İSTANBUL TECHNICAL UNIVERSITY ★ INSTITUTE OF SCIENCE AND TECHNOLOGY

ANALYSIS OF PARAMETERS IN LONG TERM EVOLUTION (LTE) TECHNOLOGY

M.Sc. Thesis by
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Department: **Electronics and Communication Engineering**

Programme: **Telecommunication Engineering**

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İSTANBUL TEKNİK ÜNİVERSİTESİ ★ FEN BİLİMLERİ ENSTİTÜSÜ

LTE TEKNOLOJİSİNDEKİ PARAMETRELERİN ANALİZİ

YÜKSEK LİSANS TEZİ

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FOREWORD

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Volkan ÇINAR

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ABBREVIATIONS

AAA	: Authentication, Authorization and Accounting
AGW	: Access Gateway
AP	: Access Point
AMPS	: Advanced Mobile Phone System
ASN-GW	: Access Service Network Gateway
BLER	: Block Error Rate
BSC	: Base Station Controllers
BST	: Base Station Transceivers
CAN	: Converged Access Network
CDMA	: Code Division Multiple Access
CRC	: Cyclic Redundancy Check
CSN	: Connectivity Service Network
CQI	: Channel Quality Indicator
DFT	: Discrete Fourier transforms
eBS	: Evolved Base Station
EDGE	: Enhanced Data Rate for Global Evolution
ePDIF	: Evolved Packet Data Interworking Function
FDD	: Frequency Division Duplex
FDMA	: Frequency Division Multiple Access
FEC	: Forward Error Correction
FFT	: Fast Fourier Transform
GSM	: Global System for Mobile communication
GPRS	: General Packet Radio System
HA	: Home Agent
HARQ	: Hybrid Automatic Repeat Request
HSDPA	: High Speed Downlink Packet Access
ICI	: Inter Carrier Interference
IMT	: International Mobile Telecommunication
IFFT	: Inverse Fast Fourier Transform
ISI	: Inter Symbol Interference
ITU	: International Telecommunication Union
LMA	: Local Mobility Anchor
LTE	: Long Term Evolution
MAC	: Media Access Control
MIMO	: Multiple Input Multiple Output
NAP	: Network Access Provider
NMT	: Nordic Mobile Telephone
NSP	: Network Service Provide
OFDM	: Orthogonal Frequency Division Multiplexing
OFDMA	: Orthogonal Frequency Division Multiple Access
PDSN	: Packet Data Serving Node
PCRF	: Policy and Charging Rules Function
PSK	: Phase Shift Keying

RBs : Resource Blocks
SAE : System Architecture Evolution
SC-FDMA : Single Carrier Frequency Division Multiple Access
SDMA : Space Division Multiple Access
SGSN : Serving GPRS Support Node
SNR : Signal to Noise Ratio
SRNC : Session Reference Network Controller
QAM : Quadrature Amplitude Modulation
QoS : Quality of Service
QPSK : Quadrate Phase Shift Keying
TB : Transport Block
TDMA : Time Division Multiple Access
TTI : Transmission Time Interval
UE : User Equipment
UMTS : Universal Mobile Telecommunication System
UMB : Ultra Mobile Broadband
WiMAX : Worldwide Interoperability Microwave Access
3GPP : Third Generation Part Project

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ANALYSIS OF PARAMETERS IN LONG TERM EVOLUTION (LTE) TECHNOLOGY

SUMMARY

Digital multimedia applications create increasing improvement demand for wireless communication systems. The requirements are so high that the current technologies are no longer feasible.

The Long Term Evolution (LTE) is one of the 4G candidate systems that challenges the disadvantages of wireless environment. Besides providing higher transmission rate also 4G aims to provide a common platform for all current technology such as 2G, 3G and fixed wired network for seamlessly internetworking with each other. The 4G offers better security, cost effective infrastructure, higher transmission rate and seamless roaming between different networks in guaranteed high speed data rate.

In this study, LTE which are candidates to provide technical requirements of 4G (Fourth Generation) networks are analyzed. The technical specifications for 4G are included in IMT Advanced which is announced by ITU (International Telecommunication Union). It is explained that LTE could not comply full requirement for 4G and it is expected LTE advanced will provide full requirements.

In this study, the purpose was to achieve simulation of LTE network to analyze the performance by using throughput and BLER findings. Before performing the simulation in LTE, the other competing technologies are also will be researched and they will be compared to LTE. These are UMB and WiMAX and they are explained briefly. Actually all these are using MIMO and OFDMA techniques so that Analyzing of LTE could be used to have an idea about 4G networks.

In this study, technical specifications which are used by telecommunication provider in TS36.942 are used then LTE network is simulated in Matlab program according to these parameters. In simulation, the bandwidth, antenna techniques and channel types are set as variables and the system was simulated according to different bandwidth, channel types and multiple antennas. Then BLER and throughput are analyzed for different situation. The maximum throughput (approximately 60 Mbps) could be obtained in AWGN channel when the bandwidth is 20MHz and BLER is smaller than 10% which was acceptable. The results were also calculated according to Shannon capacity formula (approximately 70 Mbps) and it was seen that expected throughput for AWGN was obtained in simulator. In contrast to AWGN channel high throughput could not be obtained in other channel types which are Pedestrain and Rayleigh fading channell. It is thought that the following parameters decrease the throughput in these two channel types. Multiple Users (2) were tested in other two channel types that decreases throughput. Mobility and the characteristic of channel types also decreased throughput.

In simulation, LTE recommended high data rate which is near to the target of 4G defined by ITU and also currently used LTE networks offer speeds in mobile

broadband services between 20-80 Mbit/s with a maximum speed of 100 Mbit/s for 20 MHz bandwidth in 2.5GHz frequency so it could be expected that LTE Advanced will comply full requirements of IMT-Advanced systems.

In conclusion, it is thought that LTE has become technology platform for 4G when it is compared to other competing technology, WiMAX. ITU (International Telecommunication Union) also confirmed LTE-Advanced as 4G in November 2010. Currently, nine telecommunication providers have already launched LTE in 2010 and an additional 11 expected before the end of the year. More than 250 companies have publicly expressed interest in deploying LTE networks, including CDMA, WiMAX and GSM operators. So that LTE is expected to be the leading choice for next-generation OFDMA networks over the next decade for all wireless carriers.

This thesis focuses on three technology approaches which are candidates to provide technical requirement of 4G. These are UMB, WiMAX and LTE. The development of UMB was stopped in 2008 and it was announced that LTE will be adopted for 4G deployment instead of UMB. Especially it is expected that LTE Advanced will provide full requirement for 4G deployment. All these technology are explained briefly but especially thesis focuses on LTE. Actually network architecture is basically the same for all standards and also the multiple access technique (OFDMA, SC-FDMA) and parameters such as bandwith, MIMO Antenna Array are similar for all standard.

At the beginning of the thesis, evolution of mobile communication is briefly mentioned and then network architecture, network elements and multiple access techniques are explained respectively. Finally 4G is simulated according to system parameters such as bandwith (5, 20 MHz), Antenna (1x1, 2x1, 4x2), Channel Type (Pedestrian, Flat Rayleigh). Then this simulation is analyzed according to BLER and Throughput.

LTE TEKNOLOJİSİNDEKİ PARAMETRELERİN ANALİZİ

ÖZET

Sayısal çoklu ortam uygulamaları kablosuz iletişimde daha da iyileştirme ihtiyacını yarattı. İhtiyaç o kadar fazladır ki şu anki teknolojiler artık ihtiyacı karşılayabilir gözükmemektedir.

LTE kablosuz ortamın dezavantajlarına meydan okuyan en yeni 4.nesil aday teknolojilerinden birisidir. 4. nesil yüksek iletim hızları sağlamak yanında şu anda kullanılan 2.nesil, 3.nesil ve kablolu ağların arasında sorunsuz geçiş için ortak bir platform sağlamayı hedeflemektedir.

4. nesil daha iyi bilgi güvenliği, düşük maliyetli altyapı, daha yüksek iletim hızı ve yüksek hızı garantileyen farklı ağlarda dolaşım olanağı önermektedir.

Bu çalışmada, 4. nesil iletişimde kullanılmaya aday LTE teknolojisi incelenmiştir. 4. nesil iletişimin teknik spesifikasyonları ITU tarafından IMT-Advanced sistemleri olarak belirlenmiştir. LTE'nin tam olarak IMT-Advanced sistemlerinin ihtiyaçlarını karşılaması için LTE Advanced olarak 3GPP tarafından tekrar düzenlenmiştir.

Bu çalışma da amaç LTE' nin benzetiminin gerçeklenerek veri hız ve hata bit oranlarıyla performans analizini yapmaktır. Benzetim gerçekleştirilmeden önce diğer 4. nesle aday WiMAX ve UMB gibi teknolojiler de kısaca anlatılmıştır. Aslında 4. nesil iletişimde kullanılmaya aday diğer WiMAX ve UMB teknolojilerinde de aynı çoklu anten tekniği (MIMO) ve OFDMA gibi aynı çoklu erişim tekniği kullanılmaktadır. Bu nedenle LTE teknolojisinin analizi sonrasında 4.nesil iletişim için genel bir fikir sahibi olmak mümkündür.

Bu çalışmada LTE teknolojisinin şu anda telekomünikasyon servis sağlayıcılarının da kullandığı TS36.942 de yer alan parametreler kullanılarak Matlab programında benzetimi gerçekleştirilmiştir. LTE ve LTE Advanced temel olarak aynı yapıyı içerdiğinden. LTE sisteminin analizinin LTE Advanced hakkında da fikir sağlayacağı düşünülmüştür. Benzetimde band genişliği, anten sayısı ve kanal yapısı gibi değişkenler kullanılıp bit hata oranı ve veri hızı grafikleri çıkartılmıştır. Sonuç olarak kanal yapısının AWGN ve band genişliğinin 20 MHz seçildiği durumlarda en yüksek veri hızı değerlerine kabul edilebilir bit hata oranlarında (%10) ulaşılmıştır. Ulaşılan veri hızı değerleri Shannon'un sistem kapasite formülü ile hesaplanıp(yaklaşık olarak 70 Mbps) karşılaştırma yapılmıştır. Böylece benzetimde bulunan throughput değerlerinin teorik olarak ta yeterli yakınsaklığı sağladığı gözlenmiştir. Diğer kanal yapılarında kullanıcıların hareketli olduğu (Pedestrain) veya kanalın Rayleigh sönümlenmesine uğradığı durumlarda AWGN kanal yapısı kadar yüksek throughput değerleri elde edilememiştir. Bunun nedeni bu kanal tiplerinde çoklu kullanıcı yer alması ve kanal karakteristiklerinin veri hızını azaltması olarak düşünülmüştür.

Servis sağlayıcılarının hali hazırda kullandığı LTE teknolojisi 20 MHz band genişliği ve aynı çoklu anten yapısında elde ettiği throughput değerleri 20 ile 80 Mbps olduğu düşünüldüğünde LTE Advanced teknolojisinin tam olarak IMT-Advanced

sistemlerinin ihtiyacı olan 100 Mbps throughput değerlerini karşılayabileceği düşünülmektedir.

Bu tez 4. nesilin ihtiyaçlarını karşılamaya aday teknolojilere odaklanmıştır. Bunlar UMB, WiMAX ve LTE’dir. Bunlardan UMB teknolojisinin gelişimi 2008 yılında durdurulmuş ve UMB yerine LTE 4. nesil için kabul edilmiştir. Özellikle LTE Advanced 'in 4. nesilin tüm ihtiyaçlarını karşılayabileceği tahmin edilmektedir.

Bütün bu teknolojiler kısaca açıklanacaktır fakat özellikle LTE üzerinde odaklanılmıştır. Aslında ağ mimarisi, çoklu erişim teknikleri (OFDMA ve SC-FDMA) ve band genişliği, MIMO anten dizileri gibi parametreler bütün bu teknolojiler için temel olarak aynı veya benzerdir.

Tezin başlangıcında mobil iletişimin evriminden kısaca bahsedilmiş ve daha sonra ağ mimarisi, ağ elemanları ve çoklu erişim teknikleri sırasıyla açıklanmıştır. Son olarak LTE band genişiliği, çoklu anten seçimi (1x1. 2x1, 4x2), kanal tipi (yaya, Flat Rayleigh) gibi parametrelere göre benzetimi yapılmış. Sonra bu benzetimin blokbit hata oranı ve gerçek veri hızına göre analizi yapılmıştır.

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1. INTRODUCTION TO MOBILE COMMUNICATION

Multimedia is effectively an infrastructure technology with widely different origins in computing, telecommunications, entertainment and publishing. New applications are emerging, not just in the wired environment, but also in the mobile one.

Digital multimedia applications as they are getting common lately create an ever increasing demand for wireless communication systems. The requirements are so high that the standard solutions are no longer feasible or lead to sub optimal results.

This thesis discusses possible ways to enable multimedia communication better in the mobile environment. The 4G is the latest candidate in emerging wireless technology to satisfy a requirement for mobile requirement. Long Term Evolution (LTE) is accepted as brand name for 4^{th} Generation. In the following section both LTE and 4G can be used interchangeably.

1.1 Evolution of Mobile Communication to 4G

1.1.1 1G

The first target of wireless technology is to provide communication with anyone with anytime from anywhere. If we start to use the term 'mobile communication' in wireless technology, it was December 1979 when the first public communication was started in Japan.

The first generation (1G) system was based on analog transmission technique primarily used for voice. In 1G system, the voice signal is transmitted using a form of analog modulation. The standard used in 1G is called as Advanced Mobile Phone System (AMPS) in USA and Nordic Mobile Telephone (NMT) in Europe. These systems used as phone service based on analog technology and provided bandwidth from 16 KHz to 30 KHz. The mechanism is to initialize the channel after searching and choosing the strongest one by the control channel. The poor voice quality and unprotected communication were main gap in this old technology.

1.1.2 2G

Digital cellular systems were first developed at the end of the 1980s. In the 1990s the 'second generation' (2G) mobile phone systems emerged, primarily using the GSM standard. In 1991 the first GSM network launched in Finland. In Europe GSM (Global System for Mobile communication) was the first commercially operated digital cellular system which is based on Time Division Multiple Access (TDMA). In America the IS-54 standard was deployed in the same band as AMPS which is based on Code Division Multiple Access (CDMA) and it displaced some of the existing analog channels.

These 2G phone systems differed from the previous generation in their use of digital transmission instead of analog transmission. These systems digitized not only the control link but also the voice signal. The new system provided better quality and higher capacity at lower cost to consumers.

TDMA and CDMA are multiple access techniques which are used in 2G systems. TDMA divides the available spectrum into multiple time slots, by giving each user a time slot in which they can transmit or receive. However CDMA is a spread spectrum technique that uses neither frequency channels nor time slots. In CDMA, the narrow band message (typically digitized voice data) is multiplied by a large bandwidth signal which is a pseudo random noise code (PN code). All users in a CDMA system use the same frequency band and transmit simultaneously. The transmitted signal is recovered by correlating the received signal with the PN code used by the transmitter. The architecture of the GSM is depicted in the Figure 1.1. The data rate of GSM is 9.6 kbps and 14.4 kbps for GSM900 and GSM1800.

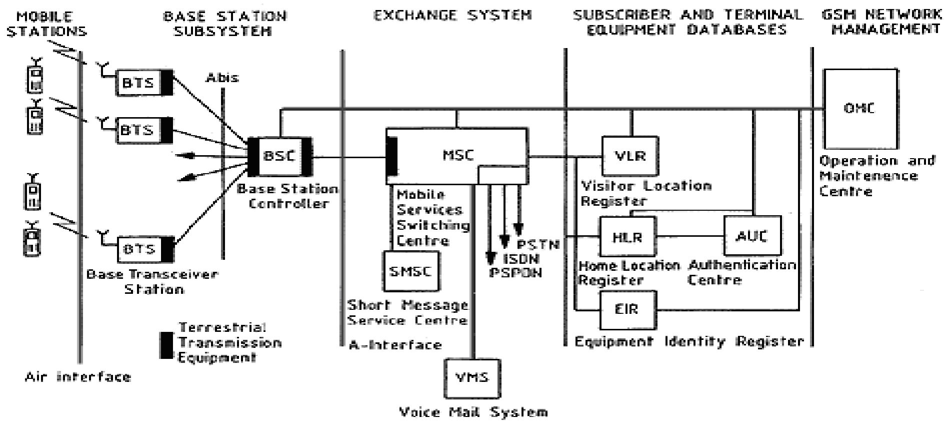

Figure 1.1: Basic GSM Architecture

1.1.3 2.5G

The 2G still provides less speed which can not support user needs so that 2.5G was emerged as a result of user needs. GPRS technology is used in 2.5G. GPRS stands for General Packet Radio System. GPRS provides packet radio access for GSM and time-division multiple access (TDMA) users. GPRS is a data network that overlays a second-generation GSM network. Multiple users can share the same air-interface resources simultaneously. It provides moderate-speed data transfer by using unused time division multiple access (TDMA) channels in. It provides packet data transport at rates from 9.6 to 171 kbps.

GPRS is important as a migration step toward next-generation networks and allows network operators to implement IP-based core architecture for data applications, which will continue to be used and expanded for integrated voice and data applications.

1.1.4 2.75G AND 3G

Data rate which is supported by GPRS is also not enough for multimedia services. Hence Enhanced Data Rate for Global Evolution (EDGE) was developed. This system referred as 2.75G which provides data rate up to 384 Kbps. EDGE (also known as EGPRS) is a superset to GPRS and it can function on any network which GPRS is working on. EDGE provides that the carrier implements the necessary upgrades. EDGE is a technology that gives GSM the capacity to handle services for the third generation of mobile telephony. EDGE provides three times the data capacity of GPRS.

International Mobile Telecommunications-2000 (IMT—2000), better known as 3G or 3rd Generation, is a generation of standards for mobile phones and mobile telecommunications services fulfilling specifications by the International Telecommunication Union (ITU). The Universal Mobile Telecommunication System (UMTS) is 3G standard which is regulated by 3GPP. The first 3G network was launched in Japan. WCDMA and CDMA2000 are used as multiple access techniques. ITU has not provided a clear definition of the data rate users can expect from 3G equipment or providers. While stating in commentary that it is expected that IMT-2000 will provide higher transmission rates; a minimum data rate of 2 Mbit/s for stationary or walking users, and 384 Kbit/s in a moving vehicle. The ITU does not actually clearly specify minimum or average rates or what modes of the interfaces qualify as 3G. 3G networks offer greater security than their 2G predecessors. By allowing the UE (User Equipment) to authenticate the network it is attaching to, the user can be sure the network is the intended one and not an impersonator.

The HSPA (High Speed Packet Access) is the further enhancement in 3G systems that extends and improves the performance of existing WCDMA techniques. A further standard, Evolved HSPA (also known as HSPA+), was released in 2008 with subsequent adoption worldwide beginning in 2010. HSPA supports increased peak data rates of up to 14 Mbit/s in the downlink and 5.8 Mbit/s in the uplink. It also reduces latency and provides up to five times more system capacity in the downlink and up to twice as much system capacity in the uplink, reducing the production cost per bit compared to original WCDMA protocols. HSPA increases peak data rates and capacity in following ways; shared channel transmission, shorter transmission time interval (TTI), link adaptation, fast scheduling, fast retransmission and 16QAM (Quadrature Amplitude Modulation) channel coding [1].

Evolved HSPA (also known as HSPA+) is a wireless broadband standard defined in 3GPP release 7 and 8 of the WCDMA specification. Evolved HSPA provides data rates up to 42 Mbit/s in the downlink and 11 Mbit/s in the uplink (per 5 MHz carrier) with multiple input, multiple output (MIMO) technologies and higher order modulation [2].

Thus far the technologies in the evolution of mobile communication which is depicted in Figure 1.2 were briefly mentioned [3].

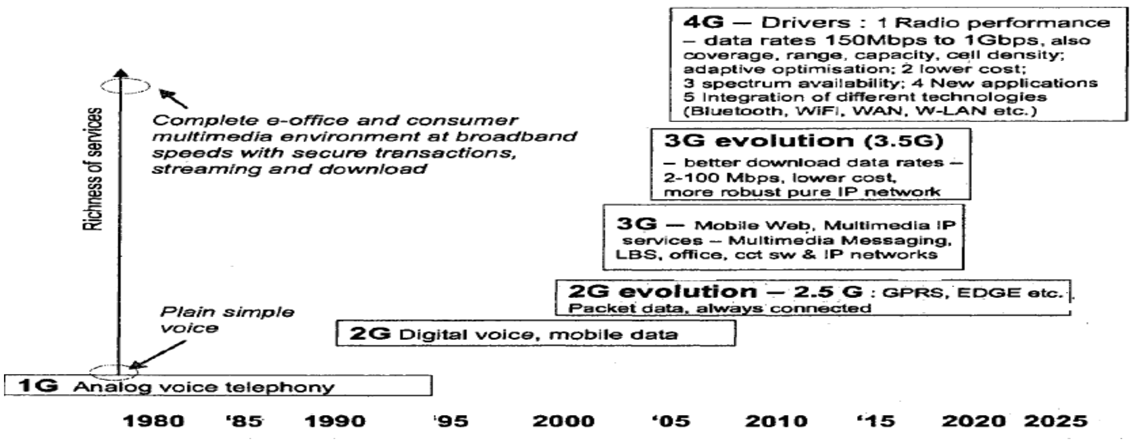

Figure 1.2: Evolution of Mobile Communication

2. INTRODUCTION TO 4G

2.1 Overview

The 4G system will be a combination of various technologies and networks that provides user connectivity at any place and at any time with high data transfer rate so that the network for 4G is called as heterogeneous network. The aim is to provide 100 Mbps data rate between two users anywhere in the world. The main function of 4G will be the integration of present technology to work with each other with new interface. To perform this integration, Quality of Service (QoS), security and mobility management must be taken into consideration. The ultimate goal is to provide a mobile network with a single worldwide standard

The followings are defined as objects of the 4G wireless communication standard. These objectives are determined by International Telecommunication Union which is included in International Mobile Telecommunications-Advanced (IMT-Advanced) systems. These objects are flexibility in channel bandwidth between 5 MHz and 40 MHz, nominal data rate for client who is in relatively moving position is 100 Mbit/s at least and 1 Gbit/s if client are relatively in fixed position, smooth handoff across heterogeneous networks, seamless mobility and roaming between networks, interoperability with existing wireless standards, high quality of service for next generation multimedia support, full IP Model which is packet switch network. (IPv6 based and also compatible with IPv4), frequency band is planned between 2- 8 GHz [4].

There are 3 technologies defined as 4G predecessors which are Long Term Evolution (LTE), Worldwide Interoperability for Microwave Access (WiMAX) and Ultra Mobile Broadband (UMB) [3].

2.2 4G Predecessors and Candidate Systems

UMB was the brand name for a discontinued 4G project within the 3GPP2 standardization group to improve the CDMA2000 mobile phone standard. It was announced that the development of this technology was ended and it will be focused on LTE instead.

