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Assessing the effects of Queuing Algorithms and Routing Protocols on Quality of Service for Video Traffic over IPv4 and IPv6 networks.

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Highlights

- **Queueing discipline—not routing protocol nor IP version—dominates video QoS performance under congestion.**
- **Priority Queuing (PQ) consistently minimizes delay and jitter, making it optimal for real-time video traffic.**
- **Weighted Fair Queuing (WFQ) improves aggregate throughput, exposing a clear fairness–latency trade-off.**
- **IPv6 demonstrates greater reliability than IPv4, with lower packet loss and more stable delay behavior.**
- **Results provide deployment-level guidance for selecting queueing strategies in next-generation IP video networks**

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ABSTRACT

This paper investigates the impact of two widely used routing protocols—Open Shortest Path First (OSPF) and Enhanced Interior Gateway Routing Protocol (EIGRP)—and two queueing techniques—Priority Queuing (PQ) and Weighted Fair Queuing (WFQ)—on the Quality of Service (QoS) of video conferencing traffic in both IPv4 and IPv6 networks. Through extensive simulations, we evaluate the performance of each queueing method in combination with each routing protocol under conditions of heavy traffic and moderately large network topologies. The analysis focuses on key QoS parameters, including throughput, end-to-end delay, packet delay variation, packet loss, and point-to-point delay. The objective is to identify the optimal pairing of routing protocol and queueing mechanism that yields the best QoS performance for video traffic in TCP/IP environments. Furthermore, the scalability of the results is assessed by replicating the scenarios in IPv6 networks. All simulations were conducted using OPNET Modeler 14.5A, providing a comprehensive evaluation of the interactions between routing protocols and queueing disciplines across diverse network conditions.

1. Introduction

The rapid growth of video traffic can be attributed to the increasing reliance on real-time communication tools, video conferencing, online education, and entertainment streaming platforms. This surge in demand places significant pressure on network infrastructures to deliver seamless, low-latency video transmission—an inherently challenging requirement. The performance of such networks largely depends on two critical factors: the queuing algorithms that manage packet scheduling and the routing protocols that determine data paths across the network.

Video streaming and conferencing applications require consistent bandwidth and minimal delay to maintain acceptable quality levels, making Quality of Service (QoS) a fundamental consideration for network operators. Queuing algorithms, which control the order and priority of packet transmission, play a vital role in alleviating congestion and ensuring timely delivery of video data. Common techniques such as Priority Queuing (PQ) and Weighted Fair Queuing (WFQ) each possess distinctive characteristics that can significantly influence video traffic performance. Similarly, routing protocols such as Open Shortest Path First (OSPF) and Enhanced Interior Gateway Routing Protocol (EIGRP) govern the selection of optimal paths for data delivery, directly affecting network efficiency and stability under varying load conditions.

With the transition from IPv4 to IPv6, additional mechanisms—such as the flow label feature—have been introduced to enhance QoS support and traffic management. Understanding how different queuing and routing strategies interact within both IPv4 and IPv6 environments is therefore crucial for optimizing performance in modern networks. Research in this area is essential to provide practical insights for network engineers and administrators tasked with designing and maintaining high-performance infrastructures that can accommodate the growing demand for high-quality video services. By evaluating the performance of various queuing algorithms and routing protocols, this study contributes to the development of best practices and informs future approaches for managing video traffic in increasingly complex network environments.

2. Related work

The measurement of Quality of Service (QoS) in networks remains an active area of research, closely associated with overall network performance. Numerous studies have examined QoS and its related parameters, with the most relevant findings summarized below. Sugirtham and Jenny (2021) investigated the behavior of various queuing techniques in managing multi-class traffic. Their analysis demonstrated that Weighted Fair Queuing (WFQ) and Priority Queuing (PQ) outperform other queuing disciplines in terms of average throughput and queuing delay. Similarly, Hossain *et al.*,

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(2018) evaluated several queuing algorithms (FIFO, PQ, WFQ) under high video traffic loads and reported that PQ consistently achieves lower delay and jitter compared to other techniques.

Gupta *et al*., (2015) shifted the focus to routing protocols, comparing OSPF and EIGRP for video streaming applications. Their results indicated that OSPF generally maintains lower packet loss rates and improved load balancing, thereby enhancing QoS for video services. In a related study, Harly, S. (2025) compared IPv4 and IPv6 networks in managing multimedia traffic, concluding that IPv6 networks exhibit lower latency and improved QoS support. Furthermore, Al-Habashneh and Al-Qadi (2017) analyzed QoS parameters for multimedia traffic across diverse network configurations, emphasizing the significance of selecting appropriate queuing disciplines and routing protocols. Kumar and Syed (2015) extended this work by examining traffic-shaping techniques in conjunction with various queuing algorithms, concluding that effective traffic shaping substantially enhances video quality.

Alshahrani and Alhassan (2016) explored the inherent advantages of IPv6, highlighting the role of flow labels and traffic classes in improving video traffic management under high-demand scenarios. Neha *et al.* (2015) revisited the comparative performance of FIFO, PQ, and WFQ, analyzing their impact on application performance and network resource utilization. Their findings suggest that WFQ does not consistently yield superior QoS outcomes for video conferencing traffic, challenging common assumptions in prior studies.

