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Low Inter Symbol Interference SDR Receiver using LMS Algorithms

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Abstract— In this paper, the low Inter Symbol Interference (ISI) Software Defined Radio (SDR) receiver using Least Mean Square (LMS) algorithms is developed and investigated. The transmission medium will distort the incoming digital signals and the aim of the (LMS) is to remove the effect of Inter Symbol Interference (ISI) in the Software Defined Radio (SDR) receivers. The resulting of this process is an important act in digital signal processing performance techniques. The development of adaptive equalizer to follow the raped growth in modern information systems is the key of error reduction. To increase the data transmission rate, an efficient adaptive equalizer should be used in radio communication systems. In this paper, the channel selection LMS equalizer filter is developed to eliminate the ISI from the software defined radio (SDR) receiver channel. The LMS equalizer is presented and investigated according to MATLAB simulation. The proposed equalizer shows better performance than the conventional filter with AWGN distortion and inters symbol interference (ISI). The results show a promising basis for rising code, algorithms in 16-QAM modulation scheme.

Keyword – LMS, ISI, SDR Receiver, 16-QAM

I. INTRODUCTION

The communication channel is mostly affected by the presentation of communication systems working in excess of the medium. The transmitted signals over a channel are distorted by both amplitude and phase functions. Additionally, the delay spread of the channel introduce the well known inter symbol interference (ISI) to the received signals which is one of the main difficulty to dependable and high speed data transmission channel equalization in this process of compensating for the unconstructive outcome in the channel on the transmitting signal and take away the ensuing ISI [1]. To attain this objective, the equalizer uses an estimate of the channel frequency response, thought the fading channel vary all through the broadcast phase. The equalizer required here to study the frequency response in an adaptive style to be talented to incessantly alleviate the unenthusiastic outcome in the channel. The obtainable equalization channels techniques like LMS with other types shows better performance [2]. Inter symbol Interference (ISI) is a form of distortion in which one symbol interferes with subsequent symbols. The ISI is an unwanted occurrence as the preceding symbols has alike effects as noise, hence making the communication fewer dependable. The multipath propagation is almost caused the ISI in the channel and the inherent non-linear frequency response produce successive symbols [3]. The attendance of ISI in the system introduces errors in the decision device at the receiver output. Consequently, the goal is to minimize the effects of ISI. There are two ways to fight with ISI, the error correction and the adaptive equalizer techniques. Also. ISI could be removed by using cyclic prefix before transmitting. The equalizer algorithms of LMS will used as simple techniques, low computational complexity and better performance in many running environments [4]. Thus, the system is motivated by the insist for better quality of service, high data rate and high mobility [5]. As illustrated in Figure 1, suppose x(n) is the original signal and m(n) is the combined complex baseband impulse response of the transmitter. Then the received signal at the equalizer is expressed as:

$$Y(n) = X(n).M^{*}(n) + N(n) ------------(1)$$

Y(n) is the input of adaptive equalizer, $M^{*}(n)$ is complex conjugate of M(n) denoted as baseband noise in the receiver path and $N(n)$ is the convolution operation impulse response of the equalizer then the output of equalizer is given by;

$$Z(n) = X(n).K(n) + N(n).H(n) ----------(2)$$

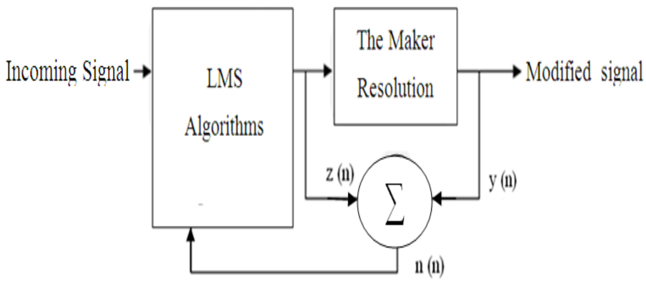

Figure 1 Adaptive equalizer Modified signal

The LMS algorithms are a type of adaptive filter used to reproduce the preferred filter by discovers the filter coefficients that relay to create the least mean squares of the error signal by mean of the different between the desired and the actual signal. The LMS algorithms include an iterative process that creates succeeding alteration to the weight vector in the direction of negative slope vector which ultimately guide to minimum fractional spaced error.

