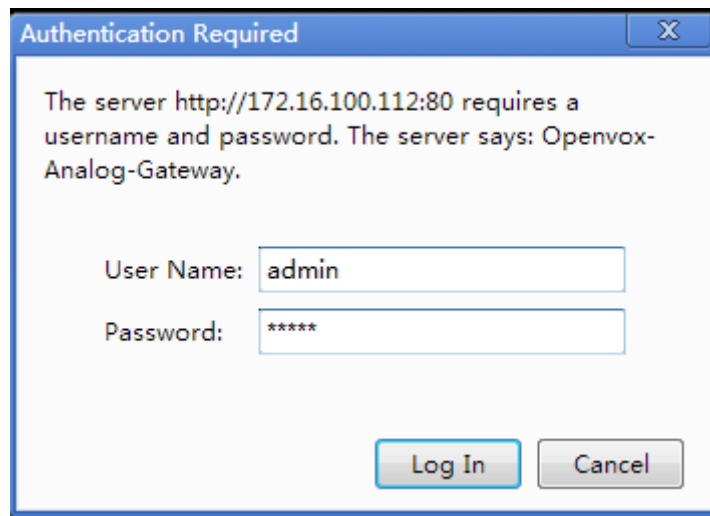


You can quickly configure your gateway as follow steps.

Step1. Log in your gateway Web GUI.



The server http://172.16.100.112:80 requires a username and password. The server says: Openvox-Analog-Gateway.

User Name:

Password:

Step2. Network Settings

If your system topology like the figure described, please enter the gateway default IP address to login web, and click "NETWORK—>LAN Settings" to set network parameters such as IP.

| LAN IPv4 | |
|------------|-------------------|
| Interface: | eth0 |
| Type: | Static ▼ |
| MAC: | A0:98:05:01:0B:27 |

| IPv4 Settings | |
|------------------|----------------|
| Address: | 172.16.100.112 |
| Netmask: | 255.255.0.0 |
| Default Gateway: | 172.16.0.1 |

Step3. Create a SIP Endpoint in Web

Please select "SIP—>SIP Endpoints—>Add New SIP Endpoint" to set a new SIP endpoint. The following figure shows detail information about how to set it.

Add New SIP Endpoint



▼ Main Endpoint Settings

| | |
|--------------------------------|---|
| Name: | <input type="text" value="501"/> |
| User Name: | <input type="text" value="501"/> <input type="checkbox"/> Anonymous |
| Password: | <input type="text" value="501"/> |
| Registration: | <input type="text" value="This gateway registers with the endpoint"/> ▼ |
| Hostname or IP Address: | <input type="text" value="172.16.8.112"/> |
| Transport: | <input type="text" value="UDP"/> ▼ |
| NAT Traversal: | <input type="text" value="Yes"/> ▼ |
| SUBSCRIBE for MWI: | <input type="text" value="No"/> ▼ |

About other parameters in SIP, please set by your requirements for there is no need to set them in simple calls.

Then you should modify your Channel Settings, "ANALOG -> Channel Settings" to set Sip

Account. You can press the button  .

| Port | Type | Name | Caller ID | Sip Account | CID signalling | Actions |
|------|------|---|-----------------------------------|-------------------------------------|-------------------------------------|---|
| 1 | FXS | <input type="text" value="board1-port1"/> | <input type="text" value="301"/> | <input type="text" value="301"/> ▼ | <input type="text" value="bell"/> ▼ |  |
| 2 | FXS | <input type="text" value="board1-port2"/> | <input type="text" value="8002"/> | <input type="text" value="None"/> ▼ | <input type="text" value="bell"/> ▼ |  |
| 3 | FXS | <input type="text" value="board1-port3"/> | <input type="text" value="8003"/> | <input type="text" value="None"/> ▼ | <input type="text" value="bell"/> ▼ |  |
| 4 | FXS | <input type="text" value="board1-port4"/> | <input type="text" value="8004"/> | <input type="text" value="None"/> ▼ | <input type="text" value="bell"/> ▼ |  |
| 5 | FXS | <input type="text" value="board1-port5"/> | <input type="text" value="8005"/> | <input type="text" value="None"/> ▼ | <input type="text" value="bell"/> ▼ |  |
| 6 | FXS | <input type="text" value="board1-port6"/> | <input type="text" value="8006"/> | <input type="text" value="None"/> ▼ | <input type="text" value="bell"/> ▼ |  |
| 7 | FXS | <input type="text" value="board1-port7"/> | <input type="text" value="8007"/> | <input type="text" value="None"/> ▼ | <input type="text" value="bell"/> ▼ |  |
| 8 | FXS | <input type="text" value="board1-port8"/> | <input type="text" value="8008"/> | <input type="text" value="None"/> ▼ | <input type="text" value="bell"/> ▼ |  |

