Comparison of WiMAX and ADSL Performance when Streaming Audio and Video Content

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Abstract
The IEEE 802.16 Worldwide Interoperability for Microwave Access (WiMAX) standard is widely used for fixed and mobile Internet access. WiMAX provides maximum data rate of 75 Mbps and high-speed Internet access to a wide range of devices used by clients over the last mile. Asymmetric Digital Subscriber Line (ADSL) is widely used to provide guaranteed service.

In this paper, we compare performance of WiMAX and ADSL by streaming audio and video contents. File Transfer Protocol (FTP), Hyper Text Transfer Protocol (HTTP), and electronic mail have also been used for the comparison. We used OPNET Modeler versions 15.0 and 16.0 to evaluate packet loss, delay, delay jitter, and throughput with various design parameters to determine whether WiMAX exhibits performance comparable to ADSL.

1. Introduction
In 1998, the Institute of Electrical and Electronics Engineers (IEEE) established the IEEE 802.16 working unit that developed a standard for wireless metropolitan area networks. Its primary aim was to provide high-speed fiber access solution using high frequency line of sight (LoS) fixed wireless connections. Initially, IEEE 802.16 was developed to support fixed broadband wireless access over lower frequency non line of sight (NLoS) wireless connections. This standard was modified to support fixed WiMAX. WiMAX technology employs the Orthogonal Frequency Division Multiple Access (OFDMA) air interface and provides support for mobility. Since its emergence, the WiMAX technology has been widely adopted in wireless networks [1].

ADSL technology is used for broadband Internet access. WiMAX is designed to replace ADSL and, hence, comparing the performance of WiMAX and ADSL provides better understanding of the new technology. We simulate performance of the two access networks by streaming video/audio files and by transmitting HTTP, FTP, and email data to measure:

- video packet loss
- end-to-end delay
- video packet jitter
- throughput.

Video/audio streaming is becoming widely adopted 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 deals with streaming services using an Internet topology and evaluating video/audio performance. The simulation model incorporates genuine movie video/audio traces. It streams 25 minutes of the Matrix III movie [2] to three WiMAX client stations and one ADSL client from a video/audio content Internet services provider.

The outline of the paper is as follows. In Section 2, we provide description of WiMAX and ADSL technologies and describe the network design. In Section 3, the OPNET model validation is discussed. Simulation results are described in Section 4. We conclude with Section 5.

2. Background
In this Section, we present an overview of WiMAX and ADSL technologies and the video content. We also briefly describe the network configurations that are simulated in this project.

2.1 WiMAX Overview
The recent rapid growth of wireless communication technology greatly improved the transmission data rates and communication distances. WiMAX is one of the emerging technologies in broadband wireless systems. Its transmission rate and distance may reach up to 75 Mbps and 50 km, respectively [3]. Compared to other wireless networks, WiMAX has the advantage of higher transmission speed and larger transmission coverage. With its high bandwidth and capacity for long distance transmission, it solves the last mile problem in the metropolitan networks [4].

The IEEE 802.16a standard outlines NLoS communications in 2–11 GHz band. It uses one of three air interfaces: single carrier (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 orthogonal subcarriers closely together. WiMAX operates in 10–66 GHz band with LoS communications. It uses SC air interface [5]. Channel bandwidth ranges between 1.25 MHz and 20 MHz in 2–11 GHz band. By allocating various sub-carriers and using various modulation schemes within this channel bandwidth, WiMAX may achieve data rates between 1.5 Mbps and 75 Mbps [1].

Generic topology of a WiMAX network is shown in Figure 1. It consists of Base Station (BS), Subscriber Station (SS), and clients. It has two transmission modes: Point to Multi Point (PMP) and mesh [3]. WiMAX uses bandwidth request and granting scheme at the SSs to achieve required Quality of Service (QoS). These schemes help the WiMAX BSs prevent over-subscribing their available resources [1], [6].

Cell sizes in WiMAX vary from 7 km to 10 km. WiMAX supports wireless backhaul links for Wi-Fi hot spots and redundant wireless Internet backup links for commercial businesses. It enables residential and commercial subscribers to attain high-speed Internet access. In this study, we examine WiMAX as an access network technology alternate to ADSL.

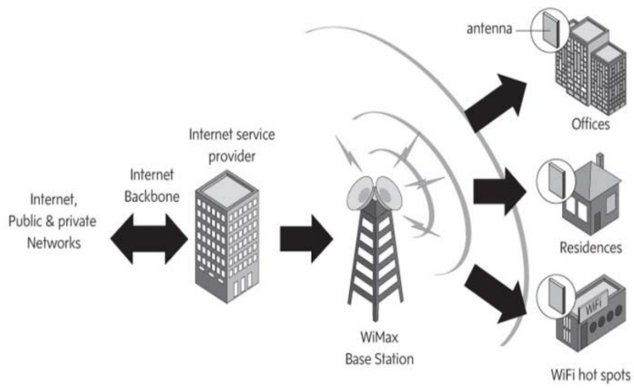

Figure 1: Generic topology of a WiMAX network.

2.1.1 Model Design

We use OPNET version 16.0 to create the simulation models. Two scenarios with distinct buffer sizes (128 kilobytes and 1,024 kilobytes) are implemented. The client stations are fixed.

2.1.2 Network Topology

The network topology of the model is shown in Figure 2. The client and the server subnets are geographically separated: The server subnet is located in Toronto while the video client subnet is located in Vancouver [7].

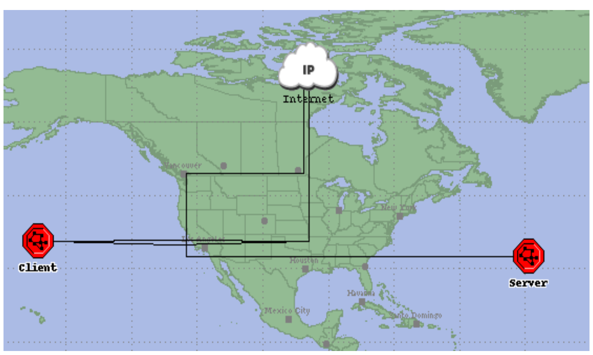

Figure 2: Network topology of the simulation model.

The inner view of the server subnet is shown in Figure 3. Server subnet is designed with a server configured to stream stored audio and video contents, HTTP, FTP, and email. It contains a 100 Mbps IP network and a firewall. An access router is connected to the firewall. This router connects the Internet cloud to the server subnet through a 45 Mbps Digital Signal (DS3) wide area network (WAN) link. A local video client in the server network is used for troubleshooting and traffic validation. It encompasses four video client stations that access the same video on demand (VoD) services from Toronto. In this subnet, three fixed wireless WiMAX stations are located 2 km, 4 km, and 6 km from the WiMAX BS. The BS is 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 for comparison with WiMAX stations.

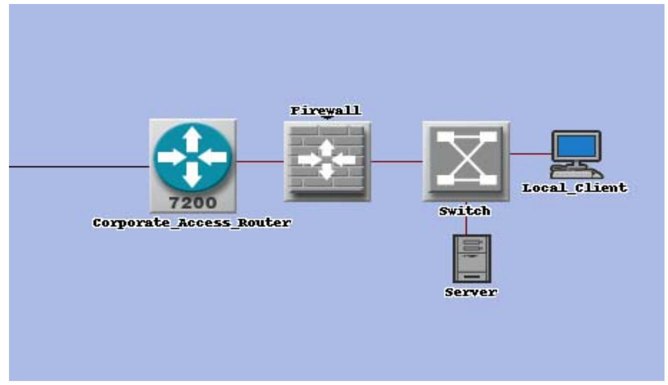

Figure 3: Video services subnet.

