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Comparison Study of Linear Adaptive Equalization Filters for Image Transmission in OFDM Systems

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Abstract: Orthogonal frequency division multiplexing (OFDM) is the most commonly used transmission technique in multicarrier systems in all types of modern communication technologies. Fading is a great dispute in wireless communication systems. A linear adaptive equalization filter at the receiver can be used to minimize frequency selective fading. In this paper, image transmission through OFDM system with IEEE 802.11 under the influence of Rayleigh fading channel with linear adaptive equalizers (LMS, NLMS, and RLS) is presented. The performance of these filters is evaluated in terms of BER performance, bit errors for different step sizes using LMS, NLMS filters and different forgetting factors for RLS filter in MATLAB simulation environment.

Keywords: LMS, NLMS, RLS, OFDM, DFE, GSM

I. INTRODUCTION

In a communication system, higher transmission rates can be achieved by setting up orthogonal channels to each user by time/frequency/code division multiplexing. Orthogonal Frequency Division Multiplexing (OFDM) system is generally used as a multi-carrier scheme to have high data rate transmission capability. An OFDM system has high spectral efficiency, less interference from adjacent channels, robustness to multi-path delays and fading, ease of system scalability and flexibility under congested and noisy environment.

These features make OFDM an essential technology for next generation mobile wireless communication networks [1]-[2]. This OFDM technology is presently being incorporated in all latest mobile wireless communication systems. A few of them are 802.16 d/e based WLANs, Wi-Fi, Wi-Max, 4G LTE, etc [3] - [4]. During transmission process, the signal is distorted due to multipath fading; adaptive channel equalization can be used to compensate signal distortion at the receiver.

From the literature survey it is observed that, less work has been reported on image transmission through OFDM system using Adaptive Equalization. In this work, an OFDM system has been implemented using different linear adaptive equalization algorithms (Least Mean Square (LMS), Normalized Least Mean Square (NLMS) and Recursive Least Square (RLS)) to minimize the ISI.

The objective of this paper to improve the performance of OFDM system in terms of filter parameters for image transmission through Rayleigh Fading channel having GSM model for mobile communication system. Channel bandwidth of 200 KHz with1.8GHz centre frequency and base band sampling frequency is 20MHz is chosen for simulation. A channel with frequency selective fading with urban or terrain scenario with a delay spread of 5-25μs is considered [9].

The rest of the paper is organized as follows: section II describes about OFDM system, section III presents about Multipath Fading, Section IV discusses the Adaptive Equalization Filters, Section V discusses the experimental results. The final conclusions are discussed in section VI.

II. ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING (OFDM)

Multi carrier modulation schemes are most suitable for high data rates in real time applications. The most frequently used techniques are code division multiple access (CDMA) and orthogonal frequency division multiplexing (OFDM), affirming better bandwidth utilization, noise immunity and reliability. Fig 1 shows the block diagram of an OFDM System with adaptive equalization for Image Transmission.

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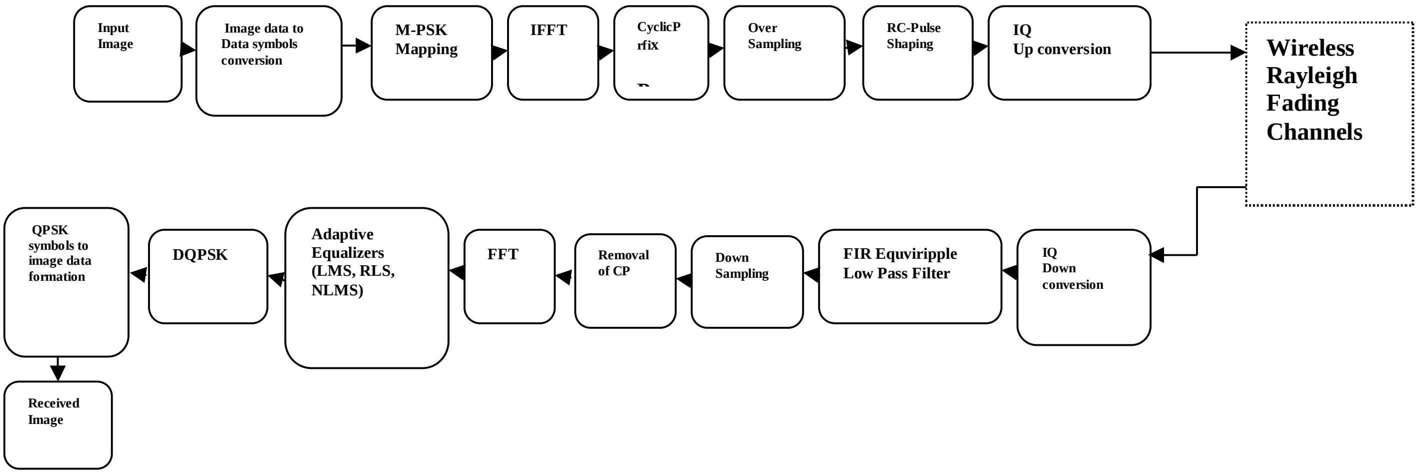

