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VoIP (Voice over Internet Protocol) Based on Asterisk Using Trixbox at SMKN 1 Bangkinang

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Abstract: The rapid advancement of informatics technology in the telecommunications sector has remarkably mitigated the challenges of distance and time, rendering them inconsequential in today`s dynamic landscape. Presently, long-distance conversations are effortlessly facilitated through telephone media, predominantly employing the Public Switch Telephone Network (PSTN) network technology. However, at SMKN 1 Bangkinang Kota, conventional communication practices persist, leading to inefficiencies, particularly in locating teachers, staff, and employees within the school premises. This search process, whether for determining their whereabouts or utilizing smartphones, often necessitates additional time and may incur expenses related to data quotas or credits. In response to these challenges, the primary objective of this research is to conceptualize and establish a VoIP communication system, strategically designed to alleviate operational expenditures associated with communication. The study is conducted within the framework of an existing internet network established on a local scale, involving the integration of wireless and LAN cable technologies. Notably, two distinct operating systems are evaluated, and their performance is compared to identify the optimal choice for serving as a VoIP server. Through this comparative analysis, the study seeks to determine the most suitable operating system that aligns seamlessly with the project's objectives. Central to the research is the successful deployment of a VoIP server utilizing the available local network infrastructure, a feat achieved through the implementation of the Trixbox Linux server. This innovative approach harnesses the existing network facilities to facilitate cost-free communication, effectively bypassing the need for additional expenses typically associated with conventional communication tools like smartphones and PCs. A notable aspect of the system is the pivotal role played by the VoIP server, particularly the Trixbox variant, in managing Session Initiation Protocol (SIP) calls emanating from the diverse client base registered on the Trixbox server. In summation, this research showcases the transformative potential of VoIP technology within educational institutions such as SMKN 1 Bangkinang Kota. By adopting innovative strategies like deploying the Trixbox Linux server, the study underscores the capability of VoIP to streamline communication processes, optimize resource utilization, and ultimately enhance operational efficiency.

Keywords: VoIP; Server Trixbox; Internet; Smartphone.

1. Introduction

The development of informatics technology in the field of telecommunications is highly advanced and rapid. Distance and time are no longer obstacles due to the swift progress in this domain. Presently, long-distance conversations are easily conducted using telephone media, which relies on the Public Switch Telephone Network (PSTN) technology. PSTN employs a specialized cable network for voice transmission. This data communication network facilitates circuit-switched voice data transmission. Telephones are directly linked to the Private Automated Branch Exchange (PABX), wherein telephone numbers are organized based on coverage areas. However, a drawback of this medium is the elevated cost associated with long-distance connections. Voice Over Internet Protocol (VoIP) is a technology that utilizes the Internet Protocol for real-time electronic voice communication. VoIP is the prevailing telecommunications technology, and its infrastructure costs are considerably lower compared to PSTN-based technology [1].

SMKN 1 Bangkinang Kota offers a range of majors, including Audio Video Engineering, Multimedia, Refrigeration and Air Conditioning Engineering, Computer Engineering, Motorcycle Engineering, Building Drawing Engineering, Light Vehicle Engineering, Mechanical Engineering, and Electrical Engineering. As a vocational high school, the institution is equipped with a local network and an interlocal network, which serve as communication tools for data, image, and sound exchange. However, current communication methods remain conventional. Locating teachers, staff, and employees within the school can be time-consuming, as it necessitates searching for their whereabouts or utilizing smartphones that require an internet connection.

To address these issues, the author proposes a solution by designing an Asterisk-based VoIP system using Trixbox at SMKN 1 Bangkinang Kota, offering communication services through a Public IP. VoIP operates by transmitting voice data over the internet using the IP protocol, providing a cost-effective alternative to wired telephony. VoIP can significantly reduce call expenses by up to 70%. The technology works by converting analog signals into digital signals. Utilizing IP networks minimizes costs, as it eliminates the need to establish new voice communication infrastructure, resulting in lower bandwidth consumption compared to traditional telephony [2].

