Quest Journals
Journal of Software Engineering and Simulation
Volume 8 ~ Issue 6 (2022) pp: 07-15
ISSN(Online) :2321-3795 ISSN (Print):2321-3809
www.questjournals.org

Research Paper

Assessment of the Impact of Different Queuing Techniques on Different Network Traffic Services.

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Abstract: *A few years ago, the internet evolved into a communication environment used not only for human and social contact but also for commercial and educational purposes in colleges and schools. As a result, a tremendous amount of new multimedia content has been created, such as interactive environments, 3D videos network gaming, virtual worlds, and other programs that demand a higher bandwidth to function properly. Multimedia applications via IP networks are becoming increasingly popular. The bandwidth consumed has become a major issue for Internet service providers (ISPs) and online communities. Since bandwidth is a limited network resource, numerous types of traffic are employed across the network, including video conferencing, VoIP, and file transfer. As a result, routers employ various traffic management techniques known as queuing approaches to administer these services by managing how packets are delayed while waiting to be dispatched. This article employs three distinct queuing systems, FIFO, PQ, and WFQ, and we attempted to set up three networks with varying traffic volumes and designed network topologies using OPNET tools. These simulations illustrate that the WFQ algorithm performs better under various traffic loads*
Keywords: *Healthy dietary habits, cardiovascular disease, diabetes, obesity*

R*eceived 28 May, 2022;* ***R****evised 05 June, 2022;* ***A****ccepted 07 June, 2022 © The author(s) 2022.*
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I. Introduction

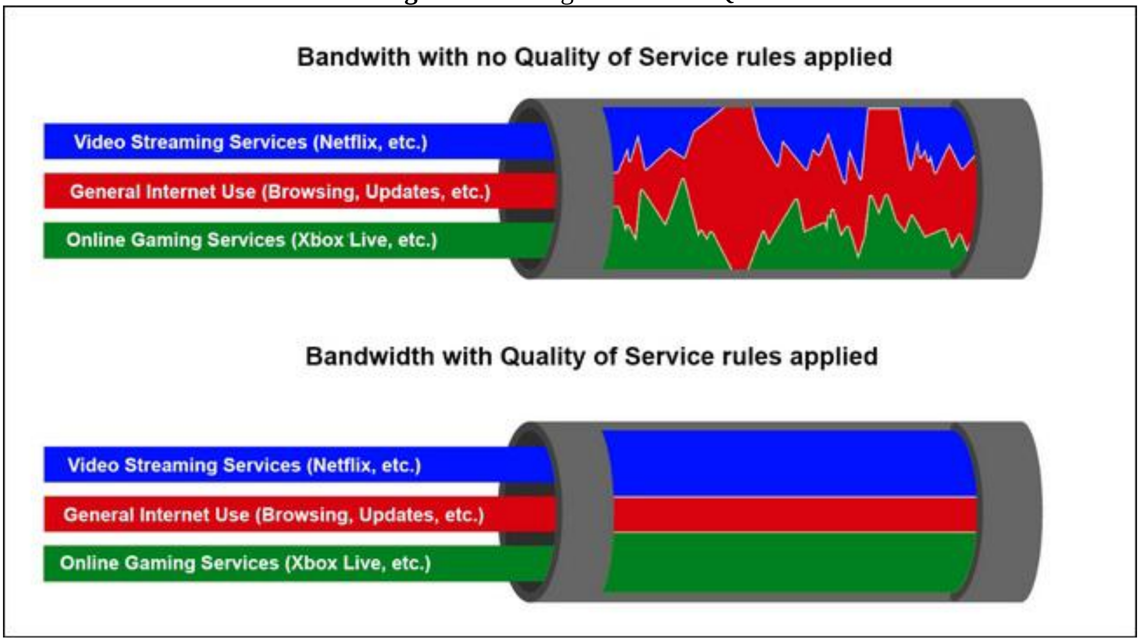
At that moment, the internet's rapid growth is posing the most serious data transmission and reception issues throughout the world's diverse networks. The capacity of a network to deliver high-quality service and accept high-quality consumers is referred to as service quality (QoS). The term "quality of service" refers to a network's ability to serve various forms of network traffic, including video conferencing, VoIP, FTP, and HTTP. The capacity to ensure the delivery of large data volumes is referred to as Quality of Service (QoS). The critical nature of Qos is demonstrated in Assurance when network capacity is insufficient, particularly for multimedia applications that frequently require a fixed bit rate and are not permitted to delay. In contrast, other applications such as FTP, HTTP, and E-Mail are less sensitive to delay and can be delayed for a few seconds without causing users any inconvenience. Distortion. In interface with multimedia apps, a delay of less than a second is acceptable **(Gozdecki,2003).** Routers use queue techniques to manage how queued packages are sent depending on the resource allocation algorithm they implement. One of the strategies that give QoS in the organizations is a lining discipline that has straightforwardly impacted information streams, as per the need that gives by what lining is utilized by hubs to give a proper portion of assets, for example, interface data transmission and computer processor process for different associations in a reasonable way. Furthermore, the scheduler, for example, while administering packet flows, follows the policy of queuing orders(**Kurose,2005**). WFQ, PQ, FIFO, and other scheduling algorithms achieve router behaviors and queue outputs. The algorithm's FIFO (First In **-** First Out) approach is as follows. Packets are queued in the case of congestion, and when the overload is eliminated or reduced, the packets are output in the order in which they were received (first come & first out)(**Morgan,1991**). All packet switching devices utilize this approach as default queuing. It is easy to set up and does not require any configuration, but it has a major flaw: it is hard to handle packets from various flows differently. FIFO queues are required for network device operation; however, they do not support (QoS) for different services. The priority service (PQ) enables the administrator network to categorize network traffic. Priority is assigned on a four-point scale: high, medium, normal, and low. Priority is tightly enforced in the router queues. Priority packets are processed first in a queue until no more queues exist, at which point medium priority packets are processed. Packets from the lowest priority queue are processed only when the queue with the next greatest priority becomes available. Is empty. Low priority traffic can be fully stopped under certain conditions, resulting in packet loss(**Kurose & Ross**,**2010**). PQ is typically used when critical applications cause delays. Weighted fair queuing (WFQ) is a technique that can ensure a minimum bandwidth by weighing (sharing) queues in a network with an overload (the number of which is specified in the Weight field). The proposal's primary objective was to ensure that all traffic types were treated equally and that burst data did not utilize more bandwidth than was allocated. The WFQ algorithm is engaged only when the capacity of a port is exceeded. More bandwidth is guaranteed for queues with greater weight than for queues with a lower weight. This queuing technique effectively distributes any available bandwidth through several traffic queues. Although the guaranteed delay boundaries offered by WFQ may be higher than those of other queue disciplines, the primary goal of WFQ is to protect each service class by ensuring a fixed level of output port bandwidth independent of the performance of other services.

