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DESIGN OF CHANNEL EQUALIZATION USING LEAST MEAN SQUARE (LMS) ALGORITHM WITH VARIABLE STEP SIZE

Mohammed Hussein Ali, Daniya Amer Jassim, Noora H. Sherif
Al-Turath University College, Baghdad, Iraq.
[1]mohammed.ali@gmail.com , [2]daniya.amer@yahoo.com, [3]noora.sherif@gmail.com
(+96407723300315) For correspondence: mohammed.ali@gmail.com

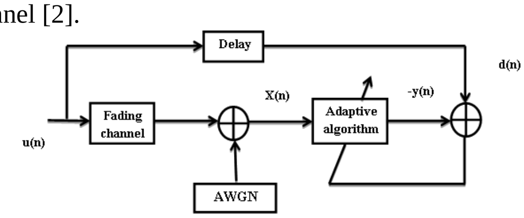
ABSTRACT: *In this paper, we present the overall structure of the least squares equalizer algorithm with different step sizes. Once the transmitted signal arrives at the receiver, the distortion is interrupted due to ISI (inter-symbol interference) interference that may occur in the symbol transmission channel of the transmitted data and the fading effect of the AWGN. As a result, many advanced symbol detection methods in the receiver must be applied to recover the originally transmitted data symbols. The coefficients of the FIR filter can be adjusted by the LMS algorithm to minimize noise and intersymbol interference. The simulation results show that the size of the step affects the stability, convergence speed and steady-state error of the adaptive filter.*

Key Words: Adaptive Equalizer, Least Mean Square (LMS) algorithm, step size, Gradient algorithm, steepest descent, Finite Impulse Response (FIR) filter.

1. 1 INTRODUCTION

What is important in what system is that the message bit sequence in the issuer can be a modulation symbol, a transmission bath on the fading wireless channel. The received signal is received by the deformed multipath fading path channel. In order to retrieve the segments sent to the receiver, channel effects must be estimated. The received signal can be defined as the convolution between the frequency response of the channel and the transmitted signal. Various techniques can be used to estimate the response channel. In any system, it is necessary to take a certain number of factors for the number of techniques for channel estimation. These factors include changes in channel time, performance, and mathematical complexity [1]. In this paper, the impulse response filter coefficient (RIF) defined by Ajustes can be the LMS of the algorithm.

This is visible in the figure. In Figure 1, the signal u(n) is the signal transmitted on the channel, and the attenuated signal x(n) is considered to be the signal input through the input of the FIR filter. The adaptive filter adjusts the value-based filtering algorithm between the output of the adaptive filter y(n) and the desired signal d(n) and the coefficients of the adaptive error signal (n). Finally, the error value e(n) is reduced and the filter coefficient w(n) is similar to the ideal channel [2].



Figure(1): adaptive algorithm[2].

The adaptive filter is considered the inverse filter which means that this filter gives the inverse coefficient coefficients for the channel and the process for canceling the effect of the channel. Adaptive filter also known as a convolution filter where the FIR filter convolves the filter coefficients with a sequence of input values and produces a sequence of output values equivalently numbered, so that the FIR filter is used to filter the signals using convolution.

FIR impulse response (FIR) filters are finite impulse response digital filters. The FIR filter is the simplest filter to design. FIR filter also known as the convolution filter, where the FIR filter convolves the filter coefficients with a sequence of input values and produces a sequence of output values with the same number so that the FIR filter is used to filter the signals using convolution. The term taps refers to the number of filter coefficients for a FIR filter.

