MODELING AND ANALYSIS OF OFDM WITH ADAPTIVE CLIPPING TECHNIQUE FOR PAPR REDUCTION

A THESIS SUBMITTED IN PARTIAL FULFILLMENT
OF THE REQUIREMENTS FOR THE DEGREE OF

Master of Technology

In

Telematics and Signal Processing

By

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Department of Electronics & Communication Engineering

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NATIONAL INSTITUTE OF TECHNOLOGY
ROURKELA

CERTIFICATE

This is to certify that the Thesis Report entitled ***"Modeling and Analysis of OFDM with Adaptive Clipping Technique for PAPR Reduction"*** submitted by Mr. **S Venkateswara Rao (20507031)** in partial fulfillment of the requirements for the award of Master of Technology degree in Electronics and Communication Engineering with specialization in "Telematics and Signal Processing" during session 2006-2007 at National Institute of Technology, Rourkela (Deemed University) and is an authentic work by him under my supervision and guidance.

To the best of my knowledge, the matter embodied in the thesis has not been submitted to any other university/institute for the award of any Degree or Diploma.

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CERTIFICATE OF WORK

This is to certify that the Dissertation titled "**Modeling and Analysis of OFDM with Adaptive clipping technique for PAPR reduction**" is a bonafide work carried out by **Mr. Venkateswara Rao Samineni** (Roll No. **20507031**), under our guidance during the period from August 07, 2006 to May 07, 2007. This is in partial fulfillment of the requirements for the award of the Degree of Master of Technology in Telematics and Signal Processing, and is submitted to the Department of Electronics and Communication Engineering, National Institute of Technology Rourkela, Orissa.

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Contents

Abstract

Orthogonal Frequency Division Multiplexing (OFDM) systems are better than single-carrier systems in multipath fading channel environment. OFDM systems are being adapted in many wire-line and wireless high data rate transmission systems of digital video broadcasting (DVB), IEEE 802.11, IEEE 802.16, HIPERLAN Type II, Digital Subscriber Line (DSL), and Home networking etc. There is also strong interest to use OFDM systems in 4G wireless systems.

OFDM has recently received increased attention due to its capability of supporting high data rate communication in frequency selective fading environments which cause Inter symbol Interference (ISI). In order to take advantage of the diversity provided by the multi-path fading, appropriate frequency interleaving and coding is necessary. Therefore, coding becomes an inseparable part in most OFDM applications and a considerable amount of research has focused on optimum encoder and decoder design for information transmission through OFDM over fading environments.

The OFDM systems use multiple orthogonal subcarriers. Transmission data is loaded on each subcarrier and transmitted after summation. When all subcarriers have same phase than instantaneous power of transmitted signal is very high. The peak power of OFDM scheme is higher than average power. This phenomenon is called PAPR problem. This is one of the main disadvantages of the OFDM system. Power amplifier characteristics are linear until some input value, so for higher peak powers the amplifier characteristic may be nonlinear. If peak powers are not handled in linear part, OFDM signals will get distorted. A definition of PAPR is log-scale of peak power over average power, and PAPR problem appears in all multicarrier systems. Traditionally several techniques are used for reducing PAPR instead of catering for higher peak powers in amplifiers. First, clipping technique is the most famous and simple technique. But it has BER (Bit Error Rate) performance degradation. Second, peak power avoidance precoding technique is used. It has some coding gain but it decrease data rate or increase bandwidth. Third, scrambling technique is used. With the scrambling technique, probability of peak power occurrence goes low, but hardware architecture is more complex.

In this thesis, a joint solution is proposed with RS coding, OFDM, and PAPR clipping. We implemented the hybrid method which consists of RS coding and adaptive clipping technique over an additive white Gaussian noise (AWGN) channel. Reed Solomon RS (255, 239) coding can correct 8 symbol errors from 239 symbols data. This capability can effectively compensate for the performance degradation resulted by setting PAPR threshold to 5 in case of 256 QAM, and RS (63, 47) and threshold of 4 incase of 64 QAM.

Binary data are grouped into ‘x’ bits and encoded by RS (255, 239) encoder and then modulated by 256 QAM. For 64 QAM, RS (63, 47) is used. In a typical OFDM system consists of N = 52 subcarriers and 64 point IFFT is used. The adaptive clipping technique is used with clipping ratio of 5 for 256 QAM, and clipping ratio of 4 for 64 QAM is used. The symbols are transmitted through AWGN channel. The receiver structure has reciprocal to the transmitter architecture.

The implemented hybrid technique based on RS coding and adaptive clipping technique method to compensate the performance degradation caused by clipping. From the simulation results, by using hybrid technique the clipping distortion can be removed when CR = 5 and SNR = 26.5 dB for 256 QAM, and CR = 4 and SNR = 20.5 dB for 64 QAM. The simulation results show that the hybrid method is an effective technique to mitigate the clipping distortions.

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Legends

Ψ	set of signals
Ψ_p	p^{th} element in the set of signals
$S_c(t)$	carrier signal
$A_c(t)$	amplitude of the carrier signal
$\Phi_c(t)$	phase of the carrier signal
I	inphase component amplitude
Q	quadrature component amplitude
τ	symbol duration period
ω_n	n^{th} carrier frequency
ω_0	fundamental carrier frequency
d_n	constellation symbol
$g(x)$	generator polynomial
$q(x)$	quotient polynomial
$r(x)$	received polynomial
$m(x)$	message polynomial
S_t	t^{th} syndrome
$\Lambda(x)$	error locator polynomial
α^j	j^{th} test root
$S(x)$	syndrome polynomial
$\Omega(x)$	error magnitude polynomial

Abbreviations Used

ADSL	asymmetric digital subscriber line
AP	access point
ASK	amplitude shift keying
AWGN	additive white Gaussian noise
BER	bit error rate
BPSK	binary phase shift keying
CR	clipping ratio
CTS	clear to send
CSMA-CA	carrier sense multiple access with collision avoidance
CCK	complementary code keying
DFT	discrete Fourier transform
DCF	distribution coordination function
DSP	digital signal processing
DVB	digital video broadcast
ETSI	European telecommunications standards institute
FCC	federal communication commission
FDM	frequency division multiplexing
FEC	forward error correction
FFT	fast Fourier transform
FSK	frequency shift keying
HDSL	high bit rate digital subscriber line
HDTV	high definition television
ICI	inter carrier interference
IFFT	inverse Fourier transform
ISI	inter symbol interference
ISM	industrial scientific and medical band
LLC	logical link control
MAC	medium access control
MIMO	multi input multi output
OFDM	orthogonal frequency division multiplexing
QAM	quadrature amplitude modulation
QPSK	quadrature phase shift keying

PAPR	peak to average power ratio
PSK	phase shift keying
RS	Reed Solomon
RTS	request to send
UNII	unlicensed national information infrastructure band
VHDSL	very high speed digital subscriber line
VLSI	very large scale integration
WLAN	wireless local area network

Chapter 1

INTRODUCTION

1. Introduction

1.1 Introduction

Wireless communications is one of the fastest growing segments of the communications industry. As such, it has captured the attention of the media and the imagination of the public. Wireless communication mainly categorized for media (voice and video), and data. Under media, cellular systems have experienced exponential growth over the last decade and there are currently about two billion users worldwide. Indeed, cellular phones have become a critical business tool and part of everyday life in most developed countries, and they are rapidly supplanting antiquated wire line systems in many developing countries. For data applications, wireless local area networks currently supplement or replace wired networks in many homes, businesses, and campuses. Many new applications – including wireless sensor networks, automated highways and factories, smart homes and appliances, and remote telemedicine – are emerging from research ideas to concrete systems. The explosive growth of wireless systems coupled with the proliferation of laptop and palmtop computers suggests a bright future for wireless networks, both as stand-alone systems and as part of the larger networking infrastructure.

However, many technical challenges remain in designing robust wireless networks that deliver the performance necessary to support emerging applications. The gap between current and emerging systems and the vision for future wireless applications indicates that much work remains to be done to make this vision a reality. We describe current wireless systems along with emerging systems and standards.

1.2 Motivation

Multi-carrier modulation (MCM) has recently gained fair degree of prominence among modulation schemes due to its intrinsic robustness in frequency selective fading channels. This is one of the main reason to select MCM a candidate for systems such as Digital Audio and Video Broadcasting (DAB and DVB), Digital Subscriber Lines (DSL), and Wireless local area networks (WLAN), metropolitan area networks (MAN), personal area networks (PAN), home networking, and even beyond 3G wide area networks (WAN). Orthogonal Frequency Division Multiplexing (OFDM), a multi-carrier transmission technique that is widely adopted in different communication applications. OFDM systems support high data rate transmission.

However, OFDM systems have the undesirable feature of a large peak to average power ratio (PAPR) of the transmitted signals. The transmitted signal has a non-constant envelope and exhibits peaks whose power strongly exceeds the mean power. Consequently, to prevent distortion of the OFDM signal, the transmit amplifier must operate in its linear regions. Therefore, power amplifiers with a large dynamic range are required for OFDM systems. Reducing the PAPR is pivotal to reducing the cost of OFDM systems.

Wireless systems always give several errors to the transmitted bits due to several transmission and system impediments. The techniques of power control also increase the bit error rate in end to end transmission. To address this need, communication engineers have combined technologies suitable for high rate transmission with error correction codes. Forward error correction (FEC) is one of the popular techniques. FEC or similar coding techniques allow the system to operate with lower power, allow the system to give more range even under other uncontrolled impediments of the system and the transmission.

In this work, main focus is given for the multi-carrier modulations along with PAPR and FEC error correction method. This is one of the useful solutions in building the wireless or other LAN based systems with better operating conditions and through-put.

1.3 Background Literature Survey

It is well known that Chang proposed the original OFDM principles in 1966[1], and successfully achieved a patent in January of 1970. Later on, Saltzberg analyzed the OFDM performance and observed that the crosstalk was the severe problem in this system. Although each subcarrier in the principal OFDM system overlapped with the neighborhood subcarriers, the orthogonality can still be preserved through the staggered QAM (SQAM) technique. However, the difficulty will emerge when a large number of subcarriers are required. In some of the early OFDM applications, the number of subcarriers can be chosen up to 34 allowing 34 symbols appended with redundancy of a guard time interval to eliminate intersymbol interference (ISI)[2].

However, should more subcarriers be required, the modulation, synchronization, and coherent demodulation would induce a very complicated OFDM scheme requiring additional hardware cost. In 1971, Weinstein and Ebert proposed a modified OFDM system [3] in which the discrete Fourier Transform (DFT) was applied to generate the orthogonal subcarriers waveforms. Their scheme reduced the implementation complexity significantly, by making use of the inverse DFT (IDFT) modules and the digital-to-analog converters. In their proposed model, baseband signals were modulated by the IDFT in the transmitter and then demodulated by DFT in the receiver. Therefore, all the subcarriers were overlapped with others in the frequency domain, while the DFT modulation still assures their orthogonality.

Cyclic prefix (CP) or cyclic extension was first introduced by Peled and Ruiz in 1980[4] for OFDM systems. In their scheme, conventional null guard interval is substituted by cyclic extension for fully-loaded OFDM modulation. As a result, the orthogonality among the subcarriers was guaranteed. With the trade-off of the transmitting energy efficiency, this new scheme can result in a phenomenal ICI (Inter Carrier Interference) reduction. Hence it has been adopted by the current IEEE standards.

In 1980, Hirosaki introduced an equalization algorithm to suppress both inter symbol interference (ISI) and ICI[5], which may have resulted from a channel distortion, synchronization error, or phase error. In the meantime, Hirosaki also applied QAM modulation, pilot tone, and trellis coding techniques in his high-speed OFDM system, which operated in voice-band spectrum.

In 1985, Cimini introduced a pilot-based method to reduce the interference emanating from the multipath and co-channels [6]. In 1989, Kalet suggested a subcarrier-selective allocating scheme [7]. He allocated more data through transmission of “good” subcarriers near the center of the transmission frequency band; these subcarriers will suffer less channel distortion. In the 1990s, OFDM systems have been exploited for high data rate communications. In the IEEE 802.11 standard, the carrier frequency can go up as high as 2.4 GHz or 5 GHz. Researchers tend to pursue OFDM operating at even much higher frequencies nowadays. For example, the IEEE 802.16 standard proposes yet higher carrier frequencies ranging from 10 GHz to 60 GHz.

Coded OFDM allows the exploitation of frequency diversity and it provides a greater immunity to impulse noise and fast fades. One of the major drawbacks of OFDM is its high PAPR. This limits the transmission range and requires that the transmit amplifiers have a large input power back-off. In many low-cost operations, the disadvantages outweigh all the benefits of OFDM

The idea of jointly solving the PAPR and the error correcting code design problem is first addressed using block codes in [8]. Similarly, Davis *et al.* [9] obtain a class of low PAPR codes with a large minimum distance based on cosets of Reed-Muller codes. In this thesis the PAPR problem is tackled in yet another way [10] [11]. The hybrid method which consists of RS coding and adaptive clipping technique.

1.4 Thesis Contribution

This section outlines some of major contributions of the study presented in this thesis. The thesis mainly combined three techniques of OFDM, PAPR, and FEC. This thesis presents adaptive clipping technique for reducing PAPR on OFDM system. The OFDM signal is corrupted with AWGN channel. It is seen that the advantage provided by adding RS coding to OFDM system for PAPR reduction problem can give better performance in terms of BER. The blocks of FEC, PAPR, and FEC are mapped in the overall system. In the process of evolution Bit Error Rate (BER) has been used as performance measure. To create enough motivation of the topic, various WLAN schemes and modulations are presented.

1.5 Thesis Outline

Following the introduction, the remaining part of the thesis is organized as under;

Chapter 2 discusses introduction to IEEE Wireless LAN standards. Chapter 3 discusses different modulation schemes used in OFDM. Chapter 4 discusses the fundamental concepts of OFDM and principles behind OFDM. Chapter 5 discusses the Reed Solomon (RS) coding techniques implemented with OFDM. Chapter 6 discusses the PAPR Schemes OFDM and its implementation. Chapter 7 discusses and analyzes the results obtained with the combination of OFDM, PAPR, and FEC. Chapter 8 summarizes the work undertaken in this thesis and points to possible directions for future work.

Chapter 2

IEEE 802.11a, 802.11b, 802.11g WLANs

2. IEEE 802.11a, 802.11b, 802.11g WLANs

2.1 Introduction

The 802.11 architecture features an LLC (logical link control) specification for identifying the service address point in the source and destination computers and informing the device's operating system of those sending and receiving applications. Where there is no contention with other users, a point coordination function is employed. However, WLAN access tends to be a contention prone service. As such, the upper layer LLC communicates directly with the lower distribution coordination function (DCF) and the specific MAC function. These MAC implementation include the original 802.11 modes and the newer 802.11 *a*, *b*, and *g* modes.

IEEE 802.11 Protocol Architecture

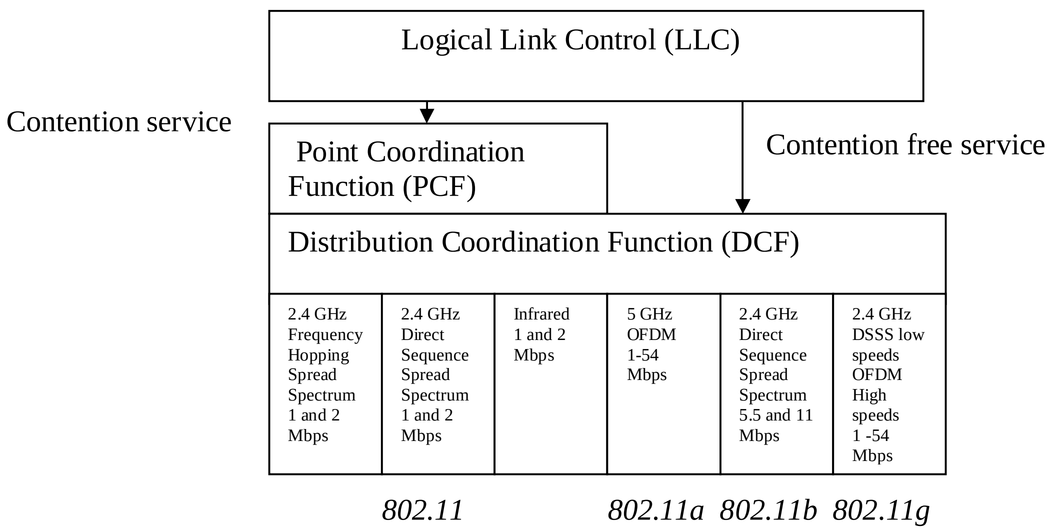


Figure 2.1: The IEEE 802.11Protocol Architecture Layer Model

Current implementation of the 802.11 protocol began with version '*b*' operating at the 2.4 GHz frequency, followed by version '*a*', which operates at the 5 GHz frequency as shown in *Figure 2.1*.

