Mobile Service Development and Analysis Of Quality of Service In IP-PBX Based On SIP Protocol at PT. Melvar Lintasnusa

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ABSTRACT

PT. Melvar Lintasnusa is a company that has long been engaged in the Internet Service Provider. The company has many employees and some of them are often conducted field visits to the office / building / home / hotel in the city of Bandung. On that basis, media communication becomes the most important means to maintain communication between employees in the central office with employees on duty in the field.
Media communication used by PT. Melvar Lintasnusa is wired telephone. However, the application is still frequent disturbances. Therefore we need a media that is able to optimize the system of communication between employees at PT. Melvar Lintasnusa. One alternative way to optimize the means of communication by developing IP-PBX systems that exist.
Melsa IP-PBX using Asterisk as a proxy server based on sip protocol. In the implementation and testing, analysis of service QoS (Quality of Service) and MOS (Mean Opinion Score). Thus the implementation of priority queueue routing for voice packets for creating sounds good results and stable, compared to the default routing First IN First Out (FIFO) in accordance with the value of test measurements ITU-T standards

Keywords: computer networks, VoIP, IP-PBX, Asterisk, Quality of Service, Mean Opinion Score, mobile service

1. INTRODUCTION

PT. Melvar Lintasnusa (Melsa) is an authorized licensee of the government as the first internet service provider in Bandung. Since 1995, Melsa continue to develop a wide range of products with leading technology to meet the needs of the people of Bandung internet are higher. With a combination of wired infrastructure and wireless reliable, product development regularly, support Technical Support of experienced and service Helpdesk 24 hours a day, now Melsa has served more than 500,000 customers from various backgrounds such as government agencies, private companies, individuals and hotspot service free in public areas in the city. In addition to Internet access services, Melsa also provide internet services other derivatives, such as hosting, colocation, domain name, intranet data and CCTV, internet filter, the design and development of web applications, which can provide a complete solution to meet the needs of Internet users.

1.1 Background

Melsa is currently using server IP-PBX (Internet Protocol-Private Branch Exchange) to regulate voice communications, an incoming call-out through the PSTN (Public Switched Telephone Network) and among employees in the office. Runs on the Linux operating system with an asterisk and clients using Cisco IP Phone 7940 series.
Starting from the frequent communication using cellular phone with a field officer and frequent phone interruptions caused disruption PSTN in the office or cellular provider who used a field officer, hence the importance of alternative communication that is Voice over Internet Protocol (VoIP) server IP-PBX Melsa via smartphone. So that employees who served in the field such as, offices, homes, hotels or hospitals are not fixated only rely on cellular or PSTN provider, but can take advantage of the Internet for voice communication facilities.

1.1 Objectives

The purpose of this study was to design an alternative communication system among employees through IP-PBX server, among employees who are in the office and the field. For employees who are in the field, can use android smartphone with apps CsipSimple for media communication via IP-PBX. Last to analyze performance on a side in each of the networks that have been created, using a network analyzer tool.

1.3 Problem Formulation

Based on the background of the problems that have been described, it can be formulated into some of the following:

1. How to design a system that can meet the needs of communication with employees who are on the field?
2. How to run the communication between employees, using VoIP technology on the server IPPBX Melsa via smartphone?
3. How does the performance of VoIP services on the server IP-PBX Melsa using a network analyzer?

1.4 Stages of Research

Here is the stage and the explanation:

1. Data collection
   Data collection methods used in this study are as follows:
   a. Study of literature
      Data collection techniques by collecting and studying literature, journals, papers and readings that has nothing to do with the research.
   b. Interview
      In this stage of data collection is done by a question and answer directly wisely related in the study.
2. Analysis of Current System
   In this stage, the analysis of the system is running, because the topic of research is the optimization and analysis qualitiy of service analysis is carried out on systems running information to determine the current business process.
3. Architecture Network Topology
   In this stage, the design of a new system architecture that will be applied to VoIP IP-PBX server. The design includes business process systems that have been developed and implemented VoIP system overview.
4. Implementation and Testing
   At this stage the results of the design of the system will be applied to then testing whether it can meet the objectives of the study.