WiMAX provides 128 Mbit/s for downlink and 56 Mbit/s for downlink over 20 MHz wide channels but it can not exactly comply IMT-Advanced criteria so that the IEEE 802.16m is under development to provide 1 Gbit/s for stationary clients and 100 Mbit/s for mobile clients.

LTE is generally named as 4G but it does not fulfill all requirements of IMT-Advanced systems. It provides 100 Mbit/s in downlink and 50 Mbit/s in uplink over 20 MHz channels. Now LTE Advanced is a candidate for IMT Advanced standard to comply full requirements. Table 2.1 shows the comparison between LTE, UMB and WiMAX [3].

Table 2.1: Comparison between LTE, UMB and WiMAX

Parameters/Technology	Mobile WiMAX 802.16e	LTE	UMB
Availability	2008	2010	2010
Standard developed By	IEEE and WiMAX forum	3GPP	3GPP2
Duplexing	TDD/FDD	FDD,TDD	FDD,TDD
Data Rate	10 Mbps in UL @10MHz 70 Mbps in DL @10MHz	75 Mbps in UL @20MHz 275 Mbps in DL @20MHz	75 Mbps in UL @20MHz 275 Mbps in DL @20MHz
Type of spectrum	Licensed	Licensed	Licensed
Channel Bandwidth	5,10 MHz	1.4, 3, 5, 15, 10, 20 MHz	1.25, 2.5,3.5, 5, 10 to 20MHz
Handoff Latency	<30ms	<50ms	<50ms
Core Architecture Technology	All-IP based network	All-IP based network	All-IP based network
Application layer	IMS	IMS	IMS
Downlink Multiple Access	OFDMA	OFDMA	OFDMA
Uplink Multiple Access	OFDMA	SC-FDMA	OFDMA

In thesis the WiMAX and UMB will be explained in the following section but thesis will mainly focus on LTE and LTE-Advanced. They will be explained in detail in the rest of the thesis.

2.2.1 Ultra Mobile Broadband (UMB)

Ultra Mobile Broadband technology will be explained briefly. Since November 2008, UMB's lead sponsor announced it was terminating development of the technology. It is explained that they will focus on development of LTE instead of UMB [3].

The UMB used OFDMA solution with MIMO in downlink connectivity and space division multiple access in uplink connectivity. The UMB technology provided high diversity due to use of advanced antenna techniques and proper mobility management. The main advantage of UMB is to provide seamless mobility and handoff and it supported CDMA2000 technology. The UMB was based on all IP-based network architecture and it simplified the interface between the network elements and provided interoperability of different equipments [5].

In summary, UMB was developed to achieve the following standard; data rates of over 275 Mbit/s in the downlink (base station to mobile) and over 75 Mbit/s in the uplink (mobile to base station), OFDM / OFDMA, frequency division duplex (FDD) would be used, full IP network architecture and it had a scalable bandwidth between 1.25- 20 MHz, it supported flat, mixed and distributed network architectures.

The UMB network which is depicted in Figure 2.1 was based on Converged Access Network (CAN) architecture which does not require a centralized entity such as a BSC in the GSM architecture. An eBS has functions of a traditional BS, BSC, and some functions of the packet data serving node (PDSN) into a single node. An eBS can be connected directly to the internet whereas in legacy networks, the BS, BSC, PDSN and mobile IP home agent all cooperate to serve user traffic [3].

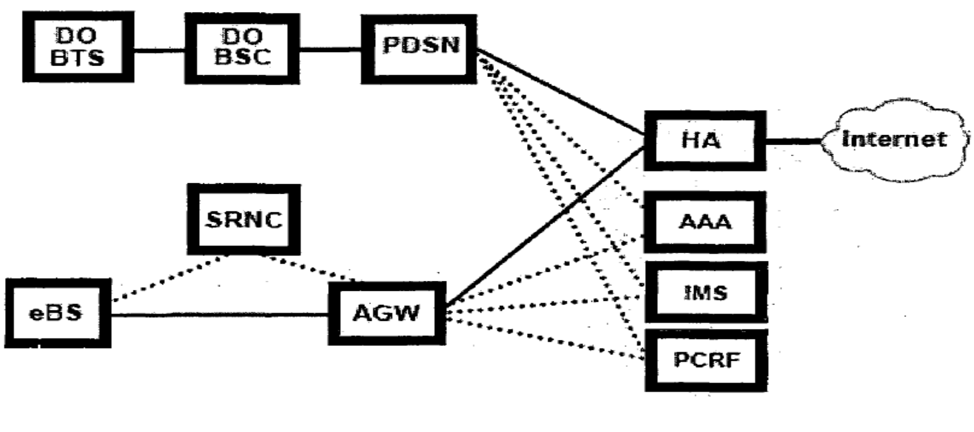

Figure 2.1: UMB Network

The followings are elements of UMB Network:

Access Terminal (AT): It is subscriber device which can be PDA, mobile phone or mobile enabled laptop.

Evolved Base Station (eBS): It provides over the air signaling and user data transport that is used by the AT for connectivity to the radio access network.

Access Gateway (AGW): It is an entity that provides the user's IP connectivity to the network. The AGW behaves like in effect the first hop router for the mobile terminal.

Session Reference Network Controller (SRNC): It is responsible for supporting idle state management of the AT and for providing paging control functions when the AT is idle.

Authentication, Authorization and Accounting Function (AAA): According to the AT's use of the network resource, functional data is provided by AAA.

Home Agent (HA): It is used to provide a mobility solution to the AT in a 3GPP2 packet data network.

Local Mobility Anchor (LMA): It is the home agent for the mobile node in the Proxy Mobile IPv6 domain.

The evolved Packet Data Interworking Function (ePDIF): It is an interworking function for connectivity between a 3GPP2 network and an untrusted non 3GPP2 network which can be Wi-Fi and Access Point (AP).

Packet Data Serving Node (PDSN): The PDSN is the node that provides the user's point of IP connectivity in the legacy packet data network.

Policy and Charging Rules Function (PCRF): PCRF provides service based bearer control rules to the AGW for the following purpose for detection of a packet belonging to a service data flow, for providing policy control for a service data flow, and for providing applicable charging parameters for a service data flow.

In conclusion, UMB was the leading OFDMA solution, used sophisticated control and signaling mechanisms, radio resource management, adaptive reverse link interference management, and advanced antenna techniques, such as MIMO, SDMA and beamforming. Also, the UMB can achieve seamless and fast handoffs, whether they are inter-eBS, inter-AGW or inter-system transitions for the AT, all while minimizing overhead. The UMB air interface and architecture was designed to provide QoS for a wide range of applications including those which requires extremely low latencies, low jitter and increased spectral efficiencies [6].

In November 2008, UMB's lead sponsor announced that it was terminating development of the technology [6].

2.2.2 WiMAX

WiMAX forum based on IEEE 802.16e-2005 developed WiMAX which stands for Worldwide Interoperability Microwave Access. The main advantage of WiMAX is to have a capability for deploying in large spectrum above 10 MHz by providing high bandwidth rate. It also provides delivering high bandwidth at low cost. The WiMAX supports the various applications from real time voice applications to real time data applications with great coverage of areas and excellent QoS at very low cost [7].

There are 4 logical parts in WiMAX network which is depicted in Figure 2.2. These are Mobile Station, Network Access Provider, Network Service Provider and Internet.

Mobile Station (MS): User mobile devices like mobile phone, laptop or pda.

Network Access Provider (NAP): NAP provides radio access functionality between MS and Base Station (BS), Access Service Network Gateway (ASN-GW) and BS. **Access Service Network (ASN):** The ASN includes base station and access gateway which allows entry for mobile station into a WiMAX network, it also provides functional use for providing radio access to the MS. It provides 802.16 layer 2 interfaces to the mobile station during entry and handover. It performs selection of the Network Service Provider (NSP) for the subscriber according to their preference. It also transfers the AAA messages to the home agent located at NSP in order to generate a subscriber session. The mobile station is connected to the Connectivity Service Network (CSN) through the ASN gateway. The ASN-GW acts as a foreign agent and authenticates the user using authentication mechanism. The ASN-GW sends access request message to the AAA (Authentication, Authorization and Accounting) home agent located in CSN and then home agent sends access accept messages and session ID to the ASN-GW.

Connectivity Service Network (CSN): The CSN provides IP connectivity services to the subscribers. A CSN includes many network elements which are AAA (Authentication, Authorization and Accounting) proxy/servers, routers, user databases and gateway. The CSN also provides per user policy management of QoS and security. The CSN is also responsible for IP address management, support for roaming between different NSPs, location management between ASNs, and mobility and roaming between ASNs [3].

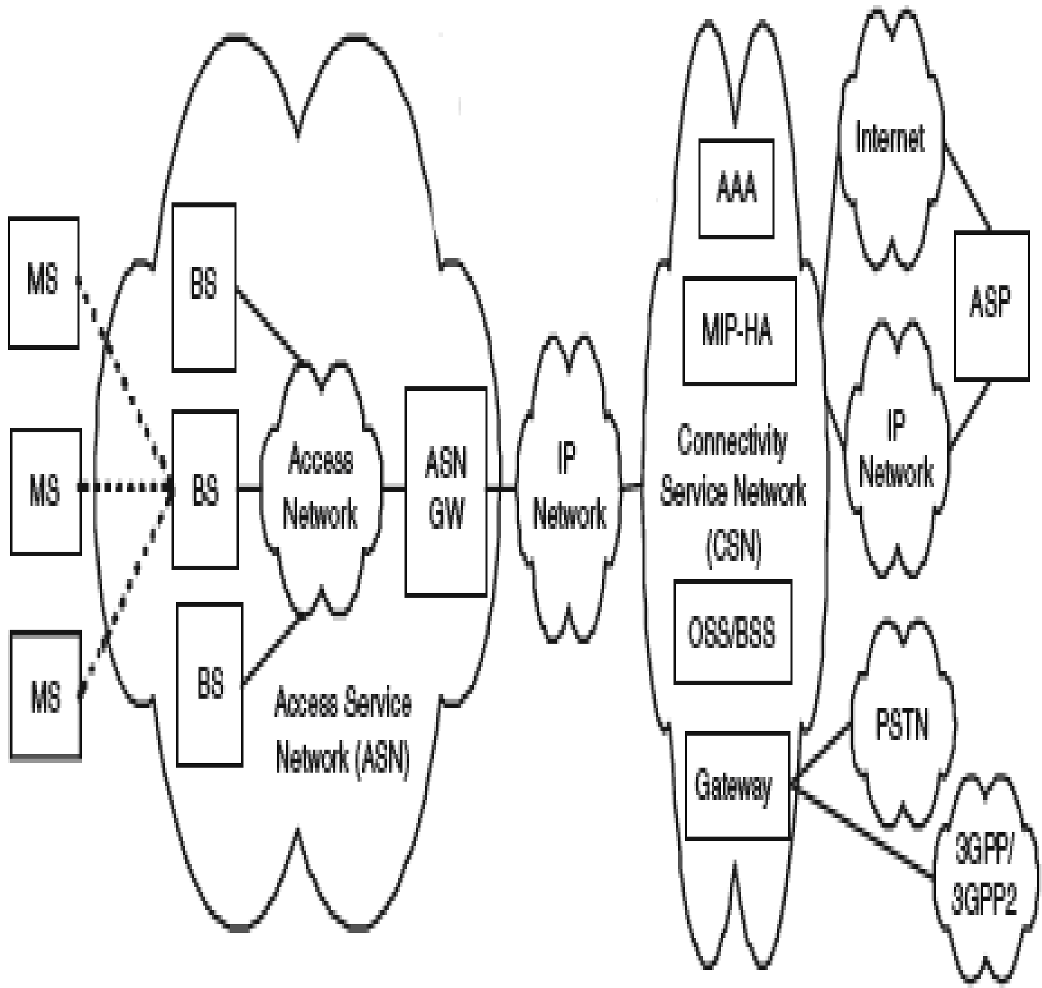

Figure 2.2: WiMAX Network

The WiMAX network framework allows for the flexible decomposition and/or combination of functional entities when building the physical entities. For example, the ASN may be decomposed into base station transceivers (BST), base station controllers (BSC), and an ASNGW analogous to the GSM model of BTS, BSC, and Serving GPRS Support Node (SGSN).

2.2.3 3GPP Long Term Evolution (LTE) and LTE Advanced

The 3GPP Long Term Evolution (LTE) is often used instead of 4G but the first LTE release does not exactly support the IMT-Advanced requirements. LTE Advanced (Long Term Evolution Advanced) is a candidate for IMT-Advanced standard, formally submitted by the 3GPP organization to ITU-T in the fall 2009, and expected to be released in 2011. The target of 3GPP LTE Advanced is to reach and surpass the ITU requirements. LTE Advanced should also be compatible with first release LTE equipment, and should share frequency bands with first release LTE. First set of 3GPP requirements on LTE has been approved in June 2008. LTE Advanced is standardized in 2010 as part of the Release 10 of the 3GPP specification. LTE Advanced is fully built on the existing LTE specification Release 10 and is not defined as a new specification series [8, 9]. The followings are targets for LTE; significantly increased peak data rates, improved spectrum efficiency, improved latency, scalable bandwidth, acceptable system and terminal complexity, cost and power consumption and compatibility with other systems [4]. The targets for LTE and LTE Advanced are basically the same but LTE-Advanced will extend performance through more powerful multi-antenna capabilities. For the downlink, the technology will be able to transmit in up to eight layers using an 8X8 configuration for a peak spectral efficiency of 30 bps/Hz that exceeds the IMT-Advanced requirements, conceivably supporting a peak rate of 1 Gbps in just 40 MHz and even higher rates in wider bandwidths. This would require additional reference signals for channel estimation and for measurements such as channel quality to enable adaptive, multi-antenna transmission. LTE-Advanced will also include four-layer transmission in the uplink resulting in spectral efficiency exceeding 15 bps/Hz.

The multiple access techniques used in design of LTE are OFDMA on downlink and SC-FDMA on uplink. In theory net bit rate for LTE is up to 100 Mbps for downlink and 50 Mbps for uplink if the channel used is 20 MHz. In LTE Advanced, downlink is up to 1 Gbps and uplink is up to 500 Mbps.

The LTE is a pure packet core network which is depicted in Figure 2.3 having two nodes, the LTE base station called eNodeB and the System Architecture Evolution (SAE) Gateway. The SAE have two main functionalities which are packet data network (PDN) and serving gateway. The PDN acts as the access point for all access technologies and provides stable IP connectivity for users during mobility between the access technologies. The S1, core network-RAN interface is used to connect LTE BS to core network. All control signaling are handled by entity nodes separated from the gateway which makes the network fully flexible. In LTE network, the interface based on diameter is adopted for connection between home subscriber server and packet core. The network signaling for policy control and charging is also based on diameter.

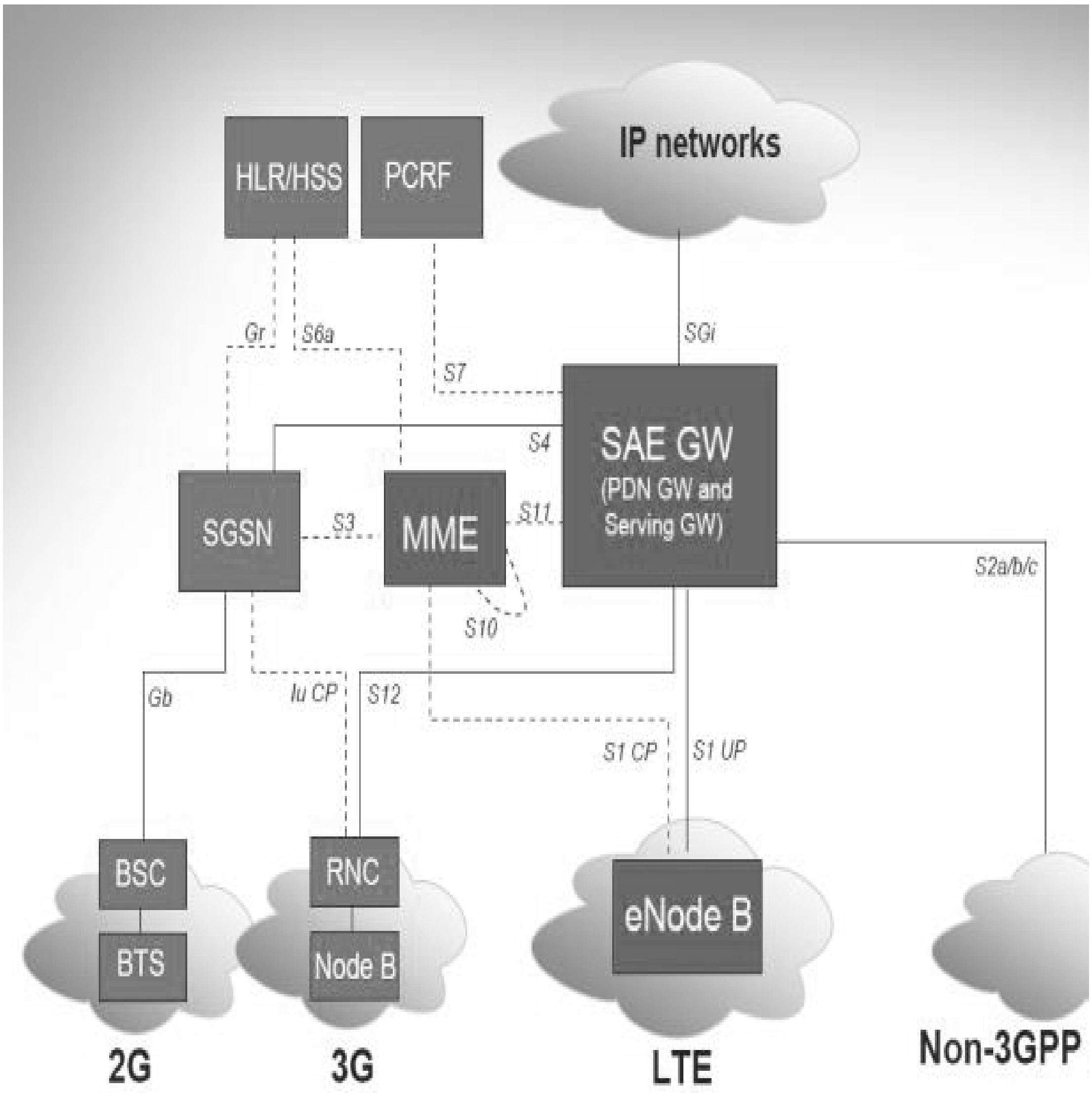

Figure 2.3: LTE Network

3. 4G ARCHITECTURE

The 4G is a heterogeneous network which combines wireless access technologies into a one network and also supports connectivity access to the application anywhere. The QoS provisioning and security support to the users are the main challenges in development of heterogeneous infrastructure. The 4G systems overcome these challenges and provide excellent QoS and security to their users.

The main goal of the 4G systems is to achieve seamless mobility in a 4G infrastructure by combining all wireless technologies and mobile network in a heterogeneous network. The 4G is going to be IP based technology integrated with different access technologies so there will be different types of data transmission and data protection technologies integrated to support different types of technologies. The possibility of variety of approaches towards quality of assurance also varies. Also architectural considerations in IP layer become so critical that it becomes difficult to handle different data transmission and data protection technologies. Therefore a basic model for 4G networks should be developed in which QoS, mobility and security are treated as three different aspects.

In 4G network the main goal is to access the service at anytime and anywhere basis in a transparent way. The 4G terminal should have ability to choose proper access technology at current location and move to different technology seamlessly. The 4G technology will use IPv6 technology as common platform for transferring data packets over IP protocol. To provide following functionality like QoS, mobility, Authentication, Authorization and Accounting (AAA) and charging is a real tough challenge for 4G providers [3].

Targets in designing the basic network architecture elements; providing a real time service with excellent quality and providing uninterrupted service at the time of handoff.

The basic network elements in 4G are mobile end user systems and access router. Network management server handles AAA service and mobility management service.

3.1 Network Elements of 4G

The main elements in 4G network which are depicted in Figure 3.1 are mobile terminal (MT), access router (AR), QoS broker and AAA server.

Mobile Terminal (MT); the MT is the user terminal from which the user accesses the services provided by subscriber.

Access Router (AR); the AR is the point of attachment between MT and the core network. The QoS broker is responsible for providing the user access to the network with the help of information provided by AAA server.

AAA Server; it provides the network management services like authentication, authorization, accounting and security. AAA server provides a functionality to support the functions like authentication, authorization, and accounting. It uses RADIUS (Remote Authentication Dial-In User Service) protocol for communication between user and network service. The RADIUS is the protocol that is used as a centralized management service which handles and manages access, authorization and accounting functions for users which try to connect network services.

In 4G, the AAA server uses diameter protocol which is successor of RADIUS. The AAA server communicates with service provider through application specific modules (ASM) using application program interface or protocol based interface. The ASM manages the resources and provides authorization decision to access services. The ASM technology is either integrated in AAA server or it can be external entity. When user wants to access the services he/she has to send a request to AAA server and then AAA server with help ASM verifies the user identity and authenticates user to use that services.

The DIAMETER is protocol between access client and access server use to communicate between them. The user credential is send over DIAMETER protocol to AAA server. The AAA server authenticates and authorizes the user to use the application at access server side. When the user credentials are verified using authentication scheme then AAA server checks the user profile stored in policy repository in order to grant authorization [10].

The AAA support basic component like session model, user profile, service definitions and QoS model. The Session model produces a session ID for each, user having unique user ID. The provider uses these ID's to combine accounting data for various activities. The accounting of data helps in generating a bill for services utilize by the user. The session model for the user is created every time after each action. The user profile (UP) contains data record of all user related information like authentication data, service level agreement data and auditing information. The UP is defined only after the user does agreement with service provider. The AAA server stores UP information of the user who has service level agreement. It is used to identify the user properly during authentication process [10].