More recently, Mresa and Ergheegh (2025) investigated the combined effects of queuing algorithms (WFQ, PQ, FIFO) and routing protocols (RIP, OSPF, EIGRP) on QoS for VoIP traffic over IPv4 networks. Using extensive simulations across different topologies and traffic loads, they found that EIGRP paired with WFQ effectively minimizes packet delay and improves QoS, with PQ serving as a viable alternative. Similarly, Balasundaram *et al.* (2014) examined the performance of queuing algorithms for multimedia applications—particularly video conferencing and VoIP—and concluded that PQ achieves lower end-to-end delay, especially for high-resolution video transmissions.

Collectively, these studies underscore the intricate relationship between queuing algorithms, routing protocols, and Quality of Service (QoS) in multimedia communications. They highlight the continuing need for comprehensive evaluations as network technologies evolve, particularly with the increasing adoption of IPv6. Despite extensive prior work, limited studies have examined the combined effects of routing and queuing mechanisms under both IPv4 and IPv6 environments using identical topologies and traffic conditions. Addressing this gap, this study conducts a detailed simulation-based assessment of selected routing protocols and queueing techniques to determine their impact on QoS performance for real-time video traffic.

3. Routing Protocols & Queueing Techniques

This section provides an overview of the routing protocols and queueing techniques used in the simulation experiments. The selection of Open Shortest Path First (OSPF), Enhanced Interior Gateway Routing Protocol (EIGRP), Priority Queuing (PQ), and Weighted Fair Queuing (WFQ) is motivated by their widespread adoption and proven relevance in prior performance-evaluation studies (Mresa and Ergheegh (2025), Ezeddien *et al.,* 2014; Thorenoor, 2010; Jalendry *et al*., 2016; Hussein *et al*., 2013). The following subsections briefly describe their operational principles and characteristics pertinent to Quality-of-Service (QoS) evaluation.

Routing Protocols (OSPF, EIGRP)

Shortest Path First (OSPF) is a link-state routing protocol that constructs a comprehensive topology database of the network to determine the most efficient paths. It employs Dijkstra's shortest-path algorithm, accounting for link costs that may include bandwidth, latency, and reliability metrics.

Enhanced Interior Gateway Routing Protocol (EIGRP) is a hybrid routing protocol combining the advantages of distance-vector and link-state approaches. It features rapid convergence, efficient bandwidth utilization, and supports load balancing and route summarization.

Queueing Techniques (PQ, WFQ)

Priority Queuing (PQ) classifies traffic into multiple priority levels, transmitting high-priority packets—such as those from video conferencing and Voice-over-IP (VoIP) applications—before lower-priority traffic. This mechanism effectively reduces delay and jitter for time-sensitive flows.

Weighted Fair Queuing (WFQ) by contrast, allocates bandwidth proportionally among competing traffic flows based on predefined weights. It ensures fairness and enables QoS differentiation while maintaining efficient resource utilization.

Together, these mechanisms constitute the basis for effective network performance management and QoS optimization. The routing protocols determine how forwarding decisions are made, whereas the queueing techniques dictate how packets are prioritized during transmission. Understanding their interactions is essential for configuring networks capable of sustaining high-quality real-time multimedia services under varying load conditions.

4. Performance Metrics (QoS)

Quality of Service (QoS) metrics in networking refer to the parameters used to evaluate and quantify the performance and reliability of a network in delivering traffic to end users. These metrics assess how effectively the network infrastructure meets the requirements of different applications, ensuring acceptable service levels and user experience. In the context of real-time multimedia traffic—such as video conferencing and Voice over IP (VoIP)—specific QoS metrics are critical for maintaining consistent quality and minimizing disruptions. The primary QoS metrics considered in this study are outlined below.

4.1.Packet Loss

Packet loss represents the proportion of data packets that fail to reach their destination during transmission. Even a small degree of packet loss can noticeably degrade the quality of multimedia applications, resulting in dropped audio frames or distorted video. Minimizing packet loss is therefore essential to maintaining clear, uninterrupted communication and ensuring reliable packet delivery.

4.2 Delay or Latency

Delay, or latency, denotes the time required for a packet to travel from the source node to the destination. In real-time applications, excessive delay can cause perceptible lag between audio and video streams, leading to synchronization problems and impaired interaction. Maintaining low latency is crucial to achieving smooth and responsive communication in multimedia systems.

4.3 Delay Variation (Jitter)

Jitter refers to the variation in packet arrival times, or the inconsistency in delay between successive packets. High jitter values can cause audio distortions or uneven video playback in multimedia transmissions. Minimizing jitter ensures stable and synchronized delivery of media streams, thereby preserving quality and continuity of service.

4.4. Traffic Dropped

Traffic dropped refers to packets intentionally discarded by network devices due to congestion, buffer overflow, or resource limitations. This metric indicates network inefficiency or overload conditions. Reducing dropped traffic is vital for maintaining reliable data transmission and preventing service interruptions in real-time communication scenarios.

