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Evaluating the Impact of Queuing Algorithms and Routing Protocols on QoS for VoIP Traffic over IP Networks

Elfurjani Sasi Mresa [1 *], Abdelfattah Saleh Ergheegh [2]
University of Tripoli
Department of Electrical and Electronic Engineering
Tripoli, Libya
e.mresa@uot.edu.ly , a.ergheegh@uot.edu.ly

[*] *Corresponding author: e.mresa@uot.edu.ly*

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Abstract

As internet usage and digital applications continue to surge, ensuring high-speed connectivity with dependable Quality of Service (QoS) for multimedia communications—particularly VOIP—has become increasingly vital. To meet this demand, various traffic management systems have been explored. This study conducts an in-depth analysis of widely adopted queuing methods (FIFO, Priority Queuing, and Weighted Fair Queuing) in conjunction with routing protocols (RIP, EIGRP, and OSPF), assessing their impact on QoS for VOIP traffic over IPv4 networks. Through comprehensive simulation experiments using Opnet 14.5A, the research evaluates each combination under varying traffic loads and network scales. Key QoS indicators such as jitter, packet loss, end-to-end delay, packet delay variation, and traffic throughput were measured. The objective is to determine which routing protocol most effectively complements each queuing strategy in optimizing QoS, and to identify the most robust pairings of routing and queuing approaches for enhanced VOIP performance in TCP/IP networks
Keywords: QoS, VOIP, Routing Protocols, Queuing Strategies, RIP, EIGRP, OSPF, FIFO, PQ, WFQ

1. Introduction and Background

The rapid expansion of real-time multimedia services and increasing reliance on Voice over IP (VoIP) technologies have intensified the complexity of maintaining service quality across IP networks. Ensuring uniform performance in multimedia communications—especially Voice over IP (VoIP)—necessitates rigorous management of Quality of Service (QoS) metrics, including latency, jitter, packet loss, and bandwidth efficiency. Addressing these challenges necessitates a detailed assessment of queuing algorithms and routing protocols to identify strategies that uphold QoS standards.

Effective VoIP communication depends on selecting optimal routing paths and managing traffic prioritization. Routing protocols govern packet delivery across network nodes, while

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queuing mechanisms determine how packets are processed and prioritized based on predefined rules. Aligning these technologies is essential to minimize service disruptions and enhance overall network performance.

This study investigates the impact of routing protocols and queuing algorithms on QoS performance for VoIP traffic over IP-based networks. Specifically, it assesses the relative performance of Routing Information Protocol (RIP), Open Shortest Path First (OSPF), and Enhanced Interior Gateway Routing Protocol (EIGRP), each implemented alongside distinct queuing models—namely, First-In-First-Out (FIFO), Priority Queuing (PQ), and Weighted Fair Queuing (WFQ)—to determine their effect on service quality metrics.

To evaluate these combinations, simulation experiments were conducted using the OPNET Modeler. This approach provides a controlled environment for testing various network configurations, enabling accurate performance comparison across routing-queuing pairings. Simulations were executed on two network topologies with different scales and exposed to varying traffic loads—categorized as Light and Heavy loads. This multidimensional setup allows investigation into how network scale and traffic intensity influence observed outcomes.

The objective of this research is to identify the routing protocol and queuing algorithm combinations that yield the most favorable QoS metrics, including throughput, end-to-end delay, jitter, traffic received, and packet loss. The results aim to inform network engineers in developing efficient, resilient configurations for VoIP communication and to bridge theoretical models with practical insights for optimizing multimedia delivery in real-time IP networks.

Previous studies have explored the roles of routing protocols and queuing mechanisms independently. For instance, Sugetha and Sherine Jenny [1] examined how different queuing disciplines affect multi-class traffic performance, emphasizing the importance of queuing strategy in optimizing network efficiency. Sllam et al. [3] analyzed QoS improvements in IP/MPLS networks through routing-queuing optimization, while Balasundaram et al. [4] highlighted service quality variation across queuing disciplines via OPNET simulations. In another study, Sllam and Ammarah [6] assessed VoIP performance over MPLS networks employing SIP and H.323 signaling protocols, supported by RSVP to improve QoS, and concluded that protocol selection should be tailored to specific network requirements. Other works, such as those by Mahdi et al. [9], Mahmood [10], and Pimwong and Sharma [11], addressed VoIP QoS in hybrid environments, IPv6 transitions, and buffer optimization respectively.

