User Manual

One-Channel GSM VoIP Gateway

Model: GoIP1

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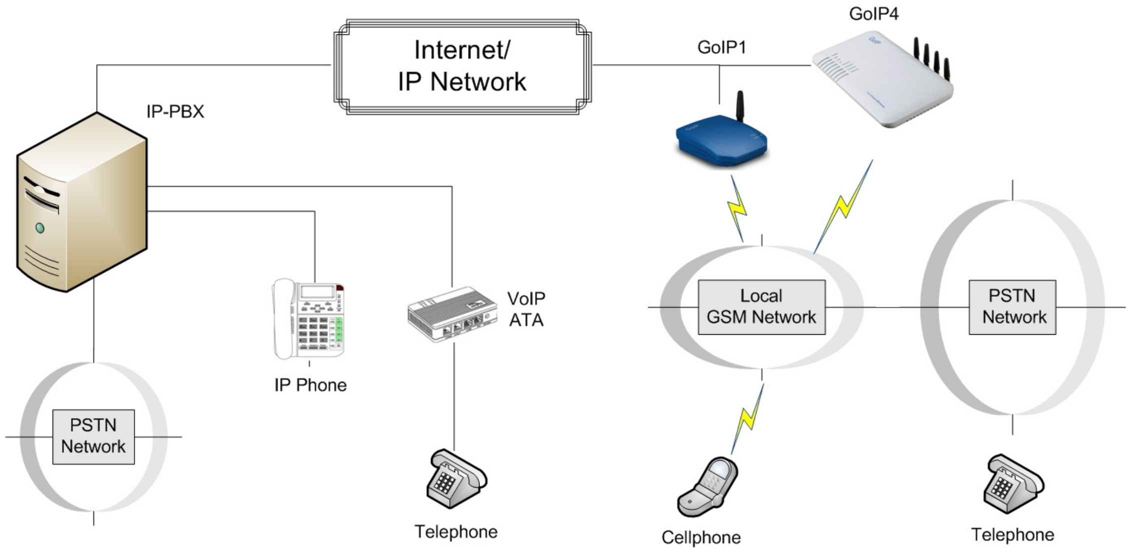
VADcore Innovative®

Contents

1 Product Introduction

1.1 General Information

A VoIP GSM Gateway enables direct routing between IP and GSM network without the use of a FXO port or the PSTN network. With this device, the usage of VoIP is greatly enhanced with significant savings on long distance and roaming charges.



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

- ARM9E 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 Channels' Connection

1.4 Software Specification

- LINUX OS
- Built-in HTTP Web Server
- PPPoE Dial-up
- NAT Broadband Router Functions
- DHCP Client
- DHCP Server
- Firmware On-line upgrade
- PSTN Caller ID transmit
- Multiple Language Support
- Supported call divert
- Supported PSTN auto call out to PSTN
- Supported Multi_devices Cooperate Mode(Group Mode)
- Supported SMS call out

1.5 List of the Package

a) One GoIP1 Gateway main unit
b) One DC12V/500mA power adaptor
c) One Ethernet cable (3M)

1.6 Appearance

VoIP GSM Gateway (GoIP1) – Front View

VoIP GSM Gateway (GoIP1) – 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 (DC12V/500mA)**

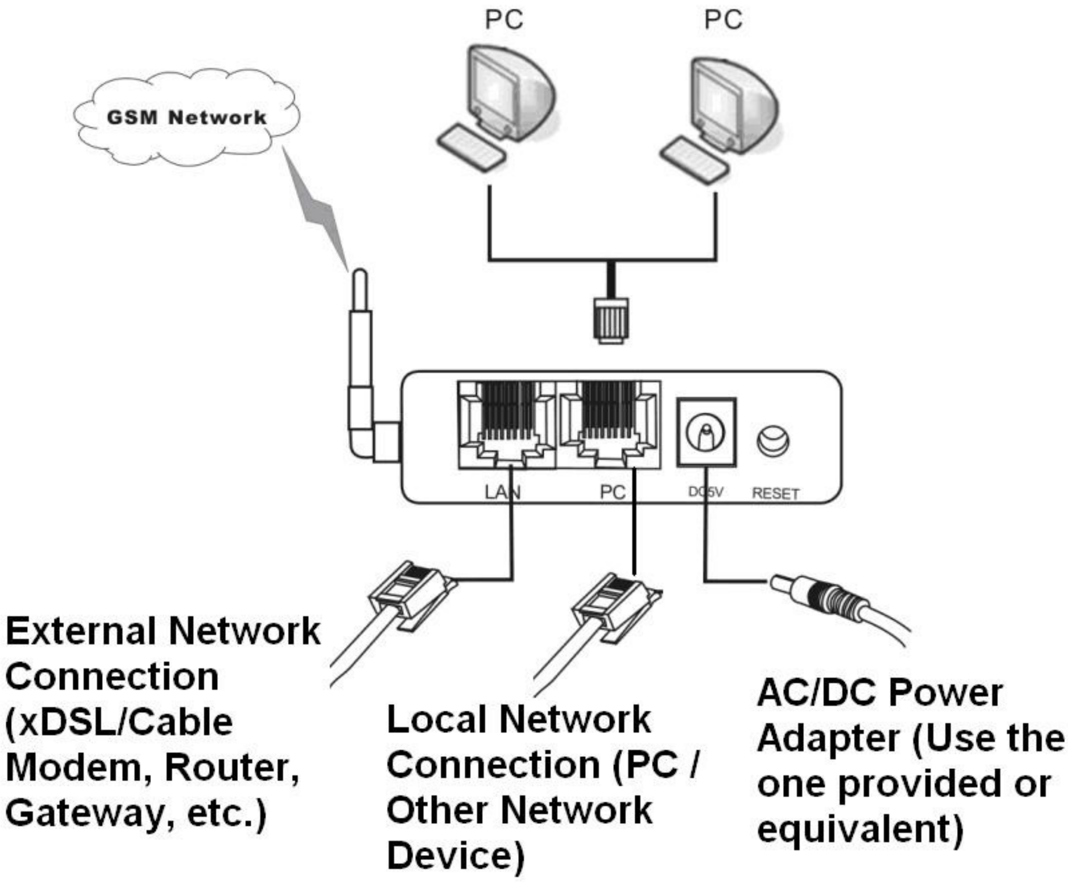
Connect the 12V/500mA Adapter provided to this power jack.

4） **Reset**

Press this button to reset the GoIP1 Gateway to factory defaults.

2 Installation

2.1 Installation Steps



Please follow the connection diagram above to install the GoIP1 Gateway.

a) Insert a GSM SIM card in the SIM card compartment located at the bottom of the GoIP1 Gateway.
b) Connect an Ethernet cable the LAN port of the GoIP1 Gateway and the other end to your existing network equipment.
c) (Optional) Connect an Ethernet cable to the PC Port of the GoIP1 Gateway and the other end to a PC or other network device.
d) Connect the power adapter provided to the power jack of the GoIP1 Gateway.

2.2 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 GoIP1 is booting, this LED will flash 100ms ON and 100ms OFF. 2. When the GoIP1 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.

2.3 SMS Commands

GoIP1 supports some maintenance commands from SMS.

FUNCTION	SMS CONTENT	REMARK
Obtain LAN Port Info	**INFO**	Not case-sensitive
Reset device	**RESET** Password	Not case-sensitive
Reboot device	**REBOOT** Password	Not case-sensitive

Note： In command **Reset** and **Reboot**, the Password is the GoIP1 device's admin password. The command keywords can be uppercase and lowercase, but the password is case-sensitive.

1》Obtain LAN Port IP Address

Once the GSM SMS with message content "info" or "INFO" is received, the GoIP1 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 GoIP1 resets its configurations to factory defaults.

3》Reboot GoIP

Upon receiving the SMS message "REBOOT <password>" or "reboot <password>", the GoIP1 reboots itself automatically.

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

3 Configuration Guide

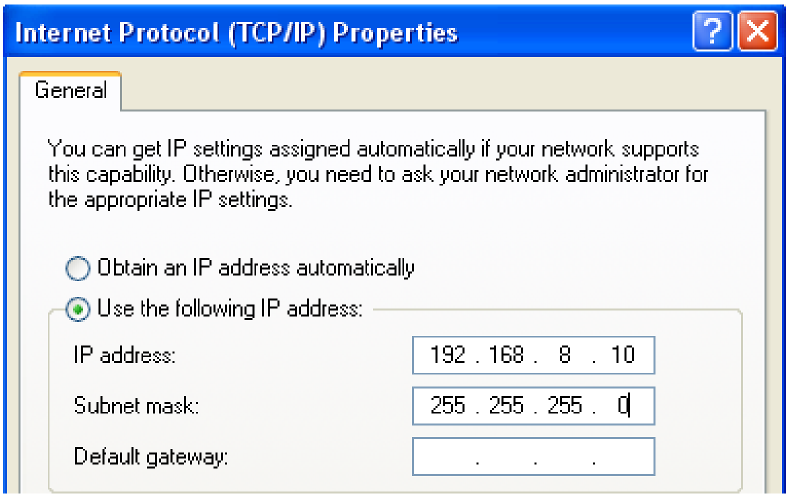
To configure the GoIP1 Gateway, you must login to its Web server via the LAN or PC port. The LAN port is factory preset to IP address 192.168.0.100 and the PC port is set to the fixed IP 192.168.8.1.

