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DESIGN AND IMPLEMENTATION OF ASTERISK-BASED VOIP SERVER WITH TOP-DOWN METHOD

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ABSTRACT

Asterisk is one of the most popular and versatile open-source VoIP platforms. This research aims to design and implement Asterisk-based VoIP server services using a top down method approach. The top down approach starts from planning and designing the entire system before proceeding to the detailed implementation stage. The first step involves identifying the organization's communication needs and determining the technical specifications required to meet those needs. Next, hardware and software selection, network topology, and basic configuration of the Asterisk server are carried out.

After the system architecture is determined, the implementation stage is carried out by installing and configuring the Asterisk server according to the design that has been created. The configuration process includes setting up communication protocols, setting extensions, system testing is carried out to ensure that all functions run well and meet user needs.

The results of this research show that the top down method is effective in building Asterisk-based VoIP server services that are reliable and suit organizational needs. The system built is capable of providing good sound quality and ease of management. Asterisk-based VoIP implementations also offer significant cost savings compared to traditional telecommunications systems.

ARTICLE INFO

Keyword:

VoIP, Asterisk, Top Down Method, Communication, Open-source.

1. Introduction

The development of information technology in the telecommunications sector is very advanced and rapid. Distance and time are no longer an obstacle due to the rapid progress in this field. Today, Voice Over Internet Protocol (VoIP) is a technology that leverages the Internet Protocol for real-time electronic voice communication. (Personal, 2023).

The DCI College of Informatics and Computer Management, abbreviated as STMIK DCI Tasikmalaya, is a college under the auspices of the Digita Loka Foundation. The Informatics Engineering Study Program is a study program for the Bachelor of S1 level and the Informatics Management Study Program is a study program for the D3 Associate Expert level. As a College of Informatics and Computer Management, DCI is equipped with local networks and interlocal networks, which function as data and voice communication tools. To be able to communicate with each other Lecturers, students and staff on campus need a SIP (Session Initiation Protocol) application This session is an exchange of data between users which includes voice, video, and text. For example, WhatsApp that requires an internet connection. Thus, to reduce the level of communication with expensive costs, we can use VoIP (Voice Over Internet Protocol) technology which can be used by utilizing the UDP (User Datagram Protocol) Protocol, data transmission is fast so that it is more efficient and does not take a long time.

To overcome these problems, the author proposes a solution with DESIGNING VOIP SERVER ASTERISK IN STMIK DCI USING THE TOP DOWN METHOD at the DCI College of Informatics and Computer Management. VoIP operates by transmitting voice data using the IP protocol, providing a cost-effective alternative to landlines.

2. Theoretical basis

2.1. Method Top Down

The Top Down method is a method used to build, namely a local network where each item is built simultaneously using the same specifications in a work project (Fauzi et al., 2022).

The author designed as follows:

1.Needs Analysis The need to collect data on all components that will be used to build a VoIP network.

2.Design At the design stage, the design of the network schema, network topology and IP address allocation along with the extension number to be used.

3.Testing At the testing stage where testing is carried out on the communication network that has been created.

4.Implementation is the last stage that is carried out, namely installing a communication network that has been completed and tested.

2.2. Virtual Box

VirtualBox is a free, open source and multi-platform Virtual Machine application that can be used for simulators running other operating systems in an already installed operating system. Of course, this is different from dual boot OS or multi boot OS in one pc (Manalu & Sitanggang, 2019).

Some of the features and advantages of VirtualBox include:

1.Free: VirtualBox is available for free for virtualization purposes.

2.Cross-platform: Compatible with various operating systems such as Windows, Linux, and macOS.

3.Snapshot: The snapshot feature allows users to save the state of the virtual system at a specific point in time, making it easier for users to back up or restore the virtual system to its previous state in case of a problem.

