ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(A High Impact Factor, Monthly, Peer Reviewed Journal)

Website: www.ijareeie.com

Vol. 6, Issue 11, November 2017

A Modified Algorithm with Variable Step Size for Blind Equalization of QAM Signals

Mohammad .S. T. Badran

Associate Professor, Department of Electronics & Communication, Al-Obour Academy for Engineering, Egypt

ABSTRACT: Adaptive channel equalization accomplished without a training sequence is known as blind equalization. A Modified Algorithm with a variable Step Size (MA_VSS) Control Technique can be employed MA which uses a combination of DDA and GA in addition to VSS. MA_VSS has been proposed as a solution to the problem of slow convergence of blind equalization algorithms and resolves the conflict between the convergence rate and low steady state error of the fixed step-size conventional blind equalization algorithms, such as Constant Modulus Algorithm. The proposed technique has been compared with the popular blind equalizers, the decision directed algorithm (DDA), Godard algorithm (GA), Sato algorithm (SA), Benveniste and Goursat algorithm (BGA), and the stop–and-go algorithm (SNGA).Computer simulations have been performed to verify the performance of the proposed method dominates both of the linear equalizer and that with decision feedback and single prediction error algorithm for symmetric constellation QAM.

KEYWORDS: Cross correlation, MA _ VSS, Constant Modulus Algorithm, blind equalization.

I.INTRODUCTION

Inter symbol interference is one of the greatest impediments of high data rate digital communication systems. In order to overcome the effects of the impairment, several channel estimation and equalization methods have been developed in the last few decades. One of the best ways to cancel the effects is to use an equalizer which minimizes the ISI while combining the multi path energy [1]- [3]. In practice, Linear Equalizers (LE) and Decision Feedback Equalizers (DFE) are the most common structures used [4], [5]. But, in suppressing the ISI, the LE inevitably enhances the channel noise. This basic limitation of a LE's ability to cope with severe ISI has motivated a considerable amount of research into suboptimal nonlinear equalizers with low computational complexity such as the DFE.

In order to achieve high speed reliable communications, channel identification and equalization are necessary to overcome the effects of ISI. Conventionally, channel equalizers are of two types: as blind and non-blind. The non-blind channel equalizers waste bandwidth by their dependence on a training sequence. On the other hand, blind channel equalization is one of the most important process during which an unknown input data sequence is recovered from the output signal of an unknown channel.

The blind channel equalizers do not require any training sequence. Instead, the statistical properties of the transmitted signals are exploited to carry out the equalization at the receiver without access to the transmitted symbols. Hence, they are capable of saving valuable bandwidth that is wasted by sending training sequence. The popular constant modulus algorithm (CMA) proposed by Sato [6] in 1975 and the famous least mean squares (LMS) algorithm proposed by Widrow [7] in1966 are widely employed in communications such as blind and non-blind channel equalization and identification for their low computational complexity and simple structure. However, due to using fixed

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(A High Impact Factor, Monthly, Peer Reviewed Journal)

Website: www.ijareeie.com

Vol. 6, Issue 11, November 2017

step size, the CMA and LMS algorithms suffer from a conflict between convergence rate and steady-state error. A larger step size can speed up the convergence rate, but at the same time it increases the steady-state error. A smaller step size can decrease the steady- state error, but the convergence rate will be poor. Thus, with the help of proposed technique the performance of the conventional CMA and LMS algorithms have become comparable to other blind adaptive MA_VSS and Conventional Blind Equalization.

Simulation results have shown that the proposed MA_VSS algorithm perform better than the conventional blind equalizers. The rest of the paper is organized as follows: The following section summarizes the Conventional Blind Equalization. Section 3 explains the proposed algorithm for blind equalization. Computer Simulation results obtained in Section 4, and finally, the paper is concluded in Section 5.