Fig. 1. Block Diagram of OFDM System

The entire block diagram can be divided into three sections. They are Transmitter Section, Channel Section and Receiver Section respectively. The main processing blocks present in transmitter section are: Source Signal Generator (Image Data to Data symbol conversion), Signal Modulator (M-PSK), and N-Point IFFT Block (for transmitting image data in a parallel fashion on different sub-carriers), cyclic prefix, over sampling, RC-pulse shaping, and IQ-Up conversion. The Channel Section is modelled by using Rayleigh Fading channel model. The receiver section contains the processing blocks which perform inverse operations corresponding to the operations performed in the transmitter section. The processing blocks present in the receiver section are: IQ-Down conversion, FIR equiripple low pass filter, down sampling, Removal of CP, N-Point FFT Block, Adaptive Equalizers(LMS,NLMS,RLS), Signal Demodulator, Source Signal Recovery Block (Recovered Data symbol conversion to Image Converter). For the evaluation of OFDM system for image transmission, we choose a Test image is as shown in Fig 2, which can be accessed by name "Baby". This is available in .JPG format with a size of 256 x 256. Fig 2 shows a noisy image which is received at the receiver's input. Table I shows the MATLAB simulation parameters.

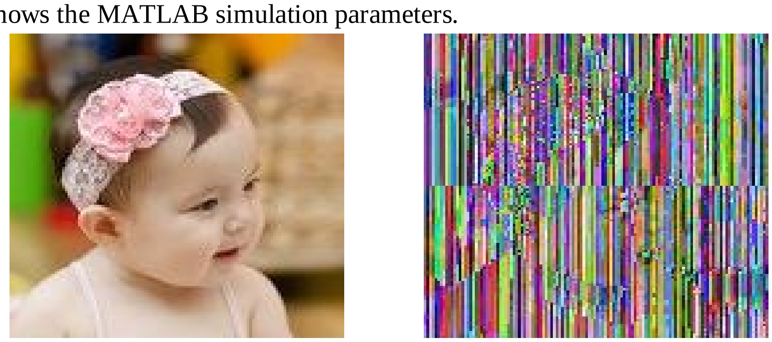

Fig. 2. Baby Test Image and Noise Image

TABLE I
MATLAB SIMULATION PARAMETERS

Property	Value
Total Number of Sub-Carriers	64
Cyclic Prefix Length	16
Modulation Schemes	M-PSK
Channel	Rayleigh Fading
Range of SNR in dB considered for evaluating BER	0-30 dB

III. MULTIPATH FADING

In wireless communications, the transmitted (Radio) signals arrive at the receiver by two or more paths, i.e., Multipath propagation. This multipath affect can cause errors and quality of communication. If a signal under goes rapid fluctuations in amplitude/phase or multipath delays for a short period of time or travel distance is termed as fading/small scale fading and is influenced by Speed of the mobile, speed of the surrounding objects and transmission bandwidth of the signal. For example the mobile user covers distances well in excess of a wave length in urban environment (changes in the signal power occur as the user passes buildings of different heights, vacant lots, intersections, and so forth). Over these longer distances, there will be a change in the average received power level about which the rapid fluctuations in the signal power occurs [8]. Frequency selective fading occurs when the signal bandwidth is more than the coherence bandwidth of the mobile radio channel or equivalently the symbols duration of the signal is less than the rms delay spread.

$$B_S >> B_C \qquad (1)$$

$$T_S << \sigma_\tau \qquad (2)$$

Where B_S is the signal bandwidth, B_C is the coherence bandwidth, T_S is symbol period and σ_τ is rms delay spread. To combat the distortion in frequency selective fading by using adaptive equalizer techniques and spread spectrum techniques can be used to mitigate frequency-selective ISI distortion.

TABLE II
DELAY SPREAD

Environment	σ_τ
Urban	1-25 µsec
Suburban	0.2-2µsec
Indoor	25-250nsec

The simulated GSM model has the carrier frequency is 1.8GHz, bandwidth of each channel is 200 KHz, Ts =5 µsec symbol period and considered urban/terrain environment (RMS delay spread is 1-25µsec) [9]. Table I gives the typical ranges for standard deviation of the delay spread for three different scenarios [10].

