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Channel equalization techniques to mitigate inter-symbol interference in wireless communication

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Abstract

The major limitation in current wireless communications is the dispersion of time and the interference of the symbols. To counteract these problems, diverse adaptive equalization techniques are used. The equalization technique is to curb time dispersion is introduced by the communication channel and counteracts the resulting effect of inter-symbol interference (ISI). Obviously, some form of blind equalization must be established in the receiver design. Blind equalizers estimates transmitted signals and channel parameters simultaneously, and may also be time-varying. The aim of this research paper is to study the performance of various adaptive filter algorithms for both blind channel and Non-blind channel equalization such as Recursive Least means Square (RLS) algorithm equalizer, the Least Mean Square (LMS) algorithm equalizer and the Constant Modulus Algorithm (CMA) equalizer in terms of the symbol error rate and convergence speed. The Quadrature Phase Shift Keying (QPSK) and 16-Quadrature Amplitude Modulation (16-QAM) is used as a modulation scheme.

Keywords: Adaptive Equalizer, Constant modulus algorithm, Recursive Means Least Square

1. Introduction

In telecommunication, inter-symbol interference (ISI) is a form of distortion of a signal in which one symbol interferes with subsequent symbols [1]. This is an undesirable phenomenon because the previous symbol has a noise-like effect, thus reducing the reliability of communication. ISI is mostly caused as results of the multipath propagation or the inherent nonlinear frequency response of the channel, which results in "fuzziness" of consecutive symbols. ISI in the system creates errors at the receiver in the decision device. Therefore, the purpose of the transmitting and receiving filter is to reduce the ISI so that the transmission can happen with minimum errors. Ways to fight inter-symbol interference include adaptive equalization and the raised cosine filter. The receiving filter is designed to reduce the distortion produced by both the channel and the transmitter; it is often stated as Channel equalizing filter or a receiving equalizing filter. Minimization of ISI can be accomplished by applying different types of equalizers [2]. In simple terms an equalizer is the technique of altering the presence of certain frequencies inside a signal. The implementation of filters with static coefficients needs proper and prescribed specifications. However, there are situations where the specifications are not available, or is time varying. Therefore there is a solution known as adaptive filtering which has coefficients which are time varying and can adapt to the situation of changing environment [3]. Demands of high data rate transmission with an acceptable error in the presence of available bandwidth are the main targets for each operator, so we need the proper channel equalization.

2. Channel Equalization

Channel equalization is an alternative to the techniques of channel identification described previously for decoding of transmitted signals across non-ideal communication channels. In both cases, the transmitted sequences $s(n)$ is known to transmitter and receiver. However, in adaptive equalization, the received signal is used as input signal $s(n)$ to an adaptive filter, which modifies its properties so that its output thoroughly matches a delayed versions $s(n - \Delta)$ of the known transmitted signal. Translation of the appropriate period,

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system coefficients are set and used to decode messages transmitted in the future or adapted using a rough estimate of the operation known as direct adaptation to decisions. The goal of channel equalization is to remove the effects of the channel on the transmitted symbol sequence i.e., inter-symbol interference (ISI), which can be done by inverse filtering, linear equalizer or decision-feedback equalization or by applying sequential detection. The following cost functions below explain how equalizer filter can be optimized.

- Zero forcing criterion: It inverses the channel impulse response
- MMSE criterion: It minimize the mean-squared-error
- Min. Bit-Error-Rate (BER) criterion.

In the following discussion on different equalizers only the first two criteria are used [4].
In adaptive equalization, the received signal is applied to a receiver filter. The output of the receiver filter is at a symbol rate. The sampled signal is then added to the adaptive filter and these equalizer coefficients apply to minimize output noise and ISI. The adaptability of the equalizer is guided by the error signal. As mentioned in the introduction, interference between symbols is a major obstacle to the accuracy required to achieve an increase in digital transmission speed. ISI problem is fixed by channel equalization in which the purpose is to design an equalizer so the impulse response of the channel is as close to $z^{-\Delta}$ as possible, where Δ is a delay. In general, the channel parameters are not known in advance and may change over time and are important in some application. Therefore, adaptive equalization provides us means to track the channel characteristics. Following is the channel equalization system diagram depicted.