The client subnet structure is shown in Figure 4. It contains three WiMAX client stations, one ADSL client station, and one WiMAX base station. WiMAX client stations are located 2 km, 4 km, and 6 km from the base station.

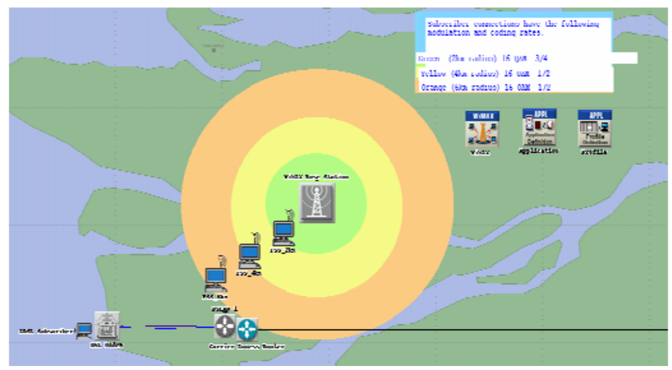

Figure 4: Video client subnet model.

The approximate distance between the two subnets is 3,342 km, which corresponds to approximately 13.3 ms propagation delay. The local area network (LAN) and WAN links are configured to utilize 10% and 20% utilization traffic loads, respectively. The packet discard ratio in the Internet "cloud" is 0.001%. The cloud has 1 ms additional delay along with the propagation delay of the WAN link.

2.1.3 WiMAX Configuration

The WiMAX configuration in the developed OPNET model consists of:

- service class/service flows
- media access control (MAC) scheduler
- burst profiles
- air Interface
- operating frequency
- channel bandwidth and subcarrier allocation
- transmit power
- path loss model.

The WiMAX QoS requirements are inferred by traffic flows between the BS and the SS. They are captured by the service class attribute. The downlink flow is from BS to the SS while the uplink flow is from SS to BS [8].

One of the key parameters of service flows is the type of a MAC scheduler. It allows WiMAX to provide QoS and support delay sensitive traffic such as voice and video. Four scheduler types are:

- UGS (unsolicited grant service)
- rtPS (real time polling service)
- nrtPS (non real-time polling service)
- BE (best effort).

The available bandwidth is allocated first to UGS and then to rtPS, nrtPS, and BE flows. In the OPNET model, two service classes were created for the downlink using BE scheduling with 3.0 Mbps and 640 Kbps minimum sustainable data rates. We assumed that both uplink and downlink channels have similar properties. The WiMAX configuration attributes are shown in Figure 5.

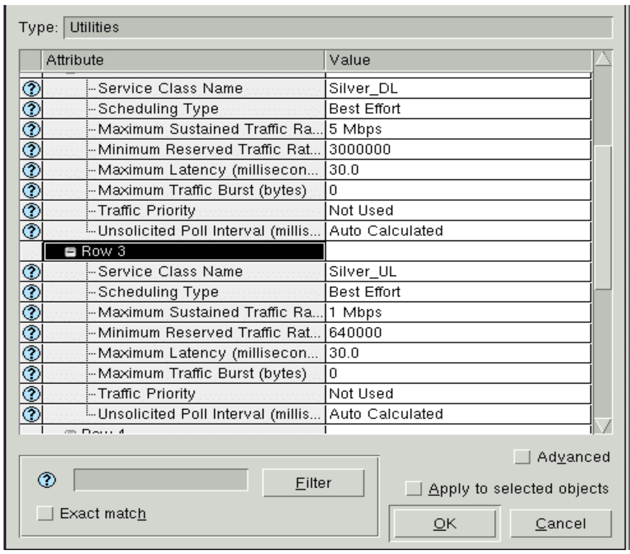

Figure 5: WiMAX service class configuration.

WiMAX client stations were configured to support 16-Quadrature Amplitude Modulation (QAM) modulation/coding schemes. The available coding rates for a given modulation scheme and the minimum signal to noise ratios (SNR) are shown in Table 1.

Table 1: Modulation/coding rates [1].

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

The modulation and coding rates for both uplink and downlink service flows for the WiMAX client located at 2 km radius are shown in Figure 6. The physical (PHY) layer access is configured to utilize OFDM with a 2.5 GHz base frequency using a 5 MHz channel bandwidth that provisions 512 subcarriers allocated as shown in Table 2. The WiMAX client station transmit power is configured to 33 dBm (2 Watts). The BS transmit power is configured to 35.8 dBm (3.8 Watts).

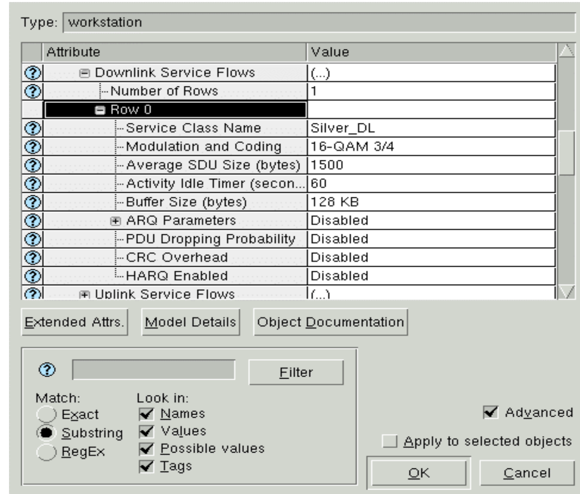

Figure 6: Configuration of the WiMAX client at 2 km radius.

Table 2: PHY layer frame division pattern [1].

	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

2.2 ADSL Configuration

The ADSL client has 3.0 Mbps downlink channel and a 640 Kbps uplink channel. The client is located 5 km from the central office. The configuration of various application services is shown in Figure 7.

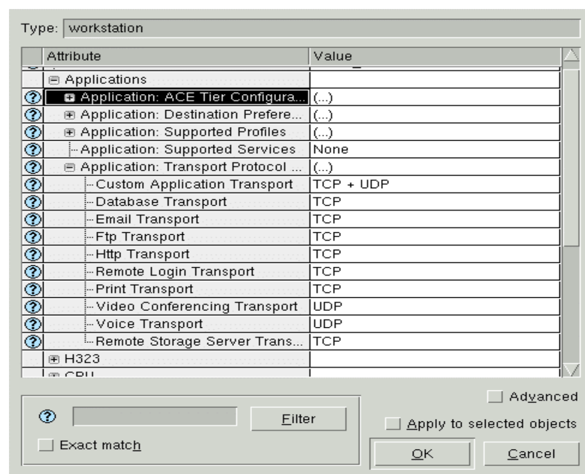

Figure 7: ADSL service configuration.

2.3 Video Content Overview

Video content refers to both the audio and the visual information. Examples of media service providers are newscasts, sporting events, movies in real time, VoD formats, and wide range of sitcoms. They are also known as real-time multimedia services. Real-time transport of live video or stored video is a significant part of Internet traffic. This project focuses on data streaming over Internet. There are two methods for transmission of stored video over the Internet: the download method and the streaming method (video streaming). In video streaming, full download of the video content is unnecessary and video data are received, decoded, and played-out at the same time [2].

