Adaptive Mobility and Resource Management for 4G Wireless Inter-networking

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CERTIFICATE

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DEDICATION

To Almighty Allah and His Beloved Prophet Muhammad (PBUH)

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With the blessings of Almighty Allah we have been able to complete our project in time, we pray to Almighty Allah for His mercy and blessings, so as to have successful life in future.

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Muhammad Tayyab Malik

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LIST OF ACRONYMS

ACK	Acknowledgement
AM	Acknowledged Data Transfer Service at the RLC
ARQ	Automatic Repeat Request
AS	Access Stratum
BCCH	Broadcast Control Channel
BCH	Broadcast Channel
BMC	Broadcast/Multicast Control
BSS	Base Station Subsystem
CBR	Constant Bit Rate
CBS	Cell Broadcast Service
CCCH	Common Control Channel
CDMA	Code Division Multiple Access
CN	Core Network
CPCH	Common Packet Channel
CS	Circuit Switched
CTCH	Common Traffic Channel
DCCH	Dedicated Control Channel
DCH	Dedicated Channel
DPCCH	Dedicated Physical Control Channel
DPDCH	Dedicated Physical Data Channel
DSCH	Downlink Shared Channel
DTCH	Dedicated Traffic Channel
EURANE	Enhanced UMTS Radio Access Network Extensions for NS2
FACH	Forward Access Channel
FEC	Forward Error Correction
GGSN	Gateway GPRS Support Node
GMSC	Gateway MSC
GPRS	General Packet Radio System
GSM	Global System for Mobile communication

HHO	Hard Handovers
HLR	Home Location Register
HO	Handovers
HS-DSCH	High Speed Downlink Shared Channel
MAC	Medium Access Control
PC	Power Control
PCCH	Paging Control Channel
P-CCPCH	Primary Common Control Physical Channel
PCH	Paging Channel
PCPCH	Physical Common Packet Channel
PDCP	Packet Data Converge Protocol
PDU	Protocol Data Unit
PDSCH	Physical downlink Shared Channel
PICH	Paging Indicator Channel
PRACH	Physical Random Access Channel
PS	Packet Switched
RACH	Random Access Channel
RLC	Radio Link Control
RNC	Radio Network Controller
RNS	Radio Network Subsystem
RRC	Radio Resource Control
RRM	Radio Resource Management
S-CCPCH	Secondary Common Control Physical Channel
SDU	Service Data Unit
SHO	Soft Handovers
SRNC	Serving RNC
TD-CDMA	Time Division Code Division Multiple Access
TF	Transport Format
TR	Transparent Data Transfer Service at the RLC
TTI	Transmission Time Interval
UE	User Equipment

UMTS	Universal Mobile Telecommunication System
UM	Unacknowledged Data Transfer Service at the RLC
USIM	UMTS Subscriber Identity Module
UTRAN	Universal Terrestrial Radio Access Network
VBR	Variable Bit Rate
VLR	Visitor Location Register
WCDMA	Wireless Code Division Multiple Access
3GPP	3rd Generation Partnership Project
AP	Access Point
BS	Base Station
CN	Correspondent Node
ND	Neighbor Discovery
MIH	Media Independent Handover
RA	Router Advertisement
RS	Router Solicitation
SAP	Service Access Point

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ABSTRACT

This research is aimed to explore innovative solutions to network inter-connectivity and wireless resource management, so as to allow efficient and transparent services to multimedia users across heterogeneous networking platforms. This project, in particular, will focus on the integration of UMTS system and WiMax. A mobility and resource adaptation frame work based on Mobile IPv6 and its enhancements will be developed to glue together the different radio networks to provide pervasive access to the Internet in the near future. In this project a simulations based approach has been adopted to determine the total cell throughput by implementing different traffic models (CBR, FTP, and VOIP) for different number of users on both Wimax and UMTS networks. Also in UMTS network the performance of HSDPA is evaluated with both round robin scheduling and best channel first scheduling. Simulation results show that under ideal condition high cell throughput is obtained by round robin scheduling. In WiMax throughput performance of different modulation schemes and same modulation schemes with different channel coding is determined. Simulations results show that throughput performance is higher for higher channel coding. In the end a simple simulations scenario is carried out on integrated model in which UMTS and WiMax networks are integrated in which vertical handover is triggered by the speed of the user who is moving from one network to the other. Simulation results shows that vertical handover is triggered quickly when user is moving fast and traffic flow is seamlessly switched to the other network .

Chapter 1

INTRODUCTION

Although it has history of more than a century, wireless transmission has found widespread use in communication systems only in the last 15-20 years. Currently the field of wireless communications is one of the fastest growing segments of the telecommunications industry. Wireless communication systems, such as cellular, cordless and satellite phones as well as wireless local area networks (WLANs) have found widespread use and have become an essential tool in many people's every day life, both professional and personal. To gain insight into the wireless market momentum, it is sufficient to mention that it is expected that the number of worldwide wireless subscribers in the years to come will be well over the number of wireline subscribers due to its advantages compared to wireline systems. The most important of these are mobility and cost savings.

1.1 EARLY MOBILE TELEPHONY

In 1946, the first public mobile telephone system, known as Mobile Telephone System (MTS), was introduced in 25 cities in the United States. Due to technological limitations, the mobile transceivers of MTS were very big and could be carried only by vehicles. Thus, it was used for car-based mobile telephony. MTS was an analog system, meaning that it processed voice information as a continuous waveform. This waveform was then used to modulate/demodulate the RF carrier. The system was half-duplex, meaning that at a specific time the user could either speak or listen. To switch between the two modes, users had to push a specific button on the terminal.

1.2 ANALOG CELLULAR TELEPHONY

IMTS used the spectrum inefficiently, thus providing a small capacity. Moreover, the fact that the large power of BS transmitters caused interference to adjacent systems plus the problem of limited capacity quickly made the system impractical. A solution to this problem was found during the 1950s and 1960s by researchers at AT&T Bell Laboratories, through the use of the cellular concept, which would bring about a revolution in the area of mobile telephony a few decades later. It is interesting to note that this revolution took a lot of people by surprise, even at AT&T. They estimated that only one million cellular customers would exist by the end of the century; however today, there are over 100 million wireless customers in the United States alone.

Originally proposed in 1947 by D.H. Ring, the cellular concept replaces high-coverage BSs with a number of low-coverage stations. The area of coverage of each such BS is called a 'cell'. Each BS is connected via wires to a device known as the Mobile Switching Center (MSC). MSCs are interconnected via wires, either directly between each other or through a second-level MSC. Second-level MSCs might be interconnected via a third-level MSC and so on. MSCs are also responsible for assigning channel sets to the various cells

The low coverage of the transmitters of each cell leads to the need to support user movements between cells without significant degradation of ongoing voice calls. However, this issue, known today as handover, could not be solved at the time the cellular concept was proposed and had to wait until the development of the microprocessor, efficient remote controlled Radio Frequency (RF) synthesizers and switching centers.

1.3 1G CELLULAR

The first generation of cellular systems (IG systems) was designed in the late 1960s and, due to regulatory delays, their deployment started in the early 1980s. These systems can be thought of as descendants of MTS/IMTS since they were of also analog systems. The first service trial of a fully operational analog cellular system was deployed in Chicago in 1978. The first commercial analog system in the United States, known as Advanced Mobile Phone System (AMPS), went operational in 1982 offering only voice transmission. Similar systems were used in other parts of the world, such as the Total Access Communication System (TACS) in the United Kingdom, Italy, Spain, Austria, Ireland, MCS-Ll in Japan and Nordic Mobile Telephony (NMT) in several other countries. AMPS is still popular in the United States but analog systems are rarely used elsewhere nowadays. All these standards utilize frequency modulation (PM) for speech and perform handover decisions for a mobile at the BSs based on the power received at the BSs near the mobile. The available spectrum within each cell is partitioned into a number of channels and each call is assigned a dedicated pair of channels. Communication within the wired part of the system, which also connects with the Packet Switched Telephone Network (PSTN), uses a packet-switched network.[3].

1.4 2G CELLULAR

Second-generation cellular systems are well-established, with over two billion users throughout the world. Technologies that qualify as 2G systems include GSM, CdmaOne and TDMA. Although capabilities vary between technologies and continue to evolve, the key attributes of 2G systems are typically:

- Digital transmission.
- Voice capabilities
- Circuit switched data capabilities up to around 60 kbits/s
- Packet switched data capabilities are also up to around 60 kbits/s.

When 2G was deployed it was predicted that 2G systems might have a lifetime of around 20 years. Many of the licenses given to operators reflected this, with end dates ranging from 2008 to 2020, but with many clustered around 2010 to 2015. When setting this end date it was thought that the spectrum might be re-farmed for 3G usage and indeed this thinking still persists widely among many regulatory bodies. However, it now appears that complete re-farming is unlikely for some considerable time. Main reasons for that are First, 3G systems are taking longer to introduce than previously expected, so that even if 3G were to entirely replace 2G, this might take another 10 years.[3] Second, most operators now expect to deploy 3G in cities and along major communication corridors, but not in the more rural areas where coverage will continue to be provided by 2G for some time. Third, operators see 2G as being the 'roaming standard' for quite some time since, regardless of which technology their local operator has deployed, most handsets will retain 2G capabilities. It is likely that, in 2015, 2G will still be widely deployed.

Indoor solutions struggle to get footings. Though 2G systems have had great success in providing wireless communications outside of buildings and in providing coverage inside the building based on signals penetrating from outside. Their success in dedicated in-building deployments has been much more limited. Many attempts have been made by manufacturers to deploy 2G in-building solutions based on a range of different technologies, but these have not had any real success. It is more likely that a short range wireless solution will be deployed.

1.5 3G CELLULAR

In 2006, 3G systems were starting to be widely deployed. There is a range of technologies for 3G, all based around CDMA technology and including W-CDMA (with both FDD and TDD variants), TD-SCDMA (the Chinese standard) and Cdma2000 (the US standard). The key characteristics of 3G systems include the ability to carry video calls and video streaming material and realistic data rates extending up to 384kbits/s in both packet and circuit switched modes.

For some time there was heated discussion about the likelihood of 3G being successful. With slow deployment and limited subscriber growth between 2000 and 2004 there were many who were quick to predict disaster. Others pointed to the growth of WiFi and the imminent arrival of WiMax and predicted that this would significantly disrupt 3G deployment. However, between 2004 and 2006 these doubts slowly faded. Operators continued to deploy 3G networks and subscriber numbers started to rise more quickly. Handsets became more attractive and the lower cost per call of 3G compared to 2G became increasingly clear. In 2006, few would now doubt that 3G networks will be widely deployed, that they will eventually take over from 2G in most areas and that subscribers will increasingly migrate from 2G. However, growth may occur more because the cellular operators 'push' the new technology than the subscribers demand it.

The original premise of 3G was that end users would like to be able to do much more than just voice and low-speed data with their mobile phones. In addition, they might like to have video calls, watch video clips such as key sporting highlights, and transfer volumes of data requiring more than 100 kbits/s, browse websites when outdoors and much more. It was suggested that 2G systems were unable to do this because they had an insufficiently high data rate for some applications and insufficient capacity to add video calls and other new applications on to the network. Early launches of 3G networks did indeed stress the ability to have video calls, see video clips and so on. However, in 2006, the situation was that by and large the benefits of 3G were being realized as increased voice call capacity, allowing more calls for less cost. Video calling capabilities were not being widely promoted, instead downloads of music videos were offered. 3G networks will be deployed by virtually all of the major operators and subscribers will be transferred over to the new networks. 3G will probably have a lifetime similar to that of 2G - in the vicinity of 20 years. With enhancements to the standard this lifetime could be even longer.

1.6 4G CELLULAR

With the introduction of 1 G in 1982, 2G in 1992 and 3G around 2004, the deployment of 4G some time around 2014-2018 might look like a fairly certain bet. In the last few years there has been much discussion around 4G, with conferences and books published. Some thought that it was appropriate to start discussion of 4G in 2002 given the likely ten years it would take to complete the standard and have equipment developed. An example of this was the Japanese authorities who announced a research program aimed at producing a system capable of delivering 100 Mbits/s to end users. Others argued that 3G had failed, or was inappropriate and 4G should be introduced rapidly in its place. Figure 1.1 suggests why 4G is likely to be different from 3G, and indeed perhaps may not even emerge.

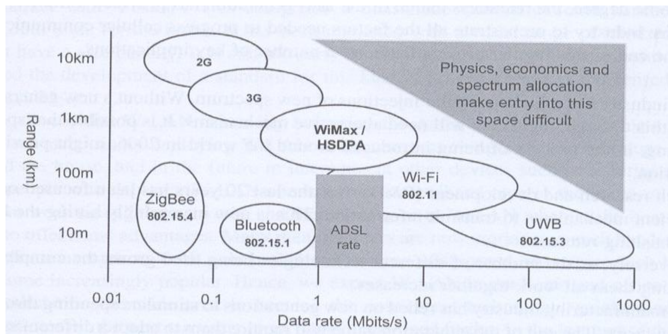

Figure 1.1 Range versus data rates for each of the generations [14]

Figure 1.1 illustrates that there is always a trade-off between range and data rate. Broadly, higher data rates require more spectrum. More spectrum can only be found higher in the frequency band but higher frequency signals have a lower propagation range. Each generation has accepted a shorter range in return for a higher data rate. But, as Figure 1.1 shows, the next 'step' in the process, where 4G might logically fit, is already taken by a mix of 3G enhancements, WiMax and WiFi. Indeed, interestingly, the Japanese plans for a 4G technology talk of an OFDM-based solution in the 3-6 GHz band providing up to 100 Mbits/s of data. This is almost identical to the specification for the latest W-LAN system, 802.1la, which is already available and exceeded by proposals for 802.11n.

Figure 1.1 also tells us much about the structure of future communications networks. Higher data rate systems are preferred but with their shorter range can be economically deployed only in high-density areas. So we might expect a network where 2G is used to cover rural areas, 3G to cover urban and suburban areas along with key transport corridors, and W-LANs providing very high data rates in selected 'hotspot' locations such as airports.

An argument is that there may not be another generation of cellular technology because there may not be sufficient economic justification for the development of a completely new standard. Instead, we might expect to see enhancements to all the different standards making up the complete communications network, with perhaps some new niche standards emerging in areas such as ultra-short range communications. This has not stopped some claiming that they already have a 4G technology. Since there is no widely agreed definition of what comprises a new generation, anyone is free to claim that their technology meets whatever criteria they regard as important, and with the definition of 3G already confused, perhaps we should expect 4G to be even more opaque.

As a final conclusion on the developments taking place in 2006, many of the proposed '4G' solutions were based on OFDM modulation. Claims were being made that OFDM was the new modulation of choice for 4G in the way that CDMA had become for 3G. Therefore, we are not of the view that there needs to be a change to from CDMA to OFDM in order to realize significant gains in throughput. Equally, we are not suggesting that OFDM is inferior to other technologies, it just offers a different mix of trade-offs.

1.7 WIMAX FOR MOBILE APPLICATIONS

Around 2004, WiMax (IEEE 802.16) became increasingly discussed. Although initially designed as a fixed technology, it was being adapted in the standards to become a mobile technology (also known as 802.16e). Major claims were made for WiMax. Some claimed it would have significantly greater range than 3G, higher data rates, would be built into all computing devices and as a result would dominate the market for 3G-like services.

In 2006, the view was that WiMax mobile networks might not be launched until 2007 or 2008. It would then take some time to deploy the networks and seed terminal devices, so it might not be until 2010 that it would be possible to see whether WiMax was having any impact on 3G. Nevertheless, some simple observations suggest that its impact will be limited.

Range The range of a technology is typically dominated by the frequency of operation, the power transmitted and the height of the transmitter and receiver. All but the frequency are likely to be similar between 3G and WiMax. Although the frequency for deployment of WiMax is not certain, the most likely band at 3.5 GHz is higher in frequency than the 3G bands at around 2.1 GHz. Range will, as a result, be lower - perhaps somewhere between 50% and 75% of the range of 3G.

Data rates The technology used for WiMax (OFDM) is not significantly more spectrally efficient than the technology used for 3G (W-CDMA).However, OFDM coupled with a high channel bandwidth will allow greater peak per-user data rates. So, on average, for an equivalent spectrum allocation, users will see similar data rates. In specific situations, where there are few users, it is possible that WiMax will provide a higher data rate. However, in commercial systems, such situations are likely rare.

Cost The network costs of WiMax will likely be higher than for 3G because of the reduced range and hence the need to build more cells. The subscriber subsidy costs may be lower if WiMax is built into processor chips, although this may not apply if users wish to have WiMax handsets.

Hence, our view is that WiMax will not have a major impact on the success of 3G. It is most likely to be deployed by those operators who have not yet rolled out 3G networks.

1.8 MOTIVATION

The second generation (2G) of wireless mobile communication systems was a big hit story only because of the services that it brought to its users and because of its revolutionary technology. At that time along with high-quality speech service, global mobility was the most strong and convincing reason for users to use 2G terminals. Although the third generation (3G) has been launched in most of parts around the globe, but the success story of 2G is almost impossible to repeat. One of the reasons for this assertion is that this evolution from the second generation towards 3G has brought just few novel additional services ,which has left the business model largely unchanged and this thing may not be enough to attract the customers to change their equipment. 3G was too late to consider the lack of innovative and appealing services. Partnership Project (3GPP), the 3G standardization body, which attempted to incorporate in the latest standards some recent services, such as the multimedia broadcast and multicast service (MBMS) center in combination with the IP multimedia system (IMS). However, these minute improvements were made without the possibility to change the access technology properly.

Following the similar paradigm of generational changes, it was thought that the fourth generation (4G) would follow after 3G in a sequential way and it will appear on scene between 2010 and 2015 as an ultra-high-speed broadband wireless network. For example in Asia, the Japanese operator NTT DoCoMo explains 4G by introducing the concept of mobile multimedia: Anywhere, Anytime, Anyone also adding global mobility support; integrated Wireless solution; and customized personal services, which mainly concentrates on public systems and envisions 4G as the extension of 3G cellular service. The emergence of new mobile broadband technologies such as WiMax,ever increasing growth of user demand and limitation of 3G have brought industries and researchers to a thorough reflection on the fourth generation and this was the main motivation for us to choose this project.

Chapter 2

OVERVIEW OF UMTS

2.1 INTRODUCTION

This chapter is intended to be an overview of the 3G Mobile Communication, the UMTS networks. First of all the main parts of UMTS will be depicted. After that, we will speak about the Wireless Code Division Multiple Access (WCDMA) protocol stacks. Then the physical layer and the different channels will be described, and finally some remaining parts of the 3G Mobile Communication basic knowledge will be briefly explained.

The revolution of wireless networking continues to unfold with the gradual emergence of WCDMA-based third generation (3G) cellular technologies, such as Universal Mobile Telecommunications System (UMTS) [16]. The range of offered services in such systems has already extended from basic speech telephony to multimedia and interactive data transfers. In such an environment one can expect that a major portion of the overall traffic will be carried by TCP/IP. Thus, particular attention must be paid to the performance of TCP/IP over 3G wireless networks.

2.2 UMTS ARCHITECTURE

According to the 3rd Generation Partnership Project (3GPP) [8], a UMTS network is a" network operated by a single network operator and consisting of UTRAN access networks (WCDMA and/or TD-CDMA), optionally GSM BSS access networks, and UMTS core network". The main parts of the UMTS specific network components that are shown in Figure 2.1 will be described in the following sections.

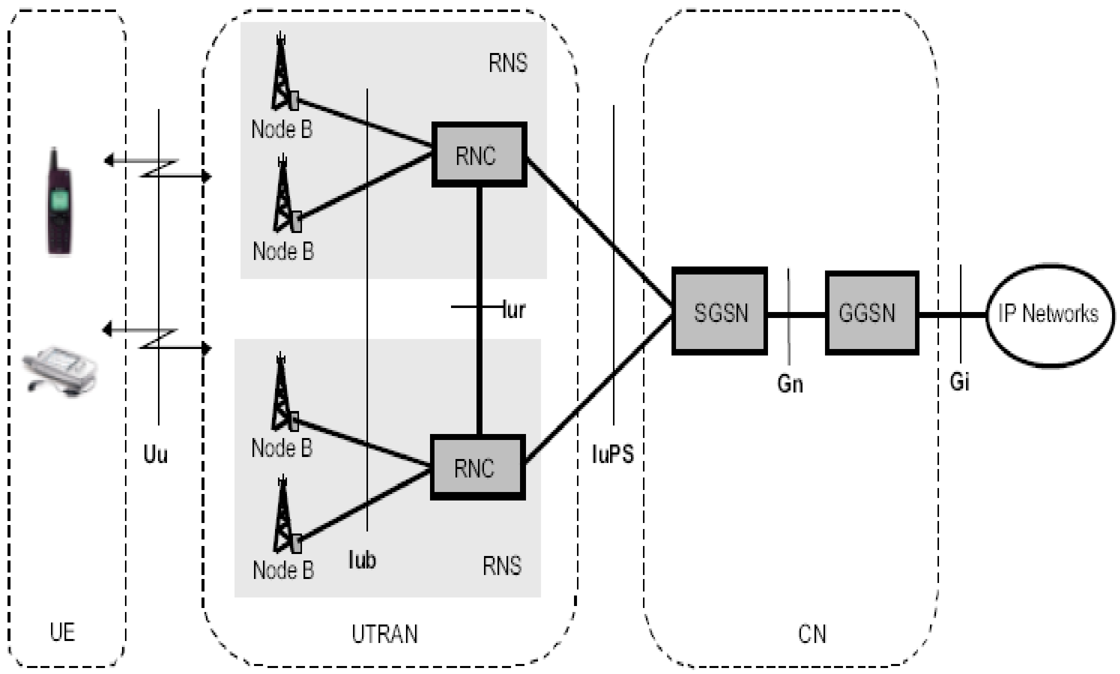

Figure 2-1 .1 UMTS architecture [11]

2.2.1 User Equipment and External Networks

The user equipment is divided in two parts. The UMTS Subscriber Identity Module (USIM) holds the subscriber identity and stores other subscription information. The Mobile Equipment (ME) is the radio terminal used for radio communication over the Uu interface. On the other extremity, we can see the two big existing network types: Circuit Switched (CS) and Packet Switched (PS) networks. The Public Switched Telephone Network (PSTN) is a common example of CS network, while the Internet is PS.

2.2.2 UTRAN

The Universal Terrestrial Radio Access Network (UTRAN) is the most important part of the UMTS networks. It handles all radio-related functionality. It is split in two components: Node B and Radio Network Controller (RNC).

- **Node B**

  A node B (similarly called Base Station (BS) in this report) is a logical node responsible for radio transmission / reception in one or more cells to/from the User Equipment [16]. The Node B is responsible for coding/spreading on the physical channel. It also participates in Radio Resource Management (RRM). For example, it makes radio measurements for the upper layers, Forward Error Correction (FEC), encoding/decoding of transport channels and Inner loop Power Control [16].

- **RNC**

  The RNC is in charge of controlling the use and the integrity of the radio resources [8].The Radio Network Controller (RNC) controls one or more Node Bs.It is the access point between the Core Network (CN) and UTRAN. It is responsible for the traffic management of the common/shared channels and the Admission control. It also performs Soft Handovers (SHO).RNC also has a logical role. The RNC controlling a Node B is called Serving RNC (SRNC), and is responsible for the load and control congestion in its own cells. Further, a RNC can be a SRNC or a Drift RNC (DRNC).

The SRNC for one mobile is the RNC that terminate the Iu link for the transport of user data. All traffic from the CN to the mobile will go through this RNC. On the other hand, a DRNC is any other RNC that controls a cell used by the mobile. All communications received by the DRNC of this mobile will be redirected to his SRNC [16].

2.2.3 Core Network (CN)

The CN is responsible for switching, routing procedures and data connections to external networks. It makes the link between the existing external CS and PS networks such as PSTN and Internet. So, the core network is splitted into several CS and PS entities. [16]

- **MSC/VLR and GMSC**

  The Mobile Switching Centre (MSC) and the Visitor Location Register (VLR) are the switch and the local database that serve the User Equipment (UE) in its current location for CS services. GMSC is the Gateway MSC.

