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Construction and Simulation Model of LMS Linear Equalizer Based Simulink

Jabbar Shatti Jahlool
Department of Computer Technique's Engineering, Dijlah University College, Baghdad/Iraq
E-mail: jabbar.shatti@duc.edu.iq

Abstract: *The interference of the transmitted signals together generate dispersion in phase and amplitude, this interference can be compensated by using equalizers. In this review paper, popular types of equalizer will steady as well as building and simulation a model of Least Mean Square (LMS) linear equalizer based Simulink has been achieved with different setting of each components factor. To confirm the stages of the proposed device's work, three checks were presented during different time periods and the equalizer response was plotted to purify the signal from interference. The results of the three tests are documented in three plots as shown in the Figures 8, 9, 10. Three scenarios for testing were presented, first used to plot the shape of the signal before entered to the equalizer, second when running the model and passed short period of time, there'd when a longer period of time passed. The response of the proposed system has been noted through plotting the improvement of the signal.*

Keyword: *Equalizer, Adaptive Equalization, Simulink, Fractionally Spaced Equalizer, Least Mean square*.

1. INTRODUCTION

Overlap between symbols is the interference that occurs when a signal is transmitted during the time dispersal channels. Adaptive filter based on the LMS algorithm has been to focuses, the results indicate that the LMS algorithm convergence performance is ideal, and the expected signal can converge to the input signal. The adaptive filter application includes correcting channel mismatch by an adaptive linear filter. System performance improvement through frequency signal filter by adjustment degree filter and adaptive equalizer [1].

The transmitted during a transmission channel is often subjected to interference; this distortion can be compensating by using adaptive equalizer [2]. There are different types of equalizers such as decision-feedback equalization, fractionally spaced equalizer, linear phase equalizer, blind equalization, T-shaped equalizer and so on [3]. Design and implement of an effective approach adaptive equalizer with LMS algorithm for Inter symbol interference cancelation that is caused by multipath propagation. In other words, to reduce the effects of interference between the symbols in the transmitter and the receiver as soon as possible [4].

Study of investing convergence of three types of signed decision directed LMS equalizer, which are, sign LMS algorithms, sign error LMS algorithm and sign error LMS algorithm [5].

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The principle and application of adaptive filter should be introduced, in order to analyze the adaptive filter based on LMS algorithm, based on the statistical experimental method simulation results are presented according to the principle and structure of LMS algorithm [6-8].

A filter mechanism that automatic modifies its transfer function according to algorithm driven by an error signal is called an adaptive filter. while filters represent a non-adaptive because it's have a static transfer function. most types of adaptive filters are digital because of the complexity of the optimization algorithms. One of the characteristics of the adaptive filter is its use of feedback in the form of an error signal in order to refine its transfer function for the purpose of matching changing parameters. Some applications required adaptive filters because unknown in advance some parameters of the desired processing operation [9].

With the adaptive notch filter, a signal with two different frequencies can be eliminated at the same time [10-12]. Digital identification system has been described and implemented based on real-time processing through adaptive filtering algorithms over the DSK C6713 hardware platform. The evaluation of the work of this system is supported with numerical and graphical results [13].

Innovative detection technology for MIMO systems has been developed. In this advanced technology, through the introduction of constellation constraints performance has been improved. In each time instance, a number of tentative decisions were produced through the estimates of the symbols made by the filter are refined by a scheme that uses multi selected constellation points [14 – 18].

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In the presented review, the simulation model of LMS linear equalizer depend on Simulink has been develop and implemented. To ensure that each sub-block is working properly and that it matches what is needed and planned, each one setting to special parameter. Three checks were performed for different periods of time to ensure the system was operating properly. The tests proved as shown in the Figures 8, 9 and 10 that the output signal improves with the advance of the time period.

2. CONCEPT OF ADAPTATION PROCESS

Adaptive filters are automatic parameter estimation methods that appear in many communication systems and signal processing for applications such as echo cancellation, system identification, noise reduction, channel equalization, the tracking of the velocity and position of motion objects or vehicles e.g., in computer vision, GPS system, beam-forming, space time signal processing in mobile communication video coding and audio coding [19].

