CHANNEL ESTIMATION IN MULTI CARRIER CODE DIVISION MULTIPLE ACCESS

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CERTIFICATE

This is to certify that the thesis entitled "**Channel Estimation in Multicarrier Code Division Multiple Access**" submitted by **Amit Kumar Bhuyan** and **Iswar Chandra Patel** in partial fulfillment for the requirements for the award of Bachelor of Technology Degree in **Electronics & Communication Engineering** at National Institute of Technology, Rourkela (Deemed University) is an authentic work carried out by them under my supervision and guidance.

To the best of my knowledge, the matter embodied in the thesis has not been submitted to any other University / Institute for the award of any Degree or Diploma.

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ABSTRACT

The concepts of OFDM and MC-CDMA are not new but the new technologies to improve their functioning is an emerging area of research. In general, most mobile communication systems transmit bits of information in the radio space to the receiver**.** The radio channels in mobile radio systems are usually multipath fading channels, which cause inter-symbol interference (ISI) in the received signal. To remove ISI from the signal, there is a need of strong equalizer.

In this thesis we have focused on simulating the OFDM and MC-CDMA systems in MATLAB and designed the channel estimation for them.

Key words: OFDM, MC-CDMA, channel estimation, pilot symbols, DFT,BER, SNR, LS estimator , LMMSE estimator, AWGN.

LIST OF ACRONYMS

AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
CDMA	Code Division Multiple Access
DFT	Discrete Fourier Transform
FIR	Finite Impulse Response
FFT	Fast Fourier Transform
IDFT	Inverse Discrete Fourier Transform
ICI	Inter Carrier Interference
ISI	Inter Symbol Interference
LSE	Least Square Estimation
LMMSE	Minimum Mean Square Estimation
MC-CDMA	Multicarrier Code Division Multiple Access
MMSE	Minimum Mean Square Estimation
MSE	Mean Square Error
OFDM	Orthogonal Frequency Division Multiplexing
PSK	Phase Shift Keying
QPSK	Quadrature Phase Shift Keying
SER	Symbol error rate
SNR	Signal to Noise Ratio

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CHAPTER-1
INTRODUCTION

1.1 INTRODUCTION

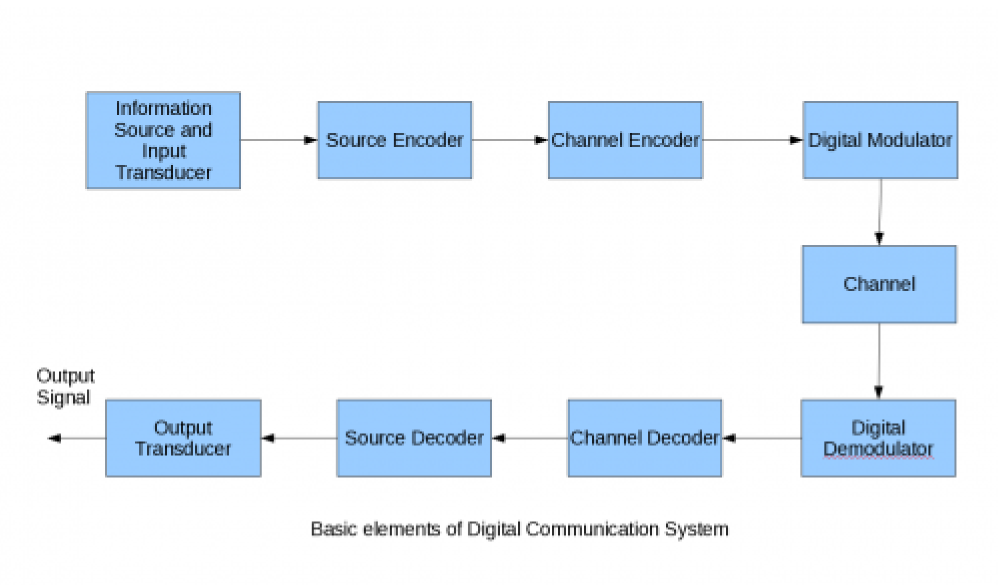
Guglielmo Marconi first discovered radios ability to provide continuous contact with ships in 1897,since then new wireless communication methods have been evolving year by year throughout the world. Particularly during the last decade the mobile radio communication industry has boomed ,this trend is expected to continue during the next decade.

In the current and future mobile communications systems, data transmission at high bit rates is essential for many services such as video, high quality audio and mobile integrated service digital network. When the data is transmitted at high bit rates, over mobile radio channels, the channel impulse response can extend over many symbol periods, which leads to Inter-symbol interference (ISI). Orthogonal Frequency Division Multiplexing (OFDM) is one of the promising candidate to mitigate the ISI.In an OFDM signal the bandwidth is divided into many narrow sub-channels which are transmitted in parallel. Each sub-channel is typically chosen narrow enough to eliminate the effect of delay spread. By combining OFDM with CDMA dispersive fading limitations of the cellular mobile radio environment can be overcome and the effects of co-channel interference can be reduced.

1.2 Digital Communication Systems

A digital communication system is often divided into several functional units.

The task of the source encoder is to represent the digital or analog information by bits in an efficient way. The bits are then fed into the channel encoder, which adds bits in a structured way to enable detection and correction of transmission errors. The bits from the encoder are grouped and transformed to certain symbols, or waveforms by the modulator and waveforms are mixed with a carrier to get a signal suitable to be transmitted through the channel. At the receiver the reverse function takes place. The received signals are demodulated and soft or hard values of the corresponding bits are passed to the decoder. The decoder analyzes the structure of received bit pattern and tries to detect or correct errors. Finally, the corrected bits are fed to the source decoder that is used to reconstruct the analog speech signal or digital data input. The main question is how to design certain parts of the modulator and demodulator to achieve efficient and robust transmission through a mobile wireless channel. The wireless channel has some properties that make the design especially challenging: it introduces time varying echoes and phase shifts as well as a time varying attenuation of the amplitude (fading).



(Fig1. 1)

CHAPTER-2

BASICS

2.1 BASICS

This chapter explains the main concepts needed to understand an OFDM system. The principles behind OFDM are quite simple but it is important to thoroughly understand these. It is the basics we have spent most time to understand. The following background sections are not frequently referenced as they are mostly basic *open* knowledge which we attempt to insightfully explain.

2.2 DFT(Discrete Fourier Transform)

In mathematics, the **discrete Fourier transform (DFT)** is a specific kind of Fourier transform, used in Fourier analysis. It transforms one function into another, which is called the frequency domain representation, or simply the DFT, of the original function (which is often a function in the time domain). But the DFT requires an input function that is discrete and whose non-zero values have a limited (*finite*) duration. Such inputs are often created by sampling a continuous function, like a person's voice. Unlike the discrete-time Fourier transform (DTFT), it only evaluates enough frequency components to reconstruct the finite segment that was analyzed. Using the DFT implies that the finite segment that is analyzed is one period of an infinitely extended periodic signal; if this is not actually true, a window function has to be used to reduce the artifacts in the spectrum. For the same reason, the inverse DFT cannot reproduce the entire time domain, unless the input happens to be periodic (forever). Therefore it is often said that the DFT is a transform for Fourier analysis of finite-domain discrete-time functions. The sinusoidal basis functions of the decomposition have the same properties.

The input to the DFT is a finite sequence of real or complex numbers (with more abstract generalizations discussed below), making the DFT ideal for processing information stored in computers. In particular, the DFT is widely employed in signal processing and related fields to analyze the frequencies contained in a sampled signal, to solve partial differential equations, and to perform other operations such as convolutions or multiplying large integers. A key enabling factor for these applications is the fact that the DFT can be computed efficiently in practice using a fast Fourier transform (FFT) algorithm.

2.3 Convolution

The concept of convolution is important because the property of circular convolution will come into play. We all know that a property of the Fourier transform is that multiplication in the frequency domain has the effect of convolution in the time domain. Meaning

$$y\,(\dot{\omega})= h(\dot{\omega})\; x(\dot{\omega})$$

and $$y[n] = h[n] * x[n]$$

where convolution is defined as $(h * x)$
Convolution of two discrete sequences of length M and N results in a sequence of length $M + N - 1$.

The DFT has a similar but different property. Multiplication in time results in a circular convolution in the frequency domain and produces a sequence of length N for a DFT defined by $0 \cdot n < N$. This is a property important in the use of the Cyclic Prefix.

2.4 Multipath Channels

The concept of multipath propagation is very simple, there is no need for a diagram. Whenever a signal is transmitted and there are obstacles and surfaces for reflection, multiple reflected signals of the same source can arrive at the receiver at different times. Such “echoes” will no doubt effect other incoming signals. These echoes are directly influenced by the material properties of the surface it reflects or permeates. These can be relative to dieletric constants, permeability, conductivity, thickness, etc. These echoes can contain useful information but we will not concern ourselves with that field of study in this thesis.Multipath propagation in indoor environments are traditionally modeled using ray tracing which just calculates all the reflections. If no measure are taken to counter the echoes, then these echoes will interfere with other signals causing ISI. We will now describe multipath with some mathematical terminology.

CHAPTER-3

Orthogonal Frequency Division Multiplexing

3.1 INTRODUCTION

The principles of orthogonal frequency division mulitplexing (OFDM) modulation have been around for several decades. However in recent years, this technology has quickly moved out of the academia world into the real world of modern communication systems. New advances have brought a fresh face to the benefits of OFDM in data delivery systems over phone lines, digital radio, television and, most importantly, wireless networking systems. In recent years OFDM scheme has become the underlying technology for various emerging applications such as digital audio/video broadcast, wireless LAN (802.11a and HiperLAN2), broadband wireless (MMDS, LMDS), xDSL, and home networking. Programmable logic devices (PLDs) are playing a fundamental role by facilitating the deployment of OFDM based systems worldwide by making it easier for the engineers to integrate complex intellectual property (IP) blocks and utilize the benefits of high-performance PLD architecture.

3.2 BASICS

OFDM is the acronym of "Orthogonal Frequency Division Multiplexing". There are two key terms here: orthogonality and multiplexing.

The basic principle of OFDM is to split a high-rate data stream into a number of lower rate streams that are transmitted simultaneously over a number of subcarriers. Unlike conventional single-carrier modulation schemes that send only one signal at a time using one radio frequency, OFDM sends multiple high-speed signals concurrently on specially computed, orthogonal carrier frequencies. The result is much more efficient use of bandwidth as well as robust communications during noise and other interferences.
In OFDM design, a number of parameters are up for consideration, such as the number of subcarriers, guard time, symbol duration, subcarrier spacing, modulation type per subcarrier. The choice of parameters is influenced by system requirements such as available bandwidth, required bit rate, tolerable delay spread, and Doppler values.

