Optimizing Branch Telephone Networks for Campus VoIP with Mobile Clients

Rahmad Hidayat*, Ninik Sri Lestari, Ahmad Sujana, Herlina and Givy Devira Ramady

Sekolah Tinggi Teknologi Mandala Bandung, Bandung, Indonesia

*rhidayat4000@gmail.com

Abstract. During working hours, employees on campus are not always in their room because of other activities in other rooms both in one building and in different buildings. In this study, the development of a branch telephone communication network or private branch exchange (PBX) uses voice over internet protocol (VoIP) technology with a mobile-based client device and utilizes the internet as a media. For system design, network development life cycle (NDLC) method is used where VoIP communication network is monitored with paessler router traffic grapher (PRTG) software to see quality of service (QoS) values with packet loss, delay, and jitter parameters. The results of the monitoring of each parameter are of perfect average which can be seen from the amount of packet loss value of 0%; delay below 150 msec for sending data packets from the transmitter to the receiver; and jitter in a proper category with a time below 75 msec. However, for better development, the installation and configuration of servers that have been created virtually can be implemented on the actual server system; IP usage which is devoted to VoIP communication networks; bandwidth management to provide a separate traffic line for the VoIP communication network that is created.

1. Introduction

Some buildings on campus are generally separated from each other so that a communication network is needed for information delivery. Meanwhile, the use of one branch telephone from a PBX for one staff member in several rooms such as academic rooms, general sections, finance and other rooms is still applied on many campuses. Also, during working hours employees are not always in the room because of activities that allow employees to go to other staff rooms. Sometimes there is a telephone call, but the intended employee is not in place, or there are no employees in his room, so the caller must make a redial or ask the employee who picks up the telephone to deliver information to the intended employee.

Voice over Internet Protocol is a technology that can configure a separate communication network in the form of a voice call to handle communication networks. The use of this VoIP communication network can reduce operating costs because the software used for servers and clients can be downloaded for free on the internet.[1] The problem is how to design and develop solutions in the form of a PBX branch communication network using VoIP where the client can be mobile-based to communicate between employees who are connected on a campus network. This study aims to develop a PBX branch communication network on the STT Mandala campus by utilizing internet facilities as a medium for branch communication networks and using VoIP so that calls can be made to targeted employees via a smartphone. Thus, it is expected that the delivery of information through
communication between employees will be more effective and can reduce operational costs; the use of internet network in the form of voice calls will be able to provide easy and inexpensive communication services; and can streamline call service to targeted employees even though the employee is outdoors.

In the research [2] conducted the making of VoIP server that uses the Linux operating system as the main foundation with opensource-based Asterisk and RasPBX applications which are integrated into mini-PCs in the form of Raspberry Pi. Trial of QoS performance and VoIP service server performance when making calls, the average results are sufficient to meet the QoS standard, and the percentage of server CPU usage (57%) with the number of clients that can be served is 12 clients by simultaneously communicating. Then in [3], it is stated that VoIP can be built using freeware applications, such as X-Lite and Axon. To test sound quality, VQManager is used so that it can analyze delay, jitter and packet loss. The WAN network is simulated using a router. The process model used in the development of the VoIP system is a classic life cycle, in which the model proposes a systematic and sequential approach in its development. Although using applications that are freeware, in general, VoIP systems that are built can operate well and have satisfying sound quality. Meanwhile [4] concluded based on the manufacture of an Asterisk-based VoIP server with the Fedora 8 operating system, which VoIP servers could be implemented at the Sahid University of Surakarta. VoIP has six lines that can be used simultaneously. VoIP is installed on 22 points of the room which are equipped with speakerphone and microphone as audio. VoIP requires for one user per call is 25.5 kbps for incoming and outgoing. VoIP here can also do out of the Sahid University Surakarta network by connecting it to the internet. Based on the related studies above, it can be said that the development of VoIP trials using a client is required to be mobile, especially in anticipating the use of smartphones in the next generation of 5G cellular technology with increasing speed. [5]

2. Material and method
The method of network development life cycle (NDLC) as a key model behind the process in computer networks design. [6]

![Figure 1. Model of network development life cycle.](image-url)
to be built, and at this stage, the teamwork will be tested in the field to solve technical and non-technical problems. After implementation, the monitoring phase is also an important stage, so that computer and communication networks can run according to their original wishes and goals. And finally at the management or regulation level, one of the special concerns is the policy issue. Policies need to be made to regulate the system that has been built to run well and last a long time. [7]

2.1. Data Collection and Analysis
Steps taken in this stage include interviews with employees of the academic section and the general section of STT Mandala Bandung to find information about the current communication network system in the form of verbal information or documents; observation or observation of research objects to understand the needs of the system being built; and third, document study. [8] Next, data analysis with a qualitative approach is by interpreting tables, graphs, or figures that are there and then doing description and interpretation.

2.2. VoIP Communication Network Design with Mobile Based Clients
In designing, the VoIP communication network scheme was first designed as a general description of the existing design.

