ENSC-835: Communication Networks

Comparison of WiMAX and ADSL by Streaming Audio and Video Content

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Final Project

Team-2

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1. Abstract

The IEEE 802.16 standard is known as WiMAX (Worldwide Interoperability for Microwave Access). Now a day this protocol is widely used for fixed and mobile Internet access. WiMAX provides a theoretical maximum data rate of 75 Mbps. On the other hand, Asymmetric Digital Subscriber Line (ADSL) embodies the RFC 2662 is widely used connection and provides guaranteed service. However characteristics of ADSL limit the even distribution of fixed broadband services. WiMAX has surfaced to substitute ADSL, which is designed to provide high-speed Internet access to a wide range of devices such as laptops, cell phones, cameras, music players, etc. which are being used by clients over the last mile.

In this project an attempt has been made to compare the performance of WiMAX and ADSL by streaming audio and video content. Standard and custom applications like File Transfer Protocol (FTP), Hyper Text Transfer Protocol (HTTP), and Electronic mail (Email) have also been used for this comparison. OPNET Modeler 15.0 and OPNET Modeler 16.0 are used to simulate networks. Analysis is intended for four performance matrices: Packet loss, Delay, Jitter, and Throughput to determine whether WiMAX can give performance comparable to ADSL for all applications. In order to determine the performance of WiMAX more comparable to ADSL, under differing conditions, simulations were run with varied values of parameters like buffer size.

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2. Introduction

The IEEE originally formed the IEEE 802.16 working group in 1998 to provide a standard for wireless metropolitan area networks. The primary application was for high-speed fiber access solutions using high frequency line of sight (LOS) fixed wireless connections. The original standard was referred to as 802.16 and evolved to support fixed broadband wireless access over lower frequency non line of sight (NLOS) wireless connections. The evolved standard, 802.16-2004, is often referred to as fixed WiMAX. WiMAX technology is based on the IEEE 802.16 standards. In particular, the current Mobile WiMAX technology is mainly based on the IEEE 802.16 amendment (IEEE, 2006a), approved by the IEEE in December 2005, which specifies the Orthogonal Frequency Division Multiple Access (OFDMA) air interface and provides support for mobility. With such emerging growth, it is reasonable that the technical community seek and quantify application performance across these dissimilar access technologies using a bandwidth intensive application load such as streaming video to understand any potential tradeoffs by moving to WiMAX [1].

WiMAX Broadband Access meets the performance of ADSL broadband access for streaming video/audio applications, HTTP, FTP, and Email by identifying four metrics to measure the resulting video transmission performance over these access networks.

- Video packet loss
- End-to-end delay
- Video packet jitter
- Throughput

Video/audio streaming is gaining wider adoption in the Internet community. Unmanaged services refer to Internet services that have little control over the end-to-end performance between the subscribers and corresponding services. This project is designed around streaming services using an Internet topology on expected video/audio performance.
The simulation model incorporates actual movie video/audio trace. Specifically, it will stream the Matrix III movie [15] for a 30-minutes interval to three WiMAX client stations and one ADSL client from video/audio content services provider on the Internet.

2.1 Motivation

As the number of Internet hosts, offered services, router switching speeds, and link transmission capacities, continue to increase, multimedia rich applications such as video streaming are gaining wider adoption in the Internet community. Media providers are exploring new and innovative applications over core Internet Protocols (IP) networks.
Observing the demand of the WiMAX technology in the real world, studying and comparing this technology with the existing technologies, seemed evident.
This project enhanced the previous work [1] that employed the OPNET Modeler to compare the performance of fixed WiMAX and ADSL access technologies by streaming Motion Picture Experts Group 4 (MPEG-4) video content, to several WiMAX and ADSL client stations. This model was developed using OPNET Modeler 12.0 and 14.5.A.

The previous model was upgraded to OPNET version 15.0.A and then to version 16.0. In the previous model the Matrix III movie video traces were added as data traffic. In the upgraded model Matrix III movie audio trace, HTTP, FTP, and Email were incorporated.
Ultimately, the objective of this project was to gain greater insight and clarity into fixed WiMAX system performance using emerging, load intensive and delay sensitive Internet Protocol television (IPTV) technology.

3. Background Knowledge

3.1 Video Content Overview

Video content consists of both the audio and the visual information. This information is available in media service providers; like newscasts, sporting events, movies in real time, video on demand (VoD) formats, and wide range of sitcoms. This is known as real-time multimedia services over the Internet. Real-time transport of live video or stored video is the significant part of real-time multimedia. This project focuses on video streaming, which refers to real-time transmission of stored video. There are two modes for transmission of stored video over the Internet: the download mode and the streaming mode (video streaming). In the video streaming mode, it is not essential to download the full video content, but it is being played-out while parts of the content are being received and decoded. As it is real-time, video streaming has bandwidth, delay and loss requirements [15].
For video streaming the video content is organized as a sequence of frames or images that are sent to the subscriber and displayed at a constant frame rate. The video component is coupled with a multi-channel audio component that is also structured as a series of audio frames which is included in the video content. While streaming real-time video there are different transmission and buffering requirements from the network and the client station video player. The video content may be characterized by several parameters including video format, pixel colour depth, coding scheme and frame inter arrival rate. Due to these characters the raw video size becomes very large, which affects transmission and buffering requirements from the network. To reduce their traffic load requirements, streaming services encode uncompressed content using MPEG-x and H.26x codecs. While these encoded streams are marginally loss-tolerant, their performance is inherently a function of available link bandwidth and delay characteristics.
Video frame inter-arrival rates can range from 10 frames per second (fps) to 30 fps. This parameter can be especially critical as network conditions can influence the frame inter-arrival rates and which if left uncompensated, significantly degrades the video playback quality. Figure 1 illustrates the necessity of the client video system to playback frames at a constant rate amidst variable delays in video frame packet arrivals [1].

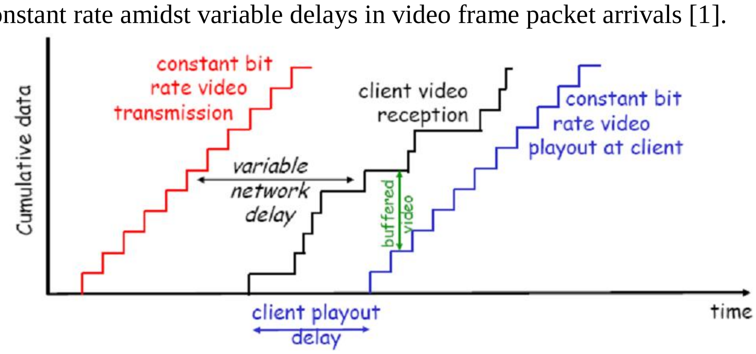


Figure 1: Video client buffering [1]

Figure 2 illustrates architecture for video streaming. The raw video and audio data are pre-compressed by video/audio compression algorithms and then saved in storage devices. Upon client's request, a streaming server retrieves compressed video/audio data from storage devices and then the application-layer quality of service (QoS) control module adapts the video/audio bit-streams according to the network status and QoS requirements. After the adaptation, the transport protocols packetize the compressed bit-streams and send the video/audio packets to the

Internet. Packets may be dropped or experience excessive delay inside the Internet due to congestion. For packets that are successfully delivered to the receiver, they first pass through the transport layers and are then processed by the application layer before being decoded at the video/audio decoder. To achieve synchronization between video/audio presentations, media synchronization mechanisms are required [15].

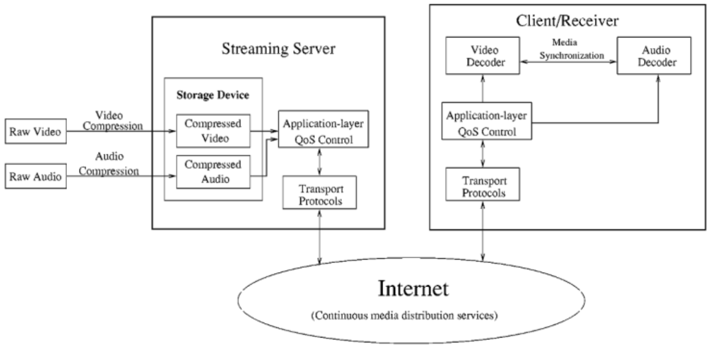


Figure 2: Architecture of video streaming [15]

Figure 3 shows the protocol stack for streaming video services includes the Real Time Protocol (RTP) that provisions a packet structure for video/audio data above the transport layer protocol. RTP specifies a twelve-byte header with protocol fields to describe the type of content being carried (MPEG-4), packet sequencing, and time stamping. Since RTP resides on top of the transport protocol, it is deployed in the end-systems rather than in the network core. RTP does not provide mechanisms to guarantee bandwidth or packet delays [1]. But it provides services like time-stamping, sequence numbering, payload type identification, source identification [15].

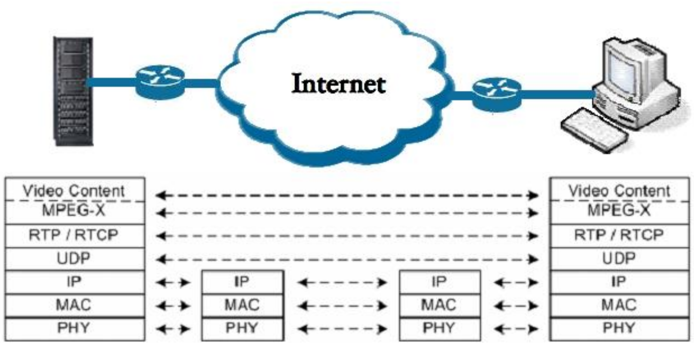


Figure 3: Video streaming network topology

Below the RTP layer, usual streaming services utilize the User Datagram Protocol (UDP) because it provides best effort service without delay, loss, or bandwidth guarantees. UDP is connectionless, unreliable and it does not provide flow control or congestion control. The lack of reliability and congestion control mechanisms are desirable properties in media content streaming because video servers can stream their content at the native video/audio source encoding rates without being constrained by congestion control when packet loss occurs. UDP segments are then encapsulated into unicast (or multicast) IP packets for proper addressing and routing to the video client stations. IP packets can be lost due to router buffer overflows or delayed due to router congestion, which impacts the client station playback rate as outlined earlier. IP packets pass through appropriate media access control (MAC) and physical (PHY) layers and then propagate through the Internet and access networks, which can be wired or wireless, to the client subscribers. Subscriber stations buffer, decompress and playback the video/audio frames at a constant rate.

By observing communication performance between the server and the client, four performance metrics with appropriate thresholds may be used to measure streaming performance. Furthermore, these metrics enable comparisons between WiMAX and ADSL connected clients because they access the same VoD services over the same wired network infrastructure. The performance metrics are:

- Packet loss
- Delay
- Jitter
- Throughput

3.2 WiMAX Overview

In recent year, the rapid growth of wireless communication technology improves the transmission data rate and communication distance. WiMAX, based on the IEEE 802.16 (PHY and MAC layers), is one of the emerging technologies of broadband wireless system. Its transmission rate and distance can reach up to 75 Mbps and 50 km. Compared with other wireless networks, WiMAX has the virtues of higher transmission speed and larger transmission coverage. It can solve the last mile problem of the metropolitan network because of the features of high bandwidth and long distance [17].

WiMAX operates in the 10–66 GHz band with LOS communications using the single carrier (SC) air interface. The IEEE 802.16a standard outlined NLOS communications in the 2 – 11 GHz band using one of three air interfaces: SC, Orthogonal Frequency Division Multiplex (OFDM), and OFDMA. OFDM and OFDMA enable carriers to increase their bandwidth and data capacity. This increased efficiency is achieved by spacing subcarriers very closely together without interference because subcarriers are orthogonal to each other. Channel bandwidths range between 1.25 MHz and 20 MHz in the 2 – 11 GHz band. Within this channel bandwidth by allocating various sub-carriers and using various modulation schemes a data rates between 1.5 to 75 Mbps are achievable for WiMAX [1].