4.5. Throughput

Throughput measures the actual rate at which data is successfully transmitted across a network within a given time interval. It reflects the effective data-carrying capacity achieved between source and destination. High throughput is necessary for multimedia applications, as it supports the continuous transfer of large audio and video streams, minimizing buffering and playback interruptions.

4.6. Bandwidth

Bandwidth refers to the maximum data rate that can be transmitted through a network connection over a specified period. Adequate bandwidth availability is essential for supporting high-quality multimedia communication, as insufficient bandwidth can result in congestion, reduced video resolution, or dropped calls. Maintaining sufficient bandwidth allocation ensures smooth transmission of real-time audio and video traffic without bottlenecks or performance degradation.

By continuously monitoring and optimizing these QoS metrics, network administrators and service providers can enhance network performance, ensuring that real-time multimedia applications—such as VoIP and video conferencing—operate with the required reliability, responsiveness, and overall service quality.

5. Network Scenario Diagram

Network simulations were conducted using OPNET Modeler version 14.5A, a comprehensive simulation environment widely employed for modeling and performance analysis of communication networks. The tool supports the design and evaluation of heterogeneous network topologies incorporating multiple protocols and traffic types. In this study, OPNET Modeler was used to construct the network scenario, configure routing protocols and queueing mechanisms, and capture performance metrics relevant to Quality of Service (QoS) evaluation. This platform enables accurate modeling of network behavior and interactions, making it suitable for assessing the impact of routing and queueing strategies on real-time multimedia traffic.

The paper employs the simulated network model shown in Fig.1 to conduct a comparative analysis of different combinations of routing protocols and queueing techniques to evaluate the effectiveness of these combinations in delivering optimal Quality of Service (QoS) for video conferencing traffic across both IPv4 and IPv6 networks. Specifically, the study seeks to identify the pairing of routing protocol and queueing mechanism that yields the highest QoS performance under varying network conditions. To achieve this objective, all possible protocol–technique combinations were systematically configured, simulated, and analyzed within the OPNET Modeler environment. The simulated topology represents a typical enterprise network consisting of multiple interconnected routers and end-user stations.

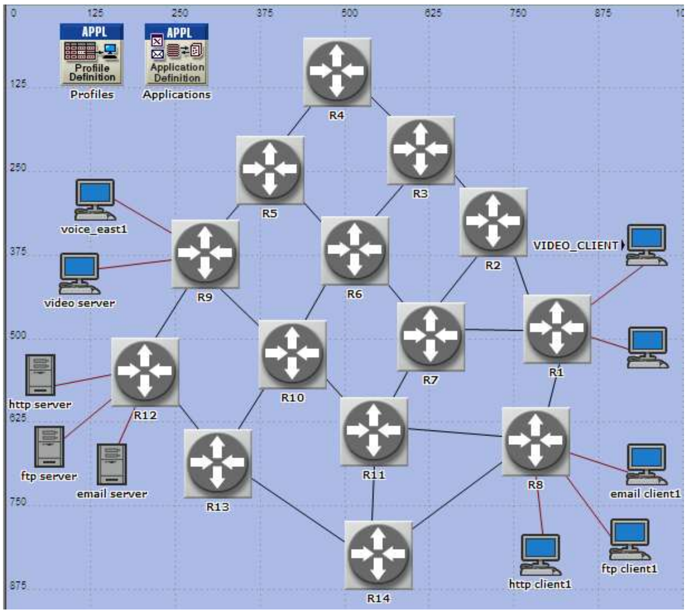


Fig. 1: Topology of the network case study

Fig. 1 illustrates the case study network. The topology includes fourteen routers (R1–R14), VoIP and video-conferencing stations, FTP and Email clients and servers, HTTP endpoints, and PPP_DS1 links (1.544 Mbps). This configuration reflects a typical enterprise network carrying diverse application traffic. It allows for the simulation of various traffic types, including real-time multimedia, data transfer, and web-based communication, under consistent network conditions. The inclusion of multiple applications ensures realistic traffic diversity and enables comprehensive QoS performance evaluation for each routing and queueing combination.

At the application layer, multiple traffic types were configured to create realistic load conditions and ensure a comprehensive evaluation of Quality of Service (QoS) under varying conditions. Table 1 summarizes the configurations used consistently across all simulations, including high-load HTTP, G.711 voice, and low-resolution video conferencing. These configurations were applied uniformly across all test cases to maintain consistency and enable fair comparison between routing protocol and queueing technique combinations.

Table 1. OPNET Application Configurations

High Traffic	
Email	High Load
HTTP	HTTP, HEAVY BROWSING
Video Conferencing	128*240 PIXELS, 15 FRAMES/SEC
Voice	PCM Quality Speech Encoder G.711

The selected application parameters represent typical high-traffic enterprise conditions and were chosen to create realistic congestion levels for performance assessment. This ensures that the resulting QoS measurements accurately reflect how each routing and queueing configuration performs under multimedia traffic loads. The following section details the simulation setup and performance metrics used for evaluation.

