GSM VOIP
Gateway Series

User Manual

1. OVERVIEW

1.1 Introduction

The GSM VOIP Gateway is a broadband relay gateway newly developed by VinTelecom Technology. It is a new product for seamless connection between the GSM network and VoIP network. When the mobile phone SIM card is installed in the GSM VOIP Gateway, users can register the GSM phone to the VoIP softswitch system. Through the GSM VOIP Gateway, users can realize the uplink and downlink call s between the GSM network and the VoIP network. In addition, the GSM VOIP Gateway supports the transparent transmission of the call er number from the PSTN to the VoIP. The
GSM VOIP Gateway features embedded SIP and H.323 protocols with flexible setting. The bi-directional password authentication (call authorization) and trust list authentication greatly minimize the risk of charge losses and the flexible routing function can meet special requirements of various call forwarding. In particular, the GSM VOIP Gateway supports multi device groups, with flexible setting of large GSM gateway groups with different channel numbers. With its low price, excell ent voice quali ty, and powerful features, the GSM VOIP Gateway is the first choice for system integrators, traffic operators, and softswit ch manufacturers.

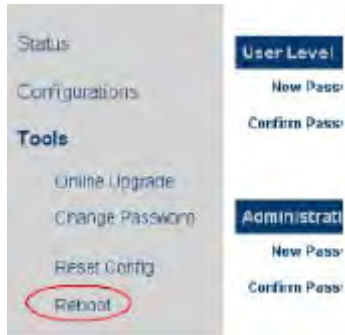The GSM VOIP Gateway includes WT-2201, WT-2204 and WT-2208.

Color	Gray	Gray	
VoIP Channel	1	4	
GSM Channel	1	4	

Hardware	Parameter	Remark
Model	WT-2208	Customized
Processor	ARM9E 133Mhz	
DSP	VPDSP101 95Mhz	2 pieces
RAM	16M	
FLASH	4M	
Power	DC12V/3A±10%	Input 100V to 240V
GSM Band	Default 900M/1800M	Default
	Optional 850/1900M	Customized
Power Consumption	Max. 20W	
LED	RUN, GSM, LAN, PC	
Network Adapter	2	100/10BASE-T
Weight	1.204Kg	Without AC Adapter
Operating Temperature	0-40°C	
Operating Humidity	40%-90% Not Congealed	
Color	Gray	
VoIP Channel	8	
GSM Channel	8	

5 Factory Default Parameter Table

Parameter		Default Value
Network	LAN	Dynamic IP (DHCP)
	PC	Fixed IP 192.168.8.1
Password	Admin	Admin
	user	1234
Default Time Zone		GMT+8

Select Tool > Reset to restart the GSM VOIP Gateway.

4 Parameters Of Equipment

Hardware Feature	Parameter	Parameter	Remark
Model	WT-2201	WT-2204	WT-2208
Processor	ARM9E 133Mhz	ARM9E 133Mhz	
DSP	VPDSP101 95Mhz	VPDSP101 95Mhz	2pcs
RAM	16M	16M	
FLASH	4M	4M	
Power	DC12V/2A ±10%	DC12V/2A ±10%	Input 100V to 240V
GSM Band	Default 900M/1800M	Default 900M/1800M	Default
	Optional 850/1900M	Optional 850M/1900M	Customized
Power Consumption	Max. 5W	Max. 12W	
LED	RUN, GSM, LAN, PC	RUN, GSM, LAN, PC	
Network Adapter	2	2	100/10BASE-T
Weight	0.10KG	0.45KG	Without
Operating Temperature	0-40°C	0-40°C	
Operating Humidity	40%-90% Not Congealed	40%-90% Not Congealed	

1.2 Protocols

- TCP/ IP V4 (IPV6 Automatic Adaptive)
- 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, G.723.1 and GSM voice coding
- RFC1889 real-time digital transmission protocol
- Firewall penetration technology
- SIP V2.0 standard
- STUN
- Network management protocol (NMP)
- PPPoE
- PPP authentication protocol (PAP)
- Internet control message protocol (ICMP)
- TFTP agent protocol
- Hypertext transfer protocol (HTTP)
- Dynamic host configuration protocol (DHCP)
- Domain name system (DNS)
- User account authentication (via MD5)
- Out-band DTMF relay: RFC 2833 and SIP INFO

1.3 Hardware Feature

- ARM9E high-speed processor
- Voice coding and voice digital signal processor
- Two 10/100MB Ethernet ports that support the IEEE 802.3 standard and connect the LAN and
- PC
- LED that dis plays the status of Ethernet ports
- Ethernet cable
- SIM card that supports the GSM 900M/1800M and GSM 850M/1900M bands

1.4   Software Feature

- LINUX OS
- Embedded HTTP that accesses internal parameters
- PPPoE diali ng
- NAT broadband routing function
- DHCP cli ent
- DHCP server
- Software online upgrade
- Automatic call ing
- Supporting multiple languages
- Supporting outgoing SMS calls

1.5   Product Package List

1) A GSM VOIP Gateway
2) 12V/2A DC(For WT-2204) or 12V/3A DC (For WT-2208) transformer
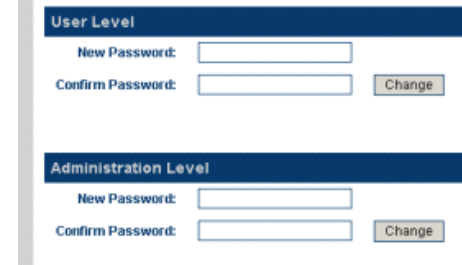3) An Ethernet cable (2 m)

1.6   Product Appearance

Note: During the upgrading, do not cut off the power. Otherwise, the GSM VOIP Gateway will be damaged.

3.13.2   Modification of Password

You can modify the password of the user and administrator. Select Tool > Modify Password. The password modification page is displayed, as shown in the following figure. Enter a new password and click "Change". Then, the password is successfully modified.

Note: The password modified by users wil l be cleared and restored to the factory default password after the factory settings are restored.

3.13.3   Restore Factory Settings

Select Tool > Restore Factory Settings. The following prompt is displayed.

Click OK. All the parameters of the gateway will be cleared and the gateway will automatically restart. After the gateway is restarted, all the settings restore to the factory default settings. This feature can be completed by using the asterisk command. For details about this operation, see the section of "Asterisk Instructions".

3.13.4   Reset

Note: Some of parameters of the gateway wil l not be valid until the gateway is restarted. Therefore, you are advised to restart the gateway after the parameters are modified, so that the modification can take effect.

3.12   Abandon the change

When the new setting is not saved, you can clear all the unsaved parameters.

3.13   Tool

Select Menu > Tool. The following page is displayed.
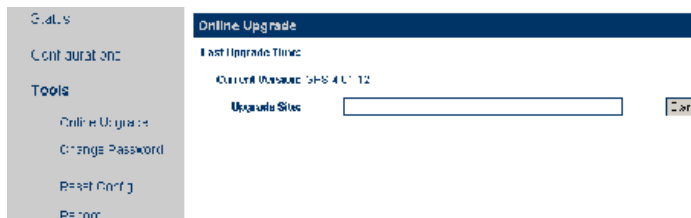
3.13.1   Online Upgrading

Warning! Only experienced users and administrators can implement the online upgrading.

Select Tool > Online Upgrading. The online upgrading page is displayed, as shown in the following figure. Enter the complete name and path of the upgrade package, such as http://118.142.51.162/update/GHS-4.01-12.pkg, and then click "Start". The gateway begins to upgrade. After the GSM VOIP Gateway is successfully upgraded, the gateway will automatically restart.
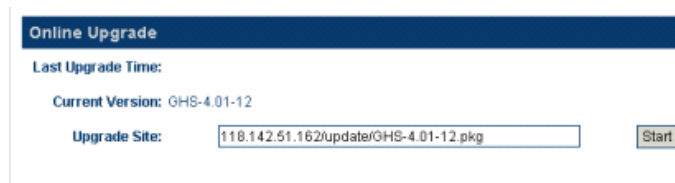
1) **LAN -** The network input port that is connected to the router, Modem, and switch
2) **PC** - The network output port that is connected to network sharing equipment (less than 100 terminals)
3) **DC4.5V/2ADC** - The output terminal that connects the transformer equipped with delivery
4) **Reset -** The reset switch for quick restart of the GSM VOIP Gateway

2   Installation

2.1   Installation Procedure

The GSM VOIP Gateway has 1/8 SIM card slots, an LAN port, and a PC port.

