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Effect of BER Performance in RLS Adaptive Equalizer

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Abstract

Wireless communication systems operating over time-varying fading channels require adaptive signal processing to equalize the channel variations at the receiver. In wireless applications, the received signal is typically affected by frequency-selective fading and channel equalization is required to mitigate the resulting inter symbol interference (ISI). In this paper an adaptive model has been proposed for a digital communication system based on RLS algorithm with binary input signal. Also, the LMS (Least mean Square), RLS (Recursive Least Square) structures are simulated for linear and nonlinear channels. Convergence characteristics, along with bit-error-rates are analysed for better performance of these equalizers than the standard equalizers.

Keywords

Digital communication, Channel Equalization, BER, LMS, RLS.

1. Introduction

In modern digital communications, the channel equalization plays an important role in compensating channel distortion. But, most channels have time varying characteristic and their transfer functions change with time. Furthermore, time-varying multipath interference and multiuser interference are the major limitations for high speed digital communications. One of the major limiting factors is inter symbol interference (ISI). ISI may be due to one or more of the factors like: frequency selective characteristics of the channel, time varying multipath propagation that is prominent in mobile communication, carrier phase jitter, symbol clock residual jitter and limited channel bandwidth. ISI is a major problem for single input single output (SISO) and single input multiple output (SIMO) channels. In case of multiple input multiple output (MIMO) channels, multiple access interference (MAI) or co-channel interference (CCI) is also equally problematic. This thesis deals with SISO and MIMO channels. Adaptive channel equalization is a major issue in digital communication systems. Adaptive equalizers are used to reduce channel disturbances such as noise, ISI, CCI and adjacent channel interference (ACI), nonlinear distortions, fading, time-varying characteristics of channels, etc. Equalizers are usually used to compensate the received signals which are corrupted by the inevitable noise, interference and signal power attenuation introduced by communication channel during transmission [2]. Traditionally liner transversal filters (LTF) [1] are commonly used in design of channel equalizers. It is mentioned in [4] that the bit error rate (BER) and mean-square-error (MSE) performance of MLP is much better than LTF. The linear equalizers, however, fail to work well when transmitted signals have encountered severe nonlinear distortion. Most of the work for channel equalization using standard methods as well as adaptive method s has been done [5-11].

2. Chanel Equalization

Unfortunately most often the digital transmission of information is accompanied with a phenomenon known as inter symbol interference (ISI). Briefly this means that the transmitted pulses are smeared out so that pulses that correspond to different symbols are not separable. Depending on the transmission media the main causes for ISI are:
Cable lines – the fact that they are band limited;
Cellular communications – multipath propagation.
For a reliable digital transmission system it is crucial to reduce the effects of ISI and it is where the adaptive equalizers come on the scene. A processor unit carries out filtering process, such as amplitude equalization, delay equalization, line equalization, etc., in accordance with parameters read from a memory.

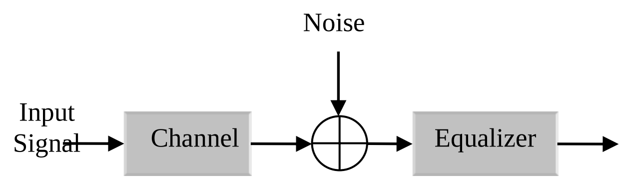

Fig 1: Block of Fixed Equalizer

An adaptive equalizer is an equalization filter that automatically adapts to time-varying properties of the communication channel. It is frequently used with coherent modulations such as phase shift keying, mitigating the effects of multipath propagation and Doppler spreading. Many adaptation strategies exist. A well-known example is the decision feedback equalizer, a filter that uses feedback of detected symbols in addition to conventional equalization of future symbols. Some systems use predefined training sequences to provide reference points for the adaptation process. Adaptive equalizers are used to reduce channel disturbances such as noise, ISI, CCI and adjacent channel interference (ACI), nonlinear distortions, fading, time-varying characteristics of channels, etc. Fig 1 shows the block diagram of the system used to carry out the study.

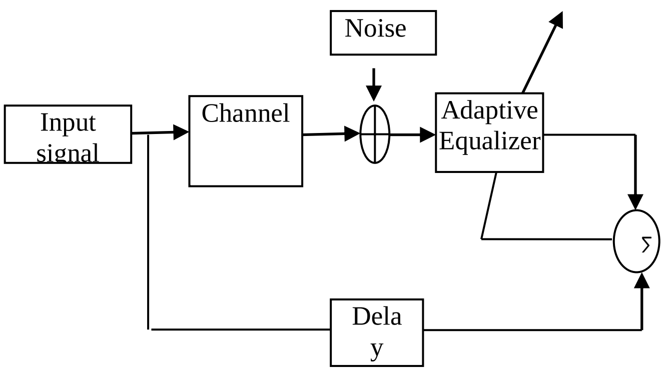

Fig 2: Block diagram of Adaptive Equalizer

3. Proposed Channel Model

The following are different channel models [3] are studied and verified.

