PERFORMANCE IMPROVEMENT OF A WCDMA SYSTEM USING PULSE SHAPING FILTER AND COMPENSATION FOR ERRORS IN THE PRESENCE OF INTERFERENCE CHANNELS

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Performance Improvement of a WCDMA system using pulse shaping filter and compensation for errors in the presence of interference channels

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DECLARATION

This thesis is my original work and has not been presented for a degree in any other university.

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This thesis has been submitted for examination with our approval as University supervisors.

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DEDICATION

I dedicate this thesis to Emmah and my parents Mr. and Mrs. Ombongi for their love, support, and encouragement during the research.

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This research has been accomplished through the contribution of many people who need to be acknowledged for the time they spared to offer what was required. Firstly, glory to the Lord because all that has been achieved was through Him. My deepest appreciation goes to supervisors; Dr. Philip Kibet and Prof. Stephen Musyoki for their invaluable contribution, careful guidance and constant support for the entire period of research. I would like also to thank my parents, brother and sisters for all the support and encouragement they gave throughout the time I started the masters programme and development of this thesis. Thanks are owed to Masinde Muliro University for their much appreciated financial support towards this study. In addition, I sincerely thank Jomo Kenyatta University of Agriculture and Technology for granting me the opportunity to pursue my studies.

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LIST OF ABBREVIATIONS

3G	Third Generation
3GPP	Third Generation Partnership Project
AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
BTS	Base Transceiver Station
CDMA	Code Division Multiple Access
CN	Core Network
DQPSK	Differential Quadrature Phase Shift Keying
DSP	Digital Signal Processing
DSSS	Direct Sequence Spread Spectrum
EDGE	Enhanced Data Rates for Global Evolution
FDD	Frequency Division Duplex
FFT	Fast Fourier Transform
FIR	Finite Impulse Response
FPGA	Field Programmable Gate Array
FSK	Frequency Shift Keying
GGSN	Gateway GPRS Support Node
GMSK	Gaussian Minimum Shift Keying
GPRS	General Packet Radio Service
GUI	Graphical User Interface

HLR	Home Location Register
ICI	Interchannel Channel Interface
IIR	Infinite Impulse Response
IMT-2000	International Mobile Telecommunication of 2000
IP	Internet Protocol
ISI	Inter-Symbol Interference
ITU	International Telecommunication Union
LUT	Look Up Table
MS	Mobile Station
MSC	Mobile Switching Centre
MSK	Minimum Shift Keying
ODQPSK	Offset Differential Quadrature Phase Shift Keying
OQPSK	Offset Quadrature Phase Shift Keying
PSK	Phase Shift Keying
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RAN	Radio Access Network
RF	Radio Frequency
RNC	Radio Network Controller
SGSN	Serving GPRS Support Node

SNR	Signal to Noise Ratio
SRC	Square Root Cosine
SRCF	Square Root Cosine Filter
SRRC	Square Root Raised Cosine
TDD	Time Division Duplex
UE	User Equipment
UMTS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Radio Access Network
VLR	Visitor Location Register
VSA	Vector Signal Analyzer
VSG	Vector Signal Generator
WCDMA	Wideband Code Division Multiple Access
WiFi	Wireless Fidelity

ABSTRACT

Most mobile operators have rolled out 3G networks, which have more benefits compared to the second and first generation networks. This is because they can support voice, data and multimedia applications. The signal transmitted over these mobile systems suffers from interference due to thermal noise, propagation loss in space and distortion of bit sequence for higher bit rate applications. The current work focussed on performance improvement in a Wideband Code Division Multiple Access (WCDMA) system using pulse shaping filter and compensation for errors technique (channel coding) over the Additive White Gaussian Noise (AWGN) channel. The system model was simulated for a data rate of 2Mbps with and without convolutional coding scheme and using QPSK, 16-PSK, 16-QAM and 64-QAM modulation techniques to modulate the signal at the transmitter. The WCDMA system model was developed and simulated using computer simulation tool MATLAB 7.8 for different digital modulation techniques applied in turns and variation of filter roll off factor from 0.1 to 0.9. For the case of variation of roll off factor, the roll off factor with the minimum error rate in each case was noted.

These results show that QPSK modulation format had better performance, with and without convolutional coding, when comparison was made with other modulation schemes making it an efficient modulation scheme at 2Mbps data rate. Therefore, it can be applied only for a system with poor channel conditions and other modulation schemes applied in systems with better channel conditions. The application of convolution coding reduced the BERs encountered in the system at 2Mbps which increased capacity due to lower error rate. The convolutional coding scheme also improved the power efficiency of the system, which meant that the power required to transmit a signal reduced for this high bit rate application. The variation of the pulse shaping filter roll off factor showed that the performance of a wireless communication system employing this type of filter depends on the value of its roll off factor. The filter roll off factor, α, of 0.3 was found to give a lower error rate in many levels of the bit energy to noise power spectral density ratio, which made it an most favourable value at 2Mbps data rate to achieve a better power efficiency and BER level without increasing the complexity of the filter.

CHAPTER ONE

INTRODUCTION

1.1 Background of the Study

Wireless mobile communication allows voice and multimedia data transmission from a computer or a mobile device without a physical fixed link connection. Most consumers have acquired devices which can receive services from various evolutions of mobile communication. This has resulted in mobile communication technologies that have made businesses conduct their operations faster and efficiently which in turn raises the standards of living. The mobile technologies have evolved from the first generation (1G) in 1970 to the fifth generation (5G) which is expected to start implementation in some countries from 2015 [1]. These technologies are based on different security standards and transmission protocols. The current study focuses on the Third Generation (3G) Wideband Code Division Multiple Access (WCDMA) which can offer augmented bandwidth capability, multiple mobile applications and clarity of the digital signals transmitted across its network infrastructure. The technology can also allow transmission of packet switched data economically at a better and increased bandwidth giving more advanced services to mobile users. In addition, many multimedia applications can be transmitted with a better spectral efficiency realized.

The WCDMA technology is an air interface technology whose specifications were developed by the 3G Partnership Project (3GPP) in the year 2000. It is a standard of the International Telecommunication Union (ITU) derived from the CDMA called International Mobile Telecommunication of 2000 (IMT-2000) direct spread spectrum which has received wide adoption all over the world. The bandwidths allocated for this system are 5MHz, 10MHz and 20MHz and are used to offer flexibility of operation. The current WCDMA utilizes 5MHz. [2].

The technology is based on direct sequence spread transmission and therefore a wideband transmission scheme. In Direct Sequence Spread Spectrum (DSSS), the user data is directly multiplied by a code which is a pseudo-random sequence of ±1 and on reception the same code is used to extract the original signal from the incoming wideband signal. A bit of the code is called a chip and the defining parameter for such a system is the chip rate. The frequency of the pseudo random sequence is very high compared to the frequency of the data. The higher frequency of the PN sequence which is given in terms of mega chips per second ensures that there is widening of the occupied spectrum of the data signal. The wider bandwidth can then be used to implement variable rate services as the data can be spread over the 5MHz bandwidth per carrier. Therefore, the high chip rate provides the ability to accommodate greater number of users who can access integrated services (voice, text, video and pictures) at different transmission rates [3].

The DS-CDMA is the form used for the air interface in the UMTS known as WCDMA with a chip rate of 3.84Mcps (mega chips per second). The variable spreading and multicode connection are used in WCDMA to make the system support high data rates up to 2Mbps. The chip rate of the pseudo-random sequence is used to lead a carrier of 5MHz bandwidth. Since the WCDMA has a wider bandwidth it has more benefits compared to narrowband systems that were used earlier. The larger bandwidth of 5MHz enables the WCDMA system to accommodate higher bit rates and makes it possible to divide and combine reception signals propagated through multipath fading channels into more multipath components which help to improve reception quality. In addition, the wider bandwidth allows multiplexing of services with different quality requirements on a single connection such as speech, video and packet data. In DSSS the signal is spread over a large frequency range. For example, a voice speech signal with a bandwidth of 3.1 KHz would be spread over 5MHz when transmitted over the WCDMA system. The bandwidth increases but the information transfer rate remains constant.

This is achieved by using a technique which has a code to represent a symbol of the transmitted message. The code is made up of a number of binary digits (bits) each one of which is referred to as a chip. The whole code consisting of all of the chips representing a symbol takes up the same span as the original symbol. The WCDMA technology supports broadband, packet based transmission of voice, text, multimedia and video data sets from a data rate of 384kbps to 2Mbps which means the subscriber can access high speed internet, videoconferencing and basic video or TV services.

These wireless communication systems suffer from multipath fading, time dispersion or time delay caused by spreading which often leads to inter-symbol interference (ISI) and lower bit rate capacity. The Bit Error Rate (BER), which is also affected by the transmit power and bandwidth, requires that the power level of the system is increased and bandwidth reduced for a better BER to be obtained. The reduced bandwidth admits less noise into the system, thereby increasing the signal-to-noise ratio; this in turn decreases the BER. This depends on the modulation scheme and data rate. When data rate is increased, the power per bit is also increased which makes higher data rate applications to require large transmit power for performance, in terms of BER, to be improved. The performance of a system on utilization of bandwidth is measured as bandwidth efficiency or spectral efficiency which needs to be low for the system to minimize the noise received for better BER at higher data rates.

1.2 Evolution of Wireless Communication Networks

The first-generation (1G) of wireless telephone technology and the mobile telecommunication was using the analog telecommunications standards. It was an analog cellular telephone system which used voice only during 1980s. The voice was frequency modulated with Frequency Division Multiple Access (FDMA) capability which was an analog modulation scheme. The 1G system had 30 kHz channels occupying the 824MHz – 894MHz frequency band and its usage of an analog modulation technique is what made this technology referred to as an analog system.

The Second Generation (2G) technology brought an improvement to the 1G technology with the introduction of digital modulation where the voice generated was converted into a digital code in the phone and then into an analog signal for transmission. The two technologies were both using digital signalling for connection of the radio towers to the rest of the telephone system. During a call conversation in 2G, voice was encoded to digital signals while in 1G the voice underwent a modulation to a higher frequency of 150MHz and above. The speed of 1G varies in the range of between 28kbit/s and 56kbit/s with the actual download speeds of 23.2kbits/s to 44.8kbits/s [4] and its data capacity is 2kbps in the 800-900MHz band.

The 2G had three major merits over the 1G which included the digital encryption of the phone conversation. This improved the privacy which was not possible in the 1G technology where the calls in 2G were almost impossible to eavesdrop by the use of radio scanners which ensures the privacy and safety of data that is transmitted is upheld. The 1G phone was very insecure because anyone with an all-band radio receiver can listen to the conversions going on in the network.

The second merit involved the allowance of greater penetration of mobile phones which was achieved by subdividing the available 25MHz of bandwidth into 124 channels each with a bandwidth of 200 kHz. Each frequency is then subdivided in the air interface of this technology based on narrowband Time Division Multiple Access (TDMA) technology into 8 time slots with each having access to one time slot at regular intervals. The narrowband TDMA allowed eight simultaneous calls on the same frequency. This protocol allowed large number of users to access a single radio frequency by allocating time slots to multiple voice or data calls. The TDMA breaks down data transmission such as a phone conversation into fragments and transmits each fragment in a short burst with the assignment of a time slot to each fragment. The sharing of the channels dynamically amongst a number of users, which was achieved through allowing multiple users on a single channel via multiplexing, results in greater penetration level of mobile phones. This also ensured that there is efficient utilization of spectrum.

The third merit was introduction of data services such as the Short Message Service (SMS) and email. This was achieved through the connection of a phone to a computer which acted as a modem for email, fax and internet browsing. The SMS was also cheap and an easier way to communicate with anyone.

The technologies in this generation can be classified into TDMA and Code Division Multiple Access (CDMA) standards when the type of multiplexing is considered.

The main standard was Global System Mobile (GSM) where 8 time slotted users for each 200 kHz radio channel was supported. The standards included three TDMA standards and one CDMA standard. The set of wireless interface standards that relied on digital modulation and superior digital signal processing techniques in the handset and base stations was covered by the 2G [4]. Its data capacity was 10kbps.

The 2G+ or 2.5G systems are based on technologies which are High Speed Circuit Switched Data (HSCSD), General Packet Radio Service (GPRS) and Enhanced Data Rates for Global Evolution (EDGE). The HSCSD is next to the 3G wideband mobile data networks. This circuit switched technology enhances data rates up to 57.6kbps with the introduction of 14.4kbps data coding and aggregation of 4 radio channels time slots of 14.4kbps. The data capacity for the GPRS technology is 200kbps and EDGE is 473kbps.

The third generation (3G) is a mobile telecommunications technology that represented those networks supporting services with an information transfer rate of at least 200 kbps. Although, most services offered by service providers might be having higher speeds when compared to minimum technical specifications of 3G services. The latest releases of this technology which are sometimes referred to as 3.5G and 3.75G offer mobile broadband access of several megabits per second to smart phones and mobile modems in laptop computers.

This technology is mostly applied in wireless voice telephony, mobile Internet access, fixed wireless Internet video calls and mobile TV. The 3GPP and 3GPP2 are developing standards which extends this technology based on all-IP network infrastructure with advanced wireless technologies such as MIMO applied. The features of this generation includes high bandwidth (5MHz), high speed of up-to 2Mbps [5] and always-online devices.

1.3 Multipath Propagation Effects

The wireless communication system is required to overcome the effects of multiple users with different propagation characteristics transmitting simultaneously. This is often referred to as the near far problem where a remote user can easily be drowned out by a user that is physically much closer to the base station. The drowning out is a worst case scenario that occurs in mobile communications in areas where mobile phones are incapable of transmitting to a nearby base station or repeater. These areas are often referred to as dead zones and mobile phones under their coverage are said to be in outage. The mobile phone service is not present because the signal between the handset and the base stations is blocked or severely reduced by hilly terrain or physical distance.

There are wide swaths of dead zone where there is no coverage available from cell towers in remote areas due to fewer numbers of inhabitants such as in desert areas. The dead zones can still exist in dense populated areas due to improperly aligned towers, which result in areas not receiving proper coverage, as well as in locations which are obstructed by obstacles like hills, mountains, trees, large buildings. They are also found underground in places like basements and parking garages.

The main features of multipath propagation effects include attenuation due to increase in distance from the receiver, fading variations due to specific features of the environment and fading variations due to the movement of the mobile device. The fading weakens a signal at the receiver due to different signal paths during transmission which makes the received signal to have different values of the phases. Fading can be either slow or fast fading. The multipath propagation brings about distortion of the signal during transmission over a medium. This in turn degrades the BER performance and therefore there is need to compensate using appropriate methods at the receiver.

The time variations due to relative motion of the transmitter and receiver which need acquiring of mobile channel states also need optimization. The multipath interference can be Rayleigh or Rician fading which can degrade a signal during propagation from the transmitter to the receiver. Fading causes ISI at the receiver which can be minimized by using a pulse shaping filter. The filter then plays a crucial role in communications to reduce spectral leakage, ISI and channel width [6]. The present study involved the simulation of WCDMA system at 2Mbps (indoor environment) over AWGN channel. The performance of the system model was analysed with and without channel coding.

1.4 Wireless Communication Pulse Shape Filtering

The increase in demand for mobile radio services has created a series of technological challenges. This is due to the need for power and spectrally efficient modulation schemes to meet the spectral requirements of mobile communications. Linear modulation methods such as Quadrature Amplitude Modulation (QAM), Quadrature Phase Shift Keying (QPSK) and Offset Quadrature Phase Shift Keying (OQPSK) have been found to offer high spectral efficiency. However, for efficient amplification of transmitted signal, the RF amplifier should be driven near the saturation region which exhibits non-linear behaviour [7].

As a result, significant spectral spreading occurs, when a signal with large envelope variations propagates through such an amplifier and creates large envelope fluctuations. Therefore, pulse shaping plays a significant role in spectral shaping in the modern wireless communication to reduce the spectral bandwidth. Pulse-shaping is a spectral processing technique where the fractional out of band power is reduced for low cost, reliable, power and spectrally efficient mobile radio communication systems. Not only does the pulse-shaping filter reduces inter symbol interference, but also reduces adjacent channel interference [8][9].

1.5 Problem Statement

In the recent times, there has been a rising demand for mobile radio services that integrate voice, data and multimedia applications which is made possible by WCDMA technology. This implies that the number of users accessing the network has gone up which give rise to interference among the users referred to as inter-channel interference.

During propagation, the signal follows multiple paths from the transmitter to the receiver, a phenomenon known as multipath fading. At the receiver, the symbols received will be having errors due to inter-symbol interference since symbols arrive at different times at the receiver. There is also noise in the path of propagation of a signal from the transmitter to the receiver and noise due to the motion of electrons in the internal circuitry of the transmitter and receiver known as thermal noise which can be modelled as additive white Gaussian noise.

The increase of data rate in a WCDMA system accounts for the errors generated during signal transmission which calls for increase of the bandwidth as the pulse width of digital symbols decreases. The BER is related to user data rate because the bit error probability depends on the energy per bit (E_b). When the data rate increases, energy per bit increases which requires that the transmission power be increased for reduced BER to be obtained. If the transmission power is not increased due to the high cost of power amplifiers, then the number of errors in the receiver goes up. This calls for incorporation of channel coding in the system to reduce errors at the receiver and at the same time improve power efficiency.

Since the bandwidth in a wireless communication system is fixed, the bits are transmitted in the limited bandwidth with collisions between the bits resulting in distortion of the bit sequence which introduces errors in the bit stream that is received. The collisions of packets occur when two or more nodes are transmitting a packet across the network at the same time. This also occurs when the receiver is within the communication range of two senders that are transmitting simultaneously which make some frames to interfere with each other. In wireless communication, the received power of each packet is dependent on the position of the access terminal and the condition of the communication channel. The packet with the largest power is received and the others are discarded which makes the bits to be received in error. The higher data rate also requires that the transmission power be increased to raise the power per bit for reliable communication in this system.

The WCDMA technology employs Direct Sequence Spread Spectrum (DSSS) technique where the information signal is multiplied by a pseudo-random noise code which gives rise to spread delay when the transmission is serial. This time delay is supposed to be smaller compared to the symbol period in the information signal. This condition cannot be met in high data rate transmissions in WCDMA which makes this time delay to be greater than the information signal symbol period that is transmitted. This spread delay causes inter-symbol interference (ISI) since in WCDMA the signal undergoes serial transmission.

For better transmission quality and delivery of quality services to the consumers there is need to develop systems that meet the spectral requirements of mobile communications with an improved performance at higher data rates and in the presence of these interferences. Therefore, this study is carried out to develop and analyse the WCDMA system at a data rate of 2Mbps subjected to additive white Gaussian noise. Pulse-shaping was done by placing a square root raised cosine filter, which is a pulse shaping filter in the transmitter and receiver side of the system to reduce the ISI which is caused by the delay spread that arises due to the serial transmission experienced in WCDMA.

Since the transmission is done at a higher data rate in WCDMA which requires an increase in the transmission power, the channel coding scheme (convolution coding) is incorporated in the system to improve the power efficiency of the system instead of increasing the transmission power. In addition, the convolution coding scheme compensates for errors in the signal received which improves the overall performance and quality of communication in the system model.

1.6 Objectives

1.6.1 Main Objective

To improve the performance of WCDMA system model that is subjected to AWGN using convolution coding scheme at a data rate of 2Mbps.

1.6.2 Specific Objectives

i. To develop a 3G WCDMA system that incorporates AWGN channel and square root raised cosine filter in the transmission and reception end of the system using its optimized parameters.

ii. To reduce the interference and errors due to the effect of increased data rate and additive white Gaussian noise in a WCDMA system by compensating for the errors encountered in the system using convolution coding scheme.

iii. To increase the rejection loss in the stop band of the filter and decrease the pass band errors by using the optimized values of the pulse shaping filter parameters or coefficients and choosing a parameter that gives better results at 2Mbps.

iv. To validate the achieved performance in a WCDMA system model by comparison with literature review of publications.

The study with a data rate of 384kbps considered the AWGN channel and Rayleigh fading channel which was carried out in [10]. The thermal noise present in all communication systems and is a major source of noise in most systems with characteristics that are additive, white and Gaussian which can be model noise in communication systems. This noise is also found to be dominant in the frequency ranges where mobile services are offered and it limits the performance of communication systems as shown in [11]. That is the reason why the current work considered the AWGN channel for simulation of the system at 2Mbps so that a comparison can be with it and since the majority of the noise found in wireless communication systems can be modelled by this channel. This channel can then be replaced with other channels such as the Rayleigh and Rician fading channels in subsequent studies to take into account the propagation losses in air.

1.7 Organization of the Thesis

The thesis is organized as follows:

- **Chapter 1** gives an overview of wireless communication technologies and their evolution, multipath propagation effects, pulse shape filtering in wireless communication, motivation, objectives and justification of this study
- **Chapter 2** provides an overview and survey of the most important literature in WCDMA. The WCDMA details and system architecture are described and the necessary aspects governing WCDMA based uplink and downlink

transmissions given. The aspects include operation modes, technical specifications and spreading operation, coding and modulation schemes applicable to this system.

- **Chapter 3** gives a detailed description of how the performance of a WCDMA system can be determined by application of signal processing techniques. The processes include spreading, modulation, pulse shape filtering and channel coding in the transmitter and the reverse of these processes occurs in the receiver. In addition, the performance analysis of a wireless communication is discussed and the parameters that are used to determine the performance of a WCDMA system.
- **Chapter 4** provides a detailed design, description and development of the proposed WCDMA system model used for the simulation and performance analysis. The transmitter and receiver parts of the system are modelled in MATLAB 7.8 Simulink which has the library of all the blocks used in digital communication systems. In addition, the MATLAB editor was used to write m-files that were used to determine the error rates and draw the performance graphs.
- **Chapter 5** provides simulation results and their discussion from the simulated WCDMA system model.
- **Chapter 6** concludes the thesis and outlines a few directions for further research in WCDMA technology and the subsequent technologies basing on this analysis.

