EE4TK4 PROJECT 1

COMPUTER SIMULATION ON ADAPTIVE EQUALIZATION

PROFESSOR K. M. WONG
DEPARTMENT OF ELECTRICAL AND COMPUTER ENGINEERING
MCMASTER UNIVERSITY, CANADA

1. INTRODUCTION

Inter-symbol Interference (ISI) is a major source of errors in determining the signal level at the receiver. When the channel bandwidth is not large enough to accommodate the essential frequency content of the data stream, it will result in signal distortion in the form of time spreading. As a consequence, the received bit is affected by its adjacent bits, resulting in the ISI. Another cause of ISI is the occurrence of multi-path during which the transmitted signals are reflected by moving and/or fixed objects. In this case, signals will arrive at the receiver via multiple propagation paths with different delays. These signals can be added destructively or constructively, causing ISI. The multi-path effect can also be looked upon as a frequency distortion and limitation in the transmission channel giving rise to ISI.

One of the methods to reduce ISI is the employment of equalization at the receiver. In this project, we would like to study the effects of adaptive equalization on transmission over multi-path channels.

2. EXPERIMENT

Consider the baseband transmission/reception system as shown in Figure 1.

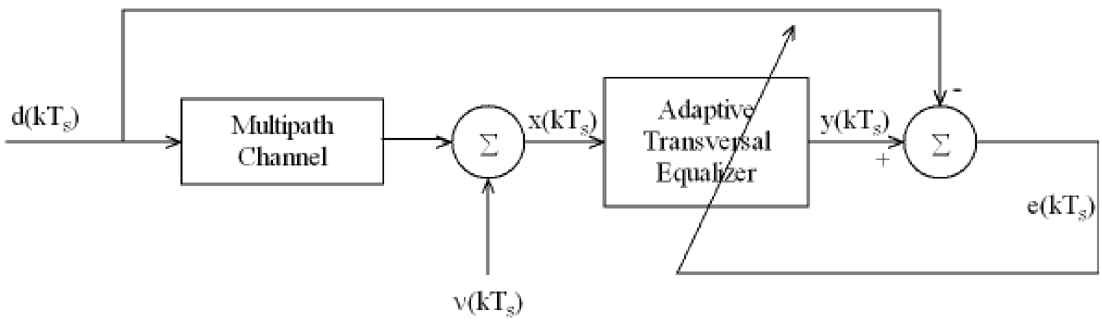

Figure 1. Baseband Transmission/Receptive System with an Adaptive Equalizer

$d(t)$ is randomly generated with equal probability and the amplitude of the pulse is either -1 or +1.

$d(kT_s)$ is the sampled signal sequence transmitted over the ISI channel. Figure 2 is the illustration of the transmitted signal $d(kT_s)$.

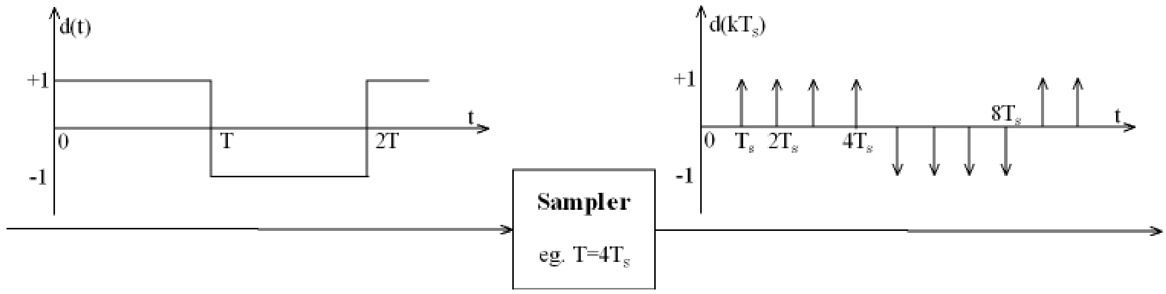

Figure 2. Illustration of the Transmitted Signal $d(kT_s)$

The impulse response of the channel is given by

$$h(kT_s)=\begin{cases}(-1)^k\exp(-0.5k) & k=0,\ 1,\ 2\\ 0 & \text{elsewhere}\end{cases},$$

At the receiver, the transmitted signal through the multi-path channel is corrupted by an additive zero-mean Gaussian noise $\nu(kT_s)$ with variance $\sigma^2=0.02$, such that the received signal is given by $x(kT_s)=d(kT_s)*h(kT_s)+\nu(kT_s)$, where $*$ denotes the discrete time convolution.

The objective of this project is to implement an adaptive equalizer that corrects the distortion introduced by the multi-path channel to the signal during the transmission.

2.1 PART A

Referring to the class notes, for a 7-tap equalizer, calculate:

1) The auto-correlation matrix of the received signal $\mathbf{\Phi}_{\mathbf{xx}}$.

2) The cross-correlation vector between the desired and the received signals $\boldsymbol{\phi}_{d\mathbf{x}}$.

3) The optimum tap weight vector $\mathbf{w}_{op}$ for the equalizer using the estimated $\mathbf{\Phi}_{\mathbf{xx}}$ and $\boldsymbol{\phi}_{d\mathbf{x}}$.

4) The corresponding minimum mean square error $\varepsilon^2_{\min}$ using the estimated $\mathbf{\Phi}_{\mathbf{xx}}$ and $\boldsymbol{\phi}_{d\mathbf{x}}$.

2.2 PART B

For the same number of the equalizer taps, apply the Least Mean Square (LMS) algorithm with step sizes $\alpha=0.1$, $\alpha=0.025$, and $\alpha=0.00625$. For each step size, calculate the minimum mean square error $\varepsilon^2_{\min}(i)$ for each i-iteration and plot it against i (learning curves). Comment on the effect of changing the step size α on the learning curves. For large number of iterations, estimate the steady state value of $\varepsilon_{\min}^2$ and compare it with the one calculated in PART A. Also compare the final values of the tap weights to the optimum ones obtained in PART A. Plot the eye diagram for the received signal before and after the adaptive equalization.

2.3 PART C

Change the number of the equalizer taps to 11, and repeat the steps 1 to 4 in PART A. Comment on the effect of the tap's number on $\varepsilon_{\min}^2$. Plot the eye diagram for the received signal before and after the adaptive equalization.

HINT

Design the adaptive equalizer by two methods:

1) Estimate the covariance matrix of the received signal and then, using the Wiener_Hopf equation, evaluate the estimated optimum weights;
2) Implement a steepest descent algorithm for the equalizer and use the least mean square method to obtain the approximate optimum weights.

The results obtained by these two methods are compared.