Synway SMG Series Wireless Gateway

SMG4002

SMG4004

SMG4008

SMG4016

SMG4032

Wireless Gateway

User Manual

Version 3.0.0

Synway Information Engineering Co., Ltd

www.synway.net

Content

Copyright Declaration

Revision History

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Note: Please visit our website http://www.synway.net to obtain the latest version of this document.

Chapter 1 Product Introduction

Thank you for choosing Synway SMG Series Wireless Gateway!

The Synway SMG series wireless gateway products (hereinafter referred to as 'wireless gateway'), as a part of the Synway gateway products, works mainly for connecting the wireless network with the VoIP network. It adopts an updated VoIP processor and the wireless module, uses the push-pull SIM card socket for easy replacement of the SIM card, quite advanced in technology. So far, only SMG4008 is available.

See below table for the modules of SMG series wireless gateway:

Series	Module & Ports	Supported Frequency Band/Code
GSM Gateway	SMG4000-C16G	**GSM: 850/900/1800/1900MHz**
	SMG4000-C16G1	
	SMG4000-C32G	
	SMG4000-C32G1	
	SMG4004-4G	
	SMG4008-8G	
	SMG4016-16G	
	SMG4016-16G1	
	SMG4032-32G	
	SMG4032-32G1	
WCDMA Gateway	SMG4002-2W	**GSM: 900/1800MHz** **UMTS: 900/2100MHz**
	SMG4004-4W	
	SMG4008-8W	
	SMG4016-16W	
	SMG4016-16W1	
	SMG4032-32W1	

	SMG4002-2WA	**GSM: 850/900/1800/1900MHz** **UMTS: 850/1900MHz**
	SMG4004-4WA	
	SMG4008-8WA	
	SMG4016-16WA	
	SMG4032-32WA	
	SMG4002-2WT	**GSM: 850/900/1800/1900MHz** **UMTS: 850/2100MHz**
	SMG4004-4WT	
	SMG4008-8WT	
	SMG4016-16WT	
	SMG4032-32WT	
	SMG4002-2WZ	**GSM: 850/900/1800/1900MHz** **UMTS: 850/900/1900/2100MHz**
	SMG4004-4WZ	
	SMG4008-8WZ	
	SMG4016-16WZ	
	SMG4032-32WZ	
CDMA Gateway	SMG4004-4C	**CDMA: CDMA 2000 800MHz**
	SMG4008-8C	
	SMG4016-16C	
	SMG4032-32C	
LTE Gateway	SMG4002-2LE	**FDD LTE: B1/B3/B5/B7/B8/B20** **TDD LTE: B38/B40/B41** **WCDMA: B1/B5/B8** **GSM: B3/B8**
	SMG4004-4LE	
	SMG4008-8LE	

	SMG4016-16LE	
	SMG4032-32LE	
	SMG4000-C1LC	**FDD LTE: B1/B3 TDD LTE: B38/B39/B40/B41 TDSCDMA: B34/B39 WCDMA: B1 CDMA2000 1X/EVDO: BC0 GSM: 900/1800MHz**
	SMG4000-C4LC	
	SMG4000-C8LC	
	SMG4000-C16LC	
	SMG4000-C16LC1	
	SMG4000-C32LC	
	SMG4000-C32LC1	
	SMG4002-2LC	
	SMG4004-4LC	
	SMG4008-8LC	
	SMG4016-16LC	
	SMG4016-16LC1	
	SMG4016B-16LC	
	SMG4032-32LC	
	SMG4032-32LC1	
	SMG4032B-32LC	
	SMG4002-2LA	**TDD LTE: B40 FDD LTE: B1/B1/B3/B4/B5/B7/B8/B28 WCDMA: B5/B8/B2/B1 GSM: 850/900/1800/1900MHz**
	SMG4004-4LA	
	SMG4008-8LA	
	SMG4016-16LA	

	SMG4032-32LA	
	SMG4002-2LT	**TDD LTE: B40** **FDD LTE: B1/B1/B3/B4/B5/B7/B8/B28** **WCDMA: B5/B8/B2/B1** **GSM: 850/900/1800/1900MHz**
	SMG4004-4LT	
	SMG4008-8LT	
	SMG4016-16LT	
	SMG4032-32LT	
	SMG4002-2LV	**FDD LTE: B4/B13**
	SMG4004-4LV	
	SMG4008-8LV	
	SMG4016-16LV	
	SMG4032-32LV	

Table 1-1 Model List

1.1 Typical Application

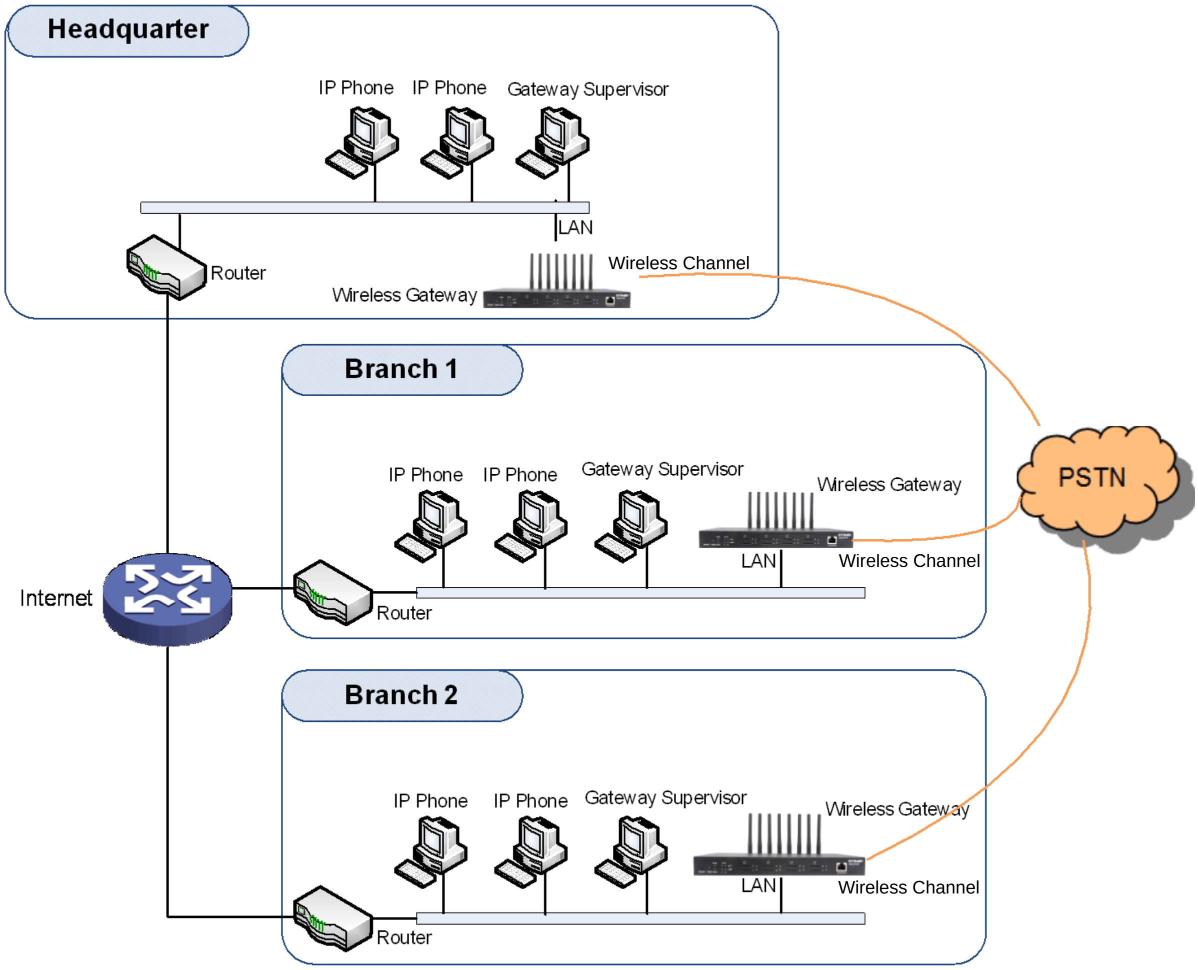


Figure 1-1 Typical Application

1.2 Feature List

Basic Features	Description
TDM Call	Call initiated from TDM to IP, via routing and number manipulation to obtain the called IP address.
IP Call	Call initiated from IP to TDM, via routing and number manipulation to obtain the call destination.
Number Manipulation	Peels off some digits of a phone number from left/right, or adds a prefix/suffix to a phone number.
Call Forward	Three options available: Unconditional, Busy, No Reply and Unreachable.
CID	Displays the CallerID.
Echo Cancellation	Provides the echo cancellation feature for a call conversation over the wireless port.

TDM/VoIP Routing	Sets a routing path: from IP to TDM or from TDM to IP.
Simultaneous Register to Multiple Servers	Registers the gateway to a master registrar server and a spare registrar server simultaneously.
IMS Network	Registers the gateway to a server under IMS network.
Custom IVR Recording	Provides the interface to customize the IVR Recording.
White/Black List	Allows the setting of the white/black list for WEB access.
Voice Gain Adjust	Supports the gain adjustment for the received or sent voice.
Receive or Send SMS/USSD	Supports the SMS sending and receiving, as well as the USSD request and response.
Auto Select Network	Supports the auto identification and selection of the network operator.
SMS CODEC	Two options available: ASCII and UCS2.
Signaling & Protocol	**Description**
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261.
Voice	CODEC: G.711A, G.711U, G.729A/B, G.723, G.722, AMR, iLBC DTMF Mode: RFC2833, SIP INFO, INBAND
Network	**Description**
Network Protocol	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN.
Static IP	IP address modification support.
DHCP	IP address dynamic allocation support.
DNS	Domain Name Service support.
Security	**Description**
Admin Authentication	Supports admin authentication to guarantee the resource and data security.
System Monitor	Monitors the running status of the system and the server.
Maintain & Upgrade	**Description**
WEB Configuration	Support of configurations through the WEB user interface.
Language	Chinese, English.
Software Upgrade	Support of user interface, gateway service, kernel and firmware upgrades based on WEB.
Tracking Test	Support of Ping and Tracert tests based on WEB.
SysLog Type	Three options available: ERROR, WARNING, INFO, DEBUG

1.3 Hardware Description

The wireless gateway supports two LANs and adopts an external 12V power supply. See below for product appearance.

Figure 1-2 SMG4008 Front View

Figure 1-3 SMG4008 Rear View

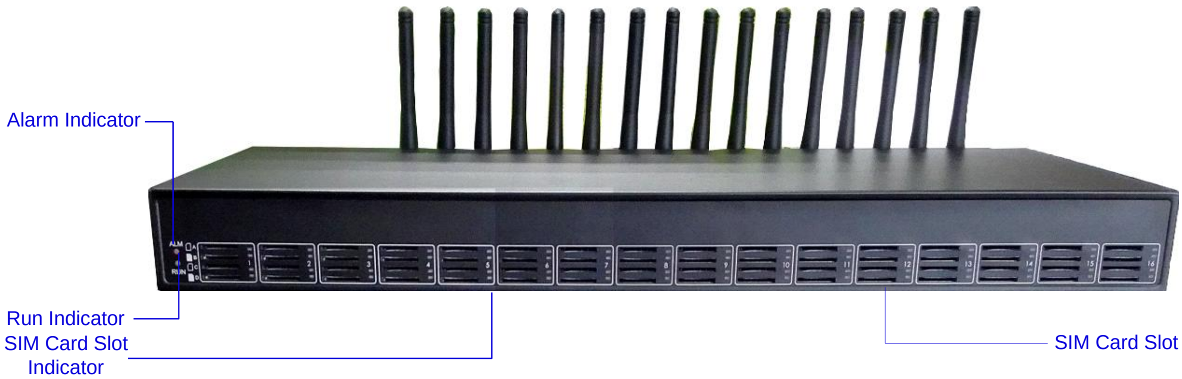


Figure 1-4 SMG4016 Front View

Figure 1-5 SMG4016 Rear View

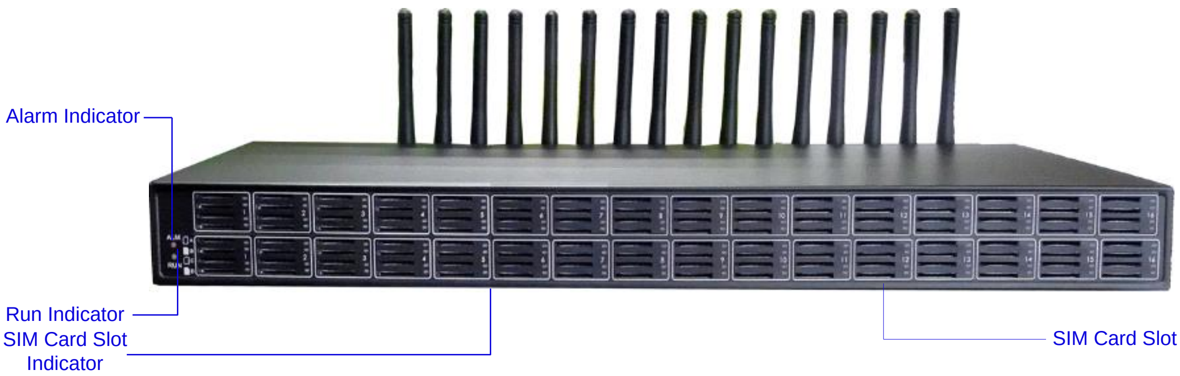


Figure 1-6 SMG4032 Front View

Figure 1-7 SMG4032 Rear View

The table below gives a detailed introduction to the interfaces, buttons and LEDs illustrated above:

Interface	Description
LAN	Amount: 2
	Type: RJ-45
	Bandwidth: 10/100 Mbps
	Self-Adaptive Bandwidth Supported
	Auto MDI/MDIX Supported
	Built-in Link indicator and ACTIVE indicator. For more details, refer to Indicator Info
SIM Card Slot	Amount: 4, 8, 16*4, 32*4
	Network Supported: GSM, WCDMA, CDMA, VoLTE

Console Port	Amount: 1
	Type: RS-232
	Baud Rate: 115200bps
	Connector: RJ45 to DB-9 Connector (4004, 4008 series), Mini-USB connecting line (4016, 4032 series)
	Data Bits: 8 bits
	Stop Bit: 1 bit
	Parity Unsupported
	Flow Control Unsupported
External Power Supply Interface	Provide the 12V voltage with positive inside and negative outside, and the current is larger than 3A
Button	**Description**
Reset Button	Restore the gateway to factory settings by pressing this button persistently for 3 seconds
LED	**Description**
Power Indicator	Indicates the power state. It lights up when the gateway starts up with the power cord well connected
Run Indicator	Indicates the running status. For more details, refer to Indicator Info.
Alarm Indicator	Alarms the device malfunction. For more details, refer to Indicator Info.
Link Indicator	The green LED on the right of LAN, indicating the network connection status.
ACT Indicator	The orange LED on the left of LAN, whose flashing tells the data are being transmitted.
Port Indicator	1. When the port is idle, the LED Lights up in green and keeps on; 2. When the port is unavailable, the LED Lights up in red and keeps on; 3. When the port is in use, the LED flashes in green 4. When the port module is disabled, the LED flashes in red 5. For SMG4016 series, only the indicator of the card slot in which the SIM card is in using lights up and other indicators will go out in the case that there are more than one SIM cards inserted in the same channel.

For other hardware parameters, refer to Appendix A Technical Specifications.

1.4 Indicator Info

The wireless gateway is equipped with two indicators denoting the system's running status: Run Indicator (green LED) and Alarm Indicator (red LED). The table below explains the states and meanings of the two indicators.

LED	State	Description
Run Indicator	Go out	System is not yet started.
	Light up and flash fast	System is starting.
	Flash slowly	Device is normal.
Alarm Indicator	Go out	Device is normal.
	Light up	Upon startup: Device is normal. In runtime: Device is abnormal.
	Flash	Device is abnormal.

Note:

- The startup process consists of two stages: System Booting and Gateway Service Startup. The system booting costs about 1 minute and once it succeeds, both the run indicator and the alarm indicator light up. Then after the gateway service is successfully started and the device begins to work normally, the run indicator flashes and the alarm indicator goes out.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Go to Appendix D Technical/sales Support to find the contact way.

Chapter 2 Quick Guide

This chapter is intended to help you grasp the basic operations of the wireless gateway in the shortest time.

Step 1: Confirm that your packing box contains all the following things.

- Wireless Gateway *1
- External 12V Power Adapter *1
- GSM/WCDMA/CDMA/LTE Rubber Antenna *2/4/8/16/32
- Standard RJ45 to DB-9 Switcher (4002/4004/4008 series) *1, Mini-USB connecting line (4016/4032 series) *1
- 8mm Antenna Wrench *1
- Rubber Foot Pad *4
- Network Cable *1
- Warranty Card *1
- Installation Manual *1

Step 2: Connect the network cable.

This product provides RJ-45 interfaces.

Step 3: Insert the SIM card (standard size) and install the antenna.

The wireless gateway provides a SIM card slot. You are required to insert the SIM card before using it. Take out the rubber antennae from the packing box, install them onto the wireless gateway, screw them up and evenly arrange them.

Step 4: Power on and start the gateway.

To use the wireless gateway, you need an external power supply. Insert it to the power interface of the wireless gateway and power it on with 100~240V AC. See the figure below:

Figure 2-1 Wireless Gateway Power Connection

Step 5: Log in the gateway.

Enter the original IP address (192.168.1.101) of the wireless gateway in the browser to go to the WEB interface of the gateway. The original username and password of the gateway are both 'admin'. For detailed instructions about login, refer to System Login. We suggest you change the initial username and password via 'System Tools → Change Password' on the WEB interface as soon as possible after your first login. For detailed instructions about changing the password, refer to Change Password. After changing the password, you are required to log in again.

Step 6: Modify IP address of the gateway.

You can modify the IP address of the gateway via 'Advanced Settings → Network' on the WEB interface to put it within your company's LAN. Refer to Network for detailed instructions about IP modification. After changing the IP address, you shall log in the gateway again using your new IP address.

Step 7: Make phone calls.

Note: For your easy understanding and manipulation, all examples given in this step do not involve registration, that is, SIP initiates calls in a point-to-point mode.

Situation 1: Call from a station to an IP phone (Tel→IP)

1. Go to 'Advanced Settings → Dialing Rule' on the WEB interface and click the 'Add New' button to add a new dialing rule. Refer to Dialing Rule for detailed instructions. Enter either a particular number or a string of 'x's to represent several random numbers. For example, 'xxx' denotes 3 random numbers. You may use the default value of 'Index' and are required not to leave 'Description' empty.

   Example: Set **Index** to **99**, fill in **Description** with **test** and configure **Dial Rule** to **123**.

2. Go to 'Port Settings → Port Group' on the WEB interface and click the 'Add New' button to create a new port group and add the corresponding ports to it. Refer to Port Group for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

   Example: Provided the added port is Port1, check the checkbox before **Port1**, set **Index** to **1**, fill in **Description** with **test**, and keep the default values of other configuration items.

3. Go to 'Route Settings → Tel→IP' on the WEB interface and click the 'Add New' button to add a new routing rule. Refer to Tel→IP for detailed instructions. Select the port group created in Step2 as 'Source Port Group' and fill in 'Destination IP' and 'Destination Port' with the IP address and the Port number you plan to call. You may use the default values of other configuration items and are required not to leave 'Description' empty.

   Example: Provided the remote IP address intended to call is 192.168.0.111 and the port is 5060. Set **Index** to **63**, **Source Port Group** to **1**, fill in **Description** with **test**, configure **Destination IP** to **192.168.0.111**, **Destination Port** to **5060**, and keep the default values of other configuration items.

4. Use an external phone to call the number of this SIM card, and then follow the cue tone to dial the number set in Step1 to ring the remote IP phone If you have set a particular number in Step 1, only this number you can dial; if you have set a string of 'x's, how many 'x's there are, how many random numbers you can dial.

   Example: The external phone dials the number of this SIM card, and then follows the cue tone to dial 123. Then the IP phone with the IP address 192.168.0.111 and the port 5060 will ring.

Situation 2: Call from an IP phone to a station (IP →Tel)

1. Go to 'Port Settings → Port Group' on the WEB interface and click the 'Add New' button to create a new port group and add the corresponding ports which are connected with stations to it. Refer to Port Group for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

Example: Provided the added port is Port1, check the checkbox before **Port1**, set **Index** to **1**, fill in **Description** with **test**, and keep the default values of other configuration items.

2. Go to 'Route Settings → IP→Tel/IP' on the WEB interface and click the 'Add New' button to add a new routing rule. Refer to IP→Tel/IP for detailed instructions. Fill in 'Source IP' with the IP address which initiates the call and select the port group created in Step1 as 'Destination Port Group'. You may use the default values of other configuration items and required not to leave 'Description' empty.

   Example: Provided the IP address of the IP phone which initiates the call is 192.168.0.111. Set **Index** to **63**, **Destination Port Group** to **1**, fill in **Description** with **test**, configure **Source IP** to **192.168.0.111**, and keep the default values of other configuration items.

3. Pick up the IP phone and call the IP address and port of the wireless gateway to make outgoing calls from the wireless channel.

   Example: Provided the IP address of the wireless gateway is 192.168.0.101, the port is 5060, use the IP phone to call the IP address 13529101232@192.168.0.101 and then the first idle wireless port in the port group of step 2 will make an outgoing call to 13529101232.

Special Instructions:

- As the device will gradually heat up while being used, please maintain good ventilation to prevent sudden failure, ensuring that the ventilation holes are never jammed.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Otherwise it may lead to a drop in performance or unexpected errors.

Chapter 3 WEB Configuration

3.1 System Login

Type the IP address into the browser and enter the login interface. See Figure 3-1.

Figure 3-1 Login Interface

The gateway only serves one user. After your first login by using the original username and password both of 'admin', the following dialog will pop up. You can change the username and the password via 'System Tools → Change Password' on the WEB interface. For detailed instructions, refer to Change Password.

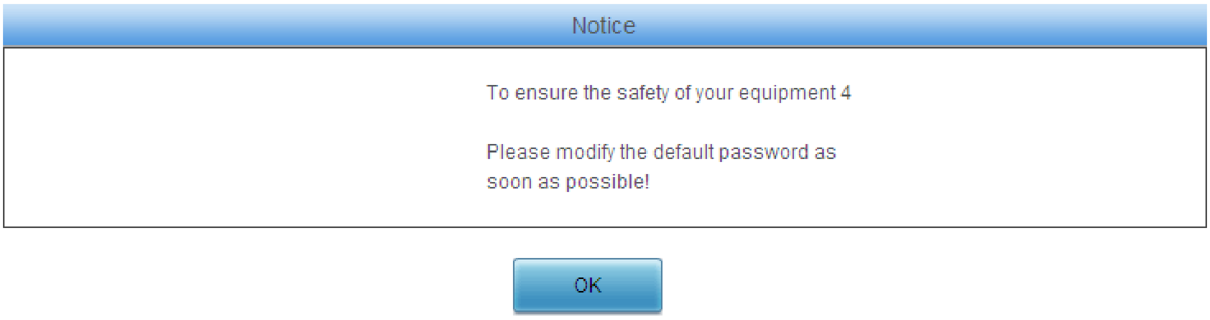


Figure 3-2 Notice Pop-up

Click OK to log in and you can see the System Info interface.

3.2 Operation Info

Operation Info includes four parts: ***System Info***, ***Port State***, ***Call Count*** and ***SIP Message Count***, showing the current running status of the gateway. You can press the following button to unfold or fold them.

Figure 3-3 Button to Unfold/fold

3.2.1 System Info

On the system info interface, you can click ***Refresh*** to obtain the latest system information. See the table below for details.

Item	Description
MAC Address	MAC address of LAN.
IP Address	The three parameters from left to right are IP address, subnet mask and default gateway of LAN.
DNS Server	DNS server address of LAN.
Receive Packets	The amount of receive packets after the gateway's startup, including three options: All, Error and Drop.
Transmit Packets	The amount of transmit packets after the gateway's startup, including three options: All, Error and Drop.
Current Speed	Show the current speed of data receiving and transmitting.
Work Mode	Show the work mode of the network, including four modes: 10 Mbps Half Duplex, 10 Mbps Full Duplex, 100 Mbps Half Duplex, 100 Mbps Full Duplex.
Runtime	Time of the gateway keeping running normally after startup, which will be automatically updated.
WEB	Current version of the WEB interface.
Gateway	Current version of the gateway service.
Serial Num	Unique serial number of a wireless gateway.
Authorization Code	The authorization codes vary from different wireless modules.
FPGA	Current version of FPGA.
U-boot	Current version of Uboot.
Kernel	Current version of the system kernel on the gateway.
Device Type	Type of the wireless gateway.

3.2.2 Port State

The Port State interface shows such information as the port type, the port state, etc. See the table below for details.

Item	Description
Port	Port number on the device.
Type	Port type on the device. So far, only GSM, WCDMA, CDMA and LTE types are supported.
State	Displays the port state in real time. You can move the mouse onto the port state icon for detailed state information.

State	Icon	Description
Idle		The port is available.
Off-hook		The port picks up the call.
Wait Answer		The port receives the ringback tone and is waiting for the called party to pick up the phone.
Ringing		The port is in the ringing state.

Talking		The port is in a conversation.
Dialing		The port is dialing.
Pending		The port is in the pending state.
Internal State		Internal state of the port.
Unusable		The port is unavailable. And the exact cause is shown in the following brackets: Module Detecting…, Module Starting…, SIM Card Detecting…, BS Connecting…, Module Disabled, SIM Card Lost, or Module Resetting….

Net Type	Displays the voice type of the current call. **Note:** For the LTE series gateway, it is Net type and will display the network type of the current call.
Direction	Displays the direction of the call on port.
CallerID	Displays the CallerID of the call on port.
CalleeID	Displays the CalleeID of the call on port.
SIM Card	Displays the real-time state of the SIM card. Move the mouse onto the corresponding icon and you can find the exact state of the SIM card. means card inserted, means no card inserted, means card in use. Note: This item is unavailable for SMG4002, SMG4004 and SMG4008 series, as well as G1, LC1 and W1 series.
Cell Phone No.	Displays the number of the corresponding channel set in Wireless Parameters.
Connection	Displays the connection status between the SIM card and the base station.
Signal	Displays the signal intensity of the wireless module. The number in the following brackets is the decibel value.
SIP Reg Status	Displays the registration status of the port.
Duration	The actual time length of each outgoing call from the gateway.
Operator	The wireless communication operator for the module, which is acquired automatically. Note: This configuration item is unsupported for the CDMA module.

3.2.3 Call Count

The Call Count Interface shows the detailed information about all the calls counted from the startup of the gateway service to the latest open or refresh of this interface. You can click ***Refresh*** to obtain the current call count information. See the table below for details.

Item	Description
Call Direction	A condition for call count, two options available: *IP→Tel* and *Tel→IP*.
Total Calls	Total number of calls in a specified call direction.
Successful Calls	Total number of successful calls in conversation.
Busy	Total number of calls which fail as the called party has been occupied and replies a busy message.
No Answer	Total number of calls which fail as the called party does not pick up the call in a long time or the calling party hangs up the call before the called party picks it up.
Routing Failure	Total number of calls which fail because no routing rules are matched.

Dialing Failure	Total number of calls which fail as the called party number does not conform to the dialing rule or due to dialing timeout.
Unknown Failure	Total number of calls which fail due to unknown reasons.
Total Calls	The total numbers of the outgoing calls.
Remote Ringing	The count of the calls which bring the remote terminal into the ringing state.
Talking Count	The count of the outgoing calls which are answered by remote terminal.
Failure Count	The count of the failure calls, i.e. the counts of the calls which cannot be made out by the port.
Continuous Failure	The count of the calls which failed continuously twice or more.
Call Completion Rate	The percentage of successful calls to total calls.
Accumulated Time	The total time of the calls which are answered by the remote terminal.
Average Time	The average time length of each call answered by the remote terminal.

3.2.4 SIP Message Count

The SIP Message Count interface. records the amount of the normal SIP messages that are sent/received or repeatedly sent/received during the period from the startup of the gateway service to the latest open or refresh of the interface. Click ***Refresh*** to refresh the count of SIP messages, or click ***Clear*** to clear the current count of SIP messages.

3.3 Quick Config

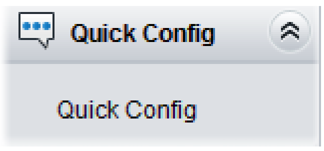


Figure 3-4 Quick Config Interface

See Figure 3-4 for the Quick Config interface. Follow the gateway Quick Configuration wizard and you can easily complete the settings on network, SIP and Port. The gateway can work normally after configuration.

Refer to Network for detailed settings. After configuration, click ***Next*** to enter the SIP Settings interface.

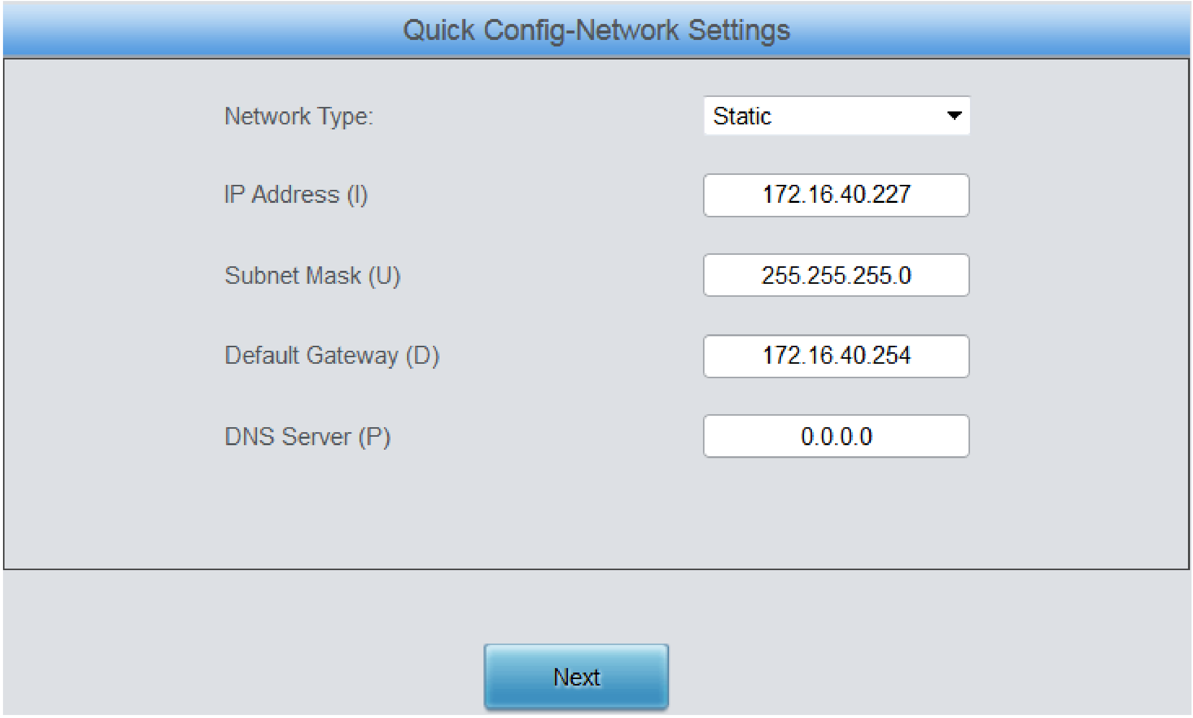


Figure 3-5 Quick Config-Network Settings Interface

The configuration items on the Quick Config-SIP Settings interface are the same as those on the SIP interface. Refer to SIP for detailed settings. You are required to fill with the information about the registrar if the gateway must be registered. After configuration, click ***Back*** to go back to the Network Settings interface; click ***Next*** to enter the Port Settings interface.