The install ation procedure is as follows:

1. Open the bottom cover of the GSM VOIP Gateway and insert an SIM card of the local GSM network
2. Connect the LAN port with the upper-layer network equipment with the Ethernet cable

3. PC port supports network sharing, so connect the PC port to the computer or lower-layer switch (HUB or router)
4. Connect the output terminal of the transformer with the power port.
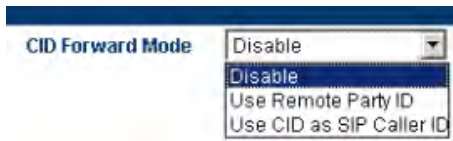
2.2   Connection Figure
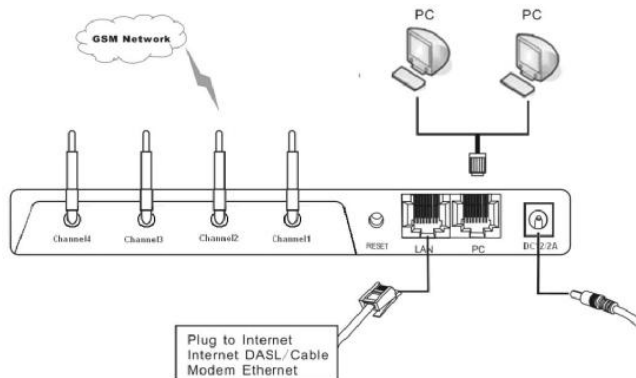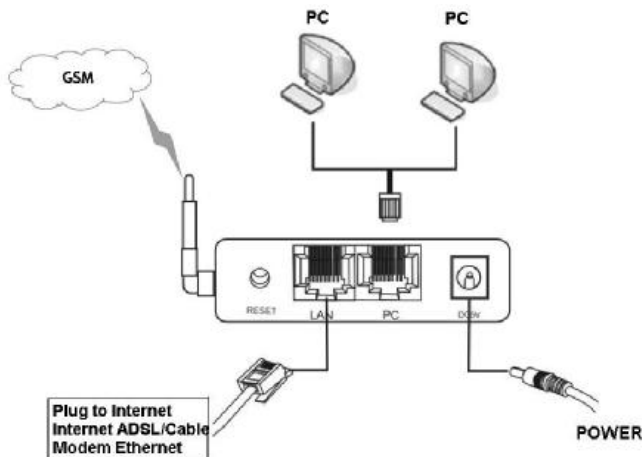
Sending Message to 192.168.2.1:5060:↵
INVITE sip:5000@192.168.2.1:5060;transport=udp SIP/2.0↵
Via: SIP/2.0/UDP 192.168.2.180:5060;branch=z9hG4bK1450498491↵
From: "13800000000" <sip:13800000000@192.168.2.1:5060>;tag=232569343↵
To: <sip:5000@192.168.2.1>↵
Call-ID: 1853068986@192.168.2.180↵
CSeq: 2 INVITE↵
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↵

3.11   Save the Change

After setting is changed, click "Save" and the new setting will be valid. Otherwise, the new setting is invalid.

SIP Terminal

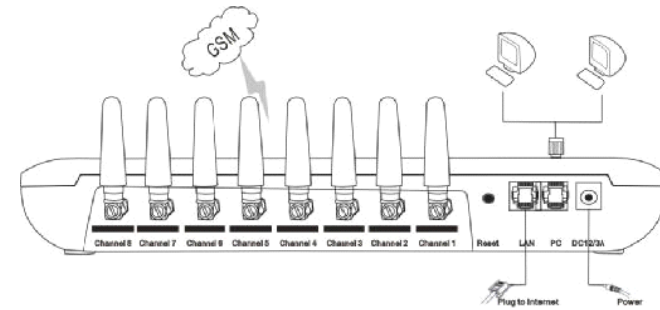**Disable:** It's not all owed to transfer the PSTN caller number to the VoIP system;

Enable: The CID is set as the SIP caller number.

Use Remote Party ID: The GSM VOIP Gateway will add the PSTN caller number to the call request signaling of the VoIP system. The signaling is as follows (provided that the PSTN caller number is 13800000000):
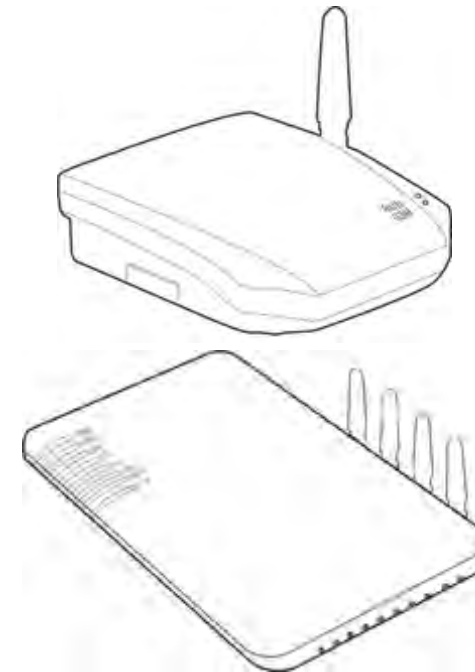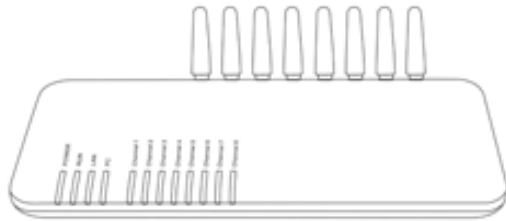
```
Sending Message to 192.168.2.1:5060:
INVITE sip:5000@192.168.2.1:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.2.180:5060;branch=z9hG4bK1645487913
From: <sip:20001@192.168.2.1:5060>;user=phone;tag=406202416
To: <sip:5000@192.168.2.1>
Call-ID: 847230278@192.168.2.180
CSeq: 2 INVITE
Contact: <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
```

Use CID as SIP caller number: The GSM VOIP Gateway directly initiates the call request to the VoIP system through the PSTN caller number and adds the relevant information to Remote Party ID option of the request signaling. The call request signaling is as follows:

2.3   LED Indicators

The description of LED indicators is as follows

LED	Description
Power indicator	Power indicator This indicator is constantly ON after connected with power
LAN indicator	This indicator is ON after connected with the network equipment
PC indicator	This indicator is ON after connected with the network equipment and blinks during data transmission.
RUN indicator	1. The RUN indicator blinks once every 100ms during startup. 2. When GSM VOIP Gateway is connected with the server, the RUN indicator bli nks once per second.
Channel indicator	1. When the GSM module of the GSM VOIP Gateway logs onto the local GSM2. 2. When this indicator blinks quickly, it indicates that the GSM VOIP Gateway is trying to log onto the local GSM network. 3. When the GSM channel of the GSM VOIP Gateway is activated, this indicator is normally ON.

2.4   SMS Instructions

To: <sip:20001@192.168.2.1>

Call-ID: 808807EB-A8B3-DD11-BBA6-005056C00008@192.168.2.89

CSeq: 3MESSAGE

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

Note: The SMS forwarding mode of the GSM VOIP Gateway is only functional under the SIP protocol.

3.10   Transparent Transmission of PSTN Caller Numbers

The GSM VOIP Gateway permit s the transparent transmission of PSTN call er numbers to the VoIP system in various methods.

H323 Terminal

The following is an example that the GSM VOIP Gateway forwards the SMS to the SIP 3999. The red part is the content of the SMS.

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: 4MESSAGE

Contact: <sip:20001@192.168.2.162:5060>

Max-Forwards: 30

User-Agent: H
Content-Type: text/plain

Content-Length: 28

8613682626865

075583185700

2. The SMS sent to the GSM VOIP Gateway from the SIP is forwarded to the specified PSTN number.

The following example is about the Hello world sent from SIP 3999 to 13682626800. Where, in the content of the SIP message (in red), the first li ne indicates the number to receive the SMS. The second line indicates the content of the SMS.

MESSAGE sip:20001@192.168.2.162:5060 SIP/2.0

From: <sip:3999@192.168.2.89>;tag=5031

Users can send instructions to the GSM VOIP Gateway through the SMS.

Function	Instructions (Short message content)	Remark
Obtain information from the LAN port	INFO or info	Case insensitive
Reset the GSM VOIP Gateway setting	RESET Password	The keyword RESET is case insensitive.
Reboot the GSM VOIP Gateway	REBOOT Password	The keyword REBOOT is case insensitive.

1) When "info" or "INFO" is sent to the GSM VOIP Gateway through SMS, the GSM VOIP Gateway will return immediately the LAN port info to the mobile phone.
2) When performing reset/reboot through the SMS, the password authentication is required. The password is identical with that of the admin. The detail ed procedure is as follows:

   a. The keyword "reset" and "reboot" is case insensitive, but the password is case sensitive.
   b. When the reset instruction is sent, the GSM VOIP Gateway will automatically reboot.

   To perform reset, the admin password of GSM VOIP Gateway user is "tengda". Input "reset tengda" or "RESET tengda" in the SMS to reset.

   To perform reboot, the admin password of GSM VOIP Gateway user is "tengda". Input "reboot tengda" or "REBOOT tengda" in the SMS to reboot.

3. Page Settings

Before setting the page, you need to have the IP address of the PC port of the gateway first. Connect the computer for setting the gateway to the PC port of the GSM VOIP Gateway. The GSM VOIP Gateway has a built-in page server that is used to accept or obtain the HTTP. You can set the related functions for the GSM VOIP Gateway through the Internet Explorer.