4.Hardware compatibility: VirtualBox supports different types of hardware such as graphics

cards, network interfaces, and USB devices.
5.Networking: VirtualBox can simulate different types of networks, including NAT, bridged, and internal networks.
6.Plugins: VirtualBox supports plugins that allow users to extend the functionality and capabilities of VirtualBox.
7.User-friendly: VirtualBox has an easy-to-use and intuitive user interface.
VirtualBox is suitable for testing operating systems, applications, or network environments before deploying them in a production environment. In addition, VirtualBox is also useful for software development and testing, as well as for isolating applications or operating systems from the surrounding environment (Pederson et al., 2023).

2.3. Linux Server

The word "Linux" for now is familiar to internet users and the student community who have a hobby to try new software. Technically and briefly, Linux is a multi-user and multi-tasking operating system, which can run on various platforms including Intel 386 processors and higher. This operating system implements the POSIX standard. Linux can interwork well with other operating systems, including Apple, Microsoft and Novell.
Debian is an open-source operating system that, in the last 25 years, has evolved independently through the collective action of countless developers. The history of Debian shows many small transitional steps but only 14 important stable releases (evolutionary steps) that progressively changed the structure of its network.(Villegas et al., 2020).

2.4. Asterisk

Asterisk is an open source framework for building communication applications. Asterisk turns a regular computer into a communication server. Asterisk supports IP PBX systems, VoIP gateways, conference servers, and other custom solutions. It is in constant development, therefore it is a very lively project with constant updates and improvements. For all this, it has established itself as one of the world's leading IP-based PBX engines.(Diamond, 2020).

2.5. VoIP

Agustin et all (2015), stated that voice communication through the telephone can be done through the internet without cost and distance limitations. VoIP (also called IP Telephony, Internet telephony or Digital Phone) is a technology that enables long-distance voice conversations over internet media. Voice data is converted into digital code and streamed over a network that transmits data packets, rather than through an analog telephone circuit. In VoIP communication, users make a phone connection through a terminal in the form of a PC or Android.

By making calls using VoIP, many advantages can be taken, including in terms of cost is clearly cheaper than traditional phone rates, because the IP network is global. In addition, maintenance costs can be reduced because voice and data networks are separate, so IP Phones can be added, moved and changed. This is because VoIP can be installed on any ethernet and IP address, unlike conventional phones that must have their own port in the central or PBX (Private branch exchange).

2.6. PortSIP

PortSIP provides a complete modern integrated communication solution to service providers, enterprises and critical infrastructure sectors globally. PortSIP clients include HPE, Qualcomm, Agilent, Keysight, CHUBB, Netflix, Nextiva, FPT, Panasonic, Softbank, Telstra, T-Mobile, Siemens, BASF, Queensland Rail, and others. We engage deeply with our customers, helping them modernize their communications to improve their competitive position and business outcomes in today's smart, always-on, and data-hungry world.

2.7. Cisco Packet Tracer

Cisco Packet Tracer is a powerful network simulation program that allows students to experiment with network behavior and ask "what if" questions. As an integral part of the Networking Academy™'s comprehensive learning experience, Packet Tracer provides simulation, visualization, authoring, assessment, and collaboration capabilities and facilitates the teaching and learning of complex technology concepts.(Sasanto, 2018).Packet Tracer complements the physical equipment in the classroom by allowing students to network with an almost unlimited number of devices, encouraging practice, discovery, and problem-solving. Simulation-based learning environments help students develop 21st-century skills such as decision-making, creative and critical thinking, and problem-solving.

3. Problem Analysis

3.1. Computer Network Analysis

At this stage of analysis, what is carried out is to analyze the research background that is adjusted to the research site of STMIK DCI Tasikmalaya City. The computer network that is currently running at STMIK DCI Tasikmalaya City is analyzed and then the results are used as research materials. The result of this analysis stage is the outline of the compressor network that exists where the research is needed and will be used in the process of making a compressor communication network.

3.1.1. Front Computer Network

Table 3.1 Campus Front Network.

