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ADAPTIVE CHANNEL EQUALIZER FOR WIRELESS COMMUNICATION SYSTEMS

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ABSTRACT

The Data rates and spectrum efficiency of Wireless Mobile Communication have been significantly improved over the last decade or so. Recently, the advanced systems such as 3GPP LTE and terrestrial digital TV broadcasting have been sophisticatedly developed using OFDM and CDMA technology. In general, most mobile communication systems transmit bits of information in the radio space to receiver. The radio channels in mobile radio systems are usually multipath fading channels, which cause inter symbol interference (ISI) in the received signal. To remove ISI from the signal there is a need of strong equalizer which required the knowledge on the channel impulse response. (LMS) Least Mean Square, (RLS) Recursive Least squares and (PSO) Particle swarm optimization algorithms are used to implement the adaptive channel equalizer. The results are measured in terms of mean square error (MSE) and bit error rate (BER) Vs the number of iterations.

KEYWORDS: LMS, RLS, PSO, MSE, BER

1. INTRODUCTION

The growth in communication services during the past five decades has been phenomenal. Satellite and fibre optic networks provide high-speed communication services around the world. Currently, most of the wired line communication systems are being replaced by fibre optic cables which provide extremely high bandwidth and make possible the transmission of a wide variety of information sources, including voice, data, and video. With the unimaginable development of Internet technologies, efficient high-speed data transmission techniques over communication channels have become a necessity of the day. As the rate of the data transmission increases to fulfil the needs of the users, the channel introduces distortions in data. One major cause of distortion is Inter Symbol Interference (ISI). In digital communication, the transmitted signals are generally in the form of multilevel rectangular pulses. The absolute bandwidth of multilevel rectangular pulses is infinity. If these pulses passes through a band limited communication channel, they will spread in time and the pulse for each symbol maybe smeared into adjacent time slot and interfere with the adjacent symbol. This is referred as inter symbol interference (ISI). Other factors like thermal noise, impulse noise, and cross talk cause further distortions to the received symbols. Signal processing techniques used at the receiver to overcome these interferences, so as to restore the transmitted symbols and recover their information, are referred to as “channel equalization” or simply equalization. In principle, if the characteristics of the channel are precisely known, then it is always possible to design a pair of transmitting and receiving filter that can minimize the effect of ISI and the additive noise. However, in general the characteristics of channel are random in the sense that it is one of an ensemble of possible channels. Therefore, the use of a fixed pair of transmitting and receiving filter designed on the basis of average channel characteristics, may not adequately reduce inter symbol interference. To overcome this problem adaptive equalization is widely used, which provides precise control over the time response of the channel. Adaptive equalizers have therefore been playing a crucial role in the design of high-speed communication systems. The data transmitted through a band limited communication channel suffers from linear, nonlinear and additive distortions. In order to reduce the effects of these distortions an equalizer is used at the receiver end. The function of the equalizer is to reconstruct the transmitted symbols by observing the received noisy signal [1,2,7]. Section 2 describes the system model, section 3 explains the LMS, RLS, and PSO algorithms, section 4 and 5 explains the simulation results and conclusion.

2. SYSTEM MODEL

In an ideal communication channel, the received information is identical to that transmitted. However, this is not the case for real communication channels, where signal distortions take place. A channel can interfere with the transmitted data through three types of distorting effects: power degradation and fades, multi-path time dispersions and background thermal noise. Equalization is the process of recovering the data sequence from the corrupted channel samples. A typical base band transmission system is depicted in Figure 1, where an equalizer is incorporated within the receiver.

