Novel Channel Estimation and Synchronization Techniques for OFDM Based Broadband Wireless Access Systems

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ABSTRACT

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Broadband Wireless Access (BWA) systems have gained a lot of attention in the industry for providing flexible and easy deployment solutions to high-speed wireless communications technology such as WiMAX. Orthogonal Frequency Division Multiplexing (OFDM) system is one of the technologies that can offer BWA efficiently. However, there are certain problems with the OFDM system that need to be solved. In this thesis, a system based on OFDM scheme and principles has been designed to deliver high-speed mobile broadband services with great efficiency and reliability, with emphasis on resolving the problems that current OFDM systems face.

A comprehensive investigation of OFDM systems are performed. We highlight and analyze in detail the current problems with channel estimation, synchronization and resource allocation of OFDM systems. We propose a new efficient channel estimation technique for the OFDM technology. Furthermore, we present an enhanced

synchronization technique to mitigate frequency errors from which OFDM systems normally suffer in certain multipath environment. It is observed that our proposed scheme has several advantages over other existing schemes due to ease of implementation, bandwidth efficiency and better performance. Finally, we apply our proposed solutions to dynamic resource allocation in OFDM systems. Dynamic resource allocation in OFDM systems with proper channel state information and synchronization is highly important for designing an efficient and reliable OFDM system. Innovative design of such system is the main focus of this thesis.

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LIST OF ACRONYMS

BWA	Broadband Wireless Access
CDMA	Code Division Multiple Access
CSI	Channel State Information
CP	Cyclic Prefix
DRA	Dynamic Resource Allocation
FDMA	Frequency Division Multiple Access
FFT	Fast Fourier Transform
IFFT	Inverse Fast Fourier Transform
ICI	Inter Channel Interference
ISI	Inter Symbol Interference
MA	Margin Adaptive
MAC	Media Access Control
OFDMA	Orthogonal Frequency Division Multiple Access
RA	Rate Adaptive
RBS	Radio Base Station
TDMA	Time Division Multiple Access
UMTS	Universal Mobile Telecommunication System
WBMCS	Wireless Broadband Multimedia Communications Systems
WiMAX	Worldwide interoperability for Microwave Access

Chapter 1

Introduction

The development of wireless communication technologies is moving forward rapidly. Increasing competition fuels this development, which strives to provide end-users with access to a wide range of efficient and reliable services at lower costs. For example, the demand for wireless Internet and Multimedia communications has increased exponentially over the past few years. Hence, a wide range of ongoing research and development around the world is aimed at defining the next generation of Wireless Broadband Multimedia Communications Systems (WBMCS) [1].

WBMCS will enable us to bring the Internet and Multimedia communications together with wireless communication and to provide services at high data rates with great reliability. Implementing systems with such requirements calls for selecting a proper modulation technique. Orthogonal Frequency Division Multiplexing (OFDM) due to its properties that we are going to discuss shortly has proven to be the right choice for such high data rate requirements.

OFDM is an advanced multi-carrier modulation technique for the next generation of wireless communication systems. With OFDM technique the entire transmission bandwidth is divided into N orthogonal sub-carriers. The sub-carriers are independent of each other, and therefore a multiple access system such as Orthogonal Frequency Division Multiple Access (OFDMA) can be designed. OFDMA technique spreads a single user's data stream over a wide bandwidth by breaking up the high-speed data into many low-speed packets and transmitting these low-speed packets over individual carriers [1].

Dynamic resource allocation based on proper channel state information and synchronization parameters is essential for achieving an optimal OFDMA system. For an advanced multiple access scheme such as OFDMA, it is crucial to have bandwidth efficiency and be properly synchronized. Novel design of such OFDMA system with efficient and enhanced solutions to channel estimation and synchronization problems is the focus of this thesis.

1.1 Overview of OFDM Technique

OFDM was first implemented using banks of sinusoidal generators, by placing them in parallel to create a multi-carrier system. The use of the Discrete Fourier Transform (DFT), suggested in 1971 by *Weinstein and Ebert*, significantly reduced the implementation complexity of multi-carrier OFDM modems [2]. This was further improved by the introduction of the Fast Fourier Transform (FFT) and the Inverse Fast Fourier Transform (IFFT) [3]. Today all OFDM signal processing is performed and implemented digitally in frequency domain with an FFT-IFFT pair, due to their efficiency and low cost of implementation.

OFDM is currently the modulation of choice for high-speed data access systems such as IEEE 802.11a/g WirelessLAN (WLAN), IEEE 802.16a/d/e WirelessMAN, ETSI HiperLAN2, HiperMAN and the Broadband Wireless Access (BWA) technology (a.k.a., WiMAX, an acronym for Worldwide Interoperability for Microwave Access). In addition, OFDM technique is used in several other applications such as the European

Digital Audio/Video Broadcast (DAB/DVB) systems. As shown in Figure 1.1, there is definitely a positive trend and most of the emerging and future wireless technologies have considered OFDM as one of the best underlying options.

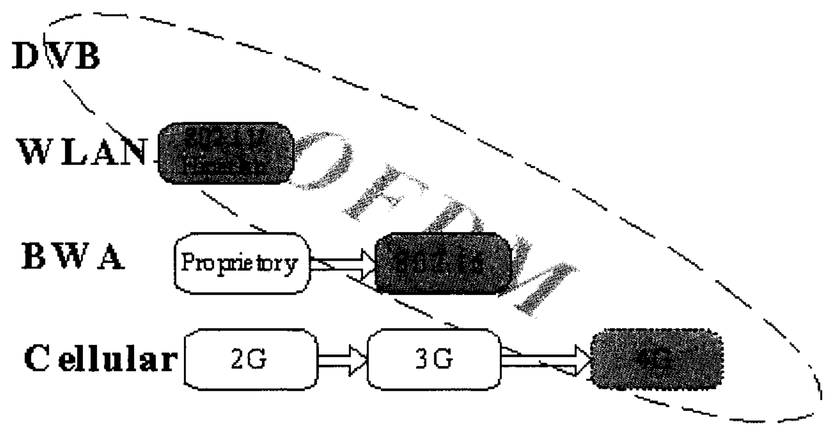

Figure 1.1 Wireless Technology Trend

1.1.1 OFDM vs FDM

OFDM is a multi-carrier multiplexing/modulation technique where the available bandwidth is divided and multiplexed into N overlapping orthogonal sub-carriers, and a single data stream is transmitted over a number of sub-carriers with lower rates [1]. An OFDM carrier signal is the sum of a number of orthogonal sub-carriers, with baseband data on each sub-carrier being independently modulated using, e.g., Quadrature Amplitude Modulation (QAM).

In original Frequency Division Multiplexing (FDM), the frequency is divided and multiplexed into N sub-carriers separated by a large guard band to detach different sub-carriers and to avoid interference. This is quite an inefficient use of bandwidth.

In Figure 1.2 we compare FDM vs. OFDM. Where $W=2R$ is the total available bandwidth, and N is the number of sub-carriers among which we want the total bandwidth to be divided. It can be observed in Figure 1.2 that with the OFDM method, the orthogonal sub-carriers overlap each other to a certain extent. Using OFDM method a lot of useful bandwidth can be saved; therefore OFDM is a highly spectral efficient technique.

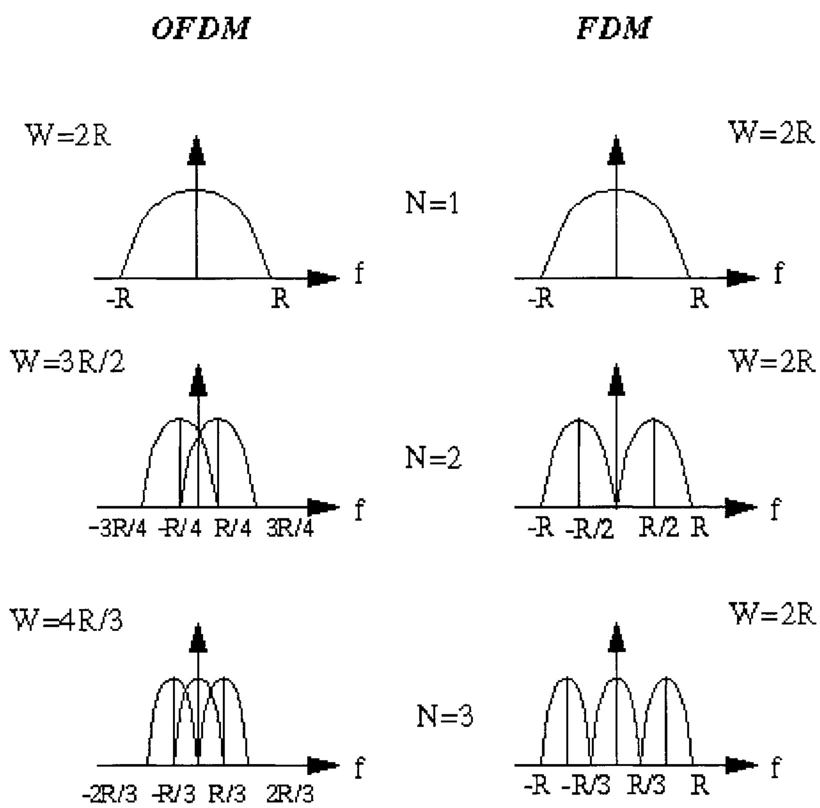

Figure 1.2 OFDM Vs FDM, Savings in bandwidth with OFDM

OFDM signal consists of narrowband carriers transmitted in parallel at different frequencies. Each individual carrier, called a sub-carrier or sub-channel, transmits information by modulating the phase or the amplitude of the sub-carrier over a symbol duration, i.e., each sub-carrier uses either Phase-Shift-Keying (PSK) or Quadrature-Amplitude-Modulation (QAM) to transmit information just as conventional single carrier systems. The minimum frequency spacing between sub-carriers is selected to be the inverse of the symbol duration so that each sub-carrier is orthogonal or non-interfering.

1.1.2 Orthogonality in an OFDM system

Orthogonality is defined as a certain type of mathematical relationship between different sub-carriers in an OFDM system [1], which we will illustrate mathematically with the help of an example from [1]. Consider an OFDM signal denoted by $s(t)$, which consists of a sum of sub-carriers that are modulated using PSK or QAM.

$$
\begin{aligned}
& s(t) = \mathrm{Re}\left\{ \sum_{i=-\frac{Ns}{2}}^{\frac{Ns}{2}-1} d_{i+\frac{Ns}{2}} \exp\left(j2\pi\left(f_c - \frac{i+0.5}{T}\right)(t-t_s)\right) \right\}, \quad t_s \le t \le t_s + T \\
& s(t) = 0, \quad t_s \le t \quad \wedge \quad t > t_s + T
\end{aligned}
\tag{1.1}
$$

where,

d_i are the complex QAM symbols

N_s is the number of sub-carriers

T is the symbol duration

f_c is the carrier frequency

t_s is the start time of OFDM symbol

Often, for simplicity reasons, the equivalent complex baseband notation is used. As shown below in Equation (1.2) [1], the real and imaginary parts correspond to the in-phase and quadrature parts of the OFDM signal, which must be multiplied by cosine and sine of the desired carrier frequency to produce the final OFDM signal.

$$
\begin{aligned}
&s(t) = \mathrm{Re}\left\{ \sum_{i=-\frac{Ns}{2}}^{\frac{Ns}{2}-1} d_{i+\frac{Ns}{2}} \exp(j2\pi \frac{i}{T}(t-t_s)) \right\}, \quad t_s \le t \le t_s + T \\
&s(t) = 0, \quad t_s \le t \quad \wedge \quad t > t_s + T
\end{aligned} \tag{1.2}
$$

So how can we achieve orthogonality in an OFDM system? If the j^{th} sub-carrier from (1.2) is demodulated by down converting the signal with a frequency j/T and then integrating the signal over T seconds, the result is written as [1]:

$$
\begin{aligned}
&\int_{ts}^{ts+T} \exp(-j2\pi \frac{j}{T}(t-t_s)) \sum_{i=-\frac{Ns}{2}}^{\frac{Ns}{2}-1} d_{i+\frac{Ns}{2}} \exp(j2\pi \frac{i}{T}(t-t_s))\, dt \\
&= \sum_{i=-\frac{Ns}{2}}^{\frac{Ns}{2}-1} d_{i+\frac{Ns}{2}} \int_{ts}^{ts+T} \exp(j2\pi \frac{i-j}{T}(t-t_s))\, dt \; = \; d_{j+\frac{Ns}{2}} T
\end{aligned} \tag{1.3}
$$

It can be observed in (1.3) that a complex carrier is integrated over T seconds. For the demodulated j^{th} sub-carrier, the integration process gives the desired output $d_{j+\frac{Ns}{2}}$ multiplied by a constant factor T, which is the QAM value for that particular sub-carrier. For all other subcarrriers, the integration is zero, because the frequency difference $i-j/T$ produces an integer number of cycles within the integration interval T, such that the integration result is always zero [1]. This way the orthognality of different sub-carriers is achieved.

Maintaining the orthogonality achieved is vital for an OFDM system. In order to ensure that orthogonality is always maintained when the signal goes under different channel degradations, we have proposed a simple and enhanced synchronization technique in Chapter 4 to mitigate the adverse channel effects.

1.2 Advantages of a Multi-Carrier System

In conventional communication systems, the information to be transmitted is modulated onto a single carrier. To obtain high data rates, the symbols must be transmitted quickly and therefore require the entire bandwidth. When the channel is of multi-path type, consecutive symbols arriving at different intervals will interfere with each other, resulting in Inter Symbol Interference (ISI). Sometimes, the signals might go through a frequency selective channel that can result in a deep fade and null signal. As a result it will be extremely difficult for the receiver to recognize the received symbol. Therefore, in a conventional communication system, multi-path environment causes severe degradation of the system performance.

One approach to the problem of eliminating ISI and achieving better system performance is to introduce a multi-carrier system. For example, in an OFDM multi-carrier system it is accomplished by splitting the available bandwidth into several orthogonal sub-carriers, and transmitting information slowly in parallel on these sub-carriers [4]. In the frequency domain each sub-carrier will occupy only a small frequency interval where the channel frequency response will be almost constant during an OFDM symbol. Therefore, each symbol often experiences an approximately flat fading channel.

In addition, if we have a constant channel during an OFDM symbol, the Inter Channel Interference (ICI) is eliminated, and the symbols transmitted on the different sub-carriers will not interfere. Hence, the problem of cross talk and interference between sub-carriers is resolved by having an OFDM system, where there is an orthogonality relationship maintained between the frequencies of the sub-carriers [1].

The main benefit of an OFDM system is derived from the fact that the system has the ability to handle the multi-path interference at the receiver. In a multi-path environment multiple copies of signals with different delay spread arrive at the receiver, as shown in Figure 1.3. The received signal can be adversely affected from multi-path effects.

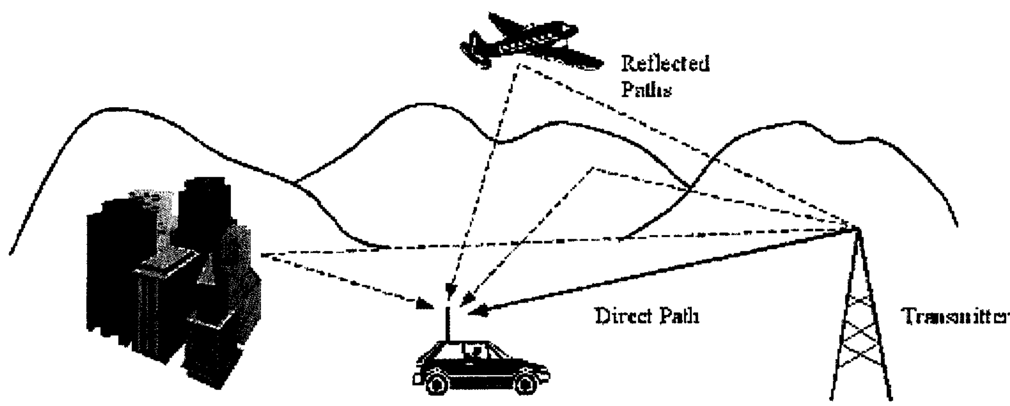

Figure 1.3 Multi-path Signals arriving at the mobile

Multi-path generates two adverse effects: frequency selective fading (deep fades) and Inter Symbol Interference (ISI). In a multi-carrier OFDM system, flat narrow-band channel response overcomes the former. Modulating at a very low symbol rate, which makes the symbols much longer than the channel impulse response, avoids the latter. Using powerful error correcting codes together with time and frequency interleaving yields even more robustness against frequency selective fading.

The effects of ISI can be further reduced with the insertion of a Cyclic Prefix (CP) [5], which is a copy of the last part of the OFDM symbol and acts as a guard space between consecutive OFDM symbols. The length of the CP is chosen such that it is longer than maximum delay spread τ_{max}. Hence, ISI can be avoided. Figure 1.4 shows the structure of an OFDM symbol with CP and all the multi-path components.

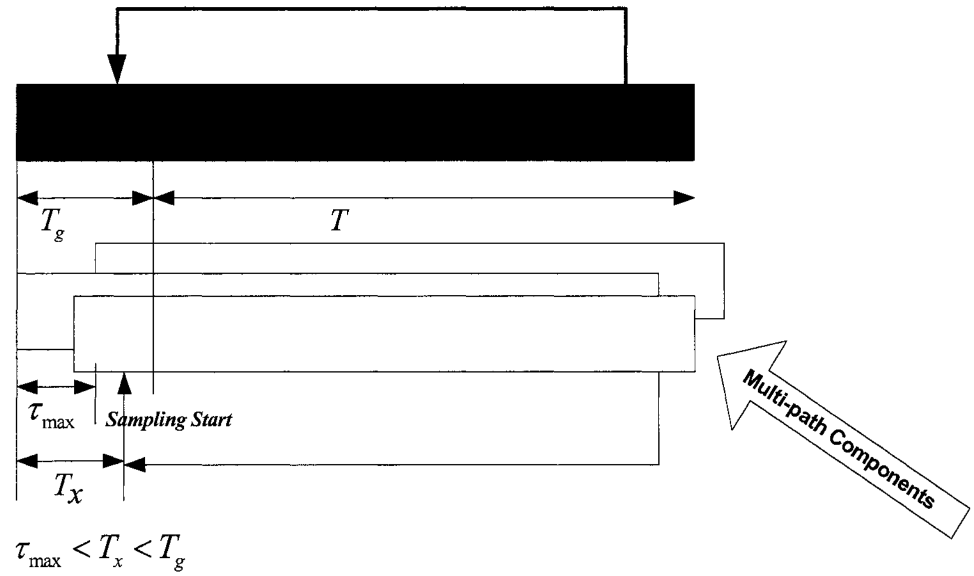

Figure 1.4 OFDM symbol with CP and Multi-path signal components

1.3 Multi-User OFDM (OFDMA)

OFDM has been proposed for multi-user systems such as the Universal Mobile Telecommunication System (UMTS) [6] and Wireless Local Area Networks (WLAN's) [7]. In a multi-user OFDM system, the orthogonality of the sub-carriers facilitates a sub-carrier division of different users. In an OFDM transmission system, the available spectrum is accessed by a large number of sub-carriers. Data symbols are efficiently modulated on these carriers by means of an FFT [4], both in the uplink and the downlink.

In a multi-user environment, an OFDM scheme has two main advantages. First, the receiver does not require an equalizer if a cyclic prefix is properly used and if the channel does not change during one OFDM symbol [5]. Secondly, dynamic channel assignment across the spectrum is simple as each user can conveniently access all of the sub-carriers by the FFT-implemented modulation.

1.3.1 Multiple Access Schemes

Since OFDM separates symbols in both time and frequency, a number of multiple access schemes such as Time Division Multiple Access (TDMA), Code Division Multiple Access (CDMA) or Frequency Division Multiple Access (FDMA) could be used. A TDMA scheme was proposed for the European WLAN standard and an OFDMA scheme combined with a time-slot structure and a frequency-hopping scheme was proposed for the UMTS radio interface [7].

In OFDM-TDMA or OFDM-CDMA structure for example, users are assigned the entire OFDM symbols, and they share the channel in a time-slot or different frequency spreading codes structure. However, each user has to transmit its signal over the entire spectrum. This leads to an averaged-down effect in the presence of deep fading and narrowband interference [8].

On the other hand, the total bandwidth can be divided into traffic channels or sub-carriers so that multiple access can be accommodated in a Frequency Division Multiple Access (FDMA) scheme, where different users are assigned different sub-carriers. We can optimize the system so that the sub-carriers can be allowed to orthogonally overlap each other, following the mathematical relationship that was explained earlier in Section 1.2. When orthogonal sub-carriers spectrally overlap, this scheme is not true FDMA, and therefore it is called Orthogonal Frequency Division Multiple Access (OFDMA) scheme. In Chapter 5, we will discuss Orthogonal Frequency Division Multiple Access (OFDMA) scheme in more detail, and we will observe how multiple users can access and obtain the systems resources dynamically.

1.4 Motivation: (Resource Allocation in OFDMA Systems)

Power and sub-carrier resource allocation is of prime importance in a multi-user OFDMA system to ensure efficiency, effectiveness and fairness in the system. Two classes of resource allocation schemes exist: fixed resource allocation [9], and dynamic resource allocation [10], [11], [12]. Fixed resource allocation schemes, such as Time Division (TD) and Frequency Division (FD) methods, assign an independent dimension (time slot or sub-carrier) to each user statically. It is obvious that fixed resource allocation scheme is not optimal, since the scheme is fixed regardless of the current channel condition and there is no multi-user diversity or spatial diversity to achieve a higher capacity.

On the other hand, dynamic resource allocation scheme allocates a dimension adaptively to the users based on their channel conditions. The wireless channel is often of varying nature; therefore, by using dynamic resource allocation we can make full use of the multi-user diversity or spatial diversity to achieve a higher capacity and an optimized system. For an OFDMA system, dynamic resource allocation seems to be the best choice. However, there are some problems with OFDMA system that need to be addressed and resolved for efficient and reliable resource allocation.

