User Manual

Single Channel GSM Gateway

Model: GoIP

Release 1.2

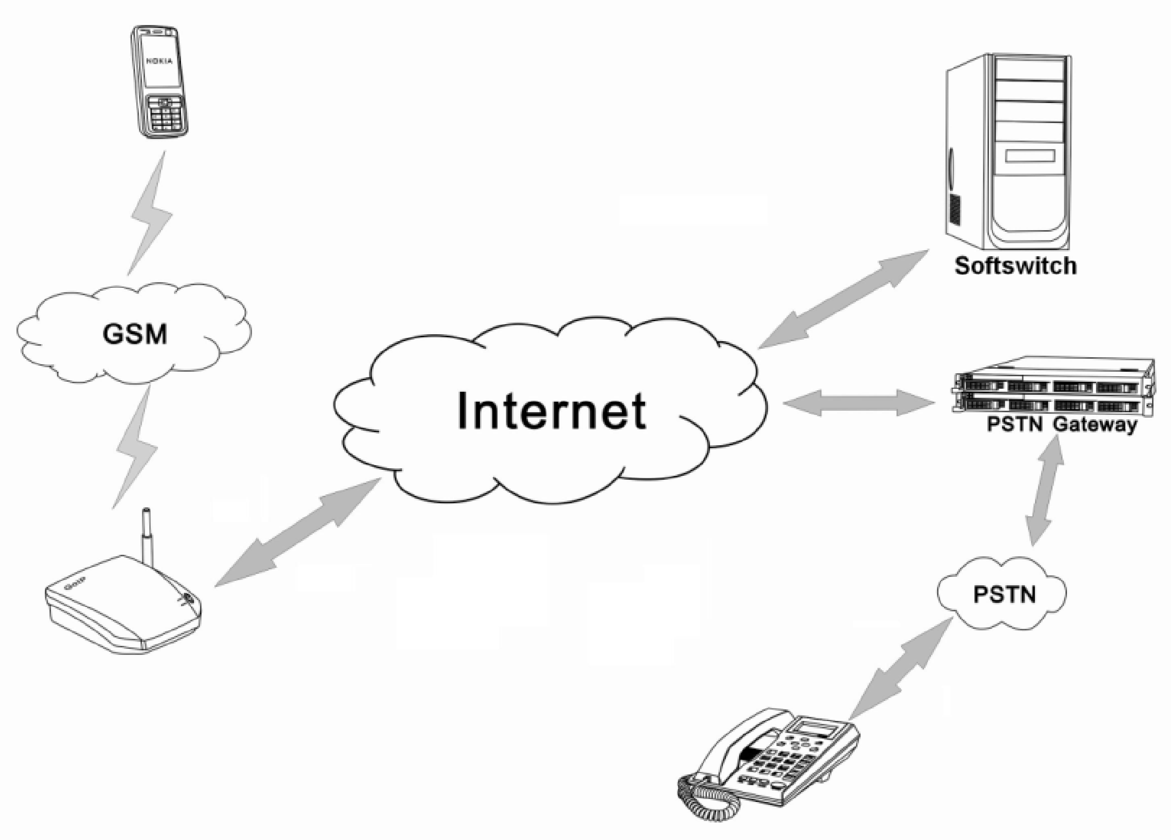
Contents

1 Introduction

1.1 Overview

A VoIP GSM gateway (***GoIP***) is an IP-based device that enables inbound and outbound VoIP and GSM cellular calls. It is an alternative to a VoIP FXO gateway, especially in area where GSM service is readily available and cheaper for VoIP call termination. Many applications can be evolved from this technology using GSM termination, for examples, distributed call centers, VoIP termination, and cell phone roaming. A VoIP GSM gateway is not only a great way to provide fast deployment but also provides significant savings in usage, infrastructure and maintenance cost compared to conventional FXO gateways.

The diagram below shows a typical topology for a VoIP GSM gateway.



1.2 Protocol

- TCP/IP V4 (IP V6 auto adapt)
- ITU-T H.323 V4 Standard
- H.2250 V4 Standard
- H.245 V7 Standard
- H.235 Standard（MD5，HMAC-SHA1）
- ITU-T G.711 Alaw/ULaw, G.729A, G.729AB, and G.723.1 Voice Codec
- RFC1889 Real Time Data Transmission
- Proprietary Firewall-Pass-Through Technology
- SIP V2.0 Standard
- Simple Traversal of UDP over NAT (STUN)
- Web-base Management
- PPP over Ethernet (PPPoE)
- PPP Authentication Protocol (PAP)
- Internet Control Message Protocol (ICMP)
- TFTP Client
- Hyper Text Transfer Protocol (HTTP)
- Dynamic Host Configuration Protocol (DHCP)
- Domain Name System (DNS)
- User account authentication using MD5
- Out-band DTMF Relay: RFC 2833 and SIP Info

1.3 Hardware Specification

- Embedded Processor
- DSP for voice codec and voice processing
- Two 10/100 BaseT Ethernet ports with full compliant with IEEE 802.3
- LEDs for Ethernet port status
- One GSM Connection:
  - Huawei GSM Module - GSM 850 MHz/GSM 900 MHz
  - Simcom GSM Module – GSM 850 MHZ, 900MHz, DCS 1800 MHz, PCS 1900 MHz

1.4 Software Specification

- Embedded LINUX OS
- Built-in HTTP Web Server
- PPPoE Dial-up
- NAT Broadband Router Functions
- DHCP Client
- DHCP Server
- Firmware On-line upgrade
- Caller ID
- Multiple Language Support
- Support Multi devices Cooperate Mode

1.5 List of the Package

a) One GoIP gateway main unit
b) One DC4.5V/2000mA power adaptor
c) One Ethernet cable (3M)

1.6 Appearance

VoIP GSM Gateway (GoIP) – Front View

VoIP GSM Gateway (GoIP) – Rear View

1） LAN

Connect this port to an Ethernet switch/router, the Ethernet of a DSL modem, or other network access equipment.

2） PC

Connect a computer or other network device to this port.

3） POWER (DC4.5V/2000mA)

Connect the 4.5V/2000mA adapter provided to this power jack.

4） Reset

Press this button to reset the GoIP gateway to factory defaults.

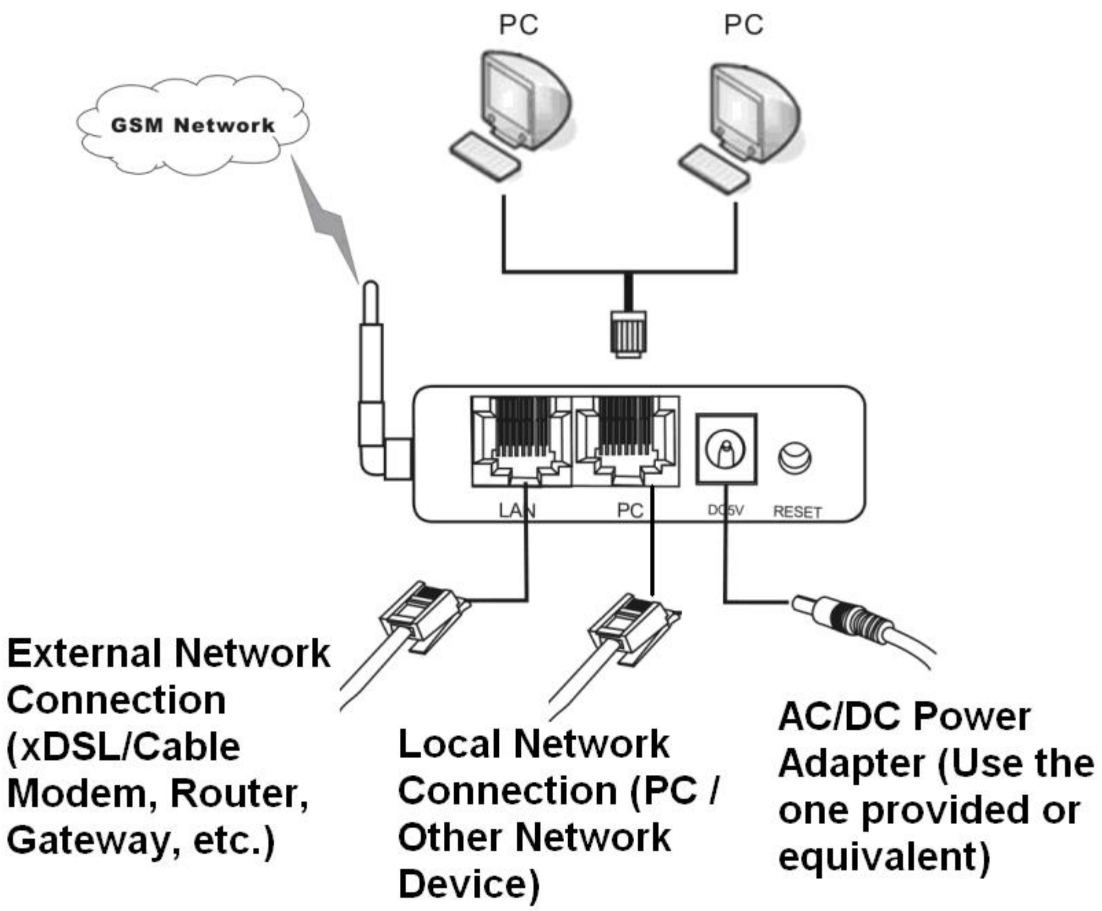
2 Installation

2.1 Connection Diagram

Please follow the connection diagram above to install the GoIP gateway.

a) Insert a GSM SIM card in the SIM card compartment located at the bottom of the GoIP gateway.
b) Connect an Ethernet cable the LAN port of the GoIP gateway and the other end to your existing network equipment.
c) Connect an Ethernet cable to the PC port of the GoIP gateway and the other end to a PC or other network device (optional).
d) Connect the power adapter provided to the power jack of the GoIP gateway.

The diagram below shows a typical installation of the device.



2.3 LED Indicators

The following table defines the status of the LEDS located on the top case and on the RJ-45 connectors.

LED	DESCRIPTION
RUN	1. When the GoIP is booting，this LED will flash 100ms ON and 100ms OFF. 2. When the GoIP is properly registered to your softswitch, this LED flashes at a rate of 1s ON and 1s OFF.
GSM	When the GSM channel is ready to sue, this LED flashes at a rate of 1s ON and 1s OFF.

2.4 SMS Commands

GoIP supports the following management commands via GSM SMS messages.

Function	Command (SMS Content)	Remarks
Obtain LAN Port IP Address	INFO or info	Case Non-sensitive
Reset GoIP Configuration	RESET<Password>	Key word “RESET” not case sensitive
Reboot GoIP	REBOOT <Password>	Key word “REBOOT” not case sensitive

1》 Obtain LAN Port IP Address

Once the GSM SMS with message content “info” or “INFO” is received, the GoIP sends back a SMS message to the sender with the message content containing the LAN Port IP address.

2》 Reset GoIP Configuration

Upon receiving the SMS message “RESET <password>” or “reset <password>”, the GoIP reset its configurations to factory defaults.

3》 Reboot GoIP

Upon receiving the SMS message “REBOOT <password>” or “reboot <password>”, the GoIP reboots itself automatically.

Note: <password> is the webpage login password as described in Section 3.1.