The video content is organized as a sequence of frames or images for video streaming. These frames are sent to the subscriber and displayed at a constant rate. The video data is accompanied with a multi-channel audio data. The audio data are structured as a sequence of audio frames. Transmission rate and buffering requirements of real-time streaming depends on the network and the client stations. The quality of video content depends on parameters such as video format, pixel color depth, coding scheme, and frame inter-arrival rate. These parameters increase the size of the raw video, which affects transmission and buffering requirements in the network. Various encoding schemes such as MPEG-x and H.26x codecs are used to reduce the traffic load created by the data. These encoding schemes are loss-tolerant. However, their performance depends on available link bandwidth and delay characteristics [9].

Video frame inter-arrival rates ranges from 10 frames per second (fps) to 30 fps. Network conditions may influence the frame inter-arrival rates. This may degrade the video playback quality. Figure 8 illustrates the necessity of the client video system to playback frames at a constant rate [1].

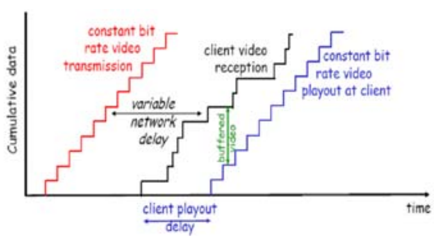

Figure 8: Video client buffering [1].

Figure 9 illustrates architecture for video streaming. The raw video and audio data are compressed by video/audio compression algorithms and then saved in a server. When requested, a streaming server resumes sending data and the application-layer QoS control module adapts the video/audio bit-streams according to the network status. The transport protocols organize the frames into packets and send the video/audio packets to the Internet. Packets may be dropped or may experience excessive delay due to congestion inside the Internet. Packets that are successfully delivered to the destination pass through transport layers and application layer. They are then decoded at the video/audio decoder. A synchronization mechanism is used to achieve synchronization between video/audio streaming [2].

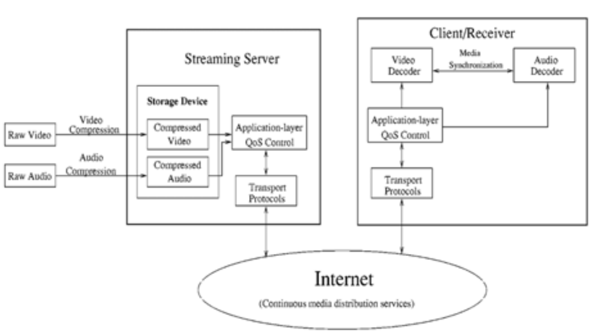

Figure 9: Architecture of video streaming [2].

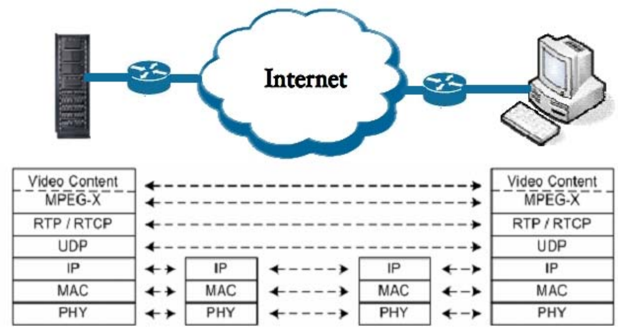

Figure 10: Video streaming network topology.

The Real-time Transport Protocol (RTP) layer operates on top of the Transmission Control Protocol (TCP) or User Datagram Protocol (UDP) layer. The UDP is used for streaming audio/video data and provides best effort service without delay, loss, or bandwidth guarantees. UDP segments are then encapsulated into unicast or multicast IP packets for addressing and routing to the video client stations. IP packets pass through MAC and PHY layers and then propagate through the Internet and wired or wireless access networks to the client subscribers. Buffers in SSs decompress and playback the video/audio frames at a constant rate [10].

To evaluate communication performance between the server and the client, four metrics are used to measure streaming performance. These performance metrics are:

- packet loss
- delay
- jitter
- throughput.

2.4 Traffic

Traffic is a key aspect of this project. The used reference model employed only video traffic [1]. In this project, we also considered audio, HTTP, FTP, and email traffic. This imposed additional load to the access links and helped observe the performance matrices [11].

The video/audio traffic source was a 2-hour MPEG-4 Matrix III movie trace which utilized a 352×288 frame format resolution and a 25 fps encoding rate. For HTTP, FTP, and email traffic, both the application attribute and the server were configured for heavy load traffic.

3. Validation of the OPNET Simulation Model

We have developed a new model based on the reference model that was developed using OPNET version 14.5.A [1]. We have upgraded the model to OPNET versions 15.0 and 16.0. The reference model employed only video streams as a traffic source during the simulation. The newly developed model incorporated audio streams, HTTP, FTP, and email traffic [12]. For model validation, we compared global statistics of the developed model with global statistics of the reference model. Analysis of network jitter, throughput, and received traffic is done for both models.

Network traffic received in the reference model and the developed model is shown in Figures 11 and 12, respectively.

Reference model shows an average of 90 packets per second (pps) while the new model shows a significantly higher rate of 165 pps. Packet jitter and network throughput exhibit similar behavior. The reference model shows network jitter of approximately 25 ms while the new model shows variation from 25 ms to 40 ms. The reference model exhibits network throughput of 24 Mbps while the throughput of the new model increased significantly to 33 Mbps. These results validate implementation of the developed model.

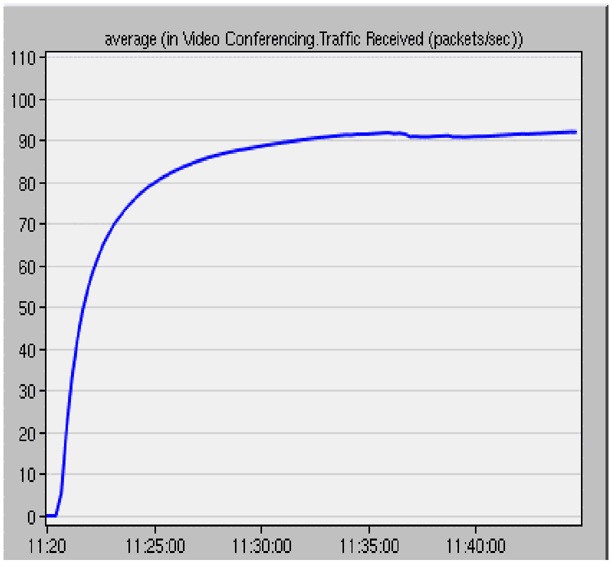

Figure 11: Reference model: average network traffic received.

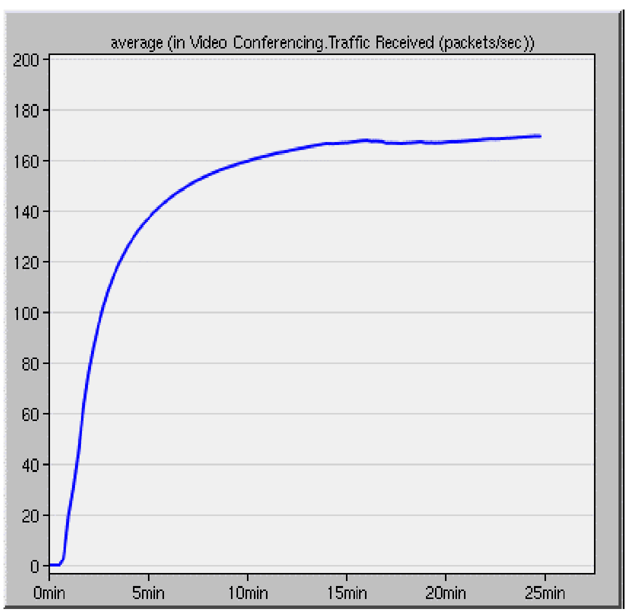

Figure 12: Developed model: Average network traffic received.