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A fractional spaced algorithm is relatively simple compared with other algorithms due to not requiring of correlation function. The step size of 0.005 is better than step size of 0.001 in LMS algorithms case. However, the different between the desired z (n) and the actual z (n) is called error signal. Hence, the desired signal is an exact replica of transmitted signal which represents the properties of transmitted signal. The weights of the adaptive filter or equalizer are updated to minimize the error signals which could be calculated as in [6].

$$N(n) = R(n) - W^t(n).Y(n) ------------ (3)$$

The weights vector of LMS algorithms is reorganized as:

$$W(n+1) = w(n) + 2\mu r(n).z(n) ---------- (4)$$

Where $W(n+1)$ represents the new weight, $w(n)$ represents the current weights, (μ) is the step size and $z(n)$ is the previous error.

II. NORMALIZED LMS

In the typical LMS filter, when step size μ is large, the algorithms will suffer from noise amplification problem. Hence, the normalized LMS could be used in this case. The weight vector $w(n)$ is corrected at iteration $n+1$ is normalized in acordance with the squared Euclidian norm of the input vector $X(n)$ at iteration n. Consequently, in normalized LMS algorithms, the step size μ could be caculated as in [7]:

$$\mu(n) = \frac{\alpha}{c + |x(n)|^2} ---------------- (5)$$

Where α is the normalized LMS adaption constant factor. The convergence rate of the NLMS algorithms could be optimized by satisfying the condition of $0 < \alpha < 2$. Hence, the filter weights are updated by:

$$w(n+1) = w(n) \frac{\alpha}{c + |x(n)|^2} e(n)\,x(n) ------- (6)$$

III. LMS EQUALIZER MODULE

The LMS linear equalizer operating on 16-QAM data source with noise and filtering introduced in the channel model. The LMS equalizer operate based on LMS algorithms and used 32-taps filter with standards constellation and eyes diagrams in MATLAB simulation model.

To minimize the error, the LMS algorithms is used in the receiver side after the transmit signal is contaminated by noise and ISI. The simulation can be broken down into three main stages, transmitter, channel and receiver. The modulation and demodulation steps are comprised of several distinct systems. The 16-QAM baseband signal is generated and passed through a channel.

When the baseband signals are transmitted over a communication channel then they are distorted by various channel imperfections.

The MATLAB model for proposed LMS equalizer has been shown in Figure 2.

