Image Transfer using OFDM System with AWGN and Rayleigh Fading Channel

Rajnath, Vijay Kumar and C. Sasivarnan
Department of Electronics & Communication Engineering,
Maharishi Markandeshwar University, Mullana (Ambala), INDIA
Email: rajnath.lamba@hotmail.com

Abstract—**In even promising multicarrier OFDM technologies; the various phenomena such as multipath wave propagation, time dispersion, and fading introduce errors like intersymbol interference (ISI), and other distortions into the signals transmitted over wireless channel. This paper, therefore, investigates the effect of these phenomenon on images transferred through an OFDM based wireless communication system including QPSK modulation, AWGN and Rayleigh fading channels and LMS equalization. The quality of the received image passed through AWGN and Rayleigh fading channels is compared with the original image. At last, the performance improvement in slow Rayleigh fading channel is done using linear equalization with LMS algorithm. The results show significant improvement in error-performance of digital image transmission. It is observed that in AWGN channel, the image is degraded by random noise; and in Rayleigh fading channel, the image is degraded by random noise, block noise, and ISI.**

Keywords— ***Channel Equalizer; Adaptive Equalizer; Least Mean Square; Slow Fading; Frequency Selective Fading.***

I. INTRODUCTION

The wireless channel is usually random and time-variant unlike the wired channels that are stationary and predictable [1]. It is well known that the wireless multi-path channel causes an arbitrary time dispersion, attenuation, and phase shift, know as fading, in the received signal. Fading is caused by interference between two or more versions of the transmitted signal which arrive at the receiver at slightly different times. Also, when digital transmission is performed through such multipath wireless channels, then the digital information usually accompanied with a phenomenon known intersymbol interference (ISI) [1-3]. The term ISI means that the transmitted pulses will be smeared out so that pulses that correspond to different symbols are not separable; and have amplitude and phase dispersion which results in the interference of the transmitted signals with one another. However, for a reliable digital transmission system it is crucial to reduce the effects of ISI. This can be reduced using equalisers, which are designed to work in such a way that Bit Error Rate (BER) should be low and Signal-to-Noise Ratio (SNR) should be high. These are preferred to filters as the transfer function of wireless channels vary with time, so it is not possible to use an optimum filter for these types of channels. More specifically, equalizers are designed to give the inverse of channel to the received signal. Thus, the combination of channel and equalizers will ideally give a flat frequency response and linear phase [3-6]. However in wireless channels, due to the multi path fading, the channel characteristics change with time. Thus, it is necessary for channel equalizer to track the time varying channel in order to provide reasonable performance [7-8]. Therefore, this paper is focused on adaptive equalisation for the unpredictable wireless channels [9-10]. More specifically, this paper focuses on using LMS based equaliser for improving the quality of images transferred over Rayleigh fading channel. This is discussed further in this paper.

II. SYSTEM MODEL

Figure 1 shows the OFDM model used in this paper; in which an image is used as an input source data. The randomly generated signal can also be used. The image data is first converted into binary data and the pilot data is inserted into head of source data in each coherence time frame. It is used to estimate the random phase shift of the fading channel and train the decision to adjust the received signal with phase recover. The data is then mapped from binary data to complex data, and each output datum represents a point in the constellation diagram. In the model shown in Figure 1, the M-ary phase shift keying (PSK) modulation is used to modulate the data source. More specifically, the QPSK modulation is used. Two different channels: AWGN channel and Rayleigh slow fading channel are simulated and used. The Rayleigh slow fading channel can be flat fading and/or frequency selective fading channel.

