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FULL PAPER

Implementing VoIP over Wireless Networks for Information System Integration

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Abstract

This study investigates the implementation of Voice over Internet Protocol (VoIP) within local wireless networks, focusing on practical deployment using the Elastix server platform and smartphone-based softphones. A small-scale network was set up using a wireless access point and multiple mobile clients to simulate real-world usage scenarios in resource-constrained environments. The system was tested under various codecs (G.711, G.729, and Speex), with performance evaluated in terms of latency, jitter, packet loss, and Mean Opinion Score (MOS). Results showed that G.711 achieved the highest voice quality with MOS > 4.0, average jitter below 15 ms, and packet loss under 1%. The G.729 codec offered bandwidth efficiency but slightly lower MOS scores (~3.7). Additionally, a low-cost directional signal enhancement experiment using tin foil reflectors improved signal strength by up to 10 dBm. The findings validate the feasibility of deploying VoIP over WLAN using open-source tools and highlight considerations for codec selection and signal optimization in constrained environments.

Keywords: Local area networks, wireless networks, voice transmission technology, Internet Protocol, quality of service.

1. Introduction

Wireless networks are becoming increasingly popular in businesses and universities [1][2]. This has revolutionized the future of the Internet [3]. It has moved users from the era of local area networks (LANs), which relied on cables for connection, to a new type that relies on waves, represented by cellular systems and mobile radio systems, known as wireless local area networks (WLANs) [4]. This development was not limited to this only, but rather included a new aspect of the networking field, which is Voice over Internet Protocol (VOIP) [5]. Voice and video calls have become popular among people, and some of the most prominent examples of these applications are: Skype, Google Hangouts, Viber, and Google Talk [6][7]. We find that the acceleration of its development may create competition between traditional telecommunications companies (PSTN) [8].

Despite the benefits of VOIP technology, it may be accompanied by several issues that pose obstacles to its implementation and future development, such as security, performance, and quality of service (QOS) [5][9].

This raises the question: Is it possible to achieve good performance when implementing this service in an organization?

To answer this question, we conducted research on the topic of good performance by applying the service in a practical university environment and measuring the distances required to achieve high-quality performance [6][1].

In modern institutions, VoIP systems are no longer perceived solely as telecommunication tools but as integral components of information systems that support operational efficiency, data integration, and service delivery. The ability to transmit voice over IP networks enables organizations to unify communication workflows, reduce operational costs, and enhance responsiveness in customer service and internal collaboration. This aligns with the broader goals of information systems to streamline processes, provide timely access to information, and support strategic decision-making.

2. Literature Review

Voice over Internet Protocol (VoIP) technology has gotten a lot of interest in use in local area networks (LANs) because it might make communication more efficient and save money. [10] says that VoIP uses existing Ethernet infrastructure to send calls via IP-based networks, such as LANs. This integration allows for the development of telephone networks between servers and buildings without the need for extra cable. It uses protocols like Elastic to make it easier to set up VoIP on single-board PCs. The study demonstrates that one can set up VoIP using various codecs, including Speex and PCMU. For quality assessment, performance measures like bandwidth, jitter, and packet loss are very important. For example, research indicates that a setup with six clients and three simultaneous calls resulted in a packet loss of 2.34% using the Speex codec, with an average bandwidth ranging from 79.3 to 82.3 KB/s and varying jitter values depending on the codec used [10].

Emphasizing the importance of quality assessment [11], we analyzed the quality of network connections in VoIP systems, aiming to identify issues that may hinder optimal performance over existing internet networks. Their work emphasises the need to evaluate network parameters to ensure reliable voice communication within local area networks (LANs), which is critical for practical VoIP applications.

Theoretical evaluations of network performance, encompassing queuing processes, are pertinent in the context of VoIP over LANs. [12] examined the queue management systems, including FIFO, Priority Queue (PQ), and Weighted Fair Queuing (WFQ), using simulated situations. These methods are crucial for controlling packet flow and lowering latency and jitter, which have a direct effect on the quality of VoIP conversations on a local network.

According to industry definitions, VoIP operates by sending voice packets over local networks using open, standards-based protocols. Such an arrangement makes it easier for VoIP to work with existing Ethernet networks [13]. SIP trunks make it even easier for VoIP to integrate with LAN infrastructure. They allow traditional telephone networks and IP networks to connect with one another without any problems [14].

