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CHANNEL ESTIMATION USING LMS ALGORITHM FOR DISCRETE WAVELET TRANSFORM BASED OFDM

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ABSTRACT

In wireless communication systems, to obtain channel state information (CSI) channel estimation is required. By doing estimation of channel bit error rate (BER) reduces. LMS algorithm gives best performance comparative to previous estimation technique LS and MMSE, the former is too sensitive to noise and later is too complex. LMS algorithm adopts the principle of steepest descent method and belongs to a group of methods known as stochastic gradient methods. This algorithm can adjust to modify the signals statistics; thus it is an adaptive filter.

Keywords: DWT-OFDM, Overview of LMS, Least mean square algorithm

I. INTRODUCTION

In wireless communication signal fading creates major problem, so the channel estimation is applied before demodulation to track the original signal. The channel estimation can be based on Least Square (LS) and Minimum Mean Square Error (MMSE) [8]. The LS is very simple but more sensitive to noise. MMSE gives improved results than LS but it is too complex [8]. As it is not easy every time to determine statistics for noise that is fairly similar with the randomly generates signal. Some of the signals change with very fast rate perspective to information in the noise cancellation process which necessitates the help of self regularized algorithms with the characteristics to converge rapidly. Least mean square (LMS) is very simple, efficient and generally used for signal enhancement.

The LMS algorithm can be distinguished from the steepest descent method with stochastic gradient that make use of the deterministic gradient for stochastic inputs in recursive computation of filter which are having the significant feature of simplicity[9]. This feature makes LMS algorithm standard over other linear adaptive filtering algorithms. Moreover, it does not require matrix inversion and measurements of applicable correlation functions.

The LMS algorithm that is an adaptive filter is also employed to estimate channel and to reduce the BER corresponds to signal to noise ratio in orthogonal frequency division (OFDM) systems. When transmitted signal passes through channel to the receiver, it becomes distorted cause of very sensitive situation, such as fading, noise and Doppler spread. That’s why LMS estimation is done at receiver initially, so that it can easily track the original signal and gives the output signal as desired.

This paper represents the channel estimation for discrete wavelet transform (DWT) based OFDM over frequency selective fading channel using LMS algorithm. The DWT-OFDM is based on wavelet transform concept and predicts best results on comparing with Fourier based OFDM over frequency selective fading channel with AWGN using time domain zero forcing equalization [3], because of rejection of cyclic prefix (CP) which was added to OFDM symbol in order to mitigate ISI (inter symbol interference) and ICI (inter carrier interference) in Fourier transform based OFDM. DWT-OFDM has one more advantage over FFT-OFDM that signal localization in time and frequency both.

Wavelet transform constructs wavelets that are small waves with limited duration, irregular and asymmetric and utilizes low pass filter s[n] and high pass filter r[n] of a Quadrature Mirror Filter bank (QMF), both assure the perfect reconstruction condition and orthonormal bases properties [2]. Down sampling is used to avoid the redundant samples. The two filtering and sub-sampling process can be expressed as [10]:

$$Y_{high}[k] = \sum_{n=0}^{\infty} x(n)s(2k-n) \quad (1)$$

$$Y_{low}[k] = \sum_{n=0}^{\infty} x(n)r(2k-n) \quad (2)$$

II. OVERVIEW OF LMS

LMS is based on stochastic steepest descent method. It consist of two basic process [5],

(1)**Filtering process**, which involves computing the output of a transversal filter produced by a set of tap inputs, and generating an estimation error between filter output and desired response.

(2)**Adaptive process**, that involves the automatic adjustment of tap weights of filter correspond to estimated error.

The combination of above two processes works together to makes up feedback loop around the LMS algorithm as shown in figure (1).

