ENSC 894 SPECIAL TOPICS II: COMMUNICATION NETWORKS

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FINAL PROJECT

PERFORMANCE ANALYSIS OF WIMAX FOR VOICE AND VIDEO CONTENT

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Abstract

IEEE 802.16 Worldwide Interoperability for Microwave Access (WiMAX) is an emerging technology, which is used as a wireless broadband access service as an alternative to the wired technologies. High speed internet mobility is the demand of the present time which is being fulfilled by the advancements in the wireless technologies, one such example being WiMAX. In this project, we aim to simulate voice and video over WiMAX. To implement this, we will be building a network model in Riverbed Modeler 18.0. Riverbed Modeler is a powerful network simulator tool that provides a user friendly environment to test various technologies.

We consider a number of network topologies and different states of the user. The user is first considered to be fixed subscriber station and then mobile. We record a number of parameters such as Jitter, Throughput, Delay and Traffic Sent/Received, to analyze the performance of this network. The mechanism of handover will be implemented to avoid loss of data during mobility.

Acknowledgement

We would like to take this opportunity to thank the people without whom this project would not have been accomplished. First of all, we would like to thank our course instructor Dr. Ljiljana Trajkovic for her kind support and help during the project. We would also like to extend our gratitude to the course TA Mr. Zhida Li for helping us solve the problems that we faced during the implementation of the project.

Arshvir Kaur Anttal Gurleen Kaur

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List of Acronyms

BS	Base Station
CSN	Connectivity Service Network
Fps	Frames Per Second
H1	Handover Threshold Hysteresis
HO	Handover
IEEE	Institute for Electrical and Electronics Engineers
IP	Internet Protocol
LOS	Line of Sight
MAC	Medium Access Control
MS	Mobile Station
NLOS	Non Line Of Sight
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
PSTN	Public Switched Telephone Network
QoS	Quality Of Service
SNR	Signal To Noise Ratio
SOFDMA	Scalable Orthogonal Frequency Division Multiple Access
SS	Subscriber Station
VoD	Video On Demand
VoIP	Voice Over Internet Protocol
Wi-Fi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access

Chapter 1.

Introduction

WiMAX (Worldwide Interoperability for Microwave Access) is a widely used, IEEE 802.16 standard, wireless network technology for commercial and residential use. WiMAX provides services to fixed as well as mobile users. It is of two types: Fixed and mobile. Infact, it is able to reach the areas where setting up a wired infrastructure is not possible or is relatively tough. It is based on the wireless Metropolitan Area (MAN) Technology. 802.16d standard specification allows WiMAX to operate in 2-11 GHz range. It thus works at different frequencies in different countries depending on the spectrum allocation. Its biggest feature is of interoperability, which allows the service providers to easily switch between various networks as per the customer demand [5]. It provides a wider coverage area, of up to 50 km, and a wide range of applications as compared to the present wireless technologies. WiMAX provided services include Voice over IP (VoIP) calling and efficient audio and video streaming to fixed and mobile stations.

As such, Audio/Video Streaming at high speed and quality is highly in demand due to the growth in the fast paced Internet services. All end to end services in WiMAX are based delivered over IP Architecture. As the next generation is being anticipated to have exponentially large amount of audio/visual content [2], it is very important to put into use efficient systems that can carry large amounts of data. Users can use the internet for voice communication by sending voice data in packets (VoIP), using IP contrary to the traditional Public Switched Telephone Networks [4].

In this project, we aim to simulate voice and video over WiMAX to observe the performance over our example network. The video traffic source is a trace file for movie ‘Tokyo Olympics’ made available online by Arizona State University [14][15]. We simulate the WiMAX wireless technology for different topologies of the network and for different states of the user. The model proposed in [1] has been taken as our reference model. We implement mobility in the system as an enhancement of the existing model. Further we to obtain graphs for some measurement factors such as delay, jitter and throughput. We use Riverbed Modeler 18.0 to implement our project, which is an efficient network simulation tool. Various configurations needed to setup the environment to implement the project are also shown in the further sections.

Chapter 2.

Background Knowledge

2.1. WiMAX

WiMAX is a wireless metropolitan access network (MAN) technology that is based on the standards defined in the IEEE 802.16 specification [6]. IEEE 802.16-2004d was published for fixed services and in 2005 new standard (802.16e) was amended for mobile services. The standard 802.16e uses Scalable Orthogonal Frequency-division Multiple Access (SOFDMA) rather than Orthogonal Frequency-division Multiplexing (OFDM). WiMAX provides a maximum transfer speed of 75Mbps per channel, and a maximum range of 50km. Thus, WiMAX provides higher transmission speed as well as greater transmission coverage, as compared to other communication technologies like ADSL and Wi-Fi. This powerful technology was predicted to be on its way to replace Wi-Fi. Users would find it very similar to Wi-Fi in terms of usage method. It would be connected to a WiMAX antenna as we are now connected to our Wi-Fi modems [7]. WiMAX has features like larger bandwidth and larger transmission coverage, because of which it is able to solve the last mile problem [2].

WiMAX provides services in the frequency range up to 10-66 GHz using Line-of-sight (LOS) communication with a single base station and single carrier (SC) air interface. The Non-line-of-sight (NLOS) communication was also made possible using WiMAX in standard 802.16a. Using this standard, 2-11 GHz band provided NLOS communications using any of the three air interfaces: Single Carrier, Orthogonal Frequency Division Multiplex (OFDM), and OFDMA. The OFDM and OFDMA helped in providing higher bandwidth and higher data capacity. Further, in these services, the subcarriers are orthogonal to each other, thus they can be placed closer to each other without any interference. The bandwidth of the channel for transmission ranges between 1.25 MHz and 20 MHz in the 2 – 11 GHz band. Data rates between 1.5 to 75 Mbps can be achieved by WiMAX by placing various subcarriers in the 2-11 GHz channel and by using various modulation techniques [1]. The subcarriers within a channel bandwidth can be allocated as: Null Subcarriers, Data Subcarriers, Pilot Subcarriers and DC subcarriers. Further, these subcarriers can be modulated by using various modulation schemes provided by the WiMAX:

- BPSK
- QPSK
- 16 QAM
- 64 QAM
- 256 QAM

WiMAX system has two major parts, namely, WiMAX base station and a receiver. Within a WiMAX cell, one base station can provide services to many users. The radii of the WiMAX cells can range between 7km and 10km. WiMAX can provide backhaul links to Wi-Fi and various redundant wireless Internet backup links for commercial business purposes. The following figure shows the reference model developed by the WiMAX Forum's Network Working Group in 2005[5].

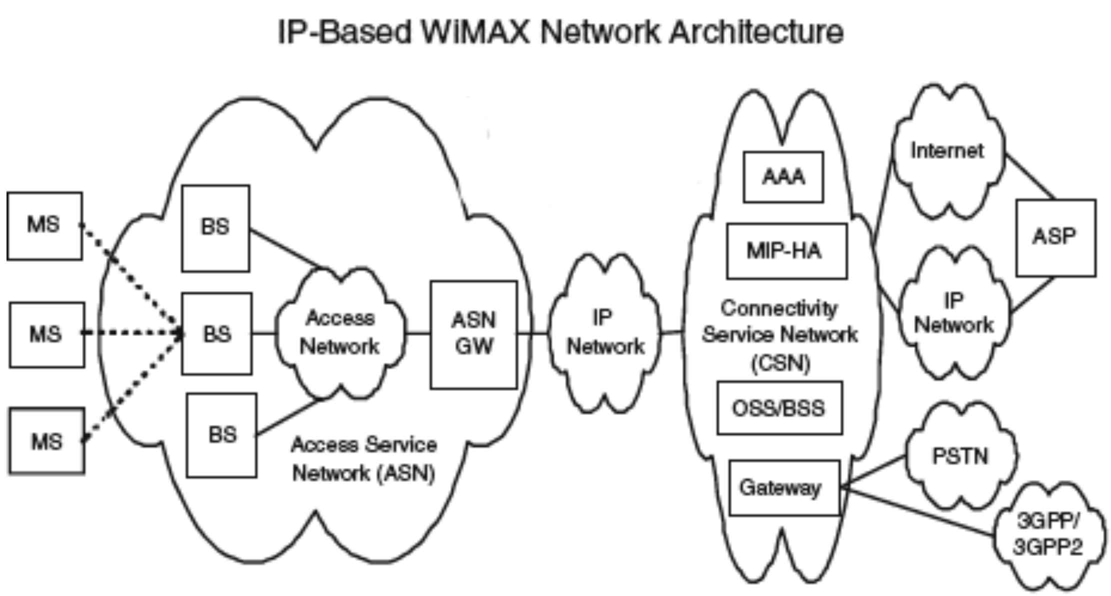


Figure 2.1. WiMAX Architecture [5]

This figure shows a number of mobile stations connected to various base stations. The base station provides air interface to the Mobile Station [5]. Base station, just as in other network technologies, has a number of functions to perform such as handoff management, Radio Resource Management, QoS limits enforcement among users and many more services. The Access Service Network is composed of groups of base stations and Access Service Network Gateways. The gateway is typically an aggregation point for traffic to and from the access service network. Further, the Connectivity Service Network is a medium to connect to other networks such as Internet, PSTN, IP Networks and all 3GPP Networks. CSN has various components that serve the various functional responsibilities of the CSN such as IP Address Management, policy management of QoS and security.

2.2. Voice over Internet Protocol (VoIP)

A rapid growth in the technology in recent time has led to the increased demand and thus development of various multimedia applications such as VoIP, Audio/Video Streaming, etc. Voice over Internet Protocol (VoIP) is a technology that uses Internet broadband services to enable voice calls instead of the conventional phone calls. As the name suggests, the voice traffic is transmitted over the network in the form of packets. VoIP can be implemented on any most of the available IP based networks like Internet, Ethernet, Fiber Optics, WiMAX and 3G[4]. Some VoIP services allow users to call those who have same services, but some services allow to call who have a telephone number, including local, international, mobile or long distance.

VoIP application is in core a digital transmission of voice signals, thus the voice signals are sampled and then converted to digital format [6]. The signal is again converted to analog form by the decoders at the receiver’s end. As the VoIP packets travel across the medium, they incur losses, which can be in the form of delay in the reception, or partial or full loss of signal. These factors adversely affect the performance of the network that has deployed this scheme.

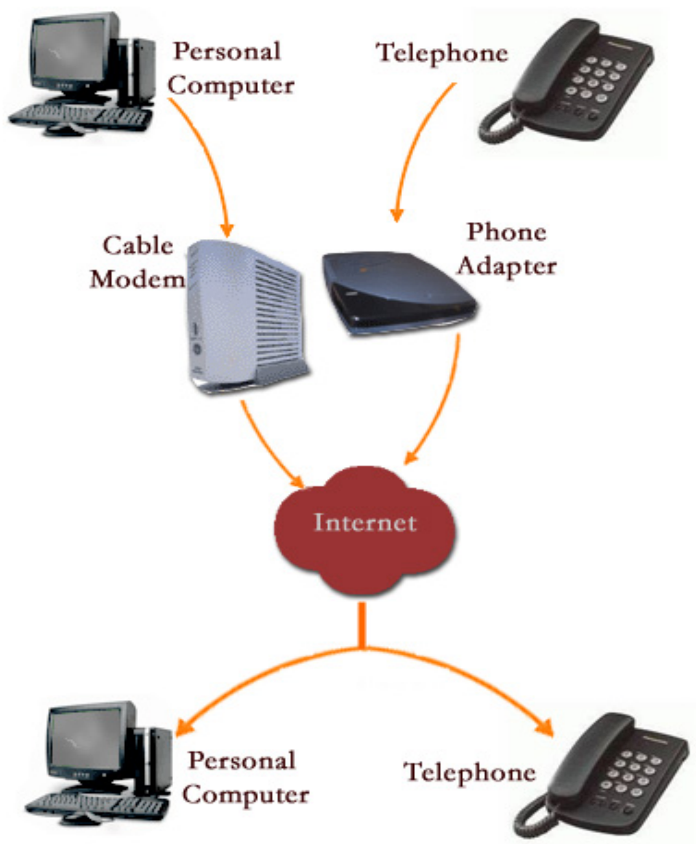


Figure 2.2. Voice over Internet Protocol

VoIP provides services at low costs as compared to the traditional phone services. It has also increased the functionality like the calls can be made while the user is mobile and has good Internet connection. Voice quality may suffer when compression is done but compression reduces bandwidth requirements. VoIP is both delay and loss sensitive. VoIP is available on many smartphones, personal computers, and on Internet access devices. In this project, we try to analyze some of these loss effects on the example WiMAX network which implements this application.

