



**Synway SMG Series Wireless Gateway**

**SMG4004**

**SMG4008**

**SMG4016**

**SMG4032**

**Wireless Gateway**

# **User Manual**

**Version 1.4.0**

**Synway Information Engineering Co., Ltd**

**[www.synway.net](http://www.synway.net)**

# Content

<b>Content</b>	<b>i</b>
<b>Copyright Declaration</b>	<b>iii</b>
<b>Revision History</b>	<b>iv</b>
<b>Chapter 1 Product Introduction</b>	<b>1</b>
1.1 Typical Application	2
1.2 Feature List	2
1.3 Hardware Description	3
1.4 Indicator Info	6
<b>Chapter 2 Quick Guide</b>	<b>7</b>
<b>Chapter 3 WEB Configuration</b>	<b>10</b>
3.1 System Login	10
3.2 Operation Info	11
3.2.1 System Info	11
3.2.2 Port State	12
3.2.3 Call Count	13
3.2.4 SIP Message Count	14
3.3 Quick Config	14
3.4 VoIP Settings	16
3.4.1 SIP	17
3.4.2 SIP Compatibility	18
3.4.3 SIP Station	20
3.4.4 SIP Server	22
3.4.5 NAT Setting	24
3.4.6 Media	26
3.5 Advanced Settings	28
3.5.1 Network	29
3.5.2 System Param	30
3.5.3 Service Config	32
3.5.4 Dialing Rule	33
3.5.5 Function Key	37
3.5.6 Cue Tone	38
3.5.7 Color Ring	38
3.5.8 QoS	40
3.5.9 Tone Generator	41
3.5.10 CDR Query	42
3.5.11 VPN	42
3.6 Wireless Settings	43
3.6.1 Basic Parameters	44
3.6.2 Wireless Param	46
3.6.3 Call Forwarding	48
3.6.4 Short Message	49
3.6.5 IMEI	52

3.6.6	USSD.....	53
3.6.7	Email.....	54
3.6.8	Balance.....	55
3.6.9	SIM Card.....	56
3.6.10	PIN Manage.....	57
3.7	Port Settings .....	59
3.7.1	Port.....	60
3.7.2	Port Group .....	63
3.8	Route Settings .....	66
3.8.1	Routing Parameters.....	66
3.8.2	IP to Tel/IP .....	67
3.8.3	Tel to IP.....	69
3.9	Number Manipulation.....	71
3.9.1	IP to Tel CallerID.....	72
3.9.2	IP to Tel CalleeID .....	76
3.9.3	Tel to IP CallerID.....	77
3.9.4	Tel to IP CalleeID .....	80
3.10	System Tools .....	81
3.10.1	Upgrade.....	81
3.10.2	Signaling Capture .....	83
3.10.3	Data Recording.....	84
3.10.4	Call Log .....	84
3.10.5	Change Password .....	85
3.10.6	Backup & Upload.....	86
3.10.7	Factory Reset .....	87
3.10.8	Restart.....	87
3.10.9	System Monitor.....	88
3.10.10	SNMP Config.....	88
3.10.11	PING Test.....	89
3.10.12	TRACERT Test .....	90
3.10.13	Wireless Network Test .....	91
<b>Appendix A Technical Specifications.....</b>		<b>92</b>
<b>Appendix B Troubleshooting .....</b>		<b>93</b>
<b>Appendix C VPN Certificate .....</b>		<b>94</b>
<b>Appendix D Technical/sales Support .....</b>		<b>95</b>

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## Revision History

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Version 1.1.0	2015-11	New Revision
Version 1.2.0	2016-1	New Revision
Version 1.3.0	2016-4	New Revision
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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:

Module	Amount of GSM Port	Amount of WCDMA Port	Amount of CDMA Port	Supported Frequency band
4016-16G	16			GSM: 850/900/1800/1900MHz
4008-8G	8			
4004-4G	4			
4008-8W		8		GSM: 900/1800MHz UMTS: 900/2100MHz
4004-4W		4		
4008-8C			8	CDMA: CDMA 2000 800MHz
4004-4C			4	

Table 1-1 Model List

## 1.1 Typical Application

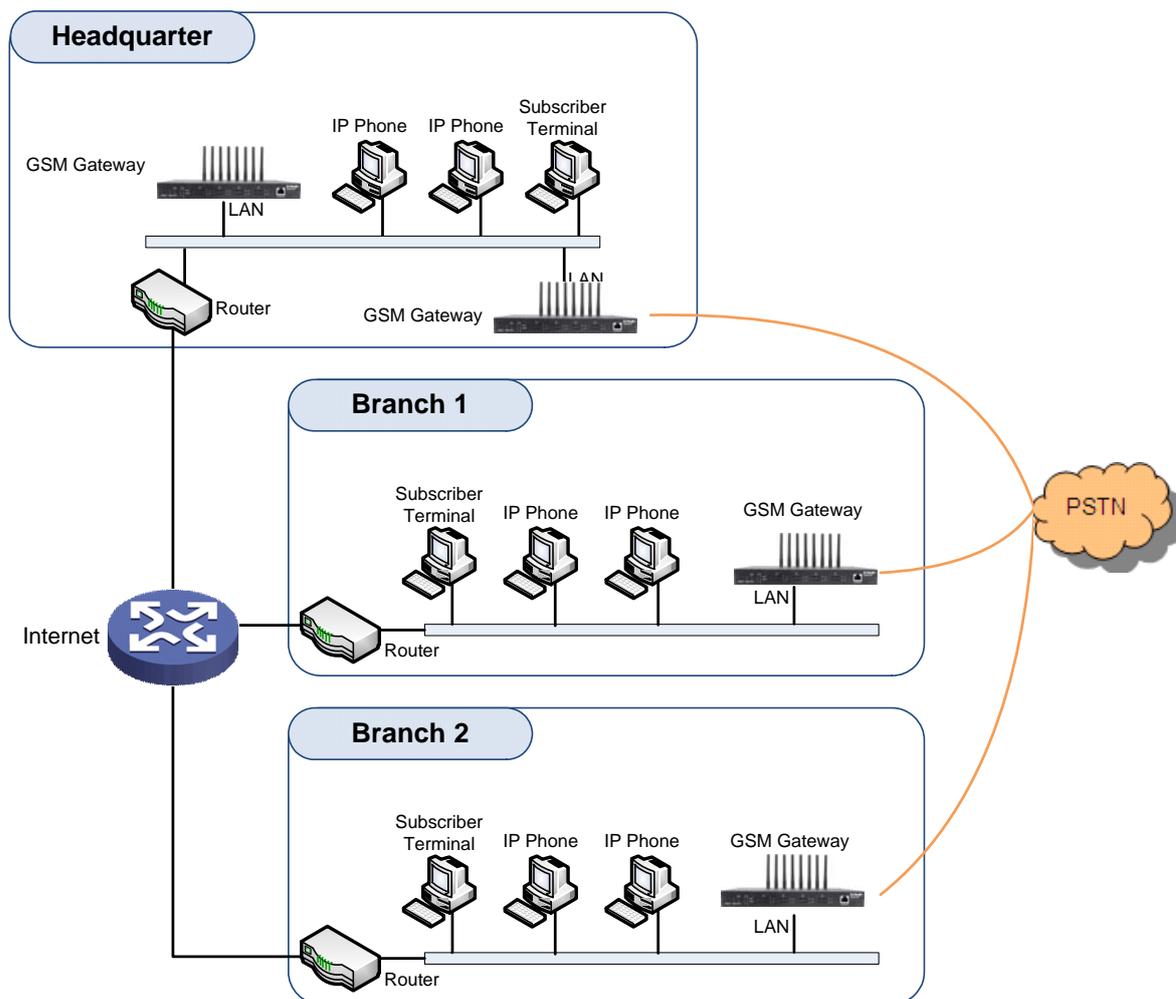


Figure 1-1 Typical Application

## 1.2 Feature List

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

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

## 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

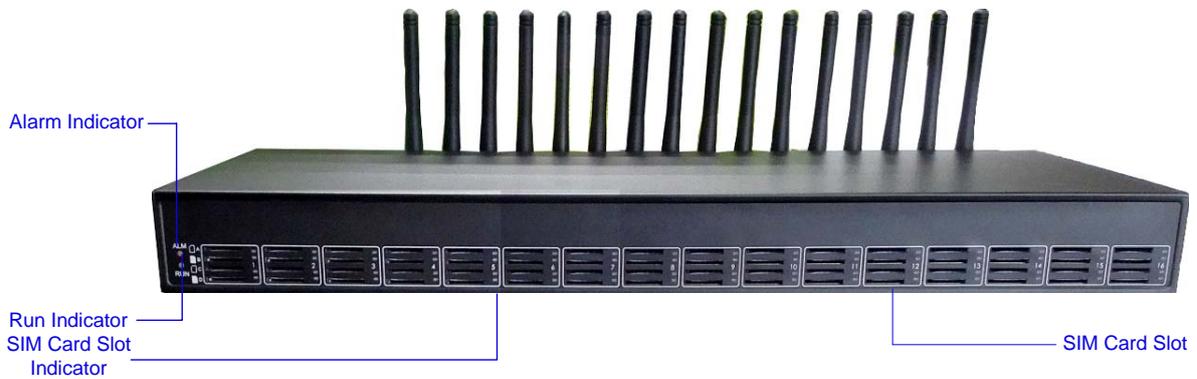


Figure 1-4 SMG4016 Front View



Figure 1-5 SMG4016 Rear View

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

Interface	Description
<b>LAN</b>	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 <a href="#">1.4 Indicator Info</a>
<b>SIM Card Slot</b>	Amount: 4, 8, 16*4
	Network Supported: GSM, WCDMA, CDMA
<b>Console Port</b>	Amount: 1
	Type: RS-232
	Baud Rate: 115200bps
	Connector: RJ45 to DB-9 Connector (4004, 4008 series), Mini-USB connecting line (4016 series)
	Data Bits: 8 bits
	Stop Bit: 1 bit
	Parity Unsupported
	Flow Control Unsupported
<b>External Power Supply Interface</b>	Provide the 12V voltage with positive inside and negative outside, and the current is larger than 3A
<b>Button</b>	<b>Description</b>
<b>Reset Button</b>	Restore the gateway to factory settings by pressing this button persistently for 3 seconds
<b>LED</b>	<b>Description</b>
<b>Power Indicator</b>	Indicates the power state. It lights up when the gateway starts up with the power cord well connected
<b>Run Indicator</b>	Indicates the running status. For more details, refer to <a href="#">1.4 Indicator Info</a> .
<b>Alarm Indicator</b>	Alarms the device malfunction. For more details, refer to <a href="#">1.4 Indicator Info</a> .
<b>Link Indicator</b>	The green LED on the right of LAN, indicating the network connection status.
<b>ACT Indicator</b>	The orange LED on the left of LAN, whose flashing tells the data are being transmitted.
<b>Port Indicator</b>	1. When the port is idle, the LED Lights up in green and keeps on;

	<ol style="list-style-type: none"> <li>2. When the port is unavailable, the LED Lights up in red and keeps on;</li> <li>3. When the port is in use, the LED flashes in green</li> <li>4. When the port module is disabled, the LED flashes in red</li> <li>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.</li> </ol>
--	--

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
<b>Run Indicator</b>	Go out	System is not yet started.
	Light up and flash fast	System is starting.
	Flash slowly	Device is normal.
<b>Alarm Indicator</b>	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 Rubber Antenna \*4/8/16
- Standard RJ45 to DB-9 Switcher (4004/4008 series) \*1, Mini-USB connecting line (4016 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 [3.1 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 [3.10.5 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 [3.5.1 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 [3.5.4 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 [3.7.2 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 [3.8.3 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 [3.7.2 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 [3.8.2 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, whose original username and password are both 'admin'. You can change the username and the password via 'System Tools → Change Password' on the WEB interface. For detailed instructions, refer to [3.10.5 Change Password](#).

After login, you can see the main interface as below.

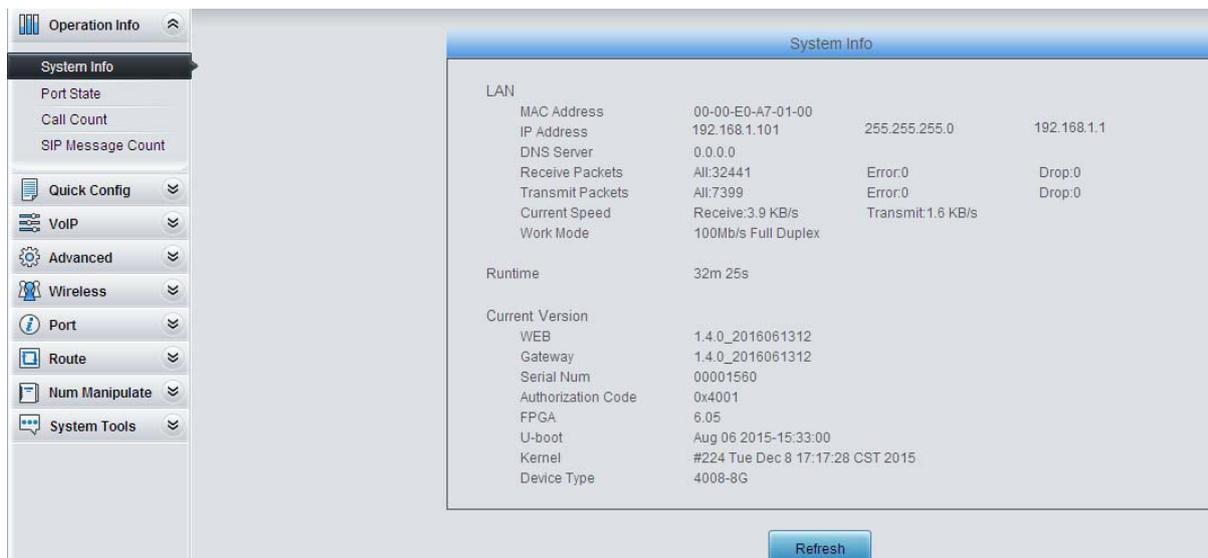


Figure 3-2 Main Interface

## 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. See Figure 3-3.



Figure 3-3 Operation Info

### 3.2.1 System Info

System Info

<b>LAN</b>			
MAC Address	00-00-E0-A7-01-00		
IP Address	192.168.1.101	255.255.255.0	192.168.1.1
DNS Server	0.0.0.0		
Receive Packets	All:32441	Error:0	Drop:0
Transmit Packets	All:7399	Error:0	Drop:0
Current Speed	Receive:3.9 KB/s	Transmit:1.6 KB/s	
Work Mode	100Mb/s Full Duplex		
Runtime	32m 25s		
<b>Current Version</b>			
WEB	1.4.0_2016061312		
Gateway	1.4.0_2016061312		
Serial Num	00001560		
Authorization Code	0x4001		
FPGA	6.05		
U-boot	Aug 06 2015-15:33:00		
Kernel	#224 Tue Dec 8 17:17:28 CST 2015		
Device Type	4008-8G		

Figure 3-4 System Info Interface

See Figure 3-4 for the system info interface. You can click **Refresh** to obtain the latest system information. The table below explains the items shown in Figure 3-4.

Item	Description
<b>MAC Address</b>	MAC address of LAN.
<b>IP Address</b>	The three parameters from left to right are IP address, subnet mask and default gateway of LAN.
<b>DNS Server</b>	DNS server address of LAN.

<b>Receive Packets</b>	The amount of receive packets after the gateway's startup, including three options: All, Error and Drop.
<b>Transmit Packets</b>	The amount of transmit packets after the gateway's startup, including three options: All, Error and Drop.
<b>Current Speed</b>	Show the current speed of data receiving and transmitting.
<b>Work Mode</b>	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.
<b>Runtime</b>	Time of the gateway keeping running normally after startup, which will be automatically updated.
<b>WEB</b>	Current version of the WEB interface.
<b>Gateway</b>	Current version of the gateway service.
<b>Serial Num</b>	Unique serial number of a wireless gateway.
<b>Authorization Code</b>	The authorization codes vary from different wireless modules.
<b>FPGA</b>	Current version of FPGA.
<b>U-boot</b>	Current version of Uboot.
<b>Kernel</b>	Current version of the system kernel on the gateway.
<b>Device Type</b>	Type of the wireless gateway.

### 3.2.2 Port State

Port State											
Port	Type	State	Voice Type	Direction	CallerID	CalleeID	SIM Card Used	Cell Phone No.	Connection	Signal	SIP Reg Status
1	GSM	Idle	---	---	---	---		13023634112	Connect		Unregistered
2	GSM	Idle	---	---	---	---		13023634183	Connect		Unregistered
3	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
4	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
5	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
6	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
7	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
8	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
9	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
10	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
11	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
12	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
13	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
14	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
15	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered
16	GSM	Unusable	---	---	---	---		---	Disconnect		Unregistered

Figure 3-5 Channel State Interface

See Figure 3-5 for the channel state interface where shows the channel type, the channel state for each channel on the gateway. The table below explains the items shown in Figure 3-5.

Item	Description
<b>Port</b>	Port number on the device.
<b>Type</b>	Port type on the device. So far, only GSM, WCDMA and CDMA types are supported.
<b>State</b>	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.
<b>Voice Type</b>	Displays the voice type of the current call.		
<b>Direction</b>	Displays the direction of the call on port.		
<b>CallerID</b>	Displays the CallerID of the call on port.		
<b>CalleeID</b>	Displays the CalleeID of the call on port.		
<b>SIM Card</b>	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 SMG4004 and SMG4008 series.		
<b>Cell Phone No.</b>	Displays the number of the SIM card inserted in current port. For SMG4016 series, the number is that of the SIM card which is in using.		
<b>Connection</b>	Displays the connection status between the SIM card and the base station.		
<b>Signal</b>	Displays the signal intensity of the wireless module.		
<b>SIP Reg Status</b>	Displays the registration status of the port.		

### 3.2.3 Call Count

Call Count							
Call Direction	Total Calls	Successful Calls	Busy	No Answer	Routing Failure	Dialing Failure	Unknown
IP->Tel	2	2	0	0	0	0	0
Tel->IP	1	0	0	0	0	1	0

Refresh

Figure 3-6 Call Count Interface

See Figure 3-6 for the call count Interface. The above list 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. The table below explains the items shown in Figure 3-6.

Item	Description
<b>Call Direction</b>	A condition for call count, two options available: <i>IP</i> → <i>Tel</i> and <i>Tel</i> → <i>IP</i> .
<b>Total Calls</b>	Total number of calls in a specified call direction.
<b>Successful Calls</b>	Total number of successful calls in conversation.
<b>Busy</b>	Total number of calls which fail as the called party has been occupied and replies a busy message.

<b>No Answer</b>	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.
<b>Routing Failure</b>	Total number of calls which fail because no routing rules are matched.
<b>Dialing Failure</b>	Total number of calls which fail as the called party number does not conform to the dialing rule or due to dialing timeout.
<b>Unknown Failure</b>	Total number of calls which fail due to unknown reasons.

### 3.2.4 SIP Message Count

Request								
Request	REGISTER	INVITE	ACK	INFO	BYE	CANCEL	NOTIFY	OPTION
Send	0	1	1	0	1	0	0	0
Send Repeatedly	0	0	0	0	0	0	0	0
Receive	0	1	1	0	1	0	0	0
Receive Repeatedly	0	0	0	0	0	0	0	0

Common Response						
Common Response	100 Trying	180 Ringing	183 Session Progress	200 OK	486 Busy	487 Request Already Terminated
Send	1	1	0	2	0	0
Receive	1	1	0	2	0	0

Figure 3-7 SIP Message Count Interface

See Figure 3-7 for the SIP Message Count interface. This is used to record 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 3-8 Quick Config Interface

See Figure 3-8 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.

See Figure 3-9 for the Quick Config-Network Settings interface. Refer to 3.5.1 Network for detailed settings. After configuration, click **Next** to enter the SIP Settings interface.

Quick Config-Network Settings

Network Type:	Static <span style="float: right;">▼</span>
IP Address (I)	192.168.1.101
Subnet Mask (U)	255.255.255.0
Default Gateway (D)	192.168.1.1
DNS Server (P)	0.0.0.0
Speed and Duplex Mode	Automatic Detection <span style="float: right;">▼</span>

Figure 3-9 Quick Config-Network Settings Interface

See Figure 3-10 for the Quick Config-SIP Settings interface. The configuration items on this interface are the same as those on the SIP interface. Refer to [3.4.1 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.

