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Techniques for equalizing channels to reduce symbol interference in wireless systems

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Abstract:

Time delays and symbol interference are now the biggest problems in wireless communications. Various adaptive equalisation strategies are used as a means of resolving this issue. The goal of the equalisation method is to reduce the inter-symbol interference caused by the temporal dispersion provided by the communication channel (ISI). For the receiver to function properly, some kind of blind equalisation must be implemented. During transmission, a blind equaliser makes estimates of both the broadcast signal and the channel characteristics, and these estimates might fluctuate over time. The purpose of this paper is to compare and contrast the symbol error rate and convergence speed of a number of different adaptive filter algorithms for blind channel and non-blind channel equalisation, including the Least Mean Square (LMS) algorithm equaliser, the Constant Modulus Algorithm (CMA) equaliser, and the Recursive Least Mean Square (RLS) algorithm equaliser. As a modulation method, we use Quadrature Phase Shift Keying (QPSK) and 16-Quadrature Amplitude Modulation (16-QAM). The staff of ijrei.com. No permission is being granted at this time.

Keywords:

Recursive Least Squares, Constant Modulus Algorithm, and Adaptive Equalizers

Introduction

Inter-symbol interference (ISI) refers to a kind of signal distortion that occurs when one symbol causes distortion to successive symbols in a transmission [1]. This is a problem because the preceding symbol acts like noise and dilutes the clarity of the message. Symbol "fuzziness" due to ISI is often the consequence of multipath propagation or the channel's intrinsic nonlinear frequency response. Systemic ISI causes erroneous decisions to be made by the receiver of the device in question. As a result, the filter at both ends of the transmission is designed to dampen the ISI so that transmission mistakes are kept to a minimum. Adaptive equalisation and the raised cosine filter are two methods that may be used to combat inter-symbol interference. Channel equalising filter, often known as a receiving equaliser, is a kind of filter used to mitigate distortion introduced by the transmission medium and the source device. Many equalisers exist for the purpose of reducing ISI [2]. Equalization, in its simplest form, is the process of adjusting the relative importance of different frequencies within an audio source. Filters using

constant coefficients need strict guidelines for their implementation. On the other hand, there are times when the requirements are either unavailable or change over time. That's why there's a technique called adaptive filtering, which uses time-varying coefficients to adjust to the ever-changing conditions it faces [3]. Since each operator primarily aims to meet the need for high data rate transmission

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with an acceptable error rate within the available bandwidth, accurate channel equalisation is essential. Equalization of Channels When decoding messages across less-than-perfect communication channels, channel equalisation may be used instead of the channel identification methods discussed above. Both the sender and the receiver will have prior knowledge of the communicated sequence s(n). However, in adaptive equalisation, a delayed version of the known transmitted signal, s (n -), is used as input to an adaptive filter, which then adapts its features such that its output completely matches the original signal s(n). By making a direct adaptation to choices based on a translation of the appropriate time period, the system coefficients
• Lower bound on the bit error rate (BER).

Only the first two criteria are considered in the following examination of several equalisers [4]. Adaptive equalisation uses a receiver filter to modify the incoming signal. The symbol rate is the output of the receiver filter. These equalisation coefficients are used to reduce output noise and intersymbol interference (ISI) by combining the sampled signal with the adaptive filter. The equalizer's sensitivity to the error signal informs its capacity to change its settings. As was noted in the introduction, one of the main roadblocks to increasing the speed of digital transmission is interference between symbols. The goal of equalising a channel is to get its impulse response as near as feasible to z -, where is a delay, hence eliminating the ISI issue. It is not always possible to predict or account for the channel's properties, and those values may change over time and be crucial to particular uses. Consequently, we may monitor the channel's properties with the help of adaptive equalisation. Channel equalisation system schematic is shown below.

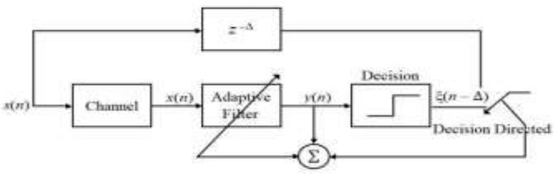

Figure 1: Digital transmission system using channel equalization

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.

Constant Modulus Algorithm (CMA)

Godard proposed an algorithm that can be used for this purpose. This algorithm introduces a different

may be established and utilised to decode messages sent at a later time. Channel equalisation employs inverse filtering, linear equalisers, decision-feedback equalisation, and sequential detection to mitigate the impact of the channel on the transmitted symbol sequence (also known as inter-symbol interference, or ISI). Optimal settings for an equalisation filter are detailed in the following cost functions.

The zero-forcing criteria inverts the impulse response of the channel.

Criteria for minimising mean-squared error (MMSE)

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.

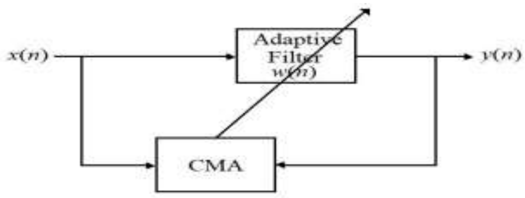

Figure 2: CMA Adaptive Algorithm Block Diagram

$$j(n) = E[\,|y(n)|^2 - R_2] \quad (1)$$

The phase update equation for carrier recovery loop is;

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

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,

$$Ex(n)^2 = 0 \quad (3)$$

And data symbols are stationary and uncorrelated i.e.

