Palestine Polytechnic University

College of Engineering & Technology
Electrical & Computer Engineering Department

Graduation Project

WIMAX based on OFDM
(Simulation & Analysis of its parameters)

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بسم الله الرحمن الرحيم

جامعة بوليتكنك فلسطين
كلية الهندسة و التكنولوجيا
دائرة الهندسة الكهربائية و الحاسوب
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اسم المشروع:

WIMAX based on OFDM
(Simulation & Analysis of its parameters)

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بناء على نظام كلية الهندسة و التكنولوجيا و إشراف و متابعة المشرف المباشر على المشروع وموافقة أعضاء اللجنة الممتحنة ، تم تقديم هذا المشروع إلى دائرة الهندسة الكهربائية و الحاسوب وذلك استكمالاً لمتطلبات درجة البكالوريوس في تخصصي هندسة الاتصالات والإلكترونيات و هندسة أنظمة الحاسوب .

توقيع المشرف

....

توقيع اللجنة الممتحنة

....

توقيع رئيس الدائرة

....

DEDICATION

To Our Families and Friends

ACKNOWLEDGMENT

We feel grateful to Our supervisor:

Dr .Osama Ata for his great help
and supports

Thanks to our teachers

Our thanks to Palestine Polytechnic University

College of Engineering & Technology

Our thanks to Electrical & Computer Engineering Department

ABSTRACT

This project comes in response to the revolution of the broadband wireless communication systems. These systems are characterized by their ability to introduce high data rate with low BER (bit error rate) at low SNR (signal to noise ratio). WiMAX system is one of those systems that is used for high data rate-based applications such as internet browsing .

For this reason , we selected WiMAX as our graduation project. In this project ,WiMAX based on OFDM is simulated using Matlab packages in order to test its performance under real channel scenarios in term of BER versus SNR plots.

The real channel models that are used in the simulation are the SUI channels(Stanford University Interim). These models characterize the most environmental fading effects. They define three terrain types ; Soft fading environment (SUI 1 and SUI 2) intermediate fading environment (SUI 3 and SUI 4) and harsh fading environment (SUI 5 and SUI 6) .

In other words testing any wireless communication system under these SUI models means that testing the communication system under real channels as if the communication system is practically tested.

To arrive the best configuration of WiMAX that introduces the optimal performance under these channel models; All mandatory WiMAX parameters(i.e BPSK ,16-QAM, Reed Solomon encoding ,... etc) are used in the simulation of WiMAX system and their results in term of BER Vs SNR plots are analyzed in detail.

Optional parameters(256-QAM, diversity , MIMO ,512 subcarriers) and known channel estimation are also used in simulation of WiMAX system. Using these parameters improves the WiMAX system in both its data rate and its performance in term BER Vs SNR plots. Conclusions are then extracted from simulation results that specify the best configuration of WiMAX system . Finally , these results led us to put required recommendations and future works can be done in this hot area.

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List of Abbreviations

AES	Advanced Encryption Standard
AMC	Adaptive modulation and coding
AP	Access point
ARQ	Automatic Retransmission Request
AWGN	Additive white Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
BW	Bandwidth
BWA	broadband wireless access
CDMA	Code Division Multiple Access
CP	Cyclic prefix
DL	down link
DSL	Digital subscriber line
FDD	Frequency division duplex
FDM	Frequency division multiplexing
FEC	Forward error correction
FFT	Fast Fourier transform
FIPS	Federal Information Processing Standard
FSK	frequency shift keying
ICI	inter-carrier interference
IEEE	Institute of electrical and electronics engineers
IETF	Internet Engineering Task Force
IFFT	Inverse Fast Fourier transform
IP	Internet protocol
ISI	inter-symbol interference
LAN	local area network
LDPC	low-density parity check
LOS	line of sight
MAC	Medium Access control
MAN	Metropolitan Area Network
MCM	multi carrier modulation
MIMO	multiple input multiple output
MISO	multiple input single output
MPH	Mile per hour
NLOS	non-line of sight
OFDM	Orthogonal Frequency Division Mulitplexing
PAPR	Peak to average power ratio
PHY	Physical layer
PMP	point to multi point
PRBS	pseudo random binary shift
PTP	point to point
QAM	Quadrature amplitude modulation
QoS	Quality of service
SER	symbol error correction

SIMO	single input multiple output
SINR	Signal to Interference and Noise Ratio
SISO	single input single output
SNR	Signal to Noise Ratio
SOFDMA	Scalable Orthogonal Frequency Division Multiple Access
SUI	Stanford University Interim
TDD	Time division duplex
TDM	Time division multiplexing
UL	up link
VOIP	Voice over internet protocol
WiMAX	Worid wide interoperability for Microwave Access
WISP	wireless Internet service provider
WLL	Wireless local loop
3DES	Triple Data Encryption Standard

Preface

Chapter one :

Brief introduction to WiMAX system . Project objectives ,Importance of project ,time plan and project costs are introduced in this chapter.

Chapter two :

Architecture of WiMAX is explained in some of details , WiMAX types are also introduced . Finally ,WiMAX features are explained to show the advantages of WiMAX over the other communication technologies.

Chapter three :

OFDM system are introduced and how solves the ISI and spread delay . WiMAX PHY main block diagram and the sub-blocks .

Also the PHY layer of WiMAX based on OFDM is explained with showing the mandatory and optional parameters .

Our simulation parameters are then introduced.

Chapter four :

Diversity and MIMO are explained . Diversity is used to improve system performance , while STBC-MIMO improves the data rate and system performance.

Chapter five :

It introduces the flow charts of the simulation functions . In addition it explains the channel modeling and how these channel models are implemented in software.

Chapter sex :

It shows and discusses simulation results in order to specify WiMAX performance under different real channel models.

Chapter seven:

Conclusions and future works are introduced in this chapter.

Chapter 1

Introduction

Chapter One

Introduction

1.1 Overview

To fully understand WIMAX, we should comprehend its background, which is broadband wireless.

Broadband access provides faster web browsing and quicker file downloading It also enables several multimedia applications, such as real time audio and video streaming, multimedia conferencing, and interactive gaming[7]. These services are available using the Digital subscriber line DSL technology, cable modem technology, voice over Internet Protocol VoIP technology, etc [9] .

Wireless broadband is available in Internet cafés, local "hot spots" within many cities, private businesses and many homes [9]. The advantage of wireless broadband is that the computer receiving the Internet signal need not be connected by an Ethernet or network cable to the broadband modem or router [7]. A wireless broadband modem receives the service and transmits it via radio waves to the immediate surrounding area. Any computer within the receiving distance can pick up the signal, making the Internet portable [7] .

1.1.1 The Term "WIMAX"

The term WIMAX means: *Worldwide Interoperability for Microwave Access.* WIMAX is a promising, standards based technology for delivering advanced fixed and mobile broadband wireless services in showing high growth and developed markets.

WIMAX technologies are widely accepted as a cost effective and reliable solution for delivering advanced communications services [9]. WIMAX, in both of its Fixed and Mobile versions, is based on a next generation all IP core network, which offers low latency, advanced security, QoS (Quality of Service), and, in the case of mobility, worldwide roaming capabilities.[7]

1.1.2 The Development of WIMAX

WIMAX technology has developed through four stages:

1.1.2.1 Narrowband Wireless Local Loop Systems[7]

The wireless local loop WLL system, and the first application were developed and deployed was voice telephony, was successful in developing countries. In 1993, after the Internet was commercialized, several companies focused on providing wireless Internet - access. These wireless Internet service provider WISP companies deployed systems which required antennas to be installed at the customer premises, (on rooftops of their buildings, for example).

1.1.2.2 First generation line of sight (LOS) broadband systems[7]

Required that subscribers install at their premises outdoor antennas high enough and pointed toward the tower for a clear LOS transmission path. Since a fairly large area was being served by a single tower, the capacity of these systems was fairly limited. Similar systems were deployed internationally in the 2.5GHz and 3.5GHz band.

1.1.2.3 second generation non line of sight (NLOS) broadband systems

They were able to overcome the LOS issue and to provide more capacity, by using a cellular architecture and implementing advanced signal processing techniques to improve the system performance under multi path conditions. NLOS problem can be solved by using techniques such as orthogonal frequency division multiplexing OFDM, code division multiple access CDMA, and multi antenna processing [4].

1.1.2.4 standards based broadband wireless systems

In 1998, the Institute of Electrical and Electronics Engineers (IEEE) formed a group called 802.16 to develop a standard for what was called a wireless metropolitan area network, or wireless MAN. As it turns out, the IEEE 802.16 specifications are a collection of standards with a very broad scope. The IEEE developed the specifications but left to the industry the task of converting them into an interoperable standard that can be certified.

1.1.3 WiMAX Forum

The WIMAX Forum, an organization of more than 400 leading operators, communications component and equipment companies, was established in June 2001[9], to remove some of the barriers to wide scale adoption of Broadband Wireless Access BWA technology[9]. Along those lines, the Forum is working to certify interoperability and compatibility of the equipment based on the IEEE802.16 technology and to ensure that WIMAX Forum certified equipment meet service provider and customer requirements [9] .

To achieve its goals, the WIMAX Forum sets up a number of working groups to address the technical, marketing, regulatory and other requirements for wide scale adoption and deployment of broadband wireless systems [9] .

1.2 Project Importance

The importance of this project basically comes from the importance of the WIMAX system in the wireless systems in the world , what the revolution that had made & will make it in the future in the data transfer field.

This importance comes from showing & explaining the effects of the most important parameters on the system & its performance by using matlab codes. Finally this project will be the starting point for the related projects in our university.

1.3 Project Objectives

- Studying the architecture of the WIMAX, specially its physical layer, how it works and the most important parameters that have the dominant effects on the data transfer under certain conditions.
- Developing the matlab code to simulate the WIMAX system using the OFDM techniques to analyze the effects of a certain parameters such as (modulation ,cyclic prefix ,type of channel ,Bandwidth (BW), signal to noise ratio(SNR)) on the bit-error-rate(BER) , the throughput of the system & the spectral efficiency .

1.4 History Of WIMAX

In the mid 1990's, telecommunication companies developed the idea to use fixed broadband wireless networks for potential last mile solutions to provide alternative[9]. means to deliver Internet connectivity to businesses and individuals. Their aim was to produce a network with the speed, capacity, and reliability of a hardwired network, while maintaining with the flexibility, simplicity, and low costs of a wireless network [9] .

This standard, which was eventually released in 2001, operated on a point-to-point radio link network means line of sight transmissions, and had a frequency range of 10 GHz to 66 GHz. [1]

In 2001, the WIMAX Forum was established with the agenda to market and encourage the 802.16 standard. There they coined the term WIMAX (Worldwide Interoperability for Microwave Access)[9]. In 2003 the IEEE came out with 802.16a, which transmitted data through non-line of sight radio channels to and from omni-directional antennas. Later on, in 2004, the 802.16-2004 standard was released. This standard combined the updates from the IEEE 802.16a, 802.16b, and 802.16c regulations [9] .

This broadband system extended the WIMAX service to a 30-mile range . The IEEE did not stop there. In 2005, they came out with the first Mobile WIMAX system: 802.16e. This version used a Scalable Orthogonal Frequency-Division Multiple Access (SOFDMA) that supported over 2,000 sub carriers, optimized handover delay and packet loss, and increased network security [7] .

1.5 Fixed WIMAX

There are two different types of WIMAX services: Fixed and Mobile[7]. The fixed WIMAX attempts to provide a set of services similar to the traditional fixed line broadband but using wireless as the medium of transmission (as shown in figure1.2)[7]. It can be thought of as a competitive alternative to DSL or cable modem.

Fixed wireless offers several advantages over traditional wired solutions. These advantages include:[7]

- Lower entry and deployment costs.
- Faster and easier deployment.
- Ability to build out the network as needed.

- Lower operational costs for network maintenance, management, and operation.
- Independence from the compulsory carriers.

1.6 Mobile WIMAX

Mobile WIMAX takes the fixed wireless application a step further and enables applications, like cell phone, on a much larger scale. This type offers additional functionalities, such as portability, nomad city and mobility. The first step toward mobility would come by simply adding nomadic capabilities to fixed broadband. Providing WIMAX services to portable devices will allow users to experience bandwidth not just at home or Work but also at other locations. Users could take their broadband connection with them as they move around from one location to another. For example, mobile WIMAX enables streaming video to be broadcast from a speeding police or other emergency vehicle at over 70 MPH [7] .

In addition to higher speed Internet access, mobile WIMAX can be used to provide voice over IP services in the future[9]. The low latency design of mobile WIMAX makes it possible to deliver VoIP services effective [9].

1.7 Activities Description and Time Plane

In this part , we will state the project activities , the time period of each activity, the starting & ending time for each of them & the activity's dependence.

Table 1.1. Activities description (first semester)

Activity ID	Activity description
A1	Selecting of the project.
A2	Collecting a surveying information , abstract & project plan.
A3	Collecting detailed information with some analyzing
A4	Designing the **OFDM** system as a block diagrams , flowcharts of some blocks with some description.
A5	Writing documentation

Table 1.2. Time plan (first semester)

Week Activity	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
A1	▨	▨	▨	▨	▨	■										
A2							■	■								
A3								■	■	▨	▨	▨	▨	▨		
A4										▨	▨	▨	▨	▨	▨	
A5									▨	▨	▨	▨	▨	▨	▨	▨

Table 1.3. Activities description (second semester)

Activity ID	Activity description
A6	Drawing the flowchart of all OFDM system blocks.
A7	Drawing the matlab function codes of all system components with testing each of them.
A8	Testing the main program that provides the initialization & calling the functions.
A9	Analyzing the most important system parameters under different conditions using the program.
A10	Concluding the results so that using them in studying system efficiency & how to improve it.
A11	What should be done to get the optimum efficiency
A12	Writing the report

Table 1.4. Time plan (second semester)

Week Activity	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
A6	▨	▨	▨	▨												
A7		▨	▨	▨	▨	▨										
A8						▨	▨									
A9							▨	▨	▨	▨			▨	▨	▨	
A10								▨	▨	▨	▨	▨		▨	▨	
A11															▨	▨
A12	▨	▨	▨	▨	▨	▨	▨	▨	▨	▨	▨	▨	▨	▨	▨	▨

1.8 Estimated Costs

In this part we will estimate the total project cost dividing into three parts as follow :

1) Software cost.
2) Hardware cost.

1.8.1 Software resources

The following table contains the software that will be used in our project & its cost

Table 1.5. Software cost

Software component	Cost
Microsoft Windows XP Professional	Available in the university LAB.
Microsoft Office 2007 enterprise	Available in the university LAB.
Microsoft Office Visio 2007	Available in the university LAB.
MATLAB 7.0.1	Available in the university LAB.
Total	**$0**

1.8.2. Hardware resources

The following table contains the hardware that will be in our project and their Costs.

Table 1.6. Hardware resources cost

Hardware component	Cost
1 PC with core 2 duo CPU 2.6 GHz, 160 GB HDD, 2 GB RAM, monitor, keyboard and mouse.	Available in the university LAB.
Printing cost	$25
Total	**$25**

Chapter 2

WiMAX

2.1. Definition of WiMAX

2.2. Features of WiMAX

2.3. Modulation and Coding in WiMAX

2.4. Network architecture

2.5. Security

Chapter Two

WiMax

This chapter provides an illustrative background for the wimax technology and IEEE 802.16d standard. Also shows the architecture of WiMAX

2.1 Definition of WiMAX

2.1.1 WiMAX technology

The term WIMAX means: ***Worldwide Interoperability for Microwave Access.***

Fixed WiMAX, based on the IEEE 802.16-2004 Air Interface Standard, has proven to be a cost- effective fixed wireless alternative to cable and DSL services.

2.2 Features of WiMAX

WiMAX offers a set of features with a lot of flexibility in terms of deployment options and potential service offerings [7] .

These features are :

2.2.1 OFDM based physical layer

The WiMAX physical layer (PHY) is based on orthogonal frequency division multiplexing, a scheme that offers good resistance to multipath, and allows WiMAX to operate in NLOS conditions[4].

2.2.2 Very high peak data rates

WiMAX is capable of supporting very high peak data rates. In fact, the peak PHY data rate can be as high as 74Mbps when operating using a 20MHz2 wide spectrum [9]. More typically, using a 10MHz spectrum operating using TDD (Time division duplex) scheme with a 3:1 downlink-to-uplink ratio, the peak PHY data rate is about 25Mbps and

6.7Mbps for the downlink and the uplink, respectively[7]. These peak PHY data rates are achieved when using 64 QAM modulation with rate 5/6 error-correction coding[7]. Under very good signal conditions, higher peak rates may be achieved using multiple antennas and spatial multiplexing [4] .

2.2.3 Scalable bandwidth and data rate support

WiMAX has a scalable physical-layer architecture that allows for the data rate to scale easily with available channel bandwidth[2]. This scalability is supported in the OFDMA mode, where the FFT (fast Fourier transform) size may be scaled based on the available channel bandwidth. For example, a WiMAX system may use 128-,512-, or 1,048-bit FFTs based on whether the channel bandwidth is 1.25MHz, 5MHz, or10MHz, respectively. This scaling may be done dynamically to support user roaming across different networks that may have different bandwidth allocations [7] .

2.2.4 Adaptive modulation and coding (AMC)

WiMAX supports a number of modulation and forward error correction (FEC) coding schemes and allows the scheme to be changed on a per user and per frame basis, based on channel conditions[1]. AMC is an effective mechanism to maximize throughput in a time-varying channel. The adaptation algorithm typically calls for the use of the highest modulation and coding scheme that can be supported by the signal-to-noise and interference ratio at the receiver such that each user is provided with the highest possible data rate that can be supported in their respective links [7] .

2.2.5 Link layer retransmissions

WiMAX supports automatic retransmission requests (ARQ) at the link layer. ARQ-enabled connections require each transmitted packet to be acknowledged by the receiver; unacknowledged packets are assumed to be lost and are retransmitted[7]. WiMAX also optionally supports hybrid-ARQ, which is an effective hybrid between FEC and ARQ [9] .

2.2.6 Support for TDD and FDD

IEEE 802.16-2004 and IEEE 802.16e-2005 supports both time division duplexing and frequency division duplexing, also a half-duplex FDD, which allows for a low-cost system implementation. TDD is favored by a majority of implementations because of its advantages [7] :

(1) flexibility in choosing uplink-to-downlink data rate ratios.

(2) ability to exploit channel reciprocity.

(3) ability to implement in nonpaired spectrum.

(4) less complex transceiver design.

All the initial WiMAX profiles are based on TDD, except for two fixed WiMAX profiles in 3.5GHz [9] .

2.2.7 Support for advanced antenna techniques

The WiMAX solution allows for the use of multiple-antenna techniques, such as beamforming, space-time coding, and spatial multiplexing. These schemes can be used to improve the overall system capacity and spectral efficiency by deploying multiple antennas at the transmitter and/or the receiver [1] .

2.2.8 Quality of service support

The WiMAX MAC layer has designed to support a variety of applications, including voice and multimedia services. The system offers constant bit rate, variable bit rate, real-time, and non-real-time traffic flows, in addition to best-effort data traffic. WiMAX MAC is designed to support a large number of users, with multiple connections per terminal, each with its own QoS requirement [9] .

2.3 Modulation and Coding in WiMAX

WiMAX supports a variety of modulation and coding schemes and allows for the scheme to change on a burst-by-burst basis per link , depending on channel conditions[9]. Using the channel quality feedback indicator, the mobile can provide the base station with feedback on the downlink channel quality [7]. For the uplink, the base station can estimate the channel quality, based on the received signal quality[7]. The base station can take into account the channel quality of each user's uplink and downlink and assign a modulation and coding scheme that maximizes the throughput for the available signal-to-noise ratio.[2]

FEC coding using convolutional codes is mandatory[1]. Convolution codes are combined with an outer Reed-Solomon code in the downlink for OFDM-PHY[1].

The standard optionally supports turbo codes and low-density parity check (LDPC) codes at a variety of code rates [1].

2.4 Network architecture

2.4.1 WiMAX architecture

WiMax base station is connected to public networks using optical fiber ,cable, microwave link, or any other high-speed (point-to-point)**PTP** connectivity, referred as a backhaul [11] . in few cases such as mesh networks,(point-to-multi point)**PMP** connectivity is also used as a backhaul. WiMAX should use **PTP** antennas as base stations across long distances [7] .

A base station servers subscriber station also called customer premise equipment **CPE** [9]for obvious reasons ,using non line of sight or line of sight point-to-multi point connectivity and this connectivity is referred to as the last mile .

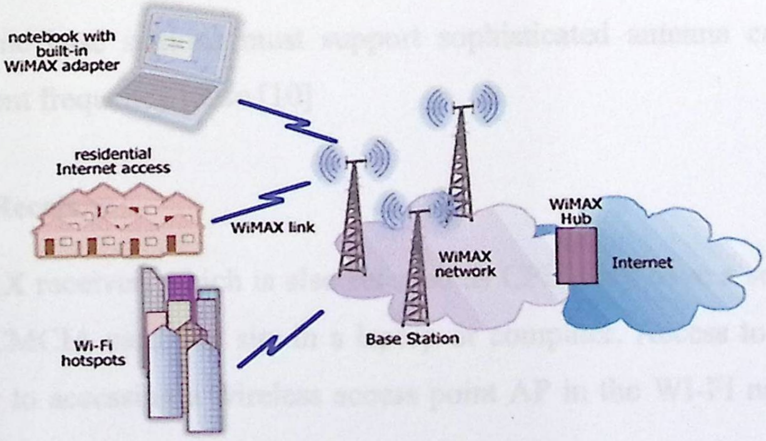

Figure 2.1 – WIMAX Network

Ideally WiMAX should use **NLOS** point-to-multi point antennas to connect subscribers to the base station , a subscriber station typically serves a building business or residential using wired or wireless LAN . WiMAX system consists of two major parts , WiMAX base station and a WiMAX receiver [10] .

2.4.2 WiMAX Base Station

A WIMAX base station consists of indoor electronics and a WIMAX tower. Abase station can cover up to 6 mile radius (theoretically, abase station can cover up to 50 Km radius or 30 mile, but particle considerations limit it to about 10Km or 6 mile) and use the media access control layer (**MAC**) defined in the standard (a common interface that makes the networks interoperable) and allocate uplink and downlink bandwidth to subscribers according to their needs.

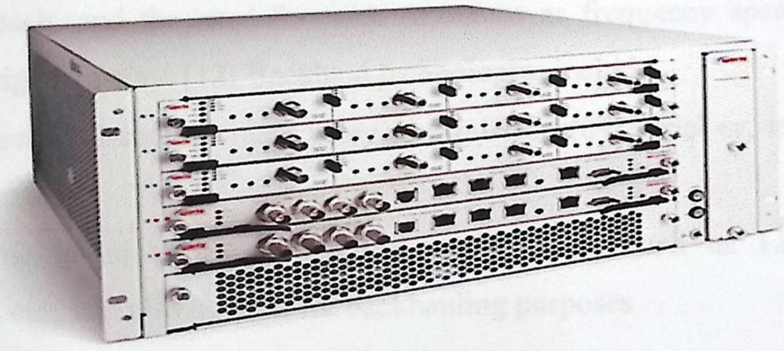

Figure 2.2 – WIMAX Base Station

WIMAX base station can range from units that support only a few subscriber stations to supports thousands of subscriber stations and provides many carrier-class features[11]. Some base stations must support sophisticated antenna capabilities and implement efficient frequency reuse.[10]

2.4.3 WiMAX Receiver

A WIMAX receiver, which is also referred as CPE, may have a separate antenna or could be a PCMCIA card that sits in a laptop or computer. Access to WIMAX base station is similar to accessing a wireless access point AP in the WI-FI network, but the coverage is larger.

Depending on the end-user needs. Three different types of CPEs [9] :

- A modem attached to an external antenna.
- A modem with an indoor antenna.
- Integrated antenna, can be integrated in to laptops, phones, and other devices.

2.4.4 WiMAX Backhaul

In a hierarchical telecommunications network the backhaul part of the network includes the intermediate links between the core, or backbone, of the network and the small sub networks at the "edge" of the entire hierarchical network. For example, while cell phones communicating with a single cell tower constitute a local sub network, the connection between the cell tower and the rest of the world begins with a backhaul link to the core of the telephone company's network (via a point of presence).

The choice of backhaul technology must take account of such parameters as capacity, cost, reach, and the need for such resources as frequency spectrum, optical fiber, wiring, or rights of way. [12] Backhaul technologies include:

- Point-to-point microwave radio relay transmission (terrestrial or, in some cases, by satellite)
- Point-to-multipoint microwave access technologies, such as LMDS, Wi-Fi, WiMAX, etc., can also be used for backhauling purposes
- DSL variants, such as ADSL and SHDSL
- PDH and SDH/SONET interfaces, such as (fractional) E1/T1, E3, T3, STM-1/OC-3, etc.
- Ethernet

2.4.5 WiMAX LOS (Line-Of-Sight)

A signal travels over a direct and unobstructed path from the transmitter to the receiver ,a LOS link requires that most of the first Fresnel zone is free of any obstruction, as shown in Figure 2.3

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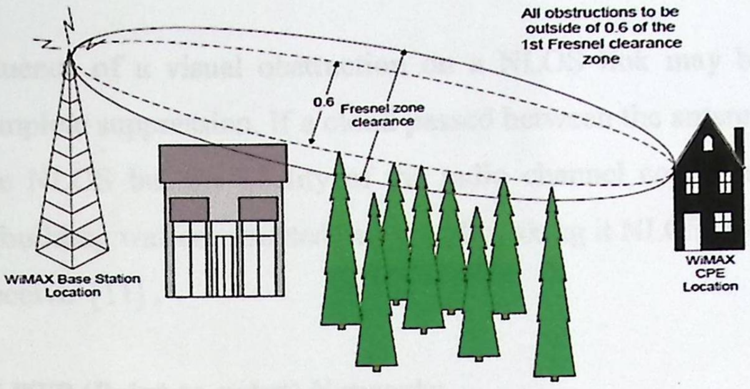


Figure 2.3 – WIMAX LOS Fresnel zone

The Fresnel clearance required depends on the operating frequency and the distance between the transmitter and receiver locations [10] .

2.4.6 WiMAX NLOS (Non-Line-Of-Sight)

Non-line-of-sight (NLOS) or near-line-of-sight is a term used to describe radio transmission across a path that is partially obstructed, usually by a physical object in the Fresnel zone.[7]

Many types of radio transmissions depend, to varying degrees, on line of sight between the transmitter and receiver. Obstacles that commonly cause NLOS conditions include buildings, trees, hills, mountains, and, in some cases, high voltage electric power lines. Some of these obstructions reflect certain radio frequencies, while some simply absorb the signals; but, in either case, they limit the use of many types of radio transmissions, including most of those used for Wi-Fi [7] .

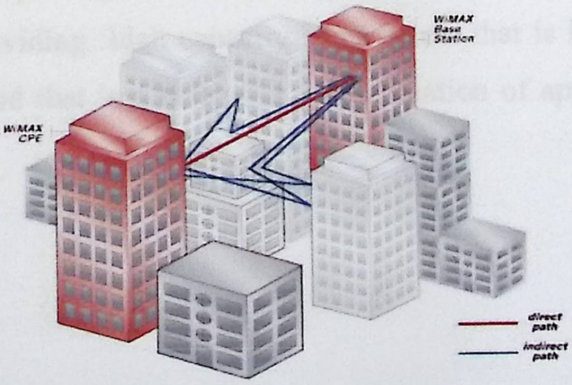


Figure 2.4 – WIMAX NLOS propagation

The influence of a visual obstruction on a NLOS link may be anything from negligible to complete suppression. If a cloud passed between the antennas the link could actually become NLOS but the quality of the radio channel could be unaffected. If, instead, a large building was constructed in the path making it NLOS, the channel may be impossible to receive [11] .

2.4.7 WiMAX PTP (Point-to-point) Networks

Point-to-point networks are fixed wireless network they offer high speed dedicated links between high density nodes in a network and the system that used point-to-point[9], it provide an effective last mile solution for the existing provider and can be used to deliver services directly to end users.

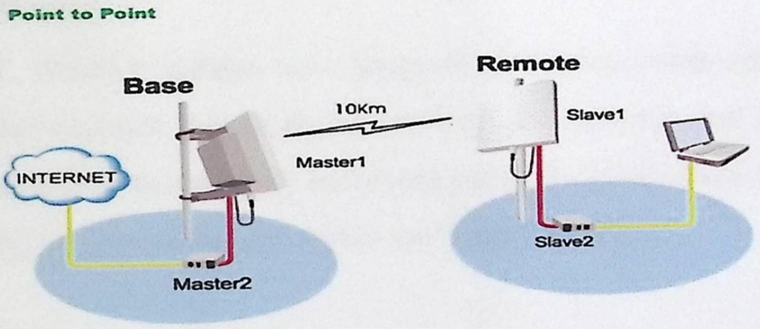


Figure 2.5 – WIMAX (Point-to-point) Networks

2.4.8 WiMAX PTMP (Point-to-multi point) Networks

In point-to-multi point system multiple subscribers can access the same radio platform using a multiplexing method and queuing .this technology offer service providers method of providing high capacity local access that is less cost and faster to deploy than wire line and that is a belt offer a combination of application as shown in Figure 2.6 [11] .

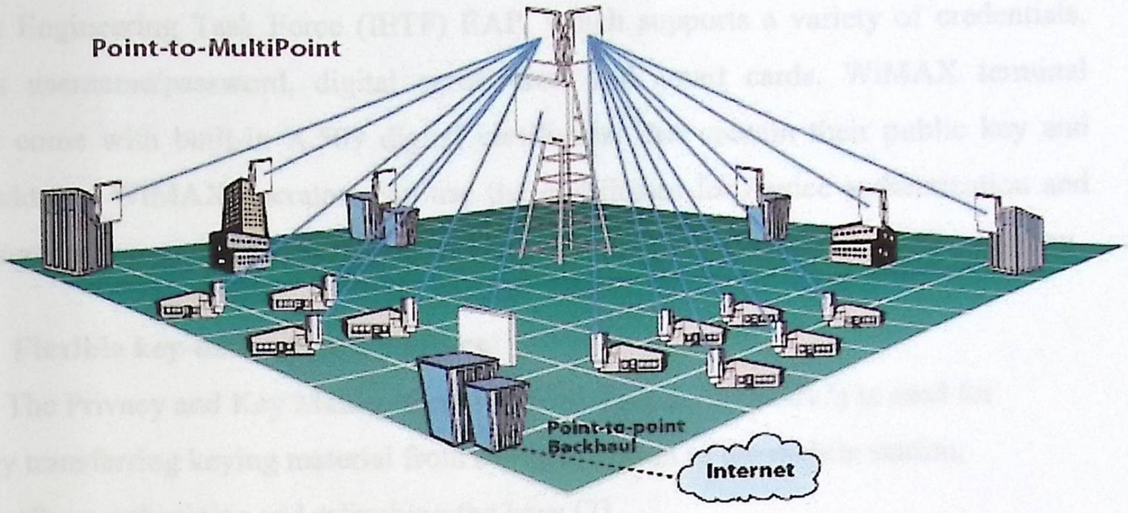


Figure 2.6 – WIMAX (Point-to- multi point) Networks [12]

2.5 Security in Wimax

Unlike Wi-Fi, WiMAX systems were designed at the outset with robust security in mind. The standard includes state-of-the-art methods for ensuring user data privacy and preventing unauthorized access, with additional protocol optimization for mobility. Security is handled by a privacy sub layer within the WiMAX MAC.

2.5.1 The key aspects of WiMAX security

2.5.1.1 Support for privacy

User data is encrypted using cryptographic schemes of proven robustness to provide privacy. Both AES (Advanced Encryption Standard) and 3DES (Triple Data Encryption Standard) are supported[1]. Most system implementations will likely use AES, as it is the new encryption standard approved as compliant with Federal Information Processing Standard (FIPS) and is easier to implement[10]. The 128-bit or 256-bit key used for deriving the cipher is generated during the authentication phase and is periodically refreshed for additional protection [2] .

2.5.1.2 Device/user authentication

WiMAX provides a flexible means for authenticating subscriber stations and users to prevent unauthorized use.[5] The authentication framework is based on the

Internet Engineering Task Force (IETF) EAP, which supports a variety of credentials, such as username/password, digital certificates, and smart cards. WiMAX terminal devices come with built-in X.509 digital certificates that contain their public key and MAC address. WiMAX operators can use the certificates for device authentication and use a username/password or smart card authentication on top of it for user authentication.

2.5.1.3 Flexible key-management protocol

The Privacy and Key Management Protocol Version 2(PKMv2) is used for securely transferring keying material from the base station to the mobile station, periodically reauthorizing and refreshing the keys.[7]

PKM is a client-server protocol The MS acts as the client; the BS, the server. PKM uses X.509 digital certificates and RSA (Rivest- Shamer-Adleman) public-key encryption algorithms to securely perform key exchanges between the BS and the MS. [8]

2.5.1.4 Protection of control messages

The integrity of over-the-air control messages is protected by using message digest schemes, such as AES-based CMAC or MD5-based HMAC.[9]

2.5.1.5 Support for fast handover

To support fast handovers, WiMAX allows the MS to use preauthentication with a particular target BS to facilitate accelerated reentry. A three-way handshake scheme is supported to optimize the reauthentication mechanisms for supporting fast handovers, while simultaneously preventing any man-in-the-middle attacks.

Chapter 3

OFDM system and WiMAX PHY based on OFDM

3.1. Introduction.

3.2. Orthogonal Frequency Division Multiplexing (OFDM) system.

3.3. WiMAX PHY based on OFDM .

3.4. OUR Model .

Chapter Three

OFDM system and WiMAX PHY based on OFDM

3.1Introduction

3.1.1 Project Objectives

- Studying the architecture of the WIMAX, specially its physical layer, how it works and the most important parameters that have the dominant effects on the data transfer under certain conditions.

- Developing the matlab code to simulate the WIMAX system using the OFDM techniques to analyze the effects of a certain parameters such as (modulation ,cyclic prefix ,type of channel ,Bandwidth (BW), signed to noise ratio(SNR)) on the bit-error-rate(BER) , the throughput of the system & the spectral efficiency .

3.1.2 system's Definition

When stream of data is transmitted from transmitter to receiver the following steps are involve :

1. Supply the system with stream data ,the data may be text file ,video, sound signal …etc. In our project we want to use random data by generating it using matlab functions .
2. The transmitter first converts the input data from a serial stream to parallel sets. Each set of data contains one symbol, S_i, for each subcarrier. For example, a set of ten data would be [S_0 S_1 S_2 …….S_9].
3. Data set is arranged on the horizontal axis in the frequency domain as shown in Figure 3.1 This symmetrical arrangement about the vertical axis is necessary for using the IFFT to manipulate this data.

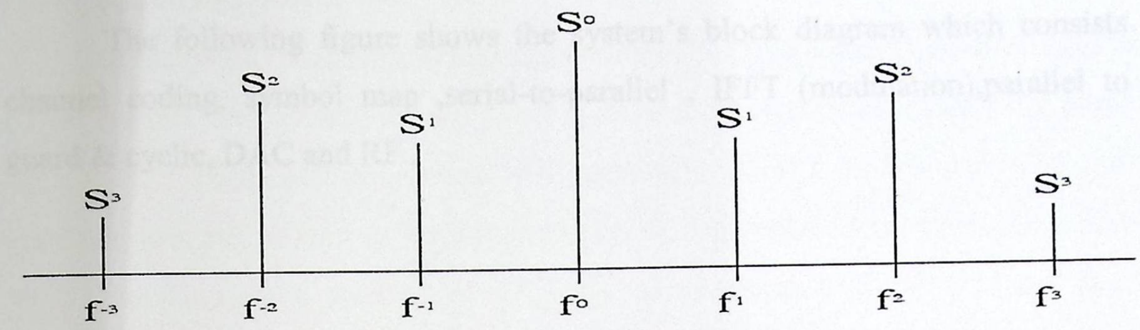
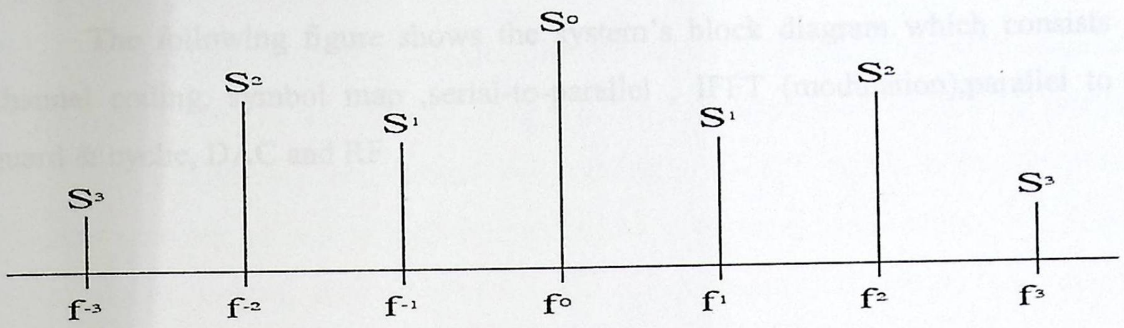


Figure 3.1 Frequency Domain Distribution of Symbols

4. An inverse Fourier transform converts the frequency domain data set into samples of the corresponding time domain representation of this data. Specifically, the IFFT is useful for OFDM because it generates samples of a waveform with orthogonal frequency components.
5. The parallel to serial block creates the OFDM signal by sequentially outputting the time domain samples.
6. The channel simulation will allow examination of the effects of noise, multi path, and clipping. By adding random data to the transmitted signal, simple noise can be simulated.
7. Multi path simulation involves adding attenuated and delayed copies of the transmitted signal to the original. This simulates the problem in wireless communication when the signal propagates on many paths.
8. Clipping simulates the problem of amplifier saturation. This addresses a practical implementation problem in OFDM where the peak to average power ratio is high.
9. The receiver performs the inverse of the transmitter. First, the OFDM data are split from a serial stream into parallel sets.
10. The Fast Fourier Transform (FFT) converts the time domain samples back into a frequency domain representation.
11. The magnitudes of the frequency components correspond to the original data and The parallel to serial block converts this parallel data into a serial stream to recover the original input data.



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3.1.3 System's Block Diagram

The following figure shows the system's block diagram which consists of the channel coding, symbol map ,serial-to-parallel , IFFT (modulation),parallel to serial, guard & cyclic, DAC and RF .

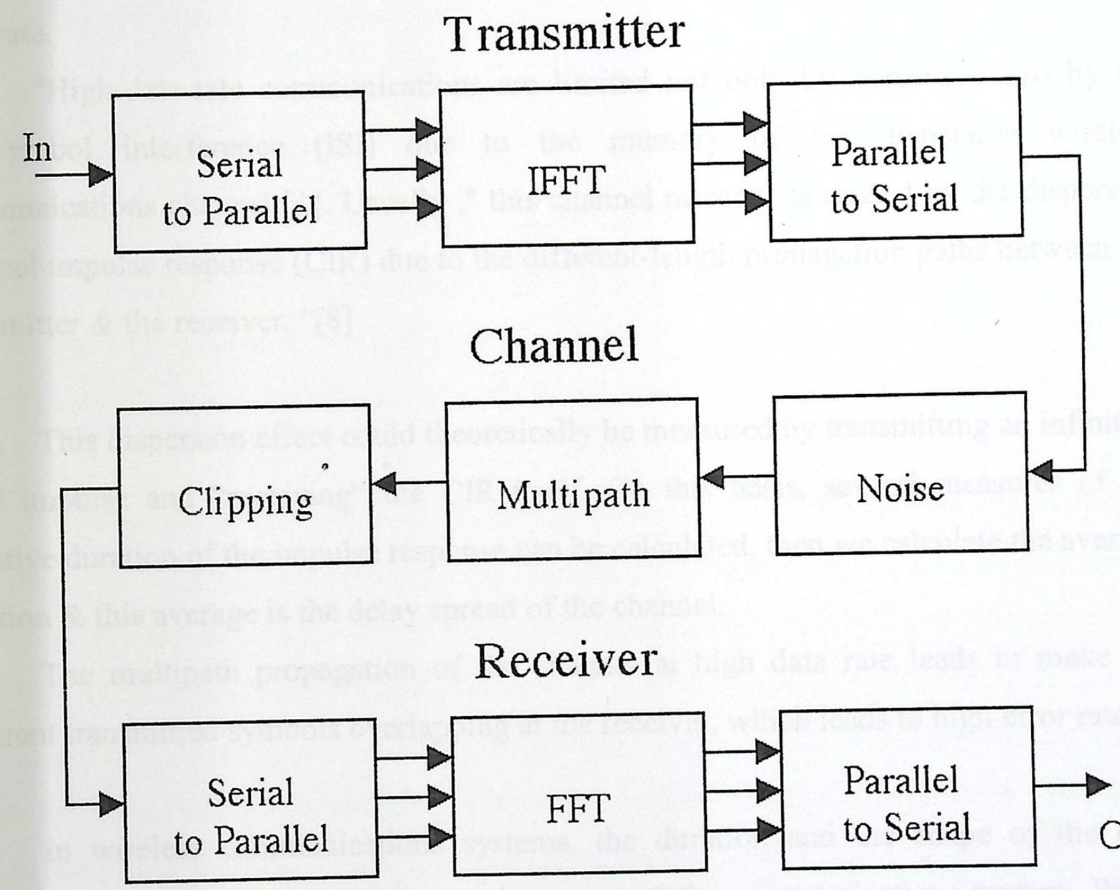

Figure 3.2 system block diagram

3.1.4 Brief introduction

In wireless broadband systems, the channel delay spread τ becomes large multiple of the symbol time ***Ts***, and the ***ISI*** (inter symbol interference) becomes a big problem. By definition, a high-data-rate system will generally have $\tau \gg Ts$, since the number of symbols sent per second is high (bandwidth>> coherence bandwidth). In a non-line of sight (**NLOS**) system, such as WiMAX, the delay spread will also be large due to high data rate.

"High-data-rate communications are limited not only by noise but also by the intersymbol interference (ISI) due to the memory of the dispersive wireless communications channel [1]. Usually ," this channel memory is caused by the dispersive channel impulse response (CIR) due to the different-length propagation paths between the transmitter & the receiver. "[8]

This dispersion effect could theoretically be measured by transmitting an infinitely short impulse and "receiving" the CIR itself. On this basis, several measures of the effective duration of the impulse response can be calculated, then we calculate the average duration & this average is the delay spread of the channel.

The multipath propagation of the channel at high data rate leads to make the different transmitted symbols overlapping at the receiver, which leads to high error rate.

In wireless communications systems, the duration and the shape of the CIR depend heavily on the propagation environment of the communications system. While indoor wireless networks typically suffer only short relative delays, outdoor networks, like the global system of mobile communications , can face delay spreads on the order of 15 µs.[9]

As a general rule, the effects of ISI on the transmission are not considerable as long as the delay spread is significantly shorter than the duration the transmitted symbol. This leads to make the symbol rate of communications systems is practically limited by the spread delay.

If symbol rates exceeding this limit are to be transmitted over the channel, an efficient techniques must be implemented in order to overcome the effects of **ISI.** Channel equalization techniques can be used to relatively cancel the effects caused by the

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If symbol rates exceeding this limit are to be transmitted over the channel, an efficient techniques must be implemented in order to overcome the effects of **ISI.** Channel equalization techniques can be used to relatively cancel the effects caused by the channel [8]. In order to perform this operation, the CIR must be estimated. Significant research efforts were invested into the development of such channel equalizers, and most wireless systems in operation use equalizers to decrease the ISI.

But there is an alternative approach toward transmitting data over a multipath channel. Instead of attempting to cancel the effects of the channel, orthogonal frequency-division multiplexing **(OFDM)** that uses a set of subcarriers in order to transmit information symbols in parallel over the channel."[8]

3.2 OFDM system

Orthogonal frequency division multiplexing (**OFDM**) is a multicarrier modulation technique that has recently used in high-data-rate communication systems, including **DSL** (Digital Subscriber Lines), wireless **LAN**s , digital video broadcasting, and now **WIMAX** and other wireless broadband systems such as the fourth generation cellular systems.

"Orthogonal frequency division multiplexing (OFDM) is a an efficient technique for achieving high data rate and overcoming multipath fading in wireless communications. OFDM can be thought of as a hybrid of multi-carrier modulation (MCM) and frequency shift keying (FSK) modulation. MCM is the principle of transmitting data by dividing the stream into several parallel bit streams and modulating each of these data streams onto individual carriers or subcarriers; FSK modulation is a technique where data is transmitted on one carrier from a set of orthogonal carriers in each symbol duration.

Orthogonality among the carriers is achieved by separating the carriers by an integer multiples of the inverse of symbol duration of the parallel bit streams. With OFDM, all the orthogonal carriers are transmitted simultaneously." In other words, the entire allocated channel is occupied through the aggregated sum of the narrow orthogonal sub-bands. By transmitting several symbols in parallel, the symbol duration is increased proportionately, which reduces the effects of ISI caused by the dispersive Rayleigh-fading environment."

In (OFDM) a single high-rate bit stream is converted to many low-rate **L** parallel bit streams and each parallel bit stream is modulated on one of N subcarriers. Each subcarrier can be modulated differently, typically using bi-phase shift keying (BPSK), quadrature phase shift ing (QPSK), or quadrature amplitude modulation (16QAM or 64QAM). To achieve high dwidth efficiency, the spectrum of the sub-carriers are closely spaced and overlapped, ere the nulls of each subcarrier's spectrum fall exactly on the centers of all other carriers. This makes them .orthogonal..

The following figures represent a simple multicarrier transmitter and receiver, where a gh-rate stream of ***R*** bps is broken into ***L*** parallel streams, each with rate ***R/L*** and then ltiplied by a different carrier frequency. Then at the receiver, each subcarrier is decoded parately, requiring ***L*** independent receivers.

