SIMULATION OF BER AND SNR OF LINEAR AND NON-LINEAR EQUALIZER

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Abstract Signal integrity has been an important topic for many years and will continue to be an important topic as data rates keep increasing. Signals have to travel through various interconnects inside a system to reach their destination so any electrical degradation induced at the transmitter, connectors, traces, cabling, and the receiver will have an effect on the timing and quality of the signal. These waveform distortions are caused by impedance mismatches like stubs and vias, frequency dependent attenuation, and electromagnetic coupling between signal traces (crosstalk).

Equalization is a signal conditioning technique in which a waveform is manipulated either at the transmitter, at the receiver, or by a signal conditioner somewhere throughout a link in order to compensate the distortions due to a channel. The scope of this paper is to examine Linear Equalizers and Non-Linear Equalizers and the benefits that it brings to Communication. In this paper the SNR and BER performance of different equalizers have been analyzed on the MATLAB platform. The simulation results suggests that in certain cases the maximum likelihood equalizer performs better with lower BER and high signal to noise ratio when compared to Linear equalizers and decision feedback equalizer.

***Key words*:** Equalizer, SNR, BER, Intersymbol interference (ISI), ZFE, MMSE-LE, DFE

1. INTRODUCTION

In broad sense, the term communications refers to the sending, receiving and processing of information by electronic means. It is the technique of transmitting a message, from one point to another, knowing how much information, if any, is likely to be lost in the process [1][2].

Hence, the term "communication" is covered all forms of distance communications including radio, telegraphy, television, telephony, data communication and computer networking. Communications started with wire telegraphy in the eighteen forties, developing with telephony some decades later and radio at the beginning of this century. More recently, the use of satellites and fiber optics has made communications even more widespread, with an increasing emphasis on computer and other data communications [1][3]. A channel is said to be non-distorting or ideal if, within the Bandwidth occupied by the transmitted signal, amplitude response is constant and phase response is a linear function of frequency. On the other hand if this is no so then the channel distorts the signal. As a result of this distortion, successions of pulses transmitted through the channel at rates comparable to the bandwidth are smeared to the point that they are no longer distinguishable as well-defined pulses at receiving terminal. Instead, they overlap, and thus we have intersymbol interference (ISI) [4]. Noise and distortion are the main limiting factors in communication and measurement systems. Therefore the modeling and removal of the effects of noise and distortion have been at the core of the theory and practice of communications and signal processing. Noise reduction and distortion removal are important problems in applications such as cellular mobile communication, speech recognition, image processing, medical signal processing, radar, sonar, and in any application where the signals cannot be isolated from noise and distortion [5]. Equalization techniques have attracted attention in wireless communications, because it offers Noise reduction and distortion removal and thereby improving the quality of signal received. And a higher spectral density (more bits per second per hertz of bandwidth) is achieved. Because of these properties, Equalizers are the important part of modern wireless communication system.

2. BACKGROUND

The equalizers have been evolved from the need to minimize transmission losses on long distance telephone lines [3]. In about 1901, as the telephone industry was expanding, it was realized that the loss in level and reduced frequency response of the long line telephone circuits could be improved by the insertion of series inductance, at critical intervals, into the long toll lines. By 1915, loaded lines were common and, with the introduction of vacuum tube repeater amplifiers, transcontinental telephone service became a reality. By the late 1920's, the new- fangled radio craze was in full bloom and improvements in amplifier design made it possible to improve both radio transmission and receiver audio quality. At the same time it became necessary to transmit radio programs by land line in support of the expanding radio networks which were being set up. In order to provide the radio interests with quality nation-wide program distribution circuits, the Western Electric gang began developing "program line filters" to "equalize" the otherwise poor frequency response given by the loaded toll lines [2][3]. The equalizers were all passive networks using RCL components to shape the frequency response of the line and get it as "flat" as possible. Amplifiers were used to "make up" for the inherent insertion loss of the filter networks.

