Implementasi *Ethernet Over IP* (EoIP) *Tunnel* Mikrotik RouterOS Pada Layanan *Voice Over IP* (VoIP) Di PT Akurasi Kuatmega

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ABSTRACT

PT. Akurasi Kuatmega is a company that has long been engaged in telecommunications and is now venturing into the world of tourism as an example of attractions Maribaya in the city of Bandung. The company has many employees and has several branch offices. On that basis, media communication becomes the most important means to maintain communication between employees at headquarters with branch office employees Maribaya. Therefore we need a media that is able to optimize the system of communication between employees at PT. Akurasi Kuatmega. One way to optimize the communication system is by implementing Voice Over Internet Protocol (VoIP) and Internet Protocol Over Ethernet as a liaison between the office network. Voice Over Internet Protocol (VoIP) is a technology that is capable of passing "voice call", video and data over IP networks by using Asterisk as a server and softphone x-lite as a client. By using a system of communication through the Internet has the advantage in terms of maintenance costs are much cheaper than the cost of the required regular telephone network.
Keywords : *VoIP, Ethernet over IP, X-Lite Softphone, Asterisk.*

1. FOREWORD

PT . AKURASI KUATMEGA INDONESIA is a company that has long specialized in telecommunications and is now venturing into the world of tourism as an example of attractions Maribaya [1] . The development of technology and telecommunications , which grew rapidly has created a world of information . This further triggers the need for ease in interacting with fellow employees , therefore, required assistive technologies that can support employee performance that is Voice over Internet Protocol (VoIP) to support communication with the form of voice and Ethernet over Internet Protocol (EoIP) to connect the two network so as - if it is on one.

1.1 Background

The network system at PT. Akurasi Kuatmega Indonesia currently has two servers that only controls the data that is disekitaran office without their relationships with other branches that have not been able to control for all branch offices . Starting from the frequent communication using the phone using either a wired telephone or a cell phone between the manager and the employee or employees and employees who are in the central office to the branch office Maribaya or vice versa and frequent phone interruptions caused interruption of telephone cables in the office network or from the network side telecoms , therefore, needed a voice communication over IP (VoIP) and connected by tunnel between mikrotik contained in each branch office so as to provide convenience in terms of data communication and control of the branch office .

1.2 Purpose

The purpose of this study was to merangcang communication system that can cover the communication needs of companies based voice over IP (VoIP) , in order to provide communication services between different client and run EoIP tunnel network between headquarters and branch offices . Being able to analyze the performance on the one hand on each network that was created when running a voice over IP (VoIP) before and after use EoIP tunnel with parameters that delay , jitter , and packet loss using a network analyzer is Wireshark that can measure the bandwidth required in VoIP technology .

1.3 Formulation of the problem

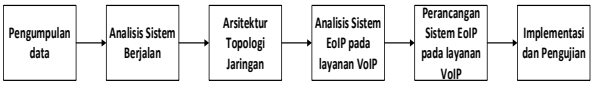
Based on the background described above , then that becomes a problem in this research are :

1. How to design a system that can cover the needs of good communication between headquarters and branch offices?
2. How to run the communication between headquarters and branch offices using VoIP technology and EoIP?

3. How is the performance and quality of this protocol on both sides of the network in the VoIP service before and after the use of EoIP tunneling using a network analyzer (wireshark)?
4. How to measure the bandwidth requirements which are well used to pass through VoIP technology?

1.4 Stages Research

The stages of the research carried out can be seen in Figure 1.1.



