Quality of Service Evaluation of a Trixbox-Based VoIP System at Sultan Iskandar Muda Meteorological Station, Banda Aceh

Devi Wulandari[1], Yeni Yanti[2*], Elvitriana[3]

[1,2] *Department of Computer Engineering, Faculty of Engineering, Universitas Serambi Mekkah, Banda Aceh, Indonesia*

[3] *Department of Environmental Engineering, Faculty of Engineering, Universitas Serambi Mekkah, Banda Aceh, Indonesia*

Keywords:

VoIP; Trixbox; QoS; Wireshark; MOS; Wireless Network

Correspondence Email:
yeniyanti@serambimekkah.ac.id

Abstract

Reliable voice communication is essential for supporting operational coordination in government institutions, particularly in meteorological services that require real-time information exchange. Voice over Internet Protocol (VoIP) has emerged as an efficient communication solution due to its low deployment cost and flexibility; however, its performance is highly dependent on wireless network quality. This study evaluates the Quality of Service (QoS) performance of a Trixbox-based VoIP communication system implemented at the Sultan Iskandar Muda Meteorological Station Banda Aceh. An experimental method with a quantitative approach was employed to measure five QoS parameters, namely throughput, delay, jitter, packet loss, and Mean Opinion Score (MOS). Packet data were captured using Wireshark under four testing scenarios based on two communication distances (10 m and 30 m) and two testing periods (morning and afternoon). The measured QoS parameters were evaluated according to the TIPHON and ITU-T standards. The experimental results show that increasing the communication distance leads to higher delay, jitter, and packet loss, while throughput and MOS tend to decrease, particularly for the client located farther from the access point. Nevertheless, the measured throughput remained above 75 kbps, delay was consistently below 150 ms, and most MOS values exceeded 4.0, indicating Good to Very Good voice quality. Overall, the Trixbox-based VoIP system demonstrated reliable communication performance under the evaluated operational conditions and satisfied the recommended QoS requirements for real-time voice communication. The findings provide practical guidance for deploying wireless VoIP systems in government institutions and contribute empirical evidence regarding VoIP performance in operational meteorological environments.

1. Introduction

The rapid evolution of Internet Protocol (IP)-based communication technologies has fundamentally transformed modern telecommunication systems by providing flexible, scalable, and cost-effective alternatives to conventional Public Switched Telephone Networks (PSTN)[1]. Among these technologies, the Voice over Internet Protocol (VoIP) has become one of the most widely adopted communication solutions because it enables voice transmission through packet-switched networks while utilizing the existing IP infrastructure[2]. Compared with traditional telephony, VoIP significantly reduces communication costs, simplifies network administration, supports service scalability, and facilitates integration with other digital communication services [3]. These advantages have encouraged its widespread implementation in educational institutions, healthcare services, government agencies, and enterprise environments, where reliable internal communication is essential[4], [5].

In government organizations, communication quality directly affects operational efficiency and service delivery. This requirement is particularly important in meteorological institutions, where continuous coordination among observation stations, forecasting divisions, aviation meteorological services, and disaster early warning units depends on reliable real-time voice communication. Any degradation in communication quality may delay information exchange and reduce operational responsiveness in critical situations[6], [7], [8]. Consequently, many government institutions have adopted VoIP technology as an economical and flexible communication platform. At the Sultan Iskandar Muda Meteorological Station Banda Aceh, internal communication is implemented using the Trixbox IP-PBX platform over a Wireless Local Area Network (WLAN)[9], [10], allowing communication among operational units without relying on conventional telephone infrastructure. However, because voice packets share the same wireless medium as data traffic, communication quality is highly influenced by signal strength, transmission distance, network utilization, and wireless interference [11], [12]. Therefore, evaluating the Quality of Service (QoS) of the deployed VoIP system is essential to ensure that communication performance satisfies operational requirements[13].

The (QoS) is widely recognized as the primary performance indicator for evaluating VoIP communication systems[14]. According to ITU-T Recommendations G.114 and G.107, acceptable VoIP performance should be evaluated using several complementary parameters, including throughput, delay, jitter, packet loss, and Mean Opinion Score (MOS)[1]. Throughput represents the capability of the network to deliver voice packets efficiently, whereas delay and jitter determine the responsiveness and continuity of real-time communication. Packet loss directly influences speech intelligibility because missing RTP packets may cause audio distortion or conversational interruptions[15]. Meanwhile, MOS provides a subjective measurement of perceived voice quality based on users' listening experience[16]. Because these QoS parameters collectively determine communication quality, comprehensive evaluation is necessary before VoIP systems are deployed for operational communication services.

Numerous studies have investigated VoIP performance under different network environments[17]. It has been reported that stable throughput, low delay, and minimal jitter significantly improve the RTP packet transmission efficiency in wireless VoIP communication[12]. Similarly, demonstrated that increasing the communication distance between wireless clients and access points degrades VoIP quality owing to increased packet loss and reduced signal stability[18]. Other researchers have evaluated Trixbox- and Asterisk-based VoIP systems by investigating the influence of codec selection, bandwidth allocation, wireless interference, and traffic congestion on communication quality[19].These studies consistently concluded that network performance strongly affects voice quality, particularly in wireless communication environments where channel conditions continuously fluctuate.

Despite these valuable findings, there are several limitations to the current literature. First, most previous investigations were conducted in laboratory environments, university campuses, or enterprise testbeds with relatively controlled network conditions, making it difficult to generalize their findings to operational government environments [[20]. Second, many studies have evaluated only selected QoS parameters, such as delay or packet loss, without simultaneously examining throughput, delay, jitter, packet loss, and MOS under identical experimental conditions[21]. Third, empirical investigations focusing on VoIP deployment in meteorological institutions remain scarce despite their dependence on reliable real-time communication for weather monitoring, aviation support, and disaster information dissemination[22]. Furthermore, only a few studies have analyzed the combined effects of communication distance and daily network utilization periods on VoIP performance in wireless operational environments. These limitations indicate the need for a more comprehensive QoS evaluation in an actual government operational setting.

