ENSC 427 Communication Networks

Analysis of Performance of VoIP

Over various scenarios

OPNET 14.0

Spring 2012

Final Report

Group 11

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List of Acronyms

Acronyms Extended Form

LAN Local Area Network

VoIP Voice over Internet Protocol

PSTN Public Switched Telephone Network

MOS Mean Opinion Score

ETE Delay End-to-End Delay

ITU International Telecommunication Union

Abstract

VoIP (Voice over Internet Protocol) is an advanced technology that has a great potential to develop new telecommunication with much lower cost and better QoS. In our project, we shall analyze the performance of VoIP over digital communication on popular application such as Skype and msn. Parameters of interest are the quality of service, the Mean Opinion Score, packet loss ratio and jitter. We also plan to examine the delays and distortion issues that VoIP might have while increasing the traffic load and generate a more realistic topology by adding extra models to the system and evaluate the impact to the overall QoS.

1. Introduction

1.1 Overview of VoIP technology and comparison between PSTN and VoIP

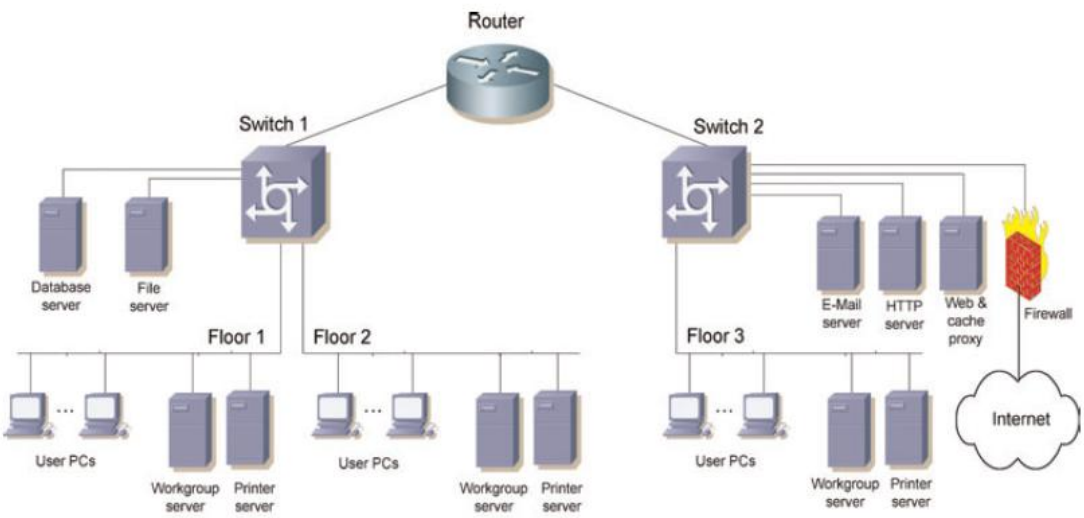


Figure 1.0 Typical VoIP network

As the advancement of technology progresses in a dramatic rate, a new age of digital communication has been established. A tremendous increase in popularity over the real-time voice communication over Internet protocol (IP) is observed in recent years. Voice over Internet Protocol (VoIP) is a high modern technology that provides the capability of users to generate telephone calls over an IP data network (Internet) using packet-switching technology instead of the traditional Public Switched Telephone Network (PSTN). With the creation of VoIP, the nowadays digital communication has greatly reduced in cost while preserving high quality service, as VoIP's capabilities to merges both data and voice in a single channel. [1]

PSTN Versus VoIP: A Feature Comparison

PSTN	*VoIP*
Dedicated Lines	All channels carried over one Internet connection
Each line is 64kbps (in each direction)	Compression can result in 10kbps (in each direction)
Features such as call waiting, Caller ID and so on are usually available at an extra cost	Features such as call waiting, Caller ID and so on are usually included free with service
Can be upgraded or expanded with new equipment and line provisioning	Upgrades usually requires only bandwidth and software upgrades
Long distance is usually per minute or bundled minute subscription	Long distance is often included in regular monthly price
Hardwired landline phones (those without an adapter) usually remain active during power outage	Lose power, lose phone service without power backup in place
When placing a 911 call it can be traced to your location	911 emergency calls cannot always be traced to a specific geographic location

Figure 1.1 Comparison of PSTN Vs VoIP

Not only limit to internet, VoIP has also shown its popularity in data networks such as Ethernet LANs. The reason is that Ethernet would be an ideal data networking platform for enterprises and other organizations to establish LAN communication [2], as it provides its user with a high level quality of services while greatly reducing cost comparing to the traditional PSTN . In addition to that, wireless Ethernet networks (IEEE 802.11) allow distance digital communication for users to connect to the network, which is ideal for locations that are difficult to setup tool such as Hospitals, two offices located far apart and conference rooms.

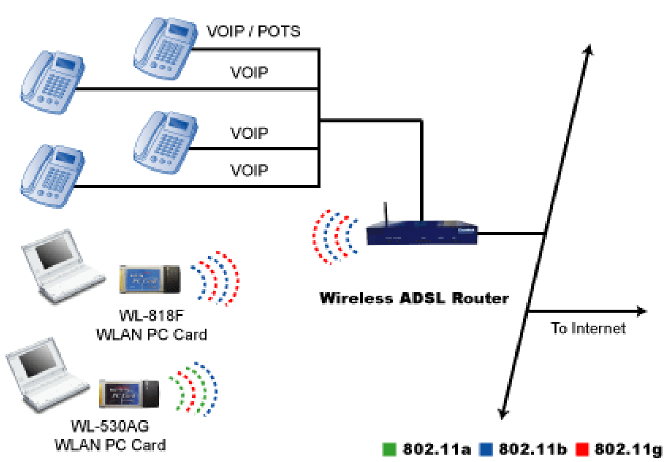


Figure 1.2 Ethernet of VoIP

Today, VoIP has becoming one of the most widely used technologies in a global aspect; as it shows an amazing grow of usage in homes and organization. Up to now, a variety of VoIP communication software has already in use on the market, such as Skype, AIM and Windows live messenger. [3].

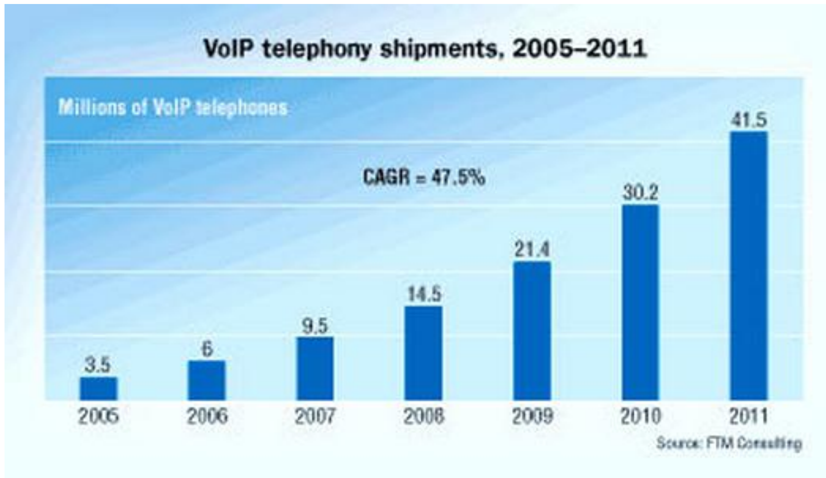


Figure 1.3 VoIP telephony shipments

1.2 Important Concepts

A .End to end Delay

End to end Delay is the total transit time for packets in a data stream to arrive at the endpoint and it is inevitable in communication system. Delay time is one of the most important factors in determining the quality of a call. Echoing has been a major problem that is caused by end to end delay. However, delay is able to be kept as small as possible by utilizing the project model/ topology. Typically, for optimum VoIP call quality, end to end delay must be less than 150ms.

To address this problem, we will simulate our model to see if VOIP can operate within the end-to-end delay of 200 ms threshold.

B. Jitter

Jitter is one the most common VoIP problems. Jitter is the undesired time delay from the packets sending end to receiving end in VoIP or other video communication network. The jitter can be affected by computer usage, the length and quality of the Ethernet cables and some other issues. The delay is inevitable and high levels of jitter leads to large numbers of packets to be discarded by the jitter buffer in the receiving IP phone or gateway. This will result in severe distortion in call quality or large increases in delay. Therefore in our case, we want to minimize the jitter as possible as we can.

C. Packet Loss

Failure or one of more packets to reach their destination across the network is recognized as packet loss. The occurrence of lost and dropped packets are extremely noticeable with real time streaming technology such as Skype and online gaming. On another hand, there is always a degree of packet loss allowance in almost every network. There are possible causes that lead to packet loss, such as channel congestion, corrupted packets rejected in-transit and poor networking hardware. To properly recover the loss packets, reliable network transport protocol such as TCP are used insure an acceptable and stable transmission. Using the Acknowledge technology, the network can reassure that the packets have been successfully delivered. In our simulation, we would try to maintain an maximum of 10% packet loss threshold.

