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VoIP

Voice over Internet Protocol, also called IP telephony, is a method and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol networks, such as the Internet.



Related Fields

- PBX
- Asterisk Server

Resources Required

Asterisk Server (<https://www.asterisk.org/downloads/>)

MicroSip (<https://www.microsip.org/downloads>)

MizuDroid (<https://playstore.google.com>)

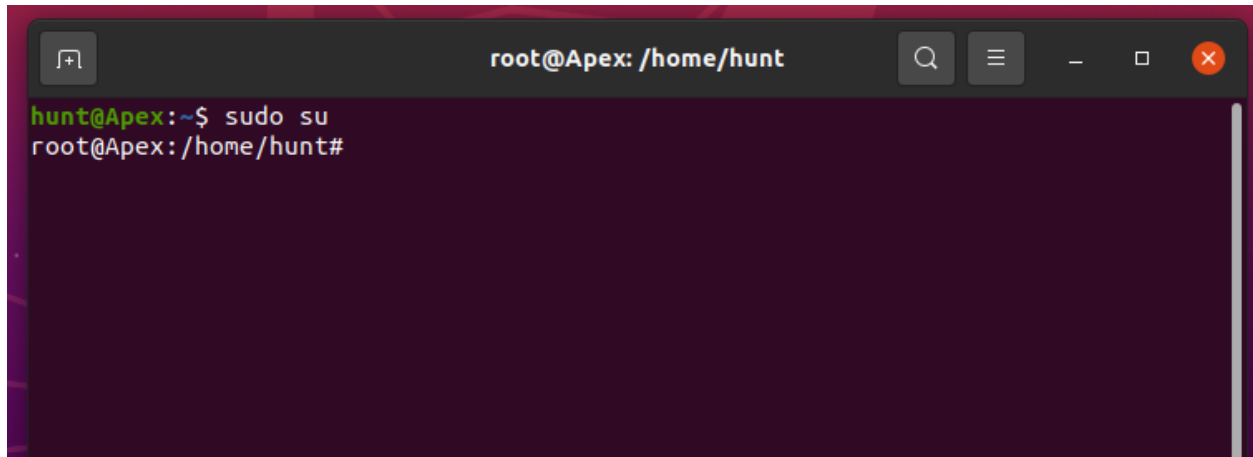
Configuration files (<https://github.com/mailrocketsystems/AsteriskVOIP>)

Configuration Step-by-Step:

To install or configure the server, we've used the ubuntu virtual machine just to make it as a practice.

If you don't have the ubuntu you can also use any other OS, but ubuntu is highly recommended.

Step 1: Open your root terminal in your Ubuntu

A screenshot of a terminal window with a dark background. The window title bar shows 'root@Apex: /home/hunt'. The terminal content shows a user prompt 'hunt@Apex:~\$' followed by the command 'sudo su'. The next line shows the root prompt 'root@Apex:/home/hunt#'.

```
root@Apex: /home/hunt
hunt@Apex:~$ sudo su
root@Apex:/home/hunt#
```

Enter the root can also ask for the password you've configured for your machine. If not then there is not much problem.

Step 2: Install Asterisk Server

Command used for the install of Asterisk Server in ubuntu: **"apt install asterisk -y"**

The '-y' at the end of the command represents that we're completely agree with installation of this package. So, we won't receive any further commands of package acceptance.

```
root@Apex: /home/hunt
hunt@Apex:~$ sudo su
root@Apex:/home/hunt# apt install asterisk -y
Reading package lists... done
Building dependency tree
Reading state information... Done
The following additional packages will be installed:
  asterisk-voicemail
Suggested packages:
  asterisk-dahdi asterisk-dev asterisk-doc asterisk-ooh323 asterisk-opus
  asterisk-vpb
The following NEW packages will be installed:
  asterisk asterisk-voicemail
0 upgraded, 2 newly installed, 0 to remove and 195 not upgraded.
Need to get 0 B/2,248 kB of archives.
After this operation, 7,778 kB of additional disk space will be used.
Selecting previously unselected package asterisk.
(Reading database ... 185022 files and directories currently installed.)
Preparing to unpack .../asterisk_1%3a16.2.1~dfsg-2ubuntu1_amd64.deb ...
Unpacking asterisk (1:16.2.1~dfsg-2ubuntu1) ...
Selecting previously unselected package asterisk-voicemail.
Preparing to unpack .../asterisk-voicemail_1%3a16.2.1~dfsg-2ubuntu1_amd64.deb ..
.
Unpacking asterisk-voicemail (1:16.2.1~dfsg-2ubuntu1) ...
Setting up asterisk (1:16.2.1~dfsg-2ubuntu1) ...
.
Unpacking asterisk-voicemail (1:16.2.1~dfsg-2ubuntu1) ...
Setting up asterisk (1:16.2.1~dfsg-2ubuntu1) ...
Setting up asterisk-voicemail (1:16.2.1~dfsg-2ubuntu1) ...
Processing triggers for systemd (245.4-4ubuntu3.11) ...
Processing triggers for man-db (2.9.1-1) ...
Processing triggers for libc-bin (2.31-0ubuntu9.2) ...
```