No	Network Hardware	type	Specifications	Purpose
1.	Switch	2950-24	It features 24 FastEthernet ports and 100 Mbps bandwidth**.**	redirects data traffic between devices in the local network.
2.	Router	CISCO2911 / K	Integrated 3-port 10/100/1000 Ethernet interface (RJ-45 only) 512 MB (installed) / 2 GB (max) 256 MB (installed) / 8 GB (max)	connect at once to the internet network.
3.	Internet Service Provider (ISP)	Biznet	Banwidth 300 Mbps.	So that the server can be accessed from the internet and the local network gets an internet supply.
4.	Mikrotik	hex RB750Gr3	- Processor Baru (MediaTek 2 Core 4 threads - 880Mhz) - RAM 256MB - Slot USB - Slot MicroSD	The internet in front of the campus is accessed using the WiFi network.

Information:

Based on Table 3.1 above, the campus front network is quite qualified for the creation of a VoIP communication network, but it is still less effective if the VoIP network is designed on a server, therefore the VoIP network will be designed by the Client PC so that it can be connected to the localnetwork.

3.1.2. Internal Computer Networks

Table 3.2 Campus Internal Network.

No	Network Hardware	Sum	Process	Purpose
1.	Pc Lab 1	24	A PC connected to the Switch will get the dhcp ip from the router and connect to the server as well as internet access.	To learn from students, and to process data.
2.	Pc Lab 2	24		
3.	Pc Lab 3	24		
4.	Pc R Theory 4	1		
5.	Pc R Theory 5	1		
6.	Office Pc	1		
7.	Pc Half	1		
8.	Pc Lecturer	1		
9.	Pc Foundation	1		
10.	Switch	4	Each Switch has 24 ports for use by all Pc Users, Accsess Point networks within the campus and connected to the router get the dhcp ip address of the router as well as internet access.	redirects data traffic between devices in the local network.
11.	Access Point	3	The access point connected to the Switch receives the IP address from the router and at the same time internet access.	Internet access inside the campus.

Information:
Based on Table 3.2 above, the campus front network is quite qualified for the creation of a VoIP communication network, but it is still less effective if the VoIP network is designed on a server, therefore the VoIP network will be designed by the PC Client so that it can be connected to the localnetwork.

3.1.3. Running Computer Network in STMIK DCI

At this stage, the computer network on campus will be designed using Cisco packet tracer software to facilitate network design. The network on campus is divided into 2 parts, namely the front and inside the campus network.