6. Simulation Scenarios

Extensive simulation scenarios were conducted using the case study network topology described in the previous section. Each simulation ran for a duration of 30 minutes in OPNET Modeler 14.5A, under identical network configurations and application settings. The primary objective of these simulations was to determine the most effective combination of routing protocol and queueing technique that yields optimal Quality of Service (QoS) performance for video-conferencing traffic over IP networks.

To replicate realistic operating conditions, the simulated environment incorporated multiple concurrent traffic types—VoIP, HTTP, FTP, and Email—in addition to video conferencing streams. This mixed-traffic design captures the heterogeneity and dynamics of real-world networks, enabling a more accurate evaluation of routing and queueing performance under congestion and cross-traffic conditions.

To achieve the research objectives, the simulation scenarios were designed to include the following elements:

- *Two Interior Gateway Protocol (IGP) routing algorithms: OSPF and EIGRP.*
- *Two queueing scheduling techniques: Priority Queuing (PQ) and Weighted Fair Queuing (WFQ), each applied in combination with both routing protocols.*
- *A high-traffic load applied consistently across all scenarios to assess performance under stress conditions.*
- *Dual IP environments: Each configuration was implemented and analyzed separately in both IPv4 and IPv6 networks to examine protocol-level influence on QoS.*

The study focused specifically on video-conferencing traffic as the primary real-time multimedia application. The simulation framework encompassed three key dimensions—routing, queueing, and IP version (IPv4 and IPv6)—allowing for a comprehensive comparative analysis of QoS performance. By evaluating every routing–queueing pairing under identical conditions, the research provides a detailed assessment of how each combination impacts throughput, delay, jitter, and packet loss.

This methodology offers a holistic understanding of how routing and queueing mechanisms interact under varying network loads and protocol environments. Evaluating QoS performance for video-conferencing applications in both IPv4 and IPv6 contexts, while subjecting the network to concurrent multimedia and data traffic, ensures that the results accurately reflect practical deployment conditions.

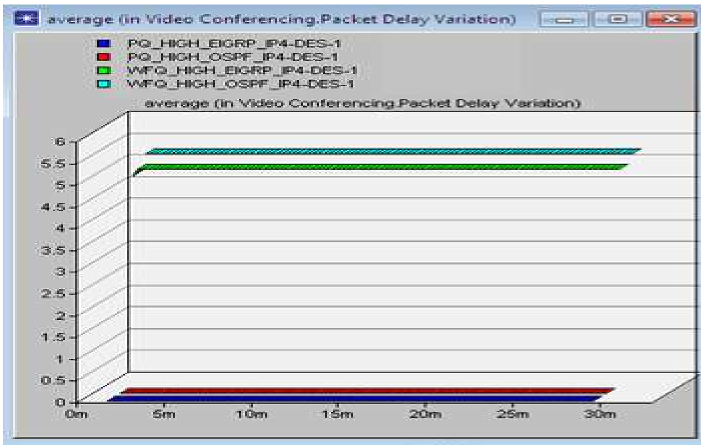
7. Simulation Results and Analysis

The simulation results presented in Figs. 2–11 illustrate the accumulated average values of the evaluated Quality of Service (QoS) metrics obtained for each routing-protocol and queueing-technique pairing—specifically, OSPF and EIGRP combined with Priority Queuing (PQ) and Weighted Fair Queuing (WFQ)—across both IPv4 and IPv6 network environments. For clarity, the quantitative results corresponding to each figure are tabulated beneath the respective plots.

7.1 Packet Delay Variation

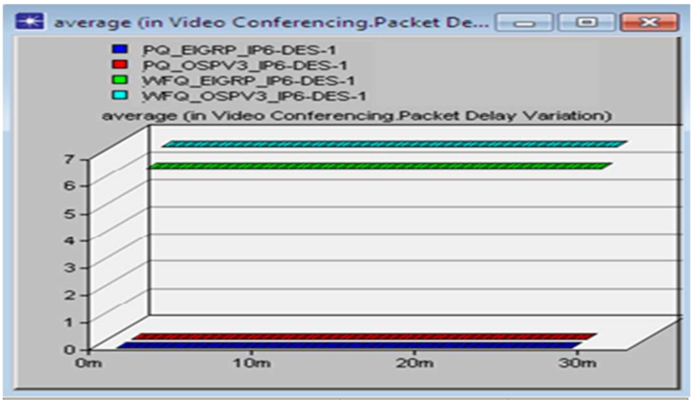
Packet Delay Variation (PDV) represents the fluctuation in the time required for packets to traverse the network. It is a critical QoS metric for real-time multimedia applications, as irregular packet arrival times can lead to playback distortion and reduced perceptual quality. Figs. 2 and 3 depict the measured PDV for IPv4 and IPv6 networks, respectively.

The simulation results show that Priority Queuing (PQ) consistently achieves lower delay variation than Weighted Fair Queuing (WFQ) in both IP versions. This outcome aligns with expectations, since PQ prioritizes time-sensitive traffic such as video frames, thereby minimizing delay fluctuation. The routing protocol (OSPF or EIGRP) exhibited negligible influence on PDV under either queuing method, indicating that delay variation is primarily determined by the queuing discipline.