3.1　　Page Setting Menu

You can access the setting page of the GSM VOIP Gateway through the IP address of the LAN port or PC port. The default factory settings are as follows:

A: The LAN port supports the DHCP (dynamic IP address). Users can dial the SIM card number of the gateway and if connected dial *00 to obtain the IP address.

B: The default IP address and mask of the PC port are 192.168.8.1 and 255.255.255.0 respectively.

Enable the DHCP service of the PC port.

Connect the computer with the PC port of the gateway through the IP address of the PC port, and set the IP of the computer to dynamic IP or fixed IP as 192.168.8.xxx and the default gateway as 192.168.8.1.

Open the Internet Explorer and enter 192.168.8.1 or http://192.168.8.1 in the address bar, then, the login page is popped up for password input. Enter the login account ("admin" as default) in the User Name, and password ("admin" as default) in the Password field.

```
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

}
```

3.9.3　　SMS Forwarding

The GSM VOIP Gateway supports the SMS forwarding through the SMS under SIP protocol. After users send the short message to the GSM VOIP Gateway through the SMS, the GSM VOIP Gateway will send the short message to the specified VoIP number automatically.

As shown in the above figure, select SMS Mode > Forwarding and enter the VoIP number that is used to receive the SMS information. The VoIP will automatically forward all the SMS information from the GSM network to this VoIP number. Similarly, the GSM VOIP Gateway will automatically forward all the SMS information from the VoIP to the specified GSM mobile phone.

1. The GSM VOIP Gateway forwards the SMS from the GSM to a specified SIP number.

[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

Click "OK" button and the gateway status page is displayed as default

WT-2201 Status Interface

WT-2204 Status Interface

WT-2208 Status Interface

The GSM VOIP Gateway adopts the tree structure. The menu is on the left and the setting parameters are on the right, as shown in the above figure.

```
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

}
```

When the user enters an SMS dial prefix, such as 999, the above call request signaling is changed to:

Send RAS Message: admissionRequest

```
admissionRequest {

requestSeqNum = 241

callType = pointToPoint NULL

endpointIdentifier = "3705_endp"

destinationInfo = 1 elements {
```

callType = pointToPoint NULL

endpointIdentifier = "3705_endp"

destinationInfo = 1 elements {

    [0] = dialedDigits "8675588228822"

}

srcInfo = 2 elements {

    [0] = dialedDigits "20001"

    [1] = h323-ID "20001"

 }
 srcCallSgnalAddress = 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.

}

You can also access the setting page of the GSM VOIP Gateway through the IP address 192.168.2.216 or 192.168.2.172 of the LAN port of the gateway. The login method is the same as that of the PC port, but you must first obtain the IP address of the LAN port.

3.2   Status

The status page contains the following contents, as shown in the above figure

3.2.1   Telephone Information

A.   Product Sequence Number

Each GSM VOIP Gateway has a factory set sequence number, such as GOIP08030031, which is used for centralized setting, technical support, and maintenance filing. The sequence number is printed on the bottom plate of the gateway and is read-only.

B.   Software Version

It displays the current version of software used by the GSM VOIP Gateway. When you want to upgrade the software, make sure the update version is newer than the current version.

C.   Hardware Version

It displays the current hardware version of the gateway.

D.   Line Register Status

It displays the login status of the line. When the line has logged into the SIP server or H.323 Gatekeeper, LOGIN is displayed, otherwise, LOGOUT is displayed.

E.   Line Use Status

It displays the use status of the line. When the line is in use, the status is ACTIVE, when the line is idle, the status is IDLE.

3.2.2   Network Information

A.   LAN Port

It displays the current IP address of the LAN port, such as 192.168.2.172.

B.   PC Port

It displays the current IP address of the PC port.

C.   PPPoE Dialing

It displays the PPPoE broadband connection condit ion. After the connection, the IP address obtained is displayed on the LAN port.

D.   Default Route

It displays the current gateway address.

E.   Domain Name Server (DNS)

It displays the current DNS address.

3.3   User Options

Click "User Options" and the following figure shows.

User Options of the WT-2201

Examples of SMS Dialing:

In the following SMS dialing examples, the H.323 number of the GSM VOIP Gateway is set as follows:

Mode 2:

When a short message of "8675588228822" is sent from the mobile phone number (+86)13800000000 to the GSM VOIP Gateway, the GSM VOIP Gateway will send the following call requests. When the GSM VOIP Gateway sends a call request through the H.323 number of the GSM VOIP Gateway, the GSM VOIP Gateway will automatically add the number of the short message sender to the PSTN Forwarding Number in Call Forwarding (VoIP Incoming Call , Forwarding to the PSTN Immediately);

In this mode, when the GSM VOIP Gateway receives the call from the H.323 GK, the GSM VOIP Gateway will forward the call to the short message sending equipment through the GSM network.

The call request signaling in this mode is as follows:

Send RAS Message: admissionRequest
         admissionRequest {
                 requestSeqNum = 241

Content-Length: 226

3.9.2   SMS Dialing under the H.323 Protocol

The GSM VOIP Gateway permits users to dial back through the SMS under the H.323 protocol. After users send the called number to the GSM VOIP Gateway through the SMS, the GSM VOIP Gateway will send a call request to the H.323 GK automatically. Users who need this function shall choose the following parameters:

SMS Mode	Dial
SMS Dial	Mode 1
SMS Dial Prefix	

Select SMS Mode > Dial and the following page is displayed.

The GSM VOIP Gateway supports three types of outgoing call via SMS:

A: Mode 1 (The current version of the H.323 protocol does not support this mode, but the later version will.)
In this mode, the GSM VOIP Gateway sets the number of the short message sender as the calling number of the call and the called number as the short message content;

B: Mode 2
In this mode, the GSM VOIP Gateway sets the H.323 number of the GSM VOIP Gateway as the calling number of the call and the called number as the short message content;

C: Mode 3 (The current version of the H.323 protocol does not support this mode, but the later version will.)
In this mode, the GSM VOIP Gateway sets the H.323 number of the GSM VOIP Gateway as the calling number of the call and the called number as the short message content and the number of the short message sender, whose format is short message content*the number of the short message sender.

D: SMS Dial Prefix
When the GSM VOIP Gateway initiates the SMS call, the GSM VOIP Gateway will change the prefix number to the called number prefix.

User Options of the WT-2204/ WT-2208

3.3.1   Language

To select a language, refresh the page to enter the language page required. For example, the current language is simplified Chinese. If you wish to display the page in English, click "English" in the menu. After your terminal is restarted, all the pages will be displayed in Engli sh.

You can also use other languages for setting, as shown in the following figure

Click "English" on the upper right corner of the setting page. Then, the setting page will display all info in English. But the shortcut will not change the language of the setting page when logged in next time.

3.3.2   Time Zone and Time Server

This item displays the adjusted time according to the selected time zone. The gateway receive time and date information from the server through the Network Time Protocol and the time difference will I be automatically adjusted. For example, the pacific standard time (PST) is GMT-8 and the pacific daylight time (PDT) is GMT-7.

Time Zone	GMT+8
Time Server	pool.ntp.org

The time zone indicates the zone where the gateway is used. You need to enter the correct time zone, so that the time of the caller ID and charging information can be displayed correctly. The time server is the address of the server that obtains the network time through the Internet. The default time server is timekeeper.isi.edu.

3.3.3   DTMF Minimum Detection Interval

This parameter is used to set the minimum interval of two DTMF signals. Packets may be lost during the data transmission over the GSM. As a result, a DTMF may be incorrectly identified as two or multiple identical DTMFs when detected by the GSM VOIP Gateway. The problem of repeated code can be solved effectively through the modification of the parameter.

DTMF Min Detect Time Gap	80

This parameter value ranges from 60ms to 120ms and the default value is 80ms. When the value of this parameter is increased properly, the repeated DTMF can be avoided efficiently. However, the packet loss may also be caused.

3.3.4   Automatic Setting

If the service provider provides the automatic setting, you can select "Enable" to start the automatic setting feature and enter the address of the server. If the service provider does provide the automatic setting, you need to select "Disable" to speed up the startup time of the GSM VOIP Gateway

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GSM VOIP Gateway sends a call request through the SIP number of the GSM VOIP Gateway, the GSM VOIP Gateway will automatically add the number of the short message sender to the PSTN Forwarding Number in Cal Forwarding (VoIP Incoming Call , Forwarding to the PSTN Immediately);

In this mode, when the GSM VOIP Gateway receives the call from the SIP server, the GSM VOIP Gateway will forward the call to the short message sending equipment through the GSM network.