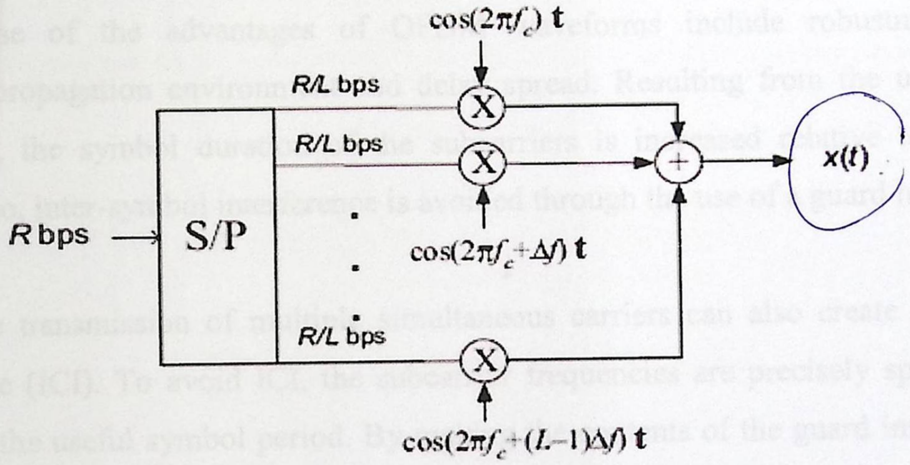

Figure 3.3 - Multicarrier transmitter

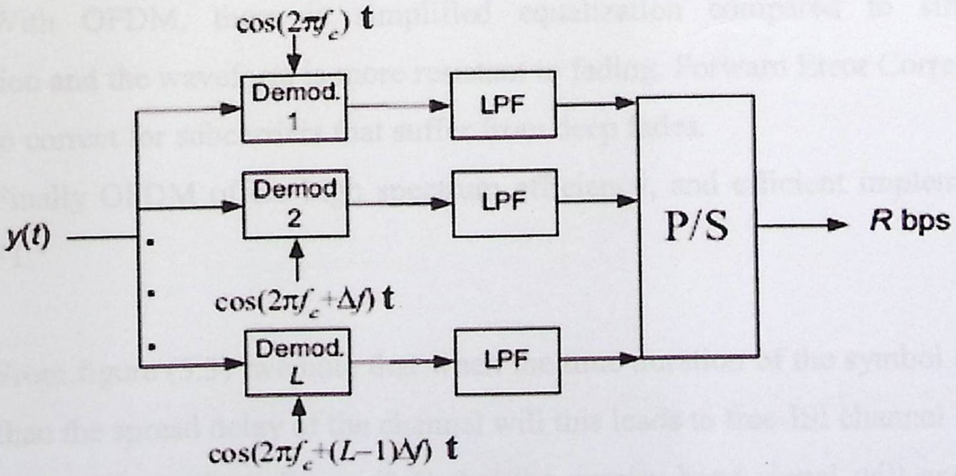

Figure 3.4 - Multicarrier receiver

The transmitted multicarrier signal experiences approximately flat fading on each subchannel, since *(B/L << Bc)*, despite the entire bandwidth experiences frequency-selective fading: *(B> Bc)*. Although this simple type of multicarrier modulation is easy to understand, it has several flaws. Very high quality and expensive low-pass filters will be required to maintain the orthogonality of the subcarriers at the receiver. This scheme requires *L* independent **RF** units and demodulation paths. later we show how **OFDM** overcomes these flaws.

3.2.1 OFDM Waveform Advantages (Why we use OFDM)

Some of the advantages of OFDM waveforms include robustness against multipath propagation environment and delay spread. Resulting from the use of many subcarriers, the symbol duration of the subcarriers is increased relative to the delay spread. Also, inter-symbol interference is avoided through the use of a guard interval.

The transmission of multiple simultaneous carriers can also create inter-carrier interference (ICI). To avoid ICI, the subcarrier frequencies are precisely spaced by the inverse of the useful symbol period. By making the contents of the guard interval a data repeated from the end of the useful symbol period(called cyclic prefix), then a time window of length equal to the useful symbol period can be applied & vary its position as much as the guard interval to recover the complete symbol without intersymbol interference.

With OFDM, there is simplified equalization compared to single carrier modulation and the waveform is more resistant to fading. Forward Error Correction (FEC) is used to correct for subcarriers that suffer from deep fades.

Finally OFDM offers high spectrum efficiency, and efficient implementation by IFFT/FFT."

From figure (3.3) ,we note that when the time duration of the symbol S(t) is much greater than the spread delay of the channel will this leads to free-ISI channel .

Also we note from figure (3.4) that the narrow band signal will experience flat fading while the wide band signal (high data rate) will experience frequency selective or frequency dependent & this leads to completely distort the signal since the bandwidth of the signal is greater than the coherence bandwidth.

For this, OFDM is considered as the most efficient method to solve the ISI & the time dispersion because of dividing the whole bandwidth of the signal into sub-channels (sub-carriers) that each of them has a bandwidth much smaller than the coherence bandwidth.

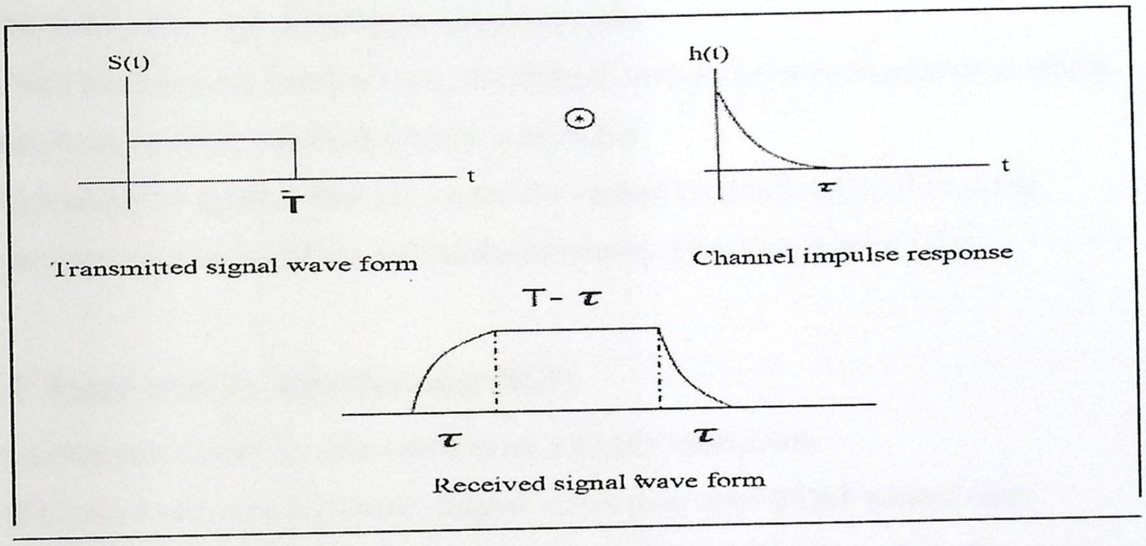

Figure3.5 **Channel and Pulse in time**

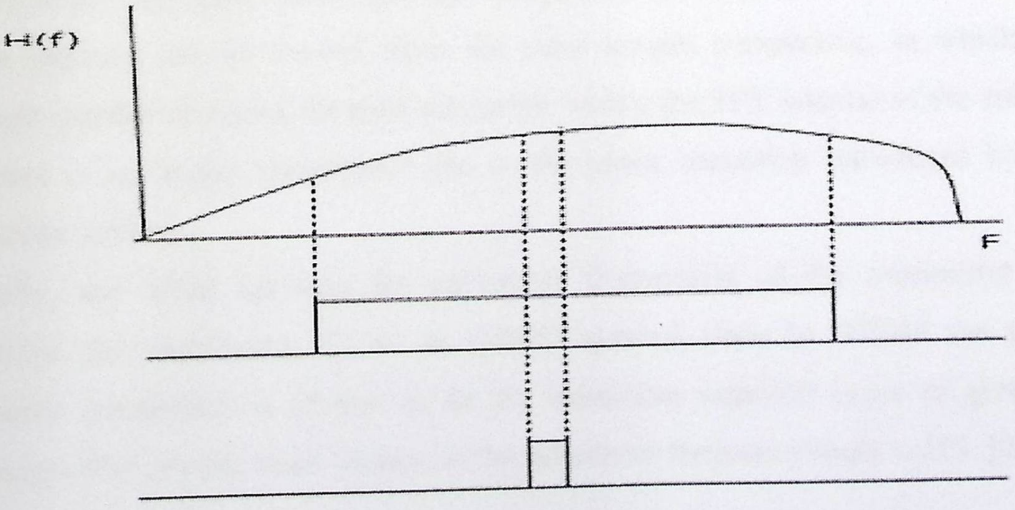

Figure 3.6 **Channel and Pulse in frequency domain** ,shows Channel Frequency Response H(f) & High rate signal spectrum (narrow pulse).& Low rate Signal spectrum (wide pulse)

3.2.2Inter-symbol interference (ISI) and inter-carrier interference (ICI) :

3.2.2.1 Inter-symbol interference (ISI)

- Refers to interference of an OFDM symbol is overlapped by previous OFDM symbols.
- In a multipath environment, a transmitted symbol takes different delays to reach the receiver through different propagation paths.
- From the receiver's point of view, the channel introduces time dispersion in which the duration of the received symbol is extended.
- Extending the symbol duration causes the current received symbol to overlap previous received symbols and results in inter-symbol interference (ISI).

3.2.2.2 Inter-carrier interference (ICI)

- Interference caused by data symbols on adjacent subcarriers.
- ICI occurs when the multipath channel varies over one OFDM symbol time.
- When this happens, the Doppler shifts on each multipath component cause a frequency shift of the subcarriers, resulting in the loss of orthogonality among them; since this shift differs from one component to another .
- This situation can be viewed from the time domain perspective, in which the integer number of cycles for each subcarrier within the FFT interval of the current symbol is no longer maintained due to the phase transition introduced by the previous symbol.
- Finally, any offset between the subcarrier frequencies of the transmitter and receiver also introduces ICI to an OFDM symbol since in OFDM the space between subcarriers is chosen to be the minimum required space to give the orthogonality ,so any small change in the subcarrier frequency leads to ICI [2] .

3.2.2.3 How to avoid interference

- Increase the number of subcarriers to ensure that the multipath channel does not vary over one OFDM symbol time.
- A smart choice of carrier frequencies can eliminate the interference between sub channels.

☐ Assume that we have two carriers exp(j2πf1t) and exp(j2πf2t) and the symbol duration is Ts .

☐ The received signal is x_1(t)exp(j2πf_1t) + x_2(t)exp(j2πf_2t) . In order to get x_1(t), multiply the received signal by exp(−j2πf1t) and integrate over Ts.

$$y = \frac{1}{Ts}\int_0^{Ts}(x_1(t) + x_2(t)\, e^{j2\pi(f_2-f_1)t})dt \tag{3.1}$$

☐ . If we assume x_1(t) and x_2(t) are constant over Ts. Then we get :

$$y= x_1 + \frac{1}{T_s}\int_0^{Ts} x_2\,(t)e^{j2\pi(f_2-f_1)t})dt \tag{3.2}$$

$$y=x_1 + \frac{x_2}{T_s}.\frac{\exp(j2\pi(f_2-f_1)T_S)-1}{j2\pi(f_2-f_1)} \tag{3.3}$$

☐ To eliminate the interference on x_1, we want the second term to be zero. This occurs when exp(j2π(f2−f1)Ts)=1. That is 2π(f2−f1)Ts=2πm ,where m is integer.

☐ The smallest separation between the two carriers that satisfies orthogonality that eliminates inter-carrier interference (ICI) is f2−f1=1/Ts .

3.2.3 Orthogonality of OFDM

☐ In OFDM, the spectra of subcarriers overlap but remain orthogonal to each other.

☐ This means that at the maximum of each subcarrier spectrum, all the spectra of other subcarriers are zero (zero crossing).

☐ The receiver samples data symbols on individual subcarriers at the maximum points and demodulates them free from any interference from the other subcarriers and thus no ICI.

☐ The orthogonality of subcarriers can be viewed in either the time domain or in frequency domain.

1. From the time domain perspective, each subcarrier is a sinusoid with an integer number of cycles within one FFT interval.
2. From the frequency domain perspective, this corresponds that each subcarrier have the maximum value at its own center frequency and zero at the center frequency of the other subcarriers as shown in the figure below.

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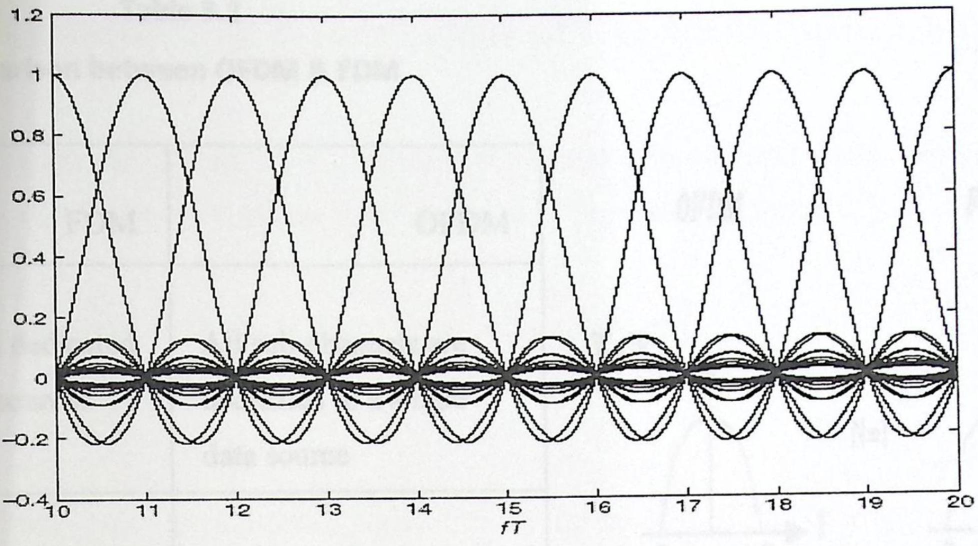


Figure (3.7) Orthogonal overlapping spectral shapes for an OFDM signal

3.2.4 Comparing FDM to OFDM

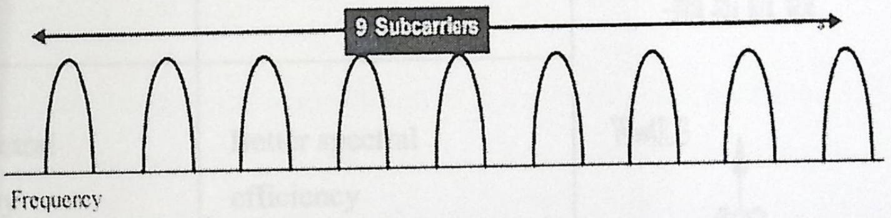


Figure (3.8) Spectrum using FDM as multicarrier technique

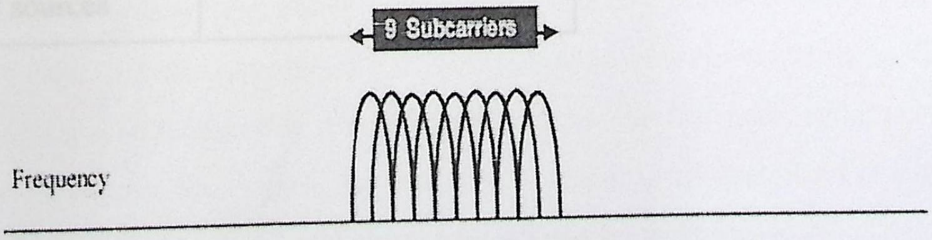
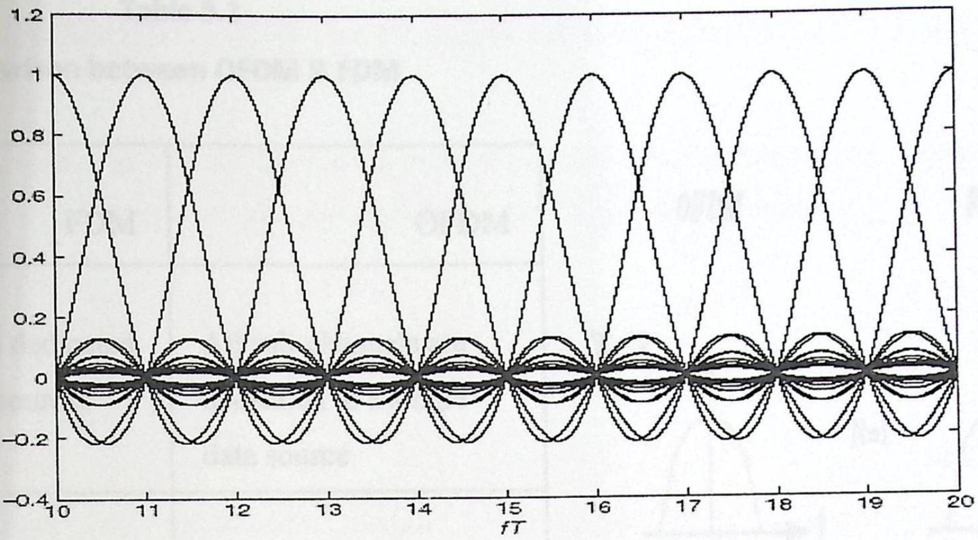


Figure (3.9) Spectrum using OFDM as multicarrier technique



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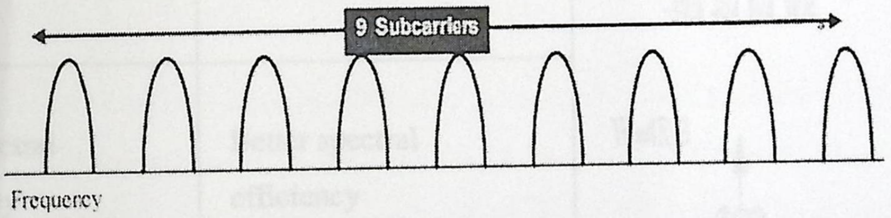


Figure (3.8) Spectrum using FDM as multicarrier technique

Figure (3.9) Spectrum using OFDM as multicarrier technique

Table 3.1

comparison between OFDM & FDM

FDM	OFDM
Bandwidth dedicated to several sources	All sub-channels are dedicated to a single data source
No relationship between the carriers	Set of an orthogonal carriers
There is a guard band between carriers	No guard band between carriers
Low spectral efficiency	Better spectral efficiency
More subject to ISI and external interference from other RF sources	Overcomes ISI and delay spread

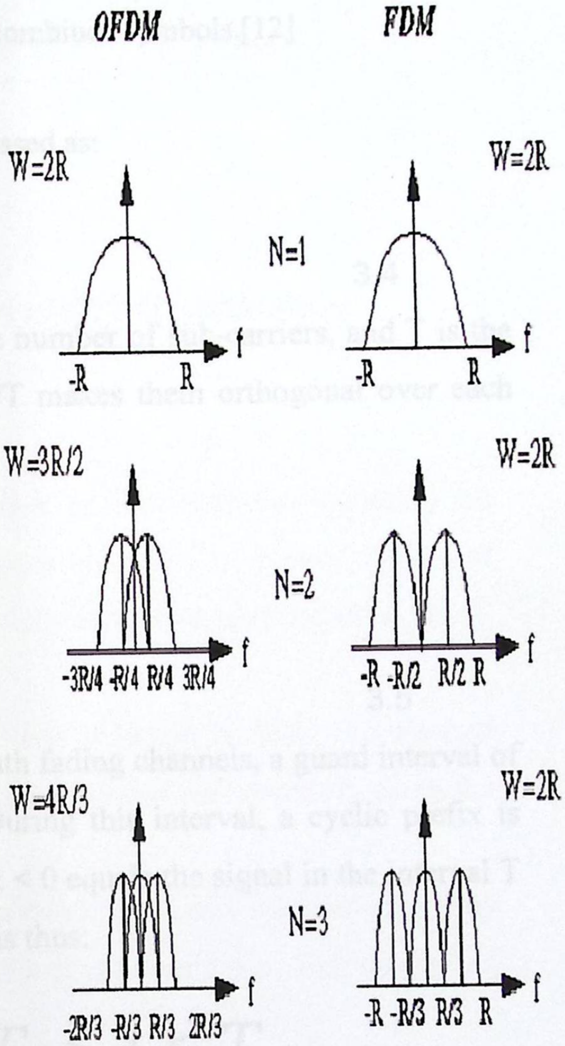


Figure (3.10)

3.2.5 Mathematical description of the transmitted signal

If N sub-carriers are used, and each sub-carrier is modulated using M alternative symbols, the OFDM symbol alphabet consists of M^N combined symbols.[12]

The low-pass equivalent OFDM signal is expressed as:

$$\nu(t) = \sum_{k=0}^{N-1} X_k e^{j2\pi kt/T}, \quad 0 \le t < T, \tag{3.4}$$

Where $\{x_k\}$ are the data symbols, N is the number of sub-carriers, and T is the OFDM symbol time. The sub-carrier spacing of 1/T makes them orthogonal over each symbol period; this property is expressed as:

$$\begin{aligned} &\frac{1}{T}\int_0^T \left(e^{j2\pi k_1 t/T}\right)^* \left(e^{j2\pi k_2 t/T}\right) dt \\ =&\frac{1}{T}\int_0^T e^{j2\pi(k_2-k_1)t/T}\, dt = \delta_{k_1 k_2} \end{aligned} \tag{3.5}$$

To avoid intersymbol interference in multipath fading channels, a guard interval of length T_g is inserted prior to the OFDM block. During this interval, a cyclic prefix is transmitted such that the signal in the interval $T_g < t < 0$ equals the signal in the interval $T - T_g < t < T$. The OFDM signal with cyclic prefix is thus:

$$\nu(t) = \sum_{k=0}^{N-1} X_k e^{j2\pi kt/T}, \quad -T_g \le t < T \tag{3.6}$$

The low-pass signal above can be either real or complex-valued. Real-valued low-pass equivalent signals are typically transmitted at baseband—wireline applications such as DSL use this approach. For wireless applications, the low-pass signal is typically complex-valued; in which case, the transmitted signal is up-converted to a carrier frequency f_c. In general, the transmitted signal can be represented as:

$$\begin{aligned} s(t) &= \Re\left\{\nu(t) e^{j2\pi f_c t}\right\} \\ &= \sum_{k=0}^{N-1} |X_k| \cos\left(2\pi [f_c + k/T] t + \arg[X_k]\right) \end{aligned} \tag{3.7}$$

.6 OFDM Transmitter :

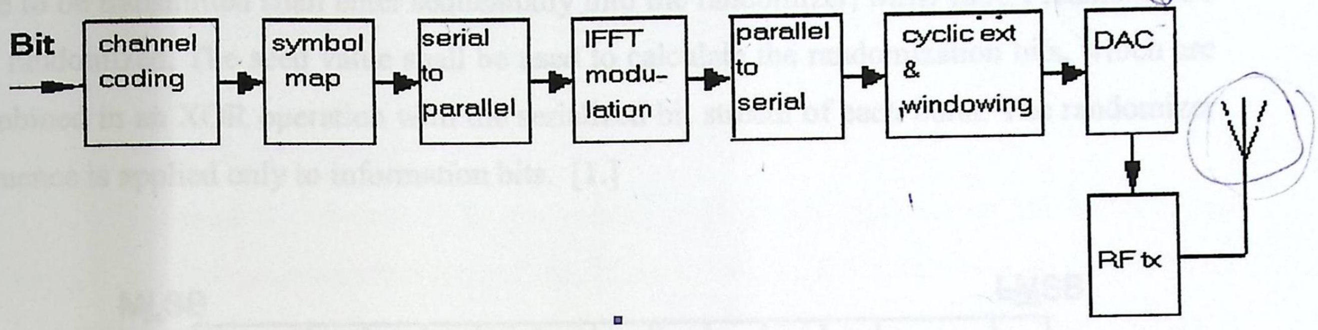


Figure (3.11) shows the block diagram of OFDM transmitter

3.2.6.1 Channel coding

Channel Coding : The channel encoder separates or segments the incoming bit stream (the output of the source encoder) into equal length blocks of L binary digits and maps each L- bit message block into an N - bit code word where N >L and the extra N – L check bits provide the required error protection.

There are $M = 2^L$ messages and thus 2^L code words of length N bits. The channel decoder maps the received N -bit word to the most likely code word and inversely maps the N -bit code word to the corresponding L -bit message.

Channel coding is composed of three steps: randomizer, FEC, and interleaving. They shall be applied in this order at transmission. The complementary operations shall be applied in reverse order at reception [1] .

3.2.6.1.1 Randomization

"Data randomization is performed on each burst of data on the downlink and uplink. The randomization is performed on each allocation (downlink or uplink), which means that for each allocation of a data block (subchannels on the frequency domain and OFDM symbols on the time domain) the randomizer shall be used independently. If the amount of data to transmit does not fit exactly the amount of data allocated, padding of 0xFF ("1" only) shall be added to the end of the transmission block. For RS-CC and CC encoded data padding will be added to the end of the transmission block, up to the amount of data allocated minus one byte, which shall be reserved for the introduction of a 0x00 tail byte by the FEC" [1] .

The shift-register of the randomizer shall be initialized for each new allocation.The pseudo random binary shift (PRBS) generator shall be as shown in Figure (3.10). Each data byte to be transmitted shall enter sequentially into the randomizer, MSB first. Preambles are not randomized. The seed value shall be used to calculate the randomization bits, which are combined in an XOR operation with the serialized bit stream of each burst. The randomizer sequence is applied only to information bits. [1.]

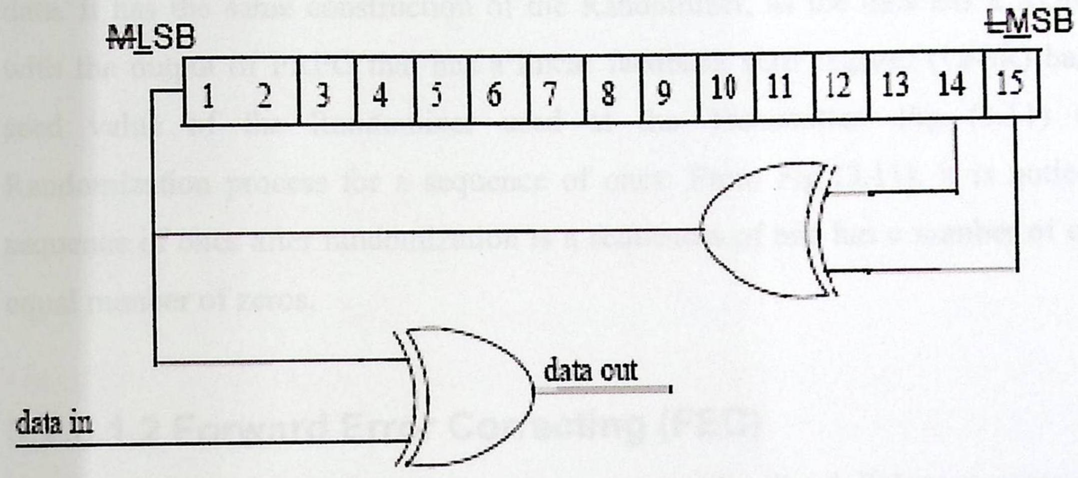


Figure (3.12)of PRBS generator

The randomizer is used in order to:

- ☐ Decrease the Peak to average power ratio (PAPR) of the transmitted data as the PAPR of the data can be quite large (e.g. more than 7 dB). It is inefficient to back to the transmitter power amplifier off enough to avoid unrecoverable nonlinear distortion at the worst possible PAPR.
- ☐ To ensure the clock synchronization at the receiver as the transition between bit values helps the receiver in synchronization.
- ☐ The randomization process ensures that there is no long runs of one's or zeros in the input bits. Since if we have long runs of ones the power of the signal will be decreases until the threshold and thus error happened [4].
- ☐ The PRBG consist of Linear-Feedback Shift Register (LFSR) has a characteristic polynomial 1 + X^14 + X^15. (As shown in the previous Fig (3.10)).
- ☐ The LFSR shall be present at the beginning of each frame to the value 100101010000000 and shall be clocked once per processed bit. [1]
- ☐ Only source bits are randomized. This includes source bits, plus uncoded (ones) bits that may be used to complete the transmission block when the amount of data

not enough . Elements that are not a part of the source data, such as framing elements and pilot symbols shall not be randomized. [1]

- On the down link the randomizer shall not be reset at the start of burst #1. [1]

The De-Randomizer:

It is used at the receiver to recover the original data again from the Randomized data. It has the same construction of the Randomizer, as the data has a XOR operation with the output of PRPG that has a linear feedback shift register (LFSR) has the same seed value of the Randomizer used at the Transmitter. Fig (3.11) shows the Randomization process for a sequence of ones: From Fig (3.11), it is noticed that the sequence of ones after randomization is a sequences of bits has a number of ones almost equal number of zeros.

3.2.6.1.2 Forward Error Correcting (FEC)

An FEC, consisting of the concatenation of a Reed–Solomon outer code and a rate-compatible convolutional inner code, shall be supported on both uplink and downlink. The Reed–Solomon Convolutional coding rate 1/2 shall always be used as the coding mode when requesting access to the network (except in subchannelization modes, which uses only convolutional coding 1/2) and in the FCH burst.[1]

The encoding is performed by first passing the data in block format through the RS encoder and then passing it through a zero-terminating convolutional encoder.

Reed-Solomon Codes & Convolutional encoder

Reed Solomon codes are *nonbinary cyclic* codes with symbols made up of **m**-bits sequence, where m is a positive integer greater than one. For the most conventional R-S (n,k) code **[10]**

$$(n,k) = (2^m - 1\ ,\ 2^m - 1 - 2t\ )$$

Where **k** is the number of data symbols being encoded , **n** is the total number of code symbols in the encoded block ,and t is the symbol error correction capability. [7]

Note that : $\mathbf{n} - \mathbf{k}$ = number of parity symbols = $\mathbf{2t}$ = twice error correction capability

In other words, for correcting t symbol errors no more than 2t parity symbols are required. R-S codes achieve the largest possible code minimum distance for any linear code. For nonbinary codes, the distance between two codewords is defined as the number of symbols in which the sequences differ. For the R-S codes the code minimum distance is given by:

$$d_{min} = n - k + 1 \tag{3.8}$$

$$t = \text{floor}\left(\left(d_{min} - 1\right)/2\right) \tag{3.9}$$

$$= \text{floor}\left(\left(n - k\right)/2\right) \tag{3.10}$$

R-S codes are used in many digital appliances such as CDs, and it is used in space and satellite communication such as (255, 223) RS code, this code is NASA standard code for satellite and space communications. The R-S code is useful for burst-error correction (for this reason it is useful for space communication). Consider an (n,k) = (255,247) R-S code : Since $n = 255 = 2^m - 1$,so m = # of bits per symbol = 8, so we can refer the symbol –in this example only- as a *byte*, Since $n - k = 8 = 2t$,so $t = 4$.

This means that this code can correct 4 symbol errors in a block of 255 symbols. Suppose the presence of a noise burst that lasts for duration of 25 bits beside each others as shown in figure (3.12):

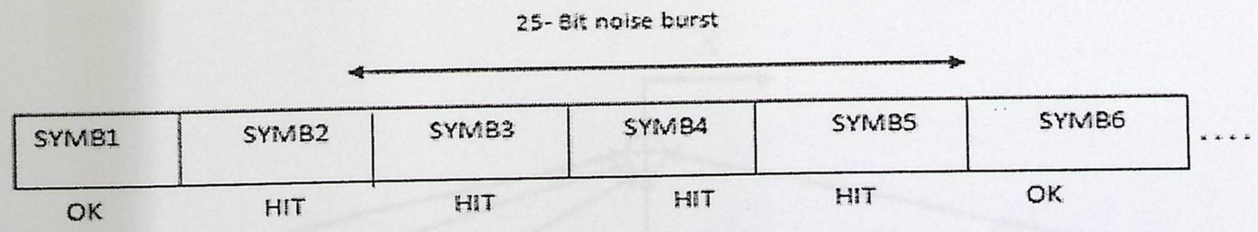


Fig (3.13) Data block disturbed by 25 bit noise burst

So this burst noise will affect (25/8 ≈ 4) symbols and this number is equal to the error correction capability (t) of the code, so we can say that code can correct 4 symbol errors regardless the damage suffered by each symbol.

The Reed–Solomon encoding shall be derived from a systematic RS (N = 255, K = 239, T = 8) code using $GF(2^8)$,[1]

where

N is the number of overall bytes after encoding, K is the number of data bytes before encoding,

T is the number of data bytes which can be corrected.

The following polynomials are used for the systematic code:

Code Generator Polynomial: $g(x)=(x+\lambda^0)(x+\lambda^1)(x+\lambda^1)(x+\lambda^2)\ldots(x+\lambda^{2T-1})$ 3.11

, $\lambda=02_{Hex}$

Field Generator Polynomial: $p(x)=x^8+x^4+x^3+x^2+x^1+1.$ 3.12

This code is shortened to enable variable block sizes and variable error-correction capability. When a block is shortened to K' data bytes, add 239-K' zero bytes as a prefix. After encoding discard these 239-K' zero bytes. When a codeword is shortened to permit T' bytes to be corrected, only the first 2T' of the total 16 parity bytes shall be employed. The bit/byte conversion shall be MSB first.[1]

Each RS block is encoded by the binary convolutional encoder, which shall have native rate of 1/2, a constraint length equal to 7, and shall use the generator polynomials codes shown in Equation below to derive its two code bits:

$G_1=171_{OCT}$ for X 3.13

$G_2=133_{OCT}$ for Y 3.14

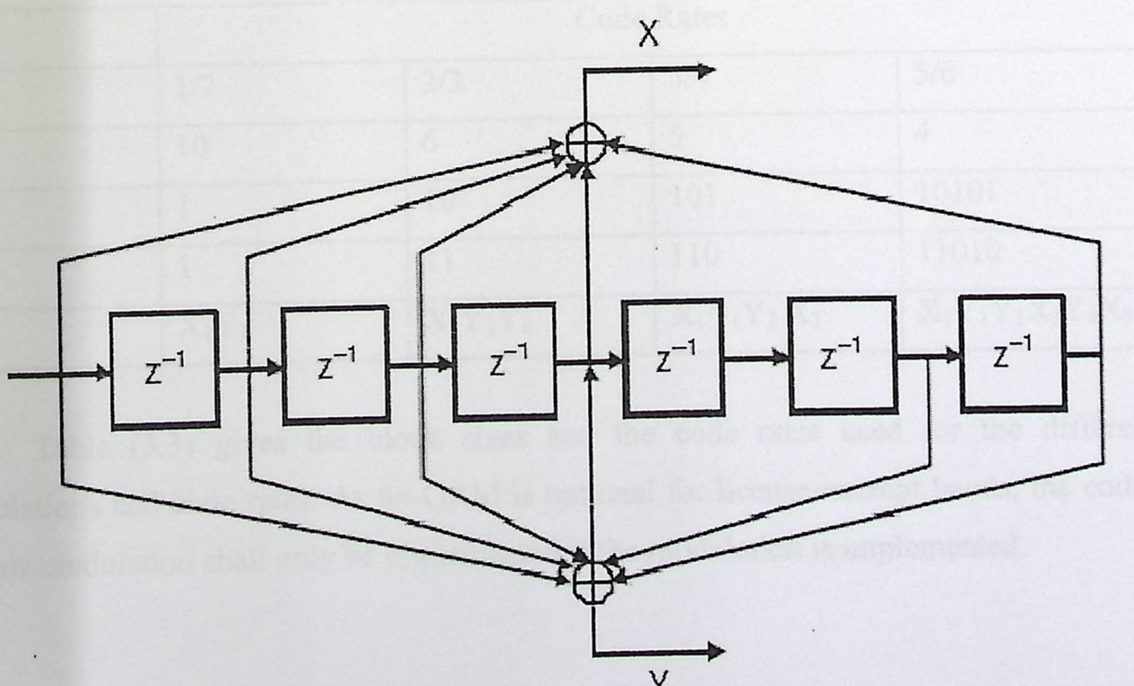


Figure (3.14)—Convolutional encoder of rate 1/2

Patterns and serialization order that will be used to realize different code rates are defined in Table (3.2). In the table, "1" means a transmitted bit and "0" denotes a removed bit, whereas *X* and *Y* are in reference to Figure (3.13). **[1]**

The RS-CC rate 1/2 will always be used as the coding mode when requesting access to the network. The encoding is performed by first passing the data in block format through the RS encoder and then passing it through a convolutional encoder. A single 0x00 tail byte is attached to the end of each burst. This tail byte will be attached after randomization. In the RS encoder, the redundant bits are sent before the input bits, keeping the 0x00 tail byte at the end of the allocation. When the total number of data bits in a burst is not an integer number of bytes, zero pad bits are added after the zero tail bits. The zero pad bits are not randomized.**[1]**

Note that this situation can occur only in subchannelization. In this case, the RS encoding is not employed.

Table 3. 2 - The inner convolutional code with shortening configuration

	Code Rates			
Rate	1/2	2/3	3/4	5/6
D_{free}	10	6	5	4
X	1	10	101	10101
Y	1	11	110	11010
XY	X_1Y_1	$X_1Y_1Y_2$	$X_1Y_1Y_2\,X_3$	$X_1Y_1Y_2X_3Y_4X_5$

Table (3.3) gives the block sizes and the code rates used for the different modulations and code rates. As 64-QAM is optional for license-exempt bands, the codes for this modulation shall only be implemented if the modulation is implemented.

Table 3.3 — channel coding per modulation

Modulation	Uncoded block size (Bytes)	Coded block size (bytes)	Overall coding rate	RS code	CC code rate
BPSK	12	24	1/2	(12,12,0)	1/2
QPSK	24	48	1/2	(32,24,4)	2/3
QPSK	36	48	3/4	(40,36,2)	5/6
16-QAM	48	96	1/2	(64,48,8)	2/3
16-QAM	72	96	3/4	(80,72,4)	5/6
64-QAM	96	144	2/3	(108,96,6)	3/4
64-QAM	108	144	3/4	(120,108,6)	5/6

3.2.6.1.3 Interleaver

"All encoded data bits shall be interleaved by a block interleaver with a block size corresponding to the number of coded bits per the allocated subchannels per OFDM symbol (Ncbps). The interleaver is defined by a two step permutation:

- ☐ The first step ensures that the adjacent coded bits are mapped onto nonadjacent subcarriers, which improves the performance of the decoder.
- ☐ The second step ensures that adjacent bits are alternately mapped to less and more significant bits of the modulation constellation, thus avoiding long runs of lowly reliable bits."[1]

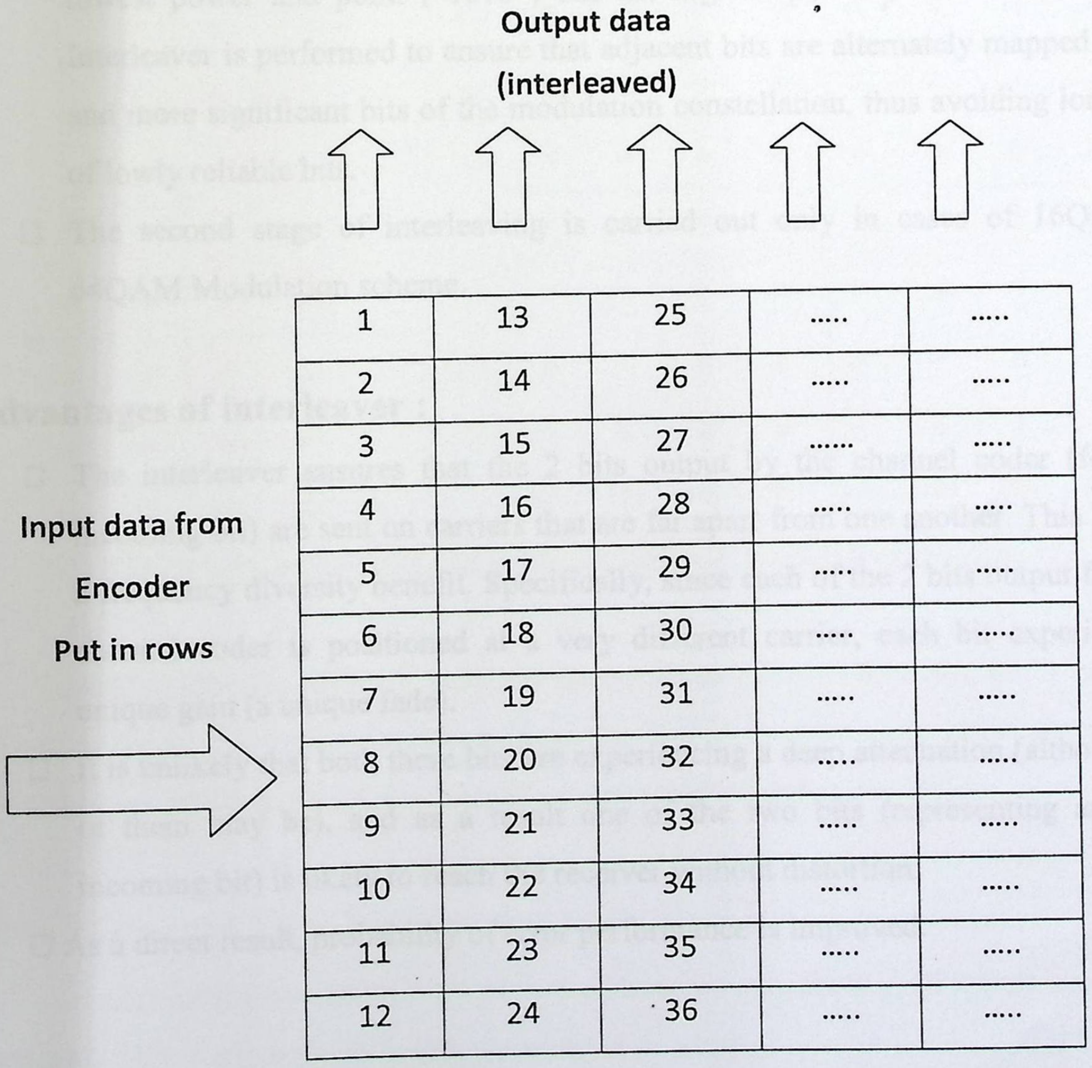
The Equations below provides the relation between k , mk, and jk , this indicates the bit index before and after the first and second steps of the interleaver, respectively, where Nc is the total number of bits in the block, and S is M/2, where M is the order of the modulation alphabet (2 for QPSK, 4 for 16 QAM, and 6 for 64 QAM), and **d** is an arbitrary parameter whose value is set to 12 (No. of rows) **:[1]**

$$mk = \left(\frac{Nc}{d}\right)*k_{\text{mod12}} + \text{floor}\left(\frac{k}{d}\right) \qquad 3.15$$

$$jk = s*\text{floor}\left(\frac{mk}{s}\right) + \left(mk + Nc - \text{floor}\left(\frac{d*mk}{Nc}\right)\right)_{\text{mod(s)}} \qquad 3.16$$

- It should be noted that interleaving is performed independently on each FEC block.
- The Interleaver of WiMAX makes a matrix of 12 rows and N columns which depends on the FEC block length .
- The separation between the subcarriers, to which two adjacent bits are mapped onto, depends on the subcarrier permutation schemes used.
- Since for 16 QAM and 64 QAM constellations, the probability of error for all the bits is not the same[1]. The probability of error of the most significant bit (MSB) is less than that of the least significant bit (LSB) for the modulation constellations , So we use the second stage of Interleaver to ensure that adjacent bits are alternately mapped to less and more significant bits of the modulation constellation .

Table (3.4) illustrates the idea of working for the first step of Interleaving :



1	13	25	.....	.....
2	14	26	.....	.....
3	15	27	......	.....
4	16	28	.....	.....
5	17	29	.....	.....
6	18	30	.....	.....
7	19	31	.....	.....
8	20	32	.....	.....
9	21	33	.....	.....
10	22	34	.....	.....
11	23	35	.....	.....
12	24	36	.....	.....

The number of column depends on the Forward Error correction (FEC) block Length. **For the Second Stage of Interleaver:** Assume the following constellation diagram for 16 QAM as shown in Fig (3.14):

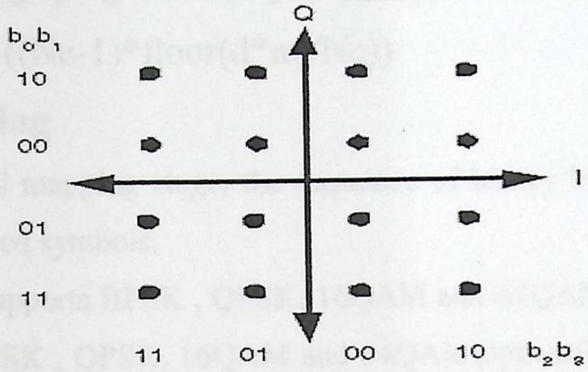


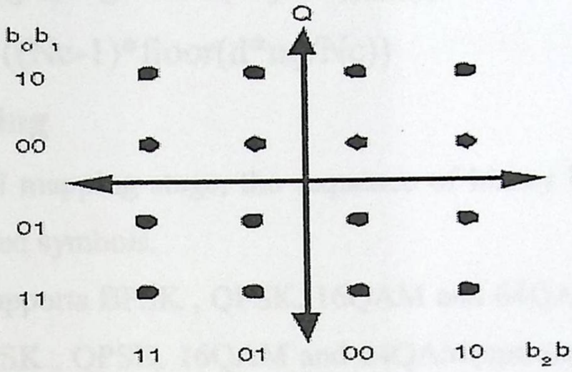
fig (3.15) Constellation diagram of 16-QAM

- From the constellation diagram , It is noticed that the point (0000) always has a lowest power and point (1010) has the highest power , So second stage of Interleaver is performed to ensure that adjacent bits are alternately mapped to less and more significant bits of the modulation constellation, thus avoiding long runs of lowly reliable bits.
- The second stage of interleaving is carried out only in cases of 16QAM or 64QAM Modulation scheme.

Advantages of interleaver :

- The interleaver ensures that the 2 bits output by the channel coder (for each incoming bit) are sent on carriers that are far apart from one another. This leads to a frequency diversity benefit. Specifically, since each of the 2 bits output from the channel coder is positioned at a very different carrier, each bit experiences a unique gain (a unique fade).
- It is unlikely that both these bits are experiencing a deep attenuation (although one of them may be), and as a result one of the two bits (representing an initial incoming bit) is likely to reach the receiver without distortion.
- As a direct result, probability of error performance is improved.

The number of column depends on the Forward Error correction (FEC) block Length. **For the Second Stage of Interleaver:** Assume the following constellation diagram for 16 QAM as shown in Fig (3.14):



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☐ As a direct result, probability of error performance is improved.

The De-Interleaver :

It is performs the inverse of the operation of the Interleaver, also works in two steps. The index of the jth bit after the first and the second steps of the de-interleaver[ieee] is given by:

$$mj=s*floor(j/s)+(j+floor(d*j/Nc))_{mod(s)} \quad 3.17$$

$$kj=d*mj-((Nc-1)*floor(d*mj/Nc)) \quad 3.18$$

3.2.6.2 Symbol mapping

During the symbol mapping stage, the sequence of binary bits is converted to a sequence of complex valued symbols.

Fixed WiMAX supports BPSK , QPSK, 16QAM and 64QAM in downlink (DL), & In the uplink (UL), BPSK , QPSK, 16QAM and 64QAM(optional) Fig (3.15) shows the constellation diagrams for different modulation schemes.**[2]**

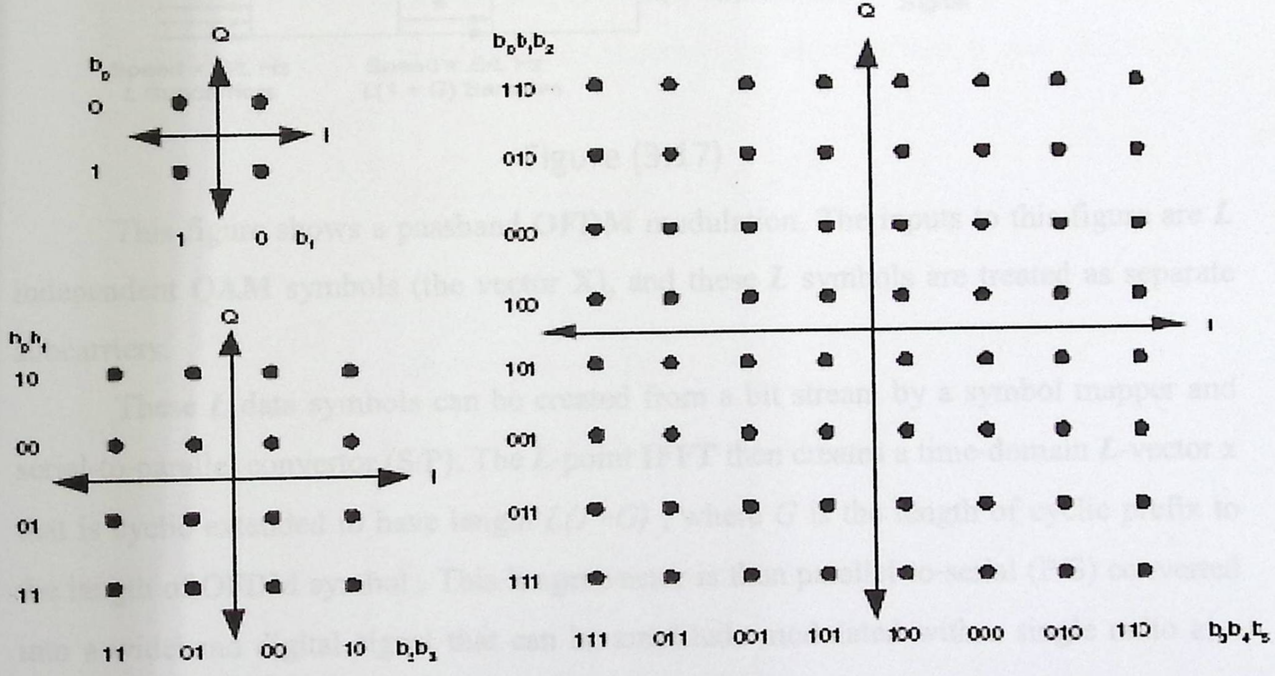


Fig (3.16) Constellation diagrams for different modulation schemes

As M increases, distance between constellation points decreases, so it is more difficult to detect the received symbols correctly. So, 64 QAM and 16 QAM are used at high SNR and QPSK & BPSK are used in a noisy environment (Low SNR).

