Chapter One

Introduction

Chapter One

Introduction

1.1 Preface

Long term evolution (LTE) is a standard for wireless data communications technology and an evolution of the Global System for Mobile/Universal Mobile Telecommunication System (GSM/UMTS) standards. The goal of LTE is increasing the capacity and speed of wireless data networks using new digital signal processing (DSP) techniques and modulations that were developed around the turn of the millennium. [1]

Single carrier Frequency Division Multiple Access (SC-FDMA) is a frequency-division multiple access scheme. It deals with the assignment of multiple users to a shared communication resource. It has drawn great attention as an attractive alternative to orthogonal frequency division multiple access (OFDMA), especially in the uplink communications where lower peak to average power ratio (PAPR) greatly benefits the mobile terminal in terms of transmit power efficiency and terminal costs. It has been adopted as the uplink multiple access scheme in 3GPP LTE since its transmitted signal has a lower peak-to-average power ratio (PAPR) than OFDMA. [2]

Since the radio channel is highly dynamic, the transmitted signal travels to the receiver by undergoing many detrimental effects that corrupt the signal and often place limitations on the performance of the system. Channel estimation (CE) techniques allow the receiver to approximate the impulse response of the channel and explain the behavior of the channel. In general, CE techniques can be divided into two major categories such as the trained and blind. The former CE algorithm requires probe sequences that occupy valuable bandwidth whereas the latter uses the received data only. Due of course to their self-sufficiency in training, blind CE techniques are considered more attractive than trained based techniques [3-6].

1.2 Objectives

- Design a simulation model for SC-FDMA system using MATLAB.
- Evaluate the performance of least mean square (LMS) and variable step size least mean square (VSS-LMS) algorithms under different channel environments and different modulation techniques.

1.3 Problem Statement

In LTE SC-FDMA the noise and channel statistics are highly dynamic and hard to be detected; therefore many researchers have proposed channel estimation and equalization algorithms. However; these algorithms require high computational complexity and the performance under high Doppler frequencies get worse.

1.4 Methodology

Using MATLAB the SC-FDMA system will be simulated. The performance of LMS algorithm will be evaluated compared with VSS-LMS algorithm in different channel environments and different modulation types by plotting the Bit Error Rate (BER) and Mean Square Error (MSE) with the SNR for every case and compute the number of addition and multiplication operations in the estimation process.

1.5 Thesis outlines

The thesis will be organized as follows:

In Chapter two the basic principles of Single Carrier Frequency Division Multiple access (SC-FDMA), transmitter and receiver structure of uplink will be described. Also propagation channel models in LTE in addition to the generation of reference signal will be covered. **In Chapter three** the model used in this work and the parameters assumption will be covered in addition to LMS and VSS-LMS channel estimation techniques. In **Chapter four** the simulations and results under different parametric conditions will be included which shows the plots of Minimum Square Error (MSE) and Bit Error Rate (BER) versus signal to noise ratio (SNR) using different modulation schemes and channel models. Finally the conclusion will be found in **Chapter five** followed by the recommendations and future work.

Chapter Two

SC-FDMA Basic Principles

Chapter Two

SC-FDMA Basic Principles

2.1 Introduction

This chapter covers the basic principles of SC-FDMA. Since it's a modified version of OFDMA; the last one will be discussed first by reviewing both of the two multiplexing techniques, OFDM and SC/FDE, and the differences between them. Then the two multiple access techniques, OFDMA and SC-FDMA, will be discussed. Finally the application of SC-FDMA in LTE uplink will be covered in details by illustrating the signal processing operations which done for every TTI duration in addition to the frame format used in this application.

2.2 Orthogonal Frequency Division Multiplexing (OFDM)

Orthogonal frequency division multiplexing (OFDM) is a transmission technique that is built-up by many orthogonal carriers that transmits simultaneously. The main idea behind OFDM is that a signal with a long symbol duration time is less sensitive to multipath fading, than a signal with a short symbol time. Hence, a gain in performance can be achieved by sending several parallel symbols with a long symbol time than sending them in a series with a shorter symbol time. [7]

OFDM transmits multiple modulated subcarriers in parallel [8]; each occupies only a very narrow bandwidth. Since the channel affects only the amplitude and phase of each subcarrier, equalizing each subcarrier's gain and phase does compensation for frequency selective fading. Generation of the multiple subcarriers is done by performing inverse fast Fourier transform (IFFT) processing at the transmitter on blocks of M data symbols; extraction of the subcarriers at the receiver is done by performing the fast Fourier transform (FFT) operation on blocks of M received samples.[9]

2.3 Single Carrier with Frequency Domain Equalization (SC/FDE)

Frequency domain equalization of single carrier modulated signals has been known since the early 1970's. Single carrier systems with frequency domain equalization (SC/FDE), which are combined with FFT (Fast Fourier Transform) processing and contain the cyclic prefix, have the similar low complexity as OFDM systems [9].

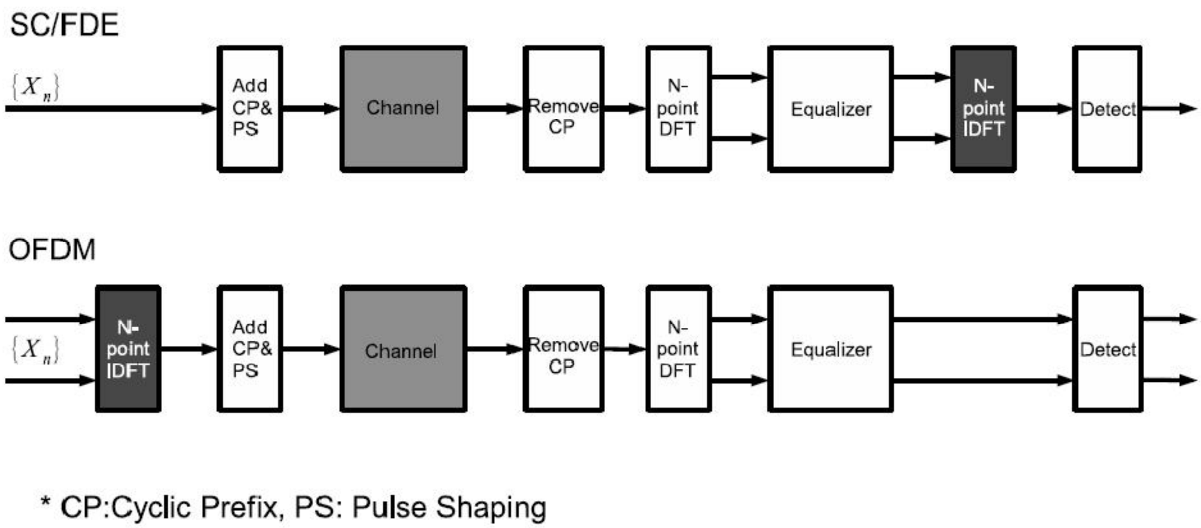

Figure 2.1: Block diagram of OFDM and SC/FDE. [9]

From Figure 2.1 it could be seen that OFDM and SC/FDE have similar structures, the only difference of their block diagram is the position of Inverse Discrete Fourier Transform (IDFT), and so it could be expected that these two have similar performance and efficiency.

However, Figure 2.2 shows differences between OFDM and SC/FDE. In OFDM, detection of signal takes place in frequency domain; all the symbols are allocated in the whole bandwidth, and extracted simultaneously. In SC/FDE, due to the IDFT processing before detection, symbols are extracted in time domain, and they are dealt with one by one.

SC/FDE has some advantages as follow:

- The inherent single carrier structure causes lower peak-to-average power ratio (PAPR) than OFDM.
- SC/FDE has lower sensitivity to carrier frequency offset than OFDM.
- SC/FDE has similar complexity as OFDM in the receiver, and even lower than OFDM in the transmitter, which will benefit the user terminals.
- SC/FDE has similar performance as OFDM.[10]

PAPR is especially important for the uplink of mobile devices. Amplifiers used in circuits today have a linear region in which they must operate so as not to introduce signal distortion, and it is ideal to run with maximum amplification. However, if there is a high PAPR, the device is forced to run with lower amplification so the peak power does not lie in the non-linear gain region. The farther these amplifiers are operated from the peak, the less power efficient the devices become, leading to increased power consumption and while this might not be very important for a base station, it will reduce drain batteries on mobile devices more quickly. Therefore it is important to keep a low PAPR on the uplink. [11]

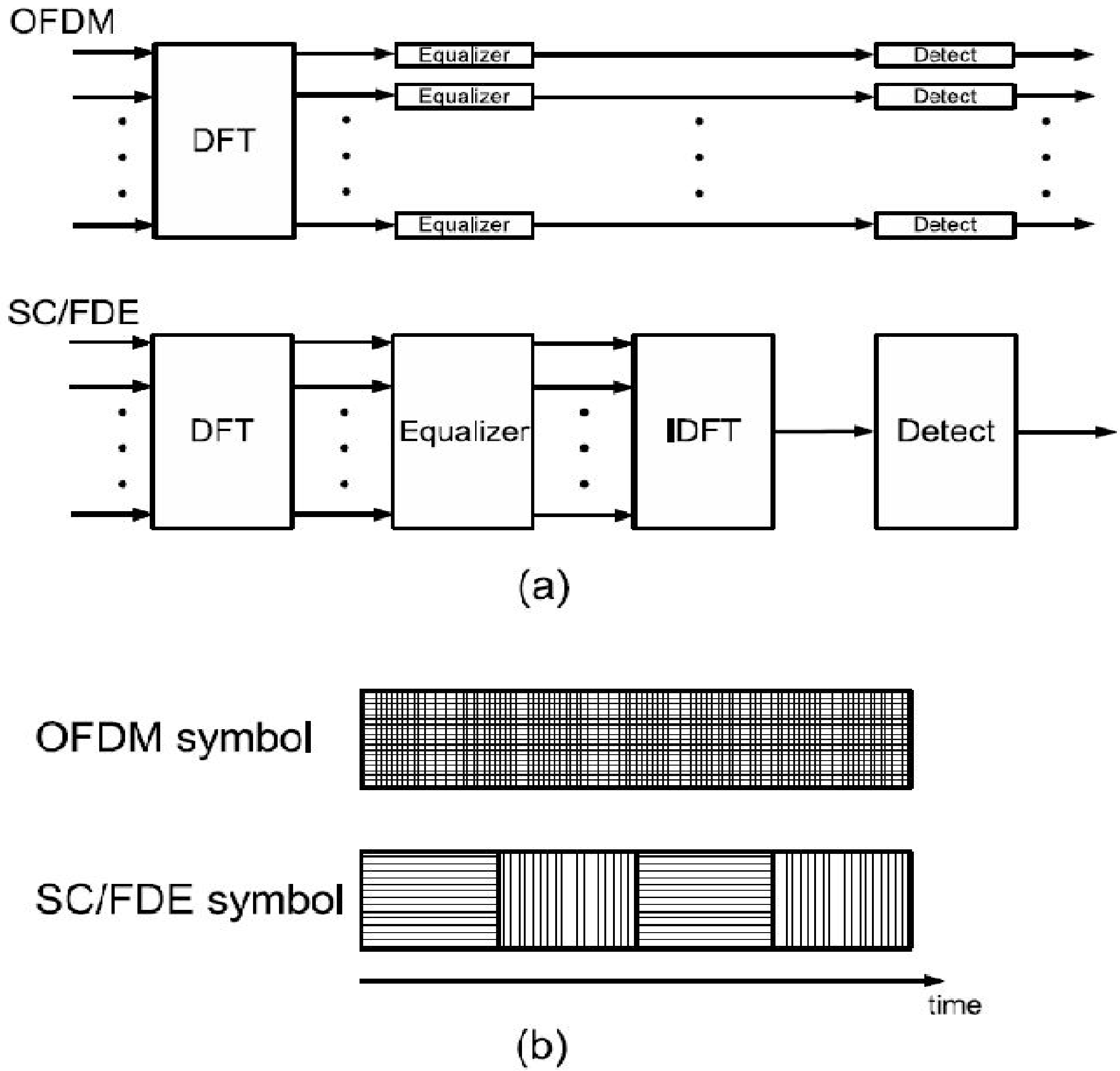

Figure 2.2: Differences between OFDM and SC/FDE. [10]

2.4 SC-FDMA and OFDMA

In cellular applications, a great advantage of OFDMA is due to its robustness in the presence of multipath signal propagation [2]. The immunity to multipath derives from the fact that an OFDMA system transmits information on M orthogonal frequency carriers, each operating at 1/M times the bit rate of the information signal. On the other hand, the

OFDMA waveform exhibits very pronounced envelope fluctuations resulting in a high peak-to-average power ratio (PAPR). Signals with a high PAPR require highly linear power amplifiers to avoid excessive inter modulation distortion. To achieve this linearity, the amplifiers have to operate with a large back off from their peak power. The result is low power efficiency (measured by the ratio of transmitted power to dc power dissipated), which places a significant burden on portable wireless terminals. Another problem with OFDMA in cellular uplink transmissions derives from the inevitable offset in frequency references among the different terminals that transmit simultaneously. Frequency offset destroys the orthogonality of the transmissions, thus introducing multiple access interference.

To overcome these disadvantages, 3GPP is investigating a modified form of OFDMA for uplink transmissions in the “long-term evolution (LTE)” of cellular systems. The modified version of OFDMA, referred to as single carrier FDMA (SC-FDMA). As in OFDMA, the transmitters in an SC-FDMA system use different orthogonal frequencies (subcarriers) to transmit information symbols. However, they transmit the subcarriers sequentially, rather than in parallel. Relative to OFDMA, this arrangement reduces considerably the envelope fluctuations in the transmitted waveform [2, 12-14].

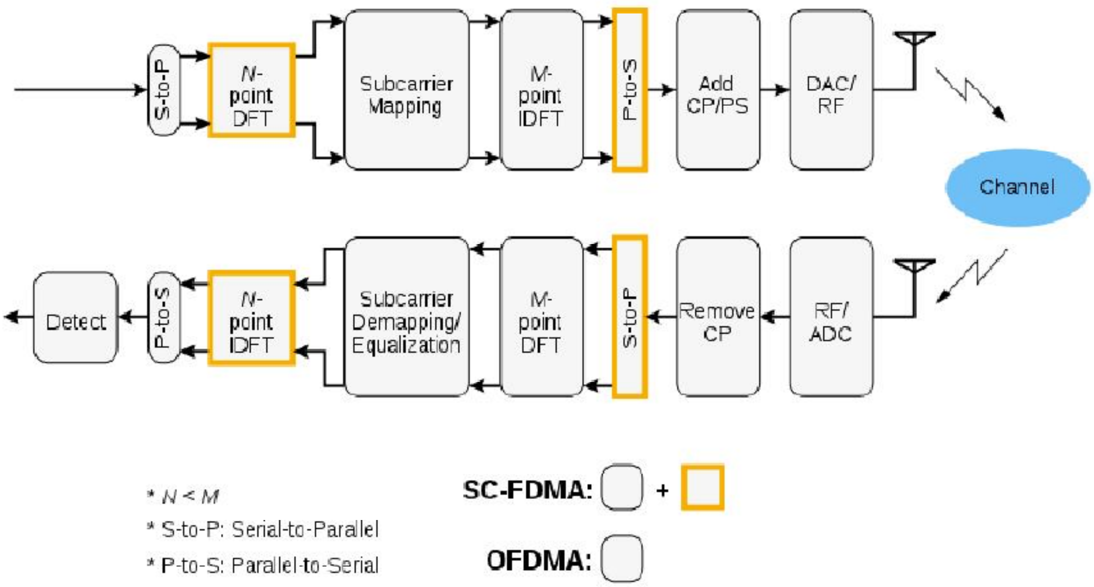

figure2.3: Transmitter and receiver structure of SC-FDMA and OFDMA systems.[14]

Therefore, SC-FDMA signals have inherently lower PAPR than OFDMA signals. However, in cellular systems with severe multipath propagation, the SC-FDMA signals arrive at a base station with substantial inter symbol interference.

The base station employs adaptive frequency domain equalization to cancel this interference. This arrangement makes sense in a cellular system because it reduces the burden of linear amplification in portable terminals at the cost of complex signal processing (frequency domain equalization) at the base station. [2]

2.5 Application of SC-FDMA in 3GPP LTE Uplink

Since SC-FDMA is utilized in the uplink of 3GPP LTE, the implementation of SC-FDMA in 3GPP LTE uplink will be described in this section.

