This document applies to OpenVox VS-GW2120 Series GSM Gateway. There are 3 RJ45 Network ports, ETH1/ETH2/ETH3. If you choose ETH1, you can access Board 1 . If you choose ETH2/ETH3, you can access different Boards with different IP addresses.

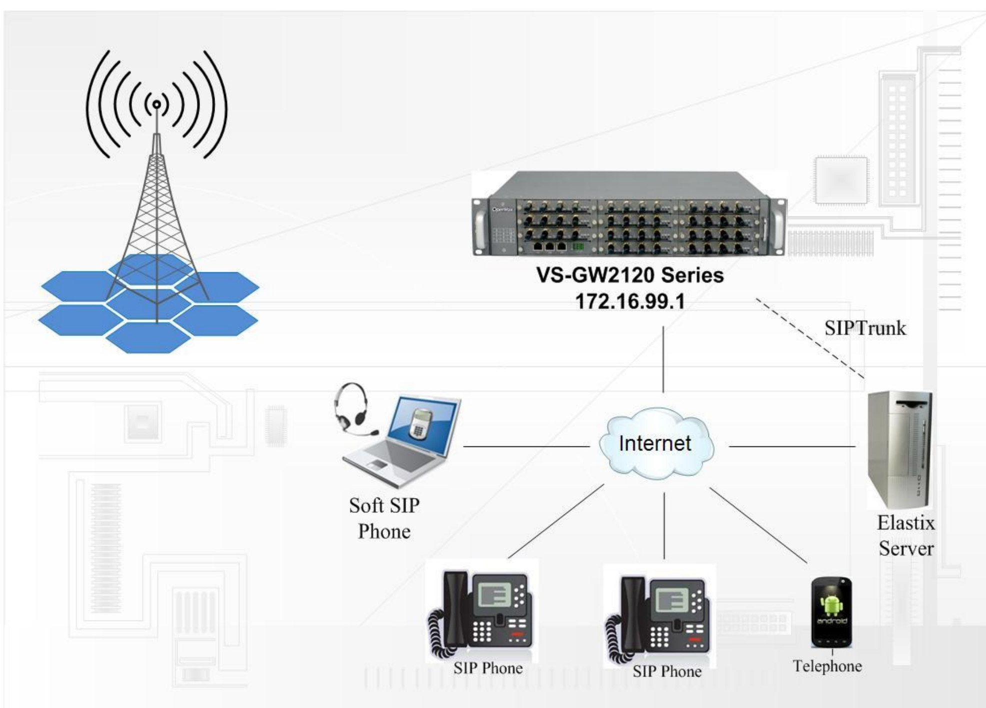
VoxStack provides 2 working modes: **Stand-alone** and **Cluster.**

⇒ 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
	...	...	...
11	172.16.99.11	admin	admin

⇒ Cluster: A single IP address manages up to 5 GSM modules (up to 44 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)

Advance Routing Rule

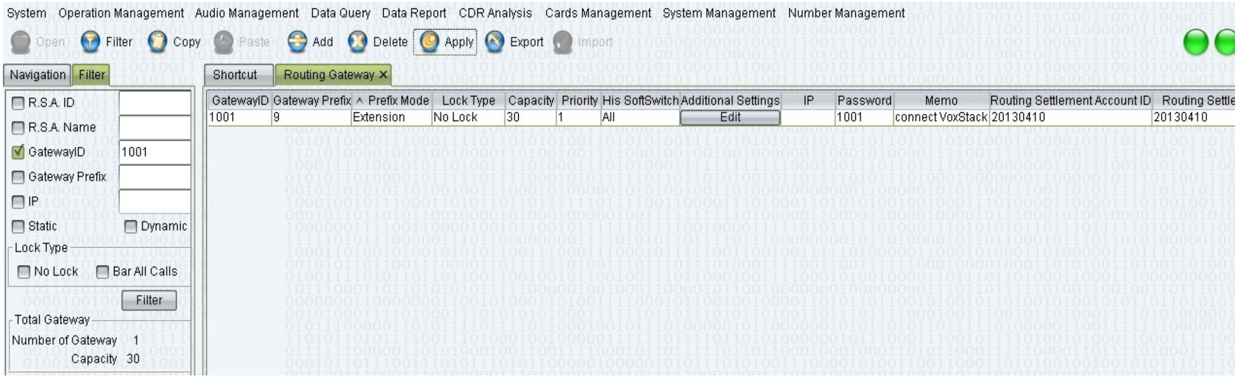
Please save all your changes to make effect.