2. REVIEW OF LITERATURE

With the rapid development of science and technology today, especially information technology computer, then followed by an increase in data processing using a computer. Many companies or institutions and government agencies as well as private use of sophisticated computer information technology to make a significant improvement in running the activities of the company which have led to improvements to the presentation of information that companies need fast, accurate, and appropriate in order to obtain more results good. In this section will discuss the notion of network, VoIP (Voice over Internet Protocol), network protocols, Quality of Service, OSI models, IP-PBX, Asterisk, Wireshark, Cisco IP-Phone, RouterOS, CsipSimple, and VQ Manager.

2.1 Definition of Computer Networks

Some experts have conducted a series of research specifically related computer networks, as well as making a conclusion that we can use to understand the definition of a computer network. These include the following:

a. Noor Jafar Yudianto (2007)
   The computer network is a system that consists of computers that are designed to be able to share resources (printer, CPU), communication (e-mail, instant messaging), and can access the information (web browser).
b. Kris (2003)
   Computer network is a group of autonomous computers interconnected with each other, using a protocol kompunikasi so selruh computers connected together they can share information, programs, resources and also are able to use other hardware devices simultaneously, such as printers, hard drives and other etc.
c. Abdul Kadil (2003)
   The computer network is the relationship of two vertices (usually a computer) or more whose main purpose is to exchange data.

Using the basics of theories have been put forward by experts of the above, I conclude that the computer network is more than a collection of computers that communicate with each other using the same network protocols, so that they can share information, sharing of data, shared resource (such as a printer or media storage).

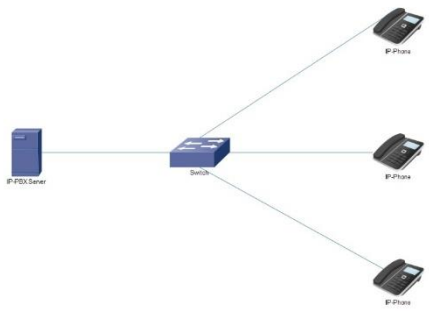
A computer network is built with the aim of bringing information secoara timely manner and without the fault of the sender (transmitter) to the receiving side (receiver) through the medium of communication. There will be many obstacles in the process of sending information, such as the use of communication facilities is not maximized, jaur transmission used is not completely free from the problem of interference (noise) [7].

2.2 IP-PBX

IP-PBX or Internet Protocol Private Branch Exchange is a technology that uses IP PABX [2]. IP-PBX is a telephone switching device and data communication technology-based Internet Protocol (IP) which controls extension analog telephone (TDM) and IP Phone extensions.

The functions that can be done such as grafting, control and termination of telephone; translational communication protocols; translation or transcoding media communication; and control devices such as

VoIP IP Telephony Gateway, Access Gateway and Trunk Gateway.
Solution-based IP PBX is the concept of communications network future generations or known as NGN (Next Generation Network) that can integrate the conventional telephone network (PSTN / POTS), a network of mobile telephony (GSM / CDMA), a network of satellite phones, network Cordless (DECT), and packet-based networks (IP / ATM).
IP-PBX brings multiservices capabilities in the world of communications networks to IP telephony, thus enabling more and more communication services that can run on top of IP networks.



Gambar 2.1. Topologi IP-PBX

2.3 Asterisk

Asterisk is an open source framework for building communication applications [4]. Asterisk has the ability to transform ordinary computer into a communications server. Strength Asterisk IP PBX systems, VoIP gateways, conference servers and more. This application is widely used by small companies, up to large enterprises, call centers, carriers and governments worldwide. Asterisk sponsored by Digium, Asterisk is able to interact with many traditional telcom protocol, VoIP protocols and codecs.
Asterisk provides a list of surprising capabilities and features include:a.

a. IVR (Interactive Voice Response)
b. ACD (Automatic Call Distributor)
c. Audio dan Video Conferencing
d. Voicemail
e. Call Recording
f. Fax Termination
g. CDR (Call Detail Record)

2.4 VoIP Voice over Internet Protocol (VoIP)

Voice over Internet Protocol (VoIP) also known as IP Telephony is defined as a system that utilizes Internet media to transmit voice data packets from one place to another using the IP protocol. This technology is capable of passing voice traffic shaped melaluiu IP network packets. IP network itself is a data communications network based packet-switch.