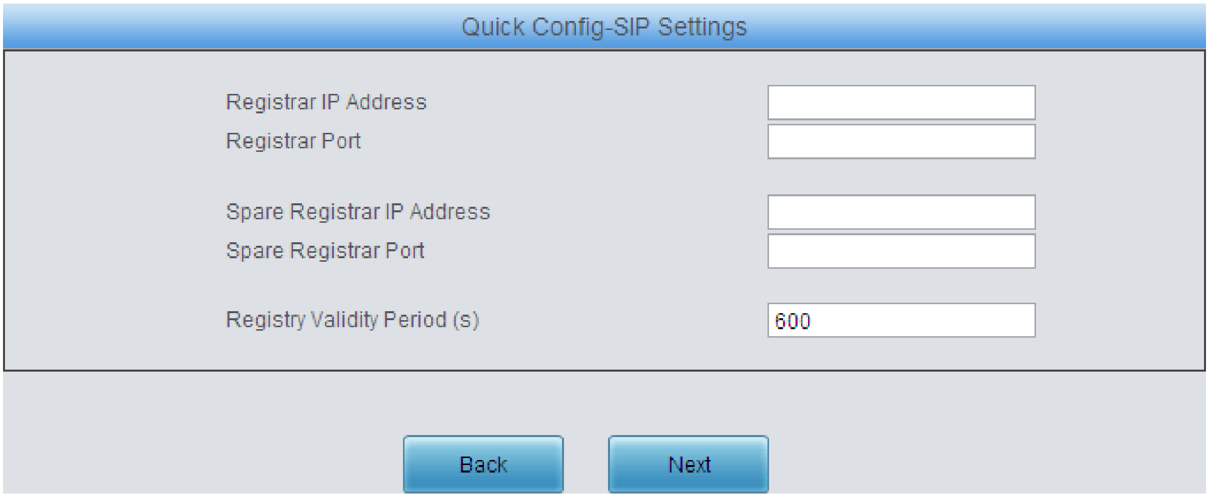
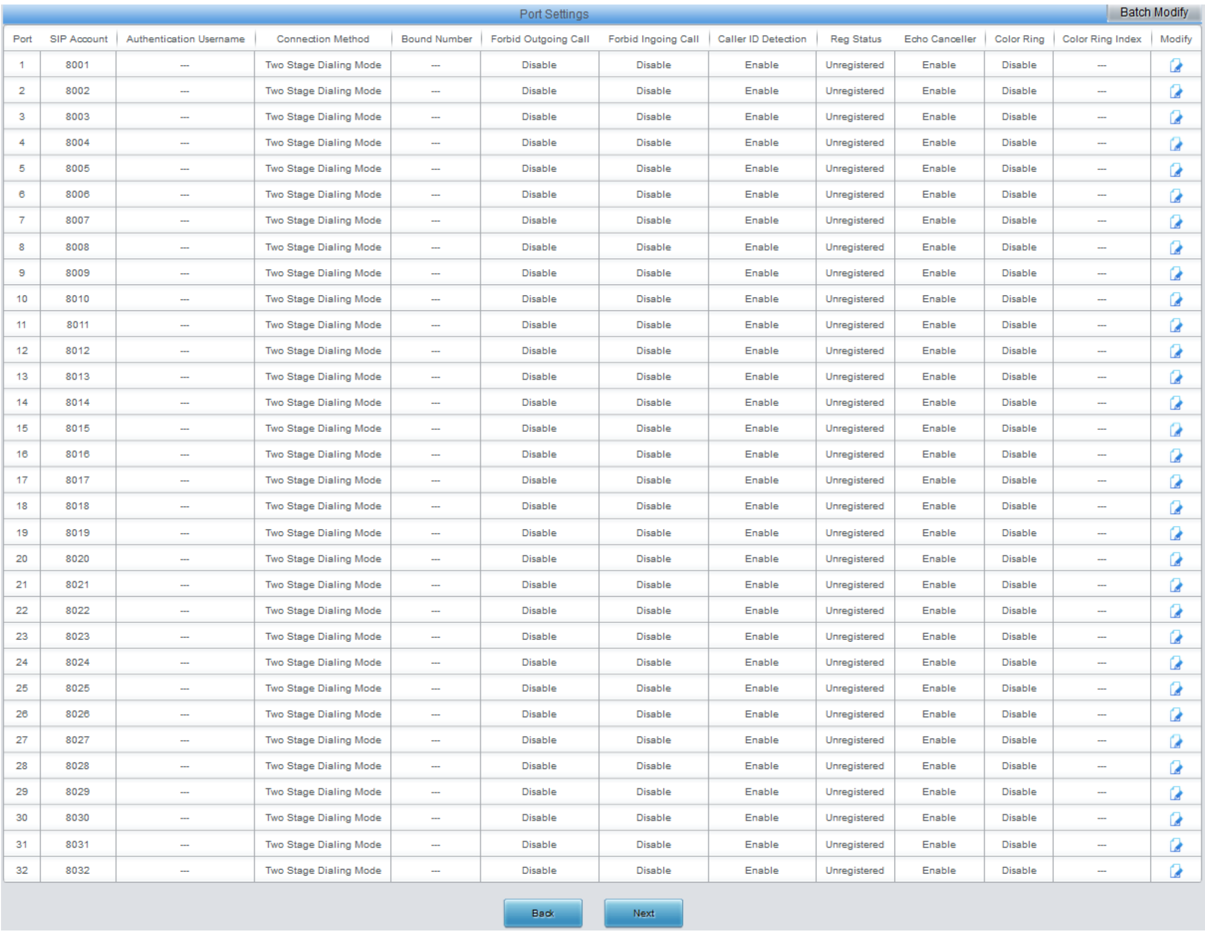


Figure 3-6 Quick Config-SIP Settings Interface

The configuration items on the Port Settings interface are the same as those on the Port interface. Refer to Port for detailed settings. After configuration, click ***Back*** to go back to the SIP Settings interface; click ***Next*** to enter the Quick Config-Completion interface.



Port Settings | Batch Modify

Port	SIP Account	Authentication Username	Connection Method	Bound Number	Forbid Outgoing Call	Forbid Ingoing Call	Caller ID Detection	Reg Status	Echo Canceller	Color Ring	Color Ring Index	Modify
1	8001	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
2	8002	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
3	8003	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
4	8004	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
5	8005	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
6	8006	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
7	8007	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
8	8008	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
9	8009	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
10	8010	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
11	8011	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
12	8012	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
13	8013	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
14	8014	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
15	8015	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
16	8016	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
17	8017	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
18	8018	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
19	8019	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
20	8020	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
21	8021	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
22	8022	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
23	8023	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
24	8024	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
25	8025	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
26	8026	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
27	8027	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
28	8028	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
29	8029	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
30	8030	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
31	8031	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	
32	8032	---	Two Stage Dialing Mode	---	Disable	Disable	Enable	Unregistered	Enable	Disable	---	

Back | Next

Figure 3-7 Port Settings Interface

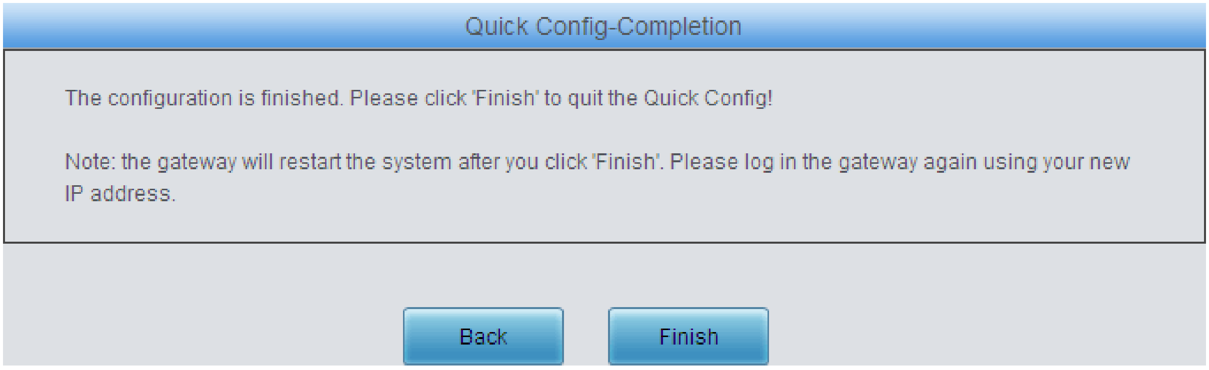


Figure 3-8 Quick Config-Completion Interface

Click ***Back*** to go back to the Port Settings interface; click ***Finish*** to finish the Quick Config wizard and now the gateway can work normally with basic configuration.

3.4 VoIP Settings

VoIP Settings includes six parts: ***SIP***, ***SIP Compatibility***, ***SIP Station***, **SIP Server**, ***NAT Setting*** and ***Media***. ***SIP Settings*** is used to configure the general SIP parameters, ***SIP Compatibility*** is used to set which SIP servers and SIP messages will the gateway be compatible with, ***SIP Station*** is to set the basic information of the SIP station, ***SIP Server*** is to set the basic information of the SIP server, ***NAT Setting*** is used to configure the parameters for NAT, and ***Media Settings*** is to set the RTP port and the payload type.

3.4.1 SIP

On the SIP Settings interface you can configure the general SIP parameters. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to Restart for detailed instructions. See the table below for details.

Item	Description
SIP Port	Monitoring port of SIP signaling. The value range of it must be greater than 1024 and less than 65535, with the default value of 5060.
Send 180	Sets whether to send the 180 message to respond to the ringing tone when the SIP end serves as the called party.
Called Number Prefix for 180 Reply	Once the feature "Send 180" is enabled, the gateway will reply the 180 message to those calls which have the calleeID with the designated prefix; otherwise, it will reply the 183 message. By default, the value is null, that is, replying the 183 message to all calls. Note: The prefix of -1 means to only reply 180.
Register Status	Registration status of the gateway. When ***Register Gateway*** is set to *No*, the value of this item is *Unregistered*; when ***Register Gateway*** is set to *Yes*, the value of this item is either *Failed* or *Registered*.
Register Gateway	Sets whether to register the gateway as a whole. The default value is *No*. Only when this configuration is set to *Yes* can you see the configuration items **SIP Account** and **Password**.
SIP Account	When the gateway initiates a call to SIP, this item corresponds to the username of SIP.
Password	Registration password of the gateway. To register the gateway to SIP, both configuration items ***SIP Account*** and ***Password*** should be filled in.
Authentication Username	Authentication username for registration.
Registrar IP Address	Address of the registry server for the gateway to register.
Registrar Port	Signaling port of the registry server.
Spare Registrar Server	Check the enable checkbox to enable the spare registrar server. By default, it is ***disabled***.
Spare Registrar IP Address	Address of the spare registry server for the gateway to register. The gateway will enable the spare registrar server if the master registrar server has no reply, or the master server is detected with no response in case the item **Detection Server Cycle** is enabled.
Spare Registrar Port	Signaling port of the spare registry server.
Registry Validity Period	Validity period of the SIP registry. Once the registry is overdue, the gateway should be registered again. This configuration item is valid only when ***Register Gateway*** is set to *Yes*. Range of value: 10~3600, calculated by s, with the default value of 600.
Multi-Registrar Server Mode	Tick the checkbox before to enable the multi-registrar server mode. By default, it is *disabled*.
SIP Transport Protocol	There are two modes *UDP* and *TCP* available for running the SIP protocol. The default value is *UDP*.

Switch Signal Port if SIP Registration Failed	If the SIP registration fails, the SIP signaling port N will switch to N+1 for a new registration. It will continue until the registration succeeds. By default, it is disabled.
IMS Network	Once this feature is enabled, the gateway will send signaling messages to the corresponding externally bound address and port when it registers to the server. By default, this feature is *disabled*. Only when this feature is *enabled* will these items ***Externally Bound Address***, ***Externally Bound Port*** and ***Authentication Username*** be shown.
Externally Bound Address	Externally bound IP address for registration.
Externally Bound Port	Externally bound port for registration.

3.4.2 SIP Compatibility

On the SIP Compatibility interface you can configure the SIP parameters to determine which SIP servers and SIP messages will the gateway be compatible with. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations.

See the table below for details.

Item	Description
Obtain CalleeID from	There are three optional ways to obtain the called party number: *Username of To Field*, *Displayname of To Field* and "*Request" Field*. The default value is "*Request" Field*.
Set CallerID Position	There are two options to set the position of the calling party number: "Displayname of From Field" and "Username of From Field". The default value is "*Username of From Field*".
Obtain CallerID from	There are two optional ways to obtain the calling party number: from "Displayname of From Field" and from "Username of From Field". The default value is "*Username of From Field*".
Use Contact Address	Sets whether to send the request message according to the content of Contact, with the default setting of *disabled*. As it is disabled, if the Contact field indicates an IP address within the LAN, the request message will be sent according to the source address; if the Contact field indicates an IP address belonging to the WAN, the request message will be sent according to this IP address.
Reply To Source Address	Once this feature is enabled, the gateway will reply the source address in the invite message. As the item ***Use Contact Address*** conflicted with this item, you may now shield the other one while enabling one of them.
Bye Message has Contact	Sets whether the Bye message sent out by the gateway carries the Contact field. It is enabled by default.
Two Stage Dialing for SIP Incoming Call	Once this feature is enabled, the incoming call from SIP should perform the two stage dialing operation. By default this feature is disabled.
Maximum Wait	Sets the maximum time for the SIP channel to wait for the answer from the called

Answer Time	party of the outgoing call it initiates. If the call is not answered within the specified time period, it will be canceled by the channel automatically. The default value is 60, calculated by s.
SIP Station Supported	Once this feature is enabled, a SIP terminal can be registered to the gateway to become a SIP station. By default this feature is disabled.
Set SIP Identifying	Sets the SIP identifying content in the SIP call message. The default setting is *Gateway*.
Maximum Wait RTP Time	Sets the maximum time for the SIP channel to wait for the RTP packet. If no RTP packet is received within the specified time period, the channel will enter the pending state automatically and release the call. The default value is *0（disabled）*, calculated by s.
Ignore ACK	Once this feature is enabled, it is not necessary for the gateway to wait for the ACK message after sending the 200OK message to establish a call. By default it is *disabled*.
Iptable for SIP Calls	Only some special SIP messages, which can be configured by users, are allowed to send to the gateway.
SIP Encryption	Once this feature is enabled, you can encrypt the SIP signal following selecting an encryption criterion and setting a key. By default it is *disabled*.
Encryption Criterion	The criterion used to encrypt the SIP signal. At present only VOS1.1 is supported.
Identifier	The identifier field of the VOS encryption, which is used to obtain the key of the SIP encryption.
Key	The key to encrypt the SIP signal.
RTP Encryption	Once this feature is enabled, you can encrypt the RTP package. By default it is *disabled*.
Reply 407 at SIP Registration Password Error	When this feature is enabled, the gateway will reply the 407 message if the registration password is incorrect. It is disabled by default and the gateway replies the 403 message.
Abnormal Call Hangup Detection	Sets the interval between checks of the remote end's abnormal hangup, with the default value of 0 (feature disabled), calculated by s. It is suggested to set to 10s if this feature is necessary to be used.
Server Status Detection	The interval of sending a heartbeat packet to detect the master registrar server status, with the default value of 0 (feature disabled), calculated by s. It is suggested to set to 15s if this feature is necessary to be used.
Available Port Negotiation	When this feature is enabled, the gateway will send messages to the preset negotiation server (e.g. VOS server) to let it know the number of available ports on the gateway. By default this feature is disabled.
Cycle, Address	Cycle means how soon will the gateway send a message; Address indicates the server address (e.g. VOS server).
Port&Group Mode	When this feature is enabled, the port and port group are registered to the set negotiation server (such as the VOS server), and the publish message is sent. The registered account is placed in the *Username of From field*. If the capacity value of the publish message for port registration is 0, it indicates the port is not idle, and if that is 1, it indicates the port is idle. The capacity value of the publish message for

	port group registration represents the number of idle ports in the port group. The default setting is no.
Custom SIP Extended Header Field	The 183 messages replied by the gateway can optionally contain one of the following four header fields.
Carrying IMEI	Add a custom header field X-IMEI to the 183 message replied by the gateway.
Carrying IMSI	Add a custom header field X-IMSI to the 183 message replied by the gateway.
Carrying Port No.	Add the custom header field X-PORT-NO to the 183 message replied by the gateway.
Carrying Phone No.	Add the custom header field X-PHONE-NO to the 183 message replied by the gateway.
Occasion to Reply 183	Sets the occasion to reply the 183 message. Two options are available: Immediately and After ringing, with the default value of *Immediately*.
Occasion to Reply 200 Ok	Sets the occasion to reply 200 OK. Three options are available: Immediately, After pickup and After ringing, with the default value of *After pickup*.
Delays	Sets the time of delay to reply the 200OK message, with the default value of 0.

3.4.3 SIP Station

A SIP terminal can be registered to the gateway to become a SIP station. Tick the option of '***SIP Station Supported***' on SIP Compatibility interface, and you will see the item SIP Station on the VoIP Settings menu. Click '***SIP Station***' to go into the SIP Station interface. By default, there is no available SIP station.

Click 

to go into the SIP Station interface. You can configure basic SIP station information on this interface. The bound port to a SIP station must be a wireless port and unique. The username must be the same as that used to register the SIP terminal to the gateway.

See the table below for details:

Item	Description
Number	The logical number for a SIP station to register to the gateway.
Username	The username used to register a SIP station to the gateway.
Password	The password used to register a SIP station to the gateway.
Bound	Two options available: *Port* and *Port Group*.
Bound Port Group	The port group which is bound to the SIP station, That is, all the ports in this group are bound to the SIP station.
Bound Port	The wireless port which is bound to the SIP station.
Description	It is user-defined, with the default value of *default*.
Batch Setting	Used to set multiple SIP stations at the same time.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings. See Figure 3-9 for the applied SIP station information.

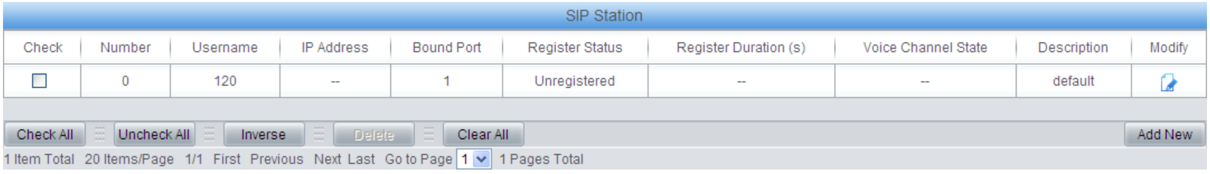


Figure 3-9 SIP Station Interface

Click ***Modify*** in the above figure to modify the configuration of the SIP station.

To delete a SIP station, check the checkbox before the corresponding index in Figure 3-9 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all SIP stations at a time, click the ***Clear All*** button in Figure 3-9.

3.4.4 SIP Server

The gateway supports the multi-registrar server feature. Enable the feature of '***Multi-Registrar Server Mode***' on the SIP interface (see 3.4.1 SIP) and you will see the item SIP Server under the VoIP Settings menu. Click '***SIP Server***' to go into the SIP Server interface. By default, there is no available SIP server.

Click 

to add SIP servers manually. Up to 32 are supported to add. You can configure basic SIP server information on this interface.

All the items except Index and Description are the same as those on the SIP interface (SIP).

Item	Description
Index	The index of each SIP server. The gateway supports up to 32 SIP servers.
Description	More information about each SIP server, with the default value of *default*.

After configuration, click ***Save*** to save the settings into the gateway or click ***Cancel*** to cancel the settings. See Figure 3-10 for the SIP server management interface.

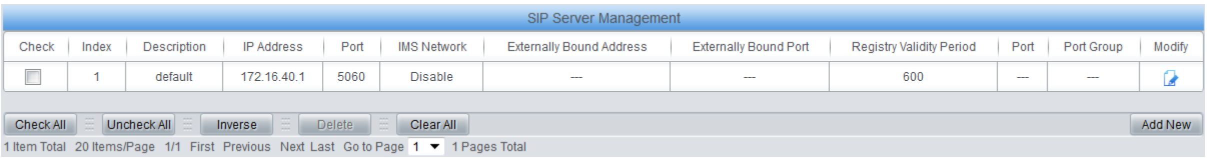


Figure 3-10 SIP Server Management

Click ***Modify*** in the above figure to modify the configuration of the SIP server.

To delete a SIP server, check the checkbox before the corresponding index in Figure 3-10 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all SIP servers at a time, click the ***Clear All*** button in Figure 3-10.

3.4.5 NAT Setting

On the NAT Setting interface you can configure the parameters for NAT. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations.

The table below explains the items shown on the interface.

Item	Description
STUN Server	Sets whether to enable the STUN server for NAT traversal. By default the STUN

	server is disabled.
NAT Type	Detected NAT (Network Address Translation) type. The gateway will return the NAT type automatically in case ***STUN Server*** is enabled. It includes 9 types: unknown; no NAT; ConeNat; RestrictedNat; PortRestrictedNat; Symmetric NAT; Symmetric NAT with firewall; can't detect over (fail to send detect message) and fail to detect (No reply from the stun server).
STUN Server Address	Address of the server for STUN traversal.
Mapping Address	It should be filled in when there exists NAT or other mapping relationships which leads to the failure of direct communication between the gateway and the destination address, so as to ask the remote end to send signaling messages or voice data to it during the signaling or voice communication between the gateway and the destination. **Note:** Once this item is filled out, it will be used as the first choice even if Rport and NAT IP are enabled.
RTP Self-adaption	When this feature is enabled, the RTP reception address or port carried by the signaling message from the remote end, if not consistent with the actual state, will be updated to the actual RTP reception address or port. By default, this feature is *disabled*.
Rport	When this feature is enabled, a corresponding Rport field will be added to the Via message of SIP. The default value is *enabled*.
Learn NAT	You should enable this feature at the same time you enable the Rport feature, and learn the Rport address to the Contact address. Otherwise, only the Rport field will be carried and the Contact address will not be processed.
Auto Detect NAT IP	When this feature is enabled, the gateway will parse the corresponding address and port in the message returned by Rport so as to use them for the following communication. By default, this feature is *disabled*. **Note:** This feature gets valid only when Rport is enabled.

3.4.6 Media

On the Media Settings interface you can configure the RTP port and payload type depending on your requirements. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to Restart for detailed instructions. The table below explains the items shown on the interface.

Item	Description
DTMF Transmit Mode	Sets the transmit mode for the IP channel to send DTMF signals. The optional values are *RFC2833*, *In-band* and *Signaling*, with the default value of *RFC2833*.
RFC2833 Payload	Payload of the RFC2833 formatted DTMF signals on the IP channel. Range of value: 90~127, with the default value of 101.
RTP Port Range	Supported RTP port range for the IP end to establish a call conversation, with the lower limit of 10000 and the upper limit of 60000 and the difference between larger than 480. The default value is 50000-50767.

Silence Suppression	Sets whether to send comfort noise packets to replace RTP packets or never to send RTP packets to reduce the bandwidth usage when there is no voice signal throughout an IP conversation. The optional values are *Enable* and *Disable*, with the default value of *Disable*. Note: This configuration item is unsupported for the B-type wireless gateway.
JitterBuffer	Acceptable jitter for data packets transmission over IP, which indicates the buffering capacity. A larger JitterBuffer means a higher jitter processing capability but as well as an increased voice delay, while a smaller JitterBuffer means a lower jitter processing capability but as well as a decreased voice delay. Range of value: 20~200, calculated by ms, with the default value of 20. Note: This configuration item is unsupported for the B-type wireless gateway.
JitterScaling	Jitter scaling factor which is used to adjust the excessive jitter of the RTP flows over IP. The more excessive the jitter is, the larger the scaling factor should be. The value range is 8~256, with the default value of 8. You'd better set this value to 8 or a multiple of 8. Note: This configuration item is unsupported for the B-type wireless gateway.
Voice Gain Output from IP	Adjusts the gain of the voice output from IP. Range of value: -24~12, calculated by dB, with the default value of 0.
AGC	If the AGC (Automatic Gain Control) feature is enabled, the gateway will automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals. Note: This configuration item is unsupported for the B-type wireless gateway.
Enable G722/8000Hz	When this feature is enabled, the sampling rate of G722 is 8000 Hz. By default it is disabled and the sampling rate is 16000Hz.
Target Energy Threshold	Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the default value of 0.
Maximum Gain Threshold	Set the maximum gain threshold that will be applied to the signal. Range of value: 0~48, calculated by dB, with the default value of 48.
Maximum Attenuation Threshold	Set the maximum attenuation that will be applied to the signal. Range of value: -42~0, calculated by dB, with the default value of 0.
Minimum Input Energy	Set the minimum threshold for the energy processed by AGC. Signals below this threshold will not be processed by AGC. Range of value: -60~ -25, calculated by dB, with the default value of -60.
Enable G729B	Set whether to enable G729B, By default it is disabled.

<table>
<tr>
<td>CODEC Priority</td>
<td>
Supported CODECs and their corresponding priority for the IP end to establish a call conversation. The table below explains the sub-items:
<table>
<tr><th>Sub-item</th><th>Description</th></tr>
<tr><td>Priority</td><td>Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.</td></tr>
<tr><td>CODEC</td><td>Three optional CODECs are supported: G711A, G711U, G729A/B, G723, G722, AMR and iLBC.</td></tr>
<tr><td>Packing Time</td><td>Time interval for packing an RTP packet, calculated by ms.</td></tr>
<tr><td>Bit Rate</td><td>The number of thousand bits (excluding the packet header) that are conveyed per second.</td></tr>
</table>
By default, all of the seven CODECs are supported and ordered G711A, G711U, G729A/B, G723, G722, AMR and iLBC by priority from high to low.
The packing time and bit rate supported by different CODECs are listed in the table below. Those values in bold face are the default values.
<table>
<tr><th>COEDC</th><th>Packing Time (ms)</th><th>Bit Rate (kbps)</th></tr>
<tr><td>G711A</td><td>10 / 20 / 30 / 40 / 60</td><td>64</td></tr>
<tr><td>G711U</td><td>10 / 20 / 30 / 40 / 60</td><td>64</td></tr>
<tr><td>G729A/B</td><td>10 / 20 / 30 / 40 / 60</td><td>8</td></tr>
<tr><td>G723</td><td>30 / 60</td><td>5.3 / 6.3</td></tr>
<tr><td>G722</td><td>10 / 20 / 30 / 40</td><td>64</td></tr>
<tr><td>AMR</td><td>20 / 40 / 60</td><td>4.75</td></tr>
<tr><td rowspan="2">iLBC</td><td>20 / 40</td><td>15.2</td></tr>
<tr><td>30 / 60</td><td>13.3</td></tr>
</table>
Note: The B-type wireless gateway only supports G711A, G711U and G729.
</td>
</tr>
</table>

3.5 Advanced Settings

Advanced Settings includes twelve parts: ***Network***, ***System Param***, ***Service Config***, ***Dialing Rule***, ***Function Key***, ***Cue Tone***, ***Color Ring***, ***QoS***, ***Tone Generator***, ***Tone Detector***, ***CDR Query*** and ***VPN***. ***Network*** is used to configure the general properties of the network port; ***System Param*** is used to configure some properties of the system; ***Service Config*** is used to configure some properties which corresponds to the service; ***Dialing Rule*** is used to set the judging conditions for dialing; ***Function Key*** is used to set a cluster of combination keys for you to query or set the network port; ***Cue Tone*** is used to set the gateway language for playing voice and the voice file used for the two-stage dialing; ***Color Ring*** is used to upload the color ring file which can be set as a ringback tone for an incoming call from IP to wireless port; ***QoS*** uses the differentiated services technology to increase the gateway's service quality; ***Tone Generator*** is used to configure some properties of tones sent from the gateway; ***Tone Detector*** is used to configure some properties of tones received by the gateway; ***CDR Query*** is used to inquire the detailed call record; **VPN** makes use of the tunnel technology to transport the data, and the methods of user authentication and data encryption to prevent the data being read and distorted when they are transported on the public network.

3.5.1 Network

On the Network Settings interface you can configure network parameters. A gateway has two LANs which can be configured with the same network type, IP address, subnet mask, default gateway and DNS server to realize the feature of hot backup. There are three options in type: Static, DHCP and PPPoE.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Reset*** to restore the configurations. After changing the IP address, you shall log in the gateway again using your new IP address.

3.5.2 System Param

See the table below for the configuration items on the System Parameters Setting interface.

