SFU

ENSC 427

Communication Networks

Simon Fraser University

Spring 2011- Final Project

Comparison and Analysis of FIFO, PQ, and WFQ Disciplines in OPNET

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List of Acronyms

10BaseT	10 Megabits per second, Baseband, Twisted Pair
ETE	End-to-End Delay
FIFO	First In, First Out Queuing
FTP	File Transfer Protocol
HTTP	Hypertext Transfer Protocol
IP	Internet Protocol
MOS	Mean Opinion Score
OPNET	Optimized Network Engineering Tools
PDV	Packet Delay Variation
PPP-DS1	Point-to-Point Digital Signal-1
PQ	Priority Queuing
QoS	Quality of Service
ToS	Type of Service
VoIP	Voice over IP
WFQ	Weighted Fair Queuing

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1. Abstract

With recent advancements in greater network speeds and higher bandwidths, the online use of applications that support services such as VoIP, video conferencing, and File Sharing Protocol (FTP) have become more prevalent. While some applications such as web browsing (HTTP), email, and FTP are insensitive to the delay of transmitted information, VoIP and video conferencing are very sensitive to delay, packet losses, and jitter [1]. Queuing disciplines are therefore implemented in routers to govern, control, sort, and to prioritize packets in the buffers prior to their transmission. FIFO, PQ, and WFQ queuing are implemented in OPNET and various parameters including but not limited to: average queuing delay, average packet drop rate, MOS, jitter, and average end-to-end delay are studied and comparison of these parameters is made for the three queuing disciplines studied in order to pick the right queue for each application.

2. Introduction and Project Scope

Traffic management is an important tool used for the allocation of network resources with the various, complex, and bandwidth demanding applications that have become common on the internet today. Congestion control mechanisms are used to govern situations with excessive network traffic; without the proper implementation of such mechanisms, network delays will cause an increase in congestion through automatic retransmission of information, thereby decreasing network throughput [2]. Real time applications such as VoIP and video conferencing are more susceptible to such delays and as such proper queuing and traffic management at the packet level must be considered to provide the required QoS to the end user.

The following project will attempt to study the effects and performances of three different queuing disciplines (FIFO, PQ, and WFQ) as applied towards some of the supported OPNET applications (FTP, voice, and video conferencing). Each queuing discipline considered will constitute a scenario in OPNET and the three applications of FTP, voice and video conferencing will be studied separately under each scenario. Comparison of various collected statistics will be made in order to justify the type of queue that would be most suitable to use under each of the applications considered. More detailed information on the scope of the project is provided in the OPNET Implementation section of the report.

3. Background on Queues Considered

3.1. First-in, First-out Queues (FIFO)

FIFO, or First-In, First-Out queuing is the simplest of the queuing disciplines studied. In FIFO queuing, the first packet to arrive at the buffer is the first packet to be transmitted. It is also important to mention that under this queuing technique, all packets are treated equally regardless of the application that is being utilized, and regardless of the importance of the packets [3]. Figure 1 illustrates the mechanism by which FIFO queuing works.

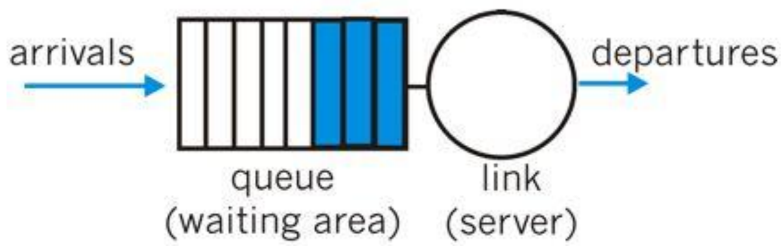


Figure 1: FIFO Queuing.

As can be observed in Figure 1, arriving packets from various applications are put into a single packet buffer in the order in which they arrive, until the buffer is full. Performance deterioration can occur in this queuing discipline as a result of shorter inter-arrival times between packets into the buffer or as a result of variable packet lengths clogging up the queue and causing various delays, jitter, and packet losses. It is also important to mention that due to the limited amount of buffer space available at each router, packets that arrive at a full buffer are dropped, regardless of the flow that a packet belongs to or its importance. These losses present significant problems and degradation of the quality of transmitted signals in real-time applications such as VoIP and video conferencing.

3.2. Priority Queues (PQ)

Priority Queues are based on FIFO queues with an important distinction; while packets are treated equally under the FIFO queues; PQs sort packets in the buffer according a priority tag which reflects the importance and urgency required in the transmission of packets. Furthermore, in contrast to FIFO Queues, Priority Queues are not made up of a single buffer. Figure 2 illustrates the mechanism in which priority queues work.

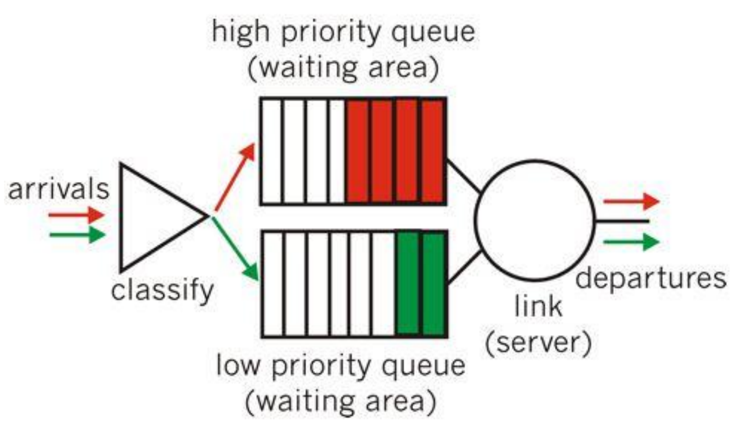


Figure 2: Priority Queuing.

As can be seen in Figure 2, arriving packets are tagged with a certain priority, thus allowing packets with higher priority to cut to the front of the line in the sorted packet buffer. Applications that require negligible delay times can use this queuing method to differentiate and prioritize their packets from other packets in order to manage the limited network resources available such as bandwidth. Real-time applications such as VoIP and video conferencing should in theory observe less delay under this queuing discipline.

3.3. Weighted Fair Queuing (WFQ)

Similar to Priority Queues, arriving packets are tagged and placed into the appropriate buffers as they wait to be serviced. One important difference between this queuing discipline and the former method is that all buffers are serviced in a circular manner by a WFQ scheduler. Figure 3 illustrates the mechanism by which WFQ operates.

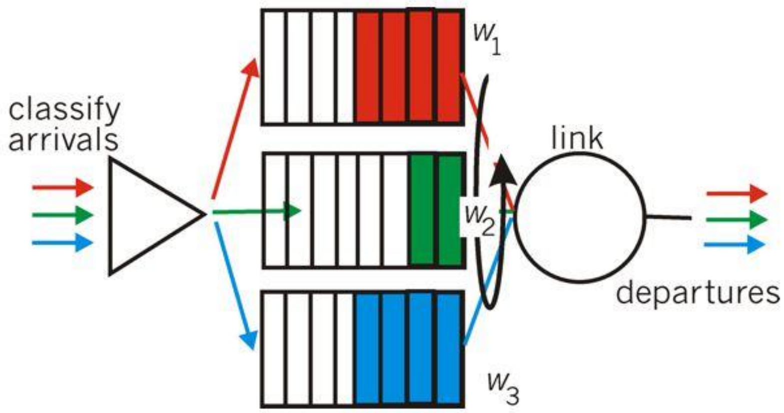


Figure 3: Fair Queuing.

In order to control the percentage of a links bandwidth that each flow will get, a weight factor is included and is assigned to the various queues based on the requirements and needs of the client. This method of queuing is called the Weighted Fair Queuing, and is widely used in QoS architectures and in various routers.

4. General Background and Definition of Terms

In order to present the results of the analysis and the effectiveness and applicability of the different queuing disciplines, a number of the parameters affected and analyzed must be defined first.

4.1. Quality of Service (QoS)

Generally speaking, the Quality of Service is a term that is used to refer to the control mechanisms for resource reservation. Real-time applications such as voice and video require a certain level of end-to-end QoS. Two main approaches to establishing Quality of Service are the differentiated service and the guaranteed service. As the name implies, the former approach allows for the relative preferential treatment of some class of traffic over other classes of traffic. Prior to entry into the routers inside the network, packets are marked with the kind of treatment that they are to receive. The required level of the Quality of Service, however, is not assured under this approach. The latter approach provides a guaranteed QoS by providing a limiting bound on the ETE delay that packets within a certain queue experience. Additionally, in order to provide a guaranteed QoS, resource reservations in the router need to be set up along the path that packets are to flow [4].

4.2.End-to-End Delay (ETE)

End-to-End delay refers to the transmission time of a packet across a network from its source to its destination. Real-time applications including voice, video, and online gaming have delay requirements that need to be met in order to provide a seamless and natural client experience. The effects of the various queuing disciplines on ETE will be studied and analyzed during simulation.

4.3.Jitter

Jitter is defined as the variation in the End-to-End delay, as packets are placed into different queues; the ETE delay in the transmission of the packets from the source to the destination will vary depending on the position of the packets in the queue and will also vary as a result of the different queue sizes; therefore, jitter should be minimized to improve the overall quality of the transmitted information, especially in applications requiring real-time data transmission.

4.4.Packet Loss

Packet loss refers to the failure of packets in reaching their destination when travelling across a network. As discussed above, one source of packet loss is due to buffer overflows that can be caused when packets entering into the queue are doing so at a faster rate than those that are leaving the buffer. Packet loss has a significant and noticeable effect on the overall quality of the received signal, especially in real-time applications. Other causes of packet loss include signal degradation and noise.

