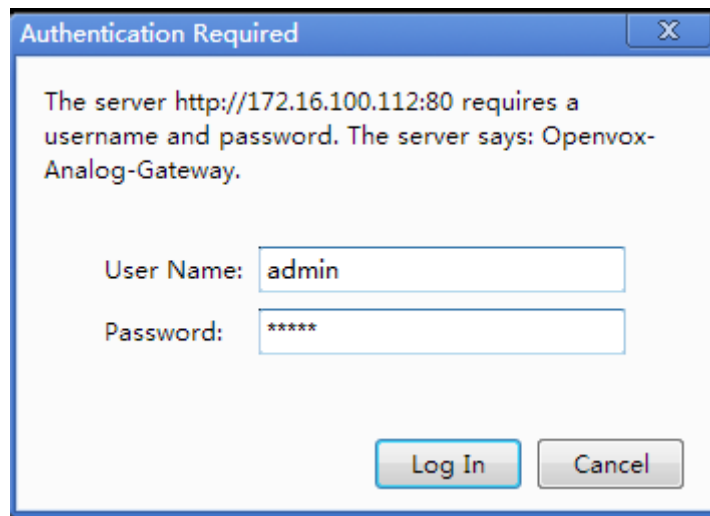




You can quickly configure your gateway as follow steps.

**Step1. Log in your gateway Web GUI.**



**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

<b>Name:</b>	<input type="text" value="501"/>
<b>User Name:</b>	<input type="text" value="501"/> <input type="checkbox"/> Anonymous
<b>Password:</b>	<input type="text" value="501"/>
<b>Registration:</b>	<input type="text" value="This gateway registers with the endpoint"/> ▼
<b>Hostname or IP Address:</b>	<input type="text" value="172.16.8.112"/>
<b>Transport:</b>	<input type="text" value="UDP"/> ▼
<b>NAT Traversal:</b>	<input type="text" value="Yes"/> ▼
<b>SUBSCRIBE for MWI:</b>	<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

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

▼ Caller ID

<b>Caller ID:</b>	<input type="text" value="501"/>
<b>Full name:</b>	<input type="text" value="501"/>
<b>CID signalling:</b>	<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

Display Name

CID Num Alias

SIP Alias

Extension Options

Outbound CID

Ring Time

Call Waiting

Call Screening

Pinless Dialing

Emergency CID

Assigned DID/CID

DID Description

Add Inbound DID

Add Inbound CID

Device Options

This device uses sip technology.

secret

dtmfmode

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.