# Asterisk labo

## Labo 3a:

Install asterisk on ubuntu vm

### SIP.conf

[1000]

type=friend

username=John

secret=test

host=dynamic

context=internal

mailbox=john@example

[1001]

type=friend

username=Jane

secret=test

host=dynamic

context=internal

mailbox=jane@example

[1002]

type=friend

username=Bob

secret=test

host=dynamic

context=internal

mailbox=bob@example

## Extensions.conf

[internal]

exten => 1000,1,Answer()

same => 2,Playback(u\_mama)

same => 3,Dial(SIP/1000,5)

same => 5,Hangup()

exten => 1001,1,Answer()

same => 2,Playback(u\_mama)

same => 3,Dial(SIP/1001,5)

same => 5,Hangup()

exten => 1002,1,Answer()

same => 2,Playback(u\_mama)

same => 3,Queue(sales)

(There were issues with my voicemail, using queue instead).

### queues.conf

[general]

autofill=yes

shared\_lastcall=yes

[sales]

strategy=ringall

timeout=15

retry=5

wrapuptime=10

maxlen=0

joinempty=yes

leavewhenempty=no

ringinuse=no

member => SIP/1000

member => SIP/1001

member => SIP/1002

## labo 3b

### Install tftp: <https://askubuntu.com/questions/201505/how-do-i-install-and-run-a-tftp-server>

### Extensions.conf:

<DIALTEMPLATE>

<TEMPLATE MATCH="\*" TIMEOUT="5"/>

</DIALTEMPLATE>

### SEP2C3F38C8FDCF.cnf.xml:

<device>

<deviceProtocol>SIP</deviceProtocol>

<sshUserId>admin</sshUserId>

<sshPassword>cisco</sshPassword>

<!-- Configuration related to the device pool -->

<devicePool>

<!-- Date and time settings -->

<dateTimeSetting>

<dateTemplate>D-M-Y</dateTemplate>

<timeZone>Central Standard/Daylight Time</timeZone>

<!-- NTP (Network Time Protocol) settings -->

<ntps>

<ntp>

<name>pool.ntp.org</name>

</ntp>

</ntps>

</dateTimeSetting>

<!-- Call Manager Group settings -->

<callManagerGroup>

<members>

<member priority="0">

<callManager>

<!-- Ports and process node information -->

<ports>

<ethernetPhonePort>2000</ethernetPhonePort>

<sipPort>5060</sipPort>

<securedSipPort>5061</securedSipPort>

</ports>

<processNodeName>10.0.10.1</processNodeName>

</callManager>

</member>

</members>

</callManagerGroup>

</devicePool>

<!-- SIP Profile settings -->

<sipProfile>

<sipProxies>

<registerWithProxy>true</registerWithProxy>

</sipProxies>

<enableVad>false</enableVad>

<preferredCodec>g711</preferredCodec>

<natEnabled></natEnabled>

<phoneLabel>Cisco Phone</phoneLabel>

<!-- SIP Lines configuration -->

<sipLines>

<line button="1">

<featureID>9</featureID>

<featureLabel>838</featureLabel>

<proxy>USECALLMANAGER</proxy>

<port>5060</port>

<name>1001</name>

<authName>1001</authName>

<authPassword>test</authPassword>

<messagesNumber>\*97</messagesNumber>

</line>

</sipLines>

<!-- Dial template configuration -->

<dialTemplate>dialplan.xml</dialTemplate>

</sipProfile>

<!-- Common Profile settings -->

<commonProfile>

<phonePassword></phonePassword>

</commonProfile>

<loadInformation></loadInformation>

<versionStamp>1143565489-a3cbf294-7526-4c29-8791-c4fce4ce4c37</versionStamp>

<servicesURL></servicesURL>

<displayIdleTimeout>NONE</displayIdleTimeout>

</device>

### switch config:

ena

conf t

hostname switch-M

ip domain-name pieter.local

crypto key generate rsa

2048

line vty 0 15

transport input ssh

login local

exit

username admin password cisco

enable secret cisco

end

!

conf t

int vlan 1

ip address 10.0.0.1 255.255.0.0

int fa0/13

switchport mode access

switchport access vlan 1

vlan 10

name voice

int vlan 10

ip address 10.0.10.2 255.255.255.0

vlan 20

name data

int vlan 20

ip address 10.0.20.2 255.255.255.0

int fa0/1

switchport mode access

switchport access vlan 10

int fa0/2

switchport mode access

switchport voice vlan 10

switchport access vlan 20

ip dhcp pool VLAN10

network 10.0.10.0 255.255.255.0

default-router 10.0.10.1

option 150 ip 10.0.10.1

exit

ip dhcp excluded-address 10.0.10.1 10.0.10.10

ip dhcp pool VLAN20

network 10.0.20.0 255.255.255.0

default-router 10.0.20.1

exit

ip dhcp excluded-address 10.0.20.1 10.0.20.10

# 3C

Capture sip packages:  
A computer screen with white text

Description automatically generated  
analysing in wireshark:

Register:

A screenshot of a computer

Description automatically generated

Start call:  
A screenshot of a computer

Description automatically generated

End call:

A screenshot of a computer

Description automatically generated

You can see the RTP messages:

A screenshot of a computer

Description automatically generated

You can use telephony in wireshark to replay the voice message:  
A screenshot of a computer

Description automatically generated

(note: one of my sip clients is on the same ip as the server)

First and second sip invite:  
A screenshot of a computer

Description automatically generated

Without sip reinvites:

A screenshot of a computer

Description automatically generated

Yes the entire call will happen through the server now, instead of through the peers.

## Extra

Creating new peer:

A computer screen with white text

Description automatically generated

New context file:

A computer screen shot of a program

Description automatically generated

I also created some new audio files with a free tts service.

## SIP trunk

Created an account on weepee.

Called their number to activate the account, this didn’t work for a day, then it did (not sure why).

Created the trunk in the sip config:  
A screenshot of a computer program

Description automatically generated

And a new context to forwared external calls to in the extensions.conf files:

A screenshot of a computer screen

Description automatically generated

