ENSC 427: COMMUNICATION NETWORKS
ANALYSIS ON VOIP USING OPNET

FINAL PROJECT

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ABSTRACT

VoIP is a technology that permits communication calls to be made over the internet and it is expected to become the mainstream for communication due to its low cost. However, the quality of VoIP is mainly impaired by jitter, delay, packet loss and many other parameters. As a case study, we simulate a VoIP network and study the behaviour and quality of VoIP under different scenarios. Furthermore, we study all the potential parameters that can deteriorate the quality of VoIP. This document presents an informative description of our VoIP network and discusses many design and technical issues pertaining to the deployment of VoIP.

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1 INTRODUCTION

VoIP application is gaining popularity recently. Many people are finding it attractive and cost effective to merge and unify voice and data networks into one. Besides the cost issues, another advantage of VoIP is portability [15]. We can make and receive phone calls wherever there is a broadband connection and it is as convenient as e-mail [15]. Furthermore, there are many other features that make VoIP attractive. Call forwarding, call waiting, voicemail, and three-way calling are some of the services that are usually provided at no extra charge [15]. We can also send data such as pictures and documents at the same time we are talking on the phone [15].

1.1 WHAT IS VOIP

The primary concept of VoIP is very similar to using a microphone to record a voice and saving it in a computer memory. However, in VoIP, the audio samples are not stored locally. Instead, they are packed into data packets and sent over the IP network to another computer [1]. With the nowadays technology, VoIP call can be made from a computer, a special VoIP phone, or a traditional phone with or without an adapter.

In packet switched network, a message is always fragmented into many data packets that are then transmitted independently from. They usually arrive at the destination in an arbitrary order. This disorder for applications such as e-mail or downloading document is not a problem since the packets will be reassembled in the correct order once they all has arrived at the destination [1]. However, due to the real-time nature of VoIP, the reassembling procedure is prohibited. Therefore, the order of received packets is a significant issue in VoIP [1]. It is inefficient to wait for all packets arriving in an organized order; therefore, some packets may be dropped if they don’t arrive in time and this can cause short periods of silence in the audio stream and causing bad VoIP quality.

1.2 VOIP DETERIORATION FACTORS

The quality of VoIP is mainly impaired by jitter, delay, packet loss. Other parameters such as quality of service (QoS) and coding scheme also play important part in the quality of VoIP. Many researches have been pursued in order to improve the reliability and quality of VoIP communication. In this project, we study all the potential parameters that can deteriorate the quality of VoIP.

1.2.1 JITTER

Jitter is the variation of delay of each packet [3]. It is a very typical problem in packet switched network due to the fact that information is segmented into packets that travel to the receiver via different paths [6]. Jitter is measured by the variance of time latency in a network. It is caused by poor quality of connections or traffic congestion [6]. Sometimes it occurs when packets take different equal cost-links. It also occurs due to the dynamic change of network traffic loads [7]. Jitter can be tolerated in data networks because arriving packets can be buffered. However, for real-time applications, such as voice, jitter has an imposed upper limit. When a packet arrives beyond the upper limit, the packet is discarded [8]. This packet loss leads to quality impairment in VoIP [5]. In order to reassemble voice signal successfully, the receiving device must account for jitter [8].

1.2.2 END-TO-END DELAY

Delay is the time interval in which a packets travels from one node to another node. It is caused by the time for endpoint to create packets, by the time needed to fill data into packets, or the time to arrange digital data on a physical link [7]. VoIP is very sensitive to delay; thus, it must be controlled and managed. As mentioned previously, it is inefficient to wait for all packets arriving in an organized order; therefore, some packets may be dropped if they don't arrive in time and this can cause short periods of silence in the audio stream and causing bad VoIP quality. Ideally, the delay constraint for VoIP packets is not above 80ms [13].

1.2.3 PACKET LOSS

Packet loss is packets that are dropped in order to manage the network traffic. It is inevitable in IP networks and occurs for various reasons. For example, it occurs when routers or switch work beyond capacity or queue buffers over flow [7]. Dropped packets in VoIP are treated as noise. Although some applications may tolerate packet loss because they can wait until packets are retransmitted, some time-sensitive applications are not tolerant to packet loss, such as text telephones (TTY) application [4]. Packet loss must be managed or controlled in VoIP since it effect voice signal distortion [8].

1.2.4 INTERNET QOS AND CODING SCHEME

Different coding schemes used in telephony can cause different delay at the sources and the destinations for audio compression and de-compression, yielding different end-to-end delay. G.711, G729, and G723 are well-known codec scheme nowadays. Furthermore, Internet QoS in term of throughput and capacity will have a significant effect on the quality of VoIP.

2 PROJECT DESCRIPTION

In the VoIP network we simulate, there are two companies that are located in two different countries such as Vancouver in British Columbia and New York in the USA as shown in **Figure 1**.

Figure 1: Two Different Companies Located in Vancouver and New York by the Communication of VoIP

Each company occupies three floors and there are fifteen workstations on each floor. The local area network (LAN) structure for both companies is the same. Workstation on each floor can communicate with workstations on different floor within the same building using VoIP. To make our VoIP network more interesting, workstations on each floor can also communicate using VoIP with workstations on any floors in the second company located in New York. **Figure 2** describes communication flow between the two companies.

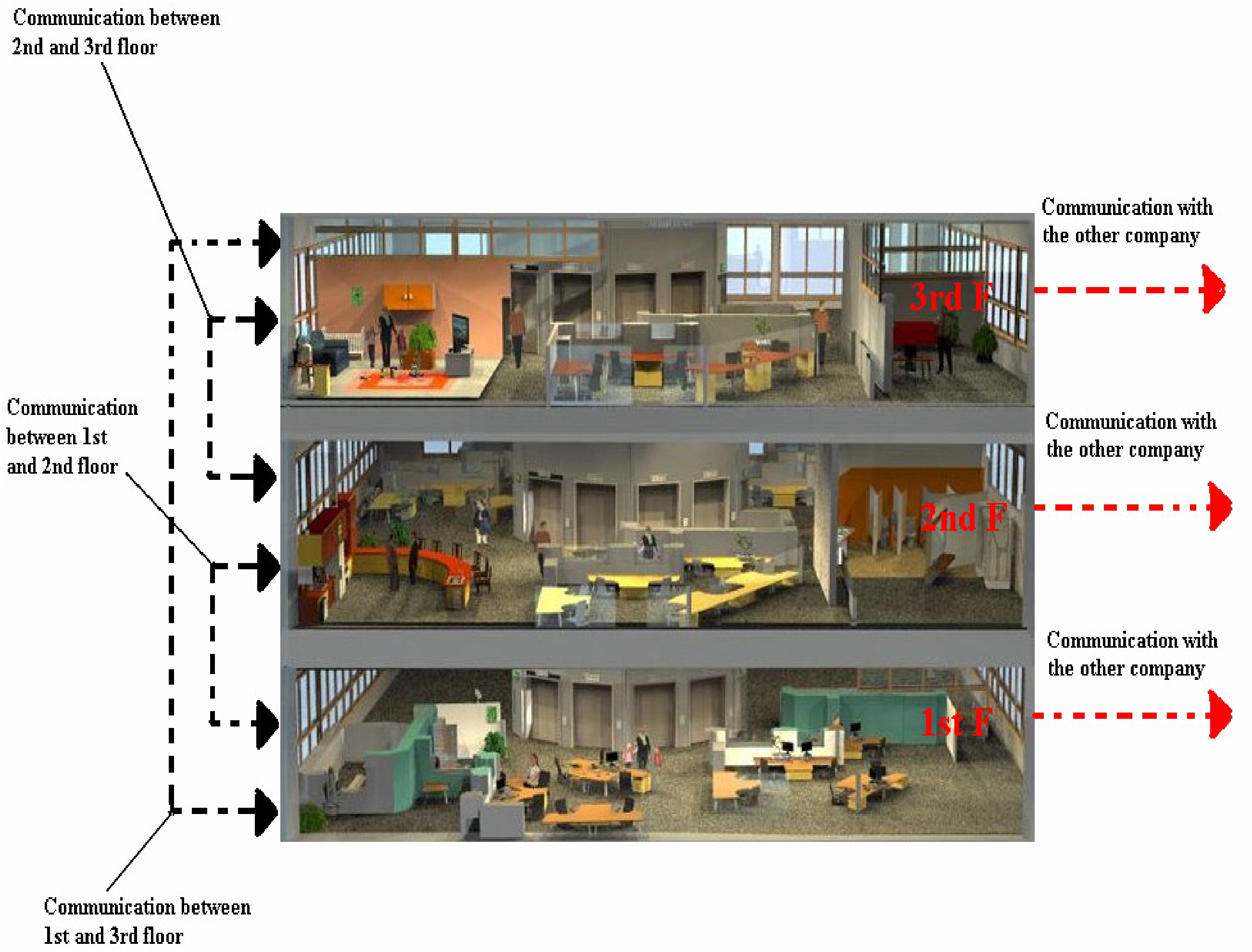


Figure 2: VoIP Communication within the Company and VoIP Communication with the Other Company

The purpose of building two LANs is because we want to simulate the communications within the same building and communications between the two different buildings as local and long-distance VoIP communication, respectively. Observing how parameters such as jitter, end-to-end delay and packet loss ratio change in both situations is a main focus of our project.

3 SIMULATION APPROACH

The simulation model for the VoIP network under study is illustrated in **Figure 3**. Two subnets are added on the map and each represents a company. The cloud symbol represents the Internet and the *Application Definition* and *Profile Definition* symbols on top are very important and will be explained later in the *Setting VoIP Application* section.

