SOCIETY: Jurnal Pengabdian Masyarakat,
Vol. 5, No. 4 (2026): Juli, pp. 477- 483
E-ISSN:2827-878X (Online -Elektronik)

Communication Technology Implementation Through Voice Over Internet Protocol at Nusa Putra University

Wira Harsa Abiyasa Rifki Ghani[1,*], Zaenal Alamsyah[2]

[1,2]Universitas Nusa Putra

wira.harsa_ti22@nusaputra.ac.id, zaenal.alamsyah@nusaputra.ac.id

ARTICLE INFO

Article history

Received : 6-5-2026

Revised : 30-7-2026

Accepted : 30-7-2026

Keywords

Communication Tecnology, Voice Over, Internet Protocol, Nusa Putra University

ABSTRACT

Efficient internal communication is essential for educational institutions to support daily operations. This study aims to implement an IP Phone system using VoIP technology as a more flexible and cost-effective alternative to conventional phones. The system was chosen for its ability to utilize existing networks without complex installation. Implementation at Universitas Nusa Putra showed stable, seamless communication across departments and wider coverage. The results indicate that VoIP-based IP Phone systems can serve as a modern, efficient communication solution in campus environments.

Komunikasi internal yang efisien sangat dibutuhkan oleh institusi pendidikan dalam mendukung kegiatan operasional. Penelitian ini bertujuan untuk menerapkan sistem komunikasi berbasis IP Phone menggunakan teknologi VoIP sebagai solusi yang lebih fleksibel dan hemat biaya dibandingkan sistem telepon konvensional. Sistem ini dipilih karena mampu memanfaatkan jaringan yang sudah ada tanpa memerlukan instalasi tambahan yang kompleks. Implementasi dilakukan di Universitas Nusa Putra dan menunjukkan bahwa komunikasi antarunit dapat berjalan lancar, stabil, dan menjangkau area yang lebih luas. Hasil ini menunjukkan bahwa sistem IP Phone berbasis VoIP dapat menjadi alternatif komunikasi modern yang efisien dan praktis di lingkungan kampus.

A. INTRODUCTION

Today's technology is developing rapidly, particularly in the field of telecommunications. Essentially, anyone connected to a network can exchange information and data with one another, whether in the form of images, audio, video, text, or other formats. Voice exchange technology is already widely used by society; one example is the telephone, where everyone can communicate using either landlines (Telkom) or mobile phones. However, when using traditional phone calls, geographic location determines the cost in addition to expensive rates. This is considered less effective because establishing local communication links requires high effort and substantial costs. The advent of Voice Over Internet Protocol (VoIP) technology brings innovation to the communication world, especially in terms of cost efficiency. Unlike conventional telephones that rely on analog circuits, VoIP works by converting voice into digital data transmitted over an internet connection or local network. This advantage is profoundly felt by sectors with high communication mobility, such as educational institutions or schools. By utilizing existing network infrastructure, institutions can significantly cut communication expenses while still obtaining high-quality service. [1]

VoIP, or Voice over Internet Protocol, is a technology capable of transmitting voice, video, and data traffic in packet form over an IP network. An IP network itself is a data communication network based on packet-switching, allowing users to make phone calls using an IP network or the internet. VoIP enables routing, access servers, and multiservice access concentrators to carry and transmit voice and faxes across IP networks. Many advantages can be gained, among which the cost is clearly cheaper than traditional phone rates. [2]

The implementation of this technology can answer or provide an appropriate solution to overcome several constraints regarding limitations in information delivery. The communication established or connected between multiple parties within the same institution can press down expenditures by utilizing this technology, where the function of the traditional telephone is replaced by VoIP IP-based technology. [3]

The implementation of a telephone communication system using VoIP on this campus was carried out to improve the quality of internal communication. The main issue faced is the effectiveness and smoothness of communication among staff members. This implementation method encompasses the stages of site surveys for IP Phone device installation, IP PBX server configuration, network connectivity testing, user training, and integration with other information technology systems on campus. The installation of VoIP based on IP Phones at this higher education institution aims to apply VoIP technology within the university environment to enhance communication efficiency between work units and to support the institution in optimizing its existing network infrastructure.

B. METHODS

In the first phase, the author configured the Panasonic KX-HTS32 IP PBX device using a web interface. An IP PABX is a combination of a Switch/Router and a PABX that handles VoIP. An IP PABX can be used to bypass circuit-switched telephone networks by utilizing data networks to connect with other data networks. [4] as presented within Figure 1 VoIP device implementation topology.