At the receiver side in Figure 1, the adaptive equalizers are used for Rayleigh fading channel. As the QPSK modulation is used in the above model, so the channel phase information in each coherence time need to be estimated and the weight of each tap in the equalizer need to be trained by pilot data. Then, source data are equalized by trained equalizer for frequency selective fading channel. Finally, the received demodulated binary data is converted back to image and is displayed. The analytical model of channel and LMS adaptive equalization is discussed below. In wireless digital communication, the impulse response is used to characterize a channel. In a fading channel with impulse response of L bins,

$$E\left[\left\|\overrightarrow{h_L}\right\|^2\right] = \sum_{i=0}^{L-1} E[|a_i|^2] \tag{1}$$

where, a_i is the ith tap of impulse response in $\overrightarrow{h_L}$. In a flat fading channel, as there is one tap in channel impulse response h for each sample, the expectation value of h is $E[|h|^2] = E[|a_0|^2]$. To get the comparable SNR to a non-fading AWGN channel, the expectation value is set to unity, i.e., $E[|h|^2] = 1$. So, in a flat fading channel, the expectation value of $\vec{h}$ with N samples can be expressed as:

$$E\left[\left\|\vec{h}\right\|^2\right] = N * E[|h|^2] = N \tag{2}$$

where, $\vec{h}$ represent the channel impulse response with N samples.

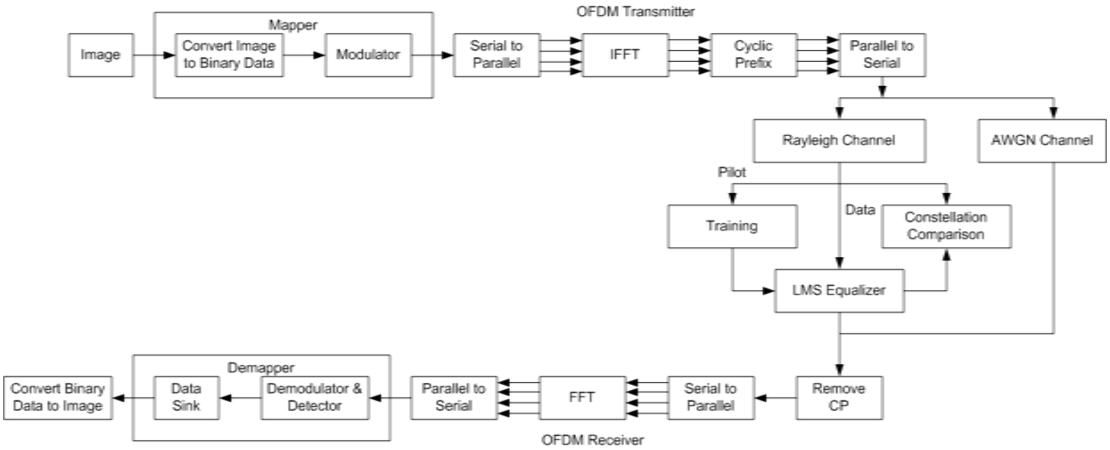


Fig. 1. OFDM model used in simulation.

When the signals are passed through this channel, the RF signal spectral shape after Doppler spread can be determined by the product of amplitude of complex Gaussian random variable and root square of Doppler power spectrum. In this paper, we denote $\vec{A}$ as amplitude of complex Gaussian random variable in frequency domain, $\vec{D}$ as root square of Doppler power spectrum and $\overrightarrow{H_b}$ as the RF signal spectral shape after Doppler spread. So, $\overrightarrow{H_b}$ is actually a scaled version of $\vec{D}$, where each point in $\vec{A}$ is a scalar of complex Gaussian random variable. To represent $\overrightarrow{H_b}$ by product of $\vec{D}$ and $\vec{A}$, we should use a diagonal matrix with component in the main diagonal come from $\vec{A}$. Therefore, we get:

$$\overrightarrow{H_b} = \sum . \vec{D}\,, \qquad where \sum \sim CN(0, \sigma^2) \tag{3}$$

where, $\sum$ is a diagonal matrix; and each component in the main diagonal is a complex Gaussian random variable, which is produced by randn() function, so variance of all components in the main diagonal are $2\sigma^2 = 2$. Thus, the expectation value of $\sum$ is $E[\sum^2] = 2\sigma^2 . I = 2. I$, where I is a identity matrix. Therefore,

$$E\left[\left\|\overrightarrow{H_b}\right\|^2\right] = E\left[\left\|\sum . \vec{D}\right\|^2\right] = \sigma^2 . \left\|\vec{D}\right\|^2 = 2. \left\|\vec{D}\right\|^2 \tag{4}$$