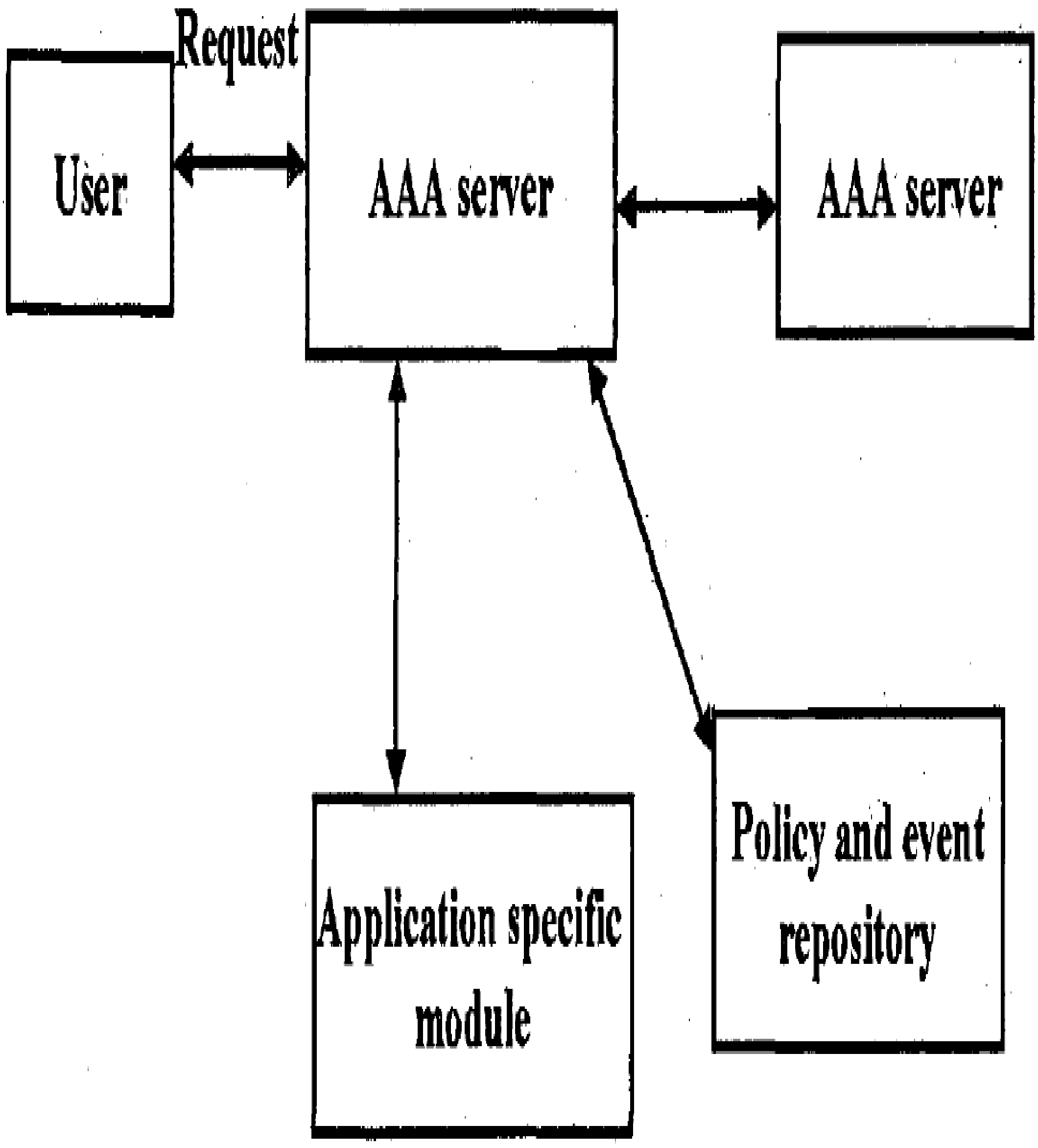

Figure 3.1: AAA Server Mechanism

3.2 Basic Network Topology of 4G

The basic 4G topology includes the following elements

1. Mobile end systems.
2. Access router provides an interfacing between MT and network.
3. Network management server handles mobility management services and AAA services.

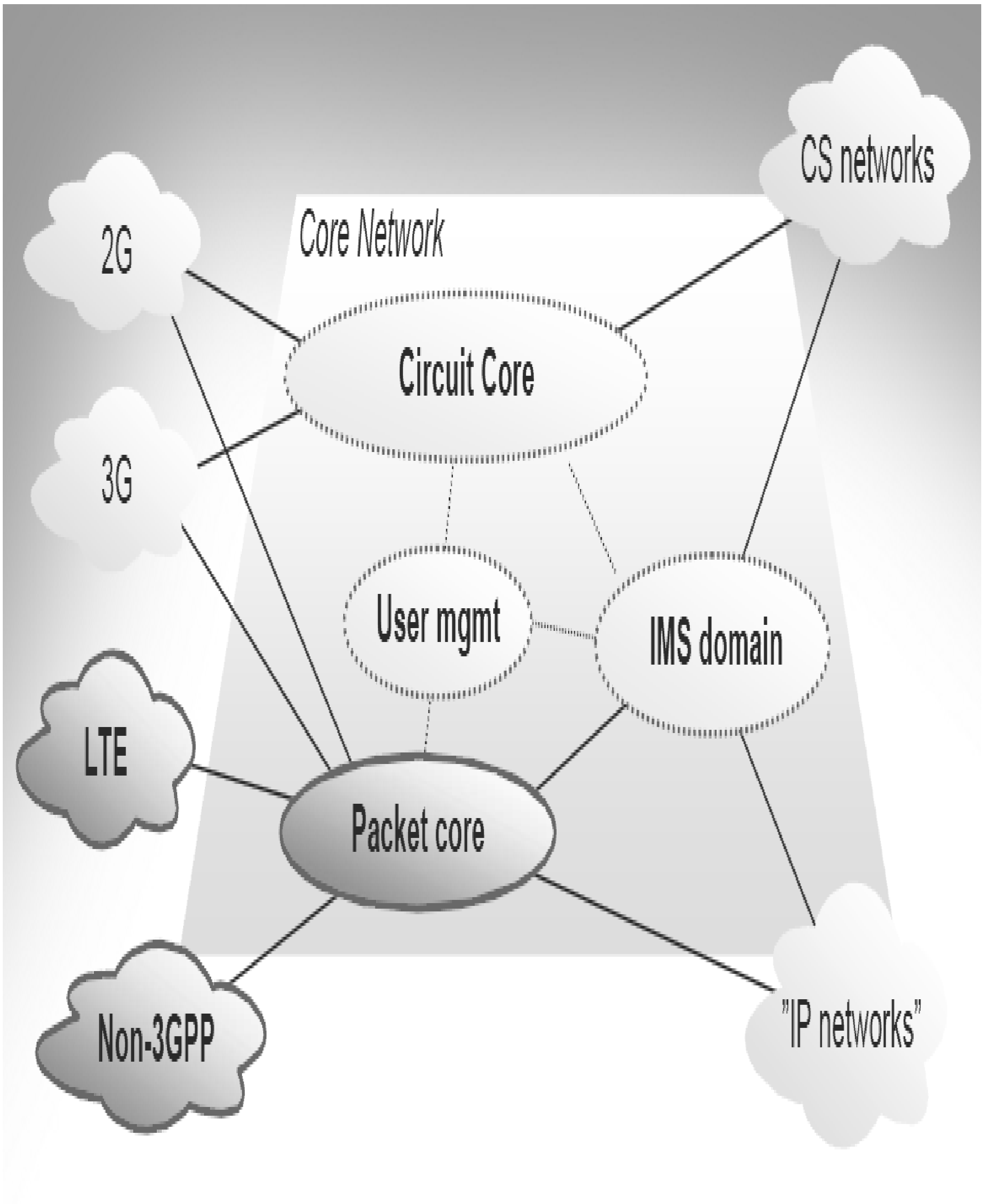

Figure 3.2: Basic Network Topology of 4G

4. MULTIPLE ACCESS TECHNIQUES

Multiple access schemes are used to allow many simultaneous users to use the same fixed bandwidth radio spectrum. Sharing of the spectrum is required in order increase the user capacity of any wireless network. In thesis OFDMA will be discussed as multiple access techniques. However, an understanding of the major methods (FDMA, TDMA) is required for understanding of any extensions to OFDMA.

4.1 Frequency Division Multiple Access (FDMA)

In Frequency Division Multiple Access (FDMA), the available bandwidth is subdivided into a number of narrower band channels which is depicted in Figure 4.1. Each user is allocated a unique frequency band in which to transmit and receive on. During a call, no other user can use the same frequency band. Each user is allocated a forward link channel (from the base station to the mobile phone) and a reverse channel (back to the base station), each being a single way link. The transmitted signal on each of the channels is continuous allowing analog transmissions. The bandwidths of FDMA channels are generally low as each channel only supports one user. FDMA is used as the primary breakup of large allocated frequency bands and is used as part of most multi-channel systems [11].

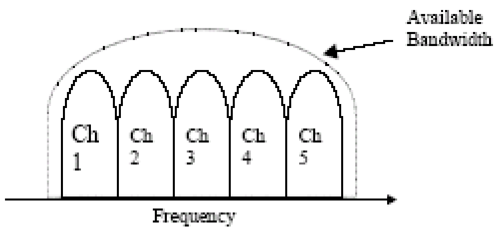

Figure 4.1: FDMA spectrum where the available bandwidth is subdivided into narrower band channels

4.2 Time Division Multiple Access (TDMA)

Time Division Multiple Access (TDMA) divides the available spectrum into multiple time slots which is depicted in Figure 4.2 by giving each user a time slot in which they can transmit or receive. Figure 4.2 shows how the time slots are provided to users in a round robin fashion, with each user being allotted one time slot per frame.

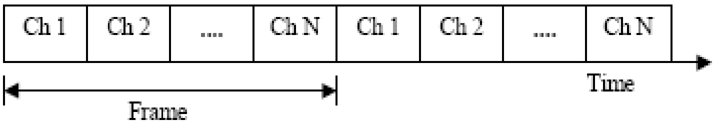

Figure 4.2: TDMA schemes where each user is allocated a small time slot

TDMA systems transmit data in a buffer and burst method, thus the transmission of each channel is non-continuous. The input data to be transmitted is buffered over the previous frame and burst transmitted at a higher rate during the time slot for the channel. TDMA can not send analog signals directly due to the buffering required, this is only used for transmitting digital data. TDMA can suffer from multipath effects as the transmission rate is generally very high. This leads the multipath signals causing inter-symbol interference. TDMA is normally used in conjunction with FDMA to subdivide the total available bandwidth into several channels. This is done to reduce the number of users per channel allowing a lower data rate to be used. This helps reduce the effect of delay spread on the transmission. Figure 4.3 shows the use of TDMA with FDMA. Each channel based on FDMA, is further subdivided using TDMA, so that several users can transmit of the one channel. This type of transmission technique is used by most digital second generation mobile phone systems. For GSM, the total allocated bandwidth of 25 MHz is divided into 125, 200 kHz channels using FDMA. These channels are then subdivided further by using TDMA so that each 200 kHz channel allows 8- 16 users [11].

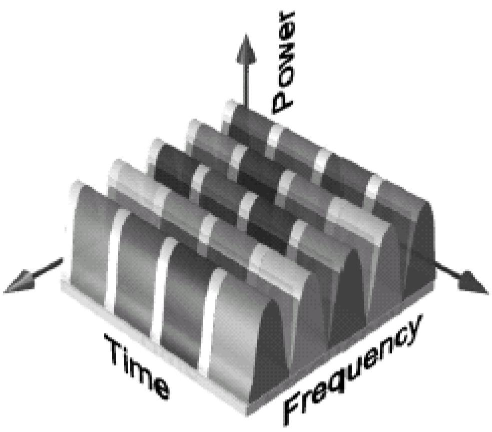

Figure 4.3: TDMA/FDMA hybrid, showing that the bandwidth is split into frequency channels and time slots

4.3 Orthogonal Frequency Division Multiplexing (OFDM) and OFDMA

4.3.1 Definition

Orthogonal frequency division multiplexing (OFDM) divides a communications channel into a number of equally spaced frequency bands. A subcarrier carrying a portion of the user information is transmitted in each band. Each subcarrier is orthogonal (independent of each other) with every other subcarrier, differentiating OFDM from the commonly used frequency division multiplexing (FDM). [12]

4.3.2 Overview

This thesis especially focused on OFDM, OFDMA and their application to mobile communications. OFDM has been explored for more than 20 years. Only recently has it been finding its way into commercial communication systems as Moore's Law has driven down the cost of the signal processing that is needed to implement OFDM based systems.

OFDM or multitone modulation as it is sometimes called is presently used in a number of commercial wired and wireless applications. On the wired side, it is used for a variant of digital subscriber line (DSL). For wireless, OFDM is the basis for several television and radio broadcast applications, including the European digital broadcast television standard, as well as digital radio in North America. OFDM is also used in several fixed wireless systems and wireless local-area network products. A system based on OFDM has been developed to deliver mobile broadband data service at data rates comparable to those of wired services, such as DSL and cable modems.

OFDM enables the creation of a very flexible system architecture that can be used efficiently for a wide range of services, including voice and data. OFDM has several unique properties that make it especially well suited to handle the challenging environmental conditions experienced by mobile wireless applications.

Figure 4.4 shows the comparison between FDM and OFDM. The idea was to use parallel data streams and FDM with overlapping subchannels to avoid the use of high speed equalization and to combat impulsive noise and multipath distortion as well as to fully use the available bandwidth.

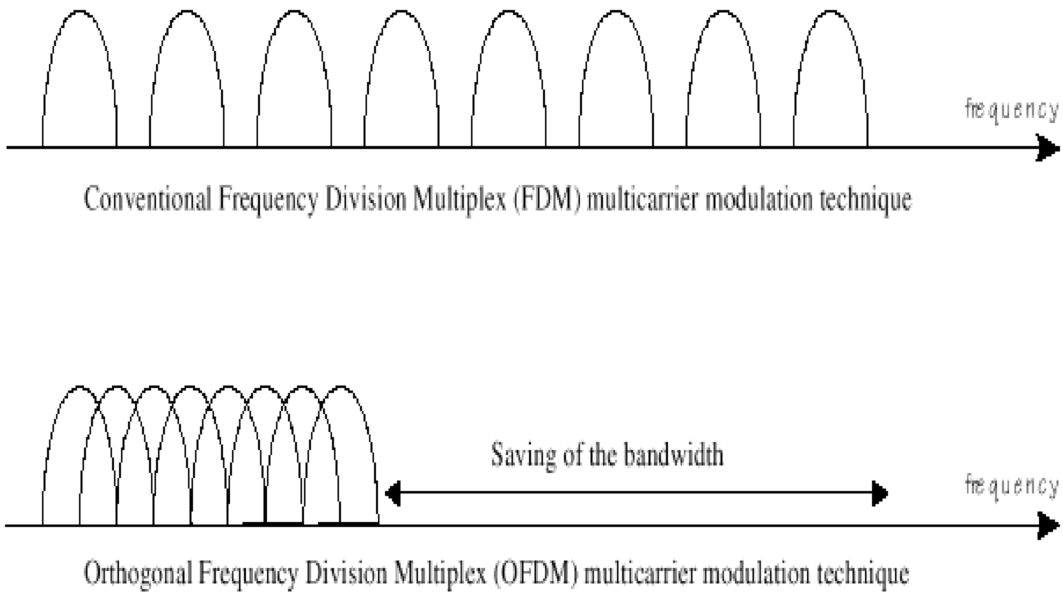

Figure 4.4: Comparison of the bandwidth utilization for FDM and OFDM

For a large number of subchannels, the arrays of sinusoidal generators and coherent demodulators required in a parallel system become unreasonably expensive and complex. The receiver needs precise phasing of the demodulating carriers and sampling times in order to keep crosstalk between subchannels acceptable. Weinstein and Ebert applied the discrete Fourier transform (DFT) to parallel data transmission system as part of the modulation and demodulation process. In addition to eliminating the banks of subcarrier oscillators and coherent demodulators required by FDM, a completely digital implementation could be built around special purpose hardware performing the Fast Fourier transform (FFT). Recent advances in VLSI technology enable making of high speed chips that can perform large size FFT at affordable price.

4.3.3 Qualitative description of OFDM

In multimedia communication, a demand emerges for high-speed, high-quality digital mobile portable reception and transmission. A receiver has to cope with a signal that is often weaker than desirable and that contains many echoes. Simple digital systems do not work well in the multipath environment [12].

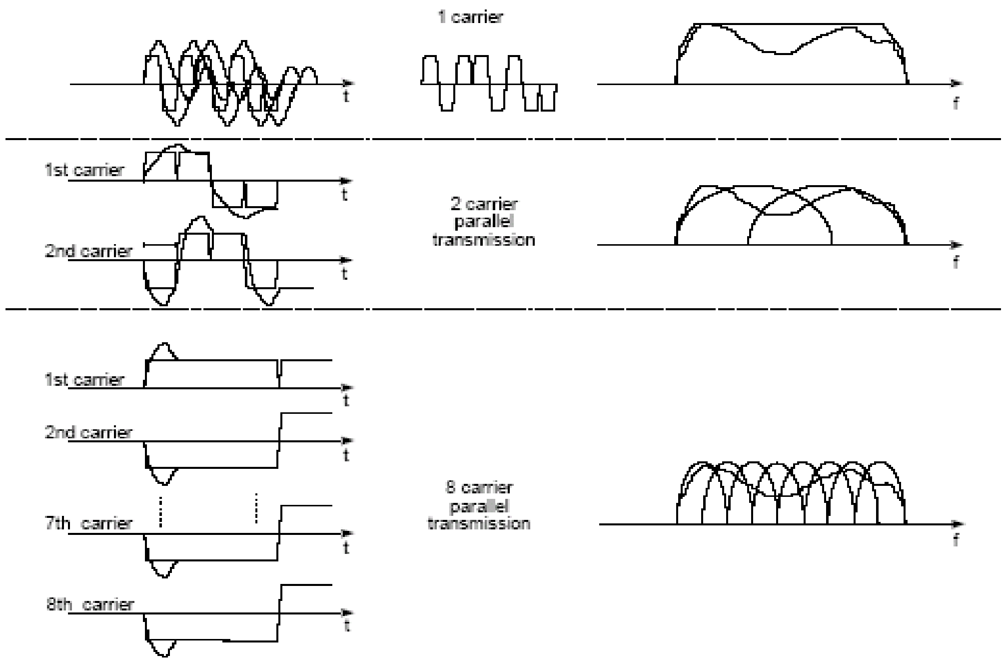

Figure 4.5: The effect of adopting a multicarrier system for a given overall data rate

In a conventional serial data system, the symbols are transmitted sequentially, with the frequency spectrum of each data symbol allowed to occupy the entire available bandwidth. In a parallel data transmission system several symbols are transmitted at the same time, what offers possibilities for alleviating many of the problems encountered with serial systems. Figure 4.5 shows the effect of adopting a multicarrier system.

In OFDM, the data is divided among large number of closely spaced carriers. This accounts for the "frequency division multiplex" part of the name. This is not a multiple access technique since there is no common medium to be shared. The entire bandwidth is filled from a single source of data. Instead of transmitting in serial way, data is transferred in a parallel way. Only a small amount of the data is carried on each carrier, and by this lowering of the bit rate per carrier (not the total bit rate), the influence of inter symbol interference is significantly reduced. In principle, many modulation schemes could be used to modulate the data at a low bit rate onto each carrier.

It is an important part of the OFDM system design that the bandwidth occupied is greater than the correlation bandwidth of the fading channel. A good understanding of the propagation statistics is needed to ensure that this condition is met. Then, although some of the carriers are degraded by multipath fading, the majority of the carriers should still be adequately received. OFDM can effectively randomize burst errors caused by Rayleigh fading, which comes from interleaving due to parallelization. So, instead of several adjacent symbols being completely destroyed, many symbols are only slightly distorted. Because of dividing an entire channel bandwidth into many narrow subbands, the frequency response over each individual subband is relatively flat. Since each subchannel covers only a small fraction of the original bandwidth, equalization is potentially simpler than in a serial data system. A simple equalization algorithm can minimize mean-square distortion on each subchannel, and the implementation of differential encoding may make it possible to avoid equalization altogether. This allows the precise reconstruction of majority of them, even without forward error correction (FEC) [13].

In addition, by using a guard interval the sensitivity of the system to delay spread can be reduced. In a classical parallel data system, the total signal frequency band is divided into N non-overlapping frequency subchannels. Each subchannel is modulated with a separate symbol and, then, the N subchannels are frequency multiplexed. There are three schemes that can be used to separate the subbands:

1. Use filters to completely separate the subbands. This method was borrowed from the conventional FDM technology. The limitation of filter implementation forces the bandwidth of each subband to be equal to (1+α) fm, where α is the roll-off factor and fm is the Nyquist bandwidth. Another disadvantage is that it is difficult to assemble a set of matched filter when the number of carriers is large.

2. Use staggered QAM to increase the efficiency of band usage. In this way the individual spectra of the modulated carriers still use an excess bandwidth, but they are overlapped at the 3 dB frequency. The advantage is that the composite spectrum is flat. The separability or orthogonality is achieved by staggering the data (offset the data by half a symbol). The requirement for filter design is less critical than that for the first scheme.

3. Use discrete Fourier transforms (DFT) to modulate and demodulate parallel data. The individual spectra are now *sinc* functions and are not band limited. The FDM is achieved, not by bandpass filtering, but by baseband processing. Using this method, both transmitter and receiver can be implemented using efficient FFT techniques that reduce the number of operations from N square in DFT, down to NlogN.

OFDM can be simply defined as a form of multicarrier modulation where its carrier spacing is carefully selected so that each subcarrier is orthogonal to the other subcarriers. As it is well known, orthogonal signals can be separated at the receiver by correlation techniques; hence, intersymbol interference among channels can be eliminated. Orthogonality can be achieved by carefully selecting carrier spacing, such as letting the carrier spacing be equal to the reciprocal of the useful symbol period.

In order to occupy sufficient bandwidth to gain advantages of the OFDM system, it would be good to group a number of users together to form a wideband system, in order to interleave data in time and frequency (depends how broad is one user signals).

4.3.4 The importance of orthogonality

The "orthogonal" part of the OFDM name indicates that there is a precise mathematical relationship between the frequencies of the carriers in the system. In a normal FDM system, the many carriers are spaced apart in such way that the signals can be received using conventional filters and demodulators. In such receivers, guard bands have to be introduced between the different carriers and the introduction of these guard bands in the frequency domain results in a lowering of the spectrum efficiency.

It is possible the carriers can be arranged in an OFDM signal so that the sidebands of the individual carriers overlap and the signals can still be received without adjacent carrier interference. In order to do this the carriers must be mathematically orthogonal which are depicted in Figure 4.6. The receiver acts as a bank of demodulators, translating each carrier down to DC, the resulting signal then being integrated over a symbol period to recover the raw data. If the other carriers all beat down to frequencies which, in the time domain, have a whole number of cycles in the symbol period (τ), then the integration process results in zero contribution from all these carriers. Thus the carriers are linearly independent (i.e. orthogonal) if the carrier spacing is a multiple of $1/\tau$ [13].