Despite these contributions, a research gap persists regarding the interaction between routing protocols and queuing strategies under varying network sizes and traffic loads. This study addresses this gap by conducting extensive simulations across diversified topologies and traffic scenarios.

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By identifying optimal protocol-algorithm combinations and evaluating their sensitivity to network conditions, this research contributes valuable guidelines for delivering high-quality, scalable VoIP services. The findings intend to empower network professionals with data-driven recommendations for robust multimedia transmission across modern IP infrastructures

2. Routing Protocols & Queueing Techniques

The research study, along with previous studies, focuses on the most commonly used routing protocols, namely RIP, OSPF, and EIGRP, as well as the widely studied queueing techniques of FIFO, PQ, and WFQ [3, 5, 8]. These protocols and techniques have been investigated in various research works, and their descriptions are briefly outlined below.

A. Routing Protocols (RIP, OSPF, EIGRP)

The Routing Information Protocol (RIP) is a distance-vector routing protocol predominantly employed within small to medium-scale network architectures. It identifies the most efficient route for data packet transmission by utilizing a hop count metric, whereby the path with the fewest intermediate routers is selected. Conversely, Open Shortest Path First (OSPF) operates as a link-state routing protocol, specifically developed to accommodate the requirements of larger and more complex network environments. It uses a more sophisticated algorithm that considers factors such as network bandwidth, latency, and link costs to determine the shortest and most reliable paths. Enhanced Interior Gateway Routing Protocol (EIGRP) represents a hybrid routing algorithm that integrates the core principles of both distance-vector and link-state methodologies. Designed to optimize routing efficiency in large-scale networks, EIGRP achieves rapid convergence and minimizes bandwidth consumption through its selective update mechanism and neighbor relationships. Furthermore, it incorporates advanced capabilities such as unequal-cost load balancing and hierarchical route summarization, enhancing scalability and performance within complex topological environments.

B. Queueing Techniques (FIFO, PQ, WFQ)

FIFO (First-In-First-Out) is a basic queuing technique where packets are transmitted in the same order they arrive. It does not prioritize any specific traffic and treats all packets equally. Priority Queuing (PQ) is a queuing technique that assigns different priority levels to traffic flows. Higher priority traffic is dequeued and transmitted before lower priority traffic, ensuring that time-sensitive packets, such as VOIP, are given precedence. Weighted Fair Queuing (WFQ) is a queuing technique that aims to provide fairness and bandwidth allocation based on traffic flow weights. It divides available bandwidth proportionally among different flows, allowing for better resource utilization and QoS differentiation based on the assigned weights.

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These routing protocols and queuing techniques play crucial roles in network management and QoS optimization.

The choice of routing protocol affects how network devices exchange routing information and make forwarding decisions, while the selection of queuing techniques determines how packets are managed and prioritized during transmission. Understanding the characteristics and capabilities of these protocols and techniques is essential for network administrators to configure and design networks that can efficiently handle real-time multimedia traffic like VOIP and ensure acceptable performance and QoS.

C. Performance Metrics (QoS)

Quality of Service (QoS) metrics in networking refer to the parameters used to assess and measure the performance and level of service provided to network traffic [4]. These metrics help evaluate the effectiveness of network infrastructure in meeting the requirements of different applications and ensure a satisfactory user experience. In the context of multimedia real-time traffic such as VOIP and video conferencing, specific QoS metrics are crucial to maintaining the desired quality and reliability of these applications. The following QoS metrics are particularly relevant:

- ***Packet Loss***: Packet loss refers to the percentage or number of packets that are lost during transmission. In multimedia applications like VOIP and video conferencing, even a small amount of packet loss can lead to noticeable degradation in audio or video quality. Ensuring reliable packet delivery while minimizing loss is vital for maintaining uninterrupted communication.
- ***Delay or Latency***: Latency, commonly referred to as delay, denotes the duration required for data packets to traverse the network from their point of origin to their intended destination. In real-time multimedia applications, excessive delay can result in noticeable delays between audio and video streams, leading to synchronization issues and communication disruptions. Low latency is crucial to ensure smooth and responsive communication, minimizing lag and maintaining real-time interactions.
- ***Delay Variation (Jitter)***: Jitter denotes the fluctuation in packet latency, specifically the inconsistency in the arrival intervals of successive data packets during transmission. In multimedia applications, jitter can cause disruptions and inconsistencies in audio and video playback, leading to poor quality and synchronization issues. Minimizing jitter is essential to ensure smooth and uninterrupted multimedia streaming.
- ***Traffic Dropped***: Traffic dropped refers to the number of packets that are intentionally discarded by network devices due to congestion or resource limitations. It indicates situations where the network infrastructure is unable to handle the incoming traffic and

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has to discard packets to prevent further congestion. Minimizing traffic dropped is crucial to ensure reliable packet delivery and minimize disruptions in multimedia real-time traffic.

