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A New Blind Equalization Algorithm for Improving the Channel Equalized Performance

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Abstract
Adaptive channel equalization accomplished without a training sequence is known as blind equalization. The decision directed algorithm (DDA), Godard algorithm (GA), Sato algorithm (SA), Benveniste and Goursat algorithm (BGA), and the stop–and-go algorithm (SNGA), are examples of Blind equalization techniques. These algorithms exhibit very slow convergence rates when compared to algorithms employed in conventional data equalization schemes. In order to speed up their convergence process, a modified algorithm (MA) can be employed which uses a combination of DDA and GA. The modified algorithm has demonstrated their effectiveness compared to other conventional techniques especially in the noisy environment.

Keyword: LE, DFE, blind equalization, adaptive equalization algorithms.

Introduction
An equalizer is necessary when the inter-symbol interference (ISI) component of a channel output becomes large relative to the signal component and causes a high symbol error rate (SER). Some equalizers filter the channel outputs directly in order to present the decision slicer with a reduced ISI estimated symbol sequence. Other synthesize an output sequence using a channel estimate and the knowledge of the symbol alphabet, and search for the closest match to the actual channel output sequence.

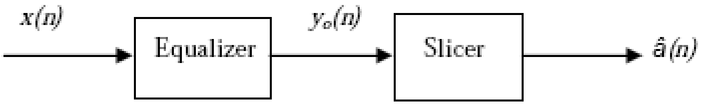

Fig.1. Simplified equalizer.

Fig.1 depicts a simplified model for the equalizer. The sampled input to the equalizer *x(n)* 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 η ***(n)*** is zero mean white Gaussian noise independent of ***a(n)***. The channel ***h(n)*** is modeled as a complex finite impulse response filter with an order L1+L2+1. The tap weight vector of the equalizer at the nth sampling instant is w_k ***(n)*** , the output of the equalizer is given by :-

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

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, $w(n) = [w_{-M1}(n),.........,w_{M2}(n)]$ 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 [1] and μ_f is the step size parameter given by :

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

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 equalization algorithms converge very slowly. In order to speed up the convergence process, the modified algorithm (MA) is introduced. Section II introduces the conventional blind equalization techniques. Section III explains the proposed algorithm (MA). Section IV employs the MA with decision feedback and single prediction error algorithm (MA-PA) for symmetric constellation QAM. Section V, provides simulation results to demonstrate the effectiveness of the MA technique. The concluding remarks are summarized in section VI.

Conventional Blind Equalization

In the following the five main blind equalization algorithms are introduced, the error signal for each algorithm is defined as in [2]-[5]: -

$$\text{DDA} : e_{DDA}(n) = y_o(n) - \hat{a}(n) \quad (5)$$

$$\text{SA} : e_{SA}(n) = [\, y_{or}(n) - \mu_f \operatorname{sgn}(y_{or}(n)) \,] + j\,[y_{oi}(n) - \mu_f \operatorname{sgn}(y_{oi}(n)) \,] \quad (6)$$

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

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

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

where *â(n)* is the slicer output , *yor(n)* and *yoi(n)* are the real and imaginary parts of *yo(n)* respectively, K1 and K2 are constants, R is the modulus defined in [2] and A,B are parameters defined as :

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

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

A Modified Algorithm for Blind Equalization (MA)

The proposed algorithm (MA) uses a combination of DDA and GA depending on the absolute value of the error signal for each algorithm, as shown in fig.2.

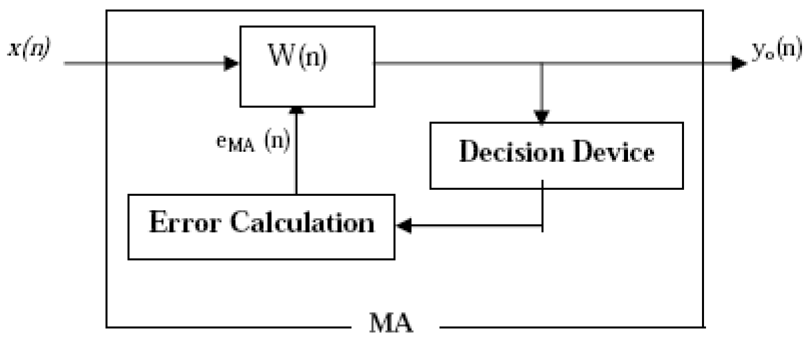

Fig. 2. The proposed MA equalizer

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

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

where $|\,.\,|$ is the absolute value

MA with single prediction error algorithm MA-PA

A decision feedback equalizer (DFE) can be divided into a linear equalizer (LE) and a prediction error filter (PEF) with feedback, which is updated by using MA and the prediction error algorithm (PA) respectively, as in fig (3),[6][7]. This technique can be called MA-PA.

