Islamic University of Technology (IUT)

PILOT BASED CHANNEL ESTIMATION IN OFDM SYSTEMS

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A Dissertation

In Partial Fulfillment of the Requirement for the Degree of
Bachelor of Science in Electrical and Electronic Engineering
Academic Year: 2014-2015

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Islamic University of Technology (IUT)
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November - 2014

Declaration

This is to certify that the project entitled "**PILOT BASED CHANNEL ESTIMATION IN OFDM SYSTEMS"** is supervised by Dr. Mohammad Rakibul Islam. This project work has not been Submitted anywhere for a degree.

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Dedicated To Our Parents

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ACKNOWLEDGEMENTS

All thanks and praise for Allah (SWT) who has enabled us finishing this thesis work. Secondly, we would like to express our gratitude to our thesis supervisor Dr. Md. Rakibul Islam for his relentless support, direction and inspiration without which this thesis work could not be perfectly accomplished. We also would like to thank our classmates and the staff members of the EEE department for assisting us in different ways even if it is sharing of a single information. Finally, the deepest gratitude would go to our beloved family for their encouragement and support mentally and economically throughout this whole work.

ABSTRACT

OFDM (Orthogonal Frequency Division Multiplexing) is becoming widely applied in wireless communications systems due to its high rate transmission capability with high bandwidth efficiency and its robustness with regard to multi-path fading and delay. High data rate and speedy communication has become the ultimate goal of any communication research work. Multiplexing of signal bits is an important part of the wireless communication system.

This thesis paper represents a total overview of OFDM system, basic principles of OFDM system, OFDM generation and reception. Though it has some salient and supreme features over other multiplexing techniques, still has some drawbacks which limit its performance in some cases.

Researches on the communications prove that to reduce the drawbacks of the overall system, the channel estimation process is an essential tool. Different channel estimation methods have been developed and applied in field presently. In this thesis work the current channel estimation methods are compared and the best possible method is proposed.

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List of Abbreviations

OFDM	Orthogonal Frequency Division Multiplexing
LPF	Low Pass Filter
DAC	Digital to Analog Converter
ADC	Analog to Digital Converter
GSM	Global System for Mobile Communication
UMTS	Universal Mobile Telecommunication System
GPRS	General Packet Radio Service
ITU	International Telecommunication Union
OLR-MMSE	Optimum Low-Rank Minimum Mean Square-Error Estimator
PCMB	Parametric Channel Modelling Based-Estimator

Chapter 1

Introduction

For the last few years, the advancement in wireless communication technology has been shooting up like a rocket to the sky. It is true that the extraordinary communication system is among the some of the few key issues for which the civilization has climbed up and is exploring at its peak now. Today's age is age of information technology and communication which could not be imagined except the radical change in communication technology.

The history of wireless communication technology is not very ancient one. The journey of it started at the mid of previous century. Gradually and rapidly this technology tree has enhanced its branches with immense opportunities like leaves and flowers and human civilization has started to enjoy its fruits. The evolution state is now transiting between the third and the fourth generation. Figure 1.1 shows the data rates of previous, current and future wireless communication technology [1]

In this introduction, we will try to have a glimpse over the major breakthroughs of wireless communication evolution.

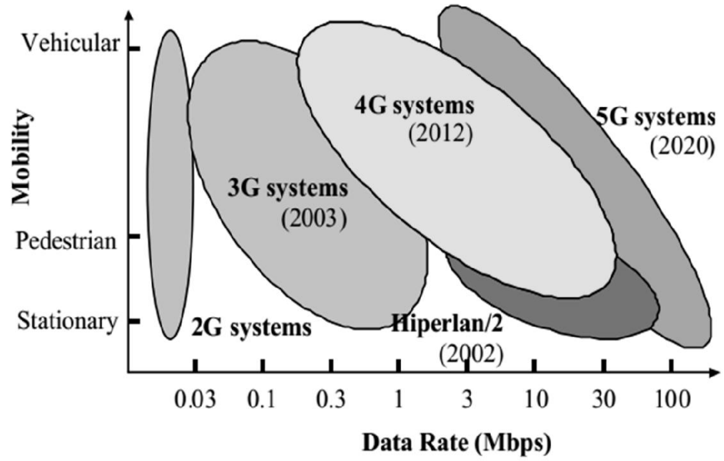


Figure 1.1: Previous, Current &Future Generation Mobile System

1.1 First Generation Wireless System

First-generation mobile systems used analog transmission for speech services. The two most popular analog systems were Nordic Mobile Telephones (NMT) and Total Access Communication Systems (TACS). Other than NMT and TACS, some other analog systems were also introduced in 1980s across the Europe. All of these systems offered handover and roaming capabilities but the cellular networks were unable to interoperate between countries. This was one of the undesired disadvantages of first-generation mobile networks. In the United States, the Advanced Mobile Phone System (AMPS) was launched in 1982. The system was allocated a 40-MHz bandwidth within the 800 to 900 MHz frequency range by the Federal Communications Commission (FCC) for AMPS. In 1988, an additional 10 MHz bandwidth, called Expanded Spectrum (ES) was allocated to AMPS. AMPS and

TACS use the frequency modulation (FM) technique for radio transmission. Traffic is multiplexed onto an FDMA (frequency division multiple access) system [2-3].

1.2 Second Generation Wireless System

Second-generation (2G) mobile systems were introduced in the end of 1980s. Low bit rate data services were supported as well as the traditional speech service. Compared to first-generation systems, second-generation (2G) systems use digital multiple access technology, such as TDMA (time division multiple access) and CDMA (code division multiple access). Consequently, compared with first-generation systems, higher spectrum efficiency, better data services, and more advanced roaming were offered by 2G systems. In Europe, the Global System for Mobile Communications (GSM) was deployed to provide a single unified standard. Global System for Mobile Communications, or GSM, uses TDMA technology to support multiple users During development over more than 20 years, GSM technology has been continuously improved to offer better services in the market.

The next advancement in the GSM system was the addition of two platforms, called Voice Mail Service (VMS) and the Short Message Service Centre (SMSC). The move into the 2.5G world began with General Packet Radio Service (GPRS). GPRS is a radio technology for GSM networks that adds packet-switching protocols, shorter setup time for ISP connections, and the possibility to charge by the amount of data sent, rather than connection time. GPRS is the most significant step towards 3G. GSM and EDGE (Enhanced Data rates in GSM Environment) with both voice and data traffic moving on the system, the need was felt to increase the data rate. Thus the advent of third generation became important.

1.3 Third Generation Wireless System

The International Telecommunication Union (ITU) defined the demands for 3G mobile networks with the IMT-2000standard. An organization called 3rd Generation Partnership Project (3GPP) has continued that work by defining a mobile that fulfills the IMT-2000 standard. In Europe it was called UMTS (Universal Mobile Telecommunication System), which is ETSI-driven.

IMT2000 is the ITU-T name for the third generation system, while cdma2000 is the name of the American 3G variant. WCDMA is the air-interface technology for the UMTS. 3G networks enable network operators to offer users a wider range of more advanced services while achieving greater network capacity through improved spectral efficiency. Services include wide-area wireless voice telephony, video calls, and broadband wireless data, all in a mobile environment.

Additional also include HSPA(High Speed Packet Access) data transmission capabilities able to deliver speeds up to 14.4 Mbps on the downlink and 5.8 Mbps on the uplink. The first commercial 3G network was launched by NTT DoCoMoin Japan branded FOMA, based on W-CDMA technology on October 1, 2001 [4]. In fact, most of these applications will not be limited by the data rate provided by 3G systems, but by the cost of the service.

1.4 Fourth Generation System & Beyond

Research are going on the development of 4th generation (4G) mobile communication systems. The commercial rollout of these systems is likely to begin around 2008 - 2012, and will replace 3rd generation technology. Few of the aims of 4G networks have yet published, however it is likely that they will be to extend the capabilities of

3G networks, allowing a greater range of applications, and improved universal access. Ultimately 4G networks should encompass broadband wireless services.

1.5 History of OFDM

The origins of OFDM development started in the late 1950's [5] with the introduction of Frequency Division Multiplexing (FDM) for data communications. In1966 , the structure of was patented by Chang [6] and published [7] the concept of using orthogonal overlapping multi – tone signals for data communications. In 1971,Weinstein [8] introduced the idea of using a Discrete Fourier Transform (DFT) for of the generation and reception of OFDM signals, eliminating the requirement for banks of analog subcarrier oscillators. This presented an opportunity for an easy implementation of OFDM, especially with the use of Fast Fourier Transforms (FFT), which are an efficient implementation of the DFT. This suggested that the easiest implementation of OFDM is with the use of Digital Signal Processing(DSP), which can implement FFT algorithms. It is only recently that the advances in integrated circuit technology have made the implementation of OFDM cost effective. reliance on DSP prevented the widespread use of OFDM during the early development of OFDM. It wasn't until the late 1980's that began on the development of OFDM for commercial use, with the introduction of the Digital Audio Broadcasting (DAB) system.

