Synway SMG Series Analog Gateway

SMG1008

SMG1016

SMG1032

Analog Gateway

User Manual

Version 1.0

Synway Information Engineering Co., Ltd

www.synway.net

Content

Copyright Declaration

Revision History

Version	Date	Comments
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Chapter 1 Product Introduction

Thank you for choosing Synway SMG Series Analog Gateway!

The Synway SMG series analog gateway products (hereinafter referred to as ‘SMG analog gateway’) are mainly used for connecting traditional phone sets, fax machines and PBXes with the IP telephony network or IP PBX. It provides a powerful, reliable and cost-effective VoIP solution for such occasions as IP call centers and multi-branch agencies.

SMG series analog gateway has three modules:

⌘ SMG1008：8 FXS/FXO

⌘ SMG1016：16 FXS/FXO

⌘ SMG1032：32 FXS/FXO

1.1 Typical Application

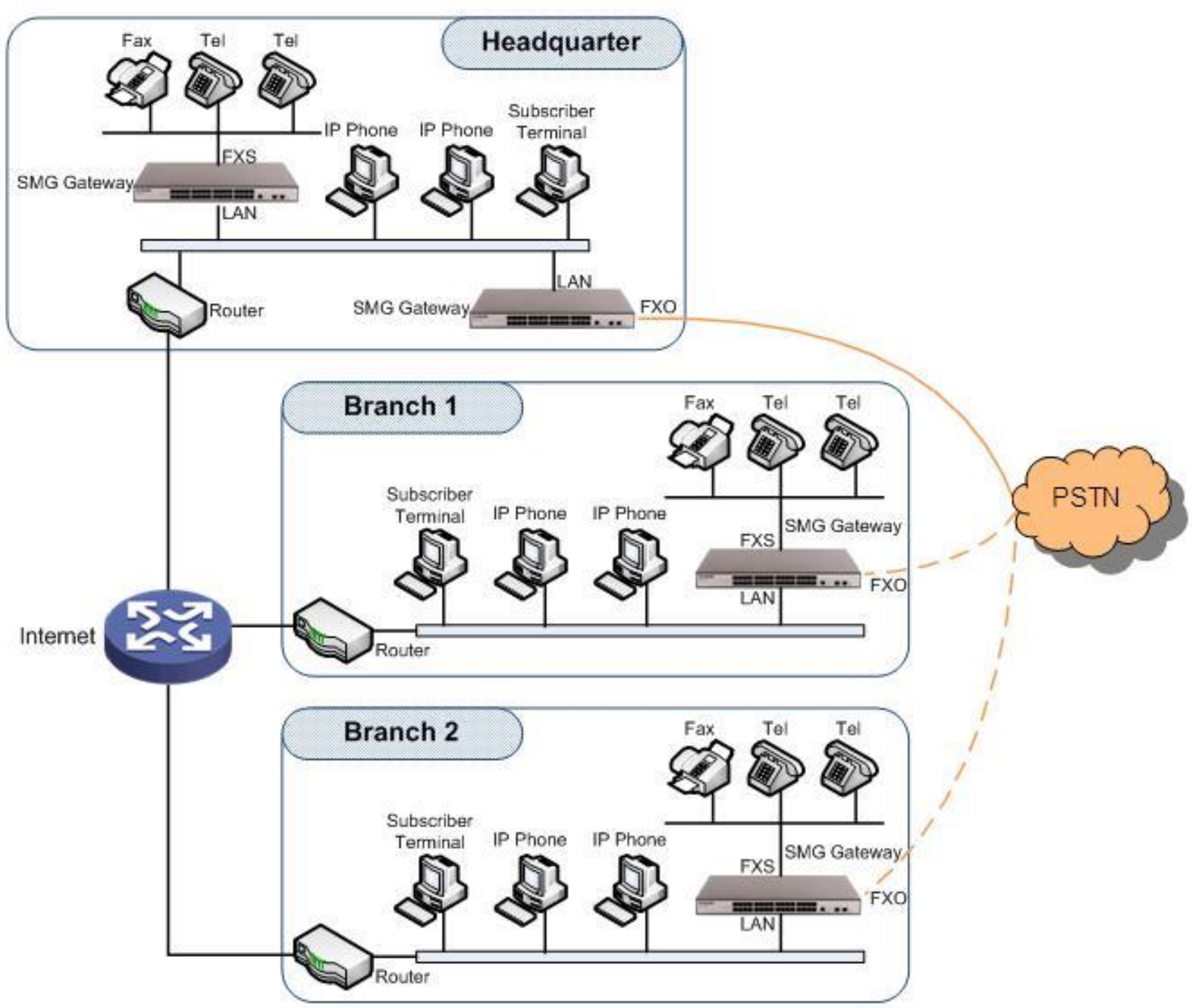

Figure 1-1 Typical Application

1.2 Feature List

Basic Features	Description
TDM Call	Call initiated from TDM to IP, via routing and number manipulation to obtain the called IP address.
IP Call	Call initiated from IP to TDM, via routing and number manipulation to obtain the call destination.
Number Manipulation	Peels off some digits of a phone number from left/right, or adds a prefix/suffix to a phone number.
Call Forward	Three options available: Unconditional, Busy and No Reply.
Call Waiting	When an FXS channel receives another call while it is in conversation, it will have the newly received call keep waiting. Once the current call is finished, the new one will ring the FXS channel and wait for its answer.
Auto Dial	If there is no dialing operation in a designated time period after pickup, the preset auto dial number will be called.
Do Not Disturb	Rejects all the incoming calls to the channel.
CID	Displays the CallerID.
Echo Cancellation	Provides the echo cancellation feature for a call conversation over the FXS/FXO channel.
TDM/VoIP Routing	Routing path: from IP to TDM or from TDM to IP.
Signaling & Protocol	**Description**
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261
Voice	CODEC: G.711A, G.711U, G.729A/B DTMF Mode: RFC2833, SIP INFO, INBAND
Network	**Description**
Network Protocol	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN
Static IP	IP address modification support
DNS	Domain Name Service support
Security	**Description**
Admin Authentication	Support admin authentication to guarantee the resource and data security
Maintain & Upgrade	**Description**
WEB Configuration	Support of configurations through the WEB user interface
Language	Chinese, English
Software Upgrade	Support of user interface, gateway service, kernel and firmware upgrades based on WEB
Tracking Test	Support of Ping and Tracert tests based on WEB

SysLog Type	Three options available: ERROR, WARNING, INFO

1.3 Hardware Description

The SMG analog gateway features 1U rackmount design and integrates embedded LINUX system within the POWERPC+DSP hardware architecture. It has 8/16/32 voice ports (FXS/FXO) and 2 LANs on the chassis. Each voice port can be configured on demand to serve as an FXS or FXO interface; however, the respective amount of FXS and FXO interfaces must be multiples of 2. See Figure 1-2, Figure 1-3 and Figure 1-4 for product appearance.

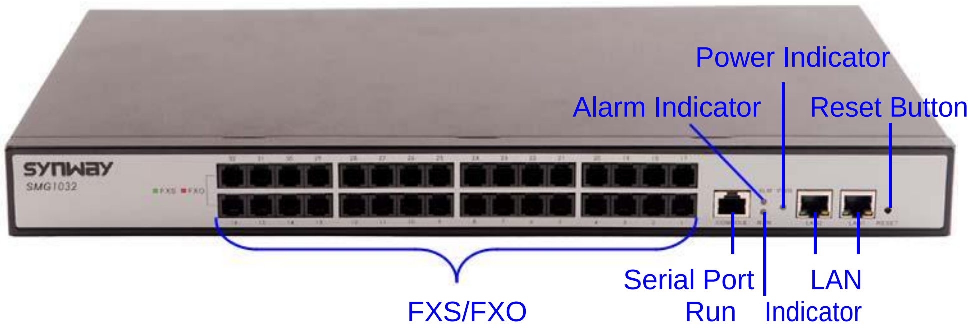

Figure 1-2 Front View

Figure 1-3 Rear View

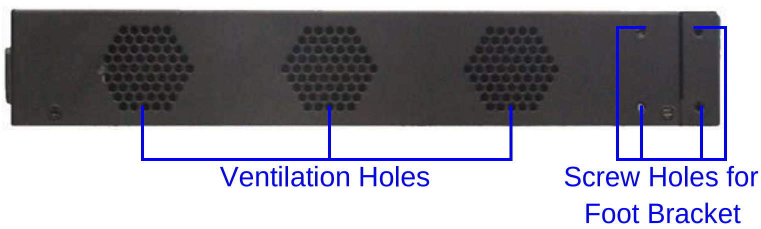

Figure 1-4 Left View

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

Interface	Description
LAN	Amount: 2
	Type: RJ-45
	Bandwidth: 10/100Mbps
	Self-Adaptive Bandwidth Supported
	Auto MDI/MDIX Supported
FXS/FXO	Amount: 8/16/32

	Type: RJ-11
	Maximum Transmission Distance: 1500m
	Charge Mode: Negative Anti-billing Supported
Serial Port	Amount: 1
	Type: RS-232
	Baud Rate: 115200bps
	Connector: RJ45 to DB-9 Connector
	Data Bits: 8 bits
	Stop Bit: 1 bit
	Parity Unsupported
	Flow Control Unsupported
Button	**Description**
Power Key	Power on/off the SMG analog gateway.
Reset Button	Restore the gateway to factory settings.
LED	**Description**
Power Indicator	Indicates the power state. It lights up when the gateway starts up with the power cord well connected
Run Indicator	Indicates the running status. For more details, refer to 1.4 Alarm Info.
Alarm Indicator	Alarms the device malfunction. For more details, refer to 1.4 Alarm Info.
Link Indicator	The green LED on the left of LAN, indicating the network connection status.
ACT Indicator	The orange LED on the right of LAN, whose flashing tells data are being transmitted.
Channel Indicator	1. Lights green for an FXS channel; 2. Lights red for an FXO channel; 3. Flashes when the channel state goes into Off-Hook.

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

1.4 Alarm Info

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

LED	State	Description
Run Indicator	Go out	System is not yet started.
	Light up	System is starting.
	Flash	System is normal.
Alarm Indicator	Go out	System is normal.
	Light up	Upon startup: System is normal. In runtime: System is abnormal.
	Flash	System 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 C 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 SMG analog gateway in the shortest time.

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

- ⌘ SMG Series Analog Gateway *1
- ⌘ Foot Bracket *2, Rubber Foot Pad *4, Screw for Foot Bracket *8
- ⌘ 220V Power Cord *1
- ⌘ Warranty Card *1
- ⌘ Installation Manual *1

Step 2: Properly fix the SMG analog gateway.

If you do not need to place the gateway on the rack, simply fix the 4 rubber foot pads. Otherwise, you should first fix the 2 foot brackets onto the chassis and then place the chassis on the rack.

Step 3: Connect the power cord.

Make sure the device is well grounded before you connect the power cord. Check if the power socket has the ground wire. If it doesn't, use the grounding stud on the rear panel of the device (See Figure 1-3) for earthing.

Step 4: Connect the network cable.

Step 5: Connect the telephone line. The line from PSTN should be connected to FXO port (port with red LED flashing); the line from station should be connected to FXS port (port with green LED flashing).

You can use a common telephone line directly or construct a telephone line by yourself according to Figure 2-1. Note that only the middle two cores in the RJ11 jack are valid for use.

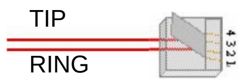

Figure 2-1 RJ11 Connection

Step 6: Log in the gateway.

Enter the original IP address (192.168.0.101 or 192.168.1.101) of the SMG analog 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 3.8.5 Change Password. After changing the password, you are required to log in again.

Step 7: Modify IP address of the gateway.

You can modify the IP address of the gateway via 'System Tools → Network' on the WEB interface to put it within your company's LAN. Refer to 3.8.1 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 8: 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 3.4.2 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 FXS ports which are connected with stations to it. Refer to 3.5.2 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 FXS port which is connected with a station 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 3.6.3 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. Pick up the station and 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: Pick up the station and 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 FXS ports which are connected with stations to it. Refer to 3.5.2 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 FXS port which is connected with a station 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' on the WEB interface and click the 'Add New' button to add a new routing rule. Refer to 3.6.2 IP→Tel 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 SMG analog gateway to ring the station.

   Example: Provided the IP address of the SMG analog gateway is 192.168.0.101 and the port is 5060, use the IP phone to call the IP address 192.168.0.101 and the station connected with Port1 will ring.

Step 9: Enable the auto dial feature. (Skip this step if not necessary.)

Go to the Port Settings interface to enable the auto dial feature and set the parameters 'Auto Dial Number' and 'Wait Time before Auto Dial'. If there is no dialing operation in a time period (i.e. Wait Time before Auto Dial) after pickup, the port will automatically call the preset number (i.e. Auto Dial Number). Refer to 3.5.1 Port for detailed instructions.

Step 10: Enable the DND (do not disturb) feature. (Skip this step if not necessary.)

Go to the Port Settings interface to enable the DND feature. Then, the FXS port will reject all incoming calls. Refer to 3.5.1 Port for detailed instructions.

Step 11: Enable the call waiting feature. (Skip this step if not necessary.)

Go to the Port Settings interface to enable the call waiting feature. Then the corresponding FXS port while in conversation can accept another call from IP and keep it in the waiting state. Once the current conversation is finished and the station hangs up, the call in the waiting state will ring the station and wait for answer. During the time in the waiting state, it will always hear the ringback tone from the FXS port. Refer to 3.5.1 Port for detailed instructions.

Step 12: Perform call forwarding. (Skip this step if not necessary.)

Situation 1: Hook-flash operation

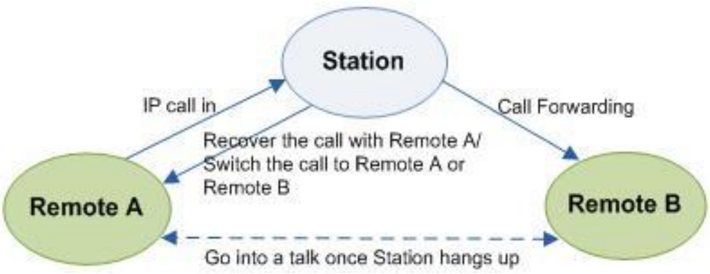

Figure 2-2 Call Forward via Hook-flash

As shown above, Remote A initiates and establishes a call with Station. Then by a hook-flash operation, that is, a rapid clap on the hook or pressing the 'flash' button on the phone set, Station can forward the call to Remote B.

Once a flash is generated, Station will go into the dialing state (the FXS port sends it dialing tones) before it dials the forwarding number.