2.3. Video Streaming

Video content is made available to the users by media services like news castings, movies, sporting events and video on demand (VoD). The significant part of real-time multimedia is real-time transport of live video or stored video. The video services can be provided to clients in two modes: downloaded video or streamed video. In video streaming, whole video is not downloaded, but the video is played out and parts of it are received and decoded. Since this is the real-time video, it requires certain bandwidth, delay and loss. Video is sensitive in terms of delay [1] as any delay in video can produce a mismatch in the audio and video being played, thus affecting the quality of video being presented to the user.

The popularity of real time video streaming is rising over the years. According to a report, the word wide revenue for video services is expected to reach $9.5 billion in 2017 which was $4.5 billion in 2013 [9]. This increase is growing further with the increase in mobile devices. Also, 25% of videos streamed are over the tablets. To meet the increasing demand tuning of the delivery networks is done by the vendors. The round trip time is being minimized and optimization of compressing/uncompressing methods is being done.

The video is organized as sequence of frames or images which are transmitted at a constant rate. When the video is being streamed, different requirements from the client station in terms of transmission and buffering. A number of parameters such as video format, pixel color depth, coding scheme, and frame inter-arrival rate [1] form the basic characteristics for video streaming. These characteristics of the video make the size of the raw video very large. To reduce the traffic load such that it satisfies the client's transmission and buffering requirements, encoding of the uncompressed video is done using MPEG-x and H.26 codecs. These encoded streams depend on the bandwidth and delay characteristics of the link.

A video server is used for streaming video to the users over the network. The server can either be placed locally or it can be placed in some other part of the world. If the server is placed in some other part, then Internet services will be used to stream the video from the server. The video is generally requested by the user using the web server. The web server sends the requested video to the video server, which then sends this video to the user in real time. For sending the video in real time, certain protocols are used like Real time protocol (RTP), Real time streaming protocol (RTSP) or Real time transport control protocol (RTCP).

Various video formats are used depending on the type of user being served. The inter-arrival rate of the video frame varies from 10 fps (frame per second) to the 30 fps. The video inter-arrival rates are affected by the network conditions which can degrade the quality of the video. The figure given below represents the need on the client's end to playback the frames at a constant rate [1] even in the events of uncompensated delays in video packet arrival rate. There is an unavoidable delay when video frame packets are sent from one end to another.

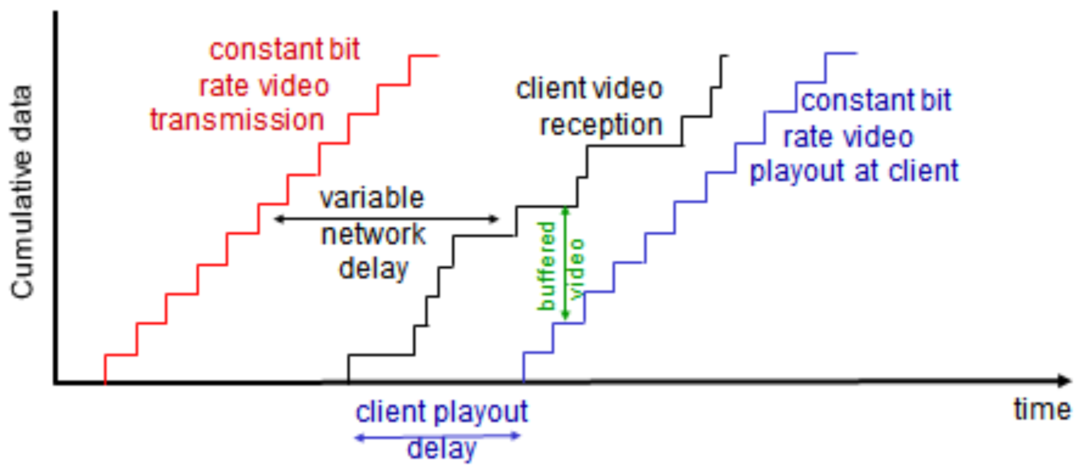


Figure 2.3. Buffering required at video client [1]

Figure 2.5 shows the architecture of video streaming. The raw audio and video data is compressed using compression algorithms and stored in storage device of the streaming server. Whenever the user requests the video, the video is retrieved by the streaming server and sent to the application layer QoS. Here, the video streams are adapted in accordance with the network and QoS requirements. The adapted video is then divided into packets and these packets are sent over the Internet. To improve the quality of delivery of video and to avoid any packet loss due to congestion, distribution services are deployed in the network. The bit streams in the form of packets are received by the client. These packets are then synchronized with respect to each other [10].

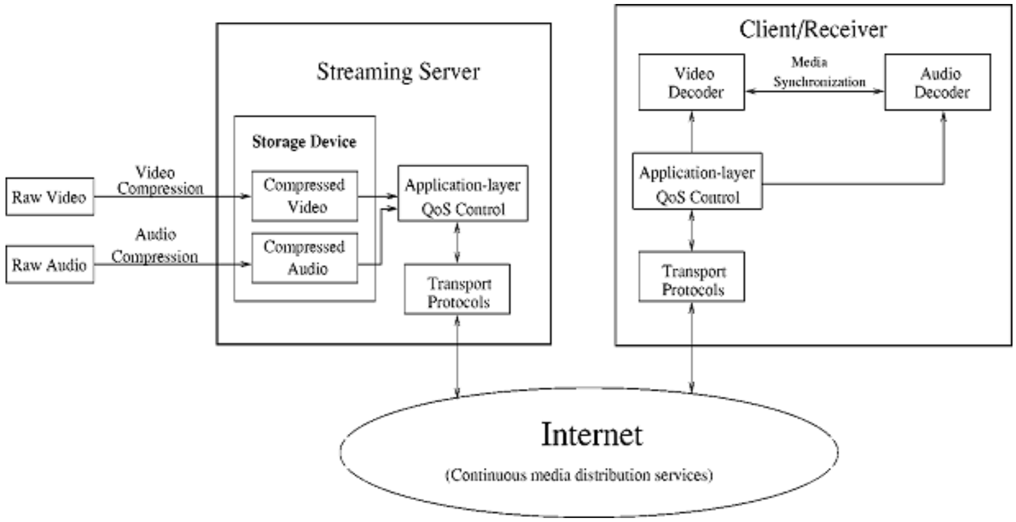


Figure 2.4. Video Streaming Architecture [10]

We aim to evaluate the performance of the video streaming application by simulating a network with certain video content and then viewing its parameters and comparing them with the theoretical outputs. The main parameters that will be observed are:

- Throughput
- Delay
- Jitter
- Traffic Received/ Sent (Packet Loss)

2.4. Handover

When we talk about mobility, we involve the transfer of a user from the range of one serving base station into the range of second one. Handover is required to reduce the chances of the signal being dropped when the user is in motion. Handover also takes place between different wireless technologies. There are two main types of handover techniques used in wireless technologies, namely, soft handover and hard handover. Most of the traffic sent over Mobile WiMAX networks is delay-tolerant. That is why it follows the approach using hard handover. During Hard Handover, the link with the serving base station is first relieved and then a connection is established between the subscriber and the target base station. This method is generally referred to as 'break-before-make' method [11]. One of the criteria used to determine the time when handover takes place is the measurement of SNR. SNR defines the strength of the signal at all times. There are different conditions for handover to take place. A handover process is initiated by the system when the signal strength received by the subscriber from the serving station falls below the signal strength received from the target base station. A certain threshold value is also selected to avoid repeated handovers. A threshold value is chosen such that the mobile station does not get attached to the target base station when it is in the range provided by the serving base station. Also, the mobile station must be handed off when it receives weak signal from the serving base station. This is where the SNR plays its part.

It should be noted that handover might be mobile or base station initiated. The mobile station continuously scans its neighbouring base stations for best signal strengths. Certain messages are exchanged between the mobile station and the base station, and based on this information, the target base station for handoff is chosen.

The handover procedure is as follows:

First, the mobile station sends its handover request to the serving base station. The serving base station contacts the neighbouring base stations by stating the QoS requirements of its mobile station [12]. Once it choses the relevant base station, it informs the mobile station of the possible handover. When the mobile station decides on which target base station it will transfer, it indicates the serving base station of its decision to disconnect its association with it. After confirmation from the target base station, it attempts to re-enter the network through initial ranging provided by the target base station, to reduce the 'handover break time' [12]. Once the mobile station connects with the target base station, the handover is established and data can be sent.

The figure below shows the mobile station initiated handover:

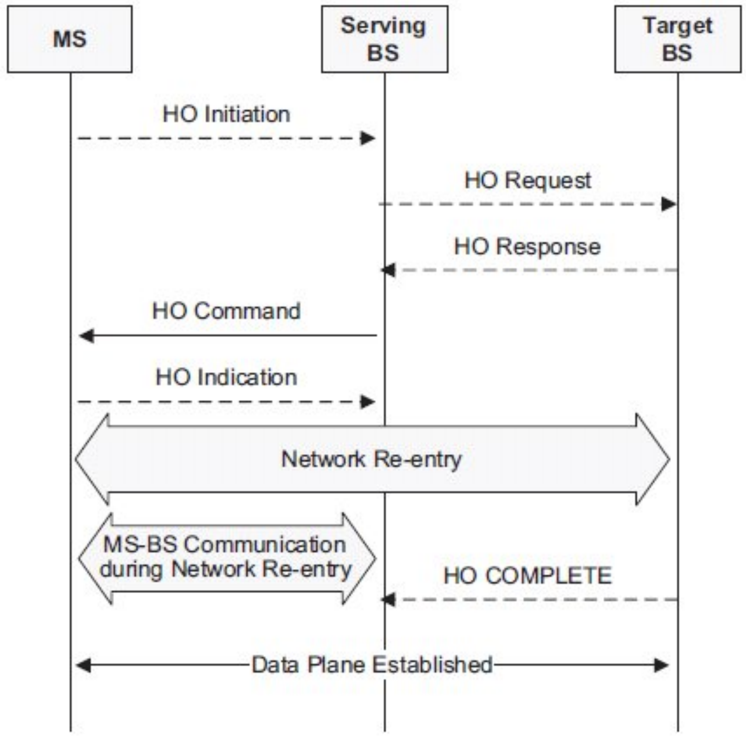


Figure 2.5. MS-initiated handover[12]

In our project, we use the criteria of SNR to trigger handover. According to this method, a handover must take place when the SNR of target base station exceeds the SNR of the serving base station. We use the following equation to justify the algorithm we use:

$$SNR_{maxDT} - SNR_{DS} \geq H_1$$

According to this equation, when the difference between the maximum Downlink SNR of target base station and Downlink SNR of the serving base station exceeds the Hysteresis threshold value, a handover is initiated by the MS [13].

