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Simulation of Video Conferencing over IP Network with QoS Using Riverbed Modeler

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Abstract *:— Riverbed is a powerful network design and simulation tool that has gained popularity in industry and academia. However, there's no any known simulation approach exist about how to deploy a popular real time network service such as video conferencing*. *Video conferencing is a telecommunication technology which can allow two or more different locations communicating each other. It has become more and more important in daily life with the trend of using smart phones and tablets. The video conferencing is supported by many network applications, the major critical issue for network researchers and service providers is how to provide high quality of communication services with limited bandwidth capacity. Also, provide a performance comparison. The purpose of this paper is to simulate the performance of video conferencing over* ***IP*** *network with* ***QoS*** *schemes***.**

Keywords- (VC) video conferencing, *(QoS) Quality of Service and IP*

I. Introduction

Today, increasing numbers of people are dependent on services provided via Internet. Some of the examples of services provided through internet are education, research, social networking, banking, communication, etc. Although all of these services , there will be seeing in the tremendous growth in using the video conferencing (**VC**) as a fundamental tool for businesses to enhance communication and collaboration between employees, partners and customers [1]. Video conferencing is reliable technology for connecting the people to communicate visually across the globe. Face-to-face interactions are achieved irrespective of any geographical locations. The

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major problem in video conferencing is quality degradation and bandwidth fluctuation among the participants. The basic of video conferencing requires three things, environment, equipment and the network that will linked the sites together. The major feature of this multipoint video conferencing is to render the high quality video images at any capture resolution. The core concept used in video conferencing is digital compression in real time.

II. Related Works

There are exists a large body of research on Multimedia Traffic characterization either on wired or wireless **LAN**, Akkus *et al.* [1] used layered video encoding techniques within peer-to-peer architecture, and they developed a multi-objective optimization approach for the **P2P** video conferencing system. Also, Domoxoudis*et al.* [2] have been made a contribute measurement and modeling results on discrete-event queuing studies on the network performance of video conference traffic encoded by variable bit rate video encoders over **IP** networks, through their paper, they found the most critical point in modeling video conference traffic is the long-term trend of the frame size sequence autocorrelation function and not the exact fit of the frame-size histogram. Moreover, Dhomeja *et al.* [3] have been made evaluation to some parameters of Quality of Service **(QoS)** related to video conferencing over three major **WLAN** Standards **802.11a, 802.11b** and **802.11g**. Furthermore, Aamir *et al.* [4] have been made **QoS** analysis in **IP** network with more realistic enterprise modeling and presents simulation results using **OPNET**.

III. Layers of Video Conferencing

The components within a Conferencing System can be divided up into several different layers: User Interface, Conference Control, Control or Signal Plane, and Media Plane. Video conferencing User Interfaces **(VUI)** can be either graphical or voice responsive. Many in the industry have encountered both types of interfaces, and normally graphical interfaces are encountered on a computer. User interfaces for conferencing have a number of different uses; they can be used for scheduling, setup, and making a video call. Through the user interface the administrator is able to control the other three layers of the system. Conference Control performs resource allocation, management and routing. This layer along with the User Interface creates meetings (scheduled or unscheduled) or adds and removes participants from a conference. Control (Signaling) Plane contains the stacks that signal different endpoints to create a call and/or a conference. Signals can be, but aren't limited to, **H.323** and Session Initiation Protocol **(SIP)** Protocols. These signals control incoming and outgoing connections as well as session parameters. The Media Plane controls the audio and video mixing and streaming. This layer manages Real-Time Transport Protocols, User Datagram Packets **(UDP)** and Real-Time Transport Control Protocol

(RTCP [5]. The **RTP** and **UDP** normally carry information such the payload type which is the type of codec, frame rate, video size and many others. **RTCP** on the other hand acts as a quality control Protocol for detecting errors during streaming.