II.CONVENTIONAL BLIND EQUALIZATION

The sampled input to the equalizer x(n) as shown in Fig.1, it may be expressed as: -

$$x(n) = \sum_{k=-L_1}^{L_2} h_k(n)a(n-k) + \eta(n) \qquad (1)$$

Where a(n) is the data symbol that was transmitted assumed to be independent and identically distributed (IID) quadrature amplitude modulated (QAM) source and $\eta(n)$ is zero mean white Gaussian noise independent of a(n). The channel h (n) is modelled as a complex finite impulse response filter with an order L_1+L_2+1.

$$y_o(n) = \sum_{k=-M_1}^{M_2} w_k(n)x(n-k) \qquad (2)$$

The equalizer taps are updated, using the adaptation algorithm for the linear equalizer (LE) as the following form:

$$w(n+1) = w(n) + \mu_f e(n) X^*(n) \qquad (3)$$

Where

$$w(n) = [w_{-M1}(n), \ldots\ldots\ldots, w_{M2}(n)]^T$$

is the coefficient vector of the $(M_1+M_2+1)^{th}$ order of LE , X^* (n) is the corresponding input vector , e(n) is the error signal [11] and μ_f is the step size parameter given by [11] :

$$\mu_f = .001/E[|a(n)|^4] \qquad (4)$$

The adaptation of the equalizer taps is carried out by minimizing the mean square value (MSE) of the difference between the equalizer output and the slicer output. When compared to conventional equalizers which employ the least mean square algorithm LMS to update their taps, blind

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(A High Impact Factor, Monthly, Peer Reviewed Journal)

Website: www.ijareeie.com

Vol. 6, Issue 11, November 2017

equalization algorithms converge very slowly.

The decision directed algorithm (DDA), Godard algorithm (GA), Sato algorithm (SA), Benveniste and Goursat algorithm (BGA), are examples of Blind equalization techniques. The error signal for each algorithm is defined as in [12]-[17]: -

$$e_{DDA}(n) = y_O(n) - \hat{a}(n) \quad (5)$$

$$e_{SA}(n) = [\, y_{Or}(n) - \mu_f \ sgn\,(y_{Or}(n))\,] + j\,[y_{Oi}(n) - \mu_f \ sgn\,(y_{Oi}(n))\,] \quad (6)$$

$$e_{GA}(n) = [\, |y_O(n)|^2 - R \,]\ y_O(n) \quad (7)$$

$$e_{BGA}(n) = K_1\,(e_{DDA}(n)) + K_2\ |e_{DDA}(n))|\,(e_{SA}(n)) \quad (8)$$

$$e_{SNGA}(n) = \begin{cases} e_{DDA}(n) & A=1,B=1 \\ \text{Real}\,(e_{DDA}(n)) & A=1,B=0 \\ \text{Img}\,(e_{DDA}(n)) & A=0,B=1 \\ 0 & \text{otherwise} \end{cases} \quad (9)$$

where $\hat{a}(n)$ is the slicer output , $y_{or}(n)$ and $y_{oi}(n)$ are the real and imaginary parts of y_o (n) respectively, K_1 and K_2 are constants, R is the modulus defined in [12] and A,B are parameters defined as :

$$A = \begin{cases} 1 & \text{if sign}\{\text{Real}\,(e_{DDA}(n))\} = \text{sign}\{\text{Real}\,(e_{SA}(n))\} \\ 0 & \text{otherwise} \end{cases}$$

$$B = \begin{cases} 1 & \text{if sign}\ \{\text{Img}\,(e_{DDA}(n))\} = \text{sign}\ \{\text{Img}\,(e_{SA}(n))\} \\ 0 & \text{otherwise} \end{cases} \quad (10)$$

III. THE PROPOSED ALGORITHM MA_VSS

The Conventional Blind Equalizers exhibit very slow convergence rates when compared to algorithms employed in conventional data equalization schemes. So, we introduce the proposed algorithm for blind equalization (MA_VSS) is two algorithms in one algorithm, the first algorithm called a modified algorithm MA [18] uses a combination of DDA and GA depending on the absolute value of the error signal for each algorithm, and the second is a variable step size algorithm VSS [19] which based on cross correlation of channel output and error signal as shown as in figure 2.

