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<u>Project 2 - Module 4</u>

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

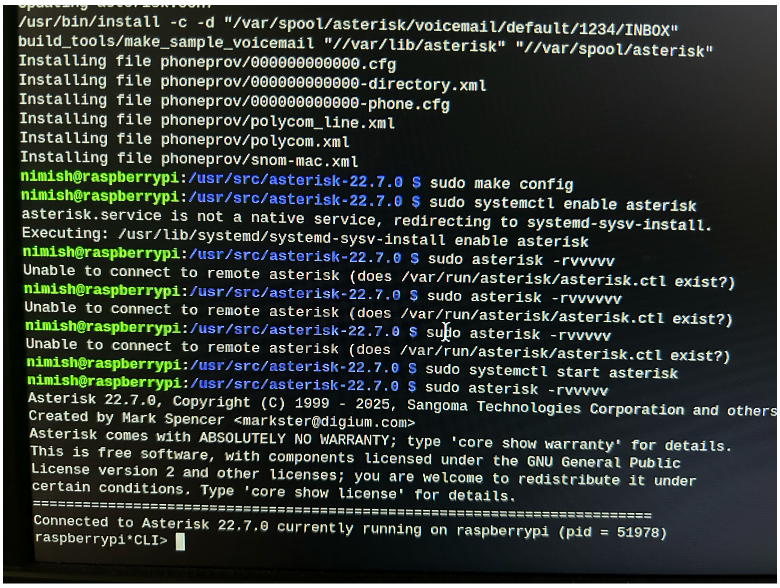

Figure 1:Asterik setup

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`

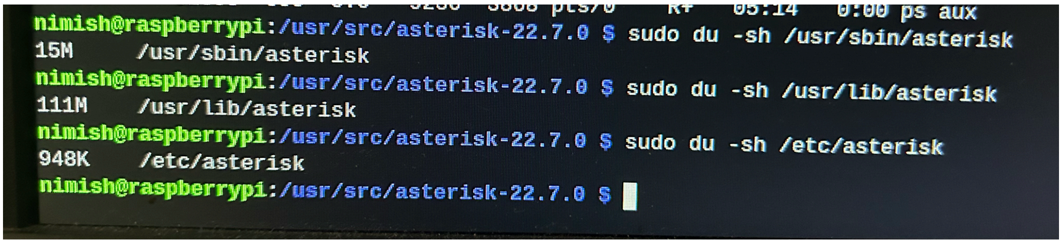

Figure 2: Memory Size

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. <u>Extension.conf changes</u>:
        ```
        [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()       > hangsup 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` > 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 -rvvvvv` (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.

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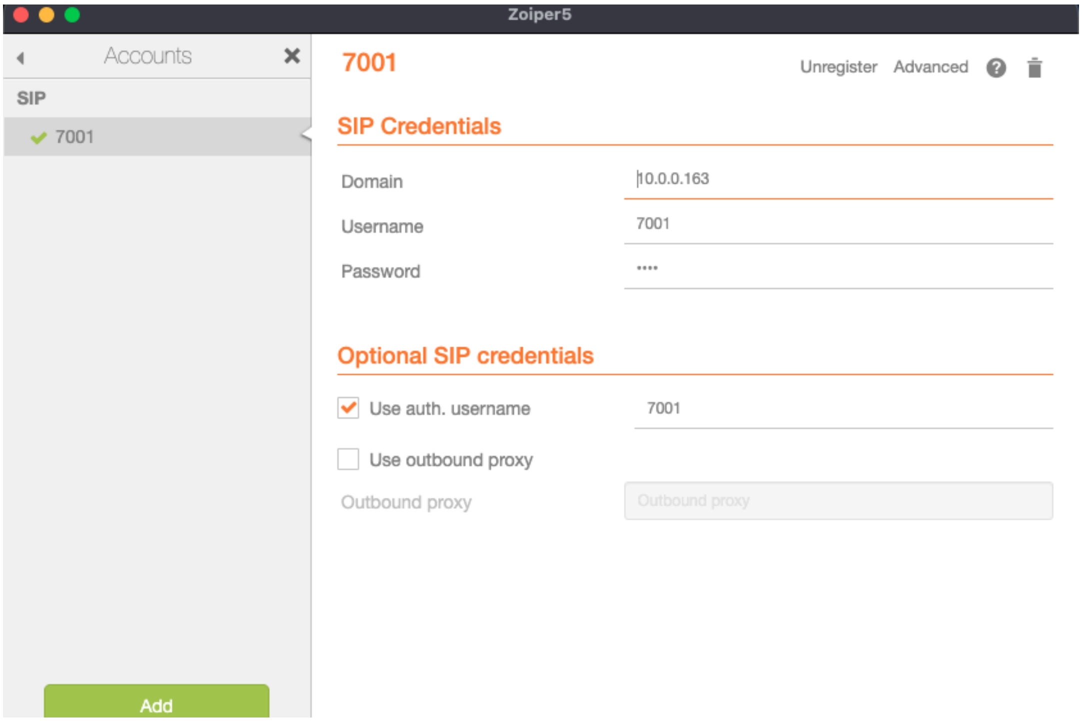

