Intersymbol Interference Reduction Technique in Wireless Channel

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Abstract - Digital Communication technologies provide the top-grade communication speed and larger data capacity. Though the Digital communication is the world-class, but it is also affected by noise and non-linearities. These noises cause disturbances in the channel making the data distort or lost. Among these interruption Inter Symbol Interference (ISI) is the most superior and disastrous phenomenon that cause huge data loss. This M-Tech dissertation provides the information on how the Inter Symbol Interference is caused and how it causes the data loss. Also, in order to reduce this ISI, a unique type of filters called Finite Impulse Response pulse shaping filters are widely used in digital system. In this M.Tech dissertation various pulse shaping filter has been realized and has been shown how they are improving the performance of digital communication.

Keywords: Roll of factor, Mean, Standard Deviation, Grey level, Sinc function, Bandwidth.

I. INTRODUCTION

The digital communication systems are fast advancing techniques that are providing the world with the quicker access to information and exchange of data in a larger scale. The line that is used to communicate within the transmitter and the receiver is generally through open air medium. This channel consist of many natural phenomenon and other naturally present signals. When the message signals are passed over such channels the data is likely to be corrupt and lost during the communication process. One such interruption is the Inter symbol Interference (ISI). In order to reduce this fault in communication the pulse shaping filters are designed. Among different pulse shaping filters the Finite Impulse Response Moving Average Filter is widely used in digital communication to reduce ISI. In this M.Tech dissertation, the causes for Inter Symbol Interference, effect of ISI and the different types of pulse shaping filters that are used to reduce the ISI are mentioned.

II. WIRELESS CHANNEL AND ISI

In communication systems, data is transmitted as binary bits (ones and zeros). It is easier to carry out a binary system using switches, where turning on a switch symbolizes '1' and turning it off symbolizes '0'. Such simple binary systems necessary represent ones and zeros as rectangular pulses of finite duration (say τ seconds). A rectangular pulse of finite duration τ manifests itself as an infinitely extending sinc pulse in the frequency domain (see figure below). This implies that a rectangular pulse requires infinite bandwidth if it is not to be distorted during its transmission. Due to the dual nature of Fourier Transform, the following figure is valid if we alternate the frequency and time domain representations. That is, a symbol with finite bandwidth will extend infinitely in time. This implies that to send a single '1' or '0' or a series of them (for a multi-level signaling), you would need infinite time duration. This is absolutely impractical. In practical terms, signals will not extend infinitely forward and backward in time. But it will definitely be non-zero after the time duration τ. This implies that the residues of adjacent symbols/signals overlap with each other giving rise to Inter Symbol Interference (ISI). If the residual energy from the adjacent symbol is very strong, it becomes impossible to distinguish the present symbol and there is a possibility of it being misinterpreted altogether. To avoid or reduce this effect, "Pulse Shaping" techniques are used to make sure that the data carried by the symbols are not affected by the overlapping effect of adjacent symbols. In a band-limited system, when we try to enhance the data rate, it may lead to Inter Symbol Interference . There are two criteria that must be satisfied for a non-interference system when pulse shaping is employed.

(1) The pulse shape attribute a zero crossing at the sampling point of all pulse intervals

(2) The shape of the pulses is such that the amplitude decreases rapidly outside of the pulse interval.

A rectangular pulse satisfies the first standard (where it contains zero crossing – see figure above) but not the second criterion (the energy of the rectangular pulse does not decrease rapidly outside the pulse interval and in fact it expand to infinite bandwidth). Pulse shapes filters like raised cosine filters, square root raised cosine filters and matched filters are employed to shape the transmitted pulses so that they will satisfy the above two criteria of providing an ISI free system.

III. LITERATURE SURVEY

Various technique and research papers has been made in improving the performance of digital communication system. We gathered information from the following research paper published by different authors. The information gathered from research papers is as follows-

Beaulieu, N.C.[2],"The evaluation of error probabilities for intersymbol and cochannel intervention", Communications, IEEE proceedings on (Volume:39 , Issue: 12). In this paper author derived an efficient series that can be used to calculate the chance of error in a binary co-channel with inter symbol interference and additive noise but the accuracy of the results is bounded for Gaussian noise.