In figure 2, blind equalization algorithm MA_VSS box has two modules MA and VSS, MA module

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(A High Impact Factor, Monthly, Peer Reviewed Journal)

Website: www.ijareeie.com

Vol. 6, Issue 11, November 2017

has two techniques one decision device technique to decide any mode (DAA/GA) will be use, the second is error calculation technique to calculate the error value for the decided mode, e_{MA} (n) , and the other module is variable step size (VSS) to determine the suitable step size will be use.

The MA operates in two modes, DD mode and GA mode [18], The equalizer taps are updated, using the error signal for MA according to equation (11) as defined below:

$$e_{MA}(n) = \begin{cases} e_{DDA}(n) \text{ if } |e_{DDA}(n)| \geq |e_{GA}(n)| \\ \text{DD mode} \\ \\ e_{GA}(n) \text{ if } |e_{DDA}(n)| < |e_{GA}(n)| \\ \text{GA mode} \end{cases} \quad (11)$$

where | . | is the absolute value

The w(n) box is the adaptation algorithm for the proposed equalizer as defined below:

$$w(n+1) = w(n) + \mu f(n)\, e_{MA}(n) X^{*}(n) \quad (12)$$

Variable Step Size (VSS) for constant modulus algorithms VSS-CMA have been used extensively in blind adaptive filtering to improve the performance of the fixed step size conventional CMA. The VSS-LMS algorithms are explained in detail in [9], [10].VSS-CMA [8] uses the autocorrelation be $\hat{e}(n)$ and $\hat{e}(n-1)$, parameter. In this way, the algorithm can effectively maintain a reasonable immunity to uncorrelated additive noise. To update the variable step size, Xiong's approach [8] considers the square of the error signal autocorrelation estimate obtained through a low- pass filter given by:-

$$c(n)= \alpha\, c(n-1)+(1-\alpha)\, \hat{e}(n)\, \hat{e}(n-1) \quad (13)$$

$$c(n)= \alpha\, c(n-1)+(1-\alpha)\, e_{MA}(n)\, e_{MA}(n-1) \quad (14)$$

Where
c(n) is the estimate value of autocorrelation of error signal and α is positive control parameter.
The setting of the step size parameter is

For VSS-CMA : $\mu(n)= \beta\, c2(n)$ (15)

For MA_VSS: $\mu f(n)= \beta\, c2(n)$ (16)

where β is a scale factor used for controlling the bounds of the step size $\mu f(n)$ .The block diagram of the MA_VSS algorithm based on cross correlation between channel output x(n) and error signal

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(A High Impact Factor, Monthly, Peer Reviewed Journal)

Website: www.ijareeie.com

Vol. 6, Issue 11, November 2017

e_{MA} (n) is given in The proposed method, considers the cross correlation function between the channel output and error signal, improving the convergence speed and performance. Moreover, the proposed method provides both noise and ISI immunity since the channel output signal includes both ISI and noise information.

IV. COMPUTER SIMULATION RESULTS

In this section, simulation results are illustrated to verify the performance of the proposed MA_VSS. The simulation studies have composed of the studies is performed using the blind channel equalization. The simulation studies of the blind channel equalizers are performed via 1000 Monte Carlo type iterations using the QAM modulation. The step size parameters were limited to 0.1and 5E-5 for conventional LMS algorithm. Positive control parameters, β was equal to 0.995, α was equal to 0.822 for the proposed VSS-CMA. The proposed algorithm MA was applied to a 16- QAM system. The parameters used in the simulation are:-

Channel no.1:
$h_1(\ell)$ = (0.1632+j0.2056), (-0.9491+j0.1524),(1),(0.2393-j0.0077),

Channel no.2:
$h_2(\ell)$ = (-0.9491+j0.1524), (0.1632+j0.2056), (1),(-j0.0077+0.2393),for ℓ= -2,-1,0,1, and M_1=M_2=N=15.

