GoIP32-X4 Quick Setup Manual

GoIP32-X4 is a series of 32-Channel VoIP to mobile network gateways embedded with SIM rotation of 4 SIM slots per channel. The triggers for SIM rotations based on the outgoing call behaviors and the SMS usage. In addition, there are restrictions that can be set to limit the SIM usage and to disable the SIM slots from SIM rotations. These could help to prevent unexpected charges from over usage and to reduce the chances of being detected for SIM blocking by the service provider.

Two models are available to support 2G GSM and 2G/3G/4G LTE (referred hereafter as "4G") mobile networks

1. GoIP32-X4-2G

This model supports 2G GSM networks with quad-band frequencies: 850/900/1800/1900MHz

2. 4G Model: GoIP32-X4-4G

This model supports 2G/3G/4G LTE mobile networks. There are a few variants of 4G LTE networks which are currently deployed in various countries or territories. Therefore, different models are available with the model number.

Type "C" supports the 4G LTE networks in China as shown in the table below.

Wireless Technology	E-UTRA Band	Frequency (MHz)
FDD-LTE	B1, B3, B5, B8	2100, 1800, 850, 900
TDD-LTE	B34, B38, B39, B40, B41	2100, 2600, 1900, 2300, 2500
WCDMA	B1, B8	2100, 900
TD-SCDMA	B34, B39	2100, 1900
GSM	B3, B8	1800, 900
CDMA/EVDO	BC0	800

Type "E" supports the 4G LTE networks deployed in the Europe, Middle East, Africa, Korea, Southeast Asia, Hong Kong, and Macau. The table below shows the details of the wireless technologies and the bands and frequencies supported.

Wireless Technology	E-UTRA Band	Frequency (MHz)
FDD-LTE	B1, B3, B5, B7, B8, B20	2100, 1800, 850, 2600, 900, 800
TDD-LTE	B38, B40, B41	2600, 2300, 2500
WCDMA	B1, B5, B8	2100, 850, 900
GSM	B3, B8	1800, 900

This document serves as a quick reference manual in setting up GoIP32-X4. For complete device configurations and functionalities, please consult the GoIP32-X4 User Manual.

1 Attentions

1) All antenna connectors are located on the back panel of the GoIP32-X4. When short attentions are used, the Right-angle short antennas shall be installed on the top row with the antennas pointing upward and the straight antennas shall be installed on the bottom row, as shown in the picture below. Please also do not swap the antenna between the 2G and 4G models. Antenna for 4G model are marked with the label "4G".


(Back Panel)

2) The 128 SIM slots are hardware preconfigured as 4 slots per line. Each slot is labelled with its designated line number and its location number ("A", "B", "C", "D"). Please note that the SIM card insertion orientation is different between odd lines and even lines, as shown in the picture below. For the odd lines, SIM cards are inserted with the cut-edge pointing upward and the metal contacts facing the front panel. For the even lines, SIM cards are installed with the cut-edge point upward but the metal contacts facing the back panel. Each SIM card must be inserted in the correct orientation and fully pressed into the slot. There is no click sound to indicate the SIM card is in the lock position (this is different from the old GoIP models). This new slot socket enables the GOIP32-X4 to support SIM Hot-swap (swapping SIM cards without powering down the slot first). Please go to the Status page to check if the SIM cards inserted are properly recognized.
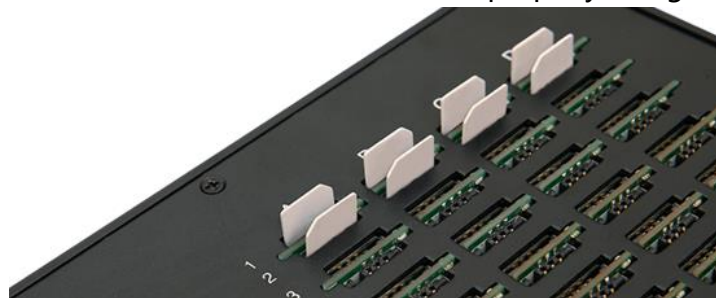
SIM Cards Insertion Orientation
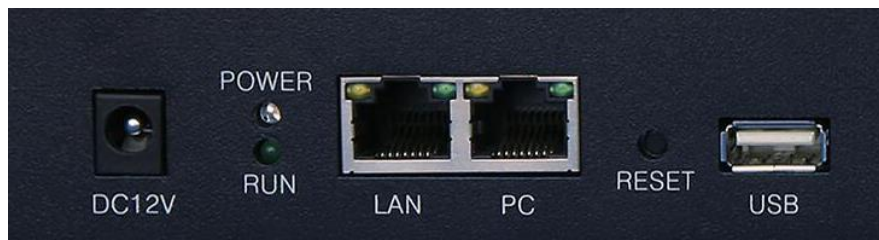
Front Panel