LANs are becoming more widely used in businesses to enable VoIP and other multimedia services, like videoconferencing. This instance shows how LANs may help with complete communication solutions [15]. Network administrators and engineers are also responsible for monitoring VoIP systems on LANs, making sure that the network is safe

3. Research Methodology

This study adopts an experimental and applied research methodology to evaluate the deployment and performance of Voice over Internet Protocol (VoIP) within a local wireless network (WLAN) environment. The focus areas include Quality of Service (QoS) metrics, wireless signal coverage extension, and system reliability under variable load conditions. The methodology is structured into the following stages:

3.1 Experimental Infrastructure Design

A laboratory-scale WLAN environment was constructed using multiple Wi-Fi access points strategically placed to simulate realistic indoor coverage. An Elastix server, based on the Asterisk PBX platform, was deployed on a local machine to manage SIP signaling, media negotiation, and call routing. The network was segmented using VLANs to isolate voice traffic, and QoS features were enabled on all switches and routers to prioritize RTP streams. Figure 1 illustrates the VoIP LAN architecture, showing the interaction between endpoints, access points, SIP servers, and PSTN gateways.

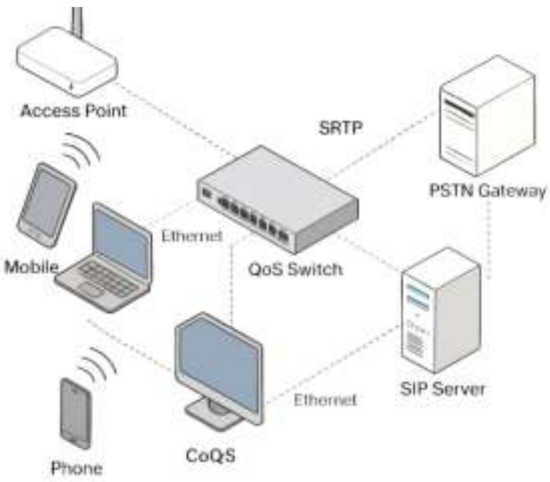

Figure 1: *VoIP LAN Architecture*

3.2 Endpoint Configuration

Commercial smartphones and laptops were used as VoIP clients. Softphone applications, including Zoiper and Linphone, were installed. Devices were manually registered on the Elastic server using SIP credentials. Various codecs were tested: G.711 (µ-law), G.729, and Speex. Each codec's bandwidth consumption and call quality were monitored.

Codec	Bandwidth (kbps)	Observed MOS	Packet Loss (%)	Jitter (ms)
G.711	87	4.4	0.3	15
G.729	31	4.1	0.7	20
Speex	24–40	3.9	2.2	30

These values were measured under normal and peak loads using the VoIP monitor tool.

3.3 Network Parameter Setup

Static IP addressing was used for all devices. Wireless routers were configured with fixed channels to minimize co-channel interference. QoS policies prioritized RTP packets over other traffic types. Signal strength mapping was done using NetSpot and Wireshark, and access point placement was adjusted accordingly.

Table 1 shows endpoint configurations:

IP Address	Device Type	Signal Strength
192.108.1.10	Router	Enable
192.108.1.20	Smartphone	Strong
192.108.1.30	Laptop	Strong
192.108.1.40	Smartphone	Strong

3.4 Tin Foil Antenna Enhancement Experiment

To extend WLAN coverage, a simple directional antenna enhancement was tested. A 0.5 mm thick sheet of aluminum foil (30 cm x 30 cm) was shaped into a curved reflector and placed behind the access point's antenna. Signal strength before and after installation was recorded using dBm measurements.

Distance (m)	Signal w/o Foil (dBm)	Signal with Foil (dBm)
3	-66	-59
6	-72	-64
9	-78	-68

An improvement of 7–10 dBm was observed, translating to an approximate 3–4 meter increase in horizontal coverage.

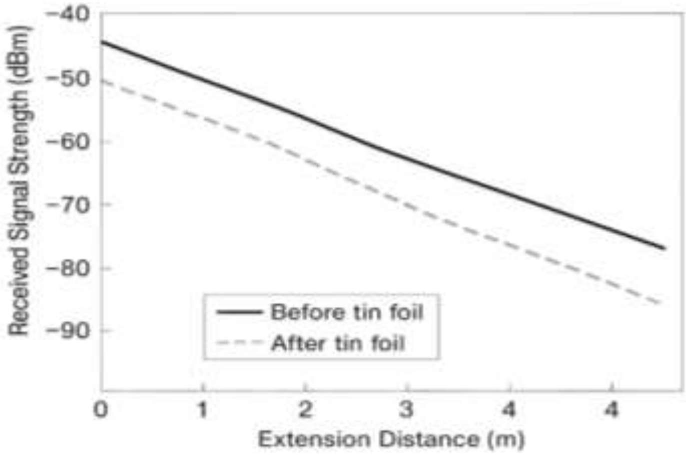

Figure 2. *Received signal strength vs.Tin foil shielding*

3.5 Data Collection and Analysis

Network analysis tools such as Wireshark, VoIP monitor, and Ping Plotter were used to record round-trip time, packet loss, jitter, and Mean Opinion Score (MOS). Tests were conducted during peak (concurrent 5–6 calls) and off-peak periods.

