

Application of Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) in Teleconference System

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Abstract – Teleconferencing is the act of meeting and exchanging information through a telecommunication medium. In teleconferencing system, network engineers are faced with different challenges like best protocols requirements; impact of Quality of Service (QoS) to be used to meet up with the latest network protocols and appliances in the design of teleconference system. This paper presented a comparative analysis on two network protocols in a teleconference system, Transmission Control Protocol (TCP) versus a User Datagram Protocol (UDP) was evaluated and compared applying Optimized Network Engineering Tools (OPNET) simulator. The experimental result showed differences in number of videoconference sessions, with few Packet Per Seconds (PPS) recorded in both the UDP and TCP, thereby reflecting the correctness of this comparative evaluation task.

Keywords: teleconference; quality of service (QoS); TCP; UDP; network protocol; simulator.

I. INTRODUCTION

Teleconferencing can be referred to a system of communication that allows its users' to exchange information across a distance through a digital medium [1]. The common types of teleconferencing system in use include: audio teleconferencing medium that enables its users to exchange audio messages among one another, audiographic teleconference system [2]. Computer teleconferencing system integrates an audio and graphic teleconferencing system in its use to produce a more sophisticated system for use with a computer system and video teleconference system [3]. The systems of video conferencing allow people at diverse geographical areas to hear and see one another, interchange information and work together using

communicating video, audio and computer technologies. The two most generally applied protocols in transport layer are User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) [4].

TCP is the most widely used protocol on the internet today. Any web page today uses TCP data packets, it is a two-way data highway of sending and receiving data in the internet. TCP packet performs error checking in the internet to reduce corrupted files and web pages. User Datagram Protocol does not perform the error checks because the error checking slows down the back and forth of sending and receiving data transmission. The UDP data packet simply sends the data and does not wait for a reply from the receiver.

The business television (BTV) used in media houses to share news and information from one studio to another and distance education are dwelt in teleconference system [5]. Teleconference is a medium employed for educational purposes where instructors or lecturers can meet their students at distance for learning purposes [6]. The types of teleconference system differs in design subject to their functions [7]. Despite their differences, the common features generally shared are: telecommunications channel, that take any platform or medium depending on the distance involved in the design of such system [8]. Also link people at different locations before a communication is established, it must involve at least two or more people trying to link up with each other.

Teleconference system is highly tasking and resource demanding on the network during designing process [9]. As such, the setting-up of a teleconference system usually pose the following problems which this seek to find a solution to, particularly at the transport layer of the Open System Internetworking (OSI) model; the most suitable network protocol to be used in deploying a teleconference system into an existing network to deliver an optimum service, effect of deploying a

teleconference system on the QoS of an already running system and how the old system of support teleconferencing to meet up with the standard QoS of such system.

This paper reflected on two types of teleconferencing system (i.e. Audio and Video Teleconference) by conducting comparative analysis on TCP and UDP network protocols in a teleconference system using different evaluation metrics in terms of throughput, latencies for their uplink, and downlinks when hosting a teleconferencing system.

II. RELATED WORK

[4] proposed adaptation for UDP and TCP protocols in video streaming using VANET. Comparison between the two protocols were conducted to select the effective protocol for transmission at transport layer for video streaming in VANET. The metrics used for evaluation were: throughput, ratio of packet delivery, end-to-end delay and PSNR. The experimental results revealed that UDP produced a faster video data transmission than TCP. While the higher quality of video was achieved in TCP than UDP because TCP is more reliable than UDP.

[10] designed video multicasting system for IntraLAN, the system was particularly made for real-time video streaming over Local Area Network (LAN). The system allowed video transmission directly from computer. It is not like other API's, the software developed client for RTP that can be used in any company and configured in the intranet of that company. The system functionality could reach distance education, e-learning, meetings through video conferencing on the internet.

