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SIP SERVER

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Abstract- *In ours paper, For a fully functional voice exchange we require to set up a server based on Asterisk / SIP, connecting clients to the server with the help of phones and then comes the configuration aspects of the phones with the server. Here in our implementation we have connected the clients to the server with the help of SIP Server.*

1. INTRODUCTION

Session Initiation Protocol (SIP) is an ASCII-based, application-layer control protocol that can be used to establish, maintain, and terminate calls between two or more endpoints. SIP is designed to address the functions of signaling and session management within a packet telephony networks. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

SIP is a peer-to-peer protocol. The peers in a session are called user agents (UAs). A UA can function in one of the following roles:

- User-agent client (UAC)--A client application that initiates the SIP request.
- User-agent server (UAS)--A server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.

SIP endpoint is capable of functioning as both a UAC and a UAS, but functions only as one or the other per transaction. Whether the endpoint functions as a UAC or a UAS depends on the user agent that initiated the request

2. OBJECTIVE

i. Users will be able to communicate effectively, speedily, securely. It will enhance the privacy and confidentiality of mobile communication.

ii. Implementing voice encryption on third generation GSM data or GPRS servers will provide a better encrypted voice speed and clarity.

iii. The configuration and usage of proxy server (SIP / Asterisk) through defining routing call and handset registration mechanisms.

iv. It will enable the users to communicate with each other in an encrypted format.

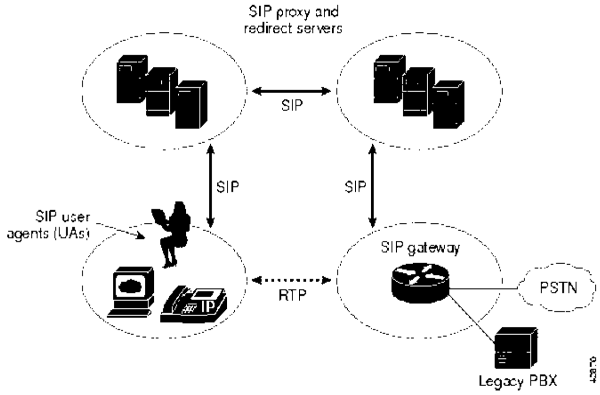

Figure 1: Block diagram of SIP Server

3. IMPLEMENTATION

Step 1: Configure Addressing

Configure the router with the IP address shown in the diagram.
R1(config)# interface fastethernet 0/1
R1(config-if)# ip address 172.16.10.1 255.255.255.0
R1(config-if)# no shutdown
Finally, configure the IP address 172.16.10.50/24 below on the interface
Click OK once to apply the TCP/IP settings and again to exit the LAN interface properties dialog box. Configure Host B similarly, using 172.16.10.60/24 as the IP address.

Step2 : Configure Router Telephony Service

Cisco's Call Manager Express (CME) is installed

R1(config)# telephony-service

R1(config-telephony)# ?

R1(config-telephony)# max-ephones 2

R1(config-telephony)# max-dn 10

R1(config-telephony)# keepalive 15

Configure a system message using the system message *line* command. This line will appear on phones associated with the CME.

R1(config-telephony)# system message Cisco VOIP

R1(config-telephony)# create cnf-files

Finally, configure the source address for SCCP using the ip source address *address* port *port* command. Use the local Fast Ethernet address with a port number of 2000.

R1(config-telephony)# ip source-address 172.16.10.1 port 2000

Step 3: Create Directory Numbers:

To configure a directory number, use the global configuration ephone-dn *tag* command. Use a tag of 1 for the first phone.

R1(config)# ephone-dn 1

At the ephone-dn configuration prompt, use the number *number* command to configure a phone number of 5001. Assign a name of "Host A" with the name *name* command.

R1(config-ephone-dn)# number 5001
R1(config-ephone-dn)# name Host A

Configure ephone-dn 2 similarly.

R1(config-ephone-dn)# ephone-dn 2

R1(config-ephone-dn)# number 5002
R1(config-ephone-dn)# name Host B

Step 4: Create Phones:

You will need to find out the MAC addresses of the hosts

In R1, enter the ephone configuration prompt by typing the ephone *tag* command in global configuration mode.

R1(config)# ephone 1

Associate the MAC address with this ephone using the mac-address *address* command. The address must be in the format HHHH.HHHH.HHHH.

R1(config-ephone)# mac-address 0002.B3CE.72A3

Use the type *type* command to configure the type of phone. Since you are configuring Cisco IP Communicator to simulate Ethernet phones, use cipc as the phone type.

R1(config-ephone)# type cipc

R1(config-ephone)# button 1:1

Apply a similar configuration for ephone 2. Change the configuration parameters where appropriate.

R1(config-ephone)# ephone 2
R1(config-ephone)#mac-address 0009.5B1B.67BD
R1(config-ephone)# type cipc
R1(config-ephone)# button 1:2

Step 5: Install Cisco IP Communicator

Step 6: Run Cisco IP Communicator.

Figure 2: Main Screen on Host A

The IP phone has successfully registered with R1. Note the correct banner at the bottom of the color display and the correct directory number in the upper-right corner. On R1, look at the debug output generated when R1 registered.

Configure Host B similarly

Figure 3: Main Screen on Host B

Step 7: Establish a Call from Host A to Host B

On Host A, dial extension 5002 (Host B's) by typing in the numbers on your keyboard or using the visual keypad in CIPC. Then click the Dial softkey.

4. RESULT

Figure 4: Host A calling Host B

Figure 5: Host B receiving the call from Host A

SIP Server has been accepted the audio calls between two users and also audio conference calls between many users in the remote area. Using SIP Server the common message to all the users which are registered.

5. CONCLUSION AND FUTURE WORK

We expect that design and implementation presented in this paper will be a valuable developing guide for similar kind of operations. Asterisk based voice exchange provides us with a much better alternative solution. Its not only cost effective but also provides us with various features which we generally don't get with the conventional circuit switched based PBX .Moreover the system also provides for unlimited expansion and since it runs on a secure operating system like LINUX, its much less prone to viruses, worms and hackers. As far as future work is concerned, we would like to work on connecting our Asterisk PBX with the conventional circuit switched networks with the help of PCI cards like for example Digium's TDM400P[7] .

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