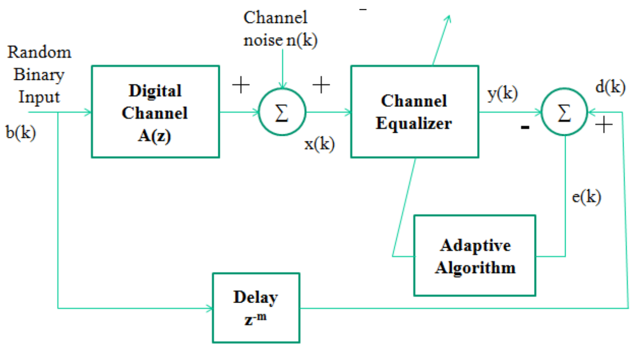

Figure 1: Block Diagram of Channel Equalization

Figure 1 shows a block diagram of a communication system with an adaptive equalizer in the receiver. If b (t) is the original information signal, and f (t) is the combined complex baseband impulse response of the transmitter, channel and the RF/IF sections of the receiver, the signal received by the equalizer maybe expressed as [6,8]

$$x(t) = b(t) \otimes f^*(t) + n_b(t) \tag{2.1}$$

where f* (t) is the complex conjugate of f(t) , $n_b(t)$ is the baseband noise at the input of the equalizer, and 0 denotes the convolution operation. If the impulse response of the equalizer is $h_{eq}(t)$, then the output of the equalizer is

$$y(t) = b(t) \otimes f * (t) \otimes h_{eq}(t) + n_b(t) \otimes h_{eq}(t) \tag{2.2}$$

where g(t) is the combined impulse response of the transmitter, channel, RF/IF sections of the receiver, and the equalizer. The complex baseband impulse response of a transversal filter equalizer is given by

$$h_{eq}(t) = \sum_n C_n \delta(t - nT) \tag{2.3}$$

where are the complex filter coefficients of the equalizer. The desired output of the equalizer is x(t), the original source data. Assume that $n_b(t) = 0$. Then, in order to force d (t) = x (t) in equation (2.2), g (t) must be equal to

$$g(t) = f^*(t) \otimes h_{eq}(t) = \delta(t) \tag{2.4}$$

The goal of equalization is to satisfy equation (2.4). In the frequency domain, equation (2.4) can be expressed as

$$H_{eq}(f)F^*(-f) = 1 \tag{2.5}$$

Where $H_{eq}(f)$ and $F(f)$ are Fourier transforms of $h_{eq}(t)$ and $f(t)$, respectively. Equation (2.5) indicates that an equalizer is actually an inverse filter of the channel. If the channel is frequency selective, the equalizer enhances the frequency components with small amplitudes and attenuates the strong frequencies in the received frequency spectrum in order to provide a flat, composite, received frequency response and linear phase response. For a time-varying channel, an adaptive equalizer is designed to track the channel variations so that equation (2.5) is approximately satisfied.

3. ALGORITHM IMPLEMENTATIONS

3.1 Least Mean Square (LMS) Algorithm

Algorithm for LMS

- Read the data signal.
- Calculate length of the data signal.
- Generate random noise and add it to data signal.
- Apply LMS algorithm on noisy data signal
- Filter output is y(n)=w(n)x(n) Where x(n) is noise data signal.
- Error e(n)=d(n)-y(n) Where d(n) is part of original data signal
- Filter coefficient updating: w(n+1)=w(n)+μx(n)e*(n)
- Update filter coefficient and minimize the error[4]

3.2 Recursive Least Squares (RLS) Algorithm

Algorithm for RLS

- Read the data signal.
- Calculate length of the data signal.
- Generate random noise and add it to data signal.
- Apply RLS algorithm on noisy data signal
- Initial weight vector w(n)=0
- Error e(n)=d(n)-y(n) Where d(n) is part of original data signal
- Filter coefficient updating h(n)=h(n-1)+k(n)e*(n)
- Update filter coefficient and minimize the error [4]

3.3 Particle Swarm Optimization (PSO)

Particle Swarm Optimization Algorithm

In PSO a swarm consists of a set of volume-less particles (a point) moving in a D dimensional search space, each representing s potential solution. Each particle flies in the search space with position and velocity which are dynamically adjusted according to its own as well as its companions flying experiences. Particles are assumed as tap weights in the adaptive channel equalizer. Initially tap weights are randomly selected. The tap weights are continuously updated by taking number of iterations. The continuous updation depends on position and velocity of the particles. Initial position error is maximum. For each iteration, position best(pbest) and global best(gbest) is calculated. The concept of movement of particles of PSO described in Figure 2.