IV. ADAPTIVE EQUALIZER

Inter symbol Interference (ISI) has been recognized as the major obstacle to high speed data transmission over mobile radio channels [11]. Adaptive equalizers are used to reduce the inter symbol interference (ISI), additive noise and Frequency selective Fading. A channel with phase only distortion, a linear fractionally spaced equalizer can be used eliminate all of the ISI without any enhancement of the noise level. [12], the block diagram of the adaptive filter is shown in fig 3.

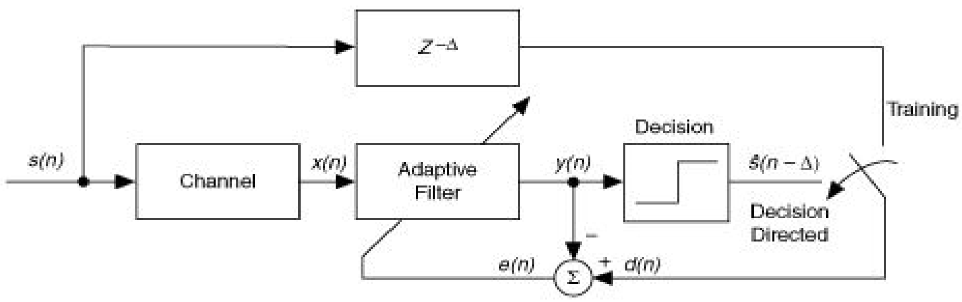

Fig. 3. Block Diagram of Adaptive Filter

In the above figure, *s*(*n*) is the signal that is transmit through the communication channel, and *x*(*n*) is the distorted output signal. To compensate for the signal distortion, the adaptive channel equalization system goes through the following two modes:

1) *Training Mode:* a training signal is needed to make equalizer to acquire the proper filter coefficients in the worst possible conditions. Therefore, the receiver substitutes known training signal in place of the slicer output. Once an agreed time has elapsed, the slicer output is substituted and the actual data transmission begins.
2) *Decision- directed Mode:* The Decision Direct mode means that the receiver decisions are used to generate the error signal [11]-[13].

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A. Least Mean Square Algorithm (LMS)

One of the most popular algorithms in adaptive signal processing is least-mean square algorithm (LMS), by Widrow and Hoff in 1959; many applications appeared to have the potential to use this fundamental concept. In the signal processing adaptive filters is a wide variety of stochastic gradient algorithm [14]-[15]. In that the LMS algorithm is an essential component of the family. An N-tap filter with weight vector W (n) at time instant 'n' denoted by

$$W(n) = [w_1(n) w_2(n) \cdots\cdots\cdots w_N(n)]^T \quad (3)$$

Let [**X (n)]** be the input sequence given by

$$\mathrm{X}(n) = [x(n) x(n-1) x(n-N+1)]^T \quad (4)$$

The superscript T in the above equation signifies matrix transpose. The filter output **y (n)** becomes

$$y(n) = W^T(n)\mathrm{X}(n) \quad (5)$$

The filter output y (n) to follow a desired signal d (n), and estimation error e(n) is defined as difference between desired response and filter output:

$$e(n) = d(n) - y(n) \quad (6)$$

The N-tap adaptive filtering algorithm adjusts the filter tap weight W(n) at each time instant of the measured value of e(n). The following equation then updates the weight vector of an LMS algorithm [16],

$$W(n+1) = W(n) + \mu e(n)\mathrm{X}(n) \quad (7)$$

In the above equation, 'μ' is a step size parameter that affects the convergence behaviour of the filter weights and the step size 'μ' introduced to control the stability of the algorithm [17].

B. Normalized Least Mean Square Algorithm (NLMS)

The NLMS algorithm is a special case of the LMS algorithm with time varying step size in which the step size varies with the input signal strength. [18]. In NLMS, the step size is denoted by

$$\mu(n) = \frac{\beta}{\| \mathrm{X}(n) \|^2} \quad (8)$$

Here 'β' is normalized step size. Its range is $0<\beta<2$. In the above equation, if x(n) is large, then LMS shows a problem with gradient noise amplification .To avoid noise amplification problem in the LMS, step size is divided by $\|x(n)\|^2$ in the NLMS [18]. The step size is an important parameter, an optimum step size chosen in NLMS, which controls the trade-off between rate of convergence and the steady-state mean-square error (MSE) of an algorithm [19].

