Installing GoIP in VOS and Elastix Platforms

This document briefly describes the basic instructions in setting up a GoIP.　It does not cover settings of all parameters. Please refer to the latest GoIP Series User Manual for more details.

1. Precaution

1) *Do not insert or remove a SIM card while the GoIP is powered up.　The SIM card may be damaged.*
2) *Do not operate the GoIP without the antenna properly connected.*

2. How to access the GoIP device webpage？

The device webpage can be accessed directly via either the LAN or PC port.　When your computer can access either network, you can just type corresponding IP in your web browser to access the device webpage.　Since the LAN port is set to obtain an IP from LAN connected, you will need to know how to find out the assigned IP address. However, for a non-technical or first time user, it is recommended to access the device webpage via the PC port (factory default IP is 192.168.8.1) as described in the instructions below.

1) Modify your computer LAN IP to 192.168.8.2.
2) Connect an Ethernet cable between your computer and the GoIP PC port directly.
3) Start your web browser
4) Type 192.168.8.1
5) A login window then pops up.　Enter the login ID as "admin" and password as "admin" to access the device webpage as shown below.

6) Click on "**Configurations**" and then "**Network**" to configure the LAN and PC ports.
7) Set the LAN port to "Static IP" mode and then assign a local IP and the default gateway.　This allows an easy to the GoIP by using a computer that is in the same LAN segment.　However, if you know how to find out the IP address that is assigned to the LAN port via DHCP, you can just skip this step.

Notes:
- ➢ LAN Port is used for network access and it is, by default, set to DHCP mode to obtain an IP from the local network automatically.
- ➢ PC Port is in general used to extend the existing LAN network (Bridge mode) or to establish a new LAN segment (Static IP mode) that is behind the existing LAN.   The default is set to Static IP mode with the default IP address as 192.168.8.1.   This enables a way to access the GoIP webpage if the LAN port IP cannot be determined.   It is recommended not to change the LAN port mode unless you know how to discover the LAN IP; otherwise, you may not be able to access the GoIP webpage.

3. GoIP Status

The Summary page is the first page shown when entering the device webpage and it shows the current status of the GoIP.   It is important to understand the meaning of the various status in order to debug and solve problems encountered when configuring or operating the GoIP.

Summary 1 2 3 4												5
CH	Line	M	SIM	GSM	VOIP	Status	SMS	ACD(S)	ASR(%)	Duration(S)	Count	CDR Start	RSSI
☐	1	Y	Y	Y	Y	IDLE		115	99	40135	349/353	2016-12-30 04:00:49	18
☐	2	Y	Y	Y	Y	IDLE		94	100	32148	342/342	2016-12-30 04:00:49	19
☐	3	Y	Y	Y	Y	IDLE		94	99	31678	337/340	2016-12-30 04:00:49	20
☐	4	Y	Y	Y	Y	IDLE		93	99	31341	337/340	2016-12-30 04:00:49	21
☐	5	Y	Y	Y	Y	IDLE		102	98	33864	332/339	2016-12-30 04:00:49	19
☐	6	Y	N	N	N	IDLE		94	100	30926	329/329	2016-12-30 04:00:49	21
☐	7	N	N	N	N	IDLE		101	100	33229	329/329	2016-12-30 04:00:49	99
☐	8	N		N	N	IDLE		89	100	28747	323/323	2016-12-30 04:00:49	99

The first page to view is the "Summary" page as shown above.   Each line status is shown horizontally.   The five columns marked in RED are described in details below.

1. **Column "M"** – This column shows the power status of the all GSM modules.   "Y" means that the power to the GSM module is on.   "N" means that the power to the GSM module is off.   Click on "Y" to turn off the power to the corresponding GSM module or vice versa.
2. **Column "SIM"** – This shows the SIM card status.   "Y" means that the corresponding GSM module can access the designated SIM card successfully.   "N" means the corresponding GSM module is unable to access the designated SIM card.   Please check if the SIM card is inserted properly or the SIM card is damaged.   If Remote SIM function is used, please check the SIM Bank and/or SIM Server configuration.   The problem could also be caused by bad network condition or improper network configuration.
3. **Column "GSM"** – This shows the GSM registration status.   "Y " means "Registered" and "N" means "Not Registered".
4. **Column "VoIP"** – This shows the VoIP registration status.   "Y " means "Registered" and "N" means "Not Registered".   If GSM Registration status is "N", VoIP registration is disabled and its status should be ignored.
5. **Column "RSSI"** – This indicates the received GSM signal strength with a valid range from 0 to 31.   It shows 99 when there is no signal at all.   For satisfactory GSM calls, the signal level should be above 15.