1.6 Thesis Layout

This thesis comprises of six chapters.

Chapter 1 represents the background of the present work, motivation, objectives and related work with this thesis.

Chapter 2 discusses about fundamental principles of orthogonal frequency division multiplexing technique and characteristics of this modulated transmitted signal.

Chapter 3

CHAPTER 2

Basic Principles of OFDM

2.1 What is OFDM

Orthogonal Frequency Division Multiplexing (OFDM) is almost identical to the popular and used technique of Frequency Division Multiplexing (FDM). OFDM uses the proposition of FDM to entitle several messages to be sent over a single radio channel. It is however in a much more dominated manner, permitting an enhance spectral efficiency.

A general example of FDM is the use of various frequencies for each frequency modulation radio stations. Because of using different transmit carrier frequencies, all stations transmit at the same time but do not constrain with each other. Moreover, they are bandwidth bounded and are spaced adequately distant in frequency so that the signals processed do not overlap in the frequency domain. At the receiver side, each signal is especially received by using a frequency adjustable band pass filter to precisely remove all the signals except for the station of concern. This filtered signal can then be demodulated to recover the original processed data.

OFDM is unlike to FDM in various ways. In typical condition broadcasting each radio station permits on a different frequency, efficiently using FDM to maintain a distinction between the stations. There is however no synchronization between these stations. With an OFDM transmission such as Digital Audio Broadcasting (DAB), the information signal from several stations is combined into a single multiplexed stream of data. This data is then processed using an OFDM ensemble that is made up from a concentrated packing of many subcarriers. All the subcarriers within the OFDM signal are time and frequency synchronized to each other, permitting the intercession between subcarriers to be carefully controlled.

Due to the orthogonal essence of the modulation, do not cause Inter-Carrier Interference (ICI) yet these several subcarriers overlap in the frequency domain. Usually with FDM the transmission signals need to have a large frequency guard-band between channels for preventing intercession. This shrinks the inclusive spectral efficiency. Anyhow with OFDM the orthogonal packing of the subcarriers significantly lowers this guard band, enhancing the spectral efficiency. All the wireless telecommunication systems use a modulation scheme to map the data signal to a form that can be especially processed over the communications channel.

With the most acceptable one, a wider range of modulation arrangements has been improved, determined on if the signal information is an analogue waveform or a digital one. Some of the regular analogue modulation arrangements contain Frequency Modulation (FM), Amplitude Modulation (AM), Phase Modulation (PM), Single Side Band (SSB), Vestigial Side Band (VSB), Double Side Band Suppressed Carrier (DSBSC) [9], [10]**.** Whereas digital communications include, Amplitude Shift Keying (ASK), Frequency Shift Keying (FSK), Phase Shift Keying (PSK) and Quadrature Amplitude Modulation (QAM) [9] – [11]

In a FDM transmission, each of the carriers can use both an analogue or digital modulation scheme. There is no synchronization between the transmission and so from one side it could be transmitted using FM and another in digital using FSK. In a single OFDM transmission all the subcarriers are synchronized to each other, regulating the transmission to digital modulation schemes.

OFDM is symbol based, and can be thought of as a large number of low bit rate carriers transmitting in parallel. To clinch that the orthogonal essence of the formation is sustained, all the carriers transmit in simultaneous using synchronized time and frequency, appearing a single block of spectrum. As these several carriers form a single OFDM transmission, they are recurrently represented to as 'subcarriers', with the terminology of 'carrier' retained for set out the RF carrier interlacing the signal from base band.

Orthogonality will be maintained if they are mutually substantive to each other. Orthogonality is a property that permits several data signals to be processed accurately over a regular channel and identified, without intercession. Dropping of orthogonality effects in obscuring between these information signals and desensitizing in communications. Several regular multiplexing arrangements are natively orthogonal. Time Division Multiplexing (TDM) permits transmission of several data signals over a single channel by allocating distinctive time slots to each separate data signal. In time of each time period only the signal from a single origin is processed intercepting any intercession between the multiple information sources. For the reason of this TDM is orthogonal in essence. In the frequency domain most FDM systems are orthogonal as each of the separate transmission signals are well expanded out in frequency intercepting intercession.

Since these methods are orthogonal, the terminology OFDM has been reserved for a special form of FDM. The subcarriers in an OFDM signal are expanded as well matched as is theoretically feasible while sustaining orthogonality between them. OFDM attains orthogonality in the frequency domain by assigning each of the distinct data signals onto various subcarriers. OFDM signals are done up from aggregate of sinusoids, with each accordance to a subcarrier. The baseband frequency of each subcarrier is selected to be an integer multiple of the inverse of the symbol time, developing in all subcarriers having an integer number of cycles per symbol.

As a consequence the subcarriers are orthogonal to each other. Figure 2-1 shows the formation of an OFDM signal with four subcarriers.

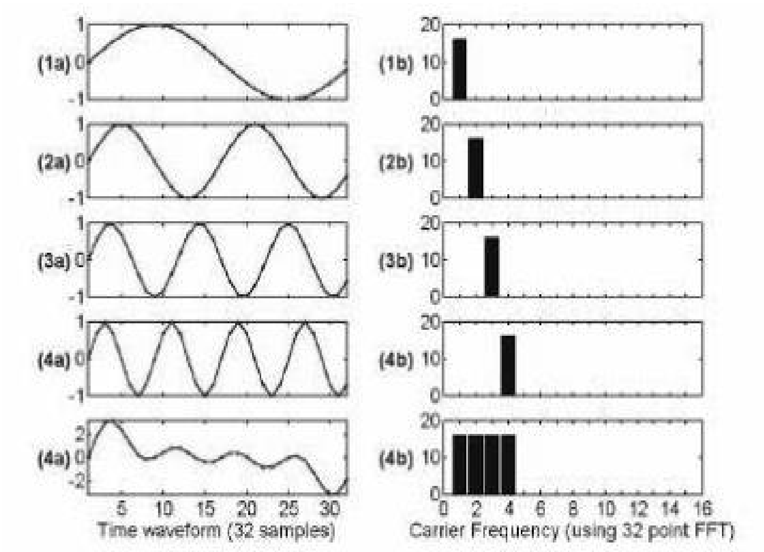


Figure 2-1: Time domain formation of an OFDM signal.

(1a), (2a), (3a) and (4a) show individual subcarriers, with 1, 2, 3, and 4 cycles per symbol accordingly. The phase on all these subcarriers is zero.

(1b), (2b), (3b) and (4b) show the FFT of the time waveforms in (1a), (2a), (3a) and (4a) accordingly.

(4a) and (4b) shows the result for the summation of the 4 subcarriers.

If any two different functions within the set are multiplied, and integrated over a symbol period, then the result will be zero, for orthogonal functions.

Further way of reasoning of this is that, at a matched receiver for one of the orthogonal functions, a subcarrier in the case of OFDM, then the receiver will only see the result for that function. The results from all other functions in the set integrate to zero, and hence have no effect.

Sets of functions are orthogonal to each other if they match the criterions in equation (2-1).

$$\int_0^T s_i\,(t)s_j(t)dt = \begin{cases} C \; i = j \\ 0 \; i \neq j \end{cases} \qquad (2\text{-}1)$$

Equation (2-2) shows a set of orthogonal sinusoids, which represent the subcarriers for an unmodulated real OFDM signal.