3) In order to fully utilize the advantages of 4G LTE network, it is recommended that the VoLTE (Voice over LTE) option is enabled in the GoIP32-X4. However, this service may not be available in your area. Please contact your local service provider for more information.

2   Device Web Login

There are two Ethernet ports on the back panel of the device.   The LAN port is intended for network connection to a VoIP Server.   In general, the designated VoIP server is not in the local network; therefore, the LAN port should be connected to a router or a LAN that has an internet access.   The PC port is intended to extend the LAN port network or to create a new LAN that is behind the LAN port.

➢ The default setting for the LAN port is to obtain an IP address from the local DHCP host. Please make sure that a local DHCP host is available when using the LAN port.
➢ The PC port is assigned with the IP address 192.168.8.1 as a factory default.   If this port is used for device web login, please make sure that the computer IP is in the same segment of 192.168.8.x.

Since it requires certain technical knowledge in order to find out the IP that is assigned to the LAN port, it is recommended that the PC port is used for the first time device web login.

   a.   Configure the computer IP to 192.168.8.2.
   b.   Connect an Ethernet cable to the GoIP32-X4 and the computer Ethernet port.
   c.   Type the PC port IP address (192.168.8.1) in address field of your computer browser.   Then enter both username and password as "admin" to login to the device web.   Try to use a different browser if abnormalities or problems encountered during configurations.

Once login, the LAN port IP can then be found in the status page when it is connected to a network that has a local DHCP host.   The LAN port IP can then be

used for web login via a computer that is connected to the same network.

3 SIM Rotation

There are a total of 128 SIM slots and each channel/line has 4 dedicated SIM slots.   However, only 1 SIM slot per channel is active at a time; the other 3 slots are the candidates for the next SIM rotations.   If there is no SIM inserted in a slot, the slot is disabled automatically for SIM rotation.   When a SIM card is inserted in a slot, the slot is then enabled automatically for SIM rotation.

3.1 SIM Manage

It is important to understand how SIM slots are managed for rotation and the conditions for triggering SIM rotation.   In the left hand menu column, click on **Configuration** and then the **SIM Manage** icon to enter the webpage shown below.

The top section defines the rules for switching an active SIM card.

1) SIM Selection Method – There are 3 choices: Round Robin, Random, Max. Talk Time Remain.
   - ➢ "**Round Robin**" method puts all available SIM cards in a circular loop and then and select the next SIM card in sequence.
   - ➢ "**Random**" selects the next SIM randomly from the SIM cards available (not including the active SIM card).
   - ➢ "**Max. Talk Time Remain**" means that the next SIM card is the one with the most remaining talk time (not including the active SIM card).

   Please select the method according to your requirements.

2) During an active SIM section, when any one of the conditions listed below is

matched, the current active SIM will then be switched to the next SIM according to the SIM Selection Method chosen.

