Implementation of the SIP Express Router with Mediaproxy Method on VoIP

Heru Nurwarsito, R. Arief Setyawan, Rakhmadhany Primananda

Abstract—Voice Over IP (VoIP) is a technology that could pass the voice traffic and data packet form over an IP network. Network can be used for intranet or Internet. Phone calls using VoIP has advantages in terms of cheaper cost of PSTN phone to more than half, because the cost is calculated by the cost of the global nature of the Internet. Session Initiation Protocol (SIP) is a signaling protocol at the application layer which serves to establish, modify, and terminate a multimedia session involving one or more users. This SIP signaling has SIP message in text form that is used for session management by the SIP components, such as User Agent, Registrar, Redirect Server, and Proxy Server. To build a SIP communication is required SIP Express Router (SER) to be able to receive SIP messages, for handling the basic functions of SIP messages. Problems occur when the NAT through which affects the voice communication will be blocked starting from the sound that is not sent or one side of the sound are sent (half duplex). How that could be used to penetrate NAT is to use a given mediaproxy random RTP port to penetrate NAT.

Keywords—VoIP, SIP, SIP Express Router, NAT, Mediaproxy.

I. INTRODUCTION

NOW days telecommunications and computer networks support each other. This can be seen from the increasing use of both these technologies are increasingly promising every year. Especially in telecommunications technology that relies on communication between the person with the other one can do that voice communication is usually by using a fixed telephone line.

Phone rates in Indonesia currently quite high, this can be proved as a Direct Dialing Long Distance (DLD) or International Direct Dialing (IDD). Phone rates are like this are usually applied to systems Public Switched Telephone Network (PSTN) and Private Automated Branch exchange (PABX) is often performed Indonesian society in general. The system is highly dependent on the distance that is used to communicate. The farther the distance, the more expensive tariffs provided. Alternative systems of a problem that higher prices can use Voice over Internet Protocol (VoIP).

VoIP is a technology that can pass voice and data traffic in the form of packets over an IP network. Network can be used

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for intranet or Internet. Phone calls using VoIP has advantages in terms of cheaper cost of PSTN phone to more than half, because the cost is calculated by the cost of the global nature of the Internet [1].

Standardization of communication protocols in VoIP technology uses the Session Initiation Protocol (SIP). SIP is a signaling protocol at the application layer that serves to build, modify, and terminate a multimedia session involving one or several users. SIP signaling is shaped in the form of a SIP message text that is used for session management by the SIP components, such as User Agent, Registrar, Redirect Server, and Proxy Server [6].

In this research would be a VoIP network architecture on the environment and the UB Center SOI contained NAT between the two, then implementing the SIP Express Router Based Session Initiation Protocol with Mediaproxy method.

To build a SIP communication is required SIP Express Router (SER) to be able to receive SIP messages, for handling the basic functions of SIP messages, and when passing Network Address Translation (NAT). Because when you pass NAT, voice communications will be blocked starting from the sound that is not sent or one side of the sound are sent. How that could be used to penetrate NAT is to use the mediaproxy and rtpproxy an SER module of the ser.cfg. However, in this study made by the module which is standard mediaproxy from SER module designed to bypass NAT in VoIP communications between users.

II. VOIP PROTOCOL

VoIP protocols are generally divided into 2 parts namely Control / signaling and voice data. Traffic used to connect and maintain the actual traffic that is, the voice data. Also keep the entire network operation (router to router communications). Known also by the term Signaling Packet.

Control protocol / signaling to VoIP user can communicate with the phone. Several signaling that there currently are H.323, SIP, SCCP, MGCP, MEGACO and SIGTRAN. But the most popular and widely used is H.323 and SIP. This research use a signaling protocol in the form of Session Initiation Protocol (SIP) which will be discussed further in the next section [2].

