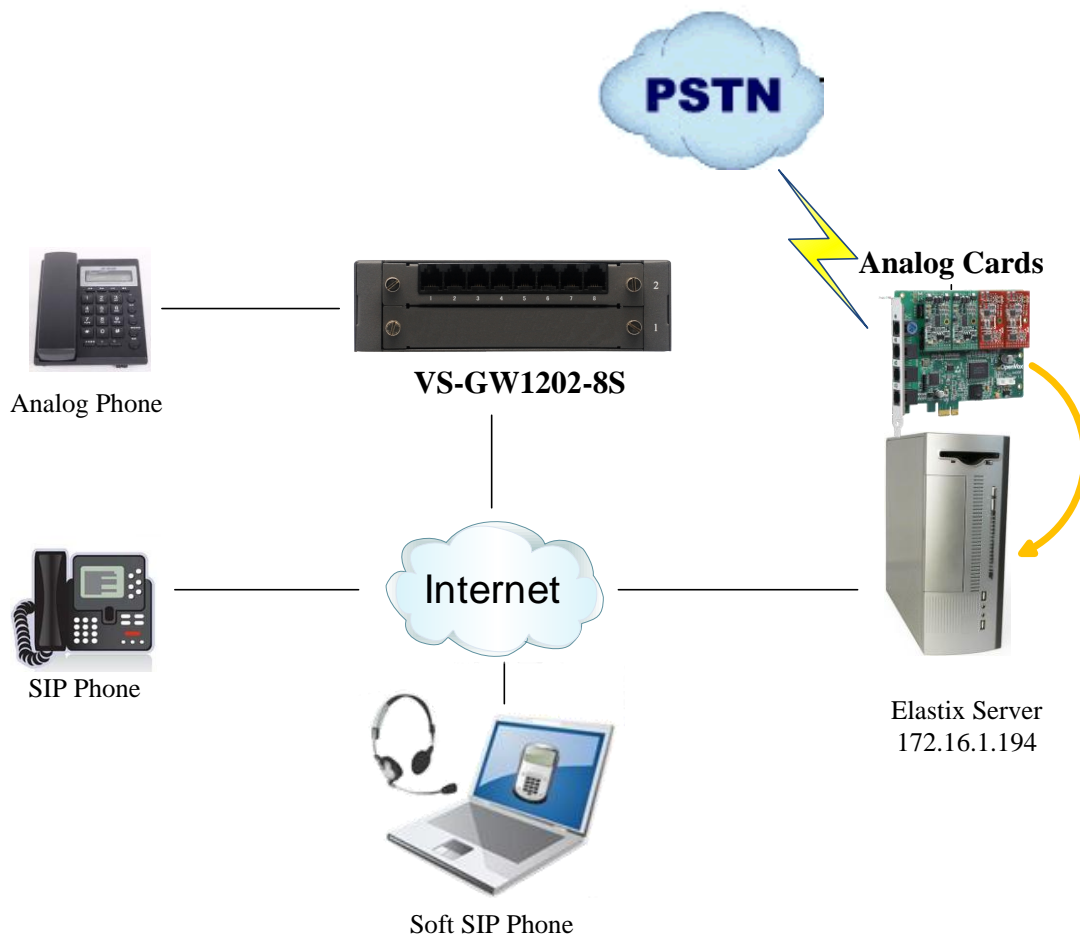




VS-GW1202-8S Connect with Elastix® Server

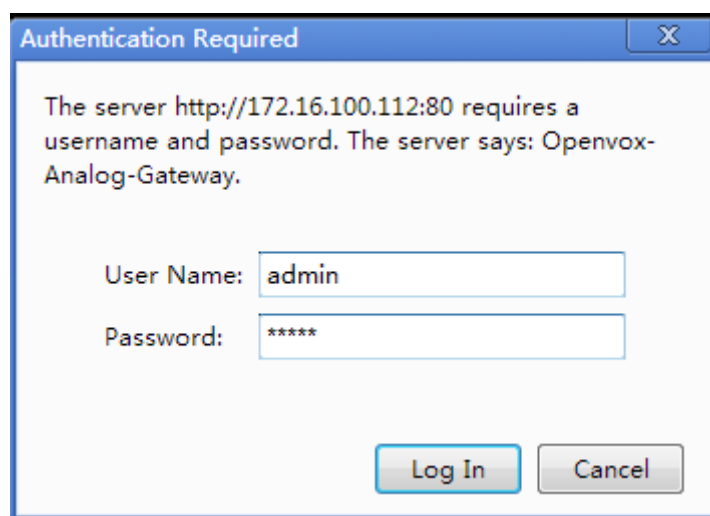
QUICKSTART GUIDE

This document applies to OpenVox VS-GW1202-8S series analog gateway. 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:	board1-port1
Rx gain:	0.0
Tx gain:	0.0
Ring timeout:	8000
Sip Account:	501 ▼

▼ Caller ID

Caller ID:	501
Full name:	501
CID signalling:	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	board1-port1	501	501 ▼	bell ▼	
2	FXS	board1-port2	502	502 ▼	bell ▼	
3	FXS	board1-port3	8003	None ▼	bell ▼	
4	FXS	board1-port4	8004	None ▼	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.

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.