The overall result was quite impressive, by the mid 1930's it was possible to have a radio circuit from New York to Los Angeles with a frequency response of 50 to 8000 Hz with a signal to noise ratio better than 50 dB. The broadcast and recording industry soon found use for the many passive filter circuits being developed by the telephone researchers. By the mid 1940's there were equalizers being offered as program filters to be used for sound effects and other program production work. As stereo and multi-track recording entered the field, the use of equalization to perk up the tracks and mix bus became common [6]. So the humble equalizer we use today to tweak up our tracks and mixes is the result of over a hundred years of fussing around, trying to get the sound we put into one end of the pipe to come out the other end sounding good.

3. EQUALIZER PERFORMANCE IN COMMUNICATION

An Equalizer is a compensator for Channel Distortion. For communication channels in which the channel characteristics are unknown or time-varying, optimum transmit and receive filters cannot be designed directly. For such channels, an equalizer is needed to compensate for the ISI created by the distortion in the channel [4][7].

The goal of equalizers is to eliminate the intersymbol interference (ISI) and the additive noise as much as possible. Inter symbol interference arises because of the spreading of a transmitted pulse due to dispersive nature of the channel, which results in overlapping of the adjacent pulses. In the above figure there is four level PAM signal x (t). This signal is transmitted through the channel with impulse response h(t). Then noise n(t) is added. The received signal r(t) is a distorted signal [8].

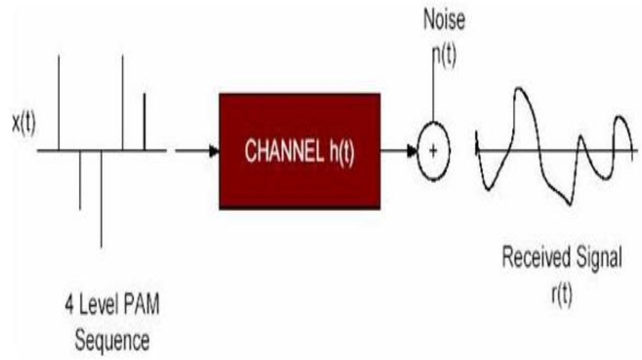


Fig.1: block diagram of equalizer

3.1 Equalization techniques

There are three important types of equalization techniques commonly used:

- Linear Equalization - Suboptimal, but simple.
- Non-Linear Equalization (DFE) - for severe ISI channels.
- Maximum Likelihood (ML) Sequence Detection - Optimal, but Impractical.

Linear Equalizers are simple to implement and are highly effective in channels where the ISI is

not severe (like the wire line telephone channel). Most linear equalizers are implemented as a linear transversal filter [5][8].

4. LINEAR EQUALIZERS

The most common type of channel equalizer used in practice to reduce ISI is a linear transversal filter with adjustable coefficients [5].

$$c(z) = \sum_k c_k\, z^{-k} \qquad (1)$$

There is no feedback path for linear equalizer. The current and the past values of the received signal are linearly weighted by equalizer coefficients and summed up to produce the output [9].

$$c(z) = h^{-1}(z) \qquad (2)$$

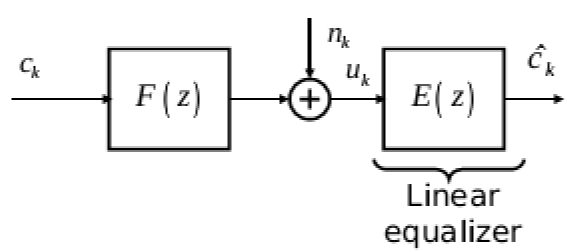


Fig.2: block diagram of linear equalizer

Now we have two different strategies used for linear equalization:

1) Design E (z) so that the ISI is totally removed: Zero-forcing (ZF)

2) Design E (z) so that we minimize the mean: MSE

Squared error $\epsilon_k = C_k - \hat{c}_k$

4.1 Zero-forcing (ZF) equalizers

The zero-forcing equalizers are designed to remove the ISI completely. As the name implies, it forces the ISI to become zero for every symbol decision.

