GoIP User Manual

VoIP GSM Gateways

Models:

GoIP

GoIP-4/4i

GoIP-8/8i

GoIP-16

GoIP-32

HyberTone
Revision: 1.4B
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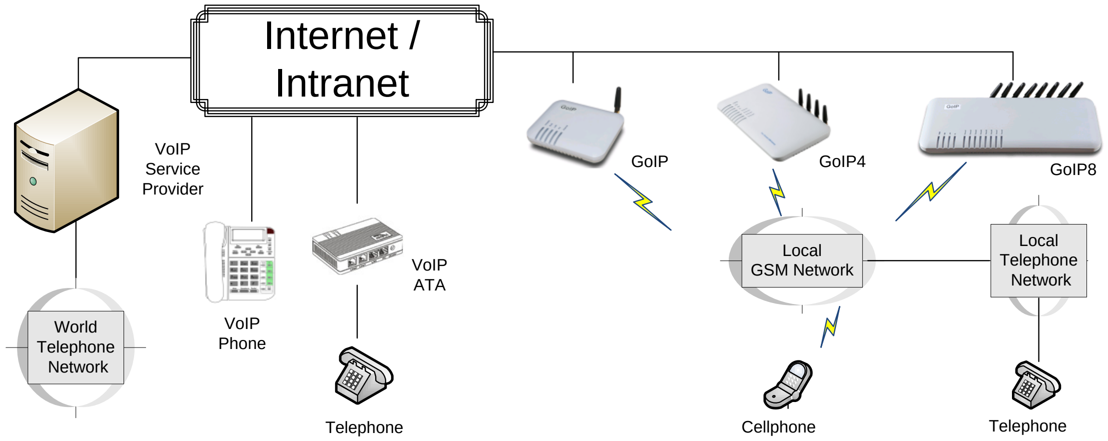
Content

1 General

1.1 Introduction

GoIP is the abbreviated from GSM over IP. It is a new type of VoIP gateway that allows call terminations from a VoIP network to a GSM network and vice versa. Call connections between IP networks and GSM networks are now bridged seamlessly to extend the voice communication coverage significantly. As the traditional PSTN lines are starting to disappear in developed countries and are not going to be built extensively in under-developed countries, GSM phones are getting more and more popular all over the world with lower and lower service charges, the emergence of GoIP bridges the gap between the traditional telephone networks and VoIP networks as shown in the diagram below. As a result, local and worldwide voice communications are more convenience, lower cost, and broader coverage.



You can now make a call from anywhere in the world via a VoIP network and then terminate the call via a GoIP to the local telephone network (PSTN). On the other hand, you can also make a call from the local telephone network to a GoIP (the GSM phone number) and then dial another number via a VoIP network to anywhere in the world. In these two cases, a VoIP Service provider is required for one side of the call termination. For two fixed locations, it is possible to setup GoIPs at both ends for call terminations without subscribing to a VoIP Service provider.

GoIP can also be used to achieve GSM roaming via VoIP. The idea is to route all your incoming GSM calls to a GoIP via call forward or simply insert your SIM card to a GoIP. You can then setup the GoIP to forward all incoming calls to another GSM number in the world via a VoIP service provider. The charge per call from a VoIP service provider is significantly lower than the roaming charge.

For office environment, GoIP offers a quick way to replace the traditional PSTN lines or T1/E1 lines to your IP PBX. There is no initial installation/reallocation charge and no need to wait for installation. Depending on our usage, you can add or remove lines as per your requirement. You can even configure the system so that everybody calls the same number regardless the number of lines available.

1.2 Protocols

- ✓ TCP/IP V4 (IP V6 automatic adaptive)
- ✓ Dual VoIP protocols: ITU-T H.323 V4, IETF SIP V2.0
- ✓ Multiple Codecs: ITU-T G.711 Alaw/ULaw, G.729A, G.729AB, G.723.1 and GSM
- ✓ H.2250 V4
- ✓ H.245 V7
- ✓ H.235 (MD5, HMAC-SHA1)
- ✓ RFC1889 real-time digital transmission protocol
- ✓ NAT
- ✓ STUN
- ✓ Network Management Protocol (NMP)
- ✓ PPPoE Dial Up
- ✓ PPP Authentication Protocol (PAP)
- ✓ Internet Control Message Protocol (ICMP)
- ✓ TFTP
- ✓ Hypertext Transfer Protocol (HTTP)
- ✓ Dynamic Host Configuration Protocol (DHCP)
- ✓ Domain Name System (DNS)
- ✓ User Account Authentication (via MD5)
- ✓ Proprietary Relay Protocol (Avoiding VoIP Blockings)

1.3 Hardware Features

- ✓ ARM processor
- ✓ DSP for voice signal processing
- ✓ Two 10/100MB Ethernet ports (IEEE 802.3 standard) with status LEDs
- ✓ Quadband GSM module (850M 900M, 1800M and 1900M)
- ✓ External Antenna (Internal Antenna option for selected models)

1.4 Software Features

- ✓ LINUX OS
- ✓ Built-in Web Server for device configuration
- ✓ Built-in SIP Proxy (Simplified)
- ✓ PPPoE Dial Up
- ✓ Router function
- ✓ DHCP client & Server
- ✓ QoS (VLAN)
- ✓ VPN (PPTP)
- ✓ Online firmware upgrade
- ✓ Remote Control Mechanism for remote technical support
- ✓ Proprietary Auto Provisioning Mechanism
- ✓ Remote SIM function
- ✓ Short Messages (SMS) support (standalone and server based)
- ✓ Call Management and Routing

1.5 Package Content

Use care when unpacking the device package in order to avoid damage to the main unit and the packing materials. Retain the packing materials in case the unit is to be transported in the future.

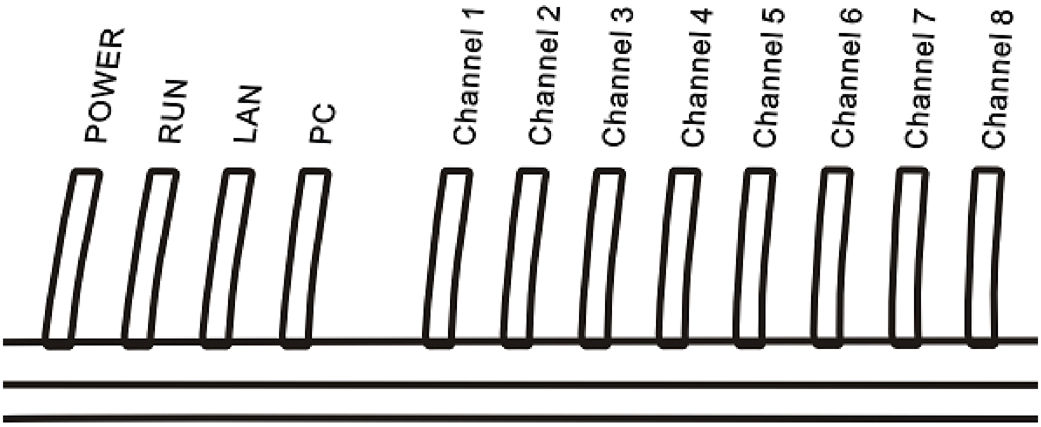
Please inspect the shipping container and the contents for any damages. If visible damages are present, please contact your vendor. Keep the shipping materials for the carrier inspection.

The package should contain the items listed in the table below

Item	Appearance	Description
1.	GoIP (1-Channel)  GoIP-4 (4-Channel)  GoIP-8 (8-Channel)  GoIP-16 (16-Channel)  GoIP-32 (32-Channel) 	1 x Main Unit

2.		AC/DC Power Adapter: ➢ GoIP1: 12V/500mA ➢ GoIP4: 12V/2A ➢ GoIP8: 12V/3A ➢ GoIP16: 12V/4A ➢ GoIP32: 12V/4.5A
3.		1 x Ethernet CAT5 Cable (2M)

1.6 LED Indicators



LED indicators (shown above for GoIP-8) are used to show the current status of the device. They are often used to determine if the GoIP is working normally or not.

LED Label	Description
Power	This LED is red and illuminates when power is connected.
LAN	This LED is red and illuminates when the LAN port is connected and blinks when data transmission occurs.
PC	This LED is red and illuminates when the PC port is connected and blinks when data transmission occurs.
RUN	This LED is green and blinks at a rate of every 100ms when VoIP is not ready for making calls. (Fast Blink) It blinks at a rate of every second when VoIP is ready for making calls (Slow Blink).

Channel "x"	Each GSM channel has its own status LED and its color is green. 1. It blinks at a rate of every 100ms (Fast blink) when the corresponding GSM channel is not yet registered to a GSM network. 2. It blinks at a rate of every second (Slow blink) when the corresponding GSM channel is ready for making or receiving calls (registered to a GSM network). 3. It illuminates when GSM call activities occurs (in use, ringing).

2 Installation

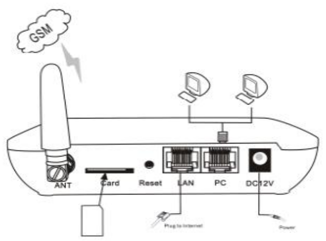
The same installation procedure applies for all models with the differences in the number of channels (ports) available and the SIM card insertion. *It is important to note that the power to the SIM slot MUST BE disconnected/removed before removing or inserting a SIM Card*. The power is removed by either disconnecting the power to the GoIP or shutting each GSM module individually via its built-in web interface.

1. SIM card slots are located either at the bottom (for old hardware) or at the back (for new hardware) of the main unit.

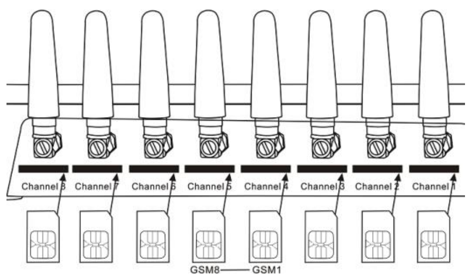
   For the models with the SIM card slots located at the bottom, you need to open the bottom SIM cover in order to install SIM cards. First slide the metal clip to the direction as indicated on the top of the clip. Insert a SIM card to each slot carefully and then place the metal clip back in place.

   For the models with the SIM card slots located at the back, just insert a SIM card to each slot as shown in the drawing on the right. Please make sure that the orientation of the SIM Card is correct before inserting the card.

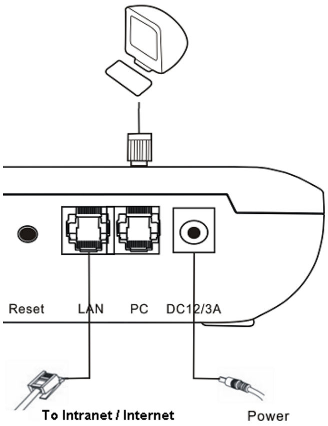
   For GoIP (1-channel), the SIM card insertion orientation is shown in the figure on the right. The metal contacts must face down and the cut corner is inserted first.



   For GoIP-4 and GoIP-8, the SIM card insertion orientation is shown in the figure on the right. The metal contacts must face up and the cut corner is inserted first.



2. The LAN port is intended for intranet or internet connection. Depending on your network environment, it can be connected various type of network equipment, such as network router, network switch / Hub, xDSL/Cable modem, etc.



3. The PC port is intended for network sharing and it supports both bridge and router modes. In Bridge mode, the PC port is connected to the same network segment as the LAN port. In Router mode, the PC port is set to a different network segment. In this case, please make sure that the PC network segment IP (192.168.x.) is different from the one in the LAN port network.
4. The DC port is for power connection. Please only use the AC/DC adapter provided. Adapter with different rating or vendor may damage the device or affect its performance.
5. The Reset button is recessed inside the GoIP cabinet. You need to use a sharp pointer to access the reset button. Press it momentarily to reboot the device. Press it for 15 seconds or more to reset the device settings including login password to its factory defaults.

3 Configuration

The device can be configured via its built-in http web server or via an Auto Provision Server. Auto Provision Server is a free utility supporting both Window and Linux OS. This utility is developed by HyberTone Technology for the sole purpose of automating the configuration of our products. It is available in our website for free download. This user manual only focuses on the device configuration via its built-in http web server.

Please note that only window based Web browsers, such as IE and Chrome are supported. Both Firefox and Mozilla may not work properly depending on the version and the operating system used. If you are having problems in configuring your device with your existing Web browser, please try one with lower version or a different Web browser and report the problem to us.

3.1 HTTP WEB Server Login

There are two methods to access the built-in web server.

1. Method 1 is to access the built-in web server via the LAN port. The LAN port is set to DHCP mode as a factory default. When you connect it to a network with a DHCP host, it will obtain an IP address from the DHCP host automatically. Via the GoIP's GSM channel(s), there are two ways to find out the IP address that is assigned to this port.

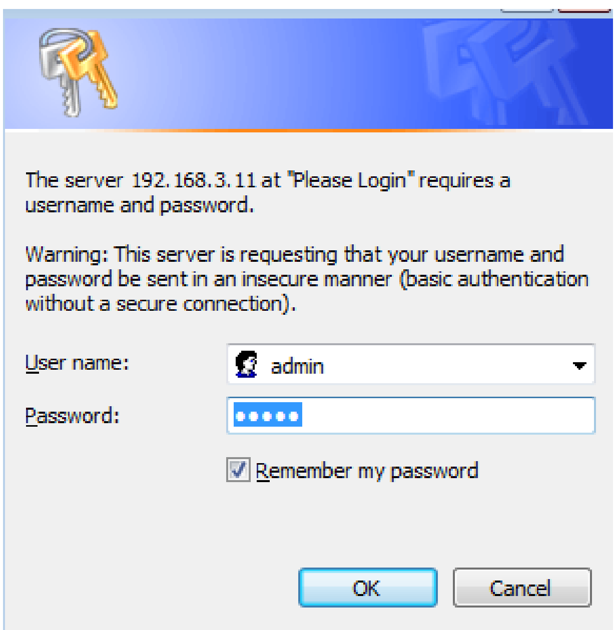
    i. Dial the SIM number of anyone of the GSM channels available. Once the call is answered, dial "*01" to hear a voice prompt reporting the LAN port IP address.
    ii. Send the "###INFO###" SMS command to one of the GSM channels available. The GoIP will then return back the LAN port IP address. Please refer to Appendix A Special SMS Commands for more information.

    Once the LAN IP address is known, you are now ready to access its built-in http web server by typing its IP address in the address field of a web browser.

2. Method 2 is to access the built-in we server via the PC port. As a factory default, the PC port IP is preset to 192.168.8.1. Connect a computer to the LAN port of the device and configure its IP to 192.168.8.x (x = 2 to 254). Type the IP address 192.168.8.1 in the address field of a web browser.

Once the IP address is entered, the login window shown on the right pops up. Enter the user name and password. There are three level of access via three different user names.

1. Administrative Level -This offers a full access right to all parameters available in the built-in webpage. The user name and password for the administrative level are "admin" and "admin" respectively.
2. User Level - This level restricts user from accessing the Call Setting page. User will not be able to change any VoIP related settings. The user name and password for the user level are "user" and "1234" respectively.
3. SMS Level - This level only allows user to access the Send SMS and SMS Box functions under the Tool menu. The user name and password for the SMS level is "sms" and "1234".



3.2 Status

There are four pages under the Status category and they are Summary, General, GSM, and SIM Call Forward. Each page is refreshed every 5 seconds; however, this feature is not supported when the Firefox browser is used. Sample pages shown in this section are captured from a GoIP-8. Please refer to the actual web pages of the models of interest. It is important to understand the information shown in these pages in order to debug or report problems encountered.

3.2.1 Summary

The current VoIP and GSM statuses are listed in the Summary page as shown below (extracted from GoIP-8) . They are very essential to display the operation status of the GoIP in order to determine if it is working properly or not.

Summary

Line	M	SIM	GSM	VOIP	Status	SMS	RSSI	Carrier	Cell ID	Idle	Remain	Reset
1	Y	N	N	N	IDLE	N	99			55	0	Reset
2	Y	N	N	N	IDLE	N	99			55	0	Reset
3	Y	N	N	N	IDLE	N	99			55	13	Reset
4	Y	N	N	N	IDLE	N	99			55	NO LIMIT	Reset
5	Y	N	N	N	IDLE	N	99			55	NO LIMIT	Reset
6	Y	N	N	N	IDLE	N	99			55	0	Reset
7	Y	N	N	N	IDLE	N	99			55	NO LIMIT	Reset
8	Y	N	N	N	IDLE	N	99			55	NO LIMIT	Reset
ALL												Reset

Here are the list of GoIP parameters shown in this page.

1. ***CH*** - GoIP channel reference
2. ***M*** - GSM module status for the corresponding CH. "Y " means "Enabled" and "N" means "Disabled". If a GSM module is disabled, all other parameters for this channel are not active. Clicking "Y" shuts down the channel selected. Clicking "N" turns on the channel selected.
3. ***SIM*** - SIM card status. "Y" means that the corresponding GSM module can access the designated SIM card successfully. "N" means 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.
4. ***GSM*** - GSM registration status. "Y " means "Registered" and "N" means "Not Registered".
5. ***VoIP*** - 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.
6. ***Status*** - VoIP line status. If VoIP registration status is "N", the VoIP line status shown should be ignored. Once VoIP registration status is "Y", the current VoIP line status is then shown in this field. Here are a list of available statuses:

   a. ***IDLE*** - The VoIP line is not engaged in any call activities.

   b. ***CONNECTED*** - An active call between VoIP and GSM is in progress.

   c. ***ACTIVE*** - A second dial tone is generated when a VoIP call is answered without making a GSM call or when a GSM call is answered without making a VoIP call. The generation of a second dial tone prompts the caller to press a phone number. The "**Status**" changes to "**ACTIVE**" since the start of the second dial tone till a phone number is received for dialing or the call is terminated.

d. ***DIALING <phone number>*** - This occurs when the GoIP is dialing out a phone number via the corresponding GSM channel or The DIALING status shows that a number is being dialed out via the corresponding GSM channel or a VoIP line. The phone number dialed is also shown in the "**Status**".

e. ***ALERTING*** - After a phone number is dialed, the "Status" changes to "ALERTING" when a ringback signal is received from the network.

f. ***INCOMING*** - This occurs when a GSM incoming call is calling and the call is not answered yet.

7. ***SMS*** - SMS Server registration status. "Y " means "Registered" and "N" means "Not Registered".
8. ***RSSI*** - This indicates the Received Signal Strength Indicator of the current cell. It ranges from 0 to 31 which represents a signal level ranging from -113 dBm to -51 dBm; each increment in rssi values means 2 dBm increment. 99 means that the signal level is unknown or undetected.
9. ***Carrier*** - This shows the name of the current GSM carrier.
10. ***Cell ID*** - This shows the Base Transceiver station (BTS) ID.
11. ***Idle*** - This shows the time elapsed since the last call.
12. ***Remain*** - This shows the time remaining if the Total Talk Time Limit (m) is set. Once the Remain time reaches zero, the corresponding channel is locked and its VoIP registration is also suspended (default setting). However, there is an option in Section 3.3.10 to enable SIP registration even when the Talk Time Limit expires ((Remain = 0).
13. ***Reset*** - Click this button to reset the Remain Timer to the Total Talk Time. Clicking on the Reset button located at the bottom (the row that is labeled "All") resets all Remain Timers of all channels.

3.2.2 General

The General page covers basic information on the hardware, network, and call status and setting. These information are useful for debugging the device operation and status.