3.3 GUARD INTERVAL

Guard intervals are used to ensure that distinct transmissions do not interfere with one another. These transmissions may belong to different users (as in TDMA) or to the same user (as in OFDM).

The purpose of the guard interval is to introduce immunity to propagation delays, echoes and reflections, to which digital data is normally very sensitive.

In FDM systems, the beginning of each symbol is preceded by a guard interval. As long as the echoes fall within this interval, they will not affect the receiver's ability to safely decode the actual data, as data is only interpreted outside the guard interval. The many carriers are spaced apart in such way that the signals can be received using conventional filters and demodulators. In such receivers, guard bands have to be introduced between the different carriers and the introduction of these guard bands in the frequency domain results in a lowering of the spectrum efficiency.

3.4 ORTHOGONALITY

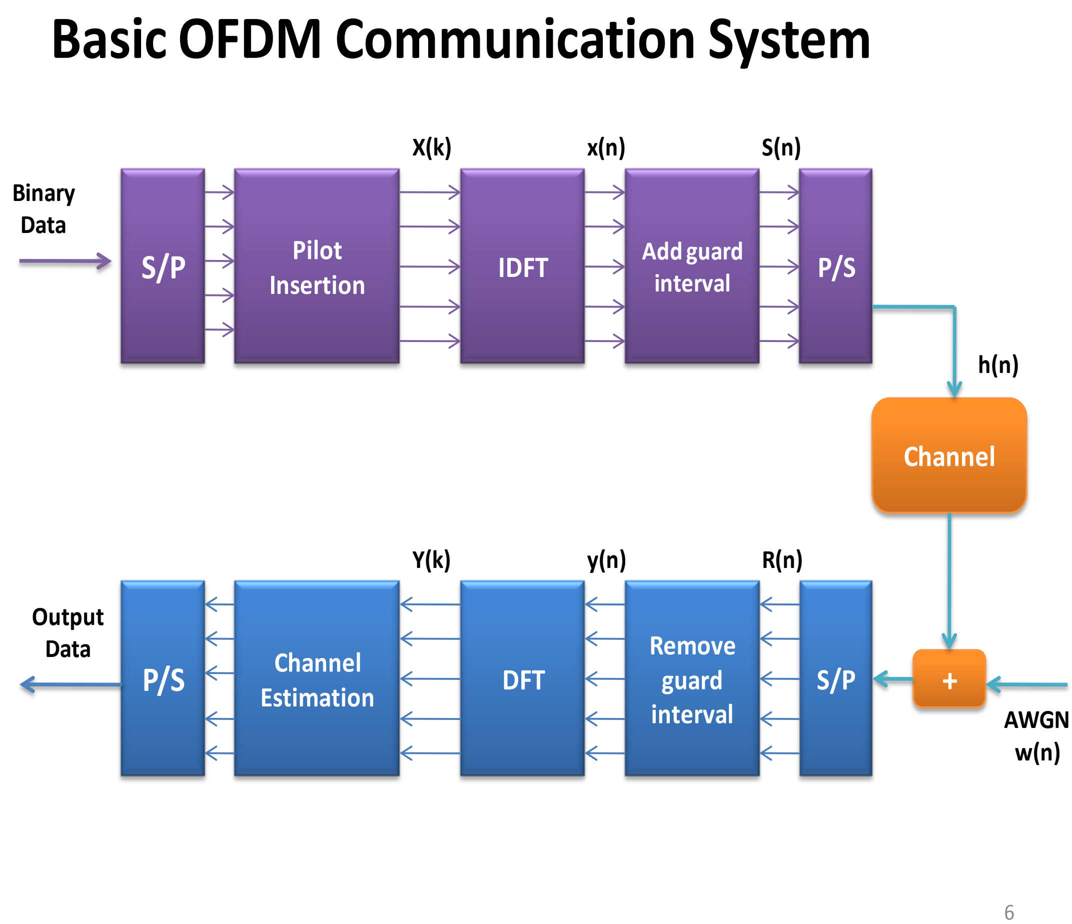
Two signals are orthogonal, if

$$\int_a^b \Psi_p(t)\Psi_q^*(t)dt = \begin{cases} K & for\ p = q \\ 0 & for\ p \neq q \end{cases}$$

where the * indicates the complex conjugate.

In place of guard interval, we can arrange the carriers in an OFDM signal so that the sidebands of the individual carriers overlap and the signals can still be received without adjacent carrier interference.

This can be done if the carriers are orthogonal to each other.



(Fig 3.1)

3.5 MATHEMATICAL TREATMENT

An OFDM signal consists of a sum of subcarriers that are modulated by using phase shift keying (PSK) or qudrature amplitude modulation (QAM). Mathematically, each carrier can be described as a complex wave:

$$s_c(t) = A_c(t)e^{j\left[\omega_c t+\phi_c(t)\right]}$$

The real signal is the real part of sc(t). Both Ac(t) and fc(t), the amplitude and phase of the carrier, can vary on a symbol by symbol basis. The values of the parameters are constant over the symbol duration period .OFDM consists of many carriers. Thus the complex signals s(t) is represented by:

$$s_s(t) = \frac{1}{N}\sum_{n=0}^{N-1} A_N(t)\, e^{j\left[\omega_n t + \phi_n(t)\right]}$$

Where

$$\omega_n = \omega_0 + n\Delta\omega$$

If we consider the waveforms of each
component of the signal over one symbol period, then the variables Ac(t) and fc(t) take on fixed values, which depend on the frequency of that particular carrier,so

$$\phi_n(t) \Rightarrow \phi_n$$
$$A_n(t) \Rightarrow A_n$$

If the signal is sampled using a sampling frequency of 1/T, then the resulting signal is represented by:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j\left[(\omega_0 + n\Delta\omega)kT + \phi_n\right]}$$

At this point, we have restricted the time over which we analyse the signal to N samples. It is convenient to sample over the period of one data symbol. Thus we have a relationship:

$$\tau = NT$$

If we now simplify eqn. 4.4, without a loss of generality by letting w0=0, then the signal becomes:

$$s_s(kT) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j\phi_n} e^{j(n\Delta\omega)kT}$$

Now the equation can be compared with the general form of the inverse Fourier transform:

$$g(kT) = \frac{1}{N}\sum_{n=0}^{N-1} G\left(\frac{n}{NT}\right) e^{j2\pi nk/N}$$

Now,S_2 equals G if:

$$\Delta f = \frac{\Delta\omega}{2\pi} = \frac{1}{NT} = \frac{1}{\tau}$$

This is the same condition that was required for orthogonality. Thus, one consequence of maintaining orthogonality is that the OFDM signal can be defined by using Fourier transform procedures.

3.6 FOURIER TRANSFORM

Fourier transform (often abbreviated **FT**) is an operation that transforms one complex-valued function of a real variable into another. In such applications as signal processing, the domain of the original function is typically time and is accordingly called the time domain. That of the new function is frequency, and so the Fourier transform is often called the frequency domain representation of the original function. It describes which frequencies are present in the original function. This is analogous to describing a chord of music in terms of the notes being played. In effect, the Fourier transform decomposes a function into oscillatory functions. The term Fourier transform refers both to the frequency domain representation of a function, and to the process or formula that "transforms" one function into the other.

USE OF FAST FOURIER TRANSFORM (FFT)

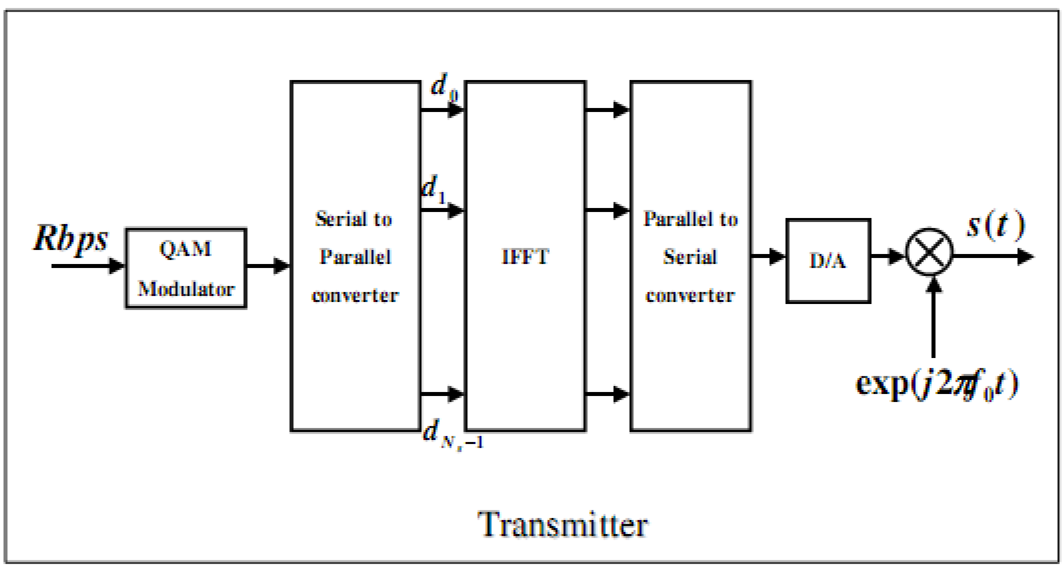
At the transmitter, the signal is defined in the frequency domain. It is a sampled digital signal, and it is defined such that the discrete Fourier spectrum exists only at discrete frequencies. Each OFDM carrier corresponds to one element of this discrete Fourier spectrum. The amplitudes and phases of the carriers depend on the data to be transmitted. The data transitions are synchronised at the carriers, and can be processed together, symbol by symbol.

