Optimizing the performance in a Wireless Channel Using Adaptive Signal Processing

Hitesh Sahoo*, Sovit Pradhan, Deepak Bhuyan
Department of Electronics and Communication Engineering
Gandhi Institute for Education and Technology, Baniatangi, Bhubaneswar

Prangya Padhi, Department of Electronics and Communication Engineering,Capital Engineering College, Bhubaneswar

Abstract – In digital communication systems, the transmission of a training sequence is either a very costly affair in terms of data output. Adaptive filters using conventional LMS algorithm depend on the use of training sequences. So, the blind adaptive channel equalization algorithms that do not depend on training signals have been used. These blind algorithms help the individual receivers to begin self-adaptation without transmitter assistance. A multiple objective optimization approach is presented in this paper to fast blind channel equalization. The performance (mean-square error) of the standard fractionally spaced CMA (constant modulus algorithm) equalizer is investigated by adding noise. The CMA local minima exist near the minimum mean-square error (MMSE) equalizers. The Fractional Spaced CMA may converge to a local minimum corresponding to a badly designed MMSE receiver with considerable large mean-square error. The multi-channel LTI representation of LTV channel is also presented.

KEYWORDS-
Fractional Spaced CMA, LMS, Adaptive MMSE, LTI, LTV

1. INTRODUCTION

One of the earliest and most successful applications of adaptive filters is adaptive channel equalization in digital communication systems. Using the standard least mean LMS algorithm, an adaptive equalizer is a finite-impulse-response FIR filter whose desired reference signal is a known training sequence sent by the transmitter over the unknown channel. The reliance of an adaptive channel equalizer on a training sequence requires that the transmitter cooperates by (often periodically) resending the training sequence, lowering the effective data rate of the communication link.

Conventional LMS adaptive filters depending on the use of training sequences cannot be used. For this reason, blind adaptive channel equalization algorithms that do not rely on training signals have been developed. Using these "blind" algorithms, individual receivers can begin self-adaptation without transmitter assistance. Accurate estimation of the communication channel greatly affects the performance of communication systems operating over the medium. The signal transmitted over a channel, such as the fading channel, is affected by many distortions that result in both amplitude and phase fluctuations. Furthermore, the delay spread of the channel introduces inter symbol interference (ISI) to the received signal, which is one of the major obstacles to reliable and high-speed data transmission. Channel equalization is the process of compensating for the negative effect of the channel on the transmitted signal and removing the resulting ISI [4]. To achieve this goal the equalizer uses an estimate of the channel frequency response; however the fading channel varies throughout the transmission cycle, requiring the equalizer to learn the frequency response in an adaptive fashion to be able to continuously mitigate the negative effect of the channel. The challenge is achieving blind equalization using only a limited amount of data. A widely tested algorithm is the constant modulus algorithm (CMA). In the absence of noise, under the condition of the channel invertibility, the CMA converges globally for symbol-rate IIR equalizers and fractionally spaced FIR equalizers. It is shown in [9] that CMA is less affected by the ill conditioning of the channel. However, Ding et. al. [2] showed that CMA may converge to some local minimum for the symbol rate FIR equalizer. In the presence of noise, the analysis of convergence of CMA is difficult and little conclusive results are available. Another drawback of CMA is that its convergence rate may not be sufficient for fast fading channels. Another approach to the blind equalization is based on the blind channel estimation. Some of the recent eigen structure-based channel estimations require a relatively smaller data size comparing with higher order statistical methods. The key idea of this paper is to combine the approach based on minimizing theconstant modulus cost and that based on matching the second order cyclostationary statistics. The main feature of the proposed approach is the improved convergence property over the standard CMA equalization and the improved robustness for ill conditioned channels.

2. Fractionally Spaced Blind Equalization

In the blind channel equalization, where the desired signal or training sequence is not available, some property of the signal is used for the calculation of the instantaneous error e(n). Figure 1 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 f (n).