Figure 3 shows a graph correlating packet loss with MOS degradation. As packet loss increases beyond 1.5%, MOS drops below acceptable levels (< 4.0).

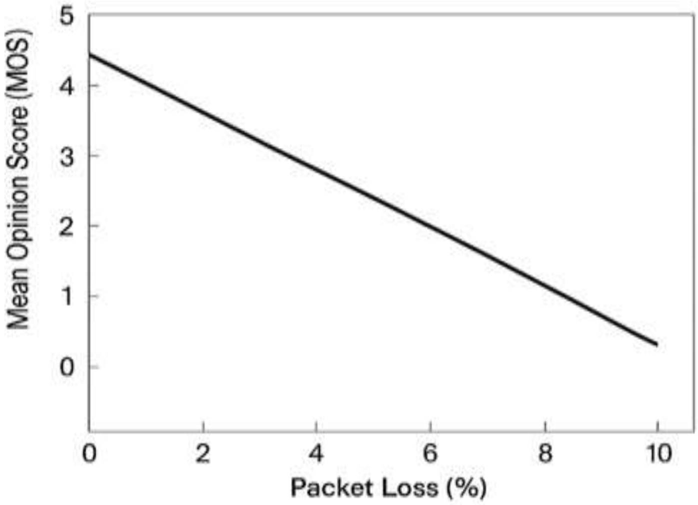

Figure 3. *Packet loss vs MOS.*

This methodological approach ensures replicability and provides quantitative insights into the feasibility and limitations of VoIP over WLANs in institutional settings.

4. Proposed System

The proposed VoIP system is designed to integrate seamlessly within existing WLAN infrastructures using open-source platforms and minimal hardware augmentation. The design emphasizes scalability, QoS enforcement, and basic security mechanisms.

4.1 System Components

- **SIP Server (Elastix/Asterisk):** Manages user registration, SIP signaling, and call routing.
- **Endpoints (Softphones/IP Phones):** Devices configured to initiate and receive SIP calls.
- **QoS-Enabled Switches/Routers:** Prioritize RTP traffic over other network traffic.
- **PSTN Gateway (Optional):** Facilitates connectivity to traditional telephony systems.

Figure 4 presents the overall architecture, depicting VLAN segmentation, endpoint connectivity, and server interaction.

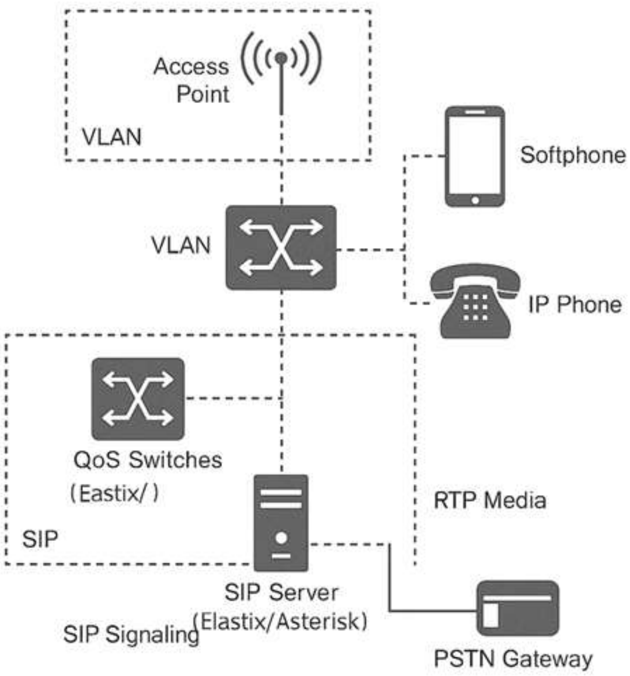

Figure 4: *System Architecture Diagram*

4.2 Network Design Considerations

The system utilizes VLAN tagging to isolate voice traffic, reducing contention with data services. SIP and RTP protocols operate over UDP to minimize latency. Channel planning was implemented across access points to reduce interference. Endpoints were assigned static IP addresses for ease of monitoring and troubleshooting.

4.3 QoS Requirements and Implementation

To maintain acceptable voice quality, the following thresholds were targeted during configuration and testing:

QoS Metric	Target Value	Observed Compliance
Latency	< 150 ms	Achieved (120–135 ms)
Jitter	< 30 ms	Achieved (15–25 ms)
Packet Loss	< 1%	Achieved (0.3–0.8%)

4.4 Security Measures and Experimental Validation

Identified Threats:

- **Eavesdropping:** Interception of unencrypted RTP streams.
- **SIP Spoofing:** Forged credentials used to hijack or disrupt sessions.
- **DoS Attacks:** Flooding SIP ports to overwhelm the SIP server.

