Design and Performance of Frequency Domain Equalization for IEEE 802.11a Physical Layer in SUI Channels

Alhamzah Taher Mohammed

Abstract— ***this research investigates the performance of adaptive equalization processing for Broadband Wireless Access (BWA) is a promising technology which can offer high speed voice, video and data service up to the customer end. Due to the absence of any standard specification, earlier BWA systems were based on proprietary standard. IEEE 802.11a Wireless MAN standard specifies a Medium Access Control (MAC) layer and a set of PHY layers to provide fixed and mobile Broadband Wireless Access (BWA) in broad range of frequencies. The OFDM has adopted in IEEE 802.11a PHY layer for the equipment manufacturer due to its robust performance in multipath environment. The paper investigates the simulation performance of IEEE 802.11a OFDM PHY layer. The overall performance analysis of the proposed equalization technique over Stanford University interim (SUI) channels was performed .The evaluation was done in simulation developed in MATLAB.***

Index Terms—***(BWA), IEEE, BWA, MAN, (MAC), OFDM, PHY, (SUI)***

I. INTRODUCTION

At current IEEE 802.11 is the standard for WLANs [1]. Both the medium access control (MAC) and the physical (PHY) layers for WLANs have been specified in this standard. This separation of 2 layers done in the standard allow a functional separation of the standard and more significantly allows a single data protocol to be used with numerous different Radio Frequency (RF) transmission methods [2]. The scope of IEEE 802.11 working groups (WGs) is to suggest and improve MAC and PHY layer specifications for WLAN to handle mobile as well as portable stations [3]. A new modulation technique called Orthogonal Frequency Division Multiplexing (OFDM) was introduced by 802.11a which allows higher data transmission rates in the smaller bandwidth. Besides proposing the new modulation method, 802.11a also switches from the rapidly getting overused 2.4GHz ISM band to 5GHz ISM band. The 5GHz ISM bandwidth is not continuous. There are two areas 5.15GHz – 5.35GHz and 5.725GHz – 5.825 GHz. Both areas are separated by 802.11a into 12 overlapping carriers (similar to 802.11 channels) spaced 20MHz. 802.11a supports bandwidth up to 54 Mbps and signals in a regulated frequency spectrum around 5 GHz. This higher frequency compared to 802.11b shortens the range of 802.11a networks [3]. The higher frequency also means 802.11a signals have further difficulty penetrating walls and other obstacles. Adaptive equalization has been a powerful area of research for many years. It begin with the invention of the zero forcing (ZF) equalizer by Lucky in 1965 [4, 5]. Lucky's equalizer is based on the peak distortion criterion. Lucky too presented decision-directed learning [5] to train an adaptive equalizer. The decision directed mode is too able to track channel variations. Later on it was seen that the ZF equalizer is inferior to equalizers based on the mean square error (MSE) criterion in terms of noise enhancement. A celebrated adaptive algorithm for tuning an equalizer based on the MSE criterion is known as the LMS algorithm[6, 7]. The LMS algorithm belongs to the family of stochastic gradient algorithms. Unluckily the LMS algorithm's convergence behavior hard depends on the eigenvalue spread of the correlation matrix of the input samples. Numerous variants of the original LMS algorithm exist, for example the normalized LMS algorithm [8-11]. The normalized LMS algorithm has benefits compared to the original LMS algorithm in terms of gradient noise amplification and convergence rate. While, the original version of the LMS and ZF algorithm are applied only to a linear transversal equalizer (LE), these algorithms can also be applied to a DFE. The DFE proposed by Austin[12] uses decision-feedback to cancel the interference from earlier detected symbols. The DFE outperforms the LE in the attendance of spectral nulls in the channel. Godard [13] applied Kalman filter theory to the problem of setting the tap values of a transversal equalizer. In this paper he treated the growing memory recursive least-square problem in a stochastic state-space framework. Afterward it was known by Sayed and Kailath [14], that the RLS algorithm [15] can be derived from Kalman theory, which lead to a correspondence between Kalman variables and RLS variables. The chief advantage of the RLS algorithm[15] compared to the LMS algorithm is that it is relatively insensitive to the eigenvalue spread of the correlation matrix of the input samples and so converges faster. However, the chief problem of the RLS algorithm is that the complexity is proportional to N2, where N is the number of taps in the equalizer. Consequently research has been dedicated into finding new RLS algorithms that exhibiting a linear increase in complexity with the number of taps, leading to so named fast RLS algorithms. Falconer and Liung proposed the fast RLS Kalman algorithm in [16] and later Ling and Proakis proposed the least square lattice DFE in [Ling85].