Figure 10 displays generic topology of WiMAX network. The WiMAX network, which consists of Base Station (BS) and Subscriber Station (SS), has two transmission modes of network topology, one is Point to Multi Point (PMP) and the other one is mesh [17]. WiMAX is able to achieve QoS by using a bandwidth request and granting scheme on the subscriber stations. This prevents the WiMAX base station from over-subscribing its available resources [1].

Cell sizes in WiMAX systems typically have radii between 7 km and 10 km. While WiMAX has numerous applications, including wireless backhaul links for Wi-Fi hot spots and redundant wireless Internet backup links for commercial businesses, this study focuses on WiMAX as an alternate access network technology to ADSL. It enables residential and commercial subscribers either outside ADSL service regions or in densely overloaded ADSL regions to attain high speed Internet access.

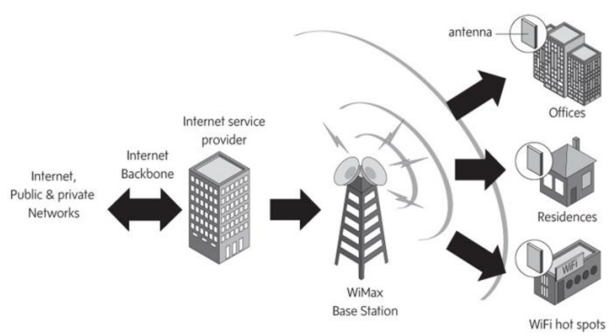


Figure 4: Generic topology of WiMAX network

4. Model Design

Will Hrudey's model was used as the reference model for the new proposed model. Some additional changes to reference model were made but the basic structure is the same. OPNET Modeler version 15 and 16 were used as simulation tool and the final results were in OPNET version 16.
The scenarios can be categorized as MPEG-4 video, audio HTTP, FTP, and Email streaming scenarios. Two different buffer sizes (i.e. 128KB, 1024 KB) were implemented in the base station. And the simulation time was 30 minutes.

4.1 Network Topology

Figure 5 captures the network topology consisting of geographically separated client and services subnets.

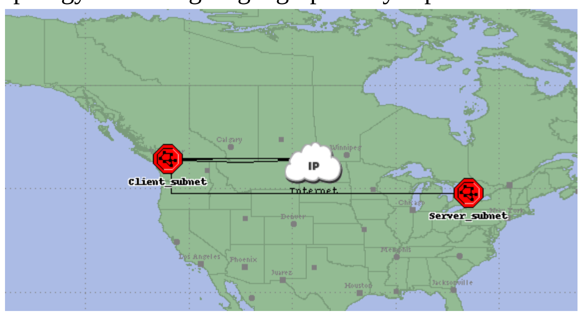


Figure 5: Simulation model network topology

The server subnet shown in Figure 6 is located in Toronto and it provisions a VoD server capable of streaming stored audio, video, HTTP, FTP, Email contents to clients on request. This subnet reflects a basic corporate architecture where the video server resides on a 100Mbps IP network behind a firewall. The firewall's outside interface connects to an access router which is connected to the Internet via a 45 Mbps DS3 wide area network (WAN) link. Additionally, the local video client was utilized for initial troubleshooting and traffic validation purposes; however it was not used in the formal simulation scenarios.

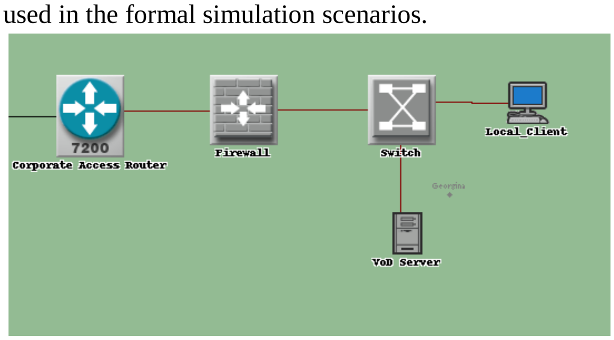


Figure 6: Video services subnet

The video client subnet is located in Vancouver and encompasses four video client stations that will access the same VoD services from Toronto. In this subnet, three fixed wireless WiMAX stations are located 2, 4, and 6 km from the WiMAX base station. The base station is subsequently connected to the Internet via a DS3 WAN link. The fourth video client is an ADSL station located 5 km from the carrier's central office and serves as the baseline reference to which WiMAX stations will be compared against.

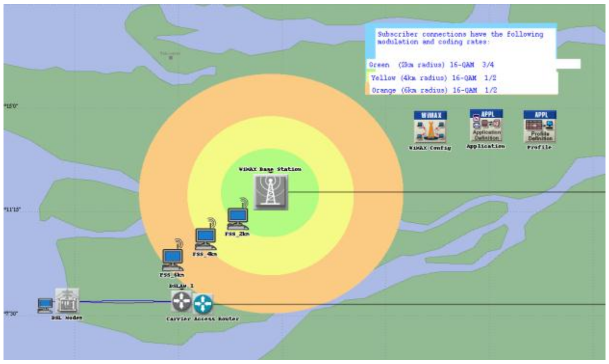


Figure 7: Simulation model video client subnet

Both subnets are connected to the Internet via DS3 WAN circuits. The approximate distance between the two subnets is 3342 km which equates to approximately 13.3 ms propagation delay. The Local Area Network (LAN) and WAN links were configured with alternating 10 and 20% utilization loads over 30 minute intervals.
Moreover, the Internet "cloud" was configured with a packet discard ratio of 0.001% which results in one packet out of every 100,000 packets is dropped in the Internet. The Internet also introduces 1 ms delay in addition the propagation delays noted on the WAN links.

4.2 WiMAX Configuration

The WiMAX specific configuration involved the following areas:

- Service Class / Service Flows
- MAC scheduler
- Burst profiles
- Air Interface
- Operating Frequency
- Channel bandwidth and subcarrier allocation
- Transmit power
- Pathloss model

In WiMAX, a service class captures the QoS requirements of service flows where service flows represent traffic flows between the base station and the subscriber stations. Service flows from the base station to the subscriber station are termed downlink flows and service flows from the subscriber station to base station are termed uplink flows. For a given service class, the key parameters are minimum sustainable data rate, which is minimum guaranteed over the air (OTA) rate, as well as the MAC scheduler type.
The MAC scheduling facility allows WiMAX to provide QoS capabilities, thereby supporting delay sensitive traffic like voice and video services. There are four scheduler types:

- UGS (ungranted service)
- rtPS (real time polling service)
- nrtPS (Non real-time polling service)
- BE (best effort)

The available bandwidth resources are allocated to UGS first, then to rtPS and nrtPS flows. Lastly, any remaining resources are then assigned to BE flows.
For this project, one service class was created for the downlink using BE scheduling and 3.0 Mbps minimum sustainable data rate. Another service class was created using BE scheduling and 640 kbps minimum sustainable data rate. Subsequently, the base station and WiMAX subscriber stations were configured to map the uplink and

down link service flows to a specific type of service (ToS) setting that was configured during the application node configuration. Moreover, each service flow (uplink and downlink) can be configured with specific burst profile. For this study, the uplink channel was assumed to have similar properties to the down channel so for a given WiMAX station, the same burst profile was used on both the uplink and downlink service flows. Figure 8 shows WiMAX configuration attributes.

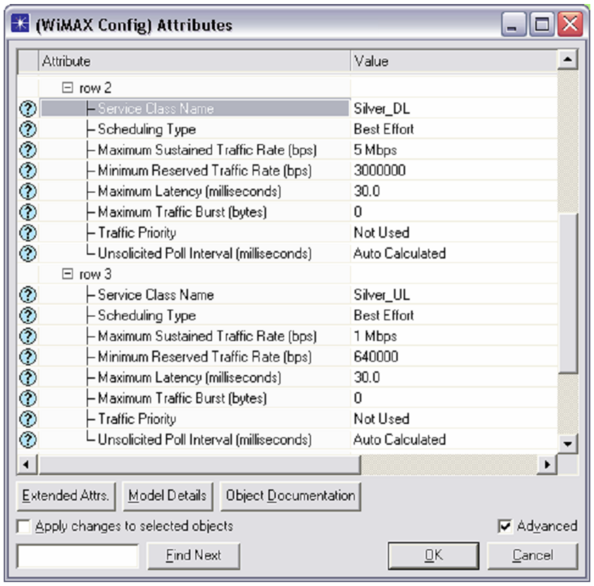


Figure 8: WiMAX service class configuration.

WiMAX client stations were manually configured with more robust modulation/coding schemes with increased distance from the base station. Table 1 detailed the available coding rates for a given modulation scheme as well as the minimum SNR.

Modulation	Coding	Information Bits/symbol/Hz	Required SNR (dB)
QPSK	1/2	1	9.4
	3/4	1.5	11.2
16-QAM	1/2	2	16.4
	3/4	3	18.2
64-QAM	2/3	4	22.7
	3/4	4.5	24.4

Table 1: Modulation / coding rates

Initially 64- Quadrature Amplitude Modulation (QAM) scheme was configured for the 2 km fixed subscriber station (FSS), but the SNR at 2 km from the base station was below acceptable levels and the resulting performance was poor. Consequently, a more robust scheme was configured at the expense of lower transmission efficiency. The 2 km FSS modulation and coding rates for both uplink and downlink service flows are shown in Figure 9.

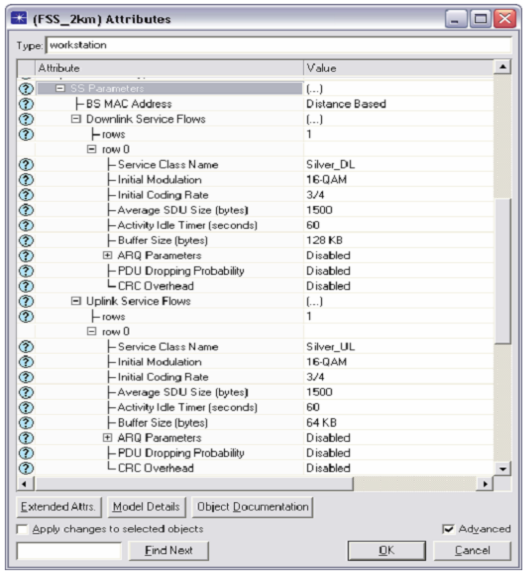


Figure 9: FSS service flow modulation and coding rates.

The air interface or PHY layer access was configured to utilize OFDM on a 2.5 GHz base frequency using a 5 MHz channel bandwidth which provisions 512 subcarriers allocated in the following manner in Table 2.

Frequency Division		
	DL Zone	UL Zone
Number of Null Subcarriers - Lower Edge	46	52
Number of Null Subcarriers - Upper Edge	45	51
Number of Data Subcarriers	360	272
Number of Subchannels	15	17

Table 2: PHY layer frame division pattern

The WiMAX client station transmit power was configured to use 33 dBm (2 watts) of transmit power over the 5MHz channel bandwidth using 14 dBi gain antennas. The base station transmit power was configured to 35.8 dBm (3.8 watts) with 15 dBi gain antenna.

4.3 ADSL Configuration

The ADSL configuration employed in this model was representative of an "enhanced" subscriber package with a 3.0Mbps downlink channel and a 640 kbps uplink channel. The modeled distance between the subscriber and the central office was 5 km.

