📘 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

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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

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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

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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

Project rerun or run after days

sudo asterisk -rx "dialplan reload"
sudo asterisk -rvvv

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