GSM Gateway Connect with Asterisk® Server QUICKSTART GUIDE

This document applies to OpenVox GSM Gateway VS-GW1200-4G/VS-GW1202-8G and VS-GW1600 series. There are two RJ45 Network ports, ETH1 and ETH2. If you choose ETH1, you can access Board 1 only, and access other boards with the same IP address, different port numbers. This will help to avoid IP conflict. If you choose ETH2, you can access different Boards with different IP addresses.

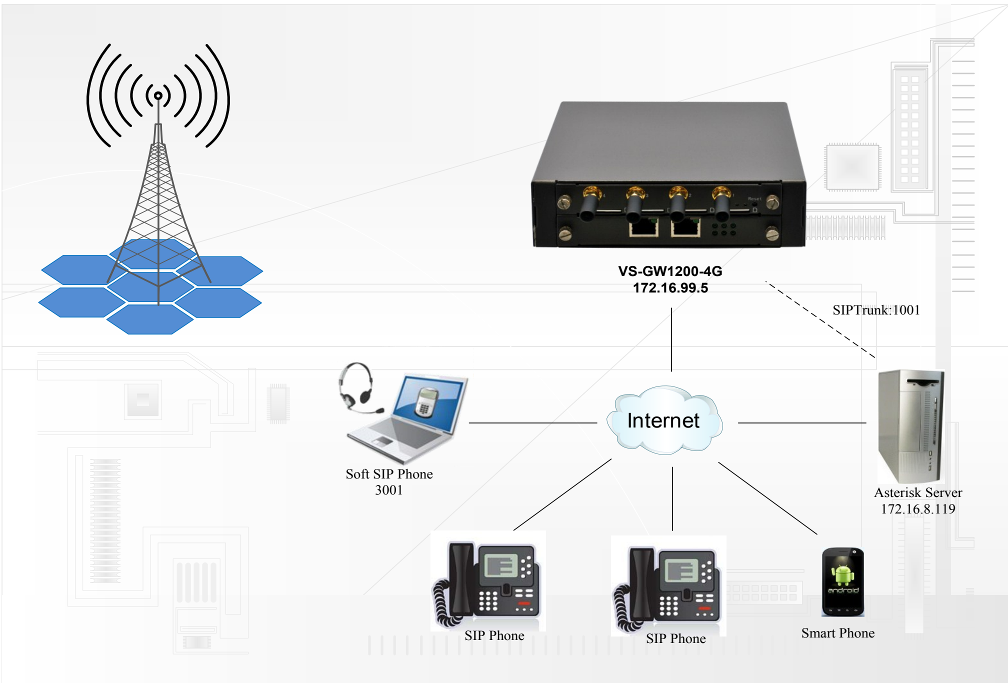
VoxStack provides 2 working modes: **Stand-alone** and **Cluster.**

⇒ Stand-alone: A single IP address manages one GSM modules (4 ports).

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

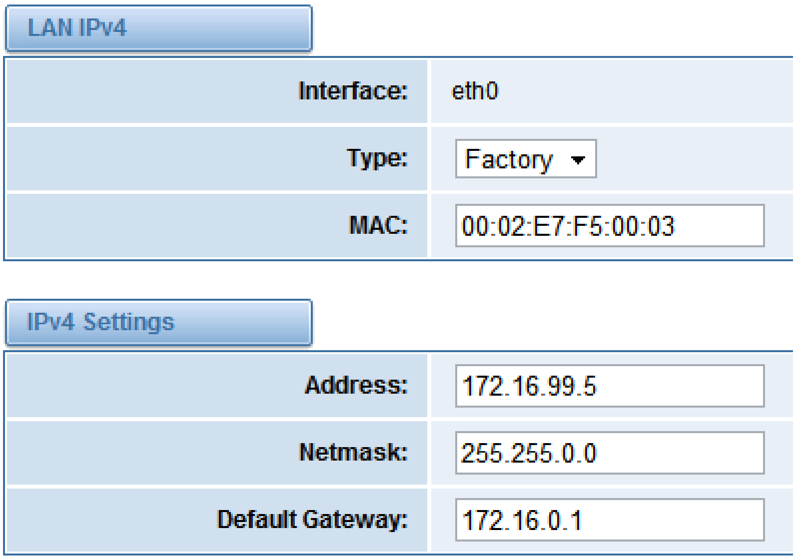
⇒ Cluster: A single IP address manages up to 5 GSM modules (up to 20 ports).