Gambar 1.1 Tahapan Penelitian

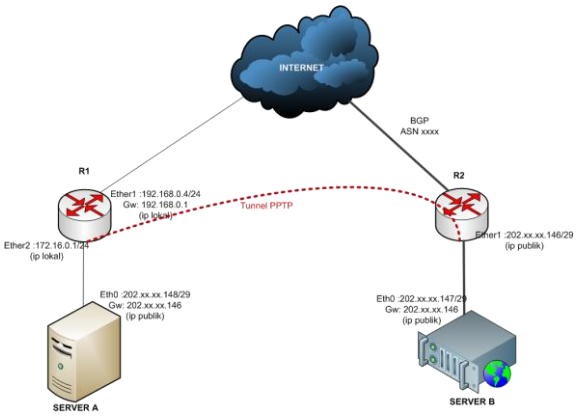
The following explanation of each of the stages :

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 to the relevant parties 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 implementation of ethernet over internet protocol (EoIP) tunnel proxy router on the voice over internet protocol (VoIP) analysis is carried out on the system information goes to see the different business processes running system and business processes must change to take effect tunneling proxy.
3. Architecture Network Topology
   In this stage, the design of a new system architecture that will be applied to the proxy tunneling EoIP on VoIP services . The design includes business process systems that have been updated as well as an overview of how VoIP system implemented.
4. Analysis System EoIP tunnel proxy on a VoIP service
   At this stage, an analysis of how EoIP tunnel proxy can be used for VoIP usage and the use wireshark tools used for the analysis of quality of service (QOS) on a service (service) network in order to know how good the quality of these services.
5. System Design EoIP tunnel proxy on a VoIP service
   At this stage, system design EoIP tunnel proxy on VoIP services and the design of the scheme will be tested.
6. Implementation and Testing
   At this stage the results of the draft proxy tunnel EoIP Systems VoIP services will be applied to then testing whether it can meet the objectives of the study.

2. LITERATURE REVIEW

2.1 Ethernet Over Internet Protocol

Ethernet over IP (EoIP) Tunneling MikroTik is a protocol that makes an Ethernet tunnel between two routers over an IP connection . EoIP interface appears as an Ethernet interface . When the bridging function of the router is enabled , all Ethernet traffic (all Ethernet protocols) will be bridged just as if there where a physical Ethernet interfaces and cables between two routers (with bridging enabled) . Understanding IPIP tunnel is a simple protocol that encapsulates IP packets in the IP to create a tunnel between two routers . IPIP tunnel interface appears as an interface in the interface list . Many routers , including Cisco and Linux based , supports this protocol . The maximum amount that can be made EoIP tunnel tunnel is 65535 .

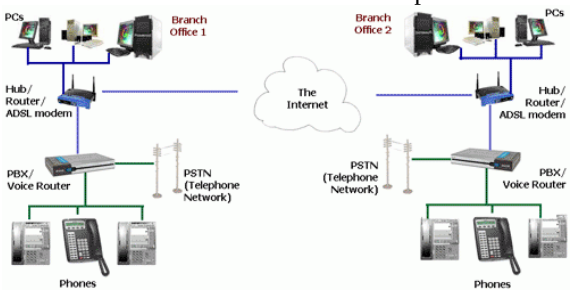


Gambar 2.1 Topologi Penggunaan EoIP
Sumber : mikrotik.com

2.2 Voice Over Internet Protocol

Voice over Internet Protocol (VoIP) also known as IP Telephony is defined as a system that uses the internet to transmit voice data packets from one place to another using an intermediary IP protocol [8] . In other words, the technology is capable of passing voice traffic in the form of packets over an IP network. IP network itself is a data

communicationsnetwork based packet-switch.



Gambar 2.2 Diagram VoIP
Sumber : cornerstonebs.co.uk

2.3 Protokol Session Initiation Protokol (SIP)

Session Initiation Protocol or SIP is an IETF standard for voice or multimedia services through the Internet . SIP [RFC 2543] filed in 1999. The creator of this standard is Henning Schulzrinne . SIP is an application layer protocol that is used for the management of the call setup 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 architecture similar to the architecture of HTTP (client- server protocol) . The architecture consists of a request sent from the user SIP to SIP server . The server processes the incoming request and provide a response to the client . Demand request , along with the other components of the response message create a SIP communication.