Hardware			
S/N	test2	Firmware	GS-4.01-61-j3
Model	GoIPx8-G610	Local Time	2013-12-23 11:28:06
Network			
LAN Port	192.168.2.250	LAN MAC	
PC Port	192.168.8.1	PPPoE	DISABLED
Gateway	192.168.2.253	DNS Server	8.8.8.8

Call Management								
VoIP				GSM				
Line	Mode	Login	Routing Prefix	CH	Login	Call In	Call Out	Remain Time
1	S	N	11,13	1	N	Y	LOCK	0
2	S	N	12	2	N	Y	LOCK	0
3	S	N		3	N	Y	Y	13
4	S	N		4	N	Y	Y	NO LIMIT
5	S	N		5	N	Y	Y	NO LIMIT
6	S	N		6	N	Y	LOCK	0
7	S	N		7	N	Y	Y	NO LIMIT
8	S	N		8	N	Y	Y	NO LIMIT

1. Hardware

a) ***S/N*** – This field shows the serial number of the device.
b) ***Firmware*** – This field shows the current firmware version.
c) ***Model*** – This field shows the model number of the device
d) ***Local Time*** - This shows the current system time. It is a good indication for normal network access provided that the network server address and time zone are set properly.

Please make sure that these information are provided when reporting a problem or requesting for technical support.

2. Network
   a) ***LAN Port*** – This field shows the IP address assigned to the LAN Port.
   b) ***LAN MAC*** - This field shows the physical hardware address (MAC) assigned to the LAN port.
   c) ***PC Port*** – This field shows the IP address assigned to the PC Port.
   d) ***PPPoE*** – This field shows the PPPoE dial up status. It is only meaningful when PPPoE is enabled.
   e) ***Gateway*** – This field shows the default gateway IP assigned for data traffic routing.
   f) ***DNS Server*** – This field shows the current DNS server assigned for domain name interpretation. It is possible that some domain names are blocked by local DNS servers. Changing this to an overseas DNS server may solve the problem.
   g) VPN Status - This shows the current VPN connection status. It only appears when VPN is enabled.

3. Call Management section summarizes the both GoIP and GSM configurations and their corresponding status. It is important to note that the VoIP lines and the GSM channels are not mapped to each other as a one to one relationship. For outgoing calls (from VoIP to GSM), the GSM channel selection is based on the **Routing Prefix**.

   <u>VoIP</u>
   a) **Line** - This is used as a reference in VoIP line configuration.
   b) **Mode** - This shows the current VoIP Registration mode. "**S**" means Single Server Mode. "**L**" means Config. By Line mode. "**Gx**" means Config. by Group mode where x is the group reference number. "**T**" means Trunk Gateway mode.
   c) **Login** - This shows the current VoIP registration status. "Y" means that the corresponding line registers to the server successfully. "N" means the corresponding line fails to register to the server.
   d) **Routing Prefix** - This shows the current setting for the Routing Prefix. Please refer to Section 3.3.3 for more information.

   <u>GSM</u>
   e) **CH** - This corresponds to the physical GSM channel number.
   f) **Login -** This shows the current GSM Registration status for voice calls.
   g) **Call In** - This shows the Call IN setting for the corresponding GSM channel. "**Y**" means incoming calls are enabled. "**N**" means incoming calls are disabled and the corresponding channel rejects all incoming calls by sending back the hangup ("ATH") command to the GSM network.
   h) **Call Out** - This shows the Call Out setting for the corresponding GSM channel. "**Y**" means outgoing calls are enabled and "**N**" means outgoing calls are disabled. When the **Remain Time** for outgoing calls reaches **zero**, the Call Out setting is set to "**LOCK**" automatically. To unlock the channel, click the corresponding [Reset] button in the **Summary** page.
   i) **Remain Time** - This is the same as the "**Remain**" shown in the **Summary** page. If the **Talk Time Limit** in the **SIM** Page is set, this parameter shows the remaining time allowed for outgoing calls.

3.2.3 GSM

The GSM page shows the current GSM channels status and information on the GSM modules and the SIM cards inserted.

GSM

Remote SIM: DISABLE

CH	SIM	GSM	RSSI	GPRS Login	GPRS Attach	Carrier	BSC Mode	Cell ID	LAC
1	N	N	99	N	N		AUTO		
2	N	N	99	N	N		AUTO		
3	N	N	99	N	N		AUTO		
4	N	N	99	N	N		AUTO		
5	N	N	99	N	N		AUTO		
6	N	N	99	N	N		AUTO		
7	N	N	99	N	N		AUTO		
8	N	N	99	N	N		AUTO		

GSM Details

CH	Module	Module Ver	SIM Number	IMEI	IMSI	ICCID
1	G610	G610_V0C.00.0D_T14		355073036020376		
2	G610	G610_V0C.00.0D_T14		355073036021077		
3	G610	G610_V0C.00.0D_T14		355073035996840		
4	G610	G610_V0C.00.0D_T14		355073036005765		
5	G610	G610_V0C.00.0D_T14		355073036003976		
6	G610	G610_V0C.00.0D_T14		355073036019600		
7	G610	G610_V0C.00.0D_T14		355073036009098		
8	G610	G610_V0C.00.0D_T14		355073036019840		

The top table shows a number of GSM parameters which are useful to determine if the GSM channels in the gateway are working properly.

1. Remote SIM - This tells if Remote SIM function is used or not. "DISABLE" means using the local SIM cards that are inserted to the GoIP.
2. SIM - "Y" means the corresponding GSM module is able to access the designated SIM card properly.
3. GSM - "Y" means the corresponding GSM module registers to the GSM network successfully.
4. RSSI - Received Signal Strength Indicator. Please see the description in Section 3.1.1.
5. GPRS Login - "Y" means access to a GPRS network. This status is obtained from the command AT+CREG.
6. GPRS Attach - "Y" means GPRS Attach is successful and is ready for PDP. This status is obtained from the command AT+CGATT.
7. Carrier - This shows the name of the current GSM carrier.
8. GSM BSC mode - This shows the current setting for the GSM BSC mode which determines how the GoIP selects a base station. For more information, please refers to the section 3.3.15.
9. Cell ID - This shows the Base Transceiver Station (BTS) ID.
10. LAC - This shows the Location Area Code.

The bottom table shows more detailed information on the onboard GSM modules and the SIM card inserted.

1. Module - The model number of the GSM module.
2. Firmware Ver - The version number of the firmware installed in the module.
3. SIM Number - The GSM number that is assigned to the SIM card. User must enter this number manually.
4. IMEI - International Mobile Station Equipment Identity
5. IMSI - International Mobile Subscriber Identity
6. ICCID - Integrated Circuit Card Identifier

3.2.4 SIM Call Forward

The table below lists the current call forward settings of the SIM card assigned to the corresponding channel. There are 3 possible status:

1. ON - This means that the corresponding Call Forward mode is enabled and this setting is sent to the GSM network when a new GSM registration takes place.
2. OFF - This means that the corresponding Call Forward mode is disabled and this setting is sent to the GSM network when a new GSM registration takes place.
3. Not Set - This means that there is no change to the current Call Forwarding mode and nothing is sent to the GSM network when a new GSM registration takes place. This is useful by leaving the current Call Forward mode unchanged.

SIM Call Forward

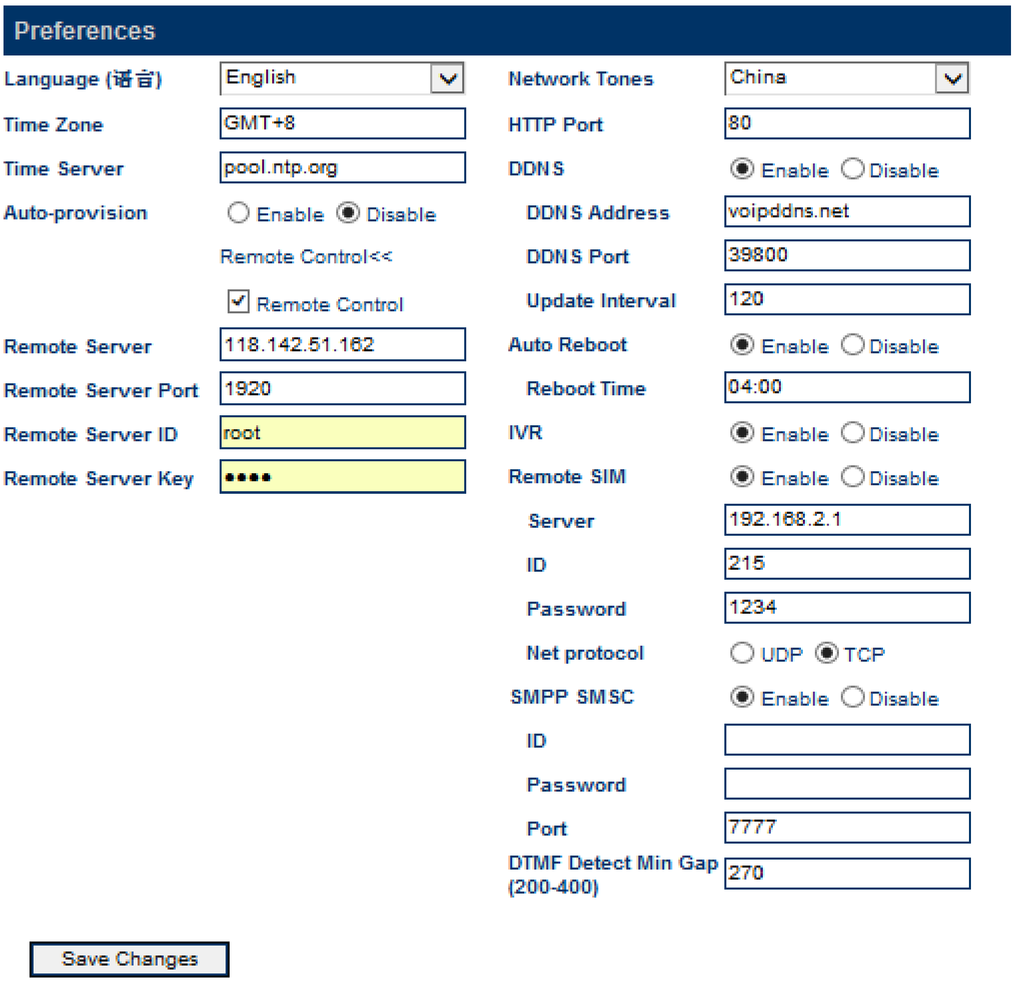
CH	Always	Busy	No Answer	No Service
1	Not Set	Not Set	Not Set	Not Set
2	Not Set	Not Set	Not Set	Not Set
3	Not Set	Not Set	Not Set	Not Set
4	Not Set	Not Set	Not Set	Not Set
5	Not Set	Not Set	Not Set	Not Set
6	Not Set	Not Set	Not Set	Not Set
7	Not Set	Not Set	Not Set	Not Set
8	Not Set	Not Set	Not Set	Not Set

3.3 Configuration

Click “Configuration” on the left hand column to display the Configuration page and the following submenu.

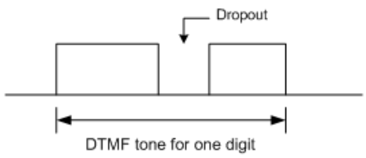
1. **Preference**
2. **Network**
3. **Basic VoIP**
4. **Advance VoIP**
5. **Media**
6. **Call Out**
7. **Call Out Auth.**
8. **Call In**
9. **Call In Auth.**
10. **SIM**
11. **SIM Forward**
12. **IMEI**
13. **SMS**
14. **GSM Carrier**
15. **GSM Base Station**

3.3.1 Preference



The preference page shown above consists of the following system level parameters and options as shown in the table below.

Parameter (Preference)	Description	Default Value
1. Language	This sets the webpage and voice prompts language. Currently, only English and Simplified Chinese (Mandarin for voice prompt) are supported.	English
2. Time Zone	This specifies the offset of the local time zone with respect to GMT. The syntax should be "GMT±x" where x is the offset.	
3. Time Server	This specifies IP address or the domain name of a network time server for computer clock synchronization. The default is "pool.ntp.org".	pool.ntp.org
4. Auto-provision	The auto provision is optional. When this option is enabled, the device downloads its configuration from the Auto Provision Server at start up or at the time interval specified by the Provision Interval. The configuration file name is <Serial Number>.cfg which is just a text file (not encrypted). If encrypted format is required, please contact technical support for further assistance. Please note that Auto Provision Server is a free utility supporting both Linux and Window environment. Please visit our website or contact technical support for more information.	
➢ Provision Server	The specifies the Provision Sever address (IP or Domain name)	
➢ Provision Interval	This specifies the interval in performing an auto provisioning event.	
5. Remote Control	This is a unique feature that allows remote access to the device's built-in Web server even when it is installed behind NAT. To achieve this function, a Remote Control Server is required to be installed. This server is a free Linux based utility and is available for download via our website. Please contact technical support for further assistance if required. Once installed, please make sure that the Remote Server Port and Password are set properly.	
➢ *Remote Server*	This specifies the IP address or the domain of the Remote Control Server.	
➢ *Remote Server Port*	Check with your Remote Server administrator for the communication port.	1920
➢ *Remote Server ID*	This specifies the name to be appeared in the Remote Control Server. It is used as a reference for the device.	
➢ *Remote Server Password*	This specifies the login password to the Remote Control Server. This is not the password to login to the built-in webpage. Please ask your Remote Server Administrator if it is not available.	
6. Network Tones	Network tones are the tones associated with the traditional (PSTN) telephone network, such as dial tone, ring back tone, busy tone, call waiting tones, etc. These tones will only be used when the device answers an incoming call and the call is not forwarded to a SIP server automatically. Predefined Network Tones are classified by country name. If the country desired is not found in the list, the "Custom" selection allows users to define the network tones individually. Please refer to Appendix B for more information.	
7. HTTP Port	This sets the port that is used to access the built-in web server. The default port number is 80. The port range is from 1 to 65535.	

8. DDNS ➢ DDNS Address ➢ DDNS Port ➢ Update Interval	This is a proprietary DDNS service offered by HyberTone. It allows HYBERTONE's products to identify each other via this DDNS service. When this service is activated, the domain name of the device is its <serial number>.com. This feature is useful to support peer-to-peer configuration. The default DDNS Address is "voipddns.net" which a free service offered by HYBERTONE. Please contact your vendor if you want to install your own DDNS server. The default communication port number is 39800. This specifies the interval between registrations to the DDNS.	 voipddns.net 39800 120 (mins)
9. Auto Reboot ➢ Reboot Time	This option allows the device to reboot itself at the time defined by Reboot Time. This parameter specifies the time to reboot the device. Two formats are supported: 1. HH:MM - When this is specified with a valid 24-hr time format (00:00 to 23:59), the goip is rebooted at this specified time. Invalid time specified has no effect. 2. M - This specifies the reboot duration in minutes. The valid range for this is from 0 to x. Changes saved are only effective after the device is rebooted.	Disabled
10. IVR	The device is equipped with a simple voice prompt. When this option is enabled and a call is answered, the device plays a voice prompt instead of a dial tone to the caller.	Enabled
11. Remote SIM ➢ Server ➢ ID ➢ Password ➢ Net Protocol	Only the GoIPs with the serial number xxxx support the Remote SIM feature. Enabling this feature allows the SIM Cards to be installed in a SIM Bank rather than in the on-board SIM slots. GoIP can either register to a SIM Bank or a SIM Server. Please refer to the SIM Bank User Manual for more information. This specifies the IP address of the SIM Bank or the SIM Server. This specifies the name to be appeared in the SIM Bank or the SIM Server. This specifies the login password to the SIM Bank or the SIM Server. Specify the network protocol (UDP or TCP) is used for Remote SIM communications.	Disabled
12. SMPP SMSC	This parameter enables the support of SMPP protocol. Please note that GoIP is acting as a SMSC (Short Message Service Center). Fill in the SMPP ID, Password, and port number for SMPP data communications.	
13. DTMF Tone Min Gap (200 - 400)	This parameter specifies the maximum dropout time for a DTMF tone. When making a call from SIP to GSM or from GSM to SIP by using the second dial method, the device needs to detect the dialing digits from the DTMF tones received via the voice data stream.  Depending on the network conditions, short dropouts may occur due to packet jitter / loss. Therefore, DTMF digit may be detected more than once if these dropouts are not taken into account. Consequently, the call is dialed to an incorrect number. To avoid this problem, a dropout window is used to avoid false detection when dropouts occur. During this window, the same DTMF digit is not recognized more than once. The range of the dropout window is specified in terms of packet timestamp value. The smaller the value is, the smaller the dropout window is. This increase the chance of detecting the same digit twice or more. However, if the value is set too large, there is a possibility that the next digit is missed.	270

3.3.2 Network

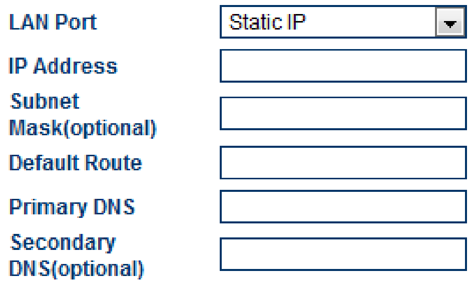
Proper network environment is the key to insure the voice call performance of the device. In general, Intranet offers a more stable network environment than Internet and it is the preferred network to be used. If Internet is going to be used, please make sure that the network can offer low packet loss, small packet jitter and low packet delay. Each voice channel requires less than 90 kbps when A-law or μ-law voice codec is used. GoIP-8 will require 8 times this bandwidth. Therefore, it is very important to make that both upstream and downstream have enough bandwidth (+ 30% headroom) in order to accommodate the data traffics for the device installed when all lines are used simultaneously.

In order to get external network access, the LAN port must be configured according to the network environment to be connected.

LAN Port

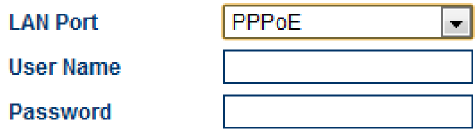
There are 3 access methods available to configure the LAN port.

1. Static IP – This mode applies to both public and private IP network environment. In the LAN port configuration shown on the left, select “Static IP” and then fill in the parameters as provided by your network administrator.



2. DHCP (default setting) – When the device is installed behind NAT and a DHCP host is available, select “DHCP” to enable the device to obtain LAN IP address and other network information automatically.

3. PPPoE – ADSL and Cable modems very often use PPPoE dial up to obtain network IPs. If this is the case, select “PPPoE” and then enter the information as provided by your ISP.

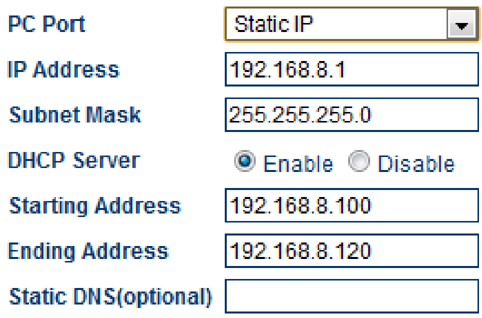


PC Port

The PC Port allows other network devices to be attached to the device in order get network connection. It offers both Router and Bridge modes to meet your requirements.

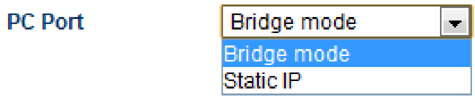
1. Static IP (default setting) – This mode enables the device to create another network segment and it then functions as a router/gateway for this new network segment. Select “Static IP” for this new segment and then enter the PC port IP address and Subnet Mask accordingly.