The main infrastructure of this adaptive filter is illustrated in FIG. 2

1.2 Least mean square algorithm (gradient algorithm)

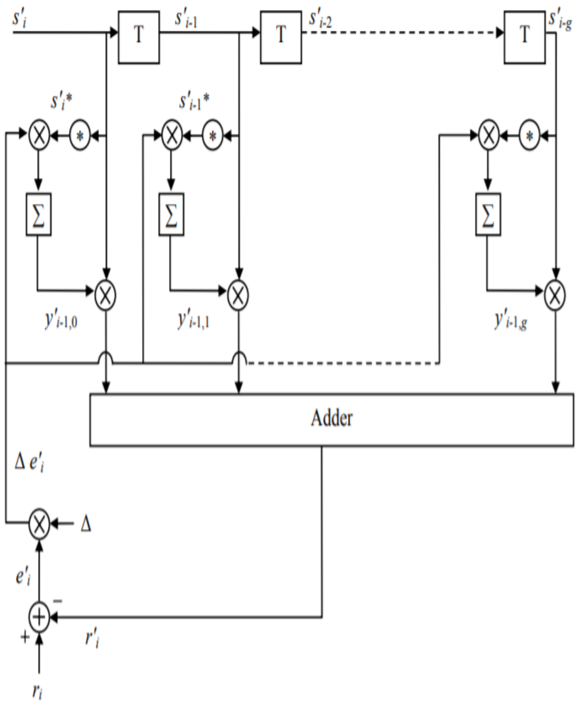
The adaptive equalizer is considered to be a time difference filter. An adaptive equalizer is called a transversal filter. Generally, an adaptive filter consists of an adaptive algorithm and a linear filter. The linear type is a finite impulse response (FIR) filter. The adaptive algorithm is the least mean square (LMS) algorithm. The finite impulse response (FIR) coefficients can be adjusted by the LMS algorithm to minimize noise and intersymbol interference. FIR filters are the simplest filter designs. The FIR filter is a finite impulse response [3]. The structure of the adaptive equalizer is shown in (2), where the subscript is a discrete time index. The value of the input s_i at any time depends on the noise value and the radio fading channel. A single input s_i has N delay elements, N + I taps and N + I complex multipliers called weights. The filter weight also has a subscript indicating that it changes over time. The weight of the filter is updated by an adaptive algorithm [4]. The error signal e_i in the adaptive algorithm is the difference between the received signal and the estimated signal reception. The adaptive algorithm uses y_h to update the equalizer weights. Therefore, the Least Mean Square (LMS) algorithm searches for the optimal value of the filter weight. This process becomes more frequent as the equalizer tries to converge, and more techniques (such as gradients or steepest algorithms) should be used to reduce the error. Finally, the adaptive algorithm freezes the filter weights to the error signal to a new level of training [5, 2].

A signal input for the equalizer defined as a vector s_i where

$s_i = [s_i \ \ s_{i-1} \ \ s_{i-2} \ \ \ldots\ldots s_{i-h}]$ ………… (1)

The weight vector (taps) of the adaptive equalizer which represent the number of filter coefficients for FIR filter can be written as:

$y_h = [y_{0i}\ y_{1i}\ y_{2i} \ldots\ldots\ldots\ y_{hi}]$ ....(2)



Figure(2): adaptive Equalizer using Least Mean Square algorithm[3].

The received signal at the receiver is given by (Where w_i is the Additive White Gaussian Noise (AWGN)):

$r_i = \sum_{h=0}^{g} s_{i-h}\ y_i + w_i$ ....[3]

$\grave{r}_i = \sum_{h=0}^{g} \grave{s}_{i-h}\ \grave{y}_{i-1,h}$ ....[4]

$e_i = r_i$ - $\grave{r}_i$ ....[5]

The adaptive filter equation is given by:

$\grave{y}_{i,h} = \grave{y}_{i-1,h} + \Delta\ e_i\ \grave{s}_{i-h}$ ....[6]

The Δ factor is the step size and it may not be constant. It can be ranging between small values to provide an excellent estimation for slow channels, or a big value to follow the fast channel change.

1.3 SIMULATION AND RESULTS

The adaptive algorithm is iteratively adjusted to minimize the coefficient of the magnitude of the error E_I. In this paper, we use variable step sizes to increase the convergence speed of the adaptive filter. When we use the LMS algorithm to create an adaptive filter, we must choose the step value. The step size is the stability effect of the adaptive filter, the convergence speed and the steady state error. When the step size is small, the convergence speed of the adaptive filter will decrease. When the step size is large, the convergence speed of the adaptive filter will be improved. Therefore, when the step size is large, this may make the adaptation unstable. In this paper, we take the step size and the least mean square algorithm results to show three different values. It is in Figure (3) and the figure is obvious (4) when the step size is equal to the value of 0.0035. The difference between Entre Rios LES is used to estimate the ratio of the channel and the actual value is wide and adaptive to the steady-state error of the curve. The magnitude of the error in the filter.

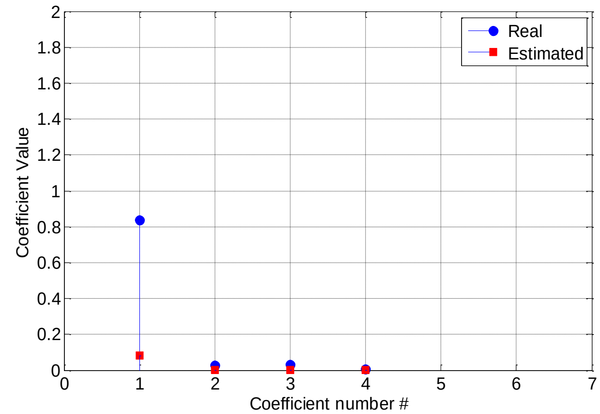


Figure (3): Difference between actual and estimated value of the coefficient of the channel when the step size = 0.0035.