Model 1: A linear time invariant channel model is a non minimum phase with transfer function $C(Z) = C_0 + C_1 Z^{-1} + C_2 Z^{-2}$, where the channel impulse response is $C = [C_0\ C_1 C_2]^T$ and $C = [0.3482\ \ 0.8704\ \ 0.3482]^T$. For the purpose of time varying channel the transfer function is modelled as

$$C(Z) = (C_0 + a_0(k)) + (C_1 + a_1(k))Z^{-1} + (C_2 + a_2(k))Z^{-1} \quad (1)$$

Model 2: A Non linear channel often used in satellite communication is $r(k) = g(r^k + v(k) = r(k) + 0.2\left(\hat{r}\,(k)\right)^2 + v(k)$ (2)

Model 3: The transfer function of a discrete time channel model is described by

$$C(Z) = a_0(k) + a_1(k)Z^{-1} + a_2(k)Z^{-2} \quad (3)$$

The non linear channel is modeled as

$$y(n) = a_0 x(n) + a_1 \tanh^2(x(n-1)) \quad (4)$$

where a_0 and a_1 are non-linear constants.

The proposed model considered in this work is Linear Transversal Filters (LTF) with time varying properties. Recursive Least Squares (RLS) algorithm is used to find the filter coefficients that relate to recursively producing the least squares (minimum of the sum of the absolute squared) of the error signal (difference between the desired and the actual signal). This is contrast to other algorithms that aim to reduce the mean square error. The difference is that RLS filters are dependent on the signals themselves, whereas MSE filters are dependent on their statistics (specifically, the autocorrelation of the input and the cross-correlation of the input and desired signals). If the statistics are known, an MSE filter with fixed co-efficient (i.e., independent of the incoming data) can be built.

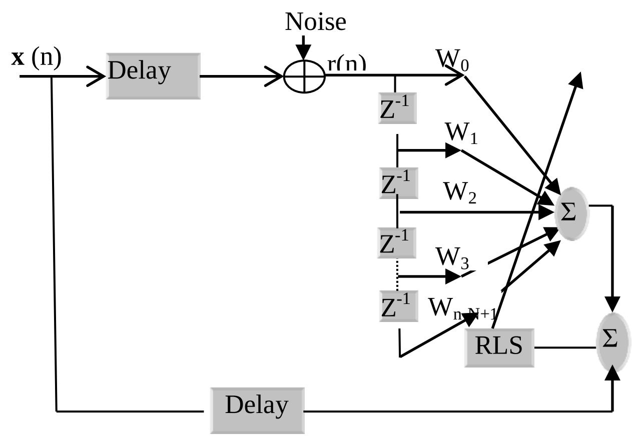

Fig. 3: LTF based Equalizer with RLS algorithm

4. Result and Discussion

It has been simulated by taking the step size parameter of μ=0.01 and the Eigen value spread W=2.9 was taken. With different value of SNR the MSE plot was presented. The same is considered with sign regresser LMS and observed that the MSE plot of the same is much better than MSE of traditional LMS. The non-linear channel given in (4) is considered. The same channel is also considered with transversal filter base RLS equaliser with SNR=20 dB, lmd =.99, and W=2.9. Bit Error rate of different equalizers are also simulated and presented.

In has been observed that the BER [7] is minimum in transversal filter based RLS equalizer as compared with other equalizers. The model is trained with a fixed SNR value. During testing, 10,000 samples of input are sent through the channel and the equalizer. The BER is calculated for each SNR value. From the plot it is seen that the BER remains constant after some SNR values. This constant value comes from the fact that there are a constant number of error bits which is introduced. These error bits are initial bits till the delay is matched. In this case the delay is 7 and hence the equalizer BER is becoming constant at -66.0980dB.

