PERFORMANCE EVALUATION OF WiMAX FOR MULTIMEDIA APPLICATIONS

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By

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UNDER THE SUPERVISION OF

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Dedicated to

My supervisor, with gratitude for the guidance and valuable suggestions.
All upcoming Research Scholars!!
My family, with all devotion for their continued love and support.
My friends who acted as a valuable support team giving me the confidence and motivation to complete this life achievement.

I send you all my deepest thanks

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CERTIFICATE

This is to certify that the dissertation entitled "**Performance Evaluation of Wimax for multimedia Applications**" being submitted by **AMIRA** to the School of Computer and Systems Sciences, **Jawaharlal Nehru University**, New Delhi, in partial fulfillment of the requirements for the award of the degree of **Master of Technology** in **Computer Science and Technology**, is a record of bonafide work carried out by her under the supervision of **Dr. D. K. Lobiyal.**

The matter embodied in this dissertation has not been submitted to any other University or Institution for the award of any other degree or diploma.

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DECLARATION

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Abstract

The new era of communication, currently employed in some parts of the world, is Worldwide Interoperability for Microwave Access (WIMAX). It is the latest technology which is approved by IEEE 802.16 group, which is a standard for point-to-multipoint wireless networking. WiMAX vision is to deliver "last mile" broadband connectivity to home or business locations, also its data rates are comparable with Cable and Digital Subscriber line (DSL) rates. It has the capability, which connects to the ISP (Internet Service Provider) even when you are roaming outside home or office. The WiMAX technology is becoming the way to avert the impending crisis of rural connectivity i.e. it will be accessible till the last mile.

The introduction of the IP multimedia subsystem on 3G cellular networks and the integration with other widely deployed wireless networks based on the IEEE 802 protocol family require support for both mobility and quality of service. When mobile systems move across heterogeneous networks, ongoing real-time sessions are affected not only by handoff delay but also by different packet delay and bit rate.

Anytime and anywhere connectivity require supporting smooth mobility with seamless handovers between the heterogeneous networks. A Media Independent Handover (MIH) standard is being developed by IEEE 802.21 to enable the handover of IP sessions from layer 2 access technology to another to achieve mobility of end user devices.

In this work using NS-2 simulator, Evalvid Framework, and NIST mobility WiMAX module, we evaluate the performance of multimedia applications. The experimental work is carried out using Wireless-Wired environment for a MultiFace node that transmits the video stream to a router while it move within the environment. The results of experiments were not suitable enough for supporting multimedia applications

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Amira Fahoud

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Abbreviations

0-9

3G	Third Generation
3GPP	Third Generation Partnership Project
4G	Fourth Generation

A

AP	Access Point
AC	Access categori
ACk	Acknoledgement
ARP	Address Resolution Protocol

B

BS	Base Station
BSS	Basic Service Set
BSSID	Basic Service Set Identifier
BER	Bit Error Rate

C

CAP	Controlled access phases
CBR	Constant bit rate
CDMA	Code-division multiple access
CDF	Cumulative distribution function
CFP	Contention-free period
CID	Connection identifier
CN	Correspondent node
CoA	Care-of address
CP	Contention period
CSMA	Carrier sense multiple access
CSMA/CA	CSMA with collision avoidance
CTS	Clear to send
CW	Contention window

D

DCD	Downlink channel descriptor
DCF	Distributed coordination function
DIFS	Distributed interframe space
DL-MAP	Downlink map message
DSL	Digital subscriber line

F

FA	Foreign Agent
FTP	File Transfer Protocol
FDD	Frequency-division duplex
G	
GPRS	General Packet Radio Service
GSM	Global System for Mobile communication
GUI	Graphical User interface
GPS	Global Positioning System
GW	Gateway
H	
HA	Home Agen
HI	Handover initiate
I	
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
ISP	Internet Service Provider
IP	Internet Protocol
IETF	Internet engineering task force
L	
Layer 2	L2 Link layer
Layer 3	L3 IP layer
M	
M-MIP	Multihomed Mobile IP
MN	Mobile Node
MAC	Medium access control
MIHF	Media independent handoff function
MIPv6	Mobile IPv6 mobility management
MSDU	MAC Service Data Unit
N	
NIC	Network Interface Card
nrtPS	Non-real-time polling service
P	
PCF	Point coordination function
PDF	Probability density function
PDU	Protocol data unit
PMP	Point-to-multipoint

Q

QoS	Quality of service

R

RTP	Real Time Protocol
RSS	Received Signal Strength
RTT	Round Trip Time
RCoA	Regional care-of address
rtPS	Real-time polling service
RTS	Request to send

S

SAP	Service Access Point
SSID	Service Set Identifier
SS	Subscriber station

T

TCP	Transmission Control Protocol
TDD	Time-division duplex
TDMA	Time-division multiple access

V

VPN	Virtual Private Network
VoIP	Voice over IP

U

UMTS	Universal Mobile Terrestrial System
UCD	Uplink channel descriptor
UDP	User Datagram Protocol
UDP	User datagram protocol
UGS	Unsolicited grant service
UL-MAP	Uplink map message

W

WMAN	Wireless Metropolitan Area Networks
WPAN	Wireless Personal Area Networks
WWAN	Wireless Wide Area Networks
Wi-Fi	Wireless-fidelity
WiMAX	Worldwide interoperability for microwave access

CHAPTER 1

CHAPTER1

INTRODUCTION

Nowadays, we are witnessing an explosion of IP connectivity demand and a rapid development of the corresponding technologies in the wireless telecommunication networks. Providing IP services for all, anywhere and anytime becomes a challenge that demands a new requirements for the network topologies. IP services deployment requires true mobile broadband IP connectivity on a globe scale. The third generation mobile cellular systems can be the main player, but its throughput is insufficient. Therefore, Third Generation Partnership Project (3GPP) considered WiMAX and Wi-Fi technologies as complementary broadband wireless access [4].

WiMAX (Worldwide Interoperability for Microwave Access) has been one of the most important technologies in telecommunication networks during last few years. Mobility is the most challenging research in WiMAX networks. This is highlighted in broadband wireless communications especially how to support mobility in WiMAX networks smoothly and seamlessly. There is also more demand for new data services which need high data rate [1], like real time services such as triple play, IPTV and VoIP which are very sensitive to packet loss especially during the handover procedure so, the long interruption of handovers is not acceptable for this kind of services. The handover should be done quickly to reduce the delay. Although, in recent years, various wireless access technologies have continuously grown as well as user demands have, but the user demands have not been satisfied by each wireless access technology alone[6]. Therefore, the Future Generation Networks (FGNs) are proposed to integrate various heterogeneous access technologies in order to support both vertical and seamless handovers. Especially, the IEEE 802.21 specifies Media Independent

Handover (MIH) services to enhance the mobile user experience by optimizing handovers between heterogeneous access networks [6].

1.1 What is WiMAX?

WiMAX stands for Worldwide Interoperability for Microwave Access (WiMAX). It is the common name associated to the IEEE 802.16a/REVd/e standards. It is wireless broadband technology for Worldwide Metropolitan Area Networks (WMAN). The industry trade group WiMAX Forum has defined WiMAX as a "last mile broadband wireless access (BWA) alternative to cable modem service, telephone company, Digital Subscriber Line (DSL) or T1/E1 service"[2]. These technologies have already being deployed in the backhaul and make it matured and reliable with large bandwidth but the last mile is still the bottleneck to enable the broadband applications like the voice over IP , interactive games, multimedia streaming and video conference. These traditional broadband Internet access are not available in the remote, less density, or rural areas because of many reasons like high cost, difficulties in installation, and low profit. But, WiMAX offers a cost-effective and a quickly deployable alternative to wired networks and to the last mile wireless connection problems in metropolitan and underserved rural areas [4] WiMAX supports applications with different QoS requirements in term of delay, jitter, and bandwidth and defines different traffic models according to the requirement of applications in terms of delay, jitter, and bandwidth [6].

1.2 WiMAX Architecture

WiMAX forum defines an architecture within which different vendors equipment can interoperate flawlessly while conforming to IEEE802.16 [3].There are three main components of WiMAX network architecture. They are mobile stations, Access service network, and connectivity service network. The main components of Access service network are the base stations, which provide the air interface for the mobile stations. It also provides mobile management functions, triggering and radio resource management, and ASN gateways which build the radio access at the end [3]. The connectivity service network is responsible for providing IP functions and internet connections [3].

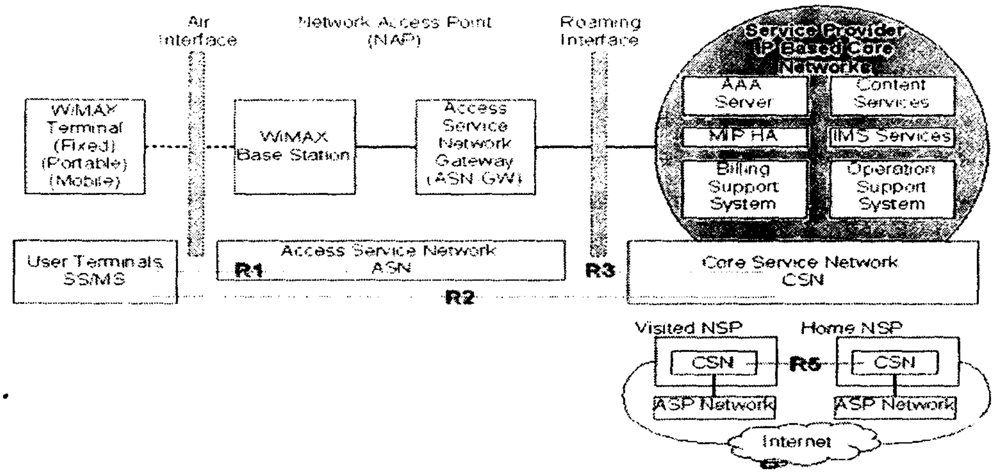


Figure 1.1: WiMAX network architecture.

In this architecture figure1.1, the Network Access Provider(NAP) is a business entity providing WiMAX radio resources to one or more WiMAX Network Service Providers and controls the Access Service Network (ASN)[3][6]. The Network Service Provider is possibly a different business entity that provides IP connectivity and WiMAX services to WiMAX subscribers and manages the Connectivity Service Network (CSN). Communication between the different elements of the network architecture is based on reference points, which form the foundation for seamless interoperability. For instance, reference point R1 describes the protocols and procedures between MS and ASN. The ASN concentrates on network access functionality, different profiles that define how the ASN-GW and the BSs are implemented in the ASN network [3]. Besides the IP connectivity assured by the CSN, IP address allocation, Internet access, billing operations and IP Multimedia Services (IMS) are also managed by the CSN.The mobility management considered by the WiMAX Forum for Mobile WiMAX supports IPv4 and IPv6 mobility management protocols as well as mechanisms for reducing packet loss and handover delay [20]. Mobility is divided into two types:

- ASN_anchored_ mobility.
- CSN_anchored _mobility.

ASN-anchored mobility, or micro-mobility, the MS moves its point of attachment between BSs of the same ASN so, there is no need for a MS Care-of-Address update while in CSN-anchored mobility, or macro-mobility there is consideration for IP mobility between ASN and CSN. Different types of Mobile IP (MIP) implementations are considered to support macro-mobility they are MIP-enabled clients and Proxy Mobile IP (PMIP). With the MIP-aware approach, the MS is compliant with MIPv4 if deployed in IPv4 networks or MIPv6 if deployed in IPv6 networks, respectively [3][6].

Two modes of operations are supported in WiMAX: the point to multi point (PMP) and mesh mode. SSs should be in line of sight with the BS in the PMP Mode. The link is single hop. But, in mesh mode SSs can communicate with the mesh BS or with each other through multihops routes via other SSs. The limitation of the mesh topology is that the interconnected devices are not able to access the web, send e-mail, or run remote applications [4].

1.3 WiMAX Infrastructure

WiMAX as we said before is standards based technology and it is the common name associated to the IEEE 802.16a/REVd/e standards. The IEEE 802.16 is a subgroup working on broadband wireless systems which specifies the air interface including the medium access control layer and multiple physical layer specifications that provide wireless technology 10 to 66 GHZ. WiMAX comes into two varieties. **Fixed WiMAX and mobile WiMAX**. Fixed **WiMAX or IEEE802.16** was first published in 2001 and had major changes since 2001.

These changes are:

- In January 2003, the IEEE approved 802.16a as an amendment to IEEE 802.16-2001, defining (Near) Line-Of-Sight capability.
- In July 2004, IEEE 802.16REVd, published under the name IEEE 802.16-2004. It consolidates IEEE 802.16-2001, IEEE 802.16c-2002, IEEE 802.16a-2003.introduces support for in (NLOS) through additional radio capabilities such as antenna beam forming and OFDM sub-channeling. It deals with stationary transmission with simple mobility [4].

Fixed WiMAX offers cost effective point to point and point to multipoint solutions. The second verity, **Mobile WiMAX**, is based on an IEEE 802.16e standard which takes the fixed WiMAX application a step farther by providing cell phone-like applications on a much larger scale. An IEEE 802.16e was published Early 2005. It deals with both stationary and full mobile transmission and Operates on frequencies below 11GHZ (2-11GHZ) with licensed and license-exempt frequencies [4].

WiMAX infrastructure has the potential to replace a number of existing telecommunications infrastructures like replacing the telephone company's copper wire networks by fixed configuration and the cellular networks by using its mobile variety.

1.4 WiMAX applications

WiMAX has the potential to provide widespread internet access and can be described as a framework for the evaluation of wireless "broadband" rather than a static implementation of the wireless technology. It provides a backhaul for cellular or Wi-Fi hotspots and high speed connectivity for business.

In corporate networks environment users can access high capacity data services along with VoIP soft phone and real time video wherever their location is and whether they are stationary or traveling. Also, they can download and upload large document along with the basic application access. They can participate in audio and video conferences. Other WiMAX applications include the security monitoring by using video cameras and other remote devices without the cost and complexity of installing cables connections. In addition to its usage in medical and other customized applications [5][7].

1.5 Comparison with Wi-Fi

- Both WiMAX and Wi-Fi are related to wireless connectivity and Internet access and both are products of IEEE802 committee.
- The Wi-Fi committee is 802.11 and was tasked with developing the standards for wireless LAN while the WiMAX committee is 802.16 and it is working on the wireless standards for metropolitan areas MAN.
- Wi-Fi transmission has a maximum rang of 100 meters while WiMAX is a long range system, blanket a radius of 30 mile or 50 km with wireless access .
- Wi-Fi uses unlicensed spectrum to provide access to a local network so it is subjected to interference while WiMAX uses licensed or unlicensed spectrum to deliver connection to a network, in most cases the Internet.
- WiMAX is not limited to dozen or so clients per access point as in Wi-Fi.

- Wi-Fi systems are not designed for high speed mobility. The new amendment of WiMAX supports the vehicle speed mobility.
- The fastest Wi-Fi connection can transmit up to 54 Mbps under optimal conditions while the WiMAX should be able to handle up to 70 Mbps.
- In Wi-Fi the MAC uses contention access while it uses the scheduling in WiMAX.
- Wi-Fi and WiMAX are complementary. WiMAX may be used in WMAN to connect Wi-Fi hot spots to the internet [4][8][9].

1.6 Comparison WiMAX with 3G

- Unlink 3G systems, which have a fixed channel bandwidth, WiMAX [26] defines a selectable channel bandwidth from 1.25 MHz to 20 MHz, which allows for a very flexible deployment.
- When deployed using the more likely 10 MHz TDD (time division duplexing) channel, 3:1 downlink-to-uplink split, and 2x2 MIMO, WiMAX offers 46 Mbps peak downlink throughput and 7 Mbps uplink[26].
- The application of OFDM modulation in WiMAX and Wi-Fi systems allows them to support very high peak rates. In addition, the OFDM physical layer used by WiMAX is more suitable to MIMO implementation than are CDMA systems from the standpoint of the required complexity over the gain. Therefore, compared to 3G, WiMAX offers higher peak rates, greater flexibility, and higher average throughput and system capacity [26].
- WiMAX could serve as 3G cellular backhaul.

1.7 The Advantages of WiMAX

- WiMAX can operate right next to cell phone towers with no interferences.
- It is a standard based technology that ensures the interoperability between multiple vendors' equipments.
- It provides an alternative wireless last mile broadband connectivity to homes, business and mobile wireless networks.

- It avoids the distance limitation of DSL and high costs of cables with high scalability, rapid deployment and lower maintenance and upgrade costs.
- It provides more protection by using licensed spectrum.
- It uses Orthogonal Frequency Division Modulation (OFDM) technology that has a lower power consumption rate and makes the network throughput performance for WiMAX to exceed that of 3G largely. [4]

1.8 Inherent Limitations

- WiMAX has the capability of reaching 30 Miles but real world testing has shown 4-8 mile working radius.
- "A recent city-wide deployment of WiMAX in Perth, Australia, has demonstrated that customers at the cell-edge with an indoor CPE typically obtain speeds of around 1–4 Mbit/s, with users closer to the cell tower obtaining speeds of up to 30 Mbit/s"[9].
- Still it is in fancy since the first piece of WiMAX-compatible equipment was just certified as just the start of 2006[4].
- There is a tradeoff between the data rates and the coverage distance like all wireless technologies.
- The available bandwidth is shared between users for a given radio sector so there is a need to ensure high performance in the case of many active users in the same time
- There is the important consideration of how to integrate end-to-end QoS between the WiMAX air link and the IP networks as their QoS protocol design strategies are not the same.
- The hard handover mode has a major disadvantage as it implies an abrupt transfer of connection from one Bs to another.

1.9 Motivation

This study is motivated due to the following reasons:

- The first and most important one is the growing interest in networks especially wireless networks to provide efficient solutions for connectivity.

- The second one and it is also important is the future of this technology. I expect by looking at the present that the future will be migrated into wireless technologies to achieve or get a totally connected world by wireless networks.
- The third is the high demand on multimedia services and application with mobility support.
- Finally, wireless technologies are less expensive than the wired ones so it can be utilized in good way to help the developing countries like Syria, my country, to have a very good network infrastructure.

1.10 Proposed work

In the current study, we propose to evaluate the performance of WiMAX based network for multimedia Applications using heterogeneous network of large number of mobile nodes. Further, the performance of video streaming from a MultiFace node while it moves between different networks is also evaluated.

1.11 Objectives

- Evaluate the performance for static networks with very little mobility support.
- Increase the mobility (full mobility).
- Understand the underlying technologies in the WiMAX network in terms of physical layer and MAC layer

1.12 Organizing the dissertation

Chapter 2 introduces the fundamental technologies of mobile WiMAX based on an amendment of the IEEE 802.16 standard (IEEE 802.16e). It introduces the technologies of WiMAX IEEE 802.16e. The features of the physical and MAC layers in WiMAX are also presented.

Chapter 3 introduces the need for the heterogeneous networks with the basic handover concepts for the cellular networks in the heterogeneous network. The specification of Multimedia application especially VoIP and video streaming are also presented.
Chapter 4 begins with the introduction of the tools used in this research and gives details about the design and implementation.
Chapter 5 presents the simulation results for the steaming Video performance in the Mobile heterogeneous WiMAX, Wi-Fi, UMTS and Ethernet based on the signal strength of handover between the three BSs.
Chapter 6 concludes the work presented in this dissertation along with suggestions to be investigated as extensions to this research.

CHAPTER 2

CHAPTER 2

MOBILE NETWORKS

2.1 Mobile WiMAX Radio Networks

Supporting mobility requires provisions for roaming and inter-cell handover and incorporating more flexibility into the standard to sustain multiple users demanding various types of services especially real time services which have strict latency tolerances and place varying demands on the system. The base station should allocate and schedule the system resources dynamically and on a frame-by-frame basis to keep up with the need of the users in the environment. WiMAX also uses a combination of adaptive modulation schemas and coding ranging from ½ rates QPSK to 5/6 rate 64QAM to approach the theoretical capacity of the system. There are many factors that determine the performance of WiMAX (throughput and range). These are:

- Frequency band it operates.
- Channel bandwidth.
- Duplexing scheme (TDD or FDD).
- Modulation (whether BPSK, QPSK, 16-QAM, or 64-QAM) and code rate.
- Antenna types.
- Whether LOS or NLOS.
- Transmit power.
- Receiver sensitivity.
- And the number of users per base station sector.

Some of the technique ,WiMAX offers and sre used to deal with above factors to improve the performance are as follows:

- Adaptive modulation and coding (AMC).
- Uplink sub-channelization (using OFDMA).
- Hybrid automatic repeat request (H-ARQ).
- Forward Error Correction (FEC)
- Smart antenna technologies (via optional AAS and MIMO features).

Further, the performance of the network can be enhancing by improving spectral efficiency or signal strength (SNR) [4] [10]

2.1.1 Mobile WiMAX physical layer

The original version of the standard specified a physical layer operating in the 10 to 66 GHz range. It provides a physical environment where the line of sight is required due to the short wavelength negligible multipath effect. The channels with 25 and 28 MHz bandwidth will be typical with 120 Mb/s raw data rate for point to multipoint (PMP) applications for small office/home office (SOHO),through the medium to large office. The modulation technique is single carrier modulation [4].