SIP architecture consists of two components as shown below:

1. *User Agent*

SIP User Agent is the end of the system (terminal end) which act on the will of the user. Consists of two parts:

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*) : This section serves to hear and respond to the SIP request

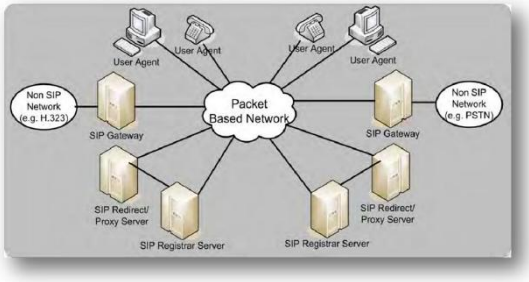
2. *SIP Server*

SIP architecture itself explain the types of servers on the network to help service and SIP call control.

a. *Registration Server* : This server receives a request from the SIP user and updating the user's location with this server.

b. *Proxy Server* : The server receives the request and forwards the SIP server to the destination that has information about the user who invoked.

c. *Redirect Server* : This server after receiving the SIP request , specify the next destination server and returns the address of the destination server to the client 's next then forwards the request to the server on the go .



Gambar **Error! No text of specified style in document.**.3 Operasi SIP
Sumber : Voice Over IP Fundamentals (2nd Edition).

2.4 Quality of Service (QoS)

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 the service. In this study the QoS parameters to be analyzed are delay , jitter and packet loss . Delay is a time delay in processing the data , which for the delay is said to be good quality if time tundanya only about 0-150 ms [4] . Some delay that can interfere with the sound quality in the design of VoIP can be grouped into :

1. *Propagation delay* (caused by transmission through the distance between sender and receiver).
2. *Serialization delay* (occurred during the process of peletan bit into the circuit).
3. *Processiong delay* (occurred during the process of coding , compression , decompression and decoding).
4. *Packetization delay* (occurred during the packetization process digital voice sample).
5. *Queuing delay* (caused by the waiting time until the packages are served).
6. *Jitter Buffer* (delay due to the jitter buffer to cope)

Jitter is the difference in arrival time interval between packets at the destination terminal , or in other words jitter is the variation of the delay . The amount of jitter resulting in the destruction of the data received , whether it's an admission that intermittent or loss of data due to overlap with other data packets . Many things can cause jitter , such as increased traffic suddenly , causing narrowing of the bandwidth and cause queues . Jitter quality is quite good for when the time is only around 0-20 ms [4] . Packet loss is the number of packets lost in a delivery of data packets on a network . Some causes of packet loss is the presence of noise , collisions and congestion caused by the occurrence of excessive queues in the network . Packet Loss in VoIP is said to be good if the number of lost packets levels range between 0 s / d 0.5 % of the data transmission [4].

Table **Error! No text of specified style in document.**.1 Parameter *Delay* berdasarkan ITU-T G.114

Nilai Delay	Kualitas
< 150 ms	Baik
150 – 400 ms	Cukup, masih dapat diterima
> 400 ms	Buruk, tidak dapat diterima

Table **Error! No text of specified style in document.**.2 Parameter *Jitter*

Nilai Jitter	Kualitas
0 – 20 ms	Baik
20 - 50 ms	Cukup
> 50 ms	Buruk

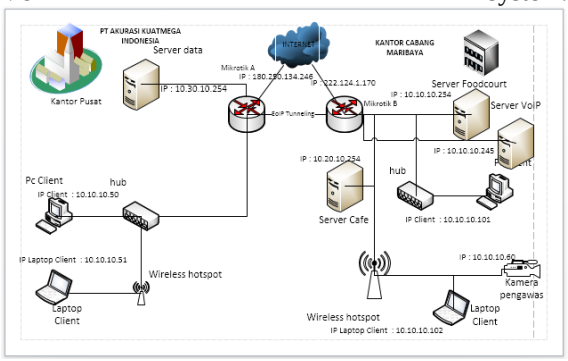
Table **Error! No text of specified style in document.**.3 Standar *Packet Loss*

Packet Loss	Kualitas
0 – 0.5 %	Baik
0.5 – 1.5 %	Cukup
> 1.5 %	Buruk

3. ANALYSIS AND DESIGN

3.1 Network Topology Design System

VoIP system that will be implemented will be applied to any type of network Wide Area Network (WAN) and EoIP are using two routers . Topology used in the WAN is a point- to-point , ie between the router A to router B for connections between them using encapsulation technology point- to-point protocol (PPP) . And for the router's LAN B topology used is a star topology (star) . Star topology has the key attributes of the concentrator , which is a hub , switch, or router. And in this study is used as a concentrator router that serves to connect multiple clients from a VoIP system.



