Module 4: Build your own PBX with Asterisk

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1. How much memory is used by the code?
   29.4 MB
   Asterisk was installed using the Debian repositories for Raspbian Linux.

```
● asterisk.service - Asterisk PBX
   Loaded: loaded (/lib/systemd/system/asterisk.service; enabled; vendor preset: enabled)
   Active: active (running) since Wed 2024-11-27 11:27:00 MST; 1h 37min ago
     Docs: man:asterisk(8)
 Main PID: 2678 (asterisk)
    Tasks: 69 (limit: 2200)
   Memory: 29.4M
   CGroup: /system.slice/asterisk.service
           ├─2678 /usr/sbin/asterisk -g -f -p -U asterisk
           └─7051 astcanary /var/run/asterisk/alt.asterisk.canary.tweet.tweet.tweet 2678

Nov 27 12:55:57 raspberrypi asterisk[2678]: [Nov 27 12:55:57] NOTICE[9647]: res_config_ldap.c:1832 parse_config: No directory user fo
Nov 27 12:55:57 raspberrypi asterisk[2678]: [Nov 27 12:55:57] ERROR[9647]: res_config_ldap.c:1858 parse_config: No directory URL or h
Nov 27 12:55:57 raspberrypi asterisk[2678]: [Nov 27 12:55:57] NOTICE[9647]: res_config_ldap.c:1776 reload: Cannot reload LDAP RealTim
Nov 27 12:55:57 raspberrypi asterisk[2678]: [Nov 27 12:55:57] NOTICE[9647]: cdr.c:4541 cdr_toggle_runtime_options: CDR simple logging
Nov 27 12:55:57 raspberrypi asterisk[2678]: [Nov 27 12:55:57] NOTICE[9648]: sorcery.c:1348 sorcery_object_load: Type 'system' is not
```

Figure 1: Snapshot of the Asterisk source code size.

2. Using either a SIP phone plugged into the same LAN as the Raspberry Pi Model 3 (you may need an Ethernet switch to create the network), or with a PC running a softphone application connected to the Raspberry Pi Model 3, configure Asterisk to provide a voicemail message at extension 100. Configure your SIP phone or softphone and register with Asterisk. Show your Asterisk setup in a screenshot.
   Zoiper was installed on the phone to place the call. To set up the account on Zoiper, 6001 was entered for the account name with the IP address of the Raspberry pi, which was 10.1.1.227 in this case.

```
[from-internal]
exten = 100,1,Answer()
same = n,Wait(3)
same = n,Playback(hello-world)
same = n,Hangup()
```

Figure 2: Snapshot of the extensions.conf file

```
[general]
context=default

[6001]
type=friend
context=from-internal
host=dynamic
secret=password
disallow=all
allow=ulaw
```

Figure 3: Snapshot of the sip.conf file

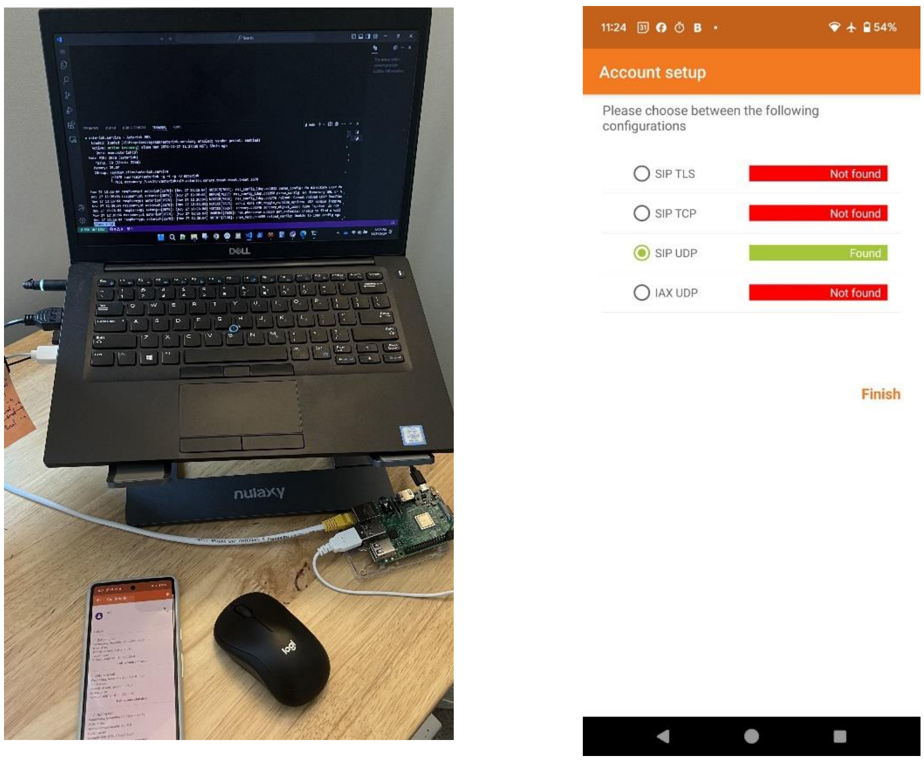

Figure 4: Asterisk setup snapshot.