If you lose the IP address information for LAN port, just dial a call to GoIP1 Gateway's SIM card phone number. When the call is connected, you will hear a dial tone. Then dial "*01#" for English voice prompt on the LAN IP and "*00#" for Chinese voice prompt on the LAN IP. The LAN IP Address can also be obtained by sending a SMS message to the GSM phone number. The GoIP1 will then reply with a SMS message containing the LAN IP address.

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

Another way to access the GoIP1 Gateway is via the PC port. You will need to change your computer's LAN configuration via the Network Connections under the Control Panel.

Windows: **Control Panel--→Network Connections--→Local Area Connection Property--→ Internet Protocol (TCP/IP)'s Property**

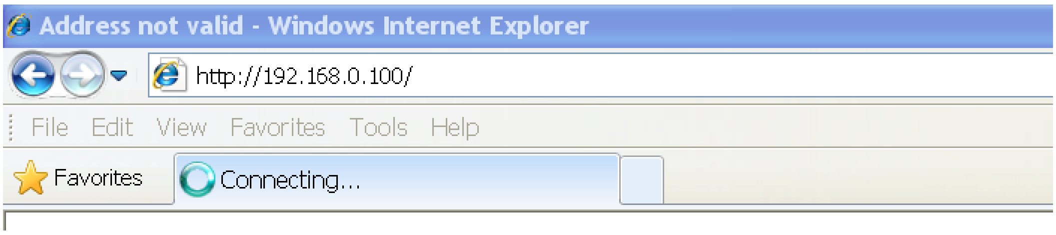


Set an unused IP address that is in the same segment as the PC port address.

Once either the IP address of the LAN or PC port is known, you are now ready to access the Web server of GoIP1 Gateway.

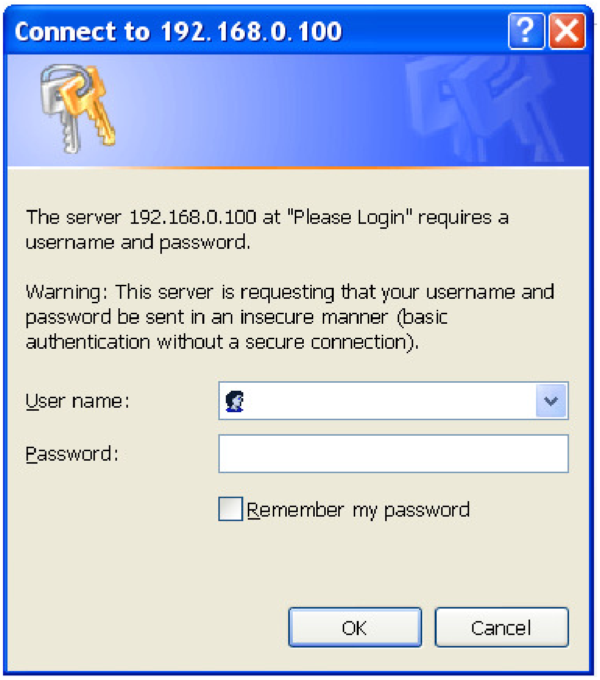
3.1 Web Configuration Menu

If your computer is connected to the GoIP1 Gateway via the LAN port, you need to type the LAN IP address of the GoIP1 Gateway in your Web Browser to access the Web server of the GoIP1 Gateway. The default IP address on the LAN port is “192.168.0.100”.



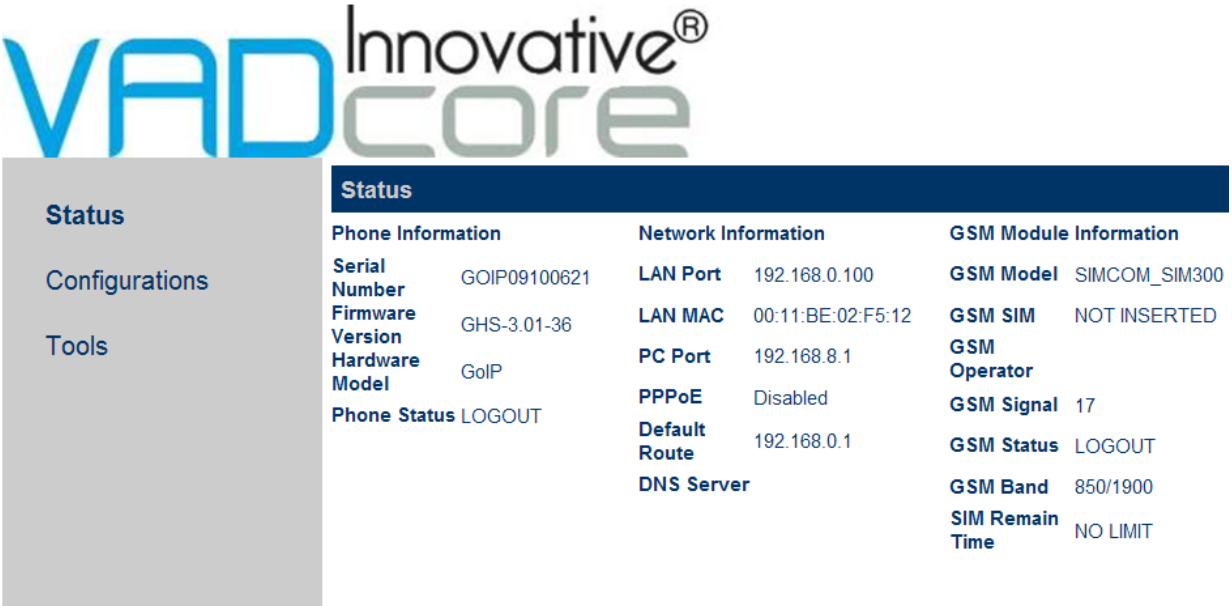
If your computer is connected to the GoIP1 Gateway via the PC port, you should type GoIP1’s PC port IP address (192.168.8.1) in the Web Browser.

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 GoIP1 Gateway Web Server. The default for both user name and password is “**admin**”.



3.2 Status

The Status page shown below is the default / home page of the GoIP1 Web server.



3.2.1 Phone Information

A. Serial Number

Each Gateway has a unique serial number assigned by the factory such as **GOIP109100019**. 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 such as **GHS-3.01-36**.

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

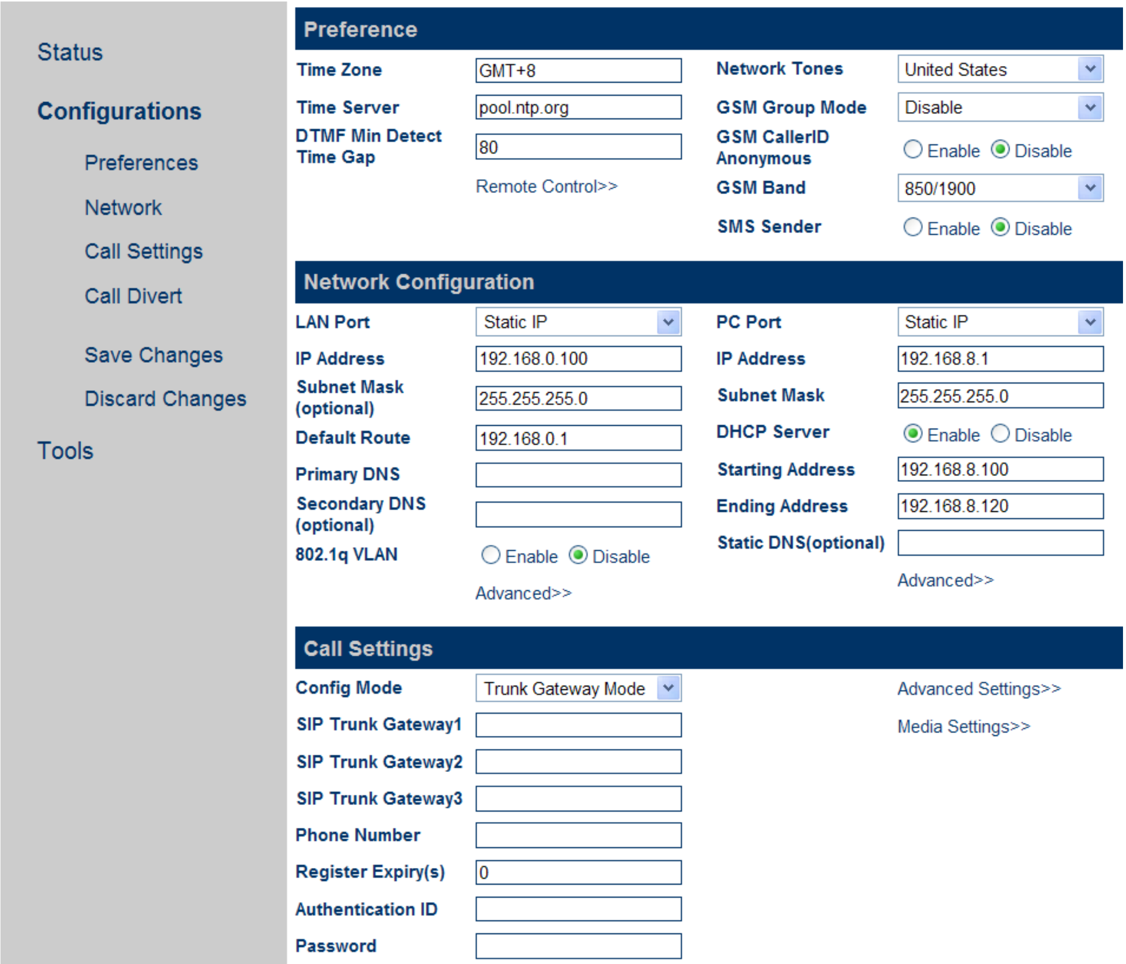
This field displays the GSM signal status. The value of GSM signal strength RSSI (Received Signal Strength Indication) is between 0 dbm and 31 dbm. The value of 99 means unknown or undetectable.

