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# Blind Equalization Based on Modified Constant Modulus Algorithm

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**Abstract:** Future services demand high data rate and quality. Thus, it is necessary to define new and robust algorithms to equalize channels and reduce noise in communications. Now a days, new equalization algorithms are being developed to optimize the channel bandwidth and reduce noise, namely, Blind Channel Equalization. Conventional equalizations minimizing mean-square error generally require a training sequence accompanying the data sequence. Considering the fact that blind equalizers do not require pilot signals to recover the transmitted data. The constant modulus algorithm (CMA) is one widely used algorithm for blind equalization of QAM signals. The algorithm exhibits slow convergence rate and large steady state mean square error and the phase-blind nature in comparison with the algorithms used in conventional data-aided equalization schemes. In this paper, a variable step size modified constant modulus algorithm is proposed. The proposed algorithm can speed up convergence rate and decrease steady state mean square error and correct phase error and frequency offset at the same time. The simulation results demonstrate the effectiveness of the proposed algorithm in improving convergence rate and reducing steady state mean square error.
Finally, a comparison of the simulation results of LMS and CMA for the test channels is provided, for different iteration rate.

**Keywords:** (Blind Equalization, CMA, LMS, MMA)

## I. INTRODUCTION

Adaptive equalizers compensate for signal distortion attributed to inter-symbol interference (ISI), which is caused by multipath within time-dispersive channels. The equalizer is the most expensive component of a data demodulator and can consume over 80% of the total computations needed to demodulate a given signal [1].

In the blind method of equalization, where the desired signal or training sequence is not available, some statistical property(mean ,varience, Kurtosis ) of the signal is used for the determination of the instantaneous error *e(n)*. It determines how much the output of the adaptive filter (equalizer) deviates from the desired property and calculates the instantaneous error. This instantaneous error is then used for updating the adaptive filter coefficient vector

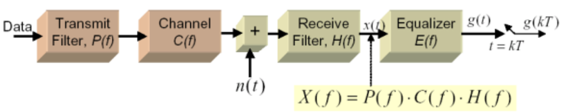


Figure.1 Communication system model

The most commonly used adaptive algorithm for blind channel equalization is the *Constant Modulus Algorithm* (CMA), which uses the constant modularity of the signal as the desired property. CMA assumes that the input to the channel is a modulated signal that has constant amplitude at every instant in time. Any deviation of the received signal amplitude from the constant value is considered a distortion introduced by the channel. The distortion is mainly caused by band-limiting or multi-path effects in the channel. Both these effects result in inter-symbol interference (ISI) and thus distort the received signal. The objective of equalization is to remove the effect of the channel from the received signal. CMA attempts to accomplish this objective by forcing the output of the adaptive filter (equalizer) to be of constant amplitude. CMA can also be used for QAM signals where the amplitude of the modulated signal is not the same at every instant. The error *e(n)* is then determined by considering the nearest valid amplitude level of the modulated signal as the desired value [1].

## II. PROBLEM STATEMENT AND MAIN CONTRIBUTION

Many algorithms have been proposed during the past few decades however due to its simplicity, good performance and robustness the constant-modulus algorithm (CMA) is most often used in practice. Maybe the most prominent disadvantage of CMA is its slow convergence [3]. The traditional LMS algorithm also suffers from slow convergence as well. Although in practice the improved Normalized LMS (NLMS) is used, which usually converges many times faster than the LMS algorithm at the cost of only a few extra computations per update [4].

## III. CONSTANT MODULUS ALGORITHM

CMA is a stochastic gradient algorithm that minimizes the dispersion of the equalizer output around a circular contour. The CMA algorithm adapts filter coefficients at each time *n* in order to minimize the modulus error. This algorithm was developed to perform the interference reduction functions on *constant-envelope signals*. The constant envelope signals, information is contained purely in the phasor angle, while the modulus or instantaneous

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amplitude is fixed.CMA seeks to minimize the cost function. Multi-Modulus Algorithm(MMA) is a variant of CMA, proposed to solve the phase ambiguity of CMA. The channel out put is defined as x(k), depends on input a(k), noise n(k),

$$x(k) = \sum_{i=0}^{L-1} h(i)a(k-i) + n(k)$$

The equalizer out put is y(k) = $W^H$(k) x (k)

$$W(k) = [w_0(k), w_1(k), \ldots, w_{N-1}(k)]^T$$
$$X(k) = [x(k), x(k-1), \ldots, x(k-N+1)]^T$$

The cost function of CMA is defined as

$$J(k) = E\left[\left(|y(k)|^2 - R_2\right)^2\right]$$

where *E* [.] indicates the statistical expectation, *y*(*k*) is the output of equalizer and *R*2 is a constant that depends on transmitted data statistics, given by

$$R_2 = \frac{E\left[|a(k)|^4\right]}{E\left[|a(k)|^2\right]}$$

The error signal is given by

$$e(k) = y(k)\left(|y(k)|^2 - R_2\right)$$

The update of tap weights vector can be written as

$$W(k+1) = W(k) - \mu \cdot e(k) \cdot X^*(k)$$

where μ is the step size parameter.

