The Comparative Study of Adaptive Channel Equalizer Based on Fixed and Variable Step-Size LMS Algorithm & its Variants for Non Stationery Wireless Channel

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Abstract-the performance of modern communication system can be reduced by non ideal characteristic of the channel, which is known as inter-symbol interference (ISI).The equalization technique is an efficient method to overcome ISI and improve the characteristics of the system. Adaptive linear filter and various categories of training algorithms are used to imitate different equalizer models. . Here we have simulated a digital communication model having quadrature amplitude modulation technique and additive white Gaussian noise channel implemented in MATLAB where Equalizer plays major role in this model. This paper presents the comparison of performance of linear & non linear adaptive channel equalizer trained by using gradient decent algorithm LMS & its variants such as NLMS, FBLMS, and SRLMS with fixed & variable step size in terms of Bit Error Rate (BER). Inspite of much prior work on this subject we reveal surprising analytical results in terms of well known BER

Keywords—**LMS, NLMS, FBLMS, BER Algorithm, Channel Equalization, DFE, VSSLMS**

I. Introduction

In any wireless communication system, the channel induced distortion results in Inter-symbol Interference (ISI), if it is not compensated, causes higher error rates. The solution of this ISI problem is to design a receiver that employs a means for reducing the channel distortion which arises due to non linear characteristic of the channel, in the received signal. An adaptive equalizer is the best compensator for the ISI problem.

Channel Equalizers are widely used in digital communication systems to mitigate the effects of ISI caused by channel distortion. Linear equalizers are most popularly used for high speed modems that transmit data over telephone channels but some channels like radio channels require a very powerful non linear adaptive channel equalizer to combat the severe ISI such as DFE adaptive channel equalizer track such variations in the channel response & adapt it's coefficients to reduce the ISI.

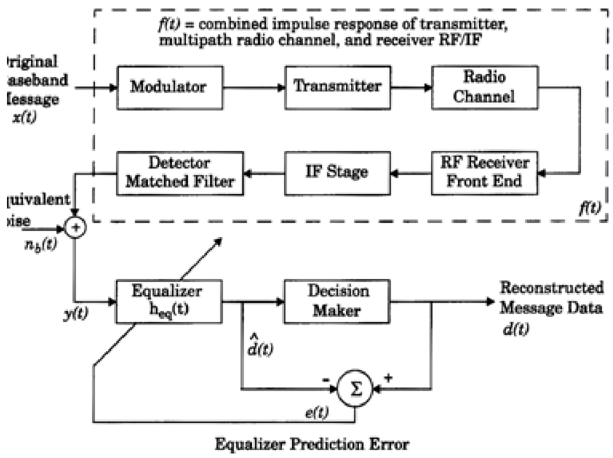

Fig (1) A simplified Block Diagram of a digital communication system using adaptive channel equalizer at the receiver [10]

Types Structures & Algorithms

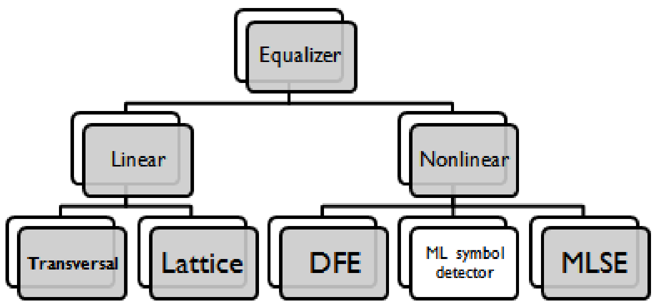

Fig (2): Equalizer classification

i. Linear Equalizer (Transversal FIR adaptive filter)

A linear equalizer can be implemented as FIR filter also called as transversal filter. In such an equalizer the current and past values of the received signal are linearly weighted by the filter coefficient and summed to produce the output. It is very effective on channels, such as wire line telephone channels, where the ISI is not severe. The basic limitation of a linear equalizer is that it performs poorly on channels having spectral nulls.

ii. Non-linear (Decision feedback) Equalizer

It arguments a linear equalizer by adding a filtered version of previous symbol estimates to the original filter output.