VoIP, as a technology, facilitates real-time voice data transmission using the Internet Protocol. The increasing interconnectivity of computers in computer networks has led to the emergence of new technologies that enable global information and data exchange, including communication through images and videos [10].

The primary objective of implementing VoIP is to diminish operational communication costs. This reduction can be achieved by leveraging existing data networks, eliminating the need for constructing new, costly infrastructure for VoIP telecommunications networks. Additionally, the performance of VoIP servers can be evaluated across wireless, internet, and wired networks. Identifying a suitable asterisk server for VoIP ensures affordable communication like conventional telephony, but at a reduced cost.

2. Research Method

The stages carried out in the research conducted are as Figure 1.

Figure 1. Stages of Research Methodology

1) Problem analysis
At this stage, analyze the needs needed for research and tool installation, such as writing needs, and collect the necessary data.
2) Literature Study
The literature study stage is used to find solutions to the problems raised by looking for references in journals and books.
3) Tools and Materials Collection
The Tools and Materials Collection stage uses laptops and Materials used in research such as Access Point, LAN Cable, PC Server, Trixbox OS, VMware Workstation application, and Zoiper application on HP and Windows.
4) Tool design
This stage makes the design of tools, such as the design and development process of tools, methods, and techniques to improve the efficiency and productivity of these tools. Figure 2 is the tool's design in the research conducted.

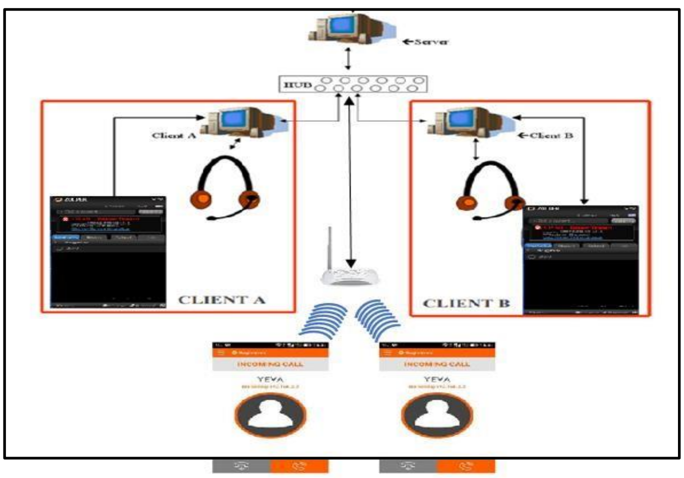

Figure 2. Tool circuit

5) Installation
The tool will be assembled with materials prepared following the design of the VoIP server via VMware Workstation, which is then connected to the PC Client and Smartphone.

6) Testing
After the tool installation process is complete, the next thing that needs to be done is testing first to check whether the tool that has been made is running as planned in the research.

3. Result and Discussion

3.1 Results

1) Testing Results
Direct video call testing results between clients can be connected for successful video calls (Figure 3).

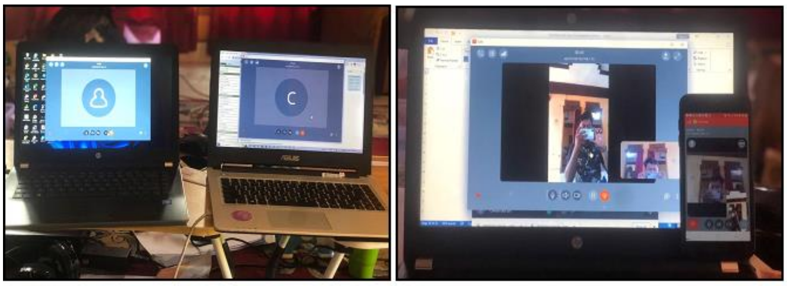

Figure 3. Testing Results

2) OS Comparison Results in VoIP Design and Implementation

In designing and implementing this VoIP, the author analyzes the comparison of the previously used Debian OS, which requires installing the asterisk package manually, with the Trixbox OS, which already has the Asterisk package in it to implement VoIP. The following is a comparison analysis table between the Operating Systems used: the Trixbox Linux server, which has an asterisk installed, with Debian 7, which must manually install the asterisk package.