Revised Version Manuscript Received on May 20, 2015.

Alhamzah Taher Mohammed, Assist Lecturer, Department of Electronic and Communications Engineering, Al-Nahrain University in Baghdad, Iraq

Also square-root RLS algorithms have been proposed [14], which show superior numerical stability than the standard RLS algorithm. A good tutorial review of these initial adaptive equalization approaches is given in [17]. Additional class of equalizers utilizes a sequence-based method known as maximum likelihood sequence estimation (MLSE). Full MLSE is very complex, but using a dynamic programming method known as the Viterbi algorithm allows a reduction in complexity, since only a fraction of all possible input sequences needs to be considered. The Viterbi algorithm was initially used for decoding convolutional codes. Only a few years later, Forney applied the MLSE via the Viterbi procedure to the problem of signal detection in the existence of ISI. In the following four paragraphs numerous equalization methods, which were not considered in this work are brief. Bayesian equalizers and MLSE implemented via the Viterbi algorithm are not considered here owing to their computational complexity. For BWA channels in which the ISI spans numerous symbols, these probabilistic procedures become impractical due to an exponential development in computational complication. The complexity trade-off can clearly be seen by considering a numerical sample. In this paper, supposing the Stanford University (SUI) [18], Phase-shift keying (PSK) modulation, quadrature phase shift keying (QPSK) modulation, Quadrature amplitude modulation 16 and 64 at 5MS/s

II. FREQUENCY DOMAIN EQUALIZERS (FEQS)

Frequency domain equalizers (FEQs) have been applied extensively in multicarrier systems to enhance transmission rate by reducing transmit redundancy in the form of guard interval. The equalization algorithm is able to remove intersymbol and intercarrier interference (ISI and ICI) incurred by the reduction or the absence of this redundancy by properly exploiting null subcarriers that are inherent in standardized multicarrier systems. The algorithm does not require additional temporal nor spatial diversity at the receiver to mitigate the channel-induced interferences.

III. PROPOSED MODEL OF IEEE 802.11A PHYSICAL LAYER IN SUI CHANNELS

At principal, a supplied a IEEE 802.11a Physical Layer model, from MathWorksTM in the MATLAB® & SIMULINK® R2014a software package, was amended and its performance measured. The proposed transceivers for the physical layer model based of Equalization Processing in diverse SUI channels will be presented in this paper. The block diagram in Figure 1 represents the whole system model for proposed design.

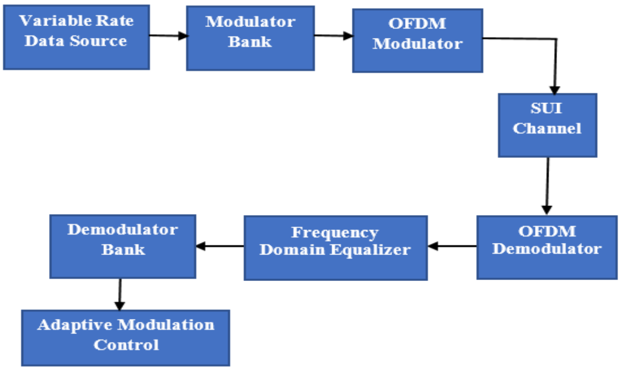

Figure 1.Proposed Model of IEEE 802.11a Physical Layer in SUI Channels

This model demonstrations an end-to-end baseband model of the physical layer of a wireless local area network (WLAN) according to the IEEE® 802.11a standard. The model supports all required and optional data rates: 6, 9, 12, 18, 24, 36, 48, and 54 Mb/s. The model also demonstrates adaptive modulation and coding over a dispersive multipath fading SUI channels, whereby the simulation varies the data rate dynamically. Note that the model uses an artificially high SUI channels fading rate to make the data rate change further rapidly and thus make the imagining more animated and useful. The model comprises components that model the important features of the WLAN 802.11a standard. The top row of blocks comprises the transmitter components though the bottom row comprises the receiver components. The communication system in this model performs these tasks: Generation of random data at a bit rate that varies during the simulation. The varying data rate is complete by enabling a source block periodically for a period that depends on the favourite data rate. Coding, interleaving, and modulation using one of numerous systems specified in the standard. Then choice any of the modulator blocks in the subsystem In specific, each modulator block in the bank performs these tasks: Convolutional coding and puncturing using code rates of 1/2, 2/3, and ¾ Data interleaving, BPSK, QPSK, 16-QAM, and 64-QAM modulation. OFDM (orthogonal frequency division multiplexing) transmission using 52 data subcarriers, 4 pilots, 64-point FFTs, and a 16-sample cyclic prefix. PLCP (physical layer convergence protocol) preamble modelled as four long training sequences. Dispersive multipath fading SUI channels. Receiver equalization and Viterbi decoding.

