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Improving the effect of ISI from optical Communication System by Using Equalizers

Ruth Chopra[*1], Sanjay Tomar[2],Neha Mehta[3]
[*1,2,3] P.D.M College of Engineering, Bahadurgarh, Haryana,India
ruthchopra@gmail.com

Abstract

The technique of compensating the effect of ISI in optical communication system described here is equalization .This technique of equalization is used for compensating the effect of ISI in the channel which causes disturbance in the signal, which is transmitted. There are various types of equalizers are used depending upon the application used and the kind of system in which it is used. The most important purpose of the equalization technique is to correct the channel frequency response. They not only correct the frequency response of the channel but also cancel the effects of multipath signal component present in the system. This paper aims at studying and simulation of equalization techniques; here two types of equalization techniques are stimulated, they are Zero Forcing Equalization and MMSE Equalization. Here in these techniques filters are used at the receiving end to cancel the effects of ISI in the received signal introduced by channel impulse response. As Inter Symbol Interference is considered to be one of the most challenging problems encountered in fiber optical communication system so as to remove the effect we can apply equalizer at the receiver to undo the effect of the channel by applying an inverse filter. The First technique used is zero forcing (ZF) equalizer to reduce the effect of inter symbol interference (ISI) introduced by the channel impulse response. The second method used Minimum Mean Square Error (MMSE) equalizer to reduce the effect of inter symbol interference (ISI) introduced by the channel impulse response. Simulation result shows that the Minimum Mean Square Error (MMSE) equalizer reduce the effect of inter symbol interference (ISI) better as compare to that of zero forcing (ZF) equalizer.

Keywords: Equalizers, Zero Forcing Equalizers, and Minimum Mean Square Error Equalizers, BER, SNR.

Introduction

Thus in designing of any communication system which can be complex and large it is necessary that one device communicate digital information to other device. This information can be digital information as well as Analog information. Digital Technology translates and reassembles data and in the process is more prone to loss of quality as compared to analog technology. Computer advancement has enabled use of error detection and error correction techniques to remove disturbances artificially from digital signals and improve quality. 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. The main components of communication system are transmitter and receiver and channel. However the process of communication using fiber-optics involves creating the optical signal involving the use of a transmitter, relaying the signal along the fiber, ensuring that the signal does not become too distorted or weak, receiving the optical signal, and converting it into an electrical signal. Here Equalizers are used which are used to estimate the transmitted bits/symbols in such a way that it eliminates the effect of the channel. The main purpose of the equalizer is to reduce the Inter Symbol Interference without enhancing the additive noise power [8]. Here two types of equalizers are used, one is Zero forcing Equalizer and the other is Minimum Mean Square Error Equalizer. These both equalizers are used to mitigate the effect of ISI.

Impact of Inter Symbol Interference

As we know that the disturbance we face while communication is some sort of disturbance or distortion and Inter symbol Interference is a (ISI) is a form of distortion of a signal in which one symbol interferes with subsequent symbols. ISI is caused by multipath propagation or the inherent non-linear frequency response of a channel causing successive symbols to "blur" together. The presence of ISI in the system introduces errors in the decision device at the receiver output. Basically when ISI occurs this energy leaks into other and this affects the system in various ways such as

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the path loss becomes more, the power penalty due to ISI increases more rapidly for highly dispersive channels due to the shorter slot duration, It becomes impossible to achieve high data rate or transmission distance in high-speed multimode Fibre-optic links for local area networks applications and these all results into degraded Performance. Thus the important performance parameters of any communication systems are signal to noise ratio and bit error rate [5] .Thus to minimize the effect of ISI , SNR is maximized in the system [9].

Bit Error Rate

With the increase in use of high speed communications systems, the use of the term bit error rate (BER) is becoming more popular. BER is commonly used as a performance metric which describes the probability of error in terms of number of bits transmitted or we can say in simple terms it is the ratio between the numbers of bits received in error to the total number of bits received .Thus BER is the probability of receiving a single bit in error [5].