Item	Description
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.
Access Setting	Sets the IP addresses which can access the gateway via WEB. By default, all IPs are allowed. You can set an IP whitelist to allow all IPs within it to access the gateway freely. Also you can set an IP blacklist to forbid all IPs within it to access the gateway.
CDR Enabled	Sets whether to enable the feature of CDR. It is required to fill in ***Server Address*** and ***Server Port*** in case CDR is enabled. By default, ***CDR*** is disabled.
Server Address	Sets the server address to receive CDR.
Server Port	Sets the server port to receive CDR.
Save CDR	Sets whether to save CDR, with the default value of *NO*.
Amount of Saved CDR	Sets the amount of saved CDR. Range of value: 200~10000, with the default value of 5000.
API Enabled	When this feature is enabled, the remote terminal can invoke the API interface. The default value is *No*.
Remote IP Address allowed to Invoke API	Sets the remote IP addresses which are allowed to invoke the API interface. Up to 5 addresses can be configured and each of them are separated by ",". "*" denotes all IP addresses are allowed.
Username for API Call, Password for API Call	The authorized username and password for calling the API interface.
Time Calibration	Sets the calibration mode for the time. Three options available: NTP, Synchronized with Operator and Close, with the default value of *Synchronized with Operator*.
NTP Server Address	Sets the Server address for NTP time synchronization.
Synchronizing Cycle	Sets the cycle for NTP time synchronization. The default value is 3600.
System Time	The system time. Check the checkbox before ***Modify*** and change the time in the edit box if *Time Calibration* is set to Close.
Time Zone	The time zone of the gateway.
Daily Restart	Sets whether to restart the gateway regularly every day at the preset ***Restart Time***. By default, this feature is disabled.
Restart Time	Sets the time to restart the gateway regularly.
Clear Call Count 2 after Restart	When this feature is enabled, the gateway will clear the data of Call Count 2 upon its restart. By default this feature is disabled.

3.5.3 Service Config

See the table below for the configuration items on the Service Config interface.

Item	Description
Enable Two Stage Dialing Mode for PSTN Outgoing Calls	Sets whether to enable the two stage dialing mode for PSTN outgoing calls. Under this mode, for an outgoing call from a wireless port, the IP side will hear the dial tone. If you fail to input the number during the schedule time, the wireless port will hang up the call automatically; otherwise, it will make an outgoing call to the number. The default value is *disabled*.
Maximum Wait Time for PSTN Outgoing Calls	Sets the maximum wait time waiting for the called party pickup during an outgoing call. Range of value: 5~120, calculated by s, with the default value of 60.
Dial Interval	Sets the largest interval between two digits of a dialing number. Range of value: 1~10, calculated by s, with the default value of 6. In case your dialing rules do not include "**.**", the call will fail if there is no digit dialed or no dialing rule matched during this interval; in case your dialing rules include "**.**", the gateway will wait until this interval ends and match to the dialing rule "**.**" if there is no digit dialed or no other dialing rule matched during this interval.
Waiting for Idle Time	Set the waiting time of the channel before it goes into the idle state after the call finishes. The default value is 1500ms, and the value range is 0~120000.
AT Busy Tone Detection Mode	Set the detection mode of the busy tone, including *Common* (hang up the call directly upon busy tone detection), *Delay* (hang up the call after a delay time upon busy tone detection), and *Ignore* (not to detect busy tone). The default setting is *Common*.
Delay Time	Sets the delay time when Delay is selected as the Busy Tone Detection mode.
Line Busy Tone Detection	Sets whether to enable the busy tone detector on the line to detect the busy tone. It is disabled by default.
Auto Check IMEI	Sets whether to turn on the switch to check IMEI automatically. It is enabled by default.
Push SMS to SMS Platform	Sets whether to push the SMS both received and sent by the gateway to the SMS platform. It is disabled by default.
SMS Platform Address	The IP address of the SMS platform connecting to the gateway.
Http Push	Sets whether to push the port status of the gateway, the SIP registration status of the port, and the carrier type of the current card. The default setting is Disable.
Http Server Address	Set the http server address to be pushed.
Identity (Default: Serial Num)	Sets the identifier in the push to distinguish the source of the push message. The default setting is the gateway serial number.
Push SMS (UDP)	Pushes SMS to the specified IP port.
Auto Hangup upon Ringback	This feature is only supported by the GSM module. Note that when it is enabled, you ***are*** required to set 'SIP compatibility-Occasion to Reply 183' after ringing.
Communication without Network	Automatically routes a call to the wireless port in case of network failure or call timeout. The default value is *disabled*.

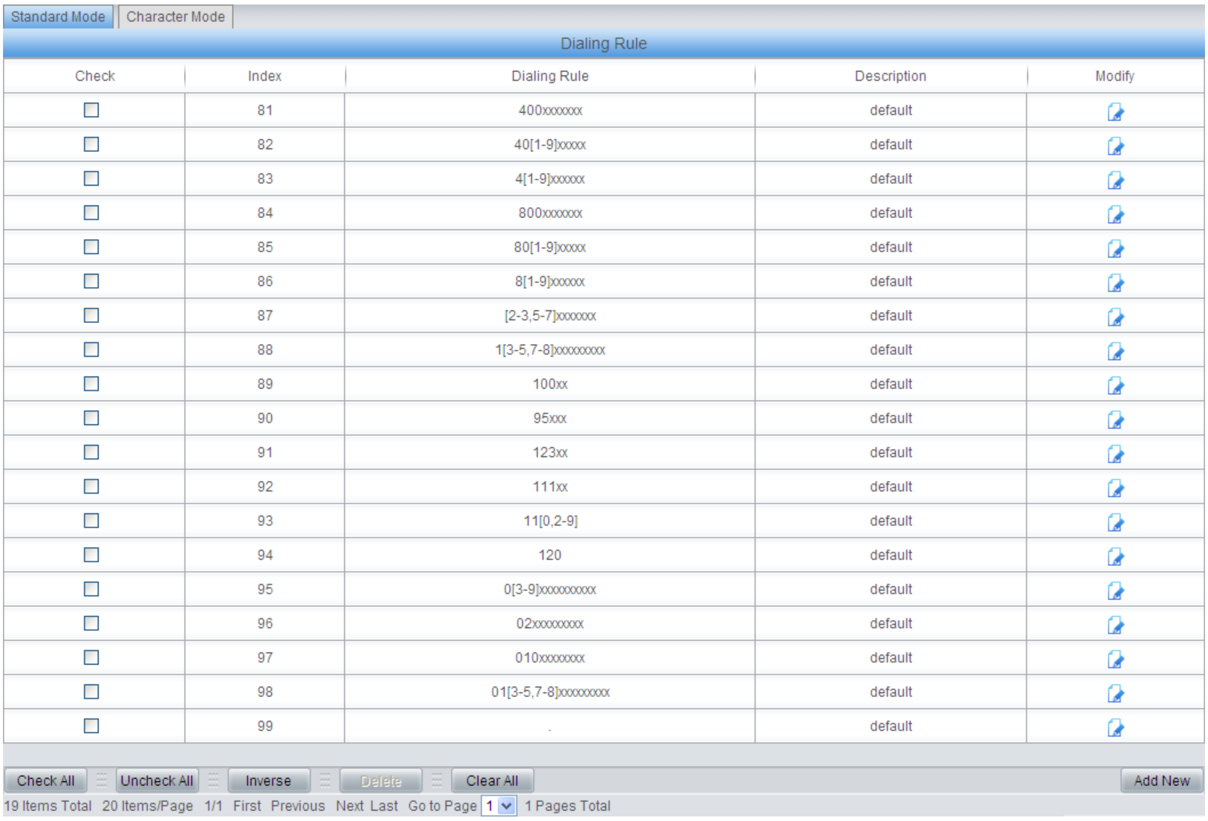
IP→Tel Call Failure, Auto Transfer	Sets whether to enable the feature of transferring the call to a designated IP automatically when a call from IP to Tel fails, with the default value of *disable*. If this feature is enabled, you are required to enter Target Number (Registered) or Target IP and Target Port (Unregistered).
Tel → IP Call Failure, Auto SMS Reply	Sets whether to enable the feature of automatic SMS reply when a call from Tel to IP fails, with the default value of *disable*. The following four options will be available if this feature is enabled. They are Unconnected, No Answer, Rejected, Fail to Connect. You can select any one of them and define the corresponding content to reply.
Auto Disable Module if Fail to Register to SIP Server	Once this feature is enabled, the gateway will automatically close this SIM card module to achieve the feature "Communication without Network" when it failed to register to the SIP server. The default value is *disabled*. It works with the feature *FWD on Unreachable*.
Auto Lock SIM Card after Consecutive Call Failure	When this feature is enabled, the times of call failure reaching the set value will trigger the operation of card locking. Call Failure Mode includes: *Busy, No Answer, Dial Failure*. Locking Time means the time of the port being locked: -1 means the card is always being locked; 0 means the card is unlocked; other values mean the exact time of the card being locked. When the feature 'Unlock by Plugging SIM Card in and out' is enabled, the port will be unlocked after you plug in and out the SIM card.
Auto Reconnect to BS after Consecutive Call Failure	When the outgoing calls from a port has failed for several times consecutively: for the SMG4002/SMG4004/SMG4008 series, the gateway will automatically reconnect the SIM card on this port to the base station; for the SMG4016/SMG4032 series, the gateway will automatically switch to other card slots available for the port and reconnect, the SIM card to the base station if there is no available card slot.
SNMP Alarm after Consecutive Call Failure	When the outgoing calls from a port has failed for several times consecutively, the SNMP alarm will be sent to the set IP and port. If this feature is enabled, you are required to fill in *Call Failure Times*, *Remote IP* and *Port*.
Record SIM Number at Gateway Restart	Once this feature is enabled, the number of the SIM card will be recorded when the gateway restarts and this SIM card will recover to work after restarting. The default value is *enabled*.
Consecutive Acquisition of QINISTAT	Set after how many times that the SIM card fails to acquire the correct QINISTAT value will it be ignored. The default setting is 0.
UDP Alarm after Consecutive Call Failure	When the outgoing calls from a port has failed for several times consecutively, the UDP alarm will be sent to the set port of the IP address in the C field for the last call. If this feature is enabled, you are required to fill in *Call Failure Times* and *Port*.
Module Exception Recovery Function	Sets whether to recover the module when it goes abnormal. This function is enabled by default and not suggested to disable.
Work Mode	Sets the work mode for the echo canceller. There are two options: *Near-end cancellation* and *Both near-end and far-end cancellation*, with the default value of *Near-end cancellation*.

Non-linear Processing	Sets whether to enable the mode of non-linear processing. By default, this feature is *enabled*.
Fixed Window Size	Sets the size of the window for the fixed cancellation.
Moving Window Size	Sets the size of the window for the moving cancellation.

Note: The echo cancellation feature is unsupported for the B-type gateway.

3.5.4 Dialing Rule

Considering efficiency, it is not acceptable that the gateway reports to the PBX or relevant devices every time it receives a number. Instead, we hope that the gateway can automatically judge the received number to see if it meets the set rule, if it is complete and if it is qualified to make outgoing calls. Therefore, a whole dialing plan, which consists of multiple dialing rules specifying the auto judging conditions, is required. Each dialing rule has a priority, which is used to restrict the sequence and avoid conflict.



Standard Mode | Character Mode

Dialing Rule

Check	Index	Dialing Rule	Description	Modify
	81	400xxxxxxx	default	
	82	40[1-9]xxxxx	default	
	83	4[1-9]xxxxxx	default	
	84	800xxxxxxx	default	
	85	80[1-9]xxxxx	default	
	86	8[1-9]xxxxxx	default	
	87	[2-3,5-7]xxxxxxx	default	
	88	1[3-5,7-8]xxxxxxxxx	default	
	89	100xx	default	
	90	95xxx	default	
	91	123xx	default	
	92	111xx	default	
	93	11[0,2-9]	default	
	94	120	default	
	95	0[3-9]xxxxxxxxxx	default	
	96	02xxxxxxxxx	default	
	97	010xxxxxxxx	default	
	98	01[3-5,7-8]xxxxxxxxx	default	
	99	.	default	

Check All | Uncheck All | Inverse | Delete | Clear All | Add New

19 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

Figure 3-11 Dialing Rule Configuration Interface (Standard)

See Figure 3-11 for the Dialing Rule Configuration interface under the standard mode. The list in the above figure shows the dialing rules with their priorities and description, which can be added by the ***Add New*** button on the bottom right corner.

The table below explains the items shown in Figure 3-11.

Item	Description
Index	The unique index of each dialing rule, which denotes its priority. A dialing rule with a smaller index value has a higher priority and will be checked earlier while matching.
Description	Remarks for the dialing rule. It can be any information, but not be left empty.
Dialing Rule	Up to 100 dialing rules can be configured in the gateway, and the maximum length of

each dialing rule is 127 characters. See below for the meaning of each character in the dialing rule. The gateway will do instant matching for your dialing number based on the dialing rule and regard your dialing as finished upon receiving ‘#’ or dialing timeout.

Character	Description
“0”~”9”	Digits 0～9.
“A”~”D”	Letters A～D.
“x”	A random number. A string of ‘x’s represents several random numbers. For example, ‘xxx’ denotes 3 random numbers.
“.”	‘.’ indicates a random amount (including zero) of characters after it.
“[ ]”	‘[ ]’ is used to define the range for a number. Values within it only can be digits ‘0~9’, punctuations ‘-‘ and ‘,’. For example, [1-3,6,8] indicates any one of the numbers 1, 2, 3, 6, 8.
“-”	’-’ is used only in ‘[ ]’ between two numbers to indicates any number between these two numbers.
“,”	’,’ is used to separate numbers or number ranges, representing alternatives.
“*”	Only represents symbol ”*”.
“#”	Only set it at the beginning of the string, representing symbol “#”.

There are 19 dialing rules already configured on the gateway for easy use. See below for detailed information.

Priority	Dialing Rule	Description
99	.	Any number in any length.
98	01[3-5,7-8]xxxxxxxxx.	Any 12-digit number starting with 013, 014, 015, 017 or 018
97	010xxxxxxxx	Any 11-digit number starting with 010
96	02xxxxxxxxx	Any 11-digit number starting with 02
95	0[3-9]xxxxxxxxxx	Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09
94	120	Number 120。
93	11[0,2-9]	Number 110, 112, 113, 114, 115, 116, 117, 118 or 119
92	111xx	Any 5-digit number starting with 111
91	123xx	Any 5-digit number starting with 123
90	95xxx	Any 5-digit number starting with 95
89	100xx	Any 5-digit number starting with 100
88	1[3-5,7-8]xxxxxxxxx	Any 11-digit number starting with 13, 14, 15, 17 or 18
87	[2-3,5-7]xxxxxxx	Any 8-digit number starting with 2, 3, 5, 6 or 7

	86	8[1-9]xxxxxx	Any 8-digit number starting with 81, 82, 83, 84, 85, 86, 87, 88 or 89
	85	80[1-9]xxxxx	Any 8-digit number starting with 801, 802, 803, 804, 805,.806, 807, 808 or 809
	84	800xxxxxxx	Any 10-digit number starting with 800
	83	4[1-9]xxxxxx	Any 8-digit number starting with 41, 42, 43, 44, 45, 46, 47, 48 or 49.
	82	40[1-9]xxxxx	Any 8-digit number starting with 401, 402, 403, 404, 405, 406, 407, 408 or 409
	81	400xxxxxxx	Any 10-digit number starting with 400

After configuration, click ***Save*** to save the above settings into the gateway or click ***Close*** to cancel the settings.

Click ***Modify*** in Figure 3-11 to modify the dialing rules. The configuration items on the dialing rule modification interface are the same as those on the ***Add New Dialing Rule*** interface.

To delete a dialing rule, check the checkbox before the corresponding index in Figure 3-11 and click the '***Delete***' button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all dialing rules at a time, click the ***Clear All*** button in Figure 3-11.

See Figure 3-12 for the Dialing Rule Configuration interface under the Character mode. You can edit the dialing rule list to add a new one or modify an old one. The exact meaning of each rule element is described on the page.

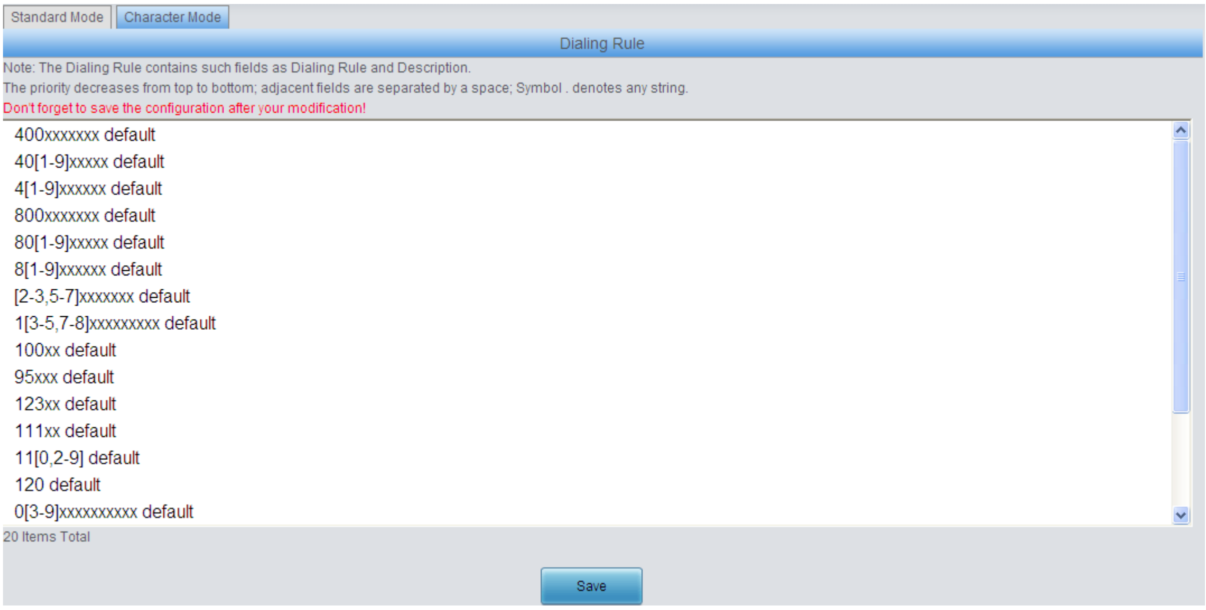


Figure 3-12 Dialing Rule Configuration Interface (Character)

3.5.5 Function Key

On the function key configuration interface you can set a cluster of combination keys. An external phone can dial the wireless port and press the combination keys after hearing the speech prompt "Please dial the extension number" to query or set the network port.

Click "Enable" to enable the corresponding function key. The gateway will use the default function keys when the mode is set to default; and it will allow you to set new function keys when the mode is set to user-defined. Click ***Save*** to save your settings into the gateway.

3.5.6 Cue Tone

See the table below for the configuration items on the Cue Tone interface.

Item	Description
Language	Sets the language for the gateway to play voice, including two options Chinese and English. The default setting is *English*.
Upload a file of cue tone	Uploads a user-defined cue tone file to the gateway.
Two Stage Dialing for PSTN Outgoing Calls Tips	Sets the cue tone of two stage dialing for the PSTN outgoing calls, including two options: Dial Tone and File Playback. You are required to upload a file for playing if File Playback is selected.

Click ***Save*** to save the above settings into the gateway.

3.5.7 Color Ring

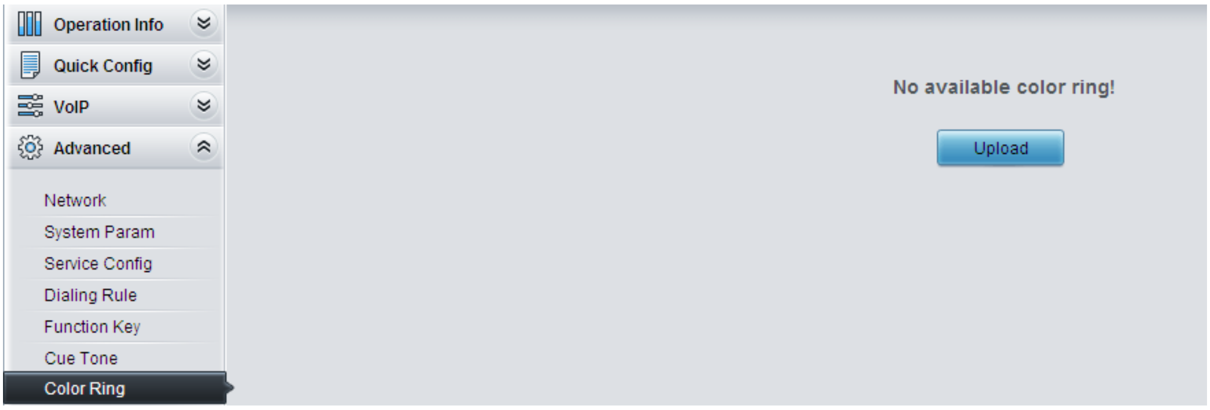


Figure 3-13 Color Ring Interface

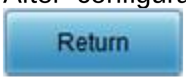
By default, there is no available color ring on the gateway. See Figure 3-13. Click 

to upload a new color ring manually.

See the table below for details.

Item	Description
Index	The unique index of each color ring to be uploaded.
Description	It is user-defined, with the default value of *default*.
Color Ring	The file of the color ring to be uploaded.

After configuration, click 

to upload the color ring file to the gateway or click 

to cancel the upload. See Figure 3-14 for the Color Ring Management interface after the upload.

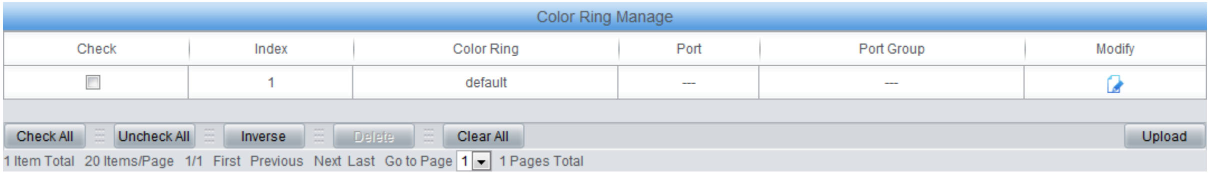


Figure 3-14 Color Ring Management Interface

Click ***Modify*** in Figure 3-14 to modify the configuration of the color ring. See below for the color ring modification interface. The configuration items on this interface are the same as those on the ***Color Ring Upload*** interface.

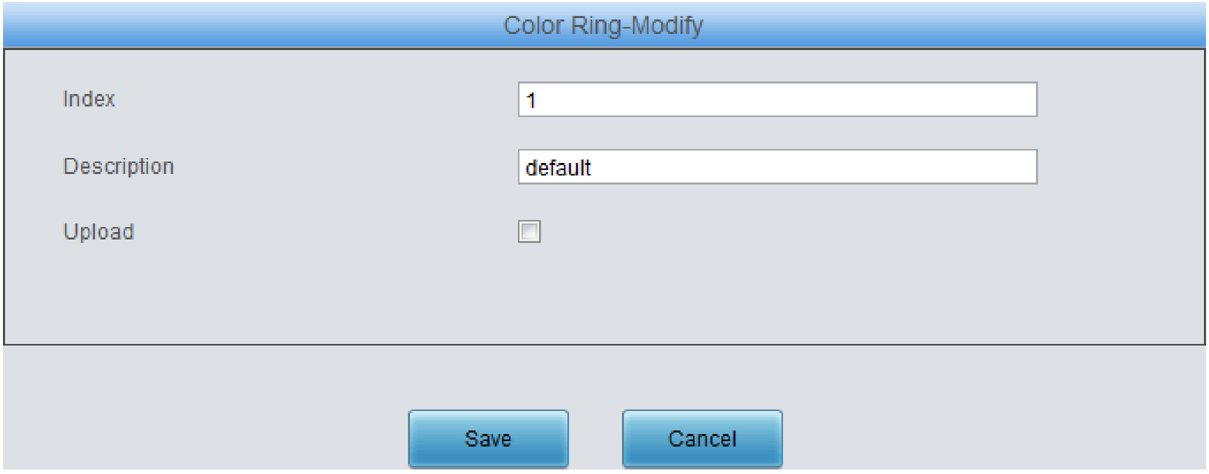


Figure 3-15 Color Ring Modification Interface

To delete a color ring, check the checkbox before the corresponding index in Figure 3-14 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all color rings at a time, click the ***Clear All*** button in Figure 3-14.

3.5.8 QoS

On the Qos interface you can enable or disable this feature. Using the differentiated service, the gateway can meet various application requirements under a limited bandwidth and ensure neither delay nor discard for important services so as to improve its quality of services.

The table below explains the items shown in the above figure.

Item	Description
QoS	Sets whether to enable the OoS differentiated services. By default, it is disabled.
Media Premium QoS	Sets the priority of the media premium for QoS. A media premium QoS with a bigger value has a higher priority. The value range is 0~63, with the default value of 46.
Control Premium QoS	Sets the priority of the control premium for QoS. A control premium QoS with a bigger value has a higher priority. The value range is 0~63, with the default value of 26.

3.5.9 Tone Generator

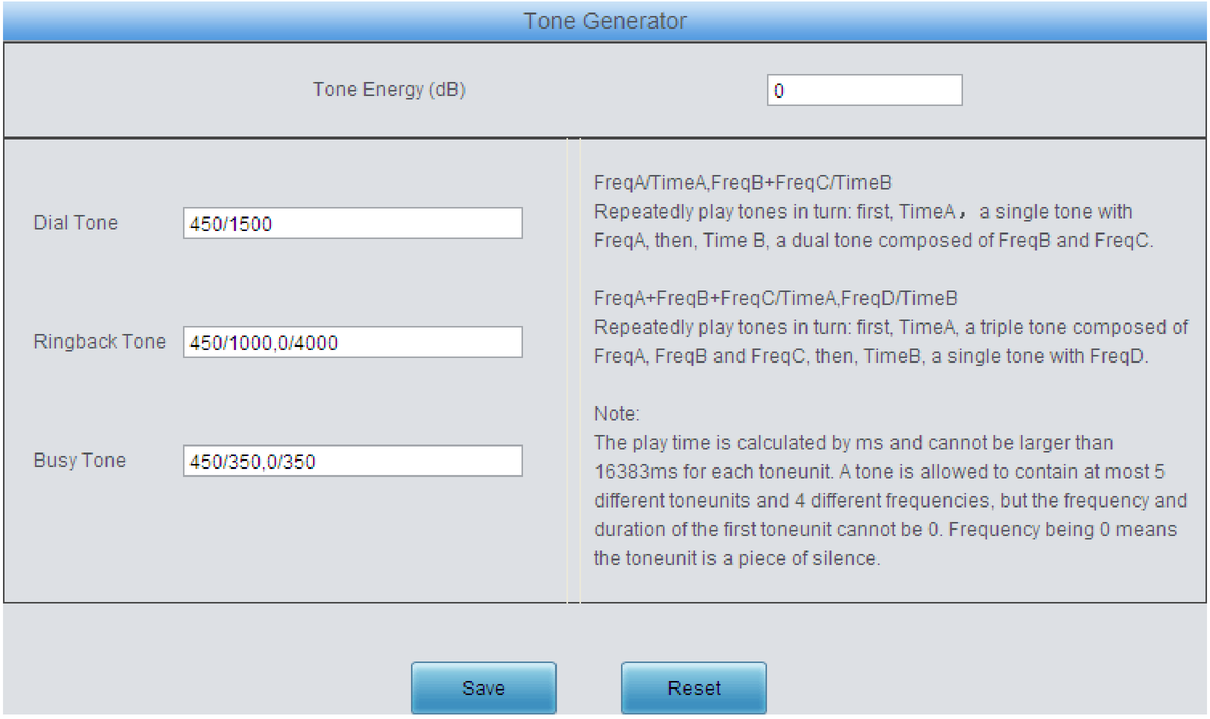


Figure 3-16 Tone Generator Setting Interface

On the Tone Generator Setting interface you can see and set parameters of the tone generator. By default, there are three tones on it: Dial Tone—a single tone with 450HZ frequency, plays continuously; Ringback Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 1s play and 4s pause; Busy Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 350ms play and 350ms pause. You can configure the tone generator manually. The exact explanation about the format and the meaning is described on the right of the interface. The value range of the tone energy herein above is -12~17, calculated by dB, with the default value of 0.

3.5.10 Tone Detector

The Tone Detector interface displays all parameters for tone detection. By default there are two pieces of settings respectively for Dial Tone and Busy Tone. No more than two can be configured. This feature is mainly used for detecting busy tones. Click ***Modify*** to go to the Tone Parameters interface.

See below for the configuration items on the Tone Parameters interface.

Item	Description
Index	The index of the tone detector. Each group of tone detectors has a unique index.
Tone	Includes three kinds of tones: Dial Tone, Busy Tone and Ringback Tone. For the wireless gateway, it is mainly for detecting busy tones.
Type	The type of tones, including Continuous Tone and Periodic Tone.
The 1st Mid-frequency	The first mid-frequency in Hz, with the value range of 200~3500 and the default value of 450.
The 2nd Mid-frequency	The second mid-frequency in Hz, with the value range of 0 or 200~3500 and the default value of 0.

Duration at ON State	The required duration of tones at ON state. By default, it is 1500ms for dial tones, 350ms for busy tones, and 1000ms for ringback tones.
Duration at OFF State	The required duration of tones at OFF state. By default, it is 0ms for dial tones, 350ms for busy tones, and 4000ms for ringback tones.
Period Count	Sets the count of periods as the condition to determine a periodic tone. By default, it is 0 for Dial Tone, 2 for Busy Tone, and 1 for Ringback Tone.
Energy	Sets the energy threshold for the tone detector to detect the on-line tone. To increase the accuracy, you can adjust the value according to the tone volume on the line. Range of value: -18~11, calculated by dB. The default value is 0.

Click ***Save*** to save the above settings into the gateway. Click ***Close*** to cancel your modification and return to the previous interface. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all tones at a time, click the ***Clear All*** button.

3.5.11 CDR Query

See the table below for the items on the CDR Query Setting interface.

Item	Description
Starting Date, Ending Date	Sets the starting and ending dates for CDR query.
Port	Sets the port on which CDR query will proceed.
Call Direction	Sets the call direction for CDR query.
CallerID, CalleeID	Sets the CallerID/CalleeID for CDR query.
Call Duration	Sets the minimum/maximum call duration for CDR query.

Click ***Query*** to query the CDR information based on the set conditions.

Note: This page will appear only when the CDR feature is enabled (set in System Param).

3.5.12 VPN

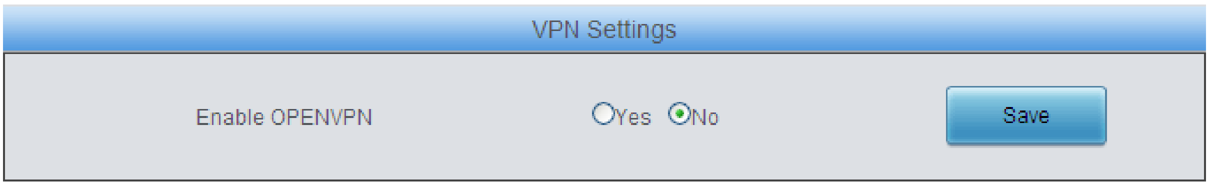


Figure 3-17 VPN Settings Interface

Thanks to the embedded VPN Client, the wireless gateway can access the VPN network via OPENVPN directly, not requiring extra VPN client, which simplifies the network deployment. Meanwhile, the design of both SIP signaling messages and voice streams transporting via VPN avoids possible problems induced by the SIP protocol in passing through the firewall and NAT. See Figure 3-17 for the VPN Settings interface. The table below gives the explanation to the items shown in the above figure.

