Design of Channel Estimation and Equalization for OFDM Systems

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by
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Ali Salah MAHDI

ABSTRACT

DESIGN OF CHANNEL ESTIMATION AND EQUALIZATION FOR OFDM SYSTEMS

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M.Sc. in Electrical and Electronics Engineering

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Orthogonal Frequency Division Multiplexing (OFDM) system is one of the multicarrier techniques which is robust against Inter-symbol-Interference, multipath fading and very easy to apply in transmitters by using inverse fast Fourier transform IFFT and at the receivers by using fast Fourier transform FFT. In a communication system, channel estimation is very important issue for the data detection. In coherent detection, one of the popular techniques is to use pilot tones as a reference signal in OFDM symbols. In the comb-type pilot tones insertion, pilot tones are inserted into each OFDM symbols, but inserting a large number of pilot tones will lead to channel capacity reduction or bandwidth expansion [1-2]. In this thesis, to overcome this transmission loss, a modified least square (ModLS) algorithm for fast time varying wireless channel at comb-type pilot arrangement in QAM signals for OFDM system is proposed. The proposed algorithm has reduced number of pilot tones in each OFDM symbol. 1/100 pilot ratio has been achieved. This ratio ranges from 1/16 to 1/18 in other studies. Also, we have improved bit error rate (BER) in low signal to noise ratio (SNR) by the proposed algorithm. The simulation results obtained from the proposed algorithm showed a good performance in noisy wireless channels. In addition, it has been compared with least square (LS) algorithm in different signal to noise ratios and different channel tabs. The comparison results showed that the proposed algorithm was superior to least square algorithm.

Key words: OFDM system, Rayleigh fading channel, Least Square algorithm, Modified Least Square algorithm.

ÖZET

DFBÇ SİSTEMLERİ İÇİN KANAL KESTİRME VE DENKLEŞTİRME TASARIMI

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Yüksek Lisans Tezi, Elektrik ve Elektronik Mühendisliği Bölümü

Tez Yöneticisi: Prof. Dr. Ergun ERÇELEBİ

Mart 2014, 64 sayfa

Dikey frekans bölmeli çoğullama (DFBÇ) sistemi hızlı Fourier dönüşümü kullanarak alıcılarda ve ters hızlı Fourier dönüşümü kullanarak vericilerde çok kolay uygulanan ve semboller arası girişime, çok yollu sönümlemeye karşı dayanıklı çok taşıyıcılı tekniklerden birisidir. İletişim sistemlerinde, veri tespiti için kanal kestirimi çok önemli bir konudur. Evre uyumlu algılamada, popüler tekniklerden biri pilot tonları DFBÇ sembollerinde referans sinyalleri gibi kullanmaktadır. Tarak tipi pilot ton eklemede, pilot tonlar her bir OFDM sembollerine yerleştirilir, ancak çok sayıda pilot tonlarının eklenmesi kanal kapasitesinin azaltılmasına veya bant genişlemesine yol açacaktır [1-2]. Bu tezde; bu iletim kaybının üstesinden gelmek için, OFDM sistemi için QAM sinyallerinde, tarak tipi pilot diziliminde hızlı zamanla değişen kablosuz ağlar için iyileştirilmiş en küçük kareler (ModLS) algoritması önerilmiştir. Önerilen algoritma, her DFBÇ sembollerinde pilot tonlarının sayısını azalttı. 1/100, pilot oranı elde edilmiştir. Bu oran, diğer çalışmalarda 1/16 dan 1/18'e kadar değişmektedir. Ayrıca, önerilen algoritma ile düşük sinyal-gürültü oranlarında bit hata oranını iyileştirdik. Önerilen algoritmadan elde edilen simülasyon sonuçları gürültülü kablosuz kanallarda iyi performans gösterdi. İlaveten, önerilen algoritma farklı sinyal gürültü oranlarında en düşük kareler yöntemi ile karşılaştırıldı. Buna ek olarak, algoritmanın performansının değerlendirilmesi için, bu gürültü oranı için farklı sinyal ile en küçük kareler (LS) algoritması ile karşılaştırılır. Karşılaştırma sonuçları önerilen algoritmanın en az kare algoritması karşı üstün olduğunu gösterdi.

Anahtar Kelimeler: DFBÇ sistem, Rayleigh sönümlü kanal, en düşük kareler algoritması, iyileştirilmiş en düşük kareler algoritması.

DEDICATION

To my parents,

my supervisor, and

my brothers.

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At the outset I thank ALLAH to complete my Master's Degree. I would like to thanks and express my gratitude to my supervisor Prof. Dr. Ergun Erçelebi for his support, idea, and guidance during the period of my MSc Program.

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LIST OF ABBREVIATIONS

OFDM	Orthogonal frequency division multiplexing
FDM	Frequency division multiplexing
TDMA	Time division multiple access
ISI	Inter symbol interference
ICI	Inter carrier interference
AWGN	Additive white Gaussian noise
BER	Bit Error Rate
SNR	Signal to Noise Ratio
CP	Cyclic Prefix
GI	Guard Interval
QAM	Quadrature Amplitude Modulation
PSK	Phase shift keying
DVB	Digital video broadcasting
HDTV	High definition television
MB-UWB	Multi-band-ultra wide band
DFT	Discrete Fourier Transform
FFT	Fast Fourier Transform
MMSE	Minimum mean-squared error
LMMSE	linear minimum mean-squared error
LS	Least square
ModLS or MLS	Modified least square
IDFT	Inverse Discrete Fourier Transform
IFFT	Inverse Fast Fourier Transform

CHAPTER 1

INTRODUCTION

1.1 General OFDM Introduction

Orthogonal Frequency Division Multiplexing (OFDM) created to get an efficient and well communications through frequency-selective fading channels. If the frequency response of the channel changes dramatically during transmission of the signal that means the channel is Frequency-selective [1]. Whereas, flat fading channel is a constant frequency response. Figures 1.1 demonstrate the frequency-selective channel and the flat fading channel as shown in figure 1.2.

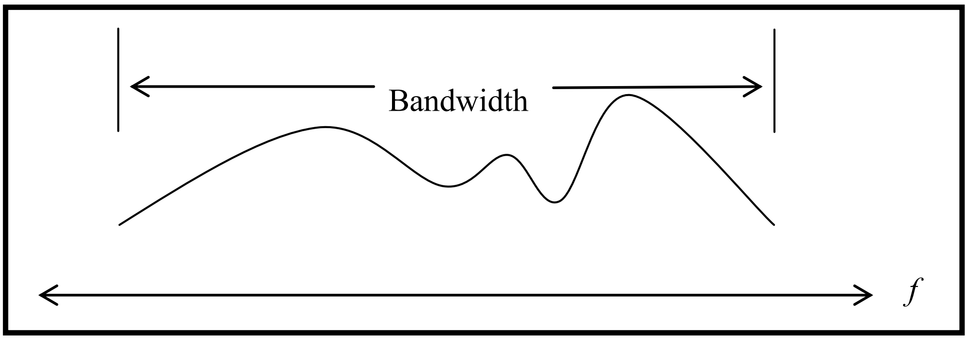

Figure 1.1 Frequency-Selective Fading Channel

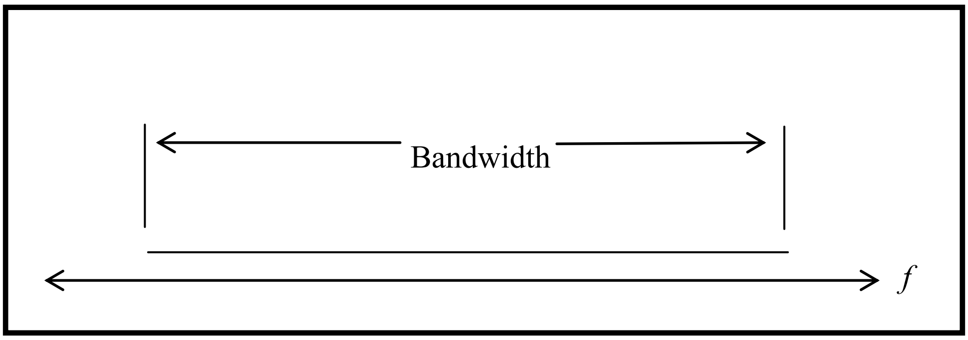

Figure 1.2 Flat Fading Channel Response

The transmitted signals can be distorted while passing through a frequency-selective channel, which led to the inter-symbol-interference (ISI). To alleviate the ISI, a guard insertion technique is used before with length must be greater than channel delay, so that the ISI-affected part of a symbol can be negligible at the receiver or complex equalizer [2] is usually used to make the frequency response of the channel flat within wanted bandwidth(during transmission). From the viewpoint of frequency-domain, the second approach (flat fading) means to transmit a narrow-band signal inside whose bandwidth the channel can be well considered to be flat fading, as shown in Figure 1.3.

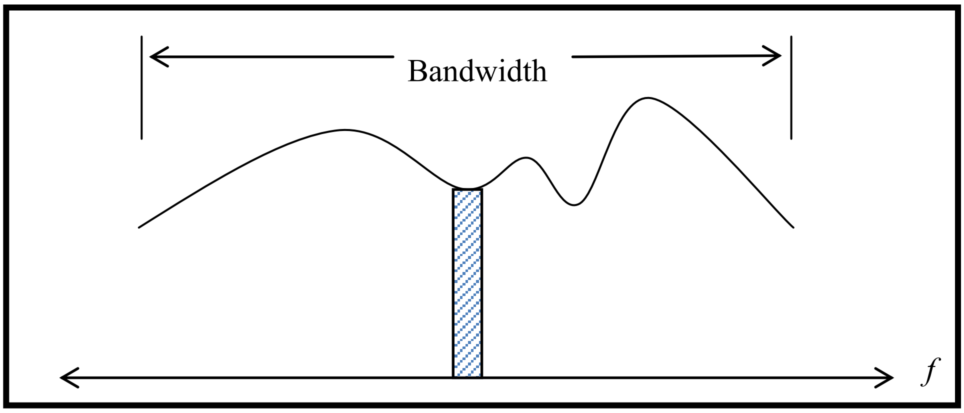

Figure 1.3 Narrow Band Signal Response

This point gives the idea that one can divided the total available bandwidth to several non-overlapping frequency sub-bands, each of which at a different carrier frequency to carry a distinct signal in parallel, and each band is ISI-free and then only one-tap simple equalizer will be used to compensate the faded signal. This idea is exemplified below as in Figure 1.4.

That is truly the idea of Frequency Division Multiplexing (FDM). However, FDM, multi-carrier transmission scheme may suffer from inter-carrier interference (ICI), i.e., the signals of close carriers may interfere with each other. One way to avoid the ICI, guard bands are used in FDM technique to prevent overlapping between different sub-carriers but this procedure led to a loss of the spectrum. Later OFDM came to applying orthogonality among sub-carrier frequencies to overcome on the ICI effect without need to use guarding bands. Figure 1.5 shows the comparison between FDM and OFDM signals. As well as, the OFDM is very similar to multi-carrier modulation (FDM) strategy.

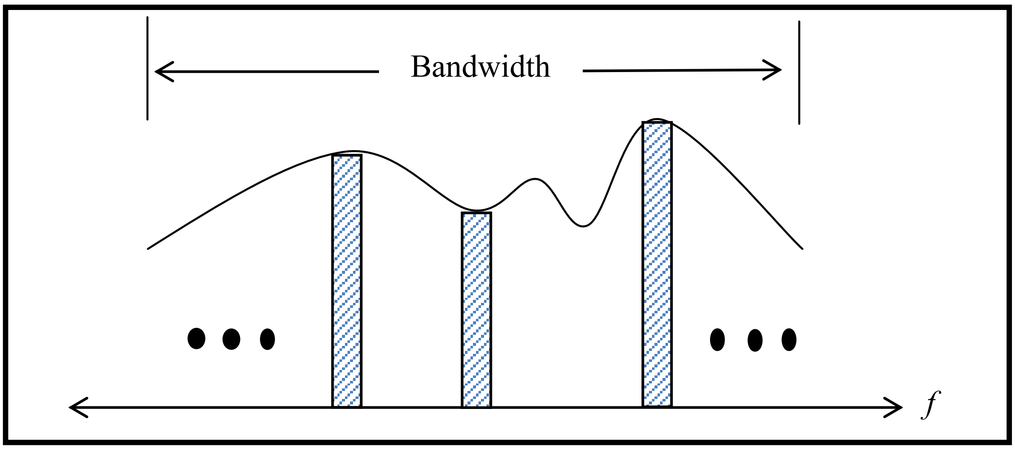

Figure 1.4 Multi-Carrier Modulation Schema

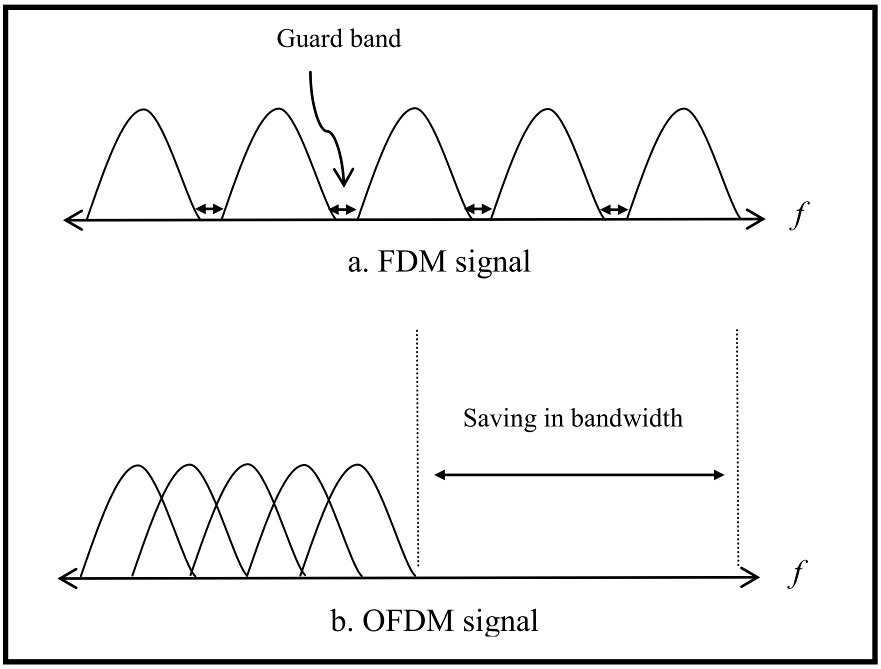

Figure 1.5 Comparison between FDM and OFDM

Nowadays, Orthogonal Frequency Division Multiplexing (OFDM) is one of the more widely used modulation technique for high speed data transmission over frequency selective channel. Mainly the wireless local area network (W-LAN) [4] systems such as IEEE 802.11a [5],IEEE 802.11g [6], (Wi-Fi), Wi-Bro, IEEE 802.16-2004 (Wi-

MAX) [7] and fourth-generation mobile systems [8] are used the OFDM schema as the core modulation technique also It was implemented as a standard for European digital video broadcasting (DVB-T) [9], in addition , it has been used in digital audio broadcasting (DAB) [10,11] and digital high-definition television (HDTV) broadcasting [12,13] standards. As well as OFDM is one of the solutions taken into account when implementing Multi-Band-Ultra Wide Band (MB-UWB) systems [14–17].

1.2 Motivation

The demand for future higher data rate communication systems always offers the impetus for this research. It is clear that a parallel system has ability to transfer more information than a cascade system, merely because it uses a variety of frequency channels. However, the significant advantage of OFDM is that it is robust in frequency-selective channels, which result from either multipath propagation or interference with other communication systems. The fast time varying channel and wide noise spectrum problems are severe especially for crowded areas. Under any of these terms, the channel has non-linear phase across frequencies, as well as non-uniform power gains. There is an example as shown in the figure 1.6.

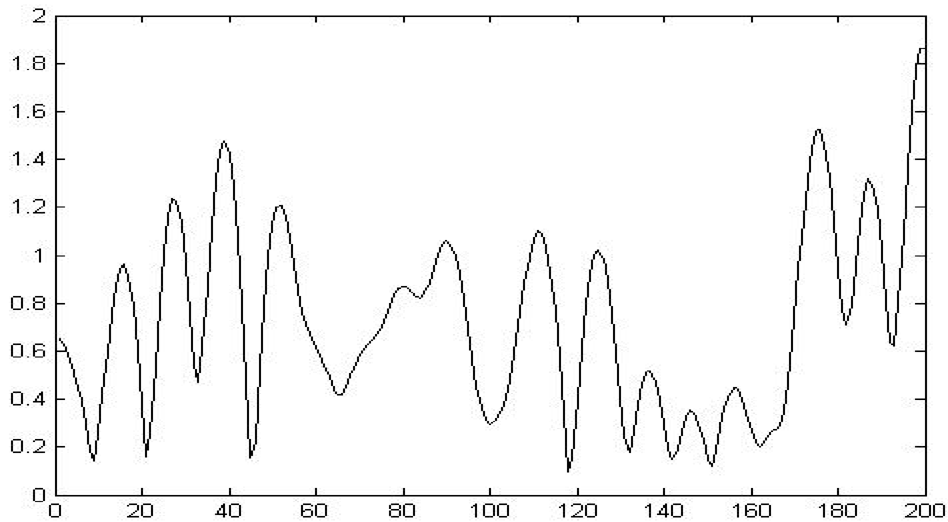

Figure 1.6 Magnitude Response of a Fading Channel

In the end, transmitted data is fully corrupted and the receiving side cannot recognize what is sent. There are many researchers have been proposed to solve these problems, but in the ways, either at the expense of reducing the data rate, or at the expense of bandwidth expansion such as in [18, 19, 20, 21, 22]. Our proposed work aim to:

a. Design an OFDM system work efficiently in the crowded city centers where there is no line of sight between the transmitter and receiver.
b. Increase data rate transmission without the need to expand bandwidth.
c. Improve estimation especially with low signal to noise ratio.

1.3 Literature survey

The early forms of multi-carrier modulation were in the date 1950 and early 1960s with military high frequency (HF) wireless links.

In the mid-60s, OFDM basic principle was presented by R. W. Chang [23]. The innovative feature of OFDM system is established by Chang related to traditional systems is that the sub-carriers are overlapped each other's. This property of OFDM systems led to needless to use the band-pass filters that were used in older multi-carrier modulation systems for the division of the spectra of the individual sub-carriers. After a short period Chang's paper, Saltzberg [24] examined the performance of OFDM and reached a significant conclusion that "the plan of designing a proficient parallel system should focus more on reducing crosstalk between sub-carriers than on perfecting the separate channels themselves, since the distortions because of crosstalk tend hegemonic".

In 1971, Weinstein and Ebert [25] presented the first suggestions to apply discrete Fourier transform (DFT) and inverse discrete Fourier transform (IDFT) to implement modulation and demodulation in OFDM baseband systems. This creativity made OFDM technique more practical. Presently, Fast Fourier Transform (FFT) and inverse FFT (IFFT) apply to implement modulation and demodulation of information data in OFDM systems.

In 1980, Peled and Ruiz [26] presented the concept of a cyclic prefix (CP) to combat the inter symbol interference that caused by effect of frequency-selective channel during transmission. Instead of using a null guard space in time among OFDM symbols, they occupied the guard space with a cyclic prefix (CP) extension of OFDM symbols, on condition, the length of cyclic prefix must be equal to or greater than maximum channel delay (channel impulse response) to prevent inter-symbol interference. The disadvantage of using a cyclic prefix is loss of signal energy proportional to the length of the CP. On the other hand, the profit from using a CP is generally greater than any loss of signal energy. All these developments make OFDM system is very widely used in many applications such as mentioned above in the OFDM introduction part.

In 1985, Cimini [27] has been applied the OFDM technique in the wireless cellular mobile communication system, and he has established the OFDM wireless mobile communication systems theory.

