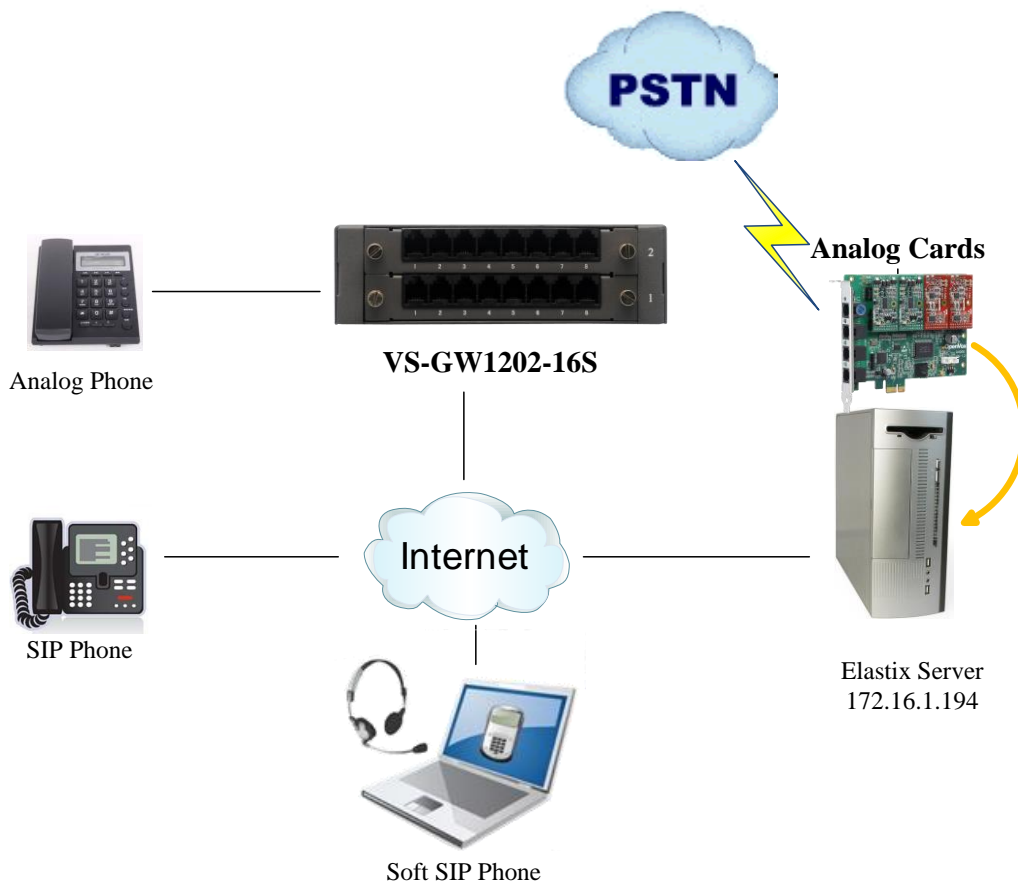




VS-GW1202-16S Connect with Elastix® Server

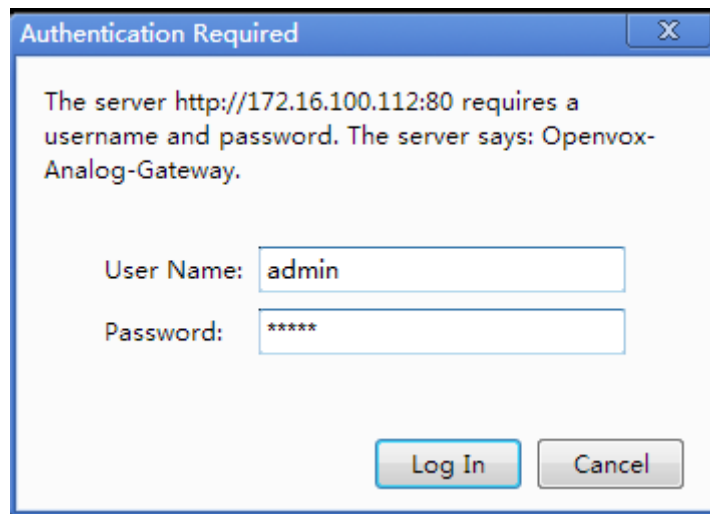
QUICKSTART GUIDE

This document applies to OpenVox VS-GW1202-16S analog gateway. 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:	<input type="text" value="172.16.100.112"/>
Netmask:	<input type="text" value="255.255.0.0"/>
Default Gateway:	<input type="text" value="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 a New SIP Endpoint

▼ Main Endpoint Settings

Name:	<input type="text" value="666"/>
Username:	<input type="text" value="666"/>
Password:	<input type="text" value="666"/>
Registration:	<input type="text" value="This gateway registers with the endpoint"/> ▼
Hostname or IP Address:	<input type="text" value="172.16.1.194"/>
Transport:	<input type="text" value="UDP"/> ▼
NAT Traversal:	<input type="text" value="Yes"/> ▼

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	Caller ID	Sip Account	Port Status	Action
1	FXS	8001		OnHook	
2	FXS	8002		OnHook	
3	FXS	8003		OnHook	
4	FXS	8004		OnHook	
5	FXS	8005		OnHook	
6	FXS	8006		OnHook	
7	FXS	8007		OnHook	
8	FXS	8008		OnHook	

Port 1





General

Port type:	FXS
Rx gain:	0.0
Tx gain:	0.0
Ring timeout:	8000
Sip Account:	666

Caller ID

Caller ID:	666
Full name:	666

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

Port	Type	Caller ID	Sip Account	Port Status	Action
1	FXS	666	666	OnHook	
2	FXS	888	888	OnHook	
3	FXS	8003		OnHook	
4	FXS	8004		OnHook	

Step4. Create a SIP Trunk in Elastix® Server

Please login your Elastix® server to create a SIP trunk (666). On Elastix® server web, please choose "PBX—>Trunks—>Add SIP Trunk" to set like that:

General Settings

Trunk Name:	666
Outbound Caller ID:	666
CID Options:	Allow Any CID
Maximum Channels:	

Outgoing Settings

Trunk Name:

PEER Details:

```

host=dynamic
username=666
secret=666
type=friend
    
```

Step5. Configure Outbound Routes in Elastix® Sever

Route Settings

Route Name:

Route CID: Override Extension

Route Password:

Dial Patterns that will use this Route

(prepend) + | [/ CallerId]

(prepend) + prefix | [match pattern / CallerId]

+ Add More Dial Pattern Fields

Dial patterns wizards: (pick one)

Trunk Sequence for Matched Routes

0

1

Add Trunk

Submit Changes

Step6. Set up a call

Please add one or more sip extensions, for example 1001.

Add an Extension

Please select your Device below then click Submit

Device

Device

Add Extension

User Extension

Display Name

CID Num Alias

SIP Alias

Device Options

This device uses sip technology.

secret

dtmfmode

After that, you can register a soft sip phone on your PC with the name "1001" to the Elastix Server. Then as configurations ahead, you can dial "6666", the analog phone "666" will rang.

If you want to dial out, you should install analog cards on your server.