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A THESIS REPORT

ON

Adaptive Equalization of Fast-Time Varying Channel for EDGE System Enhanced by Dual Symbol Rate

By

Dinesh Twanabasu

(066/MSI/604)

Thesis Submitted to

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A thesis submitted in partial fulfillment of the requirements for the degree of
Master of Science in Information and Communication Engineering

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RECOMMENDATION

The undersigned certify that they have read and recommended to the Department of Electronics and Computer Engineering for acceptance, a thesis entitled "**Adaptive Equalization of Fast-Time Varying Channel for EDGE System Enhance by Dual Symbol Rate**", submitted by Dinesh Twanabasu in partial fulfillment of the requirement for the award of the degree of "**Master of Science in Information and Communication Engineering".**

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DEPARTMENTAL ACCEPTANCE

The thesis entitled "Adaptive Equalization of Fast-Time Varying Channel for EDGE System Enhanced by Dual Symbol Rate", submitted by Mr. Dinesh Twanabasu in partial fulfillment of the requirement for the award of the degree of "Master of Science in Information and Communication Engineering" has been accepted as a bona fide record of work independently carried out by him in the department.

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ABSTRACT

Our modern society has transformed to an information-demanding system, seeking voice, video, and data in quantities with high mobility that could not be imagined even a decade ago. High data rate high mobility creates challenges to the communication system. The challenge to the system is to conceive highly reliable system unaffected by the problems caused in the multipath fading wireless channels. Broadband single carrier modulated signals experience severe multipath distortion scrambling & ISI when propagating through physical medium. To mitigate the effect of channel a transmission burst includes a training sequence which is known to the receiver and depending upon the effect of the channel in the received training sequence the equalizer updated. In mobile communication due to the relative motion between the mobile equipment and network element the channel experienced is time varying. The properties of the equalizer used for tracking such time varying channel have to be time varying hence adaptive. This thesis focuses on fast time varying channel where each radio burst experiences varying channel response. However for fast fading channel where each radio burst experiences varying channel response a single training block in a radio burst may not be sufficient to track the channel. In this thesis two training sequence are used in a single burst of a DSR EDGE. A slight modification is done in the burst structure to include two training sequence keeping the number of symbol in a frame constant. A least mean square error based algorithm is used for equalization of fast time-varying channel. The simulation of the system was done in Simulink. The simulation of the communication system showed the equalization and estimation of the symbols using two training sequence has better performance than that with single training sequence used in DSR EDGE frame.

Keywords: DSR, EDGE, adaptive, equalization, fading, channel, Training.

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ABBREVIATIONS

2G:	Second Generation
3G:	Third Generation
GSM	Global System for Mobile
UMTS:	Universal Mobile Telecommunication System
LTE:	Long Term Evolution
CIR:	Channel Impulse Response
LMS:	Least Mean Square
NMLS	Normalized Least Mean Square
RLS:	Recursive Least Square
CMA:	Constant Modulus Algorithm
MSE:	Mean Square Error
DFE:	Decision Feedback Equalizer
GERAN	General Radio Network
3GPP	Third Generation Partnership Project
HSDPA	High Speed Downlink Packet Access
GPRS	General Packet Radio Service
BER	Bit Error Rate
PSK	Phase Shift Keying
SER	Symbol Error Rate
EDGE	Enhanced Data rate for GSM Evolution

CHAPTER 1: INTRODUCTION

1.1. Background

The development of Information and Communication technology in these days increases the data requirements of users. The rapid improvement in IC technology and their capability of processing digital signals made it possible to build portable terminals to support high data rate. The applications developed for using in these mobile systems are mainly focused in data. So the service providers are now focused in providing high speed data in these portable devices. In this context, the future trend will be toward wireless broadband services.

Wireless communication system is growing rapidly, first in the form of cellular networks for voice communication and more recently for data networks. Higher reliability is required for data networks than for voice communication. Information-bearing signals transmitted between the systems in remote locations often encounter a signal-altering physical channel. The physical channels may cause ISI, signal distortion, attenuation including echoes and frequency-selective altering of the transmitted signal.

To nullify the effect of signal distorting channel and equalizer is used at the receiver of the communication system. The successful employment of most high-data-rate transmission systems has been possible by the use of equalization that counteracts the disruptive effects of the channel on the signal as it propagates from the transmitter to the receiver.

Since it is common for the wireless channel characteristics to be unknown at startup or to change over time, training sequence is used for equalization. Depending upon channel response on the training sequence the characteristics of the channel is known. The change in the surrounding environment and relative motion between the mobile system and network element, the channel response is time varying. So the equalizer that is used for equalization of such time varying channel is a structure adaptive in nature. The general operating modes of equalizer include tracking and training. A fixed length training sequence is included in the transmission burst or a frame in a wireless communication. The training sequence is known to the receiver as well. Analyzing the effect of the channel on the training sequence receiver's equalizer adapt to a proper setting for estimation of the received signal with minimum bit error rate.

ENHANCED Data Rates for GSM Evolution was introduced to achieve higher data rates and spectral efficiency in global system for mobile communications and general packet radio service systems. In order to keep a backward compatibility with the second-generation cellular systems GSM, the EDGE has a similar burst (slot) structure and system parameters as GSM.

With the use of more advanced technology such as UMTS and LTE the mobile device became more data hungry by using the applications requiring high data. However such technologies, UMTS and LTE are feasible only in highly populated urban areas due to their limitation in the coverage area. They become highly costly to be deployed to location with difficult terrain with lightly populated areas like in Nepal. From economical point of view such areas can only be covered using 2G and 2.5G. In such case user experience high difference in the data rate and the data hungry applications become unusable when a user move from areas with UMTS and LTE to areas with EDGE only. So to minimize such situations the 2.5G EDGE system has to be upgraded for high speed data.

Methods like higher modulation, increasing symbol rate or used more than one time slot for single user etc are used to enhance the data rate of EDGE. In this thesis 8-PSK modulation with double symbol rate is considered and adaptive equalization is done for such condition. Conventional GSM/EDGE training sequence doesn’t provide good estimation of the channel for EDGE system enhanced by DSR since the channel memory is doubled [1]. Training sequences different from conventional training sequences have to be used for better estimation of channel.

1.2. Problem Definition

For static or slow moving communication devices, it is reasonable to assume that the fading channel is time invariant during a burst. This assumption is adopted in [3] and [4] where various channel estimation and equalization algorithms applied for the EDGE system with slow fading channels. For EDGE system enhanced by Dual Symbol Rate the memory of the channel is increased and a high relative motion between the mobile system and the network element causes fast fading channel response. In such case it is not so wise to assume that the radio burst experiences time invariant channel. In case when each radio burst experiences varying channel response a single training block in a radio burst may not be sufficient to properly track the channel characteristics. The channel is time varying so an equalizer used to track such time variant channel should be adaptive in nature. However the equalizer used should also be simple, convergent and less complex. In this thesis a least mean square error based method is used for adaptive equalization of the fast fading channel for DSR EDGE burst structure using two training sequence in a single burst.

1.3. Objectives

The main objectives of this thesis '**Adaptive Equalization of Fast Time Varying Channel for EDGE System Enhance by Dual Symbol Rate'** are:

- To study the characteristics of wireless channel and their effects on the transmitted signal.
- To study adaptive equalizers used to mitigate the effect of time varying wireless channels
- To implement adaptive equalizers in DSR EDGE using two training sequences in a frame for tracking fast time varying channel
-

1.4. Scope of the Work

The design of the adaptive equalizer for EDGE system enhanced by the Dual Symbol Rate for fast time varying channel a mobile user can communicate with less bit-error-rate even at high Doppler frequencies. The data rate of DSR EDGE system is twice and symbol duration is half that of EDGE system. The same single block of training sequence in a burst may not be suitable for fast moving mobile user to counteract the fast time varying channel. It is not feasible for such channel response to assume as time invariant for a radio burst used in EDGE. So by using different burst structure to analyze the channel more efficient adaptive equalization for DSR EDGE can be designed with less symbol error. In this thesis DSR EDGE frame structure has been used for analysis however this type of techniques can be used in other form of communication also.

1.5. Application

The world is moving towards broadband wireless communication with high mobility between the transmitter and the receiver. The use of adaptive equalizers is to keep track of varying channel characteristics. For high speed the symbol rate has to be increased and for high mobility better training sequence has to be used. High user mobility produces fast fading channel. In cases where the single burst of data experiences varying channel response two training sequences instead of a single training sequence will be more effective to keep the track of the channel. Different methods are used to increase the data rate in EDGE like using higher order modulation, decrease the symbol duration and increase symbol rate or use of more than one time slot for communication between mobile device and mobile network. DSR uses double the symbol rate of EDGE system to enhance the throughput. So the conventional GSM/EDGE training sequences may not be sufficient for adaptive equalization for the EDGE system Enhanced by DSR since the channel the channel memory is doubled. In such cases more than one training sequence in a single burst of data produces better estimation of the channel. This type of techniques is applicable in other wireless communication system experiencing fast time varying channel.

1.6. Thesis Organization

This thesis report is divided into six chapters. Chapter 2 is review of literature on Channel equalization and DSR EDGE. Chapter 3 deals with theoretical background of this thesis in which the theory behind channel equalization is presented. The brief explanation about wireless channels, equalization techniques and EDGE frame structure are also included in Chapter 3. In chapter 4 the detail of the methodology used to equalize the DSR EDGE frame structure is presented. The results of simulations and discussions are shown in chapter 5 and finally Chapter 6 concludes this thesis work.

CHAPTER 2: LITERATURE REVIEW

To move data at high rates across wireless transmission media with limited band-width efficient equalizer is required to counter act the impairment of the transmitted symbol. Since for mobile user the wireless channel is time varying hence the equalizer has to be adaptive in nature. An equalizer can be made adaptive to the channel only when the receiver knows the channel and this is possible by the use of training sequence. The training sequence used for the EDGE system is not suitable for EDGE system enhanced by DSR [1].

For static or slow moving communication devices, it is reasonable to assume that the fading channel is time-invariant during the period of one time slot. This assumption is adopted by [3] and [4], where various channel estimation and equalization algorithms are developed or applied for EDGE system with slow fading channels. The simulation results obtained in these references used maximum likelihood sequence estimation equalizer for channel estimation. However, the algorithms developed for slow fading channels cannot be directly applied to systems with high mobile speed subscribers, where the time-invariant channel assumption is not applicable.

The original burst structure of the EDGE has 26 training symbols in the middle and DSR EDGE has 52 training symbols in each burst [1]. This burst structure is good enough to be used for estimating the time-invariant channel state information of the entire burst. However, when the mobile subscriber is moving fast, the Doppler frequency is high, and the fading within one burst is no longer constant. Under these conditions, the original burst structure of the EDGE system is not sufficient for estimating the time-varying CIR with a reasonable accuracy [2]. S.T Leong, J. Wu and C. Xiao in [2] used two training sequence in EDGE. The two training sequences are placed at either side of a block of data to produce better estimation of time varying wireless channel.