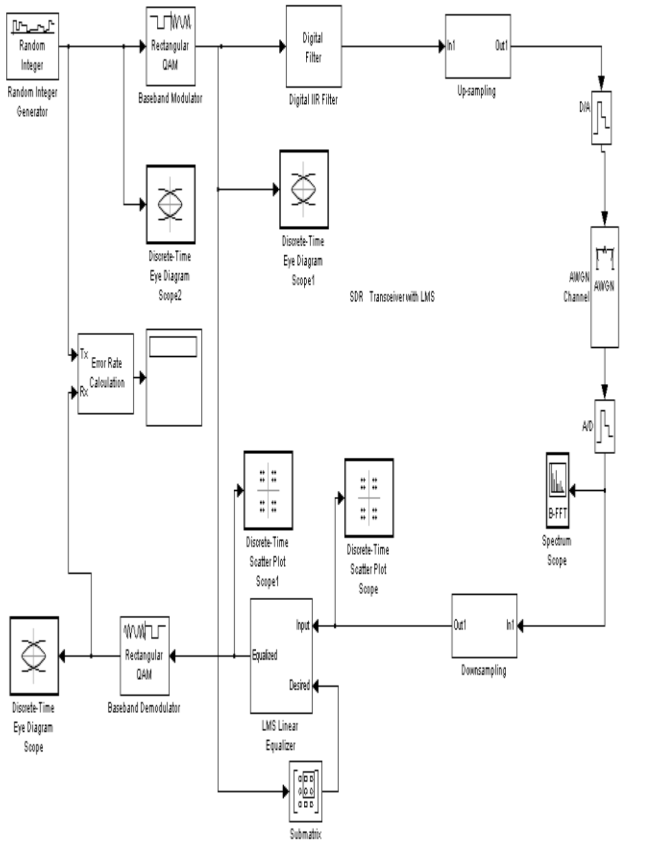

Figure 2 MATLAB simulation of QAM Modem with LMS algorithms

In the receiver side, the incoming signals is mixed with noise and down converted into intermediate frequency and fined gain filtered signal is then pass through the LMS equalizer to be compared with transmitted signal then demodulated to recovered the original data. The modulated 16-QAM signal is passed through non linear IIR filter to generate ISI noise in the transmitter path [8].

The transmitted signal is up converted and shaped by root raised cosine filter. After the transmitted signal is up converted it may lose some of properties, therefore the gain stage should be used to balance this loses before transmitted to the channel stage.

In the receiver path, the incoming signal with ISI and noise is down sampled and filtered by using root raise cosine filter and file gain before equalize stage.

The contaminated incoming signal is then passed through the LMS equalizer to remove all type of noise and ISI introduced by AWGN channel and non linear IIR filter respectively. Finally, the clean signal is demodulated and recovered to get the original data.

III. I SIMULATION RESULTS

The simulation results are given in this section for 16-QAM scheme using LMS linear equalizer and conventional one. The constellation diagram for the transmitted signal is shown if Figure 2. Obviously, the signal is free from ISI before IIR digital filter. After the transmitted signal is passed through the non linear IIR filter, it's clearly suffered from ISI distortion introduced by the IIR Filter as shown in Figure 3 and 4.

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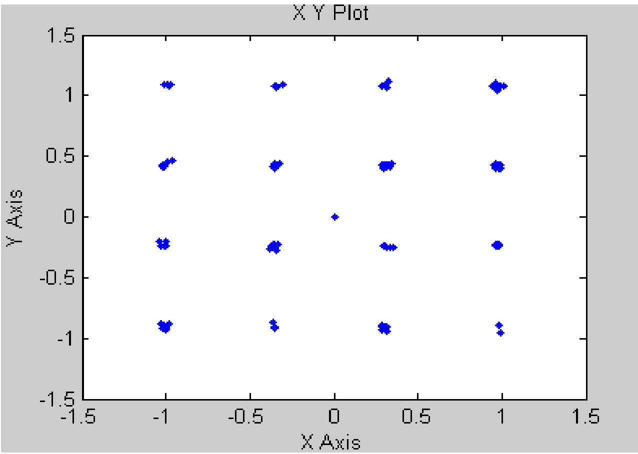