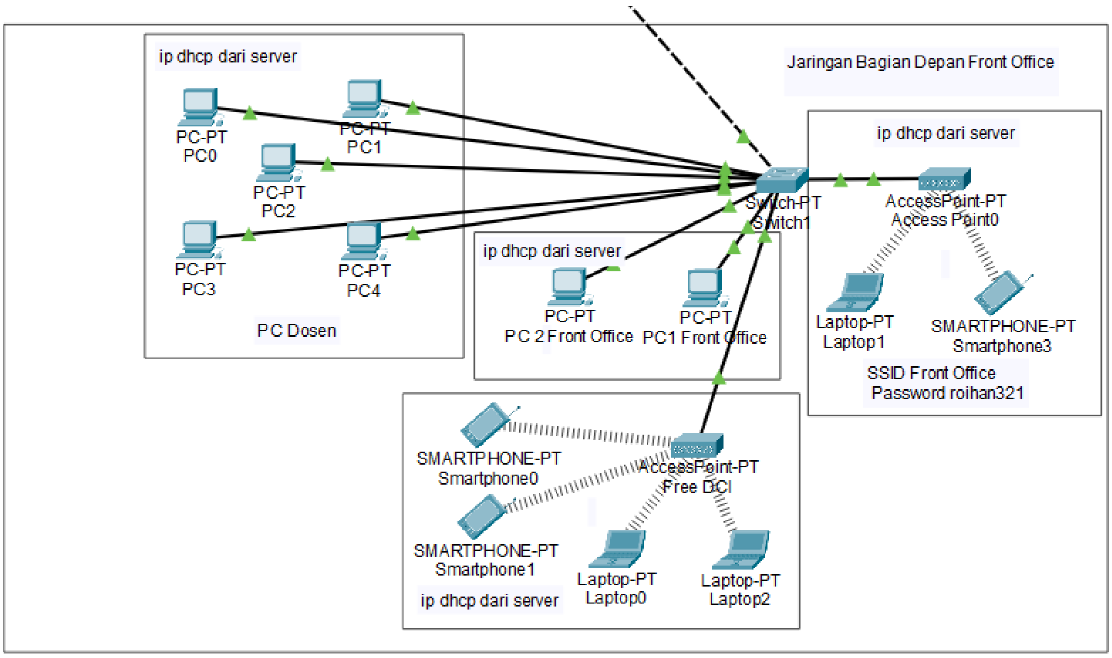

Figure 3.1 Campus Front Network.

Information:
In the picture above, the front computer network that is running at STMIK DCI consists of several hardware, namely 5 Lecturer PCs, 2 Front Office PCs, 2 AccessPoints and 1 Switch. The Lecturer's PC will receive the IP address from the swicth connected to the server and the Front Office PC will receive the same IP address from the server. For 2 AccessPoints receive the ip address of the switch connected to the server. All PCs and AccessPoints connected to the switch will receive an ip address from the server.

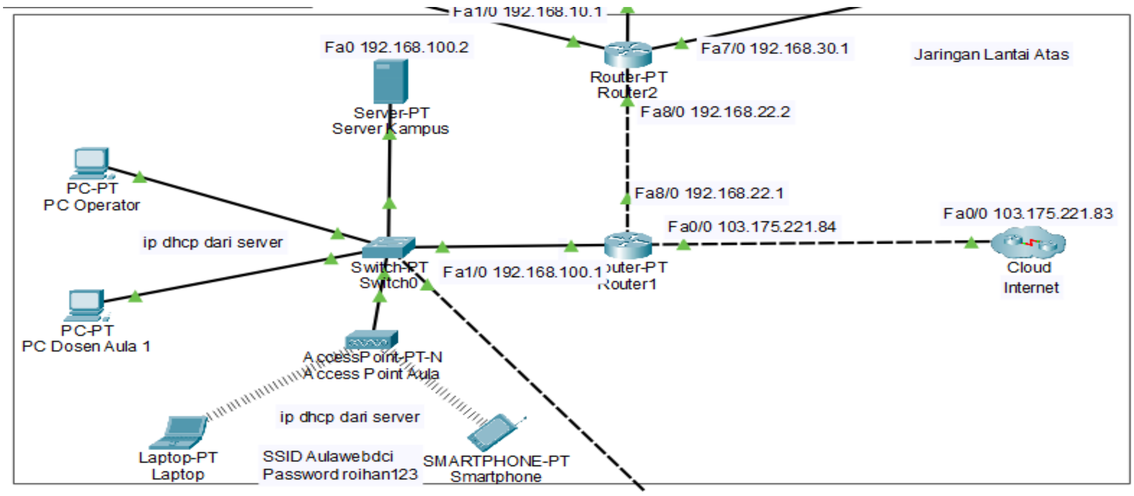

Figure 3.2 Server Network.

Information:

In the picture above, the network of the campus server section consists of several hardware, namely 1 Server, 1 Switch, 2 Routers, 1 Internet Service Provider (ISP), 1 PC Operator, 1 PC Lecturer Hall and 1 AccessPoint Hall. For servers with an ip address of 192.168.100.2 and creating a DHCP Client IP DHCP (Dynamic Host Configuration Protocol) is a protocol used to automatically configure IP addresses and other network settings on devices connected to the network.

After creating the DHCP IP the server will connect to the switch and the device connected to the swicth will receive an automatic ip address from the server. To connect to the internet, you need 1 router that is connected to the Internet Service Provider (ISP) and gets the public IP given, which is 103.177.221.83. After getting a public IP for access to the local network, RIP Routing Information Protocol (RIP) is a routing protocol used to determine the best routing path for the data packet to be sent so that the local network can access the internet.