- ***Throughput***: Throughput refers to the volume of data successfully transmitted across a network within a specified temporal interval, serving as a critical indicator of communication efficiency. It represents the actual data rate achieved between the source and destination. In multimedia applications, high throughput is essential to support the transmission of large amounts of data required for audio and video streams, ensuring smooth and continuous playback.
- ***Bandwidth***: Bandwidth refers to the amount of data that can be transmitted over a network connection in a given time period. Sufficient bandwidth is crucial for multimedia applications to transmit high-quality audio and video streams without congestion or bottlenecks. Insufficient bandwidth can result in reduced quality, buffering, or interruptions during multimedia playback.

By monitoring and optimizing these QoS metrics, network administrators and service providers can ensure that multimedia real-time traffic, such as VOIP and video conferencing, receives the necessary performance and reliability requirements to deliver a satisfactory user experience.

3. Network Scenario Diagram

We utilized version 14.0.A of OPNET Modeler for conducting network simulations. OPNET Modeler is a robust network simulation tool that offers a wide range of powerful features and capabilities. It allows for the simulation of heterogeneous networks by incorporating different protocols. With OPNET Modeler, users can accurately model and analyze network behavior, performance, and interactions [5]. It serves as a valuable tool for network engineers and researchers in understanding and optimizing network operations.

The paper introduces two simulated network models, namely "Small-Size" (SS) and "Big-Size" (BS), shown in Figure (1) and Figure (2) respectively, to conduct a comparative analysis of different routing protocol and queueing technique pairings. The main focus of the research is to assess the effectiveness of these pairings in delivering Quality of Service (QoS) for VOIP Traffic Over IP Networks. The primary objective of the study is to identify the efficient combinations of routing protocols and queueing techniques that result in high QoS performance. To achieve this, all possible combinations are thoroughly evaluated and analyzed.

However, the study goes beyond this main objective by incorporating a secondary investigation. This investigation exposes both SS and BS models to light and medium traffic

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loads, analyzing how these factors influence the efficiency of the evaluated pairings. This provides valuable insights into the scalability and adaptability of these pairings under varying network conditions.

Figure (1) shows the Small-Size network case study. The node icons in OPNET represent the following network elements:

- Six conventional IP routers: (R1, R2, R3, R4, R5 and R6)
- VoIP stations (VoIP_West and VoIP_East)
- Two FTP stations (FTP_Client and FTP_Server)
- Two Video Conference stations
- Two Email stations (Email Client and Email _Server).
- Two Http stations (Http _Client and Http _Server).
- PPP_DS1 links (1.544 Mbps).

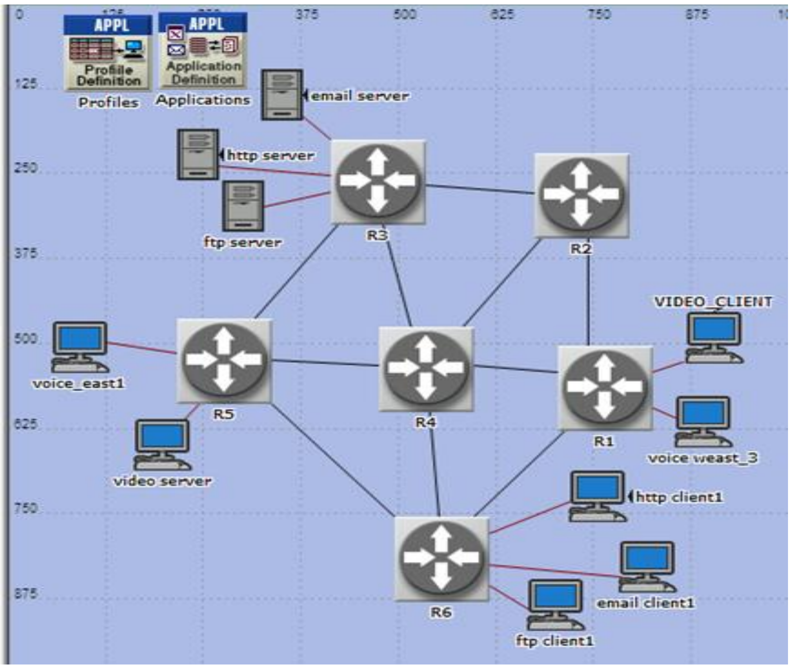

Fig. 1. Topology A (Small-Size) Network

Figure (2) shows a larger size network topology and is referred to throughout the context of this paper as the Big-Size network case study. This network although has similar elements listed above for the Small-Size network shown in Figure (1), It has fourteen routers (R1 to R14). Table (1) shows the OPNET application configurations that was adopted for each of the *low* and *High* traffic scenarios respectively.