To address these research gaps, this study presents a comprehensive QoS evaluation of a Trixbox-based VoIP communication system deployed at the Sultan Iskandar Muda Meteorological Station Banda Aceh. The evaluation was performed under actual operational conditions using two communication distances (10 m and 30 m) and two different network utilisation periods (morning and afternoon). Five QoS parameters—throughput, delay, jitter, packet loss, and Mean Opinion Score (MOS)—are simultaneously analysed to evaluate communication performance under varying wireless conditions. Experimental results indicate that increasing the communication distance reduces throughput and MOS while increasing delay, jitter, and packet loss, although the overall communication quality remains within acceptable QoS limits for operational VoIP services. These findings provide practical evidence regarding the influence of wireless network conditions on VoIP communication quality in real operational environments.

The contributions of this study are summarised as follows:

1. This paper presents one of the few empirical evaluations of a Trixbox-based VoIP system implemented in an operational meteorological institution rather than in a laboratory or campus environment.
2. This study performs a comprehensive QoS assessment by simultaneously analysing throughput, delay, jitter, packet loss, and Mean Opinion Score (MOS) under multiple communication scenarios involving different transmission distances and network utilisation periods.
3. This study provides empirical evidence regarding the influence of wireless communication distance and network conditions on VoIP performance, thereby offering practical deployment recommendations for government institutions that implement wireless VoIP communication systems.

2. Research Methods

This study employs an experimental method with a quantitative approach to analyze the quality of Voice over Internet Protocol (VoIP) communication services using Trixbox (Fig.1). This study employed an experimental method with a quantitative approach. The experimental method was selected because the research involved a series of experiments conducted on a Voice over Internet Protocol (VoIP) communication system using a Trixbox server to obtain data on network service quality. The quantitative approach was adopted because the collected data consisted of numerical values representing the Quality of Service (QoS) parameters, namely delay, jitter, packet loss, and throughput, as well as the Mean Opinion Score (MOS). These quantitative measurements were subsequently analyzed to evaluate the performance of the VoIP communication system.

The research was carried out through several sequential stages, including literature review, field observation, research equipment preparation, Trixbox server configuration, VoIP communication testing, packet capture using Wireshark, QoS and MOS measurement, data analysis, and conclusion drawing. All research stages were conducted systematically to ensure that the experimental results objectively represented the service quality of the VoIP communication system.

The data analysis process involved calculating the values of each Quality of Service (QoS) parameter, namely delay, jitter, packet loss, and throughput, based on the VoIP communication data captured using Wireshark. These calculated values were then compared with the network service quality standards defined by TIPHON and ITU-T to determine the corresponding QoS performance category. Furthermore, the Mean Opinion Score (MOS) was calculated using the standard MOS evaluation method to assess the perceived voice quality during VoIP communication. Finally, the QoS and MOS results were analyzed descriptively to evaluate the overall performance of the Trixbox-based VoIP communication syste

2.1 Hardware and Software Specifications

The hardware and software used in this study to perform the VoIP system testing are listed in Table. 1 and Tabel. 2, respectively. The hardware specifications include the server, client computers, networking devices, and other supporting equipment, while the software specifications consist of the operating system, Trixbox server, Wireshark, softphone application, and other software required for system configuration, packet capture, and Quality of Service (QoS) analysis

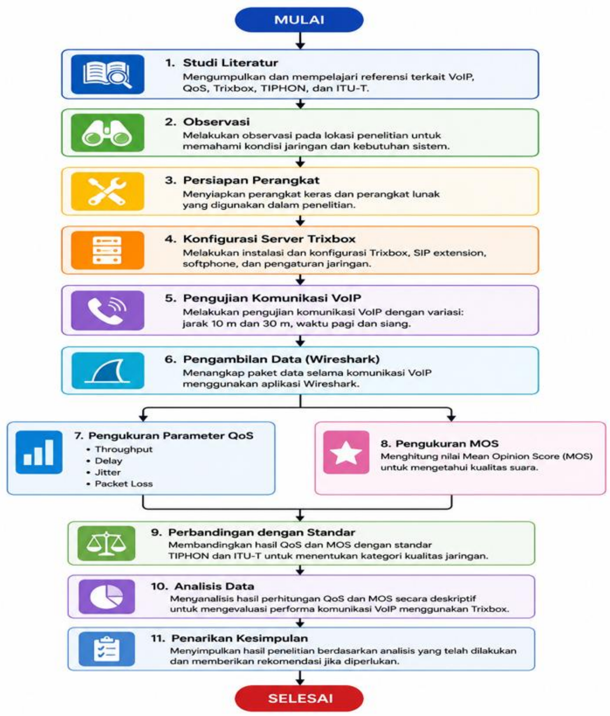

Fig. 1. Workflow of the Proposed Research Methodology

Table.1 Hardware

No	Device	Specific	Function
1	Server Laptop	AMD Ryzen 5 5500U, RAM 16 GB	Running Trixbox as an IP-PBX server
2	Client Laptop	AMD Ryzen 3 7320U, RAM 8 GB	Running the Zoiper app as a client
3	Access Point	Office Wi-Fi	Local network communication medium

Table.2. Software

No	Software	Function
1	Trixbox (berbasis Asterisk)	Manages VoIP communications, generates call logs, and tracks PBX server performance data.
2	Oracle VirtualBox	Runs Trixbox in a virtual environment to simplify server configuration and management.
3	Softphone (Zoiper)	Used to test calls between clients and record communication results.
4	Wireshark	Analyzes data packets passing through the network and calculates delay, throughput, and packet loss parameters.