4.5.Mean Opinion Score (MOS)

The Mean Opinion Score provides a numerical measurement of the quality of a voice signal that is perceived after it has been transmitted [5]. Table 1 provides the rating scheme used to determine the perceived quality of voice signals.

Mean Opinion Score (MOS)		
MOS Value	Perceived Quality	Degree of Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

Table 1: MOS Values and their Perceived Voice Quality.

4.6.Packet Delay Variation (PDV)

Packet Delay Variation is a measure of the difference in the End-to-End delay between packets in a flow, ignoring any packets that have been lost. In OPNET, PDV corresponds to the variance of the delay [6].

4.7. Utilization

Utilization represents the percentage of consumption a channels bandwidth to date, with a value of 100 indicating full usage of the channel bandwidth.

4.8. Throughput

Throughput represents the average number of messages (packets or bits) that have been successfully transmitted or received by the transmitter or the receiver channel per second. Throughput is a measure of the consumption of the digital bandwidth in a network that is being considered.

4.9. Average Queuing Delay

The Average Queuing Delay could be defined as the average time taken from the point at which a packet arrives into a queue up to the point where that packet is transmitted and leaves the queue. It is therefore desirable to keep this statistic as small as possible, especially in real-time applications such as voice and video.

4.10. IP Packet Drop

The IP Packet Drop statistic represents the number of IP datagrams dropped per second by all nodes across all IP interfaces. Insufficient space in the queue is one of the reasons why IP datagrams could be dropped.

5. OPNET Implementation

5.1. Project Scope and Overview

In order to analyze and compare the three queuing disciplines described above, a network topology had to be built initially. The scope of this project considered three separate scenarios under a $10\ x\ 10$ campus network scale. The three scenarios considered are summarized in Table 2. Simulation was run for five simulation minutes in each scenario. The results obtained provide the basis for the comparisons and for the conclusions made regarding the three queuing disciplines.

	Scenario 1	Scenario 2	Scenario 3
Queuing discipline	First-in, First-out Queuing	Priority Queuing	Weighted Fair Queuing
Simulation Time	• 5 minutes	• 5 minutes	• 5 minutes
Applications considered	• FTP • VoIP • Video Conferencing	• FTP • VoIP • Video Conferencing	• FTP • VoIP • Video Conferencing

Table 2: Three Scenarios Studied Along with Applications Considered in Each.

Tables 3 and 4 summarize the global and object statistics collected and analyzed respectively.

Statistics for	Global Statistics
FTP	• Traffic Sent • Traffic Received
VoIP	• End-to-End Delay • Jitter • Mean Opinion Score • Packet Delay Variation • Traffic Sent • Traffic Received
Video Conferencing	• End-to-End Delay • Packet Delay Variation • Traffic Sent • Traffic Received
IP	• Traffic Dropped

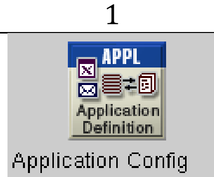
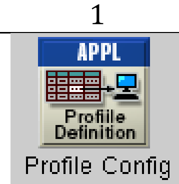
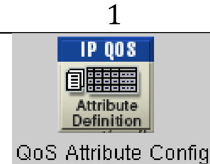
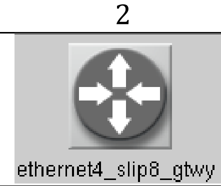
Table 3: Global Statistics Collected.

Statistics for	Object Statistics
Point-to-point	• Average Queuing Delay -> • Throughput -> • Utilization ->

Table 4: Object Statistics Collected.

5.2.OPNET Models Used

Table 5 summarizes the OPNET models used in order to create the network topology.

Model Name	Application Configuration	Profile Configuration	QoS Attribute Configuration	ethernet4_slip_gtwy
Quantity Used	1	1	1	2
Model Icon				

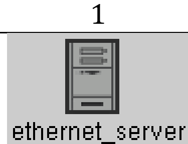
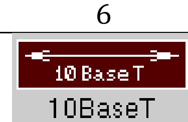
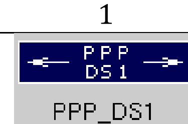
Model Name	ethernet_server	ethernet_wkstn	10BasteT	PPP_DS1
Quantity Used	1	5	6	1
Model Icon				

Table 5: Summary of Models used in OPNET.

5.3. Network Topology

The models listed in Table 5 were used to create the topology of the network considered. All three scenarios considered made use of the same network topology with some changes to reflect the different queuing disciplines considered.

Table 6 summarizes the X and Y coordinates of the models used and placed in the workspace.

Model Name	X Coordinate (km)	Y Coordinate (km)
Completed Network	10	10
FTP	1.875	2.5
VoIP A	1.875	3.75
Video Conferencing	1.875	5.0
Router A	3.75	3.75
Router B	6.25	3.75
FTP Server	8.125	2.5
VoIP B	8.125	3.75
Video Conferencing Server	8.125	5.0
Application Definition	1.875	8.0
Profile Definition	5.0	8.0
QoS Attribute Definition	8.125	8.0

Table 6: Summary of the X and Y Coordinates used in the Network Topology.

Figure 4 illustrates the network topology that was considered. As can be seen, Router A was connected to Router B using a bidirectional PPP_DS1 link, while all other workstations and the FTP server were connected to the routers by making use of the 10BaseT link. The Application, Profile, and QoS Attribute Definitions were then modified to support the network topology shown. Furthermore, the attributes of all the workstations, routers and the server were also modified to support the considered network topology. Upon completion of the network shown below, the topology was duplicated twice more to allow for the other two scenarios, and the QoS Configuration was changed to represent each of the queuing disciplines considered.

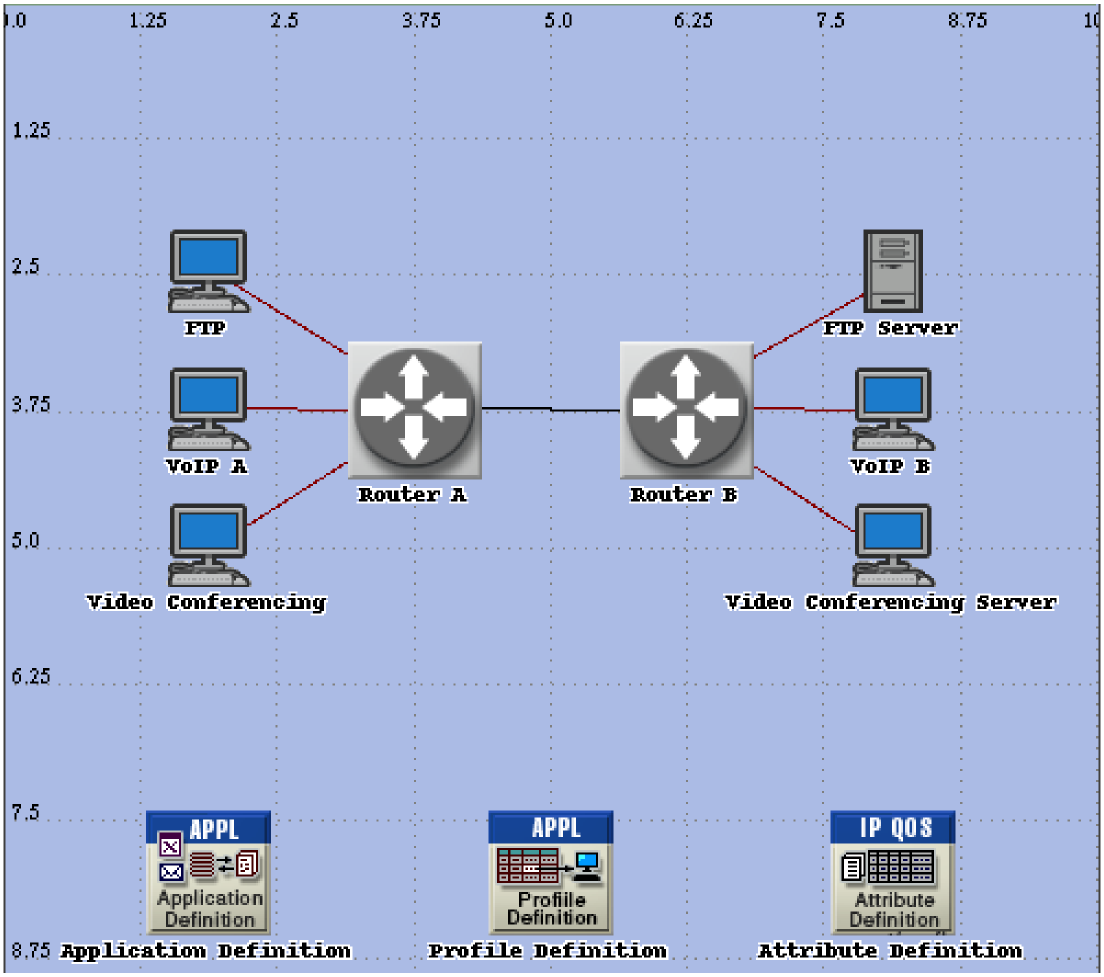


Figure 4: Completed Network Topology.

Table 7 illustrates some of the major additions and modifications made to the Application Definition.

Application Definition Attributes			
Application Name	FTP	VoIP	Video Conferencing
Discription	High Load	PCM Quality Speech	Low Resolution Video
Type of Serive (ToS)	Best Effort (0)	Interactive Voice (6)	Streaming Multimedia (4)

Table 7: Major Settings Added and Modified under Application Defintions.

Table 8 illustrates some of the major additions and modifications made to the Profile Definition.

Profile Definition Attributes			
Profile Name	FTP_Profile	VoIP_Profile	Video Conferencing_Profile
Application Name	FTP	VoIP	Video Conferencing
Start Time (seconds)	Constant (100)	Constant (100)	Constant (100)

Table 8: Major Settings Added and Modified under Profile Defintions.

