



## ADVANCED STUDY OF FUTURE GENERATION ANTENNA FOR WIRE-FREE APPLICATIONS

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### ABSTRACT

channel equalization is a process of compensating the disruptive effects caused mainly by inter symbol interference in a band-limited channel and plays a vital role for enabling higher data rate in digital communication. The development of new training algorithms, structures and the selection of the design parameters for equalizers are active fields of research which are exploiting the benefits of different signal processing techniques. Here in our paper work for equalization we use the various adaptive algorithms like least mean square (lms), recursive least square (rls), and constant modulus algorithms (cma) with 16-ary quadrature amplitude modulation (qam) and 16-ary phase shift keying (psk) modulation technique and then compare the mean square error of all these combinations of adaptive algorithms and modulation techniques. The simulation work has been done on matlab -2008 software.

**Keywords:** “equalizer”, “adaptive algorithms”, “digital modulation techniques”.

### 1. INTRODUCTION

An adaptive filter is defined as a self-designing system that relies for its operation on a recursive algorithm, which makes it possible for the filter to perform satisfactorily in an environment where knowledge of the relevant statistics is not available.

Adaptive filters are classified into two main groups: linear and non linear. Linear adaptive filters compute an estimate of a desired response by using a linear combination of the available set of observables applied to the input of the filter. Otherwise, the adaptive filter is said to be nonlinear. Adaptive filters may also be classified into:

Supervised adaptive filters, which require the availability of a training sequence that provides different realizations of a desired response for a specified input signal vector. The desired response is compared against the actual response of the filter due to the input signal vector, and the resulting error signal is used to adjust the free parameters of the filter. The process of parameter adjustments is continued in a step-by-step fashion until a steady-state condition is established.

Unsupervised adaptive filters, which performs adjustments of its free parameters without the need for a desired response. For the filter to perform its function, its design includes a set of rules that enable it to compute an input-output mapping with specific desirable properties. In the signal-processing literature, unsupervised adaptive filtering is often referred to as blind deconvolution or blind adaptation.

Gabor [1] was the first to conceive the idea of a nonlinear adaptive filter in 1954 using a Volterra series. The first algorithm used to design a linear adaptive filter is the ubiquitous least-mean-square (LMS) algorithm developed by Widrow and Hoff [2]. The LMS algorithm is often referred to as the Widrow-Hoff rule; it was originally derived by Widrow

and Hoff in 1959 in their study of a pattern recognition system known as the adaptive linear element (Adaline). The LMS algorithm is closely related to Rosenblatt’s perceptron [3] in that they are both built on error-correction learning. They both emerged about the same time in the late 1950s during the formative years of neural networks. The importance of Rosenblatt’s perceptron is largely historical today. On the other hand, the LMS algorithm has survived the test of time.

Adaptive filters find applications in highly diverse fields: channel equalization, system identification, predictive deconvolution, spectral analysis, signal detection, noise cancellation, and beamforming.

### 2. PROBLEM FORMULATION

In our paper we present the simulation of a communications system which consist various adaptive filters. The filters are used to compensate the channel effects, thus emulating a channel adaptive equalizer. In this paper we use various system parameters that can be changed. In this work we made the comparison between various adaptive algorithms with various modulation techniques. Hence with all these combinations we finally made the conclusion that which modulation technique is better for which type of adaptive algorithms. THE SIMULATION HAS DONE ON MATLAB R2008 SOFTWARE

#### • Equalisation of a Signal

Evening out a sign utilizing Communications System Toolbox programming includes taking after strides:

1.First we will make an equalizer protest that portrays the equalizer class and the versatile calculation that we wish to utilize.

2.We can change the properties of the equalizer object i.e. we can change the quantity of weights or the estimations of the weights.

3.Finally apply the equalizer item to the sign that we wish to even out, utilizing the adjust technique for the equalizer object.

**3. SIMULATION**

Here in our paper work for equalization we use the various adaptive algorithms like Least Mean Square (LMS), Recursive Least Square (RLS), and Constant Modules Algorithms (CMA) with 16-ary Quadrature amplitude modulation (QAM) and 16-ary Phase shift keying (PSK) modulation technique and then compare the mean square error of all these combinations of adaptive algorithms and modulation techniques. The simulation work has been done on MATLAB -2008 Software.

To obtain the results we have create total number of 6 cases and they are described as follows

**Case.1 :** Quadrature amplitude modulation technique with 16-ary and LMS equalization.

**Case.2 :** Quadrature amplitude modulation technique with 16-ary and CMA equalization.

**Case.3 :** Quadrature amplitude modulation technique with 16-ary and RLS equalization.

**Case.4 :** Phase shift keying modulation technique with 16-ary and LMS equalization.

**Case.5 :** Phase shift keying modulation technique with 16-ary and CMA equalization.

**Case.6 :** Phase shift keying modulation technique with 16-ary and RLS equalization.

error output of the equalizer is stored in the workspace in the matlab and then calculates the mean square error for first 50 terms. Here we find the complex error and this error is the difference between desired output and actual output. We convert the complex error into mean square error.

Scatter plots before equalization and after equalization also showing here for all above mentioned cases. By seeing the dust balls we can also conclude that which combination providing us the best result.

**Table 1 Comparison Table of Various Modulation and Adaptive Equalization Technique**

S.N O	Modulation Scheme	Adaptive Equalization Scheme	Mean Square Error
1	QAM	Linear LMS	0.5707
2		CMA	0.3040
3		RLS	0.5679
4		Linear LMS	0.0641

5	PSK	CMA	0.0104
6		RLS	0.1756

Here we can see the performance of various adaptive equalizers under various modulation schemes. Here in our paper work we are using LMS, RLS and CMA algorithms and modulation schemes are 16- QAM and 16-PSK. The measure of performance criteria is the Mean Square Error.

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We can analyze by table 5.7 that in the case of QAM, CMA algorithm has least error i.e. 0.3040 and the Linear LMS equalizer has the highest error rate i.e. 0.5707

Similarly by table 5.7 that in the case of PSK, RLS algorithm has most error rate i.e. 0.1756 and the CMA equalizer has the least error rate i.e. 0.9386.

In comparison between both two modulation techniques the performance of PSK is better than QAM.

**4. CONCLUSION**

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In future we can use more variety of adaptive algorithms like variable stepped LMS, sign LMS etc. and can use the various modulation techniques like QPSK, FSK, MSK, GMSK and can also use higher values of M i.e. 32 QAM, 64, QAM 128 QAM, 256 QAM 512 QAM etc. by using these techniques we can improve the equalization process.

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