2.2 Network Topology

In Fig. 2 illustrates the experimental network topology used in this study. The VoIP communication system consists of one Trixbox server, two client laptops equipped with SIP softphone applications, one wireless access point, and one router. All devices are connected through a WLAN infrastructure. The Trixbox server functions as the IP-PBX responsible for SIP registration and call management, while Wireshark is used to capture RTP packets during VoIP communication. Performance evaluation is conducted under two communication distances (10 m and 30 m) and two testing periods (morning and afternoon).

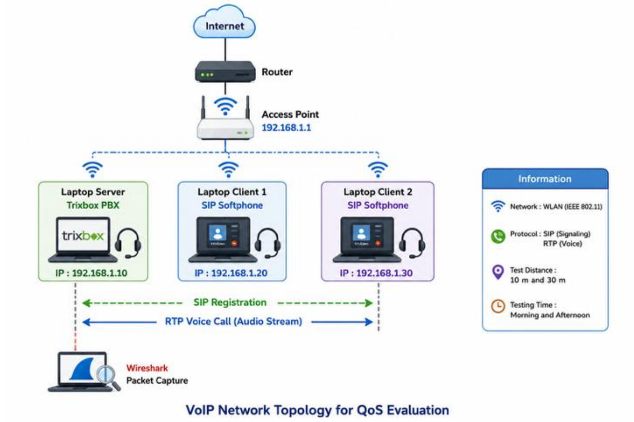

Fig 2. Network Topology

2.3. Metrik Evaluasi

- Throughput

Throughput is the effective data transfer rate in a network, typically measured in bits per second (bps), its value is calculated by dividing the number of packets that successfully reach their destination by the observation period [23]. Table 3 lists the throughput categories according to the TIPHON standard. To calculate throughput, use the formula.

Table 3. Throughput

Category Throughput	Throughput	Indeks
Excellent	100	4
Good	75	3
Fair	50	2
Poor	< 25	1

$$throughput = \frac{data\ packets\ received\ (bits)}{observation\ time\ (seconds)} \quad (1)$$

- Delay

Delay is the time it takes for a data packet to travel from the sender to the receiver during transmission over a network. This parameter is used to measure the transit time of a packet in data communication [23]. Table 4 lists the categories of delay according to the TIPHON standard. To calculate delay, use the formula.

Table 4. Delay

Category Delay	Delay (ms)	Indeks
Excellent	< 150	4
Good	150 – 300	3
Fair	300 – 450	2
Poor	> 450	1

$$average\ delay = \frac{total\ delay}{total\ packets\ received} \quad (2)$$

- Packet loss

Packet loss is the number of data packets that fail to reach their destination during transmission over a network [23]. Packet loss can be caused by factors such as noise, collisions, congestion, or excessive queuing in the network. Table 5 lists the categories of packet loss according to the TIPHON standard. To calculate packet loss, use the formula.

Table 5. Packet Loss

Category Packet Loss	Packet Loss	Indeks
Excellent	0 %	4
Good	3 %	3
Fair	15 %	2
Poor	25 %	1

$$Packet\ Loss: \frac{Data\ packets\ sent - Data\ packets\ received}{Data\ packets\ sent} x100\% \quad (3)$$

- Jitter

Jitter is the variation or difference in arrival times between data packets transmitted over a network. High jitter values can cause disruptions in real-time applications such as video conferencing or voice calls, resulting in choppy audio or out-of-sync video [24]. Table 6 lists the categories of jitter according to the TIPHON standard. To calculate jitter, use the formula.

Table 6. Jitter

Category Jitter	Jitter (ms)	Indeks
Excellent	0 ms	4
Good	0 ms s/d 75 ms	3
Fair	75 ms s/d 125 ms	2
Poor	125 ms s/d 225 ms	1

$$Jitter: \frac{Total\ Delay\ Variation}{Total\ number\ of\ packages\ received} \quad (4)$$

- Mean Opinion Score (MOS)

The Mean Opinion Score (MOS) is a metric used to assess voice quality in VoIP systems based on user perception. The MOS value indicates the extent to which speech is clear and intelligible after passing through the network [25]. Table 7 lists the categories of the Mean Opinion Score.

Table 7. Mean Opinion Score

MOS	QUALITY
5	Excellent
4	Good
3	Fair
2	Poor
1	Very Poor

3. Result and Discuss

3.1 Implementation of a VoIP System

The VoIP communication system was implemented using Trixbox as an IP-PBX server running on Oracle VM VirtualBox. The server was configured with several SIP accounts used by the Zoiper application on client devices. After the clients successfully registered with the server, voice communication tests were conducted over the local Wi-Fi network. During the communication process, RTP packets were captured using the Wireshark application to obtain Quality of Service (QoS) parameter data namely, throughput, delay, jitter, and packet loss as the basis for analyzing the quality of the VoIP communication service. And then Fig. 4 shows that the Trixbox server has been successfully launched and is ready to be used as an IP-PBX server to handle VoIP communications.

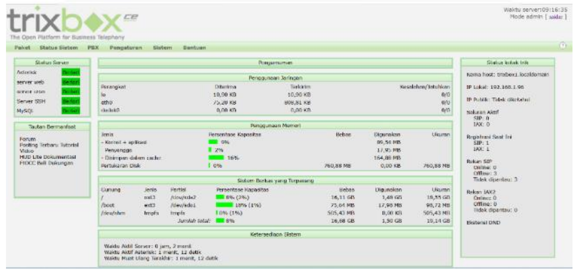

Fig. 4. The Trixbox server is active

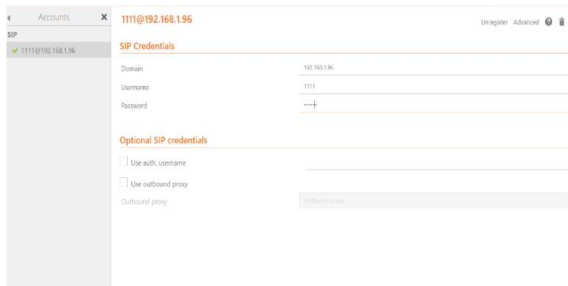

Fig. 5. SIP account configuration in the Zoiper app

Fig. 5 shows that the SIP account in the Zoiper app has been successfully configured and connected to the Trixbox server with a “Registered” status, meaning the client is ready for voice communication.