Step4. Create Gateways in VOS3000 Client

There are 2 kinds of Gateway in VOS3000 Operation Platform: Routing Gateway and Mapping Gateway. Routing Gateway is for VOS3000 to PSTN while Mapping Gateway is opposite. If you need both outbound and inbound calls from GSM gateway, then you need create both Routing Gateway and Maping Gateway to make it happen.

Section 1: Add a Routing Gateway in register mode:

Please select “Operation Management---> Gateway Operation-----> Routing Gateway" to create a Routing Gateway:



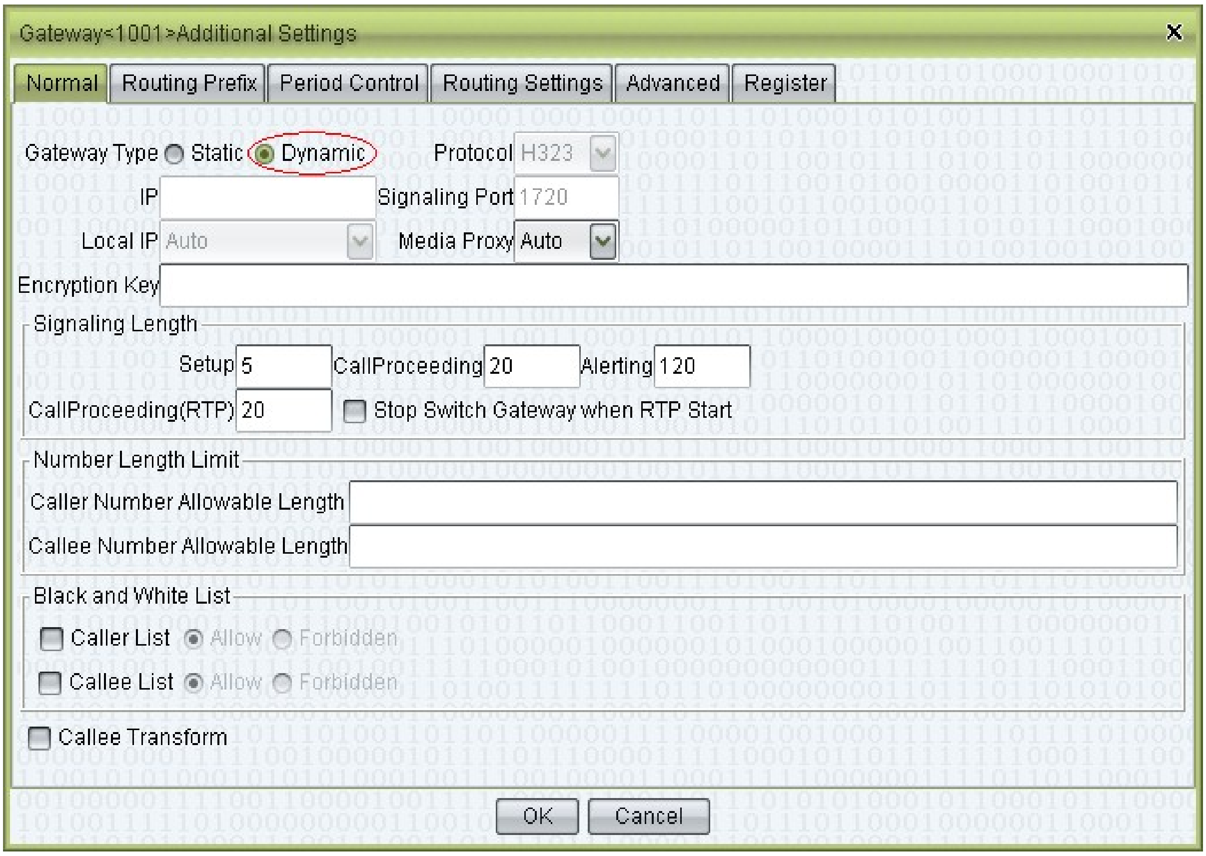
Gateway ID:1001 ---------------------Username of SIP trunk which GSM gateway registers to

Password:1001 ---------------------Password of SIP trunk which GSM gateway registers to

Gateway Prefix:9 ---------------------Specify calleeID 9+number goes out from GSM gateway

Additonal Settings:

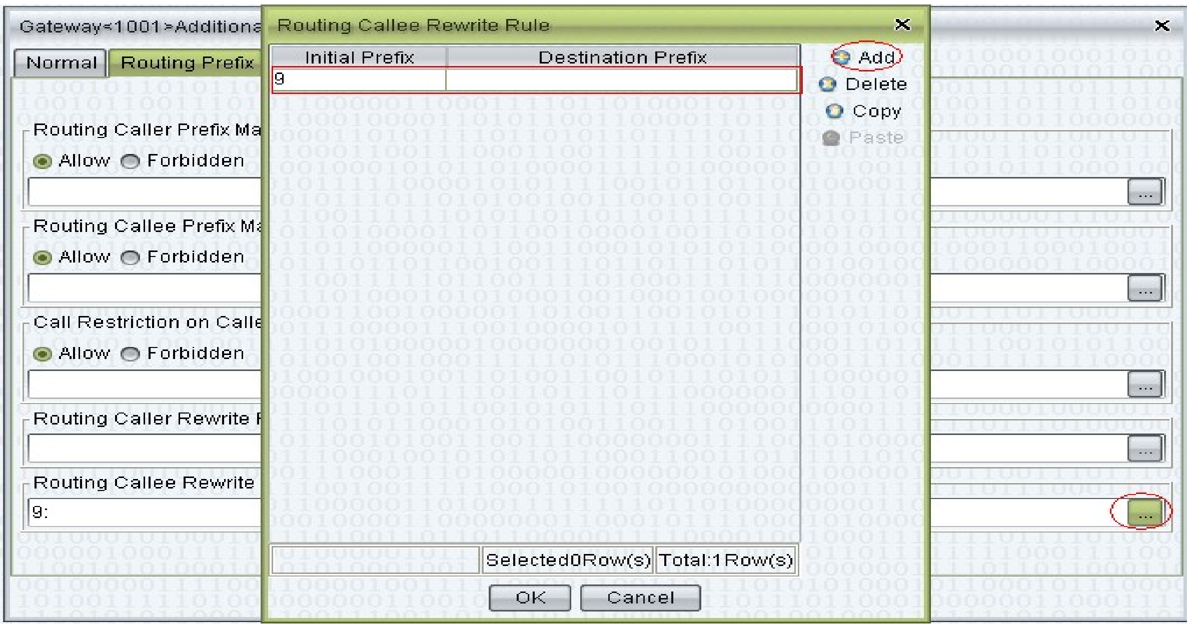
●Normal---->Gateway Type---->Dynamic



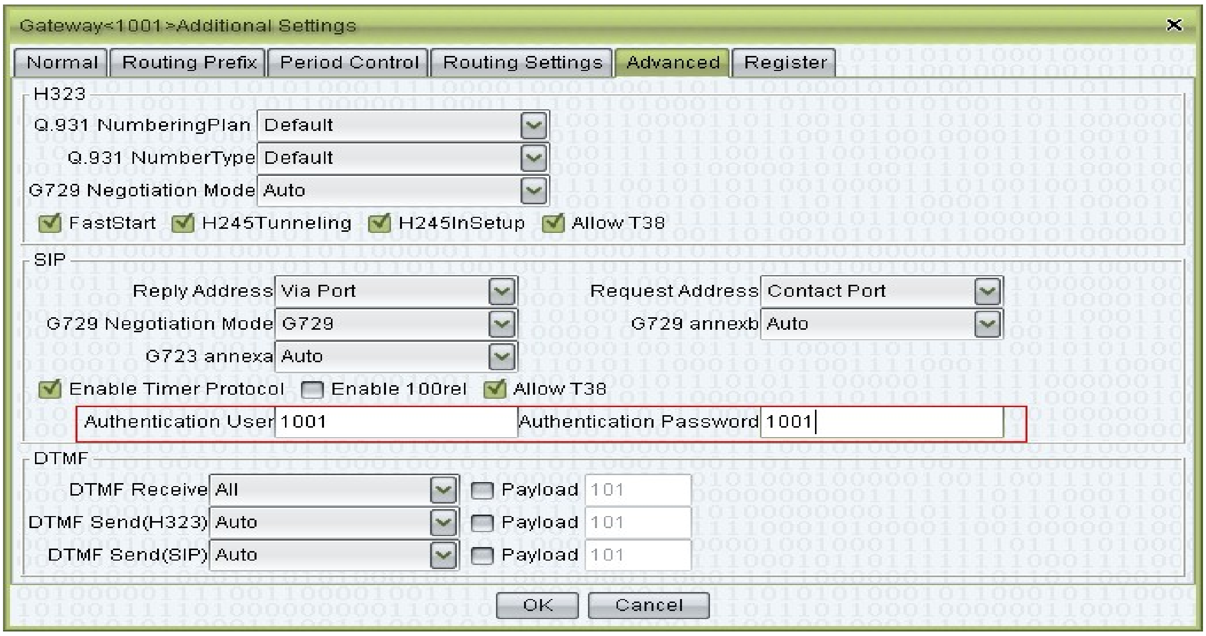
VOS3000 here acts as a SIP server, OpenVox GSM Gateway as a SIP client registers to it.