CHAPTER TWO

LITERATURE REVIEW

This chapter gives an overview of the WCDMA system and the previous research works that have been done in the field of pulse-shape filtering and WCDMA transmitting at different data rates that are relevant to this study.

2.1 Overview of WCDMA System

The growth of wireless communication technologies has been on the rise for the last one decade which has resulted in the increase of subscribers and traffic and applications which require new bandwidths. These applications include gaming, music downloading and video streaming which at the same time demands more capacity on the system. This led to the development of a new technology and spectrum which can provide a solution to these new requirements imposed on the system referred to as WCDMA. This technology was to create a standard for real time multimedia services that can support international roaming and the spectrum allocated around 2GHz. The WCDMA provided more benefits such as efficient spectrum utilisation and variable user data rates. The system performance can be improved by utilizing multipath signals as a diversity scheme [12].

2.1.1 Operation Modes in WCDMA

The WCDMA system uses two access schemes which are Frequency Division Duplex (FDD) and Time Division Duplex (TDD). In FDD the uplink and downlink traffic is separated by placing them at different frequency channels or bands which implies that an operator must have a pair of frequencies allocated to allow running of the network, hence the term paired spectrum.

In TDD only one frequency channel or band is required and uplink and downlink traffic are separated by sending them at different times also known as time slots. The frequency bands allocated for UMTS Terrestrial Radio Access (UTRA) for TDD and

FDD are shown in the Fig. 2.1. In UTRA there is one paired frequency band in the range 1920-1980MHz and 2110-2170 MHz to be used for UTRA FDD and two unpaired bands from 1900-1920MHz and 2010-2025MHz for operation of UTRA TDD [13].

The minimum allocation an operator needs is two paired 5MHz channels one for uplink and one for downlink at a separation of 190MHz. However, to provide comprehensive coverage and services it is recommended that an operator be given three channels. During frequency assignment, the FDD needs two separate 5MHz carrier frequencies for the uplink and downlink respectively. The TDD needs only one 5MHz channel which is time shared between uplink and downlink. Therefore, the operator requires a minimum of three channels to cover the two modes of operation required in WCDMA [14].

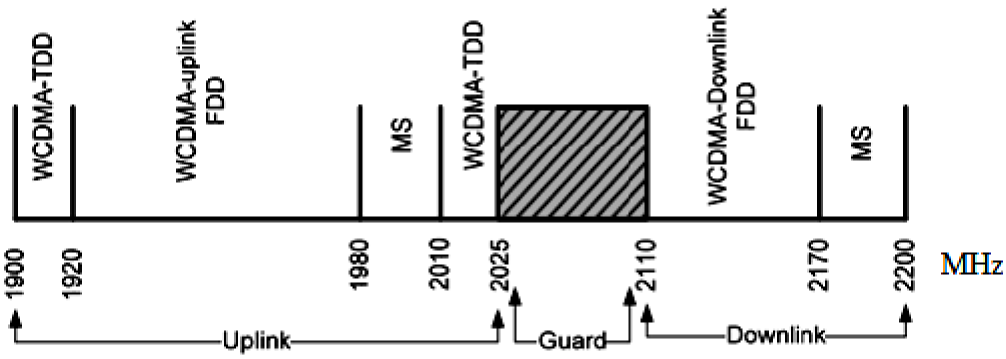

Figure 2.1: The frequency spectrum of WCDMA system

The TDD system which needs only one 5MHz channel to operate is often referred to as unpaired spectrum. The differences between FDD and TDD are only evident in lower layers (physical layer and medium access control layer) particularly on the radio interface between the two layers. At higher layers (radio link protocol) the bulk of the operation of the two systems is the same. The TDD system separates uplink and downlink traffic by placing them in different time slots.

This work deals with only the WCDMA FDD mode which has the parameters shown in Table 2.1.[15]. The major aims of 3G networks that are based on this system is to provide multimedia and high speed data services with data rates up to 2Mbps [6].

A service provider, to optimize the spectrum utilization, can adjust the bandwidth of this system, which is 5MHz and a chip rate of 3.84Mcps. The system frequency spectrum must have a centre frequency which is an integer multiple of 200 kHz. The specifications that apply to the radio transmission and reception in the UMTS FDD mode are summarised below [16] [17].

Table 2-1: Technical Specifications for FDD mode of operation

Specification	**Description**
Multiple Access Scheme	DS-CDMA
Channel Multiplexing in Downlink	Data control channels time multiplexed
Multirate/variable rate scheme	Variable Spreading Factor (SF) and Multicode
Chip rate	3.84Mcps
Channel Bandwidth	5MHz/ 200kHz carrier
Spreading factors	Uplink 4-56, Downlink 4-512
Frame length	10ms
Inter BS synchronization	No accurate synchronization needed
Data modulation	(QPSK) Downlink
Channel coding scheme	Convolutional code (rate ½ and 1/3) and Turbo code
Hand over	Soft and inter-frequency hand over
Power control	Open and fast closed loop (1.5kHz)

This system limits interference by use of the pulse shape filtering and power control mechanism. If this scheme is implemented well in a typical environment it can directly increase the capacity handled by the system.

The WCDMA system has a maximum transmitter power in the range of 125mW to 2W for the User Equipment (UE). The sensitivity of the receiver for minimum receiver input power at the antenna port for BER of 0.001 or less is -117dBm for the UE and -121dBm for a Base Station (BS).

2.1.2 Spreading Operation in WCDMA

The WCDMA spreading process involves two operations, which are channelization and scrambling. In channelization operation, the spreading code is applied to each symbol of the transmitted data which results in an increase of the bandwidth of the data signal. The number of chips per data symbol in this operation is referred to as Spreading Factor (SF).

The scrambling operation involves the application of the scrambling code to an already spreaded signal. The two spreading operations are applied to the In-phase (I) and the quadrature phase (Q) branches of the data signal. The Orthogonal Variable Spreading Factor (OVSF) codes are independently applied to I and Q branches in channelization operation [16], [18].

The long or short codes can be applied for spreading the transmitted sequence. The codes include the Walsh-Hadamard (WH), m-sequences, Gold and Kasami codes. The Walsh codes at zero code delay are orthogonal while the m-sequences, Gold codes and Kasami codes are non-orthogonal with varying cross-sectional properties. During the uplink communication of WCDMA the Gold and Walsh-Hadamard codes are used. The orthogonal variable spreading codes are used to spread the signal and then scrambled by the p-n codes that assist in differentiating different base stations. The scrambling process is done after the signal has been spread.

2.1.3 Channel Coding and Modulation

The WCDMA technical standard for error detection and correction suggests three methods of channel coding for different Quality of Service (QOS) requirements in WCDMA system [19]. The three methods of channel coding include convolution coding, Turbo coding or no coding and their selection is done in the upper layers of the WCDMA system. The convolution coding scheme is used for data rates up to 2Mbps but for higher rates above this rate the turbo coding is applied since it offers better performance above 2Mbps compared to convolution coding. This guided the selection of convolution coding for the simulation of the WCDMA system in the current work. In addition, the BER can be improved further by applying bit interleaving. The 3GPP WCDMA standard has provided for a selection of QPSK modulation scheme for downlink and BPSK for uplink.

2.2 WCDMA Network Architecture

This section provides the details of the WCDMA Radio Access Network (RAN) architecture [14]whose main function is provision of a connection between the handset and the core network and isolation of all the radio issues from the core network. The benefit of a having a core network is to support multiple access technologies.

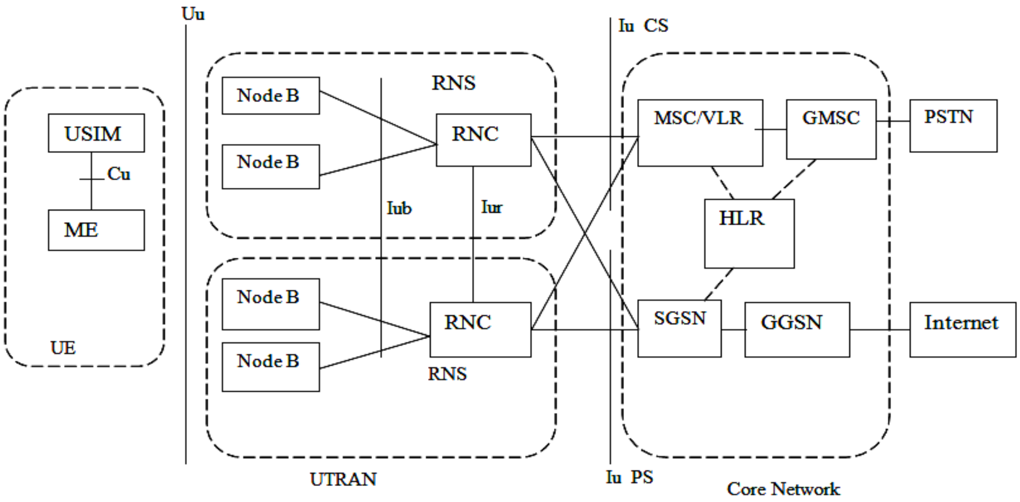

Figure 2.2: Architecture of WCDMA system

The Universal Mobile Telecommunication Service (UMTS) network is made up of three subsystems: Core Network (CN), UMTS Terrestrial Radio Access Network (UTRAN) and User Equipment (UE). These subsystems have various logical network elements which have specific functionalities and they interact with each other using different interfaces. The mobile equipment (mobile phone) and the SIM (Subscriber Identity Module) card which is also referred to as USIM (Universal SIM) form the UE. The mobile equipment is defined as a terminal that contains operating elements for user interface and radio equipment to communicate with the UTRAN over the Uu interface. The USIM contains subscriber data which allows access to the network for network authentication and security keys for data encryption [12].

The RAN consists of the Base Station or Node B and Radio Network Controllers (RNCs). The node B comprises of transceivers, processing modules to offer channel elements which service the users. The Uu interface between node B and the UE is the one that supports data rates up to 2Mbps in WCDMA. The main functions of node Bs include closed loop power control, physical channel coding, modulation/demodulation, air interface transmissions or reception and error handling. The node B is the one that performs crucial measurements on the Uu interface and reports these results to the RNC concerning the link quality. The measurements can be the Block Error Rate (BLER) and the Bit Error Rate (BER) which are required by the RNC for evaluation of the quality of service and adjust power control limits properly [14].

The RNC's main function is to control the node Bs situated within its own Radio Network Subsystem (RNS). It interfaces to the Serving GPRS Support Node (SGSN) when the service required is speech. The RNC is in charge of the load in individual cells in the RNSs and handling of admission control (traffic control) and code allocation. The signalling to the elements, which are higher in the network when a connection has been established between the UE and a node B, is done by the RNC. When it performs this function it is then referred to as the Serving RNC (SRNC). The SRNC does hand over evaluation, outer loop power control, signalling between the UE and the rest of the UTRAN [20].

Drift RNCs (DRNCs) are RNCs which enable reception of the UE signal by other node Bs which are controlled by these other RNCs in soft hand over. The DRNCs processes the uplink signal which originates from the UE and offers macro diversity and transfer of data over the Iub/Iur interface to SRNC [20].

The RNS comprises of one RNC and all connected node Bs. The UTRAN can contain many RNSs where each RNC within the RNS is interconnected over the Iur interface. The Iu interface is between the UTRAN and the core network which is standardized for compatibility of the UTRAN and core network from other manufacturers [14]. The Iur interfaces data from soft handovers between different RNCs and the Iub interface is utilized between the node B and RNC. It thus enables the handling of radio resource management and eliminates the burden from the core network. This subsystem is also standardized for different RNCs to support node Bs from various vendors.

The Core Network (CN) has the same features as any backbone network which performs switching, routing and transit for user traffic and it also contains databases and network management functions. Its architecture is based on the GSM network with GPRS technology where all the equipment has modifications for UMTS operation and services. The CN may be circuit switched (CS) or packet switched (PS) depending on the transport technology applied. The CS and PS domains form divisions of the CN where the circuit switched elements include the Mobile Services Switching Centre (MSC), Visitor Location Register (VLR) and Gateway MSC. The packet switched elements include the SGSN and the Gateway GPRS Support Node (GGSN). The Equipment Identity Register (EIR), Home Location Register (HLR), VLR and the Authentication Center (AUC) are network elements which are shared by both domains [14].

The SGSN and the GGSN supports packet switched services towards the mobile stations including mobility management, access control and control of data control contexts. The GGSN also offers internetworking with several external packet switched networks (internet). The MSC/VLR and the GMSC handle the CS services. The MSC is responsible for the control of circuit switched connections, speech and real time data applications for a UE that is active on the network. The VLR contains a database for all the UEs attached to the network [21]. The GMSC provides a connection between the public switched telephone network and other CS networks. The VLR retrieves relevant data about the user SIM from the HLR associated with the International Mobile Subscriber Identity (IMSI) when the UE registers in the network. The HLR is a central database at the CN which contains master copies of the subscriber's service profiles, roaming areas, authorization information and current location information [14].

The data service for UMTS with Release 99 which was the first data service is also referred to as R99. The theoretical maximum speed of R99 is 384 kbps for a moving vehicle and a minimum data rate of 2 Mbps for stationary or walking users [17] [22].

The detailed analysis of previous studies done on pulse-shape filtering and WCDMA system models is discussed in section 2.3. This is done so that the area of research for the present study can be identified.

2.3 Survey of Related Literature

2.3.1 Pulse Shaping Filters and Modulation Techniques

In [23], matched filters were developed using DSP based modem which offered additional analog filtering. The matched filter designed had demerits of non-uniform gain and mismatch resulting from phase analog filters which led to undesired ISI. This was later on solved by another design technique involving design of digital matched filters having spectral gain and phase compensation. It also equalized known characteristics of the analog filters in the transmission and reception paths.

In [24], differential detection of Offset Quadrature Phase Shift Keying (OQPSK) signals in a mobile radio environment was studied. It involved implementation of two schemes which were Offset Differential Quadrature Phase Shift Keying (ODQPSK) and the constant envelope ODQPSK. The schemes' analysis was done with computer simulation. It was then found that a compact spectrum and a low envelope variation of the modulated signal could be gotten by deploying the SRC-ODQPSK. This was due to the benefits of SRC and CE-ODQPSK of better spectral and envelope properties, the best error performance and simplicity of the receiver configuration. Due to these benefits they were implemented in lower and bandwidth limited mobile radio communications such as Global System for mobile communications (GSM). In addition, these schemes performed well due to small envelope fluctuation, a compact spectrum and better BER performance when compared to ¼ DQPSK and GMSK in hard limited and Rayleigh fading environments.

In [25], a description of the family of pulse shaping filters with ISI free matched and unmatched filter properties was presented. The raised cosine pulse-shaping filter played a significant role in digital communications as they reduce the ISI arising from the multipath signal reflections. From this study, a new family of pulse shaping filters was proposed. When the filter was implemented computational load and hardware cost in demodulation for modern design reduced in applications where low pass (band pass) filtering was done before the matched filtering.

In [26], an alternative approach to pulse shaping filters with ISI free matched in the presence of white noise and unmatched filter properties was proposed. It was found that the Nyquist 1 filter in closed form was ISI free with or without matched filtering. This was possible if its complex transfer function was derived from an initially given Nyquist 1 filter. With this filter there was reduction of the hardware cost that is incurred when a modern system is designed. Furthermore, the wide optimization prospects for data communications systems (transmission) might be extended. It was also found that the data transmission over band limited channels needed pulse shaping technique to eliminate or reduce ISI.

In [27], the method used was able to accommodate two different structures for a receiver. There could be a filter matched to the transmitting filter and the other without a matched filter. A generalization was done for raised cosine filters which were found to offer greater flexibility in filter design. For this design, the rate of asymptotic decay for the filter impulse response would be increased or the residual ISI added by truncation of impulse response might be reduced in return. This improved the quality and as such it became beneficial in the implementation of wireless systems.

In [28], transmission properties of Xia's pulses were studied. It was found that the family of pulses suggested by Xia were ISI-free with and without matched filtering. Their transmission characteristics were comparable to raised cosine pulse that was used for three different receiver situations. The comparison made between the two families of pulses was analyzed with eye diagrams.

The average probability of bit error due to noise was determined and from it the ISI and timing error were computed. From the research of the linear phase family, the raised cosine was then used widely and its pulses were found to contain lower probability of error than Xia pulses family.

In [29], a technique of efficient Finite Impulse Response (FIR) filters which can be implemented based on the power used was presented. These filters were found to consume less power compared to traditional FIR filter implementations in wireless embedded systems. The schemes proposed were superimposed to direct form to achieve a given margin of power consumption. In most wireless mobile systems, FIR filters are indispensable parts in applications such as image or video communication to minimize noise and enhance the specific features. Hence, low power architecture for dedicated linear phase FIR filter was proposed.

In [30], a method of designing a square root raised cosine filter with equiripple characteristics was studied. The filter design proposed had better performance in ripple compared to the square root cosine filter. However, this was done at the expense of small increase of ISI. The performance was similar to the conventional square root raised cosine filter with compatibility feature being shown.

In [7], a pulse-shaping filter applied to wireless communication systems with an improved performance was implemented. This was due to its feature of maintaining the signal level and it offered no distortion onto it. In addition, the bits or group of bits or symbols were transmitted as individual pulses of energy in modern data transmission systems. The study was derived from rectangular pulses whose Fourier transform was a spectral characteristic pulse of width t which had most of its energy on its main lobe. It spanned one sided bandwidth of $\frac{1}{\tau}$ Hz which implied that a transmission bandwidth for the data must be at least $\frac{2}{\tau}$ Hz wide.

In [31], a digital pulse shaping FIR filter design with a reduced ISI and ICI was studied. The primary goal of having data transmission filter designs was to minimize ISI, which was zero if the overall impulse response from the transmission filter to the receiver filter satisfies the first Nyquist criterion. In this regard, a significant class of transfer functions satisfying the Nyquist criterion was found to be raised-cosine filter family. For the realisation of low interference between adjacent channels, transmit and receive filters must have a high value of stop band attenuation and for a reduction on inter-channel interference to be realized as much as possible.

In [32], the design of Square Root Raised Cosine (SRRC) FIR filters by an iterative technique was also studied. The pair of matched square root raised cosine filters for the transmitter and receiver that can theoretically achieve zero ISI in a band limited digital communication system were analysed. In reality ISI cannot be zero when both SRRC filters are approximately implemented because of some numerical precision problems in the design phase as well as in the implementation phase. Hence an iterative method was proposed to design the coefficients of SRRC FIR filter.

2.3.2 WCDMA

It was observed that there was need to satisfy the upward trend in the demand for higher data rates. In addition, the designed system offering these services should have the capability of allowing multiple users to access the network simultaneously. Then research was done in [33] which led to an emergence of a WCDMA system to satisfy the needs of users. An outline of the basic features of WCDMA waveforms that enabled them to accommodate higher data rate transmissions over the wireless channels were developed.

This gave rise to an investigation on how the spread bandwidth chosen affected the BER performance of the system. In addition, a study was carried on the effect it had on the reliability of various subsystems, such as those performing coarse acquisition and adaptive power control. The potential improvements that can be done on WCDMA systems that involved the utilization of interference suppression at receiver or multiple antennas at the transmitter were proposed.

In [34], it was found that the WCDMA technology was a predominant wireless access technology for 3G systems. It was designed to offer flexible wideband services such as Wi-Fi and video transmissions where wideband was all about the data rate. It was also realised that physical limitations and the impairments to the radio channels such as the bandwidth problems, fading due to the multipath nature of the signals, noise and interference posed a difficulty in the achievement of high data rates.

In [35], an investigation on the features of a pulse shaping FIR filter applied in TDD WCDMA transmitter and a comparison between TDD and FDD channel access methods for mobile terminal and base station communication was done. Since the TDD was a new duplex scheme for WCDMA, its time slots were located on the physical layer of the WCDMA and were divided into transmission and reception parts. This made it possible for the reciprocal transmission of information on the uplink and downlink channels.

The capability of TDD was handling up to 16 users per time slot. However, 15 time slots were contained in one WCDMA cell. It was found that the two pulse shaping FIR filters attenuated the adjacent channel from in-phase and quadrature phase components. The sampling rate and the complexity of the filter depended on the interpolation factors. The different filter parameters were used for various interpolation factors. This had an effect on the sampling rate, the filter complexity and its attenuation.

The DSP implementations of 3GPP WCDMA transmissions, signal processing elements and DSP simulation of the equivalent baseband wireless channel. The signal processing elements in the transmission channels included the spectrum spreader, transmission filter and receiver filter. The spectrum spreader and transmission filter formed part of the baseband modulator and the receiver modulator was referred as a matched filter.

In [36], the simulation of WCDMA for 3G mobile systems was done. In the study, a description was given on multiuser effects and BER in 3G mobile systems computed with the help of MATLAB Simulink. The block diagrams in the Simulink library satisfying the characteristics of a digital wireless system were used to develop the model of this system. The BER in CDMA, multiple access and communication systems were defined by use of the Gaussian approximation. The BER was computed by using a comparator which checks the transmitted and received data. The analysis was done based on the multiuser effects and the BER for different data rates.