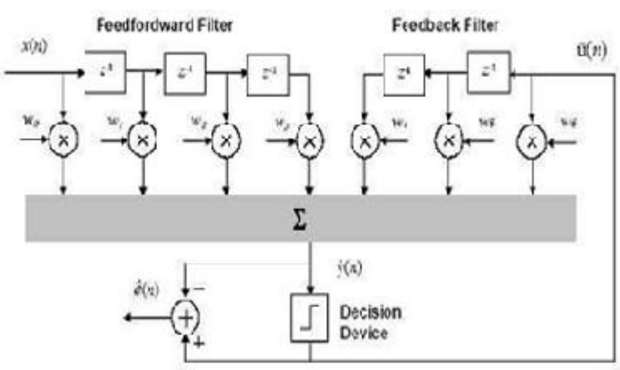

Fig (3) Simple Block Diagram of DFE[9]

Non linear Equalizer (DFE):-

The Decision feedback equalizer consists of two filters, the first filter is called a feed forward filter & it is generally a fractionally spaced FIR filter with adjustable tap coefficients. The second is a feedback filter, its input is the set of previously detected symbols, and the output of the feedback filter is subtracted from the output of the feed forward filter to form the input to the detector. The taps coefficients of the feed forward & feedback filters are selected to optimize some desired performance measure, for mathematical simplicity the MSE criteria is usually employed,& a various adaptive filtering algorithms are used to implement an adaptive DFE.

iii. Description of Adaptive Algorithms

The most prominent family of the adaptive filtering algorithm is least mean square (LMS).

Various categories of LMS Algorithm are used:

1. Normalized LMS
2. Signum LMS
3. Signum-signum LMS
4. Signum repressor LMS

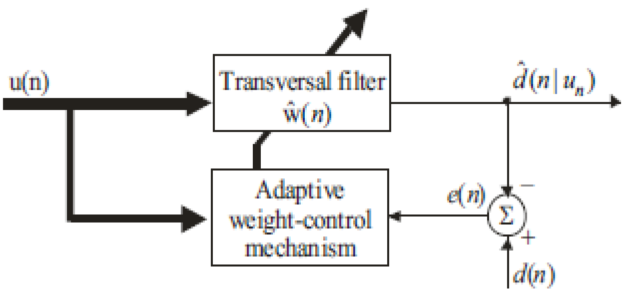

Fig (4).Basic Block diagram of Adaptive Channel Equalizer Based on gradient decent algorithms [1]

u (n)= Input signal

h (n) = system response

e (n) = Error signal

d (n)= desired signal.

$\hat{d}$(n)=Transversal FIR filter output

$\hat{w}(n)$ = weight vector of Transversal FIR filter

ξ (n) = E [e (n)] ……………………………….(1)

The error estimation e (n) is

e (n) = d(n) – $\hat{d}$(n) ……………………………...(2)

$\hat{d}$(n)=w (n)*u (n) …………………………… (3)

iv. Least mean square (LMS) with fixed step size

Least mean square filter is designed using a transversal (tapped delay line) structure. It has two practical features:

Simple to design & very effective in performance have made it highly popular in various applications. LMS algorithm is a linear adaptive filtering algorithm, which summarized as follow:

The least-mean square (LMS) algorithm updates the linear filter coefficients such that the mean square error (MSE) cost function to be minimized .it perform the following operation to update coefficients of the adaptive filter.

Calculates the error signal *e* (*n*) by using the equation (2).Coefficient updating equation is

W (n+1) = w (n) + µ u (n) e (n)……………. (4) [8]

Where µ is the fixed step size of the adaptive filter (n) is weight vector and u (n) is the input signal vector.

v. Least mean square (LMS) algorithm with variable step size (VSSLMS)

VSSLMS algorithm was introduced in which the step size was adapted that each iteration using the instantaneous power of the error. Here µ can respond to time varying channel parameters thus leading to high performance (low MSE).

VSSLMS algorithm uses the following variable step size adjustment scheme:

Let d(n) be the desired signal of the adaptive filter

$d(n) = u^T(n)h(n) + v(n)$ ……………………..….(5)

Where h(n) denote coefficient vector of the unknown system with length M.

x (n) is the input vector (n) is the (AWGN)system noise that is independent of x(n)we express a prediction error e(n) as

$e(n) = d(n) - x^T(n)w(n)$ ………………......... (6)[5]

Where w(n) denoting coefficient vector of the adaptive filter at iteration n as w(n)

By using squared instantaneous prediction error & constant parameters to update the step size as

$\mu(n+1) = \alpha\mu(n) + \gamma e^2(n)$ ………………….. (7)[5]

Where 0<α<1,γ>0 & µ(n+1) is restricted to some predefined values $[\mu_{min}, \mu_{max}]$

Filter coefficient vector update recursion is given by

$w(n+1) = w(n) + \mu(n)e(n)x(n)$ ……… (8)[5]