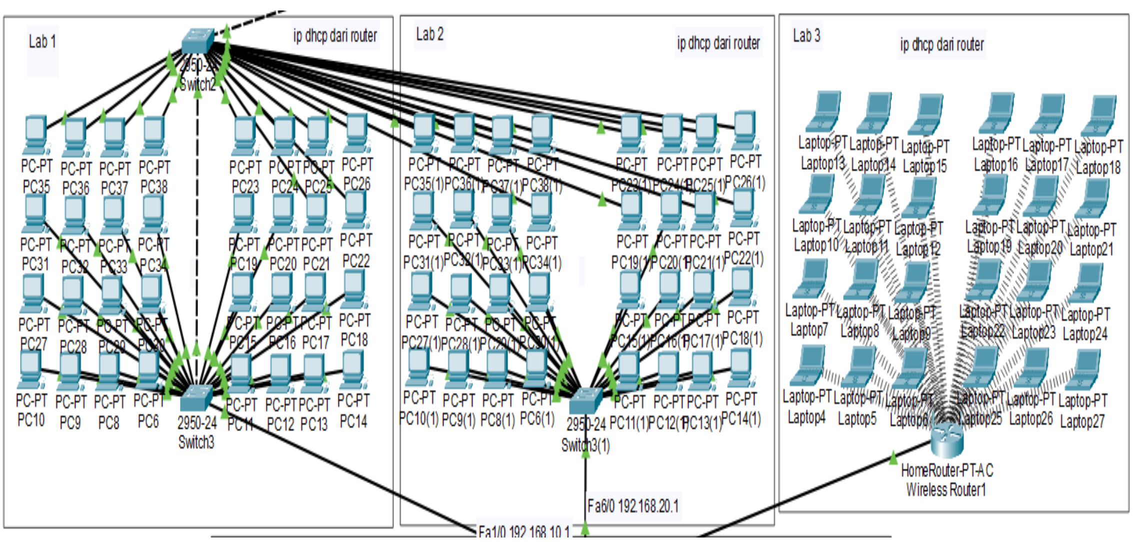

Figure 3. 3 On-campus networks

Information:

In the picture above, the network inside the campus consists of 4 parts, namely the network of Lab 1, Lab 2, and Lab 3. It consists of various hardware, namely 3 Swicth, 1 Accesspoint Router, 64 PC Clients and 24 Leptops. For Lab 1 and Lab 2 networks, it has a Swicth 24 Port Fast Ethernet. Fast Ethernet is an Ethernet network technology that has a data transmission speed of 100 megabits per second (Mbps). Each PC gets the ip address from the Router to distinguish the Lab 1 and 2 networks, it is necessary to provide the ip for the Lab 1 network is the ip address 192.168.10.1 and Lab 2 the ip address is 192.168.20.1 and for Lab 3 it is 192.168.30.1. The router will combine 3 networks to be able to connect to each other to the internet and the server is RIP routing on the router connected to the Internet Service Provision (ISP) to enter the ip address permission of the third Local Area Network (LAN) network to access the internet.

For AccessPoint Router, the configuration is to create a DHCP Client ip with IP 192.168.30.1, then the leptop in lab 3 will receive the DHCP ip from the AccessPoint Router and automatically connect to the internet because it has received the ip from the router connected to the Internet Service Provider (ISP).

