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# PERFORMANCE STUDY OF THE EFFECTS OF VARYING EQUALIZERS PARAMETERS ON MULTIPATH RADIO FADING CHANNEL

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# PERFORMANCE STUDY OF THE EFFECTS OF VARYING EQUALIZERS PARAMETERS ON MULTIPATH RADIO FADING CHANNEL

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ANWR ABD S. ELASYRI

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# ABSTRACT

## PERFORMANCE STUDY OF THE EFFECTS OF VARYING EQUALIZERS PARAMETERS ON MULTIPATH RADIO FADING CHANNEL

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The basic form of a digital communication system consists of three major components; a transmitter, channel, and receiver connected somehow together in cascade. The transmitter maps a message signal generated by a digital source into an electrical waveform suitable for transmission over the channel, the channel, which might be either a physical structure or space, suffers from two major kinds of impairments; intersymbol interference and noise. The combined effects of these two impairments are the introduction of noise and distortions to the transmitted signal at the channel output. In order to reduce the effects of these distortions an equalizer is used at the receiver end. The main role of the equalizer is to reconstruct the transmitted symbols by observing the received noisy signal. The adaptive equalizers using a least mean square (LMS) and the recursive least-squares (RLS) algorithms are investigated in this work for wireless communication systems. Both linear transversal equalizer and non-linear decision feedback equalizer (DFE) with LMS and RLS algorithms are implemented in a simplified communication system model operating over a multipath fading channel model. The impact of some influential parameters, such as the number of taps of the adaptive equalizers and the number of significant paths in a multipath fading channel were investigated using Simulink software package. The Simulink simulation program was used to implement LMS and RLS Algorithms in the simplified simulated communication system which calculates the bit error rates versus SNR. Quantitative Results obtained were presented in graphs. These graphs clearly illustrate the absence of a definite relationship between the number of multipath in the channel and the length of the equalizer being implemented. This finding is clearly reflected by the performance of the communication system in terms of bit error rate.

**Keywords:** InterSymbol Interference (ISI), multipath fading channel, LMS and RLS adaptation algorithms; linear and decision feedback equalizers.

# TABLE OF CONTENTES

**Pages**

# LIST OF TABLES

**Pages**

# LIST OF FIGURES

**Pages**

# 1. INTRODUCTION

## 1.1 OVERVIEW

Communication services have witnessed a phenomenal growth over the past five decades. Nowadays, high-speed communication services around the world would only be possible with the development of current state of art of both satellite and fiber optics networks. Gradually, fiber optic cables have replaced most of the wired line communication systems. Such cables provide bandwidth wide enough to accommodate the transmission of a large variety of information sources, such as voice, data, and video. The emerging of highly developed internet technologies necessitates the need for efficient high-speed data transmission methods over communications channels. As the rate of the data transmission increases to fulfill the needs of the users, the channel introduces distortions in data. One major cause of distortion is Inter Symbol Interference (ISI). In digital communications, the transmitted signals are coded as multilevel rectangular pulses. Such forms of pulses occupy infinite absolute bandwidth. If these pulses pass through a band limited communication channel, they will spread in time causing pulses representing symbols to smear into their adjacent time slots and hence interfering with the adjacent symbols. This is referred as intersymbol interference (ISI) [1]. In addition, received symbols suffer from further distortions introduced by other factors like thermal noise, impulse noise, and cross talk. Therefore, the use of signal processing techniques becomes inevitable at the receiver to overcome these interferences, restore the transmitted symbols and consequently, recover their information. Such processing techniques are commonly referred to as “channel equalization” or just “equalization”. For a channel with precisely known characteristics, a transmitting and receiving filters can always be adequately designed to reduce the ISI effect as well as the inevitable additive noise [2]. However, It is well known fact that the channel characteristics are random, time variant and therefore, the channel is modeled as one of an ensemble of possible channels [2], [3]. Therefore, a pair of stationary transmitting and receiving filters designed to match the average channel characteristics, may not be adequate in minimizing inter symbol interference to an acceptable level. To deal with this problem, adaptive equalization is widely used to provide a strict control over the channel time response [3]. Equalizers that employ adaptation techniques are therefore crucial in the design of high-speed communication systems [4]. It is natural that data transmitted through a band limited communication channel encounters linear, nonlinear and additive distortions. In order to reduce the effects of these distortions an equalizer is used at the receiver end. The function of the equalizer is to reconstruct the transmitted symbols by observing the received noisy signal [3][5].

## 1.2 PROBLEM STATEMENT

The basic form of a digital communication system is composed of a transmitter, channel, and receiver cascaded together to form a link (physical or logical) between an information source, where the message is generated, and a sink, the recipient of the message. The transmitter role is to convert a message signal into a waveform whose characteristics matches that of the channel over which it is transmitted. The receiver on the other side of the link reverses the conversion being made by transmitter. Typically, the channel suffers from two major kinds of impairments:

- Intersymbol interference: In a digital communication system, the channel impairment gives rise to intersymbol interference - successive pulses spread in time and hence symbols, they represent, smear into one another to the extent that they become no longer distinguishable.
- Noise: Noise, of different forms, is always introduced by a communication channel regardless of its nature.

## 1.3 OBJECTIVES

The objectives of this research are:

- Understanding radio wireless and radio propagation mechanisms.
- Investigating equalization and algorithms for adaptive equalization.
- Implementing LMS and RLS algorithm using MATLAB Simulink simulation software.
- Comparing, based on bit error rate criteria, the performance of a communication system without and with an equalizer being implemented at the receiving end and investigating the effect of a wireless channel characteristics on the propagated signal using Matlab simulation packages.

## 1.4 METHODOLOGY

- Descriptive analyses for digital data transmission and adaptive equalizations.
- Mathematical modeling for adaptive equalizations as applied to a basic digital communication system.
- Computer modeling using Matlab Simulink program to implement this computer model.
- Bit error rate performance evaluation in terms of graphs.
- Matlab Simulink models are to be designed and used to calculate the bit error rate against AWGN with different parameters associated with the equalizers being implemented.

## 1.5 THESIS LAYOUT

This thesis consists of five chapters. In chapter two, characterization of mobile-radio propagation (i.e. wireless channel) is presented. Also, concepts like channel models, such as AWGN channel, fading, multiplexing techniques, multiple access methods and intersymbol interference (ISI) are briefly reviewed.

Chapter three covers principles related to basic concepts of equalizers. Properties, advantages, and applications of equalizers are also presented. The standard least mean square (LMS) and recursive least-squares (RLS) algorithms are described in this chapter.

Chapter four encompasses the test beds implemented in this work. The test beds are specified in details and the system parameters associated with each experimental set up is tabulated.

The results obtained by simulation runs are appropriately displayed followed by a conclusion and suggested recommendations for future work in chapter five.

# 2. MECHANISMS OF SIGNAL PROPAGATION THROUGH THE WIRELESS CHANNEL

## 2.1 WIRELESS TRANSMISSIONS

The technology of wireless networking has been around since the 1800s and began to significantly penetrate the market over the last three decades. The discovery made by the well-known musician and astronomer, Sir William Herschel (1738 - 1822) who declared the existence of infrared light and was invisible to the human eye. This significant discovery had paved the way to a field of vital importance “theory of electromagnetic wave”, which was deeply investigated by James Maxwell (1831 - 1879). James Maxwell’s findings related to electromagnetism were originated from the discoveries reached; by Michael Faraday (1791 - 1867) and Andre-Marie Ampere (1775 - 1836), both came before him. Heinrich Hertz (1857 - 1894) experimentally extended the discoveries of Maxwell when he proved that electricity is carried on EM waves as they propagate at the speed of light [1], [6].

These discoveries are of great importance as they are closely related to wireless local-area networks (WLANs). To justify this argument: In standard LANs, data is propagated, in the form of electrical signals, over wires such as conventional or optical fiber cables. The contributions introduced by Hertz made possible the transfer of the same data as electrical signals but through space i.e. no physical links are involved. The relationship between WLANs and the discoveries already presented can be described simply as follows: The WLAN is a LAN that does not employ cables for data transfer between devices. This technology is prevailing now due to the tremendous efforts led by the outstanding discoveries made by Faraday, Maxwell, Ampere, and Hertz. This, of course, is only made possible through the use of Radio Frequencies (RF) [6].

With RFs, the ultimate goal of design is to transfer as much data (capacity) over farthest possible distances (coverage areas) and at the highest possible speed (bandwidth). The challenges to be taken care of are severe influences in radio frequencies which should be either overcome or reduced. The most challenging of these is interference.

## 2.2 EM WAVES PROPAGATION MECHANISMS

Radio propagation is the way or mode in which radio waves behave as they are propagated from the source to the sink. Either the transmitter and/or receiver are located on earth or into various parts of the atmosphere. Radio waves, just like light waves, are in fact a form of electromagnetic radiation and therefore, they encounter, in exactly the same fashion, reflection, refraction, diffraction, absorption, polarization and scattering [1], [6]. The troposphere contains water vapor and the upper atmosphere is, due to the Sun, continuously ionized. The level of water vapor and density of ionization changes almost daily and consequently affect radio propagation. It is therefore, crucial to understand the effects of varying conditions on radio propagation. Such affects practically influences many aspects such as the choice of international short wave broadcasters frequencies, radio navigation, designing reliable mobile telephone systems, radar systems operation, and the like.

Depending on its path from point to point, there exist many other factors which have significant effects on radio propagation. Radio waves propagation can be along a direct line of sight (LOS) path or over-the-horizon path combined with refraction caused by the ionized region of the earth atmosphere (ionosphere). The ionosphere is a region extends between approximately 550 and 650 km above the surface of earth. The ionosphere radio signal propagation might be influenced by one or more of the following factors; sporadic-E, spread-F, solar proton events, geomagnetic storms, ionosphere layer tilts, and solar flares [3].

According to their frequencies, radio waves propagate in different modes. Electromagnetic waves at extra low frequencies (ELF) and very low frequencies may propagate as in a waveguide in the region between the earth surface and the D layer of the ionosphere. This is so because the wavelength at such frequencies is very much larger than the separation being indicated. It is well known that an electromagnetic wave at a frequency less than 20 kHz propagates as a horizontally polarized single waveguide mode [4], [6].

The interaction of the radio waves with the ionized regions of the atmosphere is taking place in such a way that complicates the prediction and analysis of wave propagation as compared to that in free space. Radio propagation in the ionosphere layer is very much influenced by space weather. Sporadic solar flares give rise to the emission of x-rays which ionize the ionospheric D-region resulting in a sudden ionospheric disturbance or shortwave fadeout. Ionization enhancement in that region causing an increase in the absorption of radio signals propagating through it. When solar x-ray flares explode strongly, all ionospherically propagated radio signals in the sunlit hemisphere are virtually completely absorbed. HF radio propagation and GPS accuracy can be severely affected by these solar flares [6].

Services such as emergency locator transmitters, some television broadcasting and in-flight communication with ocean-crossing aircraft, have been moved to satellites communications systems due to the abovementioned fact that radio propagation is fully unpredictable. A satellite link is, of course, more robust as it is highly predictable, highly stable and hence can offer a satisfactory coverage over a given area [1] [6].