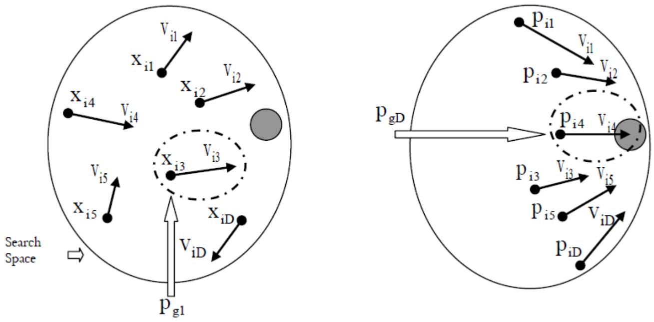

Figure 2: Procedure of Particle Swarm Optimization

The ith particles is represented by a vector: $X_i=[X_{i1}, X_{i2}, \ldots \ldots X_{id} \ldots . X_{iD}]$, the best previous position (the position giving the best fitness value) of the ith particle is recorded and represented as P = [pi1,pi2 ....pid …..piD]. At each iteration, the global at best particle in the swarm is represented by P = [pg1, pg2 ....pgd …..pgD]. The rate of change of position of the ith particle is represented as

V= [vi1,vi2 ....vid …..viD]. [3,5]

The maximum velocity and the range of particles are given by

V max = [vmax1, vmax2 ......vmaxd ....vmaxD] and

X max = [xmax1, xmax2 .....xmaxd ......xmaxD].

Each particle tries to modify its position using the following information:

- The current positions,
- The current velocities,
- The distance between the current position and pbest,
- The distance between the current position and the gbest.

The velocity and position of the dth element of the ith particle at (k+1)th search from the knowledge of previous search are modified as per the following

Vid(k+1)=w(k)*vid(k)+c1*r1*(pid(k)- xid(k))+

c2*r2*(pgd(k)+xid(k))

where

i = 1,2 ....N , d = 1,2 ....D and N is the number of particles . The symbols r1 and r2 represent random numbers between 0 and 1. Similarly c1 and c2 denote acceleration constants that pull each particle towards its best and global best positions. The acceleration constants are usually taken as 2.05 for most applications. The inertia weight, W is employed to control the impact of pervious history of velocities on the current one in order for tradeoff between the global and local exploitations. At early stage of optimization it is desirable that the individual particles wonder though the entire search space, without clustering around the local optima. On the other a hand, during later stages it is very important to enhance convergence toward the global optima so as to find optimum solution efficiently .Large inertia weight enables the PSO to explore locally. So a self-adaptive strategy is introduced such that the value of w is decreased linearly as the generation goes on increasing .The time-varying inertia weight is given by

$$w(k) = (w_i - w_f)(\frac{1-k}{k}) + w_f$$

where k is the search space number, w_i and w_f are the initial and final value of inertia weights taken values 0.4 and 0.9 respectively. I is the maximum number of search or generation.

4. SIMULATION RESULTS

4.1Simulation Results for Channel Equalization Using LMS Algorithm

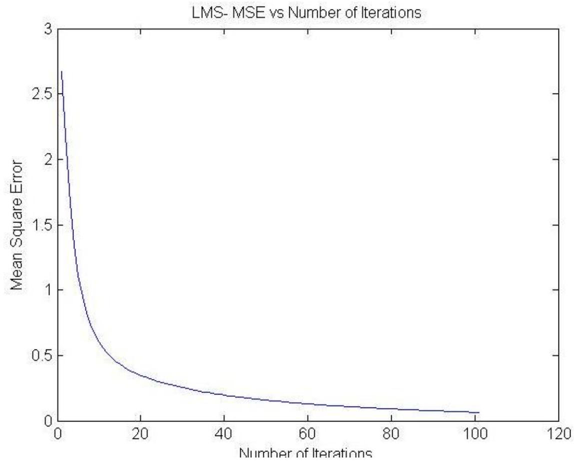

Figure 4.1(a): Error Plot Obtained Using LMS Algorithm

MSE is minimized by taking more number of iterations using LMS algorithm.

