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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

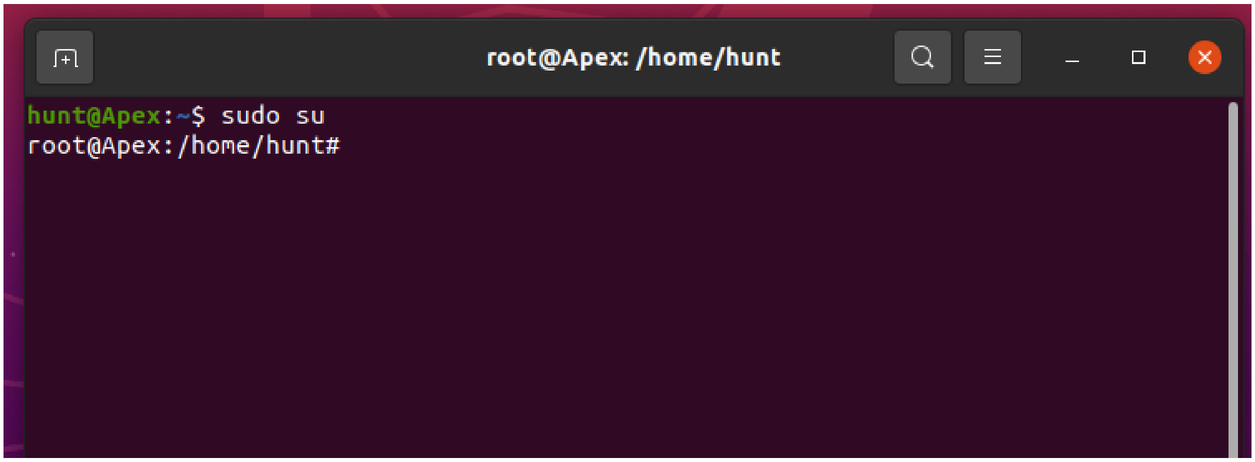
Asterisk Server (https://www.asterisk.org/downloads/)
MicroSip (https://www.microsip.org/downloads)
MizuDroid (https://playstore.google.com)
Configuration files (https://github.com/mailrocketsystems/AsteriskVOIP)

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

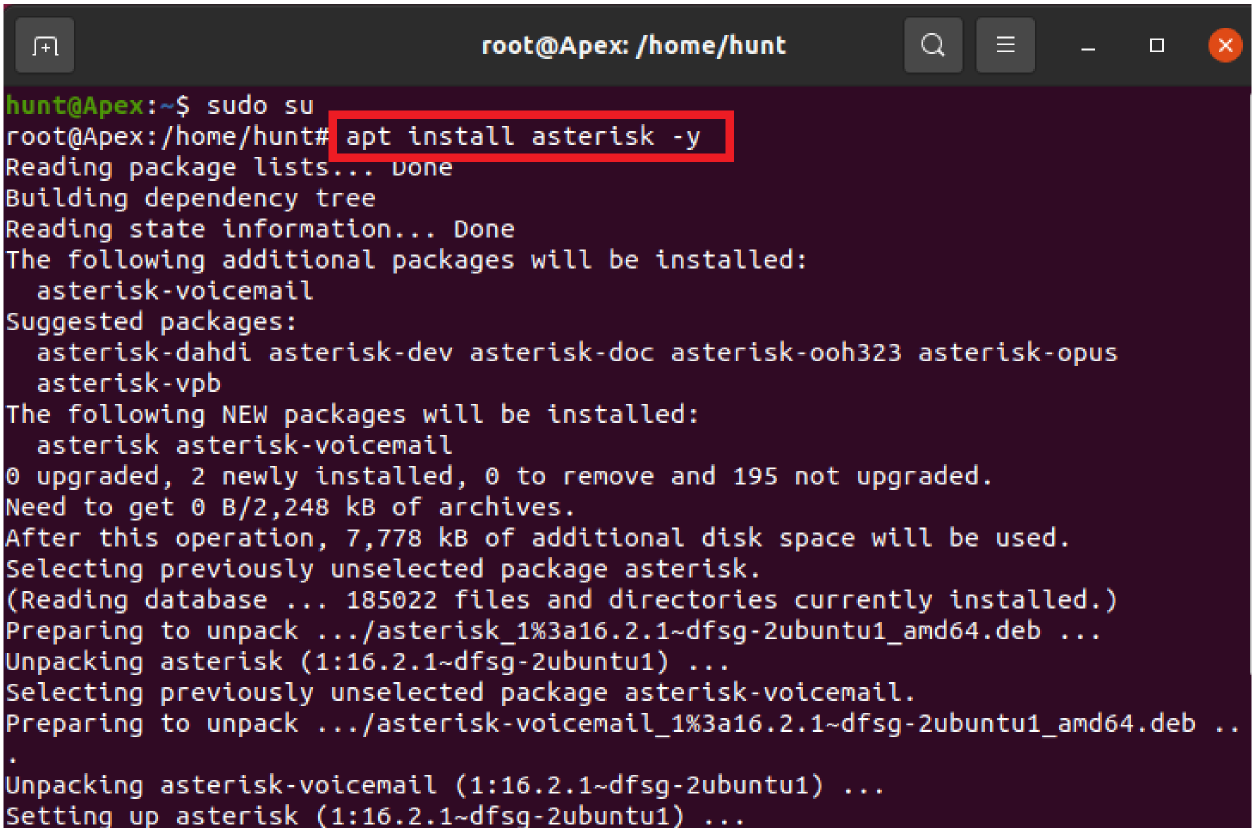


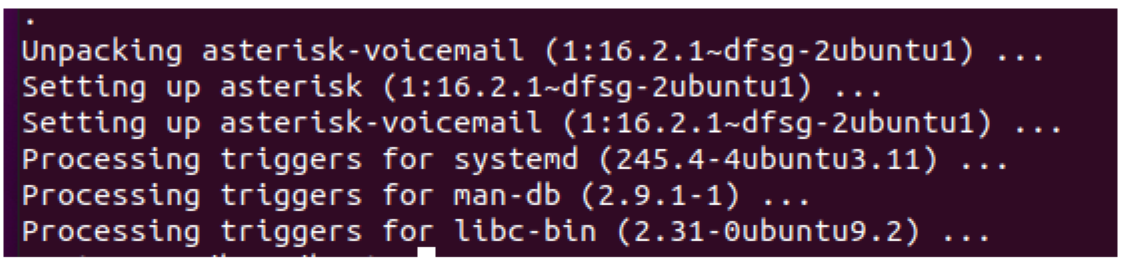
Enter the root can also ask for the password you've configured for your machine. If not then there is not much problem.