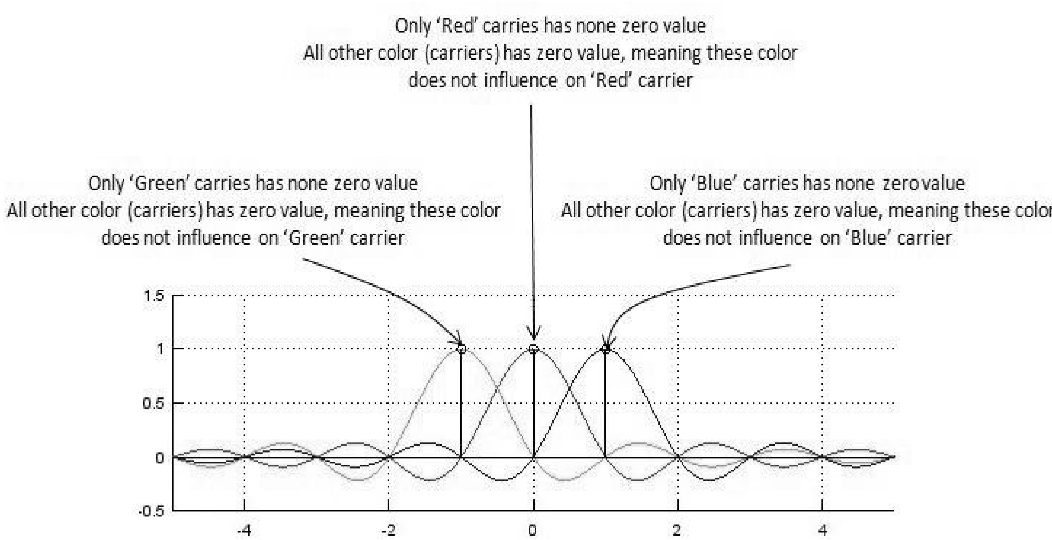
$$s_k(t) = \begin{cases} \sin(\,2\pi k f_0 t) & 0<t<T \quad k=1,2,\ldots M \\ 0 & otherwise \end{cases} \qquad (2\text{-}2)$$

where, f_o is the carrier spacing, M is the number of carriers, T is the symbol period. Since the highest frequency component is Mf_o the transmission bandwidth is also Mf_o.

2.2 Frequency Domain Orthogonality

Another way to view the orthogonality property of OFDM signals is to look at its spectrum. In the frequency domain each OFDM subcarrier has a sinc, sin(x)/x, frequency response, as shown in Figure 2-2. This is a result of the symbol time corresponding to the inverse of the carrier spacing. As far as the receiver is concerned each OFDM symbol transmitted for a fixed time (TFFT) with no tapering at the ends of the symbol. This symbol time corresponds to the inverse of the subcarrier spacing of 1/TFFT Hz 1. This rectangular, boxcar, waveform in the time domain results in a sinc frequency response in the frequency domain. The sinc shape has a narrow main lobe, with many side-lobes that decay slowly with the magnitude of the frequency difference away from the centre. Each carrier has a peak at the centre frequency and nulls evenly spaced with a frequency gap equal to the carrier spacing.

The orthogonal nature of the transmission is a result of the peak of each subcarrier corresponding to the nulls of all other subcarriers. When this signal is detected using a Discrete Fourier Transform (DFT) the spectrum is not continuous as shown in Figure 2-2 (a), but has discrete samples. The sampled spectrum are shown as 'o's in the figure. If the DFT is time synchronized, the frequency samples of the DFT correspond to just the peaks of the subcarriers, thus the overlapping frequency region between subcarriers does not affect the receiver. The measured peaks correspond to the nulls for all other subcarriers, resulting in orthogonality between the subcarriers.



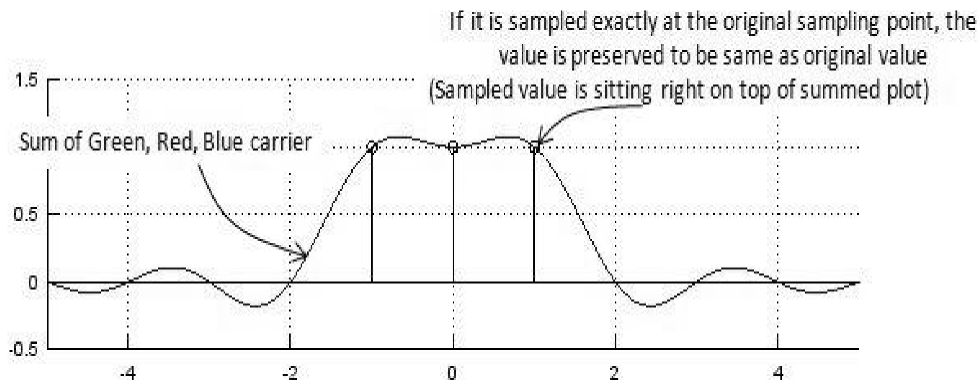


Figure 2-2: Frequency response of the subcarriers in a 3 tone OFDM signal.

2.3 OFDM Generation and Reception

OFDM signals are typically generated digitally due to the difficulty in creating large banks of phase lock oscillators and receivers in the analog domain. Figure 2-3 shows the block diagram of a typical OFDM transceiver. The transmitter section converts digital data to be transmitted, into a mapping of subcarrier amplitude and phase. It then transforms this spectral representation of the data into the time domain using an Inverse Discrete Fourier Transform (IDFT). The Inverse Fast Fourier Transform (IFFT) performs the same operations as an IDFT, except that it is much more computationally efficiency, and so is used in all practical systems. In order to transmit the OFDM signal the calculated time domain signal is then mixed up to the required frequency.

The receiver performs the reverse operation of the transmitter, mixing the RF signal to base band for processing, then using a Fast Fourier Transform (FFT) to analyze the signal in the frequency domain. The amplitude and phase of the subcarriers is then picked out and converted back to digital data. The IFFT and the FFT are complementary function and the most appropriate term depends on whether the signal is being received or generated. In cases where the signal is independent of this distinction then the term FFT and IFFT is used interchangeably.

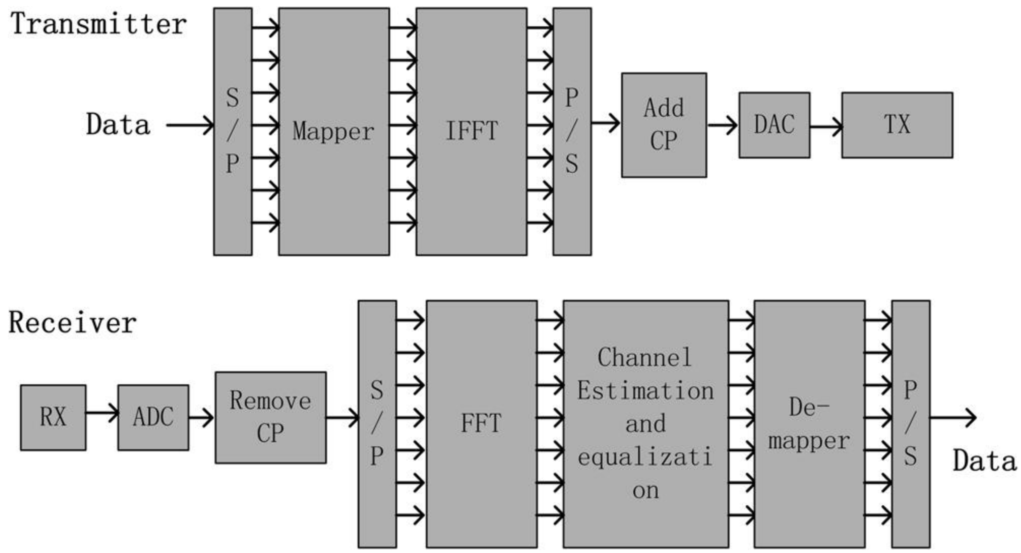


Figure 2-3: Block diagram showing a basic OFDM transceiver.

2.4 Serial to Parallel Conversion

Data to be transmitted is typically in the form of a serial data stream. In OFDM, each symbol typically transmits 40 - 4000 bits, and so a serial to parallel conversion stage is needed to convert the input serial bit stream to the data to be transmitted in each OFDM symbol. The data allocated to each symbol depends on the modulation scheme used and the number of subcarriers. For example, for a subcarrier modulation of 16-QAM each subcarrier carries 4 bits of data, and so for a transmission using 100 subcarriers the number of bits per symbol would be 400.

For adaptive modulation schemes such as described in section 4.2, the modulation scheme used on each subcarrier can vary and so the number of bits per subcarrier also varies. As a result the serial to parallel conversion stage involves filling the data payload for each subcarrier. At the receiver the reverse process takes place, with the data from the subcarriers being converted back to the original serial data stream.

When an OFDM transmission occurs in a multipath radio environment, frequency selective fading can result in groups of subcarriers being heavily attenuated, which in turn can result in bit errors. These nulls in the frequency response of the channel can cause the information sent in neighboring carriers to be destroyed, resulting in a clustering of the bit errors in each symbol. Most Forward Error Correction (FEC) schemes tend to work more effectively if the errors are spread evenly, rather than in large clusters, and so to improve the performance most systems employ data scrambling as part of the serial to parallel conversion stage. This is implemented by randomizing the subcarrier allocation of each sequential data bit. At the receiver the reverse scrambling is used to decode the signal. This restores the original sequencing of the data bits, but spreads clusters of bit errors so that they are approximately uniformly distributed in time. This randomization of the location of the bit errors improves the performance of the FEC and the system as a whole.