2.5.1 SC-FDMA Transmitter

The transmitter of an SC-FDMA system converts a binary input signal to a sequence of modulated subcarriers. To do so, it performs the signal processing operations shown in Figure 2.4. Signal processing is repetitive in a few different time intervals. Resource assignment takes place in transmit time intervals (TTIs). In 3GPP LTE, a typical TTI is 0.5 ms. The TTI is further divided into time intervals referred to as *blocks*. A block is the time used to transmit all of subcarriers once. [2]

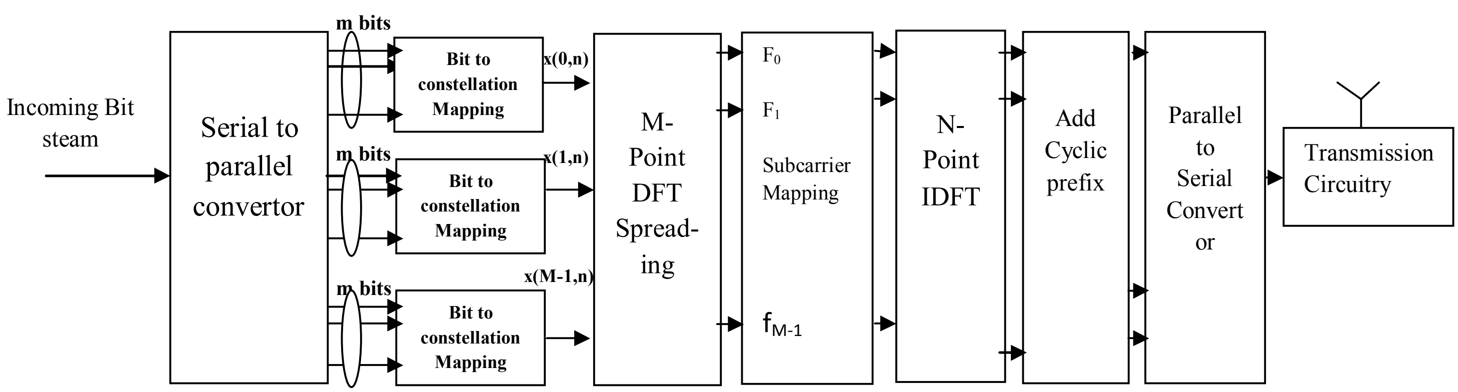

Figure 2.4: SC-FDMA Transmitter.

2.5.1.1 LTE Frame format

In order to transfer data between LTE base station called eNodeB and User Equipment terminals, a strict frame and sub-frame (slots) structure has been defined for the radio interface E-UTRA (Evolved UMTS Terrestrial Radio Access) used in LTE. Two general frame types are distinguished:

i. Type 1 - used in both LTE FDD and TDD duplexing.
ii. Type 2 - used only in LTE TDD duplexing.

Due to more frequent use [15], in this thesis mainly Type 1 will be investigated.

1. Type 1 frame Format

The generic LTE frame has duration of 10 msec. It is divided into ten sub-frames also known as TTI (Transmission Time Interval) [16]. Each sub-frame duration is $T_{subframe} = 1.0$ msec and it consists of two time slots. As it shown in Figure 2.5 each frame can also be considered as a structure divided into 20 separate time slots each with a duration of 0.5 msec.

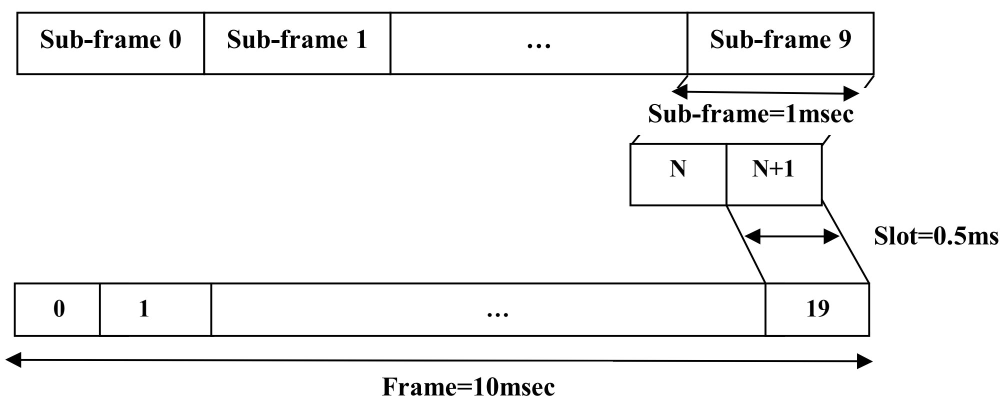

Figure 2.5: Uplink frame and sub-frame format for structure type 1.

The number of symbols (consider one symbol as DFT blocks + cyclic prefix (CP)) in one slot is determined by CP length [17]. When normal CP is used there are seven SC-FDMA symbols per slot, but for extended CP only six symbols can be transmitted. This is illustrated in Figure 2.6.

Comparing the TTI in LTE with the sub-frames in HSPA systems the LTE sub-frame is two times shorter than the sub-frame in HSPA which has duration of 2 msec. The main objective of the LTE sub-frames structure is to provide higher data rates and smaller latency. This is achieved inter alia using a shorter LTE sub-frames duration so e.g. delays caused by retransmissions are reduced.

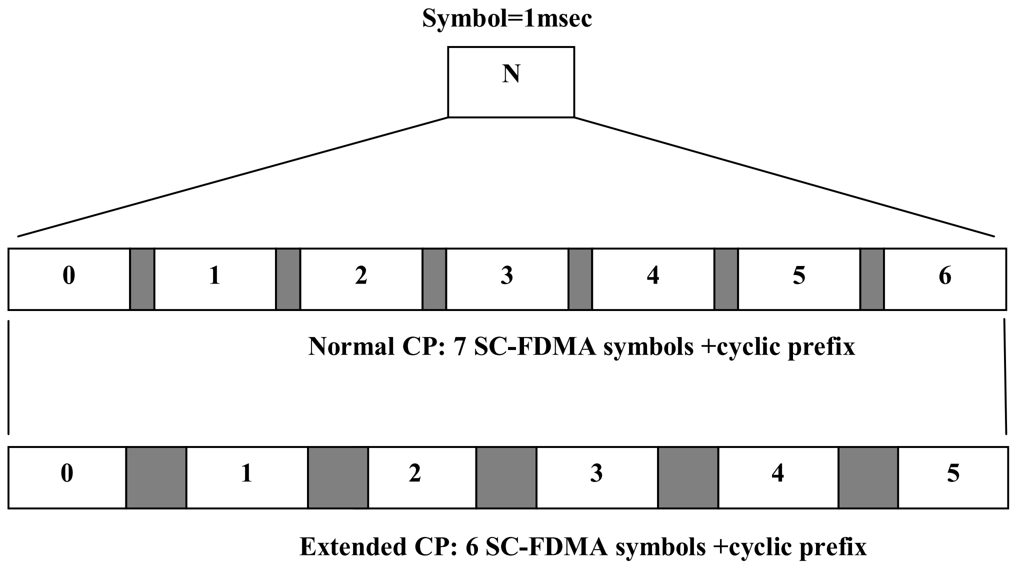

Figure 2.6: Symbol structure consisting of seven or six modulation symbols depending on the cycle prefix length.

2. Type 2 LTE Frame Format

Type 2 frames are used in LTE-Time Division Duplexing systems only. The same frame structure is used both in uplink and downlink. The difference is that main frame is divided into two half-frames, each of five milliseconds. One half-frame is then built up of five sub-frames of $T_{subframe}$ = 1.0msec each. One of the main advantages of using LTE-TDD systems is the ability to dynamically change bandwidth for uplink and downlink, depending on current needs, network load etc. [18]

In Figure 2.7, there are always some unused subcarriers at both sides of the occupied frequency, they are considered as guard band. [10]

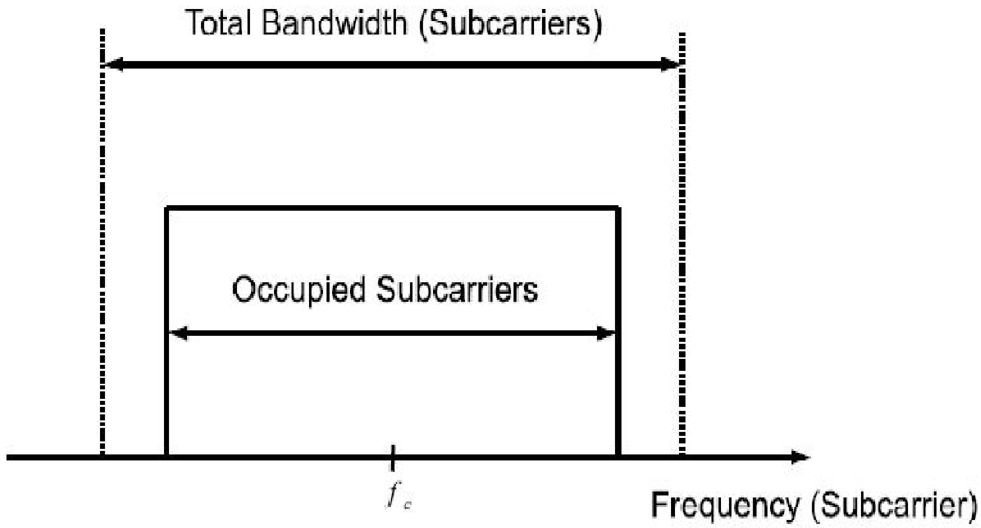

Figure 2.7: Physical Mapping of one block in RF frequency domain [10].

2.5.1.2 Modulation

At the input to the transmitter, a baseband modulator transforms the binary input to a multilevel sequence of complex numbers x_n in one of several possible modulation formats including binary phase shift keying (BPSK), quaternary PSK (QPSK), 16 level quadrature amplitude modulation (16-QAM) and 64-QAM. The system adapts the modulation format, and thereby the transmission bit rate, to match the current channel conditions of each terminal. [19]

The modulation schemes used in 3GPP LTE uplink are BPSK, QPSK, 8PSK and 16QAM [12].

2.5.1.3 N Point DFT

The transmitter next groups the modulation symbols, x_n into blocks each containing *N* symbols. The first step in modulating the SC-FDMA subcarriers is to perform an *N*-point discrete Fourier transform (DFT), to produce a frequency domain representation X_k of the input symbols. [19]

2.5.1.4 Subcarrier Mapping

There are two types of subcarrier mapping in an SC-FDMA system, localized

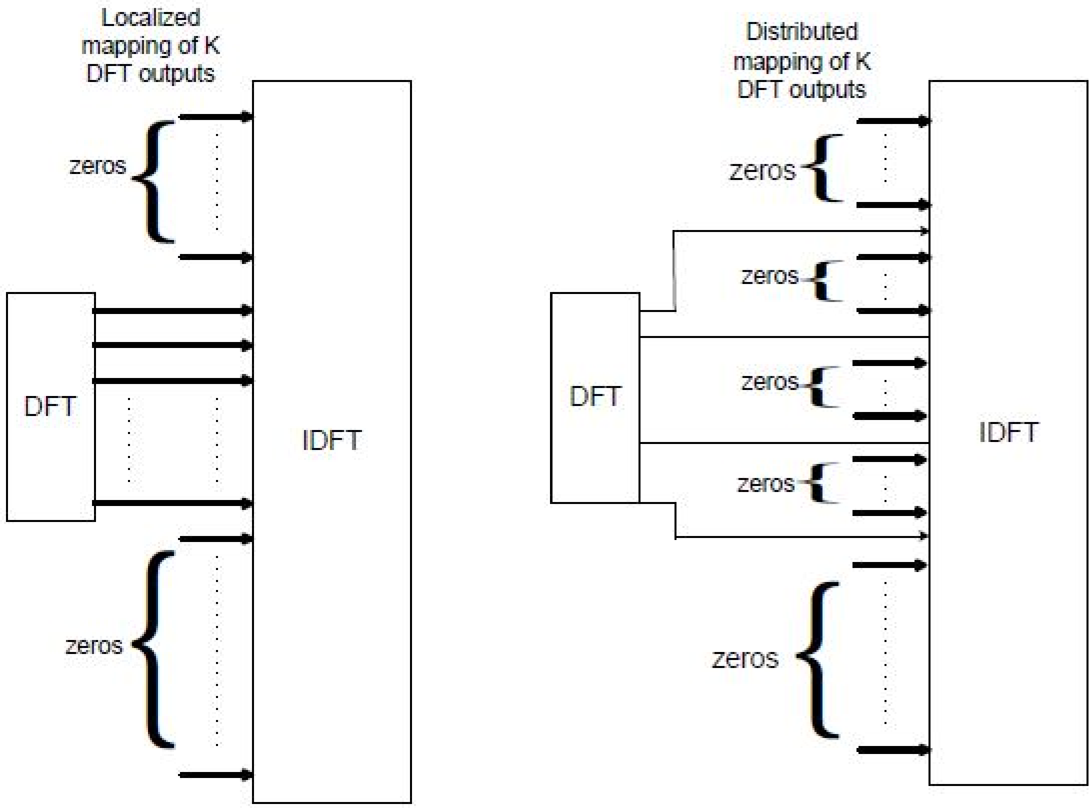

Figure 2.8: Localized and Distributed subcarrier mapping [20].

(LFDMA) and distributed (DFDMA). In LFDMA, the K outputs of the DFT block from a particular terminal are mapped to a chunk of K adjacent sub carriers, whereas in DFDMA the symbols are mapped to subcarriers which are equally spaced across a particular part of the (or the entire) bandwidth. Interleaved SC-FDMA (IFDMA) is a special case of DFDMA, where the chunk of K subcarriers occupies the entire bandwidth with a spacing of J -1 subcarriers. In both of the subcarrier allocation methods, the transmitter assigns zero amplitude to the remaining N_{total}-K unused subcarriers. Figure 2.8 illustrates the different types of subcarrier mapping methods.

Figure 2.9 demonstrates an example of the two different SC-FDMA subcarrier mapping method, for K = 3 symbols per block, N_{total} = 9 subcarriers, and J = 3 user terminals. The input time domain symbols from user terminal J_0 are u_0; u_1; and u_2, and U_0; U_1; and U_2 represent the outputs of the DFT blocks.[20]

In localized mapping, outputs of the DFT blocks will occupy the subcarriers 0; 1; and 2, and the rest of the subcarriers will have zero amplitude. In a similar manner the DFT outputs from user J1 and J2 will each occupy 3 subcarriers, starting with subcarrier number 3 and 6, respectively. In the Interleaved mapping, the DFT outputs from terminal J0 will be uniformly distributed among the 9 subcarriers starting with the 0^{th} one, and 3- 1 = 2 zeros will be assigned to the subcarriers in between the occupied ones. Similarly, the DFT outputs from user terminal J_1 and J_2 will each occupy 9 equally spaced subcarriers starting with subcarrier number 1 and 2, respectively. [20]

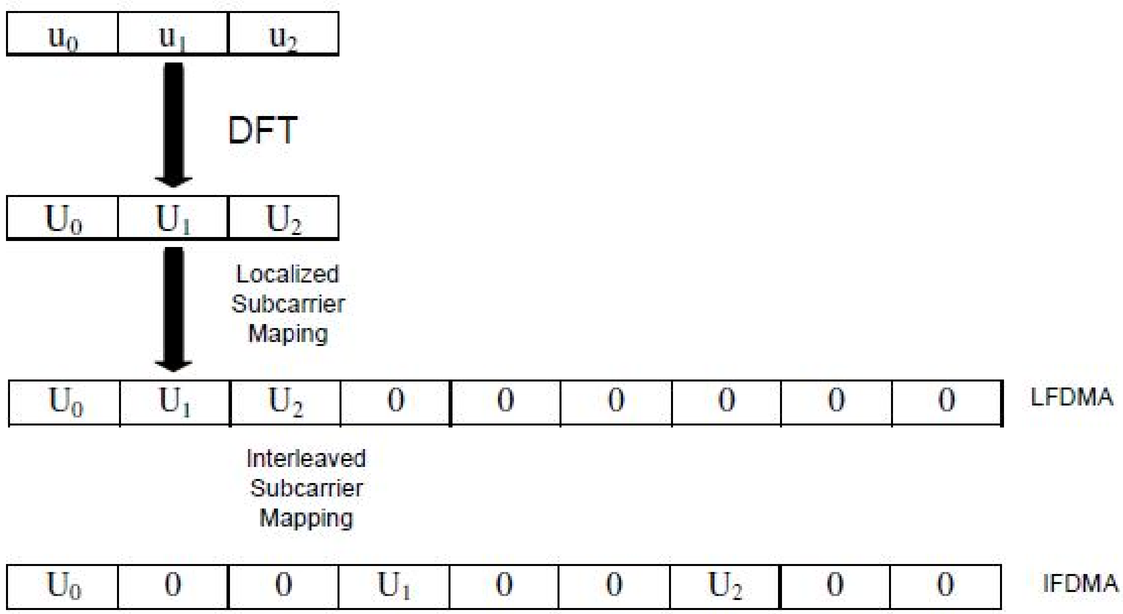

Figure 2.9: An example of localized and interleaved subcarrier mapping method [19].