Table 9 illustrates the attributes used for the workstations.

Workstation Attributes			
Workstation	FTP	VoIP A & VoIP B	Video Conferencing
Supported Profiles	FTP_Profile	VoIP_Profile	Video Conferencing_Profile
Supported Services	-	VoIP	-

Table 9: Major Settings Added and Mmodified under the Attributes of each Workstation.

Table 10 illustrates the attributes used for the servers.

Server Attributes		
Server	FTP Server	Video Conferencing Server
Supported Services	FTP	Video Conferencing

Table 10: Major Settings Added and Modified under Attributes of each Server.

The QoS Attribute Definitions settings were left as default values.

Scenario 1 considered the FIFO Queuing discipline. In order to realize this queuing method, the PPP_DS1 link was selected and the QoS Scheme was changed from its default value to FIFO. Figure 5 illustrates this configuration.

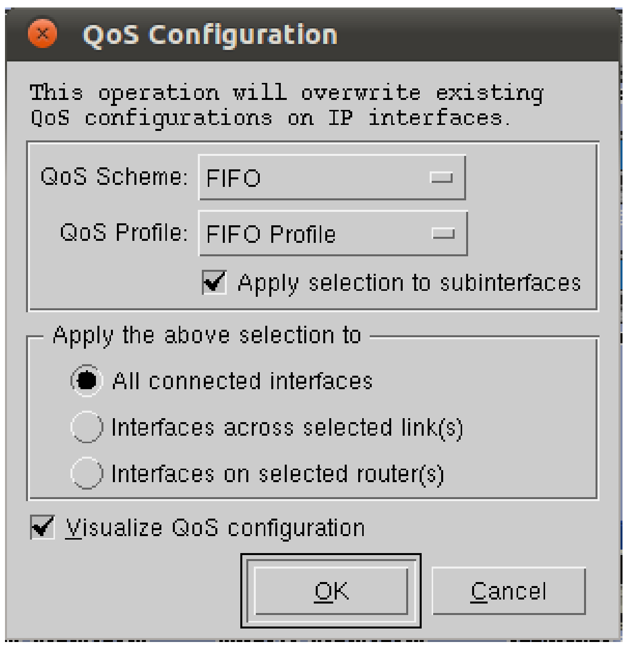


Figure 5: QoS Configuration Settings for Scenario 1.

Scenario 2 considered the Priority Queuing discipline. In order to realize this queuing method, the PPP_DS1 link was selected and the QoS Scheme was changed from FIFO to PQ. Figure 6 illustrates this configuration.

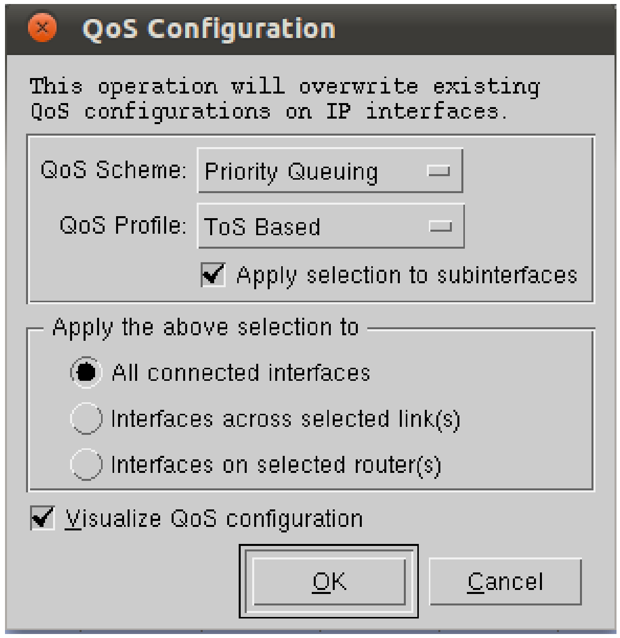


Figure 6: QoS Configuration Settings for Scenario 2.

Scenario 3 considered the Weighted Fair Queuing discipline. In order to realize this queuing method, the PPP_DS1 link was selected and the QoS Scheme was changed from FIFO to WFQ. Figure 7 illustrates this configuration.

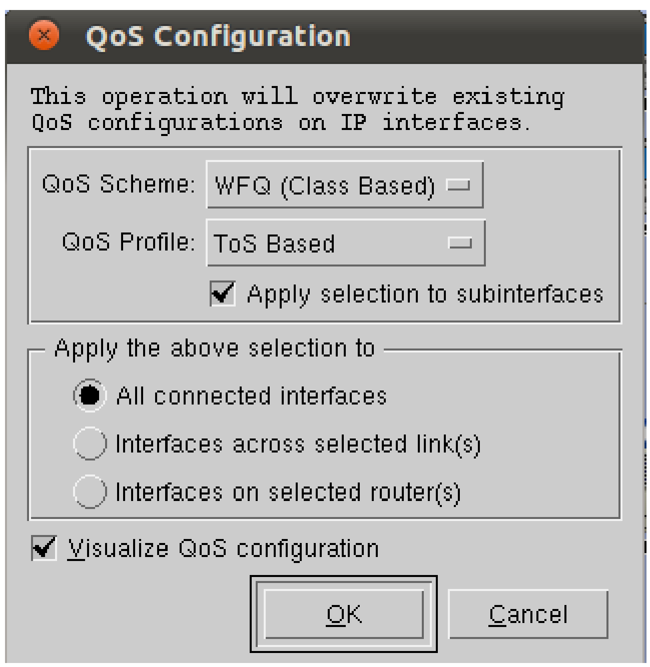


Figure 7: QoS Configuration Settings for Scenario 3.

6. Simulation Results and Discussion

The simulation results obtained for each application under the three scenarios are outlined below. The global statistics collected for FTP, voice, video conferencing and IP Packets dropped appear in sections 6.1, 6.2, 6.3, and 6.4 respectively. The object statistics collected follow.

6.1. FTP

FTP: Time Average in Traffic Sent $(bytes/s)$

Figure 8 illustrates the time average traffic in $bytes/s$ that was sent through the network by the FTP workstation. As expected for FTP, Priority Queuing had the least amount of traffic sent, while the First-in, First-out and the Weighted Fair Queuing showed a higher average amount of traffic sent.

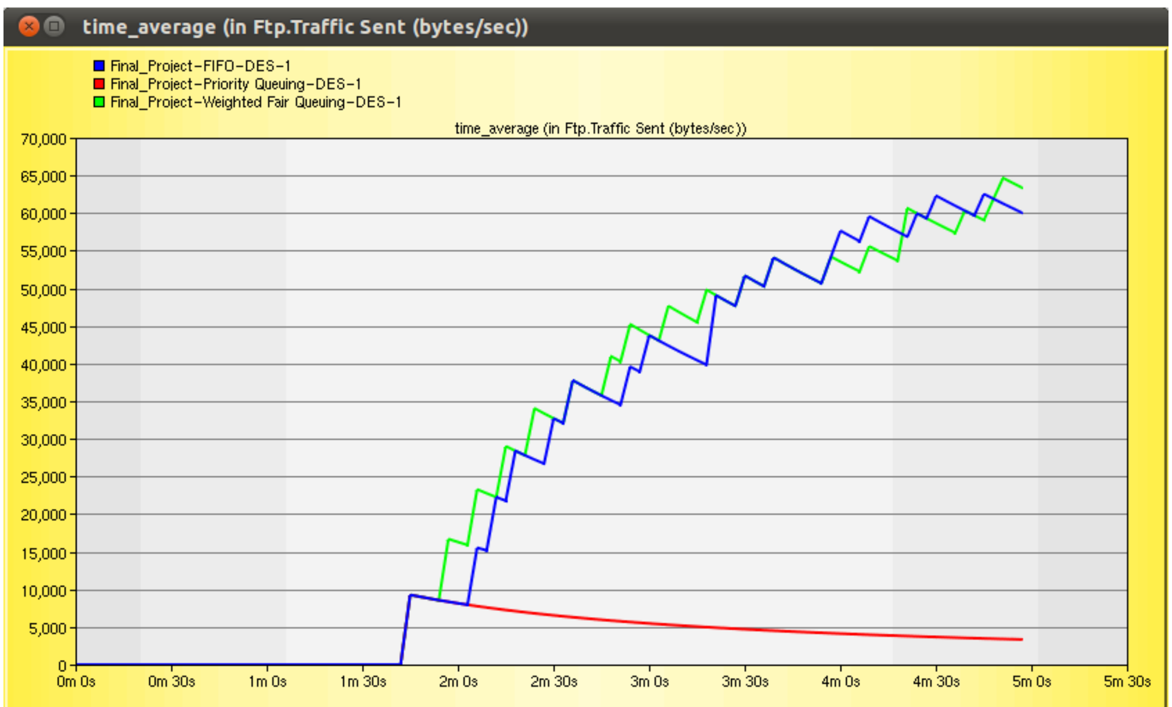


Figure 8: Time Average in FTP Traffic Sent $(bytes/s)$.

FTP: Time Average in Traffic Received $(bytes/s)$

As discussed in the background section, WFQ uses multiple queues to separate and provide equal amounts of bandwidth to each of the flows, while FIFO places all packets into one queue and then transmits those packets as bandwidth becomes available. As can be observed in Figure 9, traffic received under the FIFO scenario is initially higher than the WFQ; however over the longer run, the WFQ scenario illustrated above attains the maximum amount of traffic received. It is further concluded that the PQ scenario is the worst queuing discipline to choose for FTP, as expected, it has the least amount of traffic received.