Triggers for Initiating SIM Card Switching:

1. Total outbound Call duration ≥ [   ] mins
2. Number of outbound calls ≥ [   ] , [includes ▼] unanswered calls
3. Number of outbound SMS ≥ [   ]
4. SIM in use duration ≥ [   ] mins
5. No cell service duration ≥ [60] s

1. "**Total outbound call duration**" refers to the total talk time of all outgoing calls during an active SIM section.
2. "**Number of outbound calls**" refers to the total number of outgoing calls during an active SIM section with the option of including or excluding unanswered calls. If it is greater than the value defined, then the SIM switching process begins at the time defined in the bottom line of this section.
3. "**Number of outbound SMS**" refers to the total number of text messages sent via SMS during an active SIM section.
4. "**SIM in use duration**" refers to the duration of the current active SIM section.
5. "**No cell service duration**" refers the duration of not be able to access mobile service. This could be caused by an invalid SIM or weak RF signal.

When the above parameters exceeds the value defined, then the SIM switching process begins when there is active call in progress. If there is, the last option in this section defines what action should be taken on the active call.

Specify the action when either condition 1 or 4 occurs during an active call: [Hang up ⌄]
Specify the action when either condition 1 or 4 occurs during an active call: [Do nothing ⌄]

The bottom section, as shown below, defines the rules for disabling a SIM card slot. Once a SIM card is disabled, it can only be either manually activated via the SIM Info page (under the "Status" section) or replace the SIM card with a different one. Please note that if the same SIM card is removed and re-inserted to the same slot, the SIM card slot is still disabled. The SIM card is automatically enabled if a different SIM card is inserted.

Rules for disabling a SIM card

1. No. of consecutive registration failures ≥ [3]
2. No. of consecutive outbound calls with no ring back tones ≥ [8]
3. No. of consecutive unanswered outbound calls ≥ [15]
4. No. of consecutive short outbound calls ≥ [   ] (Duration for a short call ≥ [   ] seconds)
5. No. of quick answered outbound calls ≥ [   ] (Duration between dialing and answering ≥ [   ] seconds)

Fill in the desired value(s) for each rule.　If a field is blank, then the rule is not defined and will not be used for disabling a SIM card.　Please note that "short call" and "quick answered call" are defined in the parenthesis in rule 4 and 5.　If they are not defined, then the corresponding rule will not be used.

As shown in the rules above, when there are 3 consecutive registration failures, 8 consecutive outgoing calls with no ring back tones, or 15 consecutive unanswered outgoing calls occurs, the corresponding SIM card will be disabled.　The rule 4 and 5 are not defined.

3.2 Imposing SIM Card Usage Restrictions

Since a SIM card may come with different service plans or SIM usage may trigger blocking by the service provider, the **SIM Usage Limits** are implement to help to alleviate these issues.

To enter this page, click on **Configuration** and then **SIM Usage Limit** icon in the left hand menu column.

SIM Usage Limit

Talk Time Limit Exhausted	Maintain the active call
Call Count Limit	Ignore unanswered calls
SMS Count Limit Exhausted	Disable the SIM
Billing Increment (s)	60

How to apply the limits below For all sims

Talk Time Limit(m)	
Call Count Limit	
SMS Count Limit	

Defining the action when a limit is met:
➢ Talk Time Limit exhausted: Terminate the current call if any / Do nothing.
➢ Maximum number outgoing calls is reached: Including unanswered calls / Excluding unanswered calls.
➢ Maximum number of text messages (SMS) is reached: Disable

Defining the conditions for the limits:
➢ Billing Increment:

➢ How to apply the limits below: For all SIMs / For each SIM

Defining the limits:
➢ Talk Time Limit:
➢ Outgoing Calls Limit
➢ Outgoing Text Messages Limit

The Talk Time Limit can be specified in 3 different syntax as described below.　However, only one syntax can be used at a time.　If more than one syntax is specified, only the first one recognized will be used.

    a.　N
    b.　**D**x, N
    c.　xx:yy-xx:yy,N|xx:yy-xx:yy,N|······

Where

N is an integer to specify the talk time limit.