![VoIP communication network design configuration](image)

**Figure 2.** VoIP communication network design configuration.

Each client that will communicate (call) must pass through the transmission media, and the device that is connected with the main purpose is internet and forwarded to the line that is connected with the next destination is the virtual server, then return to the internet path and then go to the path of the connecting device the intended client. While the steps to do the system design that will be made are shown in figure 3.
In this study, existing servers use the Trixbox operating system that is installed and configured virtually using the Oracle VM VirtualBox application. Client tested using a laptop with a Windows OS and an Android OS smartphone that supports VoIP configuration. Then the measurement and analysis carried out in the form of subjective connection quality test and monitoring of commonly used QoS parameters, namely: packet loss, delay, and jitter. The parameters that are monitored are packet loss, delay, and jitter that are calculated and seen the results obtained from the PRTG software are categorized based on the standardized QoS version QoS parameter quality categories based on ETSI TR 101 329 v.2.1.1, 1999-06, page 26. [9]

Table 1. TIPHON version QoS parameter quality standardization categories.

|                | Packet loss | Delay  | Jitter |
|----------------|-------------|--------|--------|
| Perfect        | 0%          | < 150 ms | 0 ms |
| Good           | < 3%        | < 250 ms | < 75 ms |
| Medium         | < 15%       | < 350 ms | < 125 ms |
| Poor           | < 25%       | < 450 ms | < 225 ms |

The place that is used as the object of research by the author is STT Mandala which is located on Soekarno Hatta street no. 597 Bandung.

3. Results and discussion

3.1. Quality of service (QoS) monitoring

Quality of Service monitoring is done with the help of PRTG supporting software to analyze QoS with parameters of packet loss, delay, and jitter. From the results of monitoring the VoIP communication network parameters that have been obtained based on the events that occur in the field using PRTG software, then the results of the graph are also derived from the features available in the PRTG software, namely as follows:
In the experiment obtained a graphical form from the results of monitoring parameters, where: packet loss with the color of the blue chart; delay with green color graphics; while jitter with brown graph colors. The monitoring results obtained from the PRTG software above can be seen for the second test if categorized based on table 4.1, packet loss is 0% which means that the communication that has been tested is in a very good category (perfect) because there are no lost packages when do testing, for delay worth 93 msec which means it is in a very good category (perfect) because when the time of sending the data packet from the transmitter until it is received by the receiver has still less than <150 msec at the time of testing, while jitter is worth 18 msec which means it is in a good category because the interval of arrival between data packets that come and are collected first in the jitter buffer for a predetermined time until the package can be received at the receiver side has a time below <75 msec when testing.

| Value      | Explanation |
|------------|-------------|
|Packet loss (%)| 0           | Perfect    |
|Delay (ms)  | 93          | Perfect    |
|Jitter (ms) | 18          | Good       |

3.2. Connection Quality Test
Test the quality of the connection on the VoIP network is carried out as many as three scenarios to see the system that has been designed to run well or not. From the results of the trial, two problems occurred which made the process of the call failed when conducting the test. The first problem is when testing client 1001 (smartphone 1) that will make calls on client 2001 (smartphone 2), the client is connected to a hotspot network but is not connected to the VoIP server because the client is separated from the VoIP network. Then the second problem when doing client testing 2002 (softphone) which will make a call on client 1001 (smartphone 1), the access point restarts by itself, so that the internet network source is disconnected from the server and causes a system failure that should work properly on the server VoIP, where the IP obtained (as the host IP to connect each client to the server on the VoIP network) comes from the access point device that stops working. These problems arise due to access points that are overloaded in providing internet access because of a large number of users accessing the hotspot network simultaneously; servers that are still run through virtual machines and use host IPs.
4. Conclusions
In the system monitoring on the VoIP communication network that is designed, it can be proven that the system is running well, with parameter values of packet loss, delay, jitter that are within the TIPHON standardization range. However, to test the quality of the connection, the system that has been running still experiences some problems from the hardware side such as a client that suddenly detaches from the hotspot network; access points that experience more burden in providing internet access due to the number of users accessing the hotspot network simultaneously; Access point devices that restart automatically, servers that are still run through virtual machines and use host IPs.

For better development, it can be installed and configured the server that was created virtually into the actual server system; using an access point specifically for VoIP networks that have a wide and strong signal coverage to reach all buildings on a campus; activation of the auto-connect feature to anticipate if the client device is separated from the hotspot network so that it can be automatically reconnected; and managing bandwidth so that VoIP communication networks are more organized and have their traffic lines.

References
[1] Purbo, O W 2007 VoIP: cikal bakal telkom rakyat (Jakarta: PT Prima Infosarana Media)
[2] Timoryansyah, A S H, Hafidudin and Ramadhan, D N 2015 The implementation of voip server using mini pc e-Proceeding of Applied Science 1(3) 2624-2631  
[3] Wahyuddin, M I 2009 Implementasi voip computer to computer berbasis freeware menggunakan session initiation protocol Jurnal Artificial ICT Research Center UNAS 3(1) 50-59 
[4] Susilo, D, Saputro, F H and Wahono, A 2011 Analysis of infrastructure network voip availability at surakarta sahid university Jurnal Gaung Informatika 4(2) 25-34 https://journal.usahidsolo.ac.id/index.php/GI/article/view/148 
[5] Hidayat, R, Rushendra and Agustina, E 2017 Digital beamforming of smart antenna in millimeter wave communication IEEE 2017 International Conference on Broadband Communication, Wireless Sensors and Powering https://ieeexplore.ieee.org/abstract/document/8272564/  
[6] Goldman, J E, and Rawles, P T 2004 Applied Data Communications: A Business-Oriented Approach (US: Wiley & Sons, Fourth Edition)  
[7] Stiawan, D 2009 Fundamental Internetworking Development & Design Life Cycle http://unsri.ac.id/upload/arsip/network_development_cycles.pdf  
[8] Sugiyono 2013 Metode Penlitian Kuantitatif, Kualitatif dan R&D (Bandung: Alfabeta)  
[9] European Telecommunications Standards Institute (ETSI) 1999 Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); General aspects of Quality of Service. ETSI TR 101 329 V2.1.1