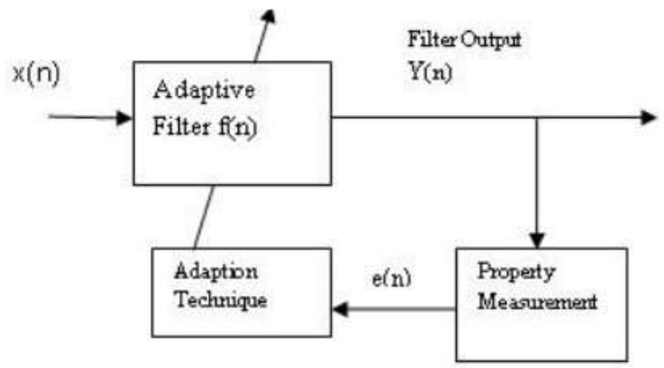

Figure1 : Adaptive Filter Model For Blind Equalization

The most commonly used adaptive algorithm for the technique used here 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 deviationof 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].

3. Blind Adaptive Algorithm: CMA

Let s(n) be the input to the channel and x(n) be its output, which is the noisy and distorted version of s(n). Let an equalizer be used to remove the channel distortion from the received signal. If f is the tap-weight vector of the equalizer of length L,
the output of the equalizer y(n) can be represented by
( ) = ( ) (1) ( ) = [( ) (− 1) . . . (− + 1)]
In (1), x(n) is the equalizer input vector and represents the Hermitian transpose of f. In the presence of noise, the main aim of designing an equalizer is to minimize the expected value of the squared value of the recovery error,
( ) = (−)− ( ) (2)
for a particular value of delay d. This criterion provides the best compromise between ISI and noise amplification in a minimum mean-squared error (MMSE) sense [1]. Therefore, the tap weight vector f of the equalizer is chosen so as to minimize
the MSE by using some optimization technique. Generally the optimization techniques are designed to minimize scalar functions that depend on the
specific criteria (e.g. MSE). This scalar function is known as the Cost Function. A cost function is defined as the transformation from a vector space spanned by the elements of the tap-weight vector to the space of a real scalar. For the MSE criterion, the cost function JMSE is defined as:
= {| (−)− ( )| }
={|()|}(3)

Even though the MSE criterion provides the optimum equalizer, this criterion cannot be used to optimize the equalizer coefficient vector when the source signal s(n) is completely unknown to the receiver. In such cases, at least one of the properties of the input signal s(n) is assumed to be known at the receiver, and the optimization
technique used to remove the channel distortion from the received signal is designed to restore the known property to the signal. The most commonly used blind

optimization technique is the Constant Modulus Algorithm (CMA), which uses constant modularity as the desired property of the output. Generally the error equation used with CMA is defined as:

()=|()|−(4)

In (4) A is the desired amplitude level. CMA optimizes the equalizer coefficients to minimize the cost function JCM, which is defined as:

={|()|}={|−()|}(5)

4. Fractionally Spaced Equalizer

4.1 Multi-Channel Model

For a fractionally spaced equalizer (FSE), the tap spacing of the equalizer is a fraction of the baud spacing (in time) or the transmitted symbol period. As the output of the equalizer has the same rate as the input symbol rate, the output ofthe FSE needs to be calculated once in every symbol period. In this situation, the FSE can be modelled as a parallel combination of a number of baud spaced equalizers. This parallel combination of baud spaced equalizers is known as the Multi-Channel Model of FSE. The Multi-Channel Model of FSE for an oversampling factor of 2 was derived in one of the earlier works [1]. The over-sampling factor determines the tap spacing of the FSE. If T is the symbol period, then

$$\text{Tap spacing} = \frac{1}{oversampling\ factor}$$

The combined effect of the linear time-invariant (LTI) channel, and the impulse response of the pulse shaping filter in continuous time will be represented by c(t).

$$r(t) = \sum_{n=-\infty}^{\infty} s(n)c(t - nt - t_0) + w(t) \qquad (6)$$

Let us consider the continuous time noise as w(t). If s(n) is the transmitted discrete symbol sequence with symbol period T and the base-band equivalent of the analog received signal is r(t), then r(t) can be expressed as:

The fractionally spaced equalizer (FSE) is an FIR filter and the tap spacing of this filter is T/M, where M is the over-sampling factor.