The SMS dial prefix is still valid in this mode;

The call request signaling in this mode is as follows:

SendingMessage to 192.168.2.1:5060:
INVITE
sip:8675588228822*8613800000000@192.168.2.1:5060;transport=udp
SIP/2.0

Via: SIP/2.0/UDP 192.168.2.237:5060;branch=z9hG4bK363969813

From: <sip:20001@192.168.2.1:5060>;user=phone;tag=65248630

To: <sip:8675588228822*8613902994477@192.168.2.1>

Call-ID: 117025903@192.168.2.237

CSeq: 2 INVITE

Contact: <sip:20001@192.168.2.237:5060>

Max-Forwards: 30

User-Agent:H

Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER, MESSAGE, INFO, SUBSCRIBE

Content-Type: application/sdp

SendingMessage to 192.168.2.1:5060:

INVITE sip:8675588228822@192.168.2.1:5060;transport=udp SIP/2.0

Via: SIP/2.0/UDP 192.168.2.237:5060;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: H

Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER, MESSAGE, INFO, SUBSCRIBE

Content-Type: application/sdp

Content-Length: 226

Mode 3:

SMS Mode	Dial
SMS Dial	Mode 3
SMS Dial Prefix	

When a short message of "8675588228822" is sent from the mobile phone number (+86)13800000000 to the GSM VOIP Gateway, the GSM VOIP Gateway wil l send the following call requests. When a short message of "8675588228822" is sent from the mobile phone number (+86)13800000000 to the GSM VOIP Gateway, the GSM VOIP Gateway wil l send the following call requests. When the

Auto-provision	⦿ Enable ◯ Disable
Provision Server	
Provision Interval	

It is a special server, which needs the support of the specific system.

3.3.5 Setting of the Remote Control

Press *20# on the terminal to initiate the request to realize the remote management of equipment. The remote control server is provided by the service provider. The default port is 1920 and the terminal is identified by the SN. The remote control password is identical with that of the server and is set as default.

Remote Control<<	
Remote Server	
Remote Server Port	
Remote Server ID	
Remote Server Key	

In the following figure, the remote control server is set as 118.142.51.162. The terminal user presses *20# and a long tone is heard, which indicates that the instruction has been successfully sent. The remote administrator access http://118.142.51.162:8086 and the model and SN of the gateway are displayed. Click the gateway SN to set the remote gateway.

Remote Control<<	
Remote Server	118.142.51.162
Remote Server Port	1920
Remote Server ID	${SN}
Remote Server Key	

Note: It is a special server, which needs the support of the specific system. For details, please contact the technical support Witura Corporation Sdn Bhd.

3.3.6 GSM Group Mode

Users can establish a GSM group containing multiple GSM VOIP Gateway. Under this mode, the administrator only needs to provide a GSM number to the user to call in the VoIP system.

Incoming Call Forward To Client GSM VOIP Gateway
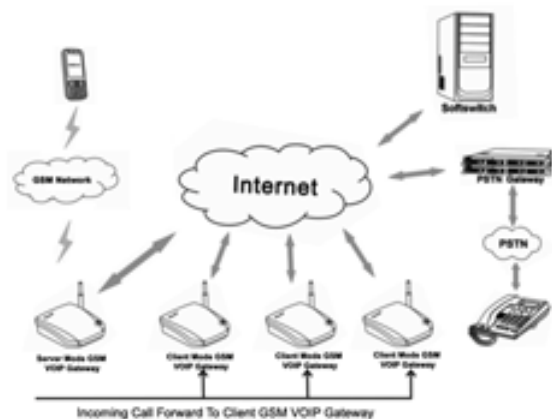
Each GSM VOIP Gateway can operate in any of the following modes:

Prohibit: This mode is used when the GSM VOIP Gateway operates independently.

Serve as the server: When GSM VOIP Gateway operates in this mode, the administrator only needs to provide the user with a GSM number of the GSM VOIP Gateway as a unique access number to the GSM VOIP Gateway group. In one GSM VOIP Gateway group, only one GSM VOIP Gateway can be used as the server. When the GSM VOIP Gateway serves as the server, the GSM unconditional call forwarding or busy call forwarding can be activated. The unconditional call forwarding is used to forward all incoming calls to other clients of the group. The busy call forwarding is used to forward incoming call s to other clients of the group when the status of the SIM card of the server is ACTIVE.

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

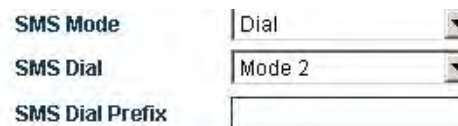**Contact: <sip: 8613800000000@192.168.2.237:5060>**

Max-Forwards: 30

User-Agent: H

Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER, MESSAGE, INFO, SUBSCRIBE

Content-Type: application/sdp

Content-Length: 226

Mode 2:

When a short message of "8675588228822" is sent from the mobile phone number (+86)13800000000 to the GSM VOIP Gateway, the GSM VOIP Gateway will send the following call requests. When the GSM VOIP Gateway sends a call request through the SIP number of the GSM VOIP Gateway, the GSM VOIP Gateway will automatically add the number of the short message sender to the PSTN Forwarding Number in Call Forwarding (VoIP Incoming Call, Forwarding to the PSTN Immediately);

In this mode, when the GSM VOIP Gateway receives the call from the SIP server, the GSM VOIP Gateway will forward the call to the short message sending equipment through the GSM network.

The SMS dial prefix is still valid in this mode;

The call request signaling in this mode is as follows:

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 192.168.2.237:5060;branch=z9hG4bK363969813

From:
<sip:861380000000@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: 861380000000@192.168.2.237:5060>

Max-Forwards: 30

User-Agent: H

Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REFER, REGISTER, MESSAGE, INFO, SUBSCRIBE

Content-Type: application/sdp

Content-Length: 226

When the user enters an SMS dial prefix, such as 999, the above call request signaling is changed to:

SMS Mode	Dial
SMS Dial	Mode 1
SMS Dial Prefix	999

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 192.168.2.237:5060;branch=z9hG4bK363969813

GSM Group Mode	As Server
GSM Forward Mode	Unconditional Forward
GSM CallerID Anonymous	Unconditional Forward / Busy Forward

Serve as the client: When GSM VOIP Gateway operates in this mode, it will send it s real-time status to the server of the GSM VOIP Gateway group, so that the GSM VOIP Gateway server can deploy the call forwarding.
Server address: It is the IP address of the GSM VOIP Gateway of the GSM VOIP Gateway group server.

GSM number: It is the telephone number of the GSM SIM card used by the GSM VOIP Gateway.

3.3.6 Anonymity Of the GSM Caller Number

GSM CallerID Anonymous	○ Enable ● Disable
GSM Band	900/1800
	☑ Auto Reboot
Reboot Time	04:00
China Phone Code	○ Enable ● Disable
	☑ IVR

The caller number can be hidden, but this needs the support of the GSM operator.

3.3.7 IMEI

IMEI	359094025200031

The IMEI (International Mobile Equipment Identity) is an electric serial number containing 15 digits.

3.3.8 SMS Send to Client

The SMS server sends the request to the mobile phone via the GSM VOIP Gateway, or the mobile phone sends the request to the SMS Server via GSM VOIP Gateway.

SMS server address: This is for filling the IP of the SMS server. Please make sure that server is installed with the GSM VOIP Gateway SMS management server software independently developed by Witura Corporation Sdn Bhd (The software can be obtained from our technical personnel).

SMS server port: It is the SMS port of the SMS server. The default value is 44444. It must be consistent with that of the server if modified on the server.

Authentication ID: It is the user ID. Make sure that the server has corresponding ID.

Authentication password: It is for filling the user password.

Note: The SMS send cli ent of the WT-2204 and WT-2208 are in the call set options, which needs the support of the GSM VOIP Gateway SMS management server.

3.3.9 GSM Band

The GSM VOIP Gateway support the GSM/GPRS 900/1800 and 850/1900 bands.

C: Mode 3

In this mode, the GSM VOIP Gateway sets the SIP number of the GSM VOIP Gateway as the calling number of the call and the called number as the short message content and the number of the short message sender, whose format is short message content*the number of the short message sender.

D: SMS Dial Prefix

When the GSM VOIP Gateway initiates the SMS call, the GSM VOIP Gateway will change the prefix number to the called number prefix.

Examples of SMS Dialing:

In the following SMS dialing examples, the account of the SIP of the GSM VOIP Gateway is set as follows:

Mode 1:

When a short message of "8675588228822" is sent from the mobile phone number (+86)13800000000 to the GSM VOIP Gateway, the GSM VOIP Gateway will send the following call request signaling:

3.9 SMS Mode

The GSM VOIP Gateway permits you to call VoIP users or forward short messages through the SMS.

3.9.1 SMS Dialing Under SIP Protocol

Under the SIP protocol, the GSM VOIP Gateway permit s users to dial back through the SMS. After users send the called number to the GSM VOIP Gateway through the SMS, the GSM VOIP Gateway will send a call request to the SIP server automatically. Users who need this function should choose the following parameters:

Select SMS Mode > Dial and the following page is displayed.

The GSM VOIP Gateway supports three types of outgoing call via SMS:

A: Mode 1
In this mode, the GSM VOIP Gateway sets the calling number of the SMS as the calling number of the call and the called number as the short message content;

B: Mode 2
In this mode, the GSM VOIP Gateway sets the SIP number of the GSM VOIP Gateway as the calling number of the call and the called number as the short message content;

3.3.10 Timing Restart

The GSM VOIP Gateway restarts at least once at the specified time every day to clear the buffer of the GSM VOIP Gateway, so that the GSM VOIP Gateway can operate normally.