User traffic information conveyed in the form of end-to-end is also known as Packet Voice [2]. Voice data protocols used include:

- Real-Time Protocol (RTP) is used the user voice. Each RTP packet contains a piece of the package of voice conversations. The amount of voice packet size depends

on the CODEC used. RTP information encapsulated in UDP packets. If the RTP packet is lost or didrop in a network, the RTP retransmission will not do (according to standard protocol UDP). This is so users do not wait too long (long pause) or delay, due to retransmission requests. RTP control protocol called RTCP (Real-time Transport Control Protocol) to control Quality of Service (QoS) and synchronization of different media streams.

- Compressed Real-Time Protocol (cRTP) which is a variant of the RTP. Compressed RTP packet headers eliminate many. By eliminating overhead, more efficient package. System with call cRTP can do 2 times more than the standard RTP.

III. VOICE CODING

ITU-T (International Telecommunication Union - Telecommunication Sector) create some standard for voice coding is recommended for the implementation of VoIP. Some standards are often known, among others [2]:

- G.711 PCM voice coding with 64 Kbps bandwidth.
- G.726 ADPCM voice coding with bandwidth 40, 32, 24, and 16 Kbps.
- G.728 voice compression coding with Code-Excited Linear-Predictive (CELP) with a bandwidth of 16 Kbps.
- G.729 voice compression coding CELP with bandwidth of 8 Kbps. The resulting sound quality to match ADPCM voice coding with 32 Kbps bandwidth.
- G.723.1 can be used to compress audio conversations or components in multimedia applications with bandwidth 5.3 and 6.3 Kbps. This standard is a component of the H.324 family.

IV. SESSION INITIATION PROTOCOL

SIP is a signaling protocol at the application layer that serves to build, modify, and terminate a session involving one or several users. These sessions include Internet multimedia conferences, Internet telephone, and multimedia distribution. SIP is a client-server protocol transported over TCP. The text such as HTTP or SMTP families [3].

There are several stages to establish and terminate a multimedia communication session:

1) User location: determine the location of the user who will communicate.
2) User availability: determine the level of the called party wishes to engage in communication.
3) User capability: the media and determine the parameters associated with the media to be used for communication.
4) Session setup: "ringing", establishment of the relationship between the caller and the called party.
5) Session management / handling: including transfer, modification, and termination of sessions [3].

Addressing
Communication on the SIP is done by sending HTTP-based message. Each user has a stated address in SIP-URI (Uniform

Resource Identification) or can be referred to as alias similar to email. its case insensitive, eg SIP-URI: sip: dhany@soi.ub which is equivalent to sip: DHANY@soi.ub [7]. In addition, the address can also be written in the tel-URL and then converted into a SIP-URI with the parameter 'user' blank 'phone'. Example: tel: +62341711599 equivalent to sip: +62341711599 @ soi.ub; user = phone. With the IP address can also be used as sip: dhany@10.100.0.45 [7]

SIP Components
1. User Agent (UA)
SIP UA is a component of the launch, receive, and close the communication session. UA consists of 2 major components [5]:

 a) User Agent Client (UAC): Components that begin the communication session.
 b) User Agent Server (UAS): Components that receive or respond to communications from the UAC session.

UA can be a hardware / software or hardphone / SoftPhone. Examples of this are hardphone USB phone, IP phone, and Analog Telephone Adapter (ATA). While this example is Idefisk SoftPhone, Kphone, Linphone, and X-Lite. For this study use X-Lite as SoftPhone.

2. Proxy Server
Intermediate components between the UA that receives request message from the UA and delivered directly to the recipient UA if it is on a single domain. If you are on a different domain name will be accepted by another proxy who then passed on to the UA in this proxy. Request can be served alone or delivered (forward) to other proxy or other servers. Translating and rewriting the request message prior to deliver on the UA goal or another proxy . In the proxy server can also to authenticate [5].

3. Redirect Server
Receiving requests from the UAC, comparing destination address to be achieved, after he was found, the address is returned to the UAC. Redirect Server does not save the state of communication sessions between the UAC and UAS when UAC mapping presented in [5].