In [37], the DS-CDMA system was analysed under different roll-off factors by using the rectangular pulse shaping and square root raised cosine pulses. It was then found that the rectangular pulses were better even with close adjacent channels. In addition, the square root raised cosine pulses outperformed the rectangular pulses with synchronization error in the order of 10% of a chip interval. The effect of square root raised cosine chip shaping on the BER of a system with and without adjacent channels for AWGN channels and Rayleigh fading channels were also studied.

In [38], the performance of chaos based asynchronous DS-CDMA with different pulse shapes was studied. The effect of pulse shaping and spreading sequence statistics on this system was also looked at. Due to this, the effect of three possible pulse shapes such as rectangular, band limited rectangular and square root raised cosine associated with three possible spreading sequences such as chaos based and optimum were analysed. It was found that the overall performance was degraded by band limitations.

In [39], the Wideband DS-CDMA for next generation mobile communication systems was presented. The review on wideband wireless access based on DS-CDMA aimed at 3G mobile communication systems was done. The WCDMA designed offered flexibility in the provision of wideband services which cannot be provided by present cellular systems with various data rates as high as 2Mbps.

In [40], an investigation on the BER obtained under different filter types for RF front ends for WCDMA-UMTS mobile radio systems was studied. The majority of base stations used Chebychev or Cauer-Chebychev filters which were analog filters. A comparison was done between the use of Bessel, Butterworth and Chebychev filters in single channel filters for a WCDMA-UMTS radio system. It was found that a significantly lower BER and insertion loss was produced by Bessel filters even though they pass more adjacent channel interference than other filter types. The group delay was an important parameter for WCDMA based communication system.

In [41], it was reported that with increase in demand for faster communication systems, the rate of data transfer over the channel also increased. The channel consisted of interferences and various other parameters which affect the transmitted bits hence resulting in corruption of message bits. From the analysis on a short wireless channel for data communication, it was observed that BER drops down with increase in SNR. In most mobile or cellular systems, message travels through various paths and reach to the receiver unit giving rise to multipath communication. The BER performance of this system for different fading parameters and roll-off factor were then evaluated.

In [42], [43] and [44], studies were carried out on square root raised cosine filter for WCDMA at 5MHz. The effect of variation of roll off factor, group delay and interpolation factor on the SRRCF was also studied. The user data rate for the model simulated was 64 kbps in a WCDMA system with a bandwidth of 5 MHz using different values of group delay. From the research done, the optimum values of the filter parameters were found as follows. At this data rate the optimum value of group delay was D=6, optimum value of interpolation factor M=5 and the optimum value of roll off factor of 0.22.

In [45], a study was done to analyse the performance of WCDMA using different spreading codes. In the study, the BER performance of WCDMA was determined and analysed by considering the various spreading codes with a reasonable Spreading Factor (SF) subjected to multipath channel (Rayleigh channel) and AWGN. A comparison was also done between the codes used for the various SFs. It was then found that the Walsh code produced lesser error compared to other spreading codes as the value of SF is increased. The Walsh code has been recommended as it produces an easy implementation of the hardware module which can be simplified later by employing FPGA design. At the same time the memory required can be reduced by using a Look-Up Table (LUT) cascade for this code generator and develop a faster working system.

In [46], a WCDMA system model was developed in MATLAB 7.3 using the blocks in the Simulink library. The model was transmitting data at a rate of 64kbps over an

AWGN channel. This model was used to analyse the performance of the system at different parameters such as roll off factor, interpolation factor, order of the filter and group delay of the square root raised cosine filter (pulse shape filtering). The BER performance was done for different values of the filter group delay parameter against sinusoidal interference. The user information signal was generated using the Bernoulli binary generator and the spreading sequence was produced by the PN sequence generator at a chip rate of 3.84Mcps. The user information and the spreading sequence were multiplied together by the XOR logic operator which acted as the spreader. The output from the logic operator was encoded and modulated using QPSK modulation scheme.

Then, the signal was up-sampled and passed through the square root raised cosine filter for band-limiting and reduction of inter-symbol interference. The filtered signal was passed through the AWGN channel to the receiver. At the receiver, the signal was filtered again with the square root raised cosine filter, down-sampled, demodulated and finally decoded. After decoding, the received signal was compared with the output of the XOR logic operator to determine the BER at different values of group delay of the filter. In this analysis of the WCDMA system model, there was no channel coding or error correction scheme that was used to improve the performance of the WCDMA system and it only considered a single user.

In [47], a simulation was done on a WCDMA system that was transmitting voice at 12.2kbps, video at 64kbps and 144kbps and 768kbps for other high data rate multimedia services. The spreading rate at this simulation was 3.84Mcps and the bandwidth was 5MHz. The model was used to study the effect of loading, interference and voice activity on the WCDMA system. The capacity enhancement analysis was also done for the different data rate services such as voice and video at a given bit energy to noise power spectral density (E_b/N_o) which in this case was 5dB.

The processing gain was also different for various data services which were varied inversely with the increase of data rate. This meant for higher rate services, the processing gain is smaller compared to lower rate services. The analysis was done by selecting an application; processing gain, E_b/N_o and the factor which affects capacity were varied. In addition, this analysis did not consider any channel coding scheme and pulse shaping of the signal which can improve performance in the provision of those services.

The bit energy to noise spectral density (E_b/N_o) was varied and the performance based on capacity of the network checked at each value of the ratio by varying data rate, loading of the cell, interference (own and neighbouring cells), voice activity factor, antenna diversity and interference cancellation. Apart from interference, majority of the factors that were considered in this study affects directly the capacity of a wireless communication system. Both capacity and BER being performance indicators in a wireless system, the study dealt only with capacity since most of the factors considered affects capacity more that the BER and that is the reason why the BER was not considered in this study.

A WCDMA system model was developed in [48] to analyse its performance using 16-QAM and QPSK modulation schemes through an AWGN and multipath fading channels in MATLAB 7.6. This analysis was done to select a better channel scheme which can suit channel quality for optimum and efficient delivery of services to mobile terminals. The performance analysis was carried out through variation of chip rate of the PN generator and comparison of a given number of users in static and dynamic environment over AWGN and multipath fading channels. The entire analysis was carried by maintaining the data rate at 384kbps without channel coding which can improve the BER performance of the system designed. At the same time there was no use of pulse shaping filter to reduce ISI in the system.

In [10], a WCDMA model was developed to study the performance of convolution channel coding scheme under different channel conditions. This model was developed in MATLAB 6.5.2 using AWGN and multipath Rayleigh fading channels. The simulation of the model only considered a single user. During the simulation, two models were developed where one was transmitting the signal through AWGN channel with and without channel coding and the other subjected to a combination of AWGN and multipath Rayleigh fading channel with and without channel coding. The analysis was done at a data rate of 384kbps and the chip rate was 3.84Mcps for the spreading sequence. This model did not employ the pulse shaping filter to reduce ISI.

Furthermore, a WCDMA model was developed in MATLAB 7.4 using the Simulink library blocks to study its performance. The generated signal was modulated by 16-QAM and QPSK modulation techniques over AWGN and multipath Rayleigh fading channels. The models that were used in this case were transmitting data at 384kbps without error correction coding to improve the performance. The same model was developed in [49] to study the performance of different channels of a WCDMA system using 16-QAM, 8-PSK and QPSK modulation techniques. This model also did not utilize the pulse shaping technique to reduce the ISI in the system. All the studies discussed did not incorporate channel coding for error detection and correction as a way of compensating for the errors to improve the performance of the system.

Therefore, in this thesis a WCDMA system model has been developed that uses a pulse shaping filter (square root raised cosine filter) to reduce the ISI caused by the time delay of spreading the signal over the AWGN at a transmission rate of 2Mbps. The system model was simulated by MATLAB which has Simulink for the development of the model using the communication and signal processing blocks from the Simulink Library. The square root raised cosine filter that was used in the transmission and reception parts of the model had the parameters as; group delay, D, interpolation factor, M, roll off factor, α, order of the filter, N and number of the filter tapings, T. The value of D=6, M=5, α=0.22. The value of N is given by the expression [44];

$$N = D \times M \tag{2.1}$$

The number of filter tapings is then given as [44];

$$T = N + 1 \tag{2.2}$$

Then substituting the value of D and M, N was found to be 30 and T=31.

The WCDMA model developed had the capability of transmitting data in the range of 64kbps to 2Mbps so that comparison can be made with the previous data rates and offer flexibility of operation. During simulation, the errors generated were determined by dividing the total number of bits received in error by the total number of bits that were transmitted from the source at the receiver end for a user data rate of 2Mbps. Then, channel coding technique was incorporated in the system for error detection and correction over AWGN. The modulation scheme used in this WCDMA model was QPSK and the error rate determined at 2Mbps. This modulation scheme was substituted with 16PSK, 16-QAM and 64-QAM so that BERs and power efficiency can be compared.

The system performance was examined due to the variation of modulation format used and implementation of convolution coding by plotting the performance graphs with the help of the MATLAB software functions. When channel coding is included in the WCDMA system model, it minimizes the errors which can be observed by a decrease in the error rates which shows an improvement in the performance of the system.

The second case involved variation of the filter roll-off factor as E_b/N_o ratio is kept constant and BER determined by dividing the number of bits that are received in error by the total number of bits sent from the source. The filter roll-off factor which optimizes the performance is determined from the results obtained with and without convolutional coding. The 16-QAM modulation technique is used in the simulation of varying the filter roll off factor.

The study was carried out on 3G WCDMA scheme because it offers more advantages compared to other previous technologies such as CDMA2000 as it has a higher capacity and increased bandwidth for data transmission. At the same time, it can integrate voice, data and other multimedia applications which can be offered at a very high transmission rate of up to 2Mbps.

Furthermore, the WCDMA operators are trying as much as possible to succeed in delivery of broadband data services and be a market leader in service innovation while increasing their data revenues. This has led to the increase in internet/data service penetration in the recent years with no guarantee of the service quality. At the same time, this technology is cheaper for transmission involving video phone handsets and for a given spectrum the carriers are bound with software upgrade and changing of the channel boards.

This improves the network access bandwidth at a very low cost while maintaining high data access rate which makes it useful in cities with dense population. Basing on these aspects, a study is necessary on WCDMA to improve its performance due to the rising number of subscribers who need integrated services (gaming, music downloading, video streaming, voice calls) at higher rates as presented in [50].

The pulse shaping filter is used in this WCDMA study because unlike other filters it has a characteristic of infinitely adjustable response. It also shows a family of responses instead of a single occurrence of a defined filter. The raised cosine filters family have a modifiable-width flat portion in the pass-band and the half power magnitude point is the same at the cosine inflection [51]. This means its parameters can be varied so that an optimum performance can be obtained in a system where it is applied.

CHAPTER THREE

PERFORMANCE OF WCDMA WIRELESS COMMUNICATION SYSTEM

3.1 Introduction

The performance of a given digital communication system is normally determined by its BER. This parameter can be determined from the receiver end by dividing the number of bit errors that are found in the received bits of a data stream at the receiver over a communication channel and the total number of bits that were transmitted from the source.

This parameter is normally defined in terms of the probability of error which can be determined from the error function, energy in one bit, E_b and the noise power spectral density, N_o. The signal that is transmitted in a digital transmission system undergoes modulation using different modulation techniques. The various modulation techniques used in digital communication systems have their own values of error function because each of them performs differently when it encounters noise in the channel[52],[53], **[**54**]**.

This implies that the higher order modulation schemes, such as 64-QAM, can support higher data rates but are not robust when they encounter noise in the channel. This occurs normally due to the amplitude variations which are associated with QAM modulation technique. Therefore, the modulation formats that are of low order such as BPSK and QPSK offer lower data rates when they are used in digital communication systems since they have a lower value of the modulation order. However, they offer better performance in wireless communication systems as they are more robust and do not have the amplitude variations which are prone to noise [55]. Therefore, the system designer will have to make a choice between the throughput required and the BER performance of the designed system depending on each modulation format used and the filter parameters which optimize the performance of the filter and the wireless system.

The following sections discuss in detail the signal processes that are encountered by a signal in the WCDMA system from generation, transmission and reception. The processes include; spreading/de-spreading, channel coding/decoding, modulation/demodulation and filtering.

3.2 Spreading of the Signal

The frequency spectrum of the data or information signal is spread by uncorrelated code with that signal which results in a bandwidth occupancy that is much higher than the required one. Spreading codes are unique to every user with low cross-correlation values. Therefore, the receiver should have the capability of identifying the code of the intended transmitter to select the desired signal. The transmission bandwidth B_t and information bandwidth B_i can be illustrated as in Fig. 3.1. It can be observed that the transmission bandwidth is larger than the information bandwidth as a result of spreading of a signal.

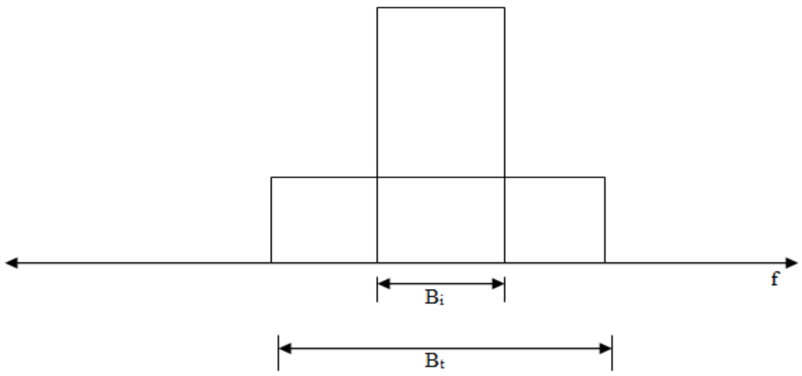

Figure 3.1: Spreading of an information signal

The processing gain (G_p) or spreading factor (SF) is given by the ratio of transmission bandwidth B_t to information bandwidth B_i [44].

$$SF = \frac{B_t}{B_i} \quad (3.1)$$

Substituting the value of B_t and B_i which are approximated to chip rate and data rate respectively, then the processing gain, G_p, can be determined in linear form as shown in Equation 3.2 or dB as expressed in Equation 3.3 [56];

$$G_p = \frac{chip\ rate}{User\ data\ rate} = \frac{3.84Mcps}{User\ data\ rate} \tag{3.2}$$

$$G_p = 10\log\frac{3.84Mcps}{User\ data\ rate}(dB) \tag{3.3}$$

In WCDMA, the number of users allowed in a system, the amount of multipath effect reduction and difficulty to jam or detect a signal are determined by the spreading factor. It is required that the processing gain be as high as possible for quality transmission of information. For an AWGN channel, the effect of jamming and multipath interference is minimal and as such the data rate can be increased and the system operates reliably. However, in Rayleigh and Rician fading channels where multipath effects are high, when data rate is increased interleaving should be implemented in the transmitter and receiver respectively to mitigate the effect of jamming and multipath interference when operating at a lower processing gain. There are different techniques which are used to spread a signal and they include;

i. Direct Sequence (DS);
ii. Frequency Hopping (FH);
iii. Time Hopping (TH);
iv. Hybrid spread spectrum.

The current study uses direct sequence spread spectrum applicable to the WCDMA technology for easier comparison with previous studies in [57], [44] and [42]. The spreading of a signal in WCDMA is carried out by the Pseudo-Random Noise (PN) sequence whose frequency is much higher compared to that of the information signal. This sequence is noise-like and periodic over one window. The sequence is generated by an *m*-bit length serial in parallel out shift registers [58].

The outputs of all the flip-flops of the shift register are fed to one logic circuit with one switch to each. The logic circuit output is then fed back to the input of the primary (left-most) flip-flop. When the flip-flops are arranged in this manner, the arrangement is referred to as feedback shift register which can easily generate the PN sequence as in Fig. 3.2.

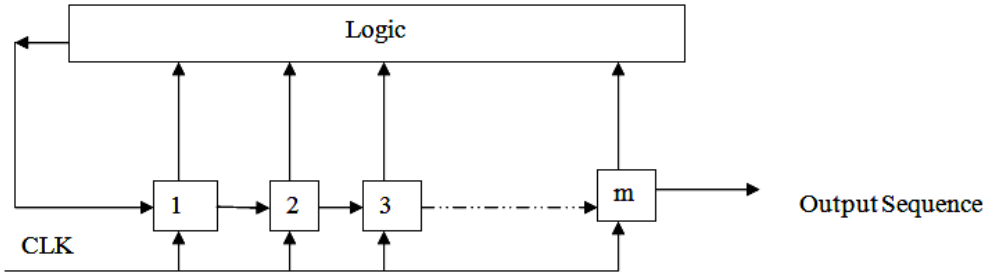

Figure 3.2: Feedback shift register

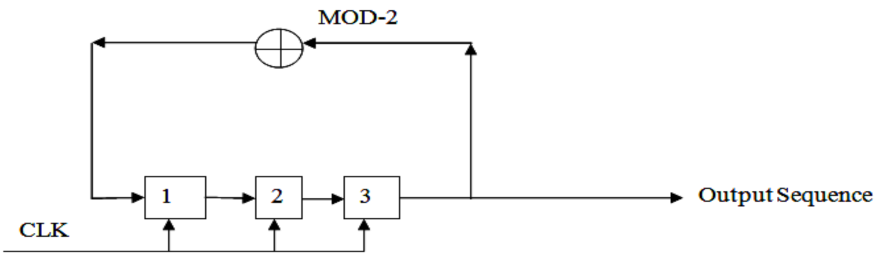

Figure 3.3: A PN sequence generator with a length of 3

The MOD-N means periodic with periodicity N. Therefore, MOD-2 operation generates a window of sequence of length 2. The bits 0 and 1 are only used as a single bit to represent 2 patterns.

Table 3-1: Truth Table of a MOD-2 Operation

A	B	$A \oplus B$
0	0	0
0	1	1
1	0	1
1	1	0

It can be seen from the truth table, in Table 3.1, that the operation follows the XOR gate truth table which implies that it can be realised practically by the gate.

The properties of the PN sequence are as follows;

i. In each period, number of 1's = number of 0's +1

ii. If the number of flip flop is m, the sequence length N must be $N = 2^m - 1$

iii. At an instant, the square of the PN sequence $c(t)$ is 1 i.e. $c^2(t) = 1$

iv. The autocorrelation is given as;

$$R_c(\tau) = \frac{1}{T_c}\int_{-\frac{T_c}{2}}^{\frac{T_c}{2}} c(t)c(t-\tau)dt \quad (3.4)$$

where T_c is the chip duration of the PN sequence and $T_c \ll T_b$ (bit duration)

In direct sequence spread spectrum, the information signal which has a bit duration of T_b is multiplied with a Pseudo-Random Noise Code (PN code) which has a chip duration T_c where $T_c \ll T_b$. This study takes the chip rate as 3.84Mcps which gives the chip duration T_c as 0.2604167μs and the bit duration depends on the data rate. The PN code which is a sequence of chips are valued -1 and 1 for polar or 0 and 1 for non-polar and with noise like properties. The process can be demonstrated as shown in Fig. 3.4.

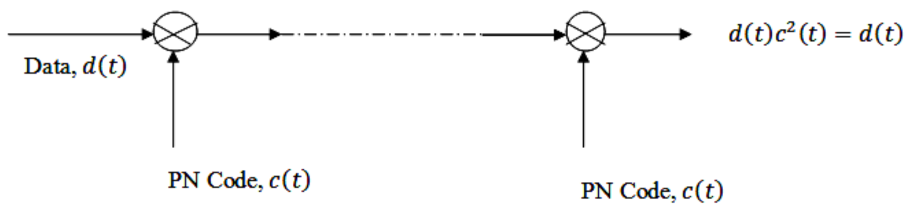

Figure 3.4: Spreading and de-spreading of an information signal

The information signal d(t) is multiplied by the high frequency chip sequence c(t) when the signal is spread in the transmitter. When the signal is again multiplied by the sequence c(t), the output of the receiver would be given as;

$$d(t)c^2(t) = d(t) \quad (3.5)$$

Since $c^2(t) = 1$

This implies the bit sequence is chopped by the chips. Since the bandwidth of the data signal is multiplied by processing gain, the power spectral density lowers but the power contents remains the same. At the receiver, the received signal is multiplied by the same synchronized PN code. This operation completely removes the code from the signal and the original data is left as the code existed as +1's and -1's. The consequence is that a possible jamming signal in the radio channel will be spread before data detection is performed which reduces the jamming effect.

A PN code sequence has chips which are valued ±1 for polar, 0 and 1 for non-polar which have the same noise-like properties. Therefore, Equation 3.5 holds for non-polar signal sequence but it is not applied in most communication systems due to its inability to include clock information especially when the bit stream consists of along sequence of 0's. In addition, the average of a non-polar is not zero which creates a significant d.c component at the receiver which is undesirable for long distance communications [59] [56].