3.2.6.3 Inverse Fast Fourier transform (IFFT)

In multicarrier modulation we have ***L*** orthogonal subcarriers , therefore we need L independent RF radio units in both transmitter & receiver.

In order to overcome the inconvenient requirement for L RF radios in both the transmitter and the receiver, **OFDM** uses a technique called Discrete Fourier Transform (**DFT**), which leads itself to a highly efficient technique known as the fast Fourier transform (**FFT**).

The **FFT** and its inverse, the **IFFT**, can create a multitude of orthogonal subcarriers using a single radio. Also creates an **ISI**-free channel, if the channel provides a circular convolution.

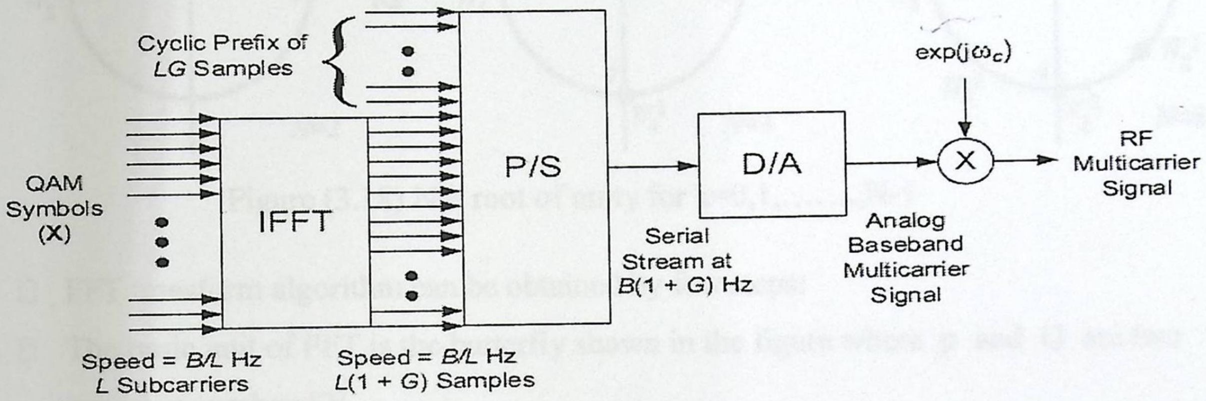


Figure (3.17)

This figure shows a passband **OFDM** modulation. The inputs to this figure are L independent **QAM** symbols (the vector **X**), and these L symbols are treated as separate subcarriers.

These L data symbols can be created from a bit stream by a symbol mapper and serial-to-parallel convertor (S/P). The L-point **IFFT** then creates a time-domain L-vector **x** that is cyclic extended to have length $L(1+G)$, where G is the length of cyclic prefix to the length of OFDM symbol . This longer vector is then parallel-to-serial (P/S) converted into a wideband digital signal that can be amplitude modulated with a single radio at a carrier frequency of $f_c = \omega_c / 2\pi$.

FFT Algorithm

DFT stands for Discrete Fourier Transform, DFT is given by:

$$A_k = \sum_{n=0}^{N-1} e^{-t\left(\frac{2\pi}{N}\right)kn} \qquad 3.19$$

Or

$$A_k = \sum_{n=0}^{N-1} w_{N}^{kn} a_n \qquad 3.20$$

Where $W_{N=}$ is the N^{th} root of unity circle :

$$W_N = e^{-t(\frac{2\pi}{N})} \qquad 3.21$$

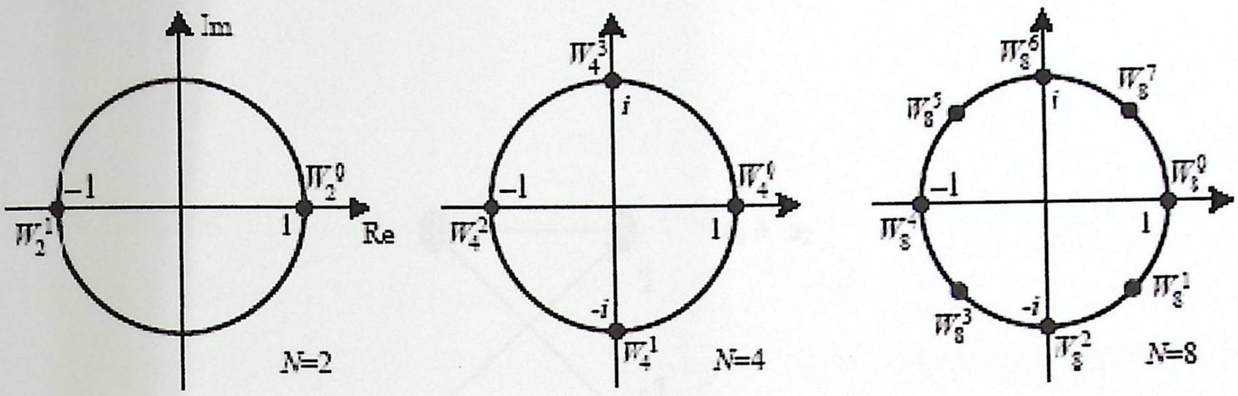


Figure (3.18) Nth root of unity for k=0,1,…….,N-1

☐ . FFT transform algorithm can be obtained by few steps:

☐ The basic unit of FFT is the butterfly shown in the figure where p and Q are two complex numbers[7]

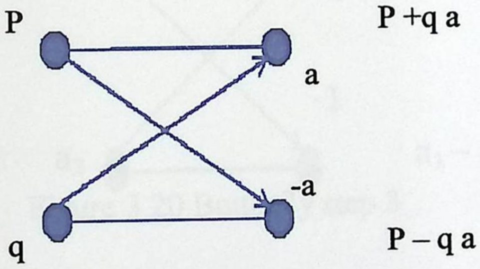


Figure (3.19) Butterfly basic unit

Steps*:* (Along with an example on how to compute FFT for N=4)

(1) Compute number of stage = $\log_2(N)$

For N=4 , No. of stages =2

(2) Perform Bit reversal of inputs : (Nj is bit reversal of j)

Table 3.4

J	0	1	2	3
N_J	0	2	1	3

(3) Apply the first stage butterfly:

Consist of N/2 butterflies using adjacent pair of numbers (Nj) in the buffer And coefficient (W_N) have exponents increasing by steps N/2 (Modulo N).

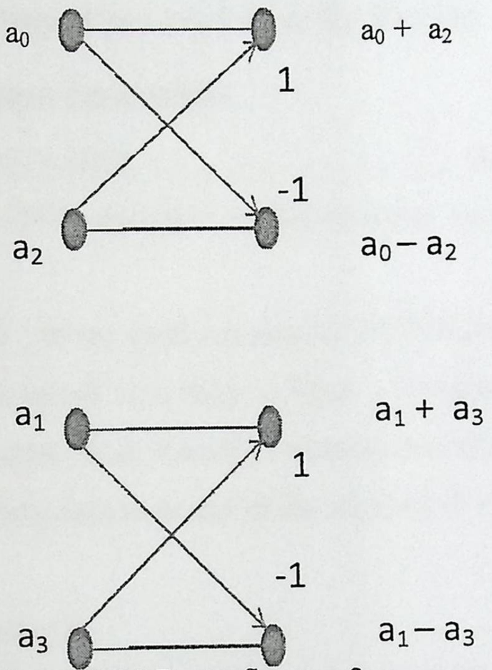
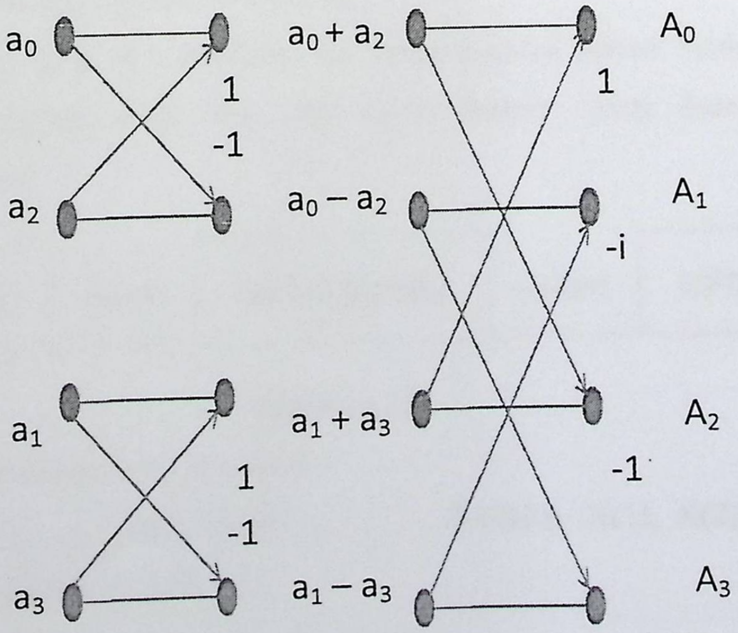


Figure 3.20 Butterfly step 3

(4) Apply the second stage butterfly:

Using pairs that are separated by two and coefficient have exponents increasing by steps N/4 (Modulo N)



i

Figure (3.21) Butterfly step 4

$A_0 = (a_0 + a_2) + (a_1 + a_3)$

$A_1 = (a_0 - a_2) - i(a_1 - a_3)$

$A_2 = (a_0 + a_2) - (a_1 + a_3)$

$A_3 = (a_0 - a_2) + i(a_1 - a_3)$

3.2.6.4 Guard Time Insertion and Cyclic Prefix

3.2.6.4.1 Guard Time Insertion

☐ We call X(0), X(1), X(2), X(3),……………………….. X(N-1) an OFDM symbol.

☐ This is result of one IFFT operation. In the receiver one FFT operation recovers the transmitted data.

☐ These is the problem ; If we send successive OFDM symbols, one directly after another. Since the channel acts like a filter ; Channel's impulse response is convolved with sequence X(n), therefore causing one OFDM symbol to interferer with another, destroying data recovery at the receiver via FFT.

☐ Analysis

- Transmitted sequence →

  X'(0), X'(1), X'(2), X'(3),….. X'(L-1), X(0), X(1), X(2), X(3),…X(L-1)

- Channel filter →

  h(v)………………… h(2), h(1),h(o) →→→→

- The out put of convolution = X(0)h(0)+X'(L -1)h(1)+X'(L -2)h(2)
- We can note that symbols are mixed.
- In order to solve this problem, we create a guard period between successive OFDM symbols such that the worst channel delay does not cause interference.

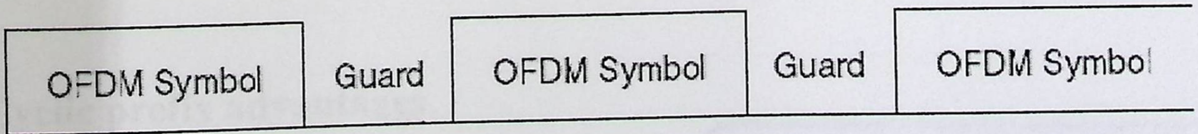


Figure (3.22)

- As the following→insert v zeros

X'(0), X'(1), X'(2),…….. X'(L-1),0000…………000X(0), X(1), X(2),…….. X(L-1)

h(v)…… h(2), h(1),h(o) →→→→

i

Figure (3.21) Butterfly step 4

$A_0 = (a_0 + a_2) + (a_1 + a_3)$

$A_1 = (a_0 - a_2) - i(a_1 - a_3)$

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☐ Analysis

- Transmitted sequence →

  X‘(0), X‘(1), X‘(2), X‘(3),….. X‘(L-1), X(0), X(1), X(2), X(3),…X(L-1)

- Channel filter →

  $h(v)$………………… h(2), h(1),h(o) →→→→

- The out put of convolution = X(0)h(0)+X‘(L -1)h(1)+X‘(L -2)h(2)
- We can note that symbols are mixed.
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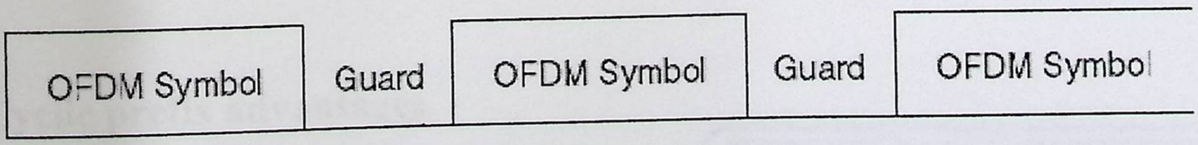

Figure (3.22)

- As the following→insert v zeros

X‘(0), X‘(1), X‘(2),…….. X‘(L-1),0000…………000X(0), X(1), X(2),…….. X(L-1)

$h(v)$…… h(2), h(1),h(o) →→→→

- Now we haven‘t mixing between X and X‘.
- Actually the guard period is not filled with zeros as previous.
- 'v' values from the end of the X sequence are copied and used in the guard region which create a circular convolution [7] and known as cyclic prefix.

3.2.6.4.2 Cyclic Prefix

- To make OFDM realizable in practice ; We the use the FFT algorithm.
- It has low complexity.
- In order for the IFFT/FFT to create an ISI-free channel, the channel must appear to provide a circular convolution. Adding cyclic prefix to the transmitted signal.

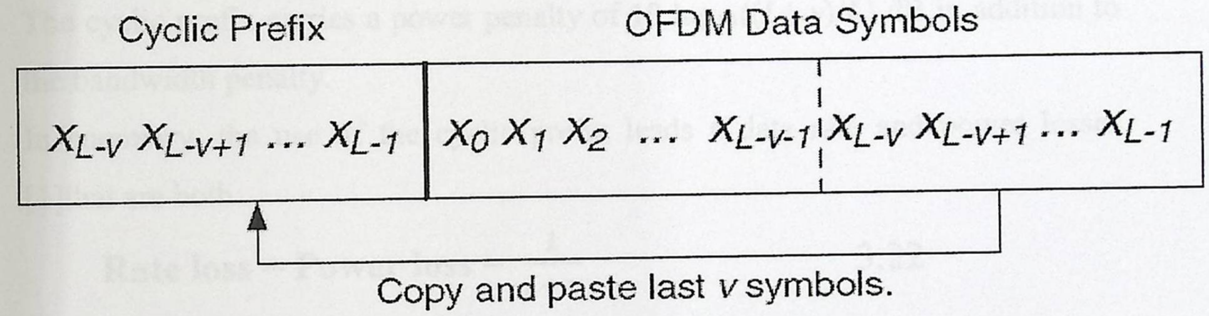

Figure (3.23)

- As the following→ copy v values from the end of the x sequence and use them in the guard region.
- X‘(0), X‘(1), X‘(2),… X‘(L-1), X(L-v),….. X(L-2),X(L-1),X(0), X(1), X(2),. X(L-1) h(v)…………..… h(2), h(1),h(o) →→→→
- In addition to preventing interference between X,X‘ the cyclic prefix above realizes the purpose of the OFDM by making each sub channel see a flat channel response.

Cyclic prefix advantages

1. The cyclic prefix is simple

2. In addition to preventing interference between X,X‘ the cyclic prefix above realizes the purpose of the OFDM by making each sub channel see a flat channel response.

Cyclic prefix disadvantages

1. The cyclic prefix is not entirely free.
2. It comes with both a bandwidth and power penalty. Since it consumes power & increases the bandwidth without providing any useful information[8].
3. Since ν redundant symbols are sent, the required bandwidth for **OFDM** increases from B to $((L+\nu)/L)B$.
4. The cyclic prefix carries a power penalty of $10\log_{10}((L+\nu)/L)$ dB in addition to the bandwidth penalty.
5. In summary, the use of the cyclic prefix leads a data rate and power losses [1]that are both

$$\textbf{Rate loss} = \textbf{Power loss} = \frac{L}{L+v} \qquad 3.22$$

6. The wasted power causes interference to neighboring users.

From above we can conclude :

1. It can be noted that for $L>>v$, the inefficiency caused by cyclic prefix can be made small by increasing the number of subcarriers (L).
2. The cyclic prefix provides a guard interval for all multipaths following the first arrival signal. As a result the time required to observe a useful OFDM symbol is quite up to $\tau_{\max}$.
3. On the other hand, time estimation often fails on the multipath signal with the highest strength[8], which in some cases may not be the first arrival signal.
4. To increase the robustness of the receiver, the guard interval is often inserted into cyclic pre-fix and post-fix as in Figure to guard the data block against (relative to the strongest path) multipath effects.

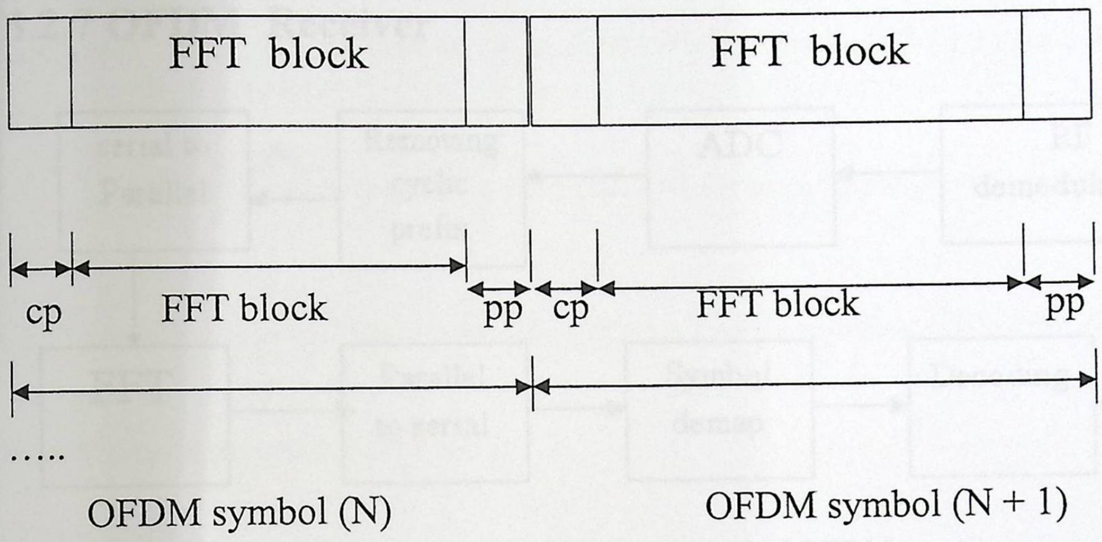
FFT block
FFT block
.....
cp
FFT block
pp
cp
FFT block
pp
.....
OFDM symbol (N)
OFDM symbol (N + 1)

Figure (3.24)

3.2.7 OFDM Receiver

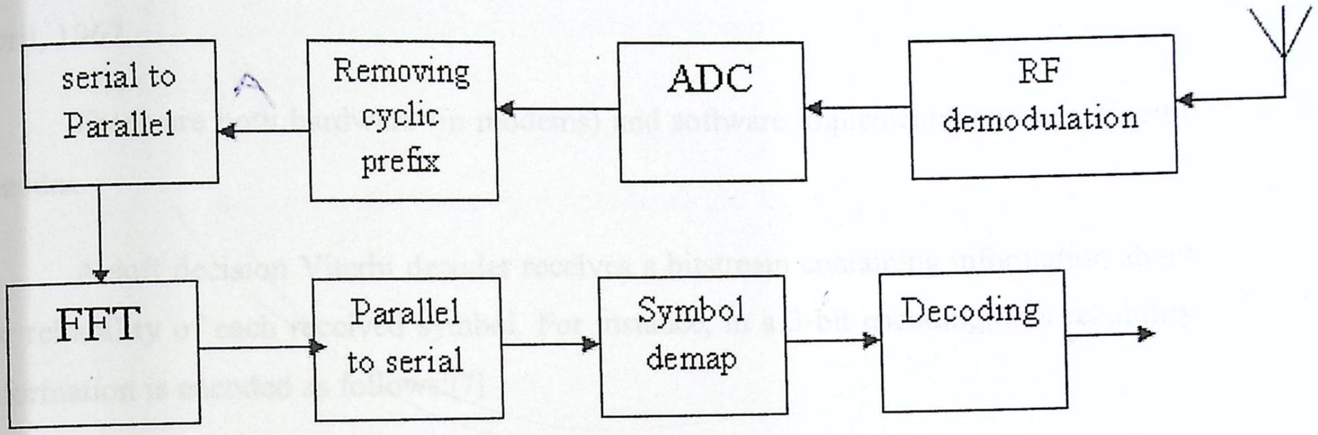

Figure (3.25) shows the block diagram of OFDM receiver

- At the receiver, first RF demodulation is performed as shown in figure (3.25).
- Then the signal is digitized using an ADC.
- Timing and frequency synchronization are performed.
- The guard time is removed from each OFDM symbol.
- The sequence is converted to parallel format.
- FFT (OFDM demodulation) is applied to get back to the frequency domain.
- The output is then serialized.
- Symbol de-mapping is done to get back the coded bit sequence.
- Channel decoding (de-interleaving , viterbi decoding, de-randomization) is done to get the user bit sequence.

3.2.7.1 Viterbi decoder

A Viterbi decoder uses the Viterbi algorithm for decoding a bitstream that has been encoded using Forward error correction based on a Convolutional code.[7]

There are other algorithms for decoding a convolutionally encoded stream (for example, the Fano algorithm). The Viterbi algorithm is the most resource-consuming, but it does the maximum likelihood decoding. It is most often used for decoding convolutional codes with constraint lengths k<=10, but values up to k=15 are used in practice.

Viterbi decoding was developed by Andrew J. Viterbi and published in the paper "Error Bounds for Convolutional Codes and an Asymptotically Optimum Decoding Algorithm", IEEE Transactions on Information Theory, Volume IT-13, pages 260-269, in April, 1967.

There are both hardware (in modems) and software implementations of a Viterbi decoder.

A soft decision Viterbi decoder receives a bitstream containing information about the reliability of each received symbol. For instance, in a 3-bit encoding, this reliability information is encoded as follows:[7]

Table 3.5

value	meaning
000	strongest 0
001	relatively strong 0
010	relatively weak 0
011	weakest 0
100	weakest 1
101	relatively weak 1
110	relatively strong 1
111	strongest 1

Of course, it is not the only way to encode reliability data.

The squared Euclidean distance is used as a metric for soft decision decoders.[7]

3.2.8 OFDM Drawbacks

☐ Despite these advantages, OFDM techniques also face several challenges.

- Multi-carrier signals with high peak-to-average power ratio (PAPR) require high linear amplifiers. Otherwise, performance degradations occur and the out-of-band power will be enhanced. [1]
- Loss in spectral efficiency due to the guard interval.[8]
- More sensitive to Doppler spreads then single-carrier modulated systems. [8]

- Out of Band radiation. [7]
- So accurate frequency and time synchronization is required.

3.2.8.1 Peak to Average Power Ratio (PAPR)

- ☐ Since the OFDM signal is the superposition of low rate streams modulated at different frequencies, its time-domain range increases with the number of subcarriers (dynamic range). [8]
- ☐ The high peak-to-average power ratio (PAPR) imposes stringent requirements on the A/Ds and D/As, and more importantly, on the linearity of the power amplifier (PA). [8]

3.3. WIMAX physical layer based on OFDM :

3.3.1. OFDM symbol parameters & transmitted signal according to IEEE 802.16d:

The basic parameter definitions

Four basic parameters characterize the OFDM symbol:

— BW : This is the channel bandwidth.

— N_{used} :. Number of used subcarriers.

— n : Sampling factor. This parameter, in conjunction with BW and $Nused$ determines the subcarrier spacing, and the useful symbol time.

— G : This is the ratio of CP time to "useful" time.

The values of these parameters are shown in the table 3.6

Derived parameter definitions

The following parameters are defined in terms of the basic parameters

— N_{FFT} : Smallest power of two greater than N_{used} (i.e. if N_{used} =200 then N_{fft} =256)

— Sampling Frequency : $F_s =$ floor(n*BW/8000)*8000

— Subcarrier spacing.:$\Delta f = F_s / N_{FFT}$

— Useful symbol time : $T_b = 1/\Delta f$

— CP Time: $T_g = G.T_b$

— OFDM Symbol Time : $T_s = Tb + Tg$

— Sampling time : $T_{\tilde{b}}\, N_{FFT}$

Table 3.6 The basic parameters of OFDM symbol

Parameter	value
N_{FFT}	256
N_{used}	200
n	For channel bandwidths that are a multiple of 1.75 MHz then $n = 8/7$ For channel bandwidths that are a multiple of 1.5 MHz then $n = 86/75$. For channel bandwidths that are a multiple of 1.25 MHz then $n = 144/125$ For channel bandwidths that are a multiple of 2.75 MHz then $n = 316/275$ For channel bandwidths that are a multiple of 2.0 MHz then $n = 57/50$ For channel bandwidths not otherwise specified then $n = 8/7$
G	1/4 , 1/8 , 1/16 , 1/32
Number of lower frequency guard subcarriers	28
Number of higher frequency guard subcarriers	27
Bandwidth	(1.75 , 3 , 3.5 , 5.5 , 7 and 10)MHz

Table 3.7 shows IEEE 802.16d block sizes of the bit interleaver.

Table 3.7—Block sizes of the Bit Interleaver

	Default (16 subchannels	**8 subchannels**	**4 subchannels**	**2 subchannels**	**1 subchannel**
	N_{cbps}				
BPSK	192	96	48	24	12
QPSK	384	192	96	48	24
16-QAM	768	384	192	96	48
64-QAM	1152	576	288	144	72

Table (3.8) illustrate the various modulations supported by **WIMAX**. In the downlink, **QPSK**, **16 QAM**, and **64 QAM** are elementary for both fixed and mobile WIMAX while **64 QAM** is optional in the uplink.[1]

Table 3.8 – 802.16 Modulation Schemes

Modulation scheme	**Required SNR (dB)**	**Description**
BPSK	6	1 bit per symbol , very robust against environment
QPSK	9	2 bits per symbol
16-QAM	16	4 bit per symbol
64-QAM	22	6 bits per symbol & only for LOS & very short distance

8.3.2.3 Transmitted signal

Equation (65) specifies the transmitted signal voltage to the antenna, as a function of time, during any OFDM symbol.

$$s(t) = \mathrm{Re}\left\{ e^{j2\pi f_c t} \sum_{\substack{k=-Nused/2 \\ k \neq 0}}^{Nused/2} c_k . e^{j2\pi\Delta fk(t - Tg)} \right\} \qquad 3.23$$

where

t is the time, elapsed since the beginning of the subject OFDM symbol, with $0 < t < Ts$,

ck is a complex number; the data to be transmitted on the subcarrier whose frequency offset index is *k,* during the subject OFDM symbol. It specifies a point in a QAM constellation. In subchannelized transmissions, *ck* is zero for all unallocated subcarriers.

3.3.2 Physical Layer Framing:

At the PHY, the flow of bits is structured as a sequence of frames of equal length.

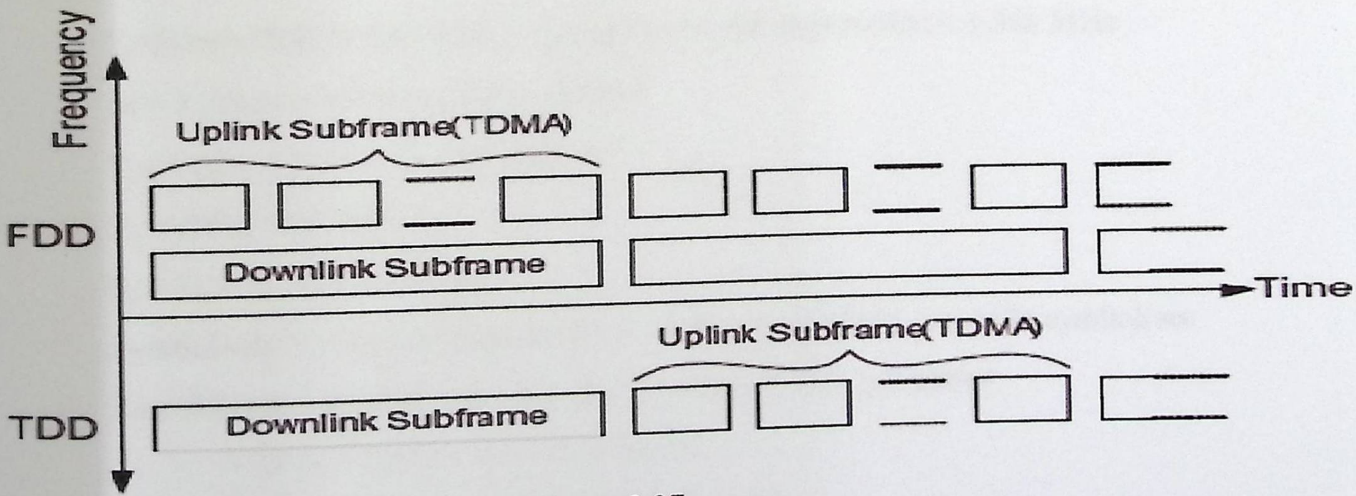

Figure 3.27

There is a down link sub frame and an uplink sub frame and two modes of operation: frequency division duplex FDD and time division duplex TDDas shown in figure (3.25). The figure above pictures these two modes. In FDD, the down link sub frame and uplink sub frame are simultaneous but do not 72 interfere because they are sent on different frequencies. In TDD, the down link sub frame and uplink sub frame are consecutive. A Frame duration

of 0.5, 1 or 2 milliseconds can be used. Frames are of equal length.**[1]**

In TDD,(this means that the uplink and down link direction between BS and MS use the same frequency band). The part allocated for the down link and part allocated to the uplink may vary. The uplink is time division multiple access TDMA, which means that the bandwidth is divided into time slots.

Transmit spectral mask

The transmitted spectral density of the transmitted signal shall fall within the spectral mask range (-50 dBr – 0 dBr)[1]. The measurements shall be made using 100 kHz resolution bandwidth and a 30 kHz video bandwidth. The 0 dBr level is the maximum power allowed by the relevant regulatory body.[1]

3.3.3 PHY Speed Calculations For PMP:

For example ; If we use 16-QAM , G= 1/4 ,N_{FFT}=256 , coding rate of 1/2 & bandwidth of 5.5MHz then n=316/275 as specified in standard .then :

F_s=floor(n*BW/8000)*8000 = floor((1/4)*5.5M/8000)*8000 = 1.368 MHz

$\Delta f = F_s / N_{FFT} = 1368000/256 = 5343.75$

$T_b = 1/\Delta f = 1/5343.75 = 187.1345\ \mu s$

$T_g = G * T_b = 46.7836\ \mu s$

$T_s = T_b + T_g = 233.9181\ \mu s$

Symbol rate = 1/symbol transmit time = $1/T_s = 1/233.9181\ \mu = 4275$ symbol/sec

Raw Bit rate = symbol rate * Number f carriers * Bits per carrier

= 4275 *256*4 = 4.3776 M bit/sec

Bit rate after coding = Raw bit rate * coding rate

= 4.3776 M *1/2 =2.1888 M bit/sec

Transmission speeds of 70 MBPS or more can be achieved with WIMAX. This value can theoretically be reached when using a 20 MHz carrier and 64 QAM modulation with a coding rate of 3/4.

Standard model :

- ✓ Bandwidth : must be a multiple of 1.5 , 1.75 , 2 , 2.5 , 2.75 or 3.5 MHz
- ✓ Mandatory modulations : (BPSK, QPSK, 16-QAM, 64-QAM (Down link)
- ✓ Option modulations : (64-QAM (up link) , 256-QAM)
- ✓ Guard width ratio :1/4, 1/8, 1/16 or 1/32
- ✓ Coding rate :1/2, 2/3 , 3/4 , 5/6
- ✓ # of subcarriers : mandatory 256 , option 512.
- ✓ Channel Models : SUI1- SUI6 .
- ✓ RS , convolutional encoding at transmitter & viterbi decoder and RS decoding at receiver. Option (Turbo coding)
- ✓ Maximum allowed spectral power density is 0 dBr .
- ✓ Diversity and MIMO are option.

3.4 Our model :

In our project we will use these parameters to analyze the performance of WIMAX:

- ✓ Bandwidth ((1.75 ,5, 10,20) MHz
- ✓ Modulation (BPSK, QPSK, 16 QAM, 64 QAM, 256 QAM)
- ✓ Guard width ratio :1/4,1/8,1/16,1/32
- ✓ Coding rate :1/2,2/3,3/4,5/6,7/8
- ✓ # of subcarriers : 256, 512
- ✓ Channels : SUI1- SUI6 under AWGN & Rayleigh fading
- ✓ RS , convolutional coding at transmitter & viterbi decoder at receiver.
- ✓ **Receiver Diversity of 1x2 and 1x4 .**
- ✓ **MIMO of 2x2 and 4x4 .**
- ✓ SNR: (0 - 40) dB with fixed step of 1 dB .

Chapter 4

The Diversity and MIMO Application to WiMAX system

4.1. Introduction

4.1.1 The effects of fading on system performance

The particularity of wireless links is that they are affected by random fluctuations of the signal level not only across time, but also across space or frequency. This behavior is known as fading, and affects the performance (in terms of symbol or bit error rate) of any wireless system. As an example, consider the simple case of binary phase-shift keying (BPSK) transmission through a SISO (single input single output) Rayleigh fading channel. In the absence of fading ($h=1$), the symbol-error rate (SER) in an additive white Gaussian noise (AWGN) channel is given by

$$p = Q\left(\sqrt{\frac{2E_s}{\sigma_n}}\right) = Q(\sqrt{2\rho}) = \frac{1}{2}\, erfc\left(\frac{\sqrt{2\rho}}{\sqrt{2}}\right) \qquad (4.1)$$

Where Q : Q - function where $Q(x_0) = \int_{x_0}^{\infty} \frac{1}{\sqrt{2\pi}} e^{\frac{-y^2}{2}}\, dy$

erfc : the error complementary function where $\mathrm{erfc}(x) = \frac{2}{\sqrt{\pi}} \int_{x}^{\infty} e^{-t^2}\, dt$

E_s : Symbol energy

σ_n : Noise energy

ρ : Signal to noise ratio

p : Probability of error under AWGN channel

When fading is considered, the received signal level fluctuates. As a result, the error rate is obtained through the following integration

$$\bar{p} = \int_0^{\infty} Q(\sqrt{2\rho}) p_s(s)\, ds \qquad (4.2)$$

where $p_s(s)$ is the fading distribution. In Rayleigh fading, the integration in above equation yields

$$\bar{p} = \frac{1}{2}\left(1 - \sqrt{\frac{\rho}{\rho+1}}\right) \qquad (4.3)$$

At large SNR, the error rate simplifies to

$$\bar{p} \cong \frac{1}{4\rho} \quad (4.4)$$

So , we note that the bit error rate decreases only inversely with the SNR By contrast, the decrease in error rate in non-fading AWGN channels is exponential with the SNR see(4.1) .

4.2. Principle of diversity

To decrease the effect of fading on increasing the bit error rate, diversity techniques are usually employed. The principle of diversity is to provide the receiver with multiple versions of the same transmitted signal. Each of these versions is defined as a diversity branch. If these versions are affected by independent fading conditions, the probability that all branches are in a fade at the same time reduces . So, diversity leads to improved performance in terms of error rate.

Because fading may take place in time, frequency and space, diversity techniques may similarly be implemented in each of these domains. As an example, time diversity can be obtained via appropriate coding and interleaving. Frequency diversity spread the channel (in the *time* domain) through equalization techniques or multi-carrier modulations. Naturally, both time and frequency diversity techniques introduce a loss in time or bandwidth to allow for the introduction of redundancy. By contrast, spatial or polarization diversity does not introduce time and bandwidth losses , since it is provided by the use of multiple antennas at one or both sides of the link. Yet the spatial dimensions are increased by the use of antenna arrays.

4.2.1 Single-Input Multiple-Output systems(SIMO) (Receiver Diversity)

Single-Input Multiple-Output (SIMO) systems rely on the use of muliple antennas at the receiver to achieve diversity. If these multiple antennas are sufficiently spaced (say, by one wavelength), one to allow sufficient decorrelation in the received signal copies different. Receiver diversity may be implemented via two rather different combining methods:

- Selection combining: the combiner selects the branch with the highest SNR among all diversity branches , which is then used for detection

- Gain combining : the signal used for detection is a linear combination of all branches. Suppose that the number of receiver antennas is n_r , then the signal (z) used for detection is [Oestges Claude , **MIMO Wireless Communications_From Real-World Propagation to Space-Time Code Design**]

$$z = \mathbf{w}^T \mathbf{y} \tag{4.5}$$

where $\mathbf{w} = [w_1, \ldots, w_{n_r}]^T$: is the combining vector.

$\mathbf{y} = [y_1, \ldots, y_{n_r}]$: is the received vector.

4.2.2 Receive diversity via gain combining

In gain combining, the signal z used for detection is a linear combination of all branches

$$z = \boldsymbol{w}^T \boldsymbol{y} = \sum_{n=1}^{n_r} w_n y_n \tag{4.6}$$

where the w_n's are the combining weights and $\boldsymbol{w} = [w_1, \ldots, w_n]^T$.

Depending on the choice of these weights, different gain combining methods have been developed. Suppose that the data symbol c is sent through the channel and received by n_r antennas. Each antenna is characterized by the channel

$$h_n = |h_n| e^{j\emptyset_n}, \tag{4.7}$$

Where $\emptyset_n$: Phases of channel

$n=1, \ldots, n_r$

assumed to be Rayleigh distributed with unit variance, all the channels being independent. The signals from all antennas are then combined and the detection variable is expressed as [Oestges Claude , **MIMO Wireless Communications_From Real-World Propagation to Space-Time Code Design**]

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Where $\emptyset_n$: Phases of channel

$n = 1, \ldots, n_r$

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$$z = \sqrt{E_s}\, \boldsymbol{w}^T \boldsymbol{h}\, c + \boldsymbol{w}^T \boldsymbol{n} \tag{4.8}$$

Where $\mathbf{h} = [h_1, \dots, h_{n_r}]^T$

E_s : symbol energy

n : Additive White Gaussian Noise vector .

4.2.2.1 Maximal Ratio Combining (MRC) :

In our project we used MRC for combining the signal copies, since the MRC maximizes the output SNR (Signal-to-Noise Ratio) , where in this method the output SNR (ρ_{out}) is equal to the summation of the SNR_{out} at all receive diversity branches . [Andrea Goldsmith , **Wireless Communications**]

The output SNR will be

$$\rho_{out} = \frac{\left(\sum_{n=1}^{n_r} (|h_n| a_n)\right)^2 / 2}{\sum_{n=1}^{n_r} \left(a_n^2 E(n_n^2(t))\right)} \tag{4.9}$$

Where h_n : fading envelop = $\sqrt{E_s}$

a_n : signal amplitude

Assume $E(n_n^2(t)) = \frac{N_0}{2}$ = noise variance , then

$$\rho_{out} = \frac{\left(\sum_{n=1}^{n_r} (|h_n| a_n)\right)^2 / 2}{\frac{N_0}{2} \sum_{n=1}^{n_r} (a_n^2)} \tag{4.10}$$

to maximize ρ_{out} , assume

$$a_n = \frac{h_n}{\sqrt{N_0}} \tag{4.11}$$

[Andrea Goldsmith , **Wireless Communications**]

Then

$$\rho_{out} = \sum_{n=1}^{n_r} \frac{h_n^2}{N_0} = \sum_{n=1}^{n_r} \rho_n \qquad (4.12)$$

So , we note that the output SNR is the summation of all branches's SNR .

4.3. Multiple-Input Single-Output systems

Multiple-Input Single-Output (MISO) systems exploit diversity at the transmitter through the use of multiple transmit antennas (n_t) in combination with pre-processing or pre-coding. A significant difference with receive diversity is that the transmitter might not have the knowledge of the MISO channel. At the receiver, the channel is easily estimated. This is not the case at the transmit side, where feedback from the receiver is required to inform the transmitter.

[Oestges Claude , **MIMO Wireless Communications_From Real-World Propagation to Space-Time Code Design**].

There are basically two different ways of achieving *direct transmit diversity*:

- When the transmitter has a perfect channel knowledge, beamforming can be performed using various optimization metrics (SNR, SINR(signal-to-Interference and Noise Ratio), etc.) to achieve both diversity and array gains

- When the transmitter has no channel knowledge, pre-processing known as space–time coding is used to achieve a diversity gain, but no array gain.

In this section, we introduce a very simple space–time coding technique known as the Alamouti scheme.

4.3.1 Transmit diversity via space–time coding

Alamouti has developed a particularly simple transmit diversity scheme for two transmit antennas, known as the Alamouti scheme, which does not require transmit channel knowledge . [Oestges Claude , **MIMO Wireless Communications_From Real-World Propagation to Space-Time Code Design**].

In this scheme , two symbols c_1 and c_2 are transmitted simultaneously from antennas 1 and 2 during the first symbol period, followed by symbols $-c_2^*$ and $-c_1^*$, transmitted from antennas 1 and 2 during the next symbol period. Assuming that the flat fading channel remains constant over the two successive symbol periods, and is denoted by $\mathbf{h}=[h_1 \;\; h_2]$ (the subscripts here denote the antenna number and not the symbol periods). The symbol y_1 received at the first symbol period is

$$y_1 = \sqrt{E_s}h_1\frac{c_1}{\sqrt{2}} + \sqrt{E_s}h_2\frac{c_2}{\sqrt{2}} + n_1 \qquad (4.13)$$

Where n_1 : is the AWGN contribution at y_1 .

and the symbol y_2 received at the second symbol period is

$$y_2 = -\sqrt{E_s}h_1\frac{c_2^*}{\sqrt{2}} + \sqrt{E_s}h_2\frac{c_1^*}{\sqrt{2}} + n_2 \qquad (4.14)$$

Where n_2 : is the AWGN contribution at y_2 .

where each symbol is divided by $\sqrt{2}$ so that the vector $c = [c_1/\sqrt{2} \quad c_2/\sqrt{2}]$ has a unit average energy (assuming that $c1$ and $c2$ are drawn from a unit average energy constellation) and $n1$ and n_2 are the additive noise at each symbol period (here, the subscripts denote the symbol periods and not the antennas).

We may express the combination of the above two equations as

$$\mathbf{y} = \begin{bmatrix} y_1 \\ y_2 \end{bmatrix} = \sqrt{E_s}\underbrace{\begin{bmatrix} h_1 & h_2 \\ h_2^* & -h_1^* \end{bmatrix}}_{H_{eff}}\underbrace{\begin{bmatrix} c_1/\sqrt{2} \\ c_2/\sqrt{2} \end{bmatrix}}_{c} + \begin{bmatrix} n_1 \\ n_2^* \end{bmatrix} \qquad (4.15)$$

We observe that the two symbols are spread over two antennas and over two symbol periods,so **H***eff* appears as a *space–time* channel. Applying the matched filter H_{eff}^H to the received vector **y** effectively decouples the transmitted symbols as shown below:

$$\begin{bmatrix} z_1 \\ z_2 \end{bmatrix} = H_{eff}^H \begin{bmatrix} y_1 \\ y_2^* \end{bmatrix} = \sqrt{E_s}\ [|h_1|^2 \quad |h_2|^2]\boldsymbol{I}_2 \begin{bmatrix} c_1/\sqrt{2} \\ c_2/\sqrt{2} \end{bmatrix} + H_{eff}^H \begin{bmatrix} n_1 \\ n_2^* \end{bmatrix} \tag{4.16}$$

$$z = \sqrt{E_s}\|\boldsymbol{h}\|^2 \boldsymbol{I}_2 c + \boldsymbol{\acute{n}} \tag{4.17}$$

where $I_2 = \begin{bmatrix} 1 & 0 \\ 0 & 1 \end{bmatrix}$

where $\boldsymbol{\acute{n}}$ is such that $E\{\boldsymbol{\acute{n}}\}=0_{2x1}$ and $E\{\, \boldsymbol{\acute{n}}\, \boldsymbol{\acute{n}}^H \,\} = \|\boldsymbol{h}\|^2\, \sigma_n^2\, I_2$. The average output SNR is equal to

$$\rho_{out} = \frac{1}{\sigma_n^2}\, E\left\{\frac{E_s\left[\|\boldsymbol{h}\|^2\right]^2}{2\|\boldsymbol{h}\|^2}\right\} = \rho \tag{4.18}$$

[Oestges Claude , **MIMO Wireless Communications_From Real-World Propagation to Space-Time Code Design**].

illustrating that the Alamouti scheme does not provide any array gain owing to the lack of transmit channel knowledge .
[Oestges Claude ,**MIMO Wireless Communications_From Real-World Propagation to Space-Time Code Design**]

4.4. Multiple-Input Multiple-Output systems

With multiple antennas at both ends of the link comes the ability to exploit other leverages than diversity and array gains – it is now possible to increase the transmission throughput via the spatial multiplexing capability of MIMO channels. we will also observe that it is not possible to maximize both the spatial multiplexing and the diversity gains.

4.4.1 MIMO without transmit channel knowledge

When the transmitter has no channel knowledge, the presence of multiple antennas at both sides may allow to extract diversity and/or increase the capacity. This is achieved through the use of so-called space–time codes, which expand symbols over the antennas (i.e. over space) and over time. In the following, we introduce space–time block codes.

[Oestges Claude , **MIMO Wireless Communications_From Real-World Propagation to Space-Time Code Design**].

4.4.1.1 Space–time block coding

Consider the Alamouti scheme of MIMO 2×2 transmissions. Analogous to the MISO case, consider that two symbols $c1$ and $c2$ are transmitted simultaneously from transmit antennas 1 and 2 during the first symbol period, while symbols $-c_1^*$ and c_1^* are transmitted from antennas 1 and 2 during the next symbol period.

Assume that the flat fading channel remains constant over the two successive symbol periods, and that the 2×2 channel matrix reads as

$$H = \begin{bmatrix} h_{11} & h_{12} \\ h_{21} & h_{22} \end{bmatrix} \qquad (4.19)$$

[Oestges Claude , **MIMO Wireless Communications_From Real-World Propagation to Space-Time Code Design**].

