Qam Receiver For Rayleigh Fading Channel

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***Abstract*: Rayleigh fading is a statistical model for the effect of a propagation environment on a radio signal, such as that used by wireless devices. The M state QAM contains M possible states for the signal. It can transmit k bits of information during each symbol period. The binary data stream is pre-processed before the modulation process. Then it performs a bit-to-symbol mapping. Binary data is then gray coded and then modulated by modulator. The modulated signal is up-sampled and filtered by square root raised cosine filter in the transmitter. Due to the channel the signal transmitted gets corrupted by the disturbances present in it. Further the received signal is filtered by the square root raised cosine filter to remove a portion of the signal in account for the filter delay in order to make a BER comparison. At receiver the signal received is down sampled and equalized by the RLS equalizer. The filtered signal is demodulated to obtain the transmitted signal.**

Keywords: Equalization; RLS algorithm; style

I. INTRODUCTION

Wireless communications is one of the fastest growing and emerging field in the modern life. While developing the systems for communication the available permissible power, inherent noise level of the system and bandwidth are the important constraints which is to be considered. The digital modulation techniques are preferred more when compared to the analog modulation techniques because of the error free capability. The performance of M-QAM over Rayleigh fading channel model and additive white Gaussian noise channel model depends on effective channel equalization technique. Inter symbol interference (ISI) could be a limiting factor in several communication systems [1]. Large stream of literature has been devoted to the analysis of adaptive RLS algorithm. Most papers deal with those variants characterized by an algorithm. However, such algorithms asymptotically behave as the standard RLS so that adaptivity is lost in the long run [2]. It is possible to transmit more number of data by choosing higher order of modulation schemes. To attain high-speed reliable communication, channel estimation and equalization are necessary to beat the effects of ISI [3]. In order to satisfy the higher demand for quality of services, the wireless communication systems require larger data transmission rates [4]. This paper mainly deals with the effect of RLS equalizer for QAM signals at the receiver. The BER curve is improved when compared to other papers observation or results.

II. QUADRATURE AMPLITUDE MODULATION

The Quadrature Amplitude Modulation or QAM is a form of modulation which is used widely for modulating the data signals onto a carrier which is used for radio communications. It is widely used because of its advantages over other forms of data modulation schemes such as PSK (Phase Shift Keying), ASK (Amplitude Shift Keying), FSK (Frequency Shift Keying). The QAM is a signal in which two carriers shifted by 90 degrees in-phase are modulated which results in the variations of both phase and amplitude. In view of the fact that both phase and amplitude variations are present it can also be considered as mixture of phase and amplitude modulation [1].

In 16 QAM, there are four I In-phase values and four Q Quadrature values which results in 16 possible states or the voltage levels for the signal. It can transmit four bits of information per symbol which consists of two bits for I and two bits for Q. The symbol rate is defined as one fourth of the bit rate. So this modulation scheme produces a more spectrally efficient transmission as compared to other. It is more efficient than QPSK (Quadrature PSK) and BPSK (Binary PSK).

III. DATA TRANSMISSION

The figure (1) describes the processing of binary data stream using a communication system which consists of modulator, pulse shaping filter, channel, root raised cosine filter and demodulator.

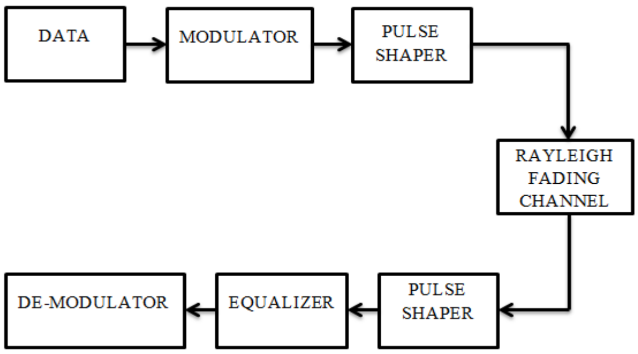

Fig 1. Block diagram

The systems bit error rate is computed for the signals which are transmitted and received. This system uses baseband 16-

QAM as the modulation scheme and Additive White Gaussian Noise (AWGN) as the channel model. The conventional way of representing a signal in MATLAB is either vector or a matrix form. In this system a column vector is used which contains the values of a binary data stream.

a. Generating Data

QAM transmits k bits of information that is it transmits k bits per symbol. To modulate a signal using digital modulation with M symbols start with real message signal whose values are integers from 0 to M-1. The MATLAB function implements a rectangular M-ary QAM modulator. By default the configuration is such that the object receives integers between 0 and 15 rather than 4-tuples of bits in figure (2). In the case of Gray code the two adjacent code numbers differs by only one bit from each other.