This shows that 'asterisk' is completely installed.

Step 3: To Insure the installation of Asterisk Package

To check that the 'asterisk' is installed or not we use "**asterisk -r**", which gives all of the perspectives of installed asterisk server.

```
root@Apex:/home/hunt# asterisk -r
Asterisk 16.2.1~dfsg-2ubuntu1, Copyright (C) 1999 - 2018, Digium, Inc. and other
S.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for detail
S.
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 Apex (pid = 6353
7)
Apex*CLI> S
```

After checking you can exit by using 'exit' command.

Step 4: Entering the 'asterisk' server

For this, we can use the command "cd /etc/asterisk"

```
root@Apex:/home/hunt# cd /etc/asterisk/
root@Apex:/etc/asterisk#
```

Now, we've got ourselves within the directory of installed asterisk package.

Step 5: Checking the main files we're going to configure

- sip.conf
- extensions.conf
- voicemail.conf

```
root@Apex: /etc/asterisk

ast_debug_tools.conf    extensions.ael          res_corosync.conf
asterisk.adsi           extensions.conf          res_curl.conf
asterisk.conf           extensions.lua           res_fax.conf
calendar.conf          extensions_minivm.conf  res_ldap.conf
ccss.conf              features.conf           res_ldap.conf
cdr_adaptive_odbc.conf  festival.conf           res_odbc.conf
cdr_beanstalkd.conf    followme.conf           resolver_unbound.conf
cdr.conf               func_odbc.conf          res_parking.conf
cdr_custom.conf        hep.conf                res_pgsql.conf
cdr_manager.conf       http.conf               res_pktccops.conf
cdr_mysql.conf         iax.conf                res_snmp.conf
cdr_odbc.conf          iaxprov.conf            res_stun_monitor.conf
cdr_pgsql.conf         indications.conf       rtp.conf
cdr_sqlite3_custom.conf logger.conf              say.conf
cdr_syslog.conf        manager.conf            sip.conf backup
cdr_tds.conf           manager.d               sip_notify.conf
cel_beanstalkd.conf    meetme.conf             sla.conf
cel.conf              minivm.conf             smdi.conf
cel_custom.conf        misd.conf               sorcery.conf
cel_odbc.conf          modules.conf            ss7.timers
cel_pgsql.conf         motif.conf              stasis.conf
cel_sqlite3_custom.conf musiconhold.conf        statsd.conf
cel_tds.conf           muted.conf              telcordia-1.adsi
chan_dahdi.conf        ooh323.conf             test_sorcery.conf
chan_mobile.conf       osp.conf                udptl.conf
cli_aliases.conf       oss.conf                users.conf
cli.conf              phone.conf              voicemail.conf
cli_permissions.conf   phoneprov.conf          vpb.conf
root@Apex: /etc/asterisk# xmpp.conf
```

All of the three files are marked with the red marker.