Channel estimation and equalization are an important issue in the design OFDM system. The main purpose of equalizer at the receiver is to calculate and compensate the effects of the channel [2]. This calculation and compensation are required at the receiving side. However, estimation of impulse response of the channel is obtainable. Frequently impulse response or frequency response of the channel is taken from pilot symbols or training sequence, on the other hand it is also possible to estimate the channel without use pilot tones or training sequence approach such as blind algorithms [28]. Channel estimation is one of the essential issues of designing an OFDM system, the performance of channel estimation accuracy will loss of nearly 3-4 dB if non coherent detection has been used rather than coherent detection [29]. If coherent OFDM system is implemented, the channel response estimation must be available and often pilot symbols are used to estimate the channel response.

In 1991, Aghamohammadi [30] et al. and Cavers [31] have presented a different type of interpolation filters for PSAM after the first studying and enhancing Pilot Symbol Assisted Modulation (PSAM). The main weakness of PSAM lies in the minor increase of bandwidth.

A superimposed pilot sequences is a totally new idea used in the channel estimation and proposed by numerous authors for many applications.

In1998 [18], Meng-Han Hsieh' and Che-Ho Wei have reduced computational complexity of pilot signal estimation based on minimum mean-squared error (MMSE) criterion by using a simplified linear minimum mean-squared error (LMMSE) estimator with low-rank approximation using singular value decomposition, the pilot ratio was 1/8.

In 1999 [32], superimposed pilot sequences took advantage in for time and frequency synchronization.

In 2000 [33], the main idea of superimposed pilot sequences is to add a known pilot tones to the transmitted signal with a linearly form and implement joint channel estimation and detection at the receiver.

In 2001, Julia [34], she has proposed an efficient approach for OFDM system in specific with symbol synchronization (frequency offsets correction), it is done easily by using pilot tones, also without any extra things added to the system. Commonest pilot tones arrangements that used in literature are block-type and comb-type [35], [36], but comb-type pattern is superior to block-type patterns in fast time varying channel [35].

In 2002, the block-type and comb-type pilot arrangements have examined by Coleri, S. [35]. Also there are several other techniques which have been proposed to estimate and calculate transfer function of the channel in OFDM systems, such as using correlation based estimators in the time domain and using singular value decomposition in the channel estimation [37].

In 2002 , Sinem Coleri and Ahmad Bahai [19] have been estimated impulse response of the channel depend on comb-type pilot arrangement by using Least square (LS) and Least Mean Square (LMS) algorithms, but the pilot ratio was 1/8.

In 2005, Farhad Tavassoli and Bahman Abolhassani [20] have investigated the effects of number of pilots as well as effects of channel Doppler frequency on the BER. Also pilot ratio was 1/16.

In 2008, Hala Mahmoud, Allam Mousa, Rashid Saleem [21], they have estimated the channel depends on Comb- type pilot frequencies arrangement, and the pilot ratio was 1/8.

In 2010, Hala Mahmoud, Allam Mousa, Rashid Saleem [22], they have explored ability of comb-type pilot arrangements to track channel time varying but in their simulation the pilot ratio also was 1/8.

1.4 Problem statement and Contribution of thesis

Due to the frequent need for high data rate communication system, orthogonal frequency division multiplexing (OFDM) was proposed, where the different carriers are orthogonal to each other. With OFDM, it is possible to have overlapping sub channels in the frequency domain, thus increasing the transmission rate. Today, OFDM has grown to be the most popular communication system in high-speed communications. OFDM presents several important advantages, some of which are: high spectral efficiency; simple implementation (with IDFT/DFT pairs); mitigation of inter symbol interference (ISI) and robustness to frequency selective fading environments. In communication systems, channel estimation very important issue to recover transmitted data. Nowadays, there are several methods to solve problem of channel, when the channel impulse response is frequency-selective fast fading channel that means channel impulse response change rapidly from one OFDM symbol to the next symbol. A coherent detection is used in specific with comb-type pilot arrangement technique to solve this problem, there is another type of coherent detection is called block- type pilot arrangement as shown in figure 1.7, it is useful to estimate slow fading channel. In comb-type technique, pilot tones are inserted into each OFDM symbol as shown below in the figure 1.8. But insert more pilot tones will causes data rate reduction or bandwidth expansion. Such as in [18, 19, 20, 21, 22] pilot ratio was ranges from 1/16 to 1/8. In our proposed algorithm [3], the pilot rate reduced to 1/100 with good estimation for the channel that means the data information in OFDM packet is increased without need to extend the bandwidth, as well, enhanced SNR especially with low values.

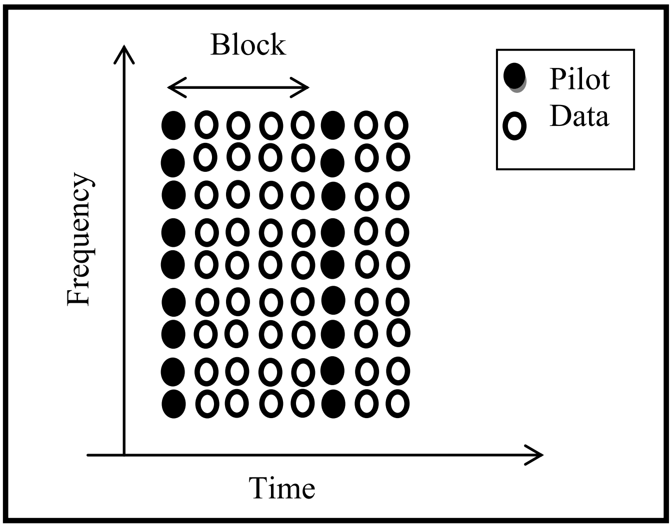

Figure 1.7 Block-Type Arrangement

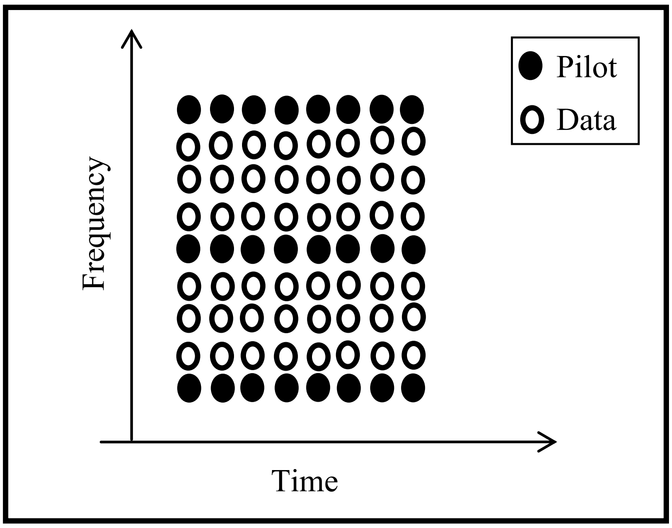

Figure 1.8 Comb-Type Pilot Arrangement

1.5 Organization of thesis

This thesis consists of five chapters:

Chapter 1: Contains OFDM introduction, motivation, literature survey, problem statement of thesis and thesis organization.

Chapter 2: In this chapter the basic principle about OFDM system is discussed and the system mathematical model is given.

Chapter 3: Presents channel estimation and interpolation in comb-type pilot arrangement by using Least Square (LS) algorithm and the proposed algorithm.

Chapter 4: Shows the simulation result.

Chapter 5: Presents conclusion of the thesis and recommendation for future studies that can be conducted in this field.

CHAPTER 2

OFDM System Fundamentals

2.1 Single-carrier and Multi-carrier Communication System

2.1.1 Single-carrier Transmission System

When data rate of a system is not very high, and inter-carrier interference ICI result of multipath signal is not predominantly serious, the single-carrier communication system is typically used as shown in Figure 2.1, where the $H(t)$ is the matching filter. Then a suitable equalization algorithm could be used to make the system work correctly. But for wideband applications of high data rate transmission, inter-symbol interference (ISI) caused by delay between received data symbols that reflected from far subjects, which call for advanced equalization requirements. As well as, a high signal bandwidth and near to channel bandwidth, time dispersion during transmission will cause frequency selective fading, these led to making frequency components of signal are different and signal characteristics will change.

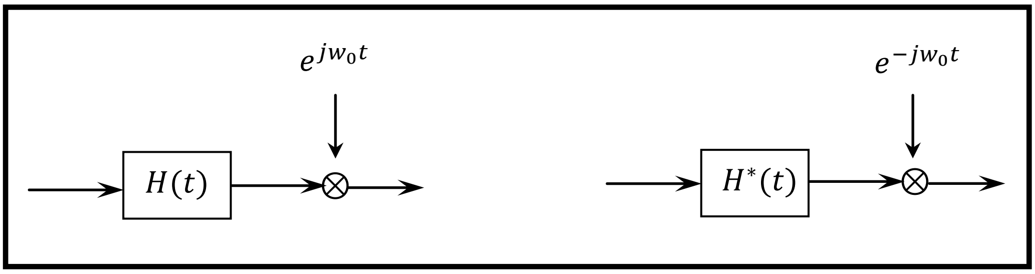

Figure 2.1 Frame of the Single Carrier

2.1.2 Multi-carrier Transmission System

Multi-carrier Transmission System have been used in the services that required high data rate transmission, the main idea of this system, divide transmitting data into.

numerous components and send these components over independent carrier signals. The independent carriers (sub-carriers) have a narrow bandwidth, but the combined signal may have wide bandwidth.

Advantages of multi-carrier transmission system

- High data rate transmission.
- It has less sensitivity than single-carrier systems to interference because of impulse noise.
- It has immunity against multipath fading.
- Improved immunity to inter-symbol-interference.

But the difficulty lies in the carriers synchronizing under marginal conditions. Figure 2.2 illustrates the basic structure of multi-carrier system. OFDM system, discrete multi-tone (DMT) and multi-carrier modulation (MCM) are examples for multi-carrier system [38].

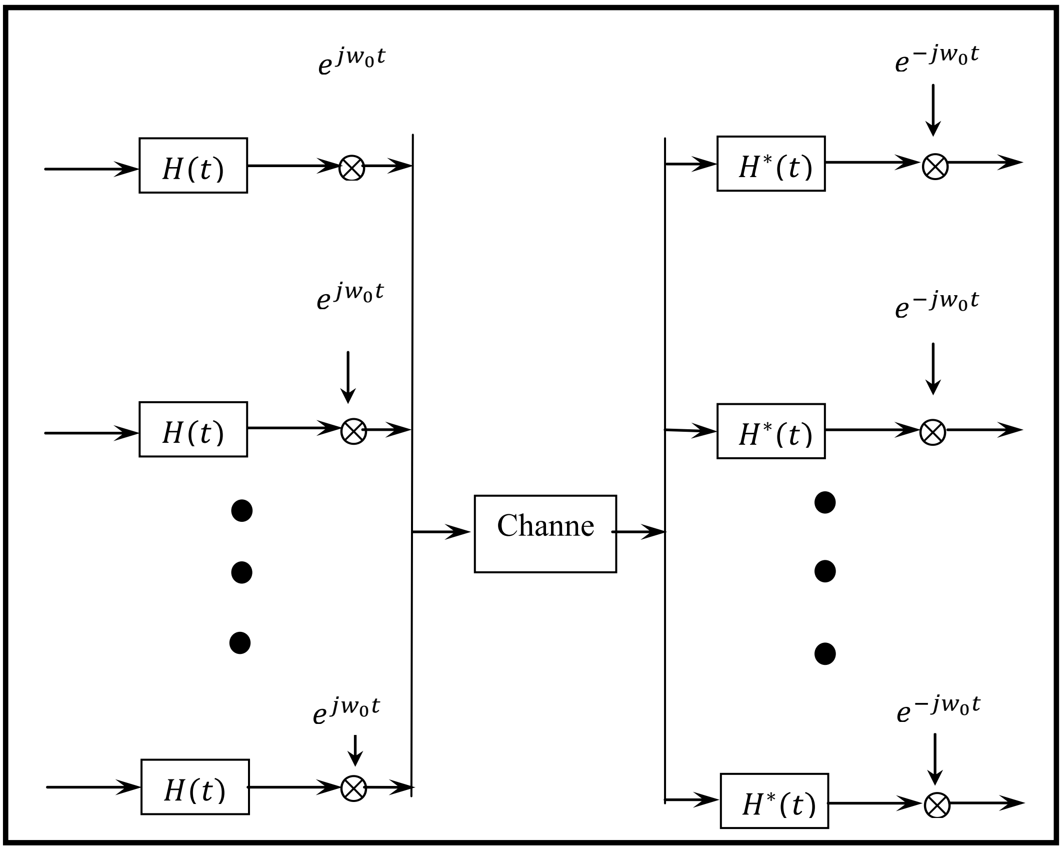

Figure 2.2 Frame of the Multi-Carriers

2.2 Frequency Division Multiplexing FDM and Orthogonal Frequency Division Multiplexing OFDM

2.2.1 Frequency Division Multiplexing FDM

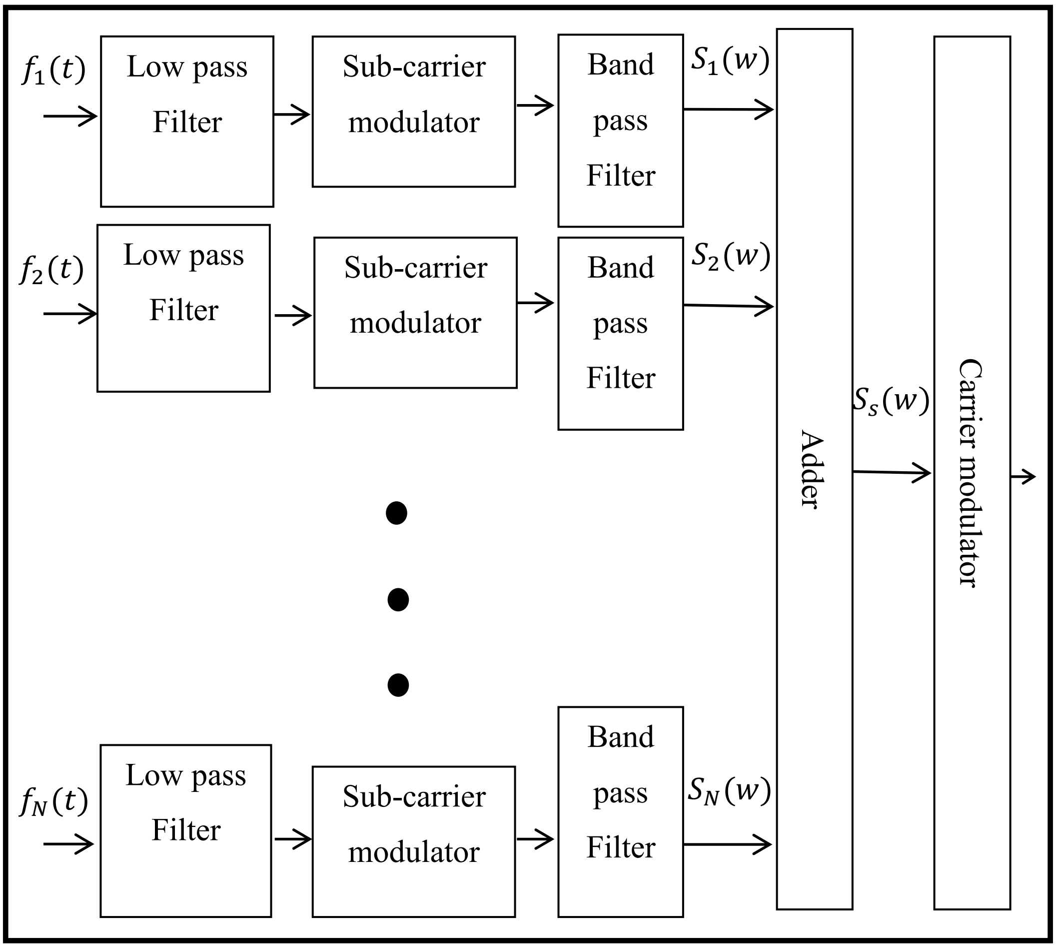

Figure 2.3 Frame of FDM System

Figure 2.3 Illustrates sending part of Frequency-division multiplexing (FDM) system, Where FDM system combines many of the signals and then send them on a single communications channel and each signal is assigned to different frequency as a sub-channels from the main channel. Here, we assume there are N signals $f_1(t), f_2(t), \ldots, f_N(t)$ with the same bandwidth. Firstly, they are passed through low pass filter separately to make sure that the bandwidth does not exceed $2w_f$, where w_f represents highest frequency of each signal and then the sub-carrier modulator affords each signal different frequency to avoid overlapping in the frequency axis, also to make receiving part able to distinguish between signals because they are in the same channel. The Band pass filter have been used before adding operation in each branch to select frequency band shared by each sub-carrier, at end we have a multichannel (base-band) signal which can be transfer directly by wire. The signals $S_1(w), S_2(w), \ldots, S_N(w)$ are non-overlapping on frequency bands at this time, so adding devices can be used to transfer N signals together. Frequency division multiplex (FDM) signal can be expressed as given below:

$$S_s(t) = \sum_{n=1}^{N} f_n(t) \cos w_n(t) \tag{2.1}$$

To achieve wireless transmission, the signal needs to be synthesized at a modulated RF carrier, and it is called the primary or secondary modulated carrier modulation. At the receiving side, received signal passes through opposite process devices to begin with demodulate the main RF carrier, the recovered signal is sent to all multi-band pass filters on each branch. The center frequency for each band-pass filter is predetermined and then to sub-carrier frequency de-modulator to achieve the division of the frequency domain [39]. After demodulating subcarrier of the signals, the spectra information can be obtained from several directions of FDM [40] as appeared in Figure 2.4.

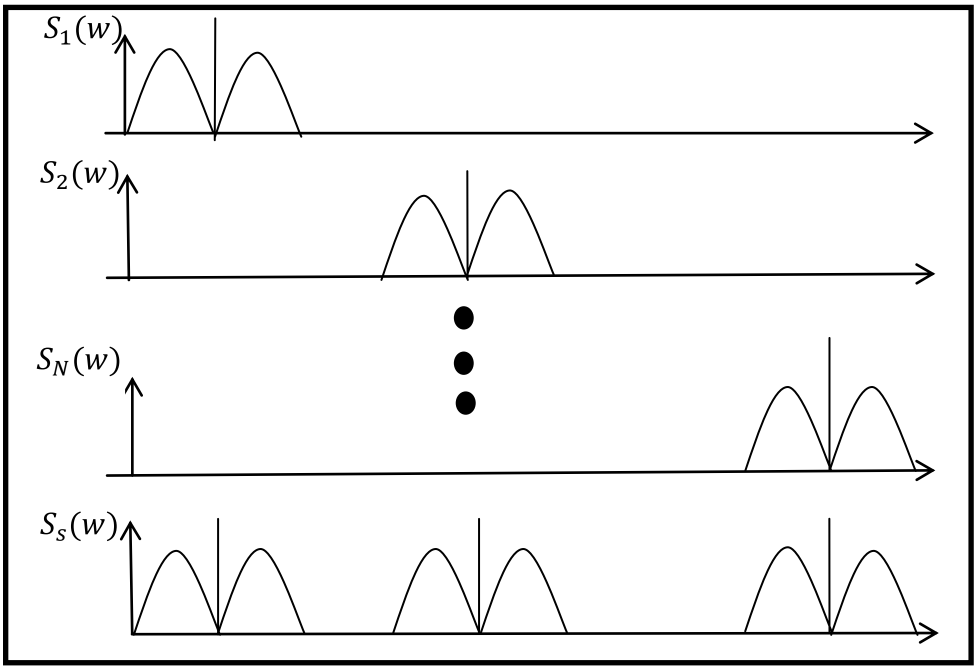

Figure 2.4 Spectrum Analysis of FDM

2.2.2 Orthogonal Frequency Division Multiplexing OFDM

On the basis of the principle of FDM, OFDM uses orthogonality condition between subcarrier groups sine or cosine function as a subcarrier groups. The orthogonality $\{\cos nwt$ and $\sin mwt$ n, m=1, 2, 3…} happens in $(t_0, t_0 + T)$ as given below:

$$\int_{t_0}^{t_0+T} \cos nwt \sin mwt \ dt = \begin{cases} 0 \ \ (n \neq m) \\ T/_2 \, (n = m) \\ T(n = m = 0) \end{cases} \quad T = \frac{2\pi}{w} \tag{2.2}$$

The cosine function is similar to sine function the difference just in the phase. In accordance with the theory, let frequencies of N sub-carriers are $f_1, f_2, \dots, f_N$ and $f_k = f_0 + \frac{k}{T_N} \quad k = 1,2, \dots, N$, where T_N is unit code duration. The signal of single sub-carrier is defined as below:

$$f_k = \begin{cases} \cos(2\pi f_k t) & 0 \leq t < T_N \\ 0 & others \end{cases} \tag{2.3}$$

As we know, the orthogonality condition is:

$$\int f_n(t) \, f_m(t) \, dt = \begin{cases} T_N & m = n \\ 0 & m \neq n \end{cases} \tag{2.4}$$

The subcarriers are orthogonal each other if the eq. (2.4) is achieved. The receiver can demodulate signal modulated by orthogonality if signal is synchronized accurately. The OFDM signal is same as FDM but the difference is spectrum as shown above in chapter 1 in Figure 1.5.