Pauluzzi D.R.[3],"A comparison of SNR approximation techniques for the AWGN channel", Communications, IEEE Transactions on (Volume:48 , Issue: 10). In this paper author compared the performances of several signal-to noise(SNR) appropriate techniques to identify the "best" estimator. The mean square error is used as a measure of execution. In addition to compare the relative performances, the absolute levels of performance are also established. Some known estimator structures are modified to execute better on the channel.

Chengshan Xiao[4], "New Add-of-Sinusoids Simulation Framework for Rayleigh and Rician Fading Channels", Wireless Communications, IEEE Transactions on (Volume:5 , Issue: 12). In this paper the author proposed an statistical reference model for the simulation of Rayleigh fading channels and Rician fading channels. It is shown that the probability density function of the Rician fading stage is not only independent of time but also uniformly distributed over [-pi, pi].The statistical properties of the new simulators are confirmed by extended simulation results, showing best agreement with theoretical analysis in all cases.

Alagha, N.S.[5], "Cramer-Rao bounds of SNR estimates for BPSK and QPSK modulated signals", Communications Letters, IEEE (Volume:5 , Issue: 1). In this paper the author derived Cramer-Rao lower bounds (CRLBs) for the approximation of signal-to-noise ratio (SNR) of binary phase-shift keying (BPSK) and quaternary phase-shift keying (QPSK) modulated signals. The bounds are compared to those for data-aided calculations (known symbols at the receiver). It is shown that at low Signal-Noise Ratio there is a significant difference between the bounds for non-data-aided and data-aided appraisal.

Wook Hyun Kwon[6],"A Receding horizon Kalman Finite Impulse Response filter for discrete time-invariant scheme", Automatic Control, IEEE proceedings on (Volume:44 , Issue: 9). Here author presented a receding horizon Kalman FIR filter which is a combination of Kalman filter and the receding horizon strategy. It is shown here that the suggested filter possesses many good inherent property irrespective of any horizon given condition.

Ursini.L.[7],"Enhancing Chaotic Communication execution by Manchester Coding", Photonics Technology Letters, IEEE (Volume:20 , Issue: 6).Here author presented an optical disorganized transmission system, based on the synchronization of two chaotic lasers. The performance are analyzed in terms of the Q-factor, considering two unlike message modulation formats: non return-to-zero and Manchester coding. The Manchester coding shows increased performances due to the transfer of the signal spectrum to higher frequencies.

