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The Affects of Busy Voip Network and Non Busy Voip Network on Qos Voip Application Using Opnet

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ABSTRACT

VoIP is an advanced technology that has a great potential to develop new telecommunication with much lower cost and better QoS. The objective of this research is to study the effect of busy VoIP network and a non busy VoIP network on the VoIP Qos. In this work, simulation tool OPNET Modeler version 14.5 is used to analyze VoIP QoS based on measuring the major factors that influence the QoS for VoIP according to international telecommunication union (ITU) standards including: delay, jitter and packet loss. In this research, a comparison was executed and it was found that when the VoIP network becomes busy; overload happens and causes increase in the end to end delay and packet loss. 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.

KEYWORDS : OPNET, VoIP, DS1 link, DS3 link, QoS

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. We can make and receive phone calls wherever there is a broadband connection and it is as convenient as e-mail. Furthermore, there are many other features that make VoIP attractive [1]. Call forwarding, call waiting, voicemail, and three-way calling are some o f the services that are usually provided at no extra charge. We can also send data such as pictures and documents at the same time we are talking on the phone. 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 [2]. QoS has become a critical issue, because some applications as FTP, HTTP and e-mail are not sensitive to delay of transmitted information, while other applications such as voice and video are more sensitive to loss, delay and jitter of the information. Therefore, QoS of VoIP is an import concern to ensure that voice packets are not delayed or lost during the transmission over the network [3].

2. Quality of VoIP

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. In this project, we study all the potential parameters that can deteriorate the quality of VoIP by compare a busy VoIP network with a non-busy VoIP network. A non busy VoIP network was created in which there is only one long-distance conversation pair with DS1 link that used to connect the subnets to the IP cloud. In order to create a busy VoIP network, all workstations in the first location will communicate with all workstations in the second location. 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.

2.1 DS1 (Digital Signal 1) is a T-carrier signaling scheme devised by Bell Labs. DS1 is a widely used standard in telecommunications in the United States and Japan to transmit voice and data between devices. DS1 is the logical bit pattern used over a physical T1 line; however, the terms "DS1" and "T1" are often used interchangeably [4].

DS1 telecommunication circuit is made up of twenty-four 8-bit channels (also known as timeslots or DS0s), each channel being a 64 kbit/s DS0 multiplexed carrier circuit [5]. A DS1 is also a full-duplex circuit, which means the circuit transmits and receives 1.544 Mbit/s concurrently. A total of 1.536 Mbit/s of bandwidth is achieved by sampling each of the twenty-four 8-bit DS0s 8000 times per second. This sampling is referred to as 8-kHz sampling. An additional 8 kbit/s of overhead is obtained from the placement of one framing bit, for a total of 1.544 Mbit/s, calculated as follows [6]:

$$\left(8\,\frac{\text{bits}}{\text{channel}} \times 24\,\frac{\text{channels}}{\text{frame}} + 1\,\frac{\text{framing bit}}{\text{frame}}\right) \times 8\,000\,\frac{\text{frames}}{\text{second}}$$
$$= 1\,544\,000\,\frac{\text{bits}}{\text{second}}$$
$$= 1.544\,\frac{\text{Mbit}}{\text{second}}$$

2.2 DS3 (Digital Signal 3) is a digital signal level 3 T-carrier. It may also be referred to as a T3 line. The data rate for this type of signal is 44.736 Mbit/s (45 Mb). This level of carrier can transport 28 DS1 level signals within its payload also can transport 672 DS0 level channels within its payload [7]. DS3 interconnect cables must be made with true 75 ohm coaxial cable and connectors. Cables or connectors which are 50 ohm or which significantly deviate from 75 ohms will result in signal reflections which will lower the performance of the connection, possibly to the point of not working [8]. GR-139-CORE, Generic Requirements for Central Office Coaxial Cable, defines type 734 and 735 cables for this application. Due to losses, there are differing distance limitations for each type of cable. Type 734 has a larger center conductor and insulator for lower losses for a given distance. The BNC connectors are also very important as are the crimping and cable stripping tools used to install them [9].

3. The Proposed Network and Components

The proposed network for the simulation is a private network for a company The LAN structure for both locations is the same. In each location (four floors and in each floor are 10 Ethernet workstations connected to Ethernet switch , the Ethernet switch in each floor connected to main switch that connected to the router and then to IP cloud that represents the Internet by PPP_DS1 link or PPP_DS3 link. All network elements have been connected using 10Base_T links. Workstations in each floor can communicate with workstations in other floors within the same location using VoIP and workstations in the first location can also communicate using VoIP with workstations in the second location. The purpose of building two locations (LAN) for simulate the communications within the same location as local and communications between two locations as long distance VoIP communication, respectively.

4. Implementing the Network with OPNET

The simulation model for the VoIP network under study is illustrated in (Figure 1). Two subnets are added on the map and each represents a location in the company. The IP cloud symbol represents the internet and the application definition and profile definition symbols on top are very important. Subnet model (LAN) to each location has four floors and each floor contains an Ethernet switch and ten workstations as shown in (Figure2).