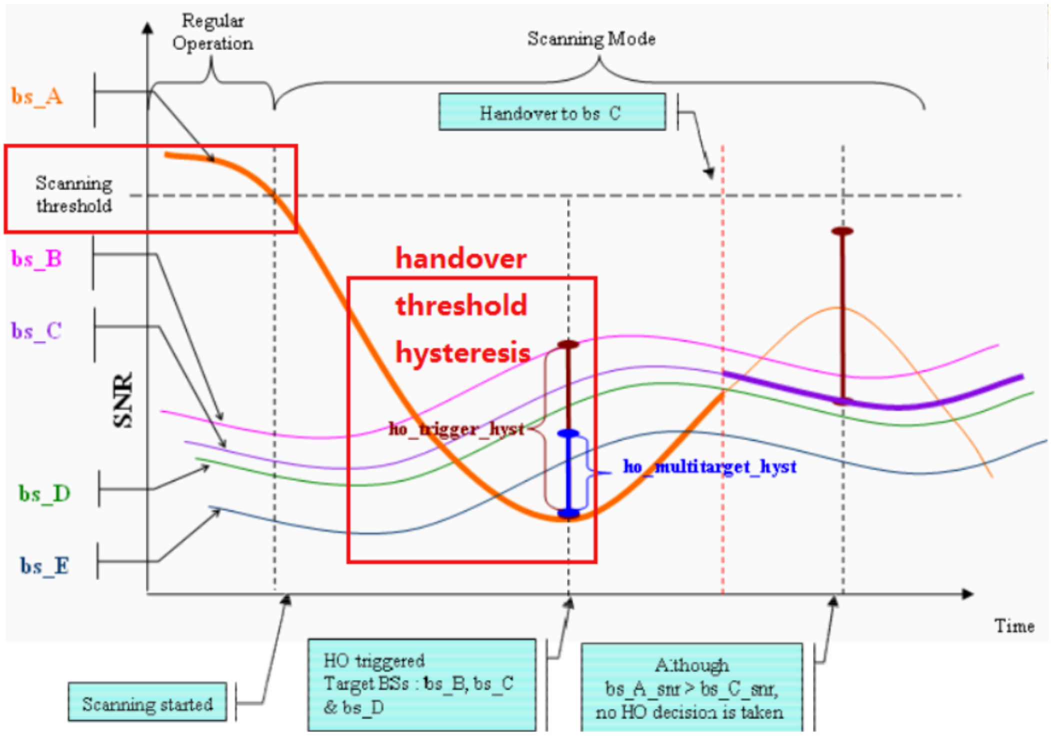


Figure 2.6. Demonstration of Handover threshold hysteresis [13]

2.5. Riverbed Modeler

Riverbed modeler is a powerful network simulator tool which is used to test network topologies before implementation. One can model and then simulate the required wired and wireless models. It allows users to create process, node, link, network models and then apply them in the network. It is a user friendly simulation tool which has a very impressive and easy to use graphical user interface. A number of inbuilt models are present in the software of various telecommunication technologies. One can also edit their process and node models to suit our need. There are a wide variety of options to play with the models, that is, modify as per our need. When a network is designed, we set the parameters we need to record to analyze the performance of our network. With a simple click of a button, we run the model and then see the results in graphical or tabulated form. We use this software to record some of the many network analysis parameters such as Delay, throughput, Jitter and Traffic sent and received. The above is the starting window of the tool. We begin our project by selecting a new project from the File Menu.

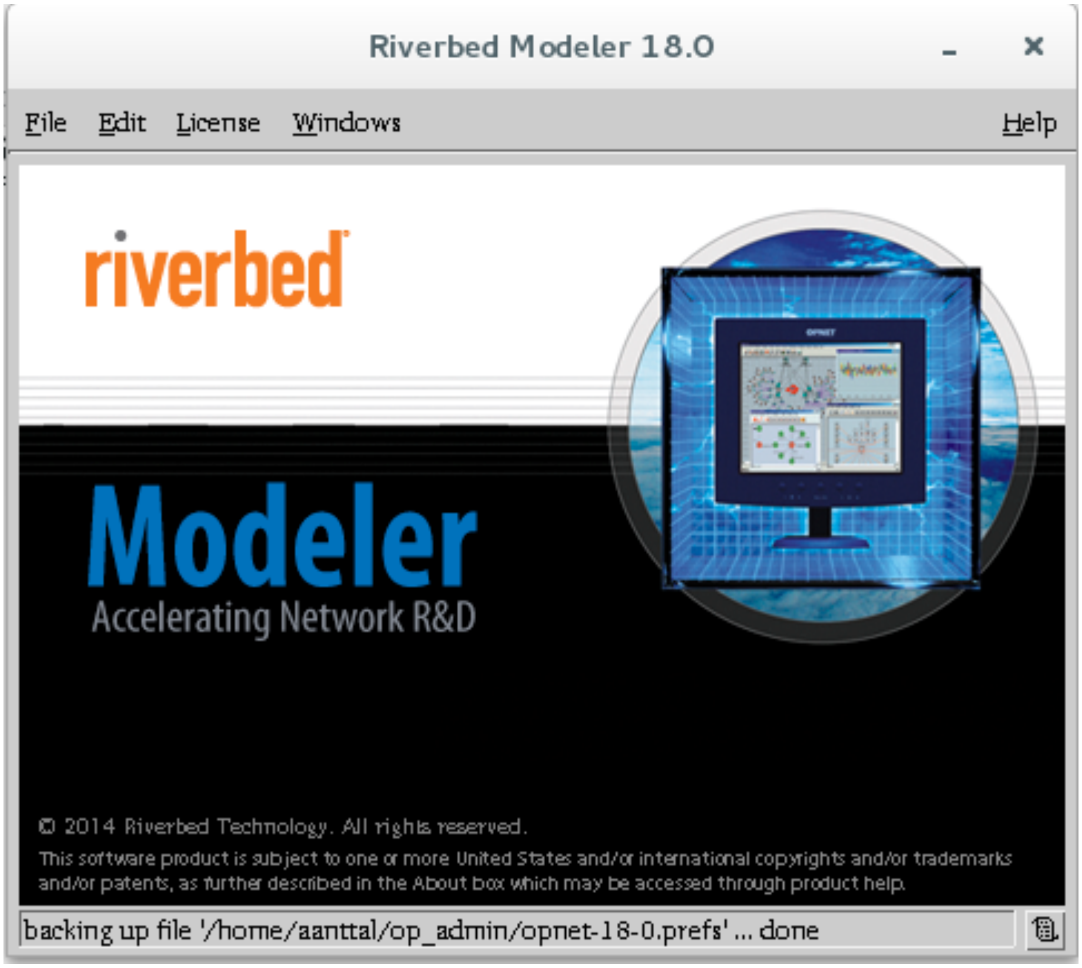


Figure 2.7. Riverbed Modeler

2.6. Related Work

Several studies have been done to study the parameters and hence the performance of WiMAX networks. In [1], the technical details and performance of WiMAX broadband access technology was studied and further compared it to ADSL technology, for video streaming applications. This study was done for fixed subscriber stations placed at different distances from the base station. Trace file of a movie was chosen as the video traffic source. Similar research has been done by [17] to compare both the technologies for audio and video content, under different conditions. In the work done in [18], the performance of WiMAX for mobile stations has been analysed. VoIP and video content was used as the traffic source and handover was shown for the MS roaming in two cells.

As mentioned earlier, the model proposed by W. Hrudey in [1] was used for our study. We try to simulate the performance of WiMAX using both fixed as well mobile nodes. The traffic source used for simulation is VoIP and Video.

Chapter 3.

Simulation Design and Results

3.1. Simulation Design

We simulate different scenarios to simulate WiMAX on Riverbed Modeler 18.0. WiMAX models are available in the software. Firstly, we analyse the performance of the network for fixed subscriber units and then we provide mobility to a subscriber node during VoIP and video streaming. We add background traffic and then see the changes in our analysis. Also, a number of nodes are added to study handover in detail.

3.1.1 Scenario 1

The scenario shown in figure 3.1 shows two fixed nodes and a base station which provides service to both. We assume that the video traffic is being sent from a server at a distance from the base station. The video server is configured to send a 70-minute video. The trace file of the movie ‘Tokyo Olympics’ was made available by Arizona State University [14], [15]. The video is sent at the rate of 30 frames per second and has a frame resolution of 352*288 (pixels). The base station is connected to the server via routers and a switch. The parameters are set in the video conferencing table of the application configuration. We also add background traffic later to see its effect on the performance of the system. The effects are discussed later in the results section. It should be noted that we have also added VoIP as one of our traffic applications. We try to simulate both the applications together. The simulation is run for 30 minutes only. The traffic characteristics are discussed later in this section.

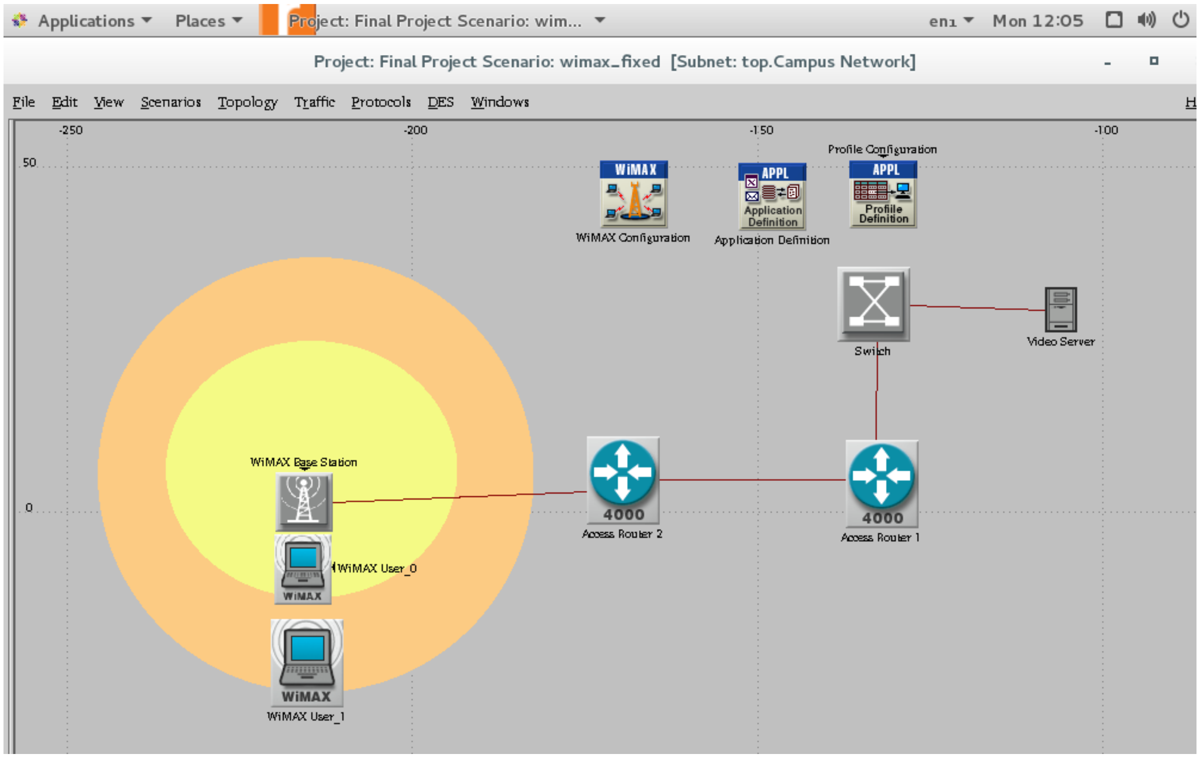


Figure 3.1. Scenario with Fixed Nodes

3.1.2 Scenario 2

The second scenario, shown in figure 3.2, is constructed to view the mobility in WiMAX subscriber stations, according to IEEE 802.16e standard. We deploy two-cell configuration, with the mobile station moving from one cell to another. A trajectory is defined which allows the user to first connect to the home base station (1) and then to the target base station (0). The mobile user is moving at the speed of 10 ms^{-1}. This topology is deployed to study the behaviour of the network when the user is in motion. We observe the process of handover when the user shifts from one cell to another, which takes place to provide good QoS to the user, that is, to provide minimal call drops. In order to enhance the performance of the network, we try to make partial use of the Dual Trigger Handover Algorithm, suggested in [16]. We only consider the effect of different values of handover threshold hysteresis on the handover mechanism.

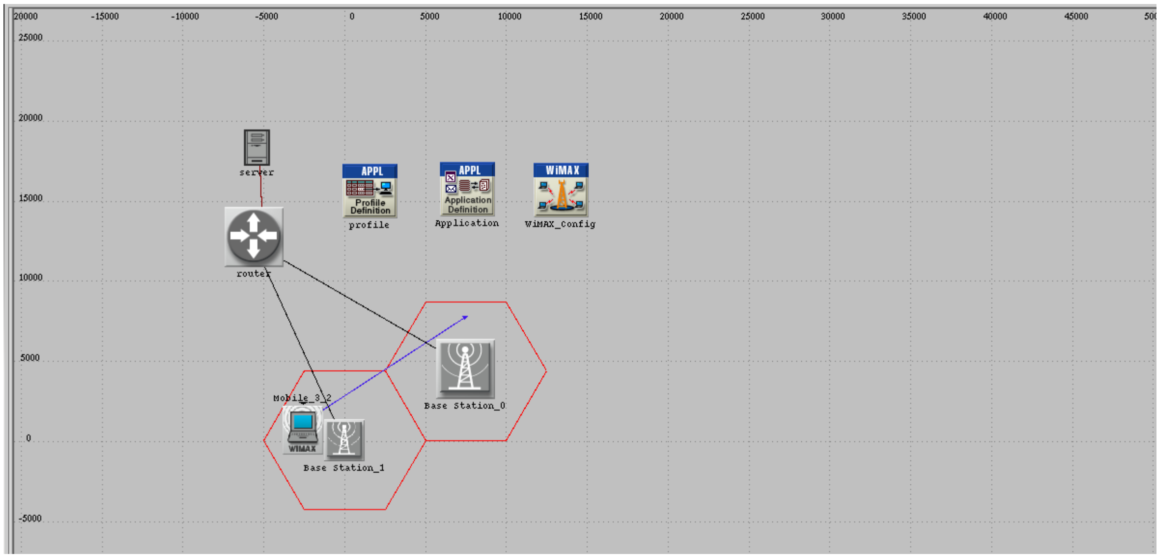


Figure 3.2. Scenario with one Mobile Node

3.1.3 Scenario 3

In this scenario, we deploy a four cell wireless network, with a mobile node. The trajectory of the subscriber unit is such that the MS first moves from the cell with BS 4to BS 1 through BS 2. We notice the handover from one base station to another when the MS follows the defined trajectory. The MS still moves with a speed of 10 ms^{-1}.

