Analysis and Synthesis of LMS and RLS Equalization under QAM Modulation Techniques

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Abstract: **The recent digital transmission system imposes channel equalization technique for transmission of more prominent data rate. In the process of recompensing the turbulent effects Channel equalization is referred, which spiels a crucial role for enabling more prominent data rate in digital communication. Channel equalizer technique is also crucial to eliminate or minimize the problem of Inter Symbol Interference in a band-limited channel. Active fields of research for example development of advanced algorithms, structures and the excerpt of the design parameters for equalizers, exploiting the benefits of different signal processing techniques. The various adaptive algorithms like LMS, RLS, with QAM modulation technique are used here. MATLAB software has been used for simulation.**

Keywords: **Channel Equalizer, Least Mean Square, Recursive Least Squares Algorithm.**

1. INTRODUCTION

In today's world, the credit goes to the development in digital communication technology for sea change in modern day living because the transmission of information has a tremendous impact. As we move towards a more information centric world, the requirement of user is higher data rate as well as efficient information signal over physical transmission channels increases with expanding the number of user over communication networks. In this case advanced services can be handled in a flexible way by more efficiently utilizing the available bandwidth.

Transmitted information may get spread and overlap over successive information signal transmitted at same time intervals due to the distortions (phase-delay variations) introduced in the communication channel. Such phenomenon is known as Inter Symbol Interference (ISI) [1]. In ISI, the transmitted symbols are subjected to other impairments in form of noise and non-linear distortions arising due to modulation and demodulation process, cross talk interference, the use of amplifiers and converters etc. To combat distortions introduced due to channel impairments and recover the transmitted symbols accurately by the signal processing techniques on received signal at the receiver end are referred to as equalization schemes [2].

Introduce an inverse filter into the receiver to equalize [3] the channel is a common means of overcoming this problem [1, 3-5]. This offers a great solution provided the channel transfer function is known and its inverse is convergent. However, in non-minimum phase channels (zeros outside the unit circle in the z-plane), such an inverse will be unstable. The structures, the learning algorithms and the use of training sequences are the basic characteristics of adaptive equalizers. Adaptive equalizer is more preferable than static equalizer in case of time varying information [6-19]. Adaptive equalizers are required in Bandwidth efficient data communication. Equalizers are used to overcome the effects of the distortion occurred due to channel.

The rest of this paper is organized as follows. In section 2, the basic concept of adaptive equalizers and different adaptation algorithms are discussed. In section 3 Results of different adaptive equalizer are discussed. In Chapter 4 consists of conclusion on the basis of different adaptive algorithm.

2. CHANNEL EQUALIZER

Channel equalization is an efficient way of extenuating the prejudicial impressions because of a poor selectivity of frequency and/or dispersive and noisy communication link for transmitting the information between transmitter and receiver. Adaptive Channel Equalizer is the technique to minimize the problem of Intersymbol Interference (ISI) [5]. In tracking, adjustment of decision equalizer can be achieved by slow variation of channel response. In this paper we mainly focus on LMS and RLS algorithms. These algorithms are used to enhance the performance of adaptive equalizer as compare to other equalizer algorithms.

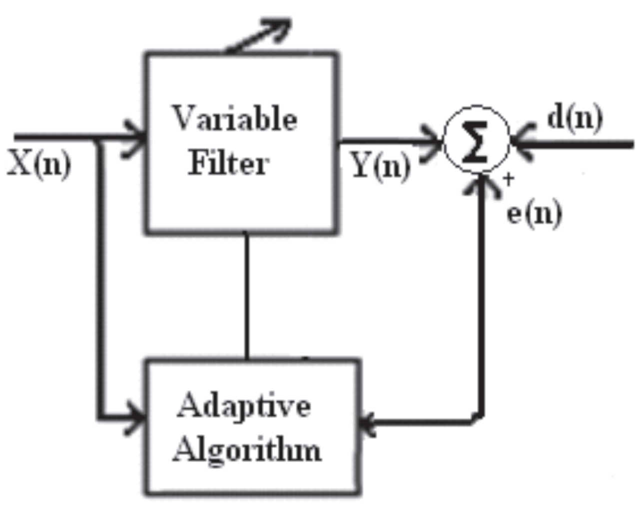

Fig.1: Adaptive filter[22]