Note that the subscripts here denote the receive and transmit antenna index and not the symbol period. The vector signal received at the receive array at the first symbol period is

$$\boldsymbol{y}_1 = \sqrt{E_s}\, \boldsymbol{H} \begin{bmatrix} c_1/\sqrt{2} \\ c_2/\sqrt{2} \end{bmatrix} + \boldsymbol{n}_1 \qquad (4.20)$$

and the vector signal received at the second symbol period is

$$\boldsymbol{y}_2 = \sqrt{E_s}\, \boldsymbol{H} \begin{bmatrix} -c_2^*/\sqrt{2} \\ c_1^*/\sqrt{2} \end{bmatrix} + \boldsymbol{n}_2 \qquad (4.21)$$

where **n**1 and **n**2 are the additive noise contributions at each symbol period over the receive antenna array (so the subscripts here denote the symbol periods, and not the antennas). The receiver forms a combined signal vector **y** as

$$y = \begin{bmatrix} y_1 \\ y_2^* \end{bmatrix} = \underbrace{\begin{bmatrix} h_{11} & h_{12} \\ h_{21} & h_{22} \\ h_{12}^* & -h_{11}^* \\ h_{22}^* & -h_{21}^* \end{bmatrix}}_{H_{eff}} \underbrace{\begin{bmatrix} c_1/\sqrt{2} \\ c_2/\sqrt{2} \end{bmatrix}}_{c} + \begin{bmatrix} n_1 \\ n_2^* \end{bmatrix} \qquad (4.22)$$

Analogous to the MISO system, both symbols $c1$ and $c2$ are spread over the two transmit antennas and over the two symbol periods. Furthermore, **H***eff* is orthogonal for all channel realizations, i.e. $H_{eff}^H H_{eff} = \|H\|_F^2 I_2$. If we compute $z = H_{eff}^H y$, we get

$$z = \begin{bmatrix} z_1 \\ z_2 \end{bmatrix} = H_{eff}^H y = \|H\|_F^2 I_2 c + \acute{n} \qquad (4.23)$$

where $\acute{n}$ is such that $E\{\acute{n}\}=0_{2x1}$ and $E\{\acute{n}\,\acute{n}^H\} = \|H\|_F^2 \sigma_n^2 I_2$.

The above equation illustrates that the transmission of $c1$ and $c2$ is fully decoupled, i.e.

$$z_k = \sqrt{E_s/2}\, \|H\|_F^2 c_k + \tilde{n}_k \qquad k=1\,,\,2 \qquad (4.24)$$

with the average output SNR given by

$$\rho_{out} = \frac{1}{\sigma_n^2} E\left\{\frac{E_s[\|H\|_F^2]^2}{2\|H\|_F^2}\right\} = 2\rho \qquad (4.25)$$

illustrating that the Alamouti scheme in a 2×2 configuration provides a receive array gain (g_a $=n_r=2$) but no transmit array gain (since the transmitter has no channel knowledge). However, it may extract the full diversity ($g_d^0 = n_t n_r = 4$)

The principle of spreading symbols over space and time is generalized through the concept of space–time block codes (STBCs). In general, these map Q symbols onto a codeword **C** of size $nt \times T$, where T is thus the duration of the codewords. The codeword **C** is usually normalized such that $E\{Tr\{CC^H\}\} = T$. As an example, the 2×2 Alamouti scheme (T =2, nt =2, Q=2) is represented by the following
codeword matrix

$$C = \frac{1}{\sqrt{2}} \begin{bmatrix} c_1 & -c_2^* \\ c_2 & c_1^* \end{bmatrix} \qquad (4.26)$$

The spatial multiplexing rate of a space–time block code is then defined as $r_s = \frac{Q}{T}$ and a space–time block code is full-rate when $r_s = n_t$. The Alamouti scheme is therefore characterized by r_s =1.

The Alamouti scheme described above is inserted under the orthogonal STBCs (O-STBCs) . O-STBCs transmit one or less independent symbol per symbol period over the n_t transmit antennas. They provide an array gain of n_r and extract the full diversity gain of $n_t n_r$. Also they allow for a direct detection since vector detections are converted into much less complex scalar detections . For complex constellations, O-STBCs with r_s =1 only exist for n_t =2. Otherwise, complex O-STBCs for arbitrary n_t offer spatial multiplexing rates r_s <1.

4.4.2. Space-frequency coding

In frequency selective channels, it is possible to exploit the additional frequency diversity by coding not only across space (i.e. across antennas) but also across the frequency band, e.g. using orthogonal frequency division multiplexing (OFDM). This technique, known as SF MIMO-OFDM, together with alternative coding schemes over frequency selective channels,

4.4.3. Space–time coded MIMO-OFDM

Space–time coded MIMO-OFDM spreads information symbols in space and time the codewords are defined in a analogous to space–time coding for flat fading channels.

different way, as they are sent on different OFDM symbols on a per tone basis. Each tone simply acts as one parallel channel. Therefore, space–time codes designed for flat fading channels can be directly used for space–time coded MIMO-OFDM.

جامعة بوليتكنك فلسطين

الدائرة المالية
محاسبة أقساط الطلبة

الفصل الدراسي الصيفي 2010/2009

إشعار الدفع " تسجيل "

1267

تاريخ طباعة الإشعار: /06/2010
5:06 am

رقم الطالب: 08/2112

اسم الطالب: عروبه سالم احمد بريوش

التخصص: إدارة الأعمال المعاصرة

عدد الساعات المسجلة: 9

الرسوم المطلوبة: 225.00 دينار أردني

الرسوم المطلوبة بالحروف دينار أردني مائتان و خمسة و عشرون دينار أردني

الرجاء التوجه الى احد فروع البنوك التالية للدفع في موعد أقصاه 72 ساعة من تاريخ طباعة الإشعار.

بنك الإسكان للتجارة والتمويل	حساب رقم 34300
البنك الإسلامي العربي	حساب رقم 15950

لإستخدام الطلبة فقط.

- الرجاء التوجه للبنك مصطحبا معك إشعار الدفع مع ضرورة ابقاء الإشعار لدى موظف البنك.
- الرجاء التأكد من صحة رقمك الجامعي على وصل الدفع الخاص بالبنك.

لإستخدام البنك فقط.

- يجب استلام كامل المبلغ حسب إشعار الدفع
- في حال الدفع في أحد الفروع خارج فرع مدينة الخليل يرجى إرسال نسخة بالفاكس الى فرع الخليل

1

Chapter 5

Detailed System Design

er Five

iled System Design

In this chapter a detailed description of the our system simulation design ,including a charts ,modules and some detailed description of the system components.

n Options

is section we demonstrate design option for the WiMAX system simulation ,and discuss ose Matlab in implementing the system ,in term of output format .

put Format

In our system, we design Simulation which represents the WiMAX based on OFDM ,to n and analyze the WiMAX system especially the physical layer (PHY) based on OFDM, matlab codes .

.1 Matlab

MATLAB stands for "Matrix Laboratory" and is a numerical computing environment ourth-generation programming language. Developed by The MathWorks, MATLAB s matrix manipulations, plotting of functions and data, implementation of algorithms, on of user interfaces, and interfacing with programs written in other languages, including +, and Fortran.

MATLAB was first adopted by control design engineers, Little's specialty, but LAB used in a wide range of applications, including signal and image processing, munications, control design, test and measurement, financial modeling and analysis, and putational biology. Add-on toolboxes (collections of special-purpose MATLAB functions, eparately) extend the MATLAB environment to solve particular classes of problems plication areas.

TLAB provides a number of features for documenting and sharing your work. You ate your MATLAB code with other languages and applications, and distribute your algorithms and applications.

Vhy MATLAB (Advantage of MATLAB)

ATLAB supports structure data types. Since all variables in MATLAB are arrays, a quate name is "structure array", where each element of the array has the same field addition, MATLAB supports dynamic field names (field look-ups by name, field ions etc). Unfortunately, MATLAB JIT does not support MATLAB structures, ust a simple bundling of various variables into a structure will come at a cost.

some important feature for the MATLAB:

atlab Perform signal processing, analysis, and algorithm development.

atlab Design and simulate signal processing systems which provides algorithms and ols for the design and simulation of signal processing systems. You can develop DSP gorithms for speech and audio processing, signal detection, radar tracking, baseband mmunications, and other applications.

Designs and analyzes algorithms for the physical layer of communication systems here computing environment with functions, plots, and a graphical user interface GUI) for exploring, designing, analyzing, and simulating algorithms for the physical yer of communication systems.

latlab supports physical layer where arrays represent the signals in your ommunication system. As a result, you can divide the physical layer into a cascade of lgorithms. Each algorithm is represented by a MATLAB function that acts on the

arrays. By cascading the output array of one function as an input array to the next
on, Communications Toolbox helps you focus your design on the transmitter,
el, or receiver.

nunications Toolbox in Matlab provides specialized plots for communication
eering, such as eye diagrams, constellation plots, and bit error rate versus signal-
se ratio plots. With the BER Tool GUI, you can combine data from several
ation runs and compare the results with theoretical benchmarks in one combined
vindow.

e many other languages, where the semicolon is used to terminate commands, in
e semicolon serves to suppress the output of the line that it concludes (it serves a
ose in Mathematica.) and this help us to test each function separately and help to
umulative error.

r important reason for choosing Matlab that it have built-in commutation
ich aims the real signals .

ject-Oriented Programming

LAB's support for object-oriented programming includes classes, inheritance,
atch, packages, pass-by-value semantics, and pass-by-reference semantics and
an call functions and subroutines written in the C programming language or
wrapper function is created allowing MATLAB data types to be passed and

ecture Of The System

WiMAX based on OFDM system simulation consists of many functions that
th the user to generate random data that passes through many processes during
channel receiver .Where all these processes were implemented in our simulation in
ement the PHY based on OFDM of WiMAX as if it is implemented in practical life

annel model represents real channel scenarios . Next Generate plots to represent ion to SNR in aim analyzing the results later .

description of the main funtions

	Description
n	used at the transmitter to Randomize data.
zer	used at the receiver to recover the original data again from the Randomized data.
	Reed-Solomon useful for burst-error correction (for this reason it is useful for space communication).
	All encoded data bits shall be interleaved by a block interleaver with a block size corresponding to the number of coded bits per the allocated sub channels per OFDM symbol (Ncbps).
aver	It performs the inverse of the operation of the Interleaver.
ping	the sequence of binary bits is converted to a sequence of complex valued symbols.
	Inverse Fast Fourier transform create a multitude of orthogonal sub carriers using a single radio. Also creates an **ISI**-free channel, if the channel provides a circular convolution.
	It performs the inverse of the operation of the IFFT.
x	preventing interference between X,X‘ ; the cyclic prefix realizes the purpose of the OFDM by making each sub channel see a flat channel response.
nal	This function perform the convolutional encoding to the serial bit stream , the convolutional encoding gives a memory encoding that reduces the effect of the selective fading channel , it uses the trellis structure over the data that will be encoded

er	A Viterbi decoder uses the Viterbi algorithm for decoding a bit stream that has been encoded using Forward error correction based on a Convolutional code.
	This function applies the effect of the fading channel to the transmitted data , so that testing the our system under real channel scenarios (SUI 1 to SUI 6) , where in this function ,we evaluate the channel impulse response (CIR) according to the mobile communication model (Jakes model) , under the different conditions of the SUI channels that represent the all environmental conditions ,also we design this function to be suitable for diversity and MIMO system , in other word ,the generated CIR depends on the number of antenna at the receiver and at the transmitter , so the user can generate the desired CIR for your simulation .
ion	This function generates the required data according the input parameters
	This function generate a BER plots under the diversity and MIMO system
ersity	This function implement the receiver diversity for 2 antennas and 4 antennas
	This function implement the transmitter diversity for 2 antennas and 4 antennas

er	A Viterbi decoder uses the Viterbi algorithm for decoding a bit stream that has been encoded using Forward error correction based on a Convolutional code.
	This function applies the effect of the fading channel to the transmitted data , so that testing the our system under real channel scenarios (SUI 1 to SUI 6) , where in this function ,we evaluate the channel impulse response (CIR) according to the mobile communication model (Jakes model) , under the different conditions of the SUI channels that represent the all environmental conditions ,also we design this function to be suitable for diversity and MIMO system , in other word ,the generated CIR depends on the number of antenna at the receiver and at the transmitter , so the user can generate the desired CIR for your simulation .
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	This function implement the transmitter diversity for 2 antennas and 4 antennas

A Viterbi decoder uses the Viterbi algorithm for decoding a bit stream that has been encoded using Forward error correction based on a Convolutional code.

This function applies the effect of the [illegible] transmitted signal . These [illegible] channel models are built in such a way that [illegible] these [illegible] values. [illegible] speed [illegible] check any [illegible] three main [illegible] these types are

[illegible] ; These channels specify the [illegible] ,in other words ; They determine [illegible] ,for example the SUI 1 represents the lowest [illegible] SUI 2 represents the highest fading effects in [illegible] standardized for broadband wireless access(BWA).

nel Model

heck the performance of any wireless communication system , the system must be real wireless channel scenarios ; these channel models are built in such a way that different propagation effects that affects the wireless transmitted signal . These effects can be classified in two main parameters ; the fading characteristic and

nown , the channel parameters are random in nature , so to characterize these nathematically , only statistical methods are used for this object.

statistical method can be characterized in term of the variance and mean values. neters depend on the environment characteristics , i.e ,the tree density , wind speed ight , building density and earth's surface characteristics .

rder to implement these channel characteristics in software design to check any tion system performance , Stanford University Interim (SUI) introduce a three main s that represent the most channel characteristics in nature . these types are by

- Soft fading (SUI 1, SUI 2)
- Intermediate fading (SUI 3, SUI 4)
- Harsh fading (SUI 5, SUI 6)

h of them is characterized by two channel models ; These channels specify the and minimum values of the fading characteristic ,in other words ; They determine f fading characteristic in each type . For example the SUI 1 represents the lowest cts in the soft fading region , while the SUI 2 represents the highest fading effects in .And these channel models are standardized for broadband wireless access(BWA).

The following table summarize the six SUI channel characteristics

Table (5.1) : SUI channel characteristics

	SUI 1	SUI 2	SUI 3	SUI 4	SUI 5	SUI 6
elay	Low			Moderate	High	
	High		low	Here no K factor since there is no LOS in this scenarios		
er	Low		High		Moderate to high	
shift	Low			High	Low	High
ype	Flat terrain , low tree densities ,		Moderate tree densities with flat terrain , or not flat terrain with light tree densities		Hilly terrain with high tree densities	

following tables show the SUI channel parameters ,where these parameters are
der the following scenario.
ceg, K.V.S. Hari, M.S. Smith, D.S. Baum et al, "*Channel Models for Fixed*
plications", IEEE 802.16.3 Task Group Contributions 2001, Feb. 01]

Size: 7Km

antenna height: 30 m

eive antenna height: 6m

antenna beamwidth: 120^0

eive antenna beamwidth: omnidirectional

arization: Vertical only

% cell coverage with 99.9% reliability at each location covered

Table (5.2) SUI channel parameters

o power 3)			Spread delay (μ sec)			K factor			Doppler (Hz)		
p1	Tap2	Tap3	Tap1	Tap2	Tap3	Tap1	Tap2	Tap3	Tap1	Tap2	Tap3
	-15	-20	0	0.4	0.9	4	0	0	0.4	0.3	0.5
	-12	-15	0	0.4	1.1	2	0	0	0.2	0.15	0.25
	-5	-10	0	0.4	0.9	1	0	0	0.4	0.3	0.5
	-4	-8	0	1.5	4	0	0	0	0.2	0.15	0.25
	-5	-10	0	4	10	0	0	0	2	1.5	2.5
	-10	-14	0	14	20	0	0	0	0.4	0.3	0.5

re Simulation

urpose of this part is to program a Matlab code that produce channel coefficients generate the channel impulse response (CIR) .

CIR is then convolved with the transmitted signals to test WiMAX performance eal channel scenarios .

Distribution

se the method of filtered noise to generate channel coefficients with the specified and spectral power density. For each tap a set of complex zero-mean Gaussian umbers is generated with a variance of 0.5 for the real and imaginary part, so that rage power of this distribution is 1.

yields a normalized Rayleigh distribution for the magnitude of the complex If a Ricean distribution (K>0) is needed, a constant path component *m* has to be Rayleigh set of coefficients.

ratio of powers between this constant part and the Rayleigh (variable) part is the K-factor.

er P of each tap is :

$$p = |m|^2 + \sigma^2 \qquad (5.1)$$

s the complex constant

the variance of the complex Gaussian set.

atio of powers is

$$k = \frac{|m|^2}{\sigma^2} \qquad (5.2)$$

quations , the power of the complex Gaussian and the power of the constant part

$$\sigma^2 = \frac{p}{1+k} \ , \qquad (5.3)$$

$$|m|^2 = p\frac{k}{k+1} \qquad (5.4)$$

eqns. 5.3 we can see that for K= 0 the variance becomes P and the constant part ishes , as expected .

er Spectrum

SUI channel model defines a specific power spectral density (PSD) function for component channel coefficients called 'rounded' PSD which is given as

$$\begin{array}{ll} - 1.72\, f_0^2 + 0.785\, f_0^4 & |f_0| \leq 1 \\ & |f_0| > 1 \end{array} \qquad (5.5)$$

$$f_0 = \frac{f}{f_m}$$

f_m : maximum frequency of the signal

t a set of channel coefficients with this PSD function ,we correlate the original with a filter which amplitude frequency response is

$$H(f) =. \sqrt{S(f)} \tag{5.6}$$

e no frequency components higher than *fm*, the channel can be represented with mpling frequency of 2 *fm*, according to the Nyquist theorem.

er of the filter has to be normalized to one, so that the total power of the signal is y it .

a Correlation

nel models define an antenna correlation, which has to be considered if multiple ceive elements, i.e. multiple channels, are being simulated.

elation is commonly defined as the envelope correlation coefficient between ed at two antenna elements.

t equivalent taps in both channels have equal power: and that taps with different correlated within a channel as well as between channels:

$$E\{g_{lk}(t) g^{*}_{ji}(t)\} = 0 \qquad \forall k \sim = l \tag{5.7}$$

$k, l \in [1..3]$; $i, j \in [1..2]$

correlation coefficient becomes:

$$\rho_{env} = \left| \frac{\rho_1 \sigma_1^2 + \rho_2 \sigma_2^2 + \rho_3 \sigma_3^2}{\sigma_1^2 + \sigma_2^2 + \sigma_3^2} \right| \tag{5.8}$$

e the correlation coefficients between each of the 3 pairs of taps $g_{1l}(t)$ and $g_{2l}(t)$. So (5.8) states that all tap correlations have to be set to the antenna correlation. lation of the SUI channel all tap correlations equal to the antenna correlation . enerate a sequence of random state vectors with mean vector μ and correlation he following transformation is used :

$$\tilde{V} = R^{1/2}V + \mu \qquad (5.9)$$

vector of independent sequences of Gaussian-distributed random numbers with identical variance.

rrelation matrix R is defined as:

$$\begin{matrix} 0 & \rho_{env} & \rho_{env} \\ 1 & \rho_{env} & \rho_{env} \\ \rho_{env} & 1 & 0 \\ \rho_{env} & 0 & 1 \end{matrix}\right] \qquad (5.10)$$

Correlating complex sequences, not all correlations between real and imaginary put sequences are set to the specified envelope correlation coefficient .

fore, we divide each complex sequence into real and imaginary part and real-valued sequences. For example, to correlate two complex sequences X and ector V and the correlation matrix R are set as:

$$\Bigg] , \qquad R = \begin{bmatrix} 1 & 0 & \rho_{env} & \rho_{env} \\ 0 & 1 & \rho_{env} & \rho_{env} \\ \rho_{env} & \rho_{env} & 1 & 0 \\ \rho_{env} & \rho_{env} & 0 & 1 \end{bmatrix} \qquad (5.11)$$

nplement correlated channels in our simulation . First two independent but equally et of channel coefficients are created .

we use eqs. 5.9 and 5.11 to correlate the random signals of equivalent taps in annels.

owchart

chapter we describe the system's flow chart for the used function using MS

al system's flow chart

low chart shows the general flow chart for the Wimax system Based on OFDM he sub-functions in our simulation model . Where in this flow chart , the most all ons appear , and it defines the data flow between these functions , also describes between these functions in a clear way .

e (5.1) shows the general flow chart ; We note that the flow chart consists of four art is the transmitter, second part is the channel ,third part is the receiver and forth ER calculation.

smitter part consists of the data generation ,then this data is randomized ,after that the forward error correction(Reed Solomon - Convolutional encoding), then the a is interleaved using block interleaver. After that is mapped using specified scheme (i.e BPSK ,QPSK ,16-QAM ,64-QAM ,256-QAM). Finally the data is analog using (IFFT) and then is transmitted at specific carrier frequency (i.e 2.5

nel part consist of generating channel coefficients (random phases and that is used next to generate a channel impulse response (CIR) . Then this CIR is vith the transmitted signal . After that an Gaussian noise is added to the faded

eceiver part , the transmitter processes are complimented and arranged in reverse

l stage ; Computing the BER (Bit Error Rate). By comparing received data with a ; so as to compute the number of errors. Then this number of errors is divided by r of transmitted bits to give the BER that will be stored in an array. n this BER is plotted versus the SNR to represent the performance of the WiMAX

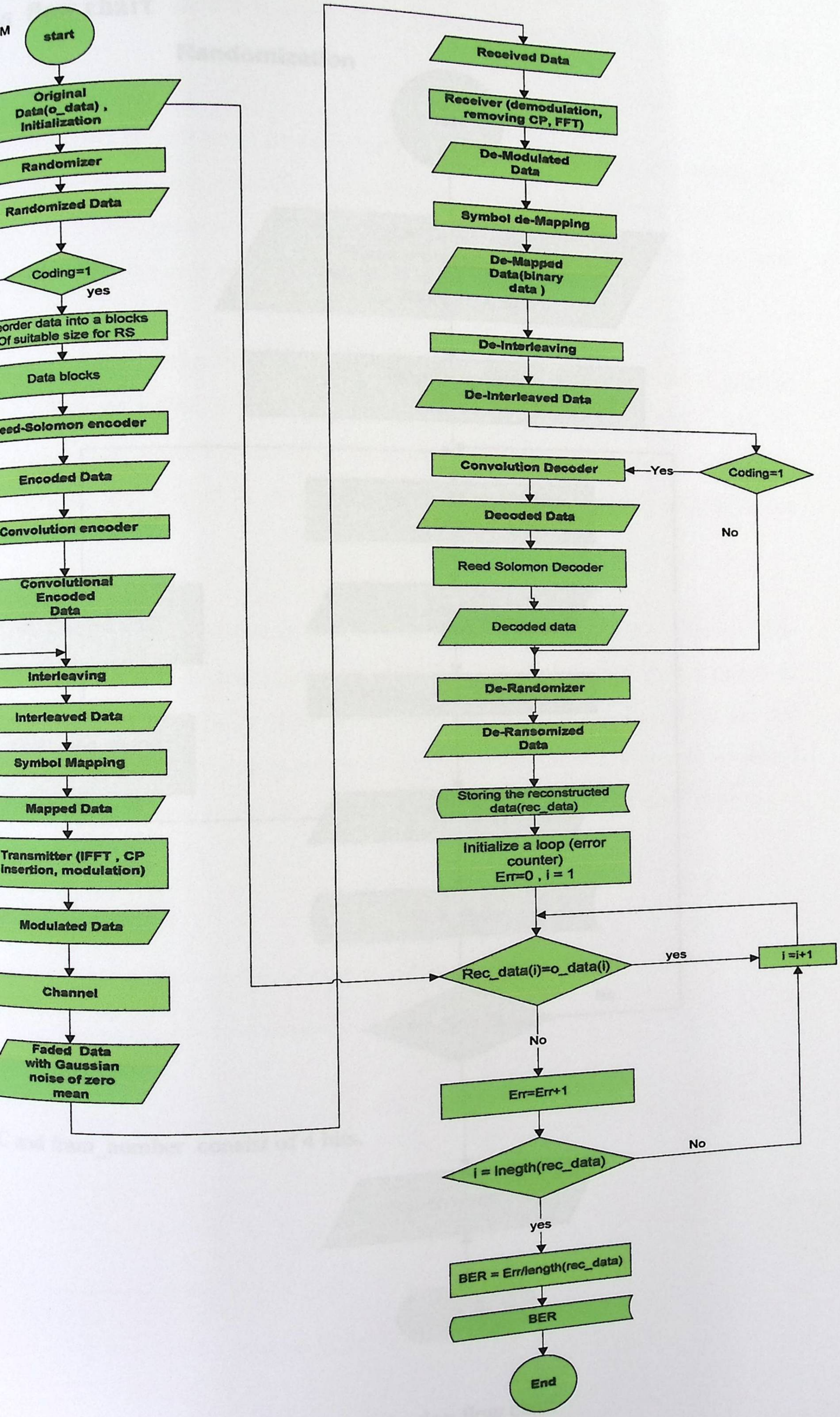
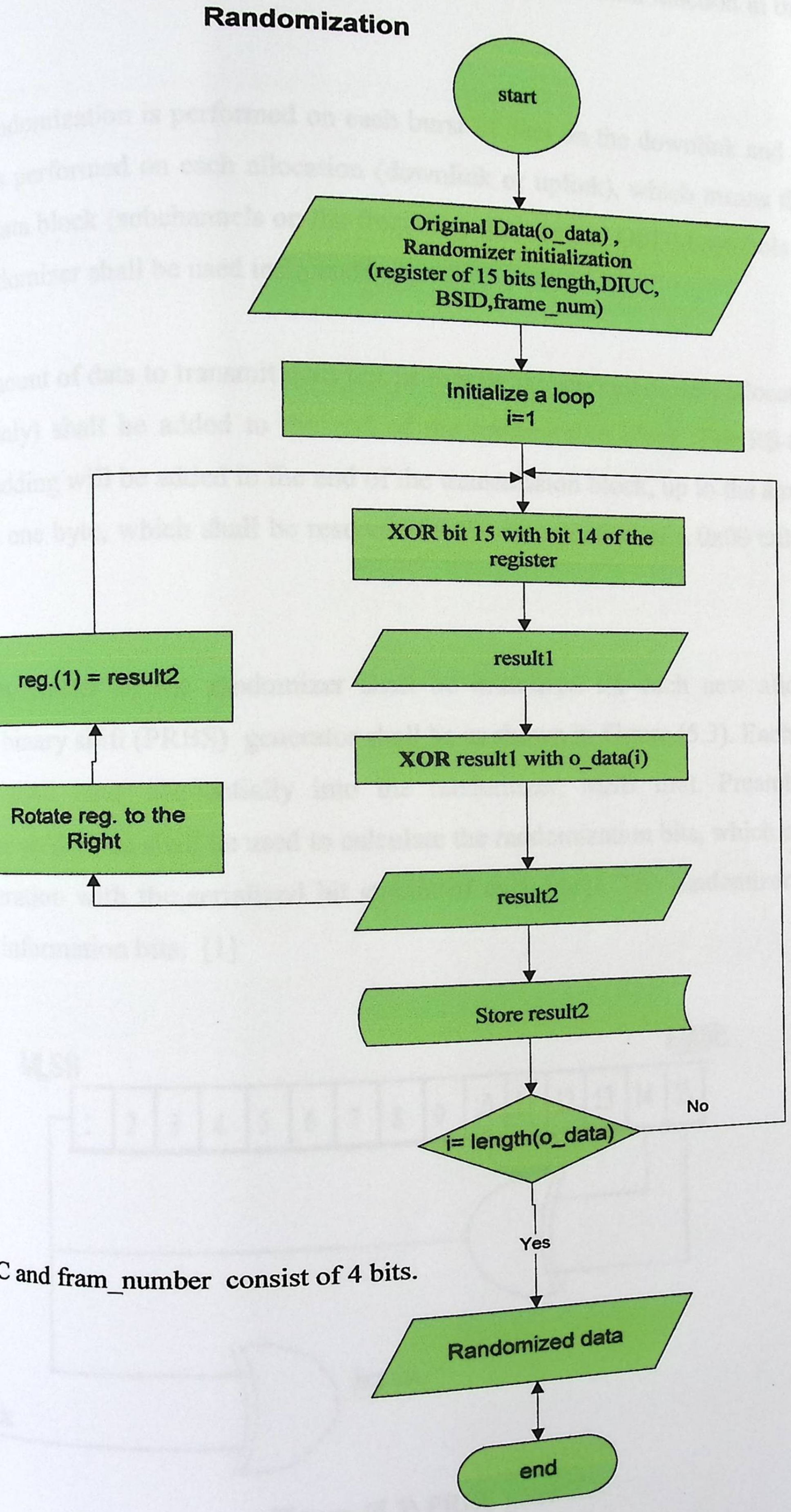


Figure (5.1) : General flow chart of WiMAX system

n flow chart



C and fram_number consist of 4 bits.

Figure (5.2) : Randomization flow chart

.2) shows the detailed flow chart for the Randomization function in the WiMAX

ndomization is performed on each burst of data on the downlink and uplink. The s performed on each allocation (downlink or uplink), which means that for each ata block (subchannels on the frequency domain and OFDM symbols on the time lomizer shall be used independently.

ount of data to transmit does not fit exactly the amount of data allocated, padding nly) shall be added to the end of the transmission block. For RS-CC and CC dding will be added to the end of the transmission block, up to the amount of data one byte, which shall be reserved for the introduction of a 0x00 tail byte by the

ft-register of the randomizer shall be initialized for each new allocation. The binary shift (PRBS) generator shall be as shown in Figure (5.3). Each data byte to shall enter sequentially into the randomizer, MSB first. Preambles are not e seed value shall be used to calculate the randomization bits, which are combined eration with the serialized bit stream of each burst. The randomizer sequence is information bits. [1]

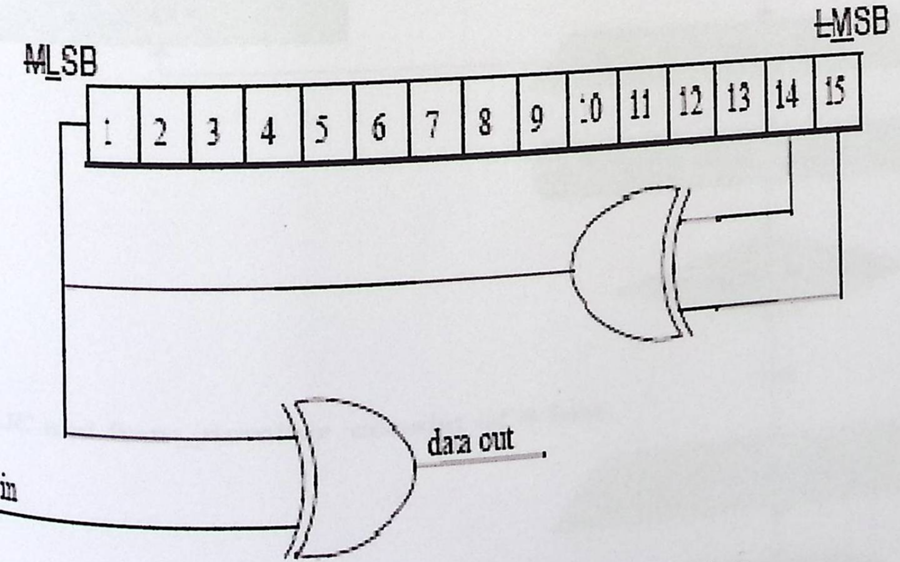
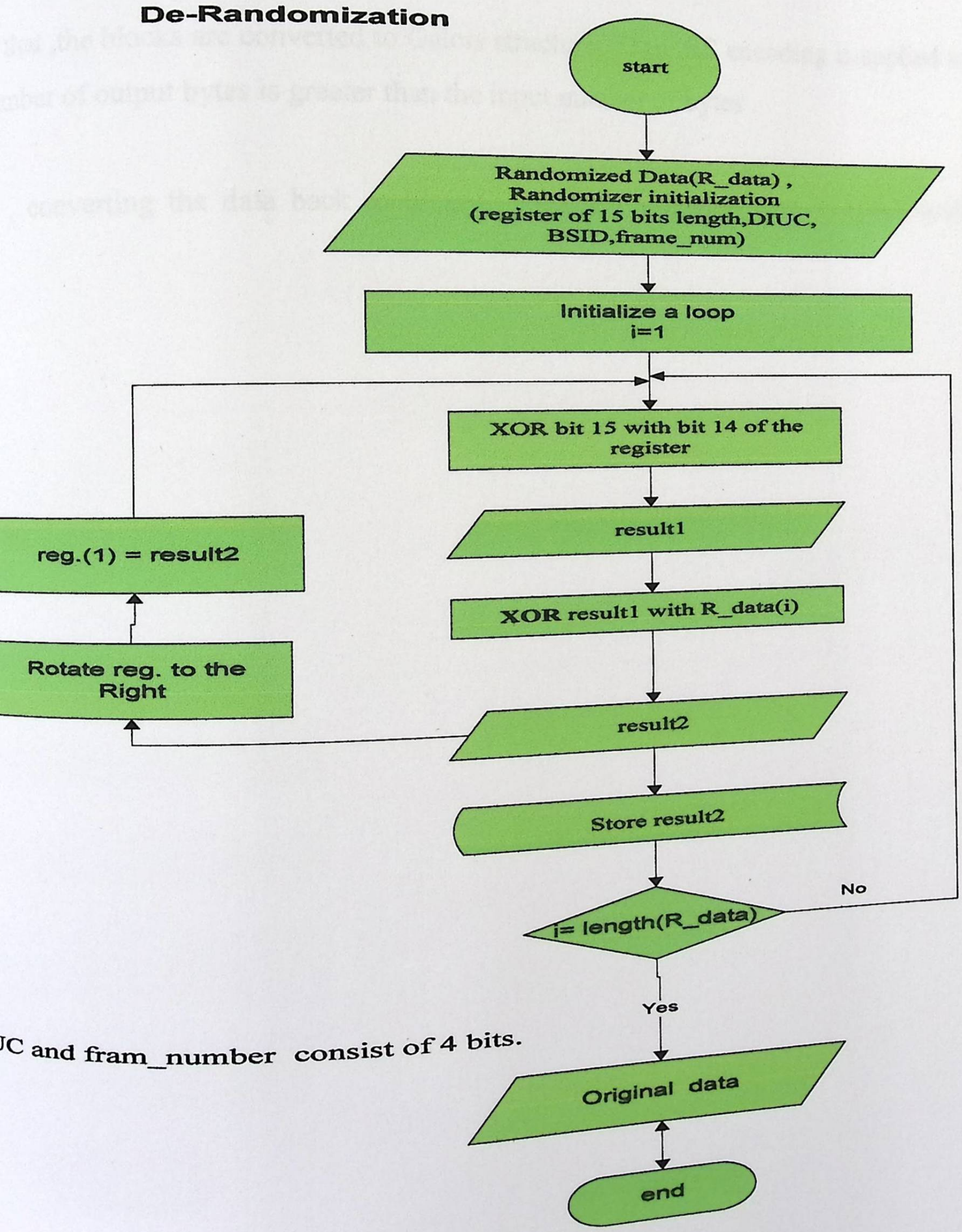


Figure (5.3) PRBS generator

ndomization Function's flow chart

ed at the receiver to recover the original data again from the Randomized data. It onstruction of the Randomizer, as the data has an XOR operation with the output has a linear feedback shift register (LFSR) has the same seed value of the sed at the Transmitter.

e (5.4) shows the detailed flow chart for the De-Randomization function in the em.



JC and fram_number consist of 4 bits.

Figure (5.4) : De-randomization flow char

Solomon(RS) -Encoder flow chart

e (5.5) shows the detailed flow chart for the Reed Solomon-Encoding
e WiMAX system.

omized data is encoded using RS encoder as shown in figure (5.5) . As we see ;
domized data is arranged in a blocks ..Then every block is converted to a one
ber that meets the binary contents of the block.

that ,the blocks are converted to Galois structure. Then ,RS encoding is applied to
mber of output bytes is greater than the input number of bytes .

, converting the data back to binary . Finally , arranging data in row (serial

nput bytes
utput bytes
its that can be
N-k)/2)
eed-salmon

array
the field

Reed Solomon Encoder

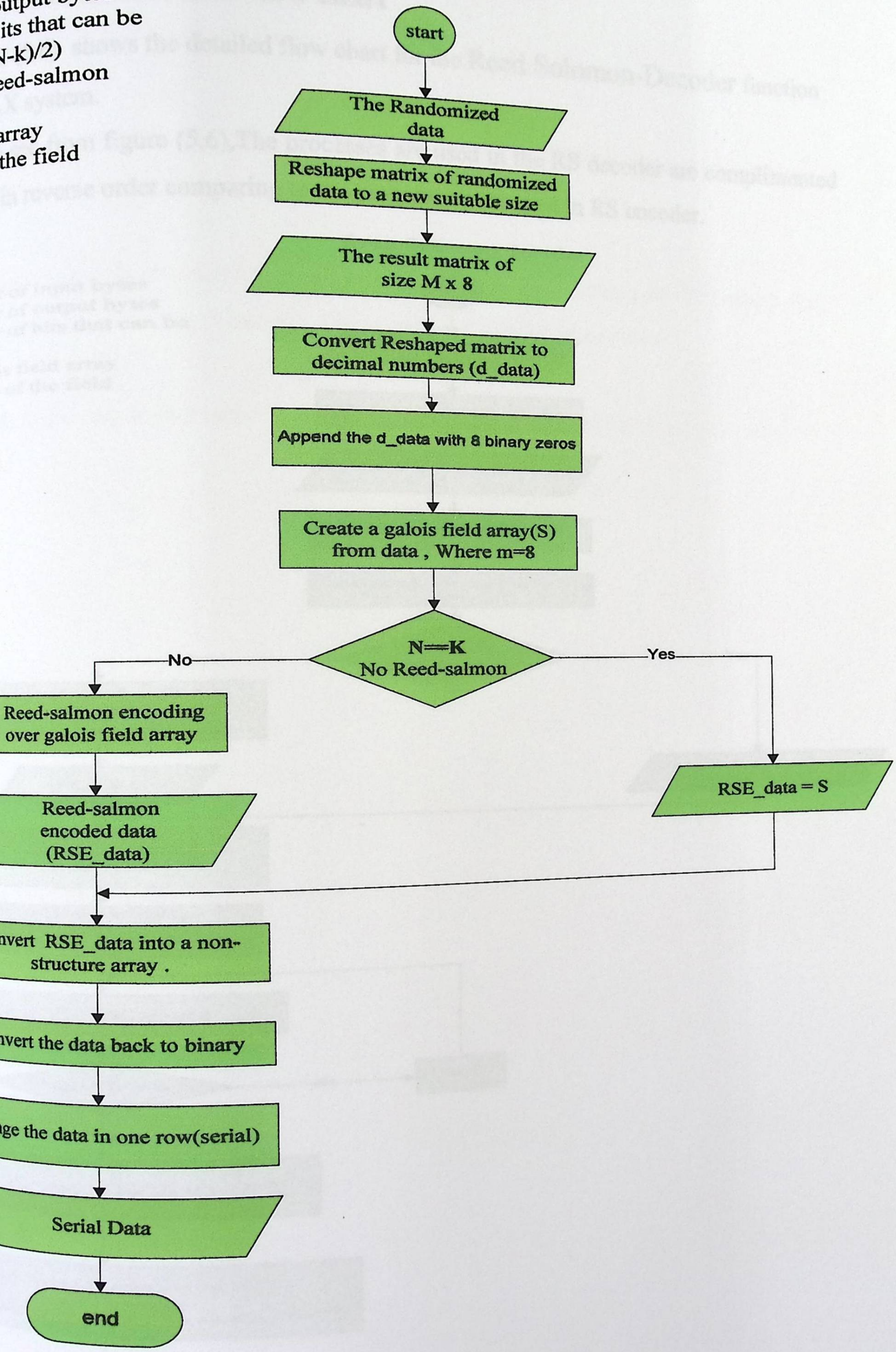


Figure (5.5) : Reed Solomon Encoder flow chart

Solomon-Decoder flow chart

e (5.6) shows the detailed flow chart for the Reed Solomon-Decoder function X system.

see from figure (5.6),The processes are used in the RS decoder are complimented in reverse order comparing to the processes were used in RS encoder.

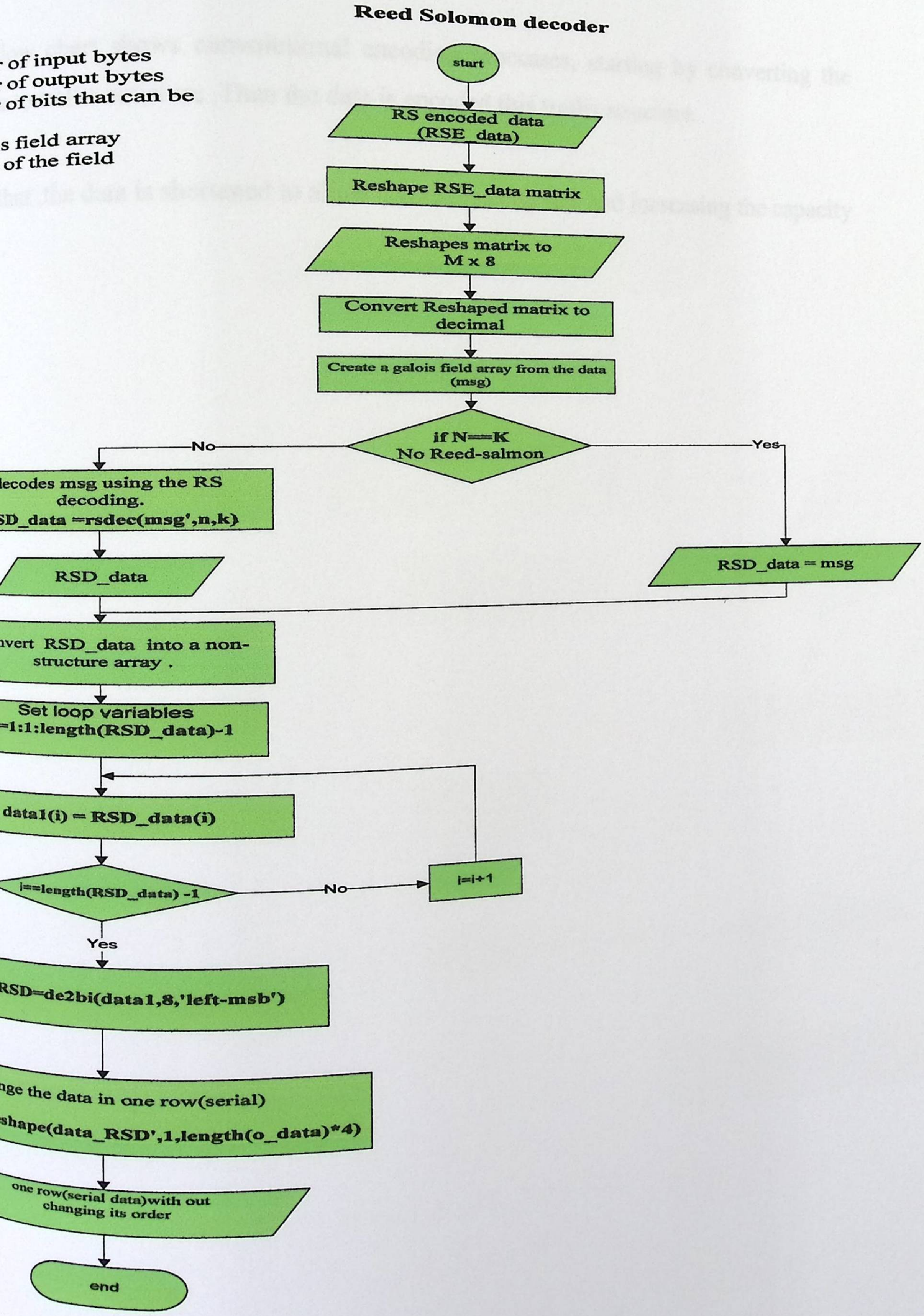


Figure (5.6) : Reed Solomon Decoder flow chart

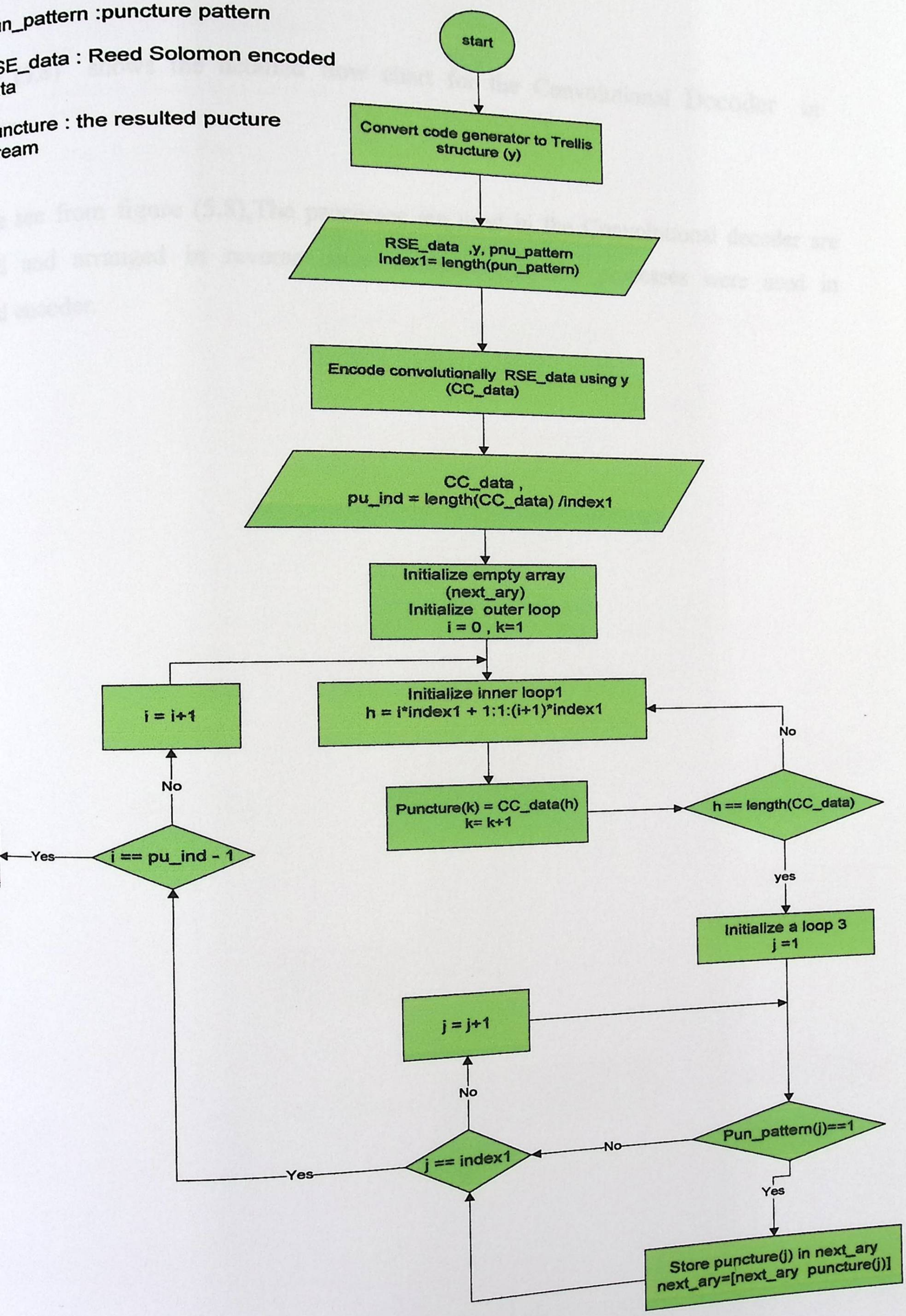
olutional encoder flow chart

e (5.7) shows the detailed flow chart for the Convolutional encoding in em.

low chart shows convolutional encoding processes, starting by converting the r to trellis structure .Then the data is encoded this trellis structure.

that ,the data is shortened to allow a variable code rate and increasing the capacity .