At Primary purpose of this study is to conduct a detailed examination of various services by determining and comparing certain metrics such as Traffic Received and Traffic Dropped when three separate scheduling disciplines are used with varying network loads. Delay variation and end-to-end packet delay. The obtained data indicate that the WFQ technique produces the highest-quality networks among the three investigated in this experiment.

II. Quality Of Service

That Quality of Service (Q.o.S) refers to technologies that require the control of data traffic (Bandwidth), and the criteria to consider are Packet Loss, Jitter, and Latency**(Zheng & Gu 2013**; **Pour & Trajkovic,2011).** IP QoS refers to the performance of IP packets flowing over one or more networks in IP-based networks. It is intended to assist end-users in being more productive by ensuring that network-based applications perform as expected. QoS refers to a network's ability to deliver better service to certain network traffic using various technologies.

Figure.1 showing the effect of QoS



The efficiency of a computer network can vary due to various issues, such as bandwidth, latency, and jitter, all of which can significantly impact numerous applications. Voice services (such as VoIP or IP Telephony) and streaming video, for example, might irritate users when application data packets are sent across a network with insufficient bandwidth, unpredictable latency, or severe jitter(**Mohammed et al., 2013; Koyuncu & Hayder,2015)** The enterprise networks used sensitive applications such as Voice or Video where Latency is very annoying. So, it is essential to give precedence to the more critical and sensitive traffic, which is the quality of service delivery. QoS helps prevent some packets from delaying or slowing down**.** The basic elements that affect QoS are described briefly in the following sections (QoS parameters):

- Delay (Latency) : Overall latency is the temporal delay experienced by a packet during its transport from one location to another. Different network delays include serialization delay, jitter buffer, and delay network are all types of delay processing. The average packet end-to-end delay is calculated using equation (1) where Packet Arrival_i is the time when packet “i” reaches the destination and Packet Start_i is the time when packet “i" leaves the source. “n” is the total number of packets**(Tawfeeq, 2009; Ray, S., & Panigrahi,2013).**

$$Average\ DeLay = \frac{\sum_{i=1}(packetArrival_i)-(packetStart_i)}{n} \quad (1)$$

- Jitter is a concept that refers to a variety of packet arrival time delays or variations. Jitter can be caused by various factors, including abrupt increases in traffic, which restrict bandwidth and induce queues. Furthermore, jitter can be caused by the pace at which each node receives and sends packets, the equation NO. (2) shows the average of calculation **(Tawfeeq, 2009).**

$$Average\ Jitter = \frac{\sum \mathrm{I}(packetArrival_{i+1}-packetStart_{i+1})-(packetArrival_i-packetStart_i)}{n-1} \quad (2)$$