Generation of subcarriers using the IFFT

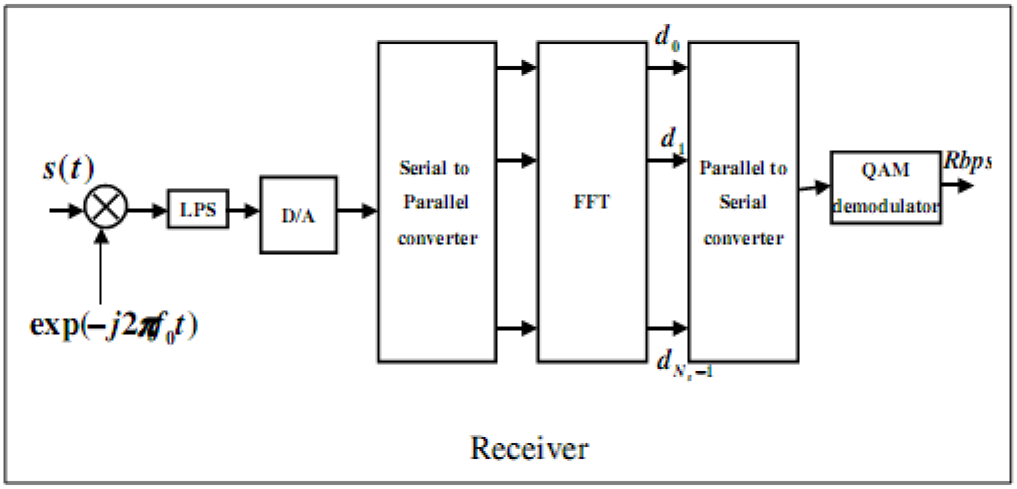
The complex baseband OFDM signal is in fact nothing more than the inverse Fourier transform of QAM input symbol. The time discrete equivalent is the inverse discrete Fourier transform (IDFT), which is given by :

$$s(n) = \sum_{i=0}^{N_s-1} d_i \exp(j2\pi \frac{in}{N})$$

Where the time t is replaced by a sample number n. In practice, this transform can be implemented very efficiently by the inverse Fast Fourier transform (IFFT) as shown in figures.



(Fig 3.2)Transmitter of OFDM



(Fig 3.3)Receiver of OFDM

3.7 CHOICE OF VARIOUS OFDM PARAMETERS

The choice of various OFDM parameters is a tradeoff between various, often conflicting requirements. Usually, there are three main requirements to start with: bandwidth, bit rate, and delay spread. The delay spread directly dictates the guard time. As a rule, the guard time should be about two to four times the root-mean-squared delay spread. This value depends on the type of coding and QAM modulation. Higher order QAM (like 64-QAM) is more sensitive to ICI and ISI than QPSK; while heavier coding obviously reduces the sensitivity to such interference.

Now the guard time has been set, the symbol duration can be fixed. To minimize the signal-to-noise ratio (SNR) loss caused by guard time, it is desirable to have the symbol duration much larger than the guard time. It cannot be arbitrarily large, however, because a larger symbol duration means more subcarriers with a smaller subcarrier spacing, a larger implementation complexity, and more sensitivity to phase noise and frequency offset, as well as an increased peak-to-average power ratio. Hence, a practical design choice to make the symbol duration at least five times the guard time, which implies a 1dB SNR loss because the guard time.

After the symbol duration and guard time are fixed, the number of subcarriers follows directly as the required –3 dB bandwidth divided by the subcarrier spacing, which is the inverse of the symbol duration less the guard time. Alternatively, the number of subcarriers may be determined by the required bit rate divided by the bit rate per subcarrier. The bit rate per subcarrier is defined by the modulation type, coding rate, and symbol rate.

An additional requirement that can affect the chosen parameters is the demand for an integer number of samples both within the FFT/IFFT interval and in the symbol interval.

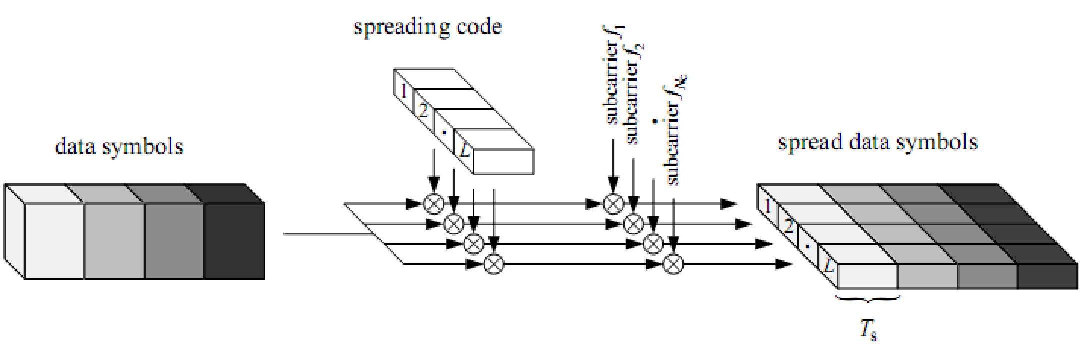
CHAPTER-4
Multi-Carrier Code Division Multiple Access

4.1 INTRODUCTION

Code division multiple access (CDMA) is a multiple access technique where different users share the same physical medium, that is, the same frequency band, at the same time. The main ingredient of CDMA is the *spread spectrum* technique, which uses high rate signature pulses to enhance the signal bandwidth far beyond what is necessary for a given data rate. In a CDMA system, the different users can be identified and, hopefully, separated at the receiver by means of their characteristic individual *signature pulses* (sometimes called the *signature waveforms*), that is, by their individual *codes*. Nowadays, the most prominent applications of CDMA are mobile communication systems like CDMA One (IS-95), UMTS or CDMA 2000. To apply CDMA in a mobile radio environment, specific additional methods are required to be implemented in all these systems. Methods such as power control and soft handover have to be applied to control the interference by other users and to be able to separate the users by their respective codes.

4.2 THE CONCEPT OF SPREADING

Spread spectrum means enhancing the signal bandwidth far beyond what is necessary for a given data rate and thereby reducing the power spectral density (PSD) of the useful signal so that it may even sink below the noise level. One can imagine that this is a desirable property for military communications because it helps to hide the signal and it makes the signal more robust against intended interference (*jamming*). Spreading is achieved – loosely speaking – by a multiplication of the data symbols by a *spreading sequence* of pseudo random signs. These sequences are called *pseudo noise* (PN) sequences or code signals.



(Fig 4.1) concept of spreading

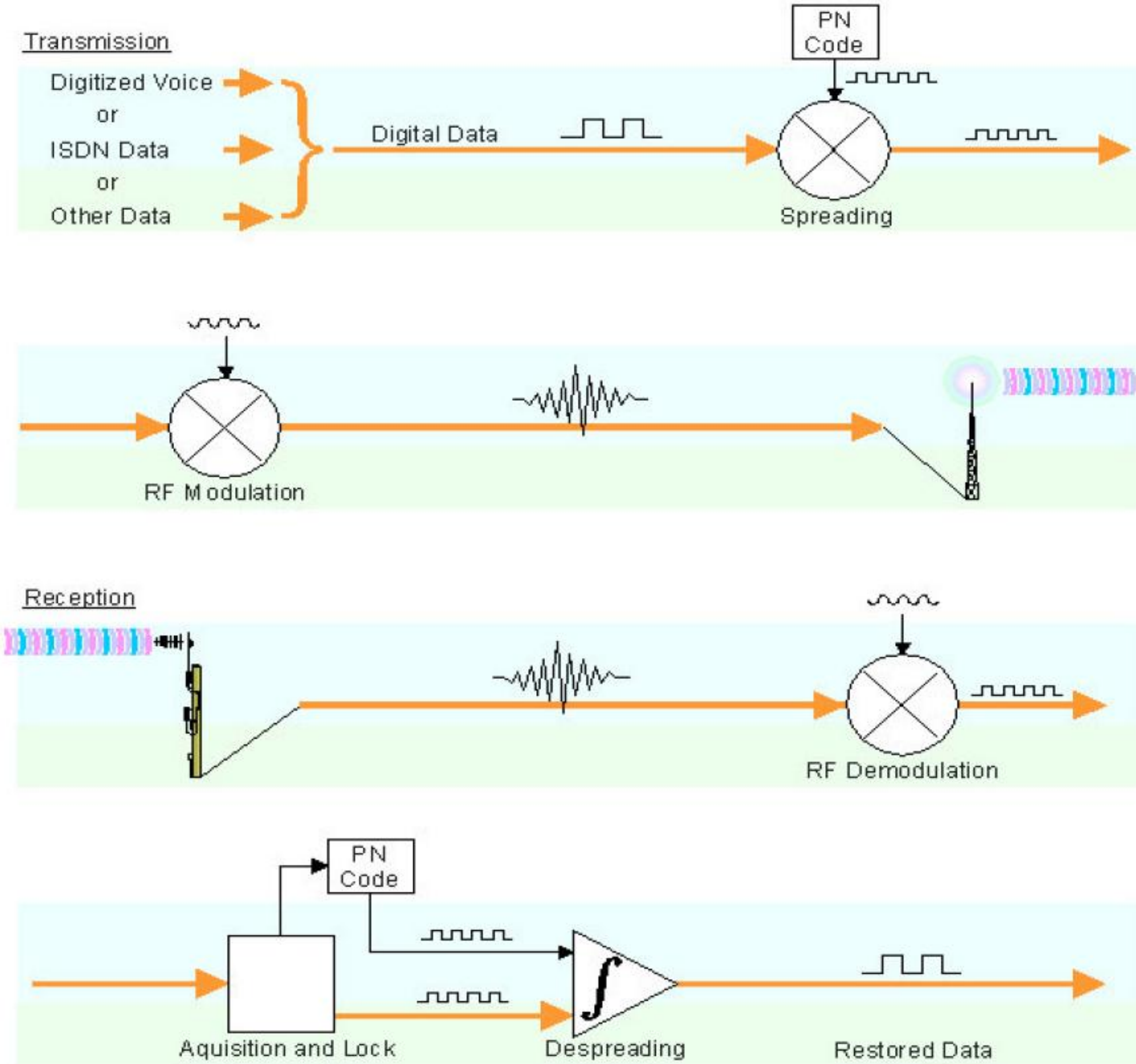
4.3 P-N SEQUENCES

In CDMA networks there are a number of channels each of which supports a very large number of users. For each channel the base station generates a unique code that changes for every user. The base station adds together all the coded transmissions for every subscriber. The subscriber unit correctly generates its own matching code and uses it to extract the appropriate signals.
In order for all this to occur, the pseudo-random code must have the following properties:

1. It must be deterministic. The subscriber station must be able to independently generate the code that matches the base station code.
2. It must appear random to a listener without prior knowledge of the code (i.e. it has the statistical properties of sampled white noise).
3. The cross-correlation between any two codes must be small
4. The code must have a long period (i.e. a long time before the code repeats itself).

Generation of PN sequence:

The p-n sequence is usually generated using a shift register with feedback-taps. Binary sequences are shifted through the shift registers in response to clock pulses and the output of the various stages are logically combined and fed back as input to the stage. When the feedback logic consists of exclusive –OR gates, the shift register is called a linear PN sequence generator.
The initial contents of the memory stages and the feedback logic circuit determine the successive contents of the memory. If a linear shift register reaches zero state at some time, it would always remain in the zero state and the output would subsequently be all zeros. Since there are exactly 2m-1 nonzero states for an m-stage feedback shift register, the period of a PN sequence produced by a linear m-stage shift register cannot exceed 2m-1 symbols which is called maximal length sequence.