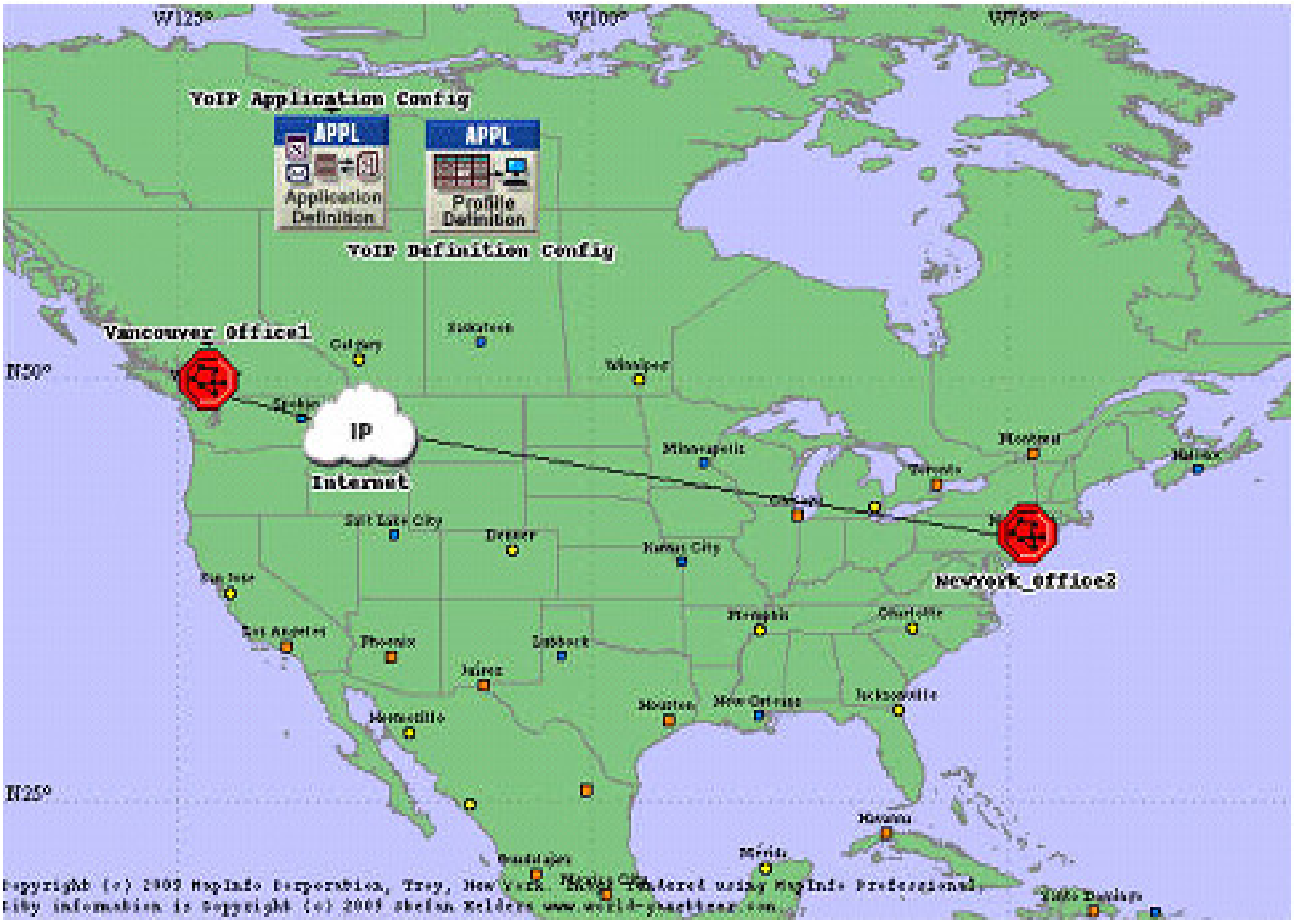
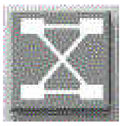
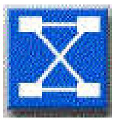
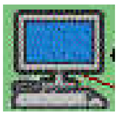
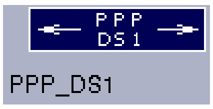


Figure 3: Simulation VoIP Network

LANs have been modeled as subnets that enclosed three floor LANs as shown in **Figure 4**. Each floor LAN contains an Ethernet switch and fifteen workstations as shown in **Figure 4**. **Table 1** shows the detail of components used in the LAN model.

Table 1: Components Used in the LAN Model

	Name	Description
	Cisco C4000 Router	This router is used as the main router in each company. It connects each company LAN to the Internet. This router has a forward rate of 14,000pps.
	Ethernet Switch	This switch is used as the switch for each floor LAN. It connects all the fifteen workstations together. This switch has a forward rate of 50,000pps
	Bay Networks Centilion100 Switch	This switch is used as the main switch in each company. It connects all floor switches. This switch has a forward rate of 6,400,000pps
	Ethernet workstation	This workstation support VoIP and it can be either the VoIP-packet sender or receiver.
	10 Base-T Duplex Link	All elements within the LAN have been connected using 10 Base-T links.
	PPP DS1 Duplex Link	This link is used to connect the main router in each company to the Internet.
	PPP DS3 Duplex Link	This Link is used to replace PPP DS1 Duplex Link in Overloaded VoIP Call scenario

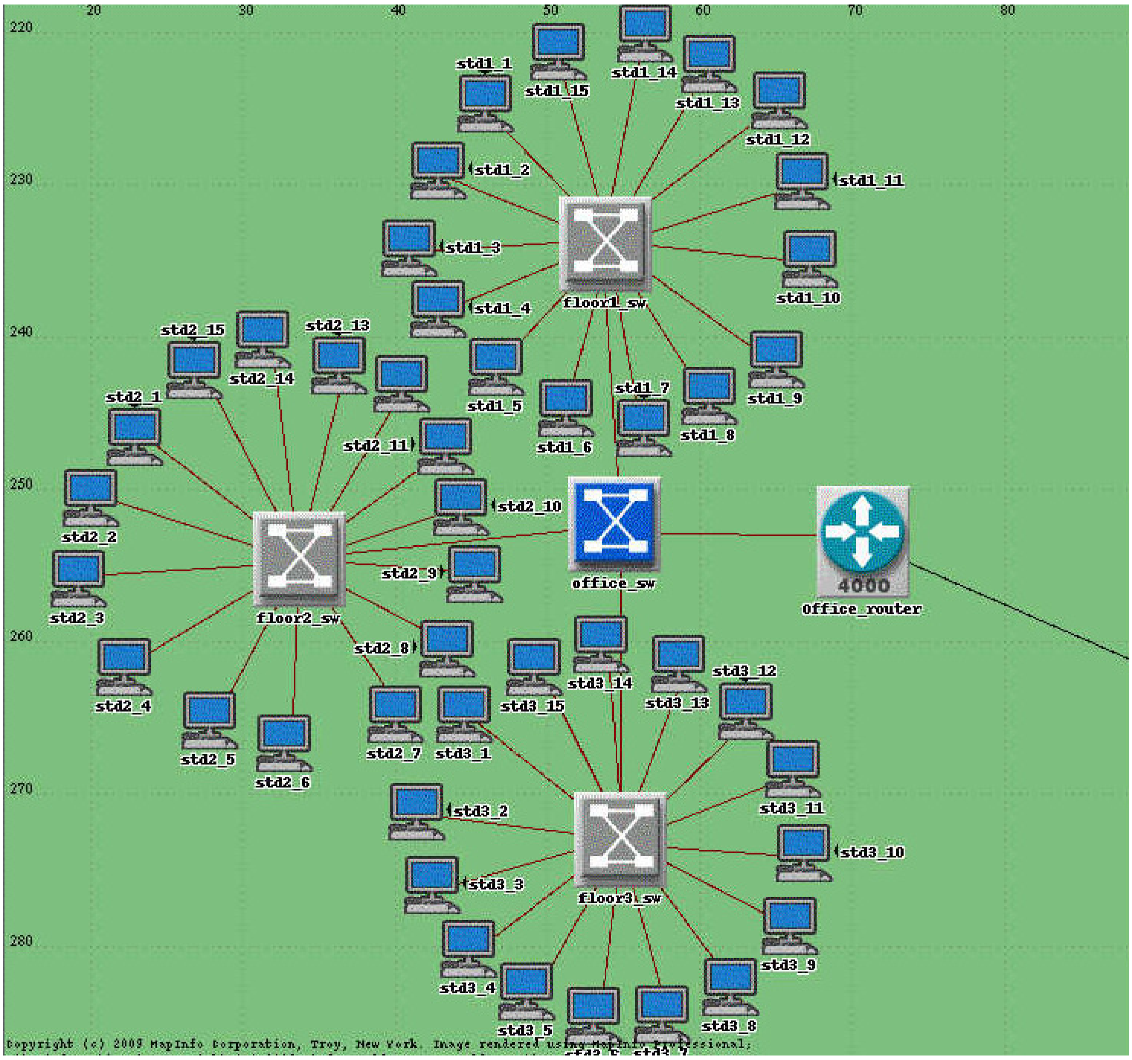


Figure 4: LAN Structure

4 SETTING VOIP APPLICATION

One way to assign the VoIP application to our model can be made under the *Application Definition*. The *Application Definition* provides a list of predefined applications as shown in the red rectangle in **Figure 5**. In the case of predefined VoIP application, users can change important attributes such as *Encoder scheme* and *Voice Frame per Packet* as shown in **Figure 6**.

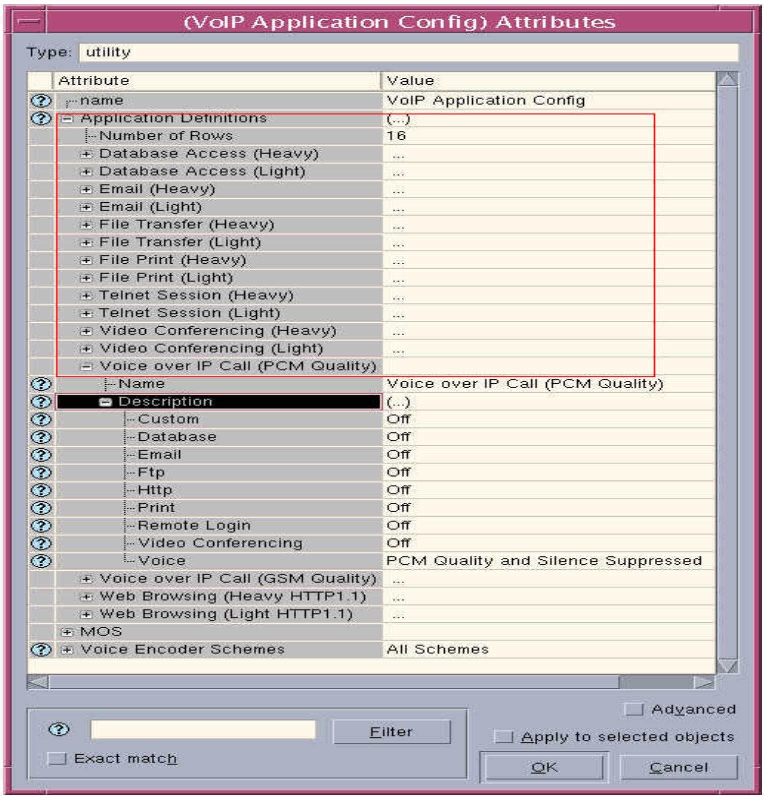


Figure 5: Application Definition

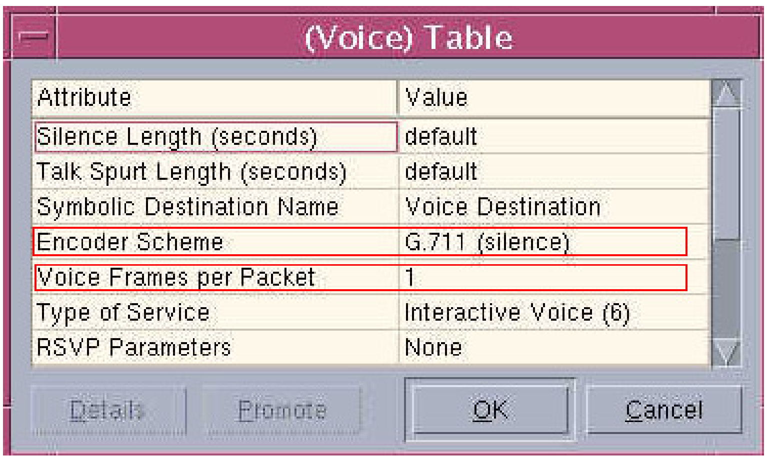


Figure 6: Voice Table Attribute

Initially, we set the *Encoder scheme* to G711 and the *Voice Frame per Packet* to one. Later in the project, we will change these attributes to see how they affect the behaviour of VoIP. A voice frame is defined as a collection of 32 audio samples of which each sample is one byte; thus, each audio sample has 32 bytes [14]. We set the *Voice Frames per Packet* to one since each packet in our VoIP network has a payload of 32 bytes.

