



# GSM Gateway Connect with 3CX<sup>®</sup> Server

## QUICKSTART GUIDE

**Default IP: 172.16.99.1**

**Username: admin**

**Password : admin**

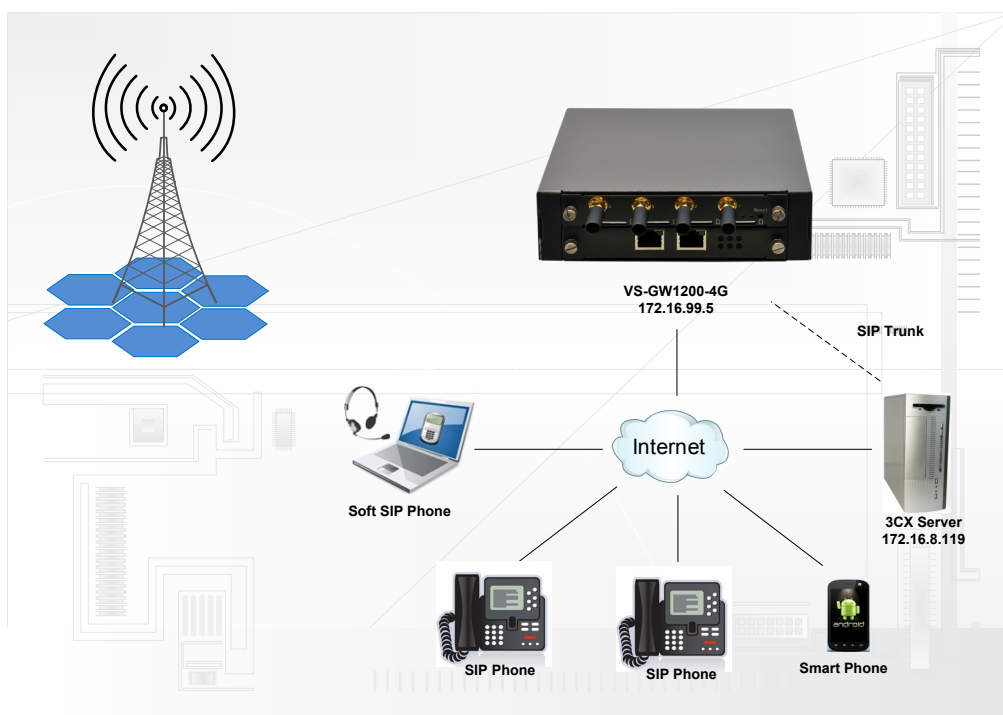
There are two LAN ports, please connect the gateway to Internet through one of LAN port and make sure connectivity by LED status.

### Configuration

⇒ Configure options in GUI:

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

⇒ Create a SIP trunk, and configure VoIP gateway in 3CX server  
⇒ 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
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(13428690093_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(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 3CX web

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

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

VOIP/PSTN Gateways

Add Gateway
Edit Gateway
Delete Gateway
Refresh Registration

Gateway Name	Host / IP Address	Type
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**VOIP/PSTN Gateways**


 Add Gateway Wizard

Add PSTN Gateway
 

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

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	<div>Custom</div>	?
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

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.

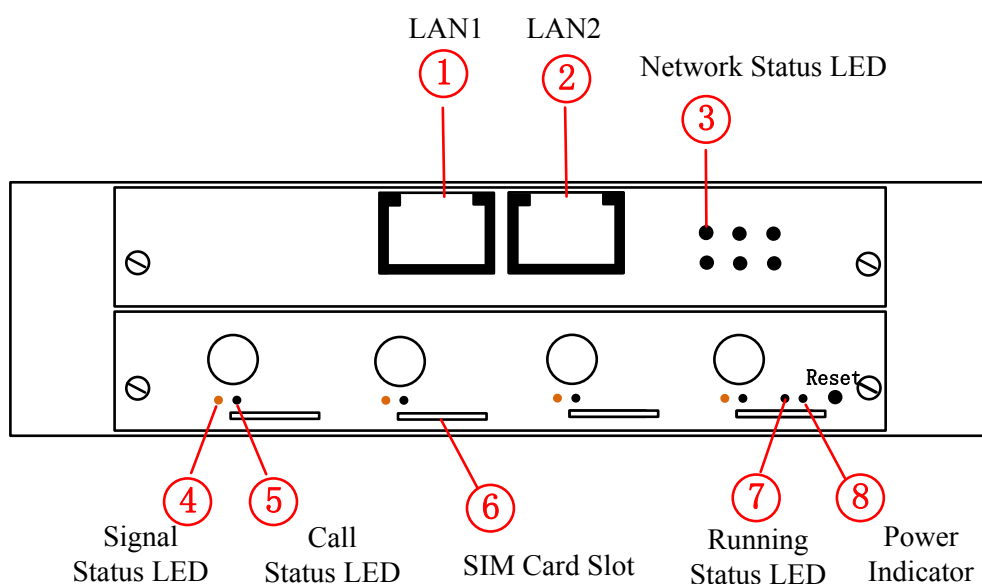
Route		Strip Digits	Prepend
1	<input type="text" value="1001"/>	<input type="text" value="1"/>	<input type="text" value=""/>



## 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



③ Network Status LED	Green and Flash	Network Connected
	Green and Flash	Module Initiating
	Red and Flash	No SIM Card
④ Signal Status LED	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	Communicating
	Blind	Normal
⑦ Running Status LED	Green and Flash	Work Normally
⑧ Power Indicator	Always Green	Supply Power
During reset, all LED indicators flash.		