LMS Equalizers for Different Blocks and over sampling factor Rayleigh Fading Channel with Normalized Impulse Response

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Abstract: Block based linear equalizers are used in communication systems for removing the effects of the channel degradation. Researchers have developed linear equalizers using different modulations techniques. This paper evaluates the BER performance of communication system for different over sampling factor of pulse shaping filters. The effect of the different number of transmitting blocks is also evaluated with different M-PSK modulations for the LMS linear equalizers over the frequency selective Rayleigh fading channel. The Rayleigh channel is modeled with multipath channels and normalized channel impulse response. For evaluating the performance of the linear equalizers M-PSK sizes are varied and BER is calculated. It is found that linear equalizers with normalized channel impulse response have better BER performance. It is observed that the BER probability is reduced with increasing the number of blocks. Performance is also compared for the different equalizer weights and block sizes.

Keywords: LMS Linear Equalizers, RLS Equalizers, Rayleigh fading channel, Bit error rate.

1. Introduction

The LMS linear equalizers are designed to give minimum error rates and to reduce the Inter symbol interference (ISI). Block based communication transmitters have gained popularities in last two decades [8]. The equalizers may be linear or non-linear in nature and are used to simplify the demodulation process. Rayleigh fading [5] phenomenon commonly occurs in communication systems due to diffraction, and scattering of the transmitted waves from the structures like vehicles or buildings as explained in Figure 1. The pulse shaping filters are widely used in communication system for reducing these ISI. Thus in this paper the performance of block based Least mean square (LMS) linear equalizers [2, 15] are evaluated using different modulation techniques. Paper also investigated the effect of the different oversampling factors of pulse shaping filters on the BER performance.

With the large number of communication users the higher size modulation methods are frequently adopted hence it is required to evaluate the performance of modulation techniques in the fading channels.

![Figure 1 Fading in Multipath communication system](image)

The LMS equalizers are used by many researchers [2, 6 and 8]. These equalizers used because of their simplicity of design and are used in linear time varying systems. Therefore in this paper the performances of the linear equalizers have been compared for block-LMS algorithm. The LMS method is used globally when desired values are known [15]. This method is computationally simple to implement. If the desired symbols are not correct, it does not converge. Usually LMS algorithm has lower computational complexity but higher convergence rate.
Researchers have used many pulse shaping filters to improve the communication system performance Viz. Gaussian filters, cosine filters, and raised cosine filters. The sampling time is inversely proportional to the oversampling factor of the pulse shaping filter. Therefore in this paper different oversampling factors are varied for evaluating the performance of the communication system. The LMS equalizer is designed in a training mode. But, since every equalizer perform differently in different working environment hence it is desired to identify and analyze the performance of equalizers over modulation techniques

In this paper performance evaluations of LMS equalizers with different M-PSK modulations and block sizes for the Rayleigh fading channels are presented. For evaluating the performance various equalizer parameters Viz. PSK size, number of blocks, are varied and Bit error rate (BER) is evaluated. In this paper after introduction section 2 discusses brief literature review of the existing work then various issues in fading channels are discussed in the section 3. The proposed communication system block diagram is explained in the section 4. Section also explained the parameters of pulse shaping filters and channel normalization. The linear equalizer along with LMS algorithm is described in the section 5. The results of performance evaluation are given in the section 6 followed by the conclusion in section 7.

2. Literature Review

The linear equalizers are commonly used by the researchers to reduce the channel distortions. The simplest algorithm from the performance and complexity point of view is the LMS algorithm [2, 8]. The LMS algorithm based equalizer is not effective when the desired symbols are incorrect such as under the presence of large noisy channels.

Alireza et al. [2] have proposed a blind linear equalization, and data detection, for an efficient, and un-coded transmission over a frequency selective Rayleigh fading channel. The method uses the LMS algorithm for their analysis. The maximum Doppler shift is taken as 20 Hz. and SNR is varied from 0-20 dB. Wing et al. [5] have presented the method for equalization of linear frequency selective fading channel which reduces the effect of inter-symbol interference. The method optimizes transmitter and receivers filters impulse response to reduce the inter symbol interferences.

Sabita et al. [4] have presented a comparative analysis of different modulation schemes is performed in a fading environment using the adaptive equalization technique for the mitigation of fading distortion. The comparison is made at a fixed SNR of 35 dB. They have concluded that QAM performs better than QPSK technique. Many researchers have adapted the variable Tap length for improving the equalizer performances [6, and 8]. Yu Gong et al. [6] have proposed a MMSE equalizer which jointly adopts the tap length and decision delay for improving the performance. But method was computationally complex. Kiran Kuchi [9] has presented the performance comparison of the zero forcing (ZF) and MMSE linear equalizers under multiantenna Rayleigh fading channel system.