Gambar 3.1 Topologi Logik Sistem VoIP

WAN connection is created between router 1 and router 2 via the Internet network connectivity in which the router 1 has a Public IP network with a subnet mask 180 250 134 246 255 255 255 248 belonging to the class B. The router 2 has a Public IP 222.124.16.170 with the subnet mask 255.255.255.248 . For LAN router 1 and router 2 has one client , which is classified into class C IP is 192.168.1.0 with a subnet mask of 255.255.255.0 . Client for PCs and laptops in mikrotik A get the IP address of 10.10.10.50 - 10.10.10.250 , PCs and laptops Client in mikrotik B get the IP address of 10.10.10.101 - 10.10.10.250.

3.2 Design Configuration Ethernet over IP

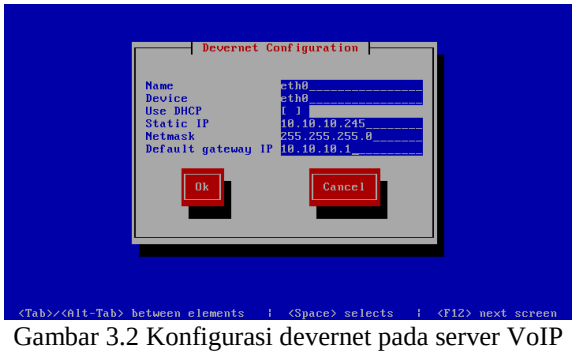
Design EoIP to perform in both RouterOS configuration , the time setting MikroTik router side of the central office, remote parameter contents with Public IP address that routers in branch offices Maribaya . Do the same when setting the router side Maribaya Branch Office , can be analogous to an exchange of information from the public network . Later on Tunnel parameter ID , make sure it has the same value between Tunnel router ID Headquarters with branch office router . The configuration of the interface EoIP on each - each proxy is as follows:

Table 3.1 Konfigurasi EoIP Pada Mikrotik

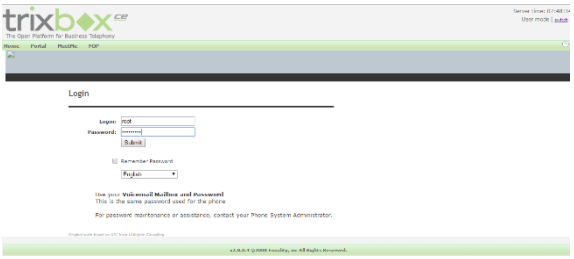
Mikrotik Kantor Pusat	Mikrotik Kantor Cabang
Name =eoip-tunnel-kantorpusat Type = EoIP Tunnel MTU = 1500 MAC Address = 02:52:5D:03:1E:FE Remote Address = 222.124.16.170 Tunnel ID = 10	Name=eoip-tunnel-kantorcabang Type = EoIP Tunnel MTU = 1500 MAC Address = 02:52:5D:03:1E:FE Remote Address = 180.250.134.246 Tunnel ID = 10

3.3 Design Configuration Server Voice over IP

Server VoIP used in the design is a Trixbox CE , Trixbox After installation is complete , the next step is to configure the VoIP gateway is to address how to configure networks that are at the VoIP server . Configuring Networks includes input IP address , IP gateway , DNS server , and Subnet Max . In designing our VoIP server provides an IP address 10.10.10.245 , Max 255.255.255.0 subnet and gateway IP 10.10.10.1 so that later can be connected to devices RouterOS .