1.4.1 Problem Specification

Dynamic resource allocation is extremely important for achieving an efficient OFDMA system. By using channel diversity we can assign resources appropriately and optimally. However, there are some issues that need to be solved. Firstly, for an OFDMA system, good knowledge of channel state information is vital in the case of allocating resources dynamically. To the best of author's knowledge, in most of previous work and investigations, such as that of [10], [11], [12], it is assumed that perfect channel information is available at the Radio Base Station (RBS). Based on the perfect channel knowledge assumption, sub-carrier and power allocation algorithms are performed and are sent to each user by a separate channel.

Perfect knowledge of the channel state information is an immense imprecise assumption, which is the root of the performance problems in OFDMA systems; because a practical mobile OFDMA wireless system operates in a time-varying fading environment. In addition, imperfect channel information arises from limitations of channel estimation techniques as well as outdated information due to time varying channel conditions. We need to apply a realistic scenario and estimate the channel state information properly and adaptively, and then evaluate the system's performance and gain. Channel estimation is an important part of any communication system; a good channel estimation scheme should provide timely and reliable information on the state of the channel at the receiver, hence, enabling the receiver to effectively compensate for the multiplicative effects of the channel.

Secondly, in a multi-user OFDM environment, uplink synchronization is a very intricate task. It is one of the most important problems that must be addressed and researched for an OFDM system. It is well known that OFDM is very sensitive frequency offsets, which destroy orthogonality among sub-carriers and thereby cause inter-channel interference. This property becomes more challenging for multi-user OFDM (a.k.a. OFDMA), where, one OFDM symbol carries data for many users. The correction of a user's time and frequency offsets cannot be accomplished easily at the base station receiver, as the correction to one user would misalign other initially aligned users. In OFDMA systems proper synchronization is, however, necessary to maintain the orthogonality of the users, which is essential for reliable transmission. Most previous work to date have assumed perfect synchronization.

Little research has been done in the field of channel estimation and synchronization problems for OFDM systems. Most analysis of OFDM systems have been done with assumptions of perfect channel knowledge and perfect synchronization. Assumptions of perfect channel knowledge and perfect synchronization can lead to many problems in practical systems. To the best of our knowledge no research has been done in order to account for these crucial constraints and then evaluate the overall dynamic resource allocation and OFDMA system's performance. The problems related to channel knowledge and synchronization are crucial and inter-related in a practical system, e.g. synchronization issues would lead to errors in channel estimation. Therefore, the aim of this thesis is to mitigate these problems by finding novel solutions that would enhance the OFDMA broadband systems.

1.5 Objective and Scope of the thesis

The objective of this thesis is to propose appropriate novel solutions to the problems related to channel estimation and synchronization of the OFDM based systems, and apply the solutions to design and evaluate dynamic resource allocation scheme for achieving an efficient, enhanced and reliable broadband wireless access system.

1.5.1 Scope of the thesis

First, we will study and analyze in detail the previous work that has been done in respect to OFDM and OFDMA systems. This will help us gain an insight into how the systems work and what are the current problems associated.

Secondly, we will investigate various factors that can cause the channel information to be imperfect. Perfect knowledge of the channel is a big assumption. This can be the root of the performance problems in practical scenarios, because the broadband wireless environment under study is of time-varying fading nature. We will find out a solution to the assumption of perfect channel information and the problems that come with it. We will propose an efficient way of estimating the channel so that precise and timely information is passed and dynamic resource allocation can be performed properly. We will present a system model. The system model will be simulated under different scenarios and compared with different existing methodologies to support our proposed solution.

Third, we will examine the synchronization problems in current OFDMA systems in time varying environment. We will propose an enhanced scheme for estimating the frequency offsets.

Finally, we will apply our proposed solutions to an OFDMA system and evaluate the systems performance and gain. We will study and analyze the previous work that has been done with respect to resource allocation problems in an OFDMA system. We will compare the system with static and dynamic resource allocation schemes. A system model will be presented formulating power and sub-carrier allocations.

We will recommend a suboptimal solution in order to balance the tradeoff between complexity and performance, and design a novel and enhanced OFDMA system based on our proposed solutions. We will implement and apply our proposed methodologies to dynamic resource allocation in an OFDMA system, which is highly important for achieving an efficient and reliable system with proper channel state information and synchronization.

1.6 Organization of the thesis

This thesis is organized as follows:

Chapter 2 explains the theory behind the OFDM technology. An OFDM system model is presented to help us visualize the basic required components in OFDM transmitter and receiver systems. In addition, we present the OFDM signal design and processing. We will prove how, by using a cyclic prefix, the OFDM systems can become resilient to wireless channel adversary such as multi-path fading effect.

Chapter 3 proposes an efficient channel estimation technique that can be applied to an OFDMA system. The channel estimation technique is based on a time varying environment. We will also perform comparisons and analysis of various system models with the previous proposed work by various researchers.

Chapter 4 proposes an enhanced frequency offset estimation technique. From our research, we have found that a time domain approach to estimate the frequency offsets, as proposed in [13], will not work in case of an OFDMA system in multi-path time varying environments. Through simulations we have found that there is an error bound and the performance of the system degraded drastically in certain multi-path environment. To mitigate this problem and propose a newly designed solution, where we shifted into frequency domain and proposed an enhanced scheme for estimating the frequency offsets [14].

Chapter 5 explains the background and theory behind multi-user OFDM or OFDMA technology. We compare OFDMA with other multiple access schemes and highlight the advantages and disadvantages of different technologies. Furthermore, we will discuss resource allocation in OFDMA, and present an OFDMA system model. We will address the problems foreseen in practical scenarios, and describe the mathematical equations and theory behind resource allocation in an OFDMA system based on previous research. We will present equations of the optimized OFDMA system and the simulation results. We will also perform comparisons and analysis of various system models with the previous proposed work by various researchers. Finally we will design an enhanced OFDMA system based on our proposed solutions.

Chapter 6 concludes the thesis, and offers suggestions and recommendations for future work.

1.7 Simulations Tool

Simulations are carried out in OFDM Digital Transmission System (ODTS) environment, which is a software system that we have designed using C/C++ and MATLAB programming. In ODTS software system modeling and simulations are designed based on the principles of communications systems described in [1], [15], [16], [17], [18] and [19]. ODTS consists of functions and classes, which model different communication blocks of an OFDM transceiver. The software design consists of a hierarchy of modules, both simple and compound, whose behavior and functionality are defined based on the requirements and specifications defined by UMTS [6], [7] and WiMAX Forum [20]. These simple modules are the fundamental building blocks of the simulator and are clearly the lowest level entity in the hierarchy. A system may easily be constructed using merely these units, however they are more commonly combined together into compound modules to more realistically model the structure of a wireless communication system.

The building block modules are interconnected through a series of gates and channels and they communicate with each other via messaging. The simple modules define the messages and they are contained in data structures. The connections are simple ideal paths that allow the transfer of these messages from one block to another; however, they may also be complex routes, such that error rates, data rates and propagation paths and delays specified to obtain certain desirable channel characteristics. ODTS software system is quite extensible and different blocks can be easily implemented and integrated into the system using predefined templates and data structures.

1.8 Summary

The growth of wireless communications has been increasing exponentially over the past decade; creating demand for high-speed, reliable and spectrally efficient communication systems over the wireless medium. There are many challenges and tradeoffs in providing a high quality service in a dynamic wireless environment. These pertain to channel time variation (leading to channel estimation and synchronization problems), and limited resources such as spectral bandwidth. Therefore, an efficient wireless system is required to overcome these challenges in the next generation of wireless communication systems. OFDM is an enhanced multicarrier modulation technique for the next generation wireless communication systems [1].

Although OFDM technology has been known for several decades, its commercialization has only recently been possible with the advent of high-speed Very Large Scale Integrated (VLSI) circuits with lower cost, size and power consumption. With conventional or classical FDM systems the sub-carriers are not orthogonal, and therefore must be separated by guard bands that waste precious spectrum. In OFDM, by using an IFFT for modulation we implicitly choose the spacing of the sub-carriers in such a way that at the frequency where we evaluate the received signal all other signals are zero thus allowing the sub-carriers to overlap and we save precious bandwidth.

However, for an OFDM system to work in this way, the receiver and the transmitter must be perfectly synchronized (i.e., the topic of research in Chapter 4), and there should not be multi-path fading. The multi-path fading problem can be resolved if we implement a guard interval larger than the expected delay spread by artificially prolonging the symbol time and then removing this extension at the receiver. Thus, the multi-path fading problem is solved but with a slight loss in bandwidth, which will be illustrated in Chapter 2.

OFDMA is an advanced multiple access technology that can dynamically and efficiently allocate the limited resources to different users. However, there are some issues in terms of channel estimation and synchronization that have not been researched with an overall system, and need to be addressed and resolved. Most research and previous work have assumed perfect synchronization and perfect channel knowledge. But there are always difficulties and performance problems in overall practical OFDMA systems that one has to analyze and investigate with realistic scenarios before implementing a real hardware system.

We have to solve the problem regarding the synchronization and channel estimation of an OFDMA system. We propose an efficient channel estimation method in Chapter 3, and an enhanced synchronization technique that provide good performance and reliability in Chapter 4, and in Chapter 5 we apply these solutions to design and simulate a simple OFDMA system that can allocate resources dynamically. Lastly, after solving these problems we will conclude this thesis with suggestions for future research.

Chapter 2

OFDM Technology

This chapter covers a detailed analysis of the theoretical aspects of the OFDM technique. An OFDM system model is presented. OFDM signal design is considered and we will illustrate theoretically how the OFDM signal can become robust against multi-path fading effects by using a Cyclic Prefix (CP, i.e., a preamble, as explained and illustrated in Chapter 1 Section 1.2, on page 8). Furthermore, we will study and analyze multi-user OFDM or OFDMA systems.

2.1 OFDM system Model

Figure 2.1 illustrates the OFDM transmission system model. Data on OFDM sub-carriers is modulated or in other words mapped using common digital modulation schemes. For instance, IEEE 802.11a/g WLANs use QPSK or QAM where the serial binary data is converted into complex numbers representing constellation points. The constellation mappings are usually Gray-Coded. After the sequences of binary data undergo QPSK or QAM mapping and Gray-Coding, the complex data is converted into a parallel stream. The complex parallel stream of $X_{k,m}$ data symbols are coherently modulated on $N-1$ sub-carriers by an Inverse Discrete Fourier Transform (IDFT). The IDFT converts the parallel data into time domain waveforms denoted as $s(n)$.

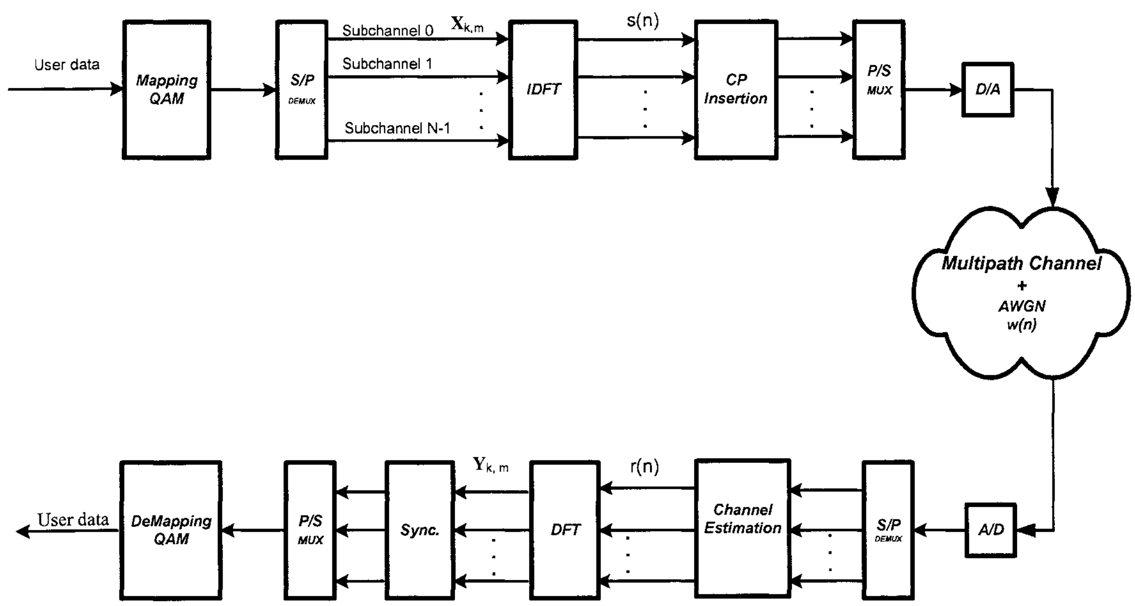

Figure 2.1 OFDM system model block diagram

At the transmitter, the last L samples are copied and put as a CP to form the OFDM symbol. The parallel data is converted into serial using a parallel to serial converter and the data vector is serially transmitted over the channel, whose impulse response is shorter than L samples. After digital to analog conversion the signal passes through a channel that will have the effects of additive noise and multi-path fading. OFDM can overcome these adversaries very easily.

At the receiver, after analog to digital conversion, the serial data is converted back to parallel. An efficient channel estimation using the Cyclic Prefix (CP) is performed, which is the topic of research in Chapter 3. Then the CP is removed. In OFDM systems that employ the CP, the frequency-selective channel distortion appears as a multiplicative distortion of the transmitted data symbols [4], and therefore, the received data symbol during the m^{th} OFDM symbol at the k^{th} sub-carrier becomes:

$$Y_{k,m} = H_{k,m}\, X_{k,m} + W_{k,m} \quad k = 0, 1, 2, \ldots\ldots, N\text{-}1 \;\&\; m = -\infty \cdots\cdots\cdots +\infty \tag{2.1}$$

where, $H_{k,m}$ is the channel gain at the k^{th} sub-carrier during the m^{th} OFDM symbol and $W_{k,m}$ is Additive White Gaussian Noise (AWGN). Equation (2.1) holds for every OFDM symbol, creating a two-dimensional grid with OFDM symbols (time) on one axis and sub-carriers (frequency) on the other axis.

The signal is demodulated with Discrete Fourier Transform (DFT) and converted back to frequency domain where synchronization is performed, which is our topic of research in Chapter 4. An interesting point to observe in OFDM signal processing is that, the sub-carrier pulse used for transmission is chosen to be rectangular by design. Therefore, Inverse Discrete Fourier Transform (IDFT) is able to perform the task of pulse forming and modulation. According to the theorems of the Fourier Transform, the rectangular pulse shape will lead to a *sin(x)/x* (or in other term *sinc*) type of spectrum of the sub-carriers as shown in Figure 2.2.

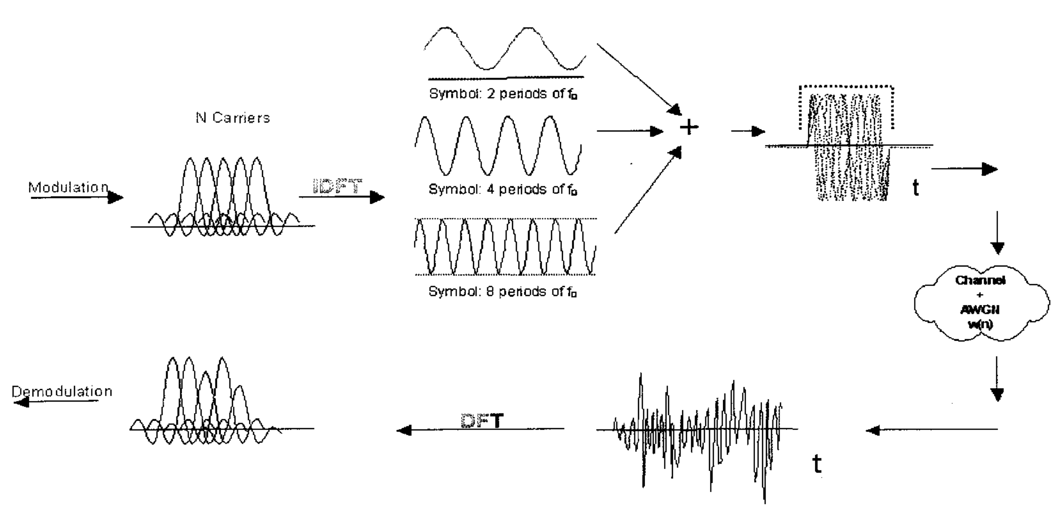

Figure 2.2 The Multi-carrier OFDM Signal processing

DFT and IDFT and can be implemented very efficiently as Fast Fourier Transform (FFT) and Inverse Fourier Transform (IFFT) [4]. These transforms are interesting from the OFDM perspective because they can be viewed as mapping data onto orthogonal sub-carriers. For example, the IFFT is used to take in frequency-domain data and convert it to time-domain data. In order to perform that operation, the IFFT correlates the frequency-domain input data with its orthogonal basis functions, which are sinusoids at certain frequencies. Hence, this correlation is equivalent to mapping the input data onto the sinusoidal basis functions.

In practice, an OFDM system treats the source symbols (e.g., the QPSK or QAM symbols that would be present in a single carrier system) at the transmitter as though they are in the frequency-domain. These symbols are used as the inputs to an IFFT block that brings the signal into the time domain. The IFFT takes in N symbols at a time where N is the number of sub-carriers in the system. Each of these N input symbols has a symbol period of T seconds. Recall that the basis functions for an IFFT are N orthogonal sinusoids. These sinusoids each have a different frequency. Each input symbol acts like a complex weight for the corresponding sinusoidal basis function. Since the input symbols are complex, the value of the symbol determines both the amplitude and phase of the sinusoid for that subcarrier. The IFFT output is the summation of all N sinusoids. Thus, the IFFT block provides a simple way to modulate data onto N orthogonal sub-carriers. The block of N output samples from the IFFT make up a single OFDM symbol. The length of the OFDM symbol is NT, where T is the IFFT input symbol period.

After some additional processing, the time-domain signal that results from the IFFT is transmitted across the channel. At the receiver, an FFT block is used to process the received signal and bring it into the frequency domain. Ideally, the FFT output will be the original symbols that were sent to the IFFT at the transmitter. When plotted in the complex plane, the FFT output samples will form a constellation, such as M-QAM. However, there is no notion of a constellation for the time-domain signal. When plotted on the complex plane, the time-domain signal forms a scatter plot with no regular shape. Thus, any receiver processing that uses the concept of a constellation must occur in the frequency-domain.

2.2 OFDM Signal Design

An OFDM signal consists of N orthogonal sub-carriers modulated by N parallel data streams. Choosing appropriate number of sub-carriers per available bandwidth is a very important point in OFDM. Let's see how we can achieve this in a most suitable way. The bandwidth of the system can be defined as $B \approx 1/T$, and the bandwidth of each sub-carrier is $1/NT$. First, the length of the CP should be chosen to be a small fraction of the OFDM symbol length to minimize the loss of SNR or the data rate. The size of the CP is directly related to the maximum delay spread τ_m of the channel. Therefore, a rule of thumb is that the length of the OFDM symbol $NT >> \tau_m$ or equivalently the number of sub-carriers $N >> \tau_m B$.

However, if the OFDM symbol length NT is too long the inter-channel interference caused by Doppler spreading in the fading channel can become performance limiting. If the inter-carrier spacing $1/NT$, is chosen much larger than the maximum Doppler frequency f_d, the system is relatively insensitive to the Doppler spread and the associated inter-channel interference. Therefore, the number of sub-carriers should also satisfy $f_d << 1/NT$ or equivalently $N << B/f_d$. The above two constraints result in the following constraint on the number of sub-carriers, i.e., an important design parameter:

$$\tau_m B << N << \frac{B}{f_d} \tag{2.2}$$

Equation (2.2) states a requirement on the delay and Doppler-spread of the physical channel for proper design of an OFDM system. Simplifying (2.2) further, the far left hand side and the far right hand side, lead to $f_d \tau_m << 1$. This means that the more the channel is under spread, i.e., the more correlated the channel is either in time or frequency, the easier it is to find a suitable number of subcarrriers.

2.2.1 OFDM signal

Let's take a look at the details of OFDM signal with a simple baseband signal example. Each OFDM baseband sub-carrier is of the form:

$$\phi_k(t) = e^{j2\pi f_k t} \tag{2.3}$$

where, f_k is the frequency of the k^{th} sub-carrier. One baseband OFDM symbol (without a cyclic prefix) multiplexes N modulated sub-carriers:

$$s(t) = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} x_k \phi_k(t) \qquad 0 < t < NT \tag{2.4}$$

where, x_k is the k^{th} complex data symbol (typically taken from a QAM symbol constellation) and NT is the length of the OFDM symbol.

We notice that as the OFDM signal is the sum of a large number of independent, identically distributed (i.i.d.) components. Therefore, its amplitude distribution becomes approximately Gaussian due to the Central Limit Theorem (CLT).

Another interesting phenomenon to notice is where the OFDM signal in (2.4) separates data symbols in frequency by overlapping sub-carriers, thus using the available spectrum in an efficient way. The sub-carrier frequencies f_k are equally spaced $f_k = k / NT$ which make the sub-carriers $\phi_k(t)$ orthogonal, on the interval, $0 < t < NT$.

Orthogonality is achieved because each sub-carrier has exactly an integer number of cycles in the interval T, and the number of cycles between adjacent sub-carriers differs by exactly one, as shown in Figure 2.3 [1].

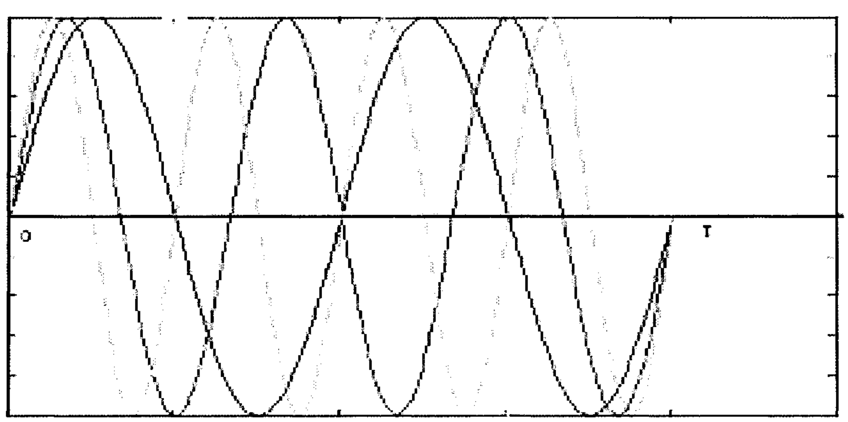

Figure 2.3 Sub-carriers that has exactly an integer number of cycles in the interval T

Each data symbol modulates the phase of a higher frequency carrier. In the frequency domain, the effect of the phase shifts in the carrier is to expand the bandwidth occupied by the modulated signal to a *sinc* function, as shown in Figure 2.4. The zeros of the *sinc* function occur at intervals of the symbol frequency; hence each orthogonal sub-carrier can be easily distinguished.