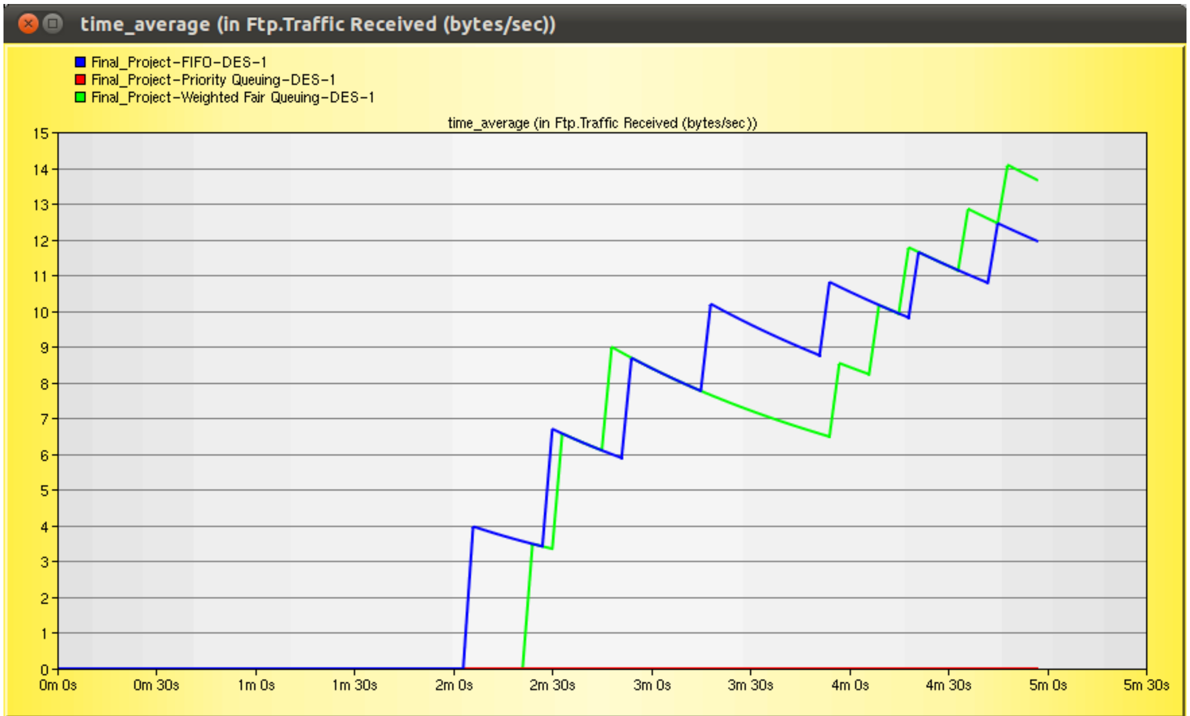


Figure 9: Time Average in FTP Traffic Received $(bytes/s)$.

6.2.Voice

Voice: Time Average in End-to-End Delay (s)

Figure 10 illustrates the time average in the End-to-End delay observed during the five minutes of simulation time for the VoIP application in each of the three scenarios.

As can be observed, the time taken for packets to be transmitted from the source to the destination, or the End-to-End delay was found average out to approximatly 2 seconds under FIFO, and approximatly 0.063 seconds under both PQ and WFQ. This result is as expected since both PQ and WFQ tag the packets with a priority or a weight, while packets in the FIFO queue are transmitted based solely on their arrival and on their position in the queue. Real time applications such as voice reaquire the ETE to be as low as possible to provide for a seamless and more natural conversation to take place, therefore it is concluded that the both the WFQ and PQ are better queuing disciplines to use for VoIP than FIFO. Table 11 summarizes the approximate End-to-End Delay observed from Figure 10.

Scenario	End-to-End Delay (s)
FIFO	2
PQ	0.063
WFQ	0.063

Table 11: Summary of the Approximate ETE Delay Observed.

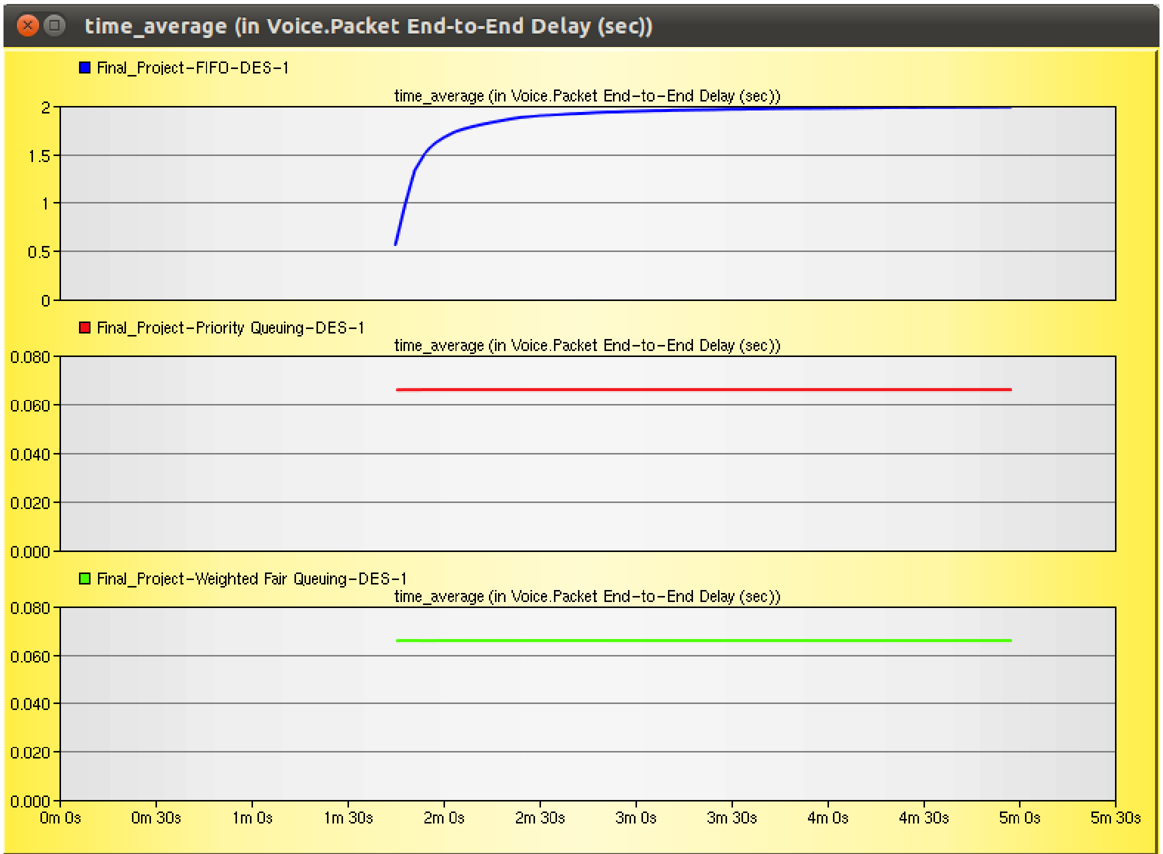


Figure 10: Time Average in End-to-End Delay for VoIP (s).

Voice: Jitter (s)

Jitter, or the variation in the ETE delay, was expected to be the highest under FIFO. As packets are placed into the three different queues studied; the ETE delay in the transmission of the packets from the source to the destination was expected to vary depending on the position of the packets in the queue.

Figure 11 illustrates the results of the Jitter obtained under the three scenarios. As expected, Priority Queuing was observed to be approximately as good as the WFQ, due to the priority and the weight factor placed upon the voice packets before transmission. FIFO displayed more noticeable Jitter due to the fact that there was no priority given to the voice packets before transmission.

Hence, it was concluded that the either the PQ or the WFQ disciplines would be a better queuing discipline to chose than FIFO in order to minimize the Jitter observed.

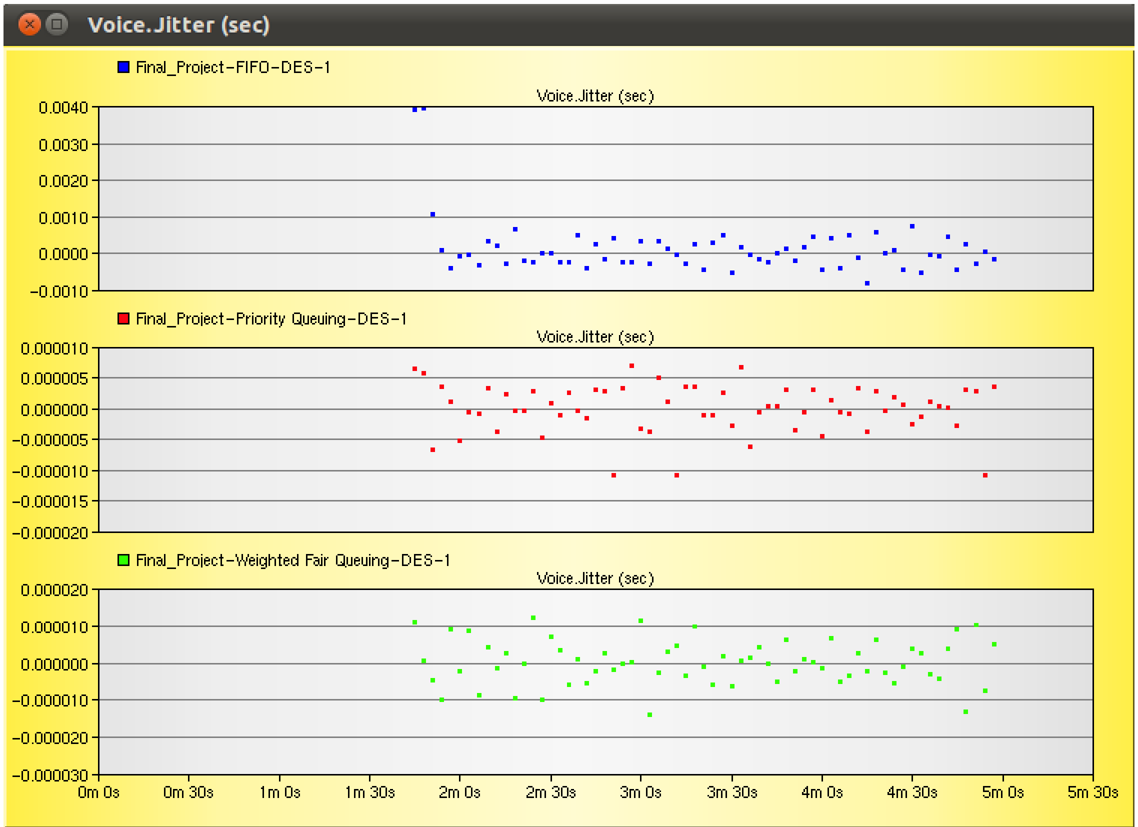


Figure 11: Voice Jitter Observed (s).

