Synway SMG-C Series Analog Gateway

SMG1008C

SMG1016C

SMG1032C

Analog Gateway

User Manual

Version 1.7.0

Synway Information Engineering Co., Ltd

www.synway.net

Content

Copyright Declaration

Revision History

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Chapter 1 Product Introduction

Thank you for choosing Synway SMG-C Series Analog Gateway!

The Synway SMG-C series analog gateway products (hereinafter referred to as 'SMG-C 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.

1.1 Typical Application

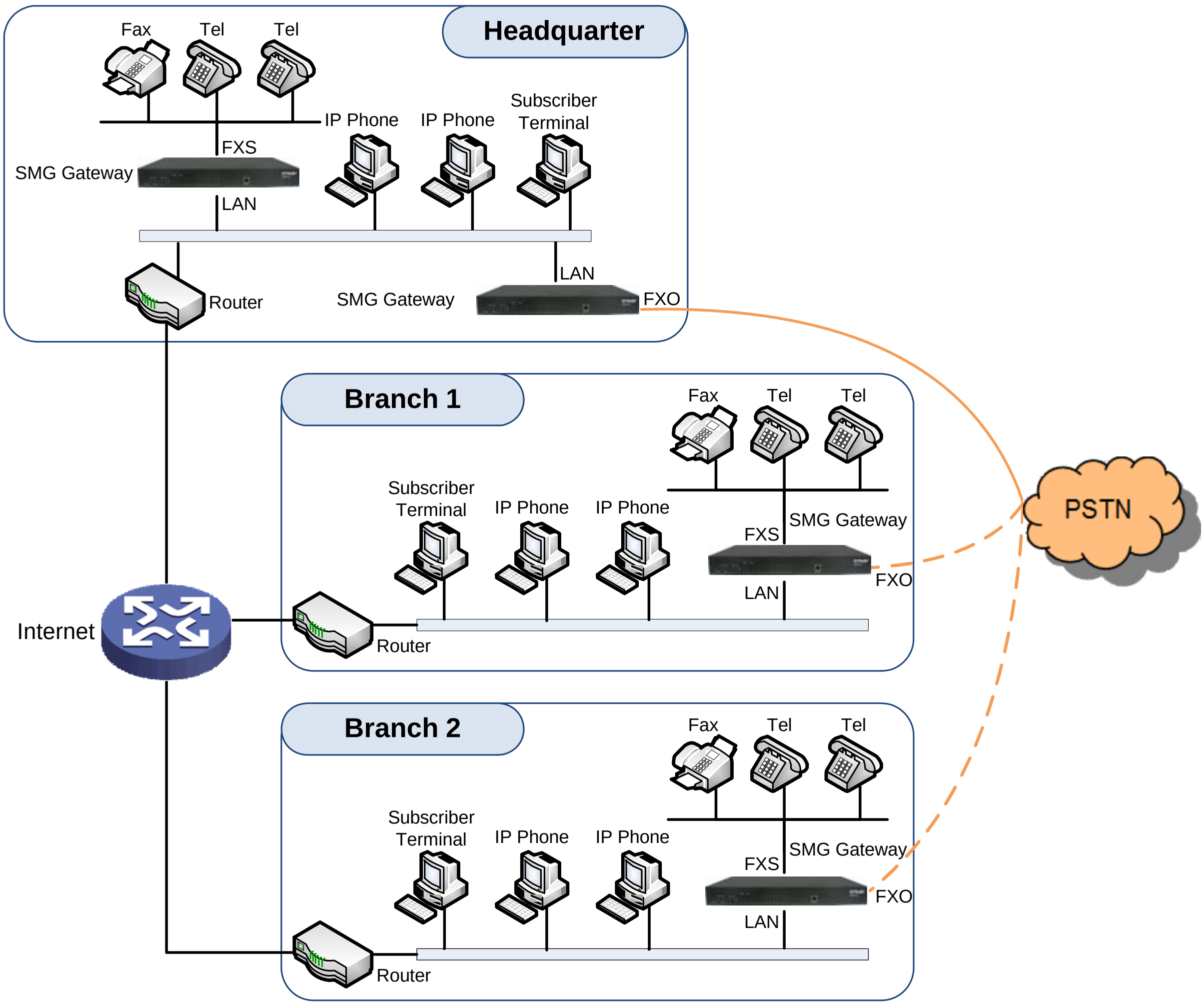


Figure 1-1 Typical Application for SMG-C Series Gateway

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 channel.
TDM/VoIP Routing	Sets a routing path: from IP to TDM or from TDM to IP.
Fax	Provides multiple fax parameters: fax mode, maximum fax rate, fax train mode, error correction mode, etc.
Communication without Power	Enable a connection of the station which is linked with the FXS port and the trunk which is linked with the FXO port to keep the calls between the FXS port and PSTN uninterrupted during power outage.
Communication without Network	Automatically routes a call to the proper port according to the configuration in case of network failure or call timeout.
Send Polarity Reversal Signal	Sends the polarity reversal signal to a corresponding FXS channel when the called party pick-up behavior is detected.
Detect Polarity Reversal Signal	Turns a corresponding channel into the talking state when the FXO port detects the polarity reversal signal.
Simultaneous Register to Multiple Servers	Registers the gateway to a master registrar server and a spare registrar server simultaneously.
IMS Network	Registers the gateway to a server under IMS network.
SIP Station	Supports a SIP terminal to be registered to the gateway and become a SIP station.
Group Ringing	Rings all the idle FXS ports in a port group.
Ringing by Turns	Rings the FXS ports in a port group by turns according to the *Rule for Ringing by Turns*.
Preemptive Answer	When a channel in a port group is ringing, another channel in the same port group can press the preemptive answer keyboard shortcut to transfer the call from the ringing channel to the current channel.
Centralized Manage	The gateway can register to Synway DCMS and accept the management of the platform.
Signaling & Protocol	**Description**
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261.
Voice	CODEC: G.711A, G.711U, G.729A/B, G.723, G.722, AMR, iLBC DTMF Mode: RFC2833, SIP INFO, INBAND

Network	Description
Network Protocol	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN.
Static IP	IP address modification support.
DHCP	IP address dynamic allocation support.
DNS	Domain Name Service support.
Security	**Description**
Admin Authentication	Supports admin authentication to guarantee the resource and data security.
System Monitor	Monitors the running status of the system and the server.
Maintain & Upgrade	**Description**
WEB Configuration	Support of configurations through the WEB user interface.
Language	Chinese, English.
Software Upgrade	Support of user interface, gateway service, kernel and firmware upgrades based on WEB.
Tracking Test	Support of Ping and Tracert tests based on WEB.
SysLog Type	Three options available: ERROR, WARNING, INFO, DEBUG.

1.3 Hardware Description

The SMG C-type analog gateway 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. The SMG-8C analog gateway adopts an external 12V power supply. See below for product appearance.

Figure 1-2 SMG1008C Front View

Figure 1-3 SMG1008C Rear View

Figure 1-4 SMG1016C Front View

Figure 1-5 SMG1016C Rear View

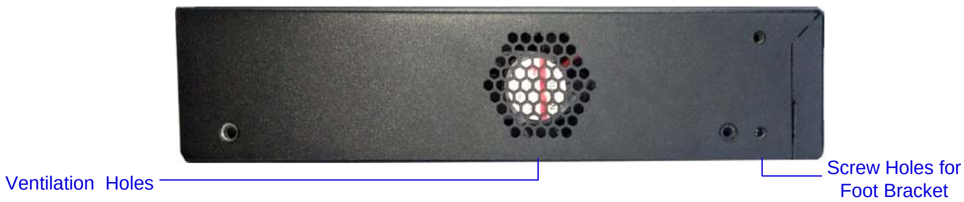


Figure 1-6 Left View

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
Console 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**
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, indicating the network connection status.
ACT Indicator	The orange LED, whose flashing tells data are being transmitted.
Channel Indicator	FXS and FXO channels are respectively marked by green and red LED after power on. 1. When the channel is idle, the LED Lights up; 2. When the channel is off-hook, the LED flashes slowly; 3. When the channel is ringing, the LED flashes fast.

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

1.4 Alarm Info

The SMG-C 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 and flash fast	System is starting.
	Flash slowly	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
- Angle Bracket *2, Rubber Foot Pad *4, Screw for Angle Bracket *8
- 220V Power Cord *1, External 12V Power Adapter *1 for SMG-8C gateway
- 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 angle 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).

These series products provide RJ11 interfaces. 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.

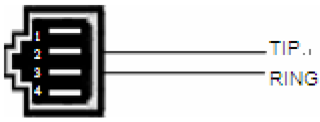


Figure 2-1

Step 6: Log in the gateway.

Enter the original IP address (LAN1: 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 3.9.15 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 3.9.2 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 another (Tel→Tel)

The gateway allows two FXS ports to call each other by default. Just use a station connected with an FXS port to dial the number of the destination FXS port and you can make a Tel→Tel call. The default number of an FXS port is 80XX, among which XX represents the corresponding port number. For example, the default number corresponding to Port 1 is 8001, and that corresponding to Port 32 is 8032.

Actually a Tel→Tel call on the gateway is accomplished via the routing of Tel→IP→IP→Tel. For detailed introductions and configuration guide, refer to Q2 in Appendix B.

Situation 2: 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 3.5.9 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 3.6.3 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 3.7.3 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 3: 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 3.6.3 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 3.7.2 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 3.6.1 FXS 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 3.6.1 FXS 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 3.6.1 FXS for detailed instructions.

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

Situation 1: Hook-flash operation

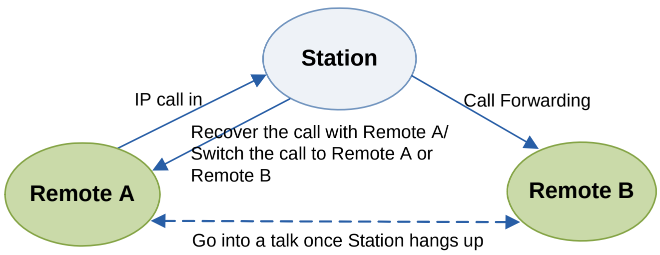


Figure 2-2 Call Forward via Hook-flash

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 3.6.1 FXS for detailed instructions.

Special Instructions:

- The chassis of the SMG-C 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.
- 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 1-6) 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.

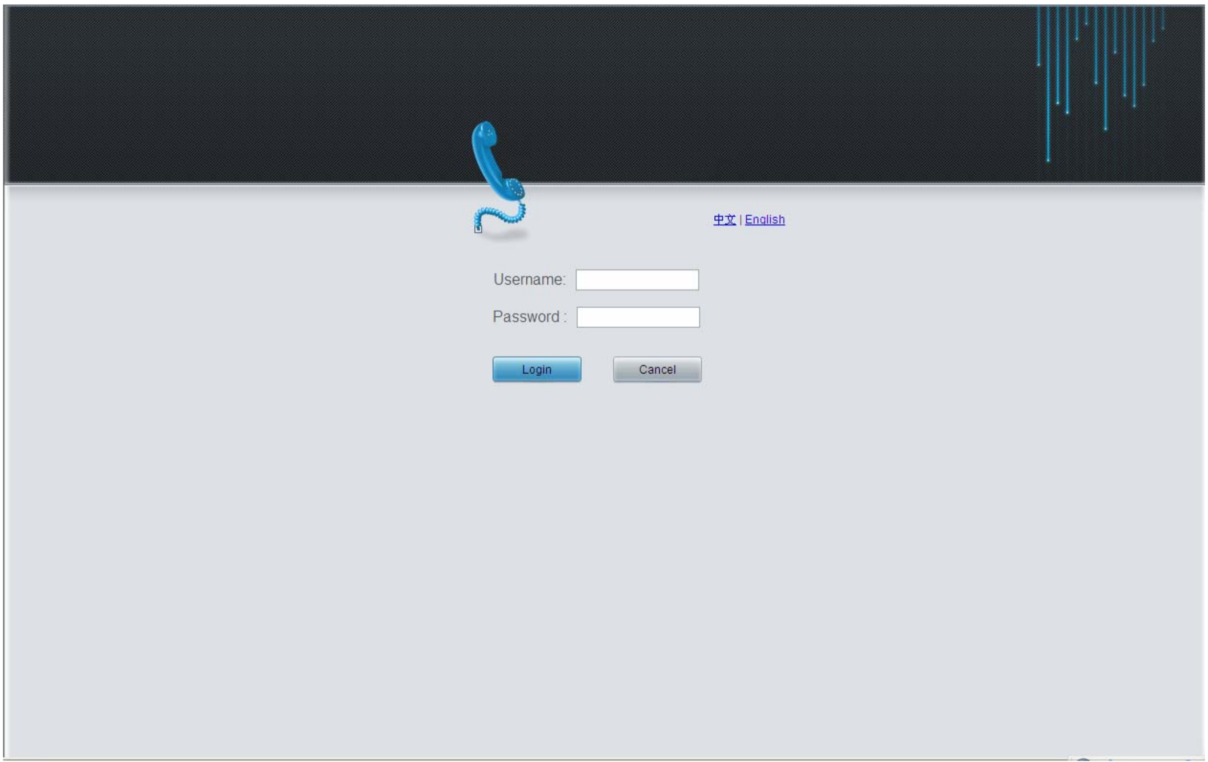


Figure 3-1 Login Interface

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 3.9.15 Change Password.

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

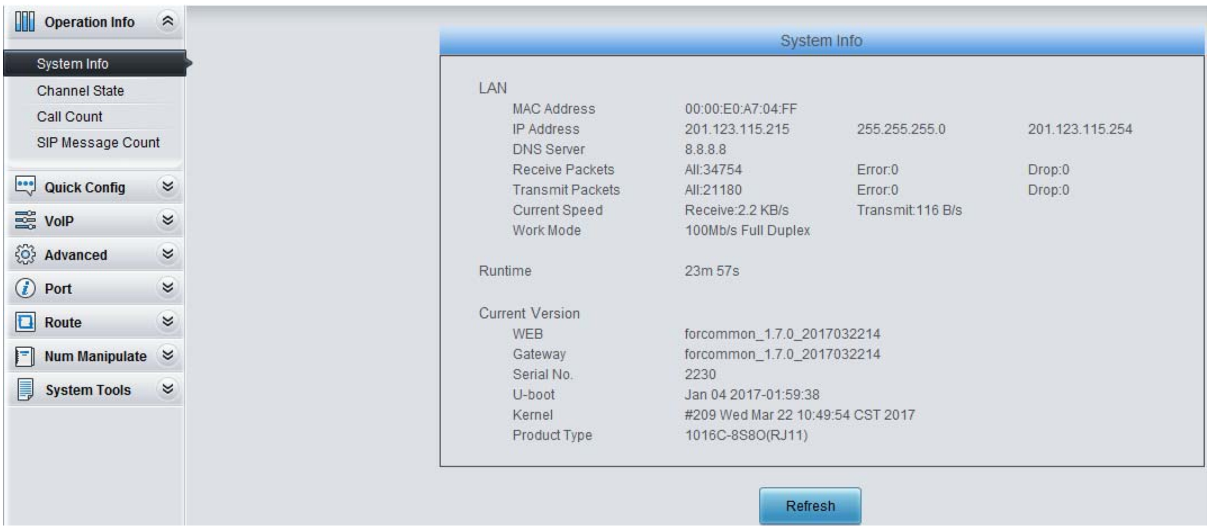


Figure 3-2 Main Interface

3.2 Operation Info

Operation Info includes four parts: ***System Info***, ***Channel State***, ***Call Count*** and ***SIP Message Count***, showing the current running status of the gateway. See Figure 3-3.

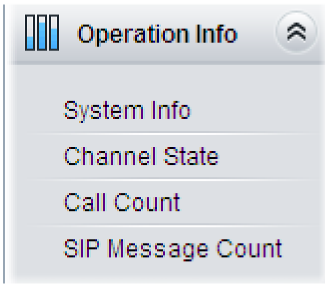


Figure 3-3 Operation Info

3.2.1 System Info

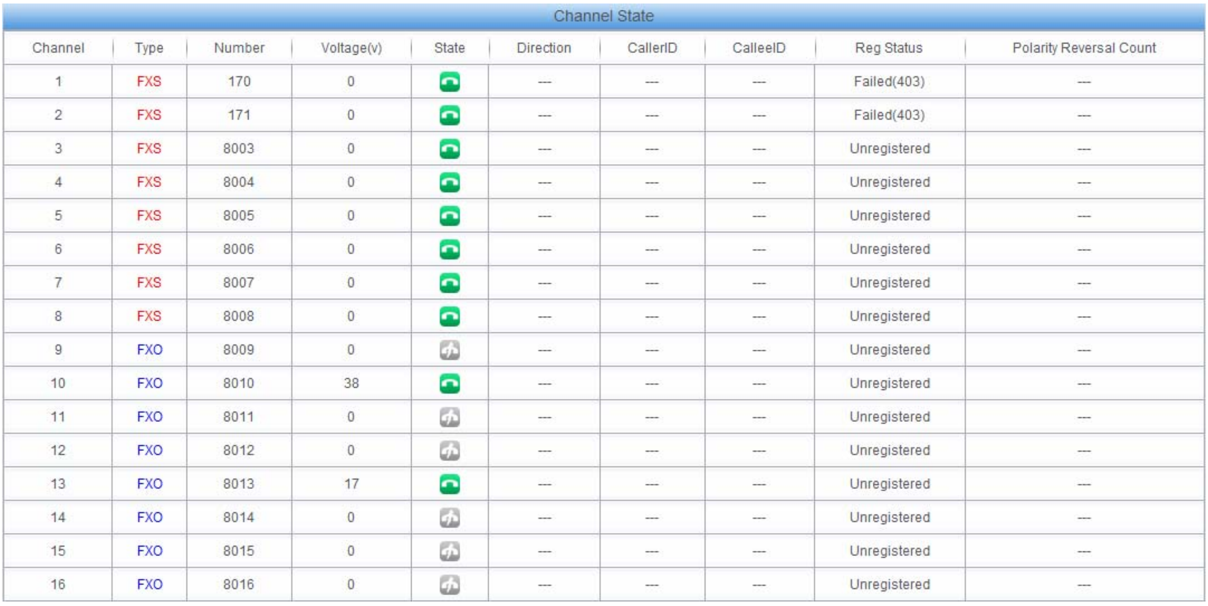
Figure 3-4 System Info Interface

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.
IP Address	The three parameters from left to right are IP address, subnet mask and default gateway of LAN.
DNS Server	DNS server address of LAN.
Receive Packets	The amount of receive packets after the gateway’s startup, including three options:

	All, Error and Drop.
Transmit Packets	The amount of transmit packets after the gateway's startup, including three options: All, Error and Drop.
Current Speed	Show the current speed of data receiving and transmitting.
Work Mode	Show the work mode of the network, including four modes: 10 Mbps Half Duplex, 10 Mbps Full Duplex, 100 Mbps Half Duplex, 100 Mbps Full Duplex.
Runtime	Time of the gateway keeping running normally after startup, which will be automatically updated.
WEB	Current version of the WEB interface.
Gateway	Current version of the gateway service.
Serial Num	Unique serial number of an SMG-C analog gateway.
U-boot	Current version of Uboot.
Kernel	Current version of the system kernel on the gateway.
Product Type	The type of current analog gateway.

3.2.2 Channel State



Channel State

Channel	Type	Number	Voltage(v)	State	Direction	CallerID	CalleeID	Reg Status	Polarity Reversal Count
1	FXS	170	0		---	---	---	Failed(403)	---
2	FXS	171	0		---	---	---	Failed(403)	---
3	FXS	8003	0		---	---	---	Unregistered	---
4	FXS	8004	0		---	---	---	Unregistered	---
5	FXS	8005	0		---	---	---	Unregistered	---
6	FXS	8006	0		---	---	---	Unregistered	---
7	FXS	8007	0		---	---	---	Unregistered	---
8	FXS	8008	0		---	---	---	Unregistered	---
9	FXO	8009	0		---	---	---	Unregistered	---
10	FXO	8010	38		---	---	---	Unregistered	---
11	FXO	8011	0		---	---	---	Unregistered	---
12	FXO	8012	0		---	---	---	Unregistered	---
13	FXO	8013	17		---	---	---	Unregistered	---
14	FXO	8014	0		---	---	---	Unregistered	---
15	FXO	8015	0		---	---	---	Unregistered	---
16	FXO	8016	0		---	---	---	Unregistered	---

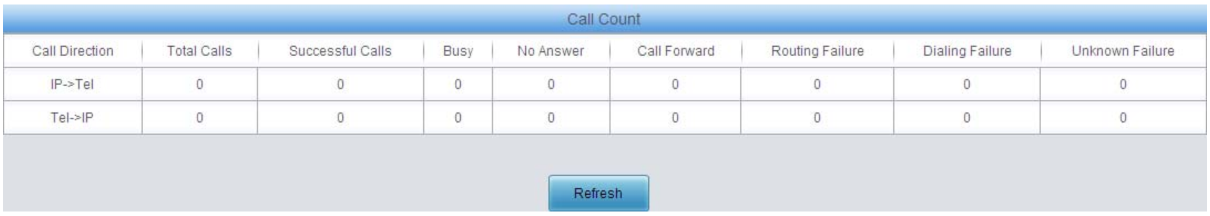
Figure 3-5 Channel State Interface

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

Item	Description
Channel	Channel number on the device.
Type	Type of the channel on the device. If this item shows ---, it means this channel is unavailable, that is, the corresponding module to this channel is not inserted or damaged.
Number	The number corresponding to the port.
Voltage	Line voltage on the channel, calculated by volt (V).
State	Displays the channel state in real time. You can move the mouse onto the channel

<table>
<tr><td></td><td>state icon for detailed state information.
<table>
<tr><th>State</th><th>Icon</th><th>Description</th></tr>
<tr><td>Idle</td><td></td><td>The channel is available.</td></tr>
<tr><td>Off-hook</td><td></td><td>The channel picks up the call.</td></tr>
<tr><td>Wait Answer</td><td></td><td>The channel receives the ringback tone and is waiting for the called party to pick up the phone.</td></tr>
<tr><td>Ringing</td><td></td><td>The channel is in the ringing state.</td></tr>
<tr><td>Talking</td><td></td><td>The channel is in a conversation.</td></tr>
<tr><td>Dialing</td><td></td><td>The channel is dialing.</td></tr>
<tr><td>Pending</td><td></td><td>The channel is in the pending state.</td></tr>
<tr><td>Internal State</td><td></td><td>Internal state of the channel.</td></tr>
<tr><td>Unusable</td><td></td><td>The channel is unavailable.</td></tr>
</table></td></tr>
<tr><td>Direction</td><td>Displays the direction of the call on channel.</td></tr>
<tr><td>CallerID</td><td>Displays the CallerID of the call on channel.</td></tr>
<tr><td>CalleeID</td><td>Displays the CalleeID of the call on channel.</td></tr>
<tr><td>Reg Status</td><td>Displays the registration status of the port.</td></tr>
<tr><td>Polarity Reversal Count</td><td>The counts of the polarity reversal detected by the FXO port.</td></tr>
</table>

3.2.3 Call Count



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

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
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.

3.2.4 SIP Message Count

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	100 Trying	180 Ringing	183 Session Prosess	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 FXS/FXO. The gateway can work normally after configuration.

See Figure 3-9 for the Quick Config-Network Settings interface. Refer to 3.9.2 Network for detailed settings. After configuration, click ***Next*** to enter the SIP Settings interface.

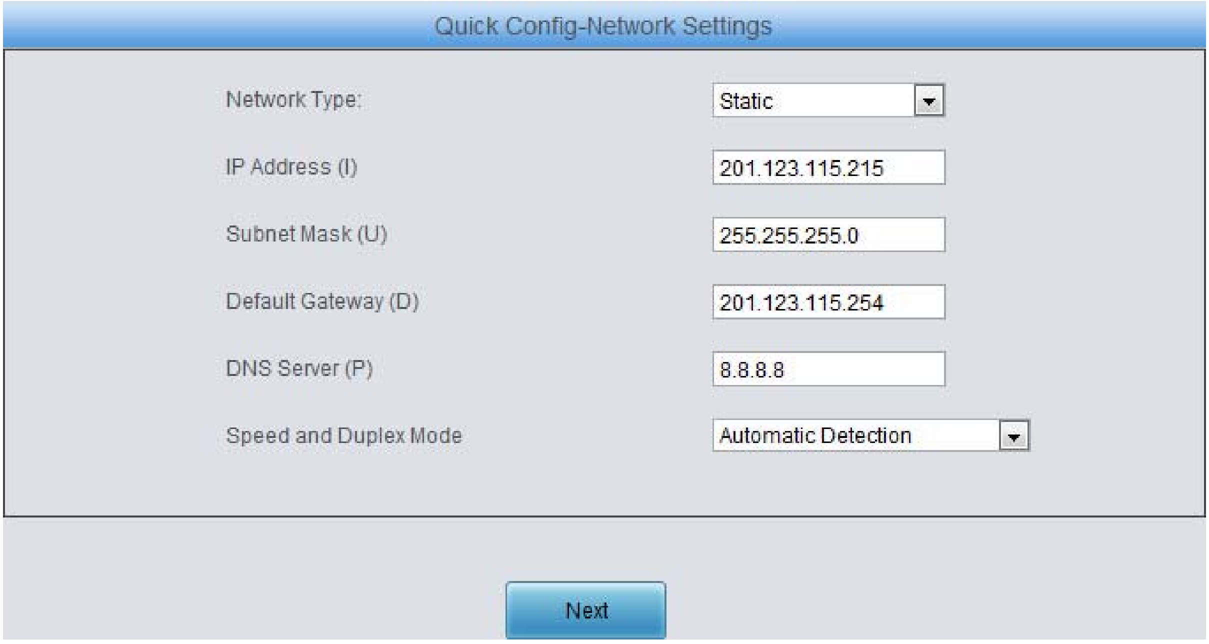


Figure 3-9 Quick Config-Network Settings Interface

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 FXS Settings interface.

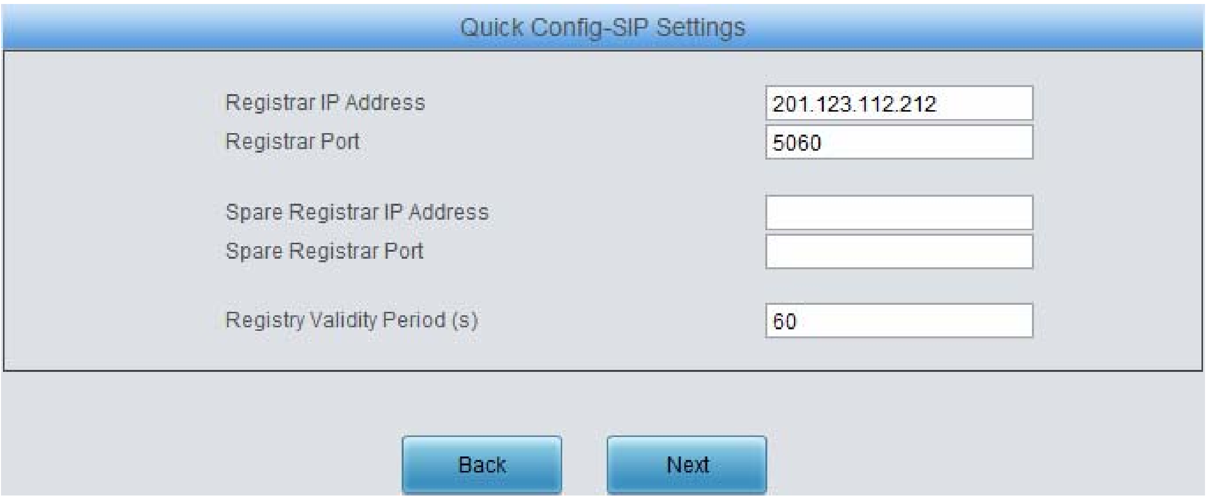
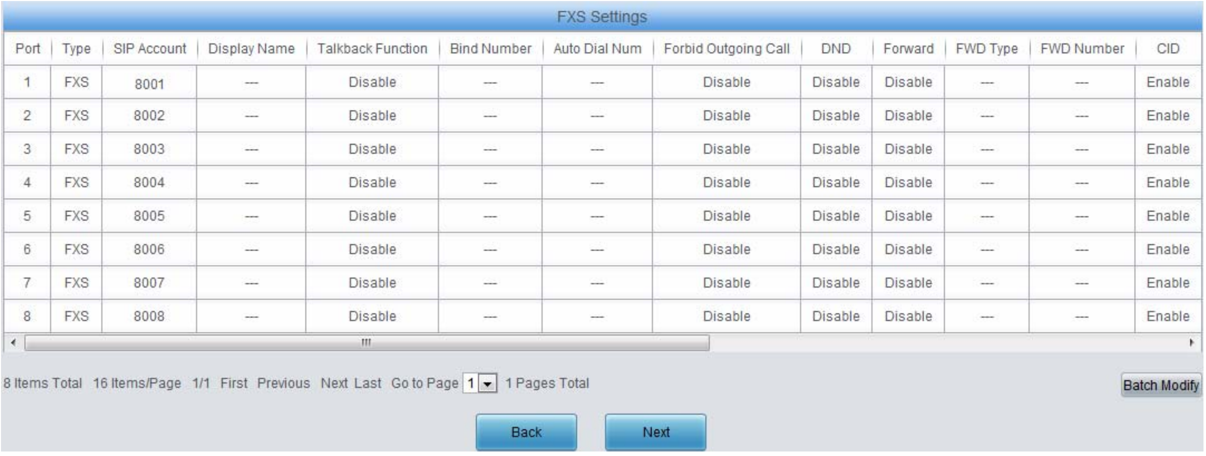


Figure 3-10 Quick Config-SIP Settings Interface

See Figure 3-11 for the FXS Settings interface. The configuration items on this interface are the same as those on the FXS interface. Refer to 3.6.1 FXS for detailed settings. After configuration, click ***Back*** to go back to the SIP Settings interface; click ***Next*** to enter the FXO Settings interface.