Quick Config-SIP Settings

Registrar IP Address	<input type="text"/>
Registrar Port	<input type="text"/>
Spare Registrar IP Address	<input type="text"/>
Spare Registrar Port	<input type="text"/>
Registry Validity Period (s)	600

Figure 3-10 Quick Config-SIP Settings Interface

See Figure 3-11 for the Port Settings interface. The configuration items on this interface are the same as those on the Port interface. Refer to [3.7.1 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	Type	SIP Account	Authentication Username	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	Echo Canceller	Color Ring	Color Ring Index	Server Index	Modify	
1	GSM	8001	---	Static Binding	180	Disable	Disable	Failed	Enable	Enable	Disable	---	---		
2	GSM	182	---	Static Binding	8003	Disable	Disable	Unregistered	Enable	Enable	Disable	---	---		
3	GSM	8003	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Enable	Disable	---	---		
4	GSM	8004	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Enable	Disable	---	---		
5	GSM	8005	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Enable	Disable	---	---		
6	GSM	8006	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Enable	Disable	---	---		
7	GSM	8007	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Enable	Disable	---	---		
8	GSM	8008	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Enable	Disable	---	---		

Figure 3-11 Port Settings Interface

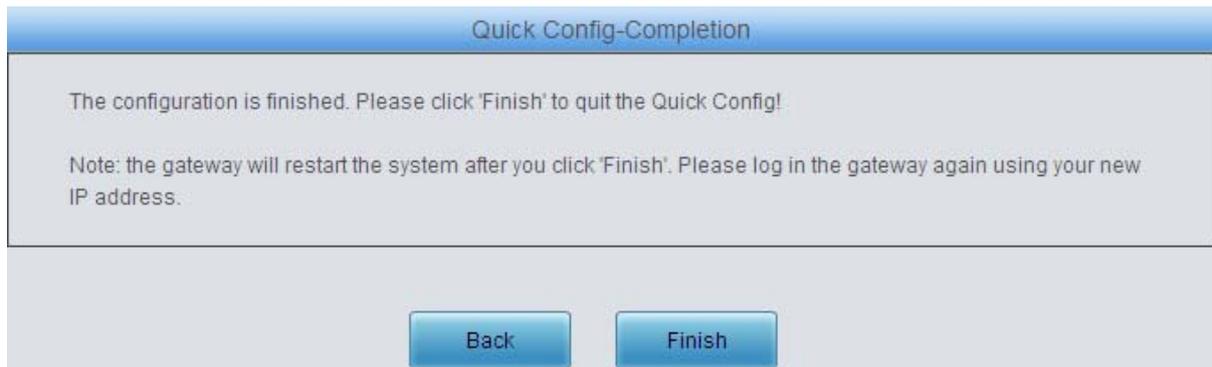


Figure 3-12 Quick Config-Completion Interface

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**. See Figure 3-13. **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.



Figure 3-13 VoIP Settings

### 3.4.1 SIP

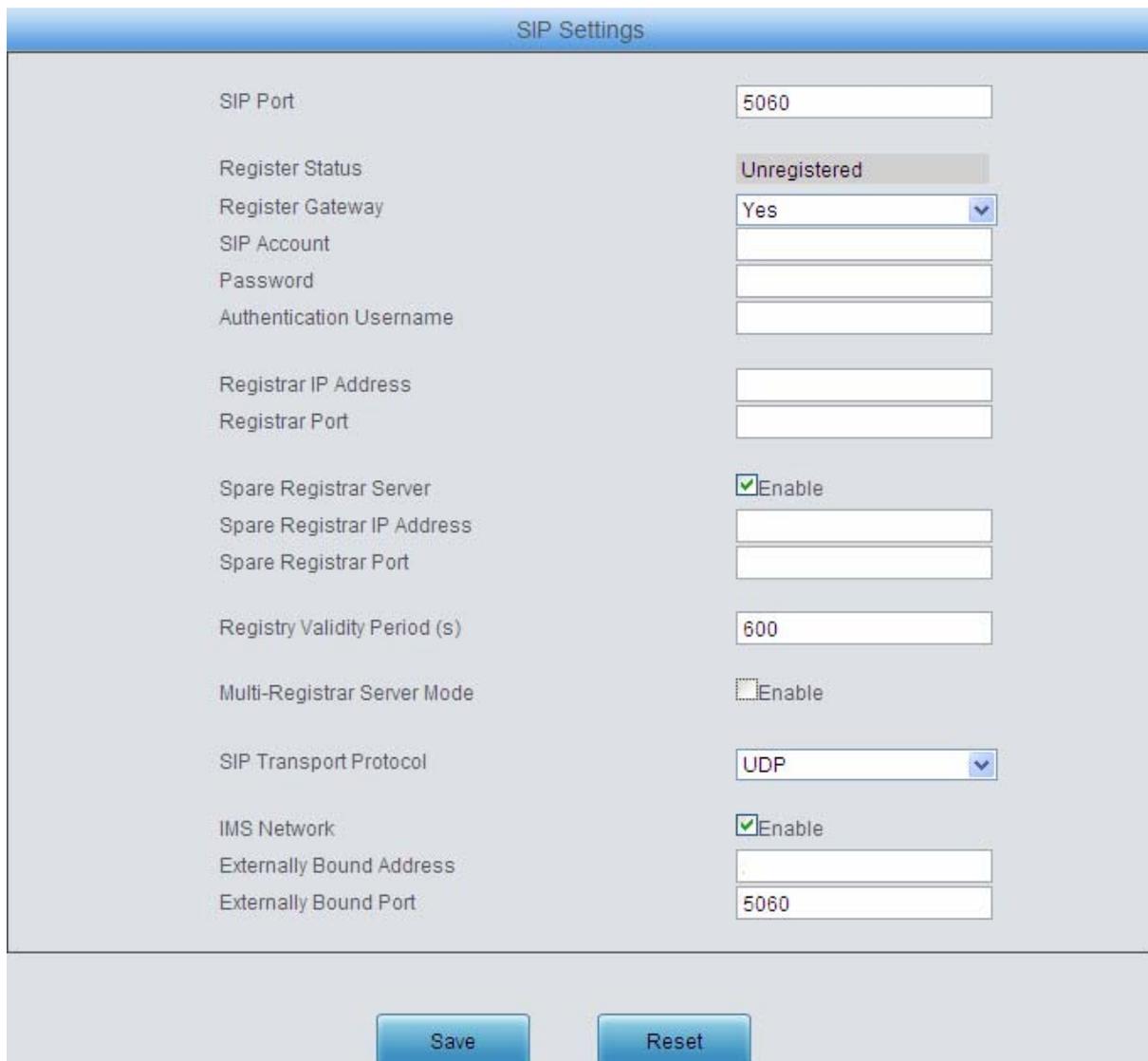


Figure 3-14 SIP Settings Interface

See Figure 3-14 for the SIP settings interface where 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 [3.10.8 Restart](#) for detailed instructions. The table below explains the items shown in Figure 3-14.

Item	Description
<b>SIP Port</b>	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.
<b>Register Status</b>	Registration status of the gateway. When <b>Register Gateway</b> is set to <i>No</i> , the value of this item is <i>Unregistered</i> ; when <b>Register Gateway</b> is set to <i>Yes</i> , the value of this item is either <i>Failed</i> or <i>Registered</i> .
<b>Register Gateway</b>	Sets whether to register the gateway as a whole. The default value is <i>No</i> . Only when this configuration is set to <i>Yes</i> can you see the configuration items <b>SIP Account</b> and <b>Password</b> .

<b>SIP Account</b>	When the gateway initiates a call to SIP, this item corresponds to the username of SIP.
<b>Password</b>	Registration password of the gateway. To register the gateway to SIP, both configuration items <b>SIP Account</b> and <b>Password</b> should be filled in.
<b>Authentication Username</b>	Authentication username for registration.
<b>Registrar IP Address</b>	Address of the registry server for the gateway to register.
<b>Registrar Port</b>	Signaling port of the registry server.
<b>Spare Registrar Server</b>	Check the enable checkbox to enable the spare registrar server. By default, it is <b>disabled</b> .
<b>Spare Registrar IP Address</b>	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 <b>Detection Server Cycle</b> is enabled.
<b>Spare Registrar Port</b>	Signaling port of the spare registry server.
<b>Registry Validity Period</b>	Validity period of the SIP registry. Once the registry is overdue, the gateway should be registered again. This configuration item is valid only when <b>Register Gateway</b> is set to Yes. Range of value: 10~3600, calculated by s, with the default value of 600.
<b>Multi-Registrar Server Mode</b>	Tick the checkbox before to enable the multi-registrar server mode. By default, it is <b>disabled</b> .
<b>SIP Transport Protocol</b>	There are two modes <b>UDP</b> and <b>TCP</b> available for running the SIP protocol. The default value is <b>UDP</b> .
<b>IMS Network</b>	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 <b>disabled</b> . Only when this feature is <b>enabled</b> will these items <b>Externally Bound Address</b> , <b>Externally Bound Port</b> and <b>Authentication Username</b> be shown.
<b>Externally Bound Address</b>	Externally bound IP address for registration.
<b>Externally Bound Port</b>	Externally bound port for registration.

### 3.4.2 SIP Compatibility

See Figure 3-15 for the SIP Compatibility interface where 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.

SIP Compatibility

Obtain CalleeID from	"Request" Field <input type="button" value="v"/>
Set CallerID position	Username of From Field <input type="button" value="v"/>
Obtain CallerID from	Username of From Field <input type="button" value="v"/>
Use Contact Address	<input type="checkbox"/> Enable
Two Stage Dialing for SIP Incoming Call	<input type="checkbox"/> Enable
Maximum Wait Answer Time (s)	<input type="text" value="60"/>
SIP Station Supported	<input type="checkbox"/> Enable
Set SIP Identifying	<input type="text" value="Gateway"/>
Call Hangup when RTP Timeout(s)	<input type="text" value="0"/>
Ignore ACK	<input type="checkbox"/> Enable
Abnormal Call Hangup Detection	<input checked="" type="checkbox"/> Enable
Cycle(s)	<input type="text" value="0"/>
Server Status Detection	<input checked="" type="checkbox"/> Enable
Cycle(s)	<input type="text" value="0"/>
Occasion to Reply 183	Immediately <input type="button" value="v"/>
Occasion to Reply 200 Ok	After pickup <input type="button" value="v"/>

Figure 3-15 SIP Compatibility Setting Interface

The table below explains the items shown in Figure 3-15.

Item	Description
<b>Obtain CalleeID from</b>	There are two optional ways to obtain the called party number: from "To" Field and from "Request" Field. The default value is "Request" Field.
<b>Set CallerID Position</b>	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".
<b>Obtain CallerID from</b>	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".
<b>Use Contact Address</b>	Sets whether to send the request message according to the content of Contact, with the default setting of <i>disabled</i> . 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.
<b>Two Stage Dialing for SIP Incoming Call</b>	Once this feature is enabled, the incoming call from SIP should perform the two stage dialing operation. By default this feature is disabled.
<b>Maximum Wait Answer Time</b>	Sets the maximum time for the SIP channel to wait for the answer from the called 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.
<b>SIP Station Supported</b>	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.
<b>Set SIP Identifying</b>	Sets the SIP identifying content in the SIP call message. The default setting is <i>Gateway</i> .
<b>Maximum Wait RTP Time</b>	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 ( <i>disabled</i> ), calculated by s.
<b>Ignore ACK</b>	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 <i>disabled</i> .
<b>Abnormal Call Hangup Detection</b>	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.
<b>Server Status Detection</b>	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.
<b>Occasion to Reply 183</b>	Sets the occasion to reply the 183 message. Two options including: Immediately and After ringing, with the default value of <i>Immediately</i> .
<b>Occasion to Reply 200 Ok</b>	Sets the occasion to reply 200 OK. Two options including: After pickup and After ringing, with the default value of <i>After pickup</i> .

### 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 [3.4.2 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. See Figure 3-16 below.

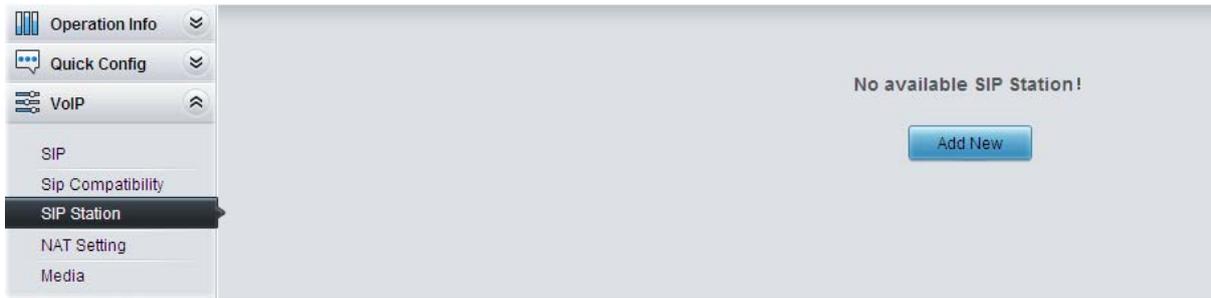


Figure 3-16 SIP Station Setting Interface

Click **Add New** to add SIP stations manually. See Figure 3-17. 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.

Figure 3-17 Add New SIP Station

The table below explains the items shown above:

Item	Description
<b>Number</b>	The logical number for a SIP station to register to the gateway.
<b>Username</b>	The username used to register a SIP station to the gateway.
<b>Password</b>	The password used to register a SIP station to the gateway.
<b>Bound Port</b>	The wireless port which is bound to the SIP station.
<b>Description</b>	It is user-defined, with the default value of <i>default</i> .
<b>Batch Setting</b>	Used to set multiple SIP stations at the same time.

After configuration, click **Save** to save the above settings into the gateway or click **Close** to cancel the settings. See Figure 3-18 for the applied SIP station information.

SIP Station									
Check	Number	Username	IP Address	Bound Port	Register Status	Register Duration (s)	Voice Channel State	Description	Modify
<input type="checkbox"/>	0	120	--	1	Unregistered	--	--	default	

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

Figure 3-18 SIP Station Interface

Click **Modify** in the above figure to modify the configuration of the SIP station. See Figure 3-19. The configuration items on this interface are the same as those on the **Add New SIP Station** interface.

SIP Station

Number:

Username:

Password:

Bound Port:  ▼

Description:

Batch Setting:  Enable

Figure 3-19 SIP Station Modification Interface

To delete a SIP station, check the checkbox before the corresponding index in Figure 3-18 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 3-18.

### 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. See Figure 3-20 below.

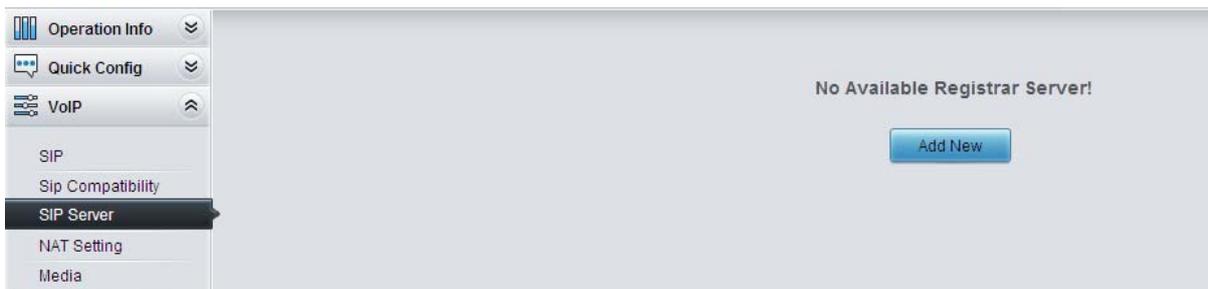


Figure 3-20 SIP Server Interface

Click **Add New** to add SIP servers manually. See Figure 3-21. You can configure basic SIP server information on this interface.

Figure 3-21 Add New SIP Server

All the items except Index and Description are the same as those on [the SIP](#) interface ([3.4.1 SIP](#)).

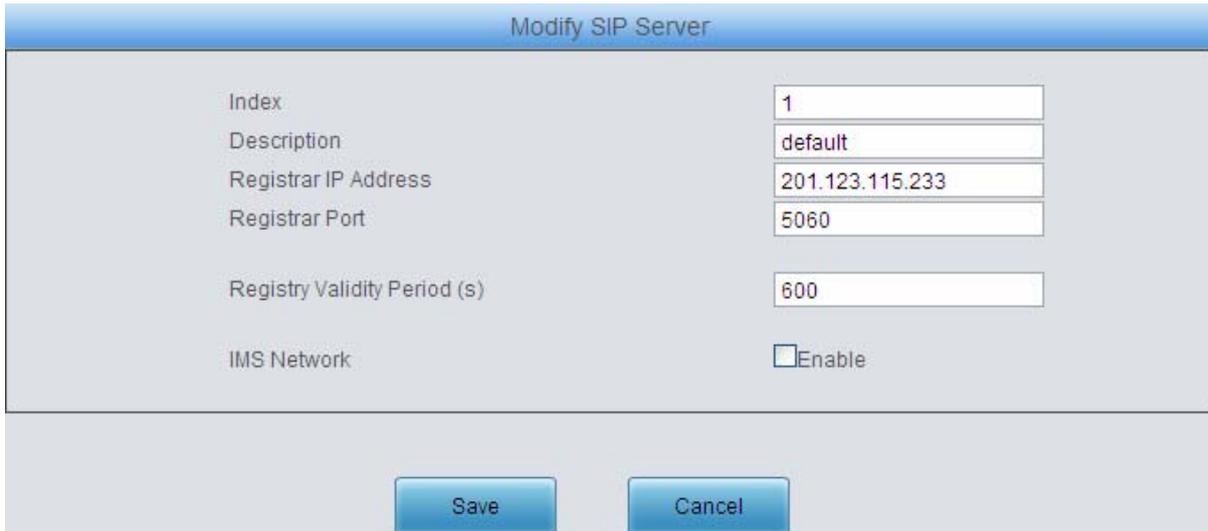
Item	Description
<b>Index</b>	The index of each SIP server. The gateway supports up to 8 SIP servers.
<b>Description</b>	More information about each SIP server, with the default value of <i>default</i> .

After configuration, click **Save** to save the above settings into the gateway or click **Cancel** to cancel the settings. See Figure 3-22 for the SIP server management interface.

Figure 3-22 SIP Server Management

Click **Modify** in the above figure to modify the configuration of the SIP server. See Figure 3-23.

The configuration items on this interface are the same as those on the **Add New SIP Server** interface.



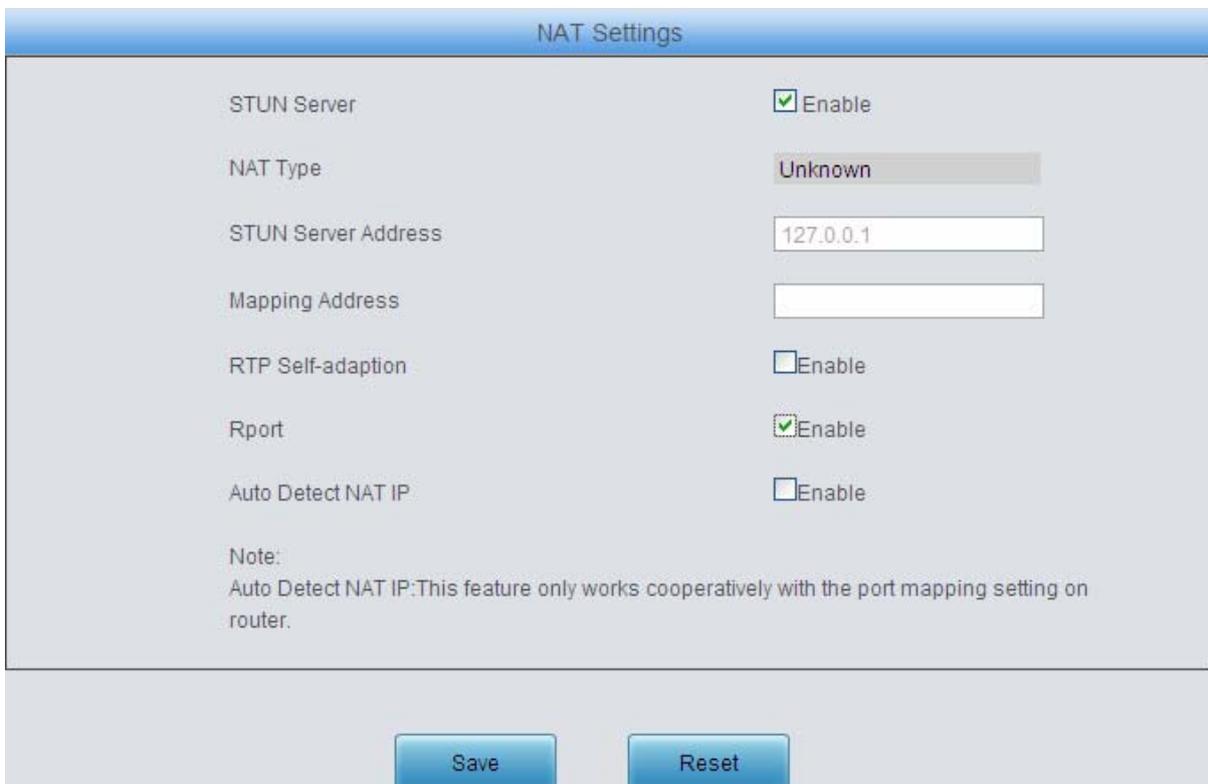
Index	1
Description	default
Registrar IP Address	201.123.115.233
Registrar Port	5060
Registry Validity Period (s)	600
IMS Network	<input type="checkbox"/> Enable

Figure 3-23 SIP Server Modification Interface

To delete a SIP server, check the checkbox before the corresponding index in Figure 3-22 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 3-22.

### 3.4.5 NAT Setting

See Figure 3-24 for the NAT setting interface where 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.



STUN Server	<input checked="" type="checkbox"/> Enable
NAT Type	Unknown
STUN Server Address	127.0.0.1
Mapping Address	
RTP Self-adaption	<input type="checkbox"/> Enable
Rport	<input checked="" type="checkbox"/> Enable
Auto Detect NAT IP	<input type="checkbox"/> Enable

Note:  
Auto Detect NAT IP: This feature only works cooperatively with the port mapping setting on router.