3.2 Performance of Direct Sequence Spread Spectrum

The performance of a wireless communication system employing direct sequence spread spectrum can be determined for any number of users accessing the wireless network. Consider a case where there are *K* simultaneous users and each user has a unique PN sequence with *N* chips per message i.e. $T_b = NT_c$. Taking $m_k(t)$ as the information or data sequence, $p_k(t)$ as PN sequence and φ_k as the phase of the k^{th} user, the signal transmitted for the k^{th} user can be given by [60];

$$s_k(t) = \sqrt{\frac{2E_b}{T}}\, m_k(t)p_k(t)\, cos(2\pi f_c\, t + \varphi_k) \qquad (3.6)$$

where f_c is the carrier frequency and *T* is the period of the information signal

The decision variable for the ith bit of user 1 can be determined after the receiver has de-correlated the received signal with the PN sequence as;

$$z_i^{(1)} = \int_{(i-1)T_b+\tau_1}^{iT_b+\tau_1} r(t)p_1(t-\tau_1)\, cos[2\pi f_c(t-\tau_1) + \varphi_1]\, dt \quad (3.7)$$

where *r(t)* is the received signal

Assuming the value of τ_1=0 and φ=0, the decision variable will then be given by;

$$z_i^{(1)} = I_1 + \sum_{k=2}^{K} I_k + \xi \tag{3.8}$$

where $I_1$the component of the desired user 1 and ξ is the Gaussian noise with a zero mean. The individual variables in $z_i^{(1)}$ are determined as;

$$I_1 = \int_0^{T_b} s_1(t)p_1(t)\cos 2\pi f_c\, tdt = \sqrt{\frac{E_bT_b}{2}} \quad when\, m_k = 1 \tag{3.9}$$

$$I_k = \int_0^{T_b} s_k(t-\tau_k)\, p_1(t)\, cos\, 2\pi f_c\, t\, dt \tag{3.10}$$

$$\xi = \int_0^{T_b} n(t)p_1(t)\, cos\, 2\pi f_c\, t\, dt \tag{3.11}$$

The variance of the Gaussian random variable ξ which has a mean of zero is given by;

$$\sigma_\xi^2 = E[\xi^2] = \frac{N_oT_b}{4} \tag{3.12}$$

Assuming that the signal of each user undergoes transmission with equal power by perfect power control in the system and each user is having random signature sequence, the variance of interference that is coming from the k^{th} user by taking $T_b = NT_c$is given as;

$$\sigma_{I_k}^2 = \frac{E_bT_c}{6} = \frac{E_bT_b}{6N} \tag{3.14}$$

When K is large and the variable I_k $(k = 2,3,4 \ldots .K)$ is taken as an independent variable, the central limit theorem can be used to approximate $\sum_{k=2}^{K} I_k$ to a Gaussian random variable with a variance $\sum_{k=2}^{K} \sigma_{I_k}^2$. The probability of bit error in this system can be determined by;

$$p_e = Q\left[\frac{\sqrt{E_bT_b/2}}{\sum_{k=2}^{K}\sigma_{I_k}^2 + \sigma_\xi^2}\right] \tag{3.15a}$$

$$= Q\left[\frac{\sqrt{E_bT_b/2}}{(K-1)\frac{E_bT_b}{6N} + \frac{N_oT_b}{4}}\right] \tag{3.15b}$$

$$= Q\left[\frac{1}{\frac{K-1}{3N} + \frac{N_o}{2E_b}}\right] \tag{3.15c}$$

Where $Q[x]$ is a Gaussian function which is the complement of the cumulative density function corresponding to a normalized random variable x. It is used to determine the bit error probability in digital communication systems.

The system modelled in this case will have only a single user ($K=1$), then the probability of error becomes;

$$p_e = Q\left(\sqrt{\frac{2E_b}{N_o}}\right) \tag{3.16}$$

The Q-function is monotonically decreasing and the value of it decreases when the value in brackets increases i.e. it tends to zero at positive infinity and 1 at negative infinity [61] as shown in Appendix 2. Therefore, to ensure that there is no error degradation in the error performance the signal-to-noise ratio (E_b/N_o) must be increased.

3.3 Channel Coding in Wireless Communication

The error correction coding or channel coding involves the transmission of additional redundant bits in the stream of information bits which allows detection and correction of some symbol errors at the receiver. This makes the information transmitted through communication systems to be safe, more efficient and improved performance which results in faster transmission and more efficient use of bandwidth when coding is applied.

The application of channel coding improves the error performance of communication systems at the expense of reduction in bandwidth efficiency due to the transmission of additional bits. These error correcting codes add redundancy into bits to improve the performance of a system and are widely adopted in almost all digital systems to provide a means of dealing with the unknown noise [62].

The error correction codes encode data in such a way that a decoder can identify and correct certain errors in the data. When the error correction coding is incorporated in a digital communication system, the performance that approaches Shannon channel capacity is achieved. The Shannon capacity theories established the fundamental limits in transmission speeds in digital communications systems which led to the search for coding techniques for approaching the limit of this capacity.

When digital information is stored in a memory, it is important to have a mechanism that can detect and correct a certain number of errors. The information strings are usually encoded by adding a number of redundant bits to them. The decoder examines the encoded message to determine if there are any errors when the original data is reconstructed.

The digital communication systems that are implemented to transmit or store digital data employ an error correction technique. Therefore, the system designers must choose a more elaborate error detection and correction method. The selection depends on the accuracy, speed, and latency requirement of the information, and whether or not there is simultaneous bidirectional communications between the information sender and recipient.

There are different error-correcting codes and their combinations which are used for particular applications and their choice is based on the physical channel characteristics (signal to noise ratio, presence of memory, fading), modulation type and the desirable channel characteristics such as information rate and BER. From the 3GPP specifications of WCDMA technology, convolutional coding is the one applied for error correction and detection in most applications under this technology.

In convolutional coding, coding and decoding can take place on a continuous data bit stream [63]. The scheme works well in those applications which require good performance with low implementation cost. The code operates on a data stream instead of a static block. They are usually denoted by (n, k, l) where n is the number of output bits, k is the number of input bits and l is the code memory depth. The data message and a given number of previously encoded messages determine the code words produced. After each message has been processed, the encoder changes state with the length of the code word remaining constant.

These codes are applied in most applications as it can be implemented easily compared to block codes and they convert the entire data stream into one single codeword. The major decoding strategy for these codes is based on Viterbi algorithm [64], [65]. The encoding process involves the division of an infinite data stream into blocks and applying the code on each individual block. The output gotten depends on the corresponding block that was sent from in the input data stream.

The stream is again divided into blocks although the blocks are much smaller. The final output for convolution codes depends on multiple blocks from the input stream. There are three parameters which define the convolution code [66]:

i. **Code Rate (R):** This is the ratio of the number of input bits to the number of output bits and is determined by the input rate *k* and output rate *n* as:

$$R = \frac{k}{n} < 1 \tag{3.17}$$

ii. **Constraint length:** This is the number of symbols in the data stream that have an effect on the output of the encoder. The constraint length k for a convolution code is defined as:

$$k = m + 1 \tag{3.18}$$

The value of m is the maximum number of stages (memory size) in any shift register. The shift registers store the state information of the convolution encoder and the constraint length relates the number of bits upon which the output depends.

iii. **Generator polynomial:** it is usually a polynomial which is used in the wiring of the input sequence with the delay elements to form the output.

The error-correction scheme is applied after the source coding and before the information signal is modulated. The coding protects the information against channel impairments such as noise, fading, interference. The important characteristic of convolutional coding which makes it different from block code is that the encoder has memory where the output from the convolutional encoder is not only a function of an input *k*, but is also a function of the previous k - *1* input. The simulation of a WCDMA system transmitting at 2Mbps was done using the code rate as ½, constraint length as 7 which gave the number of stages of the shift register as 6. The code generator polynomial was taken as 171 and 133 in octal numbering system.

3.4 Digital Modulation Techniques

The digital modulation techniques are used to transform signals into waveforms that are compatible with the nature of the communications channel. The modulation type uses a constant amplitude carrier with the information being carried in the phase (phase shift keying) of carrier amplitude variation (amplitude shift keying). The past few years has seen major transitions from the simple AM and FM to digital techniques such as QPSK, FSK, MSK and QAM [67]. The designers of wireless communication systems have the responsibility of ensuring there is good bandwidth efficiency with a low BER. These modulation techniques have the potential of having greater capacity to deliver large amounts of information than the analogue schemes [68].

A desirable modulation scheme provides low bit error rates at low received signal-to-noise ratios, performs well in multipath and fading conditions, occupies a minimum bandwidth and easy and cost effective to implement. The existing digital modulation techniques do not simultaneously satisfy all of these requirements, trade-offs are made when selecting a digital modulation scheme depending on the demand of the particular application.

The performance of a digital modulation scheme is measured in terms of its power efficiency and bandwidth efficiency. The power efficiency or energy efficiency, η_p, of a modulation scheme is a measure of the trade-off between bit error probability and signal power (or energy) and is defined as the ratio of the signal energy per bit to noise power spectral density (E_b/N_o) required at the modulator input to achieve a certain probability of error over an AWGN channel.

The bandwidth efficiency, η_B of a modulation scheme is a measure of the ability to accommodate data with a limited bandwidth and is often defined as the ratio of the throughput data rate per Hertz in a given bandwidth.

$$\eta_B = \frac{R_b}{B} bps/Hz \tag{3.19}$$

where R_b is the data rate in bits per second and B is the bandwidth occupied by the modulated RF signal. However, the bandwidth of the central main lobe is given as;

$$B_{main\ central\ lobe} = \frac{2}{T_s} = \frac{2R_b}{log_2 M} \tag{3.20}$$

Therefore, the bandwidth efficiency is given as;

$$\eta_B = \frac{Data\ Rate}{B_{main\ central\ lobe}} = \frac{R_b}{{2R_b}/{log_2 M}} \tag{3.21a}$$

$$\eta_B = \frac{log_2 M}{2}\ bps/Hz \tag{3.21b}$$

The system capacity of a digital modulation system is directly related to the modulation scheme. From the Shannon's channel coding theorem the maximum possible data rate (referred to as channel capacity) is limited by noise in the channel for an arbitrary small probability of error for AWGN channel. This theorem gives the maximum achievable bandwidth efficiency when it is upper bounded as;

$$\eta_{B(max)} = \frac{C}{B} = log_2\ (1 + \frac{S}{N}) \tag{3.22}$$

Where C is the channel capacity in bps and S/N is the signal to noise power ratio.

Besides power efficiency and bandwidth efficiency there are other factors which also affect the choice of a digital modulation scheme for a wireless system. A modulation scheme which is simple to detect is preferred to maximize the cost and complexity of the subscriber receiver. A modulation scheme is required to give a good performance under various types of channel impairments such as Rayleigh and Ricean fading and multipath time dispersion given a particular demodulator implementation.

3.4.1 M-ary Phase Shift Keying (M-PSK)

In this modulation scheme, the information is in the phase changes. The motivation behind M-PSK is to increase the bandwidth efficiency of the PSK modulation scheme. In BPSK, a data bit is represented by a symbol. In M-PSK, $k = log_2 M$ data bits are represented by a symbol, thus the bandwidth efficiency is increased to k

times. Among the M-PSK schemes, QPSK is the most often used scheme since it does not suffer from BER degradation while the bandwidth efficiency is increased. Other M-PSK schemes increase the bandwidth efficiency at the expense of the BER performance. The carrier phase in M-PSK modulation takes one of M possible values given by;

$$\theta_i = \frac{2(i-1)}{M}\pi \quad i = 1,2,3, \dots \dots . M \tag{3.23}$$

The M-PSK signal is represented by the equation [54];

$$s_{MPSK}(t) = \sqrt{\frac{2E_s}{T_s}} cos\left[2\pi f_c t + \frac{2\pi}{M}(i-1)\right] \quad 0 \le t \le T_s \tag{3.24}$$

where E_s is the symbol rate, f_c the carrier frequency and T_s the symbol period Using the trigonometric identities;

$$s_{MPSK}(t) = \sqrt{\frac{2E_s}{T_s}} \left[cos(i-1)\frac{2\pi}{M} cos\, 2\pi f_c t \; - sin\,(i-1)\frac{2\pi}{M} sin 2\pi f_c t\right] \tag{3.25}$$

Taking $\emptyset_1(t)$ and $\emptyset_2(t)$ as the basis functions which are given as;

$$\emptyset_1(t) = \sqrt{\frac{2}{T_s}} \cos 2\pi f_c t \quad \text{and} \qquad \emptyset_2(t) = \sqrt{\frac{2}{T_s}} sin\, 2\pi f_c t \tag{3.26}$$

Then, M-PSK signal equation can be reduced to;

$$s_{MPSK}(t) = \sqrt{E_s}\, \left[cos(i-1)\frac{2\pi}{M}\, \emptyset_1(t) - sin\,(i-1)\frac{2\pi}{M} \emptyset_2(t)\right] \tag{3.27}$$

The modulation scheme QPSK is a special case of M-PSK when the value of M=4 and *k*=2 and since there are four symbols for two bits, then $E_s = 2E_b$ and $T_s = 2T_b$. Using Gray coding scheme, the Symbol Error Rate (SER) can be minimized. From Equation 3.27, the envelope is constant when no pulse shaping is employed with the phase varying. The signal can be represented by equally spaced message points on a circle of radius $\sqrt{E_s}$.

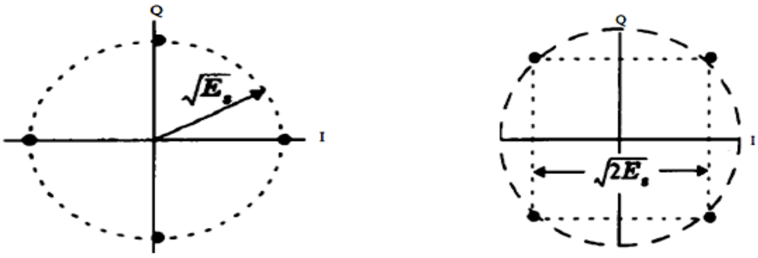

Figure 3.5: Circular and rectangular QPSK constellation

Gray coding is usually used in signal assignment in M-PSK to make only one bit difference to two adjacent signals. In gray coded QPSK, the information transmitted is carried in the phase of the carrier and occupies any of the four equally spaced phases of *π/4*, *3π/4*, *5π/4* and *7π/4* in gray coding. This modulation scheme is immune from noise that creates amplitude distortions since there is no amplitude variation. The equation for QPSK with gray coding [69] is given as;

$$s_{qpsk}(t) = \sqrt{E_s}cos[(2i-1)\frac{\pi}{4}\emptyset_1(t) - \sqrt{E_s}sin(2i-1)\frac{\pi}{4})\emptyset_2(t)]1 \leq i \leq 4 \quad (3.28)$$

The bit error probability for M-PSK modulation over an AWGN channel can be expressed as;

$$p_b = \frac{1}{m}\sqrt{\frac{mE_b}{N_o}}sin(\frac{\pi}{M}), M = 2^m, m = 2,3,4.. \quad (3.29)$$

Since the M-ary modulation scheme used in this case is 16-PSK the values of M and m are taken as 16 and 4 respectively. If the value of m is taken as 2, the error rate obtained is that of QPSK which resembles that of BPSK but utilizes half of channel bandwidth. The bit error probability in QPSK modulation for Gray Coding is given as;

$$p_e = \frac{1}{2}erfc\left(\sqrt{\frac{E_b}{N_o}}\right) \quad (3.30)$$

The bit error probability for QPSK is the same as BPSK. Therefore, the error rate can be simulated by varying the ratio E_b/N_o at 2Mbps data transmission rate. From Equation 3.16, the error probability for a direct sequence spread spectrum system resembles the error probability for QPSK and the BPSK modulation schemes which are seen to have the same error probability.

3.4.2 M-ary Quadrature Amplitude Modulation (M-QAM)

The general form of M-ary QAM is defined by the transmitted signal whose equation [48] is given by;

$$s_{MQAM}(t) = \sqrt{\frac{2E_{min}}{T_s}}\left[a_i \cos 2\pi f_c t \; - b_i \sin 2\pi f_c t\right] \;\; 0 \leq t \leq T_s \quad (3.31)$$

Where E_{min} is the energy of the signal with the lowest amplitude, a_i and b_i are a pair of independent integers selected according to location of a particular signal point.

The QAM is a modulation technique where the amplitude varies with respect to the phase. This signalling method can be seen as a combination of amplitude shift keying and phase shift keying. The QAM modulation scheme takes the value of *M*=4 and *L*=2 which is used in digital communication systems primarily because of its spectral efficiency. The rectangular signal constellation as plotted in MATLAB after modulation for 16-QAM is given in Fig. 3.6 and that of 64-QAM given in Fig. 3.7 using the codes shown in Appendix 3.

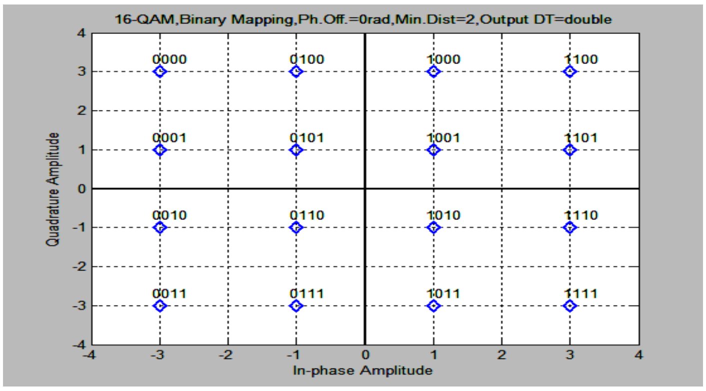

Figure 3.6: 16-QAM gray coded rectangular constellation

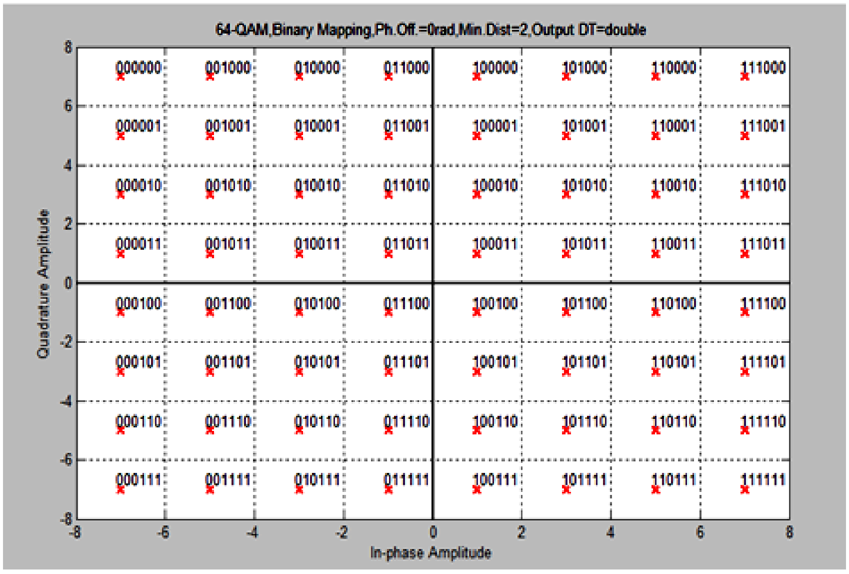

Figure 3.7: 64-QAM rectangular constellation

Since the in-phase and quadrature phase components are independent signals, the probability of correct symbol detection p_c can be determined as;

$$p_c = (1 - p_e')^2 \qquad (3.32)$$

Where p_e' is the symbol error probability for one of the components which is given as;

$$p_e' = \left(1 - \frac{1}{L}\right) erfc\left(\sqrt{\frac{E_{min}}{N_o}}\right) \qquad (3.33)$$

Therefore, the probability of symbol error for QAM is;

$$p_e = 1 - p_c = 1 - (1 - p_e')^2 \approx 2p_e' \quad (3.34a)$$

$$p_e = 2\left(1 - \frac{1}{\sqrt{M}}\right) erfc\left(\sqrt{\frac{E_{min}}{N_o}}\right) \qquad (3.34b)$$

But $E_{min} = \frac{3E_{av}}{2(m-1)}$ which gives;

$$p_e = 2\left(1 - \frac{1}{\sqrt{M}}\right) erfc\left(\sqrt{\frac{3E_{av}}{2(m-1)N_o}}\right) \qquad (3.35)$$

3.4 Pulse Shape Filtering

An efficient pulse shaping is required in communication systems to fulfil two requirements in a wireless communication channel. The need to generate band-limited channels and reduction of inter-symbol interference (ISI) which results from multipath signal reflections during propagation from the transmitter to the receiver are the two requirements that need to be satisfied by the pulse shaping filter. The pulse shaping filter is therefore applied to each symbol of the information signal to achieve both requirements. The sinc pulse meets both conditions as it uses the frequency domain to make a signal utilize the smaller portion of the frequency domain due to the windowing effect it has on each symbol period of a modulated information signal.

The information transmitted through a WCDMA wireless system at higher data rates undergoes a higher delay spread making the symbols to interfere. This causes the Inter-Symbol Interference (ISI) which is supposed to be reduced by the pulse shaping filter. This condition is unavoidable in wireless communication systems. The leakage of one symbol to the others due to high data rate is what makes the energy to be confined. The ISI can be reduced by slowing down the signal transmission with the introduction of a delay between multiple bits. The pulse shaping filter (square root raised cosine filter) is implemented to offer this delay in WCDMA and next generation networks.