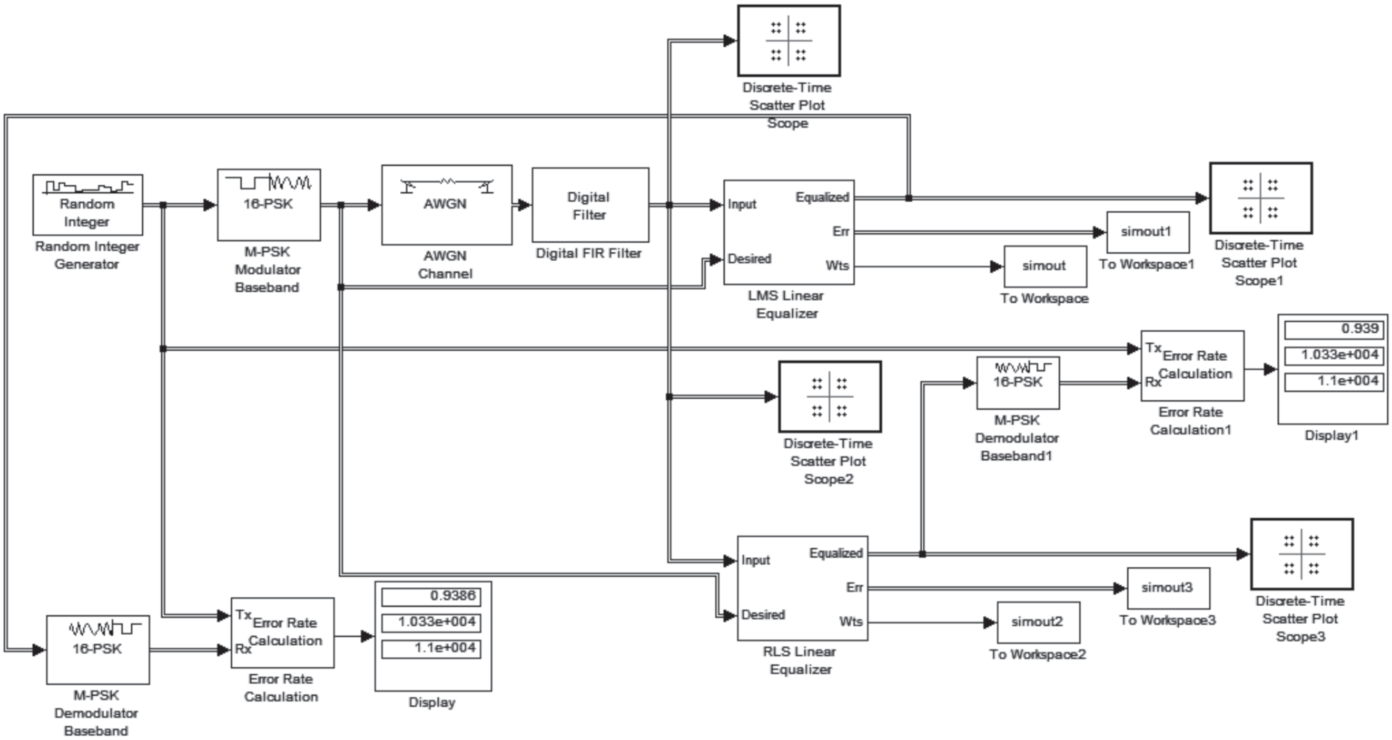

Fig.2: Block Diagram LMS and RLS Adaptive equalizer for PSK Modulation Scheme

A. *LMS Algorithm*

Least mean squares (LMS) algorithms are a class of adaptation filter algorithm. This algorithm is used to calculate the filter coefficients for design of a desired filter that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal). LMS filter is a transversal (i.e. tapped delay line) structure. Advantages of this LMS filter are easy to design and performance of this filter is highly effective.

B. *RLS Algorithm*

Recursive least squares (RLS) [15] algorithm is an adaptive filter algorithm which recursively calculate the coefficients of filter to minimize a weighted linear least squares cost function as compare to the input signals. Recursive least squares algorithm is used to reduce the mean square error. RLS adaptive filter is used a deterministic signal as input signals whereas LMS adaptive filter used stochastic signal as input signals. The RLS exhibits extremely fast convergence as compare to other adaptation filter [15]. This is the main advantages of RLS algorithm at higher computational complexity and potentially poor tracking performance

3. RESULT

In this section we show the scattered plot for LMS and RLS adaptive algorithm for QAM Modulation technique. In fig.3 we show the scattered plot between In-phase Amplitude and Quadrature Amplitude of transmitted data. This scattered data is passed through LMS equalizer block. Fig.4 shows the scattered plot of transmitted data after passing through equalizer. In this figure this transmitted data is equalize to resolve the problem of inter symbol interference problem (ISI).