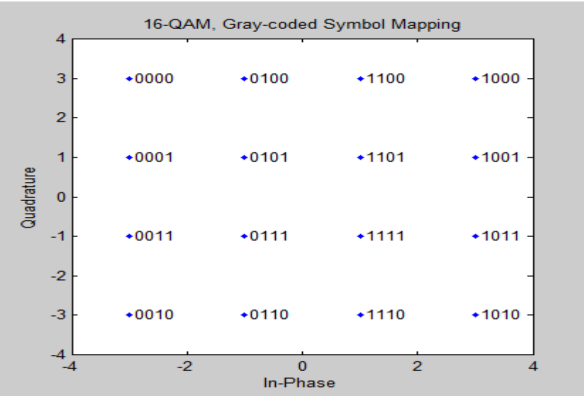

Fig 2. Gray coded symbol mapping

B. Modulation of the Data

The conversion of discrete time data signal into continuous time analog signal is called as modulation. Modulation is a process that encodes source information onto a carrier signal for the optimized transmission. During the modulation process, the analog or digital signal source is mapped onto the phase or amplitude (or both) of the carrier signal.

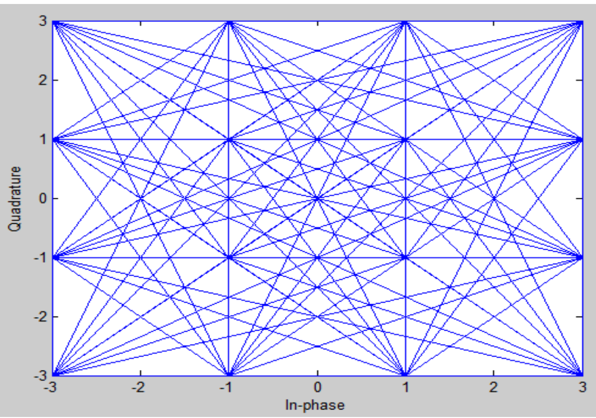

Fig 3. Data modulation output

The digital modulation also offers a number of advantages over its analog counterpart, such as accommodation to various digital signal conditioning, processing techniques, capacity and increased tolerance to channel impairments and noise. The figure (3) represents the modulated data. The modulator converts the symbols which represents the message into a continuous time analog waveform.

C. Pulse Shaping

In the digital communication systems, digital information can be transmitted on a carrier through changes in its fundamental characteristics such as frequency, phase and amplitude. These transitions can be smoothened depending on the filters implemented in the transmitter. There are two important requirements for wireless communication channel which demands the use of a pulse shaping filter. These requirements are 1) reducing inter symbol interference (ISI) from the multi-path signal reflections and 2) generating band limited channels. Both requirements can be accomplished by a pulse shaping filter which is applied to each symbol. In fact, the sinc pulse, meets both of these requirements. The root raised cosine filter is generally used in series pairs, so that the total filtering effect is that of a raised cosine filter. The advantage is that if the transmit side filter is stimulated by an impulse, then the receive side filter is forced to filter an input pulse shape that is identical to its own impulse response, there by setting up a matched filter and maximizing signal to noise ratio while at the same time minimizing inter symbol interference.