C. GSM Status

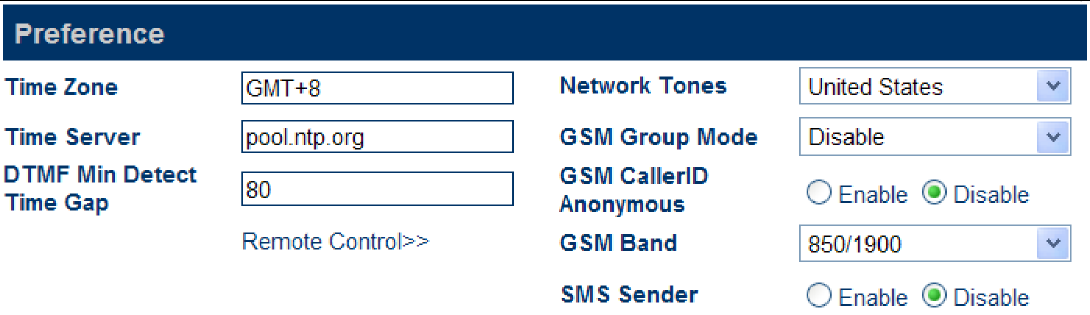
This field shows the status of GSM connection status. If the connection is successful, this field displays LOGIN; otherwise, it displays LOGOUT.

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:

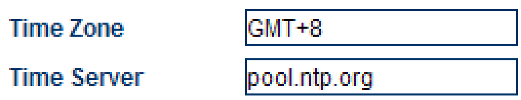


3.3.1 Language

Currently GoIP1 only supports English. VADcore also has other versions of software that support Simplified Chinese and Traditional Chinese. Contact VADcore if you need other language support.

3.3.2 Time Zone and Time Server

The GoIP1 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 GoIP1 Gateway supports CDR and Billing Information, it is important to set up these two parameters properly.

3.3.3 DTMF Min Detect Time Gap

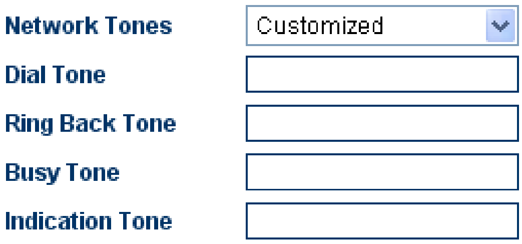
This parameter is used to limit two same DTMF digit's minimum time gap, the range is 60ms to 120ms, default is 80ms.

If you encounter double digit problem, increase this parameter. If you encounter lose digit, then decrease this parameter.

3.3.4 Network Tone

Network Tones are a set of tones used for VoIP calls. Select one of the predefined countries or select "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:

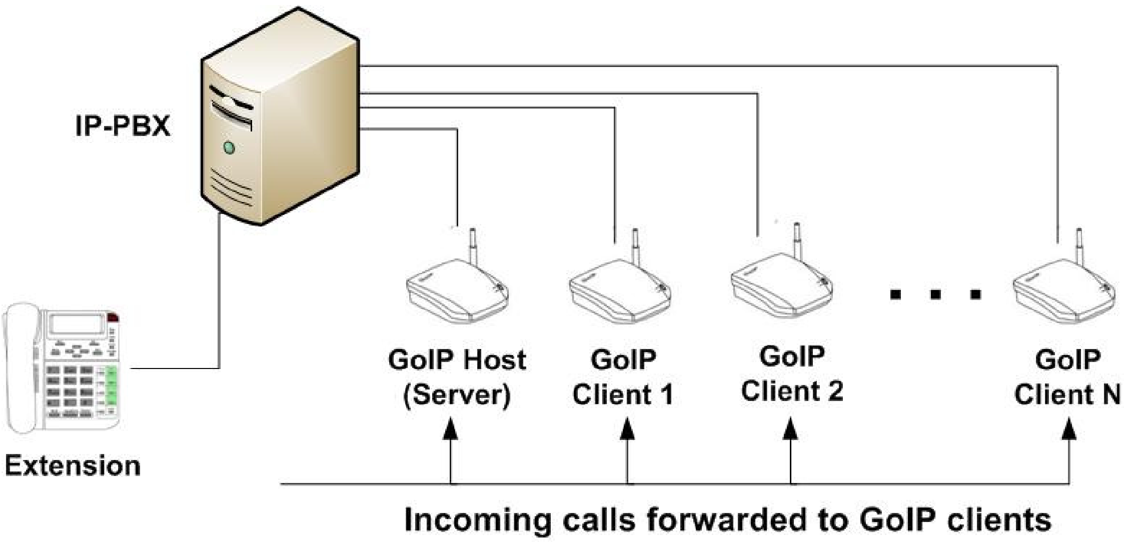
nc, rpt, c1on, c1off, c2on, c2off, c3on, c3off, f1, f2, f3, f4, p1, p2, p3, p4

Where:

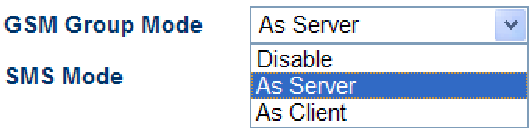
nc is the number of cadences
rpt is the repeat counter(0 - infinite, 1~n - repeat 1~n times)
c1on is the cadence one on (in milliseconds)
c1off is the cadence one off (in milliseconds)
c2on is the cadence two on (in milliseconds)
c2off is the cadence two off (in milliseconds)
c3on is the cadence three 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



GoIP1 can group multiple devices together and provide line-hunt. The Group Mode works like a multi-channels GSM gateway. Any GoIP1's channel can work as **Group Server Mode** or **Client Mode**.



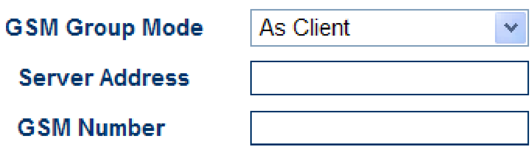
Server Mode:

Only one GSM channel runs in **Server Mode**. The GSM channel that is set in Server Mode will forward the GSM's incoming calls to other available client channels. The GSM channel that is set in Server Mode will be your main number for your customer.

Client Mode:

Other GSM channels will run in **Client Mode**. The GSM channels that are set in Client Mode will report their status to GSM channel that is set in Server Mode. The GSM channel in Server Mode then forwards phone calls to available GSM channels in Client Mode.

You must enter the GSM number for that GSM channel and IP address of the device in Server Mode into the field.

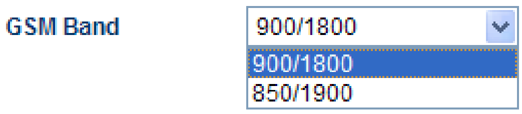


Disable: Please set all channels to **Disable Mode** if you would like to run each channel independently.

3.3.6 GSM Caller ID Anonymous

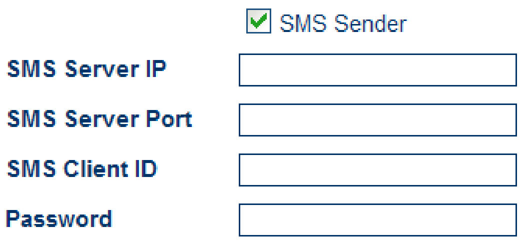
Some GSM ISPs allow the caller to disable the phone number (caller ID) when making outgoing calls. This feature must be supported by GSM ISPs.

3.3.7 GSM Band



GoIP1 Supported quad GSM bands: 850MHz, 900MHz, 1800MHz, 1900MHz. Select the correct GSM bands that are used in your country.

3.3.8 SMS Sender



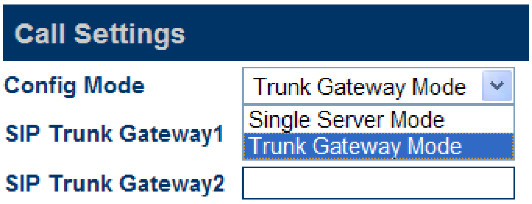
VADcore offers a software to send out SMS to GSM network through GoIP1 Gateway. A SMS server is required to work with GoIP1 Gateway for SMS Sender. Please contact VADcore for more details.

3.4 Call Settings

Click on the “**Call Settings**” to configure the VoIP call settings.

3.4.1 SIP Standard Supported

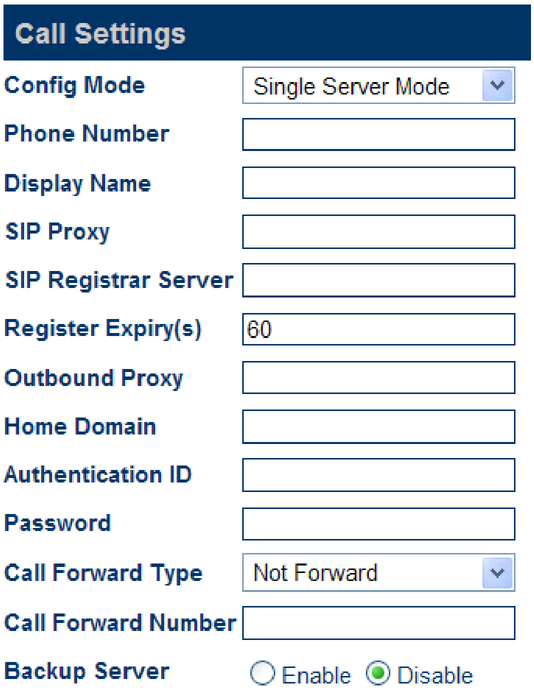
GoIP1 supports SIP standard. GoIP1 has two types of config modes for SIP protocol;



Single Server Mode: The channel uses a SIP account to connect to SIP server.