1.3 Objectives of study

The purpose of the project is to conduct several of test cases in VoIP by constructing different simulation scenarios under OPNET 14 Software. The reason behind it is that the successful implement of the project would reflect the advantage of VoIP over the traditional PSTN and thus proving that VoIP would be ideal candidate for the modern technology in network communication.

In the perspective of a good design of VoIP, the most important factor would be the quality of service (QoS) provided to all users on the network, while considering medium-to-high traffic loads that is most likely to occur in reality. Due to the fact that initially IP networks were designed to handle data traffic and not voice, there was no issue regarding to real-time communication. However, factors that might influence the overall performance of VoIP are bandwidth, end-to-end delay, jitter, packet lost rate and utilization. [4]

Base on the standard provided by the International Telecommunications Union, the acceptable Mean Opinion Score (MOS) for VoIP is in the range of 4 to 4.5. As indicated in the following figure, the latency (ETE delay) shall not be above 150ms in one way service level. As in terms of jitter, which is the measure of the inconsistency of delay of packet delivery, is approximately 20 ms in range of tolerance under the recommendation of ITU[5][6].

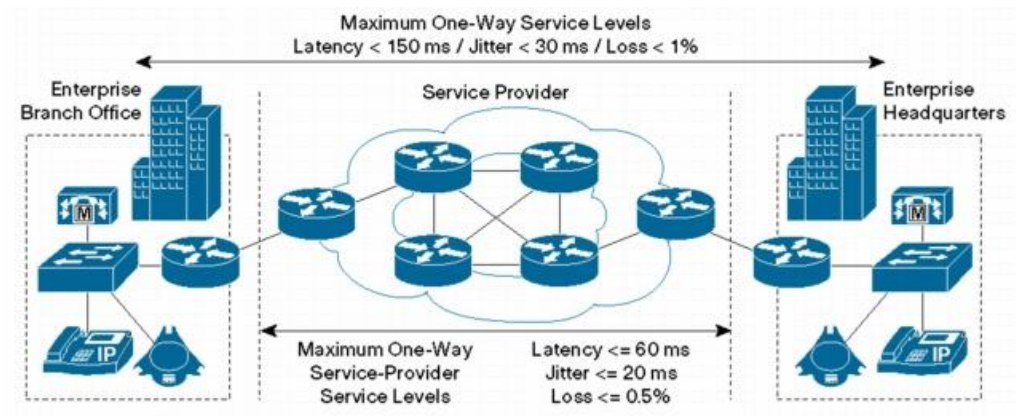


Figure 1.4 Tolerance range of VoIP

Statistics parameters to be collected and evaluated are ETE delay, delay variation, packet sent and drop rate, and jitter under WLAN802.11g and Ethernet networks under the scenarios shown in Figure 1.5. Based on the result, reflections upon the data would be made to ensure the parameters are within the standard of ITU.

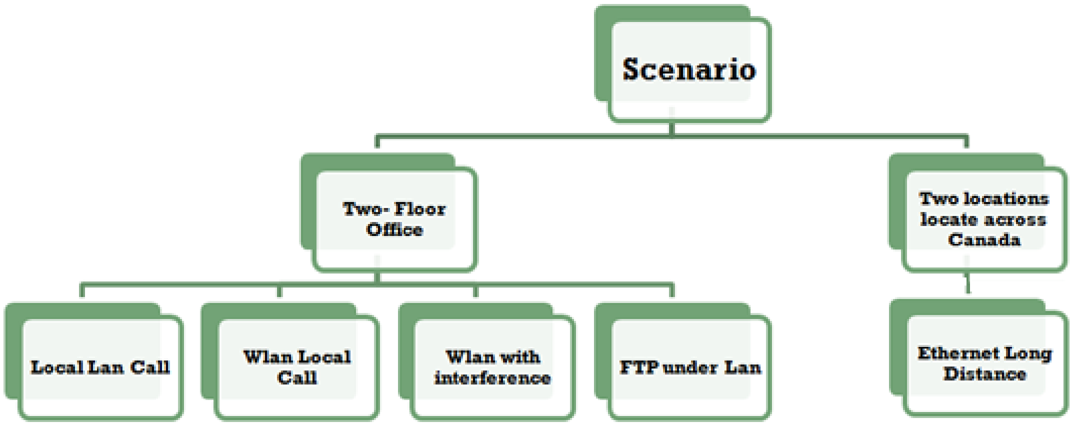


Figure 1.5 OPENT simulation scenarios break down

2. Description of overall design

2.1 Overall LAN and WLAN model

The overall network model is shown below—a model implemented in OPNET 14.

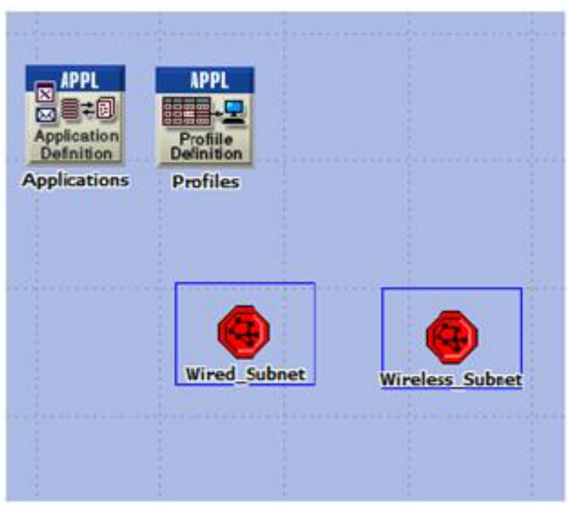


Figure 2.0 OPNET implementation of VoIP

The Project divided into two major scenarios parts: Local two-floor office model and two-locations-far model.

2.2 Configurations and Designs

The project consist total of six independent scenarios as shown in Figure 1.5. Using these models as backbone, the performance of VoIP shall be compared under the follow circumstances:

1. Local call versus Long Distance Calls

2. WLAN 802.11g versus Ethernet LAN Network

3. Increasing Traffic Load

4. LAN with FTP access during Voice transmission

5. 2.4GHz Interference to WLAN Network

The Application and Profile configuration are listed as follow:

G.711 Coding Scheme, 5 Voice Frame per Packets, interactive Voice [7]

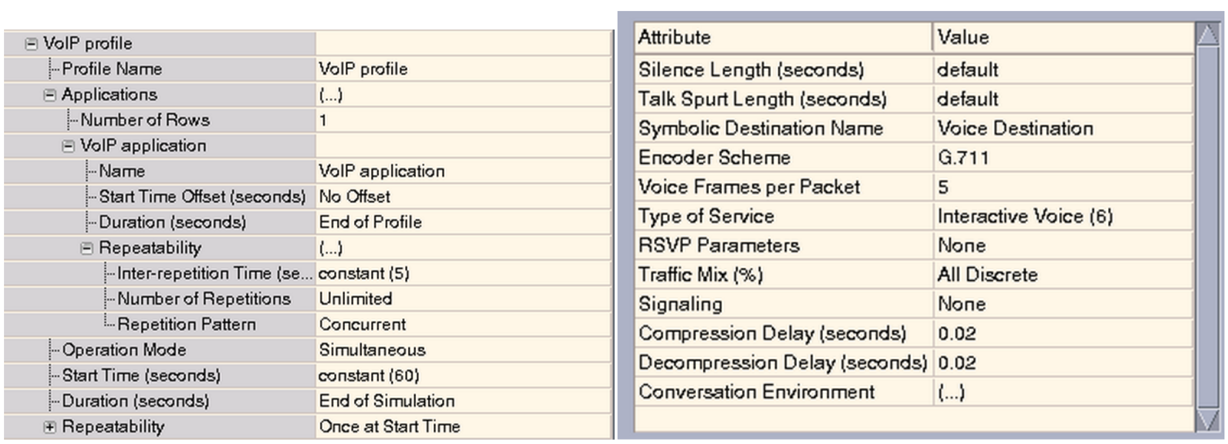


Figure 2.1 App and Profile configuration

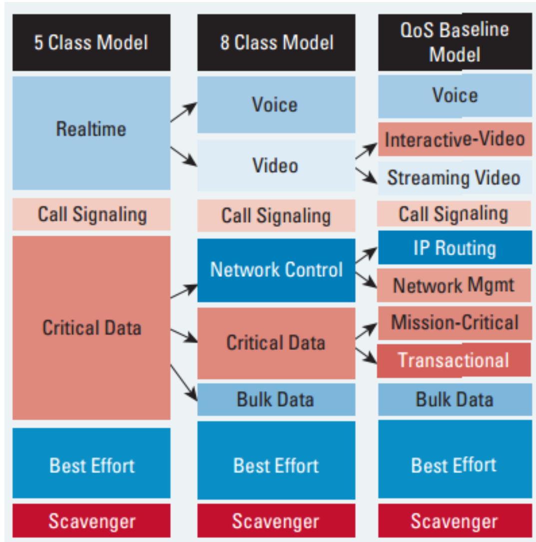


Figure 2.2 Difference of service

2.3 Topology of company with two floors

As indicated in the scenarios break down chart, this section provides the detail of the OPNET simulation with specific parameters and topology. First is the network with the size of 100 X 100 m in dimension, or categorized as office in OPNET. The office is divided into two floors with multiple work stations in each floor. The stations are located 15m apart from each other and are connected through 10 Base T line to the Ethernet switch. The overall network is hooked up to a CISCO 400 which acts has router. The topology is setup in such way to achieve an LAN environment for the small office to establish VoIP communications.