Item	Description
Enable OPENVPN	Sets whether to enable the VPN feature, with the default value of *No*. If this feature is enabled, the gateway will work as a VPN client.

You are required to upload the VPN certificate after enabling the VPN feature. See Figure 3-18.

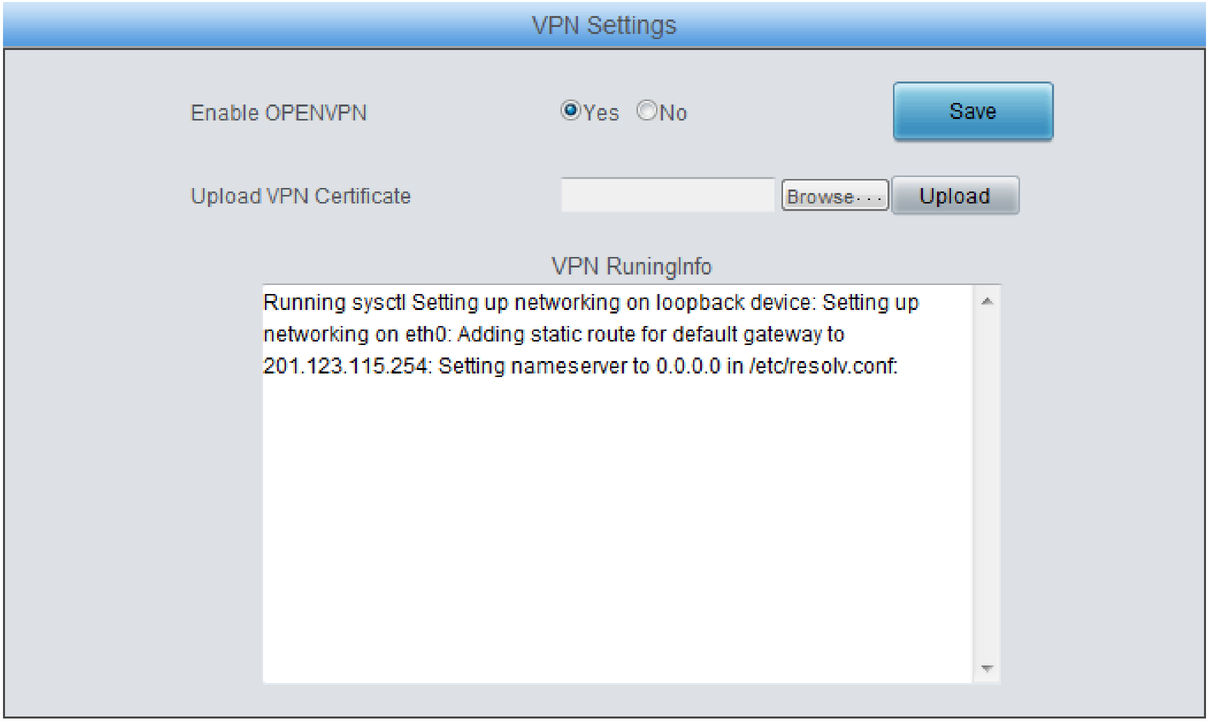


Figure 3-18 VPN Certificate Upload Interface

Note: Refer to Appendix C About VPN for how to make a VPN certificate.

3.6 Wireless Settings

Wireless Settings includes the following parts: ***Basic Param***, ***Wireless Param***, ***Call Forwarding***, ***Short Message***, ***SMPP***, ***SMS over SIP***, ***IMEI (GSM, WCDMA and LTE series)***, ***USSD (GSM, WCDMA and LTE series)***, ***Email***, ***SIM Card, PIN Manage***, ***BS Select (GSM series)***, ***SIM Mode, Call Waiting (GSM, WCDMA and LTE series), Lock SIM, Networking Setting (WCDMA and LTE series)***, ***AMD (CDMA series)*** and ***Hidden CallerID (WCDMA and LTE series)***.

3.6.1 Basic Parameters

See the table below for the items on the Basic Parameters Setting interface.

Item	Description
GSM (WCDMA/LTE) Voice Encoding	Sets the mode of the GSM (WCDMA/LTE) voice encoding. By default, the voice encoding for GSM is *Automatic* and for WCDMA/LTE is *AMR*.
Volte	Once this feature is enabled, the 4G function will be enabled when there is a call ongoing on; Otherwise, only 2G or 3G function is available.
GSM (WCDMA/CDMA/LTE) DTMF Send Mode	Sets the mode to send the GSM (WCDMA/CDMA/LTE) DTMF, three options available for GSM (WCDMA/CDMA): Voice Playback, Remote Transmission and Chip Transmission. The default value is *Voice Playback*. Two options are available for LTE: Remote Transmission and Chip Transmission. The default value is *Remote Transmission*.
DTMF Transmission Intensity	Sets the transmission intensity of the DTMF. The default values for the GSM gateway and the WCDMA gateway are respectively 6 and 1. Note: 1, This configuration item is unsupported when the DTMF send mode is set to

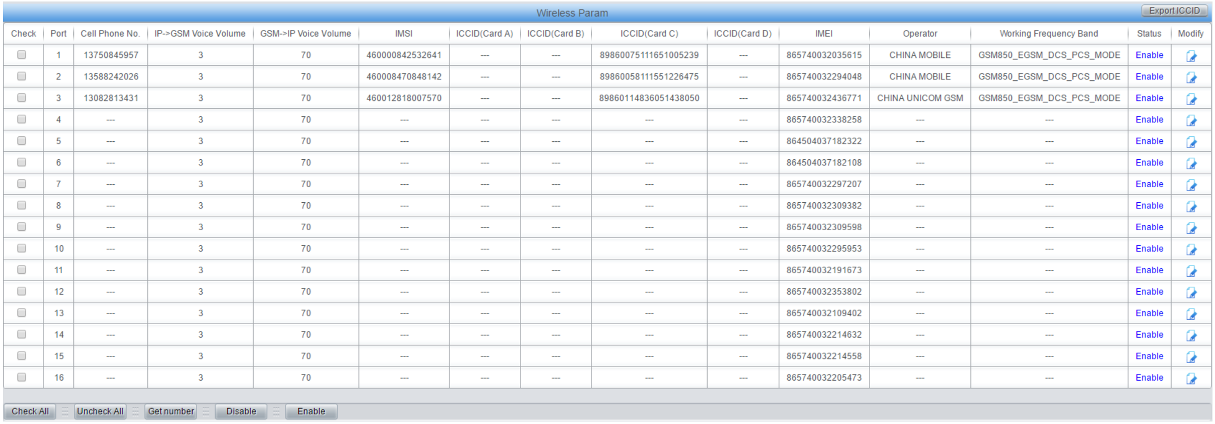
	Remote Transmission; 2, This configuration item is unsupported for the CDMA gateway.
Duration at ON	Sets the duration of the DTMF signal at ON state, calculated by ms. The default value is 120.
Duration at OFF	Sets the duration of the DTMF signal at OFF state, calculated by ms. The default value is 100.
Disconnect Voice while Sending DTMF	Sets whether to disconnect the voice channel while sending the DTMF, with the default value of *No*. Note: This configuration item is unsupported when the DTMF send mode is set to Remote Transmission;
GSM (WCDMA/CDMA/LTE) DTMF Receive Mode	Sets the mode to receive the GSM (WCDMA/CDMA/LTE) DTMF, two options available: Chip Receive and Wireless Module Receive. The default values for GSM WCDMA and LTE are *Wireless Module Receive;* the default value for CDMA is *Chip Receive*.
Minimum Duration at ON	Set the minimum duration at ON for the DTMF signal. That is, those DTMF signals staying at ON longer than the set value of this item will all be received. By default, the set value for the A-type gateway is 28 and that for the B-type is 23.
Energy Difference of High-freq minus Low-freq	Sets the allowed energy difference when the high-frequency energy in DTMF is larger than the low-frequency energy, calculated by db, with the default value of 5.
Energy Difference of Low-freq minus High-freq	Sets the allowed energy difference when the low-frequency energy in DTMF is larger than the high-frequency energy, calculated by db, with the default value of 9.
Lowest Energy Threshold	The allowed lowest energy for DTMF detection, calculated by db, with the default value of -21.
DTMF Voltage Detection for GSM	Set the On and off of the DTMF detection for GSM.
Network Scan Mode	Sets a network for the call, three options available for the WCDMA gateway: Automatic, GSM Only and WCDMA Only. The default value is *Automatic*. Nine options are available for the LTE gateway: Automatic, GSM Only, WCDMA Only, LTE Only, TD-SCDMA Only, UMTS Only, CDMA Only, HDR Only, CDMA and EVDO Only. The default value is *Automatic*
Network Scan Sequence	Sets the priority of the network, three options available for the WCDMA gateway: *Automatic*, *GSM prior to WCDMA* and *WCDMA prior to GSM*. The default value is *Automatic.* Only the option Automatic is available for the LTE gateway.
SMS Sending Type	The way to send SMS. Three options are available: Auto, Common, Flash. And the default setting is Auto.
SMS Time (Inbox)	The time display mode of the gateway inbox, two options available: Transmit Time and Receive Time. Transmit Time indicates the time of the SMS being transferred by the SMS center; Receive Time indicates the time of the SMS being received by the gateway.

SMS Sending Interval	Sets the interval to send SMS for each port. Range of value: 1~60, with the default value of *1*.
Maximum Pieces of Saved Logs	Sets the amount of the logs to be saved for each port. Range of value: 50~500, with the default value of *100*.
SMS Receipt	Once this feature is enabled, the gateway will receive a receipt upon the remote side receiving the SMS. Note: This configuration item is unsupported for the CDMA gateway.
AT Command Mode	Sets the AT command sent with the call forwarding. There are two options available: CCFC command mode and ATD command mode. The GSM and LTE LC gateways support both modes, while the WCMDA and LTE (except for LC) gateways only support the CCFC command mode and the CDMA gateways only support the ATD command mode.
Set/Cancel Service Number for FWD Unconditionally, Set/Cancel Service Number for FWD on Busy, Set/Cancel Service Number for FWD on No Reply, Set/Cancel Service Number for FWD on Unreachable	Sets or Cancels the service No. for FWD unconditionally, FWD on busy, FWD on no reply or FWD Unreachable. The former box is used to set the service No, while the latter one is to cancel the service number.
Cancel All FWD Service Numbers	Used to cancel all service numbers for FWD unconditional, FWD on busy and FWD on no reply.
Cancel All Waiting Service Numbers	Used to cancel the service number for call waiting.
SIP Answer Code	Sets the SIP answer code for each state of the called party. Busy/Rejected: 486; No Answer: 408; Other Fault: 480; Call Time Over: 503; No Idle Channel: 486; Route Failed: 488.
Timed Call Settings	Calls the set number at a specified time.
Timing Time	Sets the scheduled outgoing time.
No Response Waiting Time	The maximum waiting time for outgoing calls without response.
Suspend Mode	Timed outgoing call hangup mode: 1 Received Mo Ring; 2. After Connection. You can set the call time.

Note: *Times Call Settings* and *Number Settings* are both on the page of Port Setting.

Click Save to save the setting into the gateway, click Reset to restore the configurations.

3.6.2 Wireless Param



Wireless Param

Export ICCID

Check	Port	Cell Phone No.	IP->GSM Voice Volume	GSM->IP Voice Volume	IMSI	ICCID(Card A)	ICCID(Card B)	ICCID(Card C)	ICCID(Card D)	IMEI	Operator	Working Frequency Band	Status	Modify
	1	13750845957	3	70	460000842532641	---	---	89860075111651005239	---	865740032035615	CHINA MOBILE	GSM850_EGSM_DCS_PCS_MODE	Enable	
	2	13588242026	3	70	460008470848142	---	---	89860058111551226475	---	865740032294048	CHINA MOBILE	GSM850_EGSM_DCS_PCS_MODE	Enable	
	3	13082813431	3	70	460012818007570	---	---	89860114836051438050	---	865740032436771	CHINA UNICOM GSM	GSM850_EGSM_DCS_PCS_MODE	Enable	
	4	---	3	70	---	---	---	---	---	865740032338258	---	---	Enable	
	5	---	3	70	---	---	---	---	---	864504037182322	---	---	Enable	
	6	---	3	70	---	---	---	---	---	864504037182108	---	---	Enable	
	7	---	3	70	---	---	---	---	---	865740032297207	---	---	Enable	
	8	---	3	70	---	---	---	---	---	865740032309382	---	---	Enable	
	9	---	3	70	---	---	---	---	---	865740032309598	---	---	Enable	
	10	---	3	70	---	---	---	---	---	865740032295953	---	---	Enable	
	11	---	3	70	---	---	---	---	---	865740032191673	---	---	Enable	
	12	---	3	70	---	---	---	---	---	865740032353802	---	---	Enable	
	13	---	3	70	---	---	---	---	---	865740032109402	---	---	Enable	
	14	---	3	70	---	---	---	---	---	865740032214632	---	---	Enable	
	15	---	3	70	---	---	---	---	---	865740032214558	---	---	Enable	
	16	---	3	70	---	---	---	---	---	865740032205473	---	---	Enable	

Check All | Uncheck All | Get number | Disable | Enable

Figure 3-19 Wireless Parameters Configuration Interface

The Wireless Parameters Configuration interface displays all relative information. Click ***Modify*** in the right column to modify the properties of the corresponding module.

The table below explains the configuration items on the Wireless Parameters Modification interface.

Item	Description
Port	The number of the port corresponding to the wireless module.
Cell Phone No.	The number of the SIM card corresponding to the wireless module. This number should be configured manually.
Query Mode	It is supported to acquire the SIM card number by two modes SMS and USSD.
Destination Number	Sets the destination number to receive the short message.
Encode Mode	Sets the encoding format for sending short messages.
Content to Send	Sets the content of the short message.
Keywords to Match	Sets the keywords used to get the cell phone No. from the received SMS.
Bits of Prefix to be Removed from Phone No.	Sets the bits of the prefix to be removed from the cell phone No.. Up to 4 bits can be removed.
Get Phone No. after SIM Card Registration	Sets whether to get the cell phone No. after the SIM card being registered successfully.
Write Query Result to SIM	Sets whether to write the queried cell phone number to the SIM card. The default setting is Yes.
IP->GSM(WCDMA/CDMA) Voice Volume	The volume of the voice from IP to GSM/WCDMA/CDMA. By default, the value for GSM is 3; the value for WCDMA is 10000; the value for CDMA is 1; the value for SIMCOM is 10400.
GSM(WCDMA/CDMA)->IP Voice Volume	The volume of the voice from GSM/WCDMA/CDMA to IP. By default, the value for GSM is 70; the value for WCDMA is 3; the value for CDMA is 2; the value for SIMCOM is 7000.
Disable Call Out Un4G	Sets whether to enable outgoing calls in non 4G network. The default setting is Disable. This item is available only for the LTE gateway.
Network Scan Mode	Sets the network scan mode for a module during SIM card registration. Note: This item is only available for the LTE and WCDMA gateways.

Operator	The operator of the wireless module. It is obtained automatically. This configuration is unavailable for CDMA module.
GPRS	Set whether to enable or disable the GPRS feature. This feature is only supported by the GSM gateway.
Select Band	Sets the frequency band of the base station during SIM card registration. You shall restart the gateway after selecting a band. This feature is only supported by the WCDMA gateway.
Apply to all the modules	Sets whether to apply all the settings except for the cell phone number to all the modules.
IMSI	International Mobile Subscriber Identification Number, the unique identity of the SIM card.
ICCID	Integrate Circuit Card Identity (ICCID) is just the SIM card number which serves as he identification card of a phone number. It is the unique identification number of the IC card, consisting of 20 digits.
IMEI	International Mobile Equipment Identity. Note: This configuration item is unsupported for the CDMA gateway.
Working Frequency Band	Displays the working frequency band of the wireless module. This configuration is unavailable for CDMA module.
Status	Displays the current state of the wireless module.

Click ***Modify*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Back*** to cancel the settings.

3.6.3 Call Forwarding

See the table below for the items on the Call Forwarding Configuration interface.

Item	Description
Port	The number of the port corresponding to the wireless module.
Cell Phone No.	The number of the SIM card corresponding to the wireless module.
FWD Unconditionally	Sets whether to enable the feature of FWD unconditionally and the FWD number if it is enabled.
FWD on Busy	Sets whether to enable the feature of FWD on busy and the FWD number if it is enabled. Note: Be sure to disable the Call Waiting feature before using it.
FWD on No Reply	Sets whether to enable the feature of FWD on no reply and the FWD number if it is enabled.
FWD on Unreachable	Sets whether to enable the feature of FWD on unreachable and the FWD number if it is enabled.
FWD Setting Status	Displays the setting status of the call forwarding service.
FWD Query Status	Displays the query status of the FWD settings. This configuration is unavailable for CDMA module.
Cancel All	Cancels all the setting on call FWD service. This item will appear if none of the call FWD is selected.

Click ***Modify*** to modify the properties of the corresponding port. See Figure 3-20 for the call forwarding modification interface. Then click ***Modify*** to save the settings into the gateway. It will take some time to apply the settings, and you can check the result in the ‘FWD Setting Status’ column. Click ***Reset*** to restore the configurations, or click ***Cancel*** to cancel the settings.

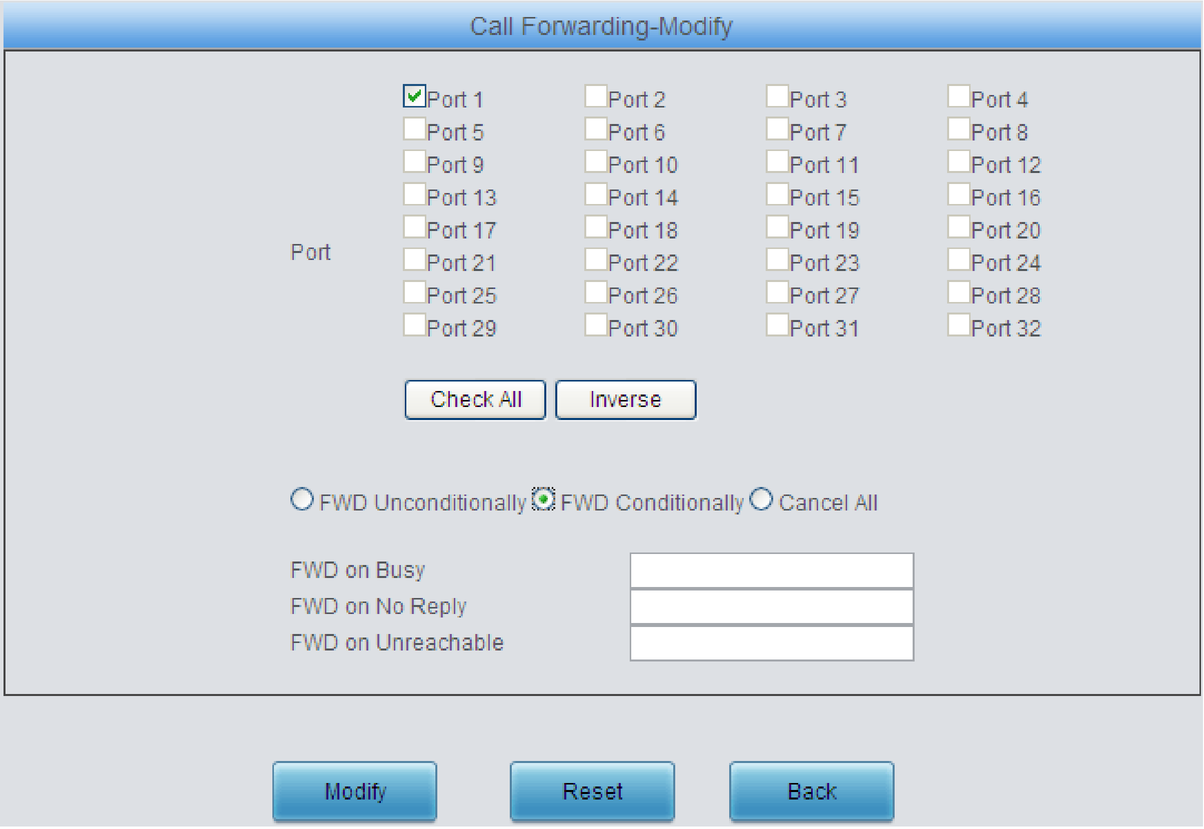


Figure 3-20 Wireless Service Modification Interface

3.6.4 Short Message

The Short Message interface displays the related information about the received/sent SMS.

Click ***SMS Center*** to go into the SMS Center Modification interface. Click ***Save*** to save the settings into the gateway, click ***Close*** to cancel the settings.

Download Inbox is used to download the SMS inbox data while ***Download Outbox*** is used to download the SMS outbox data.

Note: The configuration of SMS Center is unavailable for the CDMA gateway.

Click ***Inbox*** to go into the SMS Receiver Details interface. Such information as the remote cell phone number, the time and the content will be displayed on this page.

To delete a piece of SMS receiving detail, check the checkbox before the corresponding index on the Inbox interface and click the ***Delete*** button.

Click ***Outbox*** to go into the SMS Sending interface. Such information as the send status of the SMS, the remote cell phone number, the time, and the content will be displayed on this page.

To delete a piece of record, check the checkbox before the corresponding index on the Outbox interface and click the ***Delete*** button. To filter the receive/send short messages according to the set conditions, click the ***Filter*** button on the bottom right corner. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; to clear all records at a time, click the ***Clear All*** button; to refresh the interface, click ***Refresh***.

Click ***Send SMS*** to go into the Send SMS interface.

The table below explains the configuration items on the Send SMS interface.

Item	Description
Port	Select a port to send the SMS. There are three options available: Assignation Port， Automatic, Group Send.
Number Import	Click *Browse* to select the required number file and then click *Import* to import this file.
Send to	Enter the remote number to receive the SMS.
Sending Type	If the item ***SMS Sending Type*** on the Basic Parameters Setting interface is set to ***Auto***, this item ***Sending Type*** has two options ***Common*** and ***Flash***; if that is set to ***Common***, this item is set to ***Common*** by default; if that is set to ***Flash***, this item is set to ***Flash*** by default.
Encoding Format	The encoding format for the SMS, two options available: GSM 7bit and UCS2.
Content	The content of the SMS required to be sent.
Result	Display the send result of the SMS.

Click ***Send*** to send out the SMS, click ***Clear Result*** to clear all results. Click ***Reset*** to restore the configurations, or click ***Return*** to go back to the previous.

3.6.5 SMPP

Short Message Peer-to-Peer (**SMPP**) is an open, industry standard protocol designed to provide a flexible data communication interface for the transfer of short message data between External Short Messaging Entities (SMEs), Routing Entities (REs) and Message Centres. Our wireless gateway working as a Short Message Service Center (**SMSC**) can receive the short message from Short Messaging Entities (SMEs), transfer it to the operator and return the Delivery Report (DLR) to SMEs according to requirements; also in the opposite direction, our wireless gateway can transfer the short message from the operator to SMEs. Up to 32 SMEs are supported to connect with SMSC simultaneously and each SME can connect to corresponding channels on the gateway. See below for the configuration items on the Basic Settings interface as well as the Account Settings interface.

Item	Description
Enabled	Sets whether to enable or disable the SMPP feature, with the default setting of No.
Server Port	The port number of the SMPP server.
Transmit	Sets whether to enable the SMS forwarding, with the default setting of No.
DLR	Sets whether to report the status of SMS forwarding, with the default setting of No.
Account	Account of the Client.
Password	Password of the Client account.
Ports	Number of the bound port

Click ***Save*** to save the settings into the gateway; click ***Reset*** to restore the configurations, or click ***Return*** to go back to the previous. To delete an account, check the checkbox before the corresponding index and click the '***Delete***' button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all accounts at a time, click the ***Clear All*** button. All the configurations will apply after the gateway's restart.

You are allowed to visit the settings interface directly by entering the gateway address plus /en/imei into the address bar of your browser.

3.6.6 SMS over SIP

The wireless gateway supports the SMS over SIP feature. See below for the configuration items on the relevant interface.

Item	Description
SIP->SMS	Enable or disable.
Mode	There are two modes available: Simple and Common. In the Simple mode, the target number sent from the short message is designated by the to field of the Head part in the SIP protocol (display or SIP account), and the channel to send the short message is selected based on the SIP account that matches the to filed and the SIP account of the gateway channel. The SIP body part is just the SMS content. In the Common mode, the target number of the short message, the channel of the gateway and the SMS content are all designated in the SIP body by character strings in a similar format of json.
SMS->SIP	Enable or disable.
Identity	By default it is the serial number of the gateway.
Mode	There are two modes available: Simple and Common. In the Simple mode, the SIP account corresponding to the receiving channel of the gateway is represented in the from field of the SIP Head part. The SIP body part is just the SMS content and the identity is invalid. In the Common mode, the identity, the receiving port of the gateway, the board slot number on the gateway, the source number of the short message, the number of the SIM card receiving the short message, the receiving time of the short message, as well as the SMS content are all designated in the SIP body by character strings in a similar format of json. The identity is by default a sentence after the serial number of the gateway telling which gateway receives the short message.
SIP Address	The IP address of the SIP platform connecting to the gateway.
SIP Port	The port address of the SIP platform connecting to the gateway.
SIP Username	The username of the SIP account on the SIP platform connecting to the gateway.

Click ***Save*** to save the above settings into the gateway, or click ***Reset*** to restore the configurations. All the configurations will apply only after you restart the system.

3.6.7 USSD

The table below explains the items on the USSD Setting interface.

Item	Description
Default USSD Encoding	Sets the default encoding format for USSD, two options available: ASCII and UCS2.
Port	Sets the port used to send the USSD request. For 16/32 port gateways which have cards inserted on the port, the USSD interface will show the card slot number for the current port as well.
Request	Inputs the content of the USSD request.
Respond	Displays the result of the USSD respond.
All	Selects all the available ports to send the same USSD request.

Click ***Send*** on the USSD Setting interface. To clear all data at a time, click the ***Clear Data*** button.

Note: This configuration is unavailable for the CDMA gateway.

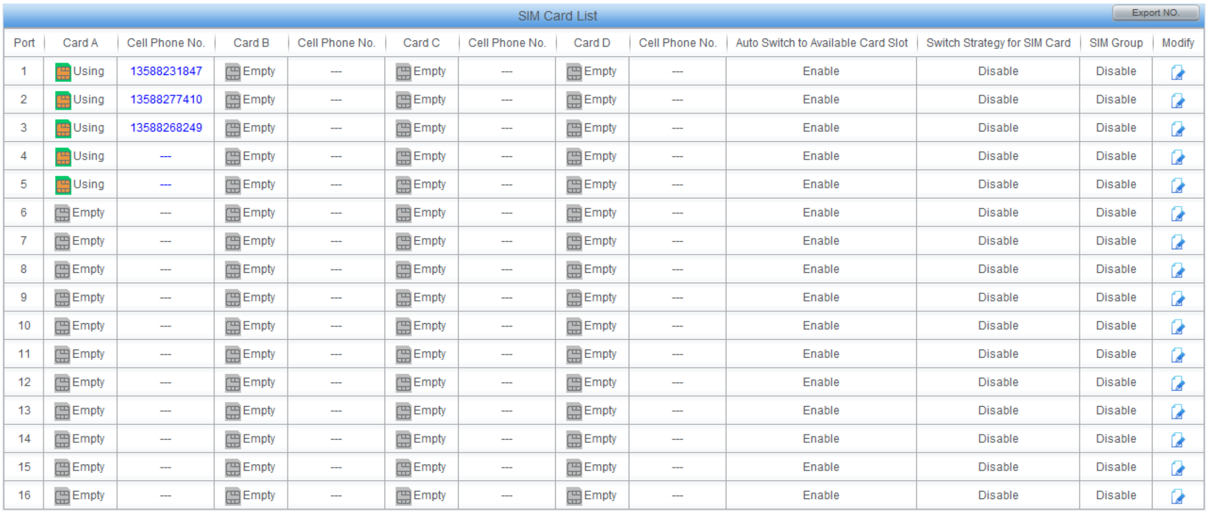
3.6.8 Email

The table below explains the configuration items on the Email Setting interface.

Item	Description
Mailbox Account, Password	Sets the account and password of the mailbox.
Outgoing (SMTP), Port	Sets the server address and port for Email sending.
Incoming (POP3), Port	Sets the server address and port for Email receiving.
SSL	Sets whether to encrypt the sending/receiving mails via SSL.
Show Log	Click it to display the log which contains the Email to SMS converted information.
Bind Mailbox to Port	Once this feature is enabled, the mailbox can be bound to the designated port. Click ***Setting*** to go into the "Bind Mailbox to Port-Settings" interface.
Convert SMS to Email	SMS can be converted to Emails if this feature is enabled.
Target Address	The target address to which the Email converted by SMS will be sent.
Subject	Sets the subject for the Email converted by SMS.
Covert Email to SMS	When this feature is enabled, the mails in a designated format can be converted to SMS.
Receiving Cycle	Sets the cycle to receive mails. Range of value: 1~60, calculated by minute, with the default value of *5*.
Mail Filtering	Sets the condition to convert the mail to SMS, two options including: Subject matching and Number matching, with the default value of *Subject matching*. If the Subject matching mode is selected, you can set the subject of your own choice, and the email format is "[Number]XXX[End] [SMS]YYY[End]; (Case Insensitive)"; If the Number matching mode is selected, the mail subject must be numbers, multiple numbers are supported which should be separated by ",", and the email format is "[SMS]......[end]".
Delete Method after Reading Mail	There are four methods to delete the mail after reading: delete all mails, delete matching mails, delete matching and size over 20k mails, never delete.
SMS Sending Port	Sets the port from which the SMS will be sent out. The default value is ***automatic***.
SMS Encoding Format	Sets the SMS encoding format used in converting Email to SMS, with the default value of UCS2. It can be configured to GSM 7bit (UCS2 is recommended in case the email contains Chinese words).
Return Receipt	Sets whether to receive a return receipt telling the mail is sent successfully or not.

After configuration, click ***Save*** to save the settings into the gateway or click ***Reset*** to reset the settings.

3.6.9 SIM Card



SIM Card List

Export NO.