Trunk Gateway Mode: The GoIP1 will act as a SIP proxy. Remote SIP clients can register to GoIP1 and GoIP1 will process SIP requests on behalf of SIP client.

GoIP1 Gateway's SIP configure page as follow:



A) Phone Number

Enter a SIP phone number.

B) Display Name

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

C) SIP Proxy

Enter the SIP proxy IP address or domain name. If the registration port is not 5060, then add “:” and the port number. For example: **sip.hybertone.com:8080**.

D) SIP Registrar Server

Enter the SIP registrar server IP address or domain name in this field. If the registration port isn’t 5060, add “:” and the port number. For example: **sip.hybertone.com:8080**.

E) Register Expiry(s)

Enter the register time (seconds) in this field. This is the maximum length of registration that SIP server will keep your registration. If SIP server does not receive another SIP registration, the current registration will time out. Check your SIP server for a reasonable value.

F) Outbound Proxy

Outbound proxy is a device that receives requests from a client, even though it may not be the server resolved by the Request-URI. Outbound proxy will forward SIP requests and frequently RTP media traffic to another SIP server. Outbound proxy is 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 IP address can be used for accessing customers, rather than the customer’s own IP address.

Check with your SIP server (SIP provider) if an outbound proxy is required.

G) Home Domain

SIP Networks sometimes use the Home Domain name as an identifier. Enter this field if it is required.

H) Authentication ID

Enter the Authentication ID as provided.

I) Password

Enter the authentication password as provided.

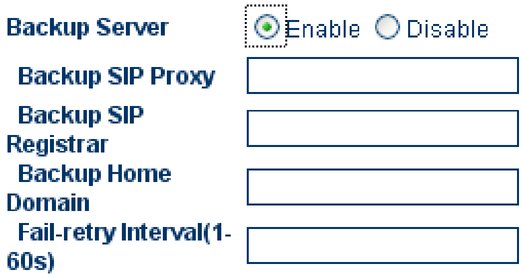
J) Call Forward Type

Call forward can be set under different conditions: Unconditional Forward, Busy Forward, No Answer Forward, Busy or No Answer Forward. Select the call forward type and enter the phone number that you would like the call to be forwarded to.

K) Call Forward Number

Enter the phone number that you would like the call to be forwarded to when Call Forward is set.

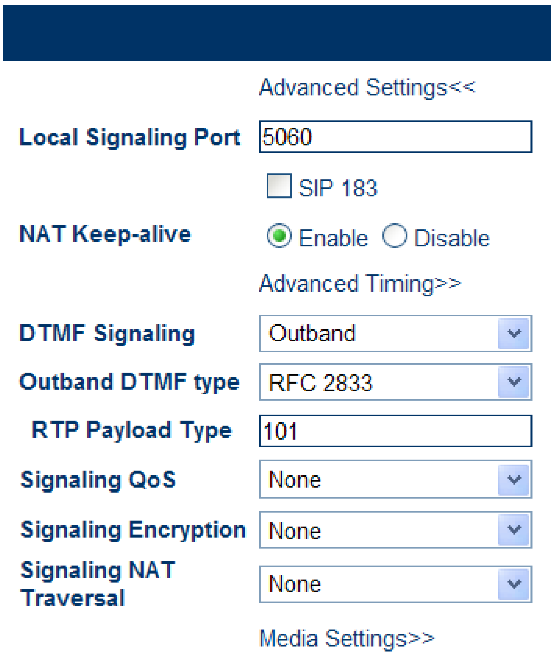
L) Backup Server



The GoIP1 Gateway supports one Backup Server as an alternative to the main server. When the registration to the main server fails, the GoIP1 Gateway will try to register to the Backup Server.

3.4.2 Advanced Settings

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



A) Local Signaling Port (SIP Local port)

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

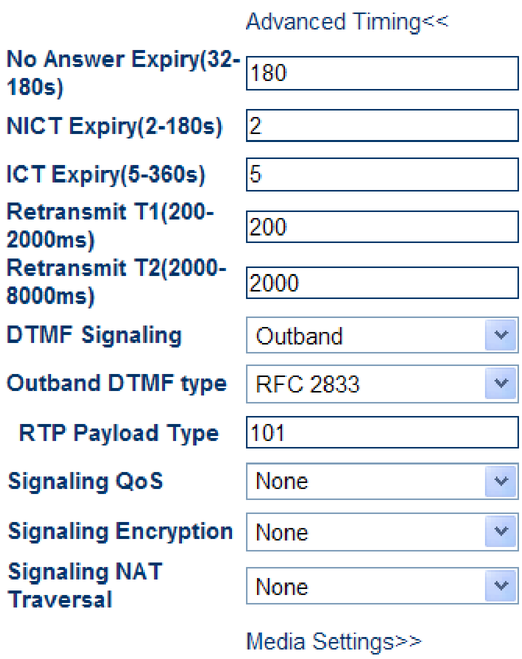
B) SIP 183

Check the box of SIP 183 if the SIP server supports this feature.

C) NAT Keep-alive

The NAT Keep-alive feature sends a null packet to the SIP Proxy periodically in order to keep the NAT open on your firewall for incoming data traffic.

3.4.3 Advanced Timing



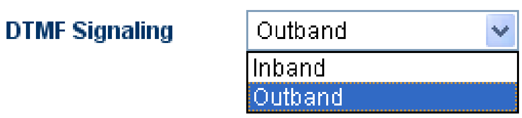
A) **No Answer Expiry(32-180s), NICT Expiry(2-180s), Retransmit T1(200-2000ms), Retransmit T2(2000-8000ms)**

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

B) DTMF Signaling

1) DTMF TYPE

DTMF signals can be sent over to the called party after a call is established. GoIP1 Gateway supports both **Inband** and **Outband** DTMF signal types.



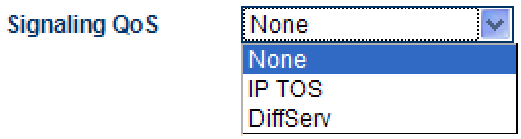
For **Inband** DTMF type, DTMF signals are generated locally at the calling phone and then send to the called party as part of the voice signals. This method is not reliable since the quality of the DTMF signals is subject to the Codec used and the quality of the networkt.

For **Outband** DTMF type, DTMF signals are independently translated and sent to the called party. After receiving DTMF signals, the called party translates and interpret based on the DTMF protocol. This method allows more reliable DTMF signaling. However, it requires the called party to support this feature in order for this to work properly. GoIP1 Gateway supports both RFC2833 and SIP INFO DTMF protocols.

2） DTMF Payload Type

DTMF Payload Type is defined by RFC2833 protocol to carry the tone definitions for various applications. The default DTMF Payload Type is 101. Please consult your VoIP service provider for the proper setting if required.

C） Signaling Qos



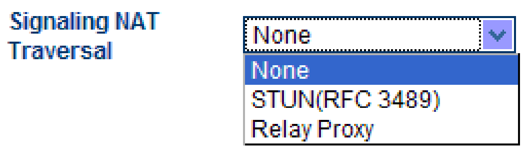
Signaling QoS improves the performance of SIP signaling. If local network device supports Qos, select this field accordingly. Please consult your network administrator for further information.

D） Signaling Encryption

GoIP1 Gateway supports different encryptions for SIP signaling. Select the one that you prefer.

E) Signaling NAT Traversal

Signaling NAT traversal may be required if the GoIP1 Gateway is put behind a NAT/firewall (or multiple NATs/firewalls). Depending on your network environment and SIP Server capabilities, this feature may or may not be turned on.



1） None

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

2） 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.

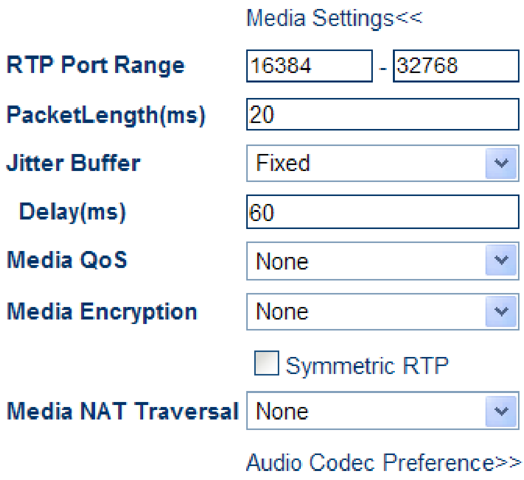
Select STUN (RFC 3489) to use a STUN server for Signaling NAT Traversal. Enter the IP Address or the domain name of the STUN server to be used.

2) Relay Proxy

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

3.4.4 Media Setting

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



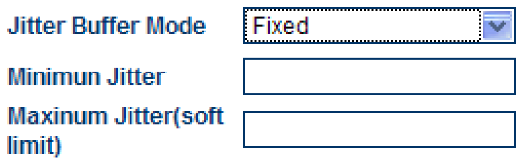
A） RTP Port Range

This parameter specifies the range of the RTP (Real Time Protocol) Ports used by the GoIP1 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(ms)

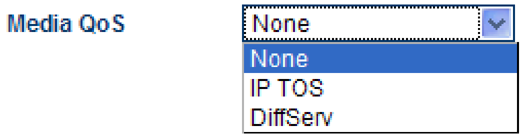
This parameter defines the voice packet length. The default setting is 20ms. The range is from 5ms to 40ms at an increment of 5ms. Please note that some codes have a minimum packet length of more than 5ms.