IV. SIMULATION AND RESULTS

In this section explain the performance of IEEE 802.11a Physical Layer in SUI Channels the Formation block named Model Parameters allows you to set parameters such as the structure of each OFDM frame, and trackback depth for the Viterbi decoder. One parameter of specific interest for the adaptive modulation and coding in this sample is the Low-SNR thresholds parameter. This is a seven-element vector that indicates how the simulation must choose a data rate based on the SNR estimate.

The model has eight modes, each associated with a particular modulation scheme and convolutional code. The seven thresholds are the boundaries between eight adjacent regions that correspond to the eight modes. Preferably, the simulation must use the highest-throughput mode that attains a desired packet error rate. Determining suitable thresholds often includes running the simulation multiple times, variable the values of the Low-SNR thresholds parameter. To view data graphically, open the presentation window by double-clicking the Signal Visualization icon. The plots within the display window demonstration a portion of the random binary data, meant to help you imagine the varying data rate. Scatter plots of the received signal before and after equalization. From the plot of the equalized signal, you can tell which modulation type the system is presently using, because the plot resembles a signal constellation of 2, 4, 16, or 64 points. The power spectrum of the received signal before and after equalization, in dB. The dynamics of the signal's spectrum before equalization depend on the Fading mode parameter in the Multipath SUI Channels block. The estimate of the SNR based on the error vector magnitude. The bit rate of the transmission. The bit error rate per packet. For greatest packets, the BER is zero. Because this plot uses a logarithmic scale for the vertical axis, BER values of zero do not look in the plot. The previous blocks display numerical results: The PER block displays the packet error rate as a percentage. The SNR block at the top level of the model demonstrations an estimate of the SNR based on the error vector magnitude. The SNR block in the Multipath SUI Channels displays the SNR based on the received signal power. The Bit Rate block demonstrations which of the bit rates quantified in the standard is presently in use. In this section the simulation of the proposed SUI channels for the IEEE 802.11a Physical Layer baseband transceiver beside the BER performance of the system regarded in SUI channel models in different modulation type

A. SUI-1 channel:

When using this type channel the BPSK 1/2, QPSK ½ and ¾, QAM-16 ½ and ¾ and QAM-64 2/3 and ¾, it be able to be understood that for BER=$10^{-3}$the SNR necessary for BPSK 1/2 is about 3 dB while in QPSK ¾ the SNR is about 7 dB and QPSK ½ =8.5dB, in QAM-16 ½ is about 13.5 dB and QAM-16 ¾ is about 16.2 dB, in QAM-64 2/3 is about 18.5 dB and QAM-64 3/4 is about 22.5 dB from Figure 2 it is found that the when using BPSK ½ outperforms important other schemes modulation for this channel model.

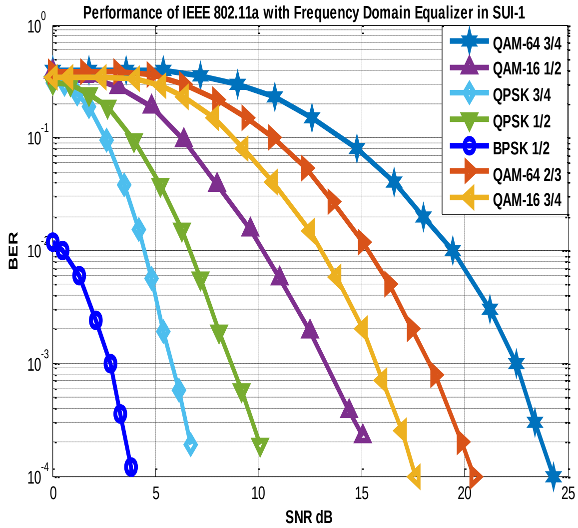

Fig .2. BER performance of proposed model in SUI-1 channel

B. SUI-2 channel:

In this simulation shape some influential results were obtained. the BPSK 1/2, QPSK ½ and ¾, QAM-16 ½ and ¾ and QAM-64 2/3 and ¾, it be able to be understood that for BER=10^{-3} the SNR necessary for BPSK 1/2 is about 4.9 dB while in QPSK ¾ the SNR is about 11 dB and QPSK ½ =14.9dB, in QAM-16 ½ is about 17.5 dB and QAM-16 ¾ is about 21.5dB, in QAM-64 2/3 is about 21.5 dB and QAM-64 3/4 is about 30dB from Figure 3 it is found that the when using BPSK ½ outperforms important other schemes modulation for this channel model.