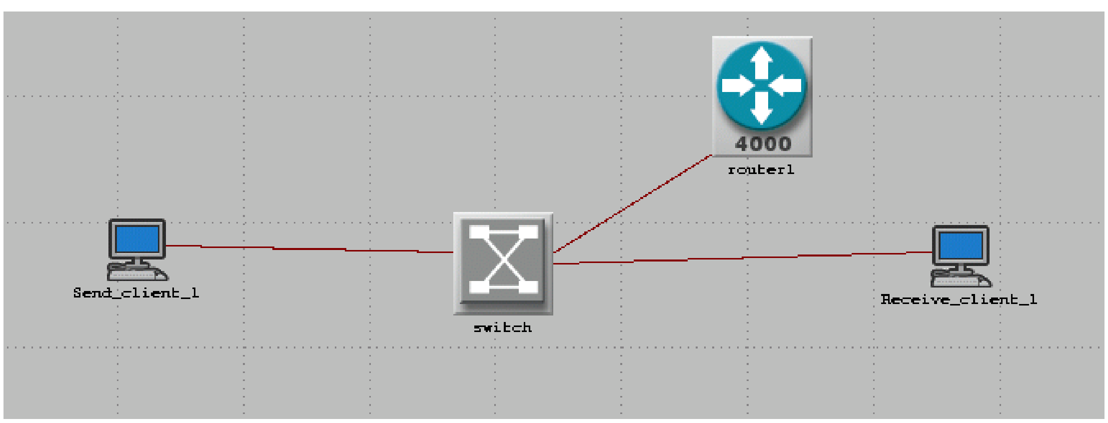


Figure 2.3 Basic LAN VoIP network

After implementing the backbone structure of the two-floor office LAN model, we increase the amount of Ethernet work station and compare the overall performance to the initial model.

One special note, although there are many Ethernet stations presents in each floor, only one work station (send client) is making a one to one VoIP call to another work station (Receive client) located in the other floor, no conference has been establish.

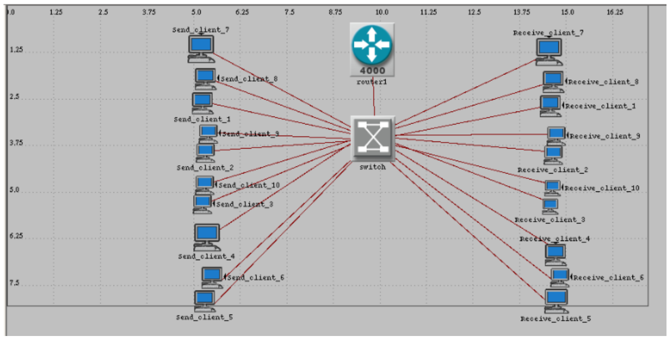


Figure 2.4 LAN 20 VoIP Network

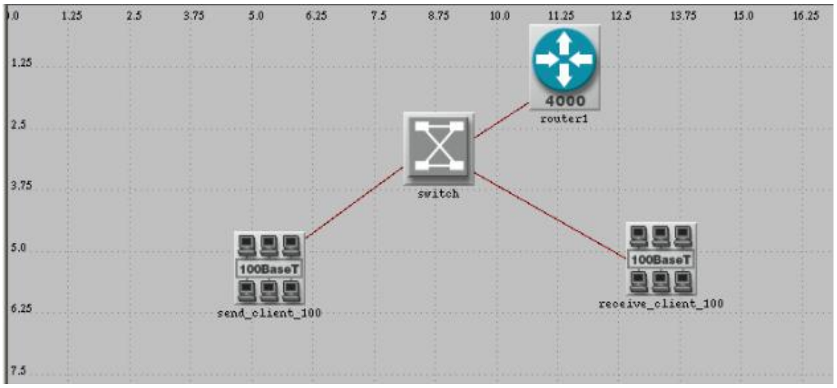


Figure 2.5 LAN 100 VoIP Network

2.4 Topology of wireless network

The topology for wireless (WLAN) VoIP connection under the standard of IEEE 802.11g between the two floor office is shown in Figure 2.1.2. The router is located in the center of the two wireless work stations. The data rate is setup to be 54Mbps. The detail configuration is listed as below:

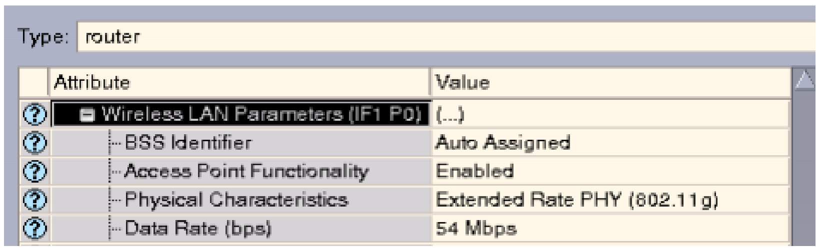


Figure 2.6 Configuration of Router

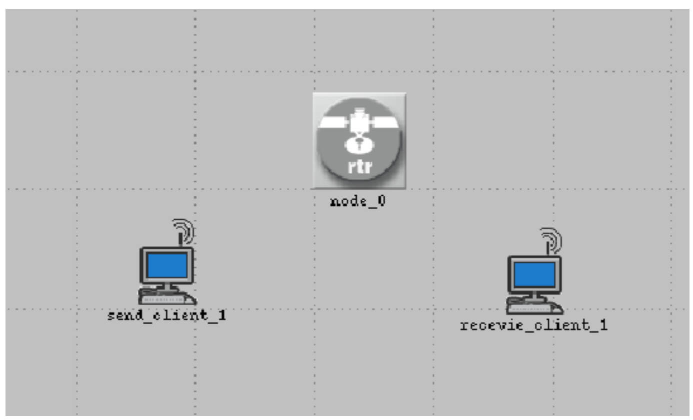


Figure 2.7 Basic WLAN VoIP Network

Applying the same method, we added more wireless work stations around the wires router and compare the result. However, there is no conference call in the WLAN network.

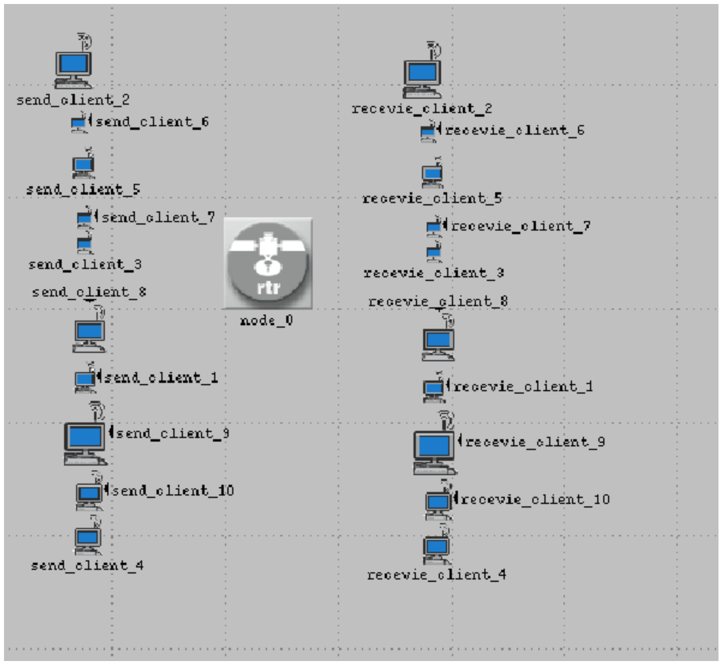


Figure 2.8 Multiple WLAN Network

2.5 Topology of two companies located far apart

The network which two locations (offices) are located relative far apart from each other (Vancouver to Waterloo) provide an realistic simulation on long distance VoIP communication under Ethernet connect. The two locations' routers are interconnected with PPP DS0. One interest thing to note, WiMAX along with WiFi would also be an ideal candidate to establish long distance VoIP calls. The topology inside the two subnet applies the same network structure as the initial two-floor-office topology.