If the dialing succeeds, the FXS port will send ringback tones to Station. Provided Remote B picks up the call, at this time Station can：

a) Directly talk with Remote B;

b) Perform another hook-flash operation to switch the call to either Remote A or Remote B.

c) Hang up to make Remote A and Remote B go into a direct talk with each other.

If the dialing fails, the FXS port will send busy tones to Station. At this time Station can：

a) Hang up to go back to the ringing state; then pick up the call again to recover the talk with Remote A.

b) Perform the hook-flash operation again without hanging up the call to recover the talk with Remote A.

Once Station recovers the call with Remote A, it can forward the call again by a new hook-flash operation.

Situation 2: Automatic call forward

Go to the port setting interface to enable the automatic call forward feature and fill in a forward number. According to what you set, the SMG analog gateway can automatically forward the incoming calls on three conditions: unconditional, busy, no reply. Note that this feature is applicable only to a single port, but not to a port group consisting of more than one port. Refer to 3.5.1 Port for detailed instructions.

Special Instructions:

⌘ The chassis of the SMG analog gateway must be grounded for safety reasons, according to standard industry requirements. A simple way is earthing with the third pin on the plug or the grounding studs on the machine. No or improper grounding may cause instability in operation as well as decrease in lightning resistance.

z As the device will gradually heat up while being used, please maintain good ventilation to prevent sudden failure, ensuring that the ventilation holes (see Figure 1-4) 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 3.8.5 Change Password.

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

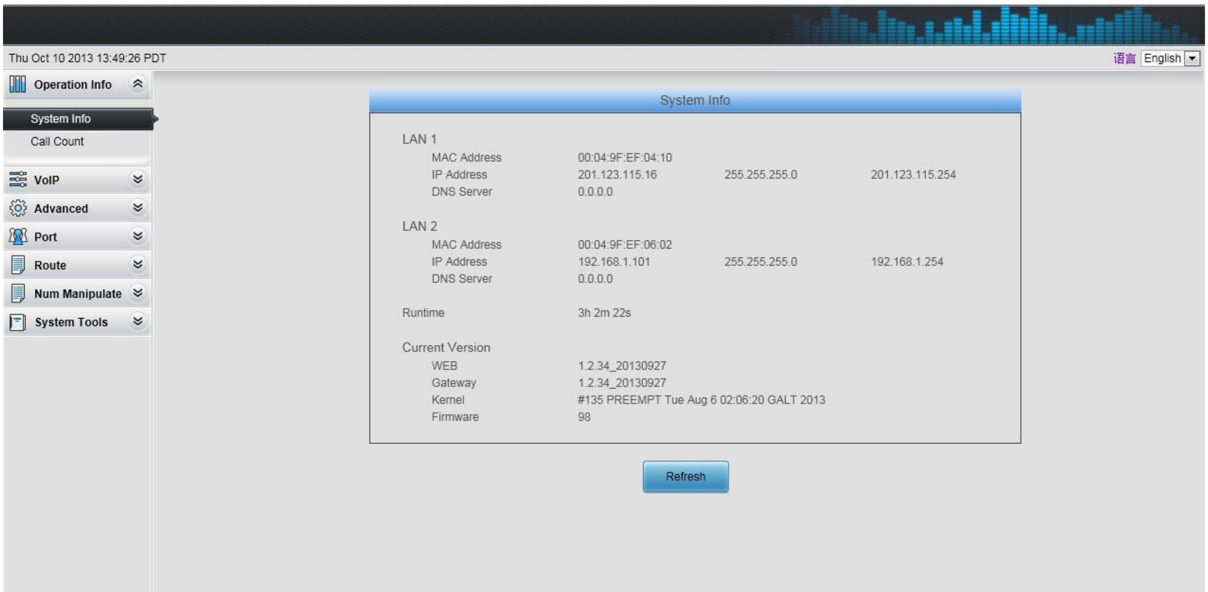

Figure 3-2 Main Interface

3.2 Operation Info

Operation Info includes two parts: ***System Info*** and ***Call Count***, showing the current running status of the gateway. See Figure 3-3.

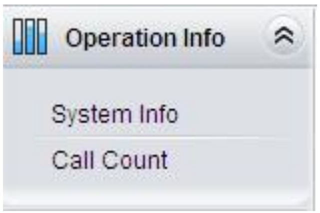

Figure 3-3 Operation Info

3.2.1 System Info

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
MAC Address	MAC address of LAN 1 or LAN 2.
IP Address	The three parameters from left to right are IP address, gateway and subnet mask of LAN 1 or LAN 2.
DNS Server	DNS server address of LAN 1 or LAN 2.
Runtime	Time of the gateway keeping running normally after startup.
WEB	Current version of the WEB interface.
Gateway	Current version of the gateway service.
Kernel	Current version of the system kernel on the gateway.
Firmware	Current version of the firmware on the gateway.

3.2.2 Call Count

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

Figure 3-5 Call Count Interface

See Figure 3-5 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 3-5.

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

3.3 VoIP Settings

VoIP Settings includes two parts: ***SIP*** and ***Media*** See Figure 3-6. SIP Settings is used to configure the general SIP parameters while Media Settings is to set the RTP port and the payload type.

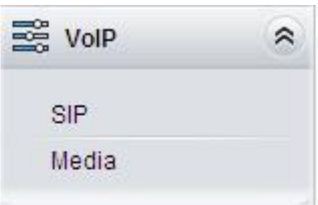

Figure 3-6 VoIP Settings

3.3.1 SIP Settings

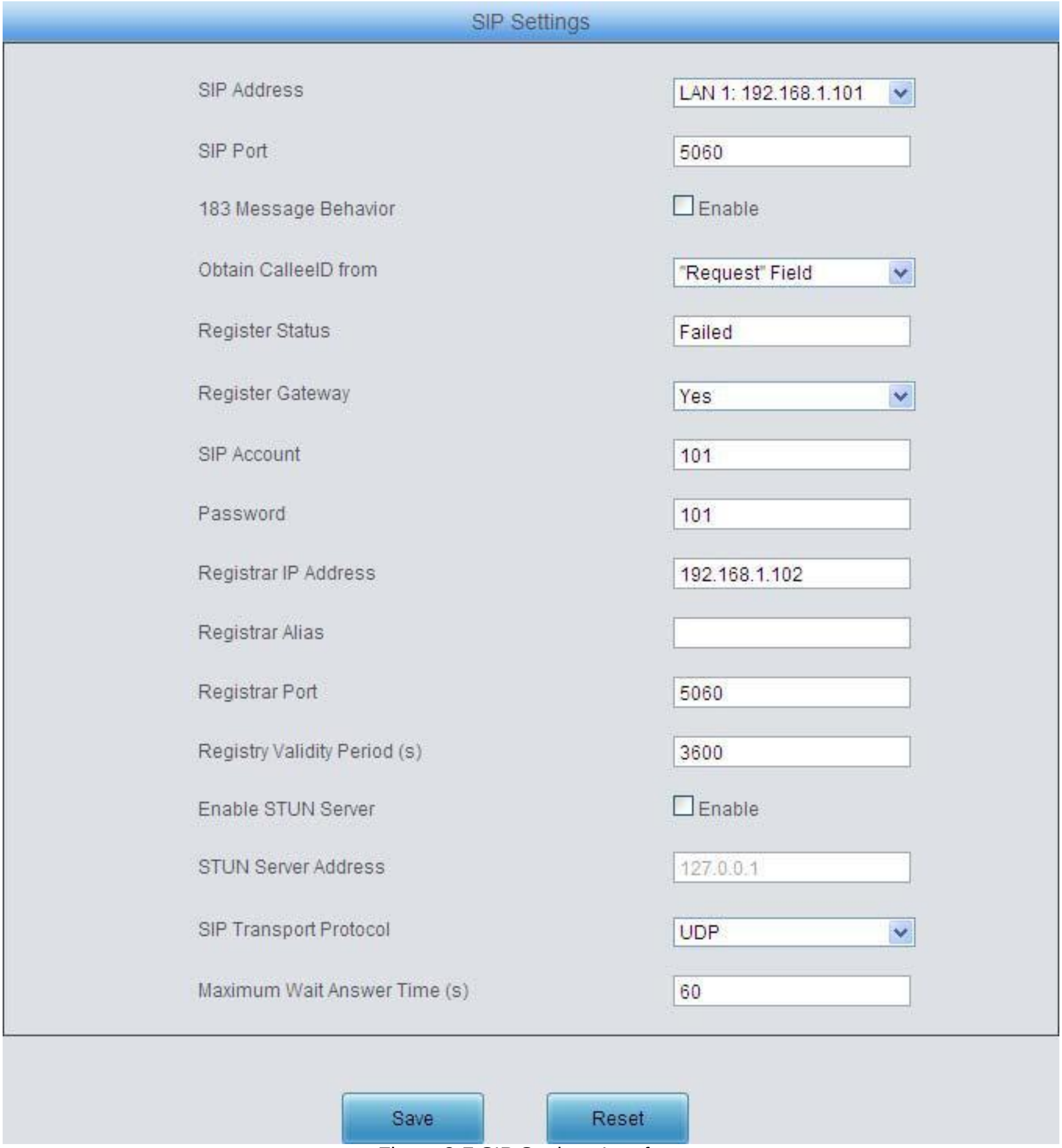

Figure 3-7 SIP Settings Interface

See Figure 3-7 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 service, do it immediately to apply the changes. Refer to 3.8.8 Restart for detailed instructions. The table below explains the items shown in Figure 3-7.

Item	Description
SIP Address	IP address of SIP signaling, using LAN 1 by default.
SIP Port	Monitoring port of SIP signaling. Range of value: 1024~65535, with the default value of 5060.
183 Message Behavior	Sets whether to send the 183 message instead of 180 to respond to the ringing tone when the SIP end serves as the called party. By default, this feature is disabled.

Obtain CalleeID from	There are two optional ways to obtain the called party number: from "*To" Field* or from "*Request" Field*. The default value is "*Request" Field*.
Register Status	Registration status of the gateway. When ***Register Gateway*** is set to *No*, the value of this item is *Unregistered*; when ***Register Gateway*** is set to *Yes*, the value of this item is either *Failed* or *Registered*.
Register Gateway	Sets whether to register the gateway as a whole. The default value is *No*. Only when this configuration is set to *Yes* can you see the configuration items ***SIP Account*** and ***Password***.
SIP Account	When the gateway initiates a call to SIP, this item corresponds to the username of SIP; when the gateway initiates a call to PSTN, this item corresponds to the displayed CallerID.
Password	Registration password of the gateway. To register the gateway to SIP, both configuration items ***SIP Account*** and ***Password*** should be filled in.
Registrar IP Address	Address of the registry server to which the gateway is registered.
Registrar Alias	Alias of the registry server. Only on some special servers does this item need to be configured.
Registrar Port	Signaling port of the registry server.
Registry Validity Period	Validity period of the SIP registry. Once the registry is overdue, the gateway should be registered again. This configuration item is valid only when ***Register Gateway*** is set to *Yes*. Range of value: 10~3600, calculated by s, with the default value of 3600.
Enable STUN Server	Sets whether to enable the STUN server for NAT traversal. By default the STUN server is disabled.
STUN Server Address	Address of the server for STUN traversal.
SIP Transport Protocol	There are two modes *UDP* and *TCP* available for running the SIP protocol. The default value is *UDP*.
Maximum Wait Answer Time	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.

3.3.2 Media Settings

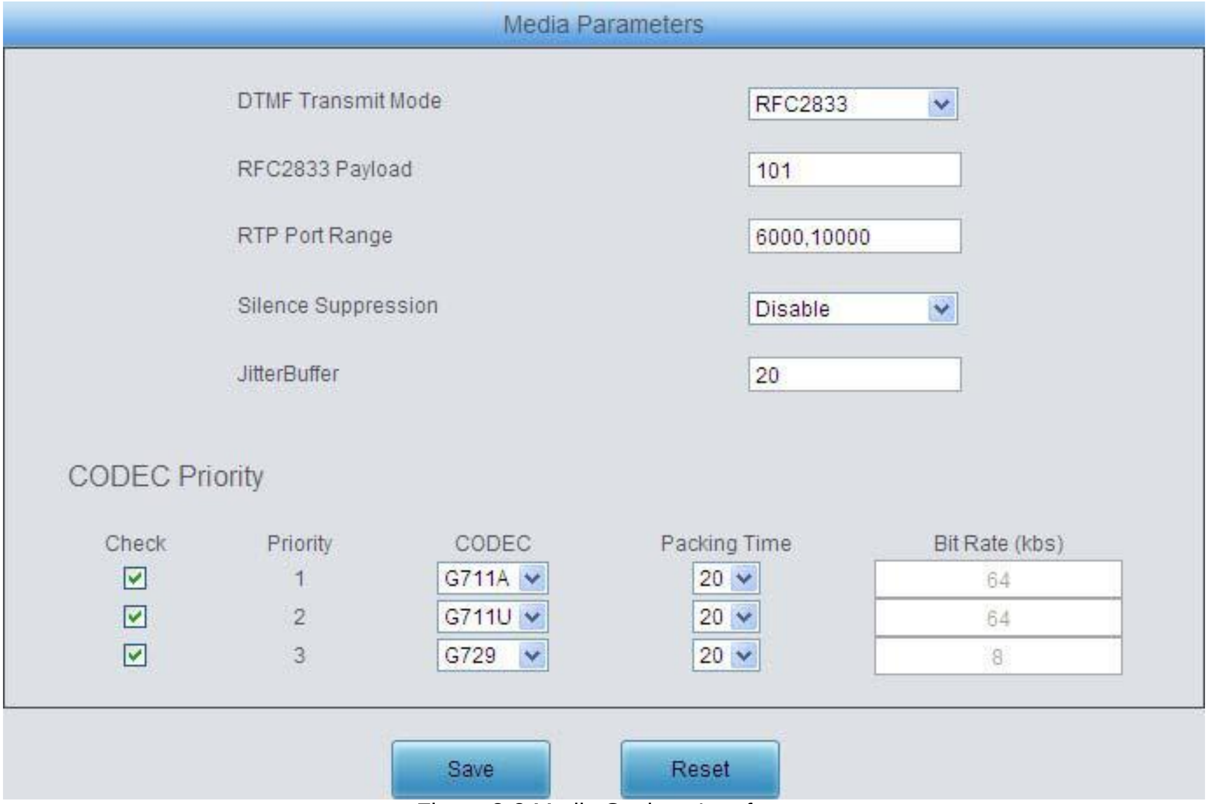

Figure 3-8 Media Settings Interface

See Figure 3-8 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 service, do it immediately to apply the changes. Refer to 3.8.8 Restart for detailed instructions. The table below explains the items shown in Figure 3-8.