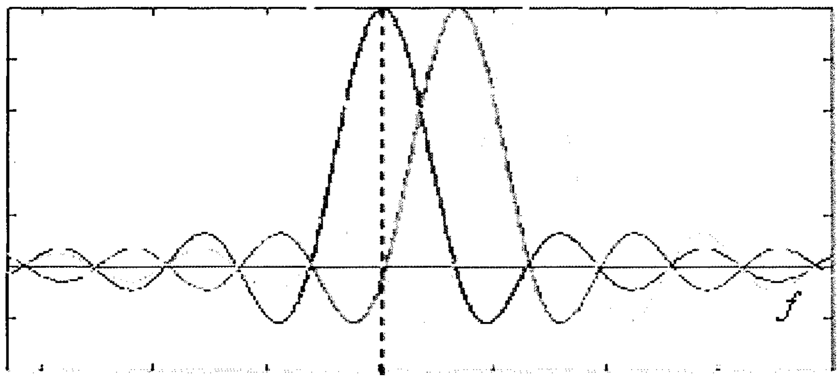

Figure 2.4 Orthogonal Sub-carriers in frequency domain

The receiver acts as a bank of demodulators that translate each carrier down to DC, with the resulting signal integrated over a symbol period to recover the raw data. The integration process results in zero contribution from all other sub-carriers, since the carriers are linearly independent, i.e., orthogonal to each other with carrier spacing $1/NT$ [11].

Let us illustrate this signal processing through a mathematical relationship. The OFDM symbol in (2.4) could typically be received using a bank of matched filters. However, an alternative demodulation is used in practice that we will illustrate. T-spaced sampling of the in-phase and quadrature components of the OFDM symbol will yield (*ignoring channel impairments such as additive noise or dispersion for simplicity*):

$$s(nT) = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} x_k e^{j2\pi\frac{nk}{N}} \qquad 0 \le n \le N-1 \tag{2.5}$$

which, is the IDFT of the constellation symbols x_k. The basis functions of the IDFT are orthogonal [15].

Accordingly, the sampled data is demodulated with a DFT. This is one of the key properties of OFDM, first proposed by *Weinstein and Ebert* [2]. The DFT, typically implemented with an FFT for faster operation and efficient implementation, actually realizes a sampled matched-filter receiver.

Two adversaries arise when the signal in (2.4) is transmitted over a dispersive channel. One is that channel dispersion destroys the orthogonality between sub-carriers and causes Inter-Carrier Interference (ICI). In addition, a system may transmit multiple OFDM symbols in a series so that a dispersive channel causes Inter-Symbol Interference (ISI) between successive OFDM symbols. The insertion of a silent guard period between successive OFDM symbols would avoid ISI in a dispersive environment but it does not avoid the loss of the sub-carrier orthogonality. Therefore, OFDM systems become quite sensitive to frequency offsets, and this calls for enhanced and efficient synchronization schemes.

The problem can be solved more effectively with the introduction of a CP, which is a copy of the few last bits added as a preamble to the OFDM symbols. The CP helps in maintaining and preserving the orthogonality of the sub-carriers and also preventing ISI between successive OFDM symbols, which promotes the use of OFDM in wireless systems in high data rate transmission systems such as WiMAX [20].

2.2.2 OFDM signal with Cyclic Prefix

The cyclic extension of the OFDM symbol is inserted as a guard period between consecutive OFDM signals. The OFDM signal (2.4) is extended over a period Δ so that:

$$s(t)=\frac{1}{\sqrt{N}}\sum_{k=0}^{N-1}x_k e^{j2\pi f_k t} \qquad -\Delta<t<NT \tag{2.6}$$

The signal then passes through a channel, modeled by a Finite-length Impulse Response (FIR) limited to the interval $[0,\Delta_h]$, where Δ_h is the length of the channel impulse response. If the length of the cyclic prefix Δ is chosen such that $\Delta>\Delta_h$, the received OFDM symbol evaluated on the interval $[0,NT]$,*(after the CP has been dropped)*, ignoring any noise effects, becomes:

$$r(t)=s(t)*h(t)=\frac{1}{\sqrt{N}}\sum_{k=0}^{N-1}H_k x_k e^{j2\pi f_k t} \qquad 0<t<NT \tag{2.7}$$

where, $H_k=\int_0^{\Delta_h}h(\tau)e^{-j2\pi f_k\tau}d\tau$ is the Fourier transform of $h(\tau)$ evaluated at the frequency f_k. Where, $f_k=k/NT$, are the sub-carrier frequencies that are orthogonal on the interval $0<t<NT$. Note that within this interval the received signal is similar to the original signal except that $H_k x_k$ modulates the k^{th} sub-carrier instead of x_k. In this way the CP preserves the orthogonality of the sub-carriers.

The OFDM signal can be demodulated as described in the previous section, taking a FFT of the sampled data over the interval $[0,NT]$, ignoring the received signal before and after $0<t<NT$. The received data (*disregarding additive noise for simplicity*) then has the form:

$$y_k=H_k x_k \qquad k=0,......,N-1 \tag{2.8}$$

The received data in Equation (2.8) can be recovered with very simple N parallel one-tap equalizers. This simple channel equalization motivates the use of a CP and often the use of OFDM itself. Moreover, the CP also acts as the above-mentioned silent guard period preventing ISI between successive OFDM symbols.

2.2.2.1 Cyclic Prefix helps in maintaining orthogonality

Adding a CP, which is a copy of the last few bits to the beginning of an OFDM symbol, makes the channel appear circular. By adding a CP, the transmitted signal will appear periodic.

The received signal can be written as:

$$r(t) = s(t) * h(t) \qquad -\Delta \le t \le NT \tag{2.9}$$

where, * denotes the convolution operator.

If the cyclic prefix added is longer than the impulse response of the channel, the linear convolution in the channel will appear as a circular convolution, from the receiver's point of view. This is shown below for any sub-carrier *n*, on $-\Delta \le t \le NT$. Where in Equation (2.10), circular convolution is denoted by $\otimes$.

$$\begin{aligned} R(n) = DFT(r(t)) &= DFT(s(t)) \otimes h(t)) \\ &= S(n)H(n) \end{aligned} \tag{2.10}$$

Therefore, by adding the CP, orthogonality is maintained throughout the transmission if the transmitter and receiver are synchronized properly. Another advantage of using a CP is that it acts as a guard space between adjacent OFDM frames, thus the problem with inter-frame interference disappears. This is maintained as long as the CP is at least the length of the channel impulse response. The drawback of using a CP is that the amount of data to be transmitted is increased, requiring additional transmit energy.

The loss of transmit energy (or loss of signal-to-noise ratio (SNR)) due to the cyclic prefix can be defined as:

$$E_{loss} = \frac{NT}{NT + \Delta} \tag{2.11}$$

This is also a measure of the bit rate reduction required by a CP. That is, if each sub-carrier can transmit b bits, the overall bit rate in an OFDM system is $\frac{Nb}{NT + \Delta}$ bits per second, as compared to the bit rate of $\frac{b}{T}$ in a system without a CP. If latency requirements allow, these losses can be made small by choosing a symbol period NT much longer than the length of the CP, i.e., Δ.

2.2.3 OFDM Channel Noise and Doppler spread

We now discuss the radio channel impairments that an OFDM signal can experience. OFDM systems often experience not only channel dispersion as addressed in the previous section, but also Additive White Gaussian Noise (AWGN), Doppler spreading and synchronization errors. Many of these channel impairments can be modeled as AWGN if they are small. However, synchronization errors, such as carrier frequency offsets, carrier phase noise, sample clock offsets and symbol timing offsets are major problems that can cause degradation in an OFDM system and are discussed in detail in Chapter 4.

We now include the Gaussian Noise in our modeled Equation (2.4), which would give us a received OFDM signal of the form $r(t) = s(t) * h(t) \; + \; n(t)$, extending this to Equation (2.8) we have:

$$y_k = H_k x_k + n_k \qquad k = 0, \ldots\ldots., N-1 \tag{2.12}$$

where, n_k is the FFT of the sampled noise terms $n_t(nT)$, $n = 0, \ldots..N-1$.

In a fading channel, the channel variations affect the performance of an OFDM system. An interesting observation is that OFDM becomes quite sensitive to channel variations, when the OFDM symbol length increases with the number of sub-carriers for a fixed sampling period. We will discuss this issue in Chapter 3, where we study and analyze the OFDM system in a time varying environment. For now, let's consider an OFDM system in a flat-fading channel, with a time varying one-tap impulse response $a(t)$. When the transmitted OFDM signal is multiplied with this time varying scalar we get a received signal $r(t) = a(t)s(t)$. The multiplication in time domain will appear as a convolution in frequency domain and thus, causing spreading of the sub-carrier and therefore yielding to adverse effect of ICI.

The sampled signal after DFT is of the form [21] :

$$y = \sum_{k=0}^{N-1} x_k A(k) \tag{2.13}$$

where, $A(k)$ is the DFT of the time varying channel tap $a(nT)$, $\quad n = 0, \ldots\ldots., N-1$

For a fixed sampling time the ICI due to the Doppler spreading increases with the number of carriers. Russell and Stuber [22], use a central limit theorem argument to characterize the effect of the ICI as an additive Gaussian noise with a variance that increases with the number of sub-carriers and with the maximum Doppler frequency.

This noise is correlated in time, but white across sub-carriers. The ICI leads to an error floor, which may be unacceptable. Interested reader may refer to [22], to see how antenna diversity and coding is discussed and demonstrated to reduce this error floor.

2.3 Channel Estimation in OFDM systems

In wireless communication systems, channel in its most general sense can be described as everything from the source to the sink of the radio signal, including the physical medium [23]. Channel estimation is a process of determining the effect of the physical channel on the input sequence. Channel estimation allows the receiver to approximate the impulse response of the channel and to characterize the effect of the channel on the signal. Channel estimation is essential for removing inter-symbol interference and noise rejection techniques. It is also used in diversity combining, Maximum Likelihood (ML) detection and angle of arrival estimation. Furthermore, it is used to bring channel diversity and to dynamically allocate a radio base station's resources such as bandwidth and power to different users based on their channel state information.

According to Peled [5], the receiver does not require a time domain equalizer or channel estimator if a CP is properly used and if the channel does not change. But in reality the channel changes due to movement of the terminals and or objects around them. If the channel changes, we need an adaptive channel estimate and this requires some type of retraining which can be quite costly and bandwidth consuming. Therefore, the CP method could be used to our advantage in the case of time varying channels to adaptively estimate the channel without additional retraining.

In a wireless OFDM system the knowledge of channel information is very important. It is shown in [24] that non-coherent detection, such as differential modulation, used in OFDM results in major performance loss compared to the coherent detection particularly at higher bit rates. At low bit rate systems differential modulation such as Differential Phase-Shift Keying (DPSK) can be used. The differential scheme has the advantage of avoiding channel estimation. However, in high bit-rate systems multi-amplitude modulation is required. We could use Differential Amplitude Phase-Shift Keying (DAPSK) to support higher bit rates, but in such a system there is an intrinsic problem, i.e., the constellation points are non-uniformly distributed in the signal space, which reduces the performance. Therefore, coherent modulation is a better choice and gives better performance, yet because of the necessity of channel estimation, it gives rise to more complexity in the receiver design.

At the OFDM receiver after down conversion and analog-to-digital conversion, the Fast Fourier Transform (FFT) is used to demodulate the N sub-carriers of the OFDM signal. For each symbol the FFT output contains N QAM values. However, these values contain random phase shifts and amplitude variations caused by the channel response, local oscillator drift, and timing offsets. It is imperative to have the knowledge of the reference phase and amplitudes for all sub-carriers so that the QAM symbols can be converted to binary decisions [1]. A channel estimation technique will be presented to find the references without introducing a great deal of overhead.

2.3.1 Channel Estimation Process

Figure 2.5 represents the channel estimation process. Estimation error is denoted by e(n). A channel estimation scheme tries to minimize the mean square error criterion, $E[e^2(n)]$.

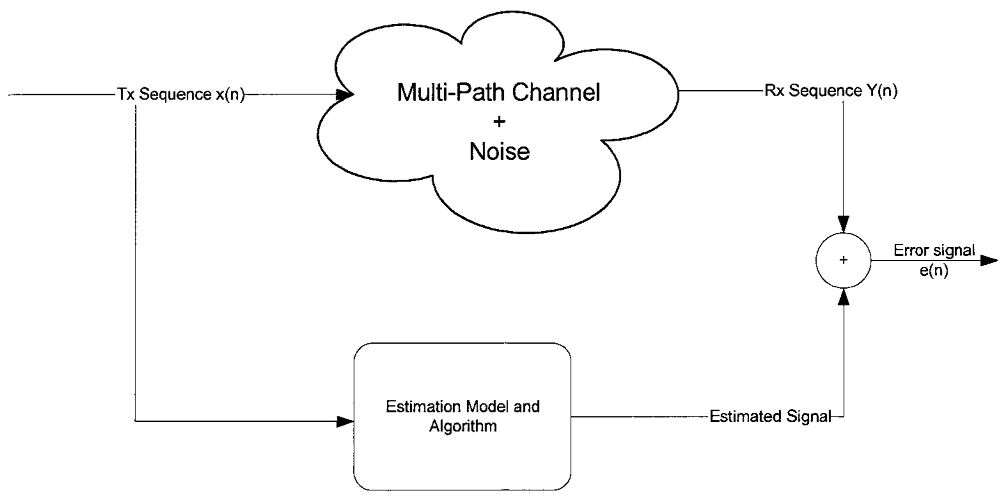

Figure 2.5 Channel Estimation Process

The channel can have a random or nonrandom impulse response. If the autocorrelation function of the transmitted signal is a unit impulse, i.e., the transmitted signal is a white noise, a correlator is used to estimate the impulse response of the time-invariant linear channel [25]. Hence, in the case where the channel is time-invariant, only some initial training or pilots are used to produce the channel estimation. However, if the channel is time varying in nature, then special techniques as shown in Chapter 3, must be applied to estimate and track the channel.

2.4 Synchronization in OFDM systems

It is well known that OFDM is very sensitive to frequency and time offsets [1], which can destroy orthogonality among sub-carriers and thereby cause Inter-Carrier Interference (ICI) and Inter-Symbol Interference (ISI) respectively. Therefore, proper synchronization is necessary to keep the orthogonality of the sub-carriers, which is essential for a reliable transmission. This property becomes more challenging for multi-user OFDM or in other words OFDMA systems, where a number of users share one OFDM symbol simultaneously. Furthermore, synchronization is crucial because synchronization problems can also lead to grave errors in channel estimation. Therefore, it is important to reduce the channel dependency in frequency synchronization, which is important for securing orthogonality and preventing ICI effects, and in order to have a reliable and enhanced system performance [14].

2.5 Summary

OFDM is an enhanced multi-carrier modulation technique that is resilient to interference and fading, which are considered as the adversaries of wireless transmission. If the CP length is longer than the longest channel delay then both inter-symbol and inter-carrier interference can be avoided. In addition, OFDM brings an improved immunity to fast fading due to its multi-carrier property [4]. OFDM is also quite versatile against frequency selective fading. In a single carrier system a single interferer can destroy the whole carrier, but in a multi-carrier system only a small amount of the sub-carriers will be affected.

In an OFDM system the benefit of adding CP are numerous. The major benefit is that it avoids or eliminates ISI that is caused by multi-path propagation. In Chapter 3, we will extend the use of CP in our research. We will propose an efficient channel estimation technique with the help of CP in order to design a more realistic and practical OFDM system in a time varying environment.

Chapter 3

An Efficient Channel Estimation Technique

This chapter discusses various channel estimation techniques and proposes an efficient channel estimation scheme for OFDM systems. We present a better way of dealing with time varying channels and estimating the channel state information efficiently using an adaptive signal processing technique. The system model will be simulated under different scenarios. Simulation results demonstrate the superiority of our proposed solution in a time varying environment.

In the presence of time-varying conditions, a time delay between the collection of the channel estimate and the adaptation of the modulation at the transmitter result in an inaccuracy between the estimate and the actual channel. A remedy to this problem is to send the retraining symbols more often so that a channel estimate is accurately taken when the channel has changed, but sending retraining symbols more often is quite bandwidth consuming and inefficient. The aim of this chapter is to evaluate the problems with estimation of the channel in a time varying environment, and to propose an improved process and efficient channel estimation technique for OFDM systems.

3.1 Brief History of Channel Estimation

The history of channel estimators started in 1950 when Lee and Wiesner [25] used a correlator for estimation of a filter impulse response. Levin [26] studied the same problem with finite data record and derived a Least Square (LS) channel estimator consisting essentially of a matched filter and an equalizer for the autocorrelation function sidelobes of the measurement signal. Kailath [27] extended the result to Minimum Mean Squared Error (MMSE) estimation. The LS and MMSE are the most commonly used techniques. These estimators can be easily interpreted as a bank of matched filters, a set of samplers and a set of Inverse Discrete Fourier Transforms (IDFT).

3.2 Different Channel Estimation Techniques

Signal reception delays in real wireless systems make only partial and delayed channel information available at the base station. With the information available, it is our job to make a proper estimate without adding too much overhead and reducing the system capacity.

A wireless channel can be estimated in time or in frequency domain. Time domain estimation error is slightly higher than frequency domain. However, the computational complexity of frequency domain method is higher than time domain method. Using frequency domain pilot symbols, the pilot symbols need IDFT computation, which may cause Peak-to-Average-Power-Ratio (PAPR) problem. Due to the complexity and PAPR that we face in frequency domain, we choose to solve the channel estimation problem in OFDM systems with the time domain estimation approach that is less complex and faster. We will determine, using our method, how we can achieve maximum efficiency in a time varying channel.

For the Channel State Information (CSI) estimation, three classes of methods are available:

1- Blind methods, which estimate CSI merely from the received symbols

2- Differential schemes that bypass CSI estimation by differential encoding

3- Input-output methods that rely on training symbols that are known to the receiver.

Relative to training-based schemes, differential approaches incur performance loss by design [24], [28], while blind methods typically require longer data records and entail higher complexity [29]. On the other hand, the insertion of known training symbols can be optimal but bandwidth consuming.

Data aided channel estimation based on pilot symbols is well motivated for the practical implementation of multi-carrier systems [30]. If pilot symbols were sent directly in time domain, it would be called training sequences. If pilot symbols were sent in the frequency domain after the IDFT operation, it would be called pilot tones. Figure 3.1, shows the block diagram of data aided channel estimation system model.

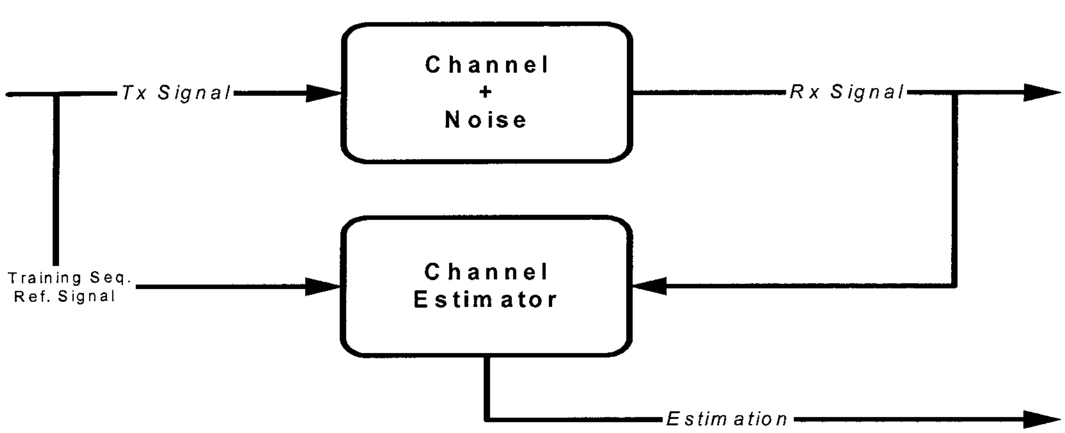

Figure 3.1 Data Aided Channel Estimation System Model

Pilot symbol can be sent every several blocks or in every block. In the first case, one OFDM symbol is dedicated to pilot tones or training sequences and the estimated channel is used for the following OFDM symbols until the next estimate is obtained. In the second case, pilot tones or training sequences are sent together with other data symbols in each block. The performance of the second case is better than that of the first case when the channel is of type time varying and fast fading. Nevertheless, more bandwidth is utilized. In order to be bandwidth efficient and estimate the time varying channel appropriately, we devise an estimation technique and perceive the CP as retraining sequences sent together with data symbols in each block.

Hence, data aided training or retraining symbol based channel estimation is less complex but adds a lot of redundancy and therefore, we often have too much redundant information to transmit. If the channel is of time varying nature then initial training method will suffer from performance and retraining methods are quite inefficient. We can achieve great results using our improved method of estimating the channel using the CP that is always there in the OFDM signal structure. Therefore, the optimal method is to use the CP that we are adding in order to combat ISI and channel impairments.

The CP can be used in an efficient way to estimate the channel conditions. J. Beek [31], has addressed this property and has proposed a scheme for slow fading time invariant channel environment. However, his method has certain performance limitations in practical time varying environment. We propose an improved and efficient method for time varying channels. We will show that our method performs very well in time varying channels and does not require any retraining or additional bandwidth to transmit the channel information.

3.3 Proposed Dynamic Channel Estimation

Dynamic resource allocation is extremely important for achieving an efficient OFDM system. By using channel diversity we can assign resources appropriately and optimally. Good knowledge of channel state information is vital in this case. Perfect knowledge of the channel is an immense imprecise assumption, which is the root of the performance problems, because a practical mobile OFDM wireless environment under study is of time varying fading nature. Imperfect channel information arises from limitations of channel estimation techniques as well as outdated information due to time varying channel conditions. We will present an efficient method where we can apply a superior dynamic estimation algorithm, without affecting the bandwidth. In addition, we will investigate various factors that can cause the channel information to be imperfect.