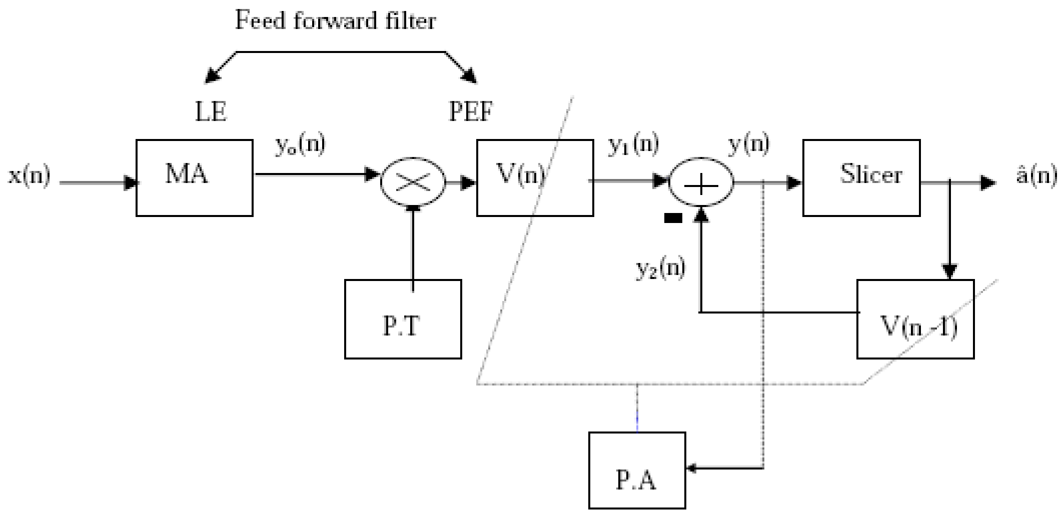

Fig.3. Structure of MA-PA equalizer

(P.T is the phase tracking) and (P.A is the prediction algorithm)

Let y1 (n) , y2 (n) and y(n) are the outputs of (PEF), the feedback filter and the overall DFE, respectively, then

$$y_1(n) = \sum_{k=0}^{N} v_k(n)\, y_o(n-k) \qquad (12)$$

$$y_2(n) = \sum_{k=1}^{N} v_k(n)\, \hat{a}(n-k) \qquad (13)$$

$$y(n) = y_1(n) - y_2(n) \qquad (14)$$

The adaptation algorithm for the feedback filter can be written in the following form:-

$$v'(n+1) = v'(n) + \mu_b\, y(n)\hat{A}^*(n) \qquad (15)$$

where $v'(n) = [\, v_1(n), \ldots\ldots\ldots\ldots, v_N(n)]^T$ $v'(0)=0$, is the coefficient vector of the N^{th} order feedback filter (FBF), $\hat{A}^*(n)$ is the corresponding input vector and μ_b is the step size given by :

$$\mu_b = .0002 / E[|\, a(n)\,|^2] \qquad (16)$$

Note that, the phase error of the carrier is not considered in this algorithm.

Computer Simulation Results

Matlab is used for the simulation to verify the performance of MA equalizer. The proposed system is applied to of 16 Rectangular QAM modulation technique with additive white Gaussian noise. The performance of the MA system is obtained via simulation for the following two channels which are given below:-

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, M1=M2=N=15, and SNR = 20 dB.

Convergence curves obtained by simulation are depicted in figures 4 – 9. In figure (4), we notice that the performance of MA is better than all the conventional equalizers as DDA, SA, BGA, SNGA, and GA. Figures 5, 6 give comparisons between GA and MA equalizers for different channels 1, 2. Figures 7, 8 show the output signal constellation diagrams after the convergence of the GA and MA equalizers. It can be seen, the constellation diagram of MA is the most compact which is due to a smaller residual error after the convergence of the algorithm. So the convergence precision of the MA is the highest after equalization. Furthermore, it has the minimum obtainable MSE. Further improvements are achieved as shown in fig.9 by the employment of MA – PA which dominates GA-PA. We can conclude that the proposed MA converges faster than other equalizers under consideration.