Video Conference [IPv4] Packet Delay Variation	OSPF	EIGRP
PQ	0.0335	0.0387
WFQ	5.2016	5.0060

Fig. 2: [IPv4] Packet delay variation (sec)



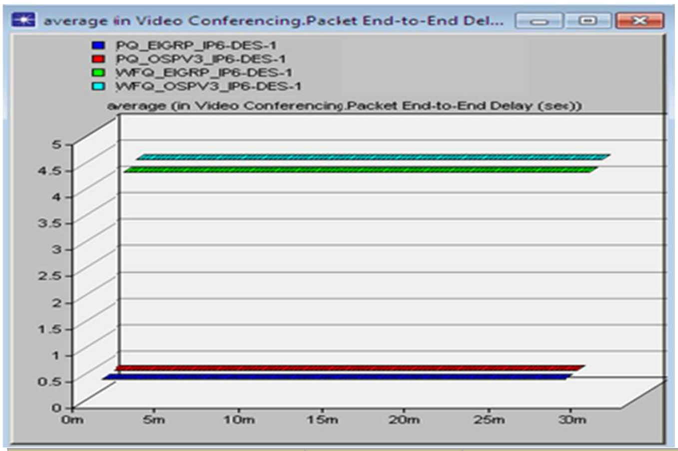
Video Conferencing [IPv6] Packet Delay Variation	OSPFv3	EIGRPv6
PQ	0.0538	0.0409
WFQ	6.4882	6.0206

Fig. 3: [IPv6] Packet delay variation (sec)

The consistent superiority of Priority Queuing (PQ) over Weighted Fair Queuing (WFQ) demonstrates PQ's ability to deliver more stable and lower delay variation—an essential characteristic for real-time video transmission. This behavior stems from the fundamental operation of the two queuing mechanisms. WFQ distributes bandwidth fairly among competing traffic flows, but such fairness can inadvertently delay latency-sensitive packets. Conversely, PQ prioritizes critical traffic classes, ensuring that time-sensitive packets, such as video frames, are transmitted promptly with minimal jitter. As a result, PQ proves more effective for real-time multimedia applications that demand continuous and synchronized data delivery. Additionally, the findings reaffirm that the routing protocol (OSPF or EIGRP) has an insignificant effect on packet delay variation in both IPv4 and IPv6 environments. This consistency highlights that PDV is primarily influenced by the queuing strategy rather than the routing mechanism, underscoring the pivotal role of efficient queue management in maintaining high-quality video communication.

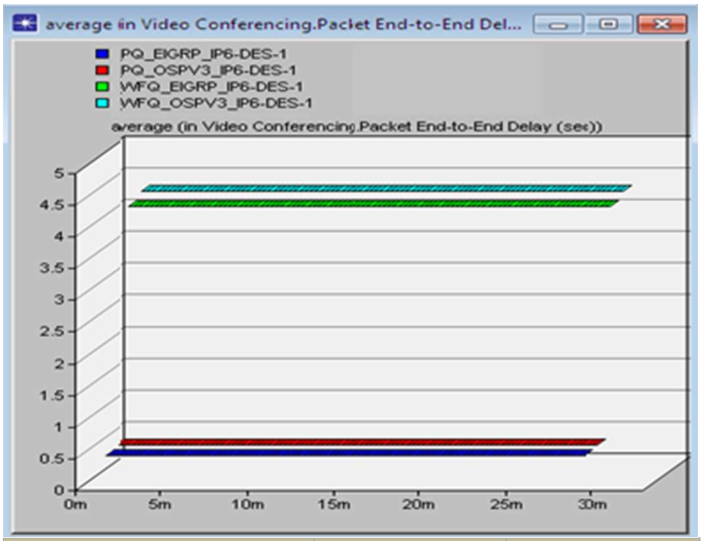
7.2 Packet End-to-End Delay

The packet end-to-end delay for video traffic in IPv4 and IPv6 networks is illustrated in Fig. 4 and 5, respectively. The results show that Priority Queuing (PQ) technique consistently achieved



Video Conferencing End To End Delay [IPv4]	OSPF	EIGRP
PQ	0.5379	0.5332
WFQ	4.17687	5.9025

Fig. 4: [IPv4] Packet End-to-End delay (sec)



Video Conferencing [IPv6] Packet End to End Delay	OSPFv3	EIGRPv6
PQ	0.5645	0.5429
WFQ	4.2785	4.1715