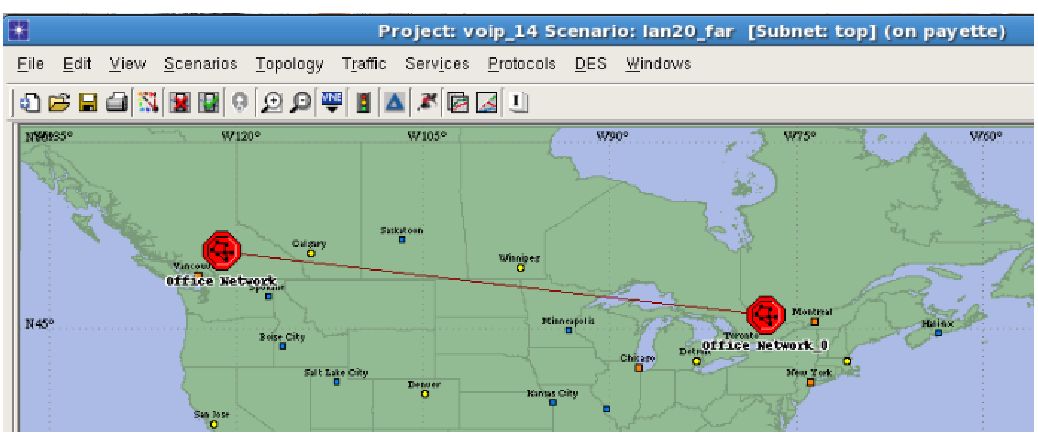


Figure 2.9 LAN VoIP for far apart

2.6 Topology of LAN with FTP

Under the FTP scenario, a **File Transfer Protocol** (**FTP**) model is added to the Ethernet switch. By adding the FTP, we would expect the traffic load to increase and reduce in bandwidth usage since the work stations are making a VoIP call and accessing the FTP simultaneously. We are interested in whether the FTP is TCP-friendliness with VoIP or not [8].

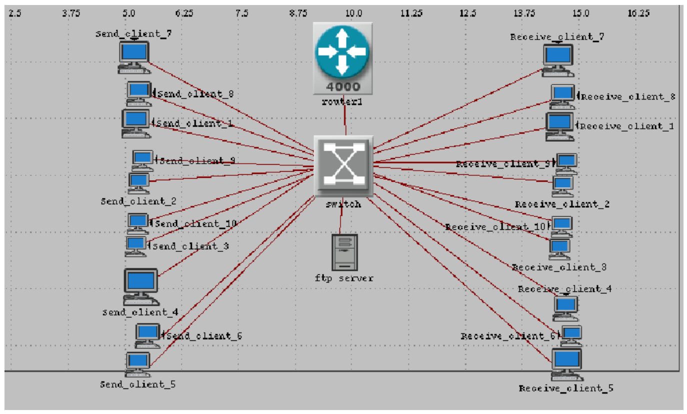
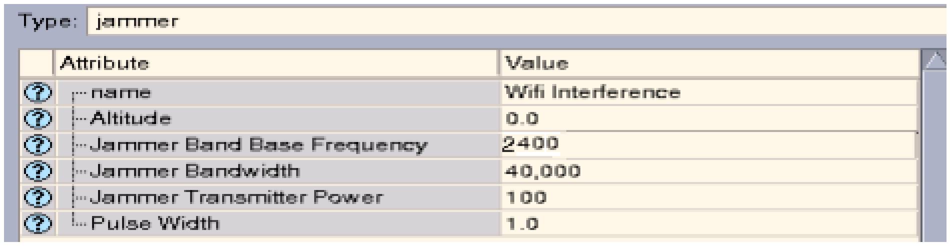


Figure 2.10 FTP in VoIP

2.7 Topology of wireless network with interference

Since the WLAN 802.11g operates in the frequency range of 2.4GHz, and it is known that many other electronic devices such microwave, Bluetooth and video devices are also potential user of the 2.4GHz band user, interference is inevitable. Under this section, a 2.4 GHz jammer next is added to the model to simulate this situation.



Type: jammer

Attribute	Value
name	Wifi Interference
Altitude	0.0
Jammer Band Base Frequency	2400
Jammer Bandwidth	40,000
Jammer Transmitter Power	100
Pulse Width	1.0

Figure 2.11 Configuration of Jammer

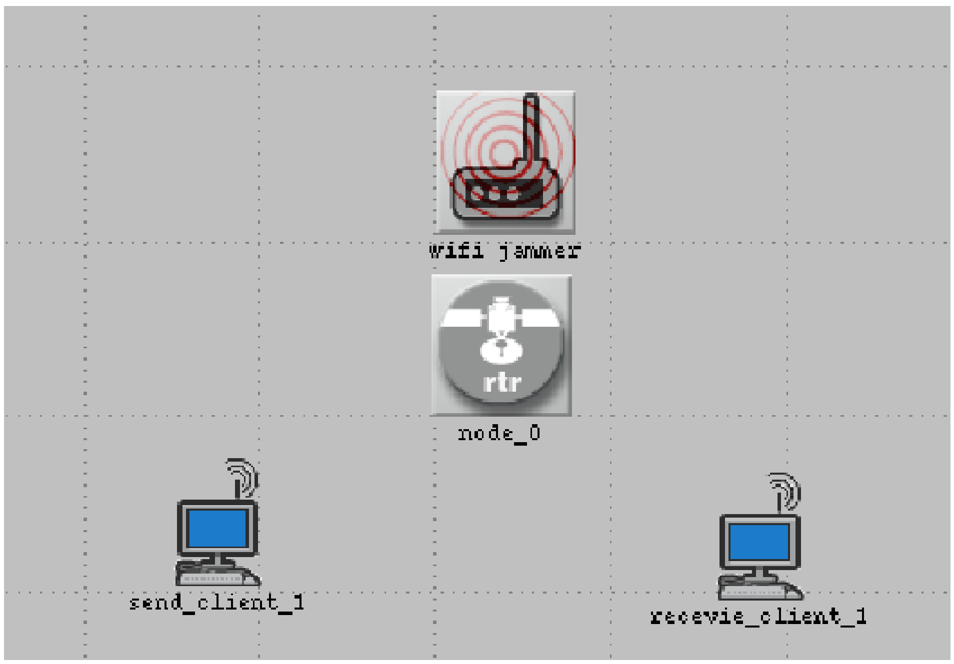
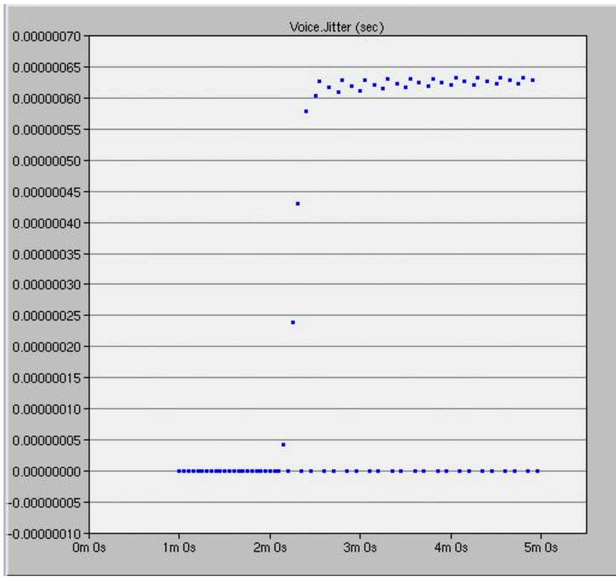