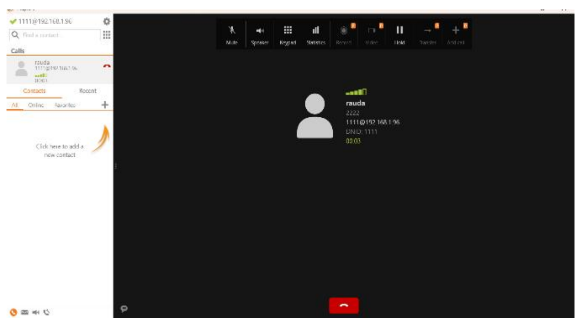

Fig. 6. The process of a VoIP call between clients

In Fig. 6. illustrates the voice communication process between clients using the Zoiper app via the Trixbox server. This process is used as a test scenario to collect network performance data.

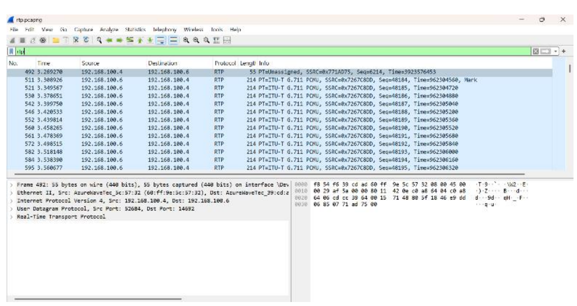

Fig. 7 RTP packet capture results using Wireshark

the process of capturing RTP packets using Wireshark during a VoIP call at shows the Fig. 7. The captured data is then used to calculate the parameters of throughput, delay, jitter, and packet loss.

3.2 Quality of Service (QoS) Test Results

- Throughput

Based on the average test results presented in Fig. 8 , the throughput values of the VoIP communication varied under different testing conditions according to the communication distance between the clients and the access point. The experiments were conducted under two testing periods (morning and afternoon) and two communication distances (10 m and 30 m). During the morning test at a distance of 10 m, the throughput reached 88.5 kbps at Client 1 and 86.5 kbps at Client 2, indicating stable voice packet transmission due to the relatively small throughput difference between the two clients. When the distance increased to 30 m, Client 1 maintained a throughput of 89.0 kbps, whereas Client 2 experienced a decrease to 81.6 kbps. In contrast, the afternoon test produced more stable throughput values, with 89.0 kbps recorded at Client 1 and 88.0 kbps at Client 2 at a distance of 10 m, suggesting that lower network utilization and reduced interference contributed to improved transmission performance.

-

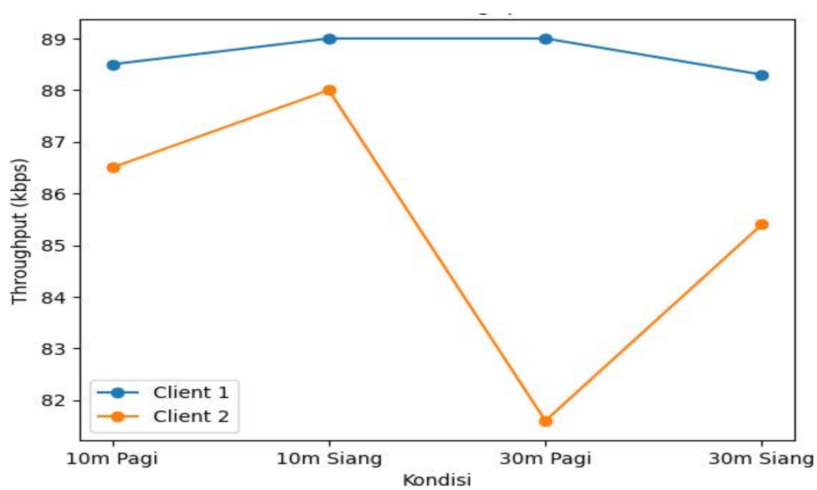

Fig. 8. Throughput

The observed throughput degradation at longer communication distances can be attributed to the reduction in Wi-Fi signal strength as the client moved farther from the access point. Weaker signal quality increases the likelihood of packet collisions, retransmissions, and transmission delays, thereby reducing the effective throughput of the VoIP communication. In addition to communication distance, network traffic conditions also affected throughput performance. The morning testing period exhibited slightly lower throughput than the afternoon session, which is likely associated with higher network utilization and greater wireless interference. These findings demonstrate that both communication distance and network load significantly influence the efficiency of Real-time Transport Protocol (RTP) packet transmission in wireless VoIP networks.

Overall, the throughput results indicate that the proposed Trixbox-based VoIP system is capable of maintaining reliable communication performance over a wireless network. All measured throughput values remained above 75 kbps, which falls within the good category according to the Quality of Service (QoS) standard. These findings are consistent with the results reported by Tiara et al [26], who concluded that stable throughput reflects the capability of a wireless network to transmit voice packets efficiently, thereby ensuring high-quality VoIP communication. Consequently, although throughput decreased under longer communication distances, the overall network performance remained sufficiently stable to support real-time voice communication in the evaluated operational environment.

- **Delay**

Based on the average test results presented in Fig. 9, the delay values varied under different VoIP communication scenarios over the Wi-Fi network. The experiments were conducted during two testing periods (morning and afternoon) with communication distances of 10 m and 30 m between the clients and the access point. Delay represents the time required for voice packets to travel from the sender to the receiver during VoIP communication. Lower delay values indicate better communication quality because voice packets are delivered more quickly with minimal transmission latency. During the morning test at a distance of 10 m, the measured delay was 20.12 ms for Client 1 and 20.65 ms for Client 2, indicating stable communication performance due to the relatively small difference between the two clients. When the communication distance increased to 30 m, the delay remained almost unchanged for Client 1 (20.13 ms) but increased to 22.11 ms for Client 2. During the afternoon session, the delay values were slightly lower, reaching 20.00 ms and 20.28 ms at a distance of 10 m, while increasing moderately to 20.19 ms and 21.00 ms, respectively, at 30 m.