4. Registar Server
Components that receive requests message REGISTER. Registrar may add a user authentication function for validation. Registrar stores the user database for authentication and location of the IP and port that a registered user to be contacted by other SIP component that serves as the Location Server also [5].

V. NETWORK ADDRESS TRANSLATION

NAT (Network Address Translation) is used to translate (translate) in a private address (private inside) to address the global external (outside global) [8]. Inside the network (internal network) is a collection of networks that became the target translation. Outside Network or outside the network show the other addresses that are usually on the Internet. The main function is to save the NAT global address of the

Internet, but it also has the function of NAT improve network security by hiding internal IP addresses from external networks [8].

NAT has many forms that can work in several ways:

1. Static NAT

Mapping an unregistered IP to IP is registered in one by one. Primarily used when a device (firewall, router, or computer) is required to be accessible from outside the network.

2. Dynamic NAT

Mapping the IP addresses that are not registered to the IP address of the registered a group of IP addresses that are registered. Dynamic NAT mapping also build one-to-one between the IP address is not listed and registered, but the mapping may depend on the registered address is available on the pool.

3. Overloading

The system of dynamic NAT, which maps multiple IP's that are not registered to a single IP listed (many-to-one) by using a different port. Therefore, also called Port Address Translation (PAT). By using PAT (NAT Overload), will have hundreds of users connected to the internet using only a single global address.

VI. SIP EXPRESS ROUTER (SER)

SIP Express Router (SER) is a SIP server and high-performance, configurable under the GNU open source. SER can be used as a registrar, proxy or redirect server to SIP applications [4].

SER built to receive SIP messages and allows the basic functions of the handling of SIP messages. Some functions SER module offers, such as the Apache web server. SER module provides capabilities that can be used in SER configuration files, ser.cfg. Ser.cfg configuration file to control the module that will contain and define how the module should be run by configuring the module variables [4].

VII. MEDIAPROXY

Mediaproxy is one solution for SER NAT Traversal. Mediaproxy known as NAT Traversal solution that handles all aspects of NAT at the SIP proxy location than the location of the SIP client [-PAU 05].

There are several advantages to deal with NAT in SIP proxy server, the main benefit is very easy clientnya configuration. Some special features of mediaproxy are [4]:

1) DNS SRV record capability for load distribution.
2) Mediaproxy can be installed on the remote server that does not run SER. To do this, SIP proxy load is reduced.
3) Web monitoring mediaproxy included with the distribution.

Mediaproxy is separate software that does not distributed with SER. Distribution SER enter only the glue that gives SER ability to communicate with mediaproxy with an example. This adhesive is known as the mediaproxy module is a module that can control the sending one or more real servers mediaproxy.

VIII. SYSTEM DESIGN

System Design SIP-based VoIP Communication is divided into three stages, namely: needs analysis, design hardware and software design. For hardware design consists of architectural design in SOI and VoIP Brawijaya Center. As for the design of software design in SOI NAT, SER server design in SOI, and the design of the server SER Mediaproxy SOI, show context diagram in Fig. 1.
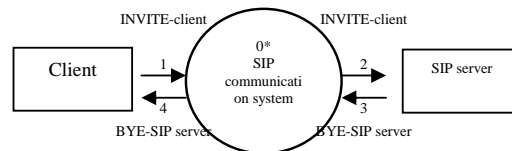

Fig. 1 DFD level 0

Diagram can extend in several sub-processes, shown in Fig. 2.
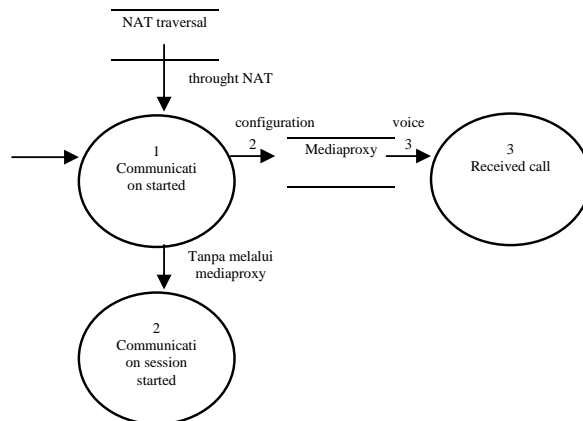