Figure 3 Constellation diagram of 16-QAM signal before IIR filter

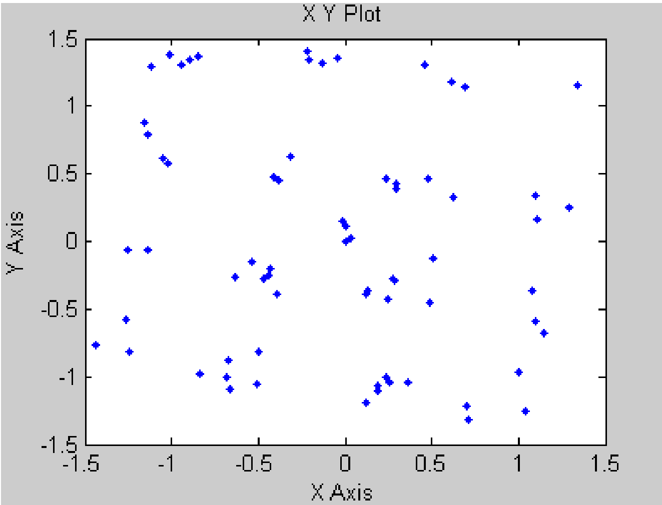

Figure 4 Constellation diagram of 16-QAM signal after IIR filter

Winning the receiver of the signal, the demodulator examine the received symbol, which may have been corrupted by the channel or the receiver noise (AWGN). As it's estimated of what was actually transmitted, that point on the constellation diagram which is closed to that of the received symbol.
However, it will demodulate incorrectly if the corruption has caused the received symbol to move closer to another constellation point than the transmitted one. In this case, the equalizer will play the main role to guide the received signal and eliminate the ISI and noise from the channel.

III.II LMS LINEAR EQUALIZER

The LMS Linear Equalizer block uses a linear equalizer and the LMS algorithm to equalize a linearly modulated baseband signal through a dispersive channel [9].
During the simulation, the block uses the LMS algorithm to update the weights, once per symbol. If the Number of samples per symbol parameter is 1, then the block implements a symbol-spaced equalizer; otherwise, the block implements a fractionally spaced equalizer.
The LMS Filter block can implement an adaptive FIR filter using five different algorithms.
The block estimates the filter weights, or coefficients, needed to minimize the error between the output signal and the desired signal.

This input signal can be a sample-based scalar or a single-channel frame-based signal. Connect the desired signal to the desired port.
The desired signal must have the same data type, frame status, complexity, and dimensions as the input signal. The Output port outputs the filtered input signal, which is the estimate of the desired signal.
The output of the Output port has the same frame status as the input signal.
The Error port outputs the result of subtracting the output signal from the desired signal. However, the desired signal is connected to the transmit signal before IIR filter.
The equalizer will compare between the transmitted symbol and the received one to correlate the correct symbol and feed to the demodulator.
This block accepts only frame-based signals. If the value of Reference tap is equal to or greater than the frame size, the block will not work properly.
The eye diagram of the received signal before the equalizer is shown in Figure 4.
If one makes a look at the received signal, the signal is affected by ISI and noise effect with SNR of less than 20 dB. When the SNR increased to 30 dB, the noise margin start to close out and the error is decreased in eye diagram. Since, the SNR decrease to 10 dB, the eye diagram has more distortion in the system.
Figure 6 shows the eye diagrams of the receiver signal when SNR = 30 dB. The eye diagram reveals less distortion given that the eye opening is more defined. The correct eye results and less bit error and hence, less transmission error [10].