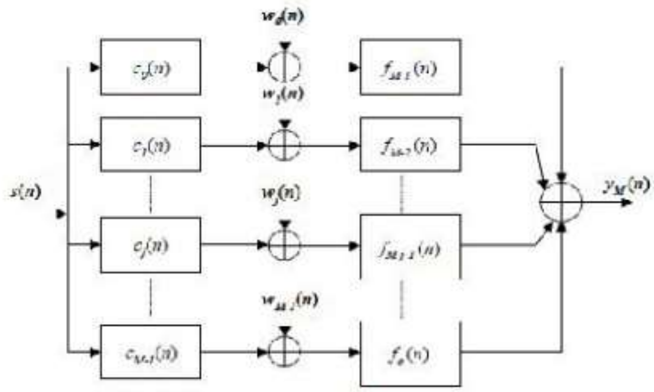

Figure 2: Multi-Channel Model of the Fractionally Spaced Equalizer

5. Simulations (I)

- Take total number of data & SNR
- Simulate Received noisy signal
- Generate four QAM symbol sequence
- Check Received noisy signal and SNR
- Apply Constant Modulus Criteria to QAM symbols
- Find instant error and update equalizer
- Estimate the symbols
- Find max. Of the composite response
- Perform symbol detection
- Calculate SER
- Show the patterns of transmitted symbols and received samples
- Show the patterns of equalized samples and equalized coefficients.

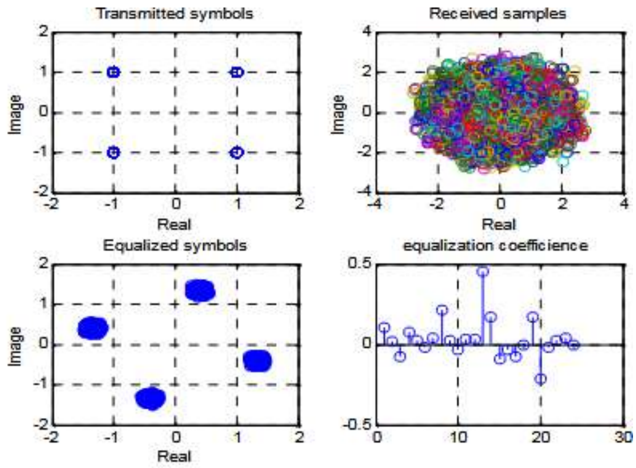

Figure 3: Transmitted symbol ,received and equalized symbol, with FSE-CMA

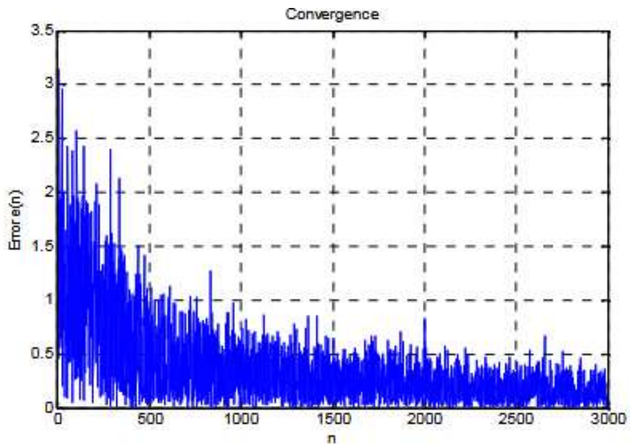

Figure 4. : Convergence plot with iteration for CMA

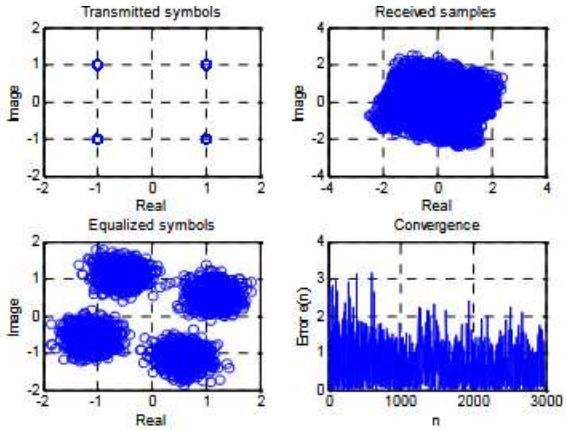

Figure 5:Transmitted symbol ,received and equalized symbol.

