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.

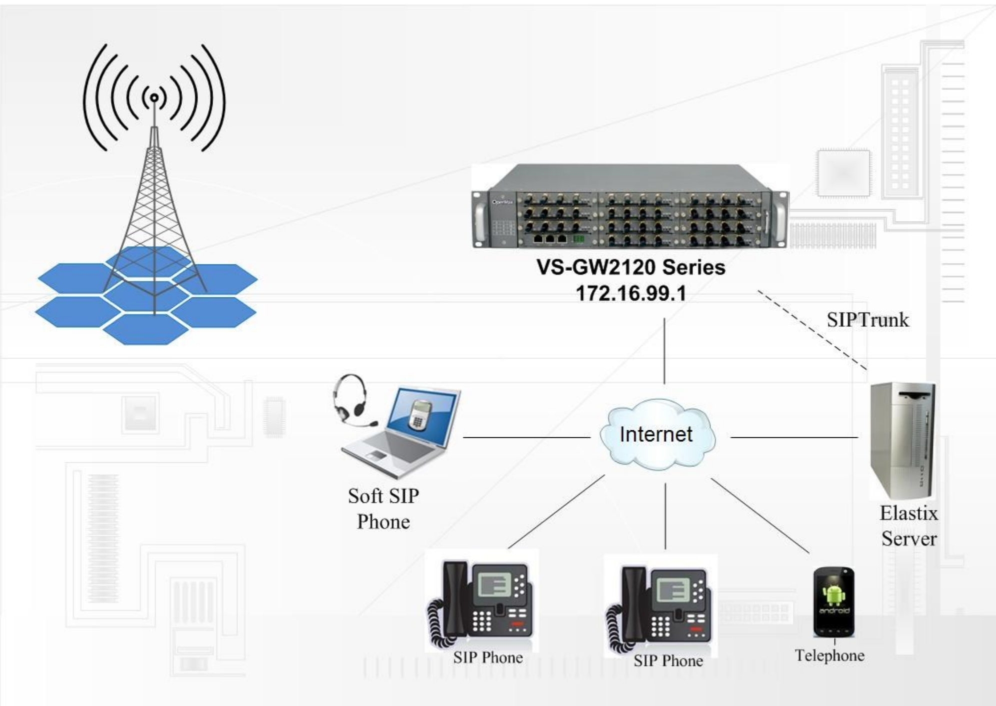
VoxStack provides 2 working modes: **Stand-alone** and **Cluster.**

⇒ 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.

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(1342869oo93_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(1342869oo93_555)

Advance Routing Rule

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.