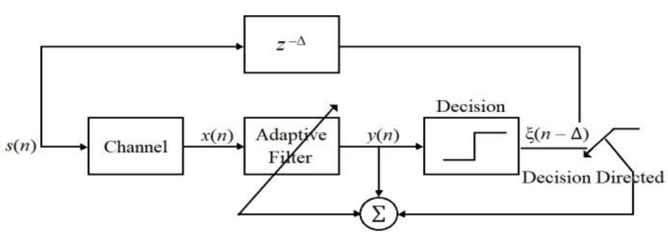

Figure 1: Digital transmission system using channel equalization

3. Adaptive Filter

There are three main adaptation algorithms used in this research, one is least mean square (LMS), constant modulus algorithm (CMA) and the other is recursive least square (RLS) filter.

3.1 Constant Modulus Algorithm (CMA)

Godard proposed an algorithm that can be used for this purpose. This algorithm introduces a different cost function that exploits the characteristics of the transmitted modulated signal. Godard's algorithm works for phased-modulated signal as it has a constant modulus and therefore is called CMA, it is very effective for achieving channel equalization. The CMA attempts to minimize the cost function $j(n)$, which depends on the difference between the received samples of squared magnitude and Godard dispersion constants.

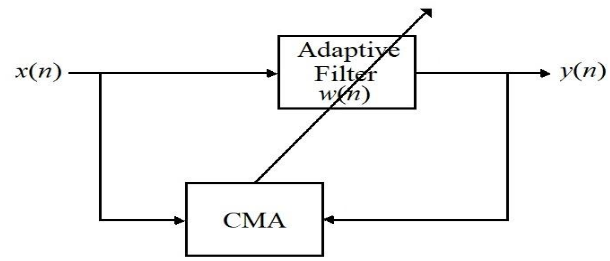

Figure 2: CMA Adaptive Algorithm Block Diagram

$$j\,(n) \;=\; E\,[\,|y(n)|^2 - R_2] \tag{1}$$

The phase update equation for carrier recovery loop is;

$$\hat{\varphi}_{n+1} = \varphi_n - \mu_2 I_m\{\,\hat{a}_n * y(n)\,exp(-j\varphi_n)\} \tag{2}$$

Comparing this to the typical LMS method where a coupling exists between the tap update and the carrier tracking loop, one can note that the coupling is removed from the CMA. This allows for the equalizer with the CMA to converge. Now the cost function of CMA is $j(n) = D^{(p)}$ where $p = 2$.
First, the data symbol constellation is assumed to be symmetric,

$$Ex\,(n)^2 = \;0 \tag{3}$$

And data symbols are stationary and uncorrelated i.e.

$$Ex(n) * x(m) = E\,|x(n)|^2\delta_m \tag{4}$$

Also, the noise can be neglected and length of the equalizer is infinite,

$$E|y(n)|^2 = E|x(n)|^2 \sum_k |s_k|^2$$

$$E|y\,(n)|^4 = \{E|x(n)|^4 - 2(E|x(n)|^2)^2\,\textstyle\sum_k |s_k|^4\; 2E\,(|x(n)|^2)^2\} \tag{5}$$

By replacing R_2 with $E|x(n)|^4/E|x(n)|^2$ the above equation becomes,

$$\begin{aligned} D^2 = \{\,E|x(n)|^4 - 2(E|x(n)|^2)^2 \sum_k |s_n|^4 \\ +\; 2(E|x(n)|^2)^2(\sum_k |s_n|^2)^2\,\} \\ -\; 2(E\,|x(n)|^4 \sum_k |x(n)|^2 \\ + constant \end{aligned} \tag{6}$$