4.4 Traffic

Traffic is a key aspect of this project. In this project with video traffic which was the only traffic of the reference model, other traffics like audio, HTTP, FTP, Email were added. All this traffic will stress the access links to a much further extent. This will help us to observe the performance matrices from more realistic prospective comparing the real world.

The video/audio traffic source was a 2-hour MPEG-4 Matrix III movie trace which utilized a 352x288 frame format resolution and a 25 fps encoding rate. For HTTP, FTP, and Email traffic both the application attribute and the server was configured for heavy load traffic.

4.5 OPNET MODELER

What is OPNET?

OPNET Technologies provide IT service assurance solutions and associated professional services. The company was founded in 1986 and went public in 2000. OPNET Modeler is the main product developed by the company. There is some free simulation software in the world like ns-2 other than OPNET. But, this software is one of the most popular, accurate and applicable in the real world in the field of network simulation and is recognized for its high reliability. Therefore, many laboratories, public institutions, and companies involved in information and communications prefer and employ this software.

The OPNET Modeler provides network and application management software and hardware. It allows for the simulation of different scenarios for a specific project and uses a project and scenario approach to modeling networks. The project approaches a collection of related network scenarios in which each explores a different aspect of network design. It contains at least one scenario, that is, a single instance of a network. Simulating a scenario can overcome constraints of proprietary hardware and software such as lack of development tools. The OPNET Modeler offers to its user a GUI interface, standards-based LAN and WAN performance modeling and detailed library model for most protocols and devices. Figure 10 captures the example of a scenario created in OPNET 16.0.

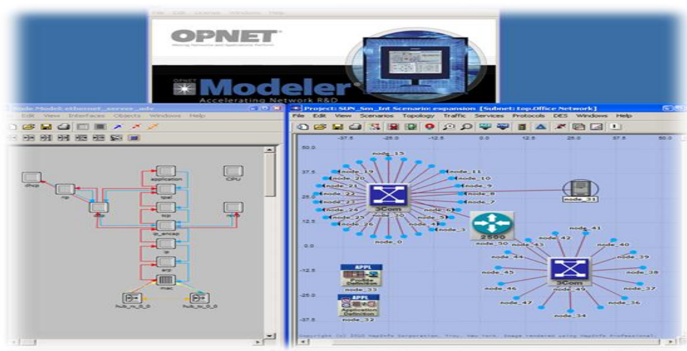


Figure 10: Example of a scenario created in OPNET 16.0

In the previous work of Will Hrudey, the video application in fixed WiMAX has been addressed but audio for fixed WiMAX was not mentioned. In order to configure the video, Will Hrudey's OPNET model is used as a reference and the attributes are edited accordingly. In order to add audio, HTTP, FTP and Email applications following steps have been used.

5. Contribution

5.1 Audio Traces

Processing Audio Traces

The audio traces data required pre-processing before they could be imported into OPNET Modeler video conferencing application (VCA) traffic data. The frame sizes need to be extracted from the audio traces and then converted from bits to bytes. The final result has to be saved as a .csv compatible format file. The output file needs to be saved into the OPNET models folder.

OPNET Setup

Configuration of Application definition tool in client subnet:

In the application definition tool, one application row has been added and it is named audio. Figure 11 shows editions in video conferencing attribute.

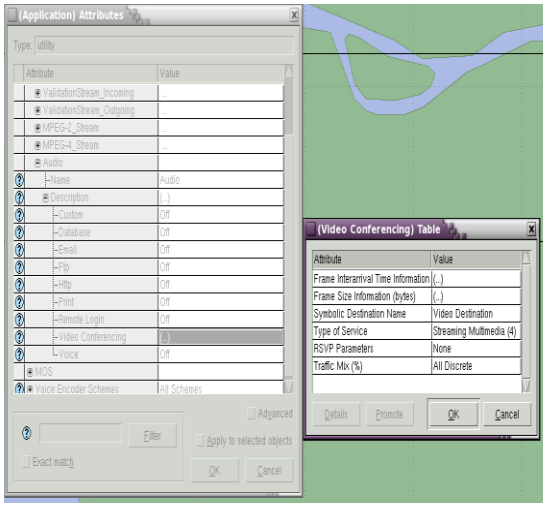


Figure 11: Video conferencing attribute table

Figure 12 shows edition of Frame interarrival time information field.

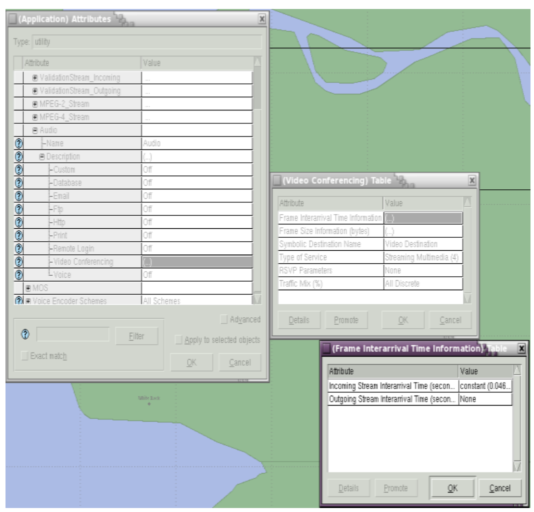


Figure 12: Frame interarrival time table

Selecting "edit" for frame size information (bytes). Then selecting "edit" for incoming and outgoing stream. Setting the distribution name as scripted for both incoming and outgoing stream, then entering the processed matrix-audio (without .csv extension). Above steps are shown in Figure 13 and 14.

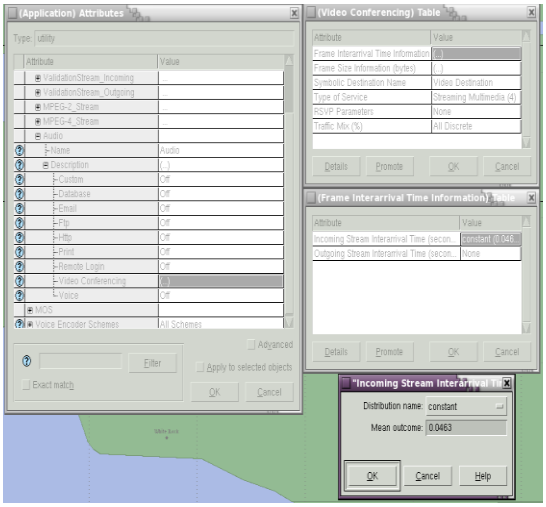


Figure 13: Frame information size table

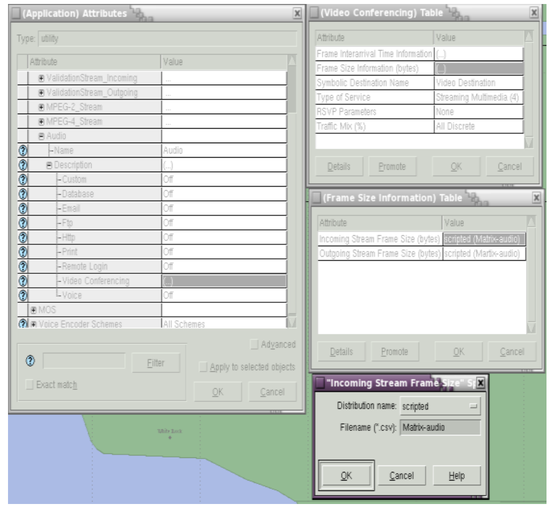


Figure 14: Incoming stream frame table

Selecting streaming multimedia 4 as type of service is displays in Figure 15.

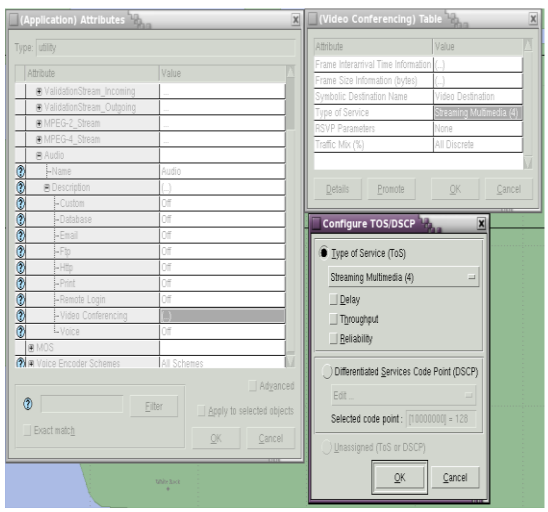


Figure 15: Video conferencing table

Configuration of Profile definition tool in client subnet:

In the Profile definition tool, one application row has been added and it is named audio. This is shown in Figure 16. Profile name is also set as audio.

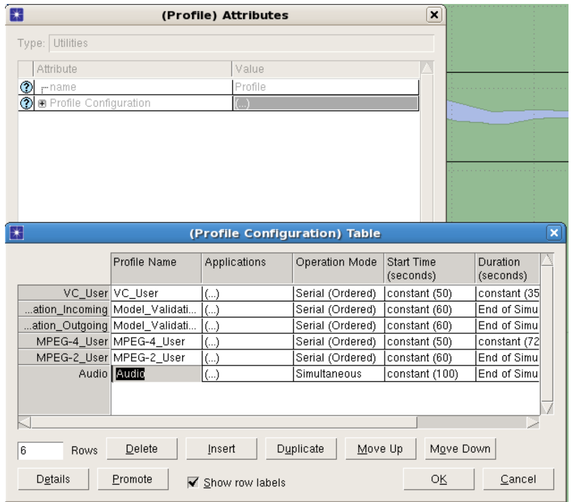
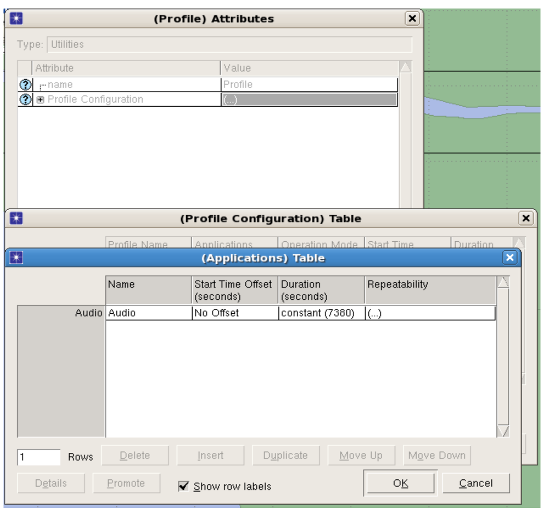


Figure 16: Profile configuration table

Editing of applications field and repeatability field is shown Figure 17 and Figure 18 respectively.



