Asterisk VOIP Server Setup

Sl. No.	Document Name	Version	Authors
1.	Asterisk VOIP Server Setup with Client	0.3	Athira Vinod, Vaishnavi G

Objective

I	Installing VOIP Server: Asterisk.
II	Desktop Client to Desktop Client Call.
III	Integration with Magma
IV	Mobile1 4G register to Server.
V	Mobile1 4G attach and Mobile2 4G attach Call.

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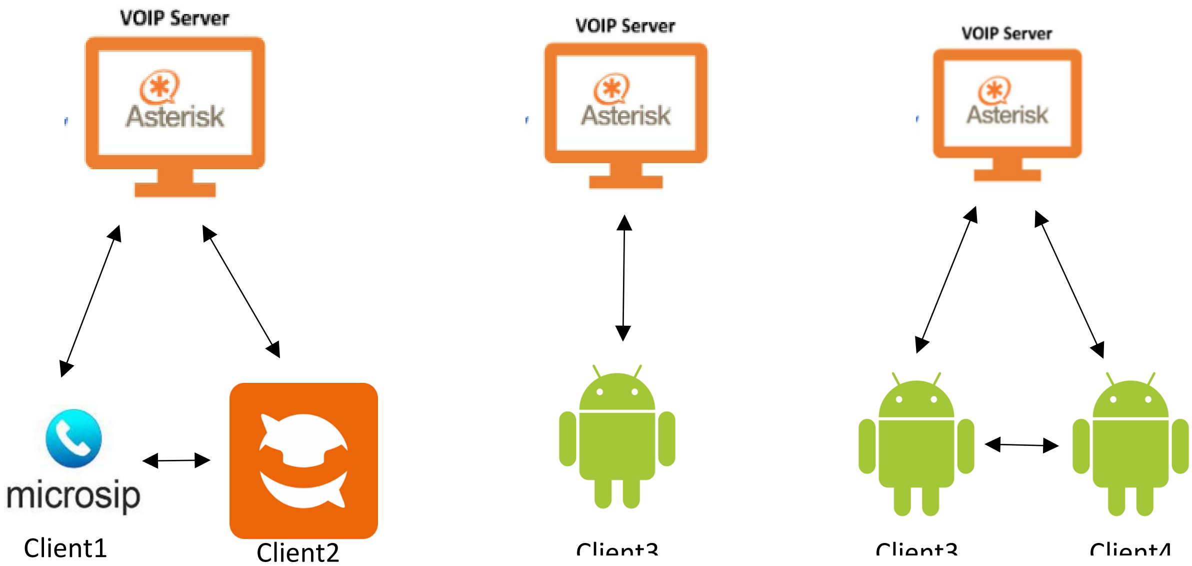
Architecture



Once these clients are connected to the server, they can initiate to any other client in the network.

I. Installing Asterisk

1. Updating the system

   sudo apt-get update

2. Install Asterisk

   sudo apt-get install asterisk –y

```
Adding system user for Asterisk
Adding user `asterisk' to group `dialout' ...
Adding user asterisk to group dialout
Done.
Adding user `asterisk' to group `audio' ...
Adding user asterisk to group audio
Done.
Created symlink /etc/systemd/system/multi-user.target.wants/asterisk.service → /lib/systemd/system/asterisk.service.
Setting up sox (14.4.2+git20190427-2+deb11u2build0.20.04.1) ...
Setting up asterisk-voicemail (1:16.2.1~dfsg-2ubuntu1) ...
Processing triggers for libc-bin (2.31-0ubuntu9.9) ...
Processing triggers for systemd (245.4-4ubuntu3.21) ...
Processing triggers for man-db (2.9.1-1) ...
Processing triggers for mime-support (3.64ubuntu1) ...
```

   The end output will be like the above snip.