For this system model the square root raised cosine filter is used in the transmitter and receiver section so that the overall response of the system resembles that a normal raised cosine filter. The impulse or time domain response of the raised cosine filter and the square root raised cosine filter in [70] are given by the equations;

$$h_{RC}(t) = \frac{Sin\,(\frac{\pi t}{T})Cos\,(\frac{\pi \alpha t}{T})}{\frac{\pi t}{T}\left(1-(\frac{2\alpha t}{T})^2\right)} \tag{3.36}$$

This expression can be simplified further by introducing the sinc function which is given as *sinc x*= $\frac{sin\,x}{x}$ which reduces Equation 3.36 to;

$$h_{RC}(t) = sinc\ (\frac{\pi t}{T})\ \frac{Cos(\frac{\pi \alpha t}{T})}{1-(\frac{2\alpha t}{T})^2} \qquad (3.37)$$

The sinc function in the response of the filter ensures that the signal is band-limited. The time domain or impulse response of the square root raised cosine filter is given as;

$$h_{RRC}(t) = \frac{\sin[\pi(1-\alpha)t]+4\alpha(\frac{t}{T})\cos[\pi(1+\alpha)\frac{t}{T}]}{\frac{\pi t}{T}[1-(\frac{4\alpha t}{T})^2]} \qquad (3.38)$$

In the development of the WCDMA system model, the square root raised cosine filter is used in the transmitter and receiver section so that the overall response of the system reduces to that of the raised cosine filter such that;

$$h_{RC}(t) = h_{RRC}(t)h_{RRC}(t) = h_{RRC}^2(t) \qquad (3.39)$$

The filter was designed in MATLAB and the visualization diagrams generated by the filter visualization tool (fv*tool)* functions. The magnitude and the phase responses of the square root raised cosine filter are shown in Fig. 3.8 for a roll-off factor α of 0.22 and group delay D of 5. The Fig. 3.9 shows the impulse response of the square root raised cosine filter and the two figures were generated by the following MATLAB functions.

```
%Definition of the filter variables
filtOrder = 80; overSamp = 8;
delay = filtOrder/(overSamp*2);
rollOff = 0.22;
```

The Communications Toolbox function `rcosine` is used to design the filter and the Signal Processing Toolbox function `fvtool` used to display the impulse response of the filter. In addition, the `fvtool` is used to display the magnitude response, phase response and other characteristics of a filter.

```
% design of the filter
rrcFilter = rcosine(1,overSamp,'fir/sqrt',rollOff,delay);
% the fvtool is used to display impulse response of the filter used on the transmitter
```

hFV = fvtool(rrcFilter,1,'Analysis','Impulse');

xlabel ('Samples');

set (gcf, 'Color', 'w')

This *fvtool* function displays the impulse response of the square root raised cosine filter shown in Fig. 3.9. The magnitude and phase response of the filter with a roll off factor 0f 0.2 and group delay of 5 is shown in Fig. 3.8. The Fig. 3.8 is displayed by clicking on the icon indicated on the filter visualization tool window of Fig. 3.10.

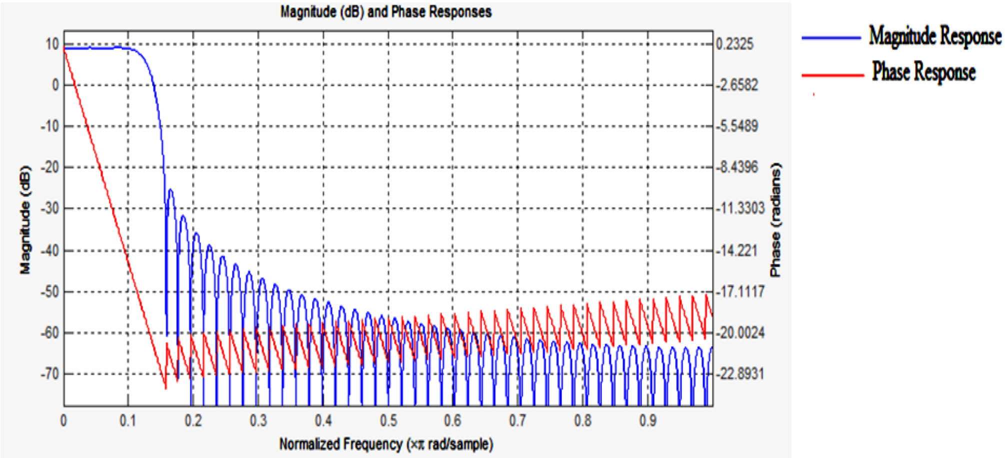

Figure 3.8: Magnitude and phase response of the filter

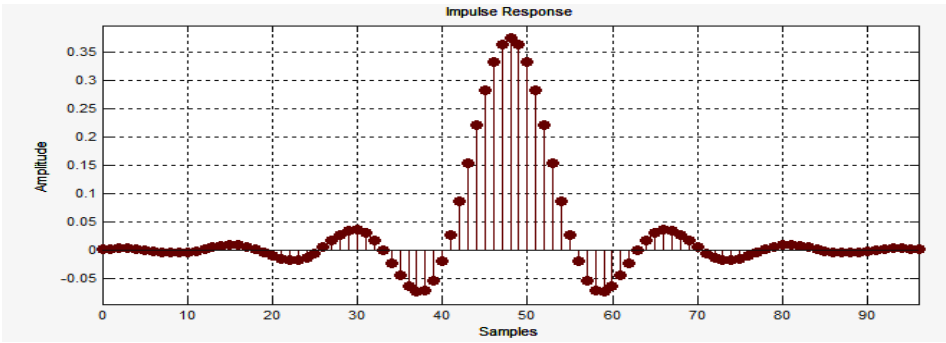

Figure 3.9: The impulse responses of the square root raised cosine filter

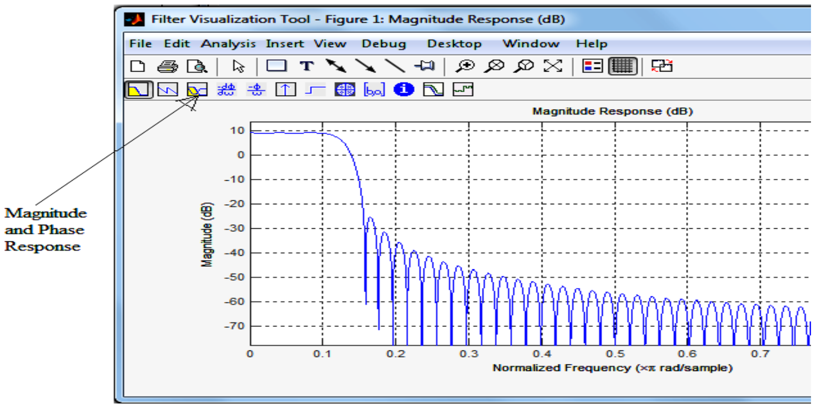

Figure 3.10: The icon used for displaying the magnitude and phase response

The Figures 3.8 and 3.10 are the expected curves when the filter response is plotted before filtering of the signal is done. They are used to check whether the response of the filter that performs the signal filtering is maintained throughout the simulation of the system. The Figure 3.10 is used to show an icon that can be used to display the magnitude and phase response on the same curve after one of the response curve has been plotted as shown in Figure 3.8.

3.5 Transmission Channel

The transmission channel used for this WCDMA system is additive white Gaussian noise channel. In this channel the additive white Gaussian noise is added to the signal that is being transmitted. The noise is generated through the thermal motion of the electrons in all dissipative electrical elements. It is modelled with zero-mean Gaussian random process where the random signal is the summation of the random noise variable (*n*) and a d.c. signal (*a*) given as [56], [71] and [52];

$$z = a + n \tag{3.40}$$

The probability distribution function for this Gaussian noise can be represented as;

$$p(z) = \frac{1}{\sigma\sqrt{2\pi}} \exp[-\frac{1}{2}(\frac{z-a}{\sigma})^2 \tag{3.41}$$

The model of this noise assumes a power spectral density $G_n(f)$ which is flat for all the frequencies denoted as;

$$G_n(f) = \frac{N_o}{2} \tag{3.42}$$

The factor 2 in the response function indicates that the power spectral density is a two-sided spectrum. This type of noise is present in all communication systems and is the major noise source for most systems with characteristics of additive, white and Gaussian. It is mostly used to model noise in communication systems which are simulated to determine their performance. The AWGN channel is used for the simulation of the system in this work of which the channel can be replaced with other channels such as the Rayleigh and Rician fading channels.

CHAPTER FOUR

PROPOSED DESIGN OF WCDMA SYSTEM MODEL

4.1 Introduction

This chapter describes the detailed method of modelling the transmission and receiving part of the WCDMA system in accordance with the specifications developed by 3GPP in MATLAB simulation software. The basic communication system consists of a source of information, encoder, modulator, transmission channel, demodulator, decoder and destination of information. There are different communication systems that transmit data from source to destination at different data rates and different channel conditions.

4.2 Development of WCDMA System Model

The current study develops a WCDMA system model that can transmit data at 2Mbps over an AWGN channel in MATLAB 7.8 simulation software. The information signal at this rate is spread by the pseudo-random noise signal which is generated at a chip rate of 3.84Mcps. The Simulink Library which has communication and signal processing blocks are used to implement the block diagram of the model in Figure 4.1.

The model should be capable of transmitting data in the range of 64kbps-2Mbps which are used for voice, data and multimedia applications. The convolution channel coding scheme is implemented at a data rate of 2Mbps to perform error detection and correction. The model uses QPSK modulation technique and pulse shaping filter. The pulse shaping filter used is the square root raised cosine filter with the parameters group delay, D=5, roll-off factor $\alpha = 0.22$, oversampling factor=4.

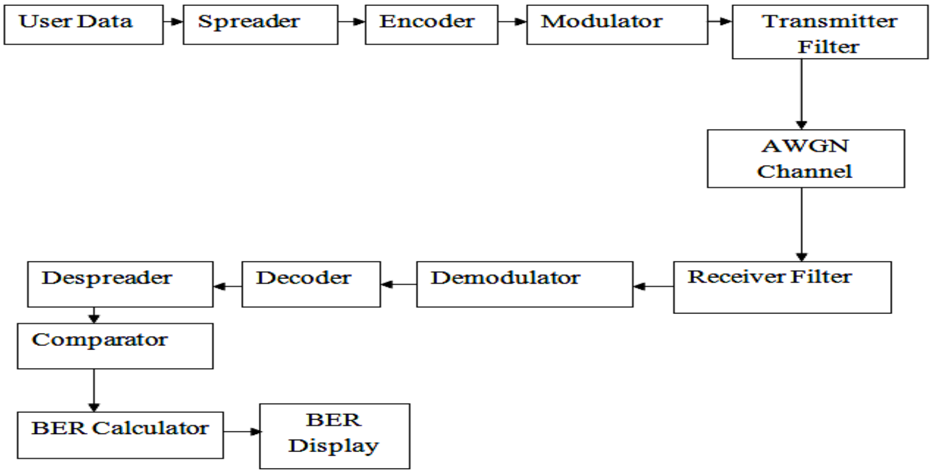

Figure 4.1: Block diagram of a WCDMA system model used for simulation

(a) Transmitter Design

The transmitter section has the data generator, PN sequence generator, spreader, encoder, modulator and square root raised cosine filter. The data generator used is the Bernoulli binary generator.

(i) Bernoulli Binary Generator

It uses Bernoulli distribution to generate random binary numbers. The distribution has a parameter *p* (probability) which produces a zero (0) and one (1) when the probability is 1-*p*. The mean value of this type of distribution is 1-*p* and the variance is given as *p**(1-*p*). When *p* is specified by a probability of a zero parameter, there can be any real number between zero and one produced.

The user data rate is set in this block in the sampling time parameter which is given as;

Sampling time=$\frac{1}{R_b}$ where R_b is the data rate

The MATLAB simulation software has the Bernoulli binary, random integer and Poisson integer generators which are used as sources of data. The Bernoulli binary generator gives an output which is binary while the other two generators give an output which is non-negative integers. When the random integer and Poisson integer generators are applied, the output needs to be converted to binary before spreading which might add some delay in this conversion process. Therefore, the Bernoulli binary generator is used as it gives out a binary data which can be fed directly be multiplied by the spreading code without any conversion [72].

(ii) PN Sequence Generator

The generator produces a PN sequence that is used for spreading the transmitted signal. This sequence has a frequency which is much higher than that of the user signal.

(iii) Differential Encoder

It encodes the binary input signal and its output is the logical difference between the present input and the previous output. In this block, the initial condition is set at zero (0).

(iv) Convolutional Encoder

It encodes a sequence of binary input vectors to produce a sequence of binary output vectors and processes multiple symbols at any given time**.** It is specified by the Trellis structure parameter which includes constraint length, generator polynomials and the feedback connection polynomials. This is done with the *poly2trellis* command which is located in the Trellis structure field of the encoder in Simulink. The convolution codes which have identical generator polynomials when used in wireless communication systems for error detection and correction, they produce poor BER. The minimum free distance for rate ½ convolutional codes is found to vary with the kind of generator functions applied. The generator functions giving out maximum free distance are found to be the optimum generator polynomials and they give better performance in terms of BER. The convolution coding with a constraint length of 7, generator polynomials (171,133) with a free distance of 10 and upper bound of 10, makes the code (7,[171,133]) an optimum generator polynomial from all the generator polynomials achievable for code rate of ½ [**11**]. Therefore, the simulation of WCDMA system at 2Mbps has used the constraint length of 7, code generator polynomials as 171 and 133 in octal numbering system so that to obtain better BER.

(v) QPSK Modulator Baseband

This modulator modulates the modulation of the signal by utilizing quaternary phase shift keying technique whose output is a baseband representation of the modulated signal.

The QPSK demodulator block is placed at the receiver to demodulate the signal that is modulated with QPSK modulation method in the transmitter section. The input to the demodulator must be a discrete-time complex signal and can be either a scalar or a frame-based column vector. The QPSK modulator is replaced by MQAM modulator (M=16 and 64) for 16-QAM or 64-QAM modulation and MPSK modulator with M=16 for 16-PSK modulation.

(vi) Raised Cosine Transmit Filter

It up-samples and filters the input signal using a normal raised cosine FIR filter or a square root raised cosine FIR filter. The following parameters are set in the raised cosine transmit filter block;

- The type of the filter is set as square root
- The roll off factor(α) as 0.22
- The group delay (D) which is the number of symbol periods between the start of the filter's response and the peak of the filter's response is set as 5. The up-sampling factor, N, is 4 and the length of the filter's impulse response can be determined from the expression $2 \times N \times D+1$. Therefore, the length of this filter is 21.
- The gain of the filter which indicates how the block will normalize the filter coefficients is also selected between '*user specified*' and '*normalized*'. This study uses normalized so that the block uses an automatic scaling.

The results given in Table 5-2, during simulation of the system with a variation of modulation schemes, the filter roll-off factor was taken as 0.22 which is the standard value given in the WCDMA technical specifications in[17] . This value is then varied for the second case to check whether it optimizes performance when the data rate is increased.

(b) Transmission Channel

The level of noise in the transmission channel is described by the quantities;

- The value of signal to noise ratio (*SNR*) which is normally the actual parameter of the AWGN channel
- The ratio of bit energy to noise power spectral density (E_b/N_o) and the ratio signal energy to noise power spectral density (E_s/N_o).

The ratio between E_b/N_o and $E_s/N_o$is given by;

$$E_s/N_o(dB) = E_b/N_o(dB) + 10\log k \qquad (4.1)$$

The value of $E_s/N_o$is influenced by the modulation technique and the code rate of the error control coding used. When the control coding scheme is implemented its value is given by the equation;

$$E_s/N_o(dB) = E_b/N_o(dB) + 10\log k - 10\log\ [Code\ Rate] \qquad (4.2)$$

Where k is the number of bits per symbol and the code rate was taken as $\frac{1}{2}$.

(c) Receiver Design

(i) Raised Cosine Receive Filter

The parameters of this filter are the same as those set on the raised cosine filter used in the transmitter.

(ii) Demodulator

The demodulator used in this case is the same modulation format as the modulator used in the transmitter section but does the reverse process of the modulator.

(iii) Differential Decoder

It does the reverse process of the differential encoder that is on the transmitter section.

(iv) Viterbi Decoder

It decodes the user information that was convolutionally encoded at the transmitter end. It can process multiple input symbols at any given time for a faster performance of the system. The decoder is also specified by the trellis structure parameter, decision type, trace-back depth and operation mode. For this study the Trellis structure is *poly2trellis (7, [171 133])*, trace-back depth as 34 and operation mode as continuous.

(v) Error Rate Calculator

The error rate calculator is used to determine the Bit Error Rate (BER). The error rate is calculated as a running statistic by dividing the total number of bits that are received in error by the total number of bits generated from the source of information.

The block can determine the symbol error rate or bit error rate. This is where the number of bits received in error and the total numbers of bits sent from the source are counted to determine the symbol or bit error rate. If the inputs are bits, then the block computes the bit error rate. If the inputs are symbols, then it computes the symbol error rate.

(vi) Display Block

This block displays the bit error rate which is determined by the error rate calculator. It displays the number of errors that are introduced in the signal transmitted by the channel noise. The block displays the Symbol Error Rate (SER) or the Bit Error Rate (BER), the total number of errors and the total number of symbols or bits transmitted. Considering the BER the information is displayed as shown in Fig. 4.2.

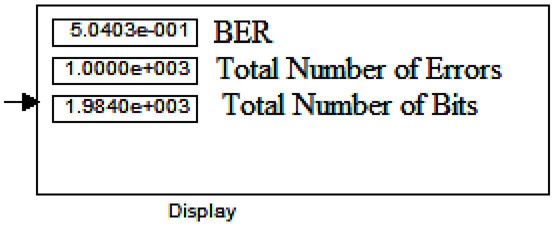

Figure 4.2: Information shown on the display block in Simulink simulation

CHAPTER FIVE

SIMULATION RESULTS AND DISCUSSION

In this chapter, the simulation results of a WCDMA system over the AWGN channel are given and discussed when there is variation of a modulation technique and the roll- off factor of the pulse-shaping filter. The simulation of wireless systems provides a powerful tool for analysis of these networks without the actual implementation of the real world systems. The WCDMA system model was simulated in MATLAB 7.8 (R2009a). The performance of the model developed was tested through different modulation techniques which were: QPSK, 16-PSK, 16-QAM and 64-QAM over the AWGN channel with and without channel coding. The same model was tested for different values of roll-off factor (α) in the square root raised cosine filter.

For validation of the implemented functions and simulation results, a comparison was made to the theoretical models without channel coding. The simulated results for the modulation schemes applied in the WCDMA system simulation were compared with the theoretical results. Then channel coding was applied to reduce the power efficiency difference between the two curves. The mapping of E_b/N_o ratio and the BER for QPSK modulation scheme is shown in Fig 5.1.

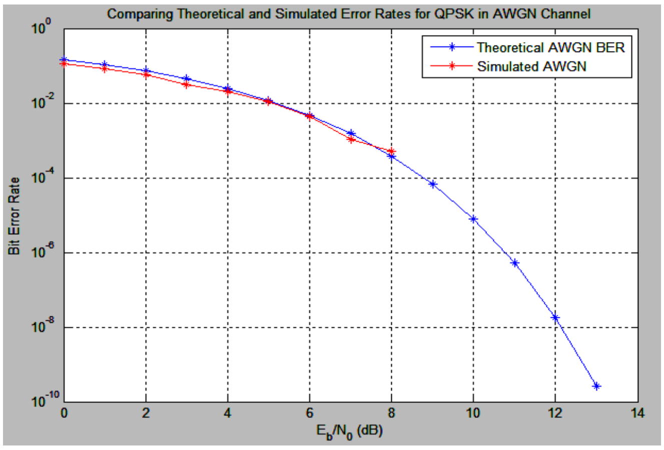

Figure 5.1: Comparison of BER in QPSK Modulation

From Fig. 5.1, it can be seen that the simulated results is very close to the theoretical plot of BER for QPSK modulation over the AWGN channel. The BER mapping for the 16-QAM modulation technique is given in Fig. 5.2.

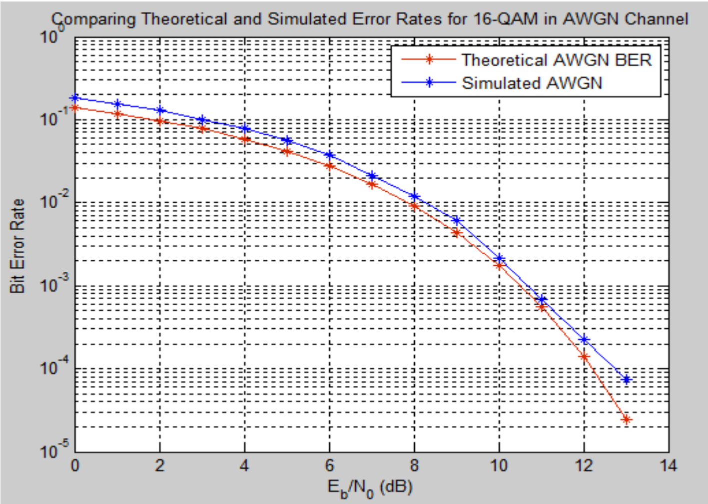

Figure 5.2: Comparison of BERs for 16-QAM modulation

From Fig. 5.2, it can be seen that the simulated BERs are slightly higher when compared to the theoretical results. The same relationship was observed in 64-QAM given in Fig. 5.3 but with a greater margin of increase of BER than in 16-QAM. Therefore, it can be said that the modulation formats that were applied in WCDMA simulation have been validated.

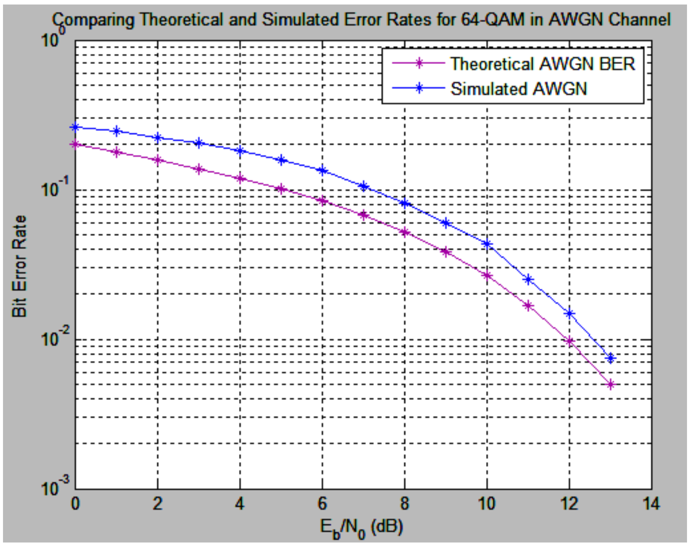

Figure 5.3: Comparison of BERs in 64-QAM modulation

The results for different simulation set ups were also verified for the modulation techniques by comparison with the results that are known under the AWGN environment and they were found to be consistent with those given in [57]. The parameters used in the simulation model are given in Table 5.1.