This Chapter discusses IEEE Wireless LAN architecture. Following the introduction this chapter is organized as under Section 2.2 describes The IEEE 802.11b Standard, Section 2.3 describes The IEEE 802.11a Standard, Section 2.4 describes The IEEE 802.11g Standard, Section 2.5 describes Comparison of these standards and Section 2.6 is presented with chapter conclusions.

2.2 The IEEE 802.11b Wireless Network Standard

The 802.11b specification provides for data transfer rates of both 5.5 Mbps and 11 Mbps.

To place user data frames, the 802.11b specification provides three different types of signal modulation methods depending upon the data rate used.

1. *Binary Phase Shift Keying (BPSK)* BPSK uses one phase to represent binary one (1) and another phase to represent a binary zero (0), so that each phase shift presents a change of a string of bits from one to zeros. No change of phase represents more of a string of the same kind of bits (zero or one). This representation approach is used to transmit data at 1 Mbps.
2. *Qudrature Phase Shift Keying (QPSK)* With QPSK, the carrier undergoes four changes in phase and can thus represent two binary bits of data. This representation approach is used to transmit data at 2 Mbps.
3. *Complementary Code Keying (CCK)* CCK uses a complex set of functions known as complementary codes to send additional data. One of the advantages of CCK over similar modulation techniques is that it suffers less interference from Multi-path distortion. CCK is used to transmit data at 5.5 Mbps and 11 Mbps.

For example, using the CCK modulation scheme, each byte (8 bits) of the user message is multiplied by a 64 bit spreading code that is uniquely assigned to that user. The result of this process is a signal that will take up a much larger portion of the frequency spectrum around the assigned 2.4 GHz band for '*b*' variation of the 802.11 protocol. This spreading process by applying a 64 bit code is shown in *Figure 2.2.*

Note on spreading code: Actually it is just like Direct Sequence Spread Spectrum (DSSS). Each user will have one different spreading code, that spreading code is XORed with the data bits to obtain the modulated signal. Each user will have different spreading code.

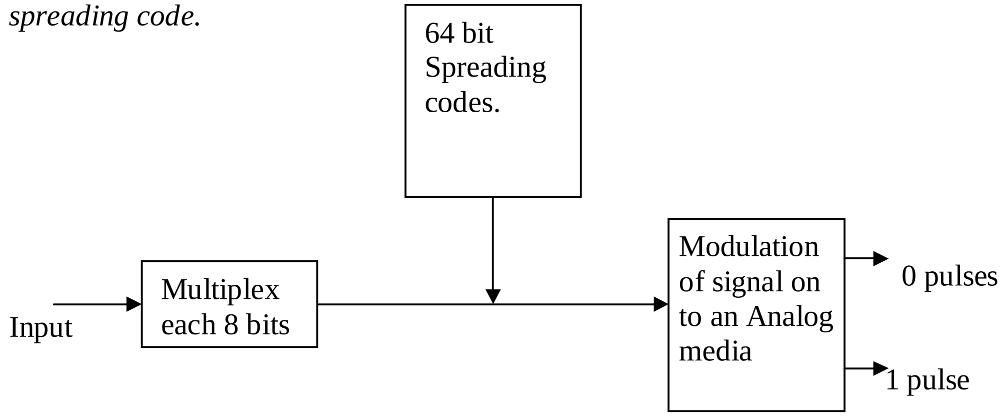


Figure 2.2: The Complementary Code Keying Spreading Process

By applying slightly different spreading codes, one individual's signal can be distinguished from anther while sharing a common frequency band. This spreading difference is shown in *Figure 2.3.*

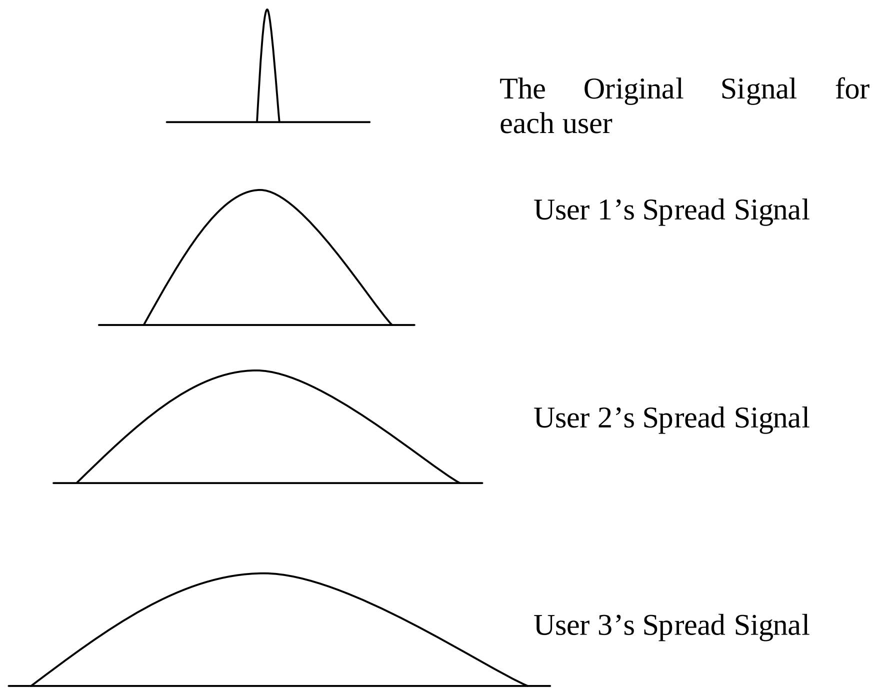


Figure 2.3: Multiple Users with Differentially Spread Signal

With DSSS, the modulated signal is dispread (spread) across one of these 22 MHz wide (0.022 GHz) channels by applying one of 64 possible codes to the signal. It is through this process of signal spreading that each signal is rendered unreadable (although transmitted over a known sub channel of the 2.4 GHz frequency range) by anyone who does not have access to the same code, to reduce the spread code to its original form. The following *Figure 2.4* displays the three sub channels of the 2.4 GHz frequency band, which fit between 2.4 GHz and 2.4835 GHz.

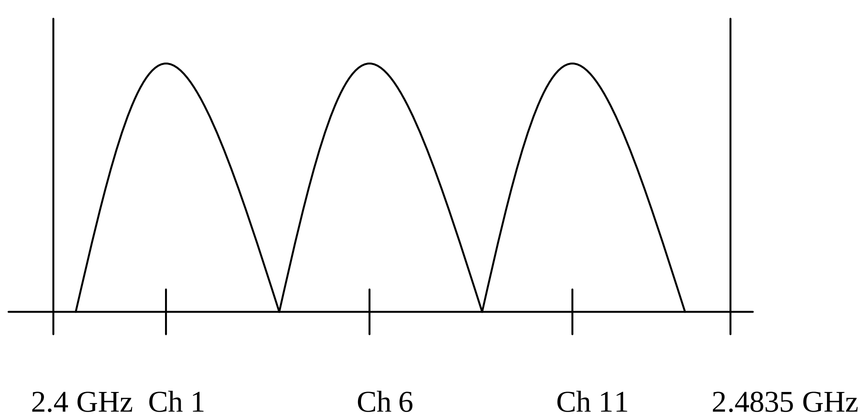


Figure 2.4: Three sub channels of the 802.11b WLAN

Regardless of the data rate (1, 2, 5.5, or 11 Mbps), the channel bandwidth is about 22 MHz (0.022 GHz) for DSSS systems. Thus, upto three non overlapping channels can be accommodated.

2.3 The IEEE 802.11a Wireless Networking

The physical layer of the 802.11a specification provides data transfer rates of 6, 9, 12, 18, 24, 36, 48, and 54 Mbps. It employs frequencies around the 5 GHz frequency band, actually frequencies between 5 GHz and 6 GHz. 802.11a uses a 300 MHz of bandwidth in the 5 GHz Unlicensed National Information Infrastructure (U-NII) band. The FCC has divided the total 300 MHz into three distinct 100 MHz domains, low, medium, high, each with different legal maximum power output as shown in *Figure 2.5*.

802.11a uses OFDM and sub carriers modulated using BPSK, QPSK, 16 QAM, and 64 QAM, 256 QAM.

1. The low band of 802.11a operates from 5.15 – 5.25 GHz and has a maximum of 2.5 mW (milliWatts).
2. The middle band is located from 5.25–5.35 GHz, with a maximum of 12.5 mW
3. The high band uses 5.725–5.825 GHz, with a maximum of 50 mW

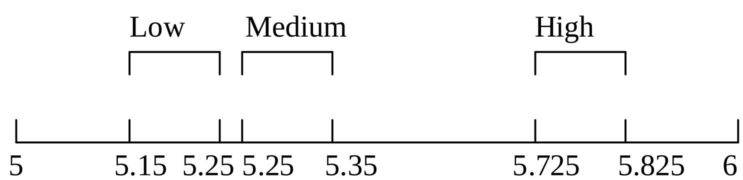


Figure 2.5: The Low, Medium, and High bands used by 802.11a

Due to the higher power output required with using this high frequency range, devices transmitting in the high band tend to be building to building devices, where as the lower power low and middle bands are used for providing access within a building.

Each of these three bands (low, medium, and high) will be divided into eight sub bands. Each of these eight sub bands will be again divided into 52 sub channels. One requirement specific to the low band is that is that all devices must use integrated antennas.

Different regions of the world have allocated different amounts of spectrum, so geographic location will determine how much of the 5 GHz band is available. In the United States, the FCC has allocated all the three bands for unlicensed transmissions.

In Europe, however, only the low and middle bands of the 5 GHz band are available for public use. Though 802.11a is not yet certifiable in Europe, efforts are currently underway between IEEE and European Telecommunications Standards Institute (ETSI) to rectify this.

In Japan, only low band may be used. This results in more contention of signal, but nevertheless, allows for very high performance.

The frequency range currently employed for most enterprise class unlicensed transmissions, including 802.11b, is the 2.4 GHz ISM band. This highly populated band offers only 83 MHz of spectrum for all wireless traffic, including cordless phones, building to building transmissions, and microwave ovens. By way of comparison, the 300 MHz offered in the U-NII band represents nearly four times amount of spectrum, all more impressive, given that there is limited wireless traffic in the band today.

2.3.1 Orthogonal Frequency Division Multiplexing Scheme

802.11a uses OFDM, an encoding scheme with distinct advantages in the number of channels that are available for use and high data rate that can be employed. Channel availability is significant because the greater the number of available independent channels, the more the scalable the wireless network becomes. The high data rate is accomplished via the combination of many lower speed sub carriers to create one high speed channel.

2.3.2 Eight Non-Overlapping 20 MHz Channels

As per IEEE 802.11a Standard the overall bands are divided as shown. Each low, medium, high band is again divided into 8 sub bands; each of eight sub bands will consist of 52 sub channels as shown in Figure 2.7.

802.11a uses OFDM to define a total eight non overlapping 20 MHz channels across the two lower bands. Each of these eight channels is divided into 52 sub carriers, each of which is approximately 300 KHz wide. The following *Figure 2.6* demonstrates this division of bands into sub bands and those sub bands into sub channels.

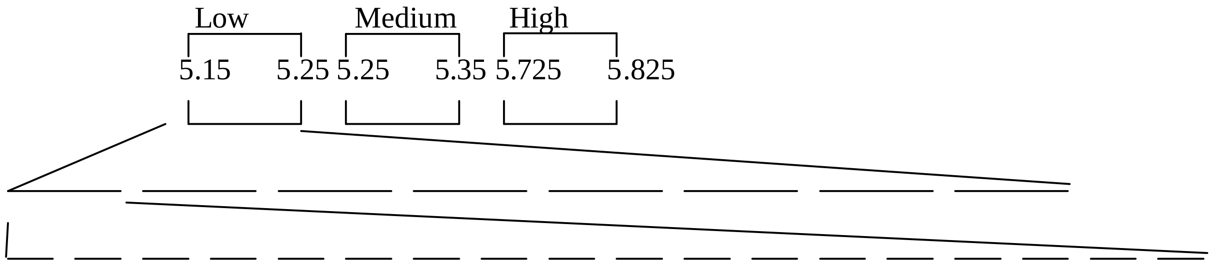
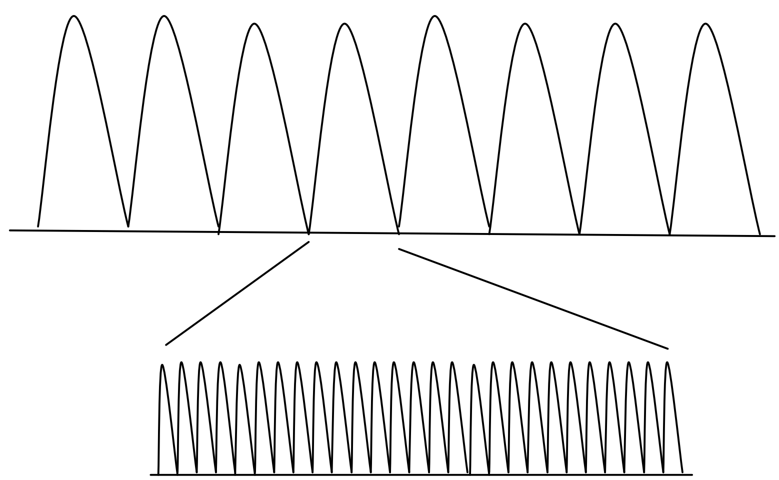


Figure 2.6: 802.11a Low, Medium and High Frequencies sub channels (some problem here)

Eight 20 MHz Channels in Low band 52 300 MHz sub channels in each of the 8 channels in each frequency band

Figure 2.7 presents in more detail how each of the eight 20 MHz channels is composed of 52 sub channels, each of which is 300 KHz wide.

8 Non Overlapping Channels



52 300 MHz sub channels in each of the 8 channels in each frequency band

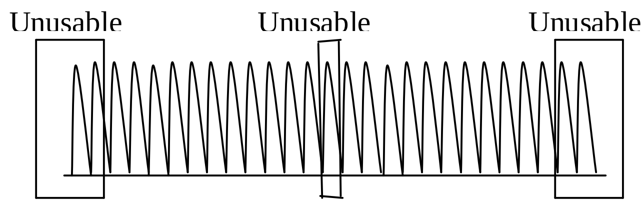
Figure 2.7: For 802.11a, the eight channels are each composed of 52 sub channels

By comparison, 802.11b, as described earlier, is more limited in that it provides only three non overlapping channels (rather than eight) and does not create sub channels with in these three sub channels. As shown in *Figure 2.4,* the three channels used by 802.11b within the 2.4 GHz to 2.4835 GHz frequency are each 0.022 GHz (or 22 MHz) wide compared to the eight 20 MHz wide 802.11a channels.

A wide channel can transport more information per transmission than a narrow one. In fact, 802.11a uses channels that are 20 MHz wide, but the protocol also breaks each of eight 20

MHz channels into 52 sub carriers. These sub carriers carry transmitted information in parallel. Information is sent and received simultaneously. The receiving device processes these individual signals, each one representing a fraction of the total data of the actual signal. With this many sub carriers comprising each channel, a tremendous amount of information can be sent at once.

Although there are 64 total sub channels available in each of the 20 MHz channels of 802.11a, 11 of the sub channels on the sides and one at the center are not usable. Furthermore, an additional four are used for pilot signals. This leaves 52 usable sub channels (out of the existing 64 sub channels) for every 20 MHz channel, as shown in *Figure 2.8*.



52 usable sub channels 300 KHz wide in each of the 8 20 MHz Channels
Figure 2.8: The 52 of 64 usable sub channels of 802.11a

2.3.3 Forward Error Correction

With such a large amount of information per transmission, it becomes crucially important to guard against data loss. FEC was added to the 802.11a specification for this purpose. FEC does not exist in 802.11b.

FEC dictates that a secondary copy is sent along with the primary information. If part of the primary information is lost, a backup copy exists to help the receiving device recover the lost data. This eliminates the need to retransmit even if a large amount of data is lost.

Because of its high speed, 802.11a can accommodate the additional overhead with negligible impact on performance. Multipath interference occurs when reflected signals cancel each other out. 802.11a uses a slower symbol rate to minimize the multipath interference.

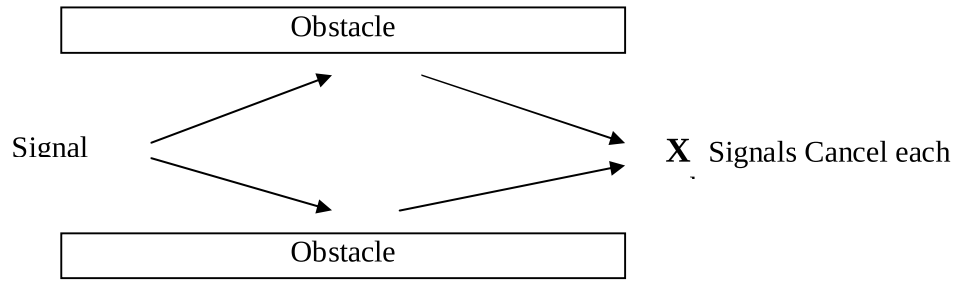


Figure 2.9: The Effect of Multipath Signaling with Collisions

2.3.4 Multipath Reflection

Another threat to transmission integrity is Multipath reflection, also called delay spread. When a radio signal leaves the sending antenna, it is radiates outward, spreading as it travels. If the signal reflects off a flat surface, the original signal and the reflected signal may reach the receiving antenna simultaneously. Depending on how the signals overlap, they can either augment or cancel each other out.