(Fig 4.2) Signal transmission and reception of CDMA

4.4 WALSH CODES

The PN Sequence codes have a disadvantage that they are not perfectly orthogonal to each other. Hence, non-zero from undesired user may affect the performance of the receiver. So Walsh function can be used in place of PN sequence codes.

Walsh codes are otherwise known as Walsh Hadamard codes. These are obtained by selecting as code words the rows of a Hadamard matrix . A Hadamard matrix is a N*N matrix of 1's and -1's, such that each row differs from any other row in exactly N/2 locations. One row contains all -1's with the remainder containing N/2 1's and -1's.

A 8 by 8 walsh code is given by

W(0,8) = 1, 1, 1, 1, 1, 1, 1, 1

W(1,8) = 1,-1, 1,-1, 1,-1, 1,-1

W(2,8) = 1, 1,-1,-1, 1, 1,-1,-1

W(3,8) = 1,-1,-1, 1, 1,-1,-1, 1

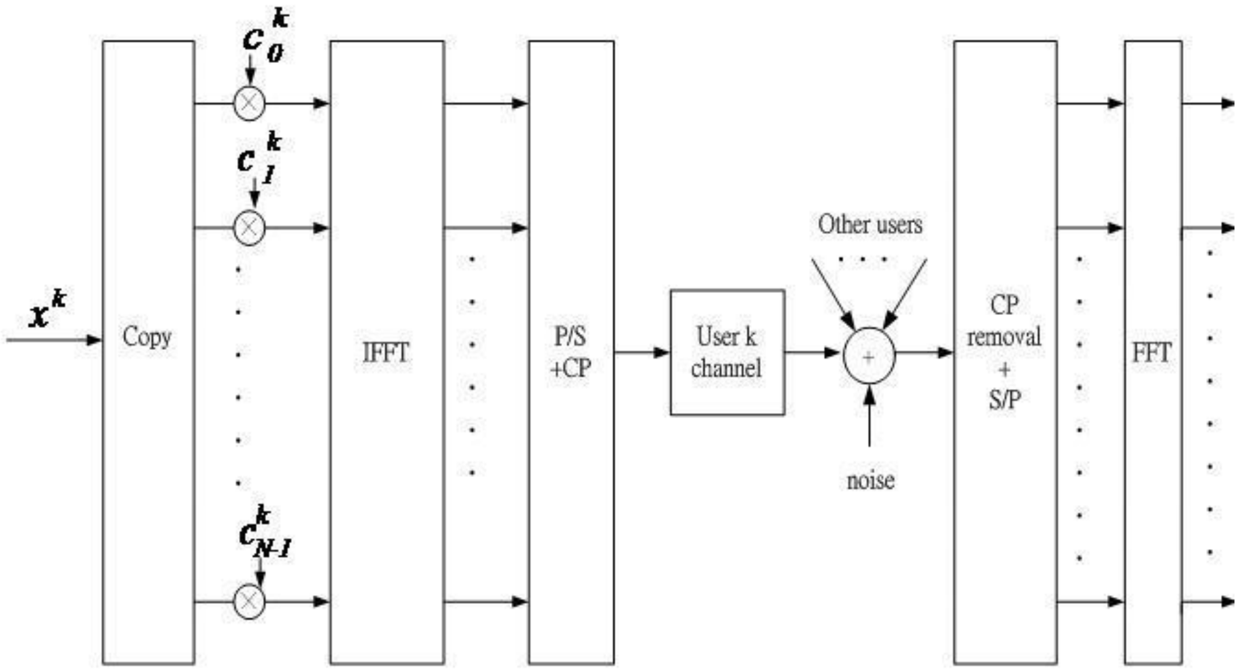
W(4,8) = 1, 1, 1, 1,-1,-1,-1,-1

W(5,8) = 1,-1, 1,-1,-1, 1,-1, 1

W(6,8) = 1, 1,-1,-1,-1,-1, 1, 1

W(7,8) = 1,-1,-1, 1,-1, 1, 1,-1

Walsh codes are mathematically orthogonal codes. As such, if two Walsh codes are correlated, the result is intelligible only if these two codes are the same. As a result, a Walsh-encoded signal appears as random noise to a CDMA capable mobile terminal, unless that terminal uses the same code as the one used to encode the incoming signal.



(Fig 4.3) block diagram of MC-CDMA

4.5 MULTI-CARRIER CODE DIVISION MULTIPLE ACCESS (MC-CDMA)

Multi-Carrier Code Division Multiple Access (**MC-CDMA**) is a multiple access scheme used in OFDM-based telecommunication systems, allowing the system to support multiple users at the same time.

MC-CDMA spreads each user symbol in the frequency domain. That is, each user symbol is carried over multiple parallel subcarriers, but it is phase shifted (typically 0 or 180 degrees) according to a code value. The code values differ per subcarrier and per user. The receiver combines all subcarrier signals, by weighing these to compensate varying signal strengths and undo the code shift. The receiver can separate signals of different users, because these have different (e.g. orthogonal) code values.

CHAPTER-5
Channel Estimation

5.1 INTRODUCTION

A wideband radio channel is normally frequency selective and time variant.For an OFDM mobile communication system, the channel transfer function at different subcarriers appears unequal in both frequency and time domains. Therefore, a dynamic estimation of the channel is necessary. Pilot-based approaches are widely used to estimate the channel properties and correct the received signal. In this chapter we have investigated two types of pilot arrangements: Block type and Comb type.

5.2 SYSTEM ENVIRONMENT

Wireless

The system environment we will be considering in this thesis will be wireless indoor and urban areas, where the path between transmitter and receiver is blocked by various objects and obstacles. For example, an indoor environment has walls and furniture, while the outdoor environment contains buildings and trees. This can be characterized by the impulse response in a wireless environment.

Multipath Fading

Most indoor and urban areas do not have direct line of sight propagation between the transmitter and receiver. Multi-path occurs as a result of reflections and diffractions by objects of the transmitted signal in a wireless environment. These objects can be such things as buildings and trees. The reflected signals arrive with random phase offsets as each reflection follows a different path to the receiver. The signal power of the waves also decreases as the distance increases. The result is random signal fading as these reflections destructively and constructively superimpose on each other. The degree of fading will depend on the delay spread (or phase offset) and their relative signal power.

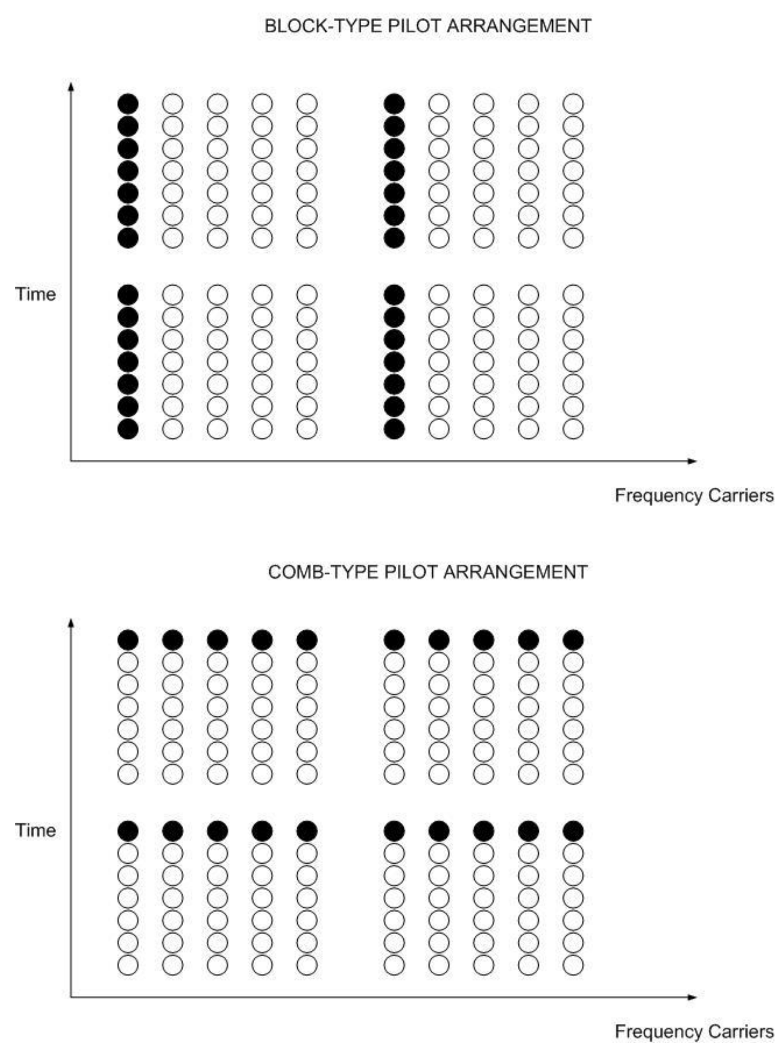
Fading Effects due to Multi-path Fading

Time dispersion due to multi-path leads to either flat fading or frequency selective fading:
Flat fading occurs when the delay is less than the symbol period and affects all frequencies equally. This type of fading changes the gain of the signal but not the spectrum. This is known as amplitude varying channels or narrowband channels, since the bandwidth of the applied

signal is narrow compared to the channel bandwidth.Frequency selective fading occurs when the delay is larger than the symbol period. In the frequency domain, certain frequencies will have greater gain than others frequencies.

White noise

In wireless environments, random changes in the physical environment resulting in thermal noise and unwanted interference from many other sources can cause the signal to be corrupted. Since it is not possible to take into account all of these sources, we assume that they produce a single random signal with uniform distributions across all frequencies. This is known as white noise.