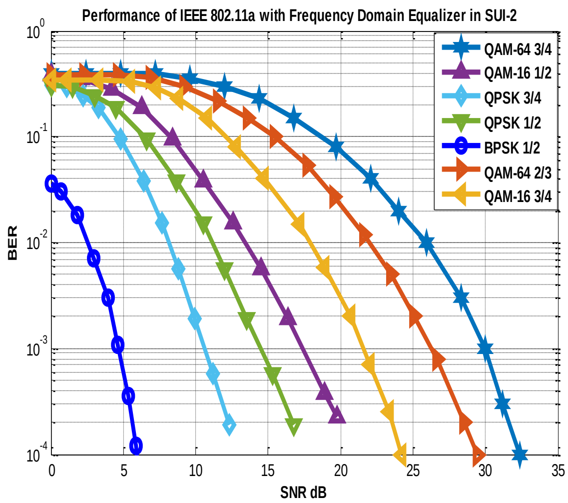

Fig .3. BER performance of proposed model in SUI-2 channel

C. SUI-3 channel:

In the SUI-3 channel, the BPSK 1/2, QPSK ½ and ¾, QAM-16 ½ and ¾ and QAM-64 2/3 and ¾, it be able to be understood that for BER=10^{-3} the SNR necessary for BPSK 1/2 is about 8 dB while in QPSK ¾ the SNR is about 12.2 dB and QPSK ½ =18.5dB, in QAM-16 ½ is about 23.5 dB and QAM-16 ¾ is about 27dB, in QAM-64 2/3 is about 30dB and QAM-64 3/4 is about 32.5dB from Figure 4 it is found that the when using BPSK ½ outperforms important other schemes modulation for this channel model.

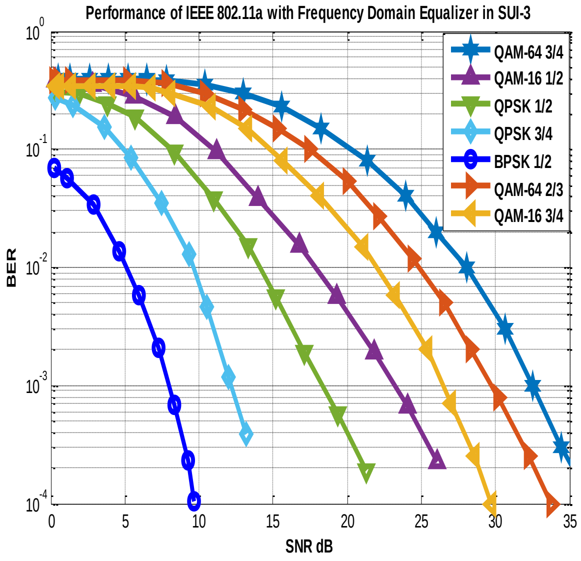

Fig .4. BER performance of proposed model in SUI-3channel

D.SUI-4 channel:

Using like methodology as in the previous section, simulations for SUI-4 channel the BPSK 1/2, QPSK ½ and ¾, QAM-16 ½ and ¾ and QAM-64 2/3 and ¾, it be able to be understood that for BER=10^{-3} the SNR necessary for BPSK 1/2 is about 10.1 dB while in QPSK ¾ the SNR is about 14.9 dB and QPSK ½ =20dB, in QAM-16 ½ is about 25 dB and QAM-16 ¾ is about 28.5dB, in QAM-64 2/3 is about 32dB and QAM-64 3/4 is about 35dB from Figure 5 it is found that the when using BPSK ½ outperforms important other schemes modulation for this channel model.

