

Asterisk + Zoiper (Beginner Setup Guide)

<https://chatgpt.com/share/6960dd50-2c30-800a-99a2-49e1ebd125dc>

Goal

- ✓ Install Asterisk
 - ✓ Create SIP extensions (1001, 1002)
 - ✓ Register phones using Zoiper
 - ✓ Make calls **through Asterisk**
-

Basic Understanding (1 minute)

Phone → Asterisk → Phone

- Phones **do NOT call each other directly**
 - **All calls go through Asterisk**
 - Asterisk controls call logic
-

System Used

- OS: Ubuntu (local laptop)
 - SIP stack: **PJSIP**
 - Softphone: **Zoiper (Free)**
 - Network: Same Wi-Fi
-

Important Asterisk Files

File	Purpose
/etc/asterisk/pjsip.conf	SIP accounts (extensions)
/etc/asterisk/extensions.conf	Call routing (dialplan)

◆ STEP 1: Create SIP Extensions (pjsip.conf)

Open:

```
sudo nano /etc/asterisk/pjsip.conf
```

Transport (only once)

```
[transport-udp]
type=transport
protocol=udp
bind=0.0.0.0
```

Extension 1001

```
[1001]
type=endpoint
context=internal
disallow=all
allow=ulaw
auth=1001-auth
aors=1001
```

```
[1001-auth]
type=auth
auth_type=userpass
username=1001
password=1234
```

```
[1001]
type=aor
```

```
max_contacts=1
```

Extension 1002

```
[1002]
type=endpoint
context=internal
disallow=all
allow=ulaw
auth=1002-auth
aors=1002
```

```
[1002-auth]
type=auth
auth_type=userpass
username=1002
password=1234
```

```
[1002]
type=aor
max_contacts=1
```



Reload SIP

```
pjsip reload
```

Check:

```
pjsip show endpoints
```

You should see:

```
1001
1002
```

◆ **STEP 2: Create Dialplan (extensions.conf)**

Open:

```
sudo nano /etc/asterisk/extensions.conf
```

Add:

```
[internal]
exten => 1001,1,Dial(PJSIP/1001)
exten => 1002,1,Dial(PJSIP/1002)
```

Reload:

```
dialplan reload
```

◆ **STEP 3: Find Your Laptop IP**

ip a

Example:

```
192.168.1.10
```

👉 This IP is used in Zoiper.

◆ **STEP 4: Install Zoiper (Free)**

- Download **Zoiper Free**
- Free version = **1 account per device**

◆ STEP 5: Configure Zoiper Accounts

Zoiper on Laptop → Extension 1001

Field	Value
Username	1001
Password	1234
Hostname	Laptop IP (e.g., 192.168.1.10)
Port	5060
Transport	UDP

Zoiper on Mobile → Extension 1002

(Same Wi-Fi)

Field	Value
Username	1002
Password	1234
Hostname	Laptop IP
Port	5060
Transport	UDP

🔍 STEP 6: Confirm Registration

In Asterisk CLI:

pjsip show contacts

Expected:

1001 → registered

1002 → registered

STEP 7: Make the Call

From **1001**, dial:

1002

-
- Phone rings
 - Audio works
 - Call is successful
-

Real Call Flow (IMPORTANT)

Zoiper (1001)



Asterisk



Zoiper (1002)

-
- Incoming call → Asterisk
 - Outgoing call → Asterisk
-

What You Have Achieved

- Asterisk installed
- PJSIP configured
- Extensions created
- Phones registered
- Calls routed via Asterisk

This is the **foundation** for:

- IVR
- AGI
- Call recording
- Real Indian numbers

- Call centers
-

For 1001 device is an AI agent, and 1002 acts as a client/user

```
*****
```

If the asterisk has any problems, then remove it and reinstall

```
sudo systemctl stop asterisk  
sudo apt remove --purge asterisk asterisk-*  
sudo apt autoremove --purge  
sudo apt autoclean
```

```
sudo rm -rf /etc/asterisk  
sudo rm -rf /var/lib/asterisk  
sudo rm -rf /var/log/asterisk  
sudo rm -rf /var/spool/asterisk  
sudo rm -rf /usr/lib/asterisk  
which asterisk
```

```
*****
```