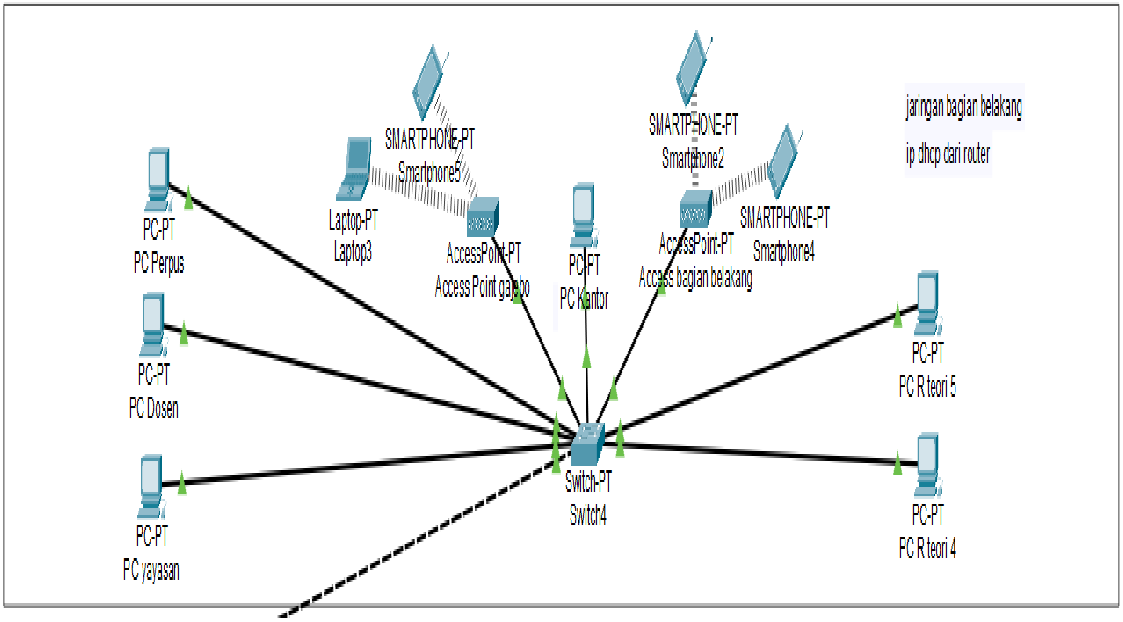

Figure 3. 4 Back-end network of campuses.

Information:

In the picture above is the network scheme of the back of the campus which is made up of several hardware, namely 1 Switch, 2 AccessPoint and 3 Lecturer PCs. For devices connected to swtich will receive an ip address from swicth lab 1 using a cross over network cable For Cross type cables there are several color sequences of Straight cables that are reversed.

Another difference is that Cross cables are used to connect the same device. AccessPoint connected to swicth will receive the ip address of the first lab router, i.e. with the automatic DHCP ip address 192.168.10.1 of the router connected to swicth lab 1.

3.2. Local Network Specifications

3.2.1. Hardware Specifications

Table 3.3 Network Hardware Specifications.

No	Network hardware	Type	Specifications
1.	Router	CISCO2911/K	Integrated 3-port 10/100/1000 Ethernet interface (RJ-45 only) 512 MB (installed) / 2 GB (max) 256 MB (installed) / 8 GB (max)
2.	Switch	2950-24	It features 24 FastEthernet ports and 100 Mbps bandwidth.
3.	Access Point	TL-WR840N	Frequency 2.4-2.4835GHz 4 10/100Mbps LAN PORTS 1 10/100Mbps WAN PORT
4.	Access Point Router	RT-AX53U	RJ45 for Gigabits BaseT for WAN x 1, RJ45 for Gigabits BaseT for LAN x 3, USB port x1 128 MB Flash 256 MB RAM Frequency 2.4G Hz / 5 GHz

Information:

Based on Table 3.3 above, the campus front network is quite qualified for the creation of a VoIP communication network, but it is still less effective if the VoIP network is designed on a server, therefore the VoIP network will be designed by the Client PC so that it can be connected to the local network. This results in the required reporting process being a little slow and inefficient.

3.2.2. Specifications Required by PC Client

Table 3.4 Specifications required for Pc VoIP Client.

No	Type	Name	Specifications
1.	Operating System (OS)	Windows 7,8,10 dan 11	Minimum 2 GB RAM, dual core processor, 250 GB storage, camera/webcam and has an audio port.
2.	Softphone on Windows	Port SIP	
3.	Softphone on Smartphone	PortSIP.	Mininmal RAM 3 GB Operating System (OS) Android dan IOS.