   It also has a built-in DHCP server to assign IPs to the devices attached to this network segment. Enable it and then enter the “Starting Address”, “Ending Address”, and “Static DNS” as required.



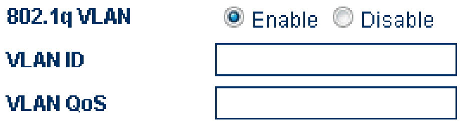
   As a factory default, the PC port is set to “Static IP” (Router mode) with IP Address set to 192.168.8.1 and Subnet Mask to 255.255.255.0.

2. Bridge Mode – Select this mode if your network topology requires the network devices attached to the PC port to be in the same network segment as the LAN port.

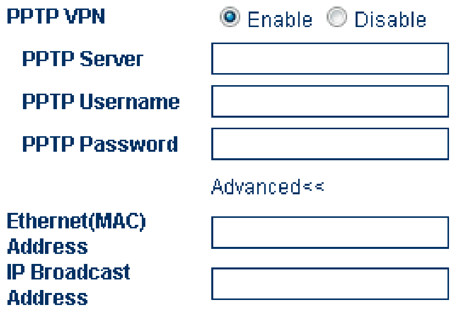


Advanced Features

1. VLAN – This is a type QoS service and is intended to give higher transmission priority to real time packets. However, your router switch and ISP network need to support this feature as well.



2. PPTP VPN – This option allows the device to create a VPN tunnel with the designated VPN Server. The VPN protocol supported is PPTP with no encryption or 40-bit encryption which is defined on the VPN server. In general, this option is used to avoid VoIP blockings.



3.3.3 Basic VoIP

The GoIP can support both SIP and H.323 VoIP protocols. For GoIP-1, both protocol are embedded in one firmware. User needs to select the VoIP protocol as shown below.

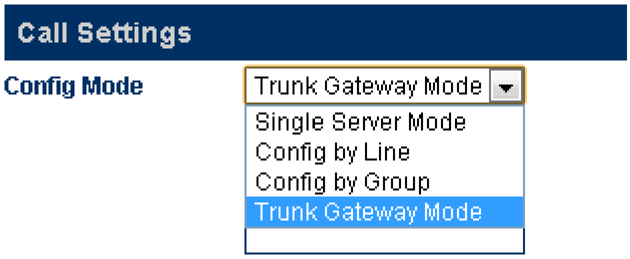
As more features are added, SIP and H.323 VoIP protocols are supported in two different firmware versions. Except GoIP-1, all other models are now shipped with the SIP protocol firmware as a factory default. If H.323 protocol is required, the firmware of the device can be changed to the one that supports H.323 protocol. Please visit our website for the latest firmware versions or contact us or your supplier for the latest firmware upgrade links available.

In general, it is important to understand your VoIP application with the device before proceeding to device configuration. If the device is going to work with a IP PBX, please make sure that you know how to configure your IP PBX. It is very important that you send us your application requirements in full details when seeking for technical support in configuring the device.

In order to simplify SIP configuration, SIP settings are categorized as Basic VoIP, Advanced VoIP and Media. In general, Basic VoIP defines how the GoIP handle SIP calls and four SIP modes are supported. It is important to understand the differences between each mode in order to select a mode that is the most suitable for your application. Depending on the SIP environment and network conditions, you may or may not need to change the default settings in the Advanced VoIP and Media pages.

Once SIP settings are completed, it is important to configure the device for making outgoing calls and receiving incoming calls. Please see section 3.3.6 and 3.3.7 for more information on Call OUT and Call IN settings.

The four modes of SIP operations are described below.



1. Single Server Mode

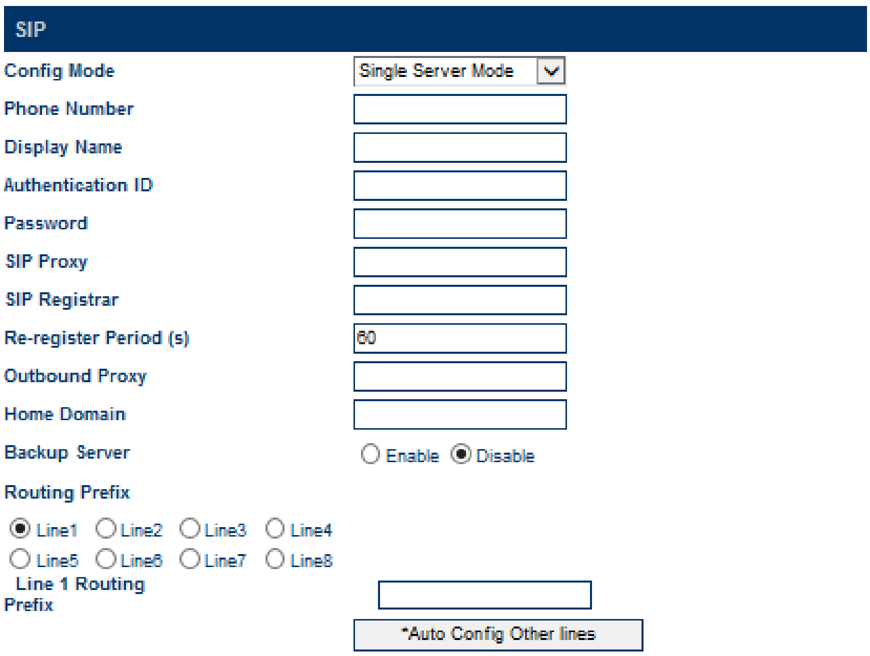
In this mode, only one SIP registration is used for single or multiple-line operation. Please make sure that your SIP server supports multiple-line operation and the SIP account is configured in the SIP server to match the number of lines available in the device. Call routing to a GSM channel is now based on the **Routing Prefix** of each GSM channel. Here is the channel selection algorithm.

a) All **Routing Prefixes** are set
   - Try to match the number received for making an outgoing call against the **Routing Prefix** of each channel. If only one match is found, the corresponding channel is used to make the outgoing call. If more than one matches are found, the best available channel among the matched channels is selected. If no match is found, the call is rejected by sending back a SIP 503 message.

b) Only some **Routing Prefixes** are set
   - Try to match the number received for making an outgoing call against those **Routing Prefix**es that are set. If only one match is found, the corresponding channel is used to make the outgoing call. If more than one matches are found, the best available channel among the matched channels is selected. If no match is found, an idle channel without a Routing Prefix is used to make the call. Otherwise, the call is rejected by sending back a SIP 503 message.

c) None of the **Routing Prefixes** are set
   - An idle channel is selected to dial out the call. If no idle channel is available, the call is rejected by sending back a SIP 503 message.

It is important to note that the Routing Prefix (P) must be removed via the dial plan before the number is dialed out. The dial plan syntax to remove the Routing Prefix is "P:-P|". Please refer to section 3.3.6 for more information on the Call Out Dial Plan.

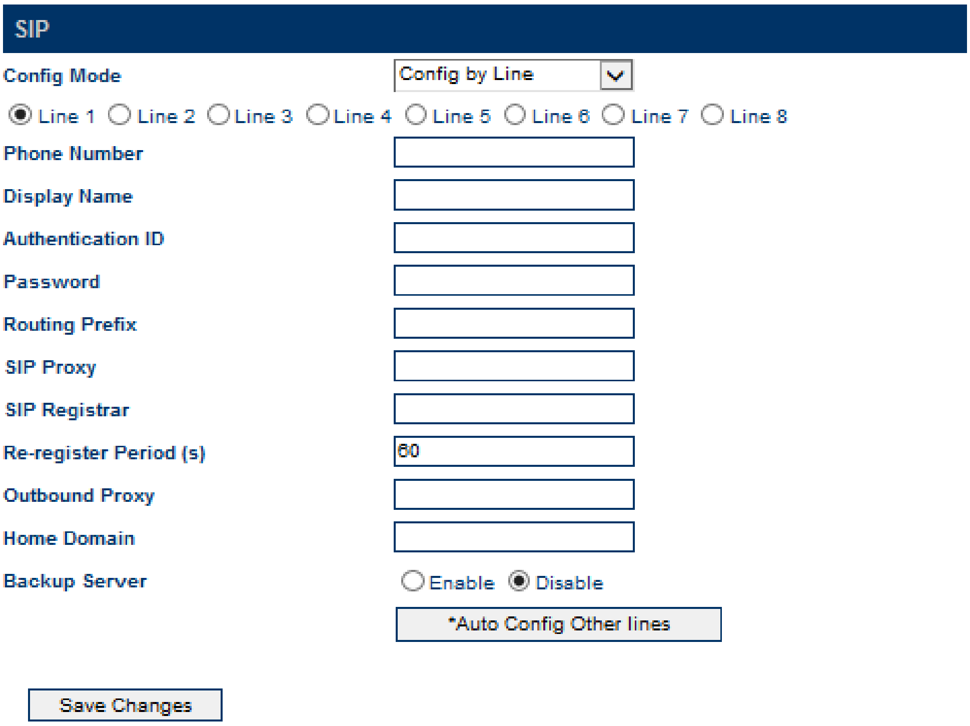


Parameter (Single Server Mode)	Description	Default Value
1. Phone Number	This is the SIP number used by the device.	
2. Authentication ID	The name of the device used in Caller Identification is defined here.	
3. Display Name	The Authentication ID used for SIP registration is specified here.	
4. Password	The password used for SIP registration is specified here.	
5. SIP Proxy	The domain name or IP of the SIP Proxy or Server is specified here. If the SIP Proxy is using the standard 5060 signaling port, then there is no need to specify the port number. Otherwise, the port number can be specified by adding ":" and then the port number at the end of the SIP Proxy address.	
6. SIP Registrar	The address of the SIP Registrar Server is specified here.	
7. Re-register Period (s)	Register to the SIP Server at an interval specified by this parameter.	60
8. Outbound Proxy	The address of the Outbound Proxy used for VoIP communication is specified here.	
9. Home Domain	Home Domain is used in SIP identification. It should be specified as required.	
10. Backup Server ➢ SIP Proxy ➢ SIP Registrar ➢ Home Domain	Backup Server improve service reliability and is used only when the primary server fails. This specifies the backup SIP Server address. This specifies the backup SIP Registrar Server address. This specifies the backup Home Domain address.	Disabled
11. Routing Prefix	This parameter is used for call routing. When this is set, the corresponding channel is only used to dial out a phone number with the matching Routing Prefix. Syntax: <Prefix1>,<Prefix2>,<Prefix3>,..... where Prefix is a text string which consists of digits, alphabets, and special characters. The maximum length of the Routing Prefix is 120 characters.	

2. Config. By Line (for all models except GoIP-1) Mode

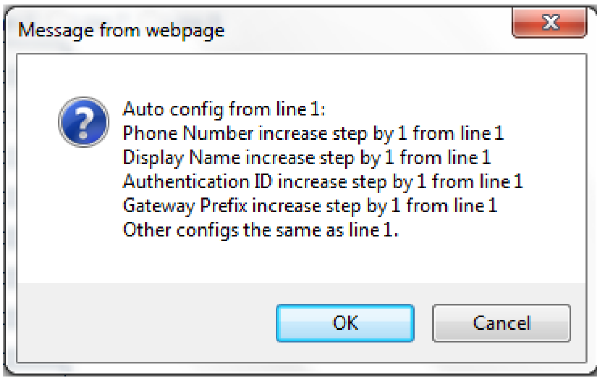
This mode is only applicable for multi-line models. Each line (associated with a corresponding GSM channel) registers to a SIP server separately and operates as an independent phone line. A Routing Prefix for each channel must be assigned in order to enable the channel to allow making outgoing calls. This allows to the SIP server to assign which channel to dial out the call for termination. If a channel does not have its Routing Prefix set, this channel will not be used to dial out any calls. In this mode, the prefix of the phone number to be dialed out must match one of the Routing Prefixes assigned. The channel with the matching Routing Prefix will be used to dial out the call. If no match is found, the call will not be dialed out and a SIP 404 message is returned to the SIP Server. If a match is found but no channel is available to dial out the call, a SIP 503 message is returned to the SIP Server. The syntax for the Routing Prefix is defined in the Parameter Table for Single Server Mode.

It is important to note that the Routing Prefix (P) must be removed via the dial plan before the number is dialed out. The dial plan syntax to remove the Routing Prefix is "P:-P|". Please refer to section 3.3.6 for more information on the Call Out Dial Plan.



When a GSM channel receives an incoming call, the call can either be answered by the device or forwarded to a SIP extension or IVR. For more details, please see Section 3.3.8 for Call In configuration.

Please note that the parameters defined in this mode are for each line. Their definitions are the same as those defined in the parameters table for Single Server mode. The "*Auto Config Other lines" option is only available for Line 1. Clicking this button after Line 1 is configured will automatically configure other lines with the Line 1 settings with the changes displayed in the following message box.



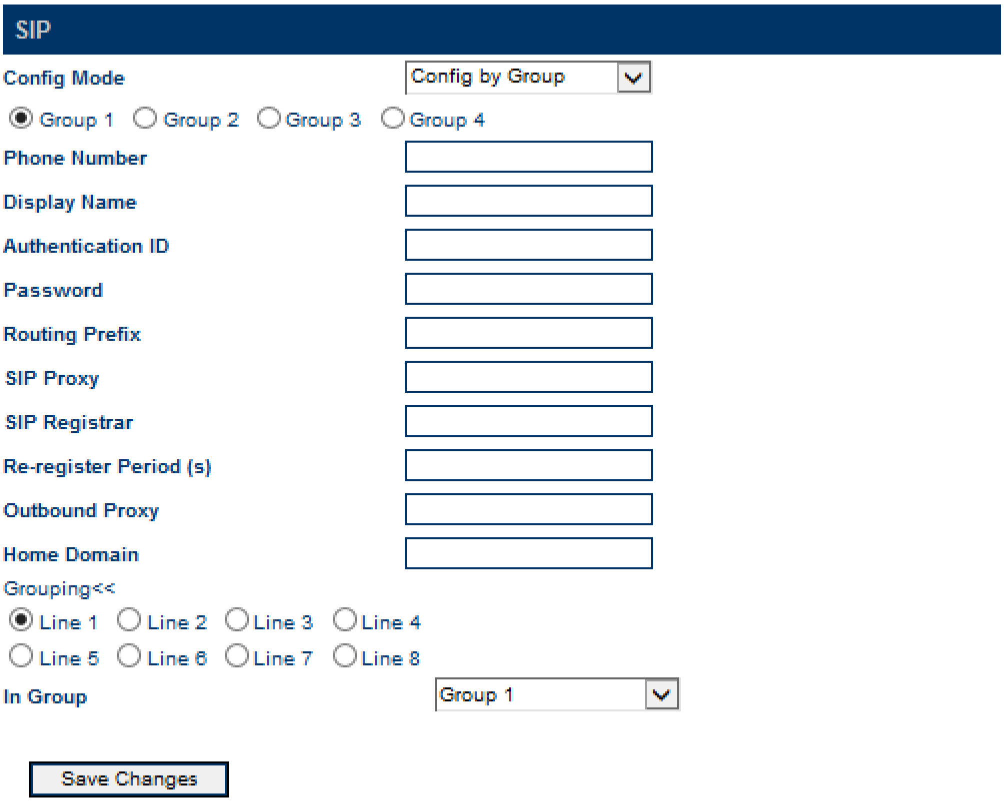
3. Config. By Group (for all models except GoIP-1)

This mode is basically a combination of Single Server mode and Config. By Line mode. It allows lines to be split up into groups. Each group only uses one SIP registration for all the lines assigned to the group.

Each line can be assigned to only one of the 4 predefined groups in the Grouping section.

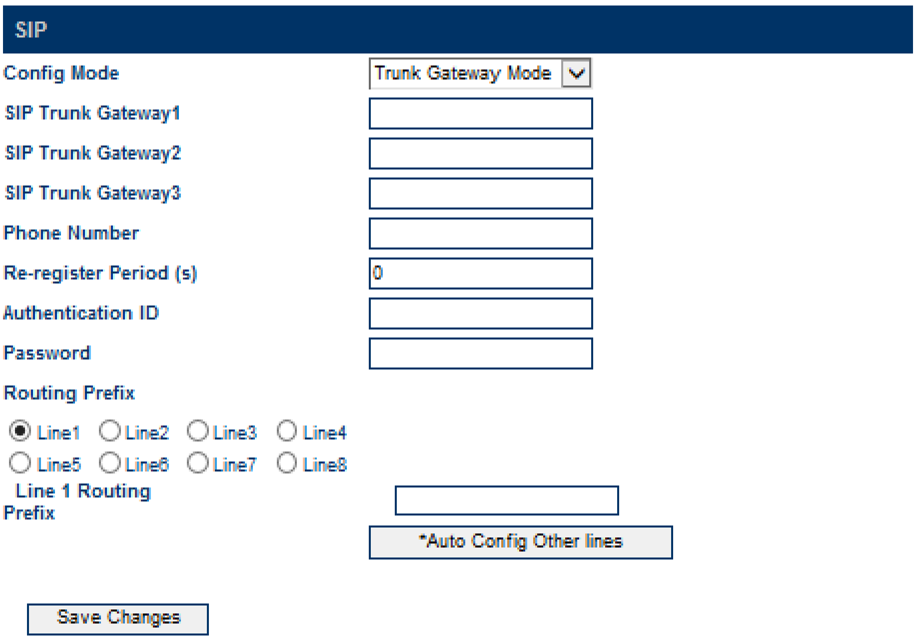
In this mode, the Routing Prefix is assigned to the Group rather than to the Channel. Its function is the same as the Routing Prefix for Config. by Line. The syntax for the Routing Prefix is defined in the Parameter Table for Single Server Mode.

Please note that the parameters listed in this figure are the same as the parameters defined in the Config. By Line mode except these are parameters for a "Group" rather than for a "Line". Backup Server and Grouping. However, these parameters are group properties rather than line properties.



4. Trunk Gateway Mode (for all models except GoIP-1)

   This mode offers a seamless interface with the SIP Trunk configuration in a SIP server. In general, SIP registration is not required in this mode; this is an advantage for a SIP system that requires a license per registration. If SIP server and the device are not in the same network segment, it is recommended that both SIP server and the device are using public IPs for reliable operation. Installing either one or both (SIP server and the device) behind NAT may or may not work properly depending on the SIP Server and the routers (on each side) as well. In this case, SIP server must support NAT. Some routers may map an internal port to a different external port number. Otherwise, VoIP calls may fail to establish properly in this network environment.



The device accepts calls from up to 3 IP addresses (SIP Trunk Gateway1, SIP Trunk Gateway2, SIP Trunk

Gateway3) specified and then dial out the call via an Idle channel that is used the least (in terms of the number of calls dialed). The last part of the SIP Trunk Gateway IP addresses can be specified as "X" or "x" to represent that the whole segment IP addresses (0 - 255). Calls originated from the IP segment are accepted.

Example: SIP Trunk Gateway2 = 123.124.125.x

This example shows that Calls originated from 123.124.125.0 to 123.124.125.255 are accepted.

Call routing to a GSM channel is now based on the **Routing Prefix** of each GSM channel (Line x). The channel selection algorithm is the same as the one described in the Single Server Mode. The syntax for the Routing Prefix is defined in the Parameter Table for Single Server Mode.For received GSM calls, they will be routed to SIP Trunk Gateway1 provided that an unique IP is used.