From Figure 1.5, FDM signal needed to wide interval protection because it occupied a wide bandwidth during transmission, the reasons of wide bandwidth occupying returns to frequency division and guard interval (GI) to achieve an accurate demodulation at the receiver. But the received OFDM signal has a small spectrum, due to subcarriers are overlapped, therefore it needs a smaller bandwidth.

2.3 OFDM Based Multiple Access Techniques

Frequency spectrum is a common resource between users, and requires a suitable technique to allow each user distinct access. Because the large number of users, they are tried to simultaneously channel access and the demand for frequency sharing, spectrum of channel becomes crowded.

Orthogonal Frequency Division Multiplexing OFDM and hybrid systems have been proposed. These hybrid systems achieved efficiently use of channel frequency spectrum and minimizing effect of interference. Here, we discuss about OFDM based multiple access techniques. Frequency division multiple access (FDMA) splits the existing bandwidth into several non-overlapping sub-channels with guard interval between adjust channels to avoid interference and assigned each sub-channel to a specific user. But the next call can't be located until a channel is available, In addition to waste of the available bandwidth caused by guard insertion. FDMA technique is effectual if the network size is small. OFDM-FDMA systems divide the entire bandwidth into several sub-bands. Adaptive bit-loading is possible in each subcarrier which assigns different modulation structures in order to assign number of different of bits to all subcarrier.

Time Division Multiple Access (TDMA) is divide time frame into integer number of time called slots, each slot is assigned to user. This grants each user to use full spectrum for a specific time interval. In TDMA systems, allocation of frequency is easier and requirements of power are low. Overhead is necessary in each frame for synchronization problem. In OFDM-TDMA systems, a number of users use all subcarriers within the predefined TDMA time interval. WLAN systems have been developed using OFDM-TDMA technology, where all users use OFDM modulation scheme for data transmitting. The Orthogonal Frequency Division Multiplexing OFDM modulation system, also it is proposed to join with code division multiple access CDMA, creates a so-called hybrid system. CDMA is a spread band technique that uses direct sequence spread spectrum. Frequency spectrum is divided both in the time and frequency directions. All users in CDMA technique can use the full channel for the entire time. The narrow-bands signals in CDMA are multiplied by pseudo-noise code sequences with higher rates and each code is unique. At the receiving side, receiver can recover the transmitted signal by using correct code sequence.

Orthogonal frequency division multiple access (OFDMA) is another promising scheme that paying attention a lot of interests. OFDMA is based on OFDM and has proved robustness against inter symbol interference caused by frequency-selective fading channels. Flash-OFDM is a one type of OFDMA systems, where the users are distributed midst the mobile stations. Flash-OFDM uses fast hopping across all tones in a predetermined pattern. Fast hopping, one user uses one tone in OFDM symbol and a different tone in the next OFDM symbol. When the flash-OFDM system is near to base stations will be higher data rate as different tones in each OFDM symbol could be modulated differently [41].

2.4 OFDM System Architecture

2.4.1 OFDM Signal Generation

The main idea of OFDM signal generation is subdivided whole bandwidth into N number of orthogonal subcarriers which are modulated by phase shift keying PSK or quadrature amplitude modulation QAM. The N number of orthogonal subcarrier is equal to $N = 2^n$ where n is an integer number, for example $N = 64$ in IEEE802.11a (WLAN) [41]. The expression of each baseband subcarriers as given below:

$$\Phi_k(t) = e^{2j\pi f_k t} \tag{2.5}$$

Here, f_k is the frequency of k^{th} subcarrier.

Each OFDM symbol consists of N subcarriers and can be expressed as below [42]:

$$S(t) = \frac{1}{N} \sum_{k=0}^{N-1} X_k \ \Phi_k(t)\ , \ 0 < t < NT_s \tag{2.6}$$

Here:

X_k : Data symbol that be taken from modulated signal PSK or QAM constellation.

NT_s : Length of OFDM symbol.

T_s : Sampling time.

The frequency of each subcarrier is given by:

$$f_k = \frac{k}{NT_s} \tag{2.7}$$

Where, $k = subcarrier\ index.$

After replacing $\Phi_k(t)$ in eq. (2.6) and then sampling at frequency equal to $\frac{1}{T_s}$, we will get the discrete version of eq. (2.6) as expressed below:

$$S(nT_s) = \frac{1}{N} \sum_{k=0}^{N-1} X_k \, e^{\frac{2j\pi kn}{N}} \;\; , \; 0 < n < N-1 \tag{2.8}$$

Thus, eq. (2.8) represented the inverse discrete Fourier transform (IDFT) of the modulated signal constellation symbol X_k. At the receiver, discrete Fourier transform (DFT) is used to demodulate OFDM signal.

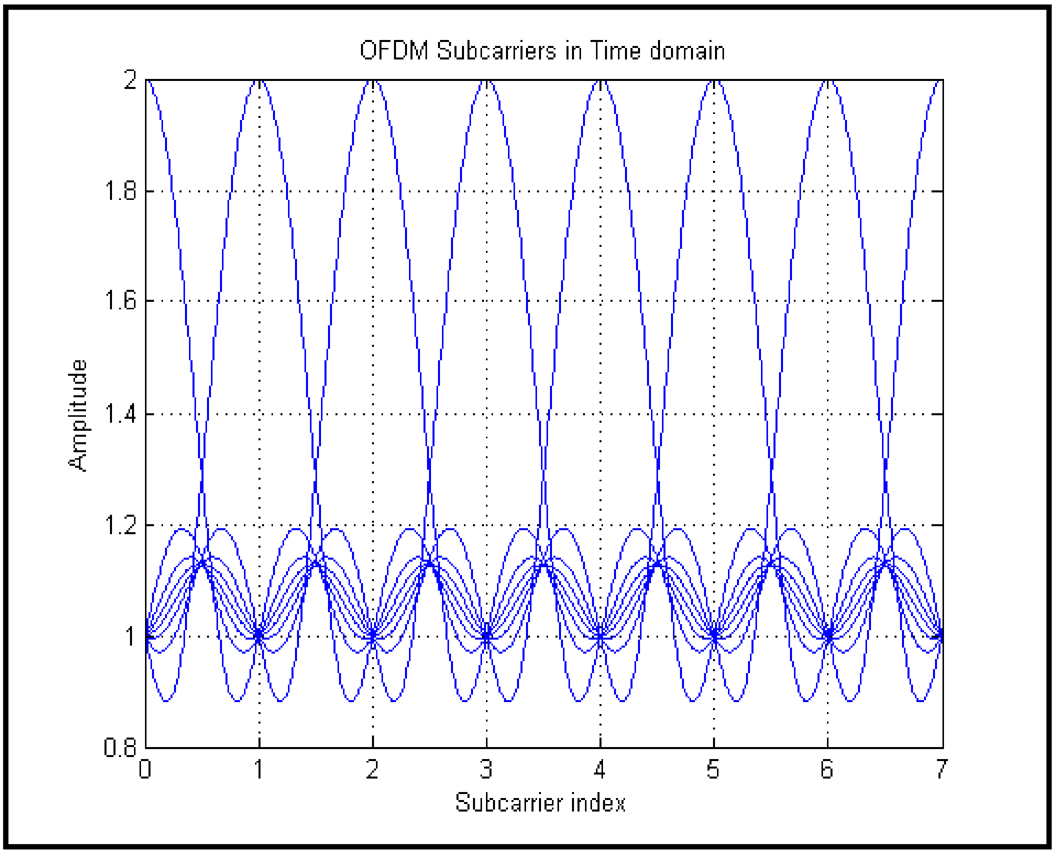

Figure 2.5 Consecutive Sub-Carriers of OFDM

Figure 2.5 shows spectrum of eight subcarriers and this figure gives each separate subcarrier within the graph through the rectangular wave forming the representation of the sinc function by the spectrum. At the peak value of the each sub-carrier frequency, all other sub-channel spectrum value is zero, accurately. At the receiver, demodulation process of OFDM symbols, these points matching to the maximum value for all subcarrier frequency must be calculated [43], so that we can extract every sub-channel spectrum symbol from several overlapping sub-channels without interference between them.

2.4.2 Basic Block Diagram for OFDM system

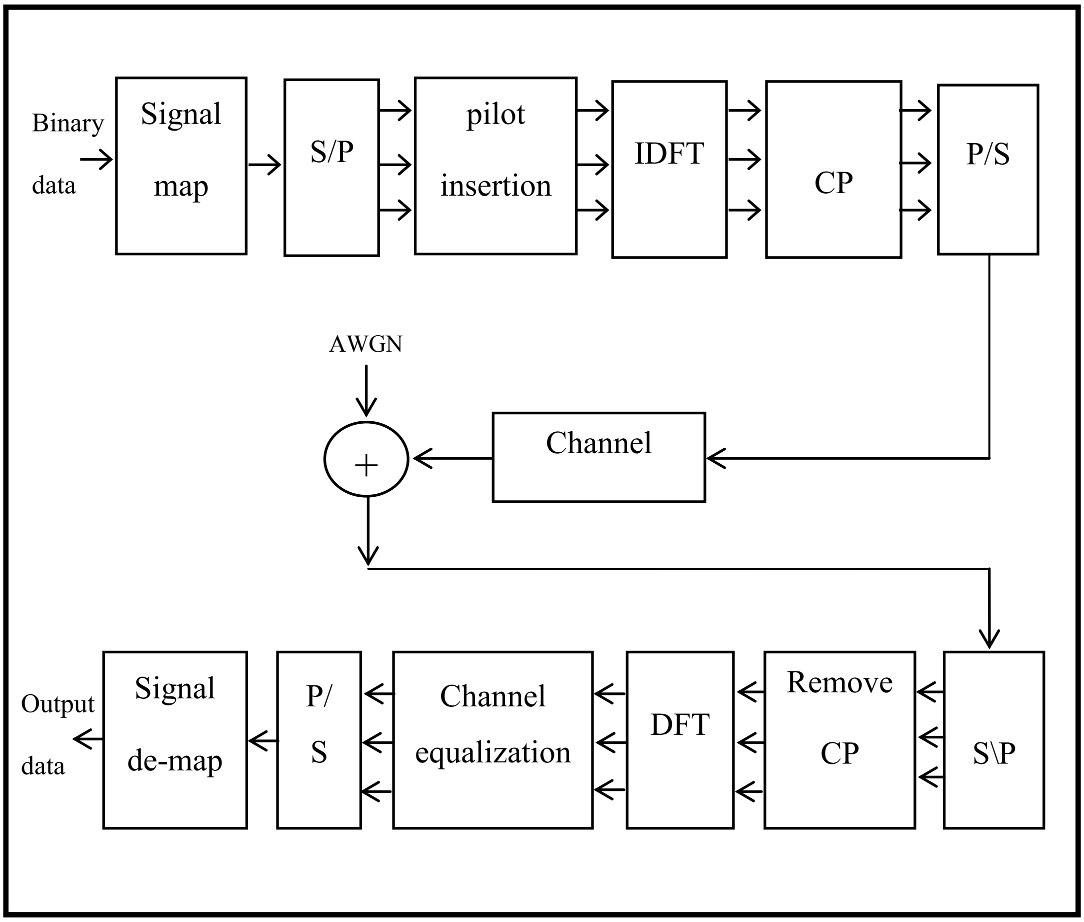

Figure 2.6 Basic Block Diagram for OFDM System

Figure 2.6 shows the basic block diagram of OFDM systems. Firstly, the binary data is grouped and mapped into complex data form in the block of signal mapper. Usually, signal mapper is phase shift keying (PSK) or quadrature amplitude modulation (QAM). After, the complex data passed through serial to parallel (S/P) converter block to convert it from serial path to parallel paths with number equal to number of sub-carriers. Pilot tones are added to parallel data (in each OFDM symbols) in the pilot insertion block for channel equalization requirements then modulated by using the inverse fast Fourier transform (IFFT) operation in the next block. Later, to mitigate the impact of inter symbol interference (ISI), guard interval is added to each OFDM symbols. The parallel data is again converted to serial path then transmitted over the frequency selective fading channel with additive white Gaussian noise (AWGN) environment. In the receiving side, received data is passed in the reverse processes. Serial corrupted data is converted into parallel paths, then the guard interval is removed and then data is converted from time domain to frequency domain by applying fast Fourier transform (FFT). In the channel equalization block, the channel effects are compensated and then complex data is obtained. At last, output binary data is recovered by applying de-modulation in the de-mapper block. Each block in the figure 2.6 has been clarified in the next section.

2.4.2.1 QAM Modulation

Quadrature amplitude modulation (QAM) has been applied through this thesis. The term of "quadrature" means that there are only four combinations of phase and amplitude which the carrier can have at a predetermined time and each combination is assigned a 2-bit digital pattern. Some times this type of modulation is called as 4-QAM. In QAM, carrier is varied in phase and amplitude, while the frequency remains constant.

Figure 2.7 shows 4-QAM constellation map, the four phases with the same amplitude are labeled as (A, B, C, D) and the phase degrees is {45, 135, 225,315} respectively. Because of the four combinations of phase and amplitude, 2-bits are transferred in each time slot.

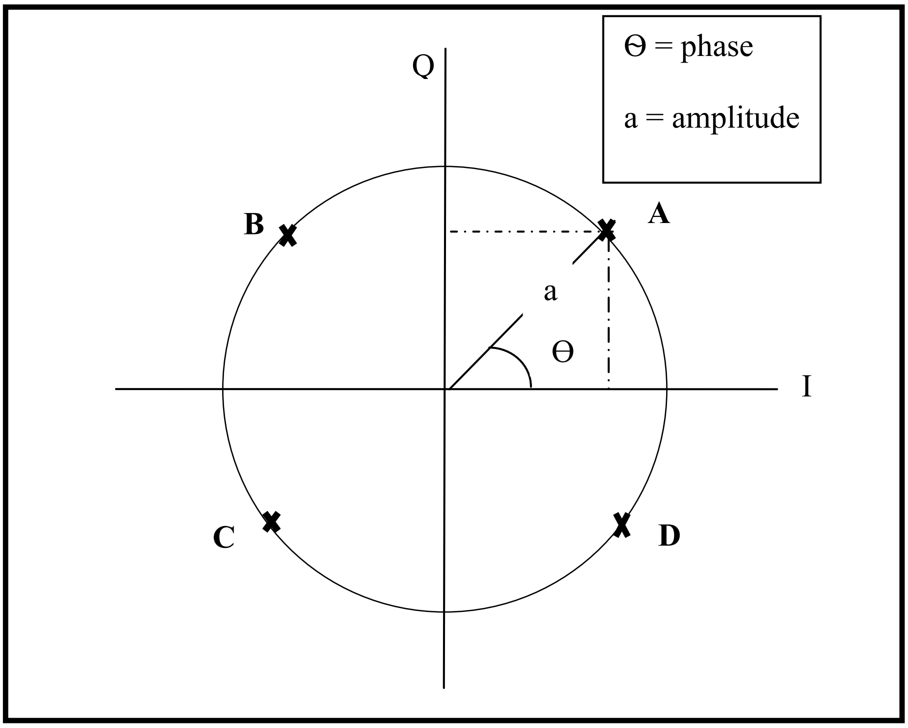

Figure 2.7 Signal Constellation of 4-QAM

Table 2.1 Signal Mapping

Phase degrees	state	Binary data
45	A	00
135	B	01
225	C	10
315	D	11

2.4.2.2 Inter-carrier Interference (ICI)

OFDM systems often suffer from sub-carrier frequency offsets because of multipath fading channels. Unsynchronized local oscillators at the transmitter and the receiver are caused frequency offset. Consequently, subcarriers can be shifted from their original positions and then it will be non-orthogonal signals at the receiver, that means inter-carrier interference (ICI) has been occurred in the receiving end. ICI causes degradation in the system performance. Also, signal transmission through time variant channel destroys the orthogonality between subcarriers and led to inter-carrier interference. Because of this ICI, OFDM systems become very sensitive to frequency offsets and reduce achievable bit error rate (BER) performance. One of the ways that have been used to combat the frequency offset is data added. In data aided techniques, known bits are used as a training sequence or pilot tones are inserted to each OFDM symbol to estimate the frequency and timing offsets.

2.4.2.3 Pilot insertion

Comb-type pilot tones arrangement is used throughout this thesis. In this type of insertion, the pilot tones have been inserted uniformly to each OFDM symbol (parallel data). In each group, the first subcarrier is used to transmit pilot signal. Eq. (2.9) expresses the insert operation as below:

$$X(K) = \begin{cases} X_p(m) & i = 0 \\ info.data & i = 1, \ldots, L-1 \end{cases} \tag{2.9}$$

Where,

L : Symbol length

$X(K)$: Data symbol is taken from modulated signal PSK or QAM constellation after S/P stage plus pilot signal.

$X_p(m)$: The pilot signal vector where m is an integer number and p is the number of pilot signal, it's depend on number of OFDM symbols or packet length.

$info.data$: Transmitted data.

2.4.2.4 IDFT/DFT

At the transmitter after pilot insertion, parallel data are arranged up into frames and then each frame is transformed from frequency domain to time domain by using the inverse discrete Fourier transform (IDFT). At the receiver, discrete Fourier transform (DFT) has been implemented to return data back to the frequency domain. After applied IDFT/DFT, that means modulation and demodulation of OFDM are performed. DFT for a finite length N is given as [44]:

$$X(K) = \sum_{n=0}^{N-1} x(n)\, W_N^{Kn}\ , K = 0,1, \ldots\ldots, N-1 \qquad \text{(Frequency domain)} \qquad (2.10)$$

Where $W_N = e^{\frac{-j2\pi}{N}}$

And the IDFT as defined in [44]:

$$x(n) = \frac{1}{N} \sum_{K=0}^{N-1} X(K)\, W_N^{-Kn}\ , n = 0,1, \ldots\ldots, N-1 \qquad \text{(Time domain)} \qquad (2.11)$$

But the disadvantage of DFT is complex and cost ineffective. Practically to solve complexity and cost issue, IFFT/FFT can be easily used for OFDM modulation and demodulation [25].

2.4.2.5 Inter-symbol Interference (ISI)

In several wireless systems, due to multipath channels the transmitted signal reflects from multiple objects such as (buildings, towers, mountains). As a consequence, multiple delay version of the transmitted signal is occurred, that means the transmitted signal is arrived at the receiver at different time periods. Because of multiple delay version of transmitted signal, OFDM symbol at the receiver becomes distorted by the formerly transmitted OFDM symbol. This problem is known as inter-symbol interference (ISI). Figure 2.8 is denotes the problem of inter-symbol interference.

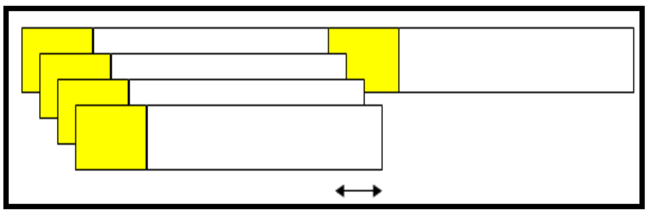

Figure 2.8 Example of Inter-Symbol Interference

Inter symbol interference is destroyed the first samples of OFDM symbol. To minimize Inter symbol interference problem, extra guard interval is used in front of each OFDM symbol.