Table 1. OS Comparison Results in VoIP Design and Implementation

Operating system		
Aspect Overview	*Linux Trixbox*	*Debian 7 and the Asterisk Package*
Make Phone and Video Calls	Same and already auto config Dial Plan.	With the command /etc/asterisk/extensions.conf.
	The Trixbox os can make telephone calls and video calls for local network scales in contrast to interlocal networks that can only make telephone calls and for interlocal network scales where the server IP is forwarded to the Public IP so that it can make calls from outside different networks with a record of having a User No. SIP Account registered to the server.	The Debian os can only make regular phone calls and cannot make Video calls because WebRTC is no longer supported in a separate Asterisk package on Debian.

3.2 Discussion

The discussion section critically examines the research findings and their implications, centering on the outcomes of direct video call testing and the comparative assessment of the Linux Trixbox and Debian 7 operating systems for VoIP design and implementation.

1) Testing Results of Direct Video Calls
The successful execution of direct video call testing between clients underscores a significant achievement in the implementation of the VoIP system. The capability to establish seamless video connections signifies a substantial advancement in communication within the context of this study. These favorable outcomes substantiate the primary aim of enhancing communication efficiency and reducing operational costs through the integration of VoIP technology. The positive testing results affirm the VoIP system's efficacy in facilitating real-time video communication, which has the potential to foster heightened collaboration and interaction among users.

2) OS Comparison in VoIP Design and Implementation
The comparative evaluation of the Linux Trixbox and Debian 7 operating systems for VoIP design and implementation offers valuable insights into their respective strengths and limitations. The Linux Trixbox server emerges as a robust choice, characterized by seamless integration with the Asterisk package and automated configuration of the Dial Plan. This inherent capability simplifies the process of establishing both voice and video calls, particularly within local network environments. Moreover, Trixbox's unique capacity to enable video calls across interlocal networks enhances its versatility and applicability.

Conversely, Debian 7 reveals constraints in its capacity for VoIP functionalities. While adept at supporting conventional voice calls, its deficiency in integrated WebRTC support through a separate Asterisk package renders it incapable of facilitating video calls. The manual configuration process, exemplified by the /etc/asterisk/extensions.conf command, introduces complexity and potential impediments to a streamlined video communication experience. This limitation may restrict the fluid adoption of video communication, especially in scenarios necessitating interlocal network connectivity.

The in-depth discussion of the operating system comparison underscores the pivotal role of platform selection in shaping the viability and extent of VoIP implementation. The comprehensive capabilities of the Linux Trixbox server position it as a favorable choice for institutions seeking a unified solution for both audio and video communication. Conversely, the limitations of Debian 7 underscore the importance of meticulous operating system consideration when aiming to optimize VoIP functionality. The successful outcomes of video call testing underscore the seamless integration of video communication within the VoIP framework. Moreover, the comprehensive comparison of operating systems underscores the paramount importance of strategic platform selection in optimizing VoIP capabilities. These findings collectively contribute to the research's overarching objective of enhancing communication efficiency and fiscal prudence within the examined context.

4. Related Work

Research by [11] Voice over Internet Protocol or VoIP is a term that describes the use of computer networks to perform voice communication. VoIP technology is now advanced enough to make it an efficient voice communication tool. In addition, the cost-saving factor of telephone conversations is also an attraction of VoIP. In this research, VoIP analysis of Quality of Service (QoS) on VoIP (Voice over Internet Protocol) communication network services using X-Lite softphone as software for making calls and Wireshark as a Network analyzer to measure QoS parameters in VoIP such as delay, jitter, and Packet loss, the network used is Indihome which has been installed at SMK Karya Mandiri with the aim of knowing the size of the network when the call takes place. The purpose of the research at SMK Karya Mandiri is to find out QoS parameters such as delay, jitter, and packet loss, by analyzing the VoIP network when making calls with varying times to find out the average value of the QoS parameters mentioned earlier with the Wireshark application as a network analyzer and XLite as an application for making VoIP calls.