Information:

Based on Table 3.4 above, the campus front network is quite qualified for the creation of a

VoIP communication network, but it is still less effective if the VoIP network is designed on the server, therefore the VoIP network will be designed by the PC Client so that it can be connected to the local network. This results in the required reporting process being a little slow and inefficient.

4. Result and Discussion

4.1. Design That Will Be Computerized

Based on the problem analysis, in the system being used there are several procedures, which are formed from several LAN networks that are located at the research site. The LAN network is still being done manually, VoIP will be installed in the Server section. The following procedures will be designed and implemented with a computer Communication Network using VoIP Asterisk as follows:

4.2. Proposed VoIP Network

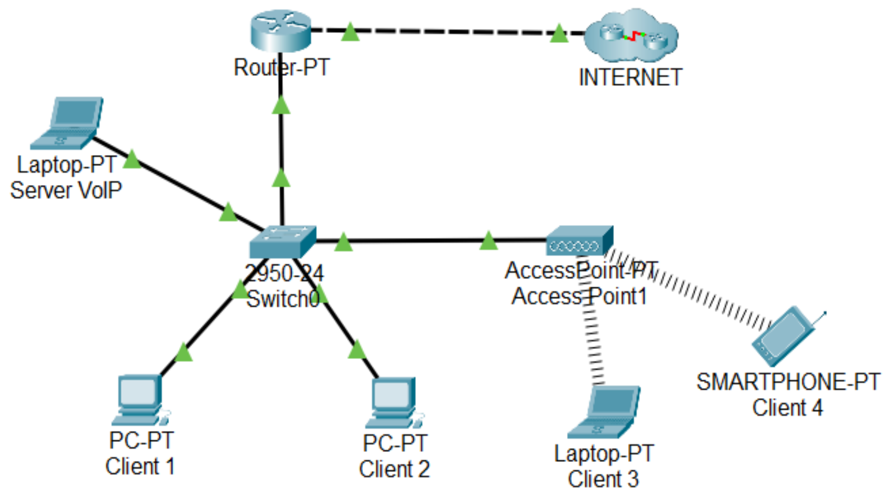

Figure 4. 1 Proposed Network.

Information:

In the network scheme, the image above consists of internet service provider (ISP), router, swicth and Access point. For the voip server, leptop is used as a service provider and 2 pcs to be used as a client. Networks connected to access points can be used as clients.

The advantage is that if the local campus network that is connected to the internet has a network problem that cannot communicate VoIP, this can be an alternative to communicating internally can be done without internet access. This VoIP can also make real-time video calls with quite good video quality, making it easier for lecturers, staff and students to carry out alternative communication in the event of network problems.

4.2.1. Installation of asterisks on Linux

```
root@roihan:/home/roihan# asterisk -rvvv
Asterisk 13.14.1~dfsg-2+deb9u3, Copyright (C) 1999 - 2014, Digium, I
nc. and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty
' for details.
This is free software, with components licensed under the GNU Genera
l Public
License version 2 and other licenses; you are welcome to redistribut
e it under
certain conditions. Type 'core show license' for details.
====================================================================
=====
Connected to Asterisk 13.14.1~dfsg-2+deb9u3 currently running on roi
han (pid = 411)
roihan*CLI>
```

Figure 4.2 Testing Asterisk.

Information:

After the server login process via cmd, whether the asterisk has been installed using the asterisk -rvvv command. To view the state of a service in Linux using systemctl, you can use the systemctl state service-name command. This command will provide detailed information about the status of the service, such as whether the service is running, error messages, and more.

4.2.2. Required File Configuration

To be able to make a phone call on the asterisk service, it is necessary to make the confirmation on the side of the asterisk service. All configurations present in this asterisk are stored in /etc/asterisk. The files we need to edit these are sip.conf and extensions.conf. which is to manage the user phone call along with the additional modules that are configured in this configuration. User creation on VoIP client Edit sip.conf file in /etc/asterisk/sip.conf.

Table 4.1 Configure the Sip.conf file.