Fig.1 depicts a simplified model for the equalizer.

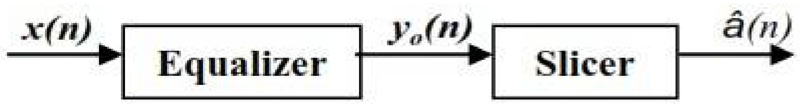

Fig.1. Simplified equalizer

Figure 2, shows a blind equalization algorithm MA_VSS

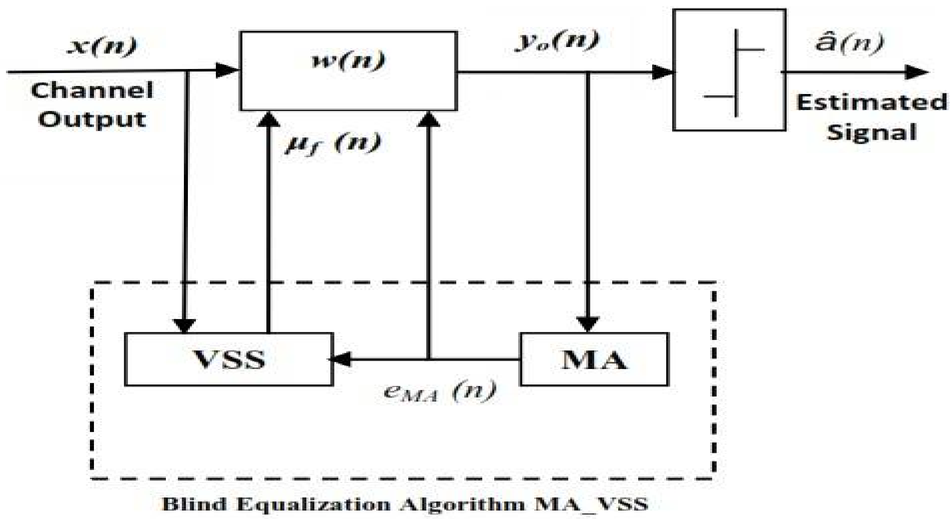

Fig.2. the proposed algorithm MA_VSS

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(A High Impact Factor, Monthly, Peer Reviewed Journal)

Website: www.ijareeie.com

Vol. 6, Issue 11, November 2017

Convergence curves obtained by simulation are depicted in figures 3 - 7 .We can conclude that the proposed MA_VSS converges faster than other equalizers under consideration. Furthermore, it has the minimum obtainable MSE. Further improvements are achieved as shown in figures when used the proposed algorithm.

Fig. 3 shows that convergence characteristics for the MA and the conventional equalizer for fixed step size,