Figure 3-24 NAT Setting Interface

The table below explains the items shown in Figure 3-24.

Item	Description
<b>STUN Server</b>	Sets whether to enable the STUN server for NAT traversal. By default the STUN server is disabled.
<b>NAT Type</b>	Detected NAT (Network Address Translation) type. The gateway will return the NAT type automatically in case <b>STUN Server</b> 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).
<b>STUN Server Address</b>	Address of the server for STUN traversal.
<b>Mapping Address</b>	<p>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.</p> <p><b>Note:</b> Once this item is filled out, it will be used as the first choice even if Rport and NAT IP are enabled.</p>
<b>RTP Self-adaption</b>	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 <i>disabled</i> .
<b>Rport</b>	When this feature is enabled, a corresponding Rport field will be added to the Via message of SIP. The default value is <i>enabled</i> .
<b>Auto Detect NAT IP</b>	<p>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 <i>disabled</i>.</p> <p><b>Note:</b> This feature gets valid only when Rport is enabled.</p>

### 3.4.6 Media

Media Parameters

DTMF Transmit Mode RFC2833

RFC2833 Payload 101

RTP Port Range 50000,50767

Silence Suppression Disable

JitterBuffer 20

Voice Gain Output from IP (dB) 0

AGC  Enable

Target Energy Threshold (dB) 0

Maximum Gain Threshold (dB) 48

Maximum Attenuation Threshold (dB) 0

Minimum Input Energy (dB) -60

**CODEC Priority**

Check	Priority	CODEC	Packing Time	Bit Rate (kbs)
<input checked="" type="checkbox"/>	1	G711A <input type="button" value="v"/>	20 <input type="button" value="v"/>	64 <input type="button" value="v"/>
<input checked="" type="checkbox"/>	2	G711U <input type="button" value="v"/>	20 <input type="button" value="v"/>	64 <input type="button" value="v"/>
<input checked="" type="checkbox"/>	3	G729 <input type="button" value="v"/>	20 <input type="button" value="v"/>	8 <input type="button" value="v"/>
<input checked="" type="checkbox"/>	4	G723 <input type="button" value="v"/>	30 <input type="button" value="v"/>	6.3 <input type="button" value="v"/>
<input checked="" type="checkbox"/>	5	G722 <input type="button" value="v"/>	30 <input type="button" value="v"/>	64 <input type="button" value="v"/>
<input checked="" type="checkbox"/>	6	AMR <input type="button" value="v"/>	20 <input type="button" value="v"/>	4.75 <input type="button" value="v"/>
<input checked="" type="checkbox"/>	7	iLBC <input type="button" value="v"/>	30 <input type="button" value="v"/>	13.3 <input type="button" value="v"/>

Figure 3-25 Media Settings Interface

See Figure 3-25 for the media settings interface where 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 [3.10.8 Restart](#) for detailed instructions. The table below explains the items shown in Figure 3-25.

Item	Description
<b>DTMF Transmit Mode</b>	Sets the transmit mode for the IP channel to send DTMF signals. The optional values are <i>RFC2833</i> , <i>In-band</i> and <i>Signaling</i> , with the default value of <i>RFC2833</i> .
<b>RFC2833 Payload</b>	Payload of the RFC2833 formatted DTMF signals on the IP channel. Range of value: 90~127, with the default value of 101.

<b>RTP Port Range</b>	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.
<b>Silence Suppression</b>	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 <i>Enable</i> and <i>Disable</i> , with the default value of <i>Disable</i> .
<b>JitterBuffer</b>	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.
<b>Voice Gain Output from IP</b>	Adjusts the gain of the voice output from IP. Range of value: -24~12, calculated by dB, with the default value of 0.
<b>AGC</b>	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.
<b>Target Energy Threshold</b>	Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the default value of 0.
<b>Maximum Gain Threshold</b>	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.
<b>Maximum Attenuation Threshold</b>	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.
<b>Minimum Input Energy</b>	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.

<b>CODEC Priority</b>	Supported CODECs and their corresponding priority for the IP end to establish a call conversation. The table below explains the sub-items:		
	<b>Sub-item</b>	<b>Description</b>	
	<i>Priority</i>	Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.	
	<i>CODEC</i>	Three optional CODECs are supported: <i>G711A</i> , <i>G711U</i> , <i>G729A/B</i> , <i>G723</i> , <i>G722</i> , <i>AMR</i> and <i>iLBC</i> .	
	<i>Packing Time</i>	Time interval for packing an RTP packet, calculated by ms.	
	<i>Bit Rate</i>	The number of thousand bits (excluding the packet header) that are conveyed per second.	
	By default, all of the seven CODECs are supported and ordered <i>G711A</i> , <i>G711U</i> , <i>G729A/B</i> , <i>G723</i> , <i>G722</i> , <i>AMR</i> and <i>iLBC</i> 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.		
	<b>COEDC</b>	<b>Packing Time (ms)</b>	<b>Bit Rate (kbps)</b>
	<i>G711A</i>	10 / <b>20</b> / 30 / 40 / 60	<b>64</b>
<i>G711U</i>	10 / <b>20</b> / 30 / 40 / 60	<b>64</b>	
<i>G729A/B</i>	10 / <b>20</b> / 30 / 40 / 60	<b>8</b>	
<i>G723</i>	<b>30</b> / 60	5.3 / <b>6.3</b>	
<i>G722</i>	10 / 20 / <b>30</b> / 40	<b>64</b>	
<i>AMR</i>	<b>20</b> / 40 / 60	4.75	
<i>iLBC</i>	<b>20</b> / 40	15.2	
	<b>30</b> / 60	<b>13.3</b>	

## 3.5 Advanced Settings

Advanced Settings includes eleven parts: **Network**, **System Param**, **Service Config**, **Dialing Rule**, **Function Key**, **Cue Tone**, **Color Ring**, **QoS**, **Tone Generator**, **CDR Query** and **VPN**. See Figure 3-26. **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 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.

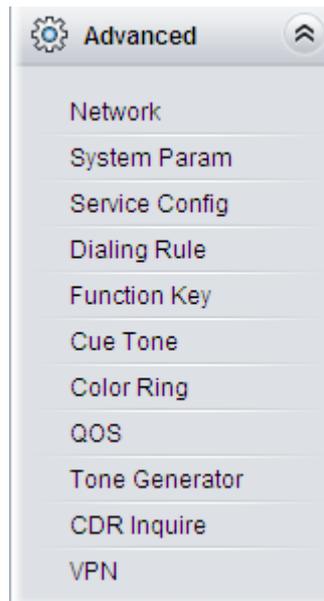


Figure 3-26 Advanced Settings

### 3.5.1 Network

A screenshot of the "Network Settings" web interface. The title "Network Settings" is centered at the top in a blue header. Below the header, there are several configuration fields arranged in two columns. The first column contains labels: "Network Type:", "IP Address (I)", "Subnet Mask (U)", "Default Gateway (D)", "DNS Server (P)", and "Speed and Duplex Mode". The second column contains input fields: a dropdown menu set to "Static", a text box with "192.168.1.101", a text box with "255.255.255.0", a text box with "192.168.1.1", a text box with "0.0.0.0", and a dropdown menu set to "Automatic Detection". Below these fields is a red text note: "Note: 1. Please log in again using your new IP address if the IP address has been modified! 2. If you select 'DHCP', your IP address will be allocated randomly. Please dial the port's number and press the corresponding function key to inquire it." At the bottom of the interface are two blue buttons labeled "Save" and "Reset".

Figure 3-27 Network Settings Interface

See Figure 3-27 for the network settings interface. 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

System Param

**WEB Management**

WEB Port

Access Setting

**SYSLOG Parameters**

SYSLOG Enabled  Yes  No

Server Address

SYSLOG Level

AT Debug Enabled  Yes  No

Echo Mode Enabled  Yes  No

Port

**CDR Parameters**

CDR Enabled  Yes  No

Server Address

Server Port

Save CDR  Yes  No

Amount of Saved CDR

**API Parameters**

API Enabled  Yes  No

Remote IP Address Allowed to Invoke API

(Separated by ','; '\*' denotes all IP addresses)

Username for API Call

Password for API Call

**Time Parameters**

Time Calibration  NTP  Synchronized with Operator  Close

NTP Server Address

Synchronizing Cycle

System Time  Modify

Time Zone

Daily Restart  Yes  No

Restart Time  h  m

Figure 3-28 System Parameters Setting Interface

See Figure 3-28 for the System Parameters Setting interface. The table below explains the items shown in the above figure.

Item	Description
<b>WEB Port</b>	The port which is used to access the gateway via WEB. The default value is 80.

<b>Access Setting</b>	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.
<b>SYSLOG Enabled</b>	Sets whether to enable SYSLOG. It is required to fill in <b>SYSLOG Server Address</b> and <b>SYSLOG Level</b> in case SYSLOG is enabled. By default, <b>SYSLOG</b> is disabled.
<b>Server Address</b>	Sets the SYSLOG server address for log reception.
<b>SYSLOG Level</b>	Sets the SYSLOG level. There are three options: <i>ERROR</i> , <i>WARNING</i> , <i>INFO</i> and <i>DEBUG</i> . The default value is <i>INFO</i> .
<b>AT Debug Enabled</b>	Sets whether to enable the AT debug feature, with the default value of <i>No</i> . Once this feature is enabled, the related information about AT will be output to the SYSLOG.
<b>Echo Mode Enabled</b>	Sets whether to enable the echo mode, with the default value of <i>No</i> . Once this feature is enabled, both the sent and received information will be displayed.
<b>Port</b>	Select the port to execute the AT debug.
<b>CDR Enabled</b>	Sets whether to enable the feature of CDR. It is required to fill in <b>Server Address</b> and <b>Server Port</b> in case CDR is enabled. By default, <b>CDR</b> is disabled.
<b>Server Address</b>	Sets the server address to receive CDR.
<b>Server Port</b>	Sets the server port to receive CDR.
<b>Save CDR</b>	Sets whether to save CDR, with the default value of <i>NO</i> .
<b>Amount of Saved CDR</b>	Sets the amount of saved CDR. Range of value: 200~10000, with the default value of 5000.
<b>API Enabled</b>	When this feature is enabled, the remote terminal can invoke the API interface. The default value is <i>No</i> .
<b>Remote IP Address allowed to Invoke API</b>	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.
<b>Username for API Call, Password for API Call</b>	The authorized username and password for calling the API interface.
<b>Time Calibration</b>	Sets the calibration mode for the time. Three options available: <i>NTP</i> , <i>Synchronized with Operator</i> and <i>Close</i> , with the default value of <i>Synchronized with Operator</i> .
<b>NTP Server Address</b>	Sets the Server address for NTP time synchronization.
<b>Synchronizing Cycle</b>	Sets the cycle for NTP time synchronization. The default value is 3600.
<b>System Time</b>	The system time. Check the checkbox before <b>Modify</b> and change the time in the edit box if <i>Time Calibration</i> is set to <i>Close</i> .
<b>Time Zone</b>	The time zone of the gateway.
<b>Daily Restart</b>	Sets whether to restart the gateway regularly every day at the preset <b>Restart Time</b> . By default, this feature is disabled.
<b>Restart Time</b>	Sets the time to restart the gateway regularly.

### 3.5.3 Service Config

Service Config

**Service Parameters**

Enable Two Stage Dialing Mode for PSTN Outgoing Calls  Disable  Enable

Maximum Wait Time for PSTN Outgoing Calls  s

Dial Interval  s

Busy Tone Detection Mode  Common  Delay  Ignore

**Abnormality Handling**

Communicate without Network  Disable  Enable

IP->Tel Call Failure, Auto Transfer  Disable  Enable

Tel->IP Call Failure, Auto SMS Reply  Disable  Enable

**Echo Canceller**

Work Mode

Non-linear Processing  Enable

Fixed Window Size (Near-end, Narrowband 8kHz)

Moving Window Size (Far-end, Narrowband 8kHz)

Figure 3-29 Service Config Interface

See Figure 3-29 for the Service Config interface. The table below explains the items shown in the above figure.

Item	Description
<b>Enable Two Stage Dialing Mode for PSTN Outgoing Calls</b>	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 <i>disabled</i> .
<b>Maximum Wait Time for PSTN Outgoing Calls</b>	Sets the maximum wait time waiting for the called party pickup during an outgoing call. Range of value: 10~120, calculated by s, with the default value of 60.
<b>Dial Interval</b>	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.
<b>Busy Tone Detection Mode</b>	Sets the busy tone detection mode, three options available: Common (hangup on busy), Delayed (Delayed hangup on busy), Undetected (no busy detection). By default it is set to Common.

<b>Communication without Network</b>	Automatically routes a call to the wireless port in case of network failure or call timeout. The default value is <i>disabled</i> .
<b>IP→Tel Call Failure, Auto Transfer</b>	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 <i>disable</i> . If this feature is enabled, you are required to enter Target Number (Registered) or Target IP and Target Port (Unregistered).
<b>Tel → IP Call Failure, Auto SMS Reply</b>	Sets whether to enable the feature of automatic SMS reply when a call from Tel to IP fails, with the default value of <i>disable</i> . 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.
<b>Work Mode</b>	Sets the work mode for the echo canceller. There are two options: <i>Near-end cancellation</i> and <i>Both near-end and far-end cancellation</i> , with the default value of <i>Near-end cancellation</i> .
<b>Non-linear Processing</b>	Sets whether to enable the mode of non-linear processing. By default, this feature is <i>enabled</i> .
<b>Fixed Window Size</b>	Sets the size of the window for the fixed cancellation.
<b>Moving Window Size</b>	Sets the size of the window for the moving cancellation.

### 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		
<input type="checkbox"/>	81	400xxxxxxx	default			
<input type="checkbox"/>	82	40[1-9]xxxxx	default			
<input type="checkbox"/>	83	4[1-9]xxxxxx	default			
<input type="checkbox"/>	84	800xxxxxxx	default			
<input type="checkbox"/>	85	80[1-9]xxxxx	default			
<input type="checkbox"/>	86	8[1-9]xxxxxx	default			
<input type="checkbox"/>	87	[2-3,5-7]xxxxxxx	default			
<input type="checkbox"/>	88	1[3-5,7-8]xxxxxxxx	default			
<input type="checkbox"/>	89	100xx	default			
<input type="checkbox"/>	90	95xxx	default			
<input type="checkbox"/>	91	123xx	default			
<input type="checkbox"/>	92	111xx	default			
<input type="checkbox"/>	93	11[0,2-9]	default			
<input type="checkbox"/>	94	120	default			
<input type="checkbox"/>	95	0[3-9]xxxxxxxxxxx	default			
<input type="checkbox"/>	96	02xxxxxxxxxxx	default			
<input type="checkbox"/>	97	010xxxxxxxxxx	default			
<input type="checkbox"/>	98	01[3-5,7-8]xxxxxxxx	default			
<input type="checkbox"/>	99	.	default			

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

Figure 3-30 Dialing Rule Configuration Interface (Standard)

See Figure 3-30 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. See Figure 3-31 for the dialing rule adding interface.

Dialing Rule

Index:  98 ▼

Description:

Dialing Rule:

Figure 3-31 Add New Dialing Rule

The table below explains the items shown in Figure 3-31.

Item	Description
<b>Index</b>	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.
<b>Description</b>	Remarks for the dialing rule. It can be any information, but can not be left empty.
<b>Dialing Rule</b>	Up to 99 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]xxxxxxxx.	Any 12-digit number starting with 013, 014, 015, 017 or 018
97	010xxxxxxxx	Any 11-digit number starting with 010
96	02xxxxxxxx	Any 11-digit number starting with 02
95	0[3-9]xxxxxxxx	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]xxxxxxxx	Any 11-digit number starting with 13, 14, 15, 17 or 18
87	[2-3,5-7]xxxxxx	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 3-30 to modify the dialing rules. See Figure 3-32 for the dialing rule modification interface. The configuration items on this interface are the same as those on the **Add New Dialing Rule** interface.

Figure 3-32 Modify Dialing Rule

To delete a dialing rule, check the checkbox before the corresponding index in Figure 3-30 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 3-30.

See Figure 3-33 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.

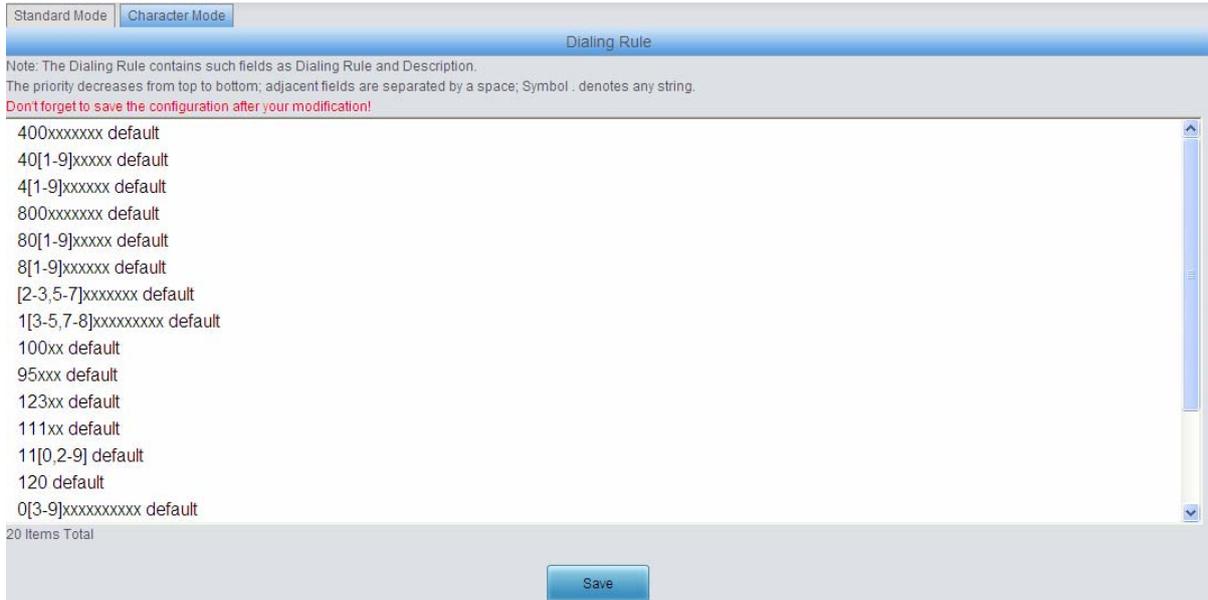


Figure 3-33 Dialing Rule Configuration Interface (Character)

### 3.5.5 Function Key

See Figure 3-34 for the function key configuration interface. Here you can set a cluster of combination keys to query or set the network port.



Figure 3-34 Function Key Configuration Interface

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

Figure 3-35 Cue Tone Interface

See Figure 3-35 for the Cue Tone interface. The table below explains the items shown in the above figure.

Item	Description
<b>Language</b>	Sets the language for the gateway to play voice, including two options Chinese and English. The default setting is <i>English</i> .
<b>Upload a file of cue tone</b>	Uploads a user-defined cue tone file to the gateway.
<b>Two Stage Dialing for PSTN Outgoing Calls Tips</b>	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 3-36 Color Ring Interface

By default, there is no available color ring on the gateway. See Figure 3-36. Click **Upload** to upload a new color ring manually. Follow Figure 3-37 to upload the required color ring file to the gateway.

Figure 3-37 Color Ring Upload Interface

The table below explains the items shown above:

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

After configuration, click **Upload** to upload the color ring file to the gateway or click **Return** to cancel the upload. See Figure 3-38 for the Color Ring Management interface after the upload.

Figure 3-38 Color Ring Management Interface

Click **Modify** in Figure 3-38 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 3-39 Color Ring Modification Interface

To delete a color ring, 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 color rings at a time, click the **Clear All** button in Figure 3-39.

### 3.5.8 QoS

Figure 3-40 Differentiated Services Setting Interface

See Figure 3-40 for the Differentiated Services setting interface. Using this technology, 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
<b>QoS</b>	Sets whether to enable the QoS differentiated services. By default, it is disabled.
<b>Media Premium QoS</b>	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.
<b>Control Premium QoS</b>	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

Tone Generator	
Tone Energy (dB)	<input type="text" value="0"/>
Dial Tone	<input type="text" value="450/1500"/>
Ringback Tone	<input type="text" value="450/1000,0/4000"/>
Busy Tone	<input type="text" value="450/350,0/350"/>
	<p>FreqA/TimeA,FreqB+FreqC/TimeB Repeatedly play tones in turn: first, TimeA, a single tone with FreqA, then, Time B, a dual tone composed of FreqB and FreqC.</p> <p>FreqA+FreqB+FreqC/TimeA,FreqD/TimeB Repeatedly play tones in turn: first, TimeA, a triple tone composed of FreqA, FreqB and FreqC, then, TimeB, a single tone with FreqD.</p> <p>Note: The play time is calculated by ms and cannot be larger than 16383ms for each toneunit. A tone is allowed to contain at most 5 different toneunits and 4 different frequencies, but the frequency and duration of the first toneunit cannot be 0. Frequency being 0 means the toneunit is a piece of silence.</p>
<input type="button" value="Save"/> <input type="button" value="Reset"/>	

Figure 3-41 Tone Generator Setting Interface

See Figure 3-41 for the Tone Generator Setting interface. 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 CDR Query

Figure 3-42 CDR Query Setting Interface

See Figure 3-42 for the CDR Query Setting interface. The table below explains the items shown in the above figure.