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.





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

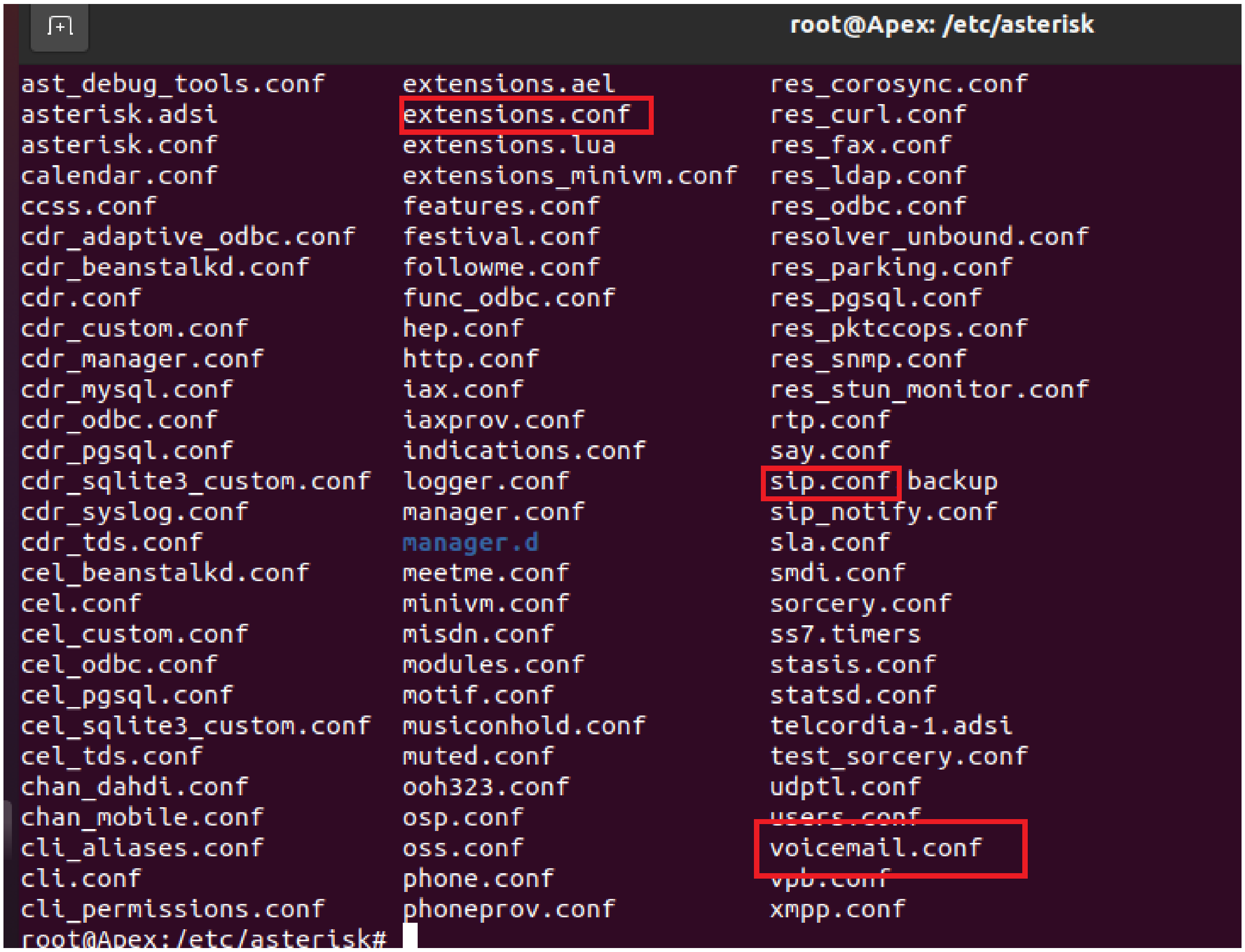
For this, we can use the command “**cd /etc/asterisk**”

```
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**



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_minivm.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                 minivm.conf             sorcery.conf
cel_custom.conf          misdn.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:

```
[internal]
exten => 7001,1,Answer()
exten => 7001,2,Dial(SIP/7001,60)
exten => 7001,3,Playback(vm-nobodyavail)
exten => 7001,4,VoiceMail(7001@main)
exten => 7001,5,Hangup()

exten => 7002,1,Answer()
exten => 7002,2,Dial(SIP/7002,60)
exten => 7002,3,Playback(vm-nobodyavail)
exten => 7002,4,VoiceMail(7001@main)
exten => 7002,5,Hangup()

exten => 8001,1,VoicemailMain(7001@main)
exten => 8001,2,Hangup()

exten => 8002,1,VoicemailMain(7002@main)
exten => 8002,2,Hangup()
```

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: Upadating 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 GVfs is not correctly installed or GVfs metadata are not supported on this platform. In the latter case, you should configure 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:

```
[general]
context=internal
allowguest=no
allowoverlap=no
bindport=5060
bindaddr=0.0.0.0
srvlookup=no
disallow=all
allow=ulaw
alwaysauthreject=yes
canreinvite=no
nat=yes
session-timers=refuse
localnet=192.168.0.0/255.255.255.0

[7001]
type=friend
host=dynamic
secret=7001
context=internal

[7002]
type=friend
host=dynamic
secret=7002
context=internal

```

Save the document after updating the files.

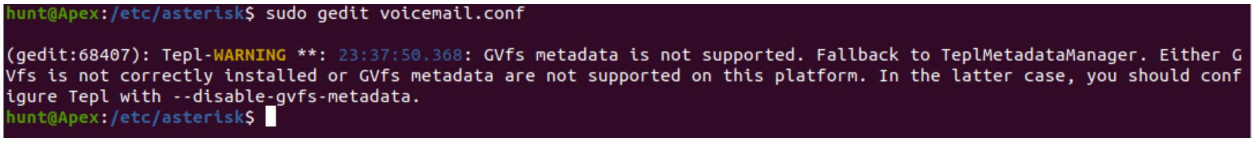
Step 11: Creating Backup file for “voicemail.conf”

```
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.