In LTE uplink each of the N DFT outputs is mapped to one of the M> N orthogonal subcarriers that can be transmitted. As in OFDMA, a typical value of M is 256 subcarriers and N = M/Q is an integer sub multiple of M. Q is the bandwidth expansion factor of the symbol sequence. If all terminals transmit N symbols per block, the system can handle Q simultaneous transmissions without co channel interference. The result of the subcarrier mapping is the set ˜X l (l = 0, 1, 2, . . . , M − 1) of complex subcarrier amplitudes, where N of the amplitudes are non-zero[19]. Figure 2.10 illustrates subcarrier mapping in SC-FDMA for k users.

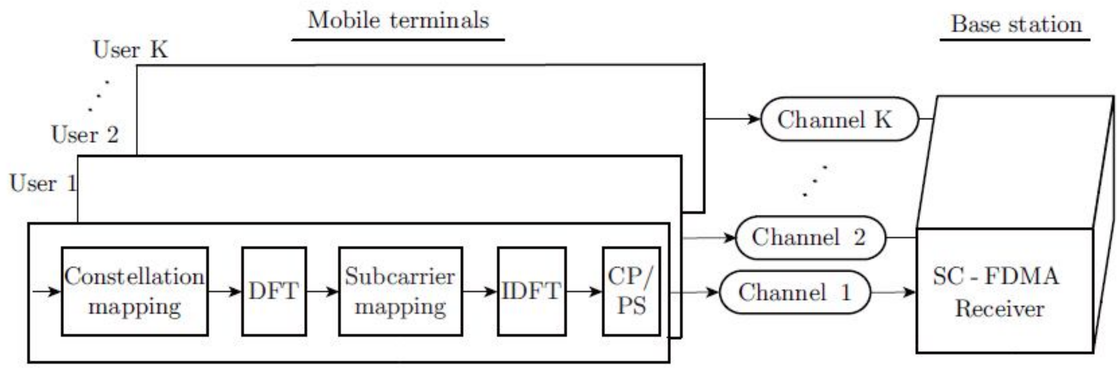

Figure 2.10: SC-FDMA system subcarrier mapping block diagram [21].

2.5.1.5 M Point IDFT

An M-point inverse DFT (IDFT) transforms the subcarrier amplitudes to a complex time domain signal $\tilde{X}_m$. Each $\tilde{X}_m$ then modulates a single frequency carrier and all the modulated symbols are transmitted sequentially. [19]

2.5.1.6 Cyclic Prefix (CP)

The transmitter performs two other signal processing operations prior to transmission. It inserts a set of symbols referred to as a cyclic prefix (CP). The transmitter also performs a linear filtering operation referred to as pulse shaping in order to reduce out of band signal energy.

The cyclic prefix is a repetition of the last data symbols in a block which is added at the start of each block. Its length in data symbols exceeds the maximum expected delay spread.

Utilizing a cyclic prefix is an efficient method to prevent IBI (Inter-Block Interference) between two successive blocks. In general, CP is a copy of the last part of the block. The existence of CP has a double effect preventing IBI [9, 19].

1. CP provides a guard time between two successive blocks. If the length of CP is longer than the maximum spread delay of channel, there won't be any IBI.
2. Because CP is a copy of the last part of the block, it will avoid the ICI (Inter Carrier Interference) between subcarriers.

However, the drawback of the cyclic prefix is that it doesn't carry any new information, so it will lower the efficiency of the transmission. [10]

2.5.2 SC-FDMA Receiver

Just like the transmitter, the two major computations required to get back the transmitted symbols in an SC-FDMA receiver are the DFT and IDFT. In an SC-FDMA receiver, after discarding the cyclic prefix, the DFT block transforms the received time domain signal into the frequency domain. Afterwards, subcarrier de-mapping is done following the same method (distributed, localized or interleaved) in which subcarrier mapping was done in the transmitter. Next, an equalizer compensates for the distortion caused by the multipath propagation channel. After the equalization process, the IDFT block transforms the signal into the time domain, and finally, a detector recovers the original transmitted symbols.

The equalization process in an SC-FDMA receiver is done in the frequency domain. Frequency domain equalization is one of the most important properties of SC-FDMA technology. Conventional time domain equalization approaches for broadband multipath channels are not advantageous because of the complexity and required digital signal processing increases with the increase of the length of the channel impulse response. Frequency domain equalization, on the other hand, is more computationally efficient and therefore desirable because the DFT size does not grow linearly with the length of the channel impulse response. Most of the time domain equalization techniques such as Minimum Mean Squared Error Equalization (MMSE), Decision Feedback Equalization (DFE), and turbo equalization can be implemented in the frequency domain. [20]

2.6 Channel Model

The realistic channel model for wireless communication is essential for the analysis, design and deployment of the communication systems. The correct knowledge of the mobile channel models are significant for testing, optimization and performance improvements signal processing algorithms. Wireless communication has the phenomenon named multi path fading. This is because of reflection from objects when the signal is transmitted in the channel. As a result signal reaches the receiver by two or more paths with some delay as shown in the figure 2.11 [23 - 26].

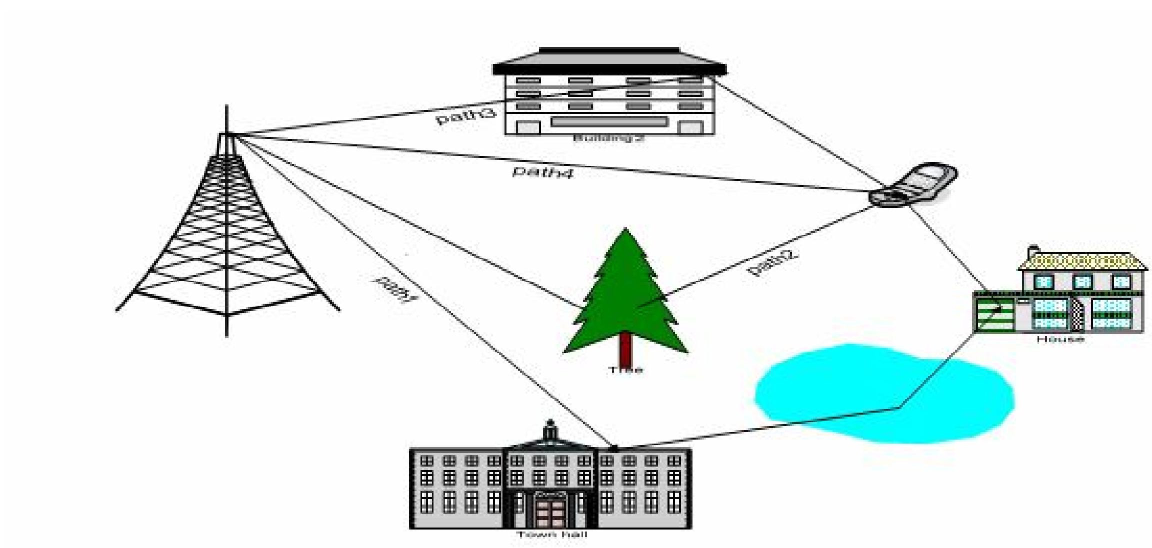

Figure 2.11: Channel Model. [23]

Multipath propagation will be modeled as

$$y(n) = h_1 s(n - t_1) + h_2 s(n - t_2) + \cdots + h_T s(n - t_T) + W(n) \qquad (2.1)$$

Where:

y(n) is the received signal.

h_T are the channel coefficient.

s(n) is the transmitted signal.

$h_T s(n-t_T)$ is delayed version of transmitted signal due to reflection.

w(n) is additive noise.

Noise is usually measured by SNR (Signal to Noise Ratio), which is defined as the ratio of the received signal power to the power of noise within the bandwidth of the transmitted signal s(n) . To make the simulation close to reality some kind of channel model should be chosen. There are different channel models like Rayleigh fading channel and Rician. In Rayleigh fading channel, there is no line of sight between transmitter and receiver and channel taps are independent where as in Rician fading channel, the fading dips are low due to presence of line of sight. [23]

2.6.1 Multipath Propagation Channel

The received signal affected by different propagation paths. These paths are called channel taps. Which cause different delayed versions of transmitted signal as described in equation (2.1). The intensity of a received signal through multi path as a function of time delay called power delay profile (PDP). PDP is an important channel characteristic parameter which is necessary for receiving techniques such as Minimum Mean-Square Error (MMSE) channel estimation. Researchers try to obtain the information of

PDP through extensive field measurements and numerical simulations but these measurements hardly provide accurate PDP information since the radio propagation environment is always changing [9]. [23]

2.6.2 Propagation aspects and Parameters

The behavior of a multipath channel needs to be characterized in order to model the channel. The concepts of Doppler spread, coherence time, and delay spread and coherence bandwidth are used describe various aspects of the multipath channel.

2.6.2.1 Delay Spread

To measure the performance capabilities of a wireless channel, the time dispersion or multipath delay spread related to small scale fading of the channel needs to be calculated in a convenient way. One simple measure of delay spread is the overall extent of path delays called the excess delay spread. This is only convenient way because different channels with the same excess delays can exhibit different power profiles which have more or less impact on the performance of the system under consideration. A more efficient method to determine channel delay spread is the root mean square (rms) delay spread (τ_{rms}) which is a statistical measure and gives the spread of delayed components about the mean value of the channel power delay profile. Mathematically, rms delay spread can be described as second central moment of the channel power delay profile [29] which is written as follows:

$$\tau_{rms} = \sqrt{\frac{\sum_{n=0}^{N-1} P_n(\tau_n - \tau_m)^2}{\sum_{n=0}^{N-1} P_n}} \tag{2.2}$$

$$\tau_m = \sqrt{\frac{\sum_{n=0}^{N-1} P_n \ \tau_n}{\sum_{n=0}^{N-1} P_n}} \tag{2.3}$$

Where (τ_{rms}) is the mean excess delay.[27]

2.6.2.2 Coherence Bandwidth

When the channel behavior is studied in frequency domain than coherence bandwidth Δf is of concern. The frequency band, in which the amplitudes of all frequency components of the transmitted signal are correlated, i.e., with equal gains and linear phases, is known as coherence bandwidth of that channel [30].

The channel behavior remains invariant over this bandwidth. The coherence bandwidth varies in inverse proportion to the delay spread. A multipath channel can be categorized as frequency flat fading or frequency selective fading in the following way.

Frequency flat fading: A channel is referred to as frequency flat if the coherence bandwidth $\Delta f >> B$, where B is the signal bandwidth. All frequency components of the signal will experience the same amount of fading.

Frequency selective fading: A channel is referred to as frequency selective if the coherence bandwidth $\Delta f <= B$. In this case different frequency components will undergo different amount of fading. The channel acts as a filter since the channel coherence bandwidth is less than the signal bandwidth; hence frequency selective fading takes place [28].

2.6.2.3 Doppler Spread

The Doppler spread arises due to the motion of mobile terminal. As the UE moves the length of path between transmitter and receiver changes, as a result the amplitude, phase and filtering applied to the transmitted signal vary with time according to the mobile speed. For an unmodulated carrier, the output is time varying and has nonzero spectral width which is Doppler spread. For a single path between the mobile terminal and the base station, there will be zero Doppler spread with a simple shift of the carrier frequency (Doppler frequency shift) at the base station. The Doppler frequency depends on the angle of movement of the mobile terminal relative to the base station [23, 27 and 29].

2.6.2.4 Coherence Time

The time over which the characteristics of a channel do not change significantly is termed as coherence time. The reciprocal of the Doppler shift is described as the coherence time of the channel. Mathematically we can describe coherence time as follows:

$T_C=1/2\pi vrms$, where υ_{rms} is root mean square vale of Doppler spread.

The coherence time is related to the power control schemes, error correction and interleaving schemes and to the design of channel estimation techniques at the receiver [27].

2.6.3 Standard Channel models

Standard channel models can be developed by setting up frame work for generic channel models and finding set of parameters that need to be determined for the description of the channel. The other method is to set up measurement campaigns and extracting numerical values of parameters and their statistical distributions [30].

When designing LTE, different requirements are considered; User Equipment (UE) and Base Station (BS) performance requirements which are crucial part of LTE standards, Radio Resource Management (RRM) requirements to ensure that the available resources are used in an efficient way to provide end users the desired quality of service, the RF performance requirements to facilitate the existence of LTE with other systems (e.g., 2G/3G) systems. The standard channel models play a vital role in the assessment of these requirements. In the following section, some standard channel models are discussed which are used in the design and evolution of the UMTS-LTE system [27, 31].

2.6.3.1 ITU Multipath Channel Models

The ITU standard multipath channel models proposed by ITU used for the development of 3G 'IMT-2000' group of radio access systems are basically similar in structure to the 3GPP multipath channel models. The aim of these channel models is to develop standards that help system designers and network planners for system designs and performance verification. Instead of defining propagation models for all possible environments, ITU proposed a set of test environments that adequately span the all possible operating environments and user mobility [27, 32].

I. ITU Pedestrian A, B

The mobile speed is considered to be 3 km/h in each of these cases. For Pedestrian models the base stations with low antennas height are situated outdoors while the pedestrian user are located inside buildings or in open areas. Fading can follow Rayleigh or Rician distribution depending upon the location of the user. The number of taps in case of Pedestrian-A model is 3 while Pedestrian-B has 6 taps. The average powers and the relative delays for the taps of multipath channels based on ITU recommendations are given in table 2.1. [27, 32]

Table 2.1: Average Powers and Relative Delays of ITU multipath Pedestrian-A and Pedestrian-B cases. [23]

Tap No.	Pedestrian-A		Pedestrian-B	
	Relative Delay(ns)	Average Power(dB)	Relative Delay(ns)	Average Power(dB)
1	0	0.0	0	0.0
2	110	-9.7	200	-0.9
3	190	-19.2	800	-4.9
4	410	-22.8	1200	-8.0
5	NA	NA	2300	-7.8
6	NA	NA	3700	-23.9

II. ITU Vehicular-A (V-30, V-120 and V-350)

The vehicular environment is categorized by large macro cells with higher capacity, limited spectrum and large transmits power. The received signal is composed of multipath reflections without LOS component. The received signal power level decreases with distance for which pass loss exponent varies between 3 and 5 in the case of urban and suburban areas. In rural areas path loss may be lower than previous while in mountainous areas, neglecting the path blockage, a path loss attenuation exponent closer to 2 may be appropriate.

For vehicular environments, we used the ITU vehicular-A channel models in our work. The mobile speed considered is 30 km/h, 120 km/h and 350 km/h. The propagation scenarios for LTE with speeds from 120 km/h to 350 km/h are also defined in [43] to model high speed scenarios (e.g., high speed train scenario at speed 350km/h). The maximum carrier frequency over all frequency bands is f=2690 MHz and the Doppler shift at speed v=350 km/h is 900 Hz. The average powers and the relative delays for the taps of multipath channels based on ITU recommendations are given in table 2.2. [27], [32]

Table 2.2 Average Powers and Relative Delays for ITU Vehicular-A Test Environment. [23]

Tap No.	Vehicular-A		Vehicular-B	
	Relative Delay(ns)	Average Power(dB)	Relative Delay(ns)	Average Power(dB)
1	0	0	0	-2.5
2	310	-1	300	0.0
3	710	-9	8900	-12.8
4	1090	-10	12900	-10.0
5	1730	-15	17100	-25.2
6	2510	-20	20000	-16.0

2.6.3.2 Extended ITU models

The analysis done by ITU-R showed that evolution of 3G systems to future generation networks will require technology changes on large scale while new quality of service (QoS) requirements will require increased transmission bandwidth. So LTE channel models require more bandwidth as compared to UMTS channel models to account that fact that channel impulses are associated to the delay resolution of the receiver. The LTE channel models developed by 3GPP are based on the existing 3GPP channel models and ITU channel models. The extended ITU models for LTE were given the name of Extended Pedestrian-A (EPA), Extended Vehicular-A (EVA) and Extended TU (ETU). These channel models are classified on the basis of low, medium and high delay spread where low delay spreads are used to model indoor environments with small cell sizes while medium and high delay spreads are used to model urban environments with large cells. The high delay spread models are according to Typical Urban GSM model [43]. The power delay profiles for these channel models are given in tables 2.3, 2.4 and 2.5, respectively [27, 33].

Table 2.3: Power Delay Profiles for Extended ITU Pedestrian-A Model. [27]

Tap No.	Average Power (dB)	Excess Delay(ns)
1	0.0	0.0
2	-1.0	30
3	-2.0	70
4	-3.0	80
5	-8.0	110
6	-17.2	190

7	-20.8	410

Table 2.4: Power Delay Profiles for Extended ITU Vehicular-A Model. [27]

Tap No.	Average Power (dB)	Excess Delay(ns)
1	0.0	0
2	-1.5	30
3	-1.4	150
4	-3.6	310
5	-0.6	370
6	-9.1	710
7	-7.0	1090
8	-12.0	1730
9	-16.9	2510

Table 2.5: Power Delay Profiles for Extended Typical Urban Model. [27]

Tap No.	Average Power (dB)	Excess Delay(ns)
1	-1	0
2	-1	50
3	-1	120
4	0	200
5	0	230
6	0	500
7	-3	1600
8	-5	2300

9	-7	5000

2.7 Channel Estimation

Channel estimation is a vital part of receivers designs used in mobile communication systems. The effect of the channel on the transmitted information must be estimated in order to recover the transmitted information correctly. The estimation of channel effects is often based on an approximate underlying model of the radio propagation channel. The receiver can precisely recover the transmitted information as long as it can keep track of the varying radio propagation channels [27, 44]. Channel estimator can be one dimension or two dimension estimator.