Table 5-1: Parameters used in the simulated model

Component	***Parameters***
Bernoulli Data generator	Sample rate=2Mbps, Generator sample time =1/sample rate
Convolutional encoder	Code length=7, code rate=1/2, code generator polynomial=(177, 133) in octal number form
Modulator	M-PSK (M=4, 16), M-QAM(M=16, 64)
Pulse Shaping filter	Square root raised cosine filter parameters: Group delay=5, roll-off factor=0.22, oversampling factor=4
Channel	AWGN channel
Demodulator	M-PSK (M=4, 16), M-QAM(M=16, 64)
Viterbi Decoder	Trace back length=34, hard decision decoding, operation mode=continuous, Code length=7, code generator polynomial=(177 133) in octal

The signals at the various stages of the system model as observed in the discrete time signal scope in Simulink are shown in Fig 5.4 to Fig 5.9.

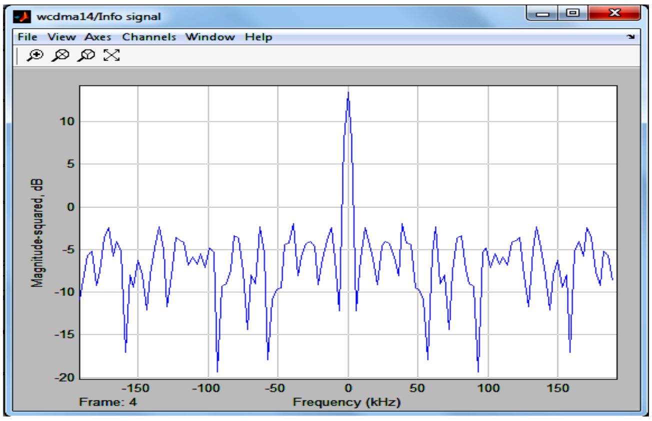

Figure 5.4: Information signal generated in MATLAB

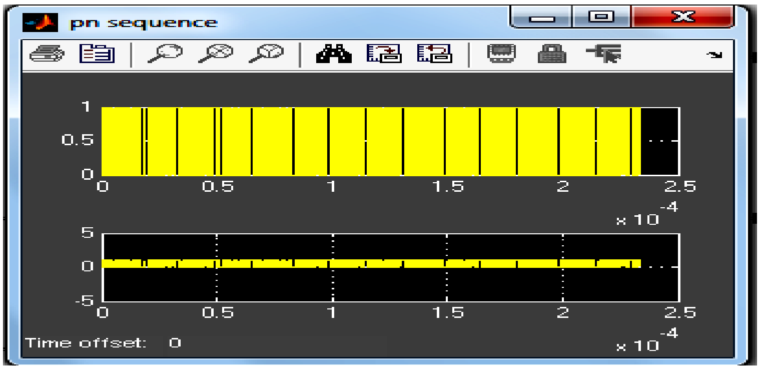

Figure 5.5: The PN sequence viewed on a vector scope with 2 channels

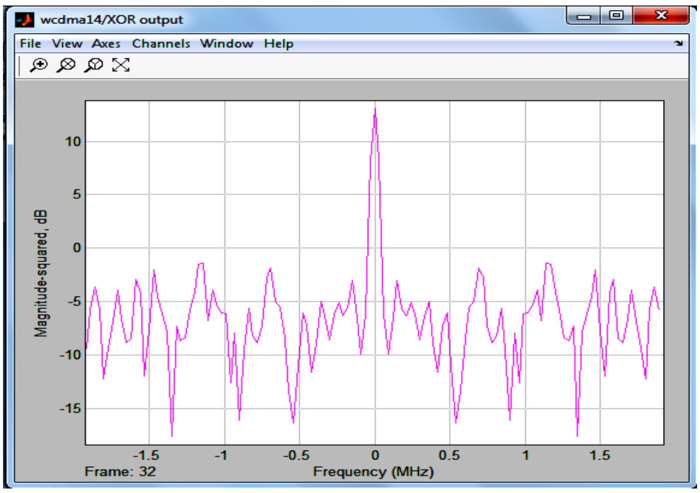

Figure 5.6: The spreaded signal which is the output of the XOR Gate

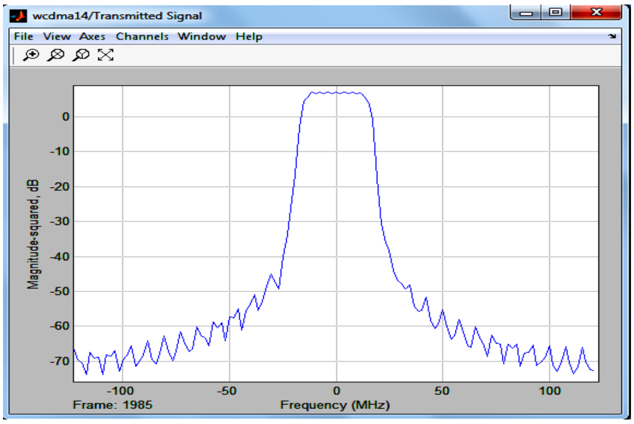

Figure 5.7: The modulated, up-sampled and filtered signal for transmission

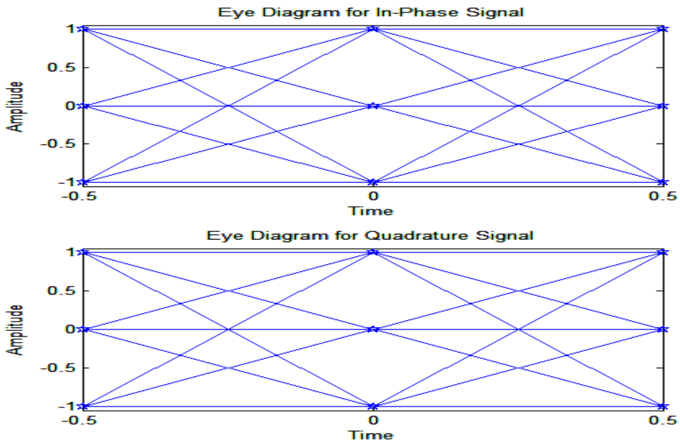

Figure 5.8: The eye diagram of the I and Q signals for QPSK modulation

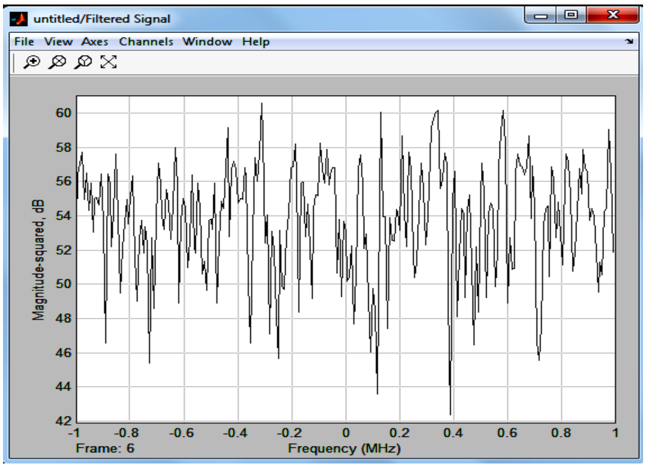

Figure 5.9: The received, filtered signal

The other signals were plotted during the simulation process using the m-files created in MATLAB as follows:

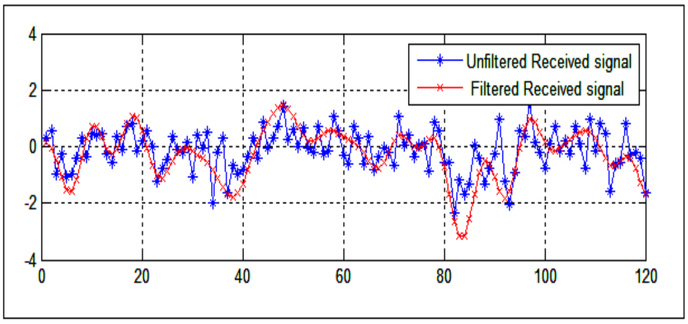

Figure 5.10: Comparison of filtered and unfiltered received signals

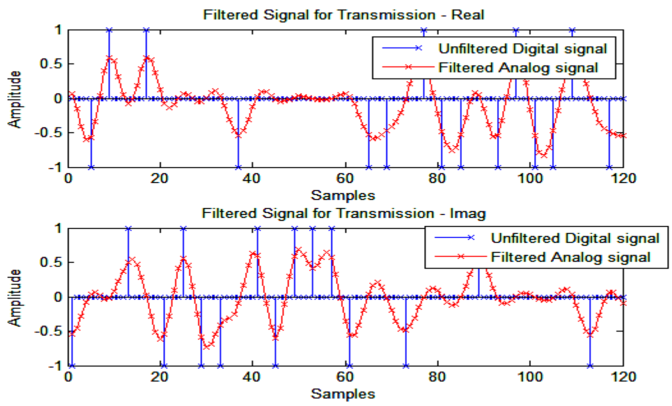

Figure 5.11: Comparison of filtered and unfiltered transmitted signal

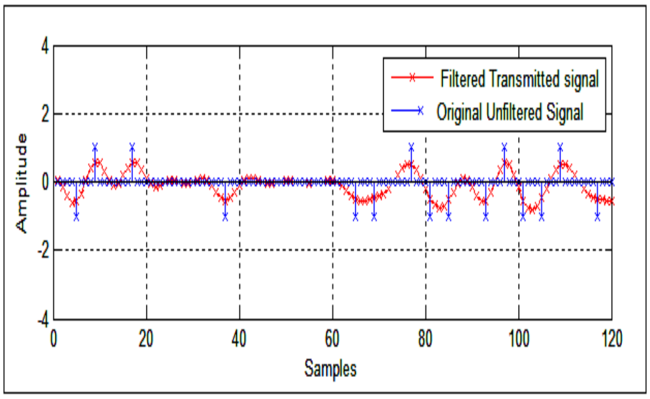

Figure 5.12: Comparison of filtered transmitted and original signal

Case I: Variation of the Modulation Formats

The simulated results of a WCDMA system at a data rate of 2Mbps with and without convolution coding due to the variation of modulation techniques are given in Table 5.2. The modulation techniques used were QPSK, 16-PSK, 16-QAM and 64-QAM over an AWGN channel.

Table 5-2: Simulated BERs over AWGN channel

	Modulation Technique							
E_b/N_o	**QPSK**		**16-PSK**		**16-QAM**		**64-QAM**	
	No Coding	*Coding*	*No Coding*	*Coding*	*No Coding*	*Coding*	*No Coding*	*Coding*
0	0.2130	0.4144	0.2830	0.4916	0.2123	0.4817	0.2689	0.4977
1	0.1793	0.3466	0.2613	0.4842	0.1874	0.4602	0.2500	0.4946
2	0.1454	0.2501	0.2387	0.4744	0.1643	0.4160	0.2310	0.49
3	0.1118	0.1432	0.2145	0.455	0.1414	0.3316	0.2116	0.4806
4	0.0817	0.0605	0.1904	0.4231	0.1194	0.2120	0.1919	0.4608
5	0.0550	0.0186	0.1668	0.3721	0.0980	0.0968	0.1727	0.4271
6	0.0339	0.0038	0.1444	0.2981	0.0777	0.0296	0.1529	0.3674
7	0.0189	6.3e-4	0.1234	0.2071	0.0588	0.0063	0.1340	0.2808
8	0.0090	6.0e-5	0.104	0.1223	0.0420	0.000856	0.1146	0.181
9	0.0036	3e-6	0.0855	0.0591	0.0281	0.000079	0.0956	0.0914
10	0.0012	0	0.068	0.0237	0.0170	0.000005	0.0772	0.0366
11	0.000286	0	0.052	0.0079	0.0093	0	0.06	0.0118
12	4.73e-5	0	0.0376	0.0024	0.0045	0	0.0443	0.003
13	6.75e-6	0	0.0257	0.00052	0.0018	0	0.0304	6.2678e-4
14	0	0	0.0157	1.03e-4	5.708e-4	0	0.0195	9.5456e-5
15	0	0	0.0089	0.000014	1.418e-4	0	0.0112	8.586e-06
16	0	0	0.0043	1.5e-6	2.55e-5	0	0.0057	1.010e-06
17	0	0	0.0019	0	2.5e-6	0	0.0025	0
18	0	0	5.883e-4	0	0	0	0.00091389	0
19	0	0	0.000156	0	0	0	0.0002452	0
20	0	0	0.000034	0	0	0	5.6818e-5	0

The improvement in performance in a wireless communication system when the BER is taken as the performance indicator is shown by the movement of the BER curve in the direction shown in Fig. 5.13 after the application of channel coding.

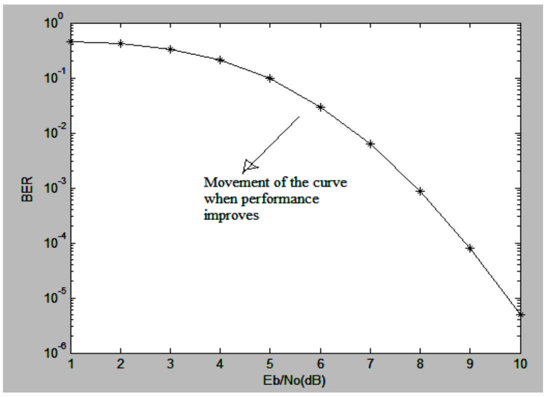

Figure 5.13: Movement of BER curve due to improvement in performance

This can be observed by plotting the BER curves for QPSK and 16-QAM modulation techniques with and without channel coding as given in Fig. 5.14 and 5.15.

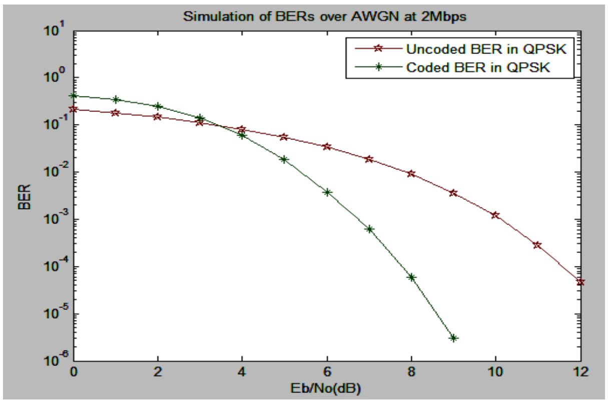

Figure 5.14: Performance improvement in QPSK modulation

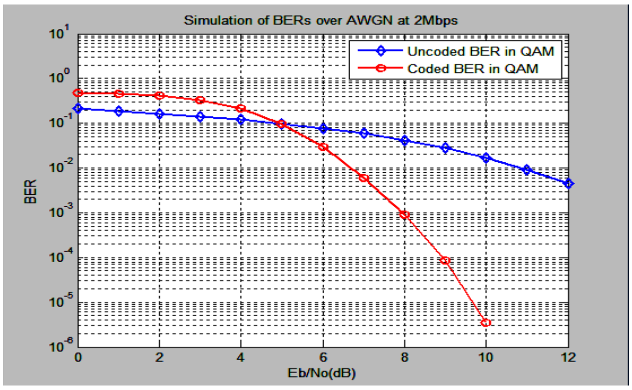

Figure 5.15: Performance improvement in 16-QAM modulation

From Fig. 5.14 and 5.15, it can be seen that the application of error correction coding reduced the E_b/N_o required for a particular BER level. For example, a BER 10^{-6} is obtained at a ratio of E_b/N_o of around 17dB without coding which is reduced to around 10dB when coding is done for 16-QAM modulation. The BER of 10^{-6} for QPSK modulation is obtained at around 13dB without coding which reduces to 9dB with error correction coding. This means that error correction coding has improved the performance in terms of power efficiency for the system. This reduces the transmitter or antenna cost which will have been incurred because the transmit power is not increased for better BER to be obtained.

The trend that can be observed with the comparison of modulation techniques performance can be validated by simulating the theoretical error rates by the *bertool* in MATLAB with and without convolution coding over the AWGN channel. The bertool plotted the theoretical BERs for the modulation schemes that were used in the simulation for the 2Mbps system as in Fig. 5.16 and 5.17 without and with channel coding (for hard decision) respectively. The simulated error rates of a WCDMA model transmitting at a rate of 2Mbps was done using the m-files created in MATLAB for a case where the modulation formats were varied.

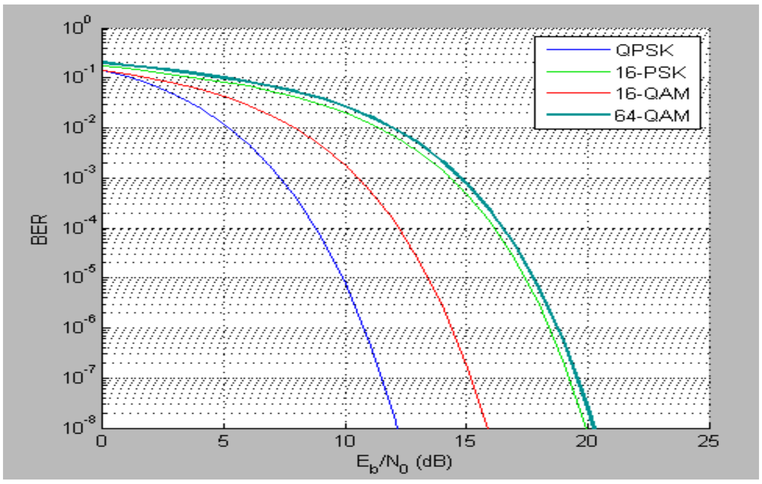

Figure 5.16: Theoretical BERs without convolution coding

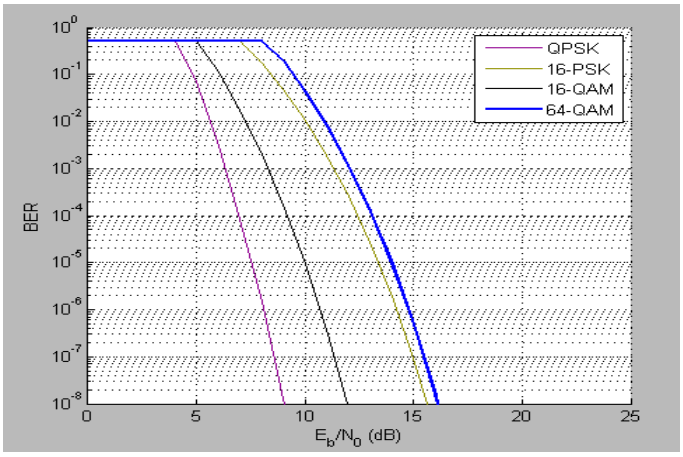

Figure 5.17: Theoretical BERs convolution coding for hard decision decoding

The comparison of the simulated BERs at 2Mbps data rate over AWGN channel was done by plotting the results given in Table 5.2 in Fig. 5.18 and 5.19 respectively.