Voice: Mean Opinion Score

As expected the MOS value, which defines the preceived voice quality after transmission shows best results with Priority Queuing and with Weighted Fair Queuing. Since realtime applications require a minimum delay in packet transmission, voice packets under the PQ and WFQ are tagged/weighed more urgent than other non-voice packets and as such are transmitted prior to non-voice packets. In FIFO, all packets are treated equally and transmission of packets does not give a higher precedicnce to the voice data thus resulting in the lower MOS value and a lower preceived voice quality. Table 12 illustrates the approximate MOS values obtained in the three scenarios considered.

Scenario	Approximate MOS Value
FIFO	1
PQ	3.7
WFQ	3.6

Table 12: Summary of the Approximate MOS Values Observed.

Comparing the values obtained in above to Table 1, the following conclusions can be made about the preceived quality of the audio after transmission. Both PQ and WFQ have preceived qualities that fall between fair and good, with the degree of impairment falling between being slightly annoying and perceptible but annoying. The overall perceived quality of the voice packets sent using FIFO are concluded to be bad and the degree of impairment was found to be very annoying. Figure 12 shows the resulting graph obtained for the MOS values.

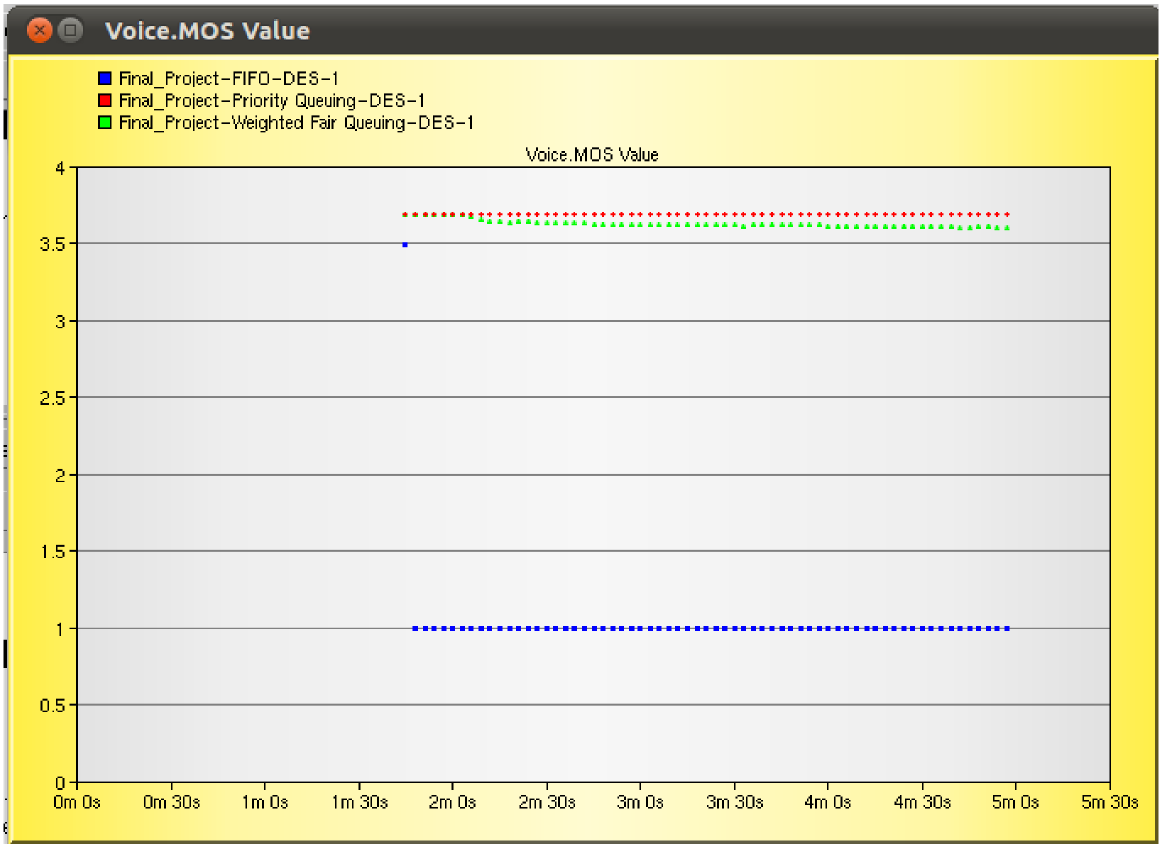


Figure 12: MOS Values Observed.

Voice: Packet Delay Variation

As discussed in the background section, Packet Delay Variation is a measure of the difference in the End-to-End delay between packets in a flow while ignoring any packets that have been lost. As can be observed in Figure 13, this statistic was found to be the highest for the FIFO queuing method. Both PQ and WFQ showed a nearly constant PDV of approximately 0.000008.

Since real-time applications require PDV to be as low as possibly, it is conclucded that either choice of PQ or WFQ are significantly better than FIFO, as applied towards multimedia applications.

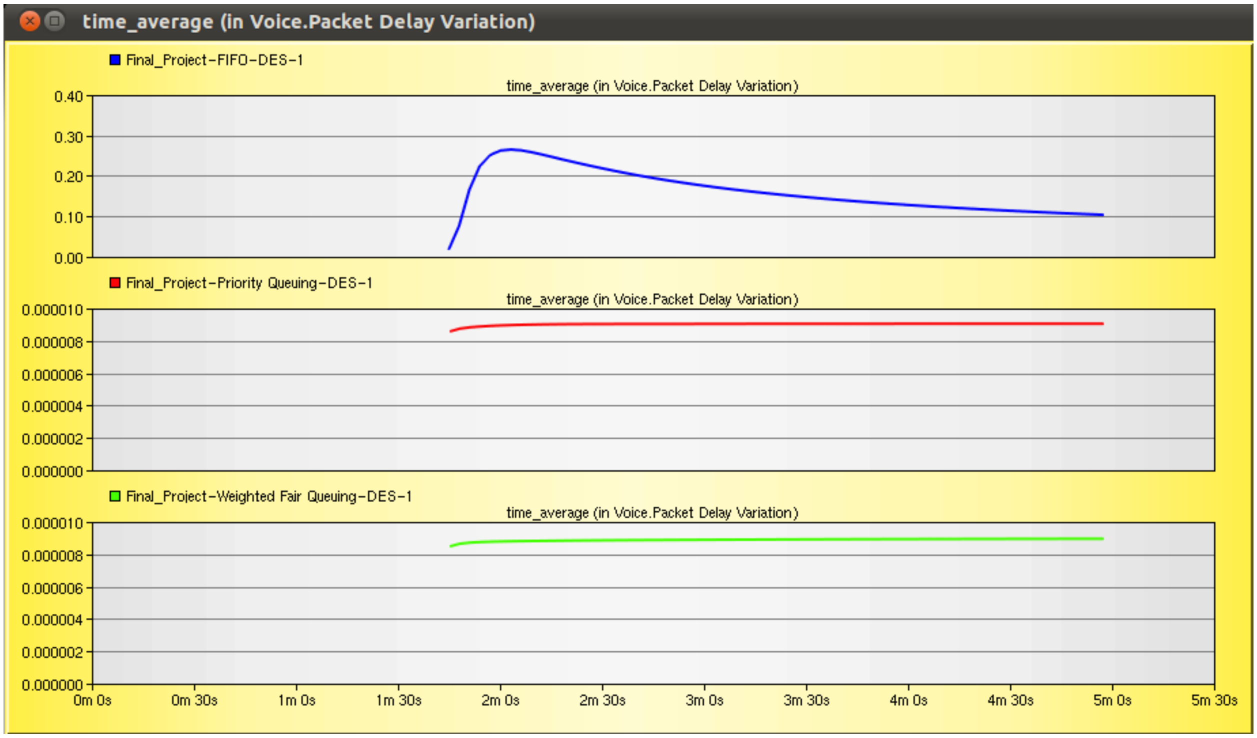


Figure 13: Time Average in Voice Packet Delay Variation.

Voice: Traffic Sent $(bytes/s)$

As expected and as can be observed in Figure 14, the time average in voice traffic that was initially sent is exactly equal for all three scenarios considered.