4. Simulation Results

Simulation results reflect the streaming of 25 minutes of MPEG-4 video/audio content to the four client subscribers. Simulation times ranged from 2 hours to 8 hours for a given scenario depending on whether incremental background traffic growth was enabled.

4.1 Throughput for Various Applications

For simulations, we assumed that an equal fragment of the total traffic is given to each traffic class and measured as the mean of the number of packets produced per unit time. The results are shown in Figures 13 and 14. The throughput of the video/audio access category is higher than the HTTP, FTP, and email access. This implies that applications such as video conferencing provide maximum throughput compared to services such as HTTP, FTP, and email.

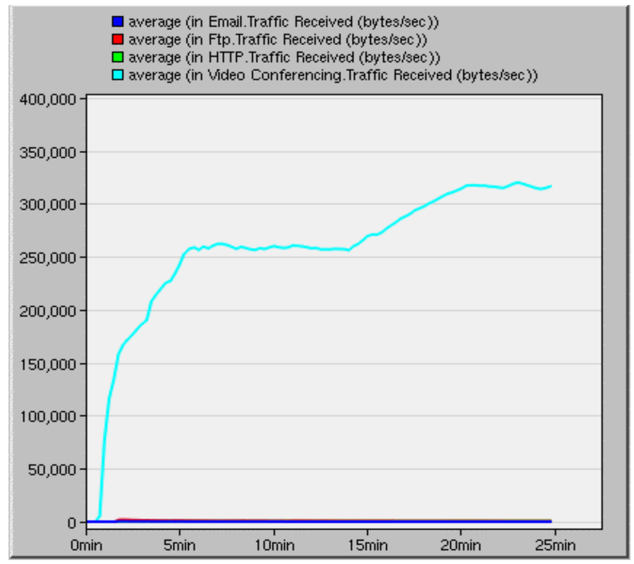

Figure 13: Average throughput for HTTP, FTP, email, and video/audio conferencing traffic.

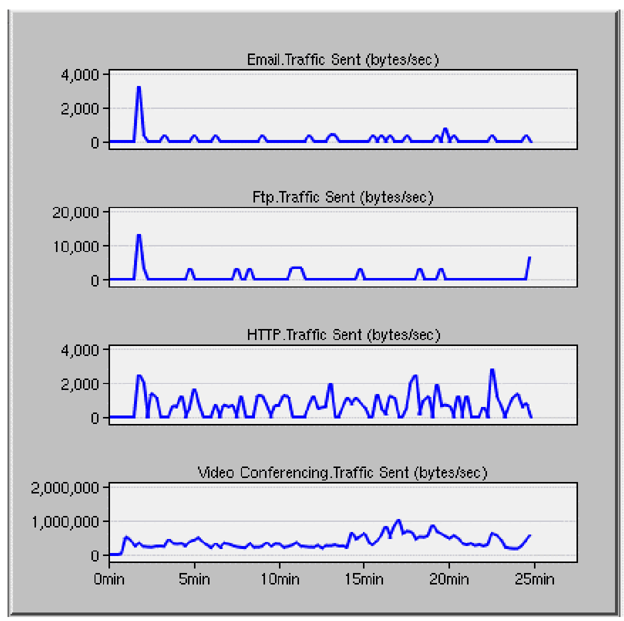

Figure 14: Instantaneous throughput for email, FTP, HTTP, and video conferencing applications.

The throughput of HTTP, FTP, and email is shown in Figure 15. All three applications are configured for heavy traffic load. Throughput of access category FTP is higher than HTTP and email. FTP is intended for transferring files and has better error checking and faster overall throughput while HTTP is designed to retrieve web pages. The model was configured to stream audio and video contents, HTTP, FTP, and email to all client subscribers. The movie was encoded at a rate of 50 fps. The VoD server sends unicast video/audio packets at a rate of 50 pps for each client. Figure 16 confirms the expected behavior of traffic sent [13].

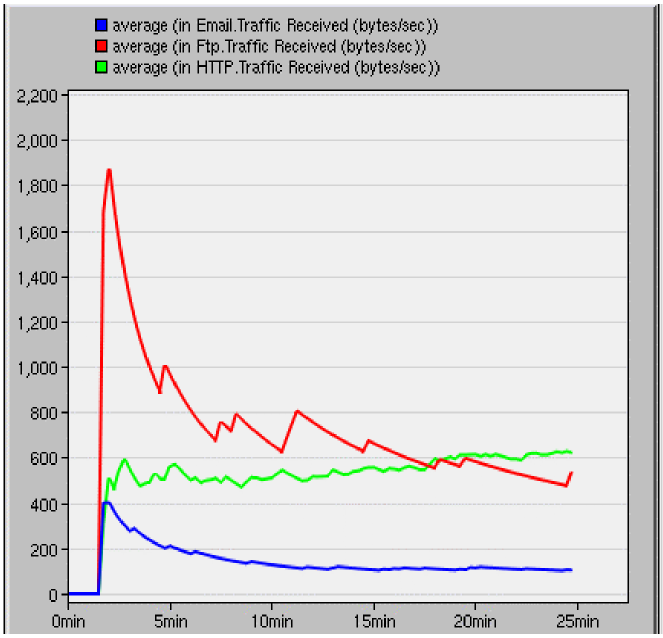

Figure 15: Average throughput for HTTP, FTP, and email application.

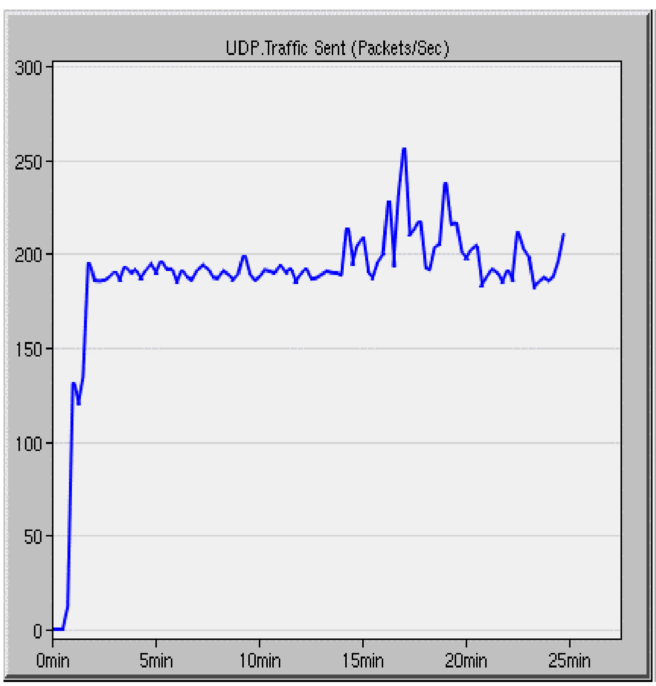

Figure 16: Traffic sent by the video server.

4.2 MPEG-4 video/audio, HTTP, FTP, and email streams

4.2.1 WiMAX Link Characteristics

The acquired PHY layer data provide insight in the operation of the WiMAX access network. The packet drop rate of the PHY layer for the three WiMAX client stations is shown in Figure 17. Smaller drop rate occurs in the WiMAX client station closest to the BS. The 6 km WiMAX station exhibits a much higher drop rate than the 2 km and 4 km stations during 25 minutes simulation interval. Figure 18 shows downlink SNR for three WiMAX stations. Note that the 6 km station has high drop rate due to low SNR, which is the necessary minimum level for 16-QAM with ½ coding [14].