- **SGSN and GGSN**

  The Serving GPRS (General Packet Radio Service) Support Node has the same function than the MSC/VLR, but for PS services. GGSN is the Gate-way GPRS Support Node.

- **HLR**

  The Home Location Register (HLR) is a database located in the user's home system that contains the master copy of all information about the user's service profile. For example, it contains information about roaming areas, allowed services, authentication keys, etc. It is created when a new user subscribes to the system and remains stored as long as the subscription is active.

2.2.4 Interfaces

An interface defines the connection between different entities. The main interfaces are well-specified by the 3GPP. The most important ones are:

- **Iu**

  This interface connects the CN to the UTRAN. It can be divided in two parts: Iu-CS for the CS domain and Iu-PS for the PS domain.

- **Iub**

  This interface is situated between the RNC and the Node B in the UTRAN.The tasks that Node B and RNC have to perform together is so complex that a proprietary solution is the most likely one. For example, the traffic management of common/shared/dedicated channels and system information management are performed on this link.

- **Iur**

  The Iur interface connects two RNCs. It is used for example for the DRNC-SRNC Connectivity.

2.3 WCDMA AIR INTERFACE PROTOCOL STACK

To help the transparency between the different Mobile Communication Generations, the concepts of Access Stratum (AS) and Non Access Stratum (NAS) were defined. The AS is a functional entity that includes radio access protocol between the UE and the UTRAN. The NAS includes CN protocols between the UE and the CN itself [6]. Thus, in theory, NAS protocols are independent of the underlying radio access technology and can be reused from older Global System for Mobile communication (GSM) networks. This section will focus on the AS protocols specific to UMTS, that are typically implemented in the UE and in the RNC. Note that the Node Bs is not aware of Radio Link Control (RLC) and upper protocols. This section will describe the UMTS AS protocols, shown in Figure 2.2.

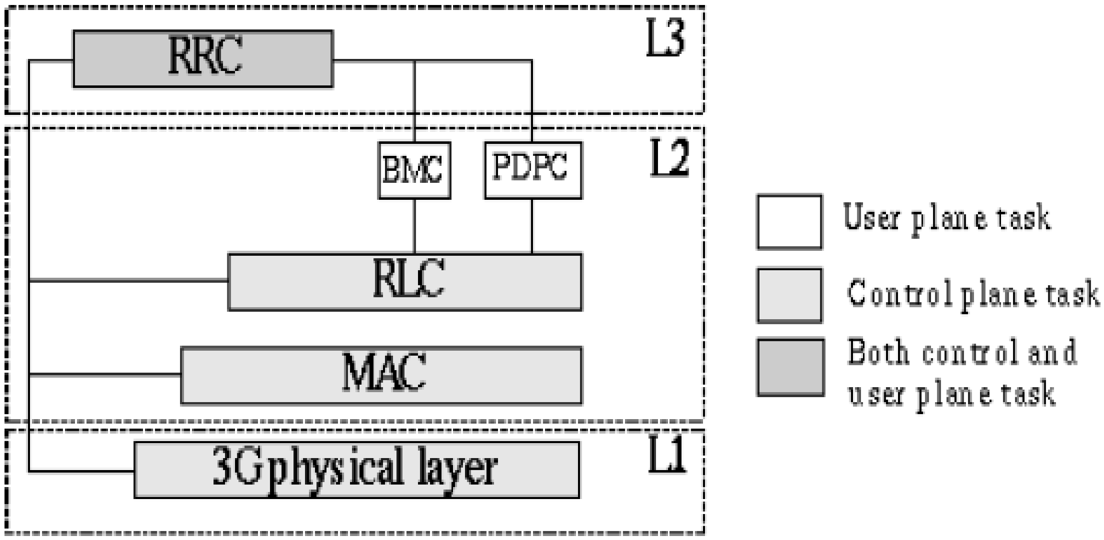

Figure 2-2 Protocol Stack in UTRAN

2.3.1 Radio Resource Control

The Radio Resource Control (RRC) is a sub layer of radio interface Layer 3 existing in the control plane only which provides information transfer service to the non-access stratum. The RRC is responsible for controlling the configuration of radio interface Layers 1 and 2 [16]. RRC functions and signaling procedures. The major part of the control signaling between UE and UTRAN is RRC messages [6]. They carry all the parameters required to set up, modify and release layer 2 and 1 protocol entities [16]. Among other things, RRC is responsible for outer-loop power control, security mode control (through ciphering and integrity protection), Handovers (HO), reception of paging messages, measurements, etc. RRC is also responsible for the establishment, reconfiguration and release of radio bearers.

2.3.2 Layer 2

BMC/PDCP

The Packet Data Converge Protocol (PDCP) must hide the particularities of each protocol from the UTRAN [6]. The PDPC functions includes, among other things, header compression and decompression of IP data streams and the transfer of user data.

The Broadcast/Multicast Control (BMC) only handles downlink broad-cast/ multicast transmission. It implements the Cell Broadcast Service (CBS) messages that are broadcast to every user in a cell.

RLC

RLC is a sub layer of radio interface layer 2 providing transparent, unacknowledged and acknowledged data transfer service [8]. A wireless physical channel is characterized by a high error rate. Normally, errors are corrected by the transport layer if needed (for example with TCP), but because errors occur often, it can be faster to do retransmissions in a lower layer. That is the goal of this sub layer. Note that in order to do this, the RLC performs segmentation and reassembly of higher-layer Service Data Units (SDU) into/from smaller RLC payload units (called Protocol Data Unit, PDU).RLC layer architecture is shown in fig 2.3.

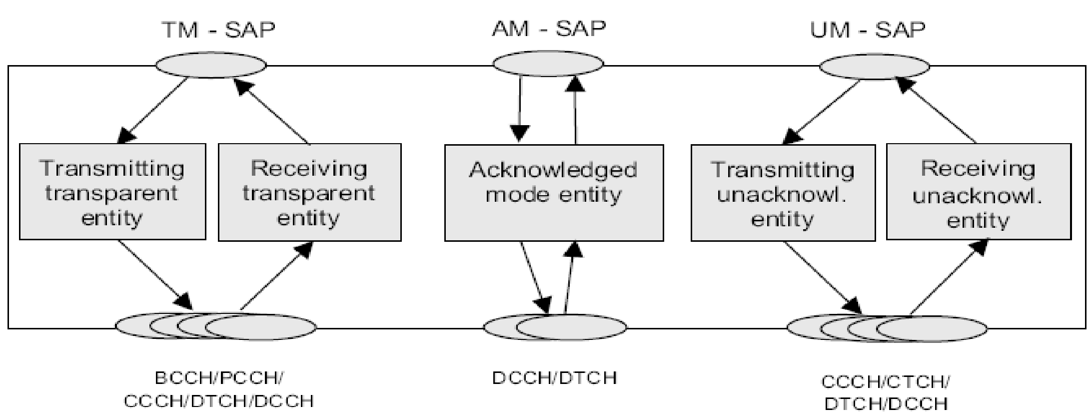

Figure 2-3 RLC layer architecture

The following services are provided to upper layers [6] [16]:

1. Transparent Data Transfer Service (TR)

In this mode, the only operation that can be done by the RLC is segmentation and reassembly, but even this functionality is very limited, since the RLC doesn't add a protocol header. If this is used, it has to be negotiated in the radio bearer set up procedure. Erroneous packets are dropped or marked.

2. Unacknowledged Data Transfer Service (UM)

In this mode, data delivery is not guaranteed. Segmentation and re-assembly is done by adding a header to RLC PDUs. Those PDUs also contains sequence numbers so that the integrity of higher layer PDUs can be checked. The packets are encrypted.

3. Acknowledged Data Transfer Service (AM)

Here, an Automatic Repeat Request (ARQ) mechanism is used to reduce the error rate for higher layers. The quality verses delay performance of the RLC can be configured by the RRC, by changing the maximum number of retransmissions (maxDAT). Moreover, the RLC can be configured for in-sequence or out-of-sequence delivery. The packets are encrypted.

MAC

The main function of the Medium Access Control (MAC) sub layer is the mapping between the logical channels recognized by the RLC and the transport channels used by the physical layer. New data is sent each Transmission Time Interval (TTI) to the physical layer. The logical channels specify the kind of data owing over the network, though the transport channels defines how the data is transferred. To do this, the MAC protocol has to select the appropriate Transport Format (TF).

Among others, the MAC layer is also responsible for:

- Some traffic volume monitoring, that will be used by the RRC for triggering reconfiguration of Radio Bearers and Transport channel
- Ciphering if the RLC is in TR
- Multiplexing/demultiplexing of higher PDUs into/from transport block
- Dynamic transport channel type switching (the switching decision between common and dedicated channel) come from the RRC.

The MAC layer consists of three logical entities (Figure 2.4). There is only one MAC-b and one MAC-c/sh in each Node B and in each UE.However, there is one MAC-d entity in each UE and one MAC-d entity for each UE in the SRNC. The possible mappings between logical channels and transport channels are shown in Figure 2.5, while table 2.1 and 2.2 briefly describe the logical and transport channels.

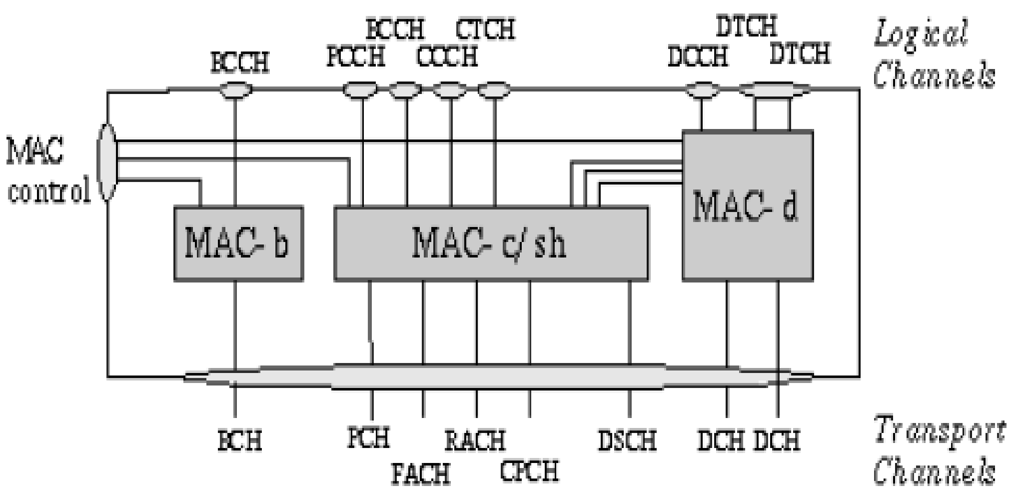

Figure 2-4 MAC layer architecture [11]

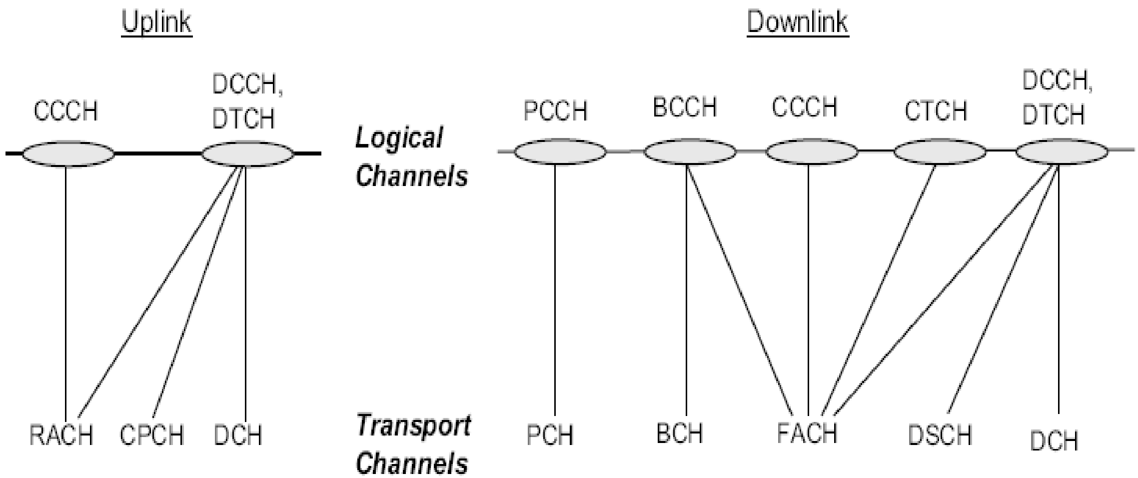

Figure 2-5 Mapping between logical channels and transport channels [11]

	Logical channels	Direction
BCCH	Broadcast Control Channel	DL
PCCH	Paging Control Channel	DL
DCCH	Dedicated Control Channel	UL/DL
CCCH	Common Control Channel	UL/DL
DTCH	Dedicated Traffic Channel	UL/DL
CTCH	Common Traffic Channel	UL/DL

Table 2-1: Logical channels

	Transport channels	Direction
BCH	Broadcast Channel	DL
PCH	Paging Channel	DL
RACH	Random Access Channel	UL
CPCH	Common Packet Channel	UL
FACH	Forward Access Channel	DL
DSCH	Downlink Shared Channel	DL
DCH	Dedicated Channel	UL or DL

Table 2-2 Transport Channels

Note that RACH is only for limited-size data fields, while FACH is in-tended for bursty data traffic.DCH is the only dedicated channel. If the data traffic over the DCH suffers from some peaks, it can be associated with a DSCH to help sending these peaks. This system allows not giving a too large bandwidth to the DCH if the peaks are not frequent.

HS-DSCH

There is a last transport channel that can be added: the High Speed Downlink Shared Channel (HS-DSCH). It is optimized for very high speed data transfer, and is always associated with a DCH.

2.3.3 Physical layer

The physical layer mainly maps transport channels into the right physical channel. Figure 2.6 shows the mapping between the transport channels and the physical channels, and Table 2.3 briefly describes the physical channels. Only channels that are relevant in term of services for the upper layer are showed here.

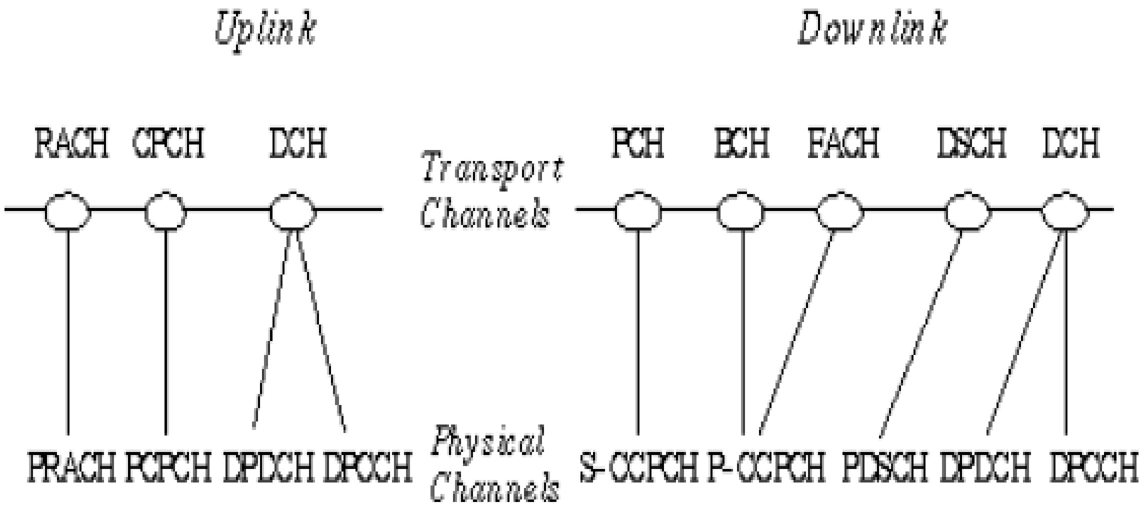

Figure 2-6 Mapping between Transport channels and physical channels [16]

	Physical channels	Direction
PRACH	Physical Random Access Channel	UL
PCPCH	Physical Common Packet Channel	UL
DPDCH	Dedicated Physical Data Channel	UL and DL
DPCCH	Dedicated Physical Control Channel	UL and DL
P-CCPCH	Primary Common Control Physical Channel	DL
S-CCPCH	Secondary Common Control Physical Channel	DL
PDSCH	Physical downlink Shared Channel	DL

Table 2-3: Physical Channels

The UMTS physical layer uses WCDMA as its air interface. Its description will be very brief, and is only present in order to make this overview complete.

Figure 2.7 shows how Code Division Multiple Access (CDMA) works for one user. It is the same when multiple users are sending data. Among other things, CDMA is used in order to send more than one data flow in the same time, to decrease the multiple access interference from many system users and to provide variable bit rates [16].

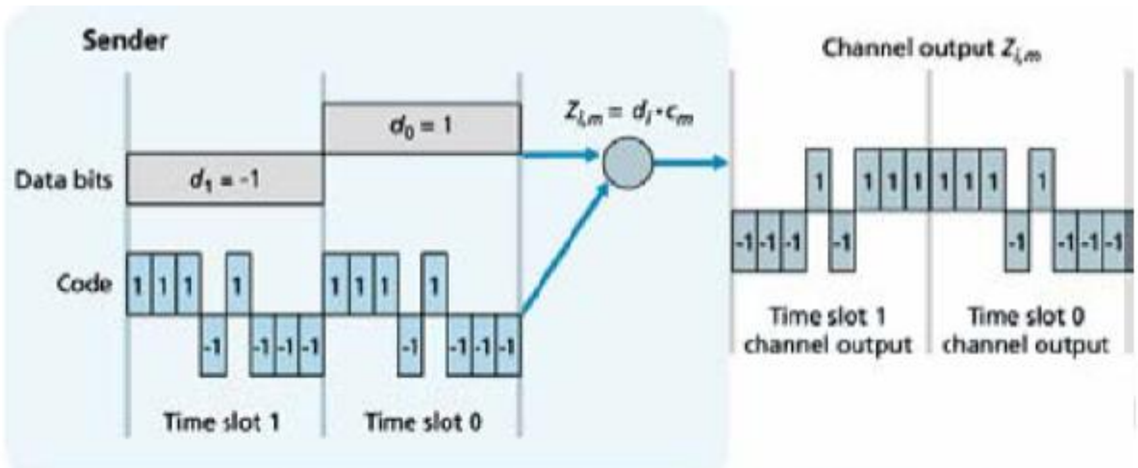

Fig 2.7 sender

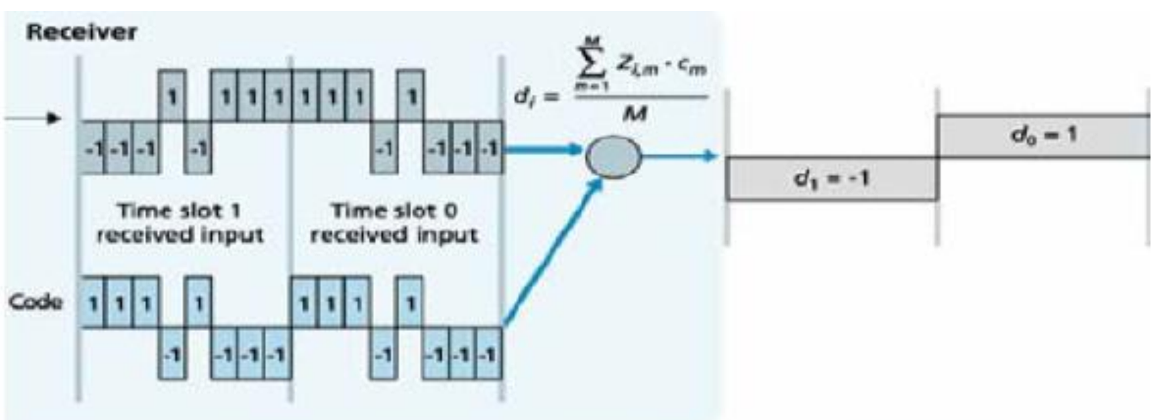

Figure 2-7 Spreading –De-spreading in CDMA single user [16]

The spreading codes are chosen through the channelization and the scrambling mechanisms.

1. Channelization

The spreading/channelization codes of the Universal Terrestrial Radio Access (UTRA) are based on the Orthogonal Variable Spreading Factor (OVSF) technique. The codes are picked from the channelization tree (see Figure 2.8). This allow UTRA physical layer to give a higher bandwidth to a connection, by giving it a higher channelization code.

However, when a code is allocated to a user, no other code from an underlying branch can be used (in order to respect the orthogonality).

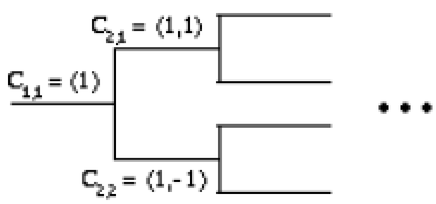

Figure 2-8 Channelization Code Tree [16]

2. Scrambling

Scrambling is needed to separate terminals or base stations from each other, and is used in top of spreading

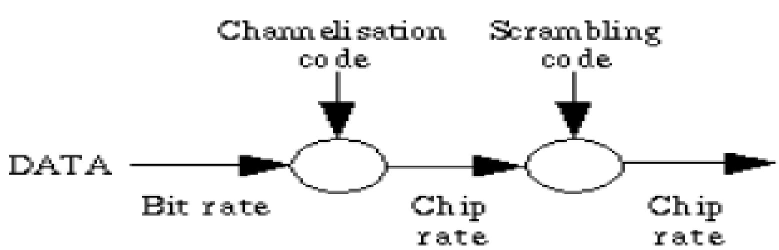

Figure 2-9 Relationship between spreading and scrambling [16]

2.4 PROCEDURES

This section summarizes two of the main UMTS procedures: the handovers (HO) and the power control.

2.4.1 Handovers

A HO is the transfer of a user's connection from one radio channel to another (can be the same or different cell) [8]. For example, a UE can maintain its connection in cellular networks when it moves from one cell area to another one [6]. The biggest problem with HO is the long delays.

There are three kinds of HO:

Soft Handovers

A Soft Handovers (SHO) takes place when a UE receives data from more than one base station at the same time [8] [16]. This requires the use of two separate codes in the downlink direction, so that the UE can distinguish the signals.

Softer Handovers

A Softer HO is a special type of SHO where only one of the possible base station is sending data to the UE [8].

Hard Handovers

A Hard Handovers (HHO) is also known as an inter-frequency HO [8]. The problem is making the required measurements on the new channel. Indeed, a CDMA handset is transmitting continuously in connected mode, so there are no spare slots available to make measurements in other frequencies. To resolve this problem, the compressed mode can be used. That means that not all time slots in downlink are used for data transmissions. When the required measurements are done, the UE stops transmitting on the old frequency and begins using the new one.

2.4.2 Power control

The main purpose of the Power Control (PC) is to minimize the interference in the system, in order to keep the Bit Error Rate (BER) as constant as possible. In the uplink direction, all signals should arrive in the Node B with the same signal power, whether the UE is close or not. In the downlink, signals sent by other Node Bs are not orthogonal and thus increase the interference level. That is one of the reasons why signals should be transmitted with the lowest possible power level [16].

There are two different ways to operate this power control:

Open-loop power control

The transmitting entity estimates the required power level by itself from the received signal.

Closed-loop power control

This power control is based on the explicit power control commands received from the peer entity. This power control can be further divided into inner-loop power control and outer-loop power control.

Chapter 3

OVERVIEW OF WIMAX

3.1 WIMAX IEEE 802.16 STANDARD [3, 4]

The 802.16 standard by the IEEE covers frequency bands in the range of 2 GHz to 11 GHz. It specifies a MAN protocol that enables a wireless alternative for Digital Subscriber Line, cable internet and T1 level services for last mile broadband access. It will also prove backhaul for Wifi hotspots. The new 802.16a standard will also specify a protocol that with other things will support bandwidth hungry applications such as voice and video. It will provide broadband connectivity without requiring a direct LOS between the base station (BTS) and subscriber terminals and will support hundreds of subscribers using a single BTS. The standard will accelerate the induction of wireless broadband equipment in the market and will speed up last-mile broadband deployment worldwide by giving opportunity to service providers to increase performance of system and reliability of system while making equipment costs less and minimum investment risks.