## 2.3 THE FADING PHENOMENA

Fading is associated with wireless communications and is defined as the fluctuation in the amount of the attenuation that affects a signal as it propagates through a given media. Fading usually varies with time; the variation depends on both; geographical location, and the frequency of the propagated wave. Therefore, it is often modeled as a random process. In wireless systems, the fading process may be either multipath induced fading or shadow fading. The former is due to multipath propagation, while the later is due to shadowing caused by obstacles along the path of wave propagation.

Reflectors existing in the premises of a transmitter and/or receiver give rise to multiple paths along which a transmitted signal propagates. As a consequence, a superposition of many copies of the transmitted signal arrives at the receiver traversing different paths. Each particular signal copy encounters a certain amount of distortion that encompasses attenuation, delay and phase shift associated with its own path of propagation towards the receiver. Due to this propagation mechanism, either constructive or destructive interference of the signal power at the receiver occurs. If strong destructive interference “deep fade” happens, a severe drop in the channel signal-to-noise ratio takes place which may result in temporary failure of communication [1], [7].

The fading phenomena that electromagnetic transmission of information encounters over the air, in both cellular networks and broadcast communications, can be modeled using fading channel models. The same models are often used to model the distortion caused to the signal by the water in underwater acoustic communications. Fading is mathematically modeled as a time-varying random fluctuation in both the amplitude and phase of the signal being propagated.

### 2.3.1 Types of Fading

Fading could be either ***Slow*** or ***fast*** fading depending on how fast the change in magnitude and phase of the signal traversing the channel. The ***coherence time*** is a figure of merit associated with a wireless fading channel. It measures the minimum time required for the change in the magnitude of the signal to become uncorrelated from its previous value [8], [7]. Depending on this measure, two types of fading can be defined. They are:

- **Slow Fading**

Fading is described as being slow if the coherence time of the channel is larger than the its delay constraint. In such fading, the amplitude and phase change experienced by the signal can be considered, approximately, stationary over the channel use interval. The main cause of slow fading is shadowing, where a signal on its main path of propagation between the transmitter and receiver encounters an obstacle; like a hill or large constructions, such as bulky buildings. The received power change caused by slow fading is often modeled using a log-normal distribution. The standard deviation is chosen according to the log-distance path loss model [7], [9].

- **Fast Fading**

Fading is considered as fast if the channel coherence time is relatively smaller than the delay constraint experienced by the propagated signal over the channel. For this scenario, the amplitude and phase change affected by the channel varies considerably over the channel use interval.

The transmitter, In a fast-fading channel, may use time diversity to increase robustness of the communication to a temporary deep fade. Error-correcting coding combined with interleaving techniques are normally used to recover the erased bits incurred by deep fading. In the case of a slow-fading channel, the transmitter sees only, within the channel delay constraints, a single realization of the channel, so error-correcting techniques coupled with interleaving is not effective in recovering bits erased by deep fading.

The channel coherence time is related to a very commonly used parameter called the ***Doppler spread***. A relative motion between a user (or reflectors in its environment) and a receiver, causes a drift in the transmitted signal frequency. The amount of this drift varies over the paths of signal propagation depending on the delay associated with each path. This phenomenon is referred to as the ***Doppler shift***. The Doppler spread, associated to a single fading channel is defined as the difference in Doppler shifts between different signal components arriving at the receiver. If the phases of the signal components change independently over time then the channel is considered to have a large Doppler spread. Such channels, by Fourier transform properties, have a very short coherence time [7], [10].

## 2.4 CHANNEL INTERFERENCE

To physically model a channel, it is required to calculate the processes which physically affect the signal passing through it. These physical processes, as far as wireless communications are concerned involve the reflections of the signal off obstacles and/or external interferences or electronic noise added to the signal in the receiver. The following three variables are usually used to statistically model a communication channel: the input alphabet, output alphabet, and a transition probability p (i, o) assigned for each pair (i, o) of input and output elements. The transition probability p (i, o) is the probability that the symbol o is received given that i was transmitted over the channel.

Statistical and physical modeling is normally combined to model a wireless communication channel. In wireless communications, for example, the channel is often modeled by a random attenuation followed by additive noise. The random attenuation is a simplification of the underlying physical processes and reflects the variation in power of the signal transmitted through the channel. Interferences, external and internal, as well as thermal noise (and other types of electronic noises) in the receiver is represented by noise in the model. A complex attenuation term describes the phase variations which reflect the relative time a signal takes to get through the multiple paths of the channel. Statistical characteristics of random attenuation are estimated based on experimental measurements and/or physical simulations.

Channel models are either continuous models or discrete-alphabet models.

When designing a communication system, it is more practical to use mathematical models for the channel that capture the most essential features of the transmission medium. This mathematical model is then used in the design of the channel encoder and modulator at the transmitter and the demodulator and channel decoder at the receiver [1], [11].

Three main channel models, namely the additive noise, linear time invariant filter and linear time-variant filter channel are briefly described in this work. These three models are closely related to the radio channels.

- **AWGN Channel**

The additive white Gaussian noise (AWGN) channel model produces simple, tractable mathematical model. The only impairment in this model is the linear addition of wideband or white noise "noise with a constant spectral density and a Gaussian distribution of amplitude". This model does not consider many aspects encountered in a fading channel such as; the phenomena of fading, frequency selectivity, interference, nonlinearity or dispersion.

Gaussian white noise is generated from many natural sources, such as the thermal vibrations of atoms in antennas, shot noise, black body radiation mainly the earth and other warm entities, and from the sun as well as other celestial sources.

Although the AWGN channel is not a good model for most terrestrial links it might be an adequate model for satellite and deep space links.

Fig. 2.1 shows the spectrum of a white noise. The power spectrum of white noise is flat over the entire bandwidth.

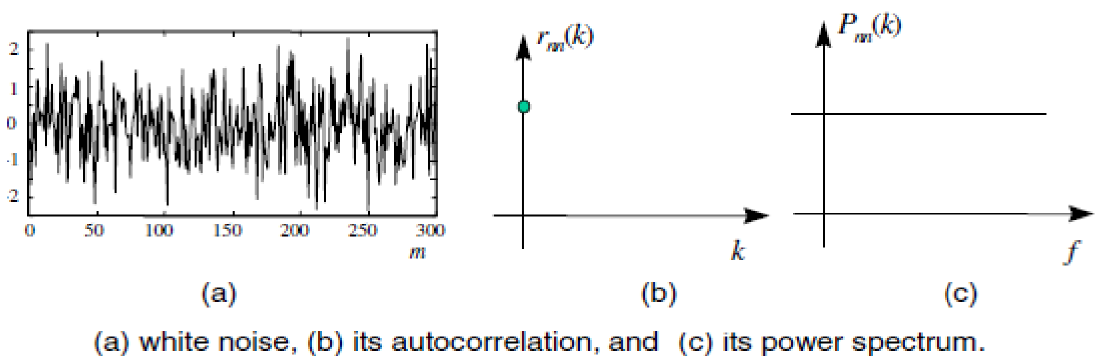


**Figure 2.1:** Power spectrum of a white noise [12]

- **Rayleigh Channel**

The line-of-sight path no longer exists; only the reflection paths are accounted for in this model shown as shown in Fig. 2.2. The process of interaction between the transmitted signal and the medium of propagation is modeled here as a sum of many (in practice few of them will be sufficient) multipath versions of the transmitted signal. These multiple copies of the signal are due to reflections from all angles (uniformly distributed) around the receiver. A mathematical expression for received signal $r(t)$ might be expressed as:

$$r(t) = a(t) * s(t) \tag{2.1}$$

Where $s(t)$ is the transmitted signal and $a(\mathrm{t})$ is the complex channel variable (expressing the fading process). For a Rayleigh fading channel $a(t)$ is a complex Gaussian process with a Rayleigh distributed envelope $|a(t)|$. The channel response in this channel is not uniformly distribute over the frequency band of interest. When a receiver moves, channel parameters vary rapidly leading to both Doppler shifting and spreading.

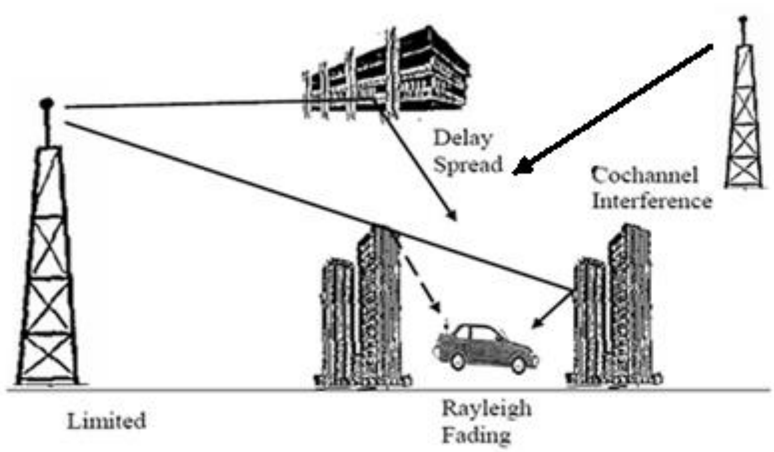


**Figure 2.2:** Illustration of Rayleigh fading [13]

- **Band limited Channel**

A band limited channel is the same as the linear filter channel, as depicted in Fig 2.3.

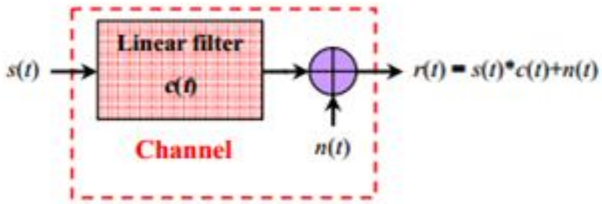


**Figure 2.3:** Linear filter model of channel with additive noise [1]

A band limited signal is either deterministic or stochastic signal, as shown in Fig. 2.4. The Fourier transform, or power spectrum, of a bandlimited signal is not zero only over a finite frequency band [1], [7]. The reconstruction of the signal from its samples can be guaranteed provided that the sampling rate is appropriately chosen (at least twice the Nyquist frequency). The highest significant frequency in the signal spectrum is referred to as the Nyquist frequency, and the minimum sampling frequency is called the Nyquist rate [1], [14].

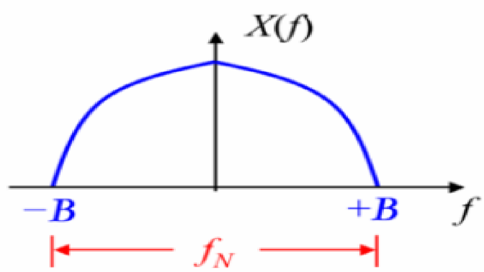


**Figure 2.4:** Spectrum of a band limited signal [1]

- **Rician Channel**

Multipath channels have a time varying characteristics. The signal in such channels arrives at the receiver over different propagation paths with various delays resulting in the arrival at the receiver of many copies of the transmitted signal. In most terrestrial channels, propagation paths are associated with the reflections of the signal by different obstructions in environment. At the receiver, it appears as a signal with some "ghosts" added to it. In such channel, one of the paths of propagation is the line-of-sight (LOS) path. More attenuation and longer delays are incurred on other paths than the main path as illustrated in Fig. 2.5. Such channel has ripples in its frequency response. The coherency bandwidth decreases when maximum channel delay increases. The characteristics of the first component received (shortest delay), associated with line-of-sight between transmitter and receiver is static and depend only on the Doppler frequency.