Signal to Noise Ratio

SNR measures the quality of a transmission channel or an audio signal over a network channel .The greater is the ratio, the easier is to identify and eliminate the source of noise. Thus signal to noise ratio is a relative measure of the signal power compared to the noise power. E_b/N_0 is a normalized signal to noise ratio. It is the energy per bit to noise power spectral density ratio. E_b/N_0 is closely related to the carrier-to-noise ratio (CNR or C/N), i.e. the signal-to-noise ratio (SNR) of the received signal, after the receiver filter but before detection:

$$C/N - E_b/N_0 \cdot \frac{f_b}{B}$$

Where f (b) is the channel data rate (net bitrate), and B is the channel bandwidth

Equalization

Equalization is the process of adjusting the balance between frequency components within an electronic signal. The most well-known use of equalization is in sound recording and reproduction but there are many other applications in electronics and telecommunications. The circuit or equipment used to achieve equalization is called an equalizer. These devices strengthen (boost) or weaken (cut) the energy of specific frequency bands. In telecommunications, equalizers are used to render the frequency response—for instance of a telephone line—flat from end-to-end. When a channel has been "equalized" the frequency domain attributes of the signal at the input are faithfully reproduced at the output. Telephones, DSL lines and television cables use equalizers to prepare data signals for transmission. In telecommunications, equalizers are used to render the frequency response for instance of a telephone line flat from end-to-end. When a channel has been "equalized" the frequency domain attributes of the signal at the input are faithfully reproduced at the output. Telephones, DSL lines and television cables use equalizers to prepare data signals for transmission.

In the field of audio electronics, the term "equalization" has come to include the adjustment of frequency responses for practical or aesthetic reasons, often resulting in a net response that is not truly equalized. The term EQ specifically refers to this variant of the term [1]. Stereos typically have adjustable equalizers which boost or cut bass or treble frequencies. Broadcast and recording studios use sophisticated equalizers capable of much more detailed adjustments, such as eliminating unwanted sounds or making certain instruments or voices more prominent.

Equalizers are critical to the successful operation of electronic systems such as analog broadcast television. In this application the actual waveform of the transmitted signal must be preserved, not just its frequency content. Equalizing filters must cancel out any group delay and phase delay between different frequency components.

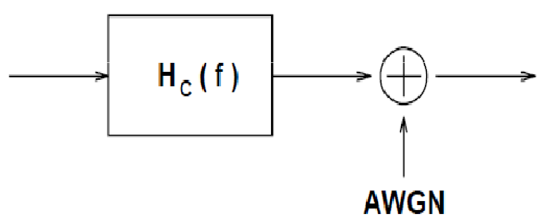

Fig.1 Dispersive Cannel Model

Principle of Equalization

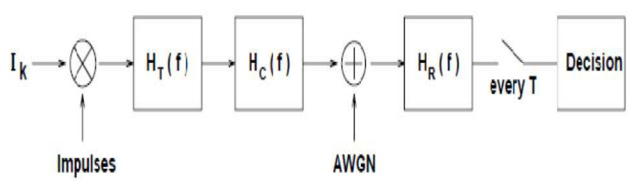

Fig.2 Communication through Equalizer.

The channel response of dispersive channel is given by H [C (f)] and H [T (f)] is the transfer function of the transmitting filter and H[R (f)] is the transfer function of the receiving filter. In order to construct the transmitting and receiving filters in advance we need to know the channel response.

Now we have to fix the values of the

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transmitting filter and choose that value of the receiving filter in such a way that the following condition is satisfied,

$$x(t) = \frac{\sin(\pi t/T)}{\pi t/T}\frac{\cos(\pi\alpha t/T}{1-4\alpha^2 t^2/T^2}$$

What we want eventually are the samples at intervals T at the receiver. Therefore, we

May choose to build a simpler (practical) filter $H_R(f)$, take samples at intervals T, and put a digital filter, called equalizer, at the output to eliminate ISI as shown below in the figure.

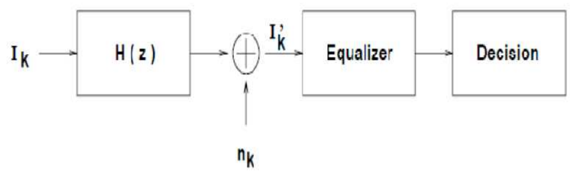

Fig.3: Equivalent discrete time channel communication system model with colored noise.

The main advantage of this approach is that a digital filter is easy to build and is easy to alter for different equalization schemes, as well as to fit different channel conditions. Here we are considering two equalization schemes one is zero forcing equalizer the other one is Minimum Mean Square Error Equalizer and then we compare these two in order to explain which one is best.

Zero Forcing Equalizer

First, let us consider the use of a linear equalizer, i.e., we employ an LTI filter with

Transfer function H [E (z)] as the equalizing circuit. The simplest way to remove the ISI is to choose H [E (z)] so that the output of the equalizer gives back the information sequence, i.e., b (Ik) = Ik for all k if noise is not present. This can be achieved by simply setting the transfer function H [E (z)] = 1=G (z). This method is called zero-forcing equalization since the ISI component at the equalizer output is forced to zero the corresponding impulse response f [h (E)]; k (g) can be an infinite length sequence. Suitable truncation and delay is applied to get an approximation. We note that the effect of the equalizing filter on the noise is neglected in the development of the zero-forcing equalizer above. In reality, noise is always present. Although the ISI component is forced to zero, there may be a chance that the equalizing filter will greatly enhancing the noise power and hence the error performance of the resulting receiver will still be poor. To see this, let us evaluate the signal-to-noise ratio at the output of the zero-forcing equalizer when the transmission filter HT (f) is fixed and the matched filter issued at the receiving filter.

