A Study on Genetic Equalization Techniques to reduce SER & Carrier phase Recovery

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ABSTRACT
The fastest growing segment of the communication industry is the wireless communication system. However, the various phenomena such as multipath wave propagation, time dispersion, and fading causes errors like Inter Symbol Interference (ISI), and other distortions into the signals transmitted over wireless channel and affect the quality of communications. To overcome these problem equalization techniques is normally used. The purpose of this paper is to review various equalization techniques in order to compensate the effect of symbol error rate. Typically, adaptive filtering algorithm (LMS & RLS) employed in the equalizer design.

Introduction

One of the most important advantages of the digital transmission systems for voice, data and video communications is their higher reliability in noise environment in comparison with that of their analog counterparts. Unfortunately most often the digital transmission of information face multi-path propagation causing problems such as arbitrary time dispersion, attenuation, and phase shift in the received signal. Also, when digital transmission is The effects of multipath include constructive and destructive interference, and phase shifting of the signal. The term ISI means that the transmitted pulses are smeared out so that pulses that correspond to different symbols are not separable; and have amplitude and phase dispersion which results in the interference of the transmitted signals with one another. ISI has been recognized as the

Major obstacle to high speed data transmission over mobile radio channel. Thus for a reliable digital transmission system it is crucial to reduce the effects of ISI. This can be reduced using equalizers, which are designed to work in such a way that BER (Bit Error Rate) should be low and SNR (Signal-to-Noise Ratio) should be high. More specifically, equalizers are designed to give the inverse of channel to the received signal. Thus, the combination of channel and equalizers will ideally give a flat frequency response and linear phase [3-6]. This implies that equalizer is an inverse filter of the channel. As the mobile fading channels are random and time varying, equalizers must track the time varying characteristics of the mobile channel and thus are called adaptive equalizers. [7-8]. Therefore, this paper is focused on adaptive equalization for the unpredictable wireless channel.

Equalizer

Equalizers are an important part of receivers, which minimizes the linear distortion produced by the channel. If channel characteristics are known a priori, than optimum setting for equalizers can be computed. But in practical systems the channel characteristics are not known a priori, so adaptive equalizers are used. .
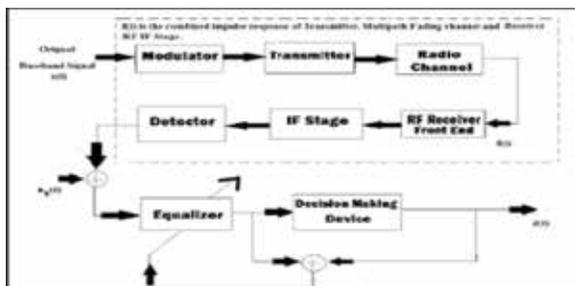

Figure 1: Block Diagram of Adaptive Equalizer [1]

Least Mean Squares Algorithm (LMS)

Least Mean Squares (LMS) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time [3]. LMS algorithm is built around a transversal filter, which is responsible for performing the filtering process.
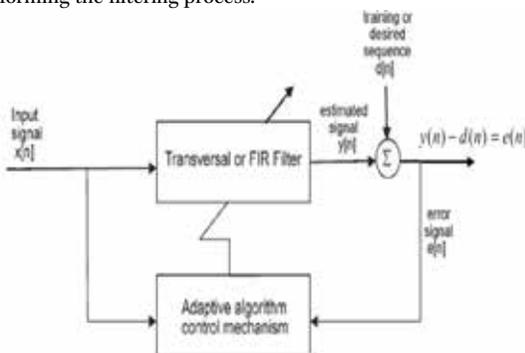

Figure 2: Block diagram of adaptive transversal filter employing LMS algorithm [3]

A weight control mechanism responsible for performing the adaptive control process on the tape weight of the transversal filter as illustrated in Figure 2.The LMS algorithm in general, consists of two basics procedure:

Filtering process, which involve, computing the input signal and generating an estimation error by comparing this output with a desired response as follows:

e(n)=d(n)-y(n)

y(n) is filter output and is the desired response at time n.

Adaptiveprocess, which involves the automatics adjustment of the parameter the filter in accordance with the estimation error.

$$\widehat{w}(n+1) = \widehat{w}(n) + \mu(u)e^{+}(n)$$

Where is the step-size, (n+1) = estimate of tape weight vector at time (n+1) and If the prior knowledge of the tape weight vector (n) is not available set (n)=0.

The combination of these two processes working together constitutes a feedback loop. First, a transversal filter, around which the LMS algorithm is built this component is responsible for performing the filtering process. Second, a mechanism for performing the adaptive control process on the tap weight of the transversal filter- hence the designated "adaptive weight -control mechanism".