(Fig 5.1)Types of Pilot arrangements

5.3 CHANNEL ESTIMATION BASED ON BLOCK-TYPE PILOT ARRANGEMENT

In block-type pilot based channel estimation, OFDM channel estimation symbols are transmitted periodically, in which all sub-carriers are used as pilots. If the channel is constant during the block, there will be no channel estimation error since the pilots are sent at all carriers. The estimation can be performed by using either LSE or MMSE .If inter symbol interference is eliminated by the guard interval, we write in matrix notation:

Y=XFh+ W

= XH +W

where

$$X = diag\ \{X(0), X(1), \ldots\ X(N-1)\}$$

$$Y = [Y(0), Y(1), \ldots\ Y(N-1)]^T$$

$$W = [W(0), W(1), \ldots\ W(N-1)]^T$$

$$H = [H(0), H(1), \ldots\ H(N-1)]^T = DFT_N\{h\}$$

$$F = \begin{bmatrix} W_N^{00} & \cdots & W_N^{0(N-1)} \\ \vdots & \ddots & \vdots \\ W_N^{(N-1)0} & \cdots & W_N^{(N-1)(N-1)} \end{bmatrix}$$

$$W_N^{nk} = \frac{1}{N} e^{-j2\pi(n/N)k}$$

5.4 CHANNEL ESTIMATION BASED ON COMB-TYPE PILOT ARRANGEMENT

In comb-type based channel estimation, the Np pilot signals are uniformly inserted into X(k) according to following equation:

$$X(k) = X(mL + l)$$
$$= \{X_p(k), \qquad l = 0$$
$$= \{\inf .data \qquad l = 1,\ldots\ldots.L-1$$

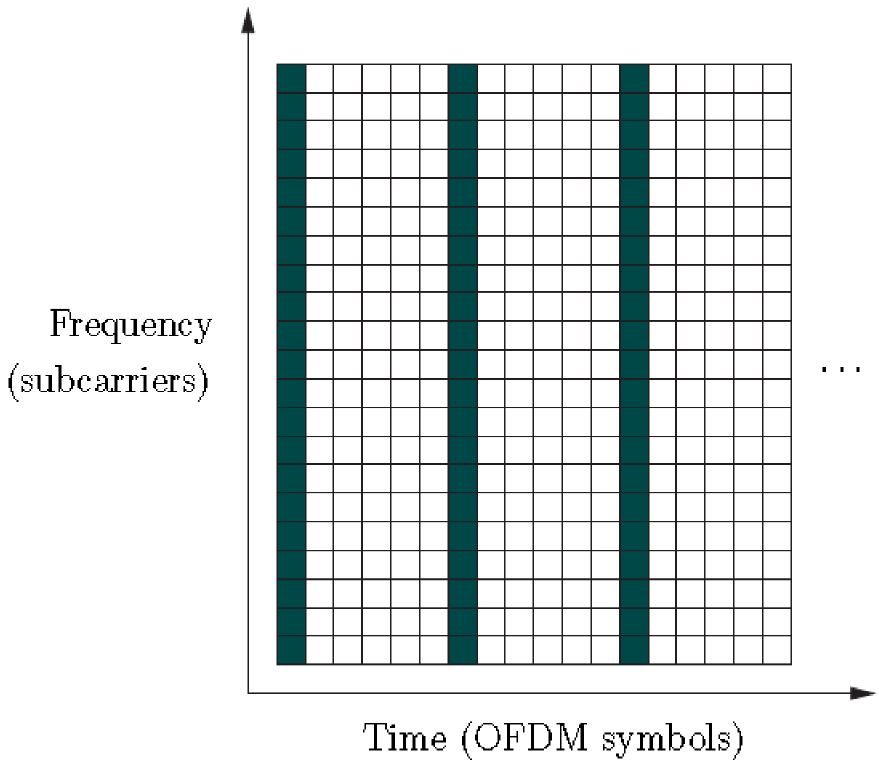
L = number of carriers*/Np*

xp(*m*) is the *mth* pilot carrier value.

5.5 PILOT BASED CHANNEL ESTIMATION

The following estimators use on pilot data that is known to both transmitter and receiver as a reference in order to track the fading channel. The estimators use block based pilot symbols, meaning that pilot symbols are sent across all sub-carriers periodically during channel estimation. This estimate is then valid for one OFDM/MC-CDMA frame before a new channel estimate will be required.

Since the channel is assumed to be slow fading, our system will assume a frame format, transmitting one channel estimation pilot symbol, followed by five data symbols, as indicated in the time frequency lattice shown in adjoining figure . Thus each channel estimate will be used for the following five data symbols.



(Fig 5.2) Block type pilot estimation

5.6 LEAST SQUARES ESTIMATOR

The simplest channel estimator is to divide the received signal by the input signals, which should be known pilot symbols. This is known as the Least Squares (LS) Estimator and can simply be expressed as:

$$HLS = y/x$$

This is the most naive channel estimator as it works best when no noise is present in the channel. When there is no noise the channel can be estimated perfectly. This estimator is equivalent to a zero-forcing estimatorThe main advantage is its simplicity and low complexity. It only requires a single division per sub-carrier. The main disadvantage is that it has high mean-square error. This is due to its use of an oversimplified channel and does not make use of the frequency and time correlation of the slow fading channel.

An improvement to the LS estimator would involve making use of the channel statistics. We could modify the LS estimator by tracking the average of the most recently estimated channel vectors.

We have to minimize

$$J = (Y - XH)^{H}(Y - XH)$$

$$= (Y^{T} - H^{H}X^{H})(Y - XH)$$

$$= Y^{H}Y - Y^{H}XH - H^{H}X^{H}Y + H^{H}X^{H}XH$$

For minimization of J we have to differentiate J with respect to H

$$\left.\frac{\partial J}{\partial H}\right|_{\hat{H}} = 0$$

$$\Rightarrow \hat{H} = X^{-1}Y$$

5.7 LINEAR MINIMUM MEAN SQUARE ERROR ESTIMATOR

The Linear Minimum Mean Squares Error (LMMSE) Estimator minimizes the mean square error (MSE) between the actual and estimated channel by using the frequency correlation of the slow fading channel. This is achieved through a optimizing linear transformation applied to the LS estimator described in the previous section. From adaptive filter theory, the optimum solution in terms of the MSE is given by the Wiener-Hopf equation:

$$h = R_{hh_{ls}} R^{-1}_{h_{ls}h_{ls}} h_{ls}$$

where X is a matrix conatining the transmitted signalling points on its diagonal,

σ^2 is the additive noise variance. The matrix R*hhls* is the cross correlation between channel attenuation vector h and the LS estimate h*ls* and R*hlshls* is the auto correlation matrix of the LS estimate h*ls*, given by

$$R_{hh_{ls}} = E\{hh_{ls}^H\}$$
$$R_{h_{ls}h_{ls}} = E\{h_{ls}h_{ls}^H\}$$

Since white noise is uncorrelated with the channel attenuation, the cross correlation between the channel h and noisy channel h*ls* is the same as the autocorrelation of the channel *h*. Thus we can replace R*hhls* with R*hh*. Also the autocorrelation of R*hlshls* is equavalent to R*hh* plus the noise power *(*σ^2*)*

and signal power. So the estimator can be expressed as:

$$h_{lmmse} = R_{hh}\left(R_{hh} + \sigma_n^2(\mathbf{XX}^H)^{-1}\right)^{-1} h_{ls}$$

5.8 SYSTEM CONSIDERATIONS

RAYLEIGH FADING

The Rayleigh distribution is commonly used to describe the statistical time varying nature of the received envelope of a flat fading signal, or the envelope of an individual multipath component. The envelope of the sum of two quadrature Gaussian noise signals obeys a Rayleigh distribution.

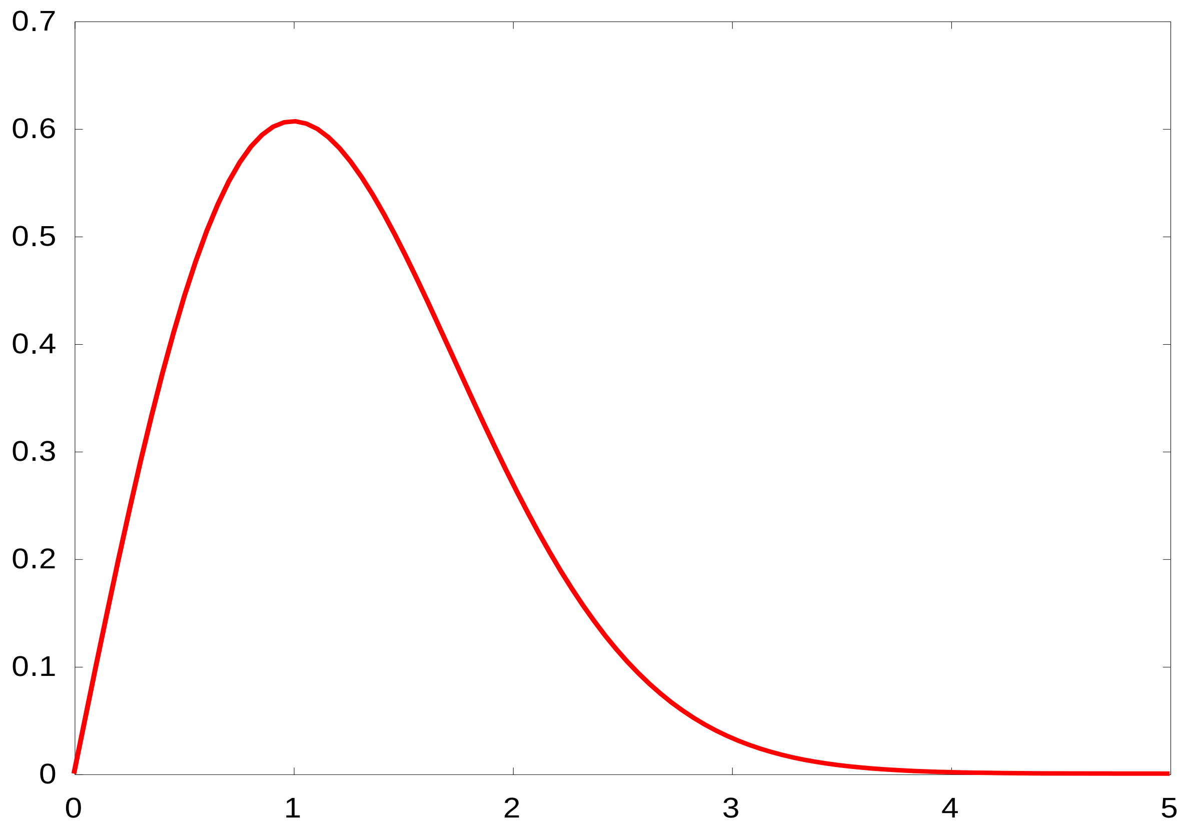
$$p(r) = \begin{cases} \frac{r}{\sigma^2}\exp(-\frac{r^2}{2\sigma^2}) & 0 \le r \le \infty \\ 0 & r < 0 \end{cases}$$

σ is the rms value of the received voltage before envelope detection, and σ^2 is the time-average power of the received signal before envelope detection.

It describes the probability of the signal level being received due to fading. Table shows the probability of the signal level for the Rayleigh distribution.

Signal Level (dB about median)	% Probability of Signal Level being less than the value given
10	99
0	50
-10	5
-20	0.5
-30	0.05

(Fig 5.3)table showing signal level for the Rayleigh distribution.