Item	Description
DTMF Transmit Mode	Sets the transmit mode for the IP channel to send DTMF signals. The optional values are *RFC2833*, *In-band* and *Signaling*, with the default value of *RFC2833*.
RFC2833 Payload	Payload of the RFC2833 formatted DTMF signals on the IP channel. Range of value: 90~127, with the default value of 101.
RTP Port Range	Supported RTP port range for the IP end to establish a call conversation, with the lower limit of 2000 and the upper limit of 60000 and the difference between larger than 240. The default value is 6000-10000.
Silence Suppression	Sets whether to send comfort noise packets to replace RTP packets or never to send RTP packets to reduce the bandwidth usage when there is no voice signal throughout an IP conversation. The optional values are *Enable* and *Disable*, with the default value of *Disable*.
JitterBuffer	Acceptable jitter for data packets transmission over IP, which indicates the buffering capacity. A larger JitterBuffer means a higher jitter processing capability but as well as an increased voice delay, while a smaller JitterBuffer means a lower jitter processing capability but as well as a decreased voice delay. Range of value: 20~200, calculated by ms, with the default value of 20.

<table>
<tr>
<td>CODEC Priority</td>
<td>Supported CODECs and their corresponding priority for the IP end to establish a call conversation. The table below explains the sub-items:
<table>
<tr><th>Sub-item</th><th>Description</th></tr>
<tr><td>Priority</td><td>Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.</td></tr>
<tr><td>CODEC</td><td>Three optional CODECs are supported: G711A, G711U and G729AB.</td></tr>
<tr><td>Packing Time</td><td>Time interval for packing an RTP packet. Range of value: 20 or 30, calculated by ms, with the default value of 20.</td></tr>
<tr><td>Bit Rate</td><td>The number of thousand bits (excluding the packet header) that are conveyed per second. The bit rate to G711A/U is 64kbs and that to G729AB is 8kbs.</td></tr>
</table>
By default, all of the three CODECs are supported and ordered G711A, G711U and G729AB by priority from high to low.</td>
</tr>
</table>

3.4 Advanced Settings

Advanced Settings includes three parts: ***FXS/FXO***, ***Dialing Rule*** and ***Dialing Timeout***. See Figure 3-9. ***FXS/FXO*** is used to configure the general properties of the analog voice ports, such as the tone detection parameters and the conditions for sending the caller party information. ***Dialing Rule*** and ***Dialing Timeout*** are used to set the judging conditions for dialing.

Figure 3-9 Advanced Settings

3.4.1 FXS/FXO

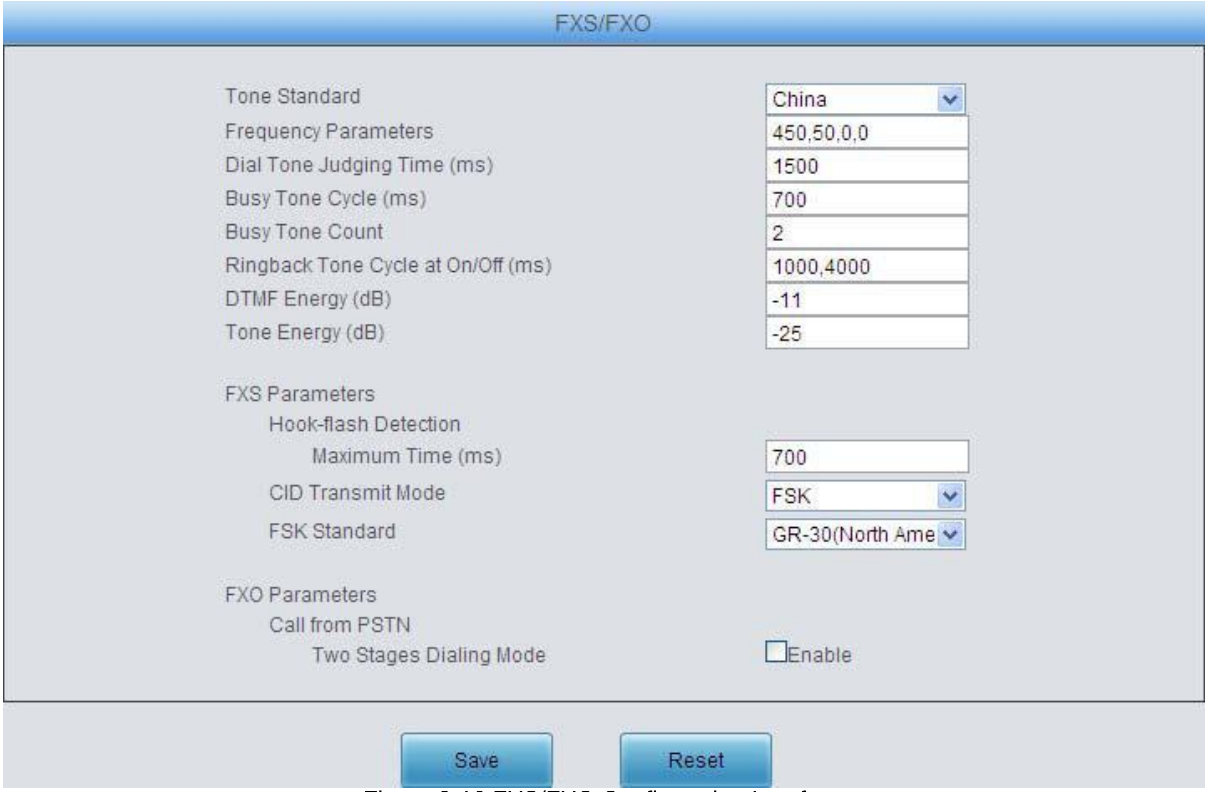

Figure 3-10 FXS/FXO Configuration Interface

See Figure 3-10 for the FXS/FXO configuration interface. The table below explains the items shown in the above figure.

<table>
<tr><th>Item</th><th>Description</th></tr>
<tr><td>Tone Standard</td><td>Standard for tone signals such as dialing tone and howler tone, which varies in different countries and districts. You can configure this parameter according to the actual location of the gateway. By default this item is set to China. (Currently, China is the only option available.)</td></tr>
<tr><td>Frequency Parameters</td><td>The value of this configuration item varies with Tone Standard. Also it can be modified. The table below explains the detailed meaning of the 4 parameters from left to right.
<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>1</td><td>The 1st center frequency. Range of value: 300~3400, calculated by Hz, with the default value of 450.</td></tr>
<tr><td>2</td><td>Allowable error of The 1st center frequency. Range of value: 10~50, calculated by ‰, with the default value of 50.</td></tr>
<tr><td>3</td><td>The 2nd center frequency. Range of value: 0 or 300~3400, calculated by Hz, with the default value of 0.</td></tr>
<tr><td>4</td><td>Allowable error of The 2nd center frequency. Range of value: 0 or 10~50, calculated by ‰, with the default value of 0.</td></tr>
</table>
Note: If it is not necessary to use the 2nd center frequency, you should set both the third and the forth parameters to 0 (default value); if only one of these two parameters is set to 0 and the other is set to other value, it will prompt error.</td></tr>
</table>

Dial Tone Judging Time	Minimum duration time for dialing tone, calculated by ms, with the default value of 1500 and the minimum value of 1300.
Busy Tone Cycle	Minimum duration time for busy tone. Range of value: 200~2000, calculated by ms, with the default value of 700. This configuration item together with ***Busy Tone Count*** judges whether a tone is busy tone or not.
Busy Tone Count	Minimum number of detected busy tone cycles for judging the hangup behavior of the remote end. Range of value: 1~10, with the default value of 2.
Ringback Tone Cycle at On/Off	Duration time at on and off states for judging whether a tone is ringback tone or not. Range of value: 300~2500 at on state, 800~6000 at off state, calculated by ms, with the default values of 1000 and 4000 respectively.
DTMF Energy	Energy of the DTMF signal sent by the gateway. Range of value: -35~15, calculated by dB, with the default value of -11.
Tone Energy	Energy of the tone signal sent by the gateway. Range of value: -35~15, calculated by dB, with the default value of -25.

After finishing the above general settings, you shall move on to configure the special parameters for FXS and FXO. The table below explains the particular configuration items for FXS.

Item	Description
Maximum Time	Time length for judging a flash operation. Only a hook-flash operation which lasts a time less than the value of this configuration item will be regarded as a valid flash operation. Those lasting a time longer than the value of this configuration item will be regarded as hangup operations. Range of value: 32~2000, calculated by ms, with the default value of 700.
CID Transmit Mode	The mode adopted by the FXS port to send the callerID. The optional values are *FSK* and *DTMF*, with the default value of *FSK*.
FSK Standard	Standard for sending FSK formatted callerID, which varies in different countries and districts. The optional values are: *ETSI (Europe)*, *GR-30 (North America, China)* and *NIT (Japan)*, with the default value of *GR-30*. This configuration appears only when ***CID Transmit Mode*** is set to *FSK*.

The table below explains the particular configuration items for FXO.

Item	Description
Two Stages Dialing Mode	Sets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.

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 service, do it immediately to apply the changes. Refer to 3.8.8 Restart for detailed instructions.

3.4.2 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.

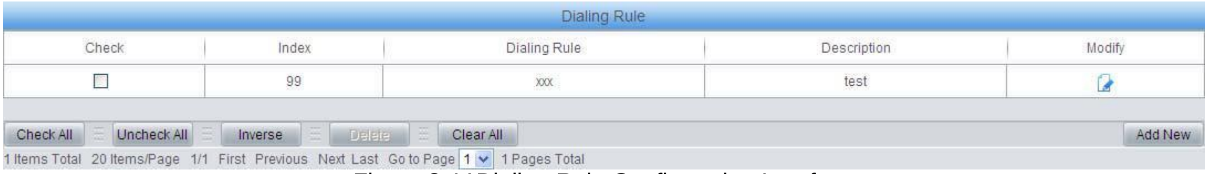

Figure 3-11Dialing Rule Configuration Interface

See Figure 3-11 for the dialing rule configuration interface. 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 3-12 for the dialing rule adding interface.

Figure 3-12 Add New Dialing Rule

The table below explains the items shown in Figure 3-12.

Item	Description
Index	The unique index of each dialing rule, which denotes its priority. A dialing rule with a smaller index value has a higher priority and will be checked earlier while matching.
Description	Remarks for the dialing rule. It can be any information, but can not be left empty.
Dialing Rule	The dialing rule can consist of the following characters: digits 0~9, letters A~D and x, punctuations '#' and '.'. The letter x indicates a random number; the punctuation '.' indicates a random amount (including zero) of characters after it; and the punctuation '#' indicates the end of the dialing operation. Up to 99 dialing rules can be configured in the gateway, and the maximum length of each dialing rule is 127 characters.

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

Click ***Modify*** in Figure 3-11 to modify the dialing rules. See Figure 3-13 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 3-13 Modify Dialing Rule

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

3.4.3 Dialing Timeout

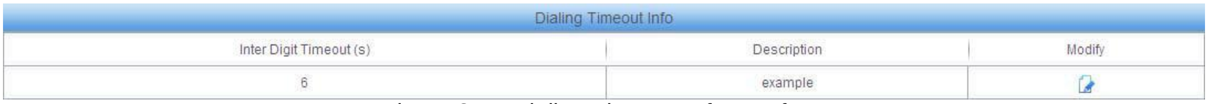

Figure 3-14 Dialing Timeout Info Interface

See Figure 3-14 for the dialing timeout info interface. The table below explains the items shown in the above figure.

Item	Description
	Sets the largest interval between two digits of a dialing number. All digits of a dialing number should be dialed one by one with intervals less than this value. If you do not ***Inter Digit Timeout*** dial a digit after previous ones until the time passing by goes longer than this value, all the previous digits you dialed will be regarded as a whole to constitute the dialing number. Range of value: 1~10, calculated by s, with the default value of 6.
Description	More information about the configuration item ***Inter Digit Timeout***, such as the reason for adopting the current value.

Click ***Modify*** in Figure 3-14 to modify the dialing timeout info. See Figure 3-15 for the dialing timeout info modification interface. The configuration items on this interface are the same as those on the ***Dialing Timeout Info Interface***.

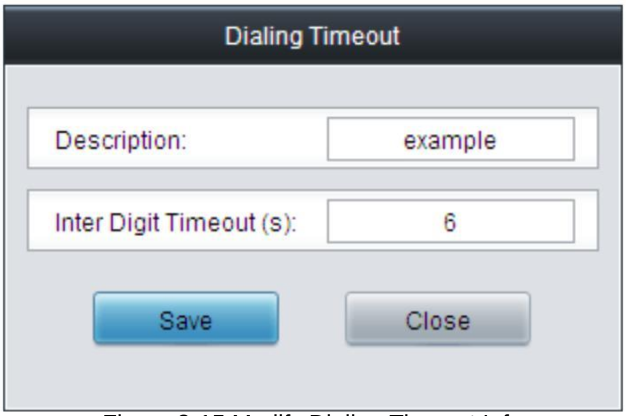

Figure 3-15 Modify Dialing Timeout Info

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

3.5 Port Settings

Port Settings includes two parts: ***Port*** and ***Port Group***. See Figure 3-16.

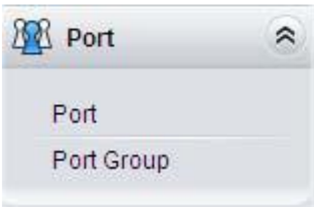

Figure 3-16 Port Settings

3.5.1 Port

Port Settings

Port	Type	SIP Account	Auto Dial	DND	Forward	FWD Type	FWD Num	CID	Call Waiting	Reg Status	Echo Canceller	Modify
1	FXS	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	
2	FXS	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	
3	FXO	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	
4	FXO	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	
5	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
6	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
7	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
8	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
9	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
10	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
11	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
12	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
13	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
14	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
15	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---
16	---	---	---	Disable	Disable	---	---	Disable	Disable	Failed	Enable	---

32 Items Total 16 Items/Page 1/2 First Previous Next Last Go to Page 1 2 Pages Total

Figure 3-17 Port Settings Interface

See Figure 3-17 for the port settings interface. The list in the above figure shows the type and properties of each port. The table below explains the items in Figure 3-17.