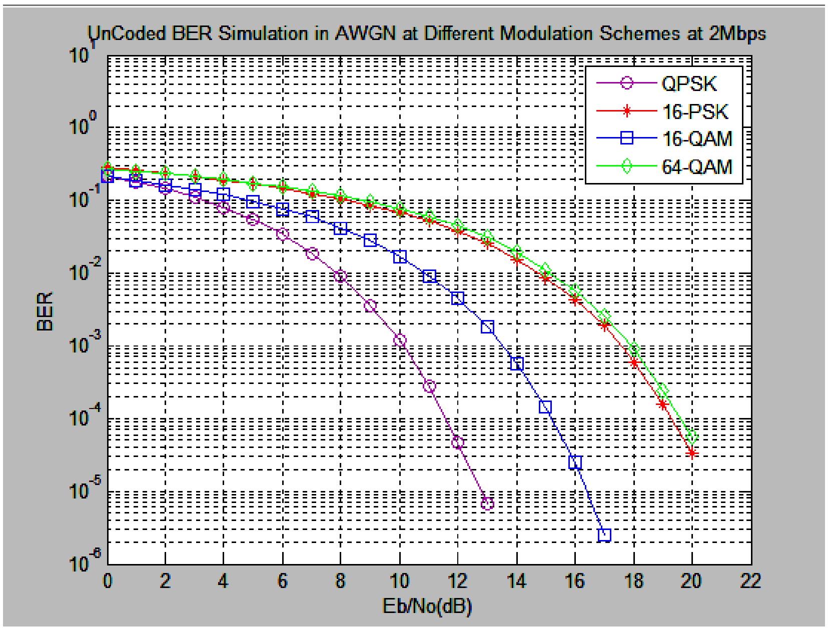

Figure 5.18: Comparison of BERs at 2Mbps without convolutional coding

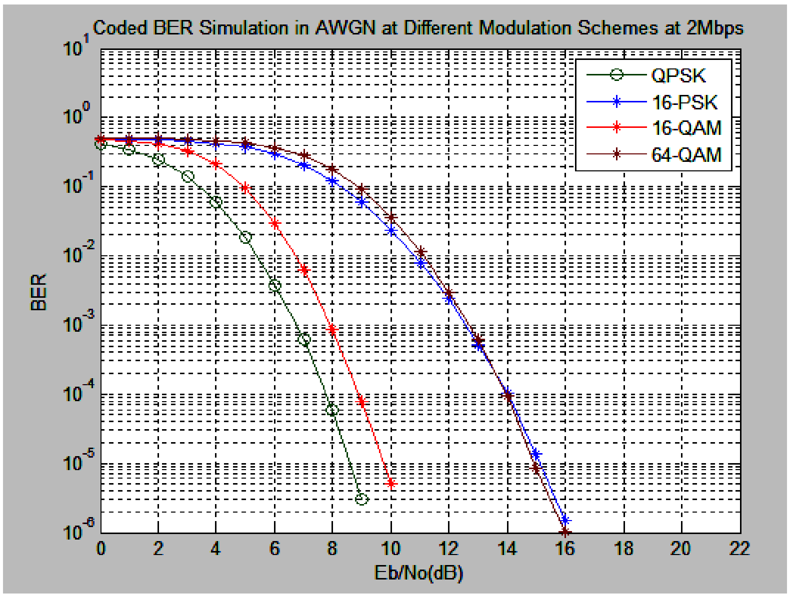

Figure 5.19: Comparison of BERs at 2Mbps with convolutional coding

The performance was analyzed by considering the BER and the power efficiency at the receiver. The Figures 5.16 and 5.17 show the theoretical error rates when the modulation formats were varied without channel coding. Fig. 5.18 and 5.19 show the simulated error rates for different modulation schemes without and with channel coding respectively. From Fig. 5.18 and 5.19, it can be seen that the power efficiency to obtain an error rate of 10^{-3} is 6.8dB, 12dB, 8dB and 12.4dB for QPSK, 16-PSK, 16-QAM and 64-QAM respectively with convolutional coding. The power efficiency is 10dB, 17dB, 13dB and 17dB for QPSK, 16-PSK, 16-QAM and 64-QAM respectively without convolutional coding. It can therefore be seen that channel coding is very important in wireless communication systems as it minimizes the required transmit power as shown by the decrease in power efficiencies. This reduces the transmitter cost and permits increased data rates for the same transmitter power and antenna size. Therefore, it can be said that the performance in terms of power efficiency of a WCDMA system transmitting data at 2Mbps has improved with the application of error correction coding.

In addition, to implement an error free communication at a data rate of 2Mbps with channel coding the power efficiency is 10dB, 17dB, 11dB and 17dB for QPSK, 16-PSK, 16-QAM and 64-QAM respectively. When all the modulation schemes are compared, QPSK gives the best result at this data rate followed by 16-QAM, 64-QAM and 16-PSK in that order up to around 13dB with convolutional coding present. The 16-PSK performs better than 64-QAM after 13dB with channel coding but 64-QAM gives better performance than 16-PSK when there is no coding for all values of E_b/N_o.

The 64-QAM and 16-QAM modulation schemes have a higher throughput at the expense of the BER while QPSK performs well over AWGN channel but has a lower throughput. Therefore, this study was important as it gives information that can be used as a guide by the designer of wireless communication systems to make a choice between modulation formats in terms of throughput required, BER level and the average signal to noise power spectral density ratio.

Case II: Variation of the Filter Roll off Factor (α)

For the second case, the system was simulated with the variation of the pulse shaping filter roll off factor with convolutional coding over the AWGN channel at a data rate of 2Mbps and 16-QAM modulation. The filter roll-off factor was varied while keeping the ratio E_b/N_o constant so that a value of a roll-off factor can be chosen which minimizes the error rate and improved power efficiency. The results were tabulated for 16-QAM modulation technique without channel coding in Table 5.3 and with channel coding in Table 5.4. The 16-QAM modulation technique was used in this case because it has a higher throughput compared with QPSK modulation. Therefore, since the margin of error between QPSK and 16-QAM is not very big as in 16-PSK and 64-QAM, the 16-QAM modulation technique, it can be easily be extended to higher data rates without increasing the simulation time.

Table 5-3: BERs for 16-QAM modulation without convolution coding

E_b/N_o	Filter Roll-off factor (α)								
	0.1	0.2	0.3	0.4	0.5	0.6	0.7	0.8	0.9
1	0.1890	0.1874	0.1880	0.1877	0.1878	0.1871	0.1878	0.1875	0.1875
2	0.1656	0.1642	0.1639	0.1639	0.1640	0.1641	0.1641	0.1640	0.1635
3	0.1436	0.1413	0.1410	0.1410	0.1412	0.1414	0.1410	0.1413	0.1407
4	0.1218	0.1193	0.1192	0.1195	0.1192	0.1192	0.1190	0.1192	0.1193
5	0.1004	0.0978	0.0979	0.0980	0.0982	0.0980	0.0981	0.0981	0.0980
6	0.0810	0.0778	0.0776	0.0777	0.0776	0.0778	0.0777	0.0774	0.0779
7	0.0626	0.0589	0.0587	0.0587	0.0589	0.0589	0.0587	0.0588	0.0588
8	0.0462	0.0423	0.0421	0.0422	0.0421	0.0422	0.0420	0.0423	0.0419
9	0.0323	0.0280	0.0283	0.0281	0.0281	0.0280	0.0281	0.0279	0.0281
10	0.0212	0.0171	0.0173	0.0171	0.0170	0.0171	0.0170	0.0170	0.0172
11	0.0129	0.0093	0.0093	0.0092	0.0095	0.0093	0.0093	0.0093	0.0093

Table 5-4: BERs for 16-QAM modulation with convolutional coding

E_b/N_o	Filter Roll-off factor (α)								
	0.1	0.2	0.3	0.4	0.5	0.6	0.7	0.8	0.9
1	0.4623	0.4604	0.4614	0.4603	0.4591	0.4591	0.4589	0.4604	0.4606
2	0.4185	0.4149	0.4139	0.4132	0.4155	0.4153	0.4149	0.4160	0.4126
3	0.3410	0.3317	0.3322	0.3314	0.3304	0.3319	0.3306	0.3311	0.3273
4	0.2267	0.2109	0.2106	0.2125	0.2108	0.2115	0.2107	0.2117	0.2118
5	0.1078	0.0961	0.0961	0.0969	0.0986	0.0973	0.0962	0.0981	0.0966
6	0.0364	0.0295	0.0295	0.0296	0.0291	0.0299	0.0306	0.0295	0.0305
7	0.0085	0.0061	0.0058	0.0060	0.0061	0.0062	0.0062	0.0061	0.0062
8	0.0015	8.401e-4	7.901e-4	8.7051e-4	8.8101e-4	9.1751e-4	8.84501e-4	9.1051e-4	8.4501e-4
9	1.83e-4	8.6501e-5	1.055e-4	7.7001e-5	8.7001e-5	8.1501e-5	9.3501e-5	7.5501e-5	8.3501e-5
10	2.5e-5	1.5e-6	1e-6	6.5001e-6	5.0001e-6	6.5001e-6	5.5001e-6	1.05e-5	6.5001e-6
11	0	0	0	0	0	0	0	0	0

The graphs of BER against the filter roll off factor at each value of E_b/N_o ratio for 16-QAM modulation when convolution coding is incorporated in the system were plotted in MATLAB as shown in Fig. 5.20 to Fig. 5.29.

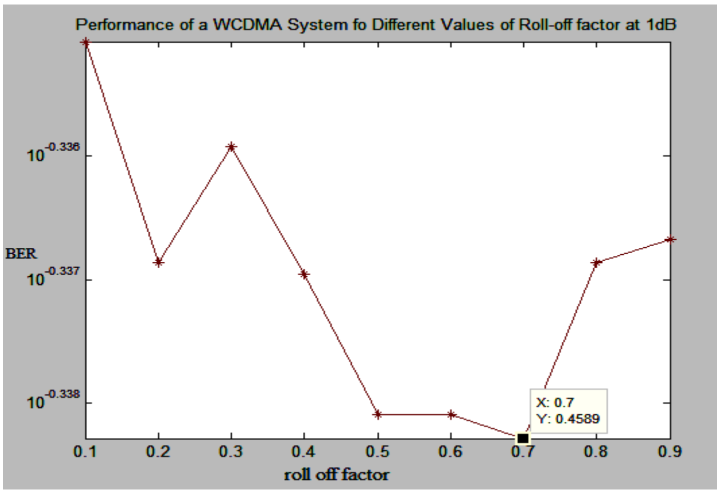

Figure 5.20: Variation of BER at 1dB of E_b/N_o ratio

From Fig. 5.20, it was found that the roll-off factor of 0.7 with a BER of 0.4589 has the minimum error rate. However, this value of roll-factor tends to 1 making the main lobe width wider and the side lobes are at their lowest. This generates many errors during detection of the signal that was transmitted and as such this value of roll-off factor is not optimum.

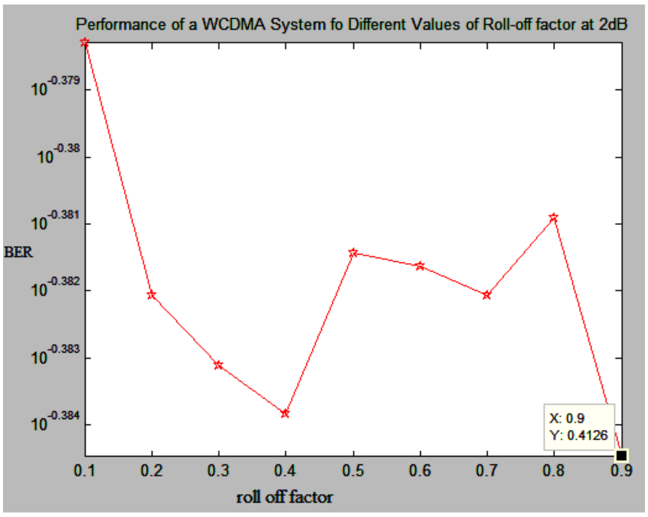

Figure 5.21: Variation of BER at 2dB of E_b/N_o ratio

At 2dB of E_b/N_o, the roll-off factor with minimum BER is 0.9 as in Fig. 5.21. This also results in many errors during detection of the signal as it makes the main lobe width to be larger which allows more stray signals to pass through it. This implies there is no optimum width of the main lobe. The same result was obtained at 3dB in Fig. 5.22 when the value of roll-off factor with the minimum BER is 0.9.

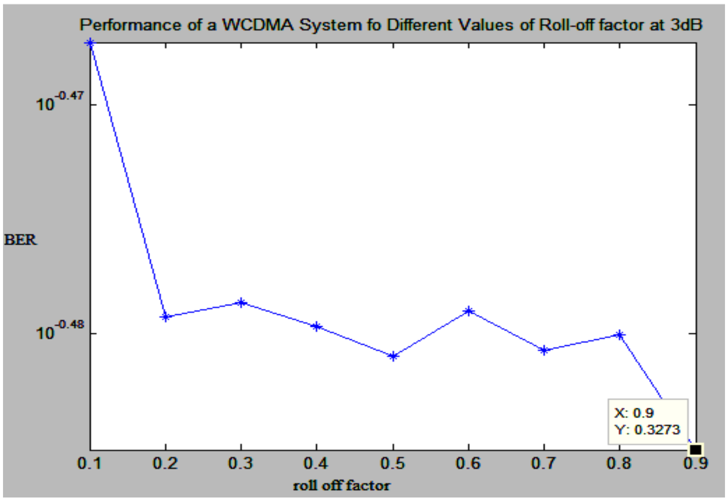

Figure 5.22: Variation of BER at 3dB of E_b/N_o ratio

As the value of the ratio E_b/N_o continues to increase, the value of roll-off factor with minimum BER decreases. This can be seen in Fig. 5.23 where at 4dB, the roll-off factor with minimum BER is 0.3. This value of roll-off factor produces a narrower width of the main lobe and the BER value decreases as lower levels of stray signal are allowed to pass through the main lobe. The same value of roll-off factor is obtained at 5dB as shown in Fig. 5.24.

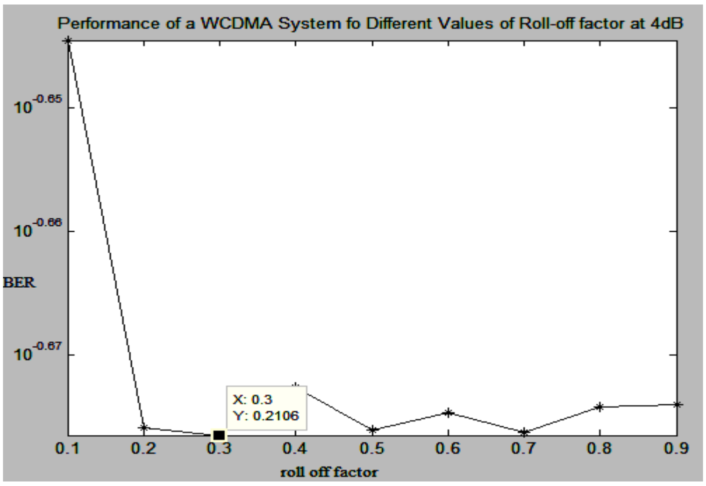

Figure 5.23: Variation of BER at 4dB of E_b/N_o ratio

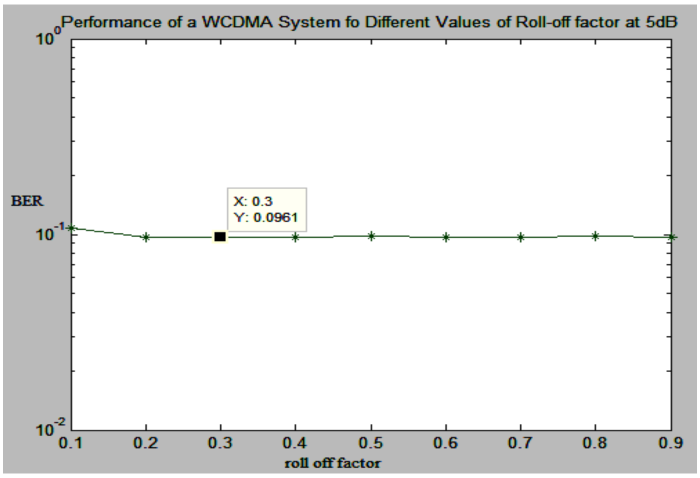

Figure 5.24: Variation of BER at 5dB of E_b/N_o ratio

In Fig. 5.25, at 6dB the value of filter roll-off factor deteriorates slightly to a value of 0.5 which outside the range that gives an optimum value of the main lobe width.

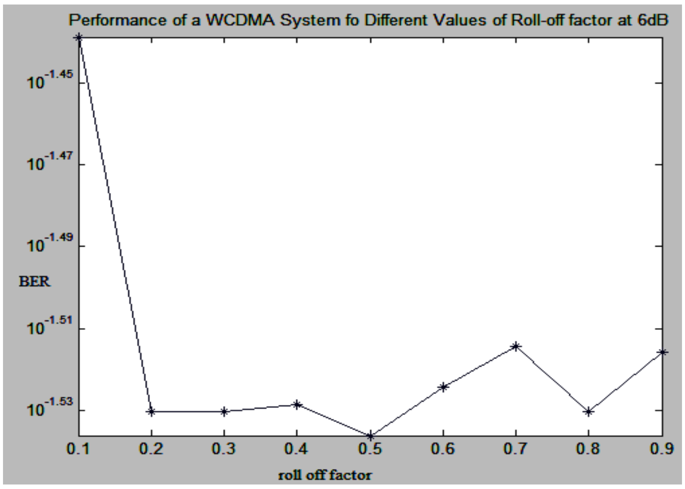

Figure 5.25: Variation of BER at 6dB of E_b/N_o ratio

In addition, the filter roll-off factor of α=0.3 gives the minimum value of BER for an E_b/N_o ratio of 7dB, 8dB and 10dB as shown in Fig. 5.26, Fig. 5.27 and Fig. 5.29. The main lobe width in this case is narrower which gives better bandwidth efficiency. This also ensures that there are fewer errors during detection of a signal at the receiver which improves the BER of the system.

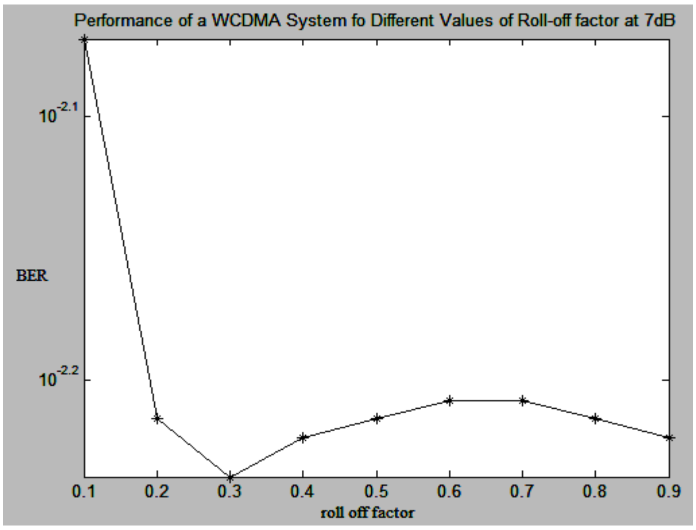

Figure 5.26: Variation of BER at 7dB of E_b/N_o ratio

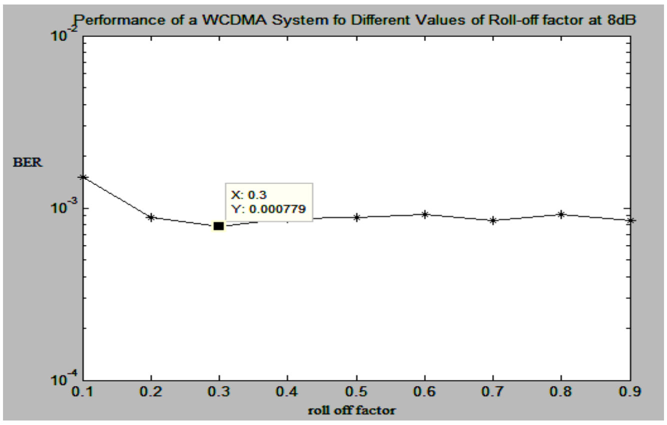

Figure 5.27: Variation of BER at 8dB of E_b/N_o ratio

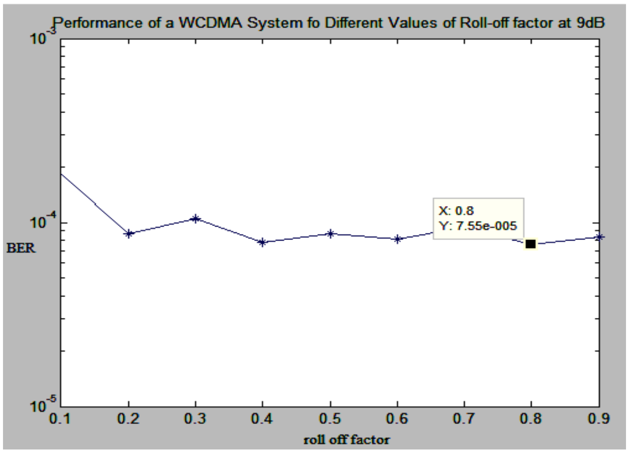

Figure 5.28: Variation of BER at 9dB of E_b/N_o ratio

At 9dB of E_b/N_o ratio, the roll-off factor with minimum value of BER is 0.8 which again does not give an optimized width of the main lobe. There is poor bandwidth efficiency in this case and the error rate of the received signal is high.

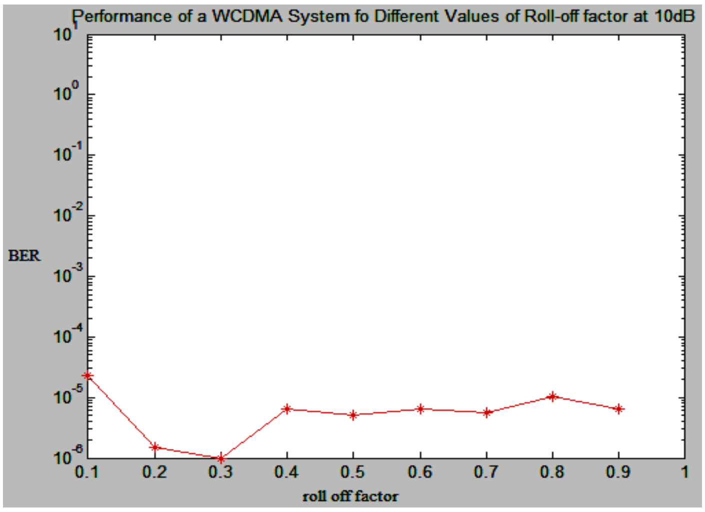

Figure 5.29: Variation of BER at 10dB of E_b/N_o ratio

The results from the simulation of the WCDMA system when the filter roll-off factor is varied can be summarised in Table 5.5 where the filter roll off factor with the minimum value of the BER for each value of E_b/N_o is indicated.

Table 5-5: Roll-off factor with Minimum BER for each value of (Eb/No)

E_b/N_o(dB)	1	2	3	4	5	6	7	8	9	10
Roll-off (Minimum BER)	0.7	0.9	0.9	0.3	0.2/ 0.3	0.5	0.3	0.3	0.8	0.3

From Table 5.5, it can be seen that for the system model that was developed and employing a pulse shaping filter the filter roll-off factor which gives the minimum BER can be taken as α=0.3 for all the values of E_b/N_o . This roll-off factor gave the minimum BER for most values of E_b/N_o and as such it can be taken as the optimum value of the roll off factor when the complexity of the filter is taken into consideration. The system data rate was taken as 2Mbps and the modulation technique was 16-QAM.