Default IP: 172.16.99.1
User Name: admin
Password: admin



Step 1. Set Network Parameters in Web

If your system topology like the figure described, please enter the gateway default IP address In your browser to login web, and click "NETWORK—>LAN Settings" to set network parameters such as IP.



LAN IPv4

Interface:	eth0
Type:	Factory
MAC:	00:02:E7:F5:00:03

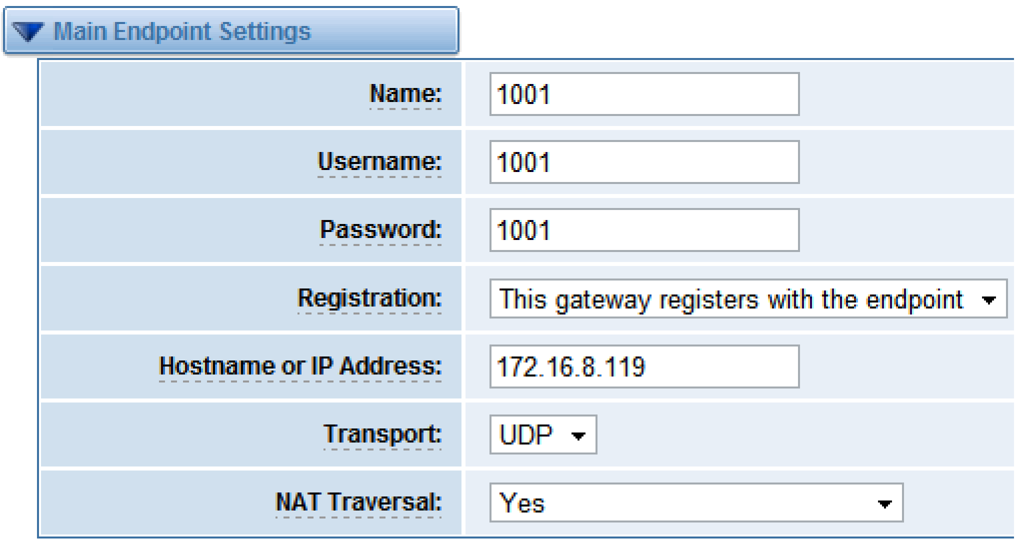
IPv4 Settings

Address:	172.16.99.5
Netmask:	255.255.0.0
Default Gateway:	172.16.0.1

Save your changes. Please type in your DNS server in "DNS Server Address".

Step 2. Create a SIP Endpoint in Web

Please select "SIP—>SIP Endpoints—>Add New SIP Endpoint" to set SIP trunk. The following figure shows detail information about how to set it.



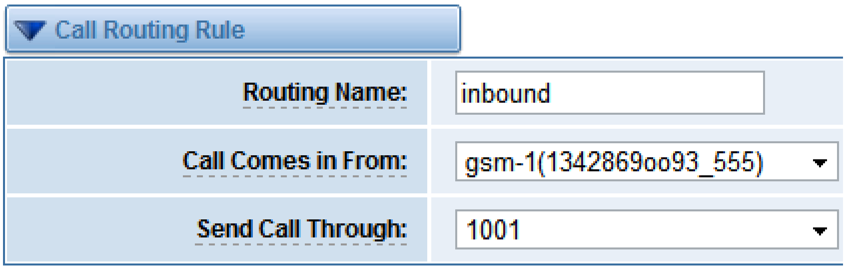
Main Endpoint Settings

Name:	1001
Username:	1001
Password:	1001
Registration:	This gateway registers with the endpoint
Hostname or IP Address:	172.16.8.119
Transport:	UDP
NAT Traversal:	Yes