Figure 2.12 Basic WLAN with Jammer

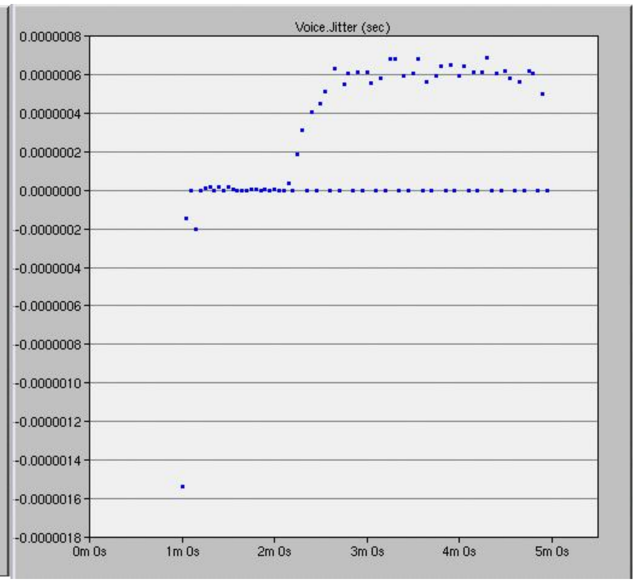
3.0 Simulation Results and Comparisons

3.1 Scenario 1: VoIP Call in LAN

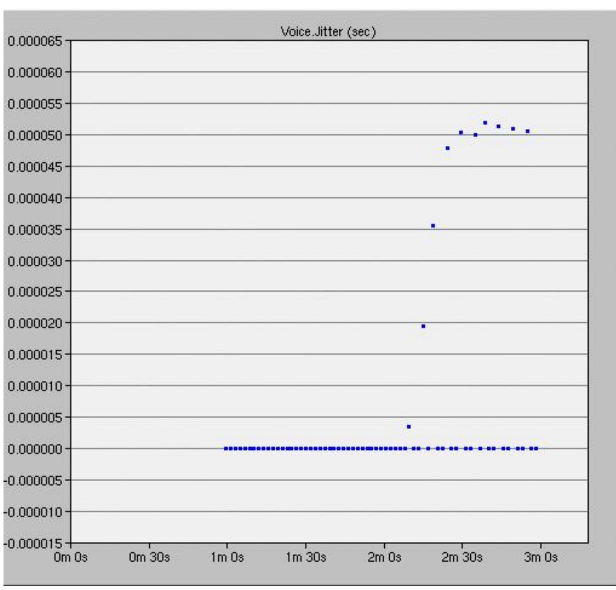
The first scenario tests the performance of VoIP call in LAN of a small office. The number of client is initially set to 2 nodes and gradually increases to 100 nodes.



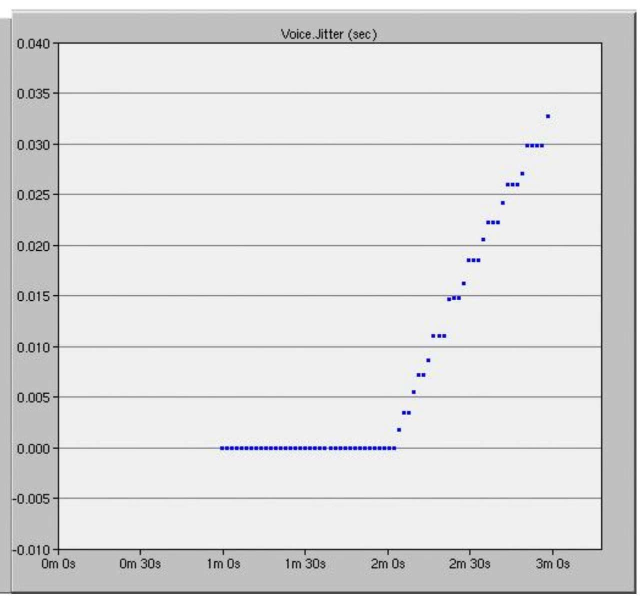
a. 2 nodes



b. 20 nodes



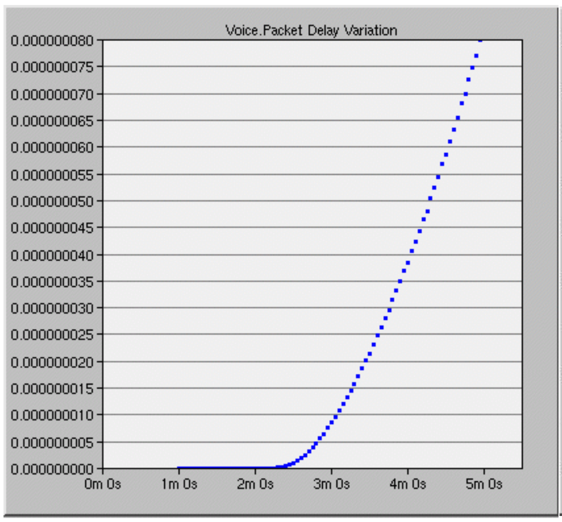
c. 50 nodes



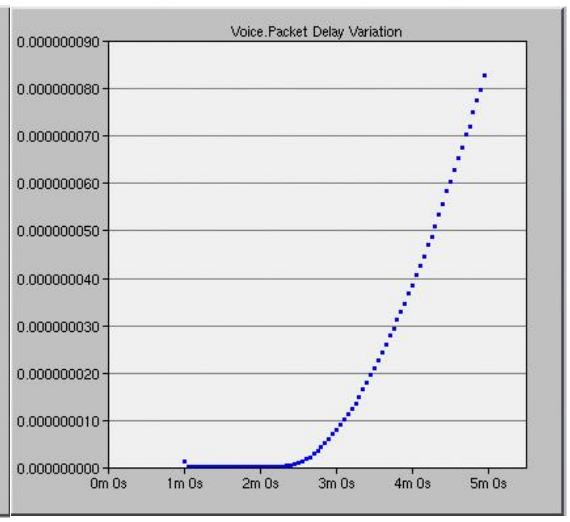
d. 100 nodes

Figure 3.1 jitter in the voice application

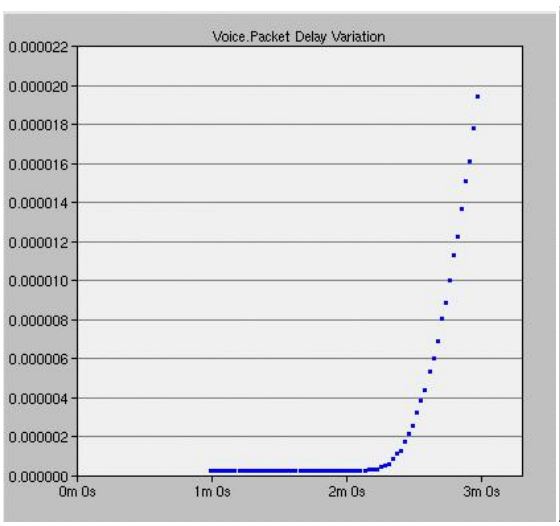
As shown in figure 3.1, the voice jitter increases as the number of nodes increase. When the number of node is between 2 to 20 nodes, the jitter is very small and unnoticeable. When the number of nodes in the LAN increased to 50, the voice jitter is 100 times larger than a LAN with 20 nodes. When the number of nodes in the LAN increased to 100, the voice jitter is 1000 times larger than a LAN with 50 nodes. Also the voice jitter in a LAN with 100 nodes seems to increase at a constant rate at the end of the simulation.