Port	Card A	Cell Phone No.	Card B	Cell Phone No.	Card C	Cell Phone No.	Card D	Cell Phone No.	Auto Switch to Available Card Slot	Switch Strategy for SIM Card	SIM Group	Modify
1	Using	13588231847	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
2	Using	13588277410	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
3	Using	13588268249	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
4	Using	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
5	Using	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
6	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
7	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
8	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
9	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
10	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
11	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
12	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
13	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
14	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
15	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	
16	Empty	---	Empty	---	Empty	---	Empty	---	Enable	Disable	Disable	

Figure 3-21 SIM Card List Interface

The SIM Card List interface displays the states of each SIM card and the strategy to switch the SIM, etc. Click the SIM card in Exist state to set it to Using state, at the same time, the SIM card which is ever in Using state at first will switch to Exist state. Click Modify to modify the parameters.

The table below explains the items on the SIM Card Management interface.

Item	Description
Port	Serial number of the port on the device.
Auto Switch to Available SIM Card	Once this feature is enabled, it will switch to other available SIM card automatically if the current SIM card is drawn out or the corresponding port is unavailable due to the SIM card is damaged. The default value is *enable*.
Switch Strategy for SIM Card	Sets the switch strategy for the SIM card. There are five options: *Based on Time, Based on Call, Based on SMS, Fixed Time* and *Disable*. Among them, the option *Based on Call* provides three count methods: *Call out*, *Ring back* and *Talking*. The default value is *Disable*.
SIM Card Grouping	Once this feature is enabled, the SIM cards in the port can be divided into groups, with the default value of *disable*.
Grouping	Sets the grouping of the SIM cards.
Use Strategy	Sets the strategy to group the SIM cards, including two options: Only One Group and Two Groups in Turn.
Apply to All Ports	Sets whether to apply the above configurations to all ports.

Click ***Modify*** to save the above settings into the gateway or click ***Reset*** to restore the configurations. Click ***Return*** to cancel the modification.

Note:

1, Only the SMG4016 and SMG4032 series gateways support this configuration;

2, The priority of these three switching modes is: Auto Switch to Available Card Slot > SIM Card Grouping > Switch Strategy for SIM Card. It is suggested not to enable them simultaneously.

3.6.10 PIN Manage

PIN Manage					
Port	SIM Card State	PIN Required	PUK Required	Setting Status	Modify
1	Unlocked	No	No	---	
2	---	---	---	---	--
3	---	---	---	---	--
4	Unlocked	No	No	---	
5	---	---	---	---	--
6	---	---	---	---	--
7	---	---	---	---	--
8	---	---	---	---	--

Figure 3-22 PIN Manage Interface

The PIN Manage interface displays the status of the SIM card and the setting status of PIN and PUK. Click Modify to go into the modification interface. See Figure 3-23.

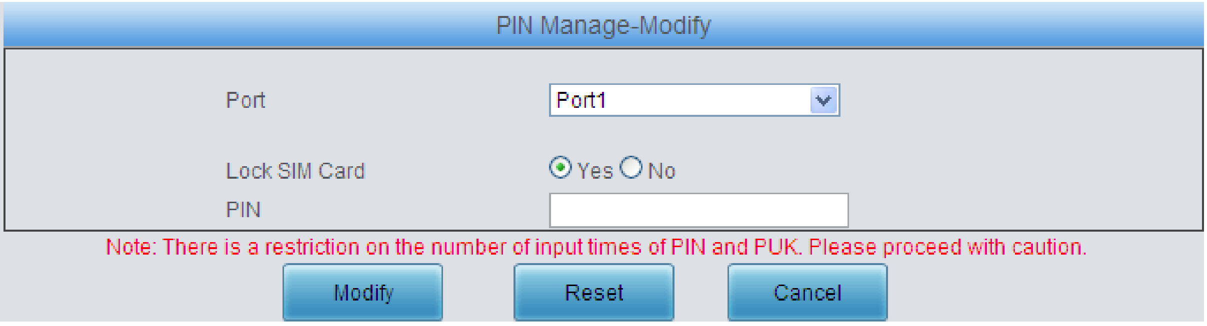


Figure 3-23 PIN Manage Modification Interface

Click “Yes” and input the correct PIN to lock the SIM card. The incoming/outgoing calls will not be initiated once the SIM card is locked. See Figure 3-24.

PIN Manage					
Port	SIM Card State	PIN Required	PUK Required	Setting Status	Modify
1	Locked	Yes	No	---	
2	---	---	---	---	--
3	---	---	---	---	--
4	Unlocked	No	No	---	
5	---	---	---	---	--
6	---	---	---	---	--
7	---	---	---	---	--
8	---	---	---	---	--

Figure 3-24 SIM Card Locked PIN Required

Click Modify in Figure 3-24, you are required to input PIN again. See Figure 3-25.

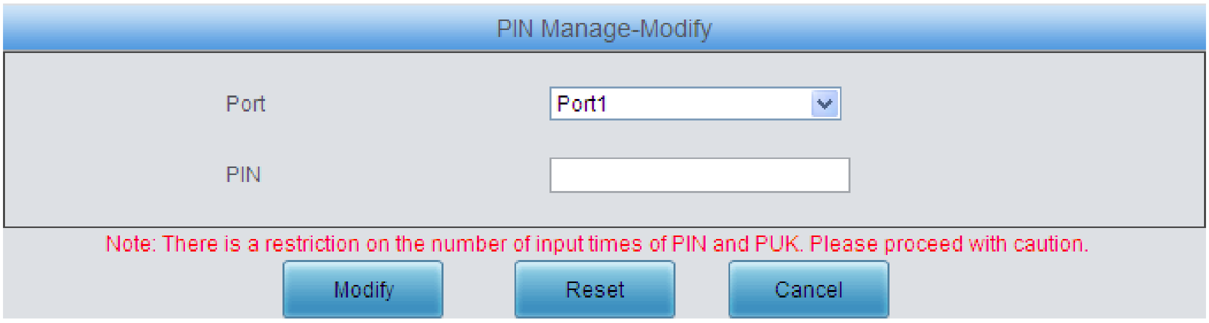


Figure 3-25 Input PIN Interface

After the correct PIN is input, the SIM card is still locked but the channel turns idle and allows the initiation of incoming/outgoing calls. See Figure 3-26.

PIN Manage					
Port	SIM Card State	PIN Required	PUK Required	Setting Status	Modify
1	Locked	No	No	Successful	
2	---	---	---	---	--
3	---	---	---	---	--
4	Unlocked	No	No	---	
5	---	---	---	---	--
6	---	---	---	---	--
7	---	---	---	---	--
8	---	---	---	---	--

Figure 3-26 SIM Card Locked without PIN

Click Modify in Figure 3-26 to unlock the SIM card or modify the PIN. See the figure below.

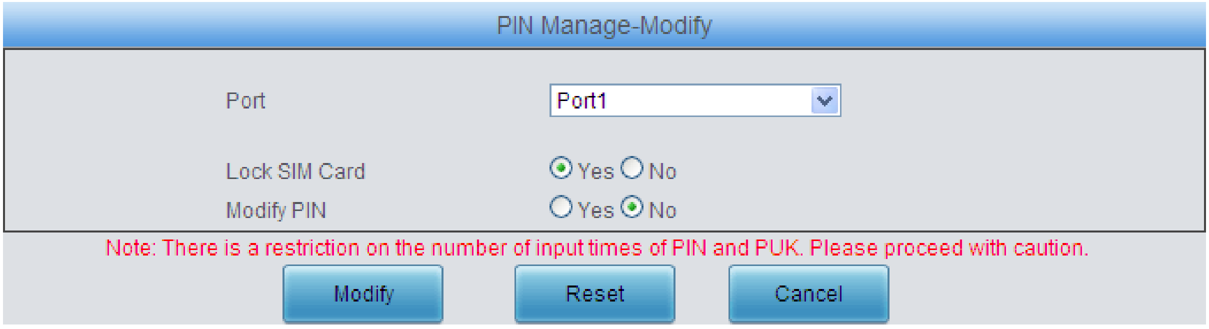


Figure 3-27 Lock SIM Card or Modify PIN Interface

The SIM card will also be locked and cannot make incoming/outgoing calls if you input a wrong PIN code three times. You are required to input the PUK to reset the PIN. See Figure 3-28.

PIN Manage					
Port	SIM Card State	PIN Required	PUK Required	Setting Status	Modify
1	Locked	Yes	Yes	---	
2	---	---	---	---	--
3	---	---	---	---	--
4	Unlocked	No	No	---	
5	---	---	---	---	--
6	---	---	---	---	--
7	---	---	---	---	--
8	---	---	---	---	--

Figure 3-28 SIM Card Locked Need PIN and PUK

Click Modify in Figure 3-28 to input PUK and reset a new PIN, see Figure 3-29.

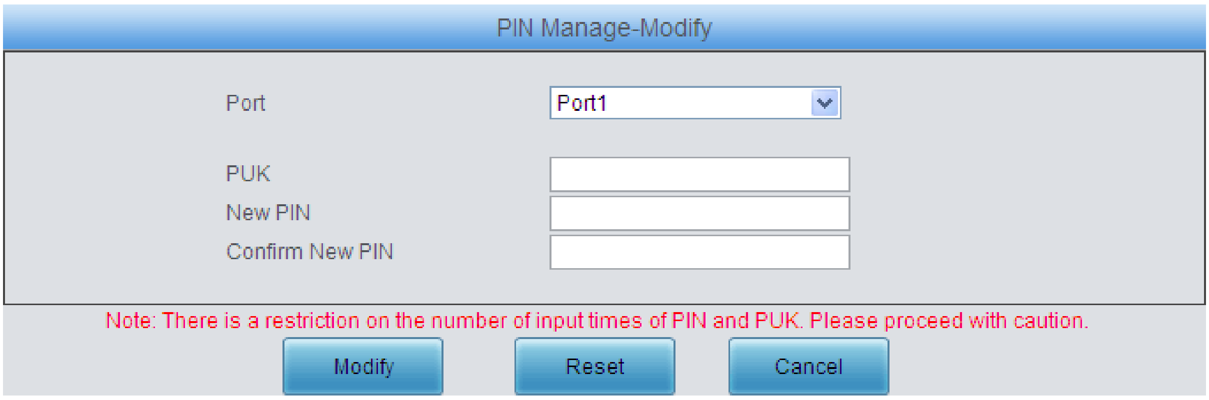


Figure 3-29 New PIN setting interface

The SIM card is still locked but do not need PIN and PUK again after inputting the correct PUK and resetting a new PIN. The status of the port displaying in Port State is idle. So the port can make incoming/outgoing calls. Click ***Modify*** to save the above settings into the gateway or click ***Reset*** to restore the configurations. Click ***Cancel*** to cancel the modification.

Note: The SIM card will be locked forever if you input a wrong PUK more than 10 times. You need to insert a new card.

3.6.11 BS Select

The Base Station Select interface displays the information of the base stations which can be searched and connected. The base station has the priority to be connected will be listed on the left according to its comprehensive ability. Click Modify to go into the Lock BS interface.

The table below explains the items shown on the Lock BS interface.

Item	Description
Port	The number of the port corresponding to that on the wireless module.
Serial No.	The serial number of the base station which can be searched.
BS	The frequency point of each base station.
LAC	The location number of each base station. It's a hexadecimal number.
CELLID	The cell number of each base station. It's a hexadecimal number.
Manual Lock	Select the serial number and click the Lock button behind to lock the base station manually. Thus, the SIM card will connect to the locked base station randomly.
Automatic Lock	Select the serial number to lock the base station automatically. The SIM card will connect to the locked base station in a cyclic order according to the set switching time.
Switching Time	Sets the switching time for connecting the base station.
BS in Use	Selects the serial number of the base station to be locked automatically.

Click ***Lock*** to save the above settings into the gateway or click ***Return*** to cancel the modification and return to the previous interface.

To cancel the lock, check the checkbox before the corresponding index on the BS Select interface and click the '***Cancel***' button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Query*** means to query the information of all the base stations which can be connected.

Note: This configuration is only supported by the GSM gateway.

3.6.12 Networking Settings

The Networking Management interface displays the networking information about the SIM card, such as Acess Mode, Month Limit, Consumed Flow, etc. Click Modify to go into the Networking Settings Modification interface. See below for the items shown on the interface.

Item	Description
Port	The number of the port corresponding to that on the wireless module.
Acess Mode	Set the Internet access mode. In the Smart mode, the set limit traffic will be uniformly consumed during the day; in the Config mode, the traffic will be consumed on the Internet once any of the trigger conditions are met in the configured Internet time; in the fast mode, the set limit traffic will be consumed as soon as possible; and the default setting is disable, which means not to enable the network access.

URL Configuration	Sets the URL address. See the following requirement before you fill in the addresses.
APN	Set the APN. Consult the corresponding operator for details. APN is automatically selected without configuration, and the card will automatically take the corresponding APN.
Month Limit	Set the amount of traffic that the SIM card needs to consume in a month. When this value is reached, the SIM card will stop accessing.
Time	In the Config mode, it is used to set the time for the SIM card to access the Internet.
SIM Card Online Trigger	In the Config mode, it is used to set the online time of the SIM card that triggers the Internet access. Once the random time within the set range is reached, the Internet access will be triggered.
Call Trigger	In the Config mode, it is used to set the count of calls that triggers the Internet access. Once the random number of calls within the set range is reached, the Internet access will be triggered.
Talk Trigger	In the Config mode, it is used to set the count of talking calls that triggers the Internet access. Once the random number of talking calls within the set range is reached, the Internet access will be triggered.
Talk Time Trigger	In the Config mode, it is used to set the talking duration that triggers the Internet access. Once the random talking duration within the set range is reached, the Internet access will be triggered.
Apply to Other Ports	Sets whether to apply the above configurations to other ports or port groups.

When this feature is disabled, you can click the test button after setting URL and APN to test the consumed traffic. And the test results will be soon displayed automatically.

Click ***Save*** to save the above settings into the gateway or click ***Cancel*** to cancel the modification.

Note:

This configuration is only supported by the WCDMA, LTE and GSM gateways.

The automatic selection of APN only supports the cards of the three major domestic operators.

3.6.13 AMD

The AMD Configuration interface is used to set the parameters for judging whether the phone is picked up by a man or not. See the table below for details.

Item	Description
AMD Detection for Outgoing Call	Sets whether to enable the AMD detection while making an outgoing call, with the default value of *Disabled*.
Line Silence Overtime after Dial Tone	Judges if the line silence after dial tone lasts overtime or not, calculated by ms, with the default value of *30000*.
Silence Overtime after tone or Color Ring Being Detected	Judges if the silence after tone or color ring lasts overtime or not, calculated by ms, with the default value of *15000*.
Overtime for a Complete AMD Detecting Process	Judges the whole AMD detecting process overtime or not, calculated by ms, with the default value of *70000*.

Upper Limit of Detected Continuous Tones	Judges if the tone detected time is overtime or not.
Shortest Voice Duration at ON State	Sets the shortest duration when the voice goes into the High voltage state, calculated by ms, with the default value of *150*.
Shortest Voice Duration at OFF State	Sets the shortest duration when the voice goes into the low voltage state, calculated by ms, with the default value of *400*.
Maximum Greeting Duration at OFF State	Sets the longest duration of the greetings at the OFF state after a call is picked up by a man, calculated by ms, with the default value of *0*.
Shortest Silence Duration before Greeting	Sets the shortest silence duration before the phone is picked up by a man, calculated by ms, with the default value of *600*.
Shortest Greeting Duration	Sets the shortest greeting duration in case the phone is picked up by a man, calculated by ms, with the default value of *180*.
Maximum Greeting Duration	Sets the longest greeting duration in case the phone is picked up by a man, calculated by ms, with the default value of *1200*.
Shortest Silence Duration after Greeting	Sets the shortest silence duration after the phone is picked up by a man, calculated by ms, with the default value of *1200*.
Silence Energy Threshold	Sets an energy value that can judge the voice is silence or not, calculated by ms, with the default value of *180*.
Energy Difference Proportion of Tone	Sets the difference proportion of the high and low energies in the signal.
Output AMD Debugging Info to Syslog	Sets whether to output the AMD debugging information to Syslog.
Do not Detect Other Pickup Signal	Sets whether to detect other pickup signals.

Click ***Save*** to save the settings into the gateway, click ***Reset*** to restore the configurations.

Note: This configuration is only supported by the CDMA gateway.

3.6.14 Hidden CallerID

The Hidden Caller Setting interface sets whether to hide the CallerID to the called party. This feature requires the support of the operator. Select the port and click ***Open*** to enable the feature, and click ***Close*** to disable it.

Note: This configuration is only supported by the WCDMA and LTE gateways.

3.6.15 SIM Mode

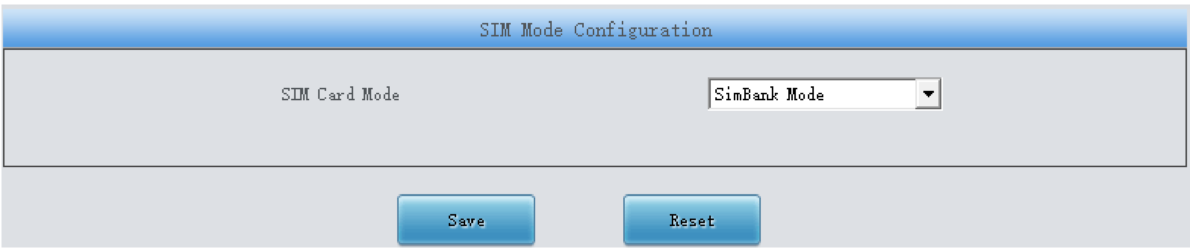


Figure 3-30 SIM Mode Setting Interface

The SIM Mode Setting interface is used to set the SIM mode of the SIMBANK, with three options available: Local, SimBank, LAN. In the Local mode, the SIMBANK is not connected to other devices; in the SIMBANK mode, you have to ensure the centralized management feature has been enabled, and then the SIMBANK can connect and work with the wireless gateway on the centralized management platform; in the LAN mode, you should configure Gateway IP address herein for the SIMBANK, select the LAN mode and configure SIMBANK IP address for the wireless gateway, and then connect the SIMBANK with the wireless gateway in the LAN.

Note: This configuration is only supported by the GSM, CDMA and LTE gateways.

3.6.16 Call Waiting

The Call Waiting Setting interface is used to enable or disable the call waiting feature for corresponding modules. Select one or more ports, click Save to enable the call waiting feature. The state column on the top shows the setting result.

Note: This configuration is not supported by the CDMA gateway.

3.6.17 Lock SIM

This feature is designed to lock the card slots on the gateway. A card slot cannot be used to start calls any more after being locked. There are three ways to lock a SIM card: auto locking based on certain call condition, manual locking, and locking via API request of the gateway. For detailed configurations, go to the Lock SIM interface.

On the Lock SIM interface, click the ***Modify*** button on the right to go into the Lock SIM Set interface and you can see the following configuration items.

Item	Description
Port	The number of the port on the gateway.
Auto Lock SIM Card after Consecutive No Answer Count	Sets how many continuous calls of no answer will cause the card lock.
Locking Time	Sets the locking time of the card slot.
Unlock by Plugging SIM Card in and out	Sets whether to unlock the card slot after pluging the SIM card in or out. Note: This configuration is unsupported by SMG4002, SMG4004, SMG4008 as well as G1, W1, LC1 series gateways.
Auto Lock SIM Card after Consecutive No Carrier Count	Sets how many continuous calls of no carrier will cause the card lock.
Auto Lock SIM Card after Successful Call Count	Sets how many continuous calls will cause the card lock.
Auto Lock SIM Card after Consecutive Short Call Count	Sets how many continuous short calls will cause the card lock. A call shorter than the set time (e.g. 5s) will be regarded as a short call.
Short Call Duration	Set the time duration for a short call.
Auto Lock SIM Card after Consecutive Fast Answer Count	Sets how many continuous calls of fast answer will cause the card lock.
Fast Answer Time	A call which is answered in the set time will be regarded as a fast answer call. This set time is the fast answer time and all answers in this time are fast answers.

Auto Lock SIM Card after Consecutive Busy Count	Sets how many continuous calls in busy will cause the card lock.
Auto Lock SIM Card after Consecutive Dial Failure Count	Sets how many continuous calls with dial failure will cause the card lock.
Auto Lock SIM Card after Consecutive Hangup Count	Sets how many continuous calls which are hung up by the caller will cause the card lock.
Switch Card after Lock	Switches the card after it is locked. Note: SMG4002, SMG4004, SMG4008 series, as well as G1, W1, and LC1 series gateways do not support this configuration.
Unlock by Plugging (Manual Lock Only)	Unlock the card after manually inserting or removing the card. Note: Only supported on the webpage.
LockCardEmailAlarm	Sends an email alarm after the card is locked. To set this feature, you need to fill in the email address.
LockCardSMSAlarm	Sends SMS alarm after the card is locked. To set this feature, you need to fill in the mobile phone number.
Apply to All Ports	Sets whether to apply these configurations to all ports on the gateway.

Click ***Modity*** to save the modified settings into the gateway, click ***Reset*** to restore the configurations, or click ***Return*** to go back to the Lock SIM interface. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page. Click ***SIM Card*** to select the corresponding card slot to lock and unlock; ***Unlock*** means to unlock the locked card slot. After that, calls are allowed on this slot as long as it is in use. ***LockUnSwitch*** means to lock those card slots which meet locking requirements or need locking manually. After that, calls are forbidden on such slot as long as it is in use. ***LockAndSwitch*** means to lock the corresponding SIM card slot and swich the card.

The SIM Lock List interface will display locked card slots at present.

Note: ***LockAndSwitch*** is not supported by single-cardslot gateways.

3.7 Call Management

Call Management includes eight parts: ***Balance***, ***Port Limit***, ***Name List Timer***, ***Tel→IP Auto Route***, ***Blacklist, SMS Count, Auto Function*** **and** ***Port Charge***. ***Balance*** is used to query the remaining time and balance of a cell phone number; ***Port Limit*** is used to calculate the call time length of the corresponding number; ***Name List Timer*** is used to set the timing rule to count and manage the call time of the target number; ***Tel→IP Auto Route*** is used to set the route for the remote end to call back; ***Blacklist*** is used to set a number table to forbid some incoming calls; ***SMS Count*** is used to calculate the number of short messages from a phone number corresponding to a port; ***Auto Function*** is used to make calls and send SMS from port to port in a special condition so as not to be blocked by the operator; ***Port Charge*** is used to count the call fees for a phone number corresponding to a port.

3.7.1 Balance

Via the Balance Query interface, you can query the balance of a designated cell phone number. Click Modify to modify the query mode.

The table below explains the configuration items on the Query Mode Modification interface.

Item	Description

Query Mode	Sets the mode to query. There are three options available: SMS, ATD, USSD.
Destination Number	Sets the destination number to query
Content to Send	Sets the content to send.
Keywords to Match	The balance matching the keywords will be displayed.
Query after SIM Card Registered	Sets whether to query the balance automatically once the SIM card is registered to the base station.
Query Regularly	Sets the time to query the balance regularly.
Alarm for Insufficient Balance	Once this feature is enabled, the gateway will notify the users by sending SMS or Email once the balance goes insufficient. The default value is *disabled*.
Alarm Threshold	Sets the threshold for the insufficient balance to send the alarm.
Alarm via SMS, Alarm via Email	Sets the addresses to receive the SMS/Email while the balance is insufficient.
Alarm via Web	Once this feature is enabled, the alarm information concerning the Insufficient Balance will be displayed on the web.
Apply to Other Ports	Sets whether to apply these query conditions to other ports or port groups.

Click ***Modify*** to save the above settings into the gateway or click ***Reset*** to restore the configurations. Click ***Cancel*** to cancel the modification. Click ***Test*** to set a balance query strategy, and then execute it to test the balance query feature. And this can help to set a proper balance query strategy. See Figure 3-31.

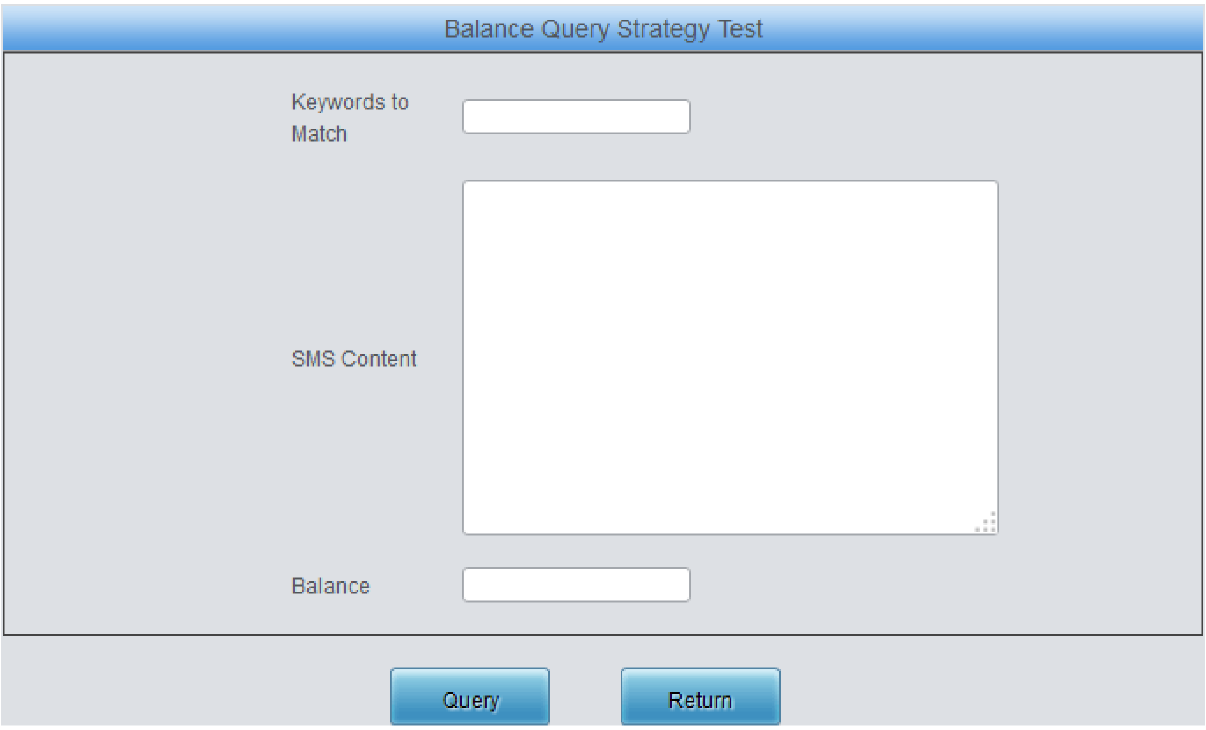


Figure 3-31 Balance Query Strategy Test Interface

Enter the ***Keywords to Match*** and ***SMS Content***, then click ***Query*** to query the information about the balance.

3.7.2 Port Limit

The Port Limit interface displays such information as the call time limit on the number corresponding to the port, the timer clear cycle as well as the alarm for the call time allowance.

Click Modify for each port to modify the port limit settings.

The table below explains the configuration items on the Port Timing Setting interface.

Item	Description
Port	The number of the port corresponding to the wireless module.
Unit	Sets the timing unit for the call，eight options available: 1s, 5s, 10s, 20s, 30s, 40s, 50s and 60s. The actual call time will be calculated as the integral multiple of the setting time. Take an example: supposed the setting time is 30s and the actual call time is 72s, thus, the gateway will consider the call time as 90s.
Store Call Time to SIM	Sets whether to write the time duration of calls on the card slot to the SIM card, with the default setting of unticked. Once it is ticked, the call time duration will be only written into the SIM card, but not saved to the Calltime file. This feature is supported by all gateways except for the ZTE module of the CDMA gateway.
Update Call Time to SIM	Sets whether to update the time duration of calls in gateway to the SIM card, with the default setting of ticked.
Time Limit on a Single Call	Sets whether to enable the time limit on a single call.
Max Call Time	Sets the maximum time length of a call.
Time Limit on a Single Day	Sets whether to enable the time limit on calls in a single day.
Switch to Other Card if no Time	Sets whether to switch to other available SIM card if the current SIM card has no available time to make calls. Note: This configuration is unsupported by SMG4002, SMG4004, SMG4008 as well as G1, W1, LC1 series gateways.
Time Limit on Total Calls	Sets whether to enable the time limit on all calls at the port.
Timing Cycle	Sets the time count cycle for the port.
Clear	Sets the time node to clear the time count.
Set Spent Call Time	Sets the spent call time length of the port.
Alarm for Call Time Allowance	Once this feature is enabled, when the remaining call time of the port is less than the alarm threshold value, the gateway will send the alarm information.
Allowance Alarm Threshold	Sets the threshold value for the remaining call time.
Alarm via SSM, Alarm via Email	Sets the way to send the alarm information. The gateway can send the alarm information via both SMS and Email or either of them.
Call Count Limit	Sets limit to calls on a gateway port by call count.
Count Method	Sets the way to count calls, four options available: *Call Out, Ring Back, Talking, Failed*.
Count Cycle	Sets the cycle to count calls. The count will be cleared automatically before a new cycle. Five settings are supported. They are *Month, Day, Hour, Halfhour, Minute*. And you can click the following '+' button to add or remove configuration boxes.
Max Call Count	Sets the maximum count of calls in a cycle.

Switch to Other Card if Count is Full	Sets whether to switch the current SIM card with full call count to another available SIM card. Note: This configuration is unsupported by SMG4002, SMG4004, SMG4008 as well as G1, W1, LC1 series gateways.
Apply to Other Ports	Sets whether to apply above settings to other ports or port groups.

Click ***Modify*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Return*** to cancel the settings.

Note: This feature is unsupported in the SimBank mode.

3.7.3 Name List Timer

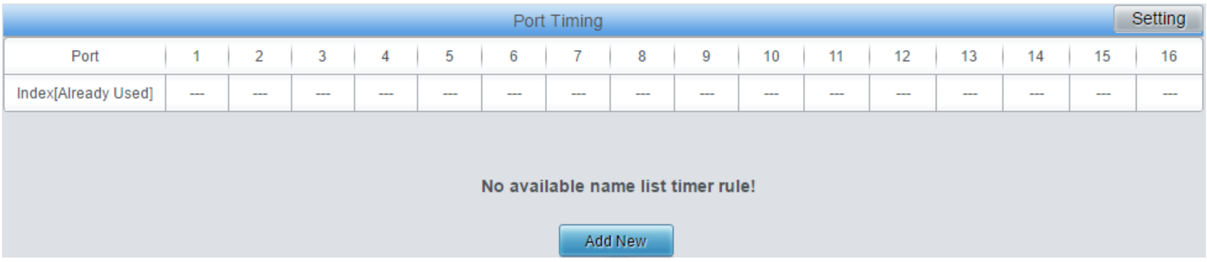


Figure 3-32 Name List Timer Interface

See Figure 3-32 for the Name List Timer interface, which contains two parts: Port Timing and Name List Timer Rule. You can add the timing rule to count the call time for the port. Click ***Add New*** in Figure 3-32 to add a timing rule.

