Sirte University Scientific Journal (Applied Sciences) Vol.12(1), June 2022

Equalization network

Omar Masoud Salih[1] and Naiema yahya[2]
E-mail: omeryomery@yahoo.com

[1],[2] Computer Department, Faculty of Science, Sirt University, Libya.

Abstract

The primary objective of this paper is to provide the requirements the network to equalize a signal and then send it without using demodulation or error-correction. Where the objective of the equalizer is to separate out the multiple paths and provide a clean, distortion free. This is one of the practical applications for adaptive processes for equalization.

Most communications systems have three processes at the receiver; equalization, demodulation and error-correction. The equalization process remove Inter-Symbol Interference (ISI) caused by the channel from a sequence of symbols. To compensate corruptions caused by ISI and to find the original information transmitted, equalization process performed at the receiver. Equalization refers to any signal processing techniques used at the receiver to combat ISI in depressive channels in the presence of additive. Equalizing filters must cancel out any group delay and phase delay between different frequency components.

Keywords: *Equalization, Demodulation, Error-correction, Inter-Symbol Interference (ISI).*

1. Introduction

An equalizer (EQ) is a network designed with lumped elements or distributed ones, as transmission lines, active elements, or adaptive system, placed at one of the two ends of trace or in both. The EQ has to amplify the signal to count the attenuation at each low and high frequencies domain. It has to amplify high frequency oscillated wave or to reduce the signal magnitude at the low frequencies wave so the EQ will compensate and balance the two frequency domains, high and low. The components or function of the EQ has different variants: active, passive, adaptive equalization. The EQ are commonly stored as active EQ or passive EQ.

2. Inter-symbol interference (ISI) and equalization

When digital signals transmitted through frequency-selective communication channels, one of the problems that arise is inter-symbol interference (ISI). When a signal is going to take different paths from the transmitter to the receiver. Each of these paths travels a different distance; and as a result, multiple copies of the same signal arrive at the receiver at different times. If the time delay between the multiple paths is approximately the symbol period, these multiple paths cause inter-symbol interference (ISI). This occurs when we are transmitting over a channel that has some echoes. These echoes cause the receiver to hear a confused signal instead of the original signal from the transmitter [2].

A loss line with both phase and amplitude distortion will have height pulse reduction and dispersion so the pulse smears outside of its assigned bit boundaries. Low-frequency energy present in a transmission line will interfere with highly attenuated energy from the high-frequency domain causing inter-symbol interference (ISI) [3].

Each channel will have different characteristics, some channels may have echoes, others may have delays, and often channels will have both. When a channel has echoes, this is called a multipath channel because there are multiple paths to reach the receiver.

The effect of ISI summarized as the spread of the pulses traveling along the line with bandwidth limitation and in worst-case scenarios; it will smear into the neighboring time slots, leading to distortion of a bit within, a symbol due to interference caused by pre-or post – bit from the transmitted stream [2].

Equalization refers to any signal processing techniques used at the receiver to combat ISI in depressive channels in the presence of additive noise; this is known as Equalization for a known channel. When the receiver does not know the channel characteristics, the process of equalization essentially has two jobs; first, identify the known channel, second remove the inter-symbol interference. If the receiver did not identify the channel first, there would be no way to remove the effects of it on the received signal [1].

3. Equalization types:

3.1 Active channel Equalization:

The EQ is designed with the use of active elements as amplifiers to improvement, where it consists of passive EQ and additionally an amplifier to enhance the signal levels in the entire frequency domain, which may be high or low. The method is done by using the split-path approach. The input signal is split into two paths: one has unity gain and another is to be a high-frequency boost. In the end, the two signals summed into a new equalized input signal into the loss line [2]. One of the disadvantages of this method is the overall cost**.**

3.2 Passive channel Equalization:

Passive EQs cannot boost or amplify frequency ranges. Often passive EQ will have an amplifier on its output stage to compensate for level lost during the filtering process.

3.3 Adaptive channel Equalization:

Adaptive channel Equalization is an equalizer that automatically adapts to the varying properties of the communication channel. It is the frequency used with coherent modulations such as phase shift keying, justifying the effects of multipath propagation and Doppler spreading, which refers to the widening of the spectrum of a narrow-band signal transmitted through a multipath propagation channel. The equalizer requires knowledge of the transmitted signal to adequately adapt to remove this distortion. This introduces overhead into the signal effectively reducing the data.

