Arquitetura de Redes Avançadas

Laboratory Guide

VoIP (part B)

SIP (Call Forwarding)

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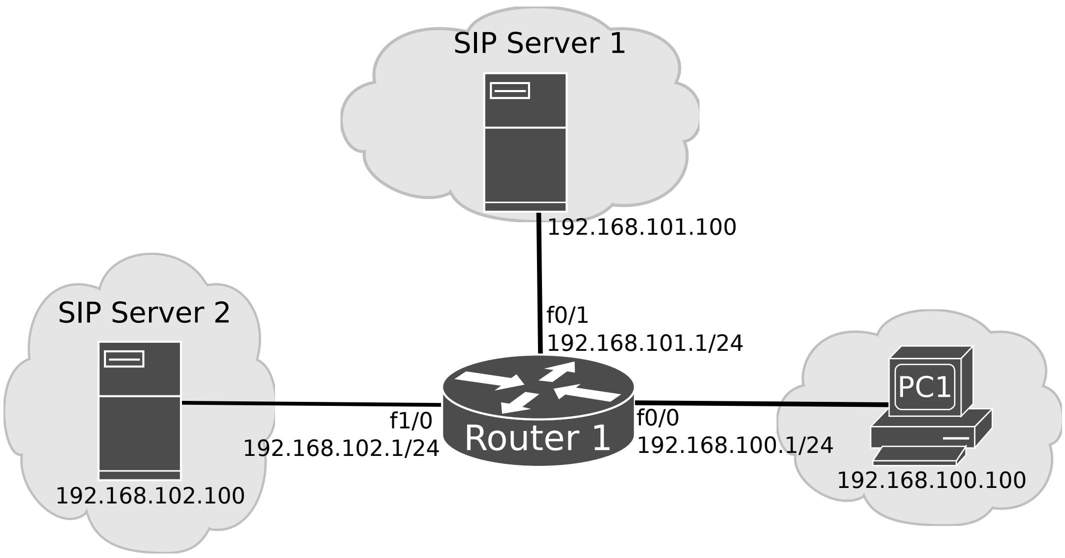

1. Assemble the above depicted network in GNS3. SIP servers should be virtual machines with a Linux server with Asterisk installed, ans PC1 could be your host machine.
Virtual machines can be connected to GNS3 by (i) bridging them to tap interfaces or using host-only adapters, and (ii) associating clouds in GNS3 with the respective interface. The host machine can be connected using tap or physical interfaces (depending on OS). Server 1 can be the SIP server configured before (part A), and Server 2 must be an Asterisk server with default configuration.
Routing must be active in Router 1 and the following additional routing rules must be added to SIP servers and PC1 (assuming Linux) :

```
Server1# route add default gw 192.168.101.1
…
Server2# route add default gw 192.168.102.1
…
PC1# route add -net 192.168.101.0/24 gw 192.168.100.1
PC1# route add -net 192.168.102.0/24 gw 192.168.100.1
```

Test interconnectivity between both servers and PC1.

2. PC1 (SIP phone/Ekiga) must be registered with SIP Server 1. Make a test call to 2000@192.168.101.100 assuming that SIP Server 1 /etc/asterisk/extensions.conf file already has extension 2000 configured as:

```
[phones]
exten => 2000,1,Answer(500)
exten => 2000,n,Playback(demo-congrats)
exten => 2000,n,PlayBack(vm-goodbye)
exten => 2000,n,Hangup()
```

Note: For better sound quality disable all audio codecs <u>except Speex</u>

3. To configure SIP Server 1 to forward all calls to any 234* number to SIP Server 2, first define the remote server type, name, address, and credentials in /etc/asterisk/sip.conf file by adding to it:

```
[Server2]
type=peer
host=192.168.102.100
secret=labcom
username=Server1
```

and, define the generic extension forwarding rule by adding to /etc/asterisk/extensions.conf file:

```
exten => _234.,1,Dial(SIP/${EXTEN}@Server2,10)
```

Note: "." represents multiple digits, and ${EXTEN} is an Asterisk variable that contains the called extension.

Start a packet captured in the link between Server 2 and Router 1. Make a test call from PC1 to 2341234@192.168.101.100. Analyze the SIP packets exchanged between Server 1 and Server 2.

4. In SIP Server 2, define Server 1's name, type, address, credentials and context in /etc/asterisk/sip.conf file by adding to it:

```
[Server1]
type=peer
host=192.168.101.100
secret=labcom
context=phones
```

and, define what to do with the received calls (234*) by adding to /etc/asterisk/extensions.conf file:

```
[phones]
exten => _234.,1,Answer(500)
exten => _234.,2,PlayBack(vm-received)
exten => _234.,3,SayDigits(${EXTEN:3})
exten => _234.,n,PlayBack(vm-goodbye)
exten => _234.,n,Hangup()
```

Start a packet captured in the link between Server 2 and Router 1. Make a test call from PC1 to 2341234@192.168.101.100.

Analyze the SIP packets exchanged between Server 1 and Server 2. Explain the SIP forwarding mechanism.