The channel impulse response before normalisation, denoted as $\vec{h}_b$, can be calculated as:

$$\overrightarrow{h_b} = \mathrm{F}^{-1}\{\overrightarrow{H_b}\} \tag{5}$$

where, F^{-1} denotes the inverse Fourier Transform. According to Parseval's theorem for Discrete Fourier Transform:

$$\left\|\overrightarrow{h_N}\right\|^2 = \sum_{n=0}^{N-1} |h_n|^2 = \frac{1}{N} * \sum_{k=0}^{N-1} |H_k|^2 = \frac{1}{N} * \left\|\overrightarrow{H_N}\right\|^2 \tag{6}$$

Combining, equations 5 and 6 to have:

$$\left\|\overrightarrow{h_b}\right\|^2 = \frac{1}{N} * \left\|\overrightarrow{H_b}\right\|^2 \tag{7}$$

With the above equation,

$$E\left[\left\|\overrightarrow{h_b}\right\|^2\right] = \frac{1}{N} * E\left[\left\|\overrightarrow{H_b}\right\|^2\right] = \frac{2}{N} . \left\|\vec{D}\right\|^2 \tag{8}$$

To calculate the channel impulse response after normalisation, denoted as $\vec{h}$, we set $\vec{h} = \lambda . \overrightarrow{h_b}$; whereby λ is calculated as below.

$$E\left[\left\|\vec{h}\right\|^2\right] = E\left[\left\|\lambda . \overrightarrow{h_b}\right\|^2\right] = \lambda^2 . E\left[\left\|\overrightarrow{h_b}\right\|^2\right] = N \tag{9}$$

Or,

$$\lambda = \sqrt{\frac{N}{E\left[\left\|\overrightarrow{h_b}\right\|^2\right]}} \tag{10}$$

And,

$$\vec{h} = \lambda . \overrightarrow{h_b} = \frac{\sqrt{N} . \overrightarrow{h_b}}{\sqrt{E\left[\left\|\overrightarrow{h_b}\right\|^2\right]}} \tag{11}$$

Using equation (8) to finally have:

$$\vec{h} = \frac{\sqrt{N} . \overrightarrow{h_b}}{\sqrt{\frac{2}{N} . \left\|\vec{D}\right\|^2}} \tag{12}$$

A Rayleigh frequency selective fading channel is produced with impulse of L bins, whereby each tap is a flat fading channel scaled by an exponential Power Delay Profile (PDP). We may use the same method as above to derive $\vec{h}$ for each tap. However in a frequency selective fading channel, $E[|a_i|^2] \neq 1$, where $a_i$is the ith tap of impulse response in $\overrightarrow{h_L}$, and instead, we should use the following equation:

$$E\left[\left\|\vec{h}_L\right\|^2\right] = \sum_{i=0}^{L-1} E[|a_i|^2] = 1 \tag{13}$$

As, the channel has an exponential PDP, the expectation value of each bin is

$$E[|a_i|^2] = \alpha . e^{\frac{i.T}{\tau}} \tag{14}$$

where, T is the sampling period, and τ is the time constant. So, we get

$$\sum_{i=0}^{L-1} E[|a_i|^2] = \sum_{i=0}^{L-1} \alpha . e^{\frac{i.T}{\tau}} = \alpha . \sum_{i=0}^{L-1} e^{\frac{i.T}{\tau}} = 1 \tag{15}$$

And,

$$\alpha = \frac{1}{\sum_{i=0}^{L-1} e^{\frac{i.T}{\tau}}} \tag{16}$$

Now we can calculate a_i from h_i; whereby, h_i is each sample in the N sample vector $\vec{h}$, using equations 13 to 16 as

$$a_i = \sqrt{\alpha . e^{\frac{i.T}{\tau}} . h_i} \quad (17)$$

Using the above values of a_i and h_i, the Rayleigh fading channel is designed. In this paper, the Matlab function Rayleighchan() is used to generate the Rayleigh fading channel. Now, the equalisers can be used to reduce the effect of ISI, which causes phase and amplitude distortion. Ideally, an equalizer needs to be designed in such a way that the impulse response of the channel/equalizer combination is as close to $z^{-\Delta}$ as possible, where Δ is delay, as shown in Figure 2. However, the wireless channel parameters are not usually known in advance and they may vary with time significantly. Therefore, it is necessary to use adaptive equalizers, which provide the means of estimating the changing channel characteristics. A typical adaptive channel equalisation system, used in this paper is shown in Figure 2.