[11] examined the collective influence in sparsity of network and node of network density on the Peak Signal Noise to Ratio (PSNR) and performance in jitter for proactive and reactive protocols in routing for ad-hoc networks. The study produced information on the collective effect of mobility and sparsity performance of these protocols. Results were validated through the application of an integrated simulator-evaluator environment comprising of the network simulator NS2, and the video evaluation framework evalvid.

[12] conducted a study to ensure optimization of bandwidth usage and as well as robust quality for a video conference, even if there are variations in network bandwidth. Three different TCP congestion avoidance algorithms were used. These algorithms observed and evaluated the network quality to adapt the video stream rate properly. One algorithm was only based on packet loss and also used as a baseline. The other two uses packet loss in conjunction with round trip time (RTT) to evaluate the network. The user experience was deemed better when using the algorithms. The algorithms that was based on both packet loss and RTT was deemed superior. There are however still a few things to adapt in the camera software and hardware before a complete system can be developed.

[13] came up with a common way out based on diverse efforts that can be employed to obtain desired

functionality over internet for the real-time applications. The system presented a basic library referred to as TCP-over-UDP for real-time applications. The optimization of the library will be completely suitable for interaction in real-time settings.

[14] investigated MANET routing protocols effectiveness for applications in video. Common routing protocols namely Dynamic Source Routing (DSR), Ad hoc On-demand Distance Vector (AODV), Temporally-Ordered Routing Algorithm (TORA), Optimized Link State Routing Protocol (OLSR), and Geographic Routing Protocol (GRP) have been considered. A performance analysis of these routing protocols were done comparatively for supporting video streaming applications. The results showed that the performances of a routing protocol differ depending on the scenario in the network as well as categories of video traffic used.

III. METHODOLOGY

In this study, the simulation of the two network protocols for use in a teleconferencing system was carried out. The procedures involved was illustrated and the result was discussed in later session. Different network topologies were used for connecting all the nodes together as shown in the Fig. 1.

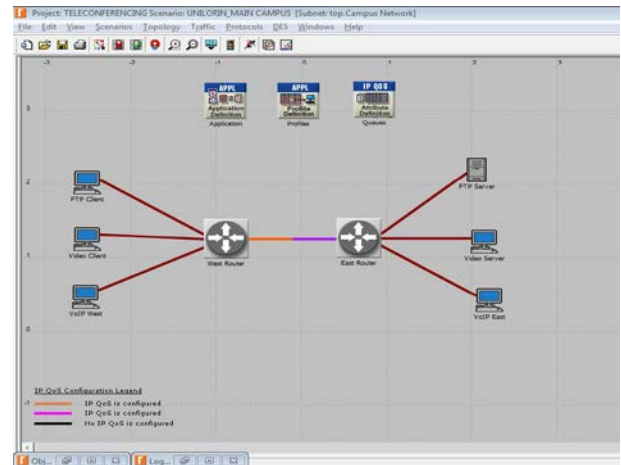


Fig. 1. Setup of the TCP and UDP connection for a Campus Area Network.

III.1. Initialization of the Network

To initialize the algorithm of the network nodes ready to support teleconference servers, the following steps were modeled and the network was set up to the following working criterion from the model formulated:

Step 1: The first step is to create a profile that can support "File Transfer Protocol" by opening the "Edit Attributes" to include FTP in the Supported profile for its application. The "row attribute" was set to "1" while the profile name was changed to "FTP Profile"

Step 2: The video client attribute is then setup to enable the simulation support video streaming application in the data exchange. The row attribute was setup to 1, while the profile name was changed to video.

Step 3: A VoIP attribute was initiated to hold the VoIP services that allows the simulation with a profile name of VoIP profile East and West.

Step 4: The VoIP was created in two axes, the East and West which the both were assigned an attribute with a profile name of name for VoIP profile East and West.

Step 4: The work is saved for further manipulation.