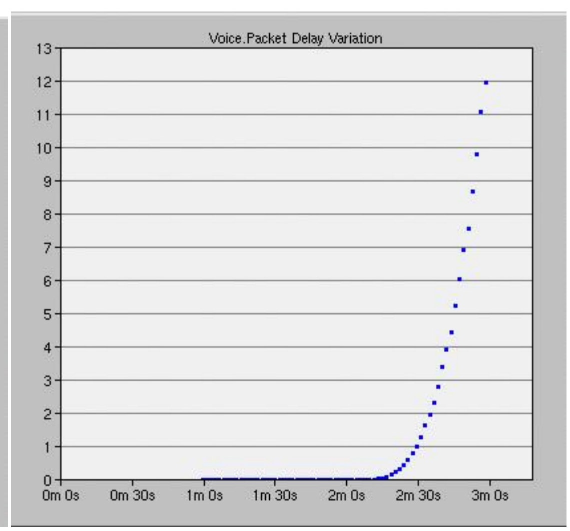
a. 2 nodes



b. 20 nodes



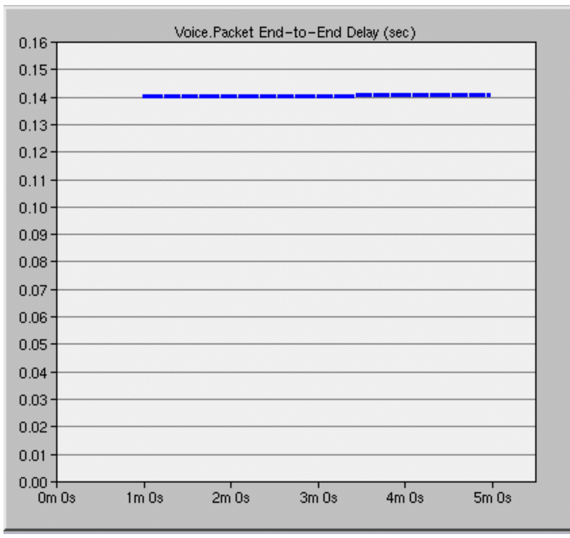
c. 50 nodes



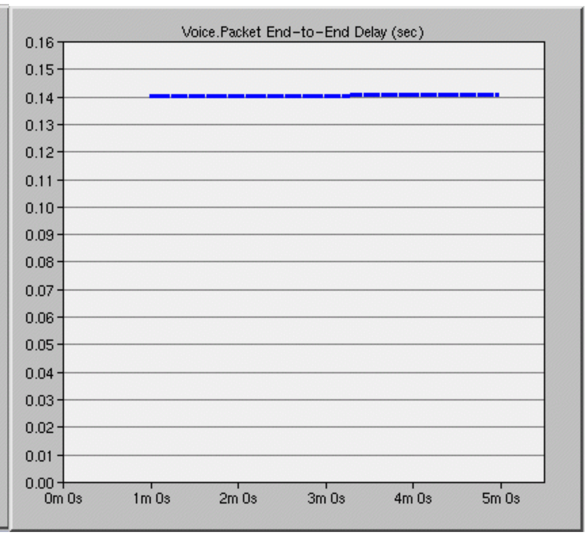
d. 100 nodes

Figure 3.2 Packet Delay Variations

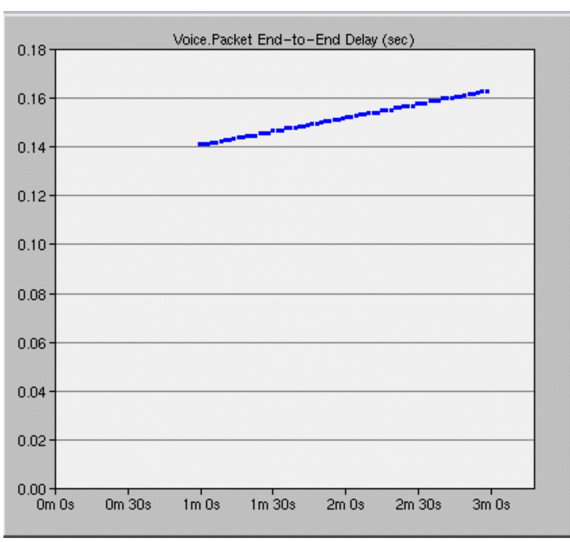
Figure 3.2 show the packet delay variation in a LAN with different number of client nodes. The packet delay variation results are very similar to voice jitter results. When the number of nodes is small, we have a very small and unnoticeable delay variation. The magnitude increases as the number of nodes increase. In the simulation with number of nodes is 100, the delay variation exceed 1 second and reach a maximum of 12 seconds. The delay variation seems to increase exponentially with the number of calls.



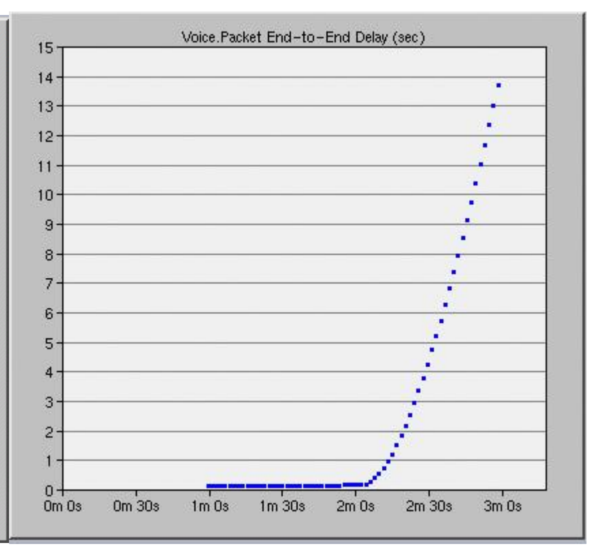
a.2 nodes



b. 20 nodes



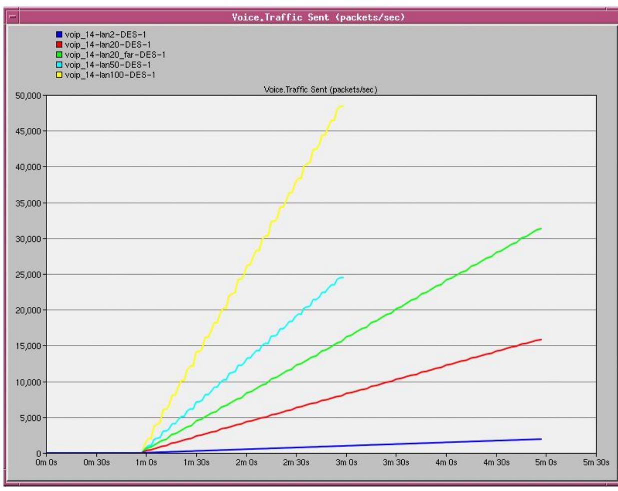
c.50 nodes



d. 100 nodes

Figure 3.3 Packet End to End Delays

Figure 3.3 shows the corresponding packet end to end delay. When the number of nodes is small, between 2 nodes to 20 nodes, we have the identical ETE delay around 140ms. ETE delay increases as the number of client nodes and calls are added into the network. When the nodes increased to 50, ETE delay increased to around 160ms. In the simulation with 100 nodes, approximately 60 seconds after VoIP calls started, ETE delay rapidly increased over 200ms and reach 12 seconds.



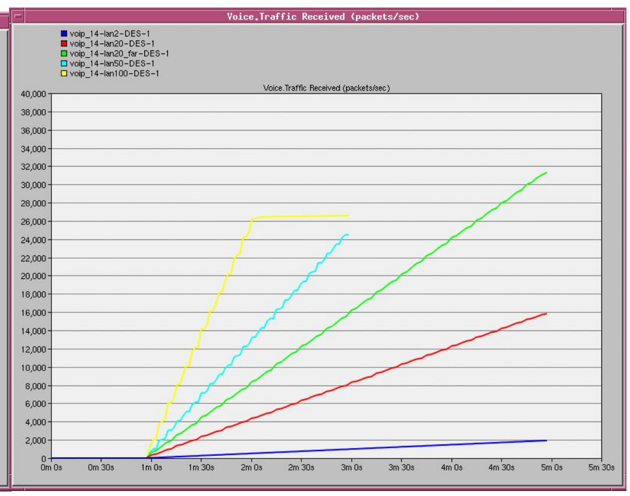


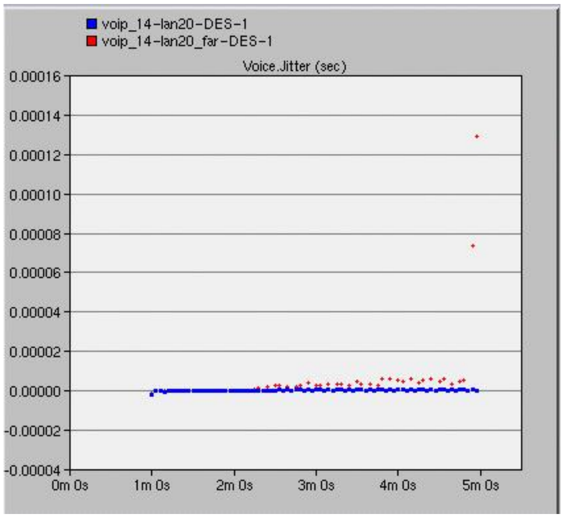
Figure 3.4 Traffic Sent **Figure 3.5 Traffic Received**

Figure 3.4 and 3.5 show the traffic sent and received with the voice application. For the simulation with less than 50 nodes, the traffic received is consistent with the traffic sent. For the 100 nodes simulation, packets are lost when the numbers of call get too high.

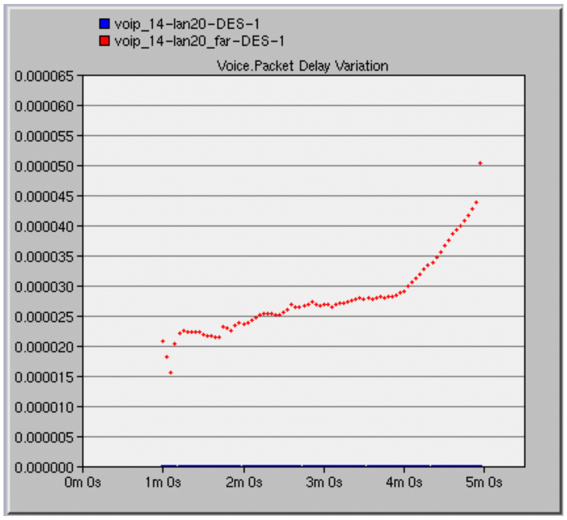
As seem from the results, VoIP calls have stable connection when the number of client is small. However, as the number of VoIP clients and calls increase, the voice jitter, delay variation, and ETE delay become significant factors to the calls quality.