(Optional) Step 6: If you want to study about the installed server

There is file within the green marker, you can access it to view or study the installed package named 'asterisk'.

```
root@Apex: /etc/asterisk
ast_debug_tools.conf extensions.ael res_corosync.conf
asterisk.adsi extensions.conf res_curl.conf
asterisk.conf extensions.lua res_fax.conf
calendar.conf extensions_minim.conf res_ldap.conf
ccss.conf features.conf res_odbc.conf
cdr_adaptive_odbc.conf festival.conf resolver_unbound.conf
cdr_beanstalkd.conf followme.conf res_parking.conf
cdr.conf func_odbc.conf res_pgsql.conf
cdr_custom.conf hep.conf res_pktccops.conf
cdr_manager.conf http.conf res_snmp.conf
cdr_mysql.conf iax.conf res_stun_monitor.conf
cdr_odbc.conf iaxprov.conf rtp.conf
cdr_pgsql.conf indications.conf say.conf
cdr_sqlite3_custom.conf logger.conf sip.conf backup
cdr_syslog.conf manager.conf sip_notify.conf
cdr_tds.conf manager.d sla.conf
cel_beanstalkd.conf meetme.conf smdi.conf
cel.conf minim.conf sorcery.conf
cel_custom.conf misdnc.conf ss7.timers
cel_odbc.conf modules.conf stasis.conf
cel_pgsql.conf motif.conf statsd.conf
cel_sqlite3_custom.conf musiconhold.conf telcordia-1.adsi
cel_tds.conf muted.conf test_sorcery.conf
chan_dahdi.conf ooh323.conf udptl.conf
chan_mobile.conf osp.conf users.conf
cli_aliases.conf oss.conf voicemail.conf
cli.conf phone.conf vpb.conf
cli_permissions.conf phoneprov.conf xmpp.conf
root@Apex:/etc/asterisk#
```

Step 7: Creating Backup file for “sip.conf”

```
hunt@Apex:/etc/asterisk$ sudo mv sip.conf sip.conf.backup
hunt@Apex:/etc/asterisk$
```

To create a backup file for sip.conf in order some inconvenience. The command we use is “mv sip.conf.backup”

Step 8: Updating Commands in “sip.conf”

To access the file in editing mode we use “sudo gedit sip.conf”

```
hunt@Apex:/etc/asterisk$ sudo gedit sip.conf
(gedit:68319): Tepl-WARNING **: 23:30:11.310: GVfs metadata is not supported. Fallback to TeplMetadataManager. Either G
Vfs is not correctly installed or GVfs metadata are not supported on this platform. In the latter case, you should conf
igure Tepl with --disable-gvfs-metadata.
```

A pop will appear which can be then edit according to your own requirements. As shown:

```

1 [internal]
2 exten => 7001,1,Answer()
3 exten => 7001,2,Dial(SIP/7001,60)
4 exten => 7001,3,Playback(vm-nobodyavail)
5 exten => 7001,4,VoiceMail(7001@main)
6 exten => 7001,5,Hangup()
7
8 exten => 7002,1,Answer()
9 exten => 7002,2,Dial(SIP/7002,60)
10 exten => 7002,3,Playback(vm-nobodyavail)
11 exten => 7002,4,VoiceMail(7001@main)
12 exten => 7002,5,Hangup()
13
14 exten => 8001,1,VoiceMailMain(7001@main)
15 exten => 8001,2,Hangup()
16
17 exten => 8002,1,VoiceMailMain(7002@main)
18 exten => 8002,2,Hangup()
19

```

After changes save the document.

Step 9: Creating Backup file for “extensions.conf”

```

hunt@Apex:/etc/asterisk$ sudo mv extensions.conf extensions.conf.backup
hunt@Apex:/etc/asterisk$

```

To create a backup file for extension.conf in order some inconvenience. The command we use is “**mv extensions.conf extensions.conf.backup**”

Step 10: Updating commands in “extensions.conf”

To access the file in editing mode we use “**sudo gedit extensions.conf**”

```

hunt@Apex:/etc/asterisk$ sudo gedit extensions.conf
(gedit:68393): Tepl-WARNING **: 23:33:13.982: GVfs metadata is not supported. Fallback to TeplMetadataManager. Either G
Vfs is not correctly installed or GVfs metadata are not supported on this platform. In the latter case, you should conf
igure Tepl with --disable-gvfs-metadata.
hunt@Apex:/etc/asterisk$

```

A pop will appear which can be then edit according to your own requirements. As shown:

```
1 [general]
2 context=internal
3 allowguest=no
4 allowoverlap=no
5 bindport=5060
6 bindaddr=0.0.0.0
7 srlookup=no
8 disallow=all
9 allow=ulaw
10 alwaysauthreject=yes
11 canreinvite=no
12 nat=yes
13 session-timers=refuse
14 localnet=192.168.0.0/255.255.255.0
15
16 [7001]
17 type=friend
18 host=dynamic
19 secret=7001
20 context=internal
21
22 [7002]
23 type=friend
24 host=dynamic
25 secret=7002
26 context=internal
27
```

Save the document after updating the files.