Fig. 2 DFD level 1

Explanation of the processes in fig. 2 are:

1. Process Communication Session Begins
   This process is the beginning of the formation of the communication session from client to server.
2. Process Connection Failed
   This process is because when the NAT, a form of data can not be passed because it does not use mediaproxy. Therefore, the connection happens to fail or not the connection was formed.
3. Receiving Process Call
   This process has occurred after successfully penetrate NAT between the client and server connections are formed.

Design of hardware computer network topology for the VoIP system to be implemented. Topology on this VoIP communication system using star topology, as in Fig. 3.
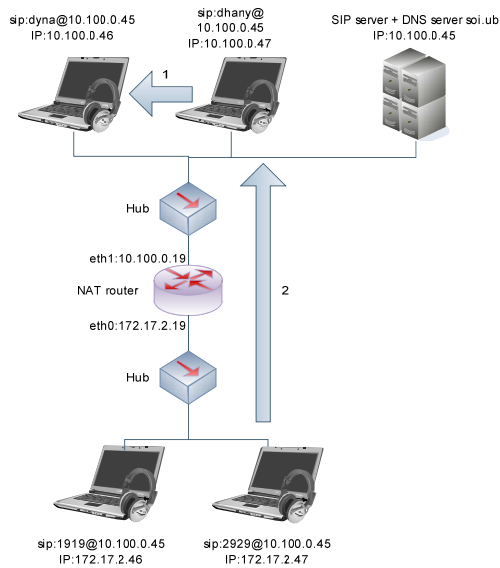
Fig. 3 Topology SIP-based VoIP network

Information flow of communication:
1) The flow of local communication (without going through NAT) either using mediaproxy or not.
2) The flow of long-distance communication (via NAT) either using mediaproxy or not.

VoIP network architecture uses the network clients SOI 2, NAT router is placed between a private network (Center) and public networks (soi) which translate the local address / internal IP addresses into globally unique. Switch is used as a liaison between the local networks with a router for directing traffic through the voice packet switch. Center client 2 on the network, DNS server hostname to name a computer, and the SIP server to manage all the servers listed in the SIP domain using soi.ub and to penetrate NAT settings with mediaproxy module.

The flow of communication made on the implementation of VoIP communications is done with 2 kinds of local (without going through NAT) and distance (via NAT) either using mediaproxy or not. The design and architecture of this system can be seen in the following table:

TABLE 1
LIST HOSTNAME AND IP OF EACH PC OR LAPTOP

Name	IP Address	Netmask	Gateway	Type
soi.ub	10.100.0.45	255.255.255.0	10.100.0.19	DNS Server
sip:soi.ub	10.100.0.45	255.255.255.0	10.100.0.19	SIP Server
sip:dyna@soi.ub	10.100.0.46	255.255.255.0	10.100.0.19	IP Public
sip:dhany@soi.ub	10.100.0.47	255.255.255.0	10.100.0.19	IP Public
Eth0 routetos	172.17.2.19	255.255.255.0	172.17.2.19	NAT Router
Eth1 routetos	10.100.0.19	255.255.255.0	10.100.0.1	NAT Router
sip:1919@soi.ub	172.17.2.46	255.255.255.0	172.17.2.19	IP Private
sip:2929@soi.ub	172.17.2.47	255.255.255.0	172.17.2.19	IP Private

The software is configured to be implemented and is designed with several sequential stages, namely the design of NAT in the SOI, SER server design in SOI, and the design of the server Mediaproxy SOI SER.