A baseband processor, or equalizer, unravels the divergent signals. However, if the delay is long enough, the delayed signal spreads into the next transmission. OFDM specifies a slower symbol rate to reduce the chance a signal will encroach on the following signal, minimizing multipath interference.

2.3.5 Data Rates and Ranges

Devices using 80.2.11a are required to support speeds of 6, 12, and 24 Mbps. Optional speeds goes upto 54 Mbps, but will also typically include 48, 36, 18, and 9 Mbps. These differences are the result of the implementation of different modulation techniques and FEC levels.

To achieve 54 Mbps, a mechanism called 64 QAM is used to peak the maximum amount of information possible on each sub carrier. Similar to operation of 802.11b, as on 802.11a client device travels farther from Access Point (AP), the connection will remain intact but the speed decreases. As *Figure 2.10* illustrates, 802.11a can have a significantly higher signaling rate than 802.11b at most ranges.

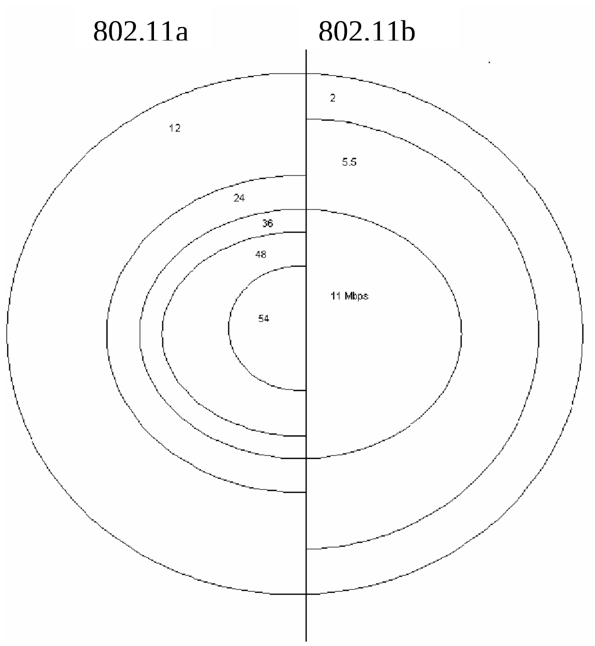


Figure 2.10: A Comparison of transmission distances and speeds for 802.11a and 802.11b

2.3.6 The MAC Layer - 802.11a

802.11a uses the same MAC Layer technology as 802.11b. Carrier Sense Multiple Access with Collision Avoidance (CSMA-CA). CSMA-CA is a basic protocol used to avoid the signals colliding and canceling each other out. Signals using CSMA/CA request authorization to transmit for a specific amount of time prior to sending information.

2.3.7 Four way Reliability Method Employed by 802.11a

The sending device broadcasts a request to send (RTS) frame with information regarding the length of its signal. If the receiving device permits, it broadcasts a clear to send (CTS) frame. Once the CTS frame goes out, the sending machine transmits its information. Any other sending devices in the area that hear the CTS realize another device will be transmitting and allow that signal to go out uncontested.

2.3.8 Compatibility and Inter Compatibility Features of 802.11a with 802.11b

While 802.11a and 802.11b share the same MAC layer technology, there are significant differences at the physical layer. 802.11b, using the ISM band, transmits in the 2.4 GHz range, while 802.11a, using the U-NII band, transmits in the 5 GHz range.

Because the 802.11a and 802.11b signals travel in different frequency bands, one significant benefit to the user who happens to implement both within a building is that they will not interface with each other. A related consequence, however, is that the two technologies are not compatible, and people who employ 802.11a equipment can connect to users of 802.11b equipment only through a wired backbone network that interconnects the two systems. There are various strategies for migrating from 802.11b to 802.11a. Moreover, there are even hybrid combination '*a*' and '*b*' APs and PC wireless cards for using both on the same network concurrently. However, they increase the cost to the enterprise.

802.11a is not the next generation of enterprise class WLAN technology, although it has many advantages over other current options. At speeds of 54 Mbps, it matches the 802.11g protocol, and both 'a' version and '*g*' version are faster than the '*b*' version. 802.11a and 802.11b have similar range, but 'a' version provides higher speed than the '*b*' version at each spot throughput the entire coverage area.

Another advantage of the 802.11a that the 5 GHz band in which it operates is not highly populated, so there is less congestion to cause interference or signal contention. And the eight non overlapping channels of 802.11a allow for a highly scalable and flexible installation.

2.3.9 Transfer Rates and Distances of the Versions of 802.11

The maximum distance that 802.11 networks can transmit from the user station to an AP is 100 meters (or 330 feet). However, the speed achieved at this distance is only 2 Mbps. For most data applications, 2 Mbps transmission is satisfactory. This transfer speed is equivalent to that offered by the telephone company with its T1 private line and interoffice truncking system. Also, distances less than 100 feet can achieve transfer speed upto 11 Mbps or even 54 Mbps (see *Figure 2.11*).

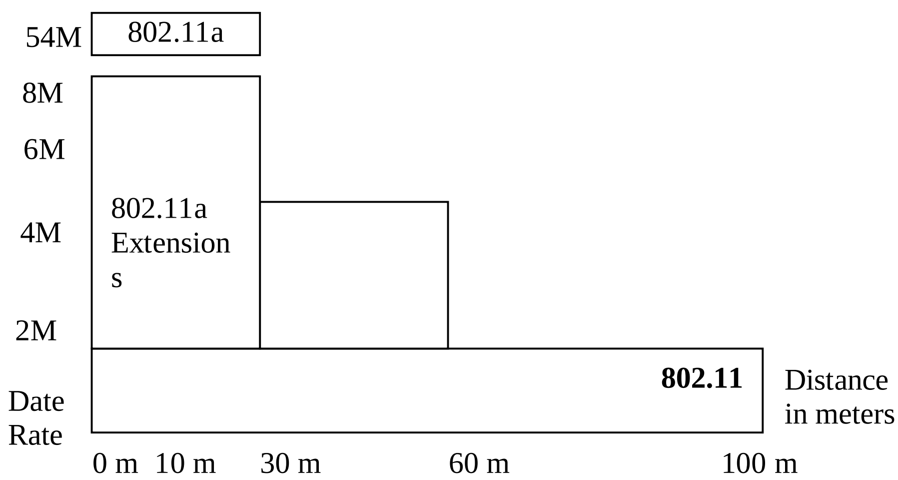


Figure 2.11: Various Speeds and Distances Achievable by 802.11 Networks

802.11a provides four basic modulation schemes BPSK, QPSK, 16 QAM, and 64 QAM with each having two channel transfer rates. These transfer rates are presented in *Table 2.1*.

Modulation Scheme	**Channel Transfer**	**Total Transfer**
BPSK	125 Kbps	6 Mbps
BPSK	187.5	9
QPSK	250	12
QPSK	375	18
16 QAM	500	24
16 QAM	750	36
64 QAM	1000	48
64 QAM	1125	54

Table 2.1: The transfer rates for 802.11a wireless networks

802.11b, on the other hand, employs three modulation schemes CCK, QPSK, and BPSK (see *Table 2.2*). CCK is used to support the higher 5.5 Mbps and 11 Mbps transfer rates.

Modulation Scheme	**Transfer Rate**
CCK	11 Mbps
CCK	5.5 Mbps
QPSK	2 Mbps
BPSK	1 Mbps

Table 2.2: The 802.11b Transfer Rates

2.4 IEEE 802.11g Wireless Network Standard

The '*g*' variation of the protocol employs the 2.4 GHz frequency band for transmission (just as the '*b*' variation does). 802.11g provides for transfer in speeds upto 54 Mbps (five times faster). To this end, 802.11g uses a variation on the DSSS spreading technique that is also used by 802.11b at lower speeds upto 11 Mbps. It switches to OFDM for the higher data rates from 12 Mbps to 54 Mbps. This array of 802.11g sub channel and data transfer rates under the two spreading techniques is shown in *Table 2.3*

Modulation Scheme	**Sub Channel Rate**	**Transfer Rate**
CCK	No Sub Channel	11 Mbps
CCK	No Sub Channel	5.5 Mbps
QPSK	No Sub Channel	2 Mbps
BPSK	No Sub Channel	1 Mbps

Table 2.3: The 802.11 g sub channel and data transfer rates using DSSS and OFDM Spreading Techniques

2.5 Comparing 802.11a, 802.11b, and 802.11g Technologies

Table 2.4 presents the broad array of similarities and differences among the three current 802.11 WLAN approaches. A critical difference is that 'a' version operates at the less congested bandwidth of 5 GHz, while the other two versions (*b* and *g*) operate at the lower 2.4 GHz bandwidth. On the other hand, the '*b'* and '*g'* versions are interoperable and can operate at distances upto 124 meters (although normally the expected range is limited to 100 meters or 330 feet).

	802.11a	**802.11b**	**802.11g**
Frequency Band	5 GHz	2.4 GHz	2.4 GHz
Spread Type	OFDM	DSSS	OFDM – Fast DSSS – Slow
Transfer Speeds	54, 48, 36, 24, 12, 9,6	11 Mbps	OFDM 54, 48, 36, 24,18 DSSS 11, 5.5, 2,1
Dist Range	54 Mbps - 13 meters 6 Mbps – 50 meters	11 Mbps – 40 meters 1 Mbps – 124 meters	54 Mbps – 27 meters 1 Mbps – 124 meters

Table 2.4: Comparison of 802.11 a, b, and g versions

From a capacity standpoint, having more channels to carry traffic is a distinct advantage. 802.11a with its 12 or 24 channels (versus 802.11b and 802.11g with only 3 channels available) provides a significantly larger capacity for carrying traffic. As an example, using a common transfer rate in the lower range of 6 Mbps or 5.5 Mbps for the '*b'* version, 802.11a can carry 72 Mbps (6 Mbps times 12 channels) while the '*b'* and '*g*' versions can only carry about 18 Mbps. This advantage of 12 carrying channels can carry 608 Mbps while the *g* version with only 3 channels can carry only 162 Mbps. *Table 2.5* presents a comparison of available capacity rates for the different protocols.

	802.11a	**802.11b**	**802.11g**
Throughput **At Maximum**	6 Mbps 54 Mbps	5.5 Mbps 11 Mbps	6 Mbps 54 Mbps
# of Channels Mixed Mode	12	3	3
Capacity – 6 Mbps	72 Mbps	16.5 Mbps	18 Mbps
At Max Capacity	648 Mbps	66 Mbps	162 Mbps

Table 2.5: Capacity Comparisons between the three (a, b, and g) 802.11 variations

A Comparison of the Transmission Distances for 802.11 *a*, *b*, and *g*

802.11g transmits at twice the distance of 802.11a for the faster transmission speeds (12 to 54 Mbps) employed by both protocols, while transmitting over the same distance that 802.11b supports when both versions are transmitting at the slower speeds of 1 to 11 Mbps.

Looking at the speed of 802.11a versus its competitor (802.11g) at equivalent distances, we can see that the *g* variation provides a better speed (sometimes two to three times the speed, as shown in *Table 2.6*).

Distance	802.11a	802.11g
1.8288 meters	16.4 Mbps	21 Mbps
10.9726 meters	7 Mbps	21 Mbps
25 meters	2.5 Mbps	14 Mbps

Table 2.6: Comparison of 802.11a versus 802.11b for transmissions speed and distance

A more complete view is to compare all three protocols together (the slower '*b'* version along with faster '*a'* and '*g'* versions), although this creates a fairly complex table. Looking at *Table 2.7,* at equivalent speeds 802.11g transfers over twice the distance of 802.11a. However, at low speeds (11 Mbps down to 1 Mbps), the '*b'* and '*g'* versions transfer at equivalent distances. And version *a* does not provide any transfer at all below 6 Mbps.

Transfer Speed	802.11a Access Distance	802.11b Access Distance	802.11g Access Distance
54 Mbps	13 meters		27 meters
48	15		29
36	26		30
24	26		42
18	33		54
12	39		64
11		48 meters	48
9	45		76
6	50		91
5.5		67	67
2		82	82
1		124	124

Table 2.7: Overall comparison of the 802.11 variations for speed and distance

2.6 IEEE 802.11n Standard

The current 802.11a/b/g WLAN standards offer the convenience of wireless connections with adequate performance for most of today's wireless networking applications, However, as next-generation wireless applications emerge, higher WLAN data throughput will be required.

In response to this need, both IEEE TGn and the Wi-Fi Alliance have set expectations for the next generation of WLAN standard 802.11n. The operating frequency of 802.11n is 2.4 GHz and/or 5 GHz. It will use Multiple Input Multiple Output (MIMO) OFDM modulation technique. The real data throughput will reach a theoretical 270 Mbit/s, and is up to 20 times faster than 802.11b, and up to 3 times faster than 802.11a and up to 4 times faster than 802.11g. In addition, 802.11n will support all major platforms, including consumer electronics, personal computing, and handheld platforms, and will be usable throughout all major environments, including enterprise, home, and public hotspots.

802.11n builds upon previous 802.11 standards by adding MIMO. MIMO uses multiple transmitter and receiver antennas to allow for increased data throughput via spatial multiplexing and increased range by exploiting the spatial diversity. This is beyond the scope of the thesis.

2.7 Chapter Conclusion

Given the data, it is rather obvious that the 802.11g specification is the current preferred choice, providing high transfer speed and decent transmission distances. However, there has been some resurgence in interest in version '*a*'. Its carrying capacity and use of the 5 GHz frequency range with low noise and competition is certainly an advantage. On the other hand, the '*b*' and '*g*' versions are interoperable since they both use the 2.4 GHz frequency and the same modulation and encoding processes at the lower transmission speed. However, they are plagued with more interface, noise, and competition from other signaling systems. So there will be some interest in the future in achieving version '*g*'transmission distances at the 5 GHz frequency.

Chapter 3

MODULATION SCHEMES

3. Modulation Schemes

3.1 Introduction

In communication, modulation is the process of varying a periodic waveform, in order to use that signal to convey a message. Normally a high frequency waveform is used as a carrier signal. The three key parameters of a sine wave are frequency, amplitude, and phase, all of which can be modified in accordance with a low frequency information signal to obtain a modulated signal.

The aim of digital modulation is to transfer a digital bit stream over an analog band pass channel or a radio frequency band. The changes in the carrier signal are chosen from a finite number of alternative symbols. Amplitude shift keying (ASK), frequency shift keying (FSK), phase shift keying (PSK), QAM are the most fundamental digital modulation schemes. In PSK and QAM, the modulation alphabet is conveniently represented on a constellation diagram, showing the amplitude of inphase (I) component on x-axis and the amplitude of Quadrature (Q) component on y-axis, for each symbol. ‘I’ and ‘Q’ signals can be combined into a complex valued signal called the equivalent baseband signal. This is a representation of the value modulated physical signal.

This chapter discusses the different modulation schemes used in OFDM. Following the introduction; Section 3.2 describes the Phase Shift Keying modulation, Section 3.3 describes Quadrature Amplitude Modulation, and Section 3.4 describes the conclusions and simulation results.

3.2 Phase Shift Keying (PSK)

PSK relies on carrier changing between distinct phases of the signal to define the status of information being transmitted. PSK default is considered for binary (two) levels of phase modulations. PSK is considered a very efficient process of data delivery because of low bit error rates in the delivery. A number of variations of PSK are used in wireless networking systems, among which are BPSK, QPSK. Most of current wireless systems employ some form of PSK.

3.2.1 Binary Phase Shift Keying (BPSK)

In BPSK, the phase of the carrier signal is switched depending upon whether the source data from the user is a zero or a one. As an example, in *Figure 3.1 & 3.2*, the signal is changing phase in the transition bits from 0 to 1 and from 1 to 0. There is no phase change for the continuity of the same bit pattern.

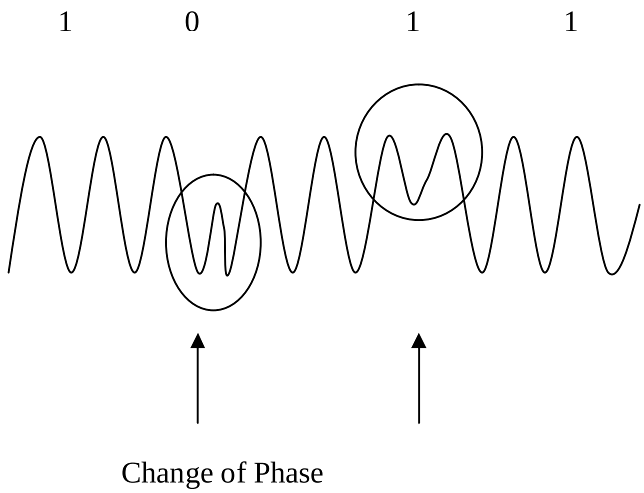


Figure 3.1: Change of phase with symbol change

Following the standard modulation process, the user's source data is modulated against the carrier signal centered on the frequency band.

