Experimental design of adaptive algorithms to reduce SER in M-ary QAM System

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Abstract: This paper presents an experimental design of adaptive algorithms to investigation of Symbol error rate for adaptive training algorithms without involving with any modified or design algorithms, two conventional adaptive equalizer training methods, the least mean squares (LMS) and recursive least squares (RLS), are implemented in a Quadrature amplitude modulation over sampled received sequence and their performances are compared in different parameters like symbol size, step size and rotation. The algorithms were evaluated using convergence SER at SNR of 20 dB over different number of iterations to determine the convergence rate and constellation diagra

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I. Introduction

A reliable digital transmission system it is crucial to reduce the effects of ISI and it is where the Adaptive algorithms come on the scene. The need for algorithms arises from the fact that the channel has amplitude and phase dispersion which results in the interference of the transmitted signals. The design of the transceiver depends on the assumption of the channel transfer function is known. But, in most of the digital communications applications, the channel transfer function is not known at enough level to incorporate filters to remove the channel effect at the transmitters and receivers[2]. For example, in circuit switching communications, the channel transfer function is usually constant, but, it changes for every different path from the transmitter to the receiver. But, there are also non stationary channels like wireless communications. These channels' transfer functions is not fix changeable with time, so that it is not possible to use an optimum filter for these types of channels [8]. So, In order to solve this problem adaptive algorithms are designed. As it is told before, most of the time, the channels and, consequently, the transmission system's transfer functions are not known. The result of this is that the algorithms cannot be designed. So, mostly preferred scheme is to exploit adaptive equalizers. An adaptive equalizer is an equalization filter that automatically adapts to time-varying properties of the communication channel. It is a filter that self-adjusts the weight its transfer function according to an optimizing algorithm.

II. System Model

The block diagram of the proposed communication system is shown in Figure 1. Proposed system use linear equalizers in a training mode operation and M-QAM modulation. The additive white Gaussian noise is added to the channel response [7]. The maximum channel response value is used to normalize the channel. Then LMS equalizers are implemented at the receiver for reducing the channel distortions.
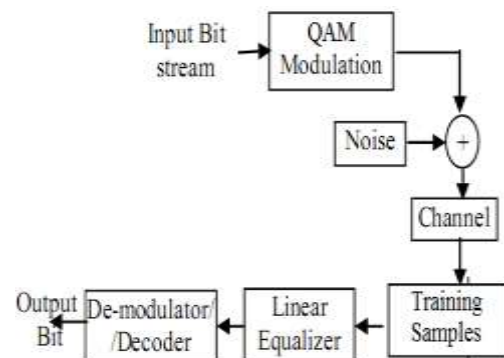

Figure 1: Proposed Communication Model

III. Problem Formulation

It is found that carrier phase recovery is very essential for wireless communication system. For this purpose, equalisers are used. Equalization is a processing technique used on any signal at the receiver to counter act the effects of ISI. In particularly, the ISI and distortion effect of channel leads difficulties in recovering the phase of the carrier at the receiver. As mentioned above equalizer used for the purpose of recovering the carrier phase at the receiver. Therefore, motivation in this area of research to investigate the performance of two LMS and RLS based decision directed (DD) equalisers.

We consider a baseband symmetric QAM communication system. Let X(k) be the kth transmitted symbol from a power normalized M-ary constellation. At the receiver side, after front-end signal processing, a complex low-pass version of the received signal is available for sampling. We assume that the received signal is already equalized and frequency-synchronized, and that timing recovery is done. We consider the following model for the phase rotated signal samples:

$$Y(k) = Ge^{j\theta} X(k) + N(k)$$

Let Y (k) is the samples of equalized signal at sample rate. G is the overall gain seen by symbols; θ is an unknown phase shift to be estimated. We further assume that X is the complex transmitted signal X(k)= a(k)+jb(k), where a(k) and b(k) are independently and based upon the training and decision direct mode find out minimum difference between actual output response and desired output response mean symbol error rate reduces to zero.

IV. Proposed work

In the communication system model implemented, two types of algorithm are used for the simulation purpose; they are Least Mean Square Algorithm and Recursive Least Squares Algorithm.