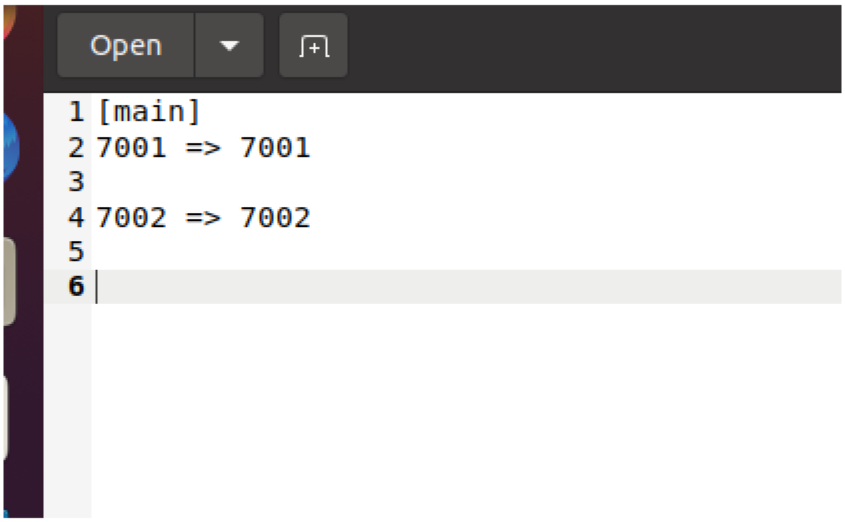
To create a backup file for voicemail.conf in order some inconvenience. The command we use is “**mv voicemail.conf voicemail.conf.backup**”

Step 12: Updating commands in “extensions.conf”



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

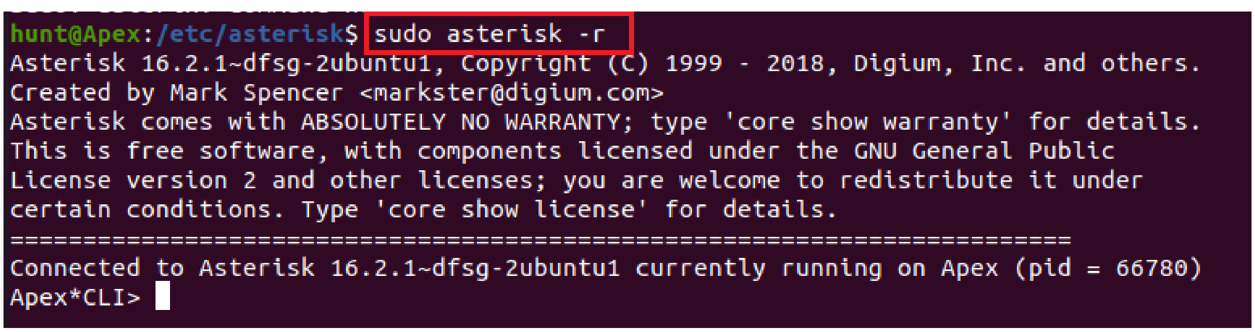
A pop will appear which can be then edit according to your own requirements. As shown:



Save the file after upgrading the data.

Step 13: Now, Accessing the server

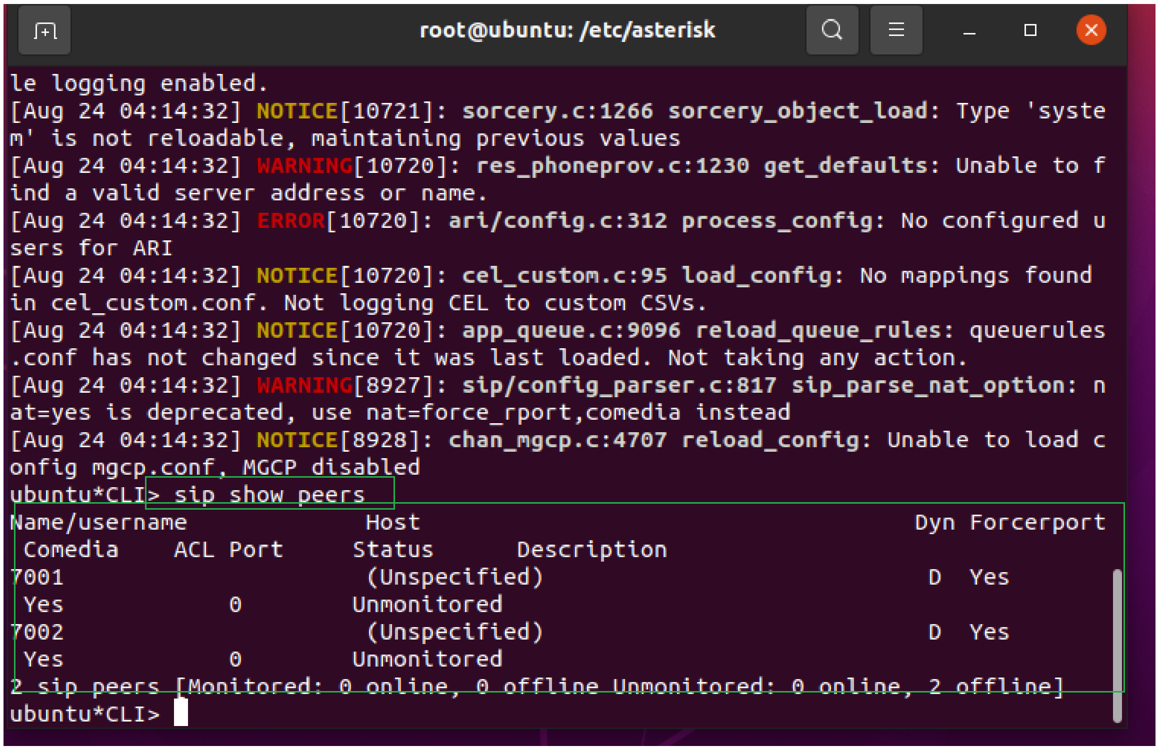
Command used: "**sudo asterisk -r**"



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.