3.2 WIMAX FLAVORS

WiMax profiles are based on IEEE 802.16 which was developed by the WiMax Forum. Now WiMax supports a big range of frequencies up to 66 GHz. It supports channel sizes from 1.25 MHz to 20 MHz and it supports applications such as Line of Site, NLOS, PTP and M. This WiMax profile is also narrowing the scope of 802.16 to focus on specific configurations. This selection is essential to make sure interoperability b/w different vendors and to generate more revenues.

The factors which affect this choice of profiles are spectrum availability, market demand, regulatory constraints, vendor interest and the services to be offered. For example, the spectrum availability for broadband wireless access services in a lot of countries lead to the creation of initial profiles in the 3.5 GHz band. The availability of exemption of spectrum license and the need for fixed services resulted in the creation of a profile in the 5.8GHz band. Similarly the need for mobile services and spectrum availability make the 2.3 GHz and 2.5 GHz bands which are targets for future 802.16e profiles.

Following parameters define WiMax profile:

- **Duplexing**

  Two options are available

1. Time Division Duplex (TDD)

   This is for operators with unpaired or license-exempt spectrum.

2. Frequency Division Duplex (FDD)

   A single channel is occupied by TDD network traffic, with uplink and downlink traffic assigned to different time slots.FDD requires two channels, one for uplink and other for downlink traffic.

Frequency (MHz)	Duplexing	Channels (MHz)	IEEE standard
3400-3600	TDD	3.5	802.16-2004
3400-3600	FDD	3.5	802.16-2004
3400-3600	TDD	7	802.16-2004
3400-3600	FDD	7	802.16-2004
5725-5850	TDD	10	802.16-2004

Table 3-1 WiMax Forum certification Profiles [12]

IEEE standard

802.16e profiles are most likely based on SOFDMA, they support mobility whereas 802.16a profiles use OFDM with 256 carriers

Channel bandwidth

This highly depends on the spectrum allocated by authorities. Developed profiles were restricted to 3.5 MHz and 7 MHz in the licensed spectrum initially as these were the prevalent spectrum channels allocated in the 3.5 GHz band. As wider channels are available to operators now, therefore the WiMax Forum members will add certification profiles with wider channel bandwidths.The initial profiles defined by the WiMax Forum (Table 3.1) support fixed and nomadic access in the 3.5 GHz and 5.8 GHz bands.

The first certification release for 802.16a is currently under way and includes products submitted within the two 3.5 GHz profiles with a channel bandwidth of 3.5 MHz's The scope of the certification and the list of tests will be extended during subsequent releases. First release will focus on the air protocol certification. Functionality needed to support outdoor services (QoS and security, for instance) will be added in second release and the third release will include support for indoor user devices.

The profiles for 802.16e have not yet been announced, as the 802.16e amendment to the standard has not yet been finalized. For the first mobile profiles the most likely bands are 2.3 GHz and 2.5 GHz. Better indoor coverage and support for mobile or portable devices make bands below 3 GHz the best targets. However, additional profiles in higher frequencies (3.3 GHz, 3.5 GHz or even 5.8 GHz) may be added if there is sufficient demand for 802.16e-based products for fixed or nomadic access.

In the coming months the WiMax Forum plans to announce new profiles supporting mobility and to open certification labs in the third quarter of 2006. The first 802.16e WiMax Forum CERTIFIED products are expected by the first quarter of 2007.

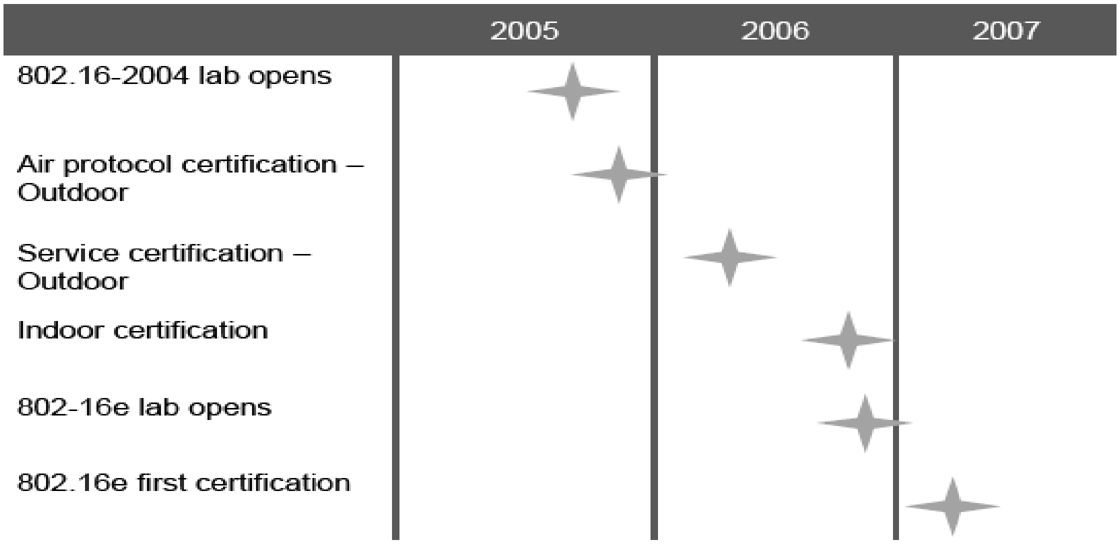

Figure 3-1 WiMax Forum timeline for product certification [12]

3.2.1 A Comparison Between 802.16a And 802.16e Profiles

The requirements for the two types of access vary and different solutions are required to meet them. Several optional features that are supported in both 802.16a and 802.16e profiles are more likely to be implemented in 802.16e products simply because mobile services stand to gain more from the added functionality. Improved support for MIMO and AAS will bring substantial increase in throughput and NLOS capabilities

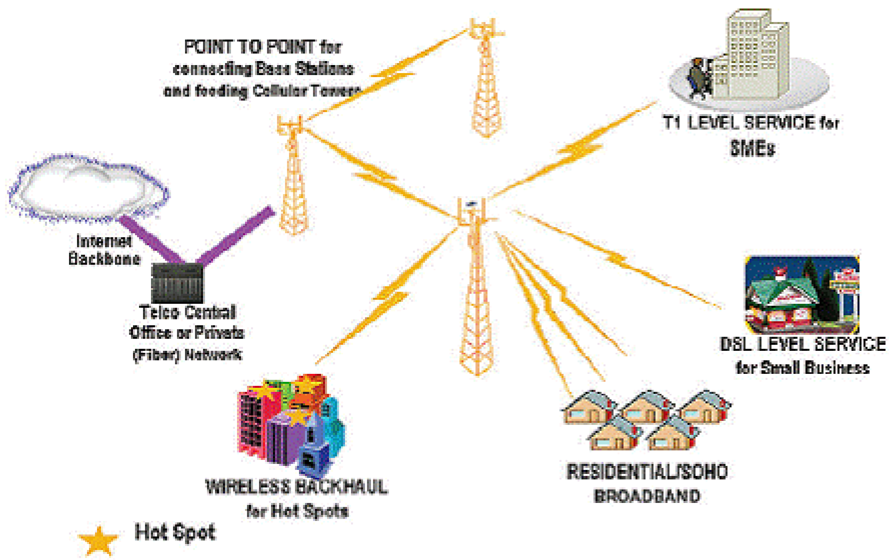

Figure 3-2 BWA (IEEE 802.16) every where [4]

3.2.2 OFDM and SOFDMA

A key difference between 802.16a and 802.16e profiles is the multiplexing technique: the first uses OFDM and the second will most likely use OFDMA.WiMax profiles based on 802.16a are better suited to fixed applications that use directional antennae because OFDM is inherently less complex than SOFDMA. As a result, 802.16a networks may be deployed faster and at a lower cost. In addition, 802.16a WiMax Forum CERTIFIED products will be available earlier and will be adopted by service providers that plan to deploy a network in the near future. OFDMA gives 802.16e profiles more flexibility when managing different user devices with a variety of antenna types and form factors. It brings a reduction in interference for user devices with omni directional antennas and improved NLOS capabilities that are essential when supporting mobile subscribers. Sub channelization defines sub channels that can be allocated to different subscribers depending on the channel conditions and their data requirements (Figure 3.3).This gives the operator more flexibility in managing the bandwidth and transmit power, and leads to a more efficient use of resources.

For instance, within the same time slot more transmit power can be allocated to a user with less favorable channel conditions, while lowering the power for users in better locations. Improved in-building coverage can be achieved by allocating higher power to sub-channels assigned to indoor user devices.

Sub-channelization in the uplink brings additional performance improvement, as transmit power from the user device is severely limited. In OFDM, user devices transmit using the entire carrier space at once (Figure 3.3). OFDMA supports multiple access, which allows user devices to transmit only through the sub-channel(s) allocated to them. In OFDMA with 2048 carriers and 32 sub-channels, if only one sub-channel is allocated to a device, all the transmit power will be concentrated in 1/32 of the spectrum available and may bring a 15 dB gain over OFDM. Multiple access is particularly advantageous when wide channels are used.

SOFDMA brings an additional advantage over OFDMA. It scales the size of the Fast Fourier Transform (FFT) to the channel bandwidth in order to keep the carrier spacing constant across different channel bandwidths. Constant carrier spacing results in higher spectrum efficiency in wide channels, and a cost reduction in narrow channels.

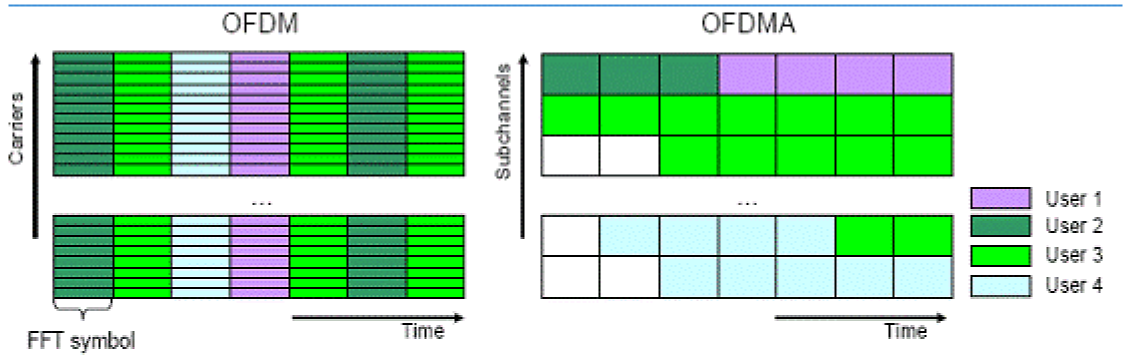

Figure 3-3 OFDM and OFDMA [12]

3.3 HANDOFFS AND ROAMING

Supporting for handoffs is a important addition in the 802.16e amendment for mobile access. A prerequisite for mobility is the ability to maintain a connection while moving across cell borders and this has been included in the 802.16e system profile. This kind of support for handoffs was not required by the 802.16a system profile.

802.16e WiMax will support both hard to soft handoffs and the operator will decide which one to use. Hard handoffs use a break-before-make approach. Soft handoffs allow the user device to retain the connection to a base station until it is associated with a new one (make-before-break approach), thus reducing latency. New applications like mobile Voice over Internet Protocol (VoIP) or gaming greatly benefit from soft handoffs, hard handoffs typically suffice for data services where as QoS and Service Level Agreements (SLAs) are maintained during handoffs.

3.4 802.16A OR 802.16E WIMAX

Fixed and mobile deployments have very different requirements and target substantially different market segments, with different usage patterns and locations, throughput needs, user device form factors, and SLAs. The two flavors of WiMax were defined to meet the distinct demands of these two market segments and the varying requirements of different applications.

In a fixed deployment with basic functionality, 802.16a and 802.126e offer similar performance. Single sector maximum throughput for both versions of WiMax is about 15 Mbps for a 5 MHz channel or 35 Mbps for a 10 MHz channel. Base station range in densely populated areas can go up to a few kilometers depending on attributes such as CPE type, frequency band, mobility, morphology and so on. In networks that are capacity constrained, the number of base stations installed depends on throughput demand, rather than range.

However, the performance of the two versions of WiMax can change substantially for specific applications, because 802.16a is optimized for fixed access and 802.16e for mobile access, although it can also be used for fixed acces.

Chapter 4

OVERVIEW OF 4 G

4.1 4 G NETWORKS

4G (or 4-G) is short for fourth-generation cellular communication system. There is no set definition to what 4G is, however the features that are predicted for 4G can be summarized in a single sentence. [7]

"The 4G will be a fully IP-based integrated system of systems and network of networks achieved after the convergence of wired and wireless networks as well as computer, consumer electronics, communication technology, and several other convergences that will be capable of providing 100 Mbps and 1Gbps, respectively, in outdoor and indoor environments with end-to-end QoS and high security, offering any kind of services anytime, anywhere, at affordable cost and one billing"[7]

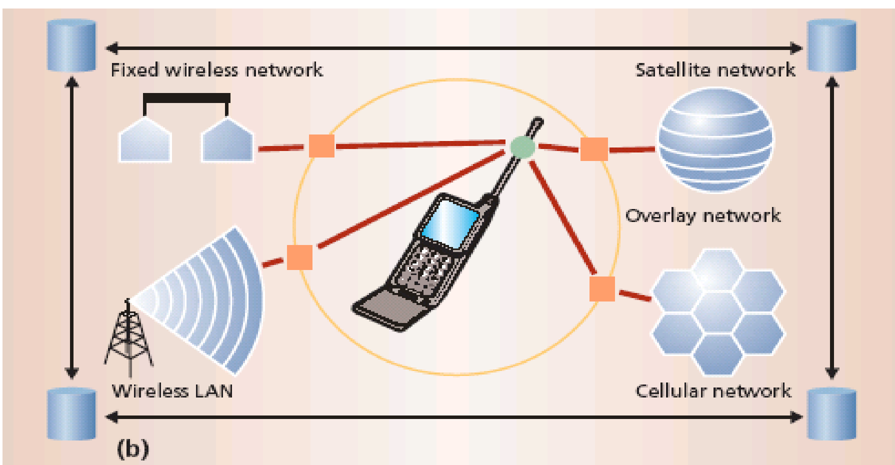

Figure 4-1 A Future 4G network [4]

We have restricted our study to two radio access networks i.e. UMTS and WiMax. The former offers a wide coverage area and thus supports a high degree of mobility, while the latter offers higher bandwidth in limited areas only.

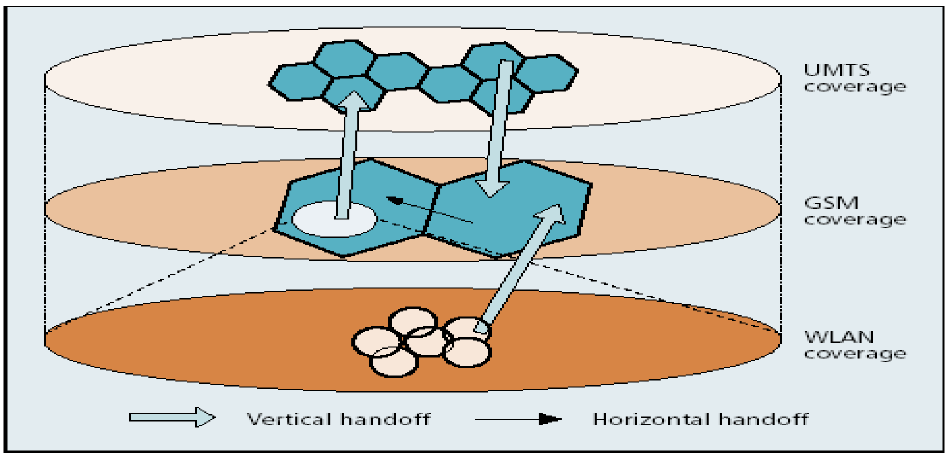

Figure 4-2 Concept of vertical and horizontal handoff [4]

There are a number of unresolved issues [15] regarding optimal wireless network selection:

- Improve the mobility models and apply other models for user movement.
- Study other, more complex handover strategies
- Study more than two radio access networks

4.2 KEY CHALLENGES AND PROPOSED SOLUTIONS

	KEY CHALLENGES	*PROPOSED SOLUTIONS*
Multimode user	To design a single user terminal that can operate in different wireless networks, and overcome the design problems such as limitations in device size, cost, power consumption, and backward compatibilities to systems.	A software radio approach can be used: the user terminal adapts itself to the wireless interfaces of the networks.
Wireless system discovery	To discover available wireless systems by processing the signals sent from different wireless systems (with different access protocols and incompatible with each other).	User-or system-initiated discoveries, with automatic download of software modules for different wireless systems.
Wireless system selection	Every wireless system has its unique characteristic and role. The proliferation of wireless technologies complicates the selection of the most suitable technology for a particular service at a particular time and place.	The wireless system can be selected according to the best possible fit of user QoS requirements, available network resources, or user preferences

Table 4-1: Key Challenges in 4G

System		
Terminal mobility	To locate and update the locations of the terminals in various systems. Also, to perform horizontal and vertical handoff as required with minimum handover latency and packet loss.	Signaling schemes and fast handoff mechanisms are proposed.
Network infrastructure and Qos support	To integrate the existing non-IP-based and IP-based systems, and to provide QoS guarantee for end-to-end services that involves different systems.	A clear and comprehensive QoS scheme for UMTS system has been proposed. This scheme also supports inter working with other common QoS technologies.
Security	The heterogeneity of wireless networks complicates the security issue. Dynamic reconfigurable, adaptive, and lightweight security mechanisms should be developed.	Modifications in existing security schemes may be applicable to heterogeneous systems. Security handoff support for application sessions is also proposed
Fault tolerance and survivability	To minimize the failures and their potential impacts in any level of tree-like topology in wireless networks.	Fault-tolerant architectures for heterogeneous networks and failure recovery protocols are proposed.

Table 4-2 Challenges in 4G systems

Service		
Multi-operators and billing system	To collect, manage, and store the customers’ accounting information from multiple service providers. Also, to bill the customers with simple but detailed information.	Various billing and accounting frameworks are proposed.
Personal mobility	To provide seamless personal mobility to users without modifying the existing servers in heterogeneous systems.	Personal mobility frameworks are proposed. Most of them use mobile agents, but some do not

Table 4-3 Challenges in 4G services

Chapter 5

SIMULATOR

5.1 NETWORK SIMULATOR V 2

Ns-2 is an event driven network simulator developed by UC Berkeley. Now it has been a part of VINT project. [9] Ns-2 implements traffic behaviors, network protocols, routing, etc. for simulation. Because it is open source software, during the development many contributions have been included from other researchers. Ns-2 has become a common tool for network researchers to simulate and evaluate network related project.Ns-2 simulator is implemented and runs with two programming languages: C++ for the object oriented simulator, and OTcl interpreter for executing user configuration scripts before the simulation begins.

There are always two corresponding hierarchies for every protocol or network objects implemented in ns-2, the compiled C++ hierarchy and the interpreted OTcl hierarchy [6]. The C++ hierarchy allows user to achieve efficiency and faster running time in the simulation especially for the detailed definition and operation of protocols since it can reduce packet and event processing time.

Through the friendly interface of OTcl language, users can define a particular network topology, the protocols and applications that they want to simulate and the form of the output that they want to obtain from the simulator quickly and clearly in an OTcl script. A simplified user view of ns-2 is given in figure 5.1.

Ns-2 is a discrete event simulator. The timing of events is maintained by the scheduler. In ns-2 an event is an object, e.g. a packet in the C++ hierarchy with a unique ID, a scheduled time and the pointer to an object handles the event.

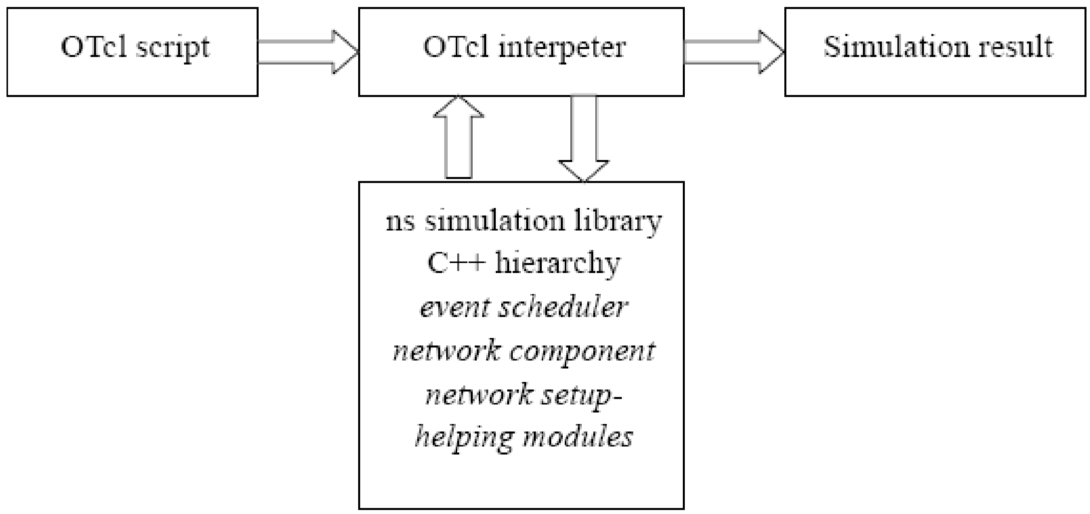

Figure 5-1 User view of NS-2 [16]

5.1.1 Network Components

NS network components mainly have two types of elements, the compound network components and the basic network components. Figure 5.2 shows a partial OTcl class hierarchy of basic network component in NS. A compound component contains several basic components with different functions, to play a complicated role during the simulation.

On the root, Ns Object class is the super class of all basic network component objects that handle packets, which may compose compound network objects such as nodes and links. Based on the number of the possible output data paths, the basic network components can be further divided into two subclasses, Connector and Classifier. Objects belong to Connector class have only one output data path, correspondingly, the objects that have possible multiple output data paths are under the Classifier class.

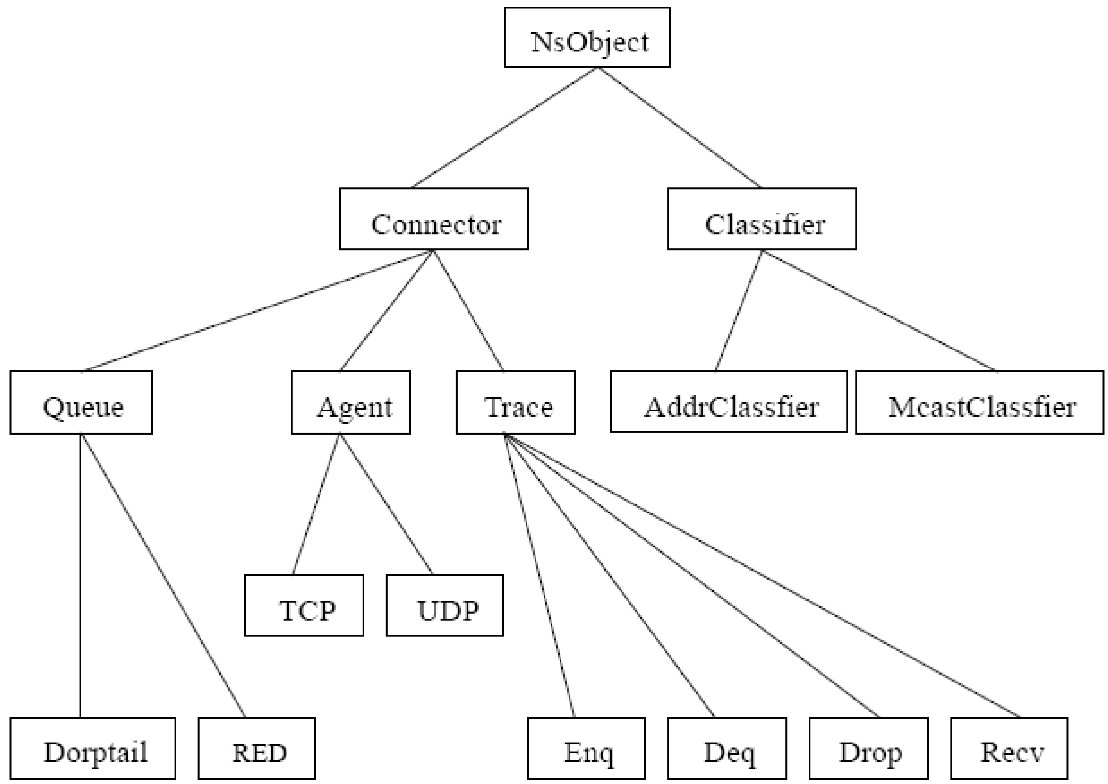

Figure 5-2 Partial class hierarchy of basic components in NS-2

Node

Node is one of the primary objects in ns-2. A node consists of a node entry object and classifiers as shown. Unicast node is the default node type in ns-2, which has an AddrClassifier that does unicast routing and a port classifier. A structure of Unicast node is shown in figure 5.3. Another type of node is multicast node, which added a McastClassifier to unicast node, for handling multicast packets.