FXS Settings

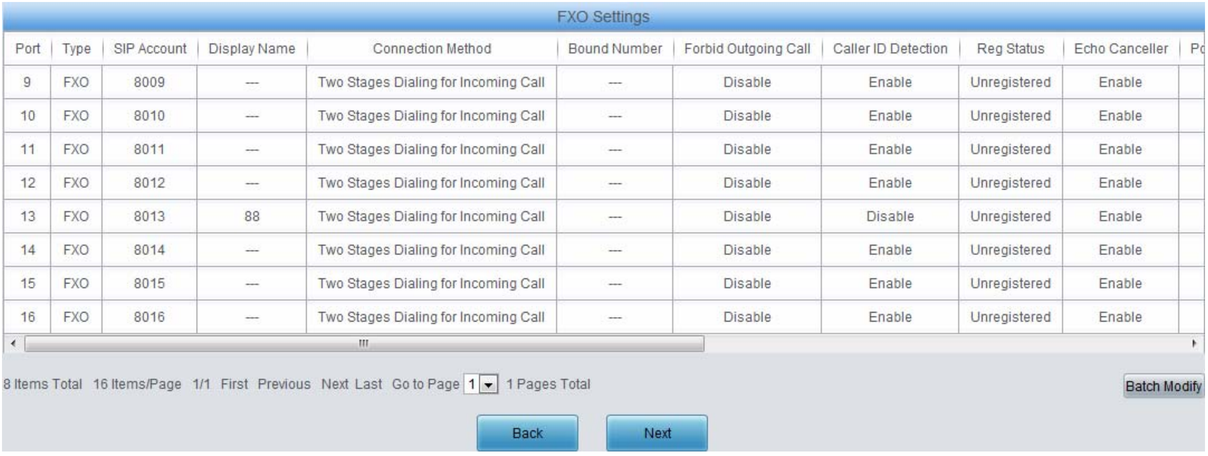
Port	Type	SIP Account	Display Name	Talkback Function	Bind Number	Auto Dial Num	Forbid Outgoing Call	DND	Forward	FWD Type	FWD Number	CID
1	FXS	8001	---	Disable	---	---	Disable	Disable	Disable	---	---	Enable
2	FXS	8002	---	Disable	---	---	Disable	Disable	Disable	---	---	Enable
3	FXS	8003	---	Disable	---	---	Disable	Disable	Disable	---	---	Enable
4	FXS	8004	---	Disable	---	---	Disable	Disable	Disable	---	---	Enable
5	FXS	8005	---	Disable	---	---	Disable	Disable	Disable	---	---	Enable
6	FXS	8006	---	Disable	---	---	Disable	Disable	Disable	---	---	Enable
7	FXS	8007	---	Disable	---	---	Disable	Disable	Disable	---	---	Enable
8	FXS	8008	---	Disable	---	---	Disable	Disable	Disable	---	---	Enable

8 Items Total 16 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total Batch Modify

Back Next

Figure 3-11 FXS Settings Interface

See Figure 3-12 for the FXO Settings Interface. The configuration items on this interface are the same as those on the FXO interface. Refer to 3.6.2 FXO for detailed settings. After configuration, click ***Back*** to back to the FXS Settings interface; click ***Next*** to enter the Quick Config-Completion interface. See Figure 3-13.



FXO Settings

Port	Type	SIP Account	Display Name	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	Po
9	FXO	8009	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	
10	FXO	8010	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	
11	FXO	8011	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	
12	FXO	8012	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	
13	FXO	8013	88	Two Stages Dialing for Incoming Call	---	Disable	Disable	Unregistered	Enable	
14	FXO	8014	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	
15	FXO	8015	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	
16	FXO	8016	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	

8 Items Total 16 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total Batch Modify

Back Next

Figure 3-12 FXO Settings Interface

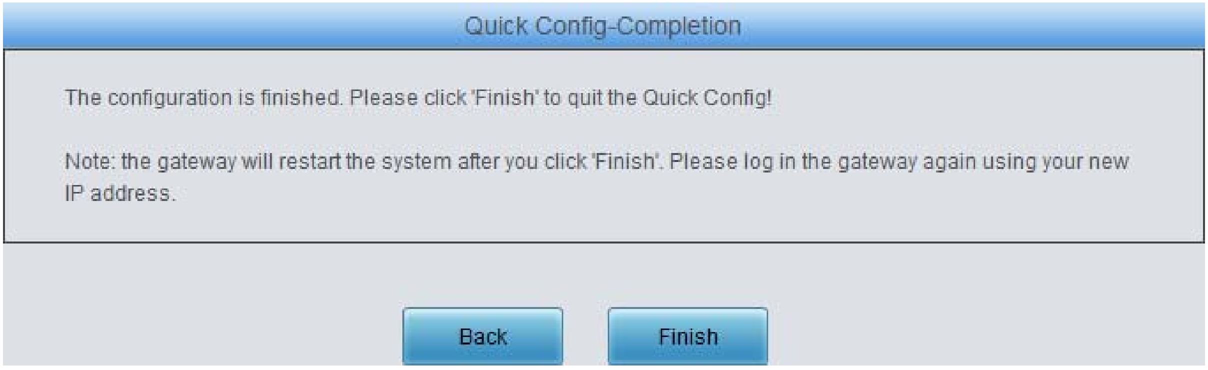


Figure 3-13 Quick Config-Completion Interface

Click ***Back*** to go back to the FXS 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 3-14. ***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 3-14 VoIP Settings

3.4.1 SIP

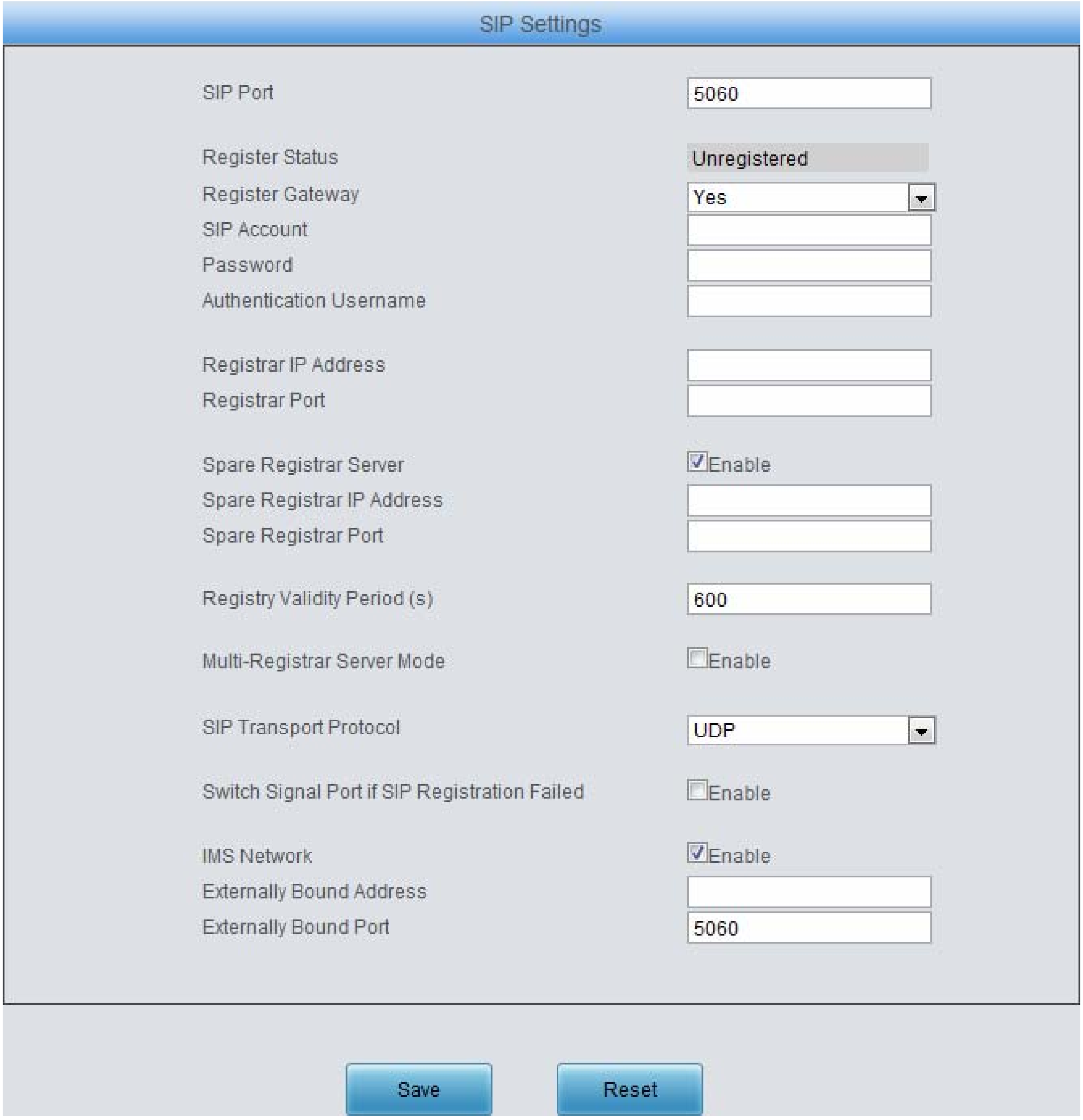


Figure 3-15 SIP Settings Interface

See Figure 3-15 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 3.9.16 Restart for detailed instructions. The table below explains the items shown in Figure 3-15.

Item	Description
SIP Port	Monitoring port of SIP signaling. The value range of it must be grater than 1024 and less than 65535, with the default value of 5060.
Register Status	Registration status of the gateway. When ***Register Gateway*** is set to *No*, the value of this item is *Unregistered*; when ***Register Gateway*** is set to *Yes*, the value of this item is either *Failed* or *Registered*.
Register Gateway	Sets whether to register the gateway as a whole. The default value is *No*. Only

	when this configuration is set to *Yes* can you see the configuration items ***SIP Account*** and ***Password***.
SIP Account	When the gateway initiates a call to SIP, this item corresponds to the username of SIP.
Password	Registration password of the gateway. To register the gateway to SIP, both configuration items ***SIP Account*** and ***Password*** should be filled in.
Authentication Username	Authentication username for registration.
Registrar IP Address	Address of the registry server for the gateway to register.
Registrar Port	Signaling port of the registry server.
Spare Registrar Server	Check the enable checkbox to enable the spare registrar server. By default, it is ***disabled***.
Spare Registrar IP Address	Address of the spare registry server for the gateway to register. The gateway will enable the spare registrar server if the master registrar server has no reply, or the master server is detected with no response in case the item ***Detection Server Cycle*** is enabled.
Spare Registrar Port	Signaling port of the spare registry server.
Registry Validity Period	Validity period of the SIP registry. Once the registry is overdue, the gateway should be registered again. This configuration item is valid only when ***Register Gateway*** is set to *Yes*. Range of value: 10~3600, calculated by s, with the default value of 600.
Multi-Registrar Server Mode	Tick the checkbox before to enable the multi-registrar server mode. By default, it is *disabled*.
SIP Transport Protocol	There are two modes *UDP* and *TCP* available for running the SIP protocol. The default value is *UDP*.
Switch Signal Port if SIP Registration Failed	If the SIP registration fails, the SIP signaling port N will switch to N+1 for a new registration. It will continue until the registration succeeds. The default value is *disabled*.
IMS Network	Once this feature is enabled, the gateway will send signaling messages to the corresponding externally bound address and port when it registers to the server. By default, this feature is *disabled*. Only when this feature is *enabled* will these items ***Externally Bound Address***, ***Externally Bound Port*** and ***Authentication Username*** be shown.
Externally Bound Address	Externally bound IP address for registration.
Externally Bound Port	Externally bound port for registration.

3.4.2 SIP Compatibility

See Figure 3-16 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.

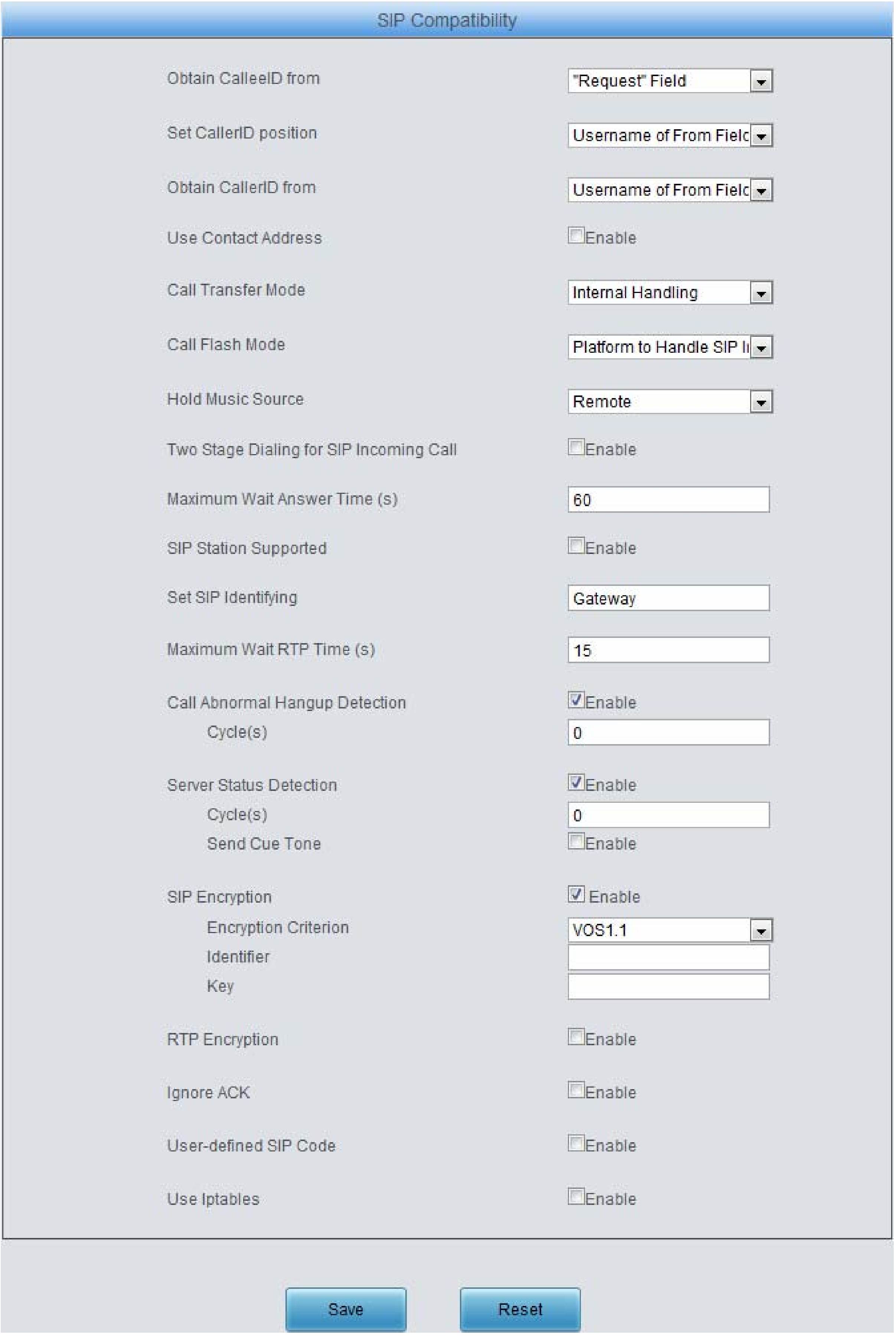


Figure 3-16 SIP Compatibility Setting Interface

The table below explains the items shown in Figure 3-16.

Item	Description
Obtain CalleeID from	There are two optional ways to obtain the called party number: from "*To" Field* and from "*Request" Field*. The default value is "*Request" Field*.
Set CallerID Position	There are two options to set the position of the calling party number: "*Displayname of From Field*" and "*Username of From Field*". The default value is "*Username of From Field*".
Obtain CallerID from	There are two optional ways to obtain the calling party number: from "*Displayname of From Field*" and from "*Username of From Field*". The default value is "*Username of From Field*".
Use Contact Address	Sets whether to send the request message according to the content of Contact, with the default setting of *disabled*. As it is disabled, if the Contact field indicates an IP address within the LAN, the request message will be sent according to the source address; if the Contact field indicates an IP address belonging to the WAN, the request message will be sent according to this IP address.
Call Transfer Mode	There are two optional ways to deal with call transfer: *Internal Handling* and *Platform to Handle SIP Info*. The default value is *Internal Handling*.
Call Flash Mode	There are two optional ways to deal with call flash: *Internal Handling* and *Platform to Handle SIP Info*. The default value is *Internal Handling*.
Hold Music Source	Sets the source of the hold music, with the default value of ***Remote***, This feature gets valid only when you choose the mode *Platform to Handle SIP Info*.
Two Stage Dialing for SIP Incoming Call	Once this feature is enabled, the incoming call from SIP should perform the two stage dialing operation. By default this feature is disabled.
Maximum Wait 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.
SIP Station Supported	Once this feature is enabled, a SIP terminal can be registered to the gateway and becomes a SIP station. By default this feature is disabled.
Set SIP Identifying	Sets the SIP identifying content in the SIP call message. The default setting is *Gateway*.
Maximum Wait RTP Time	Sets the maximum time for the SIP channel to wait for the RTP packet. If no RTP packet is received within the specified time period, the channel will enter the pending state automatically and release the call. The default value is *15*, calculated by s.
Call Abnormal Hangup Detection	Sets the interval between checks of the remote end's abnormal hangup, with the default value of 0 (feature disabled), calculated by s. It is suggested to set to 10s if this feature is necessary to be used.
Server Status Detection Cycle	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.

Send Cue Tone	Sets whether to send a cue tone once the server gets disconnected, with the default setting of *disabled*.
SIP Encryption	Once this feature is enabled, you can encrypt the SIP signal following selecting an encryption criterion and setting a key. By default it is *disabled*.
Encryption Criterion	The criterion used to encrypt the SIP signal. At present only VOS1.1 is supported.
Identifier	The identifier field of the VOS encryption, which is used to obtain the key of the SIP encryption.
Key	The key to encrypt the SIP signal.
RTP Encryption	Once this feature is enabled, you can encrypt the RTP package. By default it is *disabled*.
Ignore ACK	Once this feature is enabled, the gateway is not necessary to wait for the ACK message after sending the 200OK message to establish a call. By default it is *disabled*.
User-defined SIP Code	Once this feature is enabled, you can define a SIP code for the corresponding SIP status, with the default value of *disabled*.
Use Iptables	Once this feature is enabled, only the calls from the SIP registration server, the source IP address of the route IP->TEL and these IP addressed set in Access Control interface are permitted.

3.4.3 SIP Station

A SIP terminal can be registered to the gateway and becomes a SIP station. Enable the feature of '***SIP Station Supported***' on 3.4.2 SIP Compatibility, 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 3-17 below.

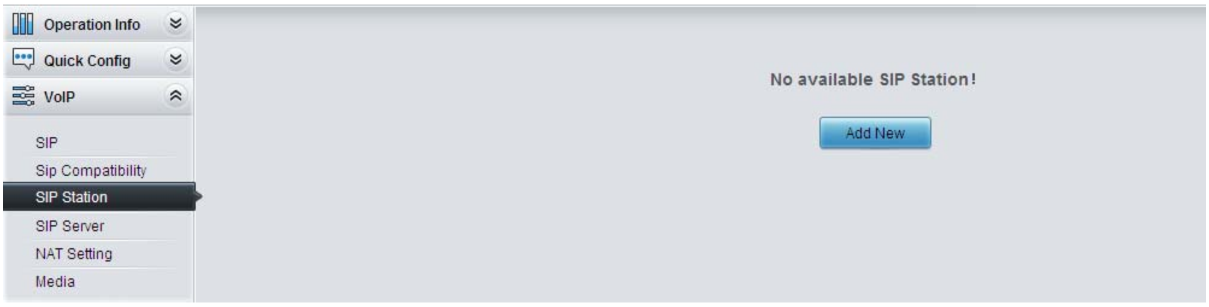


Figure 3-17 SIP Station Setting Interface

Click ***Add New*** to add SIP stations manually. See Figure 3-18. You can configure basic SIP station information on this interface. The bound port to a SIP station must be an FXO port and unique. The username must be the same as that used to register the SIP terminal to the gateway.

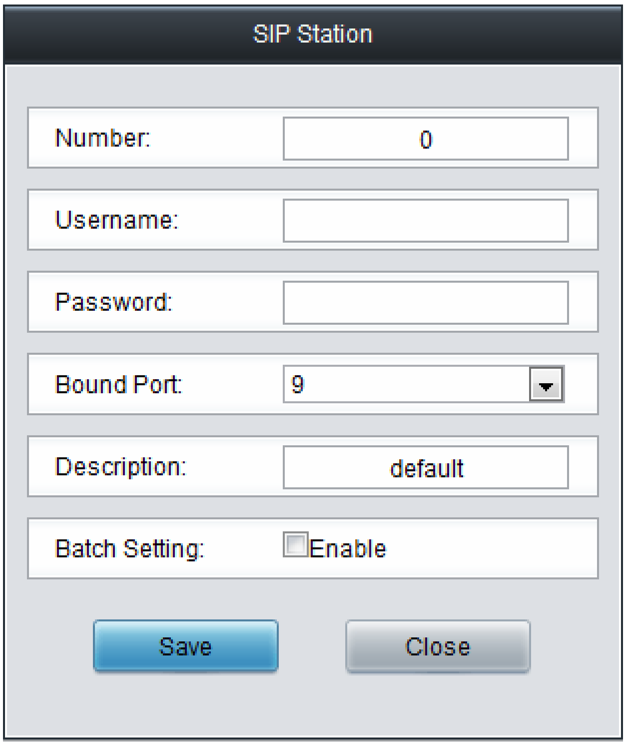


Figure 3-18 Add New SIP Station

The table below explains the items shown above:

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

After configuration, click ***Save*** to save the above settings into the gateway or click ***Close*** to cancel the settings. See Figure 3-19 for the applied SIP station information.

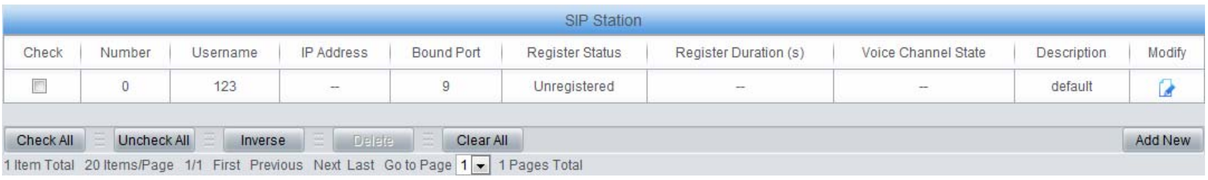


Figure 3-19 SIP Station Interface

Click ***Modify*** in the above figure to modify the configuration of the SIP station. See Figure 3-20. The configuration items on this interface are the same as those on the ***Add New SIP Station*** interface.

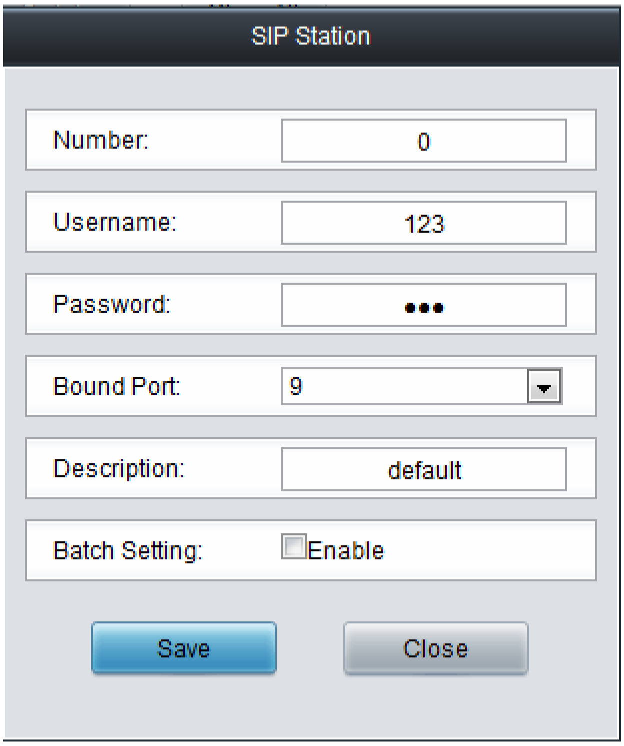


Figure 3-20 SIP Station Modification Interface

To delete a SIP station, check the checkbox before the corresponding index in Figure 3-19 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 3-19.

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 3-21 below.

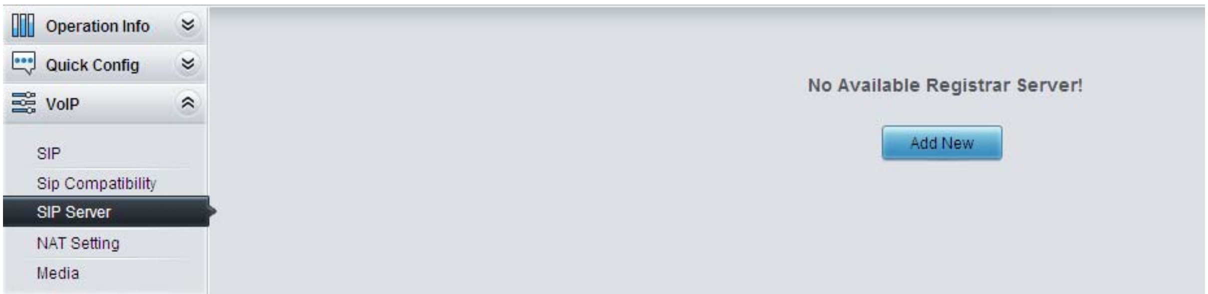


Figure 3-21 SIP Server Interface

Click ***Add New*** to add SIP servers manually. See Figure 3-22. You can configure basic SIP server information on this interface.