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

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

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

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

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

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

D^2 Can be written as the equation below.

$$\begin{aligned} D^2 = \{-|x(n)|^4 - 2(E|x(n)|^2)^2 \sum_k |s_k|^4 \\ + 2(E|x(n)|^2)^2 \left(\sum_k |s_k|^2\right)^2 \\ - 2R_2 E|x(n)|^2 \sum_k |s_k|^2 \ + \ R_2 \end{aligned}$$

It was stated that the CM cost reduction is possible, if the following satisfied when |S0| 2 is close to unity,

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

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].

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.

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

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 Windrow. 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.

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

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

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

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:

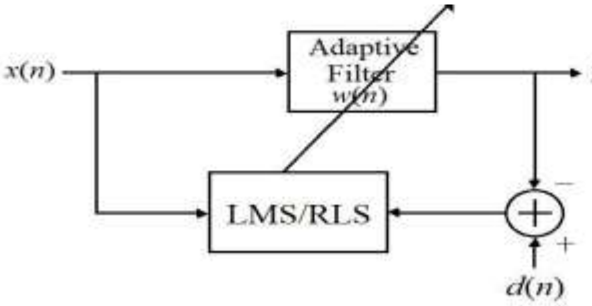

Figure 3. Block diagram for RLS adaptive equalizer

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

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...)

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

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

Time varying gain vector

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

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

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.

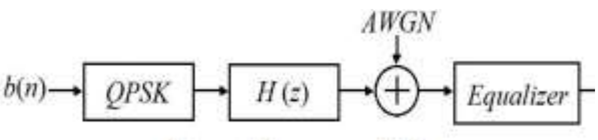

Figure 4: System model diagram

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;

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

When the signal is received at the receiver, it is denoted by the notation x(n), and the channel is denoted by H(z). The signal is subsequently corrupted by Additive White Gaussian Noise (AWGN) with a Signal to Noise Ratio (SNR) ranging from 0 dB to 30 db. Equalizers are used to negate the channel effect or reduce the impact of inter-symbol interference (ISI) on the transmitted signal and to provide an approximation for it, which is represented by the equalizer's output y.(n). Fig.4 depicts a generic equaliser model. Three adaptive FIR filter coefficients are implemented, and 300,000 transmitted samples are used. LMS uses a 0.0001-step size for CMA, 0.002 for QPSK, and 0.001 for 16-QAM. MATLAB was used to produce the simulation findings. You can see examples of the sent signal (QPSK and 16-QAM modulated signals) and the received signal (the transmitted signal after being distorted by the channel) in the figures below.

Calculation Outcomes

Analytical simulations in MATLAB are used to predict how well these adaptive algorithms will work in practise.

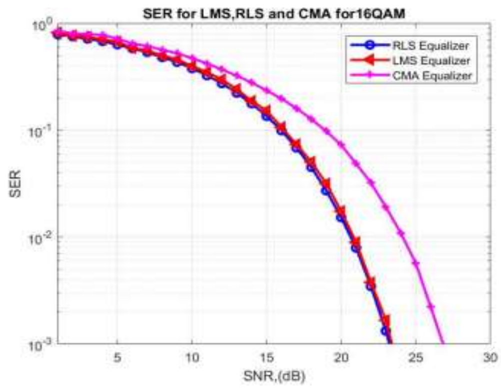

Figure 5: Results of RLS, LMS and CMA Equalization with QPSK

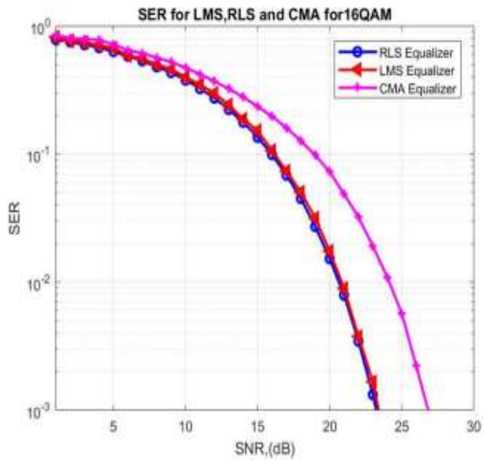

Figure 6: Results of RLS, LMS and CMA Equalization with 16-QAM

Conversations Based on Findings

The simulation results show that the proposed adaptive algorithms used to counteract ISI provide good channel equalisation for both QPSK and 16-QAM, with the RLS equaliser being the best of the three due to its high convergence and accuracy and the LMS and CMA equalisers coming in second and third, respectively. The selection of an equaliser is determined by price, convergence, and precision. Figure 5 and Figure 6 are SER vs SNR plots for (QPSK) and (16-QAM) Adaptive Equalizers operating on an AWGN channel, respectively. Results from both figures show that the CMA performs well after the SNR reaches 5db. To estimate the channel impact, the Constant Modulus Method relies on a blind algorithm, whereas Recursive Least Square and Least Means Square rely on training symbols in each frame. Compared to Constant Modulus Algorithms, Recursive Least Square and Least Means Square are superior in terms of noise power reduction.

The MATLAB environment has been used to test and analyse adaptive filters based on the Constant Modulus Algorithm, the Least Means Square Algorithm, and the Recursive Least Square Algorithm. Based on their SER, comparison diagrams between LMS, CMA, and RLS have been drawn. RLS was found to have the lowest SER, followed by LMS and CMA equalisers. By mitigating the impact of the channel, the RLS, LMS, and CMA of both (QPSK) and 16-QAM achieve channel equalisation. Compared to 16-QAM, 4-QAM or (QPSK) fared better in the simulations. In addition, the symbol error rate vs the signal to noise ratio showed that CMA worked better for 4-QAM than 16-QAM, demonstrating that a higher value of M results in a higher data rate, but that a higher value of SER results in better performance for QPSK than 16-QAM.

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