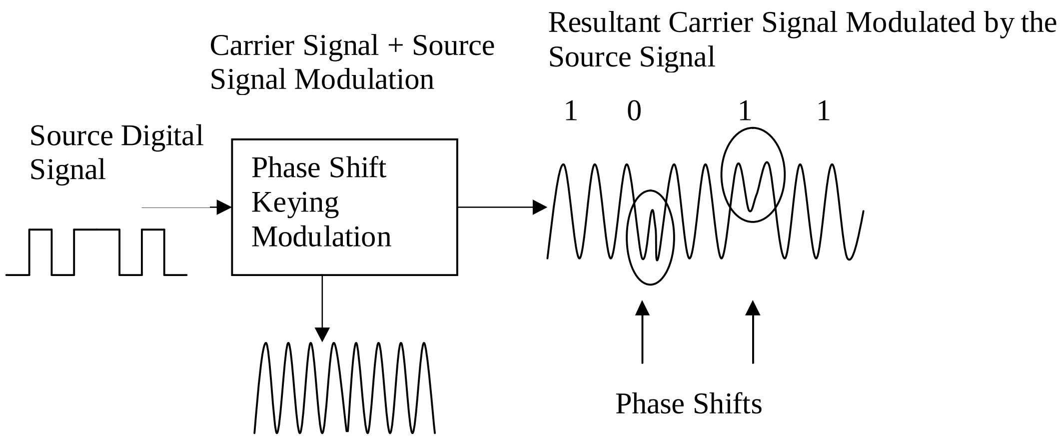


Figure 3.2: Phase Shift Modulation Process

With BPSK process, modulated wave shifts between two phase phases, which are 180 degrees apart. The zero and the one are represented by the 0 degree and the 180 degree phases. A phase diagram showing those two phases is shown in *Figure 3.3*

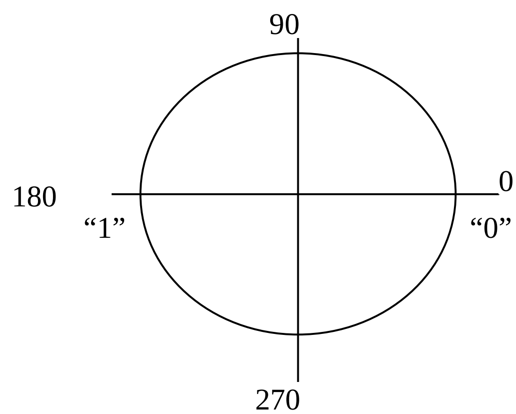


Figure 3.3: A Phase diagram showing the 0 and 180 degree phases

3.2.2 Quadrature Phase Shift Keying (QPSK)

In QPSK, each symbol is of 2-bits. This creates 4-phase combinations for modulation. BPSK is mainly used for robust signaling and low bit rate operation. QPSK provides double the bit rate than BPSK and makes use of approximately same analog signal bandwidth.

The modulated wave shifts between four phases, which are 90 degrees apart, to create phases of "zero - zero", "zero - one", "one - zero", "one - one". The process of shifting between the four phases is called 4 PSK. *Figure 3.5* portrays the phase diagram showing the shifting between 0, 90, 180, and 270 degrees based upon the codes zero – zero, zero – one, one – zero, one – one.

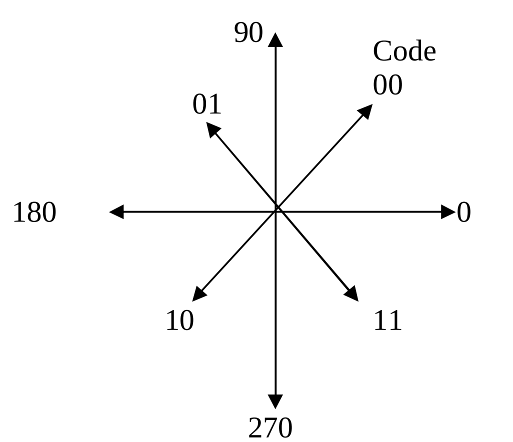


Figure 3.4: Quadrature Phase Shift Keying

3.3 Quadrature Amplitude Modulation (QAM)

For higher data rates, PSK has limitations. QAM provides the higher throughput rate required for data transfers by combining ASK and PSK. Two different signals are sent simultaneously on the same carrier frequency. The result of this combination provides two variable (amplitude and phase of the signal) to assign binary values. As the number of states are increasing, greater throughput is achieved. The number of states used in QAM ranges from 8 to 4096 in practical systems making data throughput to 100 Mbps rate in WLAN and very high speed digital subscriber line (VHDSL) systems.

The diagram in *Figure 3.5 & 3.6* represents the four quadrants of possible phase change and four groups of symbols or possible data combinations that can be delivered with the varying amplitude and phase shifts of the signal. As the complexity of QAM increases, the possibility of data loss also increases.

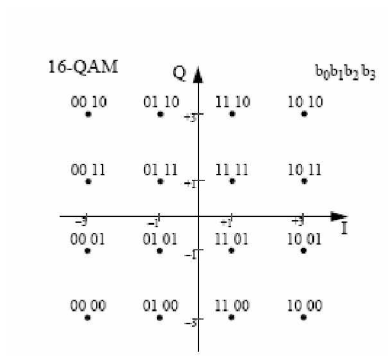


Figure 3.5: Constellation Diagrams for 16 QAM

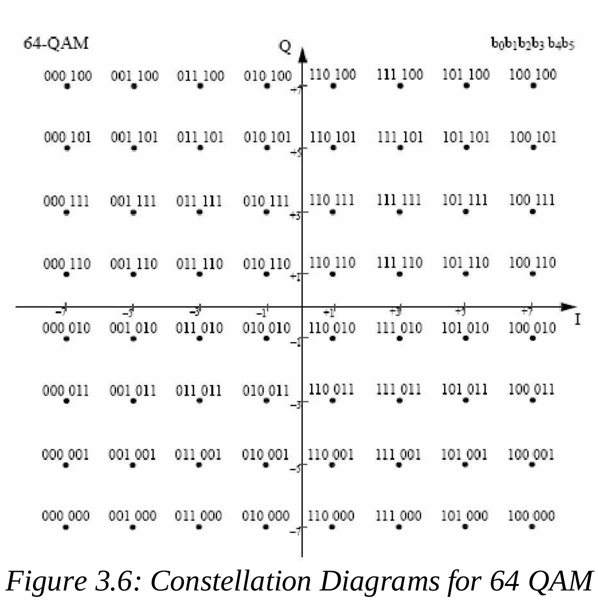


Figure 3.6: Constellation Diagrams for 64 QAM

3.4 Simulations of QAM

M-QAM has been simulated in Matlab. M used in the simulation is 4, 16, 64, and 256. Different QAM results summary is given in the Table 3.1. The results are compared for BER of 10^{-3}. For 4 QAM the SNR is approximately 10.5 dB, for 16 QAM the SNR is approximately 17.5 dB, for 64 QAM the SNR is approximately 24.2 dB, and for 256 QAM the SNR is approximately 30.2 dB. As the M value increases in order to maintain good BER, SNR also needs to be increased appropriately. For each QAM points, separate Figures are given.

- Figure 3.7 is for 256 QAM. In this figure, basic constellation, decoded constellations for two different SNRs and BER is plotted for different SNRs.
- Figure 3.8 is for 64 QAM. In this figure, basic constellation, decoded constellations for two different SNRs and BER is plotted for different SNRs.
- Figure 3.9 is for 16 QAM. In this figure, basic constellation, decoded constellations for two different SNRs and BER is plotted for different SNRs.
- Figure 3.10 is for 4 QAM. In this figure, basic constellation, decoded constellations for two different SNRs and BER is plotted for different SNRs.

Modulation	**SNR in dB**
4 QAM	10.5
16 QAM	17.5
64 QAM	24.2
256 QAM	30.2

Table 3.1: SNR comparison over AWGN channel for different QAM modulations at BER of 10^{-3}

256 QAM

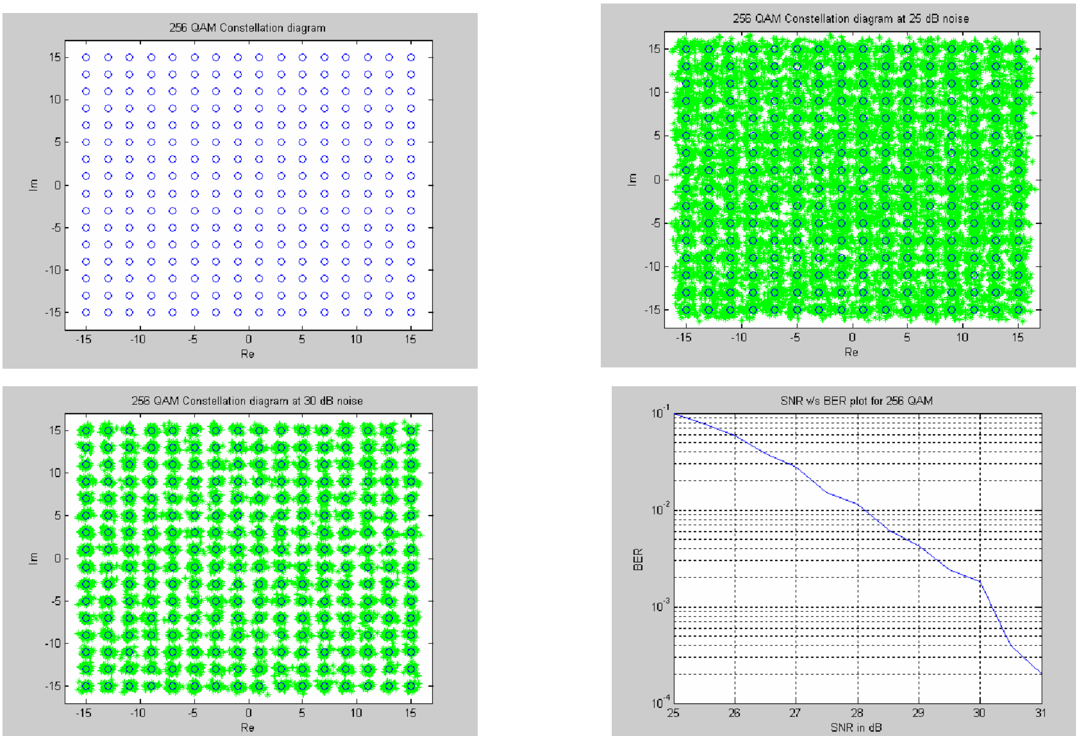


Figure 3.7: Constellation diagrams, AWGN noisy Signals, and BER plot for 256 QAM

64 QAM

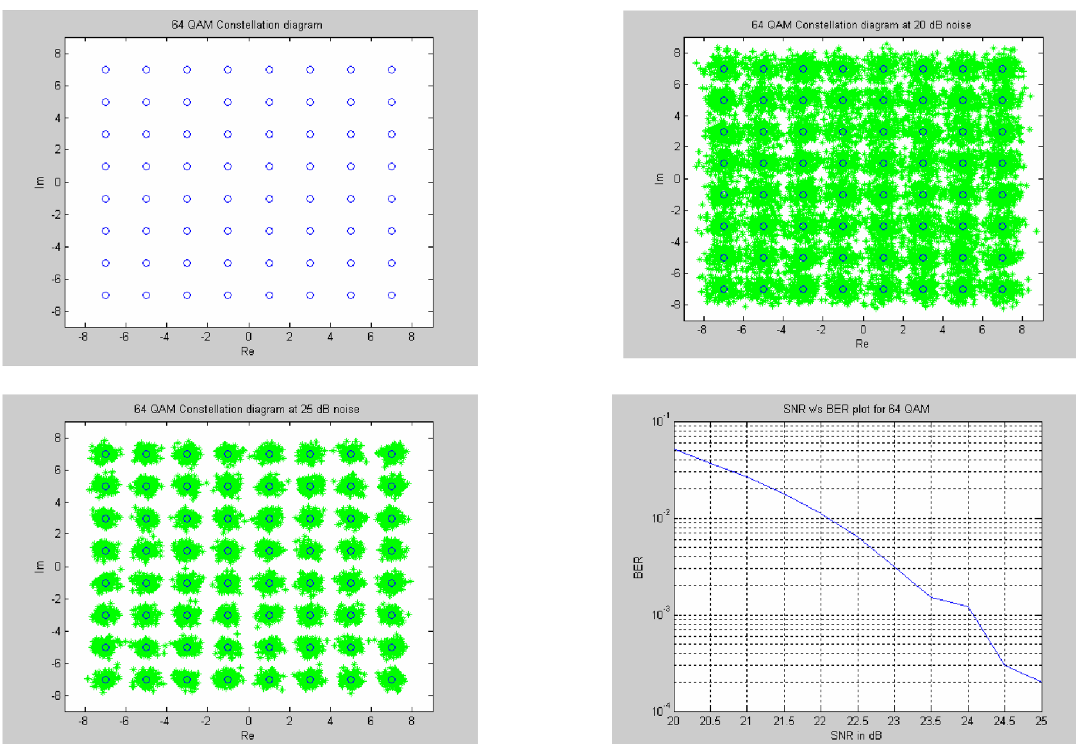


Figure 3.8: Constellation diagrams, AWGN noisy Signals, and BER plot for 64 QAM

16 QAM

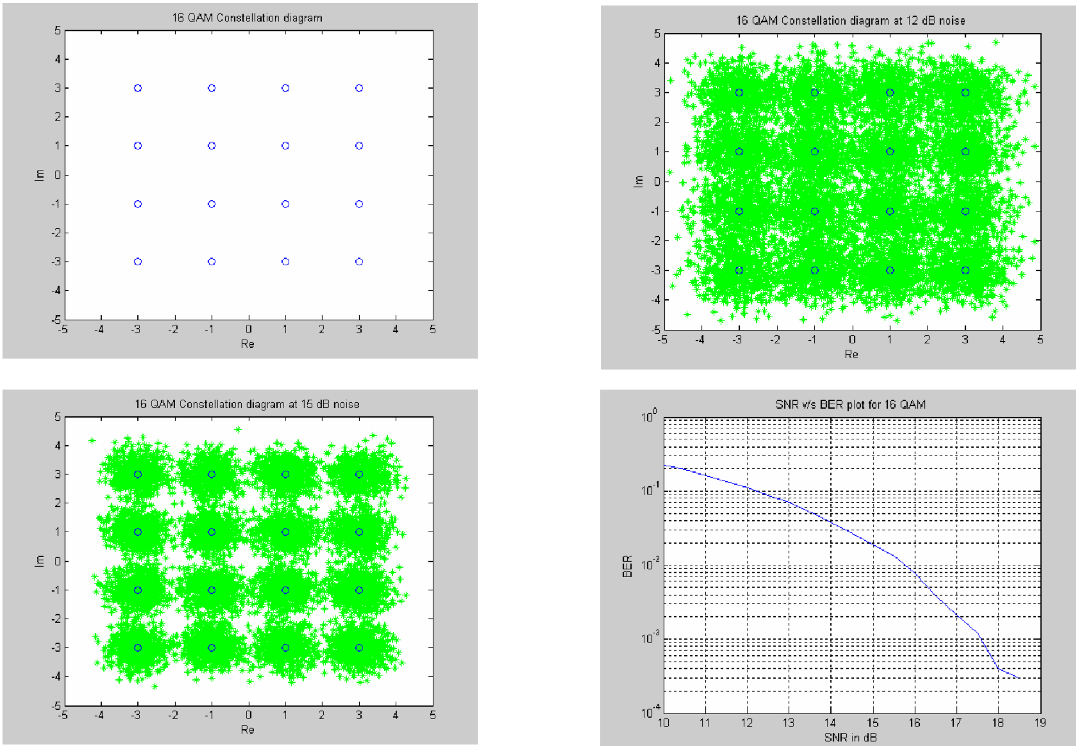


Figure 3.9: Constellation diagrams, AWGN noisy Signals, and BER plot for 16 QAM

4 QAM

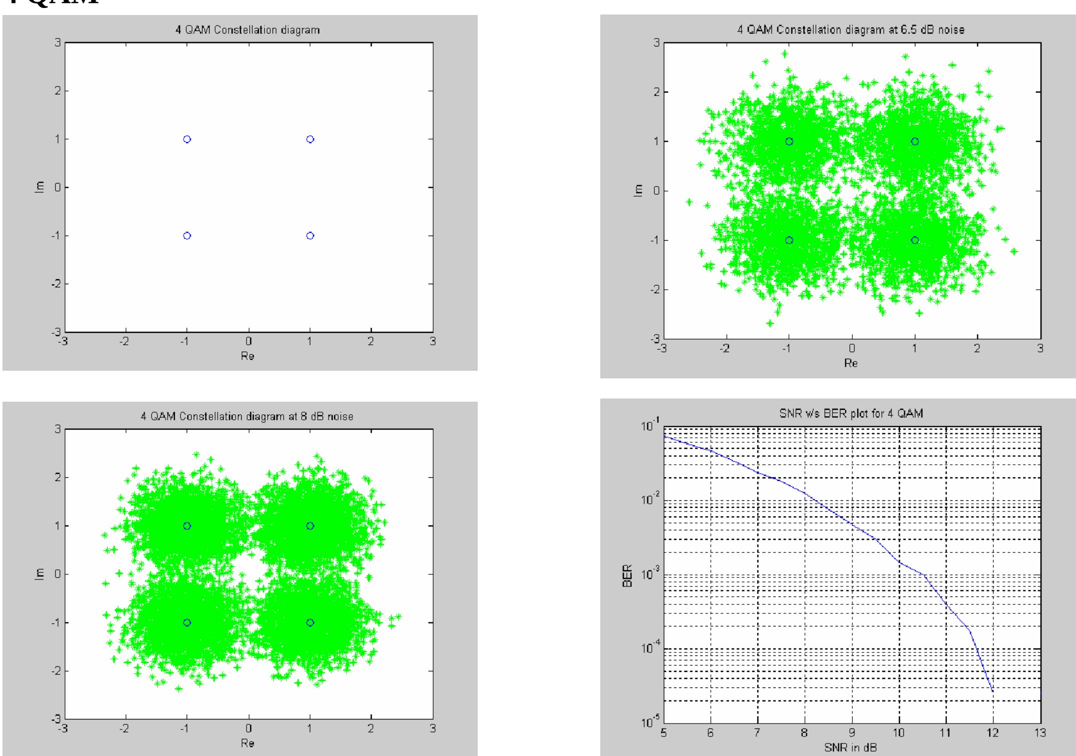


Figure 3.10: Constellation diagram, AWGN noisy signals, and BER plot for 4 QAM

3.5 Chapter Conclusion

In wireless systems PSK and QAM modulations are popularly used. In general PSK and QAM are used in several base band signal processing. PSK is mainly used for low bit rate transmissions. PSK operates at low SNR or for the given link conditions, they give better BER performance. Hence, PSK modulations are used for initial signaling, messages, and low data rates.