D is a command code for the cut off day for monthly usage.　Example: "D1,10000" means the counting period starts on the first day of the month for a maximum of 10,000 minutes of talk time.

xx:yy-xx:yy specifies an actual time interval.　Example: "08:00-12:00,200|12:00-24:00,1000" means the talk time limit is 200 minutes from 8:00 AM till 12:00 PM and the talk time limit is 1000 minutes from 12:00 PM till mid-night.　The talk time for other unspecified time interval is unlimited.　Please note that "|" is used as a separator so that more than one interval can be specified.

Note; If network time cannot be obtained, the limit for the monthly usage or the time interval usage will not work.　The Talk Time will be set to zero and this means that no outgoing calls will be allowed.　It will resume normal operation until the network time is successfully obtained.

3.3 Enabling and Disabling a SIM slot manually

SIM slots can either be enabled or disabled manually via the "Statistics" or the "SIM Info" page which can be accessed under the "Status" menu.　This implementation allows the use to enable or disable a SIM slot based on the statistics or the information of the SIM card in use.

Please note that each slot is labelled by its designated channel number and the slot location (A, B, C, or D).　To select a SIM card slot, click on the check box on the left.　Multiple slot selections are allowed.　To select all slots, click on the top check box right beside "Card" on the top row.　Then click on the "Enable" or "Disable" (below the **Statistics or SIM Info** title) to enable or disable the SIM card slot(s) selected.

Statistics															
Enable Disable		History CDR				Today			Available Resources			Alerts on consecutive calls			
☐	ID	ACD	ASR	Duration	Call Count	Duration	Call Count	SMS Count	Remain Time	Remain Count	Remain SMS	No Ring Back	No Answer	Short Call	
☐	1A	0	0%	0	0/0	0	0/0	0	300	NO LIMIT	NO LIMIT	0	0	0	
☐	1B	0	0%	0	0/0	0	0/0	0	300	NO LIMIT	NO LIMIT				
☐	1C				0/0	0	0/0	0	300	NO LIMIT	NO LIMIT				
☐	1D				0/0	0	0/0	0	300	NO LIMIT	NO LIMIT				
☐	2A	0	0%	0	0/0	0	0/0	0	300	NO LIMIT	NO LIMIT	0	0	0	
☐	2B	0	0%	0	0/0	0	0/0	0	300	NO LIMIT	NO LIMIT				

Status
Summary
General
Statistics
SIM Info
Configurations

SIM Info						
Enable Disable						
☐	Card	Num	IMSI	ICCID	Status	Disable Cause(Time)
☐	1A	+8615013484241	460023134611628	89860001191798928776	Queued	
☐	1B	+8615768271937	460077683646987	898600A6191607078549	In Use	
☐	1C				Disabled	Manually(07-19 18:03:43)
☐	1D				No Card	
☐	2A	+8615889472522	460004991858609	89860053191787732713	In Use	

Status
Summary
General
Statistics
SIM Info

If a SIM slot is disabled as described in section 3.2, please make sure that the corresponding condition is probably reset before enabling the SIM slot manually.

For the SIM info page, the "Status" column displays the current status of all slots.

➢ **In Use** – this indicates the SIM inserted is now used for mobile connection.
➢ **Queued** – this indicates the SIM slot is currently placed for SIM rotation.
➢ **No card** – this indicates that there is no SIM card inserted in the slot.
➢ **Disabled** – this indicates that the slot is disabled and will not be used for SIM rotation.
➢ **Unknown** – this indicates a SIM card is inserted but it has not been selected for mobile service connection yet. This usually occurs when the GoIP32-X4 is rebooted or a SIM card is inserted when it is in the"Disabled" status. When it is in use, the card information will be read and checked with the previous SIM card in use. If it is a different SIM card, then the SIM Info is updated and all SIM usage and limits are reset. If it is the same SIM card, then the SIM Usage Limits will be checked to determine if the SIM card is valid for SIM rotation.

4 General Concept for GoIP Deployment

The application concept described in this section is applicable to all models of GoIP. Therefore, the term "GoIP" is used to represent any GoIP model.