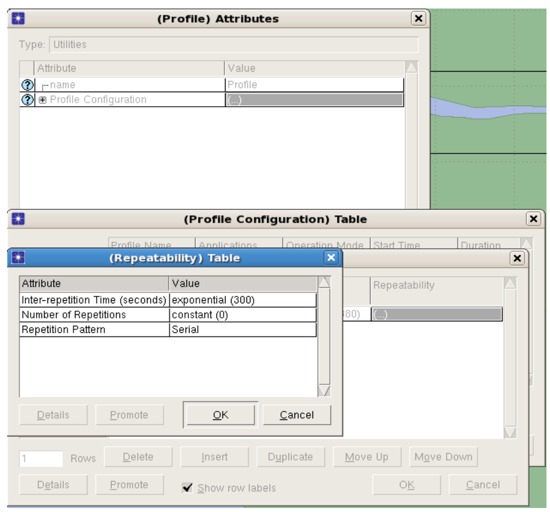
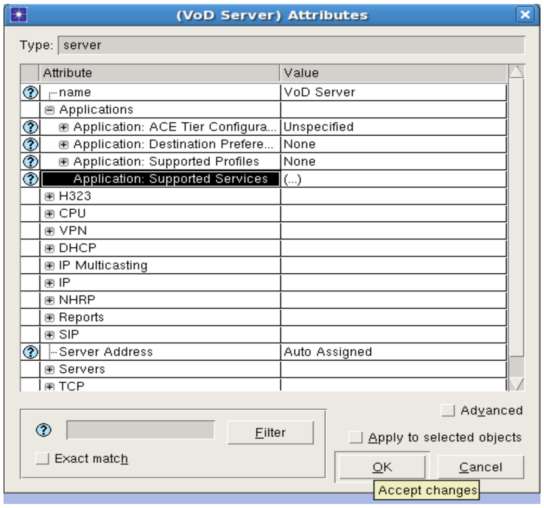


Figure 17: Application table

Figure 18: Repeatability table

Configuration of video server in server subnet:

Same server is used to configure both audio and video. In VoD server attributes, edit option of application supported services and add one more row for audio application in application supported table as shown in Figures 19 and 20.



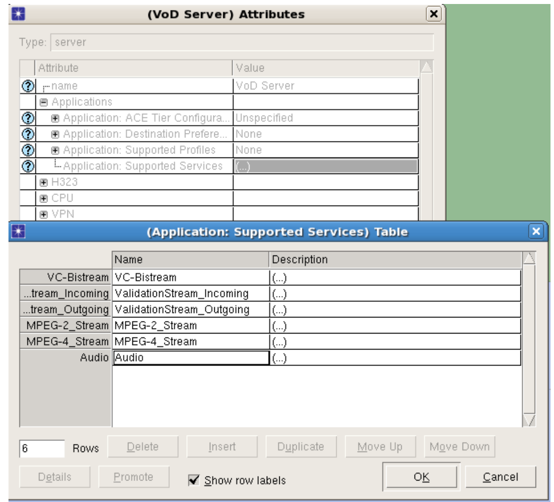


Figure 19: VoD server attributes table

Figure 20: Application supported services table

After adding row for audio application, edit description table as displays in Figure 21.

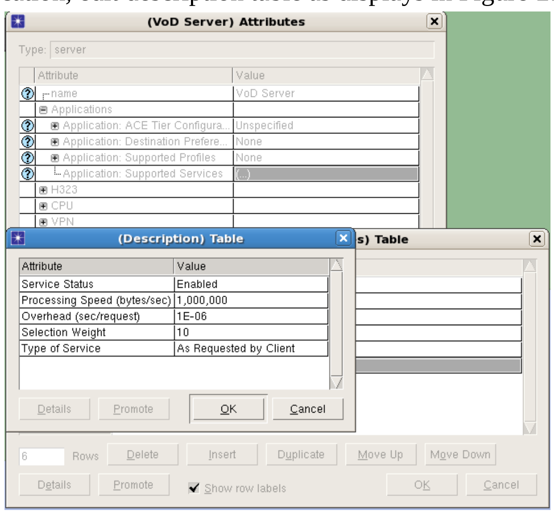


Figure 21: Description table

5.2 HTTP, FTP and Email

OPNET Setup

Configuration of Application definition tool in client subnet:

In the application definition tool, add three more application rows and named them HTTP, FTP and Email respectively. Edit attributes according to Figure 22.

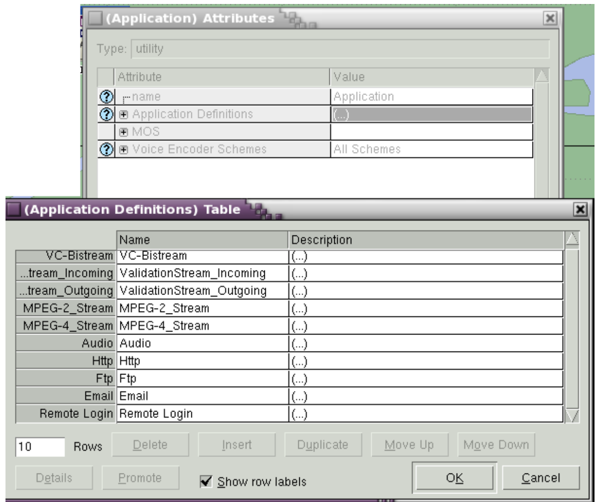


Figure 22: Application definitions table

Configuration of Profile definition tool in client subnet:

In the profile definition tool, three more application rows are added and these applications are named HTTP, FTP and Email respectively. Editions in the attributes are according to Figure 23.

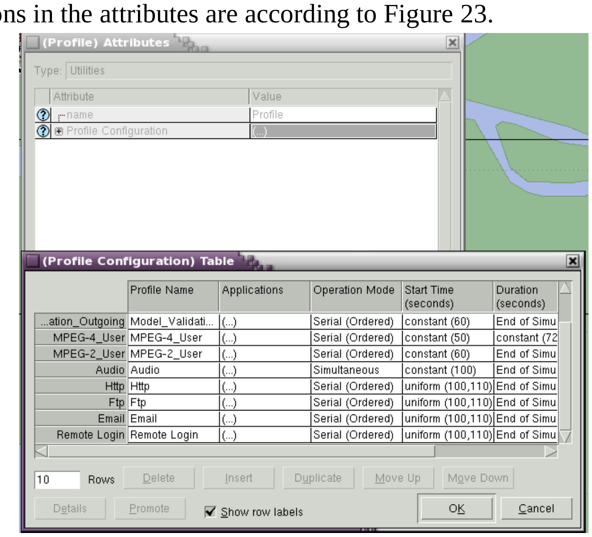


Figure 23: Profile configuration table

Configuration of subscribers in client subnet:

In the client subscriber stations (2 km, 4 km, 6 km and ADSL), three more rows are added to application supported profile table and these applications are named HTTP, FTP and Email respectively. Editions in the attributes are according to Figure 24.

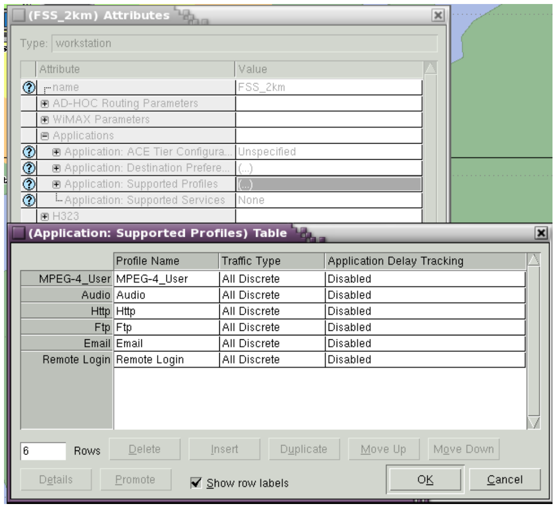


Figure 24: Application supported profiles table

Configuration of video server in server subnet:
Same VoD server is used to configure HTTP, FTP and Email. In VoD server attributes, add three more rows to application supported services and named these rows as HTTP, FTP and Email respectively. Edit attributes according to Figure 25.

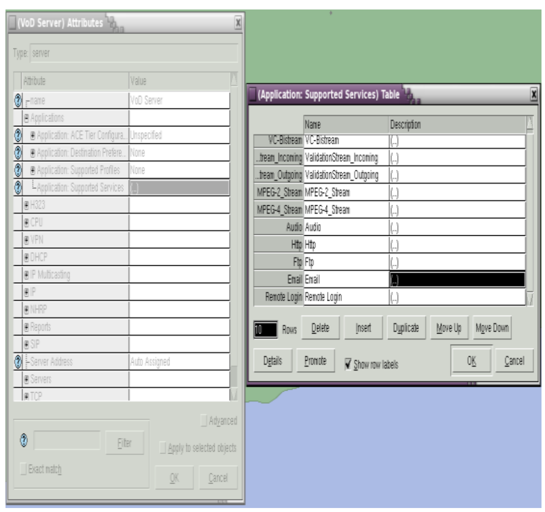


Figure 25: Application supported services table for Email

6. Model Validation

For model validation, comparison of global statistics of new model with is done with the global statistics reference model. Analysis of network jitter, throughput, and traffic received is done for both the models.

In the reference model, only video streams were added as traffic to the simulation. But for new model we have added audio streams, HTTP, FTP, Email traffic to the simulation. So the traffic increased remarkably. This increase can be seen in the simulation graphs.

Though, reference model was developed in OPNET version 14.5.A. For validation purpose simulations for the reference model were done in OPNET version 16.0. To run simulation in OPNET 16.0 the reference model was up graded to OPNET version 15.0 and then OPNET version 16.0 (as OPNET supports only one version up at a time).

Figure 26 and 27 displays network jitter for reference model and new model respectively. While the reference model shows network jitter on the order of 25 ms, new model detailed a variation from 25 ms to 40 ms.

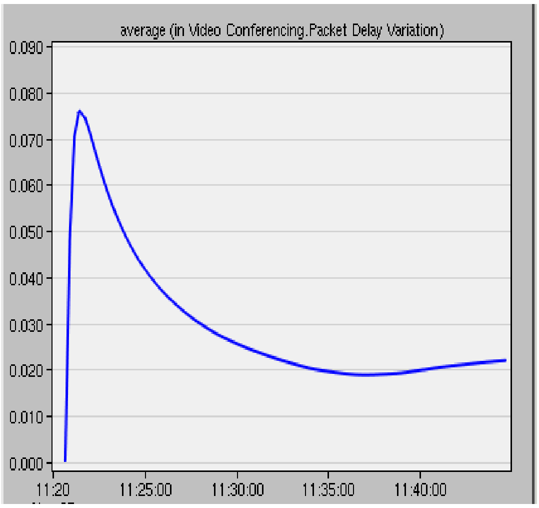


Figure 26: Network jitter (reference model)

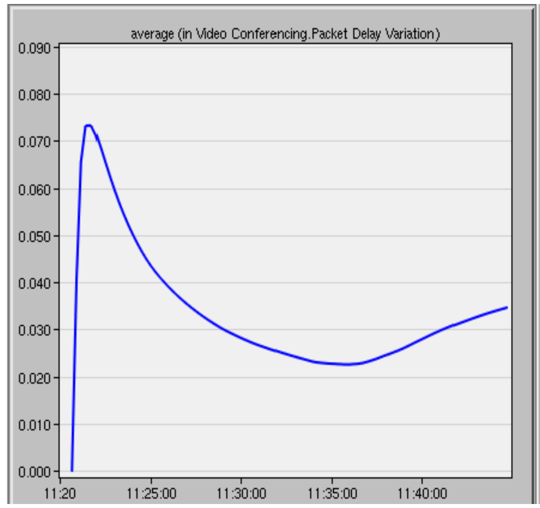


Figure 27: Network jitter

Figure 28 and 29 indicate the network throughput. As the reference model shows network throughput 24 Mbps, in new model the throughput increased significantly to 33 Mbps.