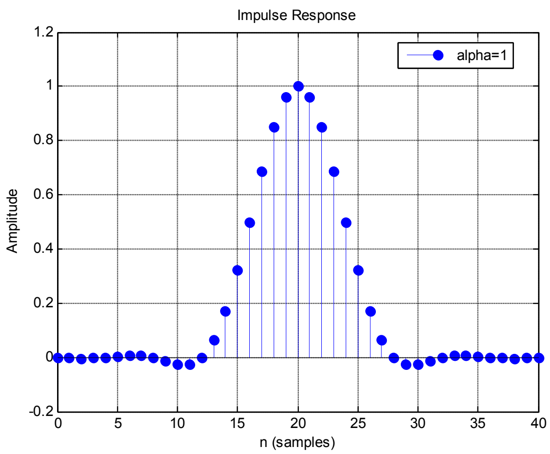
IV. SIMULATION/EXPERIMENTAL RESULTS

4.1 Raised Cosine Filters/Pulses:

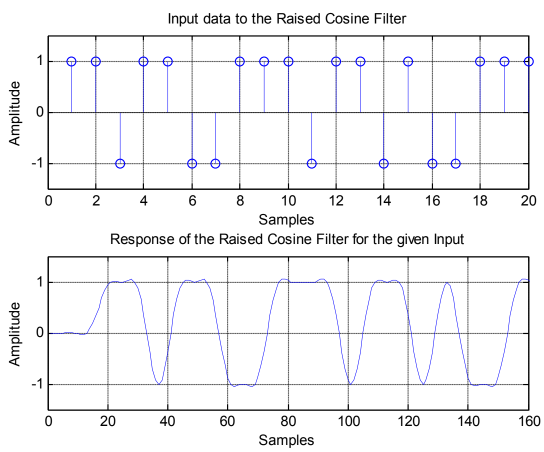
A Raised Cosine looks more like a modified sinc pulse in time domain and is given by the function (This equation is apt for digital domain and Matlab simulation, it is obtained from its analog form by substituting "t" by n*TS).

$$p(t) = \begin{cases} \frac{\pi}{4} sinc\left(\frac{nT_s}{\tau}\right), & if \left(1 - \left[\frac{2\alpha nT_s}{\tau}\right]^2\right) = 0 \\ \frac{cos\left(\frac{\alpha\pi nT_s}{\tau}\right)}{1-\left(\frac{2\alpha nT_s}{\tau}\right)^2}, & if\ n = 0 \\ \frac{sinc\left(\frac{nT_s}{\tau}\right) cos\left(\frac{\alpha\pi nT_s}{\tau}\right)}{1-\left(\frac{2\alpha nT_s}{\tau}\right)^2}, & otherwise \end{cases}$$

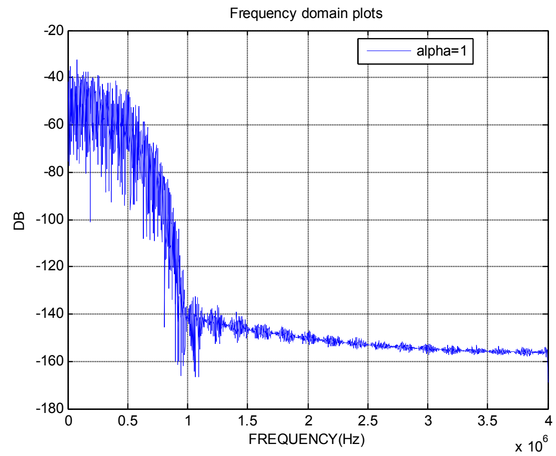
Here TS is the sampling period, n is the sample number, α is a parameter that governs the bandwidth contained by the pulse and the time at which the tails of the pulse decay. A value of α = 0 offers the narrowest bandwidth, but the slowest rate of decline in the time domain. When $\alpha = 1$, the bandwidth is $1/\tau$, but the time domain tails decay rapidly.



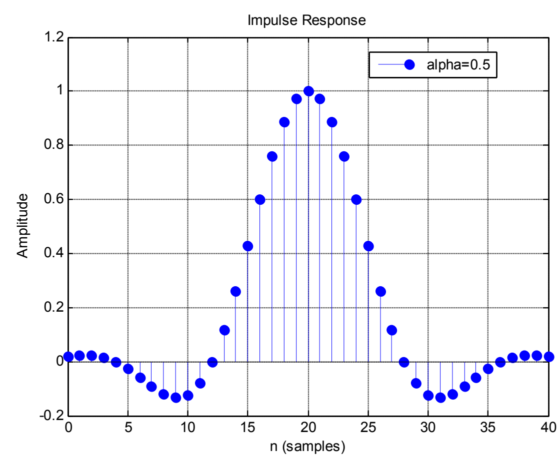
The above figure shows the impulse response of Raised Cosine signal at alpha =1 where α is the Roll of Factor. As we seen the curve at the sides contain some interference in the form of noise. The given signal is passed through the Raised cosine filter and the corresponding output is obtained.



Here the given impulse response is input to the Raised Cosine Filter at alpha =1 and corresponding output is obtained for the given no. of sample. At the output we can see that the curve we get has sharp cut off in the time domain. Now the same process is repeated for the different values of alpha.