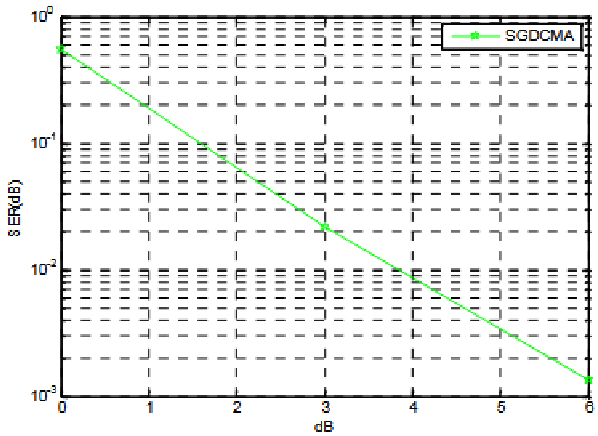

Figure 6: SER plot

Simulations (II):

- Take FSE equalizer's tap number& speed of the channel
- Take the parameter SNR, learning rate(mu),sample number with fractional
- space, source signal standard deviation(sigma)and Godard constant
- Generate source signal
- Take sampling frequency
- fill with zeros in order to make the source
- signal fractional spaced
- the source passed through the channel(LTI)
- Initialize the equalizer

- Generate the fractionally spaced sampled signal (the even sampled signal and odd sampled signal)
- calculate the recovered signal Y(i) and CMA error e(i)
- Plot the average error curve and output constellation

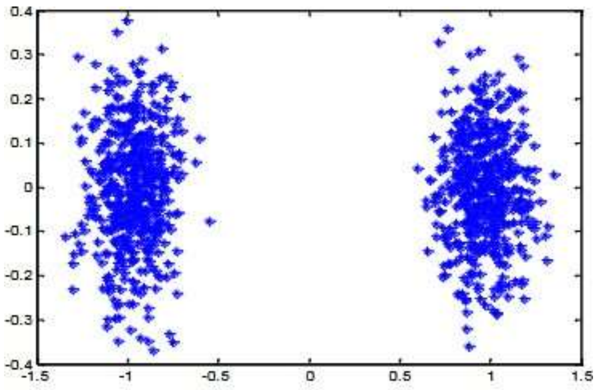

Figure 7: Output constellation Plot

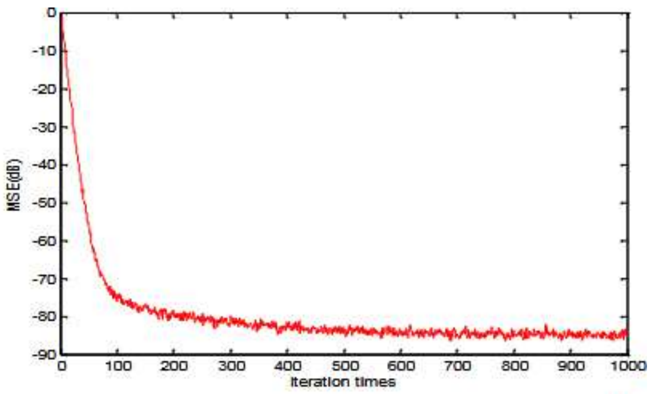

Figure 8: The average error curve with SNR=20dB, mu=0.02

6. Conclusion:

Convergence of blind equalizer CMA-FSE is based on simple length and zero condition. The need of no common zero among sub-channels may sometime be restrictive. When common zeros exist CMA-FSE may not be able to estimate the common factor among sub channels, in this case an additional linear filter may be added after the vector equalizer to minimize the remaining ISI which is computational complexity. In future conclusion, the proposed low complexity adaptive equalizer in code-multiplexed CDMA system can be proposed and this system has better practical application value.

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