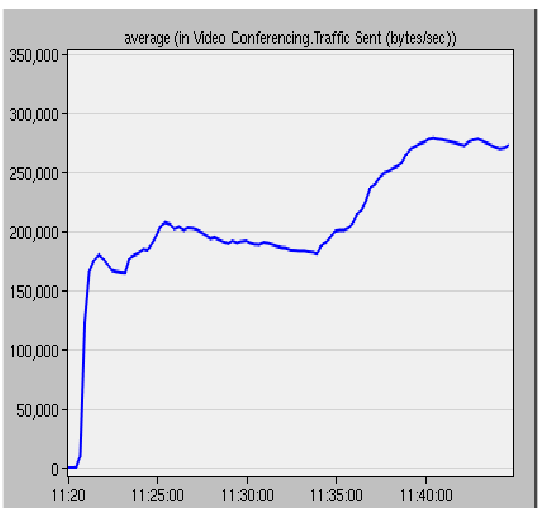


Figure 28: Network throughput (reference model)

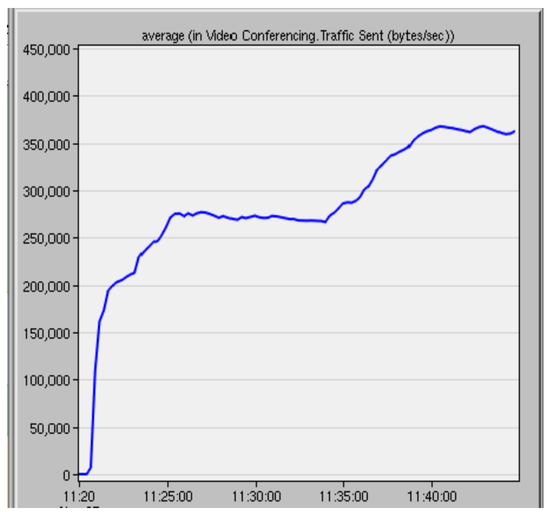


Figure 29: Network throughput

Figure 30 and 31 describes traffic receiver throughout of the network during the simulation period of reference model and new model respectively. Reference model graph reports an average of 90 packets per second, new model reports a significant high rate of 165 packets per second received.

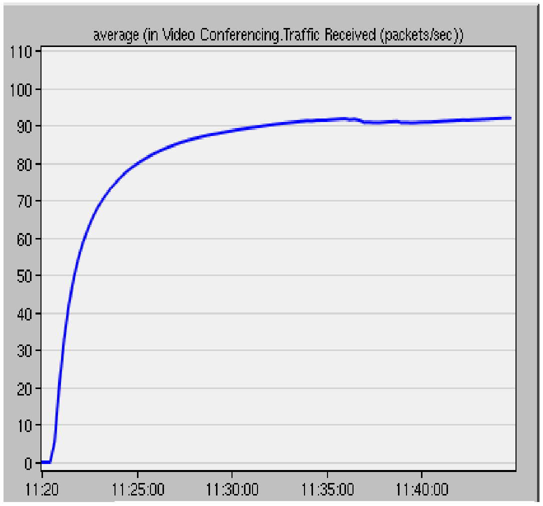


Figure 30: Network traffic received (packet/sec) (reference model)

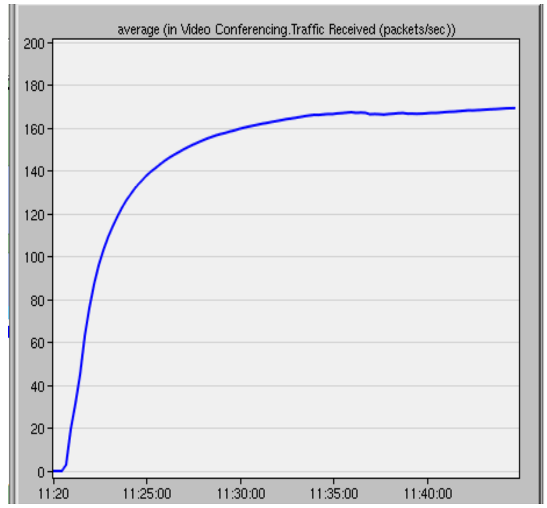


Figure 31: Network traffic received (packet/sec)

The results described in Figures 26-31 validate new model implementation.

7. Simulation Results

The reported simulation results reflect the streaming of the 30 minutes MPEG-4 video/audio content to the four client subscribers. Actual simulation times ranged from 2 to 8 hours for a given scenario depending on whether incremental background traffic growth was enabled.

7.1 Throughput for each Application

In the simulation, it is assumed that each traffic class has the equal portion of the total data traffic in terms of the average number of packets generated per unit time. The results are observed as Figure 32.

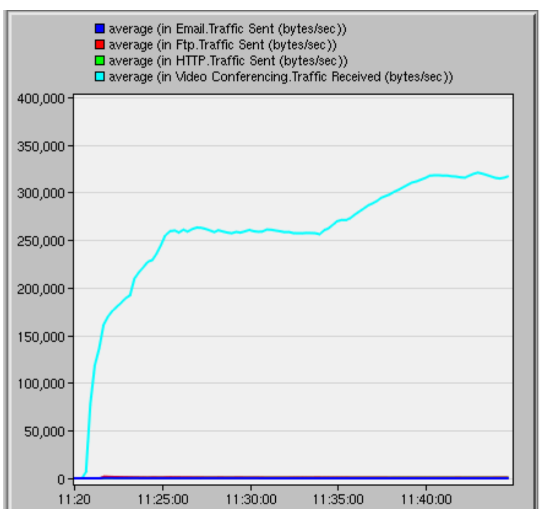


Figure 32: Averaged throughput for each Application

It is observed that the throughput of access category video/audio is way high than the access category HTTP, FTP and Email. It means that throughput for applications like video conferencing provides maximum throughput by providing them more priority over the other services like HTTP, FTP and Email.

Figure 33 reports the throughput of HTTP, FTP and Email. All three applications are configured to heavy load. Among these three simple applications, throughput of Access category FTP is higher than the HTTP and Email. HTTP is designed to retrieve web pages. It is optimized for numerous repeated fetches of small items. FTP is designed for transferring files and offers faster overall throughput and better error checking.

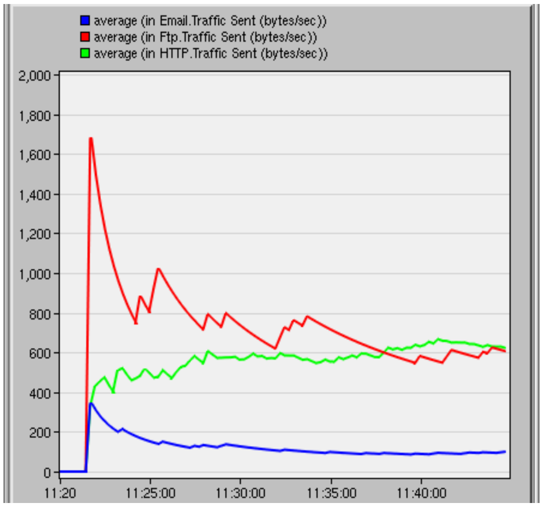


Figure 33: Averaged throughput for HTTP, FTP and Email Application

Figure 34 describes simulation result graph, it is observed that throughput of all applications starts at same time, HTTP send more bytes as comparison to other two applications. Each throughput lasts for 30 minutes of duration.

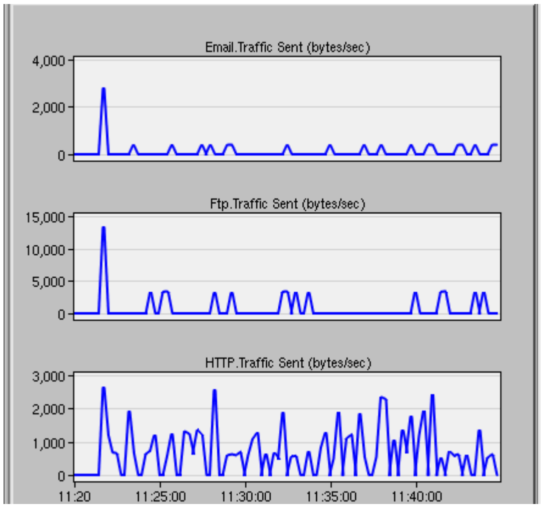


Figure 34: Instantaneous throughput for HTTP, FTP and Email Application

The model was configured to stream audio and video content, HTTP, FTP and Email to all client subscribers. Specifically, the movie was encoded at a rate of 50 fps. The VoD server is expected to send out unicast video/audio packets at a rate of 50 packets/sec for each client. Figure 35 confirms the expected behavior accordingly.

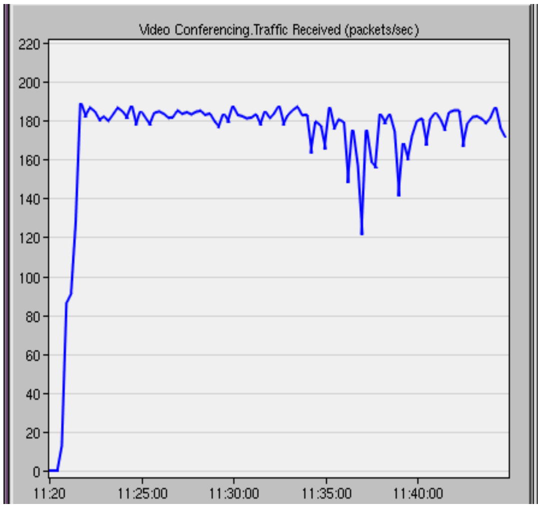


Figure 35: Video server packets/sec

7.2 MPEG-4 Video/ Audio, HTTP, FTP and Email Stream

7.2.1 WiMAX Link Characteristics

The captured PHY layer statistics provide insight into the performance of the WiMAX access network. The dropped packet rates by the PHY layer for the three WiMAX client stations is detailed in Figure .The WiMAX client station which is nearer to base station exhibits less drop rate. As noticed in Figure 36, the 6 km WiMAX station exhibits a much higher drop rate than the 2 km and 4 km stations over the 30 minutes interval. Figure 37 details the downlink SNR for the three WiMAX stations. Note that the 6 km station reports a downlink SNR that is the necessary minimum level for 16-QAM with ½ coding. Due to low SNR, 6 km station have high drop rate accordingly.

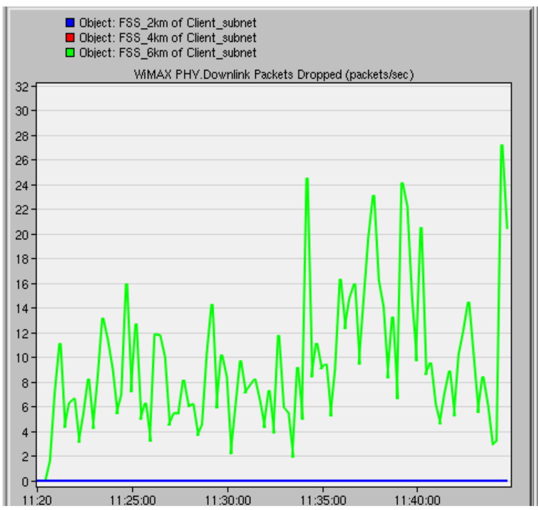


Figure 36: Downlink dropped packets/sec

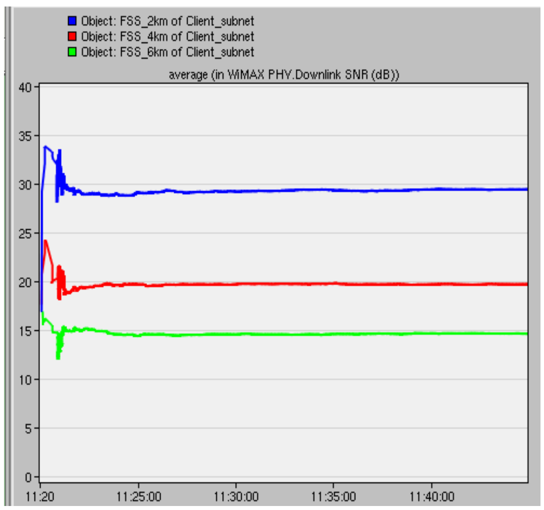


Figure 37: Downlink SNR

7.2.2 Block Error Rate (BLER)

Block error rate is the number of incorrectly transferred data packets divided by the number of transferred packets. A packet is assumed to be incorrect if at least one bit is incorrect. The downlink BLER for the 2 km and 4 km WiMAX stations is indicated in Figure 38.The WiMAX station which is nearer to base station reflects less BLER. The 4 km station is expected to reflect a higher BLER given that it is twice as far from the base station than the 2 km station. Figure 39 displays a BLER two orders of magnitude higher than the 4 km WiMAX station.