3. Launch Asterisk

sudo asterisk –r

```
athira@compute1:~/sipserver$ sudo asterisk -r
Asterisk 16.2.1~dfsg-2ubuntu1, Copyright (C) 1999 - 2018, Digium, Inc. 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 16.2.1~dfsg-2ubuntu1 currently running on compute1 (pid = 1362648)
compute1*CLI>
compute1*CLI>
compute1*CLI>
```

II. <u>Desktop Client to Desktop Client Call</u>

1. Edit the configuration files in **/etc/asterisk.**

   NB: Before editing, take backup of these files below.

```
cdr_manager.conf          codecs.conf               iax.conf                  pjsip.conf
athira@compute1:/etc/asterisk$ sudo mv sip.conf sip.conf.backup
athira@compute1:/etc/asterisk$ sudo mv extensions.conf extensions.conf.backup
athira@compute1:/etc/asterisk$ sudo mv voicemail.conf voicemail.conf.backup
```

2. Configure the files: https://github.com/mailrocketsystems/AsteriskVOIP
3. Launch the Asterisk (Step 3)
4. Reload Asterisk

```
compute1*CLI> reload
[Jun 19 12:44:01] NOTICE[1367239]: res_config_ldap.c:1830 parse_config: No directory user found, anonymous binding as default.
[Jun 19 12:44:01] ERROR[1367239]: res_config_ldap.c:1856 parse_config: No directory URL or host found.
[Jun 19 12:44:01] NOTICE[1367239]: res_config_ldap.c:1774 reload: Cannot reload LDAP RealTime driver.
[Jun 19 12:44:01] NOTICE[1367239]: cdr.c:4485 cdr_toggle_runtime_options: CDR simple logging enabled.
[Jun 19 12:44:01] NOTICE[1367460]: sorcery.c:1266 sorcery_object_load: Type 'system' is not reloadable, maintaining previous values
[Jun 19 12:44:01] WARNING[1367239]: res_phoneprov.c:1230 get_defaults: Unable to find a valid server address or name.
[Jun 19 12:44:01] NOTICE[1362887]: chan_mgcp.c:4707 reload_config: Unable to load config mgcp.conf, MGCP disabled
[Jun 19 12:44:01] ERROR[1367239]: ari/config.c:312 process_config: No configured users for ARI
[Jun 19 12:44:01] NOTICE[1367239]: cel_custom.c:95 load_config: No mappings found in cel_custom.conf. Not logging CEL to custom CSVs.
[Jun 19 12:44:01] NOTICE[1367239]: app_queue.c:9096 reload_queue_rules: queuerules.conf has not changed since it was last loaded. Not tak
ing any action.
```

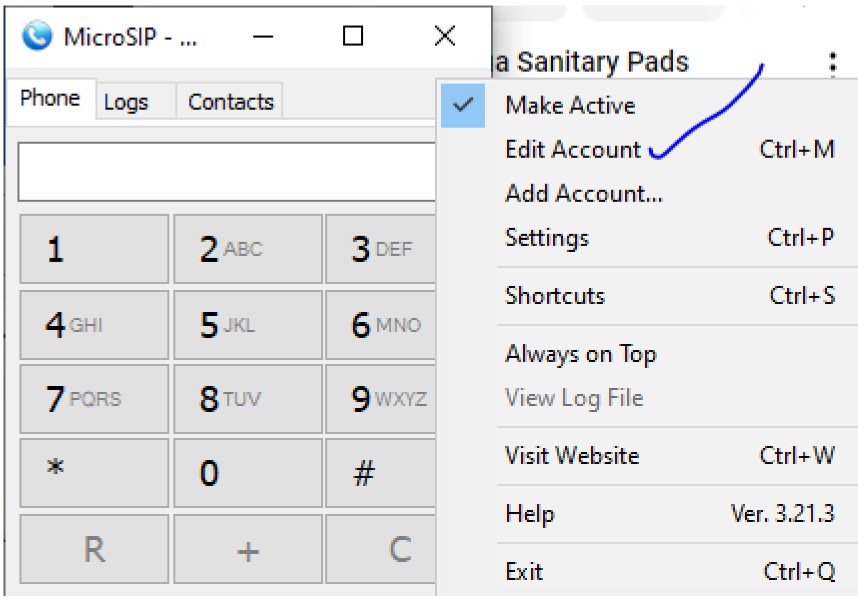
5. Check whether clients are added or not