QAM is most popular for higher data rates. Most popularly QAM is used from 16 to 1024. Some base band signal processors are also using 4096 QAM. In QAM, the margins are low between signal constellation points. Hence, any imbalances or impediments in the channel or overall system can make the BER to grow. To contain BER low, it is required to send higher power. In practical systems power is limited. Hence, QAM is used with error correction schemes like FEC with RS codes. QAM used with MCM develop higher peak powers. Sometimes, peak transmitted power is limited. This may develop more errors. Once again, peak power clipping and error correction schemes help here.

Chapter 4

ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING

4. Orthogonal Frequency Division Multiplexing

4.1 Introduction

OFDM systems are better than single-carrier systems in multi-path fading channel environment. OFDM is used in many high data rate transmission systems, for example, DVB, IEEE 802.11, IEEE 802.16, HIPERLAN Type II, many derivatives of DSL and Home networking etc. It is projected that OFDM systems are the strongest candidate for 4G systems.

OFDM is a digital carrier modulation scheme, which uses a large number of closely spaced orthogonal sub carriers. Each sub carrier is modulated with a conventional modulation scheme at low symbol rate, maintaining data rates similar to conventional single carrier modulation schemes in the same bandwidth.

The primary advantage of OFDM over single carrier schemes is its ability to cope with severe channel conditions like multipath, narrowband interference without complex equalization filters. Channel equalization is simplified because OFDM may be viewed as using many slowly modulated narrowband signals rather than one rapidly modulated wideband signal.

The orthogonality of the subcarriers results in zero cross talk, even though they are so close that their spectra overlap. Low symbol rate helps manage time domain spreading of the signal by allowing the use of guard interval between the symbols. The guard interval also eliminates the pulse shaping filter.

This Chapter discusses the implementation and simulation of OFDM system in Matlab. Following the introduction this chapter, Section 4.2 describes detailed description of OFDM, Section 4.3 & 4.4 describe importance of Fourier Transform in OFDM, Section 4.5 describes importance of Guard interval, Section 4.6 describes Choice of key elements used in OFDM, and Section 4.7 is presented with simulation results.

4.2 Orthogonal Frequency Division Multiplexing

4.2.1 The Importance of Orthogonality

The “orthogonal” part of the OFDM name indicates that there is a precise mathematical relationship between the frequencies of the carriers in the system. Considering set of signals Ψ_p , where Ψ_p is the *p*-th element in the set. The signals are orthogonal if

$$\int_a^b \Psi_p(t)\Psi_q^*(t) = \begin{cases} \text{k for p = q} \\ \text{0 for p} \neq \text{q} \end{cases}$$

where the * indicates the complex conjugate and interval [a,b] is a symbol period. A fairly simple mathematical proof exists, that the series *sin*(*mx*) for *m*=1,2,… is orthogonal over the interval -π to π. Most of transform theory makes the use of orthogonal series.

Frequency division multiplexing (FDM) is used in many communication systems. In a normal FDM system, the many carriers are spaced apart in such way that the signals can be received using conventional filters and demodulators. In such receivers, guard bands have to be introduced between the different carriers as shown in *Figure 4.1*. The introduction of these guard bands in the frequency domain result in a lowering of the spectrum efficiency.

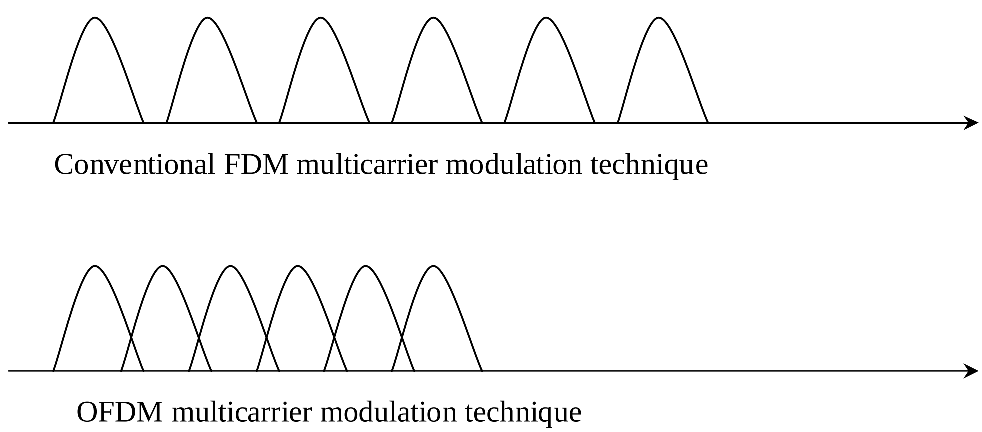


Figure 4.1: Functional bands representation of FDM and OFDM

It is possible, however, to arrange the carriers in an OFDM signal so that the sidebands of the individual carriers overlap and the signals can still be received without adjacent carrier interference. In order to do this all the carriers must be mathematically orthogonal in a symbol interval. The receiver acts as a bank of demodulators, translating each carrier down to DC, the resulting signal then being integrated over a symbol period to recover the raw data. If the other carriers all beat down to frequencies which, in the time domain, have a whole number of cycles in the symbol period (τ), then the integration process results in zero contribution from all these carriers. Thus the carriers are linearly independent (i.e., orthogonal) if the carrier spacing is a multiple of $1/\tau$.

4.2.2 Mathematical Description of OFDM

OFDM transmits a large number of narrowband carriers, closely spaced in the frequency domain. In order to avoid a large number of modulators and filters at the transmitter and complementary filters and demodulators at the receiver, it is desirable to be able to use modern digital signal processing techniques, such as fast Fourier transform (FFT).

Mathematically, each carrier can be described as a complex wave

$$S_c(t) = A_c(t)e^{j[\omega_c t + \Phi_c(t)]} \qquad (4.1)$$

The real signal is the real part of $S_c(t)$. Both $A_c(t)$ and $\Phi_c(t)$, the amplitude and phase of the carrier, can vary on a symbol by symbol basis. The values of the parameters are constant over the symbol duration period τ **.**

OFDM consists of many carriers. Thus the complex signals $S_s(t)$ is represented by

$$S_s(t) = \frac{1}{N}\sum_{n=0}^{N-1} A_n(t)e^{j[(\omega_n t + \Phi_n(t))]} \qquad (4.2)$$

Where

$$\omega_n = \omega_o + n\mathbf{S}\omega$$

The above representation is for a continuous signal. If we consider the waveforms of each component of the signal over one symbol period, then the variables *Ac (t)* and *Φ (t)* take fixed values, which depend on the frequency of that particular carrier, and so can be rewritten

$$\Phi_n(t) \Rightarrow \Phi_n$$
$$A_n(t) \Rightarrow A_n$$

If the signal is sampled using a sampling frequency of 1/T, then the resulting signal is represented by

$$S_s(kt) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j[(\omega_0 + n\mathbf{S}\omega)kT + \Phi_n]} \qquad (4.3)$$

At this point, we have restricted the time over which we analyze the signal to N samples. It is convenient to sample over the period of one data symbol. Thus we have a relationship

$$\tau = NT$$

If we now simplify *Equation 4.3*, without a loss of generality by letting ω_0=0, then the signal becomes

$$S_s(kt) = \frac{1}{N}\sum_{n=0}^{N-1} A_n e^{j\Phi_n} e^{j(n\mathbf{s}\omega)kT} \qquad (4.4)$$

Now *Equation 4.4* can be compared with the general form of the inverse Fourier transform

$$g_s(kt) = \frac{1}{N}\sum_{n=0}^{N-1} G\left(\frac{n}{NT}\right) e^{j2\pi nk/N} \qquad (4.5)$$

In *Equations 4.4*, the function $A_n e^{j\Phi_n}$ is no more than a definition of the signal in the sampled frequency domain, and *s(kT)* is the time domain representation. *Equations 4.4 and 4.5* are equivalent if

$$\mathbf{s}f = \frac{\mathbf{s}\omega}{2\pi} = \frac{1}{NT} = \frac{1}{\tau} \qquad (4.6)$$

This is the same condition that was required for orthogonality (see *Importance of orthogonality*). Thus, one consequence of maintaining orthogonality is that the OFDM signal can be defined by using Fourier transform procedures.

4.3 The Fourier Transform

The Fourier transform allows transformation from time domain to frequency domain.

The conventional Fourier transform relates to continuous signals which are not limited to in either time or frequency domains. However, signal processing is made easier if the signals are sampled. Sampling of signals with an infinite spectrum leads to aliasing, and the processing of signals which are not time limited can lead to problems with storage space.

To avoid this, the majority of signal processing uses a version of the DFT. The DFT is a variant on the normal transform in which the signals are sampled in both time and the frequency domains. By definition, the time waveform must repeat continually, and this leads to a frequency spectrum that repeats continually in the frequency domain.

FFT is merely a rapid mathematical method for computer applications of DFT. It is the availability of this technique, and the technology that allows it to be implemented on integrated circuits at a reasonable price, that has permitted OFDM to be developed as far as it has. The process of transforming from the time domain representation to the frequency domain representation uses the Fourier transform itself, whereas the reverse process uses the inverse Fourier transform.

4.4 Signal Representation of OFDM using IFFT/FFT

The definition of the N-point DFT is

$$X_p[k] = \sum_{n=0}^{N-1} x_p[n] e^{-j(2\pi/N)kn} \quad \text{(DFT)} \qquad (4.7)$$

And the N-point IDFT is

$$x_p[n] = \frac{1}{N} \sum_{k=0}^{N-1} X_p[k] e^{j(2\pi/N)kn} \quad \text{(IDFT)} \qquad (4.8)$$

A natural consequence of this method is that it allows us to generate carriers that are orthogonal. The members of an orthogonal set are linearly independent.

Consider a data sequence (d_0, d_1, d_2, …, d_{N-1}), where each d_n is a complex number $d_n=a_n+jb_n$. (a_n, b_n=± 1 for QPSK, a_n, b_n=± 1, ± 3 for 16-QAM, a_n, b_n = ± 1, ± 3, ± 5 for 64 QAM, and a_n, b_n= ± 1, ± 3, ± 5, ± 7 for 256 QAM)

$$D_m = \sum_{n=0}^{N-1} d_n e^{-j(2\pi nm/N)} = \sum_{n=0}^{N-1} d_n e^{-j2\pi f_n t_m} \quad \text{k = 0, 1, 2, 3…..N-1} \qquad (4.9)$$

where $f_n = n/(N\mathbf{s}T)$, $t_k = k\mathbf{s}t$ and $\mathbf{s}t$ is an arbitrarily chosen symbol duration of the serial data sequence d_n.

The real part of the vector D has components

$$Y_m = \mathrm{Re}\{D_m\} = \sum_{n=0}^{N-1} [a_n \cos(2\pi f_n t_m) + b_n \sin(2\pi f_n t_m)], \quad \text{k = 0, 1, 2, 3…..N-1}$$

(4.10)

If these components are applied to a low-pass filter at time intervals $\mathbf{s}t$, a signal is obtained that closely approximates the frequency division multiplexed signal

$$y(t) = \sum_{n=0}^{N-1} [a_n \cos(2\pi f_n t_m) + b_n \sin(2\pi f_n t_m)], 0 \le t \le N\mathbf{s}t \qquad (4.11)$$

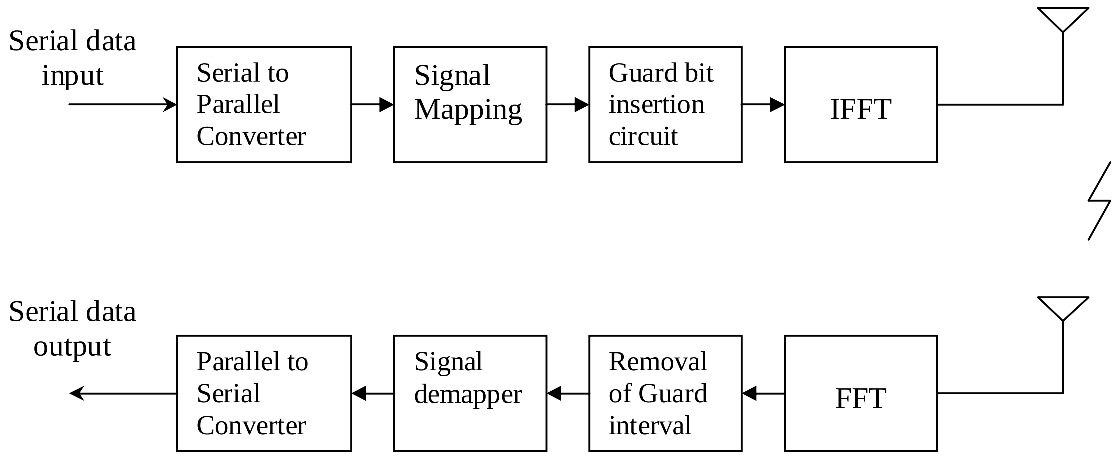


Figure 4.2: Block diagram of an OFDM system using FFT

Figure 4.2 illustrates the process of a typical FFT-based OFDM system. The incoming serial data is first converted form serial to parallel and grouped into ‘x’ bits each to form a complex number. The number ‘x’ determines the signal constellation of the corresponding subcarrier, such as BPSK, QPSK, 16 QAM, 64QAM, and 256 QAM. The complex numbers are modulated in the baseband by the inverse FFT (IFFT) and converted back to serial data for transmission. The receiver performs the inverse process of the transmitter. The Figure 4.2 on base band OFDM functions. Complete system is not shown in this Figure 4.2.

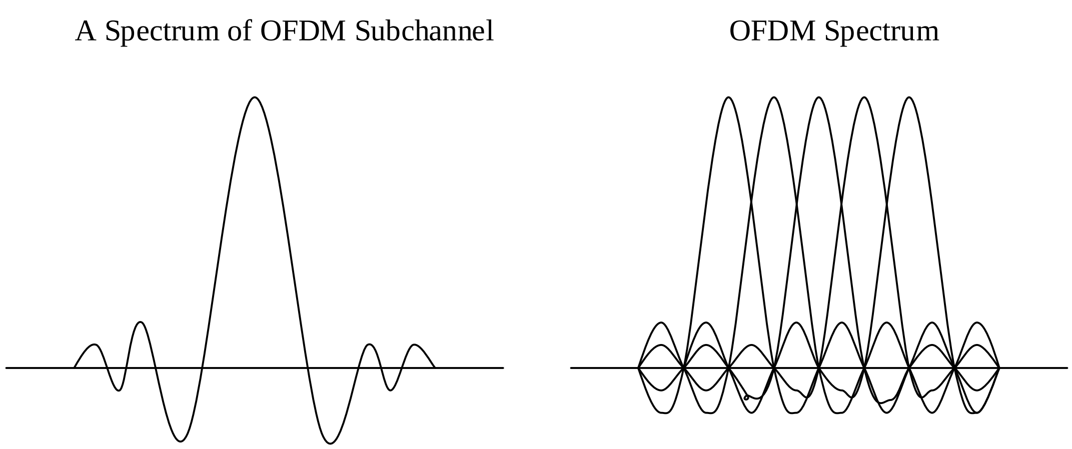


Figure 4.3: Examples of OFDM spectrum (a) a single subchannel, (b) 5 carriers at the central frequency of each subchannel, there is no crosstalk from other subchannels

Figure 4.3a shows the spectrum of an OFDM subchannel and *Figure 4.3b* shows an OFDM spectrum. By carefully selecting the carrier spacing, the OFDM signal spectrum can be made flat and the orthogonality among the subchannels can be guaranteed

4.5 Guard Interval

The orthogonality of subchannels in OFDM can be maintained and individual subchannels can be completely separated by the FFT at the receiver when there are no ISI and ICI introduced by transmission channel distortion. In practice these conditions can not be obtained. Since the spectra of an OFDM signal is not strictly band limited (*sinc(f)* function), linear distortion such as multipath cause each subchannel to spread energy into the adjacent channels and consequently cause ISI. A simple solution is to increase symbol duration or the number of carriers so that distortion becomes insignificant. However, this method may be difficult to implement in terms of carrier stability, Doppler shift, FFT size and latency.