Principle of operation of adaptive filters based on estimation hidden parameters or noise signal by reducing the objective error function, usually the mean squared difference (or error) between the target (desired) signal and the filter's output signal. it's used for identification and estimation of non-stationary channels and signals, also for unstable systems or in some of the applications that require low processing delays and/or sample-by-process modulation.

In other words, the principle of action of adaptive linear filters is based on the possibility of extracting the parameters or desired signal from the input through the process of estimation or a filtering [20].

Adaptation of filter parameters depends on the possibility of reducing the object function, often the mean squared error between the (desired) target signal and the filter output.

The following types of filters represent the combination of adaptive filters:

- Scalar-valued input or vector-valued input vectors.
- Single input or multiple-inputs filters.
- Finite impulse response (FIR) or infinite impulse response (IIR) filters.
- Linear or non-linear filters.

The sample adaptive filters namely two types which are recursive formulation of the least square error wiener filter (LMS) and the steepest descent (RLS). Sample adaptive filters have number of advantages compare with the block adaptive filters like, better tracking of the trajectory of non-stationary signals and lower processing delay.

These are essential and important characteristics in some of applications such as noise cancellation, echo cancellation, low-delay predictive coding, adaptive delay estimation, channel equalization in mobile telephony and radar where fast tracking of time-varying processes and time-varying environments and low delay are important objectives [21]. Figure 1, below represents one example of adaptive filter configuration.

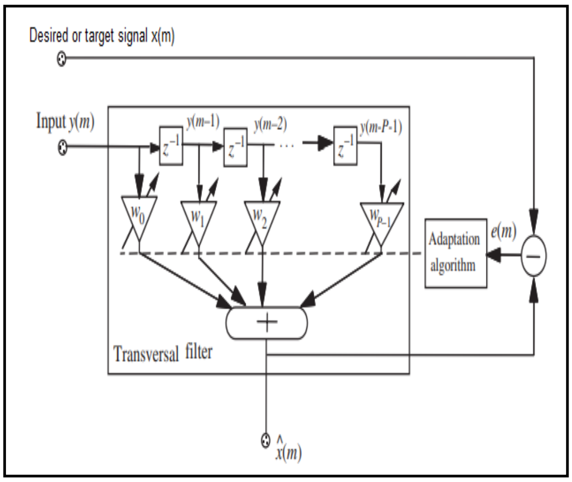

Figure 1 configuration of an adaptive filter

3. EQUALIZATION AND TYPES

Inter-symbol Interference (ISI), it's an interference can caused by a time-dispersive channels. It is one of the forms of distortion on symbols that causes the symbols to overlap so that it becomes difficult to distinguish them at the receiver. One example, in a multipath scattering environment, therefore, late versions of the symbol transmission will be seen by the receiver and these can of course interfere with other symbol transmissions.

An equalizer mission is to try to mitigate ISI and, as a consequence, to improve the performance of the receiver. There are several types of equalizer which are, first the linear equalizers, that is further divided into two categories: Symbol-spaced equalizers (SSE) and fractionally spaced equalizers (FSEs), second, decision-feedback equalizers (DFEs), there'd maximum likelihood sequence estimation (MLSE) equalizers. To make a comparison of two algorithms, the first is the classic LMS algorithm, and the second is the adaptive NLMS algorithm: (Standard LMS), its notice that the second algorithm improves the convergence speed, compared to the first algorithm, and therefore, is more powerful [22-24].

A comparative study of five adaptive filters and four performance criteria are used, which are algorithm execution time, tracking capability, convergence speed and minimum mean square error (MSE). Correlated and unrelated input data in both fixed and non-fixed environments was used to demonstrate comparisons of all algorithms [25]. Some of equalizer denoted as adaptive equalizers like linear and decision-feedback equal-

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izers, this comes from being dependent on adaptive algorithm when operating.