Problems with Zero-Forcing Equalizers

A serious problem with the zero-forcing equalizer is the noise enhancement, which can result in infinite noise power spectral densities after the equalizer. A zero-forcing equalizer enhances noise and results in performance degradation. The noise is enhanced (amplified) at frequencies where the channel has a high attenuation [10]. Another, related, problem is that the resulting noise is colored, which makes an optimal detector quite complicated. By applying the minimum mean squared-error criterion instead, we can atleast remove some of these unwanted effects [11].

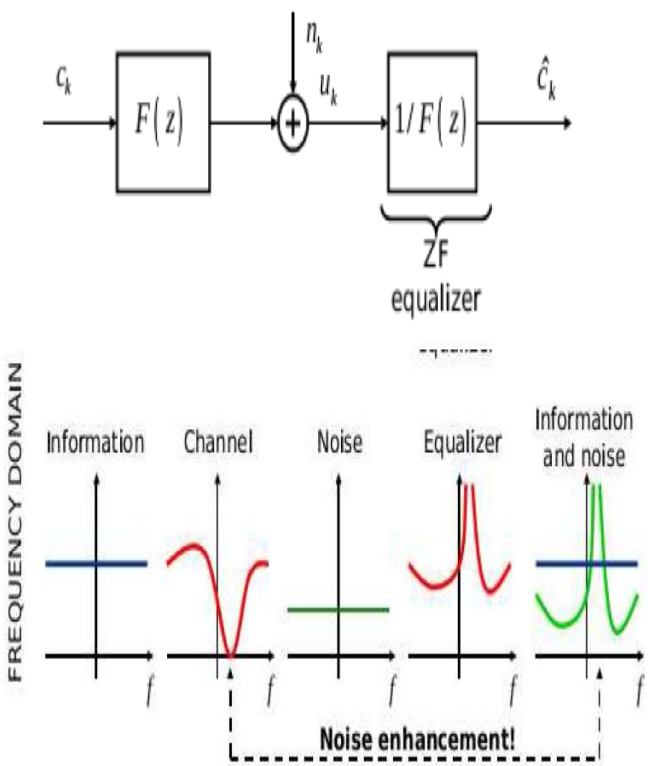


Fig.3 - Block diagram of Zero Forcing Equalizer and its waveforms

4.2 MMSE equalizers

The MMSE equalizer is designed to minimize the error variance.

In telecommunication, Intersymbol interference (ISI) is caused by multipath propagation. Therefore, the main aim is of designing the transmitter and receiver is to minimize the effects of intersymbol interference (ISI) [12]. The MSE equalizer removes the most problematic noise enhancements as compared to the ZF equalizer. The noise power spectral density cannot go to infinity any more. This improvement from a noise perspective comes at the cost of not totally removing the ISI. The noise is still colored after the MSE equalizer which, in combination with the residual ISI, makes an optimal detector quite complicated [8][11]. MMSE-LE minimizes the error between the received symbol and the transmitted symbol without Enhancing the noise.

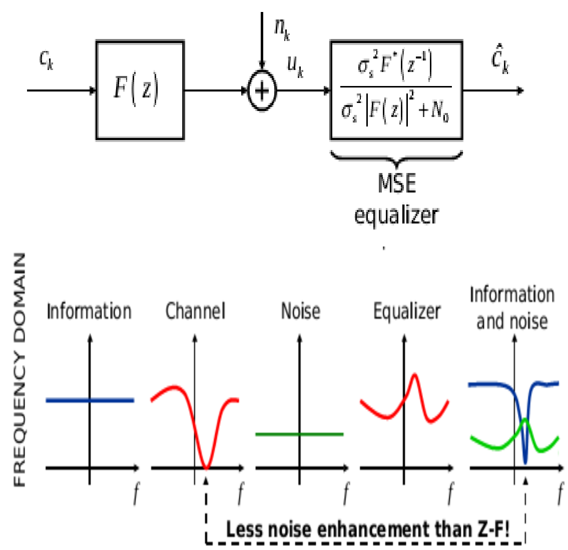


Fig.4: Block diagram of MMSE equalizer and its waveforms

Although MMSE-LE performs better than ZFE, its performance is not enough for channels with severe ISI. An obvious choice for channels with severe ISI is a non-linear equalizer. From the point-of-view of minimizing error probability, it is power. The MSE criterion attempts to minimize the total error between the slice input and the transmitted data symbol [12].

$$C(z) = \frac{1}{h(z) + \frac{N_O}{\sigma_A^2}} \qquad (3)$$

5. NON-LINEAR EQUALIZERS

Non-linear equalizers are used in applications where the channel distortion is too severe for a linear equalizer to handle, and are commonplace in practical wireless systems. Linear equalizers do not perform well on channels which have deep spectral nulls in the passband. In an attempt to compensate for the distortion, the linear equalizer places too much gain in the vicinity of the spectral null, thereby enhancing the noise present in those frequencies [13].