Modified CMA [4] modifies the cost function of CMA to the form of real and imaginary parts, the modified cost function can be written as

$$J(k) = J_R(k) + J_I(k)$$

$$J_R(k) = E\left[\left(|y_R(k)|^2 - R_{2,R}\right)^2\right]$$
$$J_I(k) = E\left[\left(|y_I(k)|^2 - R_{2,I}\right)^2\right]$$

$$R_{2,R} = \frac{E\left[|a_R(k)|^4\right]}{E\left[|a_R(k)|^2\right]}$$

$$R_{2,I} = \frac{E\left[|a_I(k)|^4\right]}{E\left[|a_I(k)|^2\right]}$$

where *R*2;*R* and *R*2;*I* are the real constants determined by the real and imaginary parts of trans-mitted data sequence respectively, the error signal $e(k) = e_R(k) + e_I(k)$ is given by

$$e_R(k) = y_R(k)\left(|y_R(k)|^2 - R_{2,R}\right)$$
$$e_I(k) = y_I(k)\left(|y_I(k)|^2 - R_{2,I}\right)$$

**The LMS Algorithm**

The LMS Algorithm is a developed form of steepest descent adaptive filter, in the family of stochastic gradient algorithms, which has a weight vector update equation given by:

$$\omega_{n+1} = \omega_n + \mu\, e(n)\, x^*(n)$$

The simplicity of the algorithm comes from the fact that the update for the kth coefficient requires only one multiplication and one addition (the value for $\mu$ * *e(n)* need only be computed once and may be used for all of the coefficients). The update equation fot the kth coefficient is given by

$$\omega_{n+1}(k) = \omega_n(k) + \mu\, e(n)\, x^*(n-k)$$

The step size determines the algorithm convergence rate. Too small step size will make the algorithm take a lot of iterations while too large step-size will not convergence the weight taps.

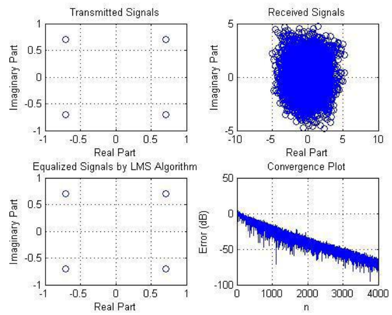


Figure.2 Signal constellation based on LMS algorithm

**Convex combination of Adaptive Equalizers**

A combination of blind equalization and LMS algorithm is proposed. The strengths of blind equalization i.e., equalization without training sequence and effective utilization of the bandwidth and the strengths of LMS algorithm i.e., good convergence of data are used together to get better results. The CMA module is designed to work in blind mode and the LMS works in the Decision-Directed mode ( DD mode). Initially, the received signal is fed to the blind equalizer and cost function is monitored. When the cost function or the error function reduces below a threshold value, the input is connected to the Decision-Directed mode. The combination is shown in the following figure 3.

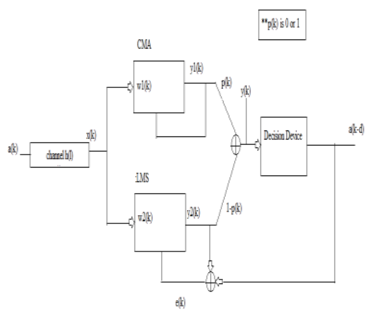


Figure.3 Block diagram of the proposed CMA

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**Modified filter with Variable Step-size**

Usually due to stability considerations, the step-size parameter which can be used in the blind equalization algorithms are much smaller than the corresponding values used in the LMS algorithm. In any adaptive filter the step-size range is given by

$$0 < \mu < \frac{2}{\lambda_{max}} \qquad -(8)$$

Where $\lambda_{max}$ is the greatest eigen value of autocorrelation matrix R. R is given by

$$R = E\{x(n)x^H(n)\} \qquad -(9)$$

The time varying step-size parameter can overcome the gradient noise amplification problem which occurs with the ordinary CMA or LMS algorithm whenever the input vector X(k) increases suddenly.