Convolutional encoder


n_pattern :puncture pattern

SE_data : Reed Solomon encoded
ta

ncture : the resulted pucture
eam

Figure (5.7) : Convolutional Encoder flow chart

olutional decoder flow chart

(5.8) shows the detailed flow chart for the Convolutional Decoder in em.

see from figure (5.8),The processes are used in the Convolutional decoder are l and arranged in reverse order comparing to the processes were used in l encoder.

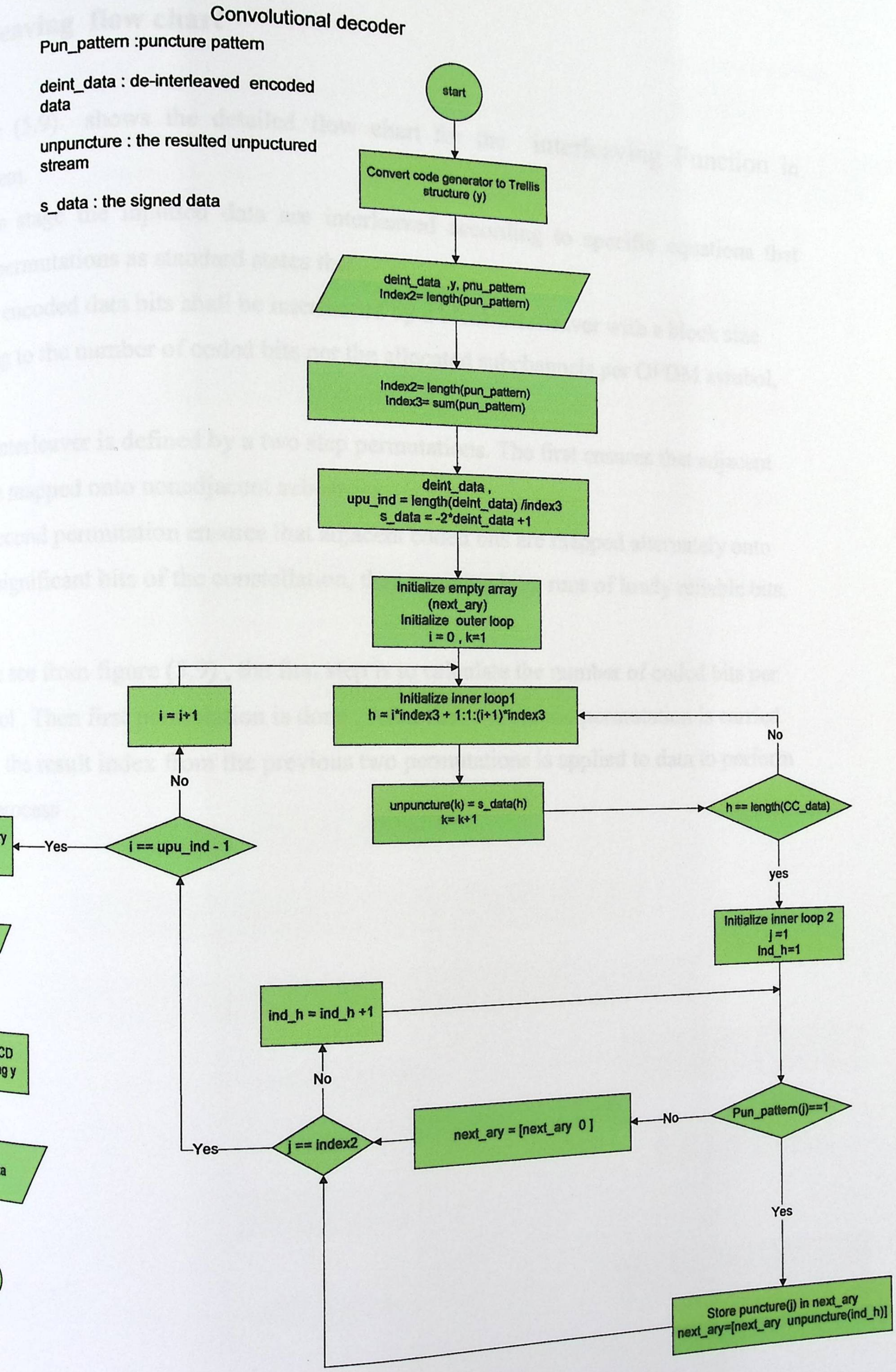


Figure (5.8) : Convolutional Encoder flow chart

eaving flow chart

(5.9) shows the detailed flow chart for the interleaving Function in
em.

s stage the inputted data are interleaved according to specific equations that
permutations as standard states that .

encoded data bits shall be interleaved by a block interleaver with a block size
g to the number of coded bits per the allocated subchannels per OFDM symbol,

interleaver is defined by a two step permutations. The first ensures that adjacent
e mapped onto nonadjacent subcarriers.

econd permutation ensures that adjacent coded bits are mapped alternately onto
significant bits of the constellation, thus avoiding long runs of lowly reliable bits.

e see from figure (5.9) , the first step is to calculate the number of coded bits per
bol . Then first permutation is done . After that, the second permutation is carried
, the result index from the previous two permutations is applied to data to perform
process .

eaving flow chart

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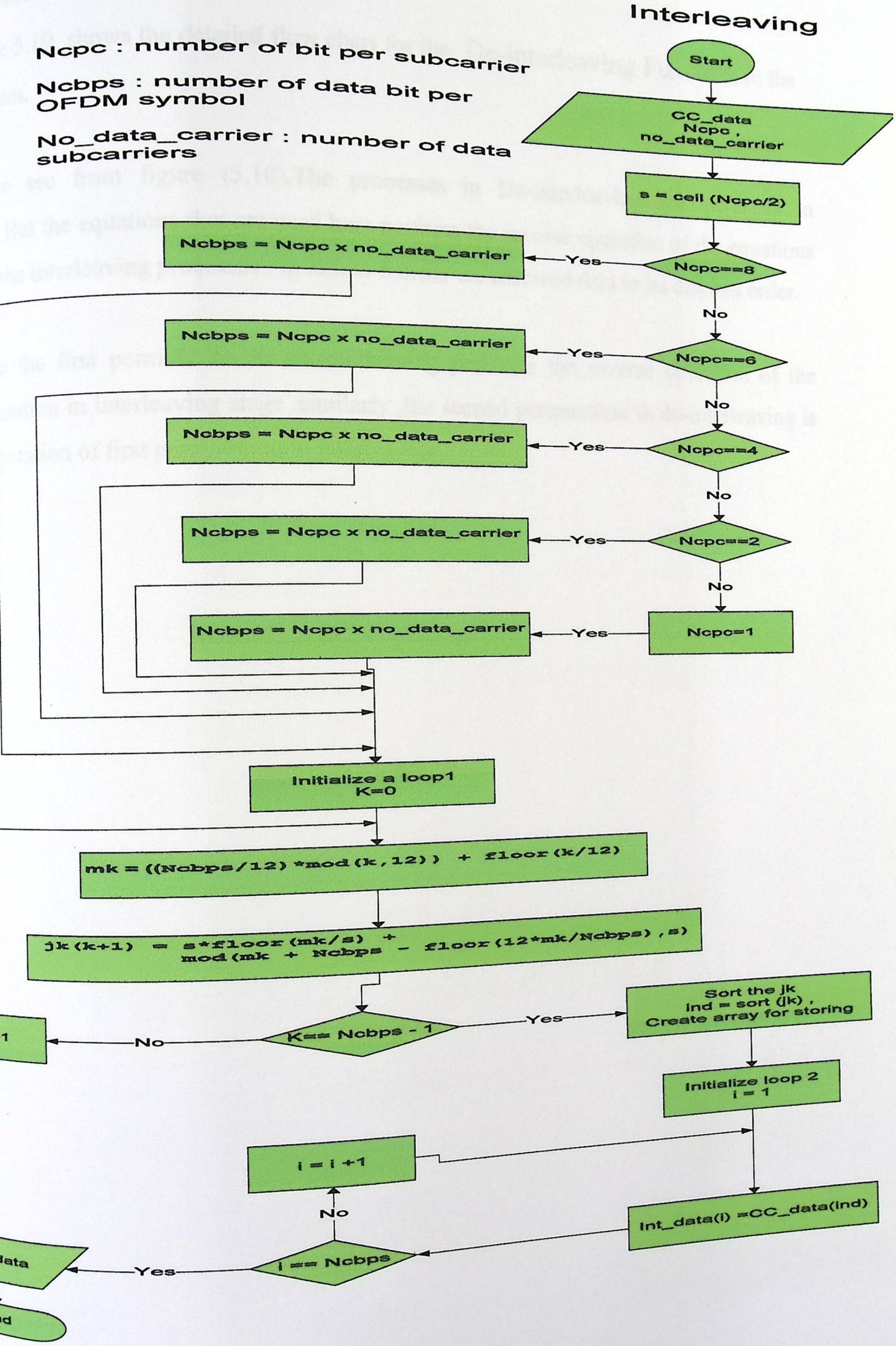

Figure (5.9) : Interleaving flow chart

nterleaving flow chart

e 5.10 shows the detailed flow chart for the De-interleaving Function in the em.

e see from figure (5.10),The processes in De-interleaving are same as in But the equations that are used here perform the reverse operation of the equations the interleaving processes . In aim to reorder the received data to its original order.

e the first permutation in de-interleaving performs the inverse operation of the utation in interleaving stage .similarly ,the second permutation in de-interleaving is peration of first permutation in interleaving.

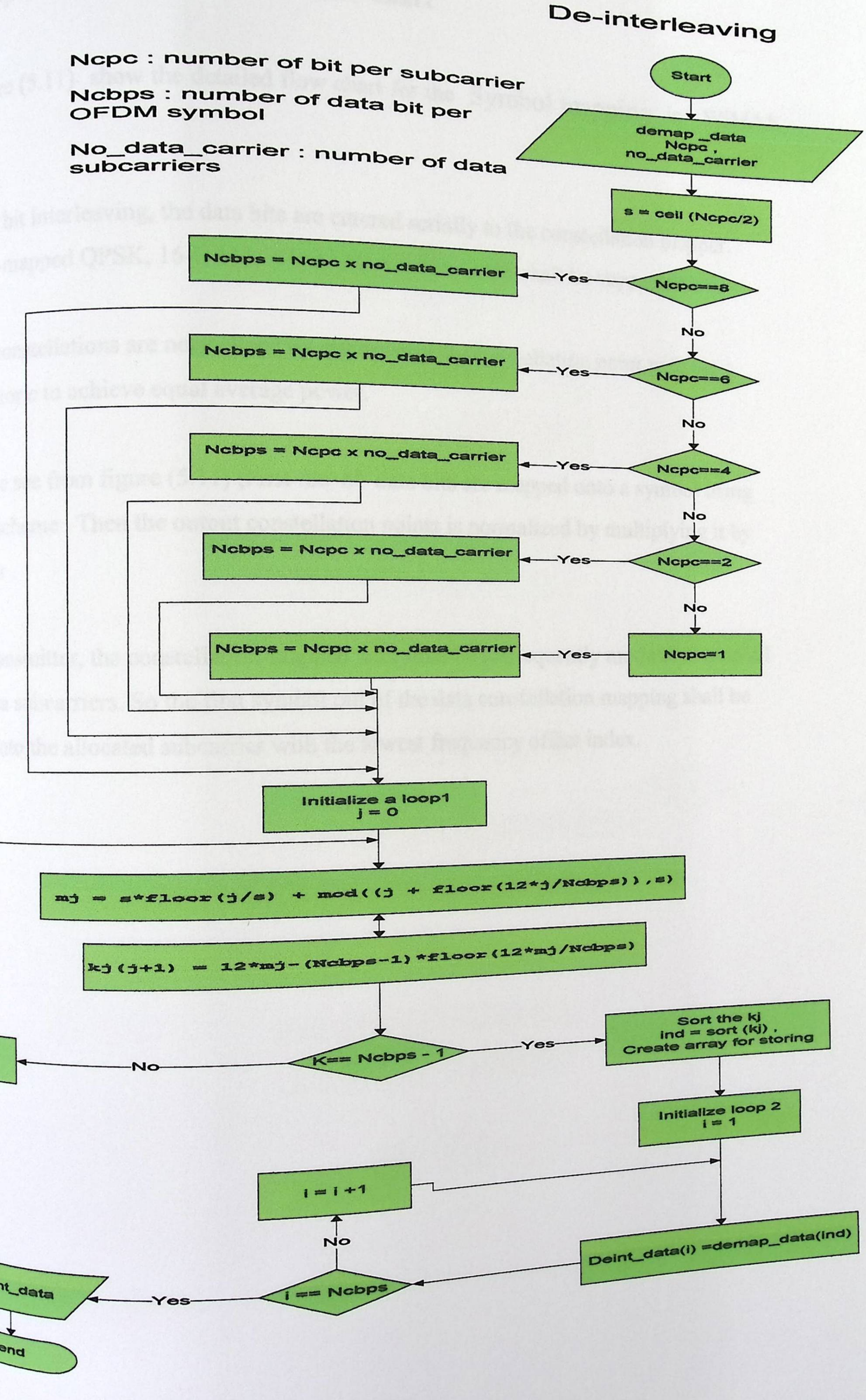

Figure (5.10) : De-interleaving flow chart

mbol mapping Function's flow chart

re (5.11) show the detailed flow chart for the Symbol mapping in WiMAX

bit interleaving, the data bits are entered serially to the constellation mapper. -mapped QPSK, 16-QAM, 64-QAM, and 256-QAM shall be supported.

constellations are normalized by multiplying the constellation point with the tor *c* to achieve equal average power.

e see from figure (5.11) ;First the M data bits are mapped onto a symbol using scheme . Then the output constellation points is normalized by multiplying it by r .

ansmitter, the constellation-mapped data shall be subsequently modulated onto all a subcarriers. So the first symbol out of the data constellation mapping shall be nto the allocated subcarrier with the lowest frequency offset index.

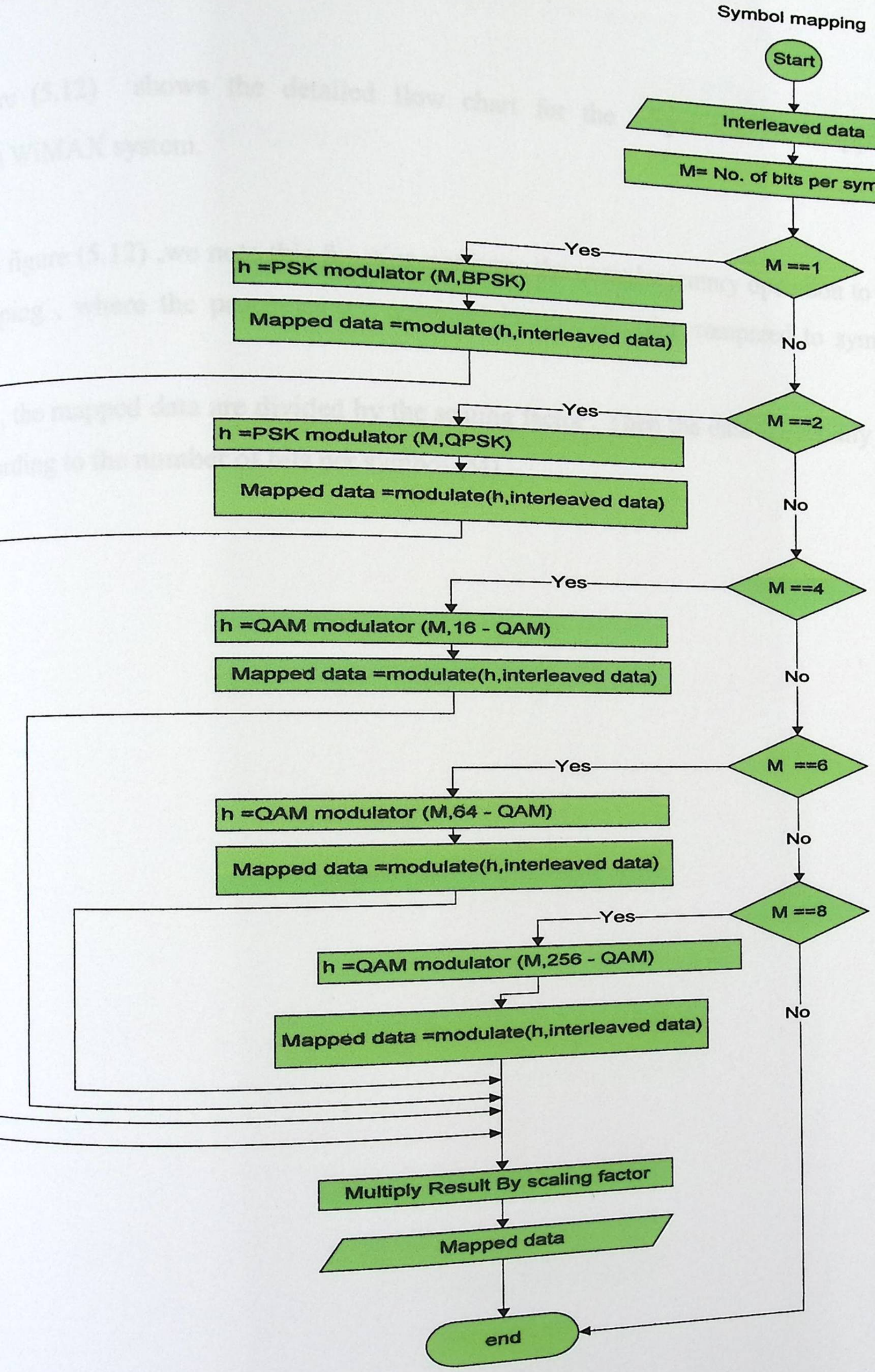

Figure (5.11) : Symbol mapping flow chart

bol De-mapping flow chart

re (5.12) shows the detailed flow chart for the Symbol De-mapping
WiMAX system.

figure (5.12) ,we note this function performs the complementary operation to the
ping , where the processes are arranged in reverse order compared to symbol

, the mapped data are divided by the scaling factor . Then the data are serially de-
rding to the number of bits per symbol (M) .

bol De-mapping flow chart

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, the mapped data are divided by the scaling factor . Then the data are serially de-
rding to the number of bits per symbol (M) .

of subcarriers

(time of cyclic
symbol time)

fix data

er frequency

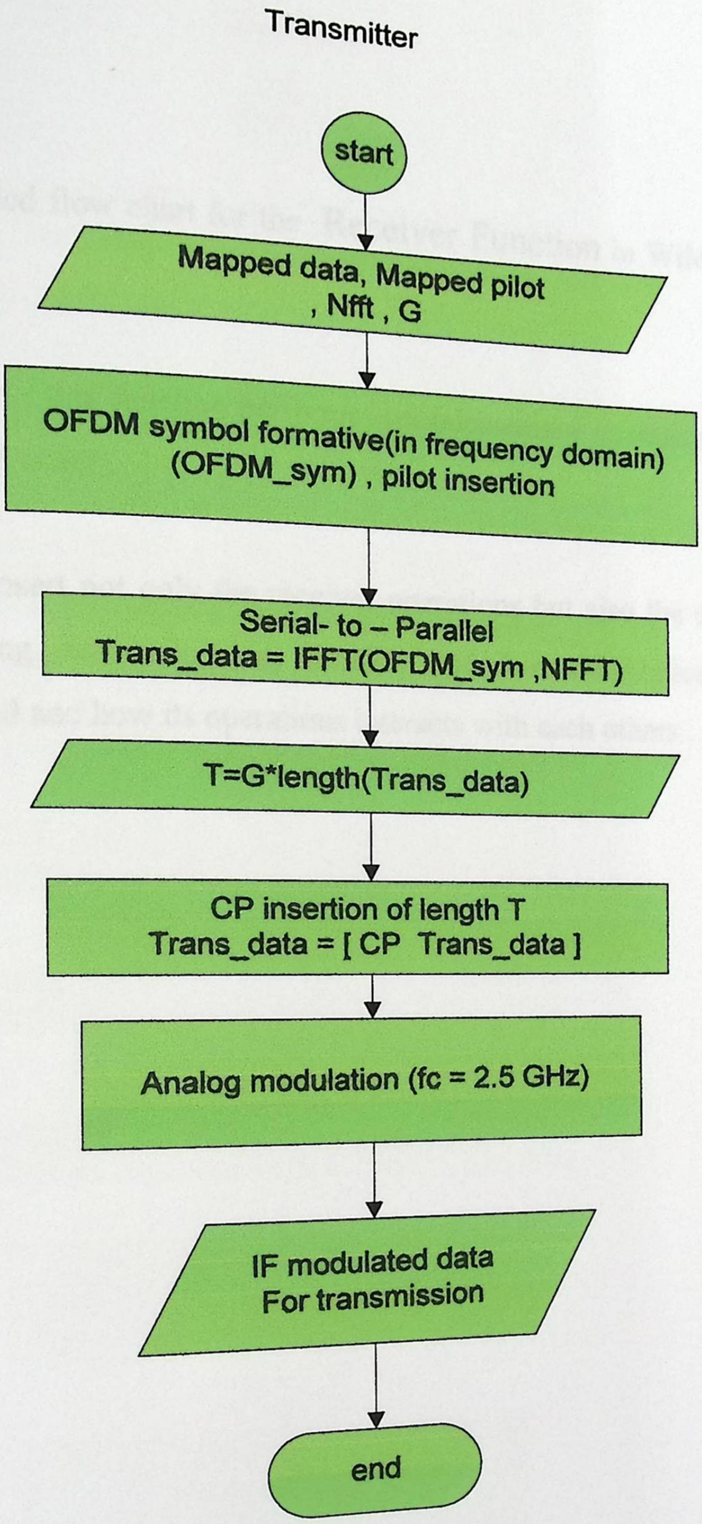


Figure (5.13) : Transmitter flow chart

ver flow chart

(5.14) shows the detailed flow chart for the Receiver Function in WiMAX

see from this flow chart , this function performs complementary operations to processes and in reverse order .

in this flow chart we insert not only the receiver operations but also the other h as symbol de-mapping, de-interleaving, decoding and de-randomization. In e complete receiving end and how its operations interacts with each others .

er of subcarriers

tio (time of cyclic
DM symbol time)

rrier frequency

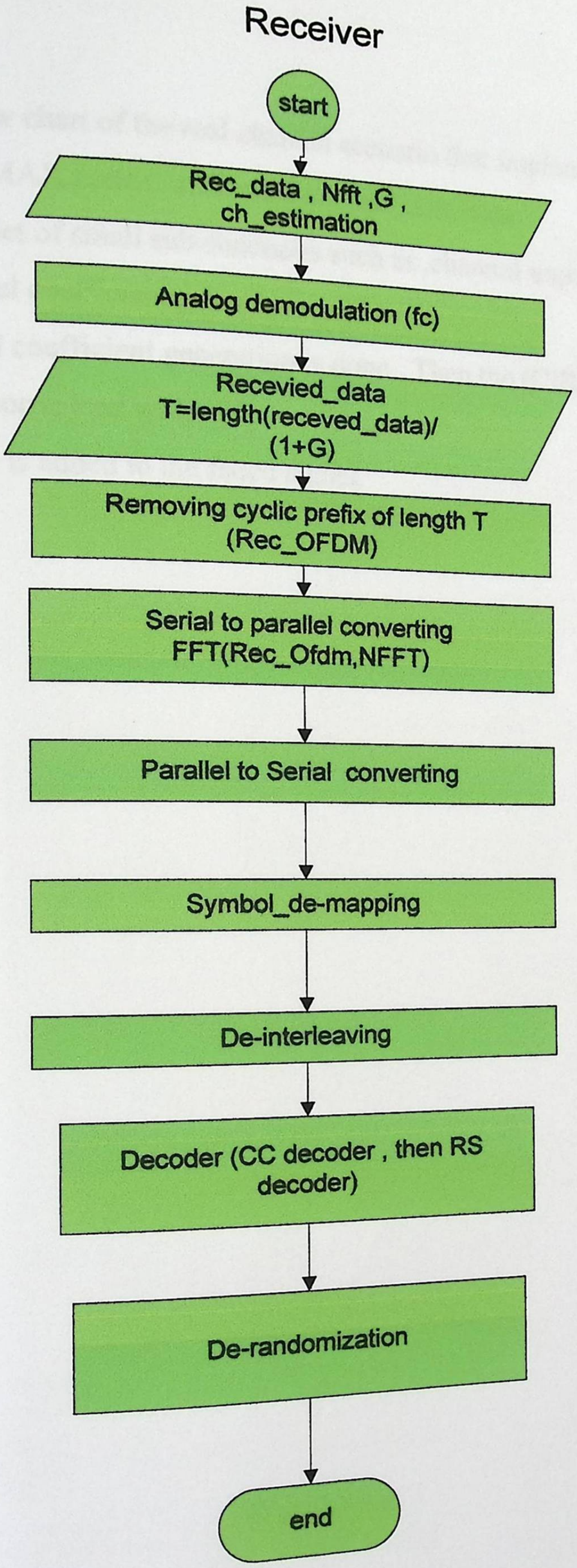


Figure (5.14) : Receiver flow chart

el flow chart

5.15) shows the flow chart of the real channel scenario that implement the SUI o as to test the WiMAX system under real channel scenarios .

ction consists of a set of small sub-functions such as ,channel impulse response n matrix and channel coefficient generation .

ee ,first the channel coefficient generation is done . Then the (CIR) is generated nsmitted signal is convolved with CIR .

the Gaussian noise is added to the faded signal.

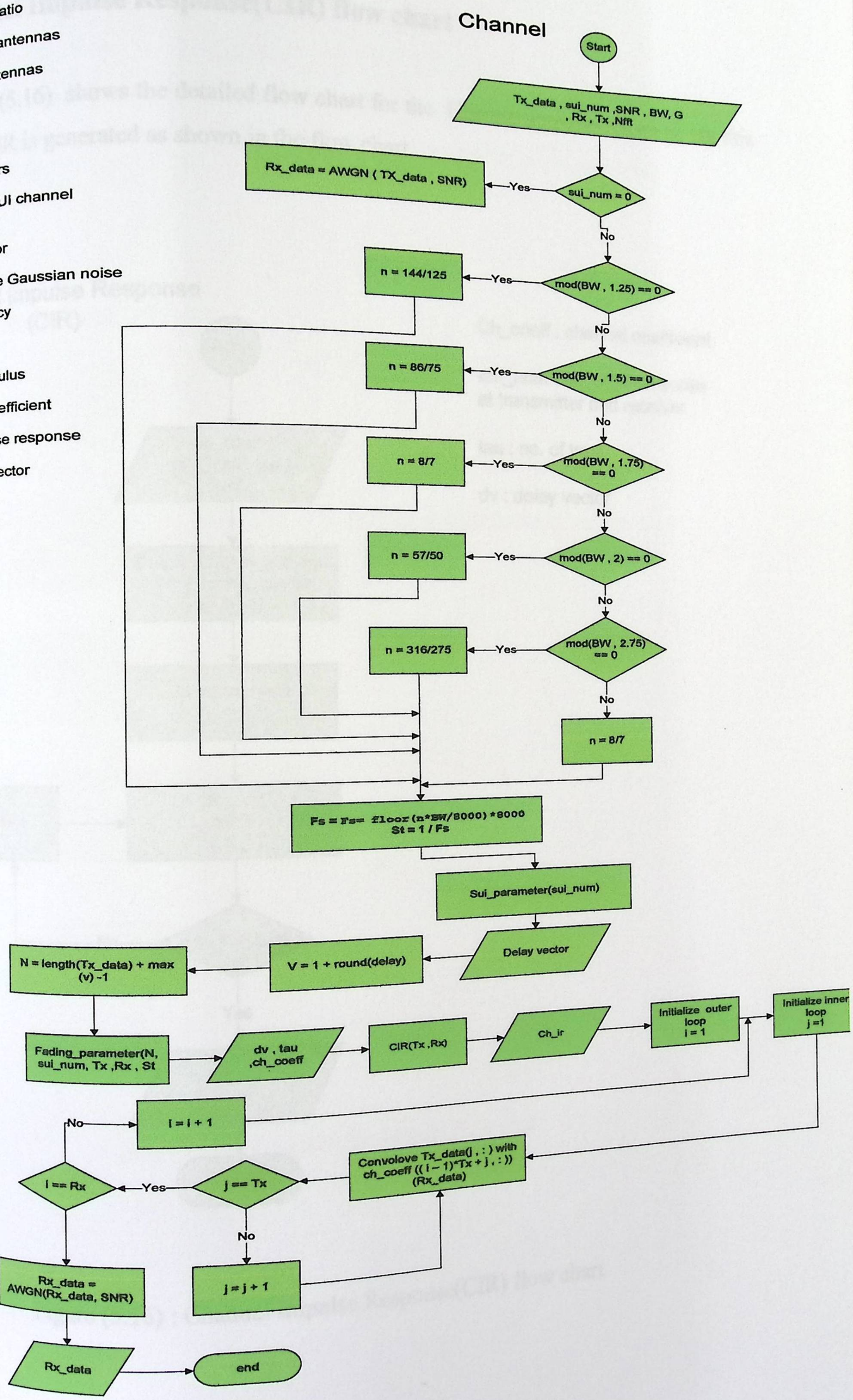
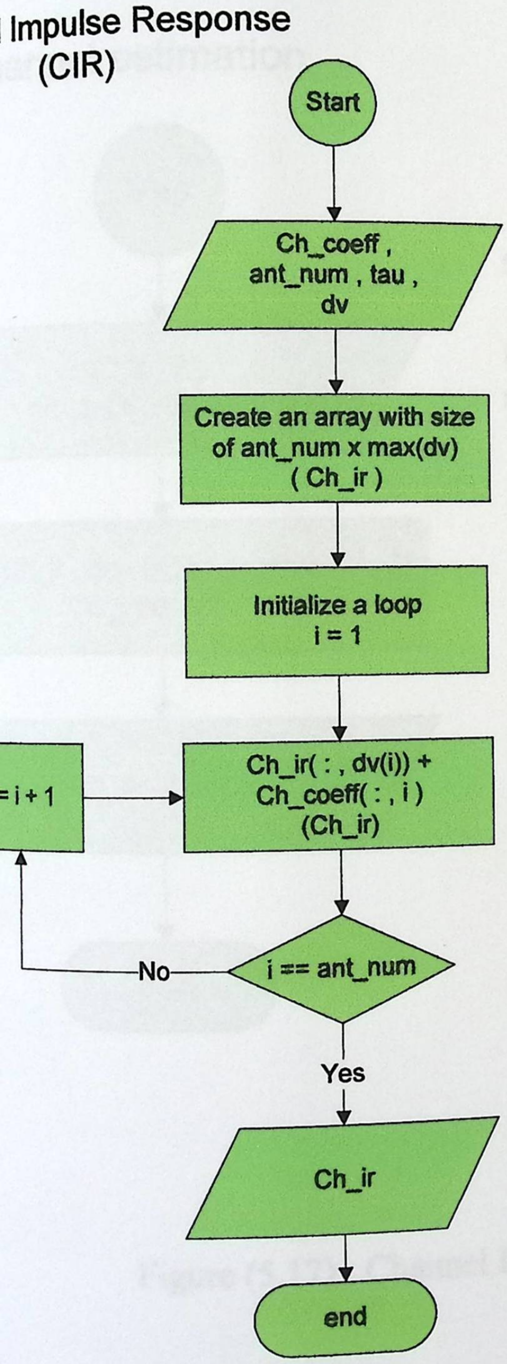


Figure (5.15) : Channel flow chart

el Impulse Response(CIR) flow chart

(5.16) shows the detailed flow chart for the Channel Impulse response .In this R is generated as shown in the flow chart.



Ch_coeff : channel coefficient

ant_num : no. of all antennas at transmitter and receiver

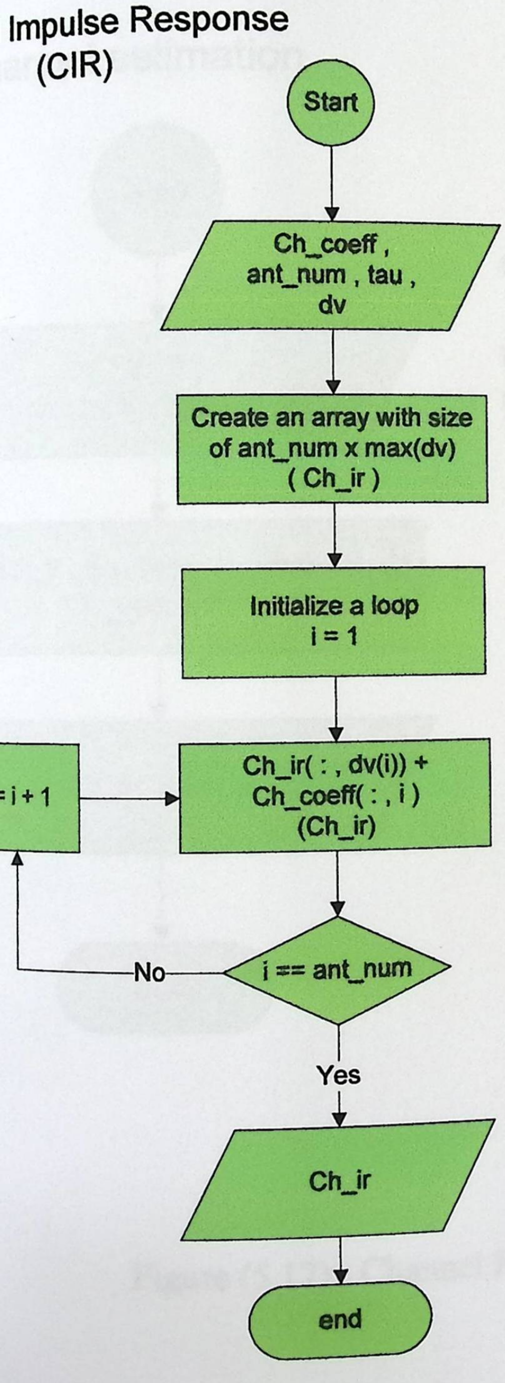
tau : no. of taps

dv : delay vector

Figure (5.16) : Channel Impulse Response(CIR) flow chart

el Impulse Response(CIR) flow chart

(5.16) shows the detailed flow chart for the Channel Impulse response .In this R is generated as shown in the flow chart.



Ch_coeff : channel coefficient

ant_num : no. of all antennas at transmitter and receiver

tau : no. of taps

dv : delay vector

Figure (5.16) : Channel Impulse Response(CIR) flow chart

el Estimation flow chart

(5.17) shows the detailed flow chart for the Channel Estimation .In this annel estimation coefficients are generated as shown in the flow chart.

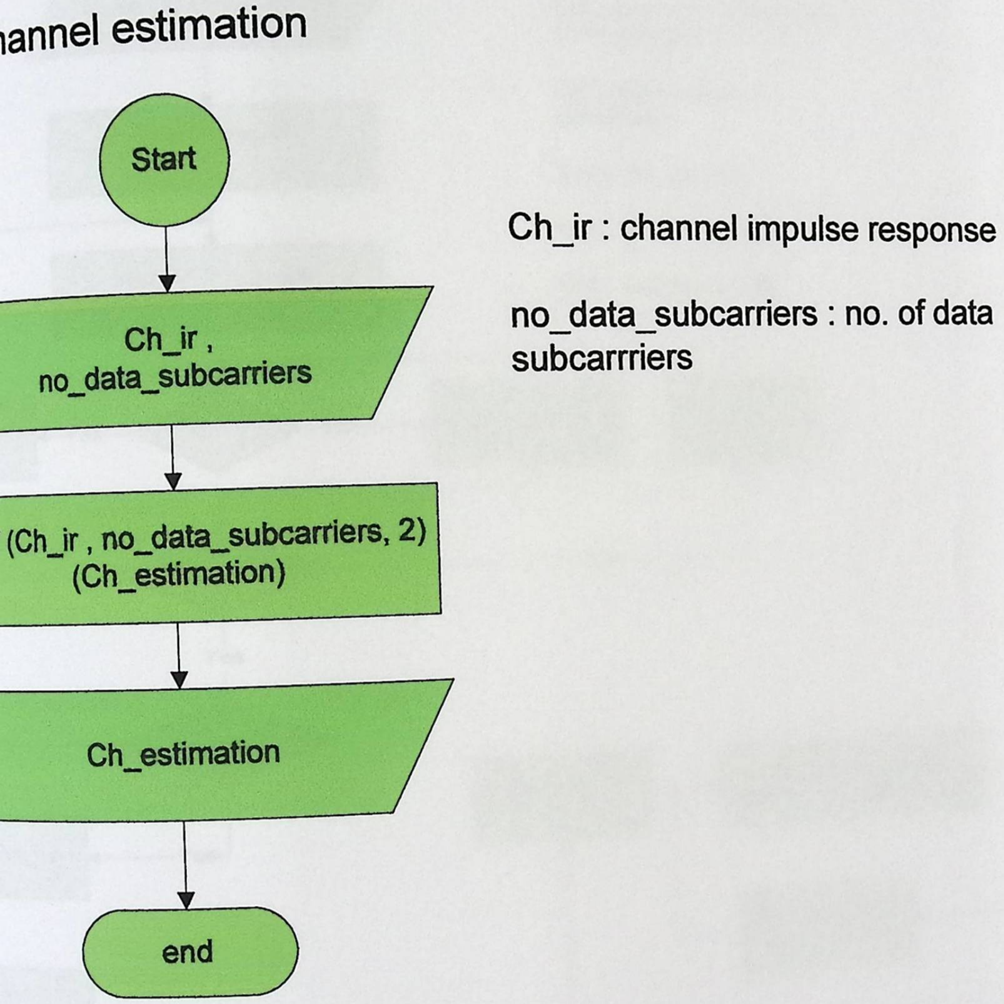


Figure (5.17) : Channel Estimation flow chart

on Matrix flow chart

8) shows the detailed flow chart for the correlation matrix .In this function trix is performed as shown in the flow chart.

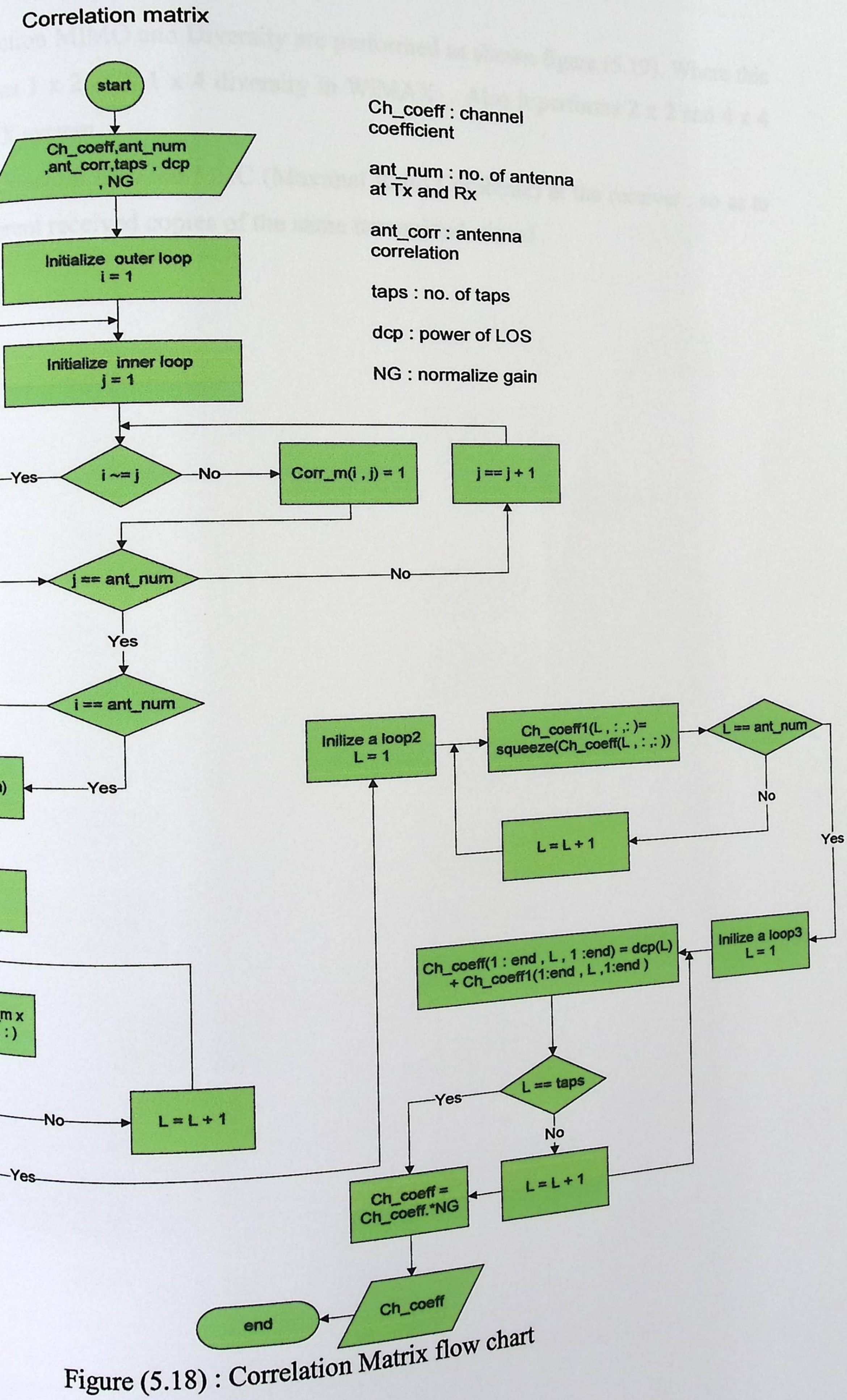


Figure (5.18) : Correlation Matrix flow chart

5.4.17 Correlation Matrix flow chart

Figure (5.18) shows the detailed flow chart for the correlation matrix .In this function the correlation matrix is performed as shown in the flow chart.

Correlation matrix

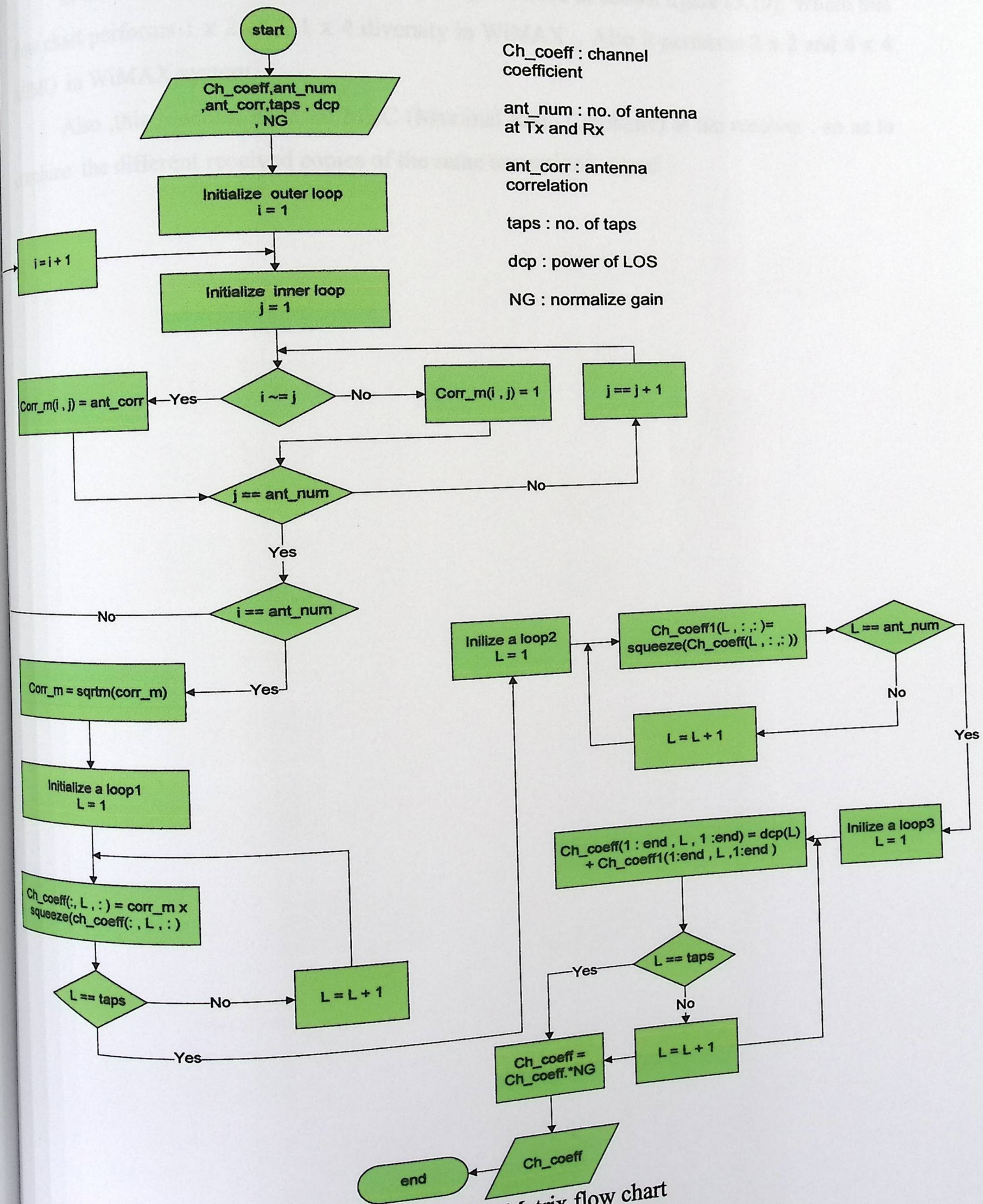


Figure (5.18) : Correlation Matrix flow chart

5.4.18 General MIMO and Diversity flow chart in WiMAX system

Figure (5.19) shows the detailed flow chart for the MIMO and Diversity in WiMAX system .

In this function MIMO and Diversity are performed as shown figure (5.19). Where this flow chart performs 1 x 2 and 1 x 4 diversity in WiMAX . Also it performs 2 x 2 and 4 x 4 MIMO in WiMAX system.

Also ,this function uses the MRC (Maximal Ratio Combiner) at the receiver , so as to combine the different received copies of the same transmitted signal .