- Packet loss :The ratio of total packets lost to total packets delivered between the source and destination is known as packet loss. The queues on each node exceed the capacity buffer, which is one reason for packet loss. Some reasons for packet loss include network congestion caused by excessive queues or nodes functioning above their capacity buffer. In some situations, network control ensures that the amount of traffic flowing across the network is within the network's bandwidth. If the traffic flowing through the network exceeds the available bandwidth, policing will regulate the excess traffic. As demonstrated in the equation, it shows the percentage of transferred packets lost**(Tawfeeq, 2009 ; Morgan,1991)**.

$$packetLoss = \frac{\sum LostpacketSize_i}{\sum packetSize_j} \times 100 \quad (3)$$

III. Setup For Simulation

The structure of the network employed in this work is shown in figure 2. As we can see, A bi-directional PPP DS1 link connects Router R1 to Router R2, while The FTP server and other workstations are connected to the routers through 10 BaseT lines that run across switches. The definitions of Profile, Application, and QoS Attributes, as shown below, have been changed to support the network design:

Table .1.The Application and QoS Attribute definitions.

	Load	Packet size	Type of Service(TOS)	Inter-request time(s)
Delay Sensitive Applications	Video Conference	High-resolution video (128*240). The Frame size is varied at Layer-4 by Compression technique	Streaming multimedia(4)	Start time: constant at 140sec duration: end the simulation
	VoIP	6000 bytes with G.711 Encoder Scheme	PCM quality speech to Voice	Start time: constant at 140sec duration: end the simulation
Delay Insensitive Applications	FTP	High load (50000) bytes size of file)	Interactive Voice (6)	Start time: constant at 140sec duration: end the simulation

This research study evaluates network performance; three network topologies were created, as shown in figures (2,3, and 4).