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

Click **Query** to query the CDR information corresponds to the above settings.

Figure 3-43 CDR Information Interface

**Note:** This page will appear only when the CDR feature is enabled (set in [3.5.2 System Param](#)).

### 3.5.11 VPN

Figure 3-44 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 3-44 for the VPN Settings interface. The table below gives the explanation to the items shown in the above figure.

Item	Description
<b>Enable OPENVPN</b>	Sets whether to enable the VPN feature, with the default value of <i>No</i> . 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 3-45.

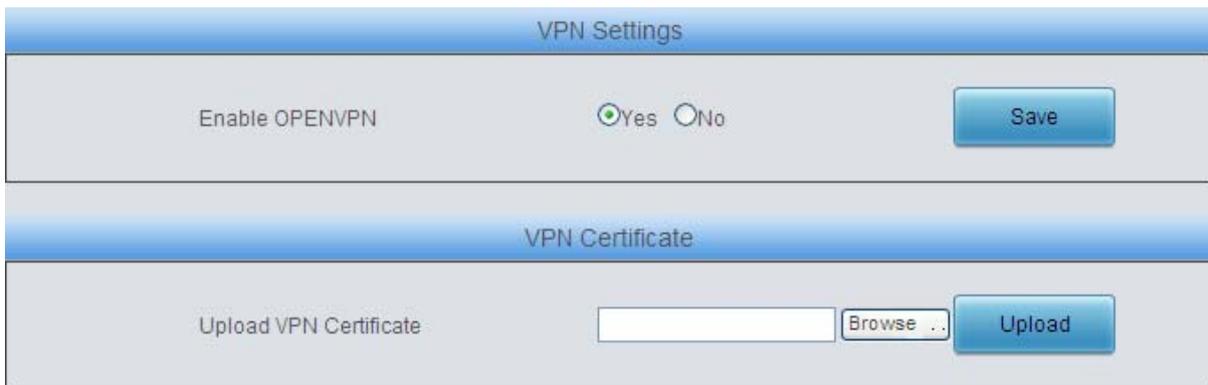


Figure 3-45 VPN Certificate Upload Interface

**Note:** Refer to [Appendix C VPN Certificate](#) for how to make a VPN certificate.

### 3.6 Wireless Settings

Wireless Settings includes ten parts: **Basic Param**, **Wireless Param**, **Call Forwarding**, **Short Message**, **IMEI**, **USSD**, **Email**, **Balance**, **SIM Card** and **PIN Manage**. See Figure 3-46.

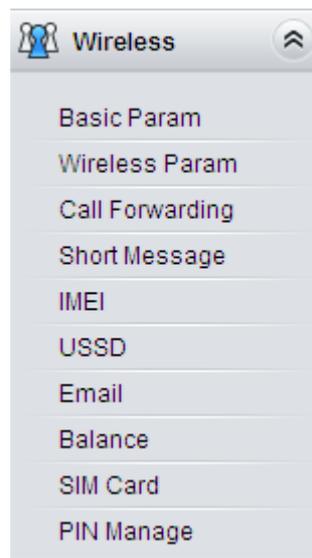


Figure 3-46 Wireless Settings

### 3.6.1 Basic Parameters

Basic Parameters		
Voice	GSM Voice Encoding	Automatic
DTMF	GSM DTMF Send Mode	Voice Playback
	GSM DTMF Receive Mode	Wireless Module Receive
	DTMF Voltage Detection for GSM	Off 0ms On 40ms
SMS	SMS Sending Interval(s)	1
	Maximum Pieces of Saved Logs	100
SIP Answer Code	Busy/Rejected	486
	No Answer	408
	Other Fault	480
		Save Reset

Figure 3-47 Basic Parameters Setting Interface for GSM

Basic Parameters		
Voice	WCDMA Voice Encoding	AMR
Network	Network Scan Mode	Automatic
	Network Scan Sequence	Automatic
DTMF	WCDMA DTMF Send Mode	Voice Playback
	WCDMA DTMF Receive Mode	Wireless Module Receive
SMS	SMS Sending Interval(s)	1
	Maximum Pieces of Saved Logs	100
SIP Answer Code	Busy/Rejected	486
	No Answer	408
	Other Fault	480
		Save Reset

Figure 3-48 Basic Parameters Setting Interface for WCDMA

Basic Parameters

DTMF	CDMA DTMF Send Mode <span style="float: right;">Voice Playback ▾</span> CDMA DTMF Receive Mode <span style="float: right;">Chip Receive ▾</span> Minimum Duration at ON <span style="float: right;">28 ms ▾</span>
SMS	SMS Sending Interval(s) <span style="float: right;">1</span> Maximum Pieces of Saved Logs <span style="float: right;">100</span>
Call Forwarding	Set/Cancel Service No. for FWD Unconditionally <span style="float: right;">*72</span> <span style="float: right;">*720</span> Set/Cancel Service No. for FWD on Busy <span style="float: right;">*90</span> <span style="float: right;">*900</span> Set/Cancel Service No. for FWD on No Reply <span style="float: right;">*92</span> <span style="float: right;">*920</span> Cancel All Service No. <span style="float: right;">*730</span> Cancel Service No. for Call Waiting <span style="float: right;">*740</span>
SIP Answer Code	Busy/Rejected <span style="float: right;">486</span> No Answer <span style="float: right;">408</span> Other Fault <span style="float: right;">480</span>

Save
Reset

Figure 3-49 Basic Parameters Setting Interface for CDMA

See Figure 3-47, Figure 3-48, Figure 3-49 for the basic parameters setting interface. The table below explains the items shown in the above figure.

Item	Description
<b>GSM (WCDMA) Voice Encoding</b>	Sets the mode of the GSM (WCDMA) voice encoding. By default, the voice encoding for GSM is <i>Automatic</i> and for WCDMA is <i>AMR</i> .
<b>GSM (WCDMA/CDMA) DTMF Send Mode</b>	Sets the mode to send the GSM (WCDMA/CDMA) DTMF, two options available: Voice Playback and Remote Transmission. The default value is <i>Voice Playback</i> .
<b>GSM (WCDMA/CDMA) DTMF Receive Mode</b>	Sets the mode to receive the GSM (WCDMA/CDMA) DTMF, two options available: Chip Receive and Wireless Module Receive. The default value for GSM and WCDMA is <i>Wireless Modulw Receive</i> ; The default value for CDMA is <i>Chip Receive</i> .
<b>Minimum Duration at ON</b>	The shortest time that a valid tone has to last at ON state, calculated by ms. The default value is 28. Note: This configuration item is only valid when the DTMF Receive Mode is set to Chip Receive.
<b>DTMF Voltage Detection for GSM</b>	Set the On and off of the DTMF detection for GSM.
<b>Network Scan Mode</b>	Sets a network for the call, three options available: Automatic, GSM Only and

	WCDMA Only. The default value is <i>Automatic</i> .
<b>Network Scan Sequence</b>	Sets the priority of the network, three options available: <i>Automatic</i> , <i>GSM prior to WCDMA</i> and <i>WCDMA prior to GSM</i> . The default value is <i>Automatic</i> .
<b>SMS Sending Interval</b>	Sets the interval to send SMS for each port. Range of value: 1~60, with the default value of 1.
<b>Maximum Pieces of Saved Logs</b>	Sets the amount of the logs to be saved for each port. Range of value: 50~500, with the default value of 100.
<b>SIP Answer Code</b>	Sets the sip answer code for each state of the calling party.
<b>Set/Cancel Service No. for FWD Unconditionally, Set/Cancel Service No. for FWD on Busy, Set/Cancel Service No. for FWD on No Reply</b>	Sets or Cancels the service No. for FWD unconditionally, FWD on busy or FWD on no reply. The former box is used to set the service No, while the latter one is to cancel the service No,.
<b>Cancel All Service No.</b>	Used to cancel all service numbers for FWD unconditional, FWD on busy and FWD on no reply.
<b>Cancel Service No. for Call Waiting</b>	Used to cancel the service number for call waiting.

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

### 3.6.2 Wireless Param

Wireless Param								
Check	Port	Cell Phone No.	IP->CDMA Voice Volume	CDMA->IP Voice Volume	IMSI	IMEI	Status	Modify
<input type="checkbox"/>	1	18143476793	1	2	460030764810073	805589A1	Enable	
<input type="checkbox"/>	2	---	1	2	---	---	Enable	
<input type="checkbox"/>	3	---	1	2	---	---	Enable	
<input type="checkbox"/>	4	---	1	2	---	---	Enable	
<input type="checkbox"/>	5	---	1	2	---	---	Enable	
<input type="checkbox"/>	6	---	1	2	---	---	Enable	
<input type="checkbox"/>	7	---	1	2	---	---	Enable	
<input type="checkbox"/>	8	---	1	2	---	---	Enable	

Figure 3-50 Wireless Parameters Configuration Interface

See Figure 3-50 for the Wireless Parameters Configuration interface. Click **Modify** in Figure 3-50 to modify the properties of the corresponding module. See Figure 3-51 for the Wireless Parameters Modification interface.

Figure 3-51 Wireless Parameters Modification Interface

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

Item	Description
<b>Port</b>	The number of the port corresponding to the wireless module.
<b>Cell Phone No.</b>	The number of the SIM card corresponding to the wireless module. This number should be configured manually.
<b>IP-&gt;GSM(WCDMA/CDMA) Voice Volume</b>	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.
<b>GSM(WCDMA/CDMA)-&gt;IP Voice Volume</b>	The volume of the voice from GSM/WCDMA/CDMA to IP. By default, the value for GSM is 40; the value for WCDMA is 3; the value for CDMA is 2.
<b>IMSI</b>	International Mobile Subscriber Identification Number, the unique identity of the SIM card.
<b>IMEI</b>	International Mobile Equipment Identity.
<b>Operator</b>	The operator of the wireless module. It is obtained automatically. This configuration is unavailable for CDMA module.
<b>Working Frequency Band</b>	Displays the working frequency band of the wireless module. This configuration is unavailable for CDMA module.
<b>Status</b>	Displays the current state of the wireless module.
<b>Apply to all the modules</b>	Sets whether to apply all the settings except for the cell phone number to all the modules.

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

### 3.6.3 Call Forwarding

Call Forwarding								
Check	Port	FWD Unconditionally	FWD on Busy	FWD on No Reply	FWD on Unreachable	FWD Setting Status	FWD Query Status	Modify
<input type="checkbox"/>	1	Close	Close	Close	+8613800571176	---	Successful	
<input type="checkbox"/>	2	Close	Close	Close	+8613800571176	---	Successful	
<input type="checkbox"/>	3	Close	Close	Close	+8613800571176	---	Successful	
<input type="checkbox"/>	4	---	---	---	---	---	---	--
<input type="checkbox"/>	5	---	---	---	---	---	---	--
<input type="checkbox"/>	6	---	---	---	---	---	---	--
<input type="checkbox"/>	7	---	---	---	---	---	---	--
<input type="checkbox"/>	8	---	---	---	---	---	---	--

Check All   Uncheck All   Query

Figure 3-52 Call Forwarding Configuration Interface

See Figure 3-52 for the Call Forwarding Configuration interface. The table below explains the items shown in the above figure.

Item	Description
<b>Port</b>	The number of the port corresponding to the wireless module.
<b>FWD Unconditionally</b>	Sets whether to enable the feature of FWD unconditionally and the FWD number if it is enabled.
<b>FWD on Busy</b>	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.
<b>FWD on No Reply</b>	Sets whether to enable the feature of FWD on no reply and the FWD number if it is enabled.
<b>FWD on Unreachable</b>	Sets whether to enable the feature of FWD on unreachable and the FWD number if it is enabled. This configuration is unavailable for CDMA module.
<b>FWD Setting Status</b>	Displays the setting status of the call forwarding service.
<b>FWD Query Status</b>	Displays the query status of the FWD settings. This configuration is unavailable for CDMA module.
<b>Cancel All</b>	Cancels all the setting on call FWD service. This item will appear if none of the call FWD is selected.

Click **Modify** in Figure 3-52 to modify the properties of the corresponding port. See Figure 3-53 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 3-53 Wireless Service Modification Interface

### 3.6.4 Short Message

Short Message						
Check	Port	Cell Phone No.	SMS Center	SMS Receiving Details	SMS Sending Details	Send SMS
<input type="checkbox"/>	1	15990156537	<a href="#">8613800571500</a>	N:5	N:56	
<input type="checkbox"/>	2	15990150207	<a href="#">8613800571500</a>	N:0	N:3	
<input type="checkbox"/>	3	--	--	N:0	N:0	--
<input type="checkbox"/>	4	--	--	N:0	N:0	--
<input type="checkbox"/>	5	--	--	N:0	N:0	--
<input type="checkbox"/>	6	--	--	N:0	N:0	--
<input type="checkbox"/>	7	--	--	N:0	N:0	--
<input type="checkbox"/>	8	--	--	N:0	N:0	--

Check All   Uncheck All   Clear All

Figure 3-54 Short Message Interface

See Figure 3-54 for the Short Message interface which displays the related information about the received/sent SMS.

Click **SMS Center** to go into the SMS Center Modification interface. See Figure 3-55. Click **Save** to save the settings into the gateway, click **Close** to cancel the settings.

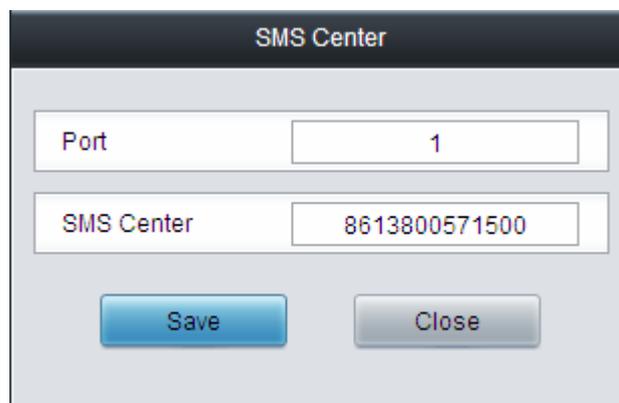


Figure 3-55 SMS Center Modification Interface

Click **SMS Receiver Details** in Figure 3-54 to go into the SMS Receiver Details interface. See Figure 3-56. Such information as the remote cell phone number, the time and the content will be displayed on this page.

Check	No.	Port	Receive/Send	Remote Phone Number	Time	Content
<input type="checkbox"/>	1	1	Receive	10010	2015-10-15 16:24:50	82.79
<input type="checkbox"/>	2	1	Receive	10010	2015-10-15 16:24:56	<a href="#">82.79</a>
<input type="checkbox"/>	3	1	Receive	10010	2015-10-15 16:28:38	82.79
<input type="checkbox"/>	4	1	Receive	10010	2015-10-15 16:31:47	<a href="#">82.79</a>
<input type="checkbox"/>	5	1	Receive	8618668137917	2015-10-19 14:31:56	<a href="#">16</a>

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

Figure 3-56 SMS Receiving Details Interface

To delete a piece of SMS receiving detail, check the checkbox before the corresponding index in Figure 3-56 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; to clear all SMS receiver details at a time, click the **Clear All** button in Figure 3-56; to go back to the previous page, click **Return**.

Click **Records** in Figure 3-54 to go into the SMS Sending interface. See Figure 3-57. Such information as the receive/send status of the SMS, the remote cell phone number, the time, and the content will be displayed on this page.

Check	No.	Port	Receive/Send	Remote Phone Number	Time	Content	Result	From
<input type="checkbox"/>	1	1	Send	135167742561	2015-10-15 09:48:50	<a href="#">coolman</a>	Successful	WEB
<input type="checkbox"/>	2	1	Send	13516774256	2015-10-15 09:48:58	<a href="#">coolman</a>	Successful	WEB
<input type="checkbox"/>	3	1	Send	10010	2015-10-15 16:22:59	<a href="#">102</a>	Successful	WEB
<input type="checkbox"/>	4	1	Send	10010	2015-10-15 16:23:04	<a href="#">101</a>	Successful	WEB
<input type="checkbox"/>	5	1	Send	13516774256	2015-10-15 16:24:27	<a href="#">123456</a>	Successful	WEB
<input type="checkbox"/>	6	1	Send	13516774256	2015-10-15 16:24:33	<a href="#">123456</a>	Successful	WEB
<input type="checkbox"/>	7	1	Send	13516774256	2015-10-15 16:24:38	<a href="#">123456</a>	Successful	WEB
<input type="checkbox"/>	8	1	Send	13516774256	2015-10-15 16:24:42	<a href="#">123456</a>	Successful	WEB
<input type="checkbox"/>	9	1	Send	13516774256	2015-10-15 16:24:47	<a href="#">123456</a>	Successful	WEB
<input type="checkbox"/>	10	1	Send	13516774256	2015-10-15 16:24:52	<a href="#">1111111</a>	Successful	WEB
<input type="checkbox"/>	11	1	Send	13516774256	2015-10-15 16:24:56	<a href="#">1111111</a>	Successful	WEB
<input type="checkbox"/>	12	1	Send	13516774256	2015-10-15 16:25:01	<a href="#">1111111</a>	Successful	WEB
<input type="checkbox"/>	13	1	Send	13516774256	2015-10-15 16:25:09	<a href="#">1111111</a>	Successful	WEB
<input type="checkbox"/>	14	1	Send	13516774256	2015-10-15 16:25:15	<a href="#">1111111</a>	Successful	WEB
<input type="checkbox"/>	15	1	Send	13516774256	2015-10-15 16:25:19	<a href="#">123456</a>	Successful	WEB
<input type="checkbox"/>	16	1	Send	13516774256	2015-10-15 16:25:24	<a href="#">123456</a>	Successful	WEB
<input type="checkbox"/>	17	1	Send	13516774256	2015-10-15 16:25:30	<a href="#">123456</a>	Successful	WEB
<input type="checkbox"/>	18	1	Send	13516774256	2015-10-15 16:25:35	<a href="#">123456</a>	Successful	WEB
<input type="checkbox"/>	19	1	Send	13516774256	2015-10-15 16:25:40	<a href="#">123456</a>	Successful	WEB
<input type="checkbox"/>	20	1	Send	13516774256	2015-10-15 16:25:44	<a href="#">123456</a>	Successful	WEB

56 Items Total 20 Items/Page 1/3 First Previous [Next](#) [Last](#) Go to Page 1  3 Pages Total

Figure 3-57 SMS Sending Interface

To delete a piece of record, check the checkbox before the corresponding index in Figure 3-57 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; to clear all records at a time, click the **Clear All** button in Figure 3-56; to go back to the previous page, click **Return**.

Click **Send SMS** in Figure 3-54 to go into the Send SMS interface. See Figure 3-58.

Send SMS

Port 
 1
  2
  3
  4
  5
  6
  7
  8

Number Import

Send to  (Separated by ',')

Encoding Format

Content

Note: 1.SMS can be sent to 50 numbers at most.  
 2.Number file must be \*.txt,number separated by ',' or 'enter'.  
 3.The length of SMS cannot exceed 600 characters.

	Time	Port	Number	Result
Result				

Figure 3-58 Send SMS Interface

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

Item	Description
<b>Port</b>	Select a port to send the SMS.
<b>Number Import</b>	Click <i>Browse</i> to select the required number file and then click <i>Import</i> to import this file.
<b>Send to</b>	Enter the remote number to receive the SMS.
<b>Encoding Format</b>	The encoding format for the SMS, two options available: GSM 7bit and UCS2.
<b>Content</b>	The content of the SMS required to be sent.
<b>Result</b>	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 IMEI



IMEI

IMEI Modification Service Agreement

Welcome to use the IMEI modification service. By using this service you accept all the following terms.

(1) Used for test only  
This service is only provided to individuals for test.

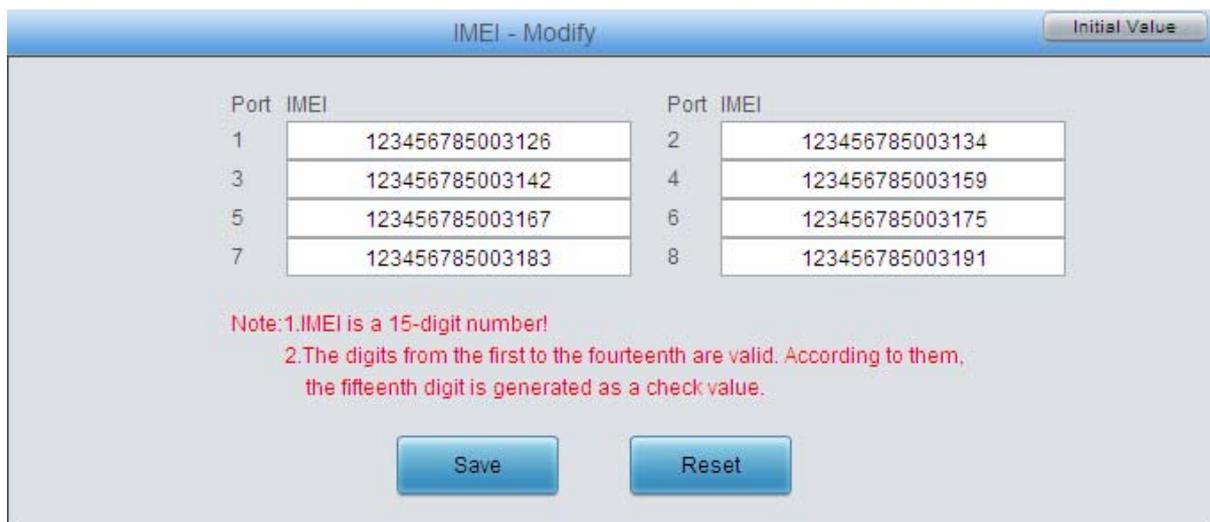
(2) Can not be used for any commercial purposes  
You should use this service under the premise of not violating any laws or regulations.

