CHAPTER 1

INTRODUCTION

1.1 WIRELESS COMMUNICATION TECHNOLOGIES

The Mobile wireless industry started its technology creation, revolution and evolution in the early 1970s. The mobile wireless technology has developed from Zero Generation (0G) to Fourth Generation (4G) in the past few decades. The large scale mobile wireless communication was introduced by the First Generation (1G) technology. The wireless communication quality was improved by the Second Generation (2G) technology, in which the analog communication was replaced by digital communication. The data communication with voice has been the main focus in the Third Generation (3G) technologies and a converged network has emerged for both voice and data communication. Later, the research has paved the opportunity for the development of 4G.

1.1.1 Second Generation (2G) Technology

Global System for Mobile communications (GSM) is a second-generation wireless telephone technology. It cannot transfer data, such as email or software, other than the digital voice. However, Short Message Service (SMS) messaging is also available as a form of data transmission for some standards. The 2G cellular telecom networks were commercially launched on the GSM standard in 1991. GSM service is used by over 2 billion people across more than 212 countries and territories. The availability of the
GSM standard makes international roaming very common between mobile phone operators, thus enabling the subscribers to use their phones in many parts of the world. Depending on the type of multiplexing used 2G technologies can be divided into Time Division Multiple Access (TDMA) based and Code Division Multiple Access (CDMA) based standards. 2G makes use of a Compression Decompression (CODEC) Algorithm to compress and multiplex digital voice data through which a 2G network can pack more calls per amount of bandwidth as a 1G network. 2G cell phone units were generally smaller than 1G units, since they emitted less radio power.

Some benefits of 2G were that the Digital signals consume less battery power which helps mobile batteries to last long. It also improves the voice clarity and reduces noise in the line. The use of digital data service assists mobile network operators to introduce short message service over the cellular phones. The data and voice calls have been provided safety and secrecy through the digital encryption. The use of 2G technology requires strong digital signals to help mobile phones work. The digital signals would be weak if there is no network coverage in any specific area.

General Packet Radio Service (GPRS) is the 2.5G in wireless communication system and it is developed between 2G and 3G. In this generation the 2G systems have implemented a packet switched domain in addition to the circuit switched domain. "2.5G" is an informal term, invented solely for marketing purposes, unlike "2G" or "3G" which are based on the official standards defined by the International Telecommunication Union (ITU). Data rates from 56 Kbps up to 115 Kbps can be provided by the GPRS. It can be used for services such as Wireless Application Protocol (WAP) access, Multimedia Messaging Service (MMS) and for Internet communication services such as email and World Wide Web access. GPRS
data transfer is usually charged per megabyte of traffic transferred, while data communication via traditional circuit switching is billed per minute of connection time, independent of whether the user actually is utilizing the capacity or it is in an idle state. The 2G networks supports the services such as WAP, MMS, SMS and mobile games.

EDGE is a digital mobile phone technology which acts as a bolt-on enhancement to 2G and 2.5G GPRS networks. This technology works in GSM networks. EDGE is a superset to GPRS and can function on any network with GPRS deployed on it, provided the carrier implements the necessary upgrades. EDGE technology is an extended version of GSM. It allows the clear and fast transmission of data and information. It is also termed as International Mobile Telecommunication-Single Carrier (IMT-SC) or single carrier. EDGE is a radio technology and is a part of third generation technology. EDGE technology is preferred over GSM due to its flexibility to carry packet switch data and circuit switch data.

The use of EDGE technology has increased the use of black berry, N97 and N95 mobile phones. EDGE transfers data in fewer seconds if it is compared with GPRS Technology. For example, a usual text file of 40 KB is transferred in 2 seconds as compared to the transfer in GPRS technology, where it is 6 seconds. The major advantage of using EDGE technology is one that does not need to install any additional hardware and software.

1.1.2 Third Generation (3G) Technology

3G is the third generation of mobile phone standards and the technology residing between 2G and 4G. It is based on the International ITU family of standards under the International Mobile Telecommunications program, International Mobile Telecommunications-2000 (IMT-2000). 3G technology enables network operators to offer the users a wider range of more
advanced services while achieving greater network capacity through improved spectral efficiency. The services include wide area wireless voice telephony, video calls, and broadband wireless data, all in a mobile environment. Additional features like High-Speed Packet Access (HSPA) data transmission capabilities are capable to deliver speeds up to 14.4 Mbps on the downlink and 5.8 Mbps on the uplink. Spectral efficiency or spectrum efficiency refers to the amount of information that can be transmitted over a given bandwidth in a specific digital communication system. HSPA is a collection of mobile telephony protocols that extend and improve the performance of existing Universal Mobile Telecommunications Systems (UMTS) protocols.

Unlike IEEE 802.11 (common names Wireless Fidelity (Wi-Fi) or Wireless Local Area Network (WLAN)) networks, 3G networks are wide area cellular telephone networks which evolved to add in a high-speed internet access and a video telephony. IEEE 802.11 networks are short range and high-bandwidth networks mainly developed for data. Wi-Fi is the common name for a popular wireless technology used in home networks, mobile phones, video games and more. The notebook is connected to the wireless access point using a PC (Personal Computer) wireless card. A videophone is a telephone which is capable of both audio and video duplex transmission. 3G technology makes use of TDMA and CDMA and also value added services like mobile television, Global Positioning System (GPS) and video conferencing. The basic feature of 3G Technology is the fast data transfer rates.

The 3G technology is flexible, because it is able to support the five major radio technologies. These radio technologies operate under CDMA, TDMA and Frequency Division Multiple Access (FDMA). CDMA holds for International Mobile Telecommunication-Direct Spread (IMT-DS) and
International Mobile Telecommunication-Multi-Carrier (IMT-MC). TDMA accounts for International Mobile Telecommunication-Time Code (IMT-TC) and International Mobile Telecommunication Single Carrier (IMT-SC). FDMA has only one radio interface known as International Mobile Telecommunication -Frequency Code (IMT-FC). The 3G technology is inexpensive due to the agreement of industry. This agreement took place in order to increase its acceptance by the users. 3G system is user-friendly to work with the 2G technologies. The aim of the 3G is to allow for more coverage and growth with minimum investment. There are many 3G technologies such as Wideband-Code Division Multiple Access (W-CDMA), GSM, EDGE, UMTS, Digital Enhanced Cordless Telecommunications (DECT), Worldwide Interoperability for Microwave Access (WiMax) and CDMA 2000. Enhanced data rates for GSM evolution or EDGE is termed to as a backward digital technology, as it can operate with older devices.