3 Built-in Web Server

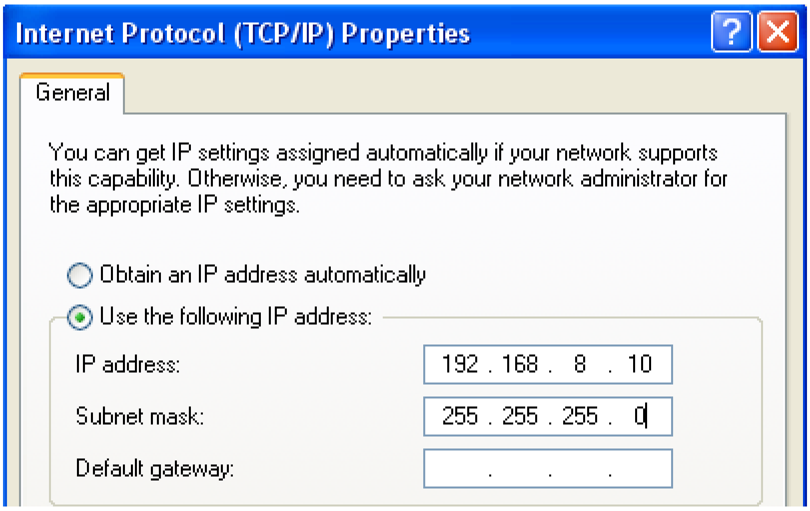
To configure the GoIP gateway, you must login to its Web server via the LAN or PC port. The LAN port is factory preset to obtain an IP address from the local DHCP host and the PC port is set to the fixed IP address ***192.168.8.1***.

If a local DHCP host is available, the LAN will obtain an IP address automatically. To listen to the IP address assigned, just dial a call via the GoIP gateway's SIM card phone number. When the call is connected, you will hear a dial tone. Just dial "***01#**" for English voice prompt for the LAN IP address and "***00#**" for Chinese voice prompt for the LAN IP address. The LAN IP address can also be obtained by sending a SMS message to the GSM phone number. The GoIP will then reply with a SMS message containing the LAN IP address.

If you want obtained LAN port IP by sending a SMS message, please send" ***INFO*** *"or"* ***info***".

If a local DHCP host is not available, you can access the GoIP gateway via the PC port. You will need to change the PC LAN configuration via the Network Connections under the Control Panel.

Windows: **Control Panel--→Network Connections--→Local Connectionism's Property--→TCP/IP Protocol's Property**



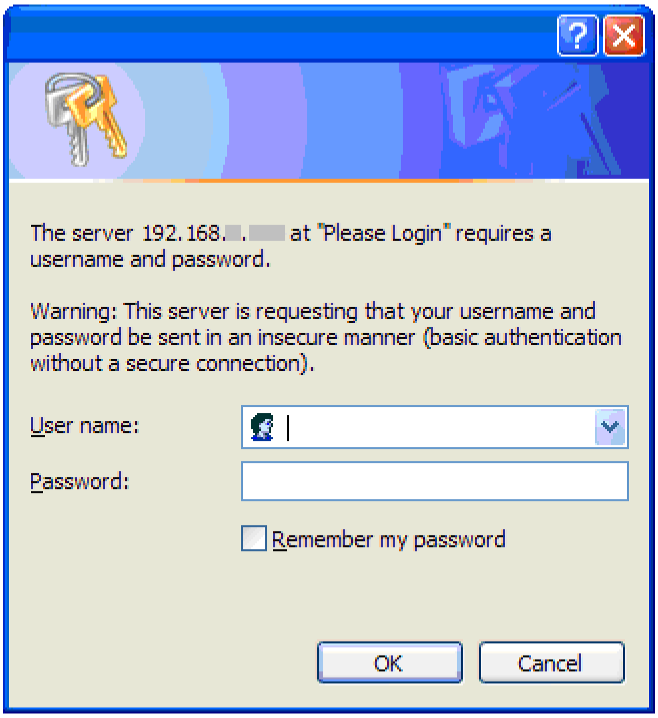
Set an unused IP address that is in the same segment as the PC port address.

Once the IP address of the LAN or PC port is known, you are now ready to access the Web server of GoIP gateway.

3.1 Web Configuration Menu

If your PC is connected to the GoIP gateway via the LAN port network segment, you need to type the LAN IP address of the GoIP gateway in your Web browser to access the Web server of the GoIP gateway. If not, you should type the PC IP address (192.168.8.1) in the Web browser.

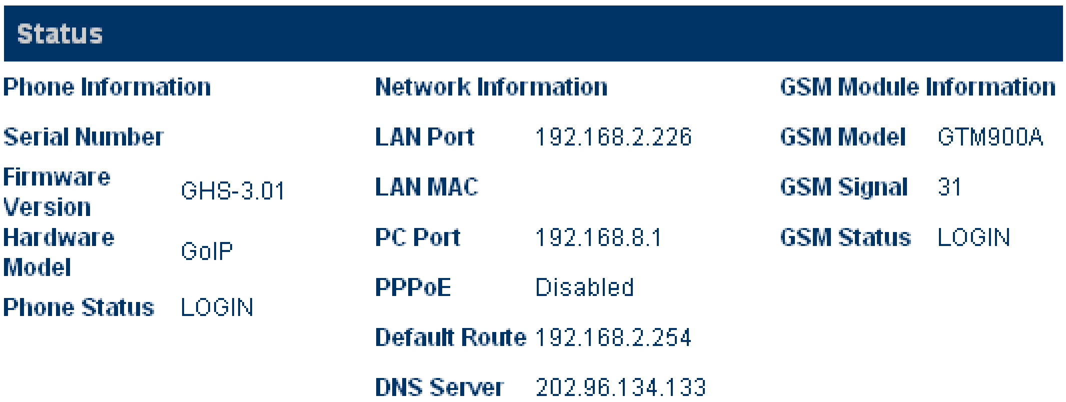
If the connection is correct, the Web browser will prompt you to enter the "User name" and "Password: as shown below.



Enter the User name and Password and the press OK to access the GoIP Gateway Web Server. The default for both user name and password is "**admin**". The default built-in webpage is then shown below.

3.2 Status

The Status page shown below is the default / home page of the GoIP Web server.



Status

Phone Information		Network Information		GSM Module Information	
Serial Number		LAN Port	192.168.2.226	GSM Model	GTM900A
Firmware Version	GHS-3.01	LAN MAC		GSM Signal	31
Hardware Model	GoIP	PC Port	192.168.8.1	GSM Status	LOGIN
Phone Status	LOGIN	PPPoE	Disabled		
		Default Route	192.168.2.254		
		DNS Server	202.96.134.133		

It consists of 3 columns status information of the GoIP and they are:

a. Phone Information
b. Network Information
c. GSM Module Information

3.2.1 Phone Information

A. Serial Number

Each gateway has a unique serial number assigned by the factory, for example, **HT304O12VTEST01**. This number is important for centralized configuration, technical support, and warranty. This number is printed on the bottom of the gateway and is associated with your software license.

B. Firmware Version

Firmware version identifies the firmware version of the gateway, for example, **GHS-3.01**.

C. Hardware Mode

This field shows terminal's hardware type.

D. Phone Status

This field shows the status of line's connection status. If the connection is successful, this field displays LOGIN; otherwise, it displays LOGOUT.

3.2.2 Network Information

A. LAN Port Configuration

This field displays the status of the LAN port.

B. PC Port Configuration

This field displays the status of the LAN port.

C. PPPoE

If PPPoE is enabled, it displays its status.

D. Default Route

This field displays the IP address of the default routing gateway.

E. DNS Server

This field displays the IP address of the domain name server.

3.2.3 GSM Module Information

A. GSM Module

This field displays the GSM module type.

B. GSM Signal

This field displays the GSM signal type.

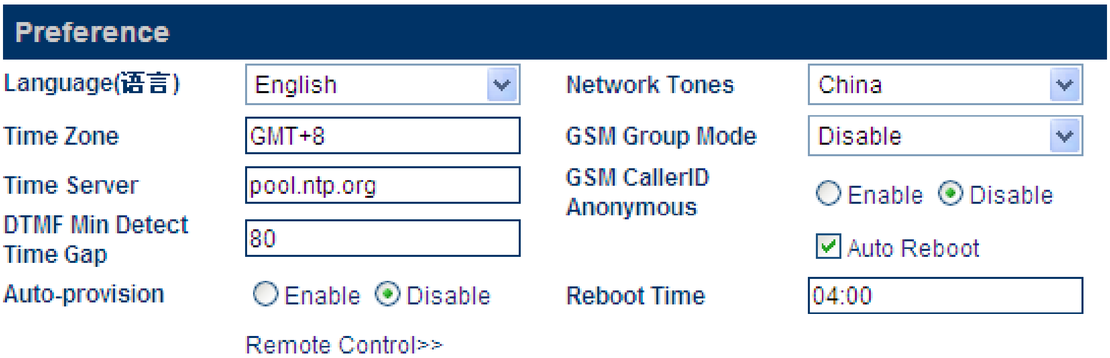
C. GSM Status

This field shows the status of GSM connection status. If the connection is successful, this field displays LOGIN; otherwise, it displays LOGOUT.

3.3 Configurations

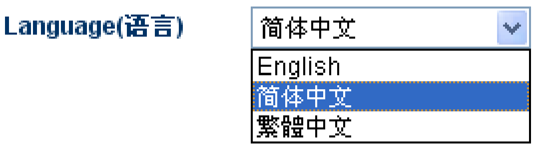
Click on the "Configurations" tab on the left hand column to access the device configuration menu: **Preference**, **Network**, **Call Settings**, **Call Divert**, **Save Changes**, and **Discard Changes**.

Click on "**Preference**" in the left menu of the configuration web, and the screen will be displayed as below:



3.3.1 Language

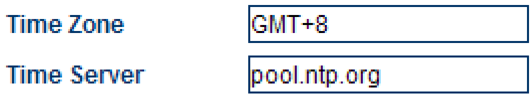
Currently GoIP supports English, simplified Chinese and traditional Chinese.   Select the language desired and the Web page will be shown in the language selected accordingly.



The language can also be selected at the top of the web page.   Once selected, the webpage language is refreshed immediately. However, the language selection is not saved until the **Save Changes** icon is clicked.

3.3.2 Time Zone and Time Server

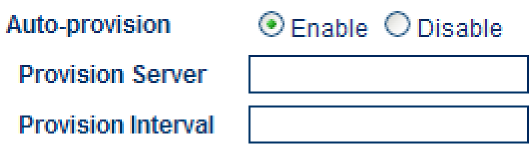
The GoIP gateway supports Network Time Protocol (NTP) to obtain the date and time information from an NTP server (Time Server). The time zone is specified as in GMT ± offset. For example, the Pacific Standard Time is GMT-8, and the Pacific Daylight Time is GMT-7.



Note: The GoIP gateway supports CDR and billing information, it is important to set up these two parameters properly.