Based upon the importance, the non-linear equalizers are classified as:

1. Decision feedback equalizer(DFE)
2. Maximum-likelihood sequence estimation(MLSE)

5.1 Decision feedback equalizer (DFE)

A decision feedback equalizer (DFE) is a nonlinear equalizer that uses previous detector decision to eliminate the ISI on pulses that are currently being demodulated. In other words the distortion on a current pulse that was caused by previous pulses is subtracted. The non-linearity of the DFE is from the nonlinear characteristics of the detector that provides an input to the feedback filter. The basic idea of a DFE is that if the value of the symbols previously detected are known , then the ISI contributed by these symbols can be cancelled out exactly at the output of the forward filter by subtracting past symbol value with appropriate weighting [13][14][15].

In the design of a ZF-DFE, we want to completely remove all ISI before the detection. Like in the linear ZF equalizer, forcing the ISI to zero before the decision device of the DFE will cause noise enhancement. Noise enhancement can lead to high probabilities for making the wrong decisions. This in turn can cause error propagation, since we may add ISI instead of removing it in the decision-feedback loop. Due to the noise color, an optimal decision device is quite complex and causes a delay that we cannot afford, since we need them immediately in the feedback loop [15][16].

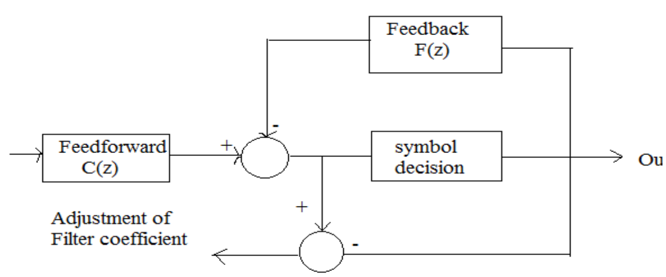


Fig.5: Simple Block diagram of DFE

The advantage of a DFE implementation is the feedback filter, which is additionally working to remove ISI, operates on noiseless quantized levels, and thus its output is free of channel noise.

A Zero Forcing (ZF) equalizer may provide adequate performance when the noise can be neglected, but at low to moderate signal-to-noise ratios the performance of an MMSE equalizer will be superior. Therefore, an MMSE equalizer is often preferred for practical implementation [16].

$$Q(K) = \sum_{I=(N-1)}^{0} c_j Y(k-i) - \sum_{I=1}^{M} f_i x(k-i) \quad (4)$$

5.2 Maximum likelihood sequence equalizer (MLSE)

Although the DFE outperforms a linear equalizer, it is not the optimum equalizer from the view point of minimizing the probability of error in the detection of the information sequence from the received signal samples. Maximum Likelihood Sequence Equalizer (MLSE) gives optimum performance [18][19][20]. It tests all possible data sequences and chooses the data with the maximum probability as the output. Generally, the Viterbi algorithm provides a solution to the problem with MLSE of a finite-state, discrete-time Markov process. However, the computational complexity of an MLSE increases with channel spread and signal constellation size. The number of states of the Viterbi decoder is expressed as ML, where M is the number of symbols in the constellation, and L is the channel-spread length [17][18].

The optimal equalizer, in the sense that it with the highest probability correctly detects the transmitted sequence is the maximum-likelihood sequence estimator (MLSE) [20].

