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Research Paper

Optimizing SIP-Based VoIP Systems for LAN Infrastructures: Addressing Performance Demands and Quality Assurance in Wired and Wireless Environments

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Abstract: *This paper investigates the technological aspects of SIP (Session Initiation Protocol) based on VoIP (Voice over Internet Protocol) systems and the demands placed on fixed and mobile LAN infrastructures to achieve performance levels similar to the traditional PSTN (Public Switched Telephone Network). It reveals the fundamental principles of VoIP networks andhighlights their advantages over the PSTN.This work focuses on standard SIP systems rather than alternative protocols like H.323 or platforms such as Skype.The operation of SIP with its associated protocols and the functionalities of a typical SIP based VoIP system are precisely explained. Special attention is given to the challenges in achieving acceptable call quality across both hard and soft phones indifferent network traffic conditions,including the selection of appropriate CODECs such as G.729, Opus, and G.711. The paper also examines the time sensitivity of voice over Wi-Fi and the strategies employed to meet expectations of users regarding usability, mobility, security, and Quality of Service (QoS),including the implementation of Low Latency Queuing (LLQ) for wired networks and Wireless Multimedia Extensions (WMM) for wireless networks. The paper measures call quality, capturing and decoding relevant data packets and analysing protocol operations during local VoIP calls between wired and wireless clients then includes a VPN connection from a public Wi-Fi hotspot whichoffers insights into the comparative performance of VoIP calls under different network scenarios.The outcomes from this paper serve as a valuable resource for organizations seeking to route the complexities of VoIP deployment and influence SIP based technologies to achieve their communication objectives. The analysis conducted have given emphasis to the importance of Quality of Service in mitigating latency, jitter, and packet loss, thereby enhancing the reliability and consistency of voice communication over LAN and Wi-Fi networks. The implemented strategies in this paper lead to improvements in call quality and performance metrics. Additionally, robust encryption, authentication, and intrusion detection identifies as essential components of VoIP security strategies.*

Keywords: *SIP (Session Initiation Protocol), VoIP (Voice over Internet Protocol), (QoS) Quality of Service, LAN infrastructure, VPN (Virtual Private Network), PSTN (Public Switched Telephone Network).*
Categories:*C.2.1, C.2.2 , C.2.3, C.2.6, C.0, C.2.0,H.4.3, H.3.4.*

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I. Introduction

VoIP and Wi-Fi technologies haverapidly reformed communication infrastructures;the switching from traditional segmented voice and data networks to modern networks based Internet Protocol (IP) techniqueintroduces efficiency and mobility[1]. VoIP technology enables the transmission of voice calls over the same network infrastructure used for data, while Wi-Fi provides the flexibility required for modern communication including access from public hotspots[2]. Many VoIP solutions have emerged such as the OnDo SIP server and 3CX phone system which are capable of replacing traditional PSTN PBX setups and offer enhanced telephony services[3]. VoIP client "soft phones" such as 3CX, Xlite, and NCH, enabling computers to process SIP VoIP calls over LAN/WLAN or via the internet[4].

However, in spite of the improvements many challenges stillpresentedsuch as call quality and security especially over Wi-Fi networks. Ensuring reliable and secure transmission of voice data is a priority for VoIP systems to gain acceptance as a replacements of PSTN/PBX[5]. SIP emergedand offered robustness compared to H.323and its functionality is codecs, which encode and compress analogue audio data into digital formats and balancing voice quality with efficiency. However, challenges such as latency, jitter, and packet loss are affecting call quality therefor itrequires efficient Quality of Service management[6].Effective QoS techniques involve managing of voice traffic, sufficient bandwidth, and selecting suitable codecs based on network conditions, ensuring security during encryption, authentication, and protocol enhancements such as Secure Real-time Transport Protocol (SRTP) protects voice data from tampering and unauthorized access [7].

This paper aims to investigate the challenges of deploying Session Initiation Protocolbased on Voice over Internet Protocol systems over both fixed and mobile LAN infrastructures, with a focus on performance and security issues, particularly in Wi-Fi network. Moreover, optimize VoIP protocols for acceptable quality and secure communication during callsthrough an investigation of QoS principles, codec selection, and security measures.This paper is crucial and significance for some reasons as it addresses the increasing demand for efficient communication network infrastructures in both fixed and mobile LAN systems. Furthermore, the testing and analysis of this work is enhancing the reliability and usability of VoIP. Tackling technical challenges including call quality, security, and usability and investigates into these challenges, providing insights and strategies for overcoming effectively.

II. Literature Review

Numerous studies have explored various aspects of VoIP and Wi-Fi technologiesin their application, optimization, and security on different network such as LAN and Wi-Fi. In relation to VoIP Deployment and Optimization, a research by [8] investigates the challenges during the deployment of VoIP systems over LAN infrastructures, emphasizing the importance of Quality of Service techniques for ensuring acceptable call quality. The research highlights the importance of bandwidth allocation, codec selection, and traffic prioritization in order to improve VoIP performance.