Step 5: End

After the application profiles for the three applications needed to host a teleconference system has been configured successfully, the other configuration of the simulation will be added. These configurations include the “QoS Attribute Config, the workstations, an Ethernet server and two gateway routers. The connection of routers using a bidirectional PPP_DS1 link to facilitate a full-duplex connection between them. The routers and the workstations were then connected using a “bidirectional 10Base_T connector as shown in Fig. 1.

III.2. Algorithm to View and Analyze the Results

The algorithm to display and perform the results analysis is described as follows:

Step 1: Access the result of the simulation above, the “Compare Results” menu is opened.

Step 2: Statistics for the “IP Traffic Dropped” is opened by selecting the “Show” icon.

Step 3: Video session of the simulation had another graph generated to display the traffic received.

Step 4: Result is collected for the jitter and network packet End-to-End Delay.

Step 5: Result for the queuing delay for the two Router links will also be deduced from the graph generated.

Step 6: End

IV. RESULTS AND DISCUSSION

IV.1. Results of Simulation

The simulation process was conducted using the OPNET Modeler Academic Edition. The simulation results were applied to conduct performance evaluation of the system as illustrated with the interfaces shown in Fig. 2-7.

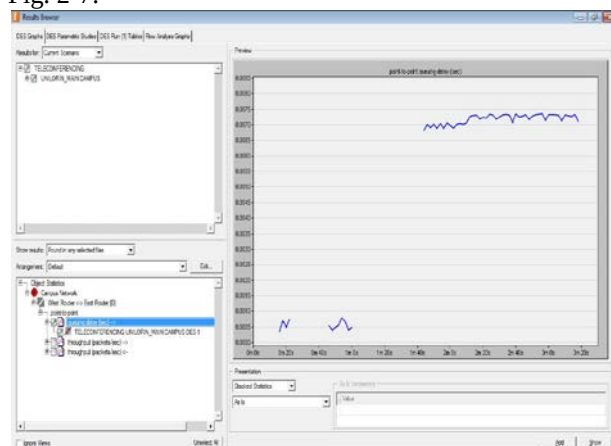


Fig. 2. Result of Latency for TCP connection.

From Fig. 2, it can be seen that TCP has a stronger connection as it is generated in the Global Statistics configuration.

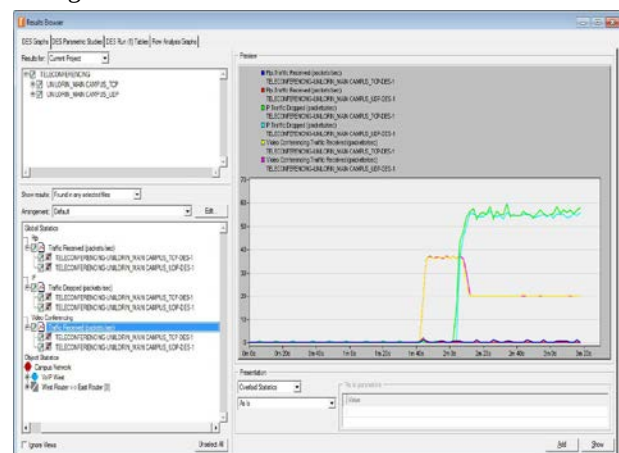


Fig. 3. Summarized Log showing the throughput, Latency, Jitter of the UDP and TCP connections comparatively.

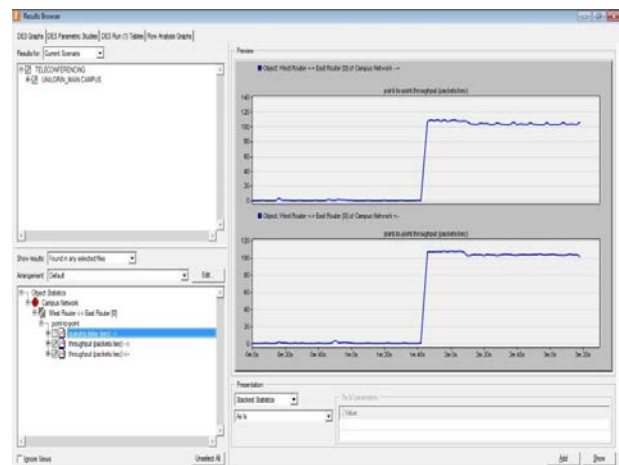


Fig. 4. Throughput result, Bi-direction.