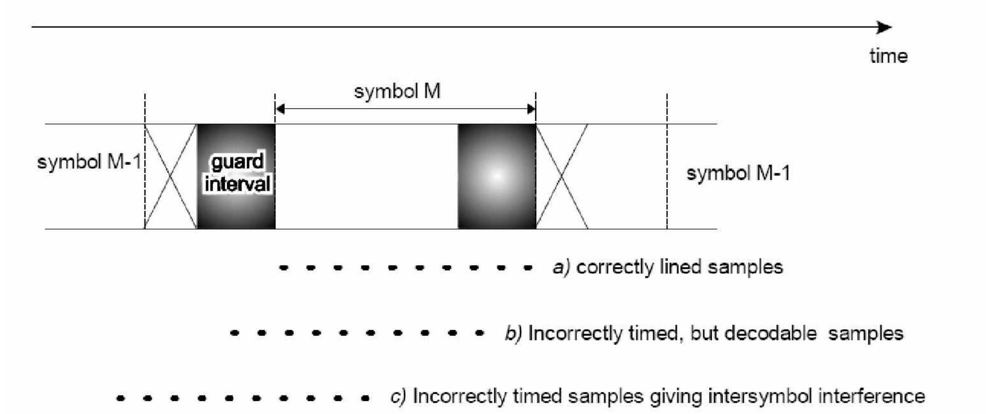


Figure 4.4: The effect on the timing tolerance of adding a guard interval. With a guard interval included in the signal, the tolerance on timing the samples is considerably more relaxed

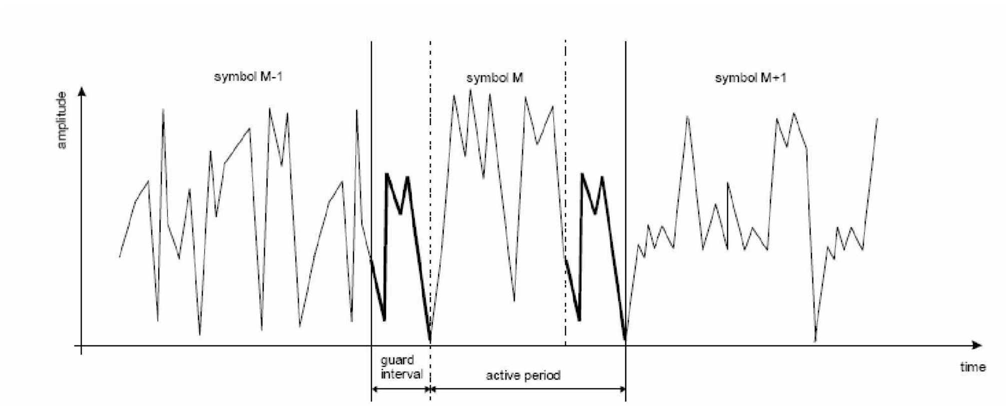


Figure 4.5: Example of the guard interval. Each symbol is made up of two parts. The whole signal is contained in the active symbol (shown highlighted for the symbol M). The last part of which (shown in bold) is also repeated at the start of the symbol and is called the guard interval

One way to prevent ISI is to create a cyclically extended guard interval (*Figure 4.4, 4.5*), where each OFDM symbol is preceded by a periodic extension of the signal itself. The total symbol duration is $T_{\text{total}}=T_g+T$, where T_g is the guard interval and T is the useful symbol duration. When the guard interval is longer than the channel impulse response, or the multipath delay, the ISI can be eliminated. However, the ICI, or in-band fading, still exists. The ratio of the guard interval to useful symbol duration is application-dependent. Since the insertion of guard interval will reduce data throughput, *T*g is usually less than *T*/4.

The Reasons to use a Cyclic Prefix for the Guard Interval are

1. To maintain the receiver carrier synchronization, some signals instead of a long silence must always be transmitted.
2. Cyclic convolution can still be applied between the OFDM signal and the channel response to model the transmission system.

4.6 Choice of the Key Elements

4.6.1 Useful Symbol Duration

The useful symbol duration *T* affects the carrier spacing and coding latency. To maintain the data throughput, longer useful symbol duration results in increase of the number of carriers and the size of FFT (assuming the constellation is fixed). In practice, carrier offset and phase stability may affect how close two carriers can be placed. If the application is for the mobile reception, the carrier spacing must be large enough to make the Doppler shift negligible. Generally, the useful symbol duration should be chosen so that the channel is stable for the duration of a symbol.

4.6.2 Number of Carriers

The number of sub carriers can be determined based on the channel bandwidth, data throughput and useful symbol duration.

$$N = \frac{1}{\tau}$$

The carriers are spaced by the reciprocal of the useful symbol duration. The number of carriers corresponds to the number of complex points being processed in FFT. For IEEE 802.11 *'a'*, *'g'* WLAN applications, the number of sub carriers are 52 (48 data carriers plus 4 pilot carriers).

4.6.3 Modulation Scheme

The modulation scheme in an OFDM system can be selected based on the requirement of power or spectrum efficiency. The type of modulation can be specified by the complex number $d_n=a_n+jb_n$. The symbols a_n and b_n can be selected to (± 1, ± 3, ± 5, ± 7) for 256 QAM, (± 1, ± 3, ± 5) for 64 QAM, (± 1, ± 3) for 16QAM and ± 1 for QPSK. In general, the selection of the modulation scheme applying to each sub channel depends solely on the compromise between the data rate requirement and transmission robustness. Another advantage of OFDM is that different modulation schemes can be used on different subchannels for layered services.

4.7 Advantages and Disadvantages of OFDM

The advantages of OFDM are

1. Efficient use of the available bandwidth since the subchannels are overlapping
2. Spreading out the frequency fading over many symbols. This effectively randomizes the burst errors caused by the Rayleigh fading, so that instead of several adjacent symbols (in time on a single-carrier) being completely destroyed, (many) symbols in parallel are only slightly distorted.
3. The symbol period is increased and thus the sensitivity of the system to delay spread is reduced.

The Disadvantages of OFDM are

1. OFDM signal is contaminated by non-linear distortion of transmitter power amplifier; because it is a combined amplitude-frequency modulation (it is necessary to maintain linearity)
2. OFDM is very sensitive to carrier frequency offset caused by the jitter of carrier wave and Doppler Effect caused by moving of the mobile terminal.
3. At the receiver, it is very difficult to decide the starting time of the FFT symbol

4.8 Simulation Results

Table 4.1 shows the configuration used for the simulations performed on OFDM signal. A 64 carrier system was used, FFT size of 64 is taken and QPSK, 16 QAM, 64 QAM, and 256 QAM modulation schemes are used. The OFDM signal is passed over AWGN channel and the received signal is demodulated and compared with the original signal to calculate BER. It is observed that for AWGN channel we have to maintain minimum of 10.5 dB for QPSK, 17.5 dB for 16 QAM, 24.2 dB for 64 QAM and 30.2 dB for 256 QAM at BER of 10^{-3}. *Figure 4.6* shows obtained simulations results for different modulation schemes in Matlab. It was found that the SNR performance of OFDM is similar to a standard single carrier digital transmission. This is to be expected, as the transmitted signal is similar to a standard Frequency Division Multiplexing (FDM) system.

Parameter	**Value**
Carrier Modulations Used	QPSK, 16 QAM, 64 QAM, and 256 QAM
FFT Size	64
Number of Carriers	64
Channel	AWGN

Table 4.1: OFDM System Parameters used for simulations

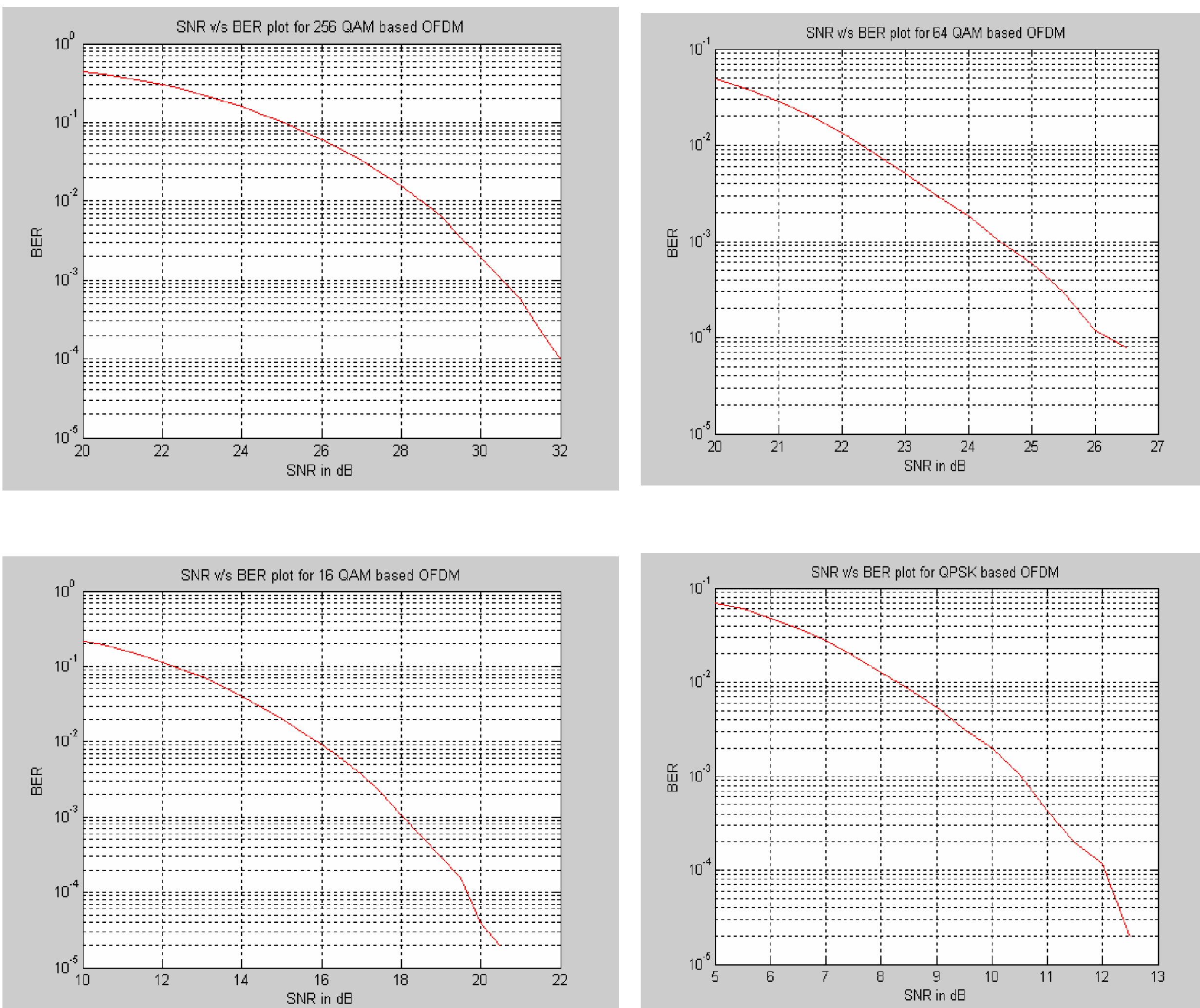


Figure 4.6: SNR v/s BER plots for OFDM system using subcarrier modulation schemes 256 QAM, 64 QAM, 16 QAM and QPSK

4.9 Chapter Conclusion

OFDM is the most popular scheme now for higher bit rate applications. This has built in orthogonality. This works based on simple frequency analysis. Spectrum utilization from OFDM is much higher. Guard band protects from various interferences. Higher bandwidth applications use up to 4096 QAM points. OFDM is easily implemented using fast computation of FFT. Several OFDM based design use dedicated hardware for FFT. The other names of Digital multi-tone (DMT) modulation, MCM are used in place of OFDM based processing. OFDM is not sufficient for end to end applications. Several error correction, interleaving schemes are used for better data throughput even under severe end to end impediments.

Chapter 5

REED SOLOMON CODED OFDM

5. Reed Solomon Coded OFDM

5.1 Introduction

OFDM has recently received increased attention due to its capability of supporting high data rate communication in frequency selective fading environments which cause ISI. Instead of using a complicated equalizer as in the conventional single carrier systems, the ISI in OFDM can be eliminated by adding a guard interval which significantly simplifies the receiver structure. However, in order to take advantage of the diversity provided by the multi-path fading, appropriate frequency interleaving and coding is necessary. Therefore, coding becomes an inseparable part in most OFDM applications and a considerable amount of research has focused on optimum encoder and decoder design for information transmission via OFDM over fading environments.

FEC is widely used in digital telecommunications systems to make it easier to encode and decode signals. Coding (FEC techniques) introduces redundancy in data and adding delay. Simplest error-detecting technique is parity check bit coding, i.e., adding an extra binary digit at the end of each word. But it will be effective only if the probability of error is very low.

Coding allows us to reduce the information bit error rate while maintaining a fixed transmission rate. In principle it allows us to reach Shannon's limit with ingenious coding.

The RS block code is organized on the basis of groups of bits. Such groups of bits are referred to as symbols. We store sequences of 'm' individual bits which appear serially in a bit stream and thereafter operate with the m-bit sequence rather than individual bits. These are called m-bit symbols. Even if an error occurs in a single bit of a symbol, the entire symbol is in error. RS code has 'k' information symbols, 'r' parity symbols and a total codeword length of 'n' (=k + r) symbols. The number of symbols in the codeword is arranged to be $n = 2^{m-1}$. The RS code is able to correct errors in'ť' symbols where t=r/2.

This chapter discusses the performance of the OFDM with RS coding technique. Following the introduction, Section 4.2 describes the encoding Section 4.3 describes about the decoding techniques. Section 4.4 describes concluding remarks.

5.2 Encoding

The encoder is the easy bit. Since the code is systematic, the whole of the block can be read into the encoder, and then output other side without alteration. Once the k^{th} data symbol has been read in, the parity symbol calculation is finished, and the parity symbols can be output to give the full n symbols.

The idea of the parity words is to create a long polynomial (‘n’ coefficients long – it contains the message and the parity) which can be divided exactly by the RS generator polynomial. At the decoder, the received message block can be divided by the RS generator polynomial. If the remainder of the division is zero, then no errors are detected. If there is a reminder, then there are errors.

The encoder acts to divide the polynomial represented by the ‘k’ message symbols d(x) by the RS generator polynomial g(x).

$$x(n-k)*d(x)/g(x) = q(x) + r(x)/g(x)$$

The term x(n-k) is a constant power of ‘x’, which is simply a shift upwards n-k places of all the polynomial coefficients in d(x). The reminder after the division r(x) becomes the parity. By concatenating the parity symbols on the end of ‘k’ message symbols, an ‘n’ coefficient polynomial is created which is exactly divisible by g(x). The *Figure 5.1* shows the Encoder.

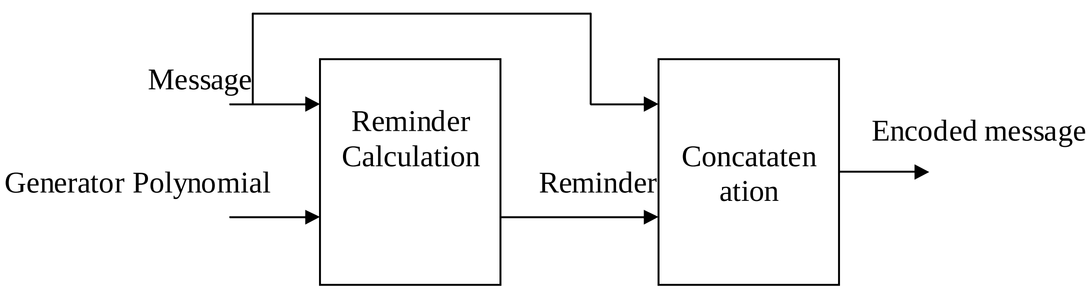


Figure 5.1: Reed Solomon Encoder

5.3 Decoding

Decoding is far harder task than encoding. Decode operation takes several stages. There are plenty of sources available for software implementations of the various algorithms required. The below block diagram (*Figure5.2*) shows the RS decoder. Each block is explained in detail below.