The estimation techniques proposed in most previous works for OFDM do not take the special structure of the OFDM symbols with a Cyclic Prefix (CP) into consideration, e.g., [24], [32], [33], and [34]. The training algorithms in [32] and [33], are not optimal because in a time varying environment [32] will perform poorly, giving too many errors, and [33] will require too many retraining of the extra pilot symbols and will be quite inefficient. Moreover, the method in [31] that takes CP into account, suffers from performance problems and is unable to track the channel variations in a time varying environment. We will introduce a new and improved method of dynamically obtaining a channel estimate in a time-varying environment by using the cyclic prefix with no additional bandwidth for retraining.

3.4 Proposed Channel Model

A channel model is a mathematical representation of the transfer characteristics of the physical medium. The channel model could be based on some known underlying physical phenomenon or it could be formed by fitting the best statistical model on the observed channel behavior. Usually, channel models are formulated by observing the characteristics of the received signal under certain environment, and the one that best explains the received signal behavior is used to model the channel. Using the channel model we apply the channel estimation techniques that are mathematical representation of what is truly happening. If the channel is assumed to be linear, the channel estimate is then the estimate of the impulse response of the system. However, for different systems we need to find the appropriate channel estimate that is reliable, efficient and is less complicated to compute.

Most of the systems designed assume that the wireless channel can be regarded to be as time invariant. This has a number of important implications and consequences such as:

I- Mobility: Mobility introduces Doppler shifts and makes the channel time varying. Consequently, the assumption that the channel is time invariant only holds approximately, and will affect wireless systems that are designed based on this assumption in the following ways:

a) Many wireless applications, such as 3G (Third Generation Systems) and WiMAX, can adapt their data rate to the link quality. If the system design is based on the time-invariance assumption, mobility will reduce the link quality and hence the data rate will drop.

b) For some systems, the violation of the time-invariance assumption can even have adverse consequences. This is for instance true for Digital Video Broadcasting (DVB) systems, which were originally developed for fixed receivers. Currently, they are also envisioned for mobile use, but due to the highly time-varying nature of the propagation environment, they basically break down at even at low speeds, e.g., a walking pedestrian.

II- Carrier frequency offsets: Current wireless systems are very sensitive to carrier frequency offsets, and require expensive oscillators and/or complex carrier frequency offset cancellation algorithms to combat this effect.

Judicious and innovative wireless system design that takes the time-varying nature of the channel into account is the key to solve the adverse effects described above. This is one of the objectives of this thesis. Consequently, in future wireless systems, the time-varying nature of the channels will become more and more important. That is why in this thesis we aim at providing solutions for future wireless system design that exploit the time-varying behavior of the environment.

In our channel modeling, we make the following assumptions:

- o Assumption: Channel is modeled as time varying.
- Justification: Our assumption is reasonable and can be justified due to the fact that the wireless channel model under research on which a practical OFDM system will operate is of time varying in nature because of the movement of the mobile station or the objects in the surroundings.

- Assumption: Channel is modeled as multi-path Rayleigh fading channel.

- Justification: Multi-path Rayleigh fading channel is assumed due to multiple copies of signal arriving at the receiver after reflection from surroundings under Non-Line of Sight (NLOS) operation.

- Assumption: The channel is flat fading multi-path channel, which means that the bandwidth of the signal is less than channel bandwidth and also the symbol period is greater than the delay spread of the channel.

- Justification: Channels are flat fading multi-path channels due to the fact that in OFDM technology, the available bandwidth is split into several orthogonal sub-carriers, and transmitting information slowly in parallel on these sub-carriers [4]. In the frequency domain each sub-carrier will occupy only a small frequency interval where the channel frequency response will be almost constant during an OFDM symbol. Therefore, each sub-carrier will experience a flat fading channel.

In order to classify the time characteristics of the channel, we define τ_c (Coherence time) and f_D (Doppler spread) parameters. The coherence time is the duration over which the channel characteristics do not change significantly [1]. The time variations of the channel are also evidenced as a Doppler spread in the frequency domain. The range of values of frequency over which the Doppler power spectrum is essentially nonzero is called Doppler spread. There is no exact relationship between coherence time and Doppler spread. However, if the coherence time is defined as the time over which the time correlation function is above 0.5, then the reciprocal of

Doppler spread is can be approximated to the coherence time of the channel as $\tau_c = 9/16\pi f_D$ [1]. Therefore, if the channel has a large coherence time, equivalently it has a small Doppler spread.

Doppler spread is caused by the differences in Doppler shifts of different components of the received signal, if either the transmitter of receiver is in motion. The frequency shift is related to the spatial angle between the direction of arrival of that component and the directions of vehicular motion. If a mobile is moving at a constant speed v, the Doppler shift $f_m = (v/\lambda)\cos\theta$ [1].

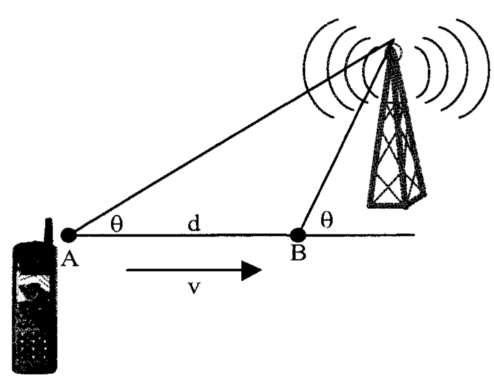

Figure 3.2 Doppler frequency shift phenomenon

The difference between the highest shift and the lowest shift will give the Doppler spread f_D. The maximum Doppler shift occurs at $\theta = 0$, and we can say that the minimum Doppler shift that occurs is when the speed of the mobile is zero. Therefore, we can approximately calculate out Doppler spread as $f_D = f_c(v/c)$, where $\lambda = c/f_c$, f_c is the carrier frequency and, $c = 3.0\times10^8\ m/s$ the speed of light [35].

We will make one more assumption to help us clearly and completely characterize the channel model.

- o Assumption: Channel is modeled as a slow fading channel
- Justification: The maximum Doppler spread that we will encounter is 300Hz moving at 100Km/hr. This will give us a coherence time of $\tau_c = 9/16\pi f_D$,i.e., 600 $\mu\,sec$, which is a lot larger than the symbol period $T_s = 160\,\mu\,sec$. Therefore our channel is slow fading with a low Doppler spread and coherence time is greater than the symbol period.

Now we are ready to model the channel based on the assumptions stated above. We will model and implement the channel as a linear Finite Impulse response (FIR) filter with length v. The multi-path fading channel can be represented as a *tapped delay line* with time varying coefficients and fixed tap spacing. If $h_{l,k}$ is the channel tap gain for the l^{th} path at time k, then the time variant frequency response of the channel for the i^{th} sub-carrier at time k is [36]:

$$H_{i,k} = \sum_{l=0}^{v-1} h_{k,l} e^{\frac{-j2\pi il}{m}} \tag{3.1}$$

An FIR filter can be easily implemented using just three types of digital hardware elements, a unit delay i.e., a *latch*, a multiplier, and an adder. The unit delay simply updates its output once per sample period, using the value of the input as its new output value. We notice that at each n we need access to $x(n)$, $x(n-1)$, $x(n-2)$, …, $x(n-v)$.

We can maintain this set of values by cascading a set of latches to form a *delay line*, as shown below in Figure 3.3 :

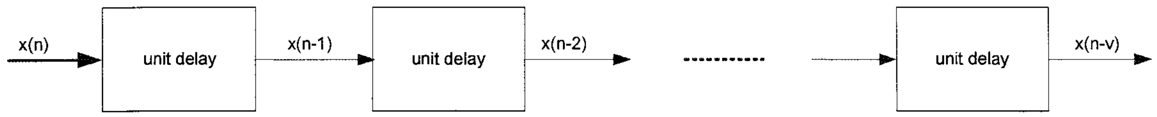

Figure 3.3 Set of latches to form a delay line

For each integer *n*, the output sample is the values in the delay line scaled by *g(1,k), g(2,k), ..., g(v, k)*. To obtain these values, we simply tap the delay line, as shown in Figure 3.4

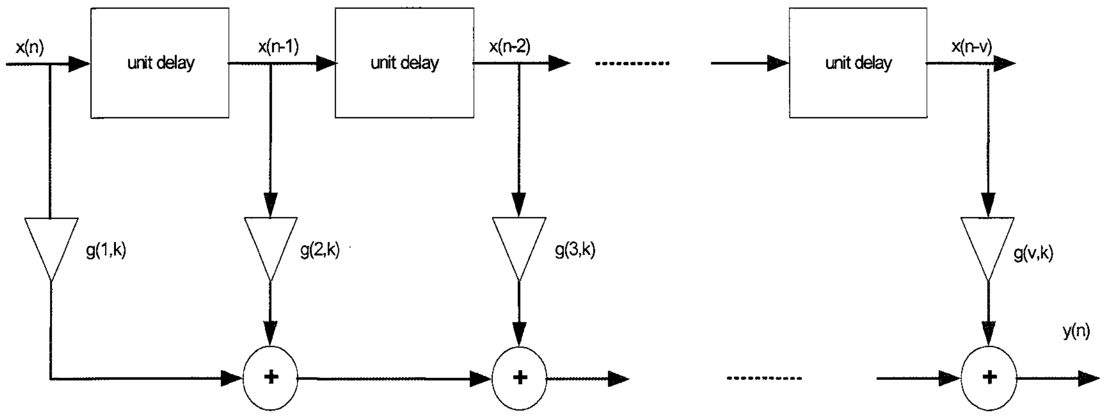

Figure 3.4 Tapped delay line implementation of FIR filter

The triangular boxes denote multipliers that multiply the channel coefficients. The circles denote adders. The above picture shows a *tapped delay line* implementation of an FIR filter.

Figure 3.5 and 3.6 shows the amplitude and impulse response of the time varying filter at different instants of times T1, T2, and T3. We can observe how the filter response changes with time.

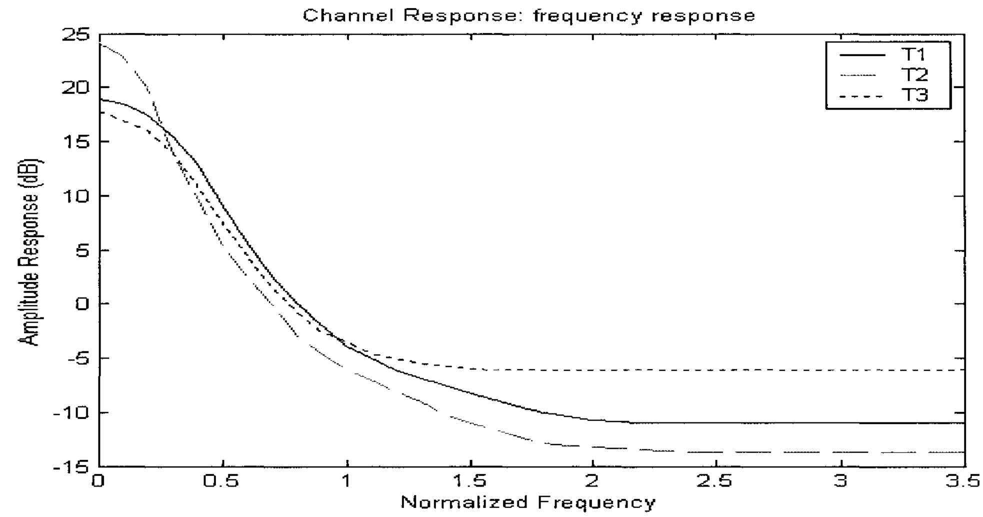

Figure 3.5 Channel Response: Amplitude Response

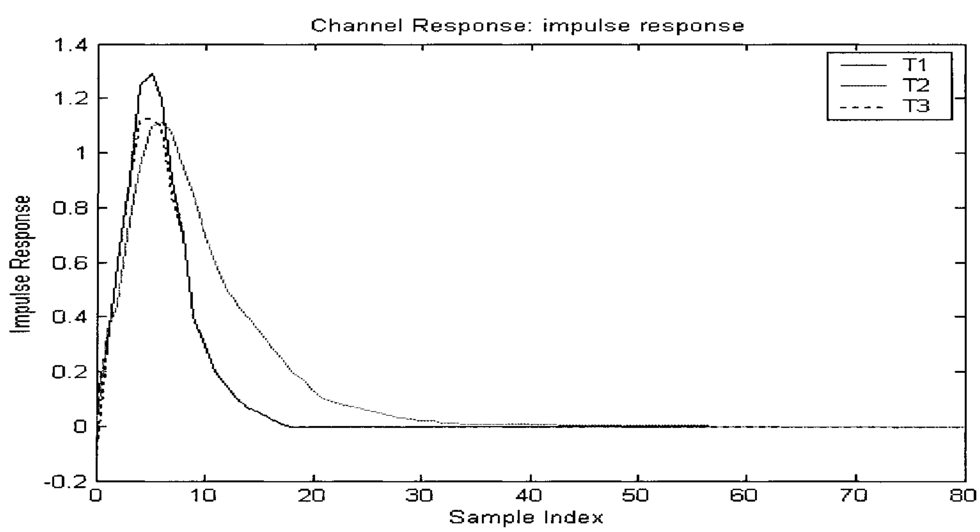

Figure 3.6 Channel Response: Impulse Response

3.5 Proposed Efficient Channel Estimation Scheme for OFDM Systems in a Time Varying Environment

Past and current research in wireless communications mainly studies transmissions over time-invariant channels. Some results in time varying area are reported, but either the channel model is restricted, optimal training designs are unknown, or the overhead on bandwidth is ignored. For example in [31], the solutions is designed only for time invariant channels and it performs poorly with an error floor if the channel characteristics changes in certain environment that wireless devices should normally operate. In systems such as [32], [33], estimation is performed by a known training sequence that is sent by the transmitter. Training algorithm is performed by the receiver based on the observed channel output and the known input to estimate the channel. The Least Squares (LS) channel identification algorithm in [33] is a simple training approach. However, it is not suited for time varying systems and dispersive channels. It has either been assumed that the channel is invariant and use the initial training to get the channel estimation [32], or training sequences has been employed periodically to track the channel variation [33]. These two solutions will either cause performance degradation or increase the overhead of the system respectively.

The CP that is constantly sent to combat ISI is normally discarded at the receiver. However, we can use it efficiently and regard it as a constant training sequence that can help in channel estimation without introducing extra overhead into the system. Therefore, we can adaptively estimate the channel without additional training sequence. We perform the estimation using an adaptive signal processing techniques using Recursive Least Square described in [17], [37]. The distinctive aspect of our methodology is that we setup a more realistic channel scenario and assess the variations and then we apply the cyclic prefix method to dynamically perform channel estimation. As a result, our algorithm can adaptively track the channel variations without additional training sequences, and has better performance then conventional estimation techniques.

3.5.1 System framework of the proposed channel estimation technique

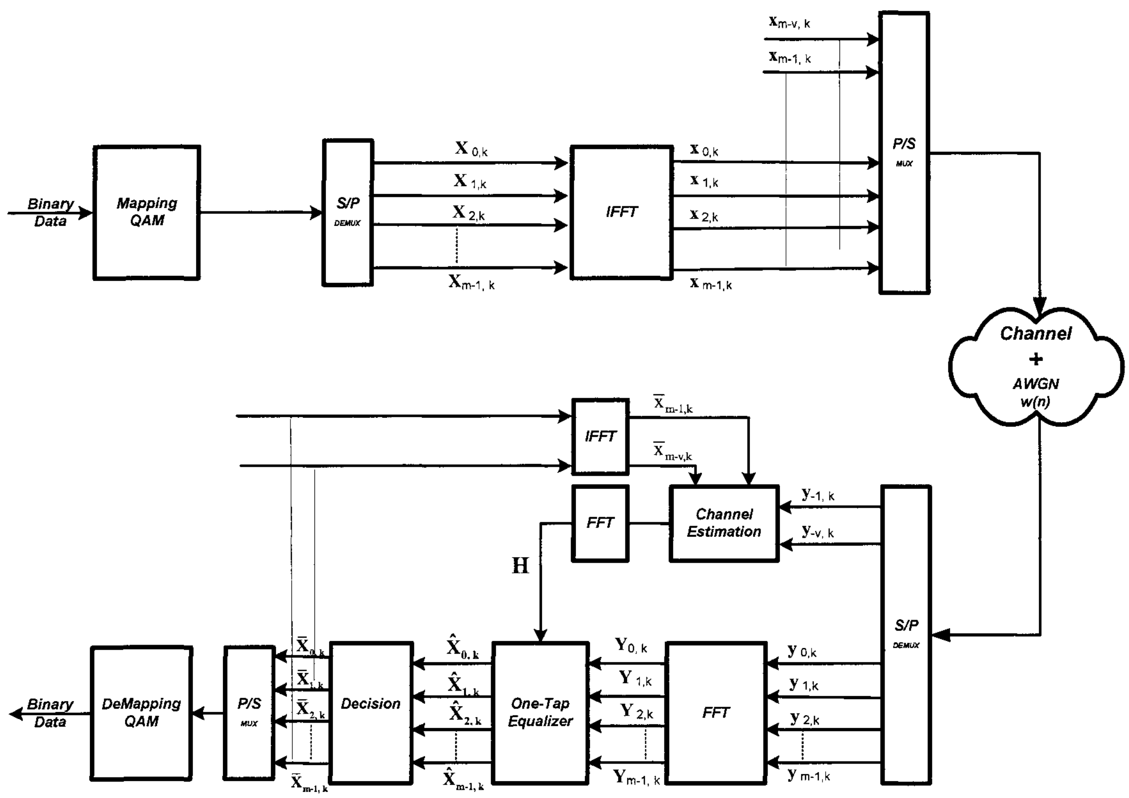

Figure 3.7 System framework for the proposed channel estimation methodology using CP

The system framework model of the proposed channel estimation is presented in Figure 3.7. The system consists of m complex parallel sub-carriers. The binary input data are divided into m bit streams and mapped into QAM constellation points $X_{i,k}$. The modulation is performed by an m-point IFFT due to the easy implementation and performance reasons as explained in Chapter 1. The output of IFFT consists of all real samples $x_k = [x_{0,k}, x_{1,k}, ...x_{m-1,k}]^T$. Further, the parallel stream is converted back into serial and then transmitted over the channel.

We model and implement the channel as a linear FIR filter with length v as described in the previous section. The impulse response of the channel is $h = [h_{0,k}, h_{2,k}, ..., h_{v-1,k}]^T$. A cyclic prefix (CP) $x_k^{(cp)} = [x_{-v,k} .... x_{-1,k}]^T$, where $x_{-i,k} = x_{m-i,k}$, $i = 1, ..., v-1$ is appended in front of x_k before transmission.

At the receiver only the prefix $y_k^{(cp)} = [y_{-v,k}, ..., y_{-1,k}]^T$ is fed into the channel estimator. We will, shortly demonstrate how the channel estimator uses this information to perform channel estimation.

The FFT demodulation is performed on $y_k = [y_{0,k}, y_{1,k}, ..., y_{m-1,k}]^T$ and we get the demodulated signal $Y_k = [Y_{0,k}, Y_{1,k}, ...Y_{m-1,k}]^T$. Since the sub-carriers in OFDM are independent of each other we have:

$$Y_{i,k} = X_{i,k} H_{i,k} + N_{i,k} \qquad i = 0 \cdots\cdots\cdots m-1 \text{ and, } k = 0 \cdots\cdots\cdots mT \tag{3.2}$$

where, $H_{i,k} = \sum_{l=0}^{v-1} h_{l,k} e^{\frac{-j2\pi il}{m}}$ is the channel frequency response, and $N_{i,k} \sim N(0, \sigma_{i,k}^2)$ is the noise of the i^{th} sub-carrier.

We can use a one-tap equalizer to obtain the optimal MMSE estimation of $X_{i,k}$. In other words, using MMSE we will obtain the estimate $\hat{X}_{i,k} = Y_{i,k} \cdot w_{i,k}$. The coefficient of the one-tap equalizer for i^{th} sub-carrier is [17] :

$$w_{i,k} = \frac{M_{i,k}^{1/2} H_{i,k}^{*}}{M_{i,k} \left\| H_{i,k} \right\|^2 + \Delta_{i,k}}, \qquad i = 0, \cdots\cdots, m-1 \tag{3.3}$$

where, $M_{i,k}$ is the transmit power of $X_{i,k}$, $M_{i,k} \triangleq E[\|X_{i,k}\|^2]$ and $\Delta_{i,k}$, can be defined as, $\Delta_{i,k} \triangleq E[\|N_{i,k}\|^2]$. Decision is made on $\hat{X}_{i,k}$ to get the final output $\bar{X}_{i,k} = q(\hat{X}_{i,k}),\quad i = 0,\cdots\cdots,m-1$, where $q(\cdot)$ is a quantization function.

For many types of channel estimation methods that employ the training sequence, we observe that channel information is obtained by training. In time invariant channels the channel information is obtained at the initial training and is kept. In time varying channels, the training and retraining must be performed periodically to get the new channel state information. This will add an overhead to the system and if it is not performed at proper times the system performance degrades. By using the CP method, the system is constantly updated and new channel estimation is performed to track the channel variations. The relationship between the cyclic prefix and the transmitted signal is:

$$y_k^{(cp)} = T_k h_k + n_k^{(cp)} \tag{3.4}$$

where, $T_k = \begin{bmatrix} x_{-v,k} & x_{m-1,k-1} & \cdots\cdots & x_{m-v,k-1} \\ \ddots & & \ddots & \vdots \\ x_{-1,k} & \cdots & x_{-v,k} & x_{m-1,k-1} \end{bmatrix}$, $n_k^{(cp)} = \left[n_{-v,k} \cdots\cdots n_{-1,k}\right]^T$ and

$h_k = [h_{0,k}, h_{2,k}, \ldots\ldots, h_{v-1,k}]^T$

The lower triangle part of matrix T_k is composed of $x_k^{(cp)}$, while the upper part is the last $v-1$ samples of x_{k-1}. The last $v-1$ samples of the x_{k-1} are also elements of the prefix $x_{k-1}^{(cp)}$.