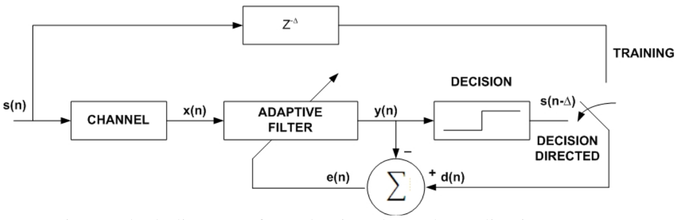


Fig. 2. Block diagram of an adaptive channel equalization system.

In Figure 2 above, $s(n)$ is the signal (image in this paper) to be transmitted through the wireless communication channel, and $x(n)$ is the distorted output signal. The distortion can be in amplitude or phase or both. To compensate for the signal distortion, the adaptive channel equalization system completes the following two modes:

- **Training mode**: This mode is required to determine the appropriate coefficients of the adaptive filter. When the signal $s(n)$ is transmitted through the communication channel, a delayed version of the same signal is also applied to the adaptive filter as shown in Figure 2. The delay function is given by $z^{-\Delta}$, where Δ is delay. In Figure 2, $d(n)$ is the delayed signal and $y(n)$ is the output signal from the adaptive filter and $e(n)$ is the error signal between $d(n)$ and $y(n)$. The adaptive filter iteratively adjusts the coefficients to minimize $e(n)$. When the power of $e(n)$ converges, $y(n)$ is almost identical to $d(n)$, which means that the resulting adaptive filter coefficients can be used further to compensate for the signal distortion.
- **Decision-directed mode:** In this mode, the adaptive channel equalization system decodes the signal $y(n)$ and produces a new signal, which is an estimation of the signal $s(n)$ with a delay of Δ taps. The structure of the adaptive filter is showed in Figure 3.

In Figure 3, the input signal is the sum of the desired signal $d(n)$ and the interfering noise $v(n)$, and is given by

$$u(n) = d(n) + v(n) \quad (18)$$

The filter has a Finite Impulse Response (FIR) structure. For such structures, the impulse response is equal to the filter coefficients.

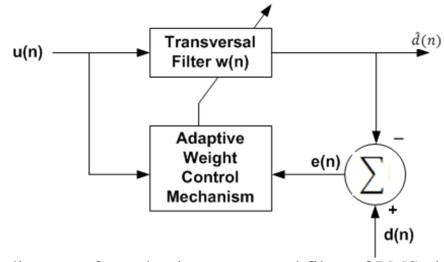


Fig 3: Block diagram of an adaptive transversal filter of LMS algorithm.

The coefficients for a filter of order p are defined as

$$w_n = [w_n(0), w_n(1), \ldots, w_n(p)]^T \quad (19)$$

The error signal is the difference between the desired and estimated signal, and is given as

$$e(n) = d(n) - \hat{d}(n) \quad (20)$$

The filter estimates the desired signal by convolving the input signal with the impulse response. In vector notation, this is expressed as

$$\hat{d}(n) = w_n * u(n) \quad (21)$$

Where,

$$u(n) = [u(n), u(n-1), \ldots, u(n-p)]^T \quad (22)$$

is an input signal vector. Moreover, the variable filter updates the filter coefficients at every time instant