3G has the following enhancements over 2.5G and previous networks:

- Enhanced audio and video streaming
- Several Times higher data speed
- Video-conferencing support
- Web and WAP browsing at higher Speed
- Internet Protocol Television (IPTV) support

High-Speed Downlink Packet Access (HSDPA) is a mobile telephony protocol which is also called as 3.5G. It provides a smooth evolutionary path for UMTS based 3G networks allowing for higher data transfer speeds. HSDPA is a packet-based data service in W-CDMA downlink with data transmission up to 8-10 Mbps (20 Mbps for Multiple
Input and Multiple Output (MIMO) systems) over a 5 MHz bandwidth in W-CDMA downlink. HSDPA implementations include Adaptive Modulation and Coding (AMC), MIMO, Hybrid Automatic Request (HARQ), fast cell search, and advanced receiver design.

The 3.75G refers to the technologies beyond the well defined 3G wireless/mobile technologies. High-Speed Uplink Packet Access (HSUPA) is a UMTS / W-CDMA uplink growth technology. The HSUPA mobile telecommunications technology is directly related to HSDPA and the two are complimentary to one another. The HSUPA enhances the one to one communication such as e-mail and real time person to person gaming applications with symmetric higher data rates. Traditional business applications along with many consumer applications will benefit from enhanced uplink speed. HSUPA initially boost the UMTS / W-CDMA uplink up to 1.4Mbps and in later releases up to 5.8Mbps.

1.1.3 Fourth Generation (4G) Technology

4G refers to the fourth generation of cellular wireless standards. It is a successor to 3G and 2G families of standards. It is the extended of 3G technology with more bandwidth and services offers in the 3G. The expectation for the 4G technology is basically the high quality audio/video streaming over end to end Internet Protocol. WiMAX or mobile structural design will gradually become more translucent. Some of the companies have been trying to achieve 4G communication at 100 Mbps for mobile users and up to 1 Gbps over fixed stations.

1.2 Orthogonal Frequency Division Multiplexing

Orthogonal Frequency Division Multiplexing (OFDM) is a parallel transmission scheme, where a high-rate serial data stream is split into a set of
low-rate substreams each of which is modulated on a separate subcarrier based on Frequency Division Multiplexing (FDM) technique.

In a usual parallel data system, the total signal frequency band is divided into ‘N’ non-overlapping frequency sub-channels. Each sub-channel is modulated with a separate symbol and then the ‘N’ sub-channels are frequency multiplexed. It is good to avoid spectral overlap of channels to eliminate inter-channel interference. However, this leads to inefficient use of the available spectrum. The bandwidth inefficiency, the ideas proposed where to use parallel data and FDM with overlapping sub-channels in which each carrying a signaling rate ‘r’ is spaced ‘w’ apart in frequency to avoid the use of high speed equalization and to combat impulsive noise and multipath distortion, as well as to fully use the available bandwidth.

![Multicarrier modulation technique](image)

**Figure 1.1 Multicarrier modulation technique**

Figure 1.1 illustrates the conventional non-overlapping multicarrier technique. In a single carrier system, a single fade can cause the entire link to fail but in multi carrier system only a small percentage of the subcarrier is affected. Error correction coding can then be used to correct the few wrong subcarriers.

The word orthogonal indicates the precise mathematical relationship between the frequencies of the carriers in the system. In a normal
frequency division multiplex system, many carriers are spaced apart in such a way that the signals can be received using conventional filters and demodulators. In such receivers, guard bands are introduced between the different carriers and in the frequency domain that results in lowering of spectrum efficiency.

![Orthogonal multicarrier modulation technique](image)

**Figure 1.2 Orthogonal multicarrier modulation technique**

As shown in Figure 1.2 by using the overlapping multicarrier modulation technique, almost 50% of bandwidth is saved. To realize the overlapping multicarrier technique, the crosstalk between subcarriers is present. However to reduce the crosstalk between subcarriers, orthogonality is maintained between the different modulated carriers. One of the main reasons to use OFDM is to increase the strength of the carrier against frequency selective fading and narrowband interference. The bandwidth of the subcarriers becomes small when compared with the coherence bandwidth of the channel, the individual subcarriers experience flat fading, which allows for simple equalization. This implies that the symbol period of the substreams is made long compared to the delay spread over the time-dispersive radio channel.

The concept of OFDM orthogonality of the subcarriers has recently been applied widely in wireless communication systems due to its high data rate transmission capability with high bandwidth efficiency and robustness to
multipath delay. It is possible to arrange the carriers in an OFDM signal the side bands of individual carriers overlap and the signals are still received without adjacent carrier interference then these carriers must be mathematically orthogonal. The receiver act as a bank of demodulators, translating each carriers to DC, with the resulting signal integrated over a symbol period to recover the raw data if the other carriers all beat down the frequencies that, in the time domain, have a whole number of cycles in the symbol period T, then the integration process results in zero contribution from all these other carriers. Thus, the carriers are linearly independent (i.e., orthogonal) if the carrier spacing is a multiple of 1/T.

Much of the research focuses on the high efficient multi carrier transmission scheme based on “orthogonal frequency” carriers. The Discrete Fourier Transform (DFT) is employed in both modulation and demodulation in parallel data transmission. The use of DFT at the receiver and calculates the correlation values with the centre of frequency of each subcarrier. Then the transmitted data is recovered without any crosstalk. In addition, using the DFT-based multicarrier technique, frequency division multiple access is achieved not by band pass filtering but by baseband processing.