Hence, if all the prefix parts concatenate to form a pair of sequences :

$$\left\{x_l^{(cp)}\right\} = \left\{\cdots x_{-v,k-1} \cdots x_{-1,k-1} x_{-v,k} \cdots x_{-1,k} \cdots\right\} \tag{3.5}$$

and,

$$\left\{y_l^{(cp)}\right\} = \left\{\cdots y_{-v,k-1} \cdots y_{-1,k-1} y_{-v,k} \cdots y_{-1,k} \cdots\right\} \tag{3.6}$$

then the relationship between these two satisfy the Equation:

$$y_l^{(cp)} = x_l^{(cp)} * h_l + n_l \tag{3.7}$$

Equation (3.7) shows that if we send $\left\{x_l^{(cp)}\right\}$ to the channel as the training sequence, the channel output is exactly $\left\{y_l^{(cp)}\right\}$. We can use the training sequence provided by the CP to estimate the channel if we can recover $\left\{x_l^{(cp)}\right\}$ correctly. The Recursive Least-Square (RLS) algorithm [17], is chosen to obtain the channel estimate from (3.7) because of the good tracking property of RLS [38]. We observe that the data in (3.4) arrive in blocks therefore we use block RLS method that updates the channel estimation by blocks.

The channel estimation at time k is $\hat{h}(k) = \Phi^{-1}(k)z(k)$, where $\Phi(k) = \lambda_1 \Phi(k-1) + \sum_{l=1}^{v} \lambda_2 u_l(k) u_l^H(k)$ is the approximation of the correlation matrix of the training input $\left\{x_l^{(cp)}\right\}$, and $z(k) = \lambda_1 z(k-1) + \sum_{l=1}^{v} \lambda_2 u_l(k) y^*_{-l,k}$ is the approximation of the cross-correlation vector between the training input $\left\{x_l^{(cp)}\right\}$ and the output $\left\{y_l^{(cp)}\right\}$. The data vector $u_l(k)$ is formed as $u_l(k) = \left[x_{-l,k} \cdots x_{-1,k} x_{-v,k-1} \cdots x_{-v+l,k-1}\right]^T$, where, $l = v, v-1, \cdots, 1$. λ_1 and λ_2 are forgetting factors[1] for the data between blocks and within the same block respectively.

[1] In least square method, forgetting can be viewed as: giving less weight to older data and more weight to recent data. Therefore, the significance of forgetting is that, older data is gradually discarded in favor of more recent information.

3.5.2 Dynamic Channel Estimation and Optimization Algorithm

(To gain a visual understanding of the algorithm's framework, please refer to Figure 3.7)

Known parameters: $M_{i,k}$ and $\Delta_{i,k}$

Selecting parameters: λ_1 and λ_2

Initialization: $k = 0$, an initial training is used to initialize $\hat{h}(0)$ and $\overline{\Phi}(0)$

Computation for $k = 1, 2, 3, 4 \cdots$

1) $H_{i,k} = \left(\frac{1}{\sqrt{m}}\right)\sum_{l=0}^{v-1} \hat{h}_{l,k} e^{\frac{-j2\pi il}{m}}$

$$w_{i,k} = \frac{M_{i,k}^{\ 1/2} H_{i,k}^{\ *}}{M_{i,k}\left\|H_{i,k}\right\|^2 + \Delta_{i,k}} \qquad i = 0, \cdots\cdots, m-1$$

2) $\hat{X}_{i,k} = Y_{i,k} w_{i,k}(k-1), \quad i = 0, \cdots\cdots m-1$

3) $\overline{x}_{i,k} = \left(\frac{1}{\sqrt{m}}\right)\sum_{l=0}^{m-1} q(\hat{X}_{l,k}) e^{j\left(\frac{2\pi}{il}\right)m}, \quad i = m-v, \cdots\cdots m-1$

4) $\overline{\Phi}(k) = \lambda_1 \overline{\Phi}(k-1) + \sum_{l=1}^{v} \lambda_2 \overline{u}_l(k) \overline{u}_l^{\ H}(k)$

$$\overline{z}(k) = \lambda_1 \overline{z}(k-1) + \sum_{l=1}^{v} \lambda_2 \overline{u}_l(k) y^*_{-l,k}$$

where,

$$\overline{u}_l(k) = \left[\overline{x}_{-l,k} \cdots \overline{x}_{-1,k} \overline{x}_{-v,k-1} \cdots \overline{x}_{-v+l,k-1}\right]^T,$$

$$\overline{x}_{-l,k} = \overline{x}_{m-l,k} \ , l = v, v-1, \cdots\cdots, 1$$

5) $\hat{h}(k) = \overline{\Phi}^{-1}(k) \overline{z}(k)$

Note: if λ_2 = 1, then all the data are weighted equally, and the algorithm has infinite memory length, which is optimal with respect to suppressing the estimation noise effect. On the other hand, if λ_1 = 0.5, with smaller lambda value, the algorithm has shorter memory length, and is better adapted to channel dynamics. A smaller value of lambda will reduce the equalization error due to lag effects. Therefore, the optimal lambda value depends on the channel fading dynamics and the extent of input noise effect on the equalization error.

3.6 Simulation Results

Simulations are carried out in OFDMA Digital Transmission System (ODTS) environment, which is a software system that we have designed using C and C++ language. The simulation parameters are shown in Table 3.1. We have used an OFDM system with 256 sub-carriers, 16-QAM constellation. The transmit power of all the used sub-carriers is equal, with the same preset error probability of 10^{-3}. The performance is evaluated by the averaged mean square error (MSE) per sub-carrier, which is defined as the $err = \frac{\sum_{i \in U} \left\| X_i - \hat{X}_i \right\|^2}{|U|}$, where $|U|$ is the number of all the used sub-carriers. We have selected forgetting factors based on the dynamics of the channel variations using trial and error method. $\lambda_2 = 1$, $\lambda_1 = 0.75$ for medium channel variations and $\lambda_2 = 1$, $\lambda_1 = 0.5$ for major channel variations.

Table 3.1 Simulation Parameters

Parameter	Value Specification
Carrier Frequency f_c	3.3GHz
Bandwidth	1.75MHz
Number of sub-carriers (Nfft)	256
Tb (seconds, Useful symbol time)	128 μsec
G (ratio of CP to Useful time)	1/4
Tcp (seconds, CP length)	32 μsec
Ts (OFDM symbol time secs/sym) Ts=Tb+Tcp	160 μsec
Symbol Rate (Rs = 1 / Ts)	6250
Delay Spread length	0 ~ 16 μsec
Channel Model	Time varying model
Signal constellation	16-QAM

Simulations are performed in a time varying environment, where we let the physical characteristics of the channel change. We have chosen to design and model three different types of channel (H_0, H_1, H_2), which are categorized as constant or minor, medium variations and major channel variation respectively. We let the channel change from one type to another and then we observe the performance of our adaptive channel estimation, and then compare it against the performance and efficiency of several conventional techniques.

The wireless channels H_0, H_1 and H_2 modeled represent different wireless environment in which the system can operate. The parameter values of the channel models is as follows:

- H_0 is modeled as a stationary wireless environment with minimal movement of surrounding objects, this represents a physical condition where the mobile user is stationary in a room. Here, the delay spread is 1 $n\,sec$, which is very small and can be negligible. There are no movements around at all and no channel variations, and there is no Doppler shift.

- The second type of wireless channel H_1, is modeled as a slow moving of the transmitter and receiver and or objects surrounding it, the motion speed is approximately around 5km/h. This represents an environment where a mobile user is walking in an urban area. The Doppler frequency is calculated as 15Hz using the formula $f_D = f_c v / c$ and simulation parameters described in Table 4.1. The normalized Doppler frequency[1], $f_d = f_D / R_s$ = 15/ 6250 = 1/416, in other words the channel changes in about every 416 symbols. The multi-path delay spread for H_1 is set to 5 μsec

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[1] To get the normalized Doppler frequency take the Doppler frequency shift in Hz , divide by symbol rate in Hz and you have dimensionless normalized Doppler frequency [18]

- The third type of wireless channel is modeled as fast moving and changing channel. The speed of the transmitter or receiver is about 100km/hr. This will represent an environment where a mobile user is driving in an urban area. The delay spread is 10 μsec. The Doppler frequency is calculated as $f_D = 300$ Hz. The normalized Doppler frequency $f_d = f_D / R_s$ = 300/ 6250 = 1/21, in other words the channel changes in about every 21 symbols.

At the beginning of the simulation the channel is H_0. The bit and power allocation are done according to H_0 and remains unchanged during the simulation. After some time the channel changes to H_1 , and then to H_2. In our algorithm, the first block is sent as a pure training sequence to get information about H_0.

The physical significance, or the practical scenario and application would be for example, the mobile broadband user is downloading large files of music or movies to a broadband portable device. The download is started at home or office (H_0), and then the user wants to travel to downtown along with the portable device. The user walks to the station (H_1), and then takes a bus or a train, (H_2), that will travel to downtown. Another application would be for instance, the user is using voice services provided by a Wireless VoIP (Voice over Internet Protocol) provider with a broadband capable mobile phone on a WiMAX network.

Then the real data is sent and the adaptive channel estimation algorithm is used to track the variation of the channel. Figure 3.8 and 3.9 show the results of the simulation where we compare the performance of retraining and initial training algorithms to our proposed algorithm. The results of using retraining algorithms in [33] and without retraining [32] are also plotted for comparison. It should be noticed that additional training sequences are used for both initial training and retraining algorithms to track the channel variations properly. While the scheme in [31] performs badly and is not able to keep up with channel variations, because it cannot estimate the channel dynamically.

Our proposed scheme performs very close to the retraining algorithm but without any loss in bandwidth. Since our scheme is estimating the channel continuously and recursively we are able to keep up the performance throughout the simulations when the channel characteristics changes. Figure 3.8 shows the results when the channel changes from H_0 to H_1, while Figure 3.9 shows the results when the channel changes from H_1 to H_2.

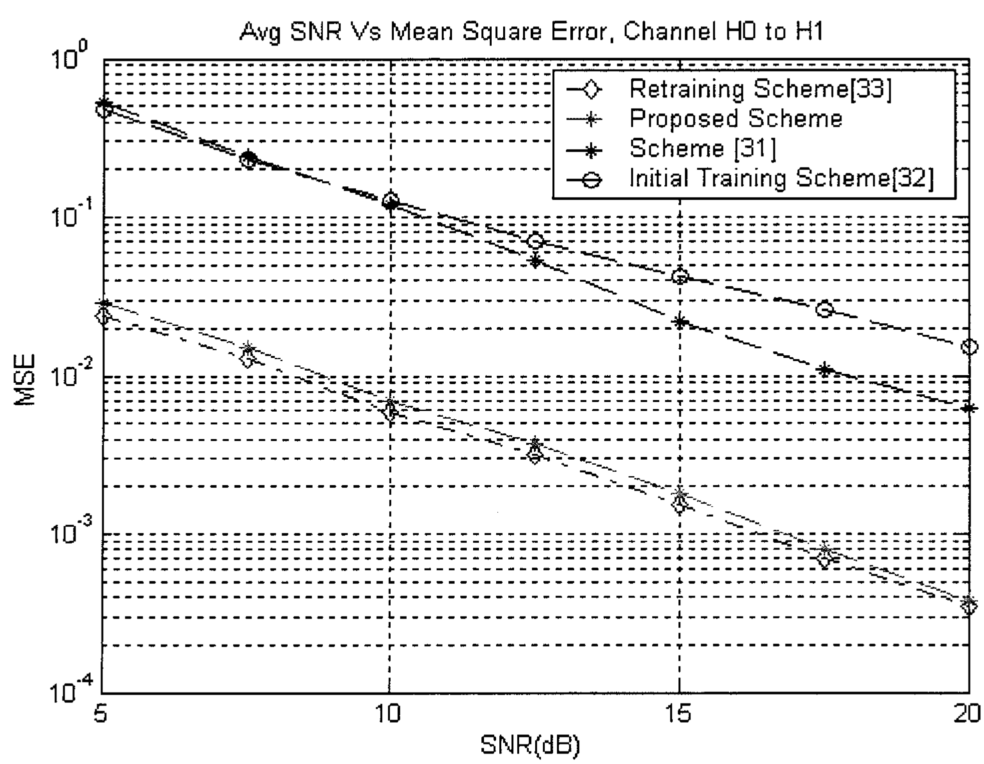

Figure 3.8 Performance when channel changes from H_0 to H_1

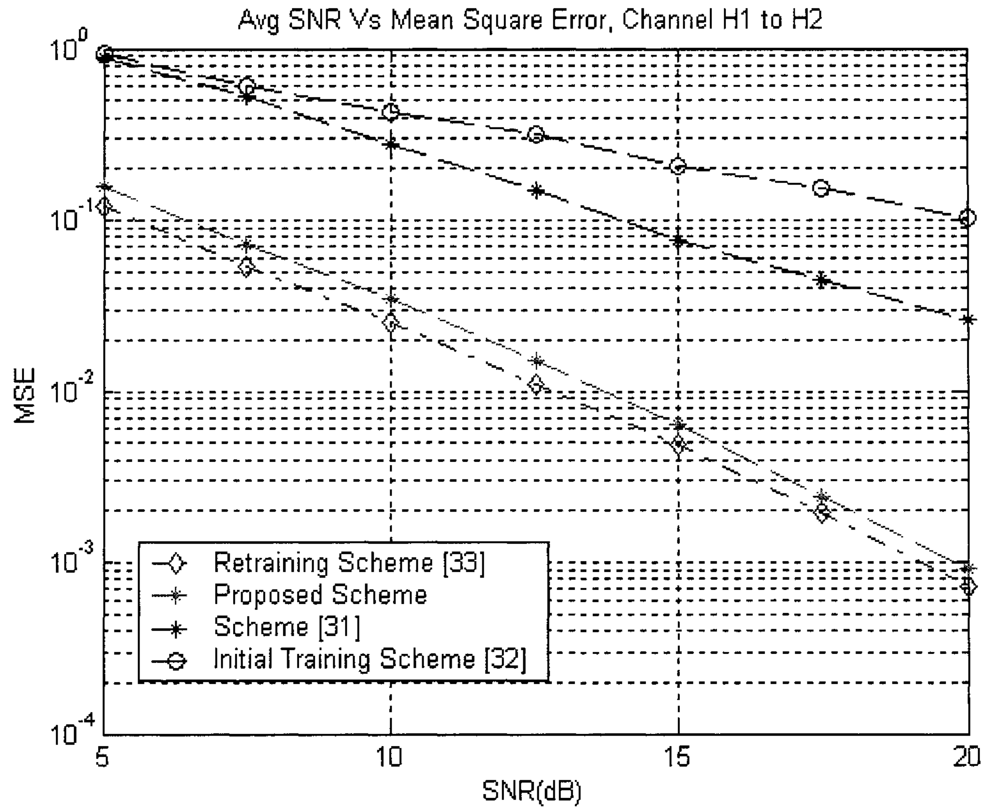

Figure 3.9 Performance when channel changes from H_1 to H_2

We can see that our algorithm can adaptively track these variations and perform well. It performs much better than the system just using initial training and is close to the system using retraining when the channel goes through major changes. Here, the advantage of our algorithm is that we are not wasting precious bandwidth in retraining every time, but we use the cyclic redundancy to track the channel without any additional training. Hence achieve better efficiency. The bandwidth efficiency can be calculated in terms of ratio of the number of useful information bits to the total number of information bits sent. Table 3.2 shows the efficiency of the proposed method in comparison to other conventional training methods. Although we have same efficiency as the algorithm in

[31], the distinctive aspect of our algorithm is that it performs better than [31] in terms of error rates. Using our algorithm we can adaptively track the channel, and estimate the channel dynamically when the channel characteristics changes.

Table 3.2 Efficiency in terms of useful information throughput (goodput)

Channel variations	**Efficiency** $\eta = \frac{\text{Total bits - Redundant bits}}{\text{Total bits}} \times 100$	
Minor (H_0)	Re-training Scheme [33]	60.7%
	Proposed Scheme	74.9%
	Scheme [31]	74.9%
	Initial Training Scheme [32]	70.1%
Medium $(H_0 \rightarrow H_1)$	Retraining Scheme [33]	37.6%
	Proposed Scheme	74.9%
	Scheme [31]	74.9%
	Initial Training Scheme [32]	70.1%
Major $(H_1 \rightarrow H_2)$	Retraining Scheme [33]	30.2 %
	Proposed Scheme	74.9%
	Scheme [31]	74.9%
	Initial Training Scheme [32]	70.1%

3.6.1 Symbol Error Rate Performance

The SER analysis is performed empirically using the parameters from IEEE 806.16 standard as shown in Table 3.1. We can see from the simulations results, in Figure 3.10, that in a time varying environment we are able to get good performance results that are very close to retraining algorithm for estimating the channel state information.

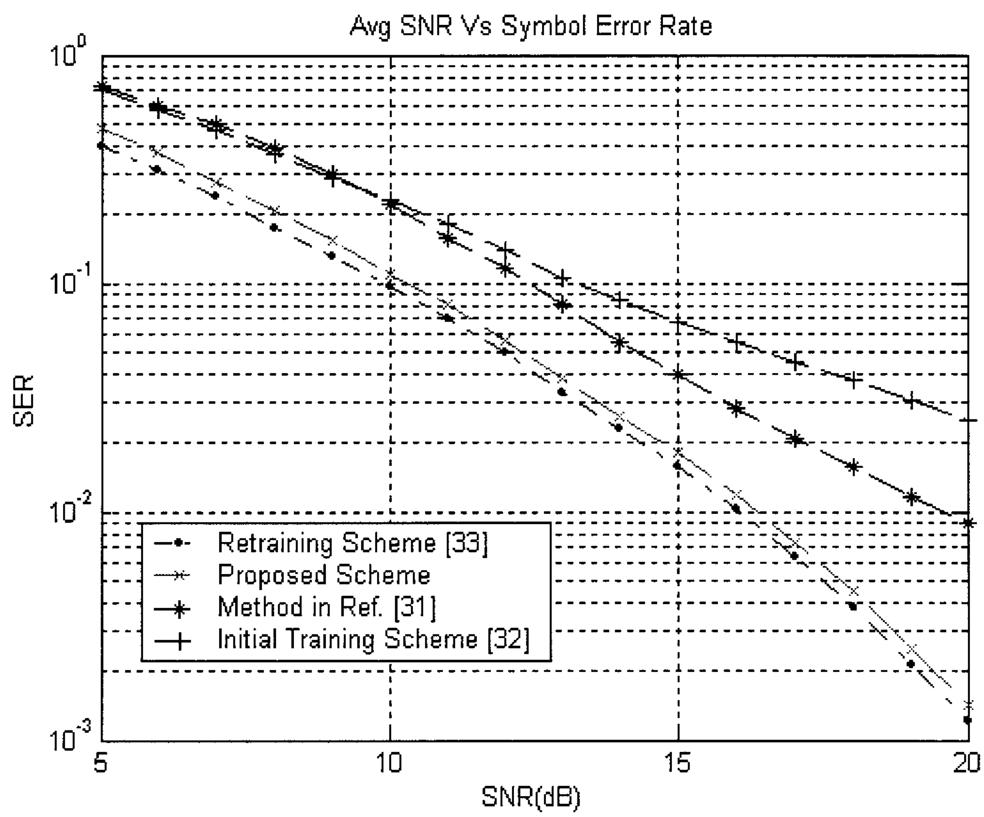

Figure 3.10 Symbol Error Rate Vs Signal to Noise Ratio

Therefore, by using the proposed adaptive channel estimation scheme the channel can be estimated dynamically without any use of pilots or retraining, thus saving a lot of precious bandwidth. Our method serves as a platform for future work in estimating the channel dynamically for the OFDMA system in time-varying channels without any loss in the available bandwidth or a tradeoff in the limited resources available.

3.7 Summary

Channel estimation is an important part of any communication system. A good channel estimation scheme should provide reliable Channel State Information (CSI), enabling us to effectively compensate for the multiplicative effects of the channel. The channel estimation in a wireless environment can serve two important purposes. First, the receivers need the estimated channel information to recover what information has been transmitted. Secondly, the CSI could be used to feed back some information to the transmitter in order to make the transmitter adapt to the changes in the channel and change the transmission performance. System performance depends on the channel estimation method and the implementation of a reliable and efficient method is crucial.

The method that we have presented is better than conventional channel estimation techniques because we have performed the channel estimation in a realistic time varying environment and have applied an adaptive signal processing technique to track the channel variations. In a time varying environment we are able to obtain good performance results that are very close to retraining algorithms for estimating the channel state information.

Therefore, using our adaptive channel estimation scheme the channel can be estimated dynamically without any need for pilots or retraining, thus saving a lot of precious bandwidth. Our method serves as a platform for future work in estimating and tracking the channel dynamically for the OFDMA systems in time varying channels without any loss or a tradeoff in the limited resources available.

Chapter 4

An Enhanced Frequency Synchronization Scheme

OFDM synchronization is an important and challenging issue. For instance, In a multi-user OFDM milieu, uplink synchronization is a very intricate task. The correction of a user's time and frequency offsets cannot be accomplished easily at the base station receiver, as the correction to one user would misalign other initially aligned users. In OFDMA systems, proper synchronization is necessary to keep orthogonality of the users, which is essential for a reliable transmission. Furthermore, synchronization is crucial because synchronization problems can also lead to grave errors in channel estimation. This chapter describes a novel scheme for proper synchronization in the uplink of multi-user OFDM systems.

We present an enhanced and novel synchronization algorithm that is demonstrated as an enhanced scheme for tracking in the uplink of multi-user OFDM systems in a multi-path environment. We have followed a simple approach, where we have the base station estimate time and frequency offsets, and then transmit the synchronization information to the mobile. The mobile can then adjust its transmitted signal so that it is in alignment with the other users' signals. This method of having the base station feedback synchronization information is also used in Time Division Multiple Access (TDMA) systems, such as the Global System for Mobile communication (GSM) [39].