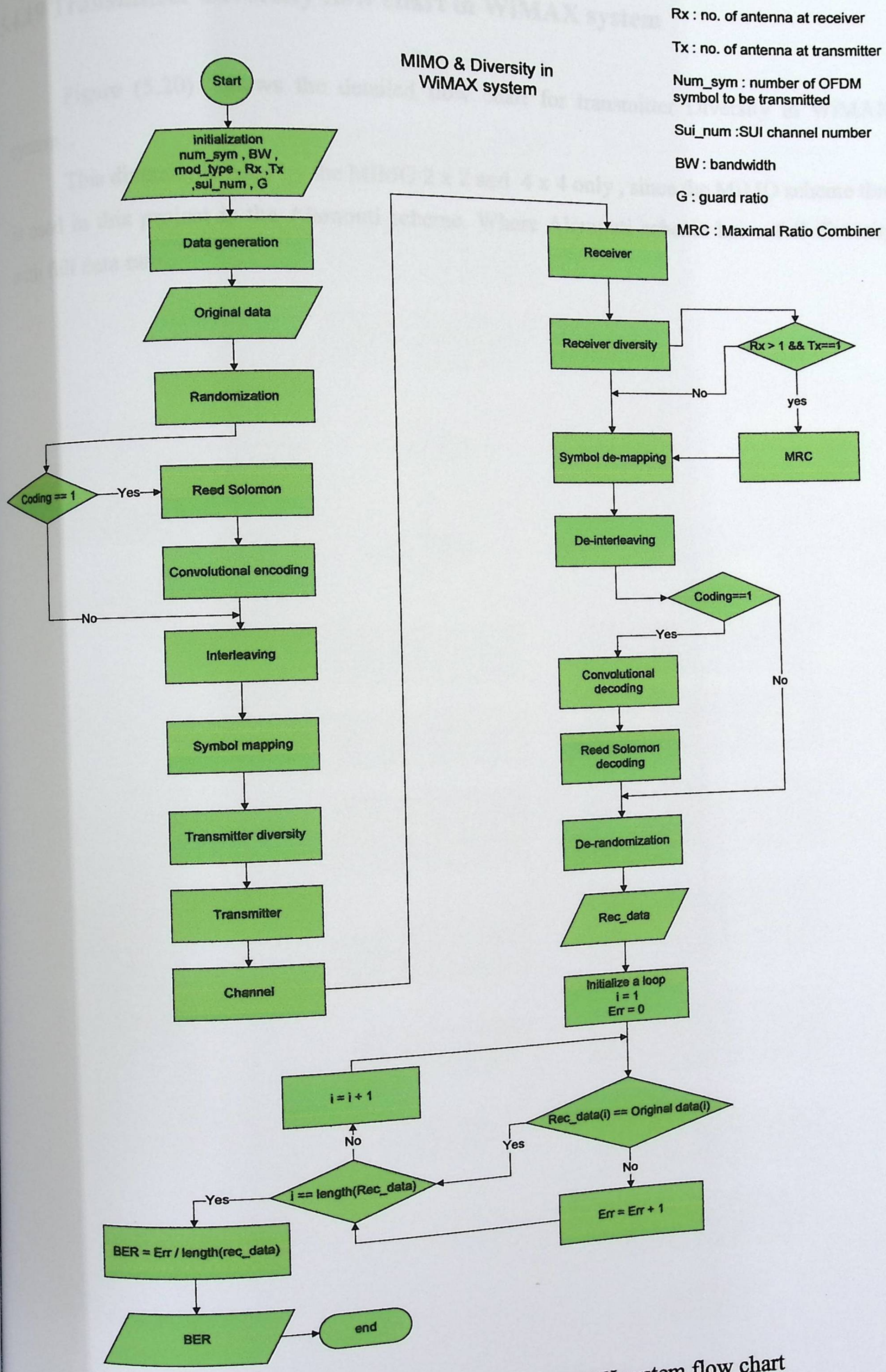


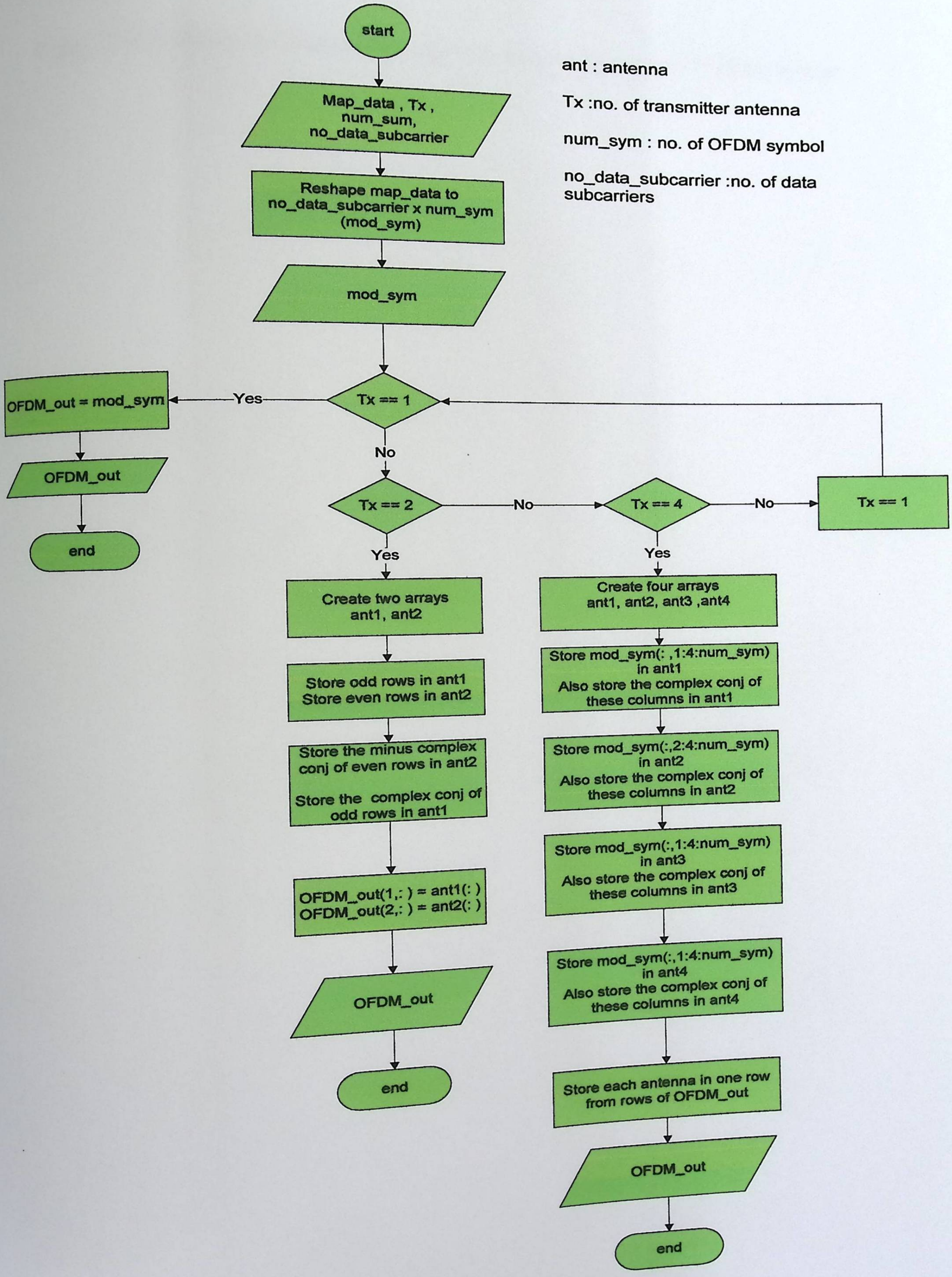
Figure (5.19) : MIMO and Diversity in WiMAX system flow chart

5.4.19 Transmitter Diversity flow chart in WiMAX system

Figure (5.20) shows the detailed flow chart for transmitter Diversity in WiMAX system .

This diversity is used by the MIMO 2 x 2 and 4 x 4 only , since the MIMO scheme that is used in this project is the Alamouti scheme. Where Alamouti scheme has a full diversity with full data rate .

Transmitter duversity



ant : antenna

Tx :no. of transmitter antenna

num_sym : no. of OFDM symbol

no_data_subcarrier :no. of data subcarriers

Figure (5.20) : Transmitter diversity

Transmitter duversity

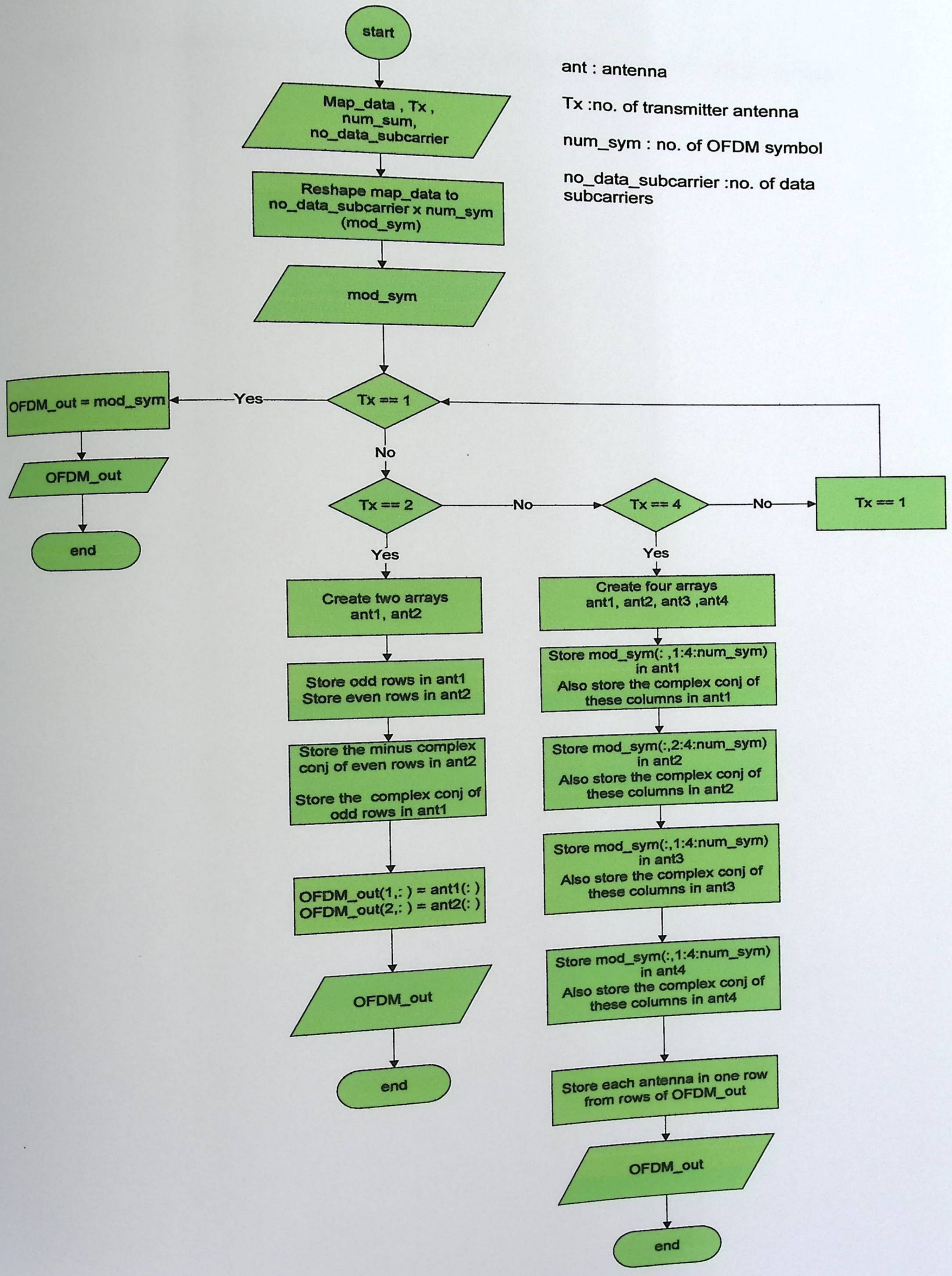


Figure (5.20) : Transmitter diversity

5.4.20 Receiver Diversity flow chart in WiMAX system

Figure (5.21) shows the detailed flow chart for receiver Diversity in WiMAX system .

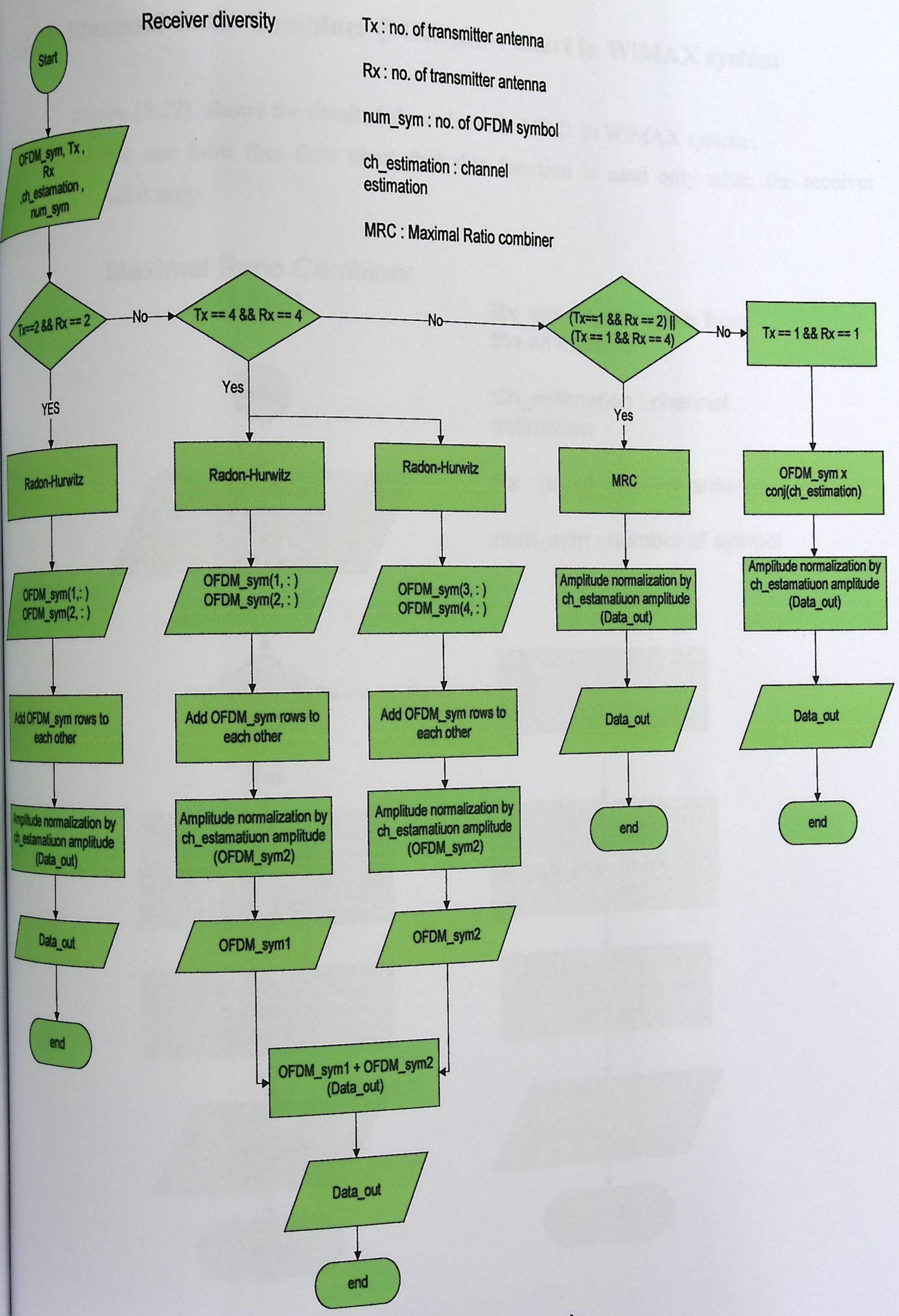


Figure (5.21) : Receiver diversity

5.4.21 Maximal Ratio Combiner (MRC)flow chart in WiMAX system

Figure (5.22) shows the detailed flow chart for MRC in WiMAX system .

As we see from this flow chart that this function is used only when the receiver diversity is used only .

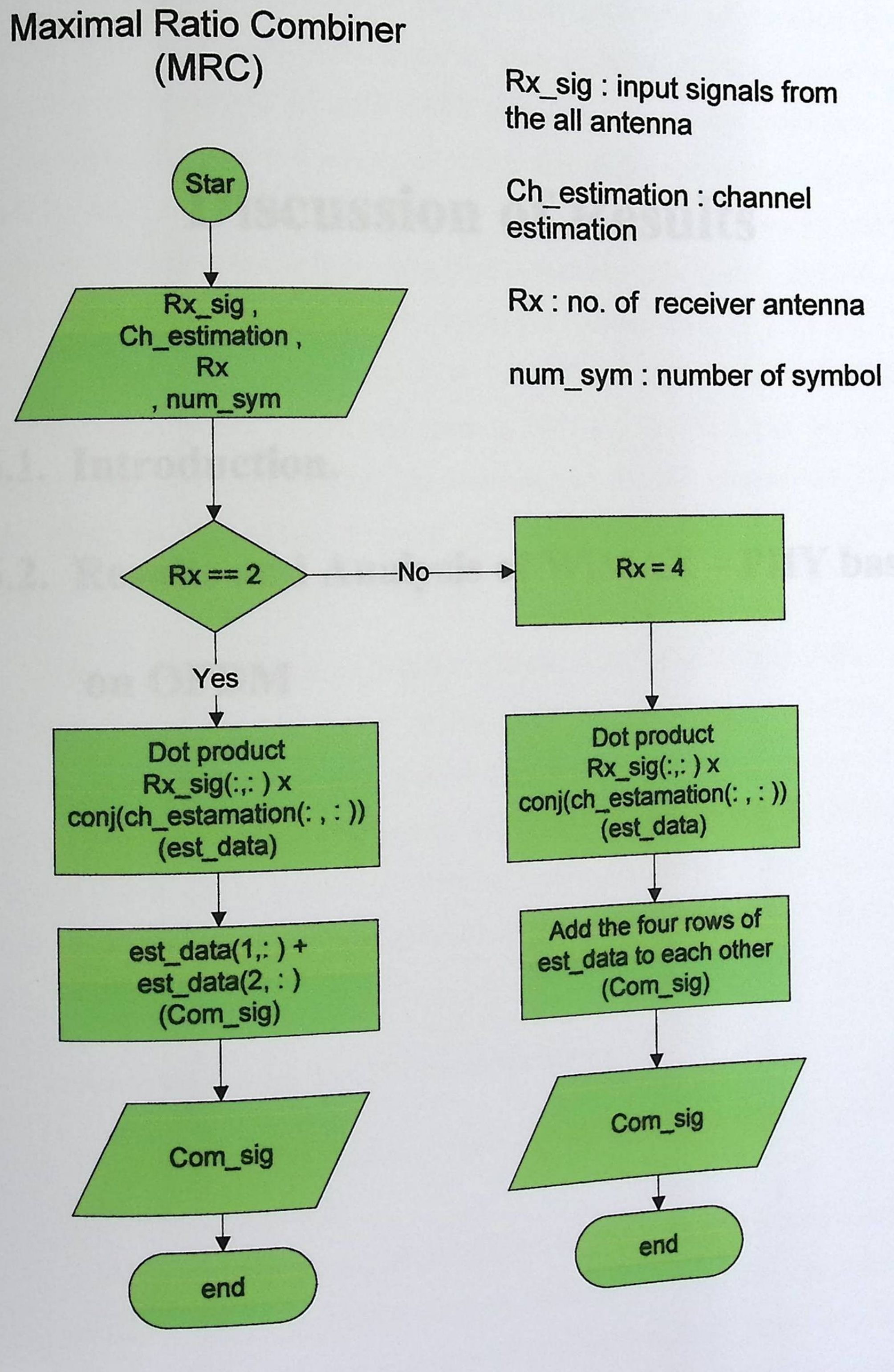


Figure (5.22) : Maximal Ratio Combiner diversity

Chapter 6

Discussion of Results

6.1. Introduction.

6.2. Results and Analysis of WiMAX – PHY based on OFDM

6.1 Introduction

In this chapter we introduce and discuss the results of the simulation that characterize the performance of the physical layer of WiMAX system based on OFDM .We examine the performance of WiMAX parameters at variable conditions ; with the aim to determine the best and worst conditions that define the correct configuration of WiMAX for each condition. In addition we improve the performance by adding some parameters to the standard ,such as ,using 512 subcarriers ,256 QAM modulation scheme with code rate of 3/4. We use the diversity at the receiver such as 1 x2 and 1 x 4, the MIMO system with different orders such as 2x2 and 4x4 to increase the data rate at high performance under worst conditions such as SUI (5) & SUI (6). In all of those simulations , we use realistic fading channel ; namely the SUI channel described in the standard IEEE 802.16 . They describe soft fading (SUI 1 & SUI 2) ,intermediate fading(SUI 3 & SUI 4) and harsh fading (SUI 5 & SUI 6) .

We built our simulation in Matlab version 7.6.0.324 - Release 2008 , using the matlab code ,where the simulation model is divided into a main function and sub functions . The main function calls the sub functions according to the entered inputs . The outputs will be a curves that specify the relationships between the BER(Bit Error Rate) and the SNR (Signal –to-Noise Ratio) according to the input specified conditions. Each sub function represents a certain block or sub block in the main block diagram of WiMAX .

The functions that are 'build in' in the Matlab are :

1. The generation of random data : this function generates a random data to represent the real data.
2. Reed Solomon encoding & decoding : but these functions need a suitable data form for applying coding process, such as ,dividing the data into a blocks ,then converting these blocks into decimal numbers and then converting these decimal numbers into " Galois structure" . The Reed Solomon function is then applied to the data.

3. Convolutional encoding and decoding : this function also needs some work for applying them ; the data first must be converted into a trellis structure , then the convolutional encoding is applied to the data , after that the encoded data must be punctured to allow for different code rates.

The results are plotted at (100 and 500) repeating times for the same point at certain SNR with different random data each time. Choosing these repetition levels result in BER point with reliable very small percentage error. Table 6.1 summarizes the percentage errors obtained for different repetitions at certain SNR relate to 1000 times repetitions . Where 100 times is used when SUI 1 and SUI 2 are used in the simulation as a channel model for the fading effects to test the WiMAX system under them ; while the 500 times is used for the SUI 3 to SUI 6 channel model.

Table 6.1

The percentage error for different repetitions with respect to 1000 times repetition

% error with respect to 1000 times at SNR = 12 dB for 16 – QAM (3/4) ,G = 1/4 ,BW = 5 MHz SUI = 1 Where the BER at SNR = 12 dB at 1000 times repetition is 0.0893						
Repetition	10 times	30 times	60 times	100 times	200 times	500 times
BER	0.15	0.142	0.123	0.0904	0.0901	0.0900
%error = $\frac{\lvert BER-0.0893\rvert}{0.0893} * 100\%$	68 %	59%	37.7%	1%	0.89%	0.79 %
SUI = 2 BER at SNR = 16 dB at 1000 times is 0.102 , *The other conditions remain same						
BER	0.1723	0.1531	0.1225	0.1034	0.1008	0.1012
%error	68.9%	50.1%	20%	1.3%	1.17%	0.784%
SUI = 3 BER at SNR = 23 dB at 1000 times is 0.03042 *The other conditions remain same						
BER	0.01801	0.0442	0.04106	0.03640	0.03350	0.0308
%error	40.7%	45.3%	35%	19.6%	10.1%	1.24%

We conclude that repetitions of 100 times for SUI-1 and SUI-2 model simulation and 500 times for SUI-3 and SUI-6 resulted in percentage error less than 1.5% which is demand satisfactory for our simulation purpose .

6.2 The results and analysis :

6.2.1. The effects of SUI channels .

In this part we will check the effect of the different SUI channel models that represent the all different environmental effects , from simple effects (low delays, low scattering(flat),there is a line–of– sight and low tap power) to the largest effects (large scattering ,larger delays ,high Doppler effect, low antenna correlation , high tap power , no line of sight) .

Figures 1 - 5 show BER plots Vs SNR for different modulation schemes under different SUI channel models .We note from figures that BER at certain SNR takes its lowest values under (SUI 1 & SUI 2) while BER take its large values under (SUI 5 and SUI 6) , but it will take a values are not considered a low or high but between of them under(SUI 3 and 4) ,where in SUI 3 ,the BER values are approximately considered a mean values of above ,but in SUI 4 ,the values not high but near of it .

The reason that makes the SUI 5 and 6 are the most worst environmental effect , is that they have a largest delays with largest tap power , these circumstances make the SNR to take a high values to suppress the effect of the high tap power and high delays, that cause a high phase variations at the receiver ; that may make the different paths of the signal to be added up out of phase at the receiver. So this will raise the required SNR to achieve a certain BER.

In this part , we take the SUI 1 , SUI 3 and SUI 5 as a channel models for testing all mandatory and option modulation schemes . Choosing these channel models , since SUI 3 approximately represents SUI 4 . SUI 1 also nearly represent SUI 2 since its behavior is near to the behavior of SUI 2 , and SUI 5 nearly represents the behavior of SUI 6 .

Also from figures we note as the SNR increases ,the BER decreases , and the reason for this that when the SNR increases ,this means that the power of the signal becomes more greater than the power of noise and the tap power ,so at the receiver ,the detection of the signal will be more efficient , and it will be easily reconstructed with better accuracy . Also for higher SNR values , the BER becomes smaller than the theoretical BER.

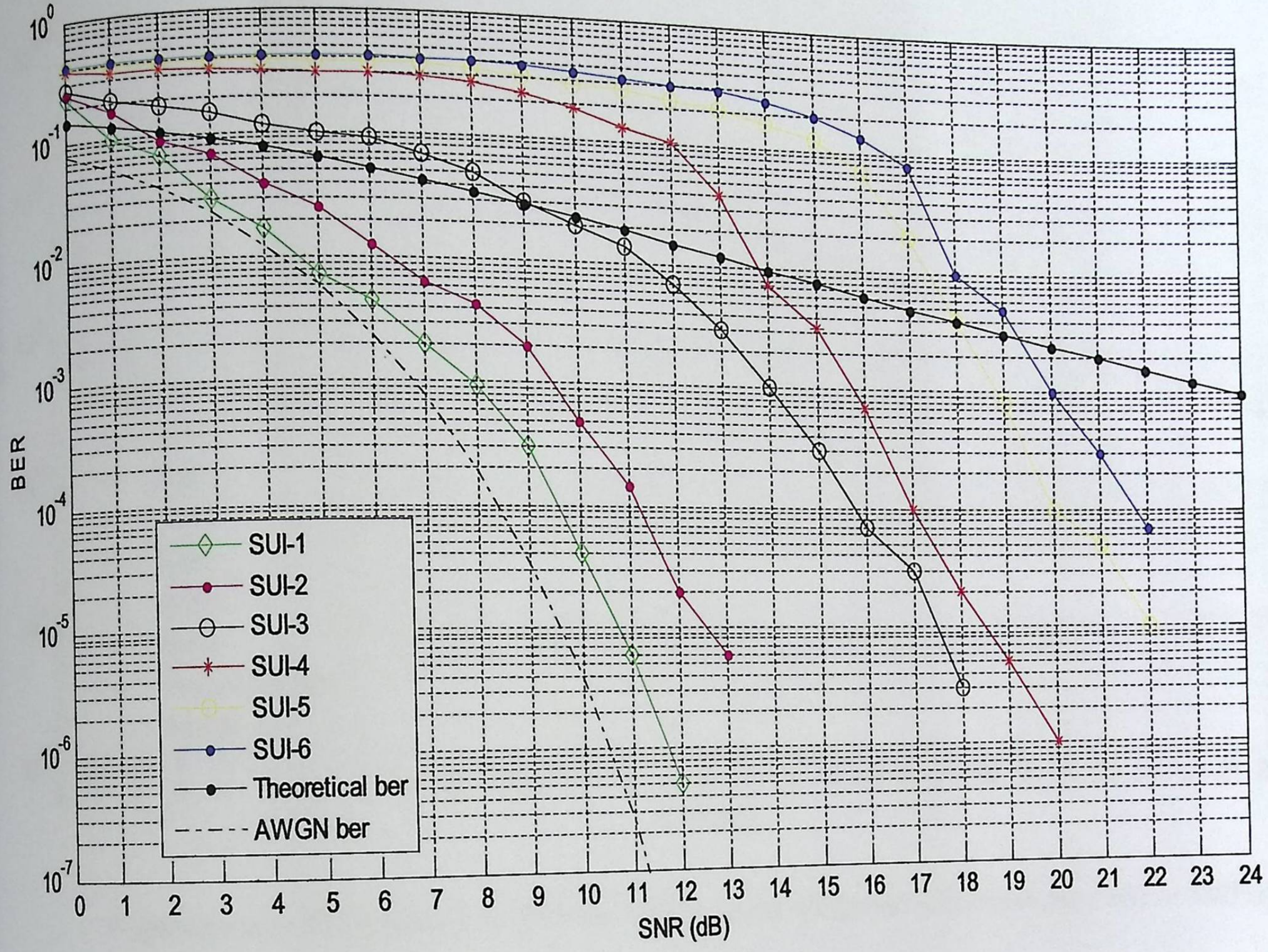


Figure (6.1) : BER plots Vs SNR for BPSK under different SUI channels , BW = 5MHz, G=1/4

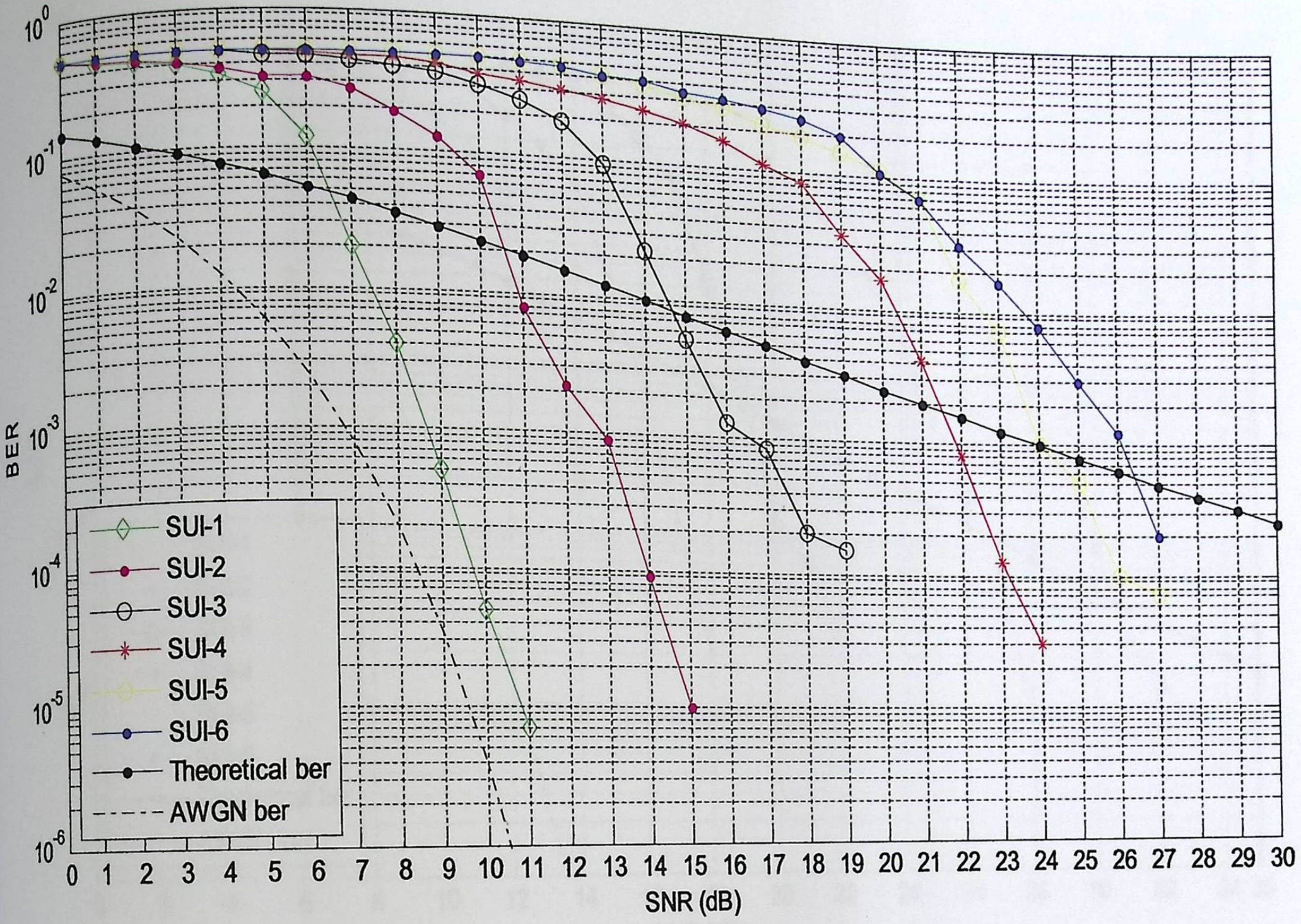


Figure (6.2) : BER plots Vs SNR for QPSK under different SUI channels , BW = 5MHz, G=1/4

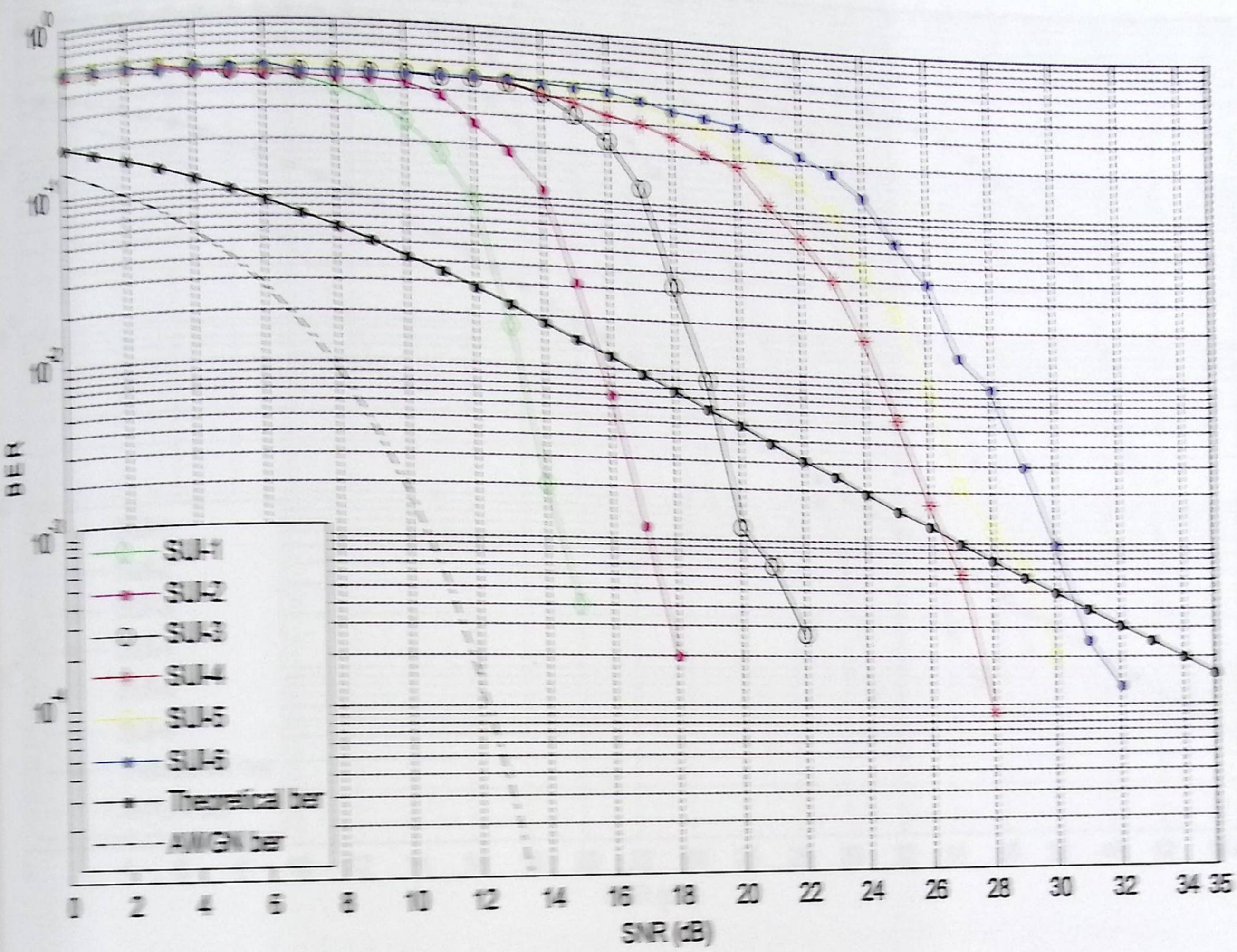


Figure (6.3) : BER plots Vs SNR for 16-QAM (3/4) under different SUI channels ,

BW = 5MHz, G=1/4

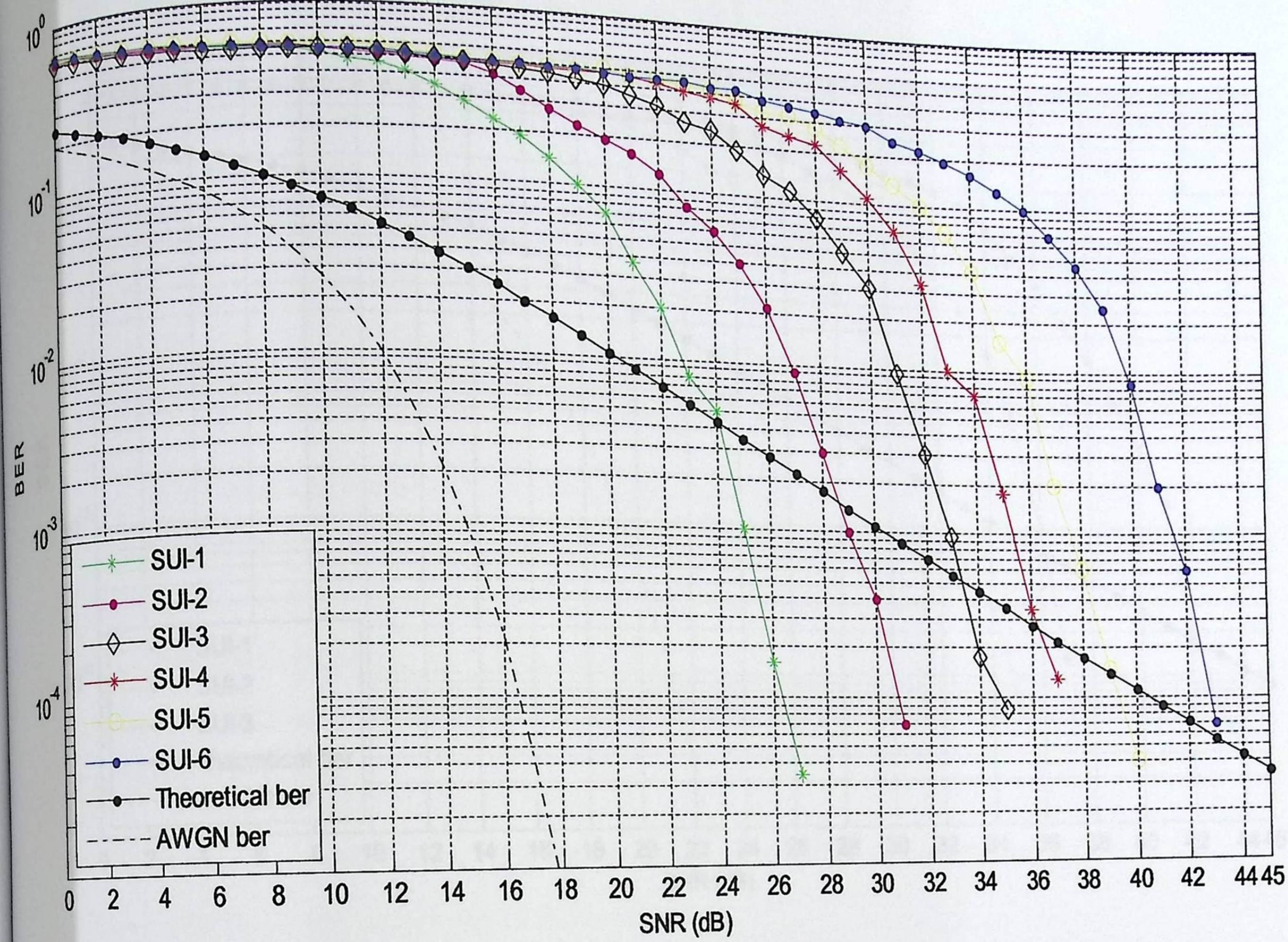


Figure (6.4) : BER plots Vs SNR for 64-QAM (3/4) under different SUI channels ,

BW = 5MHz, G=1/4

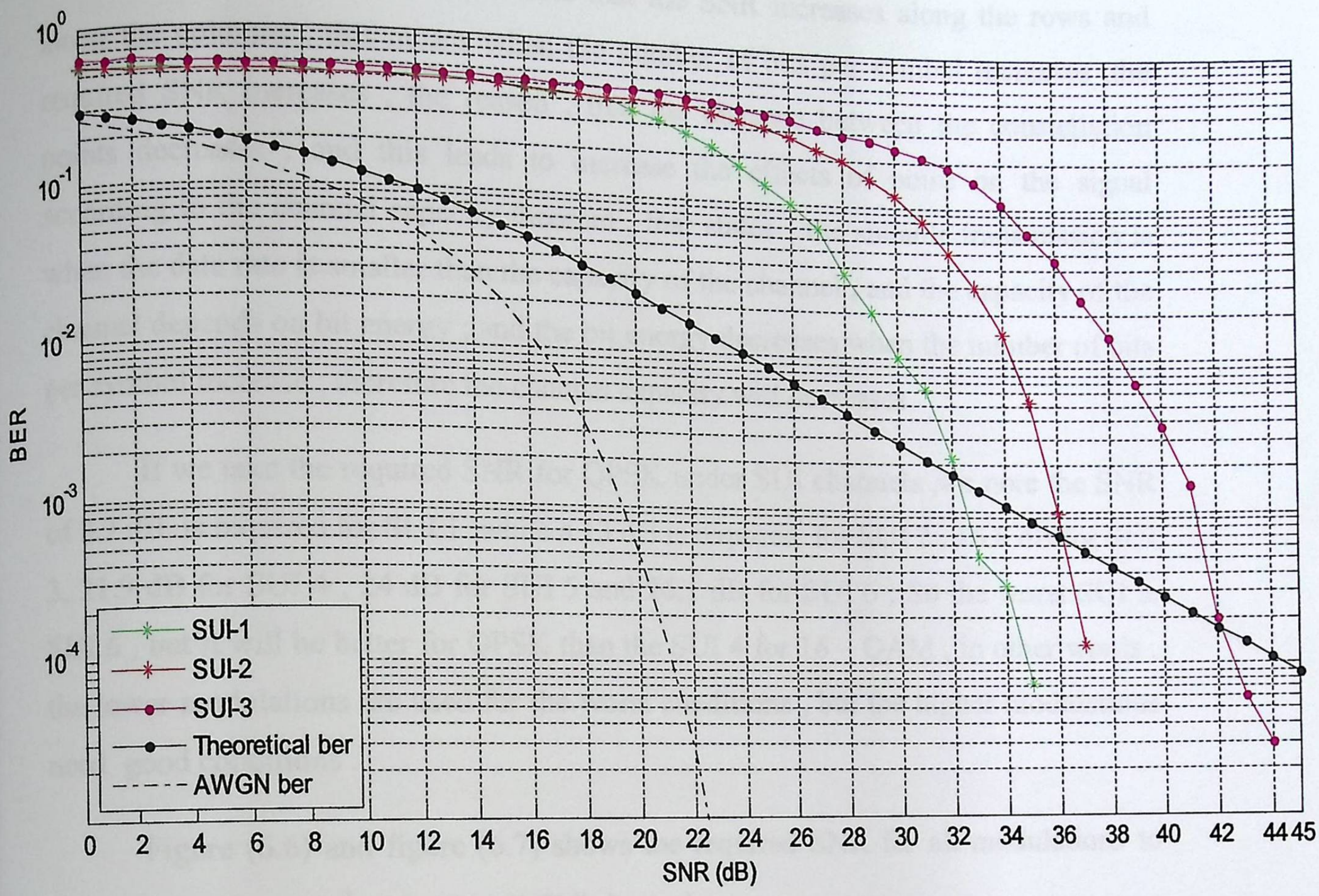


Figure (6.5) : BER plots Vs SNR for 256-QAM (3/4) under different SUI channels , BW = 5MHz, G=1/4

From above figures we can summarize the required SNR to achieve a certain BER in aim to show the effect of the different SUI channel models on different modulation schemes.

The following table shows the required SNR to achieve BER of 10^{-3} for the modulation schemes that are used in WiMAX under the SUI channels.

Table 6.2

The required SNR at 10^{-3} for different modulations under SUI channels

	SUI 1	SUI 2	SUI 3	SUI 4	SUI 5	SUI 6
BPSK (3/4)	8 dB	9.5 dB	14 dB	15.9 dB	18.9 dB	20 dB
QPSK (3/4)	8.7 dB	13 dB	16.8 dB	21.9 dB	24 dB	26.1 dB
16-QAM (3/4)	14.5 dB	17.2 dB	20.3 dB	26.5 dB	28.3 dB	30 dB
64-QAM (3/4)	25 dB	29 dB	32.9 dB	35.4 dB	37.7 dB	41.7 dB
256-QAM (3/4)	32.5 dB	36 dB	41.1 dB			

From the above table , we note that the SNR increases along the rows and along the columns , this means when the number of bits per symbol increases ,the required SNR increases , the reason , that the distance between the constellation points decreases , and this leads to increase the effects of noise on the signal according to the channel capacity theorem , that states the suitable transmission is when the data rate is smaller than the capacity of the channel , and the capacity of the channel depends on bit energy ; and the bit energy decreases when the number of bits per symbol increase ; therefore the channel capacity will decreases.

If we take the required SNR for QPSK under SUI channels ,we note the SNR of 8.7 dB is required for SUI 1 ,while a 13 dB is required for SUI 2 , 16.8 dB for SUI 3, 21.9 dB for SUI 4 , 24 dB for SUI 5 and 26.1 dB for SUI 6 ; So the worst SUI is SUI 6 , but it will be better for QPSK than the SUI 4 for 16 – QAM , in other words , the lower modulations are used for the worst conditions , but the higher modulations need good conditions .

Figure (6.6) and figure (6.7) shows the required SNR for all modulations to achieve a BER of 10^{-3} under the all SUI channels .