2.5 Frequency to Time Domain Conversion

After the subcarrier modulation stage each of the data subcarriers is set to an amplitude and phase based on the data being sent and the modulation scheme; all unused subcarriers are set to zero. This sets up the OFDM signal in the frequency domain. An IFFT is then used to convert this signal to the time domain, allowing it to be transmitted. Figure 2-5 shows the IFFT section of the OFDM transmitter. In the frequency domain, before applying the IFFT, each of the discrete samples of the IFFT corresponds to an individual subcarrier. Most of the subcarriers are modulated with data. The outer subcarriers are unmodulated and set to zero amplitude. These zero subcarriers provide a frequency guard band before the nyquist frequency and effectively act as an interpolation of the signal and allows for a realistic roll off in the analog anti-aliasing reconstruction filters.

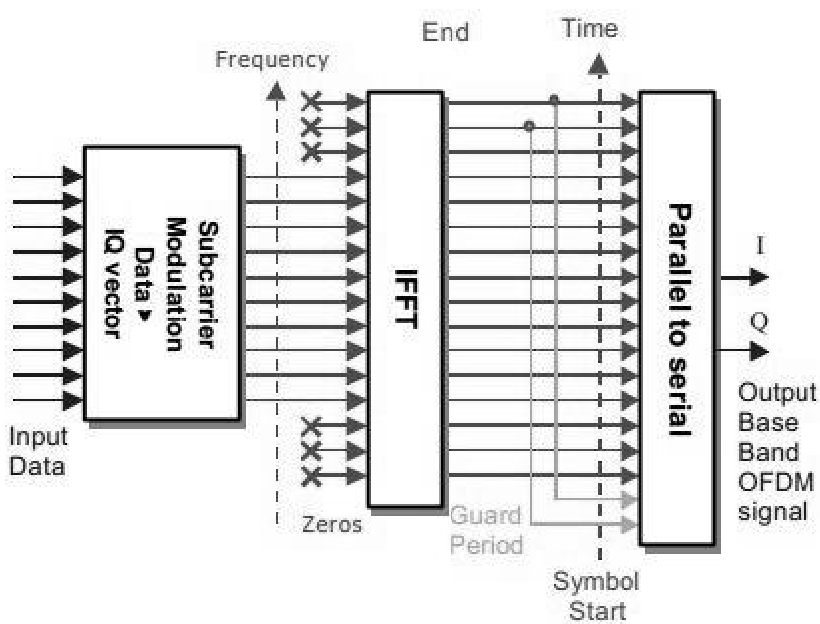


Figure 2-4: OFDM generation, IFFT stage

2.6 RF Modulation

The output of the OFDM modulator generates a base band signal, which must be mixed up to the required transmission frequency. This can be implemented using analog techniques as shown in Figure 2-7 or using a Digital Up Converter as shown in Figure 2-8. Both techniques perform the same operation, however the performance of the digital modulation will tend to be more accurate due to improved matching between the processing of the I and Q channels, and the phase accuracy of the digital IQ modulator.

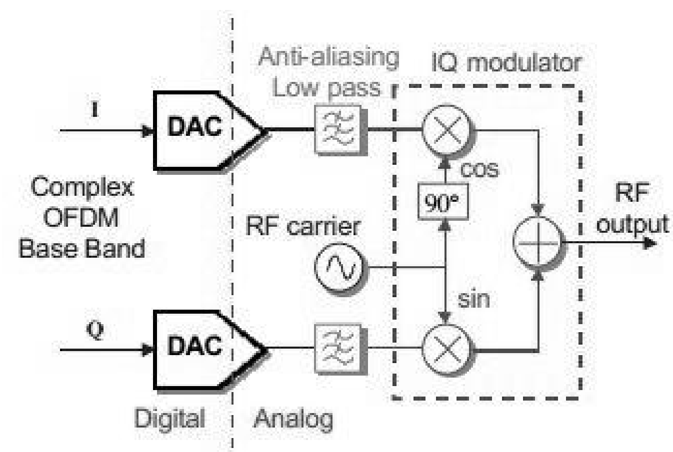


Figure 2-6: RF modulation of complex base band OFDM signal, using analog techniques.

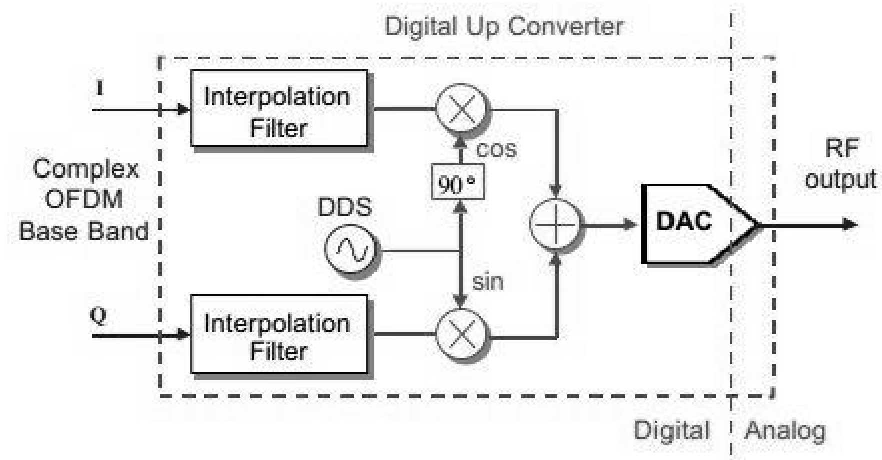


Figure 2-7: RF modulation of complex base band OFDM signal, using digital techniques.

Chapter 3

CHANNEL ESTIMATION

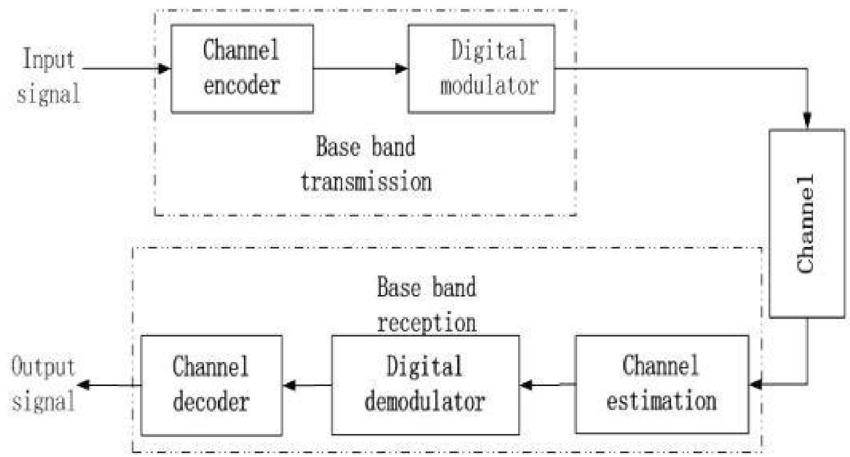
Channels are essential parts of communication systems. High speed and utilization of frequency spectrum are requested by the development of wireless communication. Mobile wireless channel is the most complex one among all kinds of channels. In reality, the state of channel is usually unknown, so channel estimation is highly requested. Radio waves propagating in wireless channels are reflected by buildings, trees and even motorcars. These form multiple propagation paths. Received waves are divided into three kinds: direct waves, diffracted waves and reflected waves, which result to different time delays, phases, amplitudes and additive noise. It will cause severe fading.

In 3G communication, higher speed of the mobile station is requested, which means the dynamic range of channel parameters is large. Thus, coherent reception is rather difficult, and the channel estimation is the key technology. The design of algorithm with high efficiency and low complexity is the foundation of channel equalization technology, joint detection and smart antenna.

And also it can greatly improve the system performance.

Channel estimation is a process to get the impulse response approximation and make it near as soon as possible in order to compensate. It is affected by different types of modulation, demodulation and detection. In communication system, the modulation mode with varying amplitude is always used in order to increase the spectrum efficiency.

In this case, the exactly channel status information should be known first by the receiver and then coherent demodulated. Channel estimation is absolutely necessary. Channel frequency response can be estimated according to the statistic information of receiving terminal. The position of channel estimation in mobile communication system is shown in the figure.



There are two main problems in designing channel estimators for wireless OFDM systems. The first problem is the arrangement of pilot information, where pilot means the reference signal used by both transmitters and receivers.