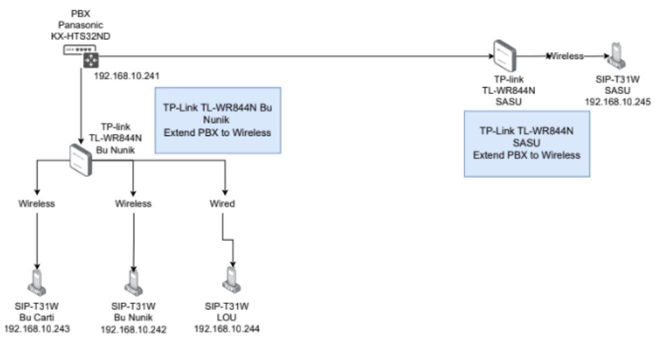

Figure 1. VoIP device implementation topology

The device was connected to the LAN network, and basic network settings—such as network activation, addressing, and the activation of the default internal Wi-Fi feature—were

SOCIETY: Jurnal Pengabdian Masyarakat,
Vol. 5, No. 4 (2026): Juli, pp. 477- 483
E-ISSN:2827-878X (Online -Elektronik)

configured via a browser. Once the device was successfully connected to the network and obtained an IP address, the next step was to connect the IP phone to the PBX's Wi-Fi network. As this documentation is being presented on Figure 2. Panasonic KX-HTS32 IP-PBX Web Maintenance Console login page. the author is prompted to enter a username and password for authentication. The default credentials for the Panasonic KX-HTS32 IP-PBX device are Username: INSTALLER and Default Password

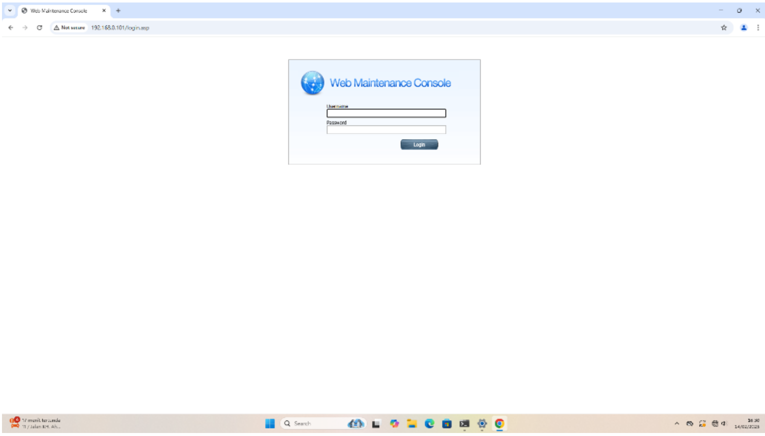

Figure 2. Panasonic KX-HTS32 IP-PBX Web Maintenance Console login page

after the login attempt author was directed onto main configuration page as presented on Figure 3 Panasonic KX-HTS32 IP-PBX Web Maintenance Console Main Page , Access to this page is gained via a browser by typing the default IP address of the IP-PBX device; in this case, the IP address of the IP-PBX is 192.168.0.101. This documentation shows the main page or dashboard of the IP-PBX after the user has successfully logged in. This page displays information regarding connected and disconnected ports, as well as the configuration menu on the left side.

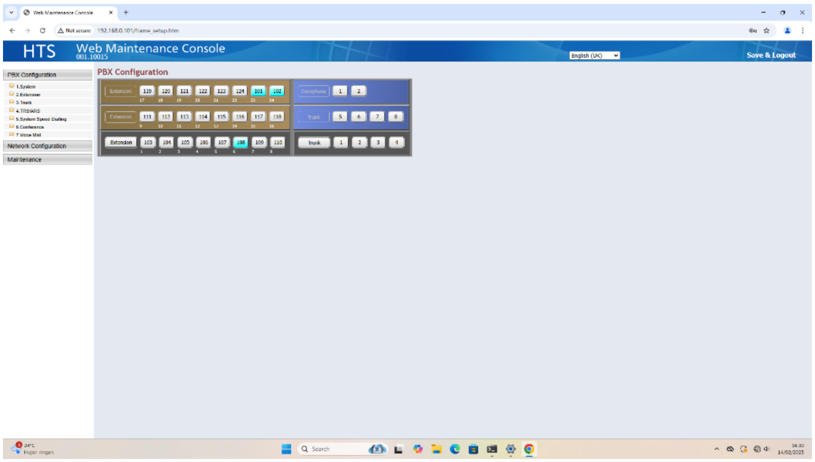

Figure 3 Panasonic KX-HTS32 IP-PBX Web Maintenance Console Main Page

Selecting menu LAN Setting page is used to configure the Wi-Fi IP of the IP-PBX, which will connect the IP-Phone devices via Wi-Fi as Presented in Figure 4 Panasonic KX-HTS32 IP-PBX LAN Setting Configuration. Here, the author used the IP address 192.168.0.101 and configured DHCP to facilitate easier device connectivity.