3.3.3 Auto-Provision

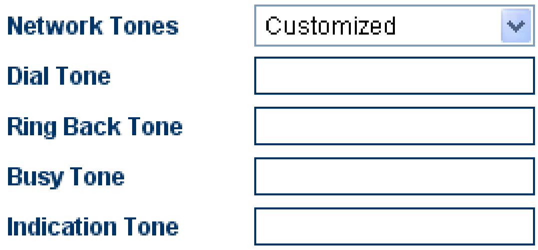
The GoIP Gateway supports Auto Provisioning which enables configuration parameters to be set automatically without human intervention. The Auto Provisioning supports both HTTP and TFTP protocols. For higher security, encrypted configuration file is also supported. This feature requires external Auto Provisioning Server. Please contact your service provider for further information on this.



3.3.4 Network Tone

Network tones are a set of tones used for VoIP calls. Select one of the countries defined or customized to define your own Network Tones.

You can configure the Network Tones as Customized option:



Each tone listed above is defined in the following format:

nc, rpt, c1on, c1off, c2on, c2off, c3on, c3off, f1, f2, f3, f4, p1, p2, p3, p4

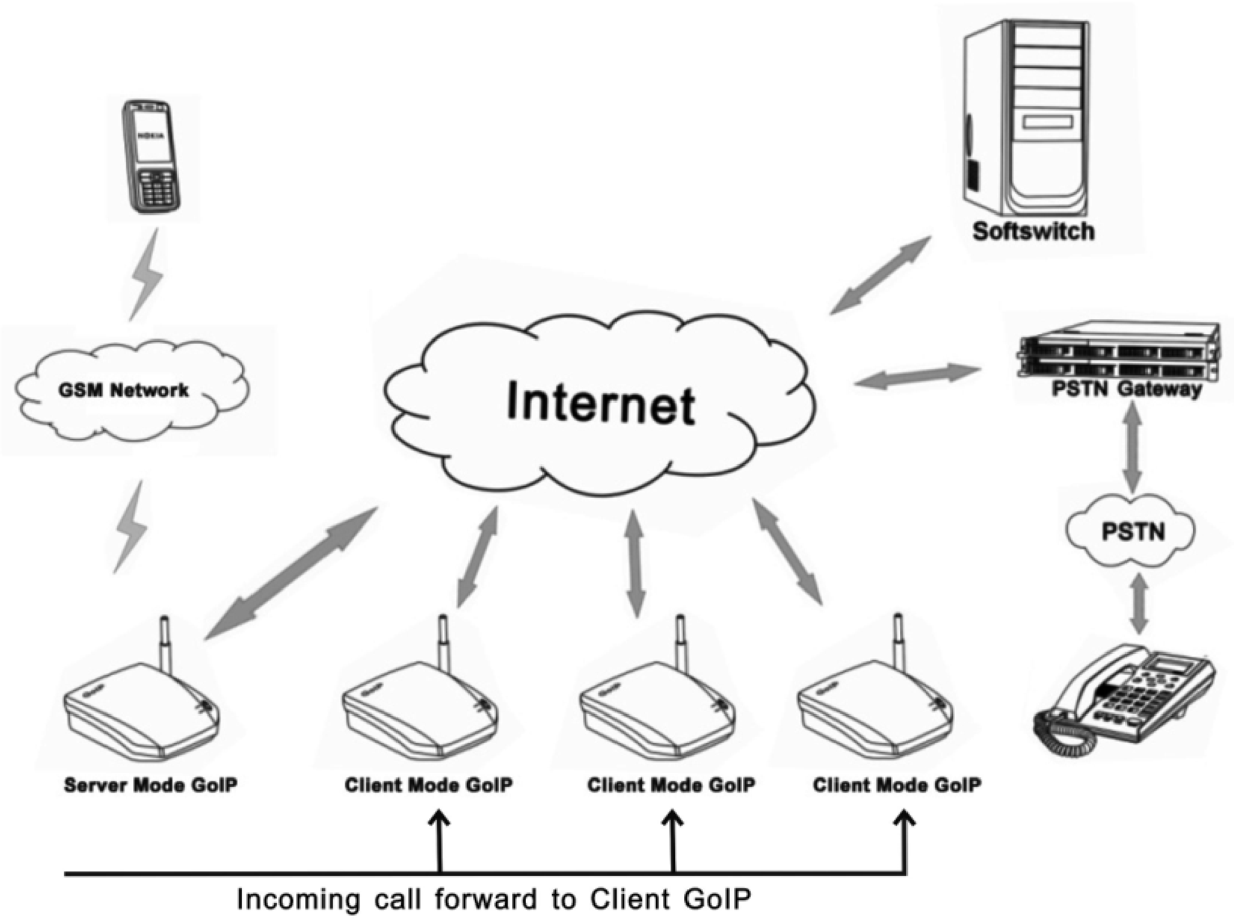
Where:

nc is the number of cadences
rpt is the repeat counter(0 - infinite, 1~n - repeat 1~n times)
c1on is the cadence one on (in milliseconds)
c1off is the cadence one off (in milliseconds)
c2on is the cadence two on (in milliseconds)
c2off is the cadence two off (in milliseconds)
c3on is the cadence three on (in milliseconds)
c3off is the cadence three off (in milliseconds)
f1 is the tone #1 frequency (300Hz-3000Hz)
f2 is the tone #2, frequency (300Hz-3000Hz)
f3 is the tone #3 frequency (300Hz-3000Hz)
f4 is the tone #4 (300Hz-3000Hz)
p1 is the attenuation index for f1, 0~31(0=3dB, -1dB increments)
p2 is the attenuation index for f2, 0~31(0=3dB, -1dB increments)
p3 is the attenuation index for f3, 0~31(0=3dB, -1dB increments)
p4 is the attenuation index for f4, 0~31(0=3dB, -1dB increments)

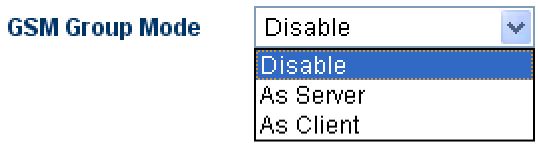
For example, the tone definition for a tone of 450Hz with a cadence of 700 ms on and 1000 ms off is **1,0,700,1000,0,0,0,0,450,0,0,0,20,0,0,0**

3.3.5 GSM Group Mode

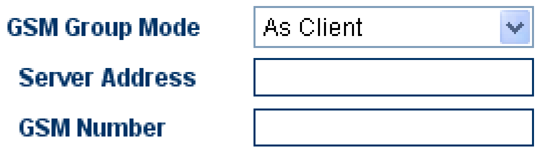
The GSM Group mode enables multiple GoIP devices to simulate a multi-channel GSM gateway.



In this mode, only one GoIP acts as a **Server** and the others act as clients of the server and reports its GSM number and status to the server. The number of clients is not restricted. When the server receives a GSM call, it finds an idle client (not engaged in a GSM call) and then forward the call to this client. This enables a scalable multi-channel VoIP GSM gateway. A typical application is to implement a call center that is accessed via a single phone number (GSM).



When **Client Mode** is selected, the **Server** IP address and the Client GSM number are required to be filled in as shown below.



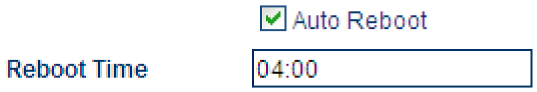
Note: Each GoIP still needs to register to VoIP server or proxy separately.

3.3.6 GSM Caller ID Anonymous

When this parameter is enabled, the GSM Caller ID is not sent; the Caller ID shown at the callee is anonymous.

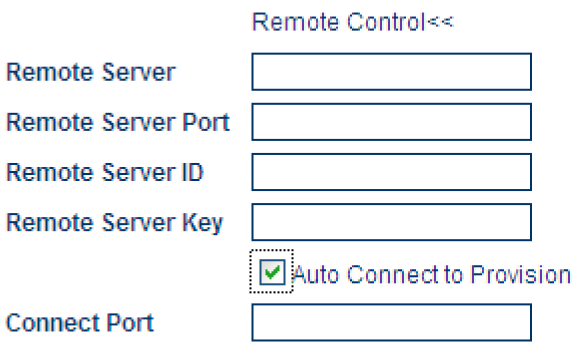
3.3.8 Auto Reboot

When the Auto Reboot box is checked, the GoIP reboots itself automatically at the time specified at the Reboot Time.



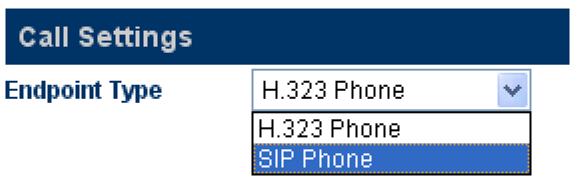
3.3.7 Remote Server

When this parameter is enabled, the GSM Caller ID is not sent; the Caller ID shown at the callee is anonymous.



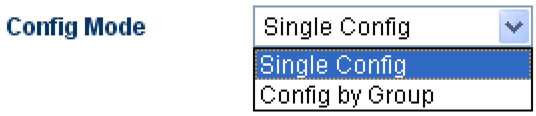
3.4 Call Settings

Click on the "**Call Settings**" to configure the VoIP call settings. The first thing to set is the Endpoint Type: H.323 or SIP.



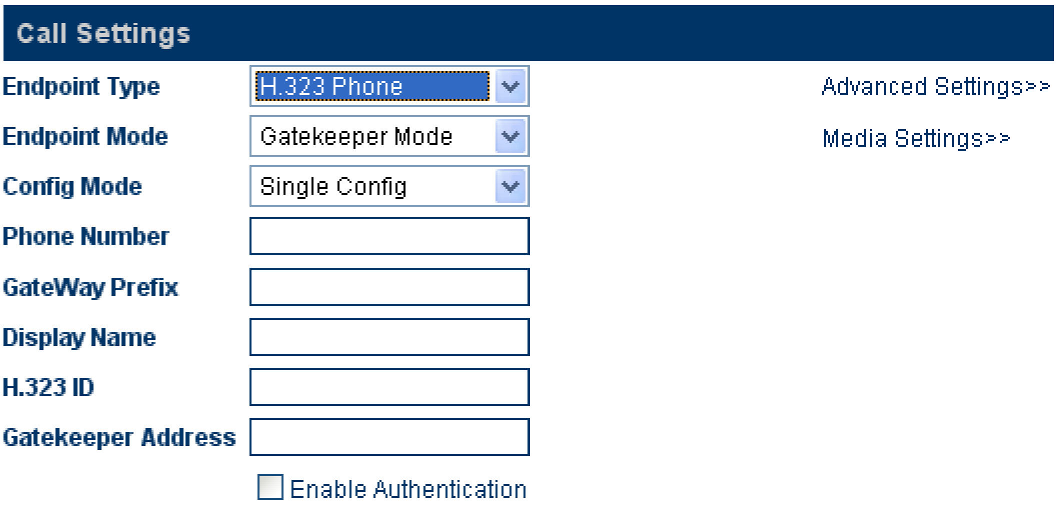
3.4.1 H.323 Phone

For H.323 protocol phone, 2 configuration modes are supported: **Single Configuration** and **Configuration by Group**.



3.4.1.1 Single Configuration

The **Single Configuration** supports only one VoIP number to a single H.323 Gatekeeper.



A. H.323 Phone Number

H.323 phone number: fill the login number (E164) here.

B. Gateway Prefix

If login with a Prefix method fill the prefix number (do not fill the Phone number)..

C. Display Name

Display name is the name to be displayed on the called VoIP party.

