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| Blind Equalizer |
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18\11\2011

# Abstract

In this project the above mentioned students are going to study the Inter Symbol Interference (ISI) on base band digital transmitted data.

ISI is one of the fundamentals problems in digital communication system that arises due to the nature of dispersive channel. One way to reduce the effect of ISI is to reduce the bit rate of the transmitted symbols. However this solution is not preferred since the trend in modern communication systems requires large transmission rate.

An alternate solution to minimize the effect of ISI is to find the inverse impulse response of the channel therefore equalize the channel and reduce the effect of ISI.

In this project equalization techniques such as decision feedback and blind equalization will be investigated over a dispersive channel. The study of these equalization techniques will be applied to quadrature phase shift keying modulated data.

In order to accomplish this project, students will simulate a randomly generated bit stream with and without the use equalizer inserted in the receiver by using MATLAB. After the successful completion of the simulation students may use real data transmission (text file between two PCs) by using DSP kits in the next semester.

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# Chapter One

## 1.1: INTRODUCTION

Information-bearing signals transmitted between remote locations often encounter a signal-altering physical channel. Examples of common physical channels include coaxial, fiber optic, or twisted-pair cable in wired communications and the atmosphere or ocean in wireless communications. Each of these physical channels may cause signal distortion, including echoes and frequency-selective filtering of the transmitted signal. In digital communications, a critical manifestation of distortion is intersymbol interference (ISI), whereby symbols transmitted before and after a given symbol corrupt the detection of that symbol. All physical channels (at high enough data rates) tend to exhibit ISI. The presence of ISI is readily observable in the sampled impulse response of a channel; an impulse response corresponding to a lack of ISI contains a single spike of width less than the time between symbols.

Linear channel equalization, an approach commonly used to counter the effects of linear channel distortion, can be viewed as the application of a linear filter (i.e., the equalizer) to the received signal. The equalizer attempts to extract the transmitted symbol sequence by counteracting the effects of ISI, thereby improving the probability of correct symbol detection. Since it is common for the channel characteristics to be unknown (e.g., at startup) or to change over time, the preferred embodiment of the equalizer is a structure adaptive in nature.

Classical equalization techniques employ a time-slot (recurring periodically for time-varying situations) during which a training signal, known in advance by the receiver, is transmitted. The receiver adapts the equalizer (e.g., via LMS) so that its output closely matches the known reference training signal. Since the inclusion of such signals sacrifices valuable channel capacity, adaptation without resort to training, i.e., blind adaptation, is preferred.

The most studied and implemented blind adaptation algorithm of the 1990’s is the constant modulus algorithm (CMA). CMA seeks to minimize a cost defined by the CM criterion. The CM criterion penalizes deviations in the modulus (i.e., magnitude) of the equalized signal away from a fixed value. In certain ideal conditions, minimizing the CM cost can be shown to result in perfect (zero-forcing) equalization of the received signal. Remarkably, the CM criterion can successfully equalize signals characterized by source alphabets not possessing a constant modulus [e.g., 16-quadrature amplitude modulation (QAM), as well as those possessing a constant modulus (e.g., 8-PSK).

## 1.2: History

Blind equalization algorithms blossomed in the 1980’s. The two principal precursors are Lucky’s blind decision-direction algorithm and Sato’s algorithm. What we term the CM criterion was introduced for blind equalization of QAM signals in and of pulse-amplitude modulation (PAM)

and FM signals. By the end of the 1980’s blind equalizers were commercialized for microwave radio. By the mid 1990’s blind equalizers

were realized in very large scale integration (VLSI) for high definition television (HDTV) set-top cable demodulators. The current explosion of interest in the CM criterion stems from blind processing applications in emerging wireless communication technology (e.g., blind equalization, blind source separation, and blind antenna steering) and from CMA’s record of practical success**([[1]](#footnote-2))**.