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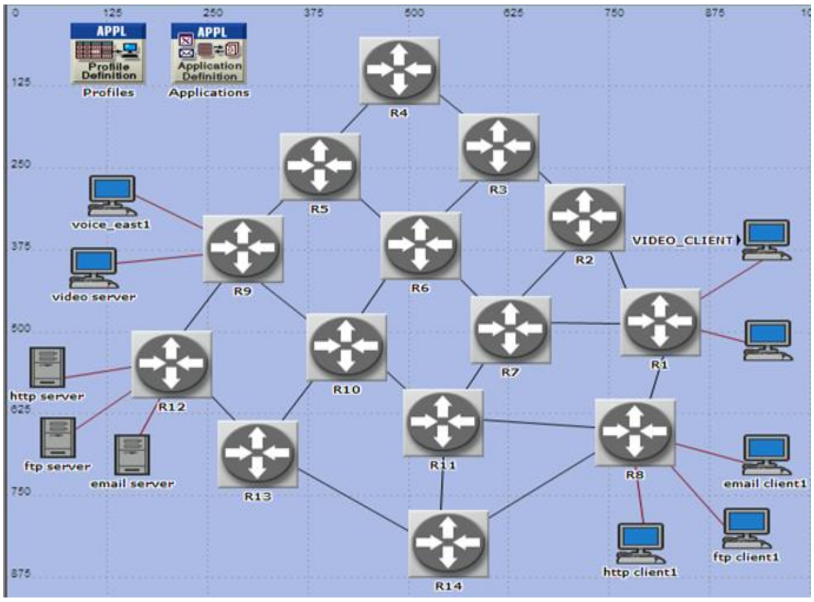

Fig. 2. Topology (Big-Size) Network

Table 1: OPNET Application Configurations

Low Traffic	
Email	Low Load
HTTP	HTTP, LIGHT BROWSING
VideoConferencing	128*120 PIXELS, 10 FRAMES/SEC
Voice	Low Quality Speech Encoder G.723 15.3 K
High Traffic	
Email	High Load
HTTP	HTTP, HEAVY BROWSING
VideoConferencing	128*240 PIXELS, 15 FRAMES/SEC
Voice	PCM Quality Speech Encoder G.711

4. Simulation Scenarios

Extensive simulation scenarios have been performed on each of the two topologies for a duration of half an hour (30 Minutes) for each scenario using OPNET network simulation tool. The main objective of this research work was to identify the right combination of a routing protocol and a queuing technique that yields the best Quality of Service (QoS) metrics for Voice Over IP networks (VOIP). To simulate realistic traffic environments and capture the complexity and dynamics of real-world IP networks, the simulations included various types of multimedia traffic (HTTP, FTP, and email) concurrently applied to the simulated network.

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Additionally, the study assessed the robustness of the results under varying network structures and levels of traffic intensity.

This objective and approach sound comprehensive and aligned with the aim of evaluating the impact of routing protocols and queuing techniques on QoS for VOIP traffic. By considering various types of multimedia traffic, the research acknowledges the importance of studying network performance in the presence of concurrent traffic. Additionally, investigating the consistency of findings across different network sizes and traffic loads enhances the generalizability and practical applicability of the research outcomes.

To achieve the objectives of this research study, the experimental simulation scenarios were designed to incorporate the following factors:

- Three different Interior Gateway Protocol (IGP) routing algorithms: RIP, OSPF, and EIGRP.
- Each routing algorithm was paired with each of the three distinct queuing scheduling techniques: FIFO, PQ, and WFQ.
- Multiple topologies were considered in all simulations.
- Each topology was subjected to both light and high traffic loads for all simulations.