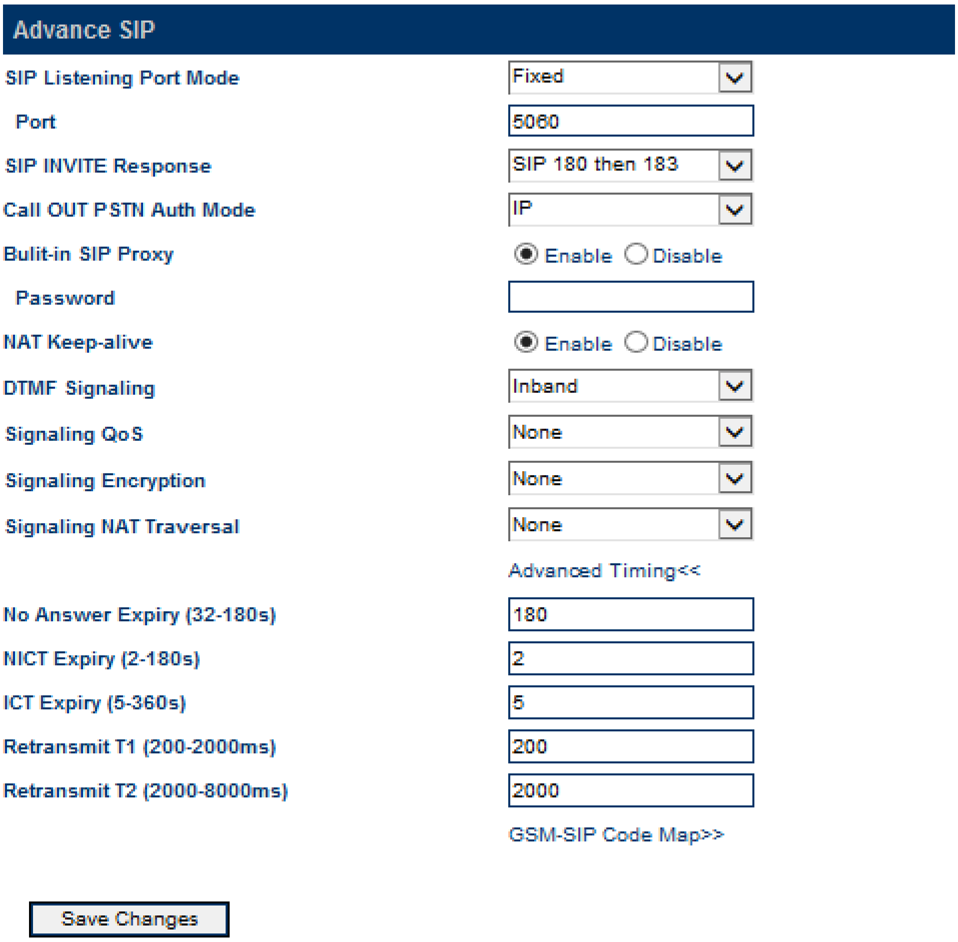
SIP Registration is only supported for SIP Trunk Gateway1. Fill in the SIP registration information as required. The Re-Register Period must be set to a non-zero value. If SIP Registration is not required, the Re-Register Period must be set to zero.

The parameters available in this mode are listed in the table below.

Parameter (Trunk Gateway mode)	Description	Default Value
1. SIP Trunk Gateway1	This specifies the first SIP Trunk Gateway address. Use "X' or "x" for the last part of the address to specify the entire segment (0 - 255).	
2. SIP Trunk Gateway2	This specifies the second SIP Trunk Gateway address. . Use "X' or "x" for the last part of the address to specify the entire segment (0 - 255).	
3. SIP Trunk Gateway3	This specifies the third SIP Trunk Gateway address. . Use "X' or "x" for the last part of the address to specify the entire segment (0 - 255).	
SIP Registration to Trunk Gateway1 Some Trunk gateway connection requires a SIP registration which can be defined via the following parameters.		
4. Phone Number	This specifies the phone number for the SIP registration.	
5. Authentication ID	This specifies the authentication ID for the SIP registration.	
6. Password	This specifies the password for the SIP registration.	
7. Re-register Period (s)	This specifies the period for sending a re-registration request.	0

3.3.4 Advanced VoIP

The parameters in the Advance VoIP section are common for all configuration modes. In general, these parameters are preconfigured with factory defaults. Users should only modify the parameters required.



The table below summaries all the parameters defined in this section.

Parameter (Advanced VoIP)	Description	Default Value
1. SIP Listening Port Mode	SIP Local port defines the network port number that the device listens for incoming SIP messages. This port number is sent to the SIP Sever/Proxy during SIP registration. This setting defines if this port is pre-assigned to a fixed number or a randomly generated port number (5060 to 6060).	
➢ Port Number	This specifies the port number when the SIP Local Port Mode is set to “Fixed”.	5060
2. SIP INVIITE Response	One of the key function of the device is to allow call terminations from VoIP to GSM. In general, a VoIP caller dials a PSTN or GSM number (E.164) number and the SIP Server routes this call to the device by sending a SIP INVITE message. This parameter specifies the response to the INVITE message. The three possible responses are described in details below. 1. SIP 200 OK - This response inform the SIP server that the call is answered and the call duration timers starts immediately. If billing applies, the call is charged (immediately) even before the call is answered. 2. SIP 180 then 183 - The device first sends back a SIP 180 Ringing to the calling SIP device to generate a local ring. The caller hears a ringback tone immediately after the call is dialed. When a ringback tone is received from the GSM network, a SIP 183 Session In Progress message is sent to the calling party to start early media (before the call is answered). This allows to the caller to hear the ringback from the GSM network. This is done in order to avoid a long silent period before a ringback tone is returned from the GSM network. 3. SIP 183 - The device sends back a SIP 183 Session In Progress message to the calling SIP device. The calling SIP device then goes into early media mode to	SIP 180 then 183

	receive audio packets. Since it may take a few to over 10 seconds for a ringback tone is returned from the GSM network, the caller may hear a long silent period	
3. Call OUT Auth. Mode	This setting defines how incoming VoIP calls are authenticated when the device is configured for using SIP registration(s). This setting applies to Single Server Mode, Config. By Line and Config. by Group modes. This prevents unauthorized calls to be dialed out via the GSM Channel(s). The following authentication methods are available: 1. None - No authentication is used for calls received. This could be a simple arrangement if calls are routed from a SIP Server in the same local network. 2. IP – only calls received from the registered SIP Server(s) are accepted. 3. Password – A SIP 401 message is sent to the SIP server for password authentication of the corresponding SIP account when a call is received. 4. IP and Password – Both authentication methods are used.	**IP**
4. Built-in SIP Proxy ➢ Password	A simple SIP Proxy is embedded in the device. Choose "Enable" to activate this SIP proxy to accept any SIP registrations with the correct password which is specified in the parameter "Password". There is no need to create a SIP account in this server. Users will have to manage the SIP numbers used on their own. This facilitates the setup of a simple SIP network for customers who do not have their own SIP servers. This sets the password for SIP registration to the built-in SIP server.	Disabled
5. NAT Keep-Alive	When enabled, NAT Keep Alive sends a NULL packet to the router regularly in order to keep the network ports used open.	Enabled
6. DTMF Signaling ➢ Outband DTMF Type ➢ RTP Payload Type	This setting specifies the DTMF dialing method. 1. Inband – DTMF tones are generated in the form of audio stream. 2. Outband – DTMF digits are sent in the form of digital commands (RFC2833 / SIP INFO). DTMF tones are actually generated by the terminating party. This parameter is for outband DTMF dialing. Select the proper format (RFC2833 or SIP INFO) as required by your SIP network. This parameter specifies the payload type in RFC 2833 commands.	Outband 2833 101
7. Signaling QoS	This specifies the QoS method used for SIP signaling. Both IP TOS and DiffServe format are supported. Select the proper setting that is compatible with your network environment.	None
8. Signal Encryption	Signaling encryption is employed to offer a more secure environment for SIP communications. The following encryption methods are supported. Consult your network/VoIP administrator for more the proper selection if required. 1. RC4 2. Fast 3. VOS 4. AVS 5. N2C 6. ECM 7. ET263 8. XOR	None

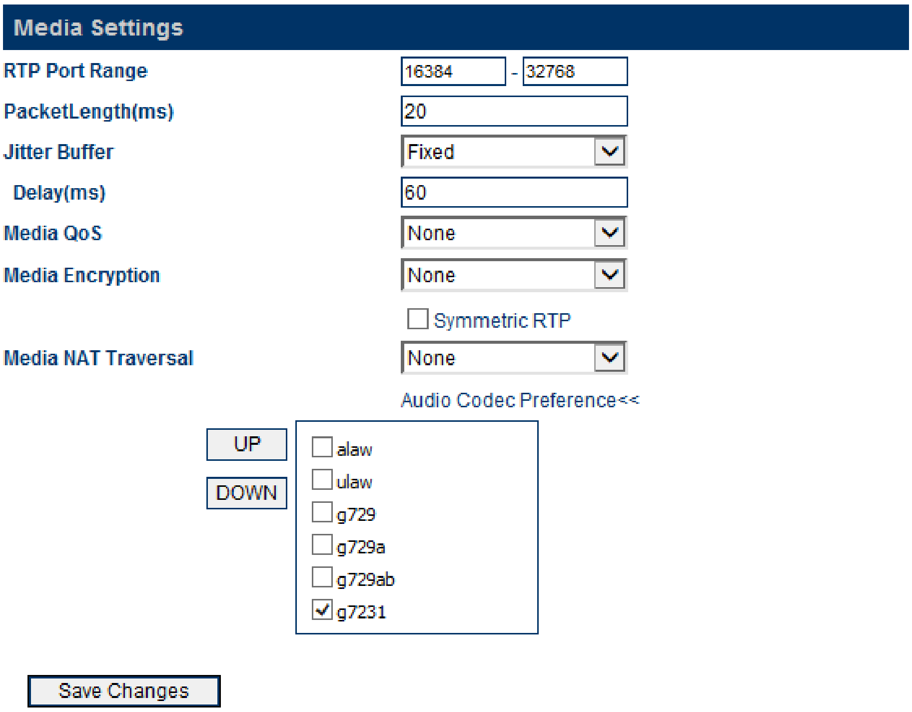
9. Signaling NAT Traversal	This setting is not required if the target SIP server / PBX supports NAT traversal. However, if your ISP blocks VoIP traffics, you could try to use Relay Proxy setting. Depending on how your ISP blocks VoIP traffics, the Relay Server method may or may not work in your network environment. Two NAT Traversal methods are supported: 1. Stun Server – An external Stun Server is required. This allows the device to obtain the public IP of the network used. 2. Relay Proxy – This is a proprietary method developed by HYBERTONE Technology. HYBERTONE's Relay Proxy server must be used. A free copy of the Relay Proxy can be downloaded from HYBERTONE's website (www.hybertone.com). Please contact support@hybertone.com for further assistance if needed.	None
10. Advanced Timings	This section consists of 5 basic timers in the SIP protocol. Configure them carefully so that they are compatible with the SIP Server and your requirements.	
➢ No Answer Expiry (32-180s)	This timer specifies the timeout for an unanswered call. A SIP 408 Request Timeout command is sent to the SIP Server when this timer expires. *Note: The default value is set the maximum value so that it will not interfere with the call unanswered timeout at SIP Proxy or PBX.*	180
➢ NICT Expiry (2-180s)	NICT: Non Invite Client Transaction (RFC 3261 Section 17.1.2)	2
➢ ICT Expiry (5-360s)	ICT : Invite Client Transaction (RFC 3261 Section 17.1.1)	5
➢ Retransmit T1 (200-2000ms)	Round Trip Time (RTT) estimate (RFC 3261 Section 17.1.1) This timer applies to the following timeout timer. 1. INVITE request retransmission interval, for UDP only 2. Non-INVITE request retransmission interval, UDP only 3. INVITE response retransmission interval.	200
➢ Retransmit T2 (20000-8000ms)	The maximum retransmit interval for non-INVITE requests and INVITE responses.	2000
11. GSM-SIP Code Map	When making a GSM call, a GSM specific cause defined in GSM04.08 Annex H is returned from the network for call control. This GSM cause must be returned back to the SIP server for call handling and control. In order to meet various application requirements, the GSM-SIP Code Map allows the user to define the corresponding SIP message for each GSM cause listed below .	

	GSM Reason	SIP Response Code	
	Unassigned (unallocated) number	404	
	No route to destination	503	
	Channel unacceptable	503	
	Operator determined barring	503	
	Normal call clearing	503	
	User busy	486	
	No user responding	503	
	User alerting, no answer	408	
	Call rejected	403	
	Number changed	404	
	Destination out of order	503	
	Invalid number format (incomplete number)	404	
	Facility rejected	503	
	Normal, unspecified	503	
	No circuit/channel available	503	
	Network out of order	503	
	Temporary failure	503	
	Switching equipment congestion	503	
	Access information discarded	503	
	Requested circuit/channel not available	503	
	Resources unavailable, unspecified	503	
	Others	486	

3.3.5

3.3.6 Media

This section allows the user to program various settings for media (voice) transmission and format. Depending on your network environment and condition, you may or may not need to change these settings. Please see the parameter table below for more information.



<table>
<tr><td>Parameters
(Media)</td><td>Description</td><td>Default Value</td></tr>
<tr><td>1. RTP Port Range</td><td>This specifies the range of RTP port to be used for audio stream.</td><td>16384 - 32768</td></tr>
<tr><td>2. Packet length (ms)</td><td>This specifies the length (in time) of each packet. However, the packet length is codec dependent as well. The minimum packet length of a codec supersedes the valued specified here. The table below summarizes the possible packet length for the codec supported.
<table>
<tr><td>Codec</td><td>Time / Frame (ms)</td><td>Time / Packet (ms)</td></tr>
<tr><td>G.711 a-law / µ-law</td><td>0.125</td><td></td></tr>
<tr><td>G.729, G.729A, G.729AB</td><td>10</td><td>10, 20, 30</td></tr>
<tr><td>G.723.1</td><td>30</td><td>30, 60</td></tr>
</table></td><td>20</td></tr>
</table>

<table>
<tr>
<td>3. Jitter Buffer

➢ Delay

➢ Min. Delay (ms)

➢ Max. Delay (ms)</td>
<td>A jitter buffer is designed to remove the effects of jitter from the decoded voice stream, buffering each arriving packet for a short interval before playing it out. This substitutes additional delay and packet loss (discarded late packets) for jitter. If a jitter buffer is too small then an excessive number of packets may be discarded, which can lead to call quality degradation. If a jitter buffer is too large, then the additional delay can lead to conversational difficulty..

A fixed jitter buffer maintains a constant size whereas an adaptive jitter buffer has the capability of adjusting its size dynamically in order to optimize the delay/discard tradeoff.
Three modes of jitter buffer are supported:
1. Fixed - The fixed mode, which is the default mode, is a simple first in first out mode, with a fixed jitter buffer delay.
2. Sequential - The sequential mode is also a fixed jitter buffer delay mode, but in this mode the jitter buffer function looks at the packet timestamp for dropped or out of sequence packet problems. The data packets are sorted based on the packet timestamp
3. Adaptive - The adaptive mode optimizes the size of the jitter buffer delay and depth in response to network conditions, in addition to the sequential mode functions.

This specifies the fixed jitter delay for both Fixed and Sequential Jitter Buffer mode.

This specifies the minimum jitter delay for Adaptive Jitter Buffer mode.

This specifies the maximum jitter delay for Adaptive Jitter Buffer mode.</td>
<td>Fixed

60</td>
</tr>
<tr>
<td>4. Media QoS</td>
<td>Similar to Signaling QoS, this parameter enables the QoS property for audio packets. Both IP ToS (Type of Service) and DiffServe (Differentiated Service) method are supported.</td>
<td>None</td>
</tr>
<tr>
<td>5. Media Encryption</td>
<td>Similar to Signaling Encryption (item 6 in this table), this parameter enables the encryption for audio packets. Encryption methods supported are RC4 and ET263.</td>
<td>None</td>
</tr>
<tr>
<td>6. Symmetric RTP</td>
<td>Network environment in some enterprise may require Symmetric RTP. Please check with your network administrator for further support.</td>
<td>Disabled (not selected)</td>
</tr>
<tr>
<td>7. Media NAT Traversal</td>
<td>This setting is not required if the target SIP server / PBX supports NAT traversal. However, if your ISP blocks VoIP traffics, you could try to use Relay Proxy setting. Depending on how your ISP blocks VoIP traffics, the Relay Server method may or may not work in your network environment.
Two NAT Traversal methods are supported:
1. Stun Server – An external Stun Server is required. This allows the device to obtain the public IP of the network used.
2. Relay Proxy – This is a proprietary method developed by HYBERTONE Technology. HYBERTONE’s Relay Proxy server must be used. A free copy of the Relay Proxy can be downloaded from HYBERTONE’s website (www.hybertone.com). Please contact support@hybertone.com for further assistance if needed.</td>
<td>None</td>
</tr>
<tr>
<td>8. Audio Codec Preference</td>
<td>Six types of audio codec are supported and they are summarized in the table below.

<table><tr><td>Codec</td><td>Raw Data Bandwidth (bps)</td><td>Ethernet 802.3 Data Bandwidth (bps)</td></tr><tr><td>1. a-law</td><td>64K</td><td>~ 85K</td></tr><tr><td>2. μ-law</td><td>64K</td><td>~85K</td></tr><tr><td>3. G.729</td><td>8K</td><td>~39K</td></tr><tr><td>4. G.729A</td><td>8K</td><td>~39K</td></tr></table></td>
<td>All codecs are enabled in the order of preference shown below.

a-law
μ-law
G.729
G.729A
G.729AB
G.723.1</td>
</tr>
</table>

5. G.729AB (with Silence Compression and Voice Activity Detection (VAD)	8K	~39K
6. G.723.1	5.3K / 6.4K	~26K / 27K

Note: Time per packet = 30ms is used for all bandwidth calculations. For more calculations with other conditions, please visit the VoIP Bandwidth Calculator website http://www.bandcalc.com/.

Place a "tick" mark in the check box enable the corresponding codec. The codecs are listed in a descending order of priority for codec selection. This means that the top one in the table will have the highest priority to be selected when establishing a call. To change the priority, select the desired codec and the click on "UP" or "DOWN" button on the left.

Note: The effective bandwidth for G.729AB is less since less data are transmitted when there is no voice activity.

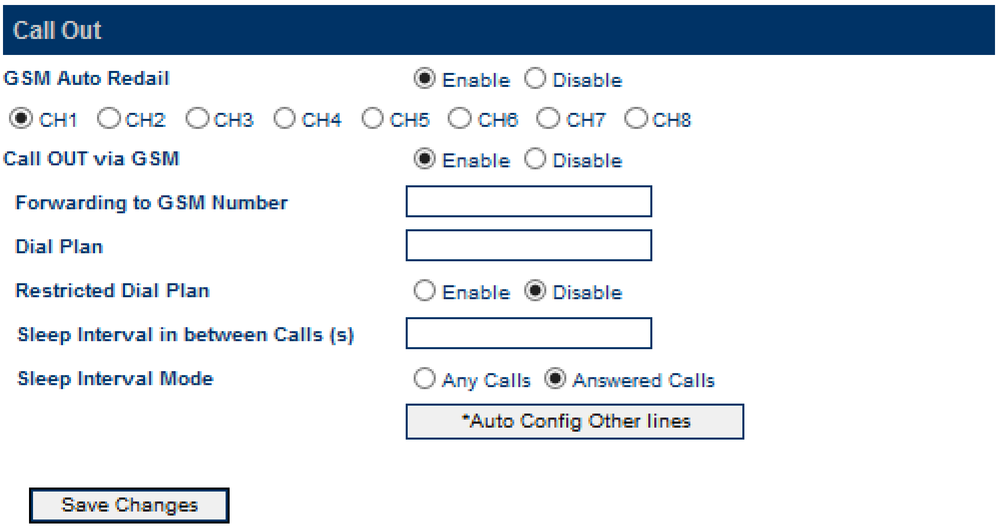
3.3.7 Call OUT

The Call Out page defines how each GSM channel handles calls when they are routed from VoIP. This section MUST Be defined properly in order to enable each GSM channel to dial out calls based on your requirements. In general, you can achieve the followings.