Gambar 3.2 Konfigurasi devernet pada server VoIP



Gambar 3.3 Tampilan Webconfig Trixbox

The next stage after we got into the Trixbox server admin menu in which there are information systems that are running , for the configuration of VoIP use at no stage - the stage is important to do them PBX extension or setting to include the client address and use generic SIP device . As for the configuration of SIP extension of each - each client is as follows :

Table 3.2 konfigurasi SIP extension

Client 1	Client 2
type = friend context = from-internal host = dynamic secret = 111 port = 5060 dial = SIP/111	type = friend context = from-internal host = dynamic secret = 112 port = 5060 dial = SIP/112

Client 3	Client 4
type = friend context = from-internal host = dynamic secret = 211 port = 5060 dial = SIP/211	type = friend context = from-internal host = dynamic secret = 212 port = 5060 dial = SIP/212

3.4 Design Configuration In Softphone

The softphone used in the design is the X - lite which is an operating system and software that can be used as a VoIP client . Before the configuration of the softphone to first install the program on the device to be tested is the client pc and mobile phones.

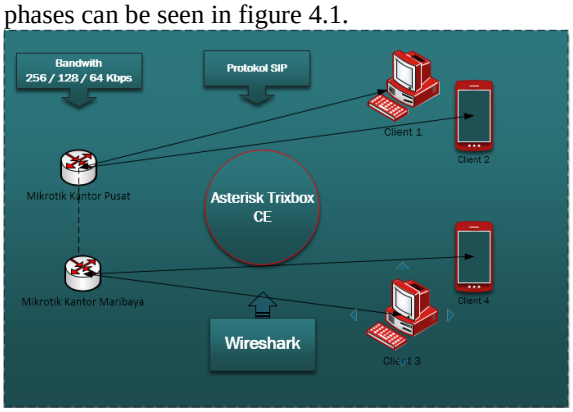
Table 3.1 Konfigurasi pada Softphone

X-lite PC Client		X-lite Handphone	
Account Name	= 111	Account Name	= 112
User ID	= 111	Username	= 112
Domain	= 10.10.10.245	password	= ***
Display Name	= 111	Caller ID	= 112
Authorization name	= 111	Host	= 10.10.10.245

4. Implementation and Analysis of Results of Testing

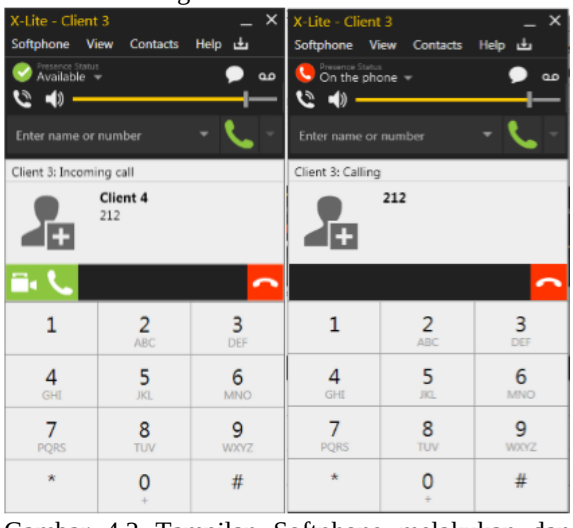
4.1 Impelemntasi Research

Implementation and testing is done in the office of PT. Accuracy Kuatmega Indonesia and branch offices Maribaya. The scenario will be that the PC client and a Mobile client is already configured SIP extension by the administrator on the server and configured VoIP bandwidth used on each -tiap testing. From such scenario would be analyzing the aspects of the performance and the impact of VoIP when using EoIP. To help the analysis of performance, then use one of the sniffing software that is Wireshark, software that can be implemented on the client computer will capture all packets transmitted to the VoIP data (VoIP data is transmitted can be recorded and playback). In addition, from the data recorded using this software can analyze the performance of VoIP by counting the delay, jitter, and packet loss. The data will be analyzed is data with RTP packets. An overview of the implementation and testing phases can be seen in figure 4.1.



Gambar 4.1 Tampilan Impelmentasi dan Pengujian

As for the view of the softphone X - lite during VoIP calls are as in Figure 4.2.