4. General Concept for GoIP Deployment

Depending on the application and network requirements, you need to first determine the connection method between GoIP and softswitch or IP PBX.   In general, if a fixed public IP can be assigned to a GoIP, then it can link up with a softswitch or an IP PBX as a SIP trunk without SIP registration.   If the GoIP is installed behind NAT, SIP trunk with SIP registration provided that the softswitch or IP PBX can support this mode.   In general, SIP trunk is a more secure method since GoIP will only accept calls originated from the IP addresses assigned.   The last resort is to use SIP registration mode only.   Please note that SIP registration in softswitch or IP PBX is generally counted as a SIP license. For multiple-channel GoIPs, you may want to use the Group SIP Registration mode in order to save on SIP licenses.

In addition, call routing rules must be correctly setup on the softswitch or IP PBX so that calls can be routed to and from the GoIP properly.

1. *Add GoIP to softswitch / IP PBX and setup call routing rules;*
2. *Configure GoIP for network connection, VoIP connection, and call routing rules if applicable.*

# 5. Example 1:   GoIP Deployment in VOS VoIP platform

VOS VoIP Platform is widely used in the call termination and private company environment.   This example only focuses on the basic configurations for both GoIP and VOS In order to establish both outgoing and incoming call connections.   Please refer to VOS user manual for account setup and other matters.
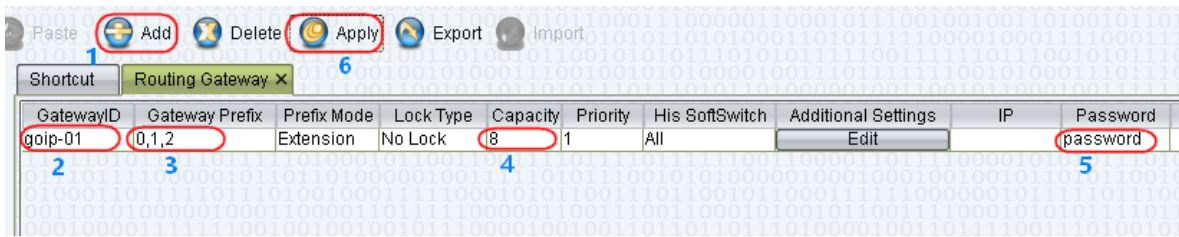
VoIP Environment:
1. VOS3000
2. GoIP-8

5.1 VOS Setup

1. Start the VOS300 VoIP Operation Platform
2. Add and configure Routing Gateway for outgoing calls.   Here are the procedures:
   a) Under Navigation, please select "Routing Gateway" as shown below.
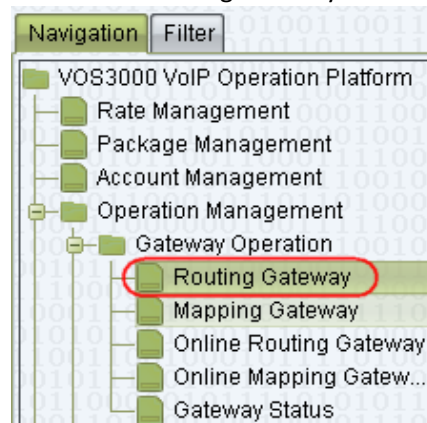
b) Follow the steps 1 to 6 (labelled in blue as shown in the screen capture below) after the Routing Gateway window pops up.

Routing Gateway in the VOS3000 is using SIP registration method and the parameters needs to be configured are:

➢ *GatewayID* – This is actually the name of the gateway and also the login ID for the gateway to register to the VOS3000.

➢ *Gateway Prefix* – The VOS3000 routes outgoing calls to the gateway when the prefix of the phone number matches the Gateway Prefix defined.   Pease note that if the prefix is only for call routing purpose, it should be removed by either the VOS3000 or the GoIP dialing rules.

➢ *Capacity* – This defines the number of lines available in the gateway.

➢ *Password* – This defines the SIP login password for the gateway.