In addition, to eliminate the banks of subcarrier oscillators and coherent demodulators required by frequency-division multiplex, completely digital implementations could be built around special purpose hardware performing the Fast Fourier transform (FFT), which is an efficient implementation of the DFT. Recent advances in Very Large Scale Integration (VLSI) technology make high speed, large size FFT chips commercially affordable. Using this method, both transmitter and receiver are implemented using efficient FFT techniques. The Inter Symbol Interference (ISI) is avoided in OFDM by the use of Cyclic Prefix (CP) which is achieved by extending an OFDM symbol with some portion of its head (or) tail. Along with ISI, the Inter
Channel Interference that arises from the loss of subchannel orthogonality in an OFDM block is found to limit the performance of OFDM systems.

The error floor created by ISI and Inter Carrier Interference (ICI) creates challenges for the success of OFDM based High-definition Television (HDTV) broadcasting. Trellis coding and antenna diversity schemes can significantly lower the error floors due to ICI. However, the very long delay of Single Frequency Network (SFN) poses the possibility of duration of the ISI which exceeds the length of the guard interval. Such ISI is known as residual ISI and can be devastating, even in small amounts. Increasing the length of the guard interval to reduce the residual ISI has its limitations because it introduces a bandwidth penalty. It has been recently applied in wireless local area network for high data rate communication. With these advantages, OFDM has adopted by many wireless standards such as Digital Video Broadcasting (DVB), Digital Audio Broadcasting (DAB), WLAN and Wireless Metropolitan Area Network (WMAN).

The OFDM transmission scheme has the following key advantages:

- Implementation complexity is significantly lower than that of a single carrier system with an equalizer.
- In relatively slow time varying channels, it is possible to significantly enhance the capacity by adapting the data rate per subcarrier according to the signal-to-noise ratio of that particular subcarrier.
- OFDM is strong against narrowband interference, because such interference affects only a small percentage of subcarriers.
- OFDM makes single frequency networks possible, which is especially attractive for broadcasting applications.
On the other hand, OFDM also has some drawbacks compared with single carrier modulation:

- OFDM is more sensitive to frequency offset and phase noise.
- OFDM has a relatively large peak to average power ratio, which tends to reduce the power efficiency of the Radio Frequency (RF) amplifier.

1.3 SELECTION OF OFDM PARAMETER

The OFDM parameters are chosen based on various requirements. Usually, there are three main requirements to start with: bandwidth, bit rate and delay spread. The delay spread directly dictates the guard time.

As a rule, the guard time should be above two to four times the root mean squared delay spread. This value depends on the type of coding and Quadrature Amplitude Modulation (QAM). Higher order QAM (like 64-QAM) is more sensitive to ICI and ISI than Quadrature Phase Shift Keying (QPSK); while heavier coding obviously reduces the sensitivity to such interference.

The Signal to Noise Ratio (SNR) loss can be minimized with larger symbol duration compared to guard time. It can’t be randomly large, however, because large symbol duration means more subcarriers with a smaller subcarriers spacing, a larger implementation complexity, and more sensitivity to phase noise and frequency offset, as well as an increased peak-to-average power ratio. Hence, a practical design choice is to make the symbol duration at least five times the guard time, which implies a 1dB SNR loss because of the guard time. After the symbol duration and guard time are fixed, the number of subcarrier follows directly as the required 3dB bandwidth divided by time subcarrier spacing, which is the inverse of the symbol duration less
than the guard time. Alternatively, the number of subcarriers may be determined by the required bit rate per subcarrier. The bit rate per subcarrier is defined by the modulation type (16-QAM), coding rate and symbol rate.

One of the main reasons to use OFDM is its availability to deal with large delay spreads with a reasonable implementation complexity. In a single carrier system, the implementation complexity is dominated by equalization, which is necessary when the delay spread is larger than about 10% of the symbol duration. OFDM does not require an equalizer instead, the complexity of an OFDM system is mainly determined by the FFT, which is used to demodulate the various subcarriers.

1.4 MATHEMATICAL DESCRIPTION OF OFDM SYSTEMS

The block diagram of the base band OFDM system model is shown in Figure 1.3.

![Figure 1.3 Block diagram of OFDM system model](image-url)
The input binary information \( b(n) \) is first grouped and the signals can be mapped according to the modulation by using signal mapper and represented by \( x(n) \). The mapped signals are then converted into parallel blocks \( x_p(n) \) for efficient high data rate communication. The data in frequency domain are transformed into time domain using an Inverse Discrete Fourier Transform (IDFT) and denoted as \( x_p(k) \). The receiver performs the reverse operation of the transmitter by using a FFT to analyze the signal in the frequency domain.

After IDFT, the Cyclic Prefix (CP) extension of guard interval is inserted between two consecutive blocks, the OFDM symbol includes the binary input information along with the cyclic prefix signal \( x_{pg}(k) \). The signals can be converted from parallel form to serial form with the help of Parallel to Serial (P/S) converter and it is represented as \( x_g(k) \). The transmitted signal \( x(n) \) passes through the frequency selective time varying channel with the effect of additive noise. At the receiver end once again serial transmission of signal is converted in parallel form and guard time is removed.

The received signal is sent to DFT block for converting time domain signals into frequency domain. The received signals are extracted using demapper and the binary information is recovered in the receiver. Figure 1.4 shows the time domain representation of an OFDM system. The \( T_{cp} \) represents the cyclic prefix duration and \( T \) is the OFDM symbol duration. The guard period length \( T_{cp} \) is equal to the one fourth of the OFDM symbol duration \( T \). The orthogonality property of OFDM signals can be verified by its spectrum. In the frequency domain, each OFDM subcarrier has a sinc function \( \sin(x)/x \) and frequency response. In the receiver part, each OFDM symbol is transmitted for a fixed time \( T_{FFT} \) with no tapering at the ends of the symbol. This symbol time corresponds to the inverse of the
subcarrier spacing of $1/T_{FFT}$ Hz. Each carrier has a peak at the center frequency and evenly spaced with a frequency gap which is equal to the carrier spacing.