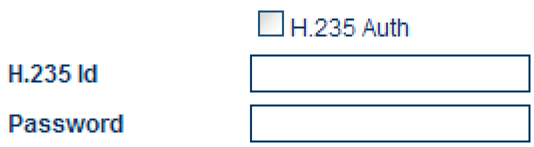
D. H.323 ID

If the system requires an H.323 ID as a method of authentication, enter the H.323 ID provided.

E. Gatekeeper Address

This field assigns the IP address or the domain name of the gatekeeper. The port number can be added with the colon ":" symbol. For example: 192.168.1.70:8080.

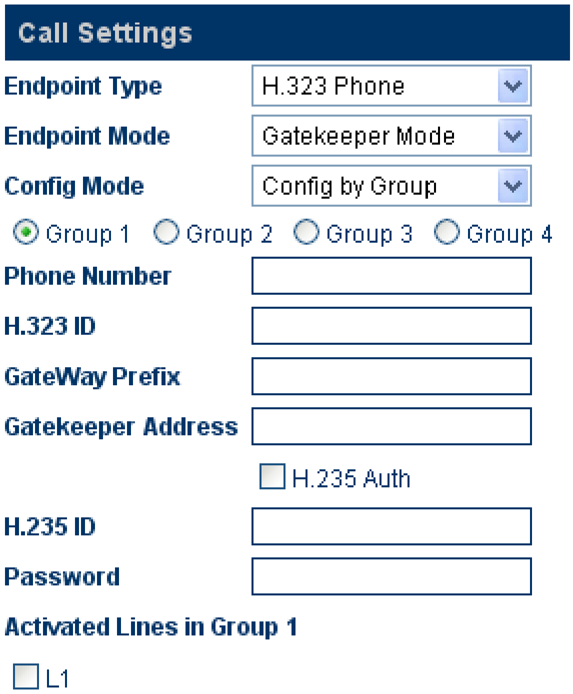
F. Enable Auth



If H.235 authentication is required, enable this field and fill in the values as provided.

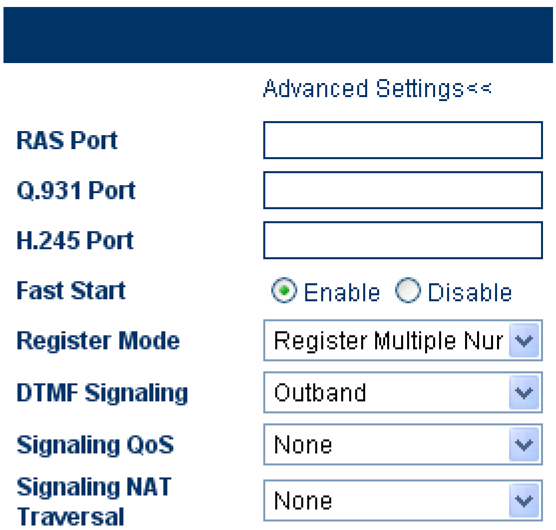
3.4.1.2 Configuration by Group

The “**Config by Group**” mode allows a user to setup the GoIP gateway to have 4 identities by registering to the same gatekeeper with different phone numbers or to different gatekeepers with different phone numbers or the same phone number, or a combination of both. The GoIP gateway can be assigned to each group individually. This allows the VoIP channel to be shared by different group.



3.4.1.4 Advance Settings

Click “**Advance Settings**” to access additional H.323 parameters as shown below.



A） RAS Port

RAS Port is an unreliable channel which is used to convey the registration, admissions, bandwidth change, and status messages between two H.323 entities.

B） Q.931 Port （ Call Signaling Port ）

Call Signaling Port is a reliable channel which is used to convey the call setup and release messages between two H.323 endpoints.

C） H.245 Port （ Media Control Ports ）

Media control port is the port or port range used by the H.245 media control protocol.

D） Fast Start

Enable or disable the Fast Start in H.225.0. Most H.323 terminals or gateways support the **Fast Start** feature.

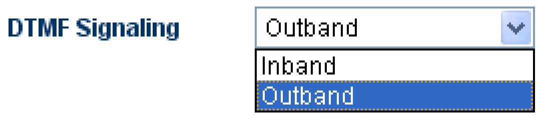
E） Register Mode

Register Multiple Numbers:	The GoIP gateway sends registration request in one signaling packet to the gatekeeper. In the mode, one signaling packet includes two VoIP line's registration information.
Register Multiple Times:	In this mode, the GoIP gateway will register like two terminals.

F） DTMF Signaling

1） DTMF TYPE

DTMF signals can be sent over to the called party once a call is established. GoIP gateway supports both **Inband** and **Outband** DTMF signal types.



For **Inband** DTMF type, DTMF signals are generated locally at the calling phone and then send to the called party as part of the voice signals. This method is not reliable since the quality of the DTMF signals is subject to the codec used and the quality of the network traffics.

For **Outband** DTMF type, DTMF signal commands are sent to the called party and the actual DTMF signals are actually generated by the called party. This method allows more reliable DTMF signaling. However, it requires the called party to support this feature in order for this to work properly. GoIP gateway supports **RFC2833** Outband DTMF protocols.

2）DTMF Payload Type

DTMF Payload Type is by RFC2833 protocol to carry the tone definitions for various applications. The default DTMF payload type is 96. Please consult your VoIP service provider for the proper setting if required.

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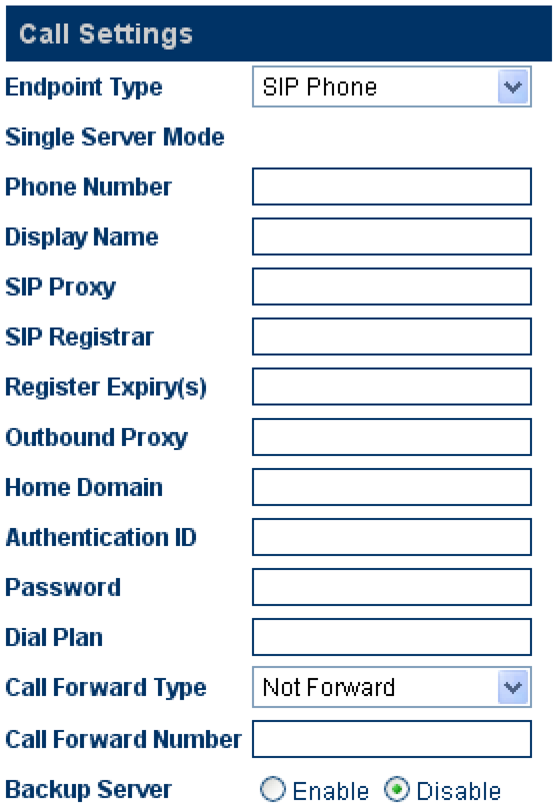
3.4.1.5 H.323 Direct Mode

The **Direct Mode** allows peer-to-peer calls without registering to a gatekeeper.

3.4.2 SIP Phone

Set the “**Endpoint Type**” to SIP phone for connections to SIP servers.
GoIP gateway’s SIP configure page as follow:



A）**Phone Number**

Enter a SIP phone number.

B）**SIP Proxy**

Enter the SIP proxy IP address or domain name. If the registration port isn’t 5060, then add “**:**” and the port number. An example is **sip.yourdomain.com:8080**.

C）**SIP Registrar Server**

If the Registrar Server is different from the SIP Proxy, enter its IP address or domain name in this field. If the registration port isn’t 5060, then add “**:**” and the port number. An example is **sip.yourdomain.com:8080**.

D）**Home Domain**

SIP networks sometimes use the Home Domain name as an identifier. Enter this field as required.

E） Authentication ID

Enter the authentication ID as provided.

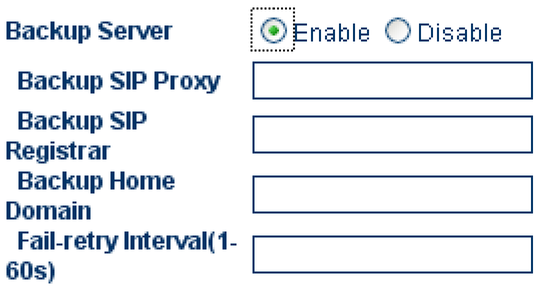
F） Password

Enter the authentication password as provided.

G） Display Name

Enter this field for the name to be displayed on the called VoIP party.

H） Backup Server



The GoIP gateway supports one backup server in case of a main server failure. Once registration to the main server fails, the GoIP gateway will try to register to the backup server.

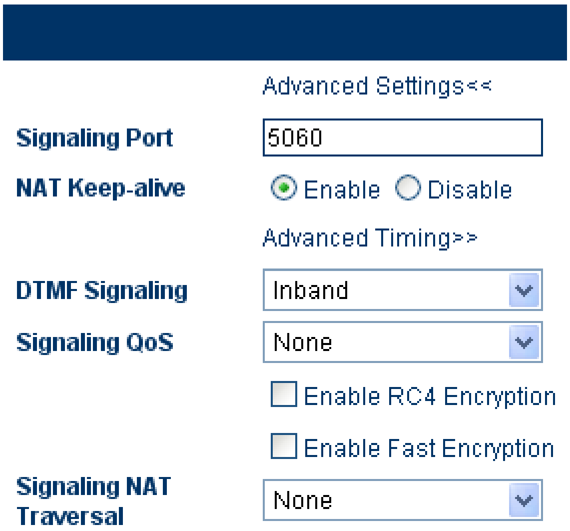
I） Outbound Proxy

OutBound proxies are devices that will forward SIP signaling (and frequently RTP media traffic too). OutBound proxies are used for a number of reasons, including, firewall traversal – both in parallel with a firewall and situated in the Internet as a Session Border Controller, and also for hiding customer IP addresses – calls are all routed through one point so that a public ITSP address can be used for accessing the customers, rather than the customer's own IP address.

If required, enter this field with the outbound proxy IP address or domain name as provided.

3.4.2.1 Advanced Settings

Click on "**Advance Settings**" tab on the top right corner of the Call Setting page to display all the parameters available, as shown below, for programming. These parameters allow more advanced control over the SIP signaling and media preference.



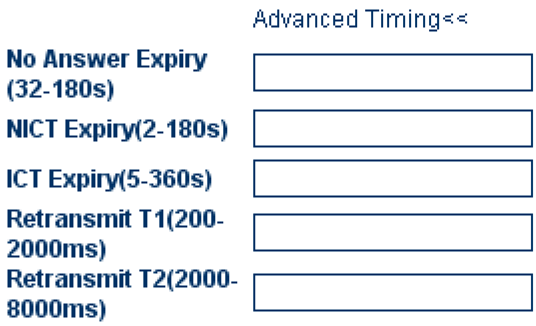
A） **Signaling Port (SIP Local port)**

The default SIP port is 5060. Change this as required.

B） **NAT Keep-alive**

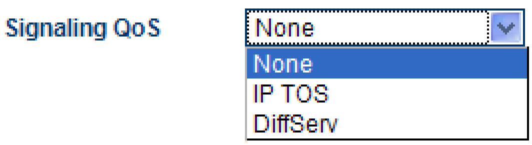
The NAT Keep-alive feature sends a null packet to the SIP proxy periodically in order to keep the NAT open for incoming data traffics.

C） **Advanced Timing Settings**



Some SIP proxies may have special timing requirements. Change these parameters as required.