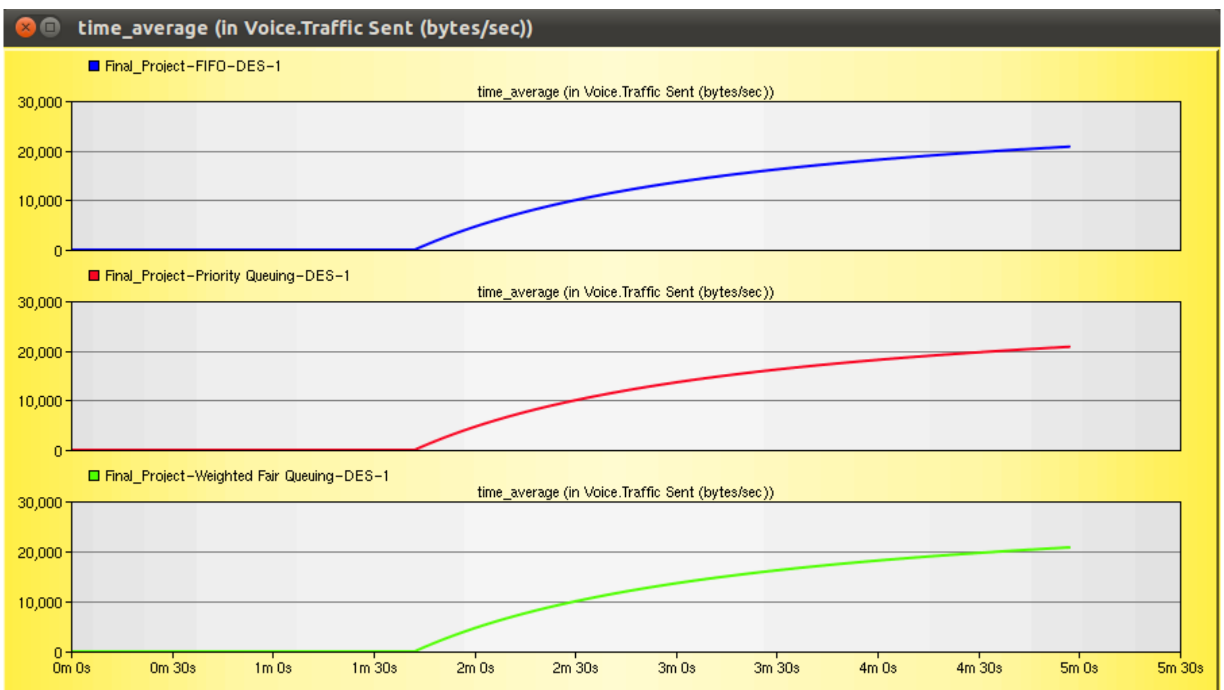


Figure 14: Time Average in Voice Traffic Sent $(bytes/s)$.

Voice: Traffic Received $(bytes/s)$

The time average in the voice traffic that was received after the packets had been sent through the three different queuing disciplines is shown in Figure 15. The results obtained show that the traffic received under FIFO was the less than both PQ and WFQ, this result was also expected and it shows that a higher volume of traffic could be obtained under either PQ or WFQ than FIFO. This is especially important for voice applications as any loss would adversely affect the overall quality of the voice signal.

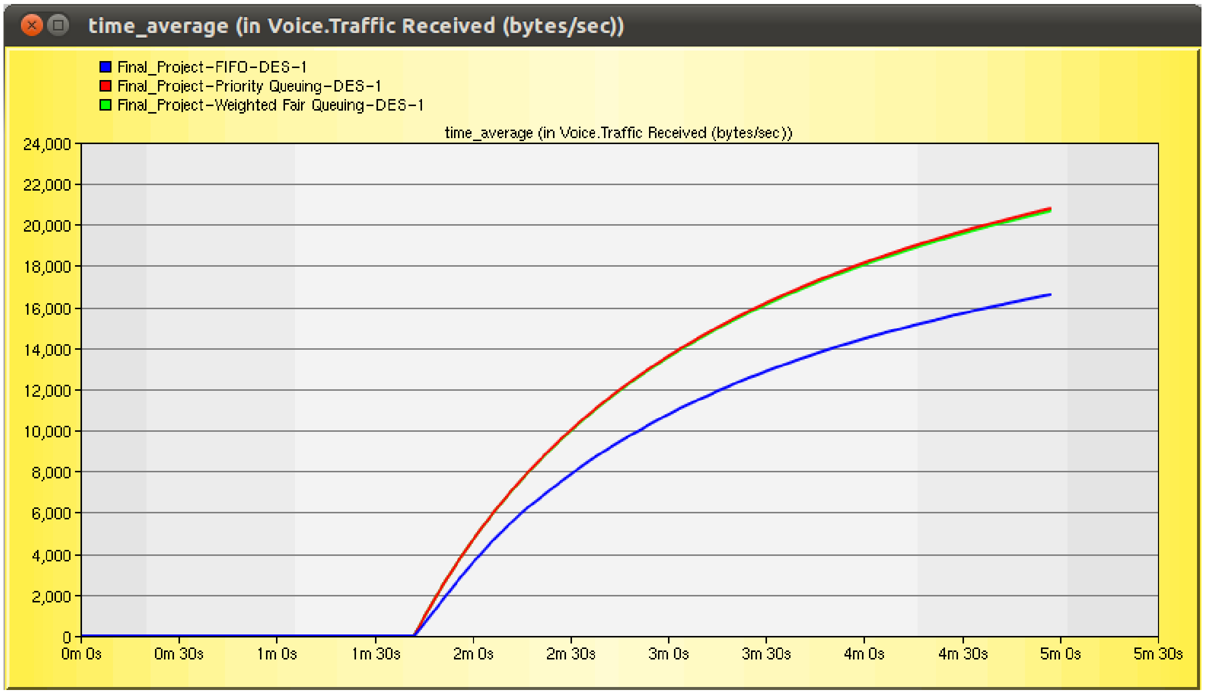


Figure 15: Time Average in Voice Traffic Received $(bytes/s)$.

6.3. Video Conferencing

Video Conferencing: End-to-End Delay(s)

The End-to-End delay observed for video packets was found to be highest for the WFQ. The results obtained can be justified as the WFQ method tries to avoid congestion and tries to maintain fairness in transmitting the packets of all the applications considered, in contrast to both FIFO and PQ.

FIFO sends packets without regard to maintaining fairness; hence it has a lower End-to-End delay than WFQ. Priority Queuing was found to be the best choice to reduce the End-to-End delay in video conferencing. This was found to be a result of the importance given to real-time multimedia packets over other application packets in the buffer such as those of the FTP. As a result, the time taken for packets to be transmitted from the source to the destination is worst for WFQ. FIFO was found to have an ETE delay between WFQ and PQ, with PQ being the best choice for reducing ETE delay observed in video conferencing. The results can be seen on Figure 16.

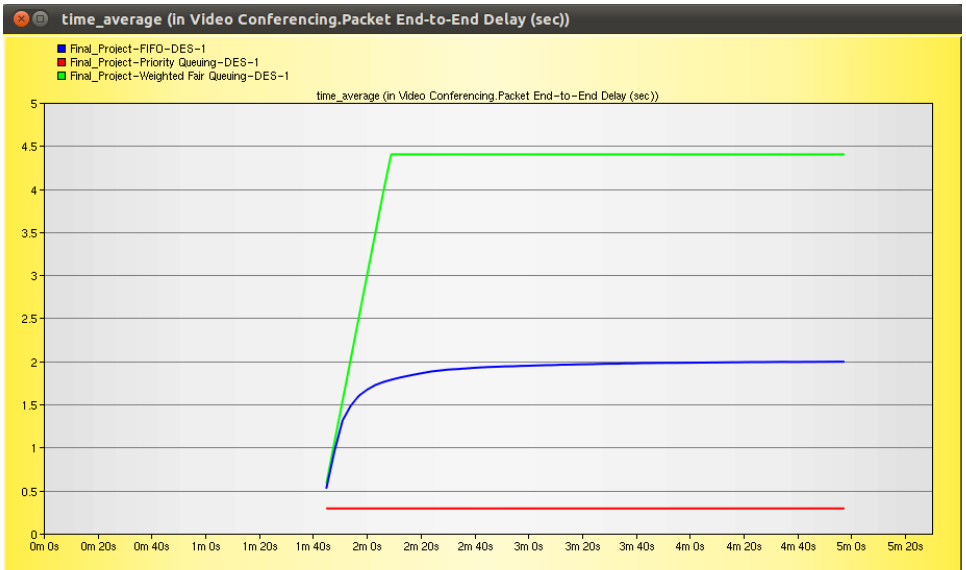


Figure 16: Time Average in End-to-End Delay (s).

Video Conferencing: Packet Delay Variation

As discussed in the background section, Packet Delay Variation is a measure of the difference in the End-to-End delay between packets in a flow while ignoring any packets that have been lost. As can be observed in Figure 17, this statistic was found to be the highest for WFQ. The results seen are directly the consequence of the End-to-End Delay observed above, and the exact same conclusions can be made here with regard to the best queuing discipline to choose. PQ is the best, while WFQ is the worst and FIFO is in between the two. It is important to note that the WFQ method should not simply be ignored because of its higher End-to-End delay. The overall packet losses and overall video quality are also important factors for video conferencing, as explained earlier those results are always worst under the FIFO queuing method. Figure 17 shows the results obtained for the PDV in video conferencing.