The second problem is the design of an estimator with both low complexity and good channel tracking ability. The two problems are interconnected. In general, the fading channel of OFDM systems can be viewed as a two dimensional (2D) signal (time and frequency). The optimal channel estimator in terms of mean-square error is based on 2D Wiener filter interpolation. Unfortunately, such a 2D estimator structure is too complex for practical implementation. The combination of high data rates and low bit error rates in OFDM systems necessitates the use of estimators that have both low complexity and high accuracy, where the two constraints work against each other and a good trade-off is needed. The one-dimensional (1D) channel estimations are usually adopted in OFDM systems to accomplish the trade-off between complexity and accuracy [1–7]. The two basic 1D channel estimations are block-type pilot channel estimation and comb-type pilot channel estimation, in which the pilots are inserted in the frequency direction and in the time direction, respectively. The estimations for the block-type pilot arrangement can be based on least square (LS), minimum mean-square error (MMSE), and modified MMSE. The estimations for the comb-type pilot arrangement includes the LS estimator with 1D interpolation, the maximum likelihood (ML) estimator, and the parametric channel modeling-based (PCMB) estimator. Other channel estimation strategies were also studied, such as the estimators based on simplified 2D interpolations, the estimators based on iterative filtering and decoding, estimators for the OFDM systems with multiple transmit-and-receive antennas, and so on.

Channel Estimation Methods

There are two channel estimation methods about channel estimation technology.

The first is an algorithm base on training sequence or pilot, it can provide rapid acquisition as well as accurate tracing. But the transmission efficiency of channel is lower, the pilot takes up some place of time slots or bandwidths. And the other one is blind channel estimation base on the statistical property of sending signal. It has high transmission efficiency and particularly can save spectrum resource.

But it requested a long time to save record of data symbols.

In a word, about channel estimation, there have been a lot of excellent methods existed. Each of them has advantages and drawbacks.

Pilot or Training Symbol-based Methods

In practical systems, channels are in variably estimated using periodic bursts of known training symbols, therefore we focus mostly on these techniques. Conventional training based methods only exploit the presence of the known training symbols. The results can be enhanced by also incorporating the convolutional properties of the surrounding unknown data symbols, which lead to so-called enhanced training-based methods. Also discussed are semi-blind methods that combined training-based criterion with a purely blind criterion. Blind techniques do not exploit the knowledge of training symbols, and focus on deterministic or stochastic properties of the system.

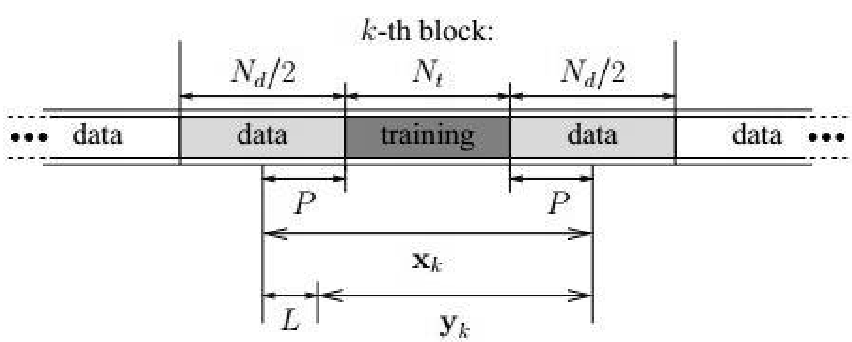


Fig.: Partitioning of the transmitted symbol vectors into blocks, each consisting of Nt training and Nd data symbol vectors.

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Conventional Training based Methods

Let us consider a convolutive MIMO system with At transmit antennas and Ar receive antennas. Suppose x(n) represents the At x 1 symbol vector sequence trans-mitted at the At transmit antennas. Assuming symbol rate sampling at each receive antenna, the Ar 1 sample vector sequence received at the Ar receive antennas is then given by

y(n) = ∑H(l)x(n l) + e(n);

where e(n) is the Ar 1 additive noise vector sequence on the Ar receive antennas, which we assume to be zero-mean white (spatially and temporally) Gaussian with variance e2, and H(l) is the Ar At MIMO channel of order L (or length L + 1). We will often make use of the vectorized form of H(l), which is obtained by stacking its columns: h(l) = vec[H(l)].

Conventional training solutions use only those received samples that solely depend on the training symbols. In other words, we consider P = 0, which allows us to simplify as

yk = (X k(t) X I)h + ek : 1.1)

where I = IAr . Although many different channel estimation procedures can be applied to (1.1), we restrict ourselves to maximum likelihood (ML) channel estimation, which neither requires knowledge of the noise variance nor any statistical information about the channel.

Basic 1D Channel Estimators

The two basic 1D channel estimations in OFDM systems are illustrated in the figure The first one, block-type pilot channel estimation, is developed under the assumption of slow fading channel, and it is performed by inserting pilot tones into all subcarriers of OFDM symbols within a specific period. The second one, comb-type pilot channel estimation, is introduced to satisfy the need for equalizing when the channel changes even from one

OFDM block to the subsequent one. It is thus performed by inserting pilot tones into certain subcarriers of each

OFDM symbol, where the interpolation is needed to estimate the conditions of data subcarriers. The strategies of these two basic types are analyzed in the next sections.

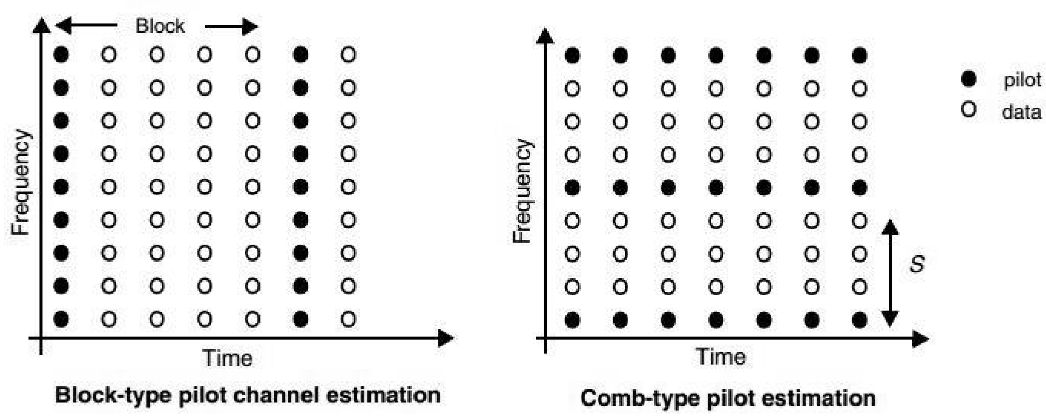


Figure: Two Basic type of pilot arrangement for OFDM systems

Block type pilot arrangement

In block-type pilot-based channel estimation, as shown in figure above, OFDM channel estimation symbols are transmitted periodically, and all subcarriers are used as pilots. The task here is to estimate the channel conditions given the pilot signals and received signals, with or without using certain knowledge of the channel statistics. The receiver uses the estimated channel conditions to decode the received data inside the block until the next pilot symbol arrives. The estimation can be based on least square (LS), minimum mean-square error (MMSE), and modified MMSE.

LS Estimator

The LS estimator minimizes the parameter $(Y - \underline{X}H)^H(Y - \underline{X}H)$, where $(\ )^H$ means the conjugate transpose operation. It is shown that the LS estimator of is given by

$$\hat{H}_{LS} = \underline{X}^{-1}\bar{Y} = [(X_k / \ Y_k)]^T$$

Without using any knowledge of the statistics of the channels, the LS estimators are calculated with very low complexity, but they suffer from a high mean-square error.

MMSE Estimator

The MMSE estimator employs the second-order statistics of the channel conditions to minimize the mean-square error.