3.3.11 China Area Code Matching

The GSM VOIP Gateway can match all area codes of China to ensure the prompt dial. The default status is Disable.

3.3.12 IVR

By default, the IVR is enabled. When a call comes in, the system prompts the user to dial a second time. When the IVR is disabled, the system will not prompt the user to dial a second time.

3.3.13 Prompt Tone System

The prompt tones are the combination of the intervals and frequencies of the dialing tones and ring-back tones when users hook off the telephone. You can select the following prompt tones for the GSM VOIP Gateway according to the countries and regions where the GSM VOIP Gateway is used, so as to remain conventional tones.

Customize:

Users can customize prompt tones according to their special requirements. Select Customize, the following setting parameters are displayed.

The setting parameters are defined as follows:

Each prompt tone involves the following parameters. If a parameter is not defined, the value of the parameter shall be 0.

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

nc: the number of tones (1-4)

rpt: the number of repeats (0 to infinity)

c1on: the duration when the frequency 1 is on (ms)

c1off: the duration when the frequency 1 is off (ms)

c2on: the duration when the frequency 2 is on (ms)

c2off: the duration when the frequency 2 is off (ms)

SIM card status reporting number: The gateway can report the status of SIM cards (remaining call duration) through the SMS. This parameter is used to specify the mobile phone number to receive the SMS.

SIM card status reporting time: This parameter is used to specify the remaining call duration and then send the report.

SIM card ID: This parameter is used to specify the ID of SIM cards in the short message report. You can specify the mobile phone number corresponding to the SIM card or any character string as the ID.

One-time call duration limit of SIM cards: This parameter is used to specify the duration of one-time calls (by minutes).

Examples and Explanations:

SIM Card Call Duration Setting   SIM Card Remaining Call Duration

The setting is same as the SIM card time limit diagram. The total call duration of the SIM card is 30 minutes. When the call duration is less than or equals 10 minutes, the gateway will send a short message to 13713652130 (the SIM card ID is 2130 in the reporting message.) to report the remaining call duration. When one-time call duration is longer than 8 minutes, the call will be disconnected.

When the call duration is over, it becomes 0. Users can dial the SIM card number by the mobile phone and when the second dialing tone is heard, press *10 to restore the value.

This mode is used to set the above password authentication and trust list authentication at the same time.

For a downlink call, the authentication mode is as follows: If the number in the trust list is used to dial the user served by the PSTN, the call will be connected. When the number you dialed is not in the trust list, you need to enter the password after the secondary dial tone is played. Then, the call will be connected.

For an uplink call, the authentication mode is as follows: If the mobile number or fixed number in the trust list is used to dial the user served by the VoIP, the call will l be connected. When the number you dialed is not in the trust list, you need to enter the password after Please Enter the Password is played. Then, the call will be connected.

3.8   Call Duration Limit

The call duration limit is to limit the call duration of SIM cards in the gateway. Through this function, you can specify the total call duration of SIM cards. When the call duration is longer than the specified value, the call s shall not be connected to prevent the unnecessary or unsafe call charging. The setting parameters are as follows:

The parameters are defined as follows:

SIM card limit time: This parameter sets the total call duration of SIM cards. When the call duration is longer than the specified time (by minutes), the call cannot be connected. When this parameter is null, the default call duration is infinite.

c3on: the duration when the frequency 3 is on (ms)

C3off: the duration when the frequency 3 is off (ms)

f1: the frequency of tone #1, 300 to 3000Hz

f2: the frequency of tone #2, 300 to 3000Hz

f3: the frequency of tone #3, 300 to 3000Hz

f4: the frequency of tone #4, 300 to 3000Hz

p1: the increment of tone #1, 0 to 31(0=3dB, -1dB increments)

p2: the increment of tone #2, 0 to 31(0=3dB, -1dB increments)

p3: the increment of tone #3, 0 to 31(0=3dB, -1dB increments)

p4: the increment of tone #4, 0 to 31(0=3dB, -1dB increments)

Example:

To add a prompt tone where f1 is 450Hz, c1on is 750ms, and c1off is 1000ms, enter the following values in the corresponding boxes:

1,0,750,1000,0,0,0,0,450,0,0,0,20,0,0,0

3.4   Network Setting

Click the "Network Setting" in the menu on the left, and the following page is displayed:

3.4.1 LAN Port Setting

The LAN port of the GSM VOIP Gateway can be set to the dynamic IP through DHCP, fixed IP, and PPPoE dialing. There are three setting modes:

A. Dynamic IP (DHCP)

This setting is default. If the network for the user provides the DHCP service, the GSM VOIP Gateway will require the network information such as IP address from the DHCP server automatically.

B. Fixed IP

Select fixed IP, and the following setting page is displayed.

Downlink   Uplink

The setting is as follows: Select "Forward to PSTN Authentication Mode" > "Trust List Authentication". Click the "VoIP Trust Number List", and the VoIP Trust Number List is displayed (Maximum 15 trust numbers can be entered). Enter the trust VoIP number in trust number sequence.

If only a VoIP trust number, such as 3306, is set, only the number 3306 can be used to dial the PSTN from the VoIP.

3.7.2.3 Password of Trust List Authentication

3.7.2 Authentication Mode Setting

The authentication mode is classified into the password authentication, trust list authentication, and password or trust list authentication.

Downlink (VoIP to PSTN)   Uplink (PSTN to VoIP)
authentication mode   authentication mode

3.7.2.1 Password Authentication

The setting is as follows: Select "Forward to PSTN Authentication Mode" > "Password Authentication". Enter the password in "Call PSTN Authentication Password".

As indicated in the above figure, for call s from the VoIP to the PSTN, when the second dialing tone is heard, dial the set password and the call will be connected. For call s from the PSTN to the VoIP), when "Please Enter the Password" is played, enter the password and then the call will be connected.

3.7.2.2 Trust List Authentication

Set these parameters according to the network the user uses.

C. PPPOE
PPPoE (Point-to-point protocol over Ethernet) is a network protocol that compresses the PPP in the Ethernet. Select PPPoE dialing, and enter the account and password provided by the network provider.
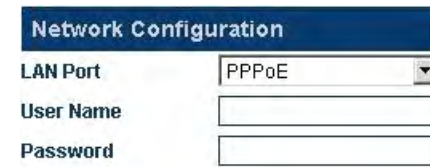
D. 802.1qVLAN
When the network serving the user provides the VLAN service, enter the parameter as required.

E. Advance…
Click Advance, and the Hardware address and Broadcast address are displayed in the page.

The hardware address is used to enter the MAC address in the format of XX: XX: XX: XX: XX: XX.
The broadcast address is used to communicate with other computers connected to the GSM VOIP Gateway.

3.4.2 PC Port Setting

The PC port can be set to connect other network equipment through the route or bridging mode. The two setting modes are as follows:

A. Bridging
When the PC port is set to the bridging mode, the relation between the LAN port and the PC port is layer 2 switching. The network equipment connected with the PC port same as the connection with the LAN port.

B. Fixed IP Address

Select the fixed IP, and the following setting parameters are dis played. Enter the IP address and subnet mask (the network section of the IP address should be different from that of the LAN port to prevent conflict).

PC Port	Static IP
IP Address	192.168.8.1
Subnet Mask	255.255.255.0
DHCP Server	⦿ Enable ○ Disable
Starting Address	
Ending Address	
Static DNS(optional)	
	Advanced>>

C. Enable the DHCP Service

This service can be enabled only when the PC port of the GSM VOIP Gateway is set as the fixed IP. To enable the DHCP service, you need to enter the start address and end address.

D. Advance...

Click Advance, and the Hardware address and Broadcast address are displayed in the page.

The hardware address is used to enter the MAC address in the format of XX: XX: XX: XX: XX: XX.

The broadcast address is used to communicate with other computers that are connected to the ATA.

3.4.3 Main DNS

The DNS (domain name system) is a database that stores the Internet names and addresses, and converts between the name and the common Internet protocol digits. The main DNS is the IP address of the main DNS (such as 202.67.156.221 or obtain from the service provider). If the PPPoE is set, the main DNS will be automatically provided by the service provider. This parameter can be null .

3.4.4 Secondary DNS

3.7   Call Forwarding (Setting on the Call Route And Authentication Mode)

The gateway provides the call routing function for users, which can be set in the Call Forwarding Setting. The call routing is to forward calls to specified numbers, so that the dialing time can be decreased. In addition, the gateway provides three authentication modes for the uplink (call s from PSTN to VoIP) and downlink (call s from VoIP to PSTN). Do not set these parameters when they are not needed.

3.7.1 Call Route Setting

Forward to PSTN	⦿ Enable ○ Disable	Forward to VoIP	⦿ Enable ○ Disable
Forward Number (VoIP To PSTN)	88290211	Forward Number (PSTN To VoIP)	3306

Downlink: from VoIP to PSTN   Uplink: from PSTN to VoIP

Note: The value of Call PSTN must be Enabling. Otherwise, The GSM VOIP Gateway prohibits any access to the PSTN. Therefore, set this parameter carefully.