D） **Signaling Qos**

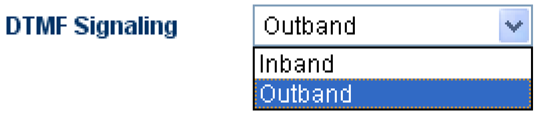


Signaling QoS improves the performance of SIP signaling. If local network device supports Qos, select this field accordingly. Please consult your network administrator for further information.

E） DTMF Signaling

1） DTMF TYPE

DTMF signals can be sent over to the called party once a call is established. GoIP gateway supports both **Inband** and **Outband** DTMF signal types.



For **Inband** DTMF type, DTMF signals are generated locally at the calling phone and then send to the called party as part of the voice signals. This method is not reliable since the quality of the DTMF signals is subject to the codec used and the quality of the network traffics.

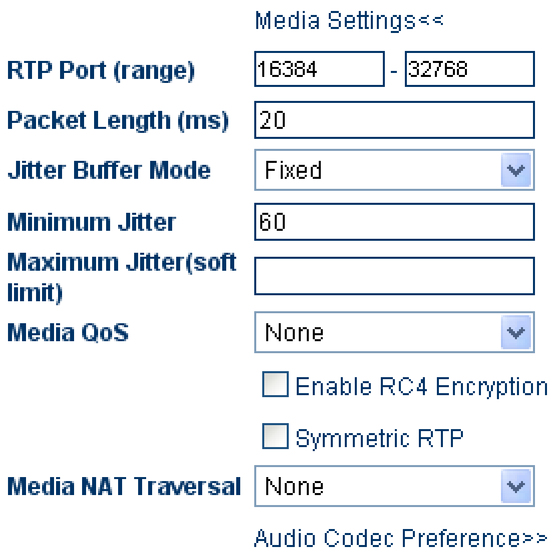
For **Outband** DTMF type, DTMF signal commands are sent to the called party and the actual DTMF signals are actually generated by the called party. This method allows more reliable DTMF signaling. However, it requires the called party to support this feature in order for this to work properly. GoIP gateway supports both RFC2833 and SIP INFO **Outband** DTMF protocols.

2）DTMF Payload Type

DTMF Payload Type is by RFC2833 protocol to carry the tone definitions for various applications. The default DTMF payload type is 96. Please consult your VoIP service provider for the proper setting if required.

3.4.3 Media Setting

Click on “**Media Settings**” in the “Call Setting” menu to access the parameters available for media settings.



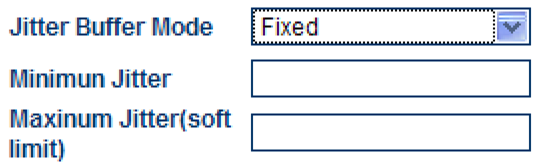
A） **RTP Port Range**

This parameter specifies the range of the RTP (Real Time Protocol) ports used by the GoIP gateway. If your network limits the usable port range, this parameter may need to be modified. Please consult your network administrator for more information.

B） **Packet Length**

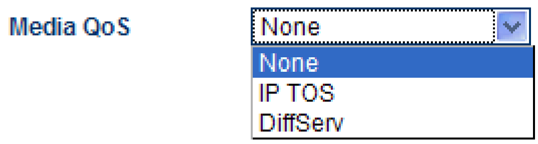
This parameter defines the voice packet length. The default setting is 20ms. The range is from 5ms to 40ms at an increment of 5ms. Please note that some codes have a minimum packet length of more than 5 ms.

C） **Jitter Buffer Mode**



Since data packets may arrives at different orders, the jitter buffer is used to hold the data packets received for re-arrangement according to the packet sequence number. Three jitter buffer modes are supported: Adaptive, Sequential, and Fixed. The default is set to Adaptive mode with a minimum jitter of 60 ms and a maximum jitter of 120ms. Please consult your network administrator for more information on the network environment in order to determine the optimal settings.

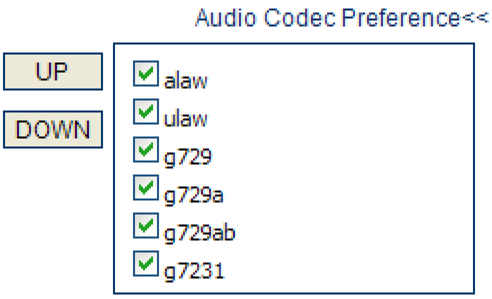
D） **Media QoS**



Similar to the Signaling QoS, the Media Qos is intended to improve the voice performance or quality if QoS is supported by your local network.

3.4.4 Codec Preference

Codec Preference allows a user to select the codes to be used and its priority to be selected for a voice call.

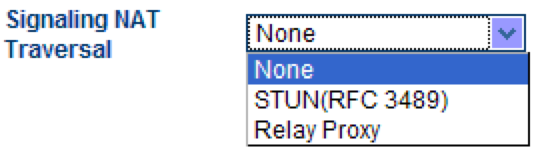


Click on the check box to enable a codec. Select a codec and then press the UP or DOWN button to move the position of the codec on the codec list with a priority in descending order.

3.4.5 NAT Traversal

3.4.5.1 Signaling NAT Traversal

Signaling NAT traversal may be required if the GoIP gateway is put behind a NAT (or multiple NATs). Depending on your network environment and SIP server capabilities, this feature may or may not be turn on.



A） None

Select **None** to turn off this feature.

B） STUN (RFC 3489)

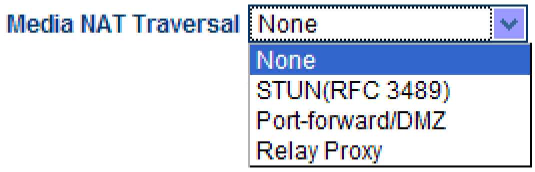
STUN (**Simple Traversal of UDP (User Datagram Protocol) through NATs (Network Address Translators)**) is a network protocol allowing a client behind a NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet-side port associated by the NAT with a particular local port.
Select **STUN (RFC 3489)** to use a STUN server for Signaling NAT Traversal. Enter the IP address or the domain name of the STUN server to be used.

C） Relay Proxy

Relay proxy is a proprietary NAT traversal technology. Please consult your service provider for more information.

3.4.5.2 Media NAT Traversal

Similar to Signaling NAT Traversal, this feature allows media packets (RTP) to be routed properly in various network environments.



A） None

Select **None** to disable this feature.

B） STUN （RFC 3489 ）

STUN (**Simple Traversal of UDP (User Datagram Protocol) through NATs (Network Address Translators)**) is a network protocol allowing a client behind a NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet-side port associated by the NAT with a particular local port.

Select **STUN(RFC 3489)** to use a STUN server for Signaling NAT Traversal. Enter the IP address or the domain name of the STUN server to be used.

C）Port forwarding Support

Port forwarding (sometimes referred to as tunneling) is the act of forwarding a network port from one network node to another. This technique allows an external user to reach a port on a private IP address (inside a LAN) from the outside via a NAT-enabled router.

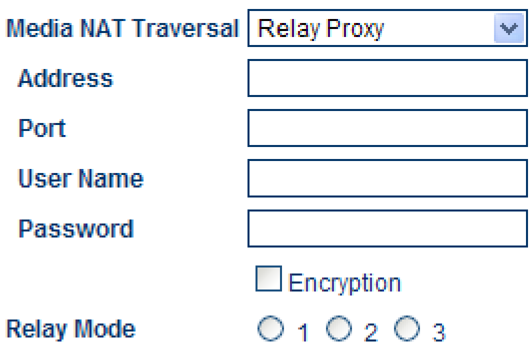
In order for this feature to work, the local network gateway must support this feature and be set up properly. Please consult your network administrator for help to enable this **Port forwarding** feature.

D） Relay Proxy

Relay proxy is a proprietary NAT traversal technology. Please consult your service provider for more information.

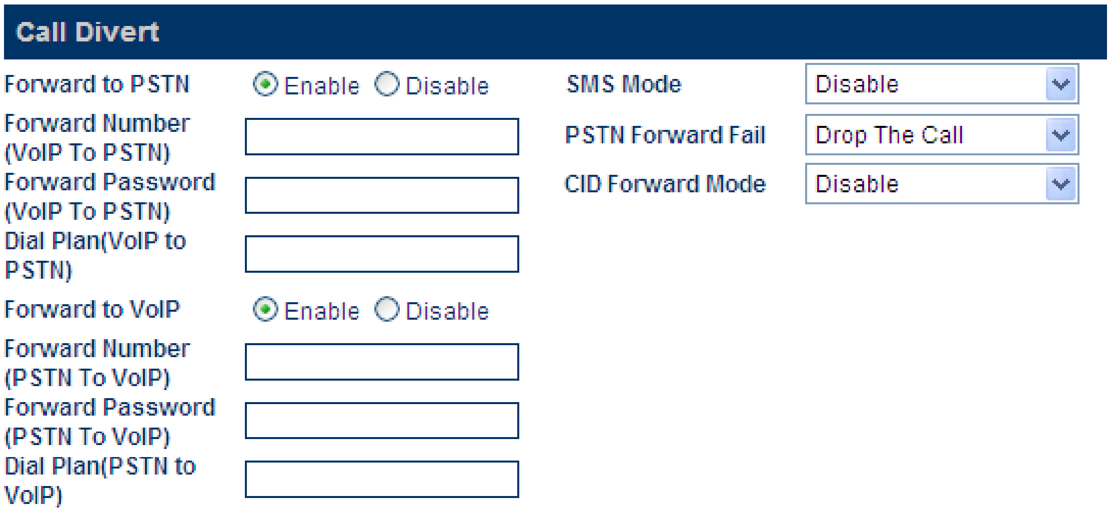
Currently, the following 3 kinds of packaging mechanism are supported:

- **Mode 1: The media uses UDP packets and (or) encrypt with multiple UDP port;**
- **Mode 2: The media uses UDP packets and (or) encrypt with single UDP port;**
- **Mode 3: The media uses TCP packets and (or) encrypt (UDP over TCP).**



3.5 Call Divert

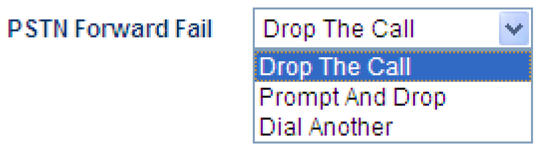
The **Call Divert** feature controls the routing of calls between VoIP and GSM.



<u>Call Forward (From VoIP to PSTN)</u>

Forward Number

Enter this field to forward all incoming VoIP calls to this number (PSTN or Mobile). Using "," to add a 500ms delay to the dialing sequence. If this field is blank, calls will not be forwarded. The GoIP gateway answers an incoming VoIP call and generates a dial tone. The caller can then dial the number (PSTN or Mobile) desired.