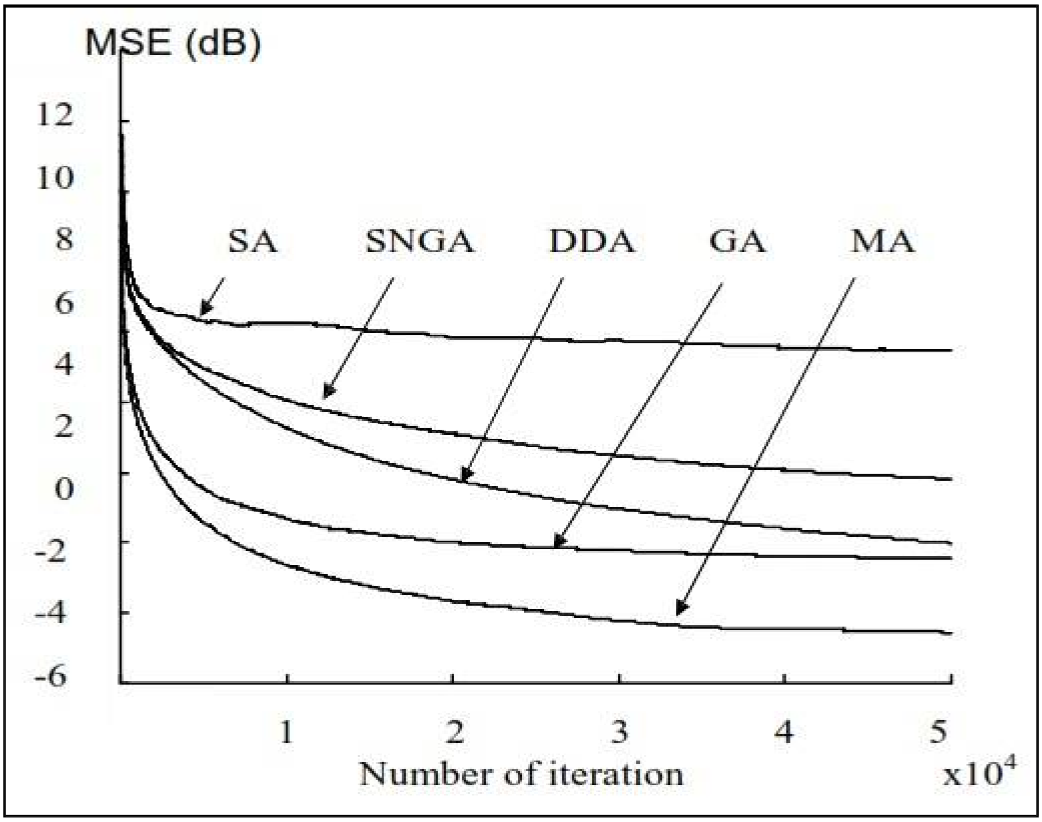

Fig.3. convergence characteristics for the MA and the conventional equalizer

Fig. 4 and Fig. 5 show that Comparison between GA and MA for two different channels

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(A High Impact Factor, Monthly, Peer Reviewed Journal)