The above Note is also suitable for Call VoIP.

1. Set a hotline number in Call PSTN Forwarded-to-number. When the user served by the VoIP network calls the GSM VOIP Gateway, the call is forwarded to the hotline number.

    When the user served by the VoIP network call s the GSM VOIP Gateway, the GSM VOIP Gateway connects the call and dials 88290211 directly. This feature is especially useful for hotline services.

2. Set a VoIP number in Call VoIP Forwarded-to-number. When the user served by the PSTN call s the VoIP network, the call is forwarded to the VoIP number. When a user calls another user served by the PSTN, the HT-342 calls the 3306 terminal of the VoIP. When the 3306 terminal answers the call , the HT-342 connects the call and the call will be connected. This feature enables international roamers to answer the phone through the VoIP anywhere.

The rule is "0:|13[0-9]xxxxxxxx:+0|[1-8]xxxxxxx:+0755".

When you dial the number 88990011 and 8899001133, the result is the same. The number actually dialed is 075588990011.

3.6   Volume Adjustment

The GSM VOIP Gateway has a built-in volume adjustment panel, which should be used carefully. When you need to adjust the volume of the gateway, change the address http://xxx.xxx.xxx.xxx/xxx/gain.html to

http://xxx.xxx.xxx.xxx/xxx/gain.html. Then, the following volume setting page is displayed.

After the volume setting is completed, click "Save" and the setting will take effect immediately (for ongoing calls, the setting will not take effect immediately).

Note: The adjustment on the output volume of the line may cause the terminal to fail in dialing numbers. Therefore, set this parameter carefully. The adjustment on the input/output volume is for VoIP lines.

When the main DNS address fail s to connect or is not avail able, the secondary DNS can be used (such as 202.67.156.222 or obtain from the service provider). If the PPPoE is set, the secondary DNS will be automatically provided by the service provider. This parameter can be null .

3.5   Call Setting

This section describes the basic setting of the network connection relating to the GSM VOIP Gateway, which supports two protocols: H323 and SIP. The setting page is as follows: You can select a protocol in the "Terminal Type".

3.5.1   H.232 Terminal Setting

The H.323 protocol involves the direct connection mode and Gatekeeper mode.

3.5.1.1   Direct Connection

Under this mode, the GSM VOIP Gateway operates in the point-to-point status. The setting parameters are as follows:

A. H.323Telephone Number
The value of this parameter is a decimal numeral string that is used to confirm the telephone number in the telephony network. For example, 5551234 is a valid telephone number. Enter the telephone number in this parameter.

B. Display Name

This parameter is used to display the name of the user who subscribes the H.323 service. For example, when you call your friend John Smith, your name will be displayed on your friend's telephone.

C. H.323 ID

H232 ID is used to verify the account. Users can set this parameter according to the requirements of the service provider.

D. Default Voice Gateway ay

This parameter is used to find the proper Gatekeeper or IP address of the equipment of the call. Enter the IP address, such as 192.168.2.197, or domain name, such as gk.yourisp.com. If the soft-switch system uses the non-standard port (1719 and 1720), you can add the detailed port number at the end of the IP address or domain name of the Gatekeeper. For example, if the port number is 7300, the IP address is 192.168.2.197:7300 and the domain name is gk.yourisp.com:7300.

Under the direct connection mode, the GSM VOIP Gateway will forward all call s to the VoIP network to this address.

Note: The value of this parameter must be standard ASCII characters (enter characters under the English input mode).

3.5.1.2   Gatekeeper

3.5.7.2   Dialing Rule with Specified Length Of Numbers

If you need to specify the length of telephone numbers matched, you can specify the dialing rule as "AAXXXXXX:-aa+bb". Where, "AAXXXXXX" indicates the number to match and the length of the number. "AA" indicates the head and other numbers are represented by X or x. The digit s after the colon indicates detailed actions to be taken on the number.

The setting is as follows:

In the above example 3, the rule is "00:|0:-0+0086|:+0086755", which can be changed to "00:|0:-0+0086|[1-8]xxxxxxx:+0086755".

It means that when you dial a number whose first digit is 1 to 8 and total length is 8, the gateway will automatically dial the number and add 0086755 before the number.

Examples:

"0:|13:+0|:+0755"

This rule enables the GSM VOIP Gateway to add 0 before the mobile phone number and 0755 before the telephone number.

The above rule can be change to:
 "0:|13[0-9]xxxxxxxx:+0|[1-8]xxxxxxx:+0755"

Similarly, this rule enables the GSM VOIP Gateway to add 0 before the mobile phone number and 0755 before the telephone number. However, the length of the mobile phone number is limited to 11 digits.

As shown above, the length of telephone numbers is limited to eight digits. 13[0-9]xxxxxxxx and [1-8]xxxxxxx represent mobile phone numbers 130xxxxxxxx to 139xxxxxxxx, and telephone number 1xxxxxxx to 8xxxxxxx, respectively.

Note: When the length of the number is specified, the exceeded numbers will be discarded if the length of the number exceeds the specified length. For example:

added. If the number fails to match, the number continues to match the next rule. If no digit after the colon is specified, such as "00:", it indicates that no actions are taken when "00" is matched and the number exit s the matching. If no digit before the colon is specified, such as "+86755", it indicates that instead of matching, actions are taken on the number directly.

4. You can specify a range for matching of dialing rules. The rule format is "[A-B]A:-aa+bb" or "A[A-B]:-aa+bb". For example, you can specify the range of numbers beginning with 2 to 8 as "[2-8]: -aa+bb" or numbers beginning with 13 to 15 as "1[3-5]: -aa+bb".

Examples:

1. Rule: 0:|:+0755

a. The input number is "02083185711" and the output number is "02083185711".

b. The input number is "83185700" and the output number is "075583185700".

2. Rule: "00:-00|0"-0+86|:+86755"

a. The input number is "008522343318" and the output number is "8522343318".

b. The input number is "02083185711" and the output number is "862083185711".

c. The input number is "83185700" and the output number is "8675583185700".

3. Rule: "00:|0:-0+0086|:+0086755"

a. The input number is "008522343318" and the output number is "008522343318".

b. The input number is "02083185711" and the output number is "00862083185711".

c. The input number is "83185700" and the output number is "008675583185700".

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

a. The input number is "076322343318" and the output number is "076322343318".

b. The input number is "13044557766" and the output number is "013044557766". Or, the input number is "13644557766" and the output number is "013644557766".

c. The input number is "23185700" and the output number is "075523185700". Or, the input number is "73185700" and the output number is "075573185700'.

Under the Gatekeeper mode, the GSM VOIP Gateway operates in the H.323 register status. When you register through the H.323 protocol, select "H.323 Terminal" in "Terminal Type", as shown in the above figure. The registration mode involves the Gatekeeper and direct connection (the direct connection mode is used for call s over the IP address). When all lines use one number, select the single server setting mode. When all lines use different numbers, select the line setting mode. When you select the line setting mode, each line can be registered to different servers. The setting parameters are as follows:

A. H.323Telephone Number
The value of this parameter is a decimal numeral string that is used to confirm the telephone number in the telephony network. For example, 191 is a valid telephone number. Enter the telephone number in this parameter.

B. Gateway ay Prefix
When you register through the gateway prefix, enter the prefix number. When the prefix number is called, a dialing tone is heard and then the secondary dialing is required. The gateway prefix enables the one-stage dialing. When users dial the gateway prefix and the telephone number, the gateway will automatically dial the number without the prefix. For example, the current gateway prefix is 123. If a user calls 075588290211, the user dials 123075588290211 on the IP phone.

C. Display Name
This parameter is used to display the name of the user who subscribes the H.323 service. For example, when you call your friend John Smith, your name will be displayed on your friend's telephone.

D. H.323 ID
H232 ID is used to verify the account. You can enter this parameter according to the requirements of the service provider.

E. Gatekeeper Address
This parameter is used to find the proper Gatekeeper. Enter the IP address of the Gatekeeper, such as 192.168.2.197, or domain name, such as gk.yourisp.com. If the soft-switch system uses the non-standard port (1719 and 1720), you can add the detailed port number at the end of the IP address or domain name of the Gatekeeper. For example, if the port number is 7300, the IP address is 192.168.2.197:7300 and the domain name is gk.youris p.com:7300.

Under the Gatekeeper mode, the GSM VOIP Gateway will forward all calls to the VoIP network to this address.

Note: The value of this parameter must be standard ASCII characters.

F. Enable VOS/AVS Encryption
You can enable the VOS/AVS encryption.

G. Enable Authentication on(Auth)
Click "Enable Authentication" and enter the following parameters when you need to set the H.235 authentication code and password.

3.5.1.3 Advance Setting of the H.232

The advance option of the GSM VOIP Gateway involves the signaling and media, corresponding to "Advance Setting" and "Media" respectively.