X. Ma and W. Zhang, [10] have explained the basic fundamental limitations of the linear equalizers: such as capacity, diversity, and computation complexity. Jaymin et al. [11] have presented a comparative analysis of the MLSE, LMS and RLS non linear adaptive equalization algorithms for the wireless digital communication. Each algorithm is tested for the BPSK, 4PSK and 16QAM modulation techniques.

Fu Shaozhong et al.[12] have updated the length of LMS equalizer for using exponential function. Method reduces the average number of iterations and thus converges faster than standard LMS algorithm. Veeraruna et al. [13] have analyzed the performance of LMS linear equalizer in the decision directed mode over the fading channel. The equalizer is approximated by the one dimensional differential equation (ODE) but method seems slightly complex. Garima Malik et al. [14] have given a brief overview of the RLS and LMS adaptive equalizers. They have concluded that bandwidth efficient communication is possible by compensating the time varying channel distortions using equalizers.

But the performance of different M-PSK modulation techniques over the fading channels is not yet evaluated for different adaptive equalizers. Also it is needed to evaluate the performance of linear equalizers for different velocities corresponding to different maximum Doppler shifts of the fading channel. These are the prime goals of this paper

3. Channel Model

In the radio communication channels multipath signals interfere with the actual signal and causes reduction in signal strength. The phenomenon is known as fading. This is the major reason of signal degradation in the wireless communication. The most common fading model is the Rayleigh fading.

3.1 Rayleigh Fading

The Rayleigh fading model assumed that communication channel induces varying amplitudes in time as per the Rayleigh distribution [15]. The Rayleigh distribution is the most widely used to describe the metropolitan environment. The received value of the faded signal at any time t is represented as \( r(t) = |x(t)| \). The Rayleigh distribution of the received resultant complex faded signal [5] is given as;

\[
p_{\chi\chi}|x\rangle = \frac{x}{\sigma^2} e^{-\frac{x^2}{2\sigma^2}} (x \geq 0)
\]  

In Rayleigh distribution described above, \( x = \text{transmitted signal} \) and \( \sigma \) is defined as the RMS value of the received. \( P_{\chi\chi} = \)
Voltage signal before signal detection, and $\sigma^2$ is the average power of the received signal before net signal detection.

4. Proposed Communication System

The block diagram of the proposed communication system is shown in Figure 3. Proposed system use linear equalizers in a training mode operation and M-PSK modulation. System uses the Pulse shaping before modulation for efficiently minimizes the inter symbol interference (ISI) induced by the channel.

4.1 Pulse Shaping

In order to reduce the ISI the paper uses the Square root raised cosine (SRRC) pulse shaping filters. The filter is spanned by 8 symbol periods, and the roll off factor of the SRRC filter is set to the 0.25. The sample duration of the each transmitted signal is defined as;

$$ T_{samp} = \frac{T_s}{OSF} $$

(2)

Where, OSF is the over sampling factor of the shaping filter and $T_s$ is the sampling time in seconds. The cutoff frequency or the sampling frequency is set to the Nyquist rate as;

$$ F_s = \frac{1}{2 \times T_s} $$

(3)

The order of the shaping filter is given as;

$$ \text{Filter order} = \text{Number of symbol periods} \times \text{OSF} $$

(4)

Paper computed the impulse response of SRRC for different symbol duration with different OSF.

4.2 Channel Normalization

The fading channel is modeled with a linear and time-varying Channel Impulse Response (CIR) function $h(t, \tau)$ is normalized with respect to the maximum absolute impulse responses.

5. Linear Equalizers

An equalizer is a adaptive filter which is capable of adapting time-varying properties of the communication channel [16]. In this paper the equalizer is designed in a training mode. It can be implemented by performing the tap weight adjustments periodically or continually. These periodic adjustments are accomplished by periodically transmitting a preamble or short training sequence of digital data known by the receiver. Continual adjustment are accomplished by replacing the known training sequence with a sequence of data symbols estimated from the equalizer output and treated as known data.

Architecture of the adaptive equalizer is shown in the Figure 4 below. The coefficients of the filters are called as weights of the system and are updated according to the type of equalizer algorithm.