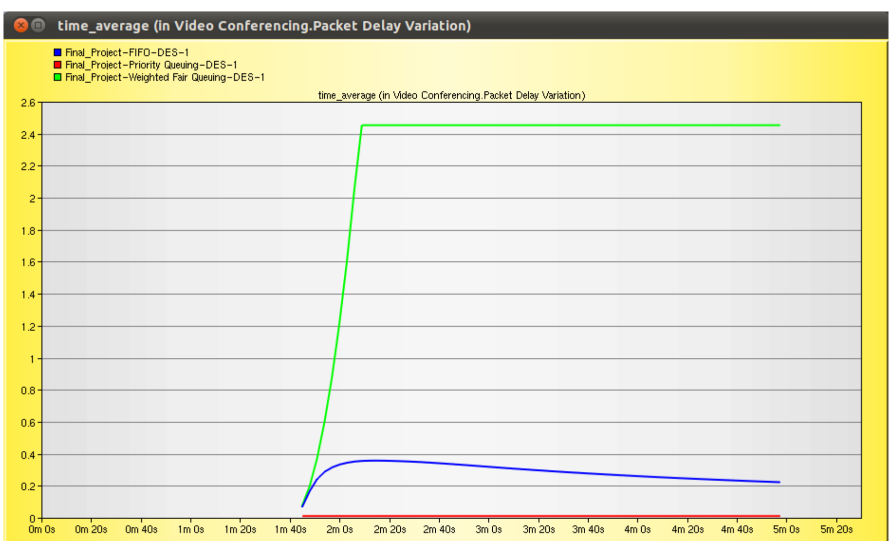


Figure 17: Time Average in Packet Delay Variation.

Video Conferencing: Traffic Sent ($bytes/s$)

The graph below is identical to the one observed for the traffic sent under the voice application and it reflects the time average in the traffic that was sent under the video conferencing application, all three queuing disciplines send the same amount of traffic into their respective queues.

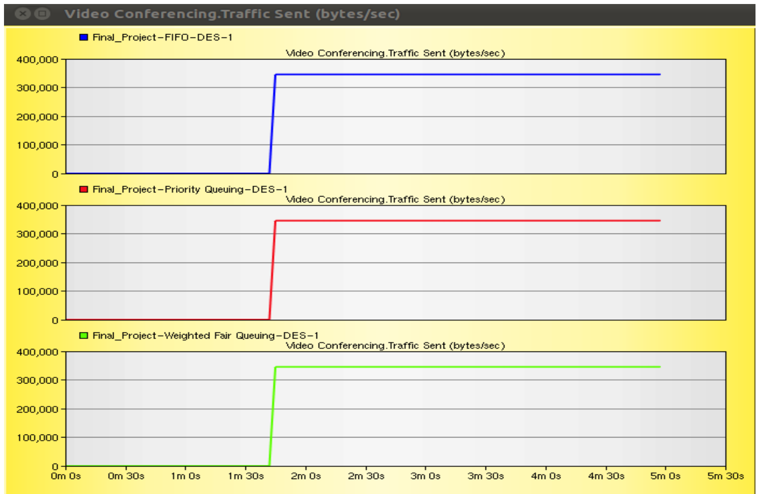


Figure 18: Video Conferencing Traffic Sent ($bytes/s$).

Video Conferencing: Traffic Received ($bytes/s$)

As can be observed in Figure 19, WFQ started to drop packets roughly about 135 seconds of network operation while reaching a maximum of about 235,000 bytes of data. PQ's drop of data was seen earlier at around 105 seconds and with a much lower traffic of about 85,000 bytes at its peak. FIFO reached a maximum of about 250,000 bytes and then started losing packets and fluctuating between 30,000 and 130,000 bytes. PQ's performance was found to be the worst, as it is only a simple variation of the FIFO queuing method. The drop in packets could be due to a number of reasons including: Insufficient queue space, Minimum number of hops exceeded by an IP datagram, absence of local router interface on non-routing nodes to be used as next hop, and failure of route table lookup on the routing nodes to yield a route to the destination [7].

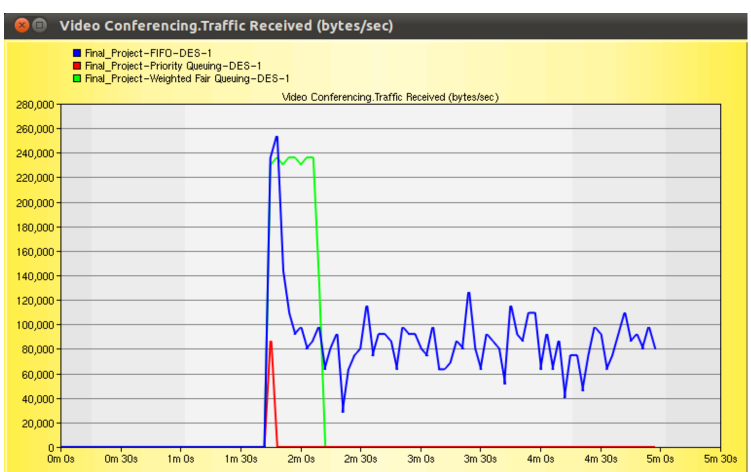


Figure 19: Video Conferencing Traffic Received ($bytes/s$).

6.4. IP Traffic Dropped $(packets/s)$

As can be observed in Figure 20, the number of IP datagrams or packets dropped per second was found to be the highest for the FIFO queuing method, while PQ and WFQ showed fewer drop in packets across all IP interfaces during the simulation. The results obtained could be explained in terms of the queue sizes of each of the queuing disciplines employed. Since FIFO has just one queue, the number of packets dropped is expected to rise as the queue becomes full. PQ and WFQ, on the other hand, employ multiple queues and so the number of packets dropped will be fewer if either of those methods is employed. This has a very important consequence on choosing the type of queue to use for real-time applications that are sensitive to packet losses. Even though the graph obtained for the End-to-End delay suggested that a FIFO queuing method would be better than the WFQ, it is clearly seen in the Figure below that choosing the FIFO discipline would result in a higher number of lost packets, thereby decreasing the overall quality of real-time applications.

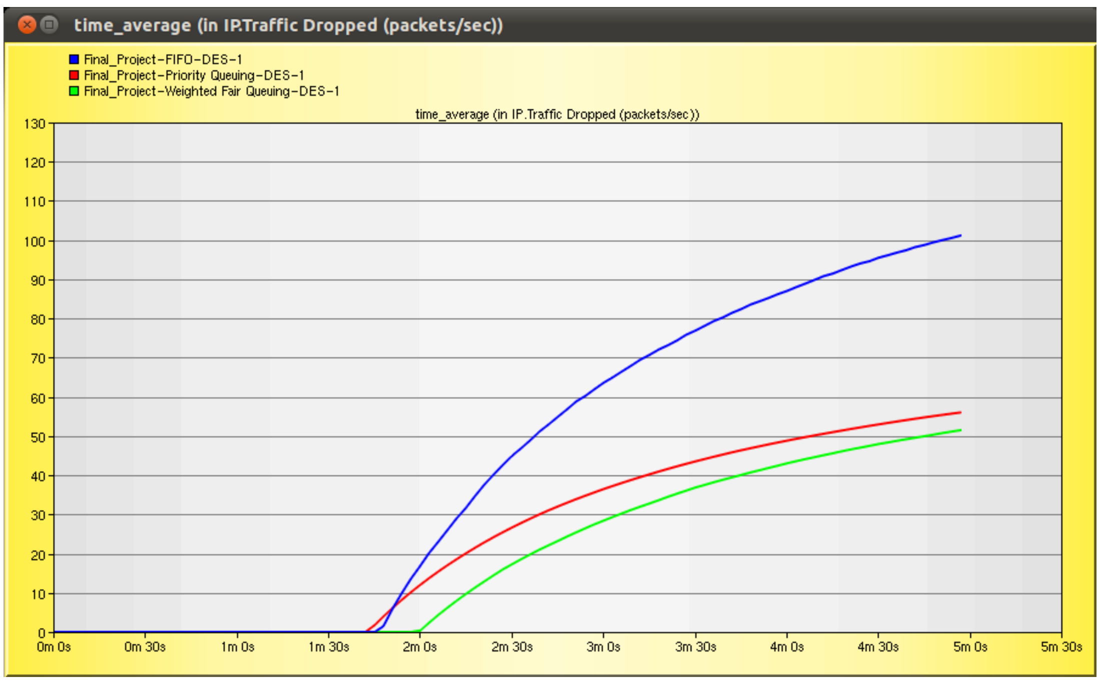


Figure 20: Time Average in IP Traffic Dropped $(packets/s)$.

6.5.Point-to-Point

Object statistics for the link connection between Router A and Router B were also recorded and analyzed, the summary of the results is provided below.

Point-to-Point: Average Queuing Delay (Router A -> Router B) (s)

The Average Queuing Delay represents the instantaneous measurement of packet waiting times in the transmitter channel's queue. The measurements of the Average Queuing Delay are taken from the moment that a packet arrives into the transmitter channel queue up to the time that the last bit of the packet is transmitted. As expected, this delay time was found to be the most for FIFO due to its single buffer, meaning that packets arriving into the channel would have to wait longer for service than either of the multi-buffered queues found in PQ or WFQ. Figure 21 illustrates the results obtained.