A. Kundu, B. K. Sarkar, and A. Chakraborty in [9] implemented LMS algorithm in adaptive channel equalization of channel. Comparative Study of LMS and NLMS Algorithms in Adaptive Equalizer is done in [10] which showed NMLS performs better than LMS with little higher complexity. Different methods are mentioned in [12] for enhancing the data rate of EDGE one of which is to increase the data rate i.e. DSR EDGE.

The proposed adaptive equalization system uses a LMS algorithm to estimate the time varying channel on a modified DSR EDGE frame structure. The frame structure is modified such that the 52 training sequence used in the middle of the radio burst is split into two and used just in front of the two data blocks one at each block. This technique will make it possible to train the equalizer parameter twice even in a single radio burst which will enhance the channel estimation capability of the adaptive equalizer for fast time varying channel. The simulation of the system is carried out in Simulink. The simulation of the equalizer was carried out according to the simulation models defined in [13] which uses Rician and Rayleigh models for modeling the wireless channel characteristics.

CHAPTER 3: THEORETICAL BACKGROUND

3.1. Propagation Characteristics of Wireless Channel

In ideal case a wireless communication channel consists of a single line of sight radio path. A signal received would be a perfect reconstruction of the transmitted signal but in real scenario the transmitted signal experiences distortion due to propagation phenomenon like attenuation, reflection refraction and Doppler shift. The received signal is modified version of the transmitted signal. Noise is added to the signal. It is also attenuated, reflected, refracted, and diffracted by the channel. The signal also experiences a shift in the carrier frequency if the transmitter and the receiver are in motion relative to one another or with the surroundings. So the performance of the communication system depends upon these effects of the channel on the signal transmitted.

Attenuation defined as the drop in the power of signal when transmitting from one point to another caused by the transmission path length, obstructions in the signal path and multipath effects. Any objects, which obstruct the line of sight signal from the transmitter to the receiver, can cause attenuation. Shadowing of the signal can occur whenever there is an obstruction between the transmitter and receiver. Shadowing is generally caused by buildings and hills, and is the most important environmental attenuation factor. It is most severe in heavily built up areas, due to the shadowing from buildings. However, hills can cause a large problem due to the large shadow they produce. The diffraction of Radio signal from the boundaries of obstructions prevents the total shadowing of the signals behind hills and buildings. However, the amount of diffraction is dependent on the radio frequency used, with low frequencies diffracting more than high frequency signals. Thus, high frequency signals, especially, Ultra High Frequencies (UHF), and microwave signals require line of sight for adequate signal strength. To overcome the problem of shadowing, transmitters are usually elevated as high as possible to minimize the number of obstructions.

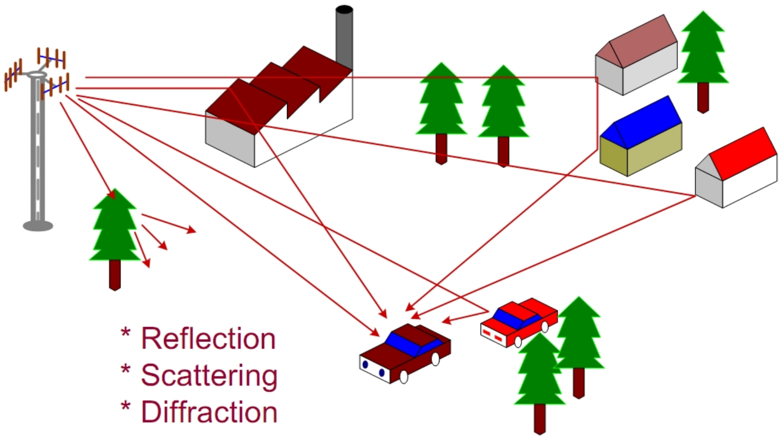

Figure 3.1 Propagation characteristics of wireless channel

The figure 3.1 taken from [6] shows different radio propagation phenomena that cause the signal to get distorted. Common propagation phenomena encountered in wireless communication are:

- Reflection: Reflection occurs when the electromagnetic waves are reflected from the objects with the physical sizes much greater than the wavelength of the electromagnetic waves.
- Diffraction: Diffraction occurs when electromagnetic waves hit an obstacle with surface irregularities such as sharp edges. It causes sharp changes in the propagation of the waves.
- Scattering: Scattering occurs when the electromagnetic waves encounter a cluster of objects smaller in size than the wavelength, such as water vapor. Scattering causes many copies of the wave to propagate in different directions.

Along with these there are other infrequent phenomena such as absorption and refraction that might take place in common wireless channels.

3.1.1. Large Scale Fading

Large-scale fading is also known as attenuation. It is characterized by the loss of the signal power usually with respect to long propagation distances. It results in the mean path loss of the signal. The average power of the received signal decreases logarithmically with the distance between the transmitter and the receiver. The attenuation caused by the distance is called large scale effect or path loss. The propagation medium and the environment would also have some effect on the total loss of the signal strength. The averaged received power at a certain distance from the transmitted is measured by keeping the distance to the transmitter constant (as the radius of a circle) and moving the mobile antenna on the circle [6].

The difference between the transmitted power *Pt* and the averaged received signal power *P(d)* (expressed in dBm) at certain distance is the path loss in dB, which is denoted by,

$$P(d) = Pt - L(d) \tag{3.1}$$

The average of the path loss in dB, with respect to a referenced distance d_0 at which the path loss is measured and is known, is given by:

$$L(d) = L(d_0) + 10_n log_{10}(d/d_0) \tag{3.2}$$

The order n has the constant value of 2 for LOS links but is usually higher than 2 for multipath channels in cities and urban areas [6].

3.1.2. Small Scale Fading

A receiver receives more than one version of the transmitted signals due to the multipath propagation. The signals from different directions arrive at different times at the receiver due to the difference in the path travelled by such signals.

Due to these multiple copies interference is induced. The interference induced by these multiple copies, also known as multipath waves, is the most significant cause of distortion in wireless communication with multipath. Such type of distortion is known as small scale fading and causes Inter Symbol Interference.

The radio signal experiences rapid changes of its amplitude over a relatively short period of time. The waves travelling different paths have travelled different distances. When these multipath waves sum up at the receiver antenna or antenna array ISI generated is of such a magnitude that the effects of large-scale path loss can be completely ignored by comparison. In most common multipath channels the ISI distortion effects are significant and dominate the channel noise. Such channels are said to be highly dispersive, and efforts are focused on eliminating ISI by equalization.

The behavior of the channel with respect to the signal bandwidth and channel delay spread the channel is divided as flat fading and non-flat fading.

When $Bs << Bc$ and $Ts >> \sigma_\tau$ the channel is known as flat fading or frequency-not-selective.

When $Bs >> Bc$ and $Ts << \sigma_\tau$ the channel is known as non-flat or frequency-selective.

Here Bs and Bc are bandwidth of the signal and the coherence bandwidth of the channel respectively. Ts is the symbol period of the transmitted signal and σ_τ is the delay spread of the channel. The delay spread is a measure of the difference between the time of arrival of the earliest significant multipath component i.e. the line of sight component and the time of arrival of the latest multipath component.

The mobility of the communicator originates Doppler shift in frequency, or simply some change in frequency due to the mobile velocity.

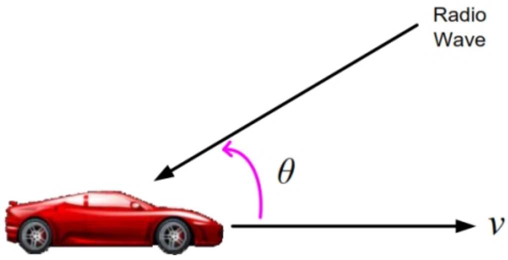

Figure 3.2 Relative motions between the mobile user and radio waves

When denoted by fd, the Doppler shift is computed as

$$fd = \frac{v}{\lambda} Cos\theta \qquad 3.3$$

Here v is the relative mobile speed, λ is the radio wavelength, and θ is the angle between the wave direction and the mobile direction. The change in frequency is positive when the mobile approaches the transmitter and negative when the mobile is departing. Higher the relative velocity higher is the Doppler shift.

The characteristics of the wireless channel can be categorized using the coherence time Tc, and the symbol time duration Ts as follows:

When $Ts<Tc$, the complete signal or symbol is affected similarly by the channel, and the channel is known as slow fading.

When $Ts>Tc$, different parts of a signal are affected differently because the channel changes faster compared to a symbol duration. Consequently, the channel is called fast fading.

The coherence time can be defined as the time for which the response of the channel for the bandwidth of interest is constant. So a wireless channels are divided into four types. Based on the delay spread, the channel is either flat (not frequency selective) or frequency selective, and based on Doppler spread the channel is known as slow or fast fading.

3.2. Channel Equalization

In wireless communication system when a signal is transmitted through a communication channel; the signal experiences a distortion when it is received by the receiver due to the effect of the channel. For efficient communication between the transmitter and the receiver the effect of such distorting channel has to be eliminated to acceptable level so that the receiver can recover the transmitted signal from the received signal. In communication system the effect of the channel is nullified or reduced using a filter called channel equalizer used at the receiver. So an equalizer is a filter used in communication system to reduce the effect of the channel in the signal transmitted so that the receiver can estimate the signal being transmitted with permissible accuracy.

In mobile communication the mobile network elements remains stationary but the mobile system and/or the surrounding that effect the signal is not stationary. Due to this relative motion the wireless channel response changes with time. Even when the mobile device and the network elements both are stationary the characteristics of the channel changes due to the change in the surroundings. It is very difficult for estimating both the channel order and the distribution of energy among the multipath signals. It is also very difficult to predict the effect of the environment on these multipath signals. Hence it is a must for the equalization process to be adaptive. The adaptive channel equalizer changes its characteristics to cope up with the changing characteristics of the channel. For proper equalization of the fast time varying channel the characteristics of the adaptive equalizer need to be updated very frequently with the changing environment.

The adaptive equalization is done using training sequence or without. Equalizer that uses training sequence which is known to the receiver is called training based adaptive equalizer. The equalizer which does not use training sequence is called blind equalizer.

Normally the training based adaptive equalization works in two phases. At first the equalizer is trained using a sequence of symbols known to both the transmitter and the receiver. Depending upon how the known symbols are affected by the channel the equalizer properties are updated. After training the weights and various parameters associated with the equalizer structure; it is frozen to function as a detector. These two processes are frequently implemented to keep the equalizer adaptive. We call the Equalizer is Frozen, if we keep the adaptable parameters of the equalizer constant [9]. In some cases after initial training of the equalizer the coefficient of the adaptive equalizer may be continually adjusted in a decision-directed manner.