4.1 OFDMA System synchronization problems

It is well known that OFDM is very sensitive to frequency and time offsets [1], which destroy orthogonality among sub-carriers and thereby cause Inter-Carrier Interference (ICI) and Inter-Symbol Interference (ISI) respectively. The frequency offset that is caused by the Doppler spread of the channels or the instabilities of oscillators produces severe side effects.

The side effects of frequency offset can be observed as the reduction of the signal amplitude in the output of the filters matched to each of the carriers. Furthermore, the introduction of ICI from other carriers, which are no longer orthogonal to the filter because of the Carrier Frequency Offsets (CFOs) between the transmitter and the receiver, destroys the orthogonality and results in multiple-access interference. The effect of frequency offset on the data transmission is described in [40]. Figure 4.1 shows the effects of frequency offset as a reduction in signal amplitude and ICI.

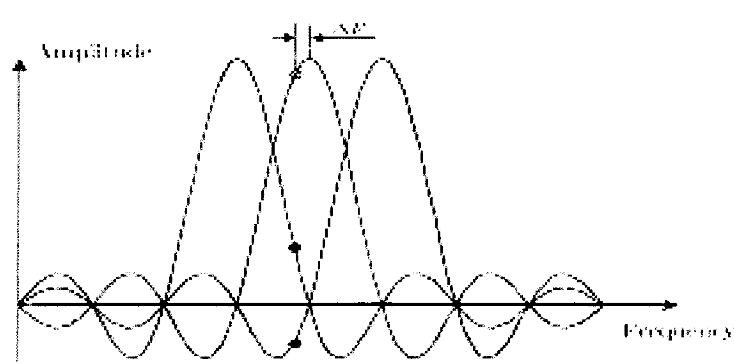

Figure 4.1 Effects of frequency offset ΔF: reduction in signal amplitude denoted by (o) and ICI denoted by (•) [40]

The side effects of time offset can be observed as different users' transmitted signals that are not time-aligned, and appear as ISI at the FFT demodulator outputs. Thus, one user's misalignment in time affects all the other users' data output, which leads to incorrect detections. Figure 4.2 shows the effects of time offset, as one user's skewed demodulated symbol could interfere with another user's data.

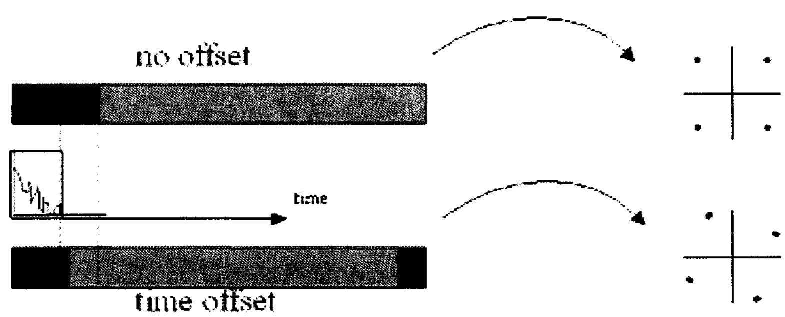

Figure 4.2 No time offset correct symbols. Time offset skewed symbols

J. Beek [13] noticed these problems and proposed a method to have all the users aligned in time and frequency especially in the uplink access. Beek's algorithm shows maximum likelihood optimality, but exhibits an estimation error floor in multi-path frequency-selective channels. His algorithm processes all the estimation in time domain at the receiver. The algorithm extracts time and frequency offset information from the cyclic prefix. The performance of the timing offset estimation is optimum even in the severely frequency-selective channel, if the length of cyclic prefix is guaranteed to be longer that the delay spread of the channel. No other algorithm has been known to outperform this estimator. However, for frequency offset estimation, it can be proved analytically and empirically through simulations that the algorithm works quite unsatisfactorily in certain multi-path environment.

Therefore, Beek's frequency offset estimation exhibits an estimation error floor in multi-path frequency-selective channels. Considering the channel situations in which Broadband Wireless Access is employed, frequency-selectivity in the channel can always be assumed because of the multi-path effects and non-line of sight operations of wireless terminals. Thus, it is important to reduce the channel dependency in carrier frequency synchronization, which is important for securing orthogonality and preventing ICI effects and in order to have a reliable and enhanced system performance.

In this thesis, we first investigate and evaluate Beek's algorithms, and then suggest a novel synchronization scheme that can be employed independently of the frequency-selectivity in the transmission channel for the uplink OFDMA systems.

4.2 Uplink multi-user synchronization

In multi-user OFDM (a.k.a OFDMA) systems, especially in the uplink of such systems, users must be aligned in time and frequency to maintain the orthogonality of the sub-carriers. If different users' transmitted signals are not time-aligned, ISI appears at the FFT demodulator outputs. Thus, one user's misalignment in time affects all the other users' data output. Meanwhile, a carrier frequency offset destroys orthogonality among sub-carriers resulting in ICI.

In the uplink, time and frequency offsets estimations are performed at the base station, and information is fed through a control channel to each user's transmitter for clock and oscillator adjustments. Namely, all users adapt to the base station's receiver clock and oscillator by adjusting their oscillator and scheduling their transmission according to the base station's information that is fed back to them via a control channel.

J. Beek introduced a Maximum Likelihood (ML) time and frequency estimator for multi-user OFDM in [13]. He used the statistical redundancy in the received signal, the cyclic prefix, and proposed an estimator, which works without the aid of pilot symbols and is independent of the modulation scheme in each sub-carrier.

Consider an OFDM symbol received by the base station, when the N sub-carriers constituting this symbol are subdivided into M bands of sub-carriers, the indexes of which are collected in the set Z_m [13]. The transmitted OFDM symbol in the m^{th} band of sub-carriers is:

$$s_m(t) = \sum_{n \in Z_m} x_n e^{\frac{j2\pi nt}{NT}} \qquad -T_g < t < NT \tag{4.1}$$

where, NT is the duration of the OFDM symbol without the cyclic prefix, and T_g is the length of the cyclic prefix [13].

- o Assumption: Note that in the Equation (4.1) above, it is assumed that the channel is slow fading and the impulse response does not change during a single time-slot.
- Justification: This assumption can be justified due to the underlying principle of OFDM where the entire transmission bandwidth is divided into N orthogonal narrow slow-fading sub-carriers [13], [40]. Furthermore, please refer to Chapter 3 Section 3.4, where we have explained the underlying principles and the channel model.

Realistically, the m^{th} transmitted signal will have a time offset relative to the receiver symbol clock and also a frequency offset relative to the receiver demodulation frequency. These offsets can have very adverse effects, as explained before. If the time offset and the frequency offset within the m^{th} transmitted signal are denoted by θ_m and ε_m, respectively, the sampled received OFDM signal is given by [13]:

$$r(k) = \sum_{m=0}^{M-1} r_m(k) = \sum_{m=0}^{M-1} s_m(k-\theta_m) e^{\frac{j2\pi\varepsilon_m k}{N}} + n_m(k) \tag{4.2}$$

Then, assuming perfect separation of each sub-band signal with the band-pass filter bank at the receiver, Beek's [13] estimator can be formalized as in (4.3) and (4.4), where L is the length of the equivalent cyclic prefix duration. His algorithm processes the estimation in time domain without involving in FFT manipulation at the receiver, and extracts time and frequency offset information using only the cyclic prefix. Therefore, it is a very fast algorithm for estimating offsets. Specifically, the performance of the timing offset estimation is optimal even in a severely frequency-selective channel, if the length of the cyclic prefix is guaranteed to be longer than the delay spread of the channel. No other algorithm has been known to outperform this estimator.

However, Beek's frequency offset estimation works unsatisfactorily in multi-path frequency selective channel, this will lead us to a definite channel dependency and poor performance in certain multi-path environment. We can illustrate this fact by theory and simulation work. Beek's [13] time and frequency offset estimation is given as:

$$\hat{\theta}_m = \arg \underset{\theta}{Max} \left\{ |\gamma_m(\theta)| - \phi_m(\theta) \right\} \tag{4.3}$$

$$\hat{\varepsilon}_m = -angle \frac{\{\gamma_m(\theta_m)\}}{2\pi} \tag{4.4}$$

where,

$$\gamma_m(\theta) = \sum_{k=\theta}^{\theta+L-1} r_m^*(k) r_m(k+N) \quad \text{and,}$$

$$\phi_m(\theta) = \frac{1}{2} \sum_{k=\theta}^{\theta+L-1} \left|r_m(k)\right|^2 + \left|r_m(k+N)\right|^2$$

Here, in Beek's frequency estimation algorithm we observe that the frequency-offset estimate is solely dependent on $\gamma_m(\theta)$, which is summed over an entire cyclic prefix interval. Any interference from the precedent symbol caused by channel delay spread in a multi-path environment deteriorates the accuracy of the estimate. From this point of view, Beek's frequency offset estimator can be said to be deficient and have performance constraints, which is shown in Figures 4.5 & 4.6 in the simulation results section of this chapter. Therefore, a novel and enhanced frequency offset estimation is proposed that will enhance the accuracy of estimation and produce better results in a frequency selective multi-path environment.

4.3 Enhanced Uplink Frequency Synchronization Algorithm for OFDMA Systems in a Multipath Fading Environment

Figure 4.3 shows the OFDMA uplink frequency synchronization system model.

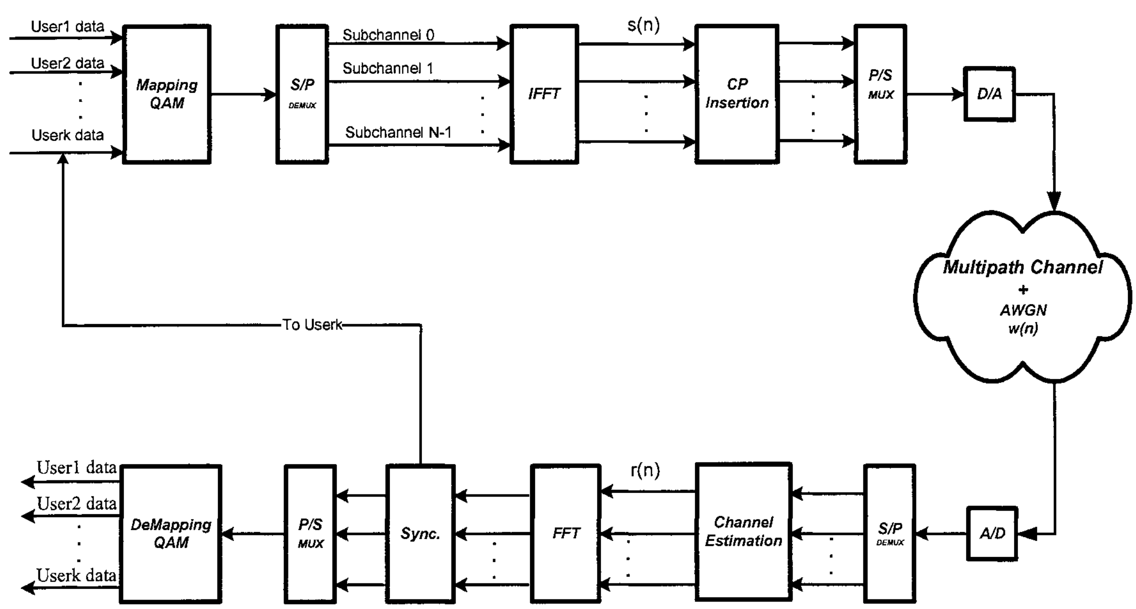

Figure 4.3 Uplink frequency synchronization system model for Multi-user OFDM

To overcome the limitations inherent in Beek's frequency offset estimator, a new frequency offset estimation scheme for multi-user OFDM is proposed in this chapter. The main idea for the proposed algorithm is processing OFDM symbols in frequency domain to estimate the frequency offset. This frequency-domain approach often employed for blind OFDM synchronizations such as in [41], [42], processes FFT outputs after the cyclic prefix removal, and thus is not affected by the delay spread of the channel that can introduce synchronization errors. This is the main advantage of our method over Beek's time-domain approach. In addition, the distinctive aspect of our algorithm is that, it does not require any over sampling and is less complex.

Our algorithm expands the single-user synchronization technique of [42] to multi-user scenario and to a new aspect. The frequency offset correction scheme in [42] basically uses two sample sets obtained by over-sampling an OFDM signal, and correlation of FFT outputs of each sample set. In single-user OFDM systems, over-sampling is not acceptable because it leads to high complexity.

Let us consider the received OFDM signal in Equations (4.2). The signal in Equation (4.2) can be sampled with sampling frequency N/T and divided into even and odd numbered sample sets. Each element of these sets is given by:

$$r_{m\_e}(k)=\sum_{m=0}^{2l} s_m(k-\theta_m)e^{\frac{j2\pi k\varepsilon_m}{N}} \tag{4.5}$$

$$r_{m\_o}(k)=\sum_{m=0}^{2l+1} s_m(k-\theta_m)e^{\frac{j2\pi k\varepsilon_m}{N}} \tag{4.6}$$

where, l takes integer values, $l = 0 \cdots M-1$

The frequency offset correction algorithm in [42] can be simplified and directly applied to each sub-band without over-sampling. Only some sample re-ordering is required to properly estimate the frequency offset in a sub-band. The proposed algorithm for the estimator is shown in Equation (4.8), where z_i^e and z_i^o are the i^{th} FFT outputs of even and odd numbered samples of $\{r_m(k)\}$ for m^{th} sub-band, respectively. This simplified and enhanced algorithm is based on the fact that, without frequency offset, data symbols can be extracted from both even-numbered samples and odd-numbered samples, which are well established by the Fourier Transform properties.

After demultiplexing, we obtain the outputs of the even and odd numbered sample sets as:

$$z_i^e(\theta) = FFT\left\{r_{m\_e}(k)\right\} = FFT\{\sum_{m=0}^{2l} s_m(k-\theta_m)e^{\frac{j2\pi k\varepsilon_m}{N}}\} = \sum_{m=0}^{2l} s_m(k-\theta_m) \tag{4.6}$$

and,

$$\begin{aligned} z_i^o(\theta) = FFT\left\{r_{m\_o}(k)\right\} &= FFT\{\sum_{m=0}^{2l+1} s_m(k-\theta_m)e^{\frac{j2\pi k\varepsilon_m}{N}}\} \\ &= \sum_{m=0}^{2l+1} s_m(k-\theta_m) \end{aligned} \tag{4.7}$$

where, l can take integer values, $l = 0 \ \cdots \ M-1$

As a result, the Maximum Likelihood estimation for the frequency offset estimation can be obtained as :

$$\hat{\varepsilon}_m = \arg \underset{\theta}{Max}\left\{\left| z_i^e(\theta)\cdot(z_i^o(\theta)e^{-\frac{j2\pi i}{N}})^* \right|\right\} \tag{4.8}$$

One remarkable characteristic for this estimator is that the performance is independent of the channel delay spread, with only additive noise determining its limit. In simulations, such characteristic is verified, and performance of the proposed estimator is evaluated.

4.4 Simulation Results

Through simulations, performance of each estimator is evaluated under 2-ray channel models. The simulation parameters are shown in Table 4.1. The number of sub-carriers is set to be 256, and in 4 user case (64 sub-carriers per user) or in 8 user case (32 sub-carriers per user). The delay spread (L) is a variable and will have $0 \sim 16\,\mu sec$, and the length of cyclic prefix is set to be $32\,\mu sec$, so that no ISI exists when perfect synchronization is achieved. Another parameter used in the simulations is R *(i.e., the power ratio of main path to delayed path)*. R ranges from 0 to 20dB to model a severely frequency-selective channel *(e.g., R = 0dB is the strongest delayed path model)*.

Table 4.1 Simulation Parameters

Parameter	Specification
Carrier Frequency f_c	3.3GHz
Bandwidth	1.75MHz
Number of subcarrriers	256
Number of users	4
Power ratio of main path to delayed path	$0 \sim 20$dB
Delay Spread length	$0 \sim 16\,\mu sec$
Length of CP	$32\,\mu sec$
Channel Model	Multipath two ray model
Signal constellation	16-QAM

The wireless channel is modeled as a two ray multi-path Rayliegh frequency selective channel. Frequency selective fading because the bandwidth of the signal is greater than the coherence bandwidth; When the delay spread is $16\,\mu sec$, we get a coherence bandwidth of $B_c \approx 0.2/\tau_m$= 12.5KHz (for a correlation of 0.5, [18]), which will be smaller than the signal bandwidth i.e., about 68KHz. The performance is determined by the Mean Square Error (MSE) of the estimators. According to [13], MSE should be less than 10^{-3}, for the receiver to experience no SNR loss due to frequency offset.

Figures 4.4 and 4.5, show the performance of Beek's algorithm in the 2-ray multi-path channel, where we can see the performance limit of the estimator. The longer the delay spread and the higher delayed-path power, the MSE shows upper limits in the estimation. For example, even with SNR of 20dB, the MSE is greater than the target of 10^{-3} , when R = 0dB and L =16 μsec, this means that the estimator cannot overcome the strong channel impairments.

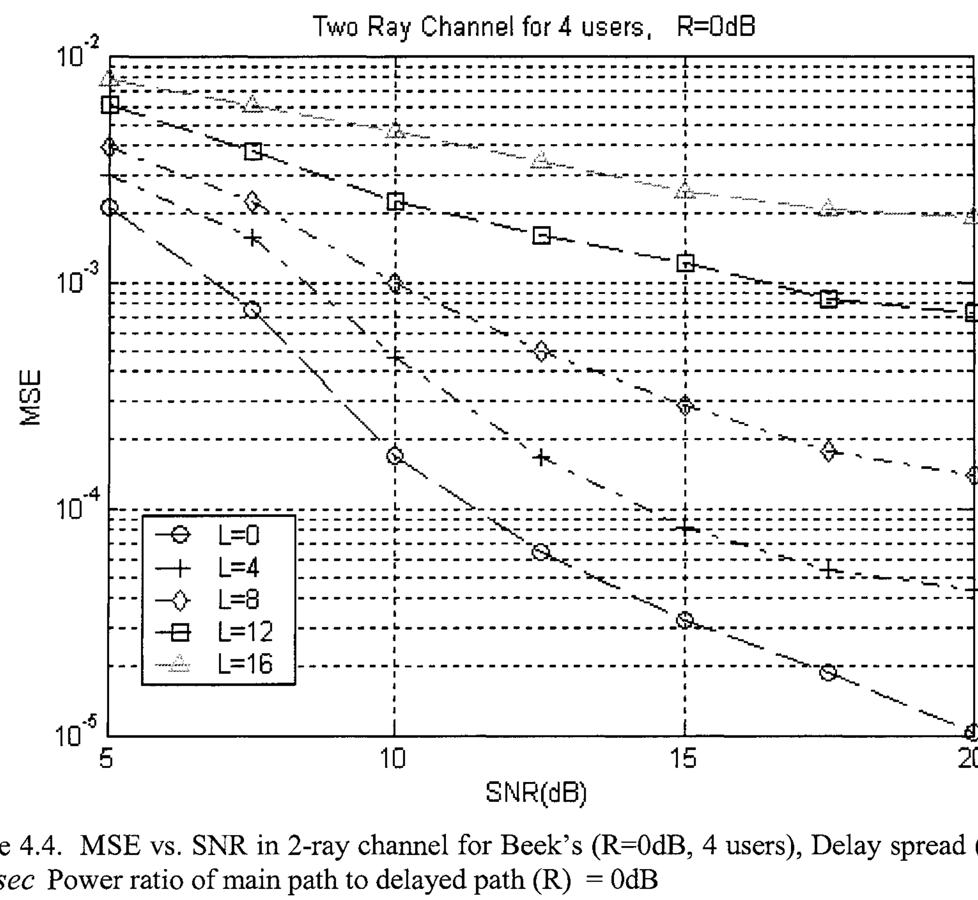

Figure 4.4. MSE vs. SNR in 2-ray channel for Beek's (R=0dB, 4 users), Delay spread (L) = 0 ~ 16 μsec Power ratio of main path to delayed path (R) = 0dB

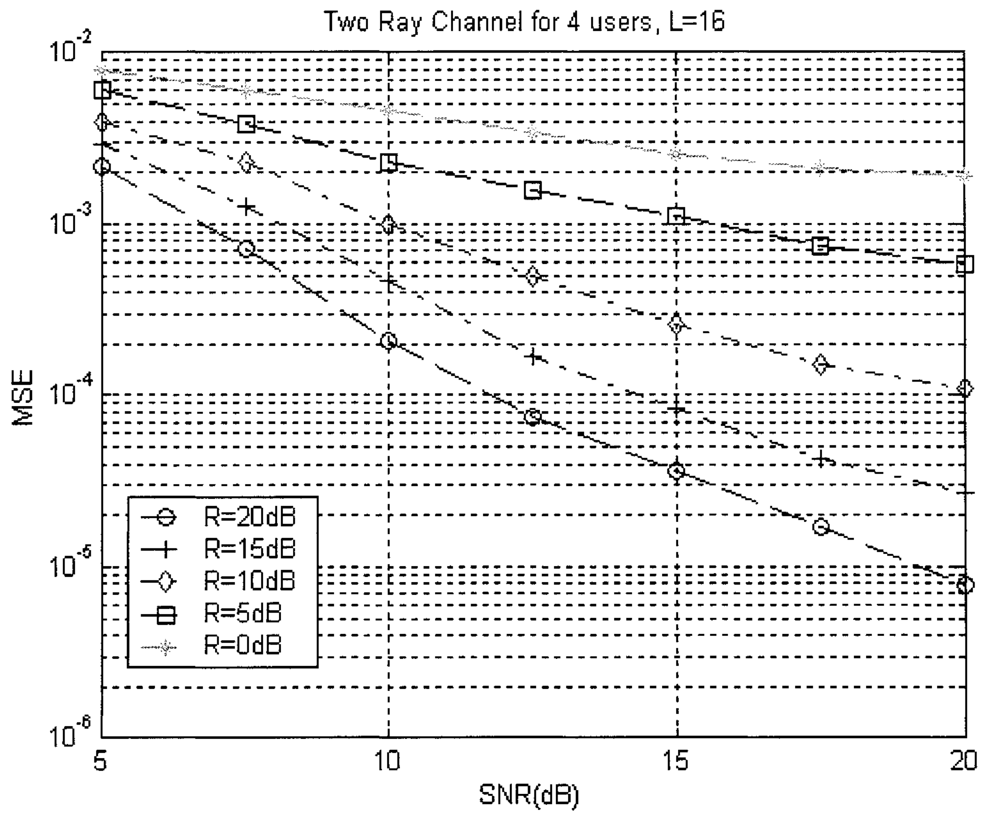

Figure 4.5. MSE vs. SNR in 2-ray channel for Beek's algorithm, 4 users, Delay spread (L) = 16 $\mu\,sec$, Power ratio of main path to delayed path (R) $= 0 \sim 20$dB

On the other hand, as is shown in Figure 4.6, our proposed estimator exhibits immunity to channel impairments in the estimation. Its performance is only dependent on SNR, which again shows the intrinsic OFDM characteristics of robustness to a multi-path fading environment.