Further, the specifications for the 2 to 11 GHz range are added. It provides licensed and license-exempt frequencies. For the licensed one it provides physical environment where the line of sight is not necessary and the multipath can be significant due to the longer wavelength. It supports both near line of sight near (LOS) and none line of sight (NLOS) wireless transmission. Therefore, for providing this support it requires additional physical functionality as if advanced power management, interference migration/coexistence, and multiple antennas .The license-exempt provide similar physical environment with additional interference and co-existence issues [4]. It introduces three air interfaces. They are:

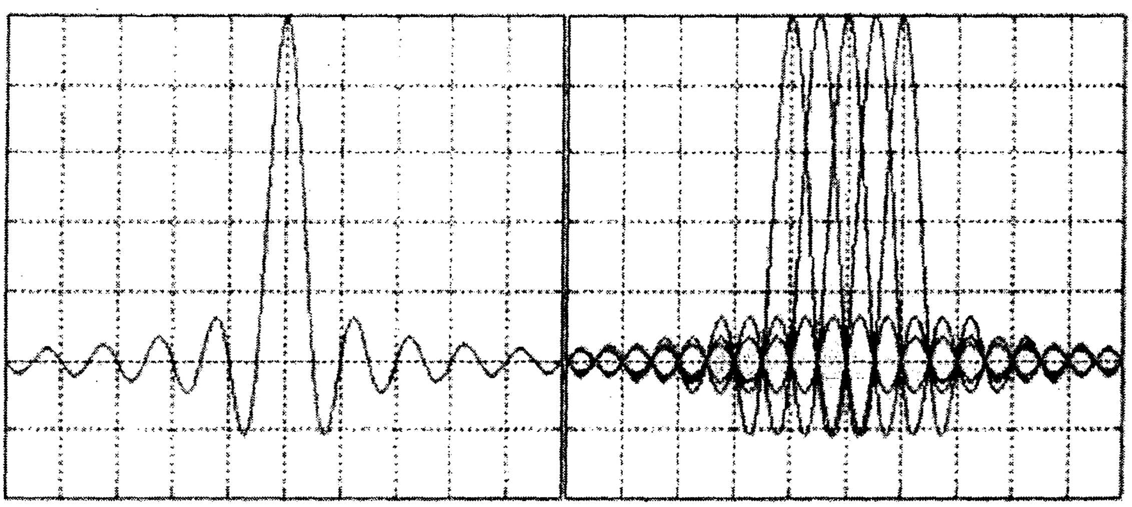
- WirelessMAN-SC2 uses a single-carrier modulation format.
- Wireless MAN-OFDM uses OFDM with a 256-point transform. Access is by TDMA .This air interface is mandatory for license-exempt bands It is also updated to use SOFDMA (scalable orthogonal frequency-division multiple

access) as opposed to the fixed orthogonal frequency-division multiplexing (OFDM) version with 256 sub-carriers [4].

- Wireless MAN-OFDMA uses OFDMA with a 2048-point transform. It addresses a subset of the multiple carriers to individual receivers to provide multiple accesses as well as supports advanced antenna systems. The OFDMA version creates the basis for the functionality of Mobile WiMAX and therefore, we discuss it in more details in this chapter.

2.1.1.1 Orthogonal Frequency Division Multiple Access Basics (OFDM).

Orthogonal frequency division multiplexing (OFDM) is a special case of multicarrier transmission. It transmits a single DataStream over a number of lower rate subcarriers. It can be seen as either a modulation technique or a multiplexing technique. In a normal frequency-division multiplex system, carries are spaced apart in such a way that the signals can be received by using conventional filters and demodulators. And in the receivers, there are guard bands between the different carriers in the frequency domain, which results in a lowering of spectrum efficiency . In an OFDM signal it is easy to arrange the carriers so that the sidebands of the individual carriers overlap and the signals are still received without adjacent carrier interference by using a mathematical relationship.



(a) (b)

Figure 2.1: Spectra of (a) an OFDM sub channel and (b) and OFDM signal

The receiver acts as a bank of demodulators, translating each carrier down to DC, with the resulting signal integrated over a symbol period to recover the raw data. The figure 2.1 shows subcarriers and how they are overlapped [11][4].one of the main reasons to use OFDM is to increase the robustness against frequency selective fading or narrowband interference. However, in terms of drawbacks, OFDM signal has a noise like amplitude with a very large dynamic range, therefore. Therefore requires RF power amplifiers with a high peak to average power ratio.

2.1.1.2 OFDMA Symbol Structure and Sub-Channelization

OFDMA splits the available spectrum into a number of parallel orthogonal narrowband subcarriers. These subcarriers are grouped into multiple sub-channels. Radio resources are thus available in terms of OFDM symbols (time domain) and sub-channels (frequency domain) [12].Three types of subcarriers are used in OFDMA symbols, data subcarrier that handle the transmission of data, pilot subcarrier that are for the estimation and synchronization use, and null subcarriers that have no transmission, but they are intended for guard bands and direct current (DC) carries as shown in figure 2.2 and figure 2.3. In OFDMA a symbol consists of several sub-channels and a sub-channel is a subset of data or pilot sub-carriers. For example, a 10 MHz channel is divided into 1024 subcarriers some of which are used for data transmission while others are reserved for monitoring the quality of the channel (pilot subcarriers), for providing safety zone (guard subcarriers)between the channels, or for using as a reference frequency(DC subcarrier)[11],[12].

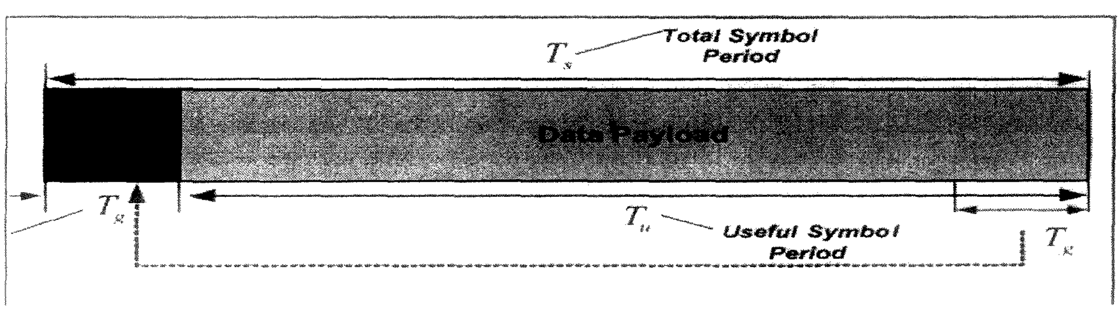


Figure 2.2: OFDMA Symbol period.

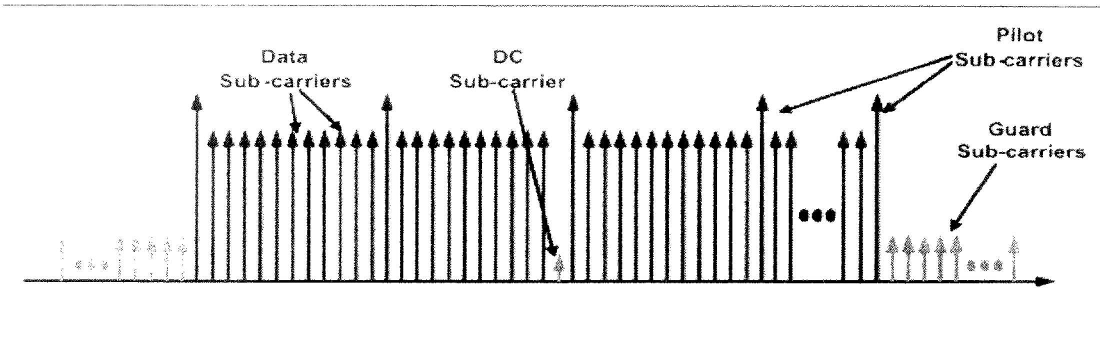


Figure 2.3: OFDMA sub-carrier structure.

2.1.1.3 Time Division Duplex Frame Structure.

In TDD, the UL subframe and DL subframe durations vary within the same-shared frame. The downlink subframe consists of one single PHY PDU while the uplink subframe consists of two contention intervals followed by multiple PHY PDUs, each transmitted by a different SS. The first contention interval is used for ranging which is the process of adjusting the Radio Frequency (RF). The second interval may be used by the SSs to request bandwidth since bandwidth since it is granted to SSs on demand. Two gaps separate the downlink and uplink subframes

- Transmit/receive transition gap (TTG).
- Receive/transmit transition gap (RTG).

These gaps prevent the DL and UL transmission collision [4][13][14]. The downlink PHY PDU consists of one or more bursts, each one is transmitted with a specific burst profile. A burst profile is a set of parameters describing the transmission properties (modulation type, forward error correction (FEC) type).The length of each burst is set by the BS. The UL bursts have the same functionality as the bursts in DL direction; hence, they provide the way to carry the different sized data from several users served by the BS. The DL-MAP, when sent, describes the location and profile of the other downlink bursts .The UL-MAP should be transmitted in each frame. It contains information elements (IE) that indicate the types and the boundaries of the uplink allocations directed to the SSs [4]. The OFDMA frame structure in TDD is shown in figure 2.4.

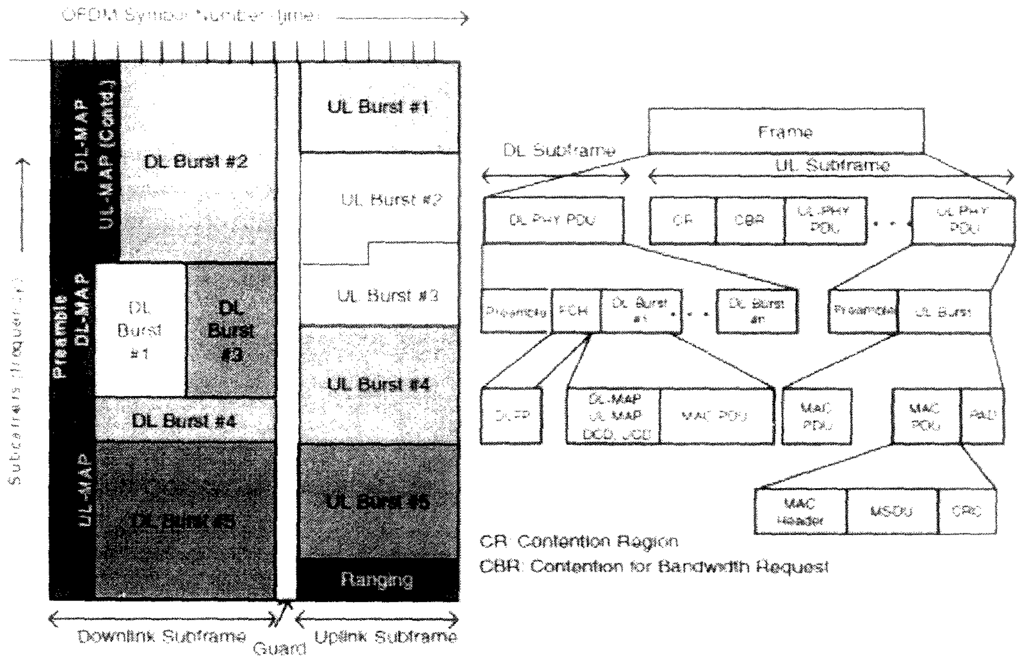


Figure 2.4: OFDMA Frame Structure in TDD

A slot is the smallest unit of resource in a frame, which occupies space both in time and frequency domain. A burst is a set of slots using the same MCS (Radio channel states).

2.1.2 Mobile WiMAX Medium Access Control (MAC) Layer

The MAC layer of WiMAX supports Point-to-Multipoint (PMP) and Mesh topologies and it is divided into following three sublayers

- CS (Convergence Sublayer), classifies the received PDUs from the higher layers into appropriate connections. Processes and delivers them to the Mac common-part sublayer,
- MAC CPS (Common-Part Sublayer) provides the fundamental MAC functionalities including connection establishment and management, generation of MAC management messages, SS initialization and registration, ranging, bandwidth management; service now management and scheduling services.
- Security Sublayer.

The WiMAX MAC uses a scheduling algorithm for which the subscriber station needs to compete only once for initial entry into the network. After network entry is allowed, the subscriber station is allocated an access slot by the base station. The time slot can enlarge and contract, but remains assigned to the subscriber station, which means that other subscribers cannot use it [4][6][14].

2.1.2.1 MAC Scheduling Services

The IEEE 802.16 MAC protocol is connection-oriented. The MAC layer schedules the usage of the air-link resources and provides Quality of Service (QoS). It has four types of service flows with distinct QoS requirement:

- Unsolicited Grant Services (UGS) is designed to support real-time flows that generate fixed-size data packet on a periodic basis, such as T1/E1 and VoIP without silence suppression. This service is tailored for carrying services that generate fixed units of data periodically. This eliminates the overhead and latency of bandwidth requests.
- Real-time polling service (rtPS) is designed to support real-time services that generate variable size data packets on a periodic basis, such as MPEG video. This service requires more request overhead than UGS, but supports variable grant sizes for data transport efficiency.
- Non-real-time polling service (nrtPS) is designed to support non-real-time and delay tolerant services that require variable size data grant burst types on a regular basis such as, FTP.
- Best effort (BE) service is designed to support data streams for which no minimum service level is required such as telnet and http. It neither provides throughput nor delay guarantee [4][16].

2.2 Advanced Techniques for WiMAX Networks.

One of the powerful features of WiMAX is its ability to use a particular digital modulation type that best meets the needs of each individual remote (mobile) unit. In case of location or propagation conditions for a low-loss RF path, higher orders of OFDM subcarrier modulation methods (16QAM, 64QAM) can be used. The choice of the modulation type to use is directly related to Carrier/Interference(C/I). The following table shows typical C/I ratio required for the basic modulation types used in WiMAX:

Modulation Type	Required C/I(dB)
BPSK	5
QPSK	8
16QAM	16
64QAM	22

Table 2.1: C/I ratios for basic modulation Types in WiMAX

Ideally, a remote unit should use as high an order of modulation type as possible consistent with a reliable connection. This enables a higher maximum data rate to be sustained by the mobile; it also aids in maximizing system capacity since the higher order types such as a 64QAM have a greater bits/Hz ratio making them more efficient. [12]

2.3. Universal Mobile Telecommunication System (UMTS)

Universal Mobile Telecommunication System (UMTS) represents a major 3G Wireless technology. It offers wireless Internet services at a worldwide scale, extending the scope of second-generation wireless networks from simple voice telephony to complex data applications including voice over IP, video conferencing over IP, web browsing, multimedia services, and high-speed data transfer. However, it is less suitable for small, indoor, and densely populated areas. A WLAN network provides high-speed data communication in restricted coverage areas at a relatively low cost. It allows users to move around in a connected area while they are still connected to the network. Therefore, WLAN could be utilized to extend 3G/UMTS services in such dense network. [25].

2.3.1 UMTS Architecture

A UMTS network consist of three interacting domains; Core Network (CN), UMTS Terrestrial Radio Access Network (UTRAN) and User Equipment (UE). The main function of the core network is to provide switching, routing and transit for user traffic. Core network also contains the databases and network management functions [25].

The basic Core Network architecture for UMTS is based on GSM network with GPRS. All equipment has to be modified for UMTS operation and services. The UTRAN provides the air interface access method for User Equipment. Base Station is referred as Node-B and control equipment for Node-B's is called Radio Network Controller (RNC).UMTS architecture is shown in figure 2.5.

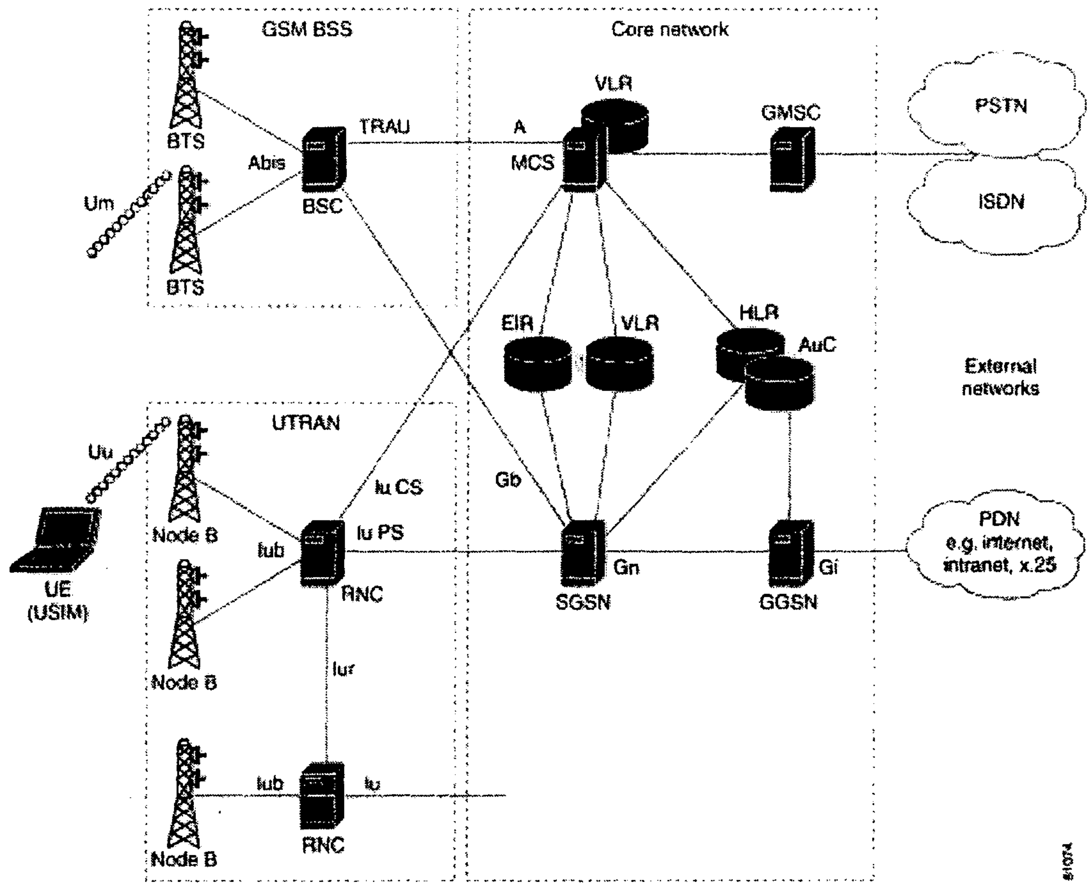


Figure 2.5: UMTS Architecture

2.3.2 Core Network

The Core Network is divided in circuit switched (CS) and packet switched (PS) domains. Some of the circuit switched elements are Mobile services Switching Centre (MSC), Visitor location register (VLR) and Gateway MSC. Packet switched elements are Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN). Some network elements, like EIR, HLR, VLR, and AUC are shared by both domains. [22]

The Asynchronous Transfer Mode (ATM) is defined for UMTS core transmission. ATM Adaptation Layer type 2 (AAL2) handles circuit switched connection and packet connection protocol AAL5 is designed for data delivery.

The architecture of the Core Network may change when new services and features are introduced. Number Portability Database (NPDB) will be used to enable user to change the network while keeping their old phone number. Gateway Location Register (GLR) may be used to optimize the subscriber handling between network boundaries. MSC, VLR and SGSN can merge to become a UMTS MSC [23].

2.3.3 Radio Access

Wide band CDMA technology was selected for UTRAN, air interface. UMTS WCDMA is a Direct Sequence CDMA system where user data is multiplied with quasi-random bits derived from WCDMA Spreading codes. In UMTS, in addition to channelization, Codes are used for synchronization and scrambling. WCDMA has two basic modes of operation: Frequency Division Duplex (FDD) and Time Division Duplex (TDD) [25].

The functions of Node-B are:

- Air interface Transmission / Reception
- Modulation / Demodulation
- CDMA Physical Channel coding
- Micro Diversity
- Error Handing
- Closed loop power control

The functions of RNC are:

- Radio Resource Control
- Admission Control
- Channel Allocation
- Power Control Settings
- Handover Control

- Macro Diversity
- Ciphering
- Segmentation / Reassembly
- Broadcast Signaling
- Open Loop Power Control

2.3.4 User Equipment

The UMTS standard does not restrict the functionality of the User Equipment in any way. Terminals work as an air interface counterpart for Node-B and have many different types of identities. Most of these UMTS identity types are taken directly from GSM specifications.

- International Mobile Subscriber Identity (IMSI)
- Temporary Mobile Subscriber Identity (TMSI)
- Packet Temporary Mobile Subscriber Identity (P-TMSI)
- Temporary Logical Link Identity (TLLI)
- Mobile station ISDN (MSISDN)
- International Mobile Station Equipment Identity (IMEI)
- International Mobile Station Equipment Identity and Software Number (IMEISV)

UMTS mobile station can operate in one of three modes of operation:

- **PS/CS mode of operation:** The MS is attached to both the PS domain and CS domain, and the MS is capable of simultaneously operating PS services and CS services.
- **PS mode of operation:** The MS is attached to the PS domain only and may only operate services of the PS domain. However, this does not prevent CS-like services to be offered over the PS domain (like VoIP)[23,25].
- **CS mode of operation:** The MS is attached to the CS domain only and may only operate services of the CS domain.

UMTS IC card has same physical characteristics as GSM SIM card. It has several functions:

- Support of one User Service Identity Module (USIM) application (optionally more than one)
- Support of one or more user profile on the USIM
- Update USIM specific information over the air
- Security functions
- User authentication
- Optional inclusion of payment methods
- Optional secure downloading of new applications

2.4 Wi-Fi

Wi-Fi connects you to your favorite content and communications over your mobile phone, computer, media players and other devices - all without cumbersome cables. When you're on the move, Wi-Fi lets you connect to the Internet or your office from an airport or coffee shop and helps you stay productive when you're away from home. Now, imagine doing all these things easily and quickly - without worrying about finding a wired network connection [24].

Wi-Fi is a globally used wireless networking technology that uses the 802.11 standard. The term Wi-Fi is an abbreviation of 'wireless fidelity'. The Institute of Electrical and Electronics Engineers (IEEE) developed the technology used in WiFi in 1997. The Wi-Fi Alliance, a trade group, later commercialized this technology.

The basic system of Wi-Fi is very simple. In a Wi-Fi enabled network, computers with Wi-Fi cards connect wirelessly to an access point or router. This access point or router is internet enabled the usual way, connected to the internet using a cable or DSL modem. Any PC or laptop with a Wi-Fi card, which is within a radius of 200 feet (60 meters) from the access point can access internet. However, a distance of 100 feet (30 meters) is considered ideal for good quality access. An area surrounding an access point, providing wireless access, is called a wireless hotspot.

2.4.1 The Technology

Wi-Fi networks use radio technologies called 802.11 to provide secure, reliable, fast wireless connectivity. A Wi-Fi network can be used to connect electronic devices to each other, to the Internet, and to wire networks which use Ethernet technology. Wi-Fi networks operate in the 2.4 and 5 GHz radio bands, with some products that contain both bands (dual band). They can provide real-world performance similar to basic wired networks.