#

# Chapter two

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## 2.1: intersymbol interference

In [telecommunication](http://en.wikipedia.org/wiki/Telecommunication), intersymbol interference (ISI) is a form of [distortion](http://en.wikipedia.org/wiki/Distortion) of a [signal](http://en.wikipedia.org/wiki/Signal_%28electrical_engineering%29) in which one [symbol](http://en.wikipedia.org/wiki/Symbol_%28data%29) interferes with subsequent symbols. This is an unwanted phenomenon as the previous symbols have similar effect as [noise](http://en.wikipedia.org/wiki/Electronic_noise), thus making the communication less reliable. ISI is usually caused by multipath propagation or the inherent non-linear frequency response of a [channel](http://en.wikipedia.org/wiki/Channel_%28communications%29) causing successive symbols to "blur" together. The presence of ISI in the system introduces errors in the decision device at the receiver output. Therefore, in the design of the transmitting and receiving filters, the objective is to minimize the effects of ISI, and thereby deliver the digital data to its destination with the smallest error rate possible. Ways to fight intersymbol interference include [adaptive equalization](http://en.wikipedia.org/wiki/Adaptive_equalization) and [the](http://en.wikipedia.org/wiki/Error_detection_and_correction) raised cosine filter.

When the receiving filter is configured to compensate for the distortion caused by both the transmitter and the channel, it is often referred to as an equalizing filter or a receiving\ equalizing filter. Figure (1): illustrate intersymbol interference.



(a)



(b)

Fig (1)- (a) what we sent, (b) what we received**([[2]](#footnote-3))**

To understand what (ISI) is**([[3]](#footnote-4))**, let us consider the transmission of a sequence of symbols with the basic waveform *u(t)*. To send the *n*th symbol *bn* we send *bnu(t - nT)*, where T is the symbol interval. Therefore, the transmitted signal is:

$\sum\_{n}^{}b\_{n}u(t-nT)$ (2.1.1)

Based on the dispersive channel model, the received signal is given by:

 r(t)=$\sum\_{n}^{}b\_{n}v\left(t-nT\right)+n(t)$ (2.1.2)

Where$v(t)= u \* h\_{c}(t)$is the received waveform for a symbol. If a single symbol, say the symbol *b0*, is transmitted, the optimal demodulator is the one that employs the matched filter, i.e., we can pass the received signal through the matched filter$ṽ(t)=v(-t)$ and then sample the matched filter output at time *t = 0*to obtain the decision statistic. When a sequence of symbols is transmitted, we can still employ this matched filter to perform demodulation. A reasonable strategy is to sample the matched filter output at time $t = mT$to obtain the decision statistic for the symbol *bm*. At $t = mT$, the output of the matched filter is:

|  |  |
| --- | --- |
| $$z\_{m}=\sum\_{n}^{}b\_{n}v\*ṽ\left(mT-nT\right)+n\_{m}$$= bm$\left‖v\right‖^{2}+ \sum\_{n\ne m}^{}b\_{n}v\*ṽ\left(mT-nT\right)+n\_{m}$ |  (2.1.3) |

where $n\_{m}$ is a zero-mean Gaussian random variable with variance$N\_{0}\left‖v\right‖^{2}/2$.

The first term in (2.1.3) is the desired signal contribution due to the symbol *bm* and the second term contains contributions from the other symbols. These unwanted contributions from other symbols are called intersymbol interference (ISI).

##

## 2.2 How we can fight ISI?

ISI can be minimized by using different types of equalizers. The concept of equalization some of the common equalizers used in the literature will be reviewed in this document.

### 2.2.1 Equalization

It is the process of adjusting the strength of certain frequencies within a signal. The most well-known use of equalization is in [sound recording and reproduction](http://encyclopedia.thefreedictionary.com/Sound%2Brecording%2Band%2Breproduction) but there are many other applications in electronics and telecommunications.

#### 2.2.1.1 What is the Equalizer?

Simply, an equalizer is a number of electronic filters which allow you to control (or adjust) the frequency response (or tone) of a sound system. The name comes from the telephone industry. The frequency response of the phone line needed to be adjusted to make the frequency response "equal" at all the frequencies of interest. There are several different types of equalizers available today and they are becoming increasingly sophisticated.