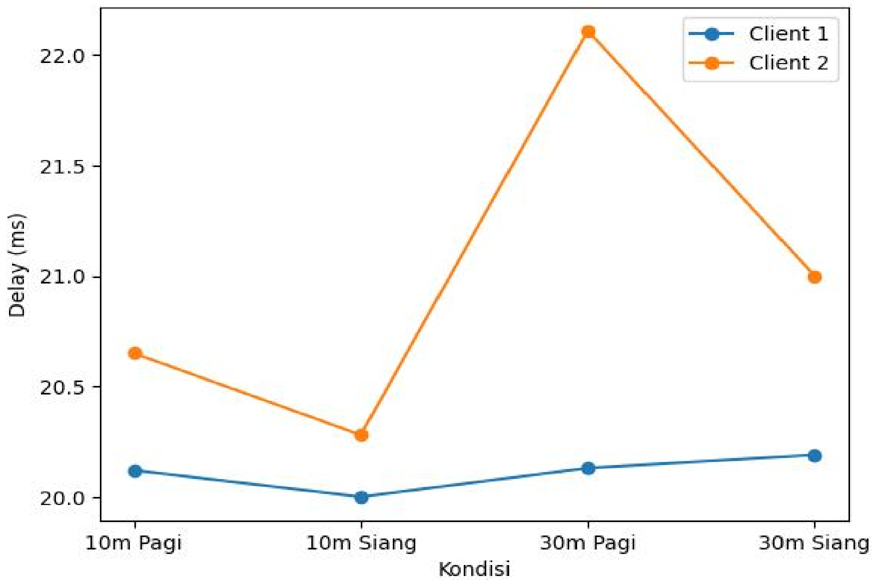

Fig. 9. Delay

The increase in delay observed at longer communication distances is primarily associated with the degradation of Wi-Fi signal strength as the client moves farther from the access point. Reduced signal quality increases packet transmission time and may lead to retransmissions, signal interference, and higher network traffic congestion, particularly during the morning when network utilization is relatively higher. Consequently, RTP packets require more time to reach the destination, resulting in increased end-to-end delay. Nevertheless, the afternoon tests consistently produced lower delay values than the morning tests, suggesting that reduced network utilization and fewer wireless interferences contributed to more stable packet transmission. These results demonstrate that both communication distance and wireless network conditions significantly influence packet transmission delay in VoIP communication.

Overall, the delay measurements indicate that the wireless network is capable of supporting reliable real-time VoIP communication. All recorded delay values remained below 150 ms, satisfying the Excellent category according to the Quality of Service (QoS) standard. These findings are consistent with the results reported by Tiara et al [26], who concluded that low delay values indicate efficient and stable RTP packet transmission in wireless VoIP networks. The study further emphasized that network quality and signal conditions play a crucial role in determining delay performance during real-time voice communication. Therefore, although delay increased slightly as the communication distance increased, the Trixbox-based VoIP system maintained stable communication performance without causing noticeable latency during voice conversations.

- **Pecket Loss**

Based on the average test results presented in Table 4.3, the packet loss values varied under different VoIP communication scenarios over the Wi-Fi network. The experiments were conducted during two testing periods (morning and afternoon) with communication distances of 10 m and 30 m between the clients and the access point. Packet loss refers to the percentage of voice packets that fail to reach the destination during transmission. Lower packet loss indicates better communication quality because RTP packets are delivered more reliably. During the morning test at a distance of 10 m, the measured packet loss was 0.50% for Client 1 and 0.99% for Client 2, indicating stable VoIP communication with only minor packet loss. When the communication distance increased to 30 m, Client 1 maintained 0.00% packet loss, whereas Client 2 experienced a significant increase to 5.79%. During the afternoon session, both clients achieved the best performance at 10 m, with 0.00% packet loss, while at 30 m, Client 1 remained at 0.00% and Client 2 recorded 2.84%, which was considerably lower than the corresponding morning measurement.

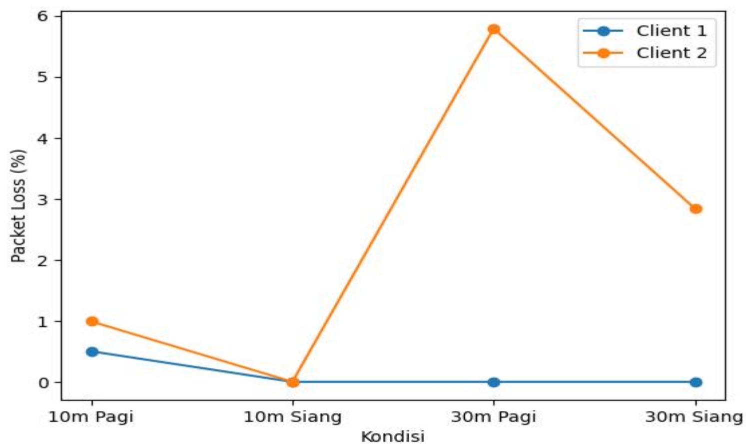

Fig. 10. Packet Loss

The increase in packet loss at longer communication distances is mainly attributed to the degradation of Wi-Fi signal quality as the client moves farther from the access point. Reduced signal strength increases the probability of RTP packet loss due to wireless interference, packet collisions, retransmissions, and higher network traffic congestion. These effects were more pronounced during the morning tests, when network utilization was relatively higher, resulting in greater packet loss than during the afternoon tests. Conversely, the more stable wireless conditions and lower network activity in the afternoon contributed to improved packet delivery performance. These findings demonstrate that both communication distance and wireless network conditions have a significant impact on the reliability of RTP packet transmission in VoIP communication.