4. Equalization structure:

Two different structures for an equalizer are available and each has its advantages and disadvantages. The **first** structure is a channel estimator followed by a minimum mean square error or a maximum likelihood equalizer constructed from this channel estimate. This equalizer structure called indirect equalization, which provides information on the channel itself. The **second** is referred to as direct equalization because it will be the directed calculation of the equalizer from the received signal[4]. Several different adaptive filtering algorithms can perform the calculation of the channel estimate in indirect equalization and the equalizer indirect equalization.

The different adaptive filter algorithms are split into two classes: **informed** and **blind**. An informed adaption uses a predetermined sequence to construct an error sequence used in a cost function that is minimized. A blind algorithm relies on some assumed 23 statistics of the received signal and possibly some minimal amount of knowledge of the signal itself such as baud rate and/or modulation technique to calculate the filter coefficients.

4.1 Indirect Equalization:

This structure uses an estimator filter to estimate the response of the channel. For informed equalizers, the error signal defined as:

$$er(n) = y(n) - \sum_{l=0}^{Lh-1} h^{\wedge}(l) a\,(n-l)$$

$^{L}h\text{-}1$ is the order of the channel impulse response.

The blind algorithms are designed to directly estimate the channel from the received signal $y(n)$

4.2 Direct Equalization:

This structure is used when the equalizer coefficients are directly computed. This eliminates the intermediate step of first estimating the channel. The basic structure used for direct equalization is shown in Figure 1, and the error signal is defined by the following equation:

$$er(n) = d(n) - \sum_{l=0}^{Le-1} e(l)y\,(n-l)$$

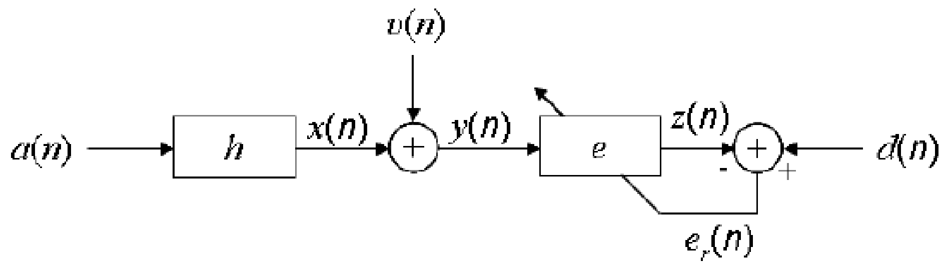

Figure 1: System model used for direct equalization.

4.3 Adaptive Algorithms:

This part focuses on three different informed adaptive filtering algorithms and three different blind algorithms.

5.1 Informed adaptive algorithms:

The main three algorithms are used for channel estimation and/or equalization, which are: the Wiener filter, the least mean square algorithm (LMS), and the recursive least square algorithm (RLS). All of these algorithms are applicable to filter estimation and The main three algorithms are used for channel estimation and/or equalization, which are: the Wiener filter, the least mean square algorithm (LMS) , and the recursive least square algorithm (RLS). All of these algorithms are applicable to filter estimation and filter De convolution [5] by changing which signal is desired signal and input signal to adaptive filter[4] by changing which signal is desired signal and input signal to adaptive filter[4]**.**

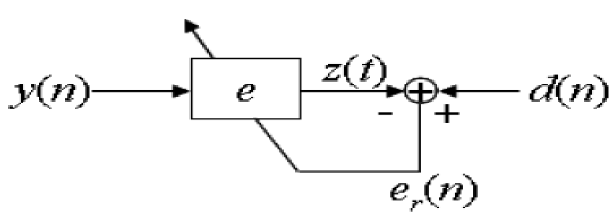

Figure 2: The basic structure used for informed adaptive filter

A channel estimate is found by the desired signal, $d(n)$,being fed into the filter and the error generated by the difference between output of the filter and the received signal, $y(n)$. As shown in the figure above.

5.2 Blind adaptive algorithms:

It can be classified into two different categories: Higher-order statistic (HOS) methods, and second-order statistic (SOS) methods, these methods can deliver excellent results provided the channel medium is slowly varying or time-invariant. The disadvantage of HOS methods is that they have a slow convergence rate, the reason for that is the amount of data required to accurate represent a cumulate increase exponentially with the order of that cumulate.

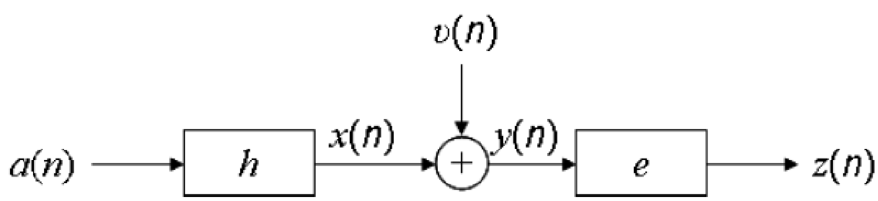

Figure 3: The basic structure used for blind adaptive filter

In this model the input source $a(n)$, h is a time-invariant complex channel, and e is the equalizer calculated from the blind algorithm.