Scenario 2: Long Distance VoIP Calls under LAN

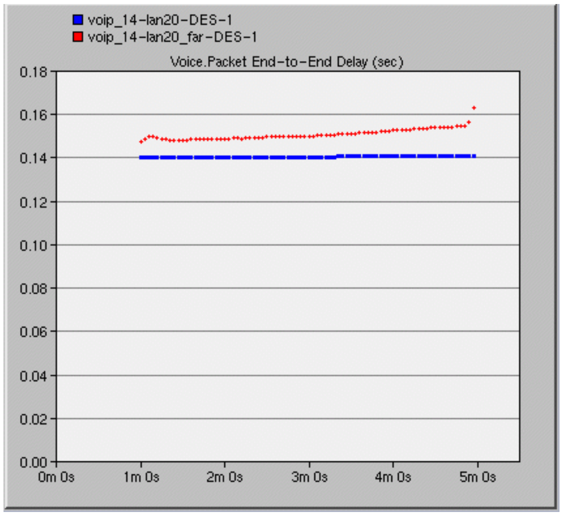
In this scenario, calls are made between two offices with LAN placed across the country. Effect on distance to calls quality will be evaluated.



a. Voice Jitter



b. Packet Delay Variation



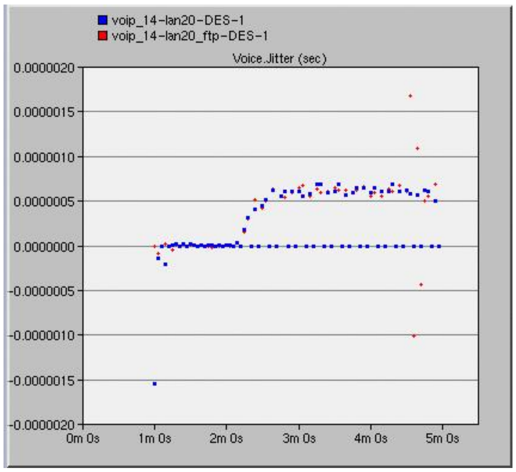
a. Packet End to End Delay

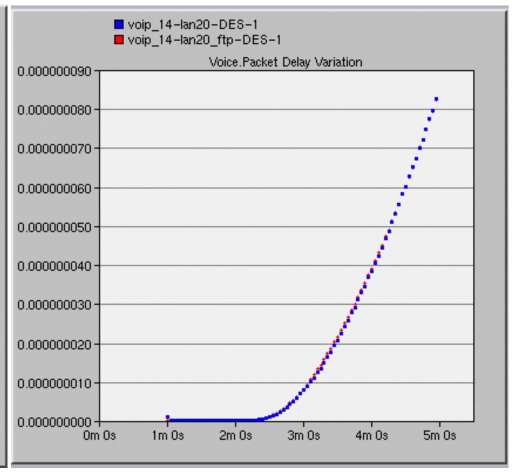
Figure 3.5 Long Distance Call under LAN

Jitter, delay variation and ETE delay increased as a result of the increased distance. However the value is still very small when compared to the simulation with large number of nodes. The ETE delay is relatively stable around 150ms throughout the simulation. Calls quality is within the acceptable range.

Scenario 3: VoIP calls in LAN with ftp server

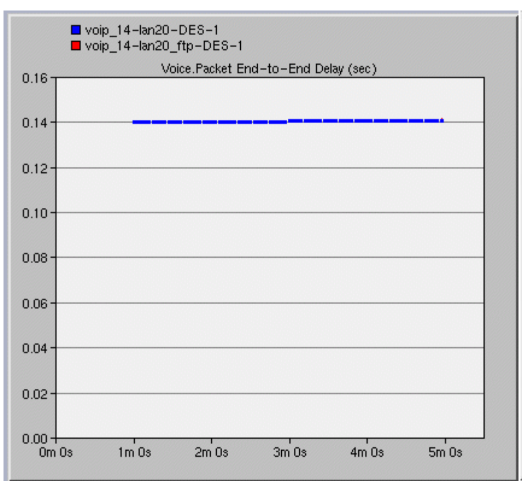
By adding an ftp server and ftp traffic into the network, we would like to produce a more realistic simulation result.

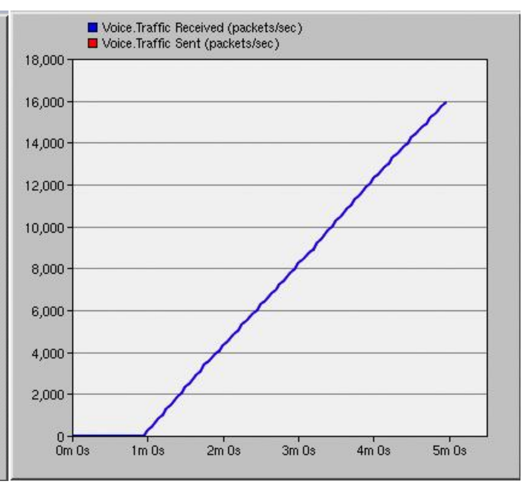




a. Voice Jitter

b. Packet Delay Variation





c. Packet End to End Delay

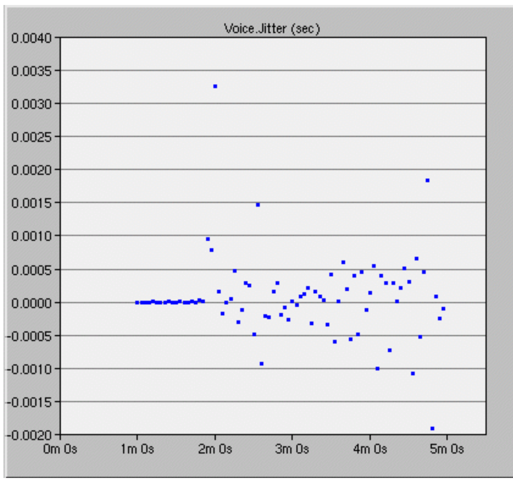
d. Traffic sent and received

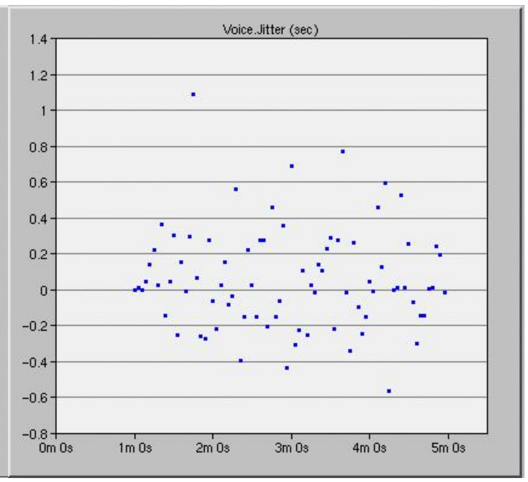
Figure 3.6 VoIP calls in LAN with ftp server

In this case, adding ftp traffic in the network does not affect the voice jitter, delay variation or end to end delay. Traffic sent is identical to traffic received. Since this FTP simulation is done on a LAN with only 20 nodes. There is not much VoIP connection and calls. We assume that there is enough bandwidth to meet the demands of both the VoIP connection and FTP traffic. Therefore, the FTP packets are not dropped at all.

Scenario 4: VoIP Calls in WLAN

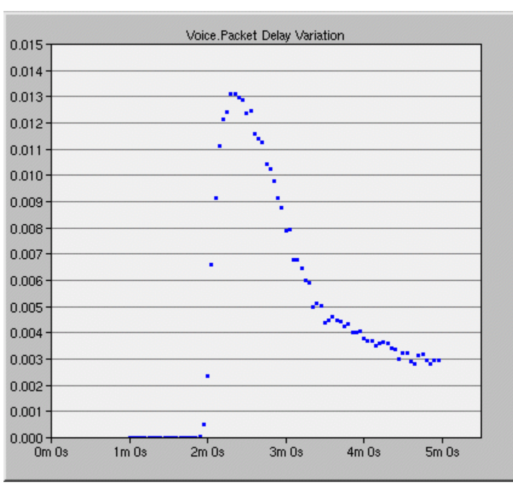
In this scenario, we will simulate a small office using WLAN for their VoIP application and compare the results with LAN.