Step 11: Creating Backup file for “voicemail.conf”

```
hunt@Apex:/etc/asterisk$ sudo mv voicemail.conf voicemail.conf.backup
hunt@Apex:/etc/asterisk$
```

Backup file created and we can use it to recover the data in any sort of inconvenience within the file.

To create a backup file for voicemail.conf in order some inconvenience. The command we use is “**mv voicemail.conf voicemail.conf.backup**”

Step 12: Updating commands in “extensions.conf”


```

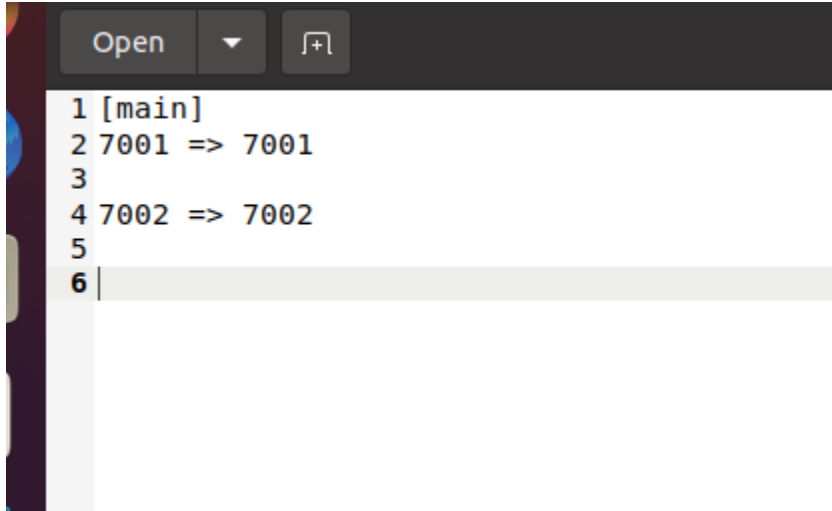
hunt@Apex:/etc/asterisk$ sudo gedit voicemail.conf

(gedit:68407): Tepl-WARNING **: 23:37:50.368: GVfs metadata is not supported. Fallback to TeplMetadataManager. Either G
Vfs is not correctly installed or GVfs metadata are not supported on this platform. In the latter case, you should conf
igure Tepl with --disable-gvfs-metadata.
hunt@Apex:/etc/asterisk$

```

To access the file in editing mode we use “**sudo gedit voicemail.conf**”

A pop will appear which can be then edit according to your own requirements. As shown:



Save the file after upgrading the data.

Step 13: Now, Accessing the server

Command used: “**sudo asterisk -r**”

```

hunt@Apex:/etc/asterisk$ 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 Apex (pid = 66780)
Apex*CLI>

```

Server will access the command line interface as CLI.

Step 14: Checking the status of peers

Command used: “**sip show peers**” within the directory of asterisk server.

```
le logging enabled.
[Aug 24 04:14:32] NOTICE[10721]: sorcery.c:1266 sorcery_object_load: Type 'system' is not reloadable, maintaining previous values
[Aug 24 04:14:32] WARNING[10720]: res_phoneprov.c:1230 get_defaults: Unable to find a valid server address or name.
[Aug 24 04:14:32] ERROR[10720]: ari/config.c:312 process_config: No configured users for ARI
[Aug 24 04:14:32] NOTICE[10720]: cel_custom.c:95 load_config: No mappings found in cel_custom.conf. Not logging CEL to custom CSVs.
[Aug 24 04:14:32] NOTICE[10720]: app_queue.c:9096 reload_queue_rules: queuerules.conf has not changed since it was last loaded. Not taking any action.
[Aug 24 04:14:32] WARNING[8927]: sip/config_parser.c:817 sip_parse_nat_option: nat=yes is deprecated, use nat=force_rport,comedia instead
[Aug 24 04:14:32] NOTICE[8928]: chan_mgcp.c:4707 reload_config: Unable to load config mgcp.conf, MGCP disabled
ubuntu*CLI> sip show peers
Name/username      Host              Dyn Forcerport
Comedia            ACL Port         Status            Description
7001                0                 (Unspecified)     D Yes
Yes                 0                 Unmonitored
7002                0                 (Unspecified)     D Yes
Yes                 0                 Unmonitored
2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 0 online, 2 offline]
ubuntu*CLI>
```

After entering the commands We've seen the available users.