Denote by $\underline{R}_{gg}$, $\underline{R}_{HH}$, $\underline{R}_{YY}$ and the auto covariance matrix of g, H , and Y, respectively, and $\underline{R}_{gY}$ by the cross covariance matrix between g and Y . Also denote σ^2_N by the noise variance $E\{\underline{N}^2\}$. Assume the channel vector g and the noise N are uncorrelated, it is derived that

$$\underline{R}_{HH} = E\{\bar{H}\bar{H}^H\} = E\{(\underline{F}\bar{g})(\underline{F}\bar{g})^H\} = \underline{F}\underline{R}_{gg}\underline{F}^H$$

$$\underline{R}_{gY} = E\{\bar{g}\bar{Y}^H\} = E\{\bar{g}(\underline{X}\underline{F}\bar{g} + \bar{N})^H\} = \underline{R}_{gg}\underline{F}^H\underline{X}^H$$

$$\underline{R}_{YY} = E\{\bar{Y}\bar{Y}^H\} = \underline{X}\underline{F}\underline{R}_{gg}\underline{F}^H\underline{X}^H + \sigma_N^2 I_N$$

Assume $\underline{R}_{gg}$ and σ^2_N are known at the receiver in advance, the MMSE estimator of g is given by $g_{MMSE} = \underline{R}_{gY}.\underline{R}_{YY}^{-1}\, Y^{HH}$

Note that if g is not Gaussian, g_{MMSE} is not necessarily a minimum mean square error estimator, but it is still the best linear estimator in the mean-square error sense. At last, it is calculated that

$$\begin{aligned}\hat{H}_{MMSE} &= \underline{F}\hat{g}_{MMSE} = \underline{F}[(\underline{F}^H\underline{X}^H)^{-1}\underline{R}_{gg}{}^{-1}\sigma_N^2 + XF]^{-1}\bar{Y} \\ &= \underline{F}\underline{R}_{gg}[(\underline{F}^H\underline{X}^H\underline{X}\underline{F})^{-1}\sigma_N^2 + \underline{R}_{gg}]\underline{F}^{-1}\hat{H}_{LS} \\ &= \underline{R}_{HH}[\underline{R}_{HH} + \sigma_N^2(\underline{X}\underline{X}^H)^{-1}]^{-1}\hat{H}_{LS}\end{aligned}$$

The MMSE estimator yields much better performance than LS estimators, especially under the low SNR scenarios.

A major drawback of the MMSE estimator is its high computational complexity, especially if matrix inversions are needed each time the data in $\underline{X}$ changes.

Modified MMSE Estimators

Modified MMSE estimators are studied widely to reduce complexity. Among them, an optimal low-rank MMSE (OLR-MMSE) estimator is proposed in this paper, which combines the following three simplification techniques:

1. The first simplification of MMSE estimator is to replace the term $(\underline{X}\,\underline{X}^H)^{-1}$ with its expectation $E\{(\underline{X}\,\underline{X}^H)^{-1}\}$. Assuming the same signal constellation on all tones and equal probability on all constellation points, we have

$$E\{(\underline{X}\underline{X}^H)^{-1}\} = E\{|1/\ X_k|^2\}\underline{I}$$

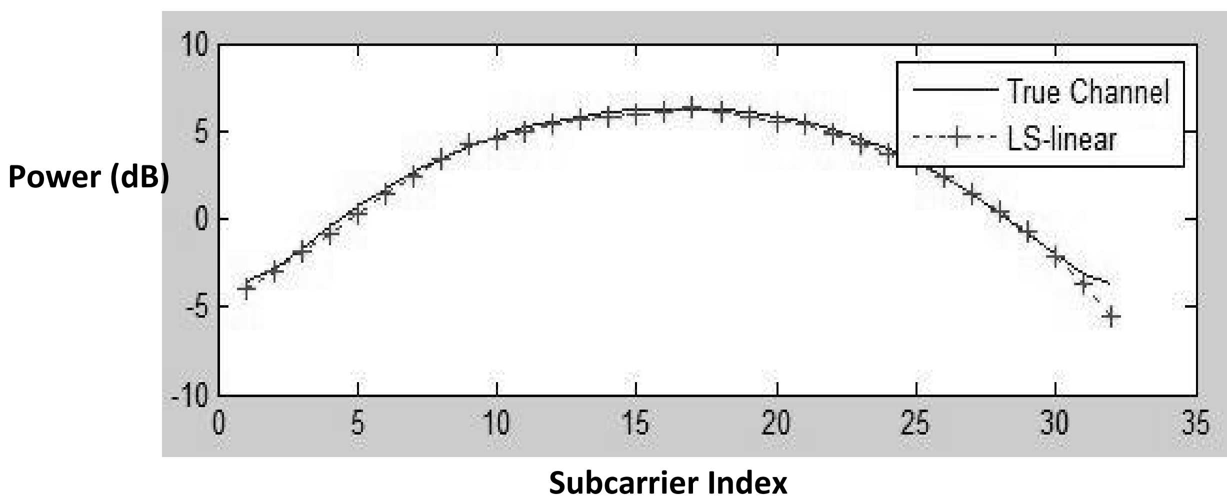
2. The second simplification is based on the low-rank approximation. Equation 1 has , $0 \leq \tau_m T_S \leq T_G$ most of the energy in g is contained in, or near, the first (L+ 1) taps, where $L = \lceil T_G / T_S \rceil N$ and N is the DFT size. Therefore, we can only consider the taps with significant energy, that is, the upper left corner of the auto covariance matrix $\underline{R}_{gg}$. In the IEEE Std. 802.11 and IEEE Std. 802.16 [13], is chosen among {1/32, 1/16, 1/8, 1/4}, so the effective size of matrix is reduced dramatically after the low-rank approximation is used.

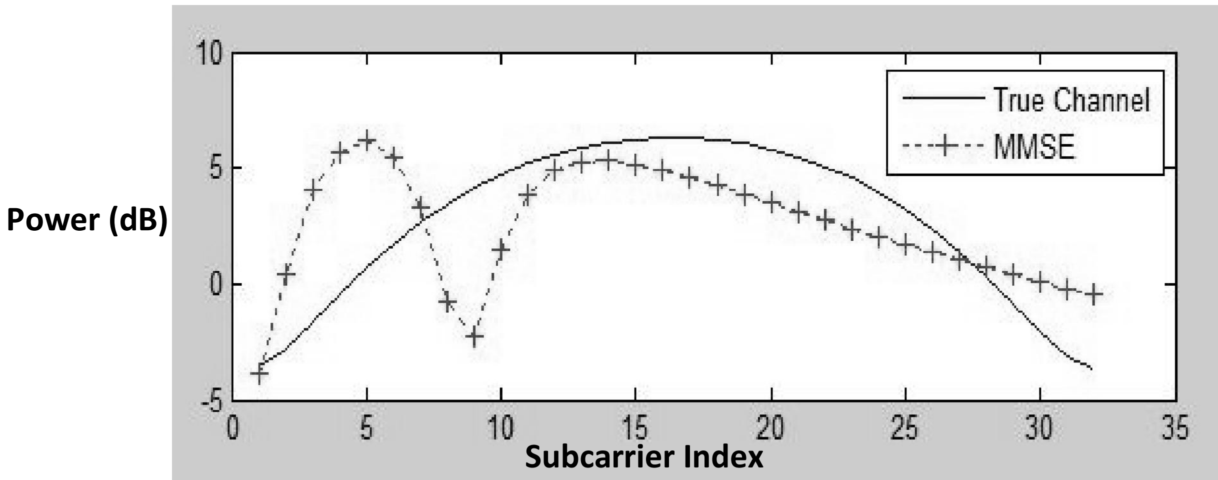
3.The third simplification uses the singular value decomposition (SVD). The SVD of $\underline{R}_{HH}$ is $\underline{R}_{HH} = \underline{U}\underline{\Lambda}\underline{U}^H$, , where $\underline{U}$ is a unitary matrix containing the singular vectors and $\underline{\Lambda}$ is a diagonal matrix containing the singular values on its diagonal. The SVD also dramatically reduces the calculation complexity of matrices.

Chapter 4

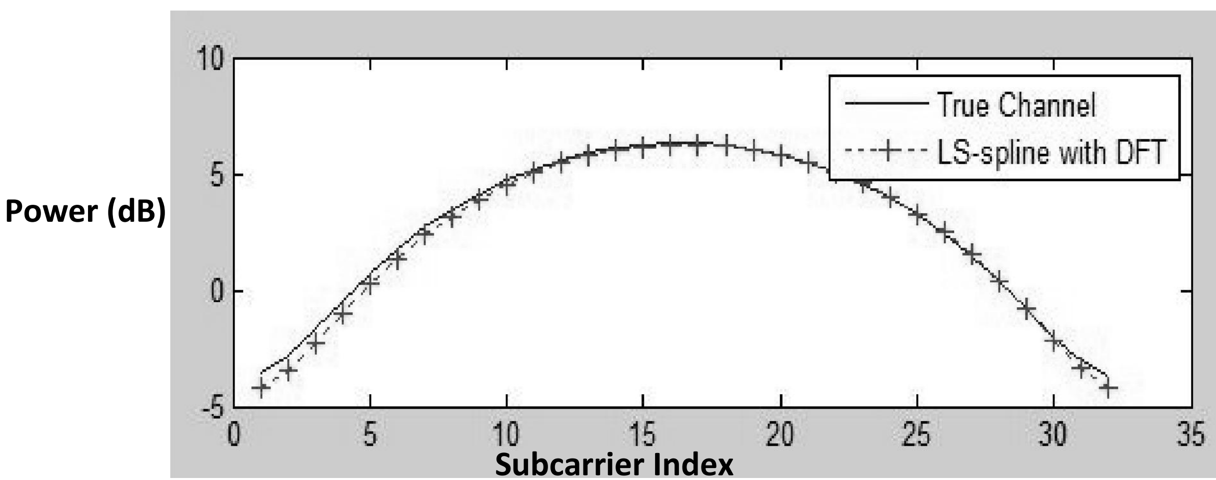
Simulation Analysis

Matlab software was used to plot the performances of different types of block type arrangements. Comparing the BER vs Simulation Index gives us the best arrangement for reducing the channel effects on the signal.