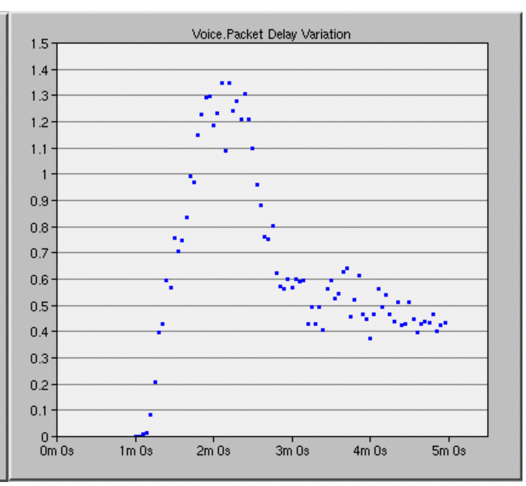




a.2 nodes Jitter

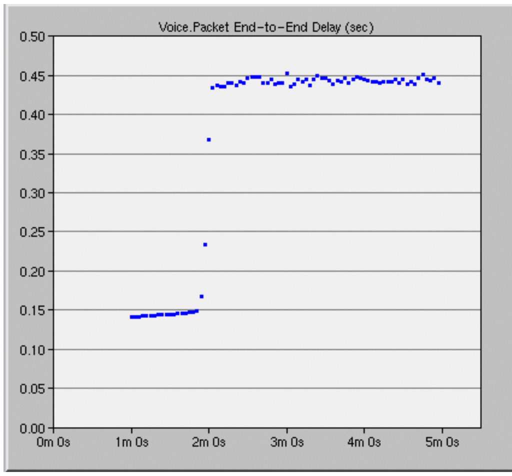
b. 20 nodes Jitter



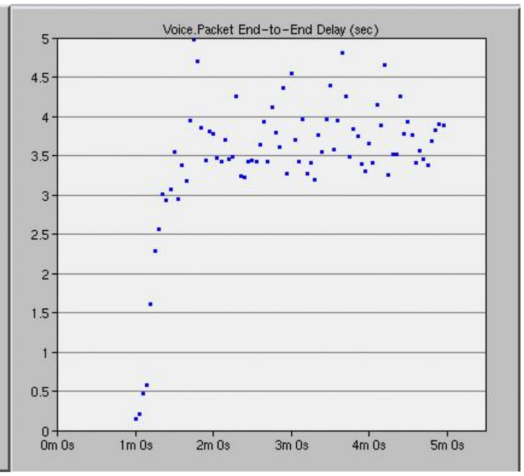


c. 2 nodes Packet Delay Variation

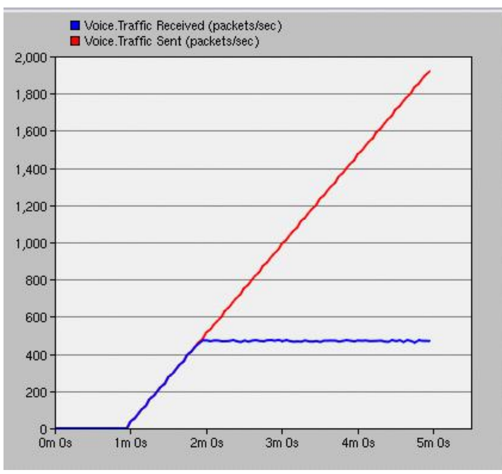
d. 20 nodes Packet Delay Variation



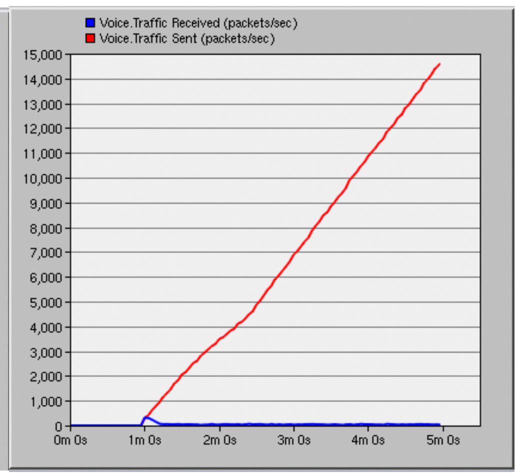
e. 2 nodes Packet ETE delay



f. 20 nodes Packet ETE Delay



g. 2 node Traffic sent and received



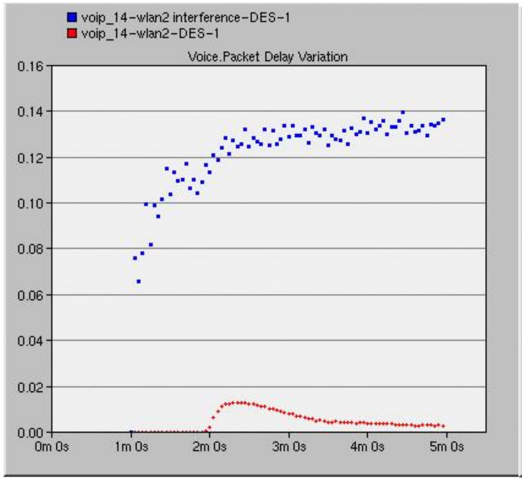
h.20 nodes Traffic sent and received

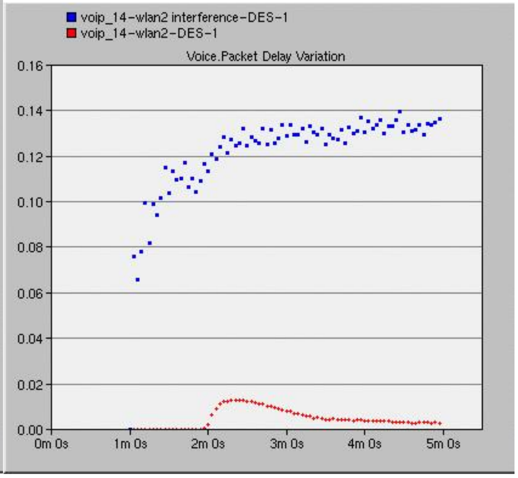
Figure 3.7 VoIP Calls in WLAN

The performance of VoIP in WLAN is quite poor compare to the wired network. The values of jitter, delay variation, and end to end delay in wireless network is multiple times larger the values of wired network. Even with only 2 nodes in the network, the ETE delay starting with 150ms will rapidly increase to approximately 450ms when the network is congested. In addition, when comparing the traffic sent and received in the simulation, we notice that, due to limited bandwidth, most of the packets sent were lost. Since we are simulating using G.711 compression codes which consume the most bandwidth, another compression code with less bandwidth consumption could be used to improve the performance of VoIP.

Scenario 5: VoIP in WLAN with interference

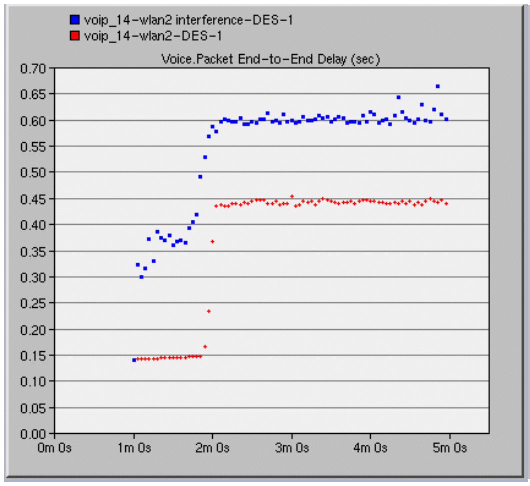
Interference exists in a network and is an inevitable factor in determining the quality of a VoIP calls. We represent the interference by adding a jammer into the network and examine the influence of interference in a network.

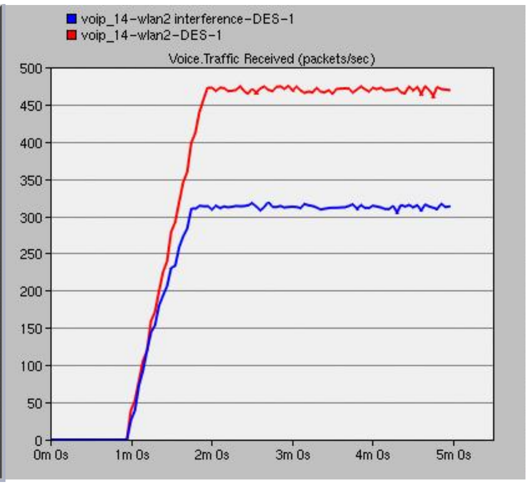




a. Voice Jitter

b. Packet Delay Variation





c. End to End Delay

d. Traffic Received

Figure 3.8 WLAN with interference

The results are consistent with our expectation. In the wireless network with interference, we have a larger jitter, delay variation and packet end to end delay. End to end delay at of the start of simulation is around 350ms which has already exceed the acceptable range. Packet received is approximately 2/3 of the ideal network which means more packets are dropped due to the interference.

4. Conclusion

In this project, we covered five scenarios: VoIP Call in LAN, Long Distance VoIP Calls under LAN**,** VoIP calls in LAN with ftp server, VoIP Calls in WLAN, and VoIP in WLAN with interference. Through the results we got found out that Ethernet has a more stable and less delay connection than wireless connection. Interference near wireless router greatly reduces QoS. Moreover, long distance VoIP introduces greater jitter, ETE and lower MOS. As a result, although VOIP has some negative sides, we consider that VOIP is a great alternative way to replace the traditional circuit switch phone network in the future.

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