So, we have configure only on two of the users.

Step 15: if the server Didn't show the configured users.

If the configure users didn't appear in the server, then you must have to reload the server. So, the users appear in a normal way.

```

2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 0 online, 2 offline]
ubuntu*CLI> reload
[Aug 24 04:20:46] NOTICE[10720]: res_config_ldap.c:1830 parse_config: No direct
ry user found, anonymous binding as default.
[Aug 24 04:20:46] ERROR[10720]: res_config_ldap.c:1856 parse_config: No directo
y URL or host found.
[Aug 24 04:20:46] NOTICE[10720]: res_config_ldap.c:1774 reload: Cannot reload L
AP RealTime driver.
[Aug 24 04:20:46] NOTICE[10720]: cdr.c:4485 cdr_toggle_runtime_options: CDR sim
le logging enabled.
[Aug 24 04:20:46] NOTICE[10741]: sorcery.c:1266 sorcery_object_load: Type 'syst
m' is not reloadable, maintaining previous values
[Aug 24 04:20:46] WARNING[10720]: res_phoneprov.c:1230 get_defaults: Unable to
ind a valid server address or name.
[Aug 24 04:20:46] ERROR[10720]: ari/config.c:312 process_config: No configured
sers for ARI
[Aug 24 04:20:46] NOTICE[10720]: cel_custom.c:95 load_config: No mappings found
in cel_custom.conf. Not logging CEL to custom CSVs.
[Aug 24 04:20:46] NOTICE[10720]: app_queue.c:9096 reload_queue_rules: queuerule
.conf has not changed since it was last loaded. Not taking any action.
[Aug 24 04:20:46] NOTICE[8928]: chan_mgcp.c:4707 reload_config: Unable to load
onfig mgcp.conf, MGCP disabled
ubuntu*CLI>

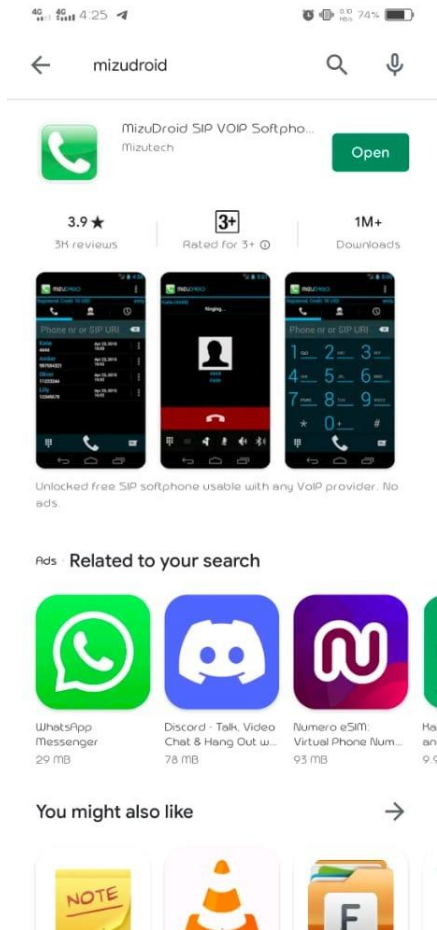
```

Configure the Users:

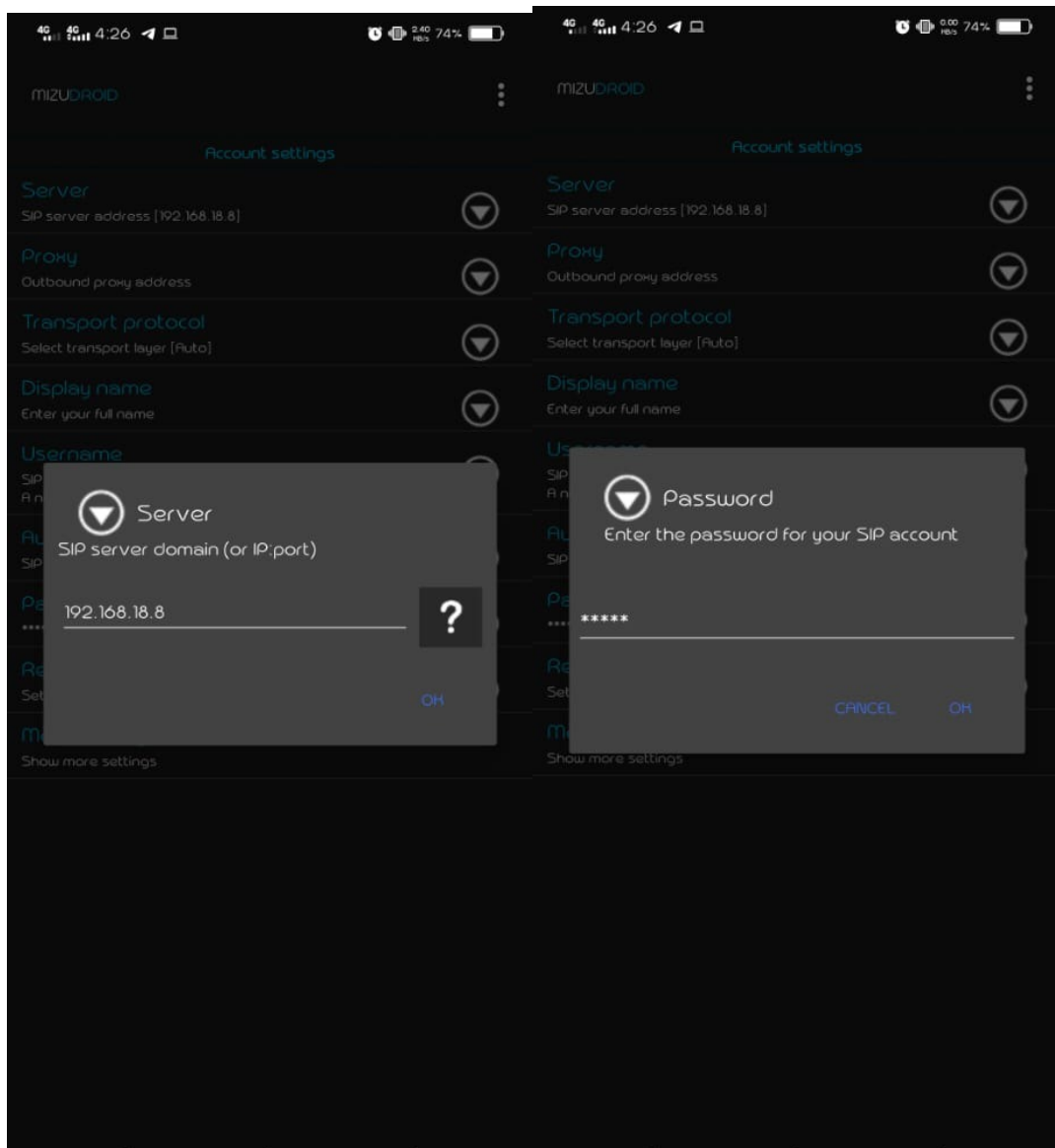
Step 16: Download any VoIP app in your mobile

You can download any VoIP app with different capabilities.

But here we've used 'MizuDroid SIP VOIP" by MizuTech from google play store



Step 17: Configure the app settings



MIZUDROID



Account settings

Server

SIP server address [192.168.18.8]



Proxy

Outbound proxy address



Transport protocol

Select transport layer [Auto]



Display name

Enter your full name



Us

SIP

A n

AU

SIP

Pa

Re

Set

M

Sho



Caller ID

SIP user name. Will be used for authentication if the Auth Username is not set. Otherwise used for AOR in the From/Contact headers. Some VoIP servers might not accept a different username and auth username.