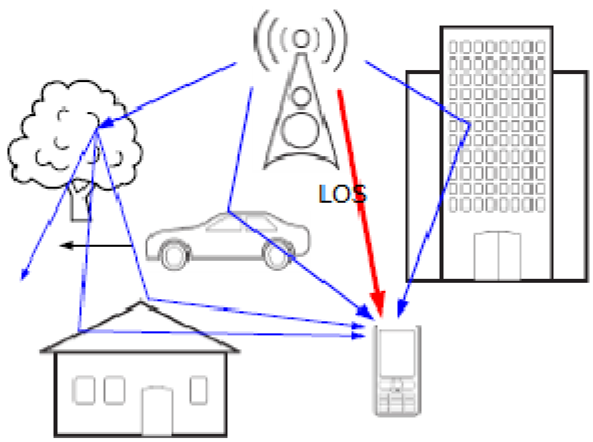


**Figure 2.5** : Illustration of Rician fading [13]

In practice the first reflected components arrives at the receiver with a very small time difference with respect to LOS component and hence the first tap in a discredited system is usually modeled as LOS component while the rest taps model fading components. The LOS component can be expressed as:

$$\rho e^{(2\pi f_p t+\theta_p)} \tag{2.2}$$

Where $\rho, f_p, and\ \theta_p$ are the amplitude, Doppler frequency and phase of that copy of the signal traversing the LOS path, respectively. The Doppler frequency is limited by the maximum Doppler frequency $f_{max}$ and hence typically the Doppler of the LOS is expressed relative to its maximum (commonly $f_p$= $0.7 f_{max}$) [14].

The phase, assumed to be random, is not normally accounted for as it has no effect on the statistics of the process and is usually set to zero.

## 2.5 MULTIPLEXING

Multiple analog message signals (or digital data streams) are combined into a single signal using multiplexing. This process facilitates sharing an expensive resource. For instance, multiple phone calls may be combined at transmitter and transferred to recipients using a single cable. Multiplexer (MUX) is a hardware device that combines data received from several sources into a single stream, and sends the data stream across a shared channel. Demultiplexer (DEMUX) is also, a hardware device or mechanism that decomposes the data stream received from a shared channel, and direct data to their corresponding recipients. The most common multiplexing techniques are:

- **FDM**

In frequency division multiplexing (FDM) several baseband messages are modulated on different frequency carrier signals and added together to form a composite signal. Carrier frequencies must be separated enough to avoid signals overlap (guard bands). Channel allocation to users is fixed, in this scheme, even if no data from a particular user exists for transmission.

- **TDM**

Most often, the data rate of transmission medium is much higher than the data rate of a digital signal to be transmitted. Therefore, it is more efficient to multiply digital signals can be interleaved in time, either at bit level or blocks, and transmitted. This is referred to as Time Division Multiplexing (TDM). In this scheme, time slots are pre-assigned to sources on fixed bases even if no data are present for transmission. Time slots allocation amongst sources is not normally evenly distributed.

## 2.6 MULTIPLE ACCESS METHODS

Multiple access schemes exist in order to facilitate sharing expensive resources such as the limited radio spectrum [9] by a number of users. This must be done without significantly degrading the performance of the system, particularly for high quality communications.

Such accessing techniques must guarantee signal integrity and therefore require that messages to the users should be orthogonal in signal space. There are four basic types of multiple access schemes: FDMA; Frequency Division Multiple Access, TDMA; Time Division Multiple Access, CDMA; Code Division Multiple Access, and SDMA; Space Division Multiple Access. A hybrid of these basic techniques might be used quite often.

- **Frequency Division Multiple Access**

Individual frequency channels are allocated to individual users in FDMA as illustrated in Fig. 2.6 [1]. Channels are assigned according to demand to users who request service. Only the assigned user can use the specified frequency band. Therefore, signals assigned to different users are clearly orthogonal. However practically, out-of-band spectral components may not be entirely annihilated leaving the signal not quite orthogonal. Therefore, guard bands should be introduced between frequency bands to suppress interferences amongst adjacent channels [1].

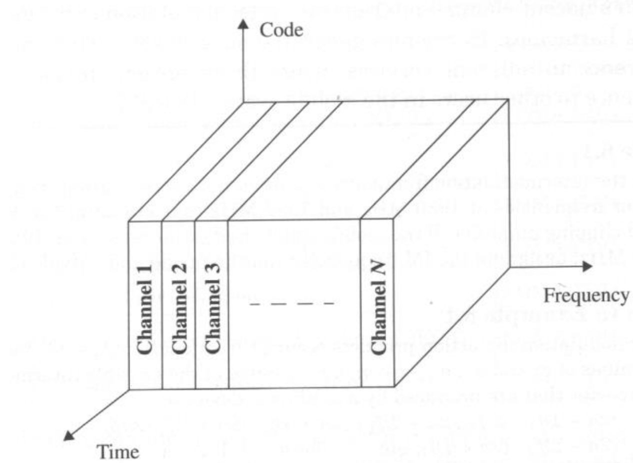


**Figure 2 6** : Frequency division multiple accesses [15]

- **Time Division Multiple Access**

The radio spectrum for this scheme is divided into time slots. Each user is assigned to one slot in the frame as depicted in Fig. 2.7. This allocation repeats over the sequence of frames. A particular user may transmit during its assigned slot in every frame. While TDMA users within a cell are separated by their time slots, different cells use different frequency channels.

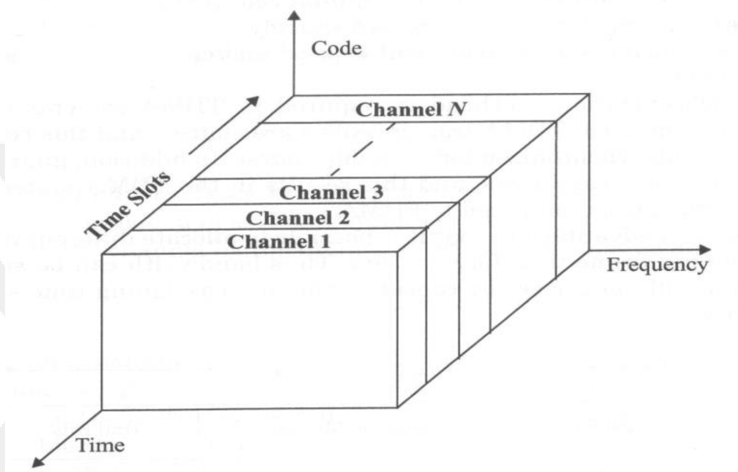


**Figure 2.7 :** Time division multiple accesses [15]

- **Code Division Multiple Access**

In this scheme, users may transmit simultaneously using the same carrier frequency [1]. The channels in this system are made orthogonal by using a different pseudonoise (PN) code sequence for each user. The set of all PN sequences in the system are made approximately orthogonal to each other as shown in Fig 2.8. A time correlation operation is implemented at the receiver in order to detect only the desired PN sequence. All other PN sequences appear as noise due to decorrelation; therefore the receiver needs to know the PN sequence used by the transmitter. User operates independently with no knowledge of each other.

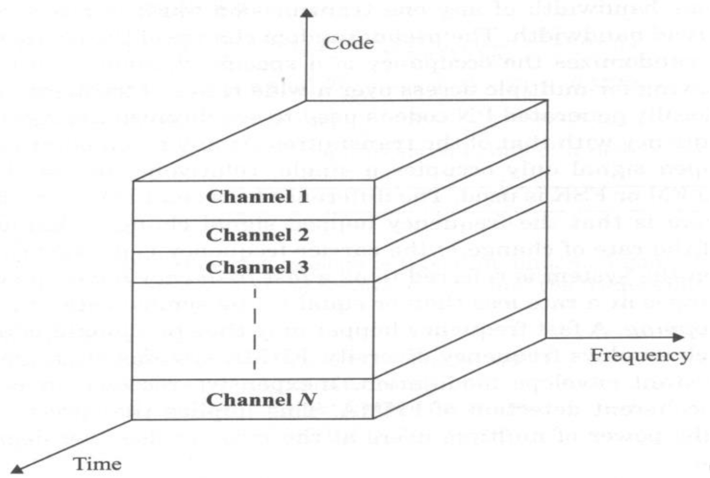


**Figure 2.8 :** Code division multiple accesses [15]

- **Space Division Multiple Access (SDMA)**

SDMA provide multiple users the opportunity to operate in the same cell, on the same frequency or time slot provided. This is accomplished by using an array of antennas to spatially separate the signals as shown in Fig 2.9. This technique is usually combined with another technique from the above-mentioned multiple access methods. This means that different areas covered by the antenna beam may be served by the same timeslots in a TDMA system, frequency in an FDMA system, or PN code in a CDMA system [1].

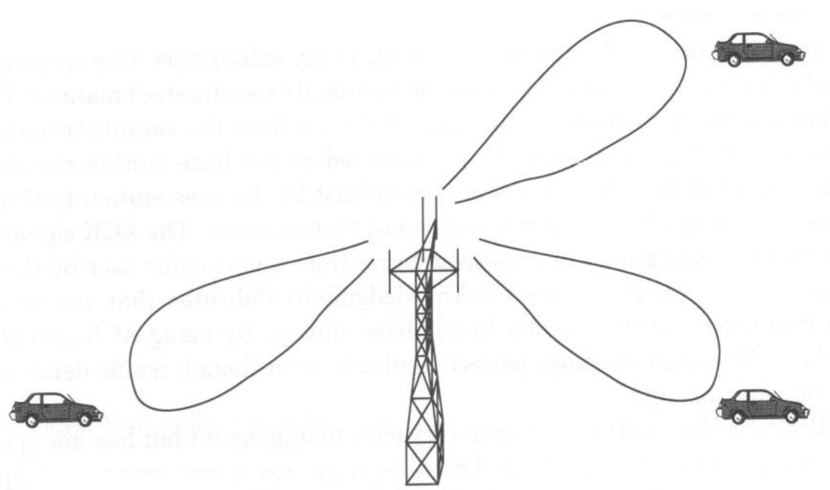

**Figure 2.9 :** Space division multiple accesses [15]

## 2.7 INTER SYMBOL INTERFERENCE (ISI)

The arrival of multiple copies of the transmitted signal traversing different paths to the receiver causes Intersymbol interference. ISI results in distortion to the transmitted signal and consequently the occurrence of bit errors at the receiver. Intersymbol interference has been recognized as the ***major constraint to high speed data transmission over mobile radio channels*** [7].

## 2.8 DIGITAL MODULATION

Modulation is the process by which certain parameters of a carrier signal are made to vary according to the information message [1], [7].Either the amplitude, phase, frequency or a combination of any two of them, suitably chosen, of the carrier signal can be modified in accordance with an information signal to obtain the modulated signal which is then further processed and transmitted. Many reasons for carrying out this modulation process before transmission does exist. These include multiplexing, which was already presented and medium compatibility; which involves making the signal properties physically compatible with the propagation medium. In digital modulation, the changes in the signal parameter are chosen from a fixed list each entry of which conveys a different possible piece of information.