Figure 3: Zoiper Menu

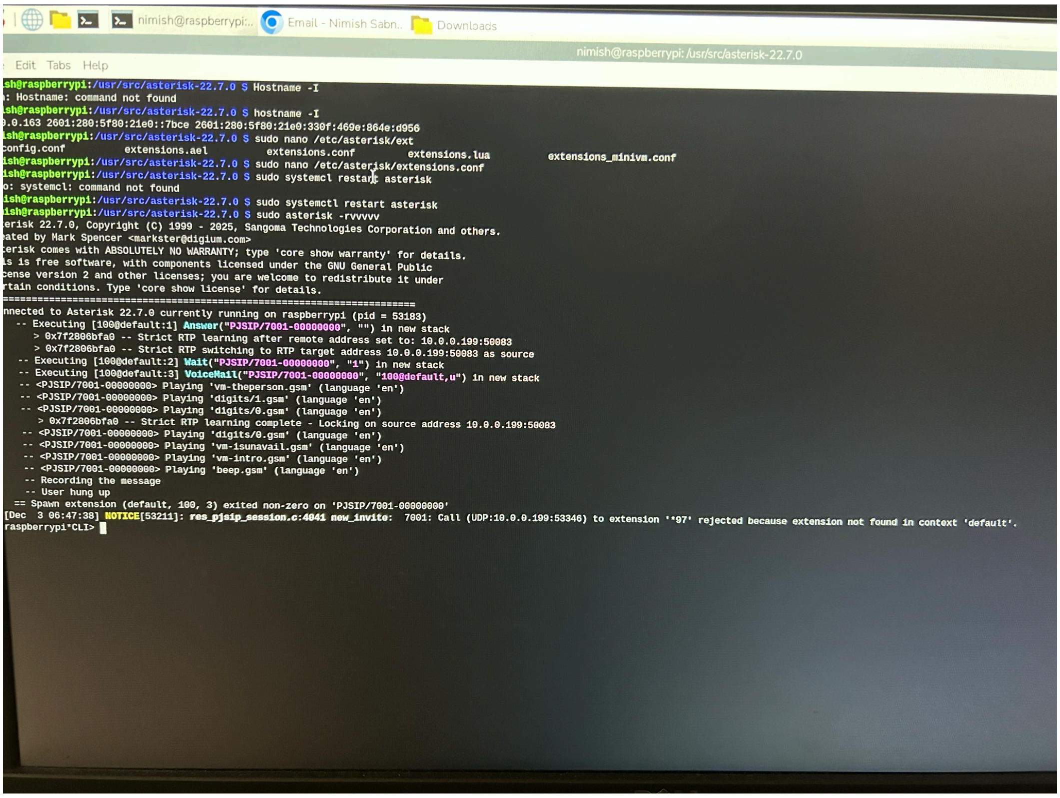

Figure 4: Asterisk Setup

Step 3: Test Phone Call

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