Website: www.ijareeie.com

Vol. 6, Issue 11, November 2017

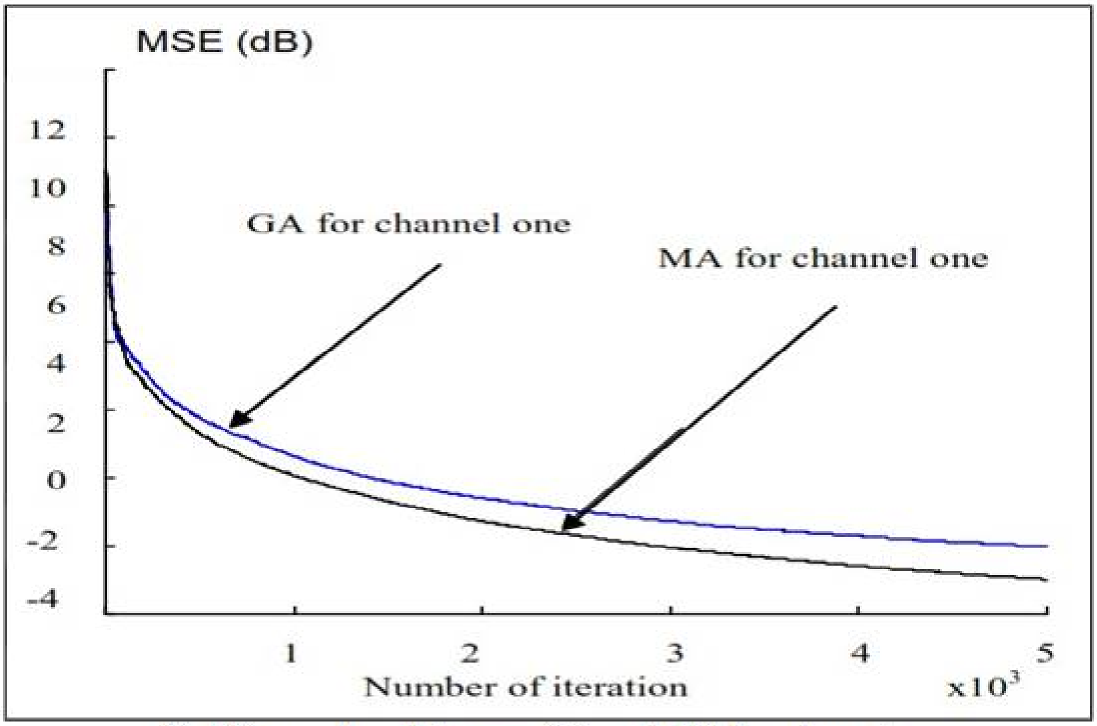

Fig.4. Comparison between GA and MA For channel one

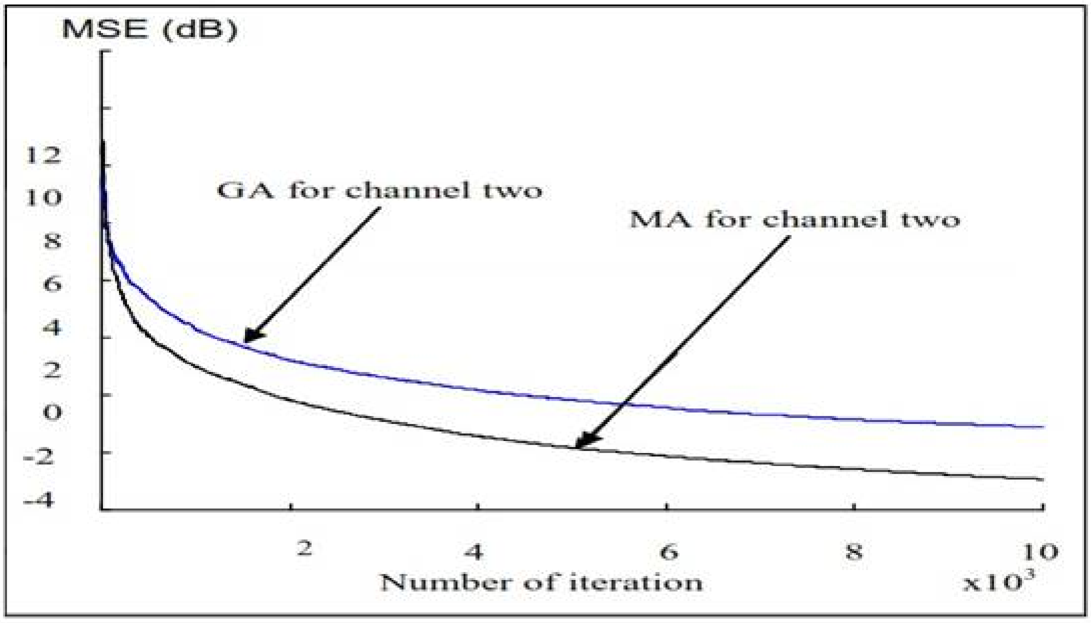

Fig.5. Comparison between GA and MA For channel two

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(A High Impact Factor, Monthly, Peer Reviewed Journal)