2.7.1 1 D estimator

An estimator that only uses information from one dimension; either time or frequency is referred to as a 1D estimator [22]. For example: Least Square(LS), FIR algorithm, LMMSE, and Gauss-Markov estimator. [10]

2.7.1.2 2D estimator

The meaning of 2D estimators is to estimate unknown data from the information both in time and frequency domain simultaneously. In frequency domain, all the pilot symbols in short blocks are known; the next step is to interpolate unknown data from two pilot symbols at the same frequency. Hence, 2D estimators are not necessary in our case; it will give the same result as 1D estimators, but having higher complexity. [10]

2.7.1.3 Adaptive estimator

Some calculations of the above channel estimators require knowledge of channel correlations R_{HfHf} . Moreover, the statistics of channels in real world change over time. To avoid these drawbacks, the adaptive estimator which is able to update parameters of the estimator continuously was introduced, so that knowledge of channel and noise statistics is not required. [10]

2.8 Literature Review

Several CE techniques have been proposed for LTE SCFDMA systems in the last years:

In 2007, A. ncora and etal has proposed the least square (LS) channel estimation method to minimize the squared differences between the received and estimated signal. This method is widely used in equalization and filtering applications because it does not require knowledge of channel, however, the statistics of channels in real world change over time. Also the inversion of the large dimensional square matrix turns out to be ill-conditioned in the straight application of the LS estimator. [34]

In 2008, L. A. M. R. D. Temino and etal has proposed two dimensional based on Wiener filtering pilot symbol aided CE. It has a good performance; however, it is more complex and requires accurate knowledge of second order channel statistics.[35]

In 2012, Yongkui Ma and etal has proposed the least mean square (LMS) method, its normalized version (NLMS) and recursive least square (RLS) CE algorithms. In these methods the estimators update coefficients continually and do not need prior knowledge of channel statistics; however, for high Doppler frequencies the performance of LMS and NLMS get worse than RLS but the last one needs longer filter for better performance. Also RLS is more complex than LMS and NLMS methods. [36]

In 2010, adaptive LMS channel estimation algorithms have been proposed for LTE uplink by Md. Masud RANA and etal. This algorithm uses adaptive estimator which is able to update parameters of the estimator continuously by periodically transmitting a training sequence which is known to the receiver. [37]

In December 2010, they have proposed the variable step size least mean square VSS-LMS CE. This time-varying step size method is re-selected at each iteration to minimize the sum of the squares of the prior estimation errors up to that current time point. Although this CE algorithm has good performance, MSE and convergence towards true channel coefficient as well as BER performance However, it requires high computational complexity. [38]

In this project an adaptive LMS based Channel Estimation Methods will be focused on since it does not require prior knowledge of channel statistics or noise and simple for practical implementation.

Chapter Three

System Design and Modeling

Chapter Three

System Design and Modeling

3.1 Introduction

MATLAB is a high performance language which has easy to use environment and has many build in functions which is used in this work to steer clear of long code.

Using MATLAB the SC-FDMA system shown in figure 4.1 will be modeled. LMS and VSS-LMS will be applied for estimating the channel coefficients.

This chapter covers the SC-FDMA block diagram used in this work with some details in the channel estimation algorithms since it's the main point of the work. In addition to the parameters and assumption used here.

3.2 Simulation Assumption and Parameters

When using SC-FDMA under LTE umbrella some parameters must be adjusted according to 3GPP specifications. The specifications used in this project for the system are shown in table 3.1and the assumptions for the channel estimation algorithms are shown in table 3.2.

Table 3.1: System assumptions.

Systems parameter	**Assumption**
System bandwidth	5 MHz
Sampling frequency	7.68 MHz
Subcarrier spacing	9.765 kHz (5 MHz/512)

Modulation data type	BPSK,QPSK,16QAM
FFT size	16
Subcarrier mapping scheme	IFDMA
IFFT size	512
Cyclic Prefix	normal
Frame Type	Type 1
Antenna Configuration	SISO
Pilot Spacing	6
Channel model	Extended Pedestrian-A
Maximum Doppler shift	5, 50, 500 Hz
Pilot	Zadoff Chu
Equalization	Zero Force
Channel Estimation	LMS, VSS-LMS

Table 3.2: Channel Estimation Algorithms' assumptions.

Channel Estimation Algorithm parameter	**Assumption**	**Channel Estimation Algorithm**
η	6.0000e-004	LMS
η_0	η_{max}	VSS-LMS
α	0.97	VSS-LMS

β	0.99	VSS-LMS
γ	7e-8	VSS-LMS
η_{min}	0	VSS-LMS
η_{max}	7e-004	VSS-LMS
Number of iterations	300	LMS/VSS-LMS

3.3 SC-FDMA

In order to model the SC-FDMA system shown in figure 3.1 a sequence of bits are generated, no matter if they are represent data or voice since we concern only about delivering them correctly. Using frame 1 as a frame type the data is concatenated. Then the data is modulated using types mentioned in table 3.1 (two types of modulation was used to obtain different data rates so as to get different cases for analysis). The symbols are then transferred to the frequency domain using Fourier transform in order to mapping them by IFDMA mapping and convert them to the time domain to add the cyclic prefix to them which is represents the final step in the transmitter.

The transmitted data is then sent via the channel, also here in this work three types of channel models are used in order to get three different cases according to Doppler shifts and delay and power gain for every tap in that model. In addition to channel, a noise is added to the data before the receiver obtain it.

In the receiver the reverse operations are made. The cyclic prefix was removed then the mapping was restored by transforming the data to the frequency domain. In this time another operation, channel estimation, is inserted before transforming the data back to the time domain. In this work two types of them are used in order to evaluate their performance. The channel estimation techniques are discussed in details in the next section. The symbols are then demodulated according to the modulation type used for that case.

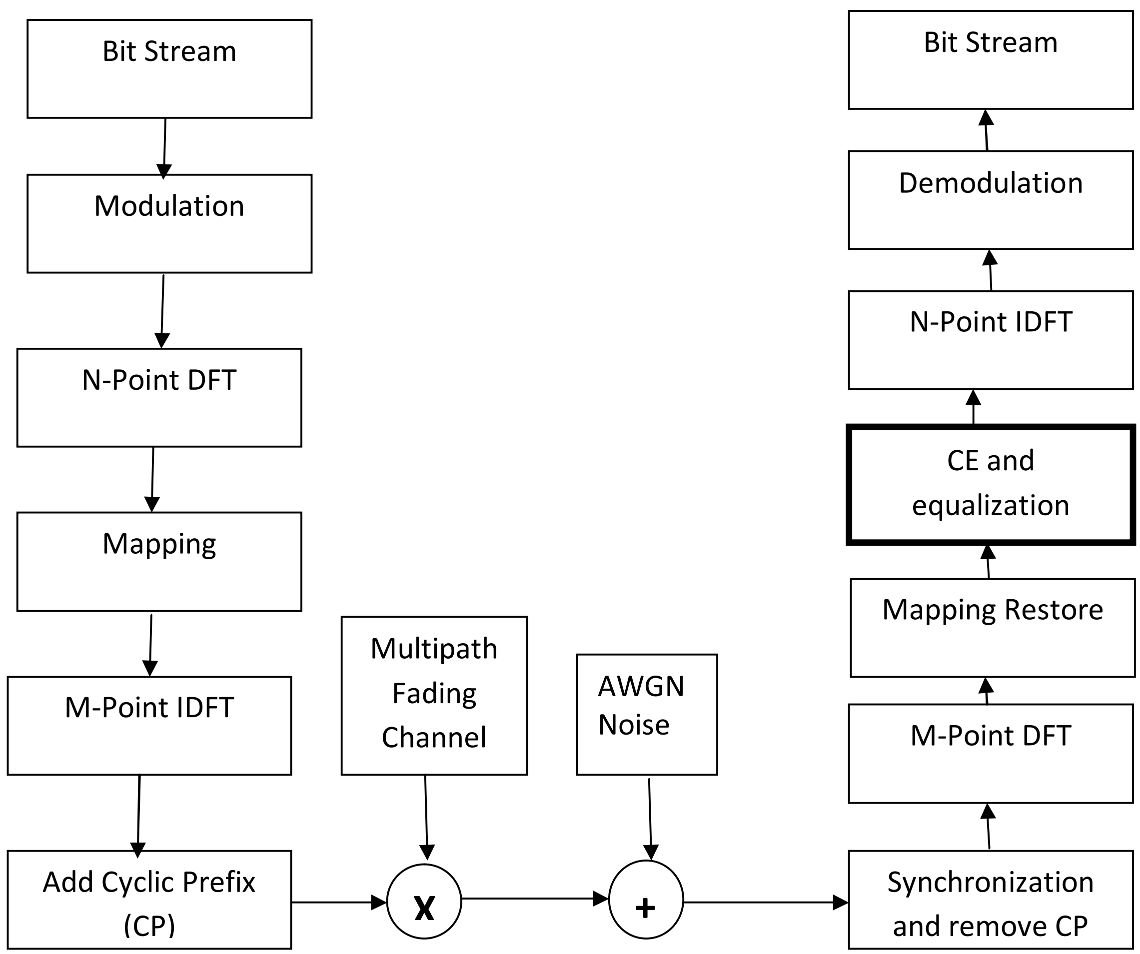

Figure 3.1: LTE SC-FDMA Block diagram.

3.4 Channel Estimation

The LMS and VSS-LMS channel estimation algorithms will be covered in this section in details since it will be used in this work.

3.4.1 LMS algorithm

Stochastic gradient based adaptive algorithms, such as the least mean square (LMS) one, are the most popular in adaptive filtering applications, due to its low computational complexity and very good stability characteristic. Moreover, in the LMS algorithm a previous knowledge of the process statistics is not required [39]. Such advantages make the LMS algorithm adequate for system identification, noise canceling, echo canceling, channel equalization, among other applications [40]. The standard LMS uses a fixed adaptation step size, determined by considering a tradeoff between convergence rate and misadjustment.[41]

If:

s(m) is the transmitted signal

z(m) is the additive white Gaussian noise (AWGN)

W(m) is the channel coefficients

The output from the channel can be expressed as:

$$r(m) = W^T(m)s(m) + z(m) \tag{3.1}$$

The output of the adaptive filter is:

$$y(m) = W_{est}^T(m)s(m) \tag{3.2}$$

Where: W_{est} (m) is the estimated channel coefficients at time m.

The priori estimated error signal needed to update the weights of the adaptive filter is:

$$e(m) = r(m) - y(m) = W^T(m)s(m) + z(m) - W_{est}^T(m)s(m) \qquad (3.3)$$

This error signal is used by the CE to adaptively adjust the weight vector so that the MSE is minimized. If w (m) is the tap-weight vector at the m^{th} iteration then the following recursive equation may be used to update W_{est} (m):

$$W_{est}(m+1) = W_{est}(m) + \eta s(m)e^*(m) \qquad (3.4)$$

Where W_{est} (m+1) denotes the weight vector to be computed at iteration (m + 1) and η is the LMS step size which is related to the rate of convergence. The smaller step size means that a longer reference or training sequence is needed, which would reduce the payload and hence, the bandwidth available for transmitting data. The term [ηs(m)e* (m)] represents the correction factor or adjustment that is applied to the current estimate of the tap-weight vector. The iterative procedure is started with an initial guess W_{est} (0). The detail steps of this CE algorithm are shown in Figure 3.2. [38]

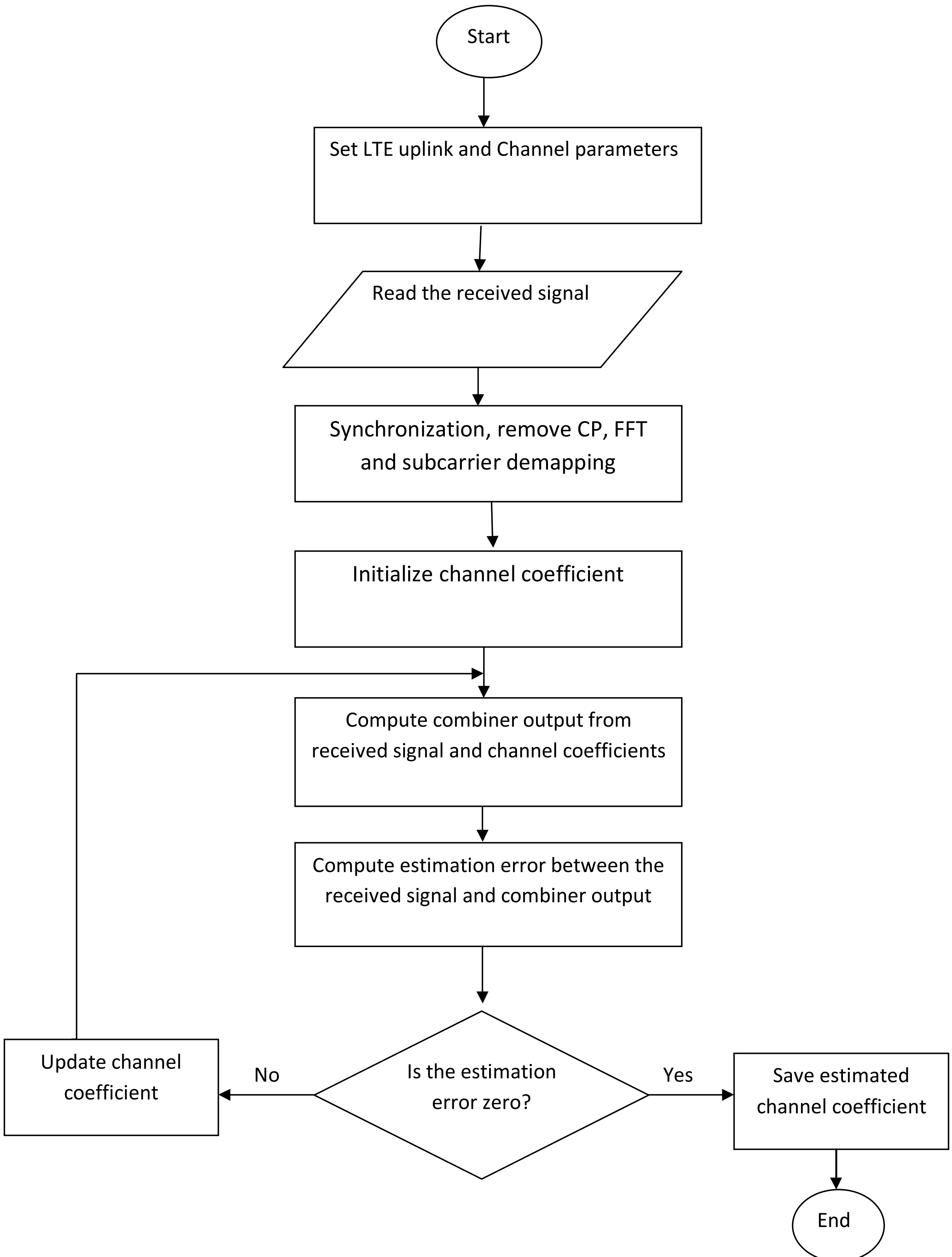

Figure 3.2: Flow Chart of LMS Channel Estimation technique.