Adaptive filtering techniques are successfully used on communication systems such as channel equalization problems, echo cancellation, smart antennas, spectral estimation for speech analysis and synthesis and interference cancellations and so on [26]. The adaptive algorithm used can be summarized as follows:

- Least mean square (LMS)
- Signed LMS, including these types: sign LMS,
- Signed regress or LMS, and sign-sign LMS
- Normalized LMS
- Variable-step-size LMS
- Recursive least squares (RLS)
- Constant modulus algorithm (CMA)

In order to simulate any type of adaptive equalizer it is important to specify information about the equalizer structure, such as the number of taps, the adaptive algorithm like the step size, and the signal constellation used by the modulator in your model. Also, important point, specify an initial set of weights for the taps of the equalizer, the block adaptively updates and the weights throughout the simulation.

There are many algorithms used for data equalization. According to the criterion which is used to optimize the coefficients of the equalizer these can be categorized in three different classes which are [27]:

- Zero-Forcing (ZF) Equalizer
- Minimum Bit Error Rate (MBER) Equalizer.
- Minimum Mean Square Error (MMSE) Equalizer

Some of equalizer types can be explained in detail.

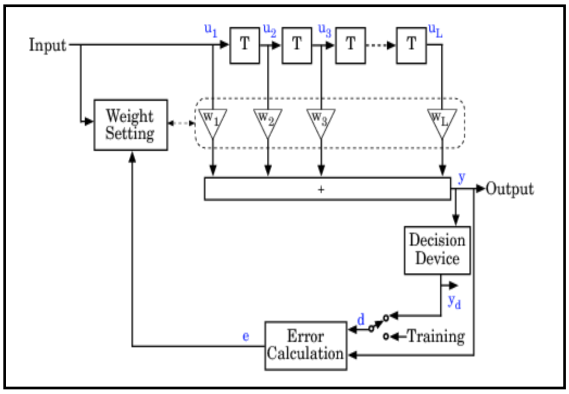
3.1 Symbol Spaced Equalizers (SSE)

This type a linear equalizer consists of a tapped delay line for stores samples from the input signal. Once per symbol period, the SSE outputs a weighted sum of the values in the delay line, as well as updating the weights for the purpose of preparing for the next symbol period. The schematic diagram of the SSE is shown in Figure 2, with N weights; where the symbol period is T and yd represent the detected version of the output signal.

This type of equalizer is called symbol-spaced because both the sample rates of the input and output are equal. Linear stabilizers are simple in structure, but perform in specific channel conditions. In a linear equation, there is no feedback path to adjust the equalizer, so the equation is linear [28, 29].

In practical applications, it is important to that training mode must be run first to gather information about the channel, and later switched to decision-directed mode. The error calculation operation produces a signal denoted as (e) given by the expression below, where R is a constant related to the signal constellation and (CMA) is the constant modulus algorithm.

$$e = \begin{cases} d - y & \text{Algorithms CMA} \\ y\,(R - |y|^2) & \text{CMA} \end{cases}$$



At simulation, The SSE operates in two modes. Training mode or decision directed mode. In training mode, the desired and transmitted symbols sequence are exactly matches i.e., the data has known to the receiver. While in decision-directed mode, the desired symbols are derived from the output of the decision device, this sign is denoted by yd. This process is done by the switch block shown in the Figure 3.

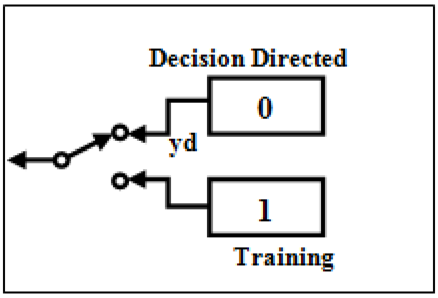


Figure 3 Selector switch between training and decision direct modes

3.2 Decision-Feedback Equalizers (DFE)

This type of equalizer represents a nonlinear equalizer that contains two filters, forward and feedback. The forward filter is similar to the linear equalizer described in symbol spaced equalizers, while the feedback filter contains a tapped delay line whose inputs are the decisions made on the equalized signal. Important feature of this equalizer to cancel ISI while minimizing noise enhancement.

Figure 4, shows the schematic diagram of DFE type that contains a fractionally spaced with L represent for forward weights and N-L feedback weights. The decision feedback equalizer will predict the channel noise level through the noise index based on the previous noise samples. For the purpose of reducing the noise level of the channel, the feedback filter is used for this

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case through subtract predicated noise from the input signal [30].