Website: www.ijareeie.com

Vol. 6, Issue 11, November 2017

Fig. 6 shows that Effect of Increasing Step Size (FSS) on MA algorithm

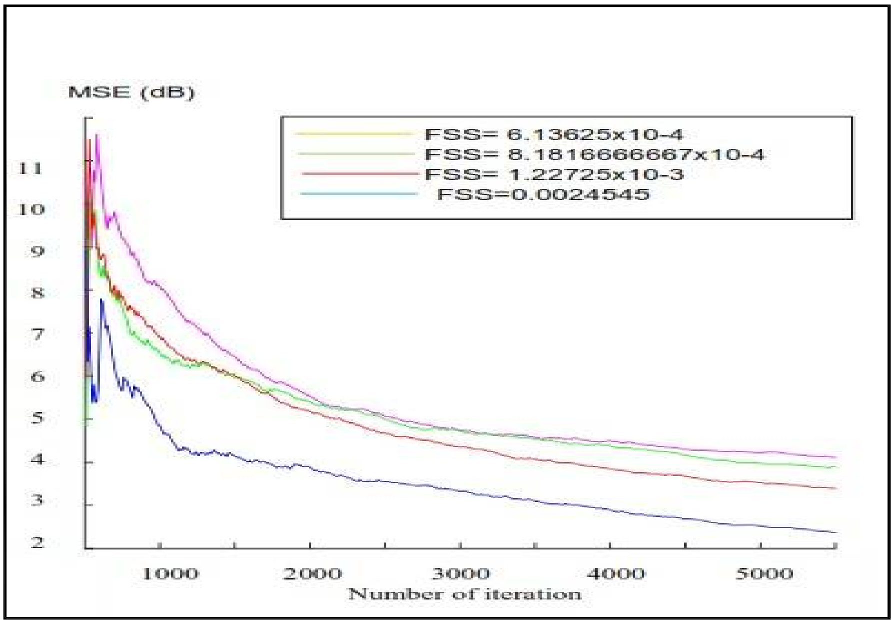

Fig.6. Effect of Increasing Step Size (FSS) on MA algorithm

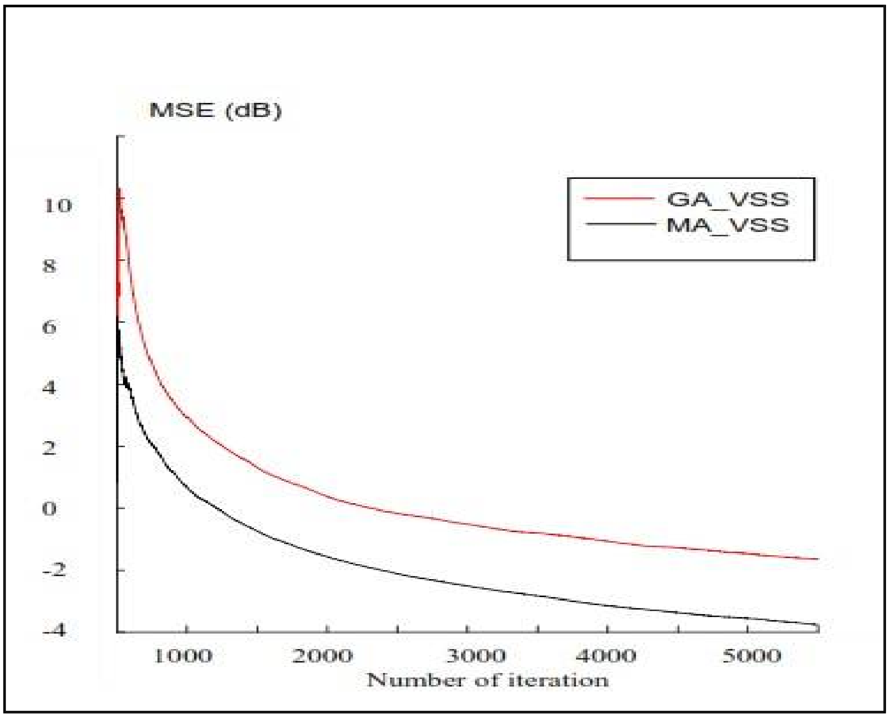

Fig.7. The proposed algorithm MA_VSS for variable Step Size (VSS) parameter and

ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(A High Impact Factor, Monthly, Peer Reviewed Journal)

Website: www.ijareeie.com

Vol. 6, Issue 11, November 2017

Comparison with CMA (GA_VSS).