Select "Advance Setting" under H.323 and the following setting page is displayed:

C. STUN (RFC 3489)
The Simple Traversal of UDP over NAT (STUN) is a protocol that enables the SIP telephone to detect the existence and type of the firewall installed in the computer. This parameter indicates the SIP address of the STUN server.
Note: The STUN protocol supports the SIP gateway only.

D. Trunk Agent
The trunk agent protocol is a firewall traversal technology developed by Witura Corporation Sdn Bhd. It enables the products of by Witura Corporation Sdn Bhd to be applicable for most LANs. It involves the address, port, user name and password. The trunk agent protocol supports encryption on communications over the gateway. This feature needs the support of the server developed by Witura Corporation Sdn Bhd.
Media agent mode:

Mode 1: media encryption and agent (supported by all versions of relay servers).
Mode 2: media encryption and agent, supporting the transit over the specified port (supported by V2 relay server).
Mode 3: media encryption and agent for the conversion of RTP data to TCP packet, supporting the transit over the specified port (supported by the relay server later thanV2).

3.5.7 Dialing Rules

The GSM VOIP Gateway supports number dialing by rules. You can specify dialing rules in the dialing rule parameter of the "Call Forwarding".

3.5.7.1 Format Of Dialing Rules

1. You can specify multiple rules, which are separated by the delimiter '|'. For example, "00:-00|0:-0+86|:+86755".

2. The number is matched from the left of the dialing rules to the right. When the number matches the correct rule, the number stops matching. Otherwise, the number continues to match the next rule.

3. The rule format is "AA:-aa+bb", such as "0:-0+86". Where, "AA" indicates the number to match and "-aa+bb" indicates detailed actions to be taken on the number. If the number is successfully matched, "aa" is deducted and "bb" is

C. Trunk Agent

The trunk agent protocol is a firewall traversal technology developed by Witura Corporation Sdn Bhd. It enables the products of Witura Corporation Sdn Bhd to be applicable for most LANs. It involves the address, port, user name and password.

The trunk agent protocol supports encryption on communications over the gateway. This feature needs the support of the server developed by Witura Corporation Sdn Bhd.

3.5.6.3   Media NAT Traversal

The media NAT (firewall) traversal is classified into four types:

A. No firewall

The firewall traversal mechanism is not supported.

B. Port Transparency/DMZ

The port transparency is used to transfer the network port on the LAN interface to the computer or the server in the LAN. This feature enables external users (through the Internet in most cases) to share the services of internal servers (such as FTP, HTTP, and Telnet).

The port transparency supports the address of the gateway and response server. The gateway is a communication device that connects two different networks. The response server is a standard service device that implements the ECHO protocol.

A. RAS Port

The RAS is the communication protocol between the terminal and the Gatekeeper. It is used to transmit the registration information, login, broadband change, and the status between two H.323. The RAS port can be used to specify the UDP and used with the router port for mapping.

B. Call Signaling Port (Q.931 Port)

H.225-Q.931 is a call control protocol of the H.323 for transmits ting the call setting and unloading information between two H.323 units. It is used to specify the Q.931 port (TCP) that receives call s and used with the router port for mapping.

C. Media Control ol Port (H.245 Port)

H.245 is the media control protocol of the H.323. It is used to specify the port that receives the H.245 connection (TCP) and used with the router port for mapping.

D. Fast Start

You can enable or disable the fast start described in the H255.0 protocol. This parameter is used to detect and solve the compatibility problem. If you are not sure, do not set this parameter.

E. Fast Start Extension

This parameter is set for the special requirement of some customers. If you are not sure, do not set this parameter.

F. H245 Tunnel

This parameter is set for the special requirements of some customers. If you are not sure, do not set this parameter.

G. Registration Mode

This parameter is used to comply with different PBXs and is not set normally.

Register Mode	Register Multiple Nur ▼
	Register Multiple Numbe
DTMF Signaling	Register Multiple Times

H. DTMF Signals

DTMF signals are used to transmit call signals to the call switching center over the audio band. The DTMF means that two different frequencies of sounds are combined to 16 types of dialing tones. The telecom office or 1860 service hotline identify these dialing tones by analyzing the DSP, and thus determining the dialing number. There are two types of DTMFs: in-band DTMF and out-band DTMF.
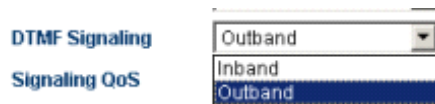
| DTMF Signaling	Outband ▼
Signaling QoS	Inband
	Outband

1) In-band DTMF

The in-band DTMF transmit s dialing tones and call voices together, without any processing. Therefore, the in-band DTMF transmits DTMF signals through a single way.

2) Out-band DTMF

The out-band DTMF transmits dialing tones over protocols, such as RFC2833, which can ensure the validity of the transmission.

I. Signaling QoS

Quality of Service (QoS) is a network's capacity to provide priority services, including the special bandwidth, jitter control and delay (used for real-time and interactive traffic), and improvement of the packet loss ratio. This parameter is used to mark the specified QoS label for the call signaling packet to increase the network service quality.

| Signaling NAT Traversal	Relay Proxy ▼
Address	
Port	
User	
Password	
	☐ Encryption
Relay Mode	1 ▼
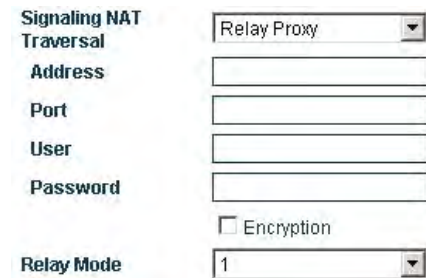
The trunk agent protocol supports encryption on communications over the gateway. The H323 trunk agent protocol supports encryption on signaling in different modes (for detail s about the agent mode, see section 3.5.6.3 "Media NAT Traversal").

Note: This feature needs the support of the server developed by Witura Corporation Sdn Bhd

3.5.6.2 Traversal Of SIP Signaling Over NAT

The traversal of SIP signaling over NAT (firewall) is classified into:

Signaling NAT Traversal	None ▼
	None
	STUN(RFC 3489)
RTP Port Range	Relay Proxy

A. No
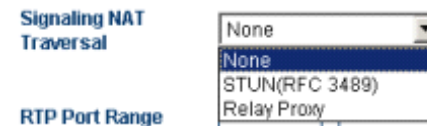
The mechanism of firewall traversal is not supported.

B. STUN (RFC 3489)

| Signaling NAT Traversal	STUN(RFC 3489) ▼
STUN Server	

The Simple Traversal of UDP over NAT (STUN) is a protocol that enables the SIP phone to detect the existence and type of the firewall installed in the computer. This parameter indicates the SIP address of the STUN server.

In the advance option of the call setting, the signaling and media have separate firewall setting, as shown in the following figures.

3.5.6.1 Traversal of H323 Signaling Over NAT

The traversal of H323 signaling over NAT (firewall) is classified into 4 categories:

A. No
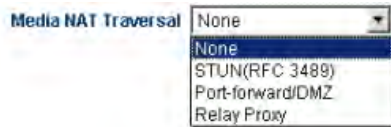
The mechanism of firewall traversal is not supported.

B. Nat Citron

The Citron is a special firewall traversal protocol for GnuGK and used with GnuGK.

C. Port Transparency/DMZ

The port transparency is used to transfer the network port on the LAN interface to the computer or the server in the LAN. This feature enables external users (through the Internet in most cases) to share the services of internal servers (such as FTP, HTTP, and Telnet).

The port transparency supports the address of the gateway and response server. The gateway is a communication device that connects two different networks. The response server is a standard service device that implements the ECHO protocol.

D. Trunk Agent

The trunk agent protocol is a firewall traversal technology developed by Witura Corporation Sdn Bhd. It enables the products of Witura Corporation Sdn Bhd to be applicable for most LANs. It involves the address of the trunk proxy server, port, user name and password.
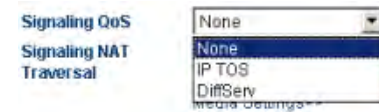
3.5.2 SIP Phone

The SIP (Session Initiation Protocol) is a simple network protocol that has less hierarchy and facilitates the initiation of calls among users. The call s may be conducted between two or more users, which include the sounds, images, session, interactive games, and virtual reality.

3.5.2.1 Setting Mode

The VoIP channel of the GSM VOIP Gateway can be set in the following three modes: single server, line setting, and trunk gateway.
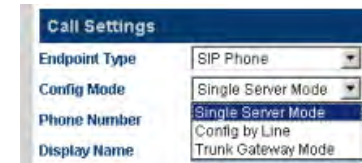

Figure 3-31 Setting Mode in the SIP Terminal

A) **Single server mode:** Multiple VoIP channels can share the same setting.
B) **Line setting:** Each VoIP channel can be served by different service providers or served by the same service provider. In the latter case, multiple different telephone numbers (accounts) can be registered on the same service, so that each telephone number is bound to the corresponding VoIP channel.
C) **Trunk Gateway:** This mode is used to establish the connection or channel between the soft-switch and the gateway to realize the transit between two ends.