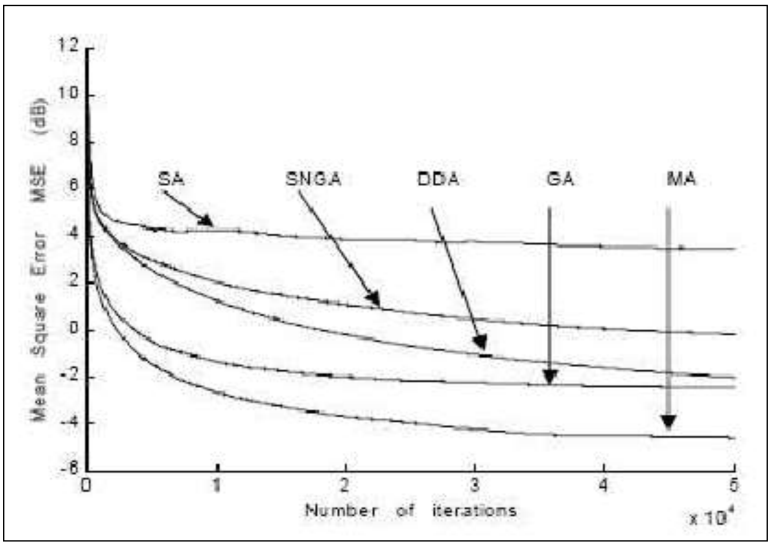

Fig. 4. Convergence characteristics for the MA equalizer and the conventional equalizers