   sip show peers

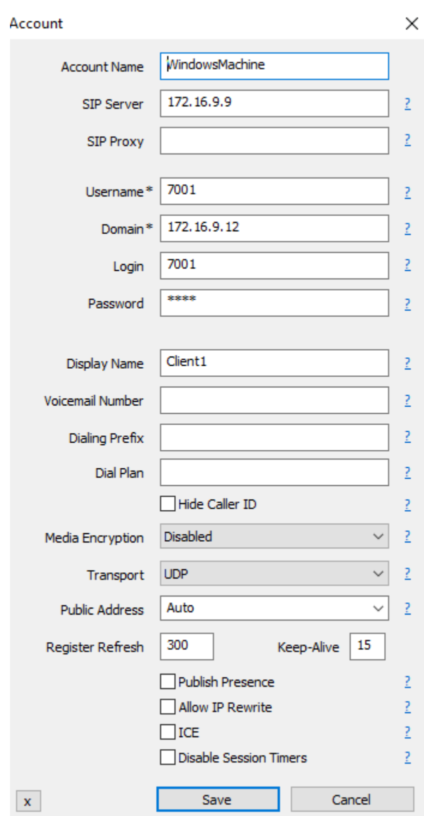
```
compute1*CLI> sip show peers
Name/username             Host                                    Dyn Forcerport Comedia    ACL Port     Status      Description
7001                      (Unspecified)                            D  Yes        Yes            0        Unmonitored
7002                      (Unspecified)                            D  Yes        Yes            0        Unmonitored
2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 0 online, 2 offline]
```

   We added two clients, and both are in offline state now.

6. Download the Windows Client: Here we used **MicroSIP** https://www.microsip.org/downloads.
7. Open MicroSip and Edit Account.



8. Give details according to your ***sip.conf*** in server and save.



9. Then again check client state.

sip show peers

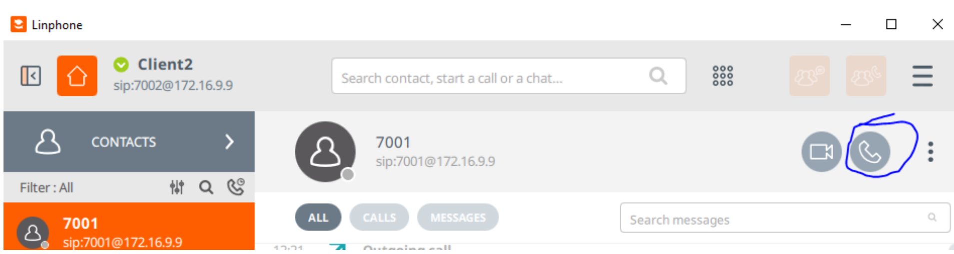
In the picture below, you can see the difference, our windows Client is connected with the IP, which is our windows machine, and it shows one offline and one online.

```
Name/username             Host                                    Dyn Forcerport Comedia    ACL Port     Status      Description
7001                      (Unspecified)                            D  Yes        Yes            0        Unmonitored
7002                      (Unspecified)                            D  Yes        Yes            0        Unmonitored
2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 0 online, 2 offline]
compute1*CLI> sip show peers
Name/username             Host                                    Dyn Forcerport Comedia    ACL Port     Status      Description
7001/7001                 10.9.9.175                               D  Yes        Yes            61494    Unmonitored
7002                      (Unspecified)                            D  Yes        Yes            0        Unmonitored
2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 1 online, 1 offline]
compute1*CLI> sip show peers
```