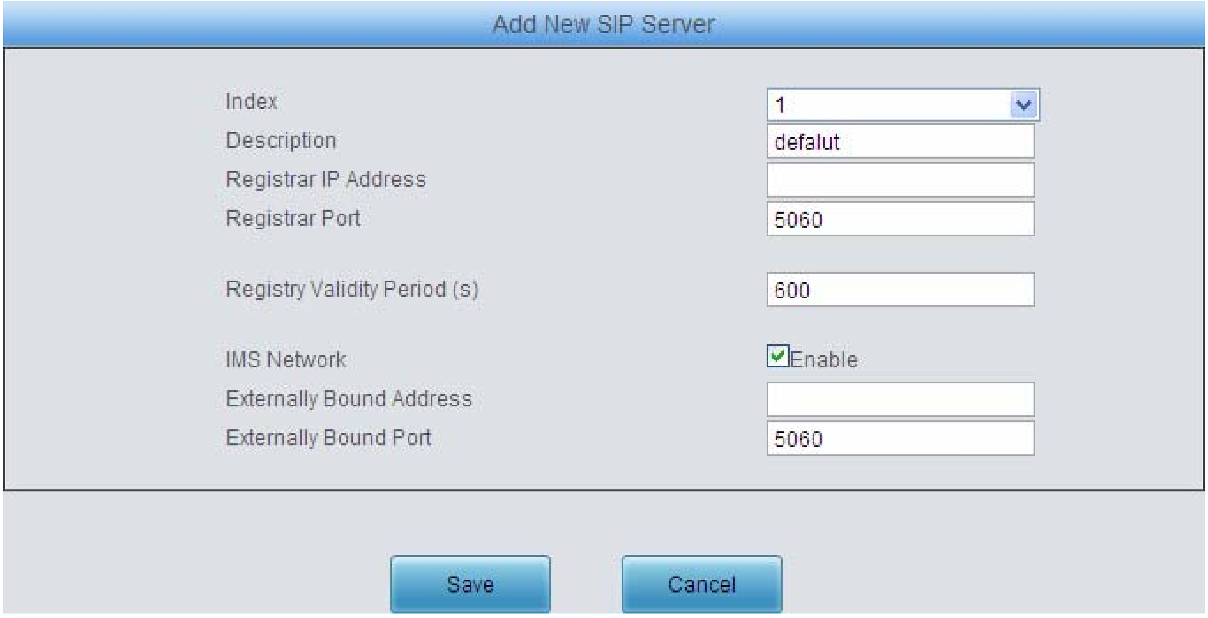
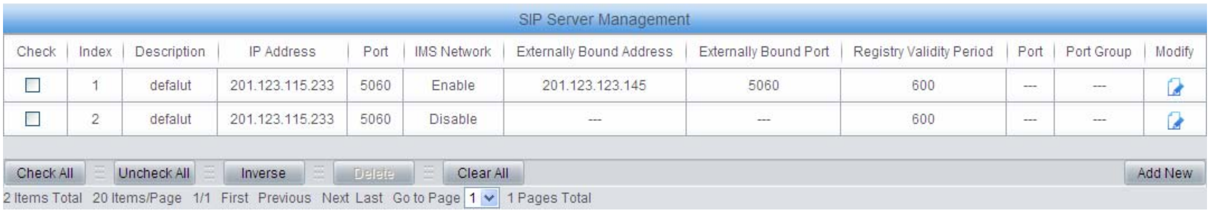


Figure 3-22 Add New SIP Server

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

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

After configuration, click ***Save*** to save the above settings into the gateway or click ***Cancel*** to cancel the settings. See Figure 3-23 for the SIP server management interface.



SIP Server Management

Check	Index	Description	IP Address	Port	IMS Network	Externally Bound Address	Externally Bound Port	Registry Validity Period	Port	Port Group	Modify
☐	1	defalut	201.123.115.233	5060	Enable	201.123.123.145	5060	600	---	---	
☐	2	defalut	201.123.115.233	5060	Disable	---	---	600	---	---	

Figure 3-23 SIP Server Management

Click ***Modify*** in the above figure to modify the configuration of the SIP server. See Figure 3-24.

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

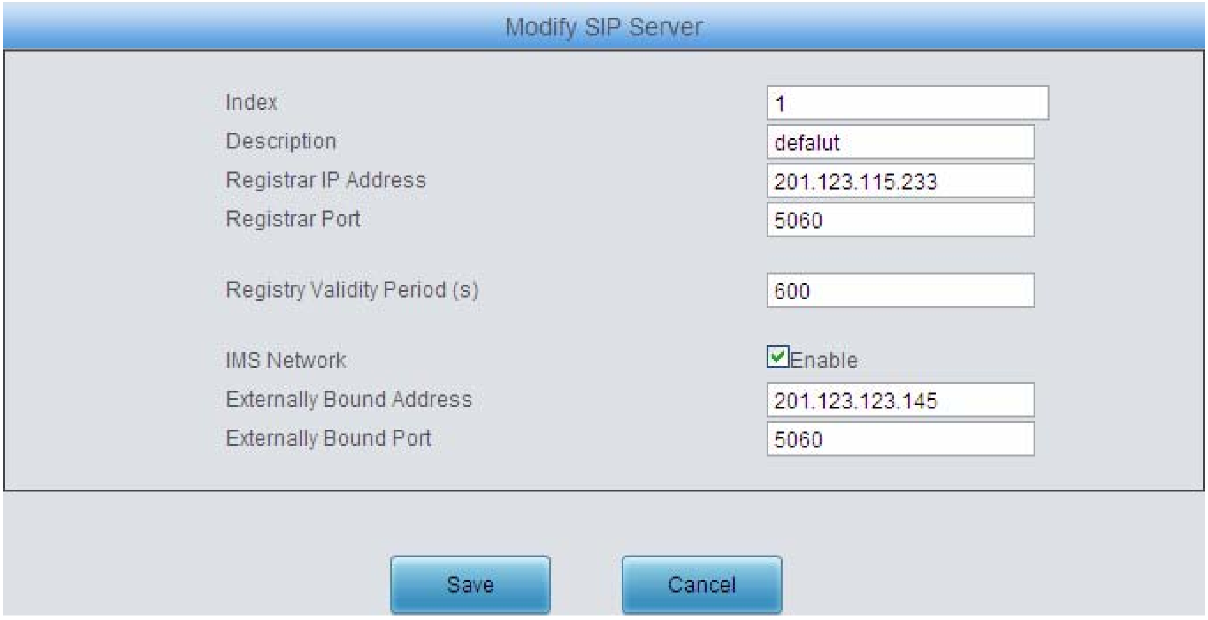


Figure 3-24 SIP Server Modification Interface

To delete a SIP server, check the checkbox before the corresponding index in Figure 3-23 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 3-23.

3.4.5 NAT Setting

See Figure 3-25 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.

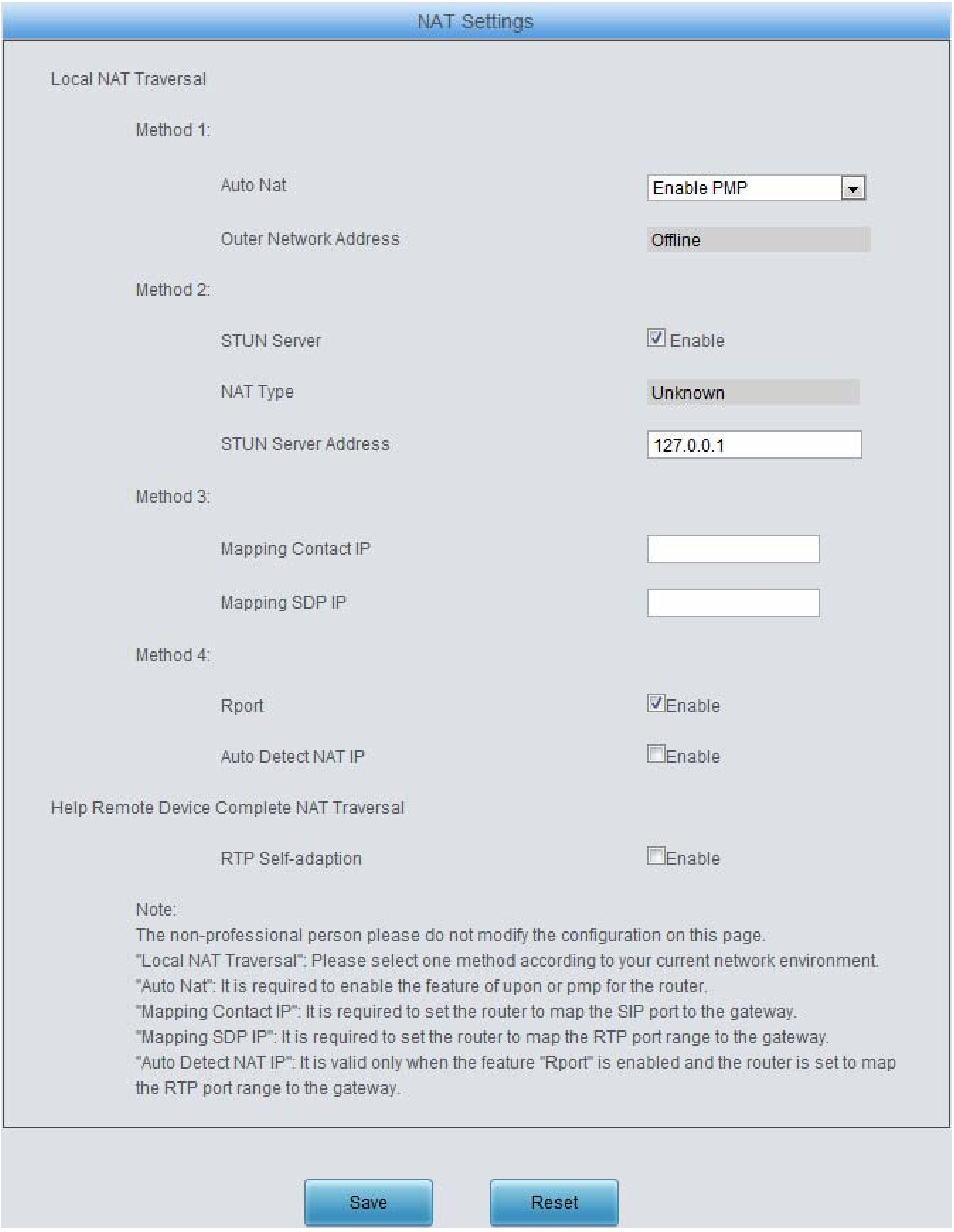


Figure 3-25 NAT Setting Interface

The table below explains the items shown in Figure 3-25.

Item	Description
Auto Nat	Sets whether to enable the Auto Nat feature. Three options are available: DisableAutoNat, Enable PMP and Enable UPNP, with the default value of *Auto Nat*.
Outer Network	The address of the outer network acquired automatically once the PMP or UPNP

Address	feature is enabled.
STUN Server	Sets whether to enable the STUN server for NAT traversal. By default the STUN server is disabled.
NAT Type	Detected NAT (Network Address Translation) type. The gateway will return the NAT type automatically in case ***STUN Server*** is enabled. It includes 9 types: unknown; no NAT; ConeNat; RestrictedNat; PortRestrictedNat; Symmetric NAT; Symmetric NAT with firewall; can't detect over (fail to send detect message) and fail to detect (No reply from the stun server).
STUN Server Address	Address of the server for STUN traversal.
Mapping Contact IP	The IP filled in here will be used in the Contact field of the SIP message.
Mapping SDP IP	The IP filled in here will be used in the SDP field of the SIP message.
Rport	When this feature is enabled, a corresponding Rport field will be added to the Via message of SIP. The default value is *enabled*.
Auto Detect NAT IP	When this feature is enabled, the gateway will parse the corresponding address and port in the message returned by Rport so as to use them for the following communication. By default, this feature is *disabled*. **Note:** This feature gets valid only when Rport is enabled.
RTP Self-adaption	When this feature is enabled, the RTP reception address or port carried by the signaling message from the remote end, if not consistent with the actual state, will be updated to the actual RTP reception address or port. By default, this feature is *disabled*.

3.4.6 Media

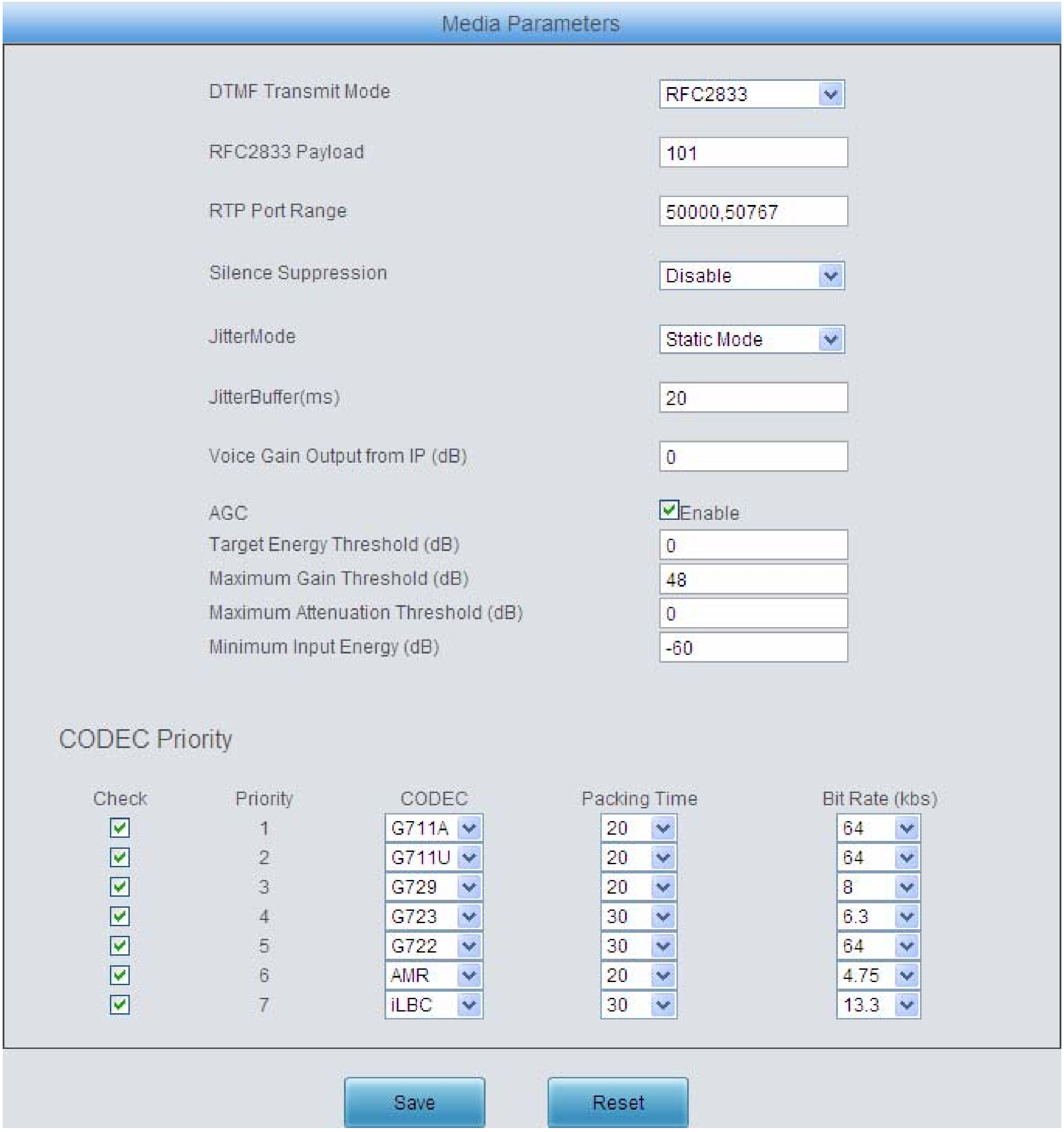


Figure 3-26 Media Settings Interface

See Figure 3-26 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 3.9.16 Restart for detailed instructions. The table below explains the items shown in Figure 3-26.

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

RTP Port Range	Supported RTP port range for the IP end to establish a call conversation, with the lower limit of 10000 and the upper limit of 60000 and the difference between larger than 480. The default value is 50000-50767.
Silence Suppression	Sets whether to send comfort noise packets to replace RTP packets or never to send RTP packets to reduce the bandwidth usage when there is no voice signal throughout an IP conversation. The optional values are *Enable* and *Disable*, with the default value of *Disable*.
JitterMode	Sets the mode for the Jitter buffer. The optional mode is *Static Mode* and *Adaptive Mode*, with the default value of *Static Mode*.
JitterBuffer	Acceptable jitter for data packets transmission over IP, which indicates the buffering capacity. A larger JitterBuffer means a higher jitter processing capability but as well as an increased voice delay, while a smaller JitterBuffer means a lower jitter processing capability but as well as a decreased voice delay. Range of value: 20~200, calculated by ms, with the default value of 20. **Note:** This is only valid if the Jitter Mode is set to Static Mode.
Voice Gain Output from IP	Adjusts the gain of the voice output from IP. Range of value: -24~12, calculated by dB, with the default value of 0.
AGC	If the AGC (Automatic Gain Control) feature is enabled, the gateway will automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals.
Target Energy Threshold	Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the default value of 0.
Maximum Gain Threshold	Set the maximum gain threshold that will be applied to the signal. Range of value: 0~48, calculated by dB, with the default value of 48.
Maximum Attenuation Threshold	Set the maximum attenuation that will be applied to the signal. Range of value: -42~0, calculated by dB, with the default value of 0.
Minimum Input Energy	Set the minimum threshold for the energy processed by AGC. Signals below this threshold will not be processed by AGC. Range of value: -60~ -25, calculated by dB, with the default value of -60.

<table>
<tr>
<td>CODEC Priority</td>
<td>
Supported CODECs and their corresponding priority for the IP end to establish a call conversation. The table below explains the sub-items:
<table>
<tr><th>Sub-item</th><th>Description</th></tr>
<tr><td>Priority</td><td>Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.</td></tr>
<tr><td>CODEC</td><td>Three optional CODECs are supported: G711A, G711U, G729A/B, G723, G722, AMR and iLBC.</td></tr>
<tr><td>Packing Time</td><td>Time interval for packing an RTP packet, calculated by ms.</td></tr>
<tr><td>Bit Rate</td><td>The number of thousand bits (excluding the packet header) that are conveyed per second.</td></tr>
</table>
By default, all of the seven CODECs are supported and ordered G711A, G711U, G729A/B, G723, G722, AMR and iLBC by priority from high to low.
The packing time and bit rate supported by different CODECs are listed in the table below. Those values in bold face are the default values.
<table>
<tr><th>COEDC</th><th>Packing Time (ms)</th><th>Bit Rate (kbps)</th></tr>
<tr><td>G711A</td><td>10 / 20 / 30 / 40 / 60</td><td>64</td></tr>
<tr><td>G711U</td><td>5 / 10 / 20 / 30 / 40 / 50 / 60</td><td>64</td></tr>
<tr><td>G729A/B</td><td>10 / 20</td><td>8</td></tr>
<tr><td>G723</td><td>30 / 60</td><td>5.3 / 6.3</td></tr>
<tr><td>G722</td><td>5 / 10 / 20 / 30 / 40</td><td>64</td></tr>
<tr><td>AMR</td><td>20 / 40 / 60</td><td>4.75</td></tr>
<tr><td>iLBC</td><td>30 / 60</td><td>13.3 / 15.2</td></tr>
</table>
</td>
</tr>
</table>

3.5 Advanced Settings

Advanced Settings includes fourteen parts: ***FXS***, ***FXO***, ***Tone Detector***, ***Tone Generator***, ***DTMF***, ***Ringing Scheme***, ***Fax***, ***Function Key***, ***Dialing Rule***, ***Dialing Timeout***, ***Cue Tone*****,** ***Color Ring***, ***QoS*** and ***Action URL***. See Figure 3-27. ***FXS*** is used to configure the general properties of the FXS port; ***FXO*** is used to configure the general properties of the FXO port; ***Tone Detector*** is used to configure some properties of detected tones; ***Tone Detector*** is used to configure some properties of tones sent from gateway; ***DTMF*** is used to set the properties related to DTMF; ***Ringing Scheme*** is used to set the ringing scheme for the FXS port; ***Fax*** is used to configure multiple fax parameters; ***Function Key*** is used to set a cluster of combination keys for you to query a related number; ***Dialing Rule*** and ***Dialing Timeout*** are used to set the judging conditions for dialing; ***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 FXS port; ***QoS*** uses the differentiated services technology to increase the gateway's service quality. ***Action URL*** is used to designate the server path to report the on-hook or off-hook state of the FXS channel.

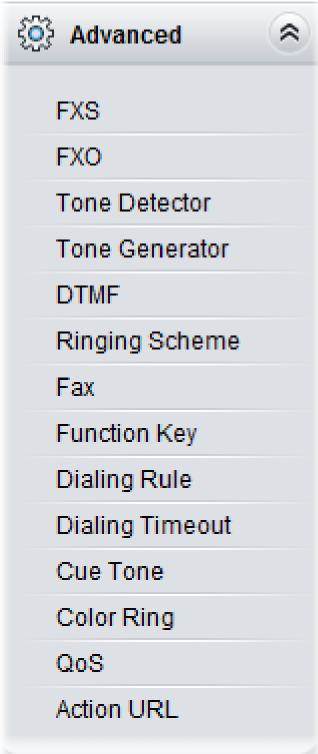


Figure 3-27 Advanced Settings

3.5.1 FXS

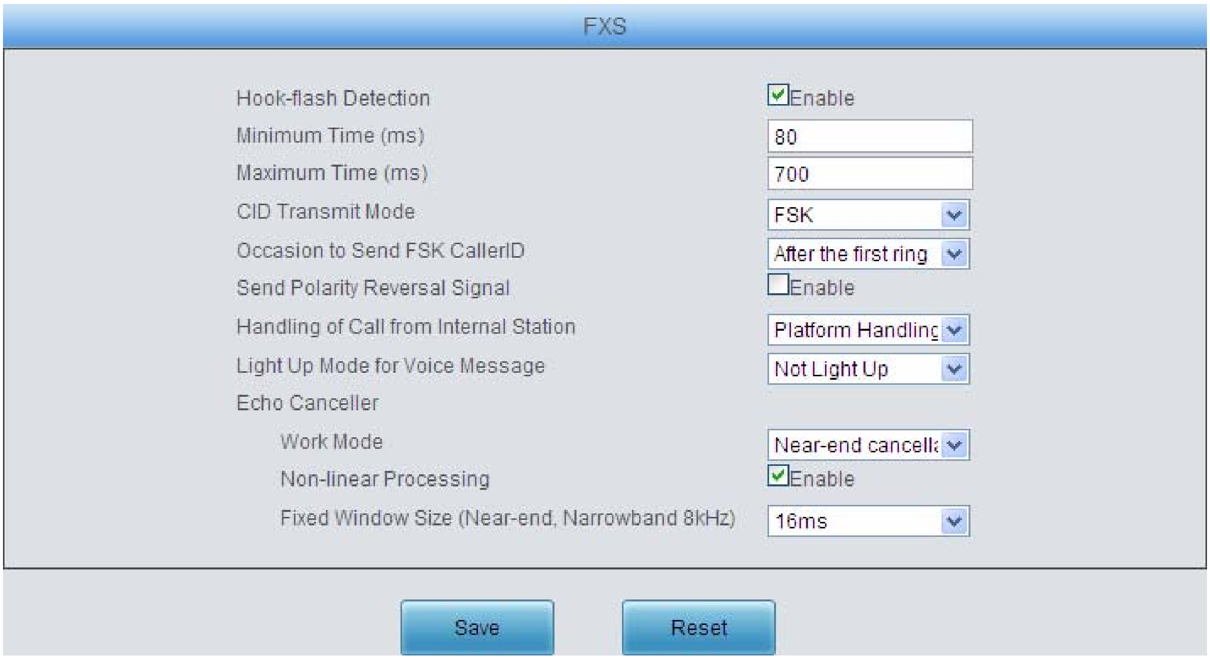


Figure 3-28 FXS Configuration Interface

See Figure 3-28 for the FXS configuration interface. The table below explains the items shown in the above figure.

Item	Description
Hook-flash Detection	Sets whether to enable the hook-flash detection feature or not, with the default setting of being disabled.

Minimum Time	Time length for judging a flash operation. Only a hook-flash operation which lasts a time more than the value of this configuration item will be regarded as a valid flash operation. Range of value: 80~ ***Maximum Time***, calculated by ms, with the default value of 80.
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.
Minimum Time Length of On-hook Detection	The minimum time length for detecting whether the phone is on-hook or not. Range of value: 64~2000, calculated by ms, with the default value of 64. **Note:** This item is valid only when the item Hook-flash Detection is disabled.
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*.
Occasion to Send FSK CallerID	Sets when to send the CallerID, before rings or after the 1st Ring. The default value is *after 1st Ring*.
Send Polarity Reversal Signal	Once this feature is enabled, the gateway will send the polarity reversal signal to a corresponding FXS channel when it detects the called party pick-up behavior. By default, this feature is *disabled*.
Handling of Call from Internal Station	Sets the handling mode for the calls from station to station, two options available: Internal Handling and Platform Handling, with the default value of Platform Handling.
Light Up Mode for Voice Message	Sets the light up mode for the voice message of the phone, There are two options: *Not Light Up* and *Light Up by FSK*, with the default value of *Not Light Up*.
Work Mode	Sets the work mode for the echo canceller. There are two options: *Near-end cancellation* and *Both near-end and far-end cancellation*, with the default value of *Near-end cancellation*.
Non-linear Processing	Sets whether to enable the mode of non-linear processing. By default, this feature is *enabled*.
Fixed Window Size	Sets the size of the window for the fixed cancellation.
Moving Window Size	Sets the size of the window for the moving cancellation.

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 3.9.16 Restart for detailed instructions.

3.5.2 FXO

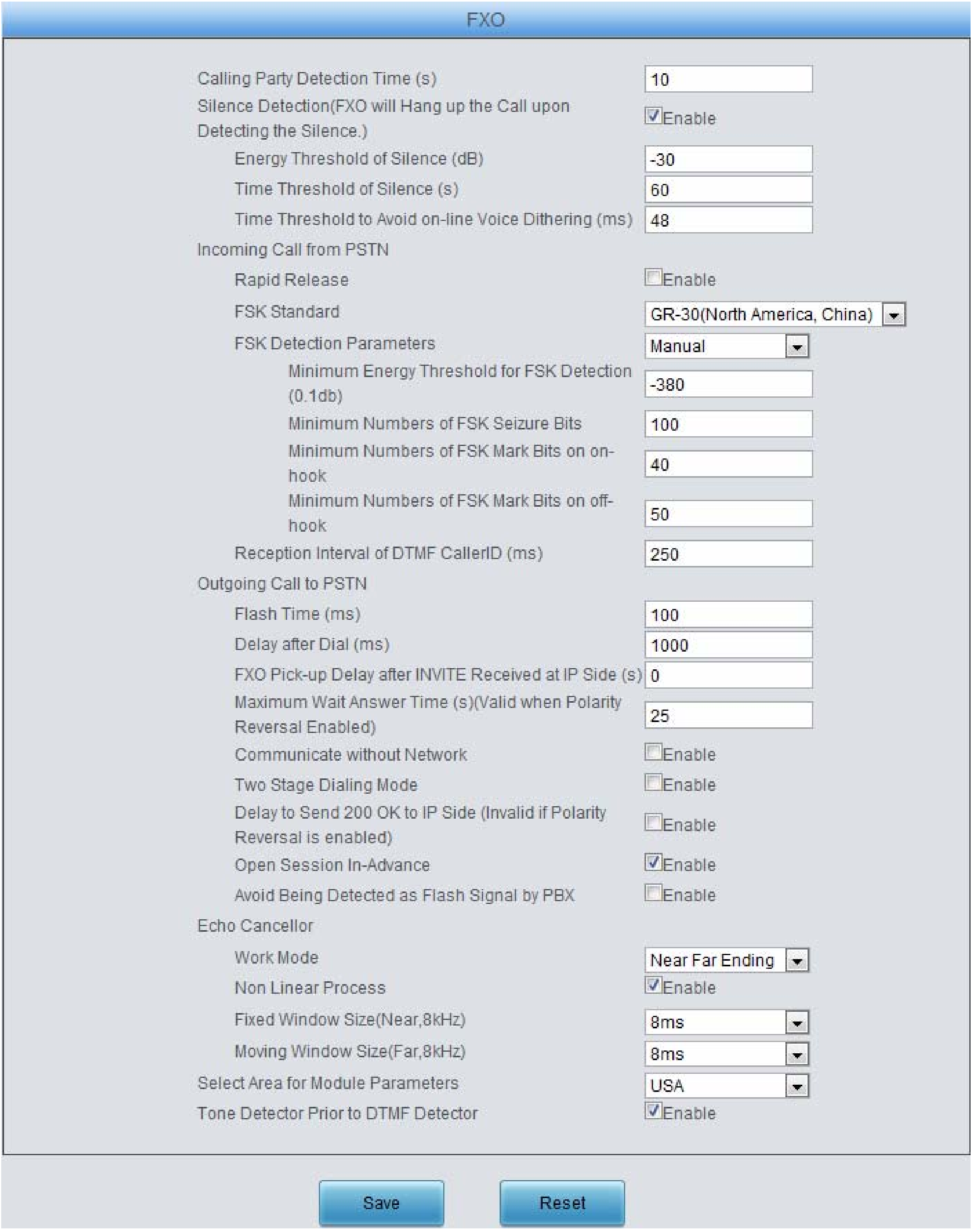


Figure 3-29 FXO Configuration Interface

The table below explains the particular configuration items for FXO.

Item	Description
Calling Party Detection Time	The maximum waiting time for the detection of the calling party number from FXO port. Range of value: 1~20, calculated by s, with the default value of *10*.