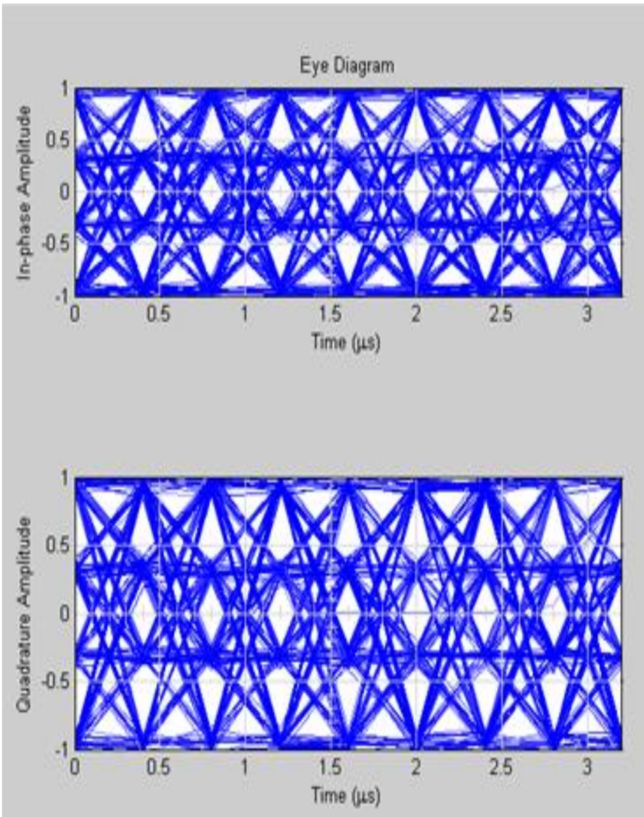

Figure 5 Eye diagram of 16QAM received signal before Equalizer

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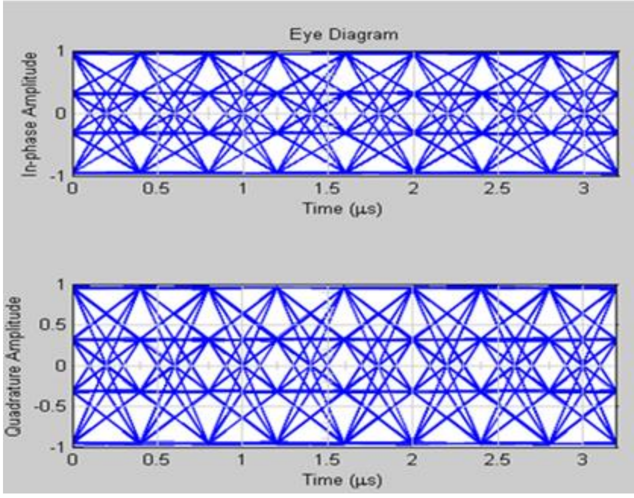

Figure 6 Eye diagram of received signal after equalizer

Figure 7 shows the input and output waveforms of the transmitter and receiver model. The received IQ signals are filtered and fine-gained to convert them into refined IQ signals. Subsequently, the signals are equalized and down-sampled by a factor of 16 to become baseband IQ symbols (2.5mega-baud).

After demodulation, the recovered 4-bit integers (2.5mega-baud) are compared with the transmitter input to confirm all system functions. The difference between the original message 4-bit integer and receiver output 4-bit integer is zero.

The LMS Decision Feedback Equalizer uses a decision feedback equalizer and the LMS algorithm to equalize a linearly modulated baseband signal through a dispersive channel. During the simulation, the block uses the LMS algorithm to update the weights, once per symbol.

If the Number of samples per symbol parameter is 1, then the block implements a symbol-spaced equalizer; otherwise, the block implements a fractionally spaced equalizer.

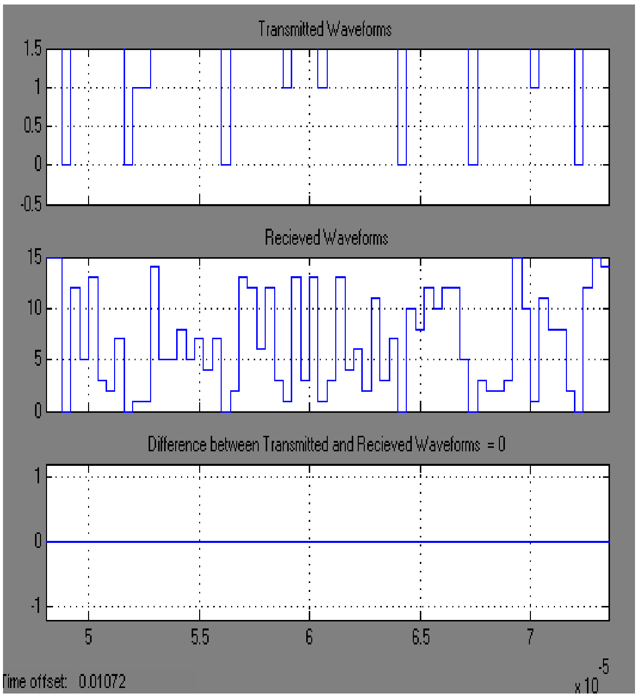

Figure 7 Transmitter input signal and Receiver Output Signal

To show the active noise control of a random noise signal that runs for 1000 samples, the M-file code is used with corresponding parameters to show the effect of proposed equalizer on the noise produced by the channel as illustrated in Figure 8.

