

# Project 2 - Module 4

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VoIP, or Voice over Internet Protocol, is a technology that transmits voice communications over the internet instead of traditional phone lines. Asterisk is a popular open-source software framework that turns a computer into a VoIP private branch exchange (PBX), enabling it to manage phone calls, voicemail, and other telephony features. The steps below show how the team configured Asterisk to receive phone calls.

## Step 1: Install Asterisk

Steps to install Asterisk onto Raspbian OS:

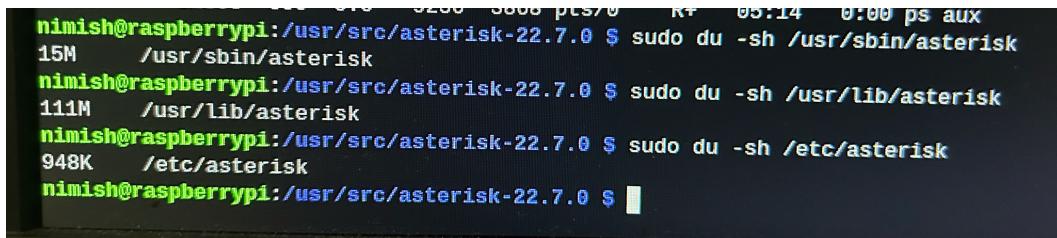
1. Extract the .tar file into a directory
  - o Sudo tar -xvf [asterisk-22.7.0.tar.gz](#) ; cd asterisk-22.7.0
2. Run the configure script
  - o ./configure
3. Then build the Asterisk .tar file using make
  - o Sudo make -j4
4. Continue with the command make for the installs, samples, and config.
  - o Sudo make install
    - i. Installs binaries into /usr/sbin
  - o Sudo make samples
    - i. Installs example config into /etc/asterisk
  - o Sudo make config
    - i. Enables Asterisks as a system Service
5. Enable/start Asterisk as a system service
  - o Sudo systemctl enable asterisk
  - o Sudo systemctl start asterisk

```
/usr/bin/install -c -d "/var/spool/asterisk/voicemail/default/1234/INBOX"
build_tools/make_sample_voicemail "//var/lib/asterisk" "//var/spool/asterisk"
Installing file phoneprov/000000000000.cfg
Installing file phoneprov/000000000000-directory.xml
Installing file phoneprov/000000000000-phone.cfg
Installing file phoneprov/polycom_line.xml
Installing file phoneprov/polycom.xml
Installing file phoneprov/snmp-mac.xml
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $ sudo make config
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $ sudo systemctl enable asterisk
asterisk.service is not a native service, redirecting to systemd-sysv-install.
Executing: /usr/lib/systemd/systemd-sysv-install enable asterisk
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $ sudo asterisk -rvvvv
Unable to connect to remote asterisk (does '/var/run/asterisk/asteriskctl' exist?)
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $ sudo asterisk -rvvvv
Unable to connect to remote asterisk (does '/var/run/asterisk/asteriskctl' exist?)
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $ sudo asterisk -rvvvv
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $ sudo systemctl start asterisk
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $ sudo asterisk -rvvvv
Asterisk 22.7.0, Copyright (C) 1999 - 2025, Sangoma Technologies Corporation and others
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=====
Connected to Asterisk 22.7.0 currently running on raspberrypi (pid = 51978)
raspberrypi*CLI>
```

Figure 1:Asterik setup

Then, using the commands below, the memory size of the Asterisk files can be seen.

- Sudo du -sh /usr/bin/asterisk
- Sudo du -sh /usr/lib/asterisk
- Sudo du -sh /etc/asterisk



A terminal window showing the output of the 'ps aux' command. It lists three processes related to Asterisk:

```
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $ sudo du -sh /usr/sbin/asterisk
15M    /usr/sbin/asterisk
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $ sudo du -sh /usr/lib/asterisk
111M   /usr/lib/asterisk
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $ sudo du -sh /etc/asterisk
948K   /etc/asterisk
nimish@raspberrypi:/usr/src/asterisk-22.7.0 $
```

Figure 2: Memory Size

## Step 2: Configure Voicemail

Zoiper was used as a softphone client, and voicemail was configured for extension 100. This required updates to pjsip.conf, voicemail.conf, and extensions.conf.

In pjsip.conf, a transport object was created to allow Asterisk to handle SIP traffic over UDP on the designated port. AOR (Address of Record) entries were added to specify the user's location and define max\_contacts, which determines how many devices can register. Authentication parameters were then added, followed by the endpoint definition used by Asterisk for SIP communication.

In voicemail.conf, a voicemail mailbox was added for extension 100. In extensions.conf, a dialplan entry for extension 100 was created to ensure proper call routing. With these configurations in place, Zoiper can dial 100 and access the associated voicemail greeting.