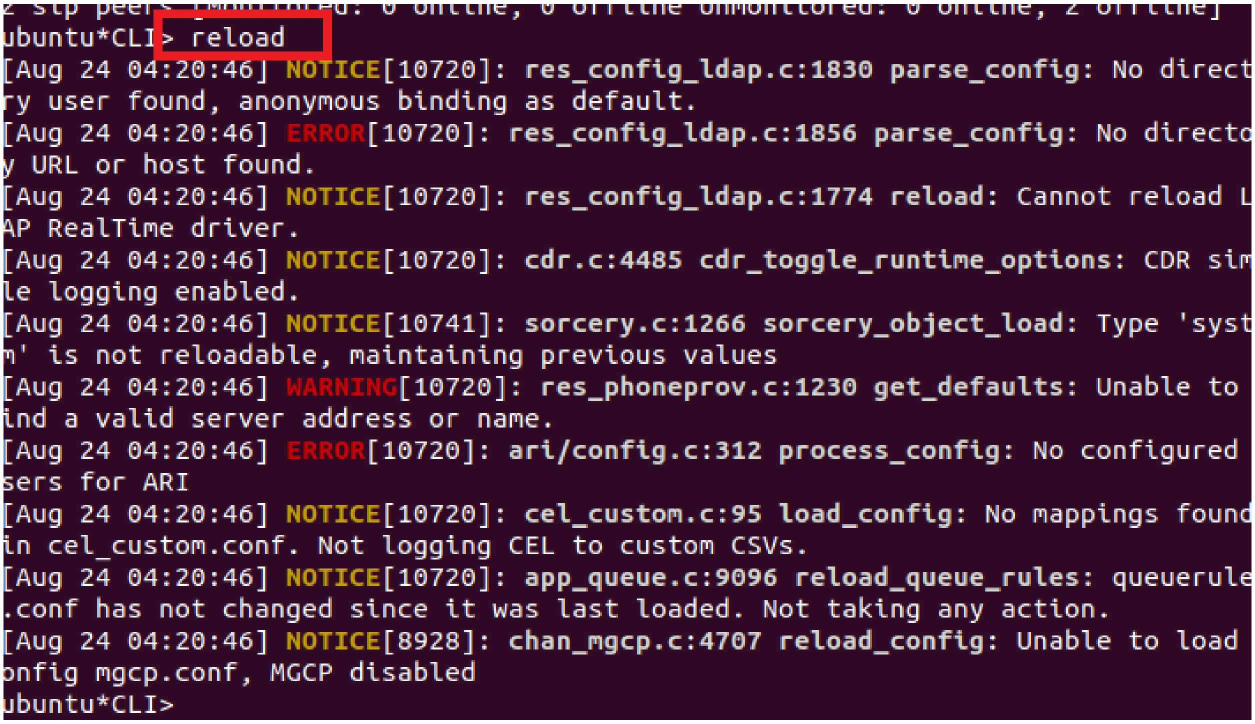


After entering the commands We've seen the available users.

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.