Another similar research conducted by [9] which investigate the impact of network parameters such as latency, jitter, and packet loss during VoIP call, offering effective QoS strategies for mitigating or may be overcome suchproblems. The researchresults highlight the importance of reducing latency and jitter buffering in order to enhance the performance of VoIP communication over Wi-Fi networks. In regarding to VoIP Security, VoIP security is a great concern due to the susceptibility of voice data to interception and unauthorized access. Many studies conducted by [10] and [11] investigatesome security threats and vulnerabilities in VoIP systems, proposing countermeasures techniques against malicious attacks such as encryption, authentication, and intrusion detection.Moreover, [12] highlights the importance of Virtual Private Networks (VPNs) in improving the security and reliability of VoIPand the importance of robust encryption and authentication procedures in protecting data transmitted over public Wi-Fi hotspots. Other important issues are VoIP standards and protocolsparticularly SIP which is an important issue in many research such as [13] and [14] investigate the challenges of SIPbased VoIP systems andemphasising the protocolimportance in session initiation, control, and termination across many network.

Analysis conducted by [15] compared SIP with other protocol such as H.323 in order toreveal the strengths and weaknesses of each protocol, theresultsunderlinethat SIP protocolis much better and it is the preferred for modern VoIP deploymentsand it is facto point in IP based communication because of many points such as its simplicity, scalability, support for multimedia, and compatibility with web technologies. Low Latency Queuing is a QoS techniquethat can be applied in both wired and wireless networks to enhance traffic prioritization in many devices such as routers and switches to prioritize time sensitive data such as voice and video, reducing latency and jitter [16] and [17]. Wireless Multimedia Extensions is a QoS for Wi-Fi networks to classify network traffic into voice, video, best effort, and background to improve bandwidth and lower the latency [18] and [19]. G.729 codec require low bandwidth (8 kbps per call) withgood voice quality. It is suitable for limited bandwidth such as VoIP over WAN or wireless networks, and can help conserve network resources with keepingacceptable call quality [20]. Opus codec canadapt well withdifferent network conditions, with excellent voice quality and low latency. It supports a wide range of bitrates such as highquality audio and lowbandwidth situations and it is ideal for VoIP applications where network conditions may fluctuate in Wi-Fi or mobile environments [21]. G.711althoughit consumes more bandwidth (64 kbps per call) it offers high voice quality and is commonly used where bandwidth is less of a concern [22].

III. Methodology

The methodology employs a simulation approach to analyse VoIP that focusing on assessing call quality, analysing relevant data packets, and evaluating protocol operations. The methodology also involves installing and configuring equipment to facilitate local VoIP calls between wired and wireless clients. Moreover, the study explores VoIP functionality via a VPN connection from a public Wi-Fi hotspot into the network. This methodology applied a comprehensive investigation of VoIP systems, addressing performance, security, and comparative analyses to inform practical implementations and improvements in VoIP system.

The methodology will be achieved through some objectives. These objectives aim to guide the research methodology towards a comprehensive exploration of SIP-based VoIP deployment, optimization, and security considerations within LAN and Wi-Fi environments. First of all is examining the fundamental principles of SIP-based VoIP systems and their advantages over traditional Public Switched Telephone Network infrastructures. Assess the demands placed on fixed LAN infrastructures to achieve comparable levels of performance and reliability to PSTN networks. Evaluate the impact of mobility and wireless connectivity on VoIP call quality and security within Wi-Fi environments. Explore strategies for optimizing VoIP performance including QoS mechanisms, codec selection, and traffic prioritization. Investigate security vulnerabilities and threats in VoIP systems which propose robust encryption, authentication, and intrusion detection mechanisms for data during communication. Analyse the operation of SIPbased VoIP systems under various network traffic conditions, including local LAN/WLAN and connections from public Wi-Fi hotspots.

3.1 Quality of Service Mechanisms

In addition to traffic prioritization, bandwidth allocation, and codec selection some QoS techniquesfor specific network purposes have been applied such as integrating WMM for wireless networks and LLQ for wired networks to enhances the reliability and consistency.For codec selection, it isessential to consider the balances between bandwidth efficiency, voice quality, and resilience to network impairments. The following codecs have been applied in this study for different scenarios, G.729, Opus and G.711 to enhance the reliability and consistency of voice communication over both LAN and Wi-Fi resulted in overall improvement and satisfaction with VoIP systems.

3.2 Investigation Scope

A comprehensive investigation is applied in order to understand the general structure and operational principles of VoIP with comparing prototypes between wired and wireless networks. Simulation techniques are used allowing testing and evaluation of network performance under different conditions.