<table>
<tr><th>Item</th><th>Description</th></tr>
<tr><td>Port</td><td>Serial number of the port on the device. This item is not configurable.</td></tr>
<tr><td>Type</td><td>Type of the port on the device, either FXS or FXO. This item is not configurable. If it shows “---”, that means the port is unavailable due to the absence or damage of the corresponding module.</td></tr>
<tr><td>SIP Account</td><td>When the port initiates a call to SIP, this item corresponds to the username of SIP; when the port initiates a call to PSTN, this item corresponds to the displayed callerID.</td></tr>
<tr><td>Auto Dial</td><td>This number will be automatically dialed by the FXS port if there is no dialing operation after pickup within a designated time period (i.e. Wait Time before Auto Dial). The parameter Wait Time before Auto Dial can be configured on the FXS port modification interface.</td></tr>
<tr><td>DND</td><td>Do Not Disturb. If this feature is enabled, the FXS port will reply the 403 message to reject all incoming calls.</td></tr>
<tr><td>Forward</td><td>The automatic call forward feature for the FXS port. Once this feature is enabled, the FXS port will forward incoming IP calls according to the preset FWD Type.
Note: To enable this feature, do not put the FXS port into a port group with other ports.</td></tr>
<tr><td>FWD Type</td><td>Forward conditions for the FXS port to forward incoming IP calls. The optional values are:
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Unconditional</td><td>The FXS port will forward all incoming IP calls to the preset FWD Num immediately when it receives them.</td></tr>
<tr><td>Busy</td><td>The FXS port will forward incoming IP calls to the preset FWD Num if busy when it receives them.</td></tr>
<tr><td>No Reply</td><td>The FXS port will forward incoming IP calls to the preset FWD Num if the corresponding station does not answer them in a designated time period (i.e. Time for No Reply Forward). Only when this forward condition is selected does the configuration item Time for No Reply Forward on the FXS port modification interface become valid.</td></tr>
</table>
This item can be configured on the FXS port modification interface and is valid only when Forward is set to Enable.</td></tr>
<tr><td>FWD Num</td><td>The number to which the incoming IP call is forwarded. If the Forward feature is enabled, this item should not be left empty.</td></tr>
<tr><td>CID</td><td>CallerID. If this feature is enabled, the FXS port will send the callerID of the incoming IP call together with the ringing tone to the corresponding station.</td></tr>
<tr><td>Call Waiting</td><td>If this feature is enabled, the FXS port while in conversation can accept another call from IP and keep it in the waiting state. Once the current conversation is finished and the station hangs up, the call in the waiting state will ring the station and wait for answer.</td></tr>
</table>

Reg Status	Registration status of the port. When ***Register Port*** is set to *No*, the value of this item is *Unregistered*; when ***Register Port*** is set to *Yes*, the value of this item may be *Failed* or *Registered*.
Echo Canceller	The echo cancellation feature for a call conversation over the FXS/FXO channel. By default, this feature is enabled and the effect can reach 128ms.

Click ***Modify*** in Figure 3-17 to modify the properties of the corresponding port. The FXO port and the FXS port have different properties as they are different in functions. See Figure 3-18 and Figure 3-19 for the FXO port modification interface and the FXS port modification interface respectively. Most configuration items on these two interfaces are the same as those on the ***Port Settings Interface***.

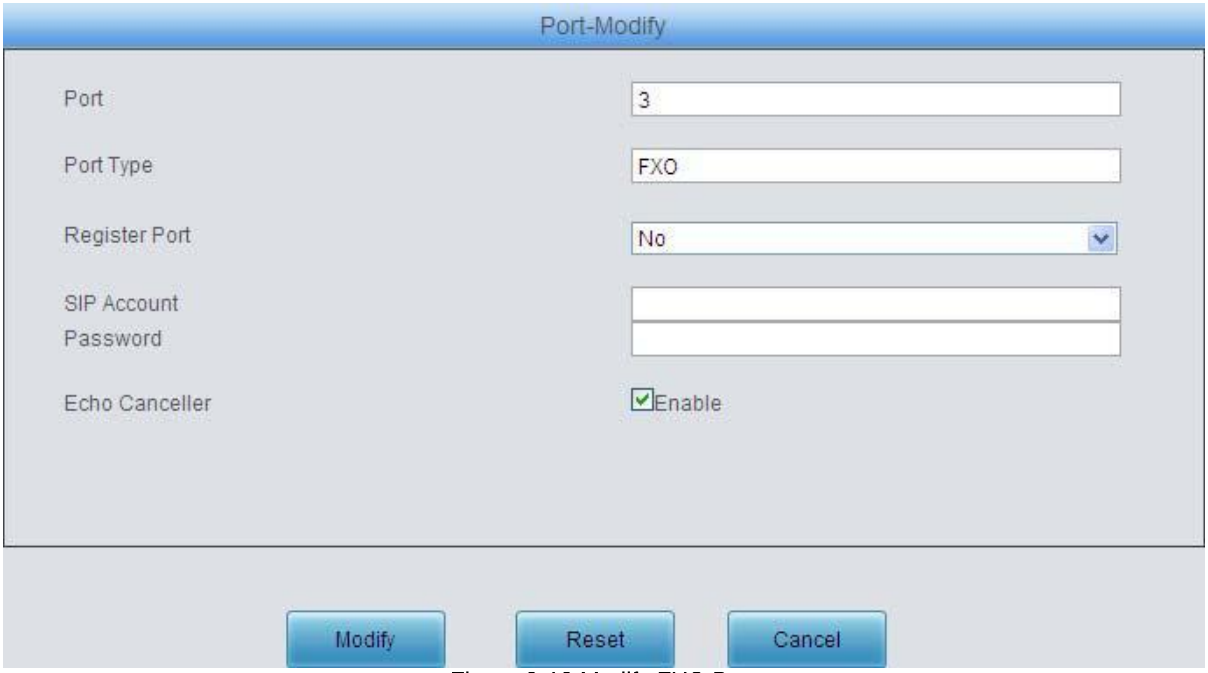

Figure 3-18 Modify FXO Port

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

Item	Description
Register Port	To register the port to the SIP server.
Password	Registration password of the port. To register the port to the SIP server, both configuration items ***SIP Account*** and ***Password*** should be filled in.

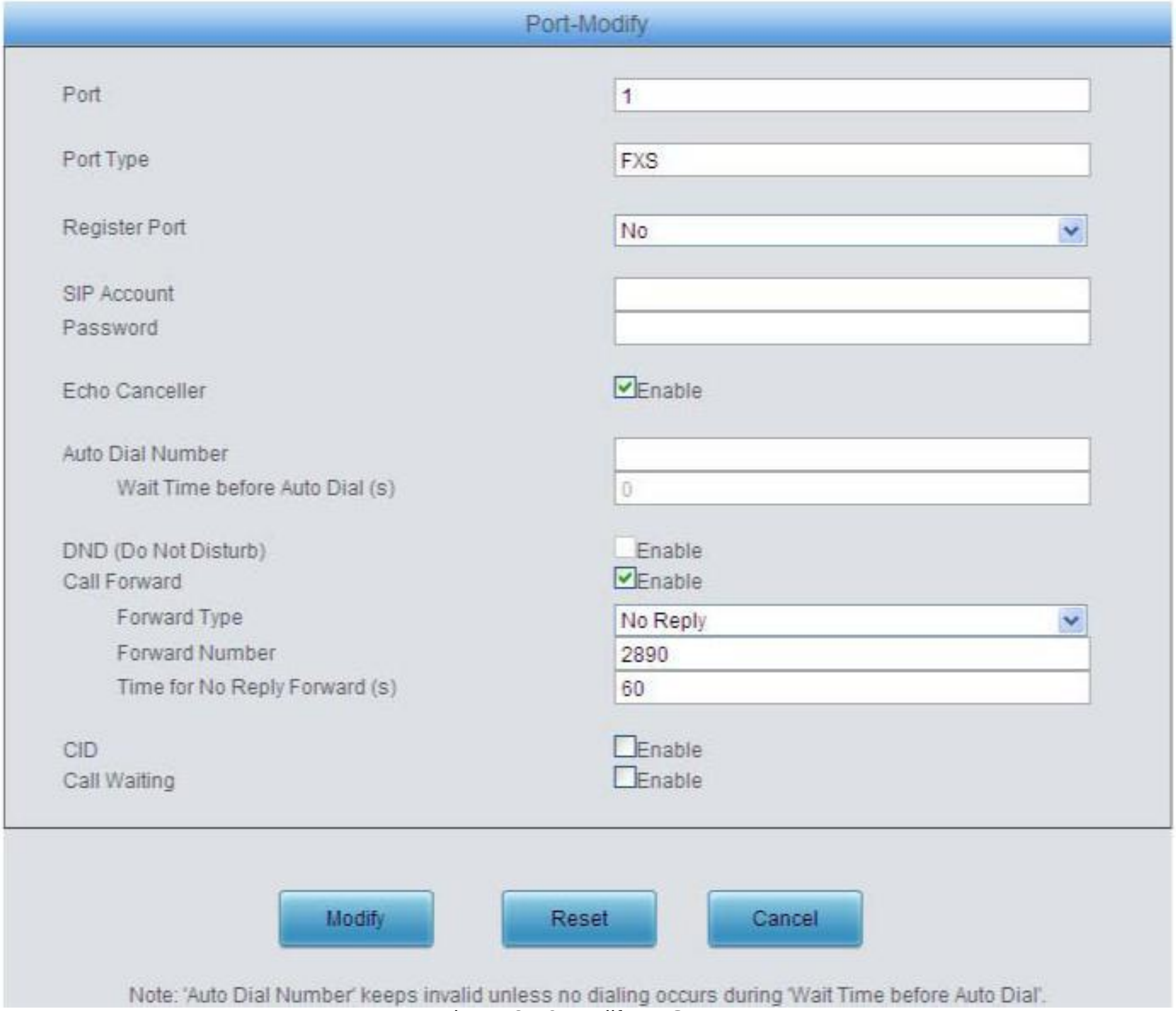

Figure 3-19 Modify FXS Port

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

Item	Description
Register Port	To register the port to the SIP server.
Password	Registration password of the port. To register the port to the SIP server, both configuration items ***SIP Account*** and ***Password*** should be filled in.
Wait Time before Auto Dial	This configuration item is valid only when the ***Auto Dial Number*** is not left empty. If there is no dialing operation after pickup within the time period set by this item, the port will automatically call the ***Auto Dial Number***.
Time for No Reply Forward	This configuration item is valid only when the ***Call Forward*** feature is enabled and the ***Forward Type*** is set to *No Reply*. If the corresponding station does not pick up the incoming call within the time period set by this item, the port will forward the call to the preset ***Forward Number***.

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

3.5.2 Port Group

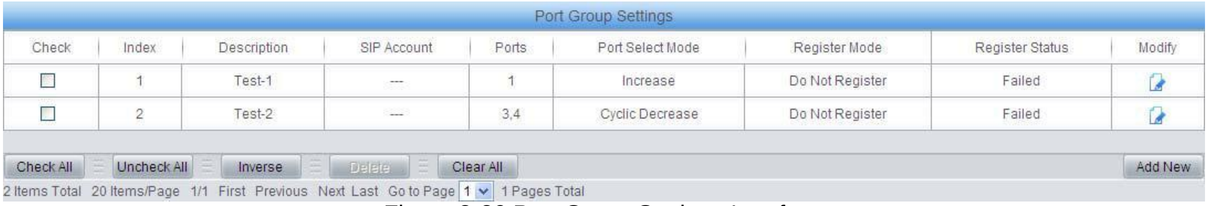

Figure 3-20 Port Group Settings Interface

See Figure 3-20 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 ***Registration 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 3-21 for the port group adding interface. Note that a port which has been occupied by one port group cannot be chosen by others.

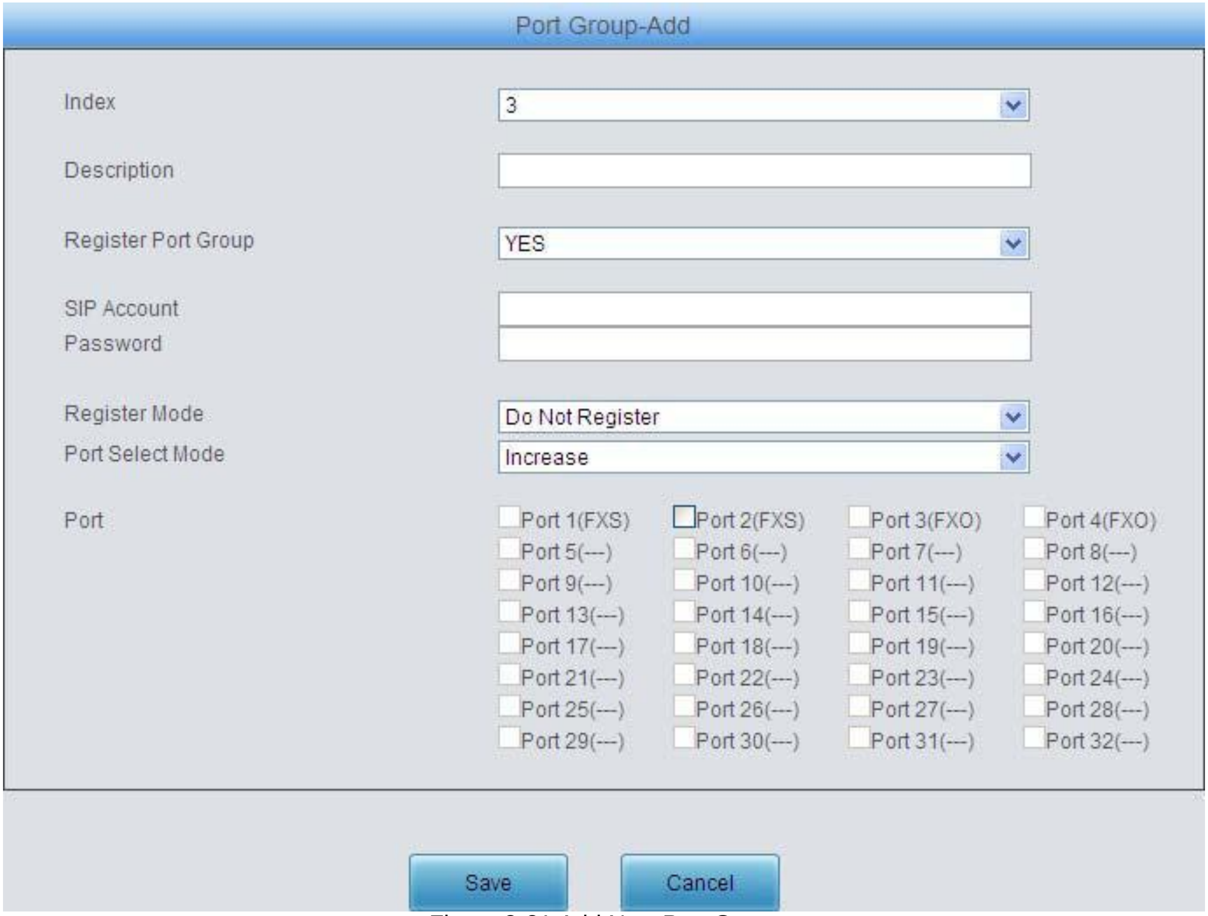

Figure 3-21 Add New Port Group

The table below explains the items in the above figure.