2.4.2.6 Cyclic Prefix

Because of multipath fading channel environment, channel fading causes following OFDM symbols to overlap and led to inter-symbol interference. ISI destroys the orthogonality between subcarriers and degrades the overall system performance. To combat effects of ISI and to maintain the orthogonality condition, a guard interval has been used in each OFDM symbol. Usually, the end of OFDM symbol has a circular shift that means the rear of OFDM symbol is repeated in the front portion in the same symbol. This technique of guard interval is called cyclic prefix. There isn't any further information which is sent through cyclic prefix (CP), the guard interval is removed at the receiver. There is a condition; the length of cyclic prefix must be equal to or greater than maximum channel delay (channel impulse response) to prevent inter-symbol interference. Because of circular insertion of CP, OFDM symbols is appeared as a periodic when convolved with impulse response of channel, because of the circler shift cyclic prefix addition, the OFDM symbol is appeared as a periodic signal when convolved with channel impulse response and then the linear convolution converts into a cyclic convolution. A cyclic convolution is only a scalar multiplication in frequency domain therefore the channel effect on the transmitted data becomes multiplicative. In order to reduce effect of the channel, another process is applied at the receiving side which works like an inverse filter of impulse response of the channel. This type of process is known as channel equalization. Figure 2.9 is explaining the technique for the cyclic prefix insertion.

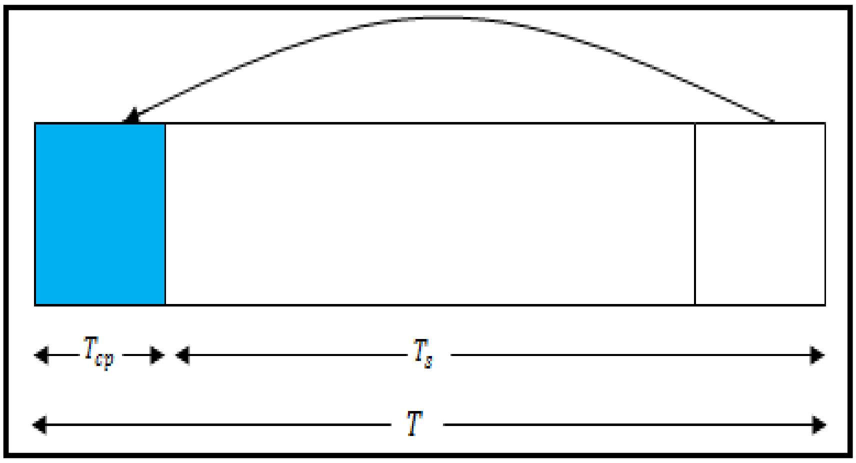

Figure 2.9 Insertion of Cyclic Prefix (CP)

Here:

T_{cp} : The length of cyclic prefix.

T_s : The length of original OFDM symbol.

$T = T_{cp} + T_s$: The length of the transmitted symbol.

The resultant samples after CP addition is $x_g(n)$ as shown in equation below:

$$x_g(n) = \begin{cases} x(N+n) & n=-N_g,-N_g+1,\ldots\ldots,-1 \\ x(n) & n=0,1,\ldots\ldots,N-1 \end{cases} \quad (2.12)$$

Where, N_g is the number of samples in the guard interval and N is the number of subcarriers. At the receiver, cyclic prefix is discarded to avoid inter-symbol interference.

Disadvantages of cyclic prefix are:-

- Increases transmitting energy because of the extra bit padding.
- Reduces the efficiency of the overall system.

2.4.2.7 Channel Model

Obstacles in many practical wireless communication systems like buildings, earth, trees, etc., are present in the propagation path. Signal suffer from diffracts, reflects and scatters, when it transmitted over this radio channel. As a result, transmitted signal arrives on the receiver out of multiple paths which is known multipath propagation. This phenomenon of this propagation is known as multipath channels. The arrived signal on the receiving end at different time, each signals may have different phase, amplitude and delay. In this case the received signals are interfere together and this interfere may add up constructively or destructively based on the signals phases. Figure 2.10 is described the multipath propagation.

The multipath channel impulse response causes time delay spread and this led to variations of the received signal strength with frequency and time. These variations are called fading characteristics. Root mean square (RMS) delay spread (T_{RMS}) is the metric to describe delay in the multipath channels [41].

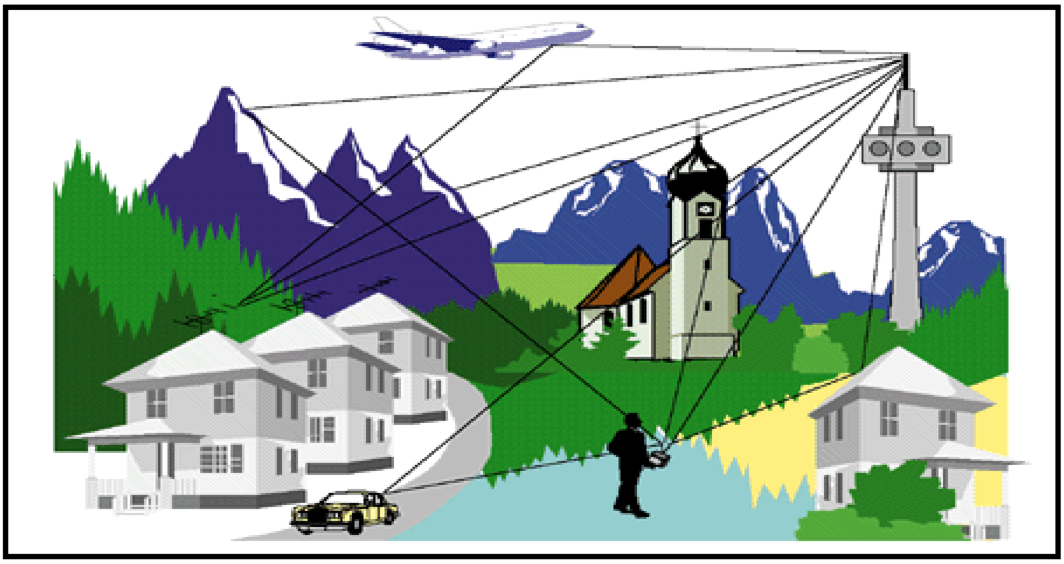

Figure 2.10 Multipath Propagation Situations

As well, coherent bandwidth B_c is one of the most important fading channel characterizations. The coherent bandwidth is the frequency interval between two frequencies of a signal having the same type of fading. A signal undergoes non-frequency-selective fading if:

$$B_s \ll B_c$$

$$T_s \gg T_{RMS}$$

Where B_s the bandwidth of received signal, B_c the coherent bandwidth as mentioned above, T_{RMS} the channel delay and T_s is the symbol time. In the case of whole signal spectrum is affected evenly. Frequency selective fading occurs if:

$$B_s > B_c$$

$$T_s < T_{RMS}$$

Due to frequency selective fading channels, the whole signal spectrum is not affected evenly.

In our simulation, we have used frequency selective fading channel where the channel impulse response is defined as [45]:

$$h(n) = \sum_{k=1}^{L} \gamma_k \, x_k(n) \, \delta(n - n_k) \tag{2.13}$$

Where $x_k(n)$ models of Rayleigh fading, L, γ_k , and $x_k(n)$ are respectively the total number of the channel paths, path loss and Rayleigh fading sequence of the k-th group of reflectors, and n_k are the groups delay.

The received signal is described as a convolution between the transmitted signal $x(n)$ corrupted by the additive white Gaussian noise $w(n)$ and the channel impulse response $h(n)$ as equation below:

$$y_g(n) = x_g(n) \otimes h(n) + w(n) \tag{2.14}$$

After that, the pilot tones are extracted and the estimated channel $H_e(K)$ for the data sub-channels is obtained in block of the channel estimation as well as the transmitted data is estimated, as in

$$X_e = \frac{Y(K)}{H_e(K)} \tag{2.15}$$

At the end, the binary information data is obtained back in de-mapping block (QAM-demodulation)

CHAPTER 3

CHANNEL DETECTION BASED ON PILOT INSERTION METHODS

3.1 Introduction

Channel estimation based on pilot tones arrangement is divided into two kinds. The 1^{st} one is the block-type pilot arrangement as showed in figure (1.7) in the chapter one, it was developed on the assumption of slow fading channel. The 2^{nd}, comb-type pilot arrangement as showed in figure (1.8) in the chapter one, it has been introduced to estimate fast fading channel. In our work, we consider the radio channel that is changing rapidly. In such an environment, the channel impulse response changes significantly from one OFDM block to the next block. Thus, the channel estimation in the current block cannot be used in the following block as a channel response. Therefore, the comb-type pilot tones arrangement is adopted to estimate the channel response in each block. In this chapter, the Least-Square LS Estimator is presented as it is required by several estimation techniques as an initial estimation. Then the modified Least-Square MLS estimator is proposed in an attempt to reduce the pilot ratio and enhance bit error rate BER especially with low signal to noise ratio SNR.

3.2 Channel Estimation Techniques

3.2.1 Least-Square (LS) Estimator

From equation (2.15), after removing guard interval from the received signal, the DFT is applied, so the received signal became as described below:

$$Y = XH + W \tag{3.1}$$

The LS estimator minimizes the cost function for the channel impulse response [46] as shown below:

$$\min_{\hat{h}} \left(Y - X\hat{h}\right)^{H} \left(Y - X\hat{h}\right) \tag{3.2}$$

Where $[\,.\,]^H$ is the Hermitian operator (Conjugate and Transpose) and $\widehat{h}$ is the estimated channel.

And it is generates the channel attenuation as given below:

$$h_{LS} = FQ_{LS}F^H x^H y \tag{3.3}$$

Here,

$$Q_{LS} = (F^H x^H F x)^{-1} \tag{3.4}$$

And F is the Discrete Fourier Transform (DFT) but in the matrix form as defined below:

$$[F]_{nk} \triangleq \frac{1}{\sqrt{N}} e^{(\frac{-j2\pi}{N})nk} = \begin{bmatrix} \frac{1}{\sqrt{N}} e^{(\frac{-j2\pi}{N})00} & \cdots & \frac{1}{\sqrt{N}} e^{(\frac{-j2\pi}{N})0(N-1)} \\ \vdots & \ddots & \vdots \\ \frac{1}{\sqrt{N}} e^{(\frac{-j2\pi}{N})(N-1)0} & \cdots & \frac{1}{\sqrt{N}} e^{(\frac{-j2\pi}{N})(N-1)(N-1)} \end{bmatrix} \tag{3.5}$$

Then, the lest square LS estimator of the channel impulse response can be written as

$$\hat{h}_{LS} = x^{-1} y \tag{3.6}$$

Equations (3.6) is the general form of the LS estimator, the computational complexity of LS is very low but it suffers from high mean-square error MSE [47], because it directly goes to estimate channel from received signal without noise control, as well as channel interpolation increase the sampling rate and thus noise will increase, therefore, it needs a sufficient number from the pilot tones to be able to estimate channel such as in [18, 19, 20, 21, 22], the pilot ratio was ranges from 1/16 to 1/8. However, increasing pilot number led to data rate reduction or bandwidth expansion. While the proposed algorithm was reducing pilot ratio to 1/100 and enhanced bit error rate BER especially with low values of signal to noise SNR ratio. The simulation result showed a good performance with very small part of OFDM symbol used as pilot tones [3].

3.2.2 Proposed Method

Modified Least-Square (MLS) Estimator Method

The LS estimator can be designed with very low complexity but it has a high bit error rate especially with low signal to noise ratio. Because it does not take into account the effects of noise on the received signal [48]. In the proposed modified Least-Square MLS algorithm, the effects of the noise has been reduced before the process of the channel estimation at the pilot tones only to combat the spread of noise over the total length of the estimated channel after the interpolation process. The noise effects are reduced by divided the received pilot tones signal by the length of the subcarriers to get a small value with the same range result as described below in equation (3.7):

$$R_{mod\ LS} = [Y_p(0), Y_p(1), \dots \dots, Y_p(k_p - 1)]/N \tag{3.7}$$

Where kp is the pilot number, N is subcarrier number and Y_p is the received signal at pilot signal only. Then multiplying the result by the inverse of known pilot signal that be generated in the receiver to minimize negative effects of the channel as explained below in equation (3.8):

$$Z_{mod\ LS} = (Q_p^T)(R_{mod\ LS} * X_p^*) \tag{3.8}$$

Where,

$$X_p = [X_p(0), X_p(1), \dots \dots, X_p(k_p - 1)] \tag{3.9}$$

Here, The operators$(\cdot)^T$and $(\cdot)$*, denote the transpose, and conjugate respectively of a vector or matrix, and Q_p as described below in equation (3.10):

$$Q_p = \begin{bmatrix} \frac{1}{\sqrt{N}} e^{(\frac{-j2\pi}{N})00} & \cdots & \frac{1}{\sqrt{N}} e^{(\frac{-j2\pi}{N})0(k_p-1)} \\ \vdots & \ddots & \vdots \\ \frac{1}{\sqrt{N}} e^{(\frac{-j2\pi}{N})(k_p-1)0} & \cdots & \frac{1}{\sqrt{N}} e^{(\frac{-j2\pi}{N})(k_p-1)(k_p-1)} \end{bmatrix} \tag{3.10}$$

And then dividing eq.3.8 by the root mean square of the subcarriers length to minimize the result as defined below in equation (3.11)

$$H_{p,mod\ LS} = Z_{mod\ LS}/\sqrt{N} \tag{3.11}$$

And then taking the mean value for the last result to achieve the maximum value as defined below in equation (3.12)

$$H_{p,mod\ LS} = (H_{0,mod\ LS} + H_{1,mod\ LS} + H_{2,mod\ LS} + \ldots\ldots.. + H_{(k_p-1),mod\ LS})/k_p \tag{3.12}$$

At the end of these processes, the channel at pilot tones is estimated in the frequency domain. After the estimation of the channel transfer function at pilot tones (H_p) based on comb-type pilot tones arrangement, an effective interpolation technique is essential to estimate channel at all data sub-carriers depend on the estimated channel information at pilot subcarriers. Figure 3.1 explain estimation process.

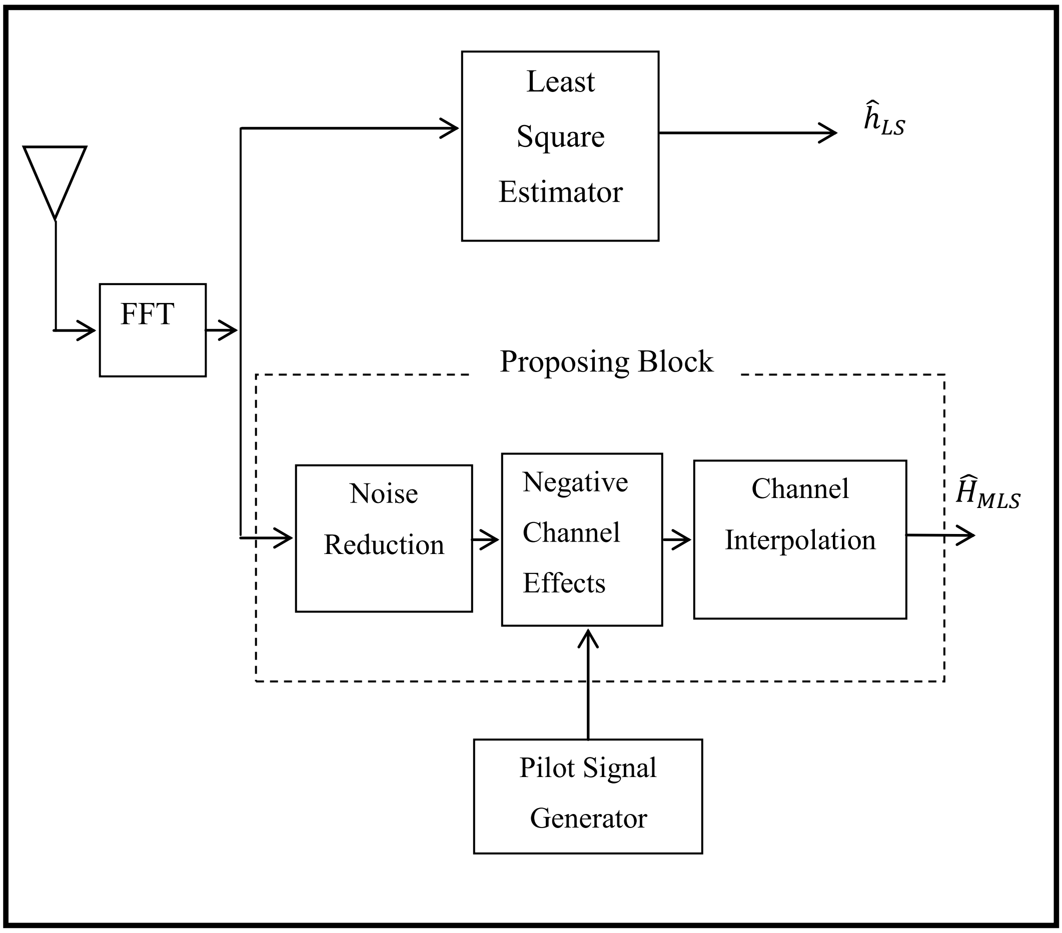

Figure 3.1 Block Diagram for least Square (LS) and Modified Least Square (MLS) Estimator

3.3 Channel interpolation

There are many interpolation techniques to estimate channel with length match to length of the data subcarriers as described below.

3.3.1 Linear interpolation (LI)

The idea of linear interpolation, the two consecutive pilots are used to estimate channel impulse response for data subcarriers that are in between these pilots, the estimation channel at data subcarrier k is defined as below [19]:

$$H_d(k) = H_d(mL + i) \qquad 0 \leq i < L - 1$$

$$= \left(H_p(m+1) - H_p(m)\right)\frac{1}{L} + H_p(m) \tag{3.13}$$

Where m is an integer number and $L = \text{number of carriers}/k_p$.

k_p: The pilot number.

3.3.2 Second-order interpolation (SOI)

The second-order interpolation is shown better than linear interpolation [18]. It depends on three successive pilots to estimate channel at data subcarrier as explained below:

$$H_d(mL + i) = c_1 H_p(mL - L) + c_0 H_p(mL) + \cdots + c_{-1} H_p(mL + L) \quad 0 \leq i < L - 1$$
(3.14)

Where, $\begin{cases} c_1 = \propto (\propto - 1)/2 \\ c_0 = -\propto (\propto - 1) \ast (\propto + 1), \propto = i/N \\ c_{-1} = \propto (\propto + 1)/2 \end{cases}$

3.3.3 Time domain interpolation (TDI)

In our computer program we have chosen time domain interpolation because it is a high-resolution interpolation based on zero-padding and DFT/IDFT [49]. And it is sufficient to work with our proposed algorithm. TDI first coverts estimated channel at pilot sub-carrier to time domain by inverse discrete Fourier transform IDFT as below:

$$G(n) = \sum_{k=0}^{N_p-1} H_p(k)e^{\frac{j2\pi kn}{N_p}} \qquad \text{n} = 0, 1, \ldots, k_p - 1 \tag{3.15}$$

And then interpolate the time domain signal to N points by using the basic multi-rate signal processing properties [35], as defined below:

$$M = \frac{k_p}{2} + 1$$

$$G_N = \begin{cases} G_p & 0 \leq n < M-2 \\ 0 & \frac{k_p}{2} \leq N - M \\ G_p & -M \leq n - N < -1 \end{cases} \tag{3.16}$$

Finally, the estimate of the channel at all data sub-carriers is obtained by discrete Fourier transform DFT as shown below:

$$H_e(k) = \sum_{n=0}^{N-1} G_N(n)\, e^{-\frac{j2\pi kn}{N}} \qquad 0 \leq k \leq N-1 \tag{3.17}$$

CHAPTER 4

SIMULATION RESULTS AND DISCUSSION

4.1 Simulation Parameters

Since the goal is to monitor channel coefficients and estimated output data, we assume to have an ideal synchronization between transmitter and receiver. In our MATLAB simulation program we have used color images as input data to the OFDM system to give us information more clarity about comparison between performances of the proposed algorithm and the original. MATLAB computer simulation has been conducted for different signal to noise ratios SNR, channel taps and images as well measuring bit error rate BER versus signal to noise ratio SNR. We have reduced pilot ratio in our proposed algorithm to 1/100 after it was ranges between 1/16 and 1/8 [3]. This will not merely make the system faster but also increase data rate efficiency. Before displaying results of the MATLAB computer simulation, we first describe the simulated OFDM system parameters in Table (4.1).