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.*

4.1 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. GoIP32-X4

4.1.1 VOS Setup

1. Start the VOS300 VoIP Operation Platform

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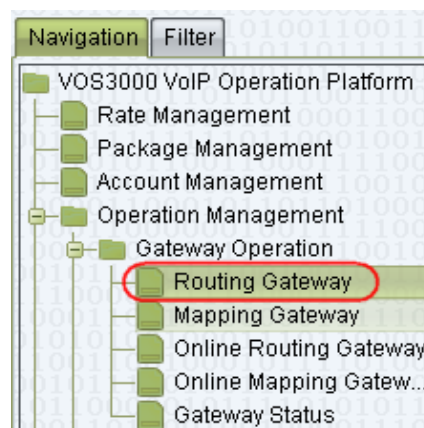
a) Under Navigation, please select "Routing Gateway" as shown below.
b) Follow the steps 1 to 6 (labelled in blue as shown in the screen capture below) after the Routing Gateway window pops up.

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.

4.2.2 GoIP32-X4 Setup

1. Access the GoIP32-X4 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 GoIP32-X4 registers to the VOS3000 successfully. This means that GoIP32-X4 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 GoIP32-X4 for termination.

5. In order for GoIP32-X4 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.

4.2 GoIP32-X4 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.   GoIP32-X4

4.2.1 Elastix Setup

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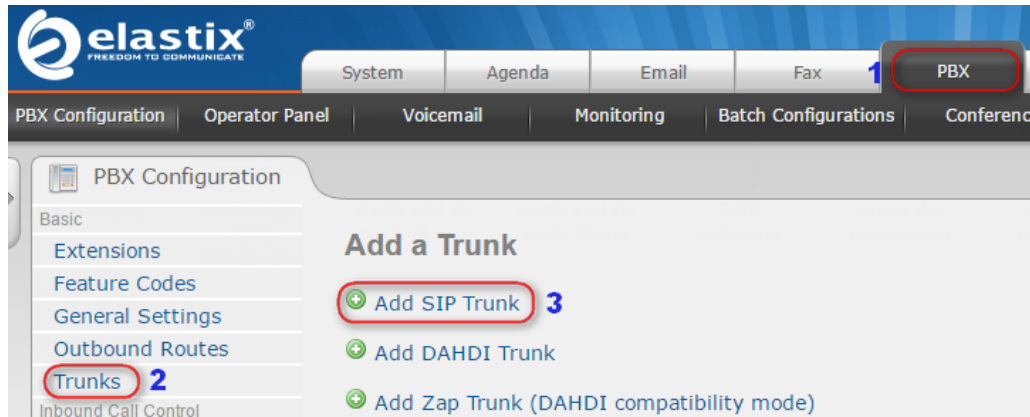
2. In the Add SIP Trunk window, program the parameters in the red circle.

➢ **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 GoIP32-X4 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.
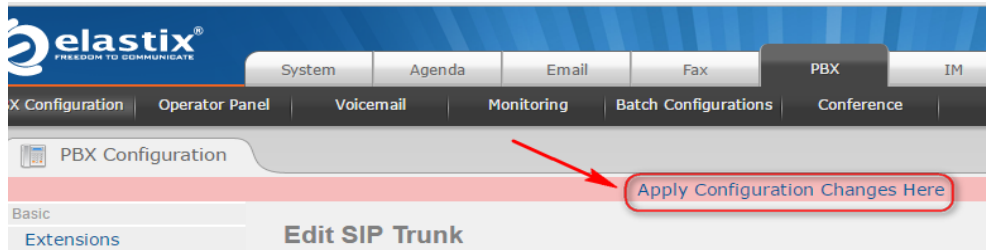
4.2.2 GoIP32-X4 Setup

1. Access the GoIP32-X4 webpage. Go direct to the Basic VoIP page as shown below.

2. 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.

3. 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 GoIP32-X4 registers to the Elastix server successfully.   This means that GoIP32-X4 is now ready to accept calls from the Elastix server.

4. In order for GoIP32-X4 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.

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.