3.4.2 Variable Step Size (VSS)-LMS Algorithm

The VSS-LMS algorithm involves one additional step size update equation compared with the standard LMS algorithm. The VSS algorithm is [30], [19]

$$\eta(m+1) = \alpha\eta(m) + \gamma P^2(m) \tag{3.5}$$

$$P(m) = \beta P(m) + (1-\beta)e^T(m)e(m-1) \tag{3.6}$$

Where $0 < \alpha < 1$, $0 < \beta < 1$, and $\gamma > 0$. When the channel is fast time-varying then algorithm cannot accurately measure the autocorrelation between estimation error to control step size update. Control parameters α and β need to be adjusted for a better performance [38]. The detail steps of this CE algorithm is shown in figure 3.3

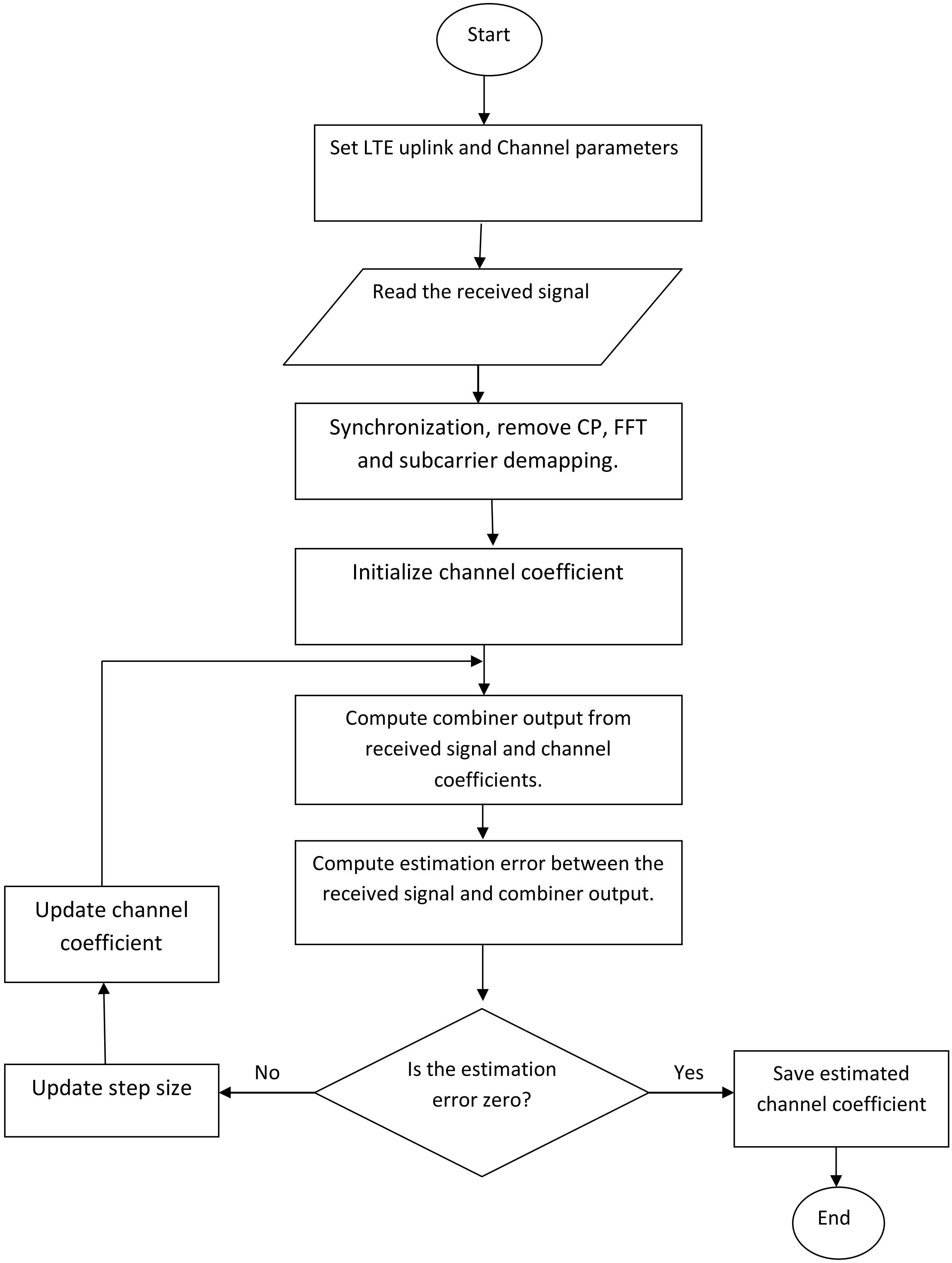

Figure 3.3: Flow chart of VSS-LMS Channel Estimation technique.

Chapter Four

Results and Discussion

Chapter Four

Results and Discussion

4.1 Introduction

In this chapter the performance of LMS and VSS-LMS algorithms will be analyzed in different channel environments and different modulation techniques in terms of Bit Error Rate (BER) and Mean Square Error (MSE), by plotting BER and MSE with the SNR for each case, and compute the complexity, number of addition and multiplication operations in the estimation process.

4.2 Bit Error Rate (BER) Analysis

Bit Error Rate (BER) is an essential parameter for performance evaluation. In this section, the BER of LMS and VSS-LMS channel estimation algorithms is evaluated as a function of SNR. The system was tested under SNR from 0-30dB, LMS's BER is plotted using solid line with star marker and VSS-LMS's one is plotted with dotted line with square marker. The performance was evaluated under eight different cases, two different modulation methods (BPSK and QPSK) and different channel environments (AWGN, Rayleigh fading channels with Doppler shifts= 5, 50 and 500 Hz).

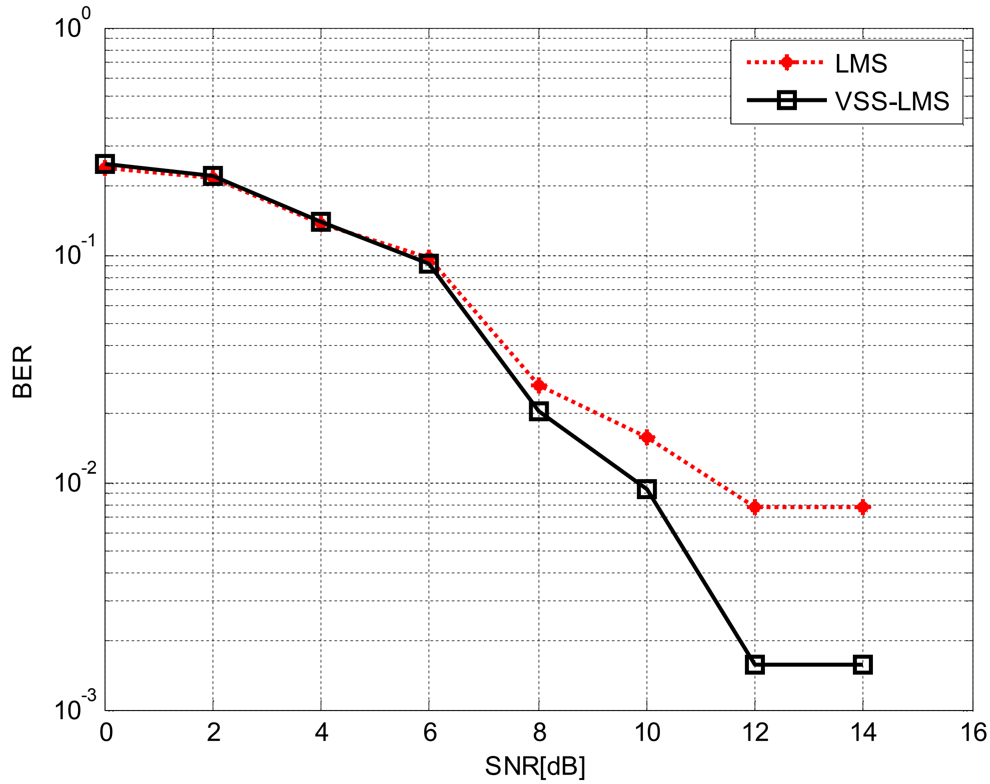

Figure 4.1: BER of LMS and VSS-LMS algorithms as a function of SNR (AWGN, BPSK).

Figure 4.1 shows the BER as a function of SNR for the LMS and VSS-LMS algorithms when using BPSK as the modulation technique and passing the signal through AWGN before it reached the receiver. As it is noticeable, at low SNR(less than 6 dB) the BER for both of the algorithms are almost the same. For higher SNR (greater than 6 dB) the BER for VSS-LMS is better than LMS algorithm, the higher the SNR the better BER for the VSS-LMS as compared to LMS one. In addition to that at SNR greater than 14 dB both of algorithms has BER equal to zero.

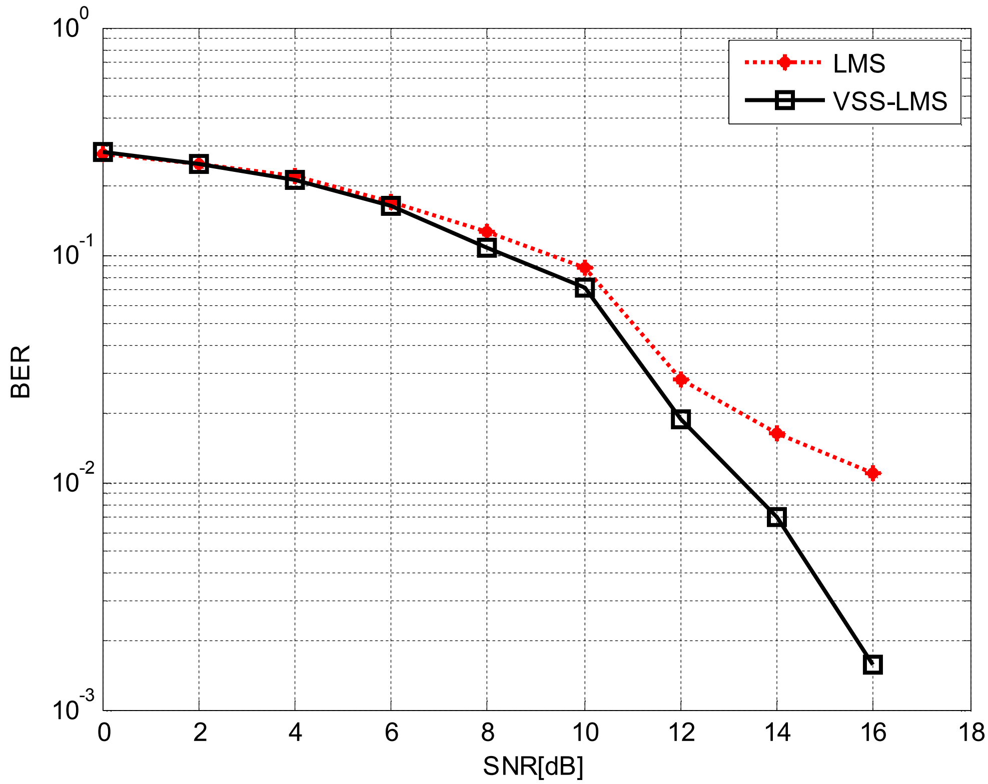

Figure 4.2: BER of LMS and VSS-LMS algorithms as a function of SNR (AWGN, QPSK).

Figure 4.2 shows the BER as a function of SNR for the LMS and VSS-LMS algorithms when using QPSK as the modulation technique and passing the signal through AWGN before it reached the receiver. As it is noticeable, at low SNR(less than 6 dB) the BER for both of the algorithms are almost the same. For higher SNR (greater than 6 dB) the BER for VSS-LMS is better than LMS algorithm, the higher the SNR the better BER for the VSS-LMS as compared to LMS one. In addition to that at SNR greater than 16 dB both of algorithms has BER equal to zero.

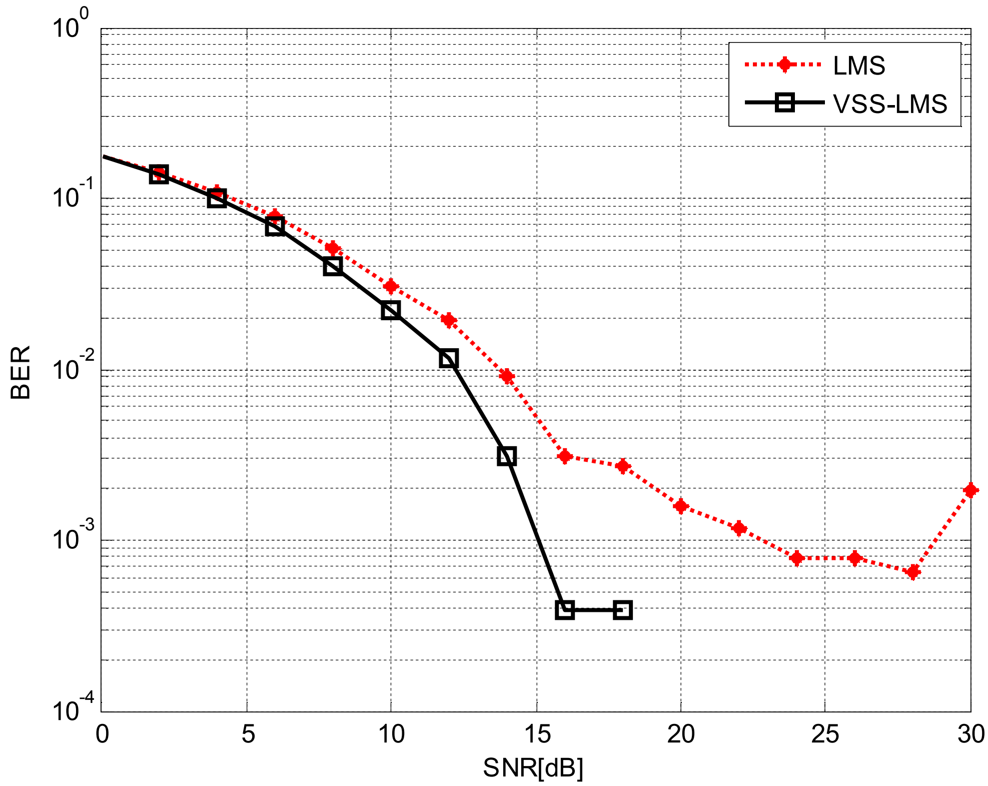

Figure 4.3: BER of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=5Hz, BPSK).

Figure 4.3 shows the BER as a function of SNR for the LMS and VSS-LMS algorithms when using BPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=5Hz. As it is clear, at low SNR(less than 6 dB) the BER for both of the algorithms are almost the same. For higher SNR (greater than 6 dB) the BER for VSS-LMS is better than LMS algorithm. In SNR greater than 12 dB VSS-LMS has BER equal to zero, while the BER of LMS equal to zero for SNR greater than 16 dB.

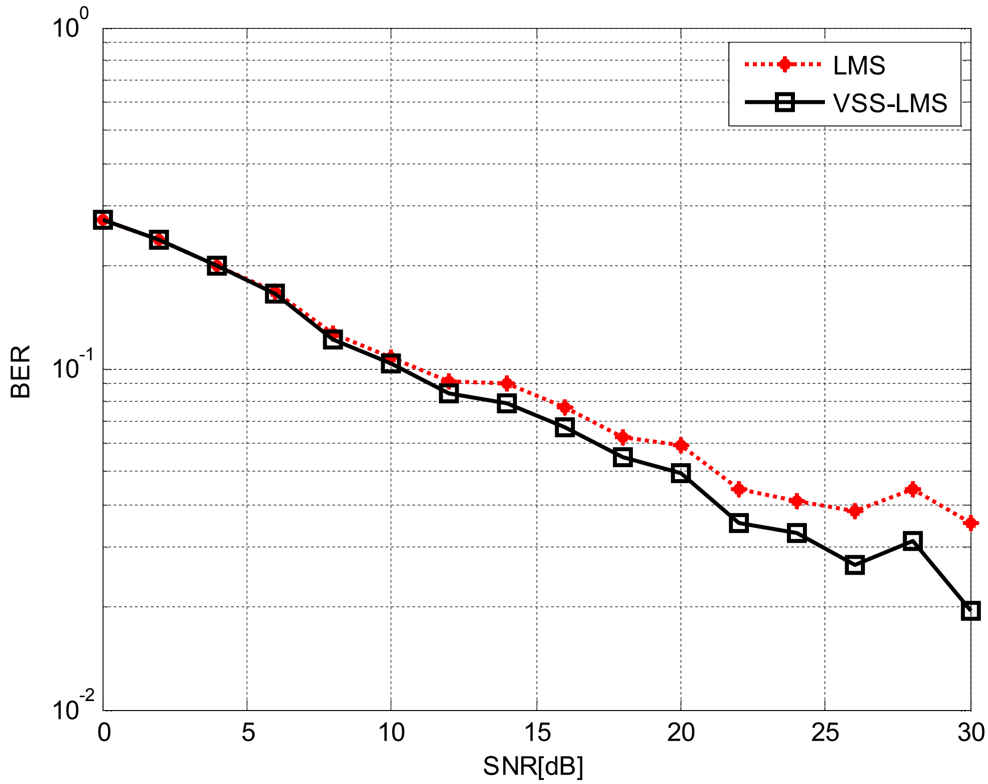

Figure 4.4: BER of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=5Hz, QPSK).

Figure 4.4 shows the BER as a function of SNR for the LMS and VSS-LMS algorithms when using QPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=5Hz. As it is clear, at low SNR(less than 10 dB) the BER for both of the algorithms are almost the same. For higher SNR (greater than 10 dB) the BER for VSS-LMS is better than LMS algorithm.