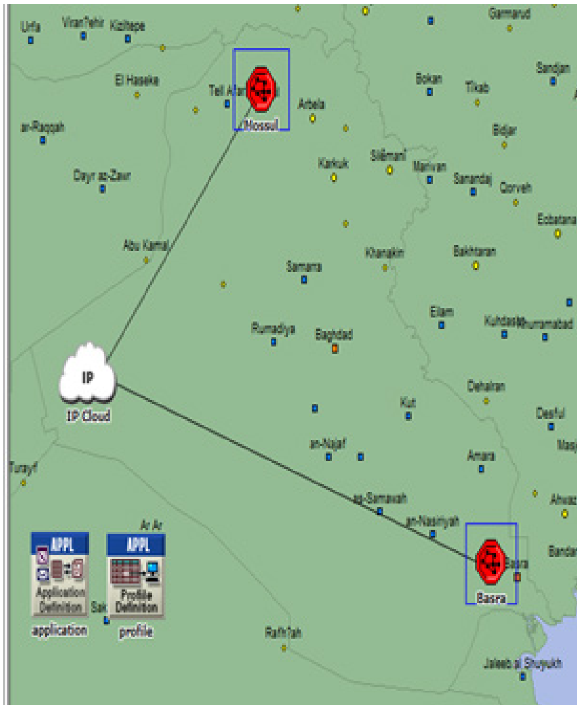

Figure 1: Simulation the Proposed VoIP Network

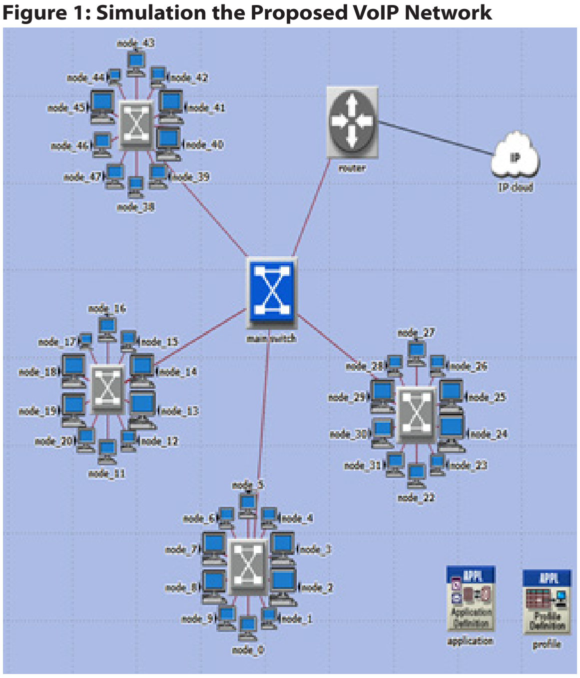

Figure 2: LAN Structure to each location in the Company

5. Network Simulation Results and Discussions

To measure the QoS of the VoIP application during collected major factors that affect the VoIP QoS such as: Voice delay (sec), Voice jitter (sec), Voice traffic sends (packet/sec) and Voice traffic received (packet/sec). The duration of simulation is 8 minutes. The blue line represent Non-busy VoIP network using DS1link; whereas the red line represent busy VoIP network using DS1link; whereas the green line represent busy VoIP network using DS3link.

Figure (3) shows the (Jitter = 0) in all three cases did not exceed the time constraint (0.075 sec).

Figure (4) shows the end to end delay in Non busy DS1 and busy DS3 VoIP network did not exceed the time constraint (0.15 sec) but in busy VoIP network DS1 exceed the time constraint (0.15 sec).

Figure (5) shows the voice traffic received in Non-busy DS1= 200 Packets/sec), the voice traffic received in busy DS1= 11640 Packets/sec) and the voice traffic received in busy DS3= 18000 Packets/sec).

Figure (6) shows the (voice traffic sent in Non busy DS1 = 200 Packets/sec), the voice traffic sent in busy DS1= 18000 Packets/sec) and the voice traffic sent in busy DS3= 18000 Packets/sec).

Figure (7) shows the bandwidth used in link called (utilization) which utilization link in Non busy DS1= 6.5%; whereas utilization link in busy DS1= 100% (overload link); whereas utilization link in busy DS3= 10.6%.

The results show that the overall traffic received rate is slower than the overall traffic sent rate in busy VoIP network using DS1 Link. The mismatch of traffic send rate and traffic received rate implies that the DS1 link is overloaded. The increase in end to end delay causes many packets arriving at the destination exceeded the time constraint (150ms). The mismatch of traffic sent and traffic received implies packet loss. More packet loss ratio in a network is less quality in that network. According to OPENT, It is the ratio of packets dropped to the total packets transferred multiplied by 100.

Packet loss ratio in busy DS1 = 6360/18000×100=35.3 %.

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 1.