C. Recursive Least Squares Algorithm (RLS)

In linear adaptive filtering algorithms, the most important and popular one is Recursive Least –Squares (RLS) algorithm. For highly correlated input signals, it has better convergence rate as compared to the LMS algorithm, and it is belongs to family of Kalman filter [20]. The cost function of the RLS algorithm is

$$\varepsilon(i) = \sum_{j=1}^{M} \beta(i,j) \,|\, e(j) \,|^2 \quad (9)$$

In above equation, 'i' is the observable data length, β (i, j) is weighting data and e (i) is error which is to be minimized. The performance of RLS algorithm in terms of convergence rate, tracking, stability and misadjustment depends on the forgetting factor. The forgetting factor or Exponential weighting factor of the RLS algorithm is defined as,

$$\beta(n,i) = \lambda^{n^{-i}} \qquad i=1, 2, 3............., M \quad (10)$$

Where parameter λ is forgetting factor that controls the convergence speed and its value is close to unity [21]. If forgetting factor 'λ' value is close to unity, the RLS algorithm gets low misadjustment and good stability but its tracking capabilities are reduced. For smaller vales of 'λ' increase the tracking capability but increase the misadjustment, and this can affect the low stability [20]. The output y(n) of the filter is defined as follows ,

$$y(i) = W^H(i)\mathrm{X}(i) \quad (11)$$

$$e(i) = d(i) - W^H(i)\mathrm{X}(i) \quad (12)$$

From values of e (i) and β (i,j) ,the exponential weighted error is

$$\varepsilon(i) = \sum_{j=1}^{M} \lambda^{i-j} \,|\, d(i) - W^H(i)X(i) \,|^2 \tag{13}$$

$$\delta\lambda^i \,\|\, W(i) \,\|^2 = \delta\lambda^i W^H(i)W(i) \tag{14}$$

In above equation' δ' is regularizing parameter and it has a real positive value. This parameter depends on tap-weight vector W (i) except δλi .Correlation vector of tap input vectors X (i) and φ (i) are given below

$$\varphi(i) = \sum_{j=1}^{M} \lambda^{i-j} X(i) X^H(i) + \delta\lambda^i I \tag{15}$$

In the above equation, where I is the identity matrix, cross correlation vector between tap inputs of filter X(i) and desired response d(i) is Z(i) is given as

$$Z(i) = \sum_{j=1}^{M} \lambda^{i-j} X(i) d^*(i) \tag{16}$$

The value of cost function ε (i) for optimum tap vector ŵ (i) attain its minimum value when

$$\varphi(i)\hat{W} = Z(i) \tag{17}$$

$$\varphi^{-1}(i) = \lambda^{-1}\varphi^{-1}(i-1) - \frac{\lambda^{-2}\varphi^{-1}(i-1)X(i)X^H(i)\varphi^{-1}(i-1)}{1+\lambda^{-1}X^H(i)\varphi^{-1}(i-1)X(i)} \tag{18}$$

$$\text{Let } P(i) = \varphi^{-1}(i) \tag{19}$$

$$k(i) = \frac{\lambda^{-1}P(i-1)X(i)}{1+\lambda^{-1}X^H(i)P(i-1)X(i)} \tag{20}$$

$$P(i-1) = P(i-1) - \lambda^{-1}k(i)X^H(i)P(i-1) \tag{21}$$

Where P(i) is inverse correlation matrix and k(i) is gain vector. By rearranging the equation, we get,

$$k(i) = \lambda^{-1}P(i-1)X(i) - \lambda^{-1}k(i)X^H(i)P(i-1) \tag{22}$$

$$k(i) = [\lambda^{-1}P(i-1) - \lambda^{-1}k(i)X^H P(i-1)]X(i) \tag{23}$$

The weight vector are updated as follows

$$\hat{W}(i) = \hat{W}(i-1) + k(i)[d^*(i) - X^H(i)\hat{W}(i-1)] \tag{24}$$

$$\hat{W}(i) = \hat{W}(i-1) + k(i)\varepsilon^*(i) \tag{25}$$

$$\varepsilon(i) = d(i) - \hat{W}H(i-1)X(i) \tag{26}$$

$$\varepsilon(i) = d(i) - \hat{W}H(i-1)X(i) \tag{27}$$

This is the prior estimation error. The inner products $\hat{W}H(i-1)X(i)$ represent an estimate of the desired response [22]-[24].

V. RESULTS AND DISCUSSION

The comparison of recovered images of LMS and NLMS at different step sizes and the recovered images for RLS filter at different Forgetting Factors is presented in Table III. From simulation result, the received image at low SNR, the image has random noise for all the algorithms. For SNR is about 10dB NLMS performance is better than the other algorithms. The signal constellation diagrams of OFDM Transmitter and Receiver for LMS, NLMS and RLS at 10 dB SNR is shown in Fig 3 & Fig 4.