(Fig 5.4)Rayleigh distribution

mean = 1.2533σ median = 1.177σ variance = $0.4292\sigma^2$

Doppler Shift

When a wave source and a receiver are moving relative to one another the frequency of the received signal will not be the same as the source. When they are moving toward each other the frequency of the received signal is higher than the source, and when they are moving away from each other the frequency decreases. This is called the Doppler's effect. An example of this is the change of pitch in a car's horn as it approaches then passes by. This effect becomes important when developing mobile radio systems.

The amount the frequency changes due to the Doppler effect depends on the relative motion between the source and receiver and on the speed of propagation of the wave. The Doppler shift in frequency can be written as:

$$\Delta f \approx \pm f_0 \frac{v}{c}$$

where Δf is the change in frequency of the source seen at the receiver,

f_0 is the frequency of the source,

v is the speed difference between the source and transmitter,

and c is the speed of light

Assumptions on channel

To simplify our simulated channel, the following assumptions were made:

- The impulse response is shorter than the Cyclic Prefix. Therefore, there is no ISI and ICI and the channel is therefore diagonal.
- The channel is a synchronised, sample spaced channel.
- Channel noise is additive, white and complex Gaussian.
- The fading on the channel is slow enough to be considered constant during one OFDM frame

CHAPTER-6

SIMULATIONS AND RESULTS

6.1 Simulation Results for single bit without noise OFDM :

Transmission data bit=1

Received data bit after OFDM simulation=1

So, no error. Simulation was successful.

6.2 Simulation Results for multiple bit without noise OFDM (10bits to 10^5 bits): (model simulation)

Transmission data=[1011001010]

Received data after OFDM simulation=[1011001010]

Error bit count=0

Simulation was successful.

6.3 Simulation Results for multiple bit with noise(AWGN) OFDM (10bits to 10^5 bits): (model simulation)

SNR=12

Transmission data=[1011001010]

Received data after OFDM simulation=[1101011010]

Error bit count=3

As noise is introduced, error in transmission of few of the bits. Simulation was successful.

6.4 Simulation Results for multiuser without noise MC-CDMA (10bits to 10^4 bits): (model simulation)

Transmission data=

```
1001   111   110
1010   001   111
1101   100   100

1000   111   110
1010   001   111
1101   100   100

1001   001   111
1101   001   010
```

Received data after MC-CDMA simulation=

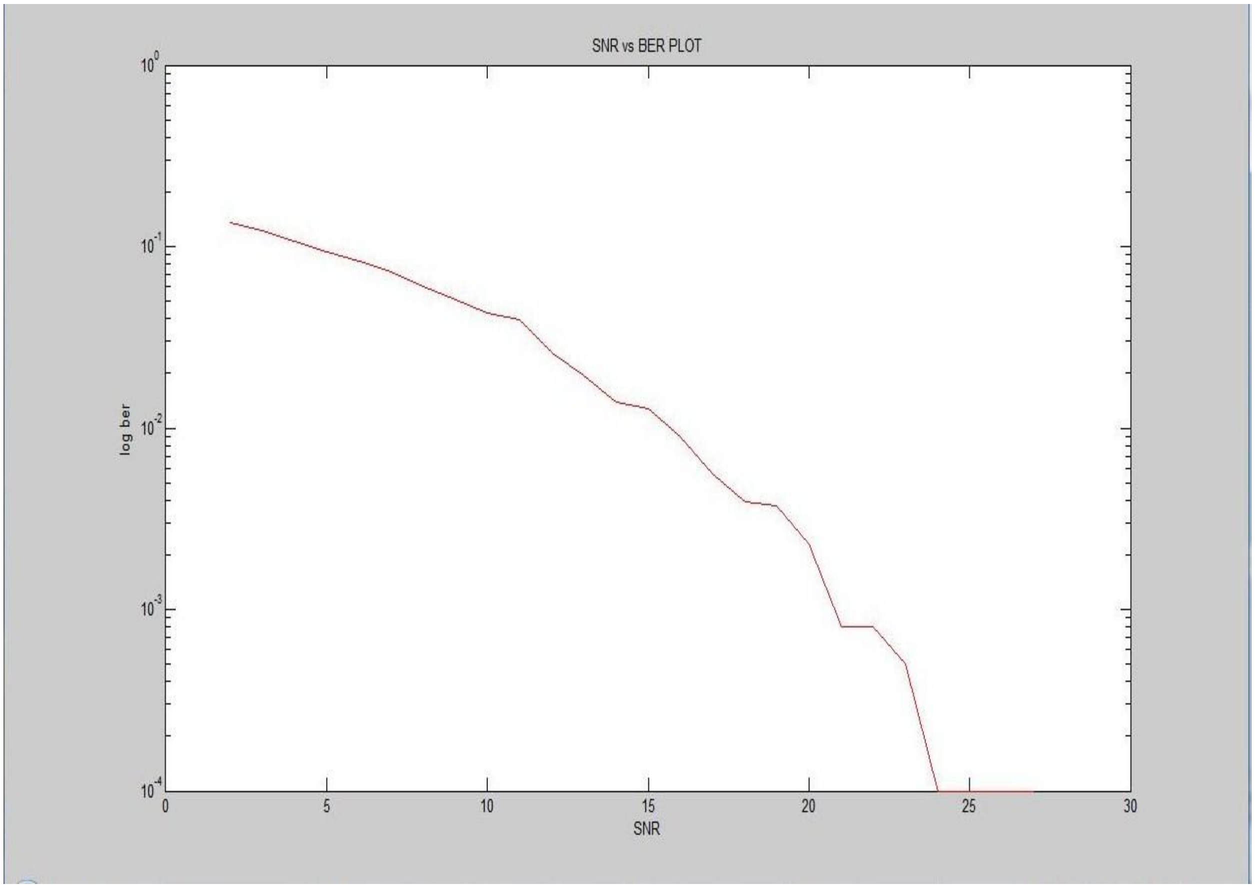
```
1001  111  110
1010  001  111
1101  100  100

1000  111  110
1010  001  111
1101  100  100

1001  001  111
1101  001  010
```

No. of users=8

Error in all 8 users =0

6.5 Simulation Results for single user with noise MC-CDMA (10bits to 10^4 bits):

BER Vs SNR plot (fig 6.1)