The study employed a comprehensive approach, which encompassed four key dimensions: routing, queuing, topology, and traffic load density. This approach enabled the creation of multiple detailed scenarios, facilitating a thorough comparison of the performance of each queuing method with the three routing protocols, as well as the impact of each routing protocol with the three queuing techniques. Additionally, the scenarios encompassed different topologies, varying traffic loads, and focused specifically on VOIP streaming.

In this paper, due to space limits, emphasis is placed on presenting figures related to a specific network configuration known as the big-size topology (Topology B) under a particular traffic condition labelled as "Heavy traffic." While various other figures detailing simulation outcomes for different topologies and traffic conditions, although discussed, are not all included[1] in this paper. However, as an illustration of the research scope, a selection of actual figures demonstrating performance concerning a single Quality of Service (QoS) metric, specifically average voice packet delay variation, are given in figures Fig.3, Fig.4, Fig.5, and Fig.6. The remaining figures (Fig.7 to Fig.10) presenting figures related to big-size topology (Topology B) under a particular traffic condition labelled as "Heavy traffic", exhibit the average values for the respective Quality of Service (QoS) metrics, obtained by applying each routing protocol (RIP, OSPF, EIGRP) with each queuing technique (FIFO, PQ, WFQ). To

[1]Excluded figures can be provided by the authors upon request

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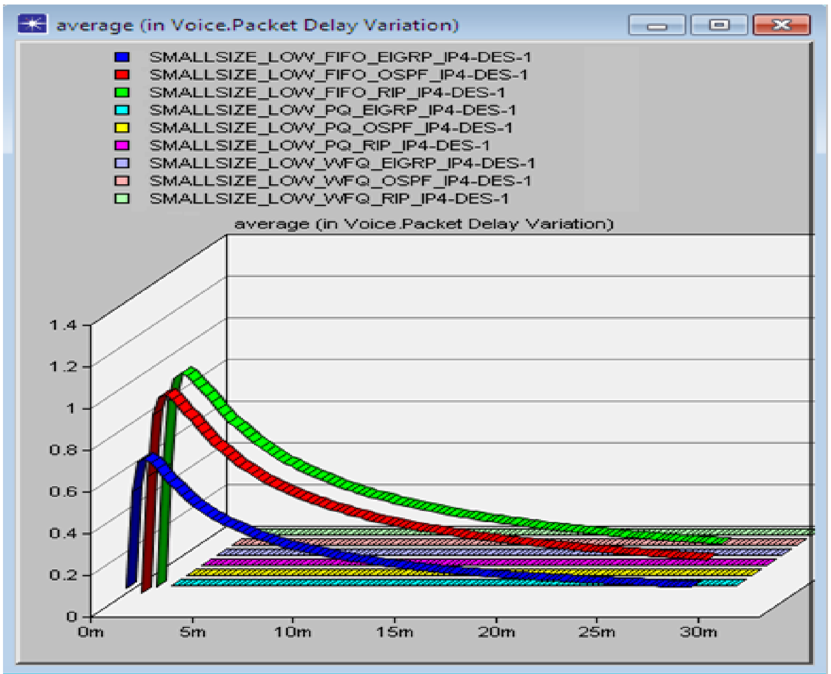
ensure clarity and easy reference, the corresponding findings are conveniently presented at the bottom of each figure

5. Simulation Results and Analysis

A. Average Delay Variation

The packet delay variation experienced in VoIP transmissions is shown in Fig.3 to Fig.6. It can be clearly seen that, when utilizing the PQ and WFQ queuing techniques across all scenarios, the packet delay variation approaches zero for all three routing protocols. Interestingly, the EIGRP protocol consistently outperforms RIP and OSPF in terms of packet delay variation across all queueing techniques. Furthermore, EIGRP and PQ queueing techniques in respective order exhibit the optimal performance, with regard to lowest delay variation among all tested configurations



Small-Size NW Low Load traffic	RIP	OSPF	EIGRP
FIFO	0.24940280	0.2240421	0.1396721
PQ	0.00001365	0.0000141	0.0000118
WFQ	0.00001428	0.0000146	0.0000126

Fig. 3. Average Delay Variation for Small-Size & Low Loaded NW (sec)