Silence Detection	Used to detect whether the line is silent or not according to the energy threshold and time threshold of silence. FXO will hang up the call automatically if these conditions are satisfied. The default setting is being disabled.
Energy Threshold of Silence	The energy threshold to judge whether the line is silent or not. The signal with the energy less than this set value will be determined to be silence. Range of value: -86~6, calculated by s, with the default value of -30. **Note:** This item will be valid only when Silence Detection is enabled.
Time Threshold of Silence	The time threshold to judge whether the line is silent or not, calculated by s, with the default value of 60. **Note:** This item will be valid only when Silence Detection is enabled.
Time Threshold to Avoid On-line Voice Dithering	Once this feature is enabled, the on-line voice will be determined to be dithering if the voice duration is less than the set value here. Range of value: 24~1000, calculated by ms, with the default value of *48*,
Rapid Release	Once this feature is enabled, the FXO port will release the source rapidly and go to the idle state when a call from PSTN to soft-terminal via FXO port is rejected by the IP soft-terminal.
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*.
FSK Detection Parameters	Sets the configuration mode of the FSK parameters, two options available: Default and Manual, with the default value of *Default*. In the Default mode, the FSK parameters use default values and cannot be modified. To modify the parameters, please select the Manual mode.
Minimum Energy Threshold for FSK Detection	Sets the minimum energy threshold for the FSK detection. Range of value: -1125~0, calculated by 0.1dB, with the default value of *-380*.
Minimum Number of FSK Seizure Bits	The FSK seizure bits (0x55) under the on-hook mode. As the protocol provides, it shall be 300 consecutive bit groups of 0 and 1. Range of value: 0~32768, with the default value of *100*.
Minimum Number of FSK Mark Bits on on-hook	The number of FSK marking signals (0xFF) under the on-hook mode. As the protocol provides, it shall be 180 consecutive bits of 1. Range of value: 0~32768, with the default value of *40*.
Minimum Number of FSK Mark Bits on off-hook	The numbers of FSK marking signals (0xFF) under the off-hook mode. As the protocol provides, it shall be 80 consecutive bits of 1. Range of value: 0~32768, with the default value of *50*.
Reception Interval of DTMF CallerID	The time interval between digits of the DTMF CallerID from FXO port, calculated by ms, with the default value of 250.
Flash Time	Sets the time for generating a flash signal on the analog trunk. Range of value: 32~1000, calculated by ms, with the default value of 100.
Delay after Dial	Sets the delay to send the CalleeID to PBX after you pick up and dial. Range of value: 200~2000, calculated by ms, with the default value of 1000.

FXO Pick-up Delay after INVITE Received at IP Side	Once this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message.
Maximum Wait Answer Time	The maximum time to wait the answer of the remote side for an outgoing call from FXO port. This item is valid only when Polarity Reversal is enabled. It is calculated by s, with the default value of 25.
Communication without Network	Automatically routes a call to the proper port according to the configuration in case of network failure or call timeout. The default value is *enabled*.
Communicate without Network Mode	Sets the mode for the communications without network, two options available: Auto Search Idle Channel and Use Current Route Setting, with the default value of *Auto Search Idle Channel*. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of Tel->IP route.
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.
Delay to Send 200 OK to IP Side	Once this feature is enabled, the gateway will delay to send 200 OK message to the IP side. The default value is *disabled*.
Open Session In Advance	Once this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is *enabled*.
Avoid Being Detected as Flash Signal by PBX	Once this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default value is *disabled*.
Work Mode	Sets the work mode for the echo canceller. There are two options: *Near-end cancellation* and *Both near-end and far-end cancellation*, with the default value of *Near-end cancellation*.
Non-linear Processing	Sets whether to enable the mode of non-linear processing. By default, this feature is *enabled*.
Fixed Window Size	Sets the size of the window for the fixed cancellation.
Moving Window Size	Sets the size of the window for the moving cancellation.
Select Area for Module Parameters	Select an area for hardware parameters of the FXO chip.
Tone Detector Prior to DTMF Detector	Sets the priorities of the Tone detector and the DTMF detector.

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 3.9.16 Restart for detailed instructions.

3.5.3 Tone Detector

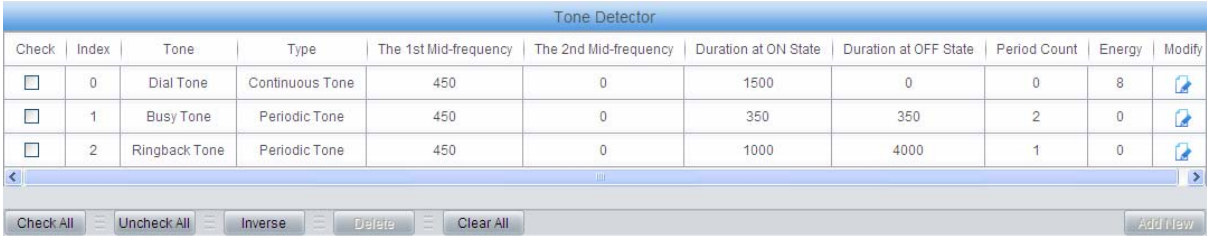


Figure 3-30 Tone Parameters Setting Interface

See Figure 3-30 for the Tone Parameters setting interface. At most three pieces of tone parameters are allowed to set. By default, there are already three pieces of tone parameters on the gateway which you can modify or delete according to your actual requirement.

Click ***Modify*** in Figure 3-30 to modify the tone parameter. See Figure 3-31 for the tone parameter modification interface.

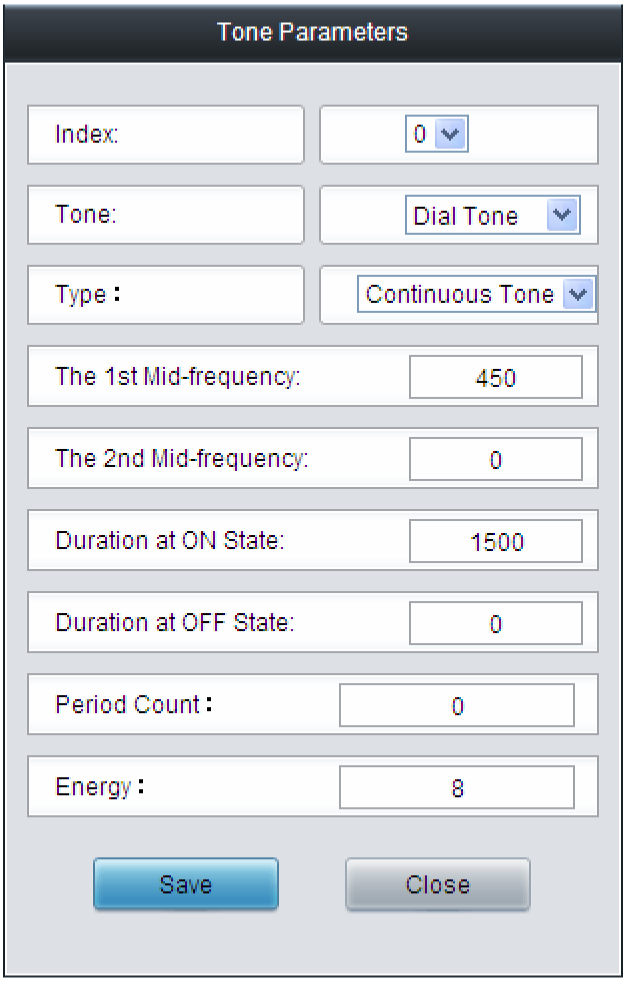


Figure 3-31 Modify Tone Parameter

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

Item	Description
Index	The unique index of each group of tone detectors.
Tone	There are three options: ***Dial Tone***, ***Busy Tone*** and ***Ringback Tone***.
Type	There are two options: ***Continuous Tone*** and ***Periodic Tone***.

The 1st Mid-frequency	The 1st center frequency. Range of value: 300~3400, calculated by Hz. The default value is 450.
The 2nd Mid-frequency	The 2nd center frequency. Range of value: 0 or 300~3400, calculated by Hz. The default value is 0.
Duration at ON State	The duration of tones at on state. The default setting: Dial Tone is 1500ms, Busy Tone is 350ms, Ringback Tone is 1000ms.
Duration at OFF State	The duration of tones at off state. The default setting: Dial Tone is 0ms, Busy Tone is 350ms, Ringback Tone is 4000ms.
Period Count	Sets the count of periods as the condition to determine a periodic tone. The default setting: Dial Tone is 0, Busy Tone is 2, Ringback Tone is 1.
Energy	Sets the energy threshold for the tone detector to detect the on-line tone. To increase the accuracy, you can adjust the value according to the tone volume on the line. Range of value: -18~11, calculated by dB. The default value is 0.

To delete a piece of tone, 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 tone at a time, click the ***Clear All*** button in Figure 3-30.

3.5.4 Tone Generator

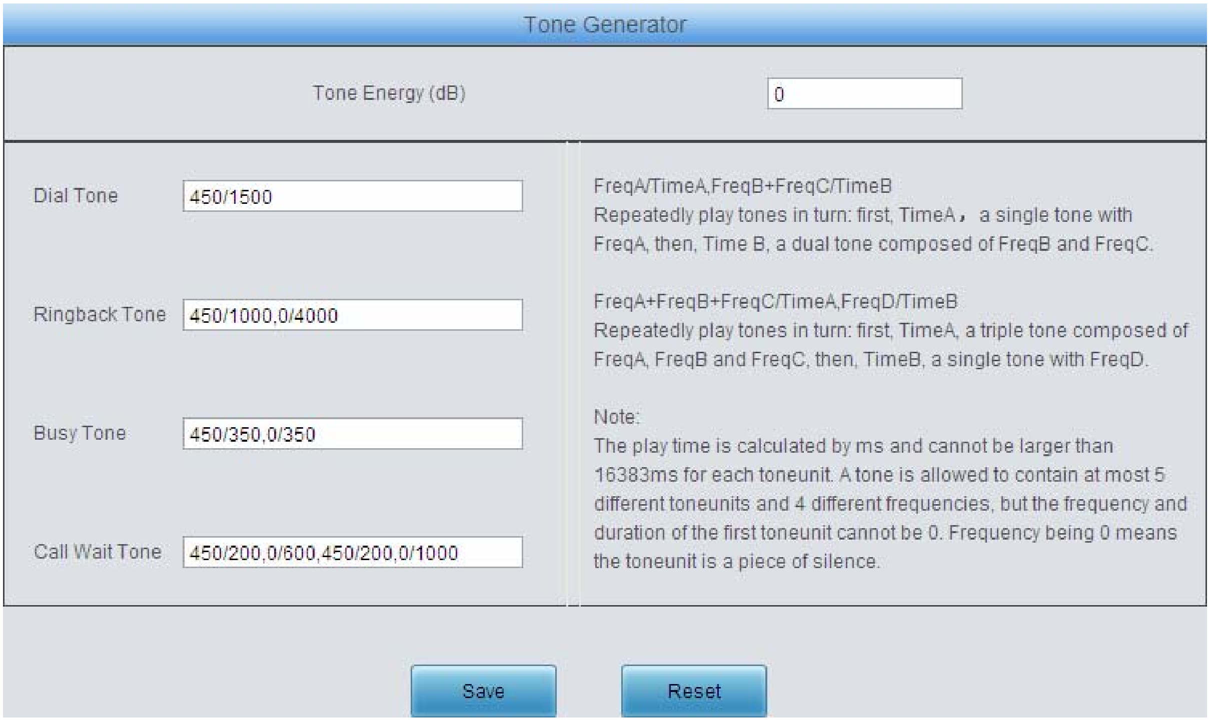


Figure 3-32 Tone Generator Setting Interface

See Figure 3-32 for the Tone Generator Setting interface. By default, there are four 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. Call Wait Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 200ms play and 600ms pause, then 200ms play and 1s 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 -18~11, calculated by dB, with the default value of 0.

3.5.5 DTMF

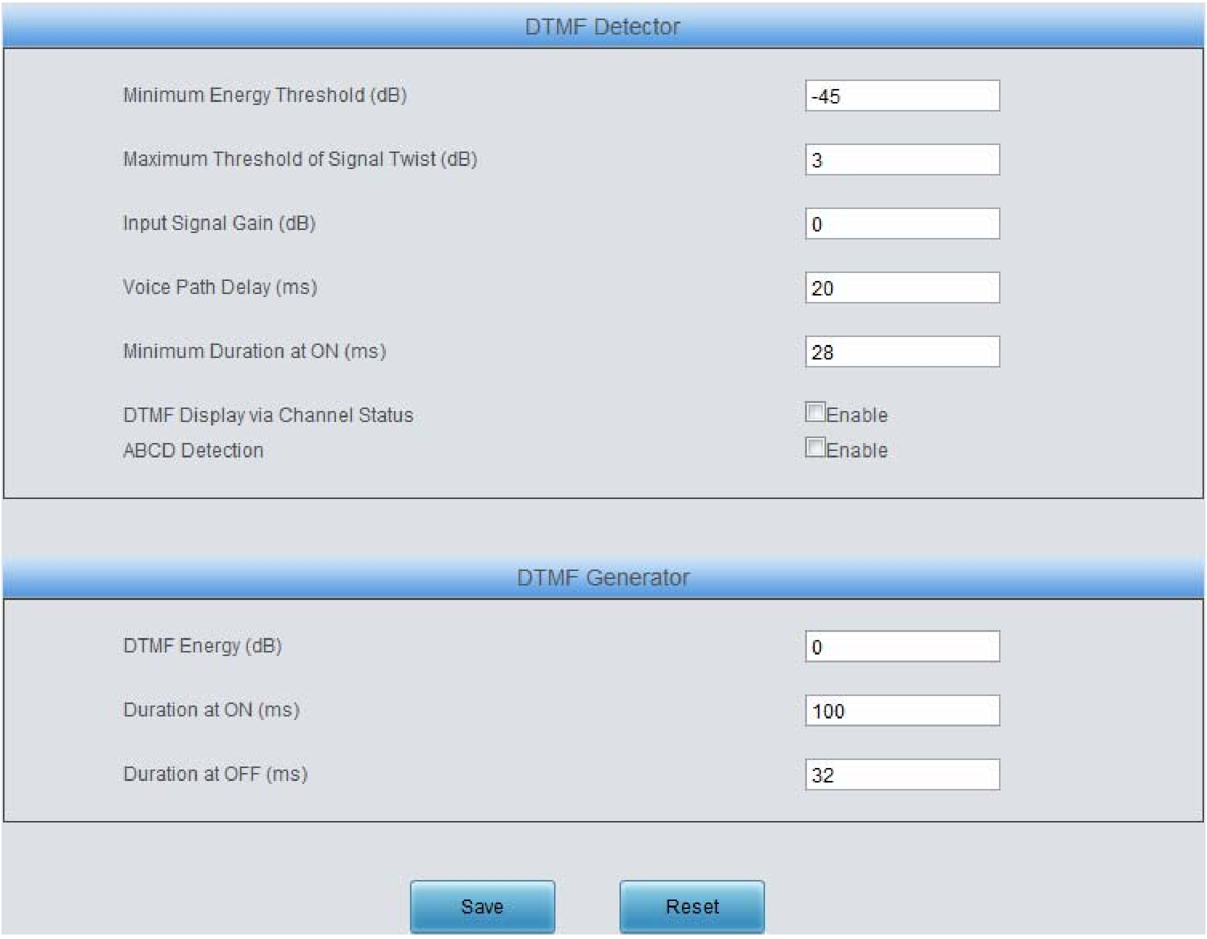


Figure 3-33 DTMF Detector Configuration Interface

See Figure 3-33 for the DTMF configuration, including two parts: DTMF Detector and DTMF Generator. The table below explains the items shown in the above figure.

Item	Description
Minimum Energy Threshold	Set the minimum energy threshold of the DTMF signal. Range of value: -96~-1. The default value is -45.
Maximum Threshold of Signal Twist	Set the maximum threshold of the DTMF signal twist. Range of value: 0~12. The default value is 3.
Input Signal Gain	Set the input gain of the DTMF signal. Range of value: -24～24, calculated by dB. The default value is 0.
Voice Path Delay	Once this feature is enabled, the DTMF in the voice data will be clamped. Range of value: 0～20, calculated by ms. The default value is 20.
Minimum Duration at ON	Set the minimum duration at ON for the DTMF signal. Range of value: 10～60, calculated by ms. The default value is 28.
DTMF Display via Channels Status	Once this feature is enabled, the received/sent DTMF will be displayed upon you putting the mouse on the icon of channel status. The default value is disabled.

ABCD Detection	Once this feature is enabled, the gateway can detect the DTMF digits A, B, C and D (Case-insensitive). The default value is disabled.
DTMF Energy	Energy of the DTMF signal sent by the gateway. Range of value: -18~11, calculated by dB, with the default value of 0.
Duration at ON	Set the duration of the DTMF signal at ON state. Range of value: 0~16383, calculated by ms, with the default value of 100.
Duration at OFF	Set the duration of the DTMF signal at OFF state. Range of value: 0~16383, calculated by ms, with the default value of 32.

After configuration, click ***Save*** to save your settings into the gateway. 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 3.9.16 Restart for detailed instructions. Click ***Reset*** to restore the configurations.

3.5.6 Ringing Scheme

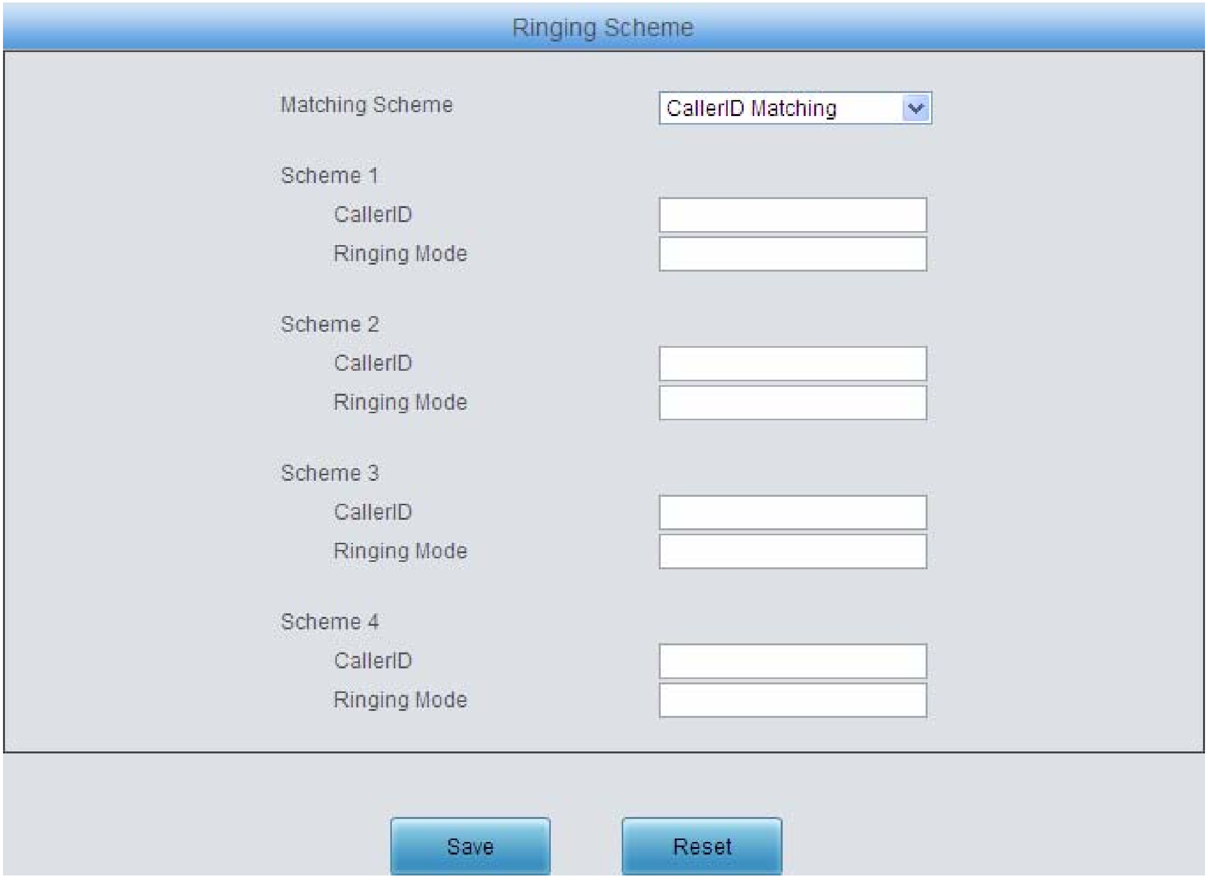


Figure 3-34 Ringing Scheme Configuration Interface

See Figure 3-34 for the Ringing Scheme Configuration interface. The gateway can execute different ringing schemes according to the callerID or Alert-Info..

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

Item	Description

CallerID	The gateway will match the CallerID set in this item to that of the incoming call. If they are matched, the current ringing scheme will be executed; otherwise, the default ringing scheme (1 sec on and 4 sec off) will work. The rule to fill in the CallerID is the same as that of 3.5.9 Dialing Rule. Multiple CallerIDs are supported; they should be separated by ","
Alert-Info Value	The gateway will match the alert-info value set in this item to that of the incoming call. If they are matched, the current ringing scheme will be executed; otherwise, the default ringing scheme (1 sec on and 4 sec off) will work..
Ringing Scheme	The ringing scheme can be "1,X,Y" or "2,X,Y,M,N", in which, the number 1 or 2 denotes one group or two groups; X, M denote the duration at on state while Y, N denote the duration at off state. Note: The duration at ON or OFF cannot be greater than 12000ms, the total duration at ON and OFF cannot be greater than 16000ms, and N - the last duration at OFF cannot be less than 1800ms if the item "Occasion to Send FSK CallerID" is set to After the first ring.

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

3.5.7 Fax

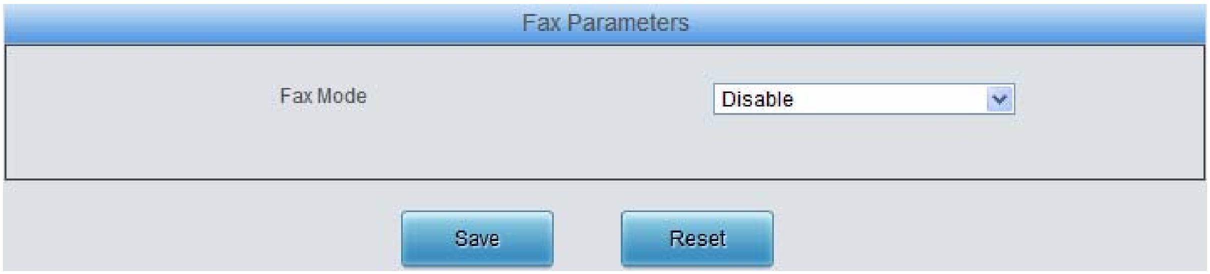


Figure 3-35 Fax Configuration Interface (Disable by default)

See Figure 3-35 for the default fax mode configuration. The table below explains the items shown in the above figure.

Item	Description
Fax Mode	The real-time IP fax mode. The optional values are *T.38*, *Pass-through* and *Disable*, and the default value is *Disable* which means to disable both T.38 and Pass-through.

See Figure 3-36 for the fax configuration under the T.38 mode.

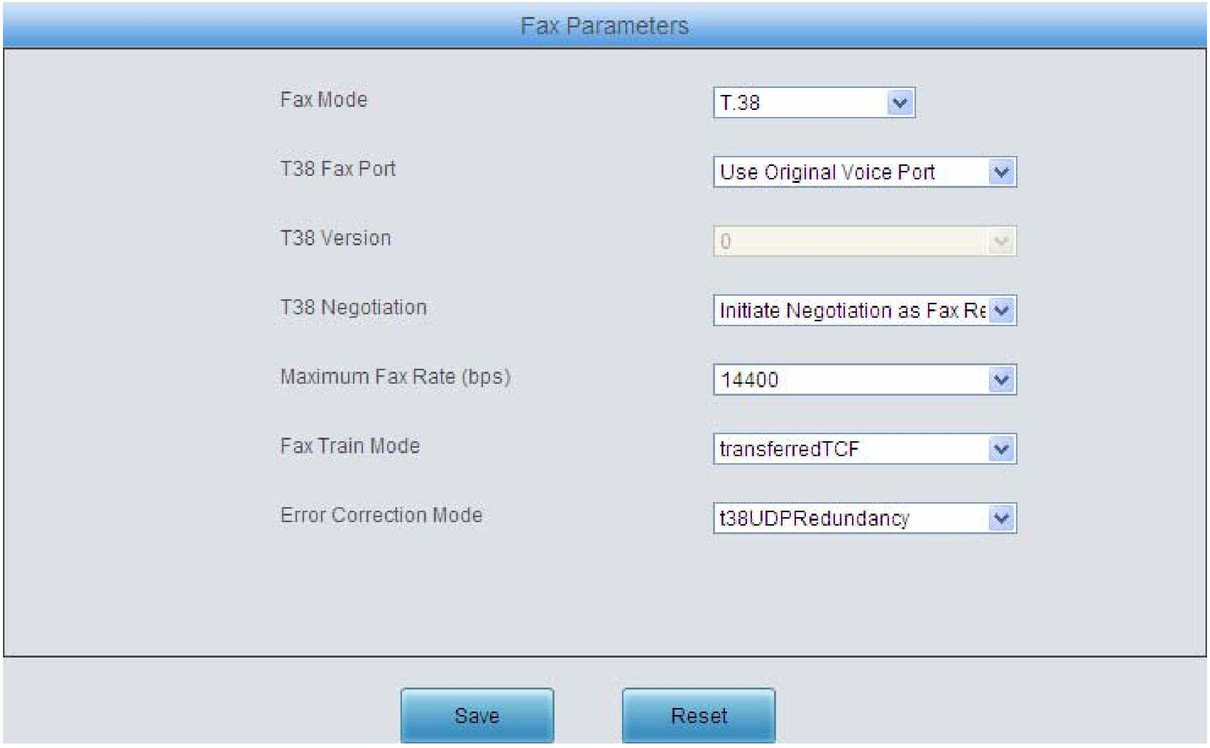


Figure 3-36 Fax Configuration Interface (T.38 Mode)

Users can configure the general fax parameters via this interface. After configuration, click ***Save*** to save your settings into the gateway. 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 3.9.16 Restart for detailed instructions. click ***Reset*** to restore the configurations. The table below explains the configuration items in Figure 3-36.

Item	Description
T38 Fax Port	The port for T.38 faxing, providing two options: ***Use Original Voice Port*** and ***Use New Port***. The default setting is *Use Original Voice Port*.
T38 Version	Version of T.38 which is defined by ITU-T.
T38 Negotiation	The Negotiation mode of T.38, providing two options: *Initiate Negotiation as Fax Sender* and *Initiate Negotiation as Fax Receiver*. The default value is *Initiate Negotiation as Fax Receiver*.
Maximum Fax Rate	Sets the maximum faxing rate for both receiving and transmitting. Range of value: 14400, 9600 and 4800, calculated by bps, with the default value of 14400.
Fax Train Mode	Sets the train mode for T.38 fax. The optional values are *transferredTCF* and *localTCF*, with the default value of *transferredTCF*.
Error Correction Mode	Sets the error correction mode for T.38 fax. The optional values are *t38UDPRedundancy* (Redundancy Error Correction) and *t38UDPFEC* (Forward Error Correction), with the default value of *t38UDPRedundancy*.

If you set ***Fax Mode*** to *Pass-through*, you can see the interface shown as Figure 3-37.

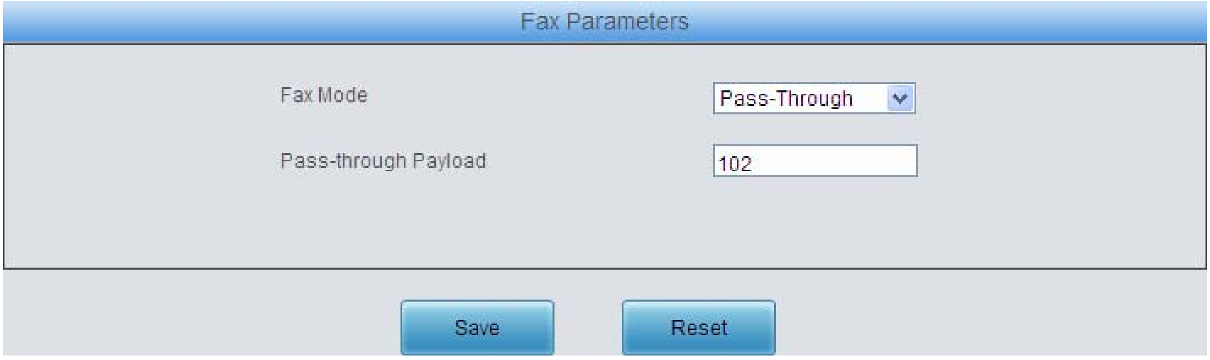


Figure 3-37 Fax Configuration Interface (Pass-through Mode)

The table below explains the configuration item in the above figure.