There are four versions of Wi-Fi radios currently available- the ones that work with 802.11b, 802.11g , 802.11a and 802.11n standards. While the first two- 802.11b and 802.11g- transmit 2.4 GHz, the radios operating at 802.11a standard can transmit at 5GHz. IEEE 802.11n-2009 is an amendment to the IEEE 802.11-2007 wireless networking standard to improve network throughput over the two previous standards—802.11a and 802.11g— with a significant increase in the maximum raw data rate from 54 Mbit/s to 600 Mbit/s with the use of four spatial streams at a channel width of 40 MHz

Another important feature of the Wi-Fi radios that enables higher data rates is the coding techniques used. The 802.11a and 802.11g standard radios use Orthogonal Frequency Division Multiplexing (OFDM) technique while the 802.11b uses Complementary Code Keying (CCK) technique.

Due to the higher frequencies and the encoding techniques, WiFi radios can transmit a very high amount of data per second. The 802.11a and 802.11g standard radios transfer between 30-54 megabits per second and the 802.11b standard typically conveys 7-11 megabits per second.

The WiFi Alliance has developed certification standards for laptops and other electronic gadgets like PDAs. Most of these appliances are today Wi-Fi certified. This means that these gadgets are interoperable regardless of the brand.

2.4.2 Mobility and Reach ability in Wi-Fi

Wi-Fi networks have limited range. IEEE 802.11b or 802.11g wireless networks have a range of 32 m (120 ft) indoors and 95 m (300 ft) outdoors. IEEE 802.11n, however, can exceed that range by more than two times. Range also varies with frequency band. Wi-Fi in the 2.4 GHz frequency block has slightly better range than Wi-Fi in the 5 GHz frequency block. Outdoor ranges, through use of directional antennas, can be improved with antennas located several kilometers or more from their base.

Due to reach requirements for wireless LAN applications, Wi-Fi has high power consumption compared to some other standards. The high power consumption of Wi-Fi makes battery life in mobile devices a concern.

Due to the complex nature of radio propagation at typical Wi-Fi frequencies, particularly the effects of signal reflection off trees and buildings, algorithms can only approximately predict Wi-Fi signal strength for any given area in relation to a transmitter. This effect does not apply equally to long-range Wi-Fi, since longer links typically operate from towers that broadcast above the surrounding foliage.

Mobile use of Wi-Fi over wider ranges is limited, for instance, to use such as in an automobile moving from one hotspot to another. Other wireless technologies are more suitable as illustrated in the figure 2.6 given below.

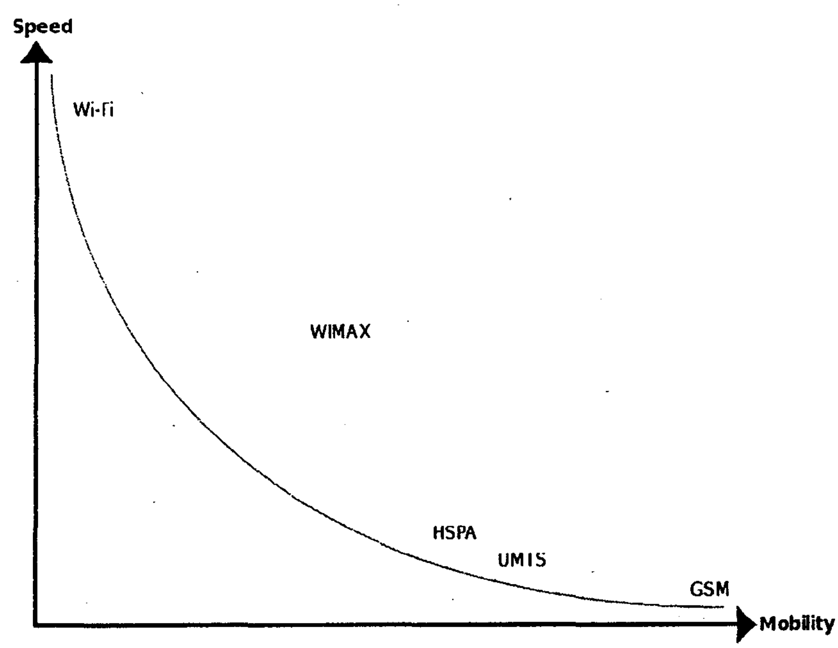


Figure 2.6: Speed vs. Mobility of wireless systems: Wi_Fi, HSPA, UMTS, and GSM

CHAPTER 3

CHAPTER 3

HANDOVER

3.1 Heterogeneous Networks

Networks today consist of different access and core network technologies. Anytime and anywhere connectivity is not achieved using a single technology. But using different networks connected together in order to form a network of networks, where users can roam seamlessly and securely across different infrastructures always utilizing the best resources available. An integration of these heterogeneous wireless networks is expected to be an effective means of providing high speed data access in wide radio coverage in the Next Generation (NG) wireless networks. When a mobile user moves across these networks, it has to perform handover to maintain its services. During a handover, it is pivotal to guarantee both service continuity and service quality, which ensure that handover can be made seamlessly. Providing seamless handover and ubiquitous services in heterogeneous wireless networks presents many new research challenges. The key challenges in order to achieve seamless mobility are how to achieve:

- Scalability: roaming is available between unlimited set of networks and operators.
- Standard handover interfaces: find interoperability between different vendor equipments.
- QoS guarantees during handover: during a handover, there is an important disruption to user traffic: significant latency, high signaling messages overhead and processing time, significant resources and routes setup delay, high handover failures and packet loss rate.

• Security: it is hard to maintain the same (if any) level of security when roaming across different access networks.[17]

3.2 Handover

Handoff is the process of changing the channel (frequency, time slot, spreading code, or combination of them) associated with the current connection while a call is in progress [18]. The handover is often initiated either by crossing a cell boundary or by deterioration in the quality of signal in the current channel and for implementing a mobile network, a handoff mechanism must be defined to maintain uninterrupted user communication session during his/her movement from one location to another. Differen techniques have been developed to addree the issue of hansoff. In general, they can be divided into following category [18]:

- Soft handoff and hard handoff according to channel usage.
- Micro cellular and multilayer handoffs.
- Vertical and horizontal.

3.2.1 Soft and hard handovers

- Soft handoff is characterized by "make before break" i.e. Both existing and new resources are released before new resources are used during the handoff process. It is used in Code Division Multiple Access (CDMA).
- Hard handoff is characterized by "break before make" i.e. current resources are released before new resources are used. It is more bandwidth-efficient than soft handoff, but it causes longer delay. A network-optimized hard handoff mechanism was developed for Mobile WiMAX to keep a handoff delay under 50 ms [18, 19].

3.2.2 Micro cellular and multilayer handoffs

- Microcellular Handoff :The microcells are cells with small radii and employed in highly populated areas such as city buildings and streets to meet high system capacity by frequency reuse it introduces two types of handoffs, Line Of Sight (LOS) and Non Line Of Sight

(NLOS), which have different characteristics. LOS handoffs try to minimize the number of unnecessary handoffs between BSs. NLOS must be as quickly as possible because of the corner effect which happen when the mobile station loss the LOS connection with its BS by turning the corner [18].

- Multilayer Handoff: In this a number of microcells are overlaid by Macrocells and the users are assigned to each layer according to their speed. Slow users are assigned to the microcells while the fast users are assigned to the Macrocells to reduce the handoffs requests. Macrocells also serve the slow users when the microcells are congested. When a microcell allocates all of its channels, the new and handoff calls are overflowed to the macrocell layer. When the microcells load decreases it is possible to assign slow users back to the microcell. This type of handoff is called take-back. There are four types of handoffs:
  - microcell-to-microcell
  - microcell-to-macrocell,
  - macrocell-to-macrocell,
  - macrocell-to-microcell

3.2.3 Horizontal vs. Vertical Handoff

Handoff between homogenous networks (where one type of network is considered) is called horizontal handoff while handoff between different types of networks is called vertical handoff. Vertical handover is a big challenge to support the multimedia applications with full mobility.

3.3 WiMAX handover scenarios

Fixed Access allows no movement. The user device is assumed to be fixed in a single geographic location for the duration of the network subscription. Nomadic access provides movement among the cells, but there is no handover support. It means that a user needs to establish a new network connection after each cell border is over run. IEEE 802.16e specifies handovers for portability, simple mobility, and full mobility of the users. Portability and simple mobility fall into a hard handover group. The moving speed is in the range of walking speed and low vehicular speed for portability and simple mobility, respectively. Full mobility comes under the group of a soft handover. Maximal supported speed matches high vehicular speed (about 160 km/h) [20]. In the IEEE 802.16e are defined three types of handover [20]: Hard handover, Macro Diversity Handover (MDHO) and Fast Base Station Switching (FBSS). Hard handover is mandatory in WiMAX systems. Other two types of handover are optional.

3.3.1 Hard Handover

During hard handover, the MS communicates with only just one BS in each time. Connection with the old BS is broken before the new connection is established. Handover is executed after the signal strength from neighbor's cell is exceeding the signal strength from the current cell. This situation is shown in Figure 3.1. The thick line at the boarder of the cells presents the place where the hard handover is realized [20].

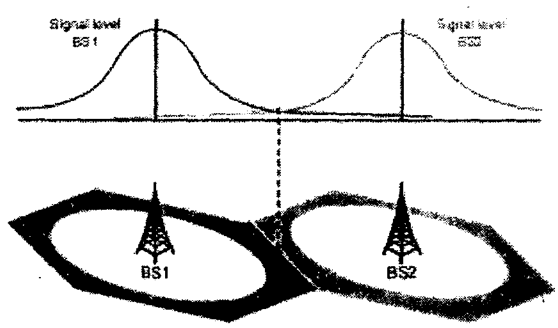


Figure 3.1: hard handover

3.3.2 Macro Diversity Handover

When MDHO is supported by Mobile Station (MS) and by Base Station (BS), the "Diversity Set" is maintained by MS and BS. Diversity set is a list of the BS's, which are involved in the handover procedure. It is defined for each of MS's in network. MS Communicates with all BS's in the diversity set (see Figure 3.2). For downlink in MDHO, two or more BS's transmit data to MS such that diversity combining can be performed at the MS. For uplink in MDHO, MS, transmission is received by multiple BS's where selection diversity of the received information is performed. The BS, which can receive communication from MS's and other BS's, but the level of signal strength is not sufficient is noted as "Neighbor BS" [20]

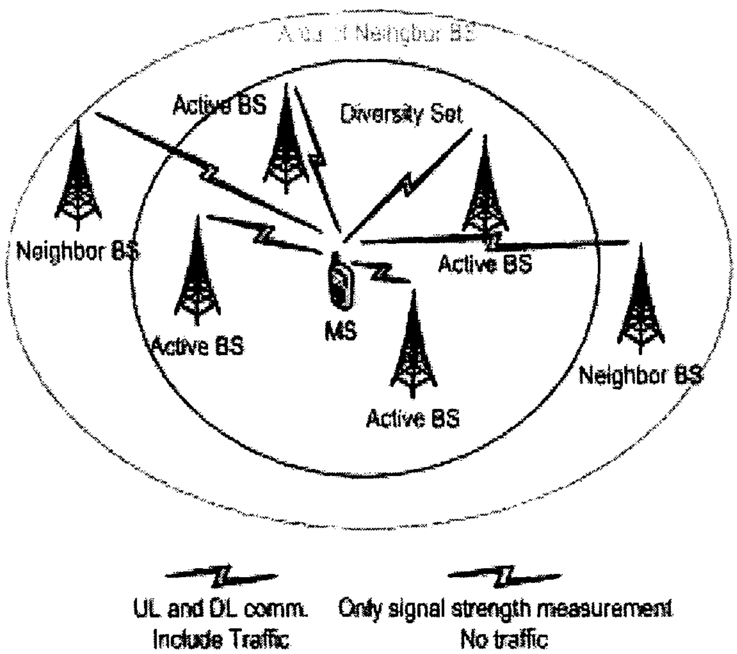


Figure 3.2 : Micro Diversity Handover

3.3.3 Fast Base Station Switching

In FBSS, the MS and BS diversity set is maintained similaly as in MDHO. MS continuously monitors the base stations in the diversity set and defines an "Anchor BS." Anchor BS is only one base station of the diversity set that MS communicates with for all uplink and downlink traffic including management messages (see Figure 3.3). This is the BS where MS is registered, and synchronized. It performs ranging and has a monitored downlink, channel for control information. The anchor BS can be changed from frame to frame depending on BS selection scheme. This means every frame can be sent via different BS in diversity set [20].

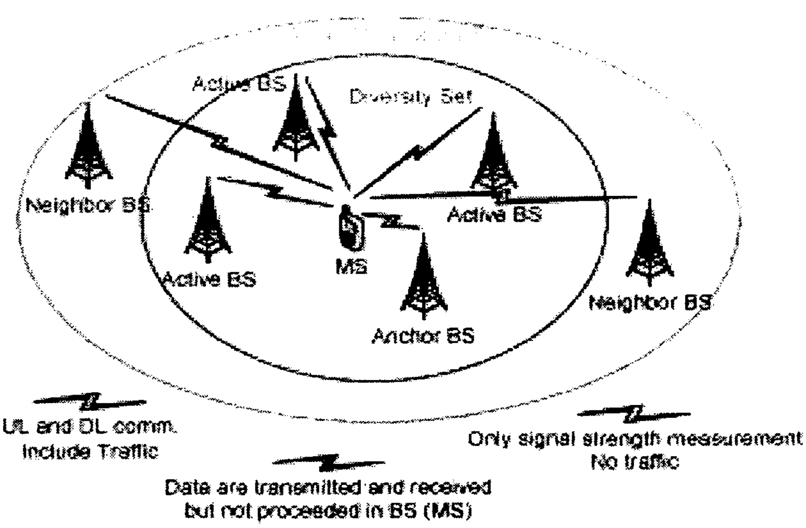


Figure 3.3 : Fast Base Station Selection

3.3.4 Handover according to mobility models

There are, as we have mentioned in the first chapter, two kinds of mobility in the mobile WiMAX network. One is ASN (Access Service Network)-anchored mobility and the other one is CSN (Connectivity Service Network)-anchored mobility [19][6]. According to these mobility , the handover scenarios supported in WiMAX is depicted in figure 3.4.

ASN-anchored mobility handover process results in comparatively low handover delay while the CSN-anchored mobility handover results in extra delay because of the need to change the IP address. The IEEE802.16e standard provides MAC layer L2 HO (handover). The basic IEEE802.16e handover operation is shown in Figure 3.5. It consists of two parts the first one is the network topology acquisition, and it can be divided into 3 sub procedures and the second is the actual handover process. The three sub procedures related to the network topology are:

- Network topology advertisement.
- MS scanning of neighbor BSs.
- Association procedure.

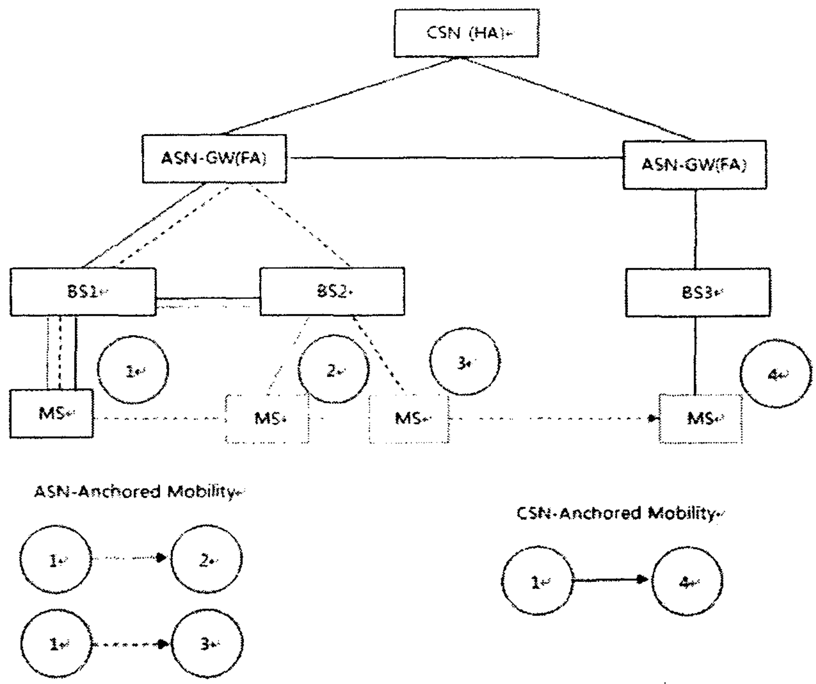


Figure 3.4 : Handover scenarios supported in WiMAX.

These procedures are executed through the backbone network. To prevent the interruption in the data transmission when a new connection is established the IEEE802.16e standard recommended that the neighboring BS scanning procedure be performed before the actual HO process.

After the acquisition process described above, several steps of the actual HO process are as following [19]:

- Cell re-selection: MS selects target BS according to neighbor BS information, such as signal strength and other QoS level parameters. This information comes from the periodic neighbor advertisement notification (MOB_NBR_ADV) or scanning requests.
- HO decision and initiation: MS can then make HO decision with MOB_MSHO_REQ or BS performs the same process with BS MOB_BSHO_REQ. According to this HO decision, an MS starts the handover procedure from a serving BS to a target one.

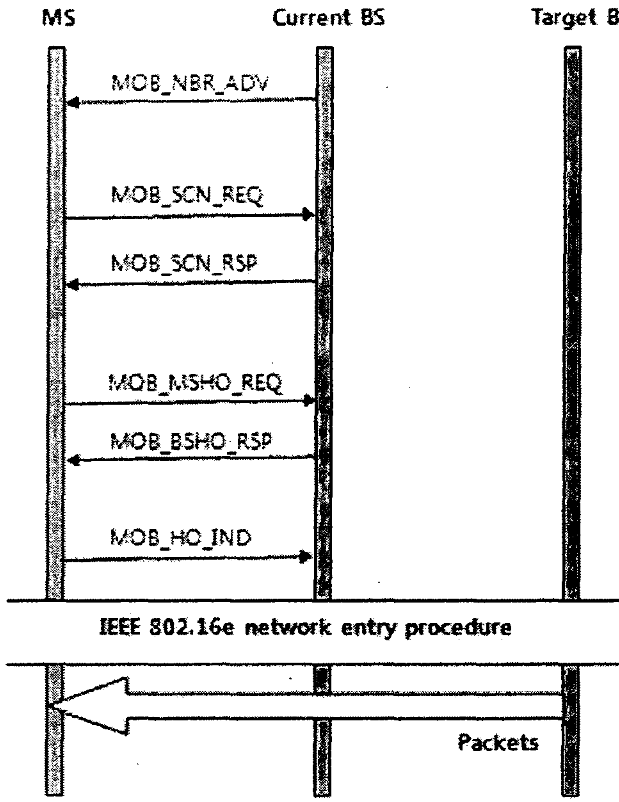


Figure 3.5 : The IEEE 802.16e handover procedure

- Synchronization with new down link and parameters acquisition: MS needs to be synchronized with target BS downlink and obtain UL and DL transmission parameters. If the previously sent MOB_NBR_ADV includes needed parameters for this job, this process can be shortened.
- Ranging and uplink parameter adjustment: MS and the target BS do initial or handover ranging.
- Ending with the serving BS: Serving BS terminates all contexts of the connections with the MS

3.4 UMTS Handover

There are following categories of handover for UMTS

3.4.1 Hard Handover

Hard handover means that all the old radio links in the User Equipment (UE) are removed before the new radio links are established. Hard handover can be seamless or non-seamless. Seamless hard handover means that the handover is not perceptible to the user. In practice a handover that requires a change of the carrier frequency (inter-frequency handover) is always performed as hard handover. Although from technical point of view inter-system handovers can be seen as a type of hard handovers[23] as depicted in the figure 3.6.

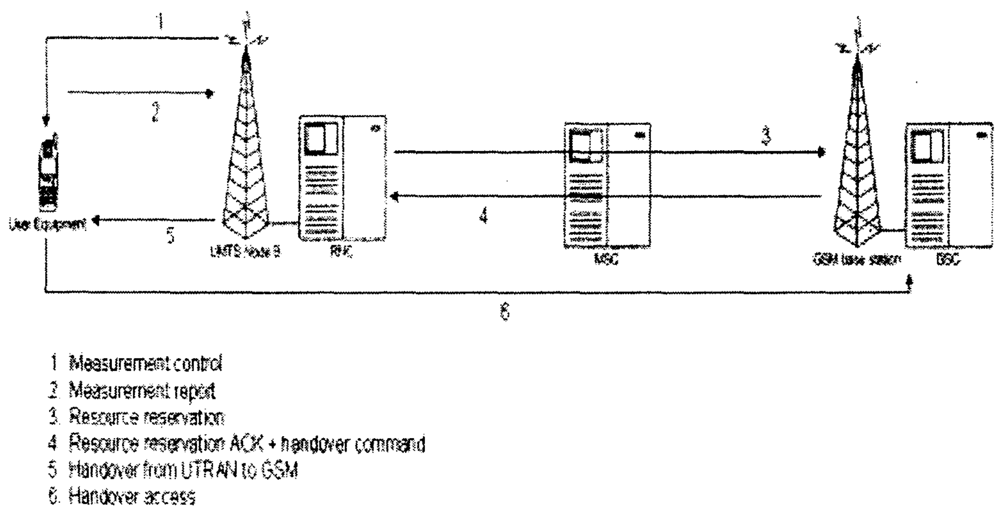


Figure 3.6: Inter-System Handover Procedure from UTRAN to GSM

3.4.2 Soft Handover

Soft handover means that the radio links are added and removed in a way that the UE always keeps at least one radio link to the UTRAN. Soft handover is performed by means of macro diversity, which refers to the condition that several radio links are active at the same time. Normally soft handover can be used when cells operated on the same frequency are changed [22, 23].

3.4.3 Softer handover

Softer handover is a special case of soft handover where the radio links that are added and removed belong to the same Node B (i.e. the site of co-located base stations from which several sector-cells are served). In softer handover, macro diversity with maximum ratio combining can be performed in the Node B, whereas generally in soft handover on the downlink, macro diversity with selection combining is applied [23]. Figure 3.7 shows the difference between the soft and softer handover.