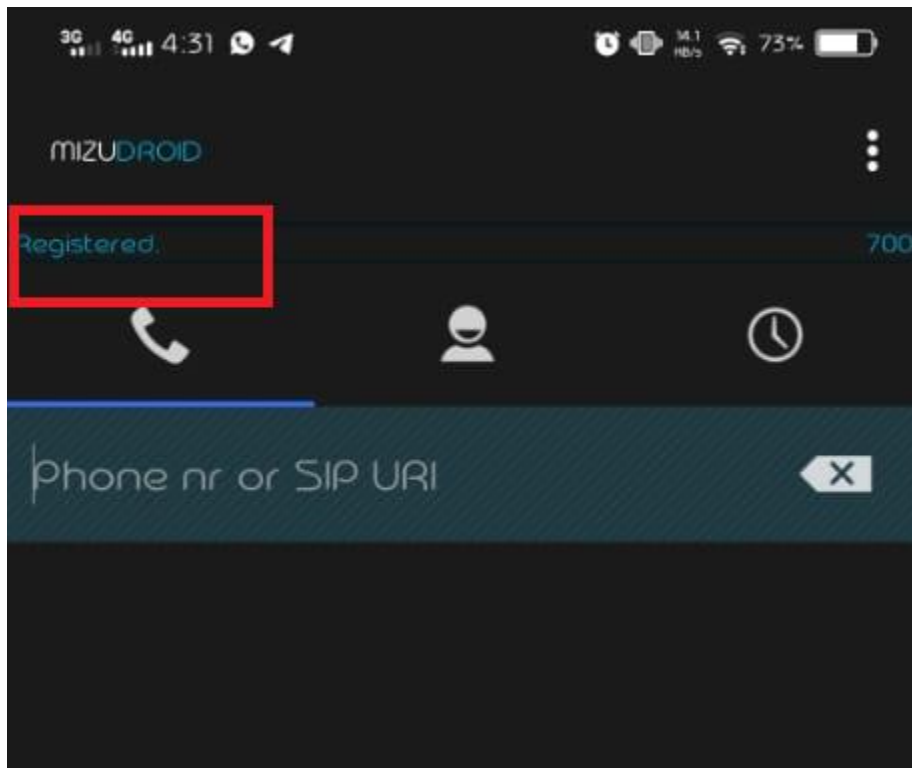
7001

CANCEL

OK

Step 18:

The first User successful added to the list



Step 19: Configure the second device

We've used a laptop for that

Download an app named MicroSip (<https://www.microsip.org/downloads>)

In this we've also to configure the domain as the network is there.

microsip.org/downloads/

microsip
Open source portable SIP softphone for Windows based on PJSIP stack

MicroSIP Home Downloads Wishes Troubleshooting FAQ Help Translate Online Source Custom Build Contact

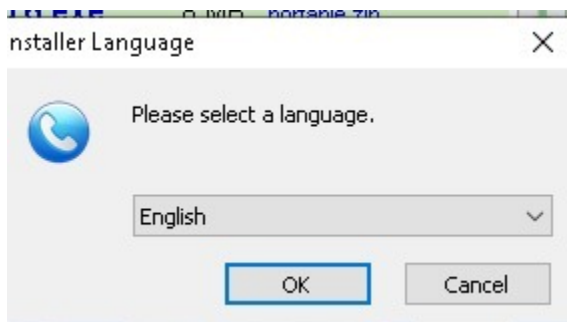
Earn up to 7,500 Qmiles and more

MicroSIP Downloads - Installer and Portable version

File	MicroSIP-3.20.6.exe	MicroSIP-Lite-3.20.6.exe
Size	8 MB	5 MB
Format	portable.zip	portable.zip
Download Count	215218	33908
Total	2,635,011	497,389
Video Support	YES	NO
Portable version	YES (see above)	YES (see above)
Unpacked size	18 MB	10 MB

Jira Software
Jira integrates with the tools you already use
Voted the #1 agile tool for teams
Start for free

Step 20: Install the related App.



Proceed the installation. So, the installation completes in a better way.