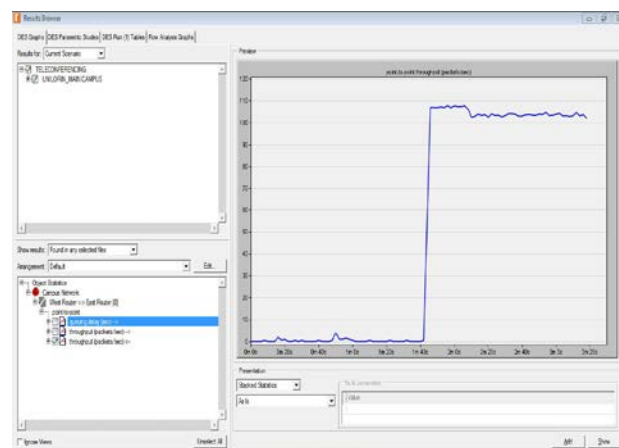


Fig. 5. Throughput Result for TCP connection.

IV.2. Generation of Traffic for TCP and UDP Video Conferencing (Packet end-to-end Delay)

The result of the generated traffic for TCP and UDP Video conferencing (Packet end-to-end Delay) is shown in Tables 1-4.

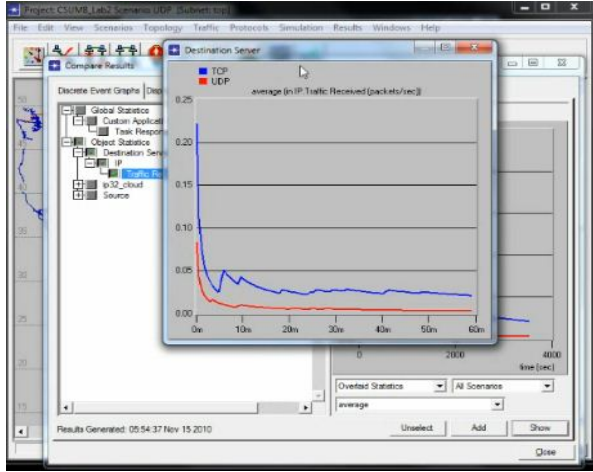


Fig. 6. IP Traffic Received (packet per seconds).

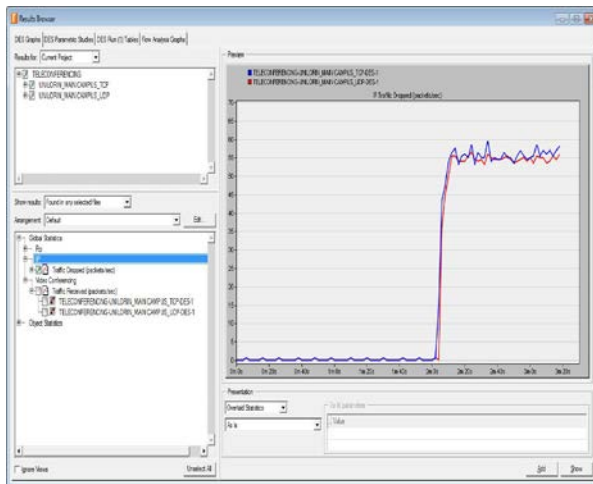


Fig. 7. UDP & TCP Traffic Dropped.