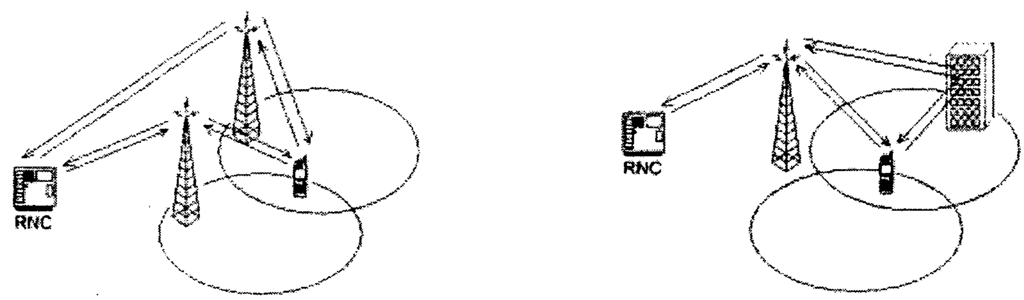


Figure 3.7 : soft vs. softer handover

Generally we can distinguish between intra-cell handover and inter-cell handover. For UMTS the following types of handover are specified:

- Handover 3G -3G (i.e. between UMTS and other 3G systems)
- FDD soft/softer handover
- FDD inter-frequency hard handover
- FDD/TDD handover (change of cell)
- TDD/FDD handover (change of cell)
- TDD/TDD handover
- Handover 3G - 2G (e.g. handover to GSM)
- Handover 2G - 3G (e.g. handover from GSM)

The most obvious cause for performing a handover is that a user moved to other cell can be served in the changed cell more efficiently (like less power emission, less interference).It may however also be performed for other reasons such as system load control[23].

- Active Set is defined as the set of Node-Bs the UE is simultaneously connected to (i.e., the UTRA cells currently assigning a downlink DPCH to the UE constitute the active set).
- Cells, which are not included in the active set, but are included in the CELL_INFO_LIST belong to the Monitored Set.
- Cells detected by the UE, which are neither in the CELL_INFO_LIST nor in the active set belong to the Detected Set. Reporting of measurements of the detected set is only applicable to intra-frequency measurements made by UEs in CELL_DCH state.

The different types of air interface measurements are:

- Intra-frequency measurements: measurements on downlink physical channels at the same frequency as the active set. A measurement object corresponds to one cell.
- Inter-frequency measurements: measurements on downlink physical channels at frequencies that differ from the frequency of the active set. A measurement object corresponds to one cell [23].
- Inter-RAT measurements: measurements on downlink physical channels belonging to another radio access technology than UTRAN, e.g. GSM. A measurement object corresponds to one cell.
- Traffic volume measurements: measurements on uplink traffic volume. A measurement object corresponds to one cell.
- Quality measurements: Measurements of downlink quality parameters e.g. downlink transport block error rate. A measurement object corresponds to one transport channel in case of Block Error Rate (BLER). A measurement object corresponds to one timeslot in case of Signal-to-Interference Ratio (SIR) (TDD only).
- UE-internal measurements: Measurements of UE transmission power and UE received signal level.

- UE positioning measurements: Measurements of UE position. The UE supports a number of measurements running in parallel. The UE also supports that each measurement is controlled and reported independently of every other measurement [22, 23].

3.5 Handover in Wi-Fi

Handover is facilitated by measuring radio parameters such as Received Signal strength (RSS), path loss, interference, and bit error rate (BER), separately or in combination. In Wi-Fi 802.11b networks a mobile terminal compares the difference between the RSS of two or more access points and initiates a link layer handover to the access point with the highest RSS .The handover process is depicted in the figure 3.8 showing the three main participants[24]:

- The supplicant mobile station (STA);
- The access point (AP);
- The authentication, authorization, and accounting (AAA) server.

The first phase, checks whether there is a need for changing the AP, In case there is a need it should be decided with which AP the STA be associated. This phase can last several seconds, but fortunately, most wireless LAN cards can do this without actually tearing down the connection with the currently used AP.

The next phase of the handover process contains an empty authentication step, the security architecture specified in the original 802.11 standard. This phase does not actually provide any security, and it takes a very short time.

The next phase is the association phase, wherein the STA establishes a logical connection to the AP. The most important task of this phase is to inform the wired network about the fact that the given STA can now be reached through the new AP. The time needed for the association is negligible.

The real authentication phase starts after the association phase. In this phase, the STA authenticates itself to the AAA server, which also helps to set up a shared session key between the STA and the AP. This phase can take a considerable amount of time, especially if the AAA server is remote [24].

Finally, the STA and the AP executes a four-way handshake, whereby they confirm the knowledge of the session key to each other, and they derive new keys from the shared session key for various purposes. The four-way handshake is a necessary in order to be compliant with the 802.11i standard. It cannot be shortened, but fortunately, it does not take too much time as it uses only local communication.

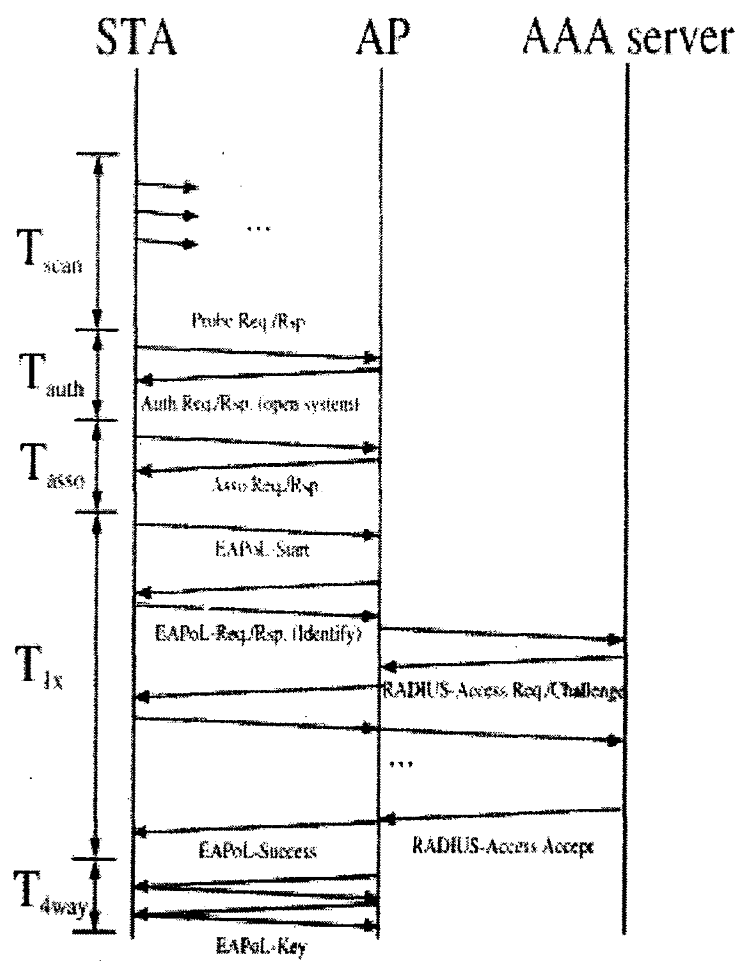


Figure 3.8: overview of the handover process in Wi-Fi networks.

3.6 Handover in heterogeneous networks

In the next generation of wireless networks, mobile users can move between heterogeneous networks, using terminals with multiple access interfaces and non-real-time or real-time services. The most important issue in such environment is the Always Best Connected (ABC) concept allowing the best connectivity to applications anywhere at any time. To answer ABC requirement, a successful handover of a terminal from one point of attachment to another should be achieved. A successful handover at least includes a new link establishment (including link-level (re)authentication), IP connectivity establishment (including network-layer (re)authentication), and new route establishment (when the handover leads to a subnet change). In addition, since a handover affects transport protocol performance, mechanisms to smoothen handover are of interest. These considerations become even more challenging when a terminal handovers across different access networks, such as from a Wireless LAN to CDMA or to GPRS. In order to provide a solution that take all the previous issues in consideration and provide enhancement to the network performance a standard called Media Independent Handover (MIH) was introduced.

3.7 Media Independent Handover

(MIH) is a standard being developed by IEEE 802.21 to enable the handover of IP sessions from layer 2 access technology to another, to achieve mobility of end user devices. The key functionality provided by MIH is communication among the various wireless layers and between them and the IP layer [6].

The IEEE 802.21 standard is a first step in order to allow mobile devices to successfully make handover (HO) between networks of different technologies, such as WiMAX, Wi-Fi, UMTS, , or Ethernet. It provides standardized interfaces between the access technologies and the mobility protocols from higher layers in the protocol stack. The heterogeneous environment is illustrated in figure 3.9. It shows multioperator, multi-technology network employing WiMAX, Wi-Fi and 3GPP UMTS/LTE. This includes the IEEE 802.21 Point of Attachment (PoA) and Point of Service (PoS) which is a MIH entity that communicates with the multimode terminal.

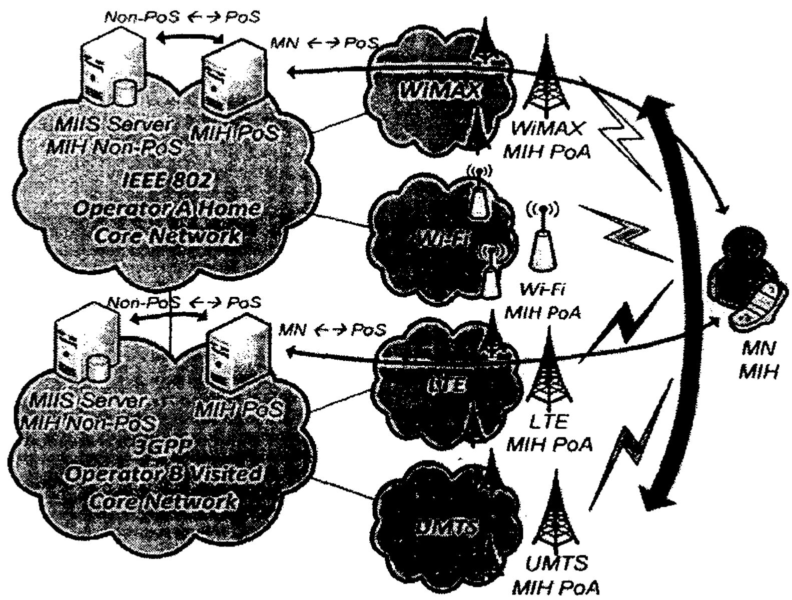


Figure 3.9: MIH in a Heterogeneous Access Network

IEEE 802.21 introduces a new entity called MIH Function (MIHF), which hides the specificities of different link layer technologies from the higher layer mobility entities. Several higher layer entities, known as MIH Users (MIHUs) can take advantage of the MIH framework, including mobility management protocols, such as FMIPv6 [27], Proxy Mobile IPv6 (PMIPv6) [28] and Session Initiation Protocol (SIP) [29], as well as the other mobility decision algorithms.

MIHF is located in the protocol stack between the layer 2 wireless technologies and IP at layer 3. The MIH function- MIHF- is implemented in

- mobile devices that have more than one wireless/wired interface;
- access points that have at least one wireless interface;
- Core network equipment that may have no wireless interface.

MIH platform provides three services to detect, prepare and execute the handovers that are

- Media Independent Event Service (MIES),
- Media Independent Command Service (MICS)
- Media Independent Information Service (MIIS).

MIES provides event reporting such as dynamic changes in link conditions, link status and link quality. Multiple higher layer entities may be interested in these events at the same time, so these events may need to be sent to multiple destinations.

MICS enables MIHUs to control the physical, data link, and logical link layer. The higher layers may utilize MICS to determine the status of links and/or control the access to different networks. Furthermore, MICS may also enable MIHUs to facilitate optional handover policies. Events and/or Commands can be sent to MIHUs within the same protocol stack (local) or to MIHUs located in a peer entity (remote).

MIIS provides a framework by which a MIHF located at the MS or at the network side may discover and obtain network information within a geographical area to facilitate handovers. The objective is to acquire a global view of all heterogeneous networks in the area in order to optimize seamless handovers while roaming across these networks [30].

3.8 Multimedia services

Multimedia can be as simple as a few images with some accompanying text to a multimedia presentation using video clips, sound, images animation and text. It uses lot of data when in a digital format. Multimedia applications, particularly those using video and images demand large bandwidths offered by network technologies like WiMAX and places new demands on the service that a network must provide. The most important of these are the bit error rate, the packet loss, delay and delay variation.

Using radio technologies for delivering real time multimedia material should be treated carefully, because radio links are susceptible to fading, interference, random delays. A real time multimedia connection is the most demanding multimedia access mode such as video on demand, video conferencing. Therefore, the adoption of multimedia requires a robust infrastructure that can support powerful multimedia applications, regardless of whether the endpoint is a traditional PC or a mobile device. In this diversity of multimedia applications, voice and video applications are the most popular and are expected to dominate the traffic mix in next generation wireless access networks. Therefore, we will go for more details about VoIP and video streaming applications.

3.8.1 VOIP

VOIP applications require assured bandwidth and specific bounds on delay and jitter (delay variation) that depend on the employed codec, which encodes human voice into samples and then it is transported over the network encapsulated in IP packets. ITU-T recommendations specify different codec including G.729, G.726, G.722, and G.711.

The choice of CODEC used is important because it determines the required bandwidth per call. G.711 [31][32] and G.729 [33] are the most widely used codec's in existing networks. However, G.729 CODEC is widely used in wireless networks. G729 has a data rate of 8 kbps and a 10-30 ms sample period (depending on implementation). Average bandwidth usage is ~40 kbps per call. The table 3.1 below describes the G.729 codec performances. The G.711 codec is currently used in a wide range of applications. Its voice sampling rate is 8 kHz and each sample is encoded with 8 bits resulting in a constant 64 kbps bit rate and offers very good voice quality. Samples can be packed into frames every 10 ms or another longer sampling rate. The G.729 codec can also generate speech frames every 10 ms or longer sample rate. Each frame contains 80 voice samples (collected at a sampling rate of 8000 samples per second), or another longer sampling rate. However, it requires a 5 ms look-ahead delay before producing any new frame. It is developed for simultaneous voice and data applications.

Codec	G.711	G.729
Sample Time (ms)	10	10
Frame Size (bits)	640	80
Packets Per Second	100	100
IP, RTP, UDP headers (bits)	320	320
IP-CS data rate (Kbps)	96	40
Data Rate Saving Efficiency with IEEE 802.16 Packet Header Suppression (PHS) (see Note-1) (see later section 4.4.9 for more details for PHS)	20.4%	40.2%
Data Rate Saving Efficiency with Robust Header Compression (RFC 3095[18]) (see Note-1) (see later section 4.4.9 for more details for ROHC)	26.7%	52.7%

Table 3.1 : G.711 vs. G.729 CODEC Performance

3.8.2 Video applications

Video traffic representative of Internet Protocol Television (IPTV) and other video-rich applications are typically encoded using MPEG-x codec's. Although marginally loss-tolerant, performance of these streams is inherently a function of available bandwidth, buffering, and delay characteristics of the underlying network. Internet Protocol Television (IPTV) technology [34], [35], [36] distributes video content over IP networks as both managed and unmanaged services. Managed services, such as IPTV and Video Conferencing, are typically provided by carriers that have provisioned the access network and therefore have control over the resulting quality of service (QoS) to their subscribers. While the unmanaged services like Skype and Google videos have little control over the end-to-end performance between the subscribers and the corresponding services. Video applications have different QoS requirements than VoIP and determined largely by:

- Data representation format (e.g. MPEG-4)
- Resolution
- Frame rate
- Compression rate

- Color spaces and stream type.

Video formats may range from 128 x 120 pixels (horizontal x vertical orientation) to beyond 1920 x 1080 pixels with various color depths. Common Internet video formats (YouTube) use 320 x 240 pixel resolutions while North American digital video disk (DVD) utilizes 720 x 480 and High Definition (HD) standards extend to 1920 x 1080 pixels. The higher the video frame resolution and/or pixel color depth, the larger the raw video content size. A video frame contains spatial (within) or temporal (between images) redundancy. Hence, various video coding schemes have evolved to reduce the raw video content size by exploiting this redundancy while balancing quality. These schemes include ITU H.26x and ISO MPEG-x codecs. Video frame inter-arrival rates range from 10 frames to 30 frames per second. four performance metrics with appropriate thresholds may be used to measure video streaming performance and these are:

- Packet loss _ number of packets dropped
  - → 1 - (# of received packets) / (# of expected packets)
  - → Avg: < 10-3; Ideal: < 10-5
- Delay: average time of transit
  - → Processing delay + propagation delay + queuing delay
  - → Avg: < 300 ms; Ideal: < 10 ms [38]
- Jitter _ variation in packet arrival time
  - → Actual reception time – expected reception time
  - → Avg: < 60 ms; Ideal: < 20 ms
- Throughput _minimum end-to-end transmission rate
  - → Measured in bytes/sec (or bps)
  - → 10 kbps – 5 Mbps [37]

WiMAX network is as any other type of networks, where users have to share the data capacity. But WiMAX's QoS features allow service providers to manage the traffic based on each subscriber's service agreement on a link-by-link basis like :

- Grant-request mechanism for letting users into the network. A grant-request mechanism allocates a small portion of each transmitted frame as a contention slot. After this, a subscriber station can enter the network by asking the BS to allocate a UL slot. The BS evaluates the subscriber station's request in the context of the subscriber's service level agreement (SLA) and allocates a slot in which the subscriber station can transmit (send UL packets).
- Link-by-link data-rate manageability: The data rate depends on the signal strength between the SS and the BS which depends on the distance between them .Therefore, the data rate is managed by using adaptive modulation techniques [6,4,16,19] .

CHAPTER 4

CHAPTER 4

LITERATURE REVIEW

&

TOOLS USED

Literature Review

4.1 Evaluation of Multimedia Services in Mobile WiMAX

A simulation was done by Bruno Sousa, Kostas Pentikousis and Marilia Curado to evaluate the performance of multimedia application over WiMAX. They discuss how the 802.21 standard can enhance the mobility management. This study shows that WiMAX support multimedia applications. Without using MIH information, the performance is poor and inflicted by high packet loss when handover accrues [6].

The simulation parameters used by this study are:

- Velocity
- MIH usage Information
- Confidence level of MIH predictive events.

The distance from the base station corresponds to the size of the cell, i.e. 2.8 km for urban macro cell and 1km for micro cell.

MIH information is employed in the following three modes:

- No events and no facilities of MIH exist.

- The links down Events are used to trigger the handover process.
- Predictive triggers link going down event is employed to start the handover process.

The predictive triggers have a confidence level associated with it. It is 60% and 80% confidence level to trigger the handover. Their scenario of simulation is composed of four WiMAX base stations as shown in the figure 4.1 below. The WiMAX network is connected to node of the core network. A mobile node (MS) is located at the beginning in the serving BS then it moves in rectangular way so it is connected to at least two base station till it reaches the final position.

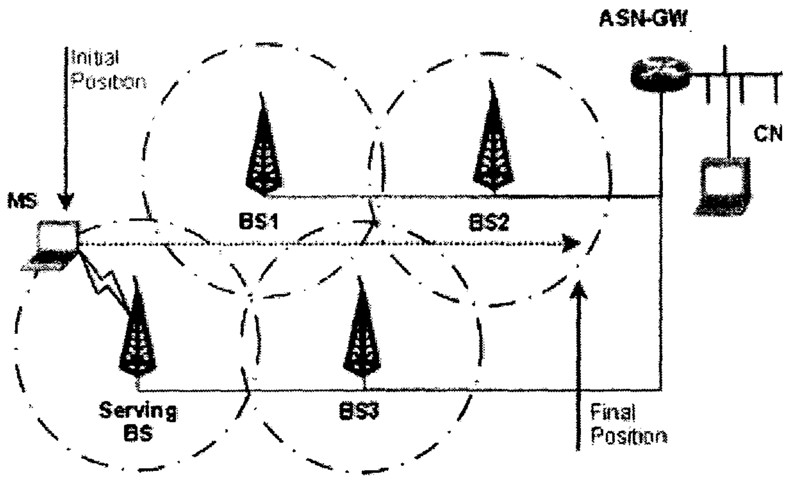


Figure 4.1 : Simulation scenario

4.1.1 VoIP

The results obtained by these authors are given in the table 4.1 below. It shows the packet loss for the G.729 codec for VoIP applications for the micro and macro cells with 30km/h and 120km/h velocity. In case of not using the MIH information, the packet loss is very high (91.29%, 30km/h, micro cell) compared to the 1.56 % in the same situation but with using LDown trigger from MIH

Velocity	Cell Size	MIH Case	Packet Loss (%)
30km	micro	NoEvents	91.29
		LDown	1.56
		LGDown 60%	43.23
		LGDown 80%	38.30
120km	micro	NoEvents	97.60
		LDown	4.62
		LGDown 60%	63.20
		LGDown 80%	34.71
30km	macro	NoEvents	88.08
		LDown	0.60
		LGDown 60%	79.57
		LGDown 80%	20.93
120km	macro	NoEvents	91.37
		LDown	2.66
		LGDown 60%	50.67
		LGDown 80%	56.08

Table 4.1: Packet Loss (G.729 codec)

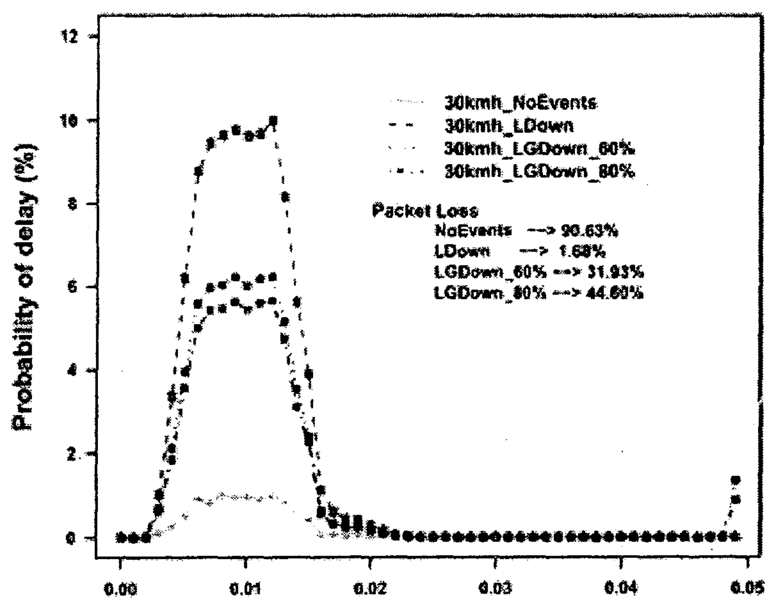
4.1.2. Video Streaming

The MOS classification for the urban macro- and micro-cell scenarios are depicted in Table 4.2 below. IT summarizes the objective quality of the video. As with voice, the test with MIH facilities and tests with higher velocities present the worst performance, mainly due to the packet loss which correlates with MOS. There is no excellent video quality (e.g. MOS=5), nonetheless the LinkDown cases present a good video quality (e.g. MOS=4), which decreases with higher vehicular speeds.