SOCIETY: Jurnal Pengabdian Masyarakat,
Vol. 5, No. 4 (2026): Juli, pp. 477- 483
E-ISSN:2827-878X (Online -Elektronik)

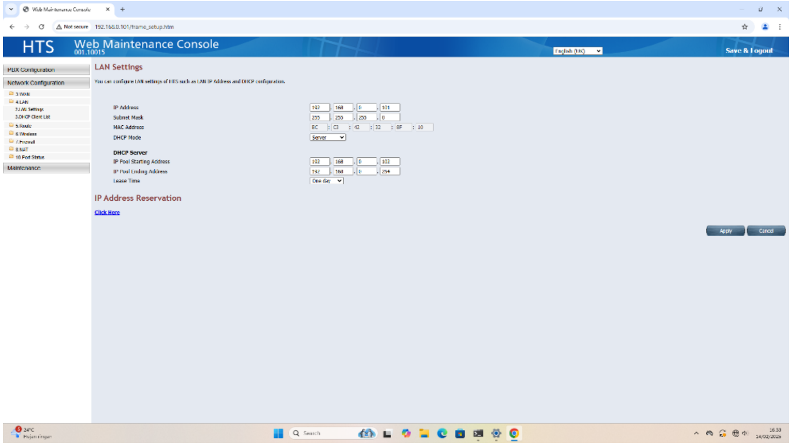

Figure 4 Panasonic KX-HTS32 IP-PBX LAN Setting Configuration

Next is the connectivity testing between the IP-PBX Wi-Fi and the IP Phones before the implementation phase as in presented on Figure 5. IP Phone device connected to the IP PBX network. The IP Phones used are four units of the Yealink SIP-T31W series; thus, with the PBX Wi-Fi IP set to 192.168.0.101, it will assign an IP range of 192.168.0.102 – 192.168.0.105. In this case, the author utilizes documentation from one of the connected IP Phones.

Figure 5. IP Phone device connected to the IP PBX network.

After all configuration steps were completed, device testing was conducted by making a call between two IP Phones that had been connected to the IP PBX Wi-Fi. As in the result presented on Figure 6 Testing IP Phone placing and receiving calls

Figure 6 Testing IP Phone placing and receiving calls

After the initial test was successful, the author conducted a network expansion simulation using a Ruijie RG-EW1200G PRO router configured as an Access Point (AP). The router was connected to the IP PBX network to extend the Wi-Fi coverage. The other two IP Phone devices were then connected to the Wi-Fi network of the router that had been set up as an Access Point, and connection testing was performed again to ensure that communication remained smooth. As presented within Figure 7 IP Phone successfully connected to the Access Point network

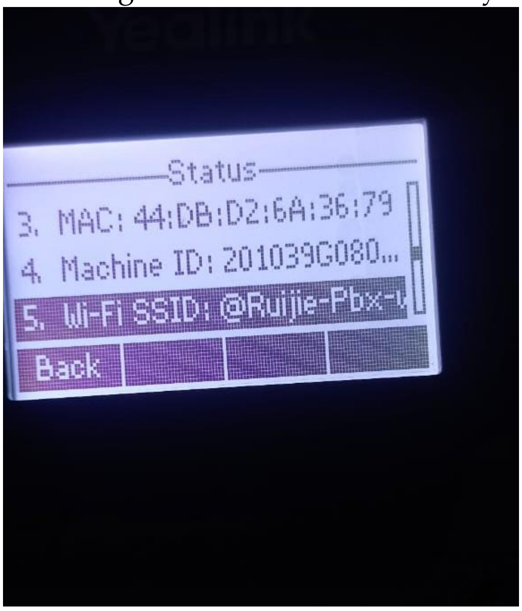

Figure 7 IP Phone successfully connected to the Access Point network

C. RESULTS AND DISCUSSION

The system was then implemented at Nusa Putra University. The implementation process followed the exact same steps as the previous trial.