![OFDM time domain representation](image)

**Figure 1.4 OFDM time domain representation**

The mathematical representation of OFDM system can be written as,

$$s(n) = \sum_{k=0}^{N-1} A_m e^{j2\pi k\delta_f} e^{j2\pi k\delta_f T_s} \quad \text{for} \quad 0 \leq k \leq T_s$$  \hspace{1cm} (1.1)

where, $N$ is number of subcarriers, $A_m$ is the $m$-ary data to be transmitted, $k$ is the discrete sample point, $T_s$ is the symbol duration and $\delta_f$ is the subcarrier frequency spacing. The $N$ different data carriers are each set apart by the frequency $\delta_f$ and the condition for orthogonality $\delta_f T_s = 1$ is considered.

The OFDM modulated block at time $n$ after cyclic prefix insertion is,

$$x_{c}(n) = T_{cp} F_N a(n)$$  \hspace{1cm} (1.2)

where, $T_{cp}$ is the cyclic prefix insertion matrix of dimension $(N+1) \times N$, $F_N$ is the $N \times N$ inverse discrete fourier transform matrix. The vector $a$ is the $N \times 1$ symbol vector and $L$ is the length of the cyclic prefix.
For designing OFDM system, the length of the information block is assumed to be \( N \), cyclic prefix length is \( L \) and the value of guard interval is zero. Then the length of the OFDM symbol becomes \( N + L \) and the symbol \( x(n) \) can be written as,

\[
x(n) = \begin{bmatrix} r_n^H \\ 0 \end{bmatrix} d_n(n) \tag{1.3}
\]

where \( F_N \) is the Fourier transform of OFDM blocks with the size \( N \times N \) DFT matrix, \( d_n(n) \) is the transmitted vector with length \( N \) and the superscript \( H \) denotes conjugate transpose of the channel matrix.

The parallel block of length \( N + L \) is converted into serial sequence and passed through the Frequency Selective time varying fading channel with additive noise. The channel impulses are considered as a finite length vector \( h \) of length \( 1 \times (L + 1) \). Then the impulse response of the channel can be written as,

\[
h = [h_1, h_2, \ldots, h_{L+1}]^T \tag{1.4}
\]

where \( h_1, h_2, h_3, \ldots, h_{L+1} \) are the channel coefficients.

The perfect synchronization between transmitter and receiver is assumed for developing the system model. The transmitted symbol \( d_n(n) \) passes through the frequency selective time varying fading channel with AWGN. The received signal from the wireless channel can be expressed as,

\[
y(n) = H F_N^H d_n(n) + w(n) \tag{1.5}
\]

where \( \hat{H} \) is the channel convolution matrix with the size of \((N + L) \times N\) and \( w(n) \) is noise term. The value of channel convolution matrix
\( \hat{H} \) can be estimated by converting the linear convolution into circular convolution matrix of size \( N \times N \). While considering Zero Padded (ZP) OFDM, the entire linear convolution of each transmitted block with channel impulse response is preserved.

The Channel matrix \( H \) with dimension \( (N + L) \times (N) \) can be written as,

\[
\hat{H} = \begin{bmatrix}
h_1 & h_2 & \ldots & h_{L+1} & 0 & \ldots & 0 \\
0 & h_1 & \ldots & h_L & h_{L+1} & \ldots & 0 \\
0 & 0 & \ldots & 0 & h_1 & \ldots & h_{L+1}
\end{bmatrix}^T
\]

(1.6)

The reverse process can happen in the receiver side.

1.5 ULTRA WIDEBAND SYSTEMS

Wireless connectivity has enabled a new mobile lifestyle filled with conveniences for mobile computing users. In recent days, WLAN and Wireless Personal Area Networks (WPAN) technologies cannot meet the needs of tomorrow’s connectivity of such a host of emerging consumer electronics devices that require high bandwidth. A new technology is required to meet the needs of high speed WPANs.

Ultra Wideband (UWB) is a fast emerging technology with unique attractive features inviting major advances in wireless communication, networking, radar, imaging and positioning system. UWB technology offers a solution for additional requirement of next generation consumer electronic devices. Using wireless connectivity, the UWB provides high data rates across multiple devices and PCs within the digital home and office, also this
technology provides the high bandwidth that multiple digital video and audio streams require throughout the home.

UWB technology is based on sending and receiving very high bandwidth carrier-less radio impulses using extremely accurate timing. It can be seen as an evolutionary replacement or complementing technology for current WLAN and especially for WPAN. UWB technology can be applicable for a very high bandwidth, short to medium range wireless connectivity at very low cost and with very low power consumption. UWB was approved by the Federal Communications Commission (FCC) in March 2002 for unlicensed operation in the 3.1-10.6 GHz band subject to modified part 15 rules. The rule limits the emitted Power Spectral Density (PSD) from UWB source measured in a 1 MHz bandwidth at the output of an isotropic transmit antenna at a reference distance is shown in Figure 1.5.

Figure 1.5 UWB Spectral mask
FCC has not mandated that any specific multiple access and modulation scheme for UWB system. However, in a single band UWB system, Impulse UWB (I UWB) and Direct Sequence UWB (DS UWB) are the two most popular modulation techniques (Wilson & Scholtz 2003). The main disadvantage of single band UWB is that building of RF and analog circuits to process the signal is challenging and usually results in high power consumption.

The maximum allowable PSD for UWB transmission of -41.3dBm/MHz corresponds to approximately 0.5mW of average transmit power when the entire 3.1-10.6 GHz band is used, effectively limiting UWB links to short ranges. The potential for exploiting such low power UWB links for high data rate wireless Personal Area Network (PAN) connectivity is up to the excess of 100 Mbps in ranges limited to 10 m particularly for in-home networking applications has led to considerable recent interest in UWB technology. The combination of the ultra-wide bandwidths and low average transmitted power presents unique challenges for UWB radio design. Such large bandwidths imply highly frequency selective multipath channels, with a large number of resolved multipath components.