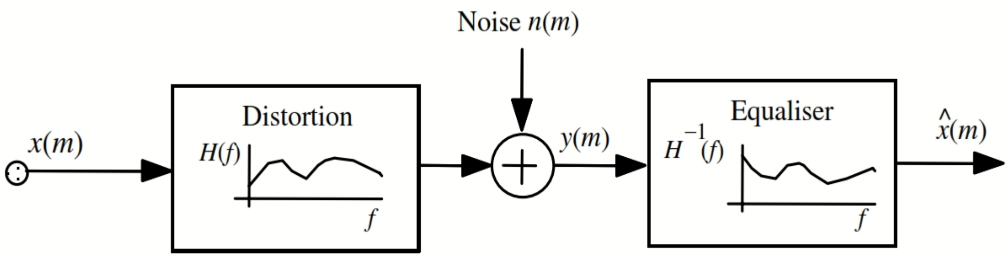

Figure 3.3 Channel effect and equalizer response

The figure 3.3 above illustrates a typical model for a distorted and noisy signal, followed by an equalizer. Let x(m), n(m) and y(m) denote the channel input, the channel noise and the observed channel output respectively. The relation between the channel input output relation can be expressed as

$$y(m) = h[x(m)] + n(m) \qquad 3.4$$

Where, the function h[·] is the channel distortion. In general, the channel response may be time-varying and non-linear. For time-varying and linear transversal filter model of the channel, Equation (3.4) becomes,

$$y(m) = \sum_{k=0}^{p-1} \mathrm{h_k(m)x(m-k)} + \mathrm{n(m)} \qquad 3.5$$

Where, $\mathrm{h_k(m)}$) are the coefficients of a P^{th} order linear FIR filter model of the channel. For a time-invariant channel model, $\mathrm{h_k(m)} = \mathrm{h_k}$.

In frequency domain, the above equation becomes

$$Y(f) = X(f)H(f) + N(f) \qquad 3.6$$

A training based equalization is used in this thesis. The following figure 3.4 shows the training based equalizer.

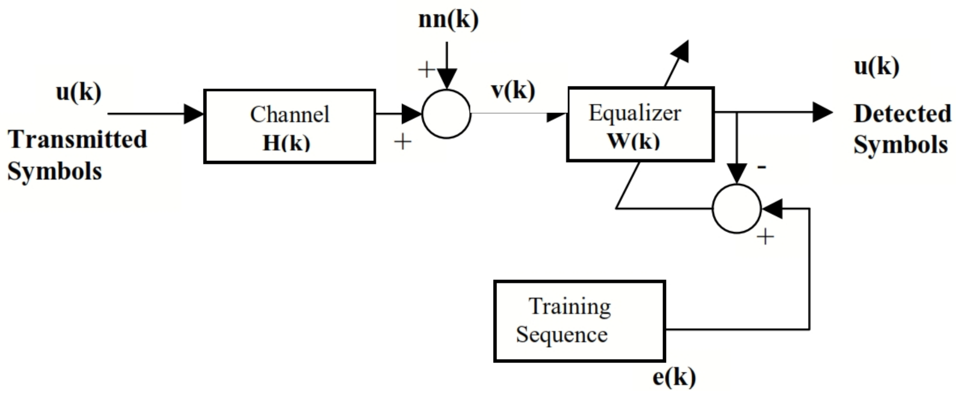

Figure 3.4 Training based channel equalization

In the above figure u(k) is the transmitted signal which is distorted by the channel having H(k) as channel response. Noise n(k) is also added to the signal which is received by the equalizer with w(k) response. The equalization system completes the two modes for the estimation of the transmitted signal. The two modes are training mode and decision directed mode.

The training mode helps to determine the appropriate coefficients of the adaptive filter used as equalizer. At first the training signal is equalized and the equalized signal is compared with the known training sequence in the receiver. The error between the estimated and actual training signal is calculated. Using the error an adaptive equalizer algorithm updates the coefficients of the adaptive filter. The coefficients are changed in such a way that the error is minimized.

After training mode the equalizer shifts into decision directed mode. In this mode the adaptive channel equalization system decodes the received signal and produces new signal, which is an estimation of the transmitted signal.

3.3. Channel Equalization Algorithms

3.3.1. Least Mean Square Algorithm

The Least Mean Square Algorithm, introduced by Widrow and Hoff in 1959 [8] is an adaptive algorithm, which uses a gradient-based method of steepest decent. LMS algorithm uses the estimates of the gradient vector from the available data. 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 nor does it require matrix inversions.

Least mean squares algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal. The error signal is the difference between the desired signal and the actual signal being received by the receiver.

Two practical features, simple to design, yet highly effective in performance have made the algorithm highly popular for adaptive equalization. LMS filter employ, small step size statistical theory, which provides a fairly accurate description of the transient behavior of the channel.

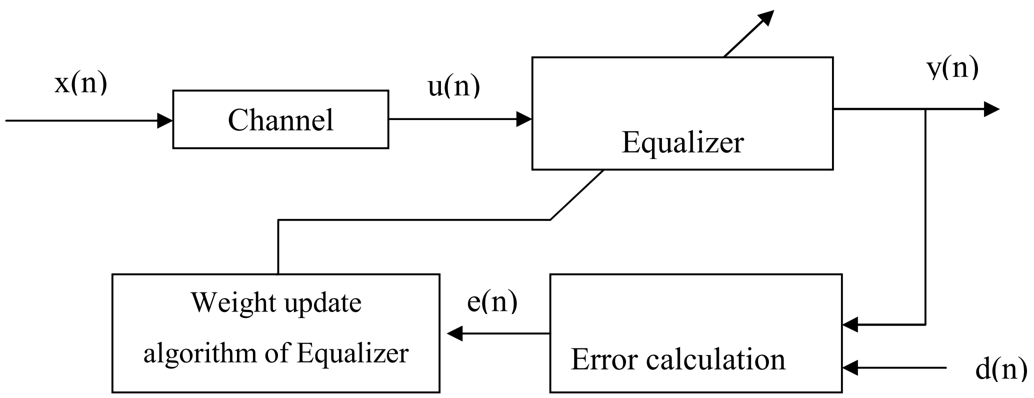

Figure 3.5 LMS algorithm based channel equalization

In the figure above x(n) is the transmitted signal from the transmitter. The signal is passed through the channel. The signal is distorted and the received signal by the receiver equalizer is u(n). The equalizer equalizes the received signal to produce estimated signal y(n). The desired signal d(n) i.e. the training sequence is then compared with the equalized signal to generate an error signal e(n) . The weights of the equalizer are updated using the error signal and the step size μ of the LMS algorithm. The step size μ of LMS algorithm determines the convergence characteristics of the equalizer.

From the method of steepest descent, the weight vector equation is given by,

$$w(n+1) = w(n) + \frac{1}{2}\mu[-\nabla(E\{e^2(n)\})] \qquad 3.7$$

Here μ is the step-size, $e^2(n)$ is the mean square error between the output y(n) from the equalizer which is given by,

$$e^2(n) = [d(n) - w^h x(n)]^2 \qquad 3.8$$

The gradient vector in the above weight update equation can be computed as

$$\nabla_w(E\{e^2(n)\}) = -2r + 2Rw(n) \qquad 3.9$$

The least mean square algorithm simplifies this by using the instantaneous values of covariance matrices r and R instead of their actual values i.e.

$$R(n) = x(n)x^h(n) \qquad 3.10$$

$$r(n) = d(n)x(n) \qquad 3.11$$

Therefore the weight update can be given by the following equation,

$$w(n+1) = w(n) + \mu x(n)[d(n) - x^h(n)w(n) \qquad 3.12$$

$$= w(n) + \mu x(n)e^*(n) \qquad 3.13$$

The LMS algorithm is initiated with an arbitrary value w(0) for the weight vector at n=0. The successive corrections of the weight vector eventually leads to the minimum value of the mean square error.

So the LMS algorithm consists of two basic procedures filtering and adaptive process.

Filtering process consists of computing the output of the linear filter

$$y(n) = w^h x(n) \qquad 3.14$$

in response to the input signal and generation of estimation error

$$e(n) = d(n) - y(n) \qquad 3.15$$

by comparing the output with a desired response.

Adaptive process, which involves the automatics adjustment of the parameter of the filter in accordance with the estimated error,

$$w(n+1) = w(n) + \mu x(n) e^*(n) \qquad 3.16$$

The LMS algorithm can be summarized in the following equations

Output, $y(n) = w^h x(n)$ 3.17

Error, $e(n) = d(n) - y(n)$ 3.18

Weight, $w(n+1) = w(n) + \mu x(n) e^*(n)$ 3.19

Here, $e^*(n)$ is the conjugate of error signal $e(n)$

The combination of these two processes working together constitutes a feedback loop, as illustrated in the block diagram of Figure above. The transversal filter, around which the LMS algorithm built, is responsible for performing the filtering process. The mechanism of updating the weight of the transversal filter performs the adaptive control process.

3.3.2. Recursive Least Square Algorithm

The Recursive least squares adaptive filter is an algorithm which recursively finds the filter coefficients that minimize a weighted linear least squares cost function relating to the input signals. This in contrast to other algorithms such as the least mean squares that aim to reduce the mean square error. In the RLS, the input signals are considered deterministic, while for the LMS and similar algorithm they are considered stochastic.

Compared to most of its competitors, the RLS exhibits extremely fast convergence. However, this benefit comes at the cost of high computational complexity, and potentially poor tracking performance when the filter to be estimated changes.

3.4. Adaptive Equalization Techniques

3.4.1. Symbol Spaced Equalizer

A symbol-spaced linear equalizer consists of a tapped delay line that stores samples from the input signal. The delay between one tap and the adjacent tap is the duration of one symbol. Once per symbol period, the equalizer outputs a weighted sum of the values in the delay line and updates the weights to prepare for the next symbol period. This class of equalizer is called symbol-spaced because the sample rates of the input and output are equal [5]. In this configuration the equalizer is attempting to synthesize the inverse of the folded channel spectrum which arises due to the symbol rate sampling of the input. In typical application, the equalizer begins in training mode which estimates the characteristics of the channel. The equalizer is then switched to decision-directed mode. In this algorithm new set of weights depends on the following quantities.

- The current set of weights
- The input signal
- The output signal
- For adaptive algorithms other than CMA, a reference signal d whose characteristics depend on the operation mode of the equalizer.

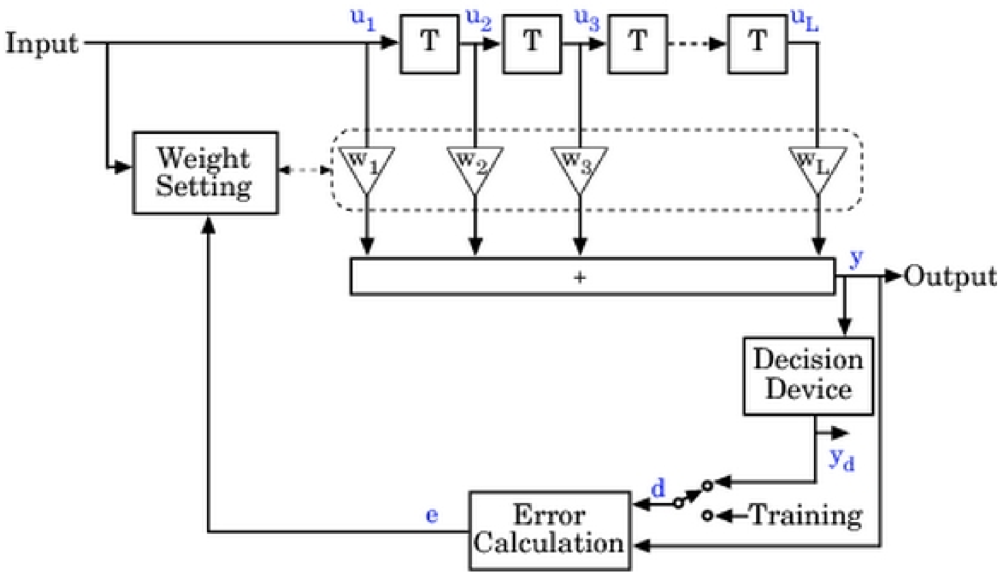

Figure 3.6 Symbol spaced equalizer

As shown in the figure above the input symbols is passed though an equalizer with delay between the one tap to other is T which is equal to the symbol period. Each delay taps are summed by multiplying each tap by using weights. The rate of output, y(n) is same as that of the input u(n) however delayed by the no of taps used in the equalizer.