II. Structure of the Proposed Model

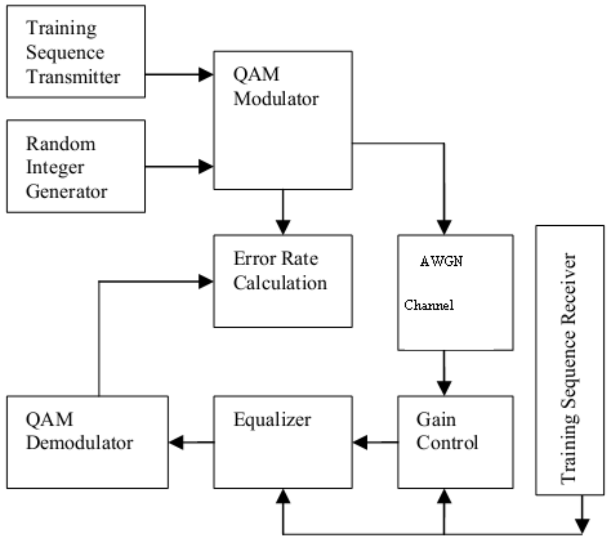

Fig (5) Structure of the proposed model [1]

The structure employed in this paper consists of random integer generator, QAM modulator, AWGN fading channel, equalizer and QAM demodulator & is implemented using MATLAB as shown in Figure (5). It follows the topology suggested in base paper [1].Simulation is being carried out by varying E_b/N_o ratio of AWGN fading channel for the linear & non linear equalizers based on fixed and variable step size LMS algorithm and its variants. The output is observed in the form of bit error rate (BER), number of errors and the number of bits processed.

III. Simulation Results and Analysis

In this section we present results of several experiments for comparison. All simulations are conducted using the same setup as given in fig (5).Moreover all parameters are chosen as per the recommendations given in their corresponding publications. All simulations plot are obtained by using QAM as a modulation technique & additive white Gaussian noise (AWGN) as a fading channel.

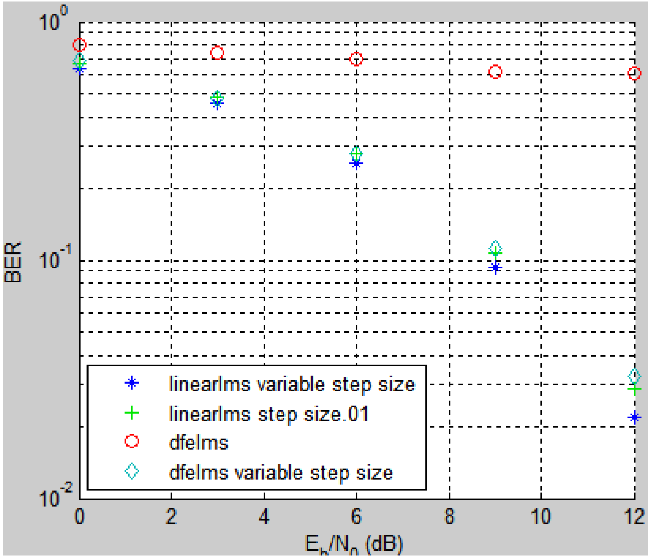

Fig (6): Linear & DFE Equalizer based on LMS algorithm for fixed μ=0.01 & variable step size.

It may be inferred from above that linear LMS (variable step size) gives most optimum results

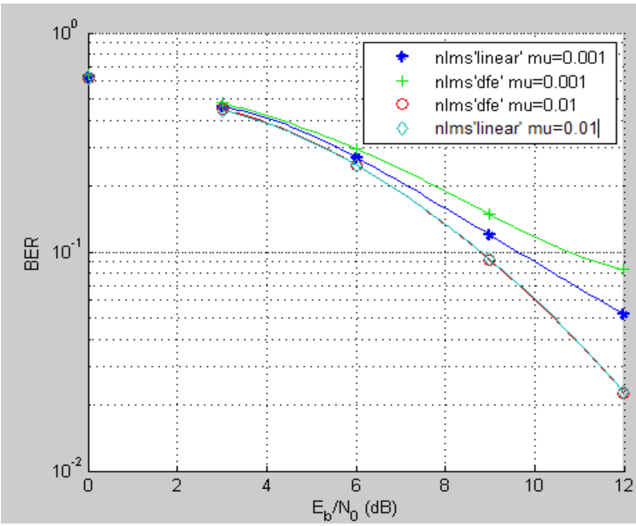

Fig (7): Linear & non linear DFE based on NLMS at different values of step size.

From above fig. it is shown that NLMS at μ=0.01 gives lowest BER amongst the four.