Item	Description
Index	The unique index of each port group, which is mainly used in the configuration of routing rules and number manipulation rules to correspond to port groups.
Description	More information about each port group.
Register Port Group	To register the port group to the SIP server. Only when this configuration item is set to *Yes* can you see the configuration items ***SIP Account*** and ***Password***.

<table>
<tr><td>SIP Account</td><td>When the port group initiates a call to SIP, this item corresponds to the username of SIP; when the port group initiates a call to PSTN, this item corresponds to the displayed callerID.</td></tr>
<tr><td>Password</td><td>Registration password of the port group. To register the port group to the SIP server, both configuration items SIP Account and Password should be filled in.</td></tr>
<tr><td>Register Mode</td><td>Sets the way for SIP to make outgoing calls (TEL→IP) on the gateway.
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Do Not Register</td><td>SIP initiates a call in a point-to-point mode.</td></tr>
<tr><td>Register Gateway</td><td>SIP initiates a call with the registered SIP account and password of the whole gateway. (Refer to 3.3.1 SIP Settings for gateway registration.)</td></tr>
<tr><td>Register Port Group</td><td>SIP initiates a call with the registered SIP account and password of the port group.</td></tr>
<tr><td>Register Port</td><td>SIP initiates a call with the registered SIP account and password of the port.</td></tr>
</table></td></tr>
<tr><td>Register Status</td><td>Registration status of the port group. When Register Port Group is set to No, the value of this item is Unregistered; when Register Port Group is set to Yes, the value of this item may be Failed or Registered.</td></tr>
<tr><td>Port Select Mode</td><td>When the port group receives a call, it will choose a port based on the select mode set by this configuration item to ring or to connect. The optional values and their corresponding meanings are described in the table below.
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Increase</td><td>Search for an idle port in the ascending order of the port number, starting from the minimum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Decrease</td><td>Search for an idle port in the descending order of the port number, starting from the maximum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Cyclic Increase</td><td>Provided Port N is the available port found last time. Search for an idle port in the ascending order of the port number, starting from Port N+1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Cyclic Decrease</td><td>Provided Port N is the available port found last time. Search for an idle port in the descending order of the port number, starting from Port N-1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
</table></td></tr>
</table>

Port	The ports in the port group. If the checkbox before a port is grey, it indicates that the port is not available or has been occupied. All selected ports for a port group will be displayed in the ***Ports*** column in Figure 3-20. Note: When a port group contains multiple ports, the automatic call forward feature is invalid.

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

Click ***Modify*** in Figure 3-20 to modify the properties of a port group. See Figure 3-22 for the port group modification interface. The configuration items on this interface are the same as those on the ***Add New Port Group*** interface.

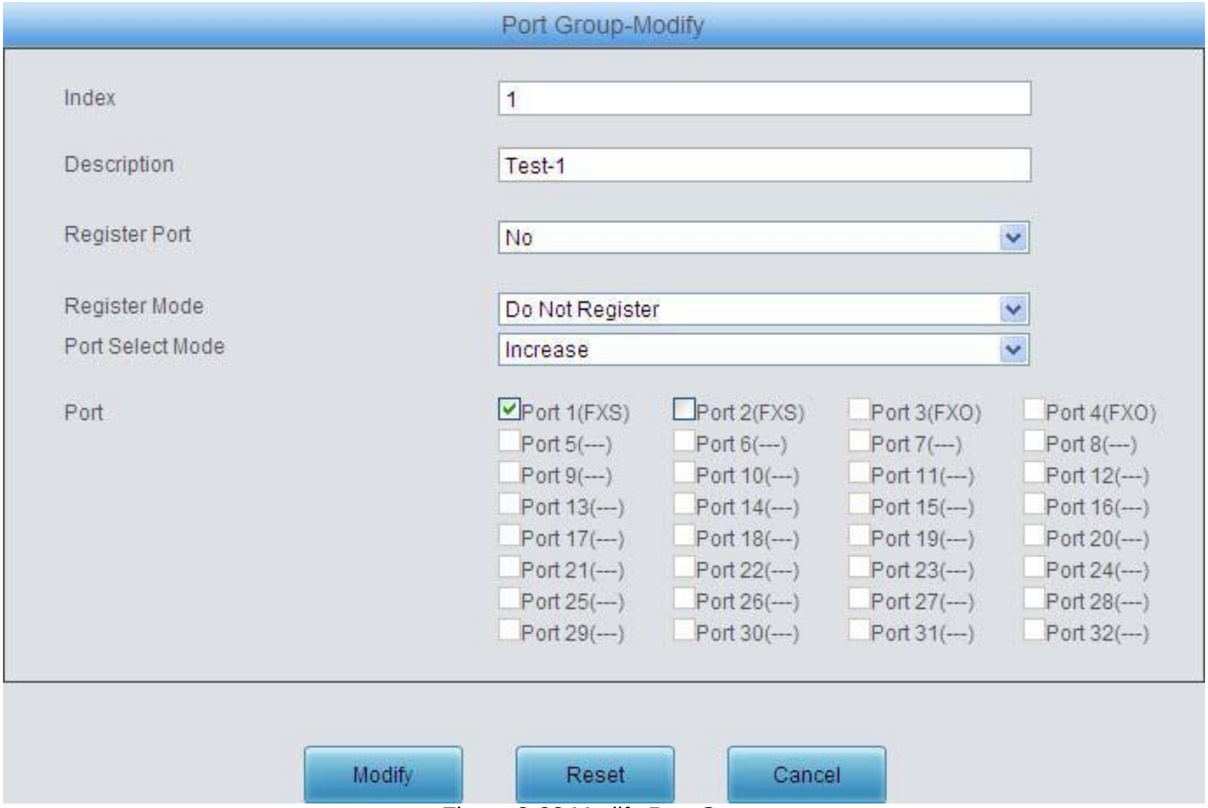

Figure 3-22 Modify Port Group

To delete a port group, check the checkbox before the corresponding index in Figure 3-20 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 3-20.

3.6 Route Settings

Route Settings is used to specify the routing rules for calls on two directions: IP→Tel and Tel→IP. See Figure 3-23.

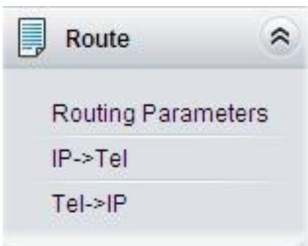

Figure 3-23 Route Settings

3.6.1 Routing Parameters

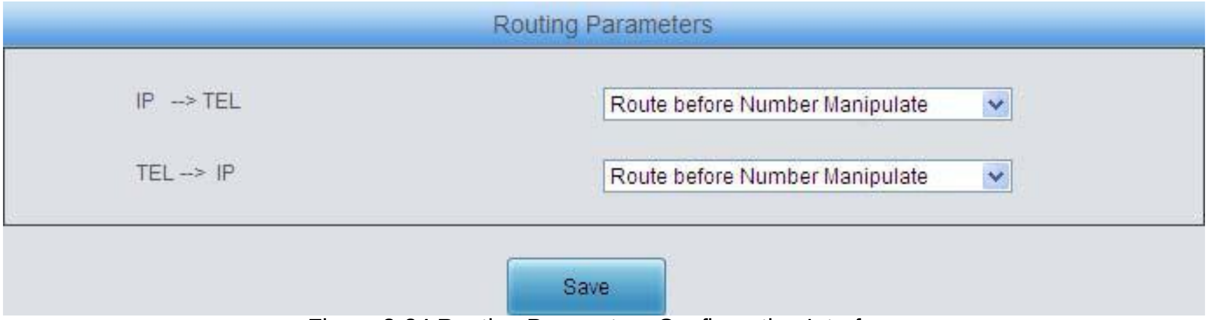

Figure 3-24 Routing Parameters Configuration Interface

See Figure 3-24 for the routing parameters configuration interface. On this interface, you can set the routing rules for calls respectively on two directions IP→Tel 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.

3.6.2 IP to Tel

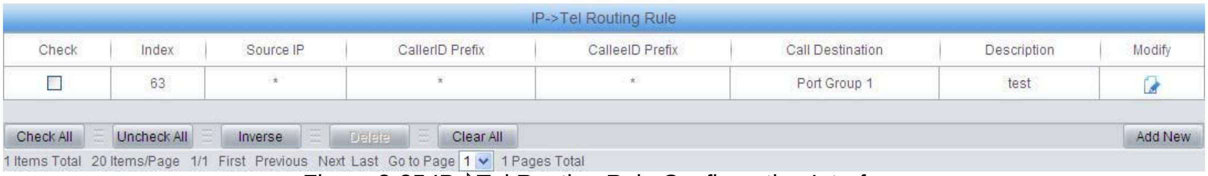

Figure 3-25 IP→Tel Routing Rule Configuration Interface

See Figure 3-25 for the IP→Tel routing rule configuration interface. A new routing rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-26 for the IP→Tel routing rule adding interface. Don't leave 'Description' empty and you may use the default values of all the other configuration items herein.

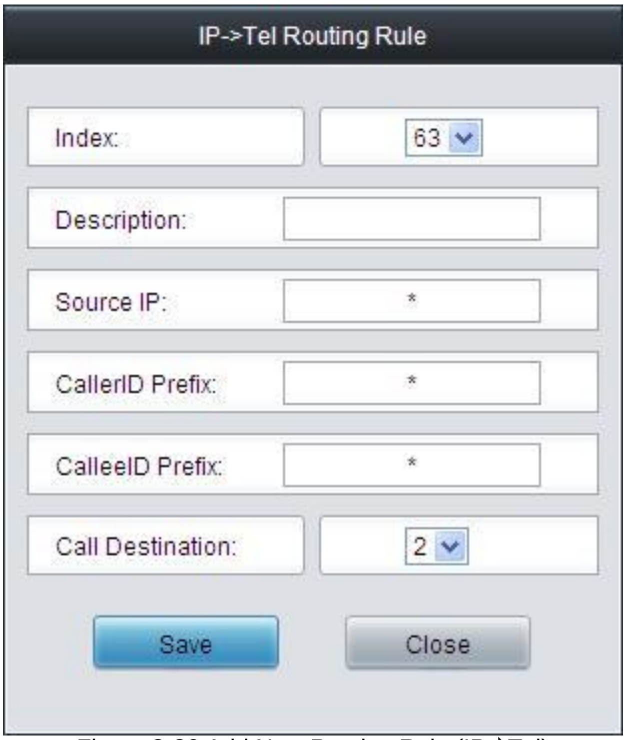

Figure 3-26 Add New Routing Rule (IP→Tel)

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

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with a smaller index value has a higher priority. If a call matches several routing rules, it will be processed according to the one with the highest priority.
Description	More information about each routing rule.
Source IP	IP address from where the call is initiated. This item can be set to a specific IP address or "*" which indicates any IP address
CallerID Prefix, CalleeID Prefix	A string of numbers at the beginning of the caller/called party number. This item can be set to a specific string or "*" which indicates any string. These two configuration items together with ***Source IP*** can specify the calls which apply to a routing rule.
Call Destination	Port group 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.

Click ***Modify*** in Figure 3-25 to modify a routing rule. See Figure 3-27 for the IP→Tel routing rule modification interface. The configuration items on this interface are the same as those on the ***Add New Routing Rule (IP→Tel)*** interface. Note that the item ***Index*** cannot be modified.

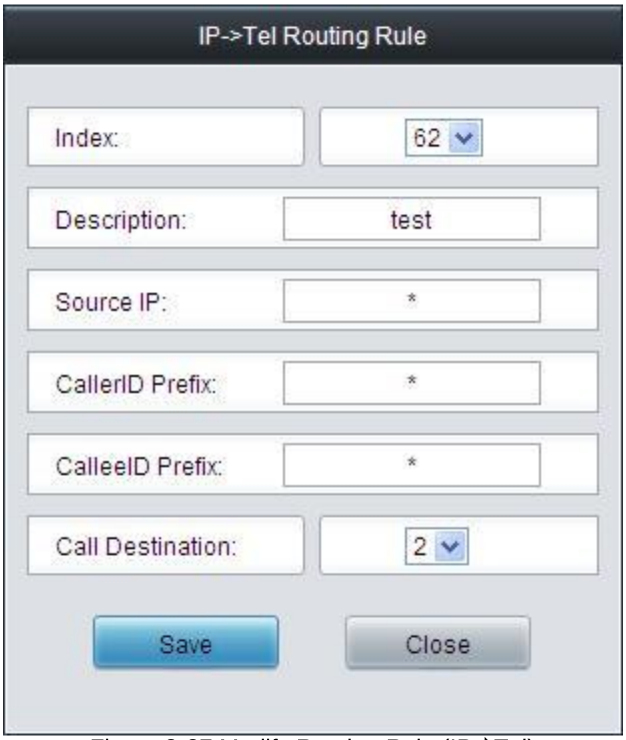

Figure 3-27 Modify Routing Rule (IP→Tel)

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-25 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 3-25.

3.6.3 Tel to IP

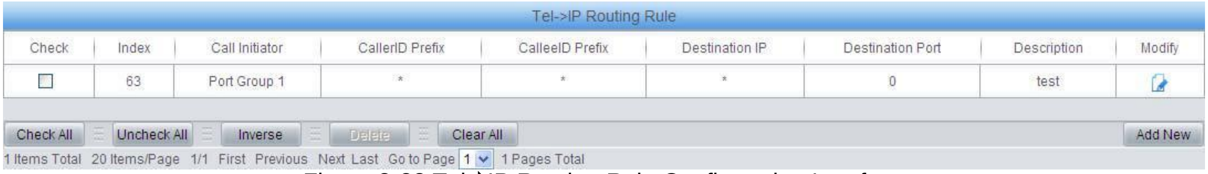

Figure 3-28 Tel→IP Routing Rule Configuration Interface

See Figure 3-28 for the Tel→IP routing rule configuration interface. A new routing rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-29 for the Tel→IP routing rule adding interface. You may use the default values of all configuration items herein except for ***Description***, ***Destination IP*** and ***Destination Port***.