3.4.2. Decision Feedback Equalization

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 figure 3.7 taken from [5] shows a simplified block diagram of a DFE where the forward filter and the feedback filter can each be a linear filter, such as transversal filter. The nonlinearity of the DFE stems from the nonlinear characteristic of the detector that provides an input to the feedback filter. The basic idea of a DFE is that if the values of the symbols previously detected are known, then ISI contributed by these symbols can be cancelled out exactly at the output of the forward filter by subtracting past symbol values with appropriate weighting. The forward and feedback tap weights can be adjusted simultaneously to fulfill a criterion such as minimizing the MSE. The DFE structure is particularly useful for equalization of channels with severe amplitude distortion, and is also less sensitive to sampling phase offset. The improved performance comes about since the addition of the feedback filter allows more freedom in the selection of feed forward coefficients. The exact inverse of the channel response need not be synthesized in the feed forward filter, therefore excessive noise enhancement is avoided and sensitivity to sampler phase is decreased. 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.

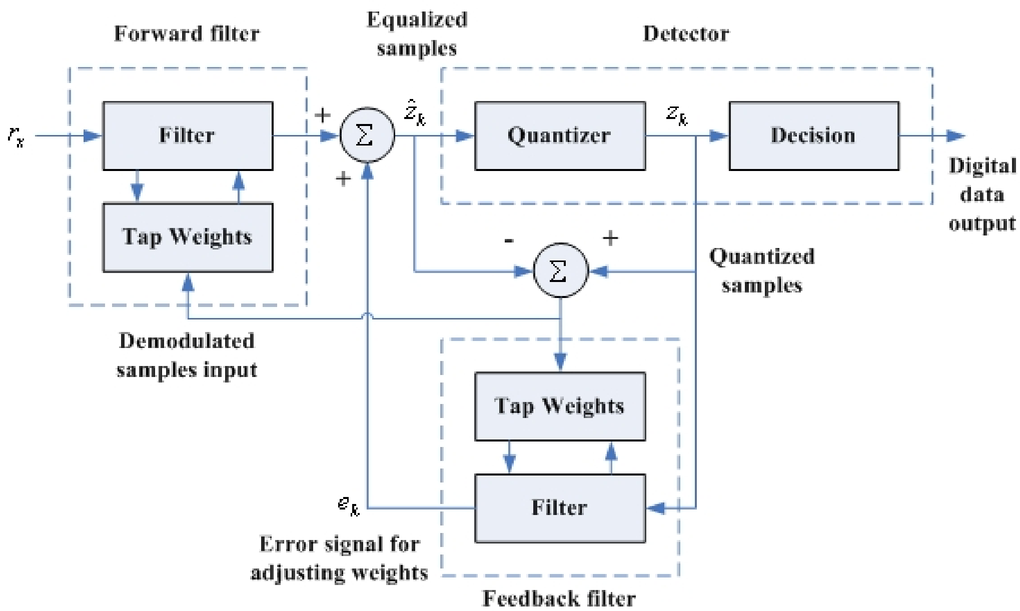

Figure 3.7 Decision-feedback equalization

One drawback to the DFE structure surfaces when an incorrect decision is applied to the feedback filter. The DFE output reflects this error during the next few symbols as the incorrect decision propagates through the feedback filter. Under this condition, there is a greater likelihood for more incorrect decisions following the first one, producing a condition known as error propagation. On most channels of interest the error rate is low enough that the overall performance degradation is slight [5].

3.4.3. Blind Equalization

Blind equalization is an equalization technique in which the transmitted signal is equalized from the received signal, while making use of only the transmitted signal statistics. Hence, the use of the word blind in the name. Blind equalization of the transmission channel equalizes the channel without the need of transmitting a training signal.

The equalizer reduces the mean-squared error (MSE) to acceptable levels. In the blind equalization algorithm, the output of the equalizer is quantized which is used to update the coefficients of the equalizer. However, the convergence property of this algorithm is relatively poor. In an effort to overcome the limitations of decision directed equalization, the desired signal is estimated at the receiver using a statistical measure based on a priori symbol properties; this technique is referred to as blind equalization.

In the Blind Equalization the error is selected as the basis for the filter coefficient update. The blind equalization directs the coefficient adaptation process towards the optimal filter parameters even when the initial error rate is large. For best results the error calculation is switched to decision directed method after an initial period of equalization, we call this the shift blind method. Referring to, the Reference Selector selects the Decision Device Output as the input to the error calculation and the Error Selector selects the Standard Error as the basis for the filter coefficient update. Constant Modulus Algorithm (CMA) and Multimodal's

Algorithm (MMA) are blind equalization algorithms [5]. The figure below shows a feed forward blind equalization.

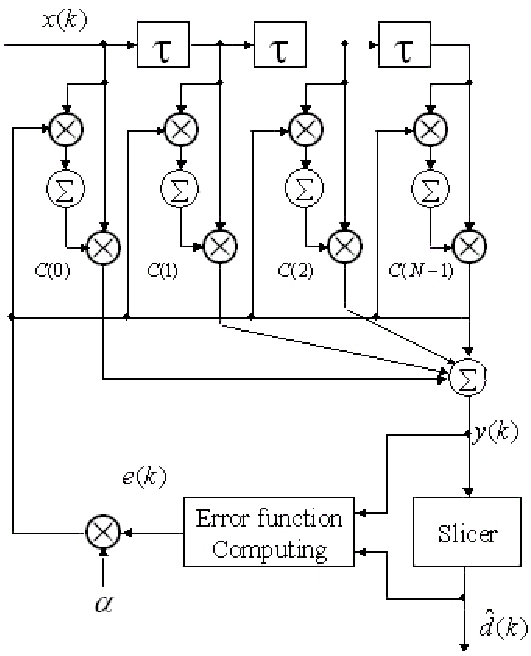

Figure 3.8 Blind equalization

3.4.4. Fractionally Spaced Equalizer

A fractionally spaced equalizer is a linear equalizer that is similar to a symbol spaced linear equalizer [5]. By contrast, however, a fractionally spaced equalizer receives say K input samples before it produces one output sample and updates the weights, where K is an integer. In many applications, K is 2. The output sample rate is 1/T, while the input sample rate is K/T. The weight-updating occurs at the output rate, which is the slower rate. Sometimes the input to the equalizer is oversampled such that the sample interval is shorter than the symbol interval and the resulting equalizer is said to be fractionally spaced. Equalizer Taps are spaced closer than the reciprocal of symbol rate.

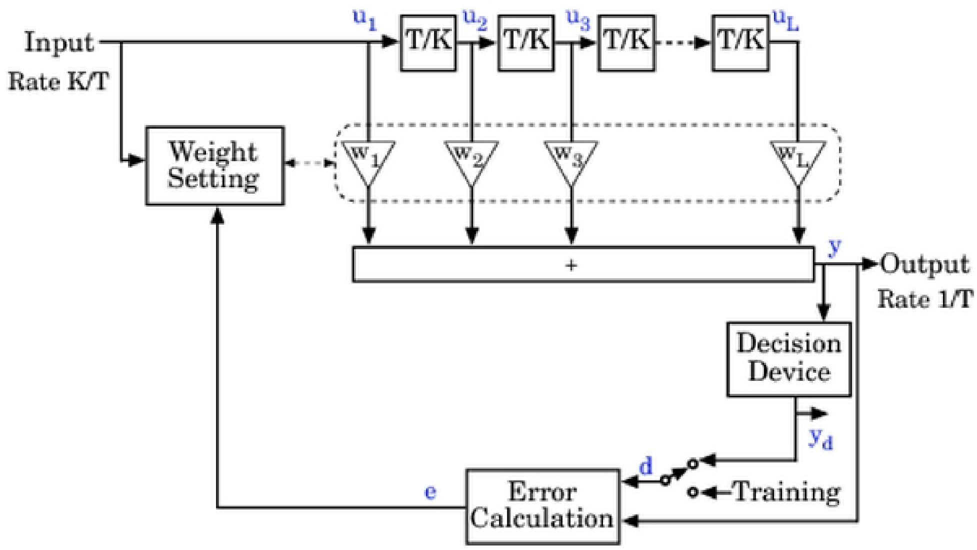

Figure 3.9 Fractionally spaced equalizer

3.5. Dual Symbol Rate EDGE

GSM/EDGE is a low cost mobile technology deployed across the globe. A review was undertaken by GERAN radio group in 3GPP to identify what kind of additional performance could be obtained in EDGE. Evolved EDGE refers to EDGE communication with enhanced performance [12]. The advantage of GSM/EDGE network is that it has very large coverage area compared to UMTS/HSPA or LTE. So it is deployed all over the world and is always has been the means of communication in areas with difficult terrain and developing countries. The UMTS/HSPA or LTE deployed will not provide full coverage in many geographical areas. However data hungry mobile applications that uses improved networks like UMTS, HSPA and LTE doesn't do well in non HSPA or LTE coverage areas. With the use of evolved EDGE the data speed of EDGE network is improved. So a user currently in an area of UMTS, HSPA and LTE moves to area without such coverage experiences severe drop in the data rate. With the enhancement in EDGE the data rate is increased to serve such areas with UMTS, HSPA and LTE. Enhancement in EDGE technology can be done in various ways. The rate disparity between HSDPA and GPRS means that many applications that may be enjoyed with HSDPA coverage may work very poorly or not at all if the mobile moves to an area with GPRS coverage only. If the mobile moves to an area with EDGE coverage, the rate change is slightly less severe but still may prohibit the use of some applications.

However with Evolved EDGE, the data rates are such that most applications will work with acceptable performance as the mobile station moves. It is logical to conclude that GSM will continue to be the baseline network technology for many years to come and that this technology will continue to be deployed and upgraded both in developing as well as developed markets. The figure below from [12] shows the speed comparison between different mobile technologies.