1. Outside the asterisk CLI, perform these steps to configure Asterisk for voicemail.
  - Open extensions.conf and edit the file to set the extension value and what happens in the extension. The line below opens the file.
    - i. Sudo nano /etc/asterisk/extensions.conf
    - ii. Extension.conf changes:

```
[default]
exten => 100,1,Answer() > answers incoming call at extension 100
same => n,Wait(1)      > delay of 1 sec
same => n,Voicemail(100@default)
                           > sends caller to voicemail box 100
same => n,Hangup()     > hangs up after voicemail
```
  - Open pjsip.conf and edit the file to configure the SIP endpoint. Defining the transport, authentication, and endpoint allows Asterisk to receive a telephone call. The line below opens the file.
    - i. Sudo nano /etc/asterisk/pjsip.conf

ii. pjsip.conf changes:

```
[transport-udp] > name of the transport profile
type=transport > Tells Asterisk this SIP transport layer
protocol=udp > SIP will use UDP
bind=0.0.0.0:5060 > listen SIP traffic here
[7001] > start of setting for SIP user 7001
type=aor > declares this block stores contact info for the user
max_contacts=1 > only one device
[7001] > Auth settings for 7001
type=auth > block handles password auth
auth_type=userpass > username + password auth
username=7001 > SIP login username
password=7001 > SIP login password
[7001] > endpoint definition label
type=endpoint > defines SIP devices' capabilities
aors=7001 > link this endpoint to AOR
auth=7001 > use auth block
context=default > calls enter [default] from extensions.conf
disallow=all > disallow all audio codecs first
allow=ulaw > enable ulaw audio codec
```

- Open voicemail.conf and edit the file to create a voicemail mailbox for extension 100 to store messages. The line below opens the file.
  - i. Sudo nano /etc/asterisk/voicemail.conf
  - ii. voicemail.conf changes:  
100 => 1234,Lab user,[lab@example.com](mailto:lab@example.com) > Defines mailbox 100 with pin and assigns the lab user + email
- Note: Outside asterisk CLI, use sudo systemctl restart asterisk after changes have been implemented

2. Get into the CLI by opening Asterisk CLI

- Sudo asterisk -rvvvv (verbose level up to the user)

3. Continuing in Asterisk CLI, reload the Voicemail and Dialplan after configuring .conf files

- Voicemail reload
- Dialplan reload

4. Create an account on Zoiper with the same username and password, and give the domain as the IP of the Raspberry Pi and set the protocol to SIP.

December 4th, 2025

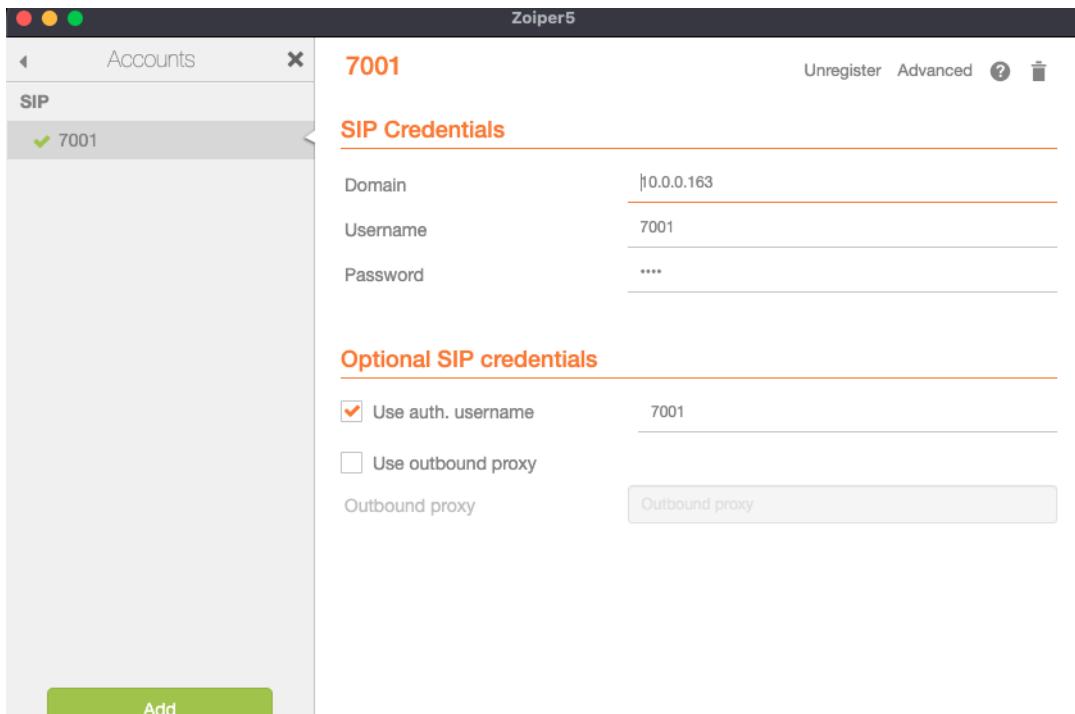


Figure 3: Zoiper Menu

The screenshot shows a terminal window with a dark background and light-colored text. The title bar indicates the session is running on a Raspberry Pi with user 'nimish' and IP '10.0.0.163'. The window displays a log of Asterisk command-line operations. It starts with 'hostname' commands, followed by 'asterisk -r' to restart the Asterisk service. The log then shows the execution of various extensions and scripts, including 'vm-theperson.gsm', 'vm-intro.gsm', and 'beep.gsm'. A warning message at the end states: '[Dec 3 06:47:38] NOTICE[52211]: res\_pjsip\_session.c:4041 new\_invite: 7001: Call (UDP:10.0.0.199:53346) to extension ''97'' rejected because extension not found in context 'default''. The session ends with a 'quit' command.

Figure 4: Asterisk Setup

### Step 3: Test Phone Call

The phone call test can be seen and heard in the MP4 file within the .zip.