It is essential to configure the workstations to adopt the VoIP application under *Application Definition*. This configuration can be made under the *Profile Definition* object. In our case, we need to define and configure only one profile as shown in **Figure 7**.

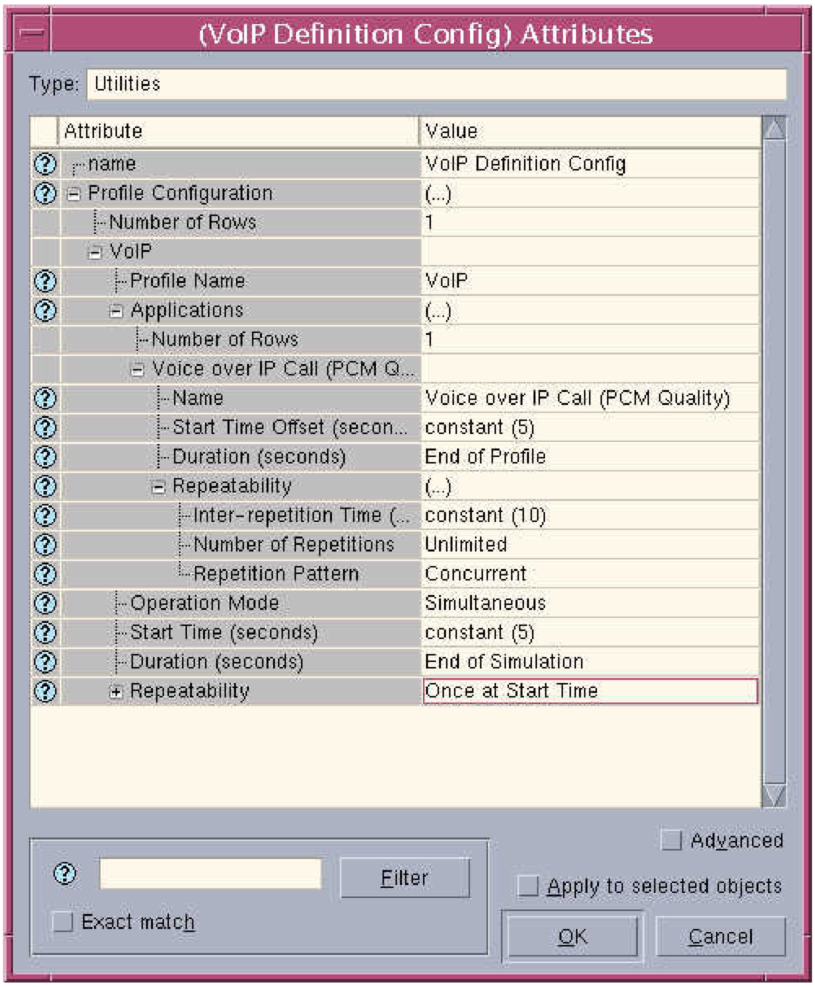


Figure 7: VoIP Definition Configuration

5 CONVERSATION PAIR

In order to implement local and long-distance VoIP calls, the first step is to define conversation pairs between the source nodes and the destination nodes. The conversation pairs can be defined in the traffic center. In the case of local VoIP calls, we set a conversation pair between two workstations within the Vancouver Company as shown in **Figure 8**. Furthermore, in the case of long-distance VoIP call, we set a conversation pair between one workstation in the Vancouver Company and another workstation in the New York Company as shown in **Figure 9**.

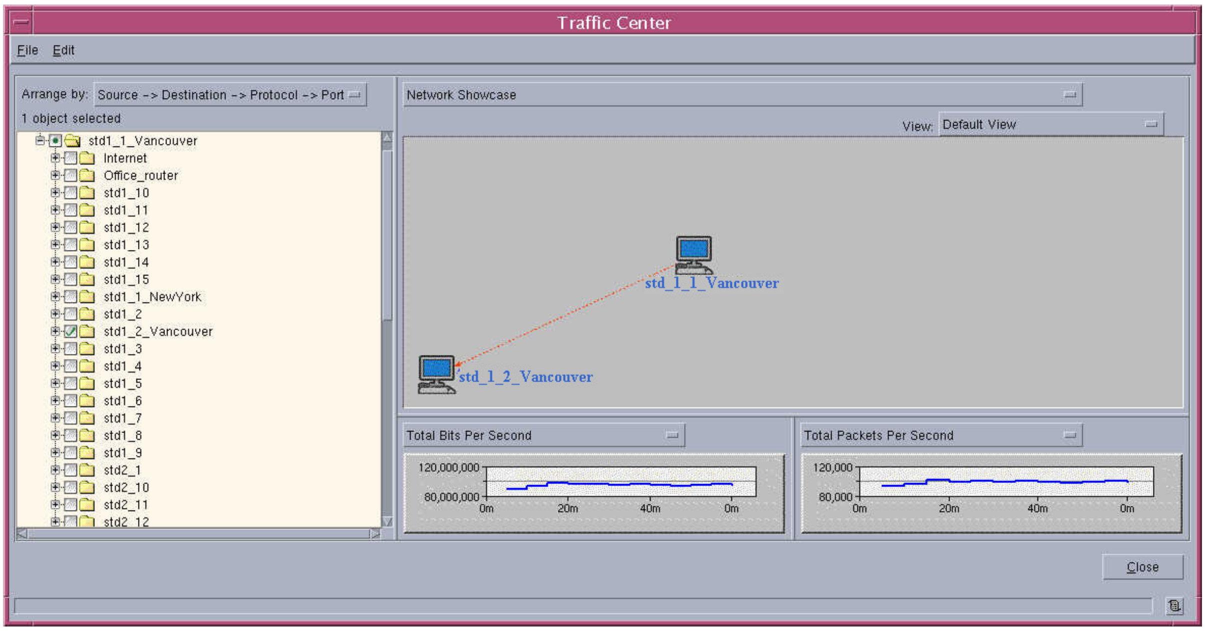


Figure 8: Local Conversation Pair Shown in Traffic Center

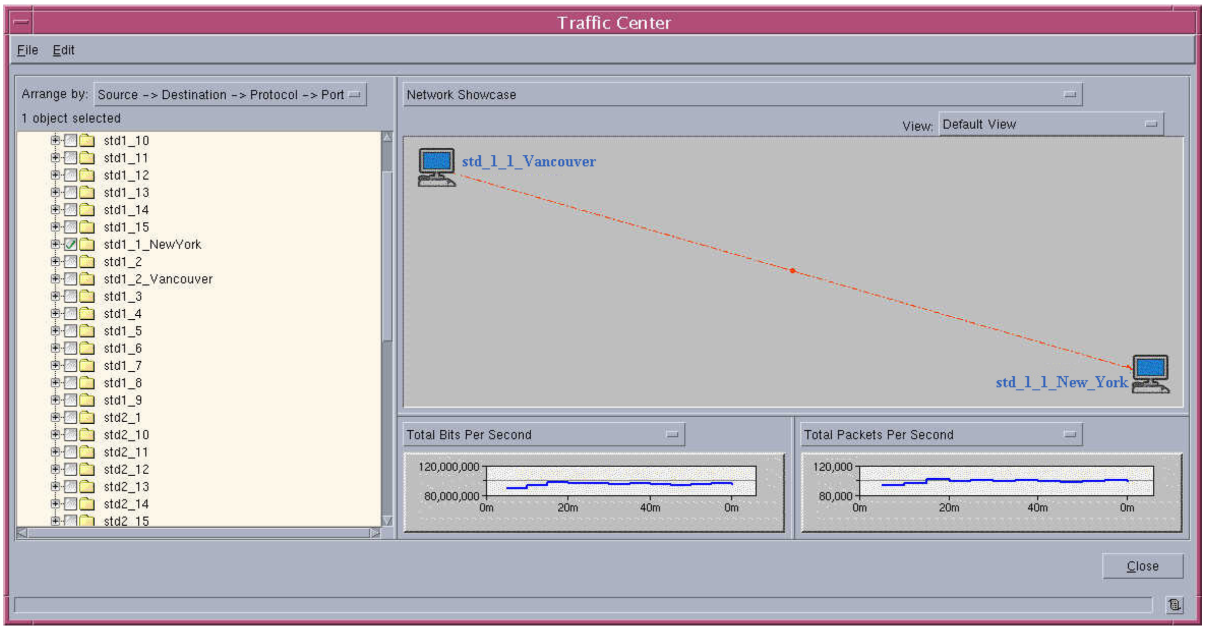


Figure 9: Long-Distance Conversation Pair Shown in Traffic Center

6 DISCUSSION

This section discusses the behaviour of VoIP under different scenarios. We create four different scenarios with names included in **Table 2**.

Table 2: Scenario Names

Scenario one	Comparison between local and long-distance VoIP communication
Scenario two	Comparison between a busy VoIP network and a non-busy VoIP network
Scenario three	Observation of VoIP quality under different discard ratio (Internet Qos)
Scenario four	Different encoder schemes usage and their effects on VoIP quality

It is a good idea to comprehend the definition of the parameters that we measure from scenario one to scenario four such as jitter, end-to-end delay, packet loss and Mean Opinion Score. **Table 3** summarizes the definition of the parameters. Further information can be reviewed in the *Introduction* section.

Table 3: Parameters to Measure for Scenarios

Jitter	Variation in packet arrival time
End-to-End delay	The time at which the source sends out the packets to the time the receiver gets the packets
Packet loss	Observation of VoIP quality under different discard ratio (Internet QoS)
MOS	Mean Opinion Score (MOS) value represents the user satisfaction. The higher the MOS value, the better the quality of the VoIP quality as shown in Figure 10.

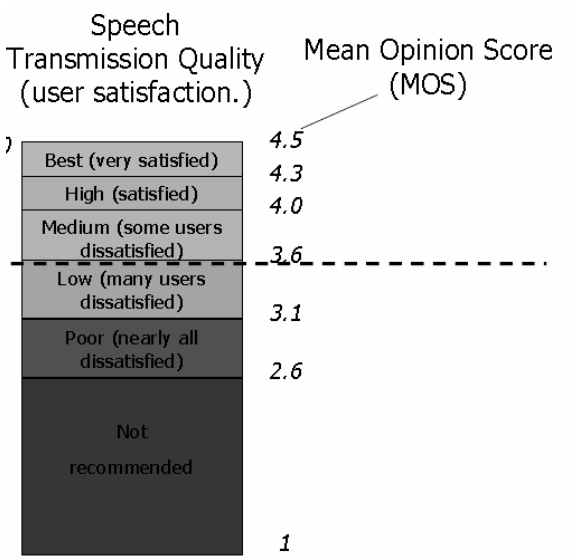


Figure 10: Speech Transmission Quality and Mean Opinion Score Ratings

6.1 SCENARIO ONE:

COMPARISON BETWEEN LOCAL AND LONG-DISTANCE VOIP COMMUNICATION

The purpose of this scenario is to compare local and long-distance VoIP calls in term of different parameters. We create one long-distance conversation pair between the two companies and two local conversation pairs within the Vancouver company – one conversation pair on the same floor and one conversation pair between two different floors as shown in **Table 4**.