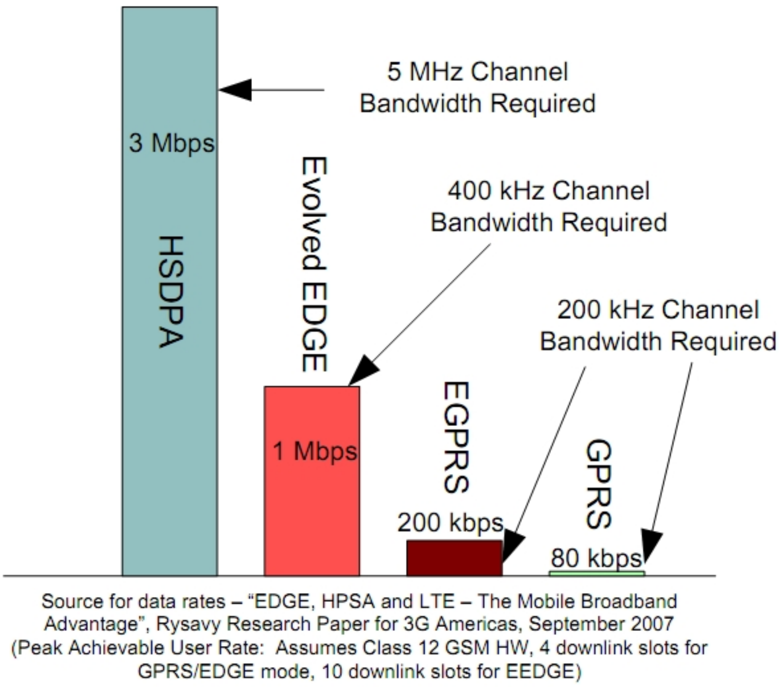

Figure 3.10 Data rate comparisons between mobile data services

Enhancement in EDGE communication can be done in different ways:

a. Using higher order modulation to 16PSK, 16QAM and 32 QAM
b. Increasing symbol rate. DSR uses 2X Symbol rate.

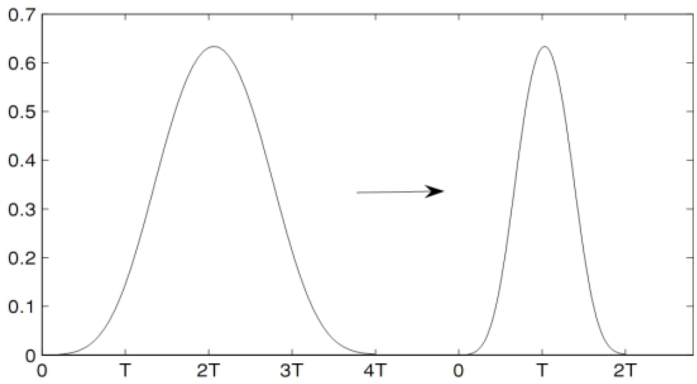

Figure 3.11 Decreased pulse duration for DSR EDGE

The figure above from [1] shows the duration of a dual symbol rate EDGE compared to the normal EDGE.

CHAPTER 4: METHODOLOGY

In this thesis wireless channel characteristics and their effect on the communication signal have been studied. Techniques that are used to mitigate the effect of such distorting channels were also studied. Different types of adaptive equalizers using different algorithms were analyzed. The use of training sequence simplifies the adaptive channel equalization. Two training sequences instead of single long training sequence is proposed in this thesis to estimate the channel with better accuracy. The simulation model was constructed using Simulink of Matlab. All the simulation are done using Simulink and Matlab version 2013. The design of the system using the required blocks of Simulink and the methodology used are described here under.

4.1. Generation of Signal

A frame structure of a DSR EDGE is generated using training sequence and data sequence. For simulation of normal DSR EDGE frame a single training sequence is placed between the two data generators. Here the length of the training sequence is 52 and the two data sequences are of 116 symbols each.

Similarly, for the simulation of modified DSR EDGE frame two training sequence are used. Each training sequence is placed in front of the data sequences. Here the length of each training sequence is 26 symbols and the two data sequences are of 116 symbols each, thus keeping the length of the symbols in normal frame structure and modified frame structure of the DSR EDGE equal.

A frame structure is created by concatenating the constant data which act as a training sequence and the random signal which acts as a data. In normal DSR, the frame structure is constructed by placing a constant sequence generator between two random signal generators. In modified DSR frame structure a random signal generator acting as data is placed after the constant sequence generator acting as training sequence.

a.

Trail (3)	Data(58)	Training Sequence (26)	Data(58)	Trail(3)

b.

Trail (6)	Data(116)	Training Sequence(52)	Data(116)	Trail(6)

c.

Trail (6)	Training Sequence (26)	Data(116)	Training Sequence (26)	Data(116)	Trail(6)

Figure 4.1 Burst Structure in (a).EDGE (b). DSR EDGE
(c). Proposed for DSR EDGE

4.2. Modulation

EDGE system enhanced by DSR uses 8-PSK modulation. So an 8-PSK baseband modulator is used to obtain the desired signal. The 8-PSK modulation is done using gray coding to reduce the bit error rate. An 8-PSK baseband modulator block from Simulink is used to generate modulated signal from the given input data.

4.3. Channel

For the simulation of the time varying channel a Rician fading channel and a Rayleigh fading channel are used one at a time. The channel models were constructed using COST 207 and GSM/EDGE channel models. The characteristics of the wireless communication channel depended upon the Doppler shift, the path delay vector and the path gain vector use to model the channel using Rician and Rayleigh channels.

4.3.1. Rayleigh Fading

Rayleigh fading is a statistical model for the effect of a propagation environment on a radio signal and assumes that the magnitude of a signal that passes through a transmission medium like air interface called channel will vary randomly, or fade, according to Rayleigh distribution [14].

Rayleigh fading is viewed as a reasonable for modeling an effect of heavily built-up urban environments on radio signals. It is most applicable when there is no dominant propagation along a line of sight between the transmitter and receiver.

4.3.2. Rician Fading

Rician fading gives a stochastic model for radio propagation anomaly caused by partial cancellation of a radio signal by itself resulting ISI [15]. It models the signal arrives at the receiver by several different paths giving multipath propagation effect. These signals from different path arrive at the receiver at different time intervals and they add up constructively or destructively. Rician fading occurs when one of the paths, typically a line of sight signal, is much stronger than the others. The amplitude gain is characterized by a Rician distribution.

4.3.3. COST 207 and GSM/EDGE Channel Models

For the simulation of multipath fading communication channel for GSM/EDGE using Matlab or Simulink a standard is used and is called COST 207 and GSM/EDGE Channel models. This standard uses Rician and Rayleigh Channel model. Four propagation models are defined in COST 207 and GSM/EDGE Channel models out of which 3 models are used for simulation.

- Rural Area (RA)
- Bad Urban Area (BU)
- Hilly Terrain (HT)

Doppler spectrum objects relevant to the given COST 207 model are constructed using a Rayleigh or a Rician multipath fading channel object. The properties of those channel objects are initialized to produce the desired channel model. Data is then processed by the channel object to give an effect of wireless channel characteristics.

4.4. Equalizer

An Equalizer is designed using a least mean square algorithm. The equalizer first takes the entire modulated frame transmitted from the transmitter after being passed through the channel. The equalizer also takes modulated training sequence and compares each received training symbol with the known training symbols. The difference between them is calculated and using the step size of the LMS equalizer its weight for the predefined no of taps is calculated. This process continues and after all the training symbols the other data symbols are equalized, the data sequence is equalized using the latest equalizer parameter. For normal DSR EDGE frame structure the equalizer enter into training mode only once using the single training sequence. However in modified DSR EDGE frame structure the equalizer enters into the training mode twice, once form each training sequence. After training mode the equalizer shifts to a decision-directed mode.

In decision directed mode the equalizer calculates the error between the equalized symbol and the quantized form of the equalized symbol. Using the error the equalizer parameters are again updated to equalize the next symbol with minimum error.

The step size of the LMS algorithm determines the convergence properties of the equalizer. The equalizer performance is observed using different values of the LMS step size. The output from the equalizer is then fed to the 8-PSK demodulator.

4.5. Demodulator

The demodulator used in the DSR EDGE is an 8-PSK demodulator. The received signal from the equalizer is demodulated using gray coding to generate the estimated symbols. An 8-PSK demodulator block from the Simulink library is used for the demodulation of the equalized symbols.

4.6. Bit Error Detector

The bit error detector compares each data transmitted from the source with the corresponding demodulated data from the receiver and calculates the number of bits transmitted, number of error bits and bit error rate. The outputs of the comparator are the total no of symbols received, no of errors in the received symbols and the symbol error rate.

4.7. Simulation in Matlab

A simulator is designed in Simulink using all the above modules and the performance of the equalizer in various scenarios is analyzed.