Figure.2. (A) Low-traffic network architecture

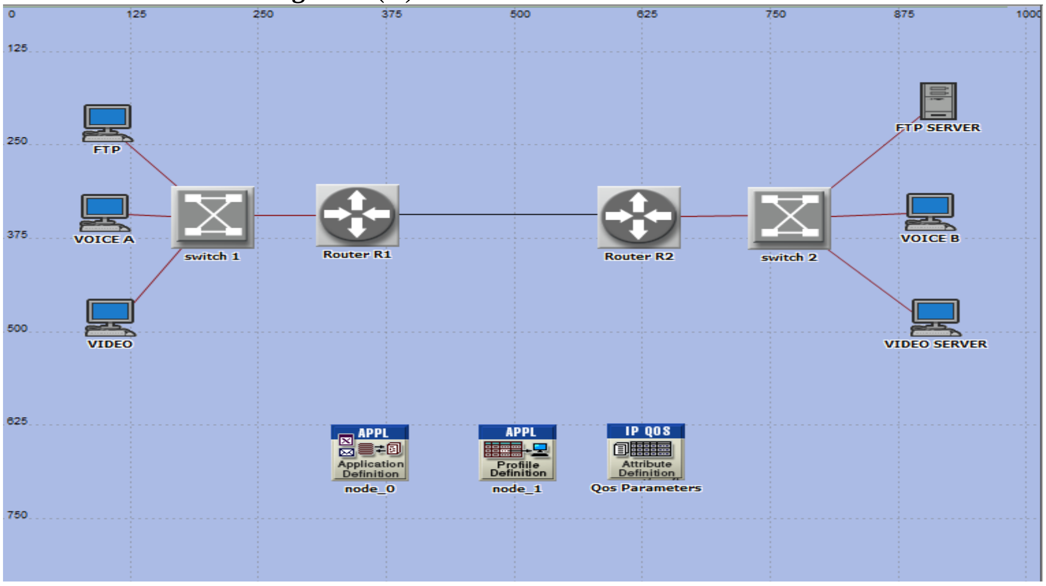


Figure.3. (B) Medium-traffic network architecture

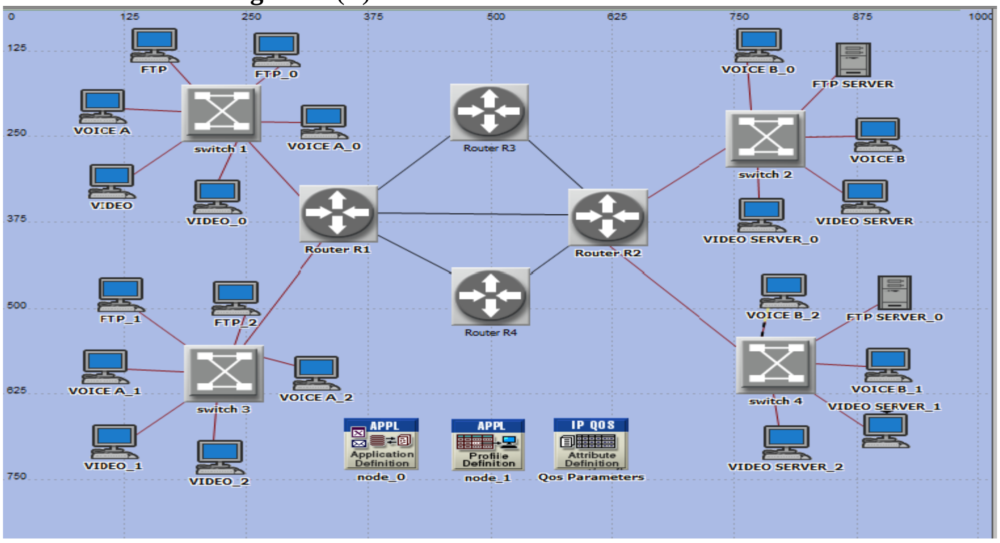
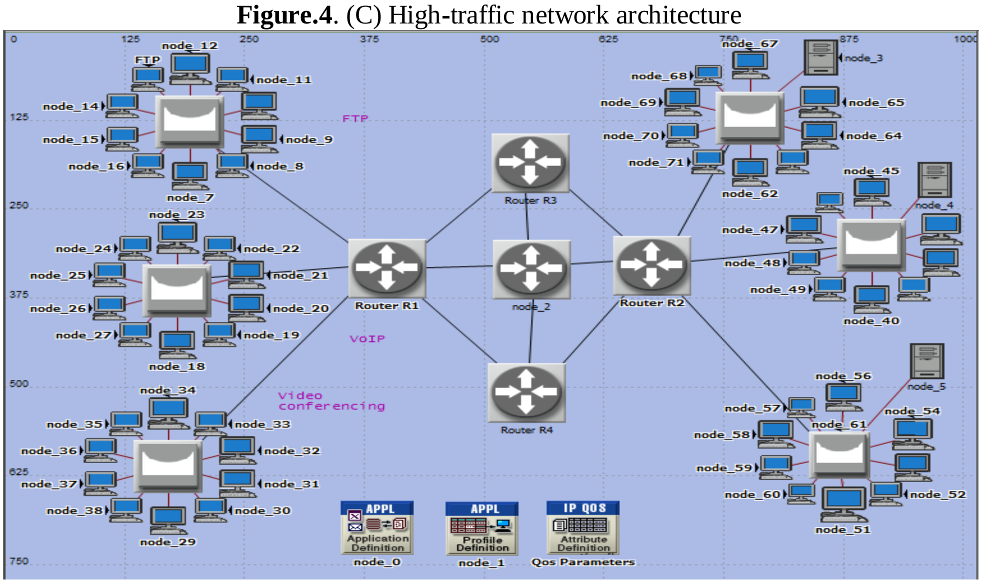


Figure.4. (C) High-traffic network architecture



Each network has a distinct number of switches and routers that have been tested with similar types of traffic to arrive at the outcome. FTP traffic is generated exponentially; three services are available: random distribution of packet arrivals, fixed packet size, and best effort. Video traffic was encoded using high-quality video at 15 frames per second with a resolution of 128x240 using the G.711 encoder. The kind of service was Streaming Multimedia for video traffic. In order to produce low, medium, and heavy traffic levels in OPNET Modeler, three different simulation methodologies were used. (FIFO, PQ, and WFQ) being employed.

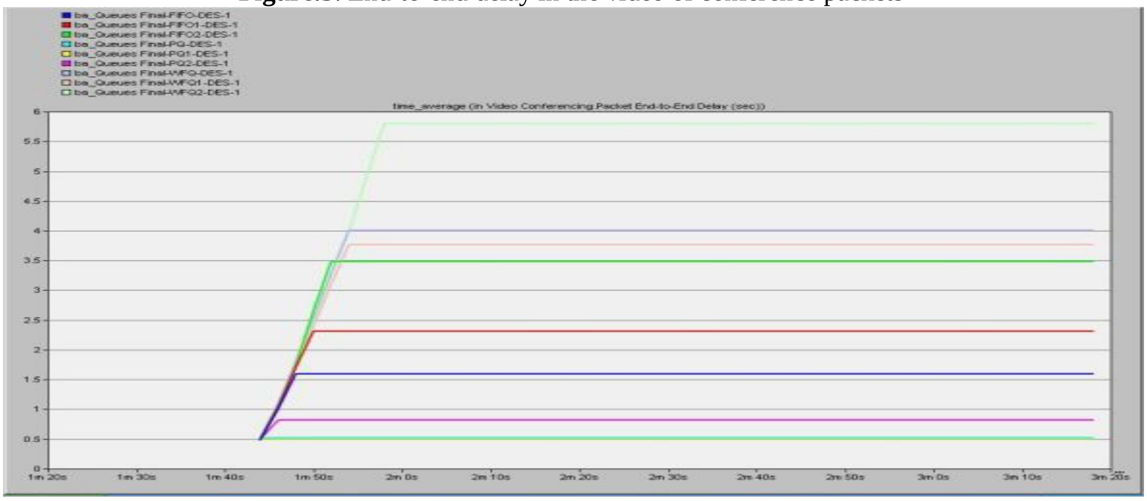
IV. Results of The Simulation

To evaluate the performance of the applications used in the networks, we collected some statistics as follows in the table below:

Video conference	***Ftp***	***Voice***	***IP***
Packet End-to-End Delay(sec)	Traffic Received(packets /sec)	Packet delay variation (PDV)	Traffic dropped
Packet Delay Variation (PDV)		Traffic Received(bytes/sec)	
		Packet End-to-End Delay(sec)	

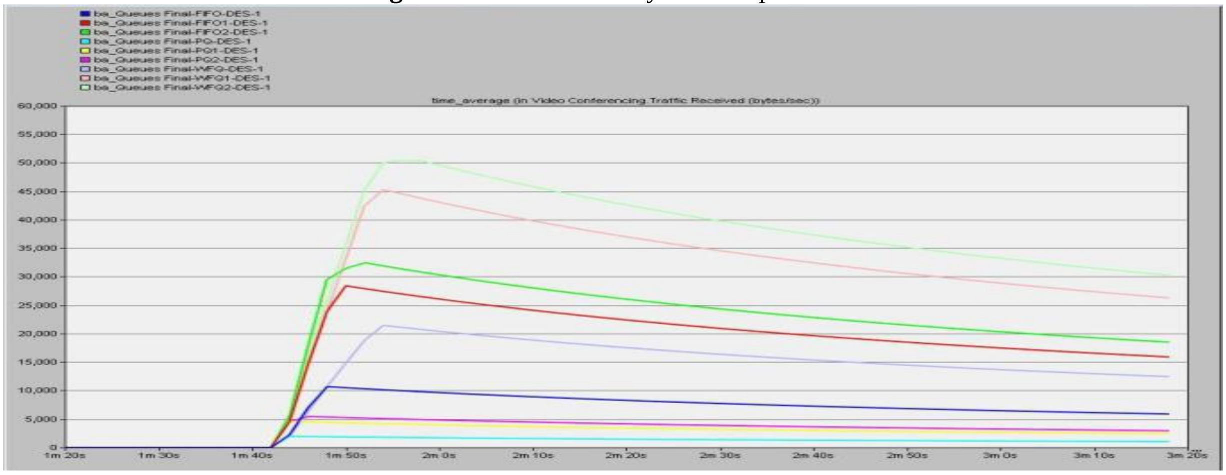
- Conference Video Packed from end to end delay:The end-to-end average time latency of video conferencing is shown in Figure 5. This diagram shows that PQ is always around 0 in all cases, even when increasing traffic. Video conferences are given higher priority, and VoIP packets are queued, separating PQ traffic from FTP traffic. As a result, as seen in Figure 5, the FTP program suffers. FIFO is maintained to rise with the greatest priority; however, in the WFQ algorithm, the delay increases with a high-resolution video utilized in a video conference. In this situation, the resulting results are consistent with those reported in[9].

Figure.5. End-to-end delay in the video of conference packets



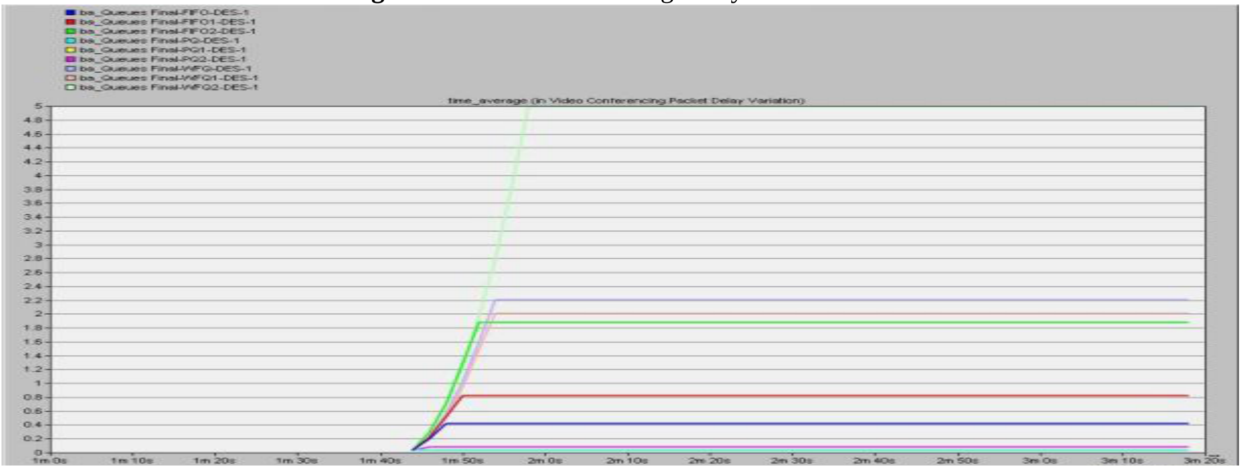
- End to End Delay VoIP Packed :Figure 6 illustrates the end-to-end Latency of VoIP packets as a function of the queuing strategy, with FIFO always being slower than the others. PQ and WFQ are always equal to zero. The size of packet end-to-end delay for FIFO is proportional to network traffic growth. The FIFO method distributes data packets via a single queue in the memory buffer. With data packets passing first based on their queue position. As a result, it has a longer delay. PQ and WFQ have more queues and a lower wait time than FIFO. The results in this graph are consistent with those in [9] and [10].