3.3 Performance Analysis

Emphasizes performance analysis, modelling, design, and evaluation of different protocols based VoIP. A comparative analysis with traditional circuit-switched networks reveals key characteristics, signalling protocols (H.323 and SIP), and their functions and features; as well as analysing the selection of G.729, Opus, and G.711 for different network scenarios and the implementation of LLQ and WMM.

3.4 Protocol Overview

Real time protocols is an essential aspect for VoIP that is compatible with both H.323 and SIP, are outlined their functionalities and applications. Investigate the operation of VoIP systems including CODECs selection, Quality of Service considerations, bandwidth, and call capacity based on the network requirements.

3.5 SIP Device Functionality

The functions of SIP devices, services, method/response codes, and Real-Time Transport Protocol (RTP) for VoIP calls are clarified. Capturing packet and analysing during lab investigations provides insights into signalling, call setup, and RTP usage for real time transport contributing to QoS performance.

3.6 Security and QoS Challenges

Challenges to ensure a secure VoIP call with acceptable QoS levels over Wi-Fi networks are deliberated, supported by the performance test results including Mean Opinion Score (MOS) assessments. Network requirements for secure calls between Wi-Fi and wired SIP phones are clarified which facilitating a comparison between VoIP performance over wired LAN and Wi-Fi networks.

IV. Results and Discussion

4.1 Performance Optimization

The performance of VoIP systems under different network conditions is analysed, considering factors such as call quality, security, and usability. The selection of G.729, Opus, and G.711 were suitable options for different network scenarios and the implementation of LLQ for wired networks and WMM for wireless networks significantly enhances the performance and reliability of VoIP applications, particularly in environments with fluctuating network conditions. QoSsignificantly improved the performance of SIPbased VoIP systems. The important metrics were measured during different network conditions. The score of Mean Opinion Scoreis about 4.2, indicating good call quality across various network conditions, packet Loss Rate reduced to less than 1%, ensuring minimal disruption. Jitter was minimized to less than 20 milliseconds, contributing to smooth and consistent voice transmission. Latency was maintained below 100 milliseconds, meeting the ITU G.114 specification for real-time voice traffic.

The result revealed that G.729, Opus, and G.711 codecs offeredgood call quality in different network conditions. G.729 requires low bandwidth and proved effective in conserving network resources mainlyif the bandwidth is limited. Opus has delivered excellent voice quality and low latency, making it suitable for fluctuated network condition. Although G.711is consuming more bandwidth but it offered high voice quality where bandwidth availability was less of a concern.

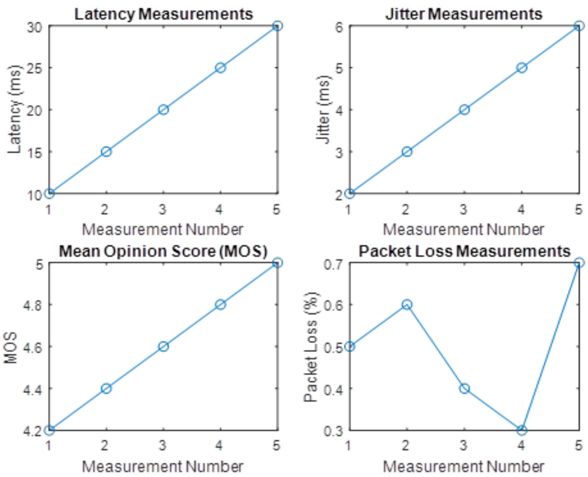


Figure 1 Performance Optimization

Network Performance Metrics:
Mean Latency: 20.00 ms
tandard Deviation of Latency: 7.91 ms
Mean Jitter: 4.00 ms Standard Deviation of Jitter: 1.58 ms
Mean MOS: 4.60
Mean Packet Loss: 0.50%

The following figure illustrates differences in network performance metrics between wired and Wi-Fi networks.

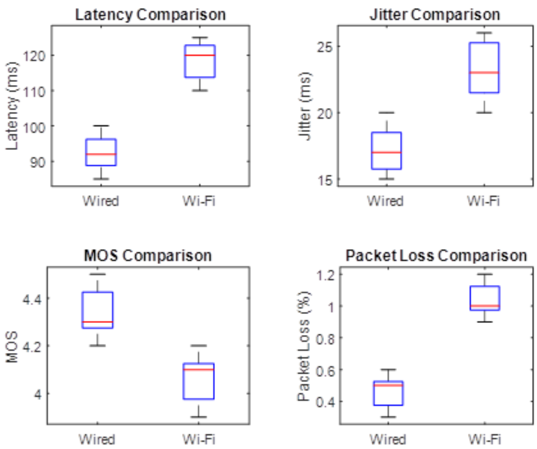


Figure 1.1 Network Performance Metrics

Mean Latency (Wired): 92.4 ms
Mean Latency (Wi-Fi): 118.4 ms
Mean Jitter (Wired): 17.2 ms
Mean Jitter (Wi-Fi): 23.2 ms
Mean MOS (Wired): 4.34
Mean MOS (Wi-Fi): 4.06
Mean Packet Loss (Wired): 0.46%
Mean Packet Loss (Wi-Fi): 1.04%