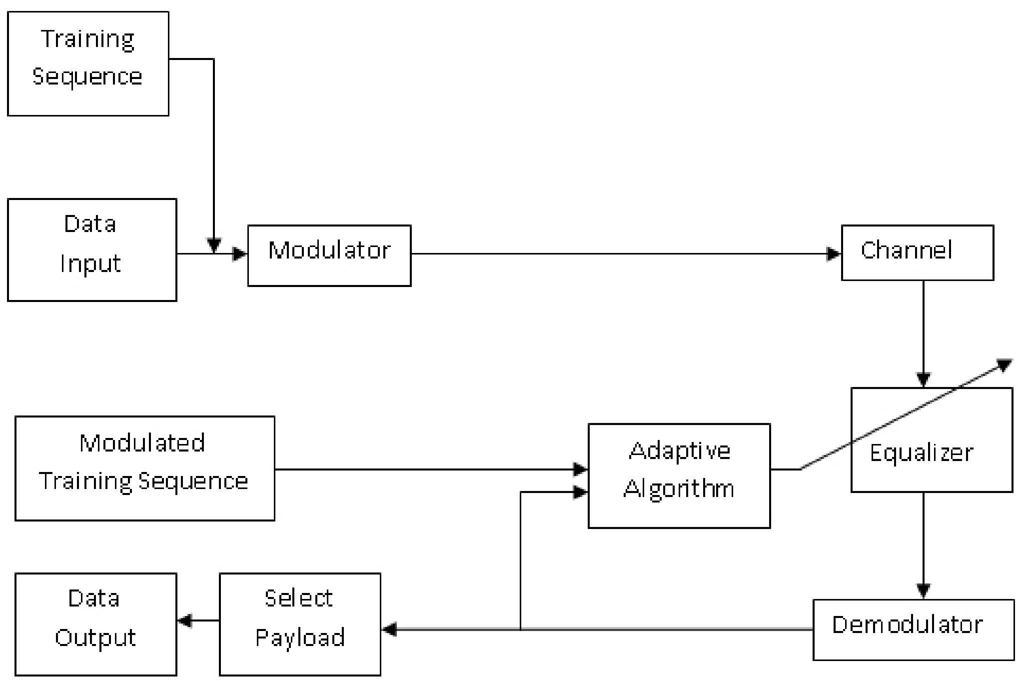

Figure 4.2 Training based channel equalization

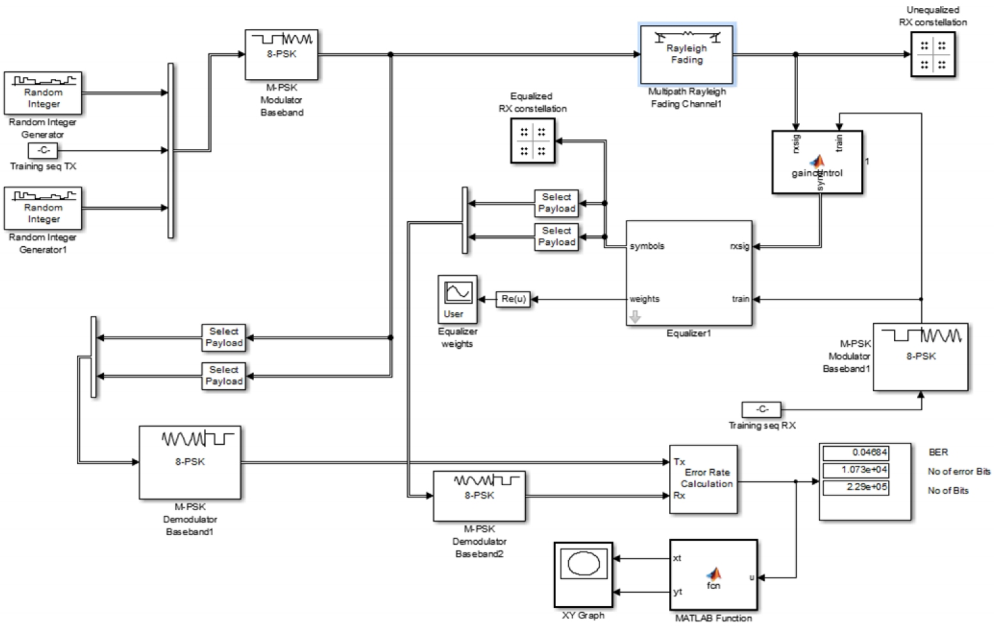

Figure 4.3 Simulink model of adaptive equalization of DSR EDGE with nor frame structure

The figure 4.3 above is a Simulink model used to simulate an adaptive equalization of a single training sequence used in normal DSR EDGE frame structure. A frame structure is constructed using two random data generator and a training sequence in the middle. The concatenated sequences of symbols are then passed through the modulator which modulates the symbols using 8-PSK modulator. The modulated signal is then passed through the wireless channel model. The signal distorted by the channel is then received by the receiver. The receiver consists of the channel equalizer. The equalizer receives the entire frame and then first updates the tap weight using all 52 symbols of the training sequence. Then the two data sequences are equalized using the updated equalizer. During the estimation of the data symbols the equalizer shift to decision directed mode and the equalizer taps are updated to give better estimation. The estimated symbols are then compared with the transmitted symbols and the error is calculated.

The following figure 4.4 is a Simulink model used to simulate an adaptive equalization of DSR EDGE with modified frame structure. Here two training sequences are used along with the two random data generator. In this case the equalizer takes the entire frame and the first training sequence is used to update the equalizer taps. After updating the equalizer taps using first training sequence the first data sequence is estimated. The equalizer shifts to decision-directed mode for better estimation. Then again the equalizer shifts to the training mode after estimation of first data sequence. So the equalizer is trained twice using 26 training symbols of modified frame structure. And at last the second data symbols are estimated. Hence by training the equalizer twice a better estimation of the channel can be done and the error is minimized.

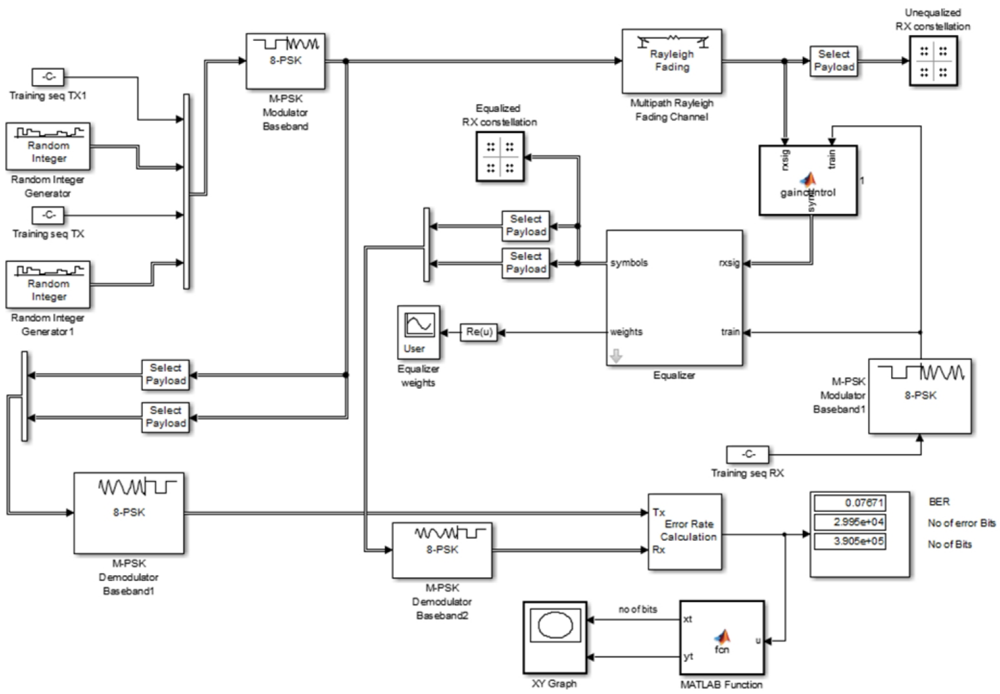

Figure 4.4 Simulink model of adaptive equalization of DSR EDGE with modified frame structure

CHAPTER 5: RESULTS AND DISCUSSION

The simulation of the equalizer system was done using Simulink in Matlab. A training sequence and a random signal generator were concatenated to give frame which was then passed through a channel model designed according to COSTS 207 and GSM channel standards. The model was simulated for different scenarios. Three different scenarios used to model the fading wireless channel were rural area (RA), bad urban area (BU), and hilly terrain (HT). The output from the channel was then equalized using LMS algorithm. With the change in the parameters of the channel the tap weight of the equalizer changed to adjust the channel response.

Signal scopes were used at various points in the model to closely analyze the signal. The constellation diagram at different points showed the condition of symbols at different points and the variation in the transmitted symbol position and received one. For calculation of the total no bits received and the error in the received signal a comparator with a display were used to give clear picture of the performance of the equalizer.

The simulation showed that the parameters of the channel models such as Doppler shift, gain vector of the multipath propagation channel, delay have large effect on the signal. The simulation also showed that the main characteristics of the equalizer using LMS algorithm depends on the step size of the equalizer and the number of taps used in the equalizer. The number of taps to be used also depends upon the scenario of the area chosen.

The use of modified frame structure for DSR showed improvement in equalization of the signal received. In modified frame structure the adaptive equalizer was trained twice in a single frame structure whereas the equalizer is trained only once in case of normal DSR frame structure. With the use of two training sequences the distance between the actual data and the training sequence was reduced.

5.1. Simulation of the Equalizer in Rural Area

S.N	Max Doppler shift Hz	Bit Error Rate					
		Step size μ= 0.100		Step size μ= 0.125		Step size μ = 0.150	
		Normal DSR EDGE	Modified DSR EDGE	Normal DSR EDGE	Modified DSR EDGE	Normal DSR EDGE	Modified DSR EDGE
1	**10**	0.003325	0.00177	0.00332	0.00178	0.003496	0.001768
2	**50**	0.0723	0.03244	0.07418	0.02962	0.07626	0.02756
3	**100**	0.2949	0.1575	0.3052	0.1406	0.3134	0.1277
4	**200**	0.4513	0.3948	0.463	0.3749	0.4726	0.3572

5.2. Simulation of the Equalizer in Bad Urban Area

S.N	Max Doppler shift Hz	Bit Error Rate					
		Step size μ= 0.100		Step size μ= 0.125		Step size μ = 0.150	
		Normal DSR EDGE	Modified DSR EDGE	Normal DSR EDGE	Modified DSR EDGE	Normal DSR EDGE	Modified DSR EDGE
1	**10**	0.008308	0.004614	0.08548	0.004578	0.0079	0.004625
2	**50**	0.1131	0.06345	0.1217	0.0604	0.1271	0.05537
3	**100**	0.2545	0.1756	0.2664	0.1608	0.2755	0.1505
4	**200**	0.4143	0.3745	0.4239	0.36	0.4346	0.3496

5.3. Simulation of the Equalizer in Hilly Terrain

S. N	Max Doppler shift Hz	Bit Error Rate					
		Step size μ= 0.100		Step size μ= 0.125		Step size μ = 0.150	
		Normal DSR EDGE	Modified DSR EDGE	Normal DSR EDGE	Modified DSR EDGE	Normal DSR EDGE	Modified DSR EDGE
1	**10**	0.009845	0.004671	0.04421	0.004807	0.2156	0.004712
2	**50**	0.2291	0.1061	0.345	0.1913	0.4213	0.3161
3	**100**	0.3521	0.2612	0.4464	0.3177	0.4326	0.384
4	**200**	0.4723	0.447	0.4866	0.4461	0.48321	0.4506

The above table tabulates the Bit Error Rate of the communication system using adaptive equalizer with LMS algorithm. Three different scenarios were used to simulate the channel model for DSR EDGE with normal frame and modified frame structure. Each of the scenarios was simulated with four different Doppler shifts 10, 50, 100 and 200 Hz.

The result in the table shows that the use of two training sequences at different location of the DSR EDGE frame the equalizer performance is better. The equalizer estimated the received signal with better accuracy in modified frame structure than the normal frame structure. In all three scenarios of GSM/EDGE channel models and all four Doppler shifts the equalizer performance is better in modified frame structure. The BER increased with increase in the Doppler shift however the performance of the modified frame structure is better compared to that of normal frame structure.

The performance of LMS algorithm also depends upon the step sizes of the algorithm. In simulation three step sizes, 0.100, 0.125 and 0.150 were used for simulation of LMS algorithm based equalization. The figures showed the BER comparison between the equalization of normal frame and modified frame. The adaptive LMS equalizer performed best at the step size of 0.125, so all the figures below are the simulation result using 0.125 step size in the LMS algorithm. The line using * shows the simulation result of normal frame structure and the thin line showed the result of modified frame structure. The BER increase at first due to the equalizer being unknown about the channel; and then decreased due to training and adaptive equalization. The bit error rate remained almost constant then after which is due to the characteristics of the time varying channel. With increased in the Doppler shift the BER increased.

