

# 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
<code>/etc/asterisk/pjsip.conf</code>	SIP accounts (extensions)
<code>/etc/asterisk/extensions.conf</code>	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 dialplan**

```
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