Overall, the packet loss results indicate that the wireless network is capable of supporting reliable VoIP communication under the evaluated testing scenarios. Although packet loss increased to 5.79% for Client 2 during the 30 m morning test, the overall communication quality remained within the good category according to the Quality of Service (QoS) standard, allowing voice conversations to continue without significant disruption. These findings are consistent with the study conducted by Goworzky et al [27], which reported that increasing the distance between wireless devices and the access point leads to higher packet loss due to signal attenuation. The study further emphasized that wireless interference and network congestion can reduce RTP packet delivery reliability, thereby affecting the stability and quality of VoIP communication.

- **Jitter**

Based on the average test results presented in Fig. 11, the jitter values varied under different VoIP communication scenarios over the Wi-Fi network. The experiments were conducted during two testing periods (morning and afternoon) with communication distances of 10 m and 30 m between the clients and the access point. Jitter represents the variation in packet arrival time during voice transmission, where lower jitter values indicate more stable packet delivery and better voice communication quality. During the morning test at a distance of 10 m, the measured jitter was 20.10 ms for Client 1 and 20.13 ms for Client 2, indicating stable packet transmission due to the minimal difference between the two clients. When the communication distance increased to 30 m, the jitter of Client 1 remained relatively stable at 19.98 ms, whereas Client 2 experienced an increase to 21.63 ms. During the afternoon session, the lowest jitter values were recorded at 10 m, reaching 19.98 ms for Client 1 and 19.88 ms for Client 2. At 30 m, the jitter increased slightly to 20.15 ms and 20.61 ms, respectively, while still maintaining stable communication performance.

The increase in jitter observed at longer communication distances is primarily caused by the degradation of Wi-Fi signal quality as the clients move farther from the access point. Weaker wireless signals lead to greater variations in RTP packet arrival times due to increased interference, packet queuing, retransmissions, and higher network traffic congestion, particularly during the morning testing period. These factors reduce the consistency of packet delivery, resulting in higher jitter values. In contrast, the afternoon tests produced lower jitter values because network utilization was relatively lighter and wireless interference was reduced, allowing RTP packets to be transmitted with more consistent timing. These findings demonstrate that both communication distance and wireless network conditions significantly influence packet transmission stability in VoIP communication.

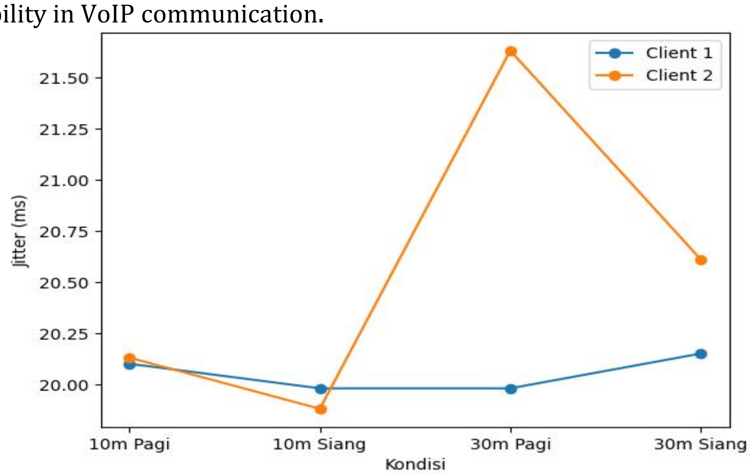

Fig. 11. Jitter

Overall, the jitter results indicate that the Trixbox-based VoIP system is capable of maintaining stable voice packet transmission over the wireless network. All measured jitter values remained within the Good category according to the Quality of Service (QoS) standard, indicating that the communication system can support real-time voice services without significant degradation in audio quality. These findings are consistent with the study conducted by Tiara et al [26], which reported that low jitter values reflect stable RTP packet transmission and contribute to clearer voice communication in wireless VoIP networks. The study further emphasized that wireless network quality and traffic conditions are key factors affecting jitter performance. Therefore, although jitter increased slightly as the communication distance increased, the overall communication quality remained stable and suitable for reliable VoIP services.

- ***Mean Opinion Score* (MOS)**

Table. 8. Analysis of VoIP Service Quality Based on MOS Scores

Conditions	Client	Min	Max	Rata-Rata
Morning 10 meters	Client 1	3,69	4,42	4,20
Morning 10 meters	Client 2	3,49	4,25	4,12
Morning 30 meters	Client 1	4,25	4,25	4,25
Morning 30 meters	Client 2	2,32	4,25	3,63
Afternoon 10 meters	Client 1	4,25	4,42	4,25
Afternoon 10 meters	Client 2	4,25	4,25	4,25

Conditions	Client	Min	Max	Rata-Rata
Afternoon 30 meters	Client 1	3,21	4,25	4,14
Afternoon 30 meters	Client 2	2,12	4,25	3,9

Based on the test results presented in Table 8, the Mean Opinion Score (MOS) indicates that the overall voice quality of the VoIP communication ranged from Good to Very Good under most testing conditions. For Client 1, the average MOS remained relatively stable, ranging from 4.14 to 4.25. The highest average MOS (4.25) was recorded during the morning test at 30 m and the afternoon test at 10 m, while the lowest value (4.14) was observed during the afternoon test at 30 m. In contrast, Client 2 exhibited greater variation in MOS values. The average MOS increased from 4.12 during the 10 m morning test to 4.25 during the 10 m afternoon test, but decreased significantly to 3.63 during the 30 m morning test before improving slightly to 3.90 during the 30 m afternoon test. These results indicate that voice quality tends to deteriorate as the communication distance increases and wireless network conditions become less favorable.