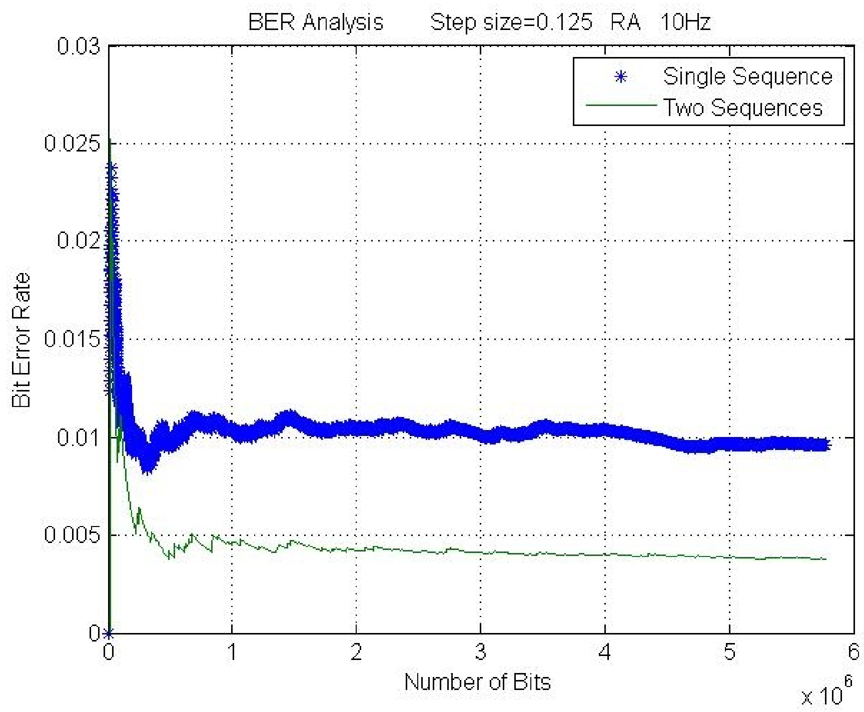

Figure 5.1 BER of equalizer in Rural Area at 10Hz Doppler shift

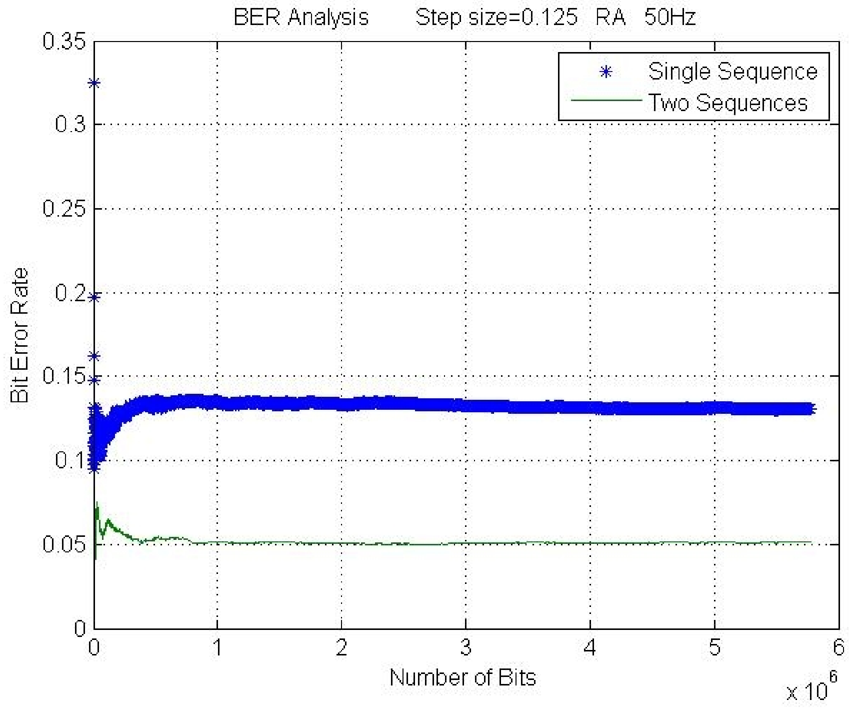

Figure 5.2 BER of equalizer in Rural Area at 50Hz Doppler shift

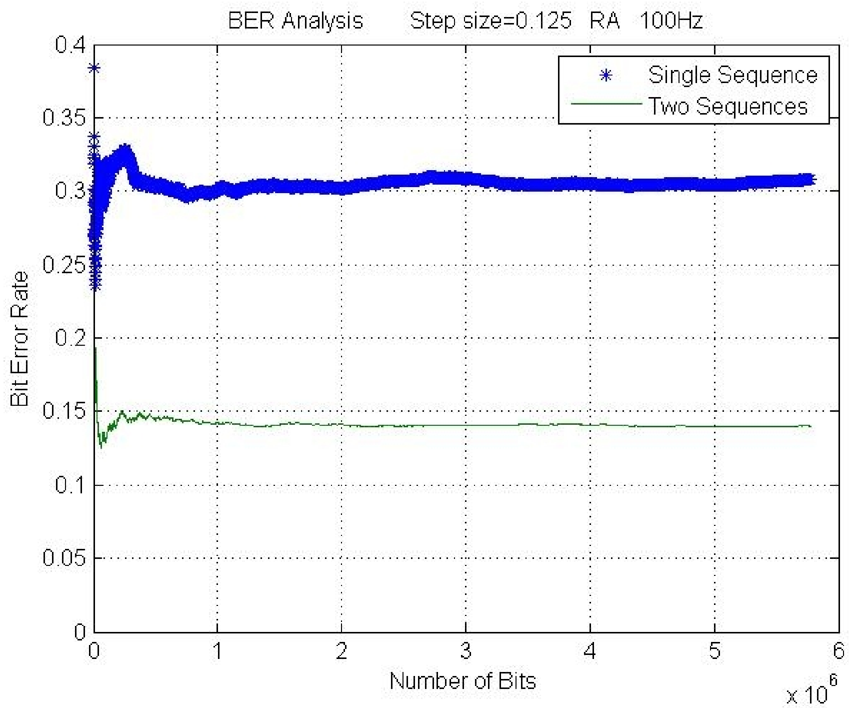

Figure 5.3 BER of equalizer in Rural Area at 100Hz Doppler shift

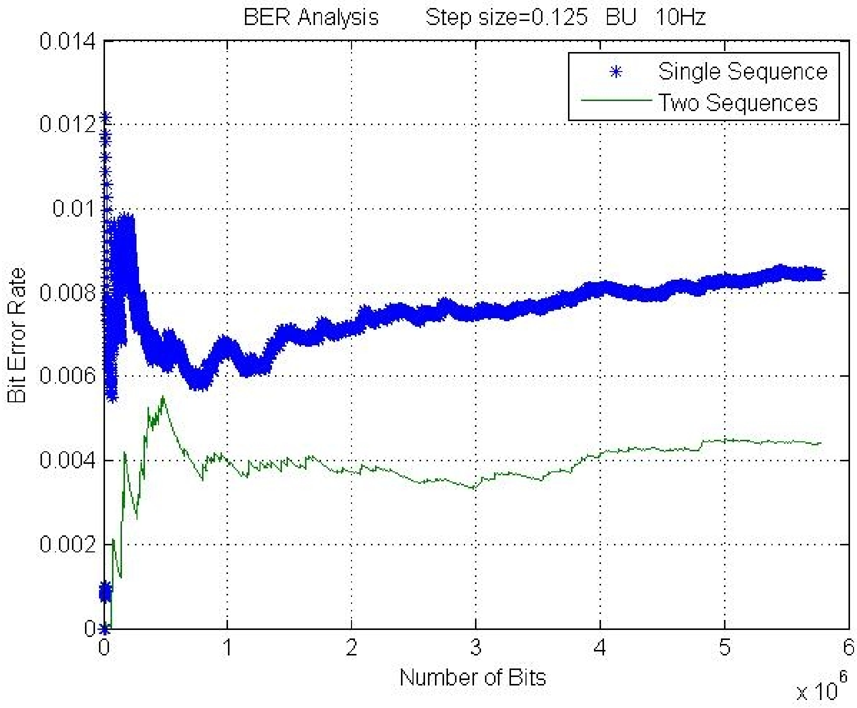

Figure 5.4 BER of equalizer in Bad Urban Area at 10Hz Doppler shift

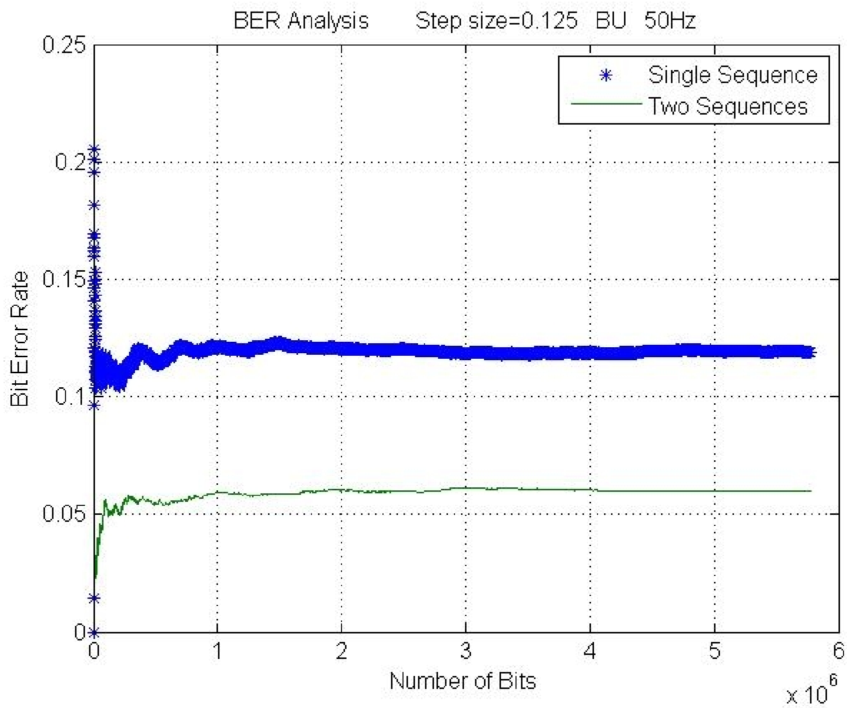

Figure 5.5 BER of equalizer in Bad Urban Area at 50Hz Doppler shift

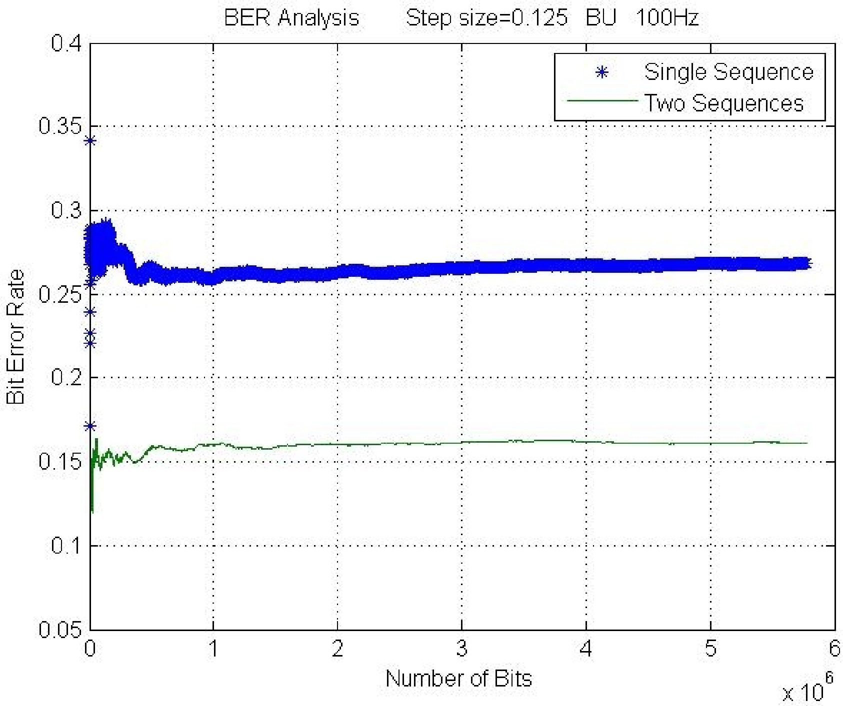

Figure 5.6 BER of equalizer in Bad Urban Area at 100Hz Doppler shift

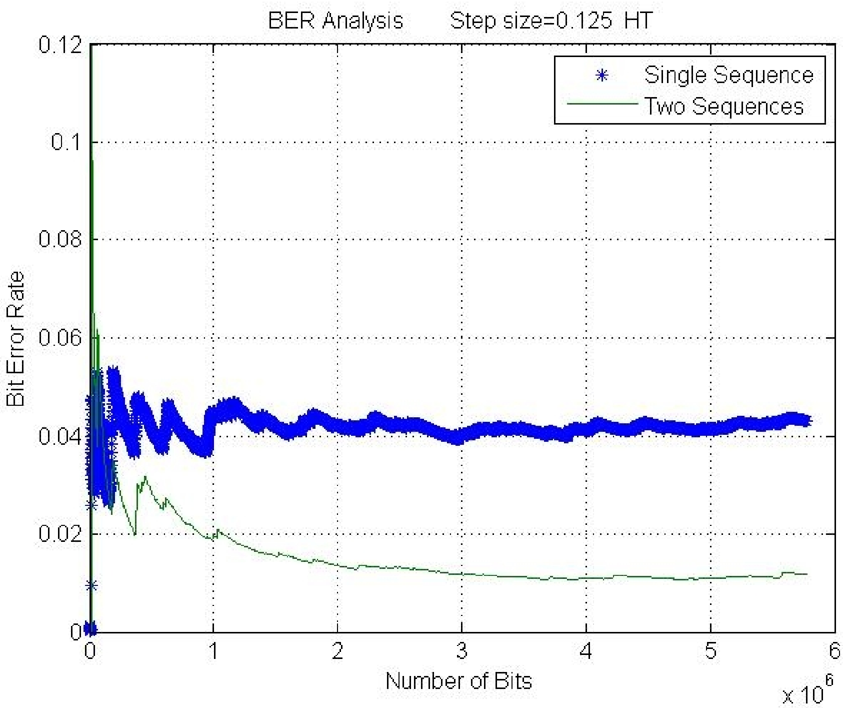

Figure 5.7 BER of equalizer in Hilly terrain at 10HZ Doppler shift

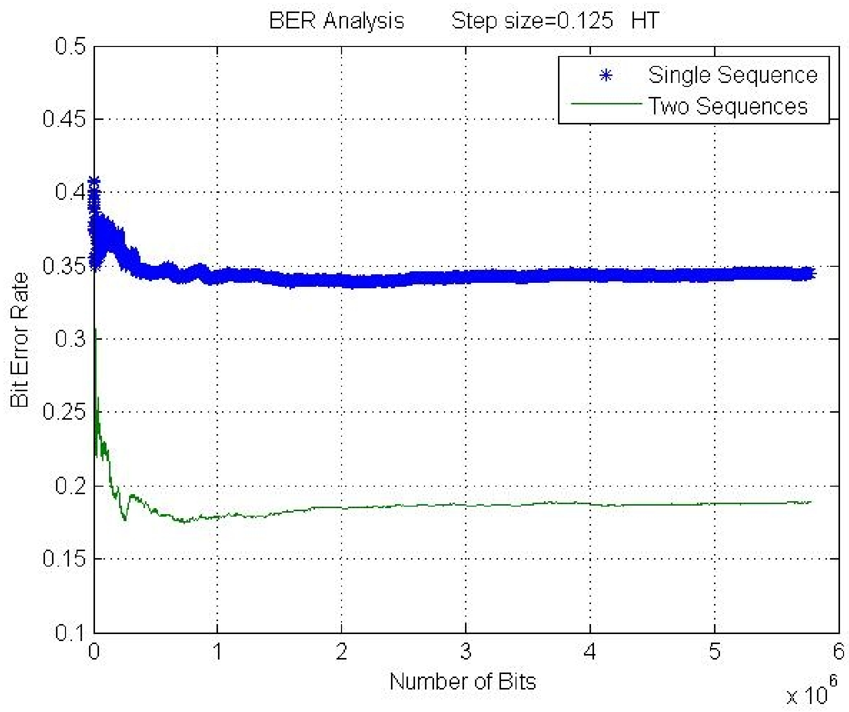

Figure 5.8 BER of equalizer in Hilly terrain at 50HZ Doppler shift

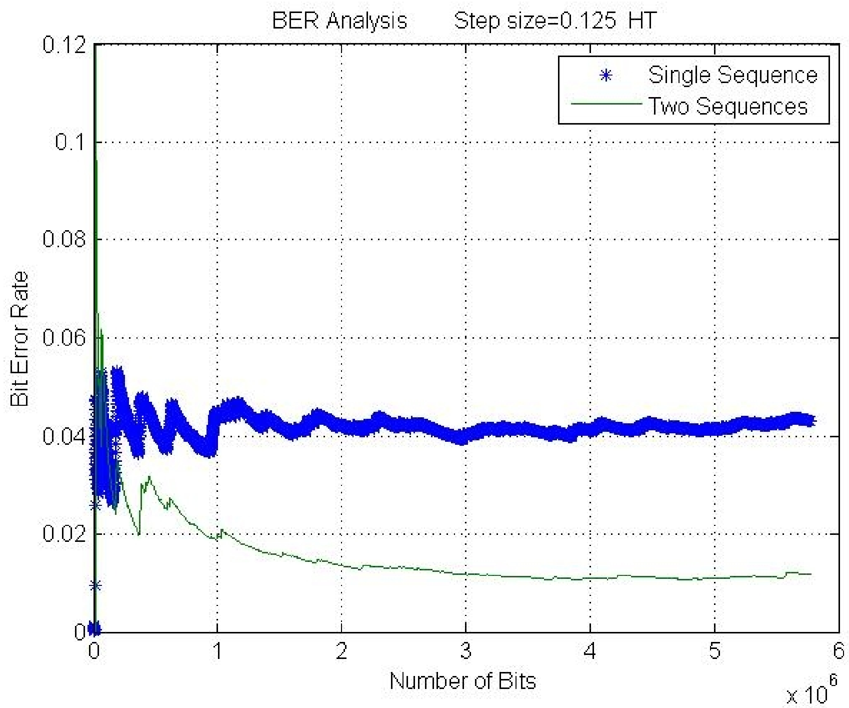

Figure 5.9 BER of equalizer in Hilly terrain at 10HZ Doppler shift

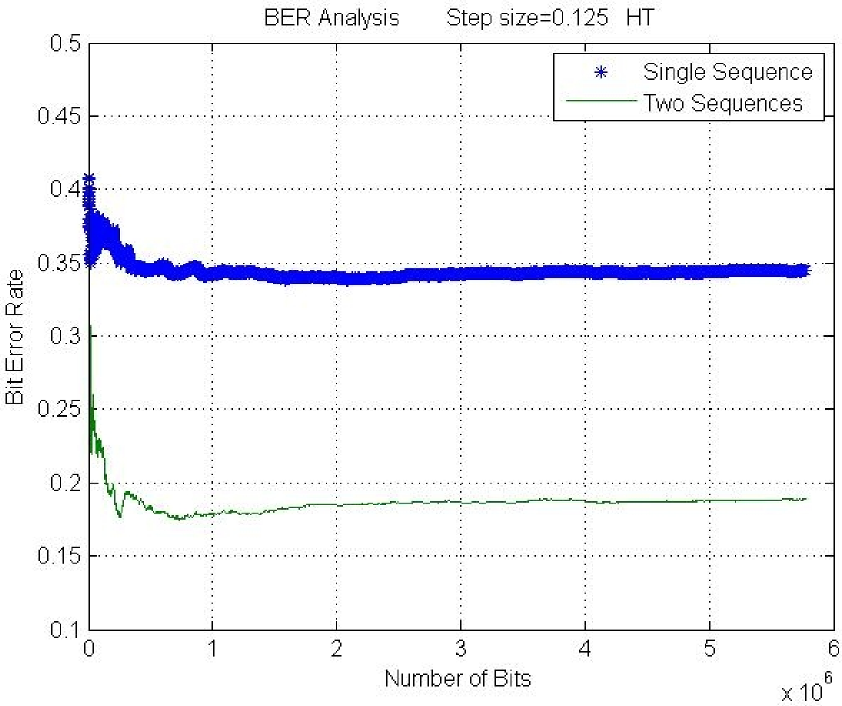

Figure 5.10 BER of equalizer in Hilly terrain at 50HZ Doppler shift

CHAPTER 6: CONCLUSION

During this thesis work study on the concepts behind the wireless channel characteristics was done and the algorithms used to equalize the wireless channel to mitigate the effect of the channel on the transmitted signal were studied. With the knowledge of the equalizer and wireless channel an adaptive equalizer was designed. For simplicity and better convergence LMS algorithm was used in adaptive equalizer.

From the simulation we can conclude that the signal transmitted through the wireless channel is affected by parameters such as path delay, gain of each path, Doppler shift and noise. An equalizer is a must to reduce such channel effect on the signal. Due to the relative motion between the mobile system and network element these channel parameters are not constant but changes over time. So an adaptive equalizer is required to estimate the time varying channel. Thus a LMS algorithm based symbol spaced equalizer was used for the estimation of the channel.