Figure.6. End-to-end delay of voice packets



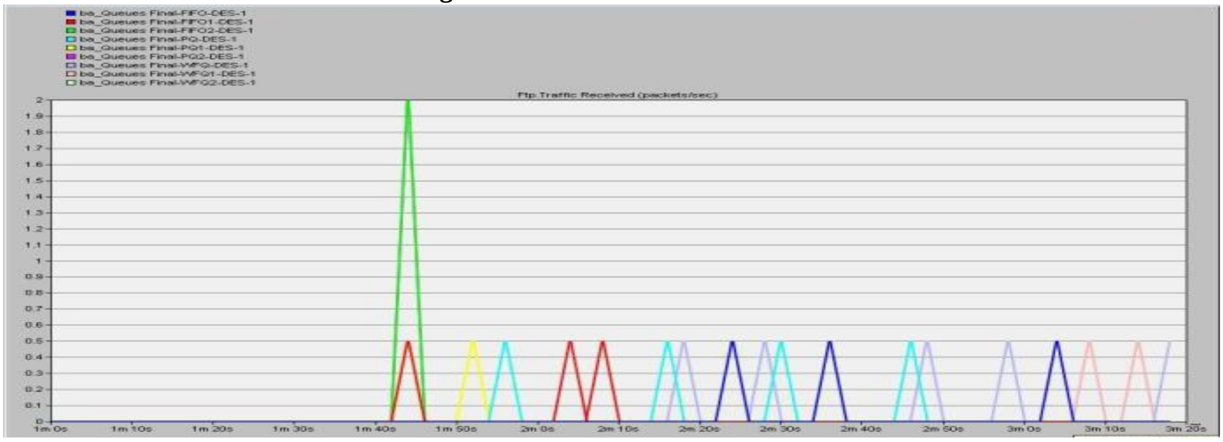
- Packet Delay Variation for Video Conferencing : Video conferencing has an average variation in the packet delay time seen in Figure 7. It depicts the packet delay variation as a function of rising traffic; therefore, WFQ and FIFO are raised as traffic increases. For all sorts of traffic, PQ is always zero. PQ implementation employed service classes with a different priority that enabled the high resolution of a video conference to preserve the quality of the video to be received. In contrast, video conferences were prioritized to help traffic flow with less jitter.

Figure.7. Video Conferencing Delay Variation



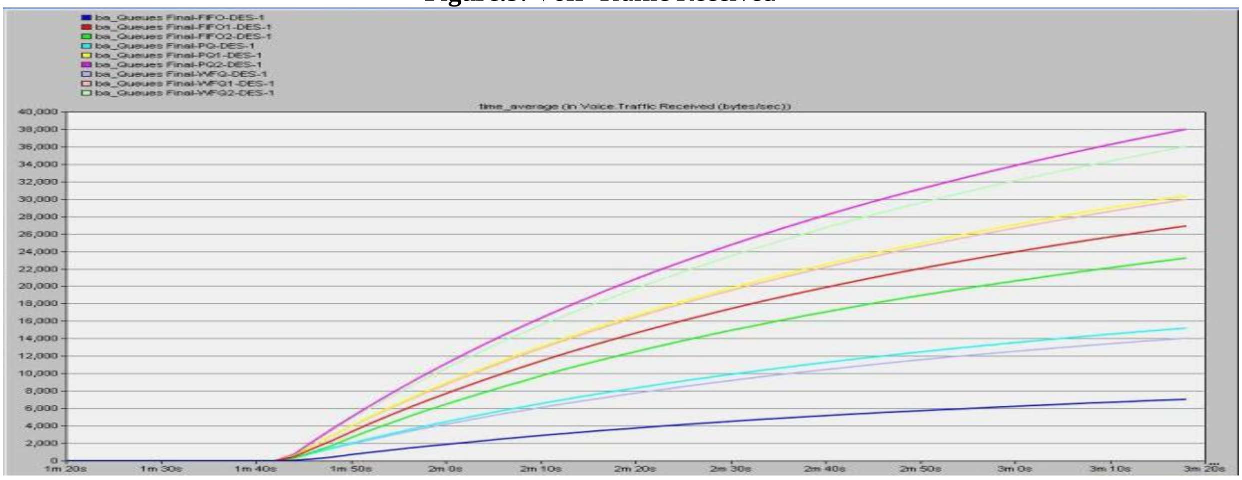
- Traffic Received For FTP **:** Figure 8 illustrates the FTP traffic received. It can be seen that the traffic received is on average 0.5 packets per second in all scenarios, even when the traffic is doubled. FIFO received 1.6 packets/sec in a medium traffic scenario, and it grew with high traffic. The FIFO method queues all received packets and transmits them simultaneously if the bandwidth is available. Therefore, it has a higher value than other queues in FTP applications.