Gambar 4.2 Tampilan Softphone melakukan dan menerima panggilan

4.2 Analysis of Testing Results

4.2.1 Testing EoIP Mikrotik

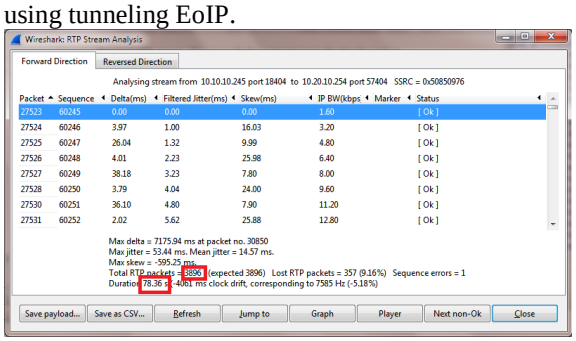
EoIP testing of the system will be performed on both mikrotik . From the side of the proxy at the head office , will be seen whether the proxy can ping the VoIP server located at the branch office mikrotik Maribaya vice versa of proxy branch Maribaya ping proxy client that is on at headquarters . Indicators testing to be performed on the system EoIP are as follows :

Table 4.1 Pengujian EoIP pada sisi mikrotik

No.	Nama Pengujian	Indikator Pengujian	Status Pengujian
1.	EoIP pada mikrotik sudah terkonfigurasi dengan benar.	Akan muncul indikator R pada menu interface pada mikrotik	mikrotik di kantor pusat bisa melakukan ping ke server VoIP di kantor cabang maribaya
2.	Komputer client sudah terhubung dengan mikrotik	komputer client bisa melakukan ping ke mikrotik	komputer client di kantor pusat bisa melakukan ping ke komputer client kantor cabang maribaya

4.2.2 Testing and Analysis of Delay

Delay is the time required by the package from the sender to get to the receiving terminal . Delay is an important parameter for determining the quality of a VoIP network . By the standards of the ITU - T G.104 for VoIP quality is good, the delay should be < 150 ms in order to avoid overlap in the communication . The following image delay measurements that have been captured by wireshark on VoIP communication using tunneling EoIP.



Gambar 4.3 Hasil Pengukuran Delay

In the picture above his 78.36 second total delay , and total packets received is 3896 , so if calculated according to the formula obtained :

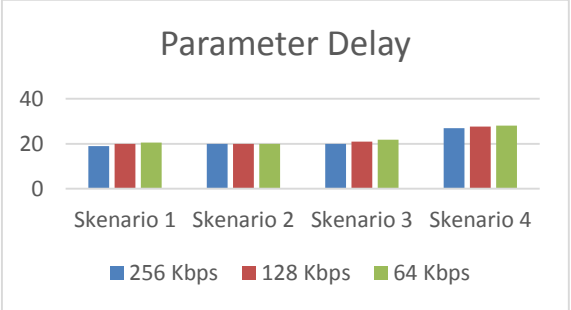
The formula for calculating the delay [20] :