The table below explains the configuration items on the Add Name List Timing Rule interface.

Item	Description
Number	Sets the number to be timed.
Import Number	Used to import the files on which the numbers need to be timed.
Number Matching Rule	Sets the rule to match the numbers, two options available: Prefix Matching and Whole Words only, with default value of *Prefix Matching*.
Max Call Time	Sets the maximum time length for a call.
Timing Cycle	Sets the timing cycle for the port, four options available: Day, Week, Month, Year.
Clear	Sets the time node to clear the timing.

Click ***Save*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Return*** to cancel the settings. After adding the timing rules, click ***Setting*** button on the up right corner in Figure 3-32 to set the timing rule for each port. See Figure 3-33 for the setting interface.

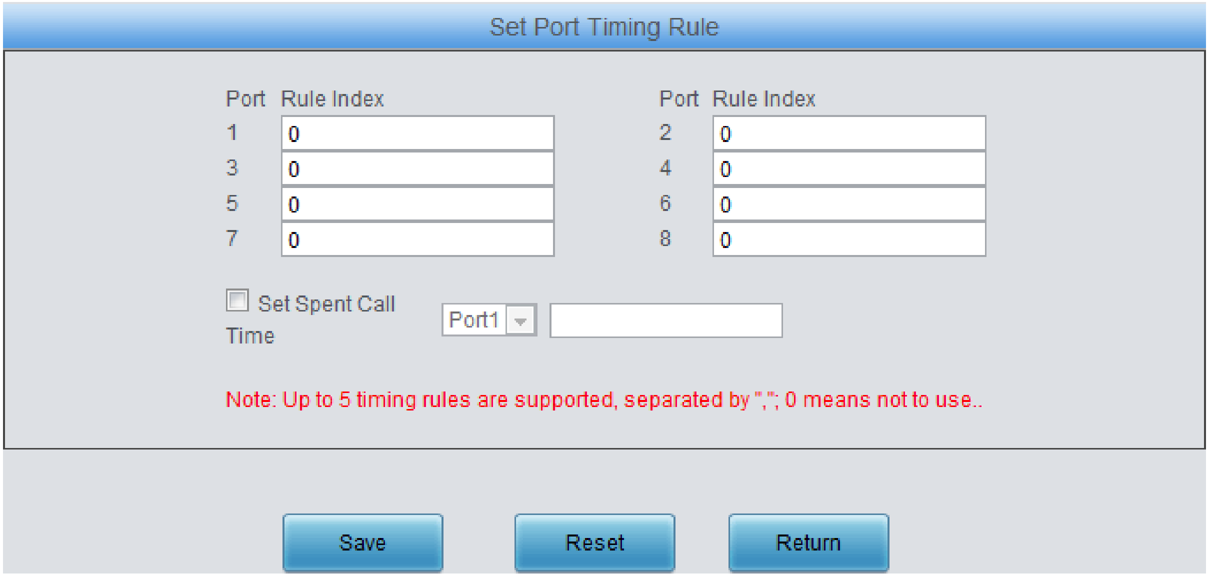


Figure 3-33 Set Port Timing Rule Interface

The table below explains the configuration items shown in the above figure:

Item	Description
Rule Index	The index number of the timing rule corresponding to the port.
Set Spent Call Time	Sets the call time already used by the port.

Click ***Save*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Return*** to cancel the settings.

3.7.4 Tel to IP Auto Route

The Tel→IP Auto Route is used to set routes for the remote phone to call back the gateway. By default, there is no available auto route for a Tel→IP call, click the ***Setting*** button to set it.

The table below explains the configuration items on the Tel→IP Auto Route Settings interface.

Item	Description
Route Holding Time	Sets the valid time of the route.
Route Calls back to IP Side	Once this feature is enabled, the calls from the PSTN terminal to the soft terminal will be routed to the soft terminal of the original call. That is, when the soft terminal A called the PSTN terminal B via our gateway, if B doesn't answer the call or A cancels the call, the later call from B back to the gateway will be routed to the soft terminal A directly. Once this feature is disabled, the later call dialed back by the remote terminal B will be routed to the original calling party A only if B doesn't answer the first call from A to B.

Click ***Modify*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Return*** to cancel the settings.

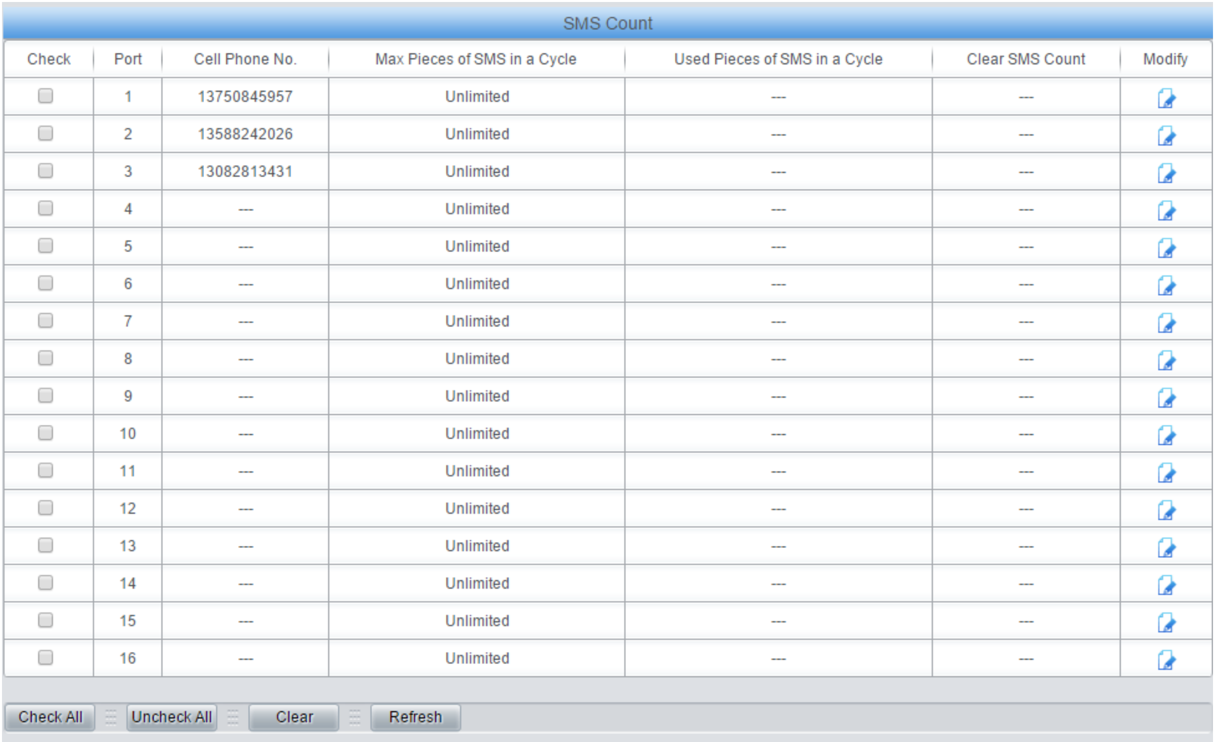
3.7.5 Blacklist

On the Incoming Call Blacklist interface, you can designate certain numbers to limit corresponding calls to go into the gateway (calls from the gateway out as well as the SMS are unlimited). The table below explains the configuration items on this interface.

Item	Description
Blacklist	Sets the number list to forbid certain calls to go into the gateway.
Processing Mode	Sets the processing mode for the calls from the numbers in the blacklist to the gateway, two options available: Hang up directly and Hang up after ringing, with the default value of *Hang up directly*.

Click ***Save*** to save the settings into the gateway, click ***Reset*** to restore the configurations.

3.7.6 SMS Count



SMS Count

Check	Port	Cell Phone No.	Max Pieces of SMS in a Cycle	Used Pieces of SMS in a Cycle	Clear SMS Count	Modify
☐	1	13750845957	Unlimited	---	---	
☐	2	13588242026	Unlimited	---	---	
☐	3	13082813431	Unlimited	---	---	
☐	4	---	Unlimited	---	---	
☐	5	---	Unlimited	---	---	
☐	6	---	Unlimited	---	---	
☐	7	---	Unlimited	---	---	
☐	8	---	Unlimited	---	---	
☐	9	---	Unlimited	---	---	
☐	10	---	Unlimited	---	---	
☐	11	---	Unlimited	---	---	
☐	12	---	Unlimited	---	---	
☐	13	---	Unlimited	---	---	
☐	14	---	Unlimited	---	---	
☐	15	---	Unlimited	---	---	
☐	16	---	Unlimited	---	---	

Check All | Uncheck All | Clear | Refresh

Figure 3-34 SMS Count Interface

See Figure 3-34 for the SMS Count interface, which displays such information as the maximum pieces of SMS in a cycle, the used pieces of SMS in a cycle as well as the clear operation. Click Modify for each port in Figure 3-34 to modify the SMS count settings.

The table below explains the configuration items on the SMS Count Configuration interface.

Item	Description
Port	The number of the port corresponding to the wireless module.
SMS Amount Limit	Sets whether to enable the limit on the amount of SMS on a port.
Max Pieces of SMS	Sets the maximum amount of SMS.
Count Cycle	Sets the SMS counting cycle for the port
Clear	Sets the time node to clear the SMS count.
Apply to Other Ports	Sets whether to apply above settings to other ports or port groups.

Click ***Modify*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Return*** to cancel the settings.

3.7.7 Auto Function

You can set via the Auto Function Settings interface to implement automatic calls and SMS from port to port in some special conditions. The table below explains the configuration items shown in the above figure:

Item	Description
Port-to-port Call	When this feature is enabled, the gateway will make calls from port to port once the set condition is triggered.
Min. Call Duration	The minimum call time for the port-to-port call.
Max. Call Duration	The maximum call time for the port-to-port call.
Auto Send SMS	When this feature is enabled, the gateway will send SMS from port to port once the set condition is triggered.
SMS-1 (random)	Once the feature ***Auto Send SMS*** is enabled, the gateway will choose one piece at random from the set SMS to send.
By Device Runtime	When this feature is enabled, as long as the device runtime reaches the set time, the gateway will automatically enable the feature ***Port-to-port Call*** or ***Auto Send SMS***.
Min. Runtime	The minimum runtime of the device.
Max. Runtime	The maximum runtime of the device.
By Accumulated Call Duration	When this feature is enabled, as long as the accumulated call time of a port reaches the set time, the gateway will make calls or send messages between this port and its bound port.
Accumulated Call Duration	The accumulated call time of a port. When it reaches or gets greater than the set value, the feature ***Port-to-port Call*** or ***Auto Send SMS*** will be triggered.
By Amount of Consecutive Calls Out	When this feature is enabled, as long as the amount of consecutive calls out from a port reaches the set time, the gateway will make calls or send messages between this port and its bound port.
Amount of Consecutive Calls Out	The amount of consecutive calls out from a port. When it reaches or gets greater than the set value, the feature ***Port-to-port Call*** or ***Auto Send SMS*** will be triggered.

Click ***Save*** to save the settings into the gateway, click ***Reset*** to restore the configurations.

3.7.8 Port Charge

The Port Charge interface displays such information as the first and second billing cycles and rates, the total expense, the spent amount of the call expense, the clear operation as well as the no balance alarm. Click Modify for each port to modify the SMS count settings.

The table below explains the configuration items on the Call Billing Settings interface.

Item	Description
Port	The number of the port corresponding to the wireless module.
Billing Limit	Sets whether to enable the cost limit on calls for a port.
Charge Mode	The way to charge, providing two options: ***General Mode***, ***Prefix Matching***.
Charge Rule Index	The index of charge rule which appears only in the ***Prefix Matching*** mode. Each rule has a unique index.

First Billing Cycle	The first period of a call to charge, e.g. the first 1 minute.
First Billing Rate	The charge for the first period of a call, e.g. 5 yuan for the first 1 minute.
Second Billing Cycle	Each period after the first one of a call to charge
Second Billing Rate	The charge for each period after the first one of a call
Max Charge	Sets the maximum charge for a call.
Switch to Other Card if No Balance	Sets whether to switch the SIM card to another automatically if it has no balance.
Billing Cycle	Sets the billing cycle for the port.
Clear	Sets the time node to clear the charge.
Set Spent Amount	Sets the spent amount of call fees for the port.
No Balance Alarm	When this feature is enabled, the gateway will send an alarm once the balance in the SIM card goes insufficient.
Alarm Threshold	Sets the threshold for the insufficient balance to send the alarm.
Alarm via SMS ***Alarm via Email***	Sets the way to send the alarm information. The gateway can send the alarm information via both SMS and Email or either of them to the corresponding number or mailbox.
Apply to Other Ports	Sets whether to apply above settings to other ports or port groups.

Click ***Modify*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Return*** to cancel the settings.

Press the ***Add New*** button on the bottom right corner of the Port Charge inteface to add a new charge rule. See below for details.

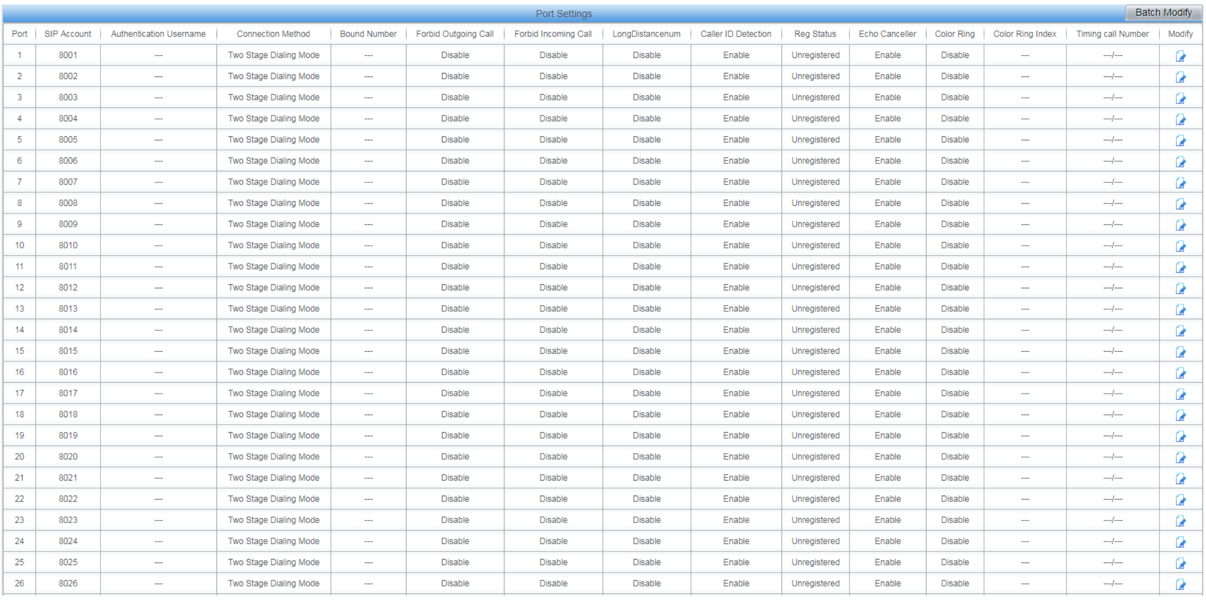
Item	Description
Index	The unique index of each charge rule.
Number Prefix	The prefix of called party numbers in the charge rule. All calls with called party numbers having such prefix will be charged by this rule.
Bill Cycle	The cycle to charge.
Bill Rate	The bill for each charge cycle.

After your configuration, click ***Save*** to save the above settings into the gateway, click ***Reset*** to restore the configurations, and click ***Return*** to cancel your modification.

3.8 Port Settings

Port Settings includes two parts: ***Port*** and ***Port Group***.

3.8.1 Port



Port Settings

Batch Modify

Port	SIP Account	Authentication Username	Connection Method	Bound Number	Forbid Outgoing Call	Forbid Incoming Call	LongDistancenum	Caller ID Detection	Reg Status	Echo Canceller	Color Ring	Color Ring Index	Timing call Number	Modify
1	8001	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
2	8002	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
3	8003	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
4	8004	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
5	8005	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
6	8006	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
7	8007	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
8	8008	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
9	8009	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
10	8010	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
11	8011	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
12	8012	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
13	8013	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
14	8014	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
15	8015	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
16	8016	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
17	8017	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
18	8018	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
19	8019	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
20	8020	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
21	8021	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
22	8022	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
23	8023	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
24	8024	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
25	8025	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	
26	8026	---	Two Stage Dialing Mode	---	Disable	Disable	Disable	Enable	Unregistered	Enable	Disable	---	---/---	

Figure 3-35 Port Settings Interface

The Port Settings interface shows the feature and properties of each port. Click ***Modify*** in Figure 3-35 to modify the properties of the corresponding port.

The table below explains the configuration items on the port modification interface.

Item	Description
Port	Serial number of the port on the device.
Register Port	Sets whether to register the port to the SIP server. When this item is set to *No*, the item **Reg Status** on the Port Settings interface (Figure 3-35) shows *Unregistered*; when this item is set to *Yes*, the item **Reg Status** shows Failed or Registered.
SIP Account	When the port initiates a call to SIP, this item corresponds to the username of SIP. The default SIP account is 80XX among which XX represents the corresponding port number. For example, the default SIP account corresponding to Port 1 is 8001, and that corresponding to Port 8 is 8008.
Password	Registration password of the port. To register a port to the SIP server, both items **SIP Account** and **Password** must be filled in.

<table>
<tr><td>Connection Method</td><td>Port connection methods include:<table><tr><th>Option</th><th>Description</th></tr><tr><td>Static Binding</td><td>Bind the number to a wireless port. The number will be listed in the Bound Number column.</td></tr><tr><td>Two Stages Dialing Mode (default)</td><td>Under this mode, an incoming call from a wireless port will go into the IVR system. Then IVR will play a speech prompt “Please dial the extension number”. If you fail to input the correct target number before IVR finishes the third repeat of the prompt, the port will hang up the call automatically; otherwise, the call goes out successfully.</td></tr></table>Note: Both items Connection Method and Bound Number will be hidden if the SIP Station feature is enabled on the SIP Settings interface.</td></tr>
<tr><td>Echo Canceller</td><td>The echo cancellation feature for a call conversation over the wireless channel. By default, this feature is enabled and the effect can reach 128ms.</td></tr>
<tr><td>Forbid Outgoing Call</td><td>If this feature is enabled, the port will be forbidden to call out. The default setting is disabled.</td></tr>
<tr><td>Forbid Incoming Call</td><td>If this feature is enabled, the port will be forbidden to call in. The default setting is disabled.</td></tr>
<tr><td>Caller ID Detection</td><td>If this feature is enabled, the port will detect the Caller IDs from the incoming calls. The default setting is enabled.</td></tr>
<tr><td>Timing Call Number</td><td>Used for regular outgoing calls, you can specify the time to call the filled number.</td></tr>
<tr><td>LongDistanceNum</td><td>You can enable this function when you need to make an international call. The default setting is Disable.</td></tr>
</table>

After configuration, click ***Modify*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or click ***Cancel*** to cancel the settings.

Or you can click ***Batch Modify*** in Figure 3-35 to modify several pieces of port settings at the same time.

Some configuration items are the same as those on the ***Port Modification Interface***. The others are described in the table below.

Item	Description
Starting Port	The starting serial number of the port on the device in the batch setting.
Ending Port	The ending serial number of the port on the device in the batch setting.
Register Port	Sets whether to register the port to the SIP server.
Starting SIP Account	The starting SIP account in the batch setting.
Starting Authentication Password	The starting authentication password in the batch setting.
SIP Account Batch Rule	The rule for batch setting the SIP account, including ***Increase*** and ***Decrease*** two options.
SIP Account Batch Step Size	Sets the increase or decrease step size of the SIP account in the batch setting.
Authentication Password Batch Rule	The rule for batch setting the authentication password, including ***Increase***, ***Decrease*** and ***All Same*** three options.

Authentication Password Batch Step Size	Sets the increase or decrease step size of the authentication password in the batch setting.
Bound Number Step Rule	It appears when the connection method is set to Static Binding, used to configure the step rule of the bound number in the batch setting, three options available: Increase, Decrease, Same.
Bound Number Step Size	It appears when the connection method is set to Static Binding, used to configure the increase or decrease step size of the bound number in the batch setting,

After configuration, click ***Save*** to save the settings into the gateway, or click ***Cancel*** to cancel the settings.

3.8.2 Port Group

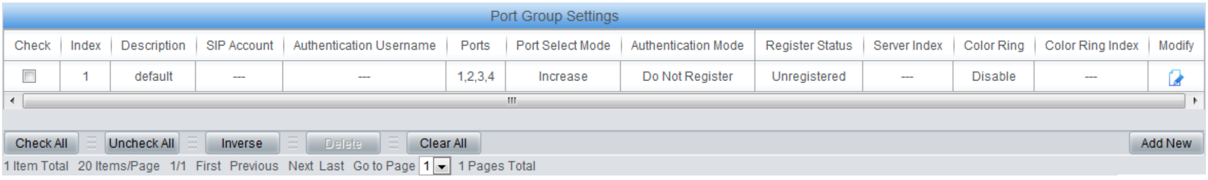


Figure 3-36 Port Group Settings Interface

See Figure 3-36 for the port group settings interface. A port group is a set containing single or multiple ports, used to specify such properties as ***Port Selection*** and ***Authentication Mode*** for all the ports in it. A new port group can be added by the ***Add New*** button on the bottom right corner of the above list. Note that a port which has been occupied by one port group cannot be chosen by others.

The table below explains the items on the Port Group Adding interface.

Item	Description
Index	The unique index of each port group, which is mainly used in the configuration of routing rules and number manipulation rules to correspond to port groups.
Description	More information about each port group, with default value of *default*.
Register Port Group	To register the port group to the SIP server. Only when this configuration item is set to *Yes* can you see the configuration items ***SIP Account*** and ***Password***.
SIP Account	When the port group initiates a call to SIP, this item corresponds to the username of SIP.
Password	Registration password of the port group. To register the port group to the SIP server, both configuration items ***SIP Account*** and ***Password*** should be filled in.
Authentication Username	Authentication username of a port, used to register the port to the SIP server when IMS network is enabled. **Note: This item appears only when IMS Network is enabled.**
Server Index	The index of the sip server which will be quoted by the current port.

<table>
<tr><td>Authentication Mode</td><td>Sets the way for SIP to make outgoing calls (Tel→IP) on the gateway.
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Do Not Register (default)</td><td>SIP initiates a call in a point-to-point mode.</td></tr>
<tr><td>Register Port Group</td><td>SIP initiates a call with the registered SIP account and password of the port group.</td></tr>
<tr><td>Register Port</td><td>SIP initiates a call with the registered SIP account and password of the port.</td></tr>
</table></td></tr>
<tr><td>Register Status</td><td>Registration status of the port group. See Figure 3-36. When Register Port Group is set to No, the value of this item is Unregistered; when Register Port Group is set to Yes, the value of this item may be Failed or Registered.</td></tr>
<tr><td>Port Select Mode</td><td>When the port group receives a call, it will choose a port based on the select mode set by this configuration item to ring or to connect. The optional values and their corresponding meanings are described in the table below.
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Increase (default)</td><td>Search for an idle port in the ascending order of the port number, starting from the minimum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Decrease</td><td>Search for an idle port in the descending order of the port number, starting from the maximum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Cyclic Increase</td><td>Provided Port N is the available port found last time. Search for an idle port in the ascending order of the port number, starting from Port N+1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Cyclic Decrease</td><td>Provided Port N is the available port found last time. Search for an idle port in the descending order of the port number, starting from Port N-1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Group Ringing</td><td>Ring all the idle wireless ports in this port group.</td></tr>
</table></td></tr>
<tr><td>Color Ring</td><td>Sets whether to enable the color ring feature or not, with the default setting of being disabled.
Note: Only when there are available color rings and the “Port Select Mode” is set to Grouping Ringing will this item appear.</td></tr>
<tr><td>Color Ring Index</td><td>The index of the color ring which is quoted by the current wireless port.</td></tr>
<tr><td>Port</td><td>The ports in the port group. If the checkbox before a port is grey, it indicates that the port is not available or has been occupied. All selected ports for a port group will be displayed in the Ports column in Figure 3-36. Note: When a port group contains multiple ports, the automatic call forward feature is invalid.</td></tr>
</table>

After configuration, click ***Save*** to save the settings into the gateway, click ***Cancel*** to cancel the settings. ***Check All*** means to select all available ports on the current page; ***Inverse*** means to uncheck the selected items and check the unselected.

Click ***Modify*** in Figure 3-36 to modify the properties of a port group. The configuration items on the Port Group Modification interface are the same as those on the ***Add New Port Group*** interface.

To delete a port group, check the checkbox before the corresponding index in Figure 3-36 and click the '***Delete***' button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all port groups at a time, click the ***Clear All*** button in Figure 3-36.

3.9 Route Settings

Route Settings is used to specify the routing rules for calls on two directions: IP→Tel/IP and Tel→IP.

3.9.1 Routing Parameters

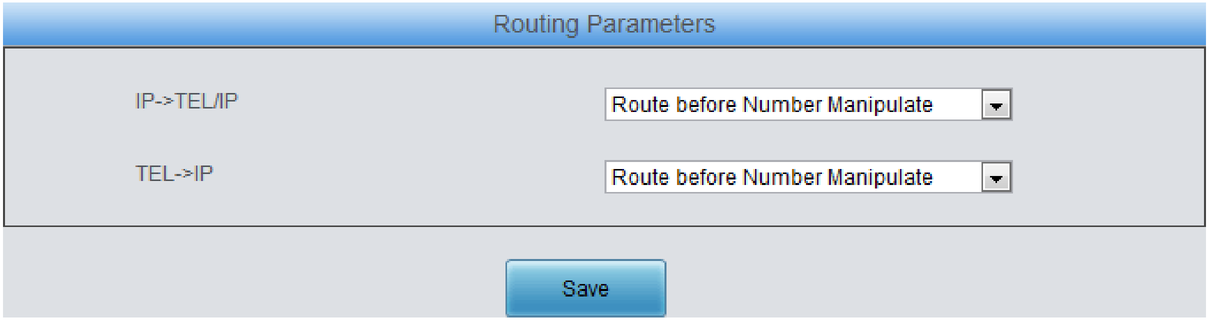


Figure 3-37 Routing Parameters Configuration Interface

See Figure 3-37 for the routing parameters configuration interface. On this interface, you can set the routing rules for calls respectively on two directions IP→Tel/IP and Tel→IP to be routing before or after number manipulation. The default value is *Route before Number Manipulate*.

After configuration, click ***Save*** to save the above settings into the gateway.

3.9.2 IP to Tel/IP

By default, there is no available routing rule displayed on the IP→Tel/IP routing rule configuration interface. And the IP→Tel/IP routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click ***Add New*** to add them manually. You may use the default values of all the configuration items herein.

The table below explains the items on the IP→Tel/IP routing rule configuration interface.

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with a smaller index value has a higher priority. If a call matches several routing rules, it will be processed according to the one with the highest priority.
Description	More information about each routing rule, with the default value of *default*.
Source IP	IP address from where the call is initiated. This item can be set to a specific IP address or "*" which indicates any IP address
CallerID Prefix,	A string of characters at the beginning of the caller/called party number. It can be a

CalleeID Prefix	specific string consisting of digits 0~9, 、"[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be characters '0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to indicates any character between these two characters. ',' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Source IP*** specify a routing rule for calls. **Note:** "[*]" represents TFM symbol *, while "*" represents any string; Multiple CallerID/CalleeID prefixes can be added simultaneously. They are separated by ":".
Route by Number	When this feature is enabled, the gateway will route a call from IP to a corresponding port based on its number. And the number of the port which this call will be routed to can be set via the item ***SIP Account*** on the Port Settings interface. In such case, the configuration item ***Call Destination*** goes invalid and shows Route by Number on the routing rule configuration interface. The default setting is *disabled*.
Call Destination	Designate a port group or an IP for the call to route.
Destination Port Group	Port group to which the call will be routed.
Destination IP, Destination Port	The IP address and port to which the call will be routed.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

See Figure 3-38 for the IP→Tel/IP routing rule configuration interface after your configuration. There is a rule displayed with Index 63 and Call Destination 'Route by Number', having no restriction on Source IP, CallerID Prefix and CalleeID Prefix, which indicates the gateway will route a call from any IP address to a corresponding port based on its number.

Press the ***Add New*** button on the bottom right corner of the list to add a new routing rule.

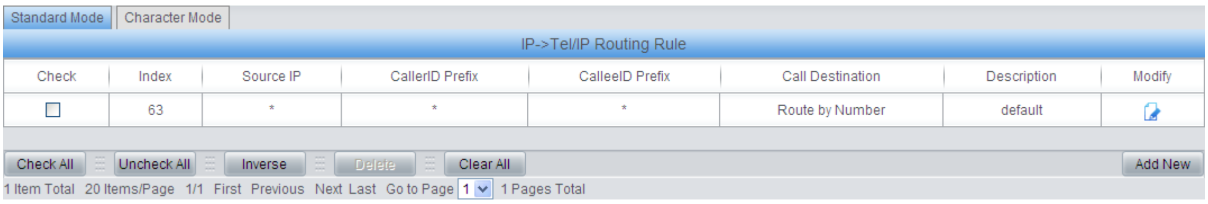


Figure 3-38 IP→Tel/IP Routing Rule Configuration Interface

Click ***Modify*** in Figure 3-38 to modify a routing rule. The configuration items on the IP→Tel/IP routing rule modification interface are the same as those on the ***Add New Routing Rule (IP→Tel/IP)*** interface. Note that the item ***Index*** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-38 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the ***Clear All*** button in Figure 3-38.