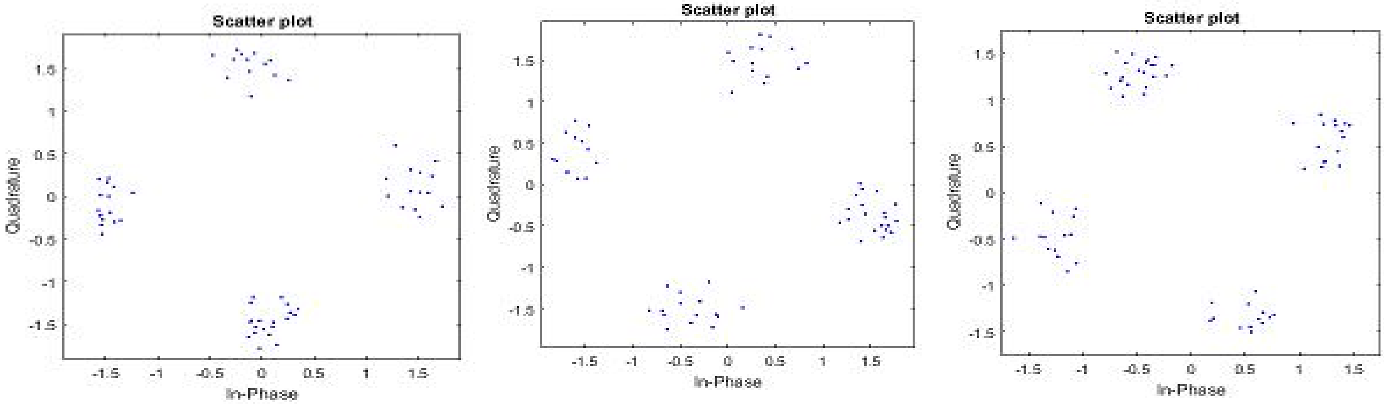


Fig. 4. Constellation Diagrams at transmitter with LMS, NLM and RLS at SNR = 10dB

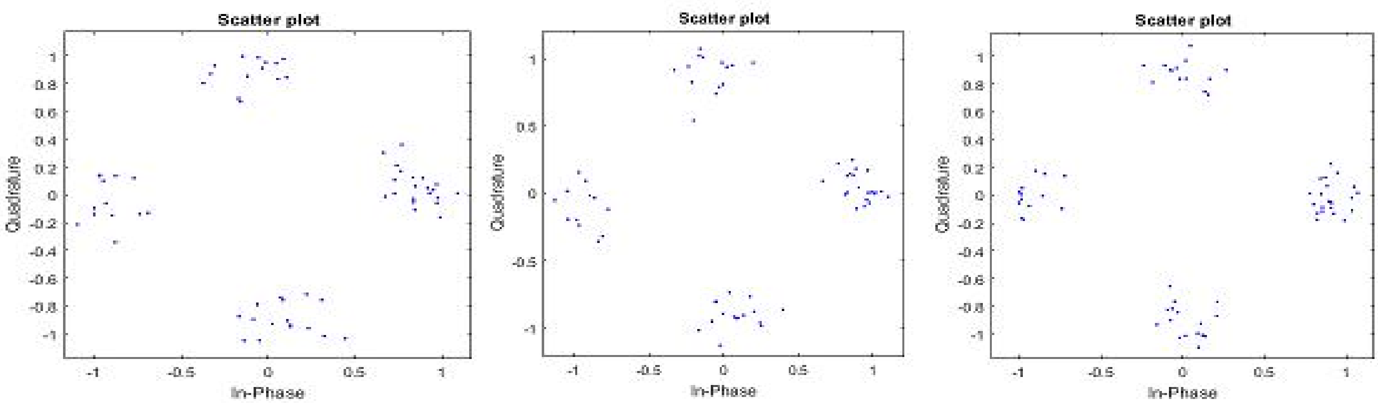
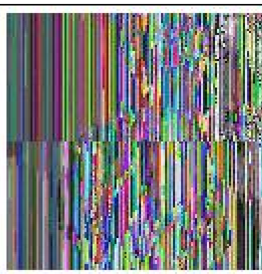
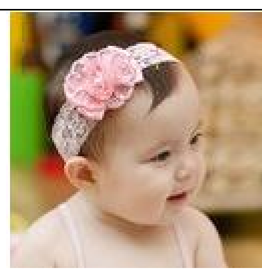
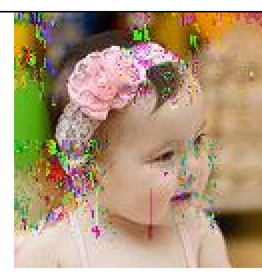
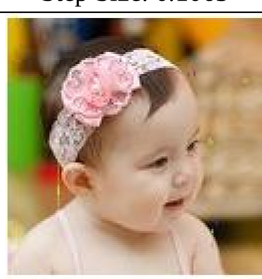
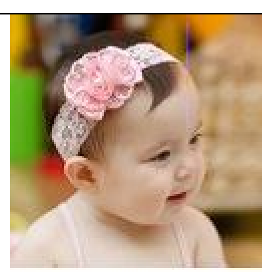