C） Jitter Buffer Mode



Since data packets may arrive 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 Fixed mode with the fixed delay of 60ms. 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 in intended to improve the voice performance or quality if your local network supports QoS.

E）Media Encryption

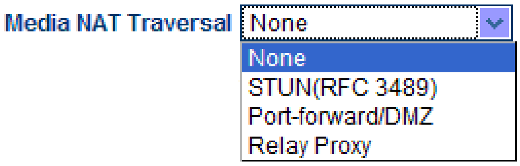
GoIP1 Gateway supports different encryptions for voice media. Select the one that you prefer.

F）Symmetric RTP

Normally GoIP1 Gateway uses RTP ports based on the configuration. If this box is checked, GoIP1 Gateway will identify RTP ports from the media traffic it has received and use the same ports when sending media traffic.

G) Media NAT Traversal

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



1）None

Select **None** to disable this feature.

2）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.

3）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 can allow an external user to reach a port on a private IP address (inside a LAN) from the outside via a NAT-enabled router.

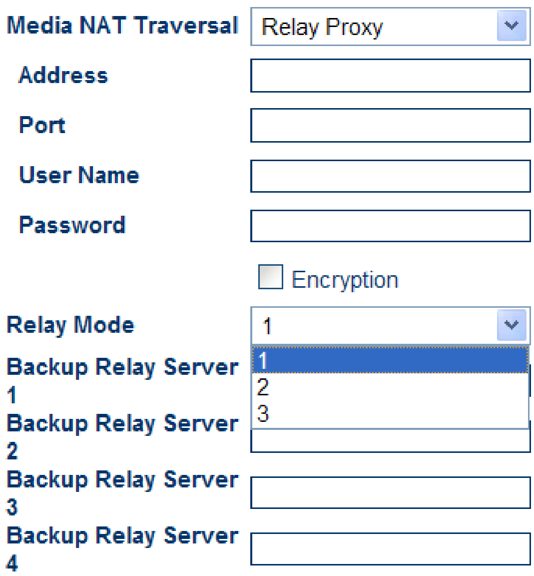
In order for this feature to work, the local network Gateway must support this feature and be set up properly. Please consult your network administrator for help to enable this Port forwarding feature.

4) **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:

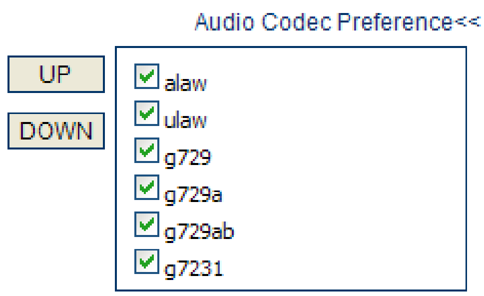
- Mode 1: The media uses UDP packets and (or) encrypt with multiple UDP port;
- Mode 2: The media uses UDP packets and (or) encrypt with single UDP port;
- Mode 3: The media uses TCP packets and (or) encrypt (UDP over TCP).



3.4.5 Codec Preference

Click on "**Media Settings**" in the "Call Setting" menu and click **Audio Codec Preference** to access the parameters.

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

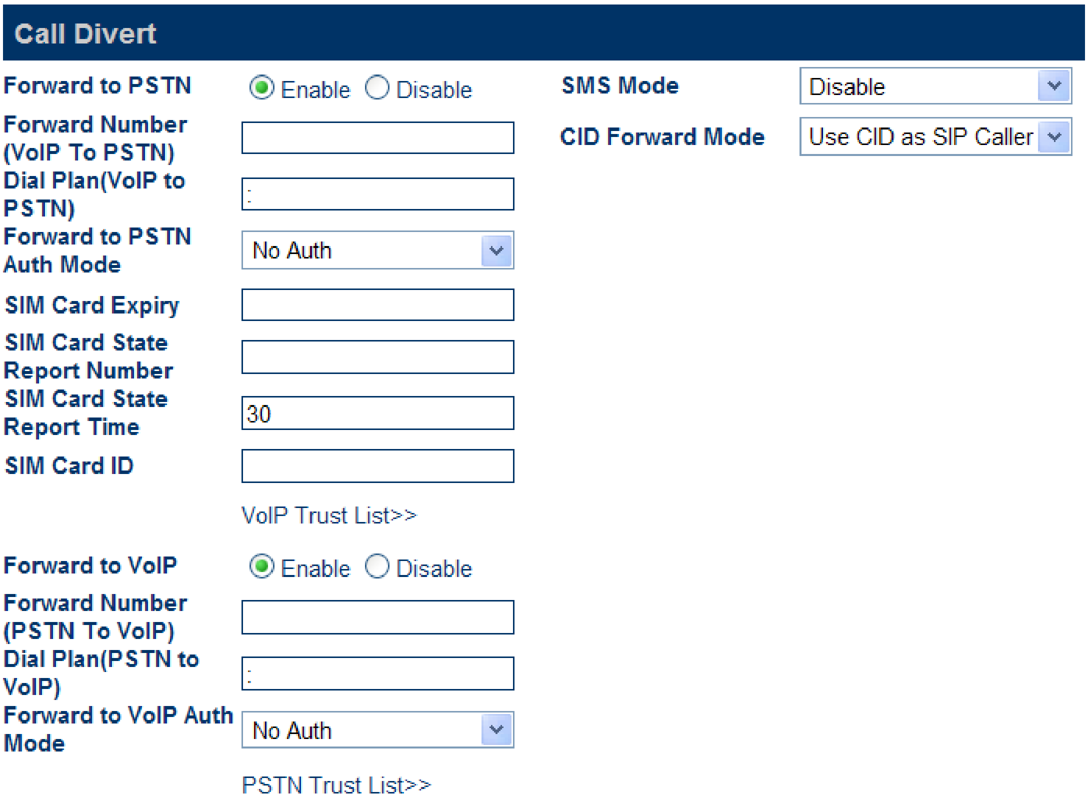


Click on the check box to enable a codec. Select a codec and then press the UP or DOWN button to move the position of the codec on the codec list with a priority in descending order.

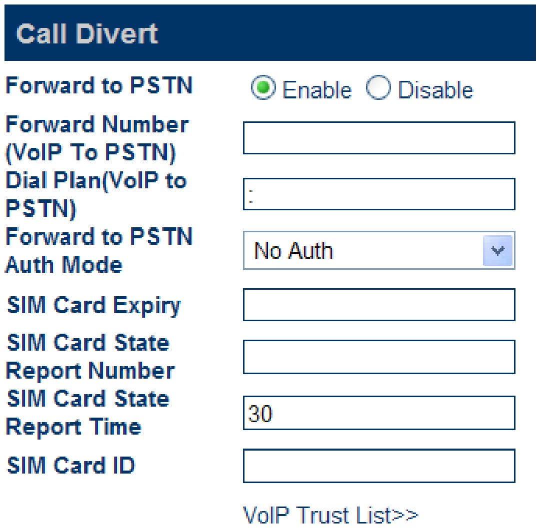
Note: The voice code alaw and ulaw is G.711a and G.711u.

3.5 Call Divert

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



3.5.1 Call Forward (From VoIP To PSTN)



Forward Number

Enter a phone number in this field will forward all incoming VoIP calls to this phone number (PSTN or Mobile). Using "," to add a 500ms delay to the dialing sequence.

When Forward Number field has a phone number, GoIP1 will automatically forward all VoIP calls to this phone number.

When Forward Number is empty, GoIP1 will route phone calls based on the following conditions.

A: When the Callee ID is GoIP1's SIP account number, GoIP1 will take the call and feed back a dial tone to VoIP caller. Then VoIP caller must dial a PSTN number when hearing this dial tone.

B: When the Callee ID is not GoIP1's SIP account number, GoIP1 will automatically dial out with this number thru GSM network, based on the rules in Dial Plan(VoIP to PSTN) field.

Dial Plan

Please refer to **3.9 Dial Plan** for details. If ":" is entered in the field, all of the phone calls will pass through.

Forward to PSTN Auth Mode

This field sets protection for using GoIP1 to connect to GSM network.

1) No Auth
   Anyone can make phone calls through GoIP1.
2) Password
   If a password is entered, the GoIP1 will generate an indication tone and wait for the caller to dial the password.
3) Trust List

VoIP Trust List<<

VoIP Trust List	
Trust Number1	
Trust Number2	
Trust Number3	
Trust Number4	
Trust Number5	
Trust Number6	
Trust Number7	

Enter the phone numbers on the Trust Number field if Trust List is used. People calling from the trust phone numbers will be able to use GSM connection.

4) Password or Trust List

Callers will be able to use GoIP1 for GSM connection if their phone numbers are on trust phone number list or if they have the password.

SIM Card Settings

SIM Card Expiry	
SIM Card State Report Number	
SIM Card State Report Time	30
SIM Card ID	

1) SIM Card Expiry - usage limit (minutes)
2) SIM Card State Report Number - the recipient phone number for the SMS report
3) SIM Card State Report Time - the time schedule to send SMS report
4) SIM Card ID - Identification sent with the sms message

3.5.2 Call Forward (From PSTN To VoIP)

Forward to VoIP	Enable Disable
Forward Number (PSTN To VoIP)	
Dial Plan(PSTN to VoIP)	:
Forward to VoIP Auth Mode	No Auth

PSTN Trust List>>

Forward Number

Enter a phone number in this field will forward all incoming PSTN (GSM) calls to this phone number (a VoIP number).