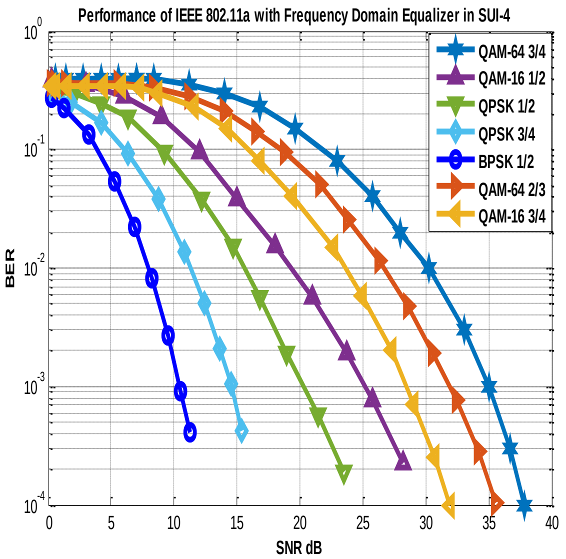

Fig .5. BER performance of proposed model in SUI-4channel

E. SUI-5 channel:

The BPSK 1/2, QPSK ½ and ¾, QAM-16 ½ and ¾ and QAM-64 2/3 and ¾, it be able to be understood that for BER=10^{-3} the SNR necessary for BPSK 1/2 is about 17dB while in QPSK ¾ the SNR is about. 20.1 dB and QPSK ½ =24.9dB, in QAM-16 ½ is about 28 dB and QAM-16 ¾ is about 23dB, in QAM-64 2/3 is about 34.9dB and QAM-64 3/4 is about 37.5dB from Figure 6 it is found that the when using BPSK ½ outperforms important other schemes modulation for this channel model.

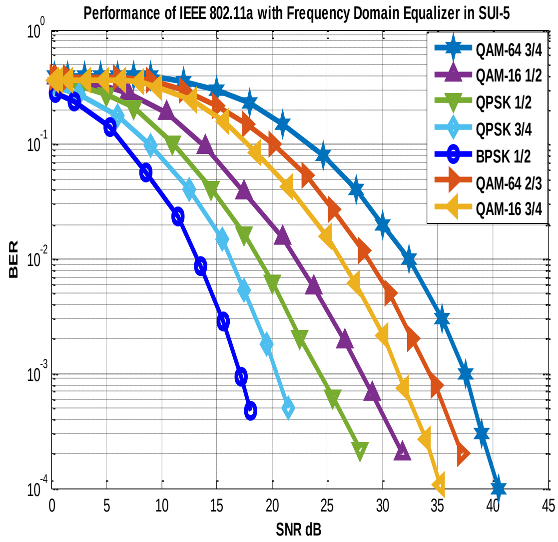

Fig .6. BER performance of proposed model in SUI-5channel

F. SUI-6 channel:

The BPSK 1/2, QPSK ½ and ¾, QAM-16 ½ and ¾ and QAM-64 2/3 and ¾, it be able to be understood that for BER=10^{-3} the SNR necessary for BPSK 1/2 is about 21.5dB while in QPSK ¾ the SNR is about. 24.9dB and QPSK ½ =29.9dB, in QAM-16 ½ is about 32dB and QAM-16 ¾ is about 35.1dB, in QAM-64 2/3 is about 39.9dB and QAM-64 3/4 is about 42.5dB from Figure 7 it is found that the when using BPSK ½ outperforms important other schemes modulation for this channel

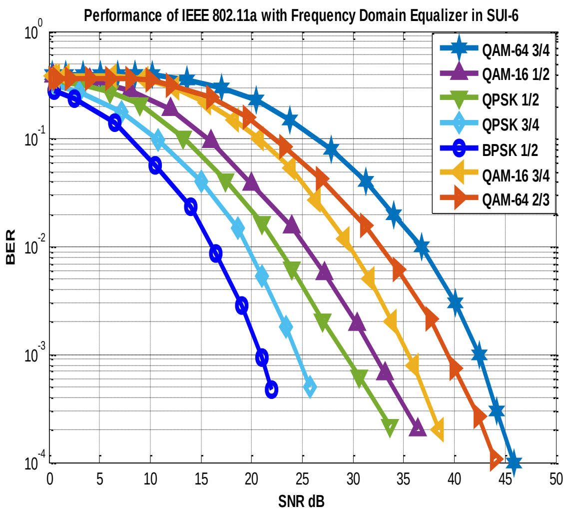

Fig .7. BER performance of proposed model in SUI-6 channel

V. CONCLUSION

This work examines the SUI channels-induced interference problem produced by insufficient CP in OFDM based systems.