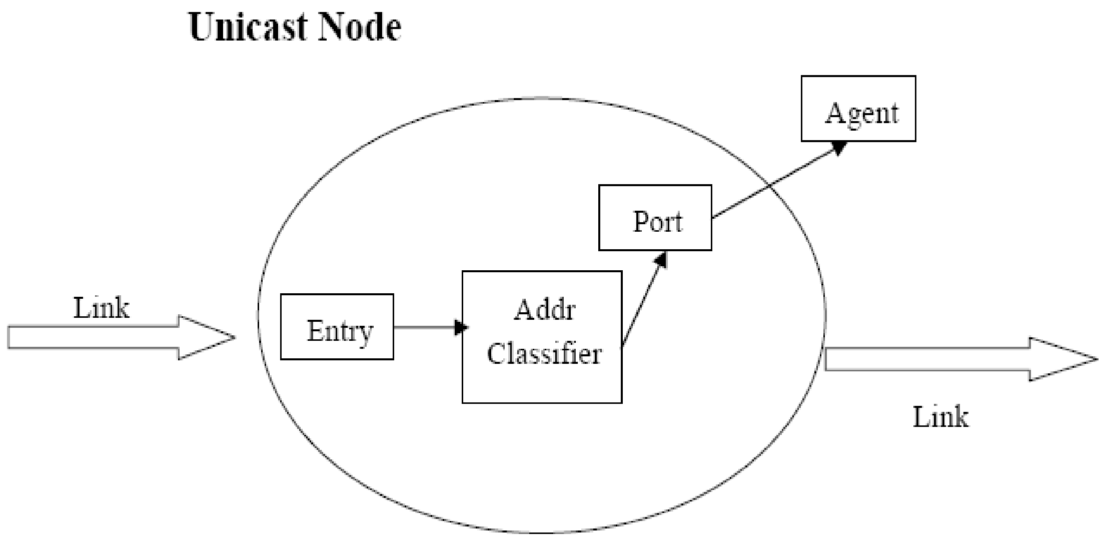

Figure 5-3 Structure of Unicast Node [16]

Link

Link is another essential object to establish a network scenario for simulation. In ns a link object represents a single direction, simplex link. To create a duplex-link for simulation, two simplex links in both direction. Figure 5.4 shows how a packet is handled in a simplex link.

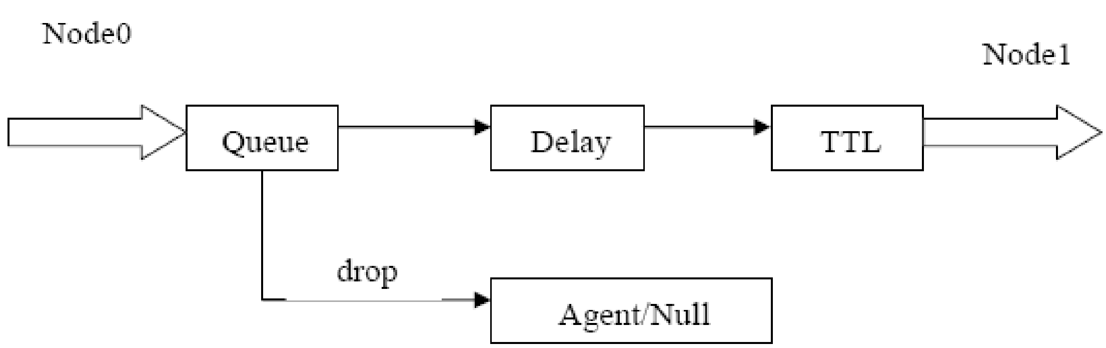

Figure 5-4 Simplex link [16]

Agent

An agent in a node can be either an active or a reactive component during network implementation. Links and nodes are reactive components that react to incoming packets and apply their behaviors to the packet. Compared to them, agents can be an active component. For instance, when transport agents TCP and UDP react to external orders for sending data, they may generate necessary packets for data transmission or generate connection control packets. Another example is routing agents which can perform dynamic route calculations. They actively generate packets and send them to other agents in same type to determine routes within the network. In the case of transport agents, they require external components to generate data to send. The function of traffic generators is to generate data in the network. It could be either a data source that sends data considering a traffic distribution, or a source that simulates an application. Two examples of the former are constant bit rate (CBR) and exponentially distributed traffic in terms of active and inactive sending. Two examples of the latter are FTP-transfer and a Telnet-connection. Traffic generators use transport agent to introduce data into simulated network.

5.1.2 Tracing

To analyze the result of the simulation, the traffic must be traced during the process. When a link is traced, the trace objects (EnqT, DeqT, DrpT and RecvT) are inserted within the link objects as shown in Figure 5.5. For every packet that passes a trace object, information about the packet is written to the specified trace file.

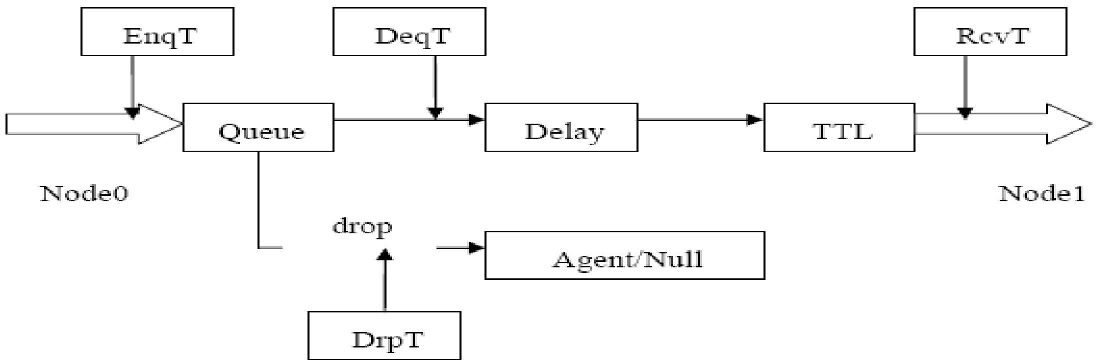

Figure 5-5 Trace objects in a simplex link

5.2 IMPLEMENTATION OF UMTS IN NS-2

This section describes the extensions to ns-2 that enable the simulation of UMTS network. This description will be based on the top level simulation environment shown in fig 5.6 and realization of that environment in ns-2 as shown in fig 5.7.

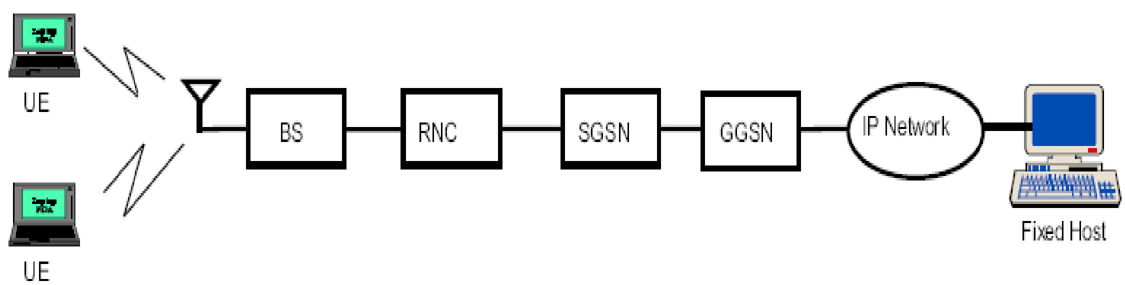

Figure 5-6 Top level simulation structure [11]

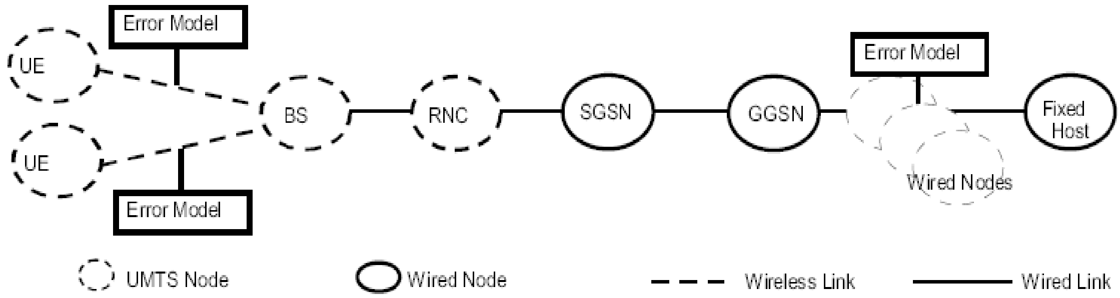

Figure 5-7 Simulation realization in ns-2 [11]

5.2.1 EURANE (Model In Ns-2) [11]

EURANE is short of

"Enhanced UMTS Radio Access Network extension for ns- 2"

It applies three new nodes

- BS
- UE
- RNC

Allow to support these transport channels

- FACH
- RACH
- DCH
- HS- DSCH

5.2.2 Transport Channel

5.2.2.1 FACH/RACH:

The link layer components of the Forward Access Channel (FACH) and Random Access Channel (RACH), which are both Common channels, are created at the time of the BS and the UE configuration, and only require an “attachment” procedure to an RLC entity.

5.2.2.2 DCH:

The Dedicated Channel (DCH) operates in both the uplink and downlink, and its entire link components, from the RLC at the RNC to the RLC at the UE, need to be created… Only in the case of the DCH is there the possibility to channel switch to the RACH/FACH.

5.2.2.3 HS-DSCH:

The HS-DSCH also has to be fully constructed, as in the case of the DCH, and although it is a downlink transport channel, an uplink return path, for RLC retransmissions and TCP acknowledgements, is also required. As the HS-DSCH always requires an associated DCH, an uplink DCH is always created alongside the HS-DSCH.

5.2.3 RLC Configuration

At the RNC, two implementations of Acknowledgement Mode (AM) are available for RLC. The type of RLC (AM or AM-HS) to use is dependent on the transport channel. Two implementation of Un-acknowledgement Mode, UM and UM-HS, are also available, and are basically a functional sub-set of the AM and AM-HS (only for HS-DSCH), respectively.

5.2.3.1 MAC/MAC-HS CONFIGURATION

There are two possible MAC architecture to choose from. The basic MAC used for the DCH and common channels (RACH & FACH), and the more complicated MAC-HS user fro the HS-DSCH.

5.2.4 Physical Layer

The physical layer is represented by a standard ns-2 channel object, which is used to connect the BS and the UE. This is combined with the attachment of an error model.

5.2.5 Trace Format of EURANE [11]

General method of tracing is not compatible with new UMTS nodes, therefore a special UmtsTrace objects is used to trace RLC PDU (i.e. MAC-d SDU) inside the UTRAN and the UE. The trace format for UMTS nodes is like

Event	Time	From node	To node	Pkt type	Pkt size	Flags	Fid	Src addr	Dst addr	Seq num	Pkt id	Rlc seq

Event:

Each trace line starts with an event (+, -, d, r) descriptor, which represent the action of enqueue, dequeue, drop and receive of packets, respectively.

Time:

The simulation time (in seconds) of that event.

From node and to node:

These two fields identify the link on which the event occurred.

Pkt type and Pkt size:

The packet type and size in Bytes. Another modification of EURANE trace file format is the "pkt type has some additional presentation to define the behavior of radio link.

Fid:

The flow id of IPv6 that a user can set for each flow.

Src addr and Dst addr:

Src addr and Dst addr are source and destination address in forms of "node.port".

Seq num:

The network layer protocol's packet sequence number. Note that even though UDP implementations do not use sequence number, ns-2 keeps track of UDP packet sequence number for analysis purposes.

Pkt id:

It shows the unique id of the packet which is the key point to trace a packet.

RLC seq:

It represents RLC sequence number is an extra object which EURANE adds to trace RLC packets.

5.3 WIMAX MODULE IN NS-2.29

Mobility extension 80216e and the IEEE 802.16 standard (802.16-2004) are the basis for the implementation of IEEE 802.16 ns-2 model. It is necessary to mention that this model does not implement all the features. Some unavailable and available features are listed below

5.3.1 Features Available:

- physical layer of WirelessMAN-OFDM with configurable modulation
- Time Division duplexing (TDD) feature
- Execution of network entry through management messages (w/o authentication)
- Round robin uplink Scheduling in case of registered Mobile Stations (MSs)
- Mobility extension of IEEE 802.16 to support handovers and scanning
- Reassembly and Fragmentation of frames

5.3.2 Features Unavailable:

- WirelessMAN-OFDM Access
- Frequency Division duplexing (FDD) feature
- Automatic Repeat Request ARQ
- QoS scheduling and Service Flow
- Power adjustments and Periodic ranging

It is worth mentioning that some of components are not in the standard. Therefore in the model one solution is implemented, which may or may not fulfill the need of users. This is the case for the flow handler and bandwidth scheduler.

5.3.3 Tracing

Some new values are introduced in the trace file. There are two new reasons for dropping a packet :

- QWI: A queue is there with each connection to store pending frames. The packet is dropped using this reason code when the queue is full.
- CID: when a packet received at the MAC layer cannot be matched to any connection, this reason code is used

A new packet type named as WimaxCtrl is introduced. When for synchronization purposes BSs need to communicate. Agent/WimaxCtrl which is a new agent handles this communication

5.3.4 Internals and Architecture

5.3.4.1 Design

This section explains the main components of the IEEE802.16 model, namely: OFDM physical layer model, MAC module, and frame structure

The OFDM physical layer transmits packets in this model. The configuration is done using TCL bindings for the cyclic prefix and the frequency bandwidth. As it inherits from the class WirelessPhy, attributes such as transmission power or frequency can also be configured by using TCL.

The physical layer can be in different states as shown in Figure 1. Packets cannot be sent in receiving mode. All incoming packets are discarded when in sending mode. Furthermore, the packet header contains information, such as cyclic prefix and frequency modulation, which are used to filter the incoming packets.

The model supports some different modulations. The MAC layer allocates bursts that can use different modulations according to interference or distance. This affects the transmission time and data rate. The physical layer includes helper functions when transmitting data, called by the MAC layers:

- getMaxPktSize is the reverse function and returns the maximum packet size given the modulation used and number of OFDM symbols available .
- getTrxTime returns the time required to send a packet given its modulation and size.

The node_off and node_on functions disable or enable blocking all transmissions and receptions of packets, but it is not currently linked to power consumption mechanisms.

5.3.4.2 MAC module structure

The Mac802_16 is a subclass of the Mac class. It is the main module where packets are sent and received. A MAC has a list of packet classifiers (SDUClassifier) which maps each outgoing packet with the proper connection identifier (CID).The user configures the list of classifiers to be used by using TCL.The destination IP address is used as the classifying element in the current implementation.

The MAC intelligence's is contained within its scheduler. There are mainly two kind of schedulers: one for the SS and one for the BS.

5.3.5 IEEE 802.16e Extensions

Some features defined in the IEEE 802.16e standard are added in the model. These features are related to handovers and scanning. When a SS wants to search for other points of access, it sends a scan request to its current Base Station. This request allows the Base Station to allocate scanning intervals so that both the Base Station and the Subscriber Station can stop transmission of packets and reduce packet loss.WimaxCtrl agent handles the allocation of scanning intervals and synchronization between BSs. It is also responsible for sending neighbor advertisement messages to the Subscribe Stations.

5.3.6 Packet Processing

Figure 5.3.1 shows the packet flows for outgoing packets and incoming packets.

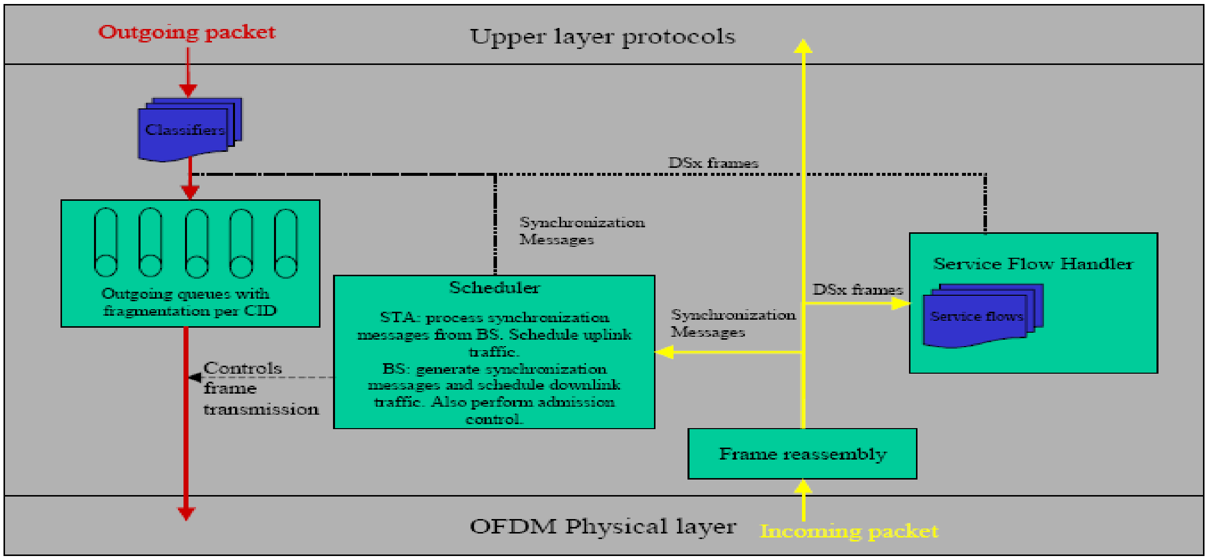

Figure 5-8 Packet processing overview [10]

The activity diagrams provide more detailed information on how the packets cross the MAC layer.

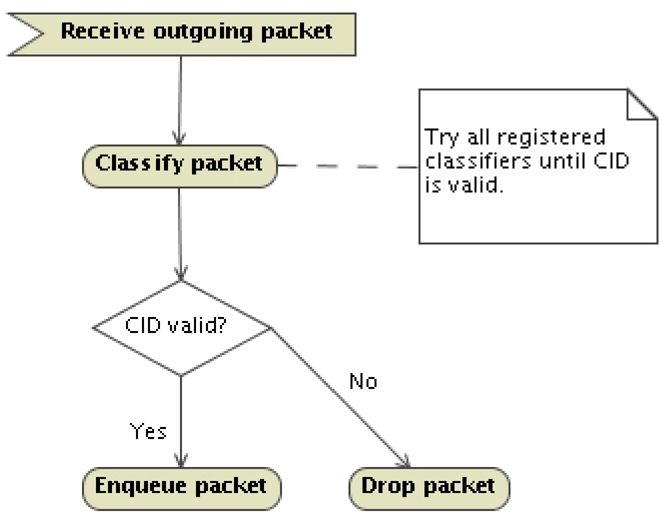

Figure 5-9 Outgoing packet processing [10]

A packet which is received from an upper layer is classified by using the registered classifiers. As there can be multiple classifiers, the MAC accesses them all of them one by one and tries until a valid CID is found, or all of the classifiers have been tested. If the CID is a valid one, that packet is added to the matching queue in other case it is dropped.

When a new packet is received, i.e. the first bit of it, some steps are executed. At the end of the reception of this packet, the packet is processed as shown in figure 5.10

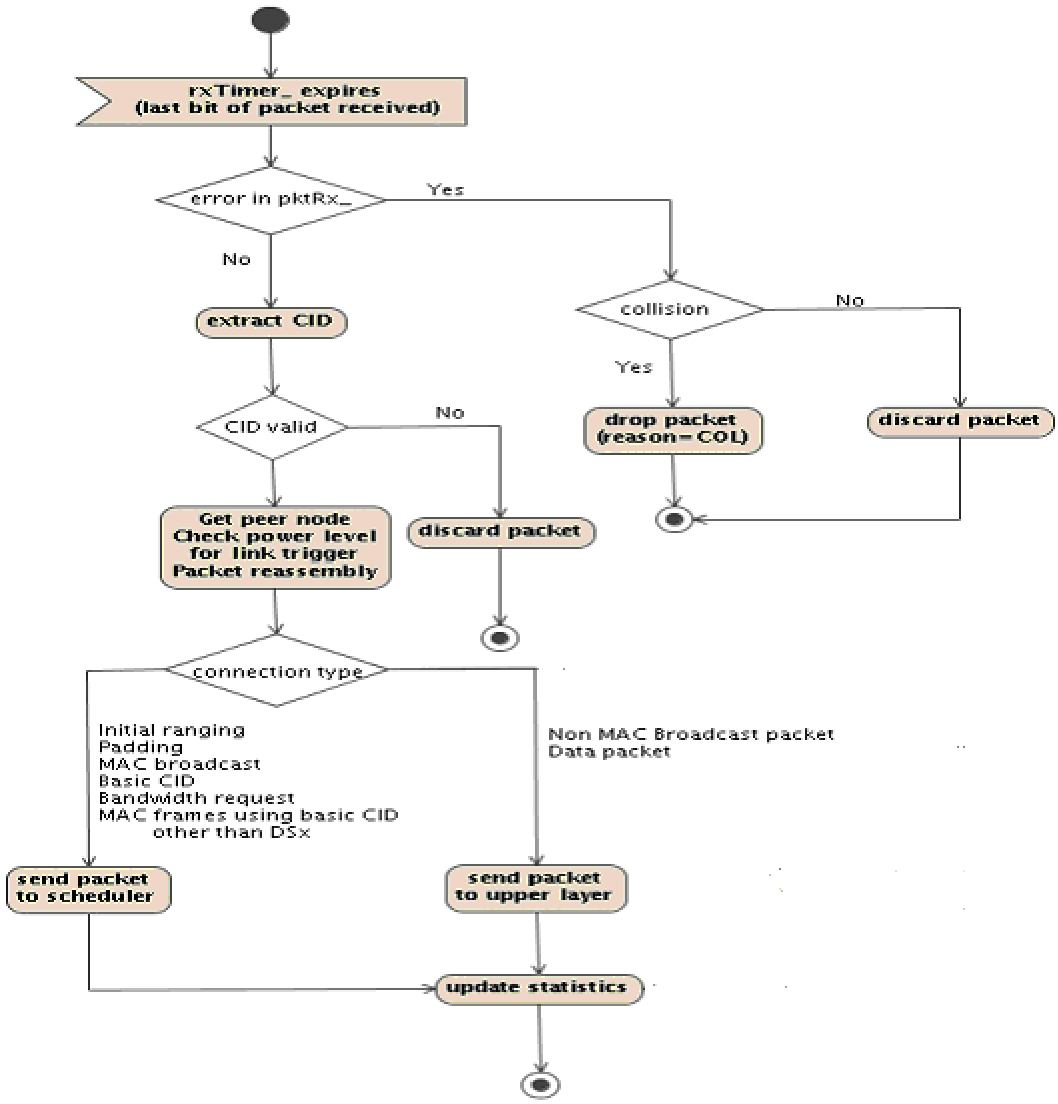

Figure 5-10 received packet processing [10]

5.4 MEDIA INDEPENDENT HANDOVER MODULE [5]

5.4.1 Overview

MIH implementation for NS-2 is presented in this section which is developed for the Seamless Mobility. It provides an overview of the mobility extensions made to the Network Simulator-2.