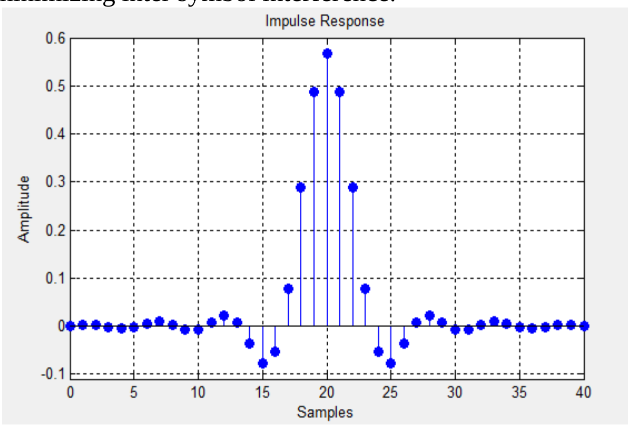

Fig 4. Impulse response

By applying a pulse-shaping filter to the modulated signal, the sharp transitions are smoothened and the resulting signal is limited to a specific frequency band. It is of course impossible to eliminate inter symbol interference at all times, but if it can be eliminated at the instant of sampling, then the problem can be dramatically reduced. The class of filters that satisfies the criteria are known as Nyquist filters. The impulse response of this filter is zero for all the values of T = nT_b, satisfying the Nyquist criterion.

$$\mathrm{sinC}\left(\pi \frac{t}{T_b}\right) = \frac{\sin\left(\pi \frac{t}{T_b}\right)}{\pi \frac{t}{T_b}} \qquad (1)$$

The Nyquist filter exhibits a sinc behavior for its impulse response, as shown in (1). The pulse shaping filter also has the desirable property that it behaves like a brick-wall filter in the frequency domain.

However, this ideal filter is physically unrealizable, since its impulse response is non-causal. In addition to it, its peak amplitude decays only as (1/t). The impulse response of the system is as represented in figure (4) and the magnitude response is as represented in figure (5). So one of the filter that satisfy Nyquist's criterion is the raised cosine filter type. The raised cosine filter has an impulse response that drops off much faster in time than the sinc response, at the expense of a small increasing bandwidth, and hence finds its applications in wireless communications.

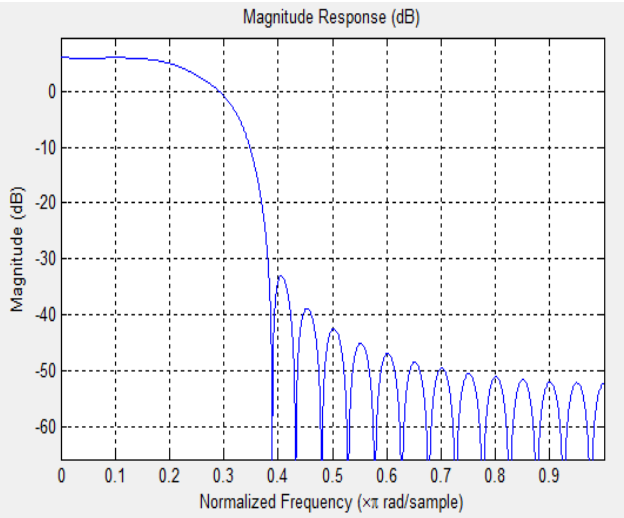

Fig 5. Magnitude response

D. Designing Channel

In the digital radio technologies, digital signals are transmitted in a bandwidth larger than the coherence bandwidth of the channel. This means that the channel is no longer "flat" across the frequency band; rather, the fading is "frequency-selective" with different signal strengths present at different frequencies across the band. Rayleigh fading is a model that can be used to describe the form of fading that occurs when multipath propagation exists. In any terrestrial environment a radio signal will travel via a number of different paths from the transmitter to the receiver. The term fading or small-scale fading means rapid fluctuations of the amplitudes, phases or multipath delays of a radio signal over a short period or short travel distance.

The received signal sample value x(t) at the sample time t, the receiver interprets it as the sum of the two components y(t) the first is the noise free component x(t) i.e. the sample value that would have been received at the sample time t in the absence of noise, as a result of the input waveform being passed through the channel with only distortion present and the second is the noise component n(t), it is assumed as independent of the input waveform. The time domain behavior of the noise, the received signal is as shown in (2).

$$y(t) = h.\ x(t) + n(t) \qquad (2)$$

Where x(t) is the original transmitted signal passed through Rayleigh fading channel and n(t) is the noise or the disturbance in the channel and h is a $e^{j\phi}$ represents the fading channel coefficient. The thermal noise is random in nature, so noise can't be deterministic otherwise it would have been subtracted the noise from the received signal y(t).