A constellation diagram is often used to represent the modulation alphabet. Any digital modulation scheme that uses a finite number of distinct signals might be used to represent digital data in each digital modulation technique. In the case of PSK, for instance, a finite number of phases are used. Each of these phases represents a unique pattern, usually equal in number, of binary bits. Each pattern of bits forms the symbol that is represented by the particular phase. The demodulator at the receiver, which is designed specifically for the symbol-set used by the modulator, determines an estimate of the phase of the received signal and maps it back to the symbol it represents, hence the sent data is recovered.

To carry out this job, the receiver has to compare the phase of the received signal to a reference signal. The QPSK system constellation diagram, for instance, consists of four point’s equispaced around a circle [1].

QPSK encode two bits per symbol using the four phases as illustrated in Fig. 2.10. Gray coding is used in the Figure. The amplitude of two orthogonal (quadrature) sinusoidal waves is varied in accordance with the information signal to be transmitted using QAM.

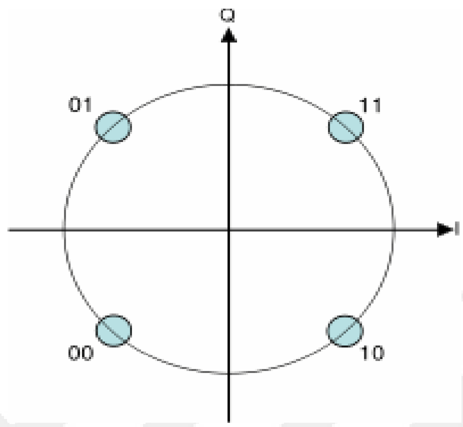


**Figure 2.10 :** QPSK constellations diagram [16]

16QAM uses sixteen points on the constellation diagram. With twelve phases and four amplitudes, 16 QAM can convey four bits per each transmitted symbol, as shown in Fig. 2.11 (Gray coding is used) [1].

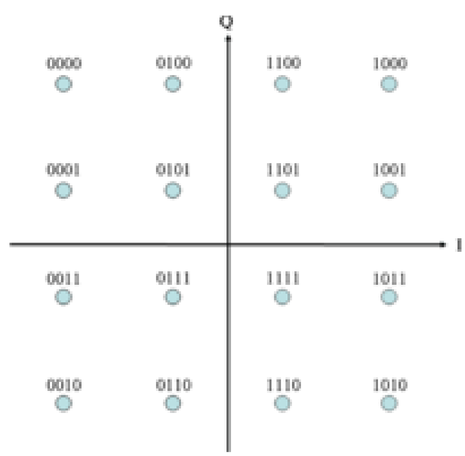


**Figure 2.11 :** 16 QAM constellations diagram [17]

# 3. FUNDAMENTALS AND IMPLEMENTATIONS OF EQUALIZERS

## 3.1 THE NEED FOR EQUALIZATIONS

Wireless channels may be modeled as AWGN channels or channels with flat fading. Other types of channels, such as wireless mobile transmission media, normally exhibit frequency selective fading and Doppler shift. Delay spread, due to multipath phenomena, and frequency-selective fading introduce intersymbol interference (ISI), which badly affect the detection of the received signal. Doppler causes broadening the spectrum of the received signal and consequently, increases the interference amongst adjacent channels (typically small at reasonable user velocities) [11][18].

Several techniques were investigated and found to be useful to counter the delay spread. These techniques can be categorized into two main parts: signal processing and antenna solutions. In a broad sense, equalization, the subject investigated in this work, involves signal processing approaches used at the receiver to suppress the ISI caused by the multipath phenomena. At the transmitter, some signal processing techniques can be used to make the signal more resistible to delay spread; spread spectrum approach and multicarrier modulation method are salient examples of such signal processing techniques [1]. Antenna solutions have proved effectiveness in dealing with delay spread and reducing or eliminating ISI: distributed antennas, directive antennas, and adaptive antennas have been used widely in this context [14].

Equalization techniques were also broadly divided into linear and nonlinear. Linear techniques are found to be more easier to understand and conceptually the simplest to implement. However, due to noise enhancement problem associated with linear equalization techniques on frequency-selective fading channels, they are not commonly implemented in various wireless applications [14], [8]. Decision-feedback equalization (DFE) is the most commonly used nonlinear equalization techniques due to being robust to noise enhancement as well as being simple to implement [11], [19]. However, error propagation is the main drawback of DFB used on channels with low SNR [20]. Maximum likelihood sequence estimation (MLSE) technique is found to be optimal amongst all techniques being implemented so far [19], [21]. However, due to its exponentially increasing complexity with memory length, its implementation on most channels of interest is rendered impractical. The MLSE is often used as a corner stone with which the performances of other equalization techniques are compared.

### 3.1.1 Adaptive Equalization

In adaptive equalization, a periodically training sequence is applied to the equalizer which is used to make its parameters adapt and track any changes in the communication channel. The transmitter sends a known training sequence of fixed length to the receiver's equalizer which then uses it to modify its weights to an optimum setting that maximize the signal by suppressing echoes resulting from multipath propagation and hence equalizes the time varying channel. Data transition starts immediately once the training sequence terminate. Figure 3.1depicts the channel and equalizer topology.

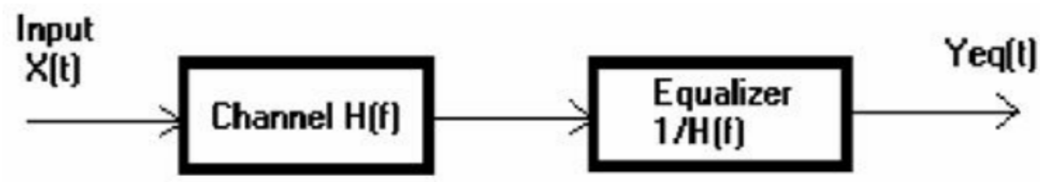


**Figure 3.1 :** General channel and equalizer pair

Equalizers can be categorized as either preset or adaptive equalizers. In preset equalizers, the channel is assumed to be time invariant and depending on H(f), the channel transfer function, the equalizer is designed. The zero forcing equalizer, decision feedback equalizer and minimum mean square error equalizer are examples of this category.

A wireless mobile channel is usually a time varying channel. For such environment it is required to design equalizers whose coefficients are time varying in order to track the continuously changing channel.

Adaptive equalizers operate in the following manner:

- The received signal is applied to a low-pass receive filter. A matched filter is not used because the impulse response of the channel of interest is not known.
- The output of the receive filter, after being sampled at either the symbol rate or twice as much, is applied to adaptive transversal filter equalizer realized as FIR discrete time filters.

- The coefficients are adapted to approach their optimal values that minimize the noise and intersymbol interference at the detector input.
- A generated error signal is derive and used to drive the adaptation process of the equalizer.

Adaptive equalizers operate in either of the two following modes:

- Decision Directed Mode: In this mode the equalizer uses the error signal produced by the receiver decisions. Although this mode is not adequate for initial acquisition, the adjustment based on it is quite satisfactory in tracking the channel response if it varies reasonably slow.
- Training Mode: A training signal is needed during initial acquisition to bring the equalizer to a suitable state. In this mode of operation, data symbol sequence known to the receiver are sent by the transmitter and substituted by the receiver in place of the actual output. After a preset duration, the actual output is substituted and transmission of data starts.

The block diagram, shown in Fig.3.2, represent the functional components of a practical adaptive filter, such as least mean squares (LMS) and recursive least squares (RLS). It is obvious from the diagram that the role of the variable filter is to extract an estimate of the desired signal.

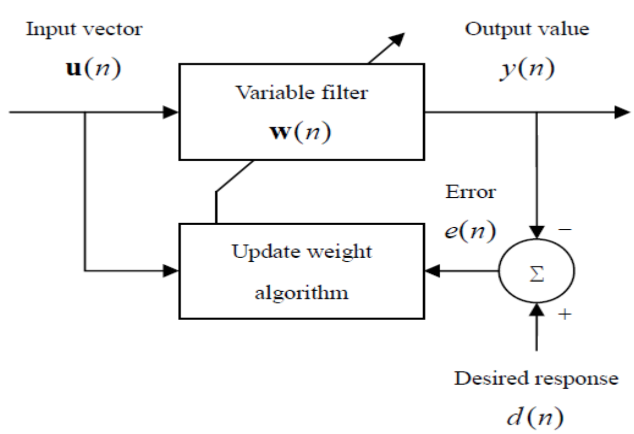


**Figure 3.2 :** A general adaptive filter functional block diagram [2]

The following parameters are used to determine which algorithm is to be a good choice:

- Rate of convergence: This defines the rate, in terms of number of iterations, at which the algorithm converge closely enough to the optimum solution in response to stationary inputs. A faster rate of convergence means the algorithm rapidly adapt to a stationary environment of unknown statistical variations.
- Misadjustment: For a particular algorithm, this parameter indicates quantitatively the amount of deviation of the final value of the mean-square error from the optimal minimum mean square error.
- Tracking: The algorithm is needed to track instantaneously fluctuations in a nonstationary environment.
- Robustness. Minimum estimation errors should only occur when small disturbances happen for a system to be robust.
- Computational requirements: The following requirements should be considered for an algorithm with effective computational cost:

  (a) The number of operations per single iteration of the algorithm.

  (b) The memory required to store both the program and the data.

  (c) The cost of programming the algorithm on a computer.
- Structure: VLSI technology favors the implementation of algorithms that possess favored characteristics such as modularity and parallelism, or concurrency.
- Numerical properties: As an algorithm is executed numerically by the computer inaccuracies are produced due to round-off approximations and representation precisions in the computer. Such practices affect algorithm stability.

## 3.2 THE TRANSVERSAL FILTER

The structures of an adaptive filter should be of finite memory for the sake of implementation feasibility [22]. A transversal filter is commonly realized as a tapped-delay line filter or finite-duration impulse response (FIR) filter. Three basic components are used to build a transversal filter, as depicted in Fig. 3.3: (a) a unit-delay element, (b) an adder, and (c) a multiplier. The finite duration of the filter impulse response is determined by the number of delay elements in the filter structure. M in Fig 3.3 is the filter order and is equal to number of delay elements used in the filter structure. The operator $z^{-1}$ in this figure is the unit-delay operator . Multipliers, also called tap weights, in the filter structure multiplies the taps input by the coefficients of the filter. The adders sum the outputs of all multipliers in the filter producing the overall output. Considering the transversal filter shown in Fig. 3.3, its output is given by

$$y(n) = \sum_{k=0}^{M} w^{*} u(n-k) \tag{3.1}$$

which is merely a finite convolution sum [1].