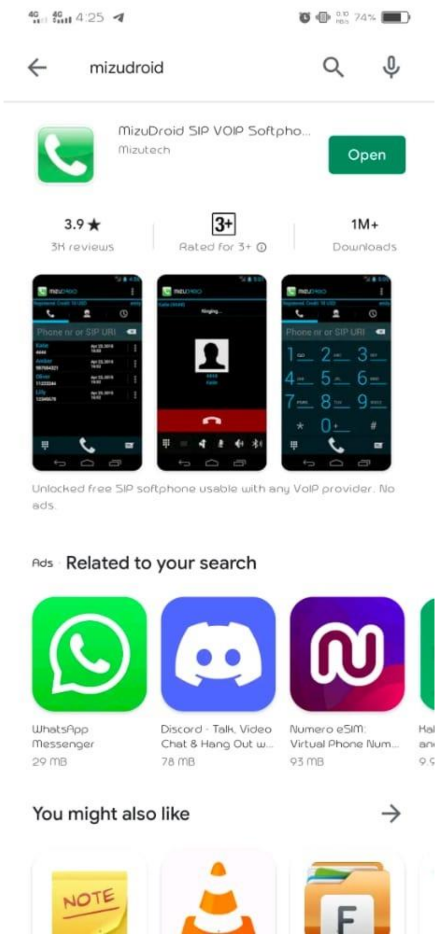
```
2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 0 online, 2 offline]
ubuntu*CLI> reload
[Aug 24 04:20:46] NOTICE[10720]: res_config_ldap.c:1830 parse_config: No direct
ry user found, anonymous binding as default.
[Aug 24 04:20:46] ERROR[10720]: res_config_ldap.c:1856 parse_config: No directo
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:

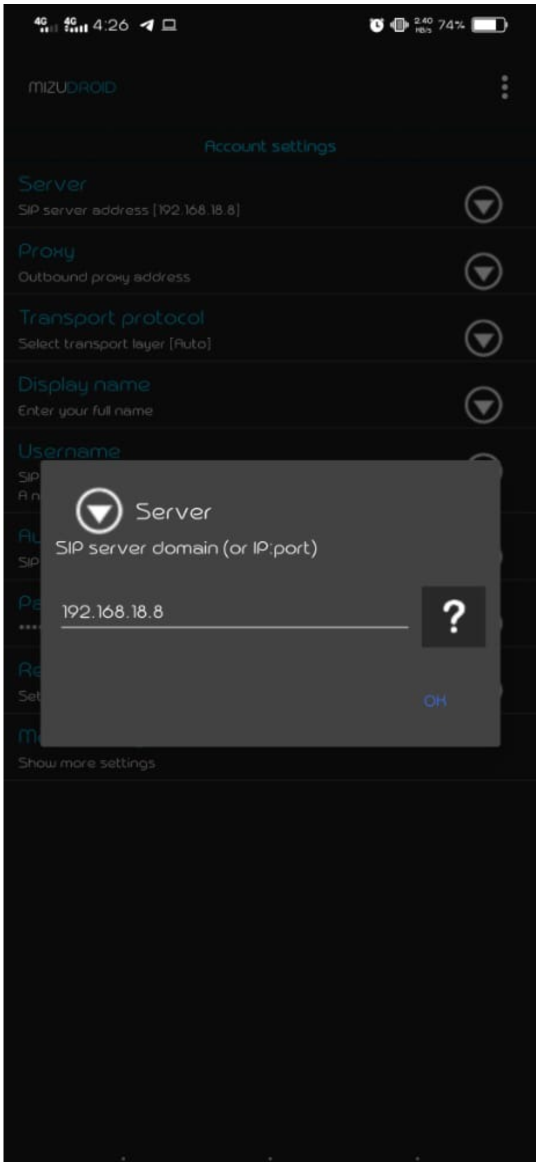
Step 16: Download any VoIP app in your mobile

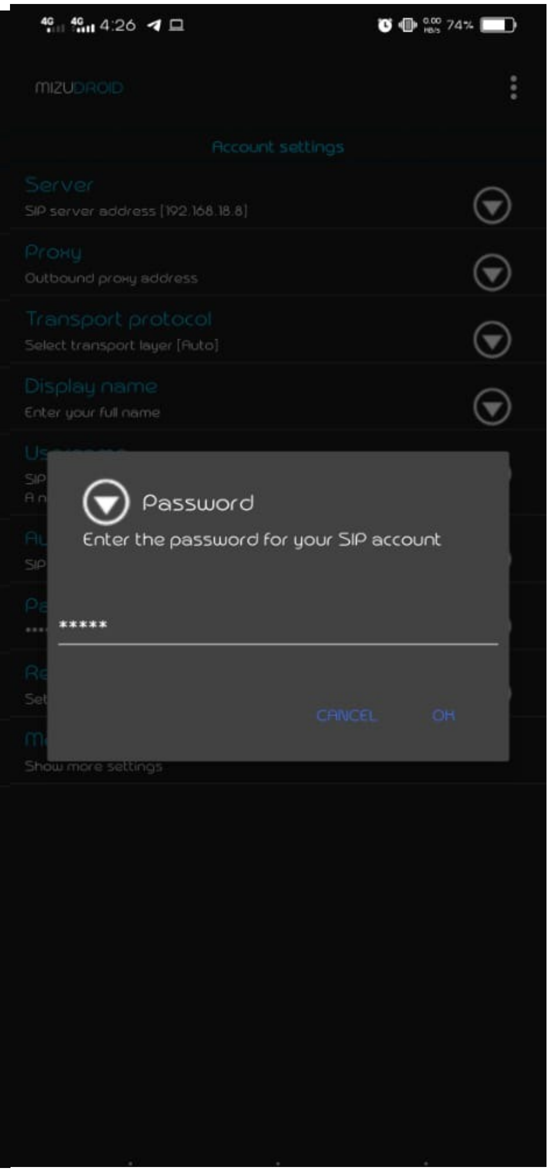
You can download any VoIP app with different capabilities.

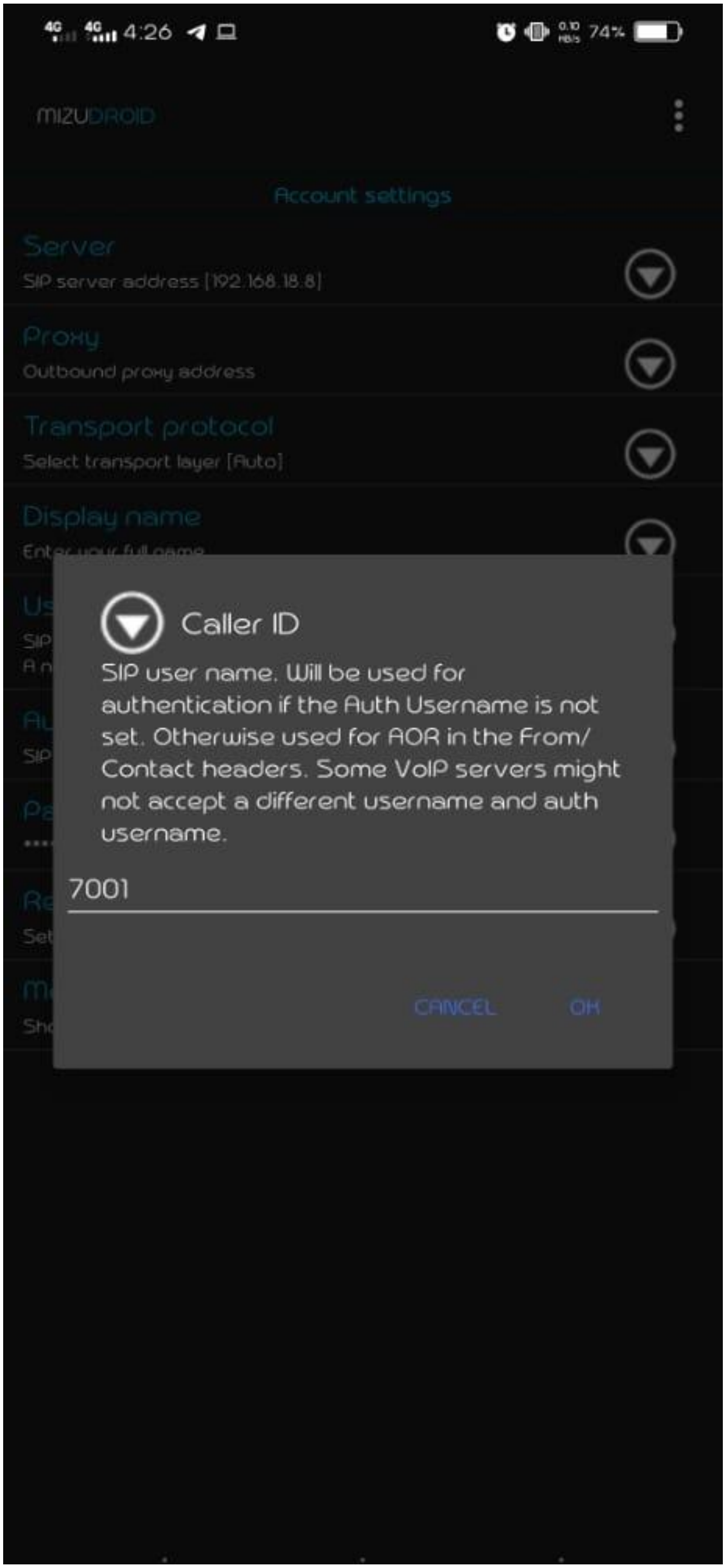