●Routing Prefix---->Routing Callee Rewrite

Add a Routing callee Rewrite Rule to remove the prefix '9' of CalleeID, then send it to GSM gateway.

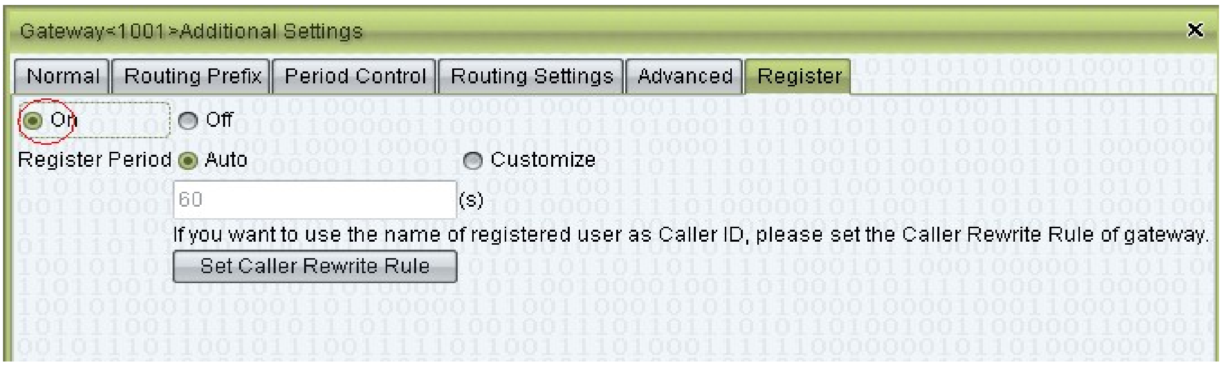


● Advanced---->SIP------Authentication User & Password



When Register enabled, Authentication Uaser and Password must be filled in.

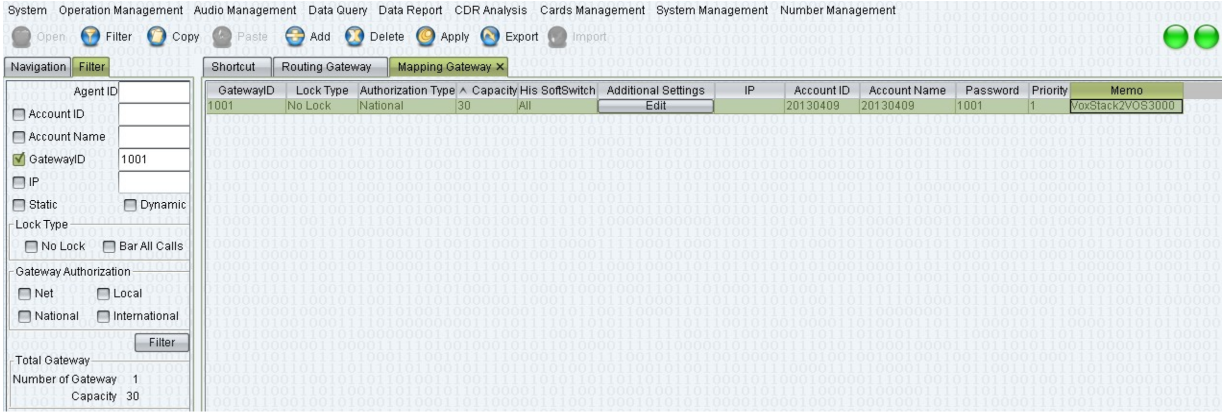
● Register------>On



This option is to choose register mode or ungister mode for GSM gateway.