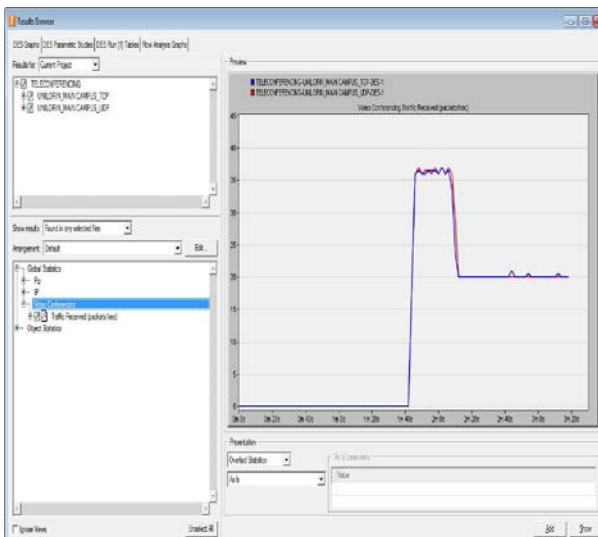


Fig. 8. UDP & TCP Traffic.

IV.3. Summary of results

Important statistics are given in Table 1 with the throughput from the measure of the delay (the difference in the data exchange of the two network protocols) at

their west and east nodes. It can be seen that the data transmission was almost at the same traffic showing that two protocols have similar throughput capacity for use in a video teleconference system.

TABLE 1. Generated Traffic for TCP and UDP Video Conferencing.

STATISTICS (bytes/sec)	MEDIUM	HIGHEST	LOWEST
TCP	49.610	107.64	0
UDP	49.565	107.63	0

TABLE 2. TCP and UDP Statistics for the Video Conference session (Jitter).

STATISTICS (bytes/sec)	MEDIUM	HIGHEST	LOWEST
TCP	0.005324	0.007368	0.00034715
UDP	0.017500	0.020000	0.0150000

TABLE 3. Generated Traffic for TCP and UDP Voice Conferencing.

STATISTICS (bytes/sec)	MEDIUM	HIGHEST	LOWEST
Traffic Received	126.427989801000	405.072	0
Traffic Sent	126.428926484000	405.100	0

TABLE 4. TCP and UDP Voice Conferencing: Statistics for the Delay Variation (in microseconds).

STATISTICS (bytes/sec)	AVERAGE	MAXIMUM	MINIMUM
TCP	35.807	75.448	0
UDP	37.445	71.346	4.96

Table 2 shows a large difference in their jitter (The jitter of a network is the difference in time stamps of two consecutive packets from the same source to the time it takes to return to that source. i.e. $\text{jitter} = (t_4 - t_3) - (t_2 - t_1)$). This was as a result of the difference in their orderliness and reliability which reflected in their simulation performance for use in a video conference session.

Table 3 presents the delay difference in the voice simulation for the teleconference system which indicates very slight or no change in the amount of packets in byte that can be broadcast at a time on TCP and UDP.

Table 4 produces the jitter in microseconds of the two networks. There also appear to be large disparity in their return time and sent of the two network protocol under consideration. But it was also observed that UDP

had an effective performance towards the end of the simulation showing its initiation is faster than TCP because it does not spend time on connection verification or challenge handshake. Hence, it is faster and more reliable as soon as the connection is established.

V. CONCLUSIONS

Network protocol remains a digital language through which communication is done with others on the internet. It is a set of mutually acknowledged and applied rules at both ends of the communications channel for the proper exchange of information. By adopting these rules, two devices can communicate with each other and can interchange information. The two popular transport layer protocols in a network include: Transmission Control Protocol (TCP) and User Datagram Protocol (UDP). The TCP provides a reliable connection that is responsible for the end to end data transmission, while User datagram protocol (UDP) offers fast but does not guaranteed data transfer. This paper performed a comparative analysis on TCP and UDP in teleconferencing platform. The experimental results were analyzed in order to show the effect of performance in teleconferencing system.

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