#### 2.2.1.2 Why we use equalizers?

The three fundamental reasons for using an equalizer are:

1. To increase the naturalness or intelligibility of the sound system.

2. To increase the gain or volume of the system before feedback occurs.

3. To minimize the Intersymbol Interference (ISI).

# Chapter Three

# Types of Equalizers

## 3.1: Adaptive Equalizer

Is an [equalizer](http://en.wikipedia.org/wiki/Equalizer_%28communications%29) that automatically adapts to time-varying properties of the [communication channel](http://en.wikipedia.org/wiki/Communication_channel); it is frequently used with coherent modulations such as [phase shift keying](http://en.wikipedia.org/wiki/Phase_shift_keying), mitigating the effects of [multipath propagation](http://en.wikipedia.org/wiki/Multipath_propagation) and [Doppler spreading](http://en.wikipedia.org/wiki/Fading).

There are four basic classes of adaptive filtering:

|  |  |  |
| --- | --- | --- |
| **Class of adaptive****filtering** | **Application** | **Purpose** |
| Identification | SystemidentificationLayered earthmodeling | Given an unknown dynamical system, the purpose system identification is to design an adaptive filter that provides an approximation to the system. In exploration seismology, a layered modeled of the earth is developed to unravel the complexities of the earth’s surface. |
| Inverse modeling | Equalization | Given a channel of unknown impulse response, the purpose of an adaptive equalizer is to operate on the channel output such that the cascade connection of the channel and the equalizer provides an approximation to an ideal transmission medium. |
| Prediction | Predictive codingSpectrumanalysis | The adaptive prediction is used to develop a model of a signal of interest (e.g., a speech signal); rather than encode the signal directly, in predictive coding the prediction error is encoded for transmission or storage. Typically, the prediction error has a smallervariance than the original signal-Hence the basis for improved encoding. In this application, predictive modeling is used to estimate the power spectrum of a signal of interest. |
| Interferencecancellation | Noise cancellationBeamforming | The purpose of an adaptive noise canceller is to subtract noise from a received signal inadaptively controlled manner so as to improve the signal-to-noise ratio. Echo cancellation,experienced on telephone circuits, is a special form of noise cancellation .Noise cancellation is also used in electrocardiography. A beamforming is spatial filter consist of an array of antenna elements with adjustable weights (coefficients). The twin purposes of an adaptivebeamformer are to adaptively control the weights so as to cancel interfering signal.  |

Table (1): FOUR BASIC CLASSES OF ADAPTIVE FILTERING APPLICATION**([[4]](#footnote-5))**

Many adaptation strategies exist, includes:

### 3.1.1 Least mean squares (LMS) algorithms

It is a class of adaptive filter used to mimics 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).(LMS) incorporates an iterative procedure that makes successive corrections to the weight vector in the direction of the negative of the gradient vector which eventually leads to the minimum mean square error. Compared to other algorithms LMS algorithm is relatively simple; it does not require correlation function calculation and does not it require matrix inversions.

#### 3.1.1.1: The LMS algorithms([[5]](#footnote-6))

***L*** *denotes the length of the filter.*

***Hn*** *the filter coefficient vector at time n*

***Yn****denotes the reference signal*

***Xn****= (Xn, X(n-1), …, X(n-L+1))t the vector of the L last input samples.*

The equations describing an LMS algorithm are:

|  |  |
| --- | --- |
| $$e\_{n}=y\_{n}-X\_{n}^{t}H\_{n}$$ | (2.2.1) |
| $$H\_{n+1}=H\_{n}+µX\_{n}e\_{n}$$ | (2.2.2) |