$$w_{n+1} = w_n + \Delta w_n \quad (23)$$

Where, Δw_n is a correction factor for the filter coefficients. The adaptive algorithm generates this correction factor based on the input and error signals. The least mean squares algorithm is a class of adaptive filter used to design the desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal (difference between the desired and actual signal). It is a stochastic gradient method in which the filter is only adapted based on the error at the current time. LMS filter is built around a transversal (i.e. tapped delay line) structure (Figure 3). LMS filter employ, small step size statistical theory, which provides a fairly accurate description of the transient behaviour. The LMS algorithm in general, consists of two basics procedure. The first step is filtering process, which involve, computing the output of a linear filter in response to the input signal, and the second step is generating an estimation error by comparing this output with a desired response as mentioned in equation 20. Adaptive process, which involves the automatic adjustment of the parameter of the filter in accordance with the estimation error, is given by:

$$\widehat{w}(n+1) = \widehat{w}(n) + \mu(u) e^*(n) \quad (24)$$

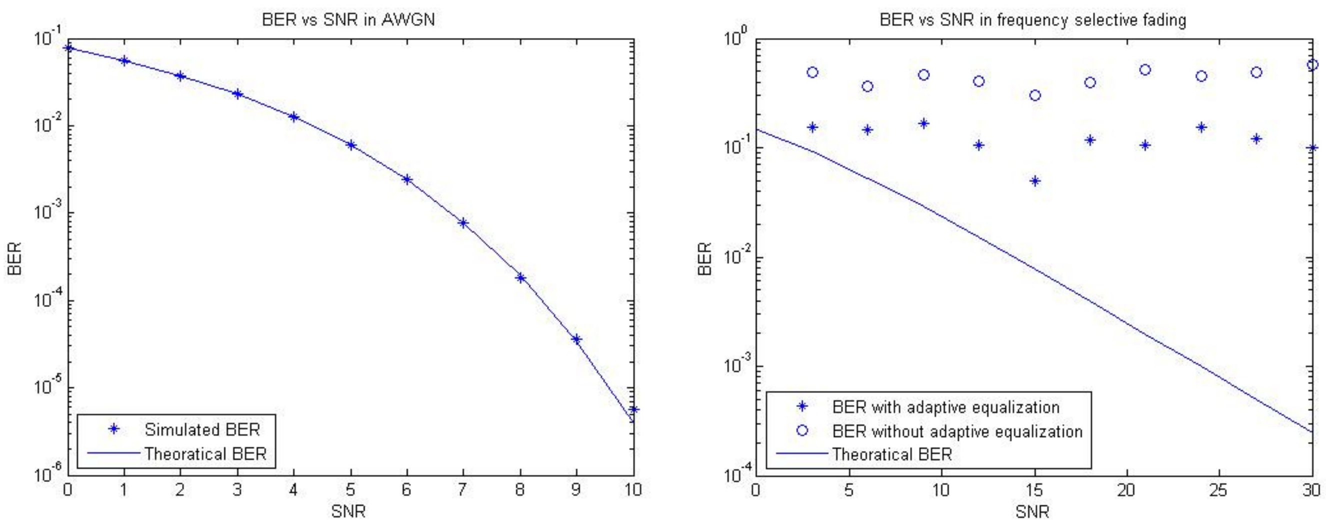
Where μ is the step size, $\widehat{w}(n+1)$ = estimate of tape weight vector at time (n+□1) and if prior knowledge of the tape weight vector (n) is not available, then n is set to 0. The combination of these two processes working together constitutes a feedback loop, as illustrated in the block diagram of Figure 3. The simulated results obtained are discussed below

III. RESULTS AND DISCUSSION

The system, shown in Figure 1, is implemented in Matlab for the image of Lena; and the results obtained for AWGN and