Fig. 5. Constellation Diagrams at receiver with LMS, NLM and RLS at SNR = 10dB

TABLE IIII
COMPARISON OF RECOVERED IMAGES FOR LMS, NLMS AND RLS AT THE OFDM RECEIVER

LMS	NLMS	RLS
 Step Size: 0.1063	 Step Size: 0.4140	 Forgetting Factor:0.5
 Step Size: 0.1663	 Step Size: 0.7650	 Forgetting Factor:0.8

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Step Size: 0.0128	Step Size: 0.9950	Forgetting Factor:1.0

TABLE IIIV
COMPARISON OF NUMBER OF ERRORS FOR LMS ALGORITHM (FILTER ORDER=8) WITH DIFFERENT STEP SIZES

S. No	SNR (dB)	Total No. Of Errors					
		LMS			NLMS		
		Step Size=0.1063	Step Size=0.0382	Step Size=0.0924	Step Size=0.4140	Step Size=0.7650	Step Size=0.9950
1	0	21288	3282	2406	1277	2567	4882
2	3	53717	1215	823	270	459	1018
3	6	107421	576	360	224	316	183
4	9	134039	409	251	54	230	286
5	12	141644	396	212	35	81	130
6	15	144953	227	238	100	108	170
7	18	145648	298	233	127	63	307
8	21	146912	360	187	154	95	178
9	24	146785	284	304	172	89	53
10	27	147459	306	350	84	198	141
11	30	147941	317	186	153	144	134

TABLE V
COMPARISON OF WEIGHT FACTORS FOR LMS (FILTER ORDER=8)

S. No	Step Size=0.08674		Step Size=0.0382		Step Size=0.0924	
	Real Weights(R)	Imaginary Weights(I)	Real Weights(R)	Imaginary Weights(I)	Real Weights(R)	Imaginary Weights(I)
1	14.4559	10.9765	-0.5019	0.1267	-0.2193	0.2651
2	-6.1025	4.8346	-0.0098	-0.0083	-0.0042	0.0018
3	-2.4062	-20.7123	-0.0034	-0.0278	-0.0109	-0.0059
4	14.6328	-6.3375	0.0127	0.0112	0.0119	0.0060
5	-9.0934	1.6194	0.0237	0.0247	0.0077	3.3189e-04
6	4.6887	-16.3585	0.0090	0.0124	0.0066	-0.0013
7	-10.6092	-10.1155	0.0016	-0.0042	2.4146e-04	-0.0053
8	-11.2397	-6.2911	-0.0138	-4.9069e-04	5.1695e-04	0.0010

The Total number of errors occurred at different SNRs and step sizes for LMS and NLMS algorithm are presented in Table IV. It is we observed from table IV for a Step size of 0.0924, LMS algorithm gives better performance and at Step size 0.9950 NLMS algorithm gives better performance. Table V and VI, shows weight factor of LMS and NLMS for different step sizes.

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TABLE VI
COMPARISON OF WEIGHT FACTORS FOR NLMS (FILTER ORDER=8)

S. No	Step Size=0.4140		Step Size=0.7650		Step Size=0.9950	
	Real Weights(R)	Imaginary Weights(I)	Real Weights(R)	Imaginary Weights(I)	Real Weights(R)	Imaginary Weights(I)
1	0.0862	0.0238	-0.1294	0.0249	1.2968	0.0616
2	0.0071	0.0058	-0.0122	-0.0031	0.0379	0.0420
3	-0.0057	-0.0060	0.0113	0.0039	0.0111	-0.0183
4	-0.0101	0.0031	0.0108	-0.0104	-0.0615	0.0261
5	0.0124	0.0017	-0.0171	0.0063	0.0512	-0.0025
6	0.0071	0.0024	-0.0106	-0.0015	-0.0146	0.0039
7	0.0022	0.0017	-0.0038	0.0011	0.0083	0.0220
8	0.0070	-0.0045	2.1132e-04	0.0070	-4.4150e-04	-0.0618

The Comparison Plot for BER vs.SNR of Linear Adaptive equalization Algorithms are LMS and NLMS is shown in Fig 6. I t is concluded that NLMS algorithm gives better performance compared to LMS algorithm.

