1 Introduction

One of the practical problems in digital communications is inter-symbol interference (ISI), which causes a given transmitted symbol to be distorted by other transmitted symbols. The ISI is imposed on the transmitted signal due to the band limiting effect of the practical channel and also due to the multi-path effects (echo) of the channel. One of the most commonly used techniques to counter the channel distortion (ISI) is linear channel equalization. The equalizer is a linear filter that provides an approximate inverse of the channel response. Since it is common for the channel characteristics to be unknown or to change over time, the preferred embodiment of the equalizer is a structure that is adaptive in nature. Conventional equalization techniques employ a pre-assigned time slot (periodic for the time-varying situation) during which a training signal, known in advance by the receiver, is transmitted. In the receiver the equalizer coefficients are then changed or adapted by using some adaptive algorithm (e.g. LMS, RLS, etc.) so that the output of the equalizer closely matches the training sequence. However, inclusion of this training sequence with the transmitted information adds an overhead and thus reduces the throughput of the system. Therefore, to reduce the system overhead, adaptation schemes are preferred that do not require training, i.e., blind adaptation schemes. In blind equalization, instead of using the training sequence, one or more properties of the transmitted signal are used to estimate the inverse of the channel.

The problem with blind adaptation techniques is their poor convergence property compared to traditional techniques using training sequences. Generally a gradient descent based algorithm is used with the blind adaptation schemes. The most commonly used gradient descent based blind adaptation algorithm is the *Constant Modulus Algorithm* (CMA). CMA exploits the constant modularity of the transmitted signal for adapting the parameters of an equalizer. The counterpart of CMA is the *Least Mean Square* (LMS) algorithm that uses a training sequence for the adaptation process. Due to the knowledge of the transmitted sequence, the LMS algorithm, if convergent, will always converge to the global minimum. Moreover, for a particular delay in the overall system, the LMS cost function is quadratic and provides only a single global minimum in the cost surface. Therefore, irrespective of the initialization, the LMS algorithm will converge to the global minimum. If the initialization is such that adaptation takes place in a major eigenspace only, the convergence is fast.

For CMA based schemes, where the receiver does not know the transmitted sequence, any sequence with a constant phase offset with the input sequence may be considered to be the right sequence at the receiver since the phase shift does not change the constant modularity property of a signal. Due to this reason, unlike the LMS cost surface, the *Constant Modulus* (CM) cost surface will have multiple minima. Each of the minima will correspond to a unique phase shift. For a length N equalizer, the number of minima of the CM cost surface is N^2; in other words, there are N^2 different phase shifts for which there exist solutions in the CM sense. For most practical purposes, all of the N^2 solutions are not equally acceptable. If the source sequence is a differentially encoded M-ary PSK signal, for any transmitted sequence, M different sequences will be acceptable at the receiver. Therefore, out of the N^2 solutions only M will be acceptable and the minima corresponding to these acceptable solutions are called global minima. All other minima are called local minima, even though the cost at all minima is the same if the equalizer is not under-modeled [1]. The initialization of the equalizer determines the minimum point on the cost surface whereto CMA will force the equalizer to converge. Therefore, depending on its initialization, an equalizer employing CMA may converge to a local or a global minimum. Another problem with the CMA algorithm is that the convergence rate is much slower than the convergence rate of any gradient descent algorithm using a training sequence [8].

The objective of this work is to improve the convergence properties of the CM schemes. The convergence properties of any adaptive algorithm depend on the cost function, which is subject to minimization during the adaptation process. The cost function is a function of the equation for the error, defined as the difference between the present and the desired value of any property of the signal that is to be restored. Therefore, the cost function of an adaptive algorithm can be changed either by changing the function itself or by changing the error equation. The most commonly used definition of the cost function is the mean squared value of the error. Changing the definition of the cost function provides a lot of advantages. For example, a non-mean-square error criterion improves the performance of the adaptive algorithm when the interfering noise distribution is non-Gaussian [15]. In this work, instead of changing the definition of the cost function, different types of error equations were used to change the CM cost surface. In Chapter 3 it is shown that the performance of CMA can be improved by only changing the equation for the error signal. Some error equations provide better convergence rate, while other error equations improve performance by eliminating the probability of converging to local minima. Convergence to global minima can be confirmed by following some steps and checks during the initialization and adaptation respectively [4]. A brief description of this procedure is given in Chapter 2.

If the channel is represented by an FIR filter, it is practically impossible to find another FIR filter which will perfectly equalize the channel. This is true if the tap spacing of the equalizer is the same as the symbol period *T*. This type of equalizer is called a *Baud Spaced Equalizer* (BSE). The limitation of the FIR equalizers while equalizing an FIR channel can be overcome by using the special type of FIR equalizer whose tap spacing is an integer fraction of the symbol period *T*. This special type of FIR equalizer is called a *Fractionally Spaced Equalizer* (FSE). The FSE uses a different technique to equalize the channel than finding the inverse of the channel. The equalization technique of FSE is discussed in Chapter 2. Chapter 2 also discusses conditions and constraints on the channel and the length of the equalizer for perfect equalization or perfect source recovery. A general overview of CMA is also given in Chapter 2, where the convergence property of CMA is discussed and compared with the LMS algorithm.

In Chapter 4, a *Recursive Linear Predictor* (RLP) is designed to work in a non-stationary case. The main advantage of the designed predictor is that it does not need any a priori information about the input source sequence. In the usual linear predictor a priori knowledge about the correlation matrix of the input sequence is required for the design of the predictor. Again, when the coefficients of the predictor are kept fixed, this usual linear predictor is not expected to perform well in a time-varying situation. The developed predictor uses a recursively estimated correlation matrix. The correlation matrix is updated for every input sample by using a forgetting window such as in the exponentially forgetting RLS algorithm. Due to this updating method, the developed predictor can predict future values of a colored sequence based on only the present and past input samples, even in the time-varying case. The predictor is then used to predict the fading envelopes generated under constant velocity and constant acceleration conditions for mobile communication scenarios. The results of these experiments suggest that, for a flat fading channel, the designed predictor can be used as an *Automatic Gain Controller* (AGC) to compensate for the effect of fading. A unique way of generating a Rayleigh fading envelope for constant acceleration is also described in Chapter 4.

In Chapter 5 a new blind equalization scheme is suggested to improve the tracking property of a conventional blind equalizer. The new scheme uses a linear predictor to predict the future value of the envelope of the equalizer output in conjunction with a commonly used fractionally spaced blind equalizer. The equalizer coefficients are normalized at every time instant so that the impulse response is a unit energy sequence. The predictor works as an AGC and compensates for the time-varying gain of the channel.