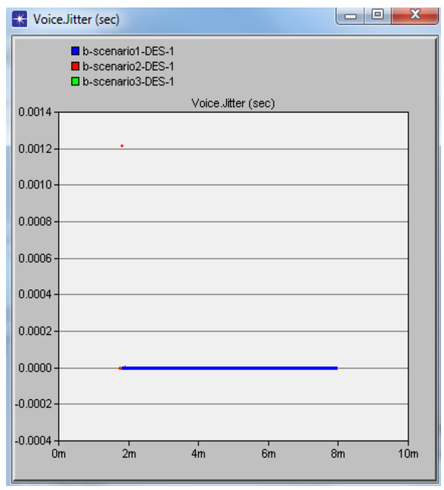

Figure 3: Voice Jitter

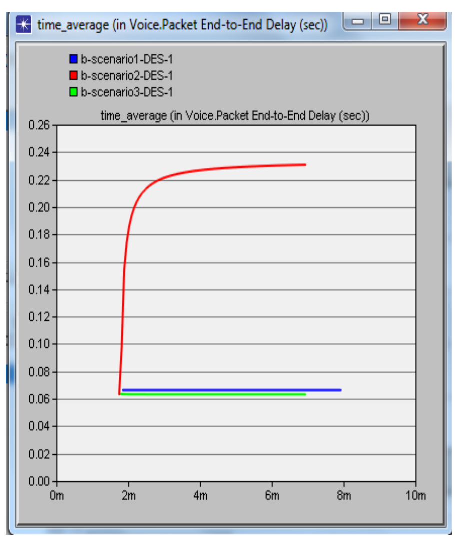

Figure 4: Voice Packet end to end Delay (sec).

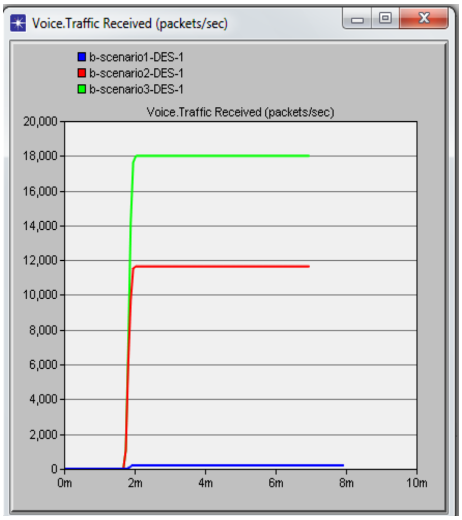

Figure 5: Voice Traffic Received (packet/sec)

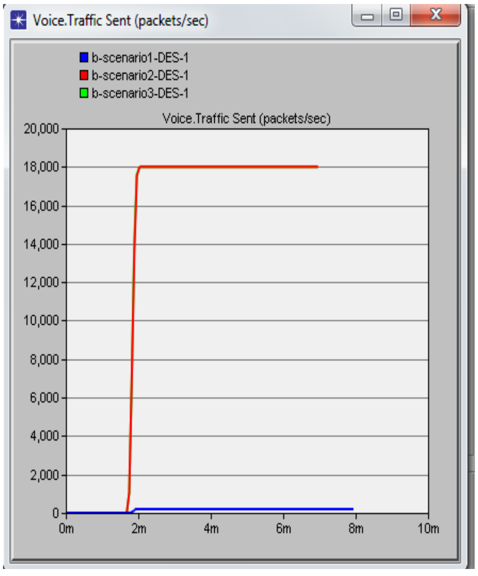

Figure 6: Voice Traffic Send (packet/sec)

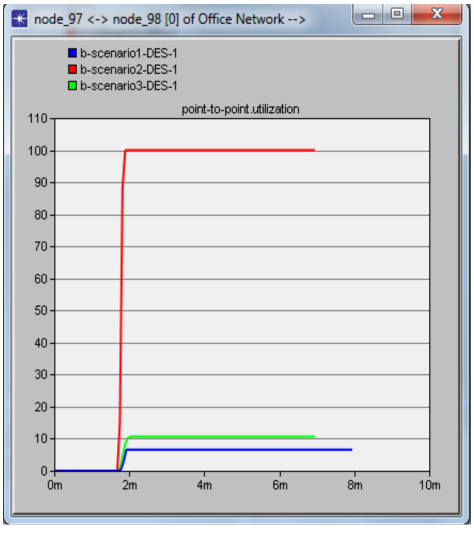

Figure 7: Link Utilization

Table 1: Comparison between non-busy VoIP Network, Busy VoIP Networks using DS1 and DS3.

Parameters	Non-busy DS1	Busy DS1	Busy DS3
Voice Jitter (sec)	0.0	0.00121	0.000000112
Voice Packet End-to-End Delay (sec)	0.066	0.268	0.063
Voice Traffic Received (packets/sec)	200	11640	18000
Voice Traffic Sent (packets/sec)	200	18000	18000
Packet loss	0.0	6360	0.0

From the result summarized in Table1, when the VoIP network becomes busy; overload happens and causes increase in the end to end delay and 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.

6. Conclusions

The presented research of the affects of busy VoIP network and non busy VoIP network on QoS VoIP application using OPNET and analyzed simulations of the results allow to be made conclusions as follows: When the VoIP network becomes busy; overload happens and causes increase in the end to end delay and 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.

7. References

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