If this field is blank, the GoIP1 answers all incoming GSM calls and then generates the dial tone. The caller can then dial a VoIP number. When finishing, a pound (#) can be dialed to activate the dialing to the VoIP number immediately. If a pound (#) is not input, the VoIP number will be dialed after a preset timeout.

When

Dial Plan

Please refer to **3.9 Dial Plan** for details. If ":" is entered in the field, all of the phone calls will pass through.

Forward to VoIP Auth Mode

This field sets protection for using the GSM connection to VoIP.

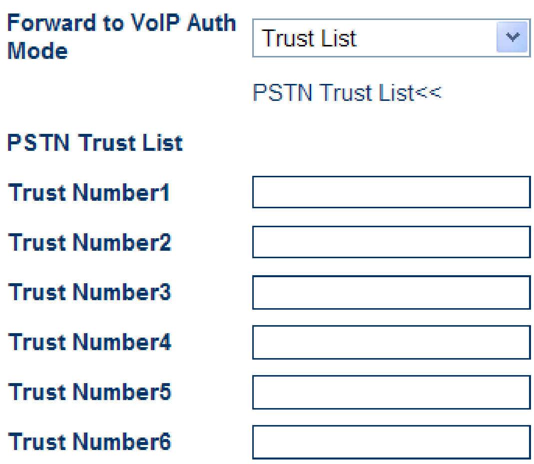
5) No Auth

   Anyone can make phone calls through GoIP1.

6) Password

   If a password is entered, the GoIP1 will generate an indication tone and wait for the caller to dial the password.

7) Trust List



   Enter the phone numbers in the Trust Number field if Trust List is used. People calling from the trust phone numbers will be able to use GoIP1 to connect to VoIP.

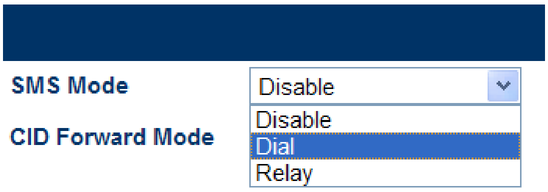
8) Password or Trust List

   Callers will be able to use GoIP1 for VoIP connection if their phone numbers are on trust phone number list or if they have the password.

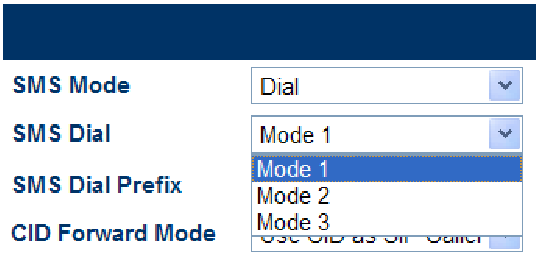
3.6 SMS Disposal

3.6.1 SMS Call Out

GoIP1 Gateway supported SMS call. In this mode, when GoIP1 Gateway receives a SMS from a mobile phone, it will automatically make a call to SIP server.
To use this function, select the **SMS Dial** option in configuration page.



GoIP1 supported three types SMS Dial：



A：Mode 1

GoIP1 dial the call use SMS sender call ID

B：Mode 2

GoIP1 dial the call via its VoIP account and add the SMS sender phone number to Call Divert option's Forward Number (VoIP to PSTN) automatically.

C：Mode 3

GoIP1 dial the call via its VoIP account and add the SMS sender phone number to SIP invites header.

D：SMS Dial Prefix

When GoIP1 dial a SMS call, it will automatically add this option's digit in be Called ID.

- **Mode 1 examples:**

A. GoIP1 use SMS Dial Mode 1:

PSTN Trust List>>

GSM Group Mode	Disable
SMS Mode	Dial
SMS Dial	Mode 1
SMS Dial Prefix	

A mobile phone's number is (86)13800000000, it sends a SMS "8675588228822" to GoIP1's GSM SIM card. When GoIP1 device receives this SMS, it will automatically call the number 8675588228822, and the caller is number 8613800000000.

The sent-out signaling as follow:

```
Sending Message to 192.168.2.1:5060:
INVITE sip: 8675588228822@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.189:5060;rport;branch=z9hG4bK1686911003
From: <sip: 8613800000000@192.168.2.1:5060>;user=phone;tag=626918067
To: <sip: 8675588228822@192.168.2.1>
Call-ID: 1835068843@192.168.2.189:5060
CSeq: 2 INVITE
Contact: <sip: 8613800000000@192.168.2.189:5060>
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. GoIP1 use SMS Dial Mode 1 and add a prefix as 999:

SMS Mode	Dial
SMS Dial	Mode 1
SMS Dial Prefix	999

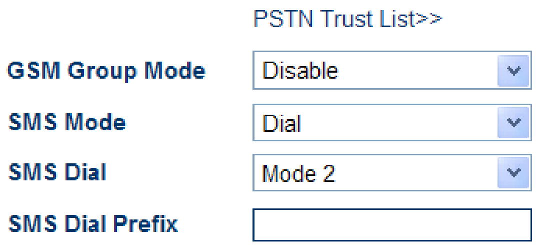
A mobile phone's number is (86)13800000000, it sends a SMS "8675588228822" to GoIP1's GSM SIM card. When GoIP1 device receives this SMS, it will automatically call the number 9998675588228822, and the caller is number 8613800000000.

The sent-out signaling as follow:

```
Sending Message to 192.168.2.1:5060:
INVITE sip: 9998675588228822@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.189:5060;rport;branch=z9hG4bK1686911003
From: <sip: 8613800000000@192.168.2.1:5060>;user=phone;tag=626918067
To: <sip: 9998675588228822@192.168.2.1>
Call-ID: 1835068843@192.168.2.189:5060
CSeq: 2 INVITE
Contact: <sip: 8613800000000@192.168.2.189:5060>
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
```

- **Mode 2 example:**

GoIP1 use SMS Dial Mode 2:



A mobile phone's number is (86)13800000000, it sends a SMS "8675588228822" to GoIP1's GSM SIM card. When GoIP1 device receives this SMS, it will automatically call the number 8675588228822, and the caller number is GoIP1's SIP account number.

GoIP1 will set the SMS sender number to "**Call Divert** "option's "**Forward Number (VoIP to PSTN**" automatically. The result is, when SIP server receives the SMS call and call back to GoIP1, GoIP1 will automatically call the SMS sender via GSM network.

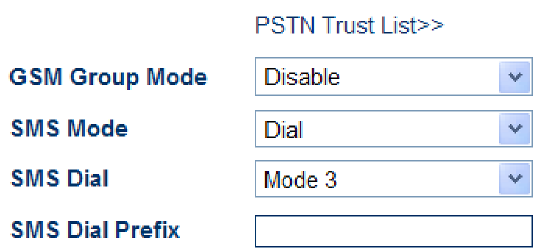
The sent-out signaling as follow:

```
Sending Message to 192.168.2.1:5060:
INVITE sip: 8675588228822@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.189:5060;rport;branch=z9hG4bK92531725
From: <sip:20001@192.168.2.1:5060>;user=phone;tag=740569827
To: <sip: 8675588228822@192.168.2.1>
Call-ID: 464713443@192.168.2.189:5060
CSeq: 3 INVITE
Contact: <sip:20001@192.168.2.189:5060>
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
```

SMS prefix can be used in mode 2 just like in mode 1.

- Mode 3 example:

GoIP1 use SMS Dial Mode 3:



A mobile phone's number is (86)13800000000, it sends a SMS "8675588228822" to GoIP1's GSM SIM card. When GoIP1 device receives this SMS, it will automatically call the number 8675588228822*(86)13800000000, and the caller number is GoIP1's SIP account number.

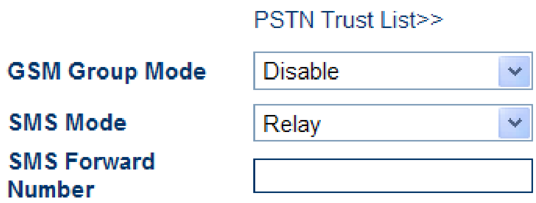
The sent-out signaling as follow:

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 192.168.2.180:5060;branch=z9hG4bK620642232
From: <sip:20001@192.168.2.1:5060>;user=phone;tag=1333994780
To: <sip: 8675588228822*8613800000000@192.168.2.1>
Call-ID: 52754291@192.168.2.180
CSeq: 2 INVITE
Contact: <sip:20001@192.168.2.180:5060>
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

SMS prefix can be used in mode 3 just like in mode 1.

3.6.2 SMS Relay

GoIP1 GSM Gateway supports SMS relay.



The **SMS Forward Number** is the receiver (ex. an extension) on your VoIP system. GoIP1 will forward the SMS it received to the number.

3.6.2.1 SMS Relay To VoIP System

When GoIP1 receives a SMS from GSM network, it will relay to VoIP system's appointed number (SMS Forward Number).
Assume the SMS Forward Number is 3999 and SMS sender number is "8613682626865 ", the SMS content is "075583185700 ". The GoIP1 will send a message to your VoIP system as below:

```
MESSAGE sip:3999@192.168.2.1 SIP/2.0
Via: SIP/2.0/UDP 192.168.2.162:5060;branch=z9hG4bK1967685528
From: <sip:1638@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:1638@192.168.2.162:5060>
Max-Forwards: 30
User-Agent: HyberTone
Content-Type: text/plain
Content-Length:     28