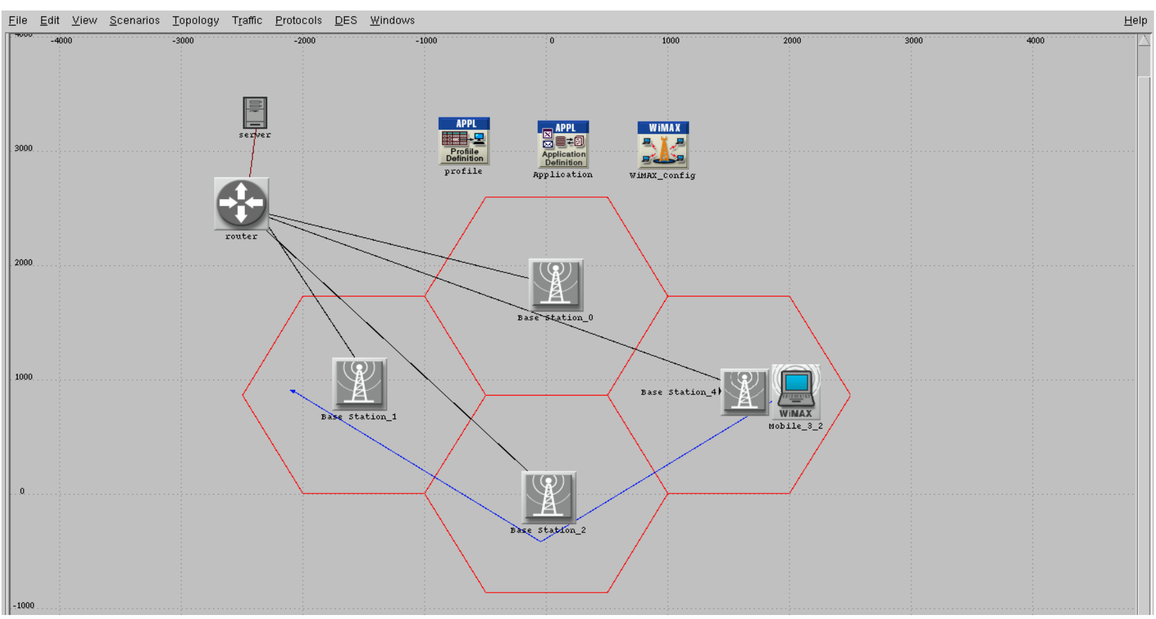


Figure 3.3. Scenario with Mobile Node moving in multiple cells

This scenario is constructed with WiMAX BS and MS and a Handover Threshold Hysteresis value of 6.0 dB is chosen. Further an enhancement in this scenario is done to obtain the scenario shown in figure 3.4. We increase the number of fixed station in one cell through which our MS passes. This is done to see the changes in traffic patterns of the mobile node when a single base station serves a large number of users. The variations in results with the respect to the original scenario are discussed in the next section of the chapter.

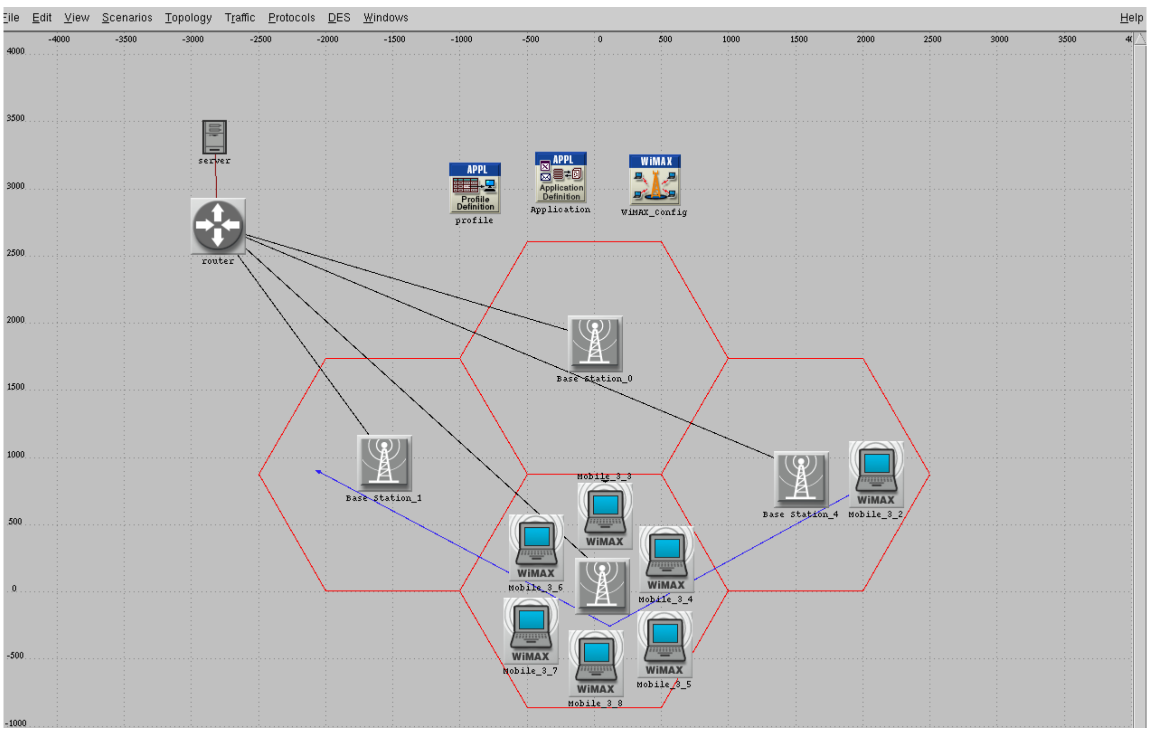


Figure 3.4 Heavy Traffic in one cell

3.1.4 Basic configurations:

The configuration of the WiMAX base station is shown in table 1 below. It can be seen the antenna gain is chosen to be 15dBi and the maximum transmission power is kept to be 3.8 W (Watt). The PHY profile for wireless is chosen to be OFDMA at 20 MHz

Antenna Gain (dBi)	15
MAC Address	1
Maximum Transmission Power (W)	3.8
PHY Profile	Wireless OFDMA 20 MHz
Perm Base	1
Receiver Sensitivity (dBm)	-200

Table 1 WiMAX parameters for BS

The WiMAX parameters in the MS are set as shown in table 2. The antenna gain is now chosen to be 14 dBi and the MAC address is auto assigned. The transmission power is chosen to be 2 watt.

Antenna Gain (dBi)	14
Mac Address	Auto assigned
Maximum Transmission power (W)	2.0
PHY Profile	Wireless OFDMA 20 MHz
Receiver Sensitivity (dBm)	-200

Table 2 WiMAX parameters for MS

The typical video traffic characteristics are shown in table 3. We simulate a 70 minute long MPEG-4 movie named ‘Tokyo Olympics’ for a time period of 30 minutes. The trace file of the video source is available at [14], [15]. The trace file is first processed and saved as a .csv file. The parameters are given to the video conferencing application in the application configuration node. The characteristics for the VoIP traffic are also shown in table 4.

Resolution	352 * 288
Codec	MPEG-4
Frame Rate (fps)	30
Compression Ratio	6.36
Mean Frame Rate (Mbps)	5.7

Table 3 Video Traffic Characteristics

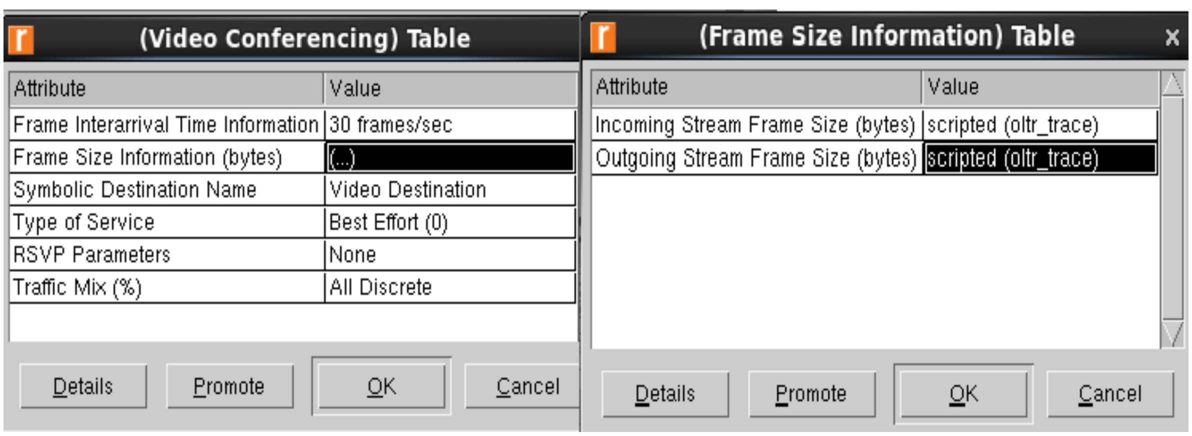


Figure 3.5: Application Configuration for Video Traffic

Voice Codec	G.729A
Voice Service	IP Telephony
Type of Service	Excellent Effort

Table 4 Voice Traffic Characteristics

3.2 Simulation Results

In this section, we show the graphical representation of the data collected to analyse the performance of the network in the scenario described in the previous section of this chapter. The performance metrics are shown to get a clearer idea of the implemented scenarios. We show the graphs in order of the scenarios created.

3.2.1 Scenario 1: The scenario with two Fixed Nodes:

This is the scenario with two fixed nodes, which are placed at approximately 2 and 4 km from the base station. The simulation is first run with the discrete traffic values. We, then add 50% background traffic to reduce the channel utilization to 50%.

Figure 3.6(a) shows the traffic received (packets/sec) by the two users. It should be noted that the traffic received by the node nearer to the base station is greater than the traffic received by the second user. When, the background traffic is added, as shown in figure 3.6(b), the traffic received reduces by almost 40%, for both the users. The blue line graph shows the traffic received over the time interval by user 0, the SS located nearer to the BS and the red line shows the traffic received by user 1, located at the farther end.