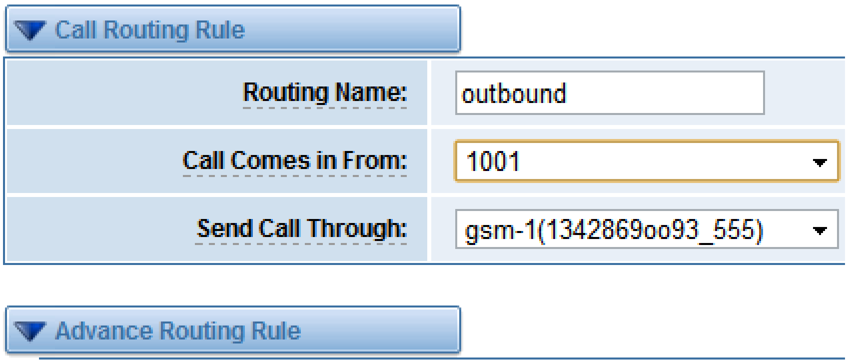
About other parameters in SIP, please set by your requirements for there is no need to set them in simple calls.

Step 3. Set Routing Rules in Web

Click “ROUTING—> Call Routing Rules—> New Call Routing Rule” to set outbound and inbound routing rules like the following:



Save the inbound call routing rules, please set the outbound rules as introduced. In order to make all calls successfully, please enable and set failover function in advanced routing rule like that:



Please save all your changes to make effect.

Step4. Create a SIP Trunk in Asterisk® Server

Please add the following lines in sip.conf to create a SIP trunk(1001):

```
[1001]
host=dynamic
username=1001
secret=1001
type=friend
fromuser=1001
context=from-gsm
```

After editing, save and exit and restart SIP service in Asterisk® Server

Step 5. Edit Dial Rules In Asterisk

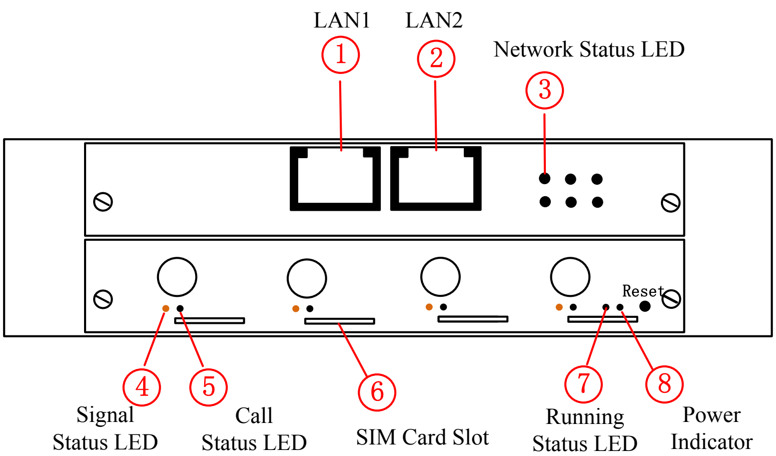
```
[from-internal]
exten => _9X.,1,Dial(sip/1001/${EXTEN:1})
exten => _9X.,n,Hangup()

[from-gsm]
exten => s,1,Dial(SIP/3001)
exten => s,n,Hangup()
```

Step 6. Register a SIP extension by software

Taking advantage of SIP software such as Xlite, eyeBeam to register a SIP extension(3001). After all above steps, you can try to make calls and send SMS.

Front Panel



③Network Status LED	Green and Flash	Network Connected
④Signal Status LED	Green and Flash	Module Initiating
	Red and Flash	No SIM Card
	Red and No-flash	Worst Signal Quality
	Yellow and No-flash	Medium Signal Quality
	Green and No-flash	Best Signal Quality
⑤Call Status LED	Flash	Communicating
	Blind	Normal
⑦Running Status LED	Green and Flash	Work Normally
⑧Power Indicator	Always Green	Supply Power
During reset, all LED indicators flash.		