Item	Description
Pass-through Payload	RTP Payload under the pass-through fax mode. Range of value: 96~127, with the default value of 102.

3.5.8 Function Key

See Figure 3-38 for the Function Key Configuration interface. Here you can set a cluster of combination keys to query a related number.

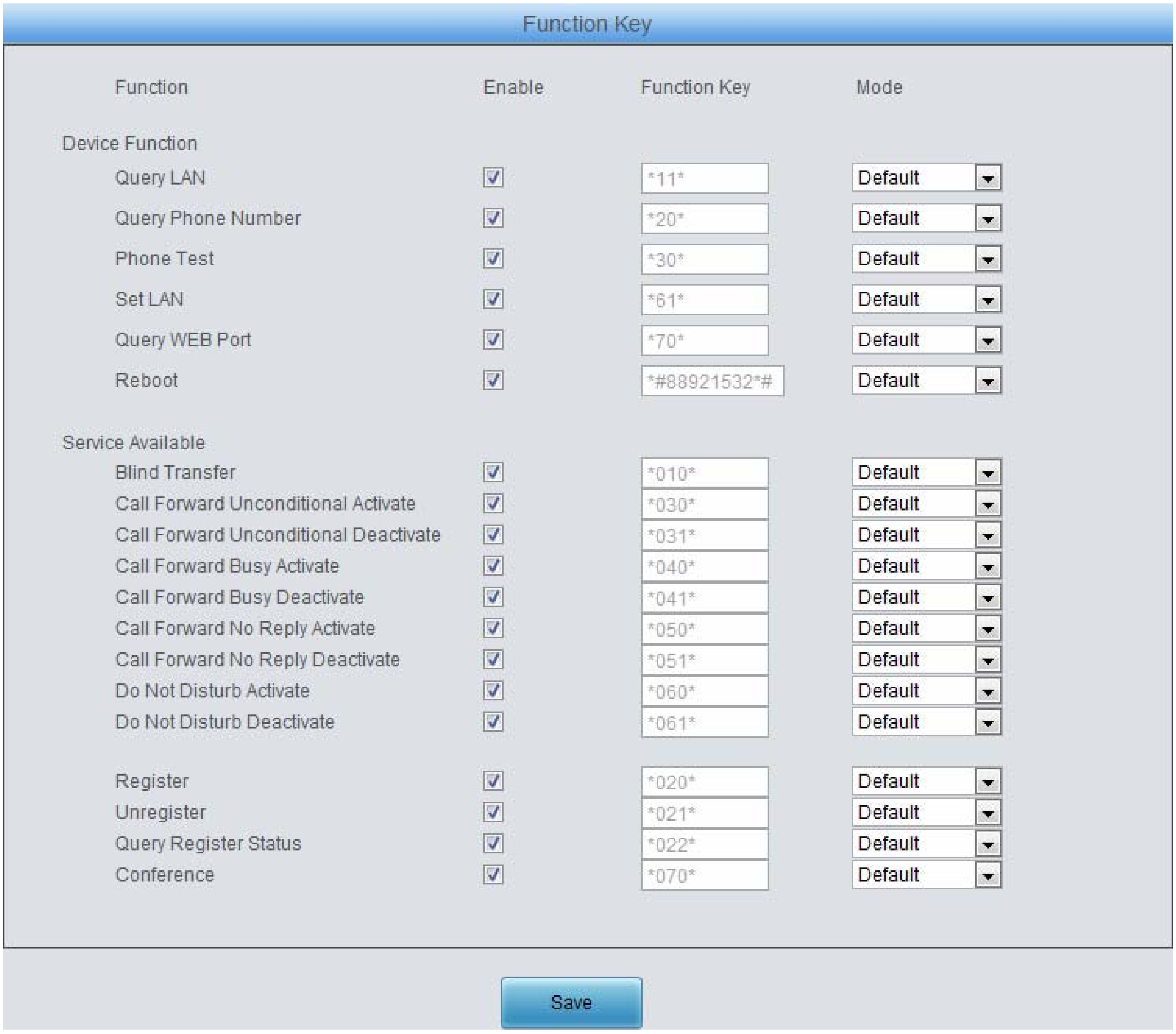


Figure 3-38 Function Key Configuration Interface

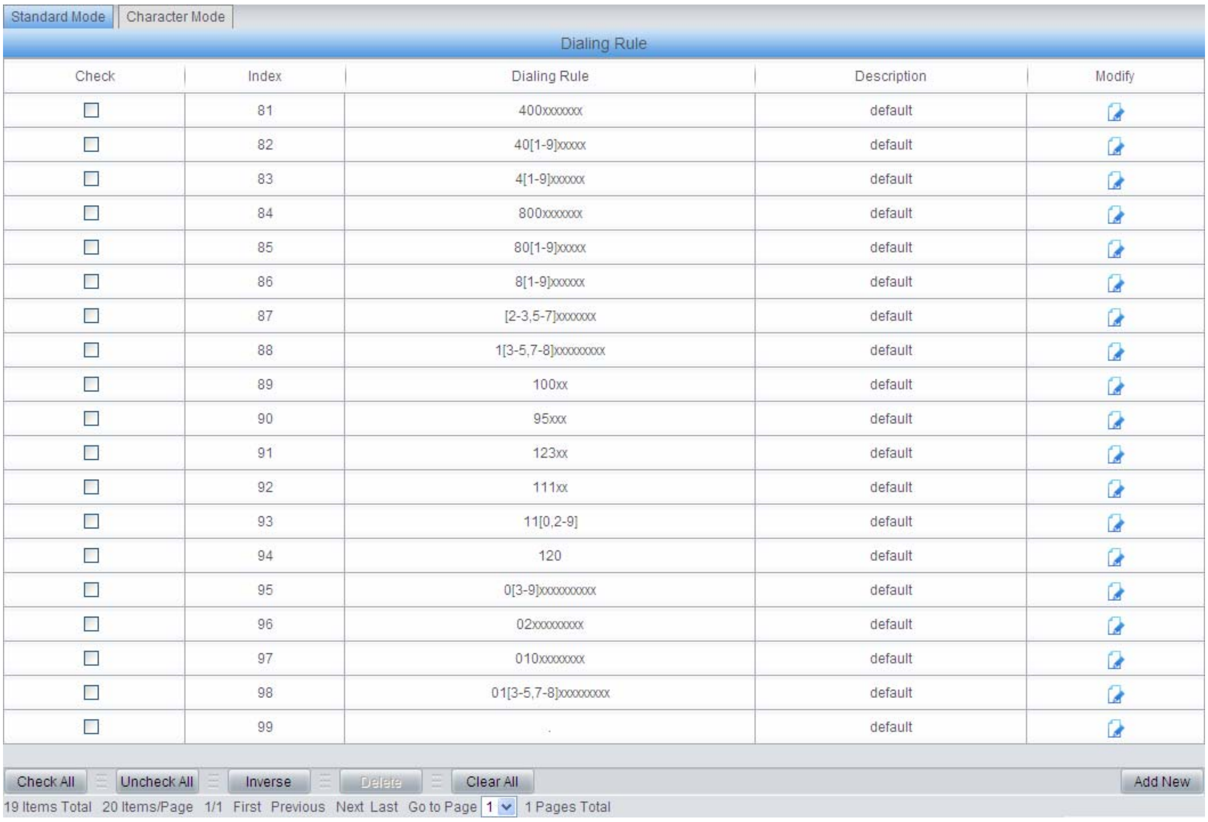
Click "Enable" to enable the corresponding function key. The gateway will use the default function keys when the mode is set to default; and it will allow you to set new function keys when the mode is set to user-defined. Click ***Save*** to save your settings into the gateway.

Note: Phone Test is used just to see if the phone can work normally. It requires you to hang up the phone after dialing the corresponding combination keys. Then the gateway will ring the phone. At that time, pick up the phone and you can hear the voice prompt played by the gateway (e.g. 'Test successful.')

When the **Blind Transfer** feature is enabled, set a corresponding function key in the box behind. After you transfer a call by rapidly clapping on the hook switch, dial the set function key for **Blind Transfer** and then the called party number. After that, hang up the call once hearing the howler tone to let the subsequent call procedure go out of your control.

3.5.9 Dialing Rule

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


Standard Mode | Character Mode

Dialing Rule

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

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

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

Figure 3-39 Dialing Rule Configuration Interface (Standard)

See Figure 3-39 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 3-40 for the dialing rule adding interface.

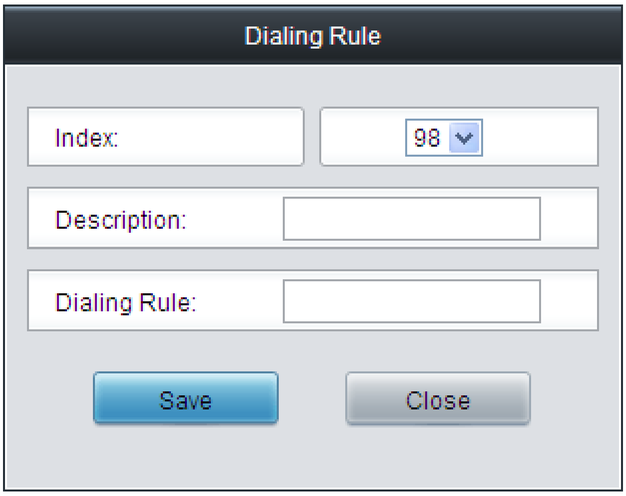


Figure 3-40 Add New Dialing Rule

The table below explains the items shown in Figure 3-40.

<table>
<tr><th>Item</th><th>Description</th></tr>
<tr><td>Index</td><td>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.</td></tr>
<tr><td>Description</td><td>Remarks for the dialing rule. It can be any information, but can not be left empty.</td></tr>
<tr><td>Dialing Rule</td><td>Up to 100 dialing rules can be configured in the gateway, and the maximum length of each dialing rule is 127 characters. See below for the meaning of each character in the dialing rule. The gateway will do instant matching for your dialing number based on the dialing rule and regard your dialing as finished upon receiving ‘#’ or dialing timeout.
<table>
<tr><th>Character</th><th>Description</th></tr>
<tr><td>“0”~”9”</td><td>Digits 0～9.</td></tr>
<tr><td>“A”~”D”</td><td>Letters A～D.</td></tr>
<tr><td>“x”</td><td>A random number. A string of ‘x’s represents several random numbers. For example, ‘xxx’ denotes 3 random numbers.</td></tr>
<tr><td>“.”</td><td>‘.’ indicates a random amount (including zero) of characters after it.</td></tr>
<tr><td>“[ ]”</td><td>‘[ ]’ 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.</td></tr>
<tr><td>“-”</td><td>’-’ is used only in ‘[ ]’ between two numbers to indicates any number between these two numbers.</td></tr>
<tr><td>“,”</td><td>’,’ is used to separate numbers or number ranges, representing alternatives.</td></tr>
<tr><td>“*”</td><td>Only represents symbol ”*”.</td></tr>
<tr><td>“#”</td><td>Only set it at the beginning of the string, representing symbol “#”.</td></tr>
</table>
There are 19 dialing rules already configured on the gateway for easy use. See below for detailed information.
<table>
<tr><th>Priority</th><th>Dialing Rule</th><th>Description</th></tr>
</table>
</td></tr>
</table>

	99	.	Any number in any length.
	98	01[3-5,7-8]xxxxxxxxx.	Any 12-digit number starting with 013, 014, 015, 017 or 018
	97	010xxxxxxxx	Any 11-digit number starting with 010
	96	02xxxxxxxxx	Any 11-digit number starting with 02
	95	0[3-9]xxxxxxxxxx	Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09
	94	120	Number 120。
	93	11[0,2-9]	Number 110, 112, 113, 114, 115, 116, 117, 118 or 119
	92	111xx	Any 5-digit number starting with 111
	91	123xx	Any 5-digit number starting with 123
	90	95xxx	Any 5-digit number starting with 95
	89	100xx	Any 5-digit number starting with 100
	88	1[3-5,7-8]xxxxxxxxx	Any 11-digit number starting with 13, 14, 15, 17 or 18
	87	[2-3,5-7]xxxxxxx	Any 8-digit number starting with 2, 3, 5, 6 or 7
	86	8[1-9]xxxxxx	Any 8-digit number starting with 81, 82, 83, 84, 85, 86, 87, 88 or 89
	85	80[1-9]xxxxx	Any 8-digit number starting with 801, 802, 803, 804, 805,.806, 807, 808 or 809
	84	800xxxxxxx	Any 10-digit number starting with 800
	83	4[1-9]xxxxxx	Any 8-digit number starting with 41, 42, 43, 44, 45, 46, 47, 48 or 49.
	82	40[1-9]xxxxx	Any 8-digit number starting with 401, 402, 403, 404, 405, 406, 407, 408 or 409
	81	400xxxxxxx	Any 10-digit number starting with 400

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

Click ***Modify*** in Figure 3-39 to modify the dialing rules. See Figure 3-41 for the dialing rule modification interface. The configuration items on this interface are the same as those on the ***Add New Dialing Rule*** interface.

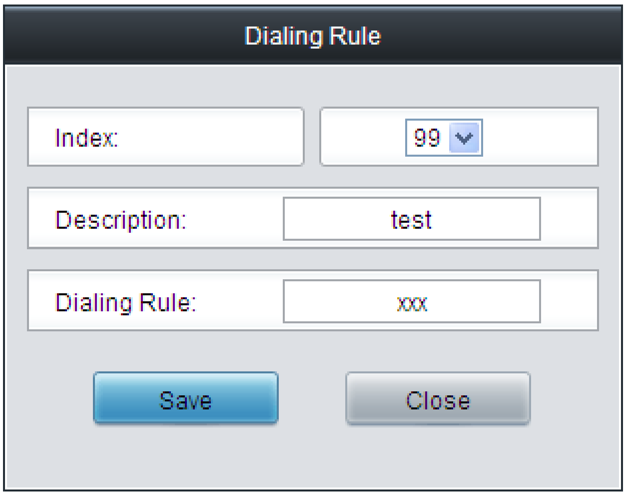


Figure 3-41 Modify Dialing Rule

To delete a dialing rule, check the checkbox before the corresponding index in Figure 3-39 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 3-39.

See Figure 3-42 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.

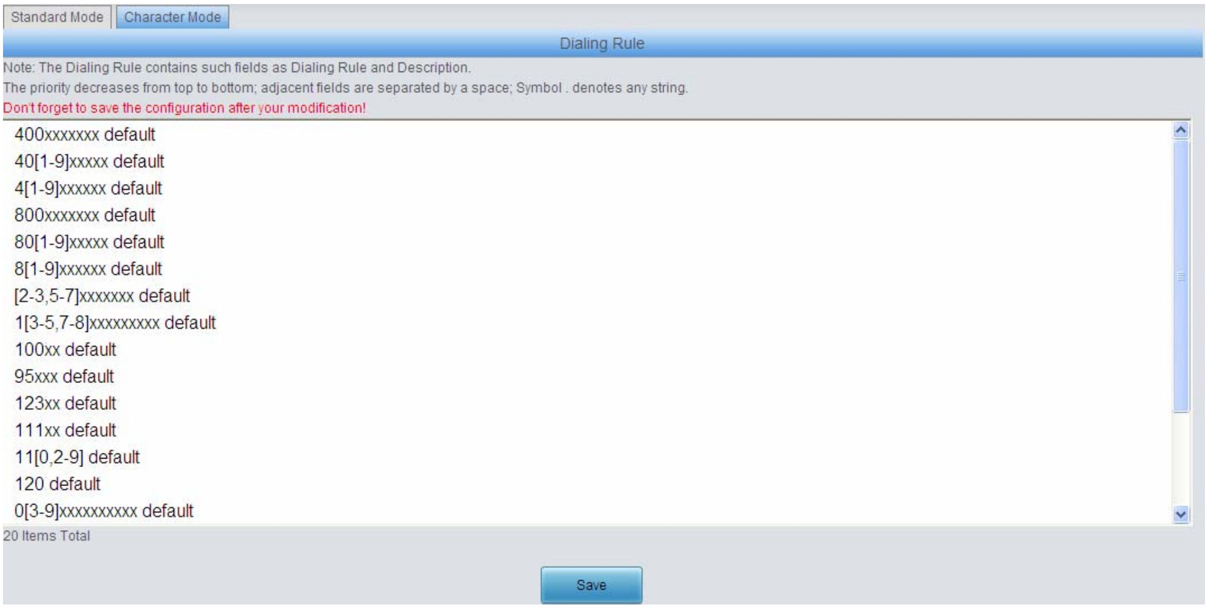
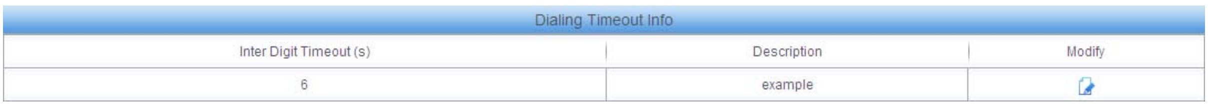


Figure 3-42 Dialing Rule Configuration Interface (Character)

3.5.10 Dialing Timeout



Dialing Timeout Info		
Inter Digit Timeout (s)	Description	Modify
6	example	

Figure 3-43 Dialing Timeout Info Interface

See Figure 3-43 for the dialing timeout info interface. The table below explains the items shown in the above figure.

Item	Description
Inter Digit Timeout	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.
Description	More information about the configuration item ***Inter Digit Timeout***, such as the reason for adopting the current value.

Click ***Modify*** in Figure 3-43 to modify the dialing timeout info. See Figure 3-44 for the dialing timeout info modification interface. The configuration items on this interface are the same as those on the ***Dialing Timeout Info Interface***.

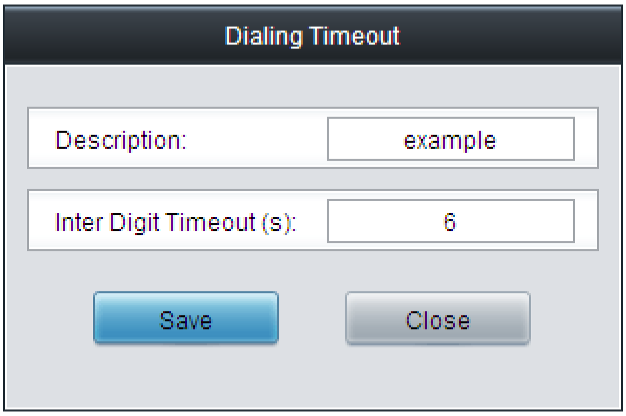


Figure 3-44 Modify Dialing Timeout Info

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

3.5.11 Cue Tone

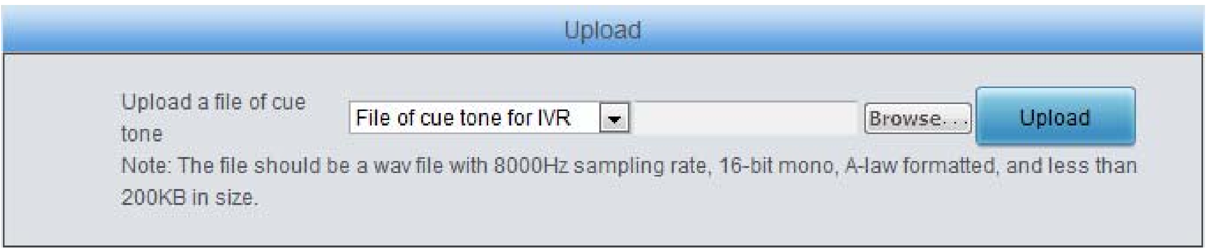


Figure 3-45 Cue Tone Interface

See Figure 3-45 for the Cue Tone interface. The table below explains the items shown in the above figure.

Item	Description
Upload a file of cue tone	Uploads a user-defined cue tone file to the gateway.

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

3.5.12 Color Ring

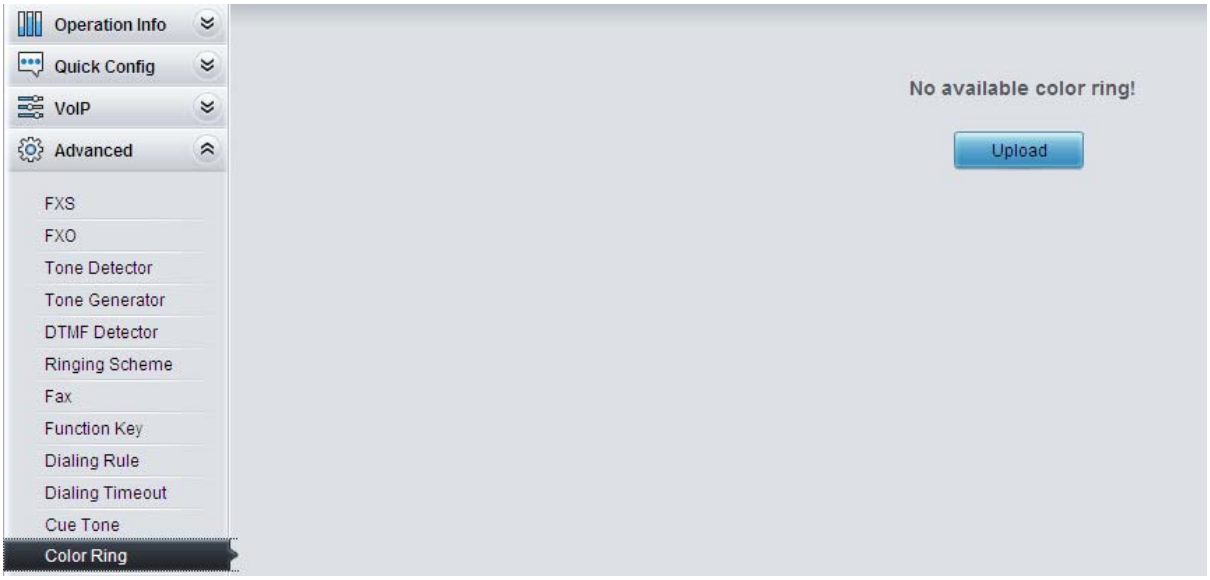


Figure 3-46 Coloring Ring Interface

By default, there is no available color ring on the gateway. See Figure 3-46. Click ***Upload*** to upload a new color ring manually. Follow Figure 3-47 to upload the required color ring file to the gateway.

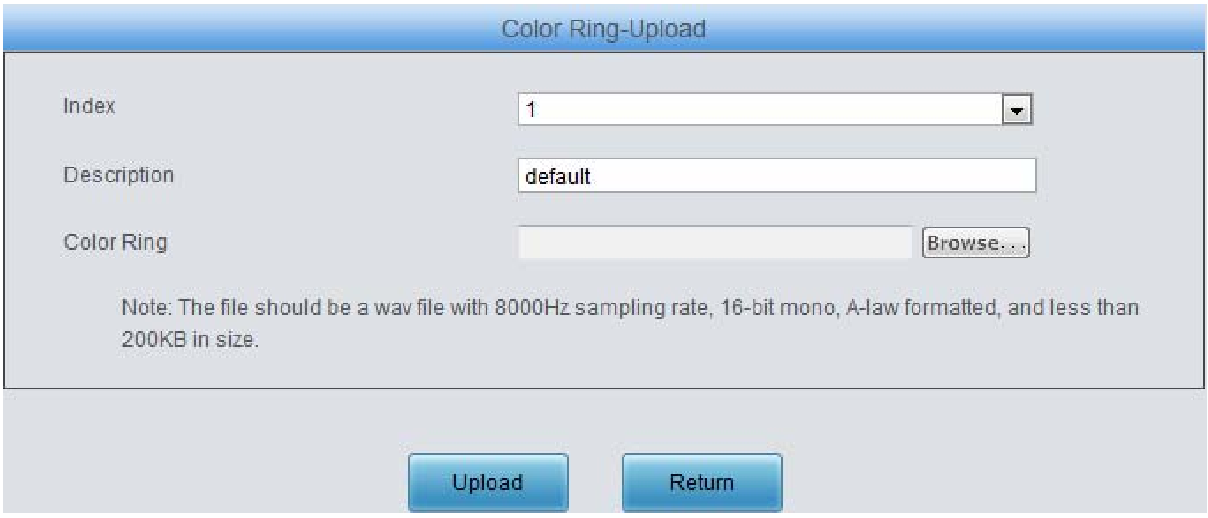


Figure 3-47 Color Ring Upload Interface

The table below explains the items shown above:

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

After configuration, click ***Upload*** to upload the color ring file to the gateway or click ***Return*** to cancel the upload. See Figure 3-48 for the Color Ring Management interface after the upload.

Figure 3-48 Color Ring Management Interface

Click ***Modify*** in Figure 3-48 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 3-49 Color Ring Modification Interface

To delete a color ring, check the checkbox before the corresponding index in Figure 3-48 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 3-49.

3.5.13 QoS

Figure 3-50 Differentiated Services Setting Interface

See Figure 3-50 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
QoS	Sets whether to enable the OoS differentiated services. By default, it is disabled.

Media Premium QoS	Sets the priority of the media premium for QoS. A media premium QoS with a bigger value has a higher priority. The value range is 0~63, with the default value of 46.
Control Premium QoS	Sets the priority of the control premium for QoS. A control premium QoS with a bigger value has a higher priority. The value range is 0~63, with the default value of 26.

3.5.14 Action URL

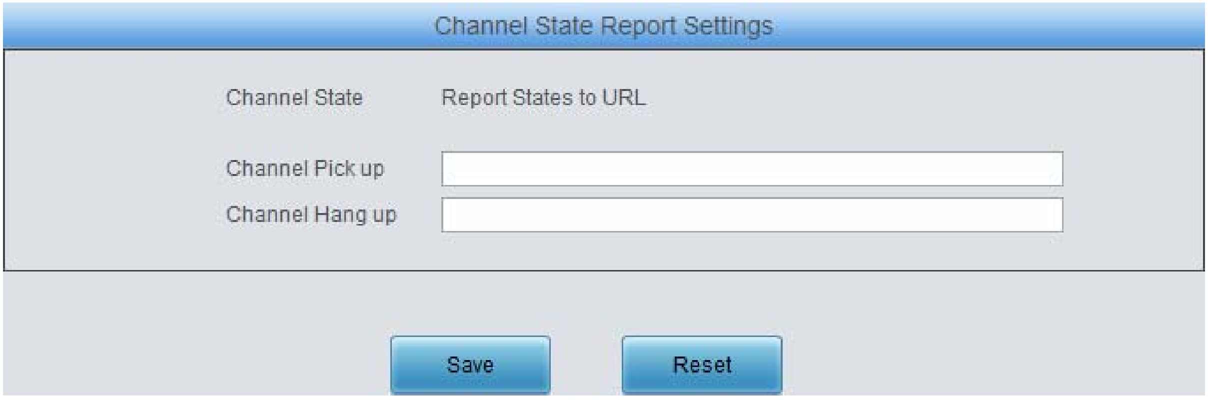


Figure 3-51 Channel State Report Settings Interface

See Figure 3-51 for the Action URL interface, which is used to designate the server patch to report the on-hook or off-hook state of the FXS channel. You are allowed to designate two different server paths. After setting, the state will be reported to the designated server once any of the FXS channel hangs up or picks up a call. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations.

3.6 Port Settings

Port Settings includes three parts: ***FXS***, ***FXO*** and ***Port Group***. See Figure 3-52.

Figure 3-52 Port Settings

3.6.1 FXS

FXS Settings

Port	Type	SIP Account	Display Name	Auto Dial Num	Forbid Outgoing Call	DND	Forward	FWD Type	FWD Number	CID	Call Waiting	Reg Status	Echo Canceller	Color Ring	Color Ring Index	Input Gain	Output Gain	Modify
1	FXS	8001	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Unregistered	Enable	Disable	---	0	0	
2	FXS	8002	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Unregistered	Enable	Disable	---	0	0	
3	FXS	8003	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Unregistered	Enable	Disable	---	0	0	
4	FXS	8004	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Unregistered	Enable	Disable	---	0	0	
5	FXS	8005	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Unregistered	Enable	Disable	---	0	0	
6	FXS	8006	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Unregistered	Enable	Disable	---	0	0	
7	FXS	8007	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Unregistered	Enable	Disable	---	0	0	
8	FXS	8008	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Unregistered	Enable	Disable	---	0	0	

8 Items Total 16 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total Batch Modify

Figure 3-53 FXS Settings Interface

See Figure 3-53 for the FXS settings interface. The list in the above figure shows the feature and properties of each FXS port. Click ***Modify*** in Figure 3-53 to modify the properties of the corresponding port. See Figure 3-54 for the FXS modification interface.