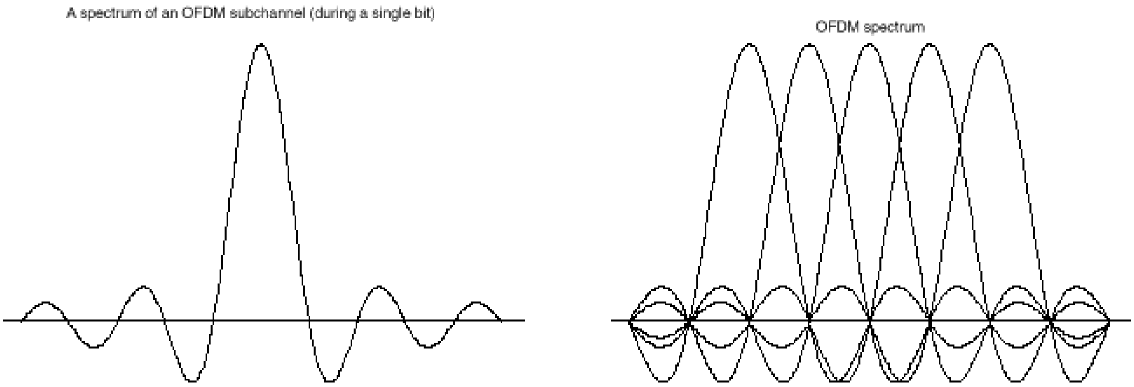

Figure 4.6: Examples of OFDM spectrum a single sub channel and 5 sub channels

4.3.5 Fourier transform

The Fourier transform allows us to relate events in time domain to events in frequency domain. There are several version of the Fourier transform, and the choice of which one to use depends on the particular circumstances of the work.

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

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

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

4.3.6 Use of FFT in OFDM

The main reason that the OFDM technique has taken a long time to become a prominence has been practical. It has been difficult to generate such a signal, and even harder to receive and demodulate the signal. The hardware solution, which makes use of multiple modulators and demodulators, was somewhat impractical for use in the civil systems.

The ability to define the signal in the frequency domain, in software on VLSI processors, and to generate the signal using the inverse Fourier transform is the key to its current popularity. The use of the reverse process in the receiver is essential if cheap and reliable receivers are to be readily available. Although the original proposals were made a long time ago, it has taken some time for technology to catch up.

At the transmitter, the signal is defined in the frequency domain. It is a sampled digital signal, and it is defined such that the discrete Fourier spectrum exists only at discrete frequencies. Each OFDM carrier corresponds to one element of this discrete Fourier spectrum. The amplitudes and phases of the carriers depend on the data to be transmitted. The data transitions are synchronized at the carriers and can be processed together, symbol by symbol [13].

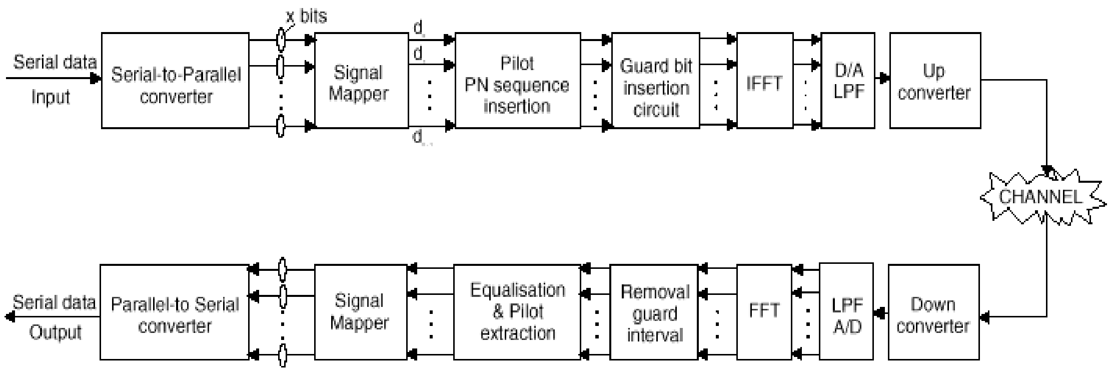

Figure 4.7: Block diagram of an OFDM

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

If these components are applied to a low-pass filter at time intervals Δt, a signal is obtained that closely approximates the frequency division multiplexed signal

Figure 4.7 illustrates the process of a typical FFT-based OFDM system. The incoming serial data is first converted form serial to parallel and grouped into *x* bits each to form a complex number. The number *x* determines the signal constellation of the corresponding subcarrier, such as 16 QAM or 32QAM. The complex numbers are modulated in the baseband by the inverse FFT (IFFT) and converted back to serial data for transmission. A guard interval is inserted between symbols to avoid intersymbol interference (ISI) caused by multipath distortion. The discrete symbols are converted to analog and low-pass filtered for RF upconversion. The receiver performs the inverse process of the transmitter. One tap equalizer is used to correct channel distortion. The tap coefficients of the filter are calculated based on the channel information.

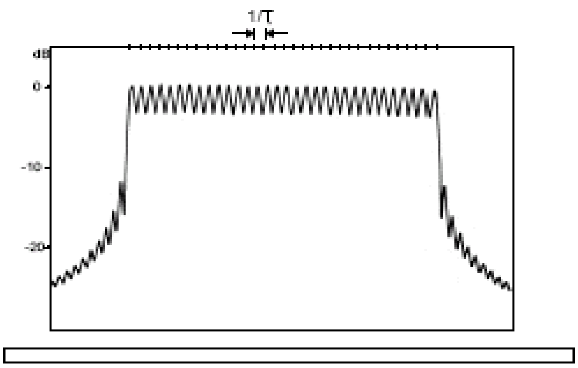

Figure 4.8: Example of the power spectral density of the OFDM signal with a guard interval (N=32)

4.3.7 Guard interval and its implementation

The orthogonality of subchannels in OFDM can be maintained and individual subchannels can be completely separated by the FFT at the receiver when there are no inter symbol interference (ISI) and inter carrier interference (ICI) introduced by transmission channel distortion. In practice these conditions can not be obtained. Since the spectra of an OFDM signal which is depicted in Figure 4.8 is not strictly band limited (*sinc (f)* function), linear distortion such as multipath cause each subchannel to spread energy into the adjacent channels and consequently cause ISI. A simple solution is to increase symbol duration or the number of carriers so that distortion becomes insignificant. However, this method may be difficult to implement in terms of carrier stability, Doppler shift, FFT size and latency.

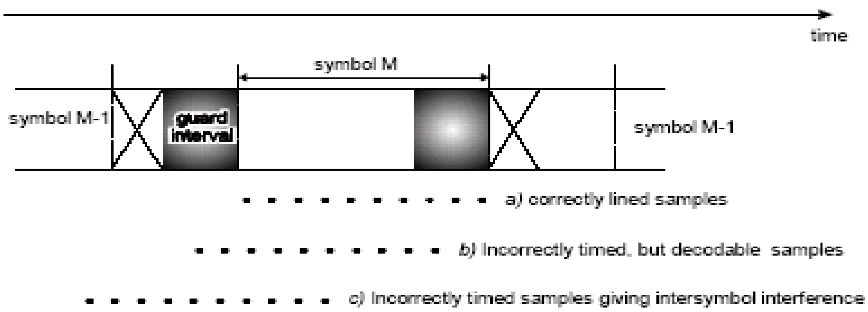

Figure 4.9: The effect on the timing tolerance of adding a guard interval.

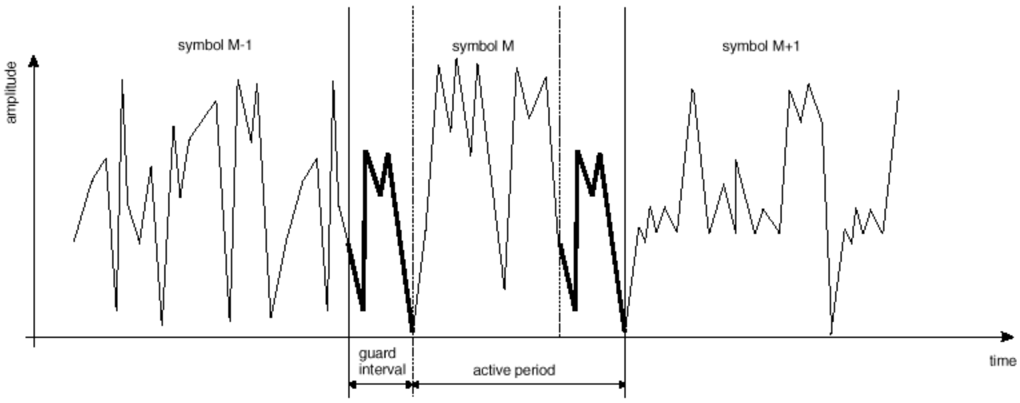

Figure 4.10: Example of the guard interval.

One way to prevent ISI is to create a cyclically extended guard interval (Fig. 4.9), where each OFDM symbol is preceded by a periodic extension of the signal itself. The total symbol duration is Ttotal=Tg+T, where Tg is the guard interval and T is the useful symbol duration. When the guard interval is longer than the channel impulse response (Fig.4.10), or the multipath delay, the ISI can be eliminated. However, the ICI, or in-band fading, still exists. The ratio of the guard interval to useful symbol duration is application dependent. Since the insertion of guard interval will reduce data throughput, Tg is usually less than $T/4$ [13].

The reasons to use a cyclic prefix for the guard interval are; to maintain the receiver carrier synchronization; some signals instead of a long silence must always be transmitted.

Cyclic convolution can still be applied between the OFDM signal and the channel response to model the transmission system.

4.3.8 OFDM for mobile communications

OFDM represents a different system design approach. It can be thought of as a combination of modulation and multiple access schemes that segment a communications channel in such a way that many users can share it then it can be called OFDMA. Whereas TDMA segments are according to time and FDMA segments are according to frequency, OFDM segments are also according to frequency but it is a technique that divides the spectrum into a number of equally spaced orthogonal tones and carries a portion of a user's information on each tone. A tone can be thought of as a frequency, much in the same way that each key on a piano represents a unique frequency. OFDM can be viewed as a form of frequency division multiplexing (FDM) however OFDM has an important special property that each tone is orthogonal with every other tone. FDM typically requires there to be frequency guard bands between the frequencies so that they do not interfere with each other. OFDM allows the spectrum of each tone to overlap (Fig. 4.11) and because they are orthogonal, they do not interfere with each other. By allowing the tones to overlap, the overall amount of spectrum required is reduced.

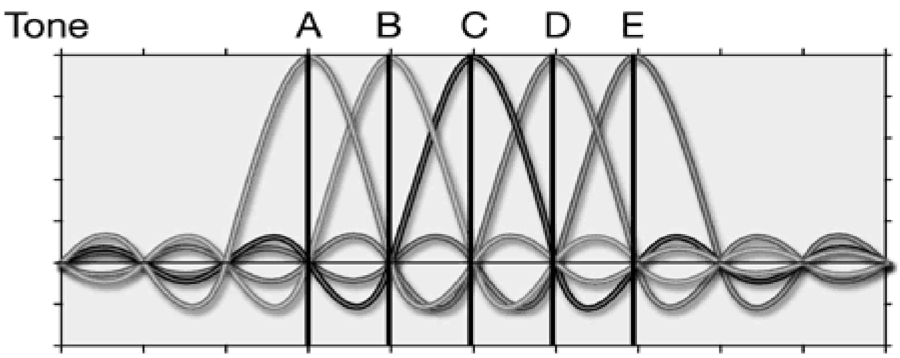

Figure 4.11: OFDM Tones

OFDM is a modulation technique in that it enables user data to be modulated onto the tones. The information is modulated onto a tone by adjusting the tone's phase, amplitude, or both. In the most basic form, a tone may be present or disabled to indicate a one or zero bit of information however either phase shift keying (PSK) or quadrature amplitude modulation (QAM) is typically employed. An OFDM system takes a data stream and splits it into N parallel data streams, each at a rate $1/N$ of the original rate. Each stream is then mapped to a tone at a unique frequency and combined together using the inverse fast Fourier transform (IFFT) to yield the time-domain waveform to be transmitted. (Fig 4.12)

For example, if a 100 tone system were used, a single data stream with a rate of 1 megabit per second (Mbps) would be converted into 100 streams of 10 kilobits per second (kbps). By creating slower parallel data streams, the bandwidth of the modulation symbol is effectively decreased by a factor of 100 or equivalently, the duration of the modulation symbol is increased by a factor of 100. Proper selection of system parameters, such as the number of tones and tone spacing, can greatly reduce, or even eliminate, ISI, because typical multipath delay spread represents a much smaller proportion of the lengthened symbol time. Viewed another way, the coherence bandwidth of the channel can be much smaller, because the symbol bandwidth has been reduced. The need for complex multi-tap time-domain equalizers can be eliminated as a result [13].

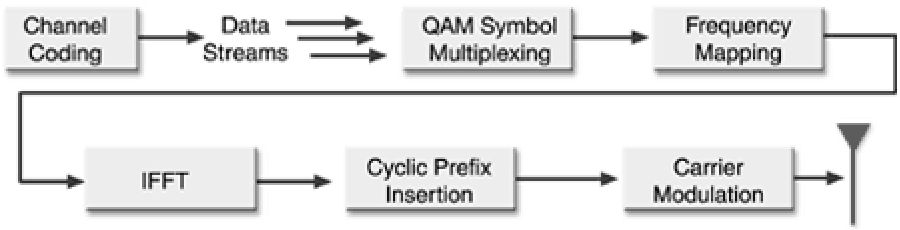

Figure 4.12: OFDM Transmitter Chain

OFDM can also be considered a multiple-access technique, because an individual tone or groups of tones can be assigned to different users. Multiple users share a given bandwidth in this manner, yielding the system called OFDMA. Each user can be assigned a predetermined number of tones when they have information to send, or alternatively, a user can be assigned a variable number of tones based on the amount of information that they have to send. The assignments are controlled by the media access control (MAC) layer, which schedules the resource assignments based on user demand.

OFDM can be combined with frequency hopping to create a spread spectrum system, realizing the benefits of frequency diversity. In a frequency hopping spread spectrum system, each user's set of tones is changed after each time period (usually corresponding to a modulation symbol). By switching frequencies after each symbol time, the losses due to frequency selective fading are minimized. Although frequency hopping they achieve comparable performance in a multipath fading environment and provide similar interference averaging benefits. Figure 4.13 illustrates the channel source for OFDM.

OFDM therefore provides the best of the benefits of TDMA in that users are orthogonal to one another while avoiding the limitations of each, including the need for TDMA frequency planning and equalization, and multiple accesses.

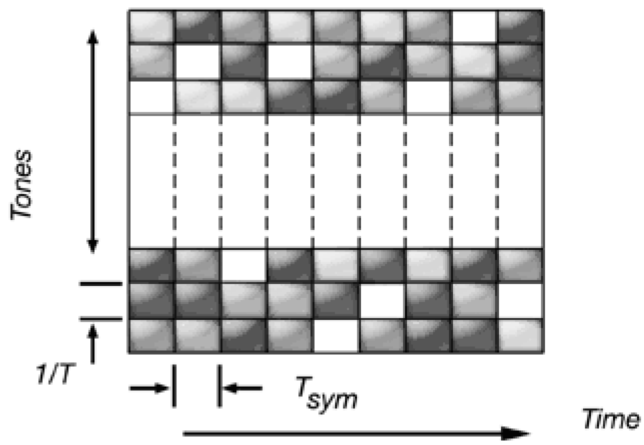

Figure 4.13: Two-Dimensional Illustration of OFDM Channel Resource

4.3.9 Theory of OFDM operation

The sinusoidal waveforms making up the tones in OFDM have the very special property of being the only Eigen-functions of a linear channel. This special property prevents adjacent tones in OFDM systems from interfering with one another, in much the same manner that the human ear can clearly distinguish between each of the tones created by the adjacent keys of a piano. This property, and the incorporation of a small amount of guard time to each symbol, enables the orthogonality between tones to be preserved in the presence of multipath. This is what enables OFDM to avoid the multiple-access interference.

The frequency domain representation of a number of tones, shown in Figure highlights the orthogonal nature of the tones used in the OFDM system. Notice that the peak of each tone corresponds to a zero level, or null, of every other tone. The result of this is that there is no interference between tones. When the receiver samples at the center frequency of each tone, the only energy present is that of the desired signal, plus whatever other noise happens to be in the channel.

To maintain orthogonality between tones, it is necessary to ensure that the symbol time contains one or multiple cycles of each sinusoidal tone waveform. This is normally the case, because the system numerology is constructed such that tone frequencies are integer multiples of the symbol period, as is subsequently highlighted, where the tone spacing is 1/T. Viewed as sinusoids (Fig. 4.14). Figure 4.15 shows tones over a single symbol period, where each tone has an integer number of cycles during the symbol.

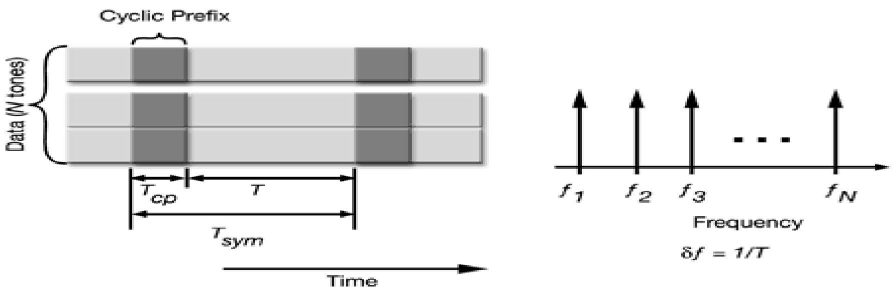

Figure 4.14: Time and Frequency Domain Representation of OFDM

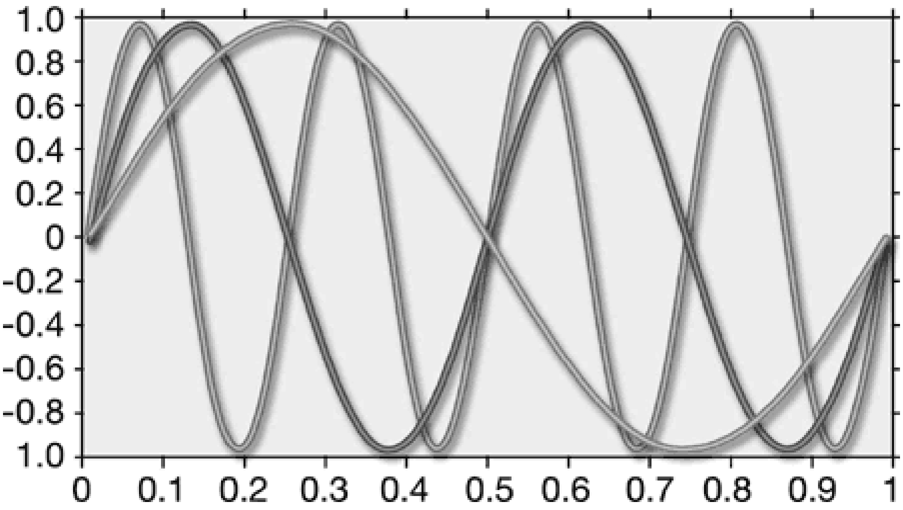

Figure 4.15: Integer Number of Sinusoid Periods

In absolute terms, to generate a pure sinusoidal tone requires the signal start at time minus infinity. This is important, because tones are the only waveform than can ensure orthogonality. Fortunately, the channel response can be treated as finite, because multipath components decay over time and the channel is effectively band-limited. By adding a guard time, called a cyclic prefix, the channel can be made to behave as if the transmitted waveforms were from time minus infinite, and thus ensure orthogonality, which essentially prevents one subcarrier from interfering with another (called inter carrier interference or ICI).

The cyclic prefix is actually a copy of the last portion of the data symbol appended to the front of the symbol during the guard interval, as shown in Figure 4.16. Multipath causes tones and delayed replicas of tones to arrive at the receiver with some delay spread. This leads to misalignment between sinusoids, which need to be aligned as in figure 4.11 to be orthogonal. The cyclic prefix allows the tones to be realigned at the receiver, thus regaining orthogonality.

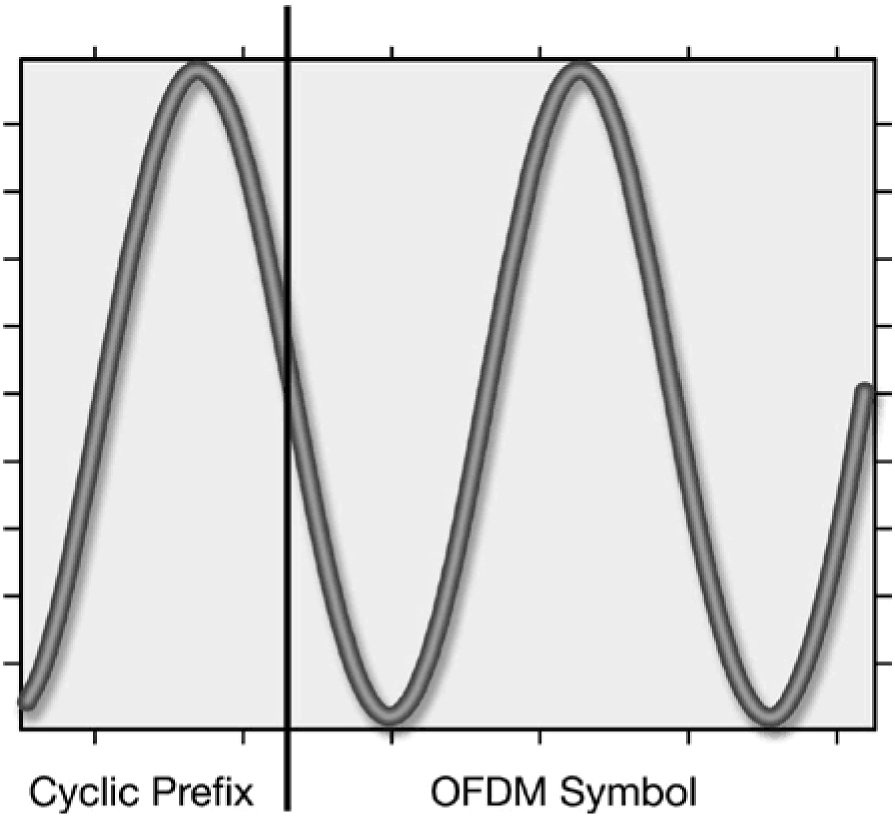

Figure 4.16: Cyclic Extension of Sinusoid

The cyclic prefix is sized appropriately to serve as a guard time to eliminate ISI. This is accomplished because the amount of time dispersion from the channel is smaller than the duration of the cyclic prefix. A fundamental trade-off is that the cyclic prefix must be long enough to account for the anticipated multipath delay spread experienced by the system. The amount of overhead increases, as the cyclic prefix gets longer. The sizing of the cyclic prefix forces a tradeoff between the amount of delay spread and the amount of Doppler shift those are acceptable.

Coded Orthogonal Frequency Division Multiplexing (COFDM) is the same as OFDM except that forward error correction is applied to the signal before transmission. This is to overcome errors in the transmission due to lost carriers from frequency selective fading, channel noise and other propagation effects. For this discussion the terms OFDM and COFDM are used interchangeably, as the main focus of this thesis is on OFDM, but it is assumed that any practical system will use forward error correction, thus would be COFDM [13].