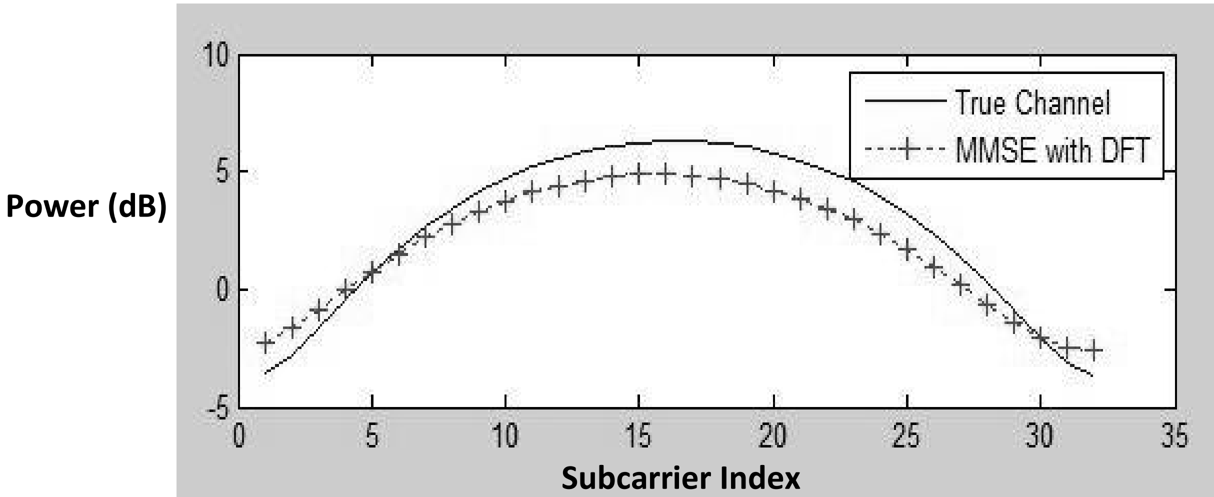




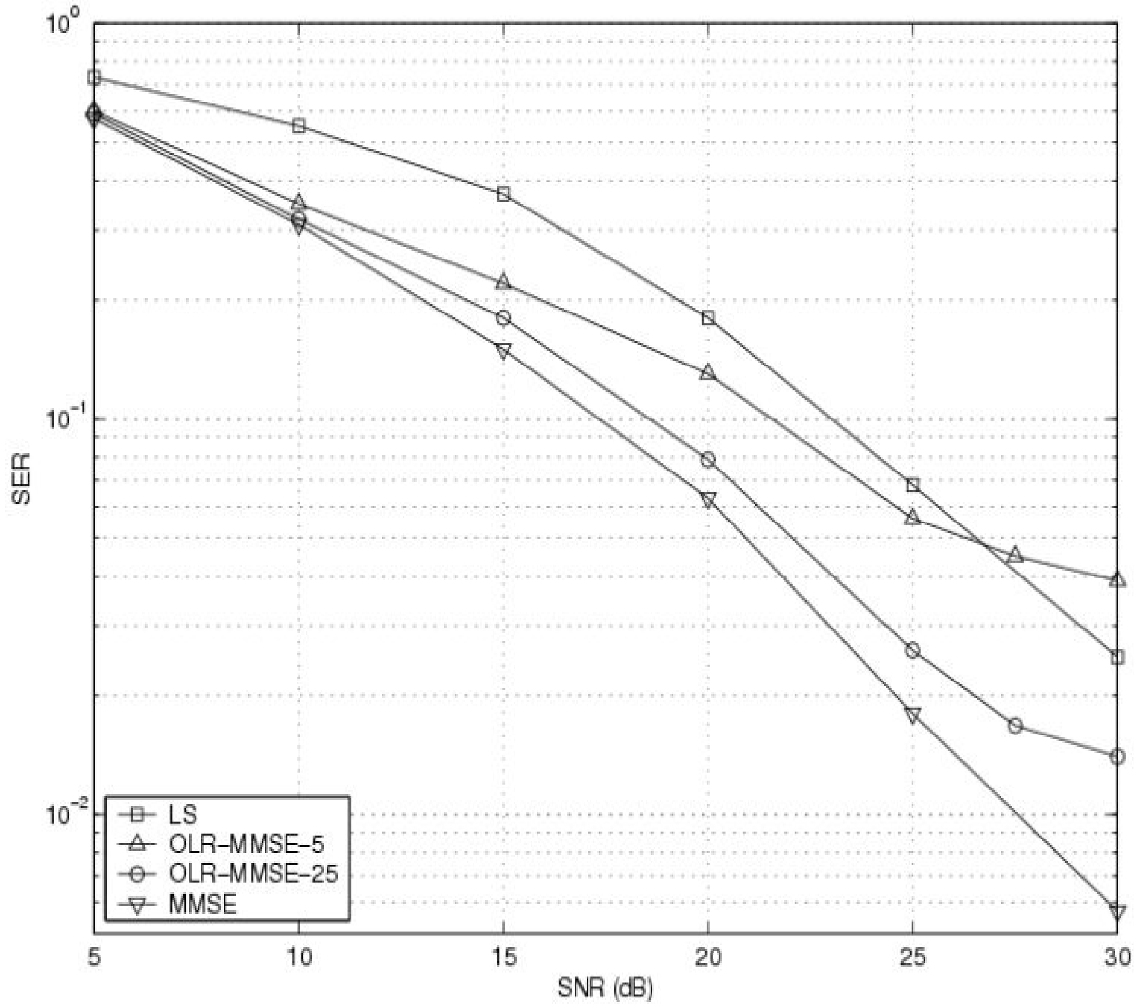
Power (dB)
Subcarrier Index
True Channel
MMSE
10
5
0
-5
0
5
10
15
20
25
30
35



Power (dB)
Subcarrier Index
True Channel
LS-spline with DFT
10
5
0
-5
0
5
10
15
20
25
30
35