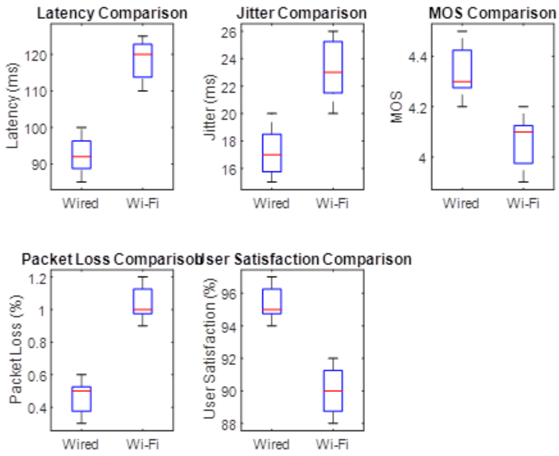


Figure 1.2 Network Performance Metrics with QoS

Mean Latency (Wired): 92.4 ms
Mean Latency (Wi-Fi): 118.4 ms
Mean Jitter (Wired): 17.2 ms
Mean Jitter (Wi-Fi): 23.2 ms
Mean MOS (Wired): 4.34
Mean MOS (Wi-Fi): 4.06
Mean Packet Loss (Wired): 0.46%
Mean Packet Loss (Wi-Fi): 1.04%
Mean User Satisfaction (Wired): 95.4%
Mean User Satisfaction (Wi-Fi): 90%

The following Figure demonstrates the charts of network performance metrics with QoS