3. Make a call to extension 100 and record what you hear. Show your Asterisk setup in a screenshot.
   When the phone dials extension 100, Asterisk plays a sound file "Hello World" to the channel and then hangs up. The CLI of Asterisk during the call is shown in Figure 5.

```
PROBLEMS    OUTPUT    DEBUG CONSOLE    TERMINAL    PORTS

pi@raspberrypi:/etc/asterisk $ sudo asterisk -rvvvvv
Asterisk 16.28.0~dfsg-0+deb10u4, Copyright (C) 1999 - 2021, 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 16.28.0~dfsg-0+deb10u4 currently running on raspberrypi (pid = 2678)
  == Using SIP RTP CoS mark 5
       > 0x74015c28 -- Strict RTP learning after remote address set to: 10.1.1.203:36433
    -- Executing [100@from-internal:1] Answer("SIP/6001-00000006", "") in new stack
       > 0x74015c28 -- Strict RTP switching to RTP target address 10.1.1.203:36433 as source
    -- Executing [100@from-internal:2] Wait("SIP/6001-00000006", "3") in new stack
    -- Executing [100@from-internal:3] Playback("SIP/6001-00000006", "hello-world") in new stack
    -- <SIP/6001-00000006> Playing 'hello-world.gsm' (language 'en')
    -- Executing [100@from-internal:4] Hangup("SIP/6001-00000006", "") in new stack
  == Spawn extension (from-internal, 100, 4) exited non-zero on 'SIP/6001-00000006'
raspberrypi*CLI>
```

Figure 5: Snapshot of the Asterisk CLI during the call.

4. Add another SIP phone or softphone to the network and make a phone-to-phone call. To make a phone-to-phone call Zoiper was installed on a laptop as shown in the setup snapshots below.

```
[from-internal]
exten = 100,1,Answer()
same = n,Wait(3)
same = n,Playback(hello-world)
same = n,Hangup()

exten => 6001,1,Dial(SIP/6001)
exten => 6002,1,Dial(SIP/6002)

~
~
~
~
~
~
~
~
~
"extensions.conf" 9L, 169C
```

Figure 6: Snapshot of the extensions.conf file for a second phone.

```
[general]
context=default

[6001]
type=friend
context=from-internal
host=dynamic
secret=password
disallow=all
allow=ulaw


[6002]
type=friend
context=from-internal
host=dynamic
secret=password
disallow=all
```

Figure 7: Snapshot of the sip.conf file with a second user added.

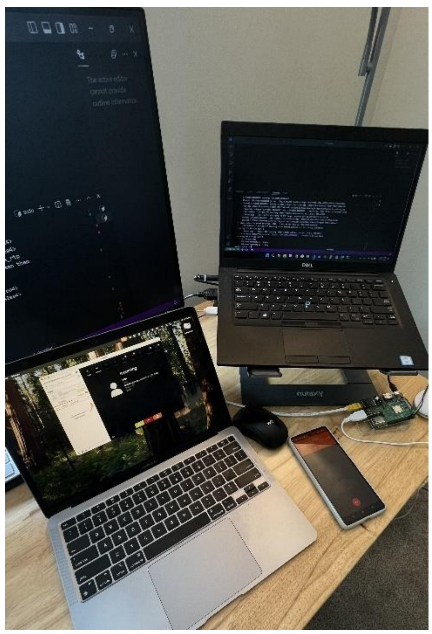

Figure 8: Snapshot of the setup to make a phone-to-phone call.

```
PROBLEMS    OUTPUT    DEBUG CONSOLE    TERMINAL    PORTS

  == Using SIP RTP CoS mark 5
       > 0x74015c28 -- Strict RTP learning after remote address set to: 10.1.1.203:34393
    -- Executing [6002@from-internal:1] Dial("SIP/6001-00000007", "SIP/6002") in new stack
  == Using SIP RTP CoS mark 5
    -- Called SIP/6002
    -- SIP/6002-00000008 is ringing
       > 0x73d085a0 -- Strict RTP learning after remote address set to: 10.1.1.239:49955
    -- SIP/6002-00000008 answered SIP/6001-00000007
    -- Channel SIP/6002-00000008 joined 'simple_bridge' basic-bridge <445effe9-af4e-4453-bb00-cf16fca42ea4>
    -- Channel SIP/6001-00000007 joined 'simple_bridge' basic-bridge <445effe9-af4e-4453-bb00-cf16fca42ea4>
       > Bridge 445effe9-af4e-4453-bb00-cf16fca42ea4: switching from simple_bridge technology to native_rtp
       > Remotely bridged 'SIP/6001-00000007' and 'SIP/6002-00000008' - media will flow directly between them
       > 0x73d085a0 -- Strict RTP learning after remote address set to: 10.1.1.239:49955
       > 0x74015c28 -- Strict RTP switching to RTP target address 10.1.1.203:34393 as source
    -- Channel SIP/6001-00000007 left 'native_rtp' basic-bridge <445effe9-af4e-4453-bb00-cf16fca42ea4>
    -- Channel SIP/6002-00000008 left 'native_rtp' basic-bridge <445effe9-af4e-4453-bb00-cf16fca42ea4>
```

Figure 9: Snapshot of the Asterisk CLI during the phone-to-phone call.

References:

[1] https://github.com/asterisk/documentation/blob/main/docs/Getting-Started/Hello-World.md