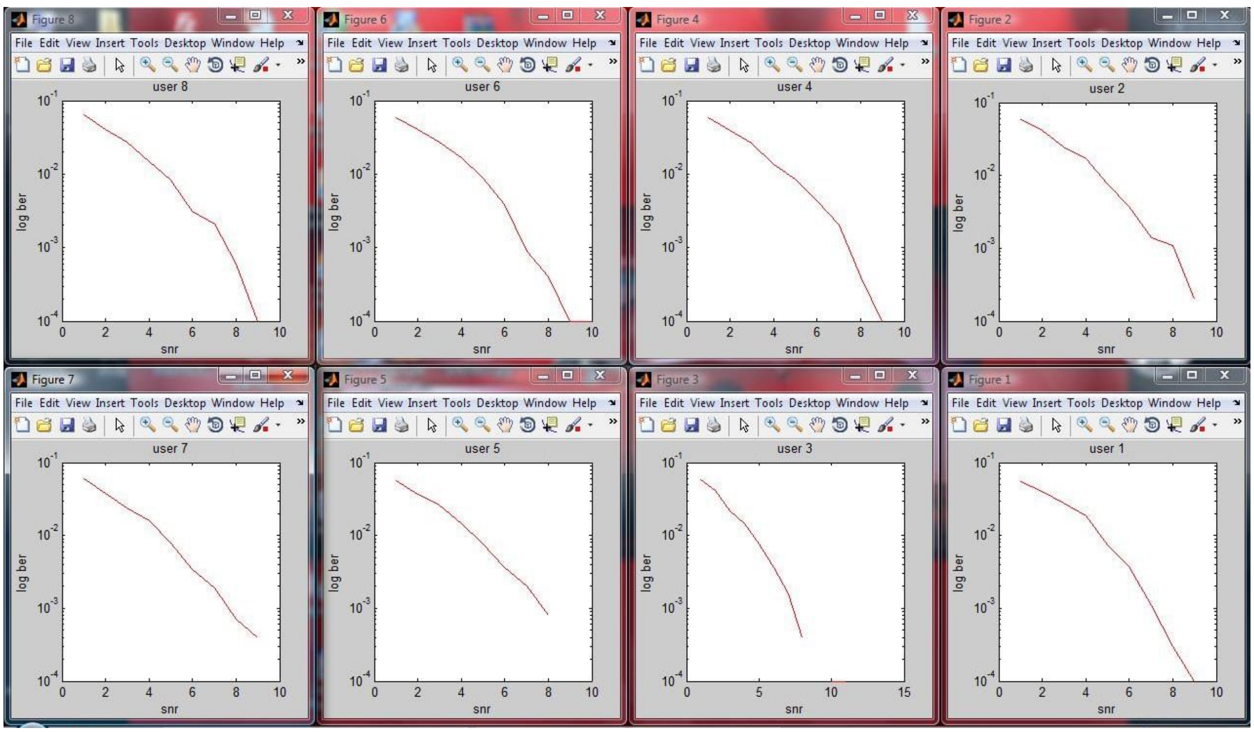


6.6 Simulation Results for multiuser with noise MC-CDMA (10bits to 10^4 bits): (model simulation)

For 10 000 bits

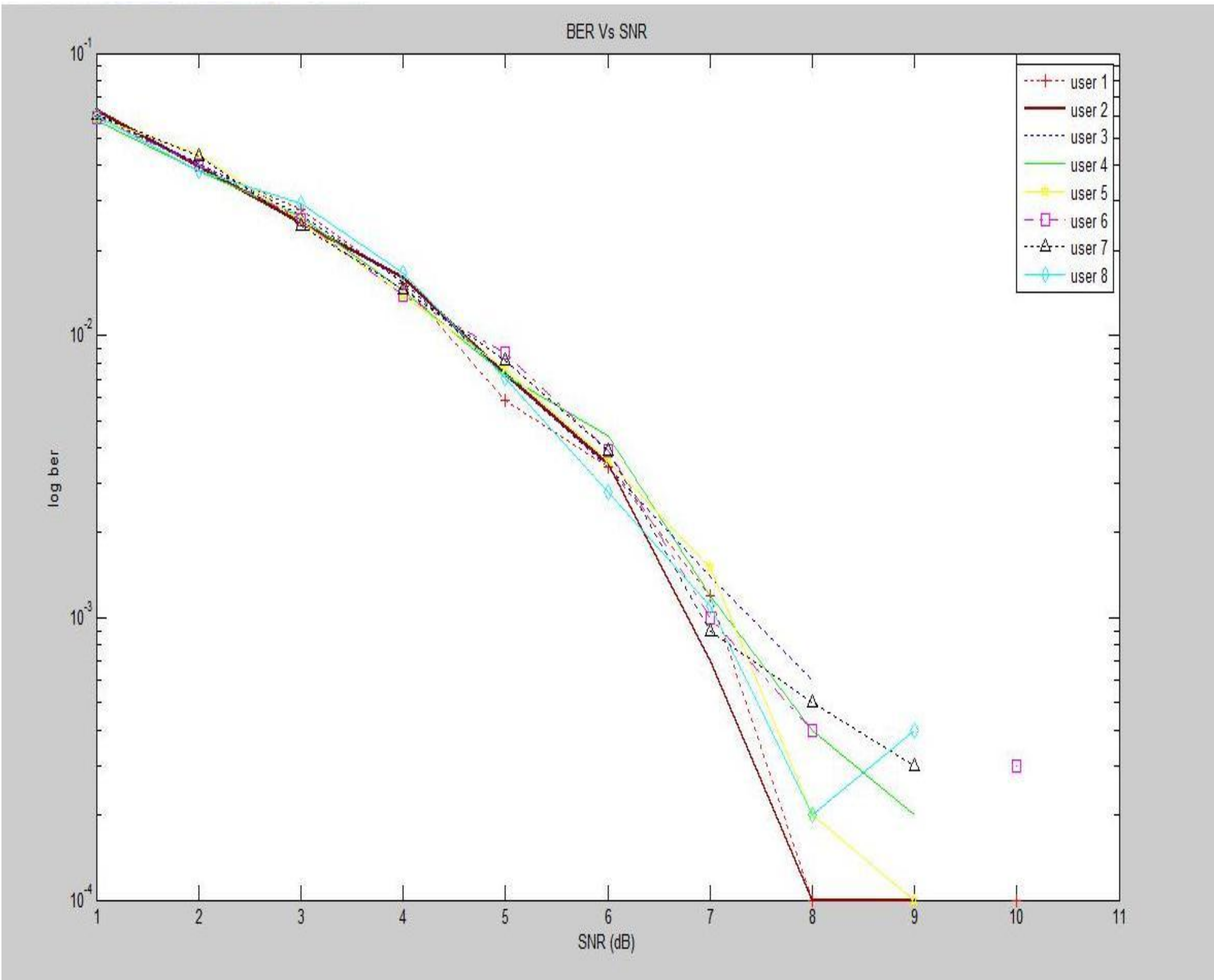
No. of users=8

User wise BER Vs SNR plot (fig 6.2)



Comparison of all the 8 users:

BER Vs SNR plot (fig 6.3)

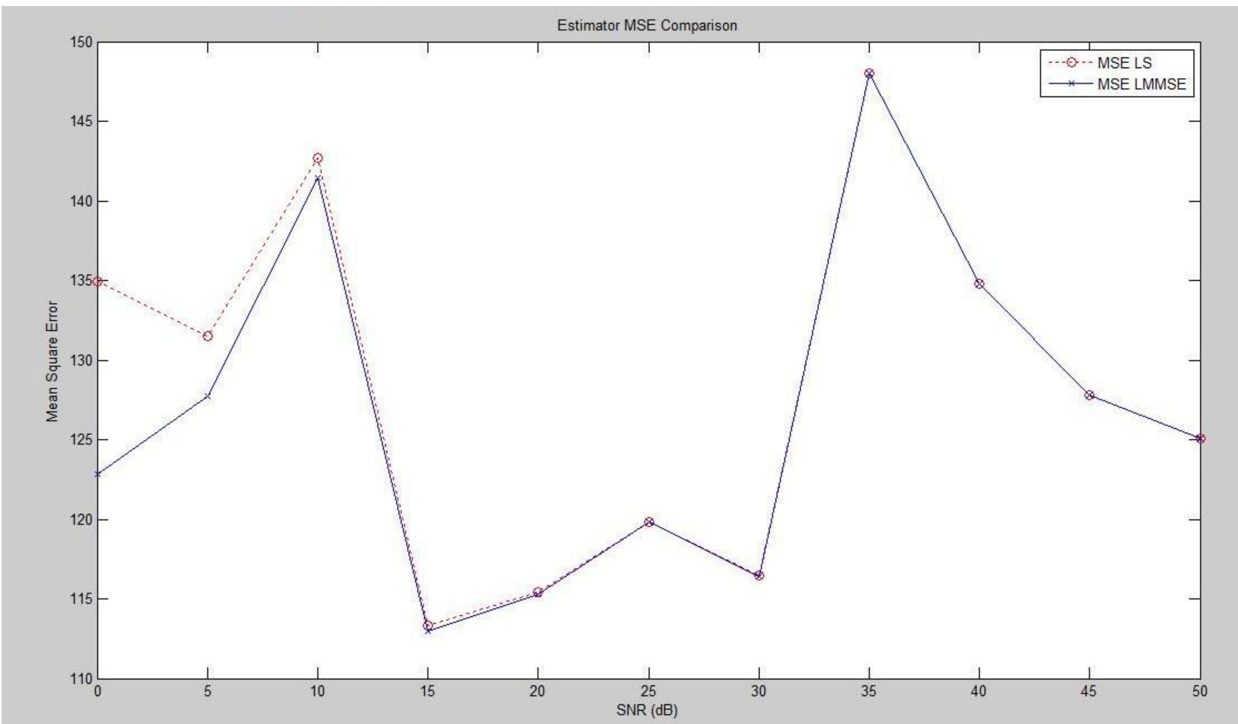


6.7 Simulation Results for channel estimation in MC-CDMA (10bits to 10^3 bits): (model simulation)

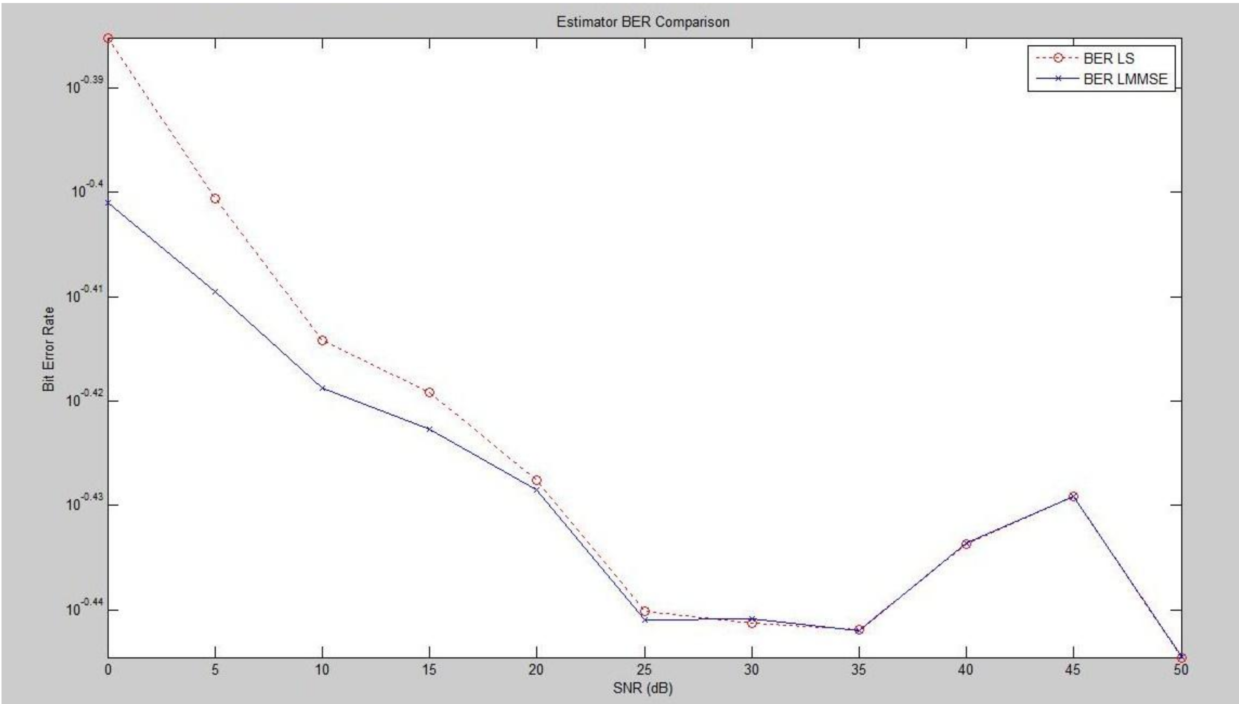
For 300 bits

Comparison between **LS estimator** and **LMMSE estimator**

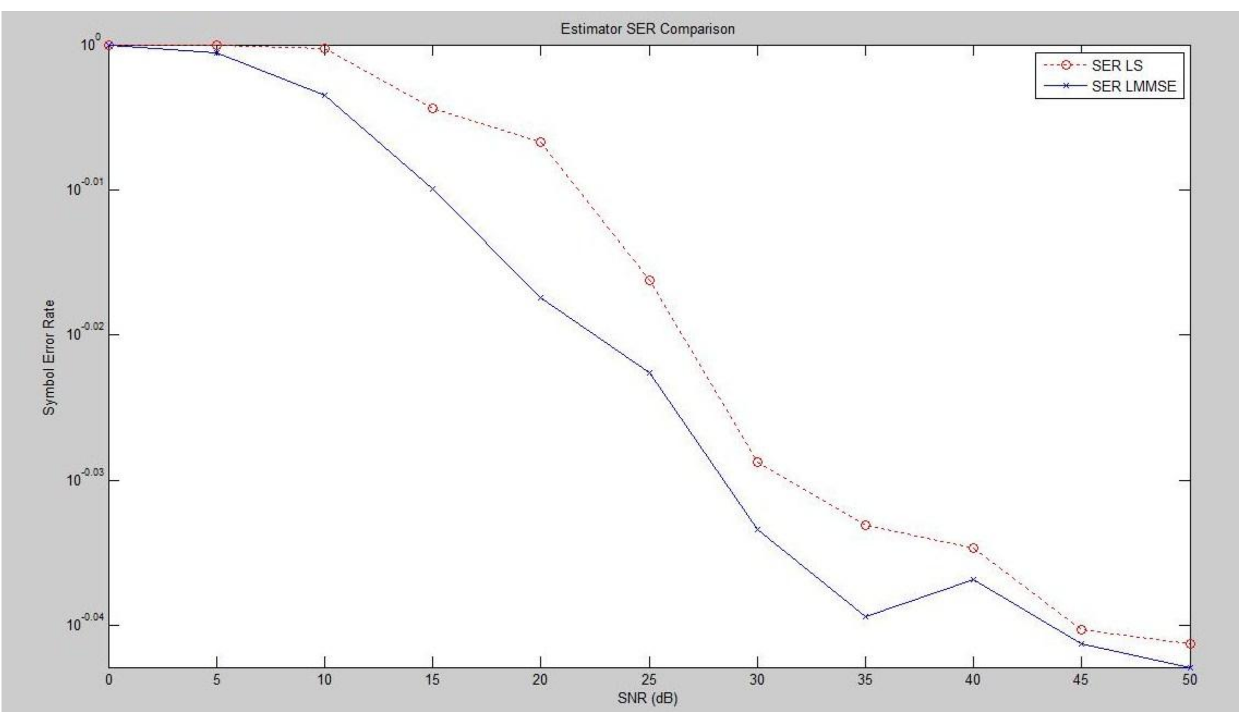
MSE comparison plot (fig 6.4)