A. The design of NAT in SOI

NAT is a network application to translate from the private ip to public ip because of the limitations of public ip and private ip to access the Internet to the outside network (internet). However, when using NAT, so if you do the voice communication voice will sound in one direction only.

- Runs on CentOS 4.4 operating system for easy configuration.
- Using iptables.
- Separating private network (Center) with a public network (soi).

B. The design of the SOI SER Server

SER is a SIP-based VoIP software to be used as a proxy server, registrar server and location server.

- SER using version 0.9.6 using Ubuntu 8.10 operating system.
- There MySQL in the SER that serves to store and load the username, domain, password, alias, etc. from a database server for VoIP.
- There is a DNS server set up one machine to see SRV of SIP protocol.

C. Designing the Server SER Mediaproxy SOI

Mediaproxy SER module separate from the functional to the media through the NAT on the protocol or the RTP.

- Using random ports 60000:65000, not 8000 as the default RTP port.
- Using socket proxydispatcher and mediaproxy for two-way communication.
- Running mediaproxy process when there is a voice communication through the NAT.

IX. IMPLEMENTATION

Topology-based VoIP communications system that was designed SIP in Chapter IV, will be implemented at this point. There are five main steps undertaken in the implementation of this system, the installation of the PC Client, the installation of the PC NAT router, DNS server installation, the installation of the SIP Express Router, and installation mediaproxy.

A. PC Client (IP Private)

- Configuration of public and private ip to each client.
- Ip address configuration, netmas, the default gateway.

B. PC NAT Router

To stage the NAT router configuration in Centos 4.4 the steps required are as follows:
1. Two pieces Ethernet Card. Since the router requires a minimum of 2 Ethernet Card in order to work.
2. Setting IP address for the uplink (10.100.0.19) and

downlink (172.17.2.119) and also for the gateway

3. Enabling this PC to function as a router with a given value of 1 in ip forwarding.
4. Restart the network service to save the configuration of IP address and ip forwarding that have been made above.
5. To enable Network Address Translation (NAT) is given in the rule or rules with MASQUERADE iptables.
6. Service iptables restart to save the iptables configuration that has been made.
7. Ping test from a client on a public ip and private ip to prove that the NAT is active.

C. DNS Server

In order for the SIP server can be accessed and recognized by the user on a VoIP network, it requires DNS server. DNS server with the SIP server is located on the same IP address, due to limited computer resources that are not so influential to the performance of the server.

1. Adding a file / etc / resolv.conf.
2. Add the following to / etc / bind / named.conf
3. Create and fill in the file / etc / bind / db.soi.ub
4. Create and fill in the file / etc/bind/db.10.100.0
5. Restart the DNS service.
6. Do ping command to check the connection to the DNS server and dig to query the DNS server.
7. Verifying SRV records in DNS server.

D. SIP Express Router

The next step is to install the package SIP Express Router (SER) on Linux Ubuntu 8.10. As described in the previous chapter, that SER serves as registrar, proxy or redirect server to SIP applications.

1. First install the packages with the following SER apt-get command
2. Then install the mysql module for SER.
3. Adding variables to identify SIP_DOMAIN domain.
4. SER to create a database using the MySQL script.
5. Completing the SER database has been created and set the user in the database with serctl SER.
6. Adding a user account to the database that is used to obtain registration of users on the server voip.

IP	Status	Username	Password
10.100.0.46	Publik	dyna	dyna
10.100.0.47	Publik	dhany	dhany
172.17.2.46	Privat	1919	1919
172.17.2.47	Privat	2929	2929

E. Mediaproxy

After the SIP Express Router software has been installed, then the next step is to do the installation for mediaproxy can penetrate NAT. Installation mediaproxy added when passing through the NAT test. Also, when testing through NAT

without mediaproxy, then mediaproxy disabled. In the implementation below is the configuration when using mediaproxy:

1. Download and extract the files mediaproxy version 1.7.2.
2. To set up the mediaproxy.ini.sample to convert into mediaproxy.ini and change settings as needed.
3. Setting the boot process mediaproxy on Debian platform, because the server is using Ubuntu.
4. Starting the process in order mediaproxy module can be compiled mediaproxy.
5. View the log file to see that mediaproxy been successfully executed.