But here we’ve used ‘MizuDroid SIP VOIP” by MizuTech from google play store



Step 17: Configure the app settings

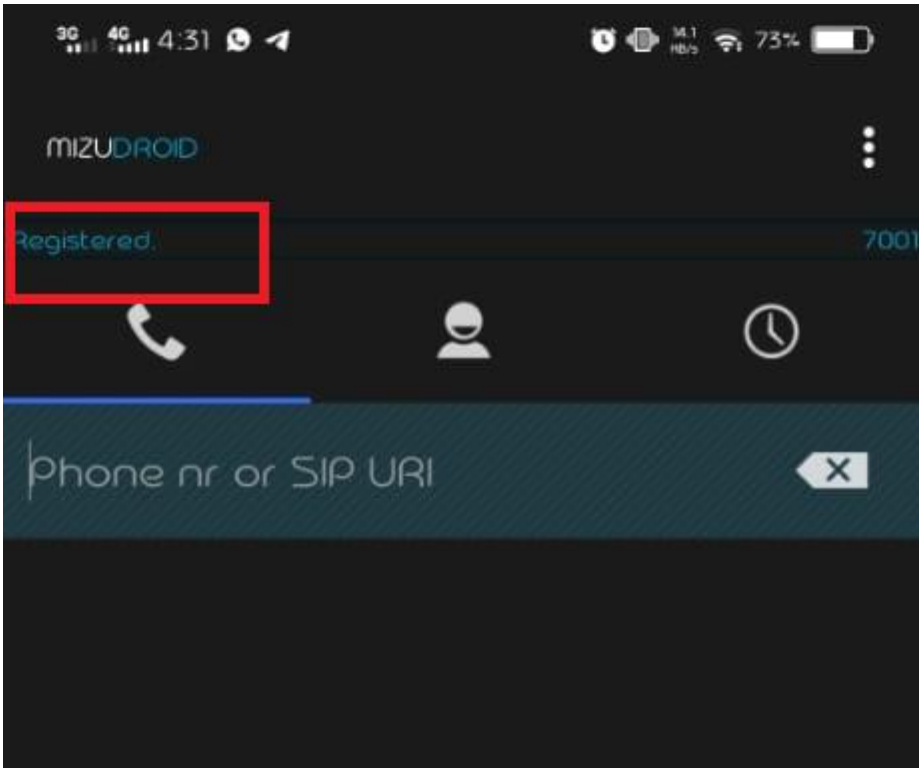

4:26
74%
MIZUDROID
Account settings
Server
SIP server address [192.168.18.8]
Proxy
Outbound proxy address
Transport protocol
Select transport layer [Auto]
Display name
Enter your full name
Username
Server
SIP server domain (or IP:port)
192.168.18.8
?
OK
Show more settings


4:26
74%
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
Password
Enter the password for your SIP account
*****
CANCEL
OK
Show more settings


4:26
74%
MIZUDROID
Account settings
Server
SIP server address [192.168.18.8]
Proxy