3.5.2.2 Single Server Mode

The setting parameters relating to the SIP are as follows:

A) Telephone Number

This parameter is used to set the telephone number of the line. The telephone number is a unique ID when the gateway serves as the caller.

B) SIP Proxy Server

This parameter is used to set the address of the SIP proxy server. If the SIP proxy server uses the special port (other than the SIP default port: 5060), you can add the detailed port number at the end of the IP address or domain name of the proxy server. For example, 192.168.2.26:3000 or hy.con.com:3000.

C) SIP Registration Server

The SIP registration server is a server used by the gateway to register the account. This parameter is used to set the IP address or domain name of the SIP login server. If the SIP registration server uses a special port (other than the SIP default port: 5060), you can add the detailed port number at the end of the IP address or domain name of the registration server. For example, 192.168.2.26:3000 or hy.con.com:3000.

D) Outbound Proxy

The outbound proxy is mainly used in the scenarios where the firewall or NAT exists, so that the signaling and media stream can penetrate the firewall .

C. Jitter Delay Processing Mode

This parameter is used to specify the algorithm model of the jitter delay buffer. The "Adaptive" mode should be set. Other modes are only used for tests and should not be set in actual applications.

D. Media QoS

Quality of Service (QoS) is a network's capacity to provide priority services, including the special bandwidth, jitter control and delay (used for real-time and interactive traffic), and improvement of the packet loss ratio. This parameter is used to mark the specified QoS label for the voice packet to increase the network service quality.

Note: For details about media encryption and media NAT penetration, refer to

3.5.5   Voice Coding And Sequence

This parameter is used to modify the compression coding according to the requirements of the service provider.

If a compression coding is ticked, it indicates that the compression coding is available. The **UP** and **DOWN** are used to adjust the priority of the selected voice compression coding.

3.5.6   Firewall Penetration

2) Out-band DTMF

The out-band DTMF transmit s dialing tones over protocols, such as RFC2833 and SIP INFO, which can ensure the validity of the transmission.

F. Registration Mode

When the registration information is sent to the platform under Mode 1, "experise" info is included. When the registration information is sent to the platform under Mode 2, "experise" variable is not sent.

3.5.4   Media Advance Setting

The media advance setting is set for the RTP media stream of the gateway. Select "Call Setting" > "Media" and the following setting parameters are displayed:

A. RTP Port (Range)

This parameter is used to specify the UDP of the RTP and used with the router port for mapping. Note: The terminal will use multiple pairs of RTPs (depending on the number of lines that the terminal supports). The value of this parameter ranges from 5500 to 5520.

B. Packet Length (ms)

This parameter indicates the duration of sending a network packet. If this parameter is null, it indicates that the default value is 20ms. It is used to specify the size of the media packet. The unit is ms (the actual number of bytes depends on the compression algorithm).

E) Homing Domain

This parameter is used for the domain management host of the SIP (a host that provides the SIP service).

F) Authentication on ID

This parameter is used to set the authentication account when the gateway logs into the SIP proxy server.

G) Password

This parameter is used to set the authentication password when the gateway logs into the SIP proxy server.

H) Display Name

When you call your friend John Smith, your name will be displayed on your friend's telephone.

I) Backup Server

This parameter is used for registration backup. When a backup registration server exists in the user's system, the user can enable this parameter. Once the backup server is enabled, the gateway will automatically log into the backup server in case of the failure of the main server.

3.5.2.3   Setting by Line (Valid for the WT-2204 and WT-2208)

WT-2204 config by line

WT-2208 config by line

You need to set parameters for each line and the setting method is the same. The setting parameters are as follows:

A) Telephone Number

This parameter is used to set the telephone number of the line. The telephone number is an unique ID when the gateway serves as the caller.

C. Timeout Setting

D. Signaling QoS

Quality of Service (QoS) is a network's capacity to provide priority services, including the special bandwidth, jitter control and delay (used for real-time and interactive traffic), and improvement of the packet loss ratio. This parameter is used to mark the specified QoS label for the call signaling packet to increase the network service quality.

E. DTMF Signals

DTMF signals are used to transmit call signals to the call switching center over the audio band. The DTMF means that two different frequencies of sounds are combined into 16 types of dialing tones. The telecom office or 1860 service hotline identifies these dialing tones through analyzing the DSP, and thus determines the dialing number. There are two types of DTMF signals: in-band DTMF and out-band DTMF.

1) In-band DTMF

The in-band DTMF transmit s dialing tones and call voices together, without any processing. Therefore, the in-band DTMF transmits DTMF signals through a single way.

3.5.3   SIP Advance Setting

The advance setting of the SIP involves the signaling and media. Users can set according to their special requirements.

Select SIP Menu > Advance Setting/Media.

A. Signaling Port (SIP Local Port)
The SIP local port is the local UDP port. It is used for communications between the SIP agent and the SIP proxy server as well as other SIP managers.

B. NAT Hold
This parameter is used to hold the port that is activated by the NAT for SIP signaling communication. The unit of the parameter is m.

B) Gateway Prefix
The gateway prefix enables the connection of call s through a particular line. It can match the first digit only. You can set a gateway prefix for multiple lines. When you set a gateway prefix for multiple lines, the call s that have the same gateway prefix will select the line set with this gateway prefix. For example, the gateway prefix is 1. When the user dials 10086, the call will be connected by the line with the gateway prefix 1. When the user dials 075588290211, the system detects whether the line with gateway prefix 0 exists. If exists, the call will be connected. Otherwise, the call will be released.

Note: When you set the GSM VOIP Gateway by lines, the gateway prefix must be set. Otherwise, the call will not be connected.

C) SIP Proxy Server
This parameter is used to set the address of the SIP proxy server. If the SIP proxy server uses the special port (other than the SIP default port 5060), you can add the detailed port number at the end of the IP address or domain name of the proxy server. For example, 192.168.2.26:3000 or hy.con.com:3000.

D) SIP Registration Server
The SIP login server is a server that the gateway registers the account. This parameter is used to set the IP address or domain name of the SIP registration server. If the SIP registration server uses the special port (other than the SIP default port 5060), you can add the detailed port number at the end of the IP address or domain name of the registration server. For example, 192.168.2.26:3000 or hy.con.com:3000.

E) Outbound Proxy
The outbound proxy is mainly used in the scenarios where the firewall or NAT exists, so that the signaling and media stream can penetrate the firewall.

F) Homing Domain
This parameter is used for the domain management host of the SIP (a host that provides the SIP service).

G) Authentication ID
This parameter is used to set the authentication account when the gateway logs into the SIP proxy server.

H) Password

This parameter is used to set the authentication password when the gateway logs into the SIP proxy server.

I) Display Name

When you call your friend John Smith, your name will be displayed on your friend's telephone.

J) Backup Server

This parameter is used for registration backup. When a backup registration server exists in the user's system, the user can enable this parameter. When the backup server is enabled, the gateway will automatically log into the backup server in case of the failure of the main server.

Backup Server	⊙ Enable ○ Disable
Backup SIP Proxy	
Backup SIP Registrar Server	
Backup Home Domain	

3.5.2.4 Trunk Gateway Mode

Call Settings	
Endpoint Type	SIP Phone
Config Mode	Trunk Gateway Mode
SIP Trunk Gateway1	192.168.2.7
SIP Trunk Gateway2	192.168.2.X
SIP Trunk Gateway3	
Phone Number	
Register Expiry(s)	0
Authentication ID	
Password	

The trunk gateway is used to connect the VoIP network with the GSM network and convert the related protocols, so that users served by the two networks can call each other.

A) SIP Trunk Gateway1

It is the IP address of the server connected to the GSM VOIP Gateway. When the registration timeout is 0, the GSM VOIP Gateway is connected to the SIP server. If the registration timeout is not 0, the GSM VOIP Gateway logs into the SIP Trunk Gateway1 server through setting the telephone number, authentication ID and password.

B) SIP Trunk Gateway2

It is the IP address of the terminal connected to the GSM VOIP Gateway, which can be an IP segment, such as 192.168.2.X. This means that all terminals connected over 192.168.2 segment can log into the GSM VOIP Gateway and land through the direct connection between the GSM VOIP Gateway and the GSM network.

C) SIP Trunk Gateway3

It is the IP address of the server connected to the GSM VOIP Gateway.

D) Telephone Number

This parameter is used to set the telephone number of the line. The telephone number is an unique ID when the gateway serves as the callee and takes effect when the GSM VOIP Gateway logs into the SIP Trunk Gateway1.

E) Registration Timeout (s)

When the registration timeout is 0, you cannot register the gateway and the gateway will be connected to the server directly. You can reference the setting parameters of the single server mode to register the gateway.

F) Authentication ID

This parameter is used to set the authentication account when the gateway logs into the SIP Trunk Gateway1 proxy server. The parameter can be null in the case of interconnection.

G) Password

This parameter is used to set the authentication password when the gateway logs into the SIP Trunk Gateway1 proxy server. The parameter can be null in the case of interconnection.