5.4.2 MIH Implementation

MIHF implementations are based on the draft 3 of IEEE 802.21 specifications. It is used to evaluate different handover decision.

5.4.3 Architecture and Internals

5.4.3.1 Design overview

Fig 5.11 represents a high level view of the MIHF interaction with the different components of the node. The MIHF is implemented as an Agent and therefore can send layer 3 packets to remote MIHF. The MIHF contains the list of local interfaces to get their status and control their behavior. The MIH User is also implemented as an Agent and registers with the MIHF to receive events from local and remote interfaces.

The cross layer information exchange has been added to the NS-2 by modify -
-ing the MAC layer and linking the MIHF to the MAC layers via TCL

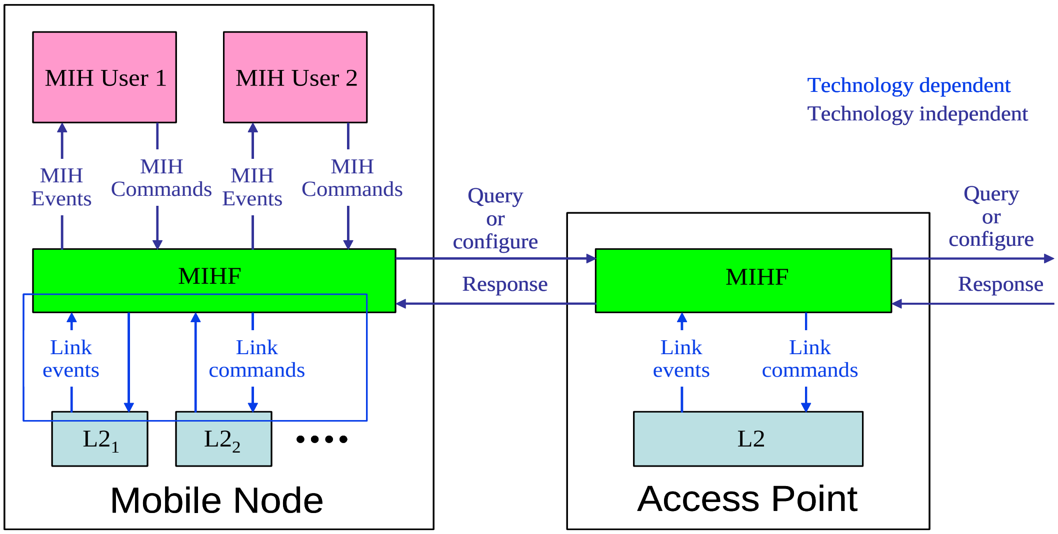

Figure 5-11 MIH design overview [5]

MIH Function

The MIHF extends the class Agent defined in NS-2. This allows each instance to send and receive packets at layer 3. The MIHAgent is at the center of the implementation. It communicates with both the lower layers (i.e. MAC) and the higher layers (i.e. MIH Users).

MIH User

MIH Users are entities that make use of the MIHF functionalities in order to enhance user performances by optimizing handovers. Since there are an infinite number of implementations depending on the user preference or network policies, the implementation provides an abstract class MIHUser that can be easily extended. See MIH Class hierarchy in fig 5.12

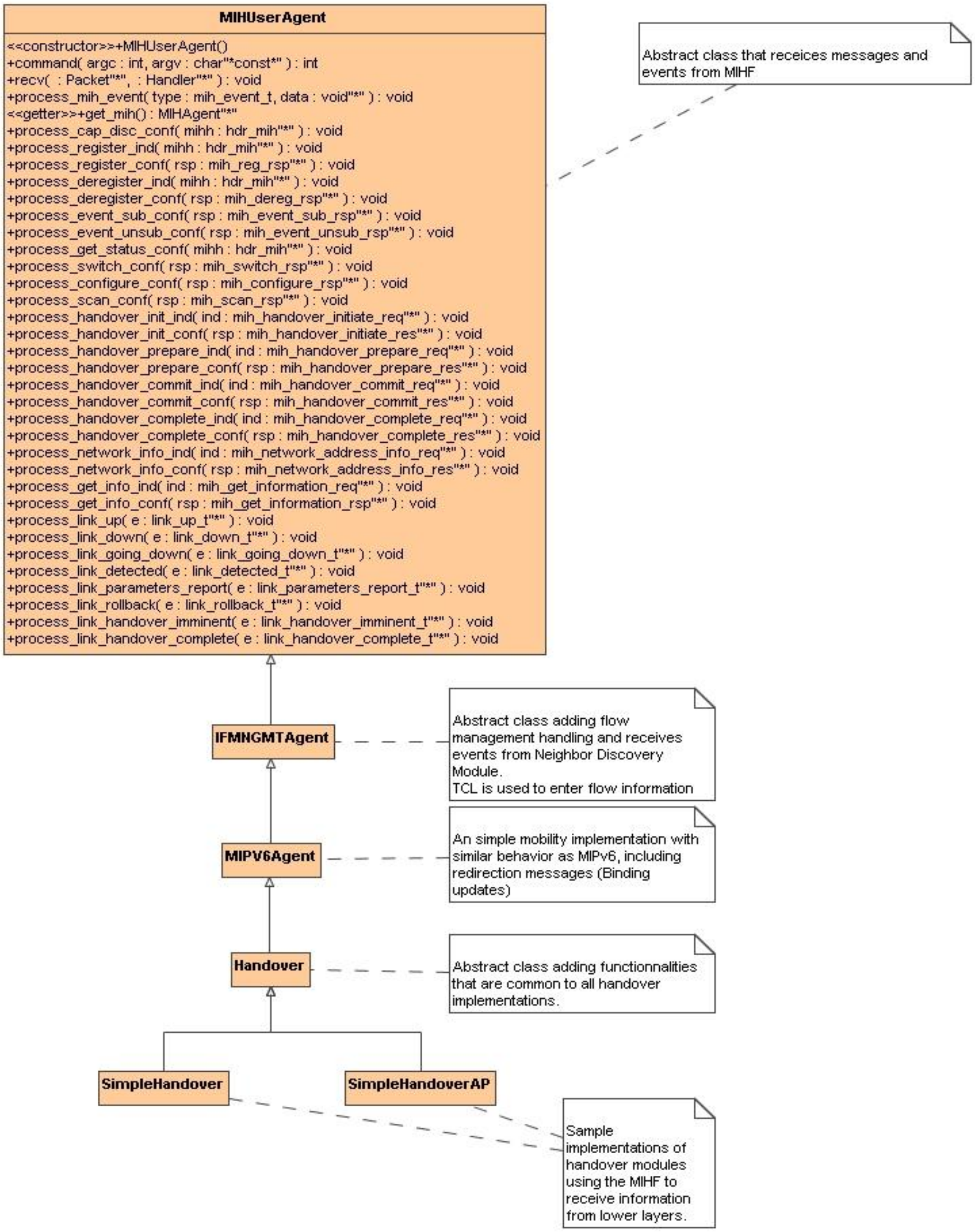

Figure 5-12 MIH User class hierarchy [5]

As show in fig 5.12, the MIHUser send command to MIHF and receive events from the MIHF.The IFMNGMT provides flow management functions. It also receives events from the ND agent when a new prefix is detected or when it expires. The MIPV6Agent adds the redirection capability to the MIH User. A message can be sent to the source node to inform him of the new address or interface to use when a flow need to be redirected. The Handover class helps in computing a new address after successful handover.

5.4.4 Mobility Extensions For NS-2

The mobility model includes enhancements to support handovers in NS-2. This includes:

- Integration of UMTS, 802.16 for heterogeneous handovers.
- Generic design for nodes with multiple interfaces.
- Support for subnet discovery and change of address.
- Support for multiple interfaces

In order to evaluate handover in heterogeneous environment, we integrated multiple packages providing additional technologies. The following is a list of technologies added to the package:

- UMTS: the source code is based on the EURANE code [11]. The modification includes support for hierarchical addressing.
- IEEE 802.16: developed internally and focusing on the mobility aspects of the technology (802.16e) [5].

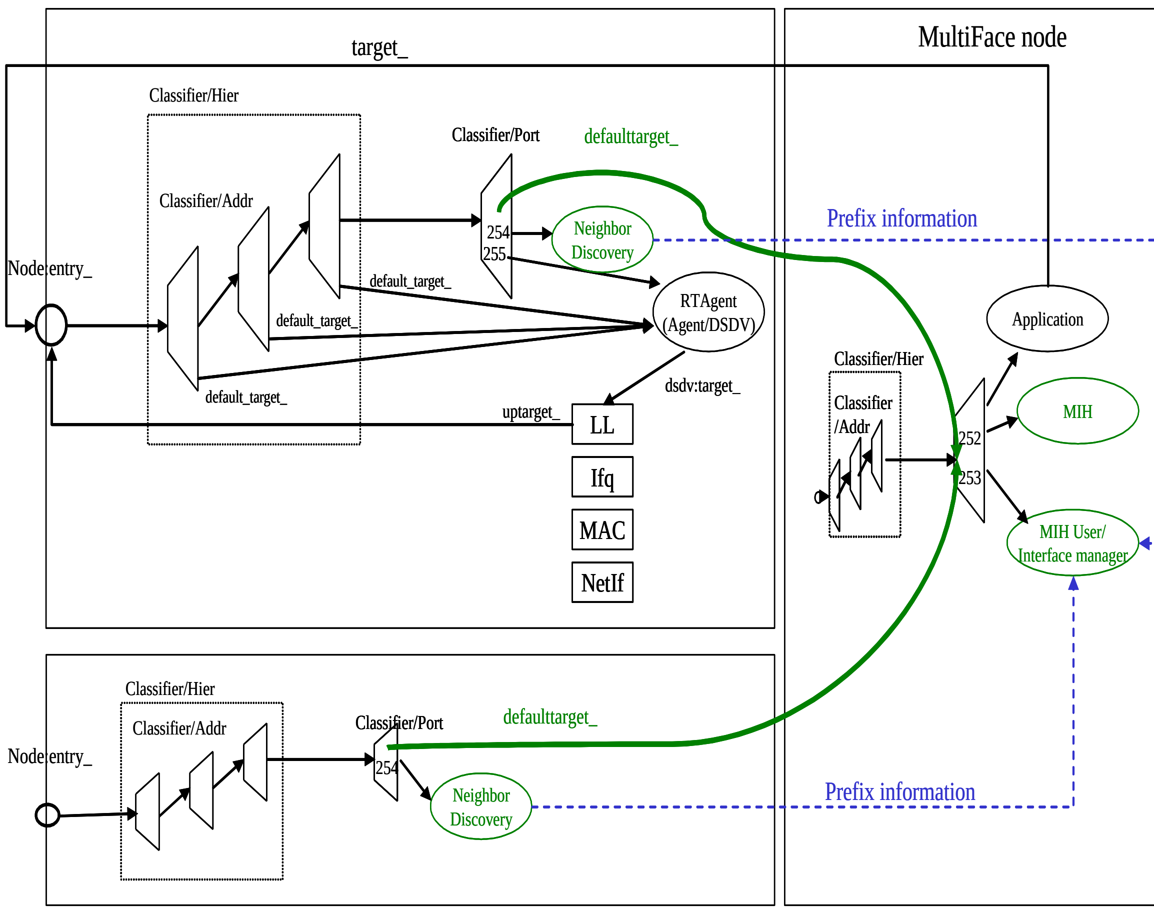

Figure 5-13 Multiple interfaces node design [5]

As show in fig 5.13 multiFace node is a virtual node linking nodes of similar or different technologies. The other nodes are interfaces for multiFace node. An ND agent is used for layer 3 movement detection .The MIH located in the MultiFace node is linked with each MAC object of the interface node.

We have been able to simulate handovers between IEEE 802.16 and UMTS with the help of this design.

5.4.5 Layer 3 Mobility

Neighbor Discovery [5]

The ND module is used to provide Layer 3 movement detection. In the network, APs and BSs send RAs periodically to inform the MNs about the prefix information. The prefix is the address of the BS. The RA is sent using broadcast messages .The ND agent located in the MN receives these RAs and determine if the message contains a new prefix and inform the interface manager. Also timer is associated with the lifetime of the prefix. the prefix expires when the MN losses its connection with the AP/BS, and a notification is sent to the interface manager. The implementation also supports RS to enable a MN to discover new AP/BS after a handover.

Address update

During a simulation, a MN can change network. When switching AP/BS, the node needs to be reachable via a new address. When receiving a new prefix the new node address is calculated. The node ID will be the same in the new network. Only the domain and cluster is updated.

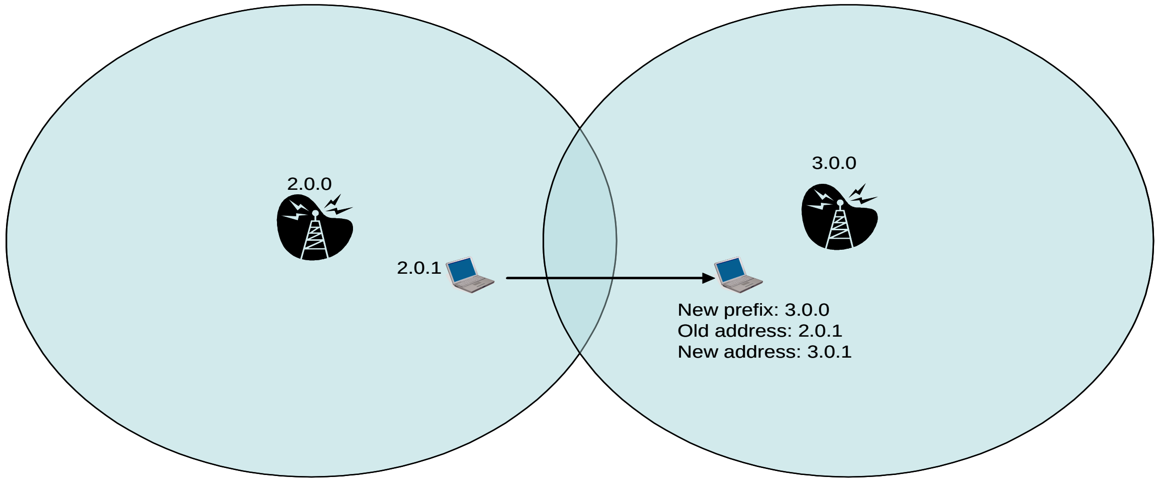

Figure 5-14 Example of node address change [5]

After computing the new address, additional changes are required:

- Update the node address.
- Update the cache (in which node address is present) of each agent.
- Update base station information.

Chapter 6

SIMULATIONS & RESULTS

6.1 UMTS

In this work an ns-2 based simulator EURANE was used, which simulates a HSDPA communication system with different traffic models using HS-DSCH functionalities. The NS-2 was developed for this type of network layer studies and focus on MAC and RLC, where versions of these protocols are implemented for HSDPA according to 3GPP specifications.

6.1.1 Network Model

For simulation purpose we have employed Enhanced UMTS Radio Access Network Extensions (EU-RANE) to ns-2 simulator, in which handover functionality has not been implemented and it is a single cell with one base station only i.e. Node-B, is serving n mobile users on a HSDPA channel using Hybrid CDMA/TDMA transmission scheme. Further, assume that at a given timeslot t, there are m, m <= n active mobile users who are competing with each other for resources at Node-B. The packets to be transmitted have been en-queued at Node-B.

In the HSDPA context, the process of scheduling refers to the process of allocation of transmitter time and power (i.e. at base station, Node-B) to the randomly time-varying mobile data connections (i.e. cellular mobile users, UE).

RR-Scheduling

The simplest type of scheduling (channel non-adaptive), is to allocate the access time fairly to all users, and this is also known as Round Robin (RR) Scheduler.

Best Channel First Scheduling

This scheduling approach can be based on transmitted power by the mobile users, as it is representative of channel conditions. In this case Node-B can compare the power transmitted by each mobile user and match it to the respective packet waiting in the queue for transmission. Thus, Node-B will transmit a packet to the user which has the best channel conditions and so on (channel adaptive). This approach is known as Best Channel First (BCF) or Maximum Carrier-to-Interference ratio (C/I). Maximum C/I is unfair as it favors the users who are closer than users who are further away from Node-B.

In our simulation experiments, the performance of TCP and UDP is determined under two extreme conditions of fairness by employing the RR and Maximum C/I scheduling schemes.

6.1.2 Constant Bit Rate (CBR) Traffic

I have simulated a scenario in which 10 mobile users are downloading data on a HSDPA link from Node-B using UDP. In this scenario, the numbers of UDP connections running CBR traffic are 10. Each of the 10 UEs are connected to Node-B through an acknowledged mode HSDPA channel. The Node-B is connected to Wired Node 1 through RNC, SGSN, GGSN and wired node 2. Thus each UE has an end-to-end UDP connection with wired node 2 as shown in Fig. 6.1. The simulations have been run under ideal environment and for this purpose I have used ideal trace. [1]

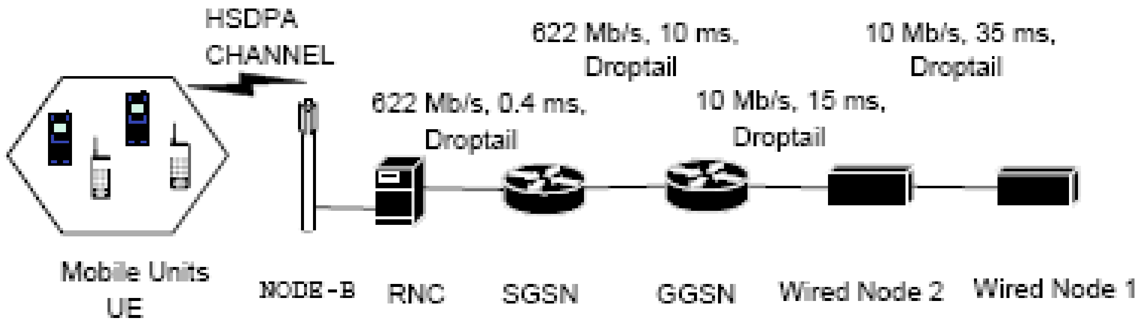

Figure 6-1 Simulated UMTS Network

The simulated network with link capacities is shown in Fig. 6.1.

Wired Link Parameters

a) **Parameters for link between node 2 and node 1**

Delay=35ms

Bandwidth=10Mbit

Queue type=Droptail

Queue size=1000

b) **Parameters for link between node 1 and GGSN**

Delay=15ms

Bandwidth=10Mbit

Queue type=Droptail

Queue size=1000

c) **Parameters for link between GGSN and SGSN**

Delay=10ms

Bandwidth=622Mbit

Queue type=Droptail

Queue size=1000

d) Parameters for link between SGSN and RNC

Delay=0.4ms

Bandwidth=622Mbit

Queue type=Droptail

Queue size=1000

Wireless Link Parameters

a) Parameters for link between RNC and NODE-B

The link between Base station (Node-B) and Radio Network Controller (RNC) is using a Dummy Droptail queue with arbitrarily large size of queue buffer, so that there is no packet loss due to overflow.

- Uplink Delay=15ms
- Downlink delay=15ms
- Uplink Bandwidth=622Mbit
- Downlink bandwidth=622Mbit
- Queue type=Dummy Droptail
- Queue size=2000

b) Parameters for HS-DSCH

- Downlink Bandwidth=64kbps
- Transmit time interaval (TTI) =2ms

As the HS-DSCH always requires an associated DCH, an uplink DCH is always created alongside the HS-DSCH.

DCH parameters

- Uplink bandwidth=64kbps
- Uplink TTI=20ms

CBR Traffic Parameters

- Burst time =0.5sec
- Idle time=0.5sec
- Data rate=150kbps

6.1.2.1 RR-Scheduling

The **throughput** obtained at each of the 10 UEs, under ideal conditions using RR-Scheduling has been plotted in Fig. 6.2.

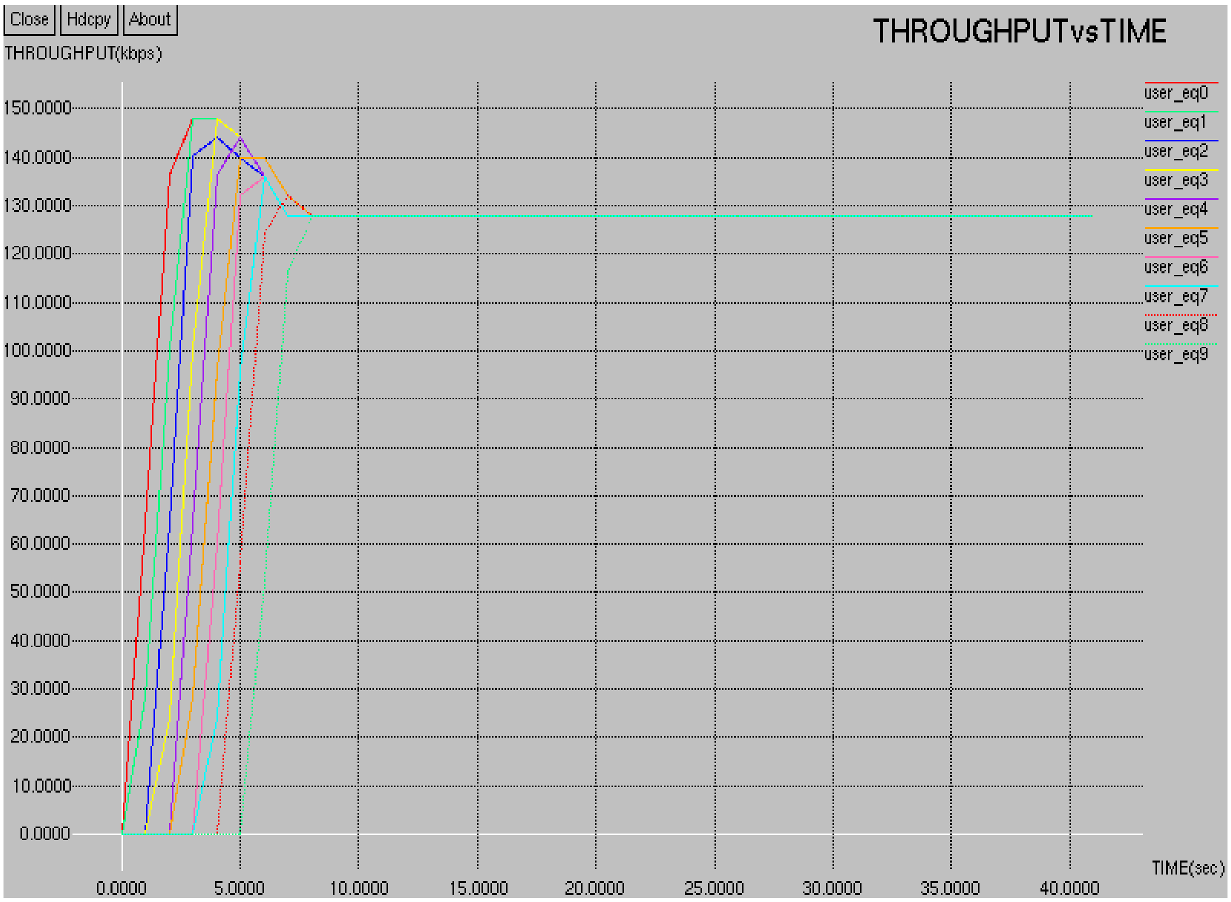

Figure 6-2 Throughput for 10 UEs under ideal conditions with RR-Scheduling.

Date rate for CBR traffic is 150kbps but the total cell bandwidth is approx 1.2Mb so for ideal conditions with RR-Scheduling, there is initial transient between 0 and 5sec. The throughput of each UE achieves a steady value between 125 and 130Kb/s after 8 sec of simulations.