Research by [4] VoIP (Voice Over Internet Protocol) is a technology capable of passing packetized voice, video, and data traffic over IP networks, enabling cost savings due to no need to create new infrastructure for voice communication and the use of smaller data width (bandwidth) compared to ordinary telephony. There are currently three different types of methods and the most used in VoIP services are ATA (Analog Telephone Adapter), IP Phones, and Computer to Computers. In addition to using conventional methods, the current infrastructure has also developed SDN (Software Defined Network) based infrastructure. SDN is closely related to OpenFlow, so many people think that SDN is OpenFlow. SDN was born from the OpenFlow protocol proposed by Nick McKeown and his colleagues. OpenFlow is a communication specification between the control plane and the data plane. OpenFlow is an open standard that is applied to SDN. An OpenFlow switch consists of two types, the first is a hardware-based switch, this type of switch has modified hardware by using a special OS to implement the OpenFlow protocol and the second type is a software-based switch that uses a Unix or Linux system to implement all OpenFlow functions. By considering the flexibility of OpenFlow-based SD), it can be implemented in VoIP services. The method used is Observation.

Research by [3] VoIP is the easiest way to make voice calls online by sending packets through a packet switched based network. The advantage of VoIP compared to GSM is that GSM service providers provide a tariff or fee for each call made. Facebook Messenger and Google Hangouts are examples of VoIP applications widely used by the public. LTE networks are designed to provide better spectrum efficiency. They can transmit large-capacity data with high data rates using OFDMA (Orthogonal Frequency Division Multiple Access) techniques on the downlink side and SC-FDMA (Single Carrier Frequency Division Multiple Access) on the uplink side so that VoIP service users can communicate comfortably. The method used to determine the Mean Opinion Score (MOS) of VoIP services generated by the two applications is the E-model objective method.

Research by [10] Voice Over Internet Protocol (VoIP) is a voice service into the Internet Protocol network so that it can communicate between users connected to the VoIP network using Softphone, Zoiper as a client and Trixbox as a server by changing IP to make client calls. The need for multimedia-based application services over IP networks has become possible, but basically, the data packets passed through IP networks are not made in real time, making IP networks reliable for sending real time data such as voice and video. IP-based telephone technology is often also known as VoIP (Voice Over Internet Protocol), with this VoIP voice call or video call communication will be cheaper and even free. This is because VoIP can be installed at any Ethernet and IP address, unlike traditional telephones that must have their ports in the center or PABX.

Research by [5] Voice Over Internet Protocol (VoIP) is a means of communication cheaper than analog telephones by applying existing office equipment such as laptops, computers, and smartphones without buying special equipment.

And can be applied using LAN and WLAN cables which are very suitable for application in companies or offices, which only require 2 models of devices that are used as servers with the Trixbox operating system and Softphones on client computers. The use of VoIP has advantages such as Skype, Google Talk, and Facetime replacing the way of communicating with each other. Due to the low cost, people use VoIP as an alternative to the expensive traditional Public Switched Telephone Network (PSTN).

Research by [7] The use of information technology as a communication medium has been widely used by various organizations, companies, or other agencies to support activities that require communication. Voice over Internet Protocol (VoIP) is an IP-based voice communication technology where users can communicate using only the internet, intranet, or ethernet networks. With wireless network infrastructure, VoIP technology can be used on network phones or mobile devices using softphones. PT National Label currently has implemented a PABX telephone as a means of communication, so more and more users need communication media, while the high price of PABX telephones can result in increased company operating costs, so companies must look for other alternatives so communication between rooms is not hampered and does not reduce company productivity. The methods used are Observation and interview.