Table 4: Conversation Pairs Set up for Scenario One

Vancouver_Floor1 workstation3 ---> Vancouver_Floor1 workstation4
Vancouver_Floor1 workstation1 ---> NewYork_Floor1 workstation1
Vancouver_Floor1 workstation2 ---> Vancouver_Floor3 workstation1

Figure 11 illustrates the jitter happens in the three conversation pairs. The blue line in **Figure 11** is the jitter in the long-distance conversation pair; whereas the red and green lines represent the jitter happen in the local conversation pairs on the same floor and the local conversation pair between different floors, respectively.

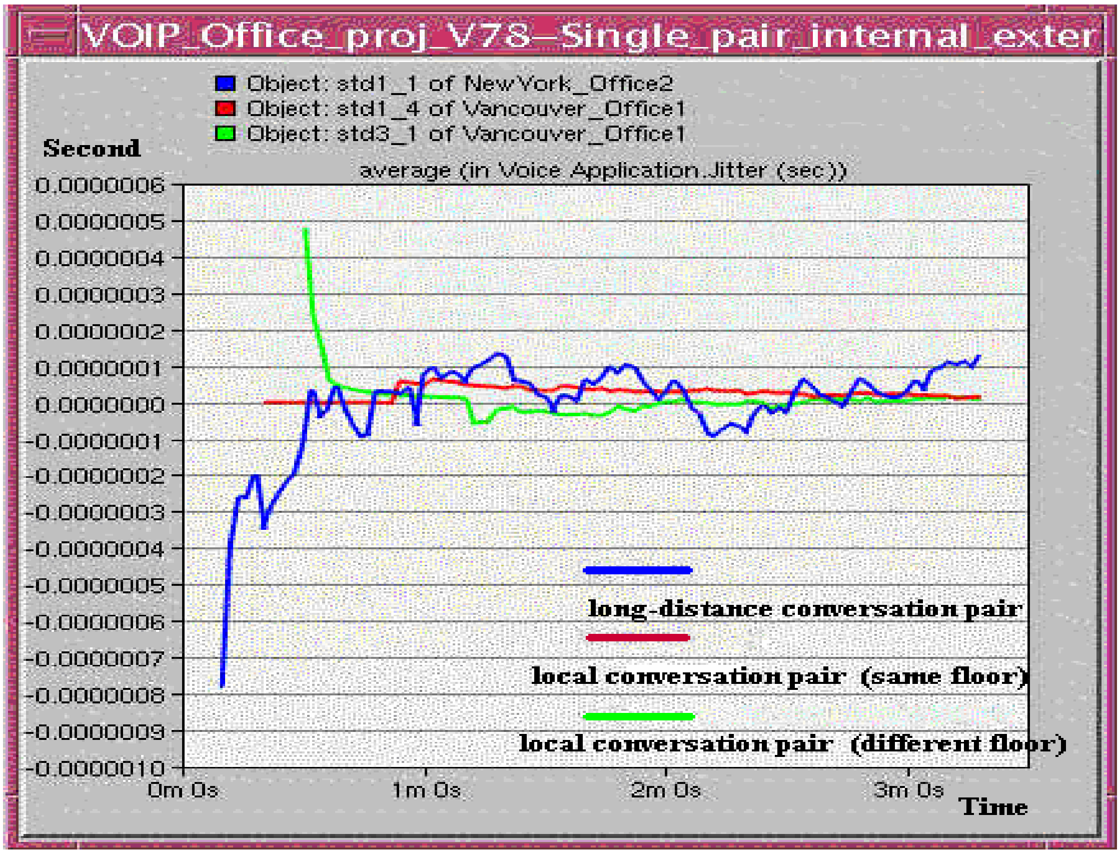


Figure 11: Jitter in Long-Distance Conversation Pair and Local Conversation Pairs

The long-distance conversation pair has the most fluctuating jitter; and the conversation pair happens on the same floor within the same company introduces the least fluctuating jitter. Jitter causes delay in the conversation. Voice packets arrived at the receivers with more fluctuating jitter have lower voice quality. If the absolute value of jitter is too large, then the callers and the receivers will notice the delay and the conversation become a walkie-talkie style conversation [15].

Figure 12 shows the end-to-end delay happen in the local conversation pairs and the long-distance conversation pair. The blue line illustrates the end-to-end delay in the long-distance conversation pair; whereas the red and green lines show the end-to-end delay in the local conversation pair on the same floor and the local conversation pair between different floors, respectively.

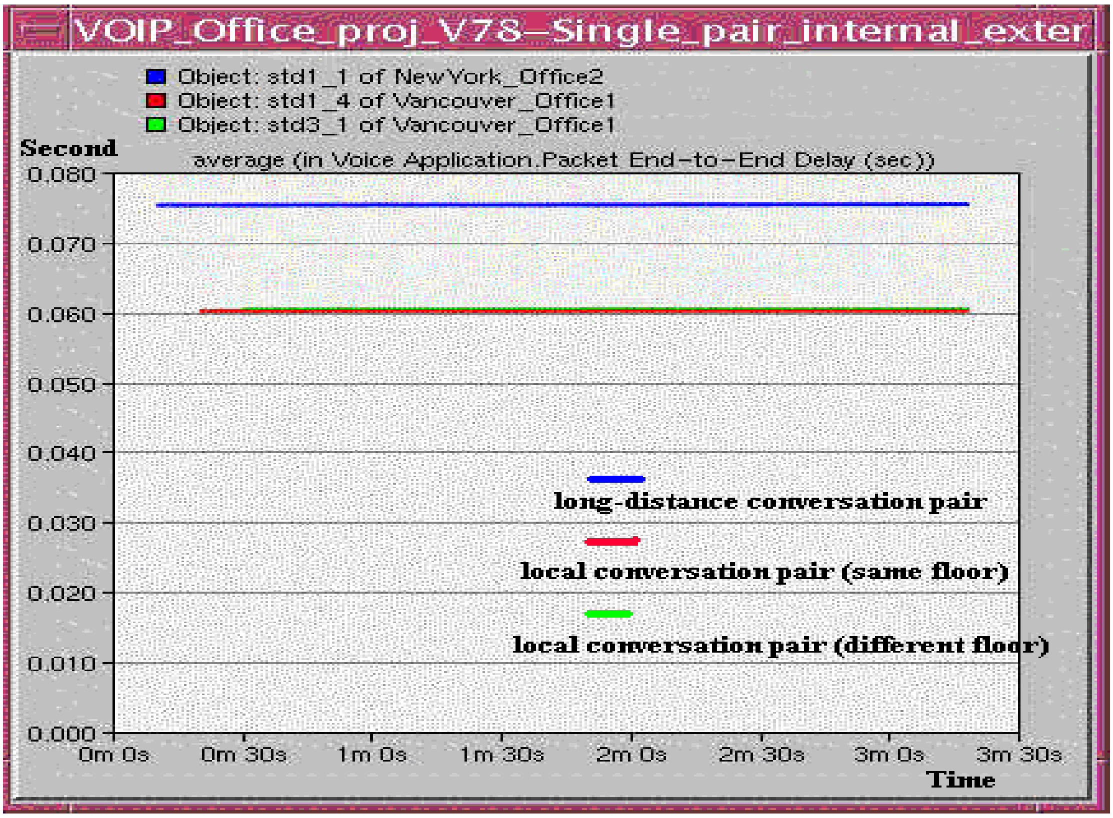


Figure 12: End-to-End Delay in Long-Distance Conversation Pair and Local Conversation Pairs

The end-to-end delay in all three cases does not exceed the time constraint – 80ms. The end-to-end delay in the long-distance call is longer than the end-to-end delay in two local conversation pairs. The result is reasonable as it is necessary to takes more time for the packets to travel from the source to the destination in the long-distance VoIP communication case.

The MOS of the long-distance conversation pair and the local conversation pairs are shown in **Figure 13**. The blue line illustrates the average MOS of the long-distance conversation pair; whereas the red and green lines show the average MOS of the local conversation pair on the same floor and local conversation pair between different floors, respectively.

The MOS of the local conversation pairs is higher than the long-distance conversation pair. This means the local conversation pairs have higher VoIP speech quality than the long-distance conversation pair.

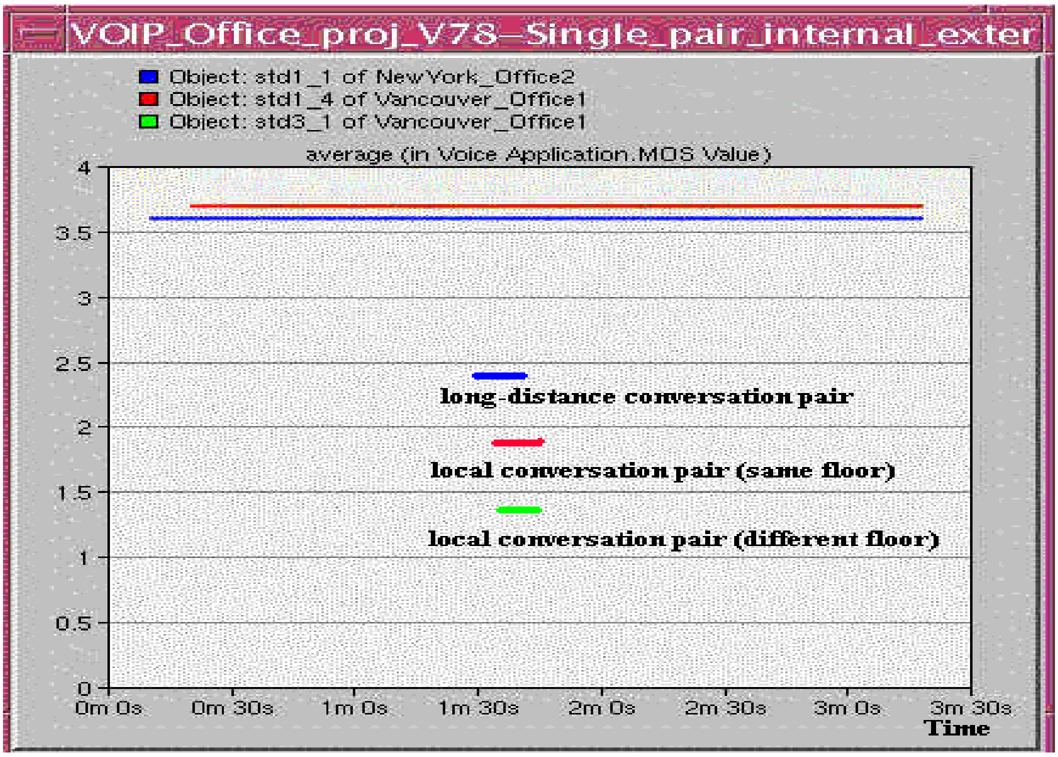


Figure 13: Mean Opinion Score of VoIP Call of Long-distance Conversation Pair and Local Conversation Pairs

No packets dropped have been observed in the long-distance conversation pair and the two local conversation pairs. Different parameters captured in the long-distance conversation pair and the local conversation pairs are summarized in **Table 5**.