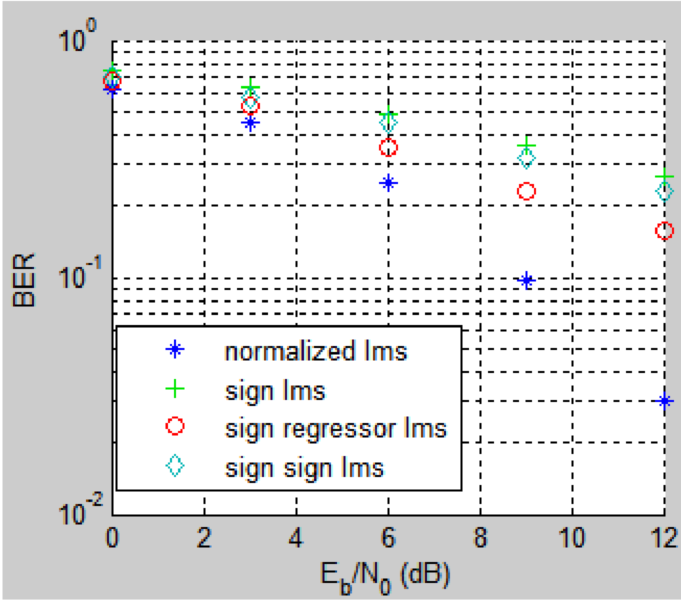

Fig (8) LMS variants (step size μ=0.01)

From above we may conclude that NLMS performs much better than others.

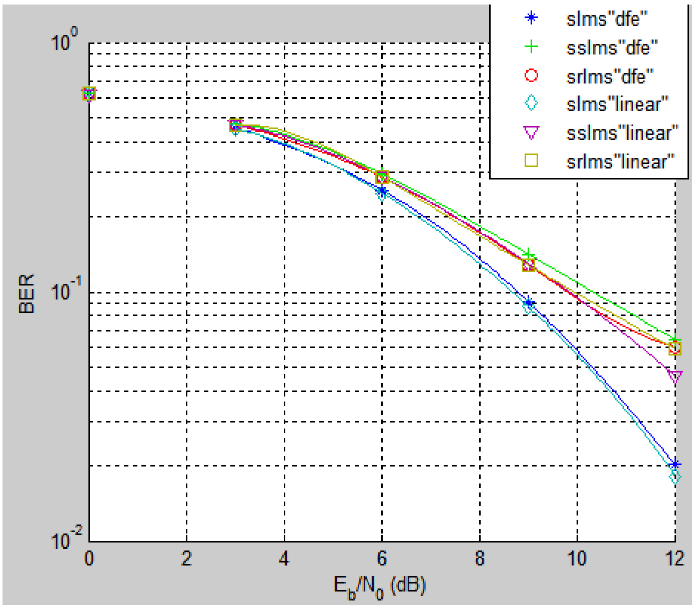

Fig (9): Linear & non-linear (DFE) Equalizers based on SLMS, SSLMS, SRLMS algorithms step size=0.001

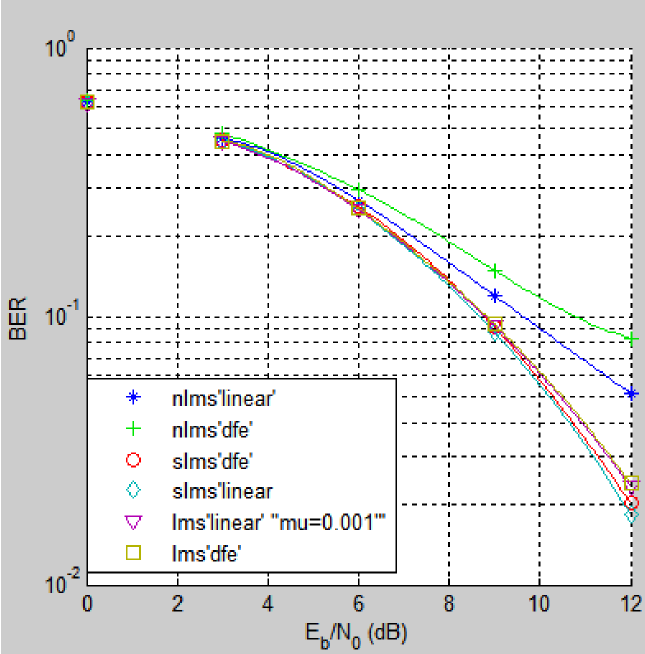

Fig (10): Linear & DFE Equalizer based on LMS, NLMS & SLMS with AWGN channel (μ=.001)

From above plot it can be concluded that SLMS linear equalizer have lowest BER over all others.