In FDMA each user is typically allocated a single channel, which is used to transmit all the user information. The bandwidth of each channel is typically 10 kHz-30 kHz for voice communications. However, the minimum required bandwidth for speech is only 3 kHz. The allocated bandwidth is made wider then the minimum amount required preventing channels from interfering with one another. This extra bandwidth is to allow for signals from neighboring channels to be filtered out, and to allow for any drift in the center frequency of the transmitter or receiver. In a typical system up to 50% of the total spectrum is wasted due to the extra spacing between channels. This problem becomes worse as the channel bandwidth becomes narrower, and the frequency band increases.

Most digital phone systems use vocoders to compress the digitized speech. This allows for an increased system capacity due to a reduction in the bandwidth required for each user. Current vocoders require a data rate somewhere between 4- 13kbps, with depending on the quality of the sound and the type used. Thus each user only requires a minimum bandwidth of somewhere between 2-7 kHz, using QPSK modulation. However, simple FDMA does not handle such narrow bandwidths very efficiently.

TDMA partly overcomes this problem by using wider bandwidth channels, which are used by several users. Multiple users access the same channel by transmitting in their data in time slots. Thus, many low data rate users can be combined together to transmit in a single channel which has a bandwidth sufficient so that the spectrum can be used efficiently.

There are however, two main problems with TDMA. There is an overhead associated with the change over between users due to time slotting on the channel. A change over time must be allocated to allow for any tolerance in the start time of each user, due to propagation delay variations and synchronization errors. This limits the

number of users that can be sent efficiently in each channel. In addition, the symbol rate of each channel is high (as the channel handles the information from multiple users) resulting in problems with multipath delay spread.
OFDM overcomes most of the problems with both FDMA and TDMA. OFDM splits the available bandwidth into many narrow band channels (typically 100-8000). The carriers for each channel are made orthogonal to one another, allowing them to be spaced very close together, with no overhead as in the FDMA example. Because of this there is no great need for users to be time multiplex as in TDMA, thus there is no over head associated with switching between users.
The orthogonality of the carriers means that each carrier has an integer number of cycles over a symbol period. Due to this, the spectrum of each carrier has a null at the centre frequency of each of the other carriers in the system. This results in no interference between the carriers, allowing then to be spaced as close as each carrier in an OFDM signal has a very narrow bandwidth (i.e. 1 kHz), thus the resulting symbol rate is low. This results in the signal having a high tolerance to multipath delay spread as the delay spread must be very long to cause significant inter-symbol interference (e.g. > 500usec).

4.3.10 OFDM generation

To generate OFDM successfully the relationship between all the carriers must be carefully controlled to maintain the orthogonality of the carriers. For this reason, OFDM is generated by firstly choosing the spectrum required, based on the input data, and modulation scheme used. Each carrier to be produced is assigned some data to transmit. The required amplitude and phase of the carrier is then calculated based on the modulation scheme (typically differential BPSK, QPSK, or QAM). The required spectrum is then converted back to its time domain signal using an Inverse Fourier Transform. In most applications, an Inverse Fast Fourier Transform (IFFT) is used. The IFFT performs the transformation very efficiently, and provides a simple way of ensuring the carrier signals produced are orthogonal.
The Fast Fourier Transform (FFT) transforms a cyclic time domain signal into its equivalent frequency spectrum. This is done by finding the equivalent waveform, generated by a sum of orthogonal sinusoidal components. The amplitude and phase of the sinusoidal components represent the frequency spectrum of the time domain signal. The IFFT performs the reverse process, transforming a spectrum (amplitude

and phase of each component) into a time domain signal. An IFFT converts a number of complex data points, of length which is a power of 2, into the time domain signal of the same number of points. Each data point in frequency spectrum used for an FFT or IFFT is called a bin.

The orthogonal carriers required for the OFDM signal can be easily generated by setting the amplitude and phase of each bin, then performing the IFFT. Since each bin of an IFFT corresponds to the amplitude and phase of a set of orthogonal sinusoids, the reverse process guarantees that the carriers generated are orthogonal [14].

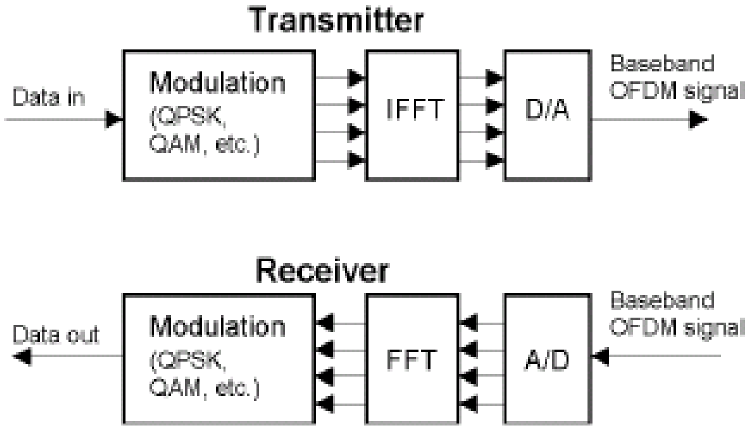

Figure 4.17: Basic FFT, IFFT OFDM transmitter and receiver

Figure 4.17 shows the setup for a basic OFDM transmitter and receiver. The signal generated is a baseband, thus the signal is filtered, then stepped up in frequency before transmitting the signal.

4.3.11 Adding a guard period to OFDM

One of the most important properties of OFDM transmissions is the robustness against multipath delay spread. This is achieved by having a long symbol period, which minimizes the inter-symbol interference. The level of robustness can be increased even more by the addition of a guard period between transmitted symbols. The guard period allows time for multipath signals from the pervious symbol to die away before the information from the current symbol is gathered. The most effective guard period to use is a cyclic extension of the symbol. If a mirror in time, of the end of the symbol waveform is put at the start of the symbol as the guard period, this effectively extends the length of the symbol, while maintaining the orthogonality of the waveform. Using this cyclic extended symbol the samples required for performing the FFT (to decode the symbol), can be taken anywhere over the length of the symbol. This provides multipath immunity as well as symbol time synchronization tolerance. As long as the multipath delay echoes stay within the guard period duration, there is strictly no limitation regarding the signal level of the echoes. They may even exceed the signal level of the shorter path. The signal energy from all paths just adds at the input to the receiver, and since the FFT is energy conservative, the whole available power feeds the decoder. If the delay spread is longer then the guard intervals then they begin to cause intersymbol interference. However, provided the echoes are sufficiently small they do not cause significant problems. This is true most of the time as multipath echoes delayed longer than the guard period will have been reflected of very distant objects. Other variations of guard periods are possible. One possible variation is to have half the guard period a cyclic extension of the symbol, the other half a zero amplitude signal. This will result in a signal as shown in Figure 4.18 using this method the symbols can be easily identified. This possibly allows for symbol timing to be recovered from the signal, simply by applying envelop detection. The disadvantage of using this guard period method is that the zero period does not give any multipath tolerance, thus the effective active guard period is halved in length. It is still not clear whether symbol timing needs to be recovered using this method. [13, 15, 16]

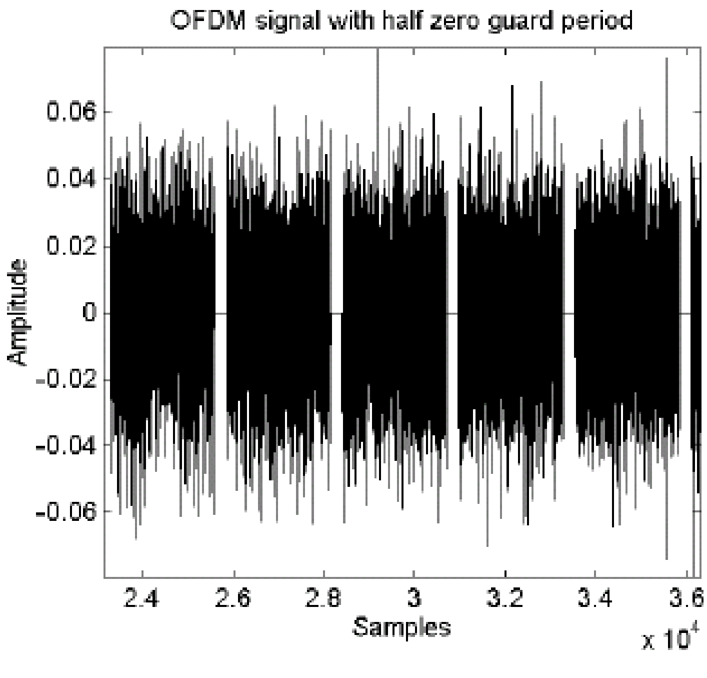

Figure 4.18: Section of an OFDM signal showing 5 symbols by using a guard period

4.4 Single Carrier Frequency Division Multiple Access (SC-FDMA)

SC-FDMA is a new multiple access technique which is currently adopted as the uplink multiple access schemes in 3GPP LTE. It has similar structure and performance to OFDMA. Actually SC-FDMA is modified OFDMA scheme to overcome the high PAPR problem. LTE is deploying SC-FDMA scheme for uplink access due to its low PAPR advantage.SC-FDMA utilizes single carrier modulation, DFT-spread orthogonal frequency multiplexing and frequency domain equalization.

As it can be seen in the Figure 4.19, SC-FDMA can be interpreted as a linearly precoded OFDMA scheme that it has an additional DFT processing preceding the conventional OFDMA processing. [17, 18]

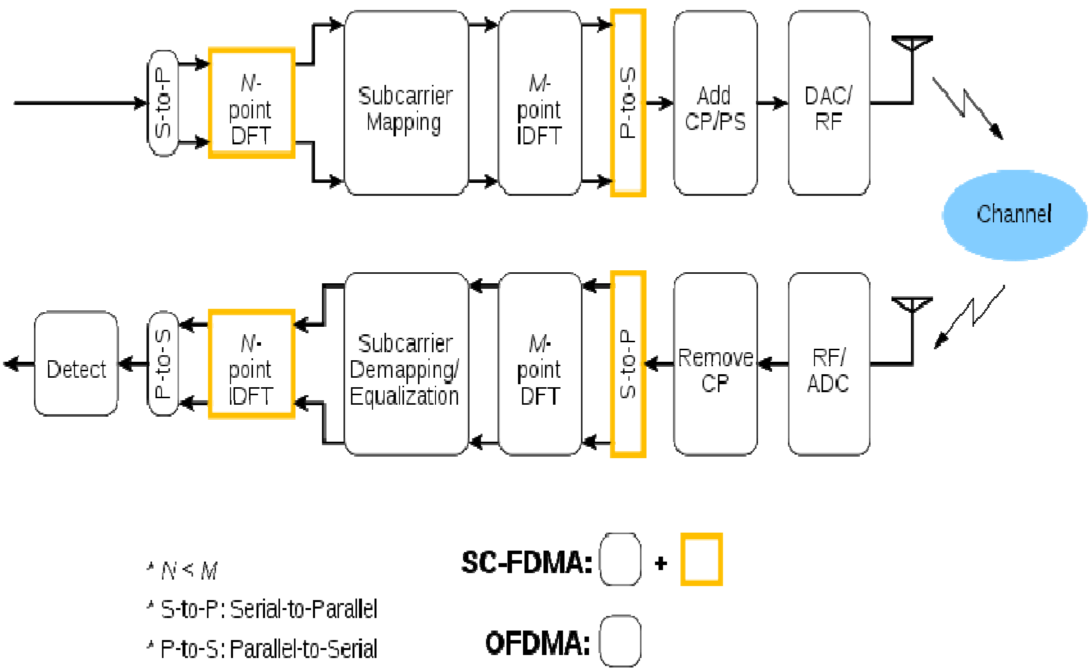

Figure 4.19: Transmitter and Receiver Structure of OFDMA/SC-FDMA

The distinguishing feature of SC-FDMA is that it leads to a single-carrier transmit signal, in contrast to OFDMA. Time domain data symbols are transformed to frequency domain by DFT before going through OFDMA modulation. The orthogonality of the users stems from the fact that each user occupies different subcarriers in the frequency domain, similar to the case of OFDMA. Because the overall transmit signal is a single carrier signal, PAPR is inherently low compared to the case of OFDMA which produces a multicarrier signal [19].

5. LTE SYSTEM ANALYSIS

Mainly there are two type analyzes for 4G network as system level and link level. The system level simulation focuses on network related issue such as scheduling, mobility handling or interference management. But the link level simulation focuses more Multiple-Input Multiple-Output, Adaptive modulation, coding feedback, modeling of channel encoding or physical layer modeling for system level [20, 21].
In thesis, link level performance is mostly analyzed although the some graphs are mentioned about the system level performance at the end of the analysis [22, 26].
In thesis, the 4G is simulated according to link layer simulation model for downlink transmission. The model is simulated in MATLAB. When the simulation model in thesis is designed, the research for Mobilcom Austria by Vienna University of Technology is mostly used and it is also stated in references [22-25].

5.1 Simulation Setup

Overall simulator structure which is depicted in Figure 5.1 includes one transmitting eNodeB, receiver user equipments (UEs), a downlink channel model. The service parameters which are determined in 3GPP technical specification (TS 36.942) are used and they are depicted in Table 5.1.

Table 5.1: Service Parameters in Technical Specification (TS 36.942)

Frequency:	2 GHz
Bandwidth:	5 MHz and 20 MHz
Transmit Mode (TxD)	Single Input Single Output (SISO), 2x1, 4x2, Open Loop Spatial Multiplexing (OLSM)
Channel Type	Flat Rayleigh, PedB (Pedestrian 10km/h), AWGN
Retransmission	0 and 3
Filtering	Block Fading
CQI(Channel Quality Indicator)	16QAM for 7, 64QAM for 15
FFT size	512 for 5 MHz and 2048 for 20 MHz
UE(User Equipment)	1,2

The bandwidth, transmit mode, channel type, CQI and FFT size are accepted as system variable. According to these variables BLER and Throughput are analyzed.

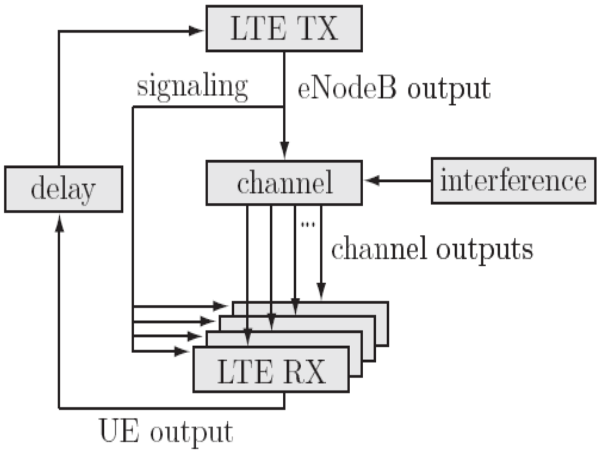

Figure 5.1: Overall simulator structure

5.1.1 Transmitter

Downlink transmission is based on OFDMA. OFDM subcarrier is carried during one OFDM symbol interval. These resource elements are grouped into Resource Blocks (RBs) that consist of six to seven OFDM symbols (depending on the cyclic prefix length utilized) and twelve consecutive subcarriers corresponding to a nominal resource block bandwidth of 180 KHz.

In the first step of the transmitter processing as it can be seen in Figure 5.2, the user data is generated depending on the previous Acknowledgement (ACK) signal. If the previous user data Transport Block (TB) was not acknowledged, the stored TB is retransmitted using a Hybrid Automatic Repeat request (HARQ) scheme. Then a Cyclic Redundancy Check (CRC) is calculated and appended to each user's TB. The data of each user is independently encoded using a turbo encoder with Quadrature Permutation Polynomial (QPP) based interleaving. Each block of coded bits is then interleaved and rate-matched with a target rate depending on the received Channel Quality Indicator (CQI) user feedback.

The encoding process is followed by the data modulation, which maps the channel-encoded TB to complex modulation symbols. Depending on the CQI, a modulation scheme is selected for the corresponding RB. Possible modulations for the DL-SCH are 4-QAM, 16-QAM and 64-QAM. The modulated transmit symbols are then mapped to up to four transmit antennas.

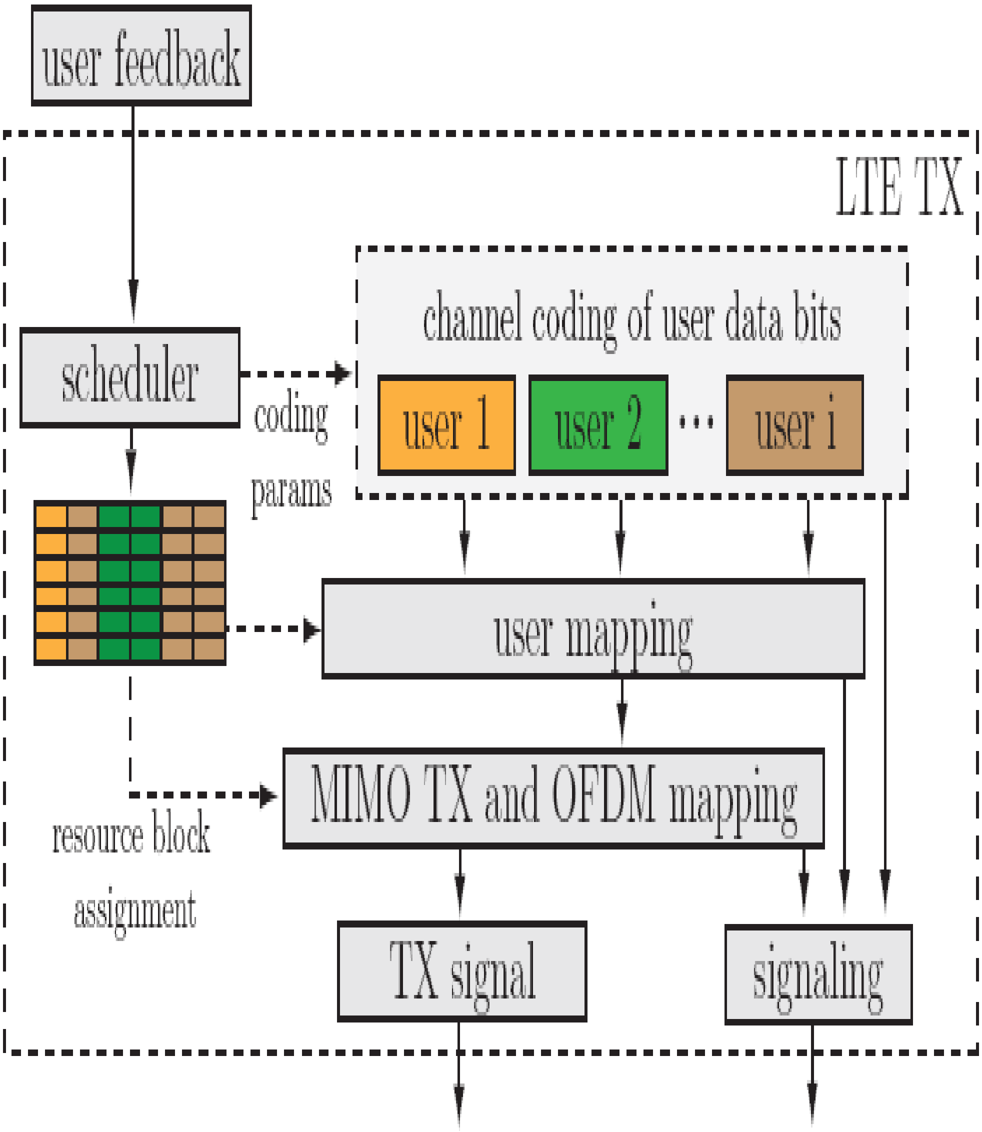

Figure 5.2: Structure of the transmitter

5.1.2 Receiver

First, the receiver has to identify the RBs that carry its designated information. The estimation of the channel is performed using the reference signals available in the resource grid. Based on this channel estimation, the quality of the channel may be evaluated and the appropriate feedback information calculated. The channel knowledge is also used for the demodulation and soft-demapping of the OFDM signal. Figure 5.3 shows the structure of the receiver.

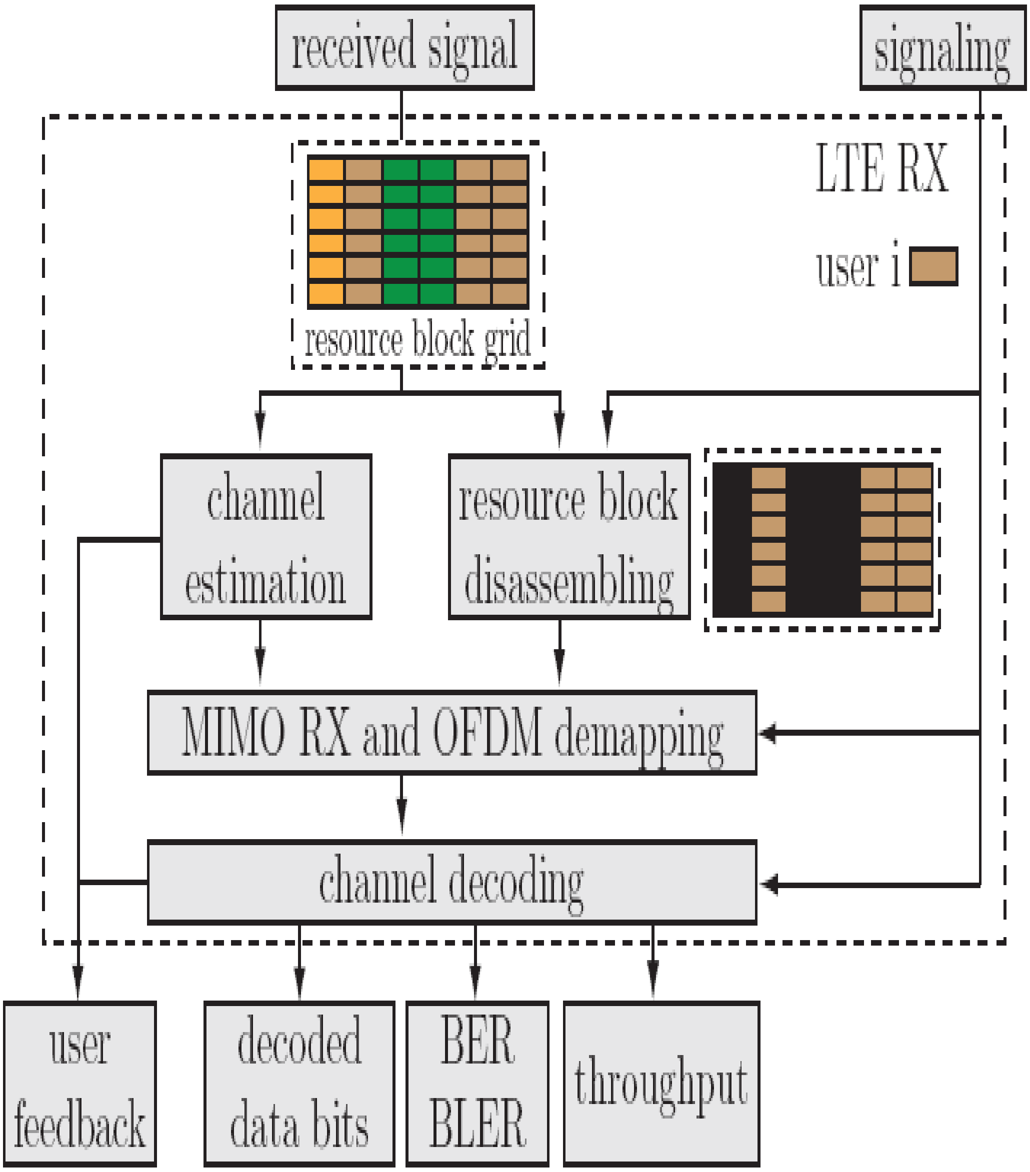

Figure 5.3: Structure of the receiver

5.2 Performance Results

When the performance results are interpreted, the following SNR-CQI figure is used to understand CQI. SNR-CQI mapping is for the model TS 36.942 by 3GPP. CQI is used as 7. As it can be seen in the Figure 5.4 when CQI is 7, SNR is around 5 dB.