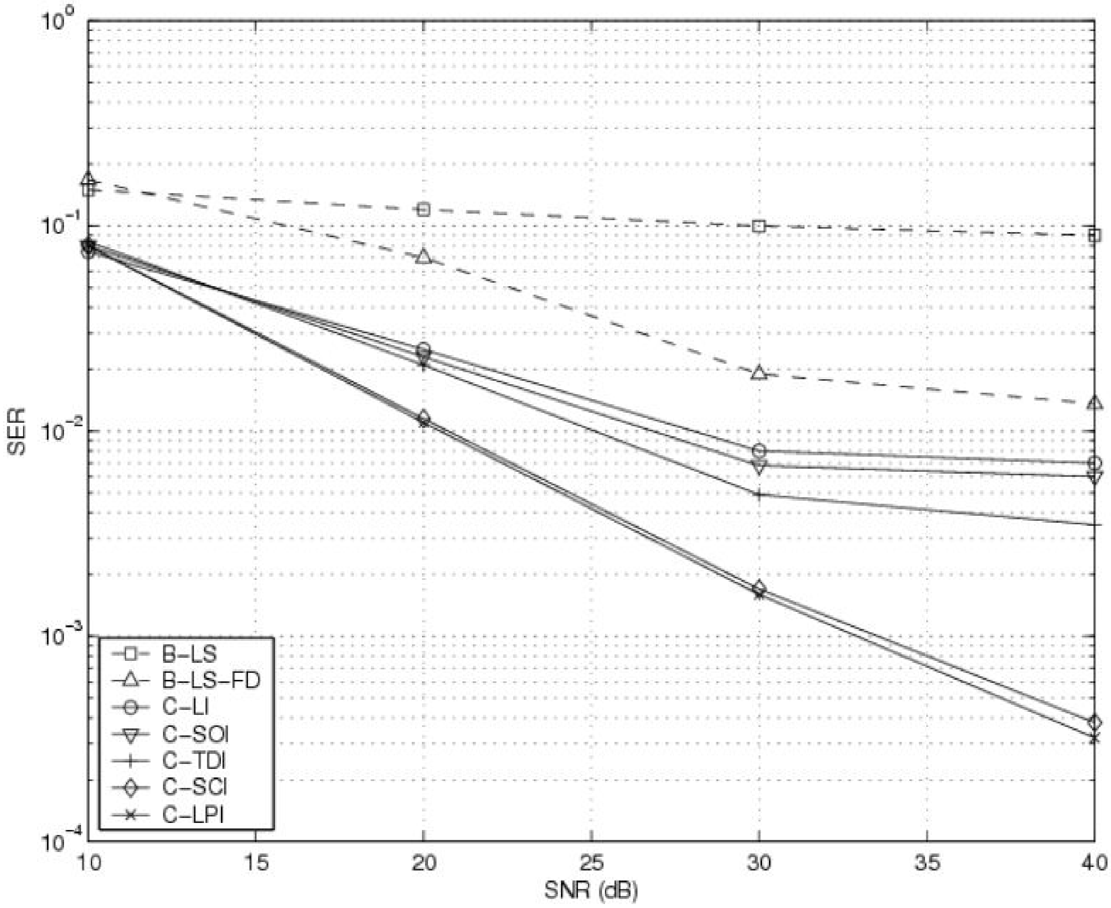


The first two, Program ("LS_CE") and Program ("MMSE_CE"), perform the LS and MMSE channel estimation, respectively. In "MMSE_CE," the channel impulse response is given as the sixth input argument. Also, the time-domain correlation $r_t[l]$ of channel is set to unity, such that $r_t[0] = J_0[0] = 1$, since the pilot symbol is inserted in every OFDM symbol. The third routine ("interpolate") uses the MATLAB built-in routine "interp1" to perform the linear or cubic spline interpolation, depending on whether the fourth input argument (method) is "linear" or "spline." These routines will be used in "channel_estimation.m" to test their validity.



10^0
10^{-1}
10^{-2}
SER
5
10
15
20
25
30
SNR (dB)
LS
OLR–MMSE–5
OLR–MMSE–25
MMSE

In above Figure, the symbol error rate (SER) versus the average SNR is plotted for the proposed block-type pilot channel estimation schemes over a slow fading channel with a bandwidth of 500 kHz, 16QAM modulation, DFT size N= 64,and a cyclic prefix L= 4. In this figure, the legends LS, MMSE, OLR-MMSE-5,and OLR-MMSE-25 present the estimators based on LS, MMSE, OLR-MMSE with rank p= 5and OLR-MMSE with rank p= 25, respectively, without the decision feedback. The MMSE estimator yields the best performance, and LS yields the worst. Also, for the OLR-MMSE estimator, a SER floor is shown due to loss of channel information by reducing the rank of the channel correlation matrix.



10^0
10^{-1}
10^{-2}
10^{-3}
10^{-4}
SER
10
15
20
25
30
35
40
SNR (dB)
B−LS
B−LS−FD
C−LI
C−SOI
C−TDI
C−SCI
C−LPI

The above figure compares the SER performance of the estimation schemes with block-type pilot arrangement and comb type pilot arrangement over a fast fading channel with Doppler frequency 70 Hz. The parameters are 17.5 kHz bandwidth, 16QAM modulation, DFT size N= 1024, the number of pilot subcarriers per symbol N_p

= 128, and a cyclic prefix L= 256. In the figure, the legends B-LS, B-LS-FD represent the block-pilot channel estimation based on LS algorithm, with and without decision feedback, respectively; and the legends C-LI, C-SOI, C-TDI, C-SCI and C-LPI represent the comb-type pilot estimation based on LS algorithm, with the linear interpolation, the second order interpolation, the time domain interpolation, the spline cubic interpolation and the low-pass interpolation, respectively. The results show that the comb-type estimation schemes outperform block-type schemes, which is because the channel changes so fast that there are even changes for adjacent OFDM symbols. It is also shown that the performance among the comb-type estimation techniques usually ranges from the best to the worst as follows: low-pass, spline cubic, time-domain, second order, and linear.

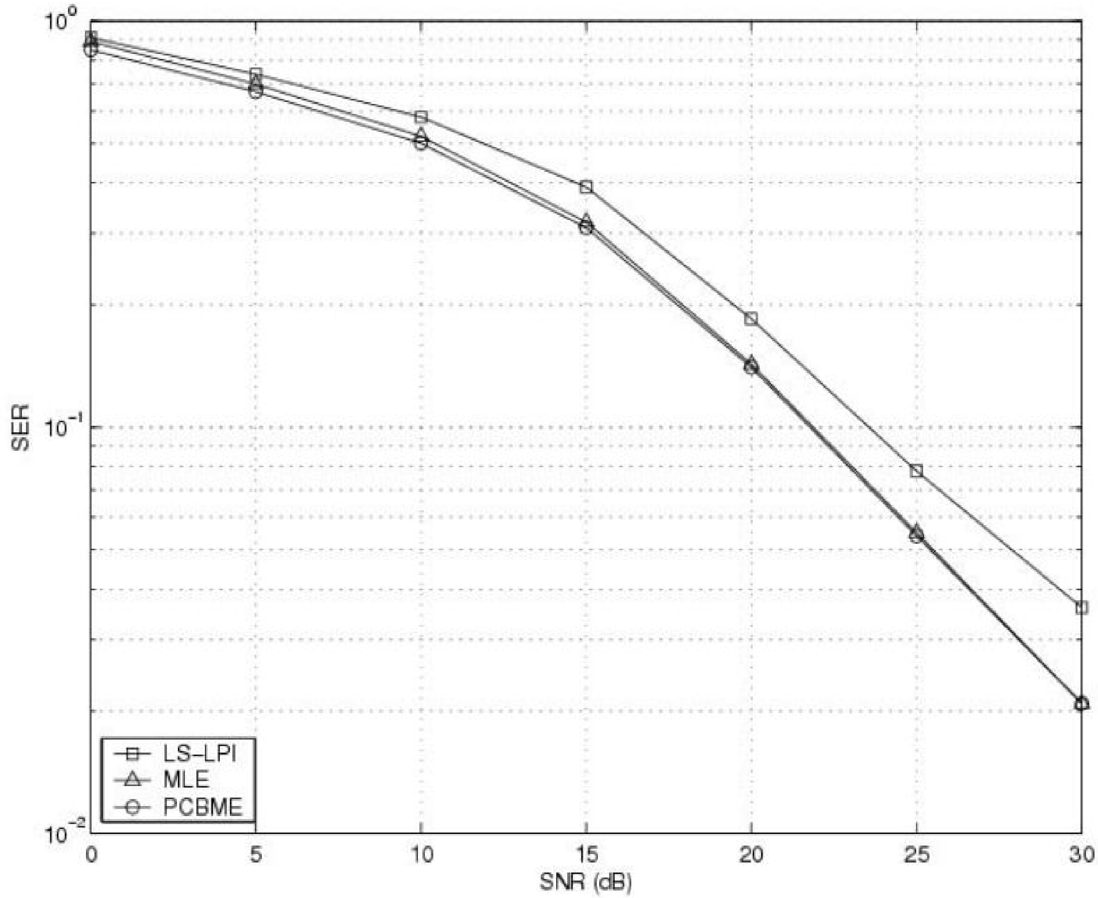

10^0
10^-1
10^-2
SER
0
5
10
15
20
25
30
SNR (dB)
LS-LPI
MLE
PCBME

Figure 5 shows the SER performance plotted for the three different estimators with comb-type pilot arrangement.

A fast fading channel with a bandwidth 5 MHz consists of 6 independent resolvable paths (that is, K= 6). Other parameters are 16 QAM modulation, DFT size N= 1024, the number of pilot subcarriers per symbol N_p= 32, and a cyclic prefix L= 16. The legends LS-LPI, MLE, and PCMBE represent the LS estimator with low-pass interpolation, the ML estimator, and the PCMB estimator, respectively. Simulation results show that the performance of LS-LPI is worse than the other two, and the performance of PCMB is slightly better than MLE at small SNRs.

Chapter 5

Conclusion

In OFDM systems, efficient channel estimation schemes are essential for coherent detection of a received signal.

After multi-carrier demodulation, the received signal is typically correlated in two dimensions, in time and frequency. By periodically inserting pilots in the time-frequency grid to satisfy the 2D sampling theorem, the channel response can be reconstructed by exploiting its correlation in time and frequency.

This paper reviews block-type pilot based channel estimation strategies in OFDM systems. It describes block-type pilot-channel estimators, which may be based on least square (LS), minimum mean-square error (MMSE) or optimal low-rank MMSE (OLR-MMSE), with or without a decision feedback equalizer.

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