A. Least Mean Squares Algorithm

LMS filter is built around a transversal (i.e. tapped delay line) structure. Two practical features, simple to design, yet highly effective in performance have made it highly popular in various application. LMS filter employ, small step size statistical theory, which provides a fairly accurate description of the transient behavior. The LMS algorithm is a linear adaptive filtering algorithm, which in general, consists of two basics procedure a filtering process, which involve, computing the output of a linear filter in response to the input signal and generating an estimation error by comparing this output with a desired response and an adaptive process, which involves the automatics adjustment of the parameter of the filter in accordance with the estimation error [3]. The combination of these two processes working together constitutes a feedback loop, as illustrated in figure 2.
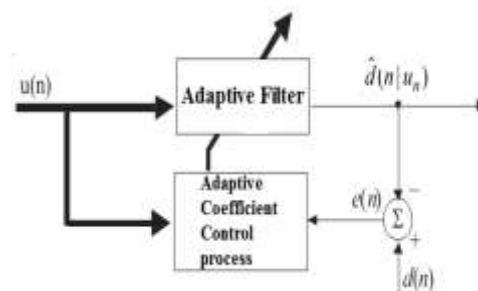

Figure 2: Block diagram of adaptive filter employing LMS algorithm [8]

B. Recursive Least Squares Algorithm

The RLS filter overcomes some practical limitations of the LMS filter by providing faster rate of convergence and good performance. In the RLS algorithm the method of least squares is extended to develop a recursive algorithm for the design of adaptive transversal filter as shown in figure 3. Given the least squares estimate of the tape weight vector of the filter at iteration (n-1), we compute the updated estimate of the vector at iteration n upon the arrival of new data[6]. An important feature of this filter is that its rate of convergence is typically an order of magnitude faster than LMS filter, due to the fact that the RLS filter whitens the input data by using the inverse correlation matrix of the data, assumed to be zero mean.
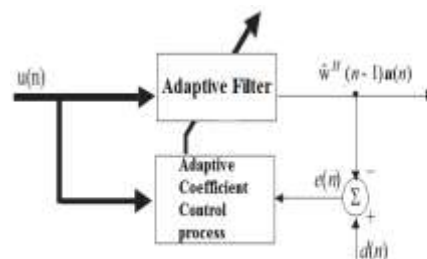

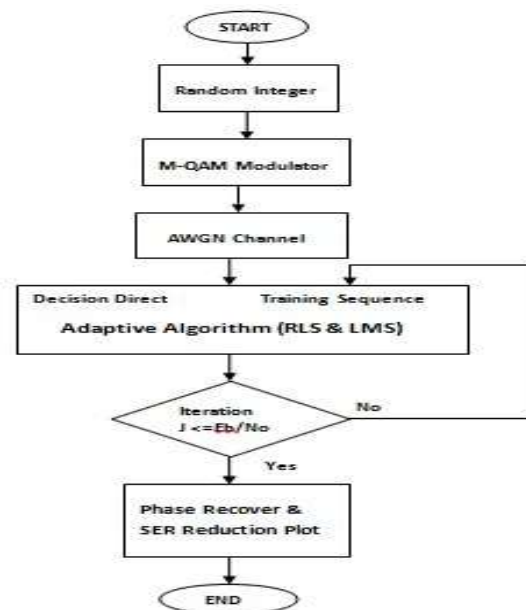Figure 3: Block diagram of adaptive transversal filter employing RLS algorithm [5]

Figure 4: Flow chart

V. Simulation and Results

Simulation is carried out in two parts. First part dealing with RLS equalizer algorithm and second part dealing with LMS equalizer algorithm Assumption made for first and second part are as follows; in random integer block M-ary number = 16 or 32. This data is modulated using QAM modulation with constellation size M- QAM. The modulated signals are processed and then passed through Additive White Gaussian Noise (AWGN) channel. In this model, the SNR is 26.0206 dB. The simulation process is carried out at different –at different step sizes, at different phase rotations (The phase noise is introduced artificially leading to a phase rotation of 10 & 20 degrees), different no. of weights. The recovered equalised signal is then demodulated using QAM demodulator and the symbol error rate (SER) is then computed between the equalised received signal and the input signal. The output is observed in the form of SER (Symbol Error Rate).

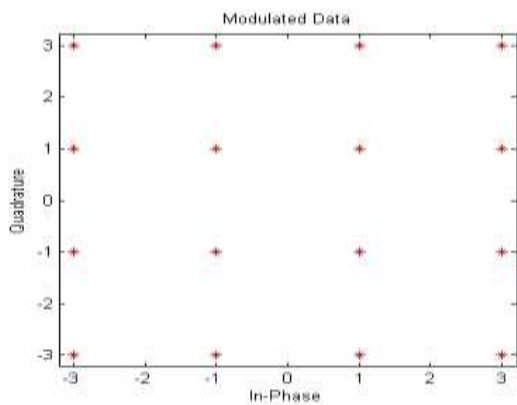The Simulation results are shown below:

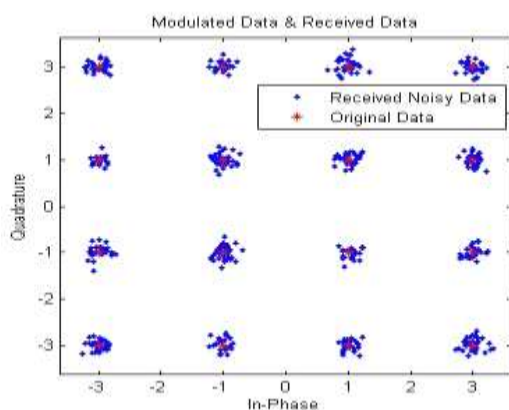Figure 5: A typical example of the QAM (Constellation Size=16) modulated data

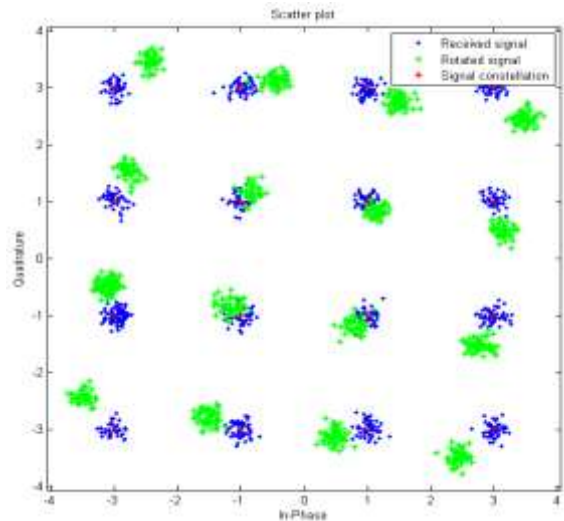Figure 6: The received modulated data with noise

Figure 7: The received noisy modulated data with rotated phase of 10 degrees
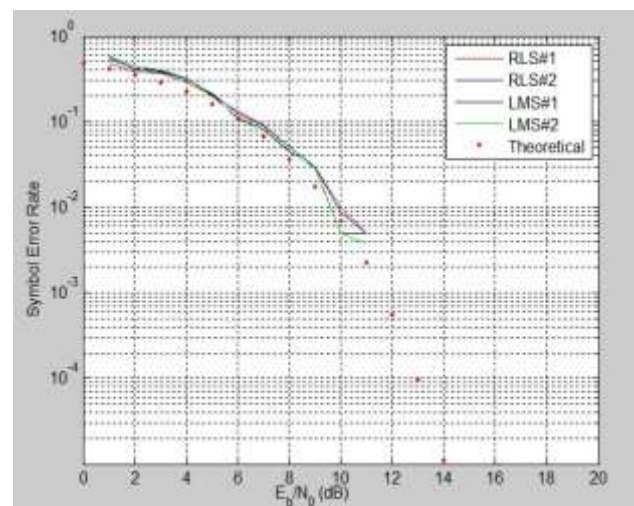

Figure 8: SER vs EbNo of the equalised signals using RLS and LMS algorithms at step size 0.0003 and Phase rotation 10 degree
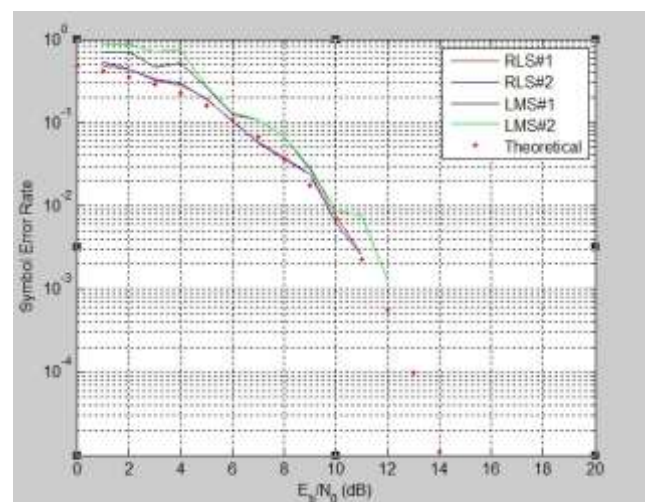

Figure 9: SER vs EbNo of the equalised signals using RLS and LMS algorithms at step size 0.003