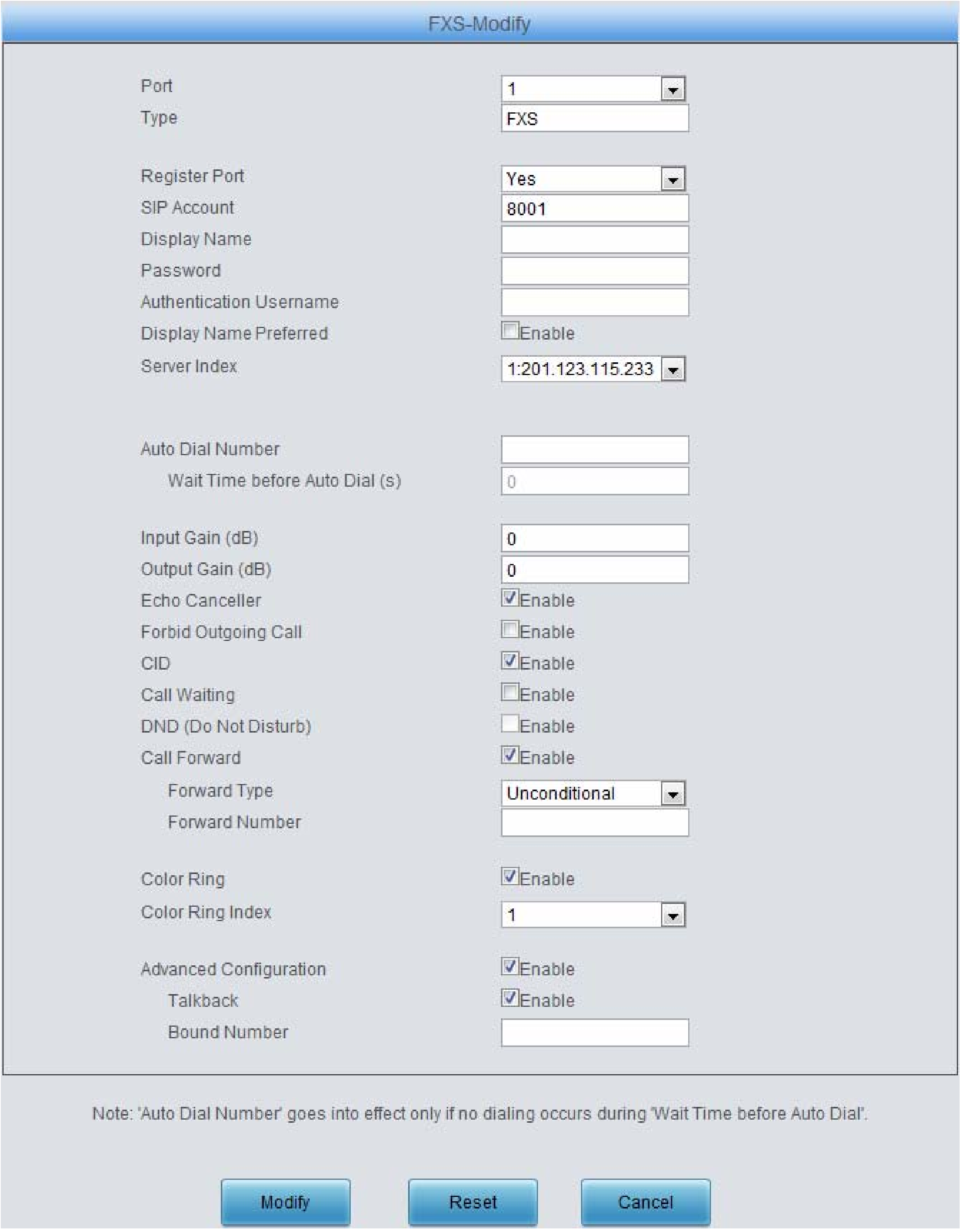


Figure 3-54 FXS Modification

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

Item	Description
Port	Serial number of the FXS port on the device.
Type	Type of the port on the device (FXS). This item is not configurable.

Register Port	Sets whether to register the port to the SIP server. When this item is set to *No*, the item ***Reg Status*** on the FXS settings interface (Figure 3-53) shows *Unregistered*; when this item is set to *Yes*, the item ***Reg Status*** shows *Failed* or *Registered*.
SIP Account	When the port initiates a call to SIP, this item corresponds to the username of SIP. The default SIP account is 80XX among which XX represents the corresponding port number. For example, the default SIP account corresponding to Port 1 is 8001, and that corresponding to Port 8 is 8008.
Display Name	Set the content of the displayname field of the SIP message. If it doesn't set with any value, the displayname field will by default display the content of callerid.
Password	Registration password of the port. To register a port to the SIP server, both items ***SIP Account*** and ***Password*** must be filled in.
Authentication Username	Authentication username of a port, used to register the port to the SIP server when IMS network is enabled. **Note: This item appears only when IMS Network or Multi-Registrar Server is enabled.**
Display Name Preferred	In case this feature is enabled and the port group or the whole gateway is registered, if the display name set by the port are different from that set by the port group, the displayname in the sent SIP message will be the one set by the port. In case this feature is disabled, if the port group is registered, the displayname in the sent SIP message will be the display name set by the port group; if the whole gateway is registered, the displayname in the sent SIP message will be the displayname of the gateway.
Server Index	The index of the SIP server which will be quoted by the current FXS port.
Auto Dial Number, Wait Time before Auto Dial	The FXS port will dial the ***Auto Dial Number*** if there is no dialing operation after pickup within a designated time period (i.e. ***Wait Time before Auto Dial***).
Input Gain, Output Gain	Adjusts the gain of the voice input to/ output from the FXS port. Range of value: -24~12, calculated by dB, with the default value of 0.
Echo Canceller	The echo cancellation feature for a call conversation over the FXS channel. By default, this feature is enabled and the effect can reach 128ms.
Forbid Outgoing Call	If this feature is enabled, the FXS port will be forbidden to call out. The default setting is *disabled*.
CID	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. The default setting is *enabled*. CallerID displays digits only and will filter out any other characters if exist.
Call Waiting	If this feature is enabled, the FXS port 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. The default setting is *disabled*.
DND	Do Not Disturb. If this feature is enabled, the FXS port will reply the 403 message to reject all incoming calls. The default setting is *disabled*.

<table>
<tr><td>Call 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 FWD Type. Note: To enable this feature, do not put the FXS port into a port group with other ports. The default setting is disabled.</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 it is busy upon receiving 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 become valid.</td></tr>
</table>
This item is valid only when Call 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 Call Forward feature is enabled, this item can not be left empty.</td></tr>
<tr><td>Color Ring</td><td>Sets whether to enable the color ring feature or not, with the default setting of being disabled.
Note: Only when there are available color rings will this item appear.</td></tr>
<tr><td>Color Ring Index</td><td>The index of the color ring which will be quoted by the current FXS port.</td></tr>
<tr><td>Talkback</td><td>With this feature enabled and a number bound, the port can talkback to its bound number. That is, they can start a call with each other as soon as picking up the phone. The default setting is disabled.
Note: This feature is only used in the case of channel registration.</td></tr>
<tr><td>Bound Number</td><td>Sets the bound number for talkback.</td></tr>
</table>

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

Or you can click ***Batch*** to modify several pieces of FXS settings at the same time. See Figure 3-55 below for the FXS batch modification interface. The configuration items on this interface are the same as those on the FXS modification interface (Figure 3-54).

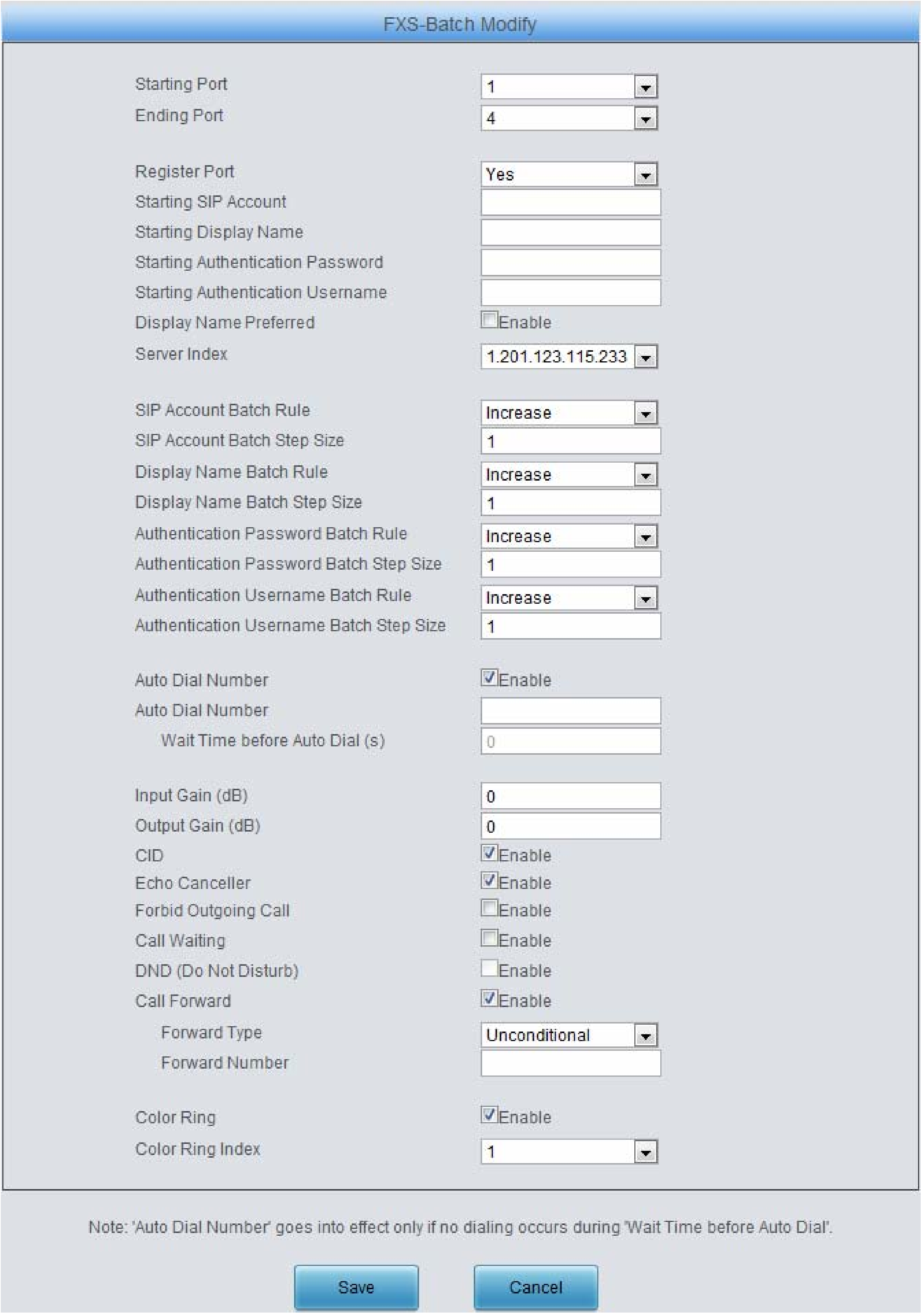


Figure 3-55 FXS Batch Modification

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

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

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

3.6.2 FXO

Figure 3-56 FXO Settings Interface

See Figure 3-56 for the FXO Settings interface. The list in the above figure shows the feature and properties of each FXO port. Click ***Modify*** in Figure 3-56 to modify the properties of the corresponding port. See Figure 3-57 for the FXO Modification interface.

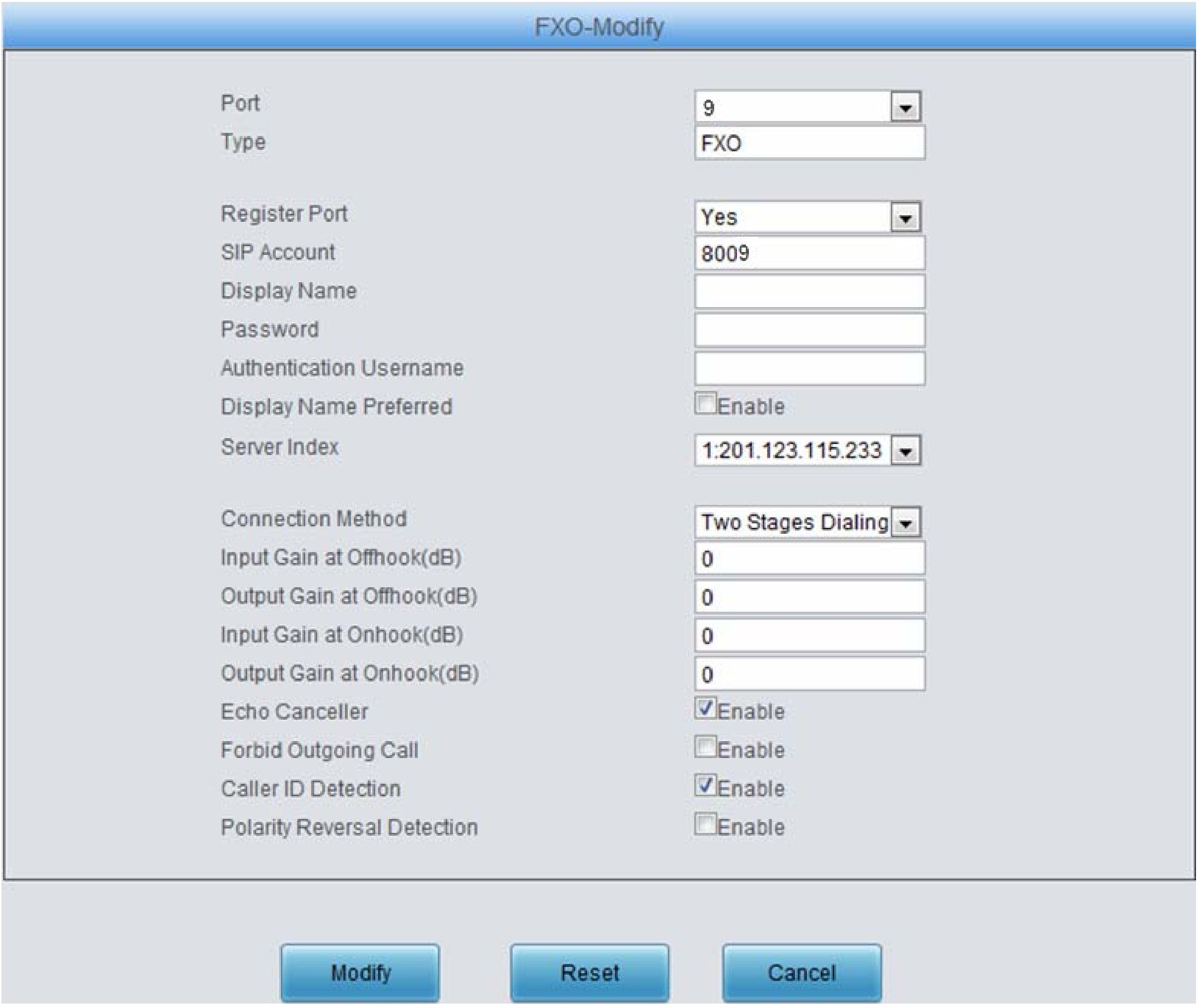


Figure 3-57 FXO Modification

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

Item	Description
Port	Serial number of the FXO port on the device.
Type	Type of the port on the device (FXO). This item is not configurable.
Register Port	Sets whether to register the port to the SIP server. When this item is set to *No*, the item ***Reg Status*** on the FXO settings interface (Figure 3-56) shows *Unregistered*; when this item is set to *Yes*, the item ***Reg Status*** shows *Failed* or *Registered*.
SIP Account	Registration account of an FXO port. 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 32 is 8032.
Display Name	Set the content of the displayname field of the SIP message. If it doesn't set with any value, the displayname field will by default display the content of callerid.
Password	Registration password of the port. To register a port to the SIP server, both items ***SIP Account*** and ***Password*** must be filled in.

<table>
<tr><td>Authentication Username</td><td>Authentication username of a port, used to register the port to the SIP server when IMS network is enabled.
Note: This item appears only when IMS Network or Multi-Registrar Server is enabled.</td></tr>
<tr><td>Display Name Preferred</td><td>In case this feature is enabled and the port group or the whole gateway is registered, if the display names set by the port are different from that set by the port group, the displayname in the sent SIP message will be the one set by the port. In case this feature is disabled, if the port group is registered, the displayname in the sent SIP message will be the display name set by the port group; if the whole gateway is registered, the displayname in the sent SIP message will be the displayname of the gateway.</td></tr>
<tr><td>Server Index</td><td>The index of the SIP server which will be quoted by the current FXO port.</td></tr>
<tr><td>Connection Method</td><td>FXO connection methods include:
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Static Binding</td><td>Bind the number which corresponds to an FXS port to an FXO port. The number will be listed in the Bound Number column. This helps to achieve the corresponding binding between an FXO port and an FXS port (two-way).</td></tr>
<tr><td>Two Stages Dialing Mode (default)</td><td>Under this mode, an incoming call from an FXO 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 station number before IVR finishes the third repeat of the prompt, the FXO will hang up the call automatically; otherwise, the corresponding station will ring.</td></tr>
</table>
Note: Both items Connection Method and Bound Number will be hidden if the SIP Station feature is enabled on the SIP Settings interface.</td></tr>
<tr><td>Input Gain at Offhook/Onhook, Output Gain at Offhook/Onhook</td><td>Adjusts the gain of the voice input to/ output from the FXO port when it is offhook or onhook. Range of value: -24~12, calculated by dB, with the default value of 0.</td></tr>
<tr><td>Echo Canceller</td><td>The echo cancellation feature for a call conversation over the FXO channel. By default, this feature is enabled and the effect can reach 128ms.</td></tr>
<tr><td>Forbid Outgoing Call</td><td>If this feature is enabled, the FXO port will be forbidden to call out. The default setting is disabled.</td></tr>
<tr><td>Caller ID Detection</td><td>If this feature is enabled, the FXO port will detect the caller IDs from the incoming calls. The default setting is enabled.</td></tr>
<tr><td>Polarity Reversal Detection</td><td>Once this feature is enabled, only when the FXO port detects the polarity reversal signal will the corresponding channel go into the talking state. The default setting is disabled. Note: This feature and the Two Stages Dialing feature cannot be enabled at the same time.</td></tr>
</table>

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

Or you can click ***Batch*** to modify several pieces of FXO settings at the same time. See Figure

3-58 below for the FXO Batch Modification interface. The configuration items on this interface are the same as those on the FXO Modification interface (Figure 3-57).

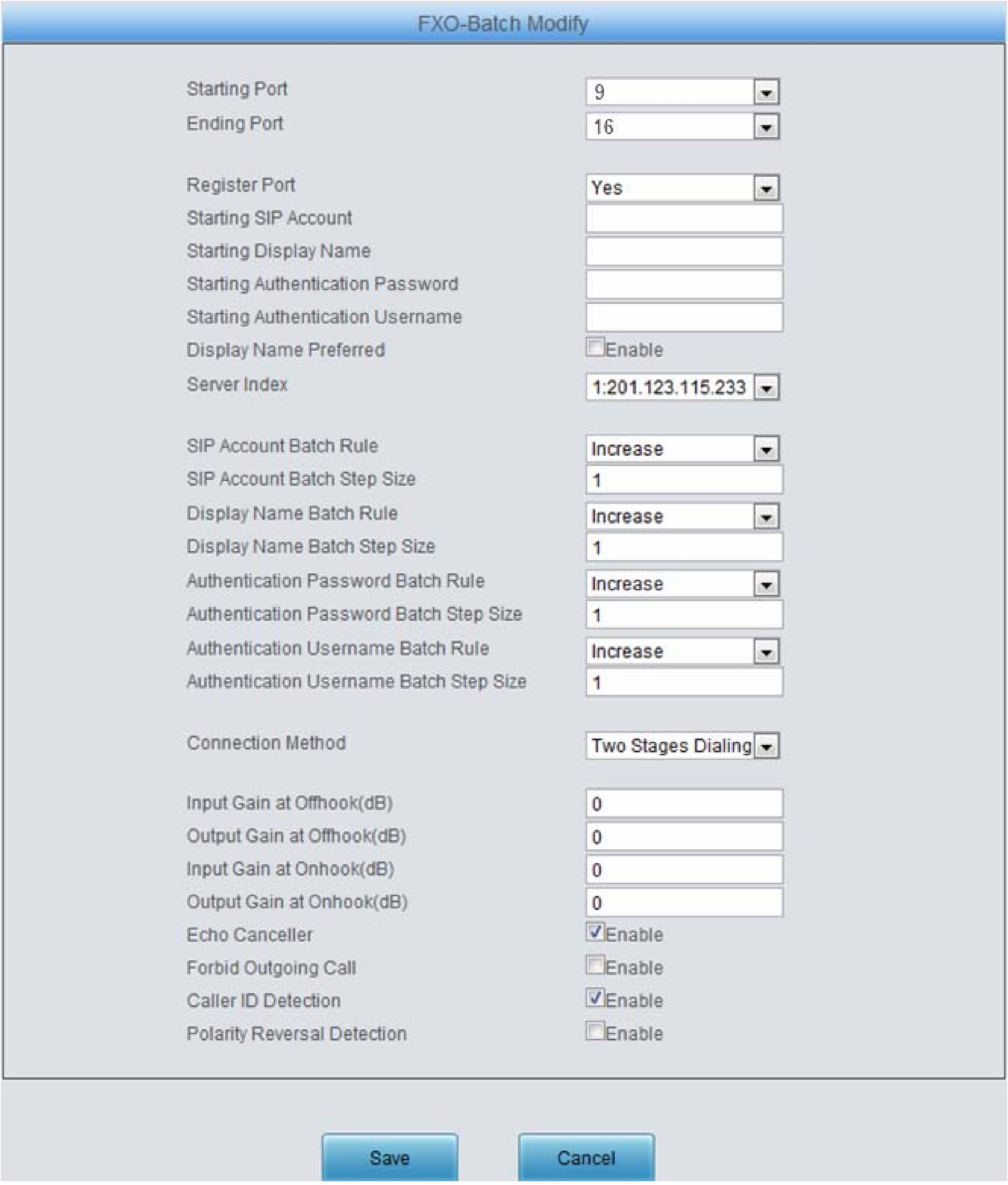


Figure 3-58 FXO Batch Modification

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

Item	Description
Starting Port	The starting serial number of the FXO port on the device in the batch setting.
Ending Port	The ending serial number of the FXO port on the device in the batch setting.
Starting SIP Account	The starting SIP account in the batch setting.
Starting Display Name	The starting displayname in the batch setting.

Starting Authentication Password	The starting authentication password in the batch setting.
Starting Authentication Username	The starting authentication username in the batch setting.
SIP Account Batch Rule	The rule for batch setting the SIP account, including ***Increase*** and ***Decrease*** two options.
SIP Account Batch Step Size	Sets the increase or decrease step size of the SIP account in the batch setting.
Authentication Password Batch Rule	The rule for batch setting the authentication password, including ***Increase*** and ***Decrease*** two options.
Authentication Password Batch Step Size	Sets the increase or decrease step size of the authentication password in the batch setting.
Authentication Username Batch Rule	The rule for batch setting the authentication username, including ***Increase*** and ***Decrease*** two options.
Authentication Username Batch Step Size	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.

3.6.3 Port Group

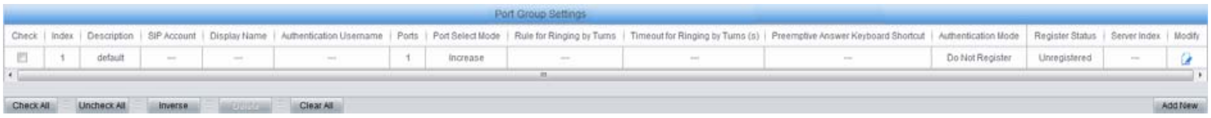


Figure 3-59 Port Group Settings Interface

See Figure 3-59 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 3-60 for the port group adding interface. Note that a port which has been occupied by one port group cannot be chosen by others.

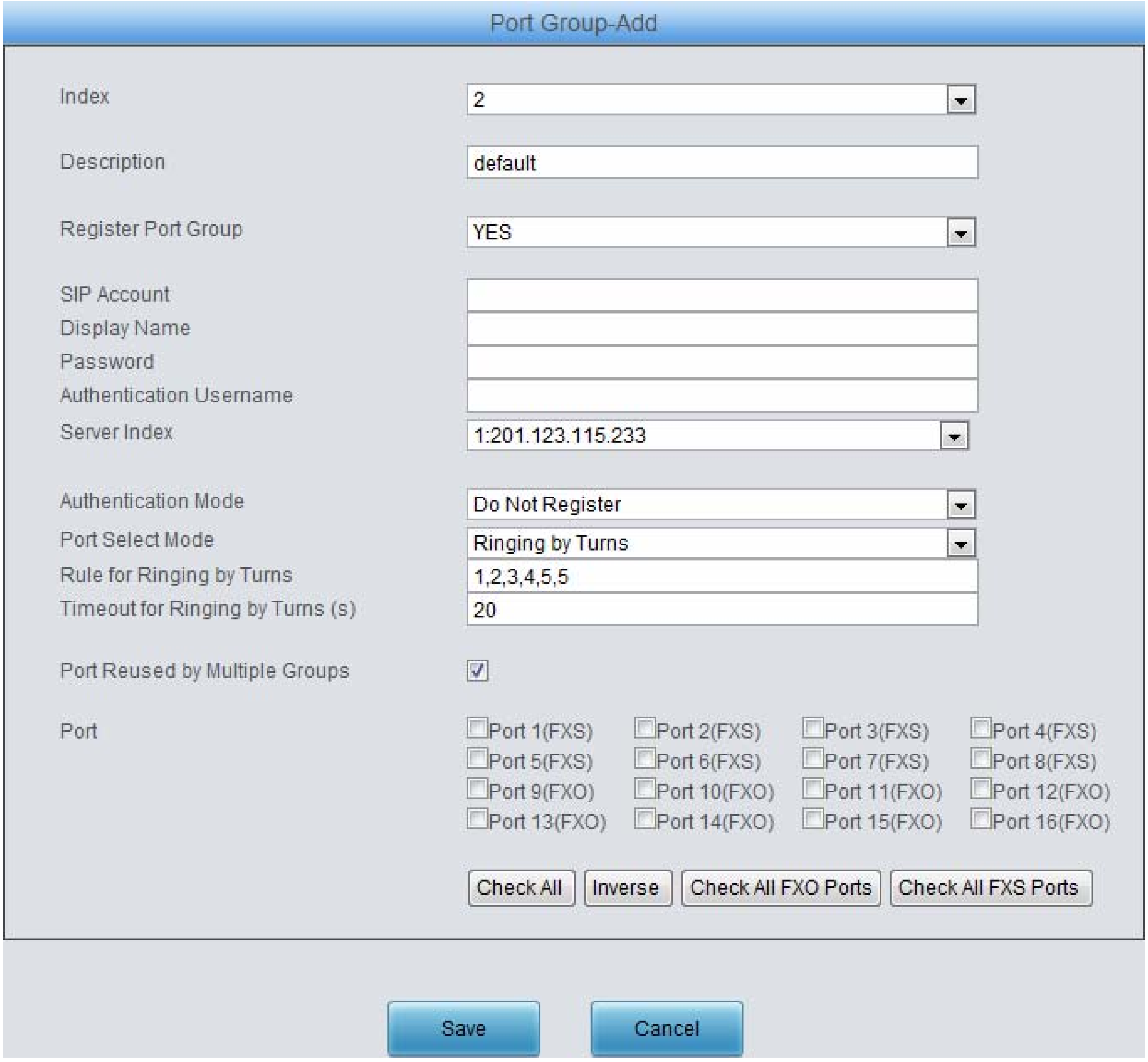


Figure 3-60 Add New Port Group

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, with default value of *default*.
Register Port Group	To register the port group to the SIP server. Only when this configuration item is set to *Yes* can you see the configuration items ***SIP Account*** and ***Password***.
SIP Account	When the port group initiates a call to SIP, this item corresponds to the username of SIP.
Display Name	Set the content of the displayname field of the SIP message. If it doesn't set with any value, the displayname field will by default display the content of callerid.
Password	Registration password of the port group. To register the port group to the SIP server, both configuration items ***SIP Account*** and ***Password*** should be filled in.