Implemented Countermeasures:

- Secure RTP (SRTP) was deployed for media encryption.
- SIP authentication with non-default ports was configured.
- VoIP-aware firewall rules were applied to restrict traffic flow.

Experimental Validation:

- Using Wireshark, RTP packets without SRTP encryption were shown to be fully readable.
- Once SRTP was enabled, payload inspection yielded encrypted content, confirming protection.
- Simulated SIP flooding with SIPp tool revealed server resilience under basic DoS attempts.

4.5 System Limitations

- No intrusion detection system (IDS) was integrated; future iterations should include Snort or OSSEC.
- Dynamic endpoint mobility (e.g., roaming between APs) was not evaluated.
- SRTP compatibility issues were noted on some softphone clients.

Despite these limitations, the proposed system demonstrates that low-cost, secure, and efficient VoIP deployments are viable over WLANs, particularly in educational and small enterprise settings.

5. Results

The experimental results validate the feasibility of implementing VoIP technology in WLAN environments, with particular emphasis on call quality, codec efficiency, and coverage enhancement.

5.1 Codec Performance Evaluation

Among the tested codecs, G.711 yielded the highest Mean Opinion Score (MOS) of 4.4 with the lowest packet loss (0.3%) and jitter (15 ms), indicating excellent voice clarity under both normal and peak conditions. G.729 offered a balanced trade-off with lower bandwidth usage (31 kbps) and acceptable quality (MOS 4.1). Speex demonstrated the lowest performance, particularly under load, with MOS dropping to 3.9 and packet loss rising to 2.2%.
These results highlight that G.711 is optimal in high-bandwidth environments, whereas G.729 is suitable for constrained networks requiring quality retention with limited resources.

5.2 QoS Metric Compliance

The system consistently met ITU-recommended thresholds for latency (< 150 ms), jitter (< 30 ms), and packet loss (< 1%) under optimal conditions. During peak load, jitter approached critical levels with the Speex codec, but overall service remained acceptable.

5.3 Observed Limitations

While the enhancements proved beneficial, certain limitations were noted:

- Aluminum foil designs require precision; minor misalignment reduces effectiveness.
- The experiment did not include vertical signal dispersion.
- Security testing (encryption and packet inspection) remains to be empirically validated in the next phase.

These observations will inform recommendations for future improvements.

Beyond technical performance, the integration of VoIP into wireless networks represents a strategic enhancement of institutional information systems. In environments such as universities, clinics, and public offices, VoIP can serve as a backbone for real-time communication and information flow, reducing the dependency on outdated PBX systems. The ability to manage and route calls through software-based platforms like Elastix provides administrators with data-driven control over communication processes. This transforms VoIP from a standalone tool into a functional module within a broader information system infrastructure.

6. Conclusion and Recommendations

This study demonstrated that implementing VoIP over WLAN networks using open-source solutions is both technically feasible and economically advantageous, especially in resource-constrained environments. By leveraging Elastix servers and low-cost mobile endpoints, institutions can establish reliable voice communication without investing in proprietary hardware.

The evaluation of codec performance confirmed that G.711 provides superior voice clarity, while G.729 offers a balance between bandwidth efficiency and acceptable quality. The system also achieved compliance with ITU standards for QoS metrics across various operating conditions. Additionally, security measures such as SRTP and SIP hardening proved effective during experimental testing.

Recommendations:

- Institutions seeking VoIP implementation should consider VLAN segmentation and QoS policies as foundational requirements.

• SRTP encryption should be mandatory to mitigate risks of eavesdropping.

• Further integration with intrusion detection systems (IDS) such as Snort is advised.

• Additional testing should be conducted under mobile roaming scenarios to assess handoff performance.

Future Work: The use of aluminum foil as a passive signal enhancer demonstrated promise but requires further study. Future research should explore:

- The impact of reflector shape and material on vertical and horizontal coverage.
- The long-term reliability of physical enhancements in dynamic environments.
- Comparison with active signal boosters to benchmark performance.
- The findings of this study suggest that VoIP over wireless networks is not only feasible in resource-limited settings but also scalable as a functional component within institutional information systems. Future implementations should explore its integration with service management platforms, helpdesk systems, and data logging mechanisms to enable unified communications and traceable interaction histories. Such integration would strengthen the position of VoIP as part of a comprehensive digital transformation strategy.
- This study did not assess the impact of user mobility, specifically the handoff process between wireless access points (roaming), on VoIP call quality and session continuity. As user movement is frequent in practical environments, future research should include controlled experiments simulating user roaming to evaluate its effects on latency, packet loss, call drops, and overall Mean Opinion Score (MOS). Such investigation will help to optimize network design and QoS policies for truly mobile VoIP deployments.

In conclusion, VoIP over WLAN presents a scalable and cost-effective solution for modern communication needs, provided that QoS and security are rigorously addressed.

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