Hr(f) = H*t (f) H*c(f)
In this case it is easy to say that the digital filter H (z) is given by:

H ($e^{j2\Omega fT}$) = 1/T ∑ = [H t (f-n/T) H c (f-n/T)] 2

N= -∞ to ∞

The noise filtering H [w (z)] can be chosen as H [w ($e^{j2\Omega fT}$)] = 1/√H [w ($e^{j2\Omega fT}$)].

Since the zero-forcing filter simply inverts the effect of the channel on the original

Information symbols Ik, the signal component at its output should be exactly Ik. If we model the Ik as iid random variables with zero mean and unit variance, then the PSD of the signal component is 1 and hence the signal energy at the output of the equalizer is just

$$\int_{-1/2T}^{1/2T} df = 1/T.$$

MMSE Equalizer

The zero-forcing equalizer, although removes ISI, but may not give the best error performance for the communication system as it does not take into account noises in the system. A different equalizer that takes noises into account is the minimum mean square error (MMSE) equalizer. It is based on the mean square error (MSE) criterion [7].Without knowing the values of the information symbols I_k beforehand; we model each symbol I_k as a random variable. Assume that the information sequence {Ik} is WS S.

We choose a linear Equalizer H [E (z)] to minimize the MSE between the original information symbols I_k and the output of the equalizer $\check{I}_k$

MSE = E[e^2k] = E [Ik-Ǐk]

Let us employ the FIR filter of order 2L+1 as the equalizer. We note that a delay of L symbols is incurred at the output of the FIR filter. Then

$$MSE = E\left[\left(I_k - \sum_{j=-L}^{L}\widehat{I}_{k-j}h_{E,j}\right)^2\right] = E\left[\left(I_k - \widehat{I}_k^T h_E\right)^2\right]$$

Where

$$\widehat{I}_k = \left[\widehat{I}_{k+L}, \cdots, \widehat{I}_{k-L}\right]^T$$

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$$h_E = [h_{E,-L}, \cdots, h_{E,L}]^T$$

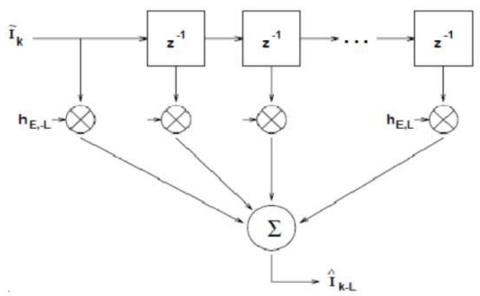

Fig 4 FIR filter as an MMSE Equalizer

We want to minimize MSE by suitable choices of:
H [E], −L, · · ·; H [E],L.
Differentiating with respect to each H [E (j)] and setting the result to zero, we get

$$E\left[\left(I_k - \tilde{I}_k^T h_E\right)\right] = 0$$

$$\mathbf{R}\mathbf{h}_\mathbf{E} = \mathbf{d}$$

Where

$$\mathbf{R} = E\left[\tilde{\mathbf{I}}_k \tilde{\mathbf{I}}_k^T\right]$$

$$\mathbf{d} = E\left[\mathbf{I}_k \tilde{\mathbf{I}}_k\right]$$

If R and d are available, then the MMSE equalizer can be found by solving the Equation matrix. It can be shown that the signal-to-noise ratio at the output of the MMSE equalizer is better than that of the zero-forcing equalizer. The linear MMSE equalizer can also be found iteratively. First, notice that the, MSE is a quadratic function of hE. The gradient of the MSE with respect to hE gives the direction to change hE for the largest increase of the MSE. In our notation, the gradient is ¡ (d ¡ RhE). To decrease the MSE, we can update h E in the direction opposite to the gradient. This is the steepest descent algorithm: At the kth step, the vector h E (k) is updated as

$$\mathbf{h}_E(k) = \mathbf{h}_E(k-1) + \mu[\mathbf{d} - \mathbf{R}\mathbf{h}_\mathbf{E}(k-1)]$$

Where,
u is the small positive constant that controls the rate of convergence to the optimal solution.