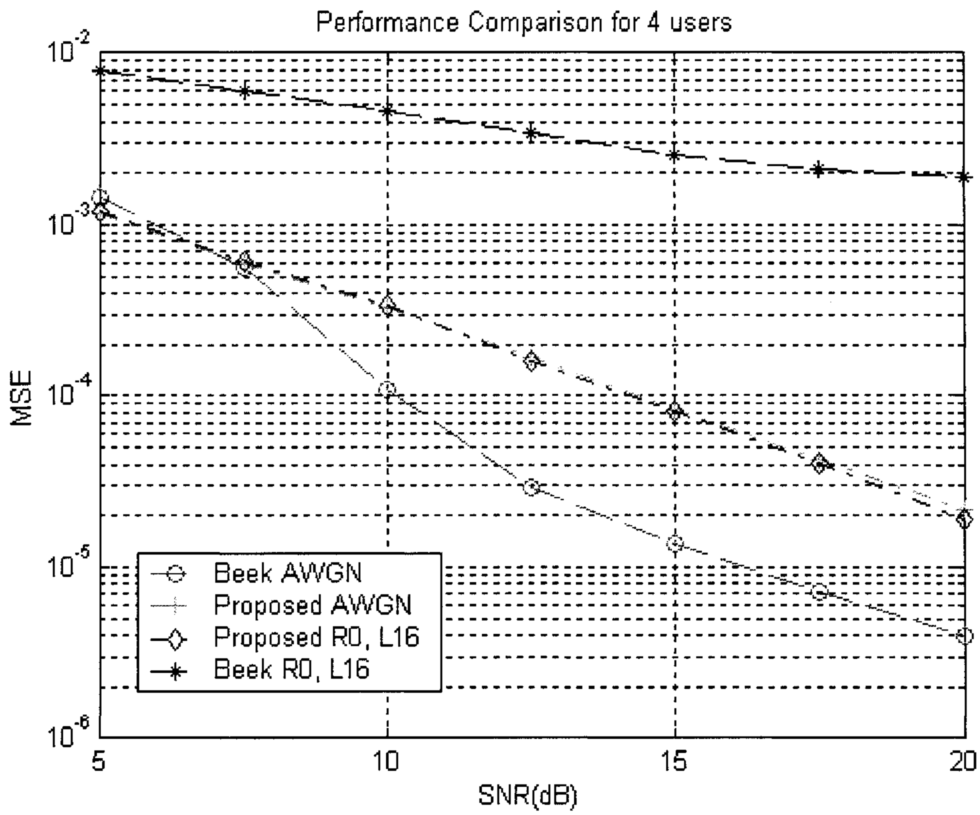

Figure 4.6. Performance Comparison, Number of Users = 4 users, Delay spread (L) = 16 μsec, Power ratio of main path to delayed path (R) = 0 dB

4.5 Conclusion

In this chapter, we have investigated a synchronization scheme for multi-user OFDM systems. Unlike the single-user case, the base station estimates frequency offset of each user's transmitter. The estimation algorithm is also applied to each subband and compensate the offset. The performance bound of the ML synchronizer of Beek [13] is evaluated under a 2-ray channel model with the delay and power varied in the scenario. From the simulations, we can observe that the ML synchronizer, using the cyclic prefix in OFDM, has some drawbacks in estimating frequency error due to delay spread in the access channel in a multi-path fading environment. The power ratio of the main path to the delayed paths is considered a major parameter in evaluating the performance. We have found that Beek's algorithm shows maximum likelihood optimality under AWGN environments, but exhibits an estimation error floor in multi-path frequency-selective channels.

We proposed an enhanced synchronization algorithm to alleviate channel impairments in estimation and verified its performance under the same channel conditions. We have simplified and enhanced uplink synchronization scheme for OFDMA in order to create a system that is robust to channel impairments. We verified its performance under different channel conditions. As expected, the ML synchronizer will show better performance in AWGN, since it is close to the theoretic Cramer-Rao bound.

However, the proposed algorithm outperforms the ML scheme, in the frequency-selective channel that is more practical in an OFDMA multi-path channel environment [14]. We found a synchronization technique for frequency offset correction in OFDM system that is an improvement over the time domain approach. Ours is a frequency domain approach, that it avoids the delay spread and related problems. It is better than other frequency domain synchronization methodologies such as the blind methods that normally require more computational complexity.

Chapter 5

Dynamic Resource Allocation

Multi-user OFDM or Orthogonal Frequency Division Multiple Access (OFDMA) is considered to be an advanced modulation and multiple access method for 4G, (4^{th} Generation Wireless Networks) [7]. OFDMA is a multiple access technique where each user is assigned with a fraction of the total available number of sub-carriers. It is equivalent to Frequency Division Multiple Access (FDMA) scheme, with the major advantage that it does not does not require large band guards to separate different users therefore, it does not waste bandwidth. Allocating resources properly to different users is important for an efficient OFDMA system. This chapter will explore OFDMA in more detail, emphasizing on the design of a realistic and practical dynamic resource allocation scheme.

In this thesis, we have proposed solutions to the problems encountered in OFDM based systems, in terms of channel knowledge and synchronization. We will apply the channel estimation and synchronization techniques proposed in this thesis and evaluate the OFDMA dynamic resource allocation system's performance and compare it with other resource allocation schemes.

5.1 OFDMA technique

In the OFDM technology, since the sub-carriers are independent of each other, a multiple access system such as OFDMA can be designed. The available spectrum is subdivided in bands of adjacent sub-carriers, and within each band a TDMA scheme is applied. Users are assigned transmission blocks consisting of adjacent sub-carriers allocated to them in different time-slots. Hence, in an OFDMA scheme, users are separated in both frequency and time. Figure 5.1 shows time-frequency allocation for users in the Universal Mobile Telecommunications System (UMTS) scenario.

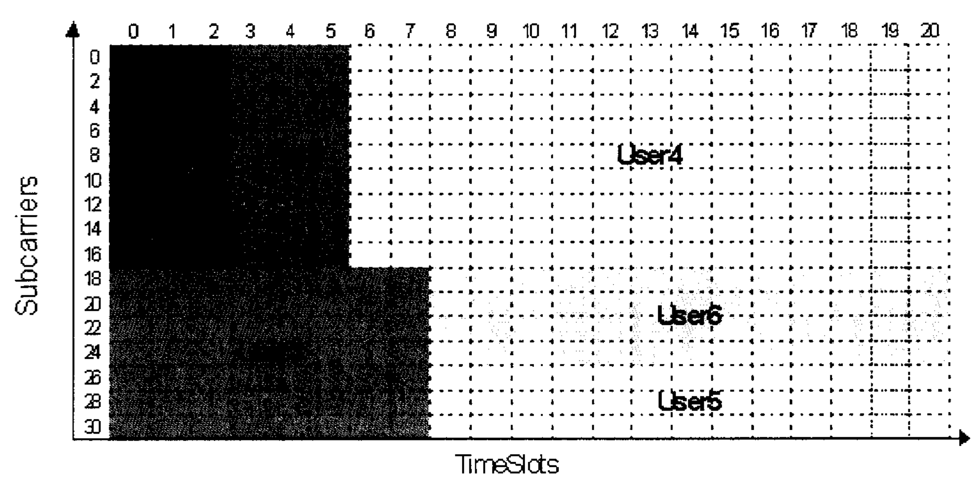

Figure 5.1 OFDMA method, users are separated in time & frequency

5.1.1 Advantages of OFDMA scheme

OFDMA technology enables simultaneous bi-directional flow of data for numerous subscribers at a high speed. This is why it has been chosen as a preferred multiple access scheme for BWA. The OFDMA technology allows the user to utilize bandwidth in an optimal manner using the given frequencies without collision between channels. This overcomes the interference and enables maximal bandwidth on demand by using logical sub-carriers that support scalability, multiple access and an advanced array of processing capabilities. The following are some of the advantages of the OFDMA system:

- OFDMA is a very flexible multiple access scheme; OFDMA is flexible in terms of the required bandwidth. It can be adjusted to a certain available piece of spectrum simply by changing the number of used sub-carriers. On the other hand, DS-CDMA and MC-CDMA require a fixed and relatively large fixed spectrum due to their fixed chip rate.
- Time-division duplex Media Access Control (MAC) with dynamic channel allocation used for unpaired spectrum allocations; asymmetrical services, and unlicensed usage is also supported.
- Simple and efficient high bit rate can be supported by allocating more sub-carriers and or timeslots. The feature is used in service level differentiation.
- No frequency planning option available; effective re-use factor of one.
- OFDMA has GSM backward compatibility requirement.
- Minimum bandwidth requirements for system deployment only 1.25MHz.
- OFDMA can provide a larger system capacity because it is not affected by intra-cell interference that is the dominant source of interference in DS-CDMA and MC-CDMA
- OFDMA can support larger data rate than DS-CDMA and MC-CDMA, because in CDMA as the data rate per user becomes larger, the spreading gain becomes

lower and the performance of the system deteriorates. Therefore, OFDMA with dynamic channel allocation is a better solution in this case [1].

The difference between OFDMA and MC-CDMA is that in MC-CDMA the users use all sub-carriers simultaneously, whereas in OFDMA each user uses a subset of sub-carriers. OFDMA systems can potentially outperform MC-CDMA, since instead of transmitting over all sub-carriers, users can choose to transmit over their best channels [54]. In MC-CDMA, orthogonal spreading codes are used to detect different users; this is not very practical. Due to code distortion by multi-path fading channels, MC-CDMA loses its orthogonality in the uplink for a cell. This gives rise to complicated equalization techniques and also introduces a loss in SNR performance and eliminates the complexity advantage of OFDM over single carrier techniques [1]. Therefore, MC-CDMA suffers from inter-carrier and intra-cell interference.

The major advantage of OFDMA over MC-CDMA and DS-CDMA is the intra-cell interference. The system capacity is inversely proportional to the total amount of interference power; therefore system capacity is increased with an OFDMA system, which has no intra-cell interference.

In OFDMA, since all the users use distinct sub-carriers, both inter-symbol and inter-carrier interference are eliminated if proper frequency and timing offsets are kept between users. We have addressed the issues of synchronization in Chapter 4, and proposed an enhanced scheme to mitigate these adverse effects properly.

5.1.2 WiMAX: From Fixed To Mobile BWA

WiMAX is one of the applications of OFDMA technology. WiMAX is an IEEE 802.16 standards-based wireless technology that provides high-throughput broadband connections over long distances. WiMAX can be used for broadband connections in hotspots, cellular backhaul, and high-speed enterprise connectivity for business. WiMAX will enable the users to be able to have high-throughput coverage of up-to 50km radius and mobile speed of up-to 120km/hr [20]. WiMAX forms the basis for future evolution to mobile broadband wireless access. Figure 5.2 shows a WiMAX Broadband Wireless Access solution that can provide tremendous amount of benefits and revolutionize the wireless access systems [20].

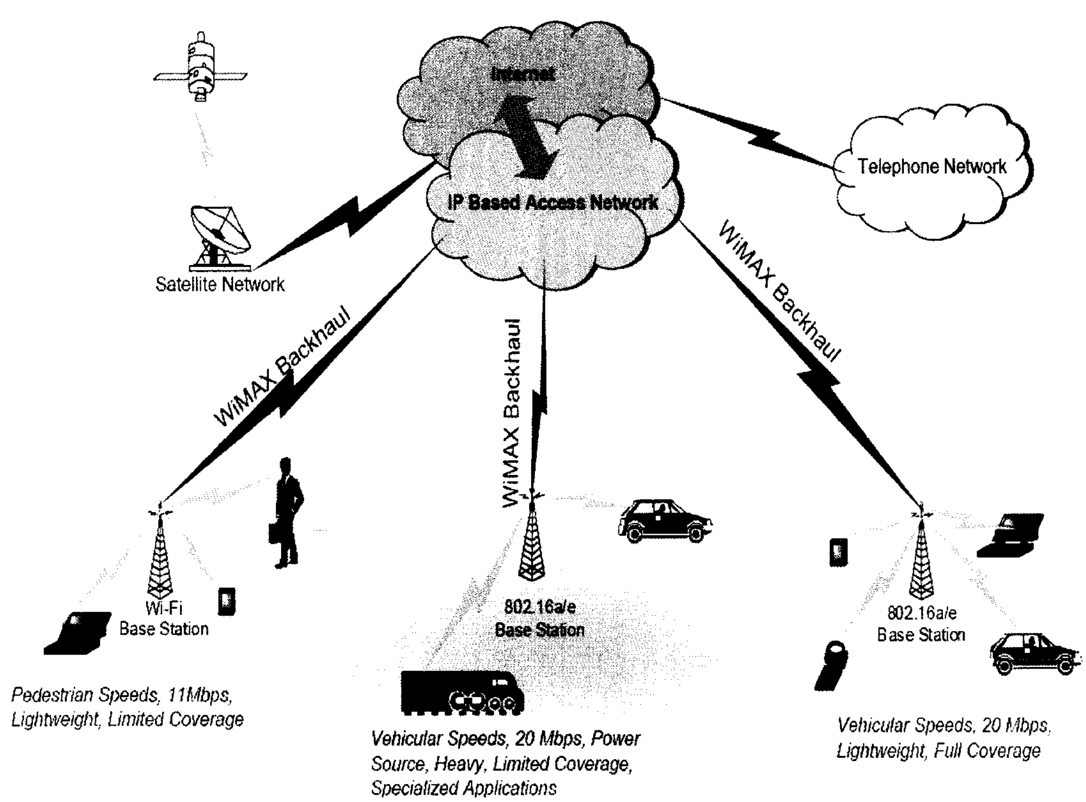

Figure 5.2 WiMAX Broadband Wireless Access (BWA) solution

WiMAX offers carrier grade QoS, prioritization of voice/video and data. Voice over IP (VoIP), wireless video and imaging, and mobile TV can be some of the direct services that WiMAX technology can provide reliably and efficiently. An important aspect of the WiMAX is that it is based on a scalable multiple access i.e., OFDMA, and a dynamic resource allocation scheme.

5.2 Resource allocation in OFDMA

Resource allocation of OFDMA system consists of allocating sub-carriers, bits and power level to each user properly, based on the user's current channel state information. In an OFDMA system, different users observe multi-path fading and have independent fading parameters due to their different locations. If a sub-carrier appears to be in deep fade for one user, it may not be in deep fade for other users. Hence, multi-user system creates channel diversity, which increases with the number of users. Therefore, in a multi-user OFDM environment, the system needs to allocate the resources to the users appropriately and efficiently.

The problem of assigning resources to the different users in an OFDMA system has recently been an area of active research. There are two classes of resource allocation scheme; fixed resource allocation [9], and dynamic resource allocation [10], [11], [12]. Fixed resource allocation scheme is not optimal because it assigns the system's resources to each user regardless of the current channel conditions. By applying fixed allocation, the system neglects the channel diversity and does not use the deep faded sub-carriers for other users, to whom they do not seem deep faded.

Dynamic resource allocation on the other hand, is an efficient method of allocating resources. Dynamic resource allocation assigns a dimension adaptively or dynamically to the users based on their channel gains and makes use of multi-user diversity to obtain higher system performance. The main focus of this section is on Dynamic Resource Allocation (DRA) techniques in an OFDMA system.

Throughput performance comparisons of static vs dynamic OFDMA resource allocation is shown in Figure 5.3, where we see a distinguishable advantage of dynamic resource allocation over fixed or static resource allocation scheme. We notice that as the number of users increases (after the second user), the dynamic resource allocation performance improves further than static scheme, this is due to fact that the dynamic resource allocation uses the channel diversity, thus gaining more capacity. The simulations were carried out with the parameter specifications described in Table 4.1

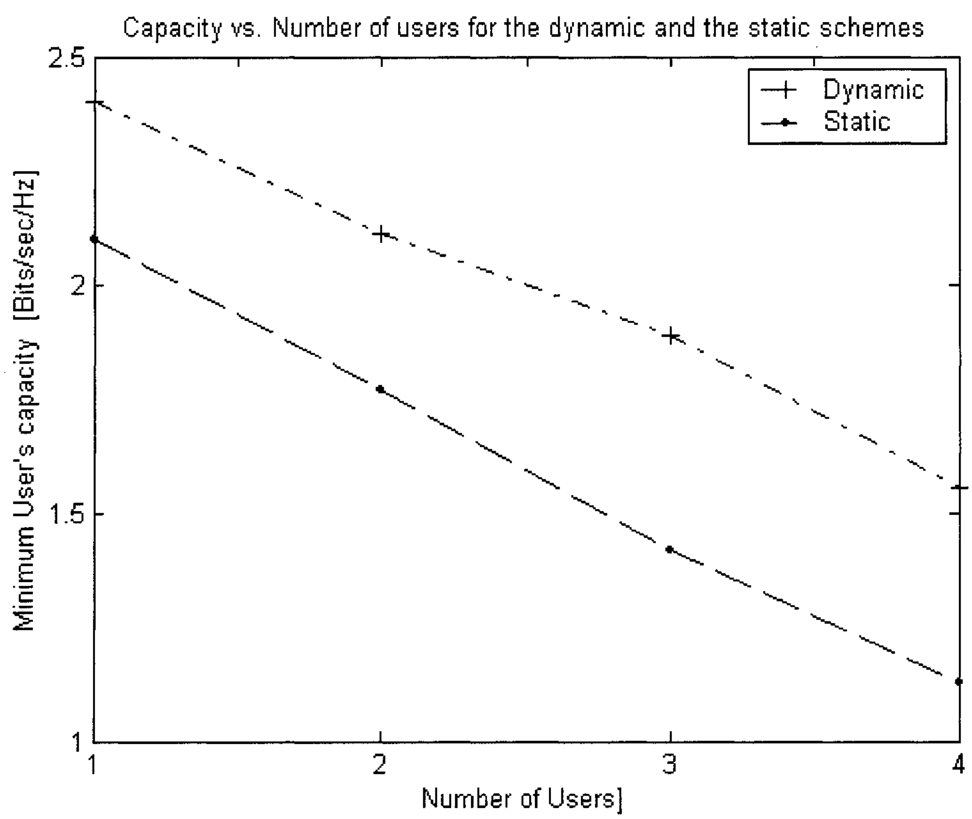

Figure 5.3 Capacity vs. number of users for the dynamic and the static schemes

5.2.1 Dynamic Resource Allocation in OFDMA

There are two types of optimization methods in dynamic multi-user OFDM literature: Margin Adaptive (MA) [12] and Rate Adaptive (RA) [10],[43],[44]. The MA optimization's objective is to achieve a minimum overall transmit power with the constraints on the user data rate and Bit Error Rate (BER). The RA optimization's goal is to maximize the minimum user's capacity within a total transmit power and BER constraint. These optimizations are non-linear, and computationally burdensome for processors to solve, and are not cost effective for a real time implementation. For example, for K users, and N sub-carriers, we need K^N computations for sub-carrier allocations.

Most dynamic resource allocation optimization methods researched have been iterative non-linear methods suitable for offline optimization. In the special high sub-carrier SNR case, an iterative root-finding method has linear-time complexity in the number of users, with $N \log N$ complexity in the number of sub-carriers [11]. Moreover, in [11], the non-linear optimization problems were transformed into linear optimization problem with integer variables. However, even with integer programming, the complexity increases exponentially with the number of users. Suboptimal algorithms are always of interest to balance the tradeoff between complexity and performance.

Therefore, we choose to use a suboptimal algorithm presented in [10] and [43] with some modification and simplification, to simulate our system and apply our proposed solutions, in order to evaluate the OFDMA system performance appropriately.

5.2.2 OFDMA System Model

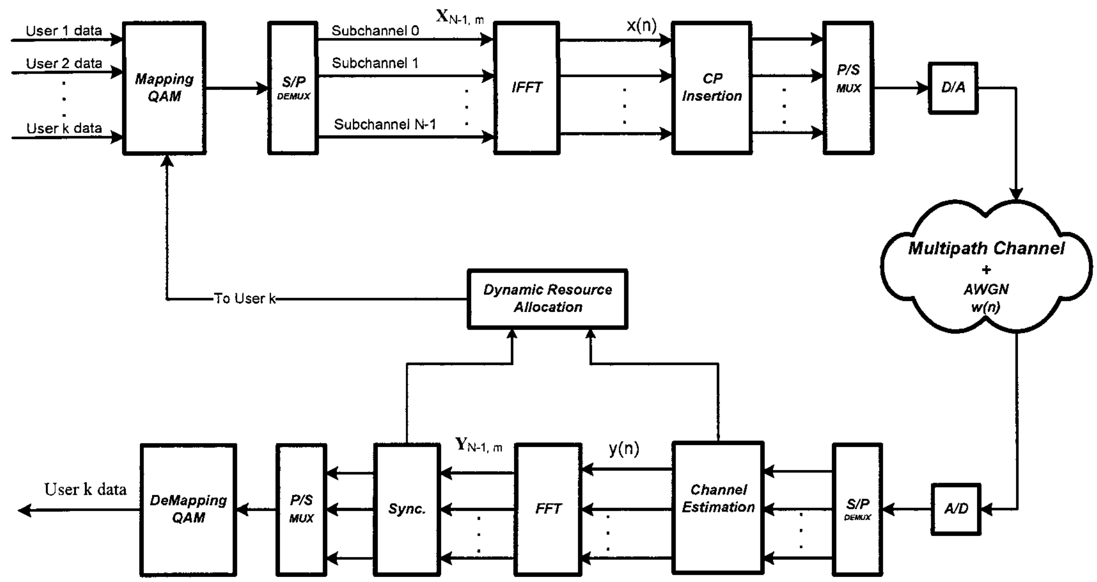

Figure 5.4: System model and framework for multi-user OFDM transreceiver

Figure 5.4 shows the system model for a multi-user OFDM or OFDMA system. This system model incorporates the proposed solutions for channel estimation (Chapter 3) and synchronization (Chapter 4). OFDMA system allows multiple users to transmit simultaneously on different sub-carriers, ensuring full utilization of the multi-user diversity gain. The Radio Base Station (RBS) station estimates each user's channel state information. With this knowledge, the radio base station then runs the optimization algorithm to allocate resources dynamically to each user, based on the estimated channel gain. Furthermore, frequency offsets are estimated and transmitted to users for frequency synchronization and correction.