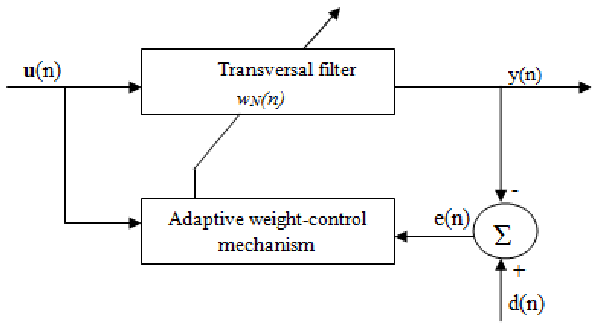

Fig.1. Block diagram of adaptive transversal filter

Where **u**(n) is N by 1 tap input vector that is wide sense stationary and $\boldsymbol{w}_N(n)$ is M by 1 tap weight vector. For wide sense stationary inputs LMS algorithm computes estimate value for tap weight vector $\boldsymbol{w}_N(n)$ whose expected value approaches the wiener solution w_o as the number of iterations approaches infinity. Desired signal d(n) is bring in filtering process along side input vector which is given to transversal filter. Transversal filter creates an output y(n) used as an estimate of desired signal d(n). An estimation error e(n) is predicted by obtaining the difference between desired signal d(n) and filter output. The input vector **u**(n) and estimation error e(n) are applied to the control mechanism and the feedback loop around the tap weights is in this manner closed. By assigning a small value of μ to the adaptive process, it is ready to progress slowly and the effects of the gradient noise on the tap weights are largely filtered out.

III. LEAST MEAN SQUARE ALGORITHM

It reduces the mean square error between desired equalizer output and actual equalizer output, which is function of tap gain vector or filter weight $\boldsymbol{w}_N(n)$ because the prediction error is dependent on $\boldsymbol{w}_N(n)$[9]. $J(\boldsymbol{w}_N)$ is cost function, denote the MSE. For steepest descent (SD) the updated tap gain vector is represented as:

$$w_N(n+1) = w_N(n) + \frac{1}{2}\mu\left[-\nabla J\left(w_N(n)\right)\right] \tag{3}$$

where

$$\nabla J\left(w_N(n)\right) = -2\,P + 2\,R\,w_N(n) \tag{4}$$

and

$$\mathrm{P} = \mathrm{E}[\mathrm{u}(\mathrm{n})\mathrm{d}^*(\mathrm{n})] \tag{5}$$

$$\mathrm{R} = \mathrm{E}[\mathrm{u}(\mathrm{n})\mathrm{u}^{\mathrm{H}}(\mathrm{n})] \tag{6}$$

For convergence of LMS there should be convergence of $\boldsymbol{w}_N(n)$ in mean and convergence of $J(\boldsymbol{w}_N(n))$ in the mean square of the error as:

$$E[w_N(n)] \to w_o \tag{7}$$

$$J\left(w_N(n)\right) \to J(w_{\infty}) \tag{8}$$

Then equation for updated filter weights becomes [9]:

$$\mathrm{w_N}(\mathrm{n}+1) = \mathrm{w_N}(\mathrm{n}) + \mu\,\mathrm{e}^*(\mathrm{n})\,\mathrm{u}(\mathrm{n}) \tag{9}$$

where

$$\mathrm{e}(\mathrm{n}) = \mathrm{d}(\mathrm{n}) - \mathrm{y}(\mathrm{n}) \tag{10}$$

$$\mathrm{y}(\mathrm{n}) = \mathrm{w_N}(\mathrm{n})^{\mathrm{H}}\mathrm{u}(\mathrm{n}) \tag{11}$$

The subscript N is number of delay stages in equalizer and μ is step size. Step size μ controls the convergence rate and stability of the algorithm [5]. The stability condition is :

$$0 < \mu < \frac{2}{\lambda_{\max}} \tag{12}$$

λ_{max} is the largest Eigen value of the matrix E= [$\mathbf{u}$ (n) $\mathbf{u}^T$(n)]. step size can be predicted as [5] :

$$\mu = \frac{2}{\mathrm{Trace(R)}} \tag{13}$$

IV. SIMULATION AND RESULTS

The comparison of Fourier based and wavelet based OFDM adopting BPSK over AWGN have been analyzed for number of subcarriers 256 using LS channel estimation is shown in figure (2).