Table 4.1 Simulation Parameters

Parameters	**Specifications**
IFFT, FFT Size	1024
Number of active carriers	1024
Data Subcarriers	1008
Pilot Ratio	1/100
Channel Model	Rayleigh Channel
Doppler Frequency	150-200 Hz
Channel Tabs	8, 16
Data Type	Color Images
Modulation Type	4-QAM
Guard Type	Cyclic Extension

4.2 Results and Discussion

The RGB image is composed of three images red, green and blue, each image is called frame in our simulation. The three frames for each image are sent sequentially with new Rayleigh fading channel generation as shown below in details.

Figure 4.1 Original Image and their Composition Red, Green and Blue Images

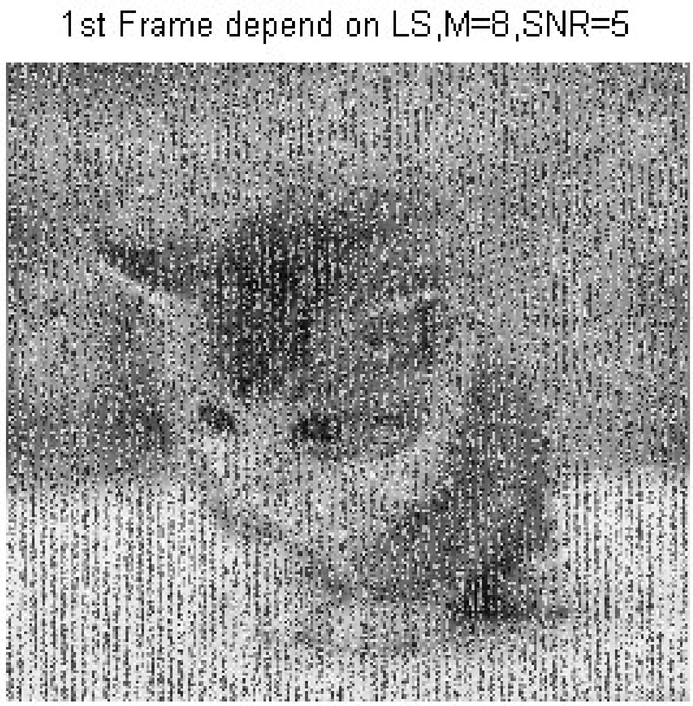

Figure 4.2 1st Frame (Red) Estimated Depend on Least Square Estimator when Channel length =8 tabs and signal to noise ratio =5 dB

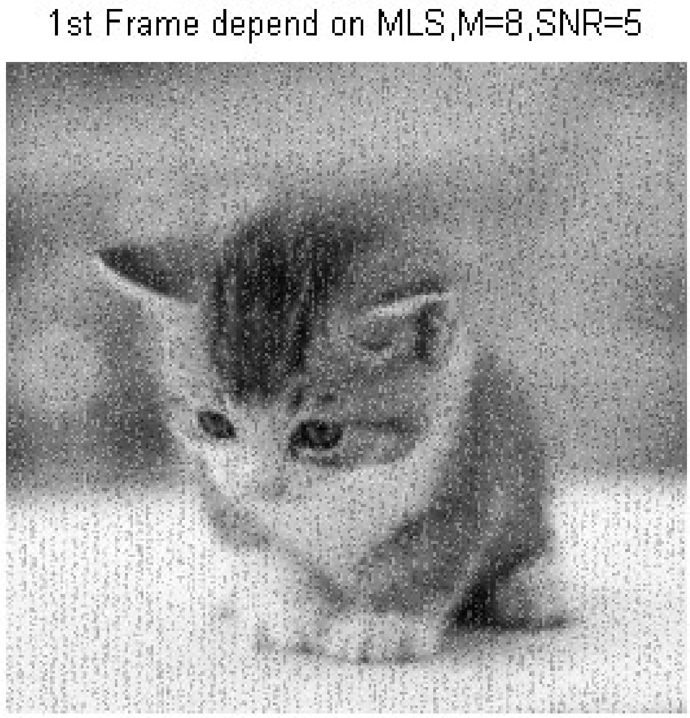

Figure 4.3 1st Frame (Red) Estimated Depend on Proposed Estimator when Channel length =8 tabs and signal to noise ratio =5 dB

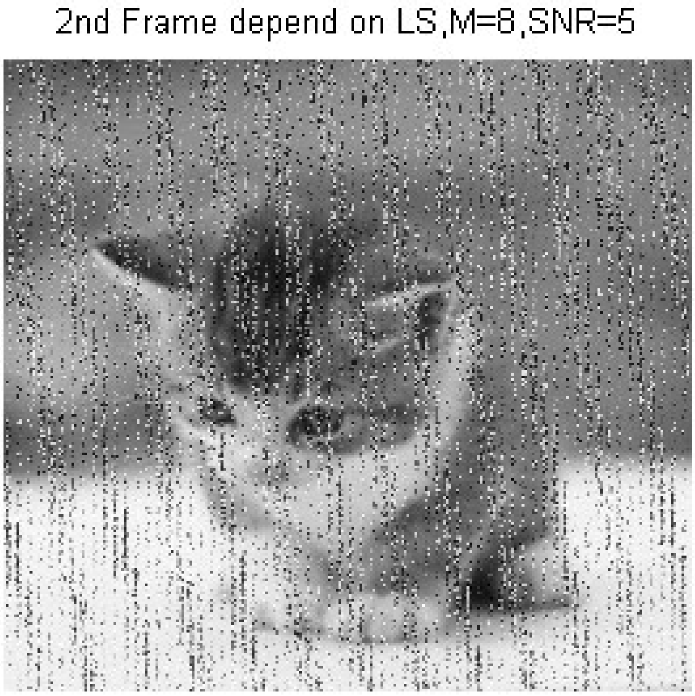

Figure 4.4 2nd Frame (Green) Estimated Depend on Least Square Estimator when Channel length =8 tabs and signal to noise ratio =5 dB

Figure 4.5 2nd Frame (Green) Estimated Depend on Proposed Estimator when Channel length =8 tabs and signal to noise ratio =5 dB

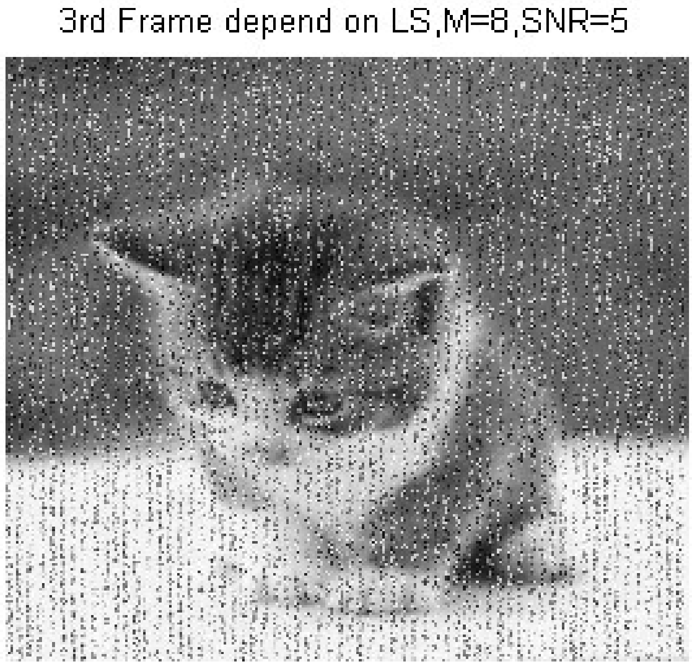

Figure 4.6 3rd Frame (Blue) Estimated Depend on Least Square Estimator when Channel length =8 tabs and signal to noise ratio =5 dB

Figure 4.7 3rd Frame (Blue) Estimated Depend on Proposed Estimator when Channel length =8 tabs and signal to noise ratio =5 dB

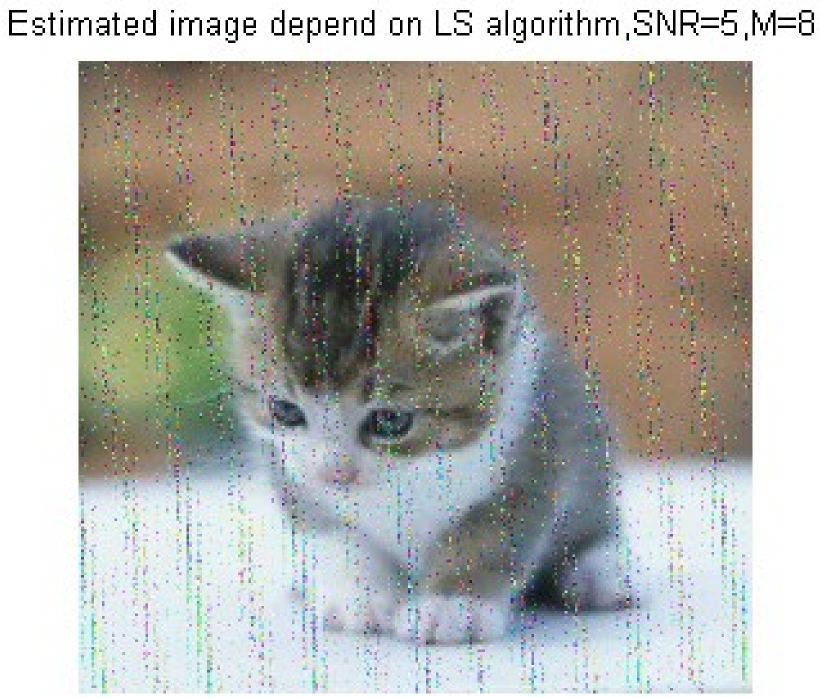

Figure 4.8 Color Image Estimated Depend on Least Square Estimator when Channel length =8 tabs and signal to noise ratio =5 dB

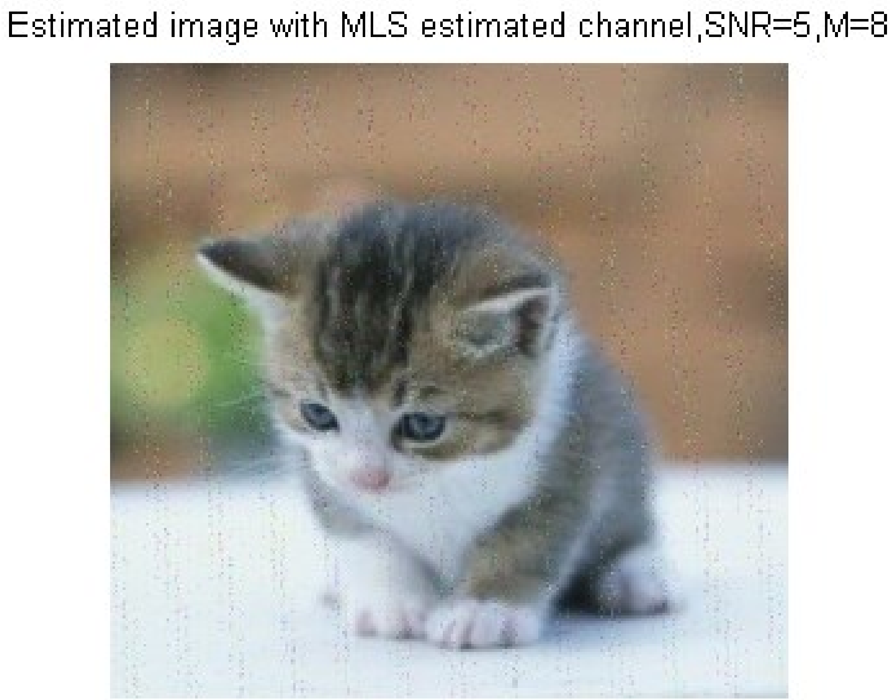

Figure 4.9 Color Image Estimated Depend on Proposed Estimator when Channel length =8 tabs and signal to noise ratio =5 dB

Figure 4.1 illustrates the original Color image (transmitted data) and their composition Red, Green and Blue images. The figures (4.2-4.9) show some of the simulation results for the modified least square MLS (proposed estimator) and least square LS estimators performance for each frame at SNR equal to 5dB and channel taps (M=8). Proposed estimator performs better than least square estimator at low signal to noise ratio as shown in the above images, also bit error rate curves that has been plotted in figure 4.10 shows the difference between estimators performance. The proposed estimator shows a perfect estimation at all values of signal to noise ratio SNR and as shown in the figure 4.10, the curve of the proposed algorithm is nearly identical to the curve of the data estimation which depended on the original channel.

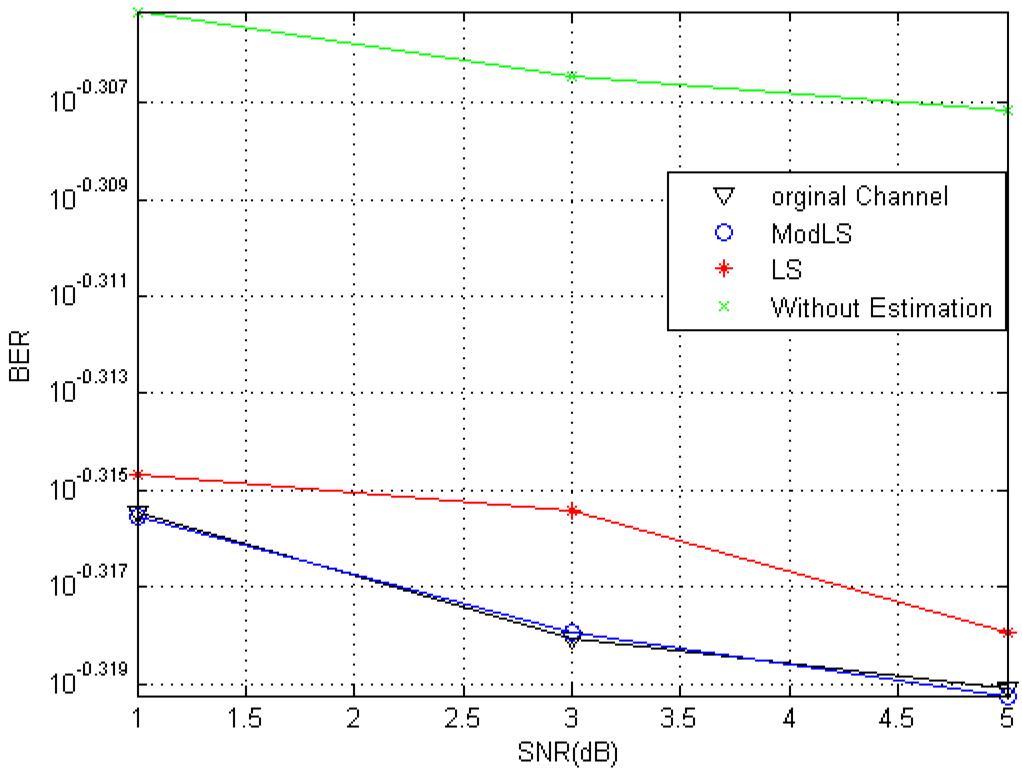

Figure 4.10 BER Performance of the Proposed (ModLS) and Least Square LS Algorithms when Channel length M=8 tabs

The frequency-selective fading channel that we use in our MATLAB simulation program is Rayleigh distribution as shown below in figure 4.11. The Rayleigh distribution is suitable for the crowded areas channels as mentioned in the chapter 1.

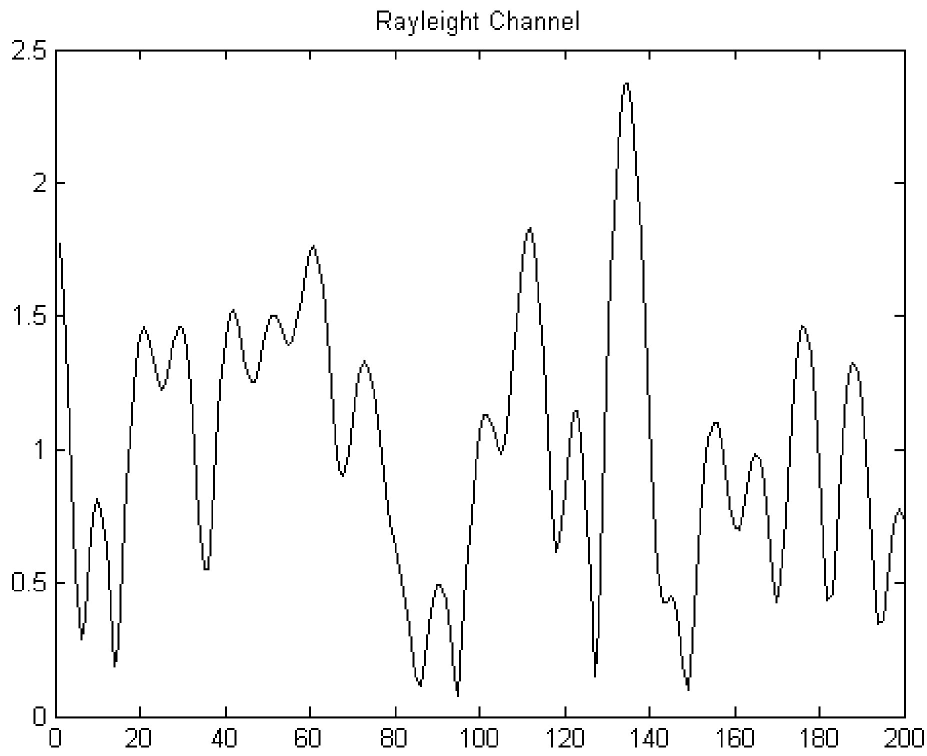

Figure 4.11 Rayleigh Fading Channel

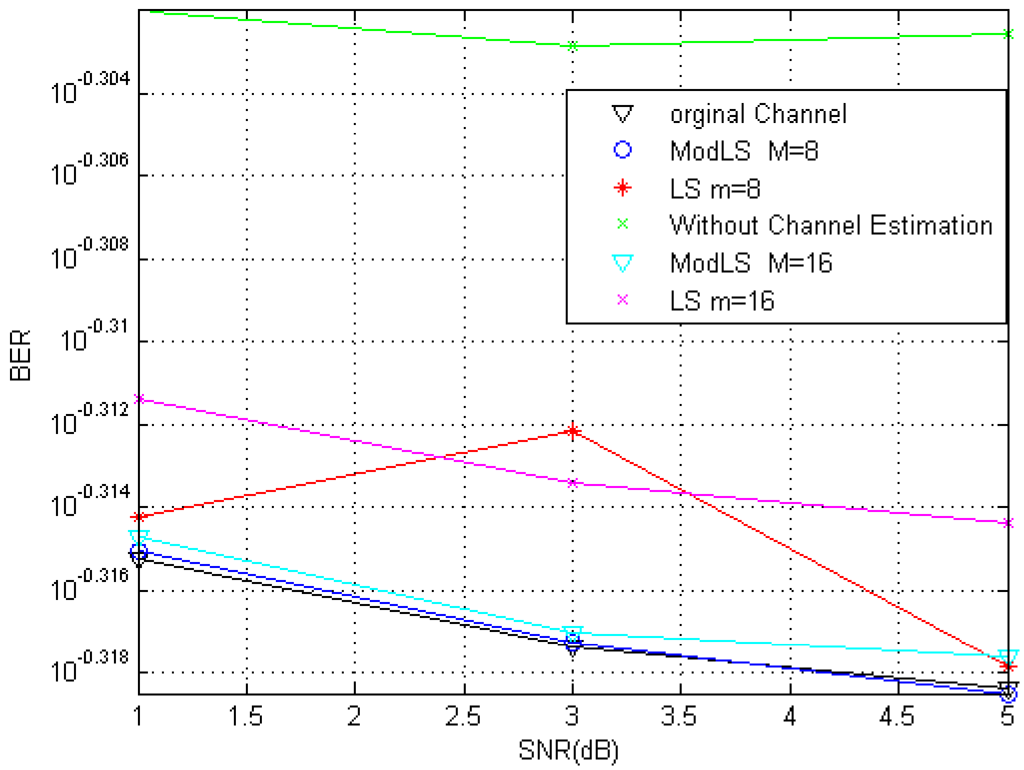

Figure 4.12 BER Performance of the Proposed (ModLS) and Least square LS Algorithms when Channel tabs M= (8,16)

Figure 4.12 shows the bit error BER performance of proposed (ModLS) and Least square LS estimators with different channel length (M=8, 16) and SNR= 5dB. Clearly, the proposed estimator has a good estimation especially with low signal to noise ratio better than least square LS estimator and the figures (4.13-4.16) below showing the difference between estimators performance with images when channel length M=16 and SNR= 5dB .