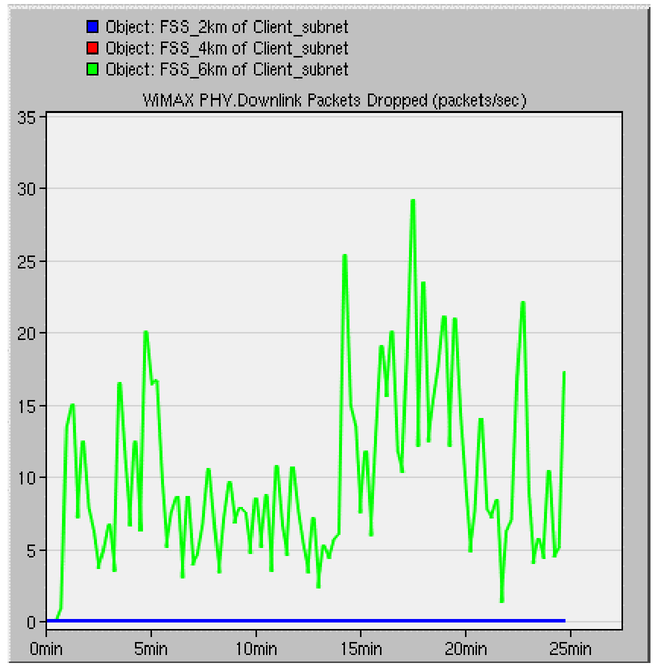

Figure 17: Packet drop rate of the PHY layer for the three WiMAX client stations.

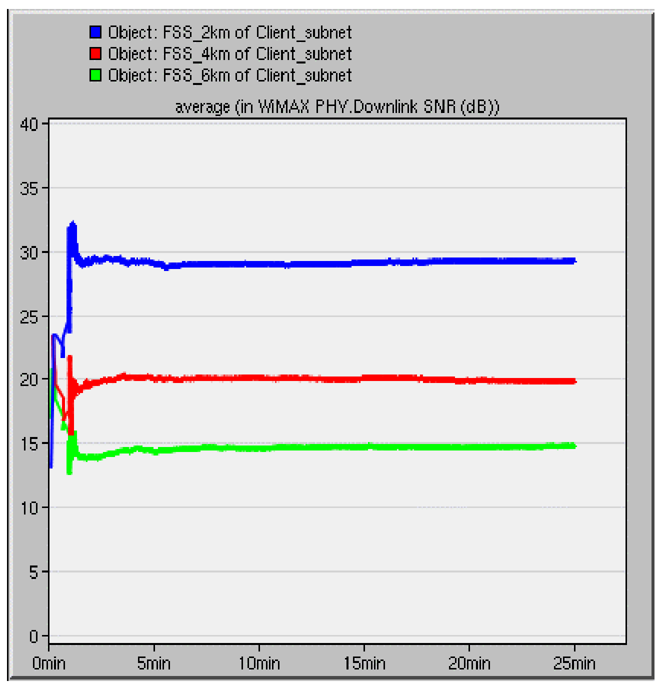

Figure 18: Downlink SNR for three WiMAX client stations.

4.2.2 Block Error Rate (BLER)

Block error rate (BLER) is the number of inaccurately transmitted data packets divided by the number of successfully transmitted packets. The downlink BLERs for the 4 km and 6 km WiMAX stations is shown in Figure 19. The WiMAX station nearest to the BS reflects smaller BLER. The 6 km station is expected to exhibit a higher BLER given that it is further from the BS than the 4 km station. Simulation results for three WiMAX client stations show that SNR is inversely proportional to BLER.

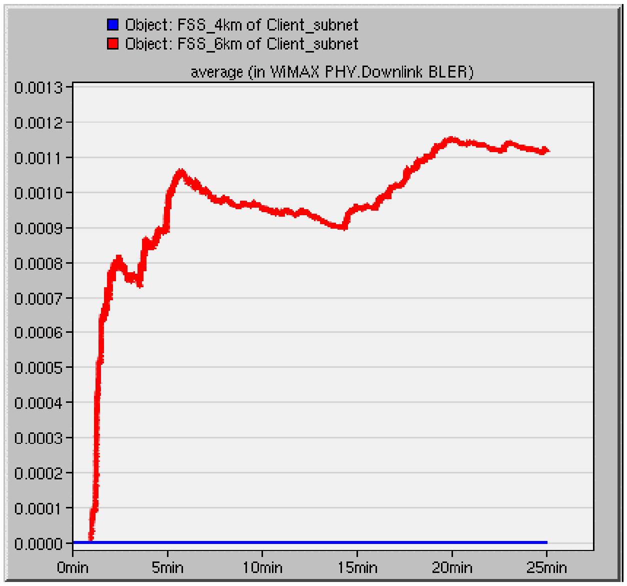

Figure 19: Comparison of downlink BLER for the 4 km and 6 km WiMAX stations.

4.2.3 Packet Loss

Packet loss for the ADSL client and the three WiMAX clients is averaged over the simulation of 25 minutes of the movie trace. Figure 20 illustrates deviation from the 50 pps value shown on the vertical axis. The VoD sending rate of 50 pps is almost matched by a received packet rate achieved by the ADSL client curve. As the WiMAX station distance from BS increases, the simulation results show expected degraded behavior. Figure 21 shows the same packet loss using instantaneous values [15].

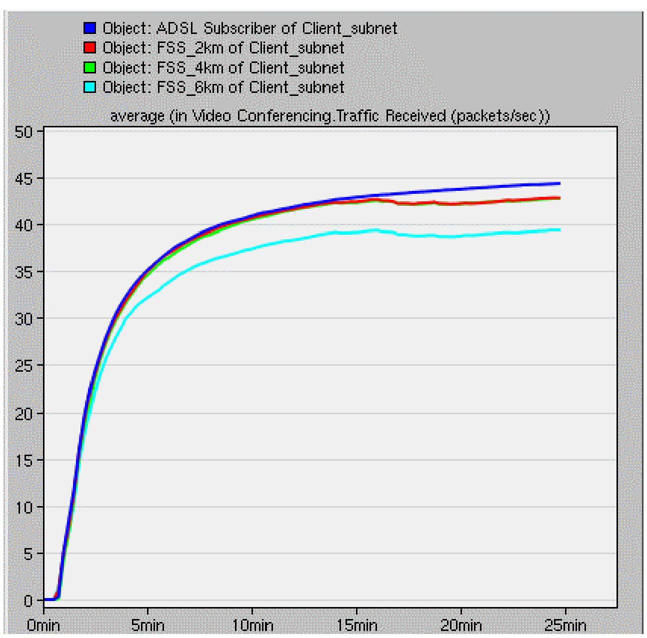

Figure 20: Average packet loss of four client stations during 25 minutes of simulation.