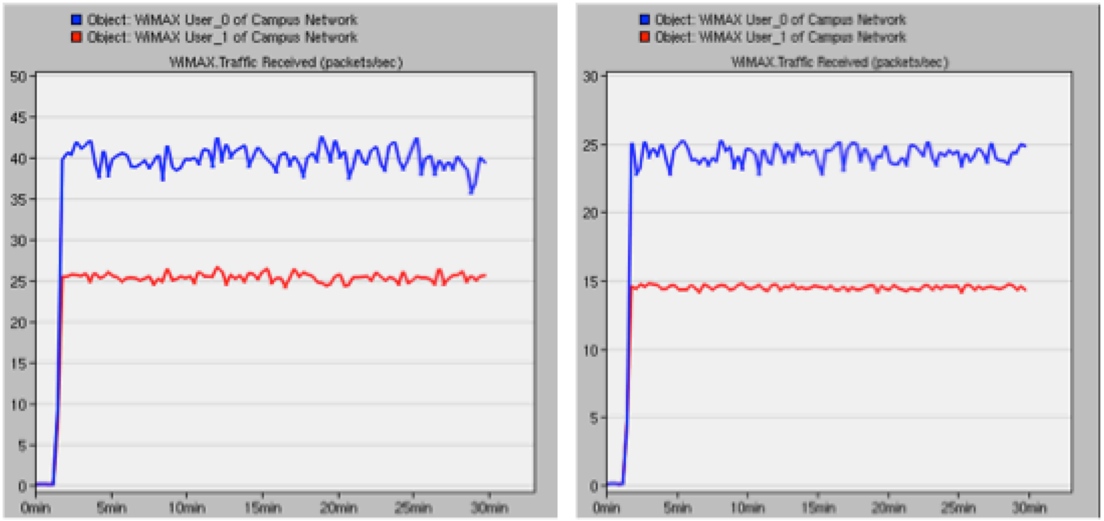


Figure 3.6(a) Figure 3.6(b)

Figure 3.6. (a) Traffic Received (packets/sec) in the absence of background traffic (b) Traffic Received with background traffic

Figure 3.7(a) and (b) show the throughput (packets/sec) of the system for both the users. Throughput can be defined as the number of packets that reach successfully to the other end i.e., the end-to-end transmission rate. Throughput is also recorded for the two cases, that is, in the presence (figure b) and absence (figure a) of background traffic.

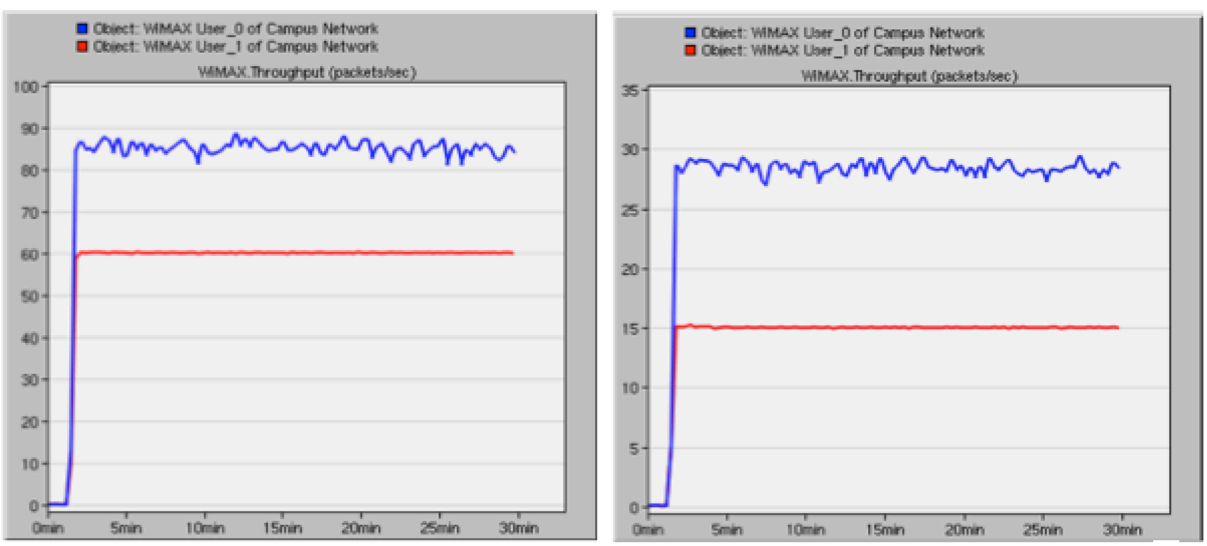


Figure 3.7(a) Figure 3.7(b)

Figure 3.7. (a) Throughput (packets/sec) without background traffic (b) Throughput with background traffic

The throughput is higher for the user located nearer to the BS. User 0 is SS nearer to the BS. The background traffic largely impacts the throughput of the system. It reduces by approximately 66% and 77% for user 0 and user 1, respectively.

Figure 3.8 shows the delay of the system, in seconds, while streaming audio and video. Delay is an important parameter as the users in the wireless network expect flawless transmissions without any wait periods in between. To give a real time feeling to the user, this property is taken into consideration. Both video and voice applications are delay sensitive applications. The user located closer to the BS experiences lesser delay than the other SS. The value of the delay is constant for user 0, however it becomes constant for user 1 as well after an increase initially.

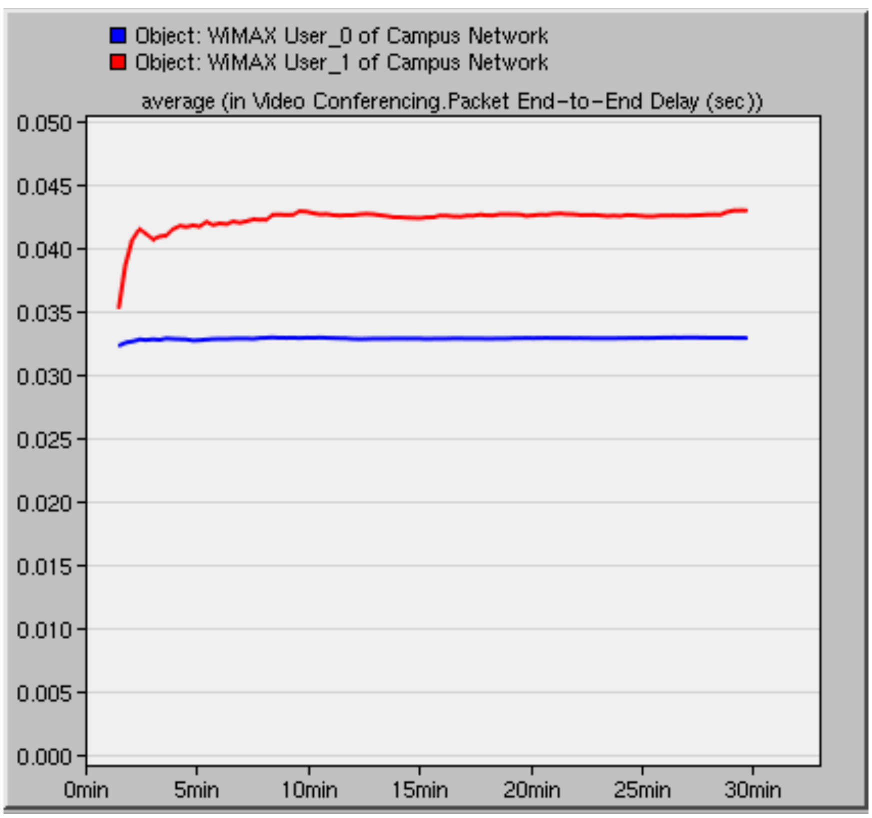


Figure 3.8. Delay (sec)

The variation in packet end to end delay is known as Jitter. It is measured in seconds Figure 3.9 shows the jitter in the video packet transmission to the two users. The red line graph shows the packet delay variation for user 1(farther one) which stabilises at around 20 ms in around 15 minutes. The jitter, for user 0, is comparatively, very less.

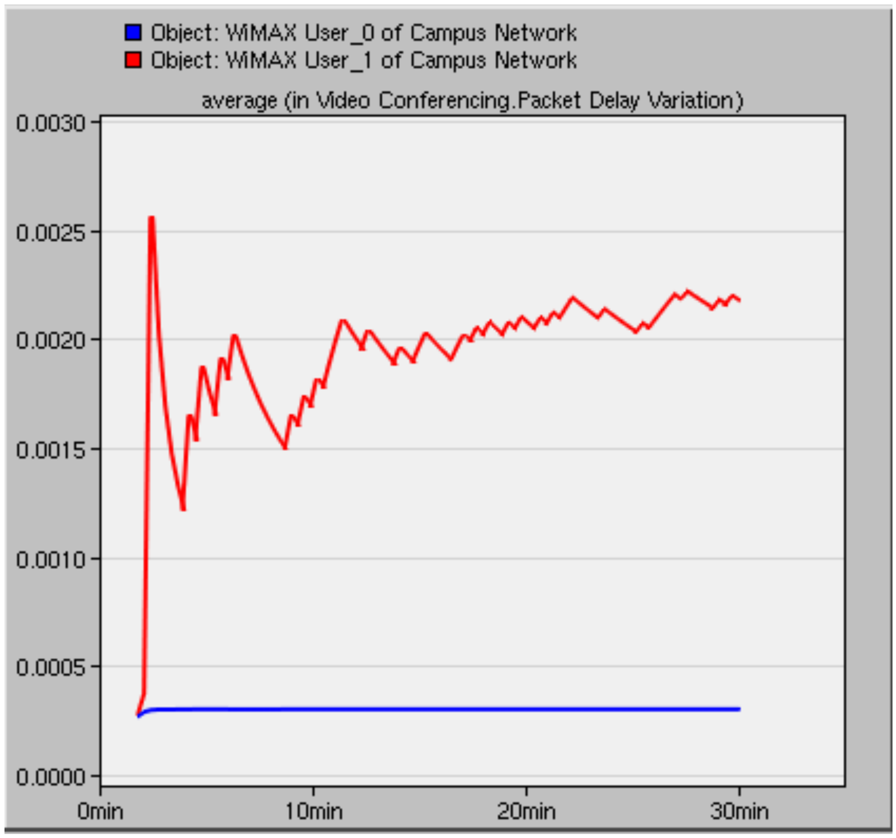
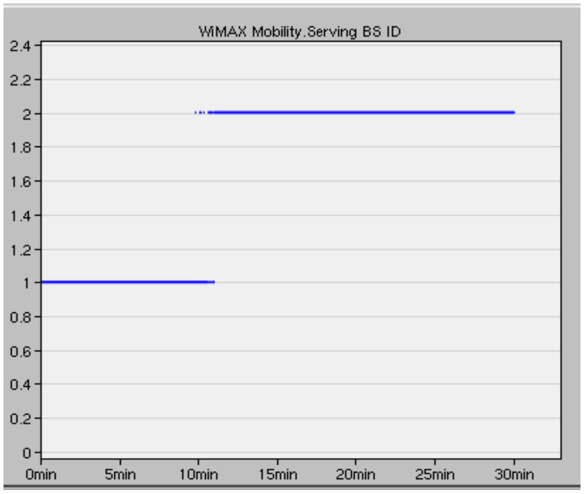


Figure 3.9. Jitter (sec)

3.2.2 Scenario 2: With Mobile node

This scenario is constructed to study handover and its effects. We also study the effect of the change in the value of handover threshold hysteresis value on the handover. Figure 3.10 shows the handover taking place at approximately 12 minutes. The x-Axis represents the time of simulation run and the y-Axis represents the base station number. The mobile user is first served by base station 1 and then by the second base station.



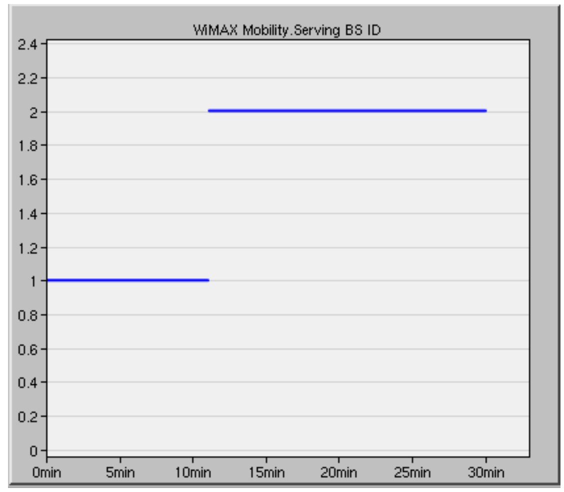


Figure 3.10 (a) **Figure 3.10 (b)**

Figure 3.10. (a) BS ID with H1 = 0.4 dB (b) BS ID with H1 = 6.0 dB

The handover is smoother when the threshold value is increased to 6.0 dB.

Figure 3.11 shows the Traffic sent (packets/sec) by the mobile node over the WiMAX network. We first record the graph for handover threshold hysteresis to be 0.4 dB and then for 6.0 dB. We increase the threshold value to reduce the number of threshold attempts and to get a better network performance. There is a large drop in the traffic sent for small threshold value as compared to the increased value. Some amount of data (almost 15 packets) is lost during handover when H1 = 6.0 dB. But, when H1=0.4, the data lost during handover is very large. The plot for the data being dropped is shown in figure 3.12, for both the handover threshold hysteresis values which shows the difference in the traffic lost during simulation for both the values.