Velocity	Cell Size	MIH Case	MOS
30km	micro	NoEvents	1.63
		LDown	3.98
		LGDown 60%	3.06
		LGDown 80%	3.02
120km	micro	NoEvents	1.36
		LDown	3.38
		LGDown 60%	2.07
		LGDown 80%	2.02
30km	macro	NoEvents	2.30
		LDown	4.11
		LGDown 60%	3.54
		LGDown 80%	3.95
120km	macro	NoEvents	1.73
		LDown	3.57
		LGDown 60%	3.59
		LGDown 80%	3.50

Table 4.2 : MOS

One way delay in the Evalvid framework is based on the probabilistic distribution function (PDF), which determines the probability of a given delay.



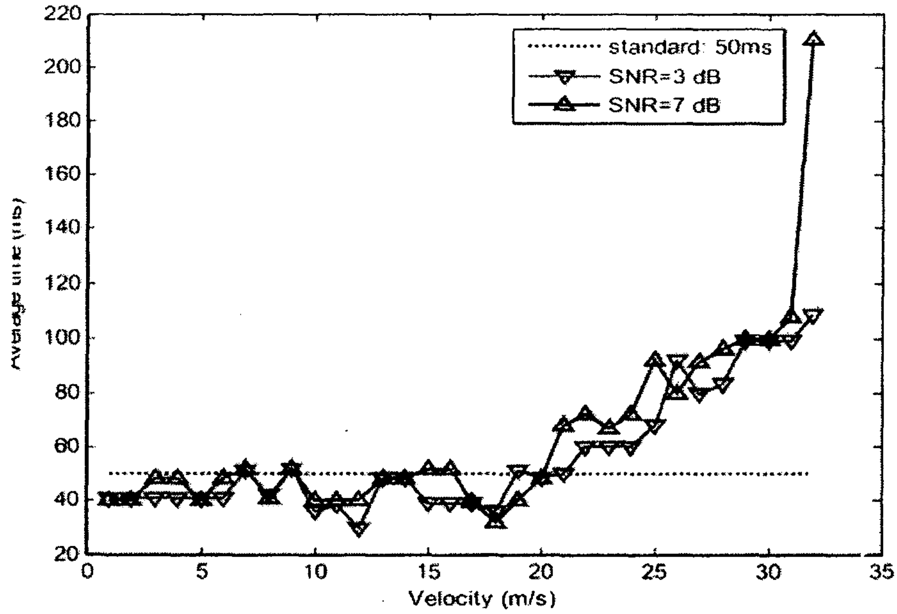
Disadvantages

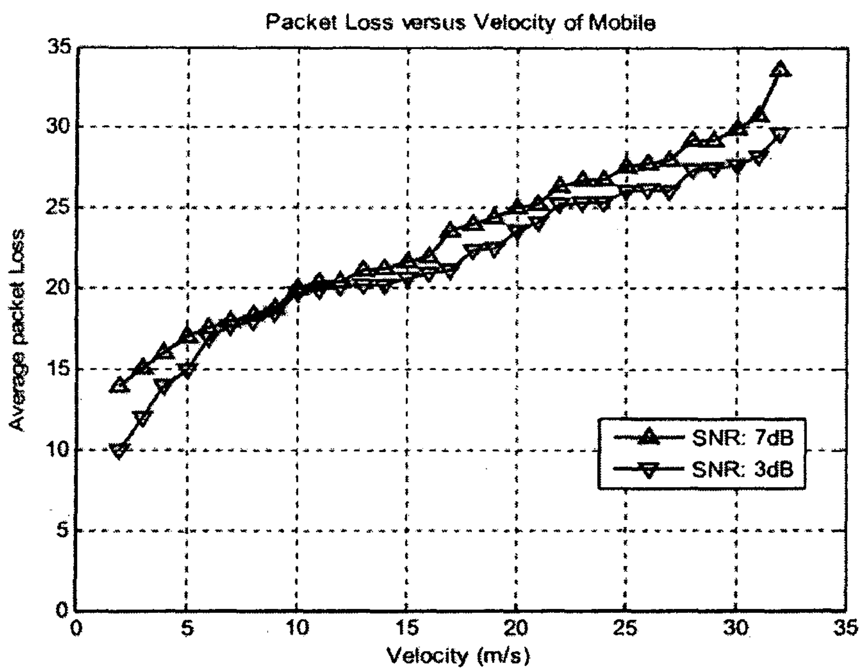
- Limited numbers of mobile nodes were considered in the study.
- No heterogeneity was considered since nodes were moving inside WiMAX cells only.

The authors has shown that the handover has high impact in the performance of multimedia applications, introducing high packet loss and intolerable delay. However, use of cross-layer information through MIH improves handover performance and significantly decreases the packet loss ratio.

4.2Handover Performance in the Mobile WiMAX Networks

YongxueYu in this paper has done an in-depth study of the handover effects of mobile WiMAX networks. The author simulated hard handover of the mobile WiMAX Network using Simulator-2 (NS-2). In this paper, "ping-pang" effect of handover has been investigated, call blocking and dropping probabilities by using MATLAB has been implemented in order to find out which parameters have the significant impact on the handover performance. It showed that the handover latency of mobile WiMAX is below 50 ms with the traveling speed of mobile station up to 20 m/s. The MS speeds between 1 and 32 m/s with 1 m/s step that for each speed. The start time for the MS is set up randomly and the simulation with corresponding speed runs 10 times. The handoff latencies first varied in the region of 40 ms, and stayed nicely below the 50 ms limit until the MS reached the velocity of 20 m/s, apart from few exceptions that exceeded the limit by only few milliseconds. After this, the times show a more or less steady growth up to 150 ms region with the 32 m/s MS speed. The average handover latencies are drawn as the velocity of 1-32 m/s also shows the handover latency with threshold level equal to 7 dB and the one with threshold level 3 dB. It seems that in order to maintain high SNR, the MS needs more time to scan channels.





4.3 Multi-client Video Streaming over Wireless MAN-OFDMA

This paper has presented an experimental performance evaluation of the Wireless MAN-OFDMA air interface when delivering multiple video streams to different clients. It use a testbed setup and empirical evaluations instead of simulations. The, study conducted has resorted to Evalvid. Experiments on the testbed were performed with videos which use different compression levels namely, CBR with 256kbps and 12kbps and VBR with peak data rate of 540kbps. The metrics used for performance evaluation delay, loss, and MOS. The combined evaluation through the delay, loss and MOS metrics has shown that by using the CBR-256kbps video, it is possible to guarantee a good quality for twice more concurrent streams than with both CBR-512kbps and VBR versions. Moreover, their results highlight the capability of WiMAX using the Wireless MAN-OFDMA air interface to support multi-user video streaming in limit conditions. For example, such as when the bandwidth requirements from the clients exceed the empirically measured maximum capacity of the BS. Overall, from the view of an ISP, a more conservative compression is able to satisfy a higher number of clients with a reasonable perceived quality, when the same capacity is available.

4.4 Simulation of 802.21 Handovers Using ns-2

In this paper authors has made an attempt to evaluate the reliability and scalability of the network simulator-2 (ns-2) [2] tool in simulating multiple vertical HOs under the scope of IEEE 802.21. Currently the 802.21 functionality can be incorporated in ns-2 by using external add-on modules developed by the National Institute of Standards and Technology (NIST). This has two main purposes:

- Evaluating 802.21 NIST add-on modules for ns-2 and
- Evaluating the reliability and scalability of ns-2 in simulating 802.21 scenarios.

The used model consisted of one WiMAX base station, three Wi-Fi access points and variable number of MNs. The variable number of MNs helped in measuring ns-2 scalability in simulating 802.21 scenarios. Figure 4.2 presents the network topology, and Table 4.3 presents values used for the most relevant variables in the paper.

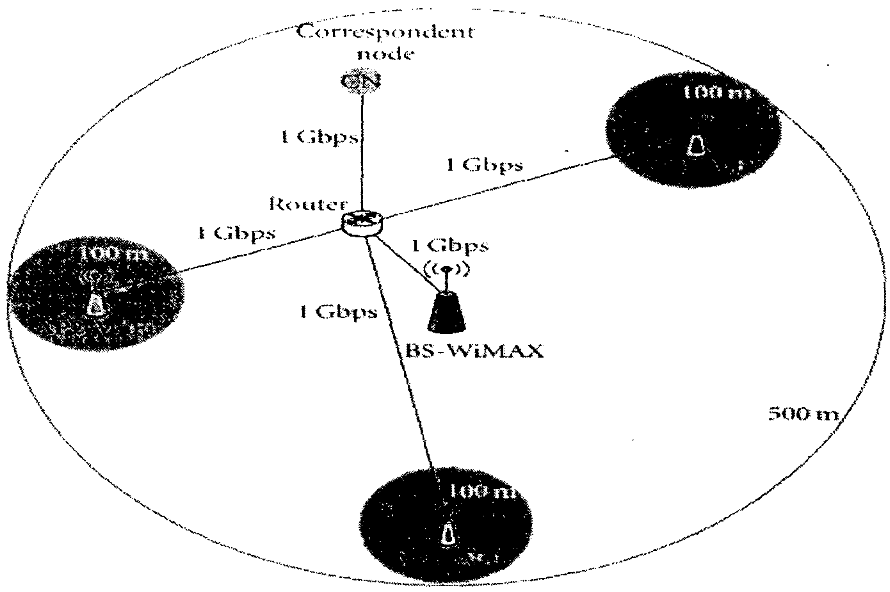


Figure 4.2 : Simulation scenario.

Number of mobile nodes	Between 1 and 100
Mobile movement	Rectilinear movement at 50 Km/h
Propagation channel model	Two-Ray Ground
Wired links	All wired links support 1 Gbps
Traffic parameters	UDP, constant bit rate at (Kbps) 50, 100, 200, 300, 500, and 1000.
WiMAX coverage	500 m
WiMAX parameters	Technology: 16QAM (10 Mbps) BS Tx power: 15 W (41 dBm) @ 3.5 GHz *RX Thresh_*: 1.215 e-9 W (~ −60 dBm) *CSTresh_*: Level 80% of *RX Thresh_*
Wi-Fi coverage	100 m
Wi-Fi parameters	Technology: 802.11 b (11 Mbps) AP Tx power: 100 mW (20 dBm) @ 2.417 GHz *RX Thresh*: 0,989 e-9 W (~ −60 dBm) *CSThresh_*: Level 90% of *RX Thresh_* *pr_limit_*: 1.2

Table 4.3 : Simulation parameters

Through their simulations, they have showed that multimedia recommendations can be achieved during handover by both WiMAX and Wi-Fi technologies, given that the number of associated MNs and traffic flow volume are in acceptable values. It also showed that ns-2 with NIST add-on modules proved to be a valuable tool to simulate IEEE 802.21 handover scenarios.

Tools used

4.5. Ns-2 simulator

Ns2 is an event driven, object oriented network simulator enabling the simulation of a variety of local and wide area networks. It implements different network protocols (TCP, UDP), traffic sources (FTP, web, CBR, Exponential on/off), queue management mechanisms (RED, DropTail), routing protocols (Dijkstra) . Ns2 is written in C++ and Otcl to separate the control and data path implementations. The simulator supports a class hierarchy in C++ (the compiled hierarchy) and a corresponding hierarchy within the Otcl interpreter (interpreted hierarchy).

The reason why ns2 uses two languages is that different tasks have different requirements: For example simulation of protocols requires efficient manipulation of bytes and packet headers making the run-time speed very important. On the other hand, in network studies where the aim is to vary some parameters and to quickly examine a number of scenarios the time to change the model and run it again is more important.

In ns2, C++ is used for detailed protocol implementation and in general for such cases where every packet of a flow has to be processed. For instance, if you want to implement a new queuing discipline, then C++ is the language of choice. Otcl, on the other hand, is suitable for configuration and setup. Otcl runs quite slowly, but it can be changed very quickly making the construction of simulations easier. In ns2, the compiled C++ objects can be made available to the Otcl interpreter. In this way, the ready-made C++ objects can be controlled from the OTcl level.

The ns-2 simulator is a discrete event network simulator developed by the University of Berkeley and the VINT project and has been contributed to by many authors [39]. Of significant note is the extensions made to ns2 by the Monarch Group from Carnegie Melon University[39]. The CMU extensions, which have since been included in the base ns simulator, enabled a more accurate simulation environment for multi-hop wireless networks. CMU included a model for the physical layer, support for MAC protocols and added ARP functionality to address some of the challenges of operating in a wireless environment.

The next section presents an MIH implementation for NS-2 developed for the Seamless Mobility project. It also provides an overview of the mobility extensions made to the standard release of NS-2.

4.6. MIH implementation

The modified version of ns-2.29 contains an implementation of MIHF based on the draft 3 of IEEE 802.21 specifications. It is a platform to evaluate the performances and find problems that could arise due to the definition of the primitives. It also serves to evaluate different handover decision engines.

4.6.1 Design overview

Figure 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.

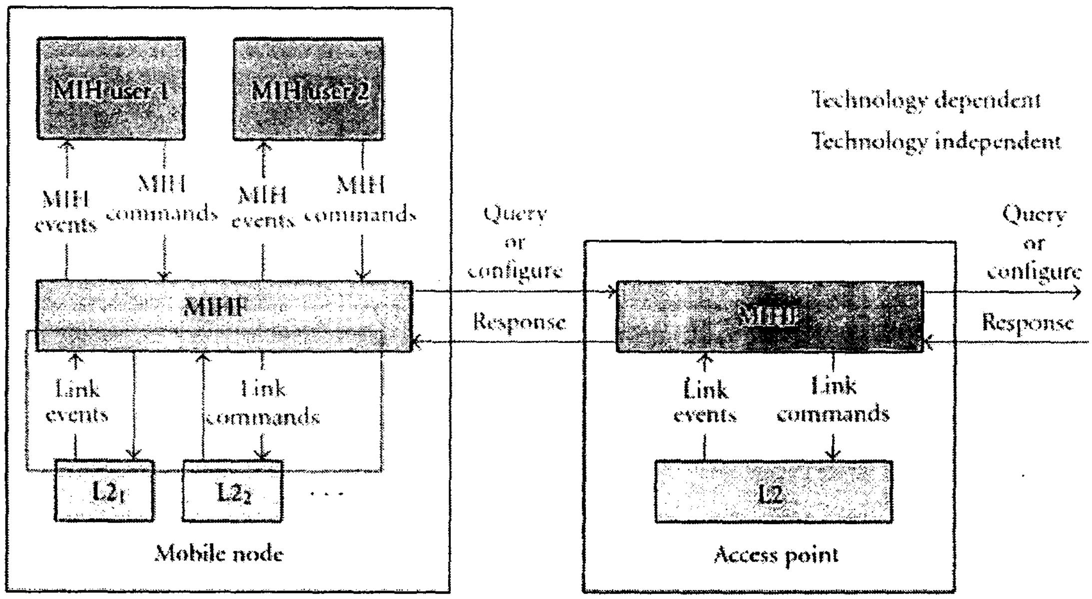


Figure 4.3 : MIH design overview

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

4.6.2. 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). The class handles the list of MIH Users and their registration information. It also takes care of handling the communication with remotes MIHFs. Finally, it provides the mapping between the media independent interface (MIH_SAP) and the media dependent interface (MIH_LINK_SAP and media specific primitives) [42].

4.6.3. 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.

4.6.4.MAC layer support for MIH

The MAC layers have been modified to include the MIH_LINK_SAP functions and to handle trigger generation.

4.7. Mobility extensions for NS-2

In addition to the MIH implementation, the mobility model includes enhancements to support handovers in NS-2. This includes:

- Integration of multiple technologies (UMTS, Bluetooth, 802.16) to allow for heterogeneous handovers.
- Modification in default implementation of 802.11 to support handovers.
- A generic design for nodes with multiple interfaces.
- Support for subnet discovery and change of address.

The difficulty encountered is that support for multiple interfaces is not intuitive in NS-2.To solve this difficulty a MultiFace node is defined.

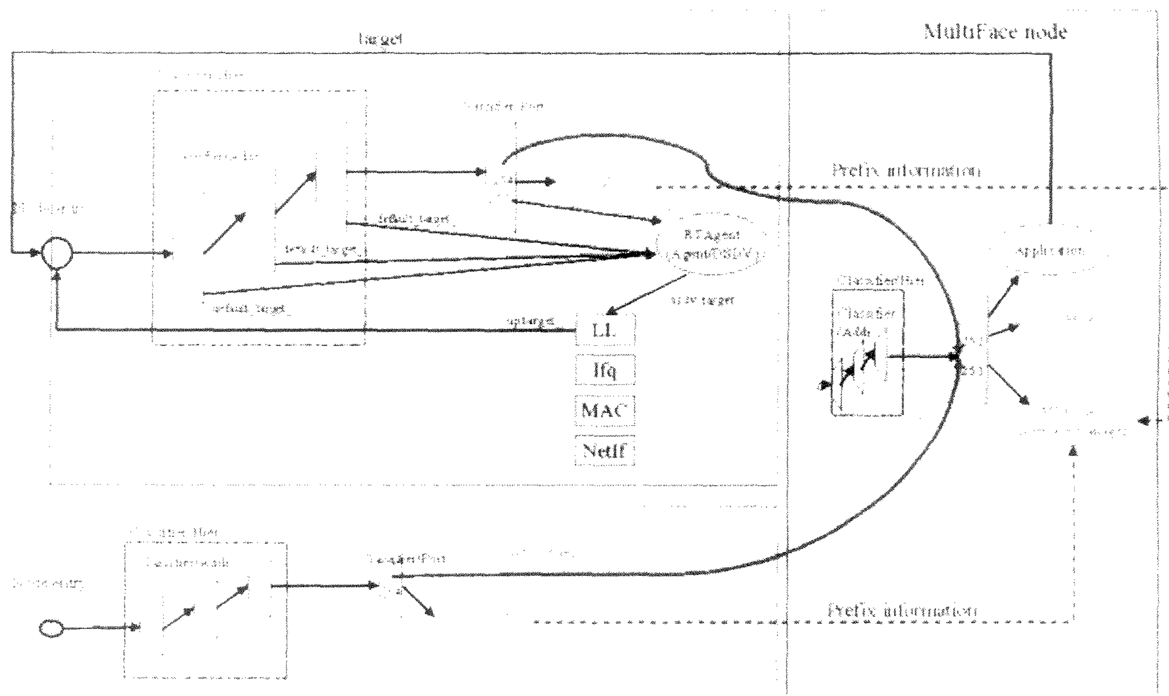


Figure 4.4: Multiple interfaces node design

MultiFace node is a virtual node linking nodes of similar or different technologies. The other nodes are considered interfaces for the MultiFace node. We define them as interface nodes. An ND agent located in each node allows for layer 3 movement detection (new and expired prefix) and notifications are sent to the interface manager (IFMNGMT). The MIH located in the MultiFace node is linked with each MAC object of the interface node. The application target_ object is dynamically assigned to the entry of the node chosen to send the traffic. It is shown in figure4.4 above[43].

4.8. MAC support of PHY

The model currently supports TDD. In this mode, uplink transmission occurs after downlink in each frame. The DL_MAP and UL_MAP messages sent every frame defines the burst allocation and transmission opportunities for each station. The information contained in the UL_MAP belongs to the same frame as shown in Figure 4.5.

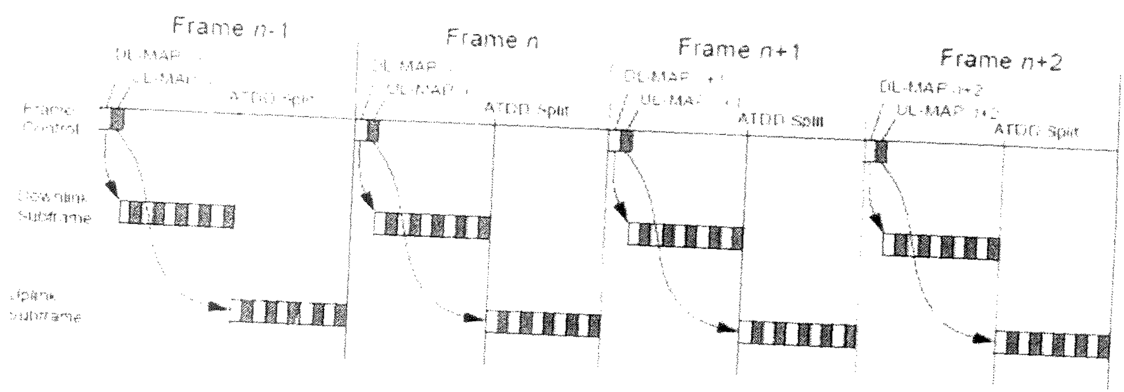
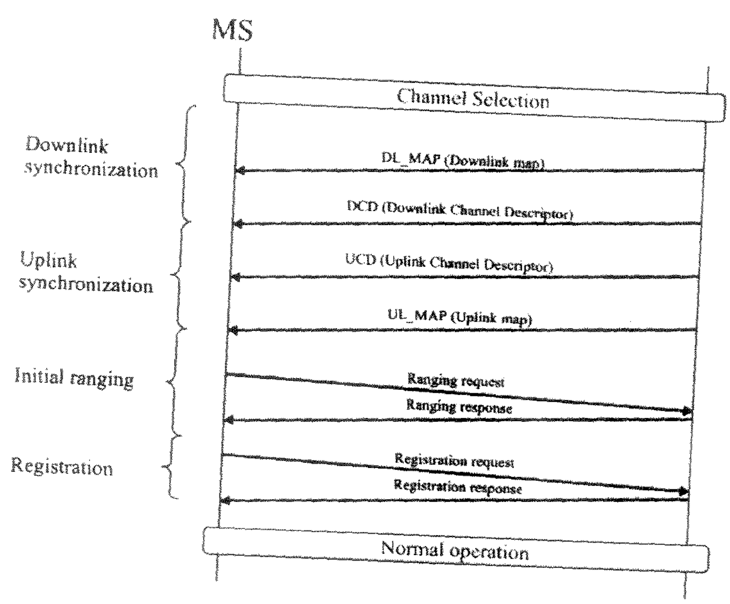


Figure4.5: Time relevance of DL_MAP and UL_MAP

4.9. Network entry and initialization

When an SS wants to join a network it needs to perform network entry. As shown in Figure 4.6 the Model implements the following components of the network entry:

- Scan downlink channel
- Obtain transmit parameters
- Initial ranging
- Registration



F gure 4.6: Network entry

The following parameters can be configured:

- Timers to perform channel scanning
- Frequency of the DCD/UCD messages
- Parameters for initial ranging (backoff window size and number of slots per frame)
- Channel allocation

Some aspects are IEEE 802.16e are implemented therefore network entry can be reduced if the SS has acquired the transmission parameters from the serving BS or during scanning.