Codec	information
[General]	The general configuration part of sip.conf includes variables.
Context=internal	This is the default context that will be used for phonetics that don't have context. The contents of the context can be set in /etc/asterisk/extensions.conf.
Port=5060	UDP Port to listen for incoming connections. Default 5060.

bindaddr=0.0.0.0	The IP address that is bound (in the bind) where the connection is listened to. default 0.0.0.0 (all interfaces).
allow=h264	Enable standards for video compression that can provide good video quality.
Srvlookup=yes	enable SRV DNS checking at the time of the call. default no.
Allowoverlap=no	yes|no enable/disable overlapping call support.
Videosupport=yes	Enables support for video SIP.
Outfocall_message_context=internal	External Context Message Call.
Accept_outfocal_message=yes	Yes is an independent protocol for processing text messages outside of calls. Messages are routed through the Asterisk dialplan.
Static=yes	Static parameter setting
Writeprotect=no	If writeprotect=no and static=yes, then you can save the current dialplan.
Clearglobalvars=no	Reloading an asterisk if it is not in the setting will remain there when reloaded.

Table 4.2 Create a User Sip.

User sip	Information
[test1] type=friend host=dynamic secret=test1 context=internal	User sip test1 Password test1
[test2] type=friend host=dynamic secret=test2 context=internal	User sip test2 Password test2

[test3] type=friend host=dynamic secret=test3 context=internal	User sip test3 Password test3

Table 4.3 Create User Extensions.

User sip	information
exten => test1,1,Dial(SIP/test1,60)	If user test1 makes a call, it will wait for 60 seconds or more, and 60 seconds will automatically terminate the call.
exten => test2,1,Dial(SIP/test2,60)	
exten => test2,1,Dial(SIP/test2,60)	

5. Conclusion

Test results show that this VoIP system functions well in terms of connectivity and voice and video quality. Furthermore, the Asterisk configuration, performed through the sip.conf and extensions.conf files, proved capable of handling multiple users (multi-client) and supporting modern features such as video calls.

This system is considered efficient, flexible, and suitable for implementation in educational settings or other organizations requiring cost-effective yet reliable communications. For further development, integration with mobile systems and strengthening of VoIP network security are recommended.

Reference

Berdiamond, B. (2020). Building an Asterisk-based VOIP Server. *Infotama Media Journal*, *16*(1), 24–31. https://doi.org/10.37676/jmi.v16i1.1117

Fauzi, A., Setiawan, A., & Utami, D. Y. (2022). Designing a Voice Over Internet Protocol (VoIP) network using the Asterisknow platform. *Computer Science (CO-SCIENCE), 2*(1), 19–25. https://doi.org/10.31294/coscience.v2i1.887

Manalu, A. S., & Sitanggang, S. S. (2019). Design and implementation of private cloud storage with owncloud on a local network using virtualbox. *Journal of Computer Networks, Architecture, and High-Performance Computing*, *1*(2), 60–71. https://doi.org/10.47709/cnahpc.v1i2.244

Pederson, M., Fitria, N., Elinda Sari, R., & Yanti, Z. (2023). DNS Server Implementation on the Ubuntu Operating System Using VirtualBox. *Journal of Network and Computer*, *2*(2), 52–62. https://jurnal.netplg.com/

Pribadi, A. (2023). VoIP (Voice over Internet Protocol) Based on Asterisk Using Trixbox at

SMKN 1 Bangkinang. *International Journal Software Engineering and Computer Science (IJSECS)*, *3*(2), 65–70. https://doi.org/10.35870/ijsecs.v3i2.1275

Sasanto, K. A. (2018). Configuration and Performance Analysis of OSPF Routing on LAN Networks with Cisco Packet Tracer Simulator version 6.2. *Electro Engineering Studies*, *1*(1), 67–78.

Villegas, P., Muñoz, M. A., & Bonachela, J. A. (2020). Evolution in the Debian GNU/Linux software network: Analogies and differences with gene regulatory networks. *Journal of the Royal Society Interface*, *17*(163). https://doi.org/10.1098/rsif.2019.0845