Section 2: Add a Mapping Gateway in register mode:

Please select "Operation Management---> Gateway Operation-----> Mapping Gateway" to create a Mapping Gateway:

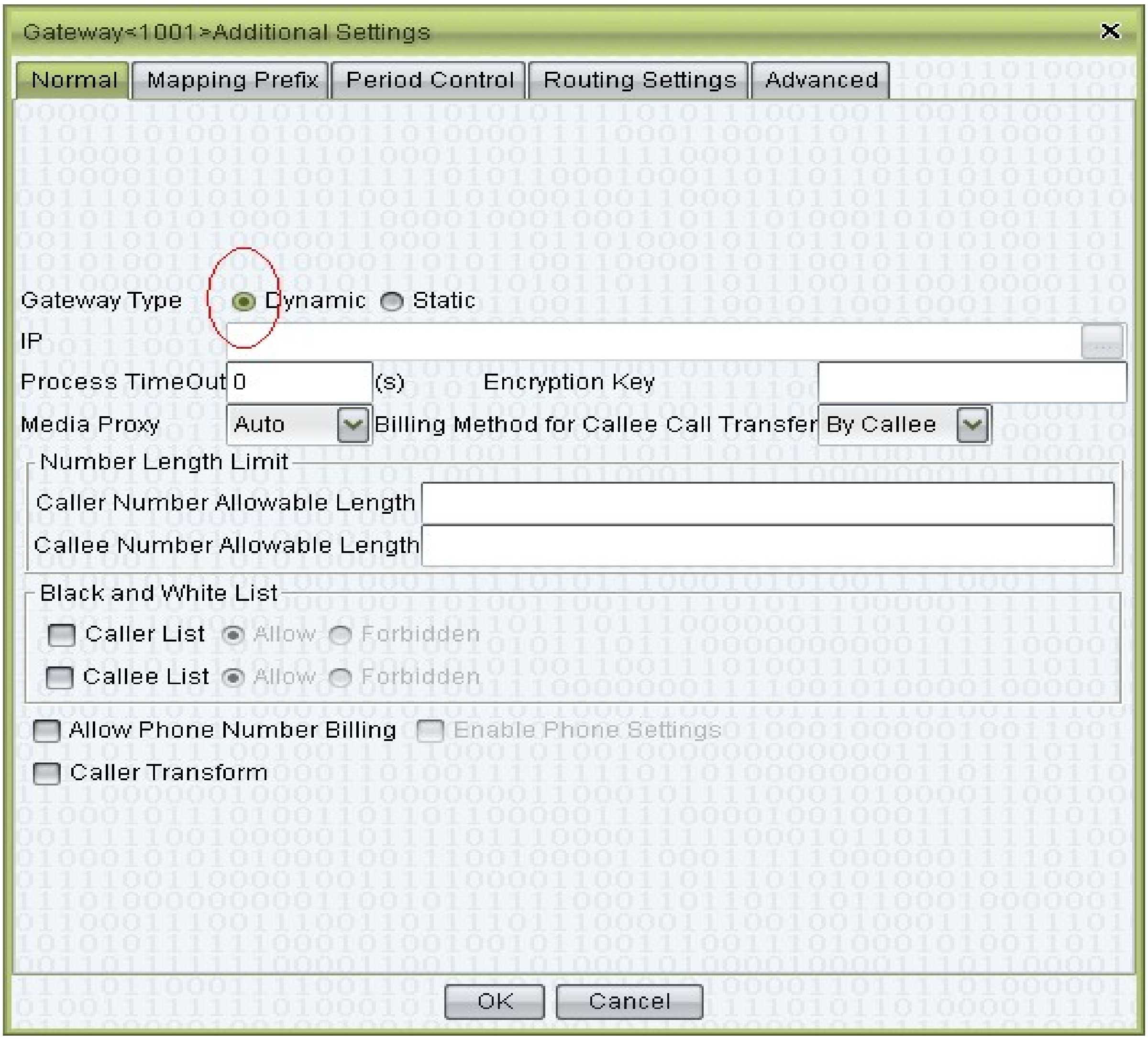


Gateway ID:1001 --------------------Username of SIP trunk which GSM gateway registers to

Password:1001 --------------------Password of SIP trunk which GSM gateway registers to