8613682626865
075583185700
```

3.6.2.2 SMS Relay To GSM Network

When GoIP1 receives a message from SIP server as below:

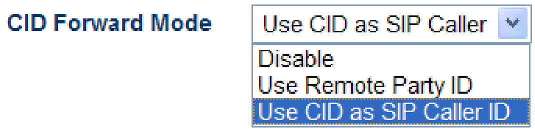
```
MESSAGE sip:1638@192.168.2.162:5060 SIP/2.0
From: <sip:3999@192.168.2.89>;tag=5031
To: <sip:1638@192.168.2.1>
Call-ID: 808807EB-A8B3-DD11-BBA6-005056C00008@192.168.2.89
CSeq: 3 MESSAGE
Contact: <sip:3999@192.168.2.89>
max-forwards: 16
date: Tue, 18 Nov 2008 06:36:37 GMT
user-agent: SIPPER for 3CX Phone
p-hint: usrloc applied
Content-Type: text/plain
Content-Length:     26

13682626800
Hello world
```

The GoIP1 will send a SMS to GSM number 13682626800, the SMS content is “Hello world”.

3.7 GSM Caller ID Transparent

GoIP1 supports GSM Caller ID transparent to VoIP via SIP Invite signaling.



A) **Disable:** Disable GSM Caller ID transparent to VoIP.

B) **Use Remote Party ID:** GoIP1 will add Caller ID in SIP invite's Remote Party ID option.

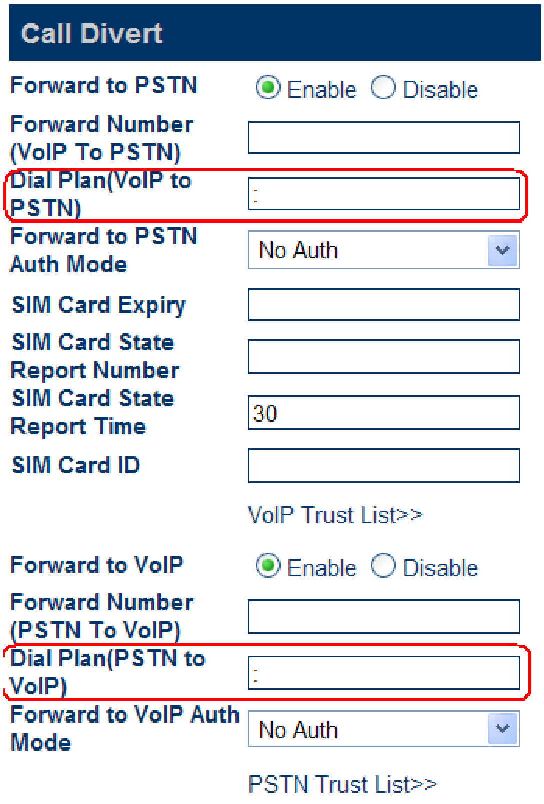
```
Sending Message to 192.168.2.1:5060:
INVITE sip:5000@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.180:5060;branch=z9hG4bK1645487913
From: <sip:20001@192.168.2.1:5060>;user=phone;tag=406202416
To: <sip:5000@192.168.2.1>
Call-ID: 847230278@192.168.2.180
CSeq: 2 INVITE
Contact: <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
```

C) **Use CID as SIP Caller ID:** GoIP1 use PSTN Caller ID in SIP invitee's Caller ID option and Remote Party ID option.