Simulation Results

The method which is used to filter the received signal to cancel the ISI introduced by the channel impulse response is generally known as equalization.
Design systems such that the impulse response is short enough that very little Energy from one symbol smears into the next symbol. Applying an equalizer at the receiver, that, broadly speaking, attempts to undo the effect of the channel by applying an inverse filter. Here we have compared two equalizers with a delay of 5 taps in both the cases and the result is shown on a graph in which y-axis is logarithmic scales and x-axis is plane scale.

In the first view, we look towards the effect of ISI on the received signal with some noise without applying the ZF equalizer at the receiver. The graph obtained in this case is as shown below. This graph is the theoretical view which is impossible to exist.

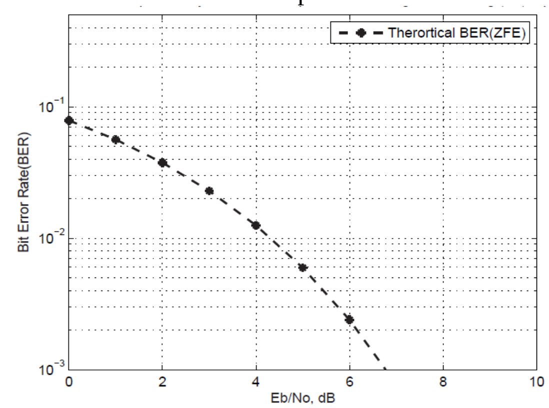

Fig.5 Theoretical Bit Error Rate probability curve.

In the second view we look towards the effect of ISI on the received signal with some noise with using Zero Forcing Equalizer at the receiver end .The graph obtained is a simulated in nature ,as shown in the figure below.

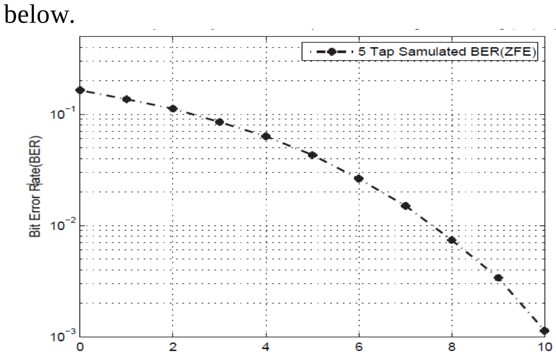

Fig 6 Bit error probability curve for Compensate ISI using Zero Forcing (ZF) equalizer.

In third view we can see that if we combine both the above mentioned graphs we see that after applying ZF Equalizer the BER VS Normalized SNR i.e.

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E_b/N_0 which can be explained as the energy per bit to noise power spectral density ratio, is better than with applying zero forcing equalizer, as to match the curve obtained from ZF Equalizer to the theoretical curve.

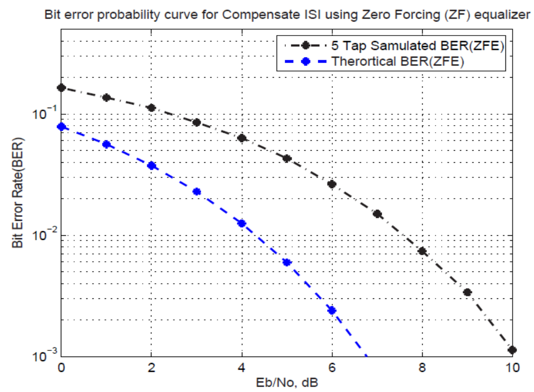

Fig 7 Bit error probability curve for Compensate (ISI)

Now in the fourth view we show bit error probability curve without using any equalizer.

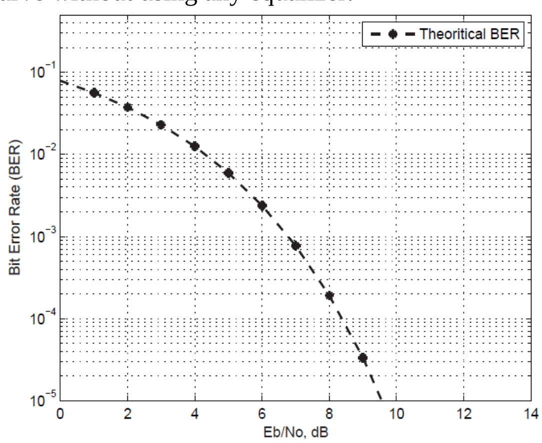

Fig.8 Theoretical Bit error probability curve for Compensated (ISI).

Now in fifth view we see the simulation curve for ZF Equalizer & MMSE Equalizer.