2.4.1 VoIP packet format

VoIP consists of the header and payload (load). The header consists of an IP header, Real-time Transport Protocol (RTP), User Datagram Protocol (UDP) header, and link header.
IP header has a role to store routing information for sending the packets to the destination. In each of the included IP header type of service or the Type of Service that allow certain packets such as voice packets that non real-time. UDP headers have certain features that do not guarantee the package will achieve the goals that are suitable for UDP real-time voice applications that are sensitive to delay and latency.
RTP Header is header is useful for framing and real-time data segmentation. A case of UDP, RTP also did not support the reliability of the package to sampau to the destination. RTP uses the control protocol called Real-time Transport Control Protocol (RTCP) that controls QoS and synchronization of different media streams. For massive link header, depending on the media used.

2.5 Session Initiation Protocol (SIP)

Session Initiation Protocol (SIP) is a protocol which is located at the application layer that defines the beginning of the process, change and termination of a multimedia communication session [1]. Session Initiation Protocol or SIP is an IETF proposed standard in 1999, for voice or multimedia services via the Internet network SIP [RFC 2543]. Henning Schulzrinne creating this standard, SIP is an application layer protocol that is used for the management arrangement calls and termination of calls. SIP is used in conjunction with other IETF protocols such as, SAP, SDP, MGCP (Megaco) to provide VoIP services more widely.
SIP has a similar architecture with HTTP (client-server protocol), consisting of a request sent from the user SIP to SIP server. Server processes the incoming request and provide a response to the client. Demand request, along with other response components make a SIP communications.
SIP architectural components:

1. User Agent
   SIP User Agent is the terminal end of the act on the will of the wearer, parts namely:
   a. User Agent Client (UAC): This section includes the user (client) that is used to initiate a request from the SIP server to UAS.
   b. User Agent Server (UAS): function to hear and respond to the SIP request.
2. SIP Server
   Architectural SIP describes the types of servers on the network to help service and setting SIP calls.

a. Registration Server: The server receives a request from the SIP user and updating the user's location with this server.
b. Proxy Server: server function to receive SIP request and then pass on to the right server according to the information about the user who invoked.
c. Redirect Server: Serves to determine the destination server after receiving reuest SIP client, and then returns the address of the destination server to the client's next then forwards the request to the destination server.

2.6 QoS dan MOS

Quality of Service (QoS) can be regarded as a term used to define the characteristics of a service (service) network in order to know how good the quality of these services [1]. Quality os Service (QoS) is a terminology that is useful to define the characteristics of a service (service) network in order to know how good the quality of the service. The parameters used are bandtwidth QoS, throughput, delay, jitter and packet loss.
Standard value by ITU-T, as follows:

Tabel 2.1. Tabel QoS 1

Delay (ms)	Kualitas
0 – 150	Baik
150 – 400	Cukup, Masih Dapat terima
> 400	Buruk

Tabel 2.2. Tabel QoS 2

Packet Loss (%)	Kualitas
0 – 0,5	Baik
0,5 – 1,5	Dapat terima
> 1,5	Tidak dapat Buruk

Tabel 2.3. Tabel QoS 3

Jitter (ms)	Kualitas
0 – 20	Baik
20 – 50	Dapat terima
> 50	Tidak dapat Buruk

Tabel 2.3. Tabel MOS

Nilai MOS	Kualitas Percakapan
5	Sangat Baik
4	Baik
3	Cukup
2	Kurang Baik
1	Buruk

Considering everyone has different standards, to achieve effective penialaian then in the current study will use three categories of MOS (Mean Opinion Score). The third category refers to the ITU standard, which is based on the quality of sound in the hearing, the effort needed someone to hear and be heard intensistas volume, I will present details in the table below [1].