UWB devices are expected to organize in typical clusters that may overlap due to close proximity of multiple clusters. These clusters are needed for effective multiple accessing techniques to manage co-existence of simultaneous users. Further, due to mobility of users, UWB enabled devices in some form of ad-hoc networking mechanism is desired to support topology changes. The UWB transmission will overlay narrowband services possibly in very close proximity. Thus UWB system design must also allow for Narrowband Interference (NBI) mitigation to preserve link integrity.
1.5.1 UWB System Considerations

Designing a UWB transceiver has several challenges, some of which are not shared with more traditional narrowband systems. This section discusses the challenges that affect the design of custom pulsed USB architecture. Each bit of information is encoded in the sign of pulse using Binary Phase Shift Keying (BPSK), which is converted up to the UWB band by multiplication with a carrier frequency. Some of the considerations addressed in this section also apply broadly to UWB system general.

The requirement for maximum total power consumption set by the 802.15.3a specification at 110 Mbps and 200 Mbps is 100 mW and 250 mW respectively. In addition, a power save mode is supported. To meet these constrains, a transceiver must either target the lowest power for all data rates or use an architecture that scales power with data rate. It is also an advantage to have an architecture that scales power consumption under optical channel condition.

UWB is an overlay technology and is susceptible to in-band interference from existing bands such as those used by 802.11a radios. The allowed power spectral density is low compared to narrow band system using overlapping bands. Therefore, interference from a UWB transmitter to a narrow band receiver arguably negligible. On the other hand the presence of narrow band interferer in the UWB band poses very stringent constraints on the filter and linearity of the UWB front-end.

For example 802.11a radios may transmit a maximum of 16 dBm/MHz in the 5.725-5.825 Unlicensed- National Information Infrastructure (U-NII) band, and may be in close proximity to a UWB receiver. In order to
cope with an 802.11a interferer at 1m and demodulate a desired UWB signal at 10 m, it is necessary to provide an interference attenuation of 65 dB if no filtering is used in the digital base band. To coexist with the larger interference, UWB system typically filters and do not use the U-NII band.

A long channel impulse response is caused by severe multipath signals which results in ISI. This limits the symbol rates for a pulsed system, or forces very complex compensation technique. UWB system has excellent multipath time resolution with large bandwidth and this information may be used to better compensate for the effects of multipath. For the purpose of simulating the system, the 802.15.3a working group proposed the use of modified Saleh-Valenzuela (S-V) channel model in the 3.1-10.6 GHz band.

1.5.2 Applications of UWB System

- Portable multimedia devices, such as camcorder, digital cameras, portable MP3 players with wireless connectivity.

- Enable high speed Wireless Universal Serial Bus (WUSB) connectivity for PCs and PC peripherals, including printers, scanners, and external storage devices.

- Next generation Bluetooth technology devices such as 3G cell phones with Internet Protocol (IP) / UPnP-based connectivity.

- Creating ad-hoc high-bit-rate wireless connectivity for PC and Mobile devices.
1.6 UWB OFDM SYSTEMS

A WPAN is also known as in-home networks which address short-range (generally within 10 m) ad-hoc connectivity among portable consumer electronic and communication devices. They are envisioned to provide high-quality real-time video and audio distribution, file exchange among storage systems and cable replacement for home entertainment systems. The ability of UWB technology to provide very high data rate for short range has made it an excellent option for physical layer of the IEEE 802.15.3a standard for wireless personal area networks. According to the FCC rules, there are two basic design approaches which are considered to obtain the single band and multiband UWB signal.

In a single band UWB, the signal is generated using very short, low duty cycle, base band electrical pulses with appropriate shape and duration. These UWB systems are also referred to as carrier free systems (Davis et al 1979). The pulse shape determines spectrum of the UWB signal and typically no additional filtering is required. In a Multiband UWB, the entire available spectrum is divided into several bands of 528 MHz each. The future UWB communications system may fulfil demands concerning the growing demands for increased bit flow such as:

(i) Ability for ad-hoc connections with Quality of Service (QoS) support for multimedia traffic

(ii) Ease of joining (or leaving) network

(iii) Advanced power management using less power compared to third-generation cellular phones
(iv) Less complexity Media Access Control (MAC) and Physical Layer (PHY) implementation optimized for short range communications

1.6.1 Applications of UWB OFDM Systems

(i) High-Data Rate Wireless Personal Area Networks (HDR-WPAN)

(ii) Wireless Ethernet Interface Link (WEIL)

(iii) Intelligent Wireless Area Network (IWAN)

(iv) Outdoor Peer-to-Peer Network (OPPN)

(v) Sensor, Positioning and Identification Network (SPIN)

1.6.2 UWB Design Issues

UWB modulation schemes can be developed based on the form of impulse modulation where the pulse shape and duration is used to control the bandwidth occupancy of the transmitted signal. The information rate can be determined using repetition frequency. The generation of an appropriate pulse shape that potentially spans 500 MHz to several GHz subject to the FCC’s spectral mask is a considerable challenge. The conflicting requirements to meet the pulse shape design are given below,

(i) Efficiently filling the FCC spectrum mask for maximum allowable transmit power.

(ii) Minimizing anticipated inter symbol interference.

(iii) Provide spectral flexibility as a method to coexist with other radio systems that may be present.
1.7 PROPAGATION CHARACTERISTICS OF MOBILE RADIO CHANNELS

In an ideal radio channel, the received signal would consist of only a single direct path signal, which would be a perfect reconstruction of the transmitted signal. However, in a real channel the signal is modified during transmission. The received signal consists of a combination of attenuated, reflected, refracted, and diffracted replicas of the transmitted signal. On top of all this, the channel adds noise to the signal and can cause a shift in the carrier frequency if either of the transmitter or receiver is moving (Doppler Effect). Understanding of these effects on the signal is important because the performance of a radio system is dependent on the radio channel characteristics.