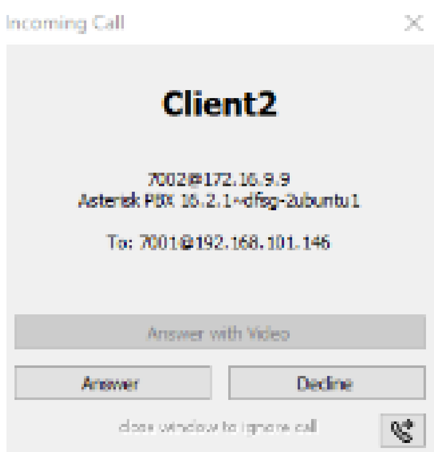
10. For Second Client, install **Linphone :** https://www.linphone.org/
11. Give Account details, as we did for the MicrSip Client.
12. Set the status Active.
13. Check the client state again (Step 9)

```
2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 1 online, 1 offline]
compute1*CLI> sip show peers
Name/username             Host                                    Dyn Forcerport Comedia    ACL Port     Status      Description
7001/7001                 10.9.8.250                               D  Yes        Yes            54733    Unmonitored
7002/7002                 10.9.8.250                               D  Yes        Yes            5060     Unmonitored
2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 2 online, 0 offline]
```

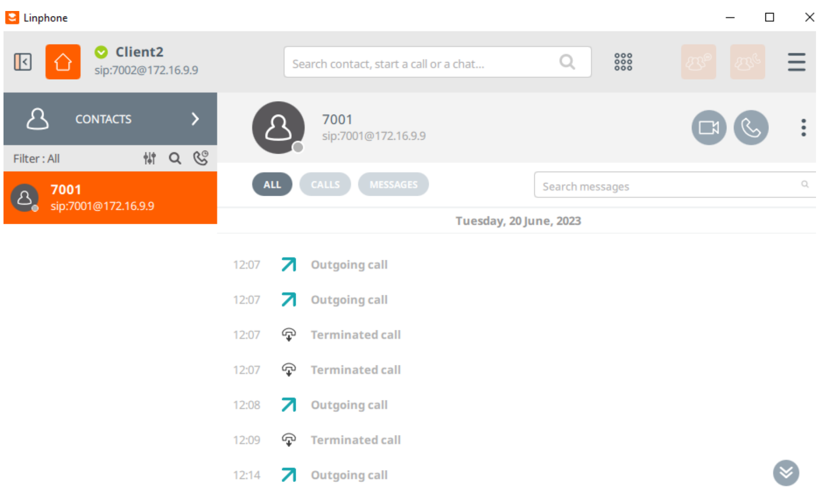
14. Make the call.



15. You will be getting calls from the other client via ringtone.



16. You can find call logs in both the clients.



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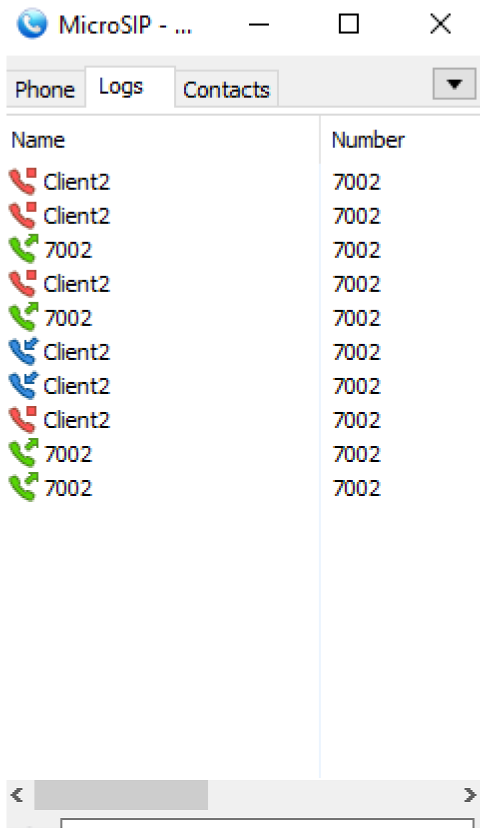


17. You can watch the pcap traces.

    sudo tcpdump -i eno1 -s 0 -w output_file_online.pcap

III. Integration with Magma

For integration of Asterisk Server with Magma, first we need to make sure these both will be under same network.