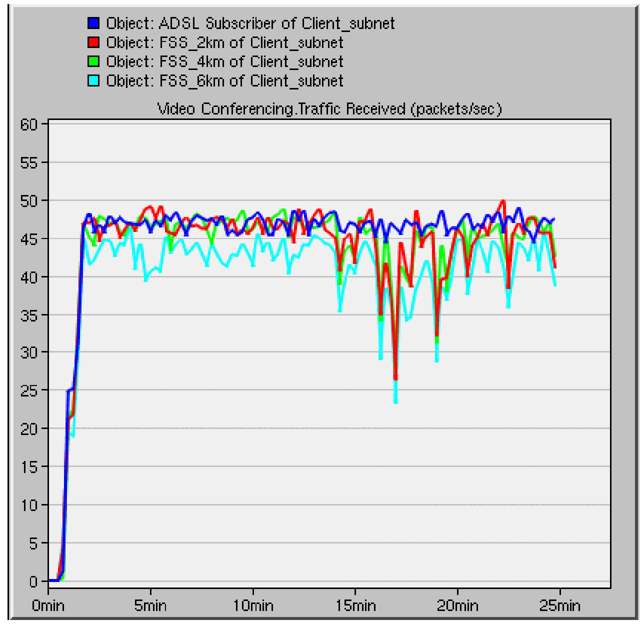

Figure 21: Instantaneous packet loss of four client stations during 25-minute of simulation.

WiMAX is designed to guarantee QoS to data flows regarding delay, packet loss, jitter, and bandwidth reservation. Additional simulations are performed in order to evaluate the effect of the packet loss in the WiMAX stations. Figure 22 captures the 2 km WiMAX station packet loss rate along with the MAC layer loss rate statistic from the BS. The MAC layer in the BS is losing a significant number of frames because the BS queue size reaches 128 kilobytes, as indicated in Figure 23. WiMAX stations at 4 km and 6 km distances exhibit similar behavior [16].

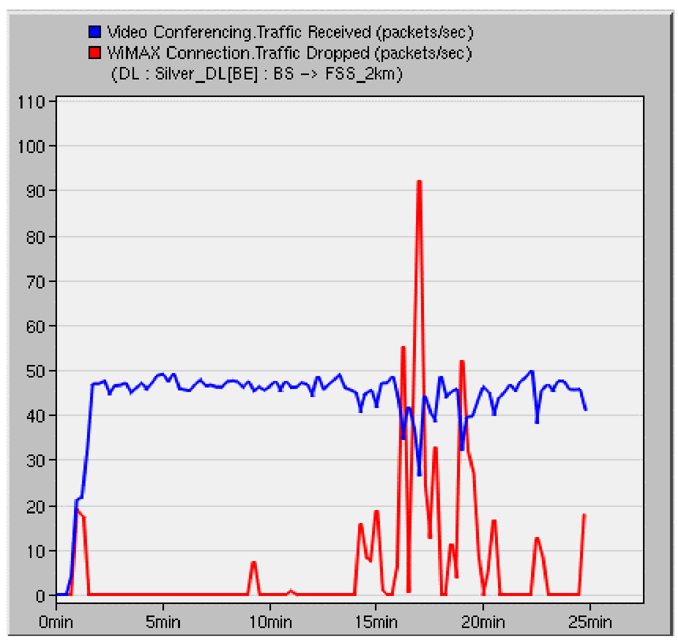

Figure 22: Received and dropped pps for the 2 km WiMAX station with 128 kilobyte buffer.

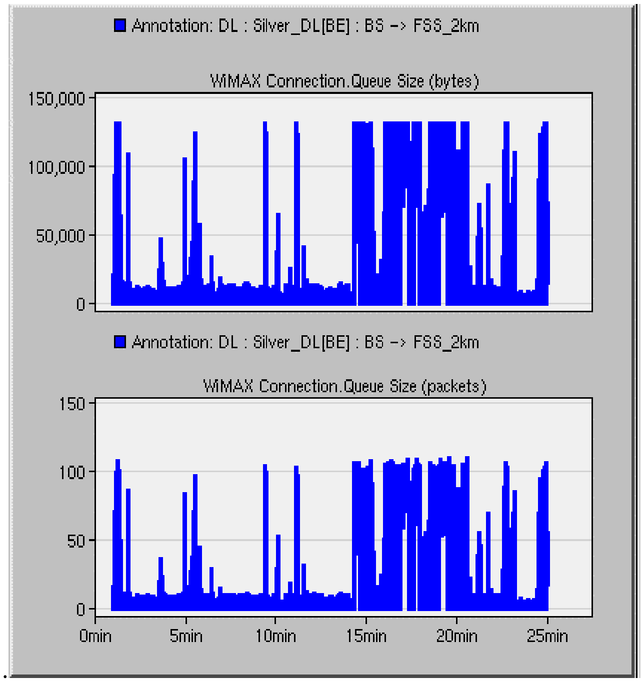

Figure 23: BS downlink queue at 2 km WiMAX station for 128 kilobytes buffer.

4.2.4 Delay

Network delay is measured as the time taken for a packet to be transmitted from source to destination. Averaged end-to-end delay for four clients over the simulation of 25 minutes movie trace is shown in Figure 24. Results show that the ADSL client experiences the delay of 10 ms.

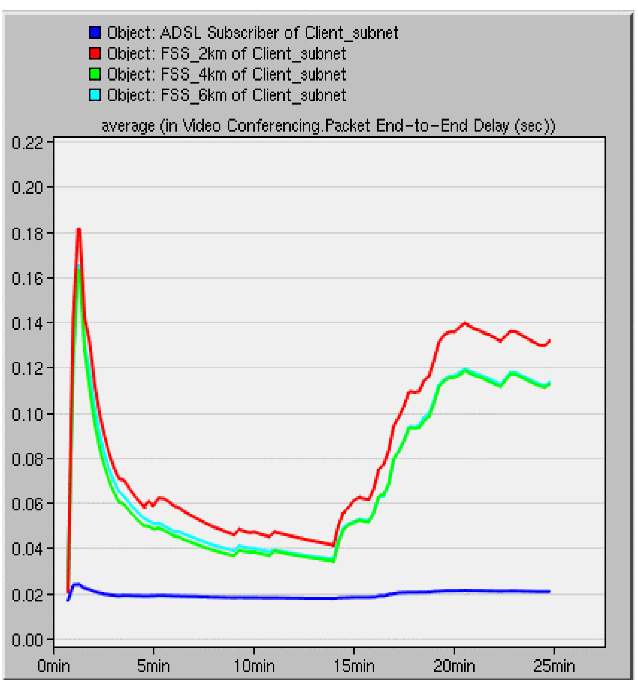

Figure 24: Average packets transmitted from source to destination in four WiMAX stations.

4.2.5 Jitter

Network jitter is an important QoS factor. The four video/audio client curves are averaged over the 25 minutes simulation of the movie trace. The simulated packet jitter is shown in Figure 25. Simulation results indicate that the ADSL client performs better. The four WiMAX client stations exhibit similar behavior and have 20 ms jitter for the movie duration.

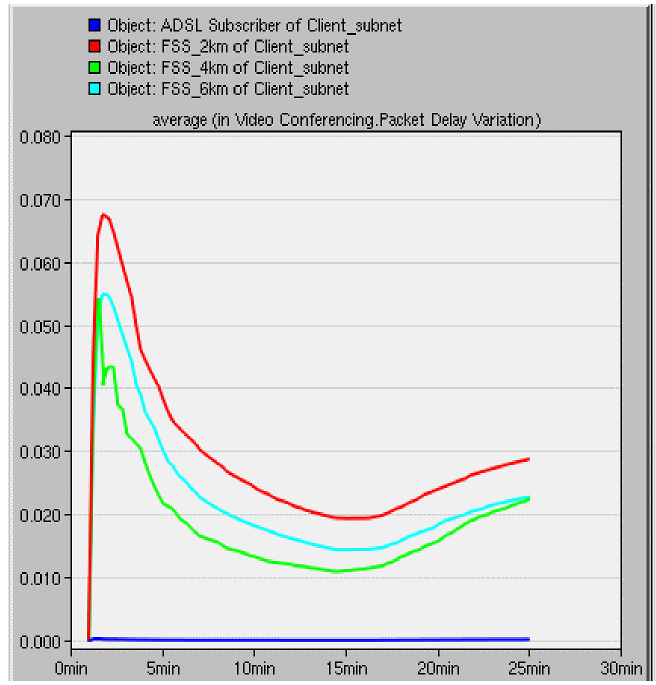

Figure 25: Packet jitter for the four WiMAX stations.