Insufficient CP state may happen when SUI channel models delay spread is very long, when the transmitter purposely shortens the guard interval to reduce transmission overhead in order to increase scheme throughput, or in SUI channel model environment when the propagation delay differences. A null subcarrier based frequency domain equalizer is proposed to moderate the adverse effects produced by the shortened guard interval as well as channel noise. The algorithm has presented to have less computational complexity and it also outclasses in terms of BER. Also, the structure is not restricted by the number of cyclic prefix and the number null subcarriers that exists, nor by their location

REFERENCES

1. IEEE 802.11 Task Group, "Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications: High-speed Physical Layer in the 5 GHz Band, 1999.," 1999.
2. B. O'Hara, and Al Petrick,, "The IEEE 802.11 Handbook: A Designer's Companion, New York, IEEE Press, 1999.," 1999.
3. “A Brief Tutorial on the PHY and MAC layers of the IEEE 802.11b standard” Benjamin E. Henry july 12.
4. R.W. Lucky, "Automatic Equalization for Digital Communications," Bell System Technical Journal, Vol. 44, No. 4, April 1965, pp. 547-588, 1965.
5. R.W. Lucky, "Techniques for Adaptive Equalization of Digital Communications," Bell System Technical Journal, Vol. 45, pp. 255-286, 1965.
6. B. Widrow, "Adaptive Filters, I: Fundamentals", Stanford Electronics Laboratory, Stanford University, Standord, CA, Technical Report No. 6764-6, December 1966," 1966.
7. D. M. T. K. Rupali V. Mane, "Implementation of Adaptive Filtering Algorithm for Speech Signal on FPGA," INTERNATIONAL JOURNAL OF INNOVATIVE RESEARCH IN ELECTRICAL, ELECTRONICS, INSTRUMENTATION AND CONTROL ENGINEERING Vol. 2, Issue 3, March 2014, 2014.
8. J.I. Nagumo and A. Noda, "A learning method for system identification”, IEEE Transactions on Automated Control, Vol. AC-12, No. 3, June 1967, pp. 282-287," 1967.
9. A.E. Albert and L.S. Gardner, " “Stochastic Approximation and Nonlinear Regression”, MIT Press, Cambridge, ," 1967.
10. R.R. Bitmead and B.D.O. Anderson, "“Performance of adaptive estimation algorithms in dependent random environments”, IEEE Transactions on Automated Control, Vol. AC-25, No. 4, August 1980, pp. 788-794," 1980.
11. N. D. V. Muhammed O. Sayin, and Suleyman Serdar Kozat,, "A Novel Family of Adaptive Filtering Algorithms Based on the Logarithmic Cost," IEEE TRANSACTIONS ON SIGNAL PROCESSING, VOL. 62, NO. 17, SEPTEMBER 1, 2014, 2014.
12. M.E. Austin, "“Decision-feedback equalization for digital communication over dispersive channels”, MIT Lincoln Lab., Lexington, MA, USA, Technical Report 437, August " 1967.
13. D.N. Godard, "“Channel Equalization Using a Kalman Filter for Fast Data Transmission”, IBM Journal of Research and Development, Vol. 18, , pp. 267-273," 1974.
14. A.H. Sayed and T. Kailath, " “A State-Space Approach to Adaptive RLS Filtering”, IEEE Signal Processing Magazine, Vol. 11, July 1994, pp. 18-60," 1994.
15. S. Haykin, "“Adaptive Filter Theory – Third Edition”, Prentice Hall, ISBN 0-13- 322760-X, 1996," 1996.
16. M. Praneet Soni "Kalman Filter Based Channel Equalizer: A Literature Review," International Journal of Emerging Technology and Advanced Engineering Website: www.ijetae.com (ISSN 2250-2459, ISO 9001:2008 Certified Journal, Volume 4, Issue 5, May 2014), 2014.
17. M. C. Ms.Manpreet kaur, "Design of an Adaptive Equalizer Using Lms Algorithm," IOSR Journal of Electronics and Communication Engineering (IOSR-JECE) -ISSN: 2278-2834,p- ISSN: 2278-8735.Volume 9, Issue 1, Ver. I (Jan. 2014), PP 25-29, 2014.
18. Daniel S. Baum " "Stanford University, Simulating the SUI Channel Models." IEEE Transactions on Consumer Electronics.," 2001.

Author profile

Alhamzah Taher Mohammed, Assist Lecturer Eng: Received his B.Sc. in Electronic and Communications Department /College of Engineering/ Al-Nahrain University (Baghdad, Iraq), 2004 received the M.Sc. degree in Electronic and Communications Department /college of Engineering/ Al-Nahrain University (Baghdad, Iraq), 2008 , Currently he is Lecturer, Researcher and training supervisor, Dep. of Electronic in Institute of Technology Baghdad