The simulation of the wireless channel was carried out in Simulink. The simulation of the wireless channel was done using COST 207 GSM/EDGE channel models. Three different scenarios, rural area, bad urban area and hilly terrain were used. The performance of the equalizer was measured based on the bit error rate. Simulation was done for normal frame structure of DSR EDGE and compared with the modified frame structure of EDGE.

The result of the simulation show the modified frame structure with two training sequence of DSR EDGE has better performance than the normal DSR EDGE frame structure with the same equalizer. When two training sequences were used the distance between the actual data and training sequence is small and so the training sequence could better estimate the channel and its effect on the data. With the use of the training sequence in front of each data sequence the adaptive equalizer performed better compared to the use of the single training sequence in between the two data sequence. The adaptive equalizer was trained twice in a single frame in modified frame structure and only once in case of normal frame structure.

The simulation of the system was carried out in Simulink of Matlab which made it easy to visualize the signal flow and the effect of the channel on the signal at every point of communication system. Clear visualization of the signal flow has been made easy using Simulink library blocks such as, signal scope, constellation diagrams, graph plotter, comparators and display.

Thus this work conclude that the training sequence used frame structure of wireless communication system greatly affect the performance of the communication system and instead of using just a single long training sequence in a frame, the training sequence can be split to give better estimation of the channel and reduce the BER. In this thesis splitting of training sequence was carried out in DSR EDGE frame structure; however this work can also be extended to other communication technologies such UMTS and LTE.

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