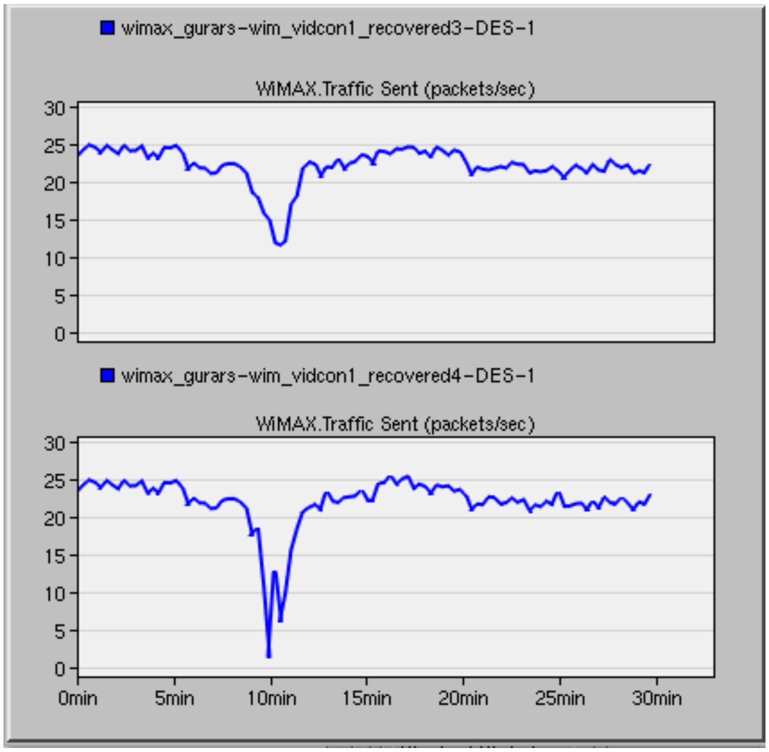


Figure 3.11 Traffic Sent: With H1=6.0 dB (upper plot) and H1= 0.4 dB (lower plot)

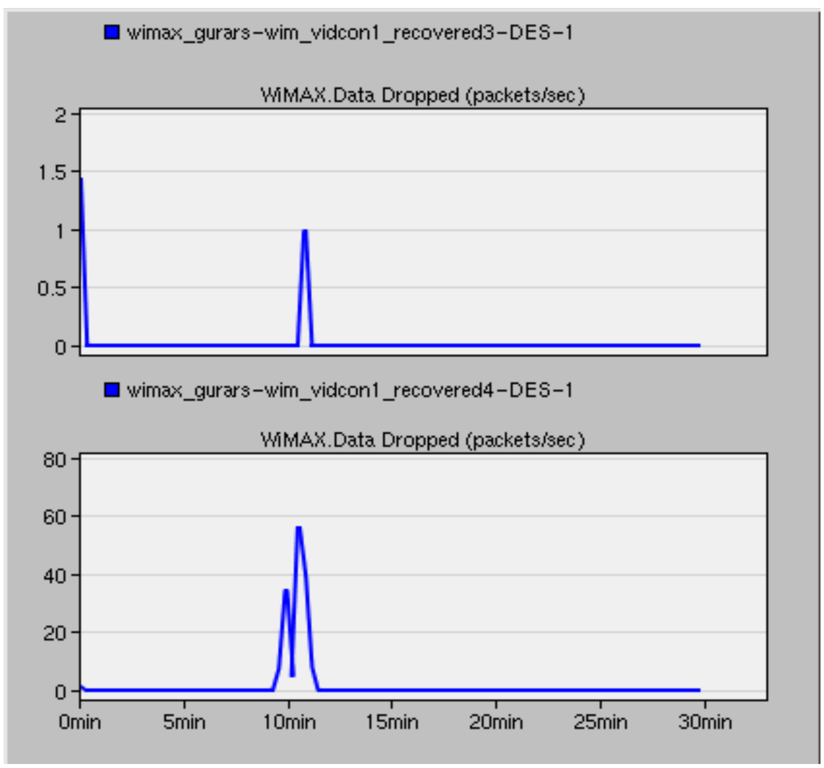


Figure 3.12. Data Dropped: With H1 = 6.0 dB (upper plot) and H1= 0.4 dB (lower plot)

The plots for Delay and Packet Delay variation in video packet transmission are shown in figure 3.13 and 3.14, respectively. The values for delay and jitter are almost constant during the simulation.

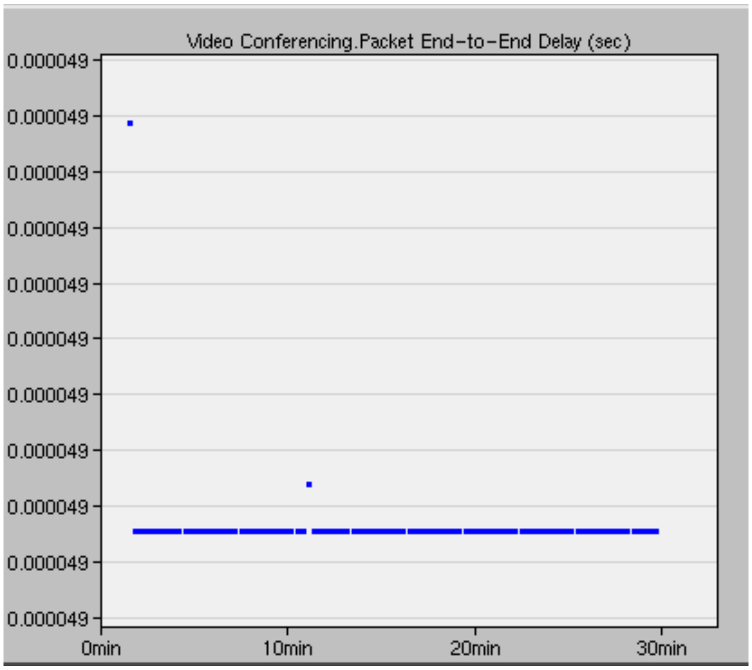


Figure 3.13. Delay (sec)

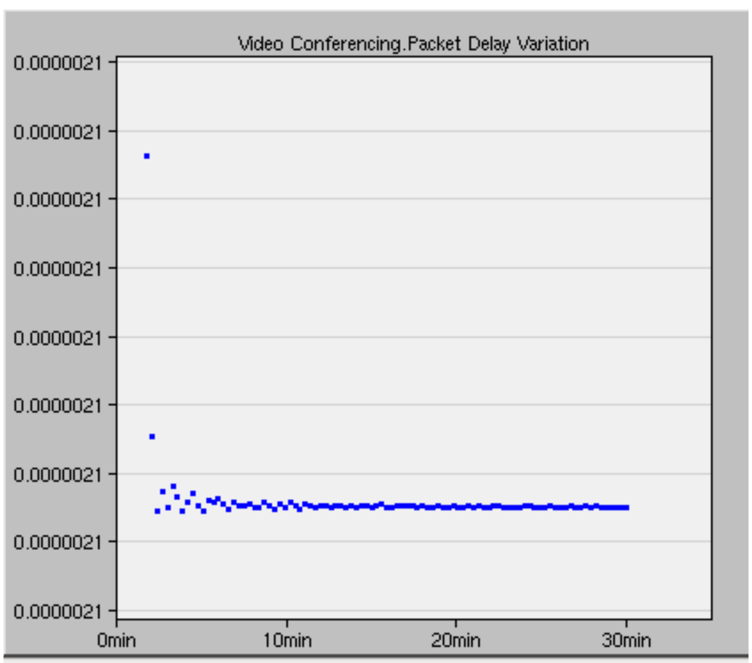


Figure 3.14. Jitter (sec)

3.2.3 Scenario 3: The scenario more cells and increased traffic:

This scenario, shown in figure 3.3 and 3.4, is created to observe the network behaviour when our MS traverses more than two cells. We provide the trajectory such that it begins moving from the cell where BS 4 is giving service and then to BS 1 through BS2. As shown in figure 3.15, the mobile station is first served by BS 4, then 2 and finally by BS1. However, the point to be noted is that when we increase the number of fixed nodes in the area served by BS 2, the mobile station still makes the handover with that node.

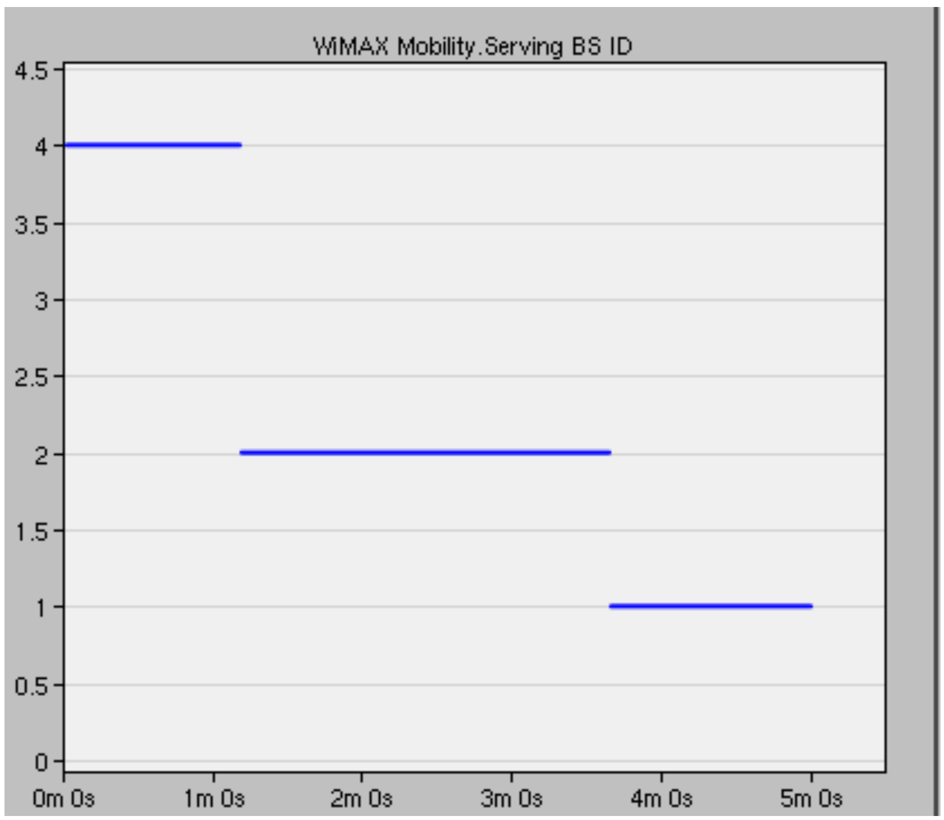


Figure 3.15. Serving BS ID

As we observed better network performance with Handover Threshold Hysteresis value = 6.0 dB, we perform the simulations for this scenario with threshold value 6.0 dB. Further, we observe the amount of WiMAX Traffic sent during the simulation. The graphs shown in Figure 3.16 show the traffic sent to the mobile user. The graph in figure 3.16 (a) is for the scenario when there is a single mobile node traversing the 4 cell network. There is a loss of data when handover takes place from node 4 to2 and from 2 to 1.

