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.
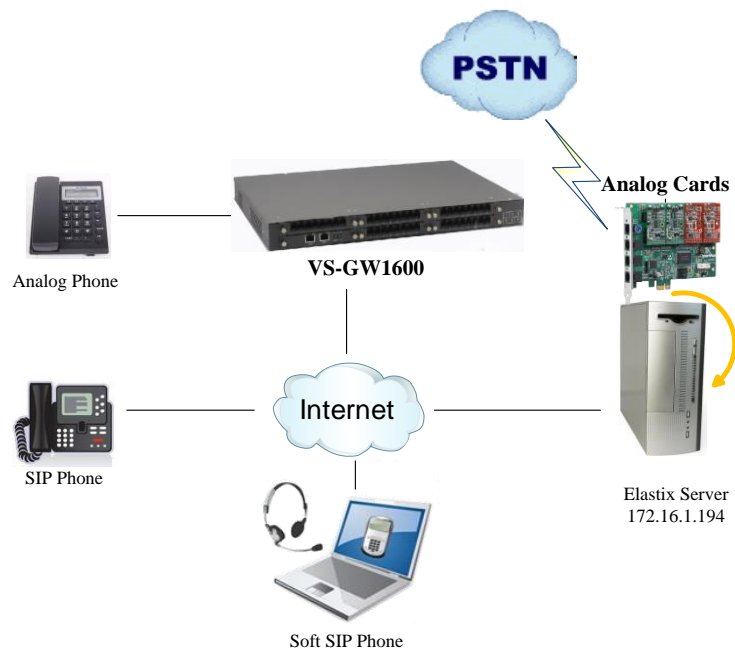
You can quickly configure your gateway as follow steps.

Step1. Log in your gateway Web GUI.
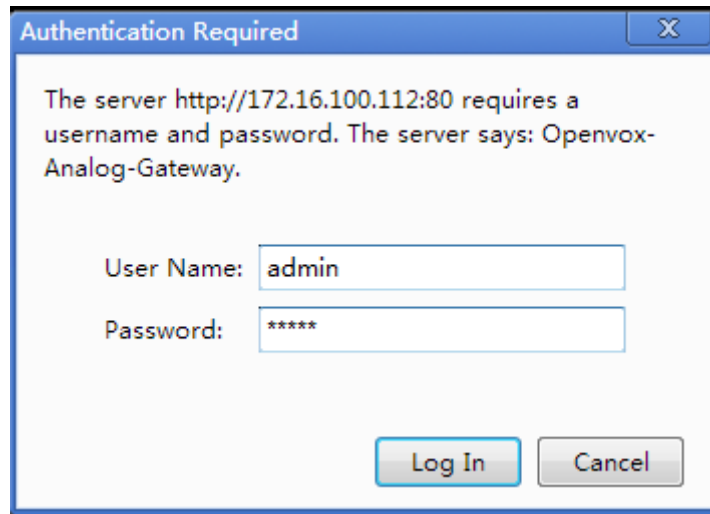
Step2. Network Settings

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.

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:	666
Username:	666
Password:	666
Registration:	This gateway registers with the endpoint
Hostname or IP Address:	172.16.1.194
Transport:	UDP
NAT Traversal:	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 [pencil icon] .

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

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

Step4. Create a SIP Trunk in Elastix® Server

Please login your Elastix® server to create a SIP trunk (1000). On Elastix® server web, please choose "PBX—>Trunks—>Add SIP Trunk" to set like that:

Outgoing Settings

Trunk Name: 666

PEER Details:

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

Step5.   Configure Outbound Routes in Elastix® Sever

Route Settings

Route Name: outgoing

Route CID: 666   ☐ Override Extension

Route Password:

Dial Patterns that will use this Route

(prepend) + 6 | [.   / CallerId   ] 🗑

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

+ Add More Dial Pattern Fields

Dial patterns wizards: (pick one) ▼

Trunk Sequence for Matched Routes

0   666   ▼ 🗑

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   Generic SIP Device ▼

Submit

Add Extension

User Extension	1001
Display Name	1001
CID Num Alias	
SIP Alias	

Device Options

This device uses sip technology.

secret	1001
dtmfmode	rfc2833

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.