The **End to End Delay** obtained for 10 UEs, under ideal conditions using RR-Scheduling has been plotted in Fig. 6.3.

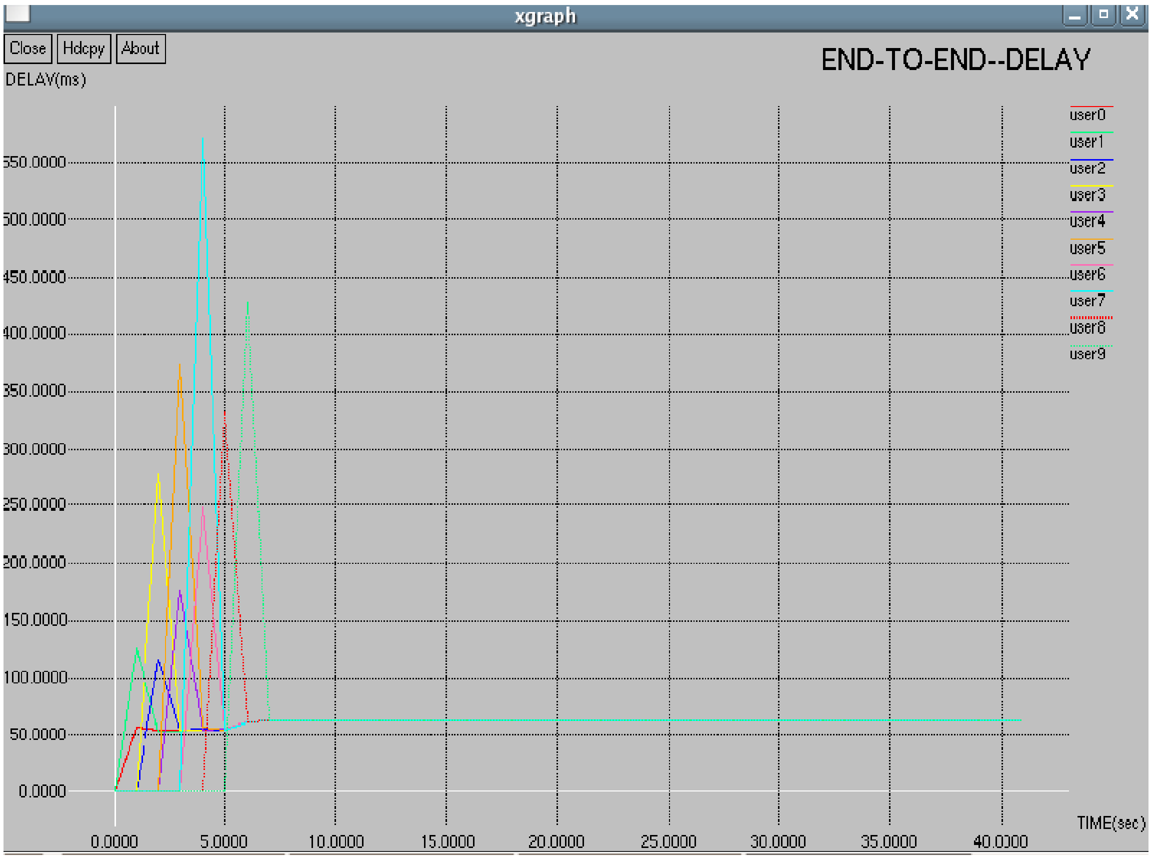

Fig 6.3: End to End Delay for 10 UEs under ideal conditions with RR-Scheduling

So for ideal conditions with RR-Scheduling, the end to end delay for each UE running CBR traffic achieves a steady value between 55ms to 60ms after 7 sec of simulations.

The **Packet loss** rate obtained for 2 UEs, 10 UEs, 20 UEs, under ideal conditions using RR-Scheduling has been plotted in Fig. 6.4

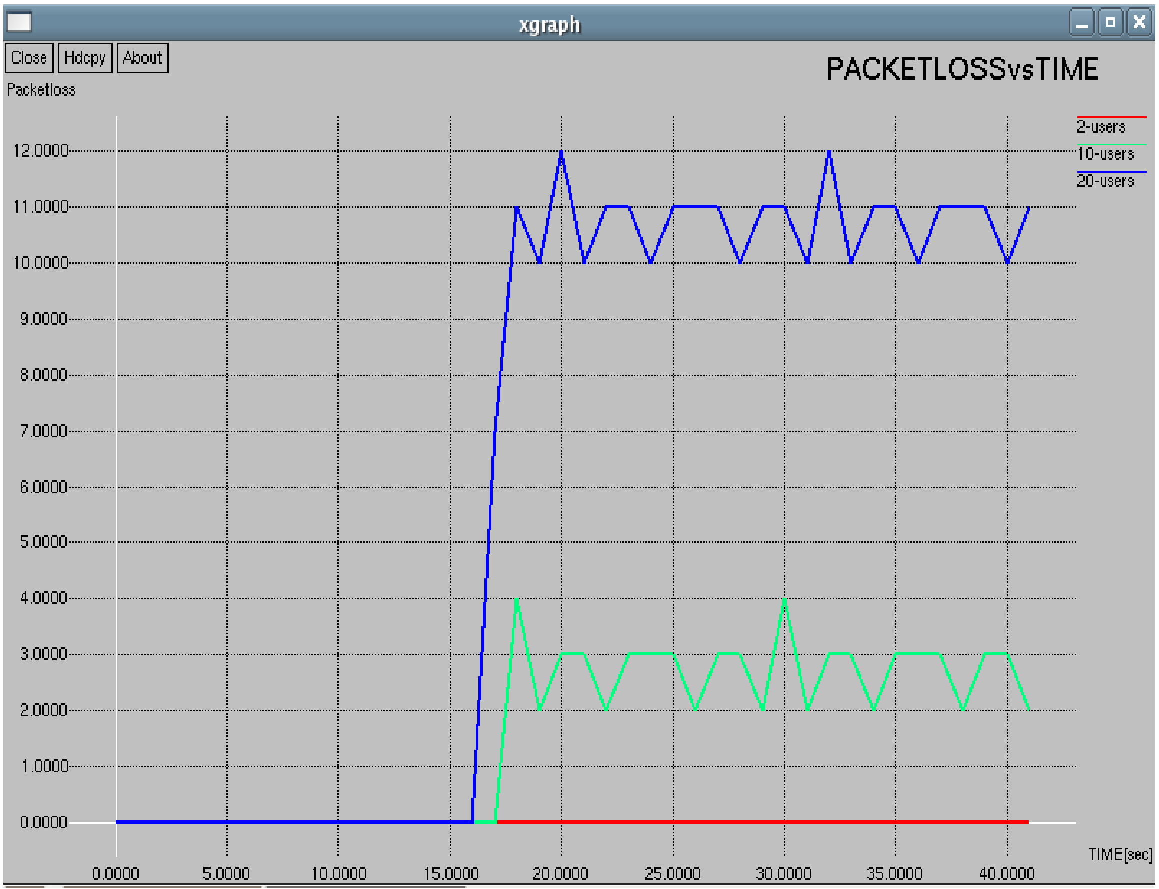

Fig 6.4: Packet loss rate for 2 UEs, 10 UEs, 20 UEs under ideal conditions with RR-Scheduling

It can be seen from fig 6.4 that packet loss rate for 20 users is greater than 10 users and there is 0 packet loss rate for 2 users under ideal conditions with RR-Scheduling. Packet loss rate for 20 users varies between 10 to 11 packets/time after 16 sec of simulation and packet loss rate for 10 users varies between 2 to 3 packets/time after 17 sec of simulations and packet loss rate for 2 users is 0.

The **End to End Delay** obtained for 2 UEs, 10 UEs, 20 UEs, under ideal conditions using RR-Scheduling has been plotted in Fig. 6.5

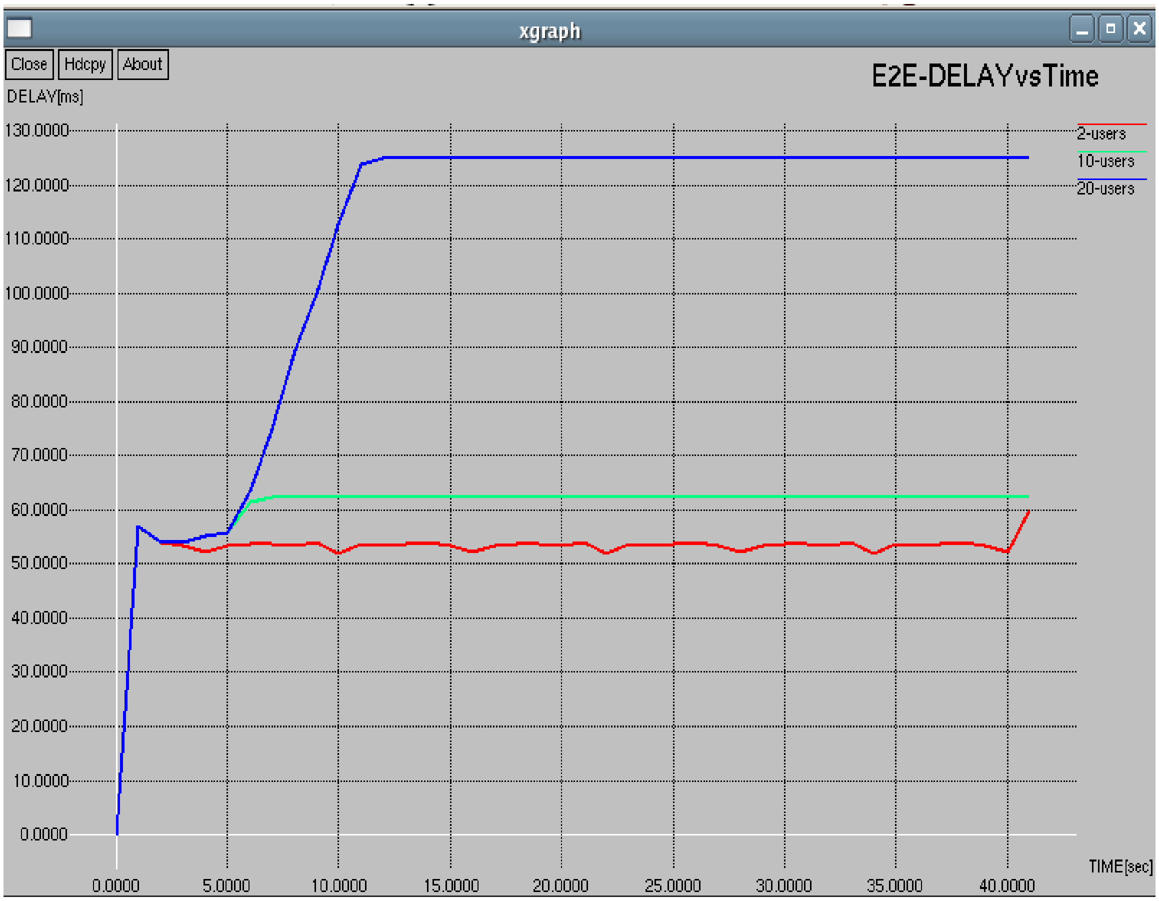

Fig 6.5: End to End Delay for 2 UEs, 10 UEs, 20 UEs under ideal conditions with RR-Scheduling

From fig 6.5 it can be seen that End to End Delay increases as the number of users increases in a cell. Under ideal conditions with RR-Scheduling, the End to End Delay for 20 users achieves a steady value of 125ms after 10sec of simulations, for 10 users it achieves a steady value of 63ms after 6sec of simulations and for 2 users End to End Delay achieves a steady value of 52ms after 5sec of simulations.

6.1.2.2 Max C/I Scheduling

The **throughput** obtained at each of the 10 UEs, under ideal conditions using Max C/I Scheduling has been plotted in Fig 6.6.

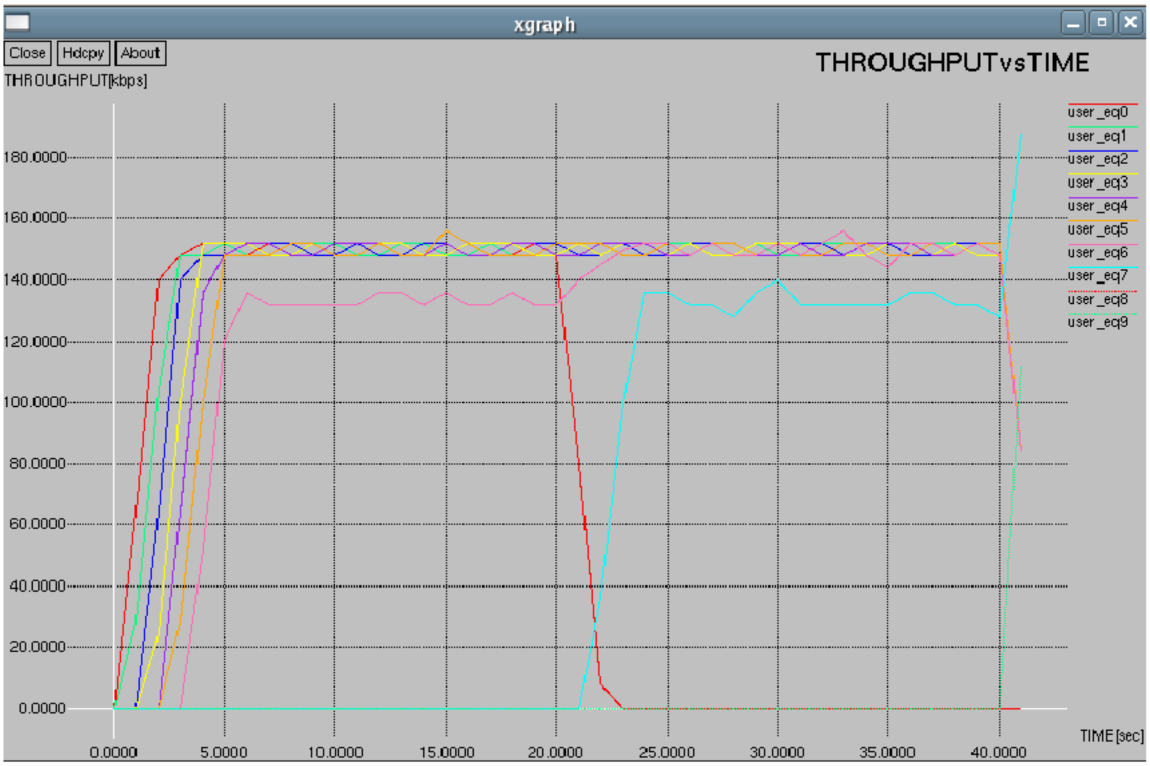

Fig 6.6: Throughput for 10 UEs under ideal conditions with Max C/I Scheduling

As the system bandwidth is approx 1.2Mb, so the cell can accommodate only 7 UEs running CBR traffic with each having rate of 150kbps.It can be seen from fig 6.6 that first 6UEs are getting throughput of approx 150kbps, 7th UE is getting throughput of 138kbps UE 7, thereby depicting the unfairness caused by Maximum C/I Scheduling (unfair to the users which are far from the Base station). Also it can be seen from fig 5 at 20ms UE 0 stop running CBR traffic and system has given admission to the next UE which initially is getting 0 through put.

6.1.2.3 Comparison between RR-Scheduling and Max C/I Scheduling

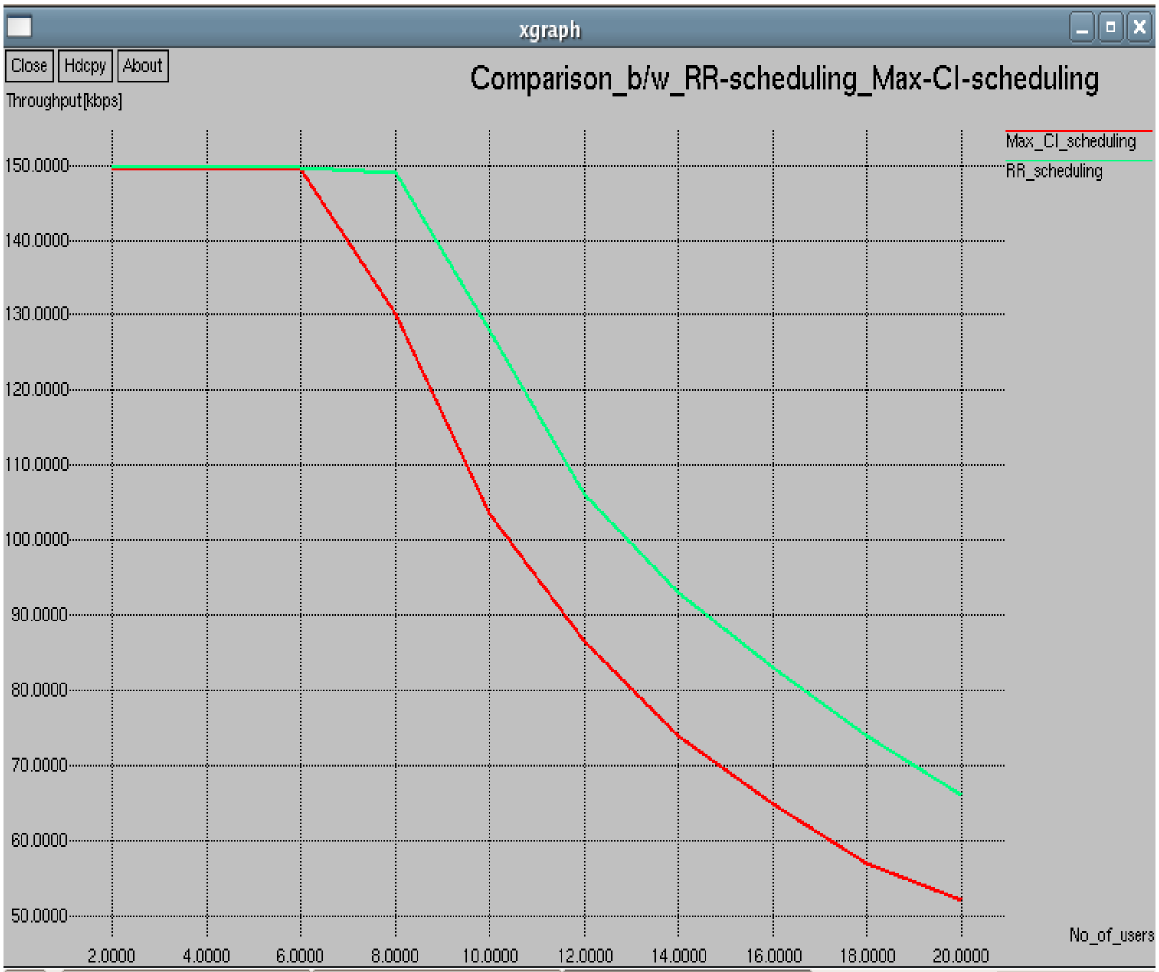

Fig 6.7: Comparison between RR-Scheduling and Max C/I Scheduling

The total cell throughput obtained by all UEs(2,4,6,8,10,12,14,16,18,20), both for RR and maximum C/I, has been plotted in Fig 6.7. It shows that maximum C/I scheduling will generate a lower value of total cell throughput as no of users are increases from 0 to 20 as compared to RR scheduling under ideal conditions.

6.1.3 File Transfer Protocol:

I have simulated a scenario in which 10 mobile users are downloading data on a HSDPA link from Node-B using TCP. In this scenario, the numbers of TCP connections running FTP traffic are 10. Each of the 10 UEs is connected to Node-B through an acknowledged mode HSDPA channel. The Node-B is connected to Wired Node 1 through RNC, SGSN, GGSN and wired node 2. Thus each UE has an end-to-end UDP connection with wired node 2 as shown in Fig.6.1. The simulations have been run under ideal environment and for this purpose I have used ideal trace. [1]

The simulated network with link capacities is shown in Fig. 6.1.For wired links the parameters are same which are for mentioned above for CBR traffic.

TCP Parameters:

- Window = 16
- Packet Size = 512

All other parameters have default values which are given in ns-default.tcl**.**

6.1.3.1 RR- Scheduling

The **throughput** obtained at each of the 10 UEs, under ideal conditions using RR-Scheduling has been plotted in Fig. 6.8.

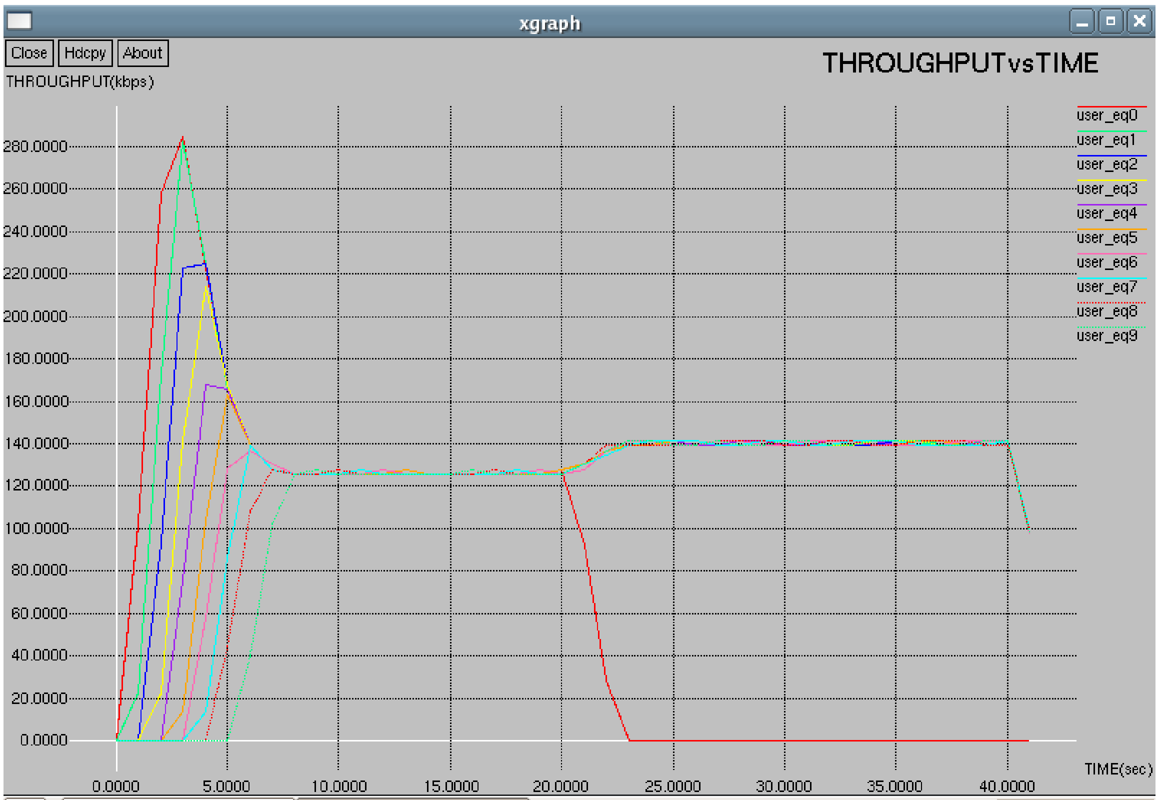

Fig 6.8: Throughput for 10 UEs under ideal conditions with RR- Scheduling

Date rate for CBR traffic is 150kbps but the total cell bandwidth is approx 1.2Mb so for ideal conditions with RR-Scheduling, the throughput of each UE achieves a steady value between 125 and 130Kb/s after 8 sec of simulations. It can be seen from fig 6.8 that at 20ms UE 0 stop running FTP as a result of this throughput of each UE increases and achieves a steady value of 140kbps.

6.1.3.2 Max C/I Scheduling

The **throughput** obtained at each of the 10 UEs, under ideal conditions using Max C/I Scheduling has been plotted in Fig. 6.9.