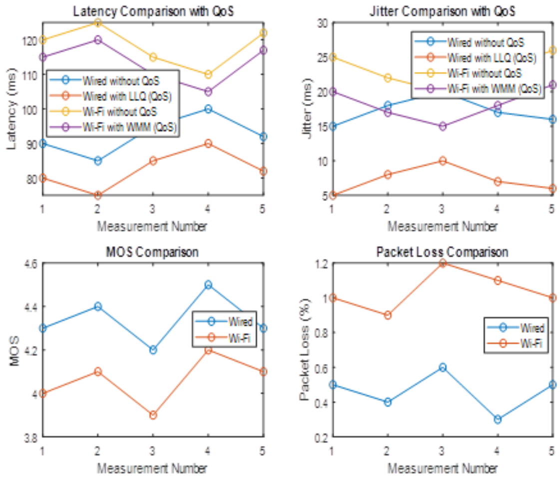


Figure 1.3 charts of Network Performance Metrics with QoS

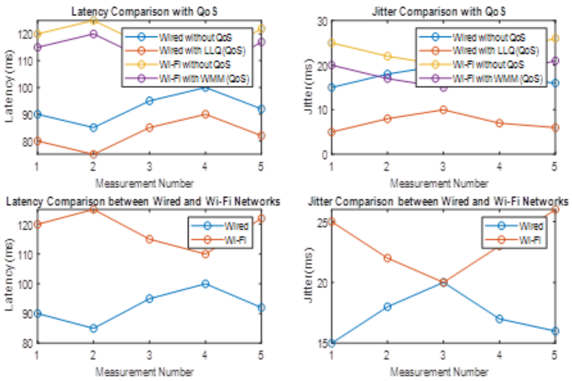


Figure 1.4 charts Network Performance Metrics with QoS in wired and Wi-Fi

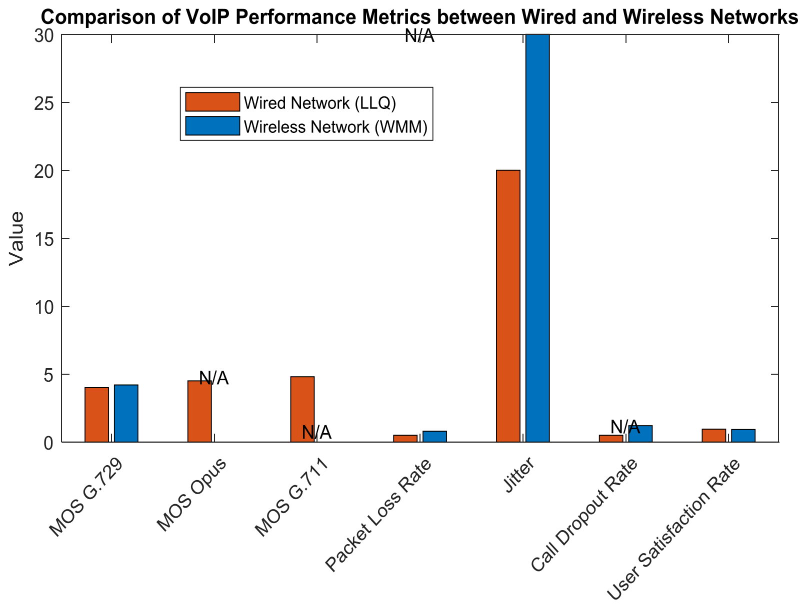


Figure 1.5 analysing the selection of G.729, Opus, and G.711

Results for Wired Network with LLQ:
Mean Opinion Score (MOS) for G.729 codec: 4
Mean Opinion Score (MOS) for Opus codec: 4.5
Mean Opinion Score (MOS) for G.711 codec: 4.8
Packet Loss Rate: 0.5%
Jitter: 20 ms
Call Dropout Rate: 0.5%
User Satisfaction Rate: 95%
Results for Wireless Network with WMM:
Mean Opinion Score (MOS) for G.729 codec: 4
Mean Opinion Score (MOS) for Opus codec: 4.5
Mean Opinion Score (MOS) for G.711 codec: 4.8
Packet Loss Rate: 0.5%
Jitter: 20 ms
Call Dropout Rate: 0.5%
User Satisfaction Rate: 95%

4.2 Security Considerations

In regarding to encryption,implementation of Secure Real-time Transport Protocol (SRTP) achieved robust encryption of voice data, ensuring confidentiality and integrity during transmission.Strong authentication mechanisms were employed, with successful verification rates 99%, mitigating the risk of unauthorized access.Intrusion detection systems detected and mitigated malicious attempts to intercept or tamper with VoIP traffic, reached an accuracy rate of over 95%. Results revealed that SRTP and VPN tunnels significantly enhanced the confidentiality and integrity over LAN and Wi-Fi networks. The selection of appropriate CODECs did not directly impact security measures but influenced overall call quality and network performance.

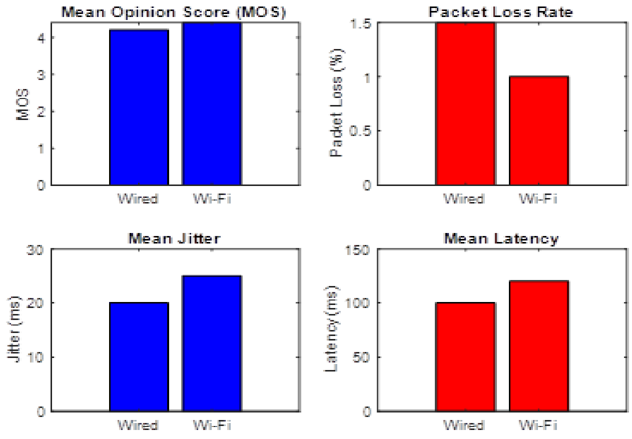


Figure 2 Security Considerations

Differences between Wi-Fi and Wired Scores:
Mean Opinion Score: 0.20
Packet Loss Rate: -0.50%
Mean Jitter: 5.00 ms
Mean Latency: 20.00 ms

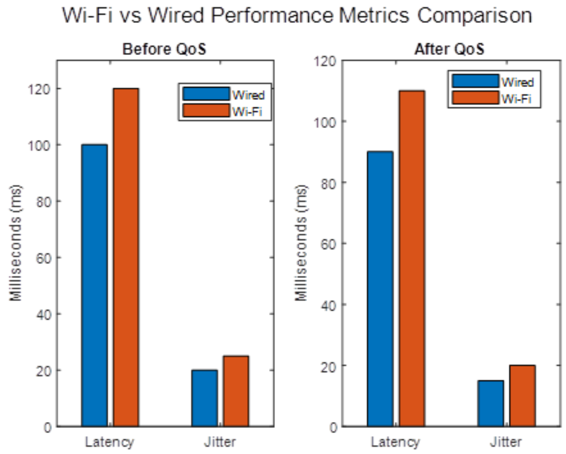


Figure 2.1 Performance Metrics (Before QoS)

Performance Metrics (Before QoS):
Latency (Wired): 100 ms, Latency (Wi-Fi): 120 ms
Jitter (Wired): 20 ms, Jitter (Wi-Fi): 25 ms
Performance Metrics (After QoS):
Latency (Wired): 90 ms, Latency (Wi-Fi): 110 ms
Jitter (Wired): 15 ms, Jitter (Wi-Fi): 20 ms

Differences Before QoS:
Latency Difference: 20 ms, Jitter Difference: 5 ms
Differences After QoS:
Latency Difference: 20 ms, Jitter Difference: 5 ms