In conclusion, when the results are analyzed, it must be taken into consideration BLER should be smaller than 10% for SNR is 5 dB as it can be seen in the Figure 5.5.

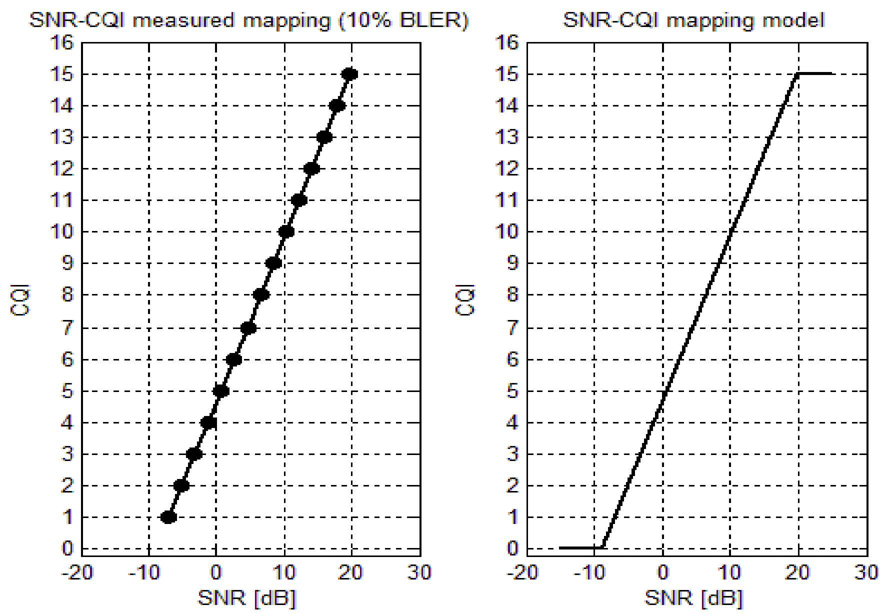

Figure 5.4: SNR-CQI Mapping Model

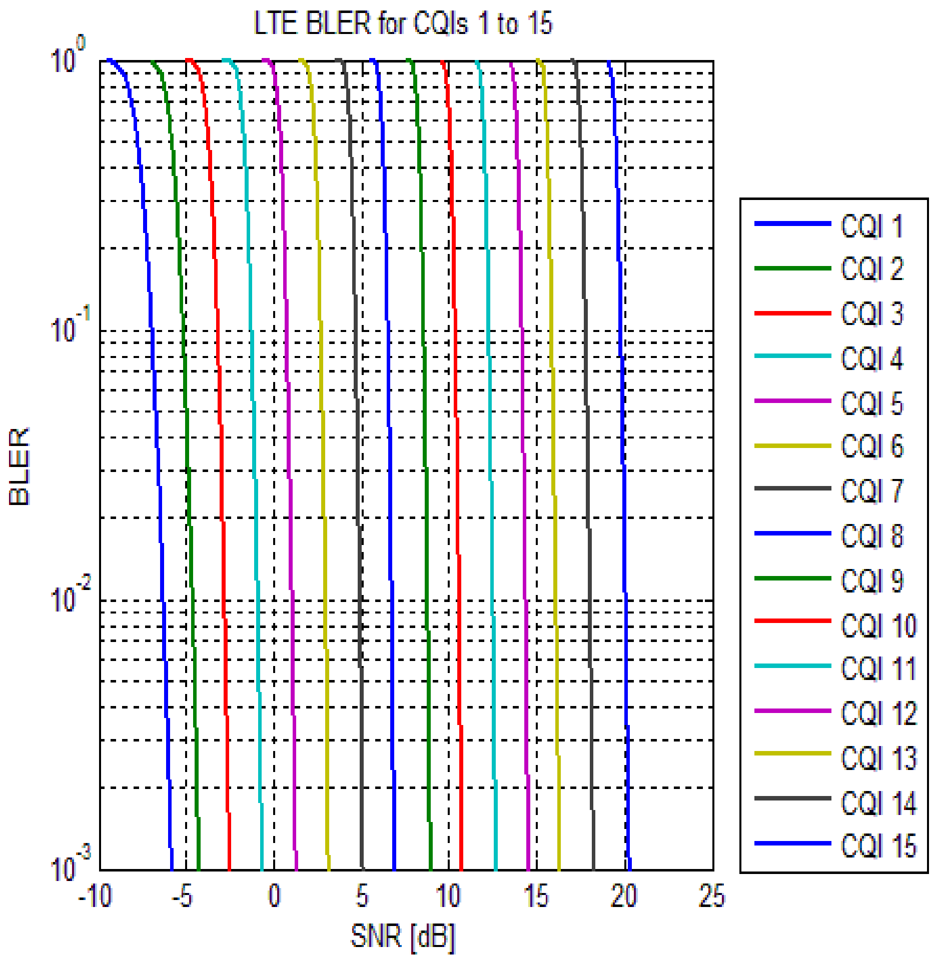

Figure 5.5: BLER for CQI 1 to 15

5.2.1 BLER results for 5 MHz

When CQI is 7, Channel type is flat Rayleigh and retransmission is 3, system is analyzed in the Figure 5.6. For CQI=7, 16QAM is used and it has to ensure a BLER value smaller than 10 % and as it can be seen in the Figure 5.4 when CQI is 7, SNR is 5 dB. According to the Figure 5.6 only for TxD 4x2, BLER is acceptable and smaller than %10 for CQI=7.

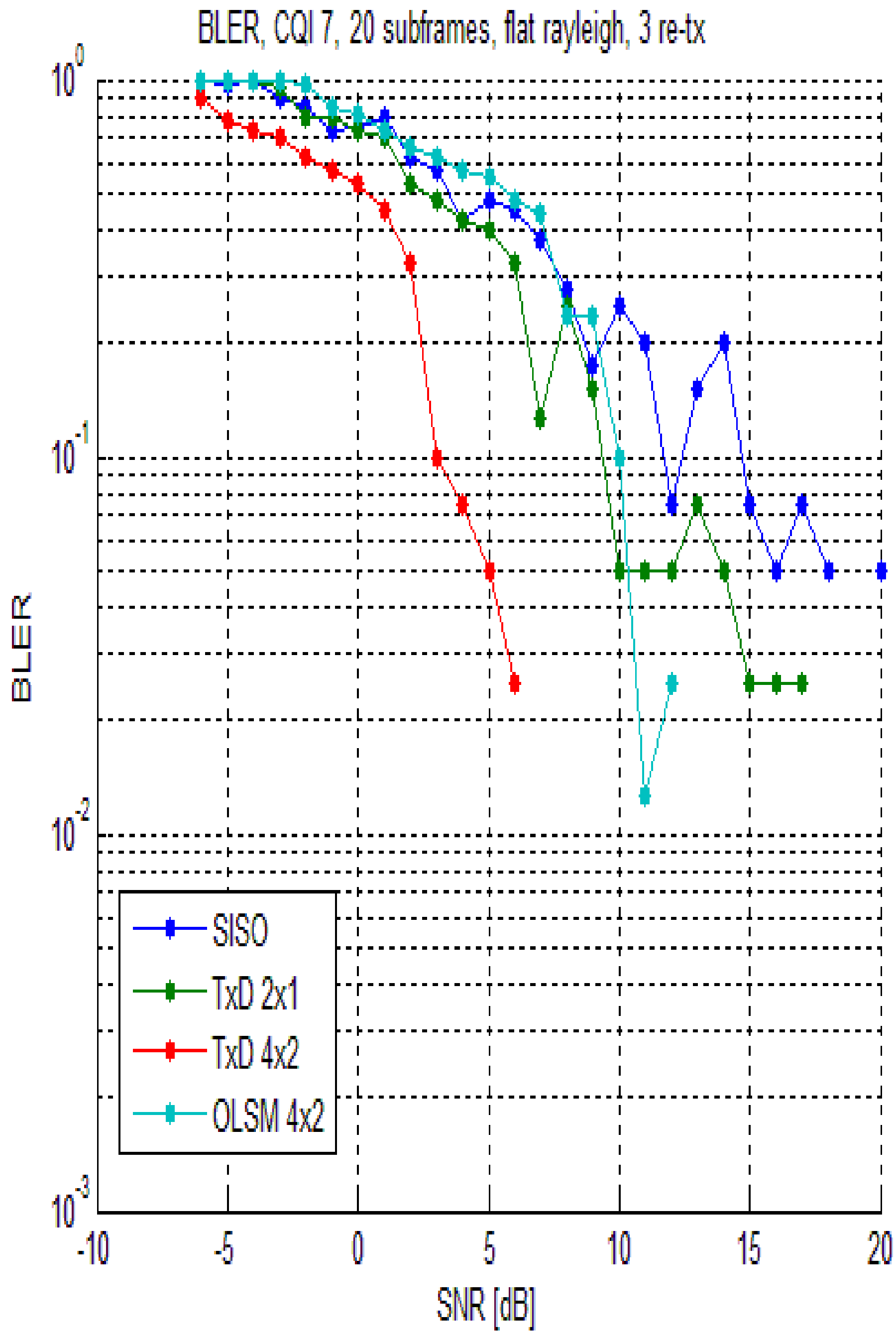

Figure 5.6: BLER, flat Rayleigh channel, 3 retransmissions

When CQI is 7, Channel type is PedB (Pedestrian 10km/h) and retransmission is 3, system is analyzed. When CQI is 7, 16QAM is used as channel coding. According to the Figure 5.7 only TxD 4x2 mode BLER is acceptable and smaller than %10 for CQI=7.

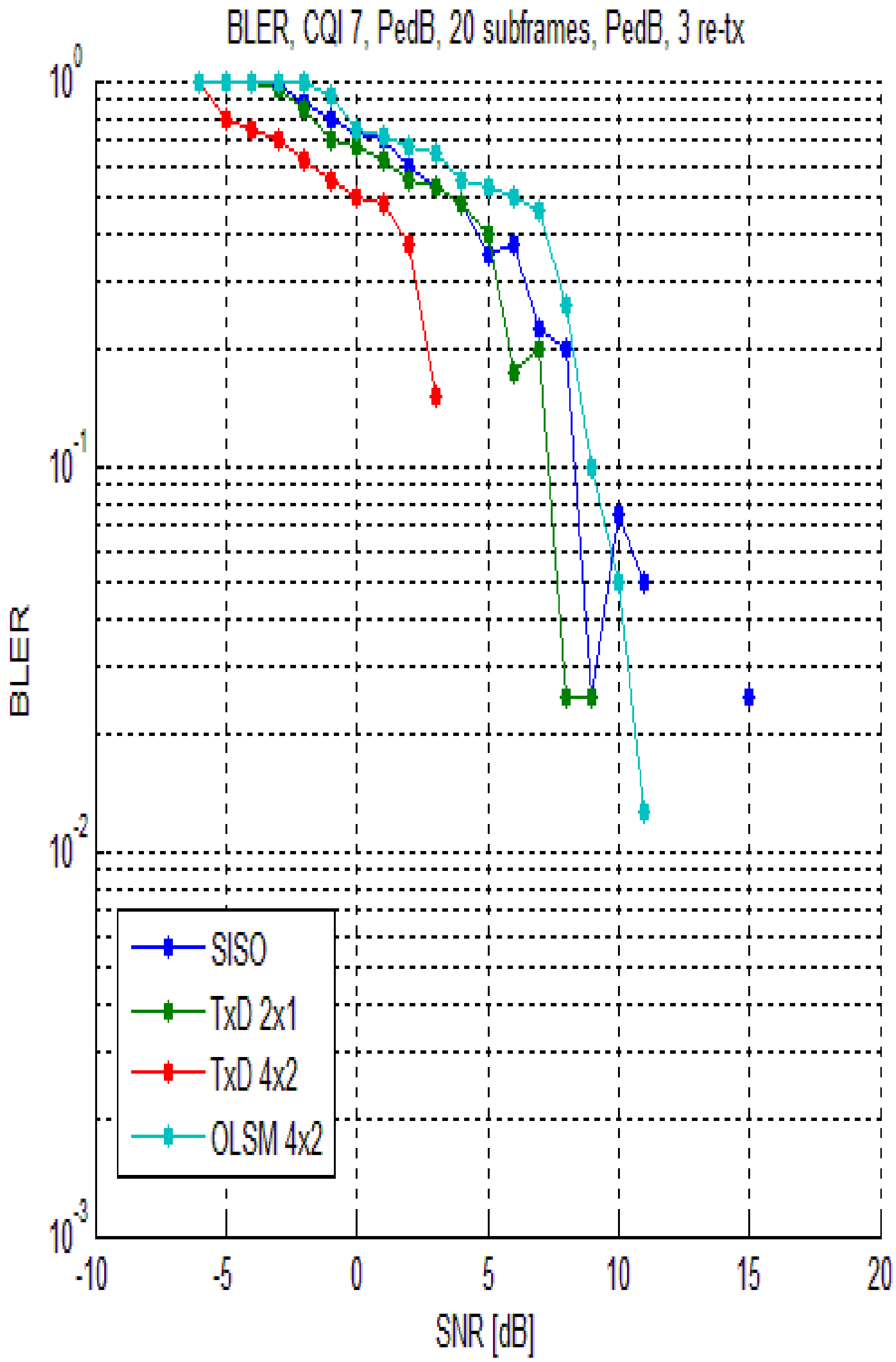

Figure 5.7: BLER, PedB channel, 3 retransmissions

5.2.2 Throughput results for 5 MHz

When CQI is 7, Channel type is flat Rayleigh and retransmission is 3, system is analyzed. For CQI is 7, 16QAM is used the maximum throughput can be obtained by using mode 3 OLSM 4x2. According to the Figure 5.4 the SNR is 5 dB when CQI is 7 and SISO, 2x1, 4x2, OLSM 4x2 provide respectively 3, 3, 5, 5 Mbps throughput.

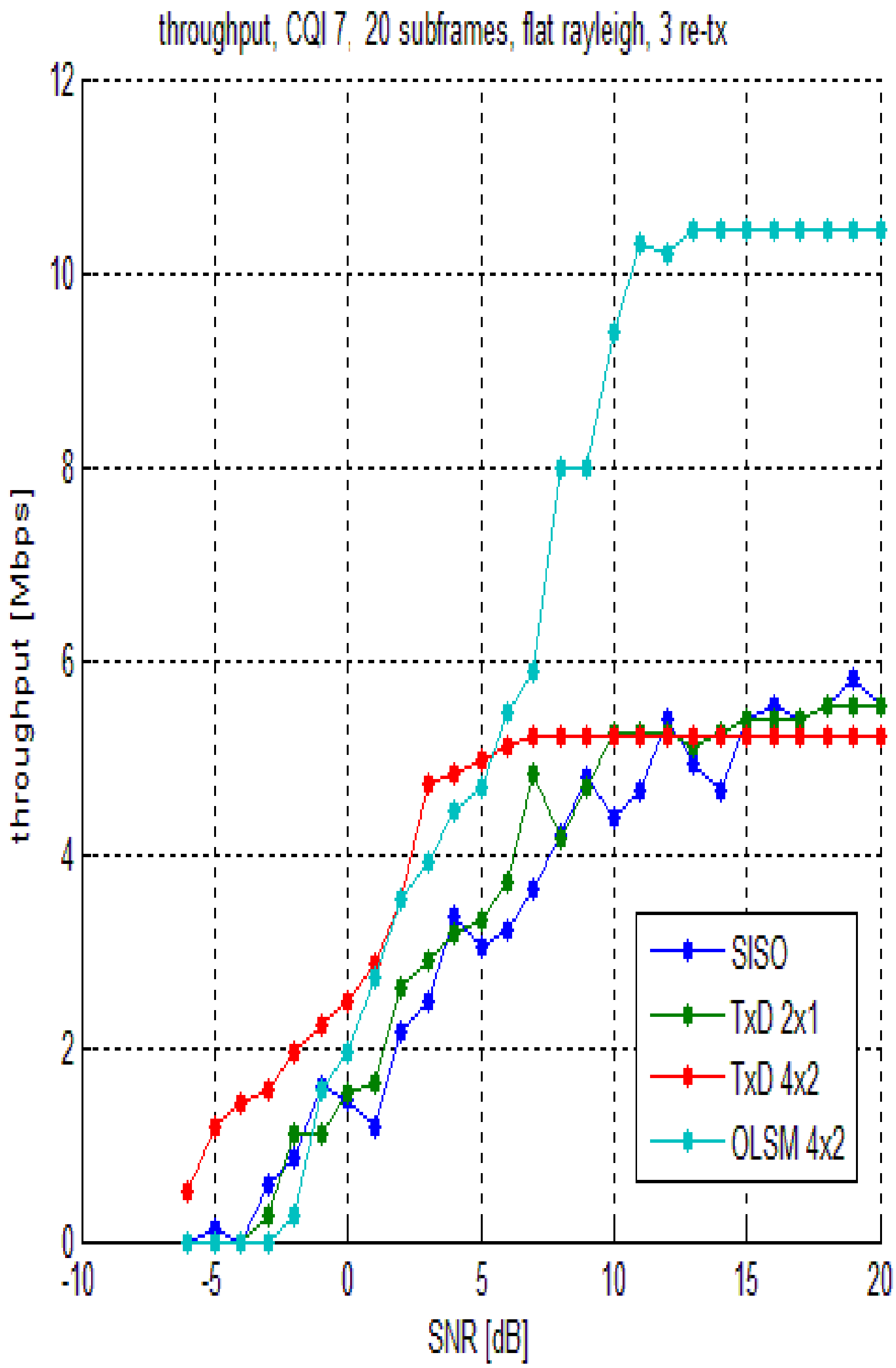

Figure 5.8: Throughput, flat Rayleigh channel, 3 retransmissions

When CQI is 7, Channel type is PedB (10km/h), Retransmission is 3, and 16QAM is used. The maximum throughput can be obtained again by using mode 3 OLSM 4x2. According to the Figure 5.9 the SNR is 5 dB for CQI=7 and SISO, 2x1, 4x2, OLSM 4x2 provides in sequence approximately 5, 5, 5, 9 Mbps throughput for 5 MHz.

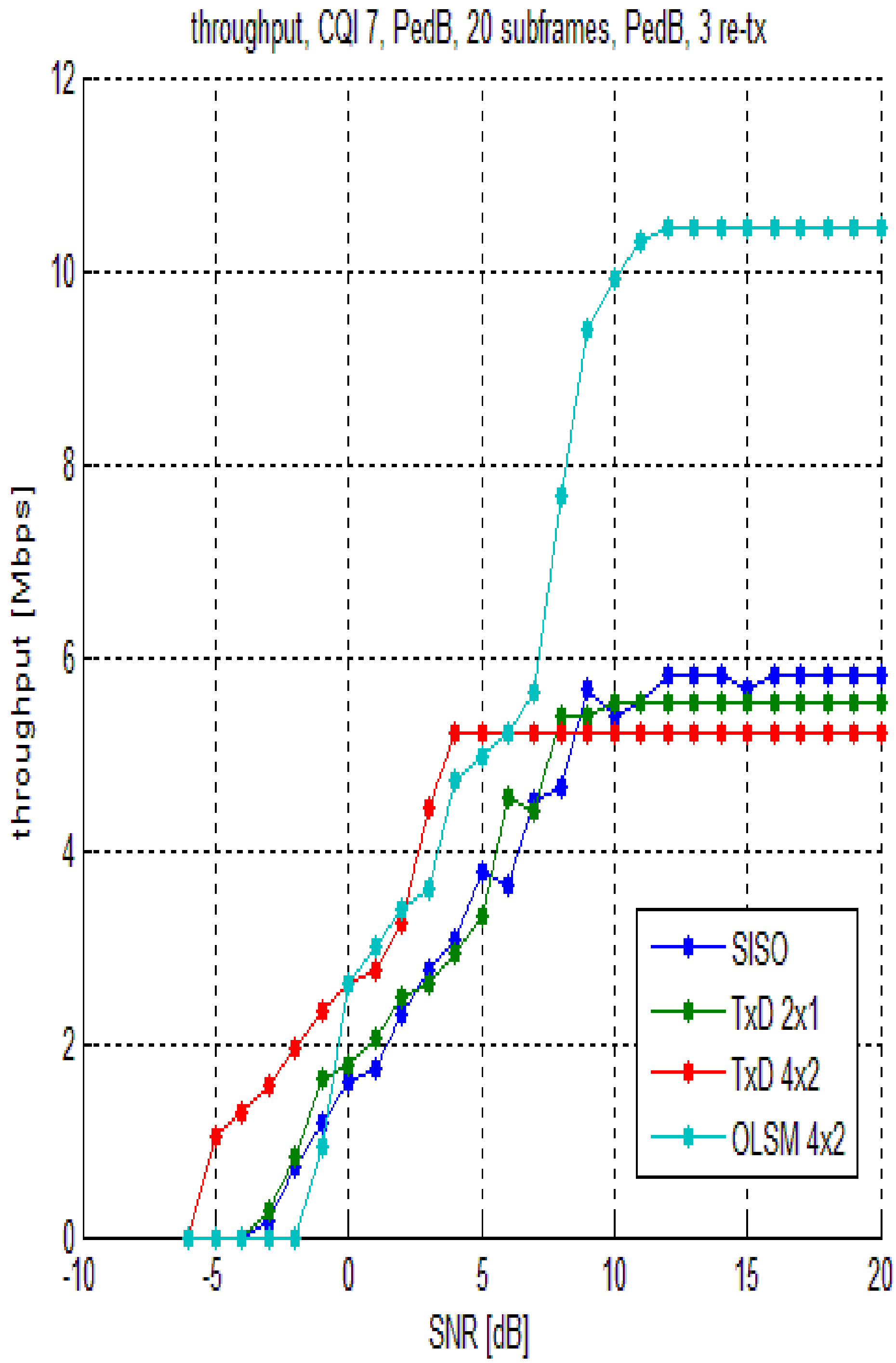

Figure 5.9: Throughput, PedB channel, 3 retransmissions

5.2.3 BLER results for 20 MHz

When CQI is 7, Channel type is flat Rayleigh and retransmission is 3, system is analyzed. When CQI is 7, 16QAM is used and it has to ensure a BLER value smaller than 10 % and as it can be seen in the Figure 5.4 when CQI is 7, SNR is 5 dB. According to the Figure 5.10 only for TxD 4x2 is acceptable and smaller than %10 for CQI=7.