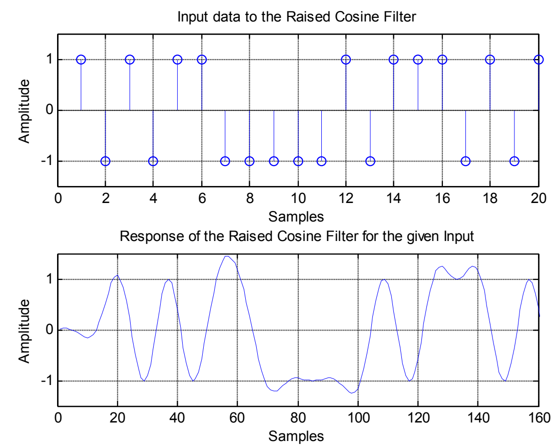
The above figure shows the frequency response of the output at alpha=1. Since we get much improved response in time domain so we get slightly distorted response in frequency domain.

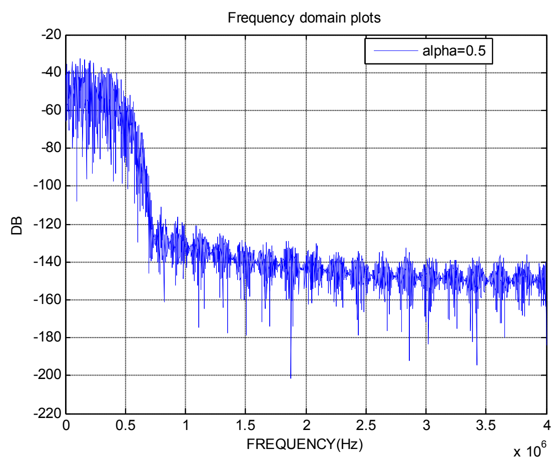


The above figure represent the impulse response of Raised Cosine signal at alpha = 0.5. In this impulse response the interference obtain at the side is more as compared to at alpha = 1.Again we passed this through the filter and see the deviation as compared to previous one.

Alpha =0.5

These figure shows the input of impulse response to Raised Cosine Filter at alpha = 0.5 and the corresponding output for the given no. of samples in time domain. Thus we can see that the output we get does not get have sharp cutoff as what we get at alpha = 1.