As explain in the schematic diagram, the forward filter is at the top side and the feedback filter is at the bottom side. If K is 1, the result is a symbol-spaced DFE instead of a fractionally spaced DFE. In each symbol period, the forward filter receives K input samples, while the feedback filter will output one decision or training sample. The DFE equalizer then outputs a weighted sum of the values in the forward and feedback delay lines, and updates the weights to prepare for the next symbol period.

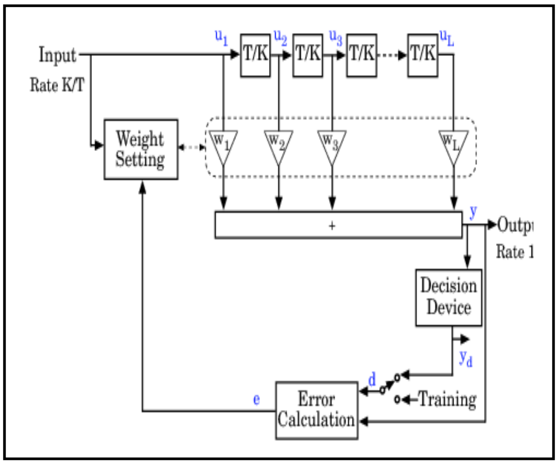

Figure 4 Schematic diagram of DFE

3.3 Adaptive blind equalization

Blind Algorithms, in this type of algorithm exploits the characteristics of the transmitted signals without the need for training sequences.

This type of algorithm has been so named because they provide a close-knife equalizer without overburdening the overhead training transmitter.

These modern algorithms are able to obtain acquire equalization through the techniques of recovering the characteristic of the transmitted signal. In general, even when the initial error rate is massive, the blind equalization technique directs modulus adjustment method towards optimal filter parameters.

A blind equalizer is in a position to compensate the distortions in the amplitude and delay of a communication channel using solely the data of the basic statistical properties of the information symbols and the channel output samples [31].

In blind equalizers there's no training sequence to calculate the tap weight coefficients, this is the important advantage of this type of equalizers, as a result there is no bandwidth is wasted by its transmission. Blind equalization is effective for digital mobile, cable TV, communication systems, a high-speed digital radio, multi-point networks and digital terrestrial TV broadcasting [32,33]. The Figure 5, represent the general block diagram for blind equalizer.

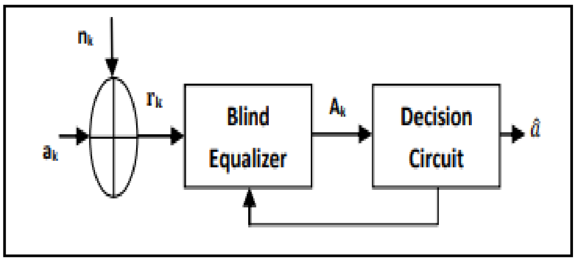

Figure 5 General block diagram of blind equalizer

Combination process adaptive equalization with the communication system is clear shows in figure (). In this system, the linear FIR filter represented by the equalizer G(z,W) with parameter vector W designed for the purpose to remove the distortion caused by channel ISI. The goal of the equalizer shown in the Figure 6, is to generate reliable output signal y(n). This signal can be quantized to get a reliable estimate of the channel input data [34].

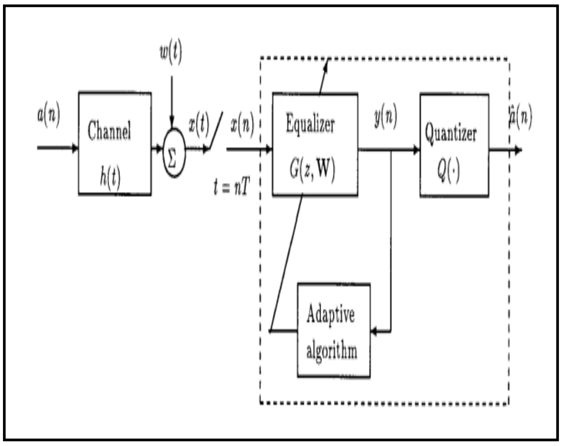

Figure 6 Adaptive blind equalization

4. PROPOSED SIMULATION MODEL OF LMS LINEAR EQUALIZER USING SIMULINK

The proposed model block diagram shown in Figure 4, the model consists of different of sub-blocks each one has a specific parameter must be right setting. The simulation transmits a 16- QAM signal, modeling the channel using a Finite Impulse Response (FIR) filter followed by Additive White Gaussian Noise (AWGN). The equalizer receives the signal from the channel and, as training symbols, a subset of the modulator's output. The equalizer operates in training mode at the beginning of each frame and then switches to decision-