The implementation activities for this VoIP-based IP Phone system were carried out in two locations: the office of PT Kaldyta Mega Tech as the initial configuration testing site, and Nusa Putra University as the direct system implementation site. This activity aimed to provide an efficient, flexible, and cost-effective internal communication medium by utilizing the existing Wi-Fi network through an IP PBX system. The stages of activities conducted during this internship and community service process can be explained as follows:

The initial stage was conducted at the PT Kaldyta Mega Tech office, located at Perum Gunung Jaya Permai, Gunungjaya, Cisaat District, Sukabumi Regency. At this stage, the configuration of the Panasonic KX-HTS32 IP PBX device was performed via a web interface. The IP PBX device was assigned a static IP address of 192.168.0.101 and accessed through a browser for network settings and its internal Wi-Fi features. Once access was successfully established, IP Phone 1 through IP Phone 4 were connected to the IP PBX's internal Wi-Fi network. These devices automatically obtained IP addresses through the PBX's DHCP feature, with the following IP ranges:

1. IP Phone 1: 192.168.0.102
2. IP Phone 2: 192.168.0.103
3. IP Phone 3: 192.168.0.104
4. IP Phone 4: 192.168.0.105

Communication testing between the devices was conducted, showing that the IP Phone devices could interconnect and make voice calls without any significant obstacles. The resulting audio was clear, with no noticeable delay, and the communication quality remained stable while connected to the IP PBX Wi-Fi.

After successfully testing the connection within the local IP PBX network, the next stage was to simulate the expansion of the IP Phone network. An additional device, a Ruijie RG-EW1200G PRO router, was configured as an Access Point (AP) and connected to the IP PBX to extend the Wi-Fi coverage. Two additional IP Phone devices were connected to the Wi-Fi network of the Ruijie AP. Call testing was conducted again between devices connected directly to the internal IP PBX Wi-Fi and those connected through the external AP. The results indicated that communication continued to function well, proving that the VoIP system's coverage could be expanded without significant disruption to call performance.

The final stage of the activity was the implementation of the IP Phone system at Nusa Putra University. The Panasonic KX-HTS32 IP PBX device was installed within the campus environment, and the configuration followed the same pattern as the previous trial stage. The IP Phone devices were reconnected to the IP PBX Wi-Fi network and to the Wi-Fi network of the external router (Access Point) to extend the coverage to a wider area. The re-testing results demonstrated that the VoIP communication system via IP Phones functioned perfectly within the campus environment. All devices could successfully place and receive calls between units smoothly. This indicates that the system is reliable for supporting internal communication within the institutional environment.

D. CONCLUSION

The implementation activities for the IP Phone-based communication system at Nusa Putra University were successfully carried out. The initial configuration process conducted at the PT Kaldyta Mega Tech office ran smoothly and served as the foundation for a broader system deployment within the campus environment. Communication testing between IP Phones connected via both the IP PBX's internal Wi-Fi network and the network expansion using an Access Point demonstrated that this VoIP system is capable of operating optimally. Communication between devices was smooth, stable, and without any significant obstacles, proving that this system can be utilized as an efficient and flexible internal communication solution.

The implementation of this system also demonstrates that IP Phone technology can be adopted by educational institutions by leveraging existing network infrastructure, without the need for complex additional cabling. Furthermore, network expansion using additional devices such as Access Points supports system flexibility in reaching wider areas. From the entire sequence of these activities, it can be concluded that the VoIP-based IP Phone system is an effective solution for enhancing internal communication efficiency, particularly for educational institutions looking to maximize the functionality of their local networks.

E. ACKNOWLEDGEMENTS

The authors would like to express their highest gratitude to Universitas Nusa Putra for providing the opportunity and network infrastructure supporting this study. Special thanks are also extended to the management and technical team of PT Kaldyta Mega Tech for granting permission, providing facilities, and guiding the authors during the initial configuration testing phase. Lastly, the authors appreciate all campus staff members who participated in the user testing and training process, making this VoIP system implementation successful.

F. AUTHOR CONTRIBUTIONS

Activity implementation: WHARG Article preparation: WHARG, Network configuration and testing: WHARG

G. REFERENCES

Toyib, R., & Wardiman, I. (2020). IMPLEMENTASI VOICE OVER INTERNET PROTOCOL (VOIP) BERBASIS LINUX (Studi Kasus SMK Negeri 03 Bengkulu). Pseudocode. www.ejournal.unib.ac.id/index.php/pseudocode

Khomar, R., & Harbani, A. (2021). Implementasi dan Optimalisasi Voice Order Internet Protocol (VOIP) Menggunakan NDLC Untuk Komunikasi Antar Lantai. Jurnal Sains dan Teknologi, 11(2), 51–60. https://doi.org/10.36350/jbs.v11i2

Fauzi, A., Setiawan, A., & Utami, D. Y. (T.t.). Perancangan Jaringan Voice Over Internet Protokol (Voip) Menggunakan Platform Asterisknow. Co-Science. http://jurnal.bsi.ac.id/index.php/co-science

Aryanta, D., Darlis, R., & Pratama, A. (2013). Implementasi Sistem IP PBX menggunakan Briker.