4.2.6 Throughput

Throughput of four clients is evaluated for simulation of 25 minutes of the movie trace and has similar behavior, as expected. The 2 km station displays better throughput performance than the ADSL station. The simulated throughput ranges between 0.40 Mbps and 0.72 Mbps, as shown in Figure 26 [17].

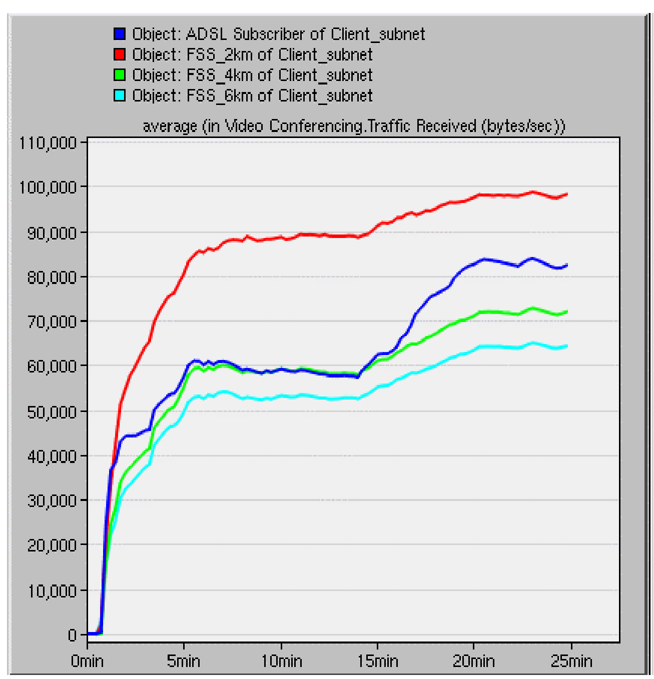

Figure 26: Minimum throughput.

4.3 Tuning the Buffer Size

WiMAX is connection oriented and, hence, a connection establishment is required before transmitting packets over the network. WiMAX substation may have several connections simultaneously. With a traffic policing algorithm, packets from a higher layer are handled and placed in a queue that shares the buffer memory with other queues. For multiple queues that are assigned with a common buffer memory, there may be degradation of QoS in terms of packet loss unless a suitable buffer management algorithm is employed. Prevention parameters for network congestion are altered by buffer tuning techniques over high latency and high bandwidth networks. Buffers up to 128 kilobytes were adequate for slow links or links with small round trip times (RTTs). However, if excessive traffic is sent, the buffer will overflow and the packet transfer will be interrupted. We introduced additional tuning of buffer size in the BS to explore its impact on packet loss rate and, ultimately, the video packet loss statistic. Various queue sizes ranging from the default value of 128 kilobytes to 1,024 kilobytes were employed. The 1,024 kilobyte buffer results in no MAC packet loss rate and, hence, it solves the buffer overflow issue. The enhanced performance of the 2 km and 4 km WiMAX stations is shown in Figure 27.

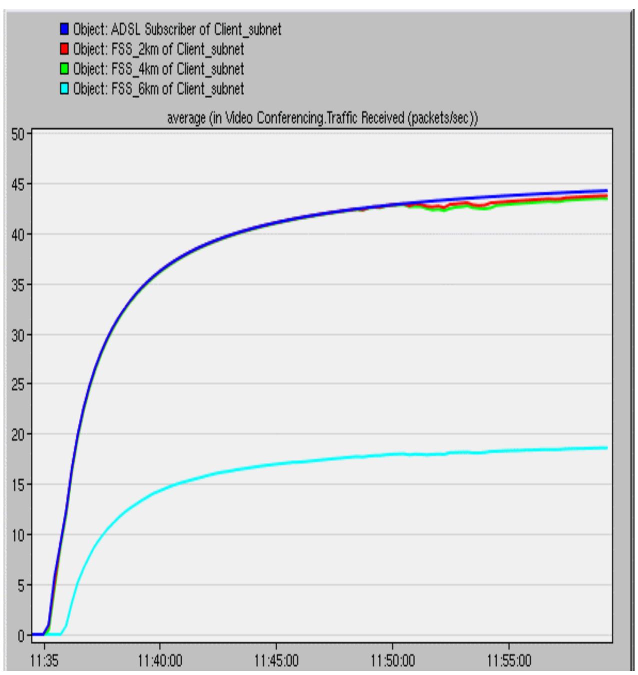

Figure 27: Average received pps with 1,024 kilobytes buffer.

The distant 6 km WiMAX station exhibits undesirable high packet loss rates, mainly due to the minimum SNR level that was essential for the configured modulation/coding pattern. The same loss performance factor using instantaneous values is shown in Figure 28.

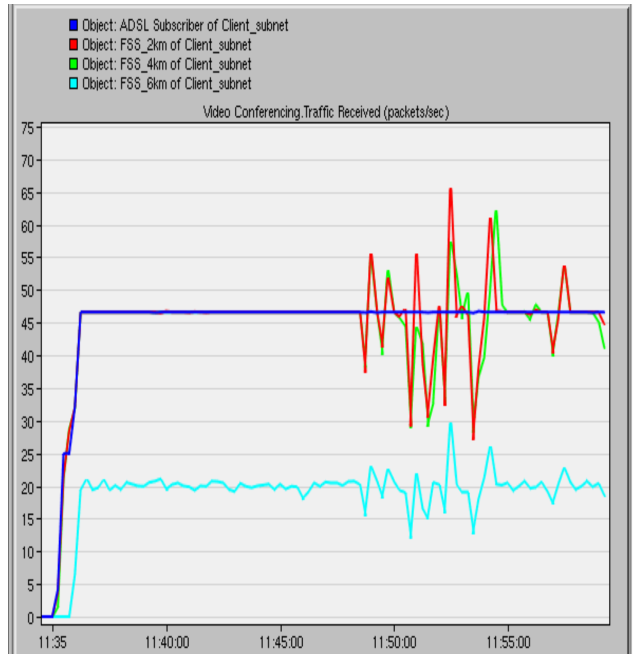

Figure 28: Instantaneous received pps with 1,024 kilobyte buffer.

Further observation of the 2 km WiMAX station reveals that the received video rate closely follows the original encoding and transmission rates, as shown in Figure 29. The BS connection queue never reaches the buffer capacity of 1,024 kilobytes, as shown in Figure 30. WiMAX stations at 4 km and 6 km distances also exhibit similar behavior.