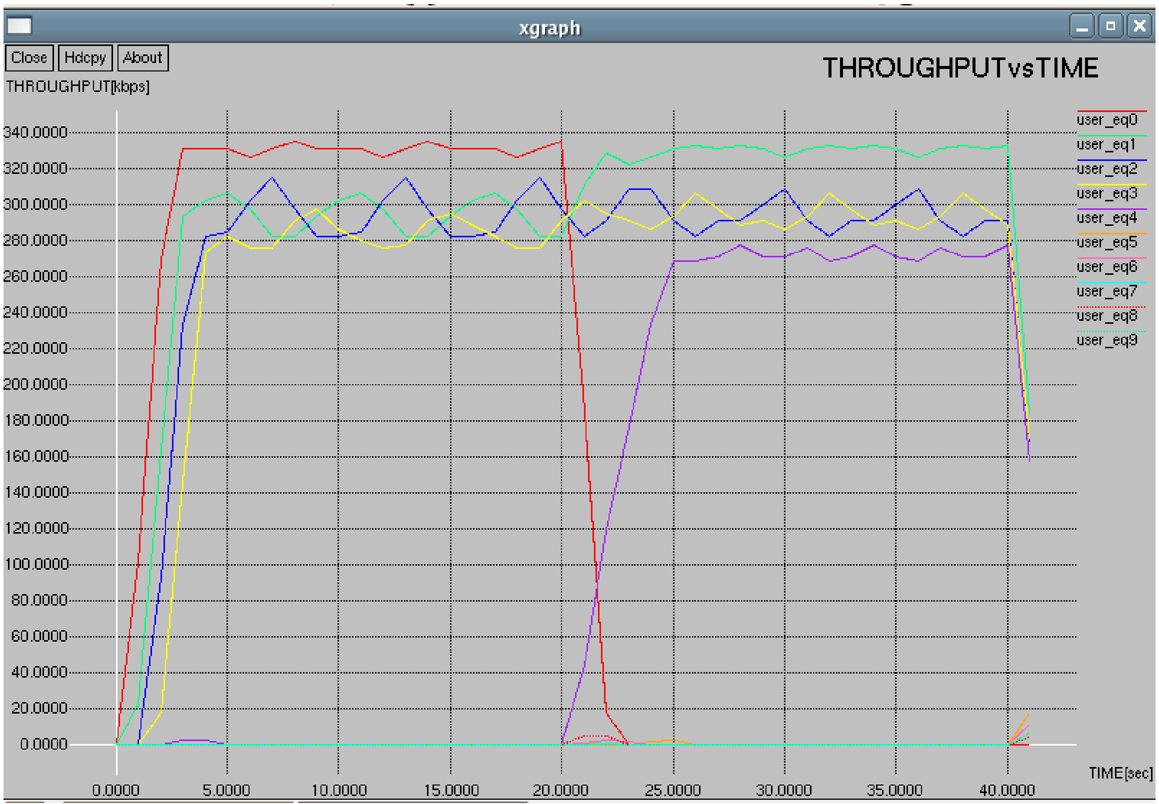

Fig 6.9: Throughput for 10 UEs under ideal conditions with Max C/I Scheduling

As the system bandwidth is approx 1.2Mb, so the cell can accommodate only 4 UEs running FTP and the rest are getting 0 through-put. It can be seen from fig 6.9 that first UE is getting throughput of approx 330kbps, next 3 UEs achieves a steady value between 280 and 300Kb/s after 5 sec of simulations, thereby depicting the unfairness caused by Maximum C/I Scheduling (unfair to the users which are far from the Base station). Also it can be seen from fig 5 at 20ms UE 0 stop running FTP and as a result system has given admission to the next UE which initially is getting 0 through-put.

6.1.4 VOIP Traffic

I have simulated a scenario in which 10 mobile users are downloading data on a HSDPA link from Node-B using UDP. In this scenario, the numbers of UDP connections running Voip traffic are 10. Each of the 10 UEs is connected to Node-B through an acknowledged mode HSDPA channel. The Node-B is connected to Wired Node 1 through RNC, SGSN, GGSN and wired node 2. Thus each UE has an end-to-end UDP connection with wired node 2 as shown in Fig. 3. The simulations have been run under ideal environment and for this purpose I have used ideal trace.[1]The simulated network with link capacities is shown in Fig. 3. For wired links the parameters are same which are for mentioned above for CBR traffic.

VOIP Parameters:

VOIP is basically just UDP packets encapsulating RTP packets with the voice data inside [2], all what is required to simulate a VOIP stream is set the correct packet size and frequency that the packets are sent out and that would simulate a stream and all of this can be done easily by using Pareto on/off generator with burst time of 500ms and idle time of 50ms.Pareto simulate bursty VOIP more precisely than exponential

- Data rate =64k
- Packet size =210
- Burst time =500ms
- Idle time =50ms

6.1.4.1 RR-Scheduling

The **throughput** obtained at each of the 10 UEs, under ideal conditions using RR-Scheduling has been plotted in Fig. 6.10.

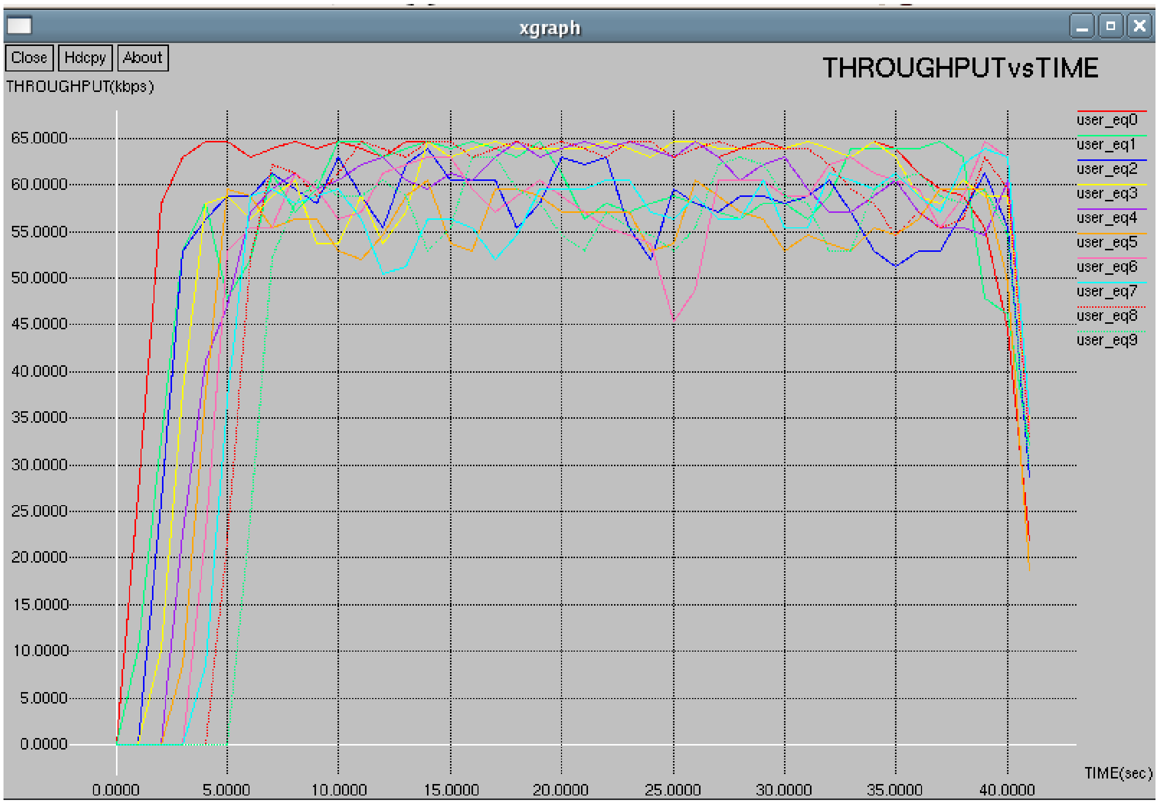

Fig 6.10: Throughput for 10 UEs under ideal conditions with RR-Scheduling

Date rate for VOIP traffic is 64kbps and the total cell bandwidth is approx 1.2Mb, which means that cell can accommodate 10 users with data rate of 64kbps. It can be seen from fig 6.10 that there is initial transient between 0 and 5sec and after 5sec throughput of each UE varies between 55kbps and 63Kb/s.

The **end to end Delay** obtained at each of the 10 UEs, under ideal conditions using RR-Scheduling has been plotted in Fig 6.11

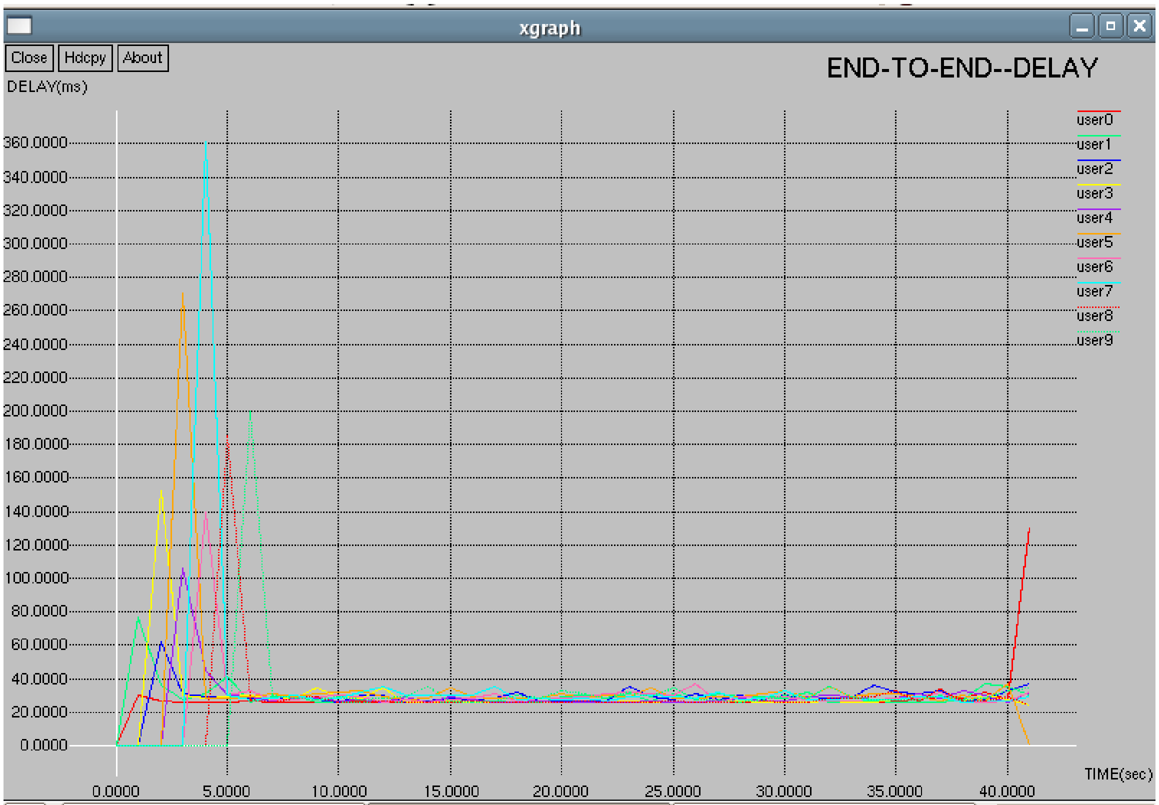

Fig 6.11: End to End Delay for 10 UEs under ideal conditions with RR-Scheduling

From fig 6.11 it can be seen that the end to end delay for each UE running VOIP (Pareto on off) traffic under ideal conditions with RR-Scheduling achieves a steady value of 30ms after 7 sec of simulations.

6.1.4.2 Max C/I Scheduling

The **throughput** obtained at each of the 10 UEs, under ideal conditions using Max C/I Scheduling has been plotted in Fig.6.12.

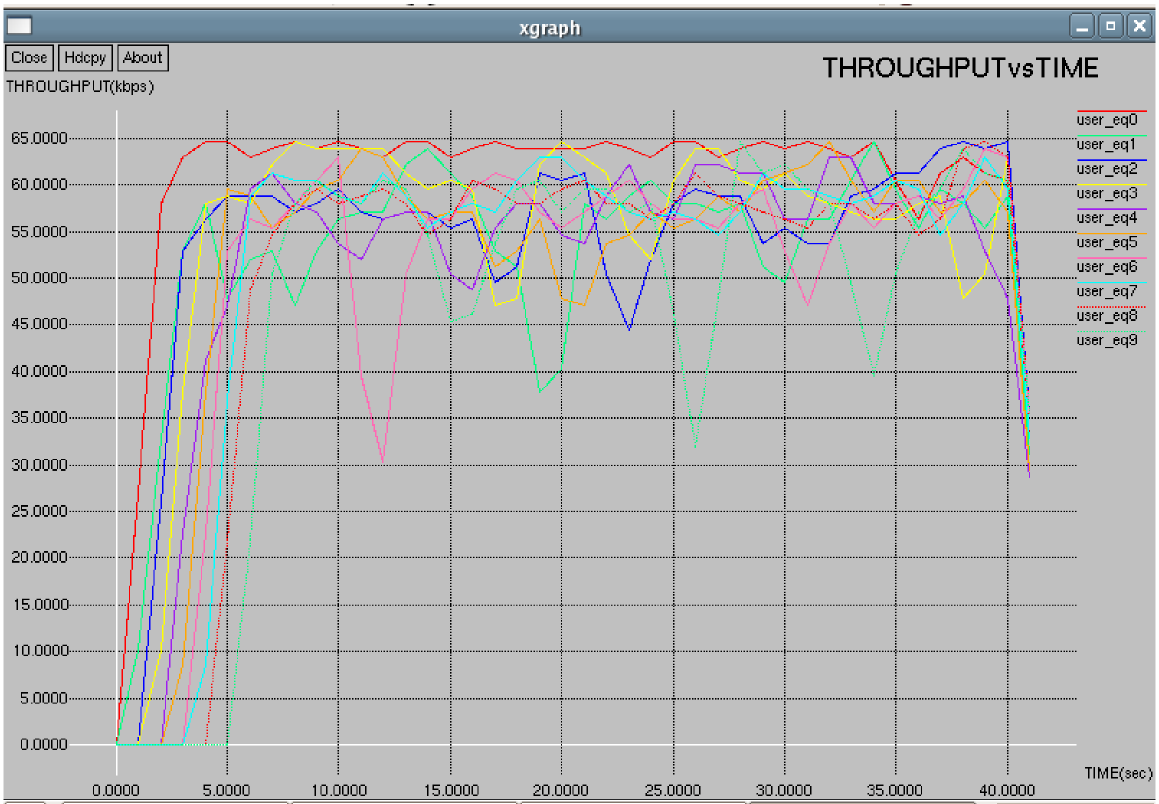

Fig 6.12: Throughput for 10 UEs under ideal conditions with Max C/I Scheduling

Date rate for VOIP traffic is 64kbps and the total cell bandwidth is approx 1.2Mb, which means that cell can accommodate 10 users with data rate of 64kbps.from fig 6.12 it can be seen that for ideal conditions with Max C/I-Scheduling, there is initial transient between 0 and 5sec and after 5sec the throughput of each UE varies between 50kbps and 63Kb/s.

The **end to end Delay** obtained at each of the 10 UEs, under ideal conditions using Max C/I Scheduling has been plotted in Fig 6.13

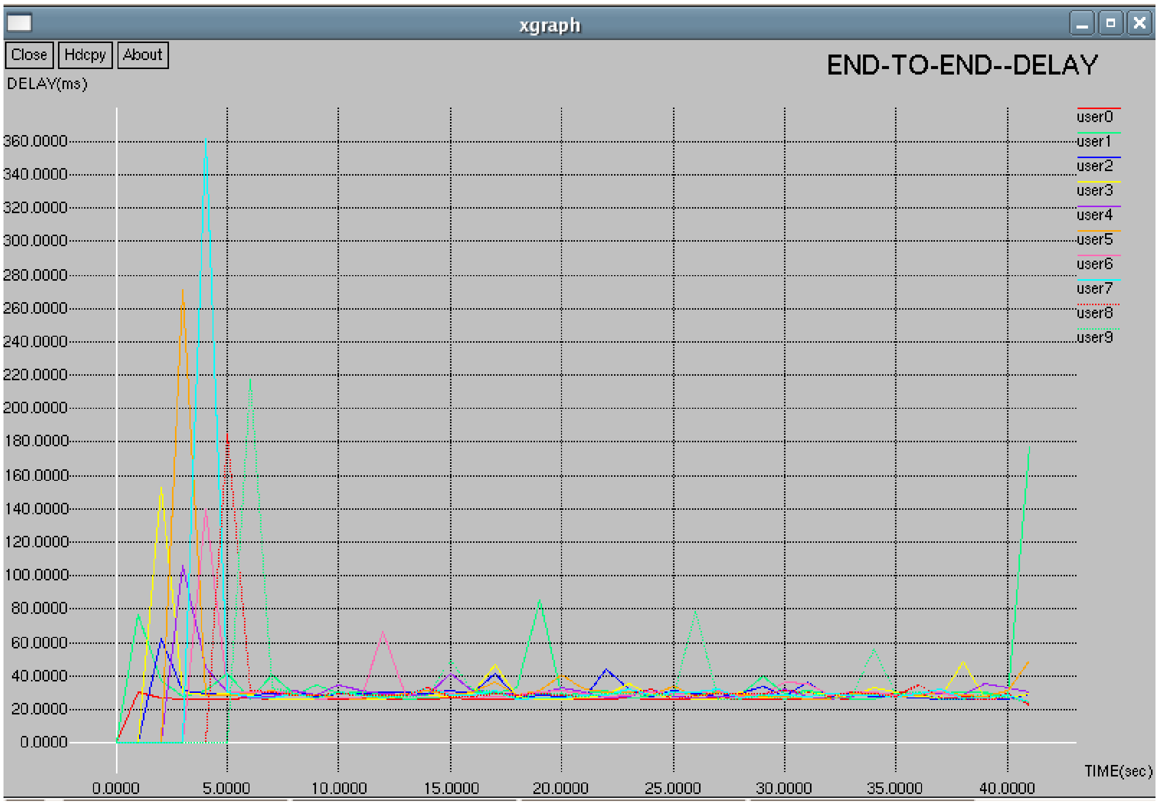

Fig 6.13: End to End Delay for 10 UEs under ideal conditions with Max C/I Scheduling

From fig 6.13 it can be seen that the end to end delay for each UE running VOIP (Pareto on off) traffic under ideal conditions with Max C/I scheduling achieves a steady value of 28ms after 6 sec of simulations.

6.2 SIMULATION FOR WIMAX

6.2.1 Topology Scenario:

In order to simulate the WiMax network we have used a topology in which we have used topography of 1100 x 1100. We have studied communication between Mobile Nodes and Sink Node through 802.16 Base Station. We have connected the Base station with the mobile nodes through wireless channel, whereas the sink node is attached to the base station through a wired link. We have used hierarchal routing in order to route over base station.

```
      |  MN  |       ; 1.0.1

          (^)
           |
    |--------------|
  | Base Station |    ; 1.0.0
    |--------------|
           |
      |-----------|
   | Sink node |     ; 0.0.0
      |-----------|
```

We have defined the coverage area of base station to be 500m.This is done by using receiver’s threshold. We have used Omni directional antenna model. The Mac layer of IEEE 802.16 is used. Under these conditions, we have studied different kind of traffic types such as CBR, FTP and VOIP.

6.2.2 CBR Traffic

CBR stands for constant bit rate traffic. In this simulation we have set up an assumption that we have to give minimum of 12k bandwidth to every user. Therefore we will setup our packet size and gap size (interval b/w two packets) to be 1.

Parameters:

Parameter	**Value**
Packet size	1500
Gap size	1
Number of users	25
Traffic start time	100
Traffic stop time	150
Simulation stop time	250
Size of Interface queue	50

Table 6.2.1 CBR with 25 users

Now as the packet size is 1500 & Gap size or interval is 1, so the rate comes out to be ((1500*8)/1)=12000 or 12 k

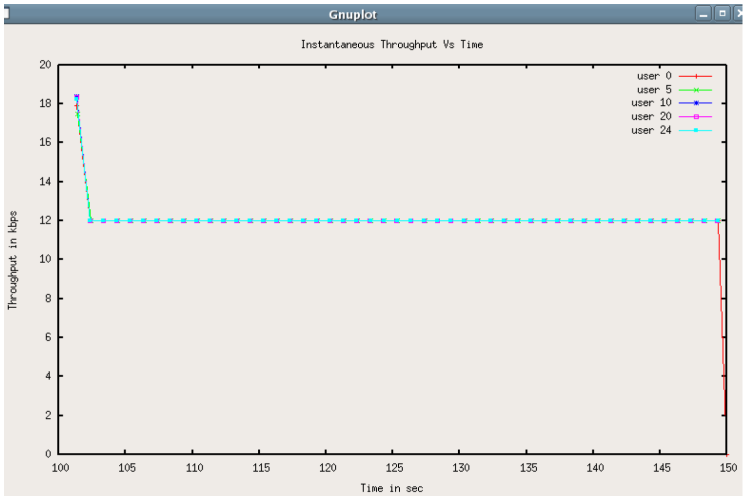

Figure 6.2.1 CBR with 25 users

Comments

The graph is clearly showing that system has sufficient bandwidth to support 25 users & give an average of 12k throughput to all of them

Parameter	Value
Packet size	1500
Gap size	1
Number of users	75
Traffic start time	100
Traffic stop time	150
Simulation stop time	250
Size of Interface queue	50

Table 6.2.3 CBR with 75 users

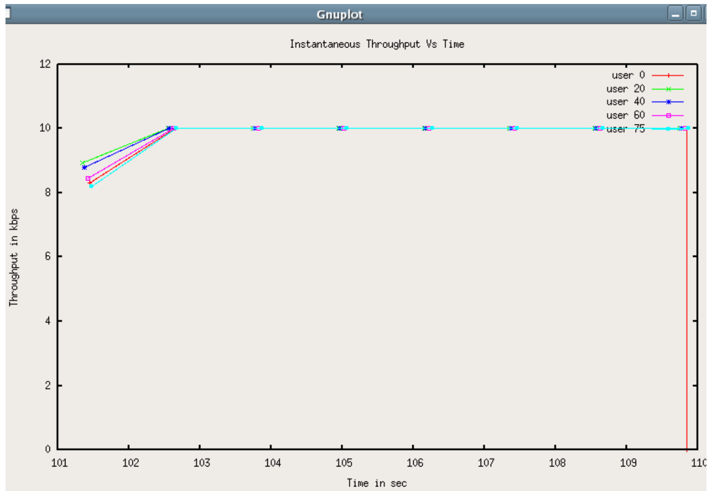

Figure 6.2.3 CBR with 75 users

Comments:

Now the average throughput has reduced & has reached to about 10k.This shows that system was not able to give the desired bandwidth of 12k to every user and as the scheduling scheme is round robin so it has given 10k average throughput to all of the 75 users.

From this we can see that total bandwidth of system comes out to be (75*10k) =750k.So by this we can calculate the capacity of system up to which it can give 12k to every user and this comes out to be 750k/12k=62 (approx).so we can say that with this rate our system can support up to 62 users and we should make such an admission control policy which will not allow more than 62 users at a time in order to ensure Qos.

6.2.3 FTP

Parameter	Value
Packet size	1500
Gap size	1
Number of users	5
Traffic start time	100
Traffic stop time	150
Simulation stop time	250
Size of Interface queue	50

Table 6.2.4 FTP with 5 users

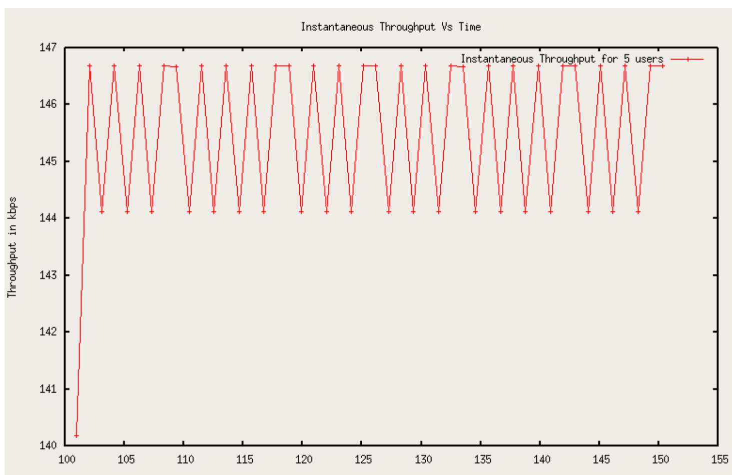

Figure 6.2.4 FTP with 5 users

Comments

Now in case of FTP traffic , as bandwidth of system is 750k so theoretically every user should get a through put of around 150k ,which is evident from the graph.