Table 5: Result For Long-distance Conversation Pair and Local Conversation Pairs

Case	Jitter	End-to-End Delay	MOS Value	Packets Loss
Long-distance conversation pair	Most Fluctuation	0.0755 s	3.6	None
Local conversation pair (same floor)	Least Fluctuation	0.0605 s	3.7	None
Local conversation pair (different floor)	Medium Fluctuation	0.0605 s	3.7	None

From the result summarized in **Table 5**, it shows that the long-distance VoIP conversation pair tends to has more fluctuation in jitter, longer end-to-end delay and smaller MOS value compared with the local conversation pairs. Since jitter, end-to-end delay and MOS value are used to determine the VoIP quality; thus, the quality of long-distance VoIP communication is not as good as the quality of local-distance VoIP communication.

6.2 SCENARIO TWO:

COMPARISON BETWEEN A BUSY NETWORK AND A NON-BUSY VOIP NETWORK

In scenario one, we create a non-busy VoIP network in which there is only one long-distance conversation pair. The purpose of this scenario is to compare a busy VoIP network with a non-busy VoIP network in term of different parameters: jitter, end-to-end delay, packet loss, and MOS value. Furthermore, different link capacity is used in the busy VoIP network to see the change in the aforementioned parameters.

In order to create a busy VoIP network, 15 workstations in each company are set to communicate with 15 workstations in the second company – 15 long-distance conversation pair. The first 15 VoIP calls start after 10 seconds and each workstation will generate an additional call every 10 second after.

First of all, DS1 link is used to connect the subnets to the IP cloud. After that DS3 is used to replace the DS1 link to connect the subnets to the IP cloud with the same load. The throughput of DS1 is 1.544 Mbps; whereas the throughput of DS3 is 44.736 Mbps

Figure 14 shows the overall traffic sent and received of the busy VoIP network using DS1 link. The overall traffic received rate is slower than the overall traffic sent rate at the time around minute one and after. The mismatch of traffic send rate and traffic received rate implies that the DS1 link is overloaded by then.

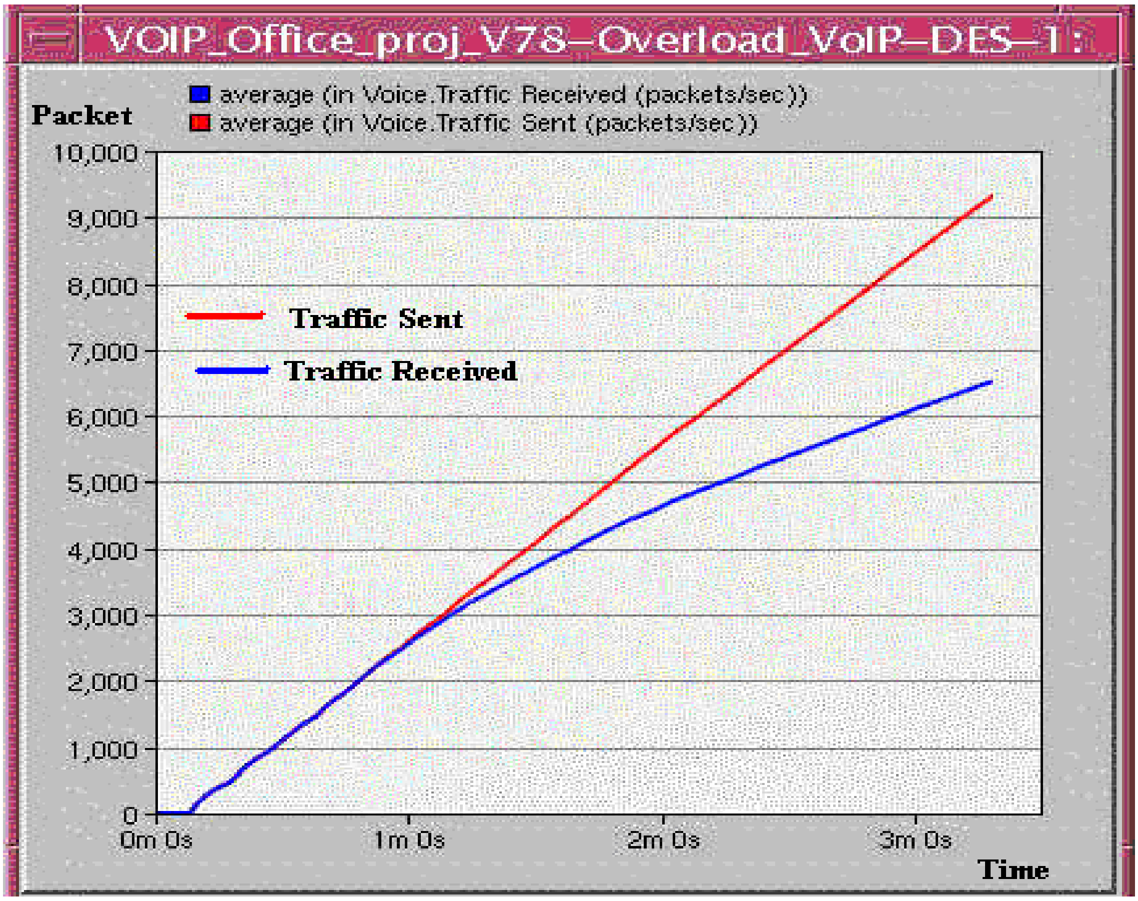


Figure 14: Overall Traffic Received Rate and Overall Traffic Sent Rate In The Busy VoIP Network Using DS1 Link

The overload slows down the throughput and increases the end-to-end delay for the packets as depicted by the red line in **Figure 16**. The increase in end-to-end delay causes many packets arriving at the destination over the time constraint, 80ms; thus, many packets are discarded. The mismatch of traffic sent and traffic received in **Figure 14** also implies packet loss.

The DS1 link is then replaced with the DS3 link in the busy VoIP network. In the case using the DS3 link in the busy VoIP network, the overload phenomenon disappears due to the larger capacity of the DS3 link. As a result, there is no rapid increase in end-to-end delay and packet loss.

Due to the light traffic in the non-busy VoIP network, there is no overload phenomenon in the network even using DS1 link.

The red and green lines in **Figure 15** represent the jitter of the busy VoIP network with DS1 link and DS3 link, respectively. The blue line shows the jitter of the non-busy VoIP network using DS1.

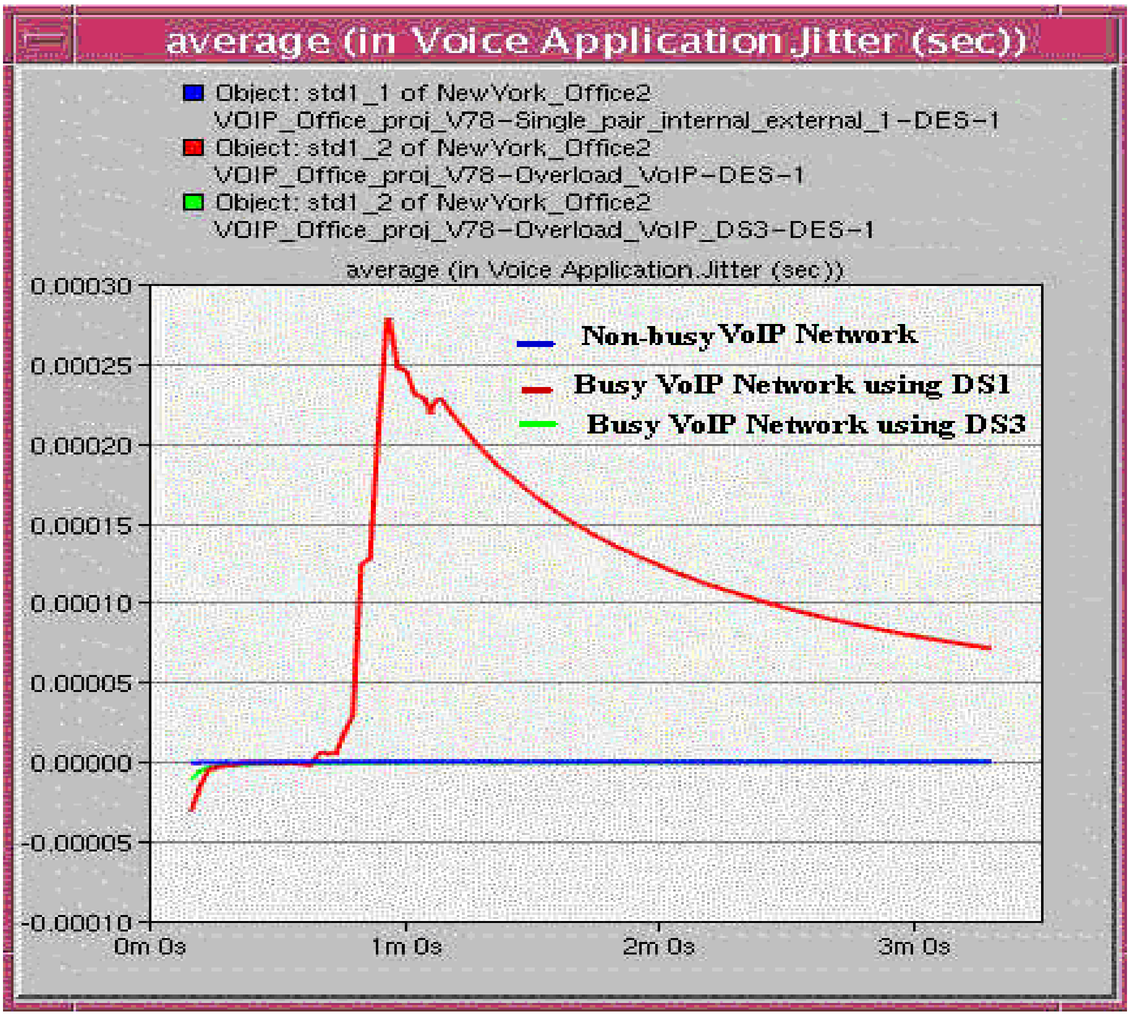


Figure 15: Jitter in Non-busy VoIP Network, Busy VoIP Network using DS1 and Busy VoIP Network using DS3

Figure 15 clearly shows that the jitter rapidly increases at the time around minute one when the DS1 link is overloaded in the busy VoIP network. This behaviour does not show in the cases of non-busy VoIP network and the busy VoIP network using the DS3 link. Similarly, in **Figure 16** there is a rapid increase in the end-to-end delay in the busy VoIP network using DS1 link at the time around minute one due to capacity overload causing slow throughput as mentioned previously.