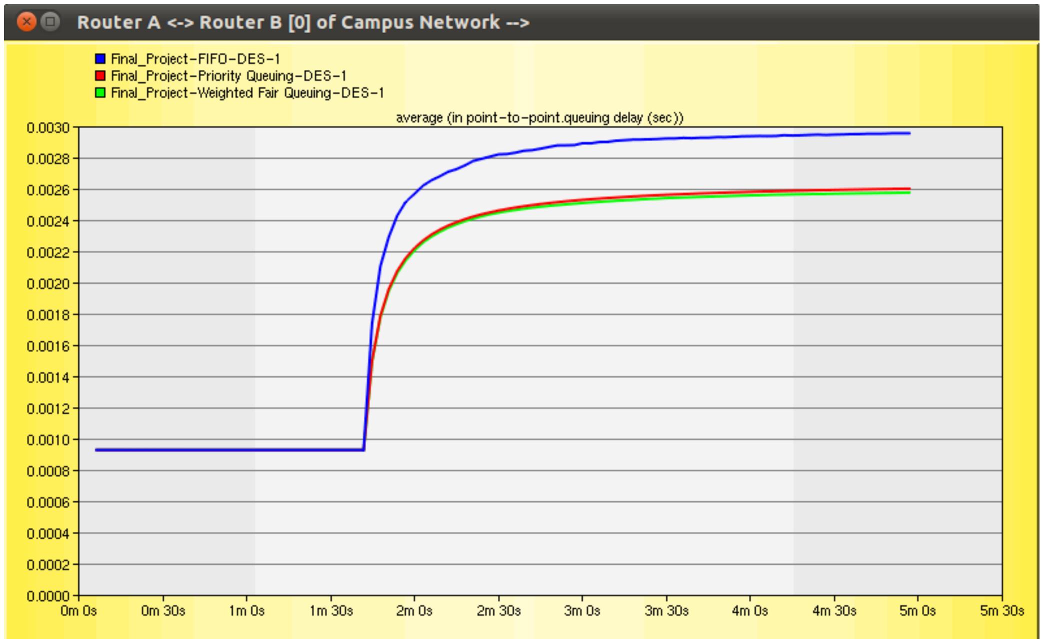


Figure 21: Average in Point-to-Point Queuing Delay (s).

Point-to-Point: Utilization (Router A ->Router B)

Figure 22 illustrates the results obtained for the Point-to-Point Utilization, or the consumption of the available channels bandwidth for the FIFO, PQ, and WFQ scenarios. As can be observed, the Point-to-Point utilization for the link between the two routers is the same across all scenarios.

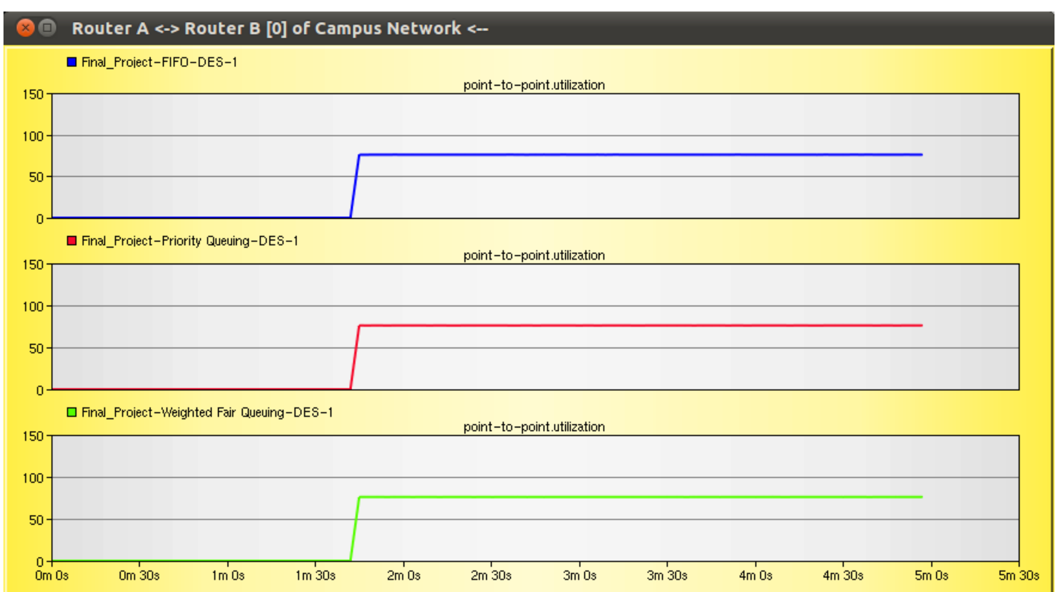


Figure 22: Point-to-Point Utilization.

Point-to-Point: Throughput (Router A -> Router B) $(packets/s)$

The throughput statistic shows the average number of packets that were successfully received and transmitted by the receiver and the transmitter channels per second. As can be seen in Figure 23, the best Throughput was found for the WFQ scenario, followed closely by the PQ scenario. As expected, Throughput was found to be the worst for FIFO.

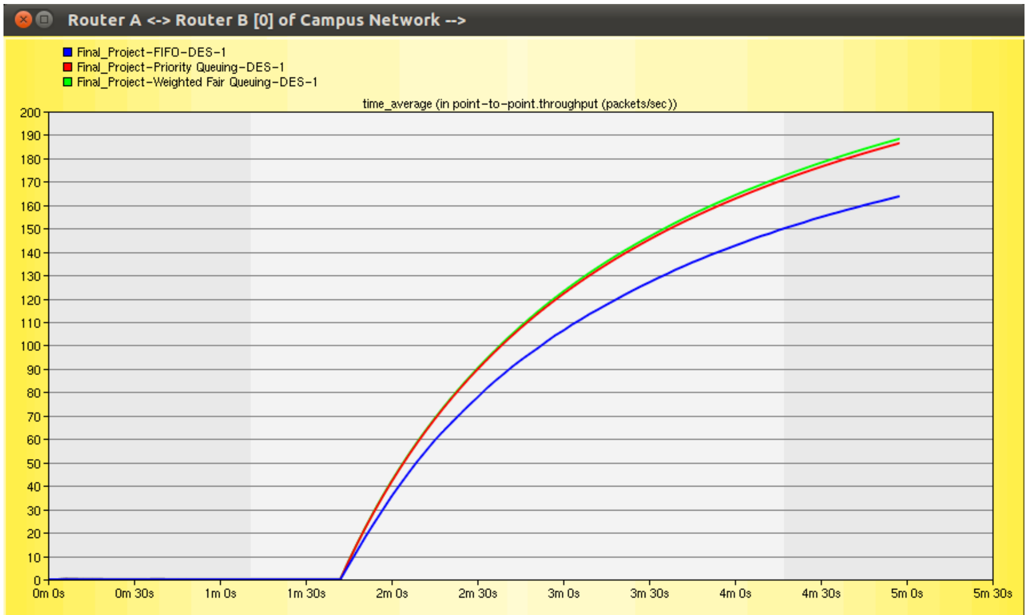


Figure 23: Time Average in Point-to-Point Throughput $(packets/s)$.

7. Conclusion

7.1. Summary of Results

Table 13 provides a ranking system that is used in order to differentiate and to select the best queuing discipline for each type of application studied in this project. The conclusions made are based on the results of the graphs obtained and their corresponding explanations given throughout this report.

Ranking System	Equivalent Numerical Value
Good	1
Worse	2
Worst	3

Table 13: Ranking System Employed to Classify Results Obtained.

Table 14 summarizes the results obtained and explained above by ranking the queuing methods employed against the global statistics collected. Each statistic collected under each scenario is given one of the numerical values presented in table 13 and the best queuing method to employ is then presented in the last column of Table 13. In cases where two scenarios are equally good, the numbering system is reduced to either a 1 or a 2. In cases where all scenarios are equally good, the numbering system reduces to just 1.

Application	Statistics Collected	Scenario 1: FIFO	Scenario 2: PQ	Scenario 3: WFQ	Best Queuing Discipline
FTP	Traffic Sent	2	3	1	WFQ
	Traffic Received	2	3	1	WFQ
Voice	End-to-End Delay	2	1	1	WFQ or PQ
	Jitter	3	1	2	PQ
	Mean Opinion Score	3	1	2	PQ
	Packet Delay Variation	2	1	1	WFQ or PQ
	Traffic Sent	1	1	1	FIFO, WFQ, or PQ
	Traffic Received	3	1	2	PQ
Video Conferencing	End-to-End Delay	2	1	3	PQ
	Packet Delay Variation	2	1	3	PQ
	Traffic Sent	1	1	1	FIFO, WFQ, or PQ
	Traffic Received	2	3	1	WFQ

Table 14: Summary and Comparisons Made Between the Three Queuing Disciplines Studied.

Before any conclusions can be made from Table 14, the statistics obtained for the IP Packet Drops and the Point-to-Point linked connection needs to be summarized as well. Table 15 uses the ranking scheme provided earlier to summarize the results.

Statistics Collected	Scenario 1: FIFO	Scenario 2: PQ	Scenario 3: WFQ	Best Queuing Discipline
IP Packets Dropped	3	2	1	WFQ
Average Queuing Delay	2	1	1	WFQ or PQ
Utilization	1	1	1	FIFO, WFQ, or PQ
Throughput	3	2	1	WFQ

Table 15: Comparison of Statistics for the Point-to-Point and IP Packets Dropped in Each Scenario.

Based on the results presented, FTP applications were found to be best supported through a WFQ discipline, even-though the results obtained under FIFO were acceptable for FTP; the statistics collected in Table 15 suggest that FIFO has the worst Throughput, and the highest number of IP Packets Dropped.

Due to their real-time nature, voice applications were found to be best supported through PQ, thus minimizing jitter, allowing more traffic to be received, and providing a slightly better MOS value than WFQ. Even-though the PQ shows slightly more IP Packets Dropped, this difference seems to be minimized as the simulation time increases.

Video conferencing was found to perform the worst under PQ, and this was perhaps the most surprising result that was obtained. Even-though PQ decreased the End-to-End Delay and the PDV, the amount of traffic that was received was found to be inacceptable under PQ. It was therefore concluded that WFQ is the best choice for video conferencing, under the three queuing disciplines studied.

7.2.Future Work

This project studied the effects of three queuing disciplines on three different applications, in order to conclude the best discipline to use for each application considered. Future work should investigate other queuing methods available such as DWRR, Custom Queuing, SPQ, and SFQ. The effects of Random-Early Drop (RED) and drop-tail policy should also be considered [8]. Other applications such as online gaming should also be considered for a more complete report. Furthermore, the study should consider different qualities (resolution, frames, speech quality) in the real-time applications sent in order to better justify the type of queue chosen.

8. References

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