1. Forward all incoming VoIP calls to a fixed GSM or PSTN number.
2. Dial out all incoming VoIP Calls based on the phone number received.
3. Use the Dial Plan to manipulate the phone number received and then dial out the modified number.
4. Use the Restricted mode to only dial out the phone numbers that match the rules defined in the dial plan.
5. Use the Idle Interval to prevent calls from dialing out during this period. This Idle Interval can be set to be active for any calls or for only answered calls.
6. Only all authorized calls to be dialed out.

The ***Auto Config. Other Line** button is provide to facilitate the programming of each channel. After the parameters for CH1 are set, clicking this button automatically duplicates the same settings to all other channels.

Please note that how a VoIP call is routed to a GSM channel depends on the Configuration mode selected. Please refer to the Basic VoIP section for more information.



The parameters available in this section are described in details in the table below.

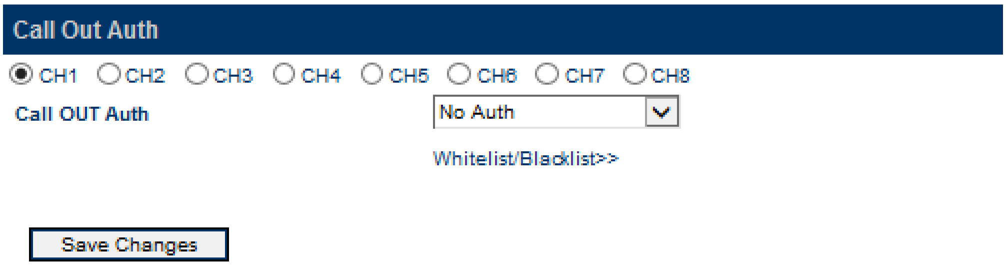
Parameter	Description	Default Value
1. GSM Auto Redial	This is a general parameter for all channels.	Enabled
2. Call OUT via GSM	This setting defines if the device is allowed to make outgoing calls via the on-board GSM channel(s). The typical application is to terminate VoIP calls via the GSM network. This setting If this parameter is specified, GoIP dials this phone number via the corresponding GSM channel whenever an incoming VoIP call is received for this line. This is a fixed forwarding method and has the highest priority. This means that the Dial Plan setting does not apply in this case. Please note that how this line is selected depends on the Config Mode and the Callee	

	Number received. The Callee Number is defined as the phone number specified in the "To:" field of an INVITE message. Please refer to Section 3.3.3.1 for more information. Calling the SIP number directly routes the call to the corresponding line immediately. If this parameter is blank and the Callee Number equals to the SIP Number defined for this line, a second dial tone is generated to wait the caller to dial a phone number. Please note that this could only happen in the Single Server mode, Config by Line mode, and Config by Group mode since SIP registration is required. If this parameter is blank and the Callee Number does not equal to the SIP Number defined for this line, GoIP dials out the Caller Number according to the Dial Plan defined.	
3. Forwarding to GSM Number	If this parameter is specified, GoIP dials this phone number via the corresponding GSM channel whenever an incoming VoIP call is received for this line. This is a fixed forwarding method and has the highest priority. This means that the Dial Plan setting does not apply in this case. Please note that how this line is selected depends on the Config Mode and the Callee Number received. The Callee Number is defined as the phone number specified in the "To:" field of an INVITE message. Please refer to Section 3.3.3.1 for more information. Calling the SIP number directly routes the call to the corresponding line immediately. If this parameter is blank and the Callee Number equals to the SIP Number defined for this line, a second dial tone is generated to wait the caller to dial a phone number. Please note that this could only happen in the Single Server mode, Config by Line mode, and Config by Group mode since SIP registration is required. If this parameter is blank and the Callee Number does not equal to the SIP Number defined for this line, GoIP dials out the Caller Number according to the Dial Plan defined.	
4. Dial Plan	The Dial Plan specifies rules to modify the Callee Number before dialing it out. Each rule is terminated with the delimiter "|". The rule matching begins from the left to the right. Once a match is found, rule matching terminates and the actions specified in the rule are executed. Syntax: a:-b+c| The portion on the left side ("a") of ":" specifies the prefix for matching starting from the beginning of the Callee Number. The right portion "-b+c" is the action to be taken. Both "-b" and "+c" are optional. "-b" means that "b" is removed from the beginning of the Callee Number. If "b" is not found starting from the beginning of the Callee Number. "+c" means that "c" is added to the beginning of the Number generated from the last action. In order for this rule to be meaning, "b" must be the same "a" or the beginning portion of "a". Example: Callee Number = 9262124567 Dial Plan = 9:-9+852| Actual Number dialed = 85226124567 Syntax: a[b-c]:-d+e "b-c" specifies the range of a single digit. Together with "a", they form a prefix for rule matching. "a" can be a single or multiple digits. "-d+e" are the actions to be taken as described in the previous syntax. Example: 913[5-9]:-9+86| In this rule, Callee Numbers starting with 9135, 9136, 9137, 9138, 9139 meet the prefix requirement. The first action is to remove the first digit "9" from the number and then append "86" to the beginning of the number. If the Callee Number is 913601234567, the actual number dialed is 8613601234567.	

5. Restricted Dial Plan	This option forces the GoIP to only dial out the phone numbers that match the rules defined in the dial plan.	
6. Sleep Interval in between Calls	This setting defines an idle interval in between calls. During this interval no outgoing calls are allowed to be made via the GSM channel.	
7. Sleep Interval Mode	This parameter defines the condition of a call when the Sleep Interval is activated. 1. Any Calls - The Sleep interval is activated whenever an outgoing call is dialed regardless whether the call is completed successfully or not. 2. Answered Calls - The Sleep Interval is activated whenever an outgoing call is answered.	

3.3.8 Call OUT Auth

The Call Out page defines how each GSM channel handles calls when they are routed from VoIP. This section MUST Be defined properly in order to enable each GSM channel to dial out calls based on your requirements. In general, you can achieve the followings.



Parameter	Description	Default Value
1. Call OUT Auth	This parameter defines how incoming VoIP calls are authenticated before dialing them out via the GSM network. Five options are available: 1. None – No authentication is required. Calls are always dialed out via the GSM network. 2. Password – The caller is prompted for entering the password before the call is dialed out. 3. Whitelist – This is part of the Call Screen function. Only the caller numbers listed on the Whitelist list are allowed to dial out via the device. 4. Password or Whitelist – Either the password or the Whitelist authentication method will be used. 5. Biacklist – The caller numbers listed on this list are rejected.	None
2. Whitelist / Blacklist	Both Whitelist and Blacklist for call screening are supported. Each list contains up to 15 entries. 1. Whitelist – This list contains a list of caller numbers that are allowed to use the device to make outgoing GSM calls when Call OUT Authentication is set to “Whitelist” or “Whitelist + Password”. Remark: Adding a "#" digit in front of an Whitelist entry enables a special call back function. When the caller ID of an incoming call is matched, the device first drops the call and then call back the caller automatically to allow the caller to dial a phone number. 2. Blacklist – This list contains a list of caller numbers that are rejected by the device to make outgoing GSM calls when Call OUT Authentication is set to “Blacklist”.	

3.3.9 Call IN

This page defines how incoming calls to each GSM channel are handled. GSM incoming calls can either be answered and prompted for second dial or forwarded to a phone extension or IVR in the VoIP network connected. The parameters in the Call In Page are divided into two sections. The top section contact the parameters that are applicable to all channels and the bottom section are for each channel individually.

1. The ***CID Forward Mode*** is applicable intended to forward the GSM caller ID to VoIP and for all channels. Please refer to Appendix E for more details'

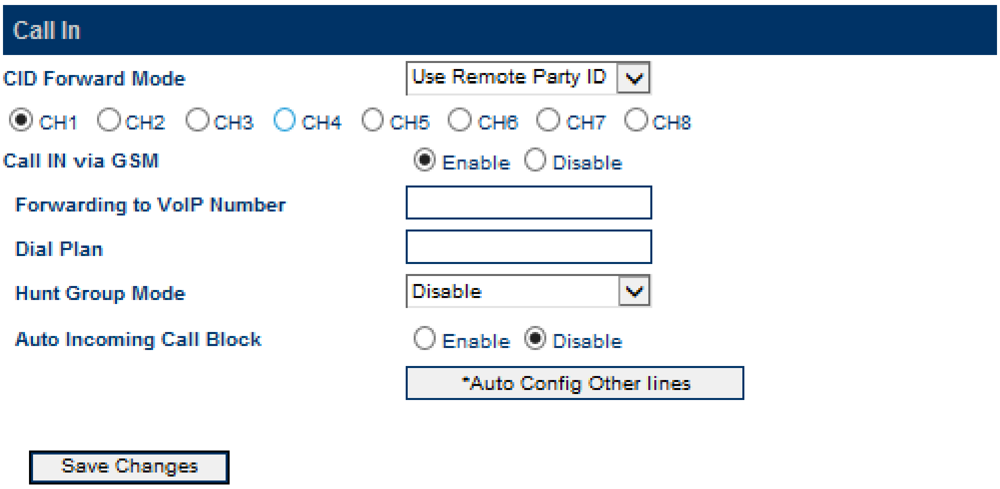
2. The ***Call IN via GSM*** parameter must be enabled in order to receive GSM incoming calls.

3. The ***Hunt Group Mode*** is use to simulate "Call Center" operation where a single GSM number is used for all incoming calls. To enable this function, set the Hunt Group mode of one GSM channel to "Host" and all other channels to "Client". Please do not set more than one channel to Host as it may cause the Hunt Group mode not to function properly. In addition, the GSM Forwarding for all "Client" channels must be disabled.

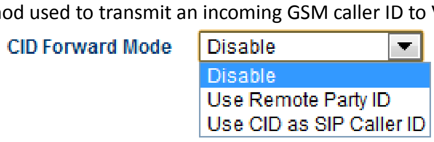
4. The ***Call IN Auth*** is used authenticate the incoming calls. Calls can be accepted or rejected based on this setting and the corresponding White list or Black list.

5. The ***Auto Incoming Call Block*** is used to block an incoming call when the same number is called consecutively for more than the limit defined. Customer has requested this feature to block test calls that are sent by the carrier.

6. The ****Auto Config. Other Line*** button is provide to facilitate the programming of each channel. After the parameters for CH1 are set, clicking this button automatically duplicates the CH1 settings to all other channels.



The parameters defined in this section are described in details in the Table below.

Parameter	Description	Default Value
1. CID Forward Mode	This specifies the method used to transmit an incoming GSM caller ID to VoIP. 	Enabled

<table>
<tr><th></th><th></th><th></th></tr>
<tr><td></td><td>1. Disabled – Incoming GSM caller ID is not transmitted to SIP.

2. Use Remote Party ID – This enables the Remote-Party-ID field is sent as part of the INVITE message. The incoming GSM caller ID is specified as part of this field.

3. Use CID as SIP Caller ID – This causes the CID information in an INVITE message is replaced by the incoming GSM Caller ID.</td><td></td></tr>
<tr><td>➢ CID Prefix</td><td>This parameter is intended to modify the incoming GSM number.

Syntax = a:-b+c|

The portion on the left side (“a”) of “:” specifies the prefix for matching. The right portion “-b+c” is the action to be taken and they are optional. When the beginning of the CID number matches “a”, the first action “-b” is to removed “b” from the CID Number. The second action “+c” is to add “c” to the beginning of the number that is produced from the first action. Please note that a, b, c could be a single digit or a sequence of digits and they are independent.

Example 1:
CID Prefix = 9:-9+852|
CID Number = 9262124567
Actual CID Number forwarded = 85226124567</td><td></td></tr>
<tr><td>2. CALL IN via GSM</td><td>This setting controls the device to accept (select “Enable”) or reject (select “Disable”) incoming calls via the selected GSM channel.</td><td>Enable</td></tr>
<tr><td>3. Forwarding to VoIP Number</td><td>This parameter defines if an incoming call to the selected GSM channel is forwarded immediately to the VoIP network or not.

If this parameter is blank, the device answers an incoming GSM call. If IVR (in the Preference section) is enabled, the device generates a voice prompt to ask the caller to dial an extension number; otherwise, it generates a second dial tone.

If a phone number is assigned to this parameter, a SIP INVITE to the “Forward Number” is sent to the SIP Server or the SIP trunk address. This number must be a number that can be recognized and accepted by the VoIP network registered. This means that it could be an extension number in the VoIP network or an E.164 number. For E.164 number, the VoIP network must be setup properly for dialing out via another trunking service. Since the device can be used for trunking, it is possible to set it up to route an incoming GSM call from one channel and dial out to another party with an E.164 number via another GSM channel.

<u>Special Feature</u>
Conditional forwarding is implemented to forward an incoming GSM call based on its caller ID.

Syntax= a>b|

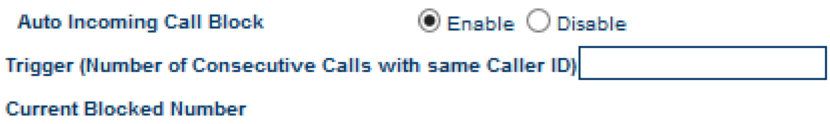
“a” is a complete or portion of a number for matching with the incoming caller ID.

“b” is the number to be dialed via the VoIP network. It could be an extension number or an E.164 number.

Example:
98765432>108|>101|
In this example, GoIP first try to match the incoming GSM caller ID with the number 98765432. If it is a match, GoIP dial the number 108. If the first rule does not match, it will continue to the second rule. There is no matching number for the second rule. It is then considered as a match and GoIP dials the number 101.

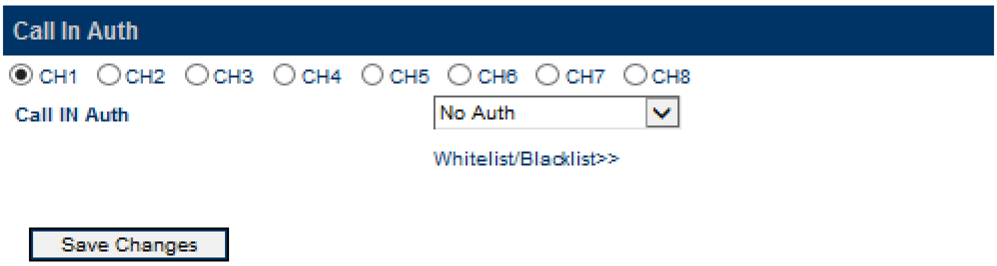
The maximum length for this parameter is 140 ASCII characters. The number of rules can be adopted is limited by this length. Each rule must end with the “|” character.</td><td></td></tr>
</table>

	When there is no match, the incoming GSM call is handled as if the Forward Number is blank.	
4. Dial Plan	The Dial Plan specifies rules to process a number dialed after the call is answered. This number is referred as a second dial number. This enables a way to recognize second dial numbers that are in a known format and dial them out immediately. Depending on the VoIP network connected, a second number could be an extension or an E.164 number. Syntax 1= a:-b+c| The portion on the left side (“a”) of “:” specifies the prefix for matching. The right portion “-b+c” is the action to be taken and they are optional. When the beginning of the callee number matches “a”, the first action “-b” is to removed “b” from the Callee Number. The second action “+c” is to add “c” to the beginning of the number that is produced from the first action. Please note that a, b, c could be a single digit or a sequence of digits and they are independent. Example 1: Dial Plan = 9:-9+852| Number received = 9262124567 Actual Number dialed = 85226124567 Syntax : XXX…XXX: | This syntax monitors the length of the number dialed when performing second dial operation. Each “X” represents a single digit. If a prefix is known, X’s can be replaced by the prefix. Example: Dial Plan: 13XXXXXXXXX: | This rule monitors the number dialed with the starting prefix of 13 and a length of 11 digits. Once this condition is met, the number will be dialed out immediately. The maximum length for the Dial Plan definition is 140 ASCII characters. There is no limit on the number of rules defined. Each rule must be ended with the “|” character. The rule matching starts from the beginning and stops once a match is found.	
5. Hunt Group Mode ➢ Forward Mode	Hunt Group operation is discussed in details in Appendix D. Please note that Hunt Group Mode is a property of each GSM channel and is required to be set individually. **Host** - This enables the channel selected to be the Host of the Hunt Group operation. All clients registers and update the host on their channel status. The host then maintains a list of clients status and selects an idle channel to receive the next incoming GSM call via GSM call forwarding. Please note that each GoIP cannot have more than one Host channel assigned. Sharing of GSM channels between two Hunt Groups is supported. When all channels in one group are in use, Call Forward will be set to the Host channel of the other group. When a free channel is available again, Call Forward is then set to the free channel instead. To enable this feature, the Backup Host Address must be set to the IP of the GoIP which contains the Host channel of the other Hunt Group.  This parameter determines if the Host channel is going to be used for incoming call or not. If the Forward Mode is set to Always, the Host channel will not be used for answering any	Disable

➢ Backup Host Address ➢ Host Address	incoming calls. If it is set to Busy, then the Host channel will always to be used to answer an incoming call whenever it is available. Setting this parameter enables Hunt Group Sharing feature. This parameter must set the IP of the GoIP which contains the Host channel of the other Hunt Group. **Client**- This assigns the selected channel to be a client in Hunt Group mode.  The Host Address field specifies the IP address of the device that contains the Host channel. If a Client and its Host belong to the same device, its device IP is entered in this parameter. A client registers and updates its channel status to the Host.	
6. Auto Incoming Call Block ➢ Trigger	This option is used to block an incoming calls when the condition specified in the **Trigger** is met. The block operation is released when an incoming call with a different Caller ID is received. This sets the number of consecutive calls with the same Caller ID required to trigger the Incoming Call Block function. 	Disable

3.3.10 Call IN Auth

This page defines how incoming calls to each GSM channel are handled. GSM incoming calls can either be answered and prompted for second dial or forwarded to a phone extension or IVR in the VoIP network connected. The parameters in the Call In Page are divided into two sections. The top section contact the parameters that are applicable to all channels and the bottom section are for each channel individually.