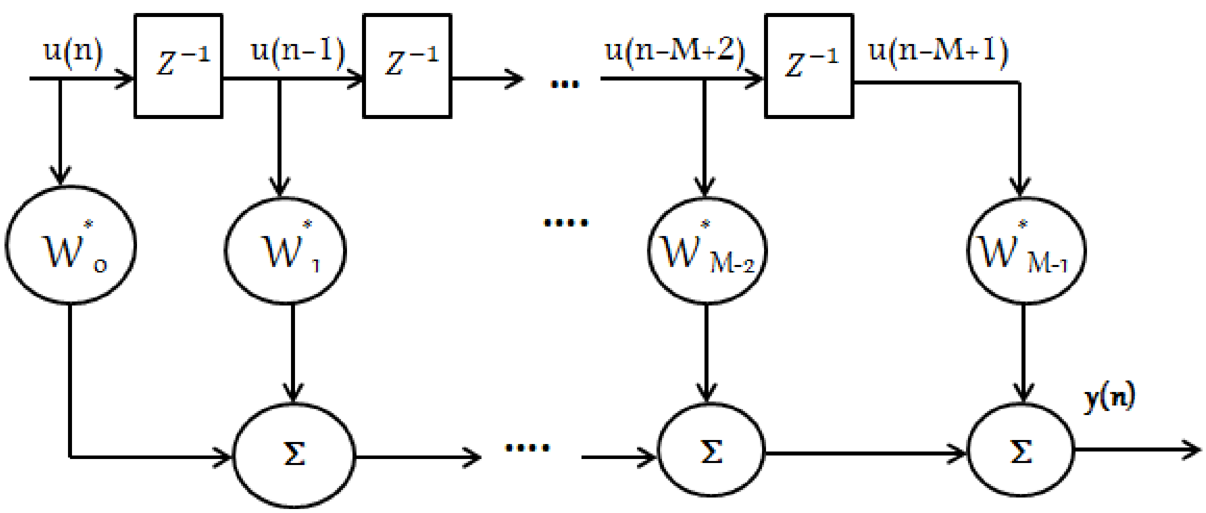


**Figure 3.3 :** The Transversal filter [15]

### 3.3 DECISION FEEDBACK EQUALIZER

In adaptive filtering a recursive algorithm is implemented in order to adapt the filter parameters continuously over successive iterations. The filter parameters are therefore data dependent. The adaptive filter is a nonlinear system since superposition principles are not obeyed by the filter.

Having said that the adaptive filter might be classified as a linear system if superposition principles apply when its parameters are held fixed.

The decision-feedback equalizer (DFE) is a nonlinear adaptive filter. It consists of two major parts; a feedforward component and a feedback part. In the feedforward part a transversal filter with taps equispaced at the reciprocal of the symbol rate is implemented. The data sequence at the output of the channel is applied to the input of this section. The feedback section consists of another transversal filter whose taps are also spaced at the reciprocal of the symbol rate.

Data obtained from decisions made by the system on previously detected symbols is applied to the feedback section of DFE. The feedback section subtracts out some of the intersymbol interference introduced by symbols, previously detected, from the estimates of future symbols. This cancellation is an old idea known as the bootstrap technique [18].

A decision-feedback equalizer yields good performance in the presence of severe intersymbol interference, as is experienced in deep fading of radio channels. Fig. 3.4 shows the block diagram of DFE system [1], [22].

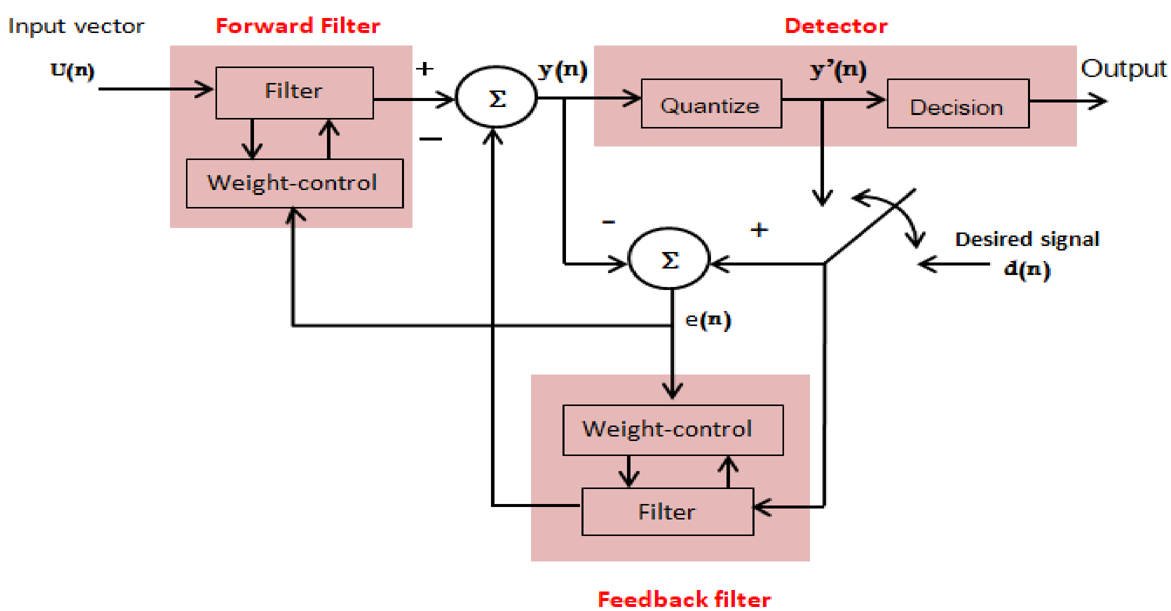


**Figure 3.4** : A block diagram of DEF [2]

## 3.4 ALGORITHMS FOR ADAPTIVE EQUALIZATION

There are many adaptation algorithms described in the literature. The two most prevailing types are the (LMS) and (RLS).

### 3.4.1 Least Mean Squares Algorithm (LMS)

Least mean squares (LMS) algorithms are used to drive the filter coefficients toward their optimum values which minimize the mean squares of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method because the coefficients of the filter are adapted using the error currently obtained.

LMS filter is realized using a transversal filter structure. The is used in various applications due to its high performance and simplicity of design [1], [21], [23]. As was mentioned earlier, the LMS algorithm is designed based on a transversal filter structure which is actually doing the filtering operation. A mechanism is organized to control the adaptation process on the taps weights of the transversal filter.

The LMS algorithm in general, is made up of two basic procedures:

- Filtering process: In this process the output of a linear filter is computed in response to the signal applied at the filter input. The error of estimation is calculated by comparing the actual filter output with a desired response.
- adaptive process: To automatically adjust the parameters of the filter under the supervision of the estimation error.

$$\widehat{W}(\text{n+1}) = \widehat{W}(\text{n}) + \mu(\text{u})e^{*}(\text{n}) \tag{3.2}$$

Where μ is the step size, $W$(n+1) is an estimate vector of taps weights at instant (n+1).

These two processes work together to constitute a feedback loop. The transversal filter, around which the LMS algorithm is built performs the filtering process. Concurrently, the adaptive control process mechanizm operates on the tap weights of the transversal filter [24]–[26].

LMS algorithm has specific characteristics such as the constant value that is proportional to the equalizer input multiplied by output error. The other important feature of the algorithm is the high speed with which the algorithm executes but at the expense of rather slow convergence rate which makes the equalizer ineffective when the channel parameters change quickly. Finally, the complexity of the LMS algorithm increases linearly with the increase in number of filter coefficients [27]–[29].

### 3.4.2 Recursive Least Square Algorithm (RLS)

This algorithm recursively modifies the filter coefficients to minimize a weighted linear least squares cost function which is related to the input signals. When deriving the RLS, input deterministic signals are assumed. This is in contrast LMS and similar algorithms which are considered stochastic. The RLS algorithm exhibits extremely fast convergence rate when compared to its competitors.

However, the convergence advantage of RLS comes at the cost of high computational complexity. Consequently, the performance of RLS in tracking is significantly poor when the filter to be estimated changes.

The RLS algorithm is known to be stable because of the covariance update formula $\rho(n)$, which is used for automatic adjustment in accordance with the estimation error as follows [22], [24], [30]:

$$\rho(0) = \delta^{-1} I \tag{3.3}$$

$\rho$ is the inverse correlation matrix and $\delta$ is a regularization parameter, it is a positive constant for high SNR and negative constant for low SNR.

At each iteration, n = 1, 2, 3, ….

$$\pi(n) = \rho(\mathrm{n}-1)\ \mathrm{u}\ (\mathrm{n}) \tag{3.4}$$

$$k(n) = \frac{\pi(n)}{\lambda + u^{H}(n)\pi(n)} \tag{3.5}$$

The gain vector, which varies with time is given by

$$\xi(n) = \mathrm{d}(\mathrm{n}) - \widehat{W}^{H}(\mathrm{n}-1)\mathrm{u}(\mathrm{n}) \tag{3.6}$$

while the priori estimation error is

$$\widehat{W}(n) = \widehat{W}(n-1) + k(n)\, \xi^{*}(n) \tag{3.7}$$

## 3.5 SUMMARY OF ALGORITHMS

The LMS equalizer operates on minimization of the mean square error (MSE) defined as the difference between the actual and the desired equalizer outputs. This strategy makes the equalizer more robust. However, the convergence rate of the LMS algorithm is quite slow which renders the LMS equalizer ineffective in fast varying channel parameters.

To allow for faster convergence rate, complex algorithms based on a least squares approach must be used in LMS algorithm, as opposed to the statistical minimum mean square error approach [31]–[33].

# 4. EXPERIMENTAL SETUPS AND SIMULATIONS

## 4.1 THE SIMULINK AS A SIMULATION TOOL

The MATLAB's Simulink is used to build the simulation models. Simulink is a software package for modeling, simulating, and analyzing dynamic systems. It supports linear and nonlinear systems modeled in continuous time, sampled time, or a hybrid of the two [34].

When modeling, the graphical user interface (GUI) provided by the Simulink is used to build, models in the form of block diagrams, using click-and-drag mouse operations. Simulink includes a comprehensive block library of sinks, sources, linear and nonlinear components, and connectors [34]–[36].

Toolboxes are available in MATLAB that contain model analysis tools such as linearization, trimming and a rich variety of other tools which can be accessed from a MATLAB command line. Models can be simulated, analyzed, and revised at any point in either MTLAB or Simulink as the two environment are integrated [30], [35], [37].

## 4.2 DESCRIPTIVE ANALYSIS

The purpose of this simulation is to consider the implementation, at the receiver side, of various equalizers for the transmission of wireless digital data and evaluate their relative performances. Before beginning, it is worth to consider the block diagram of a general digital data transmission system, shown in Figure 4.1, and discuss, in descriptive manner, each functional element in it.

- **Tx Data**: Wide variety of possible sources of information results in many different forms for messages. Here, we always discuss the transmission of digital signals. Digital signals may be divided into two main classes. In one category, signals are inherently both discrete and quantized. In the other class, signals are originally continuous (or analogue) and transformed into digital by sampling, quantizing, and coding.
- In most applications, digital signals are represented as binary numbers, so their quantization is measured in bits. The **Random Integer Generator block** of MATLAB's Simulink is used to generate uniformly distributed random integers, as shown in Fig. 4.2.