(3) Exemption  
You are liable for any possible losses in your use of this service,  
Our company will take no legal responsibility for it.

Accept

Figure 3-59 IMEI Interface

See Figure 3-59 for the IMEI interface. Read the agreement carefully and click **Accept** before you go into the IMEI Modification interface. There are two optional modes for IMEI modification: Manual Modify and Auto Modify. Click Manual Modify to go into the IMEI manual modification interface (Figure 3-60).



IMEI - Modify Initial Value

Port	IMEI	Port	IMEI
1	123456785003126	2	123456785003134
3	123456785003142	4	123456785003159
5	123456785003167	6	123456785003175
7	123456785003183	8	123456785003191

Note: 1. IMEI is a 15-digit number!  
2. The digits from the first to the fourteenth are valid. According to them, the fifteenth digit is generated as a check value.

Figure 3-60 IMEI Manual Modification Interface

The default IMEI information will be displayed after clicking Initial Value in Figure 3-60, you can save and use it according to your requirement.

Click Auto Modify to go into the IMEI auto modification interface (Figure 3-61).

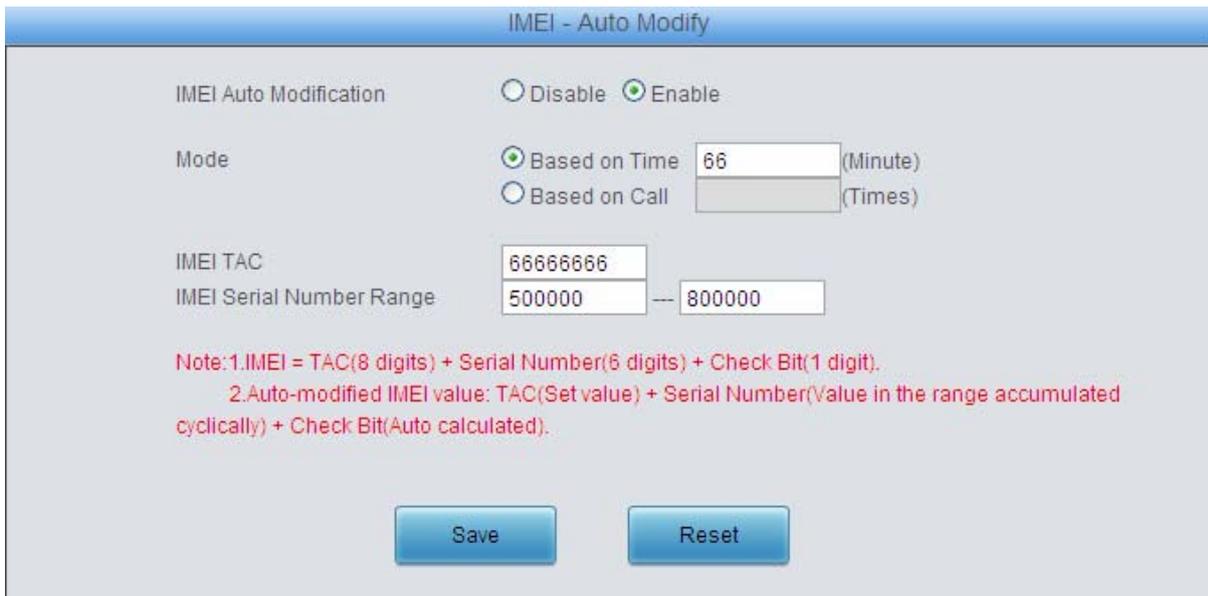


Figure 3-61 IMEI Auto Modification Interface

After configuration, click **Save** to save the above settings into the gateway or click **Reset** to restore the configurations.

**Note:** This configuration is unavailable for CDMA module.

### 3.6.6 USSD

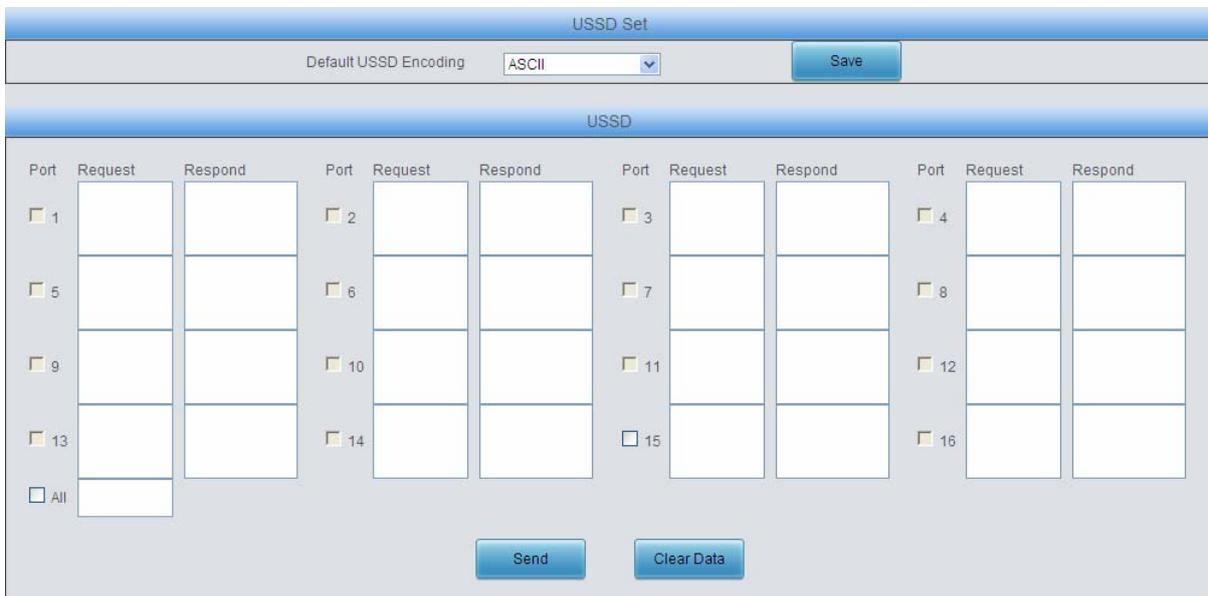


Figure 3-62 USSD Setting Interface

See Figure 3-62 for the USSD Setting interface. The table below explains the items shown in the above figure.

Item	Description
<b>Default USSD Encoding</b>	Sets the default encoding format for USSD, two options available: ASCII and UCS2.
<b>Port</b>	Sets the port used to send the USSD request.
<b>Request</b>	Inputs the content of the USSD request.
<b>Respond</b>	Displays the result of the USSD respond.

<b>All</b>	Selects all the available ports to send the same USSD request.
------------	--

Click **Send** in Figure 3-62 to send out the USSD request. Click **Clear Data** to clear all data.

**Note:** This configuration is unavailable for CDMA module.

### 3.6.7 Email

The screenshot shows the 'Email Config' interface. It is divided into two main sections: 'Mailbox Settings' and 'Conversion between Email & SMS'.  
**Mailbox Settings:**  
 - Mailbox Account: husidongtest@sanhuid.com  
 - Password: [masked]  
 - Outgoing(SMTP): 201.123.116.240, Port: 25, with a 'Send test' button.  
 - Incoming(POP3): 201.123.116.240, Port: 110, with a 'Receive test' button.  
**Conversion between Email & SMS:**  
 - A 'Show Log' button is present.  
 - 'Convert SMS to Email' is checked (Enable). Target Address is empty, Subject is 'SMStoEmail'.  
 - 'Convert Email to SMS' is checked (Enable). Receiving Cycle is '1' Minute (Range: 1~60), Subject is 'EmailtoSMS', and SMS Sending Port is 'Automatic'.  
 - 'Return Receipt' is checked for both 'Successful' and 'Failed'.  
 - A red note at the bottom provides details on supported formats and conversion rules.  
 - 'Save' and 'Reset' buttons are at the bottom.

Figure 3-63 Email Setting Interface

See Figure 3-63 for the Email Setting interface. The table below explains the configuration items on the Email Setting interface.

Item	Description
<b>Mailbox Account, Password</b>	Sets the account and password of the mailbox.
<b>Outgoing (SMTP), Port</b>	Sets the server address and port for Email sending.
<b>Incoming (POP3), Port</b>	Sets the server address and port for Email receiving.
<b>Show Log</b>	Click it to display the log which contains the Email to SMS converted information.
<b>Convert SMS to Email</b>	SMS can be converted to Emails if this feature is enabled.

<b>Target Address</b>	The target address to which the Email converted by SMS will be sent.
<b>Subject</b>	Sets the subject for the Email converted by SMS.
<b>Covert Email to SMS</b>	When this feature is enabled, the mails in a designated format (See Note 4 and 5 in Figure 3-63) can be converted to SMS.
<b>Receiving Cycle</b>	Sets the cycle to receive mails. Range of value: 1~60, calculated by minute, with the default value of 5.
<b>SMS Sending Port</b>	Sets the port from which the SMS will be sent out. The default value is <i>automatic</i> .
<b>Return Receipt</b>	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.

### 3.6.8 Balance

Balance Query					
Check	Port	Cell Phone No.	Time	Balance	Modify
<input type="checkbox"/>	1	13023634185	---	---	
<input type="checkbox"/>	2	13082814738	---	---	
<input type="checkbox"/>	3	15990152395	2016-03-16 14:29:17	20.52	
<input type="checkbox"/>	4	15990150759	2016-03-16 14:29:18	82.19	
<input type="checkbox"/>	5	15990150207	2016-03-16 14:29:19	96.15	
<input type="checkbox"/>	6	15990119352	---	---	
<input type="checkbox"/>	7	---	---	---	---
<input type="checkbox"/>	8	---	---	---	---

Check All   Uncheck All   Query   Refresh

Figure 3-64 Balance Query Interface

See Figure 3-64 for the Balance Query interface. You can query the balance for a designated cell phone number. Click Modify in Figure 3-64 to modify the query mode. See Figure 3-65.

Modify Query Mode

Port	<input type="text" value="1"/>
Query Mode	<input type="text" value="SMS"/>
Destination Number	<input type="text" value="10086"/>
Content to Send	<input type="text" value="11"/>
Keywords to Match	<input type="text" value="Blance"/>
Query after SIM Card Registered	<input type="text" value="No"/>
Query Regularly	<input type="text" value="60"/> (Minute,0:disabled)
Apply to Other Ports	<input checked="" type="radio"/> Port <input type="radio"/> Port Group <input checked="" type="checkbox"/> 1 <input type="checkbox"/> 2 <input type="checkbox"/> 3 <input type="checkbox"/> 4 <input type="checkbox"/> 5 <input type="checkbox"/> 6 <input type="checkbox"/> 7 <input type="checkbox"/> 8

Figure 3-65 Query Mode Modification Interface

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

Item	Description
<b>Query Mode</b>	Sets the mode to query the balance.
<b>Destination Number</b>	Sets the destination number to query the balance
<b>Content to Send</b>	Sets the content to query the balance.
<b>Keywords to Match</b>	The balance matching the keywords will be displayed.
<b>Query after SIM Card Registered</b>	Sets whether to query the balance automatically once the SIM card is registered to the base station.
<b>Query Regularly</b>	Sets the time to query the balance regularly.
<b>Apply to Other Ports</b>	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.

### 3.6.9 SIM Card

SIM Card List								
Port	Card A	Card B	Card C	Card D	Mobile Phone Number	Auto Switch to Available SIM Card	Switch Strategy for SIM Card	Modify
1	Using	Exist	Empty	Empty	13750845226	Enable	Disable	
2	Empty	Empty	Empty	Empty	---	Enable	Disable	
3	Empty	Empty	Empty	Empty	---	Enable	Disable	
4	Empty	Empty	Empty	Empty	---	Enable	Disable	
5	Empty	Empty	Empty	Empty	---	Enable	Disable	
6	Empty	Empty	Empty	Empty	---	Enable	Disable	
7	Empty	Empty	Empty	Empty	---	Enable	Disable	
8	Empty	Empty	Empty	Empty	---	Enable	Disable	
9	Empty	Empty	Empty	Empty	---	Enable	Disable	
10	Empty	Empty	Empty	Empty	---	Enable	Disable	
11	Empty	Empty	Empty	Empty	---	Enable	Disable	
12	Empty	Empty	Empty	Empty	---	Enable	Disable	
13	Empty	Empty	Empty	Empty	---	Enable	Disable	
14	Empty	Empty	Empty	Empty	---	Enable	Disable	
15	Empty	Empty	Empty	Empty	---	Enable	Disable	
16	Empty	Empty	Empty	Empty	---	Enable	Disable	

Figure 3-66 SIM Card List Interface

See Figure 3-66 for the SIM Card List interface, which 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. See Figure 3-67.



Figure 3-67 SIM Card Management Interface

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

Item	Description
<b>Port</b>	Serial number of the port on the device.
<b>Auto Switch to Available SIM Card</b>	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 <i>enable</i> .
<b>Switch Strategy for SIM Card</b>	Sets the switch strategy for the SIM card. There are four options: Based on Time, Based on Call, Fixed Time and Disable, with the default value of <i>Disable</i> .
<b>Apply to All Ports</b>	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.

### 3.6.10 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 3-68 PIN Manage Interface

See Figure 3-68 for the PIN Manage interface, which display the status of the SIM card and the setting status of PIN and PUK. Click Modify to go into the modification interface. See Figure 3-69.

Note: There is a restriction on the number of input times of PIN and PUK. Please proceed with caution.

Figure 3-69 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 3-70.

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 3-70 SIM Card Locked PIN Required

Click Modify in Figure 3-70, you are required to input PIN again, see Figure 3-71.

Note: There is a restriction on the number of input times of PIN and PUK. Please proceed with caution.

Figure 3-71 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 3-72.

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 3-72 SIM Card Locked Do not Require PIN

Click Modify in Figure 3-72 to unlock the SIM card or modify the PIN, see below figure.

Figure 3-73 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 3-74.

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 3-74 SIM Card Locked Need PIN and PUK

Click Modify in Figure 3-74 to input PUK and reset a new PIN, see Figure 3-75.

Figure 3-75 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.

### 3.7 Port Settings

Port Settings includes two parts: **Port** and **Port Group**. See Figure 3-76.

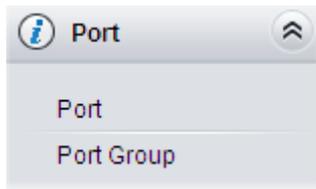


Figure 3-76 Port Settings

### 3.7.1 Port

Port Settings													Batch Modify
Port	Type	SIP Account	Authentication Username	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Cancellor	Color Ring	Color Ring Index	Server Index	Modify
1	GSM	8001	---	Static Binding	180	Disable	Disable	Failed	Enable	Disable	---	---	
2	GSM	182	---	Static Binding	8003	Disable	Disable	Unregistered	Enable	Disable	---	---	
3	GSM	8003	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Disable	---	---	
4	GSM	8004	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Disable	---	---	
5	GSM	8005	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Disable	---	---	
6	GSM	8006	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Disable	---	---	
7	GSM	8007	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Disable	---	---	
8	GSM	8008	---	Two Stage Dialing Mode	---	Disable	Disable	Unregistered	Enable	Disable	---	---	

Figure 3-77 Port Settings Interface

See Figure 3-77 for the Port Settings interface. The list in the above figure shows the feature and properties of each port. Click **Modify** in Figure 3-77 to modify the properties of the corresponding port. See Figure 3-78 for the Port Modification interface.

Port-Modify

Port	<input type="text" value="1"/>
Register Port	<input type="text" value="Yes"/>
SIP Account	<input type="text" value="8001"/>
Password	<input type="password" value="••••"/>
Authentication Username	<input type="text"/>
Connection Method	<input type="text" value="Static Binding(SIP A)"/>
Bound Number	<input type="text" value="180"/>
Echo Cancellor	<input checked="" type="checkbox"/> Enable
Forbid Outgoing Call	<input type="checkbox"/> Enable
Caller ID Detection	<input type="checkbox"/> Enable
Color Ring	<input checked="" type="checkbox"/> Enable
Color Ring Index	<input type="text" value="1"/>

Figure 3-78 Port Modification

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

Item	Description
<b>Port</b>	Serial number of the port on the device.

<b>Register Port</b>	<p>Sets whether to register the port to the SIP server.</p> <p>When this item is set to <i>No</i>, the item <b>Reg Status</b> on the Port Settings interface (Figure 3-77) shows <i>Unregistered</i>; when this item is set to <i>Yes</i>, the item <b>Reg Status</b> shows <i>Failed</i> or <i>Registered</i>.</p>						
<b>SIP Account</b>	<p>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.</p>						
<b>Password</b>	<p>Registration password of the port. To register a port to the SIP server, both items <b>SIP Account</b> and <b>Password</b> must be filled in.</p>						
<b>Authentication Username</b>	<p>Authentication username of a port, used to register the port to the SIP server when IMS network is enabled.</p> <p><b>Note: This item appears only when IMS Network is enabled.</b></p>						
<b>Connection Method</b>	<p>Port connection methods include:</p> <table border="1" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: center;">Option</th> <th style="text-align: center;">Description</th> </tr> </thead> <tbody> <tr> <td style="vertical-align: top;"><i>Static Binding (SIP Account)</i></td> <td>Bind the number to a wireless port. The number will be listed in the Bound Number column.</td> </tr> <tr> <td style="vertical-align: top;"><i>Two Stages Dialing Mode (default)</i></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> </tbody> </table> <p><b>Note:</b> Both items Connection Method and Bound Number will be hidden if the SIP Station feature is enabled on the SIP Settings interface.</p>	Option	Description	<i>Static Binding (SIP Account)</i>	Bind the number to a wireless port. The number will be listed in the Bound Number column.	<i>Two Stages Dialing Mode (default)</i>	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.
Option	Description						
<i>Static Binding (SIP Account)</i>	Bind the number to a wireless port. The number will be listed in the Bound Number column.						
<i>Two Stages Dialing Mode (default)</i>	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.						
<b>Echo Canceller</b>	<p>The echo cancellation feature for a call conversation over the wireless channel. By default, this feature is enabled and the effect can reach 128ms.</p>						
<b>Forbid Outgoing Call</b>	<p>If this feature is enabled, the port will be forbidden to call out. The default setting is <i>disabled</i>.</p>						
<b>Caller ID Detection</b>	<p>If this feature is enabled, the port will detect the Caller IDs from the incoming calls. The default setting is <i>disabled</i>.</p>						
<b>Color Ring</b>	<p>Sets whether to enable the color ring feature or not, with the default setting of being <i>disabled</i>.</p> <p><b>Note:</b> Only when there are available color rings will this item appear.</p>						
<b>Color Ring Index</b>	<p>The index of the color ring which is quoted by the current wireless port.</p>						

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** to modify several pieces of port settings at the same time. See Figure 3-79 below for the Port Batch Modification interface.

Port-Batch Modify

Starting Port	<input type="text" value="1"/>
Ending Port	<input type="text" value="8"/>
Register Port	<input type="text" value="Yes"/>
Starting SIP Account	<input type="text"/>
Starting Authentication Password	<input type="text"/>
Starting Authentication Username	<input type="text"/>
SIP Account Batch Rule	<input type="text" value="Increase"/>
SIP Account Batch Step Size	<input type="text" value="1"/>
Authentication Password Batch Rule	<input type="text" value="Increase"/>
Authentication Password Batch Step Size	<input type="text" value="1"/>
Authentication Username Batch Rule	<input type="text" value="Increase"/>
Authentication Username Batch Step Size	<input type="text" value="1"/>
Connection Method	<input type="text" value="Static Binding(SIP A)"/>
Bound Number	<input type="text"/>
Echo Canceller	<input checked="" type="checkbox"/> Enable
Forbid Outgoing Call	<input type="checkbox"/> Enable
Caller ID Detection	<input type="checkbox"/> Enable
Color Ring	<input checked="" type="checkbox"/> Enable
Color Ring Index	<input type="text" value="1"/>

Figure 3-79 Port Batch Modification

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

Item	Description
<b>Starting Port</b>	The starting serial number of the port on the device in the batch setting.
<b>Ending Port</b>	The ending serial number of the port on the device in the batch setting.
<b>Starting SIP Account</b>	The starting SIP account in the batch setting.
<b>Starting Authentication Password</b>	The starting authentication password in the batch setting.
<b>Starting Authentication Username</b>	The starting authentication username in the batch setting.
<b>SIP Account Batch Rule</b>	The rule for batch setting the SIP account, including <b>Increase</b> and <b>Decrease</b> two options.
<b>SIP Account Batch Step Size</b>	Sets the increase or decrease step size of the SIP account in the batch setting.

<b>Authentication Password Batch Rule</b>	The rule for batch setting the authentication password, including <b>Increase</b> and <b>Decrease</b> two options.
<b>Authentication Password Batch Step Size</b>	Sets the increase or decrease step size of the authentication password in the batch setting.
<b>Authentication Username Batch Rule</b>	The rule for batch setting the authentication username, including <b>Increase</b> and <b>Decrease</b> two options.
<b>Authentication Username Batch Step Size</b>	Sets the increase or decrease step size of the authentication username in the batch setting.