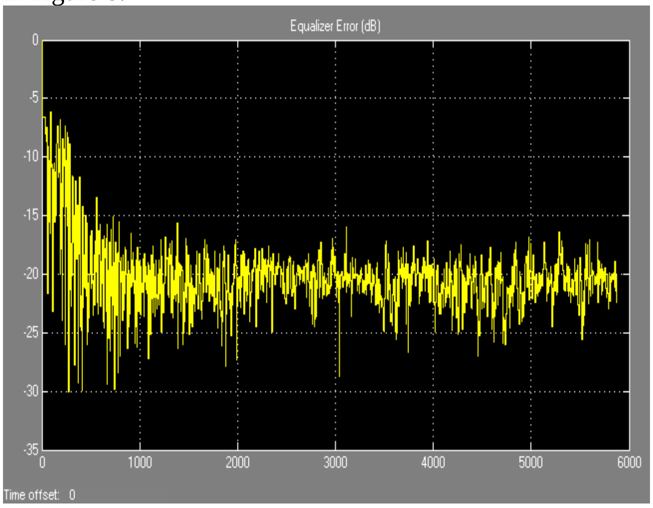

Figure 8 Noise control of random noise signal

IV. IMPLEMENTATION OF LMS EQUALIZER

The proposed LMS algorithms has been implemented using system generator block sets as illustrated in Figure 9. The white noise is feed to channel and mixed with noise before passed to the LMS normalized equalizer to compare with the desired signal. At the equalizer output, the error will separated from the original signal.

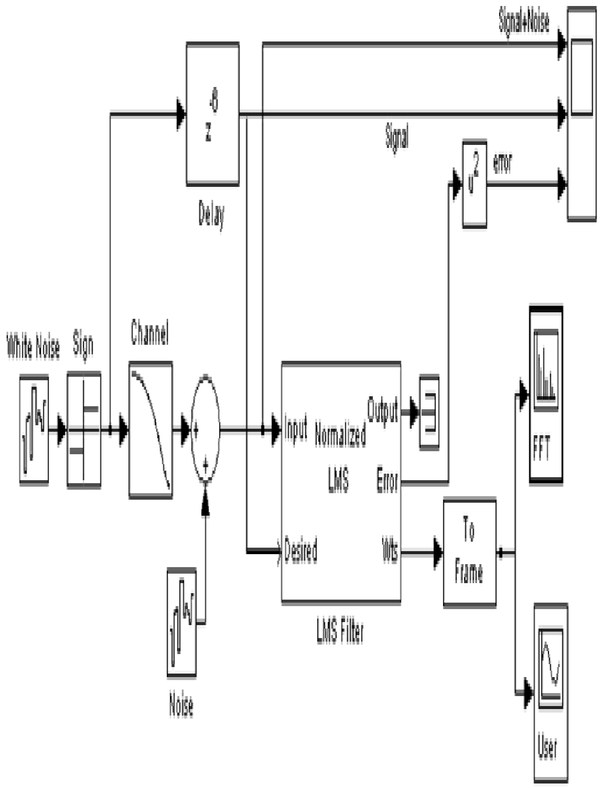

Figure 9 LMS equalizer implementation using system generator

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From the results it could be seen clearly the performance of LMS and how this equalizer effect on the signal by cancelling the error between the transmitted and received signals where the process of this equalizer is comparing the input signal shown in figure 9 with the signal after commutating ISI noise and calculate the error ratio between these two signals then updates its weights according to this error ratio tile the error between the transmitted and received signals is equal to zero. The results of multiplying the coefficients of LMS by the received samples are dot products which represents the ideal channel response for the LMS equalizer span. As it is clear from the illustrated figure10, the LMS equalizer impulse response has a symmetric impulse response and linear in phase which has a constant group delay that results in no distortion due to frequency selectivity, where the frequency components have equal delay time as show in Figure 11. Finally from figure (12) it could be seen that the process of this equalizer increase the SNR, where increasing in this ratio is desired in order to reduce the noise margin and hence decreases the error.