1.7.1 Attenuation

Attenuation is the drop in the signal power when transmitting from one point to another. It can be caused by the transmission path length, obstructions in the signal path, and multipath effects. Any objects which obstruct the line of sight of the signal from the transmitter to the receiver, can cause attenuation. Shadowing of the signal can occur whenever there is an obstruction between the transmitter and receiver. It is generally caused by buildings and hills, and is the most important environmental attenuation factor. Shadowing is the most severe in heavily built up areas, due to the shadowing from buildings. However, hills can cause a large problem due to the large shadow they produce. Radio signals diffract off the boundaries of obstructions, thus preventing total shadowing of the signals behind hills and buildings. However, the amount of diffraction is dependent on the radio frequency used, with high frequencies scatter more than low frequency signals. Thus the high frequency signals require adequate signal strength in the line of sight due to more scattering. To overcome the problem of
shadowing, transmitters are usually elevated as high as possible to minimize the number of obstructions.

1.7.2 Delay Spread

The received radio signal from a transmitter consists of typically a direct signal plus signals reflected off objects such as buildings, mountains, and other structures. The reflected signals arrive at a later time than the direct signal because of the extra path length, giving rise to a slightly different arrival time of the transmitted pulse. The signal energy confined to a narrow pulse is spreading over a longer time. Delay spread is a measure of how the signal power is spread over the time between the arrival of the first and last multipath signal seen by the receiver. In a digital system, the delay spread can lead to inter-symbol interference. It is due to the delayed multipath signal overlapping the following symbols. It can cause significant errors in high bit rate systems, especially when using TDMA. As the transmitted bit rate is increased, the amount of inter-symbol interference also increases. The effect starts to become very significant when the delay spread is greater than 50% of the bit time.

1.8 FADING

Multiplicative noise or fading can be defined as the relative difference between the power contained in a given section of the transmitted and received signals. Factors that typically contribute to the fading in wireless communication systems are the transmitter and receiver antenna and analog front-end characteristics, absorption of the signal power by the propagation media, reflection, refraction, scattering and diffraction caused by obstacles in the propagation path. The receiver experiences the combined effect of all these physical factors, which vary according to the positions of the receiver and transmitter within the propagation environment. Fading in wireless
channels is characterized as a concatenation or superposition of several types of fading processes. These processes are classified using the qualitative terms such as path loss, shadowing and multipath fading also referred to as fast fading. However, these fading processes cannot in general be considered independent of each other. The Path loss and shadowing are not considered as separate processes.

Fading is classified according to the typical variation from the mean attenuation over a spatial region of given magnitude. It is classified as large-scale, medium-scale and small-scale fading. The Small-scale fading corresponds directly to multipath fading and involves signal power variations of magnitude up to 40 dB on spatial scale of a half-wavelength (for example 50 cm at 300 MHz). Averaging the total fading in the receiver over a spatial interval significantly larger than a half-wavelength provides information on the medium-scale fading or shadowing. Over spatial intervals of magnitude hundreds of meters medium-scale fading involves signal power variations up to magnitude 20 dB. Again, averaging the total fading over a spatial interval of several hundred meters provides an estimate for the large-scale fading, which may vary up to 150 dB over the considered coverage area. These denominations do not suggest a different origin or effect for the fading types, but rather show that typically different variation around the mean attenuation is observed at different spatial scales or observation window lengths.

1.8.1 Large-Scale Fading

Large-scale fading or path loss is commonly modeled for signals at a given carrier frequency as a deterministic function of the distance between the transmitter and receiver. It is affected by several parameters such as antenna gains and properties of the propagation environment between the transmitter and receiver. Main physical factors that contribute to large-scale fading are free-space loss or the dispersion of the transmitted signal power
into surrounding space and absorption of the signal power by the propagation medium. In addition to these factors, large-scale fading is defined to include the average of the shadowing and multipath fading effects. Thus the type of propagation environment must be taken into account in the total power loss. This has been done for models by approximating the parameters for the propagation loss for specific environments and transmission setups from sets of field measurements. Similar results can be obtained using analytical methods by assuming a statistical terrain description where obstacles of suitable geometry are distributed randomly in the propagation environment. It contains a detailed description of deriving functions for path loss in various land environments using such methods.

The physical mechanisms that cause the environment-specific propagation loss are the same for large-scale fading as for medium-scale fading. Deterministic large-scale fading models are useful in applications where it is sufficient to have rough estimates on the average attenuation of signal power over a large transmission area.

1.8.2 Medium-Scale Fading

The modeling of medium-scale fading can typically be categorized as statistical or site-specific. In the statistical approach, the fading is typically assumed based on empirical data to follow a lognormal distribution. The mean for this distribution can be obtained for a given carrier frequency and distance from the transmitter using expressions for large-scale fading as outlined in the previous Subsection. The standard deviation and autocorrelation of the lognormal distribution are model parameters which must be selected according to the propagation environment. Typically, the standard deviation is of the order 8 dB. If site-specific data on the environment profile and obstructions along the propagation path from the
transmitter to the receiver are available an approximation for the medium-scale fading can be calculated as,

i. Locate the positions and heights of the antennas.

ii. Construct the great circle or geodesic path between the antennas. This represents the shortest distance between the two terminals measured across the Earth's surface.

iii. Derive the terrain path profile. It is obtained from digital terrain maps, but it is of course also possible to use traditional contour profile maps.

iv. Uplift the terrain profile by representative heights for any known buildings along the path.

v. Select a value for the effective Earth radius factor appropriate to the percentage of time being designed to modify the path profile by this value. The effective Earth radius factor is a constant used to increase the effective radius of the Earth as seen by the propagating signal.

vi. Consider the paths curve slightly towards the ground. Since the atmospheric refractivity varies with pressure, temperature and water vapor pressure of the atmosphere, the correct effective Earth radius factor will vary according to location and time.

vii. Calculate the free-space path loss.

viii. If any obstructions exist within 0.6 times the first Fresnel zone of the propagation path, calculate the diffraction over these obstructions and account for the excess loss in the fading.
ix. Compute the path length which passes through trees and add the corresponding extra loss.