V. Standards of Video Conferencing

Most major vendors now support the **H.320** suite of **ITU** recommendations that define videoconferencing mechanisms over switched digital services such as **ISDN**. Similar recommendations have also been defined for high-speed wide area networks **(H.321**), isochronous networks **(H.322**), packet-switched local area networks **(H.323)** and **POTS** phone lines **(H.324).** The three umbrellas of standards for video conferencing of **ITU** are **[5] :**

1. **ITU H.320** is known as the standard for public switched telephone networks **(PSTN)** or video conferencing over integrated services digital networks. While still prevalent in Europe, **ISDN** was never widely adopted in the United States and Canada **[5]**.
2. **ITU H.264** Scalable Video Coding **(SVC**) is a compression standard that enables video conferencing systems to achieve highly error resilient Internet Protocol **(IP)** video transmissions over the public Internet without quality-of-service enhanced lines. This standard has enabled wide scale deployment of high definition desktop video conferencing and made possible new architectures, which reduces latency between the transmitting sources and receivers, resulting in more fluid communication without pauses **[5].** In addition, an attractive factor for **IP** video conferencing is that it is easier to set up for use along with web conferencing and data collaboration. These combined technologies enable users to have a richer multimedia environment for live meetings, collaboration and presentations.
3. **ITU V.80**: video conferencing is generally compatible with **H.324** standard point-to-point video telephony over regular plain old telephone service **(POTS)** phone lines **[5].**

The Unified Communications Interoperability Forum **(UCIF),** a non-profit alliance between communications vendors, launched in May **2010**. The organization's vision is to maximize the interoperability of **UC** based on existing standards. Founding members of **UCIF** include HP, Microsoft, Polycom, Logitech/LifeSize Communications and Juniper Networks **[5].**

VI. Queuing Disciplines

When the network is designed to service widely varying types of traffic, there is a way to treat contention for network resources by queuing, and manages the available resources according to conditions outlined by the network administrator. Each router, as part of the resource allocation mechanisms, must implement some queuing (algorithm) discipline that governs how packets are buffered while wait to be transmitted. Various queuing disciplines can be used to control the transmitted packets. The queuing disciplines also affects to the packet latency by decreasing the time that packets wait to be transmitted. There are three common queuing disciplines that can be analyses, they are priority queuing **(PQ)**, first-in-first-out **(FIFO)** queuing and weighted-fair queuing **(WFQ)**. Therefore, the benefits of techniques [6] are utilized in this paper because there are the most popular approaches in real time network service

As in Figure **1**. The mechanism of **FIFO** queuing is the first packet that arrives to router, it's the first packet to be transmitted. An exception here happened if a packet arrives and the queue is full, then the router ignores that packet at any conditions.

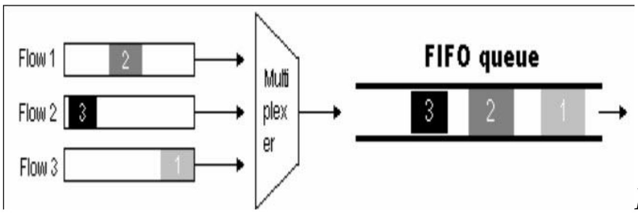

*Figure 1: FIFO queue **[6]***

The Weighted-fair queuing discipline provides **QoS** by provides fair (dedicated) bandwidth to all network traffic for control on latency and packet loss. The packets are classified and placed into queues according to information **ToS** field in **IP** header is use to identify weight (bandwidth). The Weighted-fair queuing discipline weights traffic therefore a low-bandwidth traffic gets a high level of priority. A unique feature of this queuing discipline is the real-time interactive traffic will be moved to the front of queues and fairly the other bandwidth shares among other flows, as shown in the Figure **2**.

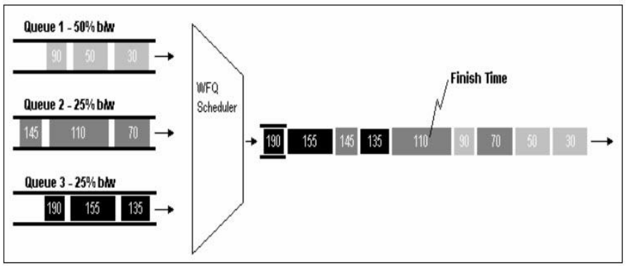

*Figure 2: WFQ queuing **[6]***

VII. Simulation Study

The simulation tool adopted in this paper is *Riverbed Modeler Academic Edition 17.5 Copyright **2014***, Riverbed Modeler is an object-orientated simulation tool for making network modeling and **QoS** analysis of simulation of network communication, network devices and protocols. Riverbed Modeler has a vast number of models for network elements, and it has many different real-life network configuration capabilities. These make real-life network environment simulations in Riverbed Modeler very close to reality and provide full phases of a study. Riverbed Modeler also includes features such as comprehensive library of network protocols and models, user friendly **GUI** (Graphical User Interface), Web report is feature allows you to organize and distribute the results of your simulations in form graphical results and statistics. Riverbed Modeler doesn’t have any programming knowledge so that it's easy to use and to deal with for any person.