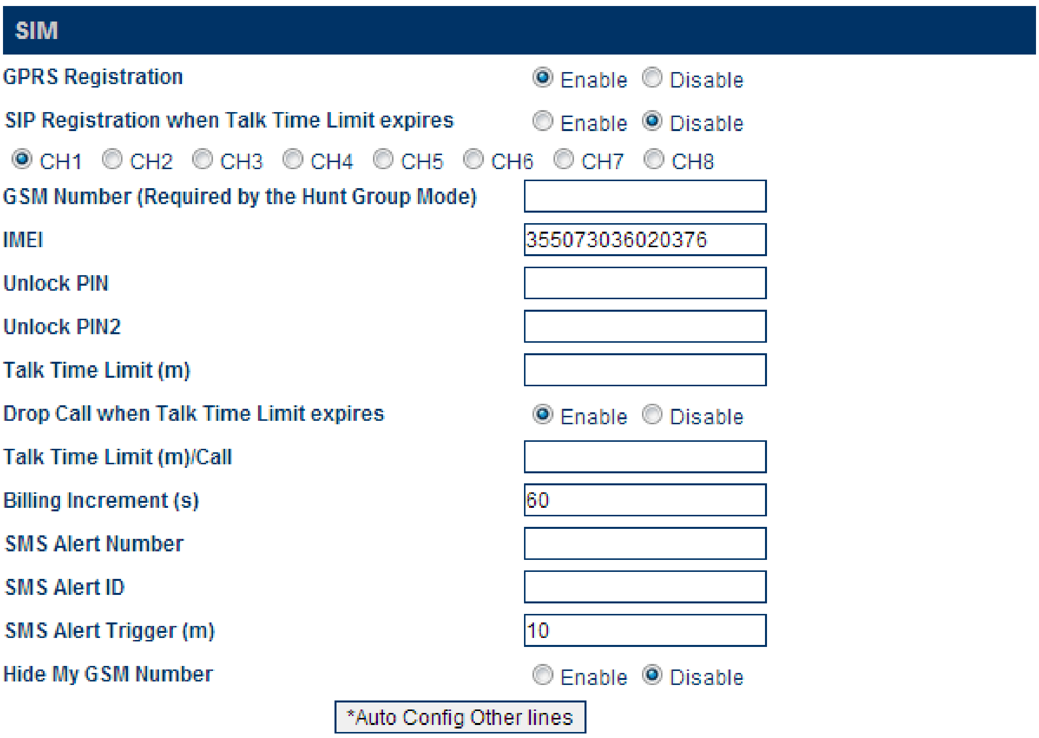


Parameter	Description	Default Value
1. Call In Auth	This parameter defines how incoming GSM calls are authenticated before routing calls to the VoIP network connected. Five options are available: 1. None – No authentication is required; all incoming GSM calls are routed to VoIP. 2. Password – The caller is prompted for entering the password before the call is routed or a second dial tone is generated. 3. Whitelist – Only calls with caller IDs that are listed on the Whitelist are accepted by the GSM channel selected. Calls with GSM numbers not on the list are not answered at all. 4. Whitelist + Password – Both Whitelist and password are used to authenticate incoming	None

	GSM calls. 5. Blacklist – Calls with caller IDs that are listed on the Blacklist are not answered by the channel selected.	
2. Whitelist / Blacklist	Call screen list can be set to Whitelist or Blacklist. A maximum of 15 entries is allowed. 1. Whitelist – This list contains a list of incoming GSM caller numbers that are accepted (answering calls from these numbers only) by the device when Call IN Authentication is set to “Whitelist” or “Whitelist + Password”. 2. Blacklist – This list contains a list of incoming GSM caller numbers that are rejected (not answering calls from these numbers) by the device when Call Authentication is set to “Blacklist”. This setting specifies the Call Forward method when the GSM channel is configured as the server in HUNT Group mode. 1. Unconditional Call Forward – Incoming calls are always forwarded to an idle channel. If all Client channels are in use, the Host channel answers an incoming call. 2. Call Forward Busy – The Host channel only forward in incoming call when the line is busy.	

3.3.11 SIM

This section contains a set of parameters are related to the SIM Card of the channel selected. The ***Total Talk Time Limit*** and the ***Talk Time Limit / Call*** can used to limit the phone usage of the SIM selected. It is useful for customers who are using a sim card that has a higher calling rates when a certain limit is exceeded or at different time periods. Please program this section as required. The ****Auto Config. Other Line*** button is provide to facilitate the programming of each channel. After the parameters for the CH1 SIM are set, clicking this button automatically duplicates the same settings to all other channels with the exception that the SMS Alert ID is incremented by 1 with respect to the ID of the previous channel.



The parameters for this section are described in details in the table below.

Parameter (SIM Card)	Description	Default Value
1. GPRS Registration	This enables the GPRS registration mode for all GSM channels.	Enable
2. SIP Registration when Talk Time Limit Expires	This parameter determines if SIP Registration is enabled or disabled when Talk Time Limit expires. If it is enabled, incoming calls are still enabled when Talk Time Limit expires.	Disable
The parameters below only apply to the channel selected.		
3. GSM Number	This specifies the phone number of the SIM Card that is inserted to the channel selected. This field **MUST** be specified when the Hunt Group mode is enabled. Otherwise, GSM Call Forward cannot be setup properly.	
4. IMEI	This specifies the International Mobile Equipment Identity number. The device comes with a factory default value. Once changed, this value cannot be restored.	Factory Default
5. Unlock PIN1	SIM Card unlock PIN code 1	
6. Unlock PIN2	SIM Card unlock PIN code 2	
7. Talk Time Limit (m)	There are two ways to define the Talk Time Limit for a SIM card. 1. Total talk time without any time limit - When the accumulative talk time since the last reset reaches this limit, the SIM card is disabled. The SIM card must be reset via the status page or via a special SMS command (see Appendix A). Syntax: <limit> <limit> is an integer. 2. Talk time allowed as per the period defined - When the accumulative talk time since the period starts reaches the Talk Time Limit defined, the SIM card is disabled for outgoing calls until the period expires. If there is an overlap between two periods defined, the first period takes priority. The next period starts from the end of the first period. If there is a gap between two periods, the talk time for this gap is unlimited. It is important to note that the Talk Time Limit is reset automatically to the one defined for the next period. There is no need to reset a SIM via the webpage in this case. Syntax: <period1, limit1>|<period2, limit2>|.... <Period> is defined in 24-hr format with starting and ending time separated by a "-". Example 1: 00:00-24:00 This specifies a time period of one day starting from 00:00 till the end of the day. Example 2: 08:00-12:00,100|11:00-20:00,200 In this example, the first period starts from 08:00 till 12:00 with a talk time limit of 100 minutes. The second period starts from 12:00 till 20:00 with a talk time limit of 200 minutes. From 20:00 till 08:00 (the next day), there is no talk time limit. Note: By default, SIP Registration is disabled when a SIM card is disabled. To change this default, select the [Enable] option for SIP Registration when Talk Time Limit Expires in the SIM Page.	
8. Drop Call (when the Total Talk Time Limit Expires)	This parameter specifies if the active call is dropped or not when the Total Talk Time Limit expires.	
9. Talk Time Limit (m) / Call	This sets the limit for the maximum talk time per call. When this limit is reached, the call is dropped automatically.	

10. Billing Increment (s)	This is a call duration measurement unit expressed in seconds. Depending on your service provider, some services are measured and billed in sixty second increments (one minute) or the billing increment may be in durations of six or even ten seconds.	
11. SMS Alert Number	This specifies the GSM number to receive a SMS Alert on the Total Talk Time. If this parameter is blank, no SMS Alert will be sent.	
12. SMS Alert ID	This parameter is used to identify the channel sending the SMS Alert.	
13. SMS Alert Trigger (Remain Talk Time (m))	The Remain Time is the Total Talk Time Limit minus the total talk time used. When this Remain Time reaches the value set in the parameter, a SMS Alert is sent to the SMS Alert Number automatically.	
14. Hide My GSM Number	This parameter determines if the caller party can receive the phone number of the caller or not. Enabling this parameter hides My GSM number from the called party. This specified if the phone number of the caller is shown at the called party or not.	

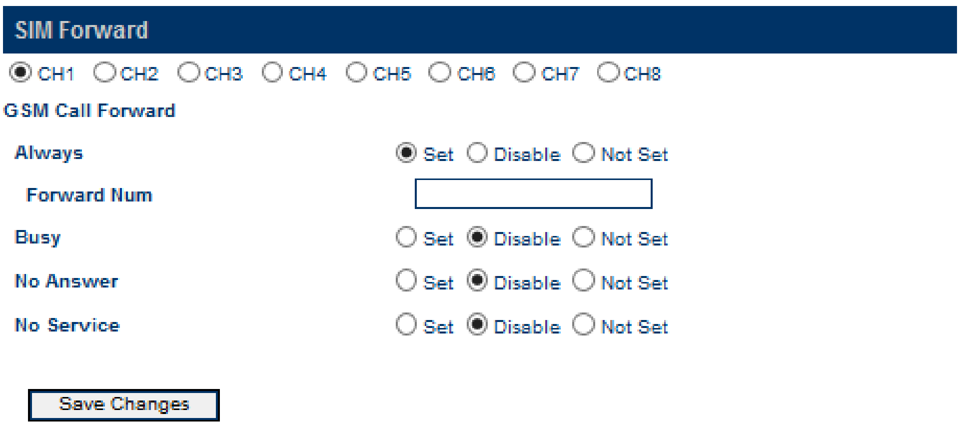
3.3.12 SIM Forward

This section allows the user to define the GSM Call Forward settings for each channel. There are four Call Forward conditions:

1. Always - Forward all incoming calls unconditionally
2. Busy - Forward all incoming calls when busy.
3. No Answer - Forward the incoming calls when it is not answered.
4. No Service - Forward all incoming calls when the SIM cannot register to the network.

There are 3 choices for each Call Forward condition.

1. Set - This enable the Call Forward function. This setting is sent to the network whenever the SIM is starting a new registration to the network.
2. Disable - This disable the Call Forward function. This setting is sent to the network whenever the SIM is starting a new registration to the network.
3. Not Set - This leaves the current SIM Call Forward setting in the network unchanged. This setting is NOT sent to the network at all.

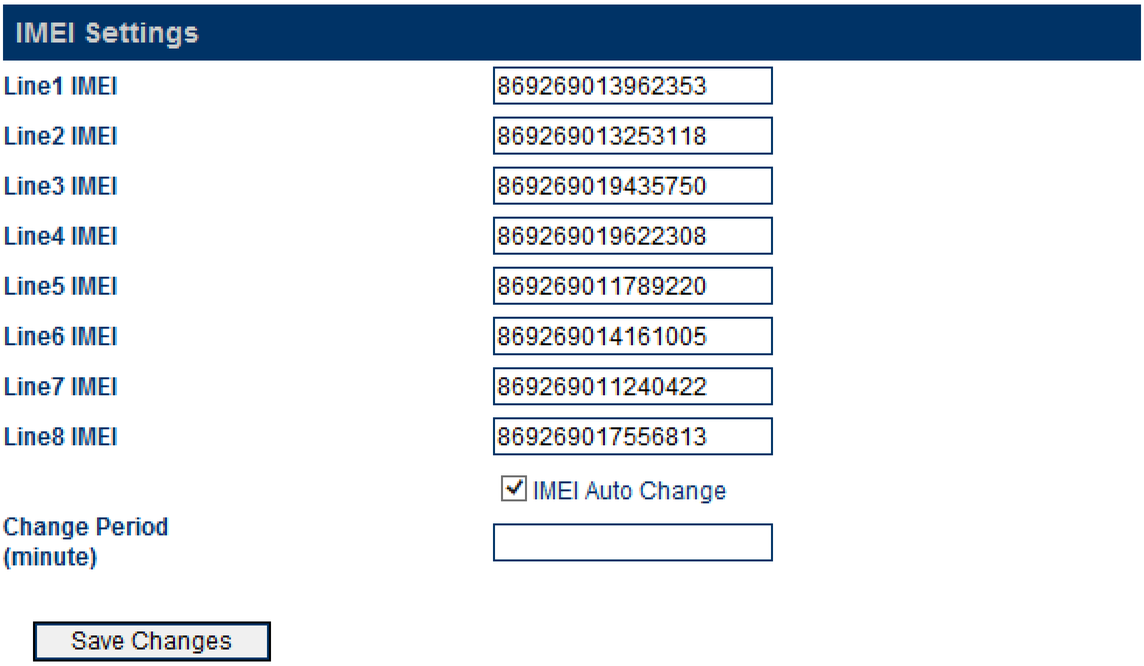


Parameter (SIM Forward)	Description	Default Value

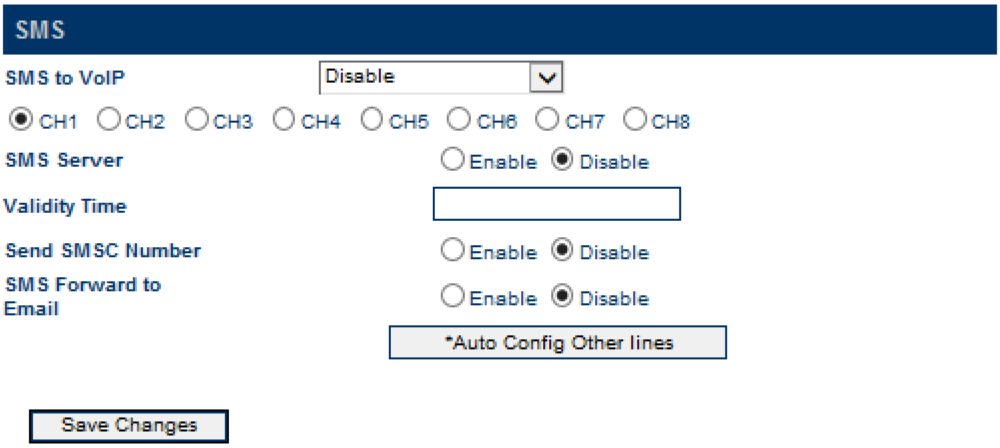
1. Unconditional Call Forward ➢ Forward Num	Forward all incoming calls unconditionally to the number specified. This specifies the phone number to receive forwarded calls under this condition.	Not Set
2. Call Forward Busy ➢ Forward Num	Forward calls when the GSM Channel is in use. This specifies the phone number to receive forwarded calls under this condition.	Not Set
3. Call Forward No Answer ➢ Forward Num	Forward the call when an incoming call is not answered. This specifies the phone number to receive forwarded calls under this condition.	Not Set
4. Call Forward Unreachable ➢ Forward Num	Forward calls when the GSM channel cannot register to the carrier. This specifies the phone number to receive forwarded calls under this condition.	Not Set

3.3.13 IMEI

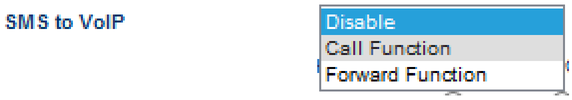
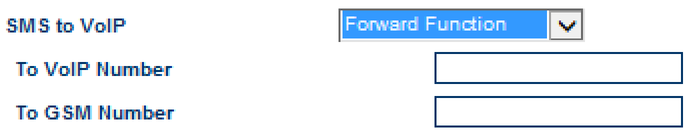
This section allows IMEI modifications for all Channel in a single page. At the bottom of the IMEI list, user can also enable an option to change the IMEI of each channel automatically at an interval defined by the ***Change Period***. The minimum Change Period allowed is 10 minutes which is currently set as the default value. Changing IMEI only occurs when the channel is idle. When it occurs, the GSM channel drops its current registration and then use the new IMEI to register to the network.



3.3.14 SMS



This page covers parameters that are related to SMS. The top parameter (SMS to VoIP) is a general setting for all channels. This defines how received SMS messages are used for Call Back or Forward functions which are described in details in the table below.

Parameter	Description	Default Value
1. SMS to VoIP ➢ Call Mode ➢ Dialing Prefix ➢ VoIP Number ➢ GSM Number	This defines how the device handles received SMS messages.  1. Call Function – This mode is used to support Call Back function via incoming GSM messages. Appendix B describes the three different modes of operations in order to meet the different requirements from various SIP servers. Please note that the SIP server registered must be configured for this operation.  This parameter specify which SMS Dial mode is used. Please refer to Appendix B for more information on the modes available. The parameter is applicable for SMS Dial mode. It allows a prefix to be added to the phone number of the called party. 2. Forward Function – This mode forwards incoming GSM SMS to a SIP terminal and a GSM number.  Incoming GSM messages (SMS) received are forwarded to the SIP Phone Number specified in this parameter. Incoming GSM messages (SMS) received are forwarded to the GSM Phone Number specified in this parameter. The channel received the message is used to forward the message.	Disabled

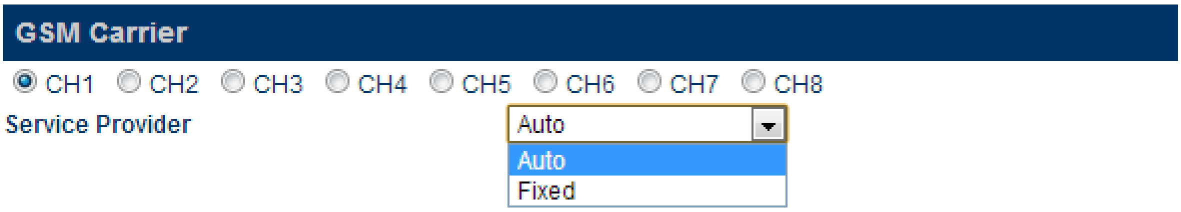
Next, the parameters defined below are channel dependent. Please make sure that the desired channle is selected before making changes.

Parameter (SIM Card)	Description	Default Value
1. SMS Server	SMS Server is a Linux based utility which is used to manage GoIPs registered to send and receive SMS messages. In addition, it also keeps an activity logs of each channel. It is a free utility offered by the GoIP's manufacturer and can be downloaded from the URL below. http://www.hybertone.com/en/download.asp For more information, please visit our website or contact *support@HyberTonetek.com*. Once the SMS Server is installed and in operation. The following parameters are require to enable the GoIP to register to the SMS Server. Please note that each channel must be programmed individually in order to register to the SMS Server.	
➢ SMS Server IP	This specifies the domain name or ip address of the SMS server.	
➢ SMS Server Port	This specifies the communication port that is used by the SMS server. This must match the port value set in the SMS server.	
➢ SMS Client ID	This specifies the login ID for the channel selected. SMS Client ID and password must first be created in the SMS Server.	
➢ Password	This specifies the login password for the SMS Client ID.	
2. Validity Period	This specifies the message expiration time in the Message Center. When this period expires, the undelivered message is discarded. The Validity Period (VP) is an integer from 0 to 255 and VP Value: 0 - 143 — Actual Time: (VP+1) x 5 minutes (i.e. 5 minutes intervals up to 12 hours) VP Value: 144 - 167 — Actual Time: 12 hours + ((VP-143) x 30 minutes) VP Value: 168 - 196 — Actual Time: (VP - 166) x 1 day VP Value: 197 - 255 — Actual Time: (VP - 192) x 1 week	
3. Send SMSC Number	When this option is enabled, the SMSC Number stored on the SIM card is read and sent to the Message Center when sending a message.	
4. SMS to Email	This enables the forwarding of SMS received via the channel selected to a designated email address.	
➢ Outgoing Email Server (SMTP)	This specifies the SMTP Server for sending emails.	
➢ User Name	This specifies the User Name for SMTP Server authentication. Leave this blank if authentication is not required.	
➢ Password	This specifies the Password for SMTP Server authentication. Leave this blank if authentication is not required.	
➢ Forwarding Email Address	This specifies the email recipient of the forwarded SMS.	

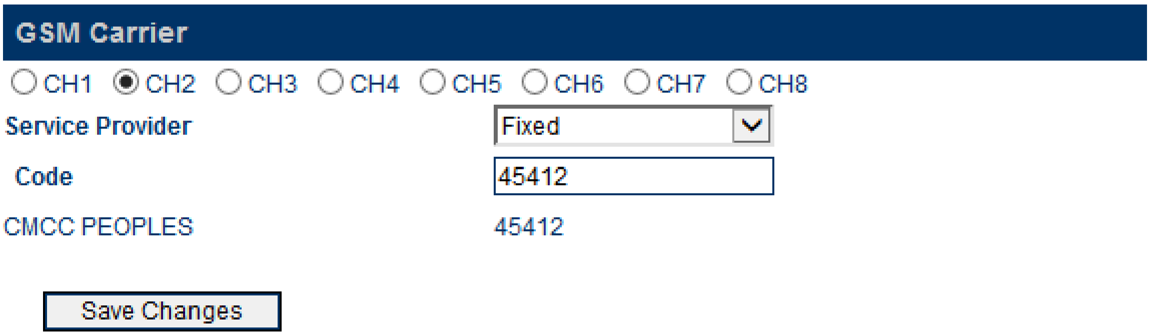
The ***Auto Config. Other Line*** button is provided to facilitate the programming of each channel. After the parameters for the CH1 SIM are set, clicking this button automatically duplicates the same settings to all other channels with the exception that the SMS Client ID is incremented by 1 with respect to the ID of the previous channel.