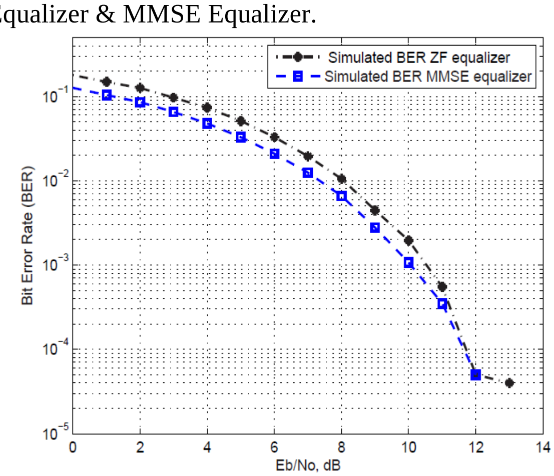

Fig.9 Simulated Bit error probability curve for Compensate (ISI) using MMSE equalizer and ZF equalizer.

In the given below graph we show the view in which we explain which equalizer is better, thus higher is the SNR i.e. if signal to noise ratio will be higher than the better will be BER. Thus SNR is one variable that determines what the likelihood of receiving correct data is. Thus from the given below we can say the curve line blue is matching with the red which is the theoretical BER which is impossible to exist in the case of MMSE equalizer than the black curve which is the BER for ZF. Thus BER is less in the case of MMSE as compared to ZF.

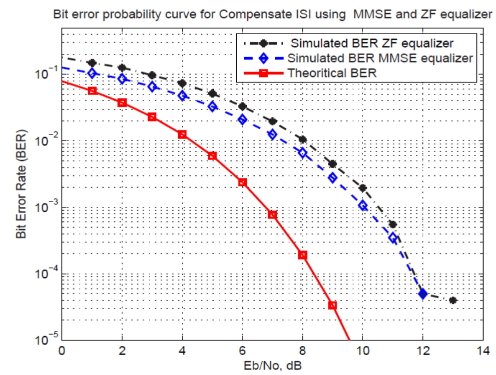

Fig: 10 Bit Error Probability curve for showing the comparison between Zero Forcing Equalizer.

Conclusion

Thus the Effect of Inter Symbol Interference over optical communication is studied and analyzed by using various equalization techniques such as zero forcing equalizer & minimum mean square error equalizer .We are using multipath channel with 5-tap . The received signal gets corrupted by Additive White Gaussian Noise (AWGN). Here we used Zero Forcing Equalizer and Minimum Mean Square Error Equalizer which reduces the effect of ISI introduced by the channel impulse response.

Zero Forcing Equalizer removes ISI but it ultimately maximizes the signal to noise ratio at the output of the equalizer which allows the signal to noise ratio to maximize at the output of the equalizer it is ideal only when the channel is noise free. When the channel is noisy the zero forcing equalizer amplifies the noise at a certain frequency f where the channel response has a very small magnitude. Thus in this way we can say Zero forcing Equalizer is good for noise free channel whereas MMSE i.e. minimum mean square error equalizer is more organized type of equalizer though it is not completely able to remove ISI but instead it minimizes

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the power associated with the noise and ISI components present in the output.

References

[1] Theodore S. Rapport, “Wireless Communications”, 2nd Edition, Prentice Hall of India, 2002.
[2] A John and C Bingham, “Multicarrier modulation for data transmission: An idea whose time has come”, IEEE Communication. May, vol: 28, no.5, pp.5-14, 1990.
[3] John G. Proakis, “Digital Communications”, 4th Edition, McGraw Hill Press, 2001.
[4] R. van Nee and R. Prasad, OFDM for Wireless Multimedia Communications, Artech House, 1999.
[5] Kevin Buchs, Pat Zabinski, and Jon Coker.” Basic Bit Error Rate Analysis For Serial Data Links”, Mayo-R-04-07-R0 June 3, 2004.
[6] Bhattacharya ,M.; Chattopadhyay, T. ”Influence of adjacent channel interference on the frequency-modulated WDM optical communication system” ,Light wave TechnologyJournal,Vol:17,Issue:12,Pag:2516-2519,1999.
[7] Wong and Lok, ”Theory of Digital Communications”, University of Florida.
[8] “Zoltowski M.D ; Balakrishnan ,V “ Trade-off between SNR and ISI in multi-channel zero-forcing equalizers, Statistical Signal and Array Processing, 1998 Proceedings.
[9] Anisul Karim and Ziauddin M. Kamran,”Optimal trade-off between SNR and ISI in Digital Communication System”, Asian journal of Information Technology, year 2004, Volume: 3, Issue: 11, Page 1.