Forward Password

This field sets the password protection for using the GSM connection. If a password is entered, the GoIP gateway will generate an indication tone and wait for the call to dial the

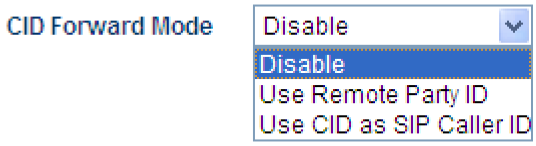
Dial Plan to PSTN

This field sets the password protection for using the GSM connection. If a password is entered, the GoIP gateway will generate an indication tone and wait for the call to dial the

<u>Call Forward (From GSM to VoIP)</u>

Forward Number

Forward all incoming calls from the GSM connection to the VoIP number specified in this field. Forward Password is not required once this field is set. If this field is blank, the GoIP answers an incoming GSM calls and then generates the VoIP dial tone. Please see below if the Forward Password is set. The caller can then dial a VoIP number manually. At the end, a pound (#) can be dialed to activate the dialing of the VoIP number immediately. If not, the VoIP number is dialed after a preset timeout.



Forward Password

This field sets the password protection for incoming GSM calls. If a password is entered, the GoIP gateway will generate an indication tone after answering an incoming call. The caller is then ready to dial the password. Once the password is correctly entered, the GoIP gateway generates a VoIP dial tone and waits for the caller to dial a VoIP number.

Dial Plan to VoIP

This field sets the password protection for using the GSM connection. If a password is entered, the GoIP gateway will generate an indication tone and wait for the call to dial the

3.5.1 Dial Plan

In Call Divert mode, dial plan is supported in order to pre-program the various call routes based on the number dialed. Dial plan can be defined individually for VoIP to PSTN/GSM calls and GSM/PSTN to VoIP calls.

3.5.1.1 Basic Syntax

1. Multiple rules are supported; the "|" character is used as a separator between two rules. For example: "00:-00|0:-0+86|:+86755"
2. The rules are examined and executed from left to right. Whenever a match is found, the rule examination terminates and the rule matched is executed immediately.
3. The dial plan syntax is "**A:-a+b**" where **A**, **a**, and **b** are single or multiple digits, for example "**0:-0+86**". The "**A**" before the colon ("**:**") is the matching condition and the "**-a+b**" after the "**:**" is the action to be executed. If the condition "**A**" is matched, the "**a**" portion of the number dialed (matched only from the first digit) is taken out and the "**b**" portion is added to the beginning of the number dialed. If a match is not found, the dial plan matching terminates and no action is performed on the number dialed. If "**A**" is not present, the rule is executed immediately after the number is dialed All subsequent rules, if defined, are ignored.

   Note: "**a**" .must be a subset of "**A**" in order for this to work.

4. Range definition is supported. Use "**[A-B]**" to specify the digit range desired. For

example, **A** = **[2-8]** means that the number dialed with the leading digit being in the range from 2 to 8 is a match and the corresponding action is executed.

Examples:

1. Dial Plan: "0:|:+0755".

a. Input: "02083185711" -> Output: “02083185711";

b. Input "83185700" -> Output: "075583185700".

2. Dial Plan："00:-00|0"-0+86|:+86755".

a. Input: "008522343318" -> Output: "8522343318";

b. Input: "02083185711" -> Output: "862083185711";

c. Input: "83185700" -> Output: "8675583185700".

3. Dial Plan："00:|0:-0+0086|:+0086755".

a. Input: "008522343318" -> Output: "008522343318";

b. Input: "02083185711" -> Output: "00862083185711";

c. Input: "83185700" -> Output: "008675583185700".

4. Dial Plan："0:|1[3-9]:+0|[2-8]:+0755|:+0755".

a. Input: "076322343318" -> Output: "076322343318";

b. Input "13044557766" -> Output: "013044557766";
   or Input: "13644557766" -> Output: "013644557766"

c. Input: "23185700" -> Output: "075523185700".
   or Input: "73185700",-> Output: "075573185700"

3.5.1.2 Advanced Syntax for Limiting Number Length

To limited the length of the number input, the syntax is in the following format:

"DDXXXXXX:-a+b"

The "DDXXXXXX" before the “**:**” is the matching condition for a number with a length of 8 digits. The first two digits, “**DD**” are predefined and the rest can be any digits. Please note that only the first 8 digits of the number entered is used and any extra digits are ignored.

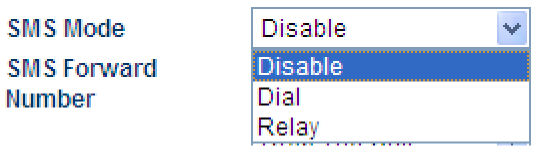
For examples:

1. Dial Plan: "00:|0:-0+0086|[1-8]xxxxxxx:+0086755".
   Input Number: 21234567
   Output Number: 008675521234567
2. Dial Plan: "0:|13[0-9]xxxxxxxx:+0|[1-8]xxxxxxx:+0755"
   Input Number: 13812345678
   Output Number: 013812345678
   Note: A “0” is added to the beginning of all China Cellphone number (11-digit) starting with the leading two digits as 13

Input Number: 83185922
Output Number: 075583185922
Note: “0755” is added to the beginning of all local number (8-digit)

3.6 SMS Mode

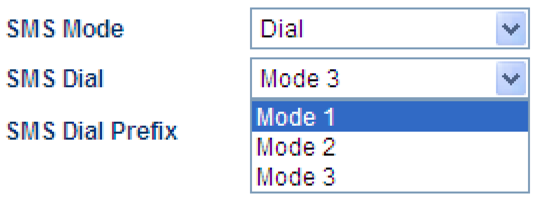
GoIP supports SMS **Dial** mode and SMS **Relay** mode. The **Dial** mode uses SMS for Call Back service and the **Relay** mode bridges the SMS messages between GSM and VoIP. Please note that **Dial** mode only supports incoming SMS messages and out going SMS from VoIP to GSM is disabled.



3.6.1 SMS Dial Mode in SIP

SMS Dial mode in SIP is commonly used for Call Back Service to save on GSM / Long Distance call charges. The caller initiates a call by sending a SMS message to the service provider via the GoIP. Once the GoIP receives the SMS, it will initiates a call to the SIP Server (Service Provider) based on the SMS contents. This is not an actual voice call that will be answered. Once the Service Provider receives the call information via the SIP call, it then connects the GSM Caller and the Callee (the number specified in the SMS content) together. Please note that this feature is intended for a Service Provider to integrate the GoIP in their system for Call Back Service.

Select the SMS Dial Mode and then select the correct mode for SMS message formats as described below:



1. **Mode 1**
   GoIP uses the Caller ID of the GSM/SMS caller to call to the number specified in the SMS message content.

   For example:

GSM Caller **(+86) 13800000000** sends a SMS to the GoIP with the message "**8675588228822**". The GoIP uses the GSM Caller ID (**8613800000000**) as its SIP ID to call the number **8675588228822** and the SIP Sever is located at 192.168.2.1 using signaling port 5060. The INVITE message 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: DBL
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER, MESSAGE, INFO, SUBSCRIBE
Content-Type: application/sdp
Content-Length: 226

2. **Mode 2**

GoIP uses its SIP number as the caller ID to call the number specified in the message content.

For example:

GSM Caller **(+86) 13800000000** sends a SMS to the GoIP with the message "**8675588228822**". The GoIP has a SIP number **20001** registered to the SIP Sever located at 192.168.2.1 (signaling port 5060). In mode 2, the GoIP uses its own SIP ID to call the number **8675588228822**. The INVITE message 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: DBL
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER, MESSAGE, INFO, SUBSCRIBE
Content-Type: application/sdp
Content-Length: 226

3. **Mode 3**

GoIP uses the GSM Caller ID and the SMS message content to form a new SIP number. It will then call this number. This SIP number is in the format shown below.

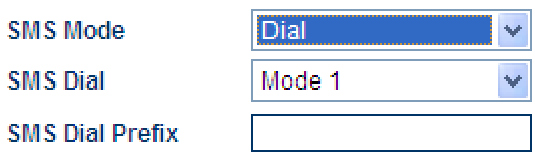
SIP number = SMS Message content + “*” + GSM Caller ID

For example:

GSM Caller **(+86) 13800000000** sends a SMS to the GoIP with the message “**8675588228822**”. The GoIP has a SIP number **20001** registered to the SIP Sever located at 192.168.2.1 (signaling port 5060). In mode 2, the GoIP uses its own SIP ID to call the SIP number **8675588228822*8613800000000**. The INVITE message is:

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

SMS Dial Mode Prefix is used to adds a prefix to the SIP number to be called in the SMS Dial Mode.



For example:

Setting the SMS Dial Prefix to 999 in Mode 1 changes the INVITE message to:

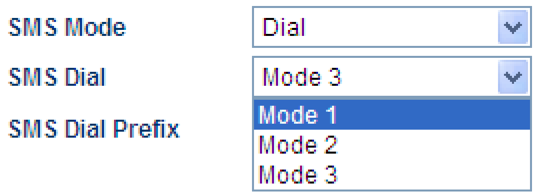
INVITE sip:9998675588228822@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK363969813

From: <sip:8613800000000@192.168.2.1:5060>;user=phone;tag=65248630
To: <sip:9998675588228822@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: DBL
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER, MESSAGE, INFO, SUBSCRIBE
Content-Type: application/sdp
Content-Length: 226

3.6.2 SMS Dial Mode in H,323

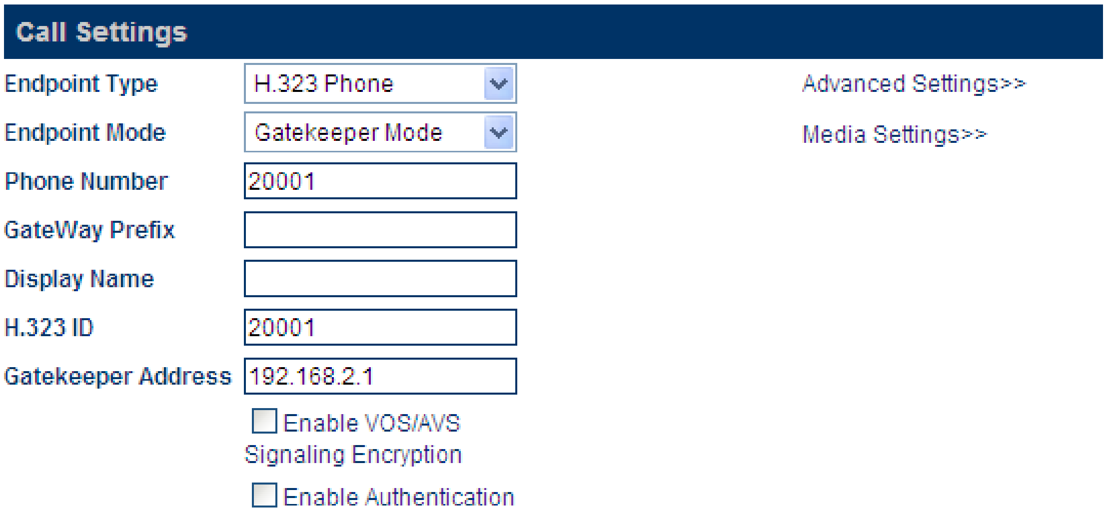
Similarly, SMD Dial Mode also works in H.323 Protocol in order to support Call Back Service. The user still sends a SMS message to the GoIP to initiate the call back. The H.323 protocol for initiating a call after receiving a Call Back SMS message Call Back has 3 different formats as described below.

Select the SMS Dial Mode and then select the correct mode for SMS message formats as described below:



1. **Mode 1 (Currently not supported)**
   GoIP uses the Caller ID of the GSM/SMS caller to call to the number specified in the SMS message content.
2. **Mode 2 (Currently not supported)**
   GoIP uses its SIP number as the caller ID to call the number specified in the message content.