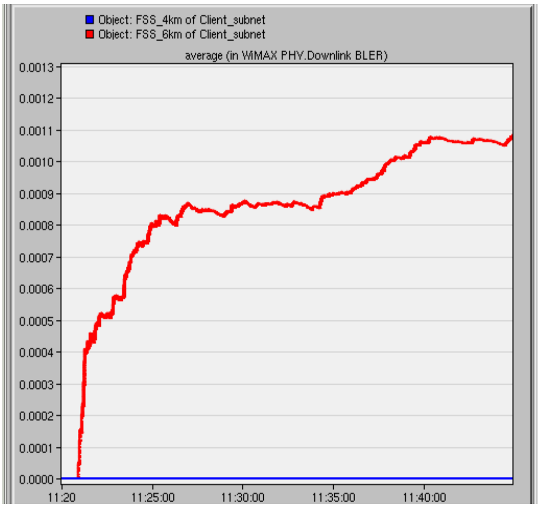


Figure 38: Downlink BLER

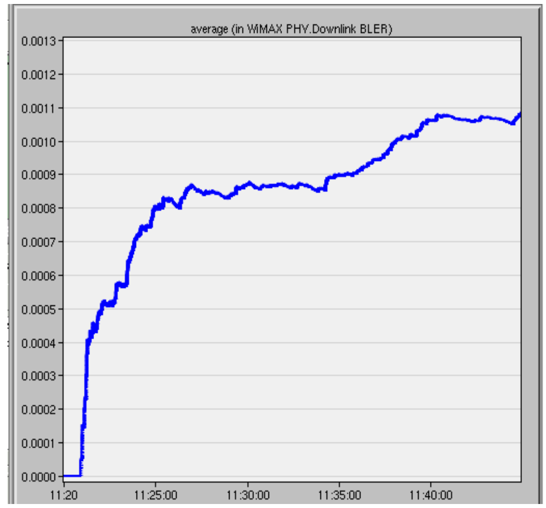
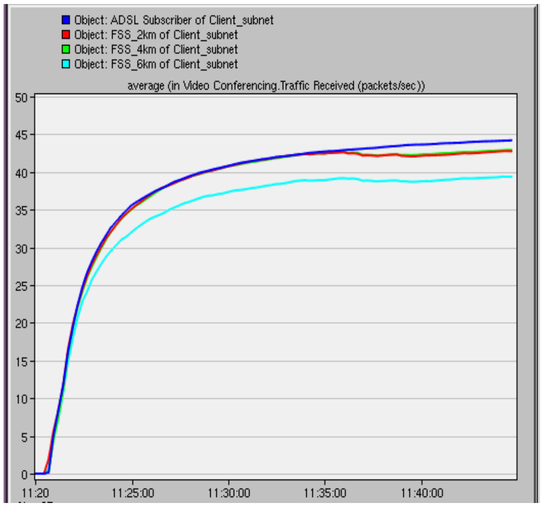


Figure 39: Averaged downlink BLER

SNR is inversely proportional to BLER. Lower the SNR for a given station, the higher BLER which is proved with simulation results for three WiMAX stations.

7.2.3 Packet Loss

All four curves are averaged over the 30 minutes movie duration. The following figures show the resulting packet loss observed on all four clients. The loss in Figure 40 is represented as the curve deviation from the 50 packets/sec position on the vertical axis. The blue ADSL client curve (top) approaches a received packet rate that matches the VoD sending rate of 50 packets/ sec. As the WiMAX station distance increases from base station, encoding rate exhibits more degradation. Results of simulation run indicate expected behavior. Figure 41 reports the same packet loss using instantaneous values.



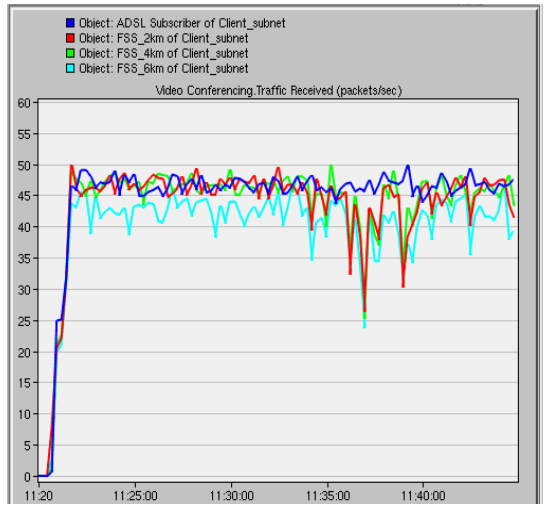
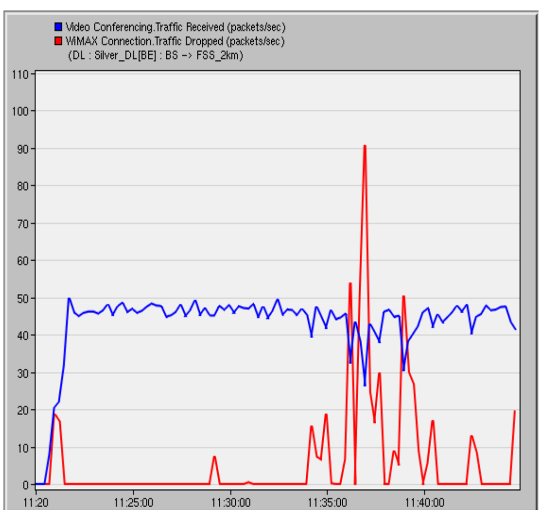


Figure 40: Received averaged packets/sec

Figure 41: Instantaneous packet loss

The main characteristic of WiMAX is the need of QoS guarantees, in terms of bandwidth reservation, delay, jitter and packet loss. One of the most important features of WiMAX is the possibility that it offers to provide QoS to data flows. In order to understand the why the packet loss on the WiMAX stations is significant, further exploration and characterization was necessary. Figure 42 captures the 2 km WiMAX station packet drop rate along with the MAC layer drop rate statistic from the base station.

The MAC layer in the base station is losing a significant number of frames because the base station queue size of 128 KB was being overrun as indicated in Figure 43.This behavior is largely in part due to the variable sized MPEG-4 video/audio frames.



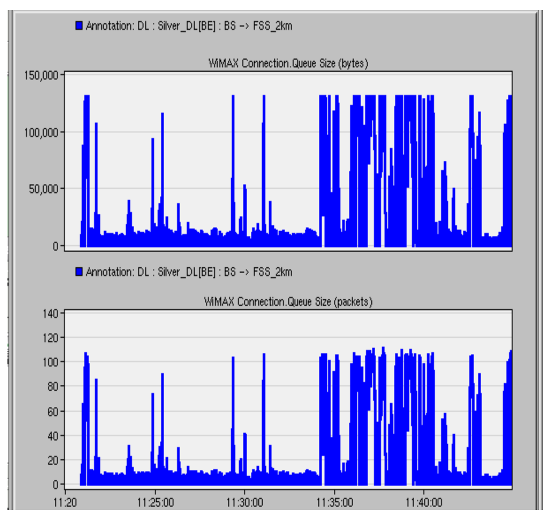


Figure 42: Received and dropped packets/sec

Figure 43: Base station downlink (DL) queue

Similarly, Figure 44-47 exhibit similar behavior for the 4 km and 6 km WiMAX stations.

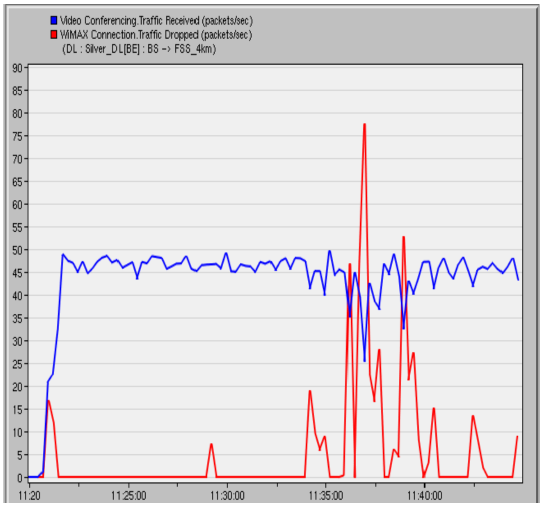


Figure 44: Received and dropped packets/sec for 4 km

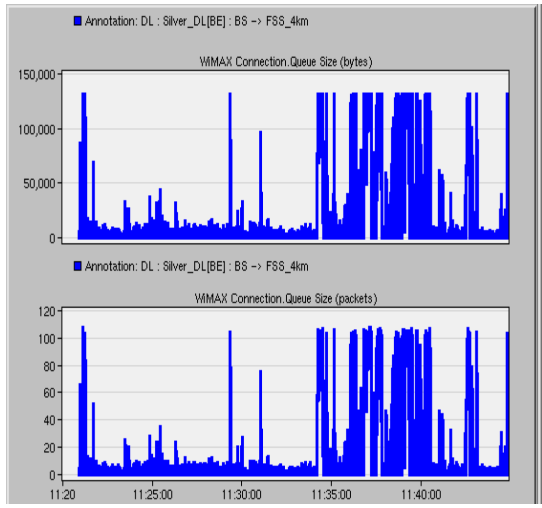


Figure 45: Base station DL queue for 4 km

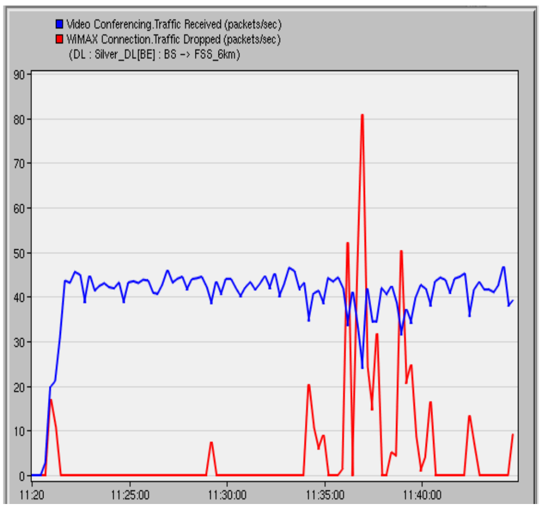


Figure 46: Received and dropped packets/sec for 6 km

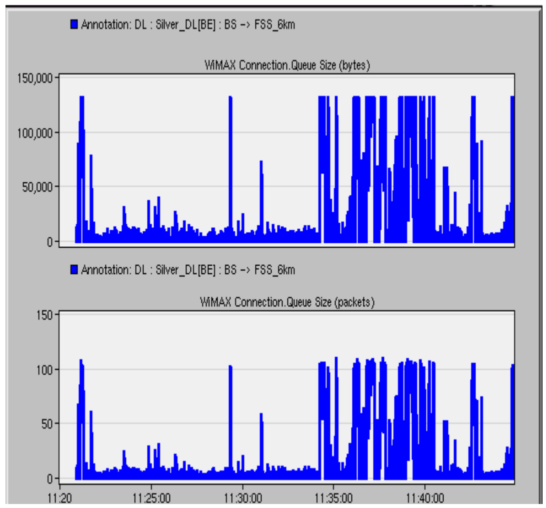


Figure 47: Base station DL queue for 6 km

7.2.4 Delay

End-to-end delay refers to the time taken for a packet to be transmitted across a network from source to destination. The end-to-end delay measured in the simulation run is detailed in Figure 48.The four client curves are averaged across the 30 minutes movie. Results indicate that the ADSL client approaches the ideal delay of 10 ms or less. All three WiMAX client station curves closely tracked each other while exhibiting a damping effect that appears to settle around 60 ms towards the end of the movie. As noticed in figure, the 6 km WiMAX station exhibits high damping effect than the 2 km and 4 km stations over the 30 minutes interval.

