

GSM Gateway Connect with 3CX[®] Server QUICKSTART GUIDE



This document applies to OpenVox GSM Gateway WGW1002G, 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 WGW1002G.

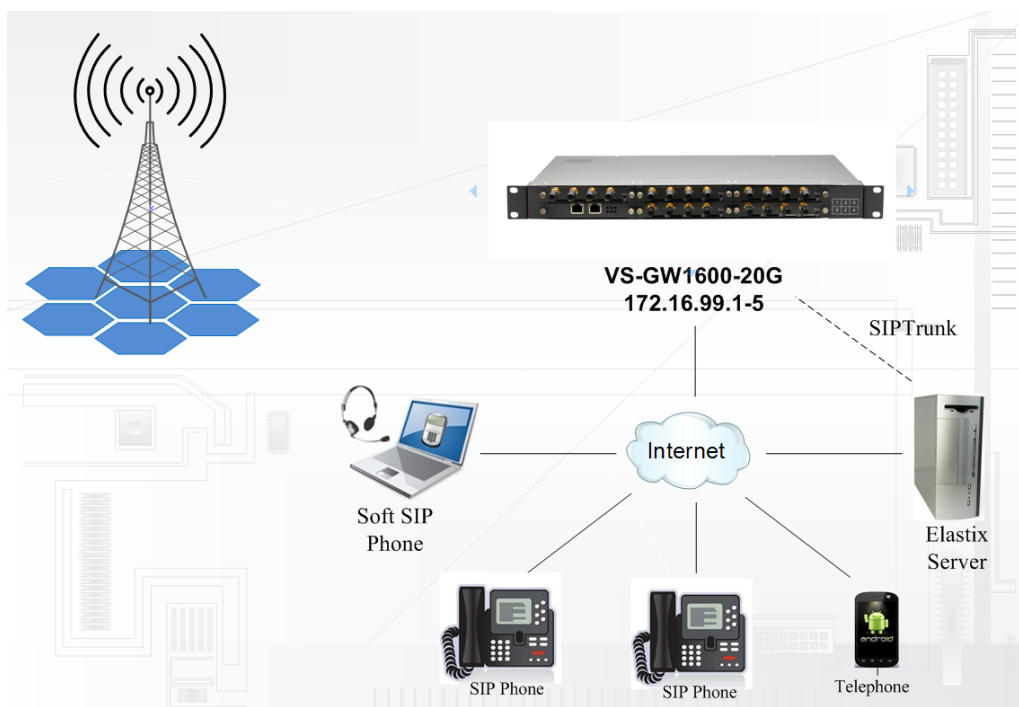
VoxStack provides 2 working modes: **Stand-alone** and **Cluster**.

⇒ 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(1342869oo93_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(1342869oo93_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:

The screenshot shows the 3CX web interface. On the left is a navigation menu with 'VOIP/PSTN Gateways' selected. On the right, the 'VOIP/PSTN Gateways' page is displayed, featuring buttons for 'Add Gateway', 'Edit Gateway', 'Delete Gateway', and 'Refresh Registration'. Below these buttons is a table with columns for 'Gateway Name', 'Host / IP Address', and 'Type'.

VOIP/PSTN Gateways

Add Gateway Wizard

Add PSTN Gateway

Name	<input type="text" value="1001"/>	
Brand	<input type="text" value="Generic"/>	
Model	<input type="text" value="Gateway Device"/>	
Description	Custom Generic Gateway Device	
URL	http://www.3cx.com	
More vendor supported gateways can be found here: http://www.3cx.com/voip-gateways/index.html		

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

VOIP/PSTN Gateways

Specify VoIP Gateway Details

VoIP Gateway

Gateway Hostname or IP	<input type="text" value="172.16.99.5"/>	
Gateway Port (default is 5060)	<input type="text" value="5060"/>	
Number of ports	<input type="text" value="1"/>	
Type	<input type="text" value="Custom"/>	
Number of channels per port	<input type="text" value="4"/>	

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

VOIP/PSTN Gateways

VoIP Gateway Created

Ports 10001 to 10001 have been created for 1001

You can find information on how to configure and provision your VoIP Gateway device at <http://www.3cx.com/blog/suppo>

The settings below are required to configure the VoIP Gateway manually

Access the VoIP Gateway web portal at:
<http://172.16.99.5>

Proxy server / SIP server / registrar: 172.16.8.148:5060

SIP User ID: 10001
Authentication ID: 10001
Authentication Password: mz1wd0j



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. For calls with a dialled number of +35722444032 in the "to" field

DID/DDI Name

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 DID/DDI number/mask.

Inbound Rule type

DID/DDI number/mask

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 with 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

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

Apply this rule to these calls
 Define to which outbound calls the rule must apply

Calls to numbers starting with (Prefix)

Calls from extension(s)

Calls to Numbers with a length of

Calls from extension group

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.

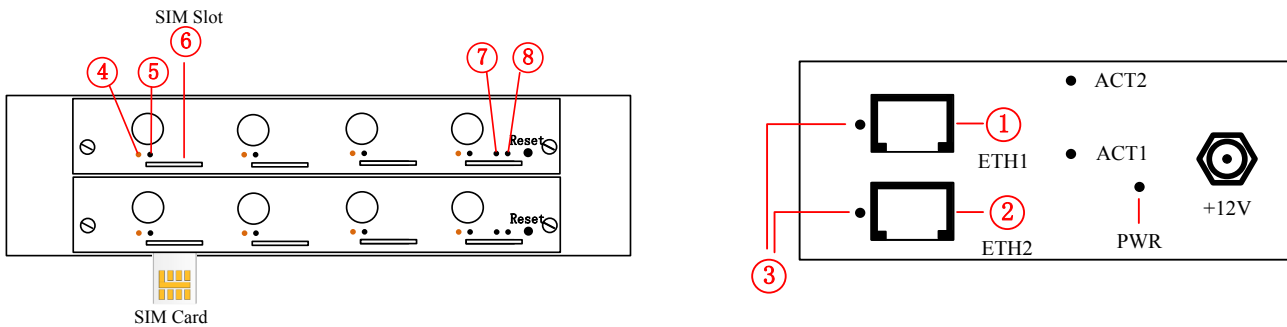
Route	1	<input type="text" value="1001"/>	Strip Digits	<input type="text" value="1"/>	Prepend	<input type="text"/>	
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Step 6. Register a SIP extension by software

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
③ Network Status LED	Green and Flash	Network Connected
④ Signal 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
⑤ Call Status LED	Flash (0.25s)	Communicating
	Blind	Normal
⑦ Running Status LED	Green and Flash(0.5s)	Work Normally
⑧ Power Indicator	Always Green	Supply Power
During reset, all LED indicators flash.		