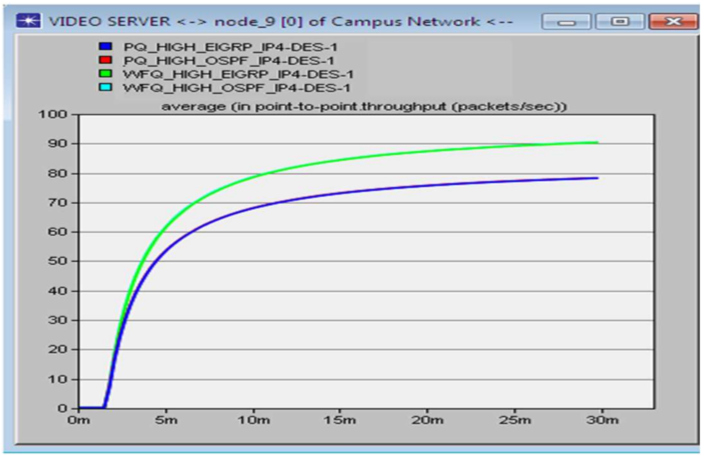
Fig. 5: [IPv6] Packet End-to-End delay (sec)

the lowest delay across all routing protocols and network environments, whereas Weighted Fair Queuing (WFQ) exhibited comparatively higher delay values. The improvement achieved by PQ is evident in both OSPF and EIGRP configurations, confirming its efficiency in supporting time-sensitive multimedia transmission.

The difference between the two queuing methods remains consistent across both IP versions, indicating that the routing protocol and IP version exert minimal influence on end-to-end delay performance. Overall, PQ maintained superior responsiveness for real-time video delivery, while WFQ's fairness-oriented bandwidth allocation introduced higher delay under congested conditions.

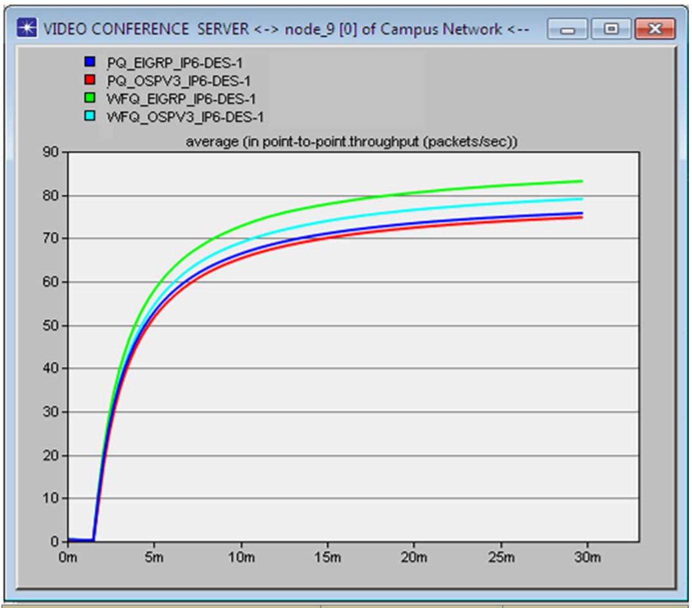
7.3 *Point-to-Point Throughput (packets/sec)*

Figs. 6 and 7 illustrate the point-to-point throughput results for IPv4 and IPv6 networks, respectively. The outcomes show that the Weighted Fair Queuing (WFQ) technique achieved higher throughput values than Priority Queuing (PQ) across both routing protocols. Throughput was also consistently greater in IPv4 than in IPv6, whereas the routing protocol—OSPF or EIGRP—had negligible influence on overall performance.



Throughput	OSPF	EIGRP
PQ	78.2911	78.2017
WFQ	90.4583	90.3133

Fig. 6: [IPv4] Point-to-Point Throughput



Point-to-Point Throughput	OSPFv3	EIGRPv6
PQ	74.79666	75.75388
WFQ	79.04666	83.14166

Fig 7: [IPv6] Point-to-Point Throughput

The superior throughput of WFQ arises from its proportional bandwidth allocation, which ensures that all traffic classes are served fairly and continuously, maximizing link utilization under high-load conditions. In contrast, PQ's strict-priority scheduling often leads to starvation of lower-priority queues, causing packet drops and reduced aggregate throughput despite its efficiency for real-time flows.

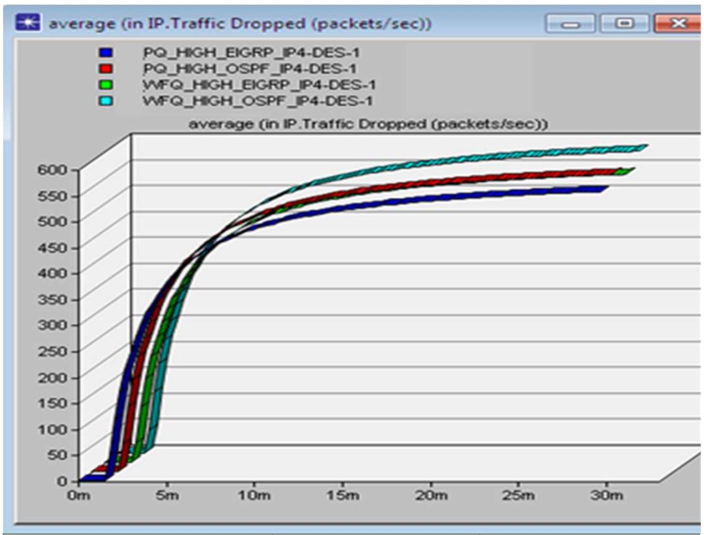
It should be noted that the throughput presented here reflects the aggregate link throughput, encompassing all traffic classes. If throughput were measured solely for the video stream, PQ would yield higher values due to its prioritization of delay-sensitive packets. Hence, WFQ demonstrates superior overall link efficiency, while PQ excels in ensuring the quality of service for high-priority applications. The minimal variation between OSPF and EIGRP further indicates that throughput is primarily governed by the queuing discipline and IP protocol overhead rather than routing strategy.