On the IP→Tel/IP Routing Rule Configuration interface under the Character mode, you can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

3.9.3 Tel to IP

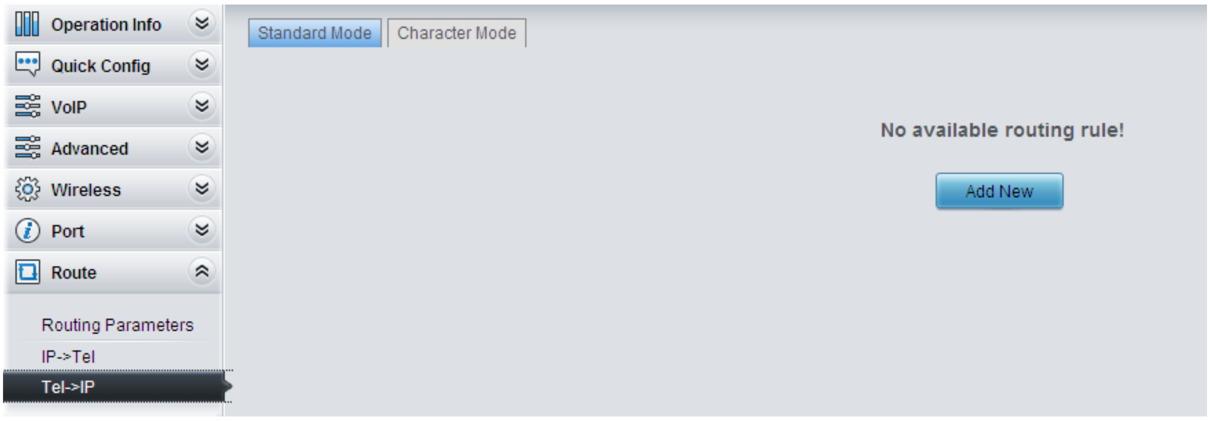


Figure 3-39 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-39 for the Tel→IP routing rule configuration interface. By default, there is no available routing rule on the gateway. The Tel→IP routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click ***Add New*** to add them manually. You may use the default values of all the configuration items herein except for ***Destination IP*** and ***Destination Port***.

The table below explains the items on the Tel→IP routing rule configuration interface.

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with a smaller index value has a higher priority. If a call matches several routing rules, it will be processed according to the one with the highest priority.
Description	More information about each routing rule, with the default value of *default.*
Source Port Group (Call Initiator)	Port group from which the call is initiated. This item can be set to a specific port group or '*' which indicates any port group.
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or characters ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '**-**' and '**,**'. ('**-**' is used between two characters to indicates any characters between these two characters. '**,**' is used to separate characters or characters ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with **Source Port Group (Call Initiator)** specify a routing rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string; Multiple CallerID/CalleeID prefixes can be added simultaneously. They are separated by ":".
Destination IP, Destination Port	IP address and port number of the remote end to which the call will be routed.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

See Figure 3-40 for the Tel→IP routing rule configuration interface after your configuration. There is a rule displayed with Index 63, Destination IP '192.168.1.101' and Destination Port '5060' (i.e. default IP address and port of the gateway), having no restriction on Call Initiator, CallerID Prefix and CalleeID Prefix, which indicates all the outgoing calls from Tel which conform to the dialing rule will be routed to the gateway.

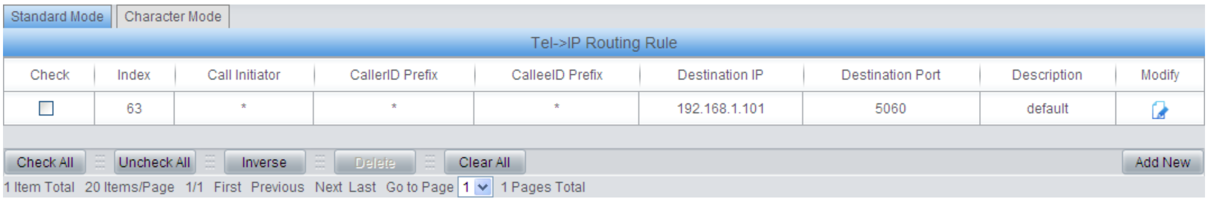


Figure 3-40 Tel→IP Routing Rule Configuration Interface

Click ***Modify*** in Figure 3-40 to modify a routing rule. The configuration items on the Tel→IP routing rule modification interface are the same as those on the ***Add New Routing Rule (Tel→IP)*** interface. Note that the item ***Index*** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-40 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the ***Clear All*** button in Figure 3-40.

On the Tel→IP Routing Rule Configuration interface under the Character mode, you can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

3.10 Number Manipulation

Number Manipulation includes four parts: **IP→Tel/IP CallerID**, **IP→Tel/IP CalleeID**, **Tel→IP CallerID** and **Tel→IP CalleeID**.

3.10.1 IP to Tel/IP CallerID

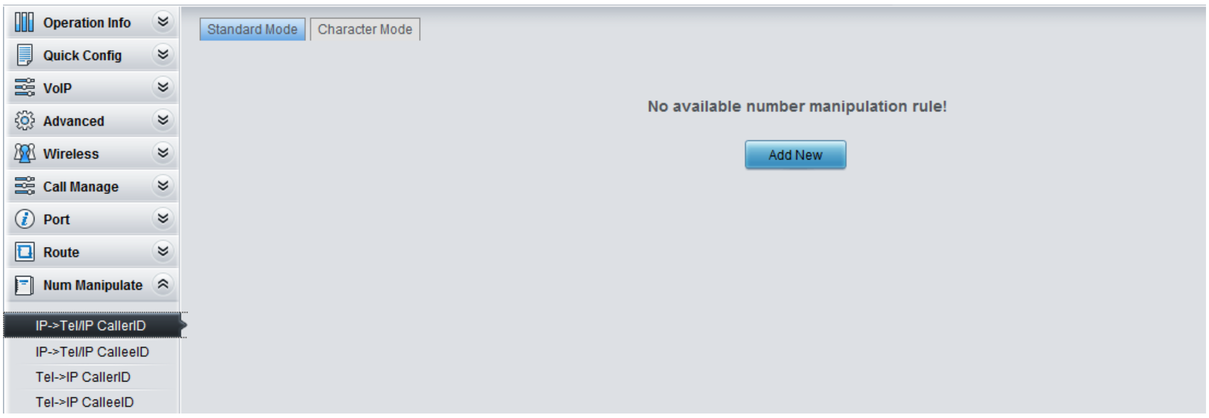


Figure 3-41 IP→Tel/IP CallerID Manipulation Interface (Standard)

See Figure 3-41 for the IP→Tel/IP CallerID manipulation interface under the Standard mode. A new number manipulation rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. On the IP→Tel/IP CallerID manipulation rule adding interface, you may use the default values of all the configuration items.

The table below explains the items on the IP→Tel/IP CallerID manipulation interface.

Item	Description

Index	The unique index of each number manipulation rule, which denotes its priority. A number manipulation rule with a smaller index value has a higher priority. If a call matches several number manipulation rules, it will be processed according to the one with the highest priority.
Description	More information about each number manipulation rule, with the default value of *default.*
Call Initiator	IP address from where the call is initiated. This item can be set to a specific IP address or "*" which indicates any IP address.
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to indicates any character between these two characters. ',' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Call Initiator*** specify a number manipulation rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string; Multiple CallerID/CalleeID prefixes can be added simultaneously. They are separated by ":".
Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

Note: The number manipulation is performed in 5 steps by the order of the following configuration items: *Stripped Digits from Left*, *Stripped Digits from Right*, *Reserved Digits from Right*, *Prefix to Add* and *Suffix to Add*.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings. See the figure below.

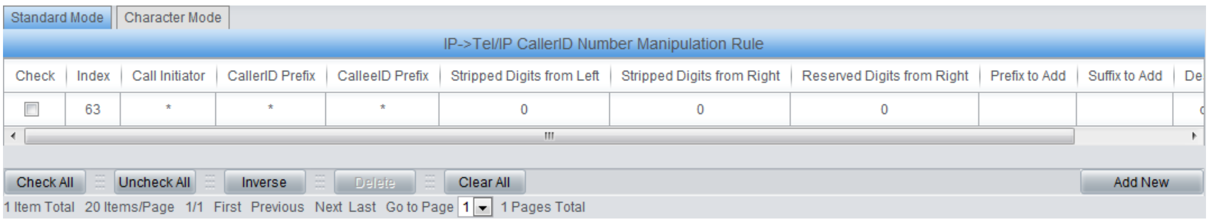


Figure 3-42 IP→Tel/IP CallerID Manipulation Interface (Standard)

Click ***Modify*** in Figure 3-42 to modify a number manipulation rule. The configuration items on the IP→Tel/IP CallerID manipulation rule modification interface are the same as those on the ***Add***

IP→Tel/IP CallerID Manipulation Rule interface. Note that the item ***Index*** cannot be modified.

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-42 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all number manipulation rules at a time, click the ***Clear All*** button in Figure 3-42.

On the IP→Tel/IP CallerID Manipulation interface under the Character mode, you can edit the number manipulation rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

3.10.2 IP to Tel/IP CalleeID

The number manipulation process for IP→Tel/IP CalleeID is almost the same as that for IP→Tel/IP CallerID; only the number to be manipulated changes from CallerID to CalleeID. The configuration items on the IP→Tel/IP CalleeID Manipulation interface are the same as those on ***IP→Tel/IP CallerID Manipulation Interface*** (Figure 3-42).

3.10.3 Tel to IP CallerID

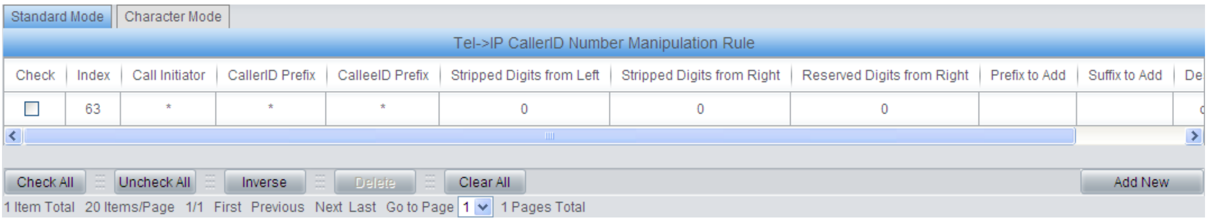


Figure 3-43 Tel→IP CallerID Manipulation Interface (Standard)

See Figure 3-43 for the Tel→IP CallerID manipulation interface under the Standard mode. A new number manipulation rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-44 for the Tel→IP CallerID manipulation rule adding interface. You may use the default values of all the other configuration items herein.

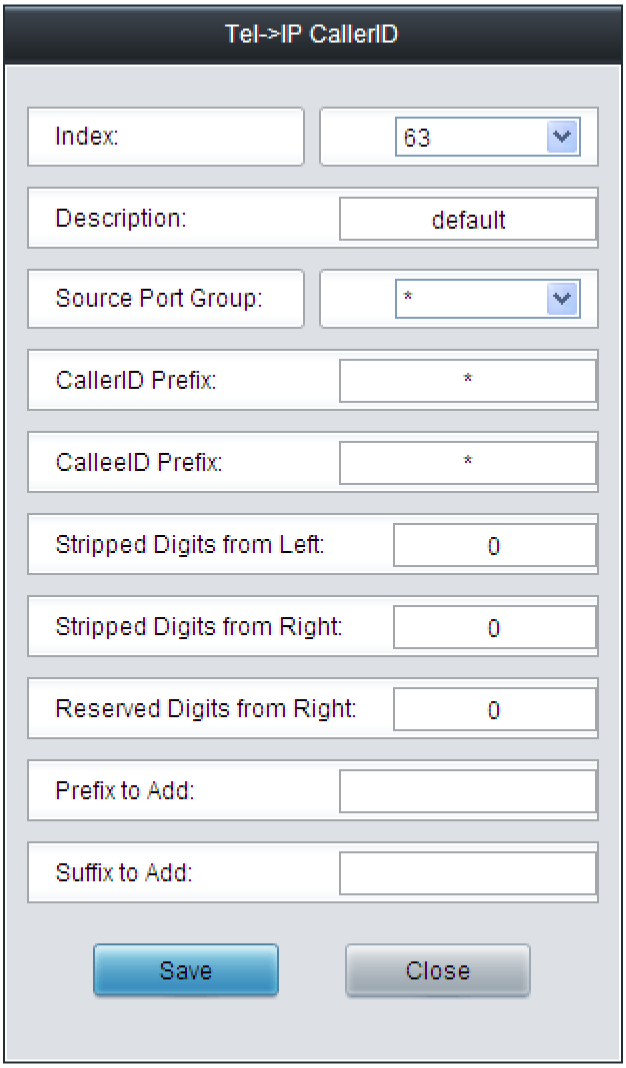


Figure 3-44 Add Tel→IP CallerID Manipulation Rule

The table below explains the items shown in the above figure.

Item	Description
Index	The unique index of each number manipulation rule, which denotes its priority. A number manipulation rule with a smaller index value has a higher priority. If a call matches several number manipulation rules, it will be processed according to the one with the highest priority.
Description	More information about each number manipulation rule, with the default value of *default*.
Source Port Group (Call Initiator)	Port group from which the call is initiated. This item can be set to a specific port group or '*' which indicates any port group.
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to indicate any character between these two characters. ',' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which

	indicates any string. These two configuration items together with ***Call Initiator*** specify a number manipulation rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string; Multiple CallerID/CalleeID prefixes can be added simultaneously. They are separated by ":".
Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

Note: The number manipulation is performed in 5 steps by the order of the following configuration items: *Stripped Digits from Left*, *Stripped Digits from Right*, *Reserved Digits from Right*, *Prefix to Add* and *Suffix to Add*.

After configuration, click ***Save*** to save the settings into the gateway or click ***Close*** to cancel the settings.

Click ***Modify*** in Figure 3-43 to modify a number manipulation rule. The configuration items on the Tel→IP CallerID manipulation rule modification interface are the same as those on the ***Add Tel→IP CallerID Manipulation Rule*** interface. Note that the item ***Index*** cannot be modified.

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-43 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all number manipulation rules at a time, click the ***Clear All*** button in Figure 3-43.

On the Tel→IP CallerID Manipulation interface under the Character mode, you can edit the number manipulation rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

3.10.4 Tel to IP CalleeID

The number manipulation process for Tel→IP CalleeID is almost the same as that for Tel→IP CallerID; only the number to be manipulated changes from CallerID to CalleeID. The configuration items on the Tel→IP CalleeID manipulation interface are the same as those on ***Tel→IP CallerID Manipulation Interface***.

3.11 System Tools

System Tools is mainly for gateway maintenance. It provides such features as change password, data backup, connectivity check, etc.

3.11.1 Upgrade

Via the upgrade interface you can upgrade the WEB, gateway service, kernel and firmware to new versions. Select the upgrade package "*.tar.gz" (The gateway will do MD5 verification before upgrading and will not start to upgrade until it passes the verification.) and click ***Update***. Then the file uploading interface will appear.

During the upgrading, you can learn the detailed upgrading information from the upgrade information box at the bottom.

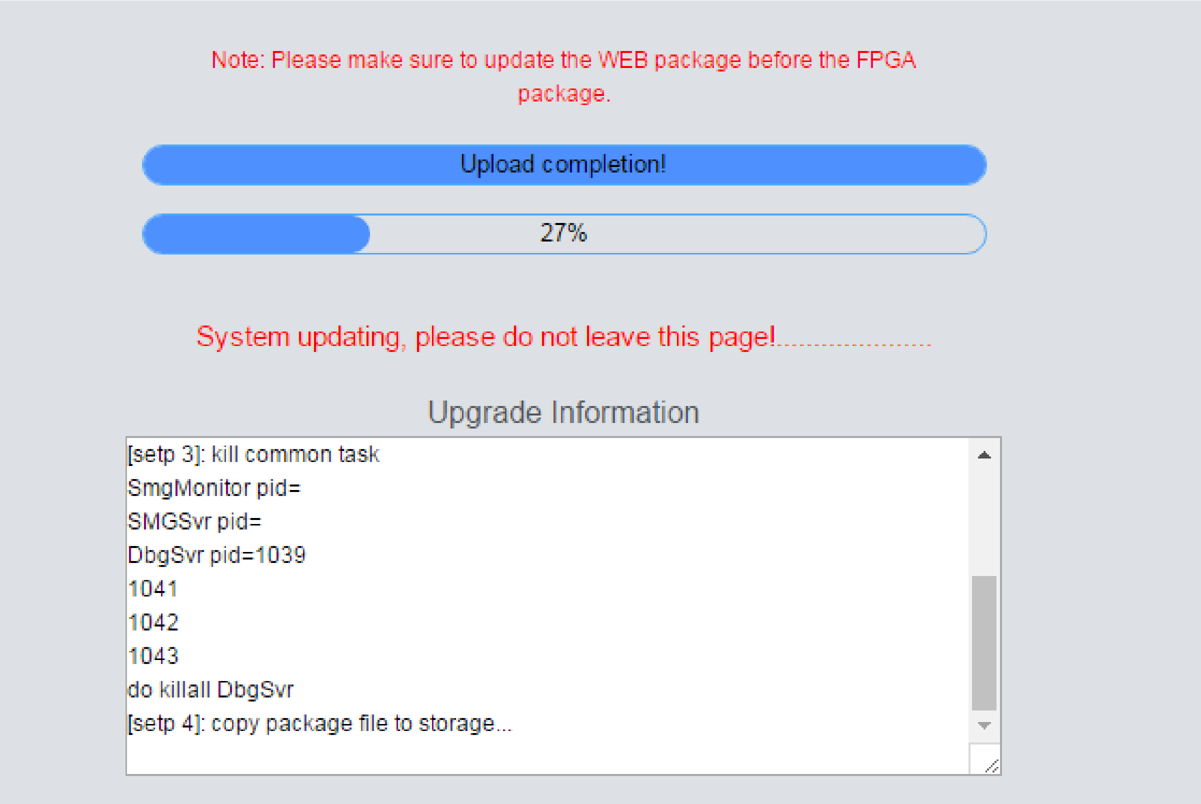


Figure 3-45 System Upgrading Interface

Note: Please contact our technicians if you need to downgrade the gateway to an old version. An improper operation may cause unexpected problems.

3.11.2 Signaling Capture

Packet capture contains Signaling Packet Capture, RTP Packet Capture. You can select either of them to start the capture according to your requirement. Click ***Start*** to start capturing packets. Click ***Stop*** to stop the capture. Click ***Download*** to download the captured packets. See below for the configuration items on this interface.

Item	Description
Signaling Packet Capture	Determines which kind of data to capture, options including: None, SIP, Syslog, Centralized Manage, SIP&Syslog&Centralized Manage.
RTP Packet Capture	Sets whether to capture the RTP data.
SYSLOG Enabled	Sets whether to enable SYSLOG. If it is enabled, you are required to fill in the server address and level of SYSLOG. By default it is disabled.
Server Address	Sets the SYSLOG server address to receive the logs.
SYSLOG Level	Sets the SYSLOG level, options including ERROR, WARNING, INFO and DEBUG. The default value is *INFO*.
AT Debug Enabled	Sets whether to enable the AT debugging feature. By default it is disabled. If it is enabled, the information about AT debugging will be sent out to SYSLOG.

Echo Mode Enabled	Sets whether to enable the Echo mode. By default it is disabled. If it is enabled, both the received and sent AT messages will be displayed at the same time.
Port	Selects the port to execute the AT debugging. You may choose a single or all of the ports.
SIM Data Debug Enabled	Sets whether to enable the SIM card data debugging.

3.11.3 Data Recording

On the Data Recording interface, click ***Start*** to start the recording; click ***Stop*** to stop the recording; click ***Download*** to download the recorded data.

3.11.4 Call Log

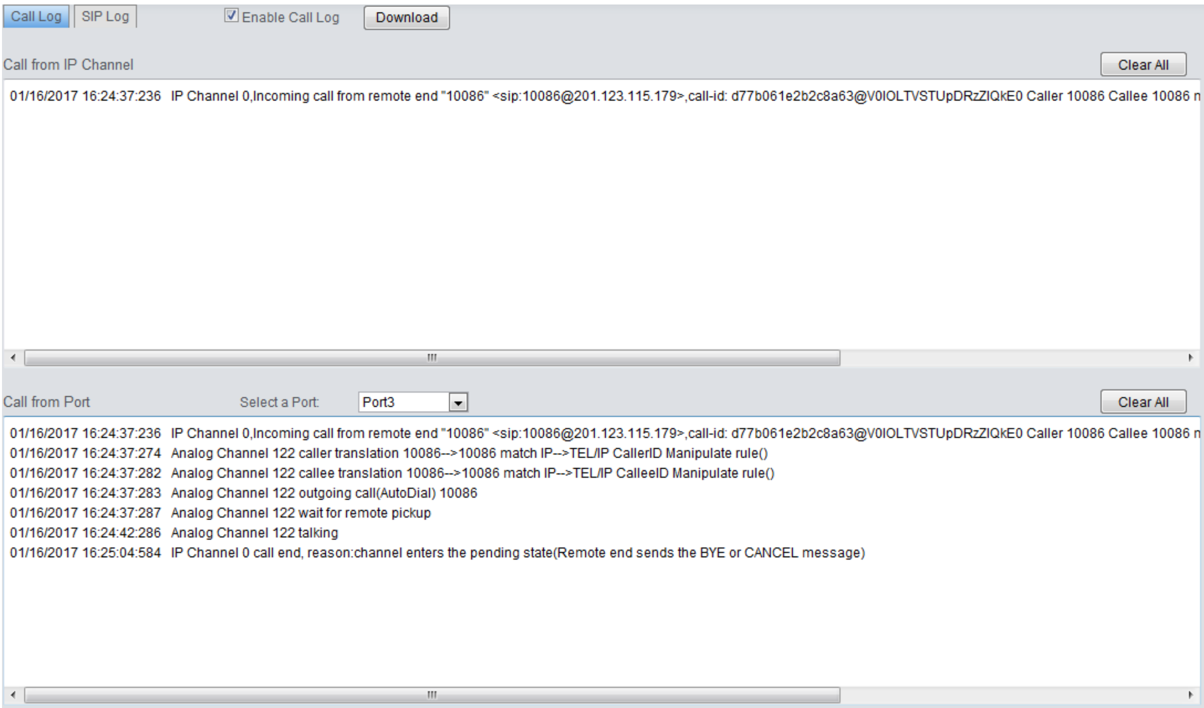


Figure 3-46 Call Log Interface

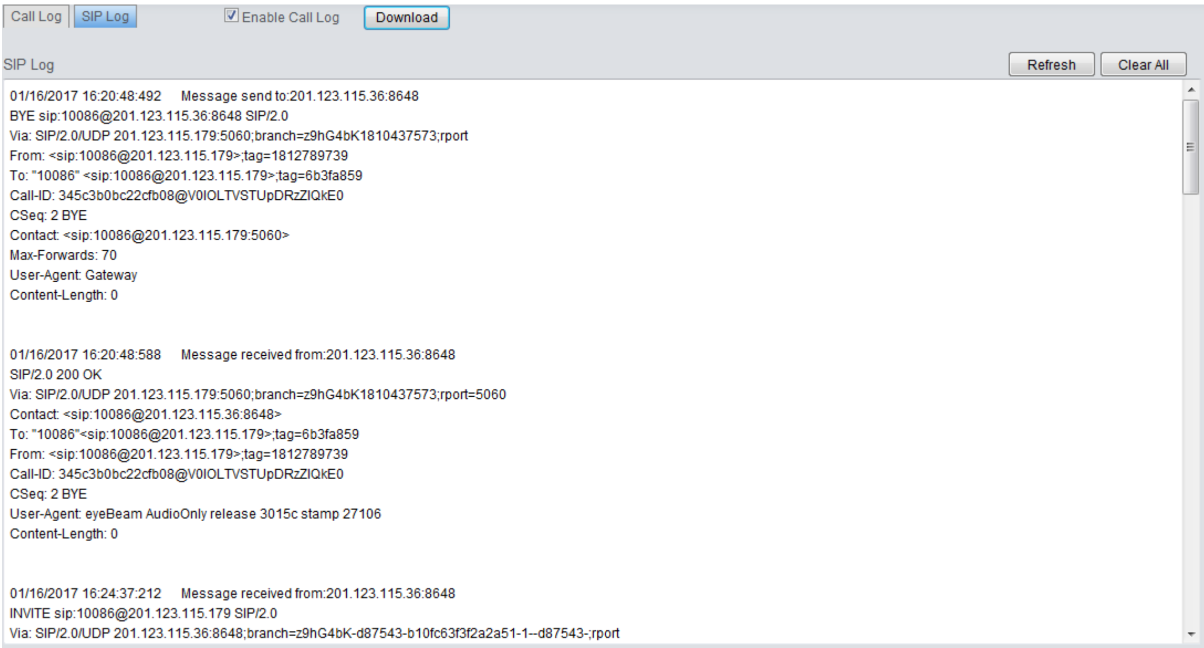


Figure 3-47 SIP Log Interface

On the Call Log interface, click the checkbox before ***Enable Call Log*** to enable the call log feature, including ***Call Log*** and ***SIP Log***. ***Call from IP Channel*** displays the call log information generated on all IP channels, and ***Call from Port*** displays the call log information generated on the port you select. All the SIP related information will be displayed in ***SIP Log***.

3.11.5 Operation Log

The Operation Log interface is used to check the operation records on WEB. Click ***Refresh*** to refresh the log; click ***Clear All*** to clear all the operation logs and click ***Download*** to download the logs. The operation log will be automatically cleared once the system restarts.

Note: The sign <@#> here means the configuration item is null.

3.11.6 Change Password

On the password changing interface you can change username and password of the gateway. Enter the current password, the new username and password, and then confirm the new password. After configuration, click ***Save*** to apply the new username and password or click ***Reset*** to restore the configurations. After changing the username and password, you are required to log in again.

3.11.7 Backup & Upload

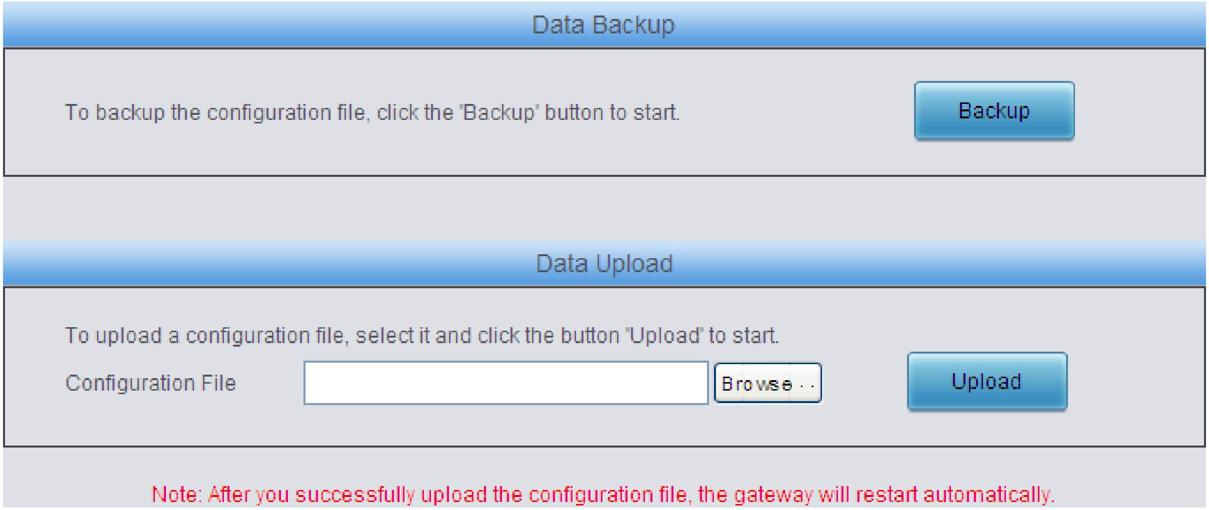


Figure 3-48 Backup & Upload Interface

See Figure 3-48 for the backup and upload interface. To back up the configuration file to your PC, just click ***Backup***. To upload a configuration file, select it via ***Browse…*** and click ***Upload***.

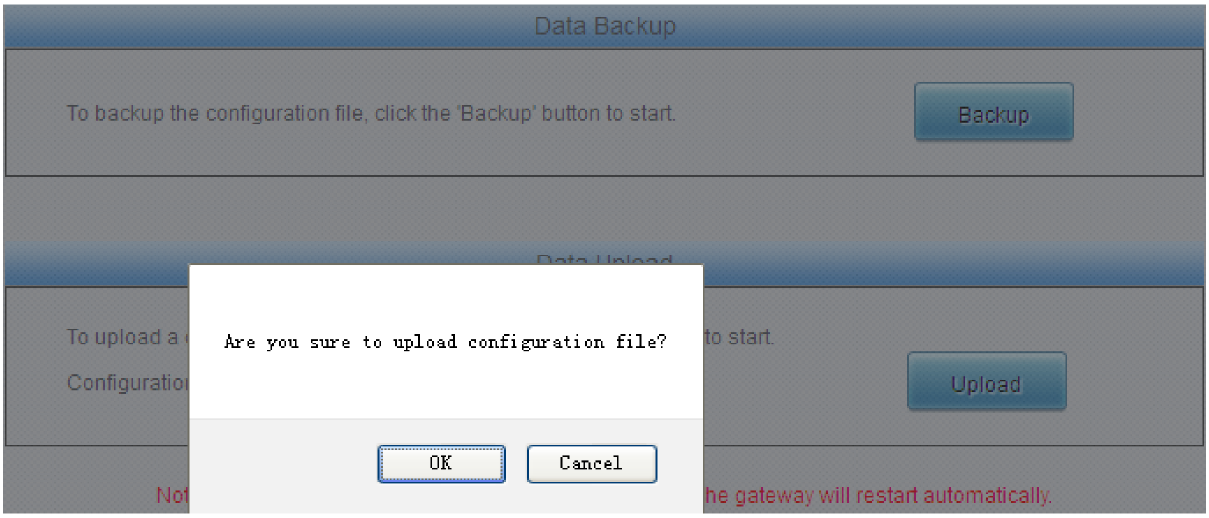


Figure 3-49 Backup & Upload & Prompt Interface

Click ***OK*** on the prompt box (Figure 3-49) to upload the configuration file to the gateway. Now the prompt information 'System is rebooting, please do not leave this page' appears. The gateway will overwrite the current configurations with the uploaded data after restart. Click ***Cancel*** to cancel this upload directly.

3.11.8 Factory Reset

Figure 3-50 Factory Reset Interface

See Figure 3-50 for the factory reset interface. Click ***Reset*** to restore all configurations on the gateway to factory settings.