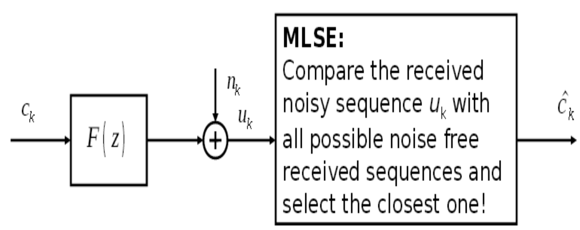


Fig.6: block diagram of MLSE equalizer

For sequences of length N bits, this requires comparison with 2^Ndifferentnoise free sequences.

Since we know the L+1 tap impulse response f_j, j = 0, 1, ... , L, of the channel, the receiver can, given a sequence of symbols {cm}, create the corresponding ¡°noise free signal alternative¡± as

$$u_m^{NF} = \sum_{j=0}^{L} f_j c_{m-j} \quad (5)$$

Where, NF denotes Noise Free.

The squared Euclidean distance (optimal for white Gaussian noise) to the received sequence {um} is

$$d^2(\{u_m\},\{u_m^{NF}\}) = \sum_m |u_m - u_m^{NF}|^2 = \sum_m \left|u_m - \sum_{j=0}^{L} f_j C_{m-j}\right|^2 \quad (6)$$

The MLSE decision is then the sequence of symbols $\{C_m\}$ minimizing this distance

$$\{\widehat{C_m}\} = arg_{\{\widehat{C_m}\}} min \sum_m \left|u_m - \sum_{j=0}^{L} f_j C_{m-j}\right|^2 \quad (7)$$

6. PERFORMANCE ANALYSIS AND COMPARISION USING MATLAB

Bit error rate or probability of bit error (BER)

BER is a performance measurement that specifies the number of bit corrupted or destroyed as they are transmitted from its source to its destination [21]. BER can also be defined in terms of the probability of error (POE). Each different type of modulation has its own value for the error function. This is because each type of modulation performs differently in the presence of noise [10].

The definition of bit error rate can be translated into a simple formula:

$$\textbf{BER} = \frac{\textbf{No.of Errors}}{\textbf{Total no of bits sent}}$$

Several factors that affect BER include bandwidth, SNR, transmission speed and transmission medium. **Signal-to-Noise density ratio (SNR)**

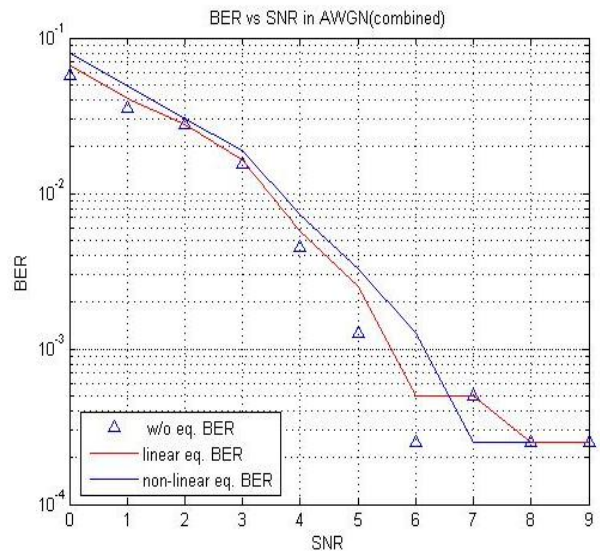
Signal to noise ratio (SNR) is a measure of the amount of signal divided by the amount of noise being received. SNR is mathematically expressed as:

$$\textbf{SNR = 10log10 } (E_b/N_o) \textbf{ dB}$$

In general a high Signal-to-Noise Ratio is good because it means we are getting more signal and less noise. SNR is usually measured using a logarithmic scale, meaning the SNR value is the logarithm of the actual ratio [22]. Generally, high noise (low SNR) can lead to high BER. High BER is bad, and usually leads to observable problems with the signal. It is important to note that probability of error (POE) is proportional to E_b/N_o and is a form of signal to noise ratio.