5. Timing analysis of equalization:

A controlled amount of ISI is based on the sampling instance of the signal introduced by the shape of the pulse[4]. The sampling is crucial to the quality of the signal. If the signal is not sampled accordingly, neighboring symbols are going to interfere with one another.

7. Sampling phase:

The sampling phase affects the performance of several different adaptive filtering algorithms, including both informed and blind at both symbol spaced and fractionally spaced rates. Sampling phase offset occurs when a phase offset is present between the sampling of the transmitter and the receiver.

In the center of the pulse, the sampling instance of the transmitter is assumed to be at the zero crossings of neighboring symbols. Where the zero sampling phase offset occurs when the signal is sampled at the exact optimal instance and no ISI is introduced by the pulse shaping.

We can define the amount of phase offset as the amount deviated from sampling at the center of the pulse. Sampling offsets of (a) -50%, (b) -25%, (c) 0%, and (d) 25% of a period *T* as shown in the Figure below for a raised cosine:

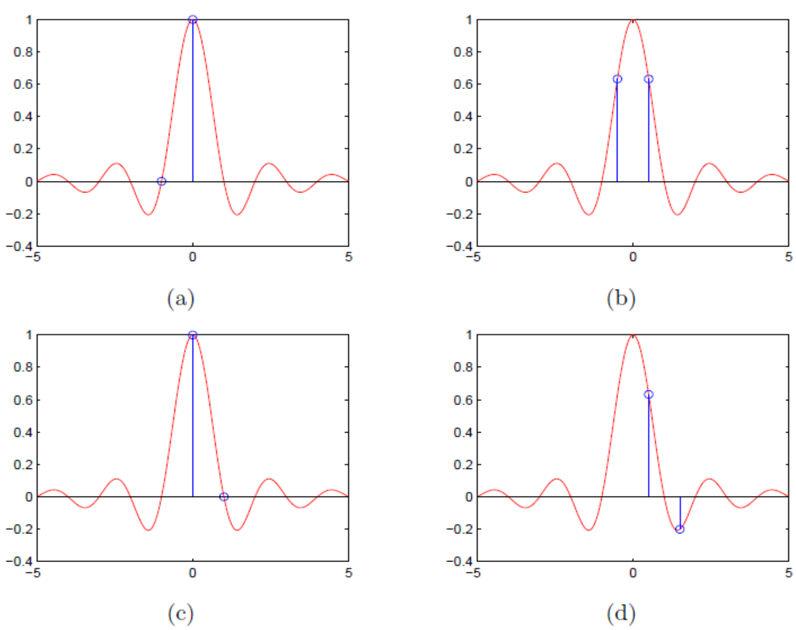

Figure 3. Sampling a raised cosine pulse

8. Conclusion

Wireless channels often distort a signal beyond the point of reliable demodulation. To account for this problem receivers, incorporate an equalizer whose objective is to remove the distortion introduced by the medium of transmission. The equalizer was designed by using periodically spaced information known by both the transmitter and receiver, i.e. probes, to form an error sequence based on the difference between the transmitted and received signals. This introduces a certain degree of overhead into the signal. The indirect equalizers obtained faster convergence because the order of the estimating filter can be less than the required length of a direct equalizer. The informed equalizer used all the available signal content for calculating the filter coefficients. A blind equalizer did not have access to all the information used by an informed algorithm and only used statistical measures of the received signal.

References:

[1] Ryanb . Casey, B.S.E.E., M.S.E.E, “ Blind Equalization of an HF channel for a passive listening system” , electrical engineering , Faculty of Texas Tech University, 05/08/2020..

[2] Anant Sahai, Ed. John Wawrzynek, Ed. Laura Brink, “Deep Networks for Equalization in Communications”, Electrical Engineering and Computer Sciences University of California at Berkeley, December 14, 2018.

[3] Diana Brinaru, “Passive Equalization Networks—Efficient Synthesis Approach for High-Speed Signal Integrity Characterization”, Faculty of Electronics, Telecommunications and Information Technology, University POLITEHNICA of Bucharest, 2021.

[4] Amiya Kumar Tripathy, Raghavendra K T “An efficient channel equalizer using artificial neural networks “, Don Bosco Institute of Technology, · January 2006.

[5] Traitement du Signal S5 Convolution et Filtrage by V. Choqueuse Département Electronique, ENIB