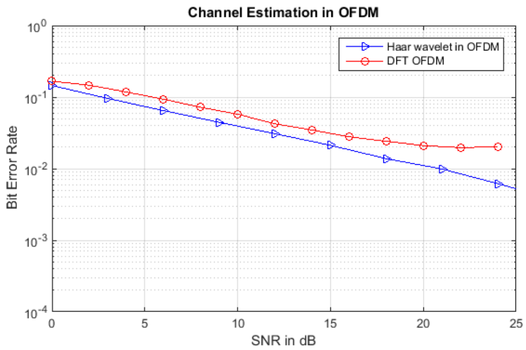

Figure 2.Comparision of DFT-OFDM and haar wavelet in DWT-OFDM using LS channel estimation.

In the previous work the BER performance Of DWT-OFDM over frequency selective fading channel with AWGN compares with FFT-OFDM is shown in figure (3)

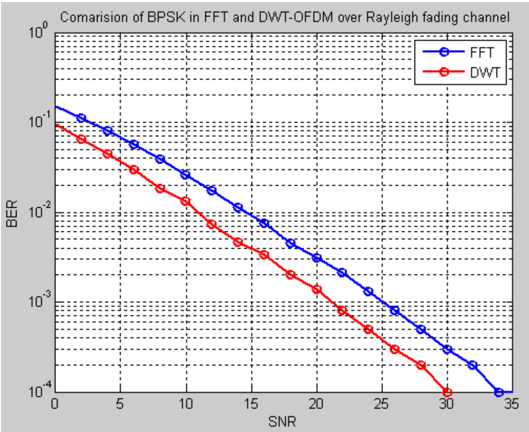

Figure 3.BER performance of DWT and FFT based OFDM over Rayleigh channel with AWGN.

In order to improve the result based on figure (3) LMS is applied in this paper for number of symbol 10 with parameters as shown in table number 1. This paper simulates the results for only DWT-OFDM using 'bior5.5' wavelet over Rayleigh with channel length L=6. In the frequency selective fading channel signals are more distorted, so channel estimation using LMS algorithm maximizes the signal to distortion ratio by minimizing the mean square error.

Table no.1.Applied parameters to evaluate the proposed method

OFDM symbol size	64
Parameters	Related Values
Number of symbols	10
Baseband modulation	BPSK
Wavelet	Bior5.5
Channel length(L)	6
Channel Model	Rayleigh with AWGN

Simulation is done by MATLAB software and results can be studied in two section

(A.) BER performance of DWT-OFDM over frequency selective fading with and without LMS.

(B.) LMS algorithm with different step size values

A. DWT-OFDM with and without LMS algorithm

LMS algorithm reduces bit error ratio response as shown in figure 4 for small signal to noise ratio.

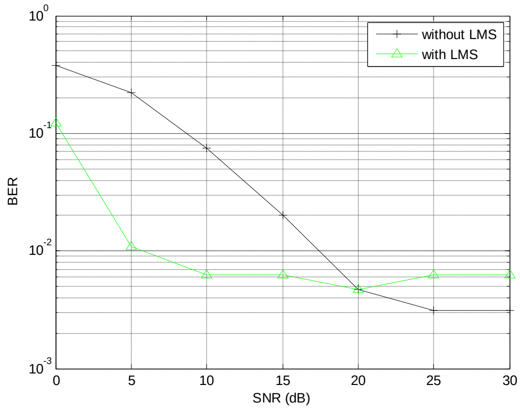

Figure 4.DWT-OFDM with and without LMS algorithm at step size 0.0056

Table no.2.Evaluated BER for DWT-OFDM with LMS algorithm at step size 0.0056

SNR (in dB)		0	5	10	15	20	25	30
BER For (μ=0.0056)	Without LMS	0.3781	0.2219	0.0750	0.0203	0.0047	0.0031	0.0031
	With LMS	0.1219	0.0109	0.0063	0.0063	0.0047	0.0063	0.0063

B. BER performance for different step size values

Performance of LMS algorithm depends on stability factor of step size parameter because as shown in equation(12). The higher value of step size improves the result but that should always be below to the upper bound. For a very small step size algorithm converges slowly and results into steady state average squared error. Whereas for higher value of step size parameter algorithm converges faster and average squared error predicts very worst steady state response.