The slightly higher throughput of IPv4 can be attributed not merely to its smaller header size but also to lower processing overhead and reduced control signaling, which together yield marginally higher efficiency under high-load conditions.

7.4 Traffic Dropped

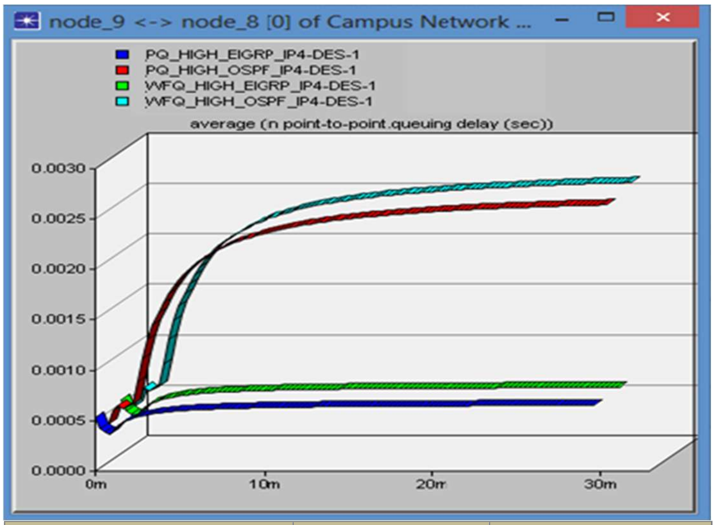
Following the throughput analysis, Figs. 8 and 9 present the traffic loss observed in the simulated IPv4 and IPv6 networks, respectively. The results show that IPv6 consistently experienced lower traffic loss than IPv4 under identical routing and queuing configurations. Differences between routing protocols (OSPF and EIGRP) and queuing disciplines (PQ and WFQ) were minimal, indicating that traffic loss was largely unaffected by these parameters.

The lower packet loss in IPv6 reflects its more controlled and adaptive congestion behavior, which stabilizes queue utilization and minimizes packet drops even under heavy load. In contrast, IPv4's faster packet-handling rate tends to generate burstier traffic patterns that can momentarily saturate router buffers, leading to slightly higher discard rates. Overall, the findings highlight IPv6's inherent efficiency in maintaining stable and reliable data delivery for real-time video applications.



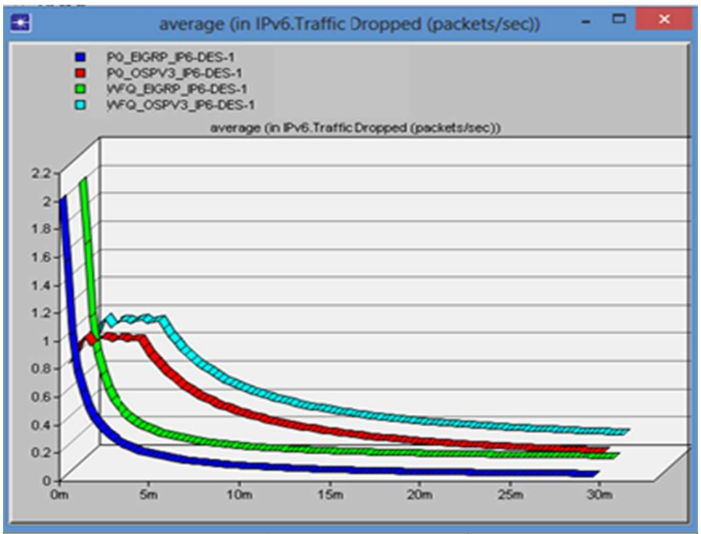
Traffic Dropped [IPv4]	OSPF	EIGRP
PQ	573.4828	557.7833
WFQ	582.8250	556.0261

Fig. 8: [IPv4] Traffic Dropped



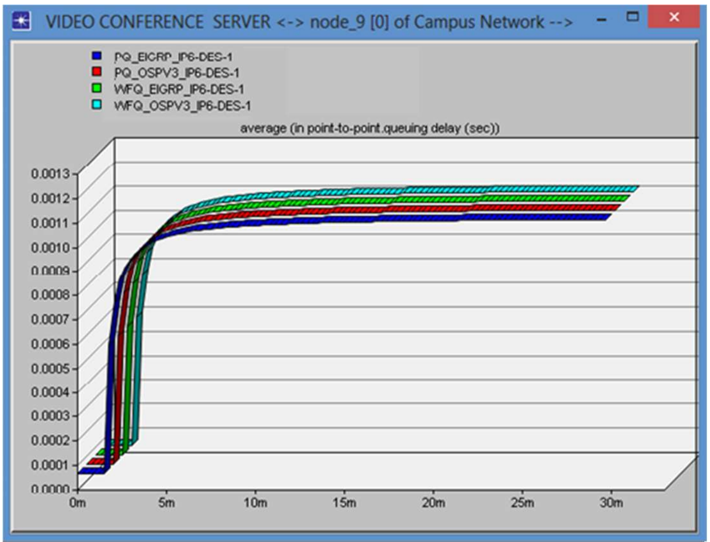
Point-to-point Queueing Delay	OSPF	EIGRP
PQ	0.0025483	0.0006460
WFQ	0.0025927	0.0006460