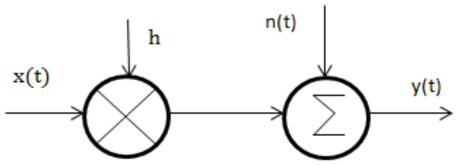

Fig 6. Channel representation

The figure (6) represents the channel. The random thermal noise has Gaussian distribution with zero mean and variance as that of noise power. If only the variance of Gaussian is high then it is not a good response so it is necessary to increase the power of x(t). The zero mean means the expected value of n(t) during any time interval t is zero. On an average n(t) will take zero value and it results in probability of n(t) = 0 is highest and probability rapidly decreases as the magnitude is increased of n(t) noise.

E. Filtering Received Signal

The receive filter block, filters the input signal using a normal raised cosine FIR filter or a square root raised cosine FIR filter. For this system the received signal is made to pass through the root raised cosine filter. It also down samples the filtered signal. This block normalizes the root raised cosine filter coefficients to unit energy. The rolloff factor should be between zero and one. The rolloff factor is a measure of excess bandwidth of the filter that is the bandwidth occupied beyond the Nyquist bandwidth of 1/2T, where 1/T is symbol rate. One of the method for achieving root raised cosine filter by taking square root of the raised cosine filter in the frequency domain and by using it in both the transmitter and the receiver part of the communication. The raised cosine filter derives its name from its shape in the frequency domain as in (3).

$$H_{rrc}(\omega)=\sqrt{H(\omega)_{rc}}=\sqrt{\tfrac{1}{2}\,(1+\cos \pi\omega/2\omega_c)} \quad |\omega|<2\omega_c \qquad (3)$$

When the transmitter and receiver filters are cascaded to obtain raised cosine filter transfer characteristics as in (4).

$$H_{rc}(\omega)=H_{rrc,tx}(\omega)\,H_{rrc,rx}(\omega) \qquad (4)$$

The equation (4) can also be expressed as (5).

$$H_{rc}(\omega) = \sqrt{H_{rc}(\omega)}\,\sqrt{H_{rc}(\omega)} \qquad (5)$$

The frequency response of a raised cosine filter can be described by (6). The raised cosine filter derives its name from its shape in the frequency domain.

$$H(f) = \begin{cases} T_s & 0 \le |f| \le \frac{1-\alpha}{2T_s} \\ \frac{T_s}{2}\left\{1+\cos\left[\frac{\pi T_s}{\alpha}\left(|f|-\frac{1-\alpha}{2T_s}\right)\right]\right\} & \frac{1-\alpha}{2T_s} \le |f| \le \frac{1+\alpha}{2T_s} \\ 0 & |f| > \frac{1+\alpha}{2T_s} \end{cases} \qquad (6)$$

Where α is the roll-off factor that determines the excess bandwidth of the filter frequency response and in the range between 0 and 1, whereas T_s is the sampling or the symbol period. A zero roll-off is essentially a sinc response. The filter response has a cosine roll-off shape in the range between (1- α)/ T_s and (1+ α)/ T_s which becomes strictly zero beyond (1+ α)/ T_s. The equation (7) and (8) is thus the bandwidth and the impulse response of the raised cosine response respectively.

$$BW = \frac{(1+\alpha)}{2} f_s \quad (7)$$

Where f = 1/ T_s is the symbol rate or the sampling rate. The corresponding filter impulse response is described by

$$h(t) = \frac{\cos\left(\pi\alpha \frac{t}{T_s}\right)}{1-\left(2\alpha \frac{t}{T_s}\right)} \frac{\sin\left(\pi \frac{t}{T_s}\right)}{\left(\pi \frac{t}{T_s}\right)} \quad (8)$$

It is clear that at the sampling instant t = nT_s, the impulse response is essentially zero except at n = 0. Therefore no inter symbol interference will be created. The peak amplitude of the raised cosine impulse response, with the modest value of α drops off much faster than the sinc response. This is due to more gradual roll-off of cosine shaping at the filter cutoff frequency.

F. Equalizing the Signal

In a high rate wireless communication system, a transmitted signal often reaches a receiver via more than one path due to reflection, refraction and scattering of radio waves by structures of the building. This results in a phenomenon known as multipath fading. Recursive Least Square (RLS) algorithm with a constant forgetting factor is often used to update the tap-coefficient vector for inter symbol interference free transmission. The forgetting factor [2] is used with the RLS algorithm for calculating the tap-coefficient vector in order to minimize the squared equalization error due to input noise and due to channel dynamics.