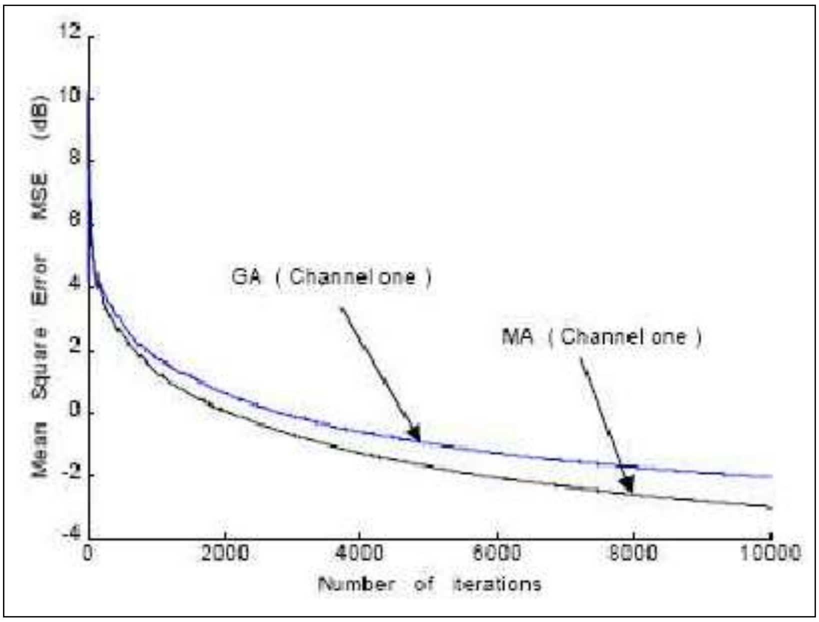

Fig. 5. Comparison between GA and MA equalizers for channel no.1

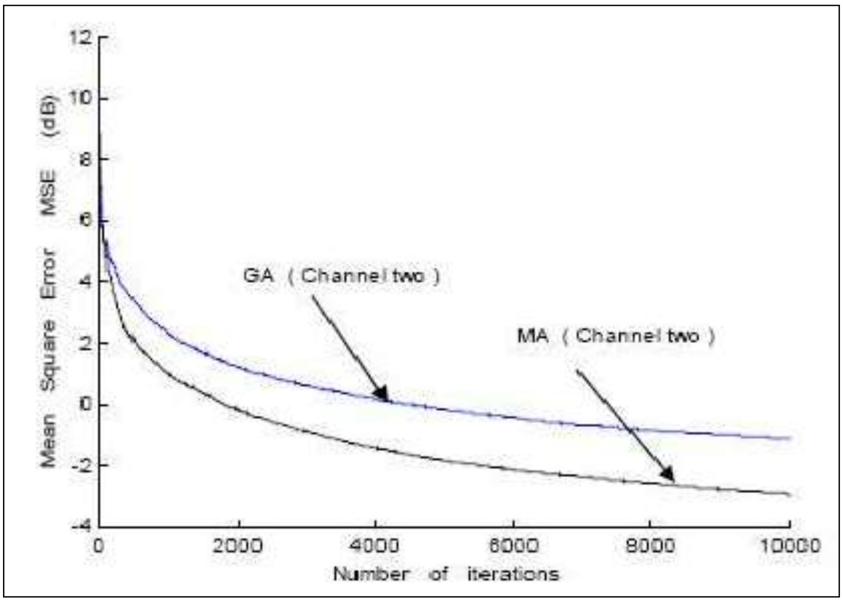

Fig. 6. Comparison between GA and MA equalizers for channel no.2

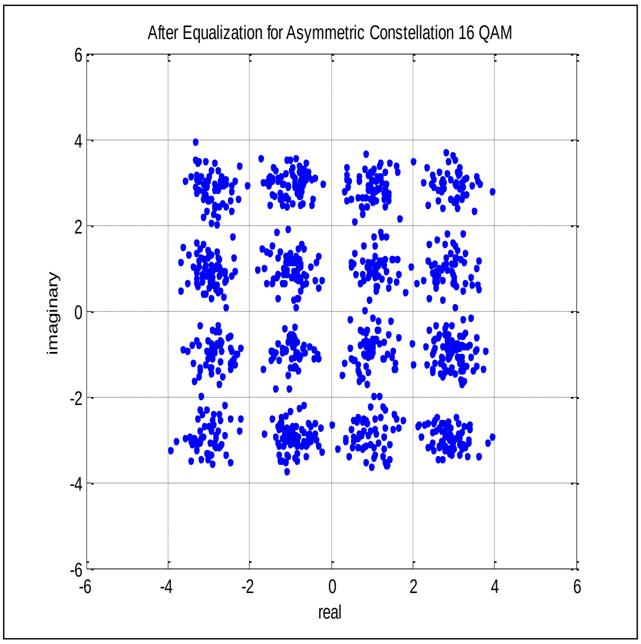

Fig.7 The output constellation of GA of 16 QAM

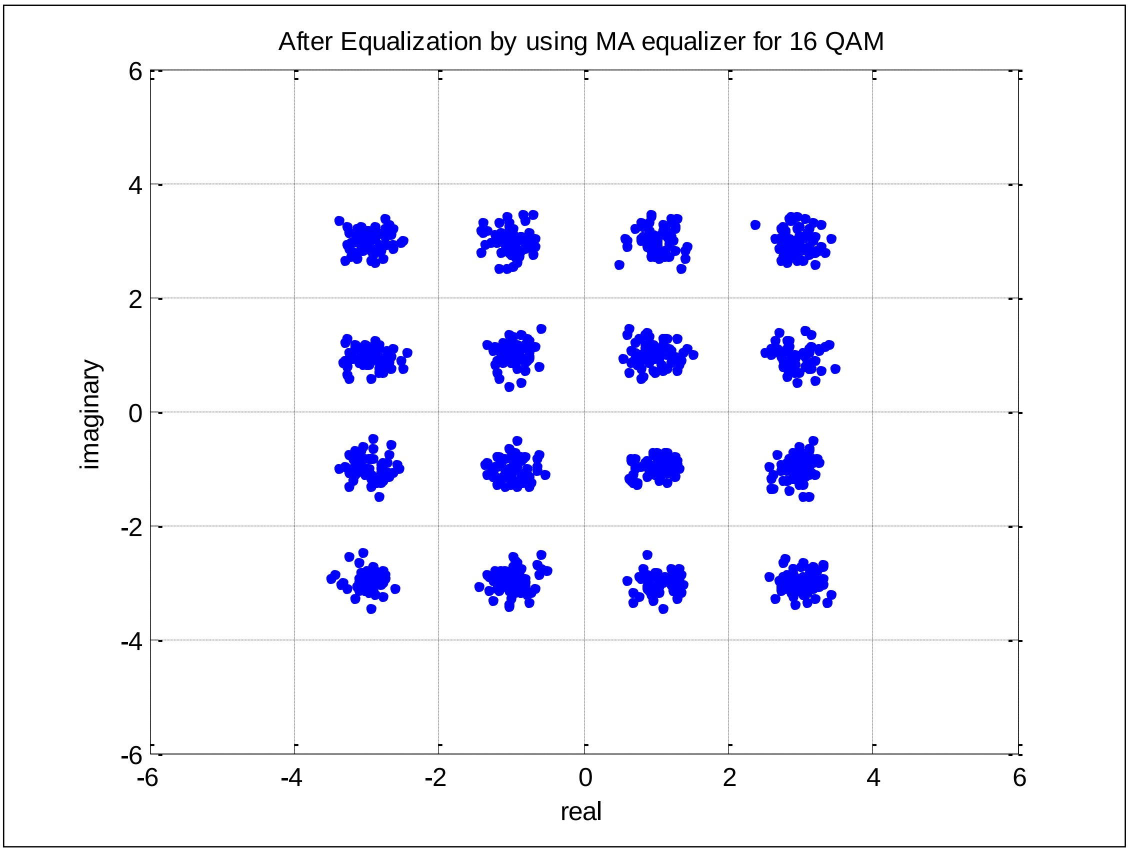

Fig.8 The output constellation of MA of 16 QAM

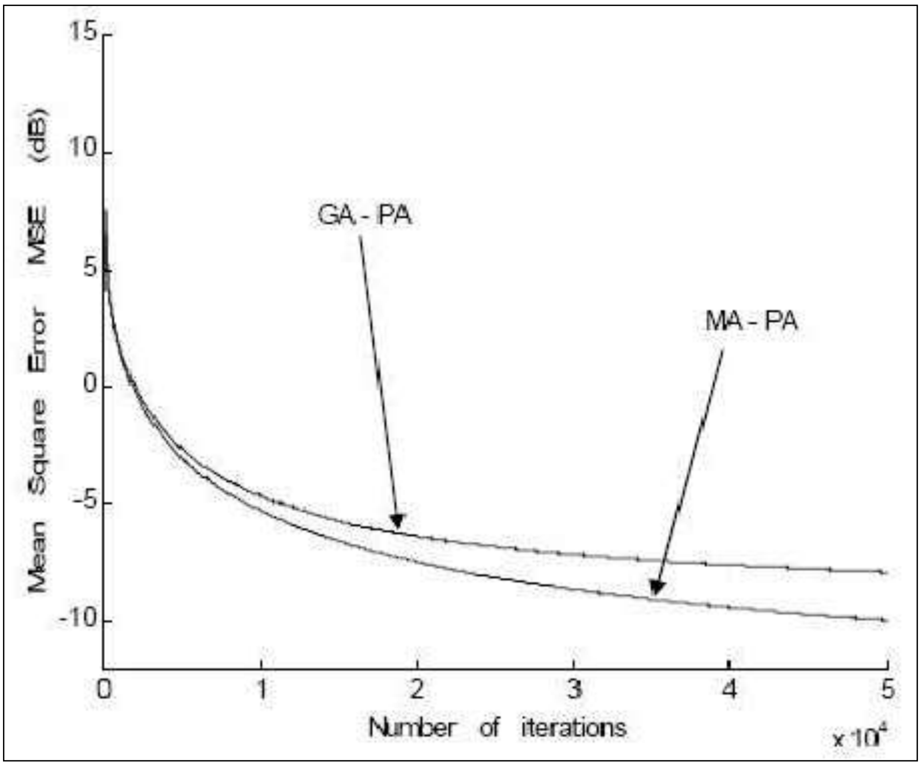

Fig. 9. Comparison between (GA-PA) and (MA-PA) equalizers

Conclusion

This paper, propose a new blind equalizer which provides an effective and robust way for adaptive blind equalization. The proposed algorithm MA algorithm has a better equalization performance compared with traditional blind equalizer. Then MA algorithm has a smaller residual error and a quicker convergence rate. We can conclude that MA algorithm is a practical blind equalization algorithm with an excellent overall performance. So the proposed algorithm MA system is strongly recommended in digital communications

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