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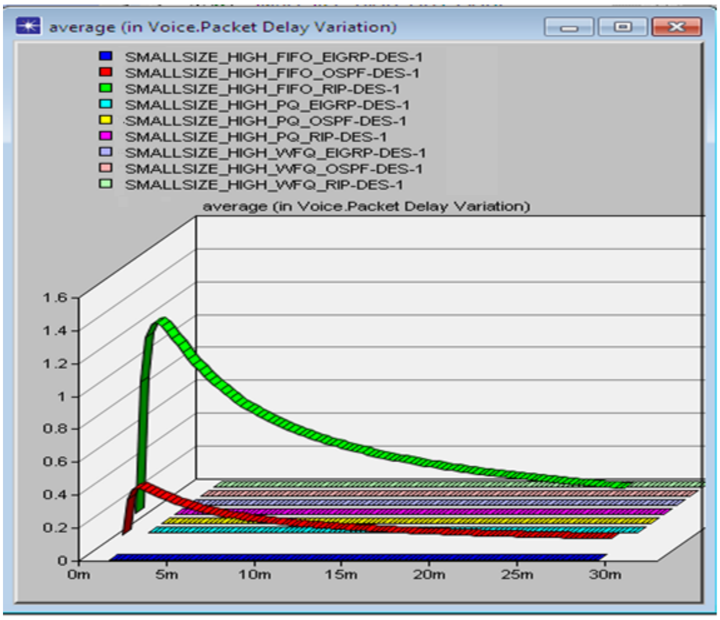
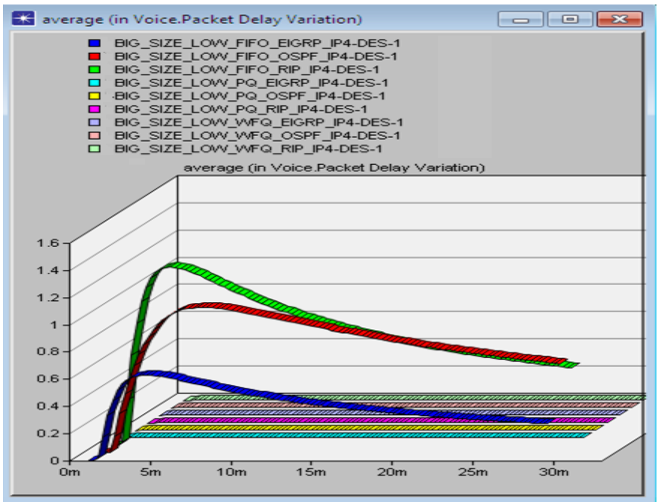


Small-Size NW High Load traffic	RIP	OSPF	EIGRP
FIFO	0.326599	0.0768581	0.000000081
PQ	0.000012	0.0000061	0.000000021
WFQ	0.000016	0.0000083	0.000000025

Fig. 4. Average Delay Variation for Small-Size & High Loaded NW (sec)



Big-Size NW Delay Variation	RIP	OSPF	EIGRP
FIFO	0.5710658	0.6567217	0.2714033
PQ	0.0000502	0.0000676	0.0000714
WFQ	0.0000511	0.0000177	0.0000251

Fig. 5. Average Delay Variation for Big-Size & Low Loaded NW (sec)

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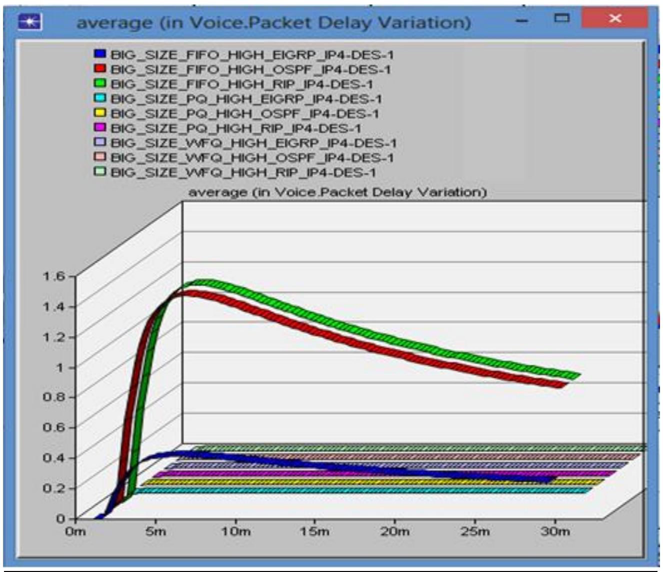

Big-Size NW Delay Variation	RIP	OSPF	EIGRP
FIFO	0.810314	0.804142	0.238442
PQ	0.000048	0.000037	0.0000245
WFQ	0.000054	0.000038	0.000028

Fig. 6. Average Delay Variation for Big-Size & High Loaded NW (sec)

B. End-to-End Delay

Fig.7 illustrates the end-to-end packet delay experienced by Voice over IP (VoIP) traffic across the simulated network scenarios. The graph clearly indicates that EIGRP outperforms RIP and OSPF across all employed queueing techniques. Additionally, the WFQ queueing technique shows slightly better performance compared to PQ for this specific QoS metric with all routing protocols (RIP, OSPF, EIGRP). As expected, FIFO exhibits the poorest performance in terms of packet end-to-end delay, leading to higher delays that could negatively impact call quality. This is because FIFO doesn't prioritize VoIP packets, leading to potential queuing behind other data types.