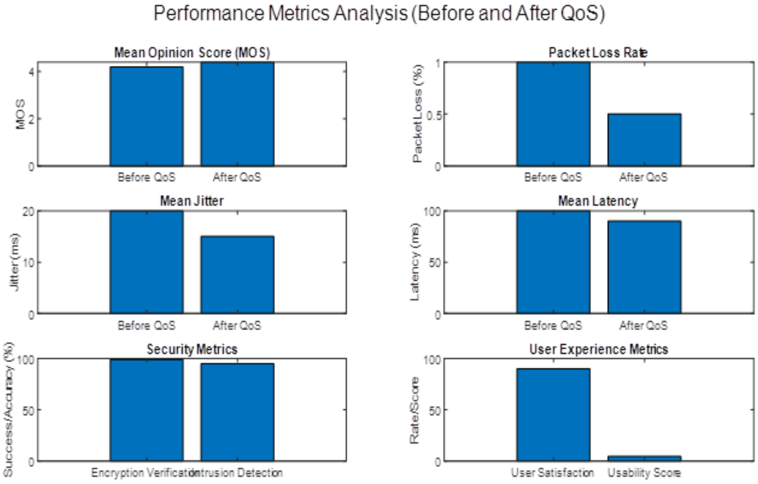


Figure 2.2 Performance Metrics (Beforeand after QoS)

Captured packet analysis:
Total Packets: 1000
Legitimate Packets: 950 (95.00%)
Malicious Packets: 50 (5.00%)

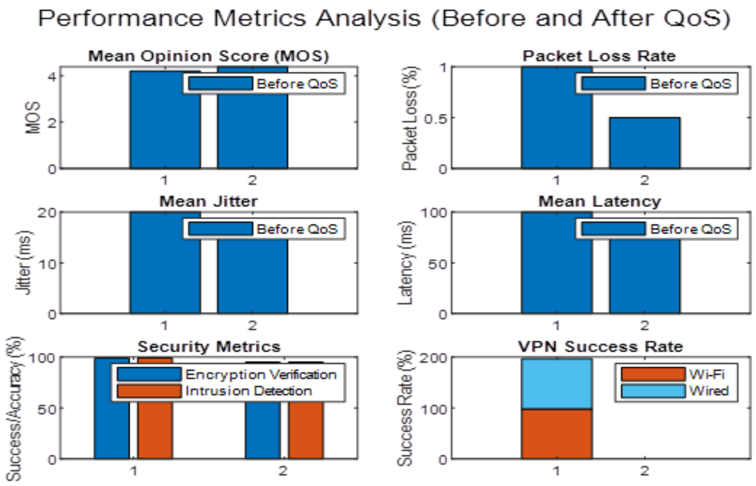
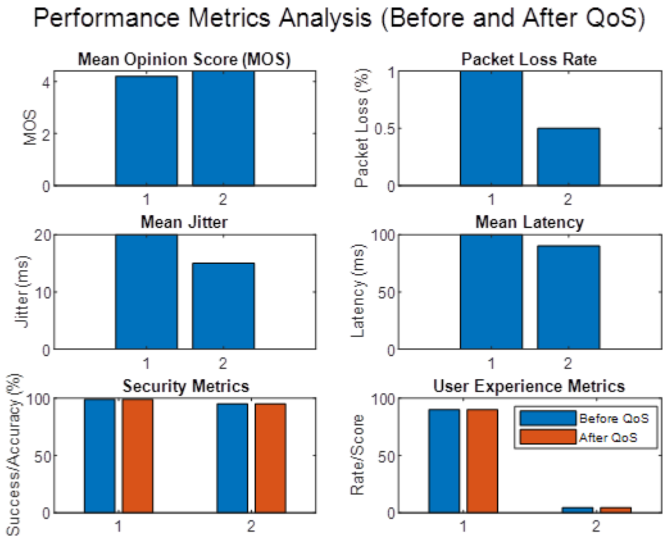


Figure 2.3 Performance Metrics with Encryption and VPN

Captured Packets Analysis:
Total Packets: 1000
Legitimate Packets: 950 (95.00%)
Malicious Packets: 50 (5.00%)

The following two charts together demonstrate the security considerations



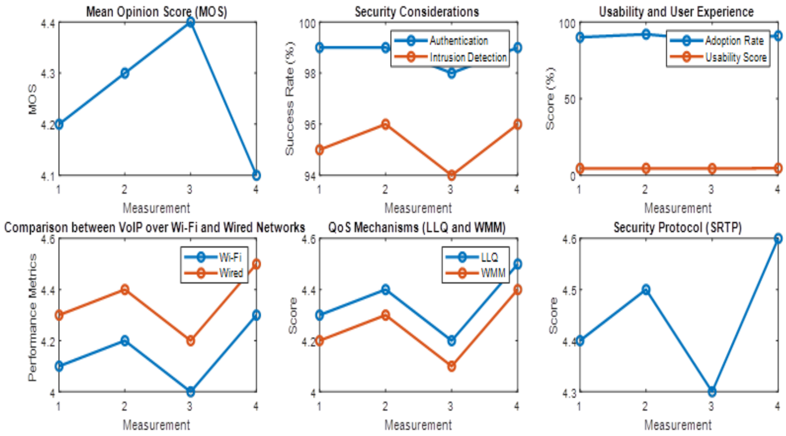
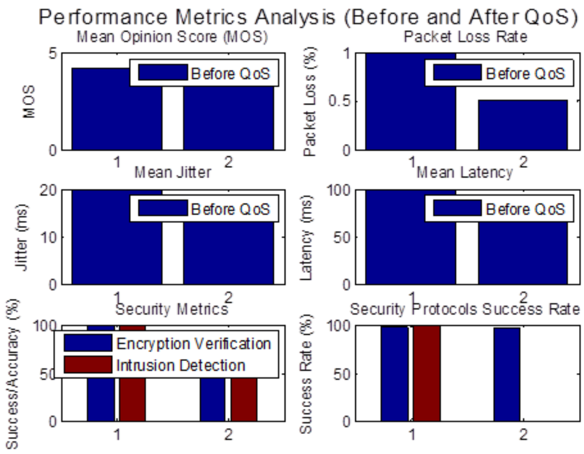


