



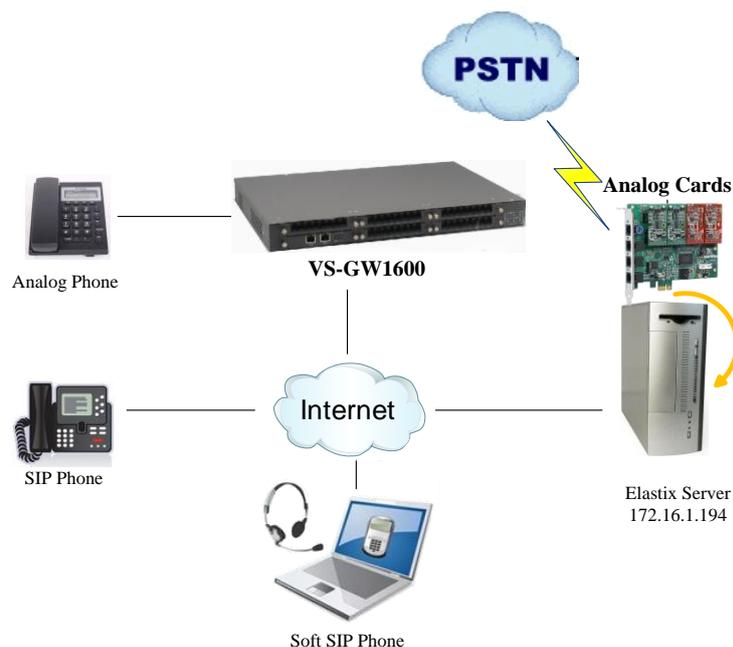
VS-GW1600-8/16/24/32/40S Connect with Elastix® Server

QUICKSTART GUIDE

This document applies to OpenVox VS-GW1600-8/16/24/32/40S series analog gateway. The figure below shows Default IP.

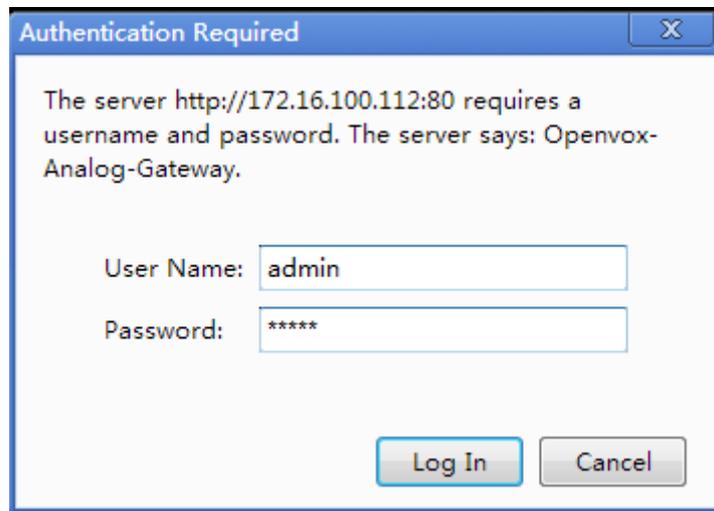
Stack Num	IP	Username	Password
1	172.16.99.1	admin	admin
2	172.16.99.2	admin	admin
3	172.16.99.3	admin	admin
4	172.16.99.4	admin	admin
5	172.16.99.5	admin	admin

This is an example with **8 FXS ports**. The Default IP is **172.16.99.1**, Username is **admin** and Password is **admin** too. There are two LAN ports, you can connect gateway to Internet through either of them and you can see the connectivity by LED status.



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.