

How to bind the gsm gateway ports with PBX sip extension

Step 1: enter the gateway via web and establish a sip trunk connecting to the PBX.

The screenshot shows the VoxStack web interface. The top navigation bar includes 'SYSTEM', 'GSM', 'SIP', 'ROUTING', and 'SM'. The 'SIP' menu is highlighted, and a sub-menu shows 'SIP Endpoints' and 'Advanced SIP Settings'. The main content area is titled 'Edit SIP Endpoint "GsmRouter"'. Under the 'Main Endpoint Settings' section, there is a form with the following fields:

Name:	GsmRouter
User Name:	gsm <input type="checkbox"/> Anonymous
Password:	...
Registration:	None
Hostname or IP Address:	172.16.8.164 → the IP or hostname of PBX
Transport:	UDP
NAT Traversal:	Yes

Below the form are sections for 'Advanced:Registration Options' and 'Call Settings'. At the bottom are 'Save', 'Apply', and 'Cancel' buttons.

Warning: the registration have three modes from which you can choose one: none, this gateway register with the endpoint, the endpoint register with this gateway.

Step 2: Open the PBX via web and create a sip trunk connecting to the gateway.

The screenshot shows the PBX web interface for 'Outgoing Settings'. The 'Trunk Name' is 'gsm'. The 'PEER Details' section contains the following configuration:

```
host=172.16.33.55
username=gsm
secret=gsm
type=friend
context=from-internal
```

The 'context=from-internal' line is highlighted with a red box. The word 'or' appears to the right of the box.

Outgoing Settings

Trunk Name:

PEER Details:

```
host=172.16.33.55 → gateway IP
username=gsm
secret=gsm
type=peer
```

Incoming Settings

USER Context:

USER Details:

```
host=dynamic
username=gsm
secret=gsm
type=user
context=from-internal
```

Step 3: create sip extensions for the PBX.

Panel | Voicemail | Monitoring | Batch Configurations | Conference | Tools | Flash Operator Panel | VoIP Provider

Add an Extension

Please select your Device below then click Submit

Device:

Submit

- Add Extension
- 16400 <16400>
- 16401 <16401>
- gsm-1 <16466372>
- gsm-4 <16466375>
- gsm-5 <16466376>
- gsm-6 <16466377>
- gsm-7 <16466378>
- gsm-8 <16466379>

Step 4: go to the gateway and create the routers:

VoxStack | SYSTEM | GSM | SIP | **ROUTING** | SMS | NETWORK | ADVANCED | LOGS

Call Routing Rules | Groups | MNP Settings

Free Communication

Move	Order	Rule Name	From	To	Rules	Actions
	1	out-1.1	sip-GsmRouter	gsm-1.1	Dial_pattern (+)[/16466372]	
	2	out-1.4	sip-GsmRouter	gsm-1.4	Dial_pattern (+)[/16466375]	
	3	out-1.5	sip-GsmRouter	gsm-1.5	Dial_pattern (+)[/16466376]	
	4	in-1.6	gsm-1.6	sip-GsmRouter		
	5	in-1.7	gsm-1.7	sip-GsmRouter		
	6	in-1.8	gsm-1.8	sip-GsmRouter		

(1) bind the gateway port when you call out. Please set the outgoing router like picture below.

VoxStack WIRELESS GATEWAY

SYSTEM | GSM | SIP | ROUTING | SMS | NETWORK | ADVANCED |

SMS Settings | SMS Sender | SMS Inbox | SMS Outbox

ROUTING DETAILS

Free Communication OpenVox

Modify a Call Routing Rule

Call Routing Rule

Routing Name: out-1.1

Call Comes in From: GsmRouter

Send Call Through: gsm-1.1

Advance Routing Rule

Dial Patterns that will use this Route

<prepend> + prefix | match pattern | 16466372

+ Add More Dial Pattern Fields

Notice: only the sip extension 16466372 coming from the trunk GsmRouter can use this route. In this way the call coming from sip extension 16466372 will be sent via gsm-1.1 port.

(2) bind the gateway port when you call in. Please set the incoming route like below.

Modify a Call Routing Rule

Call Routing Rule

Routing Name: in-1.6

Call Comes in From: gsm-1.6

Send Call Through: GsmRouter

Advance Routing Rule

Dial Patterns that will use this Route

<prepend> + prefix | match pattern | Callerid

+ Add More Dial Pattern Fields

Time Patterns that will use this Route

Time to start: - : - Week Day start: - Month Day start: -

Time to finish: - : - Week Day finish: - Month Day finish: -

+ Add More Time Pattern Fields

Change Rules

Set the Caller ID Name to

Set the Caller ID Number to

Forward Number 16466377

the sip extension you want to bind with gsm-1.6 port

Notice: in this way, when you call the gsm-1.6, the call will be sent to sip extension 16466377 via sip trunk GsmRouter.

Step 5. make a call to test it