<table>
<tr><td>Authentication Username</td><td>Authentication username of a port, used to register the port to the SIP server when IMS network is enabled.
Note: This item appears only when IMS Network or Multi-Registrar Server is enabled.</td></tr>
<tr><td>Server Index</td><td>The index of the sip server which will be quoted by the current FXS port.</td></tr>
<tr><td>Authentication Mode</td><td>Sets the way for SIP to make outgoing calls (Tel→IP) on the gateway.
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Do Not Register (default)</td><td>SIP initiates a call in a point-to-point mode.</td></tr>
<tr><td>Register Gateway</td><td>SIP initiates a call with the registered SIP account and password of the whole gateway. (Refer to 3.4.1 SIP 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>
</table>

<table>
<tr><td>Port Select Mode</td><td>When the port group receives a call, it will choose a port based on the select mode set by this configuration item to ring or to connect. The optional values and their corresponding meanings are described in the table below.
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Increase (default)</td><td>Search for an idle port in the ascending order of the port number, starting from the minimum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Decrease</td><td>Search for an idle port in the descending order of the port number, starting from the maximum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Cyclic Increase</td><td>Provided Port N is the available port found last time. Search for an idle port in the ascending order of the port number, starting from Port N+1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Cyclic Decrease</td><td>Provided Port N is the available port found last time. Search for an idle port in the descending order of the port number, starting from Port N-1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.</td></tr>
<tr><td>Group Ringing</td><td>Ring all the idle FXS ports in this port group.</td></tr>
<tr><td>Ringing by Turns</td><td>Ring the ports in this port group according to the Rule for Ringing by Turns which can be user-defined. Refer to the format of the rule in Figure 3-60. By default, the ringing will be carried out in the ascending order of the port number. Timeout for Ringing by Turns is used to set the overtime for ringing. Range of value: 15~60, calculated by s, with the default value of 20.</td></tr>
</table></td></tr>
<tr><td>Preemptive Answer Keyboard Shortcut</td><td>When a channel in a port group is ringing, another channel in the same port group can press the keyboard shortcut set by this item to transfer the call from the ringing channel to the current channel.
Note: This item will become invalid if the gateway works under the port select mode Group Ringing or Ringing by Turns.</td></tr>
<tr><td>Port Reused by Multiple Groups</td><td>Once this feature is enabled, a port can be added to different port groups.</td></tr>
<tr><td>Port</td><td>The ports in the port group. If the checkbox before a port is grey, it indicates that the port is not available or has been occupied. Once the feature “Port Reused by Multiple Groups” is enabled, a port which has been occupied is still available for other port groups. All selected ports for a port group will be displayed in the Ports column in Figure 3-59. Note: When a port group contains multiple ports, the automatic call forward feature is invalid.</td></tr>
</table>

After configuration, click ***Save*** to save the settings into the gateway, click ***Reset*** to restore the configurations, or 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. ***Check All FXO Ports*** means to select all available FXO ports on the current page; ***Check All FXS Ports*** means to select all available FXS ports on the current page.

Click ***Modify*** at the end of the list in **Port Group Settings Interface** to modify the properties of a port group. See Figure 3-61 for the port group modification interface. The configuration items on this interface are the same as those on the ***Add New Port Group*** interface.

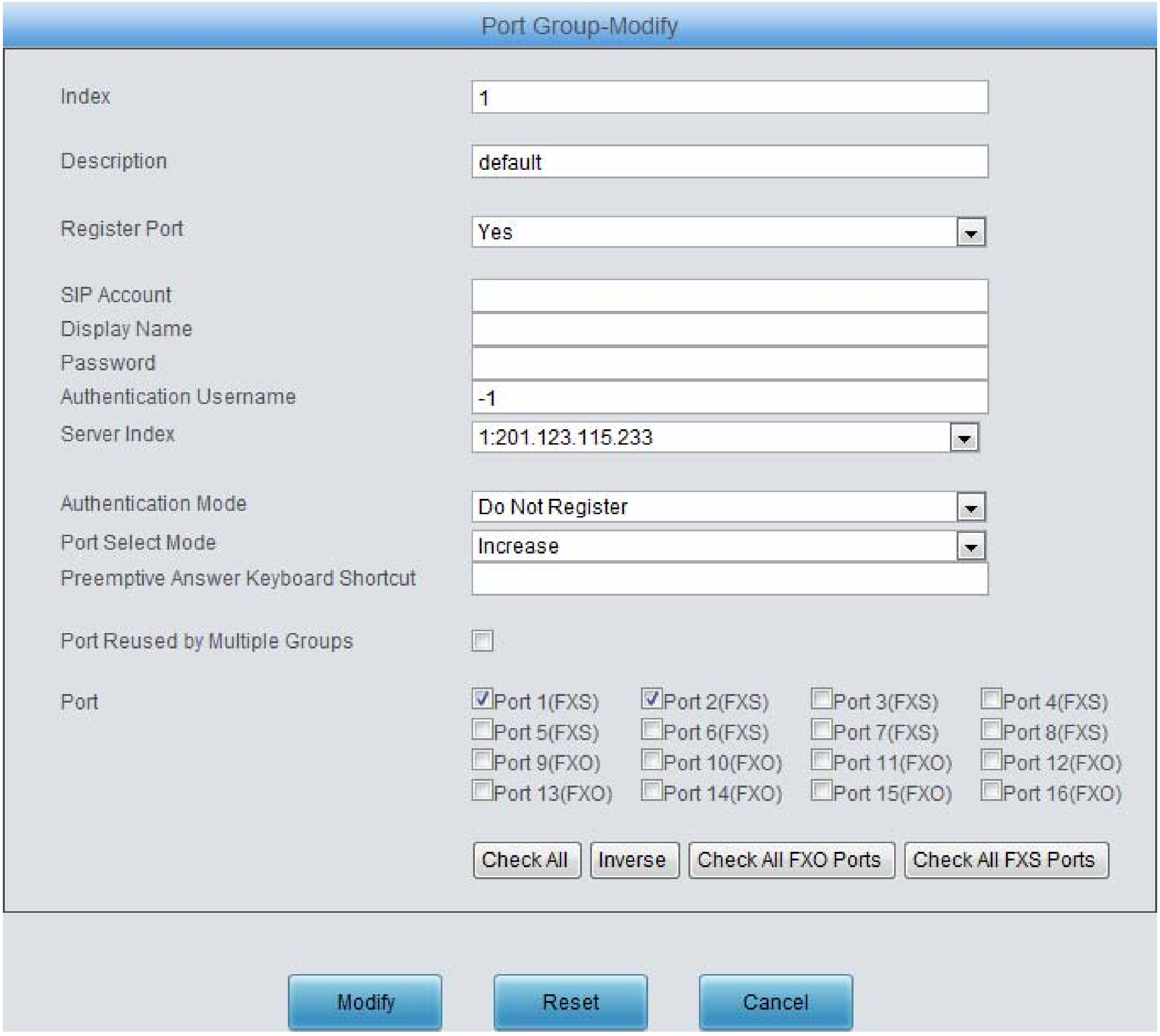


Figure 3-61 Modify Port Group

To delete a port group, check the checkbox before the corresponding index in Figure 3-59 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 3-59.

3.7 Route Settings

Route Settings is used to specify the routing rules for calls on two directions: IP→Tel and Tel→IP. See Figure 3-62.

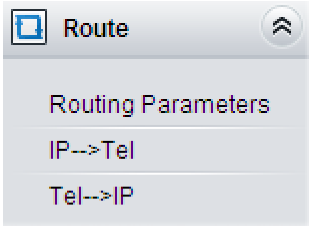


Figure 3-62 Route Settings

3.7.1 Routing Parameters

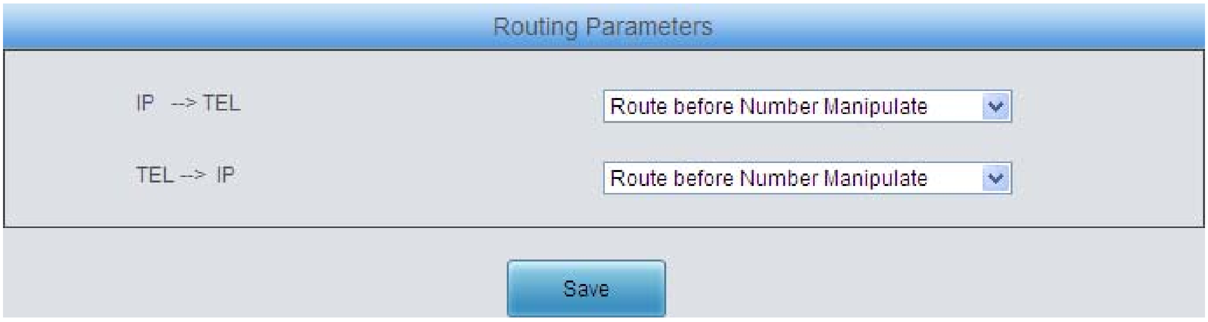


Figure 3-63 Routing Parameters Configuration Interface

See Figure 3-63 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.

3.7.2 IP to Tel

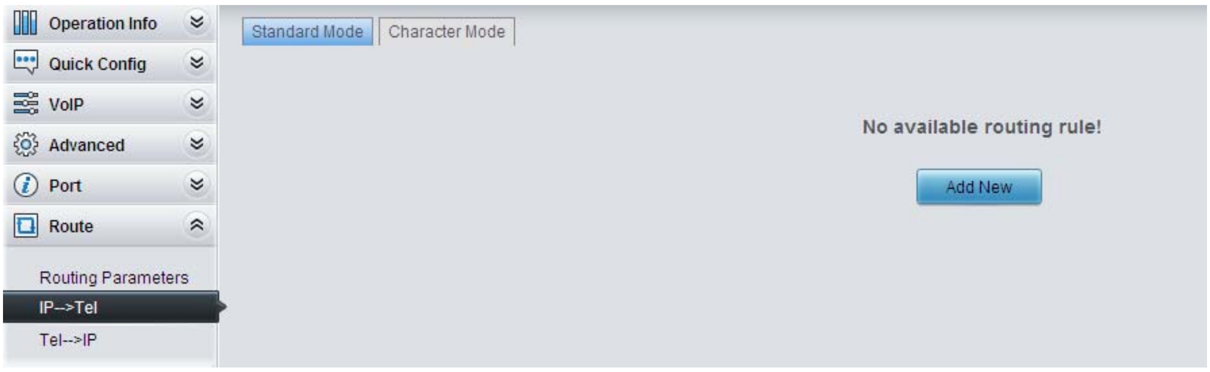


Figure 3-64 IP→Tel Routing Rule Configuration Interface (Standard)

See Figure 3-64 for the IP→Tel routing rule configuration interface. By default, there is no available routing rule on the gateway. The IP→Tel routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click ***Add New*** to add them manually. See Figure 3-65. You may use the default values of all the configuration items herein.

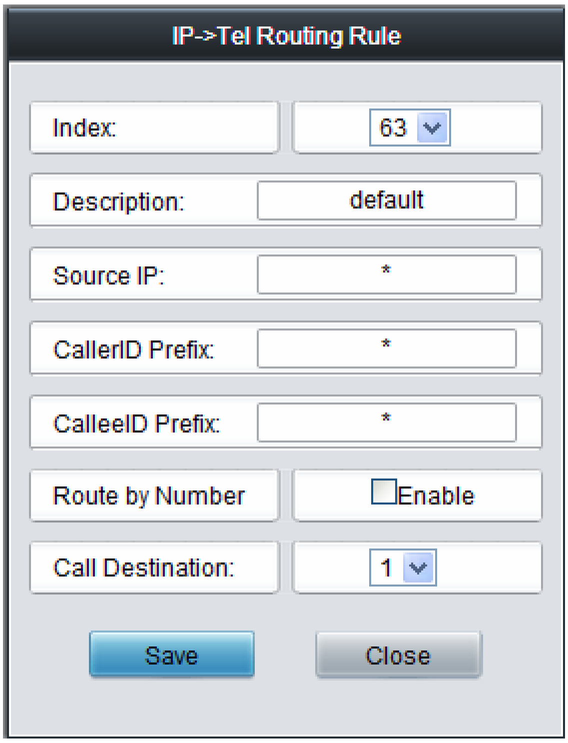


Figure 3-65 Add New Routing Rule (IP→Tel)

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, with the default value of *default*.
Source IP	IP address from where the call is initiated. This item can be set to a specific IP address or “*” which indicates any IP address
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, 、”[*]”, ”#” or character ranges defined by [ ]. ‘[ ]’ represents a character within the range it defines. Values in [ ] only can be characters ‘0~9’, ”[*]”, ”#”, punctuations ‘**-**‘ and ‘**,**’. (’**-**’ is used between two characters to indicates any character between these two characters. ’**,**’ is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to “*” which indicates any string. These two configuration items together with ***Source IP*** specify a routing rule for calls. **Note:** “[*]” represents TFM symbol *, while “*” represents any string.
Route by Number	When this feature is enabled, the gateway will route a call from IP to a corresponding port based on its number. And the number of the port which this call will be routed to can be set via the item ***SIP Account*** on the FXS or FXO Settings interface. In such case, the configuration item ***Call Destination*** goes invalid and shows *Route by Number* on the routing rule configuration interface. The default setting is *disabled*.

Call Destination	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.

See Figure 3-66 for the IP→Tel 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.

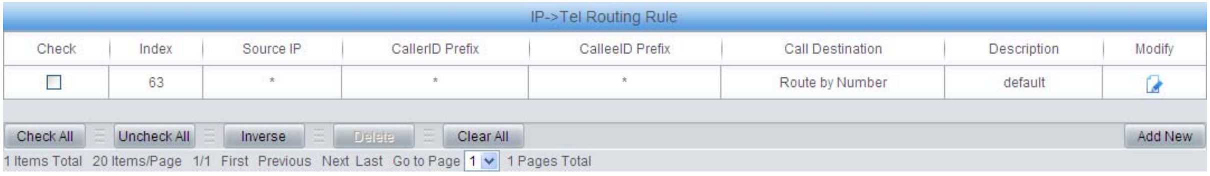


Figure 3-66 IP→Tel Routing Rule Configuration Interface

Click ***Modify*** in Figure 3-66 to modify a routing rule. The configuration items on the IP→Tel routing rule modification interface are the same as those on the ***Add New Routing Rule (IP→Tel)*** interface. Note that the item ***Index*** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-66 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 3-66.

See Figure 3-67 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.

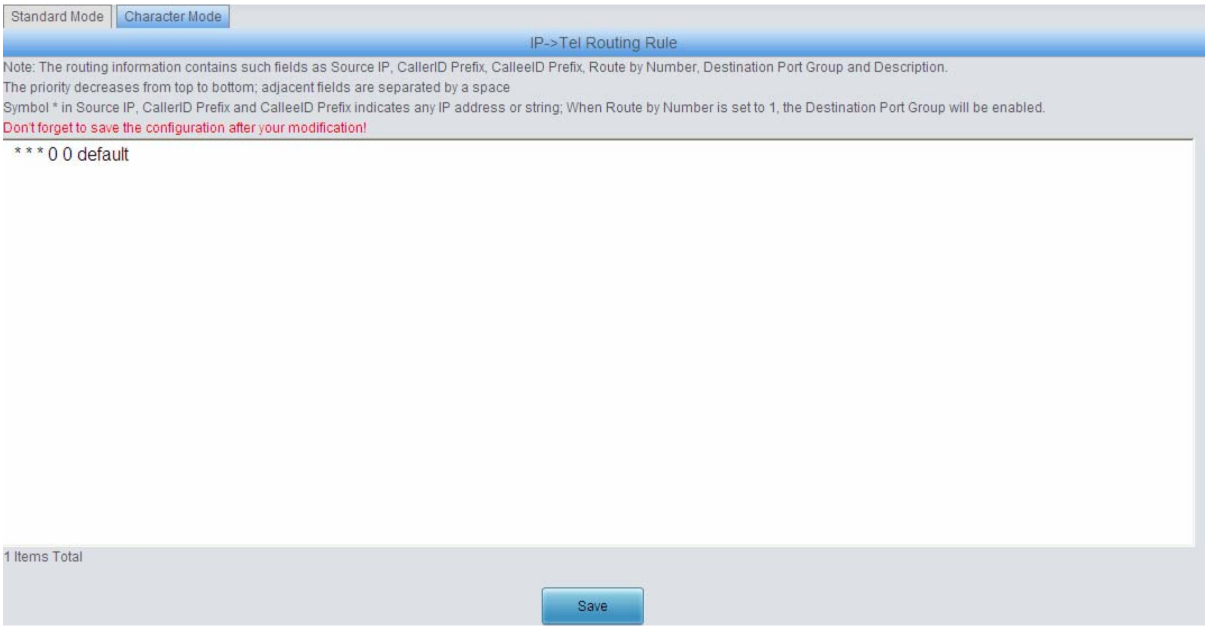


Figure 3-67 IP→Tel Routing Rule Configuration Interface (Character)

3.7.3 Tel to IP

Figure 3-68 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-68 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 3-69. You may use the default values of all the configuration items herein except for ***Destination IP*** and ***Destination Port***.

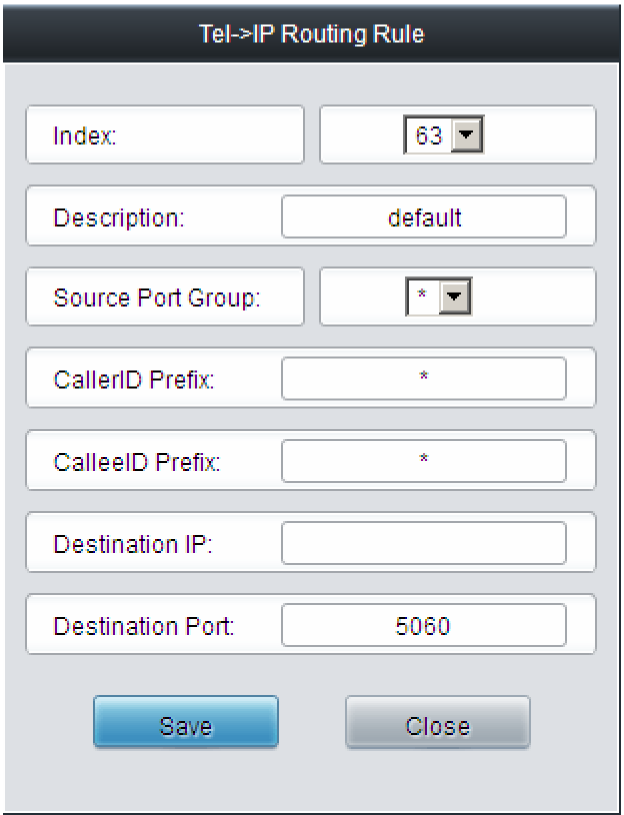


Figure 3-69 Add New Routing Rule (Tel→IP)

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, with the default value of *default.*

Source Port Group (Call Initiator)	Port group from which the call is initiated. This item can be set to a specific port group or '*' which indicates any port group.
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or characters ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '**-**' and '**,**'. ('**-**' is used between two characters to indicates any characters between these two characters. '**,**' is used to separate characters or characters ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Source Port Group (Call Initiator)*** specify a routing rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string.
Destination IP, Destination Port	IP address and port number of the remote end to which the call will be routed.

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

See Figure 3-70 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.

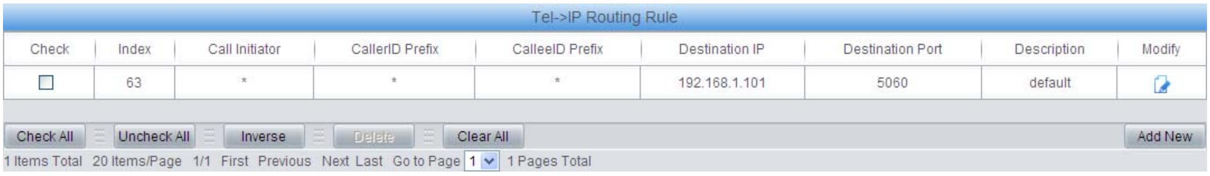


Figure 3-70 Tel→IP Routing Rule Configuration Interface

Click ***Modify*** in Figure 3-70 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 3-70 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 3-70.

See Figure 3-71 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.

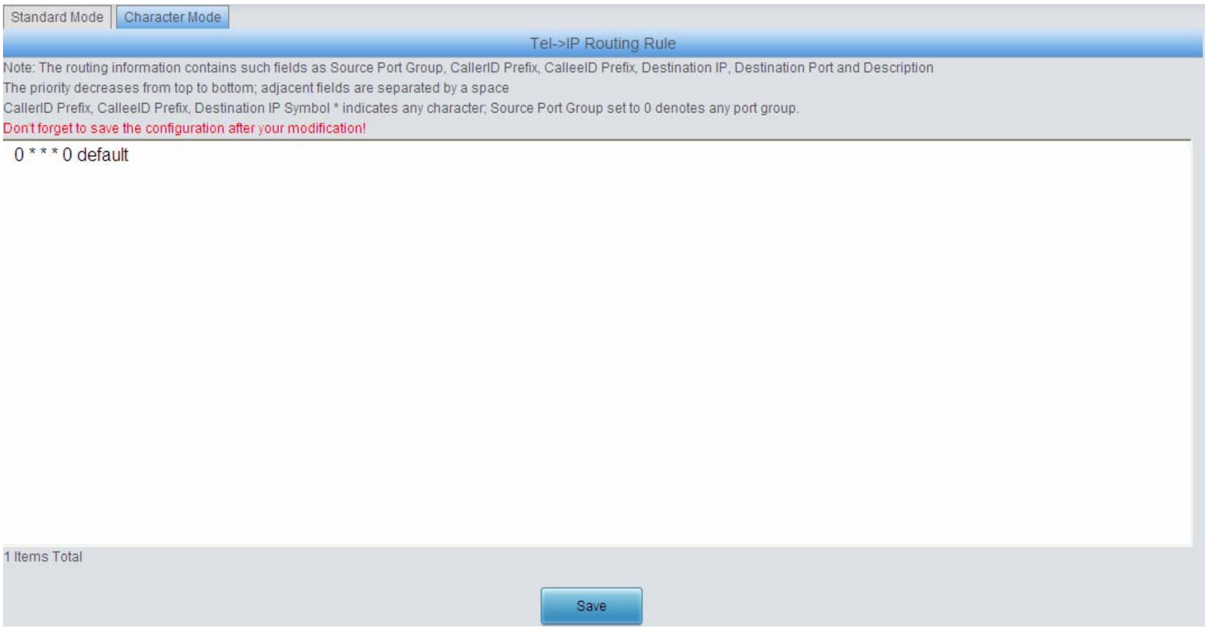


Figure 3-71 Tel→IP Routing Rule Configuration Interface (Character)

3.8 Number Manipulation

Number Manipulation includes four parts: **IP→Tel CallerID**, **IP→Tel CalleeID**, **Tel→IP CallerID** and **Tel→IP CalleeID**. See Figure 3-72.

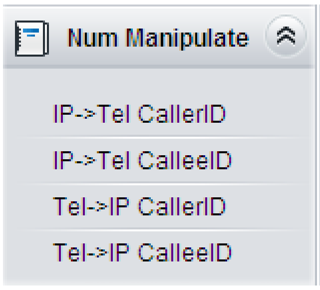


Figure 3-72 Number Manipulation

3.8.1 IP to Tel CallerID

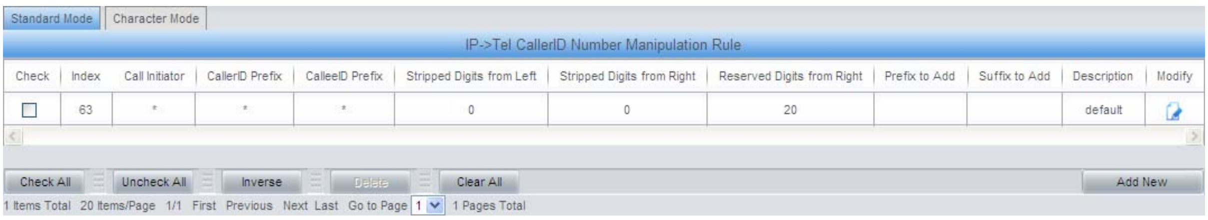


Figure 3-73 IP→Tel CallerID Manipulation Interface (Standard)

See Figure 3-73 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 3-74 for the IP→Tel CallerID manipulation rule adding interface. You may use the default values of all the configuration items herein.

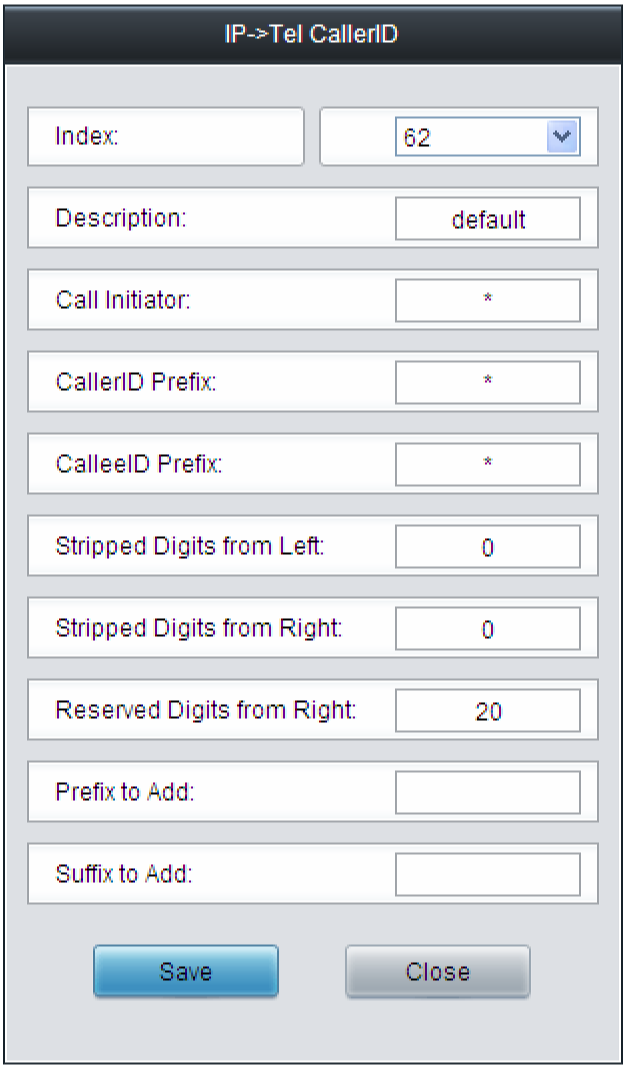


Figure 3-74 Add IP→Tel CallerID Manipulation Rule

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, with the default value of *default.*
Call Initiator	IP address from where the call is initiated. This item can be set to a specific IP address or “*” which indicates any IP address.

CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to indicates any character between these two characters. ',' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Call Initiator*** specify a number manipulation rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string.
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. The default value is 0.
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. The default value is 0.
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. The default value is 20.
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 3-73 to modify a number manipulation rule. See Figure 3-75 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.

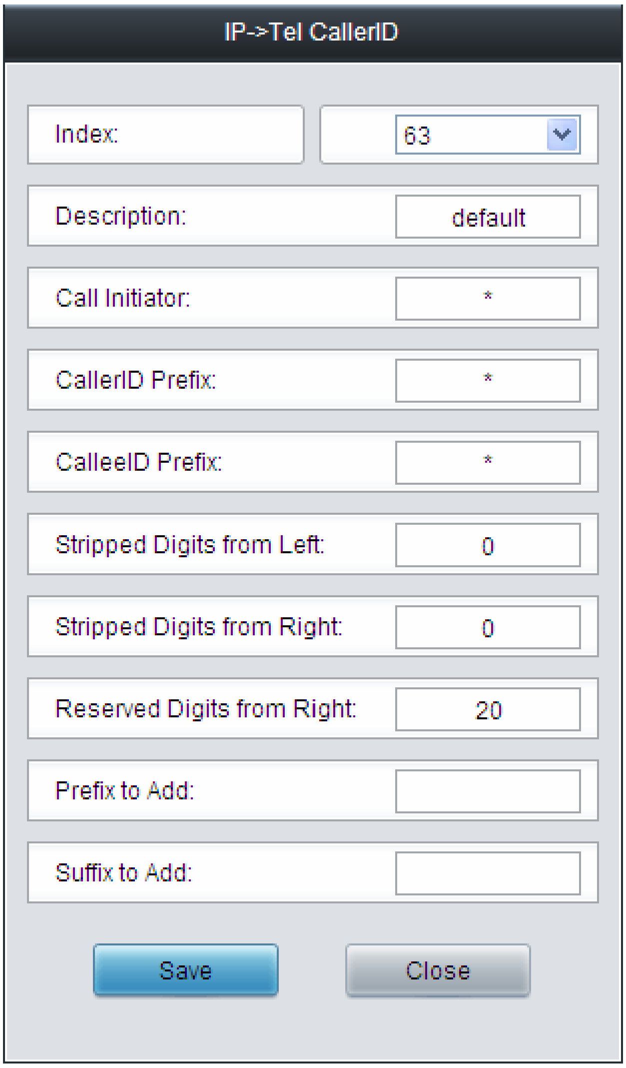


Figure 3-75 Modify IP→Tel CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-73 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 3-73.