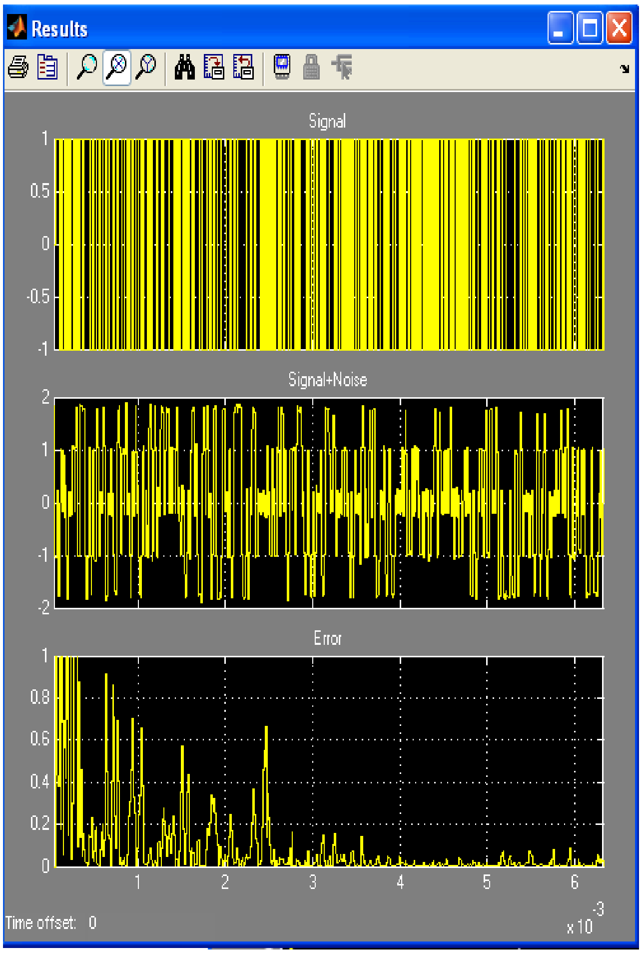

Figure 10 Compressions between the error and original signal

The experimental results obtained by simulating LMS algorithm using Matlab could be discuss in this section briefly. A random signal with amplitude ±1 is transmitted through three realizations of channel. The impulse response for each of three realizations is represented by H1, H2, and H3 respectively and their transfer functions are given by,

$$H_1(Z) = \frac{1}{4} + Z^{-1} + \frac{1}{4}Z^{-2} - - - - - - - - - (7)$$

$$H_2(Z) = \frac{1}{4} + Z^{-1} + \frac{1}{4}Z^{-2} - - - - - - - - - (8)$$

$$H_3(Z) = -\frac{1}{4} + Z^{-1} + \frac{1}{4}Z^{-2} - - - - - - - - - (9)$$

The value of μ is taken as 0.001. Tap FIR filter is used for channel equalization and initial values of this filter is set to be zero for all taps. Mean square error is calculated by simulating 100 independent realizations of input signal to each of three channels. The channel impulse response and tapped-equalizer impulse response tends to generate input-output delay. For channel 1, impulse response is symmetric around n =1; so this channel introduces a group delay of 1 unit. Since input is real data; we expect tapped equalizer to be symmetric around n = 10; to give linear phase. Hence the total delay introduced at output is 11 units. This delay has been used in our simulations for learning curves. One can conclude that any delay of more than or equal to 2 unit provides almost same mean square error. The proposed algorithm tends to coverage very fast for any delay greater than or equal to 2 units. The convergence rate is slower for delay unit of 1 unit and algorithm doesn't seen to converge when no delay line is used.