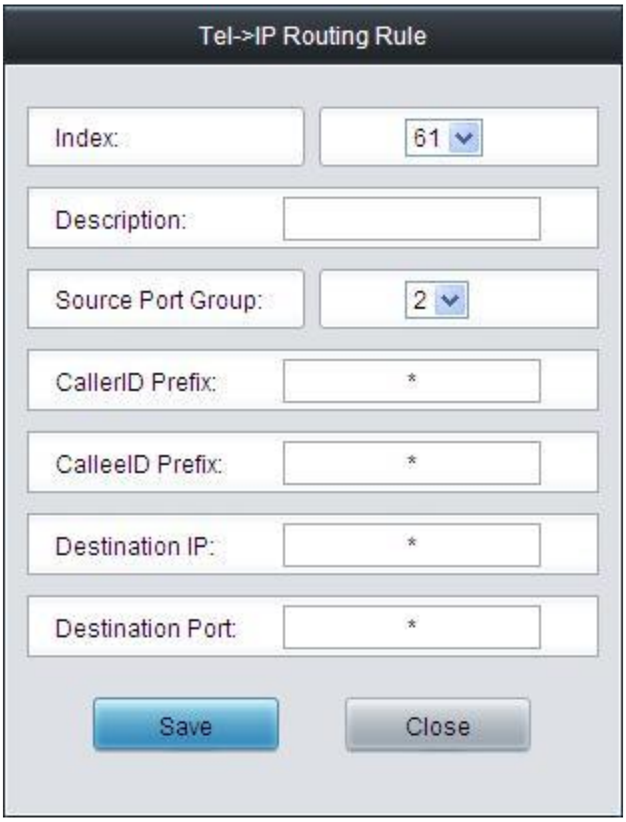

Figure 3-29 Add New Routing Rule (Tel→IP)

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

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with a smaller index value has a higher priority. If a call matches several routing rules, it will be processed according to the one with the highest priority.
Description	More information about each routing rule.
Source Port Group (Call Initiator)	Port group from which the call is initiated.
CallerID Prefix, CalleeID Prefix	A string of numbers at the beginning of the caller/called party number. This item can be set to a specific string or "*" which indicates any string. These two configuration items together with ***Source Port Group (Call Initiator)*** can specify the calls which apply to a routing rule.

Destination IP, ***Destination Port***	IP address and port number of the remote end to which the call will be routed.

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

Click ***Modify*** in Figure 3-28 to modify a routing rule. See Figure 3-30 for the Tel→IP routing rule modification interface. The configuration items on this interface are the same as those on the ***Add New Routing Rule (Tel→IP)*** interface. Note that the item ***Index*** cannot be modified.

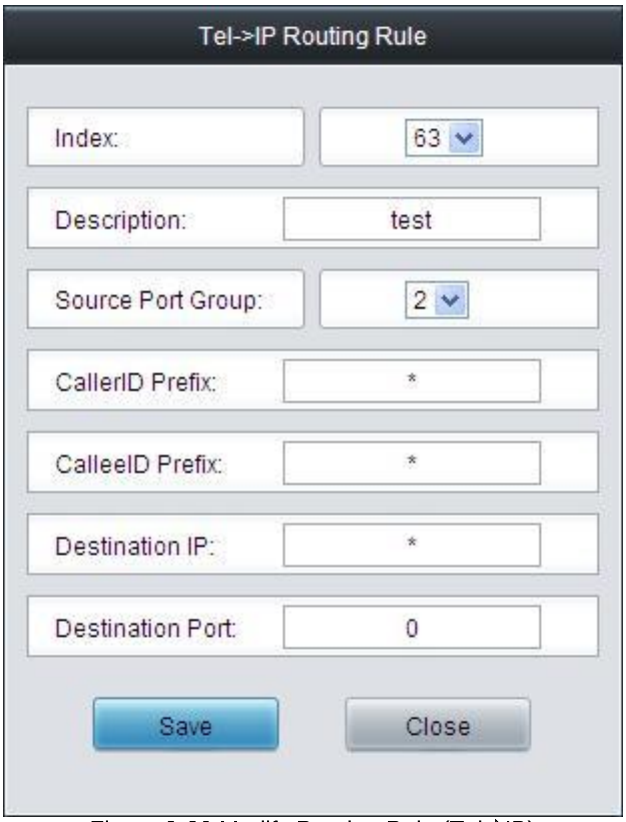

Figure 3-30 Modify Routing Rule (Tel→IP)

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-28 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 3-28.

3.7 Number Manipulation

Number Manipulation includes four parts: **IP→Tel CallerID**, **IP→Tel CalleeID**, **Tel→IP CallerID** and **Tel→IP CalleeID**. See Figure 3-31.

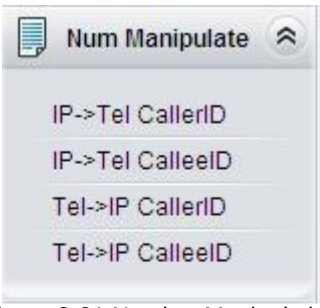

Figure 3-31 Number Manipulation

3.7.1 IP to Tel CallerID

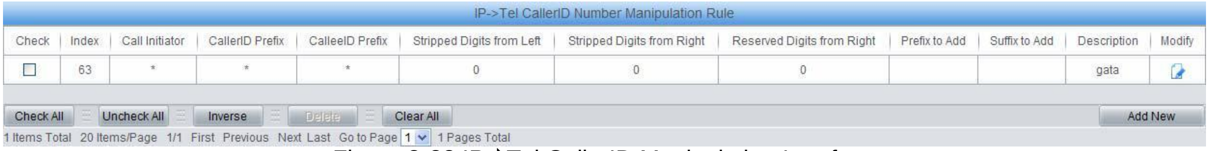

Figure 3-32 IP→Tel CallerID Manipulation Interface

See Figure 3-32 for the IP→Tel CallerID manipulation interface. 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 3-33 for the IP→Tel CallerID manipulation rule adding interface. Don't leave ***Description*** empty and you may use the default values of all the other configuration items herein.

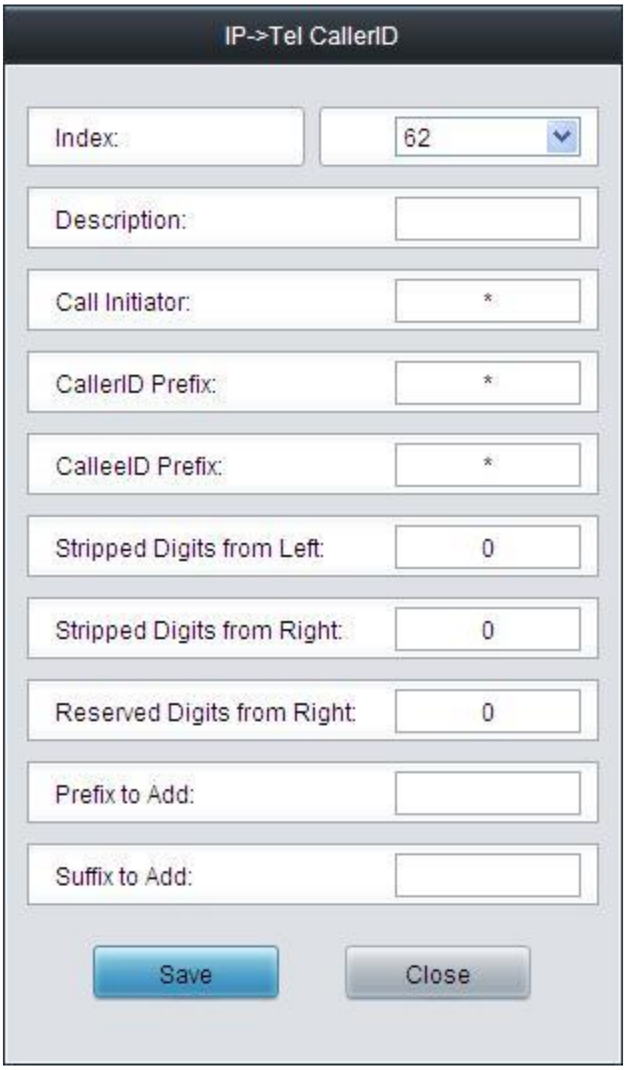

Figure 3-33 Add IP→Tel CallerID Manipulation Rule

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

Item	Description
Index	The unique index of each number manipulation rule, which denotes its priority. A number manipulation rule with a smaller index value has a higher priority. If a call matches several number manipulation rules, it will be processed according to the one with the highest priority.
Description	More information about each number manipulation rule.

Call Initiator	IP address from where the call is initiated. This item can be set to a specific IP address or "*" which indicates any IP address.
CallerID Prefix, CalleeID Prefix	A string of numbers at the beginning of the caller/called party number. This item can be set to a specific string or "*" which indicates any string. These two configuration items together with ***Call Initiator*** can specify the calls which apply to a number manipulation rule.
Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

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

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

Click ***Modify*** in Figure 3-32 to modify a number manipulation rule. See Figure 3-34 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.

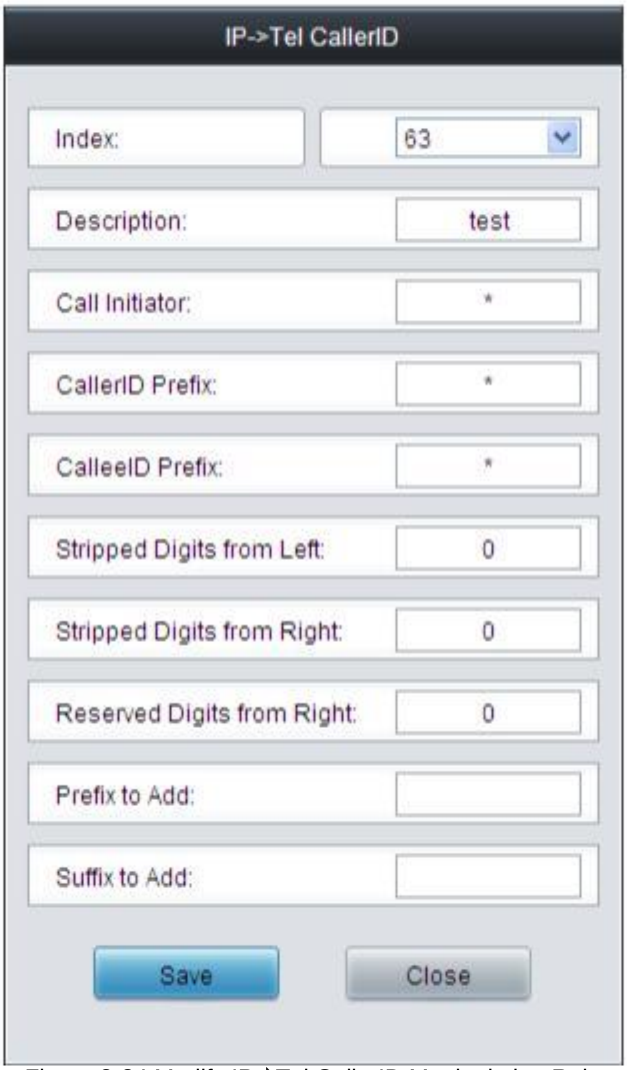

Figure 3-34 Modify IP→Tel CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-32 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 3-32.

3.7.2 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 3-35 for IP→Tel CalleeID manipulation interface. The configuration items on this interface are the same as those on ***IP→Tel CallerID Manipulation Interface*** (Figure 3-32).

Figure 3-35 IP→Tel CalleeID Manipulation Interface

3.7.3 Tel to IP CallerID

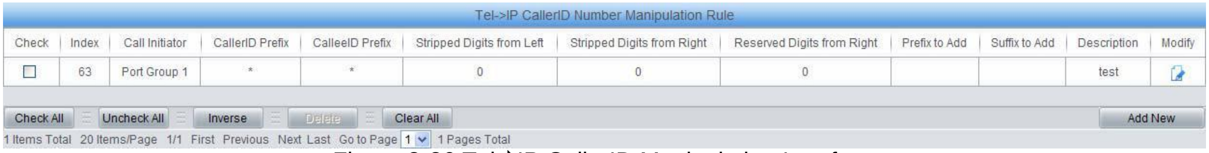

Figure 3-36 Tel→IP CallerID Manipulation Interface

See Figure 3-36 for the Tel→IP CallerID manipulation interface. 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 3-37 for the Tel→IP CallerID manipulation rule adding interface. Don't leave ***Description*** empty and you may use the default values of all the other configuration items herein.

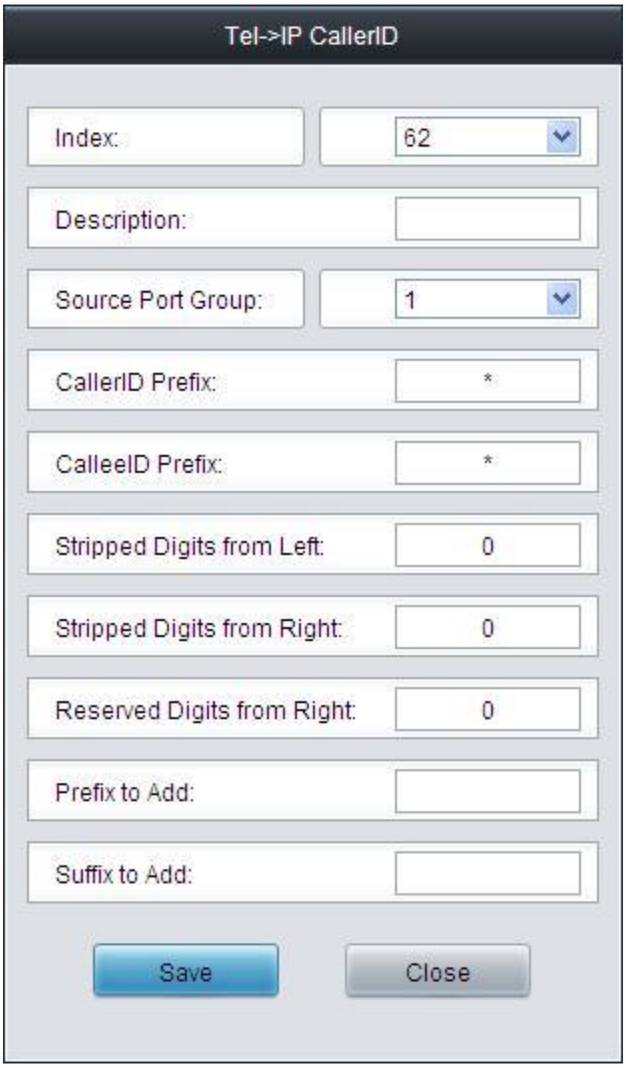

Figure 3-37 Add Tel→IP CallerID Manipulation Rule

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

Item	Description
Index	The unique index of each number manipulation rule, which denotes its priority. A number manipulation rule with a smaller index value has a higher priority. If a call matches several number manipulation rules, it will be processed according to the one with the highest priority.
Description	More information about each number manipulation rule.