$$Delay = \frac{Total\ Delay}{Total\ Paket\ yang\ diterima}$$

$$Delay = \frac{78{,}36}{3896}$$

$$= 0{,}02011s$$

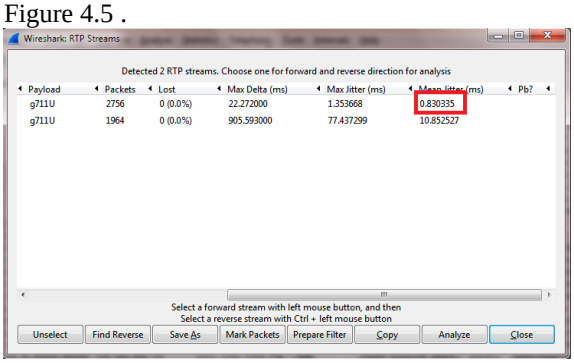
$$= 20\ ms$$



Gambar 4.4 Perbandingan Nilai *Delay*

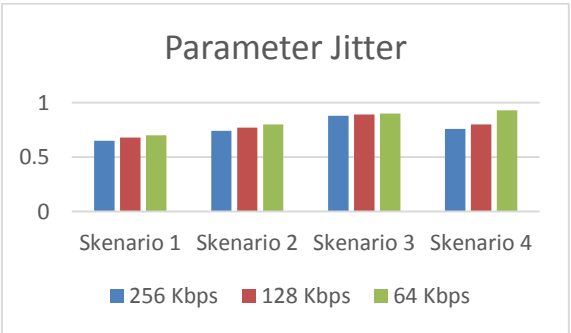
4.2.3 Testing and Analysis of Jitter

Jitter is a variation of delay caused by the long queue in a packet reassemble processing - the data packets at the end of the delivery as a result of previous failures . The larger the jitter value will result in the value of QoS getting down . To get a good network QoS value , the value of jitter must be kept to a minimum. VoIP packets passing captured and analyzed . The results of the measurements shown in Figure 4.5 .



Gambar 4.5 Hasil Pengukuran Jitter

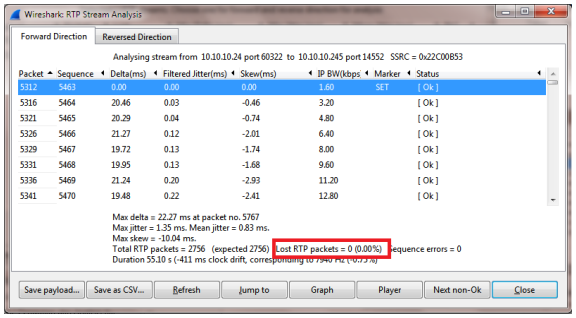
The results of the jitter measurement is according to testing done by beberpa skneario is as follows:



Gambar 4.6 Perbandingan Nilai Jitter

4.2.4 Testing and Analysis of Packet Loss

Packet Loss is defined as the number of lost packets from origin to destination. Although based on the experience that packet loss is more than 1 % can affect the sound quality . To view the capture results can be seen in Figure 4.7 wireshark capture and packet loss measurement results on VoIP communications can be seen in Figure 4.8.



Gambar 4.7 Hasil Pengukuran Packet Loss

Rumus untuk menghitung packet loss [20] :

$$\frac{Jumlah\ Paket\ yang\ dikirim - jumlah\ Paket\ yang\ diterima}{Jumlah\ Paket\ yang\ dikirim} x100\%$$

Paket Yang diterima = Paket yang dikirim – Paket Yang Hilang
= 2756 – 0
= 2756

$$Packet\ Loss = \frac{2756 - 2756}{2756} x100\%$$
$$= 0.00\ \%$$

5. CONCLUSION

From the analysis of Quality of Service (QoS) that have been carried out on the implementation of Ethernet over IP (EoIP) tunnel RouterOS on Voice over IP (VoIP) using SIP protocols this , it can be concluded that :

1. The system is able to cover the communication needs of enterprise VoIP services based on communication between different client networks.
2. The system can already run EoIP tunneling connecting the two mikrotik between headquarters and branch offices.
3. Performance VoIP with EoIP less well due to have a value of delay, jitter and packet loss and the highest sound quality is not good.
4. Bandwidth used to use a dedicated network so as to intranet to the other branches much more stable for a minimum of 1 MB of bandwidth required for VoIP services and use of the bandwidth 256 Kbps due to produce good quality QoS.
5. The role Trixbox VoIP server handles SIP calls from all clients who registered into the Trixbox server.
6. Broadly speaking Trixbox operating system can already be used to handle IP telephony network, but when it is done by using the wiretapping program Able Cain and wireshark proven voice communications can be recorded for RTP packets captured wireshark.
7. Use EoIP greatly affect the quality of the QoS obtained, the result showed that the performance quality VoIP with tunneling EoIP not as good as the quality of VoIP performance without EoIP of the result is that the use of VoIP without EoIP tunneling has a value of 20 ms delay, jitter value of 0742 ms and the value of packet lost a 0.01%, while the use of VoIP over EoIP tunneling has a value of 26.9 ms delay, jitter value of 0.75 ms and packet lost value of a 0.01%.

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