The basic parameters for the comparison of equalizers we have used here are SNR and BER [10]. The analysis shows the BER vs SNR performance of several types of equalizers in a static channel with a null in the pass band. The script constructs and implements a linear equalizer and a decision feedback equalizer (DFE) [23]. As the simulation progresses, it updates a BER plot for comparative analysis between the equalization methods. The relative burstiness of errors is shown in the plot, indicating that at low BERs, both the MLSE algorithm and the DFE algorithm suffer from error bursts.

As simulation progresses, the comparative plot between different equalizers is updated. Fig.7 shows the comparison plot between linear, non-linear and DFE equalizer. And it can be seen from the plot that as SNR or E_b/No increases, the BER of the Linear Equalizer decreases at a very slow manner. This shows that a linear equalizer requires many more taps to adequately equalize a channel with a deep null. Note that the DFE is much better able to mitigate the channel null than the linear equalizer, as shown in the BER plot.



Next the plot of maximum likelihood sequence equalizer is made between Eb/No for different parameter values. And it is observed that MLSE is even much better to mitigate the channel null as compared DFE equalizers.

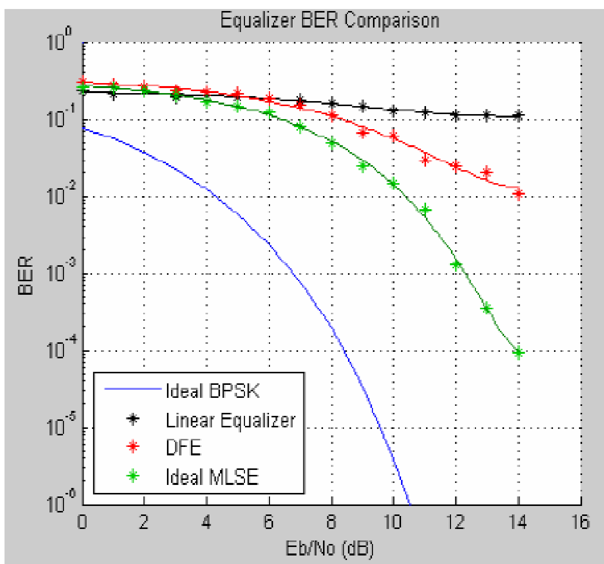


Fig.7: BER vs SNR of MLSE and of all equalizers

7. SUMMARY / CONCLUSION

- Linear equalizers suffer from noise enhancement.
- Decision-feedback equalizers(DFEs) use decisions on data to remove parts of the ISI, allowing the linear equalizer part to be less ”powerful” and thereby suffer less from noise enhancement.
- Incorrect decisions can cause error-propagation in DFEs, since an incorrect decision may add ISI instead of removing it.
- Maximum-likelihood sequence estimation (MLSE) is optimal in the sense of having the lowest probability of detecting the wrong sequence.

Equalizer Applications And Advantages In Communication

Equalizer Applications: Equalizers are built to control the loss and gain of frequencies within a sound system. This allows a sound system to sound natural and full. It also gives it the ability to maximize volume while eliminating feedback.

1. Equalizers are frequently used in public address systems to sharpen the sound and reduce echoes.
2. Churches, with their unusually angled rooms and ceilings will especially benefit from having an equalizer in the sound system.
3. Schools will want an equalizer to maximize sound output in various venues from auditoriums to gyms.
4. Bands and other live traveling shows will perhaps find the equalizer most useful, as it is nearly impossible to construct a good sound system for every venue without adjusting for frequencies that will create feedback.
5. Most studios have an equalizer, as it is very useful for coordinating the various microphones and sound inputs into the system. It can also reduce and eliminate ambient noises like air-conditioners that may hum in the background.

Advantages of Equalization: The advantages of equalization include simplification of design and verification, greater flexibility, greater reliability, reduced part count, improved testability, simplified production, reduced cost, and, especially, significantly superior sound quality.

1. **Simplification of Design and Verification** - Digital designs are less susceptible to noise and tend to be easier to layout than their analog counterparts.
2. **Greater Flexibility-** Given that the coefficients can easily be changed at any time, digital systems offer greater flexibility than their analog counterparts.
3. **Greater Reliability**- The performance of filters is independent of actual component values in the implementing circuit. Therefore, digital systems more reliably reproduce the desired filter responses inspite of temperature variations or component aging.