BER comparison plot (fig 6.5)



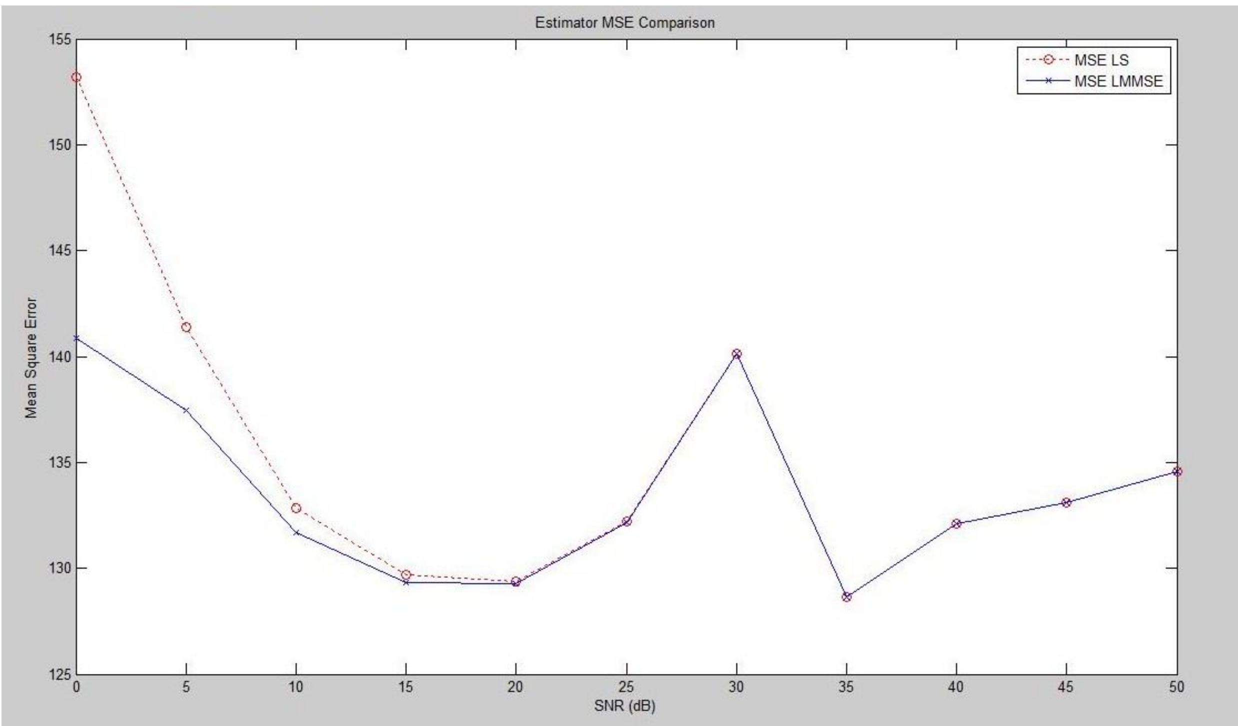
SER comparison plot (fig 6.6)



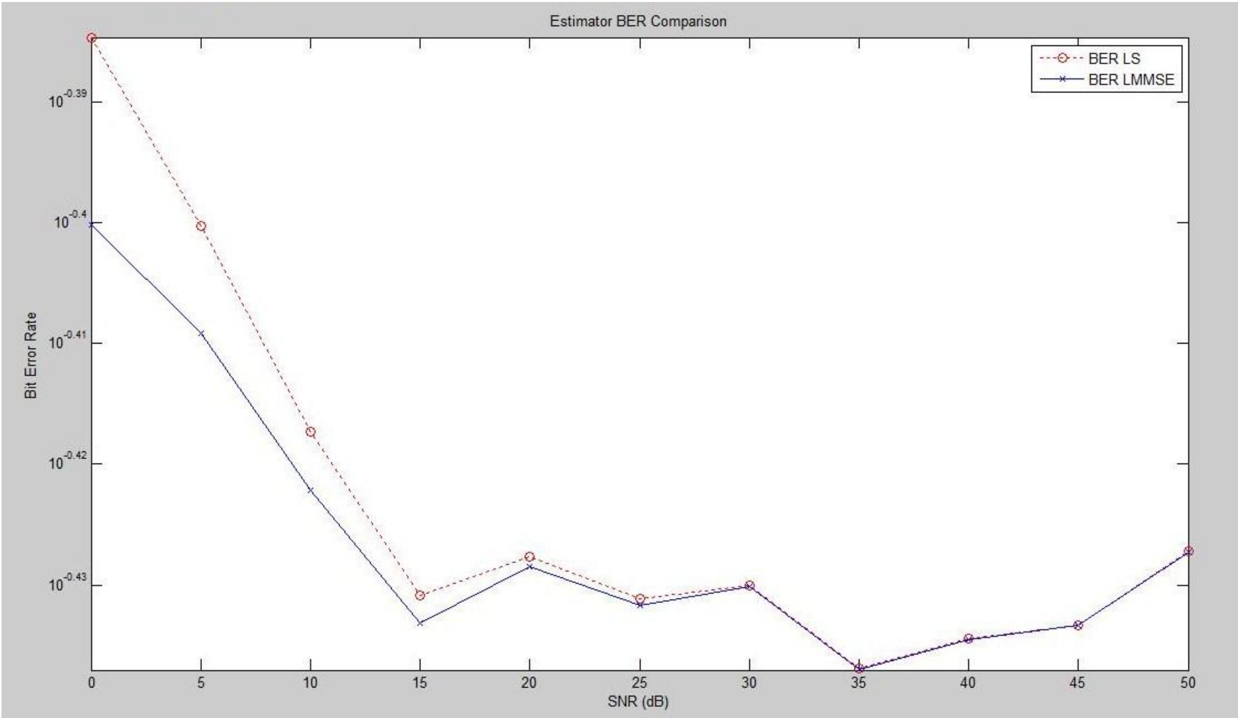
For 1000 bits

Comparison between **LS estimator** and **LMMSE estimator**

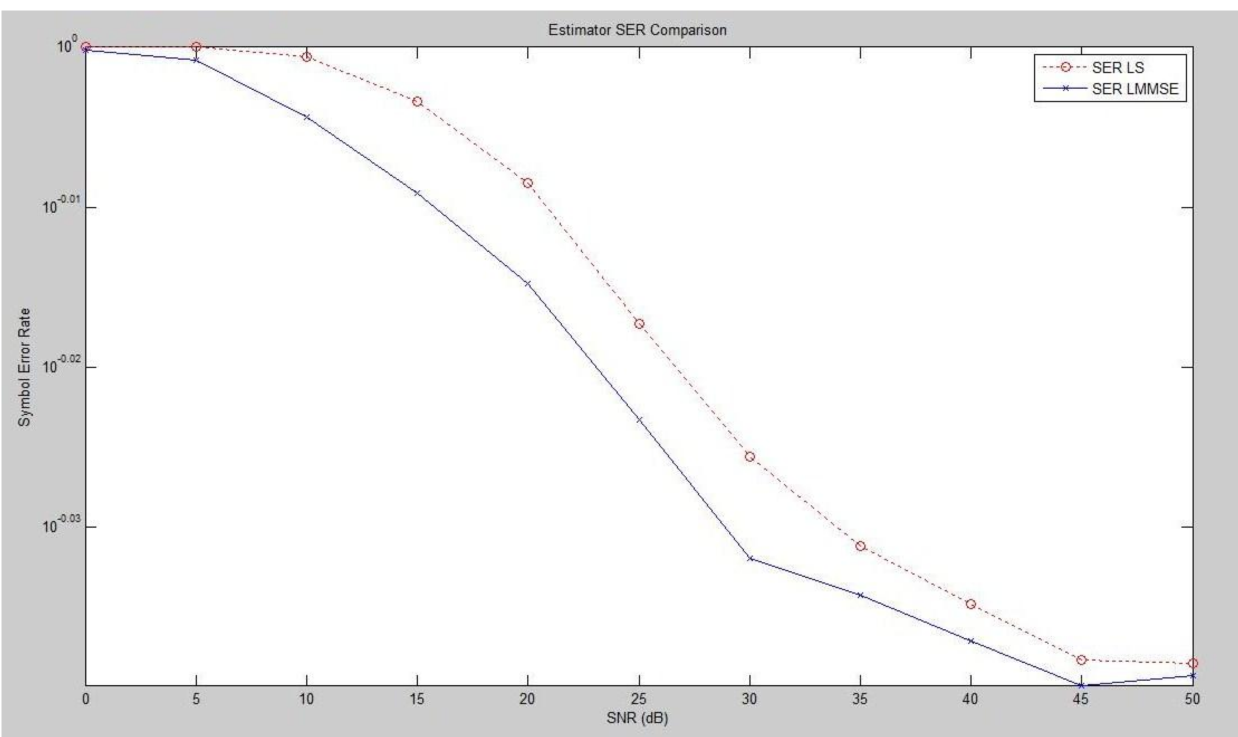
MSE comparison plot (fig 6.7)



BER comparison plot (fig 6.8)



SER comparison plot (fig 6.9)



CHAPTER-7
CONCLUSION

CONCLUSION

In this project we have simulated successively single bit without noise OFDM system , multiple bit without noise OFDM system , multiple bit with noise OFDM system for single user.

Then we simulated OFDM system for multiple user with and without noise. then we simulated MC- CDMA systems, the BER Vs SNR plots for the previously mentioned systems were obtained and were found to be satisfactory.

Then the simulation of channel estimation was carried out using **LS** and **LMMSE** estimators and the MSE,BER and SER plots were obtained.

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