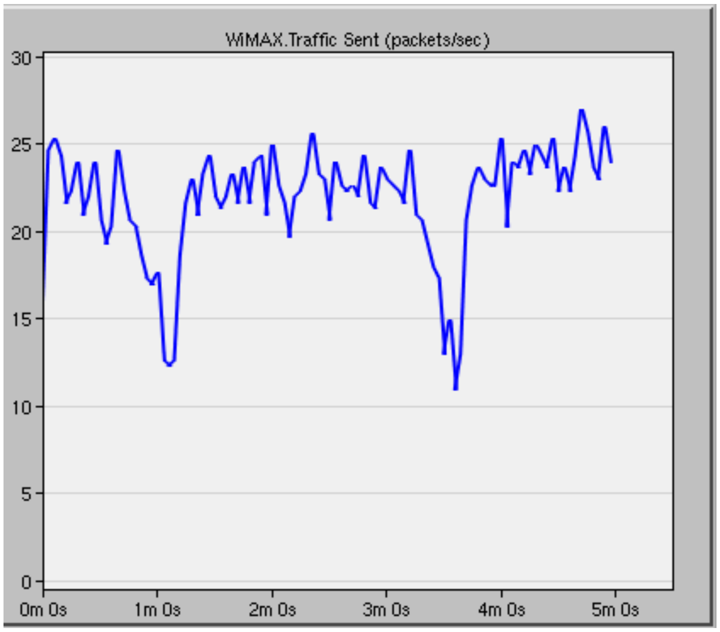


Figure 3.16 (a)

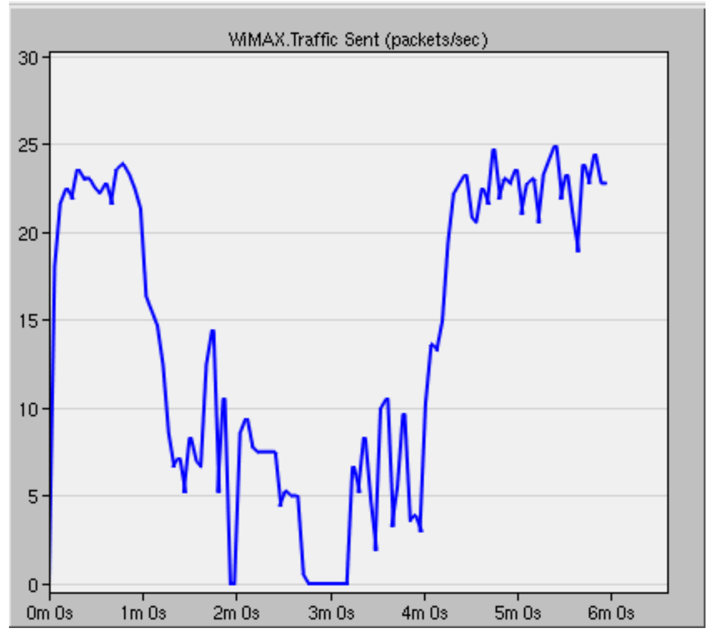


Figure 3.16 (b)

Figure 3.16. (a) Traffic Sent (packets/sec) when a single node moves through the network (b) Traffic Sent in the presence of additional nodes in one cell

Figure 3.16 (b) shows the plot for the traffic sent in the enhanced scenario, where the number of nodes accessing the network increases. There are a large number of fixed nodes in the cell region of BS 2. Since a large number of users are trying to harness the resources, the base station runs out of the available resources and there is a loss of data in that particular cell. Significantly large packet loss occurs in the BS 2 cell. The huge ditch in figure 3.16(b) shows the data loss. Figure 3.17 shown below is indicative of the number of packets dropped during the time when the MS crosses the coverage area of BS 2.

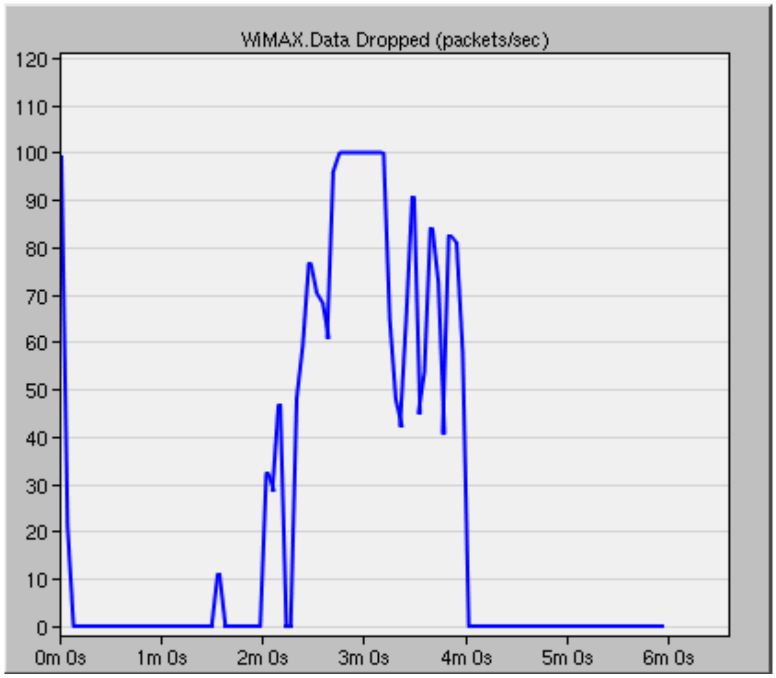


Figure 3.17. Data Dropped (packets/sec)

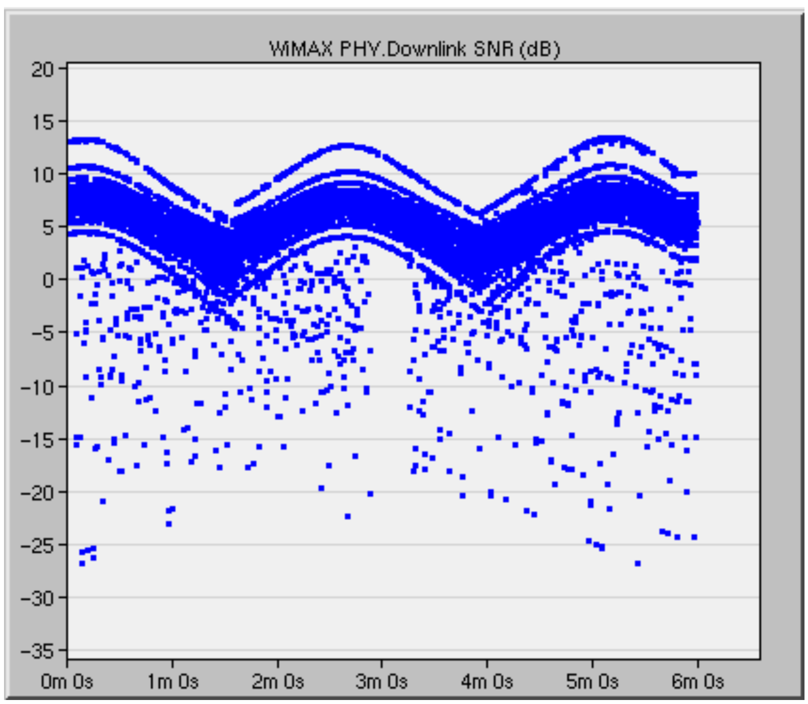


Figure 3.18. Downlink SNR (dB)

The Downlink Signal To Noise Ratio in dB can be seen in Figure 3.18. The effect of handover on SNR can be clearly seen. However, the Signal does not get dropped even during the time when the user is in motion. Thus, effective transmission is possible even in mobile workstations.

Chapter 4.
Conclusion and Future Work

4.1. Conclusion

In this project, we have investigated the characteristics of WiMAX technology and the parameters needed to efficiently implement WiMAX as a wireless technology. We use VoIP and Video content as our traffic source. To be precise, we first set up a topology with fixed workstations and observe the effect and distance from the base station on the performance metrics. The SS located nearer to the BS showed overall better performance. We further added background traffic and noticed the degradation in the performance as compared to the initial scenario.

We also attempt to deploy mobility in WiMAX stations and record the parametric outputs, first for two cell topology and then for four. In the two cell topology, the handover occurs when the MS travels from one cell to another. During this mechanism, we change the threshold value for handover to obtain a smoother handover, with lesser handover attempts. Less loss of data occurs with higher (6.0 dB) threshold value. In the four cell topology, when a large number of users try to access the network, a larger loss in data incurs for the MS travelling through that particular cell which means that there were not enough resources with the base station to facilitate another SS.

We also conclude that WiMAX is an efficient wireless technology suitable to provide service for both fixed as well as mobile users. Additionally, Riverbed Modeler 18.0 provides a suitable environment to deploy the technology and study and modify its performance metrics.

4.2. Challenges

We faced various challenges throughout the project. The lack of license availability for Riverbed Modeler in the initial days was an issue. Also, the software would occasionally hang when Object Palette was accessed during the project simulation, to change the models of the network components.

Also, learning the vast fundamentals and further implementing the WiMAX technology in the restricted time interval proved to be a challenging task.

4.3. Future Work

In this project, we successfully implemented the WiMAX technology for fixed and mobile users. On studying the characteristics of this technology, we foresee the implementation of this technology in major user market.

- We incorporated only VoIP and Video in the simulation but in real time scenario, there would be a large number of applications running simultaneously. This study can be extended for other applications, such as, HTTP, FTP, Email, Browsing, etc.

- The users in our simulation are moving at the speed of 10 km/hr. Effect on performance for high speed users can also be studied.

- The video traffic source uses MPEG-4 coding scheme. However, we can also use other coding schemes as well.

Chapter 5.
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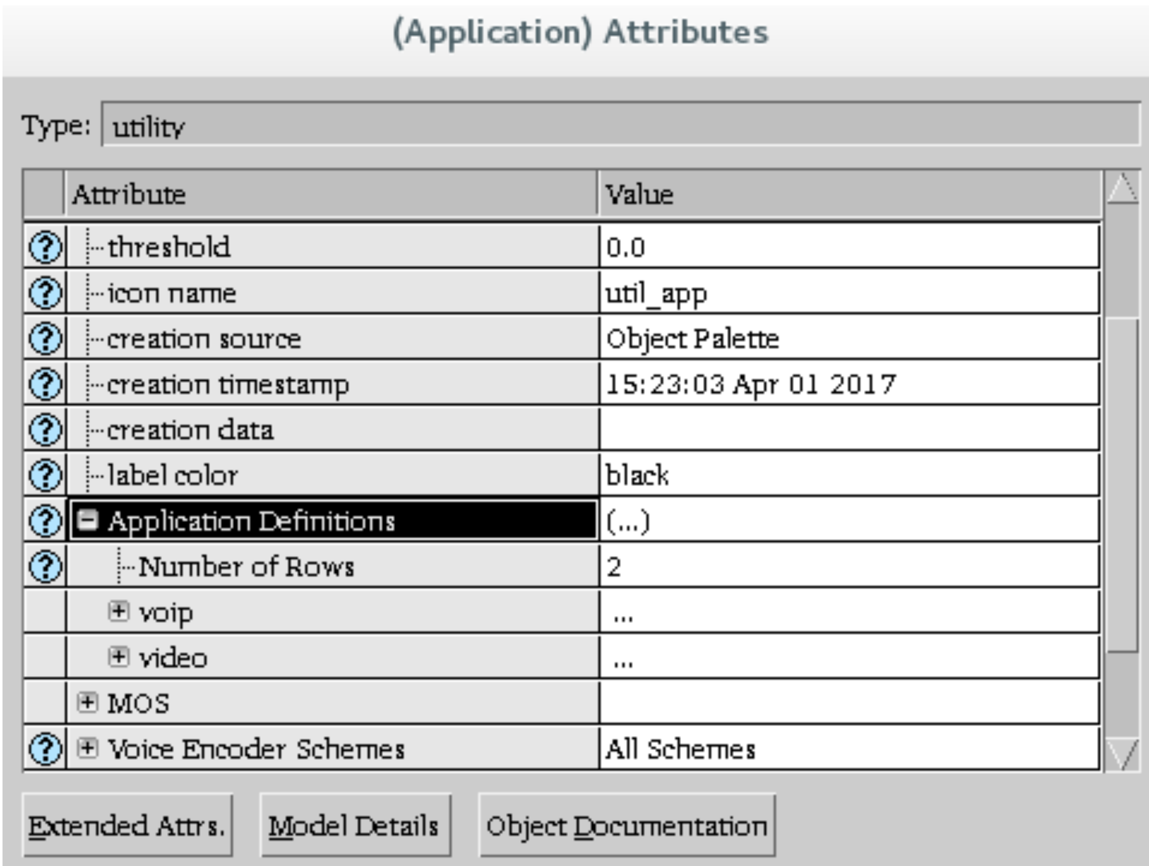
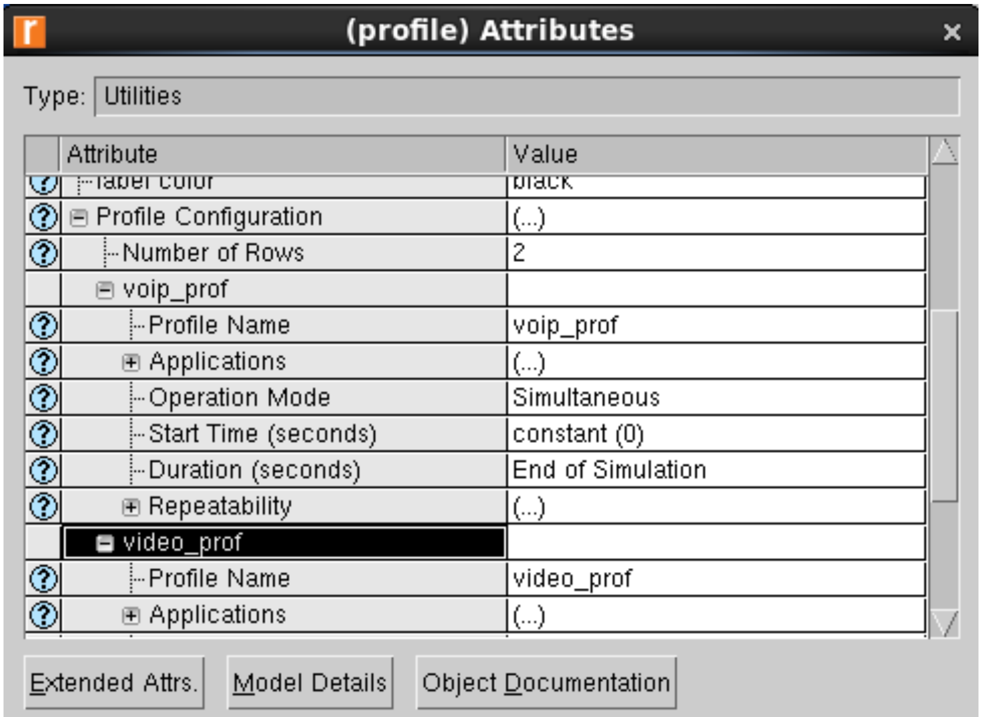
Appendix