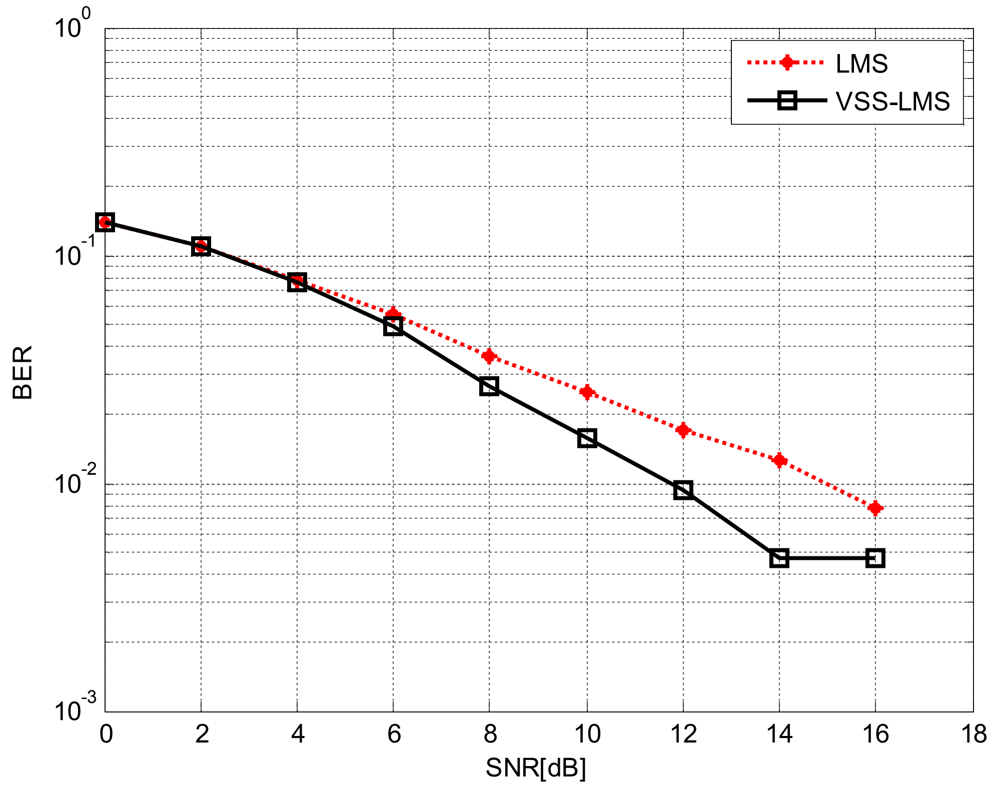

Figure 4.5: BER of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=50Hz, BPSK).

Figure 4.5 shows the BER as a function of SNR for the LMS and VSS-LMS algorithms when using BPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=50Hz. In this figure VSS-LMS is better than LMS algorithm for SNR greater than 4 dB. In SNR greater than 16 dB both of algorithms has BER equal to zero.

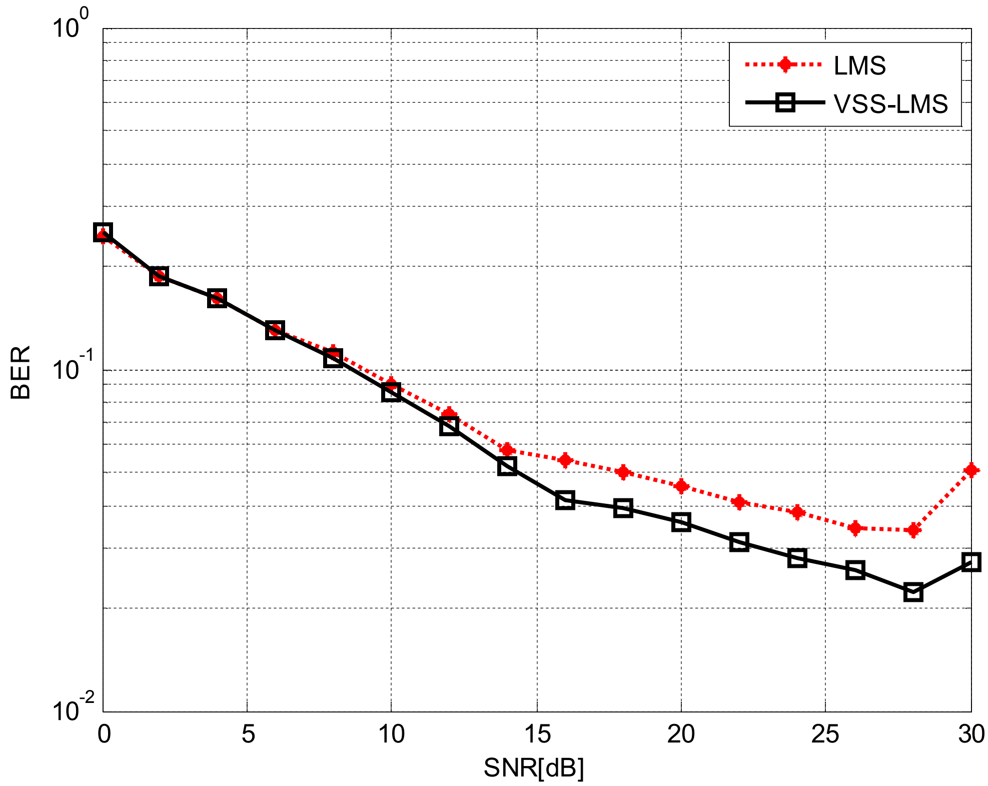

Figure 4.6: BER of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=50Hz, QPSK).

Figure 4.6 shows the BER as a function of SNR for the LMS and VSS-LMS algorithms when using QPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=50Hz. As it is clear, at SNR less than 10 dB the BER for both of the algorithms are almost the same. For higher SNR (greater than 10 dB) the BER for VSS-LMS is better than LMS algorithm, the higher the SNR the better BER of VSS-LMS compared to LMS one.

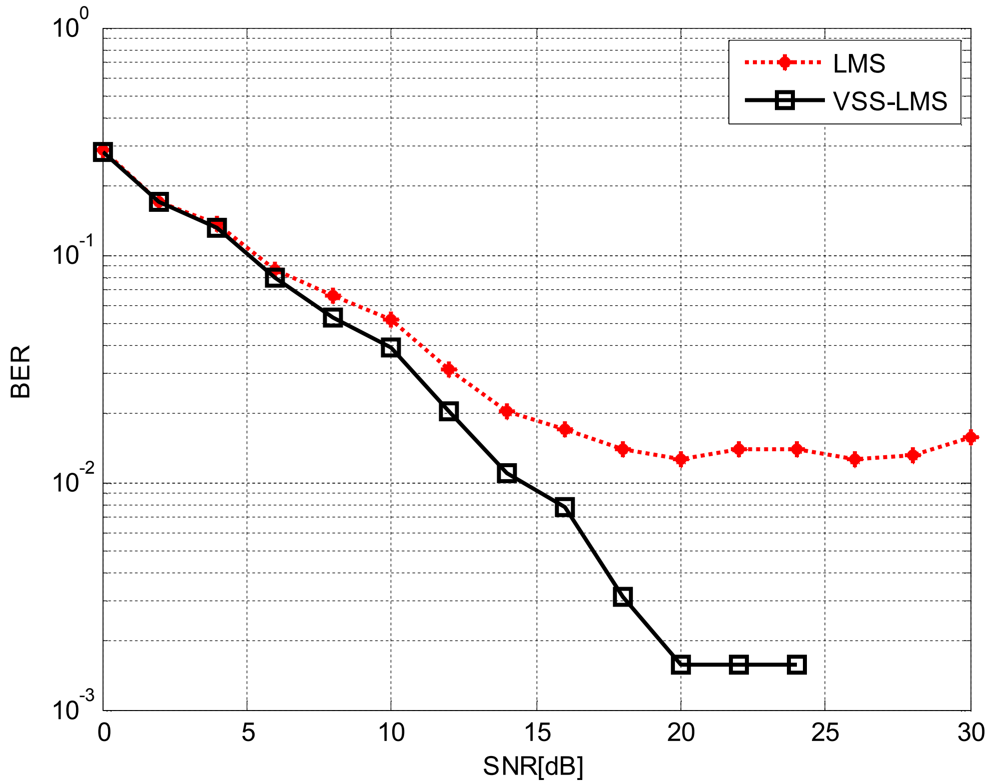

Figure 4.7: BER of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=500Hz, BPSK).

Figure 4.7 shows the BER as a function of SNR for the LMS and VSS-LMS algorithms when using BPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=500Hz. In this case the BER is greater for both algorithms if it is compared to the previous ones; however, the difference between the performances of VSS-LMS and LMS is more obvious especially at SNR greater than 10dB. At SNR greater than 24dB the BER of VSS-LMS is equal to zero while BER of LMS doesn't reach 0.01 till the end of simulation(SNR=30dB).

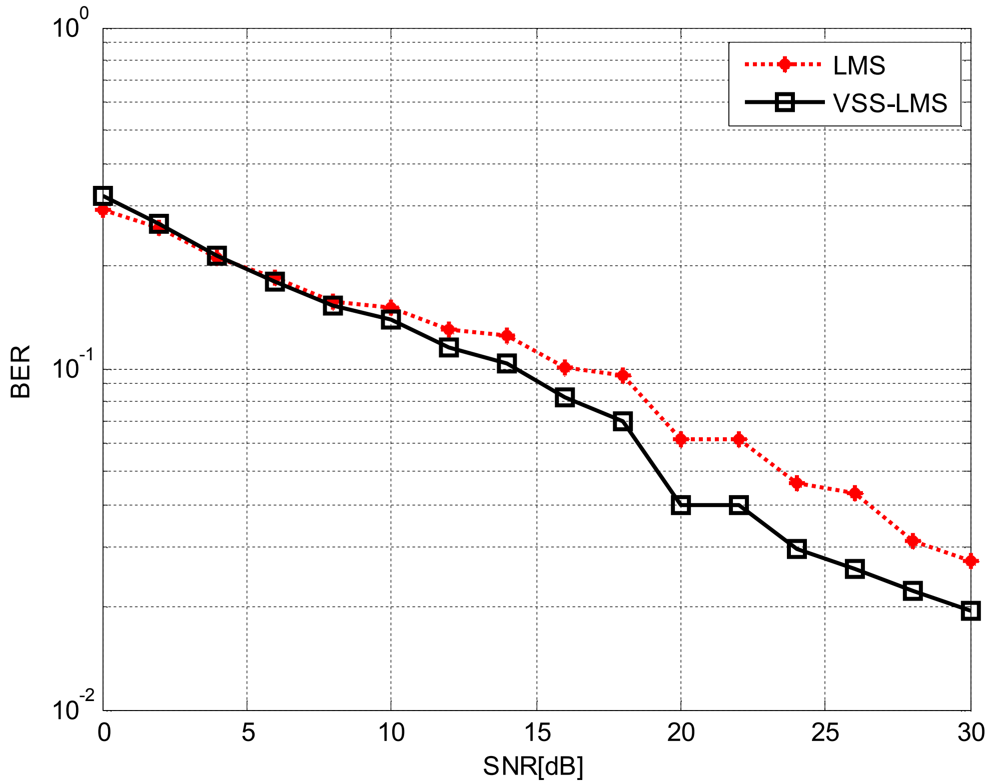

Figure 4.8: BER of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=500Hz, QPSK).

Figure 4.8 shows the BER as a function of SNR for the LMS and VSS-LMS algorithms when using QPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=500Hz. As it is clear, at SNR less than 8 dB the BER for both of the algorithms are almost the same. For higher SNR (greater than 8 dB) the BER for VSS-LMS is better than LMS algorithm, the higher the SNR the better BER of VSS-LMS compared to LMS one.

4.3 Mean Square Error (MSE) Analysis

Mean Square Error (MSE) is an important parameter for performance evaluation. In this section, the MSE of LMS and VSS-LMS channel estimation algorithms is evaluated versus SNR. The system was experienced under the same eight cases of BER ones.

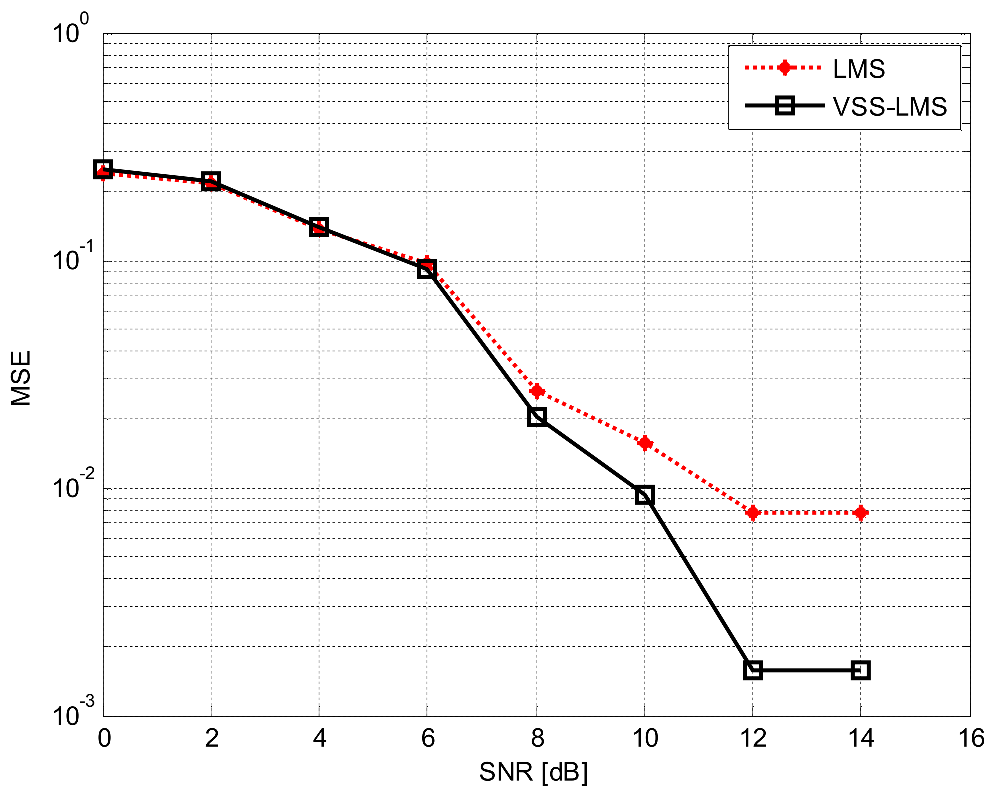

Figure 4.9: MSE of LMS and VSS-LMS algorithms as a function of SNR (AWGN, BPSK).

Figure 4.9 shows the MSE as a function of SNR for the LMS and VSS-LMS algorithms when using BPSK as the modulation technique and passing the signal through AWGN before it reached the receiver. As it is noticeable, at low SNR(less than 6 dB) the BER for both of the algorithms are almost the same. For higher SNR (greater than 6 dB) the MSE for VSS-LMS is better than LMS algorithm, the higher the SNR the better MSE for the VSS-LMS as compared to LMS one. In addition to that at SNR greater than 14 dB both of algorithms has MSE equal to zero.

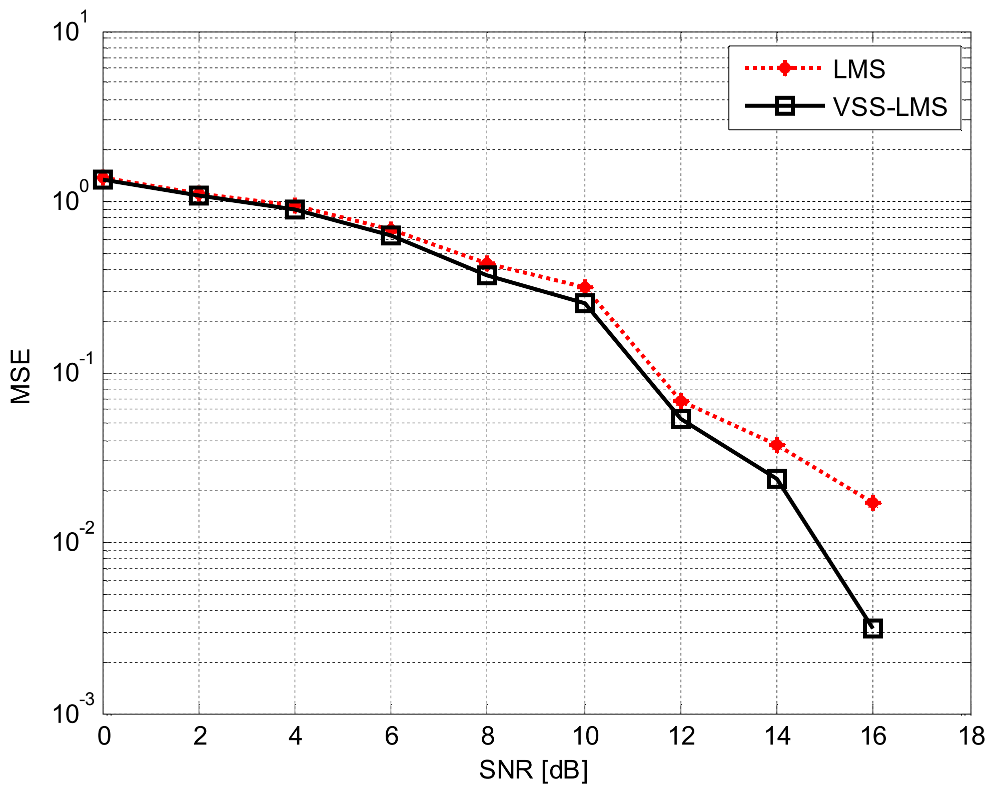

Figure 4.10: MSE of LMS and VSS-LMS algorithms as a function of SNR (AWGN, QPSK).