Tabel 2.4. Tabel MOS 1

Nilai MOS	Kualitas Percakapan
5	Sangat Jelas, Tanpa Noise
4	Jelas, Sedikit Noise
3	Cukup Jelas, Banyak Noise
2	Kurang Jelas, Sulit dimengerti
1	Tidak Jelas, Tidak dimengerti

Tabel 2.5. Tabel MOS 2

Nilai MOS	Kualitas Percakapan
5	Nyaman, Santai
4	Perlu sedikit usaha untuk konsentrasi
3	Perlu cukup usaha untuk konsentrasi
2	Perlu banyak usaha untuk berkonsentrasi
1	Tidak dimengerti dengan seluruh usaha

Tabel 2.6. Tabel MOS 3

Nilai MOS	Kualitas Percakapan
5	Sangat besar dari yang diharapkan
4	Lebih besar dari yang diharapkan
3	Sesuai dengan yang diharapkan
2	Lebih kecil dari yang diharapkan
1	Sangat kecil dari yang diharapkan

3. ANALISYS AND DESIGN

3.1. Configuration Server IP-PBX

Server ip-pbx Melsa using the FreeBSD operating system. The use of FreeBSD here specifically to run as an asterisk or ip-pbx server. In accordance with the data analysis of IP-PBX, IP-PBX server has only one NIC while so that the server can be accessed through the Internet must use public IP address. A solution that will be used is to configure an alias, so that it does not require additional cost to add NIC. IP aliasing is to associate more than one IP address to the network interface. With this, one node on the network may have some connection to the network, each serving a different purpose. In short, with IP aliasing we can add two even more IP addresses in the first NIC.

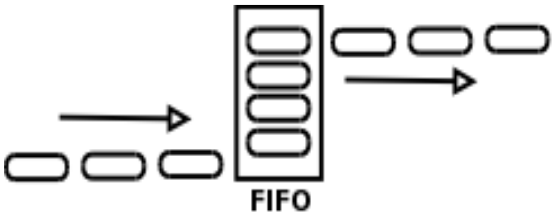
3.2. Configuration CSipSimple

Make sure CsipSimple applications already installed on the smartphone device that will connect to the IP-PBX system Melsa. Add a new account on the application by selecting the wizard "basic". Input parameters of the data requested by the application,

such as data that is already configured on the server IP-PBX. If successful, the words "display name" on the new account in setting will be green with the "registered".

3.3. Methods Differentiated Service

At the present time delivery of packets in a network is usually done best effort, in which the package will be delivered in FIFO (First In First Out) packets first to the node will be forwarded to the next node in a sequence based on arrival, so the packages will scramble the existing bandwidth.



Gambar 3.1 Antrian FIFO

This research will be using priority queing contained in RouterOS, where traffic will be packets that have a higher priority will be allocated bandwidth needed for the set on the router.

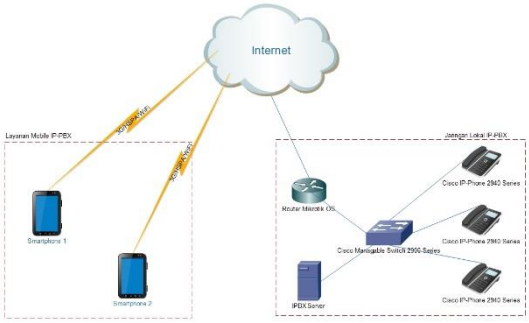


Gambar 3.2 Prioritas Queue

VoIP uses RTP protocol will get priority to pass on the transmission medium used. This research will apply the policy on proxy router should be set as a priority order for the RTP port 8000-20000 will be given priority with badwith allocation for each of the communication channels of 40 Kbps.
In this study, RTP packets will be set on port 8000 to 20000 are used for data transmission media. How sorting classification of traffic that is processed by data packet filter. The filter will sort out packets based on the protocol. Packets get first priority will be processed first. Once sorted aggregated packet delivery will be sorted in order of priority predetermined, henceforth delivered to the destination node. For the study related to noise, the bandwidth used is not so large that the current implementation of the future, the possibility is not too significant sightings over the user experience for prioritizing voice packets in the traffic condition is solid then the bandwidth allocation awake and delay too low, resulting in voice quality the good one.