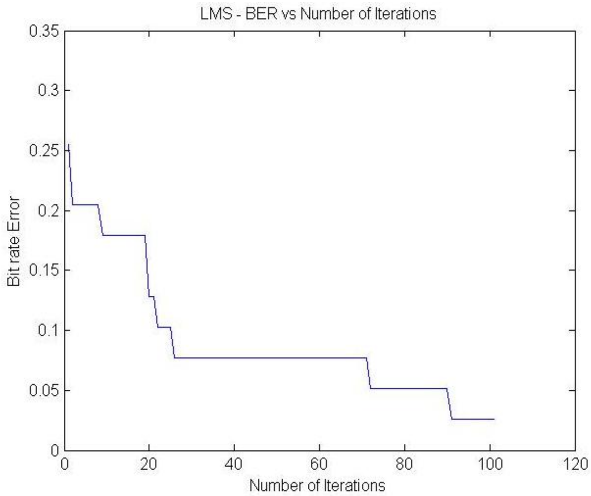

Figure 4.1(b): Bit Error Rate Plot Using LMS Algorithm

BER is minimized by taking more number of iterations using LMS algorithm.

Table 1: Tabular form for Different μ Using LMS Algorithm

μ value	0.01	0.5	0.998
Avg. MSE	0.5246	4.92e+030	5.10e+042

Avg.MSE is depends on the value of ‘ μ ’ in the LMS algorithm.

4.2. Simulation Results for Channel Equalization Using RLS Algorithm

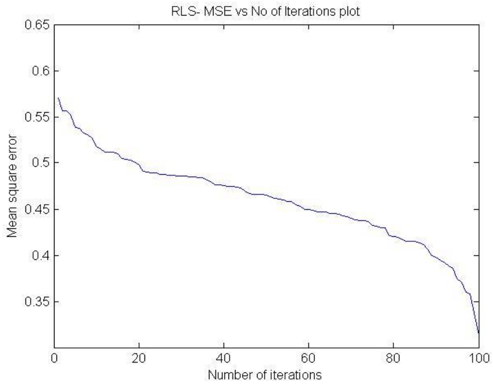

Figure 4.2(a): Error Plot Obtained Using RLS Algorithm

MSE is calculated using RLS algorithm for the forgetting factor λ =0.998.

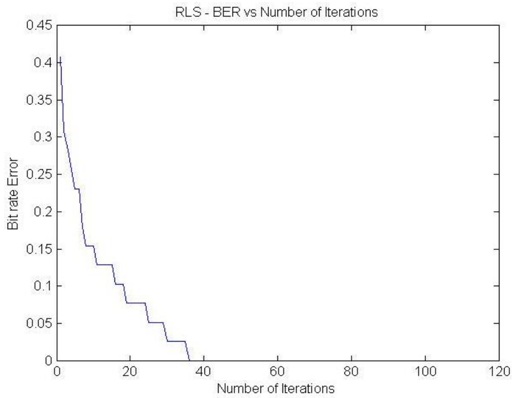

Figure 4.2(b): Bit Error Rate Plot Using RLS Algorithm

MSE, BER are reduced by increasing the number of iterations when compare with LMS, RLS shows better performance in reducing MSE, BER.

Table 2: Tabular form for Different λ Using RLS Algorithm

$\boldsymbol{\lambda}$ ***value***	0.01	0.5	0.998
Avg. MSE	30.3244	0.6011	0.4392

Avg.MSE is depends on the value of ' λ ' in the RLS algorithm.

4.3 Simulation Results for Channel Equalization Using PSO Algorithm

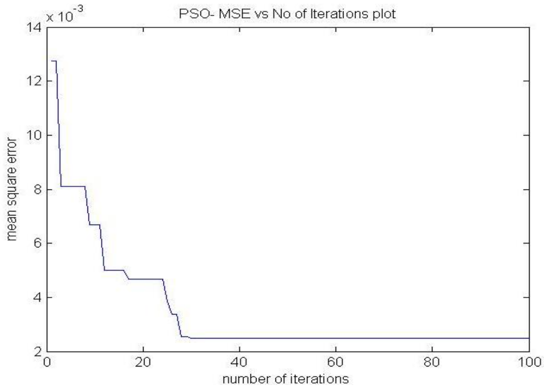

Figure 4.3(a): Error Plot Obtained Using PSO Algorithm

MSE is reducing by increasing the number of iterations.

