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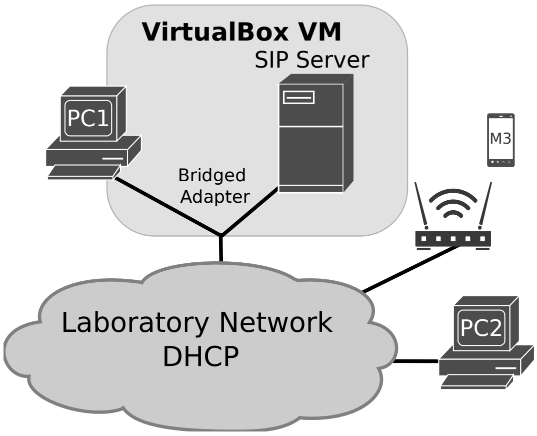
Redes de Comunicações 2



SIP, RTP and RTCP Protocols

1. The SIP server must be implemented as a virtual Ubuntu/Debian server. Connect the VM to your PC (PC1) Wi-Fi/Ethernet interface using a Bridged Adapter.

Generate a different MAC address for your VM network interface:

In VirtualBox: Settings→ Network→ Adapter/Advanced→ Click the icon right of the MAC address field.

In QEMU/virt-manager: Not required.

Boot the SIP Server and identify the DHCP acquired IPv4 address (assuming from now on 192.168.220.x).

Identify the DHCP acquired IPv4 address of your PC1. From your PC1, test the connectivity with your SIP Server (using the command ping).

Note: You may connect to your server using SSH (Linux: `ssh labcom@192.168.220.x` , Windows: install a SSH client and connect to the server), in order to make future configurations easier by command line copy and paste.

Note2: The server should have installed the VoIP gateway and conference server (Asterisk) with default configuration.

Change to root terminal with the command `sudo su`.

Editing the /etc/asterisk/sip.conf file add a new user (to the end of the file):

```
[PintoDaCosta]
type=friend
host=dynamic
secret=labcom
context=phones
allow=all
```

And editing the /etc/asterisk/extensions.conf file add a new section (to the end of the file) with a phone extension (2000) to deliver a welcome message:

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

Restart Asterisk: `service asterisk restart`

2. In your PC1, start a VoIP softphone (linphone) and login to your SIP server account with the credentials defined before
In, HOME → ACCOUNT ASSISTANT → USE A SIP ACCOUNT

```
Username: PintoDaCosta
SIP Domain: 192.168.220.x
Transport: UDP
```

or in, Preferences → SIP accounts → ADD ACCOUNT

```
SIP address: sip:PintoDaCosta@192.168.220.x
SIP Server address: <sip:192.168.220.x;transport=udp>
Transport: UDP
```

On top left status/name region, choose the active account to the one in your server. Start a Wireshark capture **on the server VM.**
On your VoIP client Preferences → SIP accounts → Edit → Register (toggle), unregistered and register in SIP server a couple of times.
>> Analyze the exchanged SIP messages and identify the purpose of the REGISTER messages (and if present, SUBSCRIBE and PUBLISH messages).

3. Start new Wireshark capture **on the server VM** and change the status in your VoIP client (top left) to BUSY. Change the status back to AVAILABLE. Analyze the exchanged SIP messages.
>> Identify the purpose of the PUBLISH messages (based on the XML contents of the message body).

4. Start new Wireshark capture, make an audio call to extension 2000 (sip:2000@192.168.220.x) for a welcome message from server, and wait for the end of the message:
>> Analyze the exchanged SIP request and status messages and identify the purpose of the following request messages: INVITE, ACK and BYE.
Note: If the call has no sound some codecs may be missing. To solve the problem install VLC in your client (Android by default has the codecs).

5. Editing the /etc/asterisk/sip.conf file add a new user:

```
[RuiCosta]
type=friend
host=dynamic
secret=labcom
context=phones
allow=all
```

Editing the /etc/asterisk/extensions.conf file define extension numbers for all users:

```
[phones]
...
exten => 2001,1,Dial(SIP/PintoDaCosta,10)

exten => 2002,1,Dial(SIP/RuiCosta,10)
```

Restart Asterisk: `service asterisk restart`

6. Using a second PC (PC2), or a mobile phone (M3) with the linphone app installed, or making arrangements with a neighbor group, register the second user with your Asterisk server. On your PC1, start new Wireshark capture and make an audio call to the second user extension (2002@192.168.220.x). After a few seconds hang up the call. Analyze the exchanged SIP request and status messages.
>> Identify the purpose of the following SIP messages: INVITE, ACK, BYE, Trying, Ringing and OK.

OK : Sinalizações de confirmação por parte do emissor
Ringing: Estabelecimento da chamada