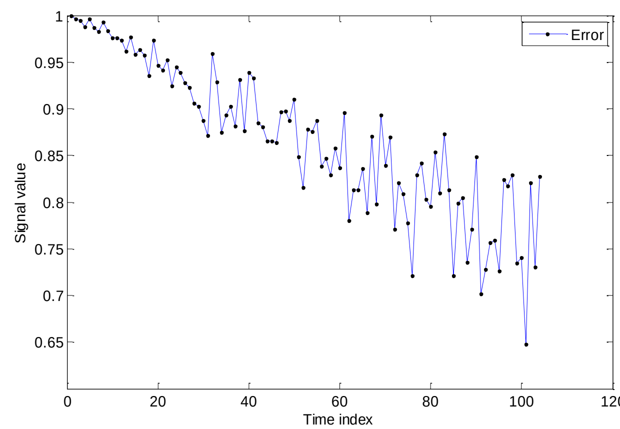


Figure (4): The magnitude of the error for the least mean square (LMS) estimator when the step size = 0.0035.

It is obvious in Figure (5) and figure (6) when the value of step size equal to 0.035 the difference between the estimated value and real value for the coefficient of the channel become small, the speed of convergence increase and the error of steady state increase.

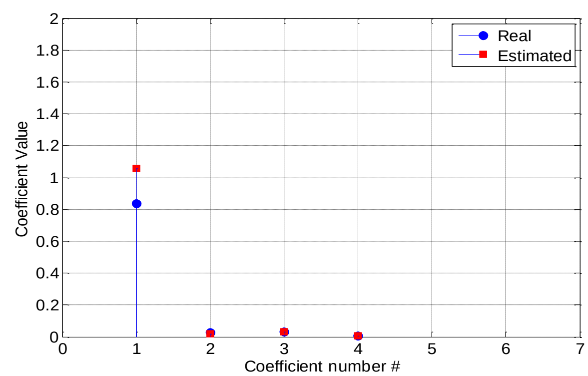


Figure (5): Difference between actual and estimated value of the coefficient of the channel when step size = 0.035

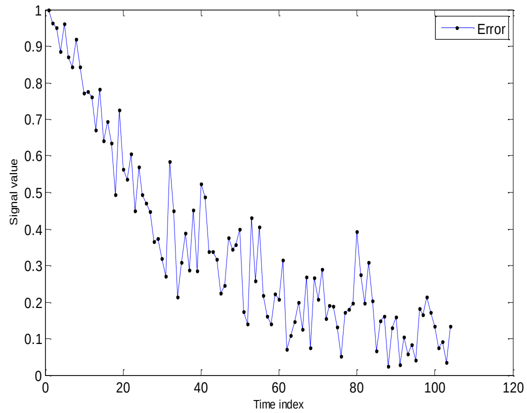


Figure (6): Error size of the smallest mean square (LMS) estimate when the size of the step = 0.035

It is evident in FIGS. 7 and 8 that when the step value is equal to 0.055, the difference between the estimated value of the channel coefficient and the actual value is small, the speed of convergence increases and the steady state error becomes no. Stable.

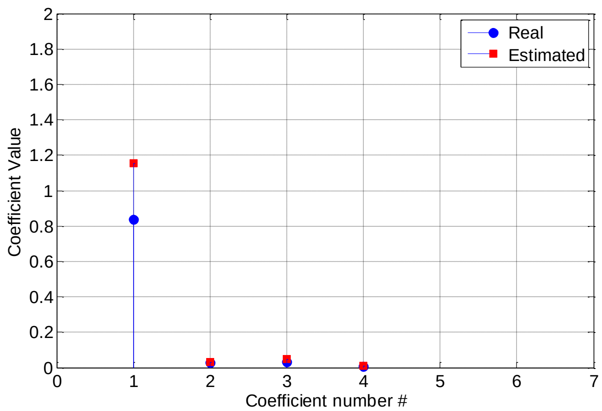


Figure (7): Difference between actual and estimated value of the coefficient of the channel when the step size = 0.055

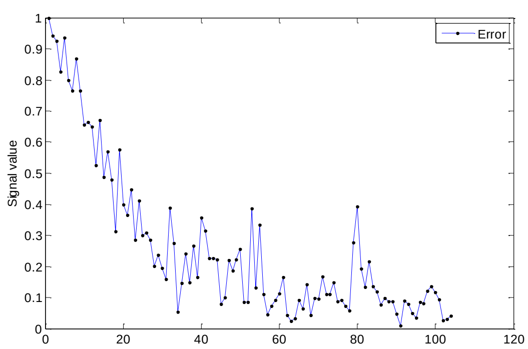


Figure (8): The magnitude of error for the least mean square (LMS) estimator when the step size = 0.055

CONCLUSION

The LMS (Least Mean Square) algorithm requires less calculation because it is simple and does not require matrix inversion. The LMS algorithm is applied to equalize the effect of the channel. In this article, we take a different value for the step size in the adaptive equalizer. The value of the step size should be used when the Least Mean Square (LMS) algorithm is used to create an adaptive filter. From the results of the simulation, we conclude that when the size of the step size is small, the system will be slow but the mean squared error at steady state will be small. Otherwise, when the size of the step size is large, the system will be fast but the steady state mean squared error will be large.

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