3.11.9 Restart

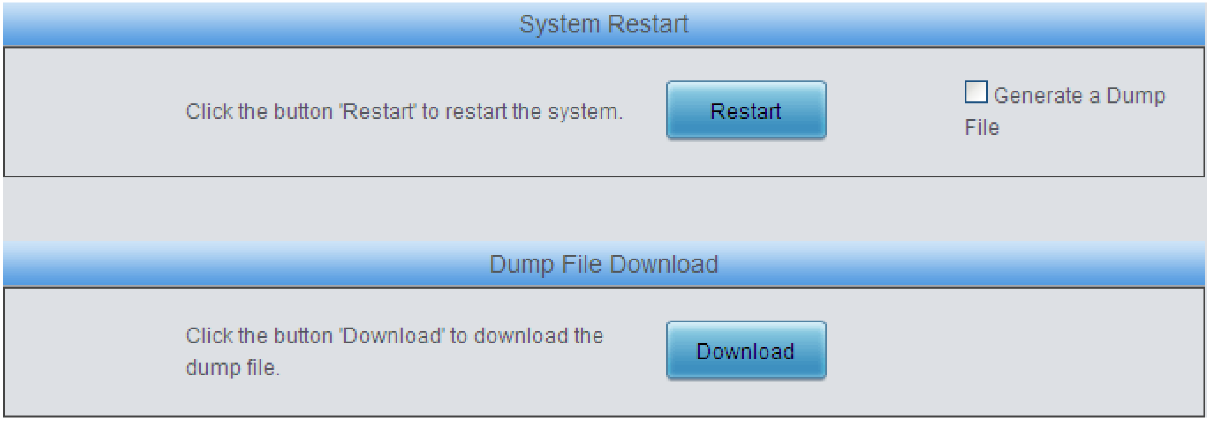


Figure 3-51 System Restart Interface

See Figure 3-51 for the restart interface. Click ***Restart*** under the service restart interface to restart the gateway service or click ***Restart*** under the system restart interface to restart the whole gateway system. A dump file will be generated each time you restart the service or the system. Click ***Download*** and you can download it to help troubleshoot issues.

3.11.10 System Monitor

See the System Monitor Configuration interface. Watchdog is a timing reset system used to avoid application crash. You can set the dog feeding interval when this feature is enabled. The feeding interval is calculated by s, with the value range of 1~15s. By default, this feature is enabled with the default value of 5s. As the feature 'Automatically restart the service if undetected' is enabled, the service application will restart automatically if it is not detected by the gateway guard application. By default, this feature is enabled.

3.11.11 Centralized Manage

Via the Centralized Manage Setting interface, the gateway can register to a centralized management platform and accept the management of the platform. The table below explains the items on this interface.

Item	Description
Management Platform	Selects a management platform for the gateway to register, including two options: DCMS and Others.
Centralized Management Protocol	Sets the centralized management protocol. It only supports SNMP currently and is valid only when Others is selected for ***Management Platform***.
SNMP Version	Sets the version of SNMP, three options available: V1, V2 and V3, with the default value of V2. This item is valid only when Others is selected for ***Management Platform***.
Server Address	The address of the server in which the DCMS locates, It can be IP or a domain name. Note: To configure the domain name, make sure the DNS is already configured and the corresponding domain name is analyzable.

Monitoring Port	Monitoring Port for SNMP on the gateway. This item is valid only when Others is selected for ***Management Platform***.
Account	The account of SNMP, valid only when the SNMP version is set to V3.
Grade	The grade of SNMP, three options available: Neither authenticated nor encrypted, Authenticated but not encrypted, Authenticated and encrypted, with the default value of *Neither authenticated nor encrypted*. It is valid only when the SNMP version is set to V3.
Community String	Community string used for information acquisition.
Company Name	The company name used to register the gateway to DCMS, only valid when DCMS is selected.
Authorization Code	The authorization code is used for the connection verification. A device can connect to the DCMS successfully only after it passes the verification.
Gateway Description	The description displayed on Synway DCMS after the gateway is registered to Synway DCMS, giving an easy identification of the gateway in device grouping. It is valid only when Synway DCMS is selected
Working Status	The status of the connection between the gateway and the centralized management server.

After your modification, click ***Save*** to save the above settings into the gateway, click ***Reset*** to restore the configurations, and click ***Download MIB*** to download the MIB file which can let the centralized management server get the data information structure of our gateway.

3.11.12 PING Test

A Ping test can be initiated from the gateway on a designated IP address to check the connection status between them. The table below explains the configuration items on the Ping test interface.

Item	Description
Destination Address	Destination IP address or domain name on which the Ping test is executed.
Ping Count	The number of times that the Ping test should be executed. Range of value: 1~100.
Package Length	Length of the data package used in the Ping test. Range of value: 56~1024 bytes.
Info	The information returned during the Ping test, helping you to learn the network connection status between the gateway and the destination address.

After configuration, click ***Start*** to execute the Ping test; click ***End*** to terminate it immediately.

3.11.13 TRACERT Test

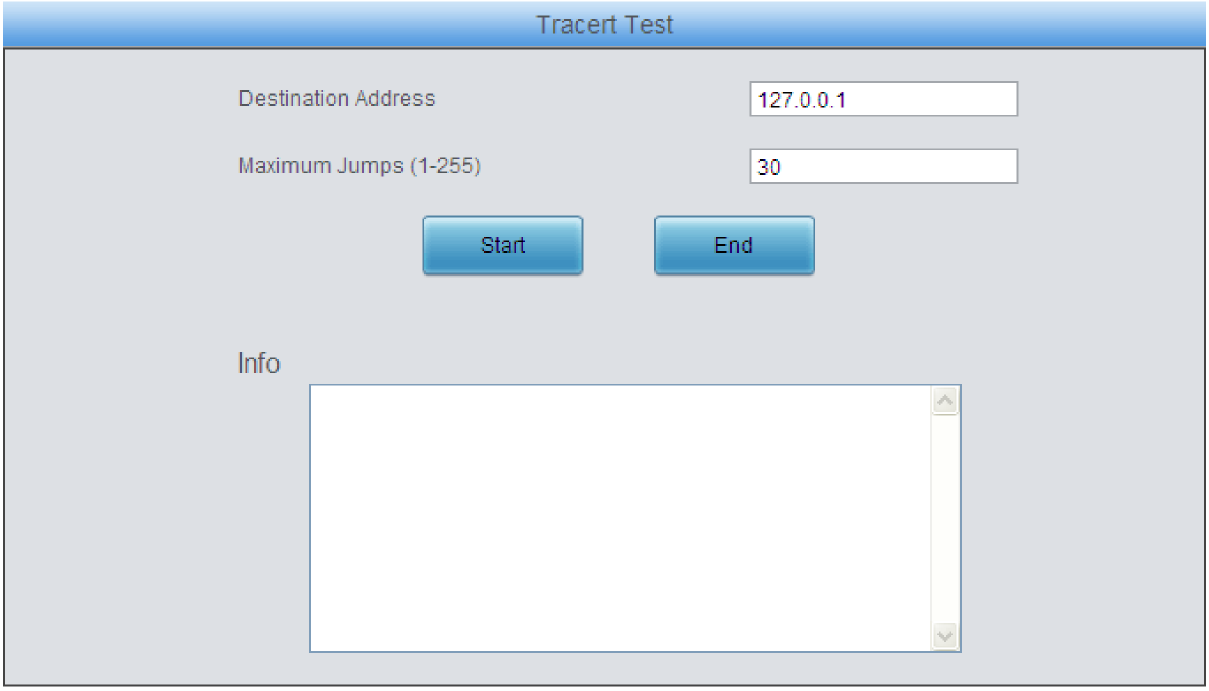


Figure 3-52 Tracert Test Interface

See Figure 3-52 for the Tracert test interface. A Tracert test can be initiated from the gateway on a designated IP address to check the routing status between them. The table below explains the configuration items shown in the above figure.

Item	Description
Source IP Address	Source IP address where the Tracert test is initiated.
Destination Address	Destination IP address on which the Tracert test is executed.
Maximum Jumps	Maximum number of jumps between the gateway and the destination address which are returned by the Tracert test. Range of value: 1~255.
Info	The information returned during the Tracert test, helping you to learn the detailed information about the jumps between the gateway and the destination address.

After configuration, click ***Start*** to execute the Tracert test; click ***End*** to terminate it immediately.

3.11.14 Wireless Test

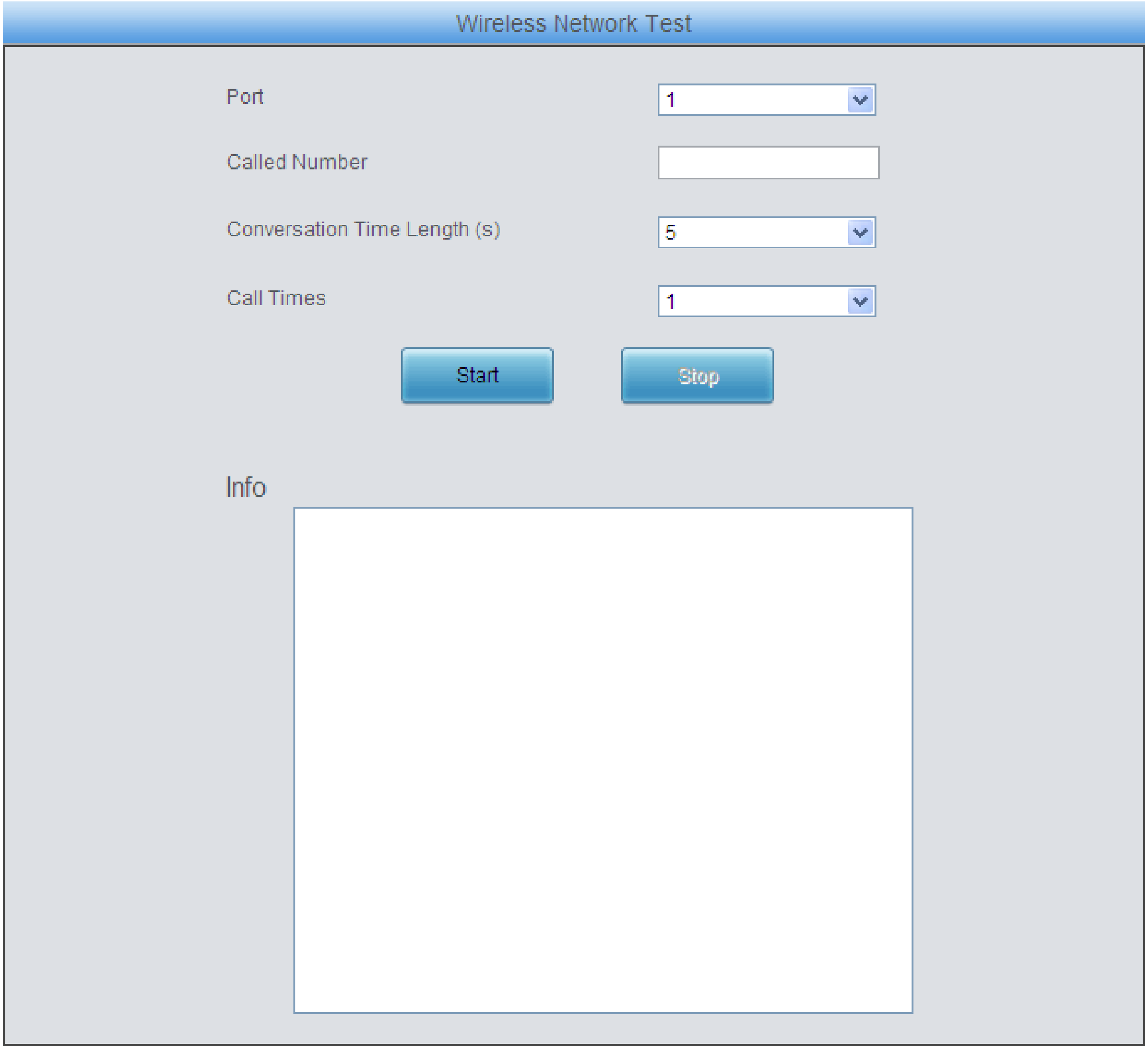


Figure 3-53 Wireless Network Test Interface

See Figure 3-53 for the Wireless Network Test interface. This test is to check whether the SIM card inserted in the gateway port can make normal calls. The table below gives the explanation to the configuration items shown in the above figure.

Item	Description
Port	The port used for the test
Called Number	The called party number which will be dialed for the test
Conversion Time Length	The time length of the conversion
Call Times	The times of the testing call

After configuration, click ***Start*** to execute the test; click ***Stop*** to terminate it immediately.

3.11.15 Module Test

Port State					
Port	Type	State	Cell Phone No.	Connection	Signal
1	CDMA	Sim Detected	660641	Connect	
2	CDMA	Sim Detected	414748	Connect	
3	CDMA	Sim Detected	884515	Connect	
4	CDMA	Sim Detected	475902	Connect	
5	CDMA	Sim Detected	15314682732	Connect	
6	CDMA	Sim Detected	13306518401	Connect	
7	CDMA	Sim Detected	864201	Connect	
8	CDMA	Sim Detected	664705	Connect	
9	CDMA	Sim Detected	15355074652	Connect	
10	CDMA	Sim Detected	874252	Connect	
11	CDMA	Sim Detected	662429	Connect	
12	CDMA	Sim Detected	15372427495	Connect	
13	CDMA	Sim Detected	18143477961	Connect	
14	CDMA	Sim Detected	111	Connect	
15	CDMA	Sim Detected	18958154838	Connect	
16	CDMA	Sim Detected	894905	Connect	

Figure 3-54 Module Test Interface

See Figure 3-54 for the Wireless Network Test interface. This test is for our manufacturers to check whether a module can detect the SIM card. Two states may appear: ***Sim Detected*** and ***Unusable***.

3.11.16 Access Control

On the Access Control List interface, you can add a piece of command to ACL to restrict the network flow. Thus only the particular devices are allowed to visit the gateway and only the data packages on the designated ports can be forwarded. Click ***Add New*** to add a new piece of command. See Figure 3-55.

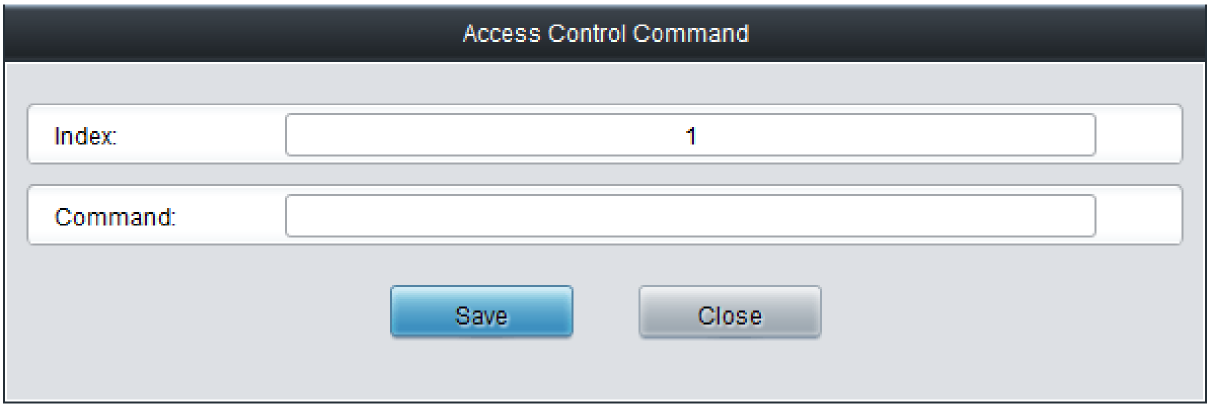


Figure 3-55 Add Access Control Command Interface

Fill in a piece of command to the item Command and click ***Save*** to save the settings to the gateway. Click ***Close*** to cancel your settings. Click ***Apply*** to make the new command valid.

Click ***Modify*** on the Access Control List interface to modify a command. The configuration items on the Access Control Command Modification interface are the same as those on the ***Add Access Control Command*** interface. Note that the item ***Index*** cannot be modified.

To delete an Access Control Command, check the checkbox before the corresponding index on the Access Control List interface , click the ***Delete*** button, and then click the ***Apply*** button to make the deleted command invalid. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all access control commands at a time, click the ***Clear All*** button on the Access Control List interface.

Note:

1, Currently, only the command iptables is supported by the gateway.

2, If you add or modify or delete commands manually, don't forget to click the ***Apply*** button to make your settings valid. However, if the gateway restarts or the configuration is leading-in, you need not click the ***Apply*** button and the commands will get valid automatically.

3.11.17 Device Lock

This feature is unopened. If you need use it, please contact our technicians to apply for a special link to access the gateway again.

Appendix A Technical Specifications

Dimensions

4002/4004/4008 series: 260×30×153mm³

4016/4032 series: 440×44×200mm³

Weight

4002/4004/4008 series Net: 1.2 kg

4016 series Net: 2.4 kg

4032 series Net: 3.1 kg

Environment

Operating temperature: 0℃—45℃

Storage temperature: -20℃—85℃

Humidity: 8%— 90% non-condensing

Storage humidity: 8%— 90% non-condensing

LAN

Amount: 2 (10/100 BASE-TX (RJ-45))

Self-adaptive bandwidth supported

Auto MDI/MDIX supported

Console Port

Amount: 1 (RS-232)

Baud rate: 115200bps

Connector: RJ45 to DB-9 Connector (4004/4008 series), Mini-USB connecting line (4016/4032 series)

Data bits: 8 bits

Stop bit: 1 bit

Parity unsupported

Flow control unsupported

Note: Follow the above settings to configure the serial port; or it may work abnormally.

Power Requirements

Input power: 12V DC ±10%

Input Current: ≥3A DC

Signaling & Protocol

SIP signaling

Supported protocol: SIP V1.0/2.0, RFC3261

Network Protocol

IP v4, UDP/TCP, PPPoE, DHCP,

FTP/TFTP ARP, RARP, NTP,

HTTP, Telnet

Audio Encoding & Decoding

G.711A	*64 kbps*
G.711U	*64 kbps*
G.729A/B	*8 kbps*
G723	*5.3/6.3 kbps*
G722	*64 kbps*
AMR	*4.75 kbps*
iLBC	*13.3/15.2 kbps*

Sampling Rate

8kHz

Wireless Feature

SMS CODEC: ASCII/UCS2

Others

The LTE series gateways support the VoLTE network so that they provide quick call establishment and stay unaffected by the Base Station capacity.

Appendix B Troubleshooting

Q1. What to do if I forget the IP address of the wireless gateway?

There are two ways to get the IP address:

1) Long press the Reset button on the gateway to restore to factory settings. The default IP address is 192.168.1.101
2) Make a call to any wireless port and press the function key to query the IP address. See Function Key for more details.

Q2. In what cases can I conclude that the wireless gateway is abnormal and turn to Synway's technicians for help?

a) During runtime, the run indicator does not flash or the alarm indicator lights up or flashes, and such error still exists even after you restart the device or restore it to factory settings.
b) Voice problems occur during call conversation, such as that one party or both parties cannot hear the voice or the voice quality is unacceptable.
c) The port of the gateway is well connected with the antenna and has a SIM card properly inserted, but the port indicator never lights up after the gateway startup or the color it lights up does not comply with the actual port state or port type.

Other problems such as inaccessible calls, failed registrations, incorrect numbers are probably caused by configuration errors. We suggest you refer to Chapter 3 WEB Configuration for further examination. If you still cannot figure out or solve your problems, please feel free to contact our technicians.

Q3. What to do if I cannot enter the WEB interface of the gateway after login?

This problem may happen on some browsers. To settle it, follow the instructions here to configure your browser. Enter 'Tools > Internet Options >Security Tab', and add the current IP address of the gateway into 'Trusted Sites'. If you changes the IP address of the gateway, add your new IP address into the above settings too.

Q4. Is there any cell-phone APP can make calls to the gateway?

Yes. Linphone is a soft SIP phone that is supported by multiple platforms, such as Linux, Windows, iOS, Android, etc. It must be registered to the SIP registrar server before dialing to other SIP devices or PSTN telephones,

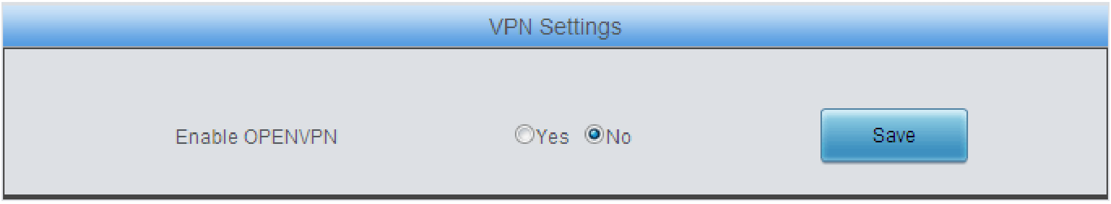
Q5. Which RTP codecs are supported by the gateway?

At present, the supported RTP codecs are: G.711A, G.711u, G.729, G.723, G.722, AMR and iLBC.

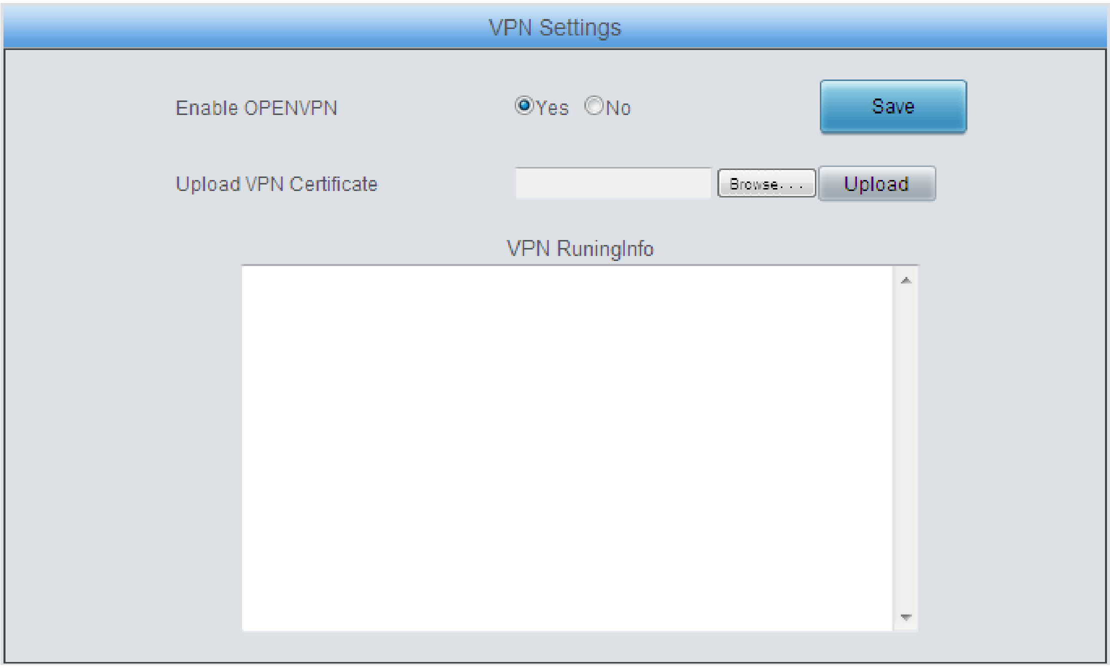
Appendix C About VPN

Part 1: Steps to Enable VPN Feature

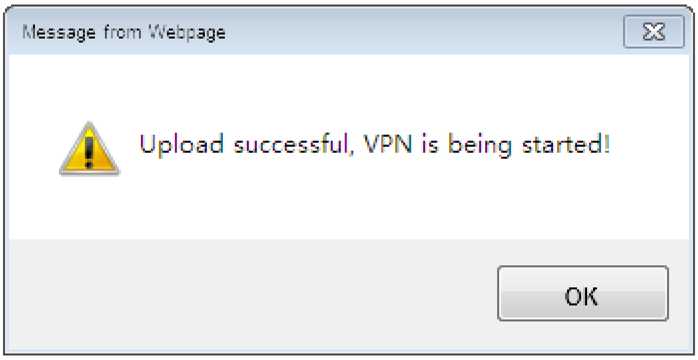
Find the VPN Settings interface under Advanced Settings on the web. This featured is disabled by default.



Step 1: Select Yes to enable this feature, click the ‘Save’ button and the following interface will appear.



Step 2: Select a certificate from the client, that is, a configuration file with the suffix of .conf, and then click the ‘Upload’ button. The following dialog will appear.



Step 3: Now you will get a virtual IP address which is allocated automatically by the VPN server. Note that each upload will lead to a new allocation of the IP address; however, restarting the gateway will not change the virtual IP address.

Then you may use the PING test under System Tool on the web to test if the client connects successfully with the server via IP, by which to check whether the VPN feature is successfully enabled or not.

Part 2: Steps to Make VPN Certificate

Step 1: Get the file of client.ovpn from the VPN server (under the 'sample-config' directory of the installation package) and rename it to "client.conf".

Step 2: Examine or add the following content into the file.

The file should contain the following content, in which the black part is fixed while the red part shall change according to the note.

client

dev tap (Note: Fill in tap or tun according to the VPN server's requirement. Currently, only tap is supported.)

proto tcp (Note: Connect via TCP which should be consistent with that of the server.)

;cipher AES-128-CBC (Note: Select an encryption algorithm which should be consistent with that of the client. It is not necessary to add if there is no algorithm at the client.)

remote 192.168.143.235 1194 udp (Note: Fill in the IP address and the port number of the VPN server, and the protocol can be left empty.)

;remote-random (Note: If there are multiple servers configured, let the client connect at random.)

resolv-retry infinite (Note: Analyze the server's domain name)

nobind (Note: Not to bind any port to the client)

persist-tun

persist-key

mute-replay-warnings (Note: Set as a flag to warn about replayed data packages.)

ns-cert-type server

comp-lzo (Note: Use the lzo compression which is consistent with the server.)

verb 3

;tls-client

;tls-auth ta.key 1 (Note: It is used to enable the feature of TLS encryption, and should be consistent with that of the server.)

<ca>

-----BEGIN CERTIFICATE-----

Note: Fill in the key copied from the file of ca.crt.

-----END CERTIFICATE-----

</ca>

<cert>

-----BEGIN CERTIFICATE-----

Note: Fill in the key copied from the file of client.crt, that is, the content inbetween “-----BEGIN CERTIFICATE-----” and “-----ENDCERTIFICATE-----”

-----END CERTIFICATE-----

</cert>

<key>

-----BEGIN RSA PRIVATE KEY-----

Note: Fill in the key copied from the file of client.key

-----END RSA PRIVATE KEY-----

</key>

Note: The following key is not necessary to add if it is never encrypted at the server.

<tls-auth>

Note: Fill in the key copied from the file of ta.key

</tls-auth>

Make sure the three key files ca.crt, client.crt and client.key are of the newest versions.

Step 3: Save the file after your examination or supplement and upload it to the device. Note that the suffix of the file must be .conf.

Part 3: Attentions

a) After the VPN featured is opened at the server, use your PCs to connect as a test. If two PCs can PING through each other, it means the server works normally.

b) Make sure the server is OK and the configuration file is ready before opening the VPN feature. The system time of the wireless gateway must be consistent with that of the server, or the connection may sometimes fails.

c) After enabling the VPN feature successfully, you can use the virtual IP of the gateway to make calls in both directions IP-->tel and tel-->IP.

Appendix D CDR Hangup Reason

```
const char* g_szSIPPendingRsn[] =   //SIP hangup reason
{
    "IP_REMOTE_CRASHED",   //Remote crash
    "IP_REMOTE_CLOSED",   //Remote hangup
    "IP_DIAL_TIMEOUT",   //Call timeout
    "IP_REMOTE_REJECTED",   //Remote reject
    "IP_REFER_SECCEED",   //Search successful
    "IP_REFER_REFUSED",   //Search rejected
    "IP_STUN_FAILED",   //Network penetration failed
    "IP_NOTRCV_ACK",   //ACK not received
    "IP_REDIRECT_FAIL"   //Redirection failed
};

const char* g_szGSMPendingRsn[] = //GSM hangup reason
{
    "GSM_NO_DIALTONE",   //No dial tone
    "GSM_BUSYTONE",   //Busy tone
    "GSM_ECHO_NOVOICE",   //No ringback tone
    "GSM_NOANSWER",   //No answer
    "GSM_TALKING_REMOTE_HANGUPED",   //Remote hangup
    "GSM_NOVOICE",   //No voice
    "GSM_NOCARRIER",   //No carrier
    "GSM_OUT_ERROR",   //Outbound call error
    "GSM_OUT_NOCLCC"   //No clcc
};

const char *g_szChFinishReason[]=   //Call end reason
{
    "STATE_IDLE",   //Channel in idle state
    "STATE_PENDING",   //Channel in suspended state
    "ROUTE_FAILED",   //Routing failed
```

```
    "BINDCH_FAILED",    //Failed to bind channel
    "MATCH_DIALDIGIT_FAILED",    //Failed to match dial digit
    "FIND_PORT_BY_NAME_FAILED",    //Failed to find port by name
    "FIND_PORT_FROM_GROUP_FAILED",    //Failed to find port by routing
    "CALL_TRANSFER",    //Call transfer
    "DNT_DISTRUBT",    //Do not disturb
    "STATE_BUSY",    //Channel busy
    "DIAL_FAILED",    //Dial failed
    "NO_IDLE_CH",    //No idle channel
    "NO_ANSWER_TRANSFER",    //Call forward with no answer
    "UNCONDITION_TRANSFER",    //Call transfer Unconditional
    "BUSY_TRANSFER",    //Call forward on busy
    "BINDCH_FIN",    //Port binding completed
    "CALLER_CANCELED",    //Canceled by caller
    "CALL_TRANSFER_FAILED",    //Call transfer failed
    "OFF_LINE",    //Offline
    "CREATE_SESSION_FAILED",    //Failed to create session
    "NO_ANSWER",    //No response
    "UNRING_NOCARRIER",    //No ringing and no carrier
    "CALLTIME_SINGLE_MAX",    //Maximum single call time reached
    "CALLTIME_TOTAL_MAX",    //Maximum total call time reached
    "CALLTIME_LIST_MAX",    //Maximum call time under the timing list rule reached
    "CALLTIME_DAY_MAX",    //Maximum call time per day
    "CALLCHARGE_SINGLE_MAX",    //Maximum single call charge limit reached
    "CALLCHARGE_TOTAL_MAX",    //Total call charge limit reached
    "NETWORK_UN4G"    //Network in non-4G state
};
```

Appendix E Technical/sales Support

Thank you for choosing Synway. Please contact us should you have any inquiry regarding our products. We shall do our best to help you.

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