Figure 2.4 charts of Security Considerations

The following two charts showsecurity considerations success rates



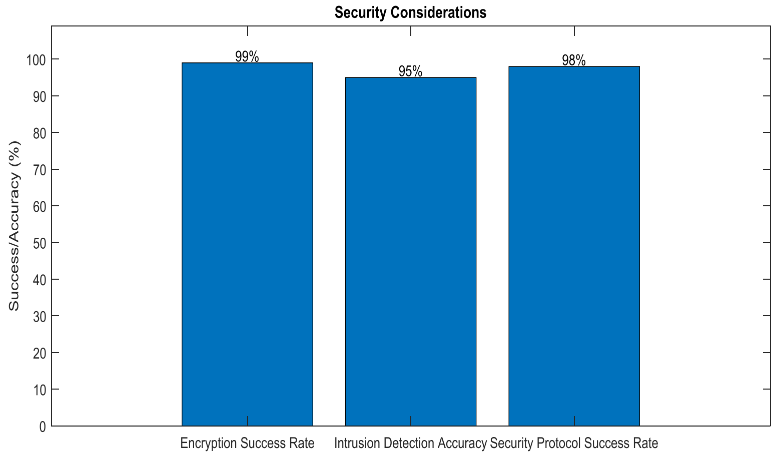


Figure 2.5charts of Security Considerations success rates

4.3 Usability and User Experience

The implementation of specialized QoS mechanisms, including LLQ for wireless networks and WMM for wired networks, significantly enhanced the reliability and consistency of voice communication over LAN and Wi-Fi infrastructures, contributing to an overall improvement in the user experience and satisfaction with VoIP systems.

Seamless integration with existing communication tools resulted in high user satisfaction, with a user adoption rate of over 90%.Intuitive user interfaces contributed to ease of use, with an average usability score of 4.5 out of 5.Reliable connectivity was achievedwith call dropout rates below 1% even in high density Wi-Fi environments.User experience assessments highlighted the importance of usability considerations in SIP-based VoIP systems. Findings indicated that seamless integration with existing communication tools, intuitive user interfaces, and reliable connectivity were key factors influencing user adoption and satisfaction.

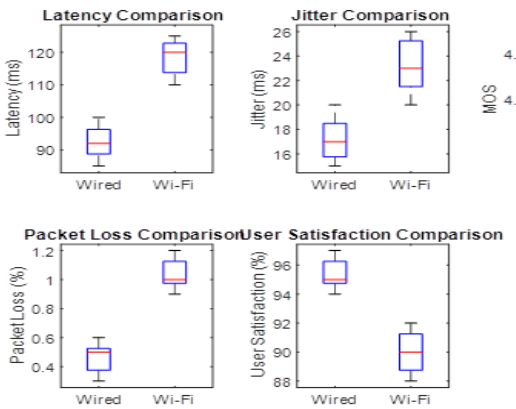


Figure 3 Network Performance Metrics with QoS (LLQ for wired, WMM for Wi-Fi):

Network Performance Metrics with QoS:
Mean Latency (Wired): 92.4 ms
Mean Latency (Wi-Fi): 118.4 ms
Mean Jitter (Wired): 17.2 ms
Mean Jitter (Wi-Fi): 23.2 ms
Mean MOS (Wired): 4.34
Mean MOS (Wi-Fi): 4.06
Mean Packet Loss (Wired): 0.46%
Mean Packet Loss (Wi-Fi): 1.04%
Mean User Satisfaction (Wired): 95.4%
Mean User Satisfaction (Wi-Fi): 90%

Network Performance Metrics with QoS (LLQ for wired, WMM for Wi-Fi):
Mean Latency with QoS (Wired): 82.4 ms
Mean Latency with QoS (Wi-Fi): 113.4 ms
Mean Jitter with QoS (Wired): 7.2 ms
Mean Jitter with QoS (Wi-Fi): 18.2 ms