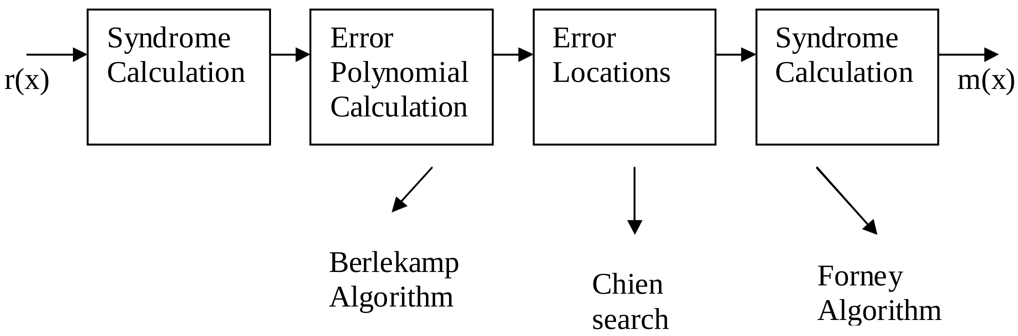


Figure 5.2: Reed Solomon Decoder

The major blocks in Reed Solomon decoder are

1. Syndrome Calculation
2. Error Polynomial Calculation
3. Error Location Calculation
4. Syndrome Calculation

5.3.1 Syndrome Calculation

Let the received polynomial be,

$$r(x) = r_0 + r_1x + r_2x^2 + \ldots\ldots. + r_{n-1}x^{n-1} \qquad (5.1)$$

and the generator polynomial has roots α, α^2 α^3 ……. α^{2t}, syndromes are calculated by substituting these roots into received polynomial in Equation (5.1). Let the calculated syndromes be S_1, S_2 … and $S_{2t.}$ From these Syndromes we will calculate error polynomial using Berlekamp algorithm.

5.3.2 Error Polynomial Calculation using Berlekamp Algorithm

For calculating error polynomial we will use Berlekamp algorithm as explained below.

Algorithm

1) Initialize parameters $i = 0$, $\Delta_0 = 1$, $u = 0$, $\Lambda^{(0)}(x) = \Lambda^{(-1)}(x) = 1$, $L_0 = 0$.

2) $i = i + 1$.

3) Compute $S_{i.}$

4) $\Delta_i = S_i + \sum_{l=1}^{L_{i-1}} \Lambda_l^{(l-1)} S_{l-1}$

5) if $\Delta_i = 0$ $\Lambda^{(i)}(x) = \Lambda^{(i-1)}(x)$ and $L_i = L_{i-1}$ and go to step 9 else continue

6) $\Lambda^{(i)}(x) = \Lambda^{(i-1)}(x) - \Delta_i \Delta_u^{(-1)} x^{(i-u)} \Lambda^{(u-1)}(x)$

7) if $L_{i-1} >= i - L_{i-1}$ then $L_i = L_{i-1}$ and goto step 9 else continue

8) $u = i$,and $L_{i-1} = i - L_{i-1}$

9) if i = n-k stop else go to step 2

The below *Figure5.3* shows the Berlekamp algorithm flowchart.

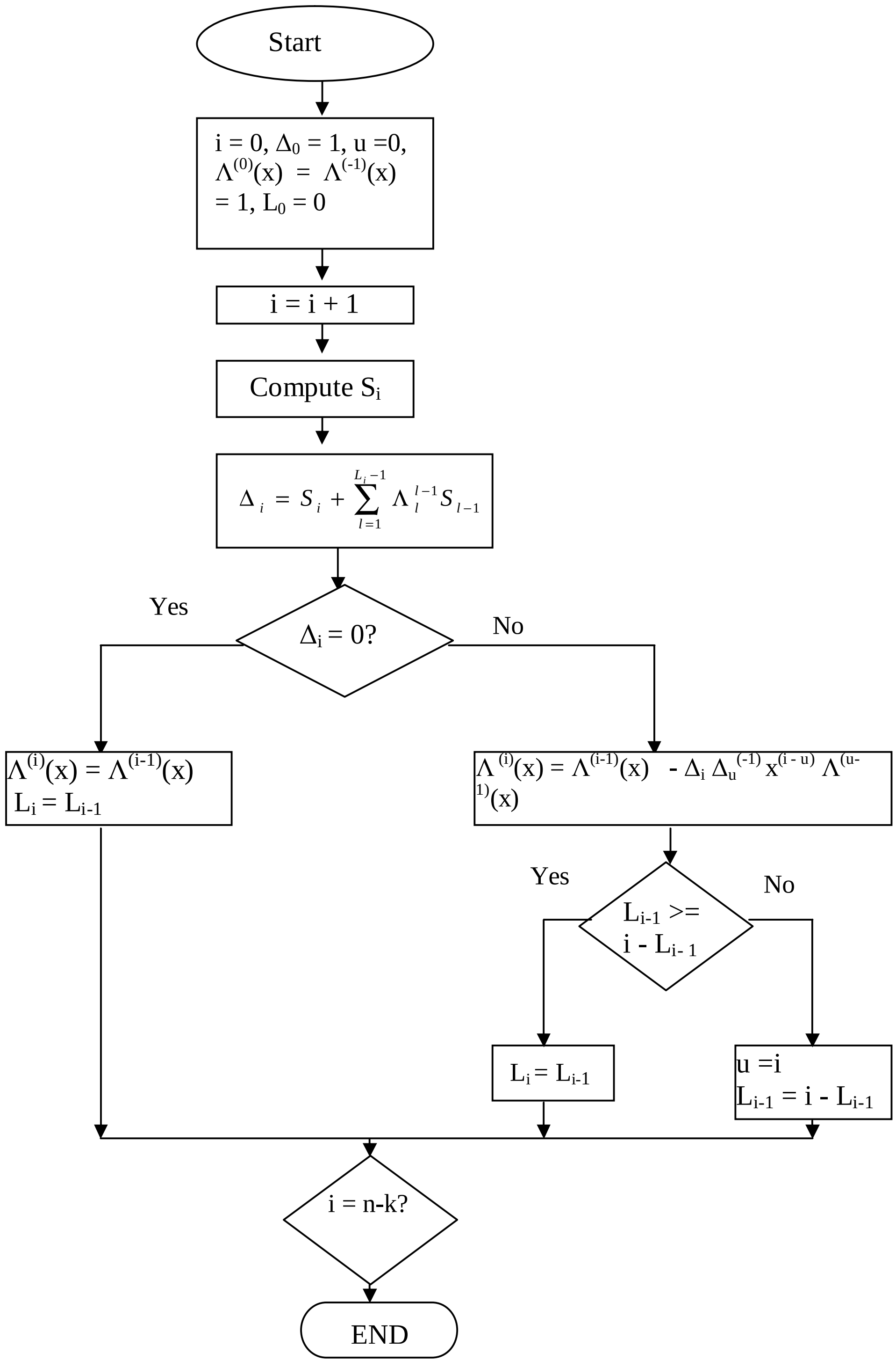


Figure 5.3: Berlekamp Algorithm Flowchart

5.3.3 Error Location Calculation using Chien Search

Let us denote error polynomial with

$$\Lambda(x) = 1 + \Lambda_1 x + \Lambda_2 x^2 + \Lambda_3 x^3 + ...\Lambda_t x^t$$

and we are interested in determining the roots of this polynomial since the polynomial is defined over finite field, with primitive element α then we show that if the order of field is n, $\alpha^{-i} = \alpha^{n-i}$ using this fact, if α^{i} is the root of the polynomial then we may write

$$\Lambda(\alpha^i) = 1 + \Lambda_1 \alpha + \Lambda_2 \alpha^2 + \Lambda_3 \alpha^3 + ...\Lambda_t \alpha^t$$

So the following steps, we may calculate the roots

1) For each power of α in the range j = 0,1, …..n-1 use α^j *as* our test root
2) calculate the polynomial coefficients at the current root using, coefficients of the past iteration using

$$\Lambda_i^{j-1} = \Lambda_i^{j-1} \alpha^i \quad \text{during j}^{\text{th}} \text{ iteration}$$

3) Calculate the polynomial coefficients, and if the sum $\sum_{i=1}^{t} \Lambda_i^j = 0$

 is equal to '0' then save α^{-j} as a root.
4) Continue to next iteration.

5.3.4 Error Magnitudes using Forney Algorithm

First, define an infinite-degree syndrome polynomial

$$S(x) = S_1 x + S_2 x^2 + ...S_{2t} x^{2t} + S_{2t+1} x^{2t+1} + ...$$

Then, define the *error magnitude polynomial* as follows

$$\Omega(x) = [1 + S(x)]\Lambda(x)$$

Given that we know only the first $2t$ coefficients of $S(x)$, the decoding problem becomes one of finding a polynomial $\Lambda(x)$ of degree less than or equal to t that satisfies

$$\Lambda(x)[1 + S(x)] \equiv \Omega(x) \bmod x^{2t+1}$$

The error magnitudes are computed using the expression

$$e_{i_k} = \frac{-X_k \Omega(X_k^{-1})}{\Lambda^l (X_k^{-1})}$$

5.3.5 Error Correction

Now we have error polynomial and received polynomial, by adding both then we can obtain the error corrected message polynomial.

$$m(x) = r(x) + \Lambda(x)$$

5.4 OFDM with Reed Solomon Encoder and Decoder

The following *Figure5.4* shows OFDM simulation model used and simulation results are discussed below.

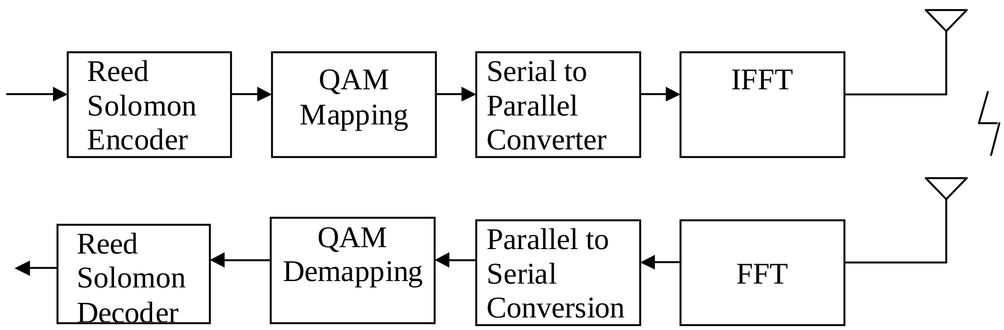


Figure 5.4: OFDM Simulation Model used

As shown in the *Figure 5.4*, initially the random data is encoded using RS encoder (RS (255,239) for 256 QAM, RS (63, 47) for 64 QAM). The encoded symbols are modulated, goes through serial to parallel conversion and IFFT is performed. IFFT signal is RF up converted and transmitted. In the receiver the received signal is down converted, FFT is performed, after FFT the signal is decoded using QAM constellation. The output of QAM demodulator is divided into 'n' (RS (n, k)) groups. Each group of 'n' symbols are applied to RS decoder to correct the errors.

5.5 Simulation Results

Figure 5.5 is shown for 256 QAM normal OFDM. For a BER of 10^{-3}, it is required to maintain SNR of 30dB. By applying RS coding, it can be observed that significant decrease in BER after 26dB. Similarly, as shown in *Figure 5.6*, for 64 QAM with RS coded OFDM, there is significant drop in BER after 17dB. RS coding improves the performance.

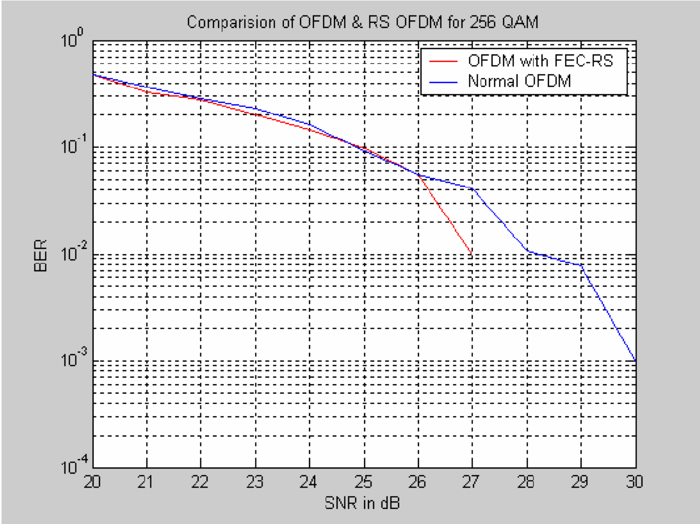


Figure 5.5: Comparison of OFDM and RS OFDM for 256 QAM

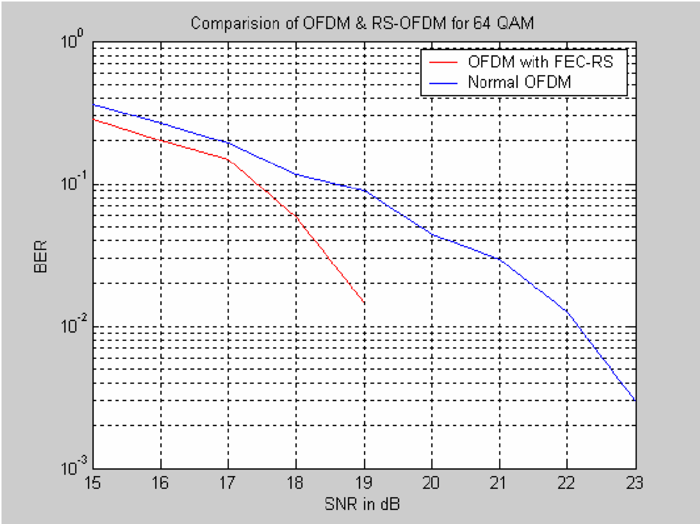


Figure 5.6: Comparison of OFDM and RS OFDM for 64 QAM

5.6 Chapter Conclusion

FEC schemes are used along with modulation schemes like OFDM. RS is one of the popular scheme. The RS codes are able to correct a corrupted symbol with a single bit error as it can be symbol with all bits in error. RS used with OFDM improves the on BER that will result in able to operate with lower SNR. This also helps in increase range. There are many other error correction codes like Turbo codes that provide more gain than RS. From complexity and easy VLSI implementation, RS is adapted popularly in base band processing.

Chapter 6

ADAPTIVE CLIPPING TECHNIQUE FOR REDUCING PAPR ON OFDM

6. Adaptive Clipping Technique for Reducing PAPR on OFDM

6.1 Introduction

This chapter is concerned about PAPR reduction technique on OFDM systems. BER performance of OFDM systems over clipping ratio (CR) are studied and simulated. We considered adaptive clipping technique for PAPR reduction on OFDM systems by modulation order.

This chapter discusses performance of the OFDM with PAPR technique. Following the introduction, Section 6.2 describes PAPR problem, Section 6.3 describes System model used, Section 6.4 describes about PAPR Reduction schemes, and Section 6.5 describes concluding remarks.

6.2 PAPR Problem

OFDM systems have the undesirable feature of a large PAPR of the transmitted signals. The transmitted signal has a non-constant envelope and exhibits peaks whose power strongly exceeds the mean power. Consequently, to prevent distortion of the OFDM signal, the transmit amplifier must operate in its linear regions. Therefore, power amplifiers with a large dynamic range are required for OFDM systems - amplifiers which are prohibitively costly and so continue to be the major cost component of OFDM systems. Consequently, reducing the PAPR is pivotal to reducing the expense of OFDM systems.

6.3 System Model

Simplified baseband transmitter and receiver block diagram of OFDM systems, described at *Figure 6.1* are considered for analyzing a PAPR characteristics of OFDM signals. First, transmitted binary source data is randomly generated. This random data is '1' or '0' and it happens equally likely. Next, this random data is mapped through QAM constellation. QAM mapped signal is complex value. Then, QAM mapped signal is loaded on each subcarrier, and goes through serial to parallel conversion. And IFFT is performed. IFFT output signal is operated with clipping for reducing PAPR, and then this signal is transmitted. The receiver structure has reciprocal architecture to the transmitter.