Recursive Least Square Algorithm (RLS)

The RLS algorithm has the same to procedures as LMS algorithm, except that it provides a tracking rate sufficient for fast fading channel, moreover RLS algorithm is known to have the stability issues due to the covariance update formula p(n), which is used for automatics adjustment in accordance with the estimation error as follows:

$$p(0) = \delta^{-1} I$$

Where p is inverse correlation matrix and is regularization parameter, positive constant for high SNR and negative constant for low SNR. For each instance time n=1, 2, 3, 4………
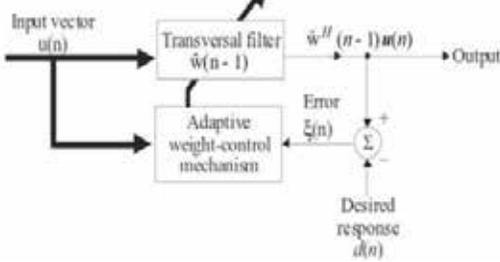
Figure 3: Block diagram of adaptive transversal filter employing RLS algorithm [3]

(III) Working Methodology

The signal received by the equalizer is given by

y(t) = x(t) * f *(t) + nb(t)
where y(t) is the input to Equalizer.
x(t) is the original signal
f(t) is the combined impulse response.
nb(t) is the baseband noise.

If the impulse response of the equalization is heq (t), the output of equalizer is
d(t) = x(t) * f *(t) * heq (t) + nb(t) * heq (t)
or
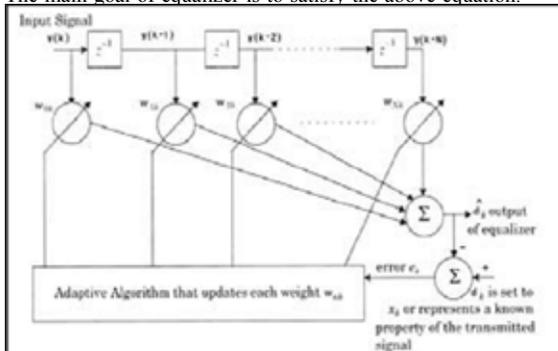d(t) = x(t) * g(t) +nb(t) * heq (t)

where g(t) = f *(t) * heq (t)

The desired output is x(t) which is the original source data.

Delay Line Structure:-

Assume that nb(t) = 0, then in order that d(t) = x(t)

g(t) = f *(t) * heq (t) = δ (t)

The main goal of equalizer is to satisfy the above equation.

Figure 4: Generic Adaptive Equalizer With Tapp [1]

In Frequency domain, it is given byHeq (f) F*(-f) this implies

that equalizer is an inverse filter of the channel. If the channel is frequency selective, theequalizer, Enhance the frequency components with small amplitude 1. And attenuate the strong frequencies in the received frequency 2, Spectrum. For a time varying channel, the equalizer is designed to track the channel variations so that the above equation is approximately satisfied.

In the block diagram, the subscript k represents discrete time index. It can be seen from the block diagram that there is a single input yk at any time instant and the value of yk depends upon the instantaneous state of radio channels and specific value of noise. The block diagram scheme shown is called transversal filter.

In this case, there are N delay elements, N+1 delay taps and N+1 tunable multipliers called weights. These weights have second subscript k to explicitly show that they vary with time and are updated on a sample by sample basis or sometimes for a whole block of weight vectors [9]. The adaptive algorithm is controlled by the error signal ek. The error signal is derived by comparing the output of the equalizer with some signal dk which is either replica of transmitter signal xk or which represents a known property of transmitted signal. The adaptive algorithm uses ek to minimize the cost function and uses equalizer weights in such a manner that it minimizes the cost function iteratively. How many iteration it will take, that depends upon the choice of algorithm. The LMS (least mean Square) algorithm searches for the optimum or near optimum weights. From classical equalization theory, the most common cost function is Mean Square Error (MSE) between the desired signal and output of the equalizer. Now let's see, how the weights are updated…

New weights = (Previous weights) + Constant * (Previous error) * (Current Input Vector)

Where, Previous error = Previous Desired Output – Previous

Actual Output Here the constant may be adjusted by the algorithm to control the variation between filter weights on successive iterations.

This process is repeated rapidly in a programming loop while the equalizer tries to converge. When the convergence is reached, the algorithm freezes the filter weights. There is Recursive algorithm try to freeze the filter weights. Rate of convergence will decide which equalization techniques to be used for real time application.

(IV) Adaptive Equalization for QAM Systems for Carrier Phase Recovery

The adaptive equalization can be used in various modulation techniques to recover the phase of the carrier at the receiver side for coherent detection receiver. As we can see in figure below that the constellation for 16- QAM are shown and whenever the symbols get distorted by noise present in channel, it's become unpredictable for the receiver to separate these symbols and interpret the information. But by using equalizer algorithm, the receiver can predict the symbols and recover the information as shown in fig. 5.

Figure 5: Adaptive Equalization Used for Phase Recovery in 16 QAM [3]
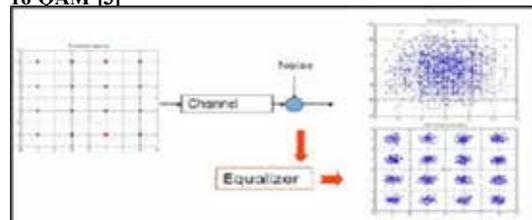
(V) Conclusion

The basic innovation in this paper is to address adaptive algorithm to the problem of carrier phase recovery. We have carried out a theoretical analysis of the two approaches. First, training sequence of input data signal. Then, we derived the suitable equations based on algorithm for input data. This algorithm was used to estimate the phase offset via a least square optimization method. Second is based on the Tracking Sequence of fir minimize the SER Symbol error rate and ISI. This theoretical approach have provided the experimental verification of this algorithm, by analyzing its error performance based on Step size for higher order Modulation Techniques.

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