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directed mode. As shown for Figure 7, two scopes are used to contrasts the signals before and after equalization to illustrate the effect of the equalizer.

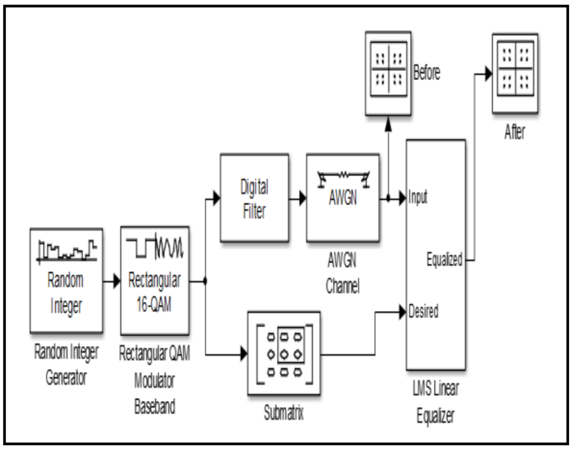

Figure 7 Block diagram of the proposed simulated model of LMS linear equalizer

To build simulated model of this type of equalizer, some of factors must be take on other consideration for configure each block of the model by setting its parameters to required values.

4.1 Random Integer Generator block:

In this block the following setting must be performs

- Set M-any number to 16.
- Set Sample time to 1/1000.
- Select Frame-based outputs.
- Set Samples per frame to 1000.

4.2 Rectangular QAM Modulator Baseband

In this block the important factors must be setting are:

- Set Normalization method to Average power.
- Set Average power to 1.

4.3 Digital Filter

The digital filter block parameters that must be setting are:

- Set Transfer function type to FIR (all zeros).
- Set Filter structure to direct form transposed.
- Set Numerator coefficients to [1 $-$0. 3 0.1 0.2j].

4.4 Sub-matrix block.

This block has the required setting:

- Set ending row to index.
- Set ending row index to 100.

4.5 AWGN Channel

The important factors must be setting at this block are:

- Set mode to signal to noise ratio (SNR).
- Set SNR to 40.

4.6 LMS Linear Equalizer Block.

The important factor must be setting at this block are:

- Set Number of taps to 6.
- Clear the Mode input port, Output error, and output weights check boxes.

5. TEST AND RESULTS

In order to check the proposed simulated model response, two plot scope are used for this purpose, and the connection points of these scope has shown in figure 4. Three test scenarios were presented, first to draw the signal shape before entering the equalizer, and the second when the model is run and the passage of a short period of time, the third when a longer period of time has passed. The result of each test is recorded in a specific plot. When comparing these three plots, the difference in improving the appearance of the signal by diminishing interventions is noted, this is evidence of equalizer work. The details functions of these three tests are explain as follows:

5.1 Behavior the system before equalization.

In this case, it's possible to plot the system behavior to scope display before equalization deviates noticeably from a 16-QAM signal constellation. The result of this case is shown in the Figure 8.

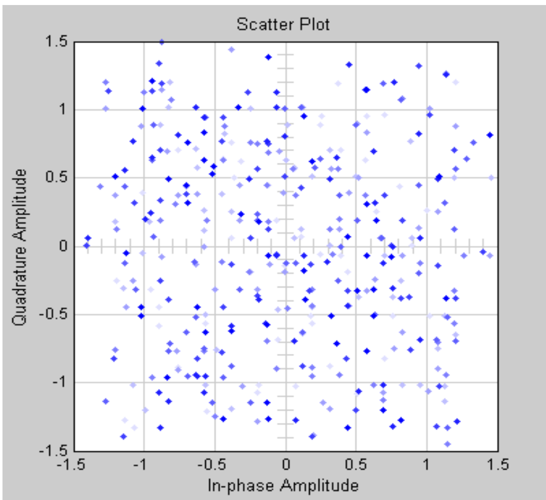

Figure 8 Scope display of signal before equalization

5.2 Signal plot at the beginning of the simulation

At the beginning of the simulation operation, a slight effect of the action of the equalizer on the signal is noted, or in other words the equalizer does not appear to improve the scatter plot. In fact, the equalizer is busy trying to adapt its weights appropriately. Figure 9,

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shows the equalized signal at the beginning of the simulation.