4.10. MAC layer handover procedures

The model supports (layer 2) mobility. Depending on the configuration, the MS may perform Scanning and handover between BSs. This section presents the configuration parameters that affect the handover capability.

4.10.1. Scanning

When the link quality deteriorates, the MS can send a MOB-SCN_REQ to the serving BS to request scanning interval for the purpose of finding surrounding BSs. Figure 4.7 shows the messages sequence during scanning as implemented in the model.

To trigger the sending of a MOB-SCN_REQ, the MS monitors the signal level of the incoming Packets. When the level crosses a threshold, the message is sent. By default, the threshold is set to the RXThreshold therefore scanning is not used. To enable scanning, we change the lgd_factor_ attribute of the MIB to a value greater than 1.0. The higher the value, the sooner the scanning will trigger.

During scanning, the MS collects RSSI values of incoming packets. These values are reported to the serving BS that uses the information to select the best target BS. After the MS receives indication of the selected BS, it waits for a few frames before indicating its intention to perform handover. The introduction of the delay is to allow the traffic buffered during scanning to be exchanged before switching BSs.

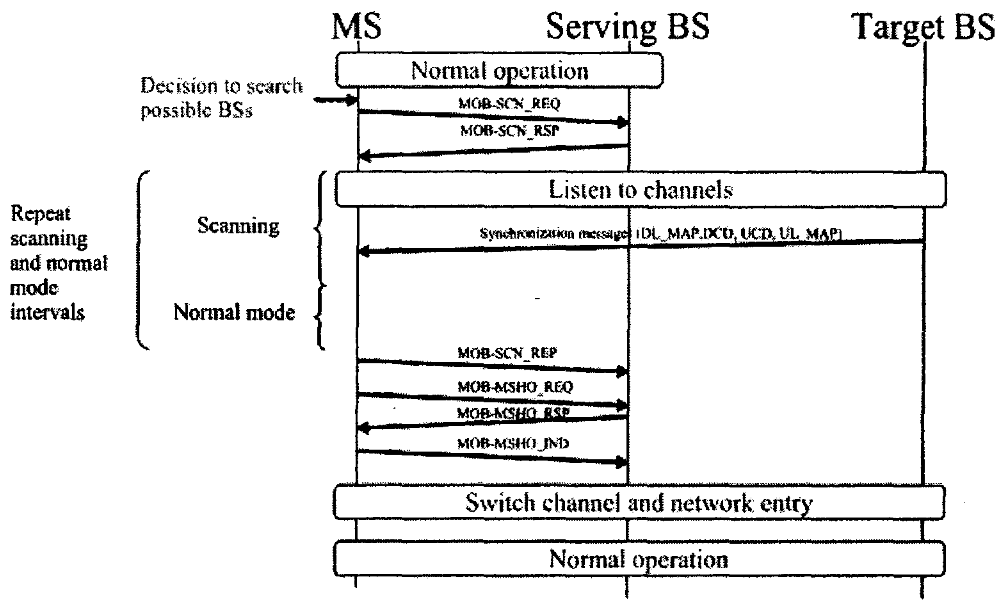


Figure 4.7: Scanning procedure

4.11. Overview of Evalvid

The structure of the Evalvid framework is shown as follows:

This section gives a description of the Evalvid framework which has been done by Jirka klaue, Berthold Rathke and Adam Wolisz [40]. In Figure 4.8 , a complete transmission of a digital video is symbolized from the recording at the source over the encoding, packetization, transmission over the network, jitter reduction by the play-out buffer, decoding and display for the user. Furthermore the points, where data are tapped from the transmission flow are marked. This information is stored in various files. These files are used to gather the desired results, e.g., loss rates, jitter, and video quality. A lot of information is required to calculate these values.

The required data are (from the sender side):

- Raw uncompressed video
- Encoded video
- Time-stamp and type of every packet sent

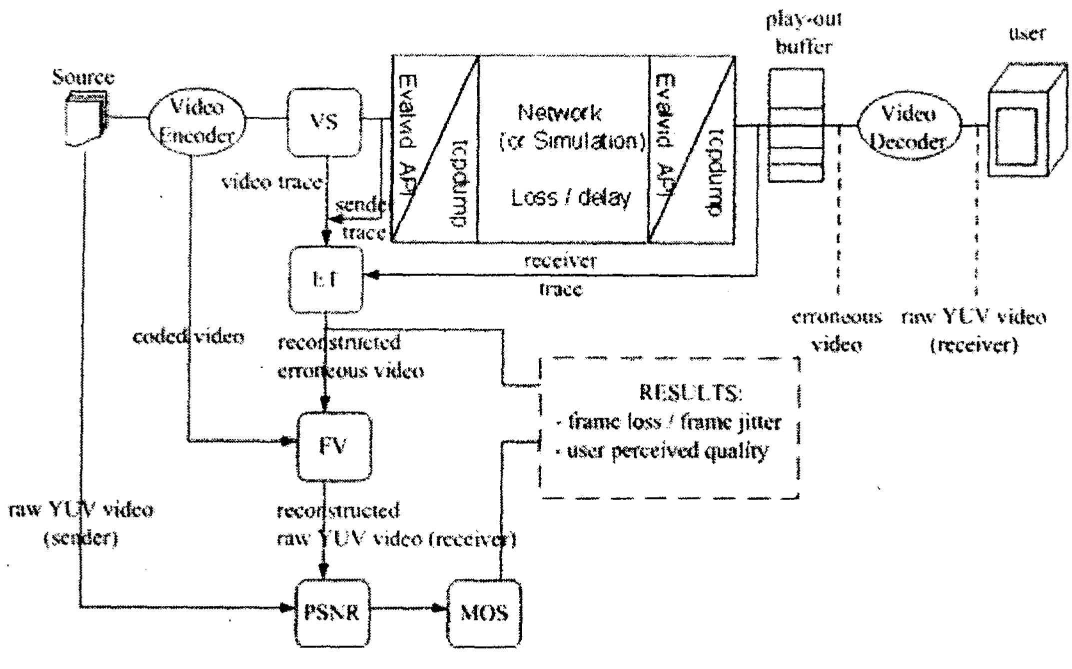


Figure 4.8 : Evalvid Framework

And from the receiver side:

- Time-stamp and type of every packet received
- Reassembled encoded video (possibly erroneous)
- Raw uncompressed video to be displayed

The evaluation of these data is done on the sender side, so the information from the receiver has to be transported back to the sender. Of practical concern is that the raw uncompressed video can be very large, for instance 680 MB for a 3 minute PDA-screen sized video. On the other hand it is possible to reconstruct the video to be displayed from the information available at the sender side. The only additional information required from the receiver side is the file containing the time stamps of every received packet. This is much more convenient than the transmission of the complete (erroneous and decoded) video files from the receiver side.

The processing of the data takes place in 3 stages. The first stage requires the timestamps from both sides and the packet types. The results of this stage are the frame-type based loss rates and the inter-packet times. Furthermore the erroneous video file from the receiver side is reconstructed using the original encoded video file and the packet loss information. This video can now be decoded yielding the raw video frames, which would be displayed, to the user. At this point a common problem of video quality evaluation comes up. Video quality metrics always require the comparison of the displayed (possibly distorted) frame with the corresponding original frame. In the case of completely lost frames, the required synchronization can't be kept up.

The second stage of the processing provides a solution to this problem. Based on the loss information, frame synchronization is recovered by inserting the last displayed frame for every lost frame. This makes further quality assessment possible. The thus fixed raw video file and the original raw video file are used in the last stage to obtain the video quality.

The boxes in Figure 1 named VS, ET, FV, PSNR and MOS are the programs of which the Framework actually consists are described as follows:

Source: The video source can be either in the YUV QCIF (176 x 144) or in the YUV CIF (352x 288) formats.

Video Encoder and Video Decoder: Currently, EvalVid supports two MPEG4 codecs, namely the NCTU codec and ffmpeg. In the present investigation, I arbitrarily choose the NCTU codec for video coding purposes.

VS (Video Sender): The VS component reads the compressed video file from the output of the video encoder, fragments each large video frame into smaller segments, and then transmits these segments via UDP packets over a real or simulated network. For each transmitted UDP packet, the framework records the timestamp, the packet id, and the packet payload size in the sender trace file with the aid of third-party tools, such as tcp-dump or win-dump, if the network is a real link. Nevertheless, if the network is simulated, the sender trace file is provided by the sender entity of the simulation. The VS component also generates a video trace file that contains information about every frame in the real video file. The video trace file and the sender trace file are later used for subsequent video quality evaluation.

ET (Evaluate Trace): Once the video transmission is over, the evaluation task begins. The evaluation takes place at the sender side. Therefore, the information about the timestamp, the packet id, and the packet payload size available at the receiver has to be transported back to the sender. Based on the original encoded video file, the video trace file, the sender trace file, and the receiver trace file, the ET component creates a frame/packet loss and frame/packet jitter report and generates a reconstructed video file, which corresponds to the possibly corrupted video found at the receiver side as it would be reproduced to an end user. In principle, the generation of the possibly corrupted video can be regarded as a process of copying the original video trace file frame by frame, omitting frames indicated as lost or corrupted at the receiver side. Nevertheless, the generation of the possibly corrupted video is trickier than this and the process is further explained in more details later. Furthermore, the current version of the ET component implements the cumulative inter-frame jitter algorithm for play-out buffer. If a frame arrives later than its defined playback time, the frame is counted as a lost frame. This is an optional function. The size of the play-out buffer must also be set, otherwise it is assumed to be of infinite size.

FV (Fix Video): Digital video quality assessment is performed frame by frame. Therefore, the total number of video frames at the receiver side, including the erroneous ones, must be the same as that of the original video at the sender side. If the codec cannot handle missing frames, the FV component is used to tackle this problem by inserting the last successfully decoded frame in the place of each lost frame as an error concealment technique.

PSNR (Peak Signal Noise Ratio): PSNR is one of the most widespread objective metrics to assess the application-level QoS of video transmissions. The following equation shows the definition of the PSNR between the luminance component Y of source image S and destination image D:

PSNR(n)dB = 20 log10

where Vpeak = 2k-1 and k = number of bits per pixel (luminance component). PSNR measures the error between a reconstructed image and the original one. Prior to transmission, one may then compute a reference PSNR value sequence on the reconstruction of the encoded video as compared to the original raw video. After transmission, the PSNR is computed at the receiver for the reconstructed video of the possibly corrupted video sequence received. The individual PSNR values at the source or receiver do not mean much, but the difference between the quality of the encoded video at the source and the received one can be used as an objective QoS metric to assess the transmission impact on video quality at the application level.

MOS (Mean Opinion Score): MOS is a subjective metric to measure digital video quality at the application level. This metric of the human quality impression is usually given on a scale that ranges from 1 (worst) to 5 (best). In this framework, the PSNR of every single frame can be approximated to the MOS scale using the mapping shown in table 4.4 as follows.

PSNR[dB]	MOS
>37 5	(Excellent)
31-37	4 (Good)
25-31	3 (Fair)
20-25	2 (Poor)
<20	1 (Bad)

Table 4.4 . : Possible PSNR to MOS conversion

The following figure 4.9 shows the evaluation process in ns-2 with Evalvid framework.

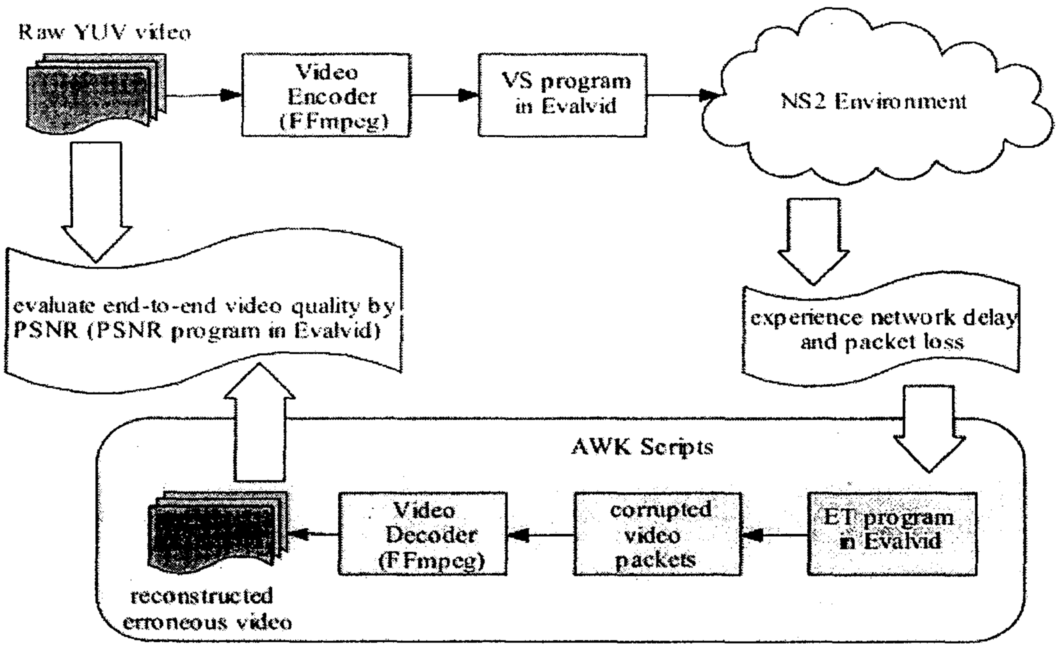


Figure 4.9 : QoS Assessment Framework including Evalvid

4.12. New network simulation agents

Three connecting simulation agents, namely MyTrafficTrace, MyUDP, and MyUDPSink, are implemented between NS2 and EvalVid. These interfaces are designed either to read the video trace file or to generate the data required to evaluate the video delivered quality.

MyTrafficTrace

The MyTrafficTrace agent is employed to extract the frame type and the frame size of the video trace file generated from the output of the VS component of EvalVid. Furthermore, this agent fragments the video frames into smaller segments and sends these segments to the lower UDP layer at the appropriate time according to the user settings specified in the simulation script file.

MyUDP

Essentially, MyUDP is an extension of the UDP agent. This new agent allows users to specify the output file name of the sender trace file and it records the timestamp of each transmitted packet, the packet id, and the packet payload size. The task of the MyUDP agent corresponds to the task that tools such as tcp-dump or win-dump do in a real network environment[40].

MyUDPSink

MyUDPSink is the receiving agent for the fragmented video frame packets sent by MyUDP. This agent also records the timestamp, packet ID, and payload size of each received packet in the user specified file[40].

4.13. Determination of Packet and Frame Loss

$$\text{packet loss} \quad PL_T = 100 \cdot \frac{n_{Trecv}}{n_{Tsent}}$$

where:

T : Type of data in packet (one of all, header, I, P, B, S)

n_{Tsent} : number of type T packets sent

n_{Trecv} : number of type T packets received

$$\text{frame loss} \quad FL_T = 100 \cdot \frac{n_{Trecv}}{n_{Tsent}} \quad \text{where:}$$

T : Type of data in packet (one of all, header, I, P, B, S)

n_{Tsent} : number of type T frames sent

n_{Trecv} : number of type T frames received

4.14 Determination of Delay and Jitter

The formal definition of jitter as used in this chapter is given by Equation 3, 4 and 5. It is the variance of the inter-packet or inter-frame time. The "frame time" is determined by the time at which the last segment of a segmented frame is received.

inter-packet time $it_{P_0} = 0$

$$it_{P_n} = t_{P_n} - t_{P_{n-1}}$$

where: t_{P_n} : time stamp of packet number n

inter-frame time: $it_{F_0} = 0$

$$it_{F_m} = t_{F_m} - t_{F_{m-1}}$$

where: t_F : time stamp of last segment of frame number m

$$\text{packet jitter } j_P = \frac{1}{N}\sum_{i=1}^{N}\left(it_i - it_N\right)^2$$

N : number of packets

it_N : average of inter-packet times

$$\text{frame jitter } j_F = \frac{1}{M}\sum_{i=1}^{M}\left(it_i - it_M\right)^2$$

M : number of packets

it_M : average of inter-packet times

CHAPTER 5

CHAPTER 5

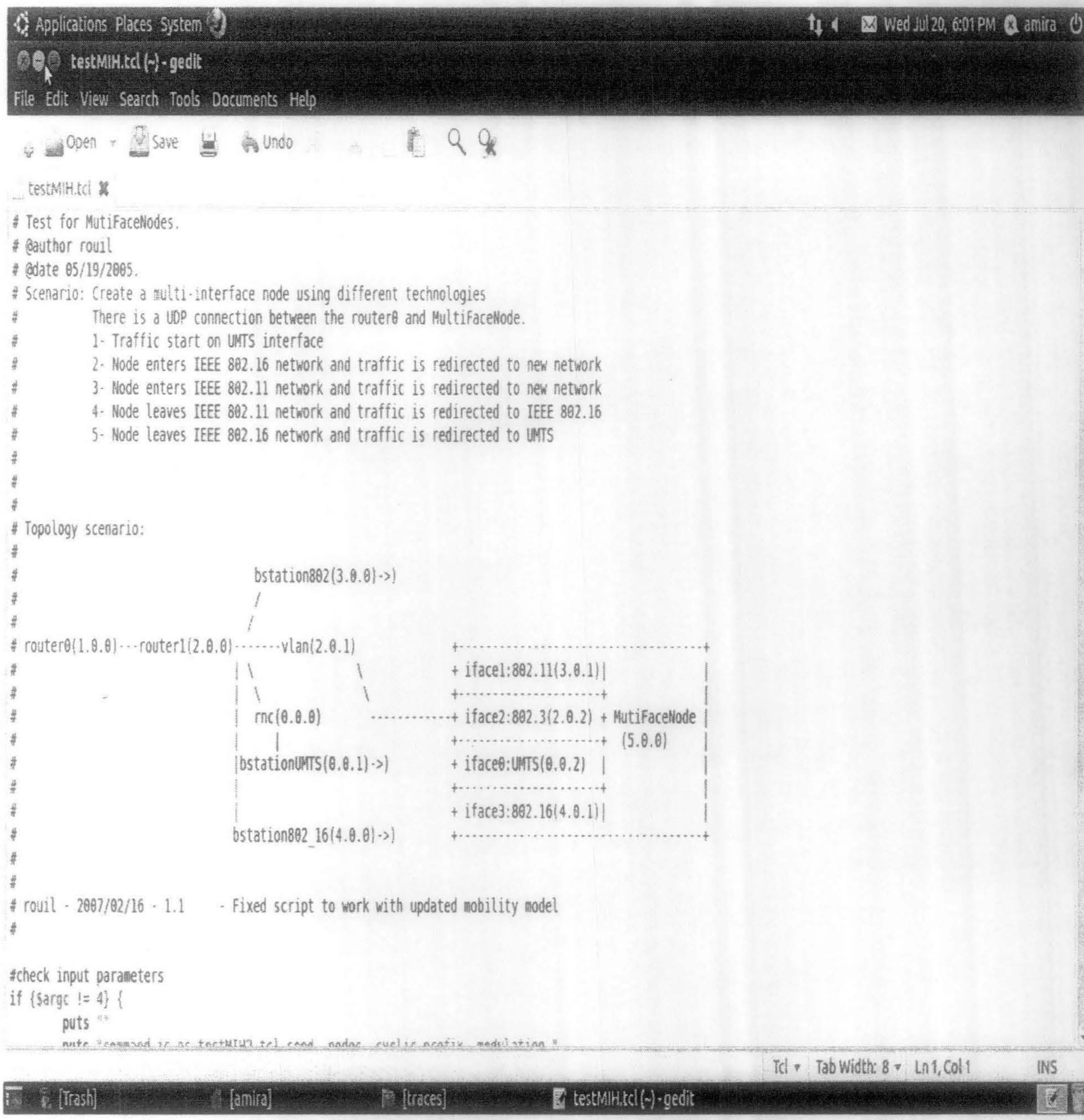
IMPLEMNTATION

&

SIMULATION RESULTS

5.1 Implementation

We are going to simulate a complex wireless-wired scenario. The topology consists of heterogeneous technologies, WiMAX base station ,Wi-Fi base station ,UMTS base station and we use Ethernet as wired part. we simulate [video sender] as a MultiFace node that transmit the video data to router0 [video receiver]. There is also number of other nodes spread in the topology within an area whose boundary is defined as 2000X2000. A MyUDP connection is setup between the video sender node and the video receiver node. Different CBR traffic patterns are being generated as background traffic between other nodes. Just as with any ns simulation, we begin by creating a tcl script for the wireless simulation. We call this testMIH.tcl



```
# Test for MutiFaceNodes.
# @author rouil
# @date 05/19/2005.
# Scenario: Create a multi-interface node using different technologies
#          There is a UDP connection between the router0 and MultiFaceNode.
#          1- Traffic start on UMTS interface
#          2- Node enters IEEE 802.16 network and traffic is redirected to new network
#          3- Node enters IEEE 802.11 network and traffic is redirected to new network
#          4- Node leaves IEEE 802.11 network and traffic is redirected to IEEE 802.16
#          5- Node leaves IEEE 802.16 network and traffic is redirected to UMTS
#
#
#
# Topology scenario:
#
#                              bstation802(3.0.0)->)
#                              /
#                             /
# router0(1.0.0)---router1(2.0.0)-------vlan(2.0.1)           +-----------------------------------+
#                             | \            \                + iface1:802.11(3.0.1)|             |
#                             |  \            \               +---------------------+             |
#                             | rnc(0.0.0)        ------------+ iface2:802.3(2.0.2) + MutiFaceNode |
#                             |     |                         +---------------------+  (5.0.0)    |
#                             |bstationUMTS(0.0.1)->)         + iface0:UMTS(0.0.2)  |             |
#                             |                               +---------------------+             |
#                             |                               + iface3:802.16(4.0.1)|             |
#                             bstation802_16(4.0.0)->)        +-----------------------------------+
#
#
# rouil - 2007/02/16 - 1.1     - Fixed script to work with updated mobility model
#

#check input parameters
if {$argc != 4} {
        puts ""
```