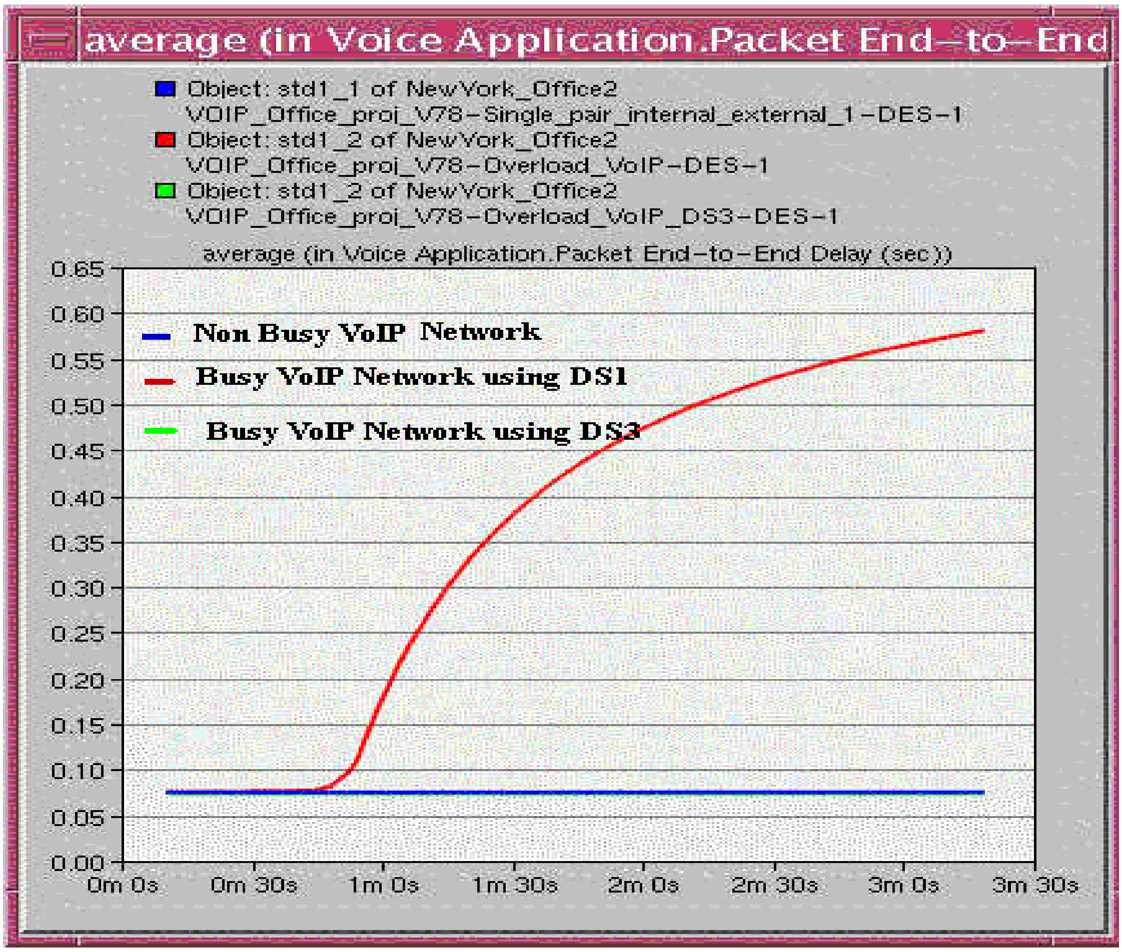


Figure 16: End-to-end Delay in Non-busy VoIP Network, Busy VoIP Network using DS1 and Busy VoIP Network using DS3

The increase in end-to-end delay and jitter around minute one results a decrease in the MOS value of the busy VoIP network using the DS1 link as depicted by the red line in **Figure 17**. The rapid MOS drop does not happen in the non-busy VoIP network and the busy VoIP network using the DS3 link due to the fact that there is no overload.

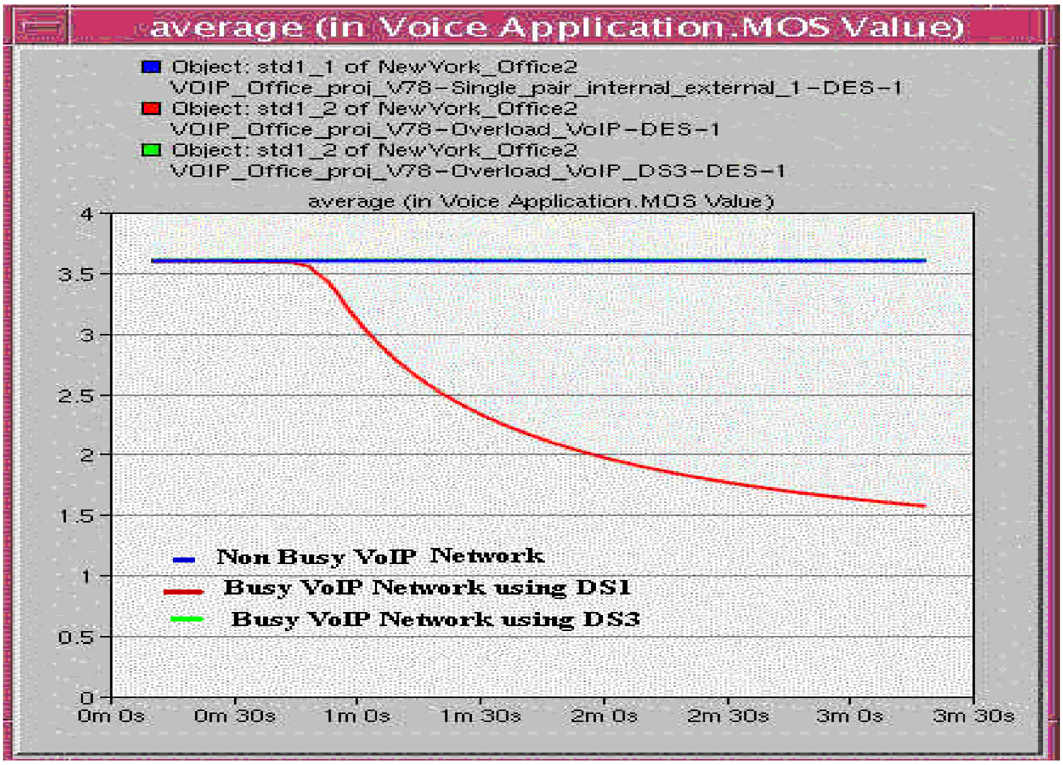


Figure 17: MOS of Non-busy VoIP Network, Busy VoIP Network using DS1 and Busy VoIP Network using DS3

The parameters captured from the non-busy VoIP network, the busy VoIP network using the DS1 link and the busy VoIP network using the DS3 link are summarized in **Table 6**.

Table 6: Result for Non-busy VoIP Network, Busy VoIP Networks using DS1 and DS3

Case	Jitter	End-to-End Delay	MOS Value	Packets Loss
Non-busy VoIP Network	Negligible Fluctuation	Constant 0.075 second	Constant 3.6	No Packets Dropped
Busy VoIP Network using DS1	Most fluctuation	Rapidly Increase When the Link Overloaded	Rapidly Decrease When the Link Overloaded	Packets Dropped Around One Minute And After
Busy VoIP Network using DS3	Negligible fluctuation	Constant 0.075 second	Constant 3.6	No Packets Dropped

From the result summarized in **Table 6**, we observe that the quality of VoIP deteriorates as the VoIP network is getting busy. When the VoIP network becomes busy, overload happens and causes larger fluctuation in jitter, longer end-to-end delay, lower MOS value and more packet loss. The solution is to change the link capacity. The replacement of DS1 link by the DS3 link eliminates the overload because the DS3 link has much faster data rate than the DS1 link. As a result, in order to improve the voice quality in a busy VoIP network, it is essential to use a high capacity link such as DS3, OC24 and OC48.

6.3 SCENARIO THREE:

OBSERVATION OF VOIP QUALITY UNDER DIFFERENT DISCARD RATIO (INTERNET QOS)

The purpose of this scenario is to observe how the Internet QoS affects the quality of VoIP. The discard ratio is used to differentiate the Internet QoS. According to OPENT, packet discard ratio specifies the percentage of packets dropped. It is the ratio of packets dropped to the total packets transferred to this cloud multiplied by 100.

We start with changing the packet discard ratio into 0.5%, 4% and 6% under the Internet Attributes of the IP cloud. From **Figure 18** to **Figure 21**, it shows jitter, end-to-end Delay, MOS value and packet loss corresponding to different packet discard ratios.

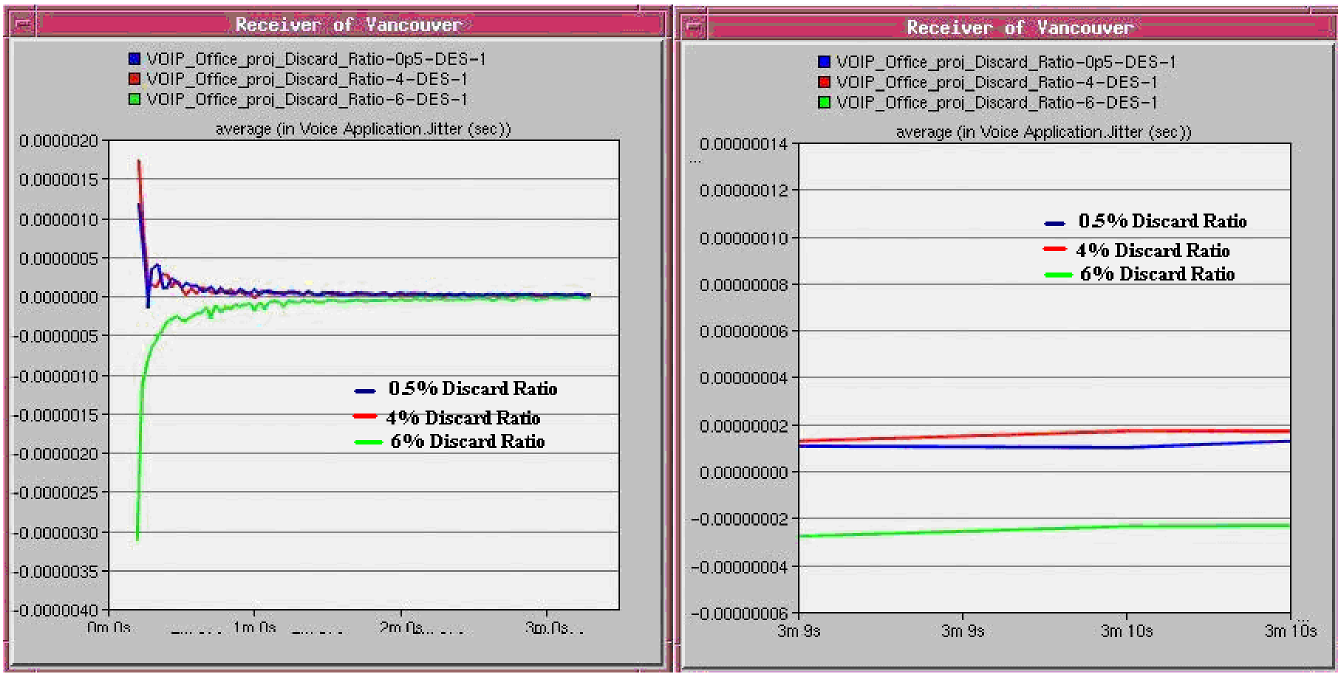


Figure 18: Discard Ratio Comparison--Voice Application Jitter (sec). Left: Original; Right: Zoom in