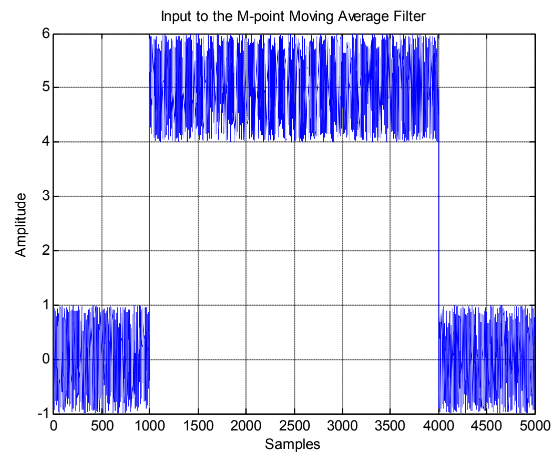


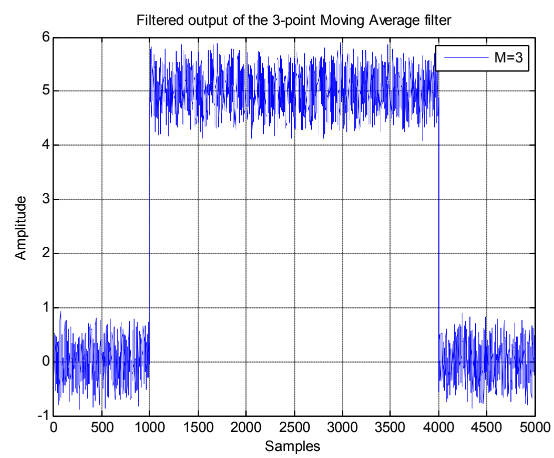


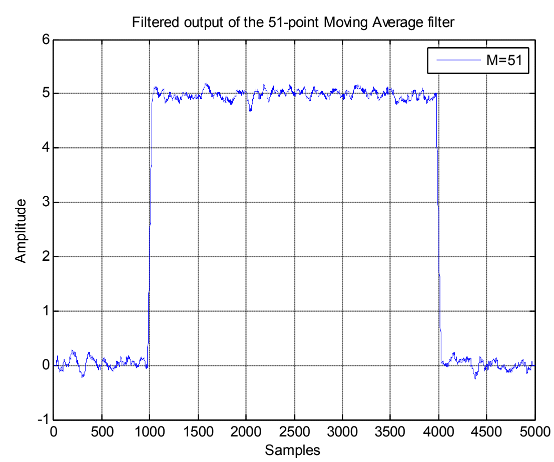
The above figure gives the frequency response of the output at alpha=0.5.Since the output response in time domain is not sharp thus the frequency response what we get contain is much improved as compared to previous one what we get in frequency response at alpha = 1. Since time and frequency has inverse relation.

4.2 Moving Average (MA) Filter

The moving average filter is a simple Low Pass FIR (Finite Impulse Response) filter usually used for smoothing an array of sampled data/signal. It takes M samples of input at a time and takes the average of those M-samples and produces a single output sample. It is a very simple LPF (Low Pass Filter) structure that comes handy for scientists and engineers to filter undesirable noisy components from the intended data. As the filter length increases, (indicated by the parameter M) the smoothness of the output also increases, whereas the sharp transitions in the data are made raising blunt. This implies that this filter has excellent time domain response but a poor frequency response.





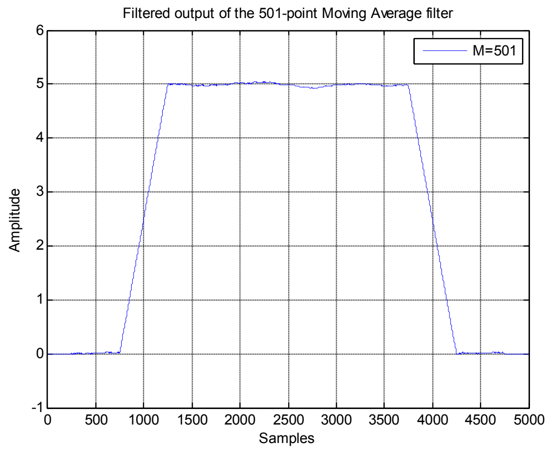


The MA filter performs three important functions:

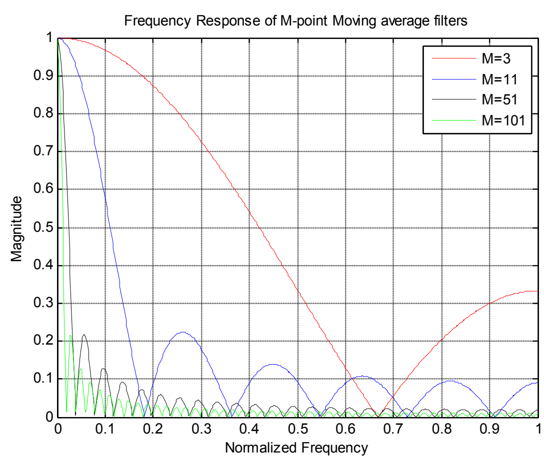
1) It takes M input points, computes the average of those M-points and gives a single output point.

2) Due to the computation/calculations involved, the filter introduces a definite amount of propagation delay.

3) The filter acts as a Low Pass Filter (with poor frequency domain response and a good time domain response).