   For example:



GSM Caller **(+86) 13800000000** sends a SMS to the GoIP with the message "**8675588228822**". The GoIP uses its own H.323 ID to call the number **8675588228822**. The H.323 protocol command involved is:

Send RAS Message: admissionRequest

```
admissionRequest {
  requestSeqNum = 241
  callType = pointToPoint NULL
  endpointIdentifier = "3705_endp"
  destinationInfo = 1 elements {
    [0] = dialedDigits "8675588228822"
  }
  srcInfo = 2 elements {
    [0] = dialedDigits "20001"
    [1] = h323-ID "20001"
  }
  srcCallSignalAddress = ipAddress {
    ip = 4 octets {
      c0 a8 02 ed                                  ....
    }
    port = 2049
  }
  bandWidth = 2048
  callReferenceValue = 7502
  conferenceID = 16 octets {
    7f f3 78 77  49 3f 4c c1  9a dc 6a 84  12 d8 30 8f    ..xwI?L...j...0.
  }
  activeMC = FALSE
  answerCall = FALSE
  canMapAlias = FALSE
  callIdentifier = {
```

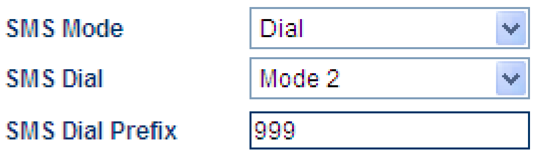
```
            guid = 16 octets {
              cb  40  a4  af    8e  9b  60  96    6b  5f  a0  03    f2  ed  55
      5b      .@....`.k_....U[
            }
          }
          gatekeeperIdentifier = "GnuGk"
          willSupplyUUIEs = FALSE
        }
```

3. Mode 3 (Currently not supported)

GoIP uses the GSM Caller ID and the SMS message content to form a new SIP number. It will then call this number. This SIP number is in the format shown below.

SIP number = SMS Message content + "*" + GSM Caller ID

SMS Dial Mode Prefix is used to adds a prefix to the SIP number to be called in the SMS Dial Mode.



For example:

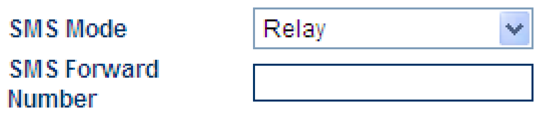
Setting the SMS Dial Prefix to 999 in Mode 1 changes the INVITE message to:

```
Send RAS Message: admissionRequest
  admissionRequest {
    requestSeqNum = 241
    callType = pointToPoint NULL
    endpointIdentifier = "3705_endp"
    destinationInfo = 1 elements {
      [0] = dialedDigits "9998675588228822"
    }
    srcInfo = 2 elements {
      [0] = dialedDigits "20001"
      [1] = h323-ID "20001"
    }
    srcCallSignalAddress = ipAddress {
      ip = 4 octets {
        c0 a8 02 ed                                        ....
      }
      port = 2049
```

```
      }
      bandWidth = 2048
      callReferenceValue = 7502
      conferenceID = 16 octets {
        7f f3 78 77   49 3f 4c c1   9a dc 6a 84   12 d8 30 8f      ..xwI?L...j...0.
      }
      activeMC = FALSE
      answerCall = FALSE
      canMapAlias = FALSE
      callIdentifier = {
        guid = 16 octets {
          cb  40  a4  af    8e  9b  60  96    6b  5f  a0  03    f2  ed  55
   5b      .@....`.k_....U[
        }
      }
      gatekeeperIdentifier = "GnuGk"
      willSupplyUUIEs = FALSE
    }
```

3.6.3 SMS Relay Mode (FOR SIP ONLY)

The SMS Relay mode bridges SMS messages between GSM and VoIP. This means that SMS messages can send from GSM phone to SIP phone and SIP phone to GSM.phone. This feature is only available for SIP protocol.



Select the Relay mode as shown above and enter the SMS Forward Number. Please make sure that this extension phone must support SMS since all GSM SMS received will be forwarded to this phone. Please also note that the SIP server must support this SMS feature as well.

1. Relay a GSM SMS to a SIP Phone

Here is an example of relaying a received GSM SMS to a SIP Phone:

a) GSM SMS is sent from **8613682626865.**
b) GSM SMS content is **075583185700.**
c) SMS Forward Number is set to **3999**
d) GoIP creates a new message containing the GSM Caller ID and the GSM SMS received

   8613682626865
   075583185700
e) The SIP Message sent from GoIP to the SIP Server is:

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: DBL
Content-Type: text/plain
Content-Length: 28

8613682626865
075583185700

2. Relay a SIP SMS to a GSM Phone

Here is an example of the SIP SMS sent to the GoIP

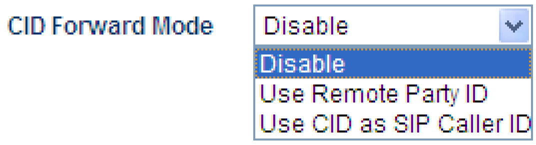
a) SIP number or extension of sending SMS is 3999
b) GoIP SIP number is 20001
c) Designated GSM number for the SMS is **13682626800**
d) SMS Message content entered to 3999 is
   13682626800
   Hello world
e) SMS Message content to GSM phone is **Hello world**
f) The SIP Message sent from GoIP to the SIP Server is:

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

3.7 Relay Incoming Caller ID (GSM to VoIP Call)

For SIP mode, the GoIP allows the incoming caller ID from a GSM Call to be transferred to a VoIP terminal. This parameter is called **CID Forward Mode** and it can be accessed under the **Call Divert Page** as shown below.



Three selections are available:

1. **Disable** – This mode disables the incoming caller ID from a GSM call to be forwarded to a VoIP terminal.

2. **Use Remote Party ID** – This sets the SIP method used to relay the incoming caller ID to a VoIP terminal. The method is called Remote Party ID. For example, if the incoming caller ID is 13800000000, the content of the SIP INVITE message is show below. Please note the section is marked in red.

```
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
```

3. **Use CID as SIP Caller ID** – This sets the SIP method used to relay the incoming caller ID to a VoIP terminal. The method is similar to the Remote Party ID except that the SIP ID is replaced with the incoming caller ID.. For example, if the incoming caller ID is 13800000000, the content of the SIP INVITE message is show below. Please note the section is marked in red.

```
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
```

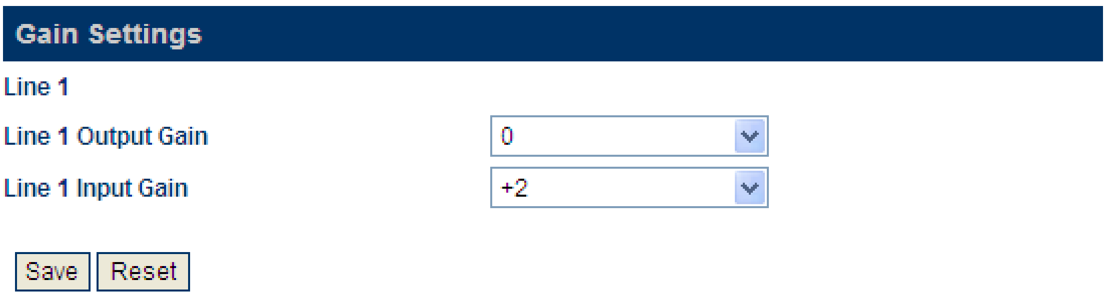
For H.323 mode, this feature is currently not supported and it will be supported in future firmware releases.

3.8 Gain Settings…

A hidden webpage is provided to set the receiving and transmit gains of VoIP Channel. The URL link is:

[http://xxx.xxx.xxx.xxx/default/en_US/**gain.html**](http://xxx.xxx.xxx.xxx/default/en_US/gain.html)

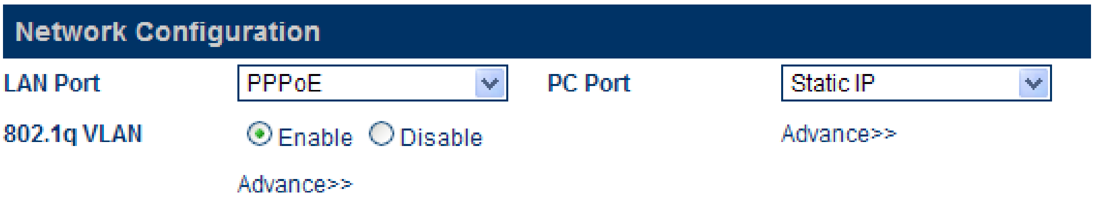
THIS PAGE IS INTENDED FOR AN EXPERIENCED USER OR AN ADMINISTRATOR ONLY. PLEASE SET THE GAINS WITH CAUTIONS.



Note: A too low or too high input gain MAY affect the sensitivity of DTMF detections

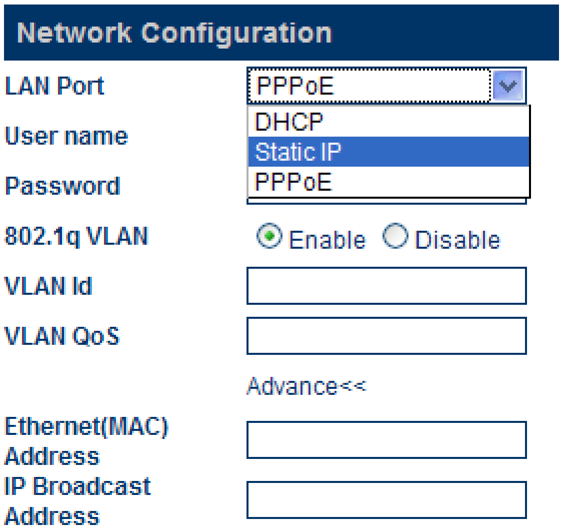
3.9 Network Configuration

Click on “**Network**” tab in the left menu column to configure the **LAN** and **PC** ports.



3.9.1 LAN Port

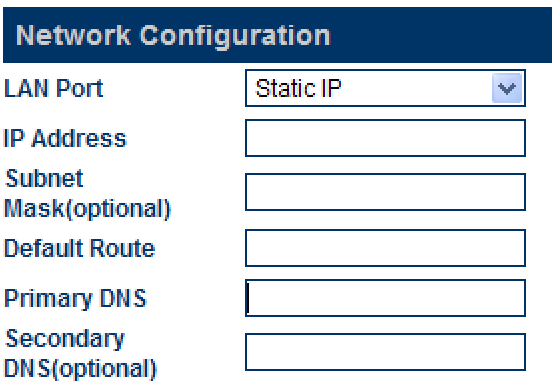
Three LAN Port modes are supported: **DHCP**, **Static IP**, **PPPoE**.



1） DHCP

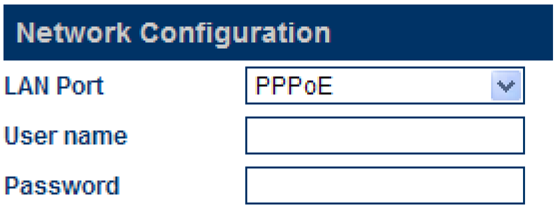
Choose **DHCP** if a local DHCP host is available. This allows the GoIP gateway to obtain network information (IP Address, Subnet Mask, Default Route, Primary DNS, Secondary DNS, and other DHCP options) from the DHCP host.