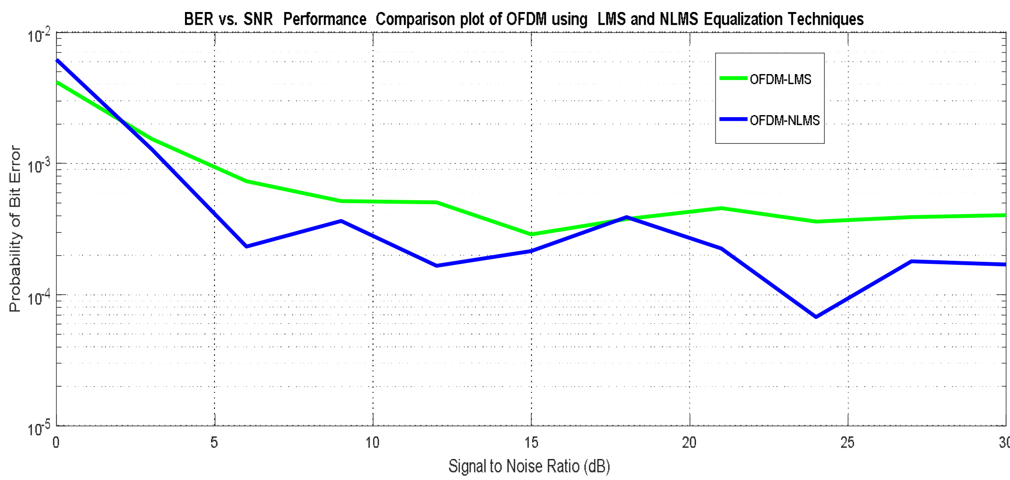

Fig. 6. BER vs. SNR Comparison plot of LMS and NLM

The Total number of errors for different SNRs and Forgetting Factor for RLS algorithm is shown in Table VII. From Table it can be observed that at Forgetting Factor 1.0 RLS algorithm gives better performance. The weight factors of RLS algorithm and BER vs. SNR curve as shown below.

TABLE VIII
COMPARISON OF NUMBER OF ERRORS RLS (FILTER ORDER=8) WITH DIFFERENT FORGETTING FACTOR

S. No	Total Image Pixels	SNR (dB)	Total No. Of Errors		
			Forgetting Factor=0.5	Forgetting Factor=0.8	Forgetting Factor=1.0
1	196608	0	2285	7779	1637
2	196608	3	13902	3422	1198
3	196608	6	9744	2036	962
4	196608	9	7764	1475	1047
5	196608	12	6820	852	1013
6	196608	15	6243	618	1046
7	196608	18	5765	500	1053
8	196608	21	5598	359	970
9	196608	24	5588	427	896
10	196608	27	5544	356	1012
11	196608	30	5539	309	761

TABLE IX
COMPARISON OF WEIGHT FACTORS FOR RLS (FILTER ORDER=8)

S. No	Forgetting Factor =0.5		Forgetting Factor =0.8		Forgetting Factor =1.0	
	Real Weights	Img Weights	Real Weights	Img Weights	Real Weights	Img Weights
1	-0.1566	0.3149	-0.2156	-0.3427	-0.7670	0.0985
2	6.7814e-01	0.0058	-0.0036	-0.0033	-0.0223	-0.0076
3	-6.1599e-04	-0.0011	0.0040	0.0015	0.0097	0.0136
4	0.0047	-0.0029	-4.6183e-04	0.0088	0.0215	-0.0214
5	0.0011	0.0020	-0.0034	2.9574e-04	0.0152	0.0069
6	-0.0016	-0.0050	0.0097	-0.0016	-0.0021	-0.0013
7	0.0030	0.0028	-0.0025	-2.7942e-04	-0.0099	-0.0080
8	0.0028	5.8435e-04	-7.115e-04	0.0033	0.0094	0.0106