F. X-lite

1. Setting on each user agent in X-lite as it has been made in the design chapter.
2. Setting using the G.711 codec with a-law and u-law because it requires the bandwidth of 64 Kbps for audio

X. TESTING

In general, the purpose of this test is to determine if the sound or voice when passing through the NAT may occur two-way communication (full duplex) than without using mediaproxy. After testing, the performance or the so-called voice QoS as it passes the NAT with or without mediaproxy can be determined by analyzing the results. Parameters of this analysis is to calculate delay, jitter, and packet loss. This analysis will be compared by using the method without using mediaproxy and mediaproxy method as through NAT. This test is done by through several stages of data acquisition at each voice performance parameters.

A. Testing and Analysis Using VoIP with Mediaproxy

This test is performed to determine the influence of the use of methods in mediaproxy when local or network without going through NAT, and when passed NAT on SIP-based VoIP with. Then it can be seen from the performance results of this test. The test is conducted with 3 stages of testing, namely:

1. Testing mediaproxy activation.

The test is conducted to enable the process to mediaproxy through NAT and then compared to without using mediaproxy.

```
root@serbuntu:# cat /var/log/syslog | grep proxy

Dec 15 22:28:55 serbuntu mediaproxy[6627]: Listening for commands on
local socket `/var/run/mediaproxy.sock'
Dec 15 22:28:55 serbuntu mediaproxy[6627]: Listening for remote
commands on `10.100.0.45:25060'
Dec 15 22:28:55 serbuntu mediaproxy[6627]: Remote commands are
allowed from: 10.100.0.45, 127.0.0.1
Dec 15 22:28:55 serbuntu mediaproxy[6627]: Using IP address
10.100.0.45' for the RTP/RTCP proxy
Dec 15 22:28:55 serbuntu proxydispatcher[6630]: Listening for
commands on local socket `/var/run/proxydispatcher.sock'
Dec 15 22:28:55 serbuntu proxydispatcher[6630]: Listening for
mediaproxy media timeout requests on 0.0.0.0:25061
Dec 15 22:28:55 serbuntu proxydispatcher[6630]: Default mediaproxy
server is on `/var/run/mediaproxy.sock'
```

2. Call without going through NAT, or from the public ip to public ip.

The test is conducted to determine if a VoIP call while without going through NAT and use mediaproxy or from a public IP to public IP can work well in both directions and to determine the QoS performance in each parameter.

3. Calls to pass NAT or from private IP to public IP.

The test is conducted to determine if a VoIP call as it passes the NAT and without mediaproxy or from a public IP to public IP can work well in both directions and to determine the QoS performance in each parameter.

XI. CONCLUSION

Base on results of testing the implementation of the SIP Express Router is based on Session Initiation Protocol by using the method in VoIP mediaproxy can be concluded that:

- Voice communication without going through NAT will be two-way communication or full duplex, whereas when the voice communication through the NAT will occur one-way communication only, or half duplex, because the 8000 RTP ports are blocked by NAT.

- For media or SIP signaling there is no problem even if through NAT or not, because the only opening, building, and close the communication session only. Another case with RTP could not penetrate NAT.

- QoS parameters when no through NAT and mediaproxy obtained with the average end-to-end delay 35.78 ms, 26.38 ms jitter, packet loss 0% which means judged to be "good" for end-to-end delay and packet loss, while the "acceptable" to jitter. QoS parameters when passing through NAT and mediaproxy obtained with the average end-to-end delay 35.93 ms, 25.28 ms jitter, packet loss over the 1.5% which means judged to be "good" to delay end to end, "acceptable" for jitter, and "poor "to packet loss. Large packet loss is caused by damage RTP packets through NAT by RTP port change given by mediaproxy.

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