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

### 3.7.2 Port Group

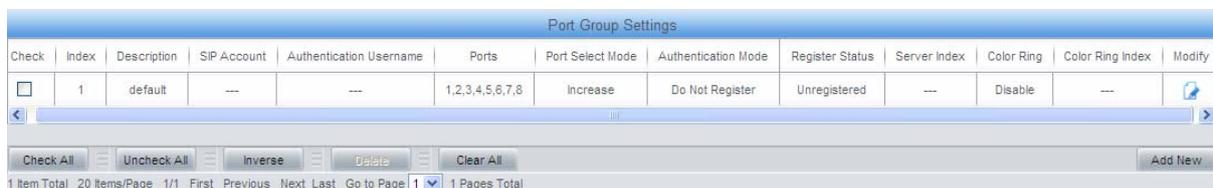


Figure 3-80 Port Group Settings Interface

See Figure 3-80 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. See Figure 3-81 for the port group adding interface. Note that a port which has been occupied by one port group cannot be chosen by others.

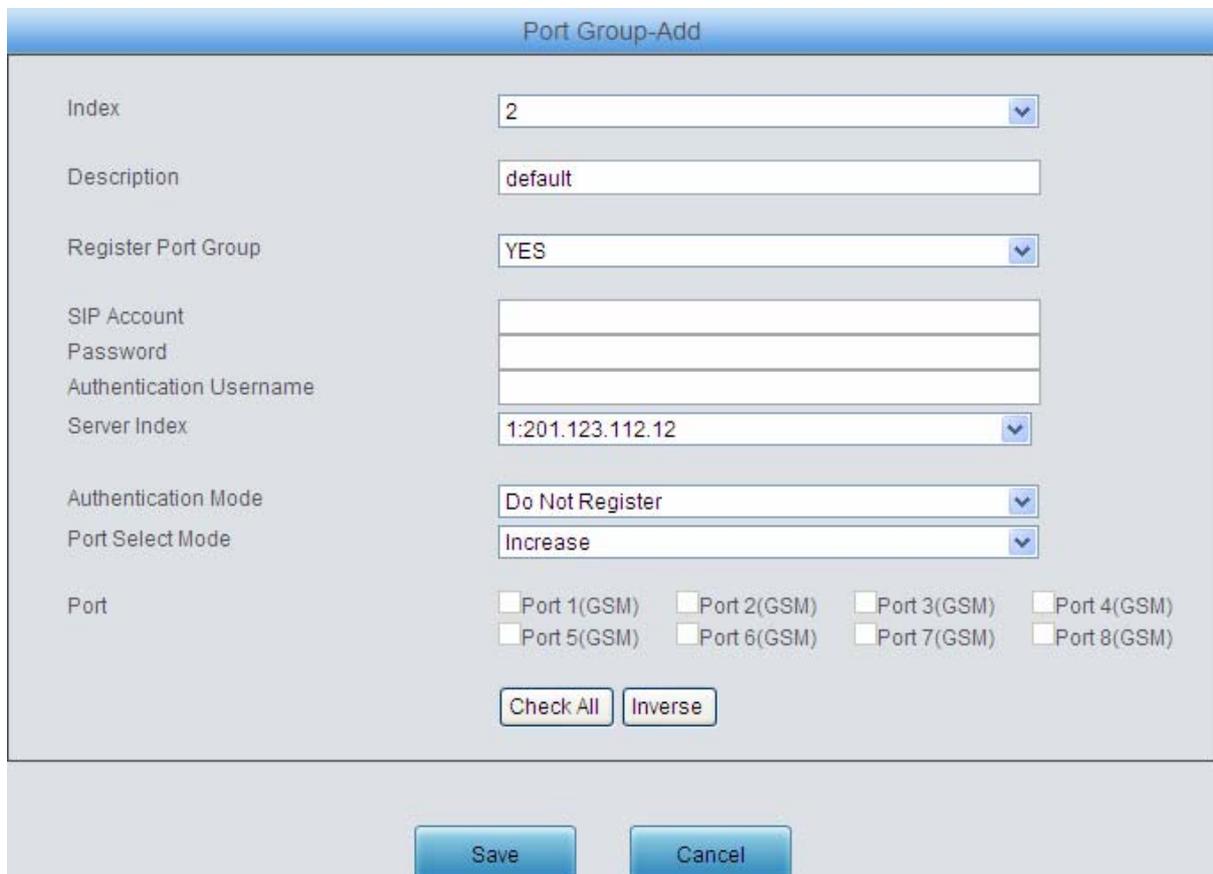


Figure 3-81 Add New Port Group

The table below explains the items in the above figure.

Item	Description												
<b>Index</b>	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.												
<b>Description</b>	More information about each port group, with default value of <i>default</i> .												
<b>Register Port Group</b>	To register the port group to the SIP server. Only when this configuration item is set to Yes can you see the configuration items <b>SIP Account</b> and <b>Password</b> .												
<b>SIP Account</b>	When the port group initiates a call to SIP, this item corresponds to the username of SIP.												
<b>Password</b>	Registration password of the port group. To register the port group to the SIP server, both configuration items <b>SIP Account</b> and <b>Password</b> should be filled in.												
<b>Authentication Username</b>	Authentication username of a port, used to register the port to the SIP server when IMS network is enabled. <b>Note: This item appears only when IMS Network is enabled.</b>												
<b>Server Index</b>	The index of the sip server which will be quoted by the current port.												
<b>Authentication Mode</b>	<p>Sets the way for SIP to make outgoing calls (Tel→IP) on the gateway.</p> <table border="1" data-bbox="485 931 1374 1406"> <thead> <tr> <th data-bbox="485 931 730 976">Option</th> <th data-bbox="730 931 1374 976">Description</th> </tr> </thead> <tbody> <tr> <td data-bbox="485 976 730 1061"><i>Do Not Register (default)</i></td> <td data-bbox="730 976 1374 1061">SIP initiates a call in a point-to-point mode.</td> </tr> <tr> <td data-bbox="485 1061 730 1191"><i>Register Gateway</i></td> <td data-bbox="730 1061 1374 1191">SIP initiates a call with the registered SIP account and password of the whole gateway. (Refer to <a href="#">3.4.1 SIP</a> for gateway registration.)</td> </tr> <tr> <td data-bbox="485 1191 730 1276"><i>Register Port Group</i></td> <td data-bbox="730 1191 1374 1276">SIP initiates a call with the registered SIP account and password of the port group.</td> </tr> <tr> <td data-bbox="485 1276 730 1361"><i>Register Port</i></td> <td data-bbox="730 1276 1374 1361">SIP initiates a call with the registered SIP account and password of the port.</td> </tr> <tr> <td data-bbox="485 1361 730 1406"><i>Group Ringing</i></td> <td data-bbox="730 1361 1374 1406">Ring all the idle wireless ports in this port group.</td> </tr> </tbody> </table>	Option	Description	<i>Do Not Register (default)</i>	SIP initiates a call in a point-to-point mode.	<i>Register Gateway</i>	SIP initiates a call with the registered SIP account and password of the whole gateway. (Refer to <a href="#">3.4.1 SIP</a> for gateway registration.)	<i>Register Port Group</i>	SIP initiates a call with the registered SIP account and password of the port group.	<i>Register Port</i>	SIP initiates a call with the registered SIP account and password of the port.	<i>Group Ringing</i>	Ring all the idle wireless ports in this port group.
Option	Description												
<i>Do Not Register (default)</i>	SIP initiates a call in a point-to-point mode.												
<i>Register Gateway</i>	SIP initiates a call with the registered SIP account and password of the whole gateway. (Refer to <a href="#">3.4.1 SIP</a> for gateway registration.)												
<i>Register Port Group</i>	SIP initiates a call with the registered SIP account and password of the port group.												
<i>Register Port</i>	SIP initiates a call with the registered SIP account and password of the port.												
<i>Group Ringing</i>	Ring all the idle wireless ports in this port group.												
<b>Register Status</b>	Registration status of the port group. See Figure 3-80. When <b>Register Port Group</b> is set to No, the value of this item is <i>Unregistered</i> ; when <b>Register Port Group</b> is set to Yes, the value of this item may be <i>Failed</i> or <i>Registered</i> .												

<p><b>Port Select Mode</b></p>	<p>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.</p>										
	<table border="1"> <thead> <tr> <th data-bbox="486 327 730 365">Option</th> <th data-bbox="730 327 1375 365">Description</th> </tr> </thead> <tbody> <tr> <td data-bbox="486 365 730 539"><i>Increase (default)</i></td> <td data-bbox="730 365 1375 539">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 data-bbox="486 539 730 714"><i>Decrease</i></td> <td data-bbox="730 539 1375 714">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 data-bbox="486 714 730 920"><i>Cyclic Increase</i></td> <td data-bbox="730 714 1375 920">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 data-bbox="486 920 730 1131"><i>Cyclic Decrease</i></td> <td data-bbox="730 920 1375 1131">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> </tbody> </table>	Option	Description	<i>Increase (default)</i>	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.	<i>Decrease</i>	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.	<i>Cyclic Increase</i>	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.	<i>Cyclic Decrease</i>	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.
	Option	Description									
	<i>Increase (default)</i>	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.									
	<i>Decrease</i>	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.									
<i>Cyclic Increase</i>	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.										
<i>Cyclic Decrease</i>	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.										
<p><b>Port</b></p>	<p>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 <b>Ports</b> column in Figure 3-80. Note: When a port group contains multiple ports, the automatic call forward feature is invalid.</p>										

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** at the end of the list in **Port Group Settings Interface** to modify the properties of a port group. See Figure 3-82 for the Port Group Modification interface. The configuration items on this interface are the same as those on the **Add New Port Group** interface.

Figure 3-82 Modify Port Group

To delete a port group, check the checkbox before the corresponding index in Figure 3-80 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 3-80.

### 3.8 Route Settings

Route Settings is used to specify the routing rules for calls on two directions: IP→Tel/IP and Tel→IP. See Figure 3-83.

Figure 3-83 Route Settings

#### 3.8.1 Routing Parameters

Figure 3-84 Routing Parameters Configuration Interface

See Figure 3-84 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.

### 3.8.2 IP to Tel/IP

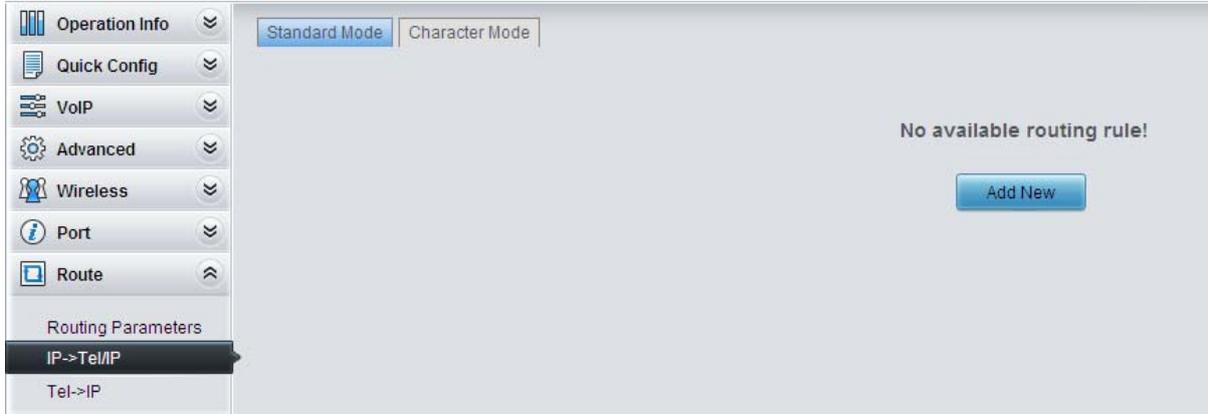


Figure 3-85 IP→Tel/IP Routing Rule Configuration Interface (Standard)

See Figure 3-85 for the IP→Tel/IP routing rule configuration interface. By default, there is no available routing rule on the gateway. The IP→Tel/IP routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click **Add New** to add them manually. See Figure 3-86. You may use the default values of all the configuration items herein.

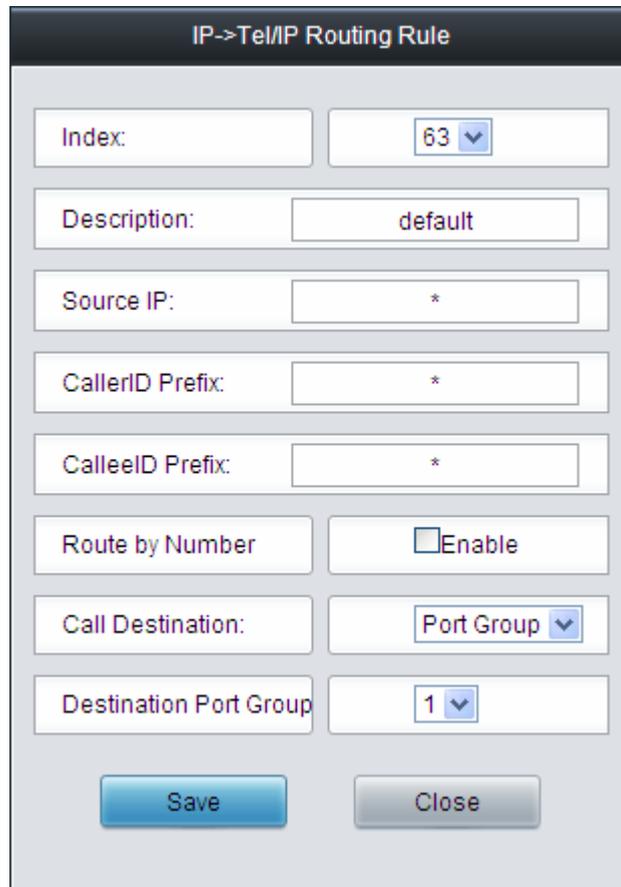


Figure 3-86 Add New Routing Rule (IP→Tel/IP)

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

Item	Description
<b>Index</b>	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.
<b>Description</b>	More information about each routing rule, with the default value of <i>default</i> .
<b>Source IP</b>	IP address from where the call is initiated. This item can be set to a specific IP address or "*" which indicates any IP address
<b>CallerID Prefix, CalleeID Prefix</b>	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 characters '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 <b>Source IP</b> specify a routing rule for calls. <b>Note:</b> "[*]" represents TFM symbol *, while "*" represents any string.
<b>Route by Number</b>	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 <b>SIP Account</b> on the Port Settings interface. In such case, the configuration item <b>Call Destination</b> goes invalid and shows <i>Route by Number</i> on the routing rule configuration interface. The default setting is <i>disabled</i> .
<b>Call Destination</b>	Designate a port group or an IP for the call to route.
<b>Destination Port Group</b>	Port group to which the call will be routed.
<b>Destination IP, Destination Port</b>	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 3-87 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.

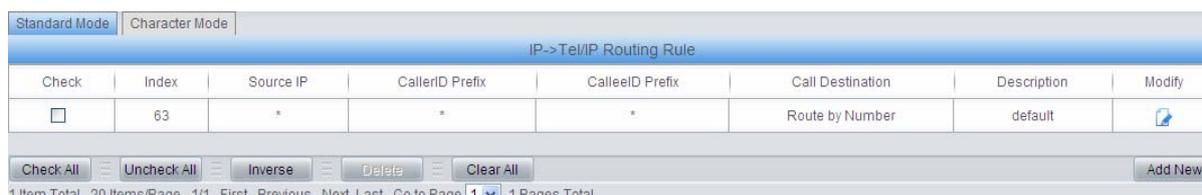


Figure 3-87 IP→Tel/IP Routing Rule Configuration Interface

Click **Modify** in Figure 3-87 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 3-87 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 3-87.

See Figure 3-88 for the IP→Tel 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.

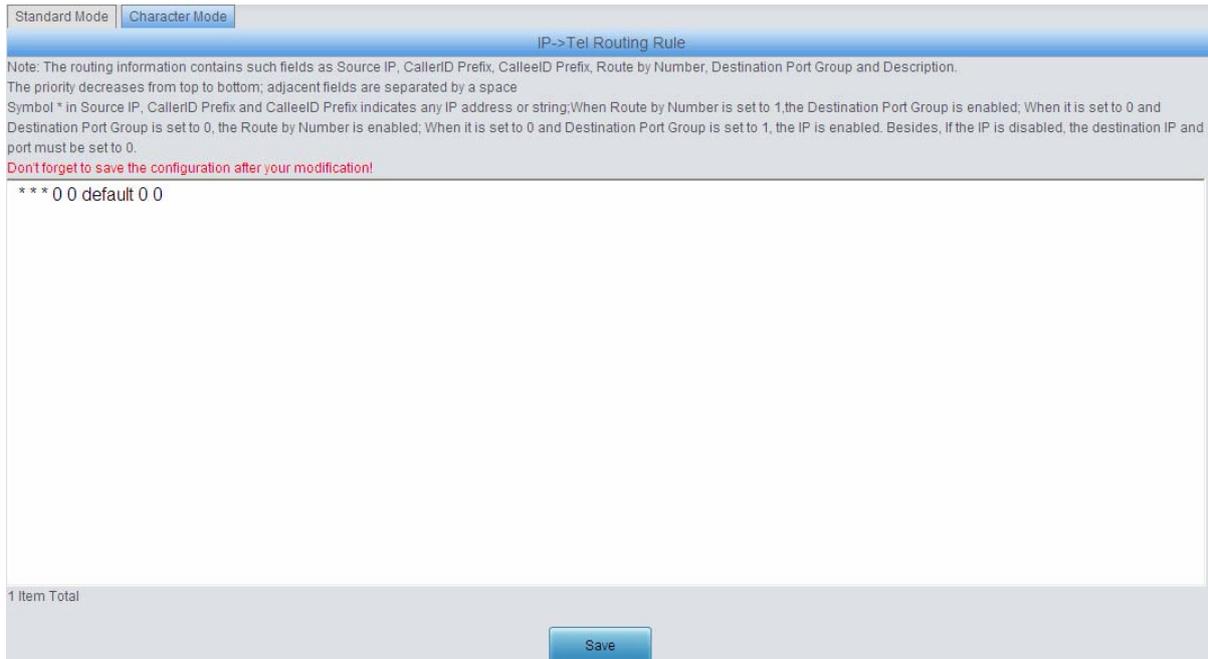


Figure 3-88 IP→Tel/IP Routing Rule Configuration Interface (Character)

### 3.8.3 Tel to IP



Figure 3-89 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-89 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. See Figure 3-90. You may use the default values of all the configuration items herein except for **Destination IP** and **Destination Port**.

Figure 3-90 Add New Routing Rule (Tel→IP)

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

Item	Description
<b>Index</b>	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.
<b>Description</b>	More information about each routing rule, with the default value of <i>default</i> .
<b>Source Port Group (Call Initiator)</b>	Port group from which the call is initiated. This item can be set to a specific port group or '*' which indicates any port group.
<b>CallerID Prefix, CalleelD Prefix</b>	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 indicate 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 <b>Source Port Group (Call Initiator)</b> specify a routing rule for calls. <b>Note:</b> "[*]" represents DTFM symbol *, while "*" represents any string.
<b>Destination IP, Destination Port</b>	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 3-91 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.

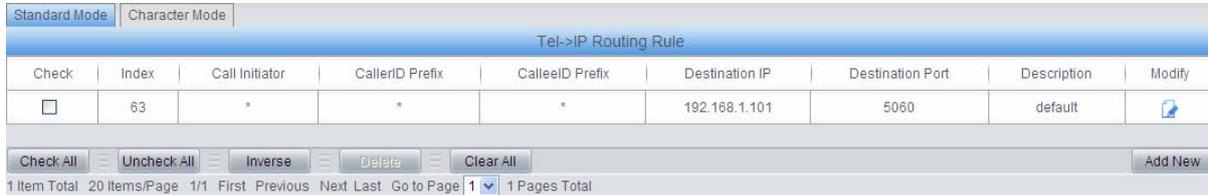


Figure 3-91 Tel→IP Routing Rule Configuration Interface

Click **Modify** in Figure 3-91 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 3-91 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 3-91.

See Figure 3-92 for 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.