Ref: https://kwlug.org/node/812

We need to make changes in *sip.conf* . While the other files like *extension.conf* and *voicemail.conf* remain the same as we did for communication between windows clients.

```
[general]
context=internal
externip=172.16.5.18        a
localnet=172.16.5.31/255.255.255.0        b
srvlookup=yes
qualify=no

;[magma]
;type=friend
;host=magma
;username=testclient
;secret=pwd123
;context=default

[7001]
type=friend
secret=7001
;qualify=200                  ; Qualify peer is no more than 200ms away
host=dynamic                 ; This device registers with us
context=internal                 ; This device registers with us

[7003]
type=friend
secret=7002
qualify=200                  ; Qualify peer is no more than 200ms away
host=dynamic                 ; This device registers with us
context=default                 ; This device registers with us

[7002]
type=friend
secret=7002
;qualify=200                  ; Qualify peer is no more than 200ms away
host=dynamic                 ; This device registers with us
context=internal                 ; This device registers with us
"sip.conf" 1654L, 96023C
```

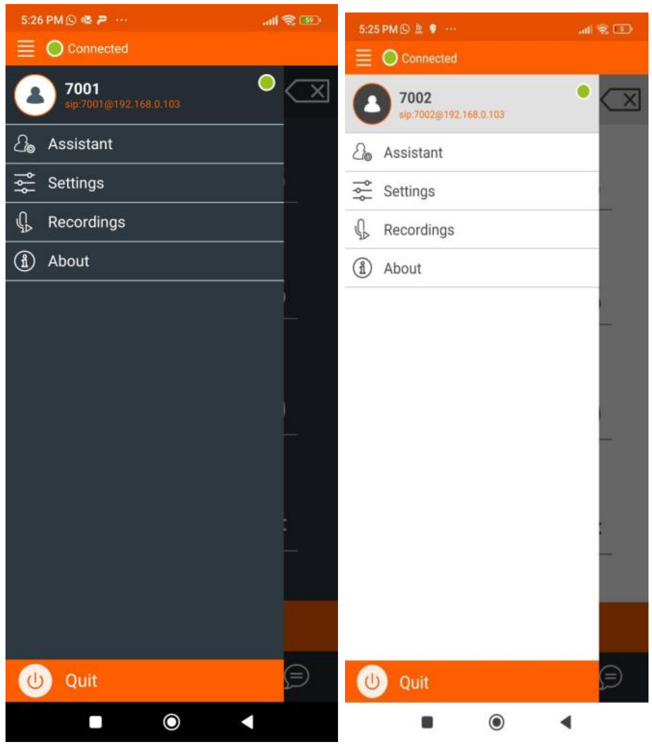
a. Magma IP
b. SIP Server IP

After configuring, please restart your asterisk server and check the status of clients added.

IV. Mobile1 4G register to Server

Like windows client configuration in *sip.conf*, two clients will be considered as android phones where sip client apps are downloaded. In this case, **Linphone** is taken as client.

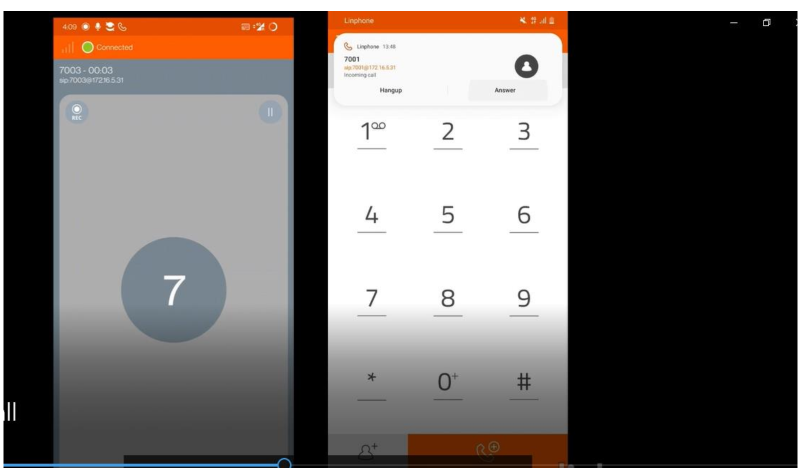
Give the account details like Step 11 in Section II and the app will be active on two android phones (4G). Now the setup is ready for call.



