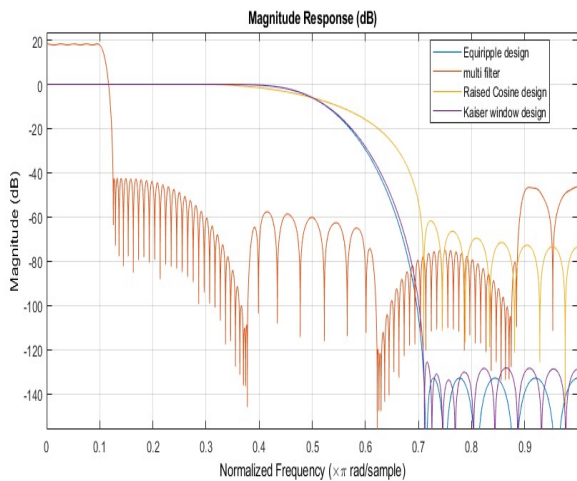


Fig. 17:- Roll off factor 0.42 using Magnitude Response in Normalized frequency-(39 dB) by Nyquist, raised cosine and Multi filter

Comparing magnitude response in 13 dB with Kaiser window filter, Nyquist filter, multi filter and raised cosine filter in normalized frequency as shown in fig.(18).

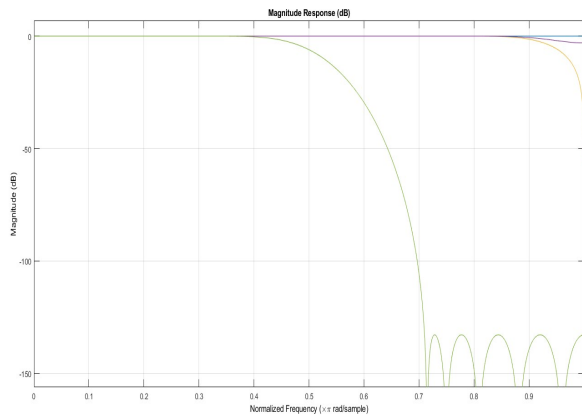


Fig. 18:-Fig. 8:- Magnitude Response in Normalized frequency-(13 dB) by Kaiser window filter, Nyquist filter, raised cosine filter, and Multi filter

N. Computation cost

In the table below, the computational costs of polyphase FIR interpolation and polyphase FIR decimation filters are compared.

IV. USING MULTISTAGE FILTER, NYQUIST FILTER, KAISER WINDOW FILTER & SRRC FILTER TECHNIQUES OF ACHIEVE EFFICIENT DESIGNS IN POLE AND ZERO

.In the receiver side filter order using 40 which uses Multistage filter, nyquist filter, kaiser window filter & SRRC filter techniques. As a result, pole-zero shown in fig(19) and it centered shown 39 is the best performance is SRRC

filter.

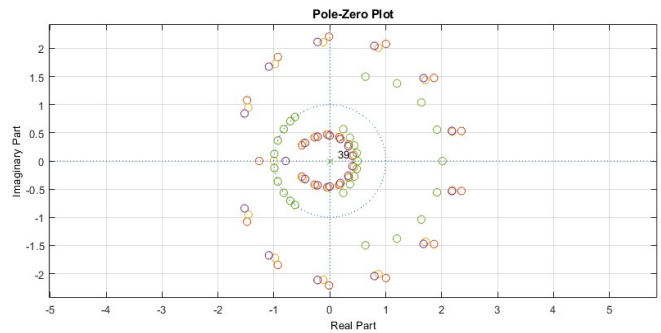


Fig.19:- Pole & Zero Plot

V. ERROR RATE CALCULATION

By comparing received data to the delayed form of transmitted data, this block aids in the computation of the error rate. The block's output is divided into four sections, each of which contains the following components: first, the error rate; second, the error rate; third, the error rate, and fourth error rate. The number of symbols that are compared in total. Table-1 shows C4 Number of the coefficient is 97, Number of State is 96, Multiplication Per 97, and Additions per input. Table-2 shows 256-QAM with Binary 4.03e-02 and error detect 9638.

VI. CONCLUSION

Transmission and receiver side QAM (Quadrature Amplitude Modulation) with SRRC (Square Root Raise Cosine) filter. In the transmission system, four signals travel in at the same time in a multiplication and addition system in one bandwidth, while the incoming signal in the receiver side has detected in simulation in MATLAB 2019a system at a lower cost per time, a lower BER (Bit Error Rate), and a lower SNR (Signal to Noise Ratio), and uses less power with high data rate. On the receiver side, we pass through the Multistage filter, Nyquist filter, and SRRC filter. As a result best filter which is QAM with SRRC filter shown in this paper using "ESTIMATING THE LOW COST PROBABILITY ERROR IN QAM USING THE SRRC FILTER WITH OTHER FILTER"

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