Parameter	Value
Packet size	1500
Gap size	1
Number of users	50
Traffic start time	100
Traffic stop time	150
Simulation stop time	250
Size of Interface queue	50

Table 6.2.6 FTP with 50 users

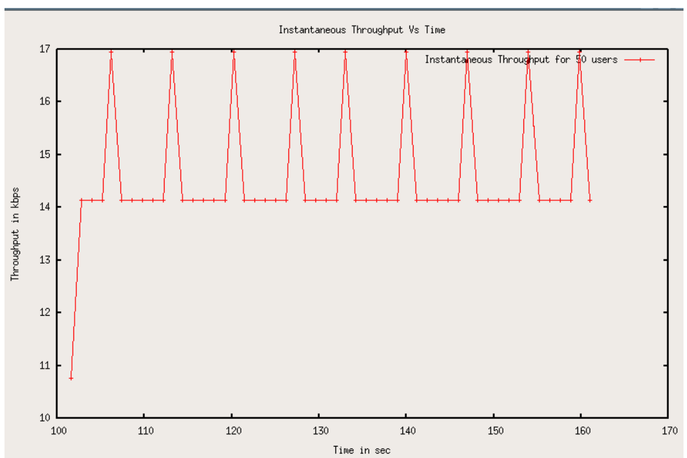

Figure 6.2.6 FTP with 50 users

Comments: similarly in this case the throughput should be around 15k and the graph is showing that

Parameter	Value
Packet size	1460
Gap size	1
Number of users	5,25,50,75
Traffic start time	100
Traffic stop time	150
Simulation stop time	250
Size of Interface queue	50

Table 6.2.8 FTP Comparison of Different users

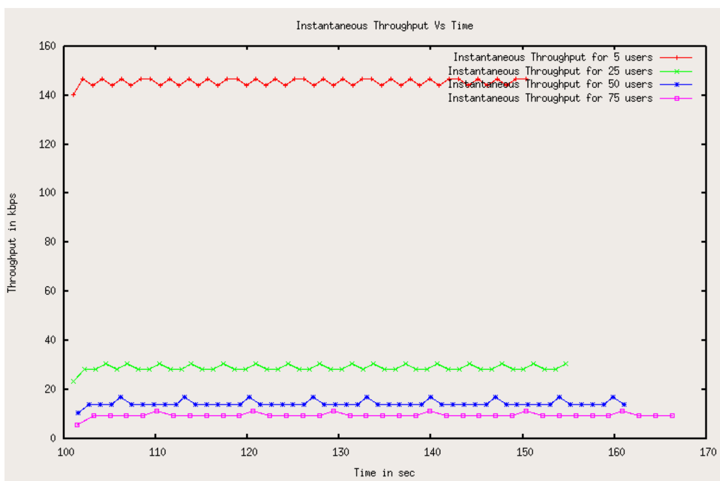

Figure 6.2.8 FTP Comparison of Different users

Comments

Now this is a graph which is showing comparison for 5, 25, 50 & 75 users. it is clearly demonstrating that as the numbers of users are increasing the throughput to each of them is visibly reducing. From this we can easily find out that how many users at any particular case the system can support and we can design our system's admission accordingly.

6.2.4 VOIP

Parameter	Value
Packet size	240
Gap size	1
Number of users	5
Traffic start time	5
Traffic stop time	150
Simulation stop time	250
Size of Interface queue	50

Table 6.2.9 VOIP with 5 users

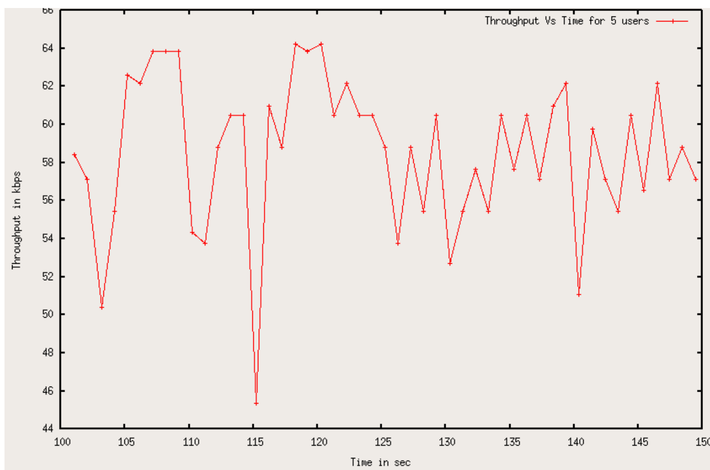

Figure 6.2.9 VOIP with 5 users

Parameter	Value
Packet size	240
Burst time	500ms
Idle time	50ms
Number of users	5,25,50,75
Traffic start time	15,25,35,50
Traffic stop time	150
Simulation stop time	250
Size of Interface queue	50

Table 6.2.11 VOIP Comparison of Different users

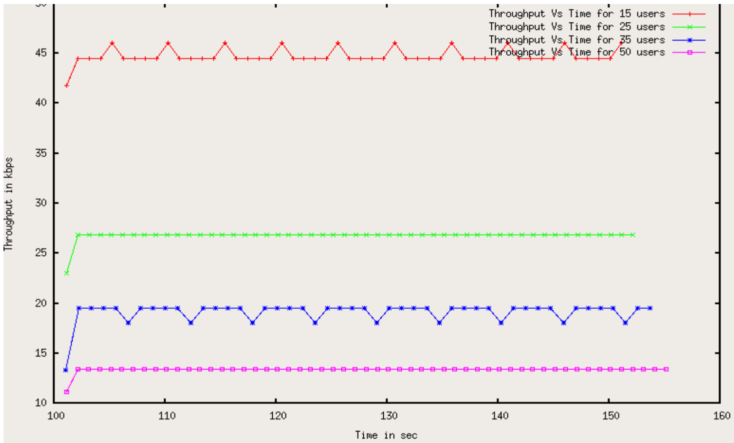

Figure 6.2.11 VOIP Comparison of Different users

Comments

The effect of number of users on system bandwidth is clearly shown by graph. Now if we set up a standard that we have to give users the quality just like normal telephone line so we will have to keep the minimum throughput up to 64k for each user , In that case we can only support up to 5 users. but as it happens in case of Voip systems now a days that normally a throughput of 12k to 16 k is enough ,so by choosing that as our standard we can support up to 35 to 40 users.

6.2.4 Effect of Channel Coding

In order to see the effects of channel coding and modulation scheme on systems performance , we have simplified our scenario .we have brought down our simulations to just 1 user for only 2 sec by using same packet size of 1500 and gap size of .0005

Parameter	Value
Packet size	1500
Gap size	.0005
Number of users	1
Traffic start time	5
Traffic stop time	7
Simulation stop time	7.5
Modulation Scheme	16QAM
Channel coding	1-2
Cyclic prefix	0.25
Maximum Throughput	7000

Table 6.2.12 16QAM Modulation with 1-2 Channel Coding

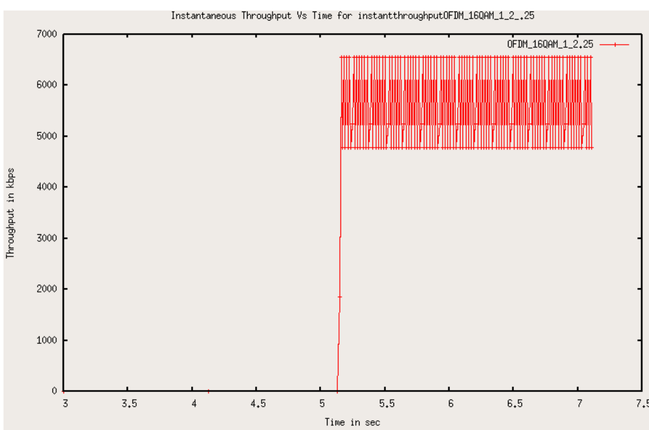

Figure 6.2.12 16QAM Modulation with 1-2 Channel Coding

Comment

The system is giving throughput of maximum 7000k to that user if we use 1-2 channel coding

Parameter	Value
Packet size	1500
Gap size	.0005
Number of users	1
Traffic start time	5
Traffic stop time	7
Simulation stop time	7.5
Modulation Scheme	16QAM
Channel coding	3-4
Cyclic prefix	0.25
Maximum Throughput	10000

Table 6.2.13 16QAM Modulation with 3-4 Channel Coding

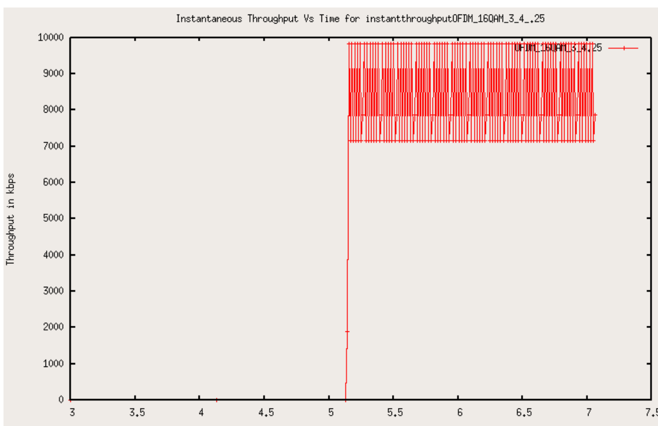

Figure 6.2.13 16QAM Modulation with 3-4 Channel Coding

Comment

The system is giving maximum throughput of 10000k to that user if we use 3-4 channel coding

Parameter	Value
Packet size	1500
Gap size	.0005
Number of users	1
Traffic start time	5
Traffic stop time	7
Simulation stop time	7.5
Modulation Scheme	16QAM
Channel coding	1-2 & 3-4
Cyclic prefix	0.25
Maximum Throughput	10000

Table 6.2.14 Comparison 16QAM Modulation with 1-2 & 3-4 Channel Coding

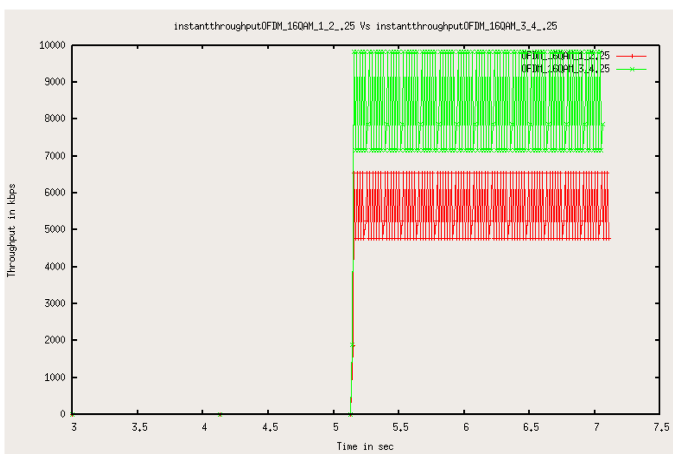

Figure 6.2.14 Comparison 16QAM Modulation with 1-2 & 3-4 Channel Coding

Comment

Now from this graph we can clearly see the effect of channel coding on systems performance. Actually while doing channel coding we add redundant bits to actual information bits. In case of 1-2 coding 1 redundant bit is added for bit is added for every information bit , whereas in case of 3-4 coding 3 bits out of 4 are information bits and only 1 bit is redundant ,so the results are better and we are getting an increased throughput by using 3-4

Parameter	Value
Packet size	1500
Gap size	.0005
Number of users	1
Traffic start time	5
Traffic stop time	7
Simulation stop time	7.5
Modulation Scheme	64QAM
Channel coding	2-3 & 3-4
Cyclic prefix	0.25
Maximum Throughput	15000

Table 6.2.17 Comparison 64QAM Modulation with 2-3 & 3-4 Channel Coding

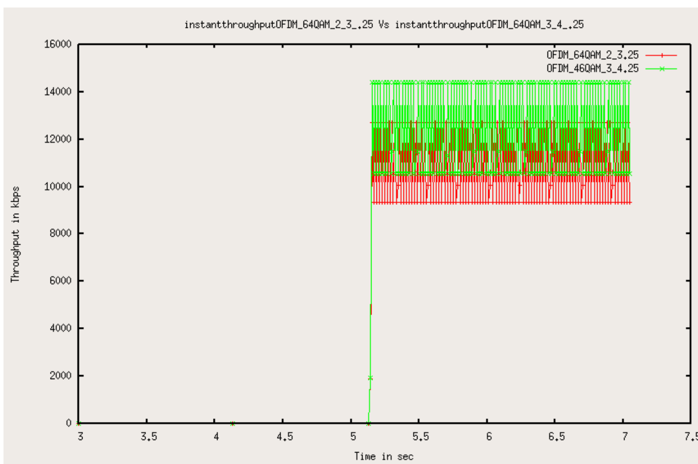

Figure 6.2.17 Comparison 64QAM Modulation with 2-3 & 3-4 Channel Coding

Similar effect of channel coding on systems performance due to channel coding can be seen by this graph. here in case of 2-3 coding 1 redundant bit is added for bit is added for every 2 information bits , whereas in case of 3-4 coding 3 bits out of 4 are information bits and only 1 bit is redundant ,so the results are better and we are getting an increased throughput by using 3-4

6.2.5 Effect of Modulation Scheme:

Parameter	Value
Packet size	1500
Gap size	.0005
Number of users	1
Traffic start time	5
Traffic stop time	7
Simulation stop time	7.5
Modulation Scheme	BPSK & QPSK
Channel coding	1-2
Cyclic prefix	0.25
Maximum Throughput	3000

Table 6.2.20 Comparison BPSK Modulation with QPSK Modulation

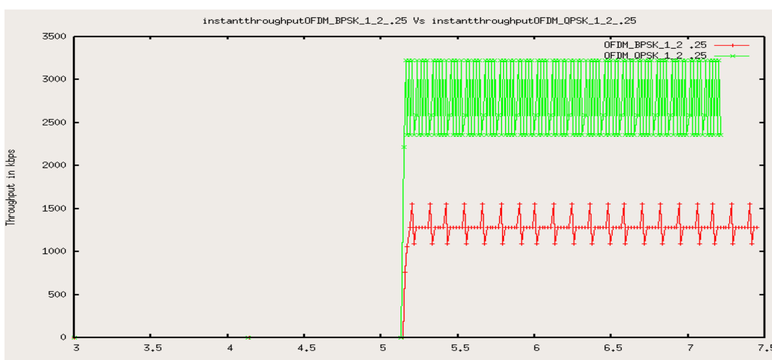

Figure 6.2.20 Comparison BPSK Modulation with QPSK Modulation

Parameter	Value
Packet size	1500
Gap size	.0005
Number of users	1
Traffic start time	5
Traffic stop time	7
Simulation stop time	7.5
Modulation Scheme	16 QAM & 64 QAM
Channel coding	3-4
Cyclic prefix	0.25
Maximum Throughput	15000

Table 6.2.20 Comparison 16QAM Modulation with 64QAM Modulation

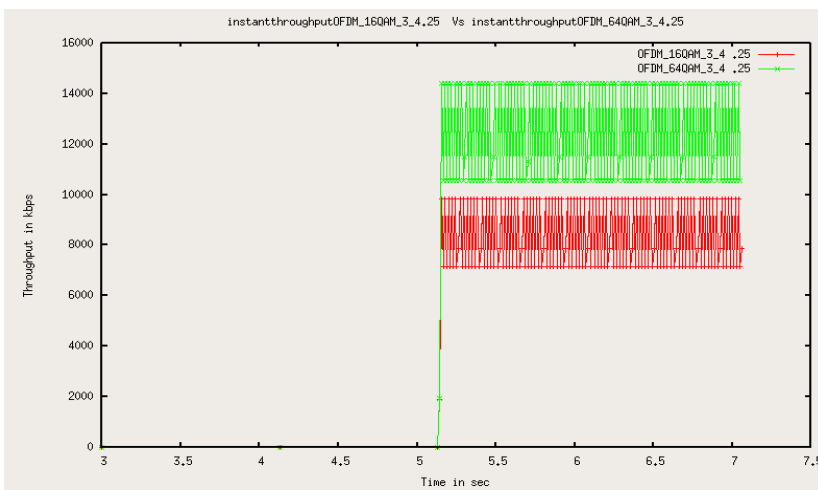

Figure 6.2.20 Comparison 16QAM Modulation with 64QAM Modulation

6.3 INTEGRATED MODULE

Scenario

Create a multi-interface node using different technologies. There is a UDP connection between the router0 and MultiFaceNode. We first use the 802.16 interface, and then we switch the traffic to UMTS when it becomes available. When the node leaves the coverage area of 802.16, it creates a link going down event to redirect to UMTS.

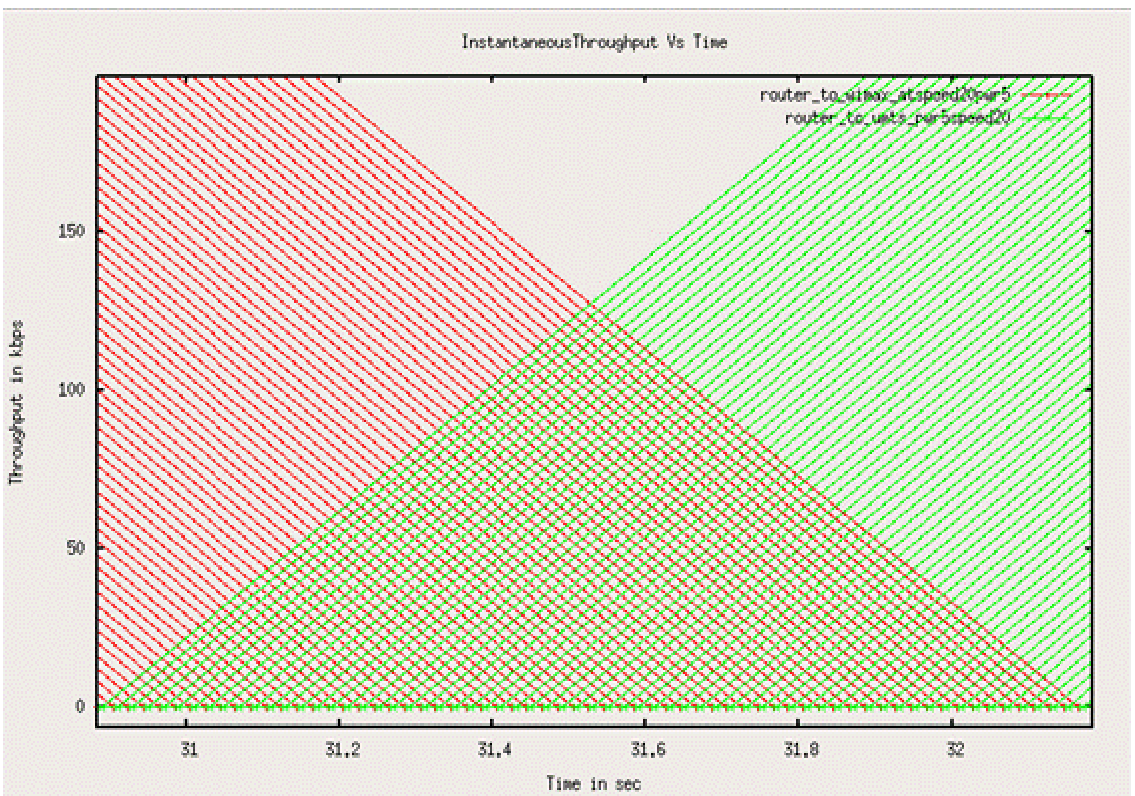

FIG 6.3.1 Handover from Wimax to UMTS at speed 5m/s, power 5 db

It can be seen from fig6.3.1 that vertical handover is taking place between WiMax and UMTS networks at 30sec when speed of user is 20 m/s.When the speed of user is decreased to 5m/s as in fig below the vertical handover is delayed and it takes place at 81sec.

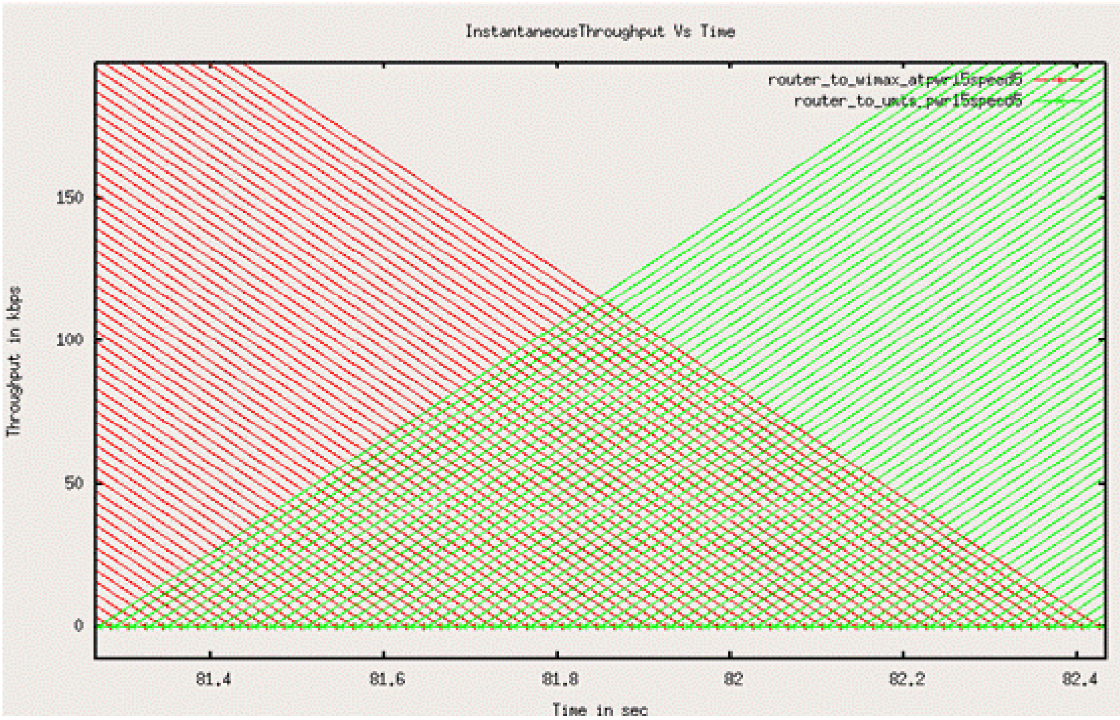

FIG 6.3.2 Handover from Wimax to UMTS at speed 5m/s, power 5 db

It can be seen from fig that vertical handover is taking place between WiMax and UMTS networks at 81sec when user is moving with speed of 5 m/s.

CONCLUSION

In this project a simulations based approach has been adopted to determine the total cell throughput by implementing different traffic models (CBR, FTP, and VOIP) for different number of users on both Wimax and UMTS networks.

Also in UMTS network the performance of HSDPA is evaluated with both round robin scheduling and best channel first scheduling. Simulation results show that under ideal condition high cell throughput is obtained by round robin scheduling. In WiMax throughput performance of different modulation schemes and same modulation schemes with different channel coding is determined. Simulations results show that throughput performance is higher for higher channel coding.

In the end a simple simulations scenario is carried out on integrated model in which UMTS and WiMax networks are integrated in which vertical handover is triggered by the speed of the user who is moving from one network to the other. Simulation results shows that vertical handover is triggered quickly when user is moving fast and traffic flow is seamlessly switched to the other network .

RECOMENDATIONS

Performance evaluation of HSDPA with other scheduling mechanisms and in case of heterogeneous networks triggering of vertical handover according to the user preferences is left for future work.

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