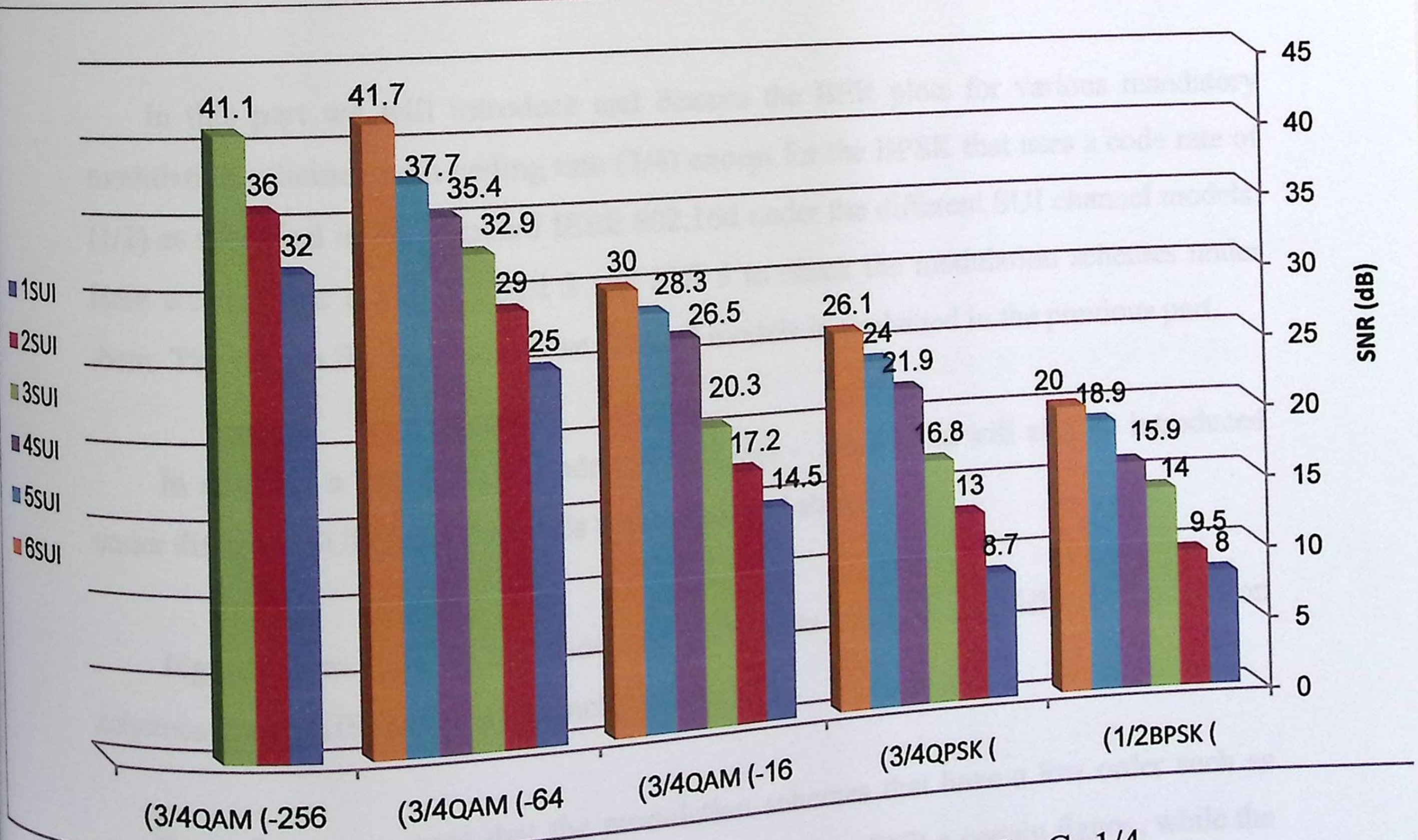


Figure (6.6) :the required SNR at BER of 10^{-3} ,BW=5MHz, G=1/4 high SNR values for high performance . for example , the BPSK needs a 9.5 dB to give a BER of 10^{-4} under SUI 1, QPSK needs a 12.8 dB at BER of 10^{-4} , while the 16-QAM needs 18.2 dB at BER of 10^{-4} , and 64-QAM needs a 24.7 dB at that BER ,while 256-QAM needs a 34 dB for BER of 10^{-4} . Also at a certain SNR the BER increases when the order of modulation increases .

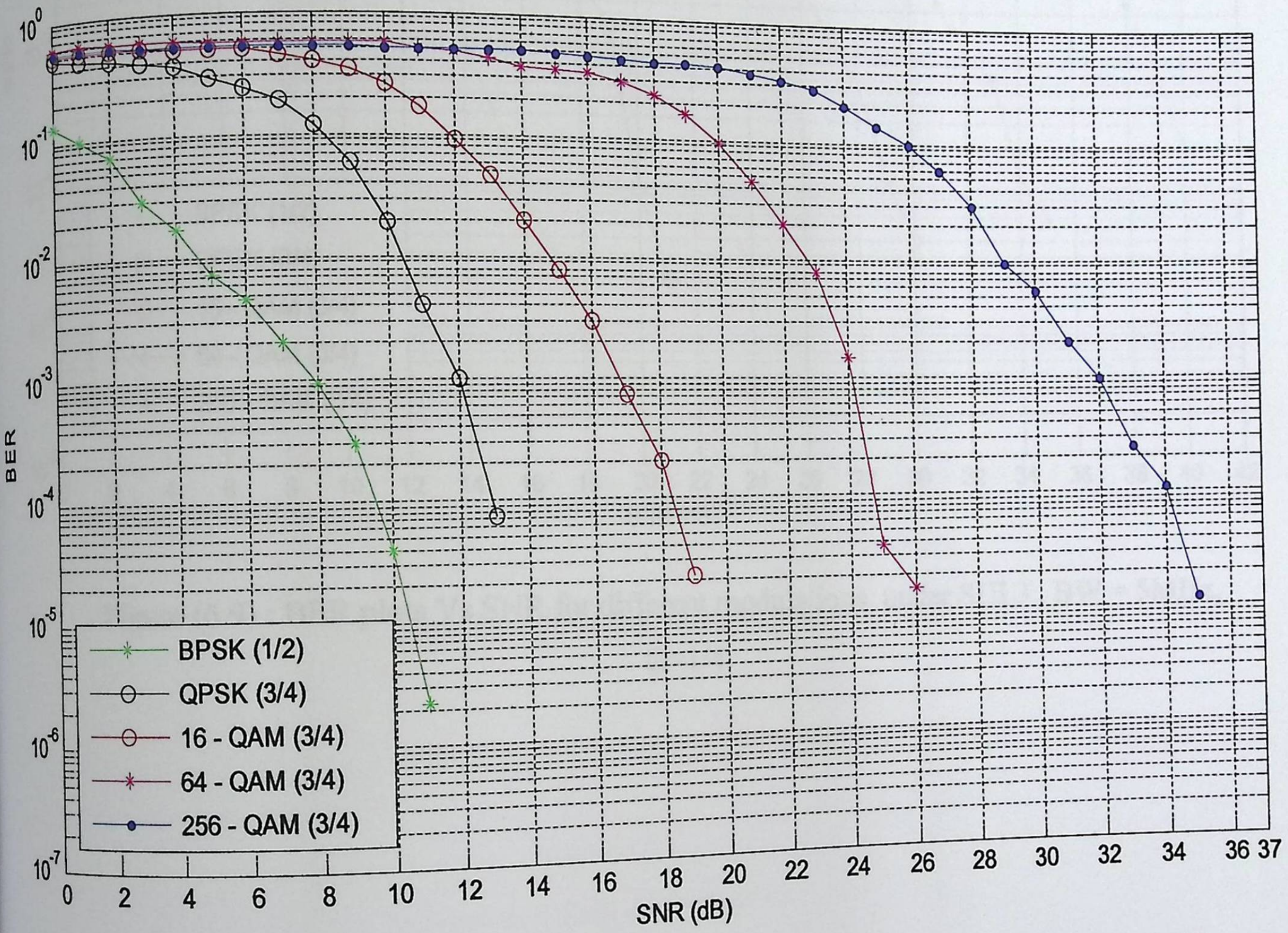


Figure (6.8) : BER plots Vs SNR for different modulations under SUI 1 , BW = 5MHz, G=1/4

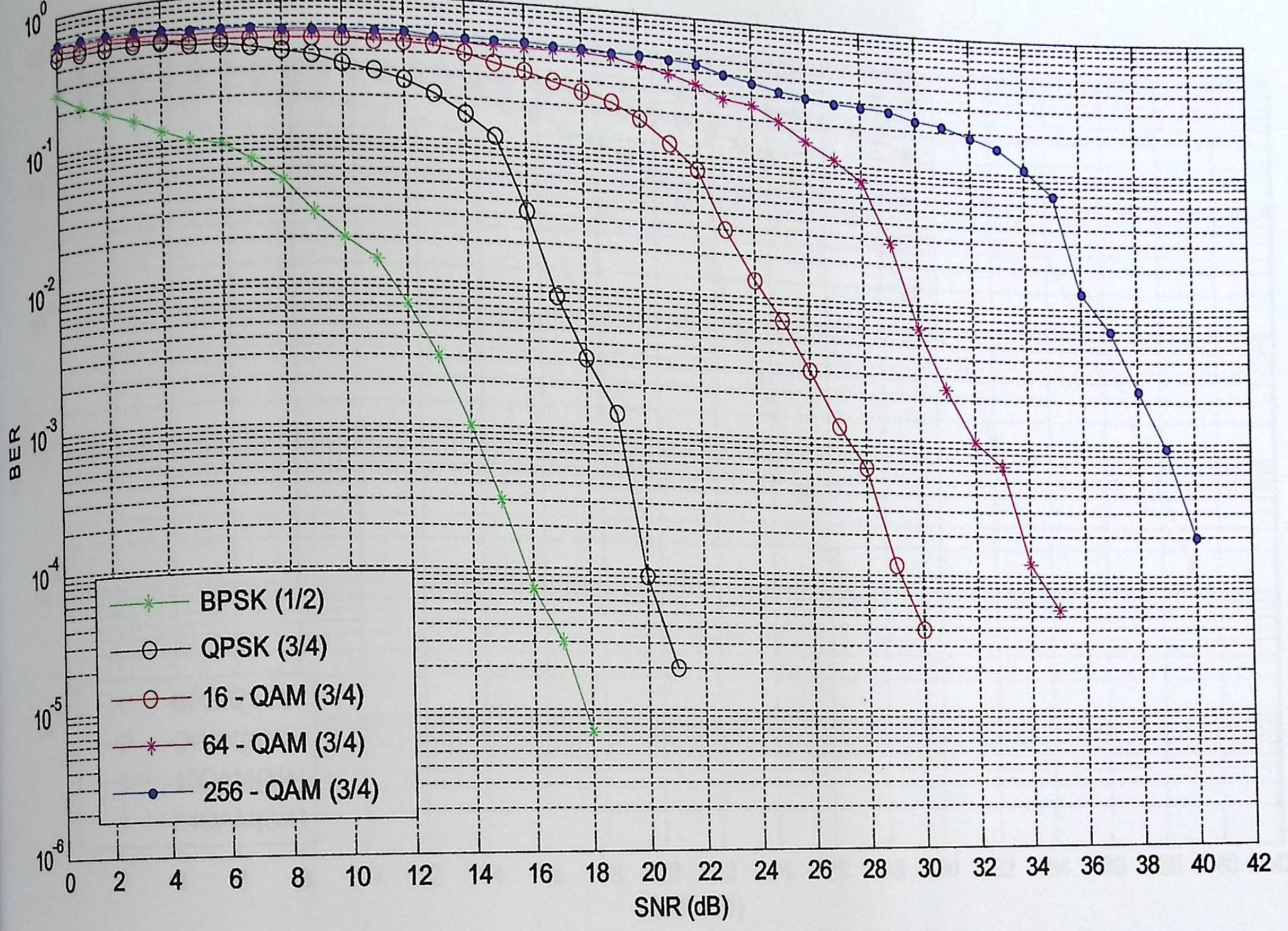


Figure (6.9) : BER plots Vs SNR for different modulations under SUI 3 , BW = 5MHz, G=1/4

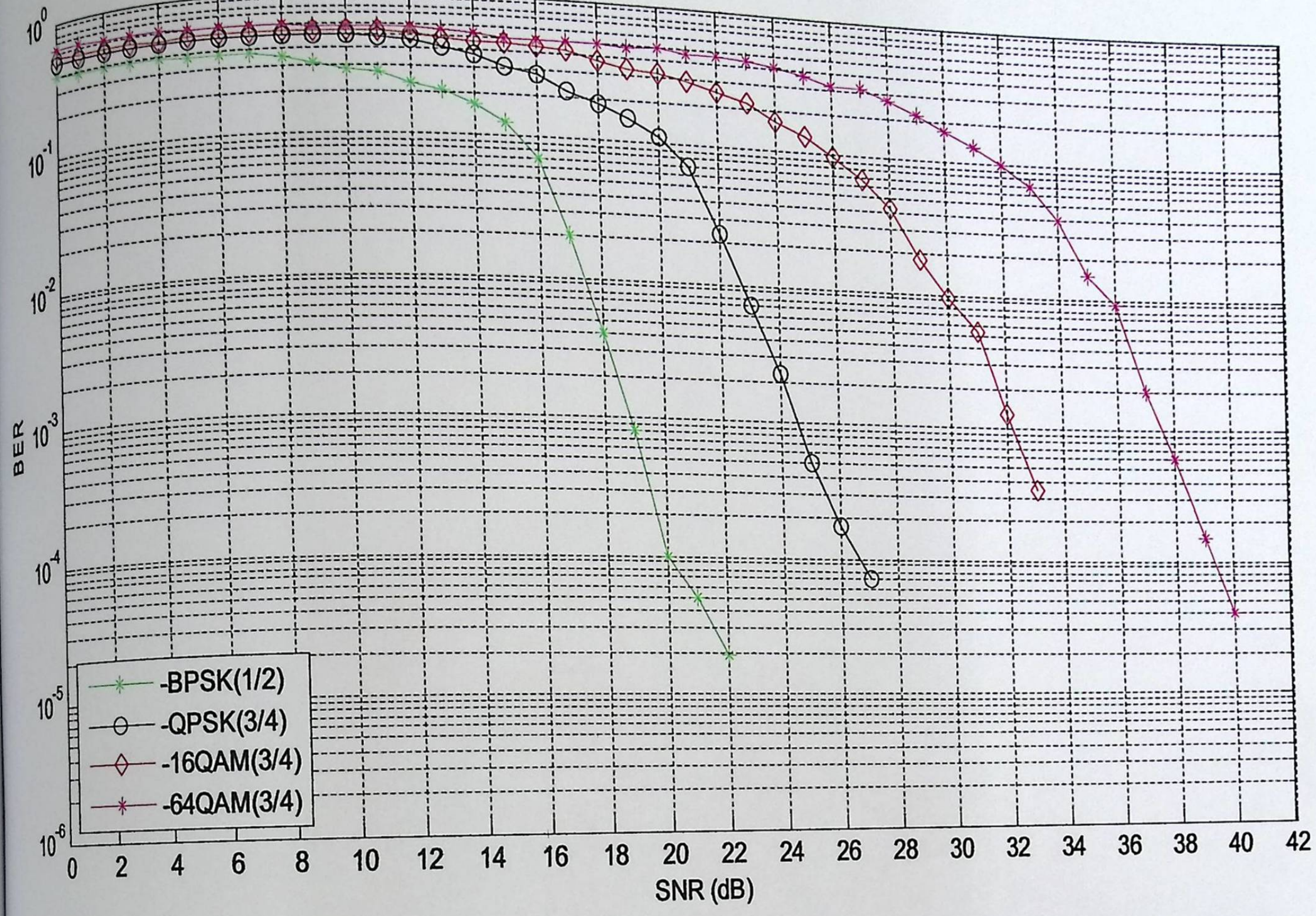


Figure (6.10) : BER plots Vs SNR for different modulations under SUI 5 , BW = 5MHz, G=1/4

Table 6.3

The required SNR at BER of 10^{-3}

	BPSK (1/2)	QPSK (3/4)	16-QAM (3/4)	64-QAM (3/4)	256-QAM (3/4)
SUI 1	8 dB	12 dB	16.8 dB	24 dB	32 dB
SUI 3	14 dB	19.1 dB	27.2 dB	32 dB	39 dB
SUI 5	18.8 dB	32.2 dB	37.5 dB	37.8 dB	

From the above table we note the SNR increases when the number of bits per symbol increases to achieve the same BER .For example , BPSK needs an 14 dB under SUI 3 at BER of 10^{-3} ,while the QPSK needs a 19.1 dB under same channel model at BER of 10^{-3} ,the 16 – QAM is more and more to reach 27.2 dB , 64 – QAM reaches the 32 dB , 256-QAM is the most that needs a 39 dB to achieve BER of 10^{-3} .

The following figures combines the all data in the above table in one graph.

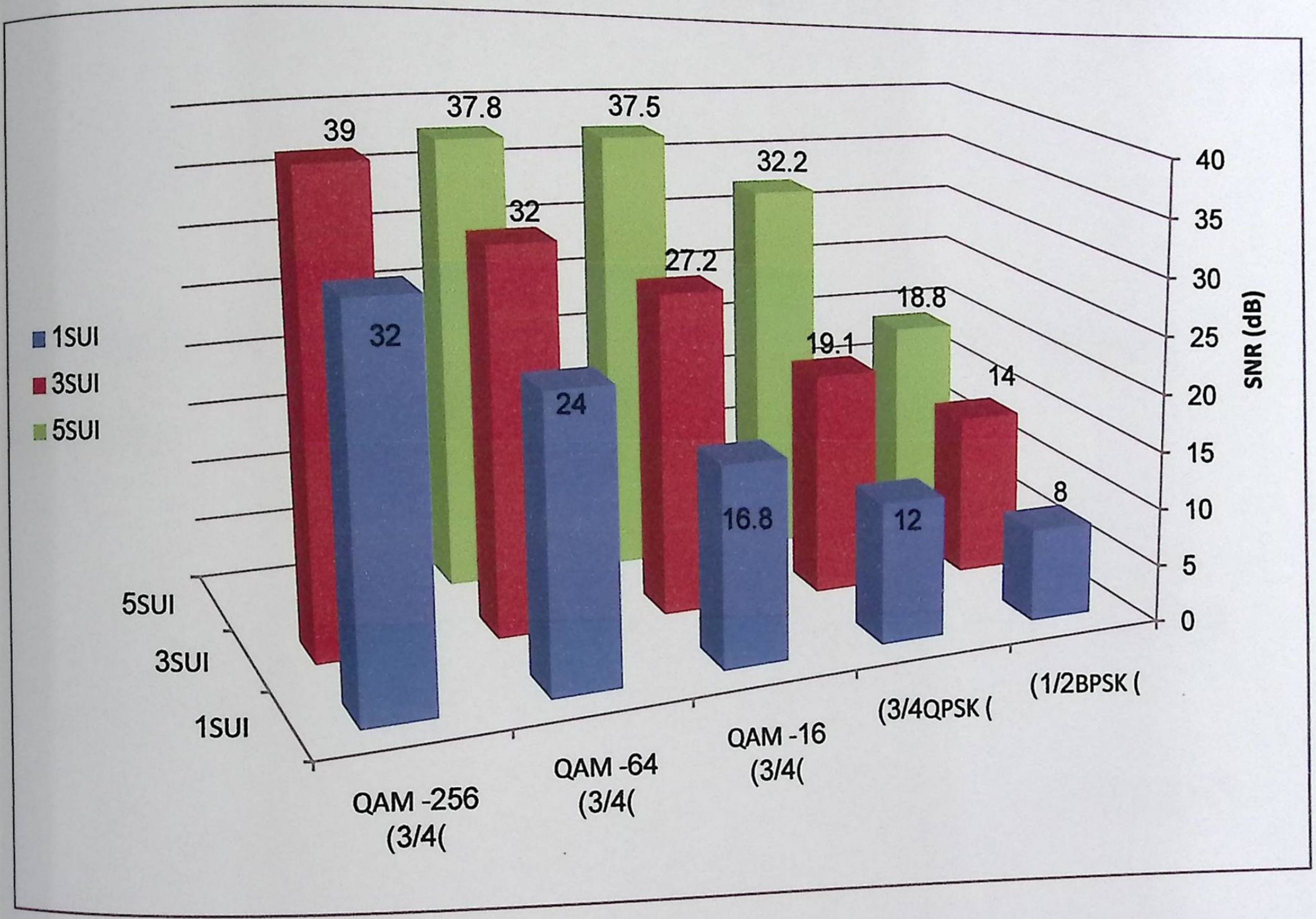


Figure (6.11) : Required SNR at BER of 10^{-3} , BW = 5MHz ,G=1/4

From the above table we note the SNR increases when the number of bits per symbol increases to achieve the same BER .For example , BPSK needs an 14 dB under SUI 3 at BER of 10^{-3} ,while the QPSK needs a 19.1 dB under same channel model at BER of 10^{-3} ,the 16 – QAM is more and more to reach 27.2 dB , 64 – QAM reachs the 32 dB , 256-QAM is the most that needs a 39 dB to achieve BER of 10^{-3} .

The following figures combines the all data in the above table in one graph.

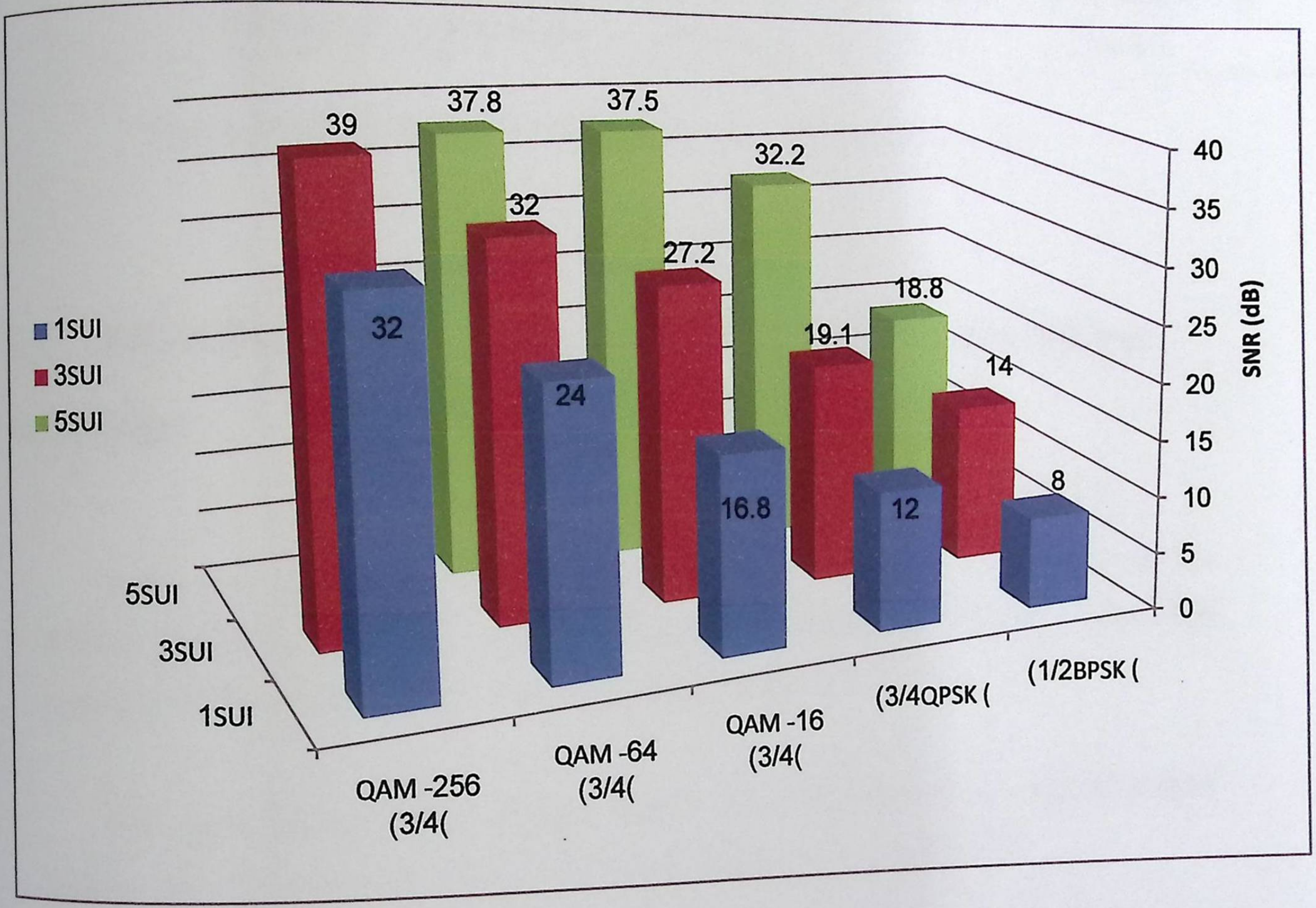


Figure (6.11) : Required SNR at BER of 10^{-3} , BW = 5MHz ,G=1/4

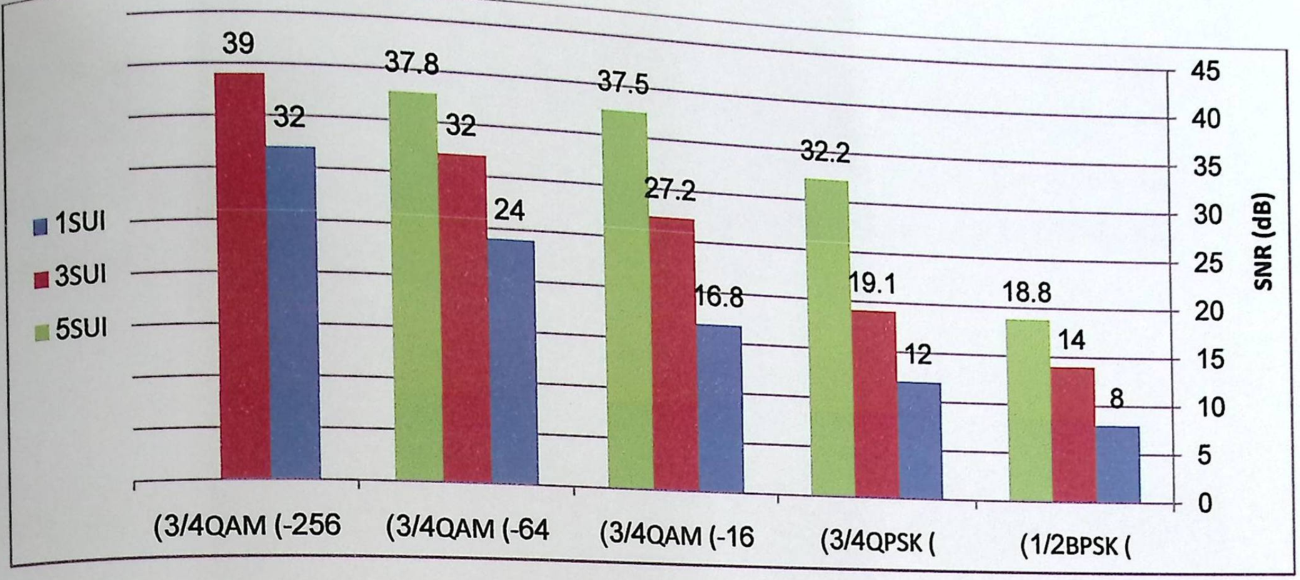


Figure (6.12) : Required SNR at BER of 10^{-3} , BW = 5MHz ,G=1/4

6.2.3 The effect of the coding (Reed Solomon – Convolutional encoding).

In this part we will introduce the results and discuss the effect of the coding on the WiMAX system ,and we will see how the coding improves the performance at low SNR compared to the required SNR when the coding is not used .

The word "coding" in this research means : the using of Reed Solomon as an outer coding , then the convolutional encoding as an inner coding , respectively.

The modulation schemes that we examined under this part are QPSK (3/4) , 16-QAM (3/4) and 64-QAM (3/4) .The SUI channel model that is used in this part is SUI 4 .

The figures from 6.13 to 6.15 show the effect of the coding on a different modulation schemes under SUI 4.

We note from the figures that the coding improves the performance by decreasing the SNR for achieving the same BER that is achieved when there is no coding is used.

For example , from figure (6.13) to get a BER of 10^{-3} in the QPSK (3/4) under SUI 4 when no coding is used, the required SNR for achieving this BER is 28.7 dB ,while when the coding is used ,the required SNR to achieve BER of 10^{-3} is 21.9 dB , so the coding decreases the SNR by a 6.8 dB to achieve the same BER . In other words, the coding gains a 6.8 dB improvement at BER of 10^{-3} .

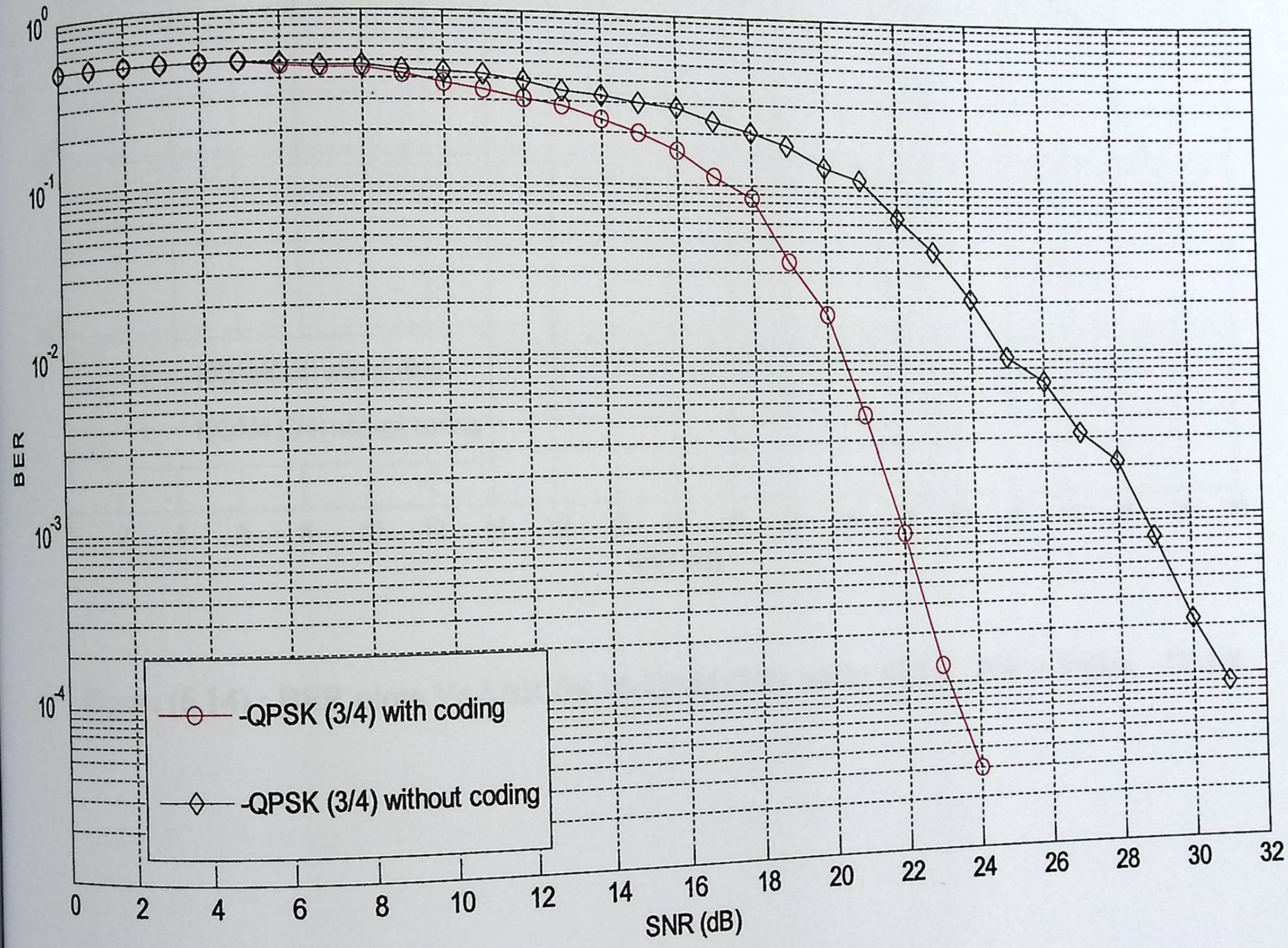


Figure (6.13) : BER plots Vs SNR for QPSK (3/4) under SUI 4 , BW = 5MHz, G=1/4

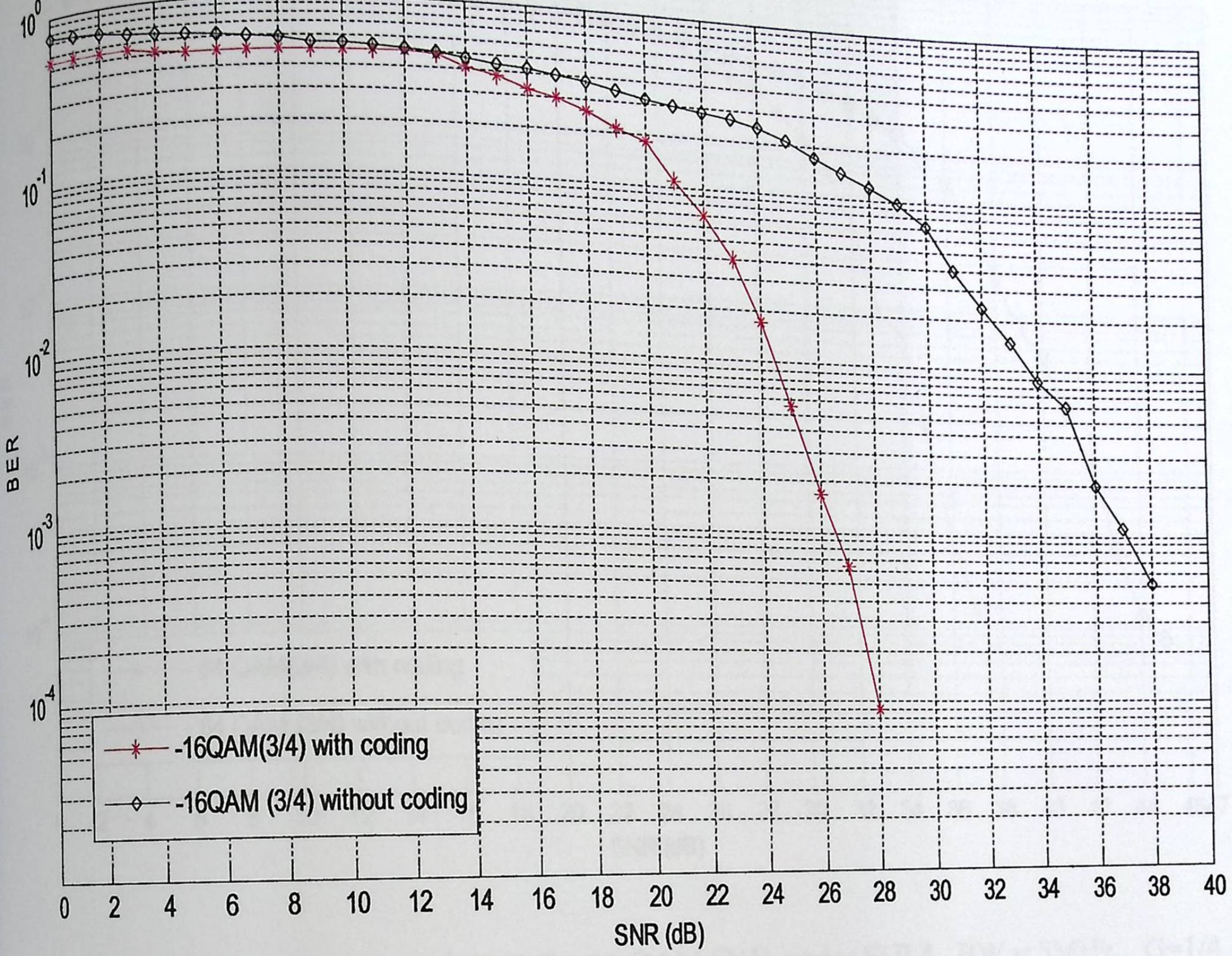


Figure (6.14) : BER plots Vs SNR for 16-QAM (3/4) under SUI 4 , BW = 5MHz, G=1/4

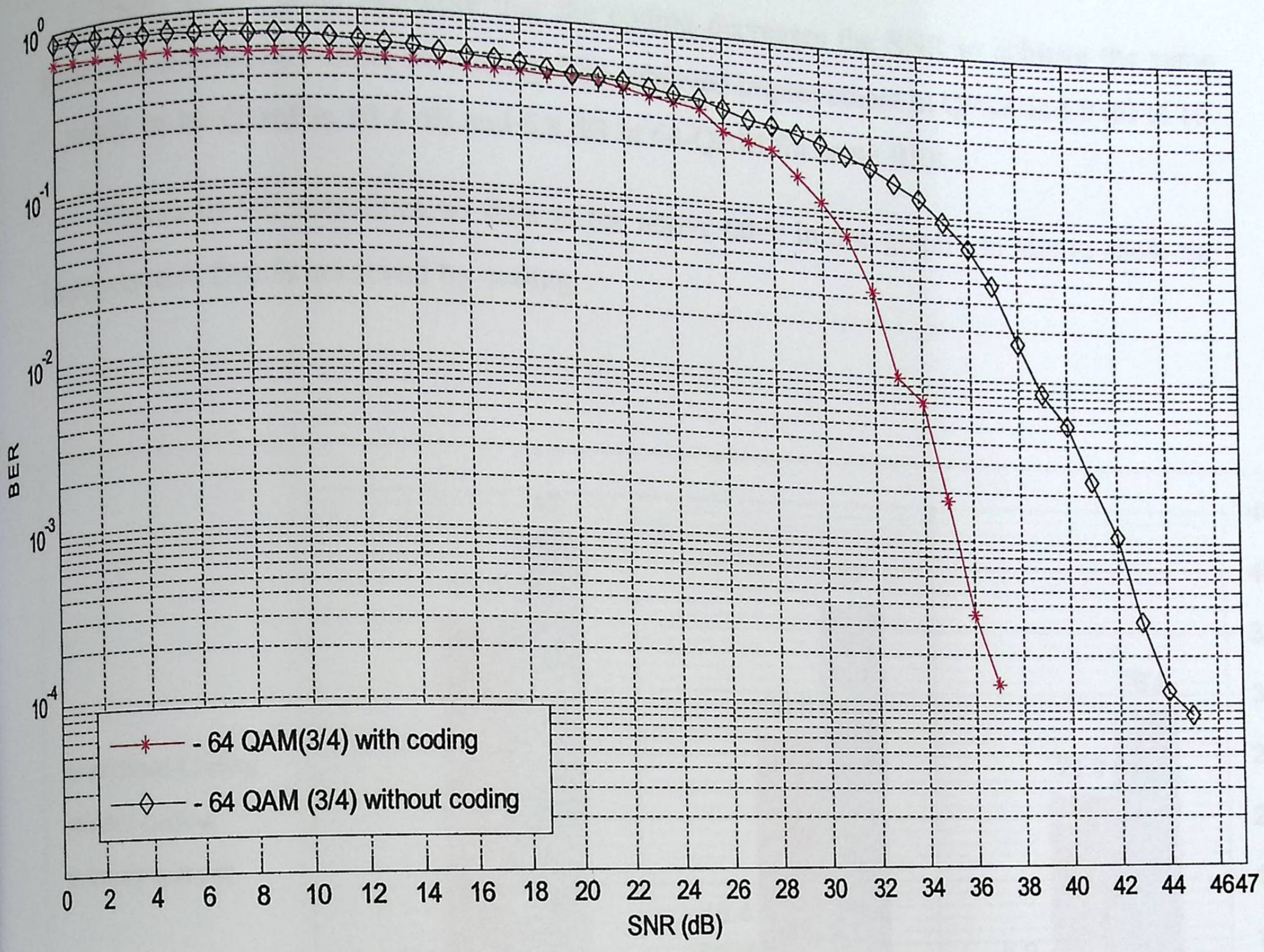


Figure (6.15) : BER plots Vs SNR for 64-QAM (3/4) under SUI 4 , BW = 5MHz, G=1/4

Table 6.4 shows the required SNR and the improvement that is achieved by the coding for different modulations at BER of 10^{-3} under SUI 4.

Table (6.4) : The required SNR at BER = 10^{-3}

	QPSK (3/4)	16 – QAM (3/4)	64 – QAM (3/4)
Without coding	28.8 dB	37 dB	42 dB
With coding	21.9 dB	26.6 dB	35.2 dB
Improvement	6.9 dB	10.4 dB	6.8 dB

From above table ,we note that the coding decreases the SNR to achieve the same BER that is achieved under no coding .The achieved improvement in QPSK is 6.9 dB at 10^{-3} ,while in 16-QAM is 10.4 dB and 6.8 dB in 64-QAM for same BER .

Figure (6.16) combines all data in the above table in one graph , in aim to show the improvement that is achieved by coding .

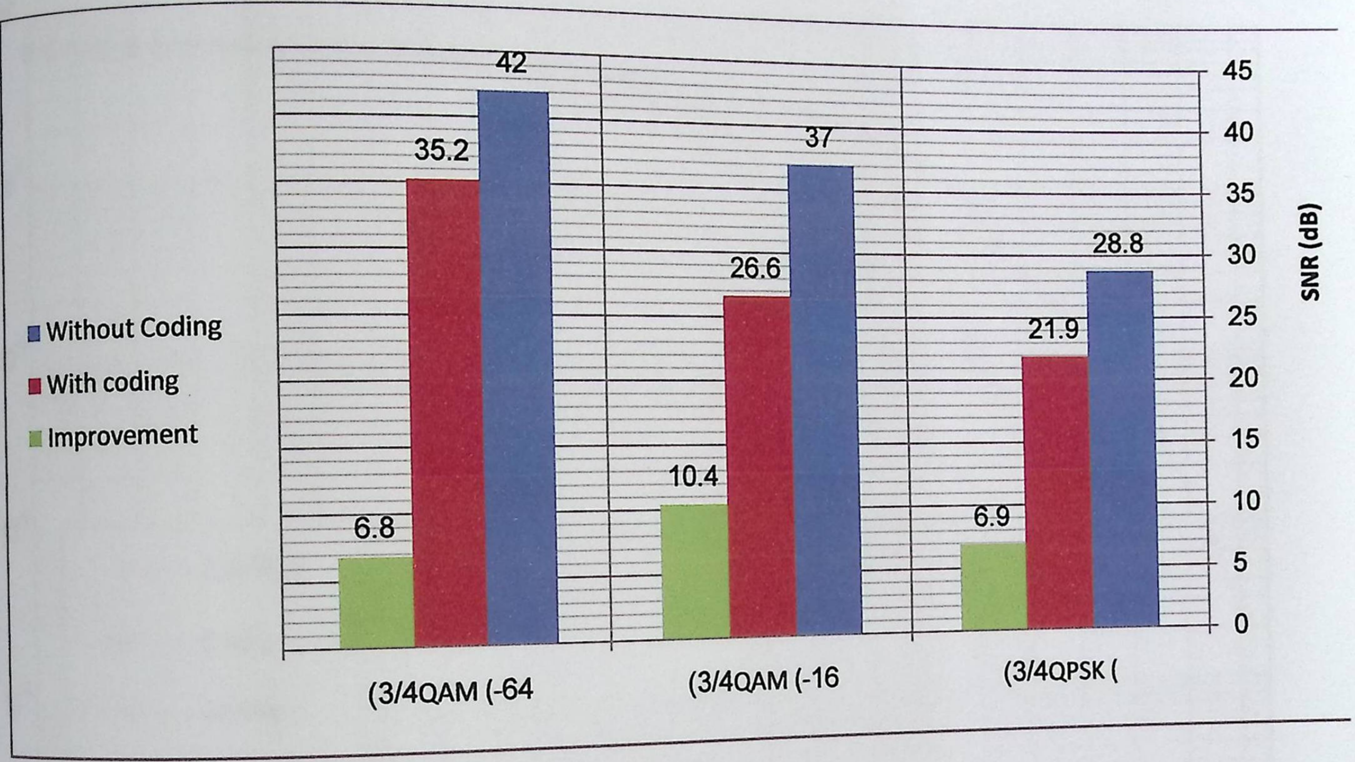


Figure (6.16) : Required SNR at BER = 10^{-3} ,BW = 5MHz , G = 1/4

6.2.4 The effect of the Bandwidth .

The results in this part show the effect of the bandwidth on the WiMAX for a certain modulation schemes under SUI 3 . The scheme that a is tested under this part is the 64-QAM(3/4) .

The figure (6.17) shows the effect of the bandwidth on 64-QAM (3/4) modulation scheme under SUI 3.

We note that when the bandwidth increases the BER increases at certain SNR ,but this increasing is not so much , for example ,from figure (6.17) ,the BER at 31 dB for

1.5 MHz is 1×10^{-3},but for 5 MHz is 10^{-2},while for 10 MHz is 0.08 . The reason for this ,that when the bandwidth increases ,the OFDM symbol time will decrease until it becomes smaller than the delay spread (BW > coherence bandwidth) , then the channel experiences selective fading . This means that the multipath spread will occur ,so the signal will reach the receiver from several different paths with different phases, and these copies of the same signal will be added up out of phase or in phase .

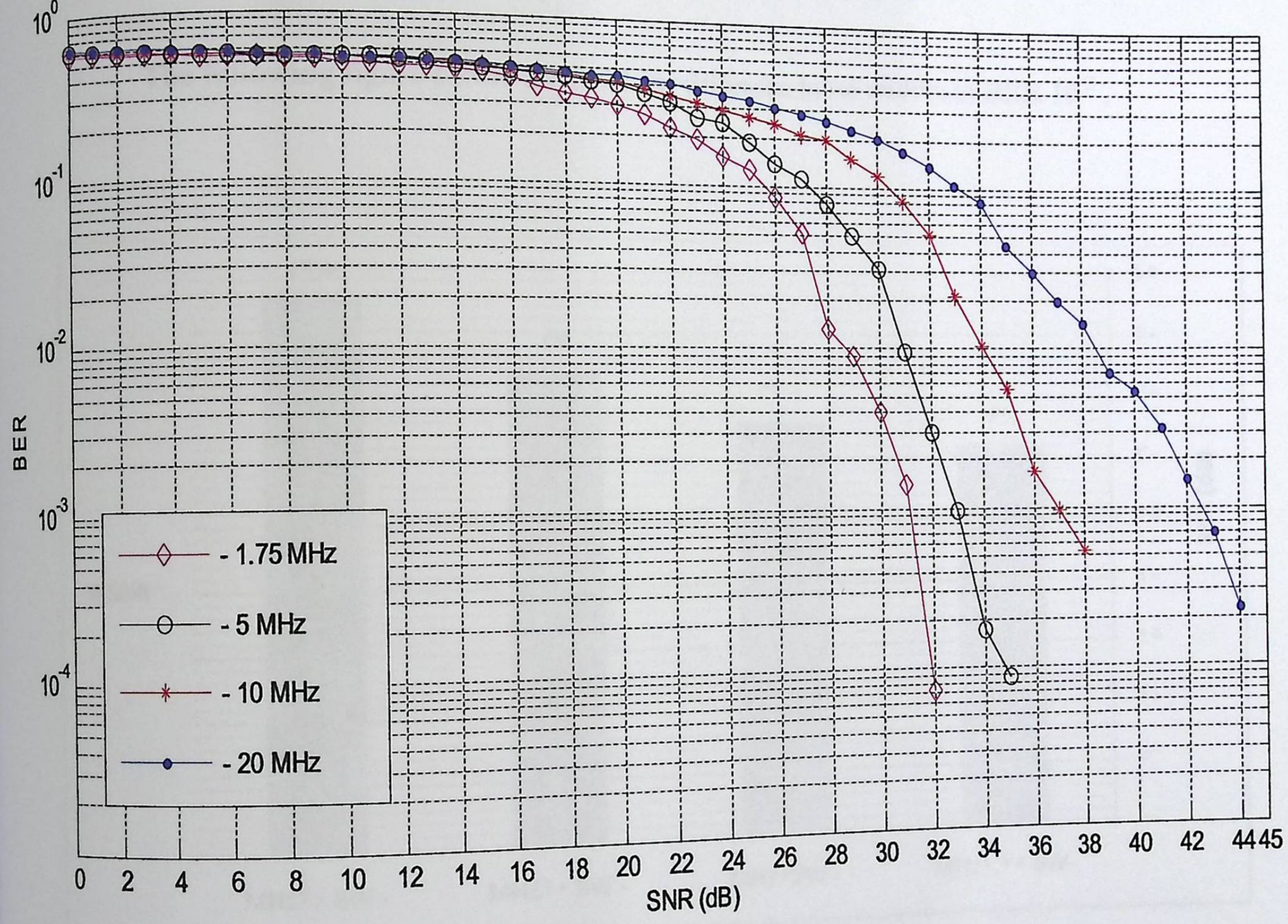


Figure (6.17) : BER plots Vs SNR for 64-QAM (3/4) under SUI 3 at different BW , G=1/4

Table 6.5 shows the required SNR for 64 – QAM (3/4) at BER of 10^{-3} at different band widths under SUI 3 .

