GSM Gateway Connect with
3CX® Server
QUICKSTART GUIDE

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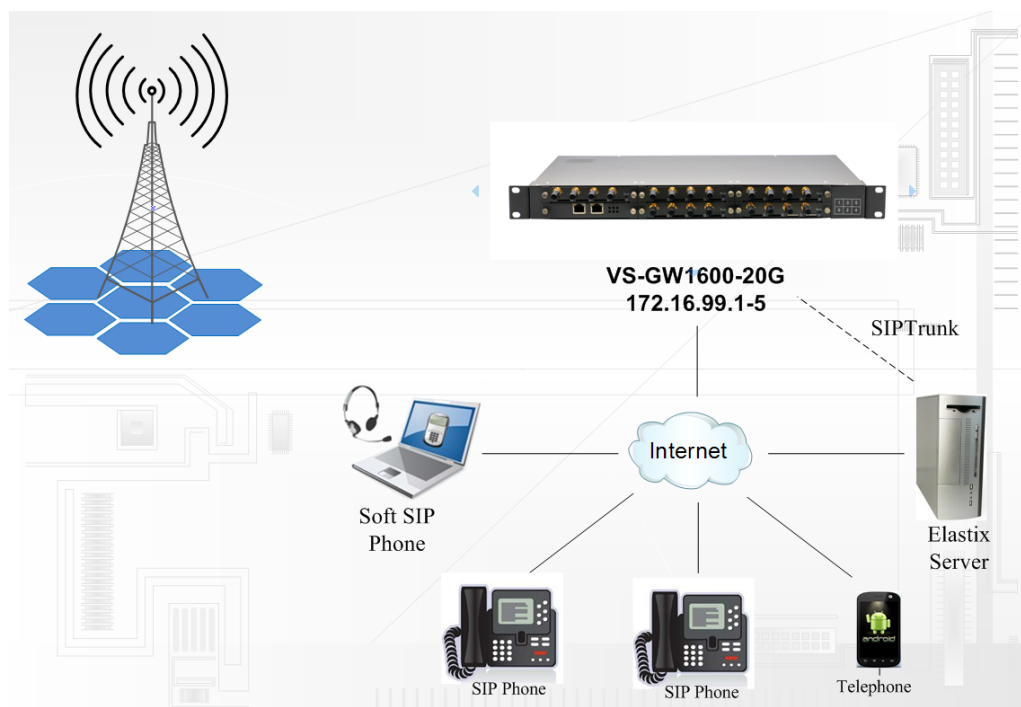
VoxStack provides 2 working modes: **Stand-alone** and **Cluster.**

⇒ 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.

LAN IPv4	
Interface:	eth0
Connection Type:	Static ▾
MAC:	00:56:64:75:7a:52

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:	10001
Username:	10001
Password:	10001
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(13428690o93_555) ▼
Send Call Through:	10001 ▼

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:	10001 ▼
Send Call Through:	gsm-1(13428690o93_555) ▼

Please save all your changes to make effect.

Step4. Create a SIP Trunk in 3CX web

Please select "VOIP/PSTN Gateways—> Add Gateway" to create a SIP trunk:

3CX

VOIP/PSTN Gateways

Add Gateway   Edit Gateway   Delete Gateway   Refresh Registration

Gateway Name	Host / IP Address	Type

▶ 3CX Phone System
   Ports/Trunks Status
   Extension Status
   System Extensions Status
   3CX MyPhone Clients
   Remote Connections
   Phones
   Server Activity Log
   Server Event Log
   Services status
▷ Extensions
▷ VOIP/PSTN Gateways

After that, please click the next button to the following page :

About other options, you can set or skip by your requirements until go to the following web which means SIP trunk is registered.

Step 5. Dial rules in 3CX web

Click Inbound Rules—>Add DID to set inbound rules

Add DID

Route calls to DID/DDI numbers directly to an extension

DID/DDI Name

Enter a DID or string to look for in the SIP "to" field. Use wildcards (*) to match any digit for that entry. Fo calls with a dialled number of +35722444032 in the "to" field

DID/DDI Name   10001_IN

DID/DDI number/mask

Select from the drop-down below the type of inbound rule you want to create and enter a mask for this D your mask.

Inbound Rule type   DID/DDI number/mask ▼

DID/DDI number/mask   10001

Apply this rule to these ports

Select the Gateway you want this DID/DDI rule to be applied to. You can select on the whole gateway wh individual ports.

Available ports   ▷ ☐ ⬒ VoxStack
                         ▷ ☑ ⬒ 1001

Office Hours

Configure where calls to this DID/DDI should be routed during office hours.

○ End Call

◉ Connect to Extension   301 voxstack openvox ▼

Click Outbound Rules—>Add Outbound to set outbound rules

OutBound Rules

Create an Outbound Call Rule to configure on which PSTN port, VOIP provider or bridge an outbound calls should be placed on

General

Rule Name   10001_out

Apply this rule to these calls

Define to which outbound calls the rule must apply

Calls to numbers starting with (Prefix)   9

Calls from extension(s)   100-999

Calls to Numbers with a length of   4-12

Calls from extension group   ...   [Select]

Make outbound calls on

Configure up to 3 routes for calls. The second and third route will be used as backup. For each route, digits can be stripped or added.
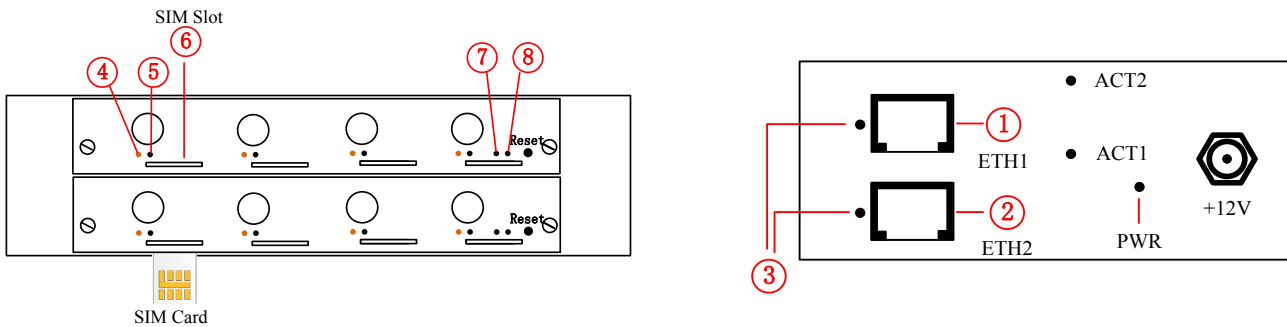
			Strip Digits	Prepend
Route	1	1001 ▼	1 ▼	

Step 6. Register a SIP extension by software

Taking advantage of SIP software such as Xlite, eyeBeam to register a SIP extension(301). 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.		