The reduction in MOS values is closely associated with the degradation of wireless network performance, particularly the increase in packet loss, delay, and transmission instability. This relationship is clearly observed for Client 2 during the 30 m morning test, where the highest packet loss and increased delay corresponded to the lowest average MOS (3.63). Under these conditions, voice quality began to deteriorate, resulting in intermittent audio interruptions and reduced speech intelligibility, although communication could still be maintained. The afternoon tests consistently produced higher MOS values than the morning tests, indicating that lower network utilization and reduced wireless interference contributed to improved voice quality. These findings demonstrate that maintaining stable network conditions is essential for achieving satisfactory VoIP communication quality.

Overall, the MOS evaluation confirms that the proposed Trixbox-based VoIP system is capable of delivering acceptable voice quality under the evaluated testing scenarios. Most average MOS values exceeded 4.0, corresponding to the Good or Very Good category, while values around 3.0 indicate acceptable but degraded voice quality. A few minimum MOS values recorded for Client 2, namely 2.32 during the 30 m morning test and 2.12 during the 30 m afternoon test, suggest noticeable audio degradation caused by increased packet loss, jitter, and delay under less favorable wireless conditions. Nevertheless, the overall results demonstrate that the VoIP system maintained satisfactory communication performance for most operating conditions. These findings are consistent with the study conducted by Goworzky et al [27], which reported that increasing the communication distance between wireless devices and the access point degrades perceived voice quality due to higher packet loss and reduced network stability. Their study further emphasized that MOS is strongly influenced by wireless network quality and serves as a reliable indicator of the overall user experience in VoIP communication systems.

4. Conclusions

This study evaluated the Quality of Service (QoS) performance of a Trixbox-based Voice over Internet Protocol (VoIP) communication system implemented at the Sultan Iskandar Muda Meteorological Station Banda Aceh under different communication distances and network utilization periods. The experimental results demonstrate that communication distance and wireless network conditions significantly influence VoIP performance. Increasing the communication distance from 10 m to 30 m resulted in higher delay, jitter, and packet loss, while throughput and Mean Opinion Score (MOS) tended to decrease, particularly for the client located farther from the access point. Nevertheless, all measured QoS parameters remained within the acceptable limits defined by the TIPHON and ITU-T standards, indicating that the implemented Trixbox-based VoIP system is capable of providing reliable real-time voice communication in an operational wireless network environment.

The findings of this study provide scientific evidence that communication distance and wireless network quality are key factors affecting VoIP service performance. From a practical perspective, the results can serve as a reference for government institutions and other organizations planning to deploy Trixbox-based VoIP systems over wireless networks by identifying communication conditions that maintain acceptable QoS performance. This study also contributes to the existing body of knowledge by providing an empirical QoS evaluation of VoIP implementation in a real operational meteorological environment, which has received limited attention in previous studies. Future research is recommended to investigate VoIP performance under more diverse network conditions, including higher traffic loads, larger numbers of concurrent users, different voice codecs, and advanced network management techniques such as Quality of Service (QoS) optimization, Quality of Experience (QoE) evaluation, and Software-Defined Networking (SDN) to further improve communication reliability and service quality.

5. References

[1] P. Susanto, "QoS measurement of VoIP codec usage on limited bandwidth network over UDP-based VPN," *Telfor J*, vol. 12, no. 1, pp. 13–17, Jan. 2020, doi: 10.5937/telfor2001013s.

[2] S. Y. Prayogi and S. B. Sinaga, "Implementasi Mos dan E-Model Untuk Mengetahui Kualitas Komunikasi Jaringan VoIP," *MEANS*, pp. 24–27, June 2021, doi: 10.54367/means. v6i1.1239.

[3] M. K. Ni'Am, A. Wijayanti, and A. Muqtadir, "PERANCANGAN DAN PENERAPAN JARINGAN KOMUNIKASI DENGAN VOICE OVER INTERNET PROTOCOL (VOIP) PADA YSPIS REMBANG," *curtina*, vol. 3, no. 2, pp. 50–56, Dec. 2022, doi: 10.55719/curtina. v3i2.611.

[4] M. E. Martínez Sánchez and J. Villoro Armengol, "The Implementation of New Technologies in Internal Communication: A Study of the Main Platforms and Applications," *Journal of Promotion Management*, vol. 27, no. 6, pp. 788–811, Feb. 2021, doi: 10.1080/10496491.2021.1888178.

[5] A. Leonard, "Empirical support of the positive impact of an Intranet on communication in the working environment," *JCSA*, vol. 24, no. 1, pp. 112–128, Oct. 2022, doi: 10.36615/jcsa. v24i1.1762.

[6] M. K. R. Farhandika and E. Christianto, "ANALISA QUALITY OF SERVICE (QOS) VOICE OVER INTERNET PROTOCOL (VOIP) PADA POWERLINE COMMUNICATION (PLC) DI ALIRAN LISTRIK 110 VOLT DAN 220 VOLT," *jipi. jurnal. ilmiah. penelitian. dan. pembelajaran. informatika.*, vol. 9, no. 4, pp. 1936–1947, Nov. 2024, doi: 10.29100/jipi.v9i4.5606.

[7] S. Ahmad and N. Awasthy, "An Information Theoretic Approach for the Development of a Framework for Improving Communication Reliability in a Mobile Network," *ijcse*, vol. 7, no. 5, pp. 114–118, May 2019, doi: 10.26438/ijcse/v7i5.114118.

[8] T. Song, G. Perna, P. Garza, M. Meo, and M. M. Munafo, "Packet Loss in Real-Time Communications: Can ML Tame Its Unpredictable Nature?" *IEEE Trans. Netw. Serv. Manage.*, vol. 22, no. 1, pp. 72–91, Feb. 2025, doi: 10.1109/tnsm.2024.3442616.

[9] S. Putra and O. Sulaiman, "PERANCANGAN JARINGAN KOMUNIKASI VOIP (VOICE OVER INTERNET PROTOCOL) MENGGUNAKAN TRIXBOX PADA UNIVERSITAS ISLAM SUMATERA UTARA," *SHILAP Revista de lepidopterología*, vol. 4, no. 2, pp. 186–190, July 2019, doi: 10.24114/cess.v4i2.14075.