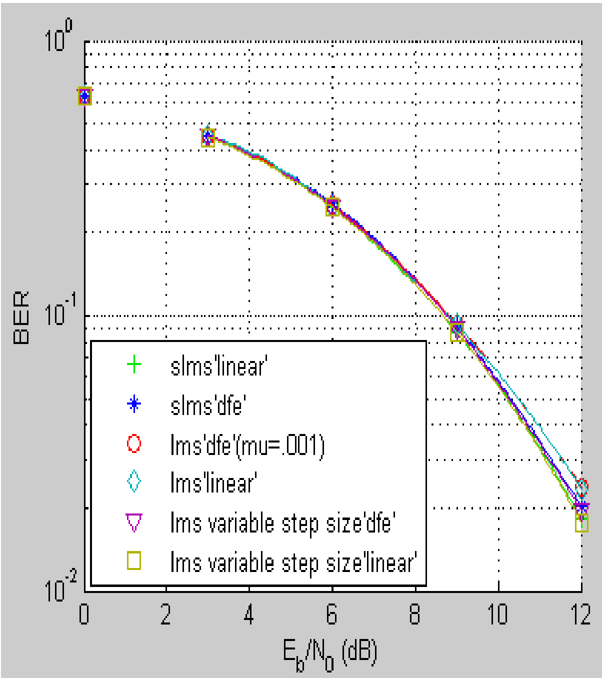

Fig (11): Linear & DFE Equalizer based on SLMS, LMS (fixed step size μ=0.001 & variable step size) algorithm

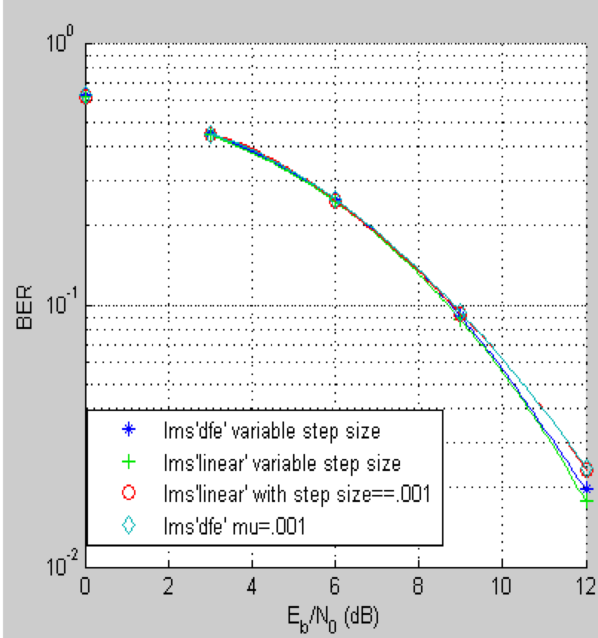

Fig (12): Linear & DFE Equalizer based on LMS algorithm for fixed step size μ=0.001 & variable step size

From above plots, it is thus clear that performance of variable step size LMS is better.

Simulation Results in Tabular Form:

Name Of Algorithm	Type of Equalizer	Step Size (μ)	Bit Error Rate(BER) values for different E_b/N_o				
			E_b/N_o 0DB	3 dB	6 dB	9 dB	12 dB
LMS	Linear	(VSS)	0.62	0.446	0.247	0.087	0.017
	DFE	VSS	0.630	0.451	0.250	0.090	0.019
SRLMS	Linear	μ=0.001	0.622	0.468	0.289	0.127	0.058
	DFE	μ=0.001	0.623	0.467	0.287	0.127	0.059
SLMS	DFE	μ=0.001	0.626	0.454	0.255	0.091	0.020
	Linear	μ=0.001	0.623	0.449	0.248	0.087	0.018
SSLMS	DFE	μ=0.001	0.622	0.472	0.297	0.141	0.064
	Linear	μ=0.001	0.618	0.466	0.290	0.130	0.046
NLMS	Linear	μ=0.001	0.629	0.459	0.270	0.119	0.051
	DFE	μ=0.001	0.642	0.479	0.296	0.148	0.082

IV. Conclusion & Further Work

Many adaptive algorithms have been proposed to achieve fast convergence rate, rapid tracking, & low BER in past two decades. This paper summarized several promising algorithms & presented a performance comparison by means of extensive simulation. Here we implemented the system identification problem using the linear and non linear (DFE) equalizer based on various adaptive filtering algorithms such as LMS & its variants with fixed step size and variable step size (VSSLMS). From the simulation results plots & table it is concluded that VSSLMS linear equalizer outperform other types of adaptive algorithms in terms of Bit Error Rate. The work done in this paper will be further optimized by some derivative free training algorithms on adaptive filters & also with neural network.

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