Source Port Group (Call Initiator)	Port group from which the call is initiated.
CallerID Prefix, CalleeID Prefix	A string of numbers at the beginning of the caller/called party number. This item can be set to a specific string or "*" which indicates any string. These two configuration items together with ***Call Initiator*** can specify the calls which apply to the number manipulation rule.
Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

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

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

Click ***Modify*** in Figure 3-36 to modify a number manipulation rule. See Figure 3-38 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.

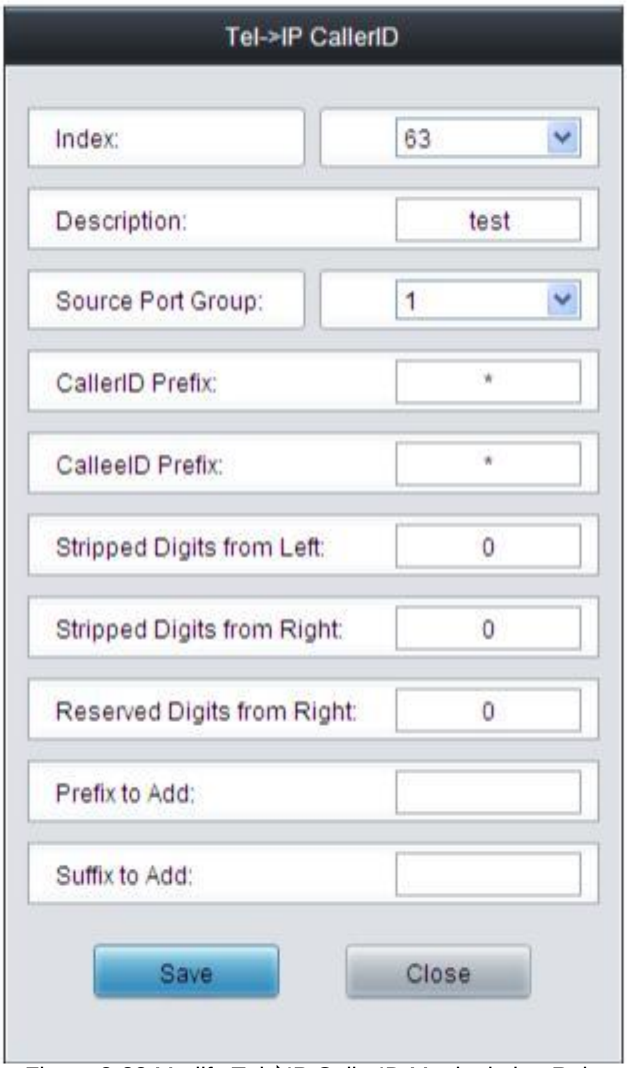

Figure 3-38 Modify Tel→IP CallerID Manipulation Rule

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

3.7.4 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 3-39 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 3-36).

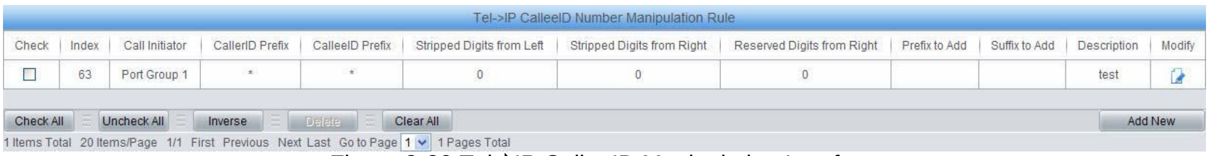

Figure 3-39 Tel→IP CalleeID Manipulation Interface

3.8 System Tools

System Tools is mainly for gateway maintenance. It provides such features as IP modification, time synchronization, data backup, log inquiry and connectivity check. See Figure 3-40 for details.

Figure 3-40 System Tools

3.8.1 Network

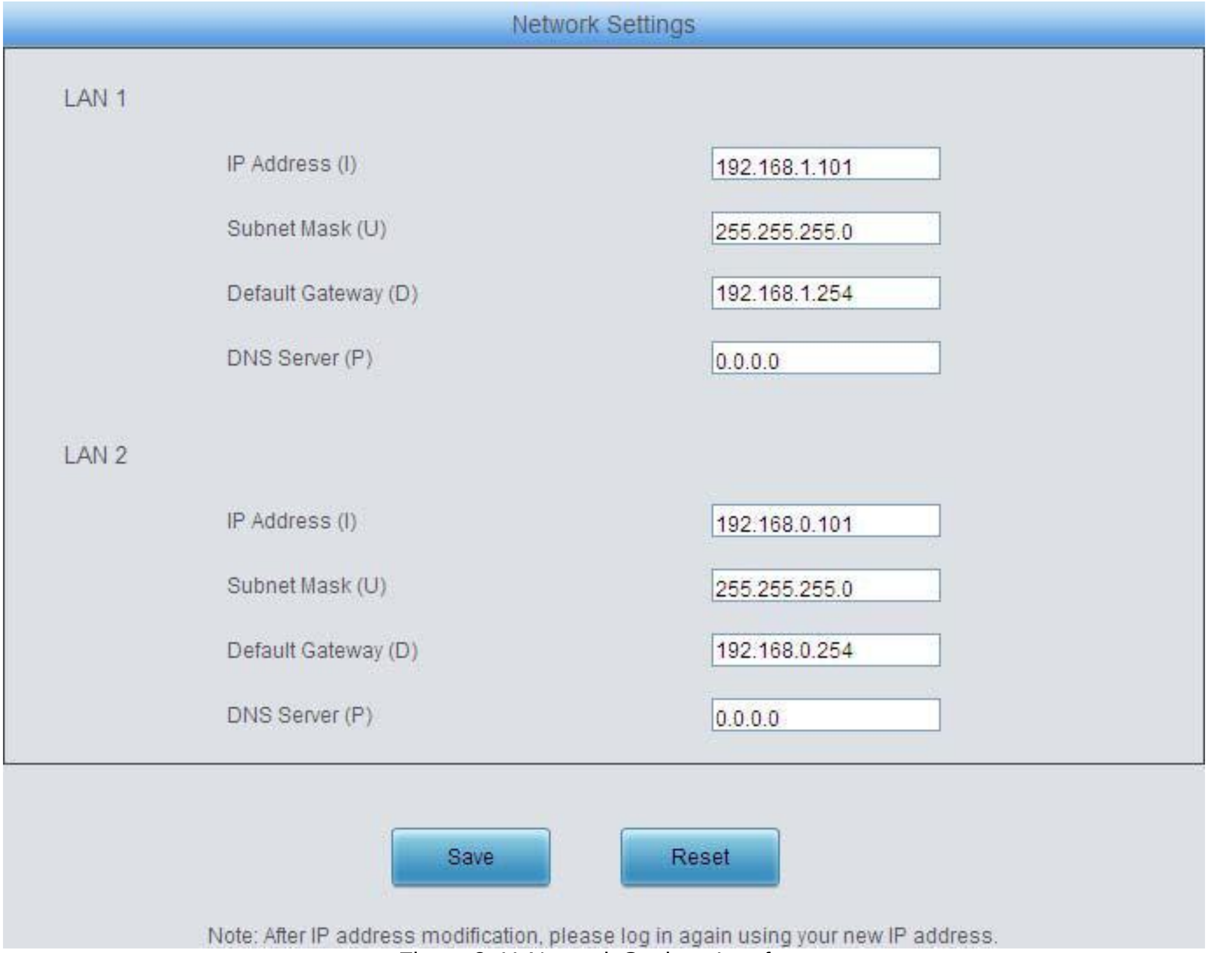

Figure 3-41 Network Settings Interface

See Figure 3-41 for the network settings interface. A gateway has two LANs, each of which can be configured with independent IP address, subnet mask, default gateway and DNS server. **Note: The two configuration items IP Address and Default Gateway cannot be the same for LAN 1 and LAN 2.**

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.

Note: The values of the *IP address*, *Subnet Mask*, *Default Gateway* and *DNS Server* shown in Figure 3-41 are all factory default settings.

3.8.2 Time

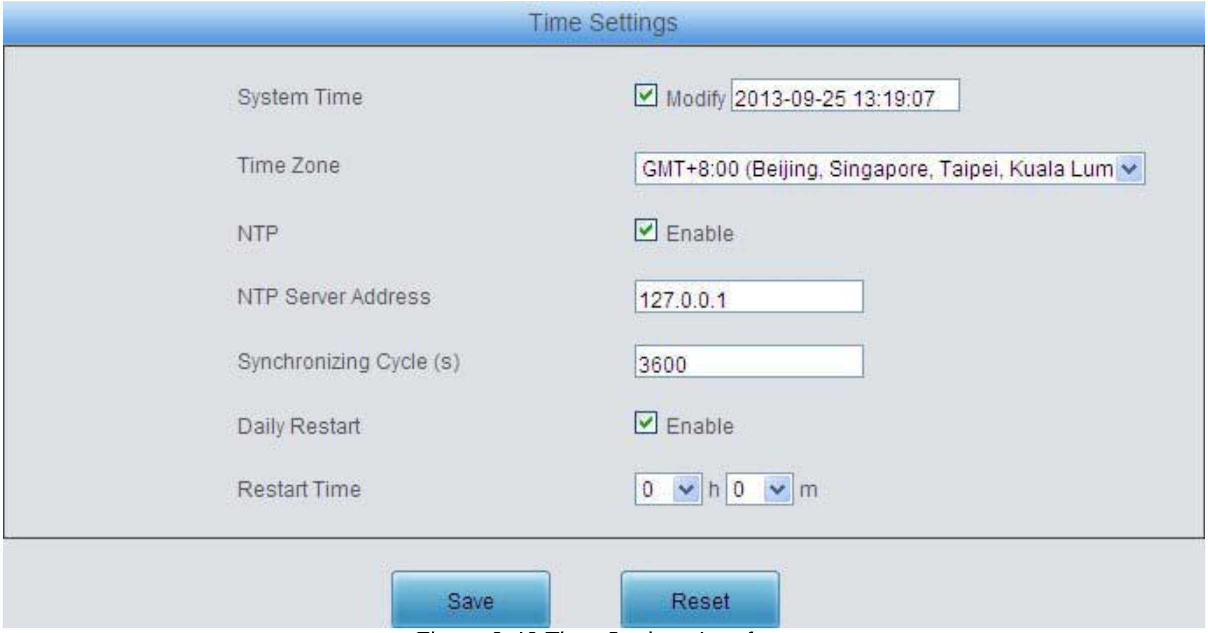

Figure 3-42 Time Settings Interface

See Figure 3-42 for the time settings interface where you can modify the system time manually or enable the NTP time synchronization feature. To set the time manually, check the checkbox for ***System Time*** and then modify the time in the edit box behind. To synchronize the system time with the NTP server, check the checkbox for ***NTP*** and configure the items ***NTP Server Address***, ***Synchronizing Cycle*** and ***Time Zone***. By default, ***NTP*** is disabled.

Daily Restart means to restart the gateway regularly every day at the preset ***Restart Time***. By default, this feature is disabled.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Reset*** to restore the configurations. The time settings will not go into effect until you save them into the gateway.

3.8.3 Upgrade

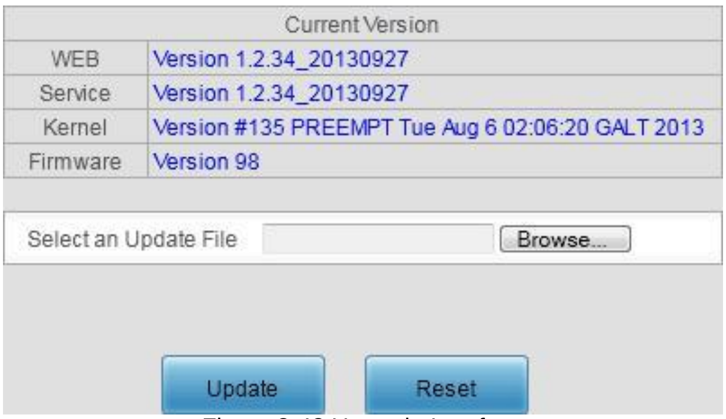

Figure 3-43 Upgrade Interface

See Figure 3-43 for the upgrade interface where you can upgrade the WEB, gateway service, kernel and firmware to new versions. Select the upgrade package "*.pack" via ***Browse…*** and click ***Update***. Wait for a while and the gateway will finish the upgrade automatically. Note that clicking ***Reset*** can only delete the selected update file but not cancel the operation of ***Update***.

3.8.4 System Log

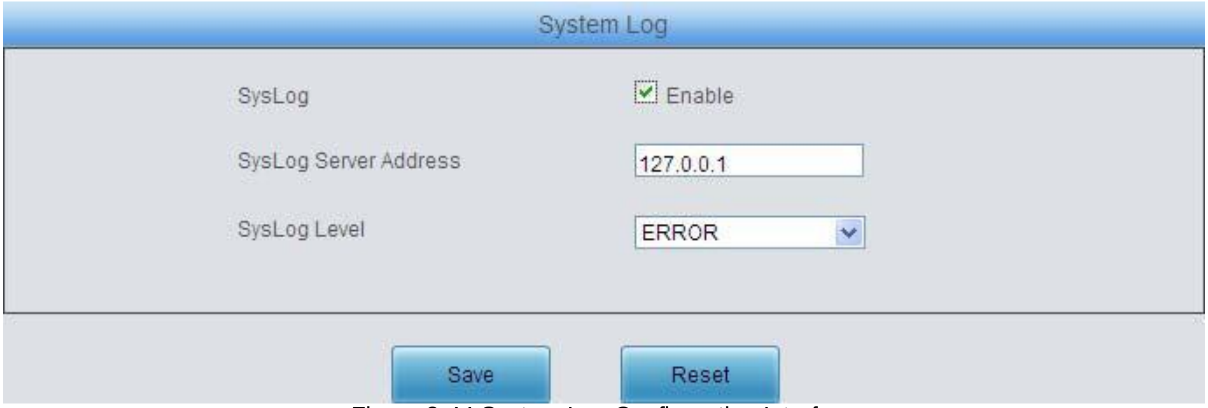

Figure 3-44 System Log Configuration Interface

See Figure 3-44 for the system log configuration interface. The logs shall be saved to a designated SysLog server as the gateway has no space for them. To enable this feature, you should check the checkbox for ***SysLog*** and configure the items ***SysLog Server Address*** and ***SysLog Level***. There are three log levels for you to choose:

Log Level	Description
ERROR	Error message
WARNING	Warning message
INFO	General information

After configuration, click ***Save*** to save the above settings into the gateway or click ***Reset*** to restore the configurations. By default, ***Syslog*** is disabled.