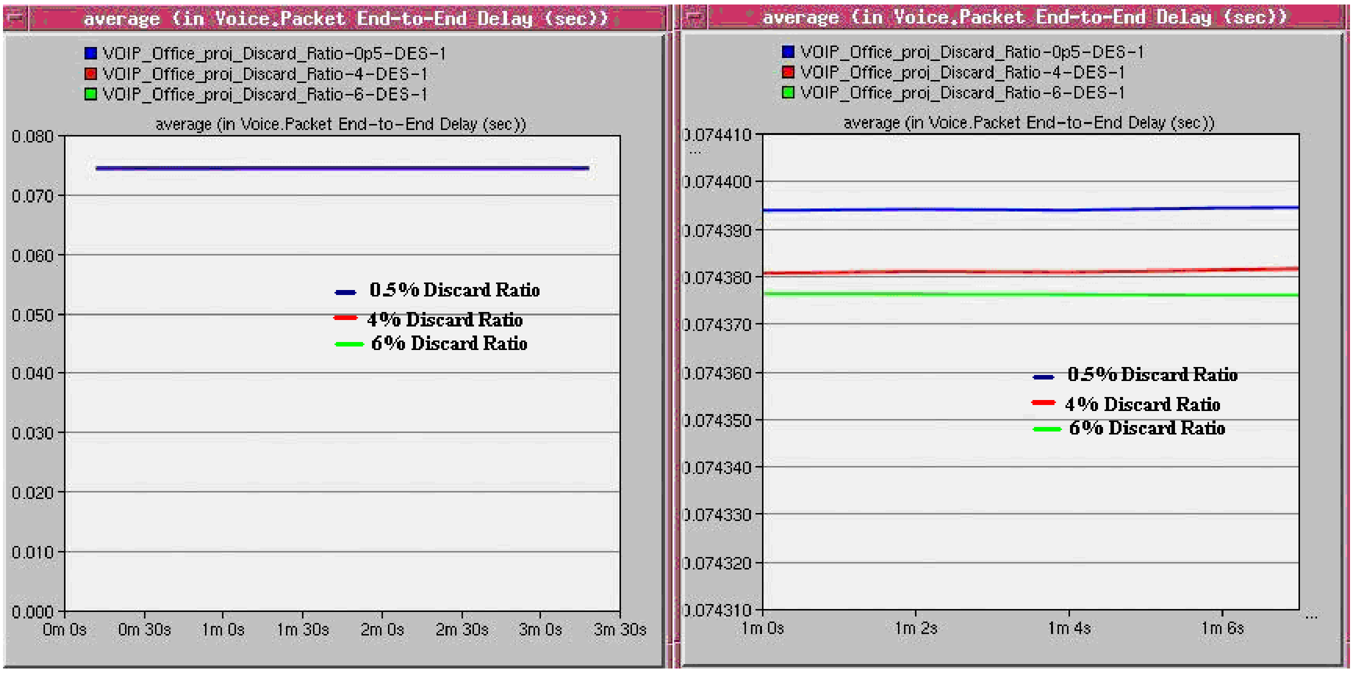


Figure 19: Discard Ratio Comparison--Voice Packet End-to-End Delay (sec). Left: Original; Right: Zoom in

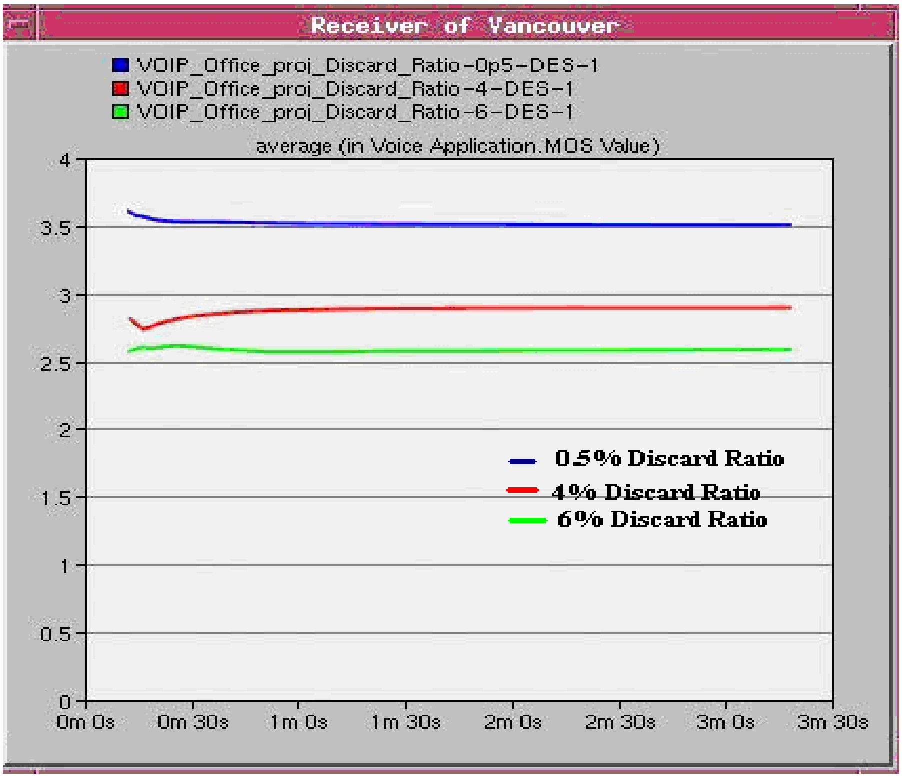


Figure 20: Discard Ratio Comparison--Voice Application MOS Value

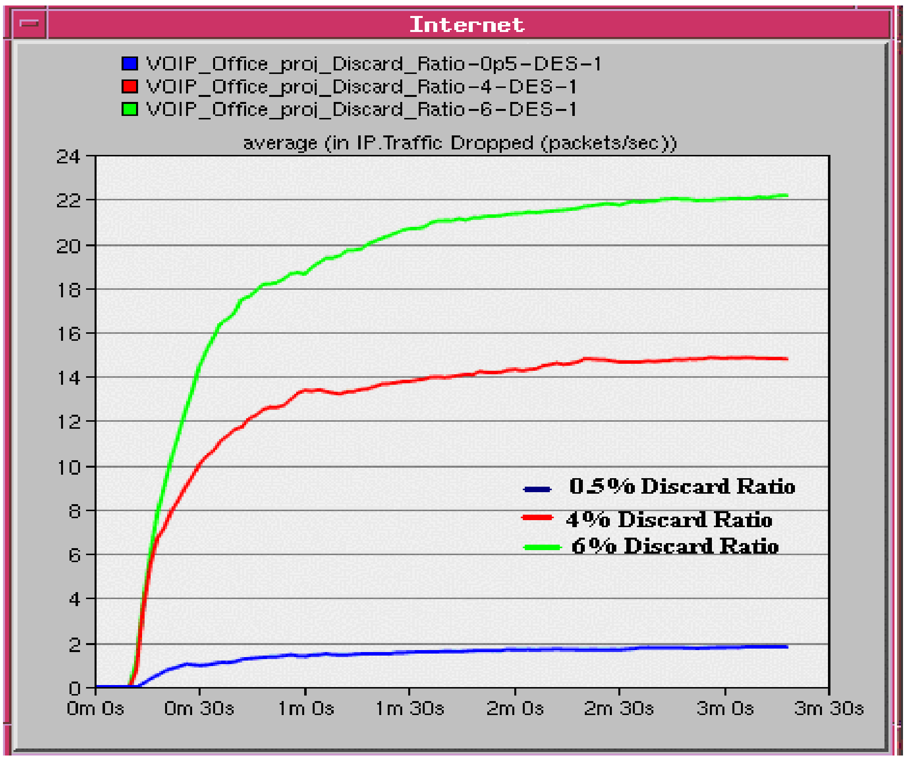


Figure 21: Discard Ratio Comparison--IP Traffic Dropped (packets per sec)

We create **Table 7** based on parameters captured from **Figure 18** to **21**. This table makes it easier to compare jitter, end-to-end delay, MOS value and packet loss under different discard ratio.

Table 7 : Different discard ratio used and the correspondent MOS values

Case	Discard Ratio	Jitter	End-to-End Delay	Voice Application MOS	Packet Loss (packets per sec)
Discard_Ratio-0.5	0.5%	Least Fluctuation	Longest	3.510	1.824
Discard_Ratio-4	4%	Medium Fluctuation	Shortest	2.896	18.411
Discard_Ratio-6	6%	Most Fluctuation	Shortest	2.584	22.024

For voice application jitter, network with 6 % discard ratio has the highest jitter fluctuation. In term of end-to-end delay, network with 6 % discard ratio has the shortest End-to-End Delay, compared with the other two discard ratio. It indicates that network with higher discard ratio has less end-to-end delay time. As the discard ratio increases, more packets are discarded during transmission causing faster link throughput. The increase in link throughput causes packets arriving at the receiver earlier than the expected time. Early packet arrival can also deteriorate the quality of VoIP as it makes the voice message incomprehensive.

From Table 7, it clearly shows that the higher the discard ratio in a network, the more packet loss occurs in that network. Furthermore, the higher the discard ratio in a network, the lower the MOS value in that network. It is reasonable as more voice packets are discarded in a network, the voice quality is greatly deteriorated and it explains the drop in the MOS value.

The Internet QoS affects the quality of VoIP since different Internet QoS tends to have different packet discard ratio. Packet discard ratio can alter jitter, end-to-end delay and packet loss which are all VoIP deterioration factors. In **Table 8**, we summary how changing packet discard ratio affect VoIP deterioration factors.

Table 8 : Different discard ratios used and the correspondent change of parameters

Packet Discard Ratio	Jitter Fluctuation	End-to-End Delay	MOS	Packet Loss
Increase	Increase	Decrease	Decrease	Increase
Decrease	Decrease	Increase	Increase	Decrease

Table 8 shows that the higher the discard ratio, the smaller the MOS value and therefore the worse the VoIP quality.

6.4 SCENARIO FOUR:

DIFFERENT ENCODER SCHEMES USAGE AND THEIR EFFECTS ON VOIP QUALITY

The purpose of this scenario is to verify if coding scheme would affect the quality of VoIP. The encoder schemes we used are Algebraic Code Excited Linear Prediction (ACELP) G 723 5.3k, Conjugate Structure Algebraic (CS-ACELP) G 729 A and PCM G 711. **Figure 22** shows the average of voice MOS value for different encoder schemes.