2） Static IP



Choose **Static IP** if your network topology requires. Please fill in the **IP Address**, **Subnet Mask**, **Default Route**, **Primary DNS**, and **Secondary DNS** (optional) as provided by your network administrator.

3） PPPoE



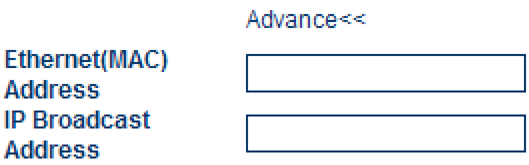
PPPoE is a common dial up method for you network modem (Cable / xDSLs). Choose this if your network environment requires. Enter the **User Name** and **Password** as provided by your ISP.

4）802.1q VLAN

This QoS feature requires your QoS support of your network to improve voice data traffics. Please consult your network administrator for proper settings.

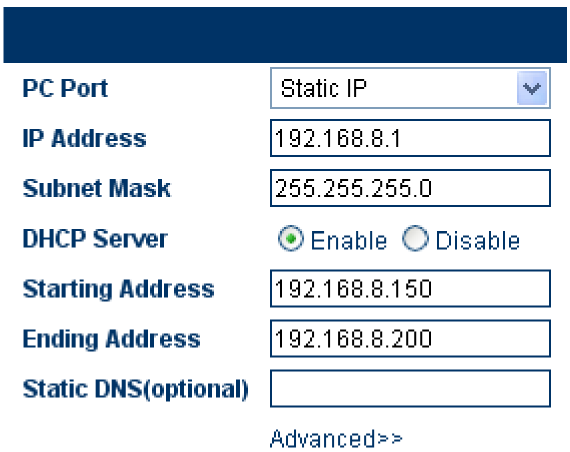
5） Advanced…

The **Advanced** settings allow the user to set the broadcast address and to clone a MAC address instead of using the factory preset MAC address. Please consult your network administrator for further information.



3.9.2 PC port configurations

The PC Port allows addition network devices to be attached behind the GoIP Gateway. It offers both Bridge and Static IP modes to meet your network topology. It is factory preset to the Static IP mode with the IP address 192.168.8.1.

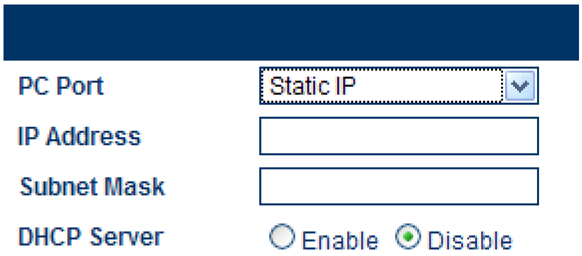


1） Bridge Mode

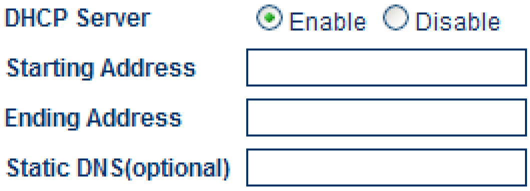
Select **Bridge** mode if your network topology requires the network devices (PC or others) to be in the same network segment as the GoIP gateway. In this case, the GoIP gateway functions as an Ethernet switch.

2） Static IP Mode (Default Setting)

Select **Static IP** mode for a new network segment for the network devices behind the GoIP gateway. In this case, the GoIP gateway functions as an Ethernet router. Fill in the **IP Address** field with a new segment address that is different from that for the LAN port. Please select the **Subnet Mask** accordingly. A commonly used value is 255.255.255.0.



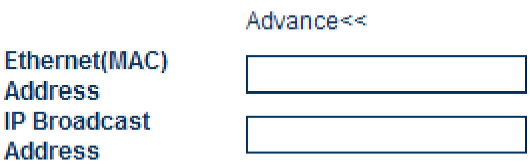
Enable the **DHCP Server** if you want the GoIP gateway functions as a local DHCP host for the PC segment. This will enables the GoIP gateway to assign IP addresses to network devices that are attached to the PC port segment.



Specify the **Starting Address**. **Ending Address**, and **Static DNS** accordingly.

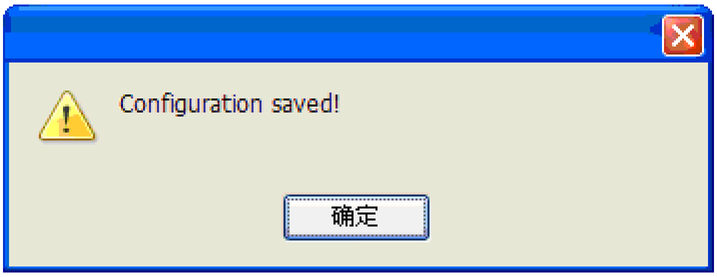
4） Advanced…

The **Advanced** settings allow the user to set the broadcast address and to clone a MAC address instead of using the factory preset MAC address. Please consult your network administrator for further information.



3.10 Save Configuration

To confirm and commit all changes made, click on the **Save Changes** tab. Otherwise, all changes will be discarded. Once all changes are saved, the following screen message is displayed.

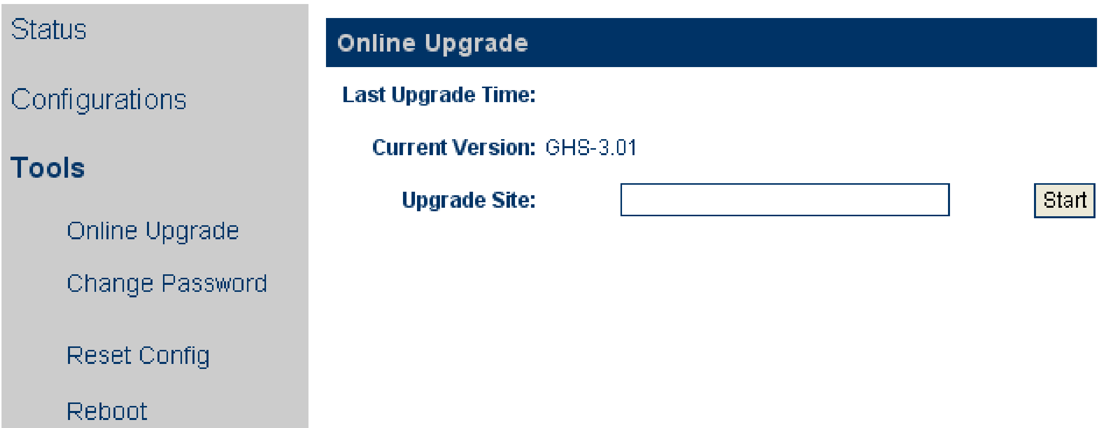


3.11 Discard Changes

To discard all changes made, click on the **Discard Changes** tab.

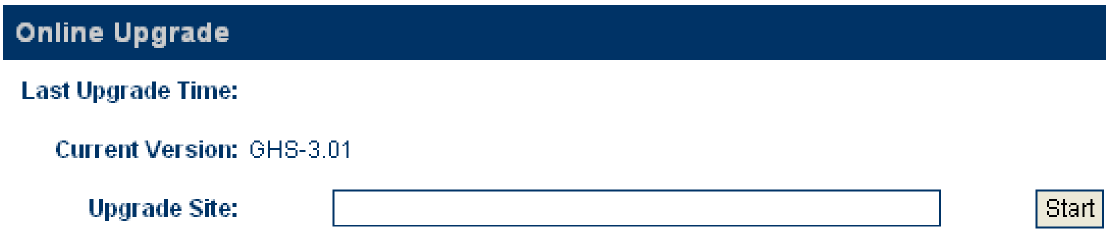
3.12 Tools Menu

Select the **Tools** to access the following functions: **Online Upgrade**, **Change Password**, **Reset Config**, and **Reboot**.



3.12.1 Online Upgrade

To perform a firmware upgrade, select the **Online Upgrade** tab to access the page below.



Enter the update link as provided by your service provider. A sample link is:

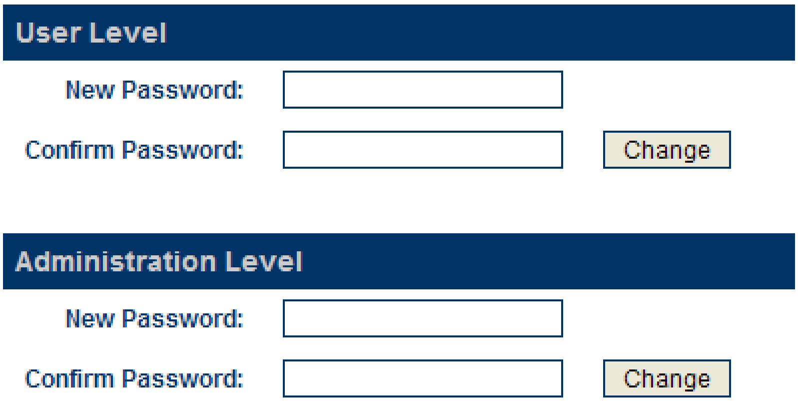
http://202.155.200.154/update/A34HS-3.07-18.pkg

Click the **Start** button to start the firmware upgrade.

WARNING: POWER SHUTDOWN / FAILURE DURING FIRMWARE UPGRADE MAY PERMINENTLY DAMAGE THE GOIP GATEWAY.

3.12.2 Change Password

Click on the **Change Password** tab to access the page below.



A） User Password

This is the password for the user name/ID “**user**”. The default password is “**1234**”. This user name is limited to access the Network Configuration menu.

B）Administrator Password (default: admin)

This is the password for the user name/ID “**admin”**. The default password is “**admin**”. This user name allows full access to all configuration settings available.

3.12.3 Reset Configuration

Click on the **Reset Config** tab to reset the GoIP gateway to its factory default settings.

3.12.4 Reboot the Device

Click on the **Reboot** tab to reboot the GoIP gateway. The web page is then not accessible until the device completes the reboot process.

4 Hardware Specifications

Item	Description	Remark
CPU	ARM9E 133MHz	
DSP	VPDSP101 95MHz	
RAM	8M	
FLASH	4M	
Power Supply	DC4.5V/2000mA +-10%	Input AC100V to AC240V
GSM Module	Huawei GSM Module - GSM 850 MHz/GSM 900 MHz	
	Simcom GSM Module: GSM 850 MHZ, 900MHz, DCS 1800 MHz, PCS 1900 MHz	Default: 900 / 1800 MHz
Power Consumption	The Maximum 3 W	
LED	RUN, GSM, LAN, PC	
Ethernet Port	Two RJ-45 Jacks	100/10BASE-T
Weight	105 Grams	Without AC/DC Adapter
Operating Temperature Range	0－40℃	
Operating Humudilty	40％－90％ Not Congealed	
Color	Blue	
VoIP Channel	1	
GSM Channel	1	

5 Useful Factory Default Settings

Parameter		Defaults
Ethernet	LAN	DHCP Client mode
	PC	Fixed IP：192.168.8.1
Login ID / Password	Admin mode:	admin/admin
	User mode	User/1234
Time Zone		GMT＋8