We defined the component of UMTS network, the radio network controller (RNC), the node_B (umts base station) and finally the user equipment nodes as follows.

```
testMIH.tcl
# Note: The UMTS configuration MUST be done first otherwise it does not work
#       furthermore, the node creation in UMTS MUST be as follow
#       rnc, base station, and UE (User Equipment)
$ns set hsdschEnabled_ 1addr
$ns set hsdsch_rlc_set_ 0
$ns set hsdsch_rlc_nif_ 0

# configure RNC node
$ns node-config -UmtsNodeType rnc
set rnc [$ns create-Umtsnode 0.0.0] ;# node id is 0.
$rnc color Red
puts "rnc: tcl=$rnc; id=[$rnc id]; addr=[$rnc node-addr]"

# configure UMTS base station
$ns node-config -UmtsNodeType bs \
                -downlinkBW 384kbs \
                -downlinkTTI 10ms \
                -uplinkBW 384kbs \
                -uplinkTTI 10ms \
                -hs_downlinkTTI 2ms \
                -hs_downlinkBW 384kbs

set bsUMTS [$ns create-Umtsnode 0.0.1] ;# node id is 1
puts "bsUMTS: tcl=$bsUMTS; id=[$bsUMTS id]; addr=[$bsUMTS node-addr]"
$bsUMTS color Red

# connect RNC and base station
$ns setup-Iub $bsUMTS $rnc 622Mbit 622Mbit 15ms 15ms DummyDropTail 2000

$ns node-config -UmtsNodeType ue \
                -baseStation $bsUMTS \
                -radioNetworkController $rnc

set iface0 [$ns create-Umtsnode 0.0.2] ;# node id is 2
puts "iface0(UMTS): tcl=$iface0; id=[$iface0 id]; addr=[$iface0 node-addr]"
$iface0 color Red
Tcl   Tab Width: 8   Ln 115, Col 1   INS
[Trash]   [amira]   [traces]   testMIH.tcl (~) - gedit
```

Then we defined the parameters used by Wi-Fi and WiMAX base stations.

```
# parameter for wireless nodes
set opt(chan)           Channel/WirelessChannel     ;# channel type for 802.11
set opt(prop)           Propagation/TwoRayGround    ;# radio-propagation model 802.11
set opt(netif)          Phy/WirelessPhy             ;# network interface type 802.11
set opt(mac)            Mac/802_11                  ;# MAC type 802.11
set opt(ifq)            Queue/DropTail/PriQueue     ;# interface queue type 802.11
set opt(ll)             LL                          ;# link layer type 802.11
set opt(ant)            Antenna/OmniAntenna         ;# antenna model 802.11
set opt(ifqlen)         50                          ;# max packet in ifq 802.11
set opt(adhocRouting)   DSDV                        ;# routing protocol 802.11
set opt(umtsRouting)    ""                          ;# routing for UMTS (to reset node config)

set opt(x)              2000                        ;# X dimension of the topography
set opt(y)              2000                        ;# Y dimension of the topography
```

In ns-2 a node consists of network components like Link Layer (LL), Interface Queue (IfQ), MAC layer, the wireless channel nodes transmit and receive signals from etc. At the beginning of a wireless simulation, we need to define the type for each of these network components. Additionally, we need to define other parameters like the type of antenna, the radio-propagation model, the type of ad-hoc routing protocol used by the nodes etc.See comments in the code below for a brief description of each variable defined.

```
# configure Access Points
$ns node-config  -adhocRouting $opt(adhocRouting) \
                 -llType $opt(ll) \
                 -macType $opt(mac) \
                 -channel $chan \
                 -ifqType $opt(ifq) \
                 -ifqLen $opt(ifqlen) \
                 -antType $opt(ant) \
                 -propType $opt(prop)    \
                 -phyType $opt(netif) \
                 -topoInstance $topo \
                 -wiredRouting ON \
                 -agentTrace ON \
                 -routerTrace OFF \
                 -macTrace OFF  \
                 -movementTrace OFF
```

Next setup, we set the traffic flow pattern between the two nodes as follows:

```
set udp1 [new Agent/myUDP]
$multiFaceNode attach-agent $udp1 $iface0          ;# new command: the interface is used for sending
$udp1 set packetSize_ $packetSize
$udp1 set_filename sd_akiyo_cif_xvid_m4v ;#sd_a01
#-------
set null1 [new Agent/myEvalvid_Sink]
$ns attach-agent $router0 $null1
$ns connect $udp1 $null1
$null1 set_filename rd_akiyo_cif_xvid_m4v ;#rd_a01

set original_file_name st_akiyo_cif_xvid_m4v ;#st_a01
set trace_file_name video1.dat

set original_file_id [open $original_file_name r]
set trace_file_id [open $trace_file_name w]

set pre_time 0
set frame_count 0

while {[eof $original_file_id] == 0} {

    gets $original_file_id current_line
    #scan $current_line "%d%s%d%d%f" no_ frametype_ length_  tmp1_ tmp2_
    scan $current_line "%d%s%d%s%s%s%d%s" no_ frametype_ length_ tmp1_ tmp2_ tmp3_ tmp4_ tmp5_
    #puts "$no_ $frametype_ $length_ $tmp1_ $tmp2_ $tmp3_ $tmp4_ $tmp5_"
    set time [expr int(($tmp2_ - $pre_time)*1000000.0)]


    if { $frametype_ == "I" } {
      set type_v 1
      set prio_p 0
   }
    if { $frametype_ == "P" } {
      set type_v 2
      set prio_p 0
```

Next we encoded a YUV sequence into MPEG4 data format using xvid_encraw decoder and recorded the trace file (st) for the sender using MP4. To Test the MPEG4 video delivery over the network described above we run the simulation in the usual way (type at prompt: "ns testMIH.tcl") with the required input parameters.

At the end of the simulation run, the trace file and nam trace file are generated with video.dat. As we have turned on the AgentTrace we see VIDEO and CBR pkts being received and sent by nodes, and Agent objects. Ns2 also creates two files, sd and rd. The file sd is to record the sending time of each packet while the file rd is used to record the received time of each packet. Using the tool eg.sh (from Evalvid) to generate the received video (err_inline.cmp), we decode the received video to yuv format using xvid_denraw., Then we fix the decoded yuv sequence and Computed the PSNR and mos using psnr and mos (from Evalvid).

5.1.1 Required steps before ruining the Simulation

- **xvid_encraw -i akiyo_cif.yuv -w 352 -h 288 -framerate 30 -max_key_interval 30 -o a01.m4v**

These examples will create compressed raw videos with 30 frames per second, a GOP length of 30 frames with no B-frames. The bit rate-control from XviD does not work, so it is omitted here.

- **MP4Box -hint -mtu 1024 -fps 30 -add a01.m4v a01.mp4**

This command lines create ISO MP4 files containing the video samples (frames) and a hint track which describes how to packetize the frames for the transport with RTP.

- **Creating Reference Videos**

For some (most) video quality assessment methods, you need a reference video. This is either the original YUV file before the encoding or the YUV file created by decoding the coded video. Whether you need a "decoded" YUV depends on objective you want to achieve. If you want to assess the quality of a video over network system and not only the encoder quality you should create the "decoded" YUV file. To produce these YUVs, you should use the corresponding video decoder (xvid_decraw). If it doesn't exist or can't output YUV, you can try this:

ffmpeg -i a01.mp4 a01_ref.yuv

The mp4trace tool from EvalVid is able to send a hinted mp4-file per RTP/UDP to a specified destination host.

mp4trace -f -s 192.168.0.2 12346 a01.mp4

It sends the H.264 track of a01.mp4 to the UDP port 12346 of host 192.168.0.2. You can watch the video with, e.g., QuickTime Player from Apple. Now you have the MP4 file and the corresponding trace files for both the sending and the receiving side.

5.1.2. Steps after running the Simulation

With the original YUV file we need to evaluate the QoS and perceptual quality with EvalVid.

akiyo_cif.yuv(andakiyo_cif ef.yuv)	raw source files (before and after encoding)
akiyo_cif.mp4	encoded, encapsulated and hinted video file
sd_ akiyo_cif	sender dump (IP packet dump sender)
rd_ akiyo_cif	receiver dump (IP packet dump receiver)
st_ akiyo_cif	sender trace (information about frame types, packet segmentation, ...)

The eg (error generator from EvalVid) tool takes a sd and st file. Given a bit error rate and an error distribution model, it generates a rd file, where the lost packets are marked.

- **eg sd_a01 rd_a01 st_a01 AWGN 250000**

This generates the rd_a01g file assuming an AWGN error model and a BER of 4E-6 (1/250000).

- **Evaluation**

The first step in the evaluation process is the calculation of the reference PSNR. This is the PSNR of the coded and decoded video without transmission errors/losses in relation to the non-coded raw video source.

- **psnr 352 288 420 akiyo_cif.yuv a01_ref.yuv > ref_psnr.txt**

The next step is the reconstruction of the transmitted video as it is seen by the receiver. For this, the video and trace files are processed by etmp4 (Evaluate Traces of MP4-file transmission):

- **etmp4 sd_a01 rd_a01 st_a01 a01.mp4 a01e**

This generates a (possibly corrupted) video file, where all frames that got lost or were corrupted are deleted from the original video track. Actually, two files are saved, a MP4-file containing the damaged video track (a01e.mp4), and a raw video file containing only the undamaged frames (a01e.264). These files are decoded to produce the YUV-file as seen at the receiver. If the appropriate decoder is not able to produce a YUV with as much frames as the original, the following can be tried:

- **ffmpeg -i a01e.mp4 a01e.yuv**

The resulting YUV file should contain exactly as much frames as the original YUV file. Unfortunately most codecs are not able to decode corrupted video files properly. E.g., ffmpeg often produces less frames or even crashes. The PSNR of the received video is calculated by:

- **psnr 352 288 420 akiyo_cif.yuv a01e.yuv > psnr_a01e.txt**

Etmp4 also creates some more files:

loss_akiyo_cife.txt	Contains I, P, B and overall frame loss in %)
delay_akiyo_cif .txt	Contains frame-nr., lost-flag, end-to-end delay, inter-frame gap sender, inter-frame gap receiver, and cumulative jitter in seconds

rate_s_akiyo_cife.txt	Contains time, bytes per second (current time interval), and bytes per second (cumulative) measured at the sender
rate_r_akiyo_cif .txt	Contains time, bytes per second (current time interval), and bytes per second (cumulative) measured at the receiver

For delay or jitter distributions, the hist tool could be of interest. E.g.,

- **awk '{print $3}' < delay_a01.txt | hist - 0 .05 50**

It gives the time, PDF and CDF of the end-to-end delay of a transmission.

- **Video Quality**

Since the PSNR alone does not mean much, you might want to use a quality metric that calculates the difference between the quality of the encoded video and the received (possibly corrupted) video. For this purpose, Evalvid provides the MOS and MIV tools. They calculate the "Mean Opinion Score" of every single frame of the received video and compare it with the MOS of every single frame of the original video. It counts (within a given interval) the amount of frames with a MOS worse than original. If you have made two measurements and calculated, e.g., ref_psnr.txt, psnr_01.txt, and psnr_02.txt, put these files into the directory "/work".

- **mos /work ref_psnr.txt 25 > mos.txt**

The above command is used to calculate the average MOS of every psnr file (last column of mos.txt) and the percentage of frames with a MOS worse than in the reference file in a sliding interval of 25 frames. These percentages are stored in miv_a01e.txt. Finally, the MIV command calculates the maximum percentage of frames with a MOS worse than original:

- **miv /work > miv.txt.**

5.2 Simulation Results

In the previous section, we have explained how the code in simulation tools has been modified to conduct simulations. WE have chosen an area of 2000 x 2000 meter simulation area. The network consists of WiMax, Wi-fi, UMTS and mobile nodes deployed randomly.MPEG video file are transmitted across the network. The following section lists the vedio sources used in this simulation.

5.2.1 Video sources

The video sources and parameters used in this work are listed below:

- Akiyo sequences in CIF (352 x 288) resolution
- MPEG-4 (XviD)

- Parameters

    - FPS=30 # frames per second
    - GOP=30 # an I-frame every second
    - MTU=1472 # max. packet size (only payload)
    - BER=100000 # bit-error rate (1/BER) for simulation with eg
    -
    - T1_DELAY=0 # minimal expected end-to-end delay [s]
    - T2_DELAY=.01 # maximal expected end-to-end delay [s]
    - T1_JITTER=0 # minimal expected inter-frame gap [s]
    - T2_JITTER=.25 # maximal expected inter-frame gap [s]
    - T_STEPS=50 # number of intervals for PDF/CDF
    -
    - YUVDIR="../yuv" # src video directory
    - RAWDIR="raw" # encoded video directory
    - MP4DIR="mp4" # encapsulated video directory
    - TRACEDIR="traces" # tcpdump and video trace files directory

- REFDIR="ref"   # reference PSNR directory
- RESDIR="results" # results directory (loss, delay, vq)
-
- YUVS="akiyo_cif "

Different methods are used to deal with the damaged video are given below:

- fx - lost frames are truncated
- f0 - lost frames are filled with zeros
- p0 - lost packets are filled with zeros

To evaluate the performance of the network the following metrics have been observed:

- End-to-end delay
- Jitter
- Throughput
- PSNR

We are using Standard MPEG encoder that generates three distinct types of frames, namely I, P, and B frames. The I frame is encoded in Intra mode, and is essential for the prediction coding of other frames. If part of an I frame is lost, then all frames in the group of pictures (GoP) including this particular frame are impaired. The P frame is encoded in prediction mode, while the B frame is encoded in double prediction mode. As with the I frame, the P frame is also important. If part of a P frame is lost, the impairment propagates the particular P frame, previous B frames, and the following frames in the GoP that includes this P frame. Conversely, if part of a B frame is lost, the impairment propagates solely within that frame.

5.2.2 End-To-End Delay

The following three figure 5.1, 5.2 and 5.3 show the results obtained in terms of end-to-end delay for the cases lost frames truncated, lost frame filled with zeros and lost packets filled with zeros, respectively. This simulations is conducted for 100 mobile nodes. In all the three figures, it is clearly visible that the end-to-end delay is high in the beginning since at the start I frames are transmitted. The delay is becomes almost constant after words except few peaks in between.

The reasons for delay becoming almost stationary is that most frames are P and B frames that are transmitted. P and B frames are small, therefore, delay is reduced. The explanation for higher peaks in the middle is that after transmitting a group of frames, in the next group it begins with I frames that re larger in size and therefore, takes more time in transmission.

The delay is for frames filled with zero is slightly less than the frames that are truncated because the processing time involved in reassembly of truncated frame may be slightly higher than the frames filled with zeros. Further, the end-to-end delay for packets will be obviously higher since packets consist of frames, therefore, reason for comparatively higher delay for packets than frames is quite understandable and requires no explanation.

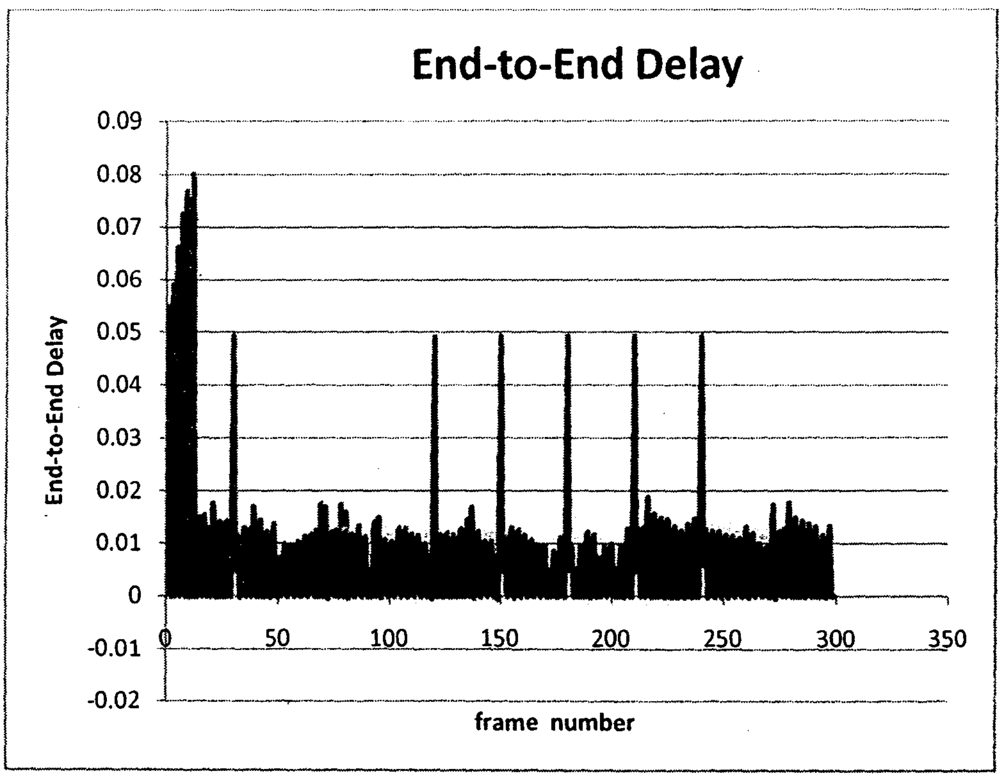


Figure 5.1 End-to-End delay for lost frame truncated

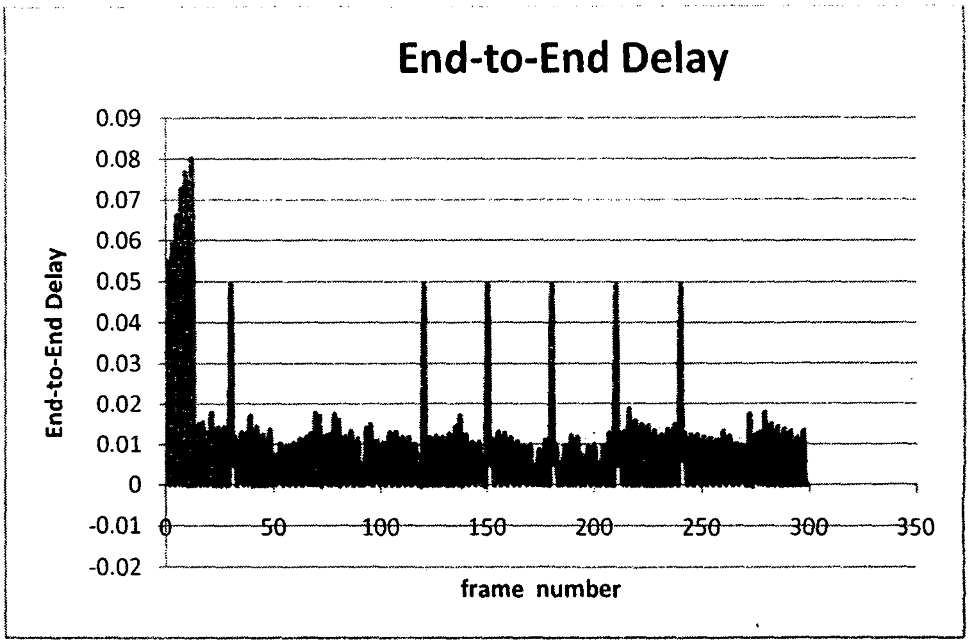


Figure 5.2 End-to-End delay for lost frame filled with zero

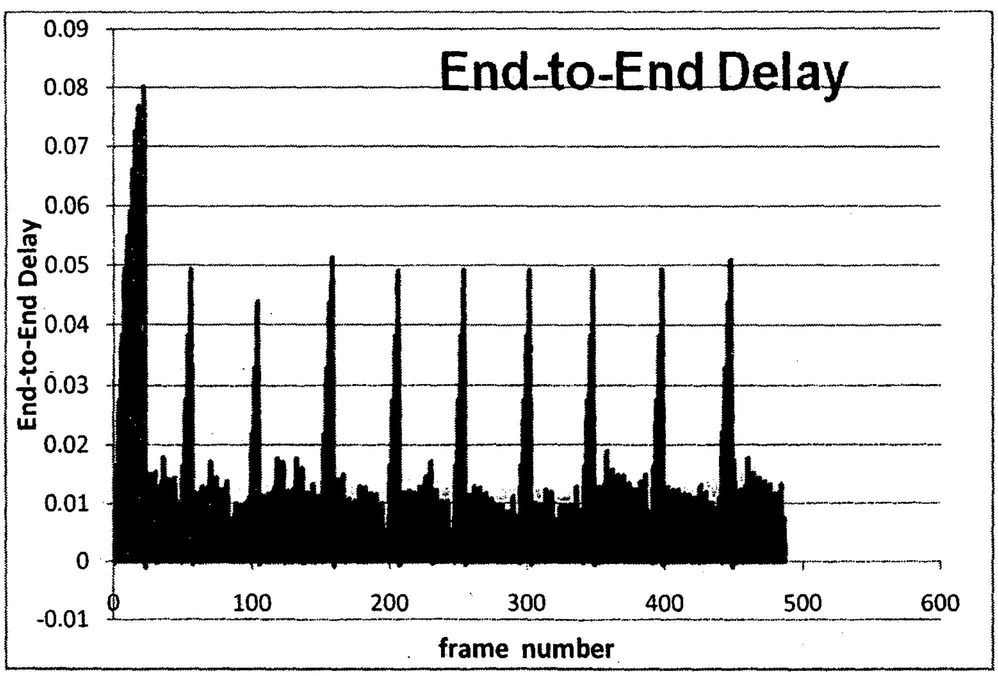


Figure 5.3 End-to-End delay for lost packets filled with zero

5.2.3 Frame/Packet Loss

In the following figure 5.4 and 5.5frame loss and packet loss for all three types of frames I, P and B are given. We see that the highest loss value is for frame B and that is because of the dependency of this frame on both I and p frames as we have mentioned above. The loss is lover for I frames since they are independent frames. P frames have loss between I and B frames since it depends on only previous frame. However, the packet loss is more that frame loss since a packet may consists of many lost frames.