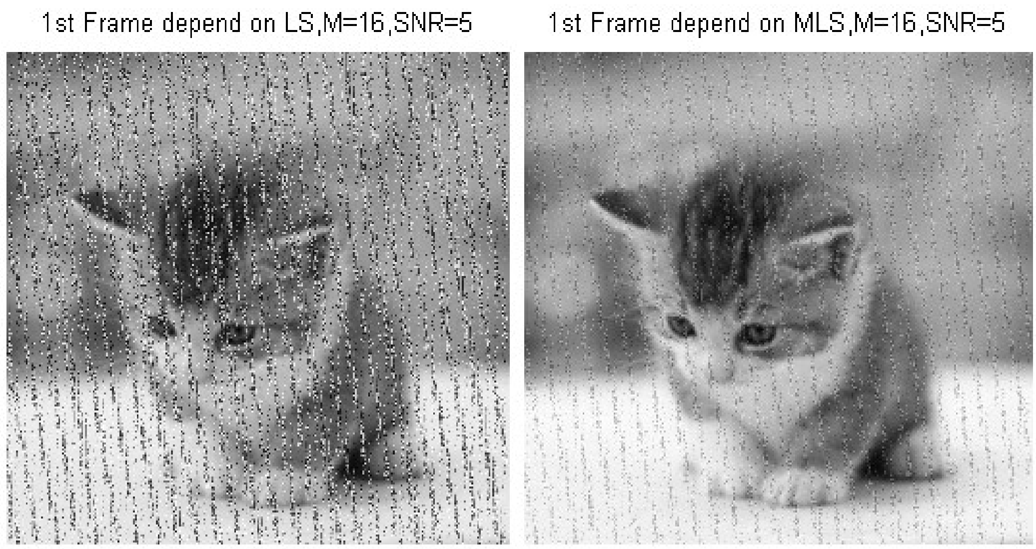

Figure 4.13 Difference between Proposed and Least Square Estimators in the 1st Frames when Channel tabs=16 and SNR=5dB

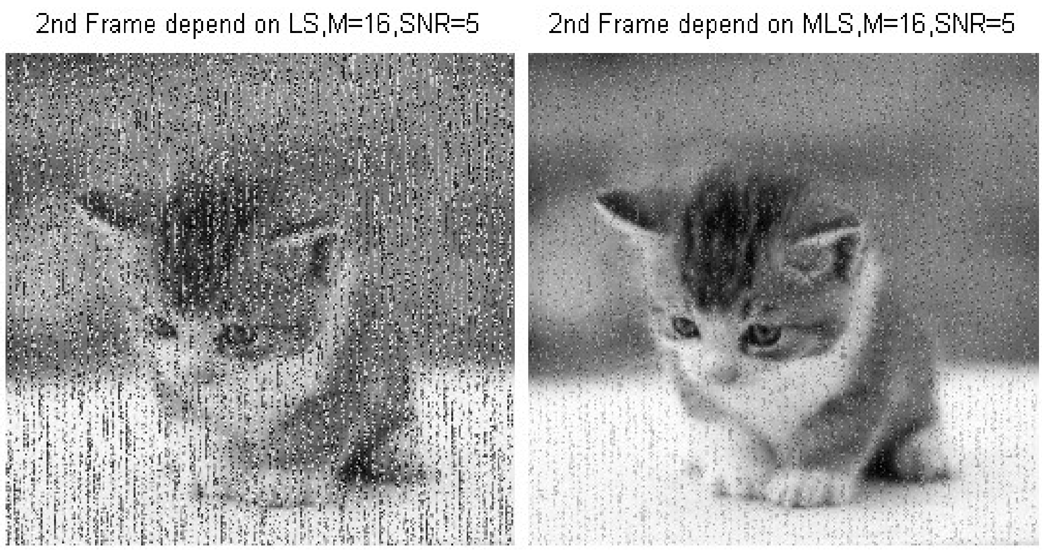

Figure 4.14 Difference between Proposed and Least Square Estimators in the 2nd Frames when Channel tabs=16 and SNR=5dB

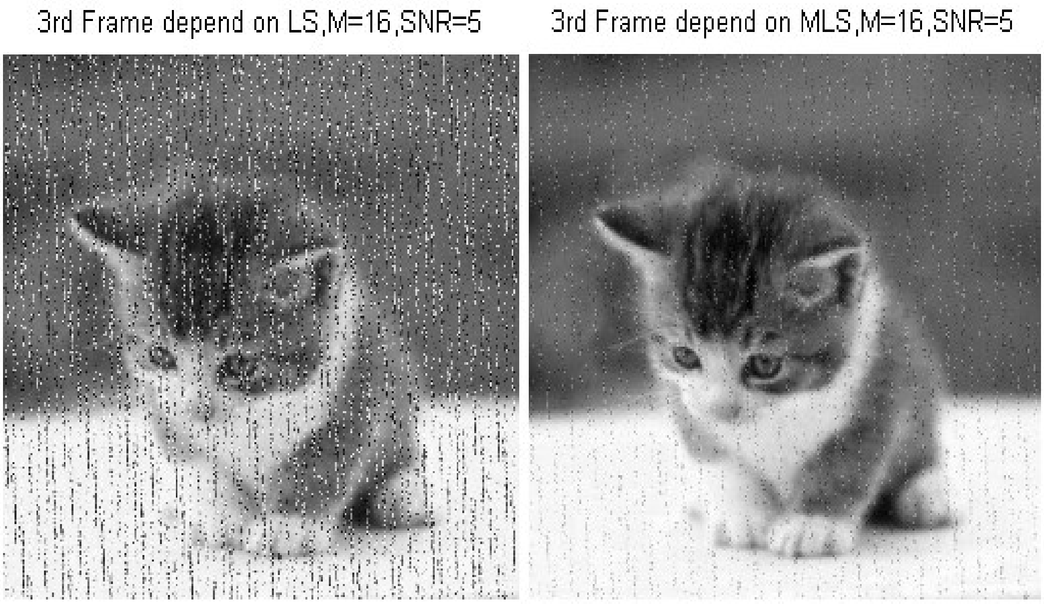

Figure 4.15 Difference between Proposed and Least Square Estimators in the 3rd Frames when Channel tabs=16 and SNR=5dB

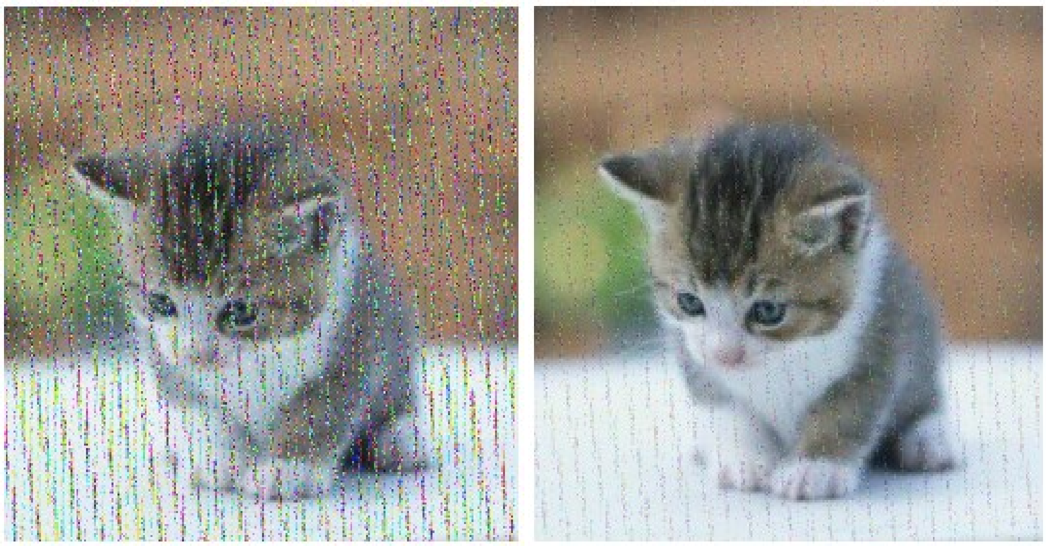

Figure 4.16 Difference between Proposed and Least Square Estimators in the Color Image when Channel tabs=16 and SNR=5dB

As shown below, figure 4.17 denotes the difference between two images. The image which on the left side has estimated without channel estimation, while the image which on right side has detected depending on the original channel and the bit error rate BER curves is shown in the figure 4.12.

Figure 4.17 Difference between Images Detection with/without Channel Estimation

Figure 4.18 Original Image and their Composition Red, Green and Blue Images

Figure 4.18 demonstrated the original Color image (transmitted data) and their composition Red, Green and Blue images.

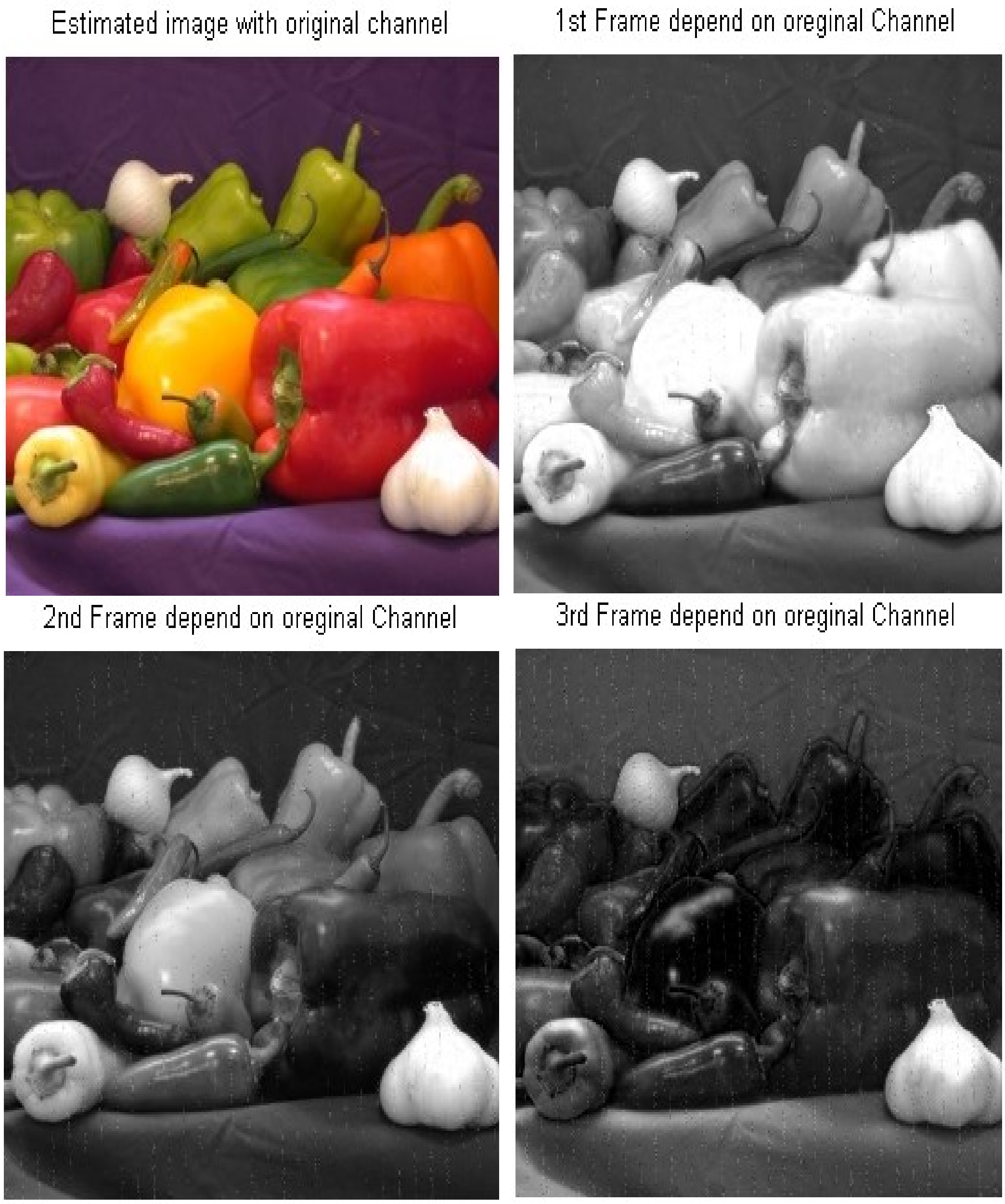

Figure 4.19 Estimated Image and their Composition Red, Green and Blue depend on Original Channel

Figure 4.19 shows the transmitted data has estimated depending on the original channel that means, the transmitted data (images) doesn't affected by the channel convolution but there are some bits in these images which have changed because of the effects of the additive white Gaussian noise AWGN.

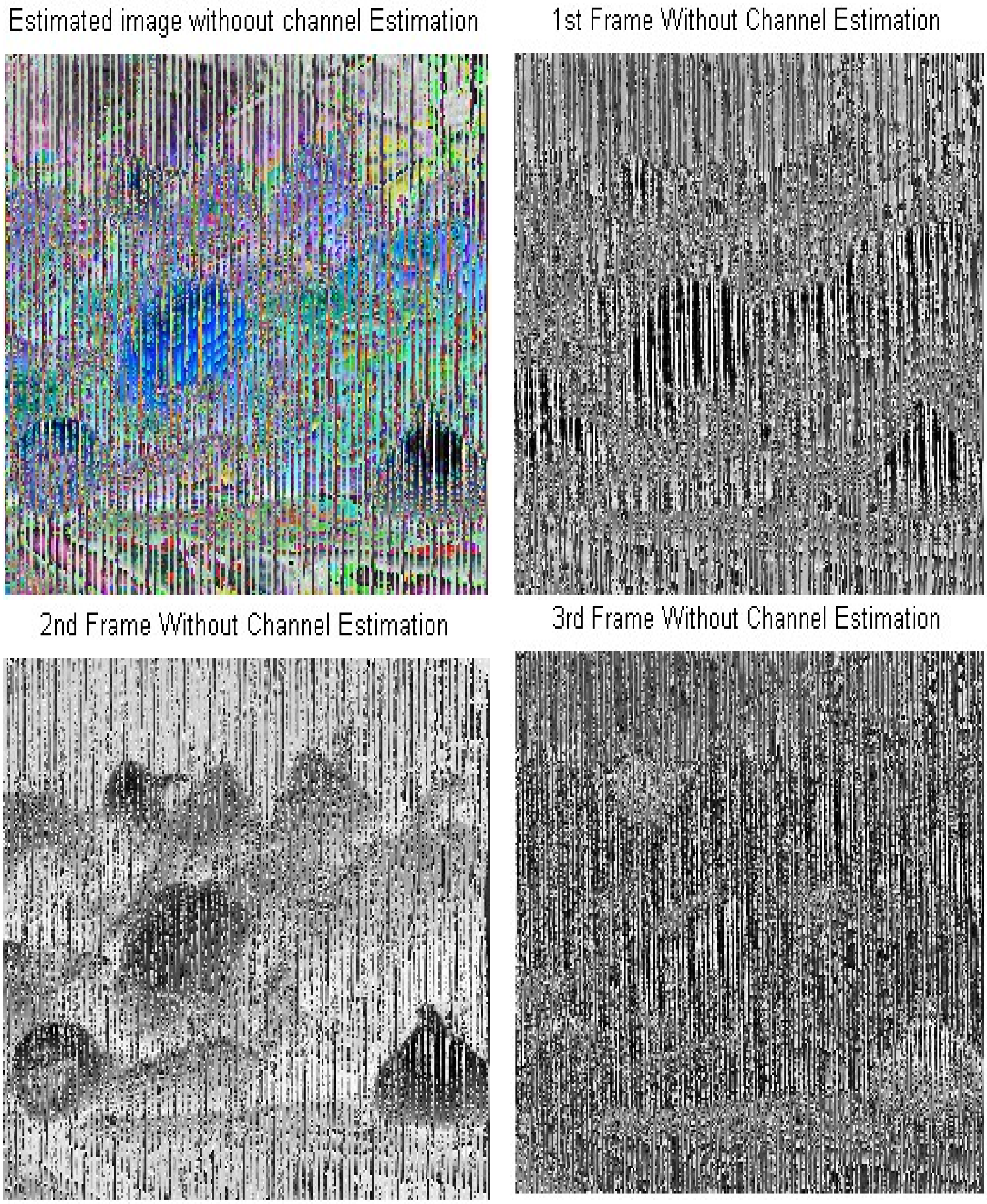

Figure 4.20 Estimated Image and their Composition Red, Green and Blue without Channel Estimation

Figure 4.20 shows the transmitted data has estimated without channel that means, the transmitted data (images) affected by the channel convolution effect plus the effects of the additive white Gaussian noise AWGN.