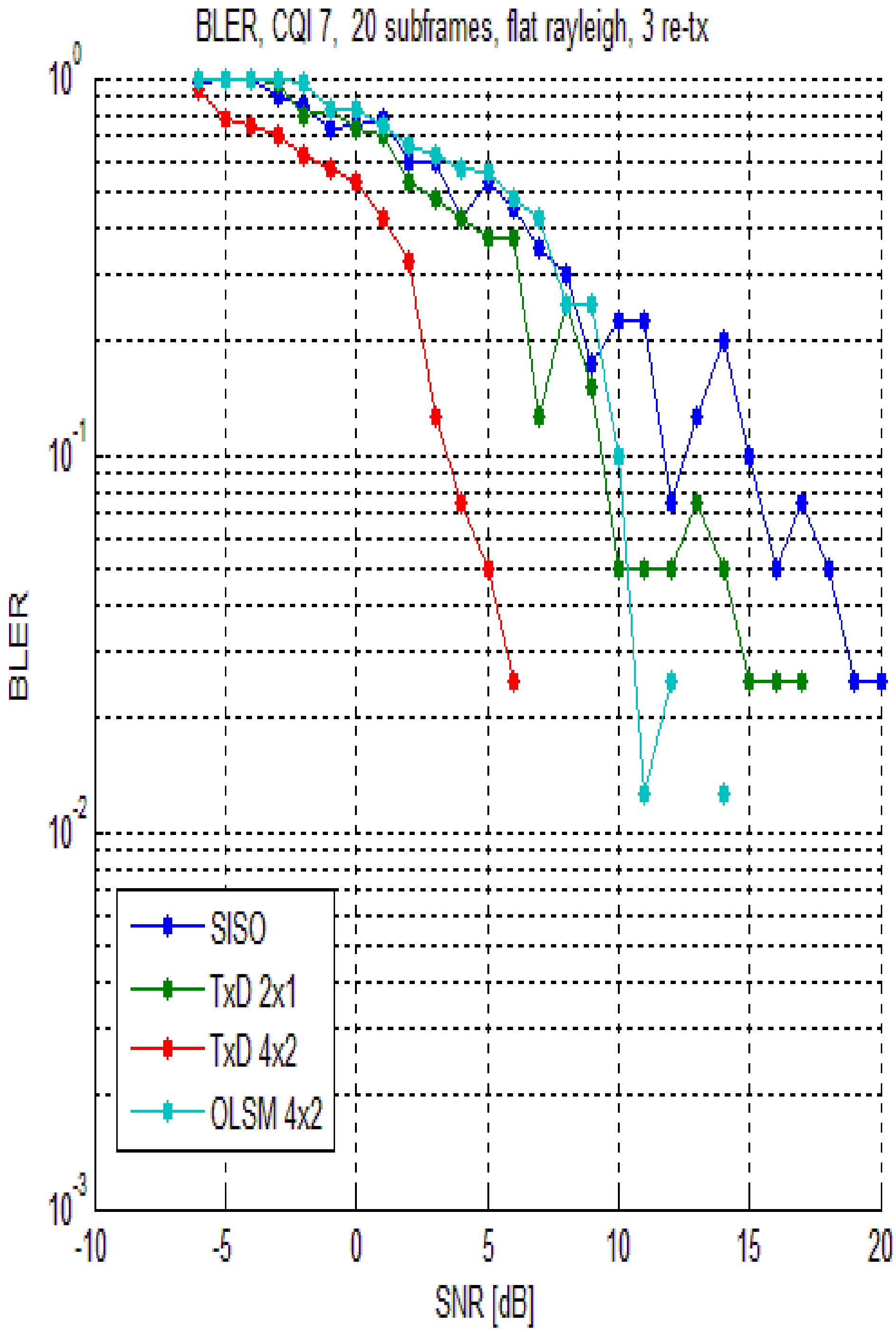

Figure 5.10: BLER, flat Rayleigh channel, 3 retransmissions

When CQI is 7, Channel type is PedB (Pedestrian 10km/h) and retransmission is 3, system is analyzed. When CQI is 7, 16QAM is used. According to the Figure 5.11 only for only TxD 4x2, BLER is acceptable and smaller than %10 for CQI=7.

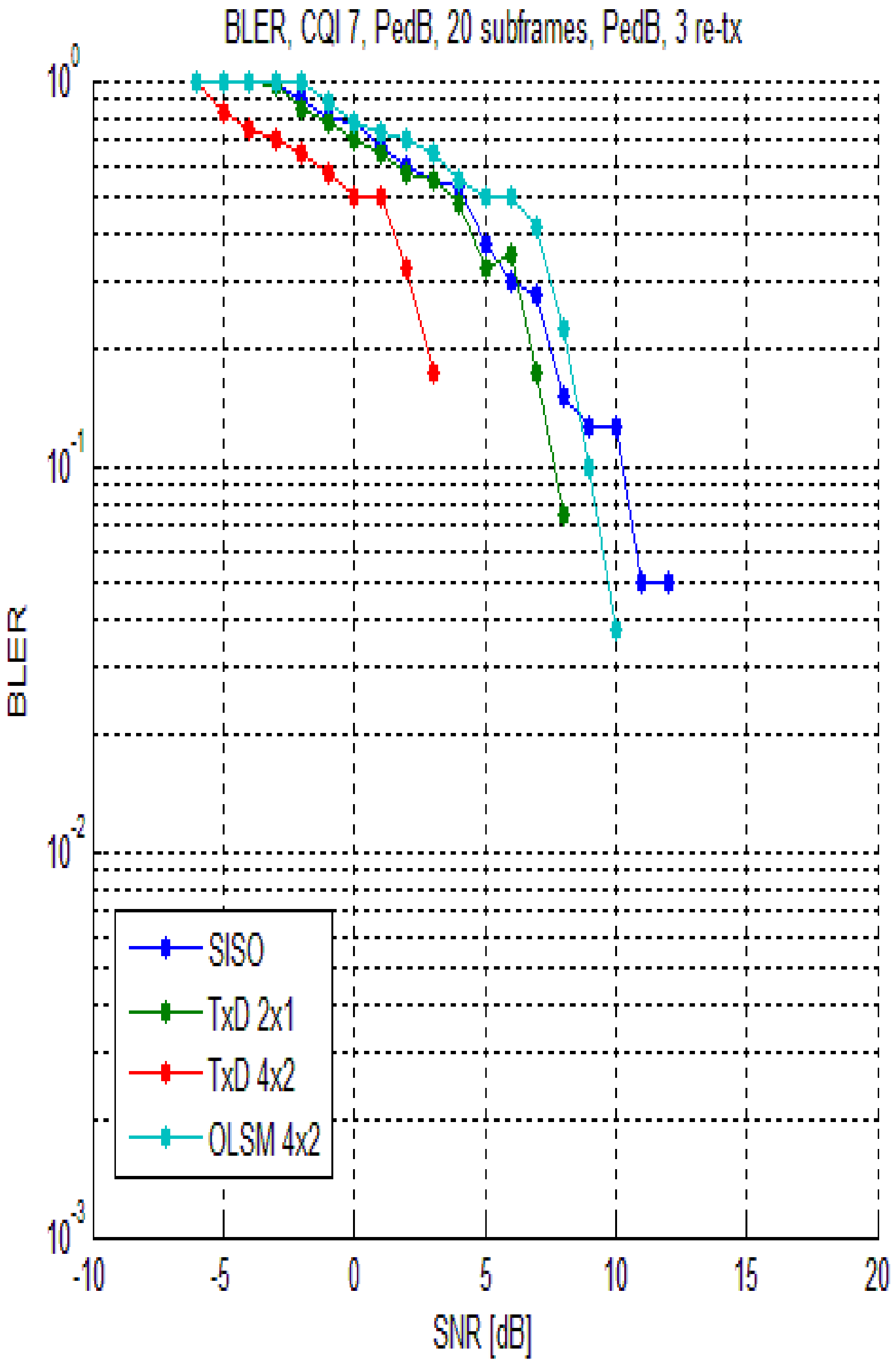

Figure 5.11: BLER, PedB channel, 3 retransmissions

When CQI is 7, Channel type is PedB (Pedestrian 10km/h) and retransmission is 0, system is analyzed. When CQI is 7, 16QAM is used. According to the Figure 5.12 only for only TxD 4x2, BLER is acceptable and smaller than %10 when CQI is 7. The difference between 0 and 3 retransmission mechanism can be seen in Figure 5.12, for all modes, in 0 retransmission mechanism, BLER could reach %10 later compared to 3 retransmission. In conclusion it is expected that BLER decreased when the symbol is retransmitted.

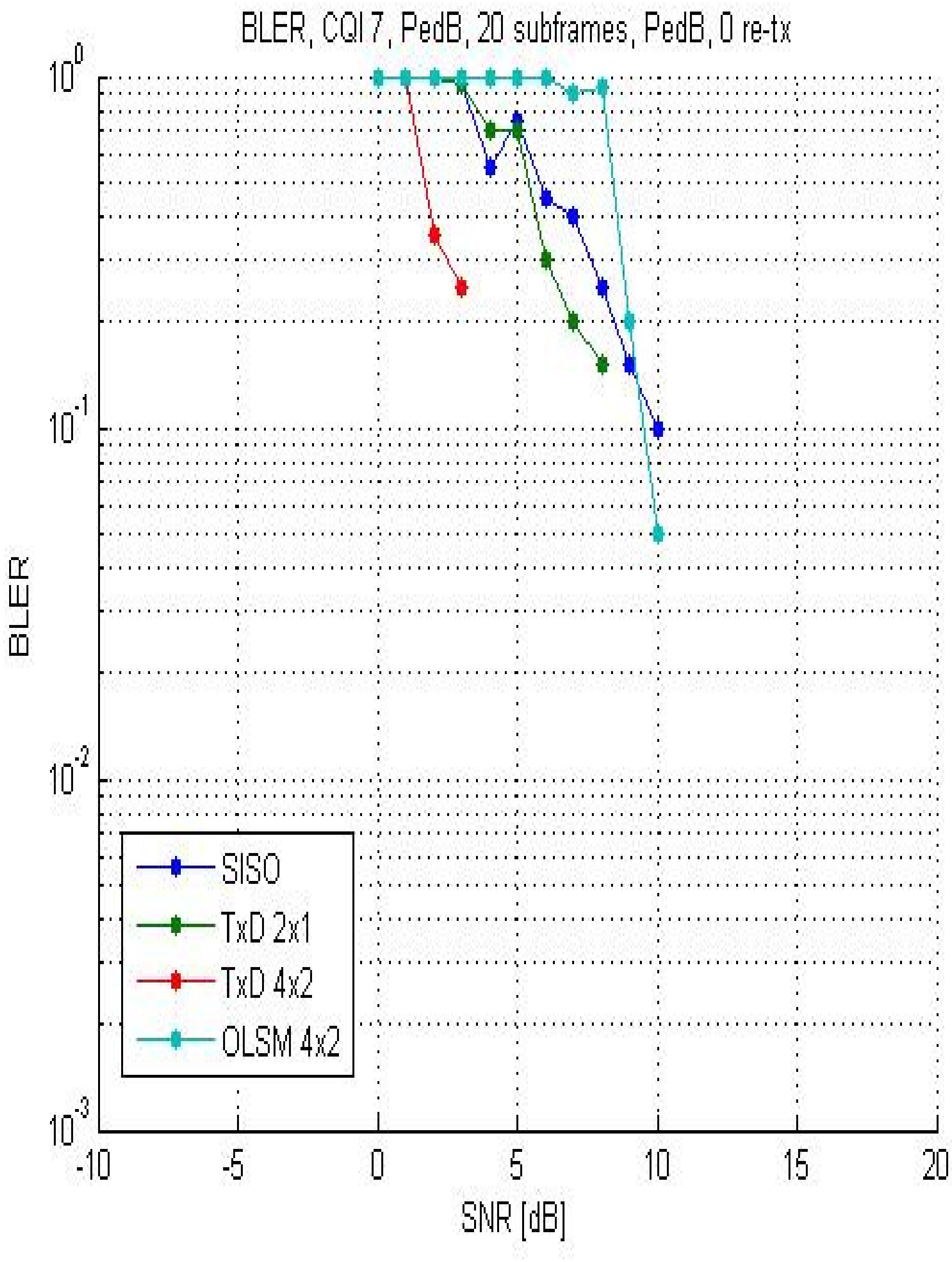

Figure 5.12: BLER, PedB channel, 0 retransmission

5.2.4 Throughput results for 20 MHz

When CQI is 7, Channel type is flat Rayleigh and retransmission is 3, system is analyzed 16QAM is used for channel coding. The maximum throughput can be obtained by using TxD 4x2. According to the Figure 5.4 the SNR is 5 dB when CQI is 7 and SISO, 2x1, 4x2, OLSM 4x2 provide respectively 12, 14, 20, 18 Mbps throughput for 20 MHz.

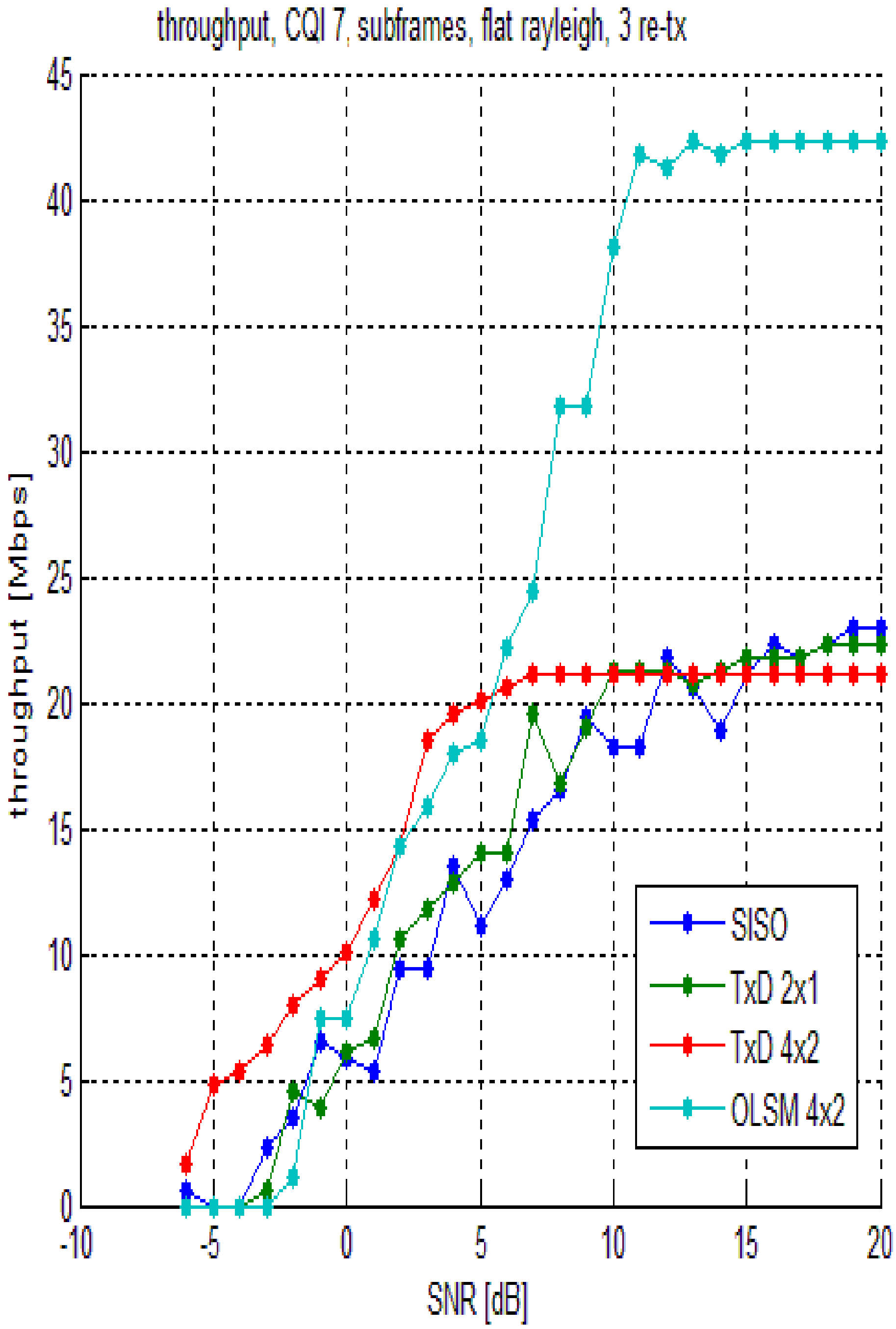

Figure 5.13: Throughput (20 MHz), flat Rayleigh channel, 3 retransmissions

When CQI is 7, Channel type is PedB (10 km/h) and retransmission is 3, system is analyzed. 16QAM is used for channel coding. The maximum throughput can be obtained again by using mode 3 OLSM 4x2. According to the Figure 5.14 the SNR is 7-8 dB when CQI is 7 and SISO, 2x1, 4x2, OLSM 4x2 provides respectively 15, 15, 21, 21 Mbps throughput for 20 MHz.

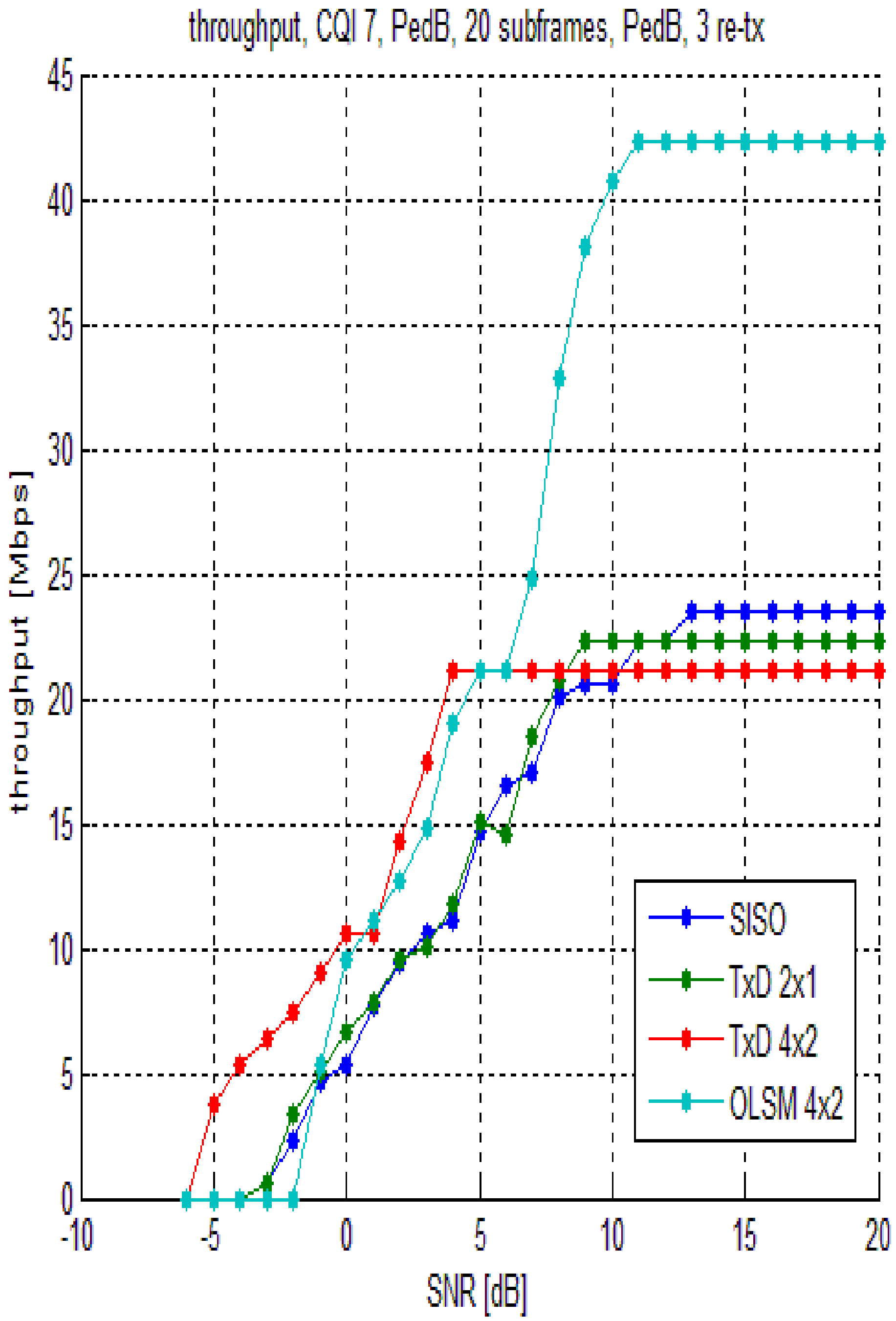

Figure 5.14: Throughput (20 MHz), PedB channel, 3 retransmissions

As it can be seen in the Figure 5.15, Throughputs are analyzed for 20 MHz, when no retransmission mechanism is used. As we expect, no transmission provided some improvement in throughput, but the performance is very similar to the system when 3 retransmissions are used. The reason for the similar performance is that in channel the switching between the modulation and coding schemes can be done perfectly and hardly any retransmissions are required.(although allowed if necessary)

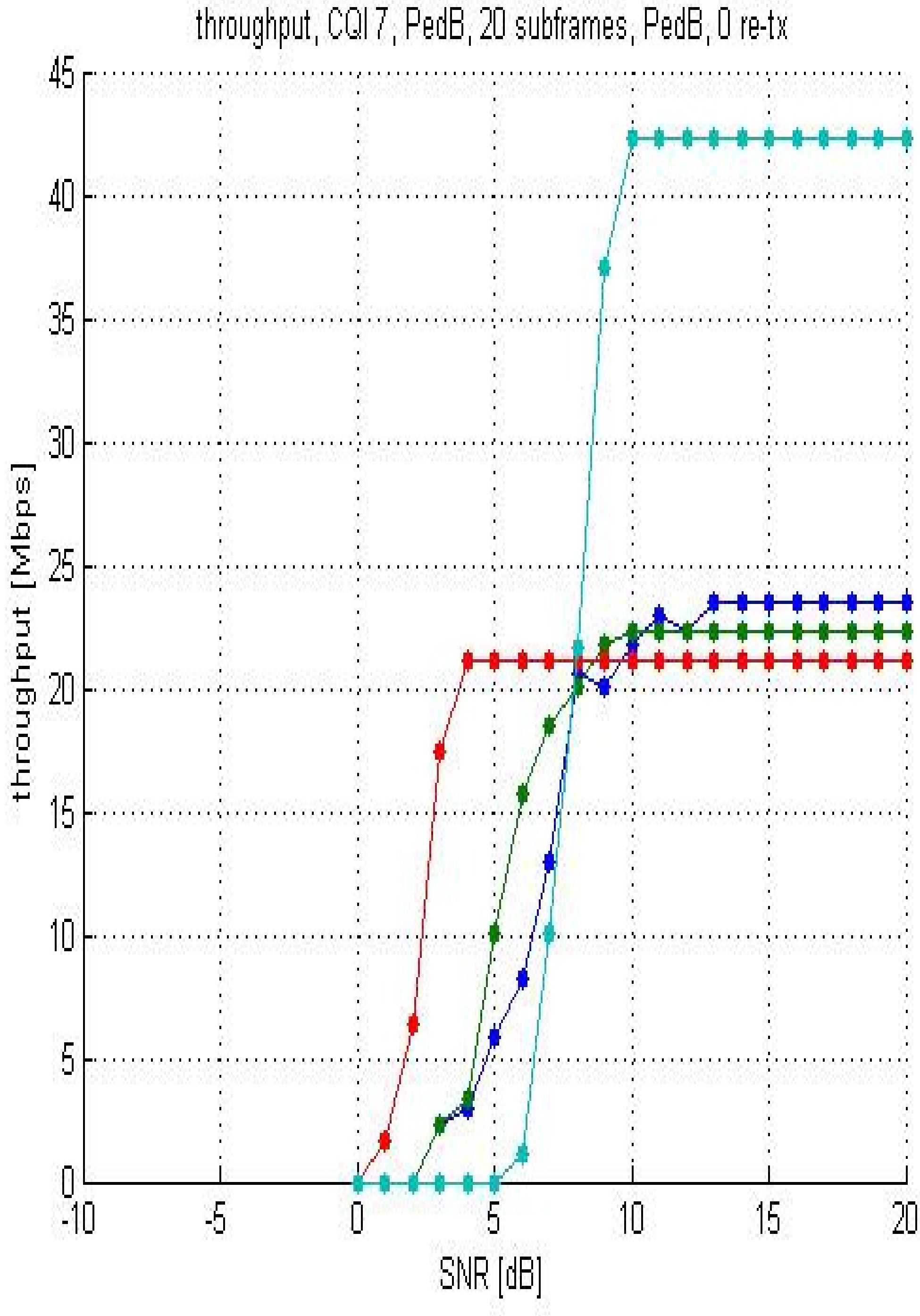

Figure 5.15: Throughput (20 MHZ), PedB channel, 0 retransmission

Finally, AWGN channel is selected to obtain best performance in LTE simulator. Retransmission mechanism, HARQ is also switched off so no retransmissions are performed and SISO (Single Input Single Output)) is used. When CQI is between 1 and 6, QPSK is used. When CQI is between 7 and 9, 16QAM is used and When CQI is between 10 and 15, 64QAM is used as coding scheme. As it can be seen in the Figure 5.16 and 5.17, we can obtain the maximum throughput which is approximately 60 Mbps when CQI is 15.