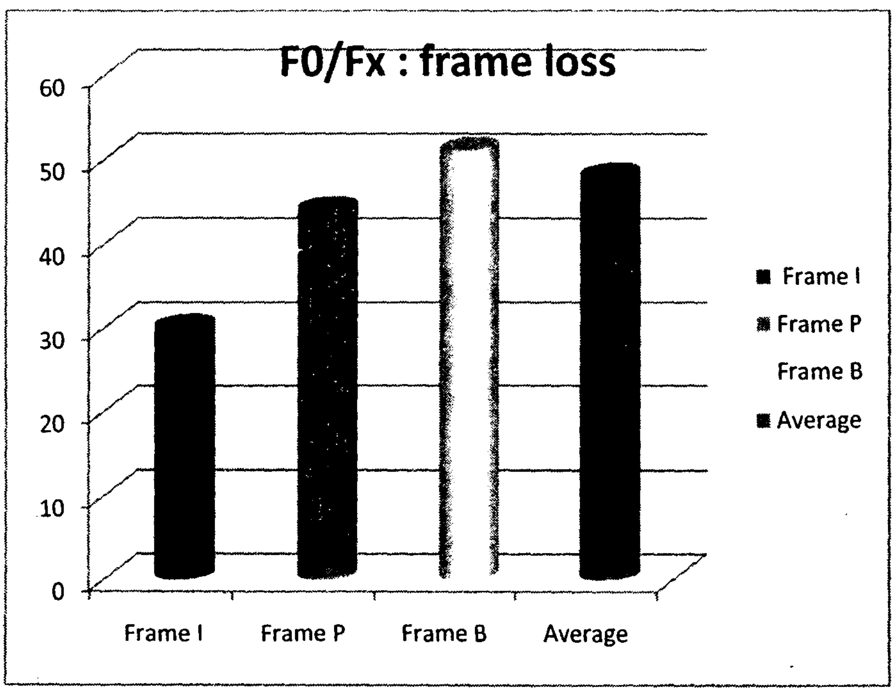


Figure 5.4 Average frames loss for F0 and Fx

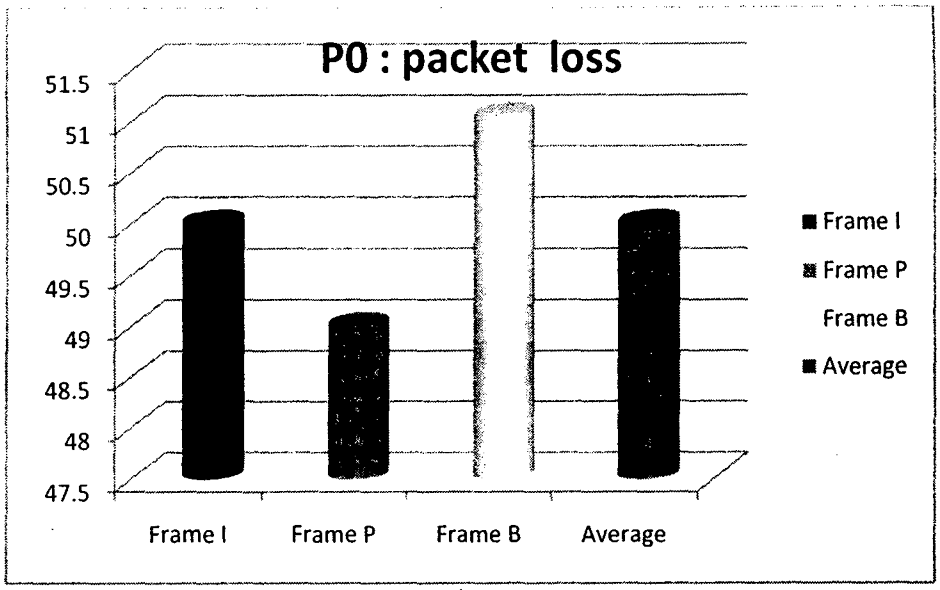


Figure 5.5 Average Packet loss

5.2.4 Jitter

Another important metrics for video is jitter that represents the variation in delay f different frame or packets. Following three figures 5.6, 5.7, and 5.8 shows the jitter as the frames/packets are transmitted continuously. Fig 5.6 shows jitter when lost frames are filled with zeros. Fig. 5.7 shows jitter when lost packets are filled with zeros, and figure 5.8 shows jitter when lost frames are truncated.

In all three figure jitter have both positive and negative values. When the delay of successive frame/packet is smaller than previous one, jitter has positive value. When the delay of previous frame/packet is smaller than the current one, jitter has negative value. Jump in the value of jitter from higher to lower or vice versa value of jitter for frames/packets shows that I frames are preceded or followed by P or B frames. The constant value of jitter shows that similar kinds of frames are preceding or following, i.e. I frames are preceded or followed by I frames or P frame are preceded/followed by P frame and B frames by B frames. However, it is clear from figures that jitter for packets has larger values than frames since packets are larger in size than frames, therefore, the difference is delay of packets containing I frames from oackets containing P or B frames is larger.

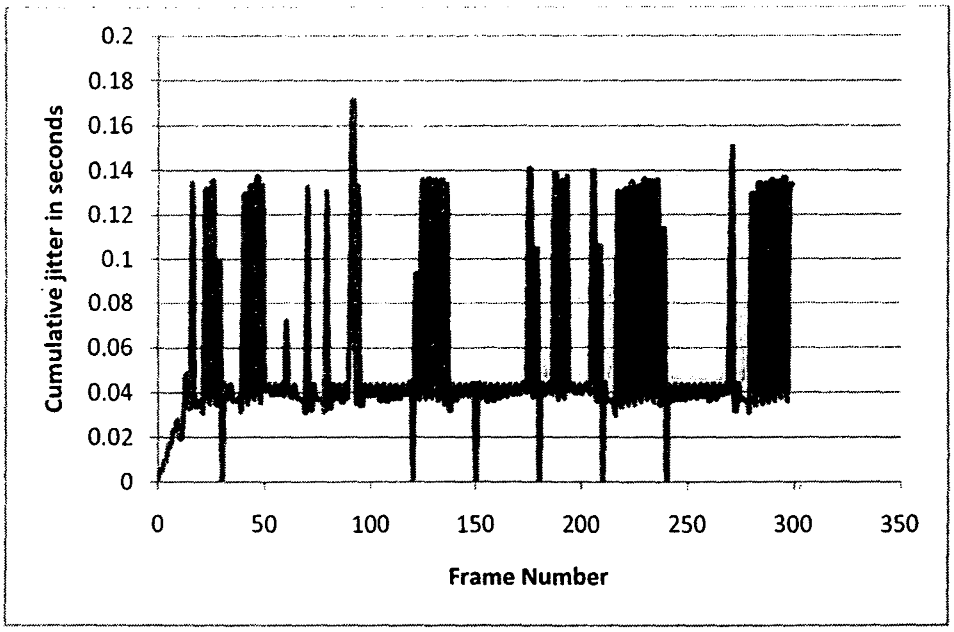


figure 5.6 jitter for frames filled with zero

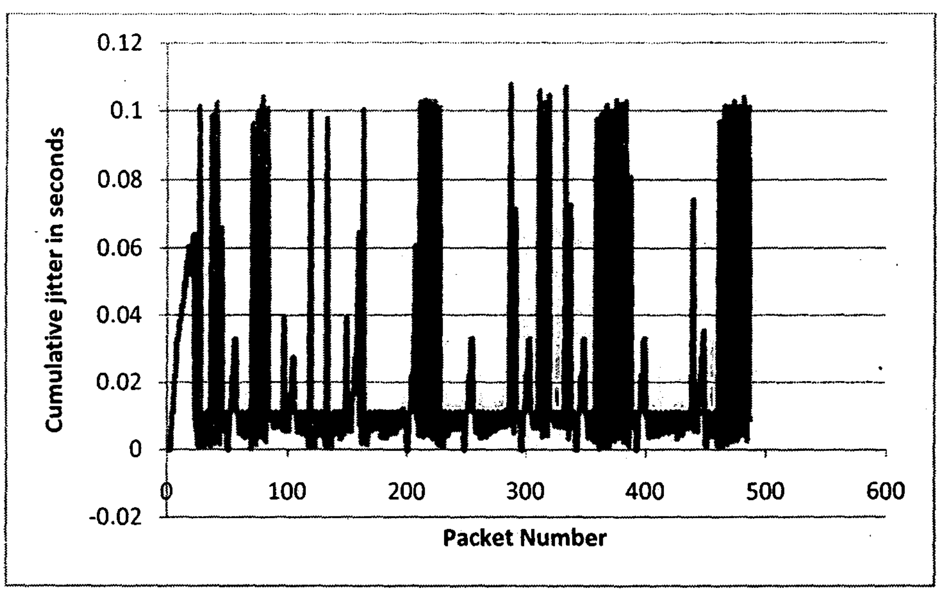


fig. 5.7 jitter for Packets filled with zero

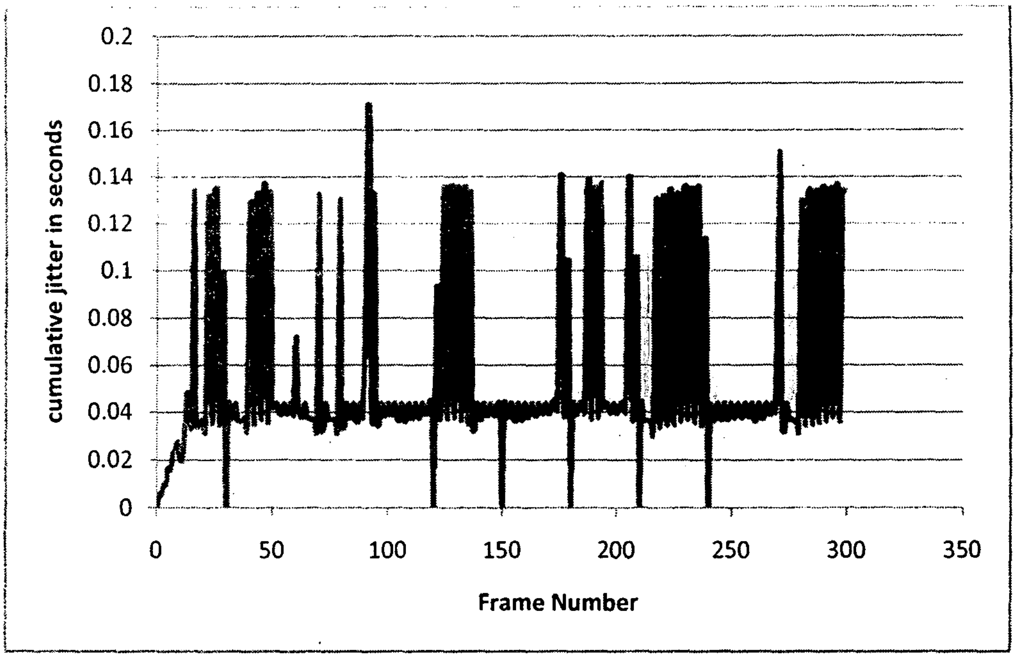


Figure 5.8 jitter for frames truncated

5.2.5 Throughput

The figure below shows the average throughput of the network with different no of mobile nodes for different executions. This is increasing steadily because of the availability of WiMax connectivity is always available and as the data comes its being transmitted without failure. The throughput needs to be computed for large size network to have better estimates of throughput.

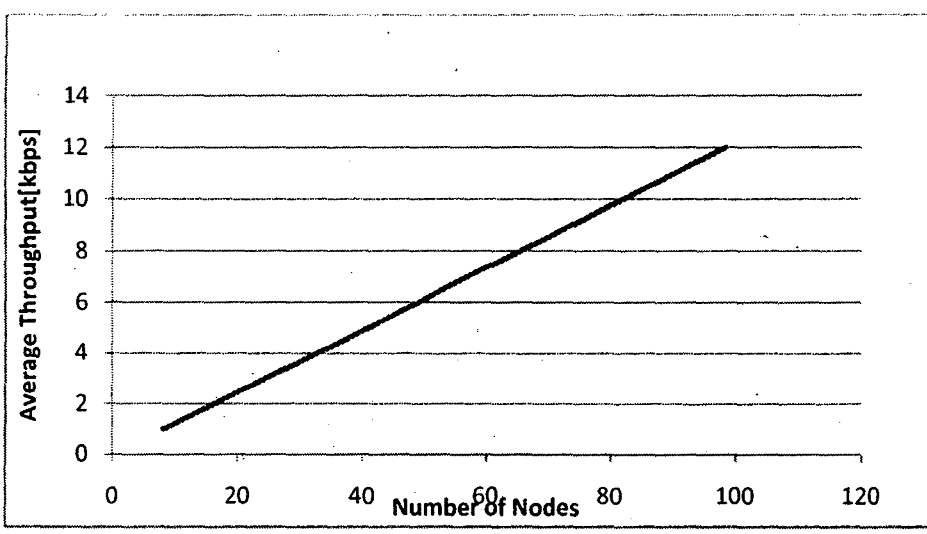


Figure 5.9 throughput

5.2.6 Signal to Noise Ratio (SNR).

In the following figure 5.10 and 5.11 SNR for before transmission (sender) and after transmission of frames is shown respectively. P and B frames are more susceptible to noise than I frames since little noise in them may make it difficult to reassemble the video. The high peaks in between shows SNR for I frames. The larger almost constant values show SNR for P and B frames. However, at receiver SNR increases as the number of frames increases. It is because as the frame increases P and B frames also increases. Therefore, small error in P or B frames has larger effect in terms of SNR.

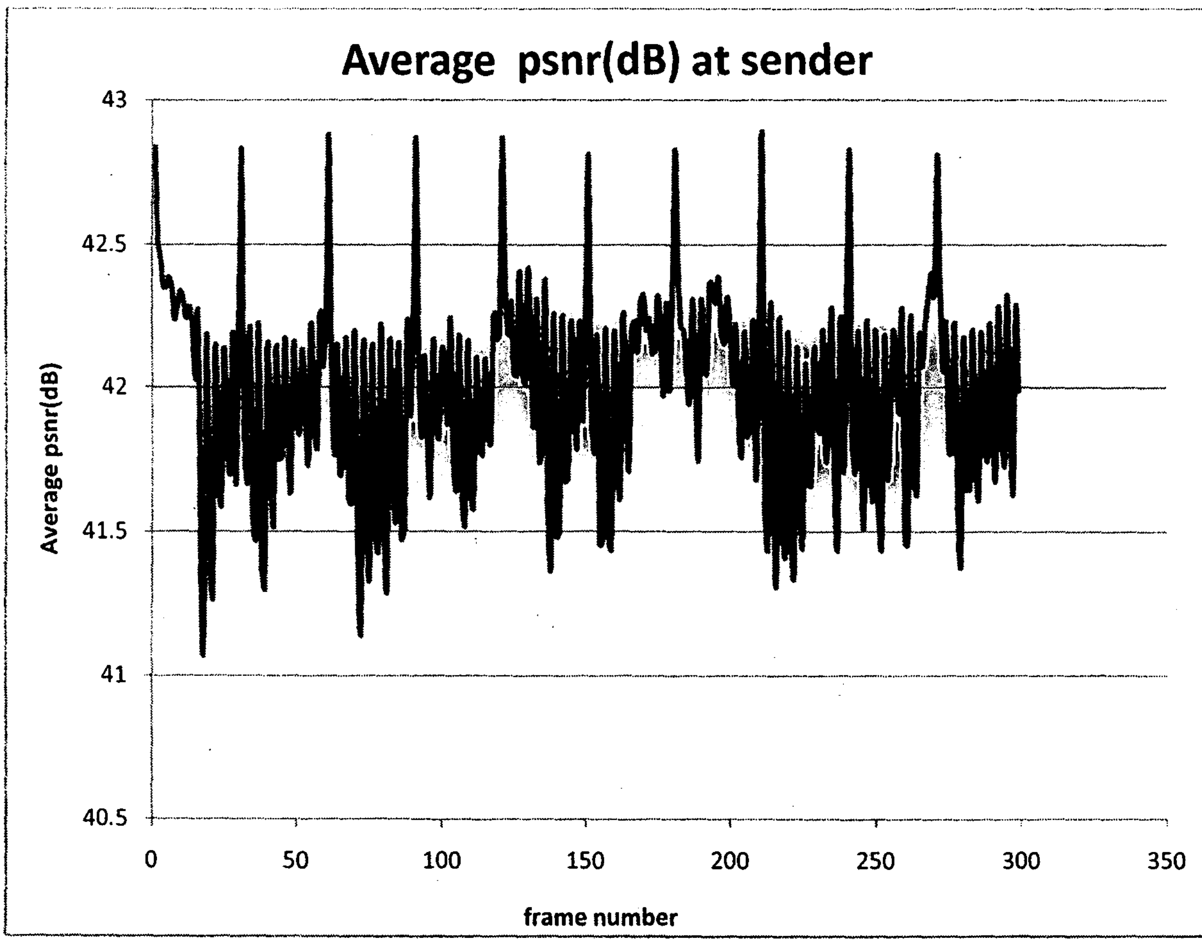


Figure 5.10 PSNR for Sender

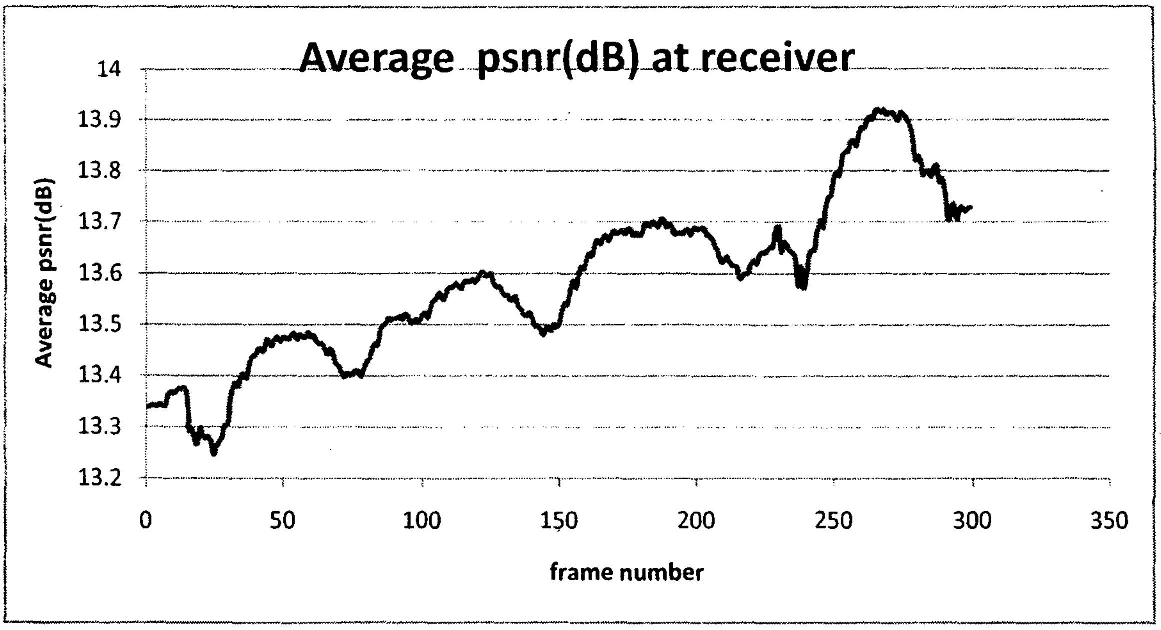


Figure 5.11 PSNR for Receiver

CHAPTER 6

CHAPTER 6

CONCLUSION

As mobile devices become increasingly multi-access-capable, users will come to expect seamless, always best connected operation, especially in the presence of overlapping network coverage. We reviewed recent developments at the IETF and the WiMAX Forum for addressing the problems caused by the mismatches in the operation of IEEE 802.16 PHY and MAC layers and IETF protocols, and summarized the salient points of their recommendations.

By integrating various modules into a single platform, we make it possible to simulate heterogeneous environments that incorporate a variety of access networking technologies. The problem we are facing is how to interconnect a wide variety of heterogeneous and un-interoperable networks including wired and wireless networks in order to provide users with ubiquitous connectivity and the ability to roam seamlessly and securely across networks of different types. Our long-term goals are to bridge the technological and measurement gaps between stovepipe and heterogeneous network technologies in order to allow for better interconnectivity, seamless interoperability, and better support for network connectivity and mobility services.

In this work, we have analyzed the performance of three wireless technologies i.e. WiMAX, Wi-Fi and UMTS for supporting MPEG-4 video traffic. The performance measures like delay, jitter, packet/frame loss and throughput have been evaluated using different simulation scenarios.

Results indicate that the End-To-End delay of video transmission is very high initially because transmission starts by sending frames I at the beginning then frames P,B,I are sending. So, the End-to-End delay is stationary with some peaks values because of sending I frames in between. Also, the number of lost packets and frames is high and hence badly affects the video quality .

Based on the analysis in this study seems that the video quality at the receiver comparing to ITU recommendation

Future Work

Clearly, there is still lots of empirical work lying ahead. We barely scratched the surface of what is possible to do with state of the art WiMAX equipment. This work can be extended by including mobility aspect, additional combination of various WiMAX physical layer parameters and considering other access technologies as well and comparing the performance of this environment using different handover technique.

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