Step 21:



This shows that device is offline

Step 22: Configure the user over the laptop

Account X

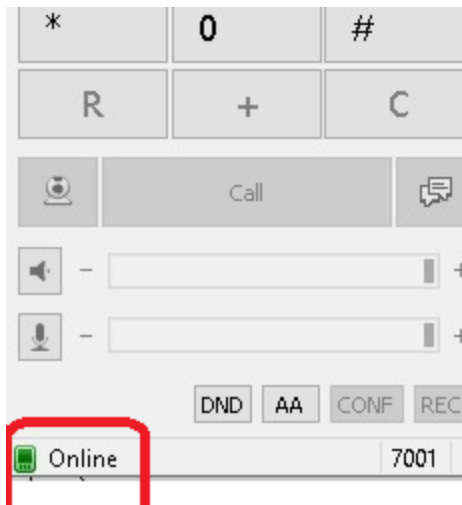
Account Name	7002	
SIP Server	192.168.18.4	?
SIP Proxy		?
Username*	7001	?
Domain*	192.168.18.0	?
Login	7002	?
Password	*****	?
Display Name	Laptop	?
Voicemail Number		?
Dialing Prefix		?
Dial Plan		?
	<input type="checkbox"/> Hide Caller ID	?
Media Encryption	Disabled	?
Transport	UDP	?
Public Address	Auto	?
Register Refresh	300	Keep-Alive 15
	<input type="checkbox"/> Publish Presence	?
	<input type="checkbox"/> Allow IP Rewrite	?
	<input type="checkbox"/> ICE	?
	<input type="checkbox"/> Disable Session Timers	?

X

Save

Cancel

Step 23: Check the availability of the device



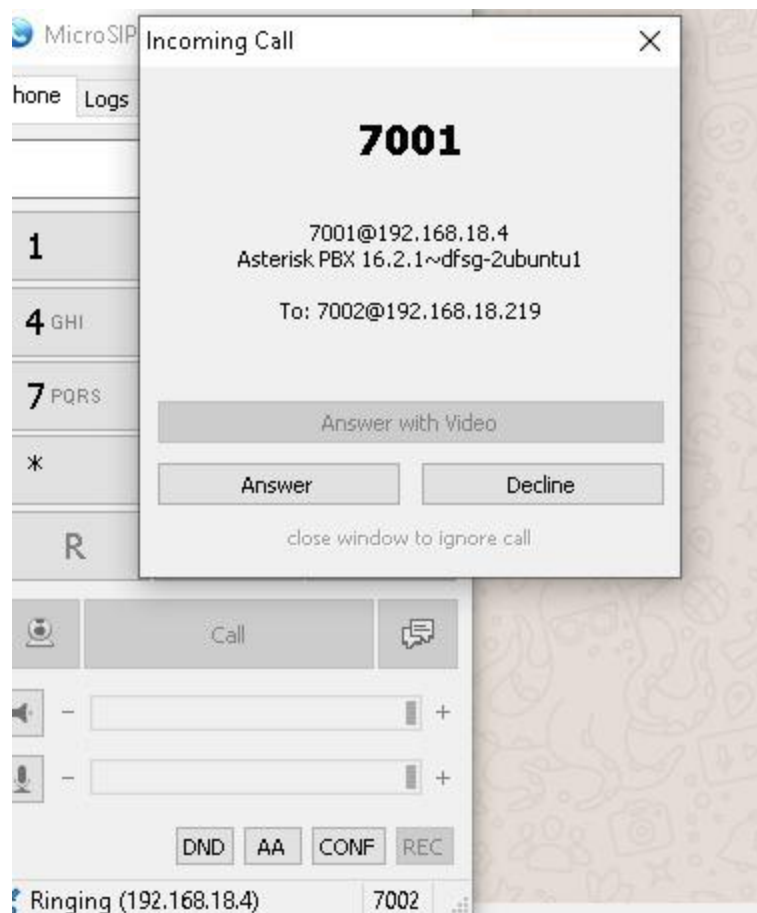
This shows that the device is completely online.

Step 24: Checking the connectivity of clients from server.

```
ubuntu*CLI> sip show peers
Name/username      Host                      Dyn Forcerport
Comedia    ACL Port    Status    Description
7001/7001           192.168.18.23           D   Yes
Yes              15476    Unmonitored
7002/7002           192.168.18.219          D   Yes
Yes              50881    Unmonitored
2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 2 online, 0 offline]
ubuntu*CLI>
```

Shows that the users are online . Now both can communicate with each other.

Conclusion



mizUDROID



7002

0.46

Speaking



7002