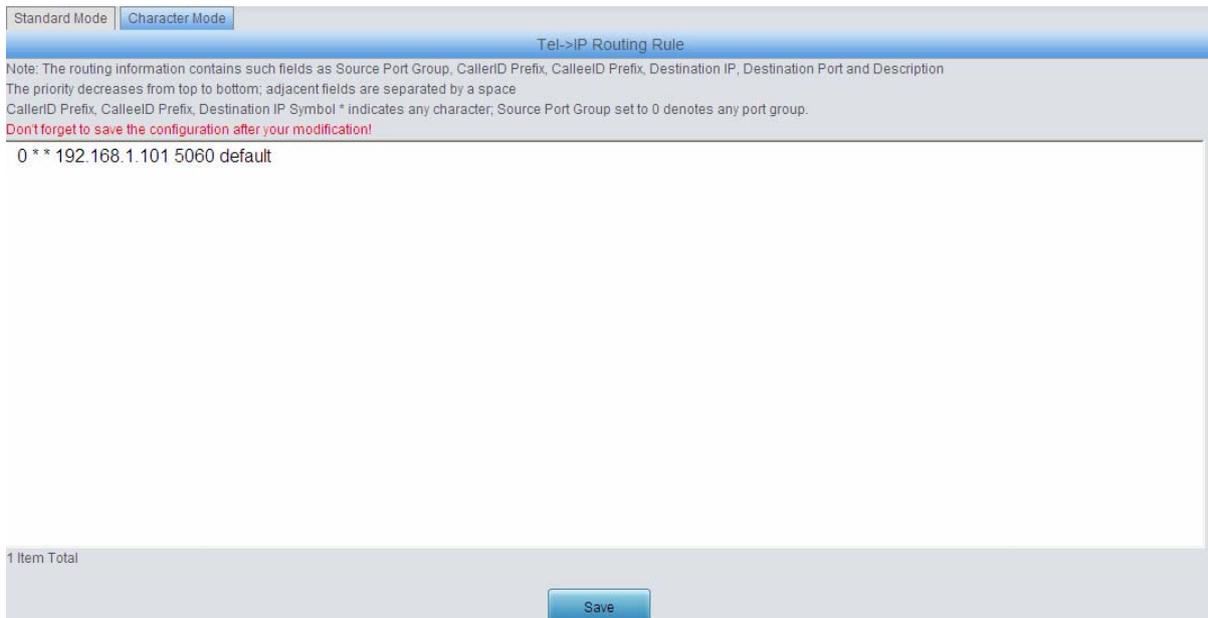


Figure 3-92 Tel→IP Routing Rule Configuration Interface (Character)

### 3.9 Number Manipulation

Number Manipulation includes four parts: **IP→Tel CallerID**, **IP→Tel CalleeID**, **Tel→IP CallerID** and **Tel→IP CalleeID**. See Figure 3-93.

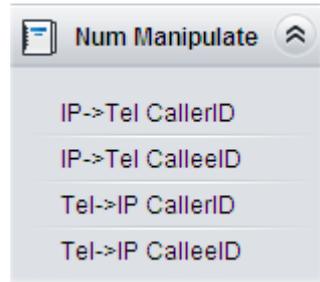


Figure 3-93 Number Manipulation

### 3.9.1 IP to Tel CallerID



Figure 3-94 IP→Tel CallerID Manipulation Interface (Standard)

See Figure 3-94 for the IP→Tel 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 3-95 for the IP→Tel CallerID manipulation rule adding interface. You may use the default values of all the configuration items herein.

IP->Tel/IP CallerID

Index:	<input type="text" value="63"/>
Description:	<input type="text" value="default"/>
Call Initiator:	<input type="text" value="*"/>
CallerID Prefix:	<input type="text" value="*"/>
CalleeID Prefix:	<input type="text" value="*"/>
Stripped Digits from Left:	<input type="text" value="0"/>
Stripped Digits from Right:	<input type="text" value="0"/>
Reserved Digits from Right:	<input type="text" value="0"/>
Prefix to Add:	<input type="text"/>
Suffix to Add:	<input type="text"/>

Figure 3-95 Add IP→Tel CallerID Manipulation Rule

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

Item	Description
<b><i>Index</i></b>	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.
<b><i>Description</i></b>	More information about each number manipulation rule, with the default value of <i>default</i> .
<b><i>Call Initiator</i></b>	IP address from where the call is initiated. This item can be set to a specific IP address or "*" which indicates any IP address.

<p><b>CallerID Prefix, CalleeID Prefix</b></p>	<p>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 <b>Call Initiator</b> specify a number manipulation rule for calls.</p> <p><b>Note:</b> "[*]" represents DTFM symbol *, while "*" represents any string.</p>
<p><b>Stripped Digits from Left</b></p>	<p>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.</p>
<p><b>Stripped Digits from Right</b></p>	<p>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.</p>
<p><b>Reserved Digits from Right</b></p>	<p>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.</p>
<p><b>Prefix to Add</b></p>	<p>Designated information to be added to the left end of the current number.</p>
<p><b>Suffix to Add</b></p>	<p>Designated information to be added to the right end of the current number.</p>

**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.

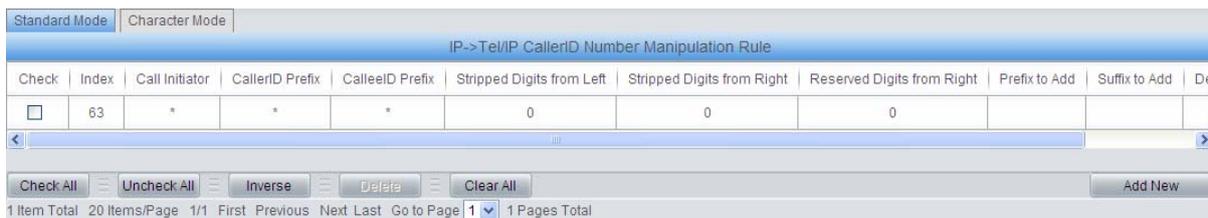


Figure 3-96 IP->Tel CallerID Manipulation Interface (Standard)

Click **Modify** in Figure 3-96 to modify a number manipulation rule. See Figure 3-97 for the IP->Tel CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the **Add IP->Tel CallerID Manipulation Rule** interface. Note that the item **Index** cannot be modified.

The screenshot shows a configuration window titled "IP->Tel CallerID". It contains several input fields and buttons:

- Index:** A dropdown menu showing "63".
- Description:** A text field containing "default".
- Call Initiator:** A text field containing "\*".
- CallerID Prefix:** A text field containing "\*".
- CalleedID Prefix:** A text field containing "\*".
- Stripped Digits from Left:** A text field containing "0".
- Stripped Digits from Right:** A text field containing "0".
- Reserved Digits from Right:** A text field containing "0".
- Prefix to Add:** An empty text field.
- Suffix to Add:** An empty text field.
- Buttons:** "Save" and "Close" buttons at the bottom.

Figure 3-97 Modify IP→Tel CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-94 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 3-94.

See Figure 3-98 for the IP→Tel 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.

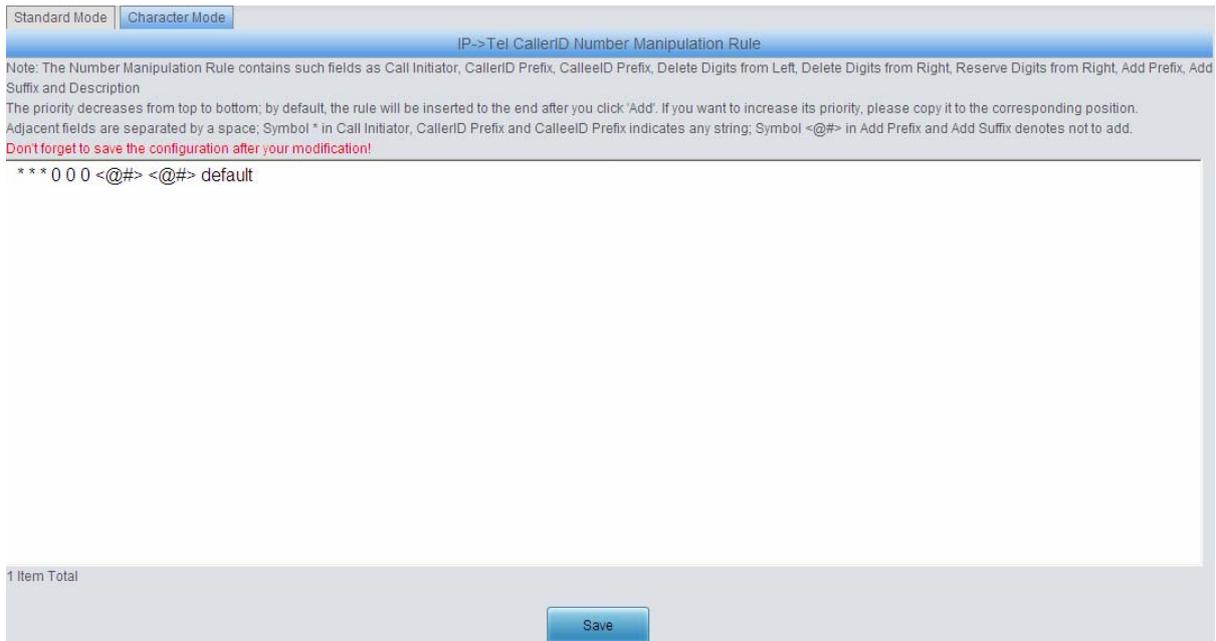


Figure 3-98 IP→Tel CallerID Manipulation Interface (Character)

### 3.9.2 IP to Tel CalleeID

The number manipulation process for IP→Tel CalleeID is almost the same as that for IP→Tel CallerID; only the number to be manipulated changes from CallerID to CalleeID. See Figure 3-99, Figure 3-100 for IP→Tel CalleeID Manipulation interface. The configuration items on this interface are the same as those on **IP→Tel CallerID Manipulation Interface** (Figure 3-94).

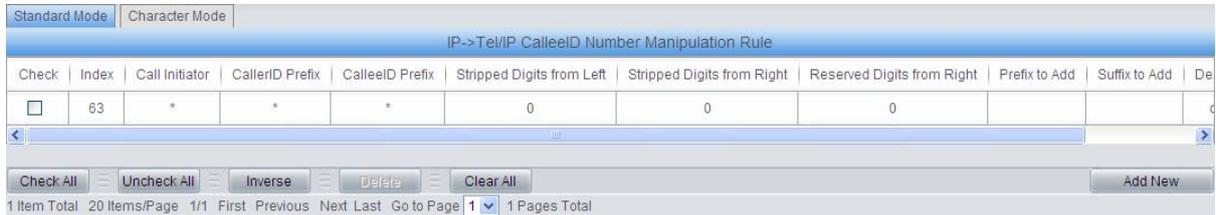


Figure 3-99 IP→Tel CalleeID Manipulation Interface(Standard)

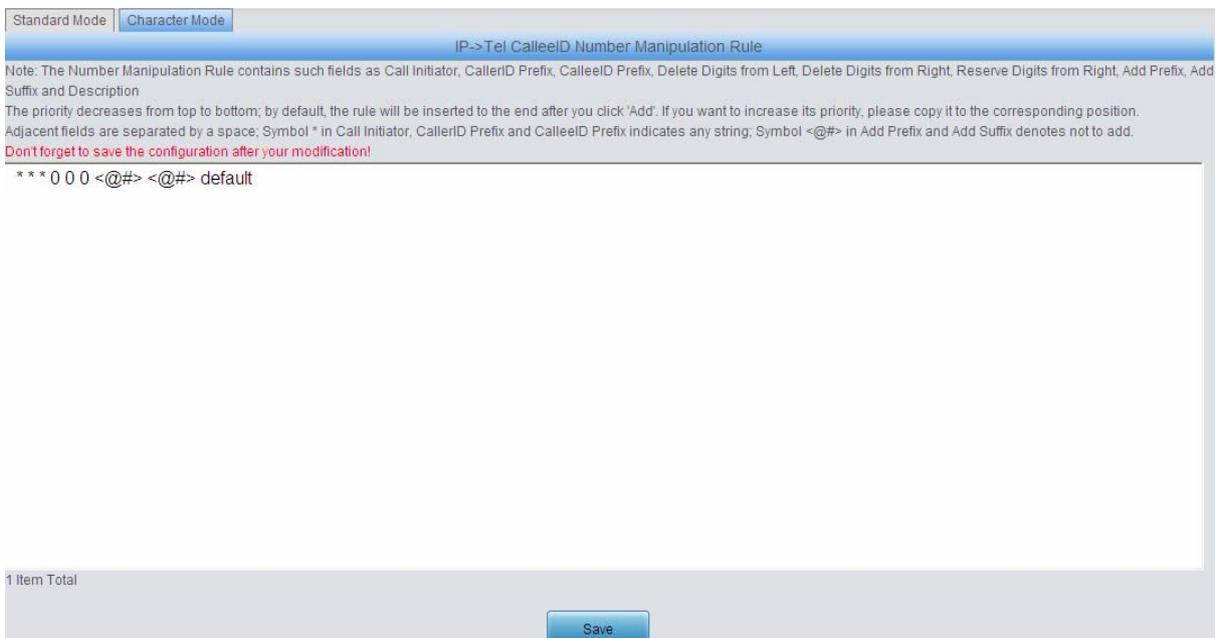


Figure 3-100 IP→Tel CalleID Manipulation Interface (Character)

### 3.9.3 Tel to IP CallerID

Figure 3-101 Tel→IP CallerID Manipulation Interface (Standard)

See Figure 3-101 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 3-102 for the Tel→IP CallerID manipulation rule adding interface. You may use the default values of all the other configuration items herein.

Figure 3-102 Add Tel→IP CallerID Manipulation Rule

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

Item	Description
<b>Index</b>	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.
<b>Description</b>	More information about each number manipulation rule, with the default value of <i>default</i> .
<b>Source Port Group (Call Initiator)</b>	Port group from which the call is initiated. This item can be set to a specific port group or "*" which indicates any port group.
<b>CallerID Prefix, CalleeID Prefix</b>	<p>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 <b>Call Initiator</b> specify a number manipulation rule for calls.</p> <p><b>Note:</b> "[*]" represents DTFM symbol *, while "*" represents any string.</p>
<b>Stripped Digits from Left</b>	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.
<b>Stripped Digits from Right</b>	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.
<b>Reserved Digits from Right</b>	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.
<b>Prefix to Add</b>	Designated information to be added to the left end of the current number.
<b>Suffix to Add</b>	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 3-101 to modify a number manipulation rule. See Figure 3-103 for the Tel→IP CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the **Add Tel→IP CallerID Manipulation Rule** interface. Note that the item **Index** cannot be modified.

Tel->IP CallerID

Index: 63

Description: default

Source Port Group: \*

CallerID Prefix: \*

CalleelD Prefix: \*

Stripped Digits from Left: 0

Stripped Digits from Right: 0

Reserved Digits from Right: 0

Prefix to Add:

Suffix to Add:

Save Close

Figure 3-103 Modify Tel→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-101 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 3-101.

See Figure 3-104 for 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.

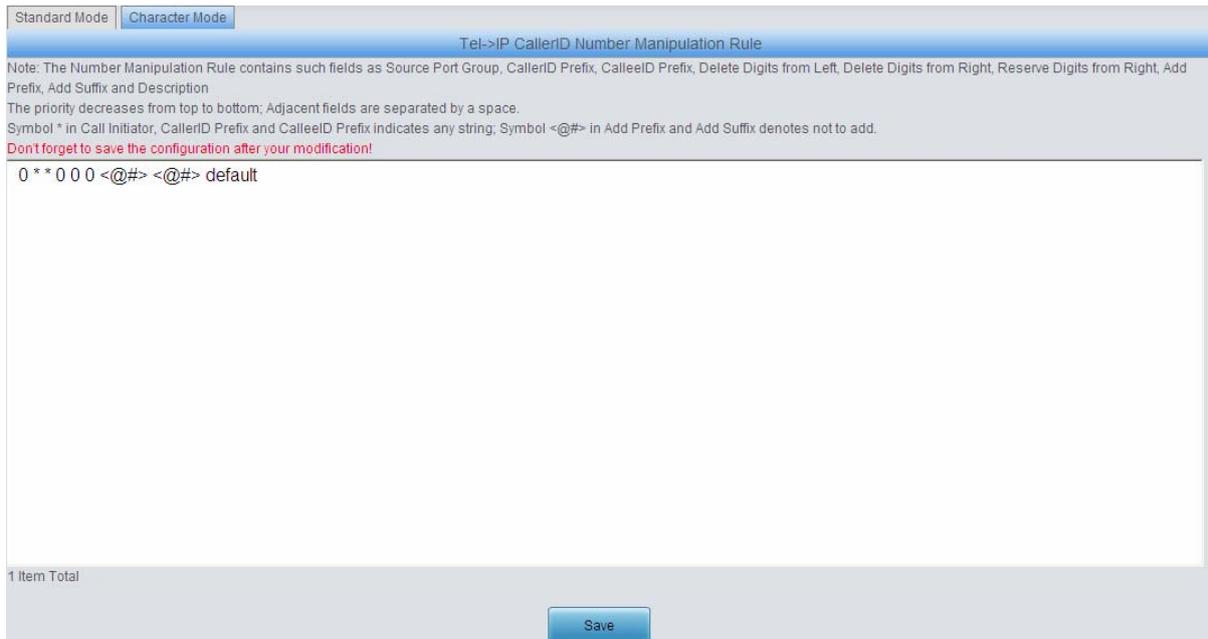


Figure 3-104 Tel→IP CallerID Manipulation Interface (Character)

### 3.9.4 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. See Figure 3-105, Figure 3-106 for the Tel→IP CalleeID manipulation interface. The configuration items on this interface are the same as those on **Tel→IP CallerID Manipulation Interface** (Figure 3-101).

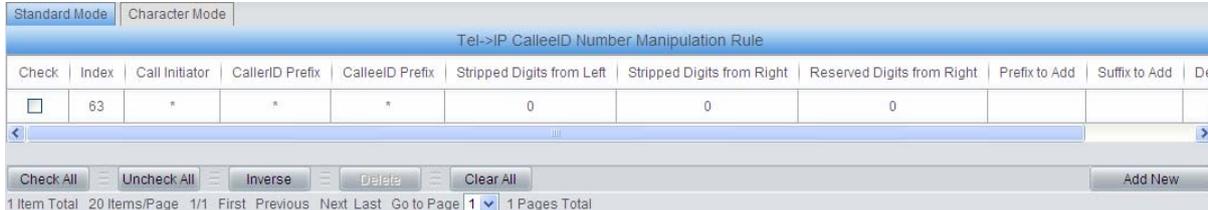


Figure 3-105 Tel→IP CalleeID Manipulation Interface (Standard)

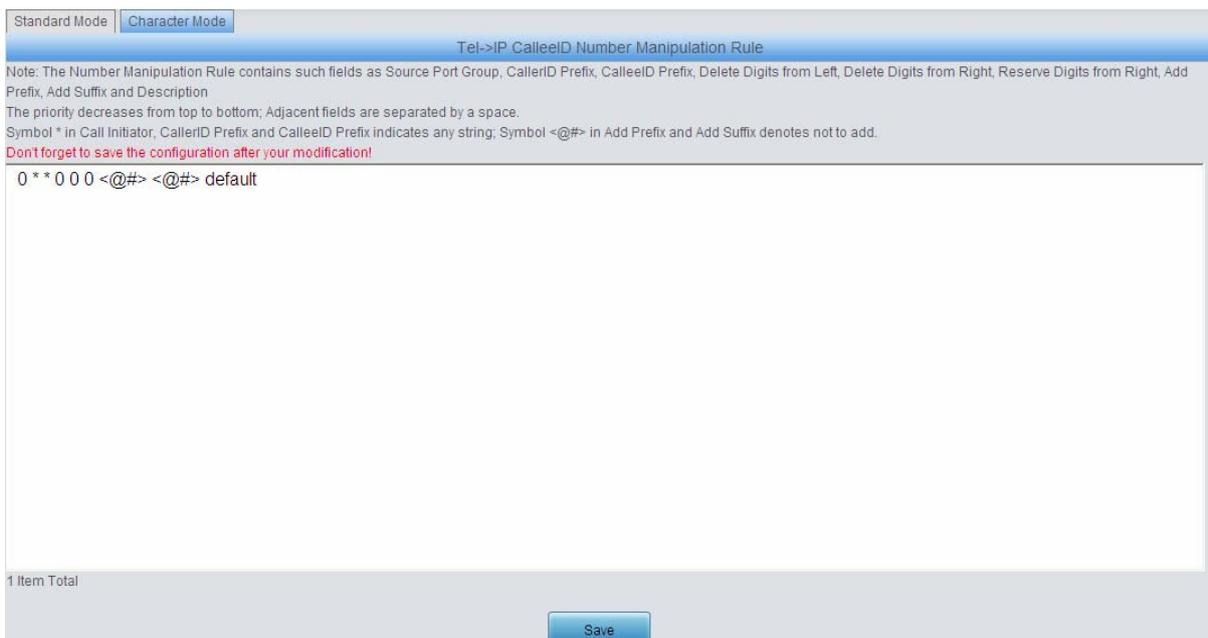


Figure 3-106 Tel→IP CalleID Manipulation Interface (Character)

### 3.10 System Tools

System Tools is mainly for gateway maintenance. It provides such features as IP modification, data backup and connectivity check. See Figure 3-107 for details.



Figure 3-107 System Tools

#### 3.10.1 Upgrade

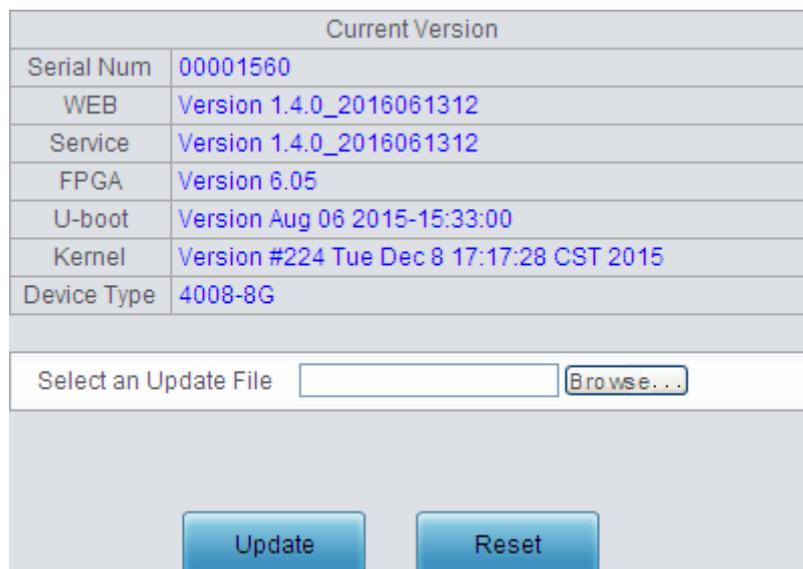


Figure 3-108 Upgrade Interface

See Figure 3-108 for the upgrade interface where 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.) via **Browse...** and click **Update**. Then the file uploading interface will appear. See Figure 3-109.