C. Variation in Packet Arrival Time (Jitter)

Figure (8) presents the variation in packet transmission delay, commonly referred to as jitter, observed in Voice over IP (VoIP) traffic. The graph demonstrates that packet delay variation tends to approach zero when using WFO and PQ queueing techniques in conjunction with all routing protocols (RIP, OSPF, and EIGRP). In contrast, FIFO exhibits noticeable packet delay variation. Additionally, it is evident that both of EIGRP and OSPF routing protocols in respective order perform expectedly well compared to RIP with all queueing techniques.

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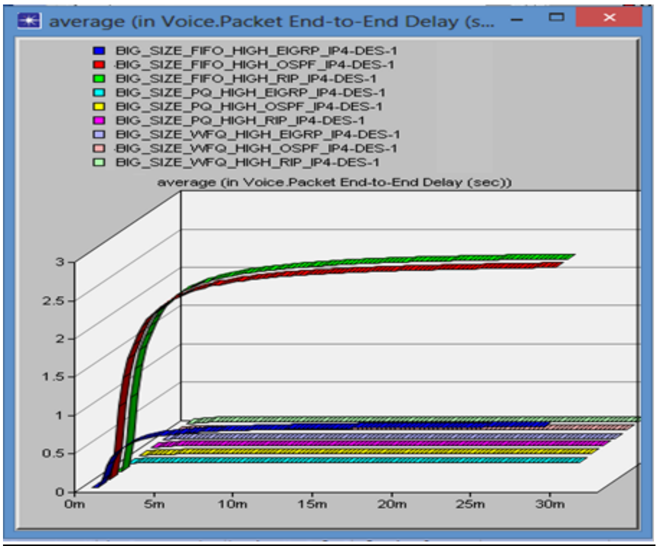
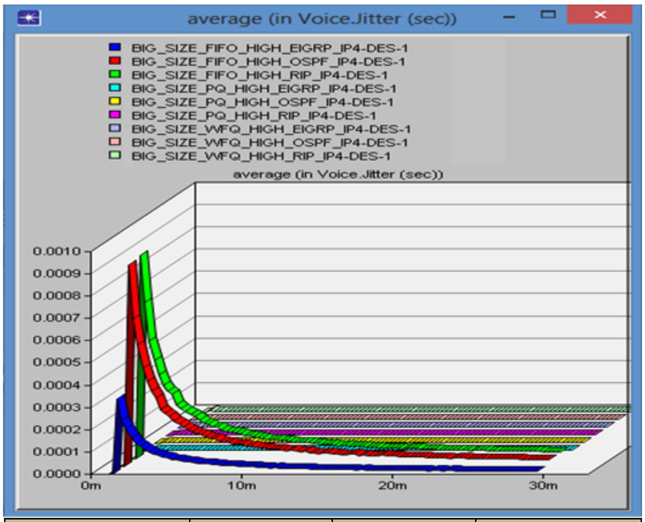


Big-Size NW End-to-End Delay High Load Traffic	RIP	OSPF	EIGRP
FIFO	2.83017	2.82221	0.84751
PQ	0.08881	0.08315	0.07594
WFQ	0.08798	0.08004	0.07513

Fig. 7. Average End-to-End Delay (sec)



Big-Size NW High traffic Jitter	RIP	OSPF	EIGRP
FIFO	0.000840921	0.000827722	0.000278990
PQ	0.000000156	0.000000111	0.000000074
WFQ	0.000000143	0.000000060	0.000000078

Fig. 8. Average Voice Jitter (sec)

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D. Average Sent and Received Traffic

Figure (9) and Figure (10) depict the average VOIP traffic sent and traffic received for different combinations of routing protocols and queueing techniques. While traffic sent remained consistent (~381.77 packets/sec) across all scenarios, the traffic received varied significantly, highlighting the impact of network configuration on VoIP delivery. From Figure (10), it is evident that the EIGRP protocol outperforms OSPF and RIP protocols in terms of the received traffic metric.