The network model for simulation consists as shown in Figure **3**.

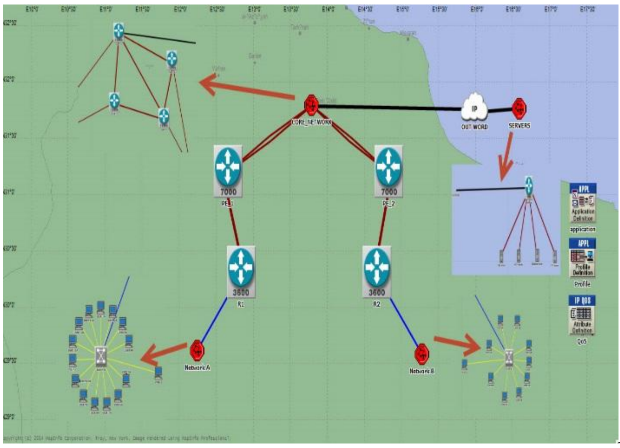

Figure 3: The Simulation Network Model

There are some important objects in Riverbed Modeler that are used to simplify and facilitate the work with it. In this paper, some of these objects that are used are: Application Configuration, Profile Configuration and **QoS** Attribute Configuration are shown in figure **4**, and figure **5** respectively :

1. Application Configuration Object: It is an object used to define and configure all Applications in the network according to the user requirements. Riverbed Modeler has most common applications like: *HTTP, E-mail, video conferencing, File transfer, Voice, and database*. In this paper, there is just a video conferencing application have been analyzed in the current network model, the other applications have been added without analyzing.

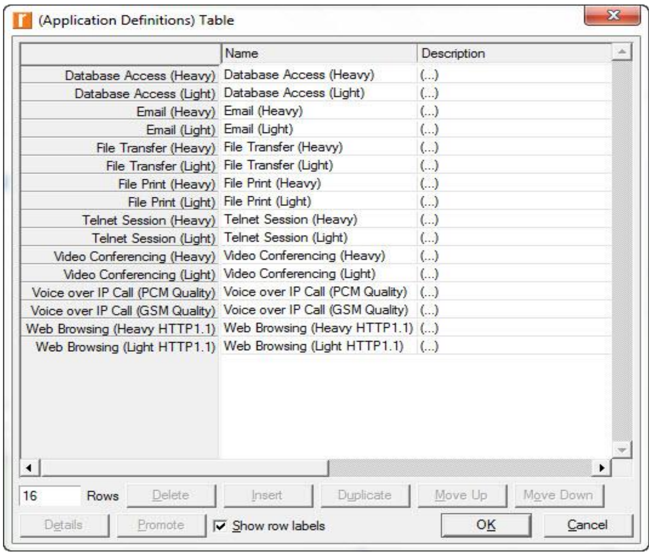

Figure 4: Application Configuration Object

2. Profile Configuration Object: It is an object that can be used to create users profiles, profile can contains one or more applications and each application can be configured by the starting time and ending time. In this paper, there are three profiles, and one of them is used to video conferencing called Video_ PF.

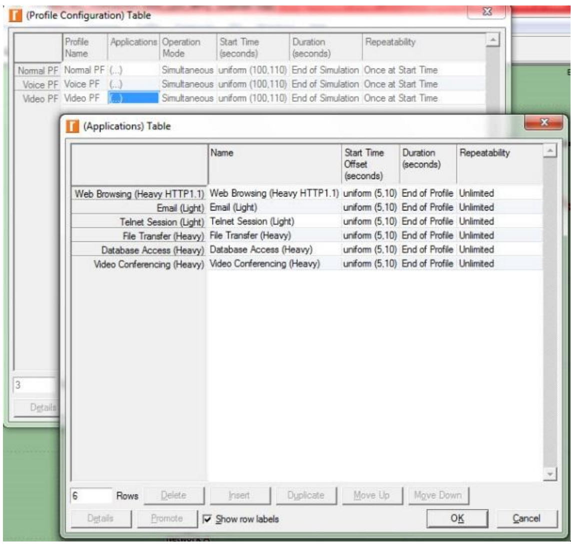

Figure 5: Profile Configuration Object

3. The **QoS** Attribute node: It is a mean of attribute configuration details that assess protocols at the **IP** layer. It deals with the two queuing profiles: **FIFO**, and weighted fair queuing **WFQ**.