Table 6.5 : the required SNR at BER = 10^{-3}

BW Modulation	1.75 MHz	5 MHz	10 MHz	20 MHz
64 – QAM (3/4) BER = 10^{-3} .SUI 3	31 dB	32.9 dB	37 dB	42.5 dB

The following figure shows the SNR for different band widths at BER 10^{-3} .

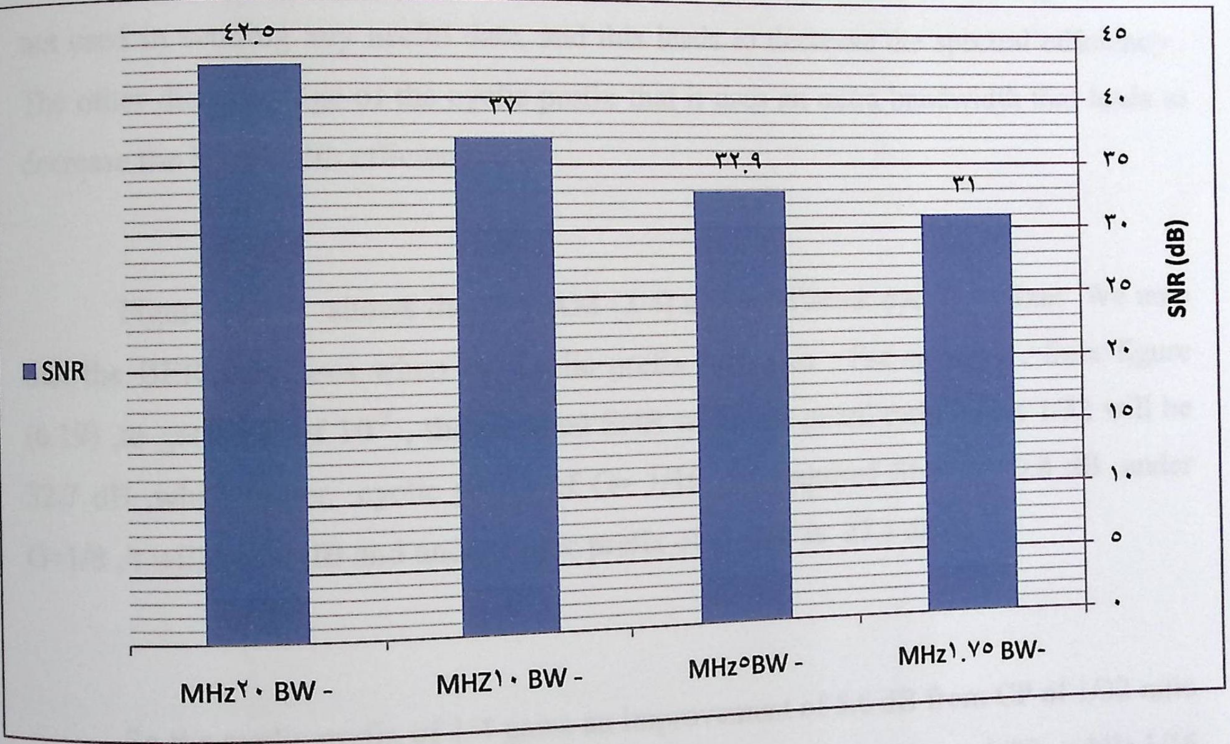


Figure (6.18) : Required SNR at BER 10^{-3} , SUI 3, G = 1/4, 64-QAM (3/4)

6.2.5 The effect of cyclic prefix .

In this part we introduce and discuss the results of the effect of the cyclic prefix and how the cyclic prefix improves performance .

The results in this part are plotted under SUI 3 , and with bandwidth of 5 MHz .Also the scheme that is tested is the 16-QAM (3/4) .

Figure (6.19) shows the effect of the different cyclic prefixes on 16-QAM modulation , where we note as the cyclic prefix increases, the BER decreases at certain SNR ,the reason for this in our opinion is when the cyclic prefix increases ,the channel will become more circulating , and this means that the circular convolution will take its place , so the signal will not be affected by the channel, and if it is affected ,then this effect will be so small since the boundaries of the signal is the cyclic prefix data that have the same data of the last data in OFDM symbol .

But the disadvantages of the cyclic prefix that it consumes an extra power that not used in sending any useful data, and this leads to decrease the spectral efficiency . The other disadvantage of the cyclic prefix that it uses an extra bandwidth that leads to decrease the bandwidth efficiency .

Figure (6.19) shows the 16-QAM (3/4) under different cyclic prefixes .We note that the BER decreases when the cyclic prefix increases . For example , from figure (6.19) ,to get BER of 10^{-3} , the required SNR when the guard ratio (G) is 1/32 will be 32.7 dB ,while under cyclic prefix of G= 1/16 ,the required SNR is 30.8 dB ,under G=1/8 ,it will be 29 dB and under cyclic prefix of G=1/4 is 27.1 dB .

So the cyclic prefix of 1/4 gains an improvement of 5.6 dB from CP of 1/32 ratio at BER of 10^{-3} , while 1/8 gains of 3.7 dB improvement with respect to 1/32 , while 1/16 gains of 1.9 dB improvement with respect to 1/32 .

From above we conclude that G = 1/4 and G = 1/8 are the best cyclic prefixes since they have a big improvement with low guard interval.

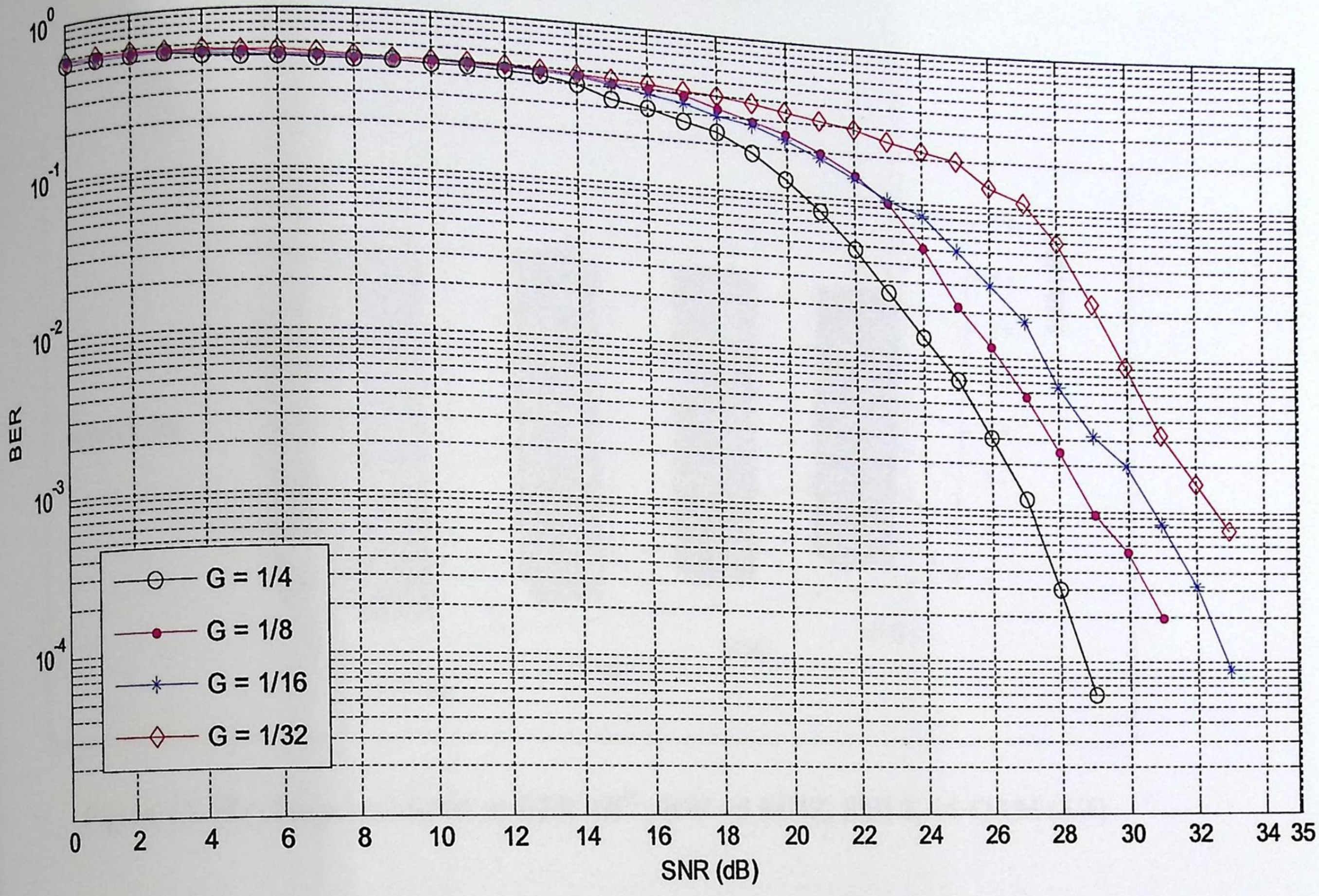


Figure (6.19) : BER plots Vs SNR for 16-QAM (3/4) under SUI 3 at different G ,BW = 5 MHZ

Table 6.6 shows the required SNR at BER of 10^{-3} for different ratio of guard intervals of cyclic prefix under SUI 3.

Table 6.6 : required SNR at BER of 10^{-3}

G Modulation	1/4	1/8	1/16	1/32
16-QAM (3/4) BER= 10^{-3}, SUI 3	27.1 dB	29 dB	30.8 dB	32.7 dB

The following figure shows the above data graphically .

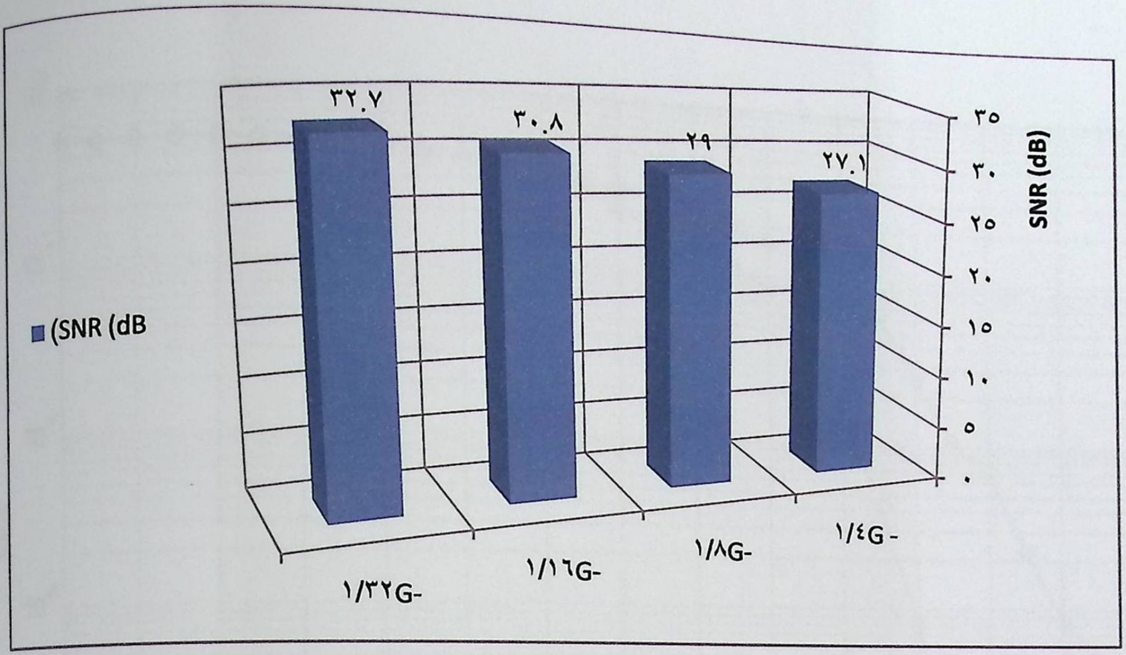


Figure (6.20) :Required SNR at BER 10^{-3} , BW =5 MHZ, SUI 3, 16-QAM (3/4)

6.2.6. The effect of Reed Solomon :

In this part we will check the effect of Reed Solomon on the performance of the WiMAX system , also the improvement that is achieved by the Reed Solomon .

The Reed Solomon is a type of block coding that uses the Galois field as a vector spaces that contain the all unique polynomials that generate a code word whose length determines the highest degree of the polynomial that is used in generating it.

Also the Reed Solomon depends on the hamming distance in its ability to correct the code , in other words ,if the number of the orthonormal bases increases in the code word ,then the correcting ability of the Reed Solomon will increase.

The SUI channel that is used in this part is SUI 4 , and the modulation schemes used in this part are QPSK (3/4), 16 - QAM (3/4), 64 – QAM (3/4).the band width is 5 MHz , and the cyclic prefix is 1/4.

The figures (6.21 – 6.23) show the different modulation schemes under the effect of the Reed Solomon .

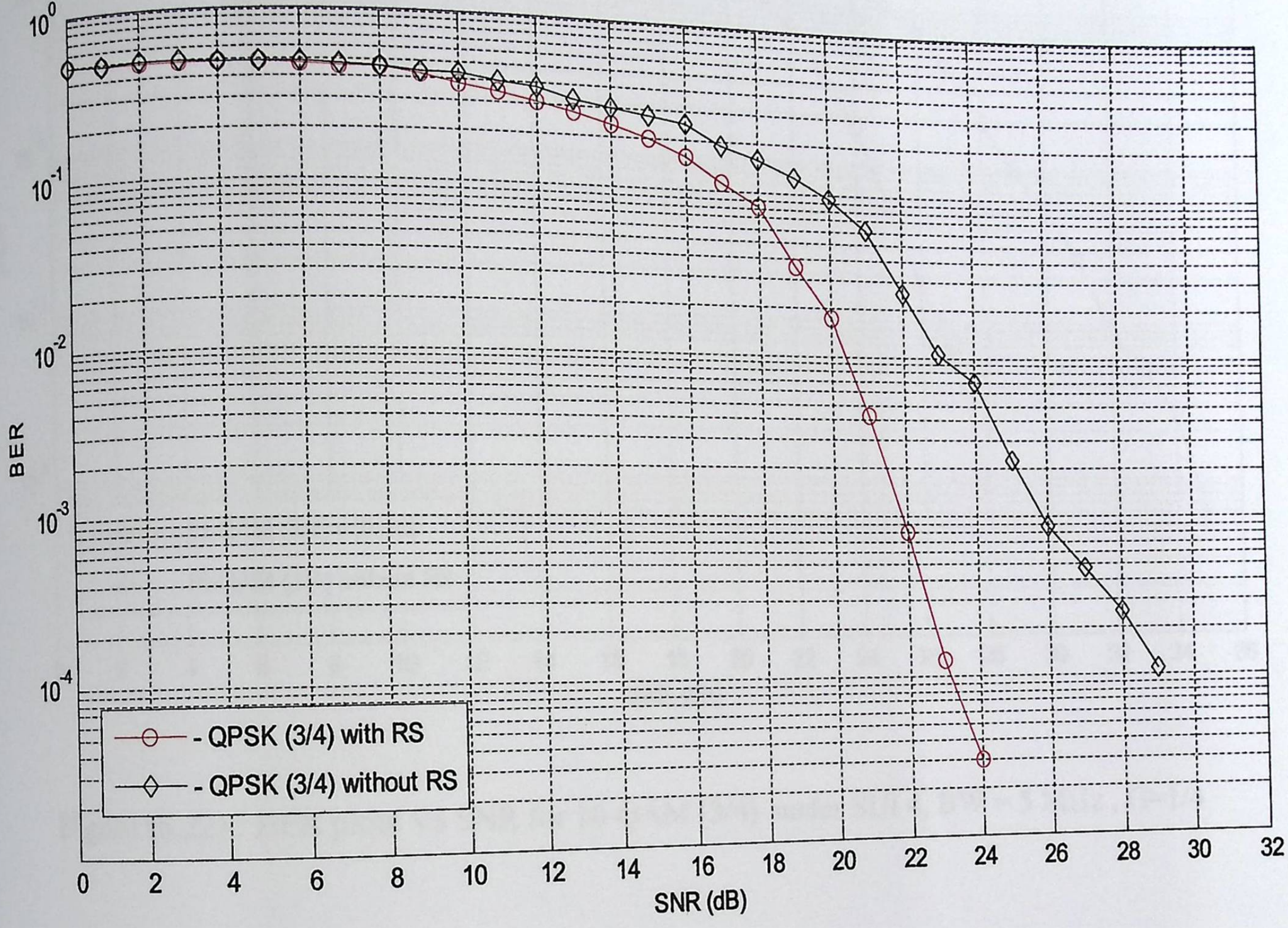


Figure (6.21) : BER plots Vs SNR for QPSK (3/4) under SUI 4, BW = 5 MHz , G=1/4

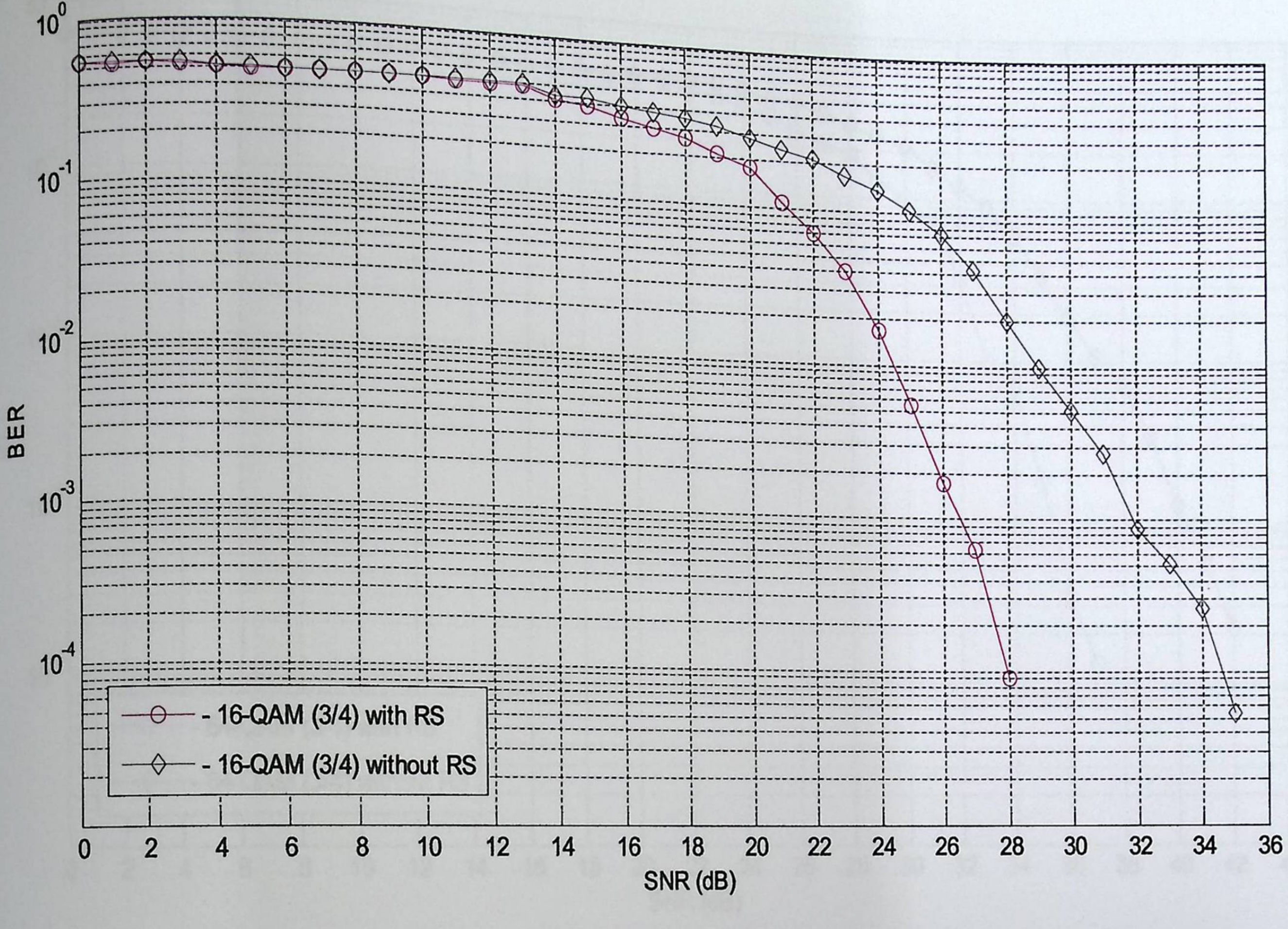


Figure (6.22) : BER plots Vs SNR for 16-QAM (3/4) under SUI 4, BW = 5 MHz , G=1/4

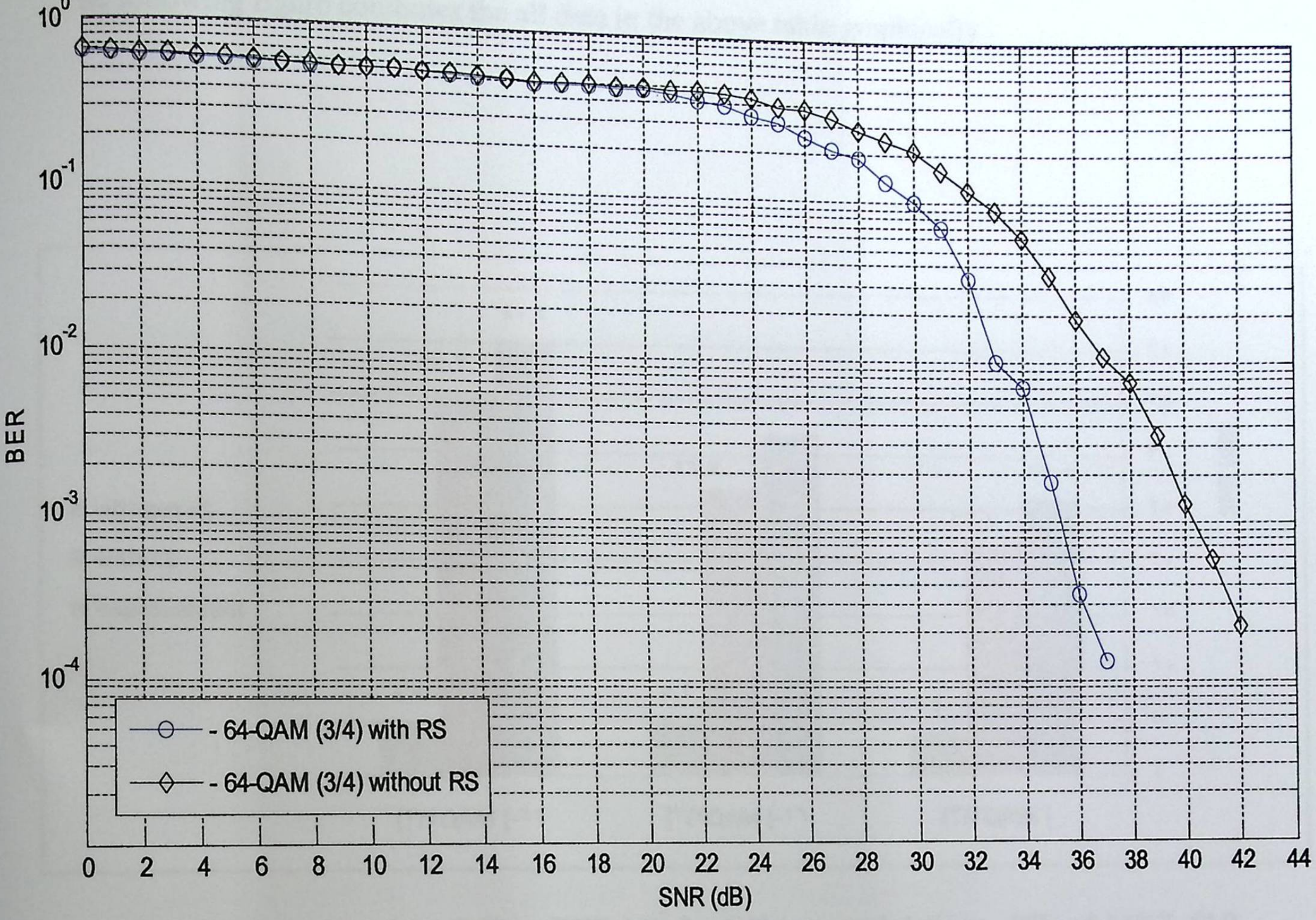


Figure (6.23) : BER plots Vs SNR for 64-QAM (3/4) under SUI 4, BW = 5 MHz , G=1/4

Table 6.7 shows the required SNR and the improvement that achieved by the Reed Solomon at BER of 10^{-3} for different modulations under SUI 4 ,with BW of 5 MHz ,G of (1/4) .

Table 6.7 : required SNR at BER of 10^{-3}

	QPSK (3/4)	16-QAM (3/4)	64-QAM (3/4)
Without using RS	25.9 dB	32 dB	40.2 dB
With using RS	21.9 dB	26.7 dB	35.3 dB
Improvement	4 dB	5.3 dB	4.9 dB

The following figure combines the all data in the above table graphically.

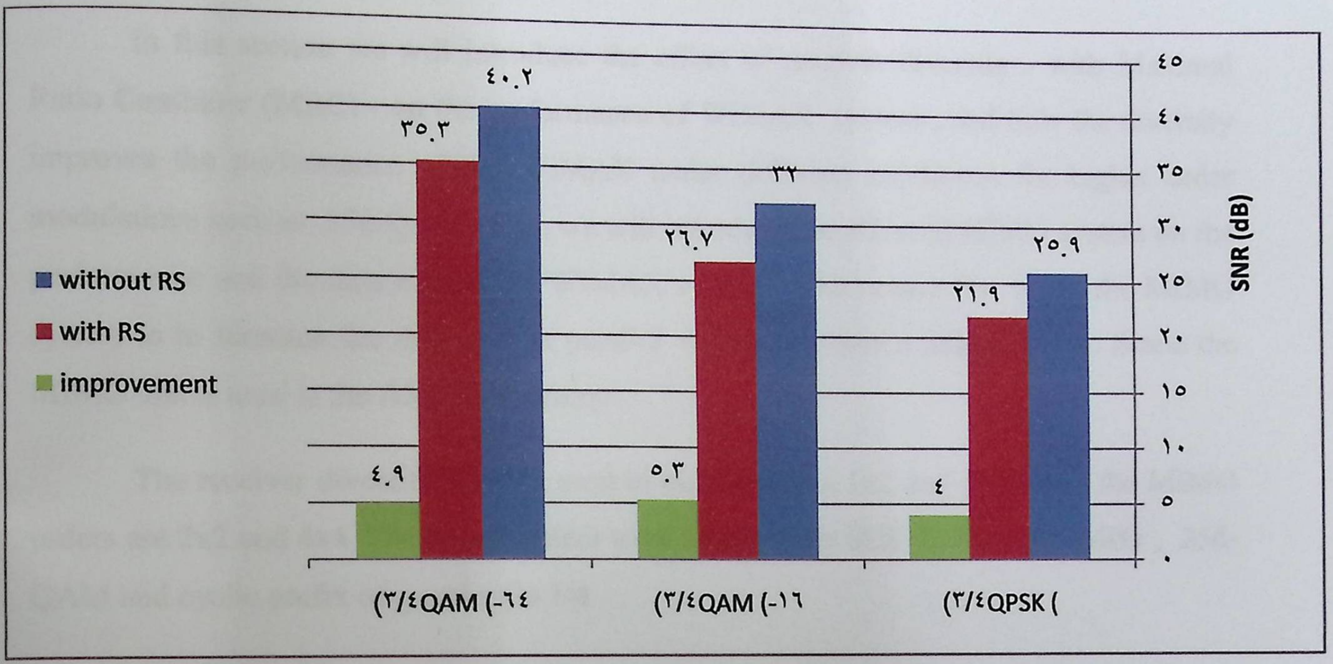


Figure (6.24) : Required SNR at BER 10^{-3} for different modulations , BW =5 MHZ, SUI 4 , BW =5 MHz , G =1/4

6.2.7. The effect of Diversity and MIMO on WiMAX system

In this section we will introduce the effect of receiver diversity - with Maximal Ratio Combiner (MRC) - on the performance of WiMAX system , and how the diversity improves the performance of the WiMAX under different conditions for higher order modulations such as 256-QAM . Also we will introduce the effect of MIMO system on the performance and the data rate of the WiMAX system . The reason for using the MIMO system is to increase the data rate in parallel with performance improvement. Since the MIMO that is used is the Alamouti scheme .

The receiver diversity that we used in our project is 1x2 and 1x4. Also the MIMO orders are 2x2 and 4x4. These parameters were tested under SUI 1, BW of 5 MHz , 256-QAM and cyclic prefix of guard ratio 1/4 .

Also we used the MRC combiner ; this combining method is the best ; since it maximizes the output SNR of the receiver , where the output SNR of this combining method equals to the summation of the output SNR for all receivers that are used in the receiving end . Therefore the MRC SNR increases when the number of antennas at the receiver increases.

The figure (6.25) shows the effect of receiver diversity with different orders on the performance of WiMAX PHY layer under SUI 1 , BW = 5 MHz and G = 1/4 for 256-QAM .

From the figure ,we note that the BER at certain SNR decreases when the diversity is used .Also the BER decreases when the diversity order increases , The reason for this ; that the receiver diversity uses multiple antenna at the receiver and this enables the receiver to receive different copies of the same signal , so the probability that all copies will be under the threshold of detection will be very small ; for example , if we assume that the probability of the signal to be under the threshold level is P , then the probability for n different copies of the same signal to reach the receiver under that threshold will be $(P)^n$, so the probability is decreased to the n^{th} power of P .

As we note from the figure (6.25) that the required SNR to achieve , i.e BER of 10^{-3} under no diversity is 35.5 dB ,while under 1x2 diversity the required SNR for the same BER is 29.8 dB and under 1x4 diversity will be 23 dB .In other word ,the diversity 1x2 gains a 5.7 dB improvement at BER level 10^{-3} ,while the diversity 1x4 gains a 12.5 dB improvement at 10^{-3} . Also 1x4 gains a 6.8 dB improvement than the 1x2 at BER level of 10^{-3} .

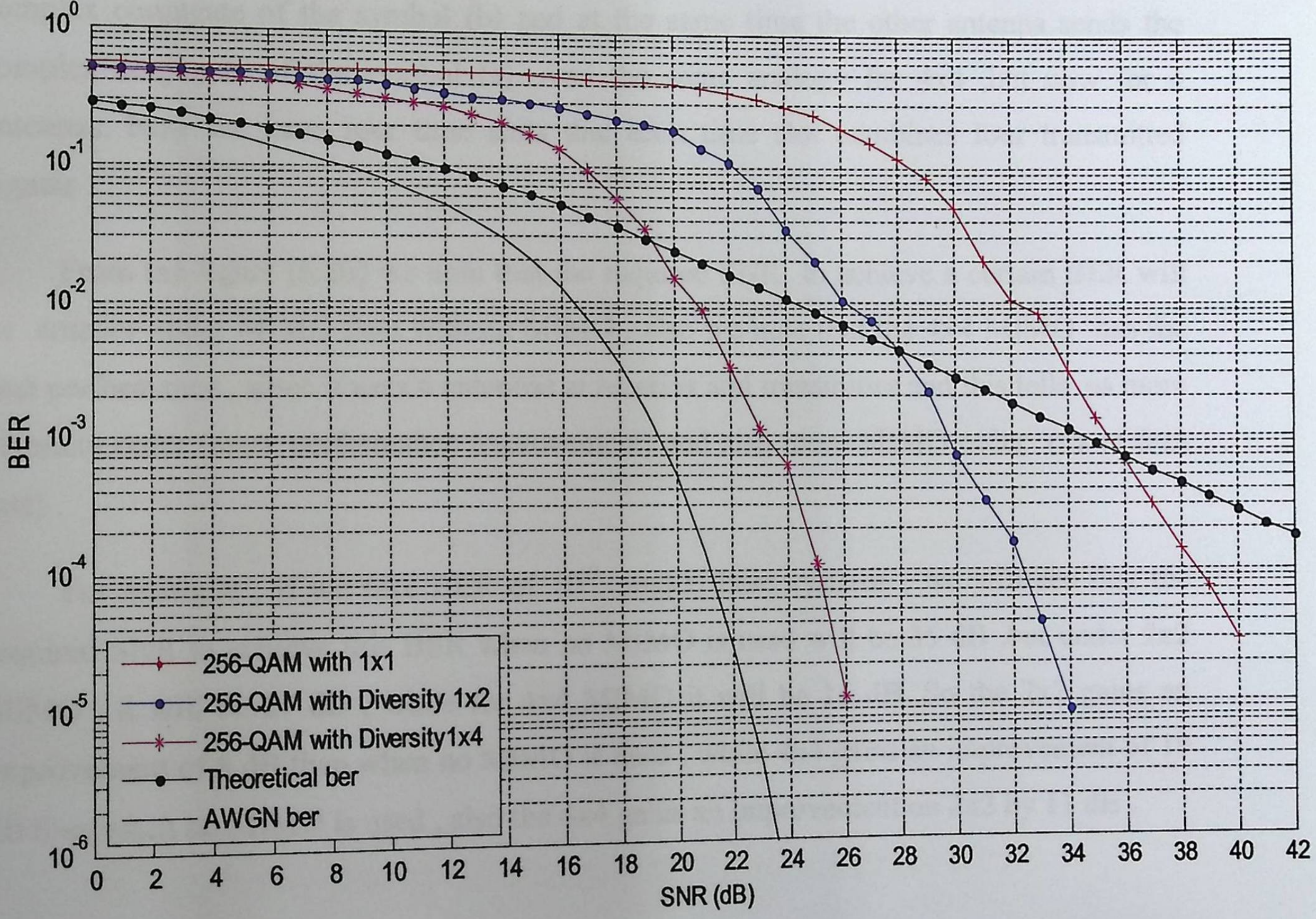


Figure (6.25) : BER plots Vs SNR for different receiver diversity orders under SUI 1, 256-QAM , BW = 5 MHz , G=1/4

The figure (6.26) shows the effect of MIMO system with different orders on the performance and the data rate of WiMAX system under SUI 1 , BW = 5 MHz and G = 1/4 for 256-QAM .

From this figure , we note that the performance of the WiMAX increases at low SNR in addition the data rate increases. The reason for this that the Alamouti MIMO scheme increases both the performance and the data rate ,since this method uses the space-time block code (STBC) . This scheme put a method that utilizes from the diversity and at the same time utilizes from the multiple antennas at the receiver and transmitter . So it combines two opposite properties in one system. This method in 2x2 order is summarized in this way ; the first antenna sends for example the symbol (a) and at the same time the other antenna sends the other symbol say (b), after that , the first antenna sends the minus complex conjugate of the symbol (b) and at the same time the other antenna sends the complex conjugate of the symbol (a). And the same analogy for 4x4 ,but now for 4 antennas. Now we have four time slots that each time slot combines four transmitted signals .

From the figure (6.26) we note that the required SNR to achieve a certain BER will be smaller under MIMO than without MIMO , also we note that the 4x4 MIMO has the best performance , since it uses 4 antennas at receiver and transmitter and this tells us more diversity order (more performance improvement) and also more MIMO order (more data rate) .

For example , if we take BER of 10^{-3} under these systems , we will see that the required SNR to achieve this BER when no MIMO is used will be 35 dB ,but under 2x2 MIMO , it will be 27 dB , while for 4x4 MIMO it will be 16 dB. So the 2x2 gains an improvement of 8 dB than when no MIMO is used , while 4x4 gains an improvement of 19 dB than when no MIMO is used , also the 4x4 gains an improvement on 2x2 by 11 dB .

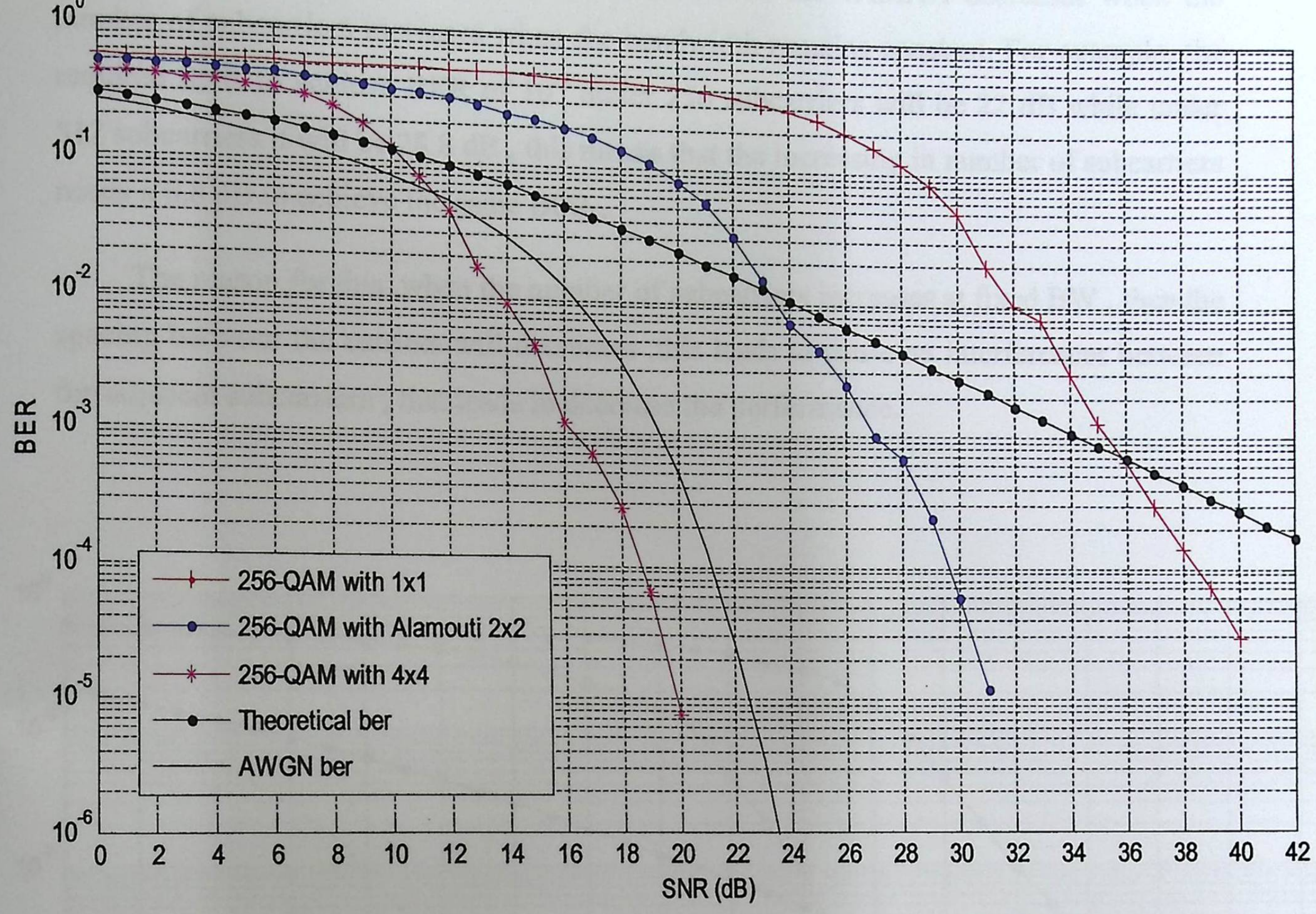


Figure (6.26) : BER plots Vs SNR for different MIMO orders under SUI 1, 256-QAM , BW = 5 MHz , G=1/4

6.2.8. The effect of number of subcarriers

In this section we examine the effect of subcarriers in order to increase the data rate. Also we assume that the bandwidth is fixed and does not increase with number of subcarriers in aim to see what happens when the number of subcarriers increases or decreases .

Under this section we use the 16-QAM as the modulation scheme ,the bandwidth is 5 MHz ,the guard ratio is 1/4 , the channel model is SUI 3 , 256 subcarriers and 512 subcarriers .

Figure (6.27) shows the 16-QAM under 256 subcarriers and 512 subcarriers.

From the figure , we note the performance of the WiMAX decreases when the number of subcarriers increases when the bandwidth remains constant. For example, the required SNR to achieve BER of 10^{-3} under 256 subcarriers will be 22 dB while under 512 subcarriers it will be 28.8 dB , this means that the increasing in number of subcarriers needs a 6.8 dB to achieve the same BER .

The reason for this ,when the number of subcarriers increases at fixed BW , then the spacing between the carriers will decreases ,this leads to cause an interferences between the adjacent subcarriers ; that leads to decrease the performance.

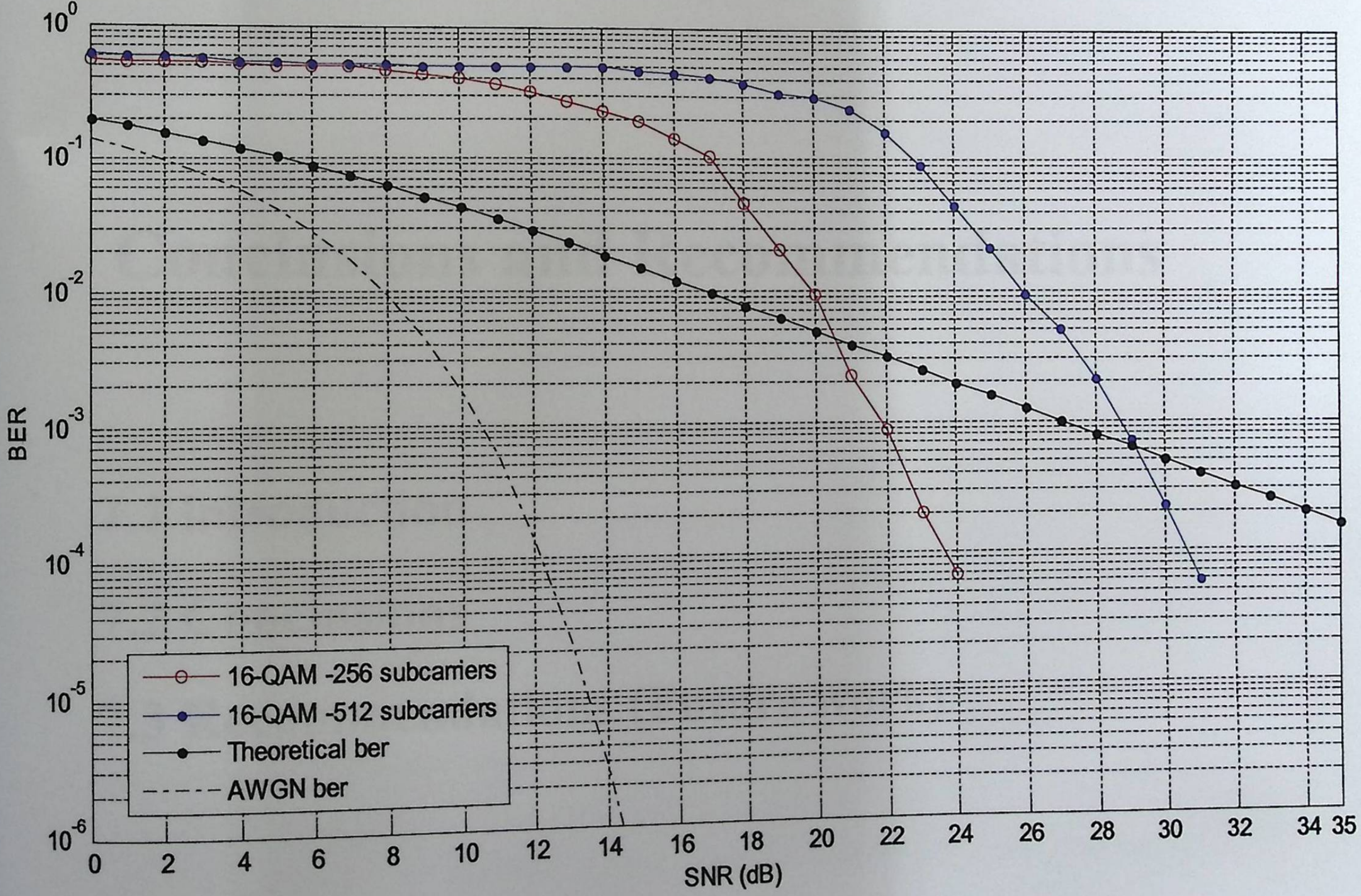


Figure (6.27) : : BER plots Vs SNR for different numbers of subcarriers under SUI 3, 16-QAM , BW = 5 MHz , G=1/4

Chapter 7

Conclusions and Recommendations

7.1 Introduction

7.2 Conclusions

7.3 Recommendations (Future work)

7.1. Introduction

In this chapter we introduce the important conclusions that define best configuration of WiMAX system . In order to achieve the best performance of WiMAX system under different fading conditions .However, this configuration is different from condition to other.

7.2. Conclusions :

1) Our results show that low order modulations (BPSK ,QPSK) are more suitable for the worst fading channels such as (SUI 4 – SUI 6) , while the higher order modulations (16-QAM, 64-QAM and 256-QAM) needed softer channel .
2) Our results showed that higher modulations are much better implemented under the worst conditions if diversity is used .
3) Forward error correction with QPSK modulation provided a coding gains of 7 dB improvement under QPSK at BER 10^{-3} ,it gains 10 dB improvement with 16 – QAM and gains 7 dB improvement with 64-QAM . The SUI channel that used in this section is SUI 4.
4) The best cyclic prefix for WiMAX system was when the time length of it is quarter the useful time length of the OFDM symbol (G = 1/4).
5) Cyclic prefix of (G=1/8 or G= 1/16) could be used with under high order modulations if good fading conditions existed.
6) 256-QAM modulation with diversity and MIMO .Results showed a good improvement in performance.
7) The best bandwidth is that ranges from 1.75 MHz to 20 MHz .
8) 512 subcarriers can be used if the low order modulations are used . also can be used if the 16-QAM is used without increasing the bandwidth , but for higher modulations ,the bandwidth must be enlarged to cope with large number of subcarriers .
9) Under MIMO 4 x 4 , the BER of 256-QAM with BW = 5 MHz ,SUI 1, G=1/4 will be better than the AWGN BER (non-fading) .
10) Receiver diversity of 1 x 2 gains a 5.7 dB improvement at BER level 10^{-3} ,and 1 x 4 gains 12.5 dB improvement at BER of 10^{-3} under 256-QAM .

7.3. Recommendations (Future work)

The following things can be done to gain more improvement on WiMAX system in both data rate and performance .

1) Implementing the Turbo coding at the transmitter and at the receiver.
2) Using WCDMA(Wide Code Division Multiple Access) with OFDM .
3) Using 1048 subcarriers or more to increase the data rate.
4) Advanced Channel Estimation can be used at the receiving end .
5) Implementing higher diversity order at receiver.
6) Implementing higher transmit diversity to enhance the downlink.
7) Using high MIMO order such as (8 x 8).
8) Using Space-Time-Frequency MIMO OFDM .

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