```
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
```

3.8 Dial Plan

Dial Plan defines how a number is processed when GoIP1 receives it. This field is located in the **Call Divert** Window. The Dial Plan is very flexible and can be configured for a wide range of dialing applications.



The basic syntax is "<**event**>:<**action**>|<**event**>:<**action**>|…", where

<**event**> defines the event to be matched. An event consists of a sequence of digits. If a specific digit has a limited range, use the syntax [A-B] where A and B are both digit (0 to 9) and B is greater than A. The length of the input number can be limited by using "X" to represent each unknown digit. If this field is omitted, it means any event.

<**action**> defines the action to be taken when a phone number is received. It consists of "-" (minus), "+" (plus), and digits. "-" followed by digits means to remove the digits from the beginning of the number. "+" followed by digits means to add the digits in front of the number.
"|" means "or" and the order of priority is from left to right.

Note: For practical use, there should be no limitation on the length of dial plan string.

Examples:

1. Dial Plan = "010:-010" means that the first 3 digits "010" of dialed number will be removed if the first 3 digits of dialed number are "010"..
   a) Number entered = "01082121234", actual number dialed = "82121234".

b) Number entered = “82121234”, actual number dialed = “82121234”.

2. Dial Plan = “1:+00” means that two digits “00” will be added in front of the number when the first digit of the dialed number is “1”.
   a) Number entered = “1082121234”, actual number dialed = “00182121234”.
   b) Number entered = “82121234”, actual number dialed = “82121234”.

3. Dial Plan = “001:-001+1751” means that the first 3 digits “001” of the dialed number will be changed to “1751” when a number with first three digits “001” is entered.
   a) Number entered = “00182121234”, actual number dialed = “175282121234”.
   b) Number entered = “82121234”, actual number dialed = “82121234”.

4. Dial Plan = “XXXX:” means that the input number is limited to 4-digit long and will be dialed out immediately when the fourth digit is entered.

5. Dial Plan = “13XXXXXXXXX:+0” means that the input number is restricted to 11-digit long and the first two digits must be “13”. When this condition is matched, the digit “0” will be added to the front of the number and then dialed out.
   a) Number entered = “13901234567”, actual number dialed = “013901234567”.
   b) Number entered = “12801234567”, actual number dialed = “12801234567”.

6. Dial Plan = “13[6-9]XXXXXXXX:+0” means that the input number is restricted to 11-digit long, the first two digits must be “13” and the third digit can be 6, 7, 8,or 9. When this condition is matched, the digit “0” will be added to the front of the number and then dialed out.
   a) Number entered = “13901234567”, actual number dialed = “013971234567”.
   b) Number entered = “13001234567”, actual number dialed = “13001234567”.

Please note that the above samples are intended to show the meaning of various rules. They may not have any practical meaning. A combination of these rules (joined with the symbol “|”) can be realized for a much more complicated dialing application.

3.9 Gain Settings

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

http://xxx.xxx.xxx.xxx/vadcore/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 gain MAY affect the sensitivity of DTMF detections.

Gain Settings	
Line 1	
Line 1 Output Gain	0
Line 1 Input Gain	+2
Save	Reset

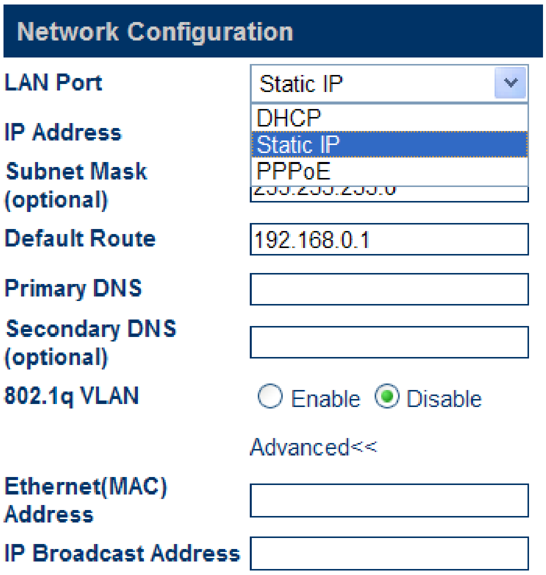
3.10 Network Configuration

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

Network Configuration			
LAN Port	Static IP	**PC Port**	Static IP
IP Address	192.168.0.100	**IP Address**	192.168.8.1
Subnet Mask (optional)	255.255.255.0	**Subnet Mask**	255.255.255.0
Default Route	192.168.0.1	**DHCP Server**	◉ Enable ○ Disable
Primary DNS		**Starting Address**	192.168.8.100
Secondary DNS (optional)		**Ending Address**	192.168.8.120
802.1q VLAN	○ Enable ◉ Disable	**Static DNS(optional)**	
	Advanced>>		Advanced>>

3.10.1 LAN Port

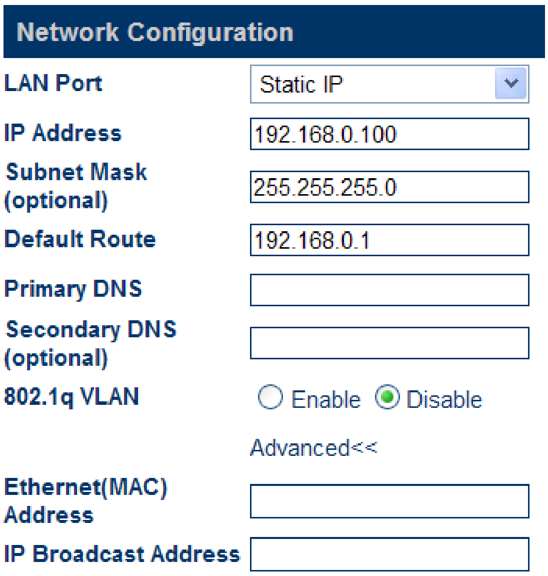
Three LAN Port modes are supported: **DHCP**, **Static IP** and **PPPoE**. The default is set for Static IP with default IP address "192.168.0.100".



1）DHCP

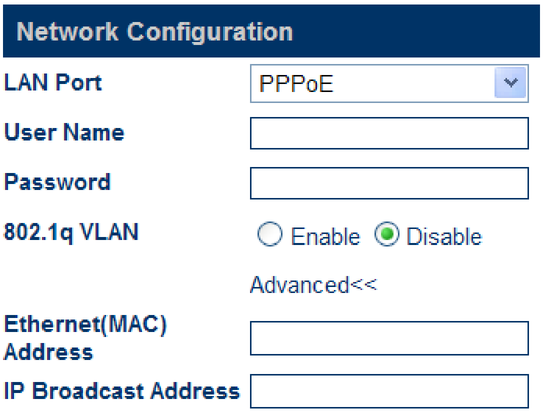
Choose **DHCP** if a local DHCP host is available. This allows the GoIP1 Gateway to obtain network information (IP Address, Subnet Mask, Default Route, Primary DNS, Secondary DNS, and other DHCP options) from the DHCP host.

2）Static IP



The default setting of GoIP1 is **Static IP** with IP address “192.168.0.100” and **Subnet Mask** “255.255.255.0”. However, **Default Route**, **Primary DNS**, and **Secondary DNS** (optional) must be manually entered according to your network configuration.

3）PPPoE



PPPoE is a common 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

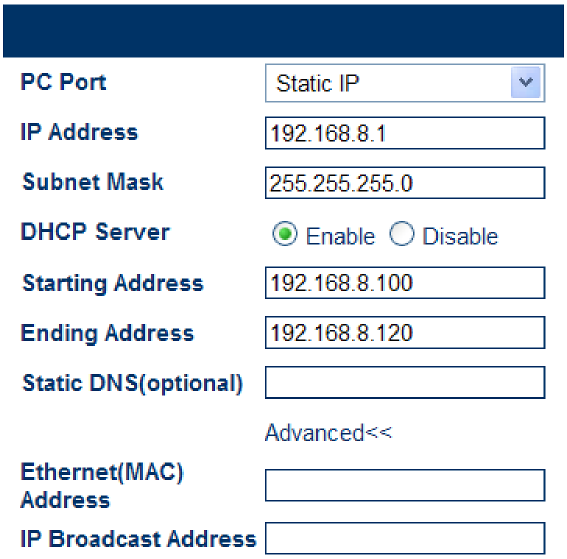
This QoS feature requires QoS support of your network to improve voice traffic. Please consult your network administrator for proper settings.

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.

3.10.2 PC Port Configurations

The PC Port allows other network devices to be attached to the GoIP1 Gateway. It offers both Bridge and Static IP modes to meet your requirements. The factory preset is Static IP mode with the IP address 192.168.8.1.

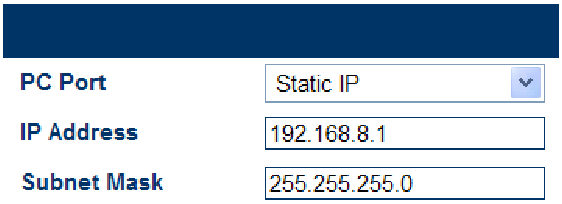


1）**Bridge Mode**

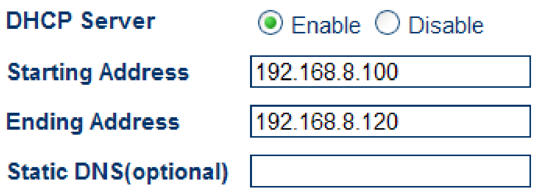
Select **Bridge** mode if your network topology requires the network devices (PC or others) to be in the same network segment as the GoIP1 Gateway. In this case, the GoIP1 Gateway functions as an Ethernet Switch.

2）**Static IP Mode (Default Setting)**

Select **Static IP** mode for a new network segment on PC port. In this case, the GoIP1 Gateway functions as Router. Enter the IP address in **IP Address** field with a new segment address that is different from that on the LAN port. Enter the subnet mask in **Subnet Mask** field accordingly. A commonly used value is 255.255.255.0.



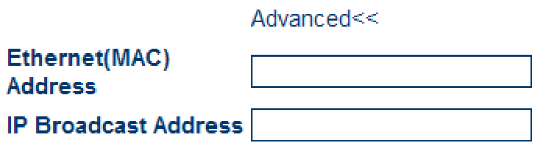
Enable the **DHCP Server** if you want the GoIP1 Gateway functions as a local DHCP host on PC port. This will enables the GoIP1 Gateway to assign IP Addresses to network devices that are attached to the PC port.



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.



3.11 Save Configuration

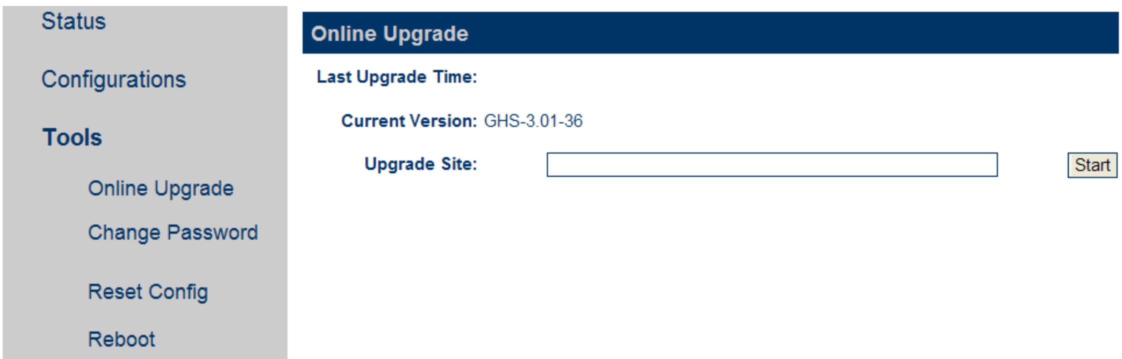
To confirm and commit all changes that have been made, click on the **Save Changes** tab. Otherwise, all changes will be discarded.

3.12 Discard Changes

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

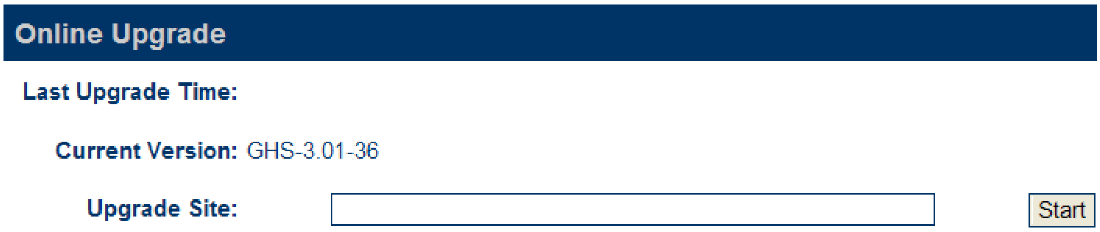
3.13 Tools Menu

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



3.13.1 Online Upgrade

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



Enter the update link as provided by VADcore. A sample link is:

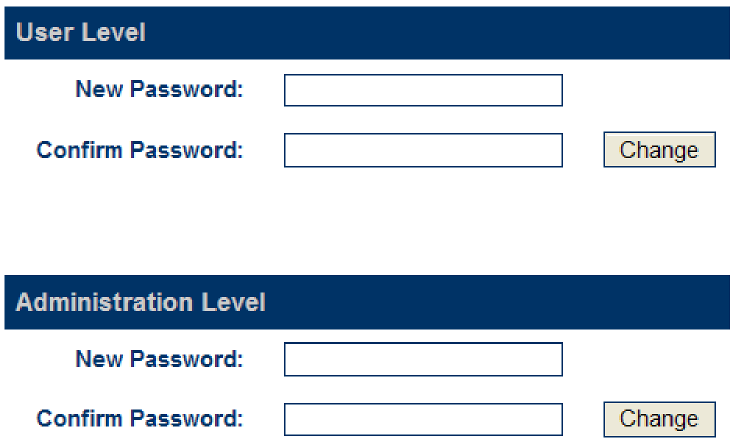
hk.ippcn.com/update/GHS-3.01-18.pkg

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

WARNING: POWER SHUTDOWN, POWER FAILURE OR UNPLUG POWER ADAPTOR FROM GoIP1 DURING FIRMWARE UPGRADE MAY PERMINENTLY DAMAGE THE GOIP1 GATEWAY AND VOID THE WARRANTY.

3.13.2 Change Password

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



A) **User Password**

This is the password for the user ID "**user**". The default password is "**1234**". This user ID has limited access to the Network Configuration menu.

B) **Administrator Password (default: admin)**

This is the password for the user ID "**admin**". The default password is "**admin**". This user ID has full access to all configuration settings available.

3.13.3 Reset Configuration

Click on the **Reset Config** tab to reset the GoIP1 Gateway to its factory default settings.

3.13.4 Reboot the Device

Click on the **Reboot** tab to reboot the GoIP1 Gateway.

4 Hardware Specifications

Characteristics of the hardware	Parameter	Remarks
Processor	ARM9E 133MHz	
DSP	VPDSP101 95MHz	
RAM	8M	
Flash	4M	
Power	DC12V/500mA +-10%	Input AC100V to AC240V
GSM Module Type	Default 850MHz/1900MHz	
	Optional 900MHz/1800MHz	
Consumption	The Maximum 3 W	
LEDs	RUN, GSM, LAN, PC,GSM	
Network Ports	2 RJ45; Supported NAT	100/10BASE-T
Weight	450 Grams	With AC/DC Adapter
Working Temperature	0—40℃	
Working Humidity	40％—90％ Not Congealed	
Colour	Blue	
GSM SIM Ports	1	
VoIP Channels	1	

5 Manufactory Parameters

Parameters		Default Setting
Network	LAN	192.168.0.100
	PC	Static IP:192.168.8.1 DHCP Server Running
Password	admin	admin
	user	1234
Time Zone		GMT＋8