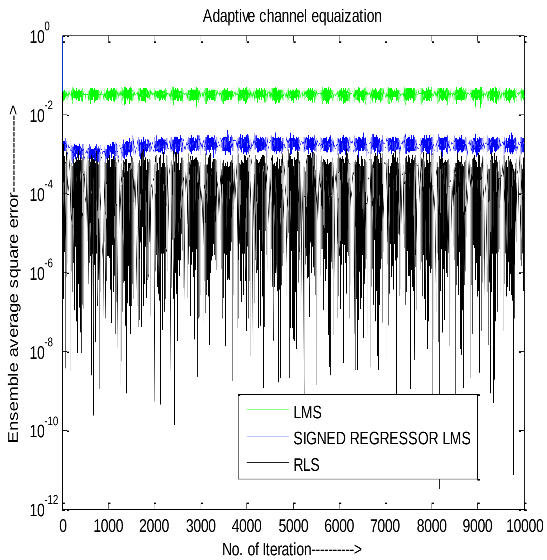

Fig. 4: MSE using LMS, Sign Regressor LMS and RLS in complex valued environment

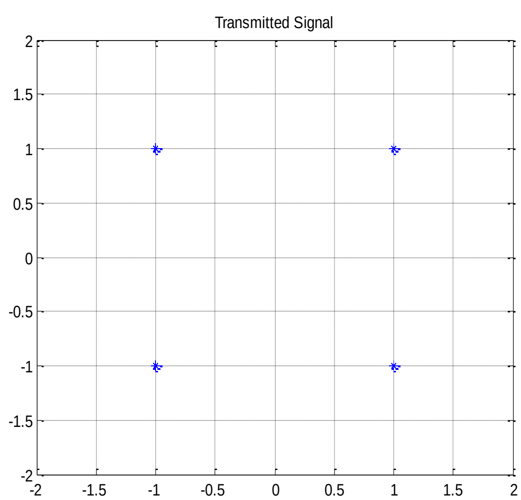

Fig. 5: Transmitted signal of QPSK in real valued environment

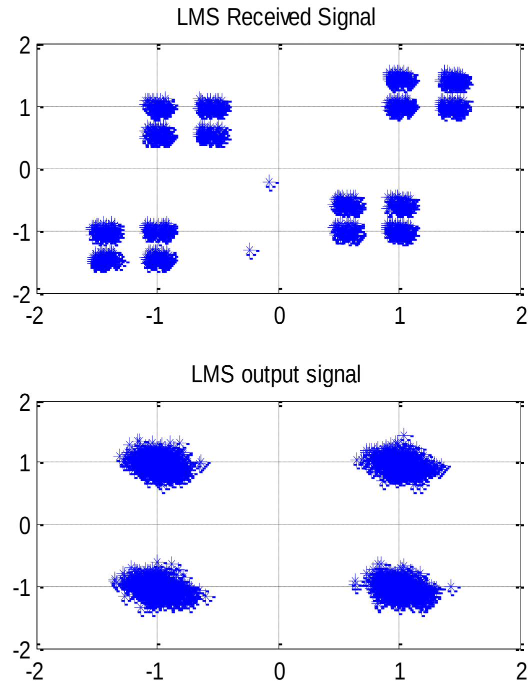

Fig. 6: Received and Equalized signal of QPSK by LMS

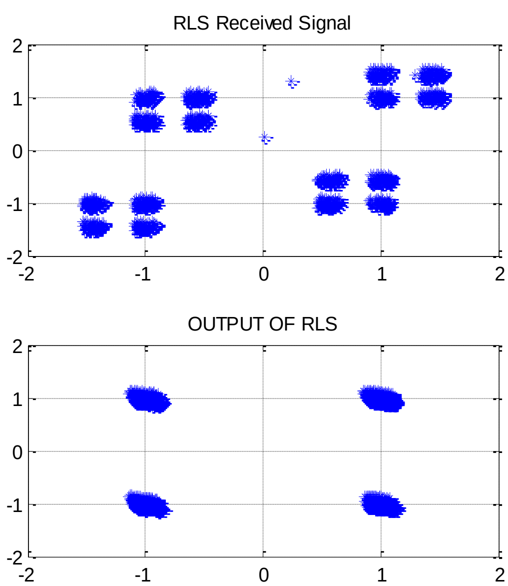

Fig. 7: Received and Equalized signal of QPSK by RLS

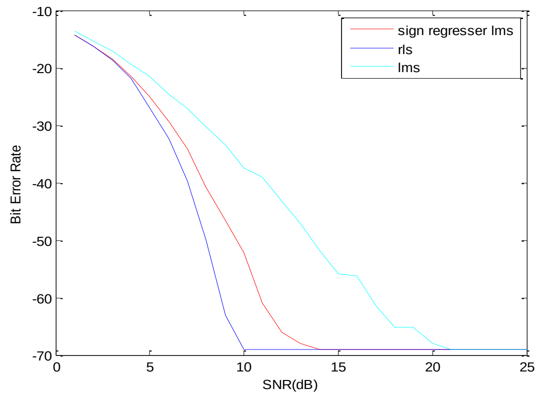

Fig. 8: Comparision of BER performance using LMS, sign Regressor LMS and RLS

5. Conclusion

The basic goal of this paper is to investigate the application of an algorithm based on adaptive filtering in channel equalization. The main objective is to achieve a high convergence rate in order to meet the requirements for short training time and good equalization properties. In this paper, we have developed a new model based on RLS algorithm. An application of this algorithm in channel equalization has been presented and its performance has been compared with those of LMS and sign regresser LMS algorithms. The proposed algorithm has good convergence rate and lower steady state mean square error.

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