The RLS algorithm in figure (7) achieves the best steady state performance in a stationary environment. In the case of a non-stationary environment, the algorithm uses a forgetting factor $\lambda \in (0,1)$ to obtain only a finite memory in order to track slow statistical variations of the channel fading status [3]. The forgetting factor gives a larger weight to more recent data in order to cope with the channel dynamics. If $\lambda = 1$, all the data are weighted equally and the algorithm has an infinite memory length which is optimal with respect to suppressing the estimation noise effect alone.

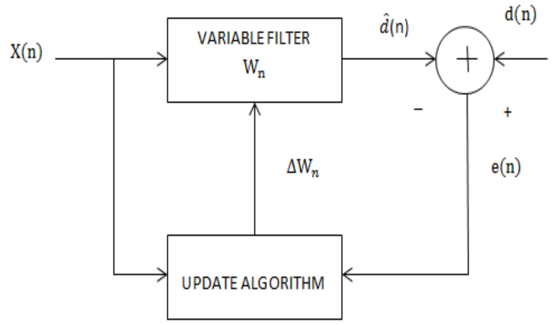

Fig 7. RLS equalizer

On other hand, with a smaller λ value the algorithm has a shorter memory length and is better adapted to channel dynamics. In other words, a larger λ will reduce the estimation noise and a smaller λ will reduce the equalization error due to lag effects.

The idea behind RLS filters is to minimize a cost function C by appropriately selecting the filter coefficients W_n, updating the filter as new data arrives. The error signal e(n) and desired signal d(n) are defined in the negative feedback in figure (7). The error implicitly depends on the filter coefficients through the estimate $\hat{d}$(n) in (9).

$$e(n) = d(n) - \hat{d}(n) \quad (9)$$

The weighted least squares error function C the cost function is desired to minimize being a function of e(n) is therefore also dependent on the filter coefficients in (10). Where $0 < \lambda \leq 1$ where is the "forgetting factor" which gives exponentially less weight to older error samples.

$$C(W_n) = \sum_{i=0}^{n} \lambda^{n-i} e^2(i) \quad (10)$$

The optimal λ value depends on channel fading dynamics and the extent of input noise effect on the equalization error. Similarly this algorithm is investigated for dynamically adjusting the forgetting factor to control the degree of the trade-off between the RLS algorithms dynamic tracking ability and its input noise suppression ability. The forgetting factor is chosen according to an inverse function of the residual power in order to achieve a constant weighted sum of the squares of a posteriori errors. The amount of forgetting at each step correspond to the amount of new information in the latest measurement ensures that the estimation is always based on the same amount of information.

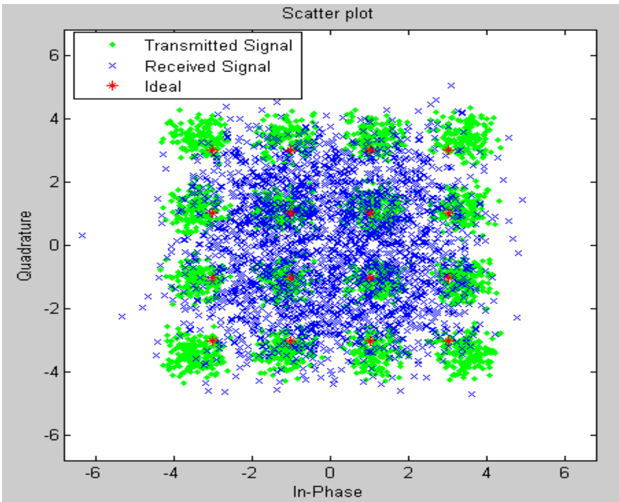

Fig 8. Before equalization

When the initial value of λ is set to unity and the error is small, it may be concluded that the estimator is sensitive enough to adjust to the variations of the system parameters and therefore to significantly reduce the estimation error. As a result it is reasonable to choose the forgetting factor close to unity. However when the estimator error is large the estimator sensitivity should be increased by choosing a smaller forgetting factor [4].