3. Add and configure Mapping Gateway for incoming calls by following the steps (labelled in blue) in the screen capture below.   Please note that Routing Gateway in the VOS design is for call termination only (outgoing calls).   In order to receive incoming calls from the same gateway, it is necessary to add a Mapping Gateway with the same ID.
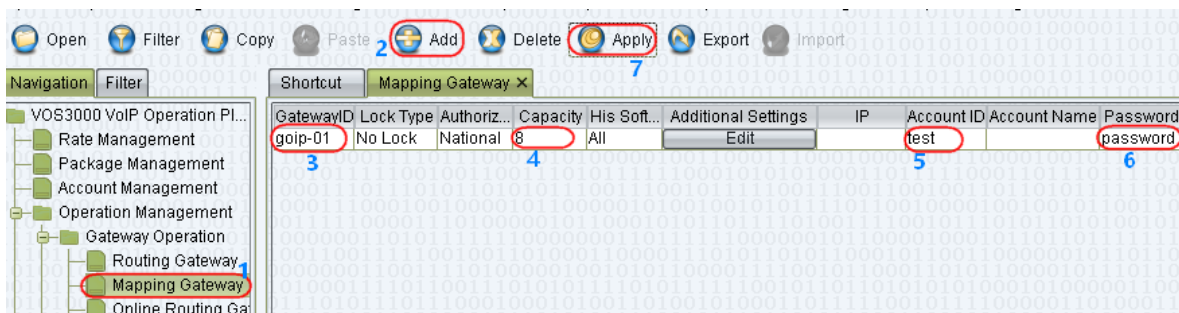
Here are the parameters for Mapping Gateway:

➢ *GatewayID* – This the Gateway ID for the VOS3000 to accept incoming calls.   Since calls are coming from the same gateway, the same ID as the one programmed in Routing Gateway must be used.

➢ *Capacity* – This defines the number of lines available in the gateway.   This must be the same as the one defined in the Routing Gateway above.

➢ *Account ID* – Assign an existing or new account for the mapping gateway and make sure that there are enough credits for the VOS3000 to handle incoming calls.

➢ *Password* – This defines the SIP login password for the gateway.   The password is the same as the one programmed in the Routing Gateway.

5.2 GoIP Setup

1. Access the GoIP webpage
2. Go direct to the Basic VoIP page as shown below.

3. Select "Single Server mode" and program the associated parameters as shown in the screen capture above. Please note that the same GatewayID and password as those defined in the VOS3000 are used for SIP registration. The SIP Proxy or SIP Registrar are the IP address or domain name of the VOS3000.

4. Save the new settings and the wait 30 seconds one before accessing the Status page as shown below. The VoIP line status is "Y" when the GoIP-8 registers to the VOS3000 successfully. This means that GoIP-8 is now ready to accept calls from the VOS3000. If the account setting with enough credits in the VOS3000 for the Routing and Mapping gateways is ready, the VS3000 can now route calls to the GoIP-8 for termination.

5. In order for GoIP-8 to route incoming GSM calls to the VOS3000, the CALL IN page must be configured properly. Access the CALL IN Page as shown below.

Here are the parameters need to be configured.

➢ **CID Forward mode** – This is a general parameter for all lines. It determines if and how the GSM Call ID is sent to the VOS.   For the VOS3000, it should be set to "Use CID as SIP Caller ID".

➢ **Call IN via GSM** – This is a property for each line and it determines if GSM incoming calls are allowed or not. Select the line first and then enable this parameter to enable GSM incoming calls are forwarded to the VOS3000.

➢ **Forwarding to VoIP Number** – GSM incoming calls for the line selected is forwarded to the phone number specified in this parameter.   It could be an extension number or IVR number in the VOS3000.   Please refer to the VOS3000 User Manual on how to configure IVR to enable second dial.

# 6. Example 2:   GoIP Deployment in Elastix VoIP platform

Elastix is another widely used VoIP platform for small business.   This example only focuses on the basic configurations for both GoIP and Elastix in order to establish outgoing and incoming call connections.   Please refer to Elastix and GoIP User Manuals other configuration matters.