1.8.3 Small-Scale Fading

Small-scale fading is caused by the interference between several reflected, diffracted or scattered signals arriving at the receiver. Since the reflected propagation paths may be of different length and corresponding to different arrival times for variously faded copies of the transmitted signal at the receiver, the effect of small-scale fading is in the digital domain similar to a Finite Impulse response (FIR) filter between the transmitter and receiver. Depending on the path delay profile of the channel, the small-scale fading may vary rapidly not only in the temporal and spatial domains and also in the frequency domain. When the waves of multipath signals are out of phase, the reduction in signal strength or fade can occur. A fade is a constantly changing three-dimensional phenomenon. Fade zones tend to be small, multiple areas of space within a multipath environment that cause periodic attenuation of a received signal for users passing through them. In other words, the received signal strength will fluctuate downward, causing a momentary but periodic degradation in quality. The small-scale signal fading due to the time dispersion and frequency dispersion mechanisms in a mobile channel could be classified into four major categories depending on the nature of the transmitted signal, the channel and the mobile velocity.

The small-scale fading models can be divided into statistical and site specific approaches. Site-specific models typically apply ray-tracing methods where detailed three-dimensional models of the propagation environment are used to calculate propagation paths between the transmitter and receiver. Such techniques were originally developed for indoor environments, but have also been extended to dense urban outdoor areas.
Especially for modeling unconfined outdoor environments, ray-tracing models require large amounts of data and are computationally demanding.

There are three basic mechanisms that impact signal propagation in a mobile communication system. They are reflection, diffraction, and scattering.

- Reflection occurs when a propagating electromagnetic wave impinges on a smooth surface with very large dimensions compared to the RF signal wavelength.

- Diffraction occurs when the radio path between the transmitter and receiver is obstructed by a dense body with large dimensions compared to single user, causing secondary waves to be formed behind the obstructing body. Diffraction is a phenomenon that accounts for RF energy traveling from transmitter to receiver without a line-of-sight path between the two users. It is often termed shadowing because the diffracted field can reach the receiver even when shadowed by an impenetrable obstruction.

- Scattering occurs when a radio wave impinges on either a large rough surface or any surface whose dimensions are on the order of l or less, causing the reflected energy to spread out (scatter) in all directions. In an urban environment, typical signal obstructions that yield scattering are lampposts, street signs, and foliage.

A mobile radio channel is a fast fading channel if the channel impulse response changes rapidly within the baseband symbol duration. Fast fading channel only deals with the rate of change of the channel due to motion. This type of fading leads to signal distortion due to Doppler
spreading and this increases with increasing Doppler spread relative to bandwidth of the transmitted signal. The channel impulse response in a slow fading channel changes at a rate much slower than the transmitted baseband signal. Hence, the channel may be assumed to be static over one or several reciprocal bandwidth intervals. Viewed in frequency domain, this implies that the Doppler spread of the channel is much less than the bandwidth of the baseband signals.

In frequency Selective fading, the transmitted signal reaching the receiver through multiple propagation paths having a different relative delay and amplitude. This is called multipath propagation and it causes different parts of the transmitted signal spectrum to be attenuated differently. It is known as frequency-selective fading. In this, the channel spectral response is not flat. It has dips or fades in the response due to reflections causing cancellation of certain frequencies at the receiver. A mobile channel creates frequency selective fading on the received signal if it possesses a constant gain and linear phase response over a bandwidth that is smaller than the bandwidth of transmitted signal. It is caused by multipaths delays that approach or exceed the symbol period of the transmitted symbol. Due to frequency selective fading, the received signal contains multiple versions of the transmitted waveform, which are attenuated (faded) and delayed in time, and hence the received signal is distorted. ISI is introduced due to time dispersion of the transmitted symbols within the channel. When translated to frequency domain, it means certain frequency components in the received signal spectrum have greater gains than others.

The spectrum of the transmitted signal for frequency selective fading has a bandwidth greater than the coherence bandwidth of the channel. Frequency selective fading channel models are very difficult to model as each multipath must be modeled and the channel must be considered to be a linear
filter. Therefore, these models are typically constructed from the wideband multipath measurements. However, when analyzing mobile communication systems, statistical impulse response models such as the two-ray Rayleigh fading model are generally used. In two-ray Rayleigh fading model, the impulse response of the channel is considered to be made of two delta functions that fade independently and have sufficient time delay between them to induce frequency selective fading upon the applied signal.

Frequency non-selective fading for all the frequency components of the signal would roughly undergo the same degree of fading, and it is called as flat fading. A signal is said to undergo flat fading if the mobile channel through which it passes has a constant gain and linear phase over a bandwidth that is greater than the baseband bandwidth of the transmitted signal. In flat fading, the spectral characteristics of the transmitted signal are preserved at the receiver but the signal strength varies with time due to fluctuations in the gain of the channel caused by multipath. The impulse response of a flat fading channel can be approximated to be simply a delta function.

1.8.4 Additive Noise

Additive noise is introduced to a wireless communication system both from outside sources such as atmospheric effects, cosmic radiation, electrical devices and from internal components of the receiver hardware, which produce thermal and shot noise. Typically, additive distortion in a received signal consists of a sum of a large number of independent components, and is modeled as AWGN. In some cases, the received signal is distorted also by a channel-induced superposition of different components of the useful transmission, or by signals from other transmission systems. Such distortions are called interference and differ from additive noise in that typically some source-specific statistical characteristics of interference are
known. Thus interference is not in all cases best approximated as an additive white Gaussian process.

1.9 CHANNEL ESTIMATION IN OFDM SYSTEMS

Channel estimation has a long and rich history in single carrier communication systems. In these systems, the Channel Impulse Response (CIR) is typically modeled as an unknown time-varying FIR filter whose coefficients need to be estimated. Many of the channel estimation approaches of single carrier systems can be applied to multi-carrier systems. However, the unique properties of multi-carrier transmission bring about additional perspectives that allow the development of new approaches for channel estimation of multi-carrier systems. In OFDM based systems, the data is modulated onto the orthogonal frequency carriers. For coherent detection of the transmitted data, these sub-channel frequency responses may be estimated and removed from the frequency samples. Similarly in single carrier systems the time domain channel can be modeled as a FIR filter. The delays and coefficients can be estimated from time domain samples of received signal which can transformed to frequency domain for obtaining the Channel Frequency Response (CFR).

