



GSM Gateway Connect with Asterisk® Server

QUICKSTART GUIDE

This document applies to OpenVox GSM Gateway VS-GW1200-4G and VS-GW1600(including 8G/12G/16G/20G). This is an example with 4G only. 8G/12G/16G/20G GSM Gateway will be automatically assigned to 2/3/4/5 different IPs in each 4 GSM channels. You can access each GW in corresponding IP from 172.16.99.1 to 172.16.99.5. Each 4 channels are separated, which is able to avoid crashing during the production.

Default IP: 172.16.99.1

Username: admin

Password : admin

Versions	Modules	IP Address
VS-GW1200-4G	1 * 4G	172.16.99.1
VS-GW1600-8G	2 * 4G	172.16.99.1-2
VS-GW1600-12G	3 * 4G	172.16.99.1-3
VS-GW1600-16G	4 * 4G	172.16.99.1-4
VS-GW1600-20G	5 * 4G	172.16.99.1-5

There are two LAN ports, please connect the gateway to Internet through one of LAN port and make sure connectivity by LED status.

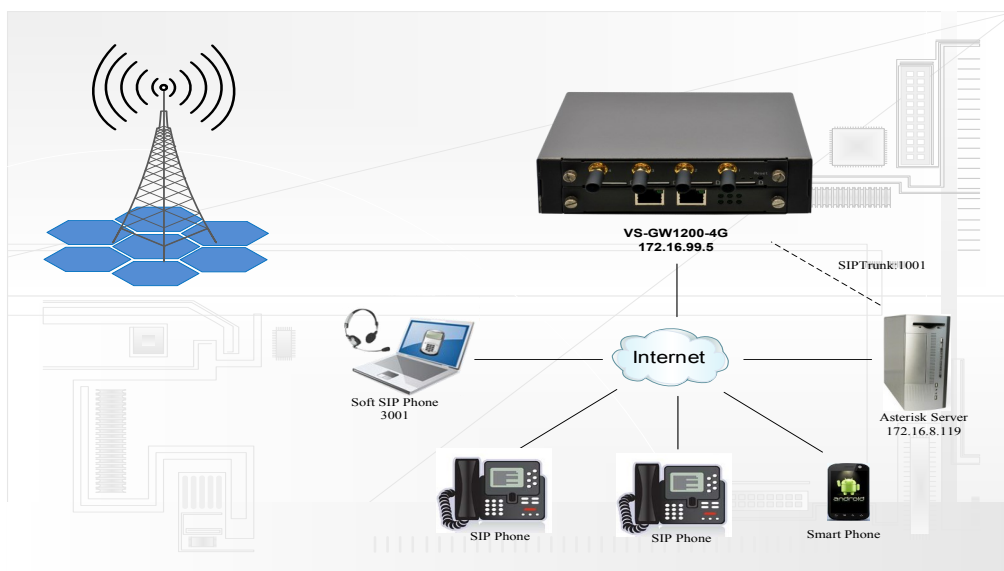
Configuration

⇒ Configure options in GUI:

1. Network parameters such as IP address;
2. SIP endpoint;
3. Routings;

⇒ Create a SIP trunk and dial rules in Asterisk server

⇒ Register SIP extensions





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	
Failover Call Through Number	
Failover Call Through Number 2	GSM-2 ▼
Failover Call Through Number 3	GSM-3 ▼
Failover Call Through Number 4	GSM-4 ▼

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()

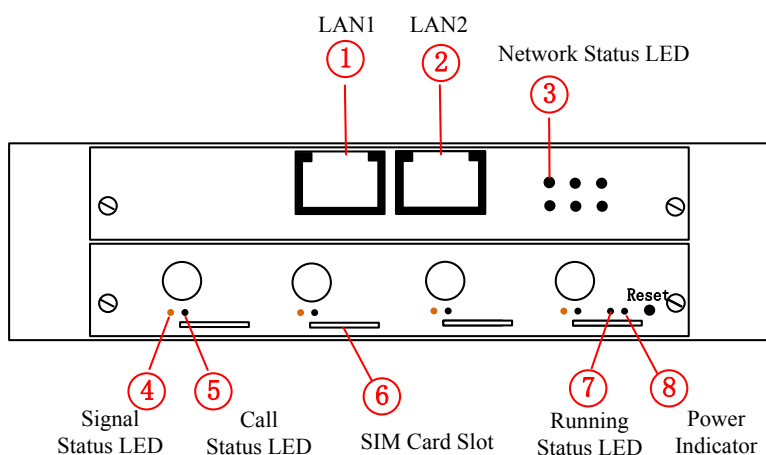
[from-gsm]
exten => s,1,Dial(SIP/3001)
exten => s,n,Hangup()
```



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



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