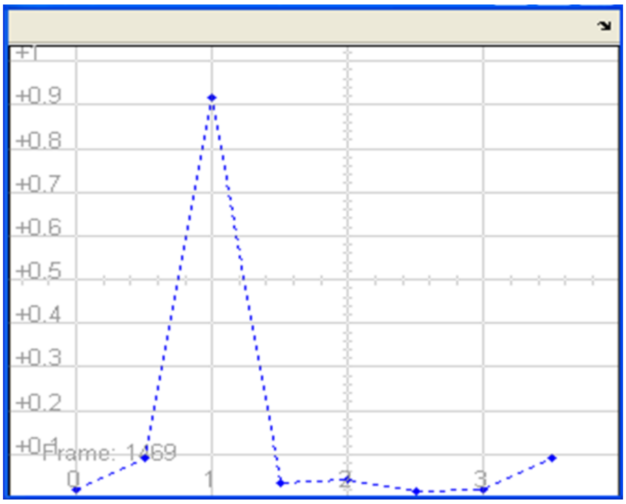

Figure 11 LMS Filter taps

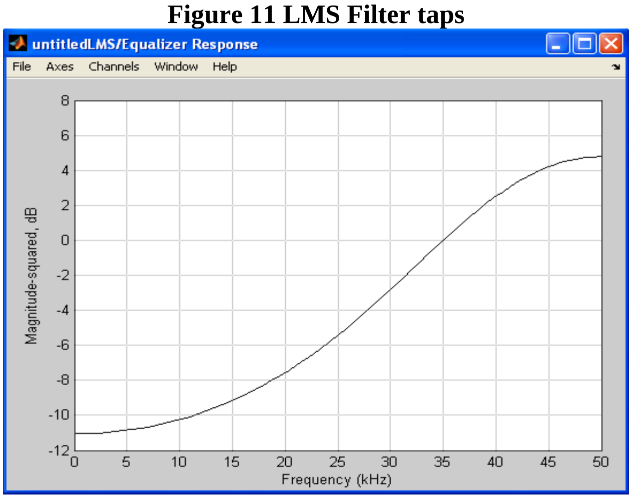

Figure 12 LMS equalizer response

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Note that all of the above mentioned simulations have been done for equalizer corresponding to channel H_2(n). The delays of 11 units have been used in our simulation, since it matches with theoretical calculation of group delay function of overall system. The converged value of minimum mean square error with respect to delay Δ of the channel input, the Mean Square error with respect to number of iterations for different delays Δ. Anything greater than $\Delta > 1$ give good convergence, we plot the draw curves for three step sizes corresponding to $\mu = 0.01$, $\mu = 0.005$ and $\mu = 0.001$. The draw curves for channel response H_1 , H_2 and H_3. By looking at these curves, we can say that algorithm converges fast as step size is increased. However, we can't increase the step size to any value as condition for convergence requires that step size should be less than the inverse of energy of the correlation matrix of received signal. One of the most important observations from these drawing curves is that the algorithm converges faster for channel corresponding to H_2 and H_3 as compared to H_1. The reason for this can be found out using the Eigen value spread analysis of each of three received signals.

V. CONCLUSION

In this paper, an efficient LMS linear equalizer is designed and an implemented using MATLAB and System Generator program has been proposed. Since LMS linear equalizer has better performance than the conventional equalizer. Software based simulation approach has made easily to change system parameters, debug and test easy. Compared to other design, this design has simple design and simulation with the reasonable performance. The proposed LMS linear equalizer could be further improvements by using new adaptation and optimization algorithms. The capability of proposed equalizer within SDR transceivers shows unlimited accuracy to recover the received symbol exactly like transmitted symbol and no error appear were the different between the transmit and received symbol is zero.

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