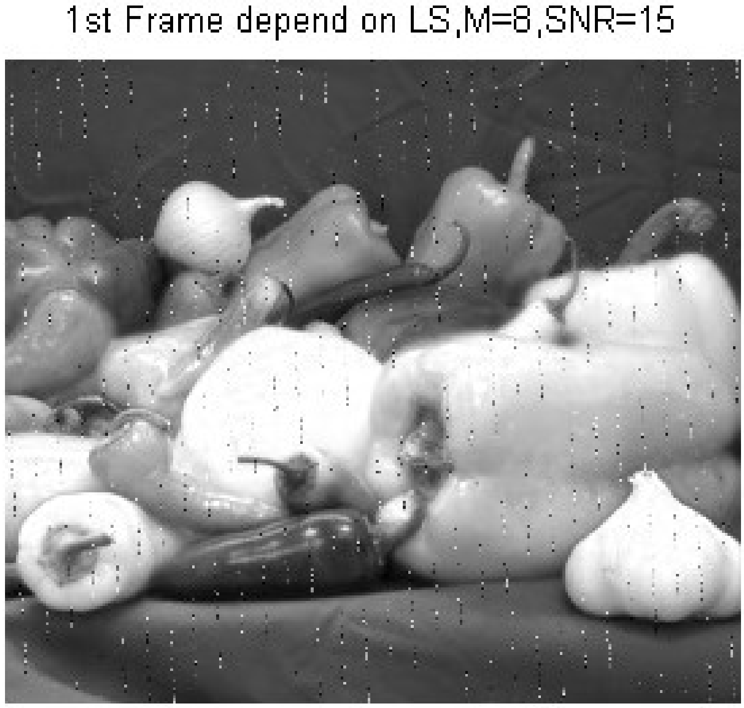

Figure 4.21 1st Frame (Red) Estimated Depend on Least Square Estimator when Channel length =8 tabs and signal to noise ratio =15 dB

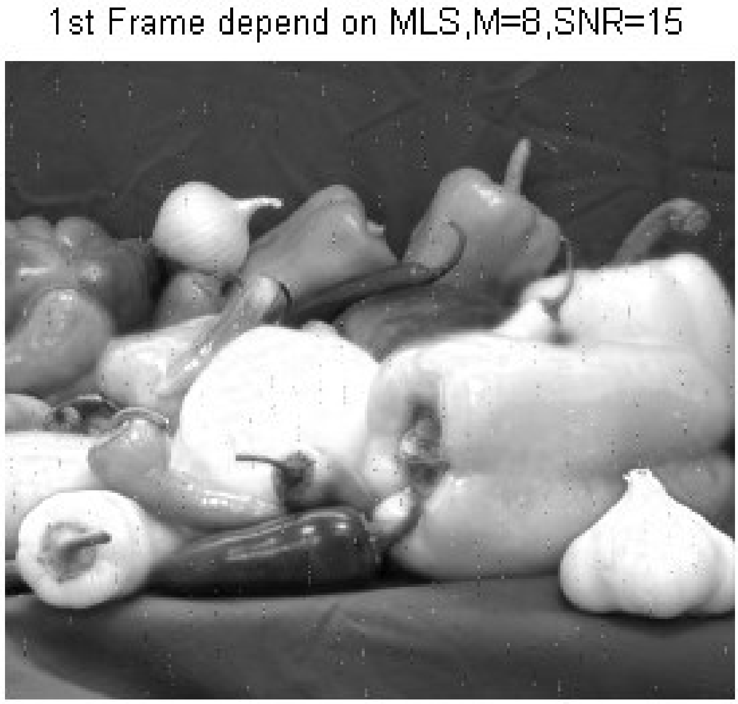

Figure 4.22 1st Frame (Red) Estimated Depend on Proposed Estimator when Channel length =8 tabs and signal to noise ratio =15 dB

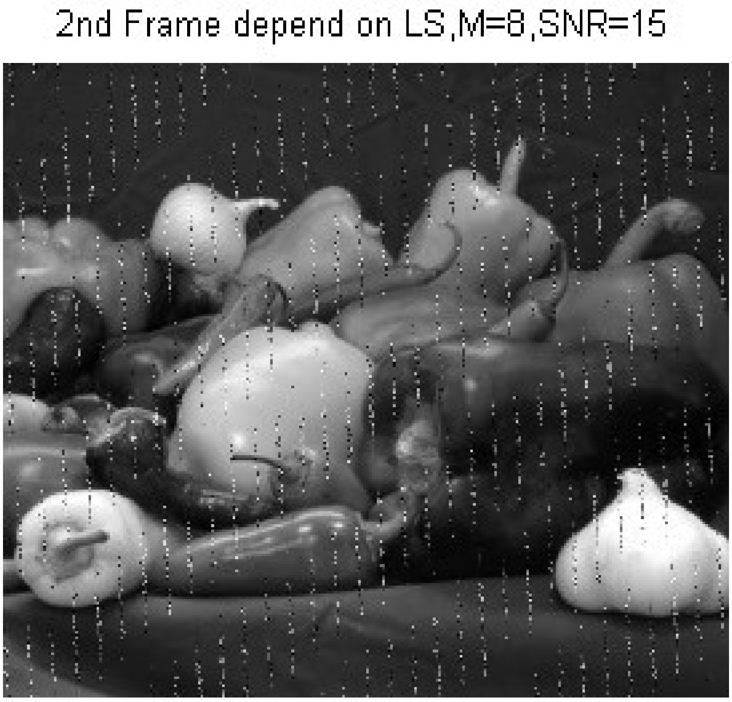

Figure 4.23 2nd Frame (Red) Estimated Depend on Least Square Estimator when Channel length =8 tabs and signal to noise ratio =15 dB

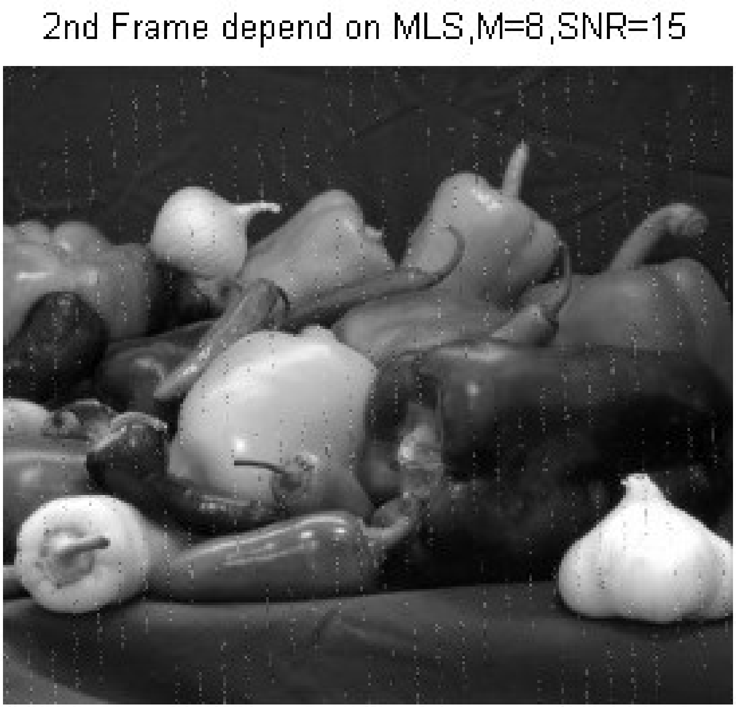

Figure 4.24 2nd Frame (Red) Estimated Depend on Proposed Estimator when Channel length =8 tabs and signal to noise ratio =15 dB

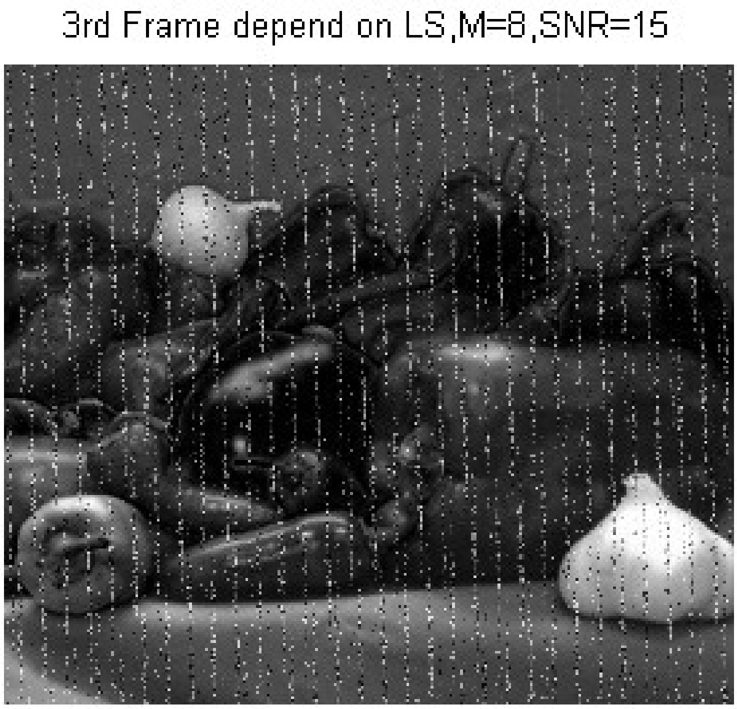

Figure 4.25 3rd Frame (Red) Estimated Depend on Least Square Estimator when Channel length =8 tabs and signal to noise ratio =15 dB

Figure 4.26 3rd Frame (Red) Estimated Depend on Proposed Estimator when Channel length =8 tabs and signal to noise ratio =15 dB

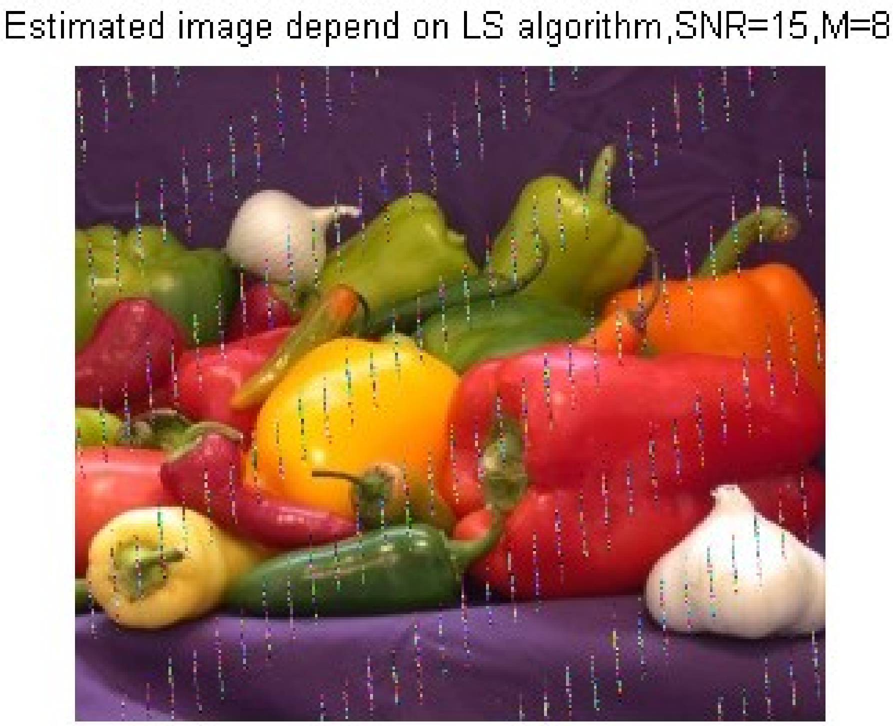

Figure 4.27 Color Image Estimated Depend on Least Square Estimator when Channel length =8 tabs and signal to noise ratio =15 dB

Figure 4.28 Color Image Estimated Depend on Proposed Estimator when Channel length =8 tabs and signal to noise ratio =15 dB

The figures (4.21-4.28) show the difference between estimators performance at high signal to noise ratio SNR=15dB and the channel length M=8 tabs. Proposed method has estimated channel and transmitted image very well for each transmission (each frame) better than least square algorithm. We also investigate the bit error performance with high signal to noise ratio and different channel tabs as shown below in the figure 4.29.

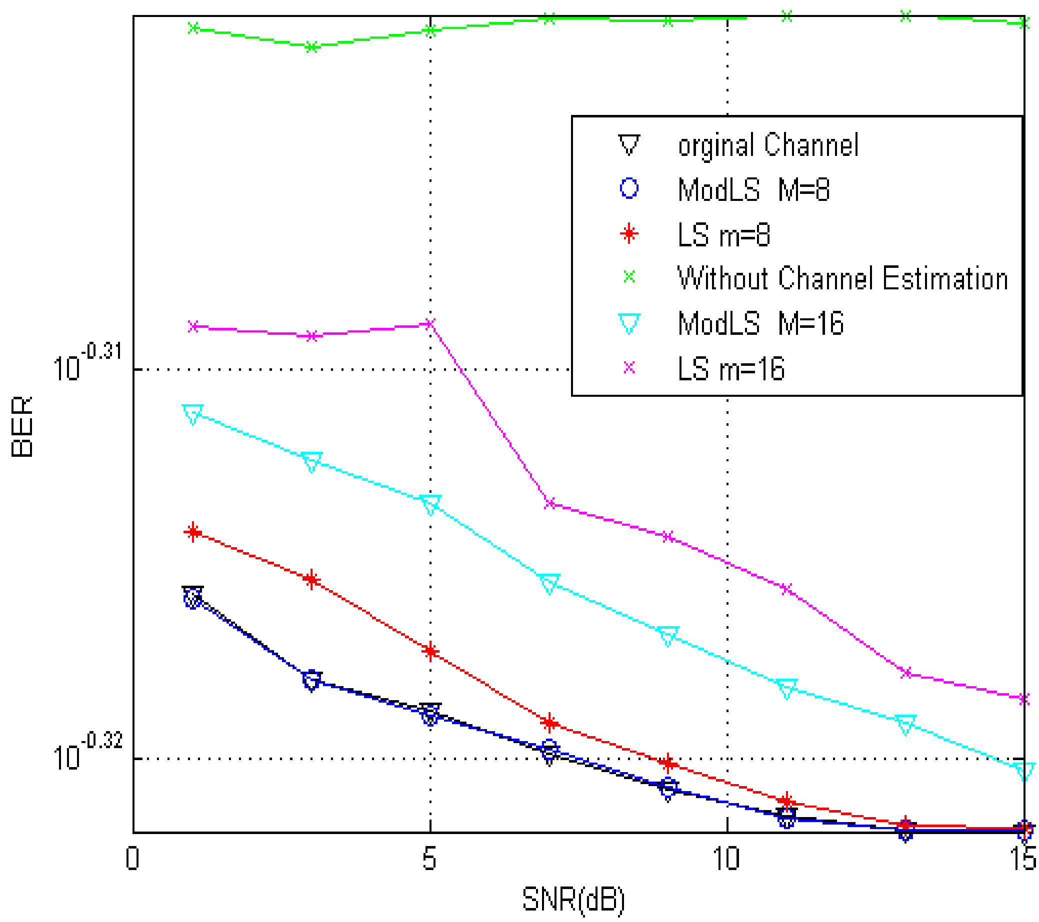

Figure 4.29 BER Performance of the Proposed (ModLS) and Least Square LS Algorithms at Different Channel length M=8, 16 tabs

In figure 4.29, naturally, if we have only a few channel tabs M=8, the estimators have performed better than in the case of the channel tabs M=16. As shown below, figure (4.30) shows difference between two images are detected with/without channel estimation. In the case of the channel length M=16 tabs also the proposed estimator has implemented better than least square estimator as shown below in figure (4.31).

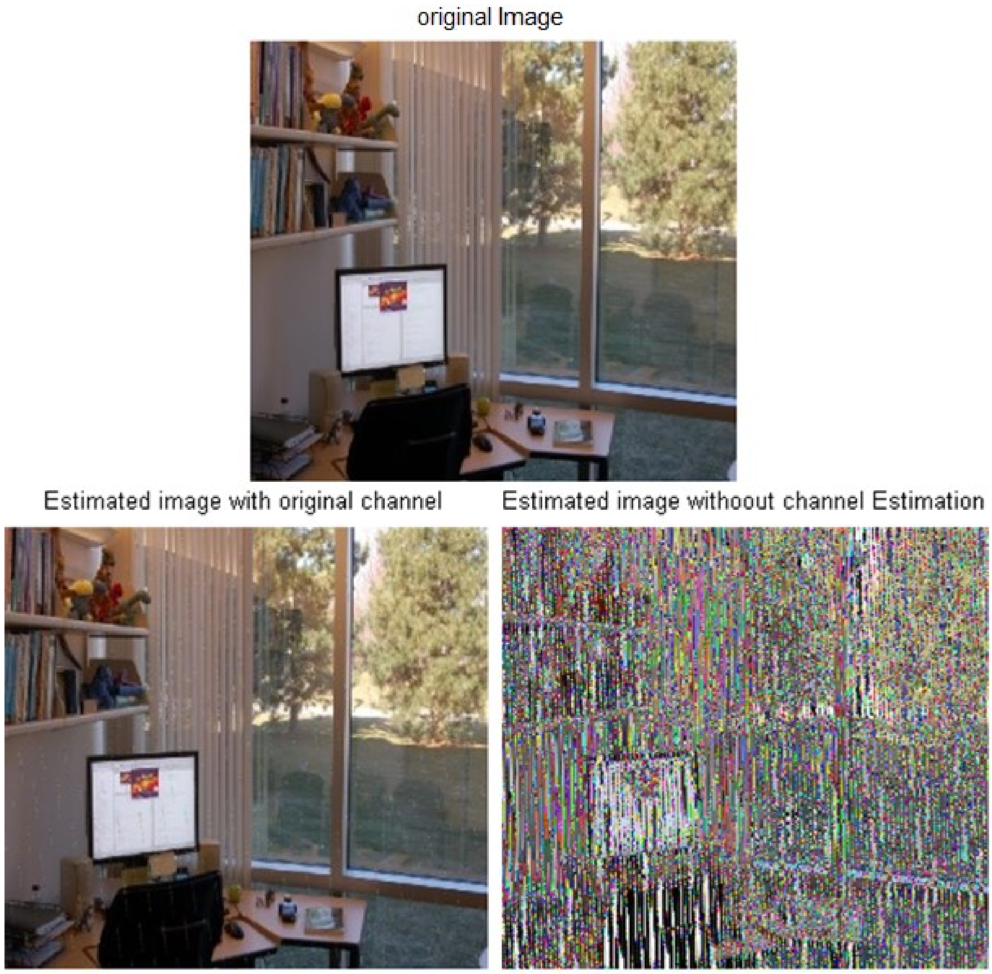

Figure 4.30 Original Image has Estimated depend on Original Channel and without Channel Estimation when Channel tabs M=16, SNR=15dB

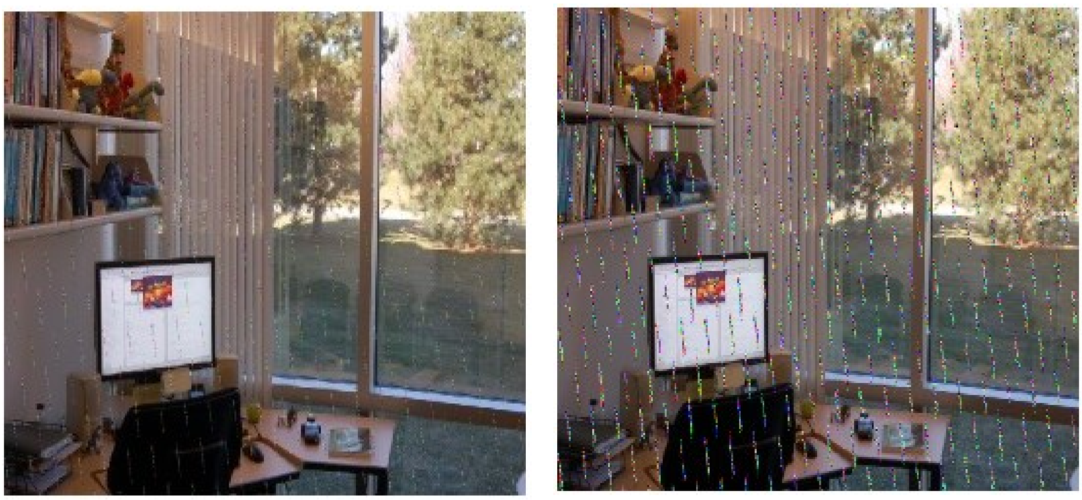

Figure 4.31 Left Image Estimated by using Proposed Estimator and Right Image Estimated by using Least Square Estimator when SNR=15dB,M=16

We also investigate channel estimation performance for the proposed and least square algorithms by increasing signal to noise ratio SNR=25 dB when channel length M=16 tabs. From the below figures (4.33-4.40) we observed that the channel estimation depend on proposed algorithms is more close to the original channel and much better than the channel has been estimated by the least square algorithm, as well we compared the bit error performance for the two estimators at SNR=25dB and channel tabs M=16 as shown in the figure (4.41).