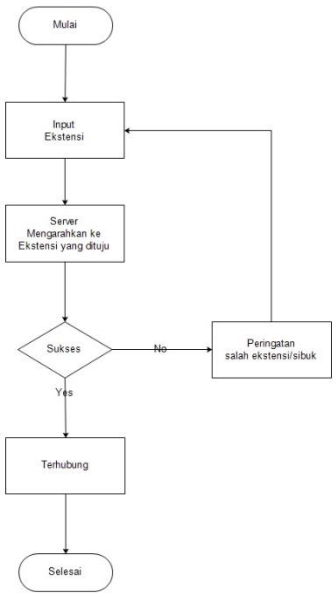
3.4. Design of The p\Proposed System

Here is a picture design proposal:



Gambar 3.3 Desain Optimasi Sistem IP-PBX Melsa

Explanation of the picture above, as follows:
Red box symbolizes a different location, the right box office LAN describe the condition, then the red box to the left is the location of the smartphone area outside the office LAN. On the LAN side IP-PBX server Office needs to be connected to the gateway Melsa (RouterOS), so that after a given server from the public network can be accessed via the Internet. For configuration on the ip-phone no IP configuration changes, because the server using IP Alias current Public IP mounting configurations. To the outside of the office LAN, a smartphone that is used is to configure the application CsipSimple as described earlier, and then directed to the Public IP server. Connectivity or media intermediary can use the HSDPA network, 4G LTE or WiFi as long as the smartphone can be connected to the Internet. Process flow between the extension as in figure 3.4.



Gambar 3.4 Proses telepon antar ekstensi

As in Figure 3.3 and 3.4, the phone is basically governed by the IP-PBX server Melsa. Router function to introduce the Public IP server to the Internet and set the packet priority. To process an extension phone has no limitation of the device used, IP-Phone or Smartphone. Flowchart of the above processes can be implemented by, among IP-Phone, inter Smartphone, or crossing IP-Phone to a Smartphone or vice versa.

4. IMPLEMENTATION AND TESTING

Based on the results of tests performed using multiple testing scenarios. We can analyze the data obtained from the results of measurements and observations.

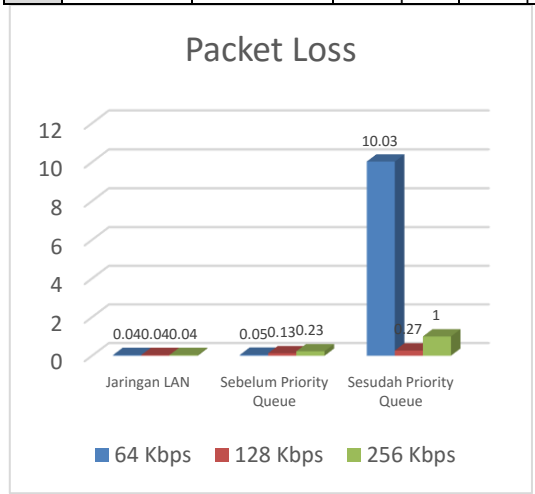
4.1 Grafik QoS

The following table charts the sum of the average yield testing, packet loss, jitter, and delay:

a. Paket Loss

Tabel 4.1 Perbandingan Packet Loss

No	Bandwidth	Jaringan LAN (tanpa setting bandwidth)	Sebelum priority queing	Sesudah priority queing
1	64 Kbps	15.55 Kbps	19.90 Kbps	20.15 Kbps
2	128 Kbps		20.84 Kbps	20.56 Kbps
3	256 Kbps		20.98 Kbps	20.27 Kbps



Gambar 4.1 Grafik Packet Loss

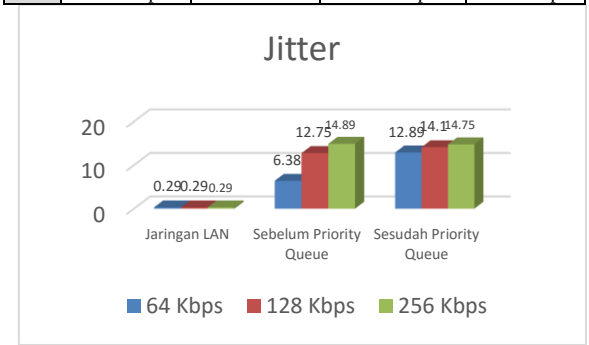
From the graph of Figure 4.1 can be seen, the number of packets lost after using a priority queue increased slightly, except at 64 Kbps bandwidth testing after using queueu priority, there is an increase in any one test among 10 stages that affect the value of the average. If the gateway router traffic load is full, theoretically the default routing networks using different will increase with the use of a priority queue will remain stable measurement and user experience of the sound would have been better.