The figure (8) represents the scattering of data before equalization and figure (9) represents the equalized output of the signal which ensures the reduction of inter symbol interference

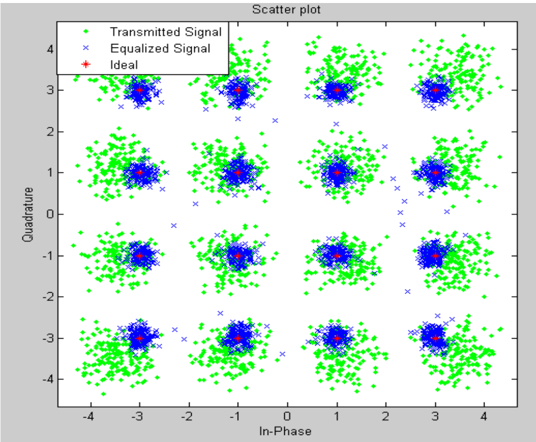

Fig 9. After equalization

G. Demodulate the Signal

The process of converting modulated signal into its original discrete form is called as demodulation. The demodulator converts the continuous time signal channel output into original form, from which the detector tries to estimate the transmitted message. As the signals enter the system, they are made to split and each side is applied to a mixer. One half has the in-phase applied by the local oscillator and the other half has the quadrature applied by the oscillator signal. The received signal and the signal from the local oscillator are mixed by the mixer. The mixer output is the input to the decision device. Recovering the phase of the carrier is important otherwise the bit error rate for the data will be compromised. The demodulation or channel decoding is the process at the receiver of converting the received waveform into a replica of the input bit sequence.

IV. BIT ERROR RATE PERFORMANCE

Noise disrupts the quality of communication between sender and receiver because the received noisy voltage samples can cause the receiver to incorrectly identify the transmitted bit, thereby generating a bit error. If a long stream of known bits of data is transmitted, the fraction of received bits that are in error, approaches the bit error rate, which is the probability that any given bit is in error P (error). The figure (10) represents the probability of bit error rate graphically. The vertical y axis represents bit error rate.

Downward direction on vertical axis represents lower BER. It is observed that as the ratio of energy per bit to spectral noise density increases, which results in the lower values of probability of bit error rate. Modulation schemes which are capable of delivering more bits per symbol are more immune to errors caused by noise and interference in the channel. The BER decreases with increase in the E_b/N_o value.

Increasing the E_b/N_o value means increasing the signal power with respect to noise energy. The error rate is increasing as the value of M increases.

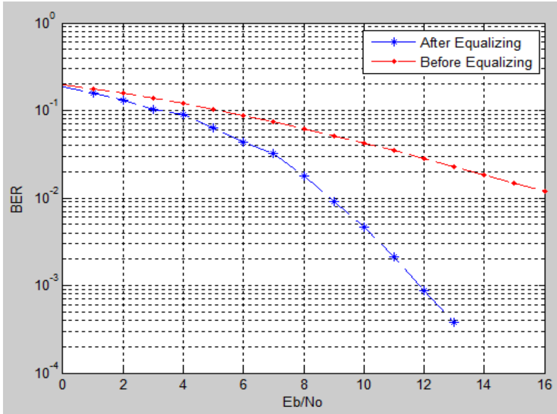

Fig 10. Probability of bit error rate graph.

The value of M increases means more number of bits is combined to make a symbol and these bits are packed more closely in signal constellation. Table (1) provides the bit error rate performance with equalizer and without equalizer for different E_b/N_o.

TABLE I. BER PERFORMANCE

$\frac{E_b}{N_o}$ (db)	BER (without equalizer)	BER (with equalizer)
0	0.1976	0.1976
2	0.157	0.1324
4	0.1199	0.08959
6	0.08784	0.04335
8	0.06199	0.01777
10	0.04237	0.00466
12	0.02825	0.0008821

V. CONCLUSION

The QAM appears to increase the transmission efficiency for radio communication by utilizing both the phase and amplitude variations. In order to obtain the transmitted signal in the receiving end, the inter symbol interference is tried to reduce. The use of RLS equalizer reduces the inter symbol interference for better communication. If the received signal is free from unwanted signals it would be easy to demodulate the signal to obtain the transmitted signal back.

ACKNOWLEDGMENT

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