VoIP Environment:
1. Elastix VoIP Platform
2. GoIP-8

6.1 Elastix Setup
1. In the Elastix main webpage, click on "PBX", "Trunks", and "Add SIP Trunk" as shown on the screen capture below.
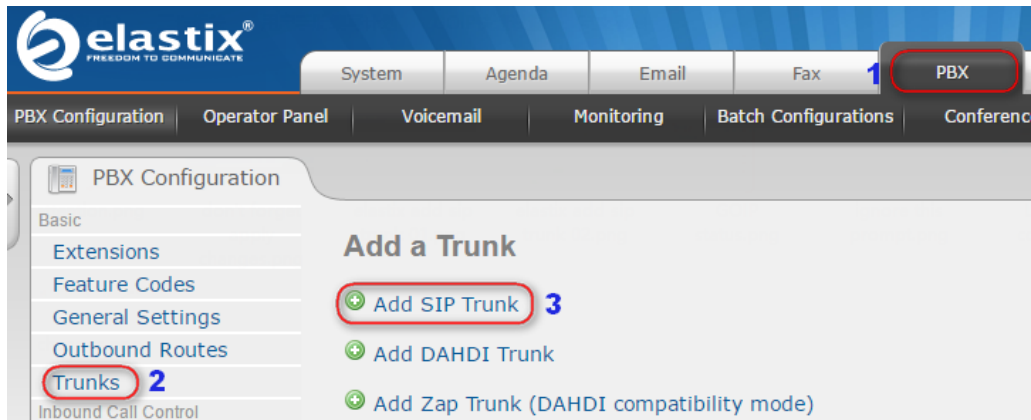
2. In the Add SIP Trunk window, program the parameters in the red circle.

- ➢ **Trunk Name** – This is the trunk name referred by Elastix.
- ➢ **Maximum Channels** - This defines the maximum number of voice channels in this trunk.   Since a GoIP-8 is used as a terminating gateway, then this value should be 8.
- ➢ **Outgoing settings** – This defines the SIP registration parameters.   The Trunk Name and secret (highlighted in green) are actually the SIP authentication ID and password for the GoIP.

3. Click "Submit Changes" to save the new settings just entered.   A warning message window is then pops up as shown below.   Ignore this warning message and click "confirm" to continue.

> It is highly recommended that you define an Outbound Caller ID on all trunks, undefined behavior can result when nothing is specified. The CID Options can control when this CID is used. Do you still want to continue?

4. Add an Outbound Route as shown below.

The Dial Pattern for this Outbound Route is shown below.   Calls with the phone number prefix "9" are routed to the trunk "goip-01" with the prefix "9" removed.   Please note that there is a "Period" mark in the field pointed by the red arrow.

5. Click "Submit Changes" at the bottom to save the Outbound Route.
6. The last step is to click "Apply Configuration Changes Here" as shown below in order to make the changes effective immediately.

6.2 GoIP Setup

1. Access the GoIP webpage

2. Go direct to the Basic VoIP page as shown below.

3. Select "Single Server mode" and program the associated parameters as shown in the screen capture above.   Please note that the same GatewayID and password as those defined in the Elastix server are used for SIP registration. The SIP Proxy or SIP Registrar are the IP address or domain name of the Elastix server.   In this case, just fill in either one or both.

4. Save the new settings and the wait 30 seconds one before accessing the Status page as shown below.   The VoIP line status is "Y" when the GoIP-8 registers to the Elastix server successfully.   This means that GoIP-8 is now ready to accept calls from the Elastix server.

5. In order for GoIP-8 to route incoming GSM calls to the Elastix server, the CALL IN page must be configured properly. Access the CALL IN Page as shown below.

Call In	
Status	
Configurations	User No Input Timeout(s) [60]
	CID Forward Mode [Use CID as SIP Calle ▼]
Preferences	CID Prefix [ ]
	○ CH1 ○ CH2 ○ CH3 ○ CH4 ○ CH5 ○ CH6 ○ CH7 ○ CH8
Network	Call IN via GSM ⦿ Enable ○ Disable
Basic VoIP	Forwarding to VoIP Number [1001]
Advance VoIP	Dial Plan [ ]
Media	GSM Call Waiting [Disable ▼]
Call Out	Hunt Group Mode [Disable ▼]
Call Out Auth	Auto Incoming Call Block ○ Enable ⦿ Disable
Call In	*Auto Config Other lines
Call In Auth	Save Changes

Here are the parameters need to be configured.

➢ **CID Forward mode** – This is a general parameter for all lines. It determines if and how the GSM Call ID is sent to the VOS.   For the Elastix server, it should be set to "Use CID as SIP Caller ID".

➢ **Call IN via GSM** – This is a property for each line and it determines if GSM incoming calls are allowed or not. Select the line first and then enable this parameter to enable GSM incoming calls are forwarded to the the Elastix server.

➢ **Forwarding to VoIP Number** – GSM incoming calls for the line selected is forwarded to the phone number specified in this parameter.   It could be an extension number or IVR number in the Elastix server.   Please refer to the Elastix server User Manual on how to configure IVR to enable second dial.