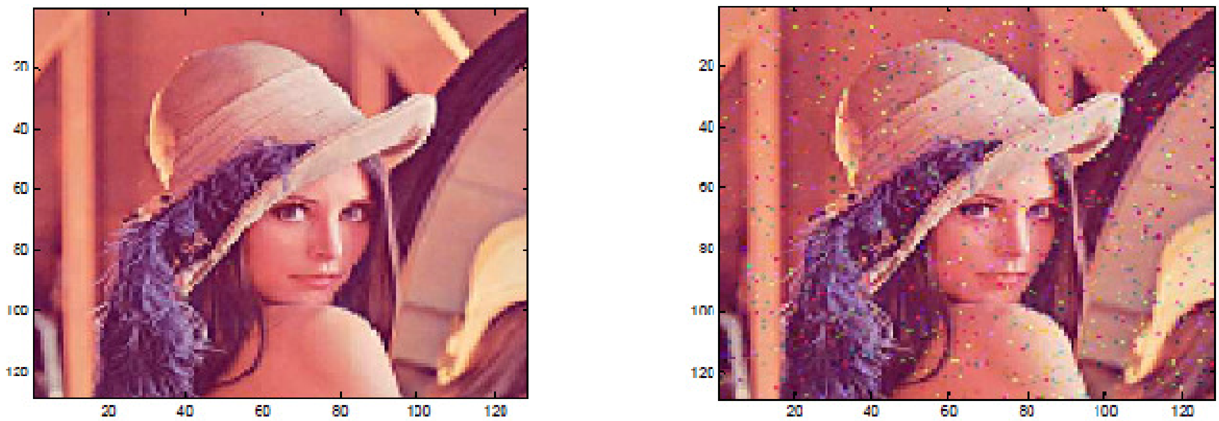
Rayleigh fading channel are shown in Figures 4 to 6. As can be seen in Figure 4(a), the BER performance of simulation result is closely identical to theoretical BER for the AWGN channel. However as observed from Figure 4(b), the BER performance of simulation result for the Rayleigh fading channel is worse than theoretical BER. This is reasonable, as the theoretical BER is based on the assumption that we know exactly the phase information of modulated signal. However, due to the time-variant channel, we always have estimation error for phase information. We also find that the BER performance is improved dramatically in low SNR, while not in high SNR. This is also reasonable, as in low SNR, white Gaussian noise dominate the BER error, which can be improved by enhancing SNR, while in high SNR, phase estimation error and ISI dominate the BER error, and the estimation error will cause even severe ISI, which cause the BER even worse.

To evaluate the image quality of received vs original images, consider the images shown in Figures 5 and 6. The received image through AWGN channel is plotted at SNR = 5dB in Figure 5. It can be seen that there are some random noises in the image passed through the AWGN channel. In Figure 6, the received image through Rayleigh fading channel is plotted at SNR = 10dB. We see that other than some random noise and block noise in the image, there are some overlaps in the image. This is due to the white Gaussian noise, phase estimation error in a coherence time, and ISI caused by frequency selective fading channel. From Figure 6, it can also be observed that using the LMS equalizer, the time-variant property of the channel change from estimation result of training data, so the image quality is inproved. In summary, it is observed that in AWGN channel, the image is degraded by random noise; and in Rayleigh fading channel, the image is degraded by random noise, block noise, and overlap. Also, the BER performance is poor in Rayleigh fading channel when compared to AWGN channel. These results obtained match the theoretical results.



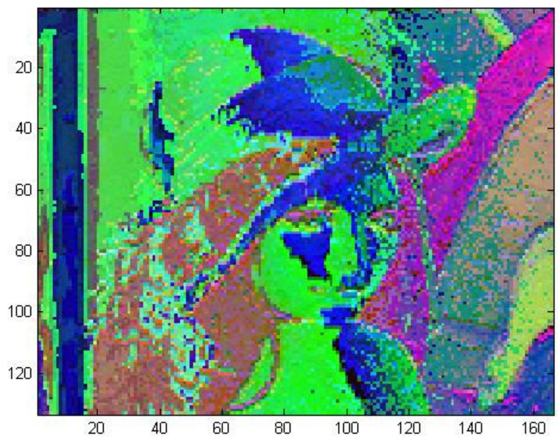
(a) (b)

Fig. 4: Theoretical Vs Simulated BER for (a) AWGN and (b) Rayleigh Fading Channel.