Outbound proxy address
Transport protocol
Select transport layer [Auto]
Display name
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

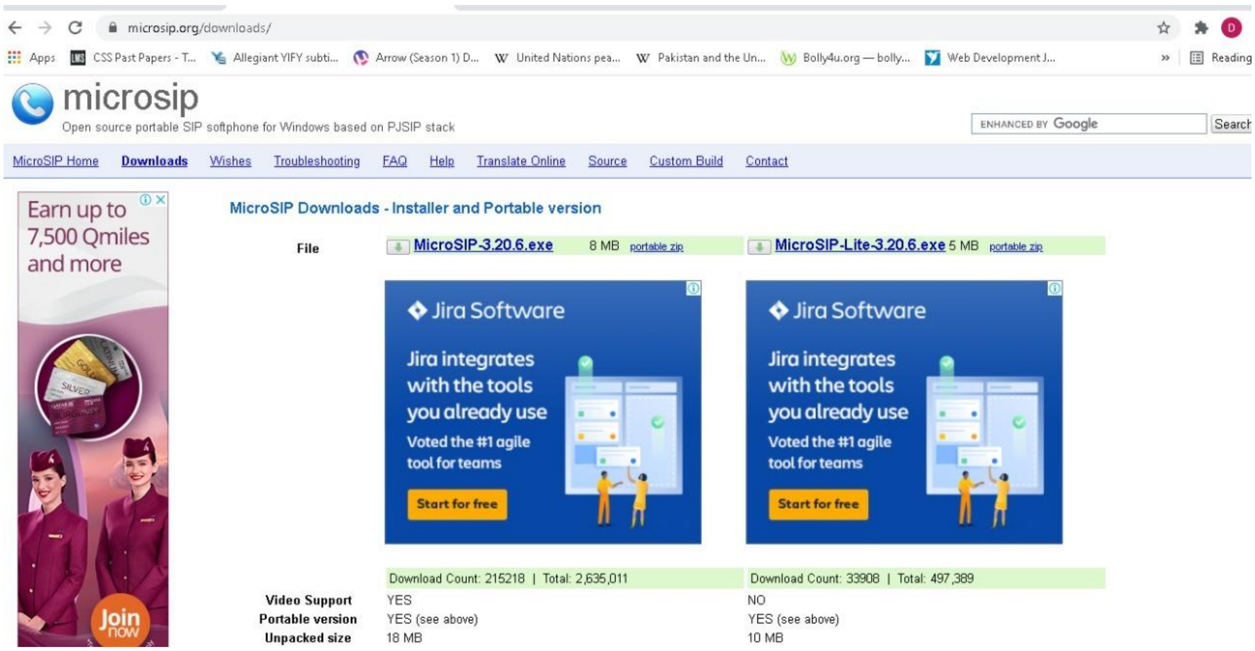


Step 19: Configure the second device

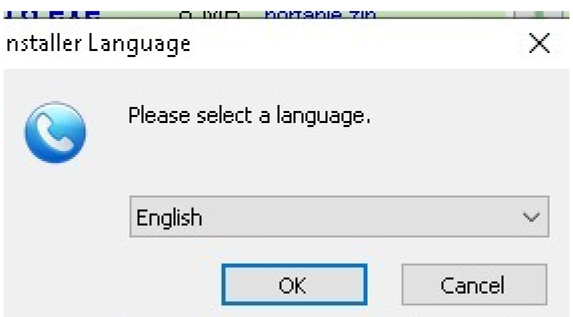
We’ve used a laptop for that

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

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

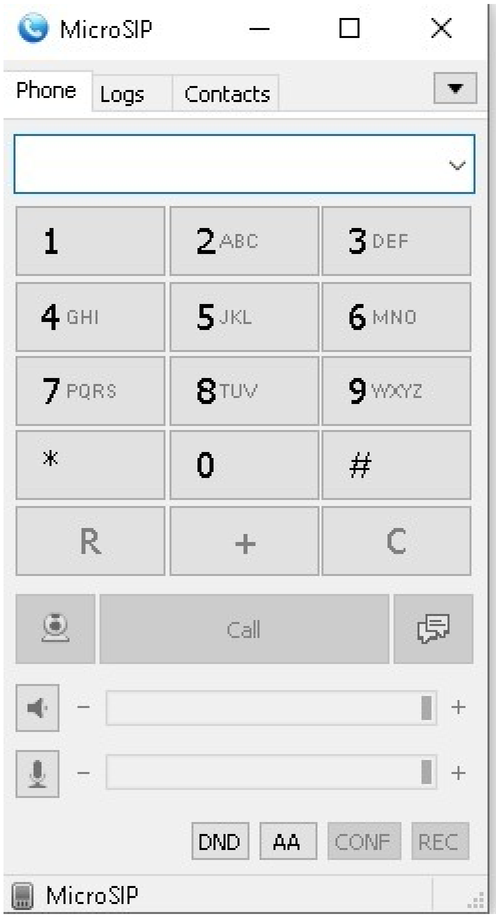


Step 20: Install the related App.



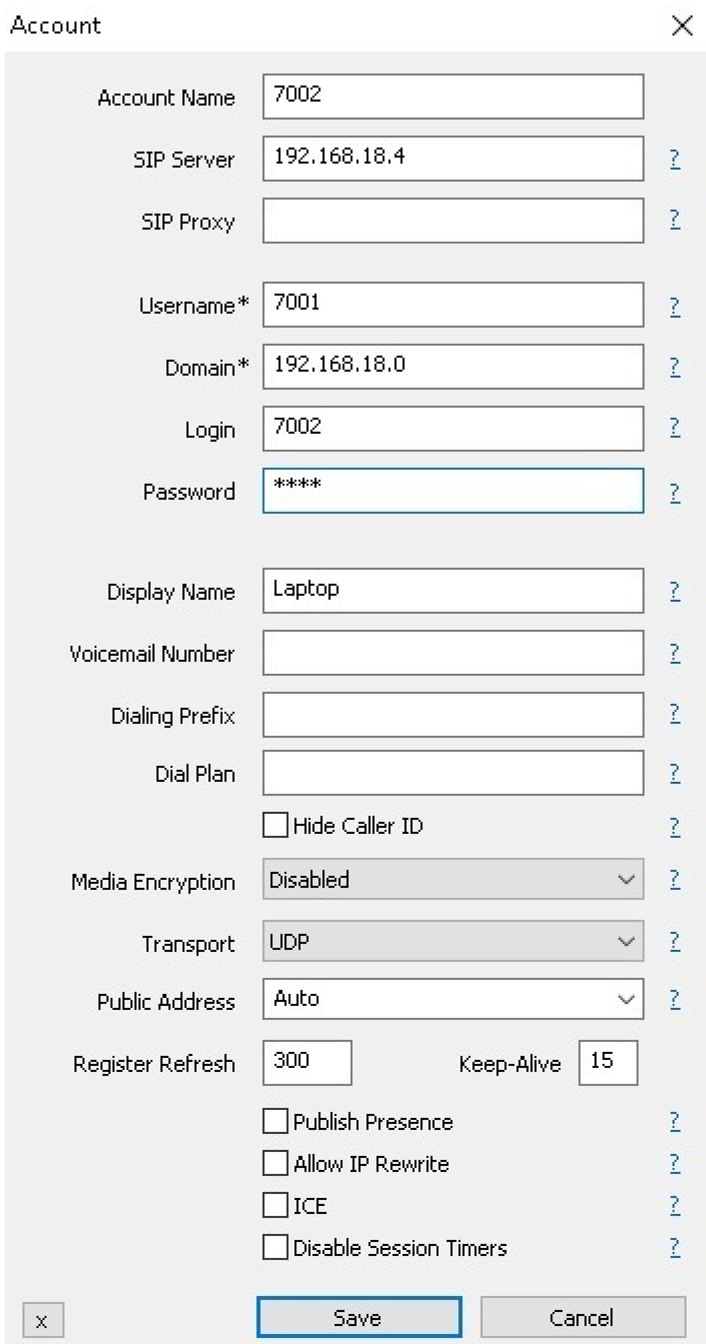
Proceed the installation. So, the installation completes in a better way.

Step 21:

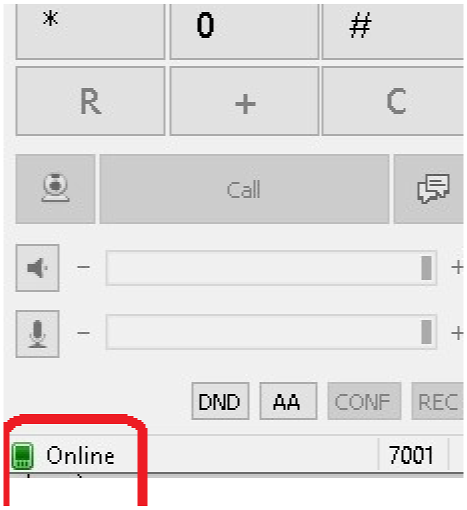