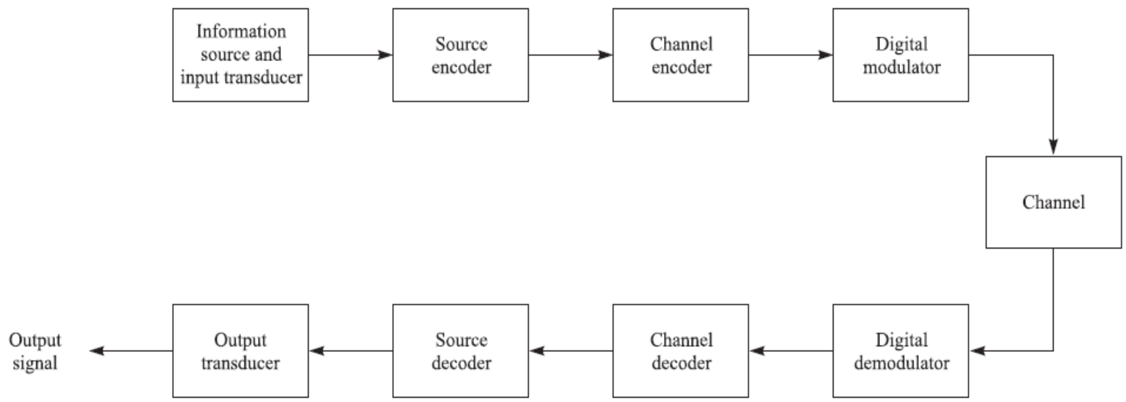


**Figure 4.1 :** A basic digital communication system implementing an adaptive equalizer [1]

- **Channel Coding**: With a sufficiently sophisticated channel encoder and decoder, digital information can be transmitted over the channel at rates bounded by the channel capacity and with arbitrarily small errors. The encoder at the transmission side converts a signal or data into a code. This code may be used to serve various purposes; for instance compressing information for transmission or storage, encrypting or adding redundancies to the input code and uses the redundancies to counter the impairments of the propagation link on the transmitted code [38]. Any appropriate encoder block can be chosen and used to map input/output sequence vectors. However, in the work presented here, equalization is the main concern and therefore channel coding is not implemented.
- **Modulation**: The general, **BPSK Modulator Baseband blocks** are used here to modulate transmitted signals using phase shift keying Methods.
- **Channel: AWGN channel, digital filter,** and **multipath fading channel blocks** are used to simulate a real channel.
- **Equalizer**: Equalizers are already described in Chapter 3. The **Linear Equalizer and the Decision Feedback Equalizer blocks** with two types of chosen algorithms for adaptation **(LMS and RLS)** are used to equalize a linearly modulated baseband signal transmitted through nonstationary dispersive channels.
- **Demodulation**: In digital communication systems, it is a common practice that the design of the modulator and demodulator must be done as a single entity. In all digital

communication systems, the demodulator at the receiver is designed so that it reverse the operations performed by the modulator at the transmitter.

***BPSK Demodulator Baseband blocks*** are used as in Fig. 4.2.

- **Rx Data:** The receiver data (Rx) input is the output of channel decoder. The detected received data are **compared** with the input data in order to evaluate bit error rate.

- **Performance Analysis:** A primary measure of system performance for digital data communication systems is the probability of bit error rate (BER). The **Error Rate Calculation block** is used to compare input data to a transmitter with output data from a receiver, as depicted in Fig.4.2. The error rate is calculated as a running statistic, by dividing the total number of erroneous received data bits by the total number of data bits transmitted to the receiver.

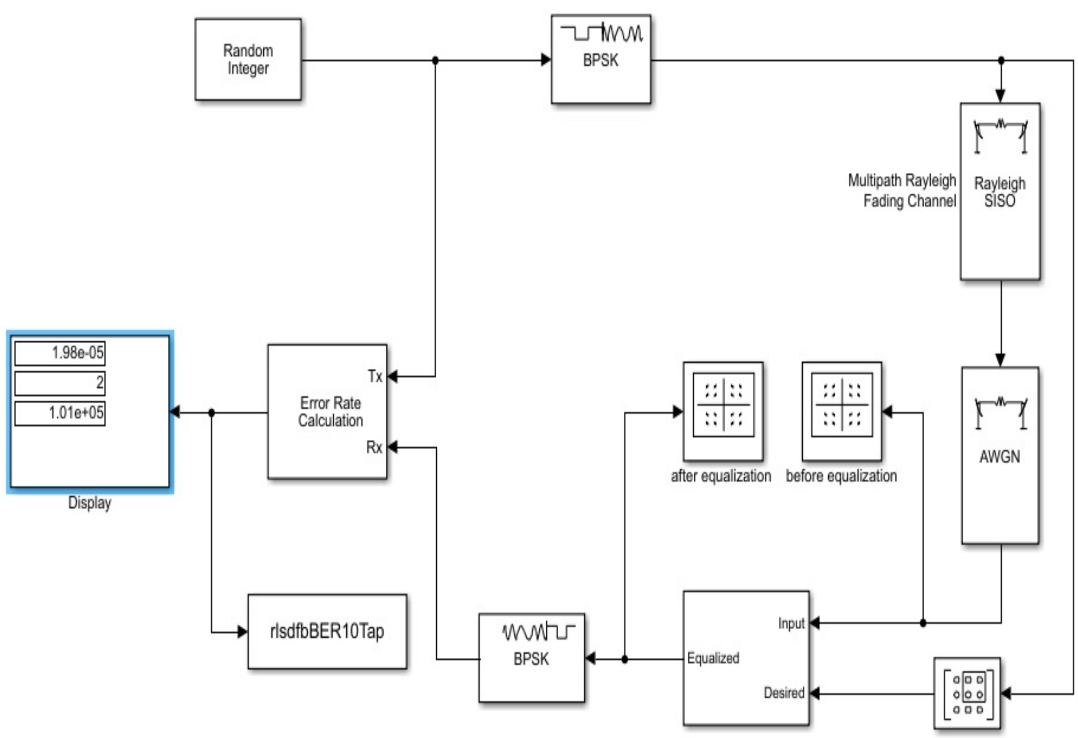


**Figure 4.2 :** MATLAB'S Simulink model used in the simulation

### 4.3 COMPUTER MODEL

Figure 4.3, presents a flow chart illustration of the simulation to be run in this work. It is executed once for each setting of the system giving a particular curve.

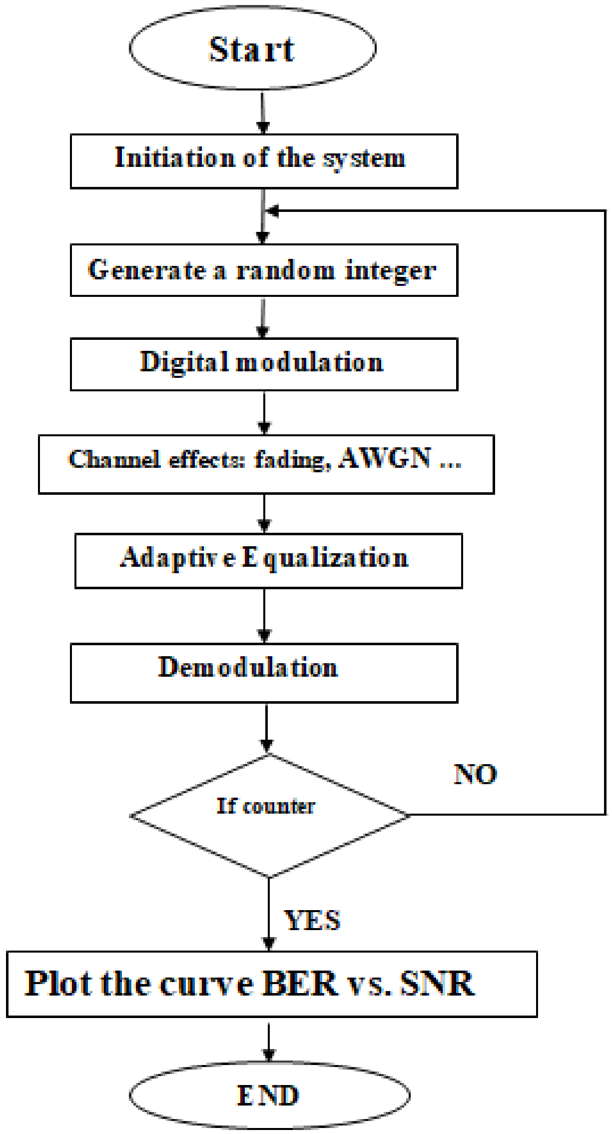


**Figure 4.3 :** The simulation Flow chart

## 4.4 SIMULATION

- **Information source**: A random generator is implemented to generate integers that are randomly distributed in the range [0, M-1]. M represents the M-ary number defined in the dialog box. For the BPSK considered here, M is set equal to 2.
- **BPSK**: The modulation used in this simulation is the simple binary phase shift keying, BPSK which is the simplest modulation scheme. The output of this block is a baseband representation of the modulated signal.
- **Multipath channel:** AWGN Channel is normally used; a Digital Filter and the Multipath Fading Channel blocks are used to simulate real channels.
- **LMS/RLS linear equalizer:** Equalizing using linear and nonlinear (decision feedback) equalizer that updates weights using LMS/RLS algorithms are employed in this investigation. These structures were used to equalize a baseband signal linearly modulated and propagated through a fading channel. LMS/RLS algorithms were simulated and used to appropriately modify the weights, once per symbol. The block implements either a symbol-spaced equalizer or a fractionally spaced equalizer for a single sample per symbol sampling or double samples per symbol sampling respectively.
- **Discrete-Time Scatter Plot Scope:** This block displays scatter plots of a modulated signal focusing on some of the modulation characteristics, such as pulse shaping or the distortions introduced by the channel on the signal.
- **BPSK Demodulator Baseband:** The BPSK Demodulator Baseband block demodulates the binary phase shift keyed signal which had been modulated and transmitted by the receiver.
- **Error Rate Calculation:** Bit error rate is a performance measure of a communication system. The Error Rate Calculation block counts the received erroneous bits by comparing input data from a transmitter with input data from a receiver output. The error rate is calculated as a running statistic, by dividing the total number erroneous received data by the total number of data generated and transmitted by the source.

# 5. SIMULATION RESULTS AND DISCUSSIONS

## 5.1 PHASE ONE; SIMULATION RESULTS

Two types of equalizers were used in this investigation; a linear equalizer and a decision feedback type of equalizer. With each equalizer two kinds of adaptation algorithm, as mentioned previously, were used. The experimental sets up parameters were almost kept the same throughout the experimentation and a table of each is given beside the corresponding result for each experiment. In phase one of experimentations the number of propagation paths is kept fixed and the performance, in terms of bit error rate, is recorded for a variety of filters lengths as the energy per bit to noise power ratio is varied. In phase two, the number of propagation paths of the simulated wireless channel is varied while the number of taps for a given equalizer used is fixed. The performance in terms of bit error rate is similarly recorded for a range of SNR values. The results obtained will be presented next in the form of graphs. Discussions, explanations and suggestions are given appropriately.

### 5.1.1 Linear equalizer with LMS algorithm

Fig. 5.1, clearly shows that at low to moderate SNR (up to about 12 dBs) the number of taps of the equalizer has no effect on the performance of the system, but beyond 12 dBs it is found that increasing the taps will adversely affect the performance. This is justified on the bases that only three paths were considered in the scenario and at higher SNR increasing the complexity of the equalizer will decrease its rate of convergence and consequently increases the bit error rate.