By writing equations (2.2.1), (2.2.2) at successive time samples i = 0,1,…n, and successively reporting H(i-1) in the next tap vector Hi, the LMS equations are seen to be equivalent to:

|  |  |
| --- | --- |
| $$e\_{n}=y\_{n}-µ\sum\_{i=1}^{n}si\left(n\right)e\_{n-i}-X\_{n}^{t}H\_{0}$$ | (2.2.3) |

With *si(n)* an estimation of the *ith* correlation coefficient at time n:

|  |  |
| --- | --- |
| $$s\_{i}\left(n\right)=X\_{n}^{t}X\_{n-i}$$ | (2.2.4) |



Fig(2): Linear adaptive equalization system**([[6]](#footnote-7))**

##

## 3.2: Blind Equalization

Blind equalization is a [digital signal processing](http://en.wikipedia.org/wiki/Digital_signal_processing) technique in which the [transmitted](http://en.wikipedia.org/wiki/Transmitter)[signal](http://en.wikipedia.org/wiki/Signal_%28electrical_engineering%29) is inferred ([equalized](http://en.wikipedia.org/wiki/Equalizer_%28communications%29)) from the [received](http://en.wikipedia.org/wiki/Receiver_%28information_theory%29) signal, while making use only of the transmitted signal statistics. Hence, the use of the word blind in the name. Blind equalization is essentially [blind de-convolution](http://en.wikipedia.org/wiki/Blind_deconvolution) applied to [digital communications](http://en.wikipedia.org/wiki/Digital_communications). Nonetheless, the emphasis in blind equalization is on [online](http://en.wikipedia.org/wiki/Online_algorithm) [estimation](http://en.wikipedia.org/wiki/Estimation) of the [equalizer](http://en.wikipedia.org/wiki/Equalization_filter) [filter](http://en.wikipedia.org/wiki/Signal_processing), which is the [inverse](http://en.wikipedia.org/wiki/Inverse_filter#Inverse_system) of the [channel](http://en.wikipedia.org/wiki/Channel_%28communications%29) [impulse response](http://en.wikipedia.org/wiki/Impulse_response), rather than the estimation of the channel impulse response itself. This is due to blind de-convolution common mode of usage in digital communications systems, as a mean to extract the continuously transmitted signal from the received signal, with the channel impulse response being of secondary intrinsic importance. The estimated equalizer is then [convolved](http://en.wikipedia.org/wiki/Convolution) with the received signal to yield an estimation of the transmitted signal.

In wireless communication systems, channel identification and equalization is one of the most challenging tasks because broadcast channels are often subject to frequency selective, time varying fading and there are several bandwidth limitations. Furthermore, each receiver channel has vastly different types of channel characteristics and signals to noise ratio. We will try to estimate blind equalization schemes particularly using constant modulus algorithm (CMA). We will try to estimate a linear channel model driven by a QAM, QPSK source and adapt an adaptive equalizer using CMA, The CMA function creates an adaptive algorithm object that you can use with the linear function(Construct linear equalizer object) or DFE function(Construct decision-feedback equalizer object) to create an equalizer object. You can then use the equalizer object with the equalize function to equalize a signal.

####

#### Algorithms used for blind equalizer:

Since they are used for blind equalization, no reference signal is used in these two algorithms.

In the Decision directed algorithm (DD), the reference signal is reconstructed from available data, usually as the constellation point which is the nearest from the actual output of the filter. Otherwise, all equations are the same ones as in classical cases (LMS or RLS, depending on the chosen algorithm). The modification of the LMS recursive scheme is due to the absence of reference signal.

# Chapter four

# Simulation Results

|  |  |
| --- | --- |
| (a) This is the original data without noise and before it's transmission In the channel | 1. This is the transmitted data after the dispersive channel with large probability of error
 |
| (c) This is the transmitted data After the equalizer | (d) nearly after 400 samples there is a convergence |

Fig(3): blind equalizer**([[7]](#footnote-8))**



1. Matlab simulation model



1. input signal



1. input signal + noise



1. output signal from the equalizer

fig(4): LMS adaptive linear prediction**([[8]](#footnote-9))**

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