Figure 4.10 shows the MSE as a function of SNR for the LMS and VSS-LMS algorithms when using QPSK as the modulation technique and passing the signal through AWGN before it reached the receiver. As it is noticeable, at low SNR(less than 6 dB) the MSE for both of the algorithms are almost the same. For higher SNR (greater than 6 dB) the MSE for VSS-LMS is better than LMS algorithm, the higher the SNR the better MSE for the VSS-LMS as compared to LMS one. In addition to that at SNR greater than 16 dB both of algorithms has MSE equal to zero.

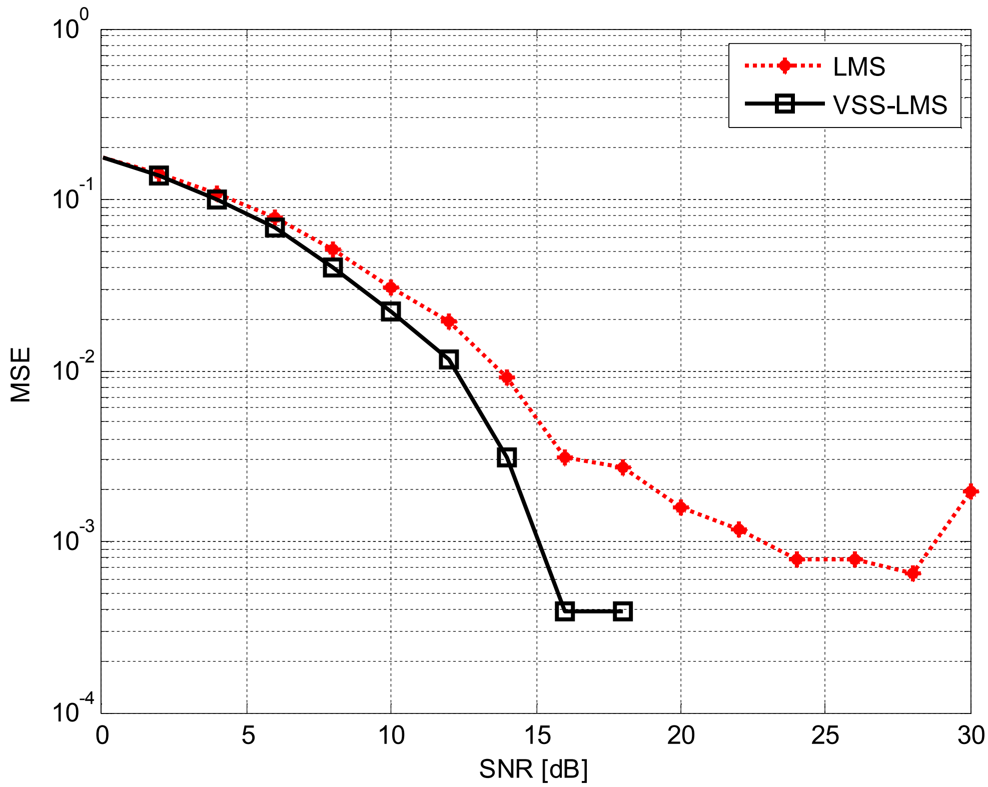

Figure 4.11: MSE of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=5Hz, BPSK).

Figure 4.11 shows the MSE as a function of SNR for the LMS and VSS-LMS algorithms when using BPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=5Hz. As it is clear, at low SNR(less than 6 dB) the MSE for both of the algorithms are almost the same. For higher SNR (greater than 6 dB) the MSE for VSS-LMS is better than LMS algorithm. In SNR greater than 12 dB VSS-LMS has MSE equal to zero, while the MSE of LMS equal to zero for SNR greater than 16 dB.

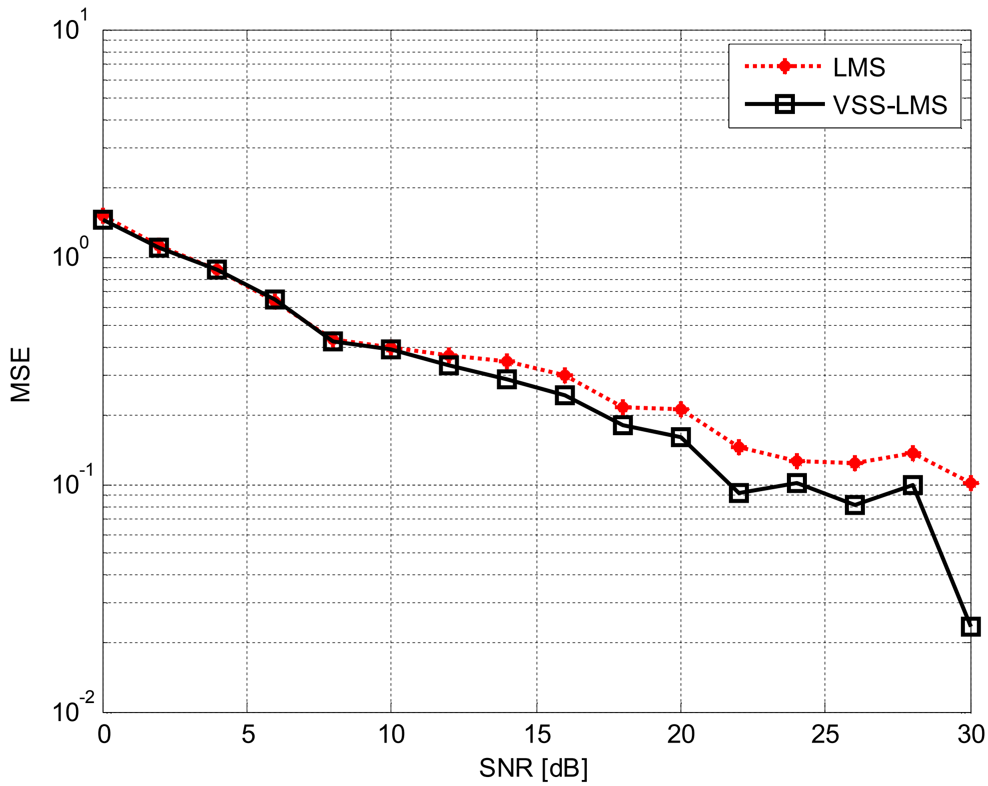

Figure 4.12: MSE of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=5Hz, QPSK).

Figure 4.12 shows the MSE as a function of SNR for the LMS and VSS-LMS algorithms when using QPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=5Hz. As it is clear, at low SNR(less than 10 dB) the MSE for both of the algorithms are almost the same. For higher SNR (greater than 10 dB) the MSE for VSS-LMS is better than LMS algorithm.

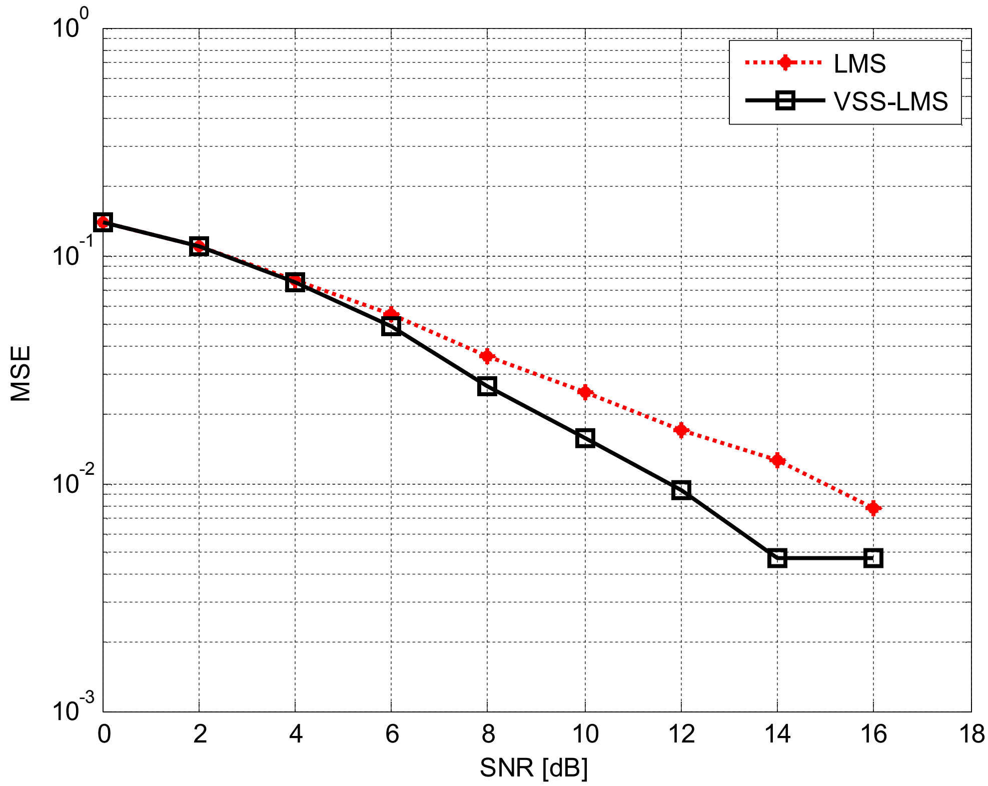

Figure 4.13: MSE of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=50Hz, BPSK).

Figure 4.13 shows the MSE as a function of SNR for the LMS and VSS-LMS algorithms when using BPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=50Hz. In this figure VSS-LMS is better than LMS algorithm for SNR greater than 6 dB. In SNR greater than 16 dB both of algorithms has BER equal to zero.

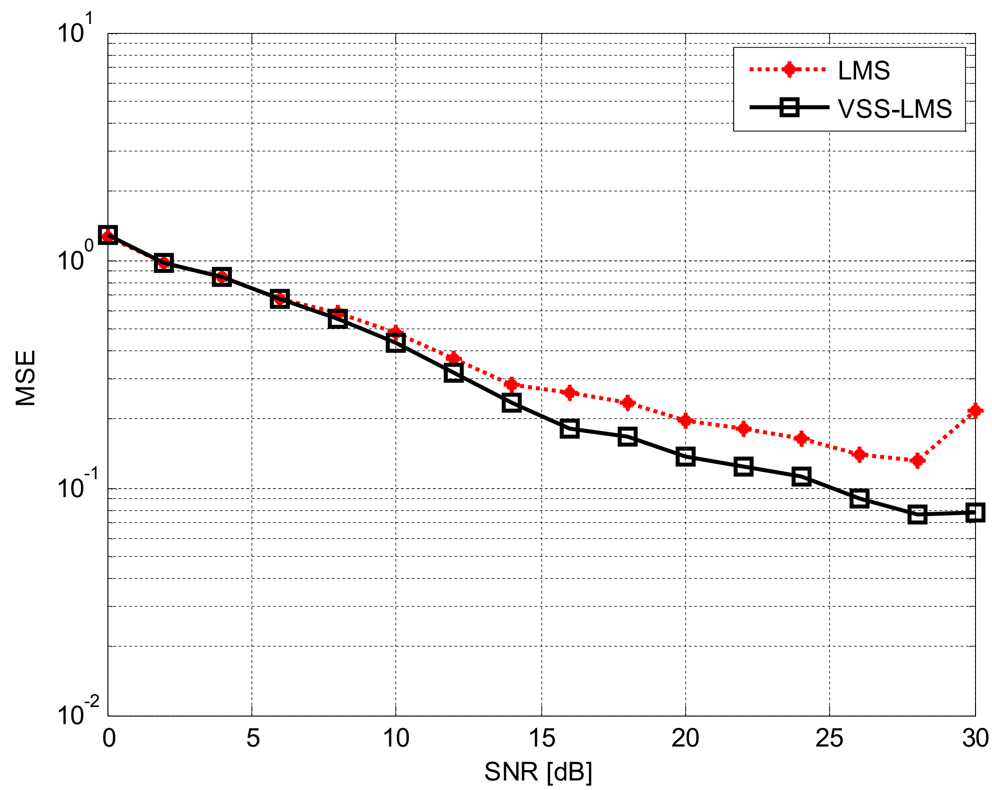

Figure 4.14: MSE of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=50Hz, QPSK).

Figure 4.14 shows the MSE as a function of SNR for the LMS and VSS-LMS algorithms when using QPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=50Hz. As it is clear, at SNR less than 10 dB the MSE for both of the algorithms are almost the same. For higher SNR (greater than 10 dB) the MSE for VSS-LMS is better than LMS algorithm, the higher the SNR the better MSE of VSS-LMS compared to LMS one.

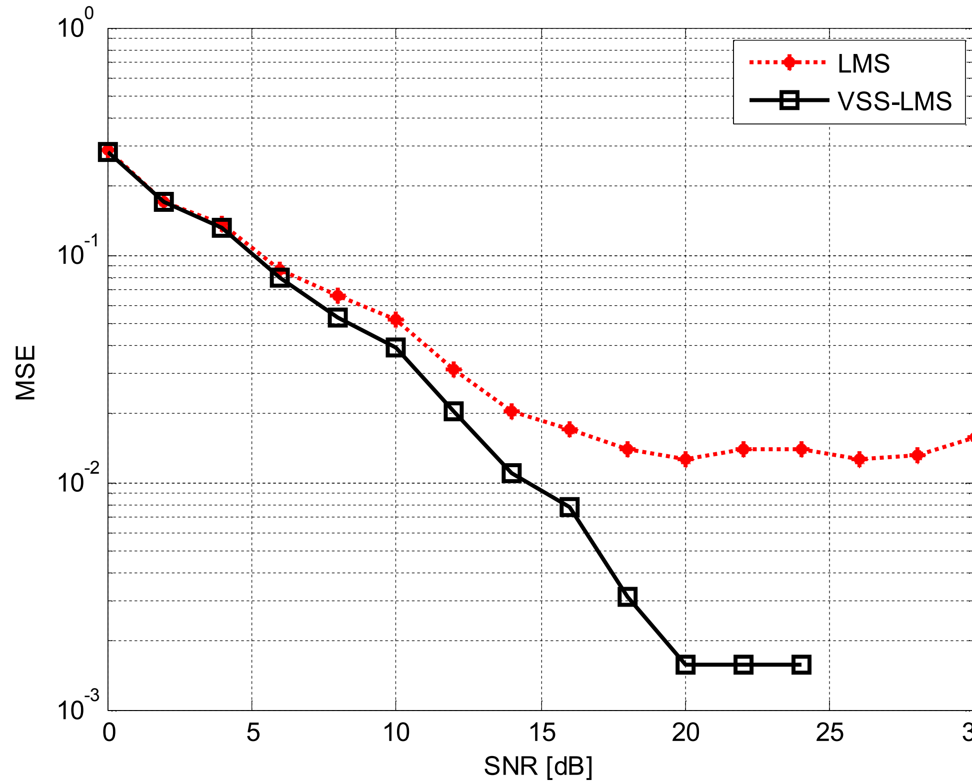

Figure 4.15: MSE of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=500Hz, BPSK).

Figure 4.15 shows the MSE as a function of SNR for the LMS and VSS-LMS algorithms when using BPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=500Hz. In this case the MSE is greater for both algorithms if it is compared to the previous ones; however, the difference between the performances of VSS-LMS and LMS is more obvious especially at SNR greater than 10dB. At SNR greater than 24dB the MSE of VSS-LMS is equal to zero while MSE of LMS doesn't reach 0.01 till the end of simulation(SNR=30dB).

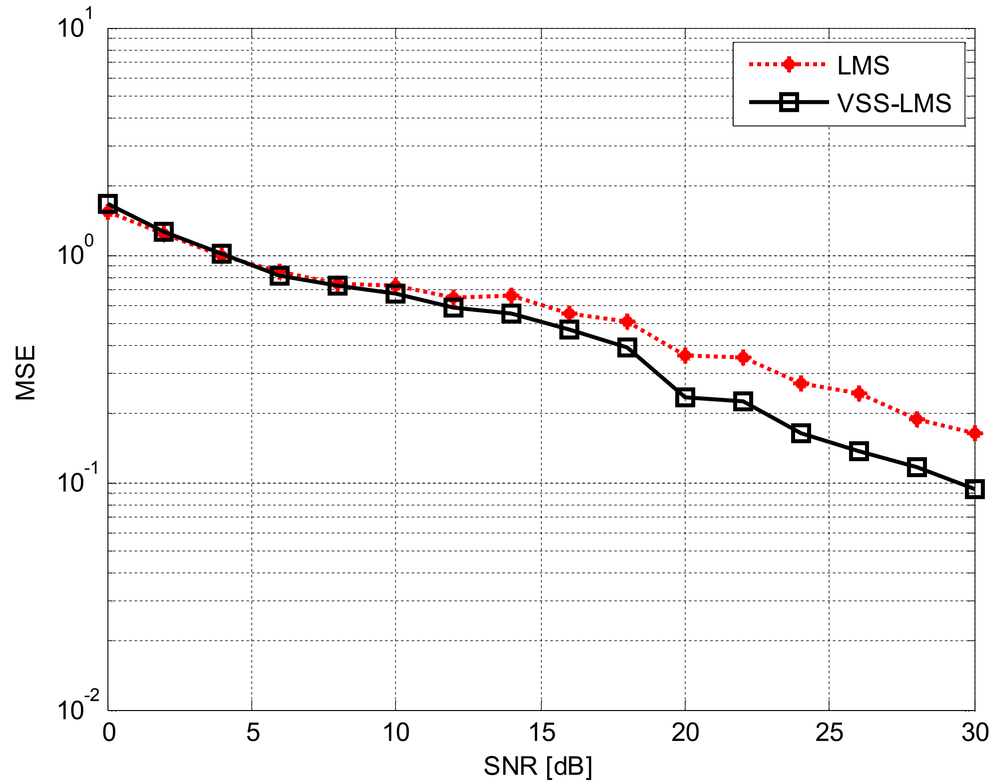

Figure 4.16: MSE of LMS and VSS-LMS algorithms as a function of SNR (Rayleigh channel with Doppler shift=500Hz, QPSK).