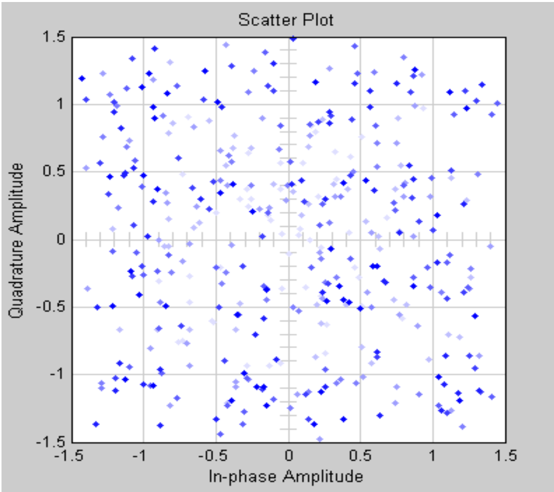

Figure 9 Scope display of the signal at the beginning of simulation

5.3 System behavior after a period of time has passed

This is the steady state of the system working, after a period of time has passed in the simulation, the equalizers weights work well on the received signal. As a result, the signal appears far more equal the constellation of 16-QAM signal is much like the received signal. Figure 10, shows the plot of the behavior of the system at this case.

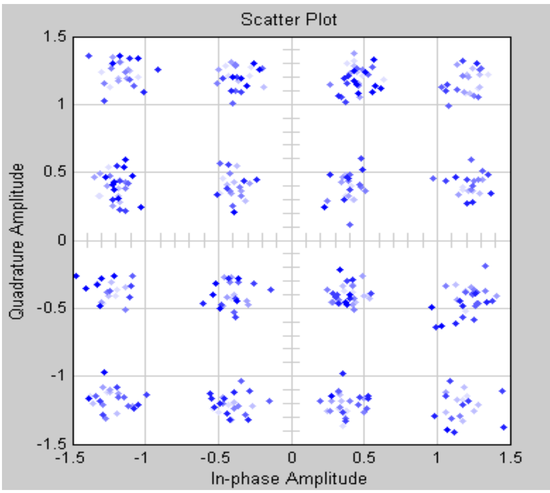

Figure 10 Scope display of signal at the steady state behavior of the system

6. CONCLUSION

During the study of different types of equalizers, it was noted that each type of equalizer has advantages, disadvantages and a different characteristics', all these can be summarized as follows. Regarding the LMS algorithm, it is very fast to implement, but on the other hand it is slowly converging, and its complexity grows linearly with the number of weights. For the RLS algorithm, it converges quickly, but on the other hand, approximately its complexity grows with square weights. Therefore, it can be said that this algorithm is unstable to implement or work with the large number of weights. For the all various types of signed LMS algorithms, it's possible and simplicity to hardware implementation. For the both types of LMS normalized and variable-step-size algorithms, it is more powerful to change the input signal statistics. Concerning the constant modulus algorithm is convenient and useful when there is no training of the signal is available, and its performance is better with constant modulus modulations like PSK. Depending on simulation techniques and programs, such as Matlab, Simulink and others, it became possible to design and implement model simulations for any of the existing type of equalizer. Also it is possible to change the parameters and note the results of operating and performing the model work. Accordingly, in this research a simulation of LMS linear Equalizer model was performed, three tests have been performed for different operating stages and different time periods. It was accurately observed that the output signal for this equalizer improved over time, this is clear by the three output response plots.