D^2 Can be written as the equation below,

$$D^2 = \{-|x(n)|^4 - 2(E|x(n)|^2)^2 \sum_k |s_k|^4 + 2(E|x(n)|^2)^2 \left(\sum_k |s_k|^2\right)^2 - 2R_2 E|x(n)|^2 \sum_k |s_k|^2 + R_2 \quad (7)$$

It was stated that the CM cost reduction is possible, if the following satisfied when $|S_0|^2$ is close to unity,

$$4(Ex(n))^2|s_0|^2 - 2E|x(n)|^4 \geq 0 \quad (8)$$

Fundamental concepts about equalizers, blind channel equalization and along with four different versions of constant modulus algorithm (CMA) have been presented that are derived from the same cost function introduced by Godard in [5].

3.2 Least Mean Square (LMS)

As described earlier the LMS algorithm is built around a transversal filter that performs a filtering process. The weighting factor mechanism is accountable for execution the adaptive control method on the tape weight of the transversal filter. The LMS algorithm is consists of two processes: Filtering process, which involves calculating the output ($d(n - d)$) of a linear filter with respect to the input signal and resulting an estimation error by subtracting this output with a desired response as shown in equation bellow:

$$e(n) = d(n) - y(n) \quad (9)$$

$d(n)$ is the desired response and $y(n)$ is filter output at time n. Adaptive process, involves the automatic adjustments of the parameter of the filter with respect to the estimation error.

$$\widehat{w}_{(n+1)} = \widehat{w}_n + \mu(n)e^*(n) \quad (10)$$

μ is the step size, (n +1) = estimate of tape weight vector at time (n +1) and if preceding knowledge of the coefficients of vector (n) is not available, set (n) = 0; additionally, an LMS adaptive algorithm having p+1 coefficients require multiplication and p + 1 additions to upgrade the filter coefficients. Hence, single addition is required to calculate the error $e(n) = d(n) - y(n)$ and single multiplication is required to perform product $pe(n)$. Finally, p+1 multiplication and p additions are required to calculate the output $y(n)$, of the adaptive filter. Thus, a total of $2p + 3$ additions per output point are required to perform the adaptive filter. The LMS algorithm [6] was by Widrow. In LMS, the weights are upgraded at every repetition by approximating the gradient of the quadratic mean square error (MSE) surface, and then stirring the weights in the opposite direction of the gradient through a small amount, known as the step size. The convergence of this algorithm is directly proportional to the step-size parameter μ. When the step size is within a range that ensures convergence, the process leads the estimated weights to the optimal weights. Stability is ensured provided that the following condition is met.

$$\lim_{x\to\infty} E\{w_n\} = w = R_x{}^{-1} r_{dx}$$

$$E\{w_{n+1}\} = E\{w_n\} + \mu E\,\{d(n)x^*(n)\} - \mu E\{x^*(n)x^T(n)\,w_n\} \quad (11)$$

$$0 < \mu < \frac{2}{(p+1)\{|x(n)|^2\}}$$

3.3 Recursive Least Square Algorithm (RLS)

This algorithm recursively finds the filter coefficients which reduces a weighted linear least squares cost function concerning to the input signals [7]. The purpose of this algorithm is to reduce the mean squares error. In RLS, the input signals are assumed deterministic. The RLS exhibits extremely fast convergence as compared to other conventional algorithms. Nevertheless, this advantage comes at the price of high computational processing complexity, and possibly not very good tracking performance when the filter to be estimated changes. RLS and LMS algorithms are similar as shown in Figure 3 but RLS algorithm gives sufficient tracking ability for fast fading channel [8]. Additionally, RLS algorithm have stability problems because of the covariance update formula $p(n)$, which is used for electronic adjustment in accordance with the estimation error as follows the figure below illustrate the RLS algorithm block diagram:

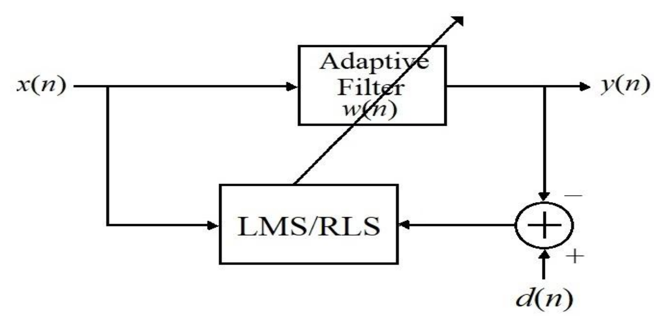

Figure 3. Block diagram for RLS adaptive equalizer

$$p(0) = \delta^{-1} I \quad (12)$$

Where; p is inverse correlation matrix and δ^{-1} is regularization parameter, positive constant for high SNR and negative constant for low SNR. (n = 1, 2, 3...)

$$\pi(n) = p(n-1)u(n) \quad (13)$$

$$k(n) = \frac{\pi(n)}{\lambda\mu^H(n) + \pi(n)} \quad (14)$$

Time varying gain vector

$$\xi(n) = d(n) - \hat{w}^H(n-1)\,u(n) \tag{15}$$

$$\hat{w}(n) = \hat{w}(n-1) + k(n)\,\xi(n) \tag{16}$$

4. General System Model

System model is shown in the following figure as block diagram which is generalized for communication channel utilizing any of the CMA, RLS or LMS equalizer to overcome ISI.

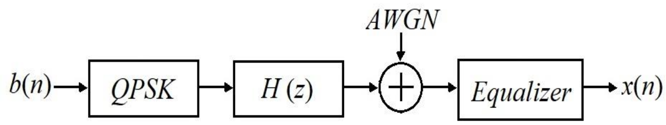

Figure 4: System model diagram

The model used and shown in Fig.4, consists of a binary data $b(n)$ and a modulated using QPSK and 16-QAM technique to produce the transmitted signal $s(n)$. $s(n)$ goes through the channel that has the transfer function. In this research work one Gaussian communication channel with its *z-transform* or *transfer function* is considered in a form as;

$$H(z) = \frac{A}{B + Cz^{-1}}$$

Where H(z) is the channel used, and Additive White Gaussian Noise (AWGN) with Signal to Noise Ratio (SNR) with a range from 0dB and 30 dB has been added, and then the receiver gets the signal which is now referred to as x(n) or the received signal. The objective of equalizer is to cancel the channel effect or minimize the effect of ISI on the transmitted signal and to also obtain an estimate for it, which is given as the output of the equalizer y(n). The equalizer models are shown in Fig .4 for a general model. Implementing the number of the adaptive FIR filter coefficients is 3 with 300000 transmitted samples is carried out. The step size for LMS is 0.002 and 0.001 for QPSK and 16-QAM respectively and 0.0001 for CMA. The simulation results were performed in Matlab. Figures below present the transmitted signal which are QPSK and 16-QAM modulated signals; also figure below shows the received signal which is the transmitted signal passed through the distortion channel.

5. Simulation Results

In order to estimate the performance of these adaptive algorithms, analytical simulations are carried out in MATLAB.