## IV. SIMULATION RESULT

Output of the conventional CMA

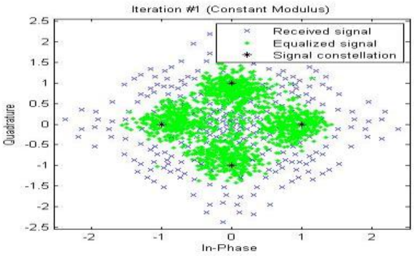


Figure 4: Output after 1st iteration

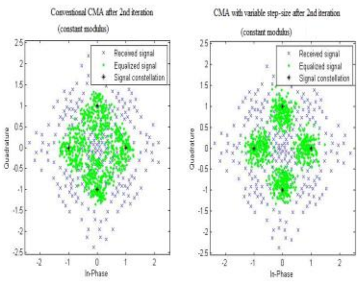


Figure 5: Output after 2st iteration

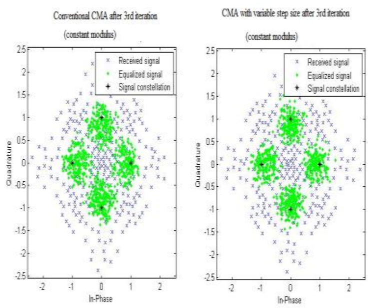


Figure 6: Output after 3rd iteration

From the output plots, it can be seen that by using variable step-size, the rate of convergence is increased. The output reaches the stable value quickly when compared to the conventional CMA algorithm.

## V. APPLICATION

Blind equalizers are used in Micro-wave radio. They were realized in very large scale integration(VLSI) for high definition television(HDTV) set-top cable demodulators. The blind processing applications are emerging in wireless communication technology.

## VI. CONCLUSION

Blind equalization is an emerging equalization technique which helps in utilizing the bandwidth efficiently. But the challenges faced are slow convergence rate and stability. Since efficient usage of bandwidth is very critical these days, researchers are coming up with new ways to overcome these challenges. By implementing the convex combination of adaptive algorithms and also by using the variable step-size, the performance of the equalizer is improved. Modifying the conventional CMA by exploiting the time-varying step-size parameter, the gradient amplification problem is handled. The simulation results show that this technique can improve the convergence speed compared to conventional CMA. It has also improved the stability of the system.

## REFERENCES

[1] J.R. Treichler, M.G. Larimore and J.C. Harp, "Practical Blind Demodulators for High- order QAM signals", Proceedings of the IEEE special issue on Blind System Identification and Estimation, vol. 86, pp. 1907-1926, Oct. 1998
[2] A Modified Blind Equalization Technique Based on a Constant Modulus Algorithm by Jia-Chin Lin and Lin-Shan Lee Department of Electrical Engineering, National Taiwan University, Taipei, Taiwan, R.O.C..1998.
[3] A Soft-Switching Blind Equalization Scheme via Convex Combination of Adaptive Filters by Magno T. M. Silva, Member, IEEE, and Jerónimo Arenas-García, Senior Member, IEEE, march 2013.
[4] Theory and design of Adaptive Filters by John R. Treichler, C. Richard Johnson Jr, Michael G. Larimore.
[5] daptive Signal Processing by Bernard Widrow and sameul D. Stearns
[6] Xiong, F. Q., Digital Modulation Techniques, Artech House, London, 2000.
[7] Proakis, J. G., Digital Communications, 4th Edition, McGraw-Hill, New York, NY, 2001.
[8] Godard, D. N., Self-recovering equalization and carrier tracking in two-dimensional data communication systems," IEEE Transactions on Communications, Vol. 28, 1867{1875, 1980.
[9] Oh, K. N. and Y. O. Chin, Modified constant modulus algorithm: Blind equalization and carrier phase recovery algorithm," 1995 IEEE International Conference on Communications,Vol. 1, 498{502, June 1995.
[10] Yang, J., J. J. Werner, and G. Dumont, \The multimodulus blind equalization and its gen-eralized algorithms," IEEE Journal on Selected Areas in Communications, Vo1. 20, No. 5,997{1015, June 2002.
[11] Weerackody, V., S. A. Kassam, and K. R. Laker, \A simple hard-limited adaptive algorithm for blind equalization," IEEE Transactions on Communications, Vol. 39, No. 7, 482{487, 1992.
[12] Thakallapalli, S. R., S. R. Nelatury, and S. S. Rao, \A new error function for fast phase recoveryof QAM signals in CMA blind equalizers," IEEE Workshop on Statistical Signal Processing,70{73, 2003.
[13] Chen, S., T. B. Cook, and L. C. Anderson, \Blind FIR equalization for high-order QAM signaling," Proceedings of IEEE ICSP'2, 2002.
[14] Ready, M. J. and R. P. Gooch, \Blind equalization based on radius directed adaptation, "Proceedings of ICASSP'90, 1699{1702, 1990.
[15] Weerackody,V. and S. A. Kassam, \Dual-mode type algorithms for blind equalization," IEEE Transactions on Communications, Vol. 42, No. 1, 22{28, 1994.
[16] Li, X. L. and X. D. Zhang, \A family of generalized constant modulus algorithms for blind equalization," IEEE Transactions on Communications, Vol. 54, No. 11, 1913{1917, 2006.