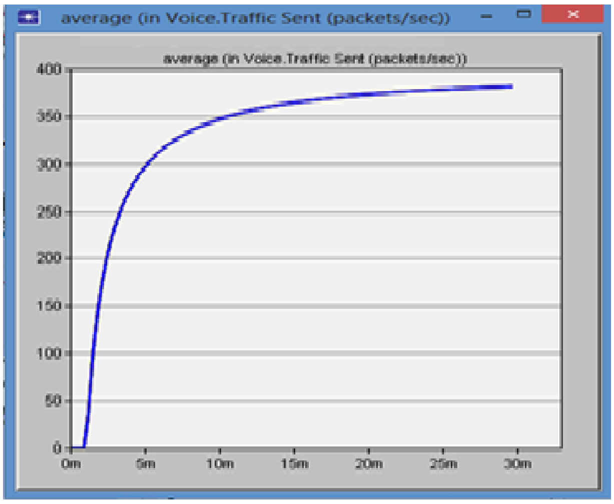
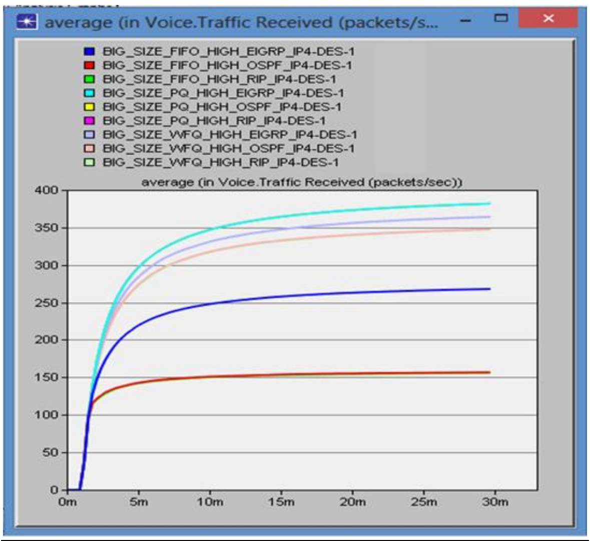


Fig. 9. Average traffic sent (packets/sec)



Bigl-Size NW Traffic Received	RIP	OSPF	EIGRP
FIFO	156.3667	156.6872	267.8883
PQ	381.7539	381.7794	381.7667
WFQ	347.1889	347.4389	364.1167

Fig. 10. Average traffic received (packets/sec)

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Additionally, PQ (Priority Queueing) demonstrates better performance compared to WFQ (Weighted Fair Queueing) in terms of the received traffic QoS metric. As expected, the FIFO queueing technique exhibits the poorest performance due to its lack of prioritization. In scenarios where the Brusly traffic dominates the queue, it can lead to dropped VoIP packets, resulting in significantly less successfully received data.

6. Conclusion

This paper has systematically examined the influence of queuing algorithms and routing protocols on the Quality of Service (QoS) for Voice over IP (VoIP) traffic across IP networks. Through comprehensive simulation scenarios, various dimensions were considered — including routing protocols, queuing techniques, network topology size, and traffic load density.

Across different phases and network scales (small and large), and varying traffic qualities, the study found no significant impact of topology size or traffic quality on the performance of queuing algorithms.

Empirical results consistently demonstrated that, under most simulation conditions, the Enhanced Interior Gateway Routing Protocol (EIGRP) outperformed Open Shortest Path First (OSPF) and Routing Information Protocol (RIP) in several key QoS metrics: packet loss, packet delay variation, end-to-end delay, received traffic volume, and jitter. These advantages held across both light and high traffic load scenarios.

Among queuing techniques, Weighted Fair Queueing (WFQ) and Priority Queueing (PQ) exhibited strong performance, with WFQ ranking highest. In contrast, First-In-First-Out (FIFO) showed the weakest performance due to its lack of prioritization, resulting in increased delays and VoIP packet loss.

The study emphasizes the critical need to prioritize VoIP traffic to maintain call quality and improve user experience. Notably, combining EIGRP with WFQ yielded optimal outcomes across nearly all tested metrics.

In conclusion, integrating EIGRP with WFQ presents the most effective approach for minimizing delay and maximizing QoS in VoIP communications. PQ also proved to be a valuable alternative. These insights offer actionable guidance for network administrators

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seeking to enhance VoIP traffic performance and overall communication quality in IP networks

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