This document applies to OpenVox GSM Gateway WGW1002, VS-GW1202-4/8G 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. But there is only one RJ45 Network port on WGW1002.

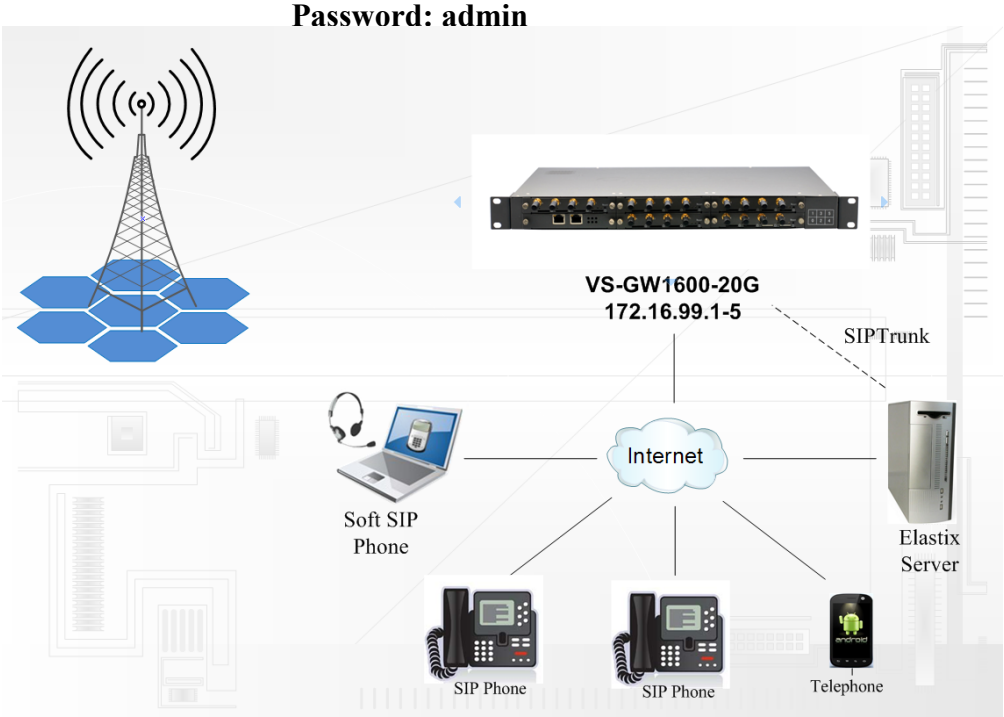
VoxStack provides 2 working modes: **Stand-alone** and **Cluster.**

⇒ 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

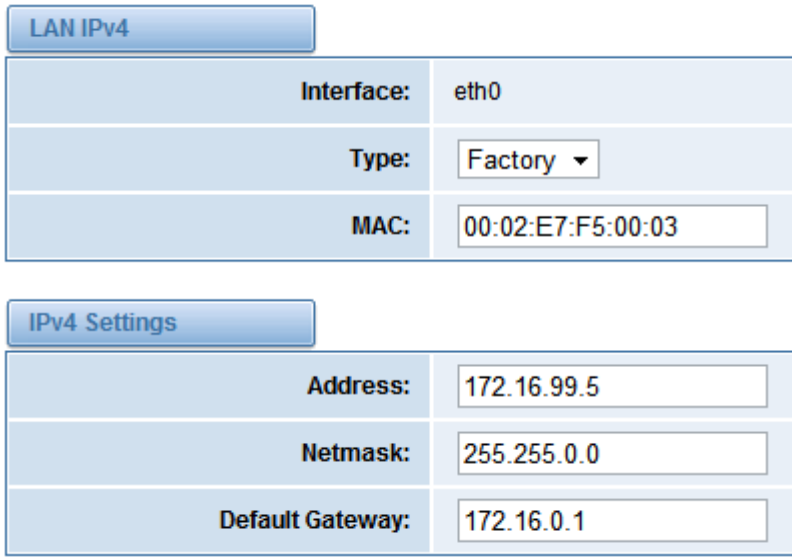
⇒ Cluster: A single IP address manages up to 5 GSM modules (up to 20 ports).

Default IP: 172.16.99.1
User Name: admin
Password: admin



Step 1. Set Network Parameters in Web

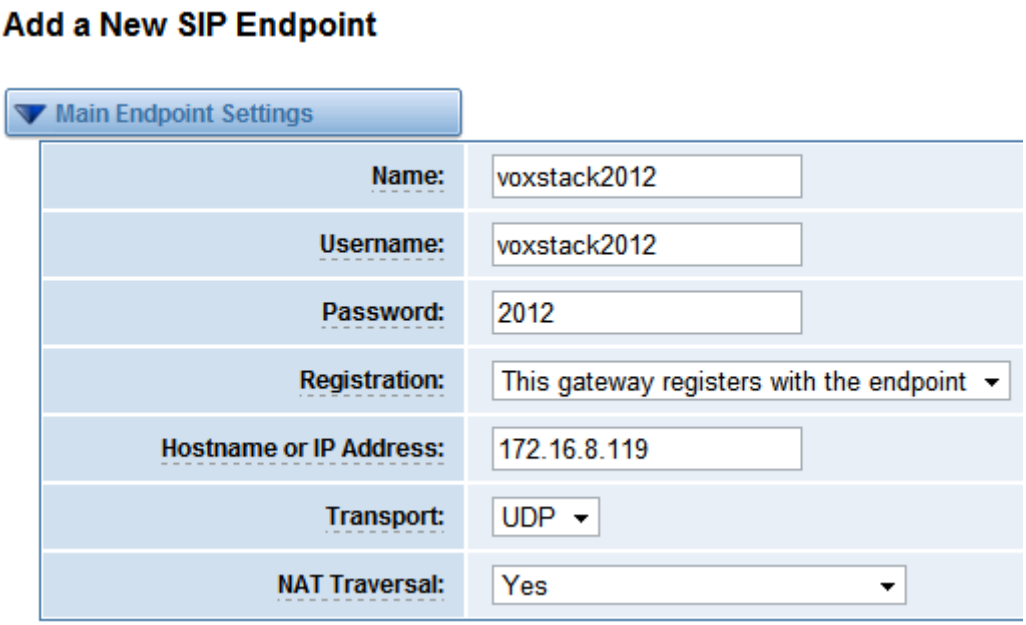
If your system topology like the figure described, please enter the gateway default IP address in your browser to login web, and click "NETWORK—>LAN Settings" to set network parameters such as IP.