The radio channel can also be estimated in frequency domain using the known (or detected) data on frequency domain sub-channels. Instead of estimating FIR coefficients, one tap CFR can be estimated. Channel estimation techniques for OFDM based systems can be grouped into two main categories: blind and non-blind. The blind channel estimation methods exploit the statistical behavior of the received signals and require a large amount of data. Hence, they suffer severe performance degradation in fast fading channels. On the other hand, in the nonblind channel estimation methods, the information of previous channel estimates or some portion of the transmitted
signal are available to the receiver which will be used for the channel estimation. The non-blind channel estimation can be classified as Data aided and Decision Direct Channel Estimation (DDCE). In data aided channel estimation, a complete OFDM symbol or a portion of a symbol, which is known by the receiver, is transmitted so that the receiver can easily estimate the radio channel by demodulating the received samples. The frequency domain pilots are employed similar to those in new generation WLAN standards (802.11a and High Performance Radio Local Area Network (HIPERLAN 2)).

The estimation accuracy can be improved by increasing the pilot density. However, this introduces overhead and reduces the spectral efficiency. In the limiting case, when pilot tones are assigned to all subcarriers of a particular OFDM symbol can be obtained using block type pilot arrangement. This type of pilot arrangement is usually considered for slow channel variation and for burst type data transmission schemes, where the channel is assumed to be constant over the burst. The training symbols are then inserted at the beginning of the bursts to estimate the CFR (e.g. WLAN and Worldwide Interoperability for Microwave Access (WiMAX) systems). When channel varies between consecutive OFDM symbols either of the training symbols should be inserted regularly within OFDM data symbols with respect to the time variation of the channel (Doppler spread), or the channel should be tracked in a decision directed mode to enhance the receiver performance.

For designing the coefficients in the channel estimation, consider ‘\( N \)’ as the total number of subcarrier in the OFDM symbol which requires of \( \log_2 M \) multiplications per data symbol in the transmitter and receiver. Since ‘\( N \)’ is proportional to the maximum expected channel response length. The OFDM is considered to be a better transmission than conventional single
carrier modulation with time domain equalization for large multipath spread. In general there are three types of channel estimation methods such as Pilot, Semi-blind and Blind channel estimation. Pilot based transmission technique is based on the concept of “Train before transmit”. In this estimation method, the channel responses are estimated by sending known pilot sequences. This method is effective when the channel does not have significant time variations. For rapidly time varying channels, such approach is not efficient because training has to be performed rapidly which reduces the throughput efficiency. In blind channel estimation, the estimation is performed when the information signals are being transmitted. The major advantage of this technique is the improved bandwidth utilization for time varying channels. The semi-blind channel estimation technique falls between pilot and blind channel estimation technique.

This technique aims to estimate the channel using not only known data in the transmitted signal but also depends on the corresponding observation on the unknown data symbols. The semi-blind channel estimation is training based when only observation corresponding to the known data is used and it becomes blind channel estimation when observation is restricted to the unknown part. Semi-blind channel estimation is motivated by the fact that, there are always some known symbols that should be incorporated to improve the performance.
1.10 **OBJECTIVES**

The objectives of the present work are

- To develop the simulation model for multiband OFDM system.
- To simulate the various parameters listed in wideband channel model and analyze the behavior in MB OFDM systems.
- To develop the modified S-V channel model with Nakagami fading distribution for MB OFDM systems and analyze the performance.
- To investigate the LS based channel estimation algorithms in order to minimize the BER and improve the performance of Multiband OFDM systems.
- To investigate the MMSE based channel estimation algorithms in order to improve the better performance of higher SNR conditions in MB OFDM systems.
- To analyse the performance of MB OFDM system over EPA, EVA and ETU channel models.
- To investigate the performance of RLS based channel estimation algorithm for improving the convergence speed and performance has been analyzed with the pilot based channel estimation techniques.
- To investigate the performance of Kalman filter based channel estimation algorithm for fast time varying channels and compare results with pilot based channel estimation techniques.
- To investigate the performance of the OFDM systems for various modulation techniques and different pilot density patterns.
1.11 ORGANIZATION OF THE THESIS

The thesis has been organized as follows

Chapter 1 : Introduction

This chapter provides the overview of the wireless technologies mathematical model of OFDM at UWB systems. The various wireless channel models and different types of OFDM based channel estimation techniques are discussed. The objectives of the research work are provided in detail.

Chapter 2 : Literature review

The concepts and methods used in the existing literatures in the area of UWB systems, MB OFDM based communication and channel models, OFDM based blind and semi-blind channel estimation techniques are reviewed and provided.

Chapter 3 : System model and channel model for UWB OFDM systems.

The details of UWB OFDM system model and various blocks in the transmitter and receiver are discussed. The channel model for UWB OFDM is discussed with channel model simulations.

Chapter 4 : Channel estimation using LS and MMSE algorithms.

The Pilot based OFDM channel estimation techniques is discussed in this chapter. The performance of these techniques is analyzed over Multi-band OFDM channel.
Chapter 5 : Channel estimation using RLS algorithm.

This chapter deals about the RLS based channel estimation techniques for MB OFDM systems over UWB channels. A brief summary of adaptive channel estimation is also given. A subspace based blind channel estimation technique is discussed and its performance is analyzed.

Chapter 6 : Kalman Filter based channel estimation algorithm.

An introduction about Auto Regressive (AR) process is given. Kalman filter equations for OFDM based communication are derived and its performance is analyzed against other channel estimation techniques for various channel models.

Chapter 7 : Conclusion and Future scope of the work.

The outcome of the research are considered and concluded. The scope for future work is also discussed.