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References

[1]. Muhammad Sana Ullah and mohammed Jashim Uddin, “Performance Analysis of Wireless MIMO system by using Alamouti’s scheme and Maximum Ratio Combining Technique”, International journal of Advanced engineering science and Technologies, Vol No.8, Issue No. 1, 019-024.
[2]. B.Vucentic and Yuan, “Space-time coding”, England, Wiley, 2003.
[3]. D.T se and Viswanath, “Fundamentals of Wireless Communication”, Cambridge University Press, USA, 2005.
[4]. J.G. Proakis, “Digital Communications”, 3rd ed., McGraw-Hill, New York, 1995.
[5]. Lucky, *R.W.*, “Techniques for adaptive equalization of digital communication”. Bell Syst. Tech. J., 45, 255–286, Feb. 1966.
[6]. J.G. Proakis “Channel Equalization” Mobile Communications Handbook Ed. Suthan S. Suthersan.
[7]. Suresh, “Adaptive Equalization”, IEEE communications magazine, March 1992, pp. 9-16.
[8]. Chanpreet kaur, Rajesh Mehra, “High Speed and Cost Efficient Adaptive Equalizer for ISI Removal”, International Journal of Engineering Science and Technology (IJEST), vol. 3 no.1 Jan 2011.
[9]. S. Karthikeyan, S. Sasikumar “comparison of linear and non-linear multiuser detectors with quantum constraints” GESJ2010.
[10]. Liatul Ferdouse, Nasrin Akhter, Tamanna Haque Nipa and Fariha Tasmin Jaigirdar, “Simulation and performance Analysis of Adaptive Filtering Algorithms in noise cancellation”. IJCSI International Journal of computer science issues. Vol. 8, Issue 1, January 2011.
[11]. Ove Edfors, Department of Electrical and Information Technology.
[12]. H. V. Poor and S. Verdu, “Probability of error in MMSE Multiuser detection”, IEEE Transactions on Information Theory, May 1997, vol. - 43, no. 3, pp. 858-871.
[13]. D.A. George, R.R. Bowen, and J.R. Storey “An Adaptive Decision-Feedback Equalizer”, IEEE Transactions on

Communications Technology, June 1971 Vol. COM-19, pp. 281–293.
[14]. A.Z Duel-Hallen, "Decorrelating decision-feedback multiuser detector for synchronous code division multiple access channels", IEEE Transactions on Communications, February 1993, vol.41 pp.285-290.
[15]. P. Monsen "Feedback Equalization for Fading Dispersive Channels", IEEE Transactions on Information Theory, Vol. IT-17, pp. 56–64, January 1971.
[16]. G. Dheepak, A. Anandraj, G. Deena, "Blind Adaptive Equalization for channel Noise cancellation", International Journal of Engineering Science and Technology (IJEST), Vol.3 No.1 Jan 2011.
[17]. Lucky, *R.W.*, "Automatic equalization for digital communications". Bell Syst. Tech. J., 44,547–588, Apr. 1965.
[18]. Forney, *G.D*, "Maximum-likelihood sequence estimation of digital sequences in the presence of intersymbol interference". IEEE Trans. Inform. Theory, IT-18, 363–378, May 1972.
[19]. S.Verdu, "Optimum Multi-User Signal Detection". PhD thesis, University of Illinois at Urbana- Champaign, September 1984.
[20]. K.S. Schneider, "Optimum Detection of Code Division Multiplexed Signals, Transactions on Aerospace and Electronic Systems", January 1979, vol.15, pp 181-185.
[21]. M.S. Chavan, R.H. Chile, S.R. Sawant, "performance analysis and comparison of Adaptive Filters for Wireless Channel Equalization", International journal of emerging technology And Advanced Engineering Volume 2, Issue 5, May 2012
[22]. Vijaya chandran "EECS PROJECT II –Equalizer Simulation"
[23]. A Duel-Hallen, J. Holtzman, Z. Zvonar, "Multiuser Detection for CDMA Systems", IEEE Personal Communications, April1995, vol2, no 2, pp46-58.