Install fresh asterisk

```
sudo apt update  
sudo apt install asterisk
```

```
asterisk -V
```

```
sudo systemctl enable asterisk  
sudo systemctl start asterisk  
sudo systemctl status asterisk
```

```
sudo asterisk -rvvv
```

```
*****
```

Add the device to the file

sudo nano /etc/asterisk/pjsip.conf

On top of the file
[transport-udp]
type=transport

```
protocol=udp  
bind=0.0.0.0
```

At the bottom of the file

```
[1001]  
type=endpoint  
context=internal  
disallow=all  
allow=ulaw  
auth=1001-auth  
aors=1001
```

```
[1001-auth]  
type=auth  
auth_type=userpass  
username=1001  
password=1234
```

```
[1001]  
type=aor  
max_contacts=1
```

```
[1002]  
type=endpoint  
context=internal  
disallow=all  
allow=ulaw  
auth=1002-auth  
aors=1002
```

```
[1002-auth]  
type=auth  
auth_type=userpass  
username=1002  
password=1234
```

```
[1002]  
type=aor  
max_contacts=1
```

Reload SIP

```
sudo asterisk -rvvv
```

```
pjsip reload
```

Check:

```
pjsip show endpoints
```

You should see:

```
1001
```

```
1002
```

```
*****
```

Add extension which device has which action, all answer, hangup

```
sudo nano /etc/asterisk/extensions.conf
```

At the bottom of the file

[internal]

; AI AGENT

```
exten => 1001,1,NoOp(AI AGENT)
```

```
same => n,Answer()
```

```
same => n,AGI(voice_agent_stt.py) same => n,Hangup()
```

; HUMAN PHONE

```
exten => 1002,1,Dial(PJSIP/1002,30)
```

```
same => n,Hangup()
```

Reload dailplan

```
sudo asterisk -rvvv
```

```
dialplan reload
```

```
# // voice_agent_stt this file calls when 1002 calls 1001 and STT ->LLM-> TTS work
```

```
*****
```

The necessary command needed to run

```
sudo systemctl restart asterisk
```

- ls -ld /var/lib/asterisk/agi-bin
If this found - ls: cannot access '/var/lib/asterisk/agi-bin': No such file or directory
- sudo mkdir -p /var/lib/asterisk/agi-bin
- sudo chown -R asterisk:asterisk /var/lib/asterisk
- sudo chmod 755 /var/lib/asterisk/agi-bin
- ls -ld /var/lib/asterisk/agi-bin
Above command returns this - drwxr-xr-x 2 asterisk asterisk 4096 Jan 12 11:49 /var/lib/asterisk/agi-bin
 - sudo ln -s /home/divya/Divya/diya/voice-agent/voice_agent_stt.py \
/var/lib/asterisk/agi-bin/voice_agent_stt.py
 - sudo chmod +x /home/divya/Divya/diya/voice-agent/voice_agent_stt.py

If still not run

- ls -l /usr/share/asterisk/agi-bin
It returns - total 0
- sudo ln -s /home/divya/Divya/diya/voice-agent/voice_agent_stt.py \
/usr/share/asterisk/agi-bin/voice_agent_stt.py
- ls -l /usr/share/asterisk/agi-bin
It returns this - total 0

lrwxrwxrwx 1 root root 53 Jan 12 11:59 voice_agent_stt.py --> /home/divya/Divya/diya/voice-agent/voice_agent_stt.py
- sudo rm -f /var/lib/asterisk/agi-bin/voice_agent_stt.py
- sudo ln -sf /home/divya/Divya/diya/voice-agent/voice_agent_stt.py \
/var/lib/asterisk/agi-bin/voice_agent_stt.py
- sudo chmod 755 /home/divya/Divya/diya/voice-agent/voice_agent_stt.py

Now you call 1002 to 1001

User speaks - hello

AI agent returns to speak - how can I assist you, etc

sudo asterisk -rvvv

In this, you can show logs and print STT and LLM response, and TTS logs