The results from the variation of the filter roll-off factor showed that the performance of a wireless communication system that is employing a pulse shaping filter depends on the value of its roll-off factor. When this system was simulated at 2Mbps over the AWGN channel, the filter roll-off factor α of 0.3 was found to give a lower error rate in many levels of the bit energy to noise power spectral density ratio which optimizes the main lobe width giving narrow transmission bandwidth and better BER level. Therefore, this value of filter roll-off factor can be taken as the optimum value that can be applied for this data rate to achieve a better BER level since the narrower the bandwidth the better the BER without increasing the complexity of the filter.

This can also be explained from the spectrum of a carrier signal used in wireless communication systems which is represented by the main lobe and a number of side lobes which have lower amplitudes. When the pulse shaping filter is applied in communication systems, it increases the width of the main lobe and/or reduces the amplitude of side lobes. When there is a higher width in the main lobe and smaller amplitude in the side lobes, the inter-channel interference reduces and better bandwidth efficiency obtained. The width of the middle frequencies is determined by the roll off factor α which lies between 0 and 1. This parameter determines the bandwidth occupied by the pulse and the rate at which the tails decay. The side lobes are undesirable in a communication system as they increase the bandwidth of the transmitted signal.

The filter roll-off factor obtained in this thesis (α=0.3) is within the range given in [73] of 0.22-0.33 for better bandwidth efficiency. The optimization of the filter roll-off factor ensures that the transmission bandwidth is reduced to an optimum value so that there is minimal reception of interference signals. Therefore, the interference signals are reduced at the receiver when the bandwidth of the main channel is reduced, which in this case is optimized when the value of the filter roll-off factor is 0.3.

This analysis on the filter roll-off factor is beneficial as it provides a guide to the system designer to choose pulse shaping filter with an optimum value of its roll-off factor depending on the BER level and system requirements. It also shows how stray signals (interference) can be reduced in a system with the optimization of transmission bandwidth in the system. Therefore, the optimization of the filter roll-off factor in this case reduced the interference encountered in the system.

Since the error rate probability for QPSK and BPSK is the same, then the simulation of WCDMA at 2Mbps in this thesis can be compared with a study of 384kbps in [57] as shown in Figure 5.30. It can be seen that this study which has modelled a WCDMA system transmitting at 2Mbps offers a better performance than that of 384kbps. Furthermore, in [10] the performance in terms of power efficiency for a transmission of 384kbps over AWGN channel with convolution coding is 7dB for a BER of 10^{-3}. However, in this study at 2Mbps with convolution coding and same transmission channel, the power efficiency is 6.8dB at a BER of 10^{-3} which is much better compared to 384kbps. Therefore, it can be said that the WCDMA system model simulated in this study which is transmitting at a higher data rate of 2Mbps has an improved performance of 0.2dB or 2.857%. The improvement in performance in a WCDMA system is important as the system can be able to deliver integrated services at a better quality and accommodate more users.

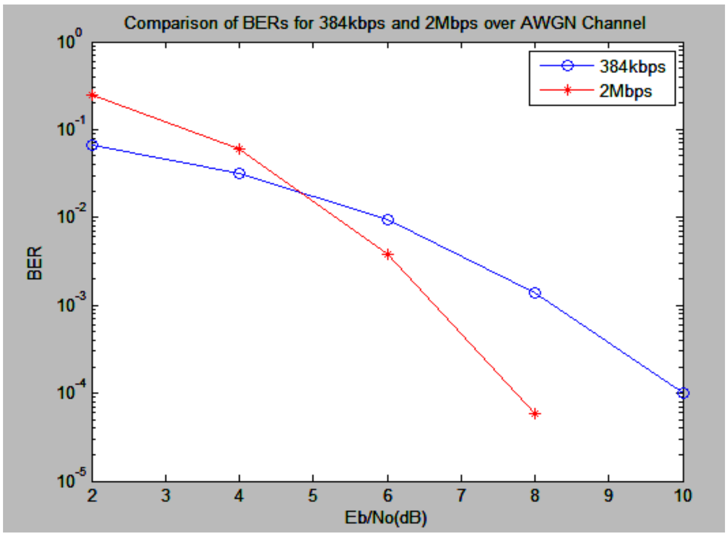

Figure 5.30: Comparison of 384kbps and 2Mbps BER over AWGN channel

CHAPTER SIX

CONCLUSION AND FUTURE WORK

This chapter gives a summary of the work that has been presented in this thesis and outlines a few directions for future research based on this study.

6.1 Conclusion

The WCDMA system model has been simulated at 2Mbps with or without convolutional coding over an AWGN channel. It has been found that when the error correction scheme is implemented the power efficiency of the system improves and the bit error rate (BER) reduced allowing for reliable and quality communication. The error rate depends on the modulation format used, ratio of bit energy to noise power spectral density, E_b/N_o and the channel conditions. The performance of this system was analysed for a data rate of 2Mbps by considering the variation of the modulation format used and the variation of the filter roll off factor with the ratio of bit energy to noise power spectral density kept constant.

The modulation formats whose performance was compared were QPSK, 16-PSK, 16-QAM and 64-QAM with and without convolutional coding. The performance was found to be better in QPSK than other modulation formats which made QPSK an efficient modulation scheme at this data rate (2Mbps) to deliver quality services. This means that QPSK can only be applied in systems with poor channel conditions and other modulation schemes in systems with better channel conditions. The QPSK modulation scheme offers a better performance because it only has the phase variations which are not prone to noise but the QAM modulation formats have the amplitude variations which are prone to noise which increases the BER. During the simulation of this system, the phase variations which might have affected the BER were minimized by compensating for the delays in the system and use of coherent detection at the receiver which minimized the phase errors. In coherent detection, the demodulator computes the phase angle of the received symbols and compares it to the phase angle of the transmitted symbols. The demodulator selects the phase that is very close to the phase of the transmitted symbols. The demodulator can also compute the difference between the two phases and choose a received phase that gives the smallest difference at the output [56]. This ensures that there is a minimal contribution of the phase variation to BER. The 16-QAM modulation has a better performance compared to 16-PSK since the symbols in 16-QAM cover all the spaces in the constellation diagram and not confined to a densely packed circle which is found in 16-PSK. Therefore, for applications which require same spectral efficiency the rectangular constellation QAM is used instead of PSK which can only be used where linear amplification is considered. In addition, the 16-PSK has phase discontinuities due to transitions from symbol to symbol.

The performance was improved with the application of convolution coding which adds redundant bits to the data stream. This enabled the receiver to identify and correct errors without requesting for a retransmission. The retransmission requires a reverse channel where the request for retransmission passes through. Since retransmission is unavailable in convolution coding, it reduces the bandwidth when this coding scheme is coupled with an M-ary modulation scheme. This has the effect of improving the BER in a digital communication system which implies an increase in capacity when the error rate is low and allowing of more users to access the services. The convolution coding scheme not only does it improve the performance but also reduces the implementation cost of a wireless communication system. The study is beneficial in network planning when a decision is supposed to be made on the type of services to be provided (data rate or throughput required), type of modulation to be utilized to achieve the required power efficiency and bit error rate.

The filter roll off factor is directly proportional to the width of the main lobe and inversely proportional to the side lobes' amplitude. This implies that when there is higher roll-off factor, the width of the main lobe increases, which reduces the bandwidth of the transmitted signal. In addition, since the side lobes increase the bandwidth of the transmitted signal, the higher number of users who would have been allowed to share the same bandwidth for communication reduces.

When the roll off factor is 1 ($\alpha=1$), the spectra of the signals has widest main lobes and lowest side lobe, which is positive and decreases the signals' bandwidth.

However, the neighbouring channels overlap the main channel very much which makes the main lobes to overlap. The wider main lobes cause many errors during detection of the main channel.

When the roll off factor is zero (α=0), the main lobes are narrowest with less overlapping of main lobes of neighbouring channels with the main lobe of the main channel. There is no optimal BER characteristic in this case because there are high-amplitude side lobes which interfere with the main signal. The transmission bandwidth is also very high in this case which increases the BER. For reliable communication, the system should have a better BER with a reduced bandwidth. The reduction in bandwidth decreases the BER as lower levels of noise are received, which results in an improvement of signal to noise ratio. Therefore, the roll off factor of 0.3 optimizes the transmission bandwidth and the BER level and that is why it gave better BER level compared to other values. From the third objective of the research, optimization of bandwidth using the filter roll off factor was important so that the errors in the pass band are decreased since small traces of noise would be allowed through this band.

6.2 Future Work

The directions for further research in the area of WCDMA system performance and next generation wireless communication systems are outlined as follows:

The AWGN channel can also be replaced by multipath fading channels so that the propagation loss can be incorporated into the performance analysis of the system. The multipath fading channel takes the Rayleigh fading and Rician fading losses and a comparison made with additive white Gaussian noise channel at a data rate of 2Mbps in WCDMA. In addition, other spreading sequences such as Walsh codes can be used to spread the signal that is transmitted through the system at the same data rate or higher. The performance of this code can then be compared with the one obtained when the PN code is applied in the system. The study can be extended to other network technologies such as Wi-Fi IEEE 802.11, 802.11b and 802.11g which also utilize direct sequence spread spectrum with speeds of up to 12Mbps.

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APPENDICES

APPENDIX 1 Publications

This thesis gave rise to the following list of conference presentations and journal publication;

1. Filbert O. Ombongi, Philip L. Kibet, Stephen Musyoki, *Performance Analysis of a WCDMA System Model in a Low Mobility and Indoor Environments with Channel Coding over Additive White Gaussian Noise Channel,* International Journal of Engineering Research in Africa, ISSN: 1663-4144 ,Vol. 15, pp 123-133 http://www.scientific.net/JERA.15, April 2015
2. Filbert Onkundi Ombongi, Philip Langat Kibet, Stephen M. Musyoki, *Performance Analysis of Pulse Shaping Filter for WCDMA Applications, International Journal of Advanced Engineering and Global Technology,* ISSN: 2309-4893, Vol. 2, No. 11, pp 88-93, November 2014, www.ijaegt.com
3. Filbert Ombongi, Philip Kibet, Stephen Musyoki, *Bit Error Rate Performance Analysis of a Wideband Code Division Multiple Access System Model in an Indoor Environment using Convolution Coding over Additive White Gaussian Noise Channel, International Journal Of Information Technology And Electrical Engineering (ITEE),* ISSN: 2306-708X, Vol. 3, No. 4, August, 2014, Pg 10-16. http://iteejournal.org/index.php
4. Filbert O. Ombongi, P.L Kibet, S. Musyoki, "*Comparison of BER Performance of a WCDMA System in Rayleigh fading and AWGN Channels at Different Data Rates*", *Proceedings of 2014 International Conference on Sustainable Research and Innovation (SRI),* ISSN: 2079-6226, held on 7th-9th May 2014 in AICAD, JKUAT, Pg 239-242
5. Filbert O. Ombongi, P.L Kibet, S. Musyoki, "*Bit Error Rate Simulation at Different Data Rates of a WCDMA System in Multipath Rayleigh Fading Channel*", *The 8th Egerton University International Conference Proceedings,*

held on 26th-28th March, 2014 at the Faculty of Education Complex (Egerton University) Njoro-Kenya, Pg 476-480

6. Filbert O. Ombongi, P.L Kibet, Dr. S. Musyoki, "*Performance Improvement in WCDMA System Subjected to Increased Data Rate and Interference Channels using Compensation for Errors*", *The 8th JKUAT Scientific, Technological and Industrialization Conference Proceedings,* held on 14th-15th Nov 2013 at AICAD (JKUAT), Pg 1122-1129. Available online at http://elearning.jkuat.ac.ke/journals/ojs/index.php/jscp/article/view/1154/956

APPENDIX 2 Q-Function Table

x	Q(x)	x	Q(x)	x	Q(x)
0.00	0.50000000	1.35	0.08850799	2.70	0.00346697
0.05	0.48006119	1.40	0.08075666	2.75	0.00297976
0.10	0.46017216	1.45	0.07352926	2.80	0.00255513
0.15	0.44038231	1.50	0.06680720	2.85	0.00218596
0.20	0.42074029	1.55	0.06057076	2.90	0.00186581
0.25	0.40129367	1.60	0.05479929	2.95	0.00158887
0.30	0.38208858	1.65	0.04947147	3.00	0.00134990
0.35	0.36316935	1.70	0.04456546	3.05	0.00114421
0.40	0.34457826	1.75	0.04005916	3.10	0.00096760
0.45	0.32635522	1.80	0.03593032	3.15	0.00081635
0.50	0.30853754	1.85	0.03215677	3.20	0.00068714
0.55	0.29115969	1.90	0.02871656	3.25	0.00057703
0.60	0.27425312	1.95	0.02558806	3.30	0.00048342
0.65	0.25784611	2.00	0.02275013	3.35	0.00040406
0.70	0.24196365	2.05	0.02018222	3.40	0.00033693
0.75	0.22662735	2.10	0.01786442	3.45	0.00028029
0.80	0.21185540	2.15	0.01577761	3.50	0.00023263
0.85	0.19766254	2.20	0.01390345	3.55	0.00019262
0.90	0.18406013	2.25	0.01222447	3.60	0.00015911
0.95	0.17105613	2.30	0.01072411	3.65	0.00013112
1.00	0.15865525	2.35	0.00938671	3.70	0.00010780
1.05	0.14685906	2.40	0.00819754	3.75	0.00008842
1.10	0.13566606	2.45	0.00714281	3.80	0.00007235
1.15	0.12507194	2.50	0.00620967	3.85	0.00005906
1.20	0.11506967	2.55	0.00538615	3.90	0.00004810
1.25	0.10564977	2.60	0.00466119	3.95	0.00003908
1.30	0.09680048	2.65	0.00402459	4.00	0.00003167

APPENDIX 3 Matlab Codes

```
%%%%% simulation of QPSK modulation%%%%%%
%%%%% Data rate 2Mbps%%%%%%%%%%%%%%%%%%
%%%%% for other modulation schemes the variation is made on the modulator and
demodulator%%%%%%%%%%%%%%%
clear all
close all
M = 4;              % Size of signal constellation for QPSK modulation
k = log2(M);              % Number of bits per symbol (QPSK)
n = 1000000;               % Number of total symbols to simulate
% Generation of Binary signal as a column vector
x = randint(n*k,1);        % Random binary signal
% Display the first 40 random bits
stem(x(1:40),'filled');
title('Random Bits');
xlabel('Number of Bits');
ylabel('Binary Value');
set(gca,'yTick',[0 0.5 1],'XLim',[-2 42],'YLim',[-0.2 1.2],'Box','on');
set(gcf,'Color','w')
% Variables for Convolution Coding
codeRate = 1/2;                         % Code Rate
constlen = 7;                          % Constraint Length
codegen = [171 133];                       % Generator Polynomials
trellis = poly2trellis(constlen, codegen);  % Trellis structure
xEnc = convenc(x, trellis);                % Encode sequence
% Generation of Gray code mapping for QPSK Modulation starting from sqrt(M)
points
grayPsk = bitxor(0:sqrt(M)-1,floor((0:sqrt(M)-1)/2))';
% Display Bit-to-Symbol mapping
% Use it to create a QAM constellation with M points
mapping = repmat(grayPsk,1,sqrt(M))+repmat(sqrt(M)*grayPsk',sqrt(M),1);
mapping = mapping(:);
t = pskmod(mapping,M);
scatterplot(t);
set(get(gca,'Children'),'Marker','d','MarkerFaceColor','auto');
hold on;
% Addition of labels showing the binary sequence at each constellation point
for jj=1:length(t)
   text(real(t(jj))-0.5,imag(t(jj))+0.5,dec2base(jj-1,2,4));
end
set(gca,'yTick',(-(k+1):2:k+1),'xTick',(-(k+1):2:k+1),...
   'XLim',[-(k+1) k+1],'YLim',[-(k+1) k+1],'Box','on',...
   'YGrid','on', 'XGrid','on');
xlabel ('In-Phase');
hold off;
set(gcf,'Color','w')
% Conversion of the sequence from a column vector to a 4-column matrix
```

```
xSym = reshape(xEnc,k,n/codeRate).';
% Conversion of each row of the matrix from binary to decimal for modulation
xSym = bi2de(xSym, 'left-msb');
%Gray code mapping
xSym = mapping(xSym+1);
% Display the first 40 transmitted symbols
figure; stem(xSym(1:40));
title('Random Symbols');
xlabel('Number of Symbols');
ylabel('Integer Value {0..4}');
set(gca,'yTick',(0:M-1),'XLim',[-2 40+2],'YLim',[-0.2 M-
1+.2],'Box','on','YGrid','on');
set(gcf,'Color','w')
% Modulation of the signal with QPSK modulator
xMod = pskmod(xSym,M);
% creation of eye diagram of the modulated symbols
eyediagram(xMod(1:100),2);
subplot (2,1,2)
xlabel ('Time');
set (gcf, 'Color', 'w')
% Design of the pulse shaping filter with the variables as shown
filtOrder = 40; overSamp = 4;
delay = filtOrder/(overSamp*2);
rollOff = .22;
% plot the filter response-magnitude and phase response
rrcFilter = rcosine(1,overSamp,'fir/sqrt',rollOff,delay);
% Display impulse response of Transmit filter
hFV = fvtool(rrcFilter,1,'Analysis','Impulse');
xlabel ('Samples');
set (gcf, 'Color', 'w')
% Upsample signal
yModUp = upsample(xMod,overSamp);
% Pad signal with zeros to flush filter
yModUp = [yModUp; zeros(filtOrder,1)];
% Filter upsampled signal
yTx = filter(rrcFilter,1,yModUp);
% signal plot before and after filtering
figure;
subplot(2,1,1);
stem(real(yModUp(1:120))); hold on;
plot(real(yTx(1+delay*overSamp:120+delay*overSamp)),'r-');
xlabel('Samples'); ylabel('Amplitude');
title ('Filtered Signal for Transmission - Real');
legend ('Unfiltered Digital signal', 'Filtered Analog signal')
subplot(2,1,2);
stem(imag(yModUp(1:120))); hold on;
plot(imag(yTx(1+delay*overSamp:120+delay*overSamp)),'r-');
```

```
xlabel('Samples');
ylabel('Amplitude');
title ('Filtered Signal for Transmission - Imag');
legend ('Unfiltered Digital signal', 'Filtered Analog signal')
hold off;
set (gcf, 'Color', 'w')
% SNR per coded bit which was varied from 0 to 20dB for all the modulation
schemes
EbNo = 2;
% conversion of bit energy to noise power spectral density to signal to noise ratio for
QPSK modulation with convolution coding
EsNo = EbNo+10*log10(k)-10*log10(overSamp)-10*log10(1/codeRate);
% Adding additive white Gaussian noise to the channel
yNoisy = awgn(yTx,EsNo,'measured');
% the received and transmitted signal are plotted on the same curve
h=scatterplot(yNoisy(1:overSamp:end).*sqrt(overSamp),1,0,'b.');
hold on;
scatterplot(xMod,1,0,'rx',h);
set(get(get(h,'children'),'children'),'MarkerSize',10,'LineWidth',4);
title('Received vs. Transmitted Signal (Downsampled for visualization)');
axis([-(k+1) k+1 -(k+1) k+1]);
xlabel ('In-Phase');
hold off;
set (gcf, 'Color', 'w')
% Filration of the received signal
yRx = filter(rrcFilter,1,yNoisy);
% Downsampling the signal
yRxDown = downsample(yRx,overSamp);
% Compensation of the filter delay which was taken as 5 in the transmitter filter
yRxDown = yRxDown(filtOrder/overSamp+1:end);
% the received signal, received analog signal after the RRC, and the resulting digital
signal are plotted
figure;
subplot(2,1,1);
plot(real(yNoisy(delay*overSamp+1:delay*overSamp+120)),'bo-');
hold on;
plot(real(yRx(2*delay*overSamp+1:2*delay*overSamp+120)),'rx-');
grid on;
axis([0 120 -4 4]);
legend('Unfiltered Received signal',' Filtered Received signal');
% the transmitted signal is plotted again for comparison
subplot(2,1,2);
plot(real(yTx(1+delay*overSamp:120+delay*overSamp)),'rd-.');
hold on;
stem(real(yModUp(1:120)),'b');
xlabel('Samples');
ylabel('Amplitude');
```

```
legend(' Filtered Transmitted signal','Original Unfiltered Signal');
axis([0 120 -4 4]);
grid on;
set(gcf,'Color','w')
%signal demodulation
ySym = pskdemod(yRxDown,M);
[dummy demapping] = sort(mapping);
demapping = demapping - 1;
ySym = demapping(ySym+1);
% Converting the signal into bits for determination of BER
yBits = de2bi(ySym,'left-msb');
yBits = reshape(yBits.',numel(yBits),1);
% Design of the Viterbi decoder
tblen = 32;     % Traceback length
%Quantizing the signal for soft decision decoding
qcode=quantiz(yBits,[0.001,0.1,0.3,0.5,0.7,0.9,0.999]);
% the signal is decoded with the assumption of all-zero initial state
y = vitdec(qcode, trellis, tblen, 'cont','soft',3);
% computation of the BER without convolution coding and the result displayed on
the command window
[numErrors_Sym_no_code, bitError_Sym_no_code] = biterr(xEnc,yBits)
% Computation of the BER with convolutional coding and the result sent to
command window
[numErrors_with_code,bitError_with_code] = biterr(x(1:end-tblen),y(tblen+1:end))
```

APPENDIX 4 WCDMA Simulation Model