Fig. 10: [IPv4] Point-To-Point Queueing Delay



Traffic Dropped [IPv6]	OSPFv3	EIGRPv6
PQ	0.13166	0.03111
WFQ	0.13277	0.03111

Fig 9: [IPv6] Traffic dropped (packets/sec)



Point-To-Point Queueing Delay	OSPFv3	EIGRPv6
PQ	0.0011126	0.0011126
WFQ	0.0011126	0.0011126

Fig. 11: [IPv6] Point-To-Point Queueing Delay

7.5 Point-To-Point Queueing Delay

Figs. 10 and 11 present the simulation results for point-to-point queuing delay in IPv4 and IPv6 network topologies, respectively. The results show that queuing delay is largely unaffected by the choice of queueing algorithm in either IP version. However, in the IPv4 environment, the EIGRP routing protocol consistently achieved slightly lower queuing delays compared to OSPF, indicating marginally better delay optimization. In contrast, IPv6 exhibited more uniform queuing delay behavior across all routing and queueing combinations, suggesting a higher degree of stability and robustness for this QoS metric under varying network configurations.

Figs. 10 and 11 illustrate the point-to-point queueing delay results for IPv4 and IPv6 networks, respectively. This metric represents the latency between two directly connected nodes, providing a localized measure of link responsiveness rather than end-to-end delay or jitter. The results show that the choice of queueing discipline had negligible influence on delay in both IP versions. Within the IPv4 topology, EIGRP achieved slightly lower delay values (0.000646 s) than OSPF (≈0.0025 s), indicating marginally higher forwarding efficiency.

In contrast, all IPv6 configurations exhibited identical and consistently low delays (≈0.0011 s), reflecting stable and predictable link behavior under varying conditions. These results confirm that point-to-point delay is mainly determined by link and processing characteristics rather than routing or queueing mechanisms. The uniform IPv6 performance underscores its reliability in sustaining steady per-link responsiveness for real-time multimedia traffic.

8. Conclusion

This paper investigated the effects of routing protocol and queueing mechanism pairings on the Quality of Service (QoS) of video-conferencing traffic over IPv4 and IPv6 networks using OPNET Modeler 14.5A. Four configurations—OSPF-PQ, OSPF-WFQ, EIGRP-PQ, and EIGRP-WFQ—were simulated under identical heavy-traffic conditions to ensure fair and objective comparison. The evaluation encompassed key QoS parameters, including throughput, end-to-end delay, packet delay variation, traffic dropped, and point-to-point queueing delay, providing a comprehensive assessment of how each combination influences video performance under congestion. The impact of routing protocol was marginal, with EIGRP exhibiting slightly lower delay than OSPF.

Regarding queueing performance, Priority Queuing (PQ) consistently delivered the lowest delay and jitter, making it well suited for real-time video streams, while Weighted Fair Queuing (WFQ) achieved higher overall throughput by distributing bandwidth more evenly among concurrent flows. This contrast highlights the inherent trade-off between responsiveness and fairness when managing multimedia traffic under congestion.

A notable outcome is the enhanced reliability of IPv6 compared with IPv4, particularly in terms of reduced traffic loss and more consistent link-level delay. These results emphasize the maturity and efficiency of the IPv6 architecture in supporting multimedia traffic through improved congestion management and streamlined header processing, reinforcing its capability to sustain heavy video loads.

Overall, the findings indicate that the queueing discipline exerts a more decisive influence on video QoS than routing protocol or IP version. For practical deployment, PQ is recommended where delay and jitter are critical, whereas WFQ is preferable in mixed-traffic environments prioritizing aggregate throughput. The superior stability and lower loss rates observed in IPv6 further highlight its growing suitability for real-time multimedia communication in next-generation IP networks.

Future Work

While this study provides valuable insights into the relationship between routing protocols, queueing techniques, and Quality of Service (QoS) in video conferencing over IPv4 and IPv6 networks, several extensions can further enrich the findings. Future research may focus on dynamic and adaptive queue management mechanisms, such as Random Early Detection (RED) or Weighted Random Early Detection (WRED), to evaluate their impact on real-time traffic under variable congestion levels.

Moreover, expanding the analysis to include larger-scale topologies, wireless or hybrid network environments, and heterogeneous traffic patterns would provide deeper insights into scalability and protocol interoperability. Finally, implementing real-time experimental validation on physical testbeds or emulated platforms could strengthen the correlation between simulation and real-world performance, supporting the development of more robust QoS frameworks for next-generation IP networks.

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