VIII. Simulation results and Discussion

This paper provides a Comparison between the effects of different queuing duplicates such as **FIFO**, and **WFQ** for video conferencing. To measure the **QoS** of the video conferencing application during collected statistics (parameters) such as: traffic dropped, traffic sent (packet/sec), traffic received (packet/sec), packet delay variation, and packet end-to-end delay. The duration of simulation is **150** seconds and the utilization of the all links almost **100%,** and the results are obtained as shown in figures below.

In figure **6,** there is a big variation between normal scenario and queuing discipline scenarios, where **FIFO** it has the smallest delay.

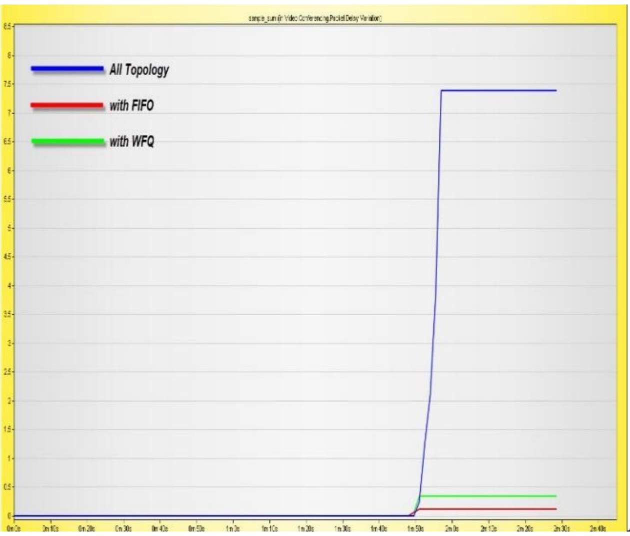

Figure 6: Packet delay variation

In figure **7** the traffic sent (packet/sec) for all scenarios almost the same; because there is no packet dropped that mean the network should be working properly. Whereas, figure **8** the traffic receiver (packet/sec), had little different between **FIFO** and **WFQ**, but in the 1st scenario the traffic is higher.

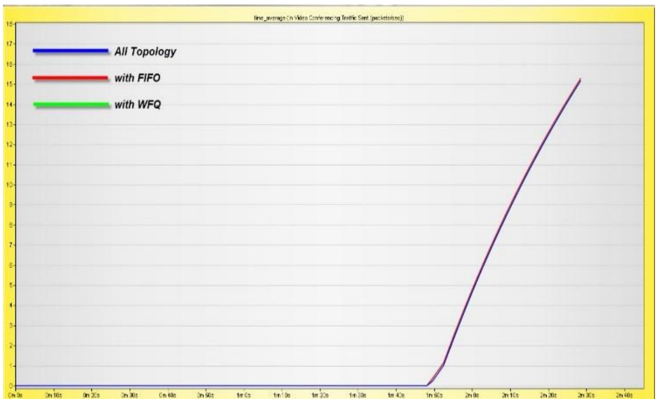

Figure 7: traffic sent of video conferencing

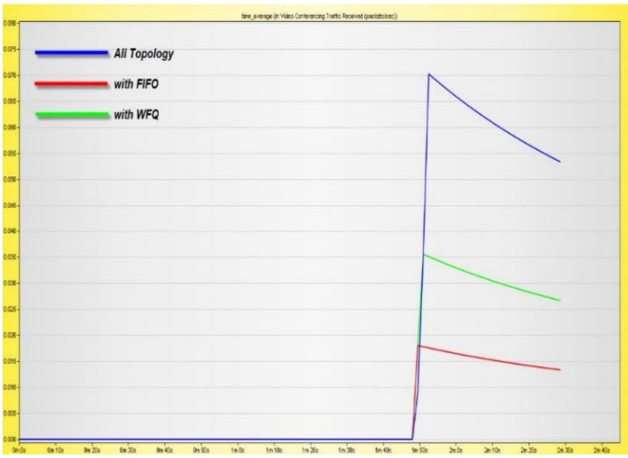

Figure 8: traffic received of video conferencing

In case of applying queuing discipline with high utilization, the congestion will occur . Then, the queuing discipline technique dropped the packets according to their roles. As a result, figure **9** is shown difference between scenarios on packet drop, and in the figure **10** is focused deeply between

FIFO and **WFQ** scenarios. In view of the fact that the first scenario doesn't contain a huge amount of packet drop compared to other scenarios.