Figure.8. FTP Received of Traffic



- Received Traffic for VoIP: Figure 9 illustrates typical traffic received in the voice application. VoIP has greater priority in FIFO. The PQ and WFQ have higher values in traffic received than FIFO. PQ receives more traffic, whereas FIFO receives the least traffic in all circumstances in all networks. PQ prioritizes VoIP above all other applications in the network, resulting in larger traffic received. In contrast, FIFO transmits packets straight to destinations without regard for application priorities, resulting in the least traffic. WFQ receives less traffic than PQ since it distributes a weight (bandwidth) to each type of application concurrently. Here, we must emphasize the significance of avoiding loss in VoIP applications and receiving all packets because any packet loss in the network will affect the quality of conversation.

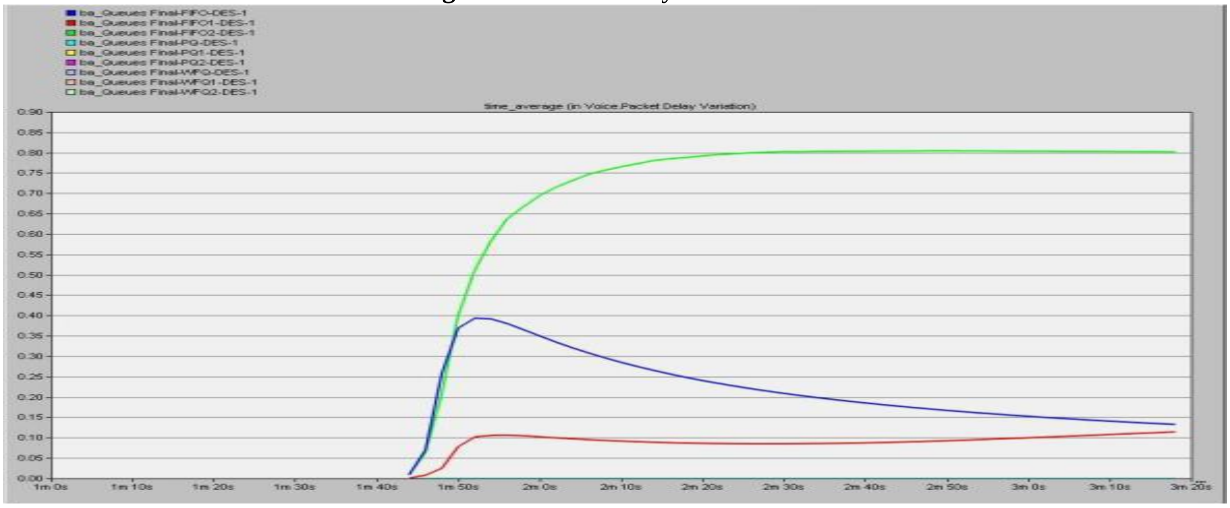
Figure.9. VoIP Traffic Received



The results for received VoIP traffic are consistent with those in [9] and [10], which evaluate the performance of queuing techniques. (WFQ, PQ, and FIFO).