5.1 LMS algorithm

The standard least mean squares (LMS) algorithm is a type of adaptive filter which adapts the filter coefficients to produce the least mean squares error signal between the desired and the actual signal. LMS is a stochastic gradient descent method in which the filter is adapted based on the error at the current time. LMS filter is built around a transversal (i.e. tapped delay line) structure. The updated weights are calculated as;

$$ w_k(n + 1) = L_{factor} \cdot w_k(n) + G \cdot e^*(n) $$

(6)

Where, $L_{factor}$ is the leakage factor which is 1 for standard LMS algorithm.
6. Experimental Results

In this paper the Bit error rate (BER) of the various M-PSK modulation techniques are compared for different linear adaptive equalizers. The simulation is performed on MATLAB software. Paper model the channel with normalized Channel Impulse Response (CIR). Figure 5 compare the BER performance of the LMS linear equalizer respectively for QPSK modulation. It is found that using the normalized CIR improves the performance of linear equalizers significantly. It is found that LMS equalizers can achieve minimum BER order of $10^{-4}$ as in Figure 5. The Figure 6 gives the comparison of the pulse shaping filter response for two different OSF’s. It can be seen that the increasing the OSF reduces the sapling time of the symbol. Figure 7 compare the BER performance of the LMS linear equalizer respectively for QPSK modulation with different over sapling time. The minimum OSF can be 2 but it is the critical limits to reduce the ISI and to satisfy the Nyquist criterion. Thus it is proposed to use the OSF of 4 for communication systems.

![Figure 5 BER for LMS algorithm with fading channel, for step size of 0.1](image)

![Figure 6 Comparison of magnitude and phase responses of the Pulse shaping filters upper response is for OSF=4 and lower is for OSF=8.](image)

The various input parameters used for the simulation are given in the Table 1.

<table>
<thead>
<tr>
<th>Variable</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>N bit PSK</td>
<td>2-10</td>
<td>Bits per PSK Symbol</td>
</tr>
<tr>
<td>Ts</td>
<td>1e-6</td>
<td>Sampling time</td>
</tr>
<tr>
<td>M</td>
<td>4 – 1024</td>
<td>Size of Modulation</td>
</tr>
<tr>
<td>NTap</td>
<td>4</td>
<td>Length of equalizer</td>
</tr>
<tr>
<td>xPayload</td>
<td>randi(1,400)</td>
<td>Number of data bits per block</td>
</tr>
<tr>
<td>Fd</td>
<td>30 Hz</td>
<td>Doppler Shift</td>
</tr>
<tr>
<td>D</td>
<td>1e-6[0 4 8 12]</td>
<td>Multipath Delay vector</td>
</tr>
<tr>
<td>G</td>
<td>[0 3 6 9] dB</td>
<td>Multipath Gain vector</td>
</tr>
<tr>
<td>nBlocks</td>
<td>30, 40 and 50</td>
<td>Number of blocks</td>
</tr>
<tr>
<td>Step</td>
<td>0.1</td>
<td>LMS Step size</td>
</tr>
<tr>
<td>OSF</td>
<td>2, 3, 4, and 8</td>
<td>Oversampling factor</td>
</tr>
</tbody>
</table>
The BER performance of the different M-PSK with $M = 4, 16, 256, 512$ and 1024 are, compared for LMS equalized frequency selective fading channels in Figure 8. It can be observed that up to around 16 dB the proposed system with normalized CIR performs approximately similar for all PSK sizes. It can be also observed that proposed method performs better for even PSK size of up to 512 and 1024. the BER is very much compatible to the smaller PSK sizes.

As another experiment in this paper the BER is calculated by varying the different number of blocks as 30, 40 and 50. The comparison of the BER is shown in Figure 9. It can be observed that the probability of error is reduced by increasing the number of blocks. The Figure 10 presents the comparison of the constellation diagram of the QPSK modulation with LMS equalizer for flat and fading channel with equalized sequences. It is clear that more scattered pattern is there without equalizer.
7. Conclusion

In this paper evaluation of the performance of linear LMS equalizer is compared for different M-PSK modulations. For improving the BER performance of PSK modulation channel is model with normalized impulse response.

It is found that proposed method improves the performance of the PSK modulations at the higher size of 512 and 1024 significantly. It is found that the RLS equalizer gives better performance than LMS in terms of minimum BER. The linear equalizers are widely used because of its simplicity. These are useful where channel parameter does not vary frequently. But the complexity increases linearly with the increasing tap weights.

Acknowledgments

I would like to thanks my guide Mr.Anoop Tiwari and Dr.Ravi Shankar Mishra for their technical support and guidance for this work. I would also like to thanks my family members for their support.

References


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