Figure 4 basically the constellation plot for 16 Qam communication systems, its show the message store in symbol in terms of phase and amplitude define the position. When this message signal passed through the AWGN channel position of symbol will distract from original position (Error) due to noise add shown in figure 5. Figure 6 shows that with the use of adaptive algorithms with 10 degree phase rotation to recover the carrier phase and reduce to phase distraction to its original position of symbols. From the matlab graphs, it is concluded that as the Step Size is decreased down towards zero, SER gets reduced and LMS and RLS tracks or match closely the theoretical predictions. When the Step Size increased towards 1, SER gets increased and Step – Size dependent LMS algorithm does not converge and resulting a poor choice for adaptive algorithms. Thus, it is observed that Step Size is an important design criterion and plays a significant role in the performance of LMS equalizer Here we have found that the LMS equalizer is giving optimal performance at step size of 0.0003.
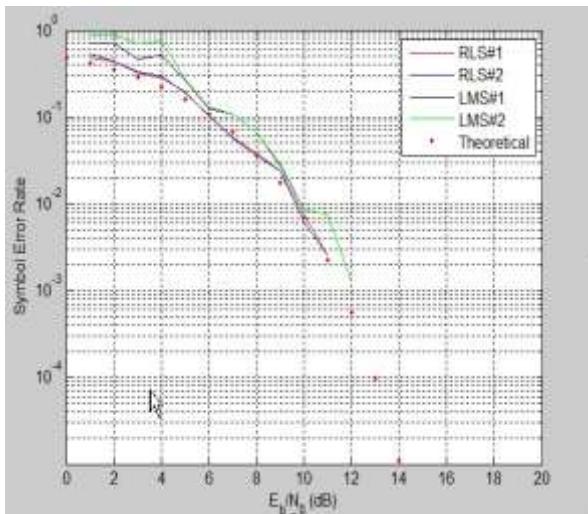

Figure 10: SER vs EbNo of the equalised signals using RLS and LMS algorithms at step size 0.003 and Phase rotation 20 degree

From the observation is made in case when the phase rotated of QAM modulated carrier is increased from 10 degrees to 20 degrees with same step size which is shown in figure 7 and figure 9, It has also been noticed that the RLS algorithm converges fastly than the LMS algorithm,. It is observed that the SER is slightly higher for situations when the phase shift is high. It is also found that the SER is slightly higher for situations when the noise is high (EbNo is between 1 to 5).

VI.   Conclusion

From the simulated results, it is noticed that the adaptive nature of the filter allows it to actively cancel out the adverse effects of the channel even without knowing the characteristics of the channel in advance.   The error performance is evaluated in terms of symbol errors rate (SER) occur in trying to recover the modulated message with and without the equalizer for the simulated digital data. Thus, overall, it can be concluded from the results that RLS converges fastly than LMS, but has comparatively higher complexity and Step Size and Phase Rotation, Forgetting Factor (Gain Factor) and the Constellation Size are the significant design criteria for both RLS equalizer and LMS equalizer and they need to be chosen appropriately to recover or estimate the phase of the carrier.

References

[1] K. V. Cartwright, "Blind phase recovery in cross QAM communication systems with eighth-order statistics", IEEE Signal Processing Letters, Vol. 8, No.12, pp. 304–306, Dec. 2001.

[2] Y. Geoffrey Li, J. H. Winters and N. R. Sollenberger, "Adaptive equilization for Wireless Communications: Signal   Detection   with   Enhanced   Channel Estimation",   in   IEEE   Transactions   on communications, Vol. 50, September   2002.

[3]   O. Rousseaux and G. Leus, "Gaussian Maximum-Likelihood Channel Estimation With Short Training Sequences",   IEEE   Transactions   on   Wireless Communications, Vol. 4, No. 6, November 2005.

[4] J. Zheng and B. D. Rao, "LDPC-Coded MIMO Systems With Unknown Block Fading Channels: Soft MIMO Detector Design, Channel Estimation,   and Code Optimization",   IEEE   Transactions   on   Signal Processing, Vol 54, No. 4, April 2006.

[5] I. H. Kacem, N. Sellami, A. Roumy and Inbar Fijalkow, "Training   sequence   optimization   for   frequency selective channels with MAP equalization", ISCCSP 2008, Malta, 12-14 March 2008.

[6] L. Zhang and T. Kwasniewski,"Optimal equalization for reducing the impact   channel group delay distortion on high-speed back plane data transmission", Journal of Electronics and Communication, 2009.

[7] W. Junfeng and Z. Bo,"Design of Adaptive Equalizer Based on Variable Step LMS Algorithm", Proceedings of the Third International Symposium on Computer Science and Computational Technology (ISCST`10), pp.256-258, Jiaozuo, P.R. China, 14-15 August 2010.

[8] C. A. N. Georghiades, "Blind carrier phase acquisition for QAM   constellations"   IEEE   PRTransactions   on Communication, Vol. COM-45, pp. 1477–1486, Nov. 1997.

[9] A. K. V. Cartwright, "Blind phase recovery in general QAM communication   systems using alternative

higher order statistics", IEEE Signal Processing Letters, Vol. 6, No. 12, pp. 327–329, Dec. 1999.

[10] K. V. Cartwright, "Fourth power phase estimation with alternative two   dimensional   odd-bit constellations", IEEE Communication Letters, Vol. 4, pp. 199–201, June 2000.