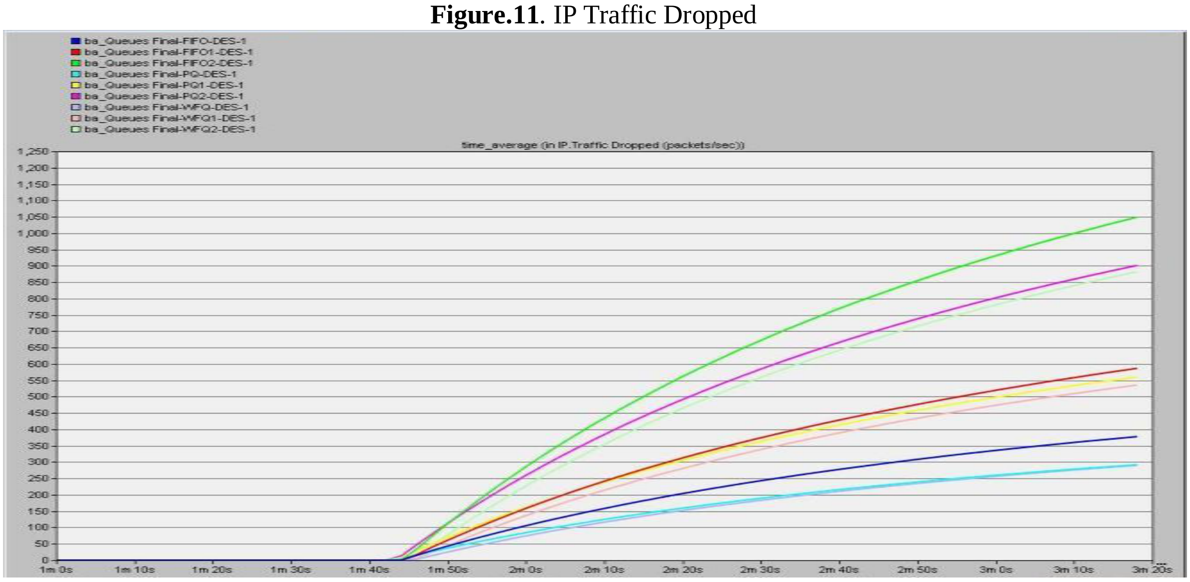
- Packet Delay Variation for VoIP: Figure No. 10 illustrates the variation in packet latency for (VoIP). The outcome of the packet delay variation for VoIP, especially when utilizing PQ and WFQ approaches, is always in line with end-to-end latency in all cases. It shows a number near zero, whereas FIFO is always greater. When we examine the first (FIFO) and third (FIFO) scenarios, we see that the packet delay variance increased with scenarios 1 to 3. Encoded and compressed packets in speech application at buffer will consume Time in FIFO. The buffering delay resulted from a voice sequence that had to wait until another speech sequence occurred in the queue.

Figure.10. Voice Delay Variation



- IP Traffic Dropped : The number of packets lost per second for all apps (Video conversing, VoIP, and FTP) is shown in Figure 11. in all scenarios. During the simulation, FIFO has a greater value for packet loss, but all the networks (WFQ and PQ) contain fewer packet losses than (FIFO).

Figure.11. IP Traffic Dropped



The number of dropped packets is higher in FIFO because it uses one queue in its system, so the buffer fills. At the same time, other algorithms (WFQ, P Q) have multiple queues in the system, packets are passed between different queues, and the number of data losses is less than other algorithms as the buffer fills. In contrast, other algorithms (WFQ, PQ) have multiple queues in the system, packets are passed between different queues, and the number of data losses is less than other algorithms as the buffer fills. In contrast, other algorithms (WFQ, PQ) contain multiple queues in the system. Also, packets are passed between different queues and numbers as the buffer fills. In contrast, other algorithms (WFQ, PQ) have multiple queues in the system, packets. Figure 11 shows that FIFO affected negatively for guaranteed (QoS) to these applications thereby affects. The dropping packets' results are similar to the results given in NO. [11] Which analysis the same queuing algorithms. So average values in the simulations show below:

Table .3. The parameters' values for all networks.

Application	***Performance Parameter***	***The network with a Low traffic***			***The network with a medium traffic***			***The network with a high traffic***		
		WFQ1	PQ1	FIFO1	WFQ2	PQ2	FIFO2	WFQ3	PQ3	FIFO3
Video	*Packet End-to-End Delay(sec)*	*4*	*0.5*	*1.6*	*3.7*	*0.5*	*2.4*	*3.5*	*0.8*	*3.5*
	Packet Delay Variation (sec)	*2.2*	*0.05*	*0.8*	*2*	*1.8*	*0.9*	*5*	*2*	*1.9*
Voice	*Traffic Received(bytes/sec)*	*14000*	*15500*	*7000*	*30000*	*30500*	*27463*	*36000*	*38000*	*23500*
	Packet End-to-End Delay(sec)	*1.3*	*0.2*	*3.2*	*1.5*	*0.2*	*3.6*	*2*	*0.4*	*3.9*
	Packet delay variation (sec)	*0*	*0*	*0.14*	*0*	*0*	*0.11*	*0*	*0*	*0.85*
IP	*Traffic Dropped(packet/sec)*	*290*	*290*	*468*	*536*	*560*	*638*	*865*	*900*	*1066*
FTP	*Traffic received(packet/sec)*	*0.5*	*0.5*	*2*	*0.5*	*0.5*	*0.5*	*0.1*	*0.1*	*0.5*

V. Conclusion

As explained previously, an OPNET modeler was used to compare the performance of several queuing algorithms in this research. We thoroughly investigated the many aspects that influence the network's QoS. We created three network topologies with many clients and routers with comparable traffic types. Since it has a reduced packet delay variation and (ETE) delay, the results indicate that PQ is the optimal strategy for video conferencing and VoIP. Compared to FIFO or PQ, the high-resolution video demonstrates that the WFQ has a larger packet delay variance and ETE delay value. The study demonstrates that FIFO causes higher packet loss and ETE delays with video conferencing and speech applications. So, FIFO has worse performance than the other two algorithms for video and VoIP used in networks and better performance for FTP applications. As a result, FIFO is considered the worst choice for multimedia applications.

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