REFERENCES

[1] Zhen Zhu and Xiang Gao, Leilei Cao, Daoyuan Pan, Yumeng Cai, Yu Zhu, (2016), "Analysis on the adaptive filter based on LMS algorithm" Elsevier Optik - International Journal for Light and Electron Optics, pp. 4698–4704.

[2] Wang Junfeng and Zhang Bo., (2010), "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, pp. 256-258.

[3] B. P. Lathi, (1998), "Modern Digital and Analog Communications system" Third edition. USA: Oxford University Press.

[4] Chanpreet Kaur and Rajesh Mehra, (2011), "High Speed and Cost-Efficient Adaptive Equalizer for ISI removal" International Journal of Engineering Science and Technology (IJEST), Vol. 3, No. 1, pp. 224 – 256.

[5] Engr. Silas A. Akaneme and Prof. G. N. Onoh, (2015), "Investigating Convergence of Different Types of Signed Decision Directed LMS Equalizer" European Journal of Engineering and Technology, Vol. 3, No. 2, pp. 62 – 69.

[6] I. Dai, P. Kim and De Wilde, (2000), "Performance analysis of the DCT-LMS adaptive altering algorithm" Signal Process, pp. 1629–1654.

[7] D.S. Brito, Enio Aguiar, Fausto Lucena, etal.(2009), "Influence of low frequency noise in adaptive estimation using the LMS algorithm" Signal Process, pp. 933–940.

[8] Lingquan Zeng and Xin Li, (2012), "Research on harmonic suppression in power system based on improved adaptive filter" Energy Procedia, pp. 1479–1486.

[9] Ch.Sridevi and K. Anusha, (2012), "Implementation of MMSE equalizer using variable of LMS" International Journal of Engineering Research & Technology (IJERT) Vol. 1 Issue 7,

ISSN: 2456 - 3935
International Journal of Advances in Computer and Electronics Engineering
Volume 6, Issue 8, August 2021, pp. 1 – 7