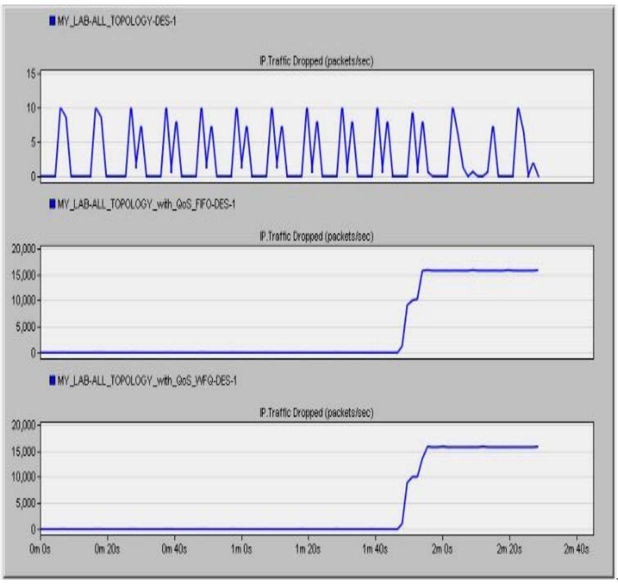

Figure 9: packet drop for video conferencing over all scenario

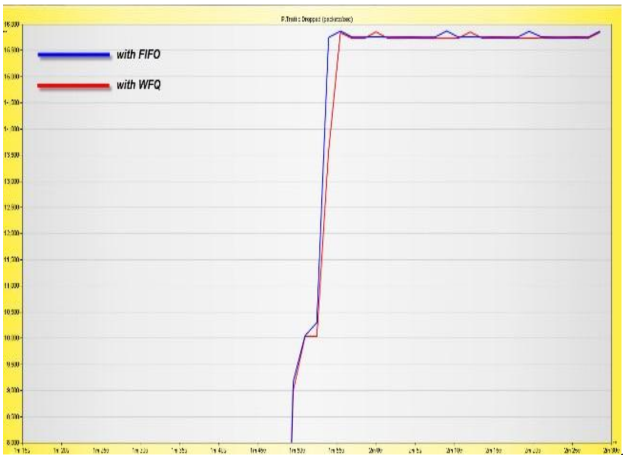

Figure 10: packet drop for video conferencing over queuing discipline

The packet delay variation over the node, and end-to-end delay, have been illustrating in the figure **11** and figure **12** respectively.

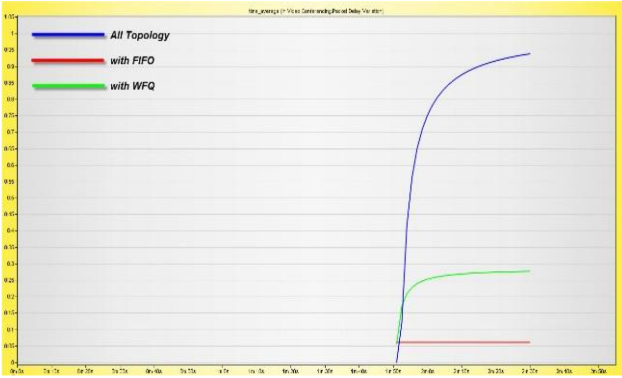

Figure 11: Packet delay variation for video conferencing over the node

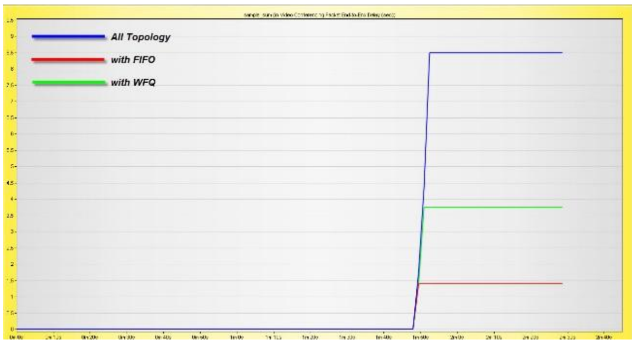

Figure 12: Packet end-to-end delay for video conferencing over the node

VIII. Conclusion

This paper presents the effects of different queuing disciplines on the performance of video conferencing using *Riverbed Modeler*. Simulations results are concluded that; improving the **QoS** of video conferencing traffic based on the Priority and Weighted-fair queues are the most appropriate scheduling schemes because the values of the parameters are within the acceptable range. The analysis was done in terms of delay and its variant for video conferencing, although the other services are present, according to our result the delay is highest for the normal scenario (without **QoS**) and the lowest for **FIFO** scenario. The simulation results show that **FIFO** is a better discipline than other scheme.

IX. References

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