This shows that device is offline

Step 22: Configure the user over the laptop



Step 23: Check the availability of the device



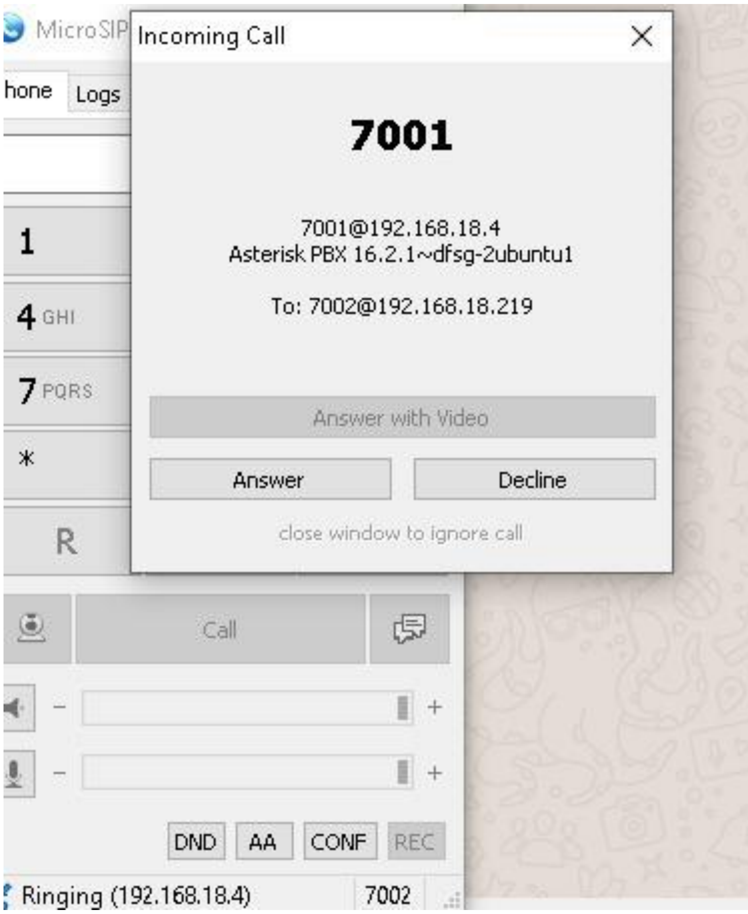
This shows that the device is completely online.

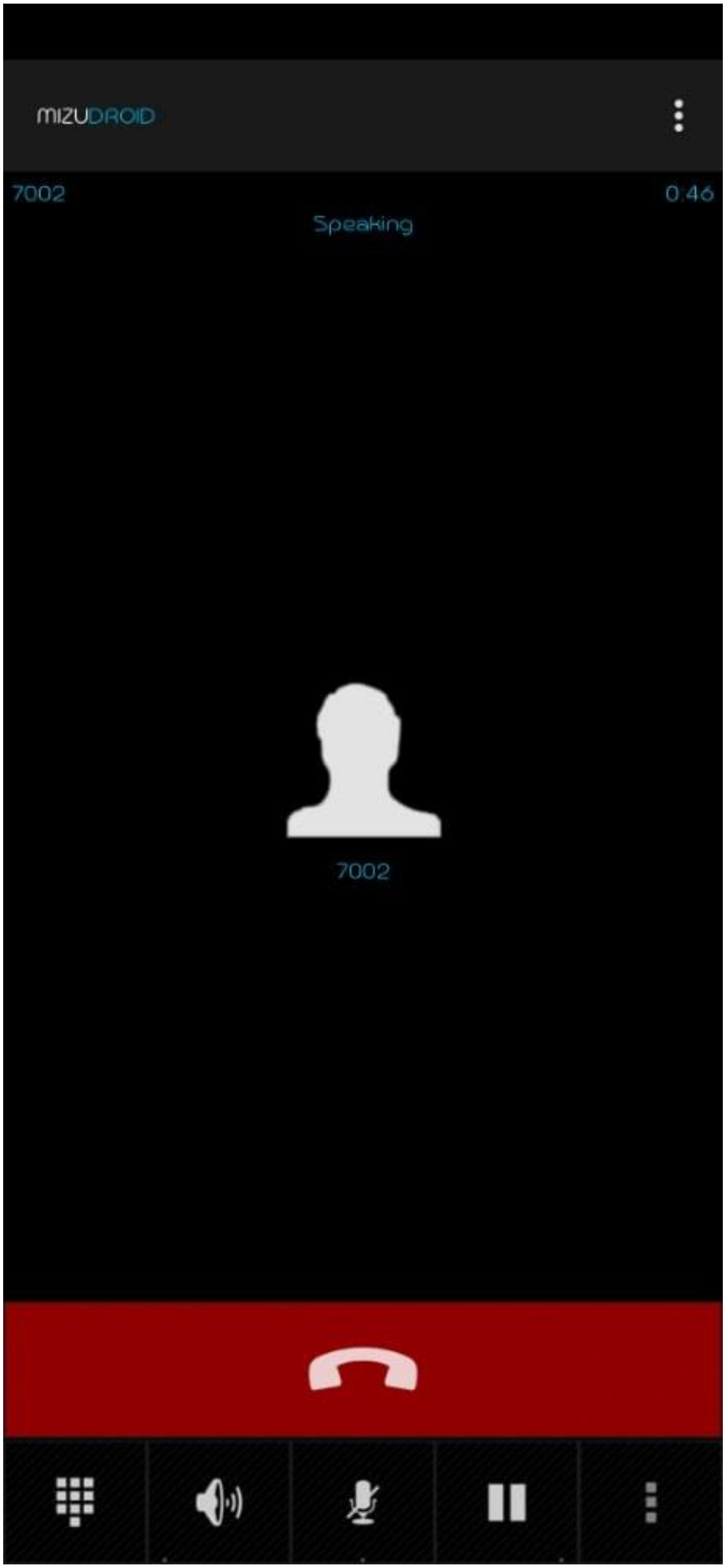
Step 24: Checking the connectivity of clients from server.

```
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


MicroSIP
hone
Logs
Incoming Call
7001
7001@192.168.18.4
Asterisk PBX 16.2.1~dfsg-2ubuntu1
To: 7002@192.168.18.219
Answer with Video
Answer
Decline
close window to ignore call
1
4 GHI
7 PQRS
*
R
Call
DND
AA
CONF
REC
Ringing (192.168.18.4)
7002


mizuDROID
7002
0:46
Speaking
7002