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

Step 2. Create a SIP Endpoint in Web

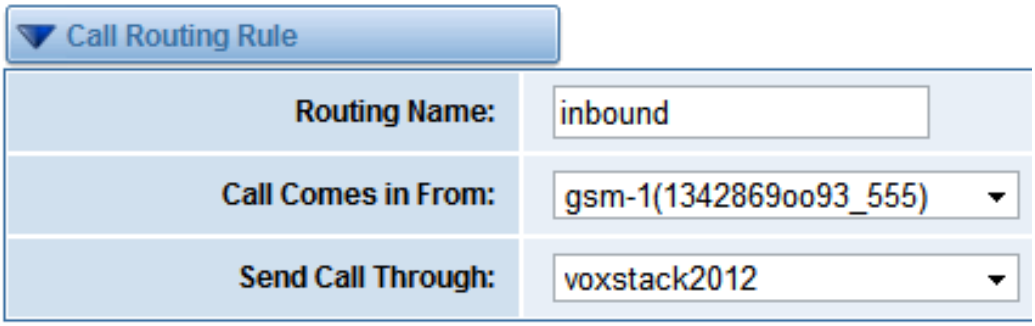
Please select "SIP—>SIP Endpoints—>Add New SIP Endpoint" to set SIP trunk. The following figure shows detail information about how to set it.



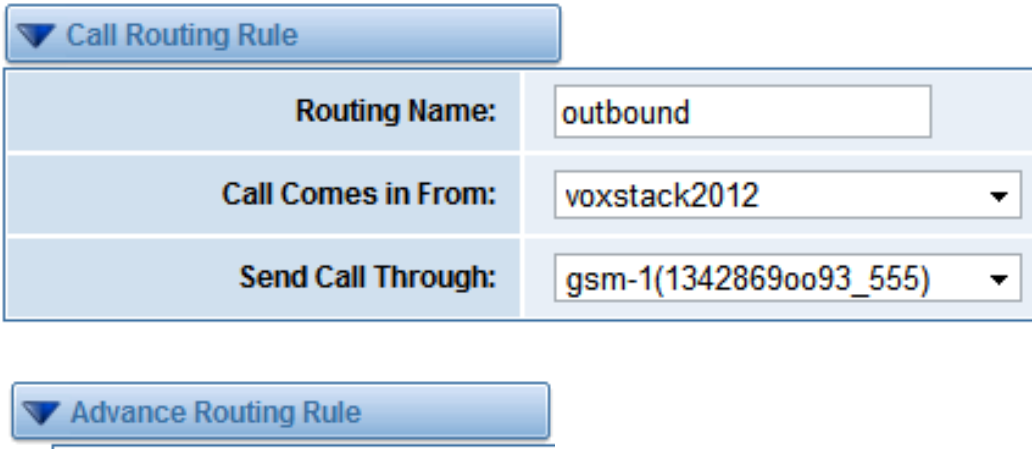
About other parameters in SIP, please set by your requirements for there is no need to set them in simple calls.

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 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>
```

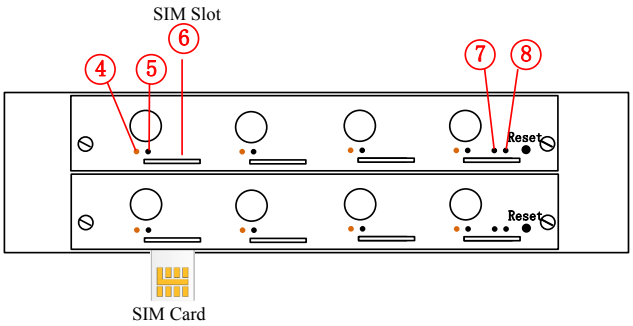
Inbound rules realize all incoming calls transfer to SIP extension 3001.
Edit the outband dialplan in /usr/local/freeswitch/conf/dialplan/public/00_inbound_did.xml.

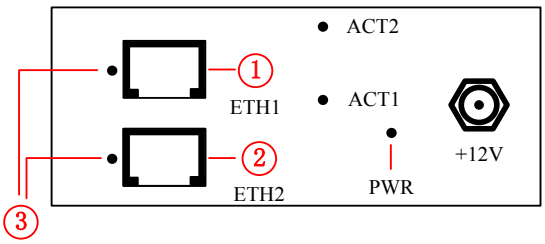
```
<include>
        <extension name="public_did">
   <condition field="destination_number" expression="^(.+)$">
        <action application="set" data="domain_name=$${domain}"/>
        <action application="transfer" data="3001 XML default"/>
   </condition>
</extension>
</include>
```

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





LED Indicator	Color	Status
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 (0.25s)	Communicating
	Blind	Normal
⑦Running Status LED	Green and Flash(0.5s)	Work Normally
⑧Power Indicator	Always Green	Supply Power
During reset, all LED indicators flash.		