V. Mobile1 4G attach and Mobile2 4G attach Call

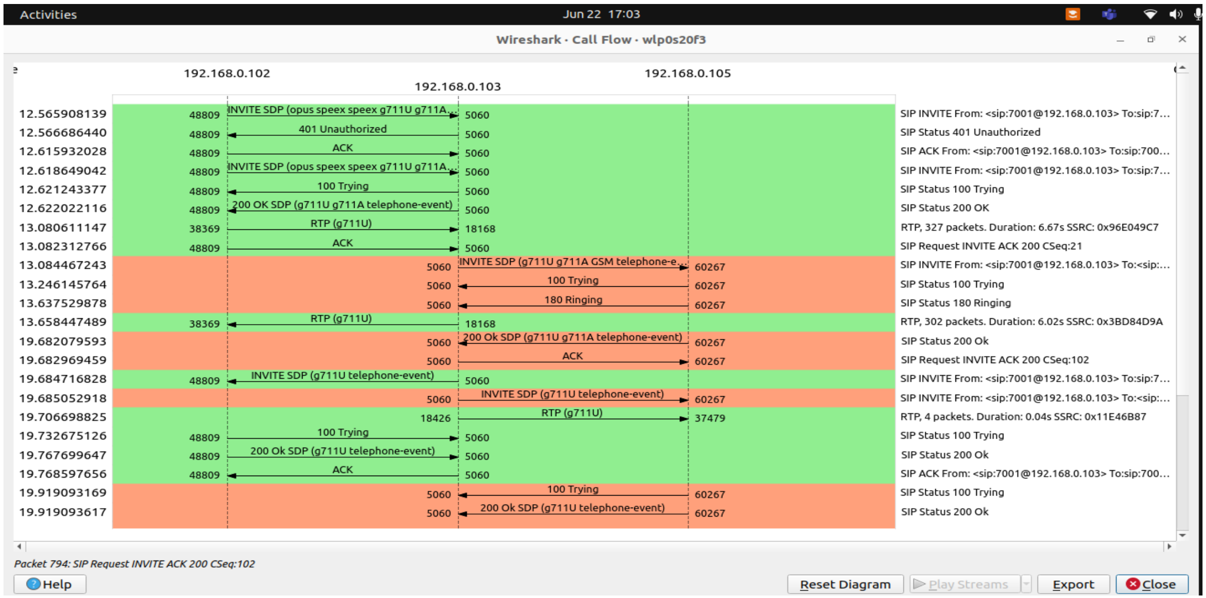
Here two 4G android phones are considered. In *sip.conf*, the the linphone clients installed on these phones are configured as 7003 and 7001. You can give username and credentials according to your requirements.

In the snippet below, you can see the call is coming from 7003 to 7001.



After the call is made from one phone client to another, we can get the information from pcap traces. It will show the connection establishment below.

As Magma is integrated, when the call is established, we can monitor the logs from Magma side also.



Reference:

1. [Asterisk_SIPCall.mp4]
2. [magma_logs_sip.txt]
3. https://wavelabstechnologies-my.sharepoint.com/:v:/g/personal/athira_vinod_wavelabs_ai/EdrCNecDK7ZGlD_kN3Jbri8Bz8E9zXba4x1-vrOk01JVCQ?e=hvbYD8

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