[10] Yegui Xiao and Yoshiaki Tadokoro, (1995), "LMS-based notch filter for the estimation of sinusoidal signals in noise" Signal Process, pp. 223–231.
[11] K.C. Ho and P.C. Ching, (1995), "Split filter structures for LMS adaptive filtering" Signal Process, pp. 255–266.
[12] Hideaki Sakai and Yoichi Hinamoto, (2005), "An exact analysis of the LMS algorithm with tonal reference signals in the presence of frequency mismatch" Signal Process, pp. 1255–1262.
[13] Fabián Rolando Jiménez-López and Camilo Ernesto Pardo-Beainy, (2014), "Adaptive filtering implemented over TMS320C6713 DSP platform for system identification" ITECKNE Journal, Vol. 11, No. 2, pp. 157 – 171.
[14] Li, P.; De Lamare and R. C., (2012), "Adaptive decision feedback detection with constellation constraints for MIMO systems" Vehicular Technology, IEEE transactions on, Vol. 61, No. 2, pp. 853-859.
[15] V. Jagan Naveen; K. Murali Krishna and K. Raja Rajeswari, (2010), "Performance analysis of equalization techniques for MIMO systems in wireless communication" International Journal of Smart Home, Vol. 4, No. 4, pp. 47- 63.
[16] Chadokar, S. and Barmashe P., (2011), "Performance Evaluation of Equalizers and Different Diversity Techniques using OFDM" International Journal of Computer Technology and Electronics Engineering (IJCTEE), Vol. 1, No. 3.
[17] Randhawa, N. S. and Khurana, S., (2014), "Error Rate Performance Enhancement Using Hybrid Equalization technique in MIMO-OFDM systems" accepted and presented in IEEE conference, ICHPCA, pp. 22 -24.
[18] Xiaojun Yuan and Junjie Ma, (7–10 April 2013), "Iterative equalization for MIMO systems: Algorithm design and evolution analysis" Wireless Communications and Networking Conference on, IEEE, pp. 3974-3979.
[19] Saeed V and Vaseghi, (2008),"Advanced Digital Signal Processing and Noise Reduction" Fourth Edition, 2008 John Wiley & Sons, Ltd. ISBN: 978-0-470-75406-1.
[20] Liaqat Mahmood, Syed F. Shirazi, Saeeda Naz, Syed H. Shirazi, Muhammad I. Razzak, Arif I. Umar and Syed S. Ashraf, (2015),"Adaptive Filtering Algorithms for Channel Equalization in Wireless Communication" Indian Journal of Science and Technology, Vol 8, Issue 17.
[21] Paulo S.R. Diniz, (2008),"Adaptive Filtering Algorithms and Practical Implementation" Third Edition, Springer Science+Business Media, LLC.
[22] H. Zhao, S. Hu, L. Li and X. Wan, (2013), "NLMS adaptive FIR filter design method" presented at IEEE Region 10 Conference TENCON, pp. 1- 5.
[23] C. Paleologu, J. Benesty, S. L. Grant and C. Osterwise,(2009),"Variable step-size NLMS algorithms for echo cancellation" presented at Conf. Record of the forty-third Asilomar Conference on Signals, Systems and Computers, pp. 633-637.
[24] X. Guan, X. Chen and G. Wu, (2009), "QX-LMS ADAPTIVE FIR filters for system identification" presented. At 2nd International Congress on Image and Signal Processing, CISP '09, Vol. 1, No. 1, pp.1–5.
[25] Sandeep Agrawal and Dr. M. Venu Gopal Rao, (2012), "Performance Analysis of Adaptive Filtering Algorithms for System Identification" International Journal of Electronics and Communication Engineering, Vol. 5, No. 2, pp. 207 – 217.
[26] Simon O. Haykin, (2013),"Adaptive Filter Theory", 5th ed., Pearson Education: Prentice Hall.
[27] Z. Ding and Y. G. Li, (2001), "Blind Equalization and Identification", ser. Signal Processing and Communication Series. Marcel Dekker, Inc.
[28] Navdeep Singh Randhawa and Shally Sharma, (2015), "A Survey of Equalization Techniques for an Effective Equalizer Design in MIMO-OFDM Systems" International Conference on Circuit, Power and Computing Technologies, IEEE, [ICCPCT.
[29] Simeone, O.; Bar-Ness, Y. and Spagnolini, U., (2004), "linear and nonlinear preequalization/equalization for MIMO systems with long-term channel state information at the transmitter" Wireless Communications, IEEE Transactions on, Vol. 3, No. 2, pp. 373 – 378.
[30] Heyi Sun, (2017),"Spectrum Access System: Design and Implementation of the Decision-Feedback Equalizer in Software" Masters of engineering, Electrical Engineering and Computer Sciences University of California at Berkeley, USA, pp. 1-59.
[31] K Suthendran and T Arivoli, (2014), "Performance Comparison of Blind Equalization Algorithms for Wireless Communication" International Journal of Computer Applications, Vol. 85, No. 13, pp. 1-6.
[32] Kil Nam Oh and Jae Hong Park, (1997), "Property Restoral Approach to Blind Equalization of Digital Transmission Channels", IEEE Transactions on Consumer Electronics, Vol. 43, No. 3.
[33] K.Suthendran and T. Arivoli, (2013), "Performance comparison of adaptive and blind equalization algorithms for wireless communication," Bonfring International journal of research in communication engineering, vol 3, pp. 1-6.
[34] Charu Sharma and, Kapil Gupta, (2012), "Performance Analysis of Adaptive Blind Equalizers Algorithms" International Journal of Electronics Communication and Computer Technology (IJECCT), Vol. 2, Issue 3.

Authors Biography

Jabbar Shatti Jahlool, was born in 1964 Iraq. In 1988, he obtained a Bachelor's degree in Electrical and Electronic Engineering from the University of Technology /Iraq. he obtained a master's degree in electronics in 2001 from Iraq University of Technology.
From 1988 to 2003, he was a assite researcher, a preliminary designer, electronic hardware designer, head of the department of programmable logic design center and director of the technology transfer and scientific research of ministry of military industrial. From 2003 to 2014, he was a director of the technology transfer and scientific research at the ministry of industry.
Since 2015 he has been a university lecturer with department of computer techniques engineering, Dijlah university college, Baghdad, Iraq.. His research interests include hardware electrical and electronic circuits design, VHDL and FPGA design, microcontrollers research and design project, power system, design, anaylsis and control, and power electronics circuits designer.

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