Research by [8] Currently, Voice Over Internet Protocol (VoIP) technology is widely used by humans to communicate. VoIP allows sending voice data packets from one place to another via an Internet Protocol (IP) based network, along with the increasingly cheap cost of Internet broadband subscriptions, the cost of conversations via VoIP technology has become cheap. One of the Linux Operating Systems that can be used for networking is Linux Red hat 9, which is proven to be cheap and reliable in doing its job as a router. Widely used in ISPs (Internet Service Providers), routers in Internet cafes, and Gateways in offices. GNU is an operating system consisting entirely of free software, an acronym for GNU's Not Unix. The project introduced the concept of copyleft which adopts the copyright principle, but the principle is used to guarantee creative freedom. The guarantee takes the form of attaching the source code and a statement that the software may be modified as long as it adheres to Copyleft principles.

Research by [1] Voice Over Internet Protocol (VOIP) is a technology that utilizes Internet protocol to provide electronic and real time voice communication. The costs incurred for this technology infrastructure are much cheaper than the telecommunications technology commonly used by the public today. Asterisk is a soft switch for operating a proxy based on session initiation protocol (SIP). Asterisk and X-Lite are open-source software used to build VOIP. Asterisk can run on various operating systems (Windows, Linux, Mac, OpenBSD, and FreeBSD). Which authorizes developers and implementers to create better communication solutions for free.

Research by [6] The results of this study indicate the design of the Voice over Internet Protocol (VoIP) communication system using the Trixbox CE Server Based on Session Initiation Protocol (SIP) can run well with reference to Quality of Service in the form of delay, jitter and packet loss parameters tested twice, namely in the morning and afternoon using 64 Kbps, 128 Kbps and 256 Kbps bandwidth. This research uses a local network to integrate existing networks in each unit (school) within the Said Na'um Education Foundation. And the communication carried out in this study is only from VoIP to VoIP, not from VoIP to GSM, or VoIP to CDMA. This research uses Engineering Methods, including planning, needs analysis, design, testing and system implementation. The system requirements needed in this research are a server computer, client computer, switch, proxy router and smartphone.

Research by [9] VoIP (Voice Over Internet Protocol) technology is a technology that offers voice data transmission services directly (real time) using the Internet Protocol. However, VoIP communication does not have a security guarantee for the data in the ongoing voice communication, it does not rule out the possibility of other unauthorized parties intercepting the communication, such as: hijacking the contents of the data (sniffing) or unable to access the server due to server overload (denial of service). This method tries to test VoIP communication using VPN PPTP and ZRTP security methods to minimize tapping, and to know the results of testing VoIP communication while in progress.

Research by [12] Voice over IP was applied to a local area network (LAN) segment. VoIP for call establishment, traffic data collection, and quality of service (QoS) parameters were analyzed for objective and subjective evaluation. Subjective investigation parameters were R-Factor and Mean Opinion Score (MOS), and objective parameters looked at jitter and packet loss. Findings were achieved with actual network topology and appropriate traffic flow software. VoIP uses several protocols that ensure voice communication is well established between parties and voice is delivered with a quality close to the public switched telephone network (PSTN) signaling protocols such as Session Initiation Protocol (SIP) and H.323.

5. Conclusion

Considering the comprehensive exploration of VoIP technology and its practical applications, several important conclusions can be drawn. The comparative analysis between the two operating systems presented by the authors reveals a range of advantages and limitations. The decision on the ideal operating system for a VoIP server is based on considerations of convenience, technical support, and overall suitability for the specific needs of the organization. Through this meticulous assessment, organizations can make informed choices that fit their communication needs.

The successful implementation of a VoIP server in an existing LAN infrastructure, as evidenced by the implementation of a Trixbox Linux server, highlights the feasibility and potential of using available network resources for efficient communication. cost results. By leveraging VoIP technology, organizations can establish communication channels at no additional cost, providing a practical alternative to conventional communication tools such as smartphones and tablets. PC. The Trixbox VoIP server's ability to handle SIP calls from a multitude of registered customers represents a breakthrough in communication capabilities, facilitating seamless and efficient voice communications. Essentially, the study highlights the transformative impact of VoIP technology in the contemporary communication landscape. Exploring the nuances of operating systems and successfully implementing VoIP servers illustrates the potential of using technology to improve the efficiency and cost-effectiveness of communications. This study provides valuable insights that can guide organizations to harness the power of VoIP to optimize their communications infrastructure and improve connectivity.

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