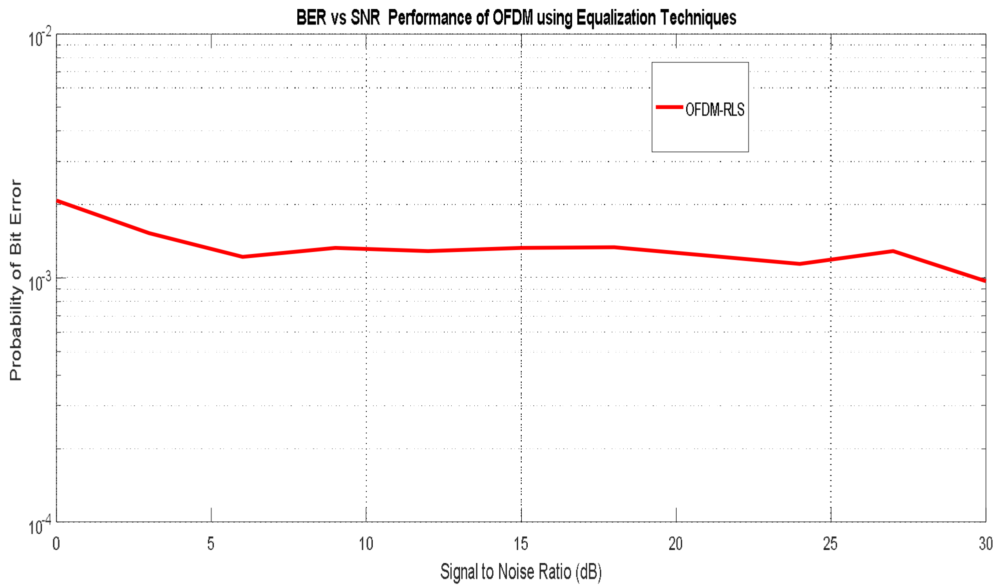

Fig. 7. SER vs. SNR plots of RLS equalization

VI. CONCLUSIONS

The OFDM system has been Implemented with LMS, NLMS and RLS Linear Adaptive Equalization Algorithms for image transmission through Rayleigh Fading channel. We conclude that from simulation results, NLMS algorithm gives better Image quality comapred to the rest of the techniques (LMS and RLS) used in this paper. In future, Image Transmission through OFDM System by using Non–Linear Adaptive Equaliztion algorithms gives better performance.

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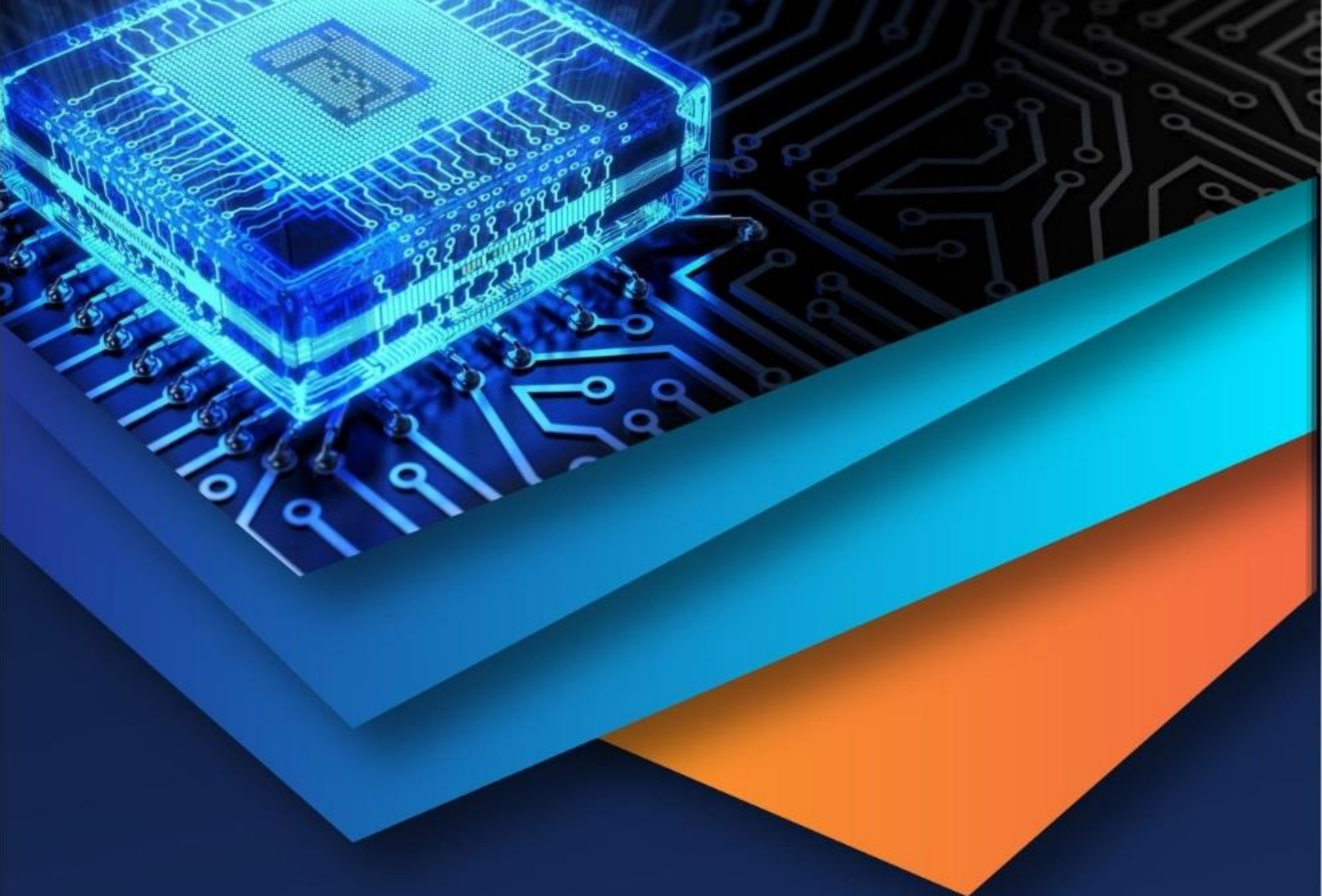

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