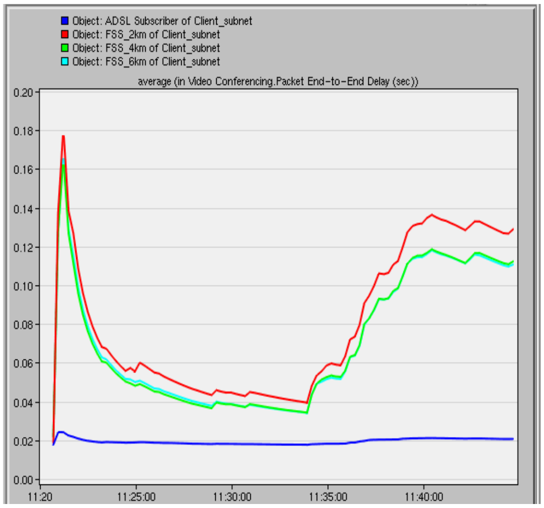


Figure 48: End-to-end packet delay

7.2.5 Jitter

A network with constant latency has no variation (or jitter).Packet delay Variation (PDV) is an important QoS factor in assessment of network performance. It is the difference in end-to-end delay between selected packets in a flow with any lost packets being ignored.The four video/audio client curves are averaged across the 30 minutes movie. The packet jitter measured in the simulation run is detailed in Figure 49.Results indicate that the ADSL client performed better than ideal value of 20 ms. The WiMAX client station curves closely tracked each other for the movie duration with a jitter approaching to ideal value of 20 ms.

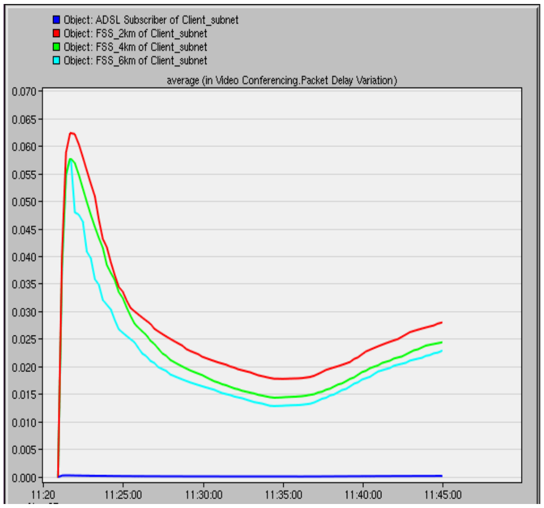


Figure 49: Video packet jitter

7.2.6 Throughput

Throughput is the average rate of successful message delivery over a communication channel. The throughput is usually measured in bits per second (bit/s or bps) and sometimes in data packets per second or data packets per time slot.The four client curves are averaged across the 30 minutes movie and tracked each other as expected in simulation results. The 2 km station surpassed the ADSL station throughput when measured in bytes/sec. The throughput measured in the simulation run is detailed in Figure 50. The observed throughputs ranged from 0.40 Mbps to 0.72 Mbps.

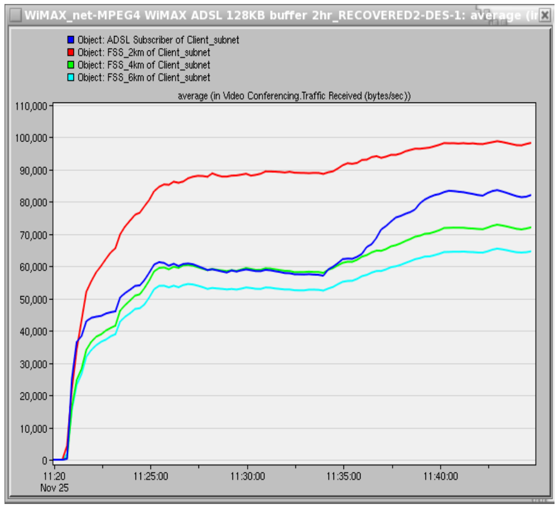


Figure 50: Minimum throughput

7.3 Tuning of buffer size effects

Since WiMAX is connection oriented, a connection establishment is required before transmitting packets over the network. WiMAX substation can have several connections simultaneously. Packets from higher layer are processed by a traffic policing module and put in an appropriate queue that shares the buffer memory with other queues. Because multiple queues share common buffer memory, there may be packet loss and degradation of QoS unless an appropriate buffer management scheme is used. Buffer tuning techniques adjust the network congestion avoidance parameters over high- bandwidth, high-latency networks. The Section 7.2 supported buffers of up to 128 Kbytes, which was adequate for slow links or links with small round trip times (RTTs).The receive window specifies the amount of data that can be sent and not received before the send is interrupted. If too much data is sent, it overruns the buffer and interrupts the transfer. Now, we conducted additional tuning of the base station buffer size to explore its impact on packet loss rate statistic and, ultimately, the video packet loss statistic. Various queue sizes ranging from the default value of 128 KB to 1,024 KB were employed. It was evident that 1,024 KB buffer resolved the buffer overflow and resulted in zero MAC packet loss rate. The improved performance of the 2 km and 4 km WiMAX stations is shown in Figure 51. The 6 km station continued to exhibit unacceptably high packet loss rates, primarily due to the SNR that was the minimum level required for the configured modulation/coding scheme. The same loss performance using instantaneous values is shown in Figure 52. Further examination of the 2 km WiMAX station reveals that the received video rate closely tracks the original encoding and transmission rate, as shown in Figure 53. Figure 54 reports that the base station connection queue size never reaches the buffer capacity of 1,024 KB. Figure 55-58 exhibit similar behavior with the 4 km and 6 km WiMAX stations.