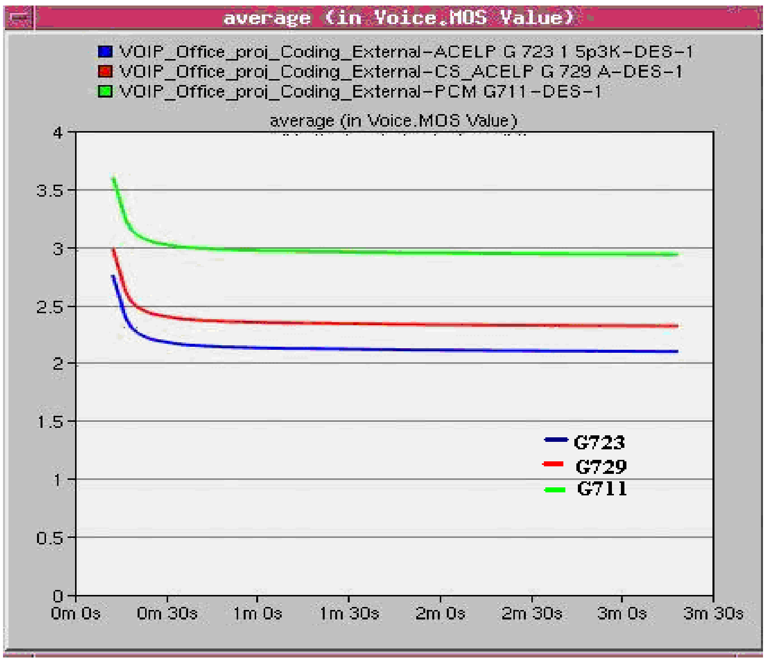


Figure 22: Encoder scheme comparison—average (in Voice.MOS Value)

The simulation result of those three encoder schemes shows that there is no difference in jitter, end-to-end delay and packet loss. Therefore, instead of creating a table to show four measured parameters as **Table 7**, we only show the MOS value in **Table 9**.

Table 9: Codec Used and the correspondent MOS values

Codec	MOS
ACELP G723 5.3k	2.097
CS-ACELP G 729 A	2.316
PCM G 711	2.935

Table 9 shows the Codec PCM 711 has the highest MOS value; whereas, Codec ACELP G 723 has the lowest MOS value. It means that Codec PCM G 711 has higher quality, compared with the other two codec. The bit rate for ACELP G 723, CS-ACELP G 729 and PCM G711 is 5.3 Kbps, 8Kbps and 64 Kbps respectively [12]. The delay for ACELP G 723, CS-ACELP G 729 and PCM G711 is 30 milliseconds, 10 milliseconds, and 0.25 milliseconds. Based on the bit rate and delay for those three codec, the rating of MOS from the simulation results makes sense because higher compression rate makes shorter delay which leads to higher voice quality.

We summary the delay and bit rate of the three codec in **Table 10**. We observe that the faster the bit rate and shorter the delay of a codec, the better the quality (MOS) of VoIP.

Table 10: Codec Used and the correspondent parameters

Codec	Bit Rate	Delay	MOS
ACELP G723 5.3k	Low	High	Low
CS-ACELP G 729 A	Medium	Medium	Medium
PCM G 711	High	Low	High

7 CONCLUSION

VoIP will continue to be widely used in the future since it has many advantages. In this project, we have successfully simulated a VoIP network and we have studied factors that deteriorate the quality of VoIP such as jitter, voice end-to-end delay, packet loss and Internet QoS. With our VoIP simulation network, we use it under four different scenarios to study how the VoIP deterioration factors change in each scenario. We found that the quality of VoIP depend on the distance between communication nodes. Therefore, the quality in long-distance VoIP communication is not as good as the quality in short-distance VoIP communication. Long-distance VoIP communication introduces longer end-to-end delay, more jitter fluctuation, more packet loss and low MOS value, compared with short-distance VoIP communication.

Considering the possible overload of the network capacity, we compared non-busy and busy VoIP networks. We found that the quality of VoIP deteriorates as the VoIP network is getting busy. When the VoIP network becomes busy, overload happens causing larger fluctuation in jitter, longer end-to-end delay, more packet loss and lower MOS value. The solution to fix these parameters is to change the link capacity. As a result, in order to improve the voice quality in a busy VoIP network, it is essential to use a high capacity link such as DS3, OC24 and OC48.

We also found that the Internet QoS affects the quality of VoIP. Poor Internet QoS introduce higher packer discard ratio; thus, more voice packets are dropped causing the voice message incomprehensive. High packet ratio also has effect on other VoIP deterioration factors such as jitter, and end-to-end delay. We also explore the effect of compression on VoIP quality by comparing the three speed codec (G 711, G723 and G729). The simulation results of those three codec match the compression theory.

8 REFERENCES

[1] Packetizer Inc., "How Does VoIP Work?" Available: http://www.packetizer.com/ipmc/papers/understanding_VoIP/how_VoIP_works.html, Jan. 2009 [Mar. 1, 2009]

[2] TSeyva Pte Ltd., "Advantage Disadvantage of VoIP." Available: http://support.tseyva.com/support/index.php?_m=knowledgebase&_a=viewarticle&kbarticleid=1, Aug.02, 2007 [Feb. 25, 2009]

[3] P. Curry, J. Hagedorn, J. Hermanowicz, and M. Sparks, Synchronized Voice Broadcast Over Congested IP Networks, Dec. 1 2007

[4] 2004 Conference Proceedings. Available: http://www.csun.edu/cod/conf/2004/proceedings/265.htm

[5] X. Chen, C. Wang, D. Xuan, Z. Li, Y. Min and W. Zhao, Survey on QoS Management of VoIP. Fed. 03 2003

[6] Jitter. Available: http://www.en.VoIPforo.com/QoS/QoS_Jitter.php

[7] S. Kemp, E. Eng and A. Hassanali, BlueS.E.A. Semester Research Project. Available: http://itom.fau.edu/jgoo/fa05/ISM4220/Blusea.pdf

[8] Noise and Voice Quality in VoIP Environments. Available: http://cp.literature.agilent.com/litweb/pdf/5988-9345EN.pdf

[9] Using OPNET Modules in a Computer Networks Class at Mercer University, ASEE Southeach Conference 2004, Donald U. Ekong, 2004

[10] The affects of different queuing disciplines over FTP, Video and VOIP performance, International Conference on Computer System and Technologies – CompSysTech'2004, Mitko Gospodinov, 2004

[11] Athina Markopoulou and Fouad Tobagi, Assessment of VoIP Quality over Internet Backbones, June 25 2002

[12] Voice Coding Algorithms. Available: http://www.nextgendc.com/?/seminar_voice_coding.htm

[13] K. Salah and A. Alkhoraidly, An OPNET-based Simulation Approach for Deploying VoIP, International Journal of Network Management Volume 16, Issue 3, Pages 159-183

[14] Monitoring and Troubleshooting VoIP Networks with a Network Analyzer, http://www.tamos.com/htmlhelp/voip-analysis/jitter.htm

[15] T. Schueneman, “The Advantages and Disadvantages of Using Voip”, 2009. Available: http://ezinearticles.com/?The-Advantages-and-Disadvantages-of-Using-VoIP&id=147921

9 APPENDIX I

This appendix presents the wireless workstation model (**Figure 23**) and the application processor model in the wireless workstation (**Figure 24**) we used in the subnets (both in Vancouver and New York company) of the network.

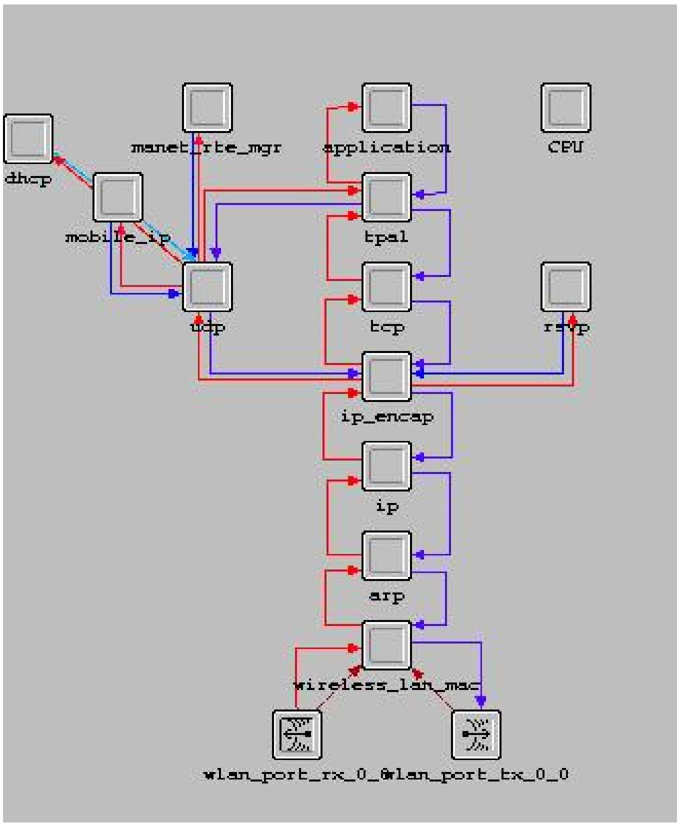


Figure 23: Wireless Workstation Model

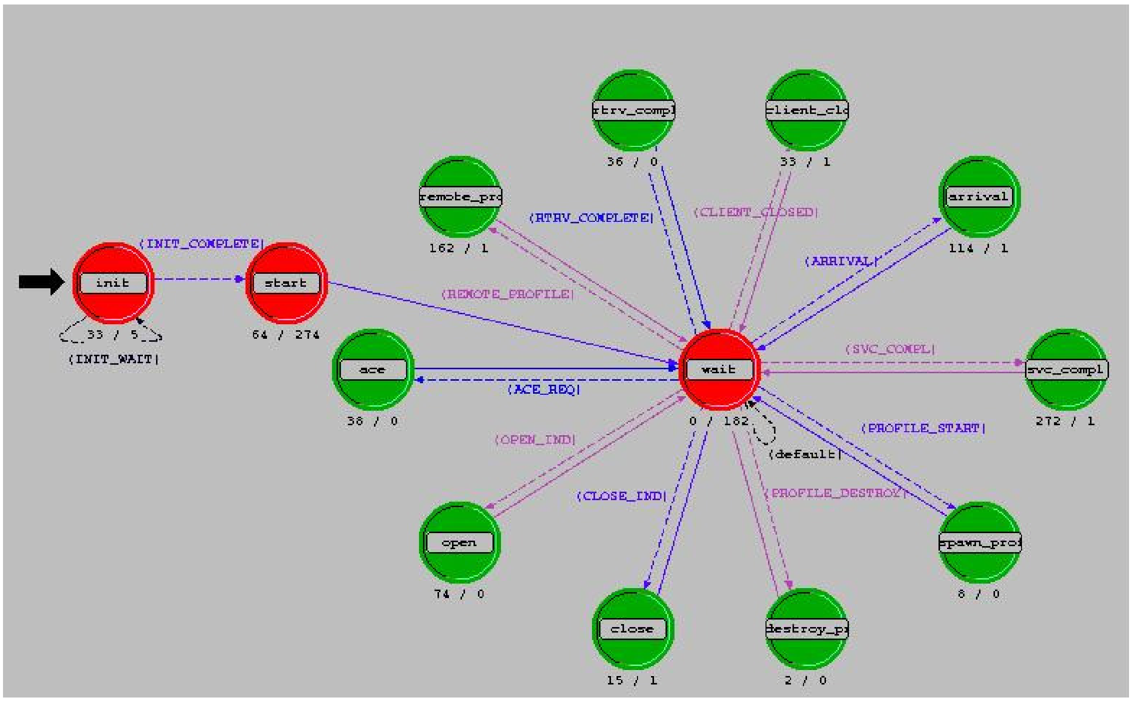


Figure 24: Application Processor Model