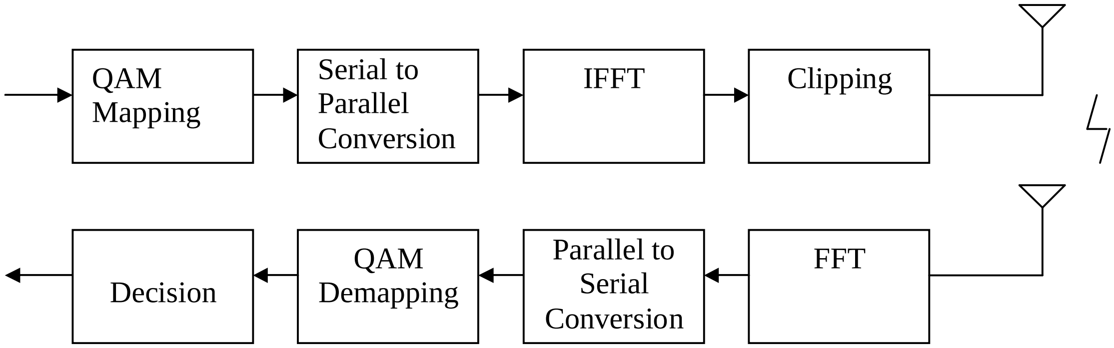


Figure 6.1: OFDM Simulation Model

Analysis of OFDM Signal

IFFT output signal at *Figure 6.1* is *Equation 6.1*.

$$x(n) = \frac{1}{N}\sum_{k=0}^{N-1} X(k)e^{j2\pi kn/N}, n = 0,1,2,...N-1 \qquad (6.1)$$

N number of subcarriers

$x\,(n)$ IFFT output signal, complex value

$X\,(k)$ IFFT input signal, complex value

If the FFT size is N, then average power is 1/N and peak power is 1 regardless of N at IFFT output side at *Figure 6.1*. So, PAPR equation can be written as *Equation 6.2*.

$$PAPR = 10\log\frac{1}{\frac{1}{N}} = 10\log N \qquad (6.2)$$

But peak power occurs very scarcely. Generally, the input signal of IFFT is a random data, so the output signal of IFFT is a random data. If N is sufficiently large then statistical characteristic of IFFT output signal is Gaussian random process by central limit theorem and it can be written as from *Equation 6.3* to *Equation 6.5*.

$$E\{x(m)x^*(n)\} = \begin{cases}\sigma^2, \text{ for } m=n \\ 0, \text{ for } m\neq n\end{cases} \qquad (6.3)$$

$$E\{x_R(m)x_R(n)\} = E\{x_I(m)x_I(n)\} = \begin{cases}\sigma^2/2, \text{ for } m=n \\ 0, \text{ for } m\neq n\end{cases} \qquad (6.4)$$

$$E\{x_R(m)x_I(n)\} = 0 \qquad (6.5)$$

$x\,(n)\,R$ a real value of $x(n)$

$x\,(n)\,I$ a imaginary value of $x(n)$

Now, we will tell a probability of peak power occurrence. When modulation is M-ary QAM and IFFT size is N then number of occasions that input OFDM symbol can have is M^N, and peak power occurs M^2 times. For example, N is 32 and QPSK modulation is used then probability of peak power occurrence is 8.7×10^{-19}. So this situation will never occur.

Therefore PAPR value is less important than probability distribution of IFFT output signal power. Now we will consider probability of IFFT output signal power.

6.4 PAPR Reduction Schemes

Clipping, the simplest technique of PAPR reduction scheme, is operated by *Equation 6.6.*

$$y(n) = \{ \begin{matrix} \frac{A}{\sqrt{|x(n)|^2}} x(n), if \, |x(n)|^2 > A^2 \\ x(n), if \, |x(n)|^2 \leq A^2 \end{matrix} \qquad (6.6)$$

A^2 is the maximum power

We define a CR to represent a clipping level. A CR is a ratio of the maximum power of clipped signal over average power of unclipped signal. IFFT output signal is normalized, so unclipped signal power is 1. If we clip a IFFT output signal at level A then a CR is $A^2 / 1$ $= A^2$. In this chapter we analyze probability distribution of IFFT output signal with FFT size and modulation order and BER with the CR by computer simulation.

6.5 Simulation Results

Figure 6.2 & 6.3 show BER performance with Modulation scheme and CR. In these figures 64-QAM and 256-QAM show an error flow phenomenon at CR. *Table 6.1* shows a required SNR when BER is 10^{-3} and we recommend a reasonable CR. When we use the 64 QAM modulations, required SNRs when CR is 3 and 4 are shown, so we can use CR 4 when 64 QAM is used. Similarly for 256 QAM, required SNRs when CR is 4 and 5 are shown, so we can use CR 5 when 256 QAM is used. So we propose the adaptive CR technique with modulation order like *Table 6.1*. If we use this algorithm we can overcome PAPR problem of OFDM systems with small BER degradation.

	64 QAM	**256 QAM**
Unclipped	24.2dB	30.2dB
CR = 5	--	31dB
CR = 4	24.8dB	> 35dB
CR = 3	> 30dB	--
Recommended	CR > 3	CR>4

Table 6.1: Required SNR with Modulation and Clipping Ratio

256 QAM

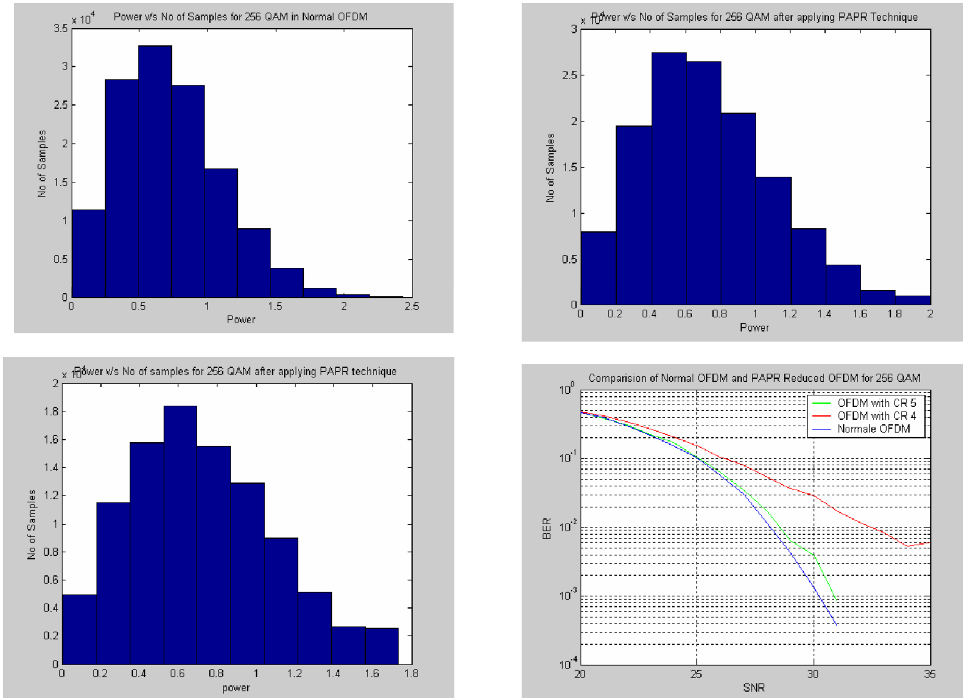


Figure 6.2: SNR v/s BER plots for Adaptive clipped OFDM system for 256 QAM at CR-4 and CR-5

64 QAM

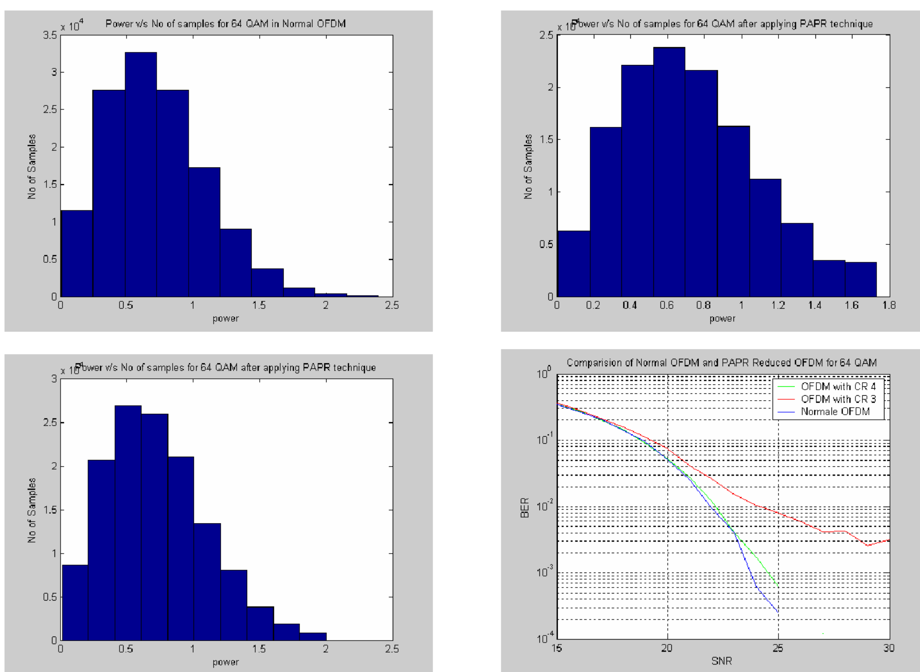


Figure 6.3: SNR v/s BER plots for Adaptive clipped OFDM system for 64 QAM at CR-3 and CR-4

6.5 Chapter Conclusion

The major drawback of OFDM system is when all the subcarriers are of same phase then the instantaneous power is very high, this leads to nonlinear characteristics in amplifier.

To overcome this earlier techniques are First, clipping technique is used, but it has BER degradation, second, peak power avoidance precoding technique, it will decreases the data rate.

This proposed technique will be similar to clipping technique but it is adaptive to modulation order. It is simple and combined advantages of both the clipping technique and peak power precoding technique.

Chapter 7

RS CODED OFDM FOR PAPR REDUCTION TECHNIQUE

7. RS Coded OFDM for PAPR Reduction Technique

7.1 Introduction

This chapter discusses the implementation and simulation of RS coded OFDM for PAPR reduction problem. In this chapter study and results of OFDM system under different noise conditions and different environments, different clipping ratios are presented. BER plots are plotted for multiple SNR conditions.

This chapter discusses performance of the RS coded OFDM with PAPR technique. Following the introduction, Section 7.2 describes System model used for simulation, Section 7.3 describes about some simulation results with discussion.

7.2 System Model Used

An RS coded OFDM for PAPR reduction problem was modeled using Matlab. In the simulation various parameters of the system are varied for testing. The main aim of doing the simulations was to measure the performance of the system under different channel conditions, different CR, and different OFDM configurations. The OFDM system was modeled using matlab and is shown in *Figure 7.1*.

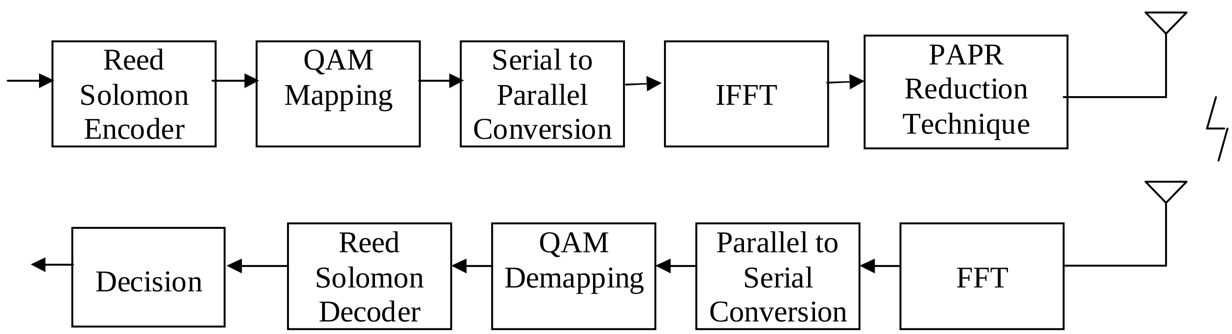


Figure 7.1: OFDM System Model used for Simulation

7.3 Simulation Results

By combining RS coding and PAPR reduction techniques, it is observed that under different channel conditions, and occurrence of peak power, the system gives more robust performance. Even though BER is slightly degraded due to PAPR technique, performance is comparable to that of RS coded OFDM.

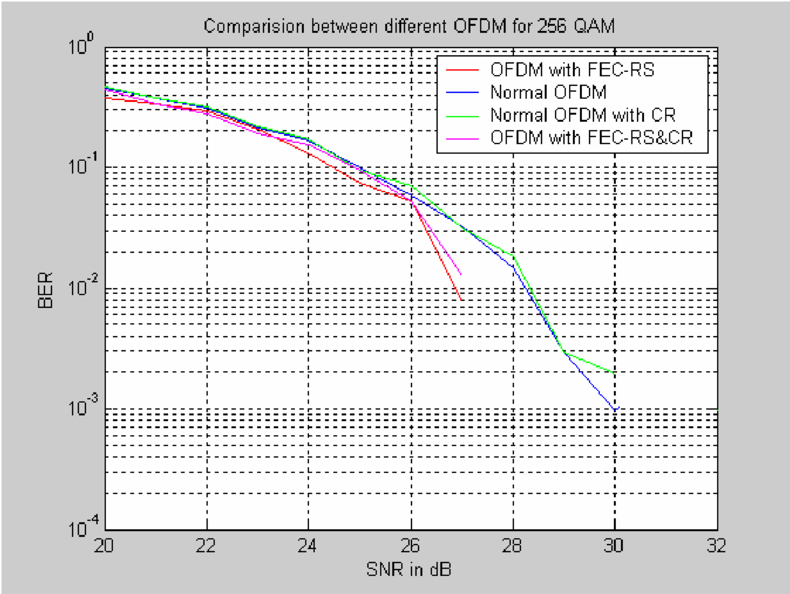


Figure 7.2: Comparison of OFDM, RS coded OFDM, PAPR Reduced OFDM, and RS Coded OFDM for PAPR Technique for 256 QAM

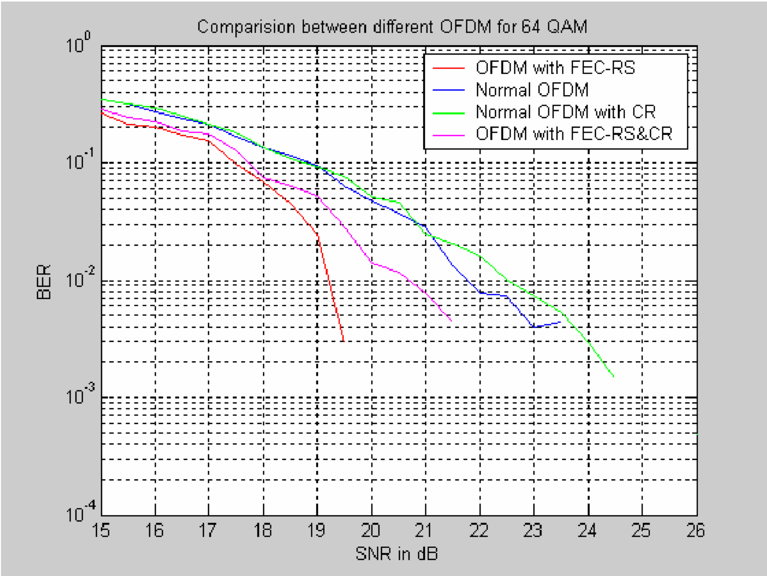


Figure 7.3: Comparison of OFDM, RS coded OFDM, PAPR Reduced OFDM, and RS Coded OFDM for PAPR Technique for 64 QAM

7.4 Chapter Conclusion

OFDM is the most popular technique for high bit rates, but in wireless transmission there are many parameters to consider for better performance. First, under severe channel conditions decoding of the signal is a problem. To overcome this channel coding is necessary. Second, OFDM has undesirable effect of occurrence of peak power which causes amplifier characteristics non linear to overcome these peak power reduction techniques are necessary.

By combing RS coding, and Adaptive clipping techniques, the system behavior is robust to channel conditions and peak power problem.

Considering from VLSI point of view, both RS coding and Adaptive clipping techniques are simple and less complex for implementation, so recommended OFDM system is also robust in terms of implementation.

Chapter 8

CONCLUSION

8. Conclusion

8.1 Achievement of Thesis

In this thesis, various WLAN standards and the system aspects are presented at first level. We then analyzed on multi carrier modulation scheme OFDM. Next, we have considered some of the parameters which effect the wireless transmission of data through OFDM. We have proposed a new system which is robust in terms of channel coding, peak power reduction, and complexity. The main disadvantage of peak power of OFDM based systems can be improved with PAPR techniques. This creates additional BER to the overall performance. Error correction schemes like Reed-Solomon coding helps in both PAPR BER degradations and also overall system performance. In this thesis, all the three techniques are combined for overall better system performance. Several details and results of the techniques used are presented in this thesis.

8.2 Scope of Further Research

- The proposed OFDM system is working well for AWGN channel, it can be tested to Rayleigh fading channels.
- The proposed system can be applied at 60 GHz, which is current research area of WLAN applications.
- The proposed system can be applied to MIMO OFDM, which is currently used in IEEE WLAN 802.11n standard for higher data rates.
- Other coding techniques like Trellis, Convoltuional, and Turbo coding techniques can be applied with adaptive clipping technique for PAPR reduction.

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