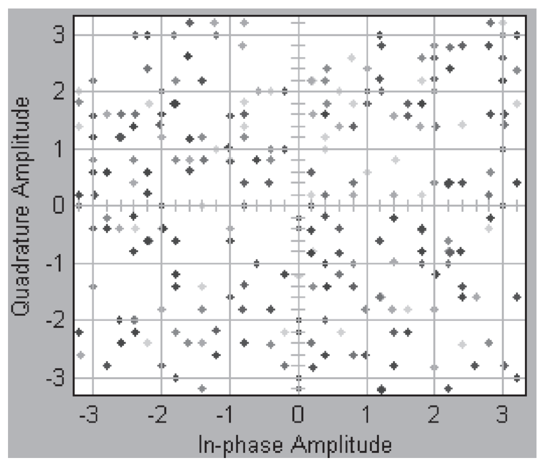

Fig.3: Scatter plot of transmitted data for LMS algorithm using QAM modulation

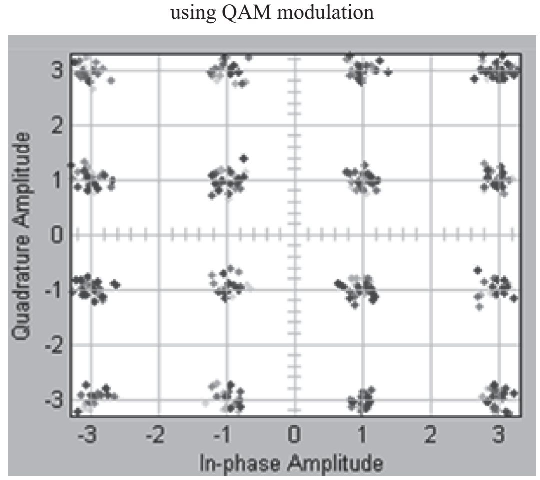

Fig.4: Scatter plot after equalization of data for LMS algorithm using QAM modulation

In fig.5 we show the scattered plot between In-phase Amplitude and Quadrature Amplitude of transmitted data. This scattered data is again passed through RLS equalizer block. Fig.6 showed the output of equalizer. Error Rate in variable step size is less as compare to other equalization algorithm

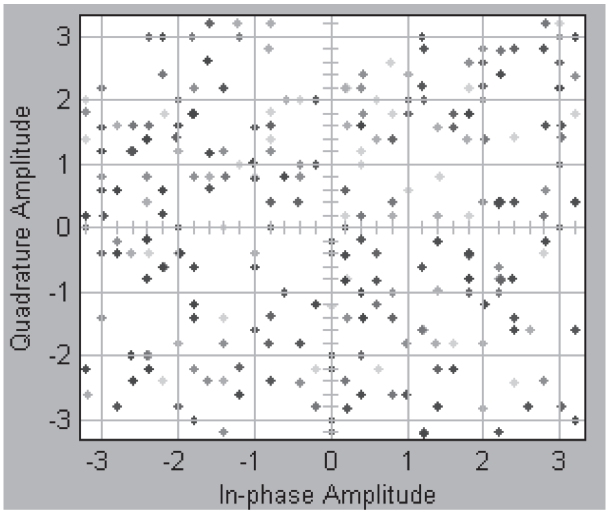

Fig.5: Scatter plot of transmitted data for RLS algorithm using QAM modulation

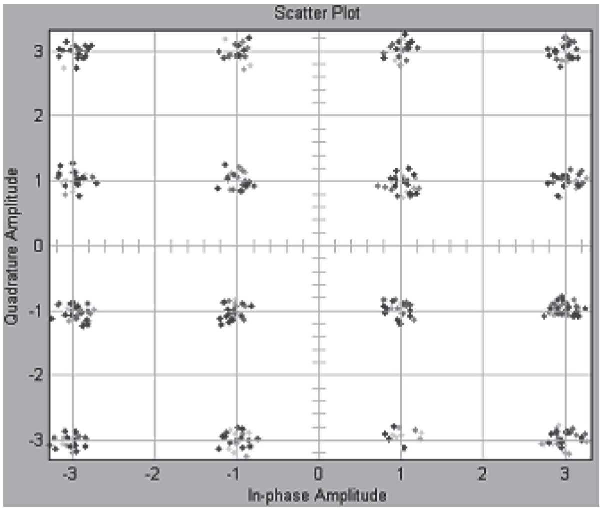

Fig.6: Scatter plot after equalization of data for RLS algorithm using QAM modulation

Table 1: Error rate comparison for QAM Modulation scheme

Error rate comparison for different Equalizer in QAM Modulation scheme			
	Error Rate	**Number of Error Detected**	**Total Number of symbols compared**
LMS	0.939	1.03E+04	1.10E+04
RLS	0.938	1.03E+04	1.10E+04

4. CONCLUSION

This work has extended to check the performance of various adaptive equalizers under various modulation schemes. In this work we mainly constraint on Quadrature Amplitude Modulation (QAM) Modulation scheme. We are using different algorithm to check the better equalization for the reduction of intersymbol interference problem. In this paper generally discuss Least Mean Square (LMS), Recursive Least Square (RLS) algorithms and check the performance on QAM modulation schemes. Table 1 shows the result of error detection in transmitted data and error rate for all adaptive algorithms for QAM Modulation scheme. Error Rate in RLS equalization is less than LMS equalization algorithm for QAM modulation scheme.

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