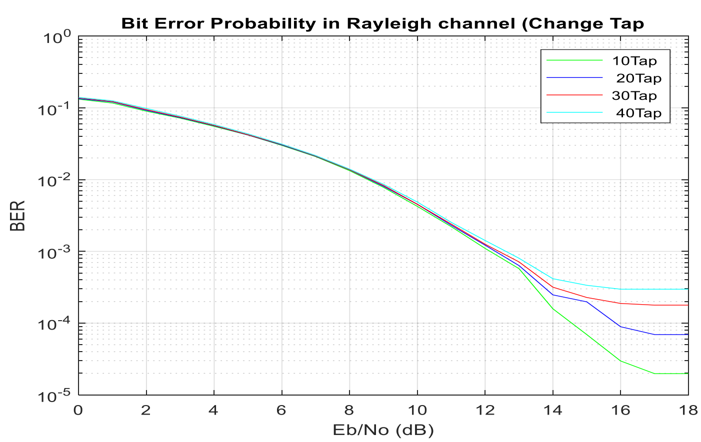


**Figure 5.1 :** BER performance for linear equalizer with LMS algorithm

**Table 5.1** Operating parameters for linear equalizer with LMS algorithm

| Component | Parameters | Value |
|---|---|---|
| Variable in Matlab model | M | 2 |
| | Tsym | 0.0001 |
| | EbNoVec | 0 : 1 : 18 |
| Random Integer Generator | Set size | M |
| | Sample time | Tsym |
| | Samples per frame | 1000 |
| BPSK Modulator Base Band | Phase offset (rad ) | 0 |
| | Output data type | Double |
| Rayleigh Fading | Maximum Doppler shift | 0.04 |
| | Discrete path delay vector | [ 0 , 0.00000002 , 0.00000003 ] |
| | Average path gain Vector | [ 0, -8 , -12 ] |
| | Initial seed | 73 |
| AWGN Channel | Number of bits per symbol | log 2( M |
| | Symbol period | Tsym |
| Equalizer | Number of taps (10,20,30,40) | 10, 20, 30, 40 |
| | Number of samples per symbol | 1 |
| | Reference tap to (10,20,30,40) | 3, 6, 7, 11 |
| | Step size | 0.0009 |
| BPSK Demodulator Baseband | Decision type | Hard decision |
| | Phase offset | 0 |
| Error Rate Calculation | Receive delay to (10,20,30,40) tap | 3, 6, 7, 11 |
| | Target number of error | 10000 |
| | Maximum number of symbols | 10000000 |

### 5.1.2 Linear equalizer with RLS algorithm

For this algorithm the behavior of the system shows a little effect when increasing the equalizer taps. Fig. 5.2 clearly illustrates this finding for SNR up to about 14 dB. Higher SNR indicates an adverse relationship between the bit error rate and the number of taps of the equalizer as was found with the LMS algorithm and as was deduced earlier.

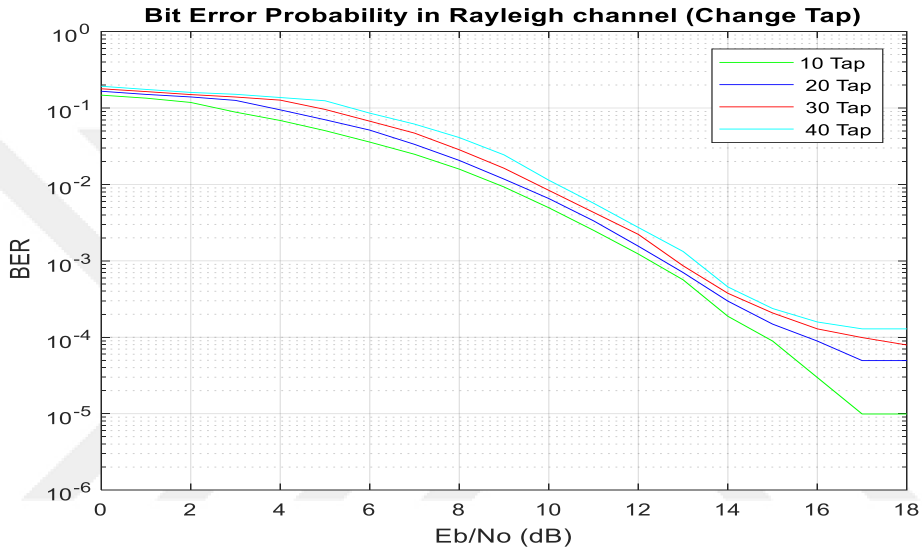


**Figure 5 2 :** BER performance for linear equalizer with RLS algorithm

**Table 5.2:** Operating parameters for linear equalizer with RLS algorithm (Other parameters are set in Table 5.1)

| Component | Parameters | Value |
|---|---|---|
| Equalizer | Number of taps (10,20,30,40) | 10, 20, 30, 40 |
| | Number of samples per symbol | 1 |
| | Reference tap to (10,20,30,40) | 3, 6, 9, 12 |
| | Forgetting factor | 0.99 |

### 5.1.3 Decision feedback equalizer with LMS algorithm

Fig. 3.4 illustrates the BER rate performance of the decision feedback equalizer for four different tap-lengths; 10 taps, 20 taps, 30 taps, and finally 40 taps. The Figure clearly shows that at low SNR (up to about 6 dB) the performance of the system is very poor. Over the range 6 to 13 dB, the performance improves significantly. However, over this range varying the tap lengths has no effect as well. Beyond 13 dB, we notice that increasing the tap-lengths will have a negative impact on the system performance, in consistence with the behavior of its previous two counterparts.

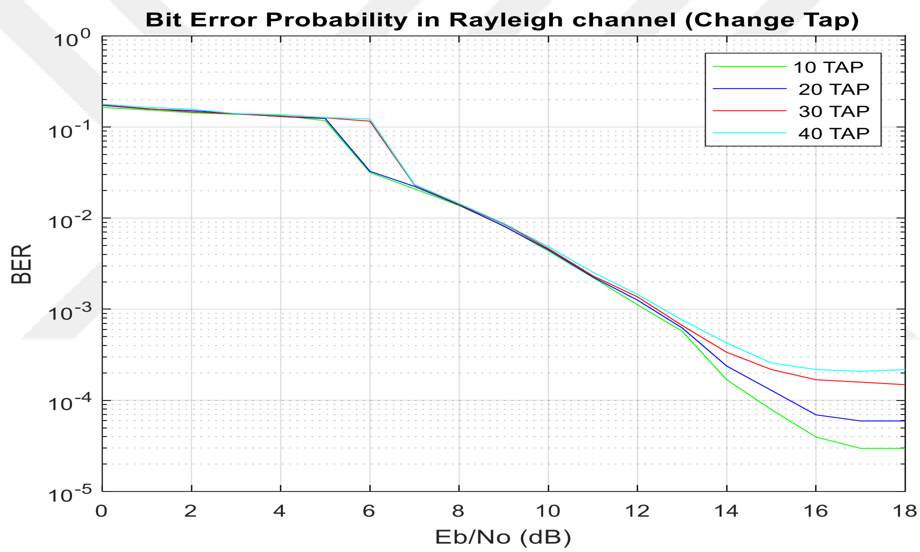


**Figure 5.3:** BER performance for a decision feedback equalizer with LMS algorithm

**Table 5.3:** Operating parameters for decision feedback equalizer with LMS algorithm (Other parameters are set as in Table 5.1)

| Component | Parameters | Value |
|---|---|---|
| Equalizer | Number of forward taps is (10,20,30,40) | 4, 9, 14,19 |
| | Number of feedback taps is (10,20,30,40) | 6, 11, 16, 21 |
| | Number of samples per symbol | 1 |
| | Reference tap is (10,20,30,40) | 3, 6, 9, 12 |
| | Step size | 0.0009 |

### 5.1.4 Decision feedback equalizer with RLS algorithm

The performance of the system incorporating this type of equalizer that adopts this kind of adaptation algorithm shows similar behavior to the last system.

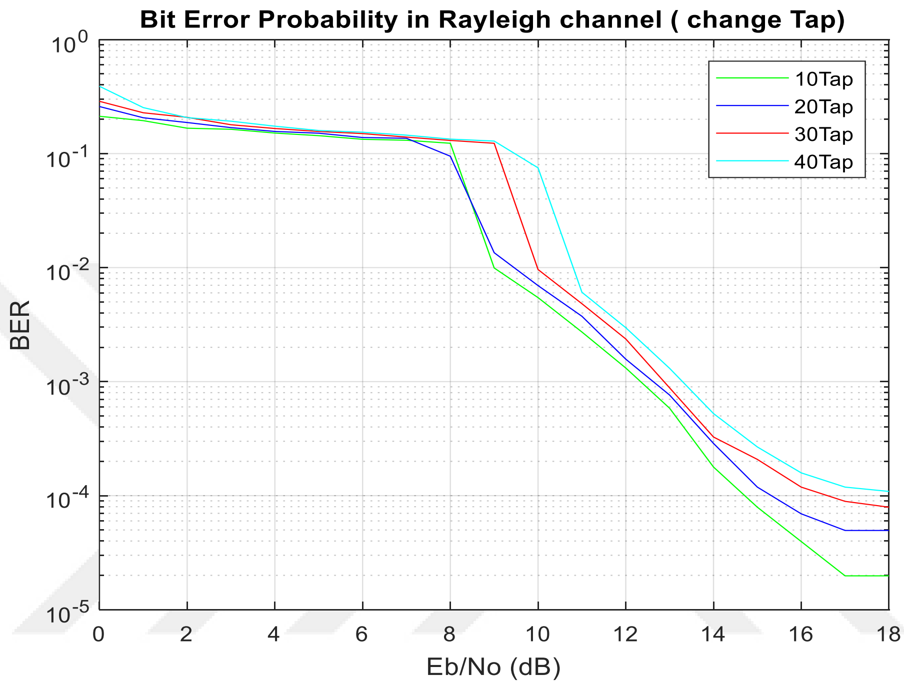


**Figure 5.4:** BER performance for decision feedback equalizer with RLS algorithm

**Table 5.4**: Operating parameters for decision feedback equalizer with RLS algorithm (Other parameters are set as in Table 5.1)

| Component | parameters | Value |
|---|---|---|
| Equalizer | Number of forward taps is (10,20,30,40) | 4, 9, 14,19 |
| | Number of feedback taps is (10,20,30,40) | 6, 11, 16, 21 |
| | Number of samples per symbol | 1 |
| | Reference tap is (10,20,30,40) | 3, 6, 9, 12 |
| | Step size | 0.0009 |

## 5.2 PHASE TWO; SIMULATION RESULTS

In this part of investigation,, the effect of the number of paths (along which the transmitted signal propagates) on the performance of a system implementing equalization is investigated. The system is modeled using Simulink software package. The performance is measured in terms of bit error rate (BER). Two different equalizer’s structures are used, as in the first part; linear (tapped line) equalizer and decision feedback (nonlinear) equalizer. With each one of them, two adaptation algorithms (the most commonly used ones) are employed. They are; the least mean squares (LMS) algorithm and the recursive least square (RLS) algorithm. The scenario is to have up to seven paths in the channel model with corresponding gains and phase delays. The length of the equalizer (number of taps) is fixed for both structures, and for each particular number of paths; a bit error rate graph is evaluated. The procedure is repeated for the four different models. The results obtained are presented as follow:

Fig. 5.5 depicts the behavior of the system implementing the linear equalizer with LMS adaptation algorithm. The length of the filter is kept fixed while the number of the channel paths is varied. The same practice is repeated with other filter structures adopting other algorithms as shown in Figures 5.6, 5.7, and 5.8, corresponding tables are given appropriately.

