



GSM Gateway Connect with FreeSWITCH® Server QUICKSTART GUIDE

Default IP: 172.16.99.1

Username: admin

Password : admin

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

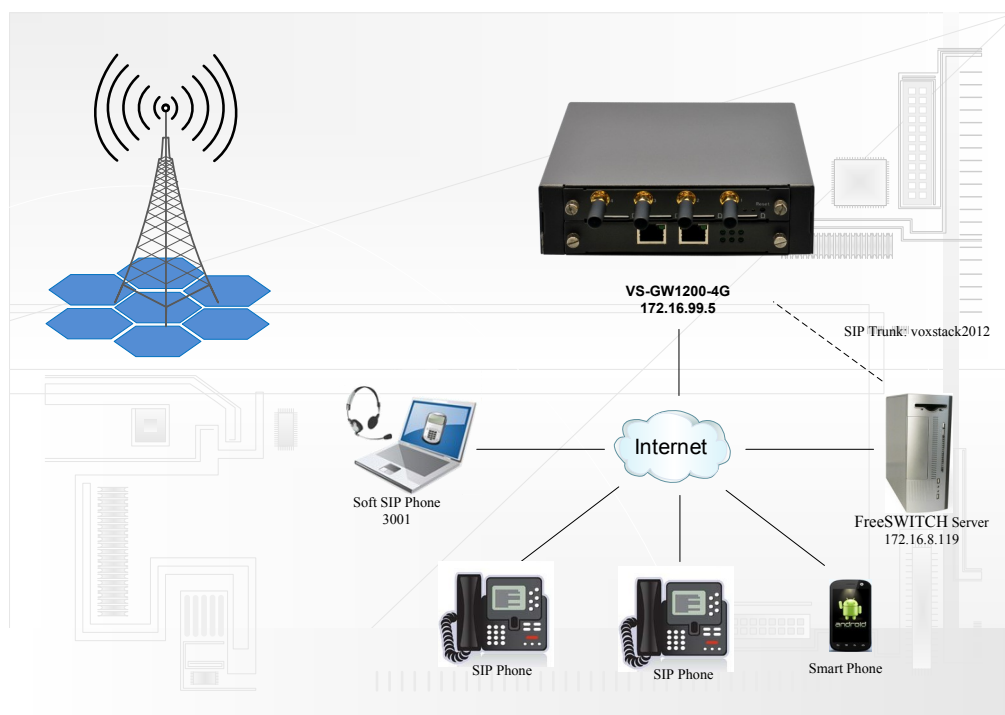
Configuration

⇒ Configure options in GUI:

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

⇒ Create a SIP trunk and edit dial rules in FreeSWITCH®

⇒ 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.

Add a New SIP Endpoint

Main Endpoint Settings	
Name:	voxstack2012
Username:	voxstack2012
Password:	2012
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:	voxstack2012 ▼

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:	voxstack2012 ▼
Send Call Through:	gsm-1(13428690093_555) ▼

Advance Routing Rule	
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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 FreeSWITCH®

Enter the directory of FreeSWITCH's default configuration directory, add the gateway's configuration in /usr/local/freeswitch/conf/directory/default/

```
vi /usr/local/freeswitch/conf/directory/default/voxstack.xml
```

```

<include>
<user id="voxstack2012">
<params>
<param name="password" value="2012"/>
<param name="vm-password" value="9999"/><!--if vm-password is omitted pass-
word param is used-->
</params>
<variables>
<!--all variables here will be set on all inbound calls that originate from
this user -->
<variable name="user_context" value="public"/>
<variable name="effective_caller_id_name" value="voxstack"/>
<variable name="effective_caller_id_number" value="2012"/>
<!-- Don't write a CDR if this is false valid values are: true, false,
a_leg and b_leg -->
<variable name="process_cdr" value="true"/>
</variables>
</user>
</include>

```



Step 5. Dialing Rules in FreeSWITCH®

Outbound rules realize dialing “9+destination number” to the remote party, and 9 can be replaced by any other digital. Edit the outband dialplan in /usr/local/freeswitch/conf/dialplan/default.xml

```

<extension name="voxstack2012_gateway">
  <condition field="destination_number" expression="^9(\d+)$">
    <action application="answer"/>
    <action application="set" data="ringback=${us-ring}"/>
    <action application="bridge" data="sofia/internal/$1@172.16.99.5"/>
  </condition>
</extension>

```

Inbound rules realize all incoming calls transfer to SIP extension 3001.

Edit the outband dialplan in /usr/local/freeswitch/conf/dialplan/public/00_inbound_did.xml.

```

<include>
  <extension name="public_did">
    <condition field="destination_number" expression="^(.)$">
      <action application="set" data="domain_name=${domain}"/>
      <action application="transfer" data="3001 XML default"/>
    </condition>
  </extension>
</include>

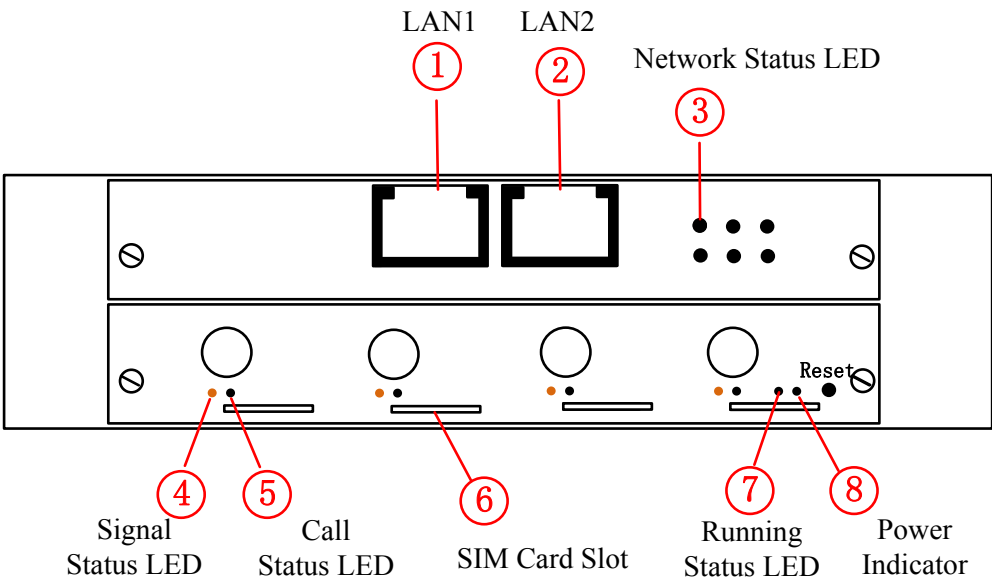
```



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
	Green and Flash	Module Initiating
	Red and Flash	No SIM Card
④Signal Status LED	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.		