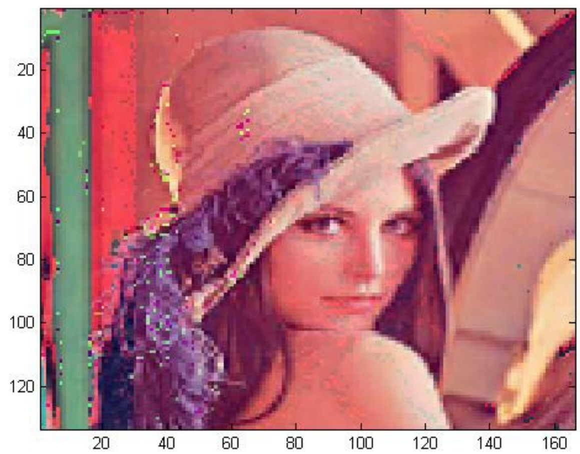


(a) (b)

Figure 5: (a) Original Image and (b) Image affected by AWGN Channel.



(a)



(b)

Figure 6: (a) Image received through the Rayleigh fading channel and (b) After adjuted using LMS equalisers.

IV. CONCLUSION

In this paper, the effect of passing an image through two different wireless channels (AWGN and Rayleigh Fading) is observed. It is observed that in AWGN channel, the image is degraded by random noise; and in Rayleigh fading channel, the image is degraded by random noise, block noise, and overlap. Also, the BER performance is poor in Rayleigh fading channel when compared to AWGN channel. We also compare and analysis the improvement of quality of received images using LMS equalization in Rayleigh fading channel. The results obtained match closely to the theoretical results.

REFERENCES

[1] A. V. Oppenheim, R. W. Schafer, J. R. Buck, "Discrete- time Signal Processing", Second Edition, 1999.
[2] T. S. Rappaport, "Wireless Communications: Principles and Practice", Second Edition, 2002.
[3] J. G. Proakis, "Digital Communications", Fourth Edition, 2001.
[4] K Banovic, "Adaptive Equalization: A Tutorial", Research Centre for Microsystems, University of Windsor, Windsor, Ontario, Canada, October 14, 2005.
[5] S. Haykin, "Adaptive Filter Theory", Fourth Edition, 2002.
[6] O.P. Sharma, V. Janyani and S. Sancheti, "Recursive Least Squares Adaptive Filter a better ISI Compensator", International Journal of Electronics, Circuits and Systems, 2009.
[7] R.H Kwong, E.W. Johnston, "A variable Step size LMS algorithm", IEEE Trans Signal Processing, Vol. 40, Issue.7, pp.1633-1642, July 1992.
[8] S. Choi, T-Lee and D Hong, "Adaptive error constrained method for LMS algorithm and application", Signal Processing, Vol. 85, Issue10, pp.1875 – 1897, Oct. 2005.
[9] W. Junfeng and Z. Bo, "Design of Adaptive Equalizer Based on Variable Step LMS Algorithm," Proceedings of the Third International Symposium on Computer Science and Computational Technology (ISCSCT '10) Jiaozuo, P. R. China, 14-15, pp. 256-258, August 2010.
[10] G. Malik and A. S. Sappal, "Adaptive Equalization Algorithms: An Overview", International Journal of Advanced Computer Science and Applications, Vol. 2, No.3, March 2011.
[11] Mahmood Farhan Mosleh, Aseel Hameed AL-Nakkash, "Combination of LMS and RLS Adaptive Equalizer for Selective Fading Channel," European Journal of Scientific Research ISSN 1450-216X Vol.43, No.1, pp.127-137, 2010.
[12] Y. Ben Jemiia**,** S. Cherif**,** M. Jaidane and S. Marcos, "Design of Decision Feedback Equalizer with short training sequence", IEEE Vehicular Technology Conference, August 2002.
[13] V. F. Pancaldi, "Frequency-domain equalization for space-time block-coded systems", Wireless Communications, IEEE Transactions, Vol 4, Nov 2005.
[14] J. Zheng and B. D. Rao, "LDPC-Coded MIMO Systems With Unknown Block Fading Channels: Soft MIMO Detector Design, Channel Estimation, and Code Optimization", IEEE Transactions on Signal Processing, Vol 54, No. 4, April 2006.