### 5.2.1 Linear equalizer with LMS algorithm

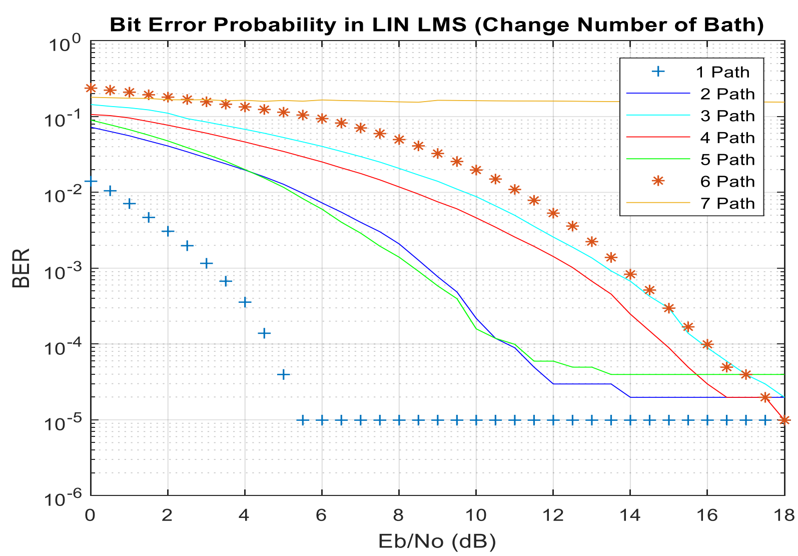


**Figure 5.5:** BER performance for linear equalizer with LMS algorithm

**Table 5.5:** Operating parameters for linear equalizer with LMS algorithm

| Component | parameters | Value |
|---|---|---|
| Variable in Matlab model | M | 2 |
| | Tsym | 0.0001 |
| | EbNoVec | 0 : 1 : 18 |
| Random Integer Generator | Set size | M |
| | Sample time | Tsym |
| | Samples per frame | 1000 |
| BPSK Modulator Base Band | Phase offset (rad ) | 0 |
| | Output data type | Double |
| Rayleigh Fading | Maximum Doppler shift | 0.04 |
| | Discrete path delay vector to number of Path ( 1 , 2 , 3 , 4 , 5 ,6 , 7 ) | [0 0.00000002 0.00000003 0.00000004 0.00000005 0.00000006 0.00000007] |
| | Average path gain Vector to number of Path ( 1 , 2 , 3 , 4 , 5 ,6 , 7 ) | [0 -6 -9 -12 -13 -17 -19] |
| AWGN Channel | Number of bits per symbol | log 2( M |
| | Symbol period | Tsym |
| Equalized | Number of taps | 5 |
| | Number of samples per symbol | 1 |
| | Reference tap | 2 |
| | Step size | 0.0009 |
| BPSK Demodulator Baseband | Decision type | Hard decision |
| | Phase offset | 0 |
| Error Rate Calculation | Receive delay | 2 |
| | Target number of error | 10000 |
| | Maximum number of symbols | 10000000 |

### 5.2.2 Linear equalizer with RLS algorithm

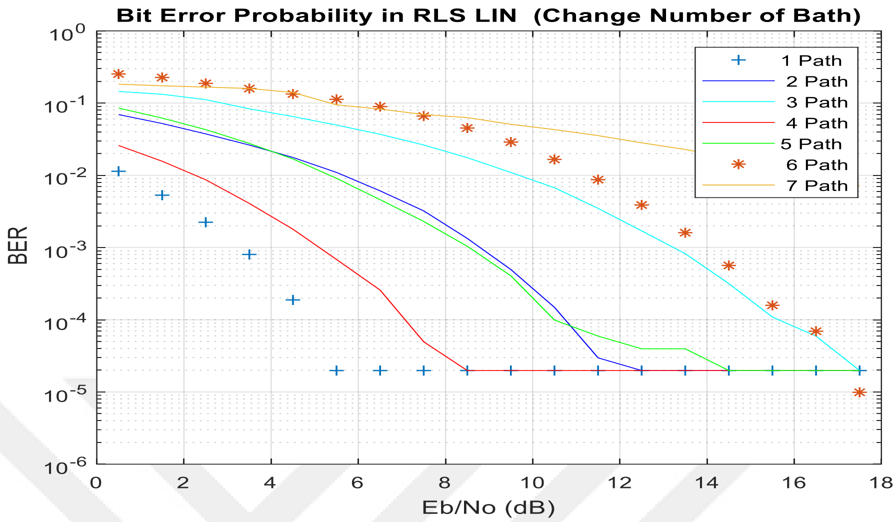


**Figure 5.6:** BER performance for linear equalizer with RLS algorithm

**Table 5.6:** Operating parameters for linear equalizer with RLS algorithm (Other parameters are set as in Table 5.5)

| components | parameters | Values |
|---|---|---|
| Equalized | Number of taps | 5 |
| | Number of samples per symbol | 1 |
| | Reference tap | 2 |
| | Forgetting factor | 0.99 |

### 5.2.3 Decision feedback equalizer with LMS algorithm

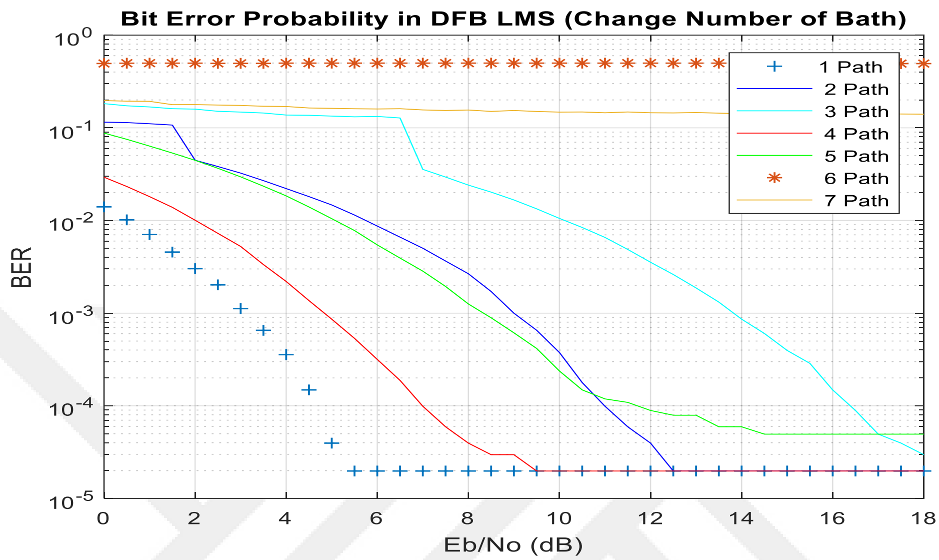


**Figure 5.7: BER** performance for DFB equalizer with LMS algorithm

**Table 5.7:** Operating parameters for decision feedback equalizer with LMS algorithm ( Other parameters are set as in Table 5.5)

| Component | parameters | Value |
|---|---|---|
| Equalized | Number of forward taps | 2 |
| | Number of feedback taps | 3 |
| | Number of samples per symbol | 1 |
| | Reference tap | 2 |
| | Step size | 0.0009 |

### 5.2.4 Decision feedback equalizer with RLS algorithm

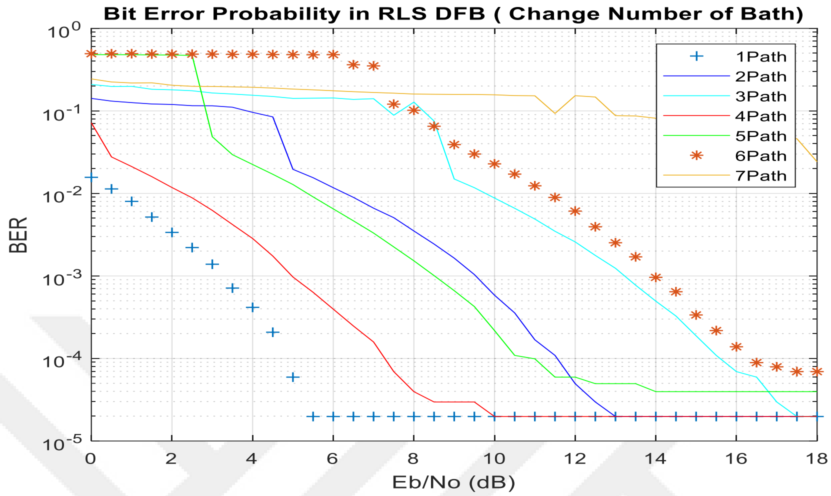


**Figure 5.8:** BER performance for DFB equalizer with RLS algorithm

**Table 5.8:** Operating parameters for decision feedback equalizer with RLS algorithm ( Other parameters are set as in Table 5.5)

| Component | parameters | Value |
|---|---|---|
| Equalized | Number of forward taps | 2 |
| | Number of feedback taps | 3 |
| | Number of samples per symbol | 1 |
| | Reference tap | 2 |
| | Step size | 0.0009 |

# 6. CONCLUSION

The simulation conducted in the first part of this work and the associated results obtained clearly indicate that in such time-varying environment there can be no definite relationship(s) among the structure (type and complexity) of the equalizer, the adaptation algorithm and their impact on the performance of the communication system that uses equalization technique to alleviate the detrimental effect (i.e. ISI) which results due to multipath phenomena. The number of dominant multipath in a particular scenario intuitively recommend a particular value of tap-length in the implemented equalizer, however the nature of the environment (i.e. nonstationary) keeps such dominating paths not fixed in number. The results obtained revealed the an excessive increase in the number of taps as compared to the number of paths of the wireless channel will decrease the equalizer efficiency. This is expected since increasing the complexity of the equalizer will slow its rate of convergence. Consequently, in a time varying environment, higher error rate occurs during the adaptation process duration.

In the second phase of experimentations, the results for the four models show that the best performance is obtained when the number of the multipath (excluding) the main (LOS) path is equal to or less than the number of taps of the equalizer. For a single path, we found best performance for all scenarios. This is intuitive since a single path means that a single copy of the signal arrives at the receiver and hence no ISI is present. These findings encourage a further investigation in implementing suitable techniques of sensing the channel to establish the number of significant multipath and devising a way of adaptively changing the number of taps in the implemented equalizer. The effect of the reference tap in the transversal structure, and the number of taps in the forward and backward in addition to the reference tap in DFB structure are to be determined for optimal performance when varying the tap-lengths of the implemented equalizer.

As recommended for any researcher interesting in pursuing this topic to use a higher order modulation such as QPSK and even higher which are currently used in common applications.

Also, some sort of practical channel coding is to be implemented in the system to make it more robust. Another issue which is worth to investigate is the criteria of rate of convergence for the adaptation algorithm. This is a crucial feature of equalization since it affects the system performance in a time-varying environment.

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