3.3.15 GSM Carrier

This section sets the mode of the GSM service provider selection. The factory default setting is "Auto" for automatic selection of GSM service provider based on the default preference set by the SIM card .

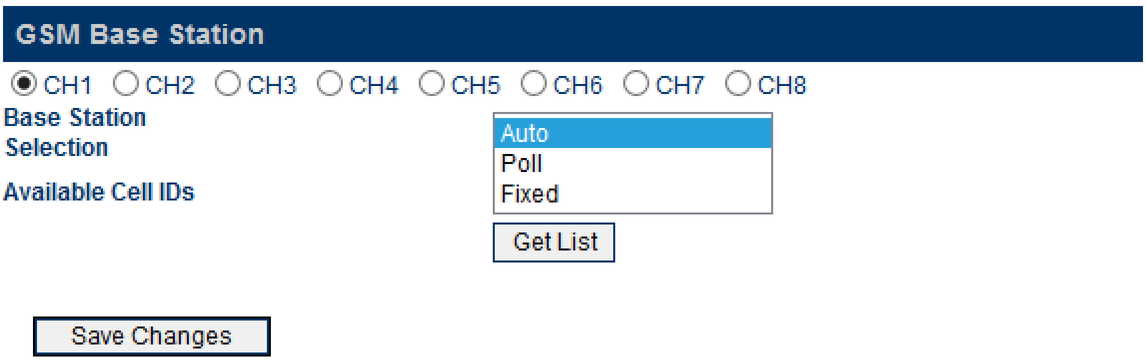


When a GoIP is installed at a location that is close to a country border, it is possible that the default service provider is not selected based on the base station signal strength. The GoIP may then register to a GSM service provider that charges for expensive roaming fees. In order to avoid this, the "Fixed" mode should be selected in order to lock the channel to a preferred service provider. When the "Fixed" mode is first selected, you must press [**Save Changes**] to save the setting and then refresh the browser after a few minutes in order to view a list of GSM service providers. Enter the provider code displayed in the Code entry and then press [**Save Changes**]. Please view the screen captures below.



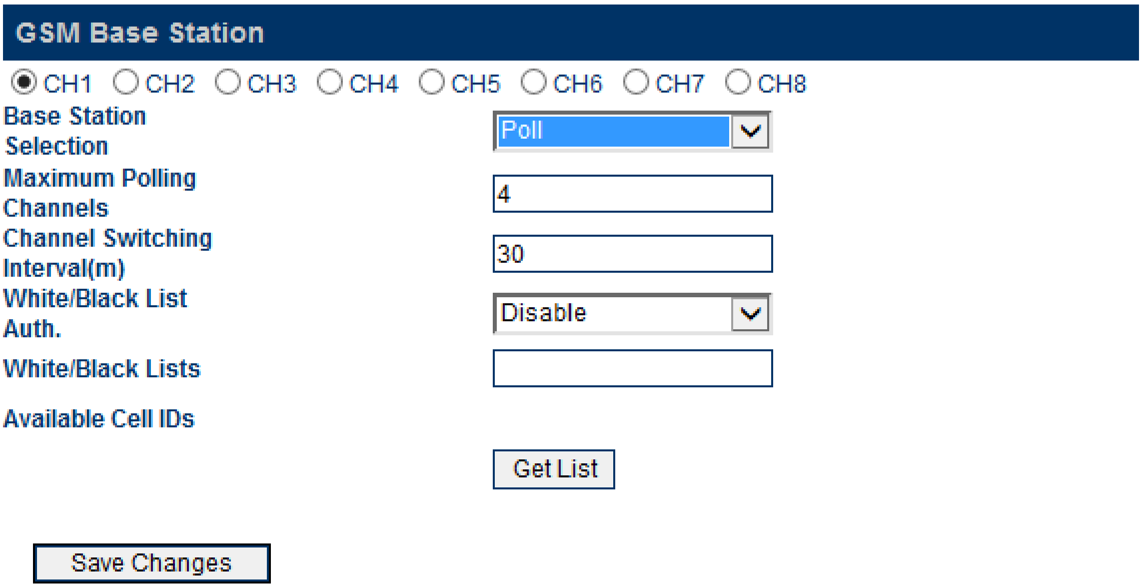
3.3.16 GSM Base Station

This feature is currently in beta testing stage with certain customers and it is intended for advanced users only. Don't attempt to change the default settings if you do not have a good understanding on the GSM network. Please contact us for help if you have a specific requirement on GSM base station settings.



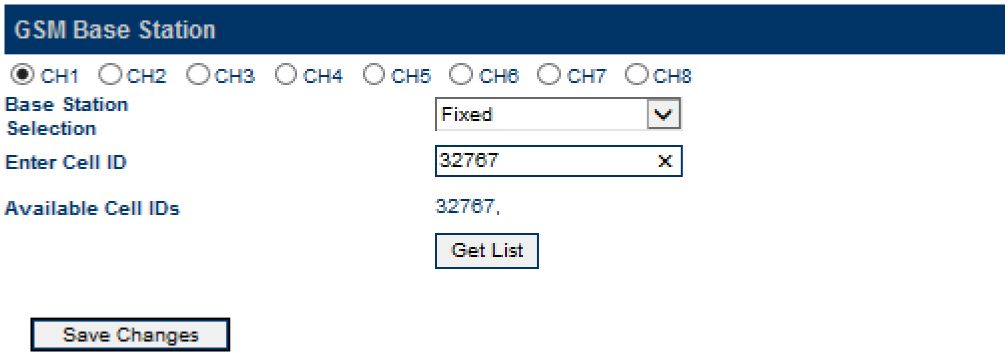
Three modes for base station selection are available:

1. Auto - This mode uses the default GSM base selection mechanism.
2. Poll - This mode limits the number and the list of base stations (BTS) that could be selected. Either Whitelist or Blacklist method can be used for BTS selection.



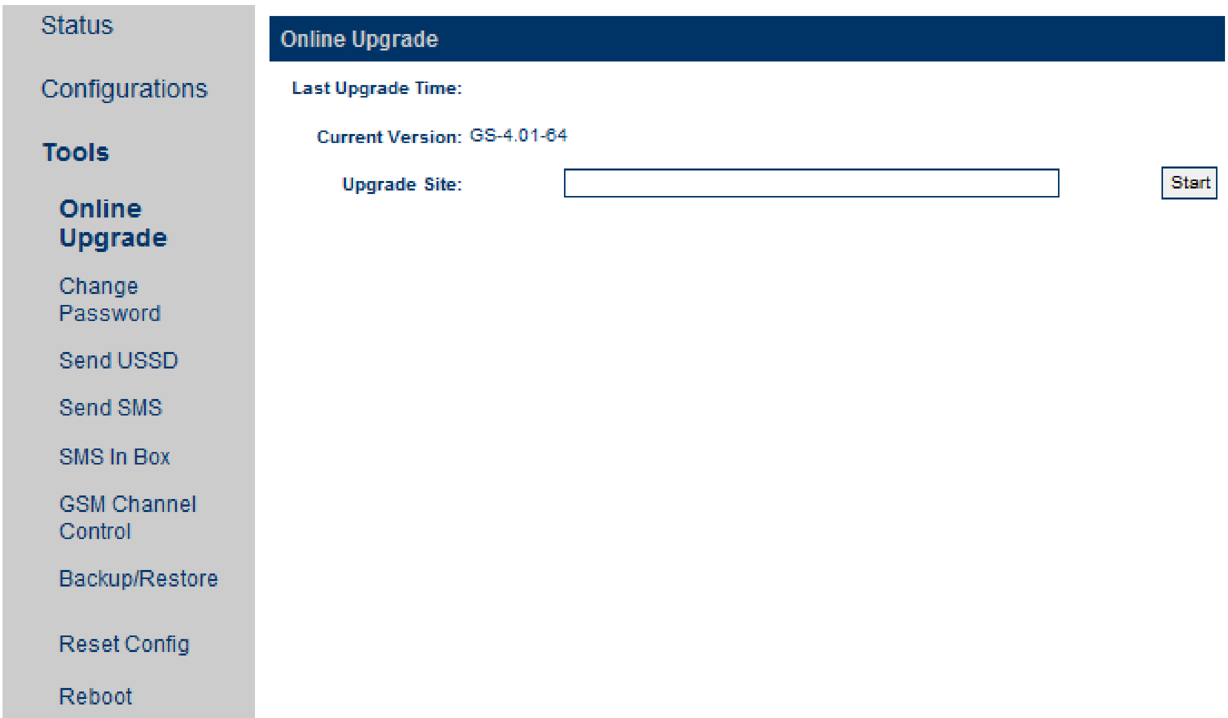
Parameter (GSM Base Station)	Description	Default Value
1. Maximum Polling Channels	This limits the maximum number of channels for polling.	4
2. Channel Switching Interval (m)	This defines the duration in minutes when the next base station switching should occur. The base station switching only occurs when the corresponding channel becomes idle.	30
3. White/Black List Auth	There are three ways to define how to switch to a different base station. If it is disabled, the base station selection cycles through the neighbor list. If the Whitelist authentication mode is selected, then the channel selection cycles through the base stations defined in the Whitelist. If the Blacklist authentication mode is selected, then the channels listed in the Blacklist will not be selected.	
4. Whitelist/Blackllist	Whitelist defines the base stations that are going to be used. Blacklist defines the base stations that are NOT going to be used.	

3. Fixed - This mode locks the base station to the Cell ID specified. Select "Fixed" to enable this mode and then press [**Save Changes**]. To view the available Cell IDs, please wait couple of minutes and then refresh the screen then select and enter the desired Cell ID as shown below. Remember to press [**Save Changes**] again to save the Cell ID entered.



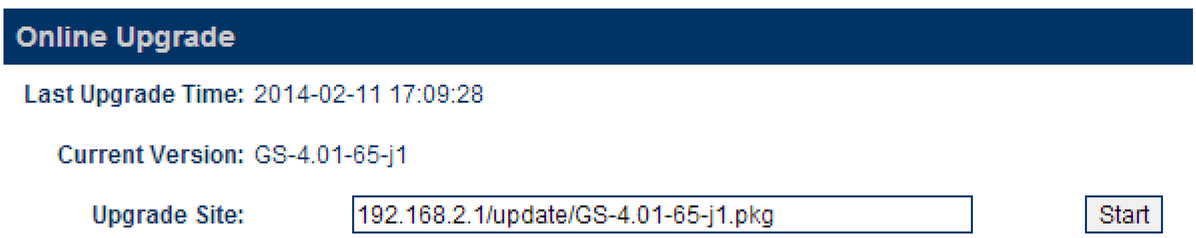
3.4 Tools

Click "Tools" on the left hand menu to access the submenu as shown below. Please note the available options under the **Tools** menu.



3.4.1 Online Upgrade

Click [Online Upgrade] to upgrade the device firmware. The current version is displayed as well as the last upgrade time.



Contact us or your local agent/supplier for the latest firmware version. Enter the firmware link (URL) and then click "Start" to begin firmware upgrade. Once the firmware upgrade is completed, the device will reboots itself automatically. Please wait patiently as this process may take a few minutes.

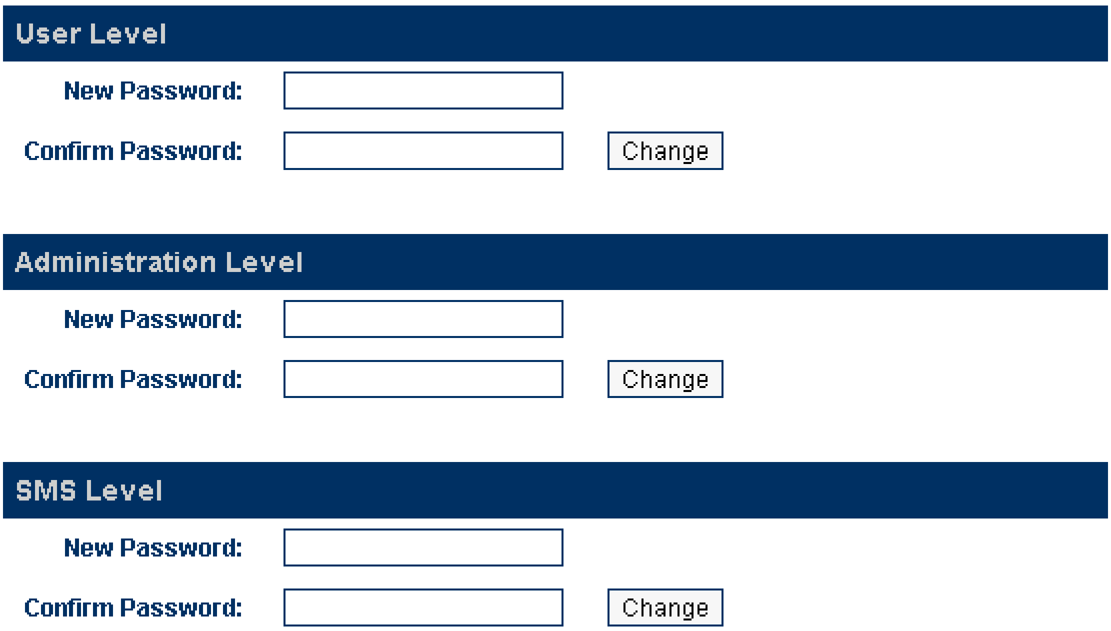
Note: It is important not to disconnect the power during a firmware upgrade since the internal Flash may be corrupted. If this happens, pleases contact technical support for assistance.c Please reboot the device if an upgrade attempt fails before performing another upgrade.

3.4.2 Change Password

Click "Change Password" to change the password with respect to the login level. There are three login levels:

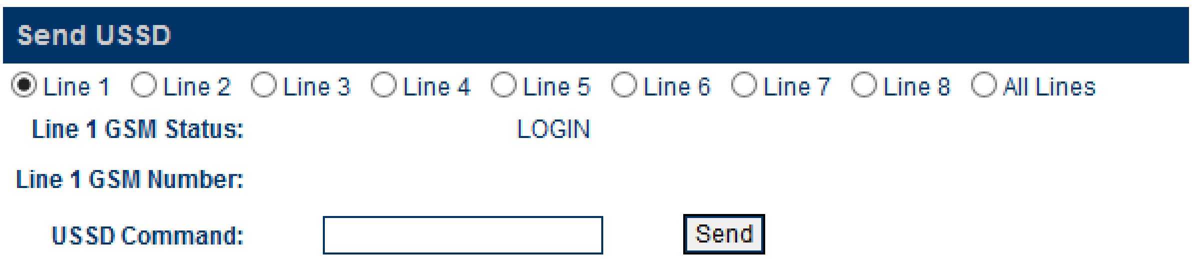
1. Administrative Level – Login ID is "admin" and the default password is "admin".
2. User Level – Login ID is "user" and the default password is "1234".
3. SMS Level – Login ID is "sms" and the default password is "1234"

Note: Administration Level allows changing the passwords for all 3 levels.



3.4.3 Send USSD

Click [Send USSD] to access the webpage (as shown below) to send USSD commands.



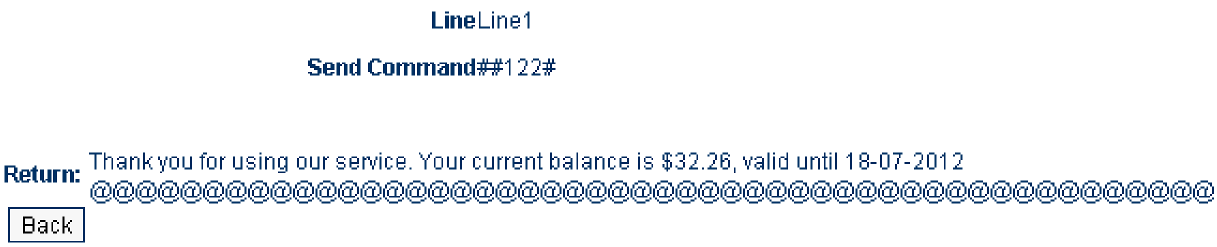
The procedures to send an USSD command are:

a) Select the Line (GSM channel) that you want to send an USSD command to the service provider. The line status and the SIM (GSM) number are displayed. The last option "All Lines" means that all channels are selected and the same USSD command is sent via all channels provided that they are in the Login Status.
b) Enter the USSD command
c) Click [Send].

Example:

For the service provider PCCW in Hong Kong, the USSD command to check balance is ##122#. Enter "##122#" and the click [Start]. The following screen is then displayed.

A few seconds later, the service provider sends back a USSD message/response as shown below.

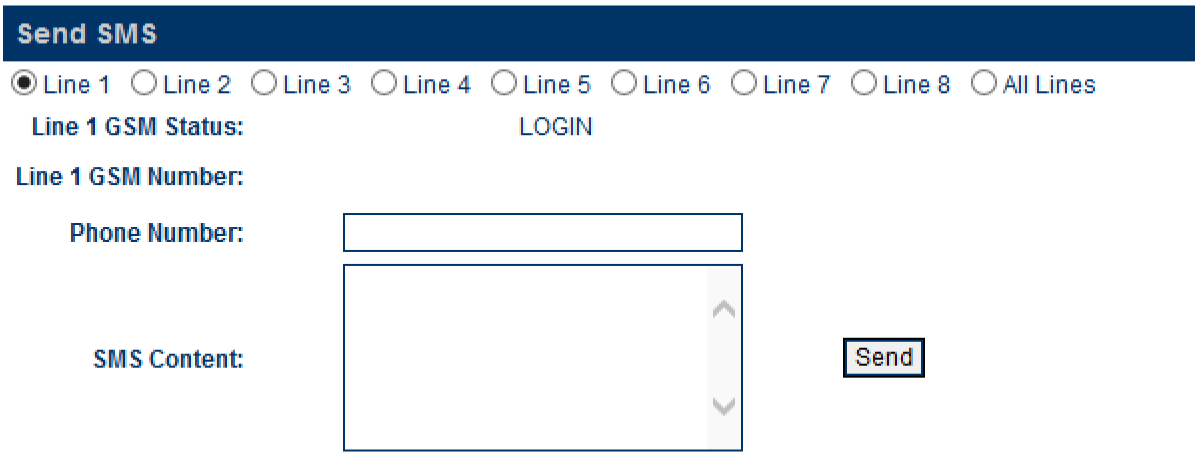


Click [Back] to return to the Send USSD command page.

For certain service requests, user responses are required. Just following USSD message and then send back a response via SEND USSD command.

3.4.4 Send SMS

Click [Send SMS] to access the Send SMS webpage as shown below



.

The procedures to send a SMS are:

a) Select the Line (GSM channel) that you want to send a SMS. The line status and the SIM (GSM) number are displayed. The last option "All Lines" means that all channels are selected and the same SMS is sent via all channels provided that they are in the Login Status.

b) Enter the recipient's phone number (GSM)

c) Type the SMS message in the SMS Content box. The maximum length of a message is 140 characters for 7/8-bit ASCII code and 70 characters for 16-bit Unicode) .

d) Click [Send] to send out the SMS.