Current Version	
Serial Num	00001560
WEB	Version 1.4.0_2016061312
Service	Version 1.4.0_2016061312
FPGA	Version 6.05
U-boot	Version Aug 06 2015-15:33:00
Kernel	Version #224 Tue Dec 8 17:17:28 CST 2015
Device Type	4008-8G

34% 619kb/s

The file is uploading. Please do not leave this page!.....

### Upgrade Information

```
start upload upgrade file...
```

Figure 3-109 File Uploading Interface

After a successful uploading of the file, the gateway will start to upgrade the system. See Figure 3-110 and you can learn the detailed upgrading information from the upgrade information box at the bottom.

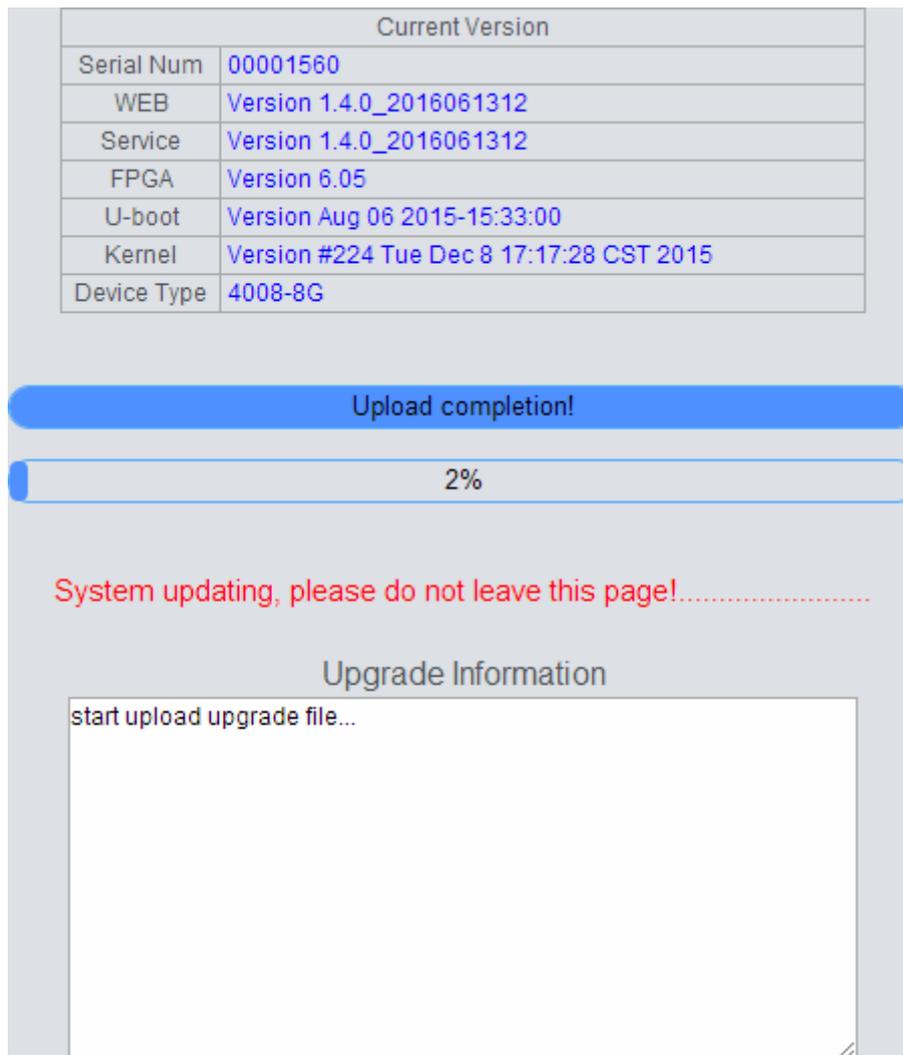


Figure 3-110 System Upgrading Interface

Note that clicking **Reset** can only delete the selected update file but not cancel the operation of **Update**.

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

### 3.10.2 Signaling Capture

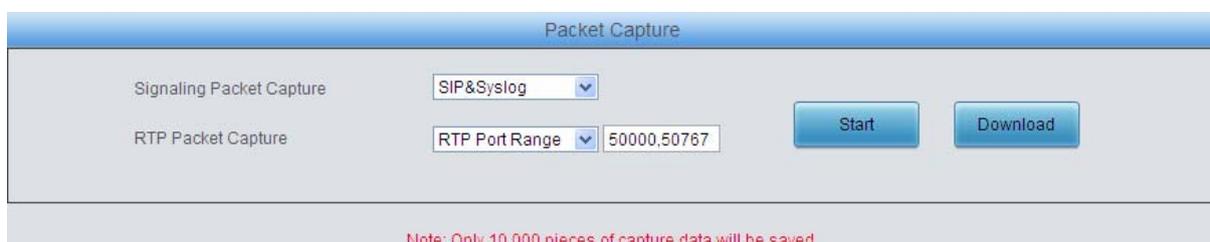


Figure 3-111 Signaling Capture Interface

See Figure 3-111 for the Signaling Capture interface. 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.

### 3.10.3 Data Recording



Figure 3-112 Data Recording Interface

See Figure 3-112 for the Data Recording interface. Click **Start** to start the recording. Click **Stop** to stop the recording. Click **Download** to download the recorded data.

### 3.10.4 Call Log

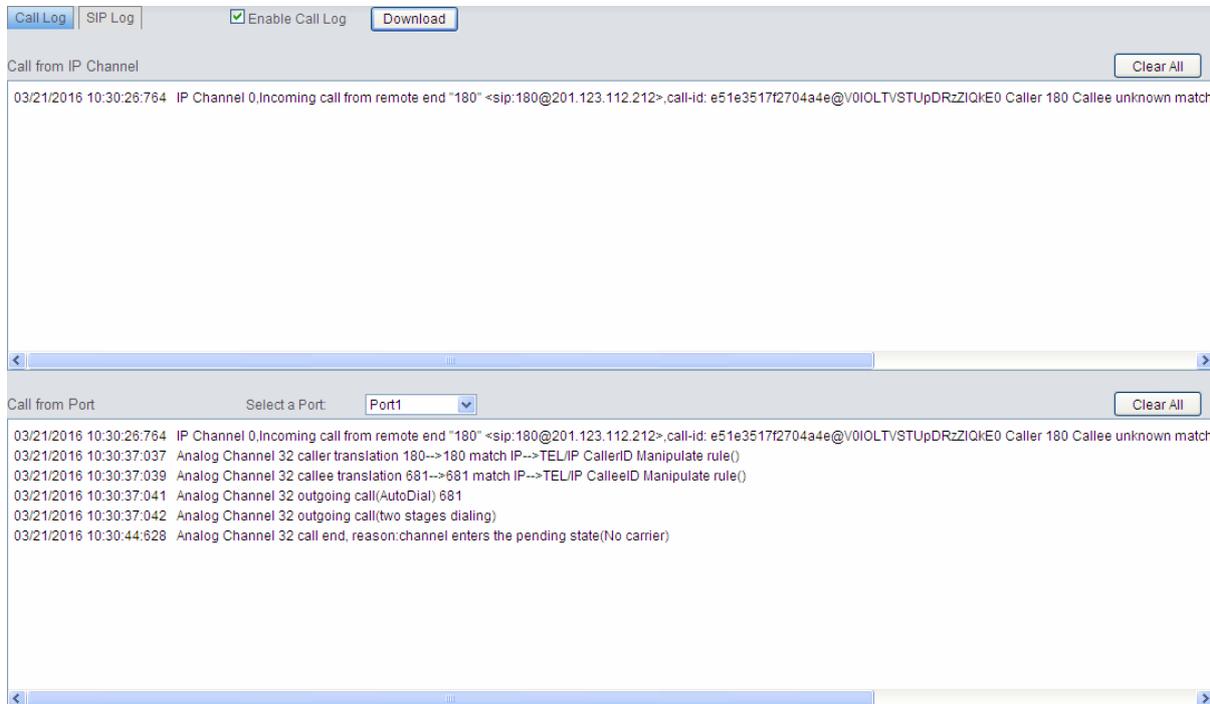


Figure 3-113 Call Log Interface

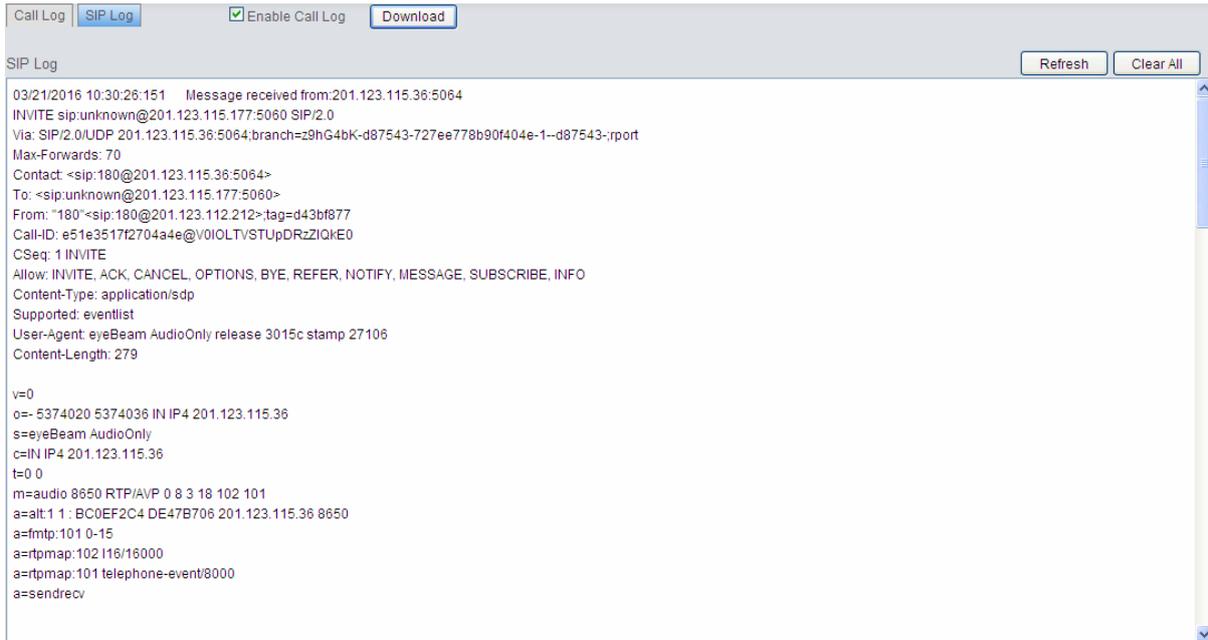


Figure 3-114 SIP Log Interface

See Figure 3-113, Figure 3-114 for 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**.

### 3.10.5 Change Password



Figure 3-115 Password Changing Interface

See Figure 3-115 for the password changing interface where 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.

### 3.10.6 Backup & Upload



Figure 3-116 Backup & Upload Interface

See Figure 3-116 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**.

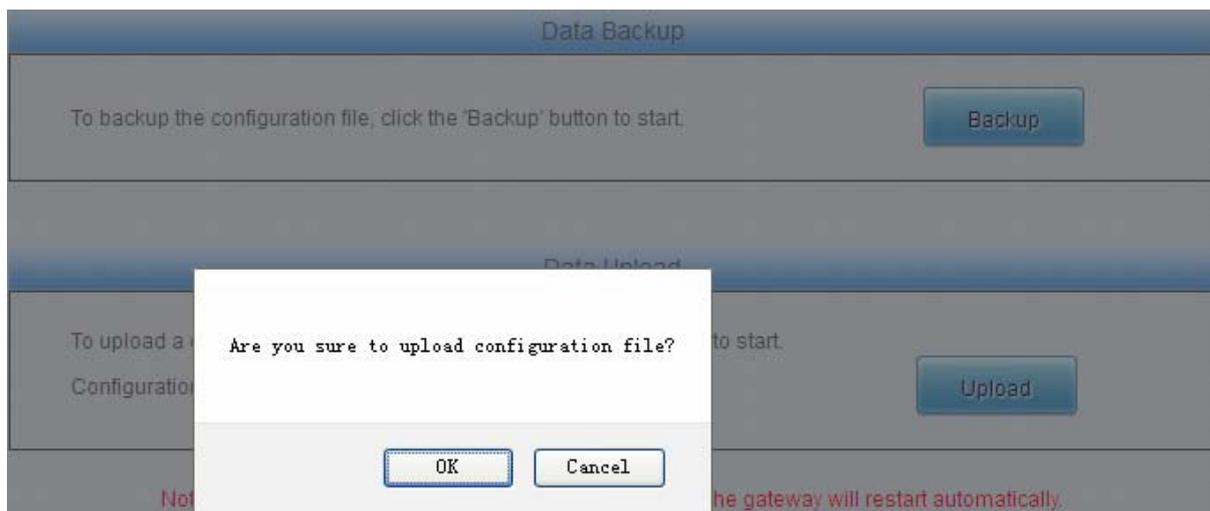


Figure 3-117 Backup & Upload & Prompt Interface

Click **OK** on the prompt box (Figure 3-117) to upload the configuration file to the gateway. Now the prompt information 'System is rebooting, please do not leave this page' appears. See Figure 3-118. The gateway will overwrite the current configurations with the uploaded data after restart. Click **Cancel** to cancel this upload directly.

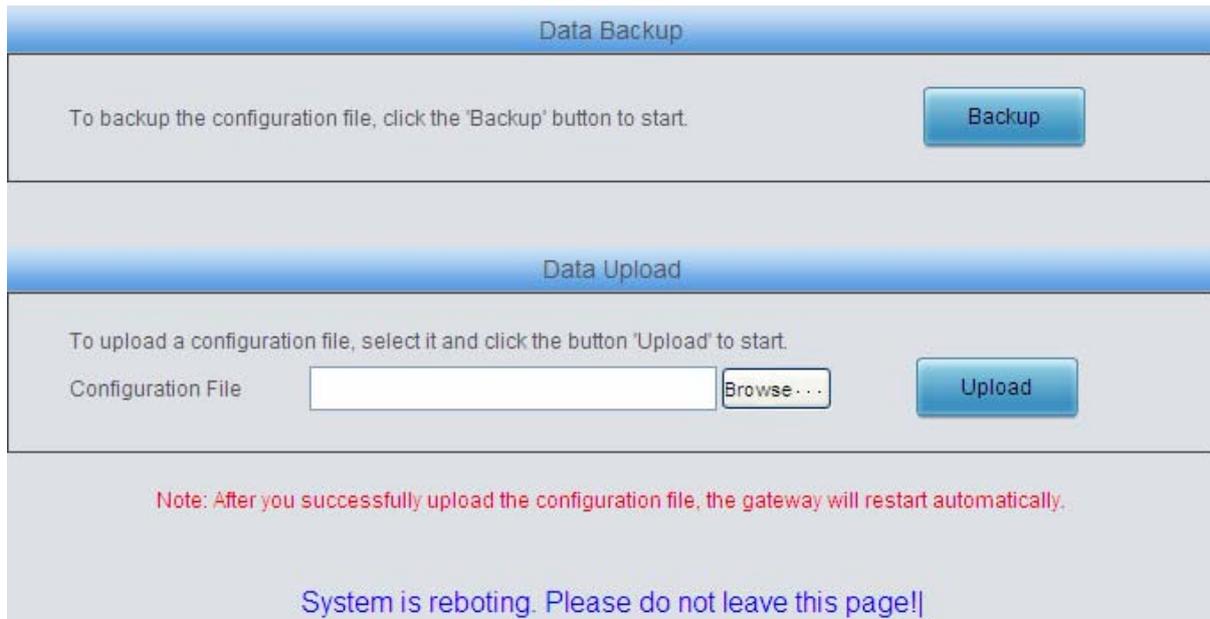


Figure 3-118 Configuration File Uploading Interface

### 3.10.7 Factory Reset



Figure 3-119 Factory Reset Interface

See Figure 3-119 for the factory reset interface. Click **Reset** to restore all configurations on the gateway to factory settings.

### 3.10.8 Restart

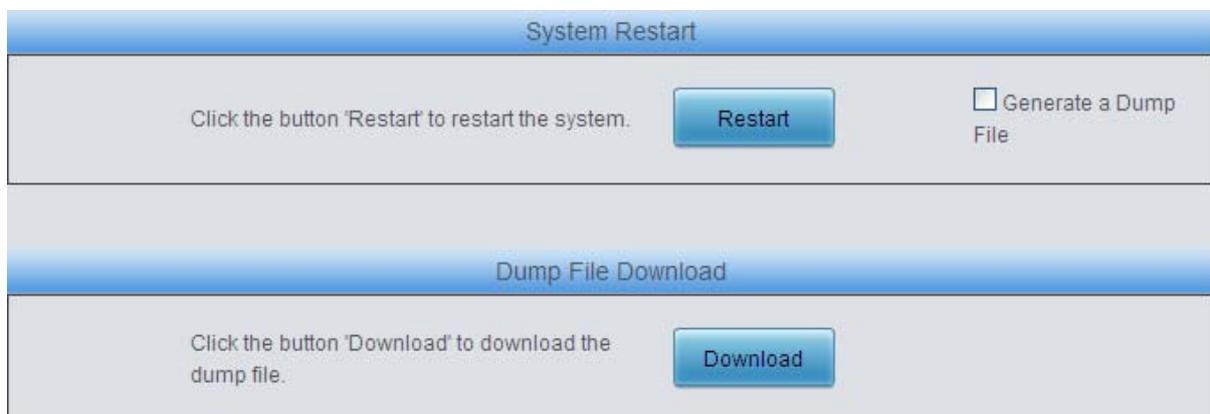


Figure 3-120 System Restart Interface

See Figure 3-120 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.

### 3.10.9 System Monitor

Figure 3-121 System Monitor Configuration Interface

See Figure 3-121 for 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.

### 3.10.10 SNMP Config

Figure 3-122 SNMP Configuration Interface

See Figure 3-122 for the SNMP configuration interface. If the SNMP feature is enabled, once the gateway receives a request from the SNMP management software, it will collect relevant information and reply them to the SNMP management software. By default, the SNMP feature is disabled. The available information includes kernel version, CPU usage, processes, memory usage, startup information, LAN status and etc. Currently, the gateway only provides the community string for information acquisition. The table below explains the configuration items shown in Figure 3-122.

Item	Description
<b>SNMP Server Address</b>	IP address of SNMP.

<b>Monitoring Port</b>	Monitoring Port for SNMP on the gateway.
<b>Access Password</b>	Community string used for information acquisition.

### 3.10.11 PING Test

Figure 3-123 Ping Test Interface

See Figure 3-123 for the Ping test interface. 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 shown in the above figure.

Item	Description
<b>Destination Address</b>	Destination IP address or domain name on which the Ping test is executed.
<b>Ping Count</b>	The number of times that the Ping test should be executed. Range of value: 1~100.
<b>Package Length</b>	Length of the data package used in the Ping test. Range of value: 56~1024 bytes.
<b>Info</b>	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.

### 3.10.12 TRACERT Test

Figure 3-124 Tracert Test Interface

See Figure 3-124 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
<b>Source IP Address</b>	Source IP address where the Tracert test is initiated.
<b>Destination Address</b>	Destination IP address on which the Tracert test is executed.
<b>Maximum Jumps</b>	Maximum number of jumps between the gateway and the destination address which are returned by the Tracert test. Range of value: 1~255.
<b>Info</b>	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.

### 3.10.13 Wireless Network Test

Figure 3-125 Wireless Network Test Interface

See Figure 3-125 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
<b>Port</b>	The port used for the test
<b>Called Number</b>	The called party number which will be dialed for the test
<b>Conversion Time Length</b>	The time length of the conversion
<b>Call Times</b>	The times of the testing call

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

# Appendix A Technical Specifications

## Dimensions

4004/4008 series: 260×153×30 mm<sup>3</sup>

4016 series: 440×44×200 mm<sup>3</sup>

## Weight

4004/4008 series Net: 1.2 kg

4016 series Net: 3.5 kg

## Environment

Operating temperature: 0 °C—45 °C

Storage temperature: -20 °C—85 °C

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 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

GSM Frequency band: 850/900/1800/1900MHz

WCDMA Frequency band: GSM 900/1800MHz,  
UMTS 900/2100MHz

CDMA Frequency band: CDMA 2000 800MHz

SMS CODEC: ASCII/UCS2

## 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 [3.5.5 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 VPN Certificate

The steps to make a VPN certificate;

Step 1 Get the file of client.ovpn from the VPN server and rename it to “client.conf”.

Step 2 Examine or add the following content in/to 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.

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

persist-tun

persist-key

cipher AES-128-CBC

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.)

client

remote 192.168.143.235 1194 udp (Note: Fill in the IP address and the port number of the VPN server.)

tls-remote yfadmin

comp-lzo

passtos

<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>

<tls-auth>

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

</tls-auth>

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.

## Appendix D 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.

### Headquarters

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