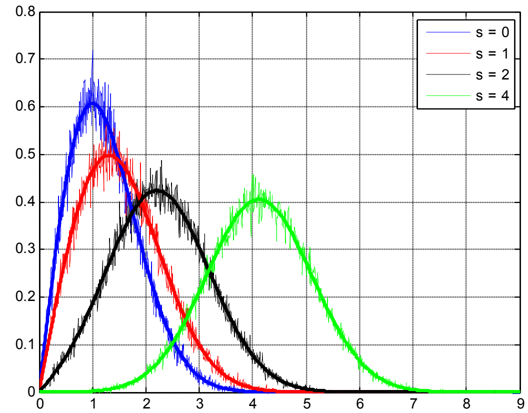


The above figures represent the input to moving average filter and the corresponding out for the different values of filter length (denoted by parameter M) in time domain. We say that the output obtain for the value of M=3 has sharp cutoff as compared to other values of M =51 and 105. Thus as the value of M goes on increasing the curve gets deviated and we get distorted output. At the same time we can also see that the noise is more at M =3 and less at M= 105.Hence we have to choose that value of M for which we get slightly sharp cutoff and least value of noise.



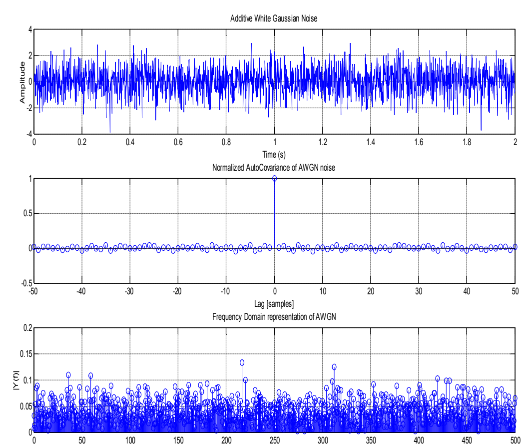
This figure represent the frequency response of M-point Moving average Filter for different values of M.

4.3 Racian fading

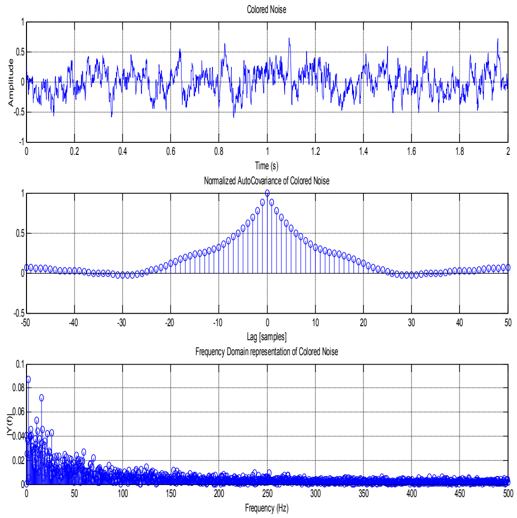


The above figure gives Rician Fading. Rician fading is a stochastic model for radio propagation unusual caused by partial cancellation of a radio signal by itself — the signal arrives at the receiver by many different paths (hence have multipath interference), and at least one of the paths is changing (lengthening or shortening). Rician fading arises when one of the paths, typically a line of sight signal, is much stronger than the others. In Rician fading, the amplitude gain is represented by a Rician distribution