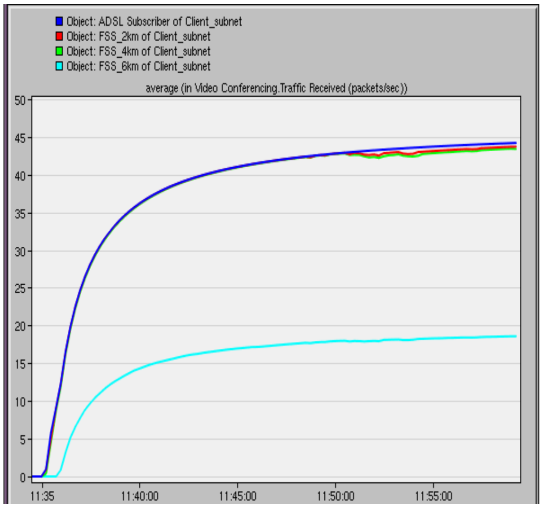


Figure 51: Average received packets/sec

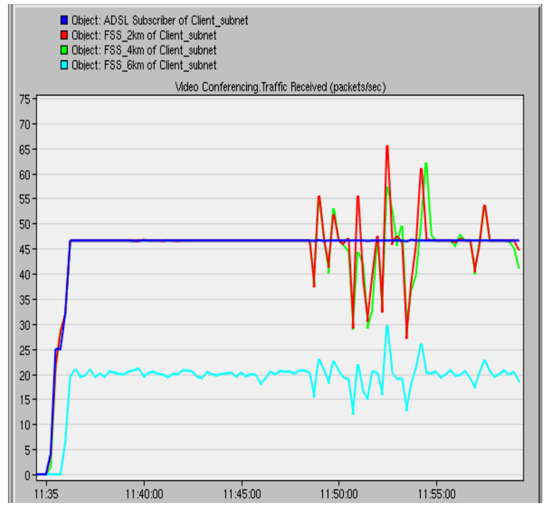


Figure 52: Instantaneous received packets/sec

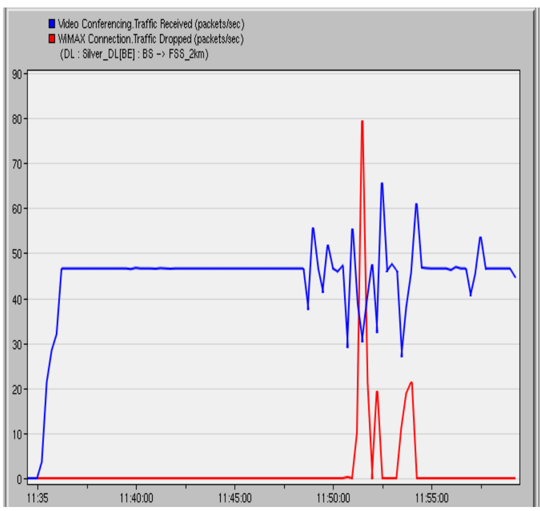


Figure 53: Received and dropped packets/sec for 2km

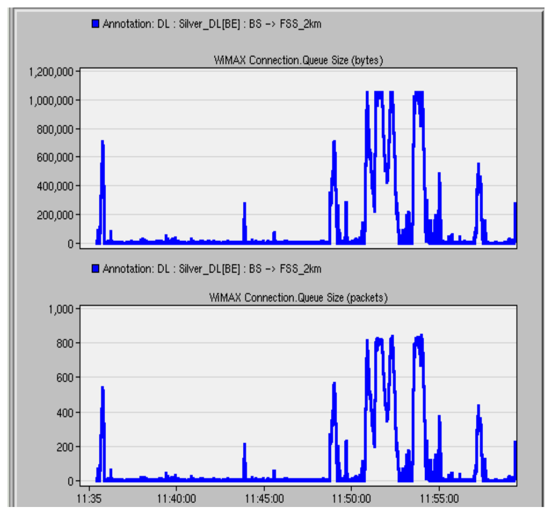


Figure 54: Base station DL queue for 2km

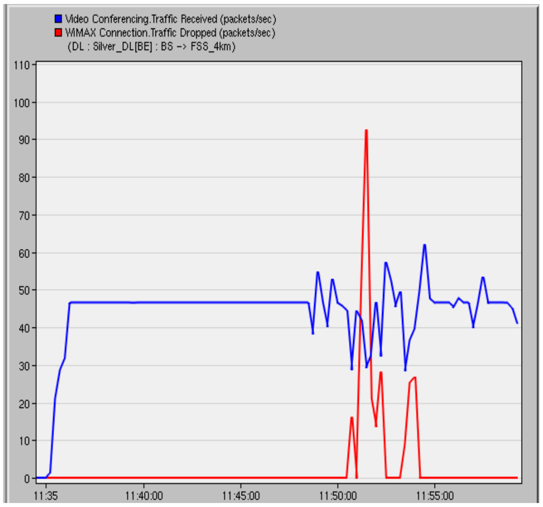


Figure 55: Received and dropped packets/sec for 4 km

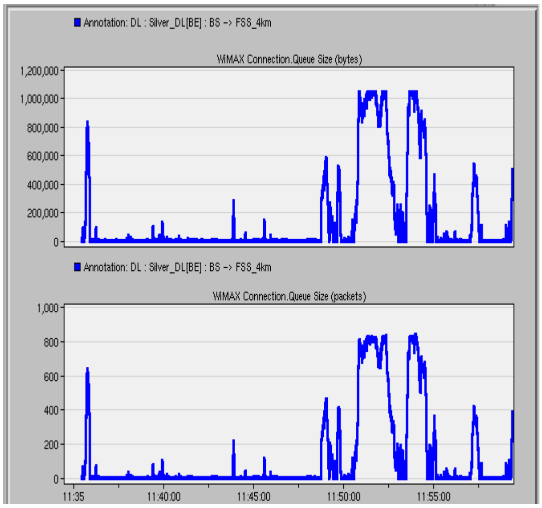


Figure 56: Base station DL queue for 4 km

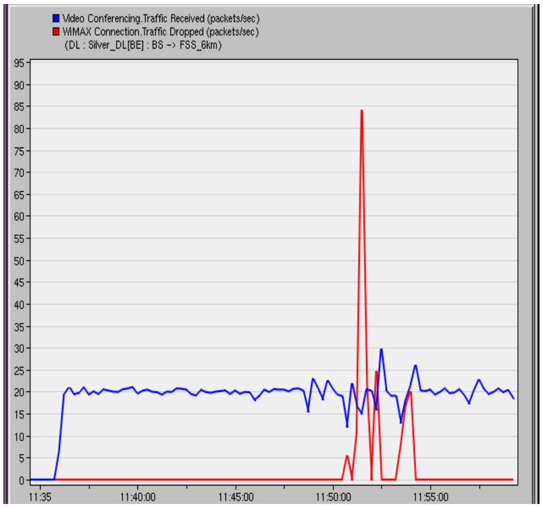


Figure 57: Received and dropped packets/sec for 6 km

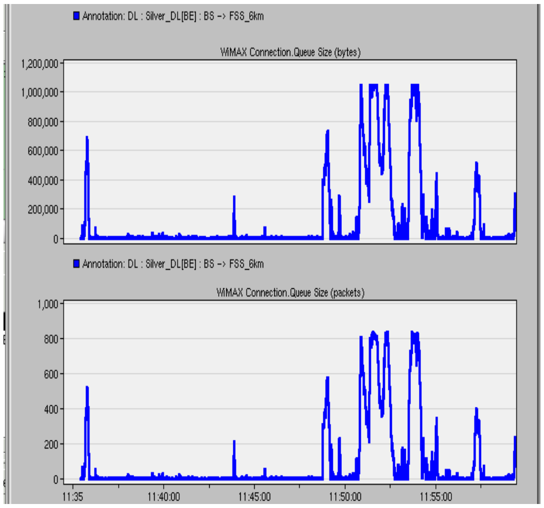


Figure 58: Base station DL queue for 6 km

8. Related Work

As stated before, the model from the first reference was the base model for this project [1]. Numerous interrelated efforts have explored WiMAX in the context of real-time and stored video applications. Another research effort [19] presented WiMAX fundamentals as a broadband access solution to support IPTV services framework. The authors discussed the considerations associated with delivery video services while minimizing video and audio quality degradation. Furthermore, they presented some key transceiver design considerations at the PHY layer. There has also been effort exploring the performance of scalable video streaming over mobile WiMAX stations using feedback control [20]. Researchers evaluated MAC layer performance by scaling video content over multiple connections based on feedback of the available transmission bandwidth.

9. Future Work

This project analyzed the performance of WiMAX broadband access to existing ADSL broadband access in terms of a throughput, delay, jitter and packet loss for video/audio streaming, HTTP, FTP and Email.

Subsequently, to increase the precision of this model, future efforts could revisit these design parameters and further characterize their impact on the system performance through isolated scenarios. During the simulation, certain key assumptions were made accordingly. These assumptions include:

- Station transmit power
- Station antenna gain
- Pathloss model and corresponding flat, low density tree terrain
- Carrier operating frequency and channel bandwidth
- WiMAX MAC scheduling type
- WiMAX Service class throughput rates
- WiMAX multi-path model disablement
- WiMAX fixed station configuration only (mobility disabled)

Comprehensive analysis of WiMAX networks and characterization of other WiMAX parameters can be conducted. Moreover, performance matrices can be studied in depth. Incorporate other applications like remote login and network printer. Lastly, WiMAX mobility and shadowing can be performed.

10. Conclusion

The aim of the paper is to highlight the research going on in the field of wireless and wired computer networks. An extensive review on wireless and wired networks using simulation has been investigated for their performance comparison by varying the attributes of network objects such as traffic load, customizing the physical characteristics to vary BLER, packet loss, delay, jitter, and throughput. As a result, the study has utilized the OPNET Modeler to design and characterize the performance of streaming a 30 minutes MPEG-4 movie to both WiMAX and ADSL subscribers.

The validation scenarios confirmed the overall design of the study was implemented successfully in the Modeler. From the simulation results, while ADSL demonstrated behavior that approached the ideal values for the performance factors. Initial simulation runs exhibited significant packet loss. To improve the overall performance of the network, simulations were run with varied values of buffer size. With further tuning, a configuration was derived that demonstrated packet loss that was more adequate of the ADSL client station. Applications that are sensitive and affected of delay and jitter of information, such as video and audio, need small queues. Small queues reduce delay, which is essential for real-time traffic. Non-real-time traffic such as electronic mail, file transfers, and backups must be serviced by large queue. The file transfer performance drop dramatically when distance between work station increases. But the WiMAX file transfer performance in is nearly perfect.

In conclusion, this study was successful in analyzing its specified objective. Overall, the OPNET Modeler has provisioned a suitable environment to design and characterize WiMAX networks. Furthermore, all applications were modeled as unicast traffic; multicast video traffic would yield better performance.

11. References

1. W.Hrudey and Lj.Trajkovic “Streaming Video Content Over IEEE 802.16/WiMAX Broadband Access,” *OPNETWORK 2008*, Washington, DC, Aug.2008.

2. What is WiMAX? [online]. Available: http://www.wimax.com/wimax-tutorial/what-is-wimax.

3. WiMAX MAPS [online]. Available: http://www.wimaxmaps.org/.

4. WiMAX Report [online]. Available: http://www.wimaxforum.org/technology/ downloads/.

5. D. M. Ali and K. Dimyati, “Performance Study of the WiMAX Uplink Scheduler,” *Communications IEEE (MICC) 2009*, Malaysia, Dec 2009, pp.831.

6. M. Hu, H. Zhang, T. A. Le, and H. Nguyen, “Performance Evaluation of Video Streaming over Mobile WiMAX Networks,” *IEEE GLOBECOM Workshops 2010*, FL, Dec 2010.

7. W. Kim and H. Song, “A Novel Combined Packet Scheduling and Call Admission Control for Video Streaming over WiMAX Network,” *IEEE GLOBECOM Workshops 2010*, FL, Dec 2010.

8. Aman, Muhammad, Sikdar Biplab, and Parekh Shyam, "Scalable Peer-to-Peer Video Streaming in WiMAX Networks," *IEEE GLOBECOM Workshops 2009*, Honolulu, HI, Nov 2009.

9. Castellanos, G.D., Khan, and J.Y. “Performance of WiMAX packet schedulers for multi-class traffic,” *Proc. IEEE LATINCOM 2010*, Bogota, Sept 2010, pp.1-6.

10. Iosif, Cirstea, E.-R., Banica, Ciochina, S, “Performance analysis of uplink resource allocation in WIMAX,” *Proc. IEEE ICCOMM 2010*, Bucharest, June 2010, pp. 351 – 354.

11. J. Chen, W. Jiao, and Q. Guo “An integrated QoS control architecture for IEEE 802.16 broadband wireless access systems,” *IEEE Global Telecommunications Conference, 2005. GLOBECOM '05*, St. Louis, MO, Dec 2005, pp. 3335.

12. D. Kim, H. Cai, M. Na, and S. Choi “Performance measurement over Mobile WiMAX /IEEE 802.16e network,” *International Symposium on a World of Wireless, Mobile and Multimedia Networks 2008*, Newport Beach, CA, USA June 2008 pp. 1-8.

13. S. Tiraspolsky, A. Rubtsov, A. Maltsev, and A. Davydov “Mobile WiMAX - Deployment Scenarios Performance Analysis,” *Wireless Communication Systems, 2006, ISWCS ' 06*, 3rd International Symposium, Sep 2006, pp. 353-357.

14. D. Wu, Y. T. Hou, W. Zhu, Y. Zhang, and Jon M. Peha “Streaming Video over the Internet: Approaches and Directions,” *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 11, no. 3, March 2001.

15. W. Hrudey and Lj. Trajkovic, “Mobile WiMAX MAC and PHY Layer Optimization for IPTV,” Special issue on Recent Advances in Simulation and Mathematical Modeling of Wireless Networks, *Mathematical and Computer Modelling*, Elsevier, vol. 53, March 2011, pp. 2119-2135.

16. S.C. Wang, K.Q. Yan and C.H. Wang "A Channel Allocation based WiMax Topology" *Proceedings of the International MultiConference of Engineers and Computer Scientists (IMECS) 2009*, vol I, Hong Kong, March 2009.
17. K. Pentikousis, J. Pinola, E. Piri, and F. Fitzek, "An experimental investigation of VoIP and video streaming over fixed WiMAX," *Modeling and Optimization in Mobile, Ad Hoc, and Wireless Networks and Workshops, 2008. WiOPT 2008. 6th International Symposium*, Berlin, April 2008, pp. 8-15.
18. F. Retnasothie, M. Ozdemir, T. Yucek, H. Celebi, J. Zhang, and R. Muththaiah, "Wireless IPTV over WiMAX: challenges and applications*," Proc. IEEE WAMICON 2006*, Clearwater, FL, Dec. 2006, pp. 1–5.
19. H. Juan, H. Huang, C. Huang, and T. Chiang, "Scalable video streaming over mobile WiMAX," *Proc. ISCAS 2007*, New Orleans, Louisiana, May 2007, pp. 3463–3466.

12. APPENDICES

12.1 Acronyms

ADSL Asymmetric Digital Subscriber Line
BS Base Station
BE Best Effort
DL Downlink
Email Electronic mail
FTP File Transfer Protocol
FPS Frames Per Second
FSS Fixed Subscriber Station
HTTP Hyper Text Transfer Protocol
IP Internet Protocols
IPTV Internet Protocol television
LAN Local Area Network
LOS Line of Sight
MAC Media Access Control
MPEG Motion Picture Experts Group
NLOS Non Line-of-Sight
nrtPS Non-Real-Time Polling Service
OFDM Orthogonal Frequency Division Multiplex
OFDMA Orthogonal Frequency Division Multiple Access
OTA Over the Air
PHY Physical Layer
PMP Point-to-Multi-Point
QAM Quadrature Amplitude Modulation
QoS Quality of Service
RTP Real Time Protocol
rtPS Real-Time Polling Service
SC Single Carrier
SNR Signal to Noise Ratio
ToS Type of Service
UDP User Datagram Protocol
VCA Video Conferencing Application
VoD Video on Demand
WAN Wide Area Network
WiMAX Worldwide Interoperability for Microwave Access

12.2 Challenges

Numerous challenges were experienced throughout this project. Initially, environment problem was faced due to which simulation and log in (license expiration) were big issues.

Additionally, the major challenge was disk quota. Linux operating system has limited disk quota and it did not support simulation for 2 hours (which was reference model run time). So, the simulation was run for only 30 minutes and analysis of the results of those 30 minutes was done.

Finally, learning WiMAX fundamentals within the duration of this project to drive the design of this simulation model proved to be challenging given the breadth and depth of this technology.