b. Jitter

No	Bandwidth	Jaringan LAN (tanpa setting bandwidth)	Sebelum priority queing	Sesudah priority queing
1	64 Kbps	0.29 ms	6.38 ms	12.89 ms
2	128 Kbps		12.75 ms	14.10 ms
3	256 Kbps		14.89 ms	14.75 ms

Tabel 4.2 Perbandingan Jitter

No	Bandwidth	Jaringan LAN (tanpa setting bandwidth)	Sebelum priority queing	Sesudah priority queing
1	64 Kbps	0.04 Kbps	0.05 Kbps	10.03 Kbps
2	128 Kbps		0.13 Kbps	0.27 Kbps
3	256 Kbps		0.23 Kbps	1.00 Kbps

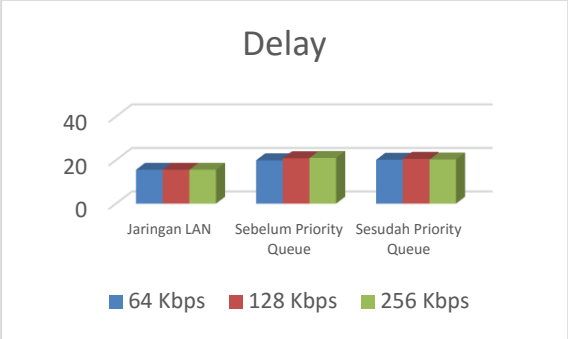


Gambar 4.2 Grafik Jitter

From the graph it can be seen figure 4.2, jitter parameter after using a priority queue to increase slightly. If the gateway router traffic load is full, theoretically the default routing networks using different will increase with the use of a priority queue will remain stable measurement and user experience of the sound would have been better. As well as on comparative measurements using a priority queue after the 256 Kbps bandwidth decreased slightly.

c. Delay

Tabel 4.3 Perbandingan Delay



Gambar 4.3 Grafik Delay

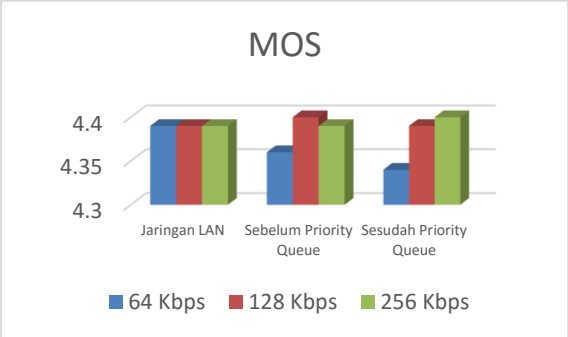
From the graph of Figure 4.3 can be seen, after a delay parameter using the priority queue to decline slightly. If the gateway router traffic load is full, theoretically the default routing networks using different will increase with the use of a priority queue will remain stable measurement and user experience of the sound would have been better.

d. Graph MOS

Here is a summary chart of the average yield testing MOS:

Tabel 4.4 Perbandingan MOS

No	Bandwidth	Jaringan LAN (tanpa setting bandwidth)	Sebelum priority queing	Sesudah priority queing
1	64 Kbps	4.39	4.36	4.34
2	128 Kbps		4.4	4.39
3	256 Kbps		4.39	4.4



Gambar 4.4 Grafik MOS

4.4 From the graph image can be seen, the result of the acquisition of MOS trials using a bandwidth of 256 Kbps after using a priority queue to increase slightly. Which states that good voice quality has a value closer to the perfect number, according to the standard ITU-T.i mendekati angka sempurna, sesuai standar ITU-T.

5. KESIMPULAN

From the results of the implementation and testing has been done, it can be concluded that:

1. System Melsa existing IP-PBX can be developed, so it can be accessed anywhere via the Internet with the help of Smartphone devices.
2. CSipSimple application is installed on your Android Smartphone to function as UAC (User Agent Client), so it can run VoIP with existing systems.
3. Obtaining the average results of the analysis show that QoS optimization results or VoIP that uses a priority queue to have VoIP without delay under the priority queue. Average MOS produced VoIP uses a priority queue is 4 or better according to ITU-T standards.
4. So the performance of IP-PBX to mobile services Melsa good after using a priority queue, because it can maintain stability and sound quality, especially if traffic bandwidth on the LAN gateway router being solid.

5. CONCLUSION

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