[10] A. Pribadi, "VoIP (Voice over Internet Protocol) Based on Asterisk Using Trixbox at SMKN 1 Bangkinang," *ijsecs*, vol. 3, no. 2, pp. 65–70, Aug. 2023, doi: 10.35870/ijsecs.v3i2.1275.

[11] E. Pertovt, K. Alic, A. Švigelj, and M. Mohorcic, "Performance Evaluation of VoIP Codecs over Network Coding in Wireless Mesh Networks," *WSEAS TRANSACTIONS ON COMMUNICATIONS*, vol. 20, pp. 185–191, Dec. 2021, doi: 10.37394/23204.2021.20.24.

[12] B. V. D. Krishna, P. Yogendrasai, S. Rajeev, and S. M. Rajagopal, "Voice Over Internet Protocol: Custom SIP Server with QoS Optimization for Real-Time Communication," Institute Of Electrical Electronics Engineers, Feb. 2025, pp. 566–571. doi: 10.1109/idciot64235.2025.10914779.

[13] M. Adibasyah, "Rancang Bangun Dan Analisa Quality Of Services (QoS) VoIP Server Menggunakan Briker Dengan Memanfaatkan Trunk Dan Interactive Voice Respons (IVR) Di AirNav Padang," *Jurnal Ilmiah Telekomunikasi, Kendali dan Elektronika Terapan*, vol. 11, no. 2, pp. 115–128, Jan. 2024, doi: 10.34010/telekontran.v11i2.11220.

[14] V. Vichev and T. Georgieva, "Behavior of VoIP Traffic QoS Metrics in Loaded Networks," Institute Of Electrical Electronics Engineers, Nov. 2024, pp. 1–4. doi: 10.1109/telecom63374.2024.10812328.

[15] Y. Daza Alava, D. M. Zambrano, E. Cedeño Palma, L. Chancay Garcia, and M. Cruz Felipe, "Evaluation of Quality of Service in VoIP Traffic Using the E Model," Springer, 2021, pp. 34–43. doi: 10.1007/978-3-030-90241-4_3.

[16] K. S. Roy and S. Malakar, "Comparative Study of Different Machine Learning Models for Measurement of QoE Based on MOS in Wireless Networks," Springer Singapore, 2021, pp. 201–213. doi: 10.1007/978-981-16-5207-3_18.

[17] J. M. Ahmad, “The Effect of Decreased Voice Quality in a VoIP Network,” *SJFSSU*, vol. 2, no. 1, pp. 57–62, Apr. 2022, doi: 10.37375/sjfssu. v2i1.140.

[18] B. Cetin, H. H. Kilinc, H. Kilimci, and Z. H. Kilimci, “Identifying Degradation in Voice over Internet Protocol (VoIP) Communications Through Audio-Based Transformer Models,” Institute Of Electrical Electronics Engineers, Nov. 2024, pp. 1–6. doi: 10.1109/ismsit63511.2024.10757222.

[19] K. Maheswari and A. Balamurugan, “Voice over internet protocol codec performance in interactive streaming environment,” *JCS*, vol. 13, no. 1, p. 16, Jan. 2024, doi: 10.26634/jcs.13.1.20435.

[20] Y. Skuratovskyi, A. Anosov, V. Strelnikov, and M. Kucheriavyi, “EXPERIMENTS AND PRACTICAL SOLUTIONS FOR BUILDING A TEST ENVIRONMENT TO ASSESS APPLICATION-LEVEL SECURITY,” *Cybersecurity: Education, Science, Technique*, vol. 3, no. 31, pp. 217–226, Jan. 2025, doi: 10.28925/2663-4023.2025.31.1014.

[21] D. K. Mwangi, A. O. Ndlovu, and S. T. Boateng, “Performance analysis of broadband communication networks using basic QoS metrics,” *Int. J. Commun. Inf. Technol.*, vol. 7, no. 1, pp. 28–32, Jan. 2026, doi: 10.33545/2707661x.2026.v7.i1a.173.

[22] P. H. Mukti, S. Ananda, and G. Kusrahardjo, “Performance Evaluation of VoIP Service for Emergency Disaster Responses over WiMAX-WLAN Integrated Network,” *JNTE*, Nov. 2022, doi: 10.25077/jnte. v11n3.963.2022.

[23] D. Ramayanti, "Analisis dan Peningkatan Kualitas Layanan pada Jaringan Komputer Nirkabel Badan Penghubung Lampung dalam Mendukung Tugas Pemerintahan," vol. 7, no. 1, pp. 1–13, 2024.

[24] F. Ardiansyah, M. M. Alamin, M. Rafli, M. F. Z. Farros, R. Hidayat, B. Al Zalzala, and N. Kurniati, "A Study of Internet Network Quality of Service (QoS) Using Wireshark in a Campus Environment," *Nusantara Computer and Design Review*, pp. 64–70, 2025.

[25] Y. W. Pratama, T. Budiman, and A. Z. Sianipar, "Analysis of Internet Network Quality Using the Quality of Service (QoS) Method at the FOLU Office, Ministry of Forestry," vol. 10, no. 2, pp. 507–516, 2025.

[26] D. R. Tiara, B. Sugiarto, and S. Nurpadillah, "Performance Analysis of VoIP Based Wireshark on Wireless Networking Using RTP Protocol," *Jurnal FUSE – Teknik Elektro*, vol. 3, no. 1, pp. 55–63, 2023.

[27] M. D. Goworzky, A. Wijaya, I. Z. Yadi, and T. D. Purwanto, "Analisa QoS pada Jaringan Voice over Internet Protocol Server Portable Berbasis Raspberry Pi Menggunakan Metode Action Research pada Daerah Tak Terjangkau Sinyal dan Sumber Daya Listrik," vol. 2, pp. 1–12, 2024.