How do we allocate the N sub-carriers and P total power to the K users to optimize the resource allocation problem stated above while satisfying the Quality of Service (QoS)?

At the RBS each user is allocated a subset of sub-carriers by an iterative optimization algorithm that we will shortly illustrate. The sub-carrier allocation is made known to all the users through a control channel. Then each user's sub-carrier is assigned to a certain power level $p_{k,n}$. Sub-carriers are independent and sub-carrier sharing among users is not allowed by the system. Each of the user's bits are modulated into M-QAM symbols, which are subsequently combined using the IFFT into an OFDMA symbol which is then transmitted through a time-varying, Rayleigh fading channel with a bandwidth B.

The RBS is able to get the channel state information by using the proposed methodology in Chapter 3, and estimate the channel and track the channel variations dynamically or adaptively. The estimate is then used as input to the resource allocation algorithms.

In [10], the authors would like to find optimal S_k and $P_{k,n}$ in order to maximize the throughput of the user with worse condition. We follow their path and choosing Rate Adaptive method, because our aim is also to provide good throughput for all users. However, the optimization technique in [10] is non-linear and difficult to solve and computationally burdensome for real time implementations. By introducing a sharing factor [10], the optimization problem can be converted to a convex optimization problem, which can be solved by some optimization tools, such as AMPL (*an Algebraic Modeling Programming Language for the linear, nonlinear and integer programming problems often encountered in optimization*) [10].

However, even with the relaxation that sharing is allowed, there is a need to perform matrix decomposition and inversion to find the optimal solution. Hence, These operations are not practical in real-time implementation of dynamic resource allocation algorithms.

Therefore, we recommend on solving the optimization problem by performing sub-carrier allocation and power allocation sequentially for simplicity and efficiency.

I. Sub-carrier allocation
   - Greedy algorithm [10]– allow the user with the least allocated capacity to choose the best sub-channel

II. Power allocation
   - Water-filling algorithm [43]– Pour more power into the sub-carriers with high channel-to-noise gain

Since rate adaptation is the main aim, maximizing all the users' capacity by allowing the user with least allocated capacity to choose the best subcarrier can provide a better solution to the rate adaptive optimization problem, and we can keep the capacity of all the users as high as possible proportionally and according to the requirement.

We can follow the following three steps algorithm to allocating the resources:

Step 1. Determine the total number of sub-carriers N [1]. Find out how many users k and also calculate how many sub-carriers N_k for each user

Step 2. Assign sub-carriers to each user (using [10], Greedy algorithm)

Step 3. Assign the power $p_{k,n}$ for each user's sub-carriers (using [43], Water-filling algorithm)

---

[1] In order to determine the total number of sub-carriers available we use the help of the method derived and explained in Chapter 1 Section 2.2, and by applying the system design constraints in terms of available bandwidth and delay profile, and Doppler spread)

The rate adaptive resource allocation problem is formulated as [43], an objective function (5.1), where we can maximize the capacity based on constraints listed below :

$$\max_{p_{k,n},c_{k,n}} \sum_{k=1}^{K} \sum_{n=1}^{N} \frac{c_{k,n}}{N} \log_2 \left(1 + \frac{p_{k,n} g_{k,n}^2}{N_o \frac{B}{N}} \right) \tag{5.1}$$

Subject to the following constraints:

C1:	$\sum_{k=1}^{K}\sum_{n=1}^{N} p_{k,n} \leq P_{total}$	**Power constraint**
C2:	$p_{k,n} \geq 0 \quad \forall\ k,n$	**Non-zero power**
C3:	$c_{k,n} \in \{0,1\} \quad \forall\ n,k$	**Exclusive sub-carrier sub-carriers assignment**
C4:	$\sum_{k=1}^{K} c_{k,n} = 1 \quad \forall\ n$	**No sub-carriers sharing**
C5:	S_k	**Subsets of sub-carriers; disjoint for all** k
C6:	$R_1 : R_2 : \cdots\ R_K = \gamma_1 : \gamma_2 : \cdots\ \gamma_K$	**Ensure proportionality among users**

where,

$g_{k.n}$ - Channel gain for user k on sub-carrier n
$p_{k,n}$ - Power for user k on sub-carrier n
S_k - Subset of sub-carriers assigned to user k
N - Number of sub-carriers
K - Number of users
P_{total} - Base station total transmit power
B - Bandwidth
N_o - Power spectrum density of Additive White Gaussian Noise AWGN
γ_k - A set of predetermined values that are used to ensure proportionality among users

R_k is the channel capacity for user k defined as:

$$R_k = \sum_{n=1}^{N} \frac{c_{k,n}}{N} \log_2 \left(1 + \frac{p_{k,n} g_{k,n}^2}{N_o \frac{B}{N}} \right) \tag{5.2}$$

When all of γ_k 's have the same value, the objective function in (5.1) is similar to the objective function for the rate adaptive method [10], because maximizing the overall capacity while making all R_k 's equal, is equivalent to maximizing the minimum user's capacity. Furthermore, the proportionality constraint in (5.1) also prevents the situation that no resource allocation scheme exists because of impractical capacity requirements from some of the users. In addition, the operators can also define certain users with higher preferences to get higher capacity, thereby differentiating the service level.

- Assumption: It is assumed that each user experiences independent fading and the channel gain of user k in sub-carrier n is unique
- Justification: No sub-carrier sharing among users is allowed by the system

The channel-to-noise gain for user k in sub-carrier n is defined as:

$$H_{k,n} = \frac{g_{k,n}^2}{N_o \frac{B}{N}} \tag{5.3}$$

5.2.2.1 Sub-carrier allocation

We use the suboptimal algorithm in [10] to allocate the sub-carriers. First, the channel gains are estimated by the RBS for each user. We can certainly say that the poorest user in terms of capacity, suffers from the lowest channel gain. We allocate the sub-carriers using the algorithm in [43], in a way to allow the user with the least allocated capacity to choose the best sub-channel.

- Assumption: Equal power distribution on all sub-carriers. Therefore, reducing the complexity of the system.
    - Implication: The algorithm in [43], assumes that equal power is distributed into every subcarrier. This is valid since the total power that is radiated by the RBS is equally distributed among all sub-carriers However, when the number of users increases, equal power distribution does not equalize every user's capacity. By transferring power from the users with high capacity to the users with low capacity, the minimum user's capacity could be even increased, which also makes sense.

Sub-carriers are initially assigned to the users who have the highest channel gain in those sub-carriers. After all the sub-carriers are assigned, the capacity for each user is calculated, assuming equal power distribution. Then iteratively, the lowest rate user is given the first choice to choose the next best sub-carrier. The process is repeated until all sub-carriers are allocated.

1. Initialization (Enforce zero initial conditions)

   Set $R_k = 0,\quad S_k = \phi\quad$ for $\quad k = 1,2,\cdots K\quad$ and $\quad I = \{0,1 \quad \cdots \quad N-1\}$

2. Allocate best subcarrier for each user i.e., sub-carriers are initially assigned differentially to the user who has the highest channel gain in that sub-carrier.

for $k=1$ to K {

find n such that $|H_{k,n}| > |H_{k,m}| \quad for\ all \quad m \in I$

put n to S_k and $I = I - \{n\}$

update R_k

}

3. Iteratively, give lowest rate user first choice to choose the best sub-carrier

while $(I \neq \phi)$ {

find k such that $R_k / \gamma_k \leq R_k / \gamma_i \quad for\ all \quad i,\ 1 \leq i \leq K$

for found k, find n such that $|H_{k,n}| > |H_{k,m}| \quad for\ all \quad m \in I$

for found k and n, let $S_k = S_k \cup \{n\}$, $I = I - \{n\}$

update R_k

}

5.2.2.2 Power Allocation

We use the optimal power allocation defined in [43]. The total power allocated for user k is defined as:

$$P_{k,total} = \sum_{n=1}^{N_k} p_{k,n} = N_k p_{k,1} + \sum_{n=2}^{N_k} \frac{H_{k,n} - H_{k,1}}{H_{k,n} H_{k,1}} \quad for\ \ k = 1, 2, \cdots, K \tag{5.4}$$

For user k, given total power $P_{k,total}$, water-filling is used to maximize capacity. Optimal power distribution strategy for a single user is derived [43].

$$\frac{H_{k,m}}{1 + H_{k,n} p_{k,m}} = \frac{H_{k,n}}{1 + H_{k,n} p_{k,n}} \quad for\ \ m, n \in S_k \quad and \quad k = 1, 2, \cdots, K \tag{5.5}$$

Equation (5.5) can be written as:

$$p_{k,n} = p_{k,1} - \frac{H_{k,n} - H_{k,1}}{H_{k,n} H_{k,1}} \quad for\ n = 1, 2, \cdots N_k\ and\ k = 1, 2, \cdots, K \tag{5.6}$$

where, N_k is the number of sub-carriers in S_k

Equation (5.6) shows the optimal power distribution for a single user. More power will be put into the sub-carriers with high channel-to-noise gain. This is water-filling in the frequency domain [43].

In general, it can be proved that there must be an optimal subcarrier and power allocation scheme that satisfies the proportionality constraints and the total power constraint [43]. Furthermore, the optimal scheme must utilize all available power. Several facts lead to the above conclusion. First, capacity of a user is maximized if waterfilling algorithm is adopted. Furthermore, the capacity function is continuous with respect to the total available power to the user. In other words, $R_k(P_{k,total})$ is continuous with $P_{k,total}$. Second, if the optimal allocation scheme does not use all the available power, there is always a way to redistribute the unused power among users while maintaining the capacity ratio constraints, since $R_k(P_{k,total})$ is continuous with $P_{k,total}$ for all k. Thus, the overall capacity is further increased [43].

5.3 Simulation Results

Table 4.1 Simulation Parameters

Parameter	Specification
Carrier Frequency f_c	3.3GHz
Bandwidth	1.75MHz
Number of sub-carriers	256
OFDM symbol time	160 μsec
Number of bits per OFDM symbol	1024
Number of users	4
Delay Spread length	$0 \sim 10\,\mu sec$
Length of CP	$32\,\mu sec$
Channel Model	Multi-path time varying
Signal constellation	16 QAM
Number of bits per QAM symbol	4
Symbol period (T)	$0.625\ \mu sec$
Max Data Rate (Mbs)	6.4

In the simulations the wireless channel is modeled as a multi-path time-varying channel. With maximum delay spread of $10\,\mu sec$ and Doppler frequency is set to 15Hz. At the RBS the synchronization is preformed and fed back to the user via a dedicated control channel. After the channel is estimated, the RBS dynamically allocates resources to the users. The instantaneous knowledge of the channel for each user is used to allocate the sub-carriers accordingly, and subsequently allocate the transmit power for each sub-carrier. Now we will evaluate the cost of this type of realistic channel knowledge by dynamically assigning the resources. Here we will compare the results with static and dynamic resource allocation schemes.

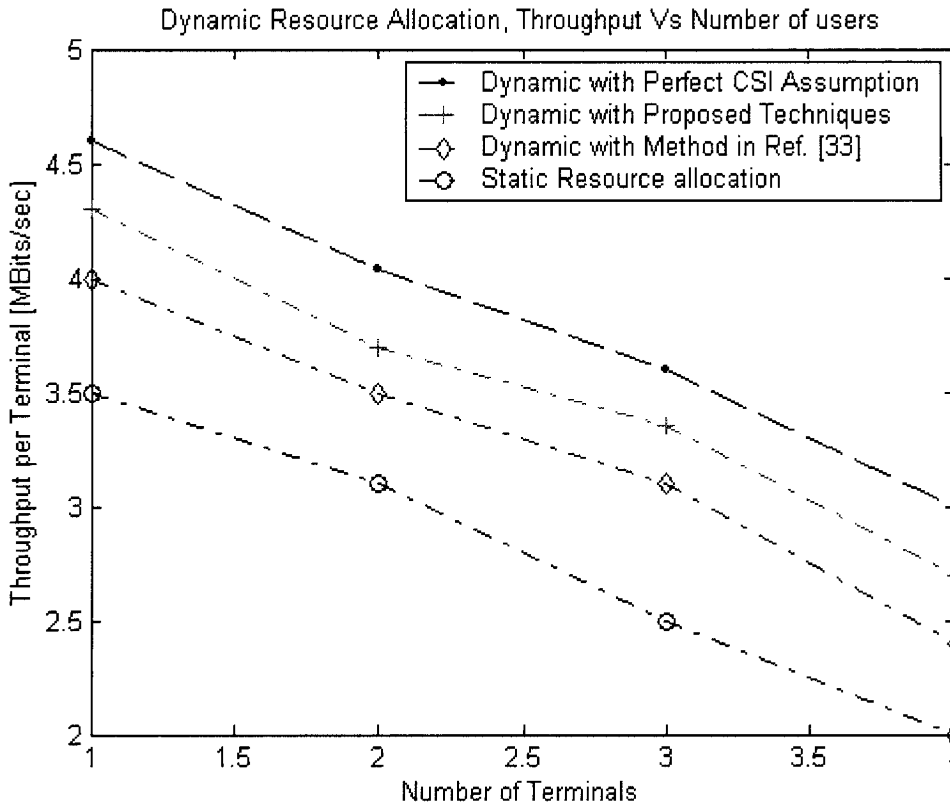

Figure 5.5: Average absolute throughput for a varying number of terminals in the cell

In essence, the success of dynamically assigning sub-carriers to terminals is directly related to the estimated channel knowledge of the RBS and to a working signaling system. Gaining the channel knowledge and having some type of signaling is costing the system performance, and we can clearly see that with respect to the perfect channel knowledge assumption. However, we can be re-assured that we still get better results than static resource allocation schemes. We have used our simple approach proposed in Chapter 3 of this thesis. Our channel estimation is less complex than most channel estimation algorithms and performs very well in a time varying environment without taking additional bandwidth.

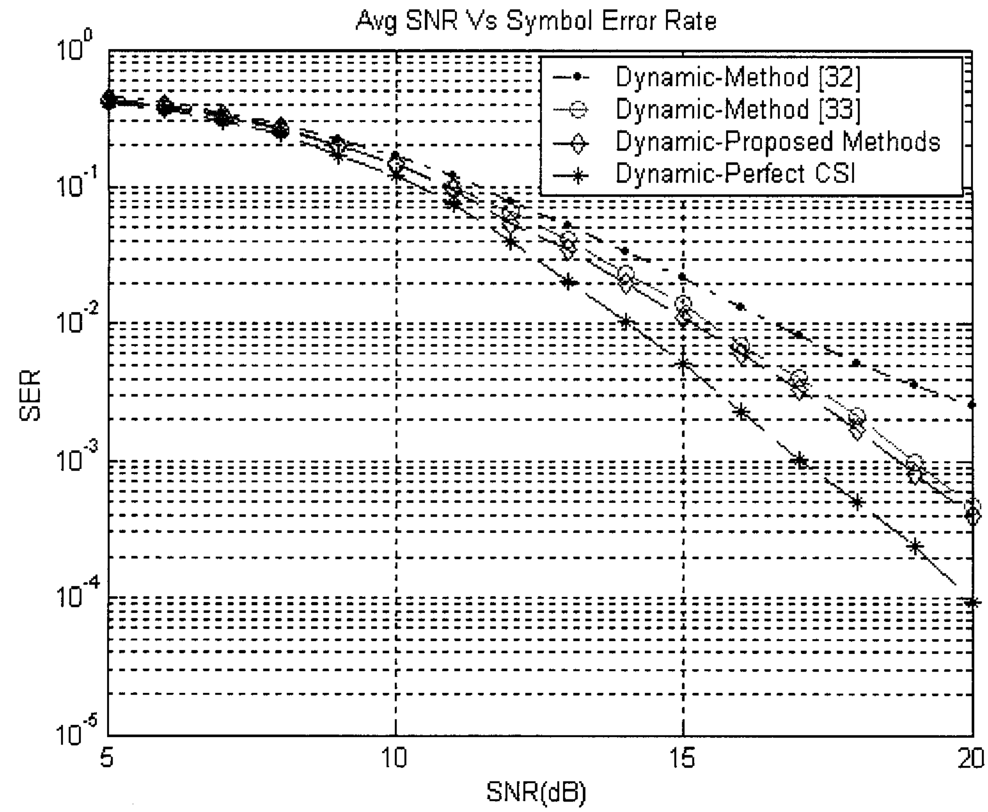

Figure 5.6: Symbol Error Rate Vs Signal to Noise Ratio

5.4 Summary

We have investigated the performance difference of static and dynamic schemes for a scenario where realistic channel knowledge is added to the system model through a signaling scheme. After analysis of various schemes a solution to the resource allocation is recommended and shown to provide good results. The added advantage of our method compared with other methods is that, it has significantly lower complexity, and simulation results clearly demonstrate that it yields higher user data rates.

The number of terminals in the cell, the transmission power per sub-carrier, the delay spread and movement speed of the terminals can be varied. We have compared average throughput and good put per terminal. We found that a realistic overhead model decreases the performance of both static and dynamic schemes such that the overall ratio favors in all cases except for very high speeds the dynamic rather than the static scheme especially in realistic system environments. The performance of an OFDMA system employing dynamic resource allocation and channel estimation using the CP scheme is virtually indistinguishable from the performance of a system employing other estimation algorithms. Our method performs the same as the retraining algorithms that take huge amount of precious bandwidth. Therefore, our method is more efficient in terms of bandwidth and yields a good solution for our high capacity dynamic resource allocation. In addition, we have enhanced the system by performing frequency synchronization, which is important for securing orthogonality and preventing ICI effects and in order to have a reliable system performance.

Chapter 6

Conclusion

This research has presented an enhanced OFDM based broadband wireless access communication system, which is efficient, has low complexity and is reliable. CP method is used to perform the channel estimation. The benefit of adding CP was two fold. First it, eliminated ISI caused by multi-path propagations. Second, we extended the use of CP in our research and proposed a channel estimation technique with the help of CP in order to enhance and design a more realistic and practical system in a time varying environment. To the best of our knowledge, most research to date on dynamic resource allocation have assumed perfect channel knowledge and perfect synchronization.

We have designed a new dynamic channel estimation scheme using an adaptive signal processing technique with which, we can estimate and track the channel in a time varying environment very efficiently without wasting any bandwidth. We proposed a simple and enhanced frequency domain approach to correct frequency offset errors. Using the CP method we have analyzed and implemented a solution to synchronization in time domain for a time varying channel. However, using time domain approach, we found that good results can be achieved only for AWGN channels and not in time varying and frequency selective channels. Therefore, in order to reduce the channel dependency, for frequency offset estimation we have proposed a simple frequency domain approach, which performs very well in any environment.

The time domain approach using the CP method that Beek has used works, but only in case of AWGN. However, in a realistic multi-path environment the time domain approach of frequency offset correction does not perform well and produces high amount of errors. Our frequency domain approach is better because it is independent of the delay spread and related problems to frequency synchronization. It is also an improvement to other frequency domain synchronization approaches such as the blind methods and methods which perform over sampling, and that normally have more computational complexity. Thus, with our scheme the frequency offsets are estimated and corrected in frequency domain, using an enhanced and fast algorithm that performs well in a multi-path frequency selective environment.

Finally we have analyzed and observed the benefits and cost of dynamically allocating resources based on proper channel state information, using multi-user diversity, and improving the overall system's performance and capacity of OFDMA systems. We conclude that years of research on OFDM technology have provided many thriving results. Large-scale commercial deployment of OFDM based broadband wireless access systems such as WiMAX is foreseeable in the near future.

6.1 Research contribution as a result of this thesis

Some of the primary contributions of this thesis are as follows:

We have devised an efficient method of estimating the proper channel state information. We have studied various types of channel estimation methods, and we found that the CP method to be the best method for an OFDM system. Firstly, because it is less complex than most other algorithms. Secondly, we are utilizing the redundancy introduced in an efficient manner that will enhance the overall performance of the system. We have proposed an efficient channel estimation technique in time varying environment without affecting the bandwidth. Using this estimated channel knowledge we could allocate resources properly.

In addition, we have investigated and analyzed the synchronization problems in an OFDMA system. We proposed an enhanced novel scheme for properly estimating the frequency synchronization parameter, in the uplink of multi-user OFDM systems [14].

This research also considered dynamic resource allocation to each user by the base station in order to maximize the sum of user data rates, subject to constraints on total power, bit error rate, and proportionality among user data rates. One of the contributions of this thesis is that we apply a realistic knowledge of the channel state information and then the actual benefits of dynamic resource allocation of OFDMA was assessed, verified and proved. Finally, we have applied our efficient and enhanced solutions for channel estimation and synchronization to a newly designed OFDMA system and evaluated its performance. In comparison to other schemes we achieved good results with less complexity.

6.2 Future work

- Frequency domain channel estimation can be investigated and implemented and compared with current time domain approach. As discussed in Chapter 3 Section 3.2 the frequency domain approach the pilot symbols need IDFT computation, which may cause Peak-to-Average-Power-Ratio (PAPR) problem and more complexity. Designing a better and less complex scheme to mitigate PAPR problem is an excellent and challenging topic of further research.
- To find out what are the problems related to estimation error if the system is not synchronized properly and if there are synchronization errors
- To find out the convergence criteria for the estimation algorithms
- Multiple antenna diversity and MIMO OFDMA will bring a lot of benefits to OFDMA systems and is a good topic of research.
- Channel coding in an OFDMA system is an interesting topic of research that can bring improvement and make the system more reliable by reducing the error rate

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