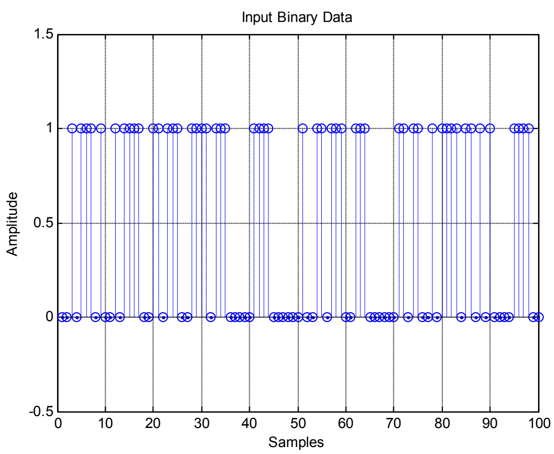
Generation of noise

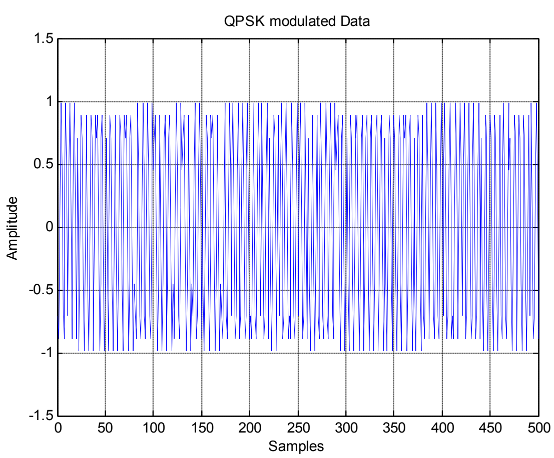


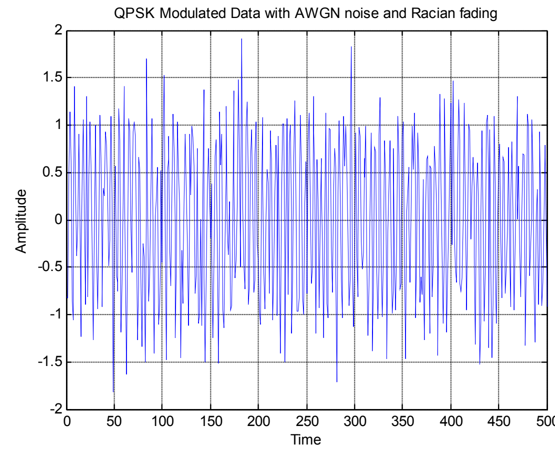
The above figure gives the generation of Additive White Gaussian Noise, its Normalized Auto Covariance and Frequency Representation for given no. of samples



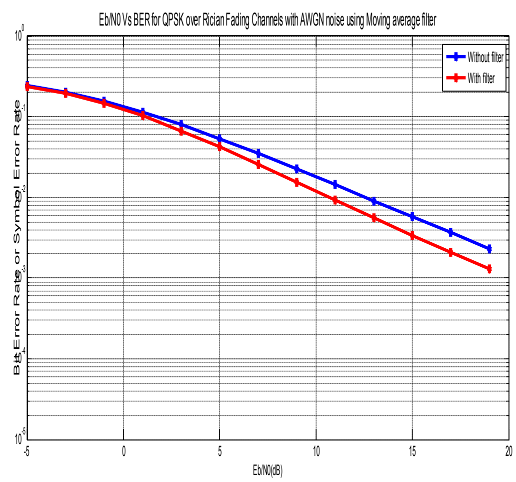
The given figure represent the Colored Noise, its Normalized Auto Covarience and Frequency Domain Representation







In the above figures the input binary data acts as a input to QPSK modulation technique and the corresponding modulated data output is obtained. This modulated data combine with noise and Fading and received at the receiver. This will then passed to moving average filter and the corresponding output is obtained which contain low intersymbol interference. This is checked by Bit Error Rate which is low when we passed it through filter which is shown below:



The above graph represent Eb/No Vs QPSK over Rician Fading Channels with AWGN noise. The Red line indicates with Moving Average Filter and Blue line indicates without filter. Thus from the above figure we can say that BER decreases with filter as signal to noise ratio increases.

V. CONCLUSION

Inter symbol Interference (ISI) is a form of distortion of a signal in which one symbol interrupt with subsequent symbols. This is an undesirable phenomenon as the previous symbols have similar effect as noise, thus making the communication least reliable. The spreading of the pulse beyond its allotted time interval causes it to interrupt with neighboring pulses.[1] ISI is usually caused by multipath propagation or the inherent non-linear frequency output of a channel causing successive symbols to "blur" together. The presence of ISI in the system introduces noise in the decision device at the receiver output. Therefore, in the design of the transmitting and receiving filters, the aim is to minimize the effects of ISI, and thereby deliver the digital data to its destination with the smallest error rate possible.

VI. FUTURE SCOPES

- We can use adaptive control technique to reduce Inter symbol Interference.
- We can also use Error Detection And Error Correction Technique.
- We can also use ARMA (Auto Recursive Moving Average) technique which is a combination of Auto Recursive and Moving Average Filter.

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