3.8.5 Change Password

Figure 3-45 Password Changing Interface

See Figure 3-45 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.

3.8.6 Backup & Upload

Figure 3-46 Backup & Upload Interface

See Figure 3-46 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***. The gateway will automatically apply the uploaded data to overwrite the current configurations.

3.8.7 Factory Reset

Figure 3-47 Factory Reset Interface

See Figure 3-47 for the factory reset interface. Click ***Reset*** to restore all configurations on the gateway to factory settings.

3.8.8 Restart

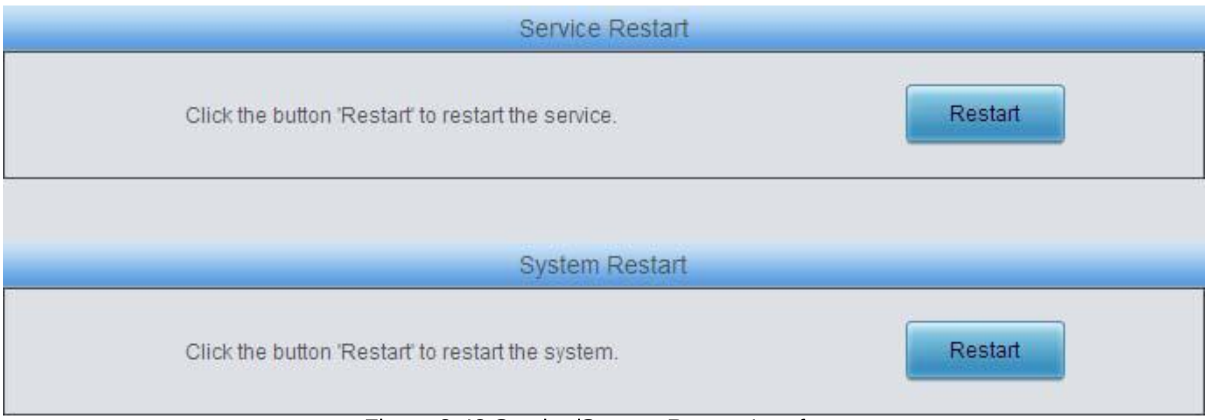

Figure 3-48 Service/System Restart Interface

See Figure 3 -48 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.

3.8.9 SNMP Config

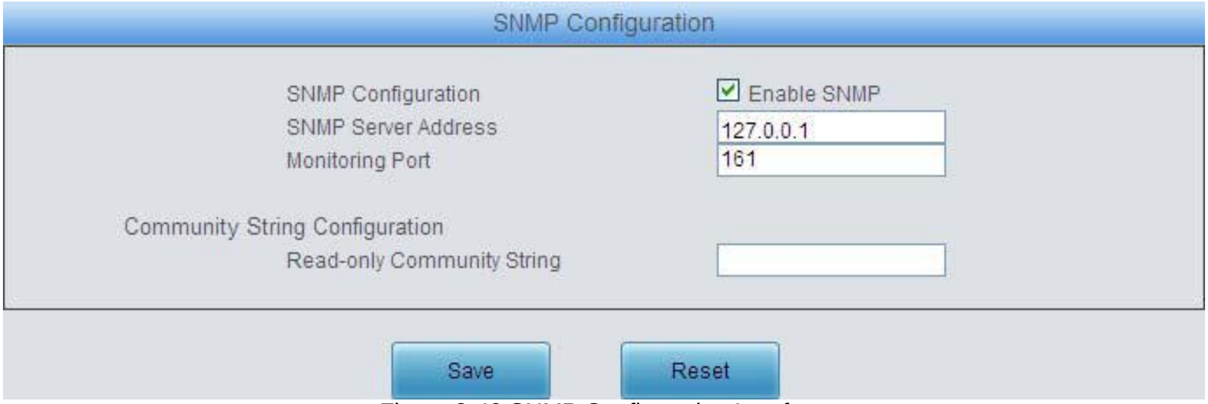

Figure 3-49 SNMP Configuration Interface

See Figure 3-49 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 3-49.

Item	Description
SNMP Server Address	IP address of SNMP.
Monitoring Port	Monitoring Port for SNMP on the gateway.
Read-only Community String	Community string used for information acquisition.

3.8.10 PING Test

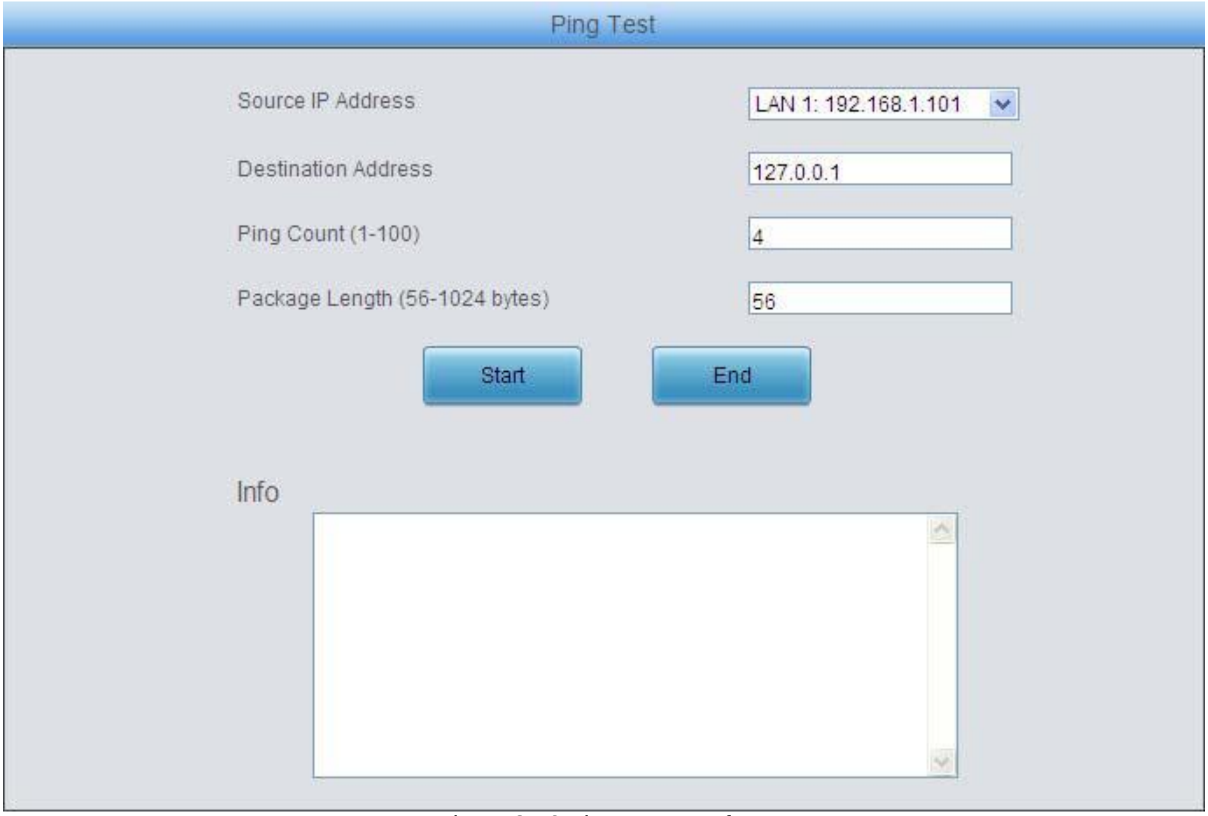

Figure 3-50 Ping Test Interface

See Figure 3-50 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
Source IP Address	Source IP address where the Ping test is initiated.
Destination Address	Destination IP address on which the Ping test is executed.
Ping Count	The number of times that the Ping test should be executed. Range of value: 1~100.
Package Length	Length of the data package used in the Ping test. Range of value: 56~1024 bytes.
Info	The information returned during the Ping test, helping you to learn the network connection status between the gateway and the destination address.

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

3.8.11 TRACERT Test

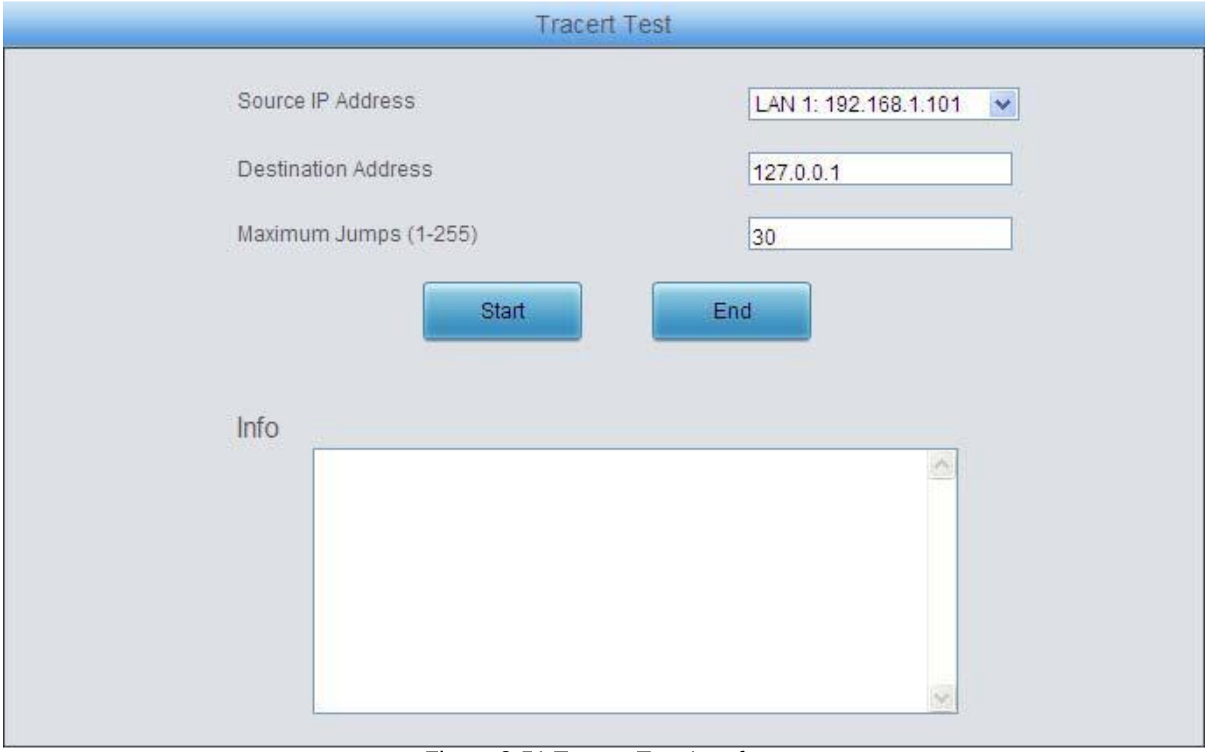

Figure 3-51 Tracert Test Interface

See Figure 3-51 for the Tracert test interface. A Tracert test can be initiated from the gateway on a designated IP address to check the routing status between them. The table below explains the configuration items shown in the above figure.

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

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

Appendix A Technical Specifications

Dimensions

440×44×267 mm^3

Weight

About 4 kg

Environment

Operating temperature: 0℃—55℃

Storage temperature: -20℃—85℃

Humidity: 8%— 90% non-condensing

Storage humidity: 8%— 90% non-condensing

LAN

Amount: 2 (10/100 BASE-TX (RJ-45)) Self-adaptive bandwidth supported Auto MDI/MDIX supported

FXS/FXO Port

Amount: 8/16/32

Type: RJ11

Maximum transmission distance: 1500m

Impedance

Input impedance:

≥1MΩ/500V DC; ≥10kΩ/1000V AC

Insulation resistance of telephone line from PC:

≥2MΩ/500V DC

Telephone line impedance: Compliant with the national standard impedance for three-component network

Serial Port

Amount: 1 (RS-232)

Baud rate: 115200bps

Connector: RJ45 to DB-9 Connector

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: 100~240V AC

Signaling & Protocol

SIP signaling

Supported protocol: SIP V1.0/2.0, RFC3261

Audio Encoding & Decoding

G.711A	*64kbps*
G.711U	*64kbps*
G.729A/B	*8kbps*

Sampling Rate

8kHz

Safety

Lightning resistance: Level 4

Appendix B Troubleshooting

1. What to do if I forget the IP address of the SMG gateway?

Long press the Reset button on the gateway to restore to factory settings. Thus the IP address will be restored to its default value:

LAN1: 192.168.1.101

LAN2: 192.168.0.101

2. The SMG gateway only supports routing on two directions, i.e. Tel→IP and IP→Tel. What to do if I want to make a Tel→Tel call?

You can make a Tel→Tel call via the routing of Tel→IP→IP→Tel. See below for detailed introductions.

Provided you are going to initiate a call from Port Group 1 to Port Group 2; the IP address and port number of your gateway are 192.168.1.101 and 5060 respectively.

a) Add a new routing rule on the Tel→IP routing rule configuration interface (See Figure 3-28). Select a port group (e.g. **Port Group 1**) as 'Source Port Group' to initiate the call and fill in 'Destination IP' and 'Destination Port' with the gateway's IP address (e.g. LAN1: **192.168.1.101**) and port number (e.g. **5060**). Then the call initiated from the station corresponding to Port Group 1 will be routed to the gateway.

b) Add a new routing rule on the IP→Tel routing rule configuration interface (See Figure 3-25). Fill in 'Source IP' with the gateway's IP address (e.g. LAN1: **192.168.1.101**) and select a port group (e.g. **Port Group 2**) as 'Destination Port Group' to be called. Then if the IP end of the gateway calls itself, the station corresponding to Port Group 2 will ring.

c) Finishing the above configurations, you can perform a Tel→Tel call from Port Group 1 to Port Group 2 simply by the way you make a Tel→IP call.

3. Does call forwarding involve routing and number manipulation?

Yes. A call forward procedure can be regarded as a Tel→IP call. It uses the routing rules and number manipulation rules in the same way as the Tel→IP call. A complete call forward is performed as follows:

a) An incoming IP call to the gateway rings the port which matches the IP→Tel routing and number manipulation rules and obtains a new callerID.

b) Then the gateway uses the newly obtained callerID and the call forward number, via the Tel→IP routing and number manipulation rules, to make another call from the port to a remote IP address.

4. In what cases can I conclude that the SMG 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, but the channel indicator never lights up after the gateway startup or the color it lights up does not comply with the actual state or port type.

Other problems such as inaccessible calls, failed registrations, incorrect numbers and abnormal dialing operations on the FXS port 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.

5. What to do if I cannot enter the WEB interface of the SMG 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.