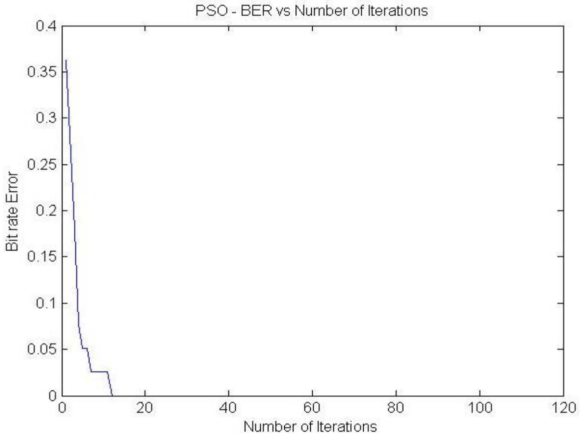

Figure 4.3(b): Bit Error Rate Plot Using PSO Algorithm

BER is reduced to zero by increasing the number of iterations.

Table 3: Tabular form for Different Iterations Using PSO Algorithm

No of Iterations	10	50	100
Avg.MSE	0.2019	0.1263	0.0732

PSO shows better performance than RLS. BER is almost reduced to zero with the same number of iterations taken in both the algorithms.

4.4 Comparison of Three Algorithms

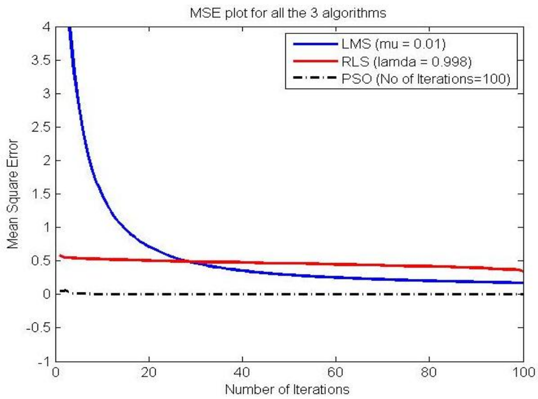

Figure 4.4: Comparison in Mean Square Error Plot of all Three Algorithms

Table 4: Tabular Form in Comparison of all the Three Algorithms for Channel Equalization

Optimization Techniques	**Number of Iterations**	**Mean Square Error(MSE)**	**Elapsed Time (sec)**
LMS (μ =0.01)	100	0.5246	0.2623
RLS (λ =0.998)	100	0.4392	0.03164
PSO	100	0.0732	0.00397

Rate of convergance is fast in PSO rather than LMS and RLS as in the table (4).

5. CONCLUSIONS

The main problem that is being addressed in this project is related the most important issues associated with the wireless communication channels, namely Inter-Symbol Interference (ISI) caused by multipath nature. ISI is causing data errors and hence the main obstacle in the way of reliable and fast wireless digital communications. The common methods to eliminate or sufficiently mitigate the effect of distortion caused by ISI are collectively known as channel equalization. Equalization techniques are of enormous importance in the design of high data rate wireless systems. They can combat for inter symbol interference even in mobile fading Channel with high efficiency. In this project performance of adaptive channel equalizer with three algorithms used to reduce ISI is presented. The analysis showed that all the three algorithms are successful in reducing ISI to some extent. The Results for all the three algorithm are compared with the help of three parameters they are Mean Square Error (MSE), Bit Error Rate (BER) and Time of Convergence (TOC). It was concluded that the Particle Swarm Optimization (PSO) offers two advantages than LMS and RLS algorithms, first one in order to reduce the MSE, however the second one to reduce the TOC.

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