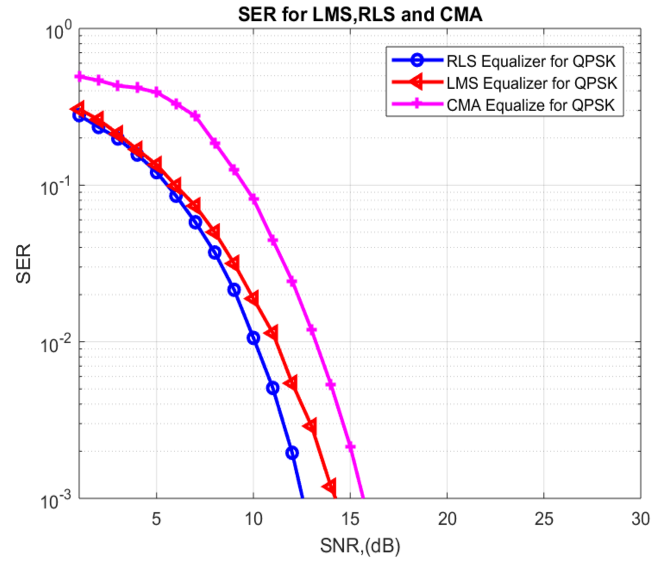

Figure 5: Results of RLS, LMS and CMA Equalization with QPSK

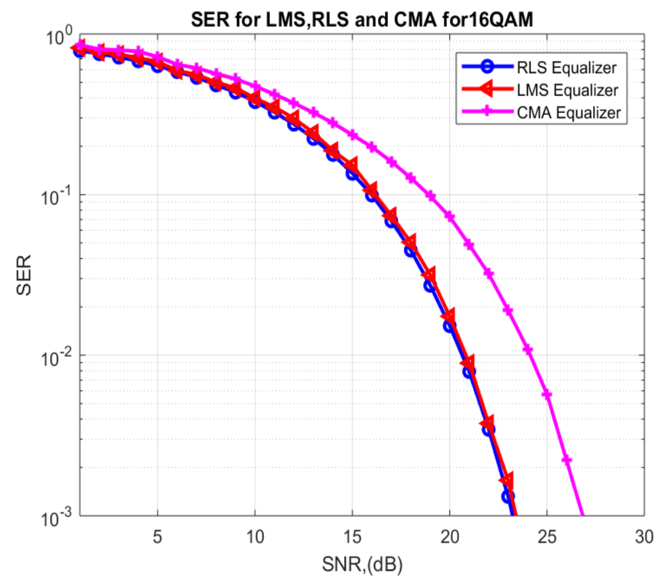

Figure 6: Results of RLS, LMS and CMA Equalization with 16-QAM

6. Results Discussions

From the simulation results, for both QPSK and 16-QAM we can notice that the proposed adaptive algorithms used to combat ISI give good channel equalization (CMA, LMS and RLS) among the three algorithms, thus the RLS equalizer is better, followed by the LMS and CMA equalizers and also RLS has high convergence and accuracy, trailed by LMS and CMA. Cost, convergence and accuracy determine equalizer choice. Fig. 5 and Fig. 6 show a SER versus SNR for (QPSK) and 16-QAM using AWGN channel by Adaptive Equalizers respectively. In both results from the figures, it can be seen that the CMA works well after SNR is 5db. This is due to the fact that Recursive Least Square and Least Means Square using training symbols in every frame to approximate channel effect then equalize it; which Constant Modulus Algorithm uses blind algorithm for its estimate. Decreasing in noise power the Recursive Least Square and Least Means Square are more efficient and better than Constant Modulus Algorithms. Adaptive filters using Constant Modulus Algorithm, Least Means Square and Recursive Least Square algorithm have been done using MATLAB environment and their responses have been studied. The comparison diagrams of the (LMS, CMA and RLS) has been carried out based on their SER. It was shown that the SER of RLS was less followed by LMS and CMA equalizer. The RLS, LMS and CMA in both (QPSK) and 16-QAM meet channel equalization by reducing channel effects. From the simulation results, for both QPSK and 16-QAM, it can be noticed that 4-QAM or (QPSK) performed better than 16-QAM. Also CMA performed better for 4-QAM than 16-QAM based on the symbol error rate against the signal to noise ratio which proved that for increased value of M the data rate will be increased but with an increasing value of SER, however QPSK will perform better than 16- QAM.

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