Comparison with GA_VSS

Fig. 7 shows that the proposed algorithm MA_VSS for variable Step Size (VSS) parameter and

VI.CONCLUSION

In this paper, a new MA_VSS blind equalizer use two techniques one uses a MA algorithm which combination of DDA and GA depending on the absolute value of the error signal for each algorithm and other technique which based on cross correlation of channel output and error signal and has been proposed as a solution to the problem of slow convergence of the fixed step size conventional CMA (GA) blind and conventional equalizers which employ the least mean square algorithm LMS. Thus, the conflict is removed between the convergence rate and low steady state error of the fixed step-size conventional CMA and LMS algorithm. It has been shown that a combination of conventional CMA and LMS with the proposed MA_VSS technique provides an effective and robust way for adaptive blind and non- blind equalization. Also dem onstrated to be very suitable for high speed blind channel tracking

REFERENCES

[1] QURESHI, S. U. H. Adaptive equalization. In Proceedings of the IEEE, September 1985, vol. 73, no. 9, p. 1349-1387.
[2] BELFIORE, C. A., PARK, J. H. Decision feedback equalization. In Proceedings of the IEEE, August 1979, vol. 67, no. 8, p. 1143 to 1156.
[3] BOYD, R. T., MONDS, F. C. Equalizer for digital communication. IEE Electronic Letters, Januar y 1971,vol. 7, no. 2, p. 58-60.
[4] HAYKIN, S. Communication Systems. Third edition. John Wiley and Sons, 1994.
[5] PROAKIS, J. G. Digital Communications.Fourth edition. Singapore: McGraw-Hill Co.,2001.
[6] SATO, Y. A method of self-recovering Equalization for multilevel amplitude modulation. IEEE Transactions on Communications, June 1975, vol. COM-23, p. 679-682.
[7] WIDROW, B. Adaptive filters I: Fundamentals. Stanford Elec- tronic Laboratories Technical Report 6764-6, December 1966.
[8] XIONG, Z., LINSHENG, L., DONFENG, Z., ZENGSHOU, D. A new adaptive step-size blind thEqualization algorithm based on autocorrelation of error signal. In 7Processing, 2004, vol. 2, p. 1719-1722.
International Conference on Signal
[9] ZIPF, J. G. F., TOBIAS, O. J., SEARA R. A VSSLMS algorithm based on error autocorrelation. In16th European Signal Process- ing Conference (EUSIPCO 2008). Lausanne (Switzerland), August 2008, p. 25-29. [10] OZEN, A. A novel variable step size adjustment method based on channel output autocorrelation forthe LMS training algorithm. International Journal of Communication Systems (Wiley-Black- well), 2011, vol. 24, no. 7, p. 938-949.
[11] Vijitha Weerack ady and Saleem A. Kassam. "Dual-Mode Type Algorithm for BlindEqualization ", IEEE, Trans., Comm., 1994, vol. 42, No.1. pp. 22-28.
[12] Godard D.N. " Self recovering equalization and carrier tracking in two dimension data communication system", IEEE, Trans., comm.1980, vol. 28, No.11.pp.1867- 1875
[13] Y. Sato. " A method of self recovering equalization for multilevel amplitude modulation system", IEEE, Trans., comm. 1975, vol. 23, No.6. pp. 679-682
[14] A. Benveniste and M.Goursat," Blind equalizers," IEEE, Trans., Comm., Aug., 1984, vol. 32,No.8.pp,871-883.
[15] Lee M. Garth,Jian Yang,and Jean-Jacques Werner. " Blind Equalization Algorithm for Dual-Mode CAP- QAM Reception", IEEE, Trans.,comm. March 2001. vol. 49, No.3. pp.455-466.
[16] Mohammad .S. T. Badran. "Three Developed Adaptive Blind Equalizers" PhD Thesis,Cairo University,Faculty of Engineering, 2004.
[17] B.S. Seo, J.H.Lee and C.W.Lee. " Approach To Blind Decision FeedbackEqualization" Electronic letters 29th August 1996. vol. 32 No.18.pp1639-1640
[18] MOHAMMED .S. T. BADRAN, "A MODIFIED ALGORITHM FOR BLIND EQUALIZATION," AL-AzaherEngineering Twelfth International conference , Decem ber 25-27,2012.
[19] A New Step Size Control Technique for Blind and Non-Blind Equalization Algorithms Emin TUĞCU, FatihÇAKIR, Ali OZEN, RADIOENGINEERING, VOL.22, NO. 1, APRIL 2013