Board-1-Port 1

▼ General

| | |
|----------------------|---|
| Port type: | FXS |
| Name: | <input type="text" value="board1-port1"/> |
| Rx gain: | <input type="text" value="0.0"/> |
| Tx gain: | <input type="text" value="0.0"/> |
| Ring timeout: | <input type="text" value="8000"/> |
| Sip Account: | <input type="text" value="501"/> ▼ |

▼ Caller ID

| | |
|------------------------|-------------------------------------|
| Caller ID: | <input type="text" value="501"/> |
| Full name: | <input type="text" value="501"/> |
| CID signalling: | <input type="text" value="bell"/> ▼ |

Save
Cancel

You can choose the Sip Account that you have set up for every port.

| Port | Type | Name | Caller ID | Sip Account | CID signalling | Actions |
|------|------|---|-----------------------------------|-------------------------------------|-------------------------------------|---------|
| 1 | FXS | <input type="text" value="board1-port1"/> | <input type="text" value="501"/> | <input type="text" value="501"/> ▼ | <input type="text" value="bell"/> ▼ | |
| 2 | FXS | <input type="text" value="board1-port2"/> | <input type="text" value="502"/> | <input type="text" value="502"/> ▼ | <input type="text" value="bell"/> ▼ | |
| 3 | FXS | <input type="text" value="board1-port3"/> | <input type="text" value="8003"/> | <input type="text" value="None"/> ▼ | <input type="text" value="bell"/> ▼ | |
| 4 | FXS | <input type="text" value="board1-port4"/> | <input type="text" value="8004"/> | <input type="text" value="None"/> ▼ | <input type="text" value="bell"/> ▼ | |

That's all. Now the board 1-port 1 phone num is 501, and the board 1-port 2 phone num is 502, you can make calls between 501 and 502.

Step4. Create Extensions in Elastix® Server

Don't forget to create Extensions 501 and 502 on your Elastix server.

Dashboard

- Extensions**
- Feature Codes
- General Settings
- Outbound Routes
- Trunks

Inbound Call Control

- Inbound Routes
- Zap Channel DIDs
- Announcements
- Blacklist
- CallerID Lookup Sources
- Day/Night Control
- Follow Me
- IVR
- Queue Priorities
- Queues
- Ring Groups
- Time Conditions
- Time Groups

Internal Options & Configuration

- Conferences
- Languages
- Misc Applications
- Misc Destinations
- Music on Hold
- PIN Sets
- Paging and Intercom
- Parking Lot
- System Recordings
- VoiceMail Blasting

Remote Access

- Callback
- DISA

Option

- Unembedded freePBX

Add SIP Extension

Add Extension

User Extension: 501

Display Name: 501

CID Num Alias:

SIP Alias:

Extension Options

Outbound CID:

Ring Time: Default

Call Waiting: Disable

Call Screening: Disable

Pinless Dialing: Disable

Emergency CID:

Assigned DID/CID

DID Description:

Add Inbound DID:

Add Inbound CID:

Device Options

This device uses sip technology.

secret: rfc501

dtmfmode: rfc2833

After that, you can register a soft sip phone with the name "1001" on the Elastix Server , the same method as above. Then you can make calls to 501 or 502 from SIP 1001.