



This document applies to OpenVox VS-GW2120 Series GSM Gateway. There are 3 RJ45 Network ports, ETH1/ETH2/ETH3. If you choose ETH1, you can access Board 1 . If you choose ETH2/ETH3, 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).

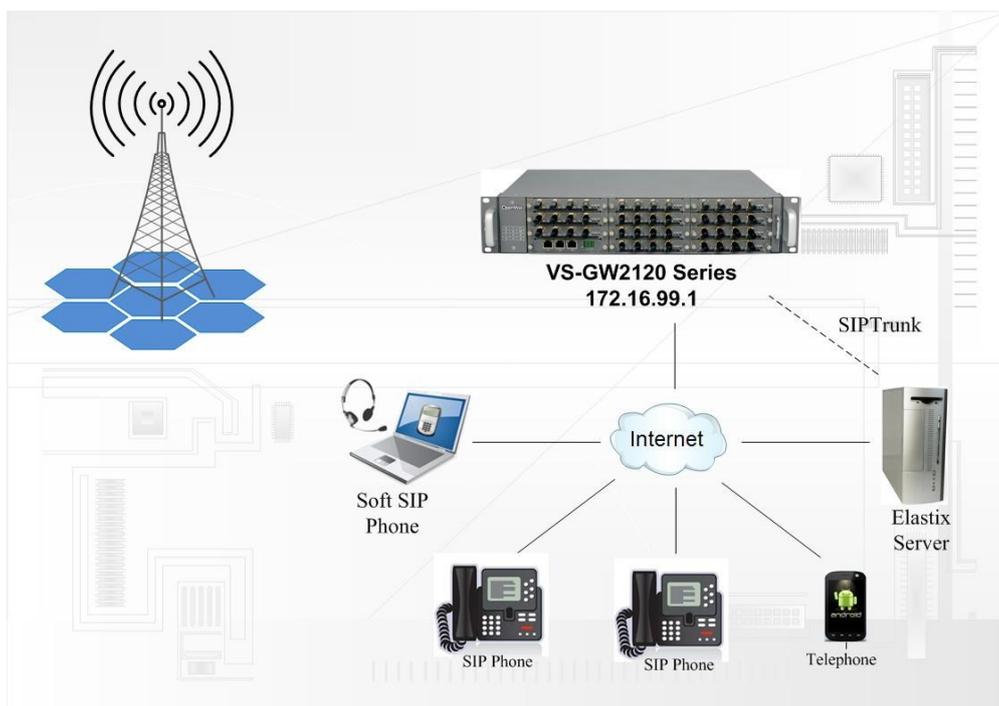
Stack Num	IP	Username	Password
1	172.16.99.1	admin	admin
2	172.16.99.2	admin	admin
3	172.16.99.3	admin	admin
...
11	172.16.99.11	admin	admin

⇒ Cluster: A single IP address manages up to 11 GSM modules (up to 44 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:

Call Routing Rule	
Routing Name:	inbound
Call Comes in From:	gsm-1(13428690093_555) ▼
Send Call Through:	1001 ▼

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:

Call Routing Rule	
Routing Name:	outbound
Call Comes in From:	1001 ▼
Send Call Through:	gsm-1(13428690093_555) ▼

Advance Routing Rule	
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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.