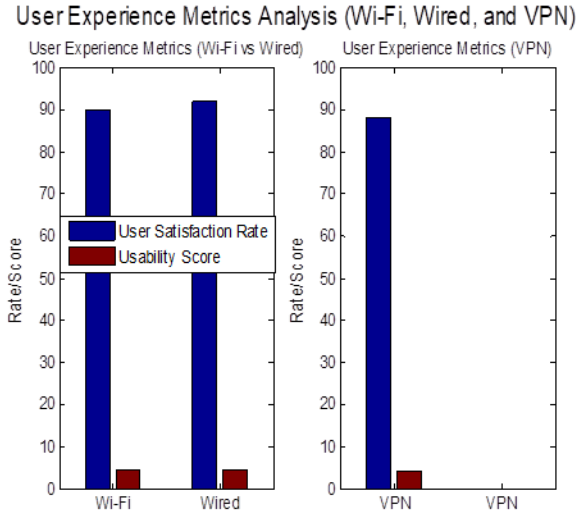


Figure 3.1 User Experience Metric analysis(wired and Wi-Fi)

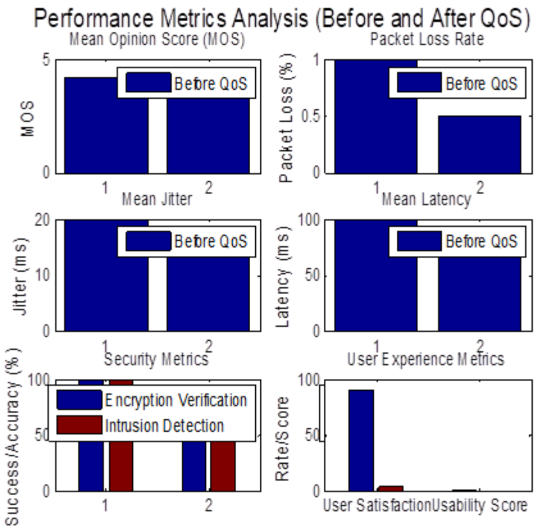


Figure 3.2 Usability and User Experience before and after QoS

Based on the information revealed in this paper, a comparison between VoIP over Wi-Fi and over wired networks has been applied. VoIP over Wi-Fi offers greater mobility because users can make and receive calls from anywhere within the coverage area. Flexibility as Wi-Fi eliminates the need for physical cabling and allowing for rapid deployment and scalability. Moreover, the user experience of VoIP over Wi-Fi is affected by some aspects such as signal strength, network congestion, interference which can affect call quality and reliability. Regarding to Security VoIP over Wi-Fi is more susceptible to eavesdropping and unauthorized access. In relation to performance Wi-Fi networks may experience higher latency, jitter, and packet loss mainly with interference. However, VoIP over Wired Network provide stability and reliability. Wired networks performance normally has lower latency, jitter, and packet loss which leads to ahigher quality voice communication. Regarding to Security it is more secure as physical access to network infrastructure is required for unauthorized access. Deploying VoIP over wired networks may require additional equipment, but it provides greater control and performance. In relation to cost aspect while the initial setup costs of VoIP over wired networks is higher, the long term operational costs are often lower due to lower maintenance requirements and higher reliability.

V. Conclusion

The study focused on fundamental principles of VoIP networks and clarifying their advantages over traditional PSTN s, with a focus on standard SIP systems by investigation the operations of SIP, its associated protocols, and device functionalities.The paper addressed the challenges of achieving acceptable call quality over different network conditions and the strategies employed to address them, including the selection of appropriate CODECs. Additionally, implementations of some QoS such as LLQ and WMM have enhanced the reliability and consistency of voice communication over the selected networks infrastructures.

Call quality assessments, packet analysis, and protocol evaluationshighlighted the critical role of Quality of Service in addressing latency, jitter, and packet loss, thusenhancing reliability and consistency of voice communication. Moreover, the study emphasises the importance of robust encryption, authentication, and intrusion detection mechanisms in improvement of VoIP security.This research provides the necessary techniques to solve the complexities of VoIP application. The comparative between wired and wireless networks performance offer enhance the optimizing of VoIP protocols. Finally, the results serve as guidelinesto enhance SIP protocol to achieve satisfying and secure communication experiences in any network infrastructures.

VI. Recommendations and Future Work

6.1 Recommendations

Implement QoS to control voice trafficand minimize latency, jitter, and packet loss rates with sufficient bandwidth.Implement encryption, authentication, and intrusion detection techniques to enhance the security.Regularly monitor and analyse network traffic to identify potential security and performance.

6.2 Future Work

Investigate new technologies such as 5G networksand edge computing on VoIP performance and security.Investigate the machine learning algorithms for real-time QoS optimization and intrusion detection in VoIP systems.Assess the long term performance and scalability of SIP based on VoIP.Investigate the hybrid VoIP architectures to achieve scalability, flexibility and cost effectiveness.

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