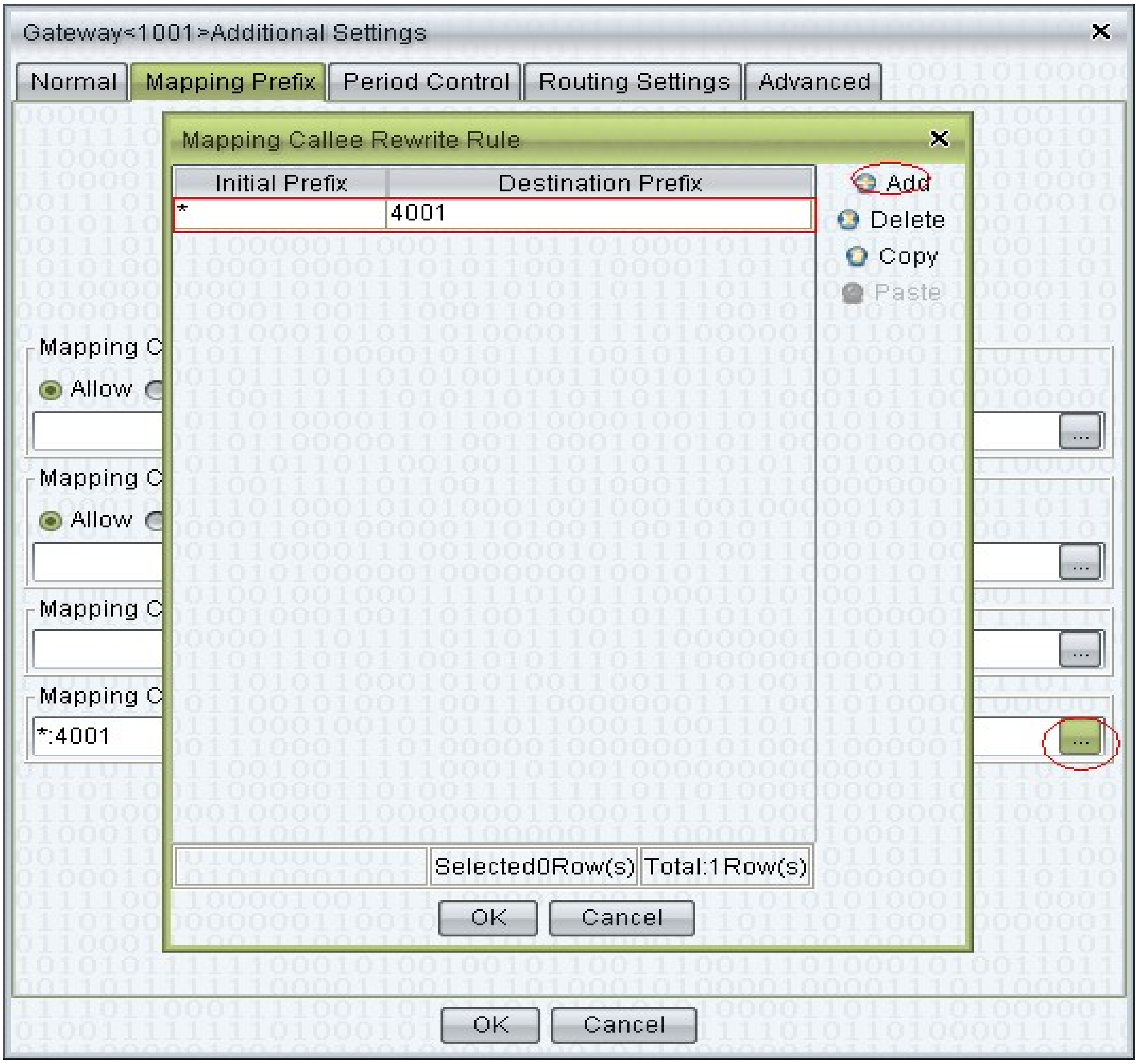
Additonal Settings:

●Normal---->Gateway Type---->Dynamic



● Mapping Prefix---->Mapping Callee Rewrite

Add a RMapping callee Rewrite Rule to match all calls from GSM gateway, then send it to extension 4001.



Step5. Call Test

Apply all changes on VOS3000 and GSM gateway, then you can try to make calls.

Taking advantage of SIP software such as Xlite, eyeBeam to register a SIP extension(4001) on VOS3000 server.

● Test call from VOS3000 to GSM geteway

Use Extension 4001 to call 9+number, then you will reach the number you want through GSM gateway, you can check it on GSM gateway.

● Test call from GSM geteway to VOS3000

Use your mobile to call numbers of SIM cards on GSM gateway, then Extension 4001will be ringing.