3.4.5 SMS In Box

Click [SMS In Box] to view the SMS messages received as shown below. Select the desired line to view the latest 5 messages received for the corresponding GSM channel.

SMS BOX

◉ Line 1 ○ Line 2 ○ Line 3 ○ Line 4 ○ Line 5 ○ Line 6 ○ Line 7 ○ Line 8

Date	Number	Content
06-01 16:55:13	+8615817459136	but its taking time to save
06-01 16:54:43	+8615817459136	i ll sent message to you from my skype
06-01 16:54:33	+8615817459136	I see you on facebook
06-01 16:53:37	+8615817459136	but i cannt logon facebook
06-01 16:53:17	+8615817459136	Public furious over alleged rape of girls by official

3.4.6 GSM Channel Control

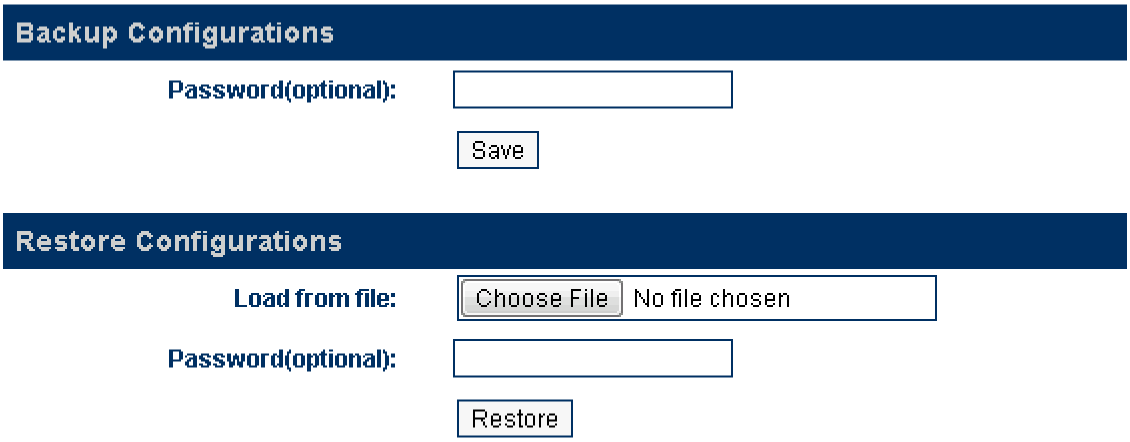
This feature is implemented for two functions:

1. Removing the power to a GSM channel before removing or inserting a SIM card. This is the recommended procedure in order to prevent damages to the SIM card.
2. Disabling a GSM channel temporary.

Click [GSM Channel Shut Down] to access the webpage below to shut each GSM module individually. Place a check mark (☑) to select the desired channel and then click [Save] to activate the shut down. Remove the check mark and then click [Save] to turn on the channel again. The "All Channels" selection is a short cut to turn on or to shut down all channels.

3.4.7 Backup / Restore

The device configuration can be backup or restore via this page. Click [Backup / Resotre Confguration] to access the page shown below.



To backup the device configuration, just click [Save] in the Backup Configuration section. If a password is required when restoring a saved configuration, enter a password before the backup.

To restore a saved configuration, choose the configuration file in the Restore Configuration section and then click [Restore]. Enter the password if required.

3.4.8 Reset

Click [Reset] to reset the device configuration back to the factory default. Click [OK] in the pop up window shown below to confirm this action.

Click [OK] to reset the device configuration back to the factory default!

3.4.9 Reboot

Click [Reboot] to restart the device. Click [OK] in the pop up window shown below to confirm this action. The reboot process will take couple of mins.

Appendix A. Special SMS Commands

In order to manage the device, special SMS commands can be sent to anyone of the GSM channel in order to read the LAN IP, reset the device and reboot the device. The table below summarizes the SMS command syntax. “<” and “>” are not part of command text.

SMS Message Content	Function
###INFO###	Sends an SMS response to the sender with the LAN port IP address.
###info###	
RESET *<password>*.	Reset the device configuration back to the factory defaults and then reboot the device. *<password>* is the password for the administration level.
reset *<password>*	
REBOOT *<password>*	Reboot the device. *<password>* is the password for the administration level.
reboot *<password>*	

Appendix B. SMS To VoIP

The device receives SMS messages from both GSM and VoIP networks and they are handled according to the modes defined below.

1. Call Function – In this mode, a received SMS is used to implement the "Call Back" function. The concept is to establish a phone call between the called party and the calling party. The phone number of the called party is specified in the GSM SMS message received. The phone number of the calling party is the SMS sender's number. The device then sends a SIP INVITE message containing these phone numbers to the SIP Server registered. Three different SIP INVITE message formats are supported and are described below.

   a) Mode 1
   SIP Message format:
   - The "To" field in the SIP INVITE message contains the phone number of the called party.
   - The "From" field contains the phone number of the calling party.

   Once the SIP server receives these two numbers via a SIP INVITE message, it then terminates the SIP call (SIP INVITE) and then call both parties via its own phone network. The device may or may not take part in the actual call conversation.

   Example:
   SMS content = 8675588228822
   Sender's number = 861380000000
   SIP Server IP = 192.168.2.1
   SIP Number = 20001
   The INVITE message sent to the SIP Server is:

```
INVITE sip:8675588228822@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK363969813
From: <sip:8613800000000@192.168.2.1:5060>;user=phone;tag=65248630
To: <sip:8675588228822@192.168.2.1>
Call-ID: 117025903@192.168.2.237
CSeq: 2 INVITE
Contact: <sip: 8613800000000@192.168.2.237:5060>
Max-Forwards: 30
User-Agent: HYBERTONE
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER, MESSAGE, INFO, SUBSCRIBE
Content-Type: application/sdp
Content-Length:   226
```

   b) Mode 2
   SIP Message format:
   - The "To" field in the SIP INVITE message contains the phone number of the called party.
   - The "From" field contains the SIP number of the line that is associated with the GSM channel
   - Note: The VoIP configuration of the device must be set to "**Config. By Line mode**".

   Only the phone number of the called party is passed to the SIP server via a SIP INVITE message. This mode is designed to use the GSM channel of the device to complete the Call Back function. Therefore, the ***Call OUT via GSM*** parameter of the device must be enabled and the ***Forward Number*** associated with this parameter is set to the phone number of the SMS sender.

To achieve the Call Back function, the SIP server calls the called party via its phone network and then calls the SIP number. Since a call to this SIP number is set to forward to the phone number of the SMS sender, both the called and calling parties can then be connected.

Example:
SMS content = 8675588228822
Sender's number = 861380000000
SIP Server IP = 192.168.2.1
SIP Number = 20001
The INVITE message sent to the SIP Server is:

```
INVITE sip:8675588228822@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK363969813
From: <sip:20001@192.168.2.1:5060>;user=phone;tag=65248630
To: <sip:8675588228822@192.168.2.1>
Call-ID: 117025903@192.168.2.237
CSeq: 2 INVITE
Contact: <sip:20001@192.168.2.237:5060>
Max-Forwards: 30
User-Agent: HYBERTONE
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER, MESSAGE, INFO, SUBSCRIBE
Content-Type: application/sdp
Content-Length:   226
```

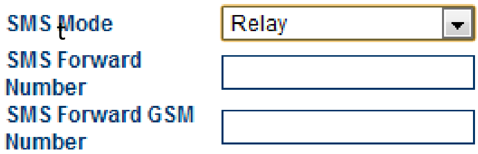
c) Mode 3
SIP Message format:
- The "To" field in the SIP INVITE message contains the phone numbers of both the called and calling parties. These two numbers are concatenated by using the asterisk (*) character with the number of the called party in the front.
- The "From" field contains the SIP number of the line that is associated with the GSM channel

Example:
SMS content = 8675588228822
Sender's number = 861380000000
SIP Server IP = 192.168.2.1
SIP Number = 20001
The INVITE message sent to the SIP Server is:

```
Sending Message to 192.168.2.1:5060:
INVITE sip:8675588228822*8613800000000@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK363969813
From: <sip:20001@192.168.2.1:5060>;user=phone;tag=65248630
To: <sip:8675588228822*8613902994477@192.168.2.1>
Call-ID: 117025903@192.168.2.237
CSeq: 2 INVITE
Contact: <sip:20001@192.168.2.237:5060>
Max-Forwards: 30
User-Agent:HYBERTONE
```

```
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER, MESSAGE, INFO, SUBSCRIBE
Content-Type: application/sdp
Content-Length:   226
```

2. Forward Function – This mode supports SMS forwarding from GSM to SIP and from SIP to GSM.
   a) Received GSM SMS messages are forwarded to both SIP and GSM depending on the settings of the SMS Forward Number and SMS Forward GSM Number.



When an incoming GSM SMS is received, it can be forwarded automatically to another GSM number as specified by the SMS Forward GSM Number. The received SMS can also be forwarded automatically to a a SIP number or extension as specified by the SMS Forward Number. If this number is not set, this feature is disabled. Forwarding GSM SMS to SIP is achieved via the SIP MESSAGE command. An example of a SIP MESSAGE is shown below. Please note that the number of the GSM SMS Sender is added as part of the message (the last two lines in the SIP MESSAGE command).

Example:

SMS SIP Recipient = 3999
SIP Proxy = 192.168.2.1
GSM SMS Sender = 861361234567
GSM SMS Content = 075583185700

SIP MESSAGE Sent to SIP Server:

```
MESSAGE sip:3999@192.168.2.1 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.162:5060;branch=z9hG4bK1967685528
From: <sip:20001@192.168.2.1>;tag=667435795
To: <sip:3999@192.168.2.1>
Call-ID: 2094144847@192.168.2.162
CSeq: 4 MESSAGE
Contact: <sip:20001@192.168.2.162:5060>
Max-Forwards: 30
User-Agent: HYBERTONE
Content-Type: text/plain
Content-Length:   28
8613682626865
075583185700
```

Please note that the SIP Server side must be programmed to process this SIP MESSAGE according to the application needed. It can forward the message to the SIP number with the Caller ID as the GSM SMS Sender. If the message content is a phone number for a called party, it is then possible to implement the Call Back function by using the content of the SIP message.

If the SMS GSM Recipient is set, the received GSM SMS is forwarded to this recipient via the same GSM channel which receives the SMS.

   b) A SMS can be sent to the device via its SIP number. The content of the SMS must be in the preset format. The first line must contain a valid GSM number and then the text message begins at the second line and must meet the restrictions imposed by a normal GSM SMS. A sample of a SIP MESSAGE sent from SIP device is shown below.

Example: A SIP SMS is sent from the SIP number 3999 to the SIP number 2001 (used by the device) and then the SMS is sent out to the phone number 1368266800 via the GSM channel associated with 2001.

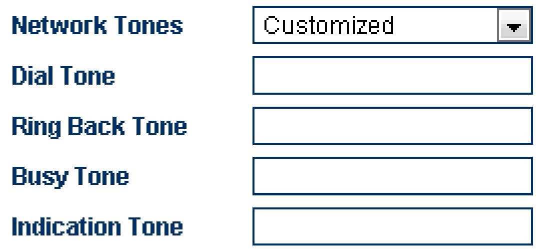
SIP SMS Sender = 3999
SIP SMS Recipient = 2001
SIP SMS Content = **13682626800**
Hello world

SIP MESSAGE Sent from the SIP Server:

```
MESSAGE sip:20001@192.168.2.162:5060 SIP/2.0
From: <sip:3999@192.168.2.89>;tag=5031
To: <sip:20001@192.168.2.1>
Call-ID: 808807EB-A8B3-DD11-BBA6-005056C00008@192.168.2.89
CSeq: 3 MESSAGE
Contact: <sip:3999@192.168.2.89>
max-forwards: 16
date: Tue, 18 Nov 2008 06:36:37 GMT
user-agent: SIPPER for 3CX Phone
p-hint: usrloc applied
Content-Type: text/plain
Content-Length:    26
13682626800
Hello world
```

Appendix C. Custom Network Tones

This section describes how to define custom network tones. The "Custom" selection allows the following tones to be defined as shown on the right.



1. Dial Tone – When an incoming call is answered, this tone is generated to indicate to the caller to dial a number.

2. Ring Back Tone – When a call is dialed from the device to VoIP and the SIP 183 is not enabled, this tone is generated to indicate that the calling is in progress.

3. Busy Tone - When a call dialed from the device to VoIP is busy, this tone is generated.

4. Indication Tone – When a call waiting call is presence, this tone is generated.

The syntax for a network tone script is defined as

<nf, rpt, p1on, p1off, p2on, p2off, p3on, p3off, f1, f2, f3, f4, l1, l2, l3, l4>

where

nf is the number of single frequency tone (1-4) to be generated.
rpt is the number of times for the tone to be repeated based on the on/off pattern defined (0 means infinite).
p1on is the tone on duration for the first frequency tone (ms)
p1off is the tone off duration for the first frequency tone (ms)
p2on is the tone on duration for the second frequency tone (ms)
p2off is the tone off duration for the second frequency tone (ms)
p3on is the tone on duration for the third frequency tone (ms)
p3off is the tone off duration for the third frequency tone (ms)
f1 is the frequency of the first tone (300 to 3000Hz)
f2 is the frequency of the second tone (300 to 3000Hz)
f3 is the frequency of the third tone (300 to 3000Hz)
f4 is the frequency of the forth tone (300 to 3000Hz
l1 is the level for tone 1 (range from 0 to 31 with 0 = 3dB, -1dB for each increment)
l2 is the level for tone 2 (range from 0 to 31 with 0 = 3dB, -1dB for each increment)
l3 is the level for tone 3 (range from 0 to 31 with 0 = 3dB, -1dB for each increment)
l4 is the level for tone 4 (range from 0 to 31 with 0 = 3dB, -1dB for each increment)

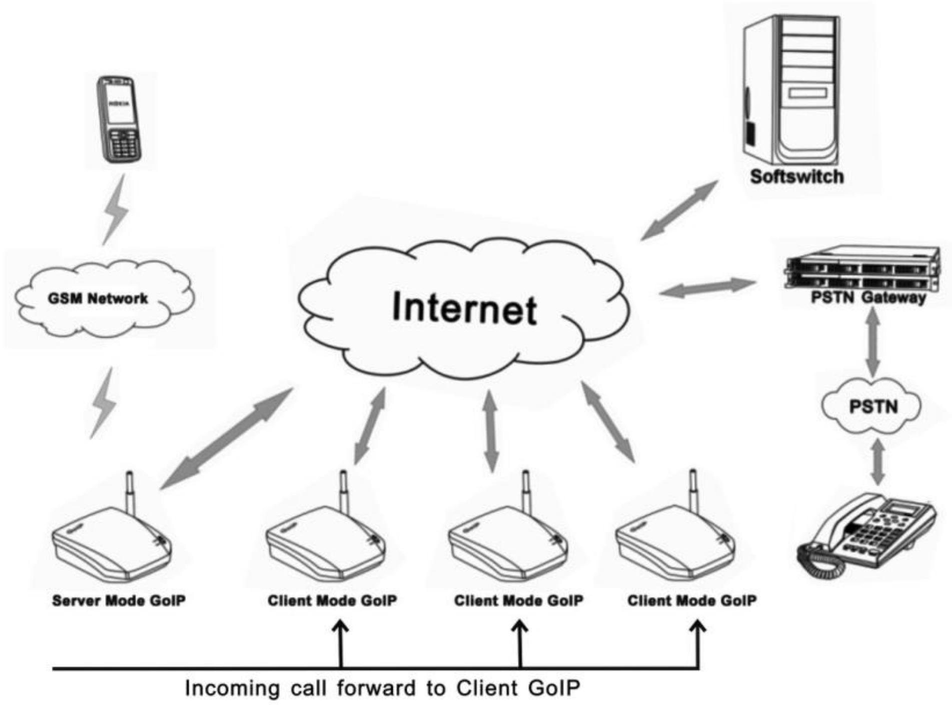
Example:

1. Dial tone definition: 450Hz@-20dB, on continuously.

The dial tone script is: 1,0,100,0,0,0,0,0,450,0,0,0,23,0,0,0

Appendix D. GSM Group Mode

The GSM Group mode is designed to simulate the function of one GSM number with multiple lines. The idea is to form a GSM group with one number being the "Server". Only this GSM number is announced to the public. Calls to this number are forwarded to other GSM numbers ("Clients") in the group until all GSM channels are used up. Effectively speaking, if there are 40 GSM channels in a group, a maximum of 40 concurrent calls can be achieved by just calling the GSM number of the Server channel. The diagram below demonstrates this concept with only single channel GoIPs. In fact, GoIP with multiple channels can also be used. Only one "Server" in a group and all the other channels must be set to "Client" individually.



Appendix E. CID Call Forward

For incoming GSM calls, the phone number of the caller can be displayed at the called party (SIP terminal). The device supports the following two methods. Unfortunately, not all SIP servers support one or both methods. Please check with the vendor of the SIP server for more information.

1. Remote Party ID - This is a parameter in a SIP INVITE message. Choose this if both SIP Server and SIP terminal support this parameter.

   Example: Caller ID / number = 13800000000
   The Remote Party ID parameter is included in the SIP INVITE Message below.

```
Sending Message to 192.168.2.1:5060:
INVITE sip:5000@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.180:5060;branch=z9hG4bK1645487913
From: <sip:20001@192.168.2.1:5060>;user=phone;tag=406202416
To: <sip:5000@192.168.2.1>
Call-ID: 847230278@192.168.2.180
CSeq: 2 INVITE
Contact: <sip:2000@192.168.2.180:5060>
Max-Forwards: 30
User-Agent: HBT
Remote-Party-ID: "13800000000"
<sip:13800000000@192.168.2.1>;party=calling;screen=no;privacy=off
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER,
MESSAGE, INFO, SUBSCRIBE
Content-Type: application/sdp
Content-Length:    226
```

2. USD CID as SIP Caller number - This parameter specifies the use of GSM Caller ID instead of its SIP number in the INVITE message when making a call. Please make sure that the SIP server supports this type of INVITE message since the call now is not originated from a valid SIP number defined in the server. Please note that the Remote-Party-ID is also included in the INVITE message.

```
Sending Message to 192.168.2.1:5060:
INVITE sip:5000@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.180:5060;branch=z9hG4bK1450498491
From: "13800000000" <sip:13800000000@192.168.2.1:5060>;tag=232569343
To: <sip:5000@192.168.2.1>
Call-ID: 1853068986@192.168.2.180
CSeq: 2 INVITE
Contact: <sip:13800000000@192.168.2.180:5060>
Max-Forwards: 30
User-Agent: HBT
Remote-Party-ID: "13800000000" <sip:
13800000000@192.168.2.1>;party=calling;screen=no;privacy=off
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER,
MESSAGE, INFO, SUBSCRIBE
Content-Type: application/sdp
Content-Length:    226
```

Appendix F. Volume Adjustment

The volume adjustment of the device can be accessed via the URL below.

http://<device address>/en_US/gaim.html

The <device address> is the IP address or domain name of the device. The volume levels of the audio streams from VoIP to GSM and GSM to VoIP are controlled by the input gain and the output gain respectively. An increase in the output gain means that the GSM / PSTN party hears a higher audio level.

An increase in the input gain means that the VoIP party hears a higher audio level.

Please note that changing these gain settings affects the DTMF tones in the corresponding path as well. As a result, DTMF tones for phone dialing may not be detected correctly. Please change these settings with great care and make sure that DTMF detections are not affected.