Riverbed Modeler Configurations to setup WiMAX

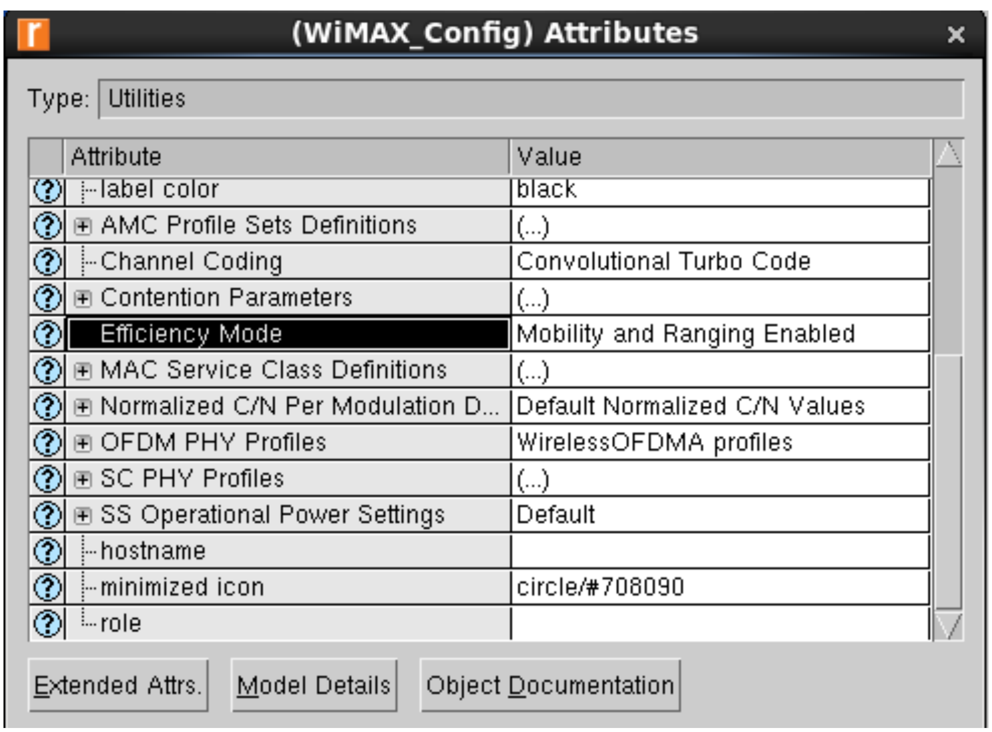
The following parameters were set to deploy WiMAX for our simulations. The parameters are set in Application, Profile and WiMAX Configuration and also for mobile as well as base stations.



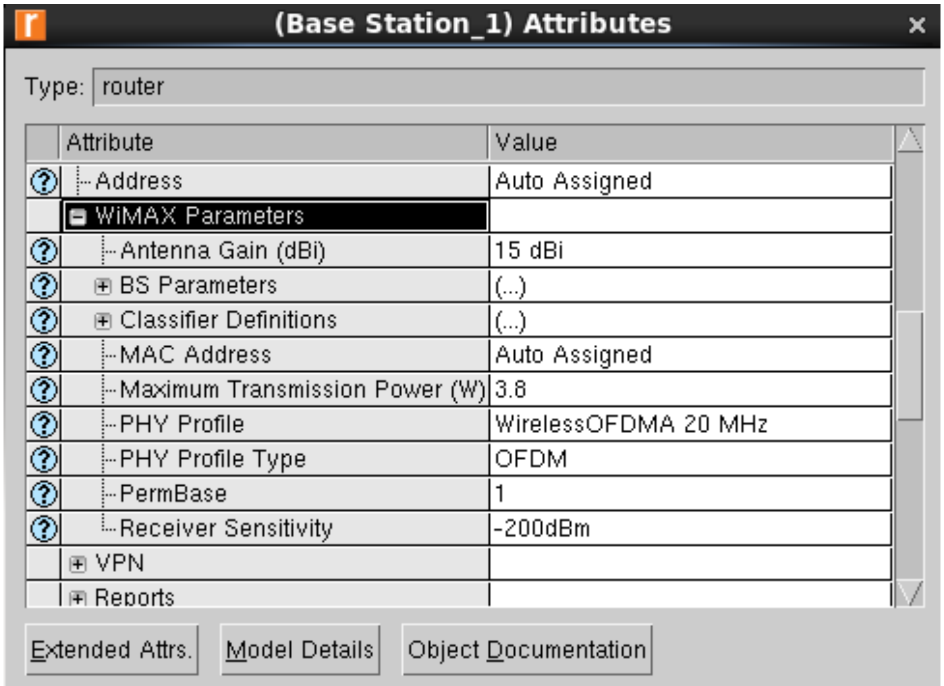
Application Configuration Attributes



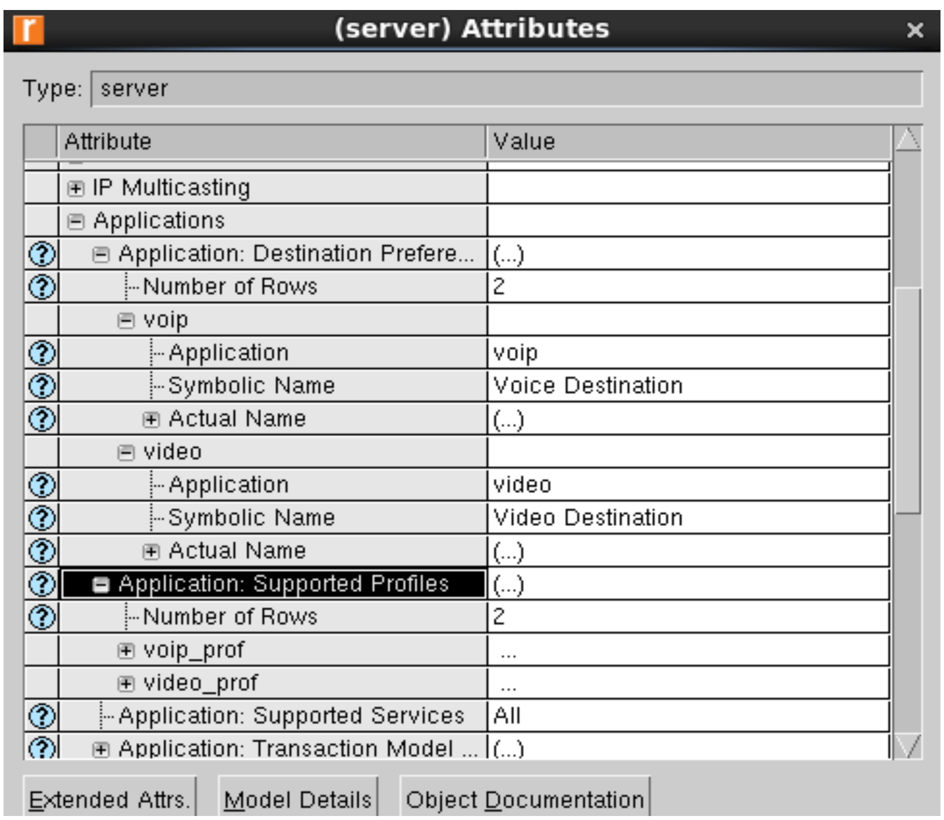
Profile Configuration attributes



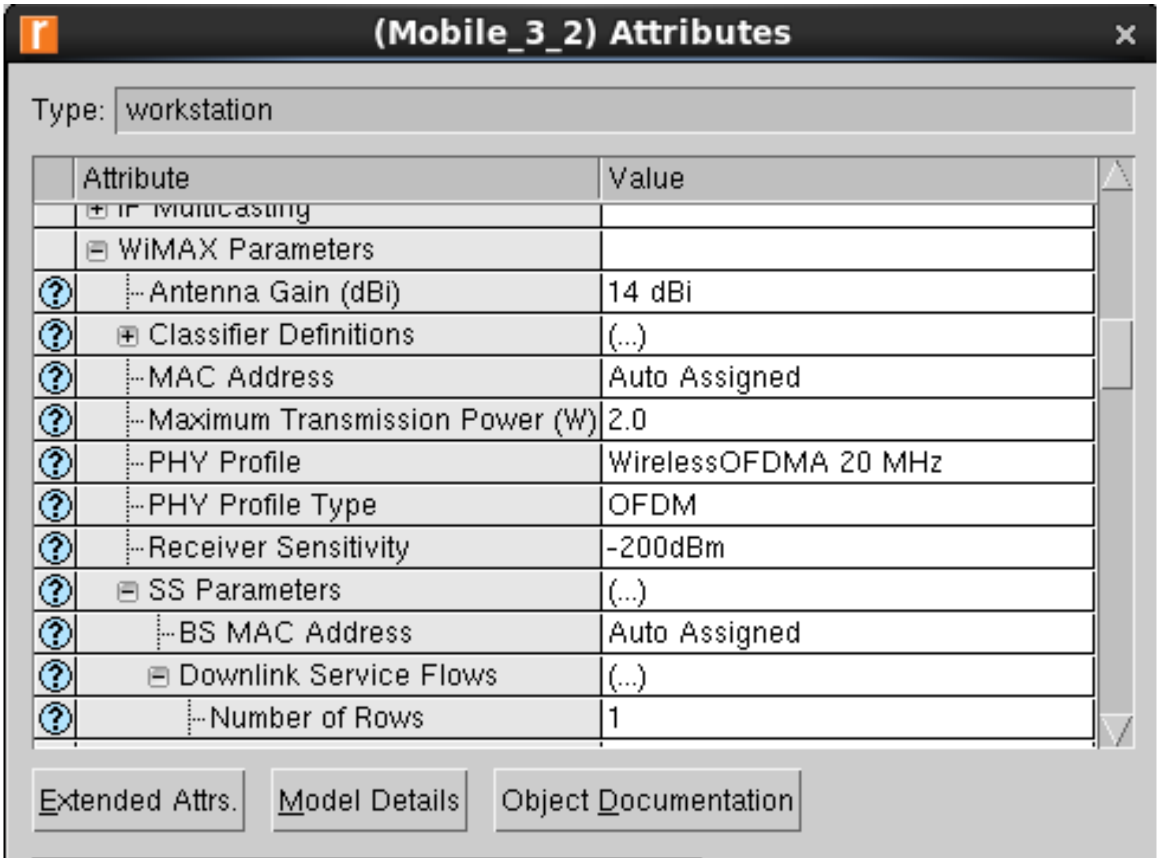
WiMAX Configuration Attributes



Base Station Attributes



Server Configuration Attributes



Mobile Station attributes