When throughput results in 20 MHz compared to the system capacity (C) which is defined in Shannon capacity formula, of an AWGN channel calculated according to the following formula.

C = F*B log2(1 + SNR) **(5.1**

SNR is the Signal to Noise Ratio, B the bandwidth occupied by the data subcarriers, and F a correction factor. The bandwidth B:

B= (Nsc* Ns*Nrb)/ Tsub **(5.2)**

Nsc is equal to 12 is the number of subcarrier in one RB. Ns is equal to 14 which is the number of OFDM symbols in one subframe. Nrb is equal to 100 (it is selected for 20 MHz) which is the number of resource blocks. Tsub is equal to 0.001 second which is the duration of one subframe

The transmission of an OFDM signal requires also the transmission of a CP to avoid inter-symbol interference and the reference symbols for channel estimation. Therefore, the Shannon formula is adjusted in equation by the factor F. This factor F accounts thus for the inherent system losses and is calculated as

F = ((Tframe – Tcp)/ Tframe) * ((Nsc*(Ns/2) -4) / Nsc*(Ns/2)) **(5.3)**

Tframe is the fixed frame duration. (10ms). Tcp is the total cyclic prefix (CP) time of all OFDM symbol within one subframe. (0.7 ms)

When we make a calculation for 20MHz,

B= (12*14*100)/0.001 ~ 17 MHz

Transmission bandwidth is not equal to 20 MHz since the number of resource blocks (100) is selected to set transmission bandwidth to 90 percent of the total bandwidth.

F= ((10-0.7)/10) * (12 *(14/2) -4) / 12*(14/2)) ~ 0.89

SNR is equal to 19.5 when CQI is 15 so the system capacity:

C= (0.89) * 17 * log2 (1+19.5) ~ 66 Mbps.

So as it can be seen in Figure 5.17, the similar result could be obtained in LTE simulator when CQI is 15 and BLER is acceptable in Figure 5.18.
In contrast to AWGN channel high throughput could not be obtained in Pedestrian and Rayleigh Fading channel. It is thought that the following parameters decrease the throughput in these two channel types. Multiple Users (2) are tested in Pedestrian and Rayleigh Fading channel that decreases throughput. Mobility and the characteristic of channel types decrease throughput. Transmission Time Interval (TTI) which is selected as 20 also affects throughput slightly for these channel types.

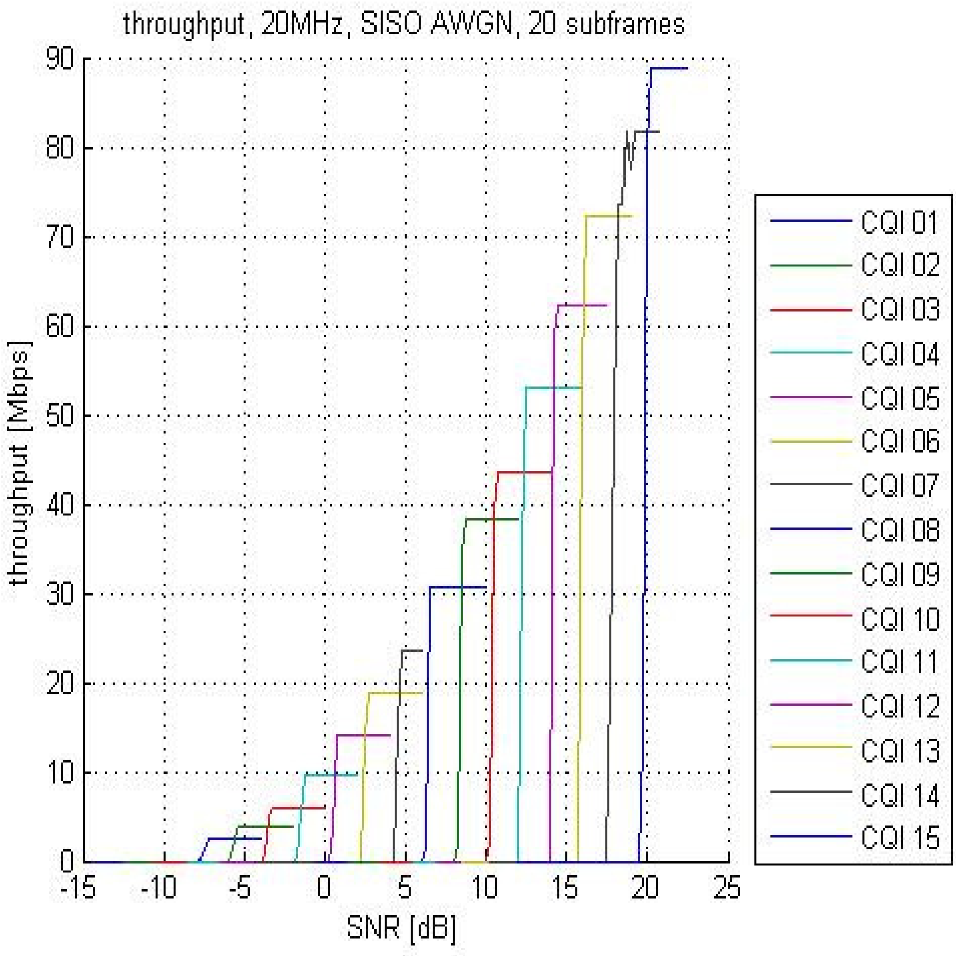

Figure 5.16: Throughput (20 MHZ), AWGN channel, 0 retransmission

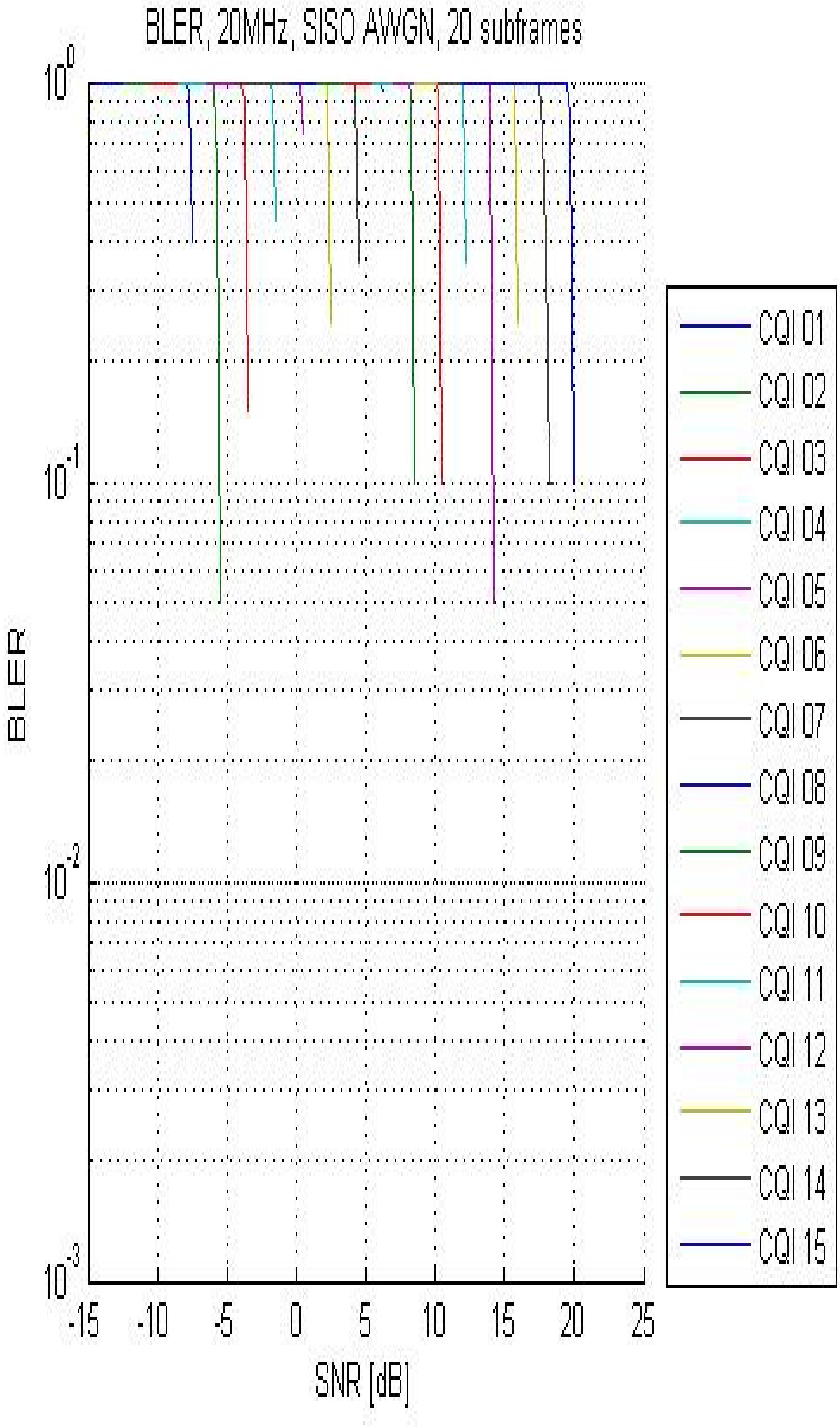

Figure 5.17: BLER (20 MHZ), AWGN channel, 0 retransmission

6. CONCLUSION

In thesis, the purpose was to achieve simulation of LTE network to analyze the performance by using throughput and BLER findings. Before performing the simulation in LTE, the other competing technologies are also were researched and they were compared to LTE. These are UMB and WiMAX and they are explained briefly. Actually LTE, UMB and WiMAX do not comply full requirements of IMT-Advanced systems which are defined by ITU as fourth generation so that for LTE technology, 3GPP has made progress on how to enhance LTE to meet requirements of IMT-Advanced in a project called LTE-Advanced and ITU confirmed LTE-Advanced as 4G. It is expected that all specifications in LTE-Advanced are completed in March of 2011, with earliest availability for deployment in 2012 and it is thought that LTE-Advanced will be first true 4G system available.

In this research, LTE was simulated in MATLAB according to the 3GPP technical specification (TS36.942). Even though LTE is accepted as pre-4G technology, LTE and LTE Advanced recommend basically same capabilities. Additionally in LTE Advanced the following capabilities are supported which are wider bandwidth supports for up to 100 MHz via aggregation of 20 MHz blocks, uplink MIMO (two transmit antennas in the device); downlink MIMO of up to 8 by 8. So that it is thought that LTE simulation could be used to have an idea for 4G technology.

In simulation, the bandwidth, antenna techniques (SISO, 2x1, 4x2), channel type, FFT size, CQI were set as variables. Rayleigh Fading and AWGN channel types are used and the system was simulated according to different bandwidth, channel types and multiple antennas. Then BLER and throughput were analyzed for different situation. The maximum throughput could be obtained in AWGN channel which is approximately 60 Mbps when the bandwidth is 20MHz, CQI is 15 and BLER is smaller than 10% which was acceptable. The results were also calculated according to Shannon capacity formula and it was seen that expected throughput for AWGN was obtained in simulator. In contrast to AWGN channel high throughput could not be obtained in Pedestrian and Rayleigh Fading channel. It is thought that the following parameters decrease the throughput in these two channel types. Multiple

Users (2) were tested in Pedestrian and Rayleigh Fading channel that decreases throughput. Mobility and the characteristic of channel types decreased throughput. Transmission Time Interval (TTI) which was selected as 20 also affected throughput slightly for these channel types.

In simulation, LTE recommended high data rate which is near to the target of 4G defined by ITU so it could be expected that LTE Advanced will comply full requirements of IMT-Advanced systems.

Currently, nine telecommunication providers have already launched LTE in 2010 and an additional 11 expected before the end of the year. More than 250 companies have publicly expressed interest in deploying LTE networks, including CDMA, WiMAX and GSM operators. So that LTE is expected to be the leading choice for next-generation OFDMA networks over the next decade for all wireless carriers.

Currently used LTE networks offer speeds in mobile broadband services between 20-80 Mbit/s with a maximum speed of 100 Mbit/s for 20 MHz bandwidth in 2.5GHz frequency. LTE has become the technology platform of choice as GSM-UMTS and Code Division Multiple Access (CDMA)/One Carrier Evolved; Data Optimized (EV-DO) operators are making strategic, long-term decisions on their next-generation platforms.

REFERENCES

[1] **Dahlman, E., Parkvall, S. and Beming, P.,** 2007: 3G Evolution: HSPA and LTE for Mobile Broadband, 1st ed. Academic Press, 2007.

[2] **3GPP**, 2008: Technical specification group radio access network; spatial channel model for Multiple Input Multiple Output (MIMO) simulations (release 8), 3GPP, Tech. Rep. 25.996 V8.0.0, Dec. 2008.

[3] **Kondra, V. A**., 2009: Technology Case Study on the Emerging 4G Broadband Wireless Network; Overview of Current Technologies and Up-to-Date Analysis of Competing Technologies, PHd Thesis, State University of New York Institute of Technology, USA.

[4] **3GPP LTE Specs: Url-1** <http://www.3gpp.org/ftp/Specs/html-info/36-series.htm> accessed at March, 2010

Url-2 <http://www.3gpp.org/ftp/Specs/html-info/25814.htm > accessed at March, 2010

[5] **Prasad, A.,** 2007: UMB Network Evolution, Tutorial, 03.11.2007, India

[6] **Qualcomm White Paper:** UMB Technology Overview, 04.12.2008, from

Url-1 <http://www.qualcomm.com/coinmon/documents/white_papers/> accessed at March, 2010

[7] **WiMax Forum**, 2010:

Url-1 <http://www.wimaxforum.org/home/> accessed at March, 2010

[8] **3GPP TR 25.913**, 2006: Requirements for Evolved UTRA (E-UTRA) and Evolved UTRAN (E-UTRAN), v.7.3.0

[9] **3GPP**, 2008: Technical specification group radio access network, (E-UTRA) and (E-UTRAN), Overall description, Stage 2, 2008.

[10]**Marques, V.,** and **Aguiar, R. 2003;** An IP-based QoS architecture for 4G operator Scenarios, Wireless Communications, IEEE, 10, pp. 54-62.

[11] **Rappaport T. S.,** 2002**:** Wireless Communications: Principles and Practice, 2nd edition, Prentice Hall.

[12]**Nee R. V.** and **Prasad, R.**, 2000: OFDM Wireless Multimedia Communications, Norwood, Artech House.

[13]**Brooks A. C., Hoelzer S.J.,** 2001: Design and Simulation of Orthogonal Frequency Division Multiplexing (OFDM) Signaling Final Report, Bradley University, Peoria, Illinois, USA.

[14]**Kyun Y., Prasad R.,** 2006: 4G Roadmap and Emerging Communication Technologies, Artech House pp 12-13.

[15] **Stüber, G. L., Barry, J. R., McLaughlin, S. W., Li, Y. G., Ingram, M. A., Pratt, T. G.,** 2004: Broadband MIMO-OFDM Wireless

Communications," Proceedings of the IEEE, Feb. 2004, vol. 92, no. 2, pp. 271-294.

[16] Sivanesskumar, S. 2009: Performance Analysis of Multi-Carrier Code Division Multiple Access System under Clipping Noise, European Journal of Scientific Research ISSN 1450-216X Vol.38 No.4 , pp 590-595.

[17] Myung H. G., Lim, J., Goodman, D. J., 2006: Single Carrier FDMA for Uplink Wireless Transmission, IEEE Vehicular Technology Magazine, vol. 3 no.1,Septemer 2006, pp. 30-38.

[18] Myung Hyung G., Junsung L., Goodman D. J., 2006: Peak-To-Average Power Ratio of Single Carrier FDMA Signals with Pulse Shaping, 17th Annual IEEE International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC '06), Helsinki, Finland, September.

[19] Soltani N., 2009**:** Comparison of Single-Carrier FDMA vs. OFDMA as 3GPP Long-Term Evolution Uplink, Project, Stanford University, USA, 4.12.2009.

[20] Ikuno J. C., Wrulich M., Rupp M., 2010: System Level Simulation of LTE Networks, 17th European Signal Processing Conference (EUSIPCO)**,** Institute of Communications and Radio-Frequency Engineering**,** Vienna University of Technology**,** Austria.

[21] Mehlfuhrer, C., Wrulich, M., Ikuno, J. C., Bosanska, D., Rupp M., 2009: Simulating the Long Term Evolution Physical layer, 17th European Signal Processing Conference (EUSIPCO), Institute of Communications and Radio-Frequency Engineering, Vienna University of Technology, Austria.

[22] LTE Simulator, 2009: **Url-1** <//www.nt.tuwien.ac.at/ltesimulator/> accessed at May, 2010

[23] Ikuno, J. C., Wrulich, M., and Rupp, M. 2009: Performance and modeling of LTE H-ARQ, in Proc. International ITG Workshop on Smart Antennas (WSA 2009), Berlin, Germany, Feb. 2009.

[24] MEX-les Guide:

Url-1 <http://www.mathworks.com/support/tech-notes/1600/1605.html> accessed at May 2010

[25] Salo, J., Galdo, G. Del., Salmi, J., Kyosti, P., Milojevic, M., Laselva, D. and Schneider, C., 2005: MATLAB implementation of the 3GPP Spatial Channel Model (3GPP TR 25.996).

Url-1 <http://www.tkk./Units/ Radio/scm/> accessed at May, 2010

[26] Slock D.T.M., 2008**:** Performance Analysis of General Pilot-Aided Linear Channel Estimation in LTE OFDMA Systems with Application to Simplified Mmse Personal, Indoor and Mobile Radio Communications (PIMRC), IEEE 19th International Symposium, Cannes, France, 15-18 September.

APPENDICES

APPENDIX A : Spectrum Analyzes for Future LTE Deployment

APPENDIX A

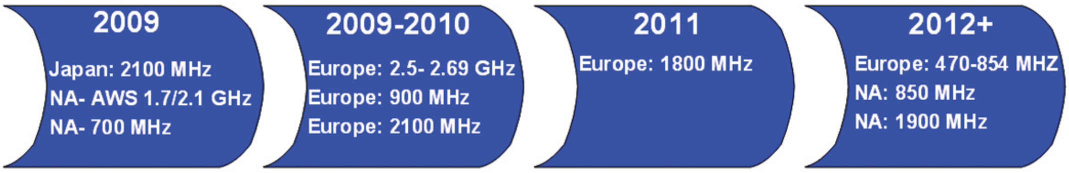

Figure A: LTE Deployment Scenario [14]

Table A: Candidate Bands for LTE [14]

Band	Uplink (MHz)	Downlink (MHz)	Carrier Bandwidth (MHz)	Comments
700 MHz	746-763	776-793	1.25 5 10 15 20	Digital Dividend. U.S. commercial spectrum auctioned Q108. "D" block to be re-auctioned. Potential future alignment with Europe
AWS	1710-1755	2110-2155	1.25 5 10 15 20	U.S. Auctions completed September 2006
IMT Extension (Paired)	2500-2570	2620-2690	1.25 5 10 15 20	Initially Western Europe. Offers a unique opportunity for the deployment of LTE in channels of up to 20 MHz.
IMT Extension (Unpaired)	2570-2620		1.25 5 10 15 20	Potential for LTE –TDD in Europe and Asia Pac.
GSM 900	880-915	925-960	1.25 5 10 15 20	Reallocate this spectrum to advanced networks, such as LTE, from 2009 onwards
UMTS Core	1920-1980	2110-2170	1.25 5 10 15 20	Europe and Asia Pac. Potential for unused WCDMA carriers
GSM 1800	1710-1785	1805-1880	1.25 5 10 15 20	Europe and Asia Pac. Refarm underutilized band along with GSM 900
PCS 1900	1850-1910	1930-1990	1.25 5 10 15 20	U.S. Refarm after new 700 MHz and AWS spectrum is consumed.
Cellular 850	824-849	869-894	1.25 5 10 15 20	U.S. Refarm after new 700 MHz and AWS spectrum is consumed.
Digital Dividend	470-854		1.25 5 10 15 20	Identified at WRC-07.

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