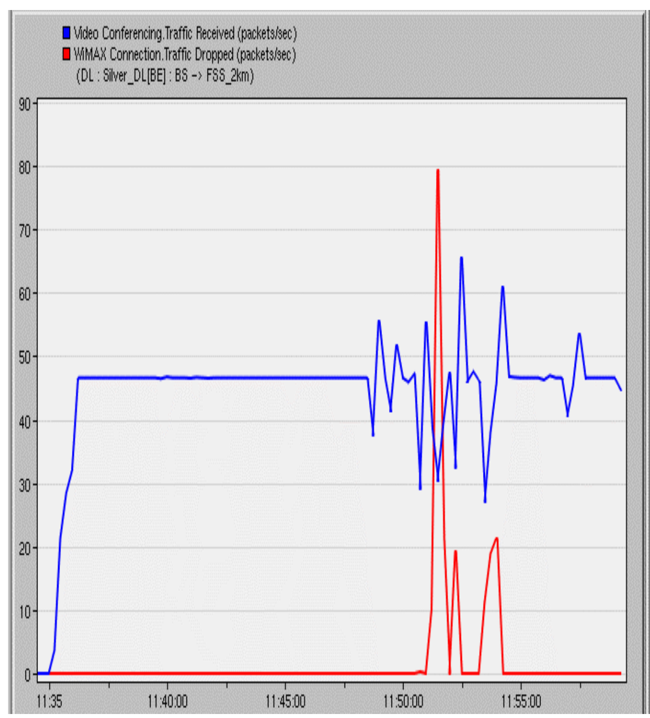

Figure 29: Received and dropped pps for 2 km WiMAX station with 1,024 kilobytes buffer.

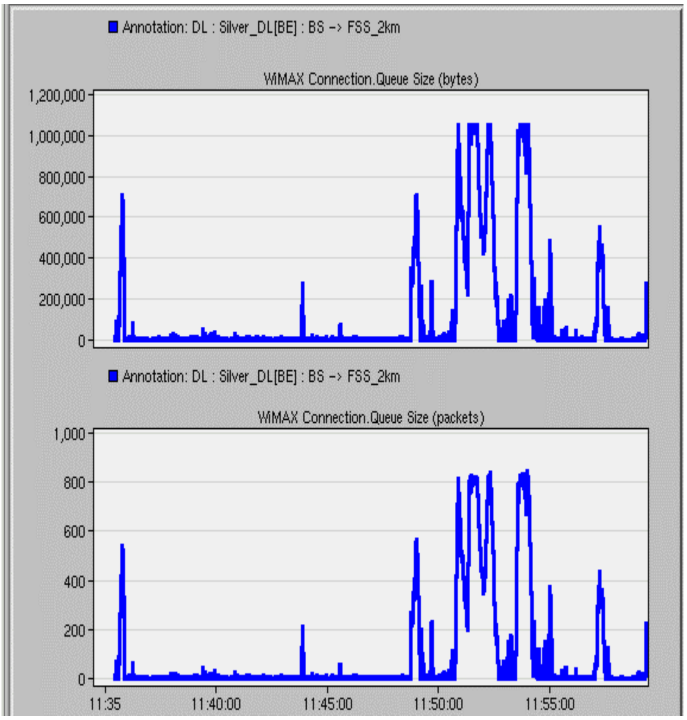

Figure 30: BS downlink queue at 2 km WiMAX station for 1,024 kilobytes buffer.

5. Conclusions

In this paper, we conducted extensive simulations of ADSL and WiMAX wireless networks and compared their performance by varying the attributes of network objects such as traffic load and by customizing the physical characteristics to vary BLER**,** packet loss, delay, jitter, and throughput. The study employed OPNET Modeler version 16.0 to design and describe the performance of WiMAX and ADSL.

The validation scenario confirms the overall design of the study that was implemented using OPNET Modeler. Simulation results demonstrated considerable packet loss. ADSL exhibited considerably better performance than the WiMAX client stations. To improve the overall performance of the network, we employed various buffer sizes. With further tuning, we derived a configuration that demonstrated packet loss that was more comparable to the ADSL client station. Small queues reduce delay, which is essential for real-time traffic. Such queues are required for video and audio applications that are sensitive to delay and jitter. Non-real-time traffic such as electronic mail, file transfers, and backups should be serviced by larger queues. The WiMAX file transfer performance, while almost ideal, degrades dramatically when the distance between workstations increases.

The OPNET Modeler provided a suitable environment to design and characterize WiMAX networks. While all applications were simulated using unicast traffic, multicast video traffic may have yielded better performance.

References

[1] W. Hrudey and Lj. Trajkovic, “Streaming video content over IEEE 802.16/WiMAX broadband access,” *OPNETWORK,* Washington, DC, Aug. 2008.

[2] W. Hrudey and Lj. Trajkovic, “Mobile WiMAX MAC and PHY layer optimization for IPTV,” *Mathematical and Computer Modelling,* Elsevier, vol. 53, pp. 2119–2135, Mar. 2011.

[3] WiMAX Report [Online]. Available: http://www.wimaxforum.org/technology/ downloads/.

[4] K. Pentikousis, J. Pinola, E. Piri, and F. Fitzek, “An experimental investigation of VoIP and video streaming over fixed WiMAX,” in *Proc. Modeling and Optimization in Mobile, Ad Hoc, and Wireless Networks* (*WIOPT),* Berlin, Germany, Apr. 2008, pp. 8–15.

[5] D. M. Ali and K. Dimyati, “Performance study of the WiMAX uplink scheduler,” in *Proc. IEEE Malaysia International Conference on Communications (MICC),* Malaysia, Dec. 2009, pp. 831–835.

[6] O. Iosif, E. R. Cirstea, I. Banica, and S. Ciochina, “Performance analysis of uplink resource allocation in WIMAX,” in *Proc. IEEE International Conference on Micro Manufacturing (ICCOMM)*, Bucharest, Romania, June 2010, pp. 351–354.

[7] S. Tiraspolsky, A. Rubtsov, A. Maltsev, and A. Davydov, “Mobile WiMAX - deployment scenarios performance analysis,” in *Proc. 3rd International Symposium on Wireless Communication Systems (ISWCS),* Valencia Spain, Sept. 2006, pp. 353–357.

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

[9] F. Retnasothie, M. Ozdemir, T. Yucek, H. Celebi, J. Zhang, and R. Muththaiah, “Wireless IPTV over WiMAX: challenges and applications,” in *Proc. IEEE Wireless and Microwave Technology Conference (WAMICON,* Clearwater, FL, Dec. 2006, pp. 1–5.

[10] H. Juan, H. Huang, C. Huang, and T. Chiang, “Scalable video streaming over mobile WiMAX,” in *Proc. IEEE International Symposium on Circuits and Systems (ISCAS),* New Orleans, LA, May 2007, pp. 3463–3466.

[11] M. Aman, B. Sikdar, and S. Parekh “Scalable peer-to-peer video streaming in WiMAX networks,” in *Proc. IEEE GLOBECOM,* Honolulu, HI, Nov. 2009.

[12] J. Chen, W. Jiao, and Q. Guo “An integrated QoS control architecture for IEEE 802.16 broadband wireless access systems,” in *Proc. IEEE GLOBECOM,* St. Louis, MO, Dec. 2005, p. 3335.

[13] M. Hu, H. Zhang, T. A. Le, and H. Nguyen, “Performance evaluation of video streaming over mobile WiMAX networks,” in *Proc. IEEE GLOBECOM,* Miami, FL, Dec. 2010, pp. 898–902.

[14] WiMAX MAPS [Online]. Available: http://www.wimaxmaps.org/.

[15] D. Wu, Y. T. Hou, W. Zhu, Y. Zhang, and J. M. Peha “Streaming video over the Internet: approaches and directions,” *IEEE Transactions on Circuits and Systems for Video Technology (CSVT)*, vol. 11, no. 3, pp. 282–300, Mar. 2001.

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

[17] W. Kim and H. Song, “A novel combined packet scheduling and call admission control for video streaming over WiMAX network,” in *Proc. IEEE GLOBECOM*, Miami, FL, Dec. 2010, pp. 960–964.