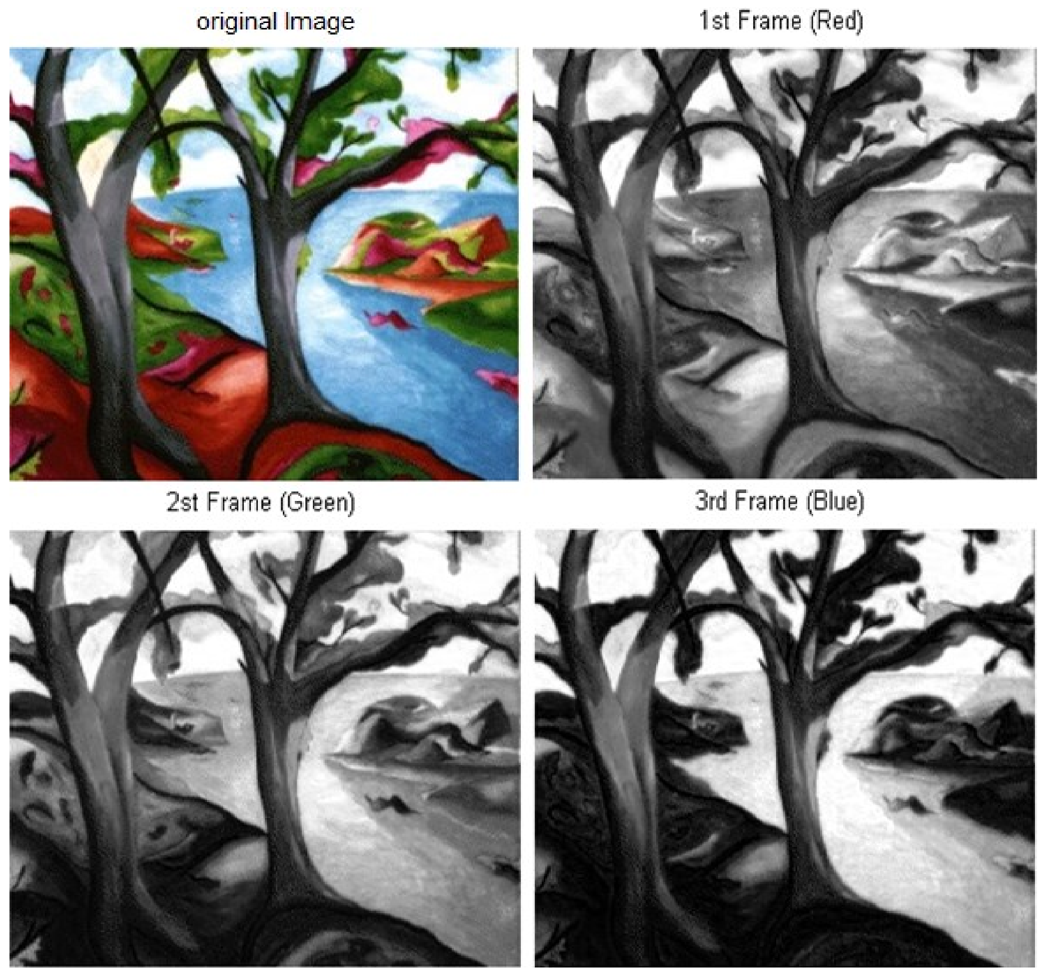

Figure 4.32 Original Image and their Composition Red, Green and Blue Images

Figure 4.32 shows transmitting data (color image) and their composition Red, Green and Blue images, these images are sending sequentially, frame by frame through Rayleigh channel.

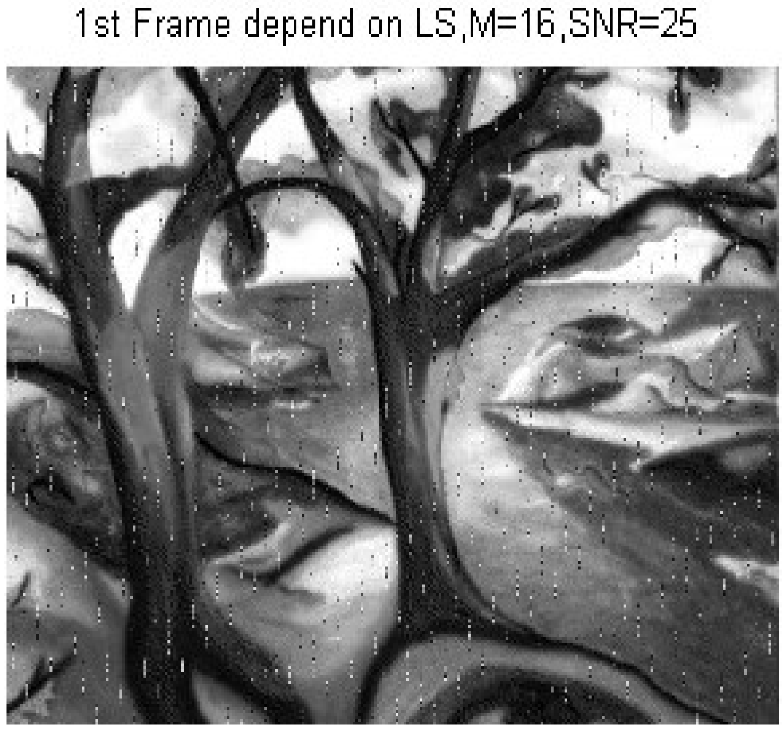

Figure 4.33 1st Frame (Red) Estimated Depend on Least Square Estimator when Channel length =16 tabs and signal to noise ratio =25 dB

Figure 4.34 1st Frame (Red) Estimated Depend on Proposed Estimator when Channel length =16 tabs and signal to noise ratio =25 dB

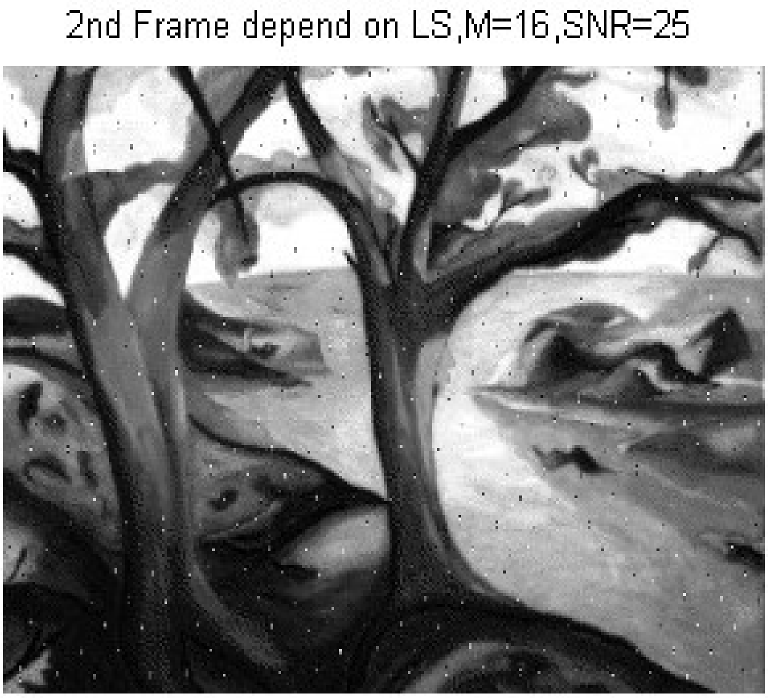

Figure 4.35 2nd Frame (Red) Estimated Depend on Least Square Estimator when Channel length =16 tabs and signal to noise ratio =25 dB

Figure 4.36 2nd Frame (Red) Estimated Depend on Proposed Estimator when Channel length =16 tabs and signal to noise ratio =25 dB

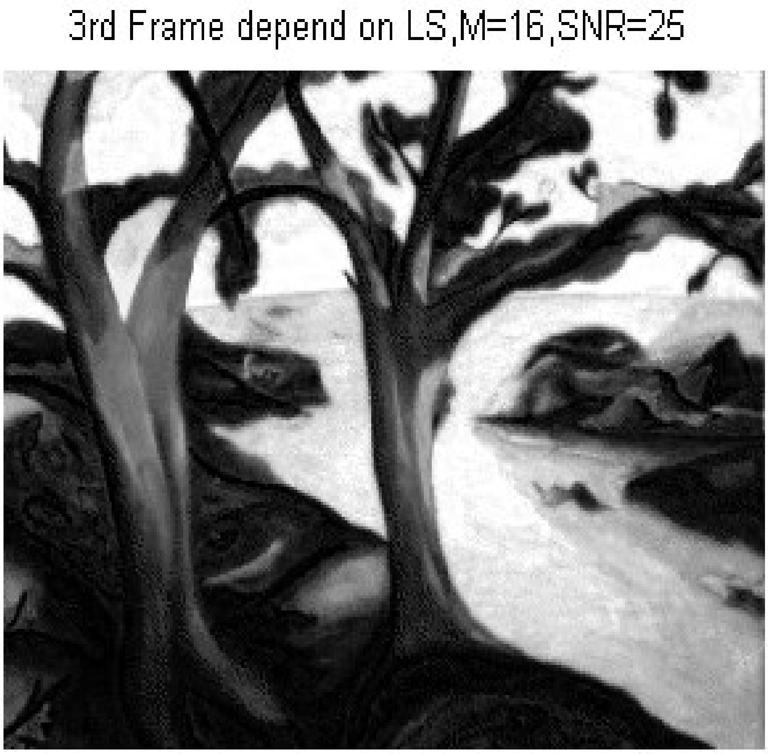

Figure 4.37 3rd Frame (Red) Estimated Depend on Least Square Estimator when Channel length =16 tabs and signal to noise ratio =25 dB

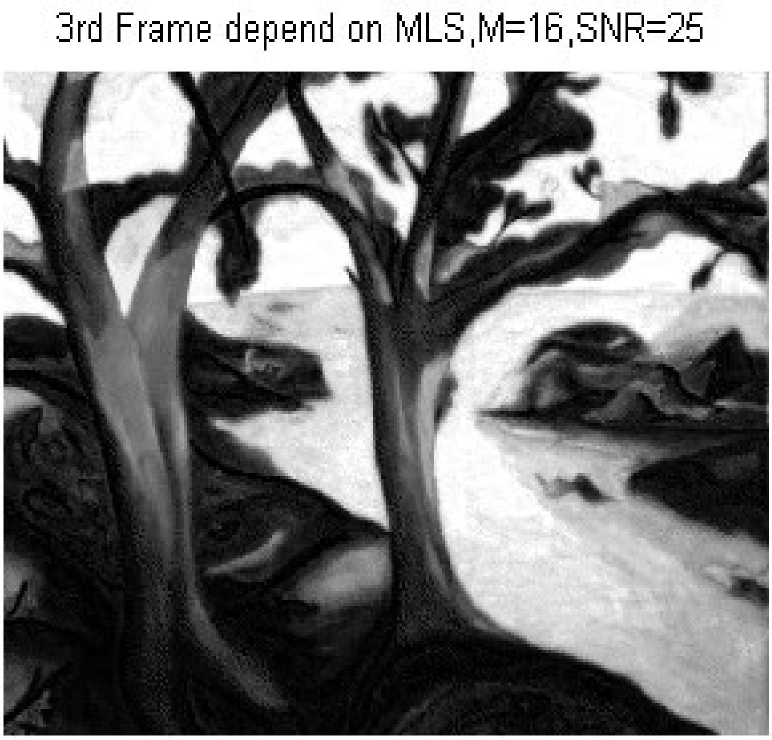

Figure 4.38 3rd Frame (Red) Estimated Depend on Proposed Estimator when Channel length =16 tabs and signal to noise ratio =25 dB

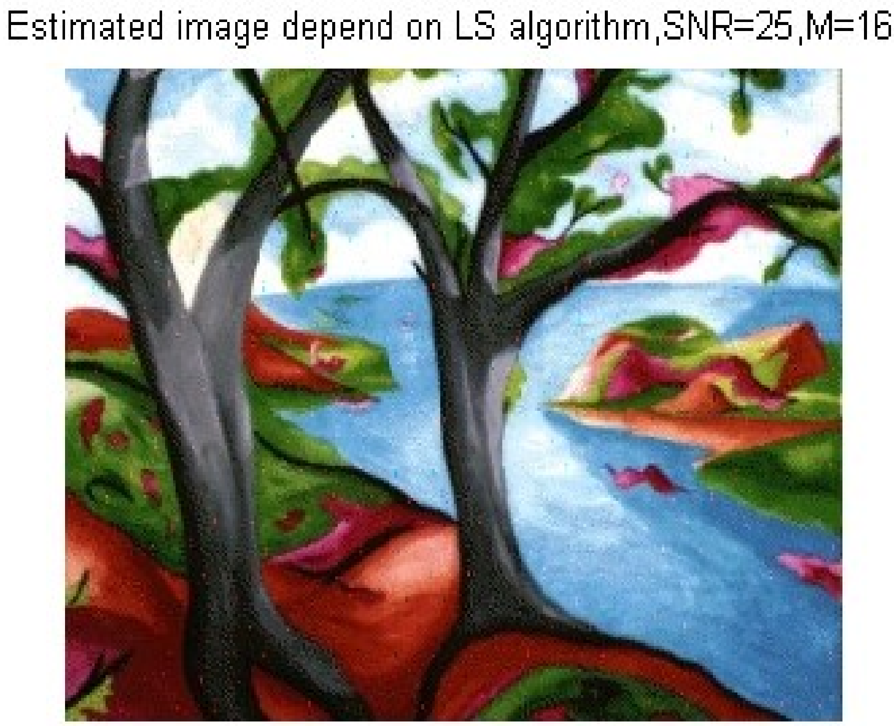

Figure 4.39 Color Image Estimated Depend on Least Square Estimator when Channel length =16 tabs and signal to noise ratio =25 dB

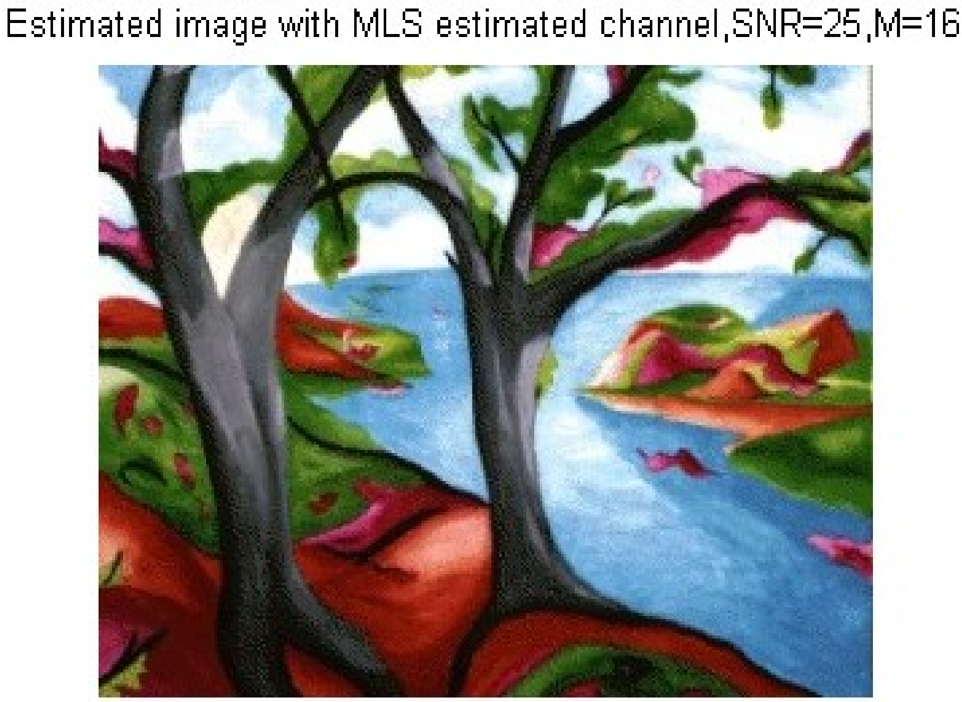

Figure 4.40 Color Image Estimated Depend on Proposed Estimator when Channel length =16 tabs and signal to noise ratio =25 dB

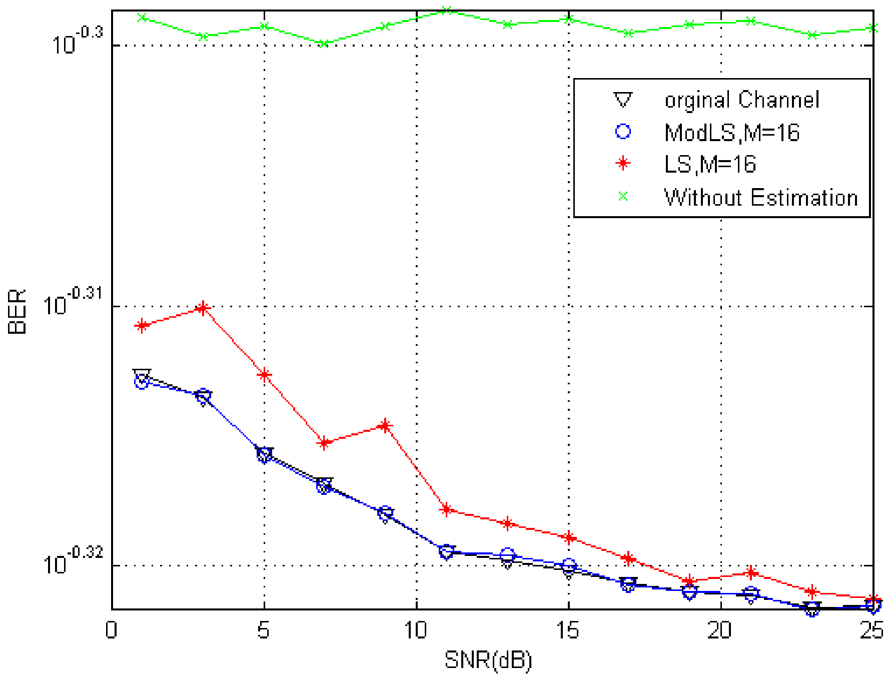

Figure 4.41 BER Performance of the Proposed (ModLS) and Least Square LS Algorithms when Channel length M=16 tabs

Figure 4.41 shows difference between two estimators, we observed that the proposed method works much better than least square estimator especially at low signal to noise ratio SNR, by this we achieved one of the our objectives.

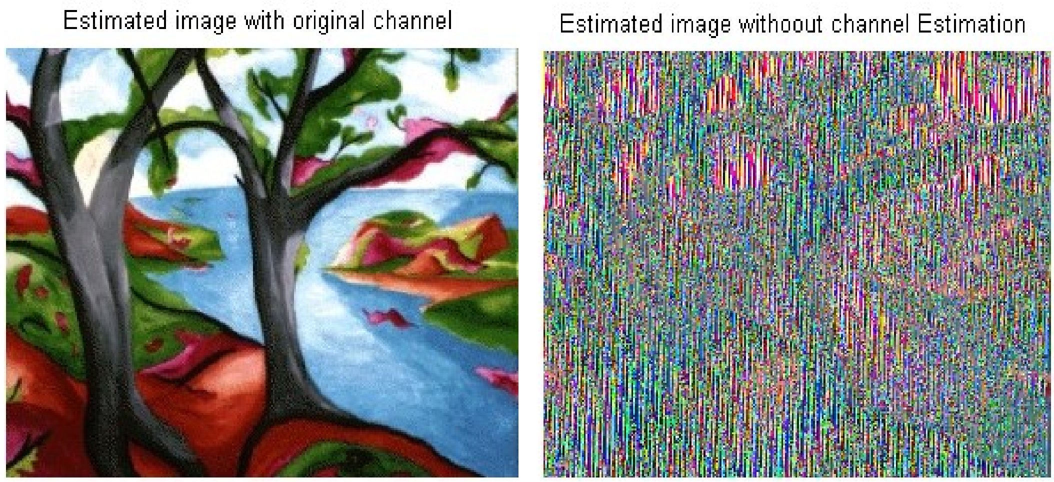

Figure 4.42 Original Image has Estimated depend on original channel and without Channel Estimation when Channel tabs M=16, SNR=15dB

Figure 4.42 shows two images have estimated with/without channel estimation.

CHAPTER 5

CONCLUSION AND FUTURE WORK

5.1 Conclusion

In this thesis, pilot-based channel estimation of OFDM system has been studied. Especially, we have focused on fast fading channel because fast fading channel represents crowded areas. Some basic methods, such as fast Fourier transform (FFT), guard interval (GI) and channel estimation has been analyzed and simulated for OFDM systems. FFT plays an essential part for decreasing the complexity of OFDM as a multi-carrier communication system. GI has been utilized for combating the inter-symbol-interference (ISI). Channel estimation is necessary to estimate the transmitting data. Least square and modified least square (proposed method) estimators with time domain interpolation have been simulated. The simulation results showed that the performance of the proposed estimator was superior to least square estimator especially in low signal to noise ratio SNR. The proposed algorithm showed good results with high FFT size, high channel taps and high Doppler frequency comparing with least square algorithm. The advantage of proposed algorithm is simple and low-complexity approach for the OFDM. Also, it reduced the number of pilot tones in each OFDM symbols to increase data rate without need to extend the bandwidth. It improved BER performance at different signal to noise ratios.

5.2 Future Work

Based on this thesis, there are wide scopes for further research as follows:

1. Modify other types of non-blind algorithms, such as minimum mean-square error (MMSE) and least minimum mean-square error (LMMSE) and compare the result with our proposed algorithm.
2. Find an alternative ways to rearrange pilot tones in OFDM symbol to increase system efficiency.

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PUPLICATION:

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