Figure 4.16 shows the MSE as a function of SNR for the LMS and VSS-LMS algorithms when using QPSK as the modulation technique and Rayleigh fading channel as a channel model with maximum Doppler shift=500Hz. As it is clear, at SNR less than 8 dB the MSE for both of the algorithms are almost the same. For higher SNR (greater than 8 dB) the MSE for VSS-LMS is better than LMS algorithm, the higher the SNR the better MSE of VSS-LMS compared to LMS one.

4.4 Complexity analysis

One of the vital analysis parameter is the complexity of the algorithm; since it affects the delay directly. Complexity is measured by the number of multiplications and number of additions.

LMS algorithm requires 2N+1 multiplication operations, N multiplications for the weights update and N+1 for error calculations, and 2N addition operations, N subtraction operations for error estimation and N addition operations for weights update. See equation 3.3 and 3.4.

VSS-LMS algorithm requires 3N+6 multiplication operations and 2N+2 addition operations. In addition to the operations used in LMS algorithm it requires N+5 multiplication operations and 2 addition operations, See equation 3.5 and 3.6.

The complexity analysis is shown in figure 4.17. This figure shows that the LMS requires as average 4.0711ms and VSS-LMS requires 195.6453ms to estimate the channel response under the specifications documented in table 3.1.

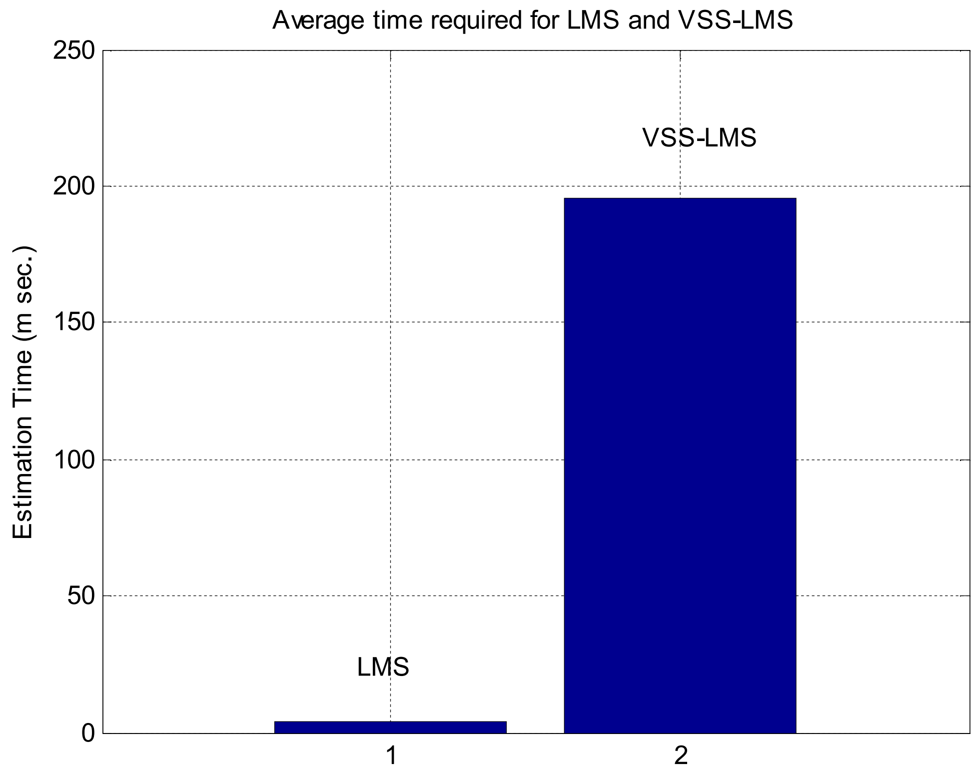

Figure 4.17: Time estimation of LMS and VSS-LMS algorithms.

Chapter Five

Conclusion and Recommendation

Chapter Five

Conclusion and recommendation

5.1. Conclusion

Channel estimation of SC-FDMA under LTE umbrella is analyzed for different modulation techniques and channel environments by modeling the system and then applying LMS and VSS-LMS channel estimation techniques using MATLAB. According to results shown in chapter four, the modulation type affects the performance of channel estimation techniques; however, channel environment has an obvious effect.

The performance degraded when using QPSK modulation compared with BPSK for both algorithms. However this degradation increased clearly under high Doppler shift. The higher the Doppler shift the worse performance of both algorithms, however VSS-LMS is less affected by Doppler shift compared to LMS.

With regard to algorithm complexity, VSS-LMS requires just two additional addition operations compared to LMS. But it requires N+5 multiplications. As it is known, multiplication operation requires more time to be accomplished (in simulation VSS-LMS algorithm elapsed approximately about 50 times the time elapsed by LMS algorithm).

In conclusion, VSS-LMS algorithm has better performance than LMS in all cases; however it requires more multiplication and addition operations.

5.2. Recommendation

Since VSS-LMS has better performance than LMS algorithm but require more operation, reducing the number of operations by modifying the cost term of the gain factor equation may be gives better results. So more studying in this area is required. Further enhancement in term of BER and MSE can be obtained by using closed loop power control (CLPC) or open loop power control (OLPC).

Single input single output (SISO) was used in this work, however, if a diversity combining technique is used, the performance will be enhanced much more since it will reduce the effect of the fading. Diversity combining techniques are needed to be investigated in the future work.

Publications

Based on this thesis the following papers have been submitted to publish:

- N. Elshaikh, M. Hussein, “Performance Evaluation of LMS and VSS-LMS Channel Estimation Techniques for LTE Uplink under different modulation types”, International Journal of Multidisciplinary and Current Research (IJMCR), (Accepted to be published on April/June 2014 Issue).
- M. Hussein, N. Elshaikh, “Performance Evaluation of LMS and VSS-LMS Channel Estimation Techniques for LTE Uplink under different Doppler shifts”, International Journal of Multidisciplinary and Current Research (IJMCR), (Accepted to be published on April/June 2014 Issue).

References

[1] http://www.etsi.org, "Work Plan 3GPP (Release 8)", 1 March 2012.

[2] H. Ekström, A. Furuskär, J. Karlsson, M. Meyer, S. Parkvall, J. Torsner, and M. Wahlqvist, “Technical Solutions for the 3G Long-Term Evolution,” IEEE Commun. Mag., vol. 44, no. 3, pp. 38–45, 2006.

[3] PARK, S. Y., KIM, Y. G., KANG, C. G. “Iterative receiver for joint detection and channel estimation in OFDM systems under mobile radio channels”. IEEE Trans. on Vehicular Technology, vol. 53, no. 2, p. 450 – 460, 2004.

[4] HSIEH, M. H., WE, C. H. “Channel estimation for OFDM systems based on comb-type pilot arrangement in frequency selective fading channels”. IEEE Trans. on Consumer Electronics, vol. 44, no. 1, p. 217 – 225, 2004.

[5] DOUKOPOULOS, X. G., MOUSTAKIDES, G. V. “Blind adaptive channel estimation in OFDM systems”, IEEE Trans. On Wireless Communication, vol. 5, no. 7, pp. 1716 – 1725, 2006.

[6] ADIREDDY, S., TONG, L., VISWANATHAN, H. “Optimal placement of training for frequency selective block fading channels. IEEE Trans. on Information Theory”, vol. 48, no. 8, p. 2338 – 2353, 2002.

[7] R.W. Chang and R.A. Gibby, “Theoretical Study of Performance of an Orthogonal Multiplexing Data Transmission Scheme”, IEEE Transactions on Communications, 16, 4, pp. 529-540, 1968.

[8] L. J. Cimini, Jr., “Analysis and Simulation of a Digital Mobile Channel Using Orthogonal Frequency Division Multipexing,” IEEE Trans. Commun., vol. 33, no. 7, July 1985, pp. 665–75.

[9] David Falconer, S. Lek Ariyavisitakul, Anader Benyamin-Seeyar, Brian Eidson, “Frequency Domain Equalization for Single-Carrier Broadband Wireless Systems", Communications Magazine, IEEE, Apr. 2002.

[10] Jin Xinzhu, “Channel Estimation Techniques of SC-FDMA”, Karlstad University, 2007.

[11] J. Li, Y. Du, and Y. Liu, "Comparison of Spectral Efficiency for OFDM and SC-FDE under IEEE 802.16 Scenario," Proceedings of the 11th IEEE Symposium on Computers and Communications (ISCC'06), 2006.

[12] 3rd Generation Partnership Project (3GPP), “Requirements for Evolved UTRA (EUTRA) and Evolved UTRAN (E-UTRAN)”, http://www.3gpp.org/ftp/Specs/htmlinfo/25913.htm.

[13] 3rd Generation Partnership Project (3GPP); Technical Specification Group Radio Access Network; Physical Layer Aspects for Evolved UTRA, http://www.3gpp.org/ftp/Specs/html-info/25814.htm.

[14] DIVYA VIJAYAN, “ADAPTIVE CHANNEL ESTIMATION FOR LTE UPLINK”, National Institute of Technology, Rourkela, 2013.

[15] Agilent Technologies, “Uplink resource block”, Agilent Technologies Inc, USA, 2007.

[16] Rohde and Schwarz, “UMTS Long Term Evolution (LTE) Technology Introduction”.

[17] Agilent Technologies, “Agilent Technologies Solutions for 3GPP LTE”, Agilent Technologies Inc., USA, 2007.

[18] Rafal Surgiewicz, Niklas Str¨om, Anser Ahmed, Yun Ai, “LTE Uplink Transmission Scheme”, 2011.

[19] Hyung G. Myung, Junsung Lim, and David J. Goodman, “Single Carrier FDMA for Uplink Wireless Transmission”, IEEE VEHICULAR TECHNOLOGY MAGAZINE, SEPTEMBER 2006.

[20] Raina Rahman, “CPM-SC-IFDMA|A Power Efficient Transmission Scheme for Uplink LTE”, University of Kansas, 2010.

[21] Yibing Li, Xu Zhang, Fang Ye, ” Improved Dynamic Subcarrier Allocation Scheduling for SC-FDMA Systems”, Journal of Information & Computational Science 9: 12, 2012.

[22] Daniel Larsson, “Analysis of channel estimation methods for OFDMA”, XR-EE-KT 2006:011, Royal Institute of Technology, 2006.

[23] Mohsin Niaz Ahmed, “LTE UPLINK MODELING AND CHANNEL ESTIMATION”, Linköping University , 2011.

[24] T. Rappaport, Wireless Communications, Principles and Practice, Prentice-Hall, Englewood Cliffs, NJ, USA, 1996.

[25] Mohamed Ibnkahla, Ed, Signal Processing for Mobile Communications, CRC Press, New York Washington, D.C., 2005.

[26] Tero Ojanpera and Ramjee Prasad, WCDMA: Towards IP Mobility and Mobile Internet, Artech House Publishers, Boston, London, 2001.

[27] Asad Mehmood and Waqas Aslam Cheema, “CHANNEL ESTIMATION FOR LTE DOWNLINK”, Blekinge Institute of Technology, 2009.

[28] S. B. Weinstein and P. M. Ebert, “Data Transmission by Frequency Division Multiplexing 634, Using Discrete Fourier Transform,” IEEE Transactions on Communications, vol. 19, no. 5, pp. 628–October 1971.

[29] James K. Cavers, Mobile Channel Characteristics, Kluwer Academic Publishers, New York, Boston, Dordrecht, London, Moscow, 2002.

[30] J. Meinilä, T. Jämsä, P. Kyösti, D. Laselva, H. El-Sallabi, J. Salo, C. Schneider, D. Baum, 'IST-2003- 507581 WINNER: Determination of Propagation Scenarios', D5.2 v1.0, 2004.

[31] Harri Holma and Antti Toskala, Ed, LTE for UMTS: OFDMA and SC-FDMA Base Band Radio Access, John Wiley & ISBN 9780470994016 (H/B) John Wiley & Sons Ltd, 2009.

[32] ITU-R M.1225 International Telecommunication Union, 'Guidelines for evaluation of radio transmission technologies for IMT-2000', 1997.

[33] Ericsson, Nokia, Motorola, and Rohde & Schwarz, 'R4-070572: Proposal for LTE Channel Models', www. gpp.org, 3GPP TSG RAN WG4, meeting 43, Kobe, Japan, May 2007.

[34] A. ncora, C. Bona, and D.T.M. Slock, "Down-sampled impulse response least-squares channel estimation for LTE OFDMA," *Proc. Int. Con. ASSP*, vol. 3, pp. 293-296, April 2007.

[35] L. A. M. R. D. Temino, C. N. I Manchon, C. Rom, T. B. Sorensen, and P. Mogensen, "Iterative channel estimation with robust wiener filtering in LTE downlink," Proc. Int. Con. on VTC, pp. 1-5, September 2008.

[36] Yongkui Ma, Ping Tan, Xiaoyu Wang, Dianwei Li," Research on Channel Estimation Based on Adaptive Filtering for LTE Uplink", IEEE,2012.

[37] Md. Masud Rana, Jinsang Kim, and Won-Kyung Cho," An Adaptive LMS Channel Estimation Method for LTE SC-FDMA Systems", International Journal of Engineering & Technology IJET-IJENS Vol: 10 No: 05.

[38] Md. Masud RANA, Jinsang KIM, Won-Kyung CHO," LMS Based Adaptive Channel Estimation for LTE Uplink", RADIOENGINEERING, VOL. 19, NO. 4, DECEMBER 2010.

[39] B. Widrow and M. Hoff, “Adaptive switching circuits,” in Proc. IRE Western Electronic Show and Convention, New York, USA, Part 4, Aug. 1960, pp. 96-104.

[40] S. Haykin, Adaptive Filter Theory, 4th edition, Upper Saddle River, NJ: Prentice Hall, 2002.

[41] José Gil F. Zipf, Orlando J. Tobias, and Rui Seara, “A VSSLMS ALGORITHM BASED ON ERROR AUTOCORRELATION”, 16th European Signal Processing Conference (EUSIPCO 2008), Lausanne, Switzerland, August 25-29, 2008.

Appendix

MATLAB Code

LMS Algorithm

```
wk=zeros(N,1);
yrxk=Y_DEMAPPING(:,2);
uk=zadoffchu_seq_rx;
ek=yrxk-uk*wk;
EEE=300;
c=0;
tic;
while abs(EEE.^2)>0 && c<c_thr
ek=yrxk-uk*wk;
e1=[e1 mean(ifft(ek))];
wk=wk+mue*uk'.*conj(ek);
c=c+1
EEE=real(mean(ifft(ek)))
end
estimation_time(sim,i_alg)= toc;
count(sim,i_alg)=c;
```

VSS-LMS Algorithm

```
e2=[];
mue2=[];
mue2=[mue2 mue22];
p=0;
wk=zeros(N,1);
yrxk=Y_DEMAPPING(:,2);
uk=zadoffchu_seq_rx;
ek=yrxk-uk*wk;
c=0;
tic;
while abs(EEE.^2)>0 && c<c_thr
ek_1=ek;
ek=yrxk-uk*wk;
e2=[e2 mean(ifft(ek))];
wk=wk+mue22*uk'.*conj(ek);
c=c+1
EEE=real(mean(ifft(ek)))
p=bet1*p+(1-bet1).*ek'*ek_1;
mue22=alp1*mue22+gam1.*p.^2;
mue22=mean(abs(mue22));
if mue22>mue_max
    mue22=mue_max;
elseif mue22<mue_min
    mue22=mue_min;
end
```

```
mue2=[mue2 mue22]
end
count(sim,i_alg)=c;
estimation_time(sim,i_alg)= toc;
```

LTE Channel

```
chan = rayleighchan(ts,fd,delay_EPA,avgpower_EPA);
ych = filter(chan,ytx);
%% Add white gaussian noise
for sim=1:numel(snr)
ynoisy = awgn(ych,snr(sim),'measured');
end
```