The evaluated BER versus SNR range 0 to 30dB for DWT-OFDM using LMS algorithm are shown in table number 2. The concept of LMS is studied in two cases by taking difference among three step size values 0.0056, 0.0142 and 0.0902. For step size mu=0.0056 LMS predicts a good response. On increasing step size to 0.0142 LMS algorithm also reduces up to low BER 0.0047 that is 0.0063 for step size 0.0056 at SNR 20dB in case-1. On further increasing the value of step size to 0.0902 at SNR 20dB LMS predicts the high BER 0.0078 which shows that at very high step size value LMS performs with bed response.

Figure (5) and (6) show the BER performance for DWT-OFDM using LMS algorithm in case-1 at step sizes 0.0056 and 0.0142 and in case-2 at step sizes 0.0142 and 0.0902 respectively.

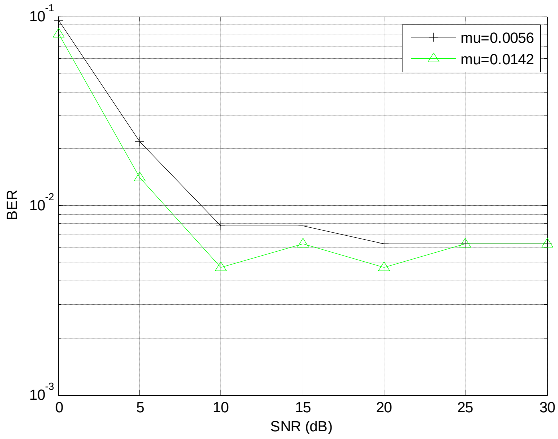

Figure 5.BER performance of DWT-OFDM using LMS algorithm for μ=0.0056 and μ=0.0142.

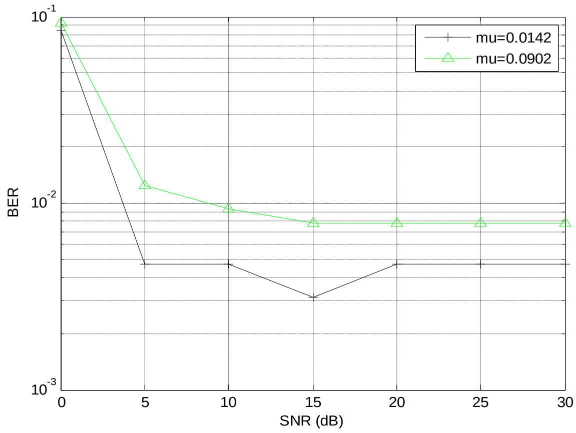

Figure 6.BER performance of DWT-OFDM using LMS algorithm for μ=0.0142 and μ=0.0902.

Table number 3.Evaluated BER for DWT-OFDM with LMS at different step sizes

SNR (in dB)	Evaluated BER			
	Case -1		Case -2	
	μ=0.0056	μ=0.0142	μ=0.0142	μ=0.0902
0	0.0953	0.0813	0.0844	0.0922
5	0.0219	0.0141	0.0047	0.0125
10	0.0078	0.0047	0.0047	0.0094
15	0.0078	0.0063	0.0031	0.0078
20	0.0063	0.0047	0.0047	0.0078
25	0.0063	0.0063	0.0047	0.0078
30	0.0063	0.0063	0.0047	0.0078

V. CONCLUSION

As when distorted signal is appeared at DWT-OFDM receiver cause of multipath fading channels of different path gains and delay spread. To recover the original signal very well or to estimate the channel LMS algorithms is applied and predicts the output by increasing the signal to distortion ratio.

The tap of filter should be small as to create an excellent response. The LMS algorithm in this paper works for 3 taps.

At small values of step size 0.0056 and 0.0142, there is slow convergence and at large value of step size 0.0902, the convergence is faster. The fast convergence results into very worst steady state averaged square error, so it predicts very worst response and reduces BER very few with respect to the performance of BER for small step size.

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