See Figure 3-76 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.

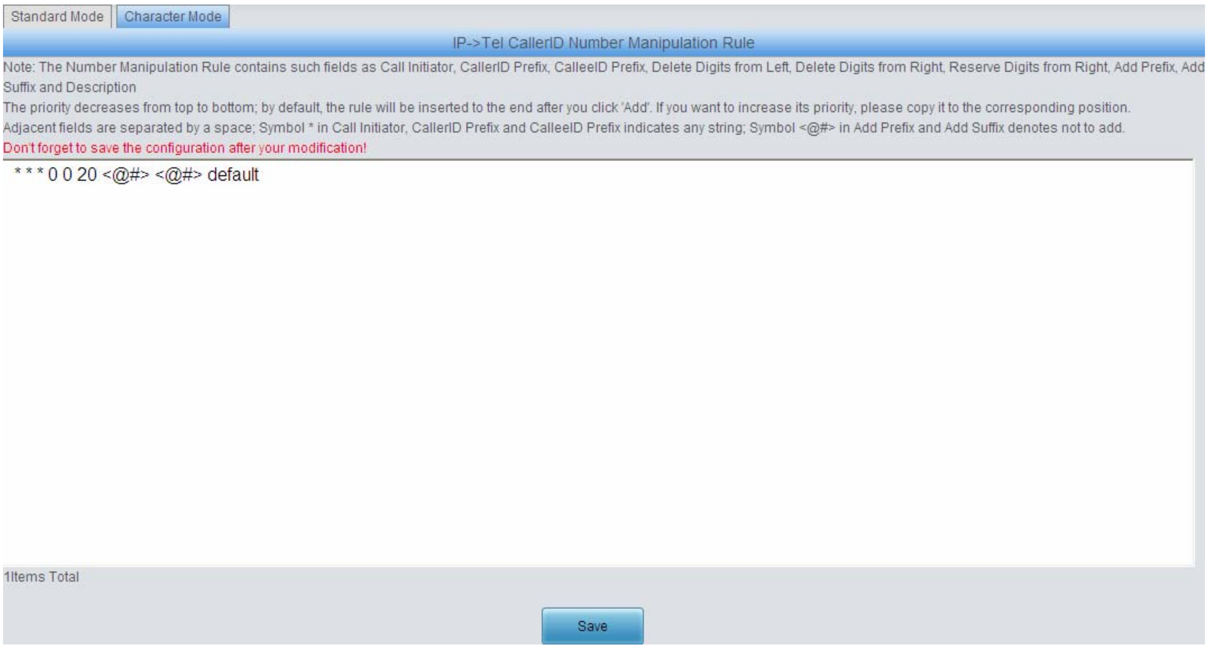


Figure 3-76 IP→Tel CallerID Manipulation Interface (Character)

3.8.2 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 3-78 for IP→Tel CalleeID manipulation interface. The configuration items on this interface are the same as those on ***IP→Tel CallerID Manipulation Interface*** (Figure 3-73).

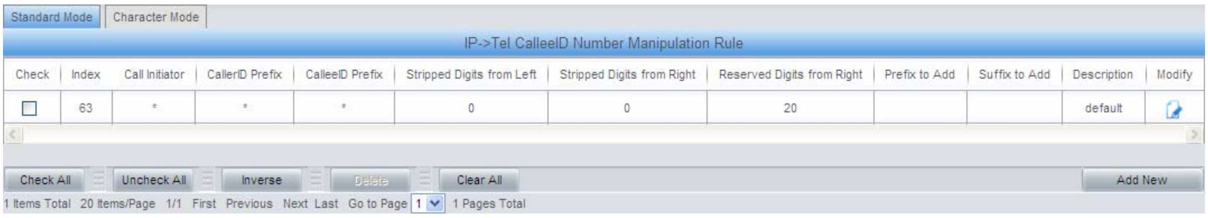


Figure 3-77 IP→Tel CalleeID Manipulation Interface(Standard)

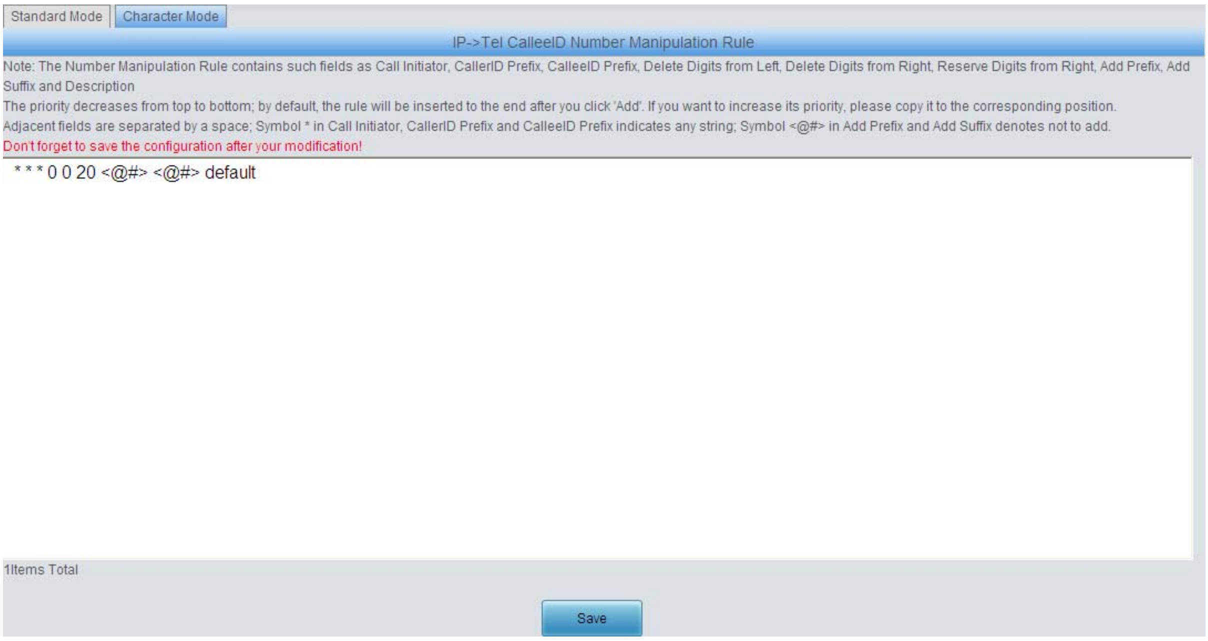


Figure 3-78 IP→Tel CalleeID Manipulation Interface (Character)

3.8.3 Tel to IP CallerID

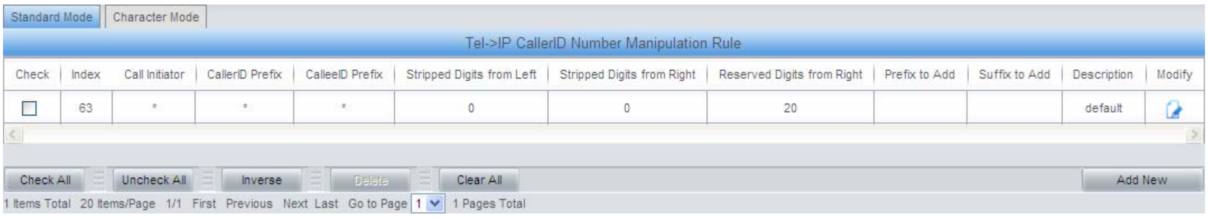


Figure 3-79 Tel→IP CallerID Manipulation Interface (Standard)

See Figure 3-79 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 3-80 for the Tel→IP CallerID manipulation rule adding interface. You may use the default values of all the other configuration items herein.

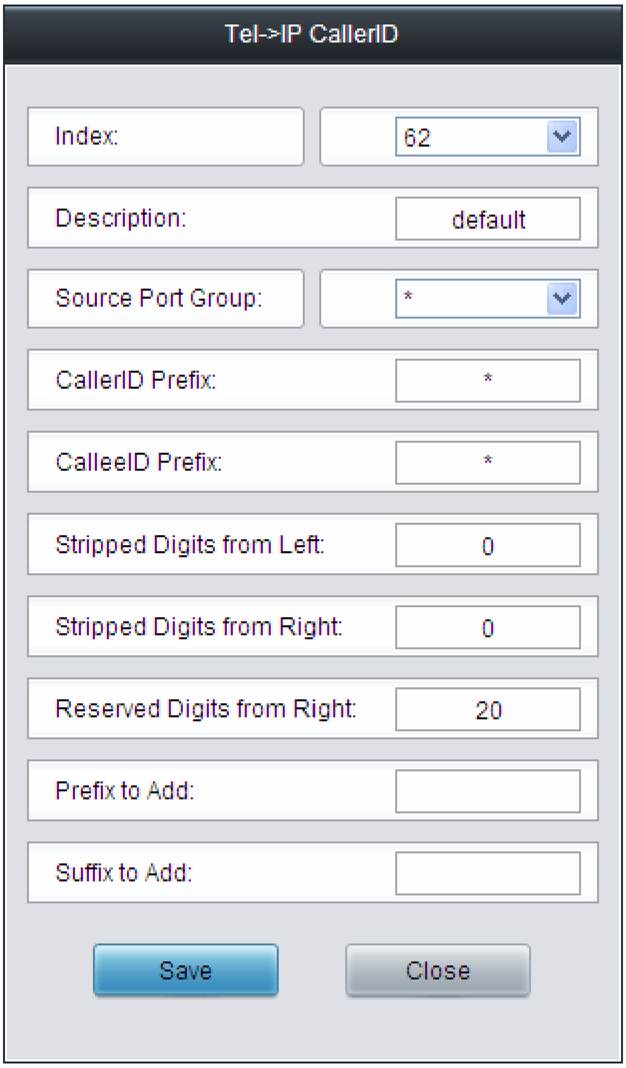


Figure 3-80 Add Tel→IP CallerID Manipulation Rule

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

Item	Description
Index	The unique index of each number manipulation rule, which denotes its priority. A number manipulation rule with a smaller index value has a higher priority. If a call

	matches several number manipulation rules, it will be processed according to the one with the highest priority.
Description	More information about each number manipulation rule, with the default value of *default*.
Source Port Group (Call Initiator)	Port group from which the call is initiated. This item can be set to a specific port group or '*' which indicates any port group.
CallerID Prefix, CalleeID Prefix	A string of characters at the beginning of the caller/called party number. It can be a specific string consisting of digits 0~9, "[*]", "#" or character ranges defined by [ ]. '[ ]' represents a character within the range it defines. Values in [ ] only can be digits '0~9', "[*]", "#", punctuations '**-**' and '**,**'. ('**-**' is used between two characters to indicates any character between these two characters. '**,**' is used to separate characters or character ranges, representing alternatives.) For example, 057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which indicates any string. These two configuration items together with ***Call Initiator*** specify a number manipulation rule for calls. **Note:** "[*]" represents DTFM symbol *, while "*" represents any string.
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. The default value is 0.
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. The default value is 0.
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. The default value is 20.
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 3-79 to modify a number manipulation rule. See Figure 3-81 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.

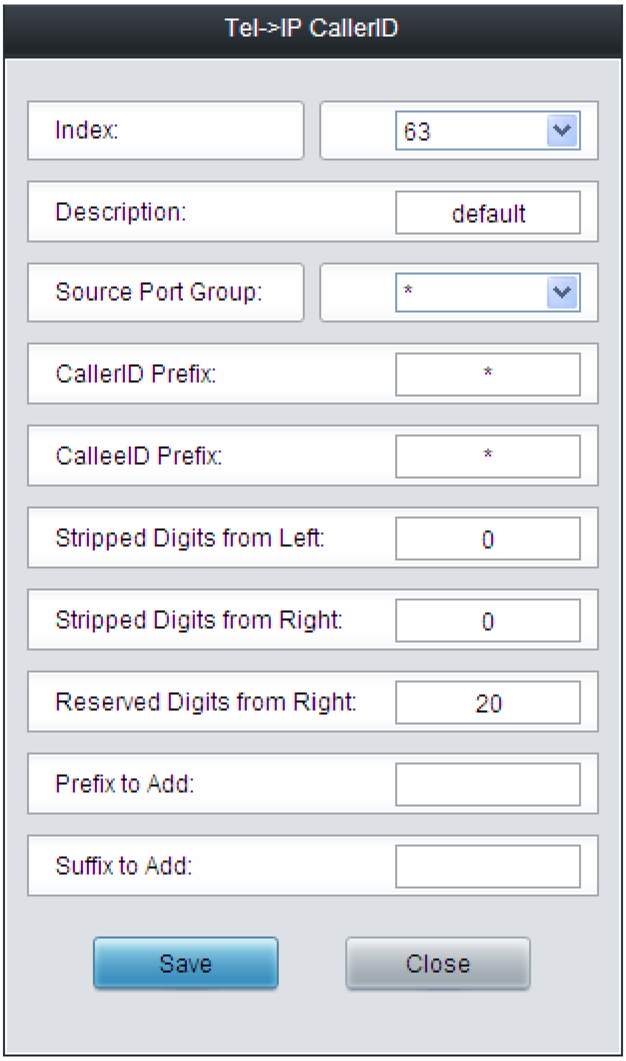


Figure 3-81 Modify Tel→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-79 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 3-79.

See Figure 3-82 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.

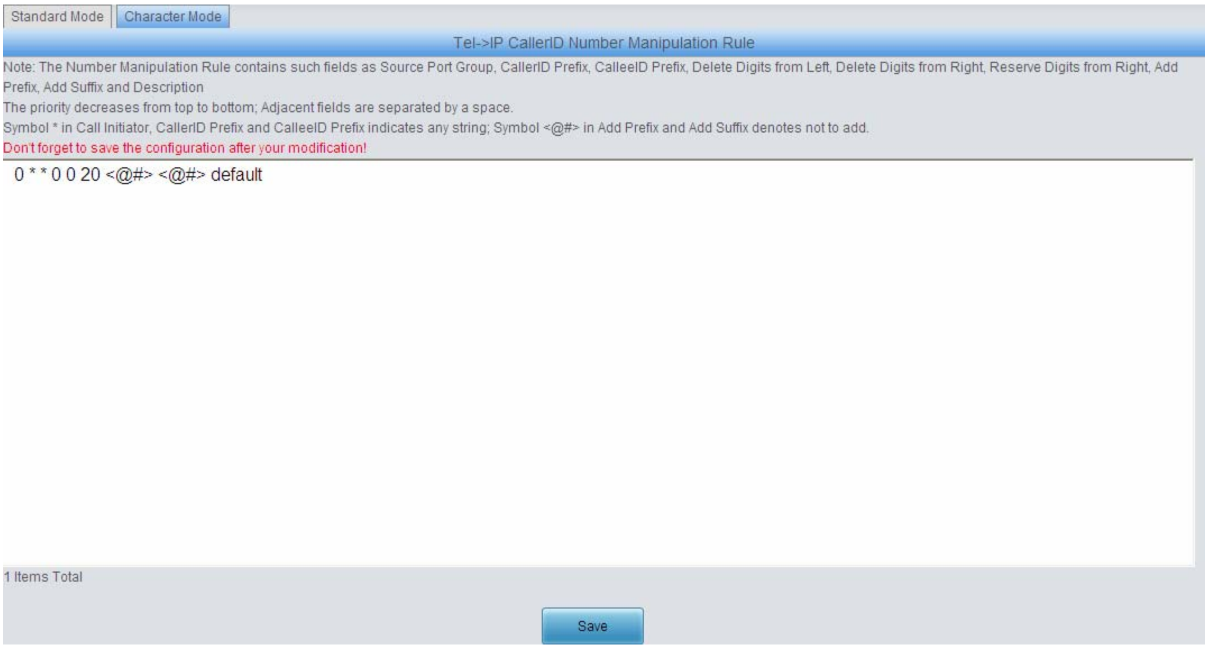


Figure 3-82 Tel→IP CallerID Manipulation Interface (Character)

3.8.4 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 3-83, Figure 3-84 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 3-79).

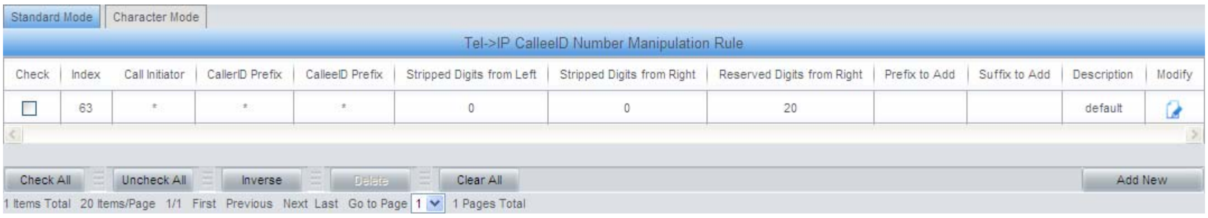


Figure 3-83 Tel→IP CalleeID Manipulation Interface (Standard)

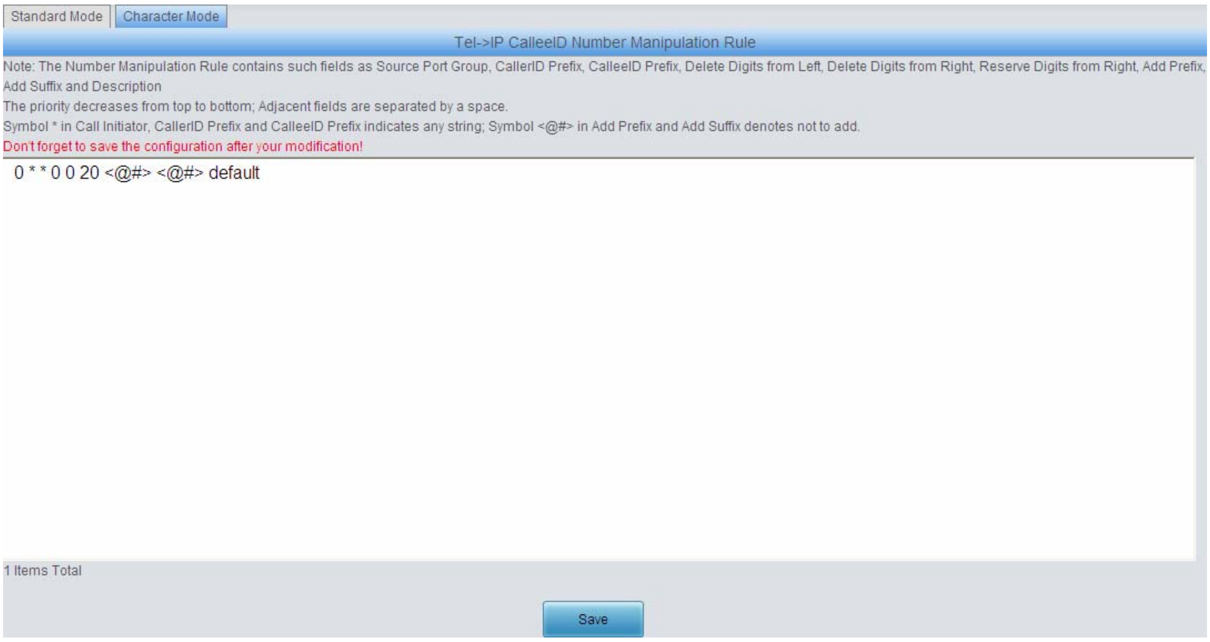


Figure 3-84 Tel→IP CalleeID Manipulation Interface (Character)

3.9 System Tools

System Tools is mainly for gateway maintenance. It provides such features as IP modification, data backup and connectivity check. See Figure 3-85 for details.

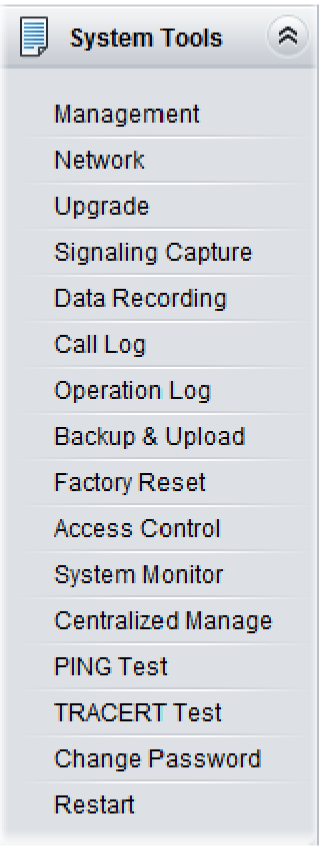


Figure 3-85 System Tools

3.9.1 Management

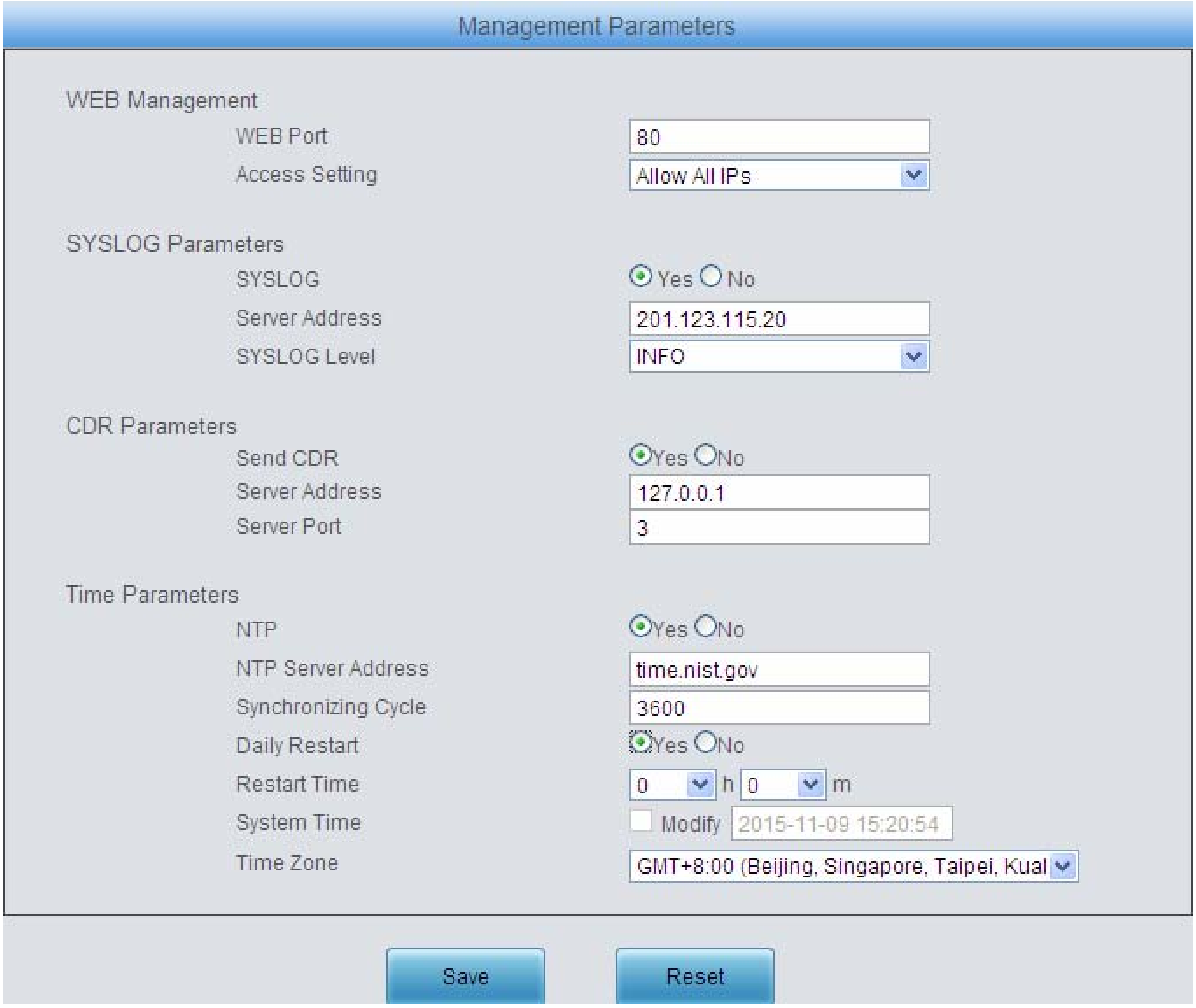


Figure 3-86 Management Parameters Setting Interface

See Figure 3-86 for the Management Parameters Setting interface. The table below explains the items shown in the above figure.

Item	Description
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.
Access Setting	Sets the IP addresses which can access the gateway via WEB. By default, all IPs are allowed. You can set an IP whitelist to allow all IPs within it to access the gateway freely. Also can set an IP blacklist to forbid all IPs within it to access the gateway.
SYSLOG	Sets whether to enable SYSLOG. It is required to fill in ***SYSLOG Server Address*** and ***SYSLOG Level*** in case SYSLOG is enabled. By default, ***SYSLOG*** is disabled.
Server Address	Sets the SYSLOG server address for log reception.
SYSLOG Level	Sets the SYSLOG level. There are three options: *ERROR*, *WARNING*, *INFO* and *DEBUG*. The default value is *INFO*.
Send CDR	Sets whether to enable the feature of sending CDR. It is required to fill in ***Server Address*** and ***Server Port*** in case Send CDR is enabled. By default, ***Send CDR*** is disabled.

Server Address	The address of the server to receive CDR.
Server Port	The port of the server to receive CDR.
NTP	Sets whether to enable the NTP time synchronization feature. It is required to fill in ***NTP Server Address***, ***Synchronizing Cycle*** and ***Time Zone*** in case NTP is enabled. By default, ***NTP*** is enabled.
NTP Server Address	Sets the Server address for NTP time synchronization. By default, the address is time.nist.gov
Synchronizing Cycle	Sets the cycle for NTP time synchronization, calculated by s, with the default value of 3600.
Daily Restart	Sets whether to restart the gateway regularly every day at the preset ***Restart Time***. By default, this feature is disabled.
Restart Time	Sets the time to restart the gateway regularly.
System Time	The system time. Check the checkbox before ***Modify*** and change the time in the edit box when NTP is disabled.
Time Zone	The time zone of the gateway.

3.9.2 Network

Figure 3-87 Network Settings Interface

See Figure 3-87 for the network settings interface. A gateway has only one LAN, which can be configured with network type, IP address, subnet mask, default gateway and DNS server. Network Type has three options: Static, DHCP and PPPoE. If PPPoE is used, it is necessary to enter the username and the password of the network.

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.

3.9.3 Upgrade

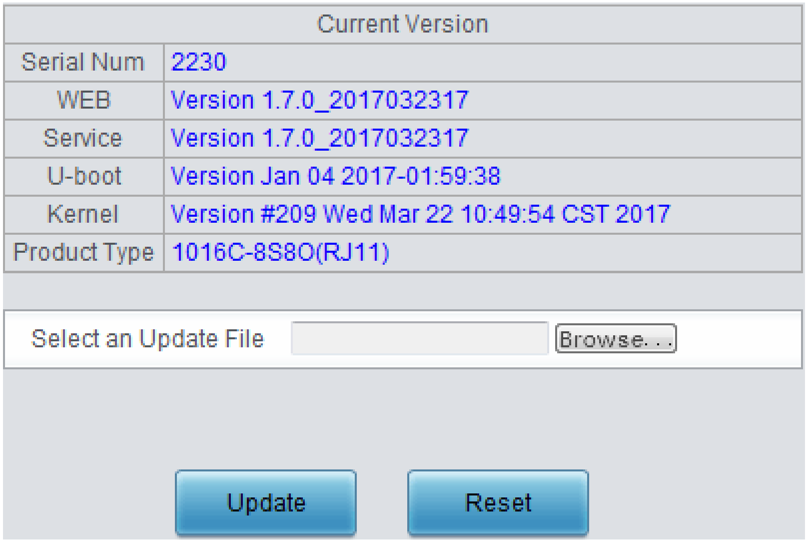


Figure 3-88 Upgrade Interface

See Figure 3-88 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 3-89.

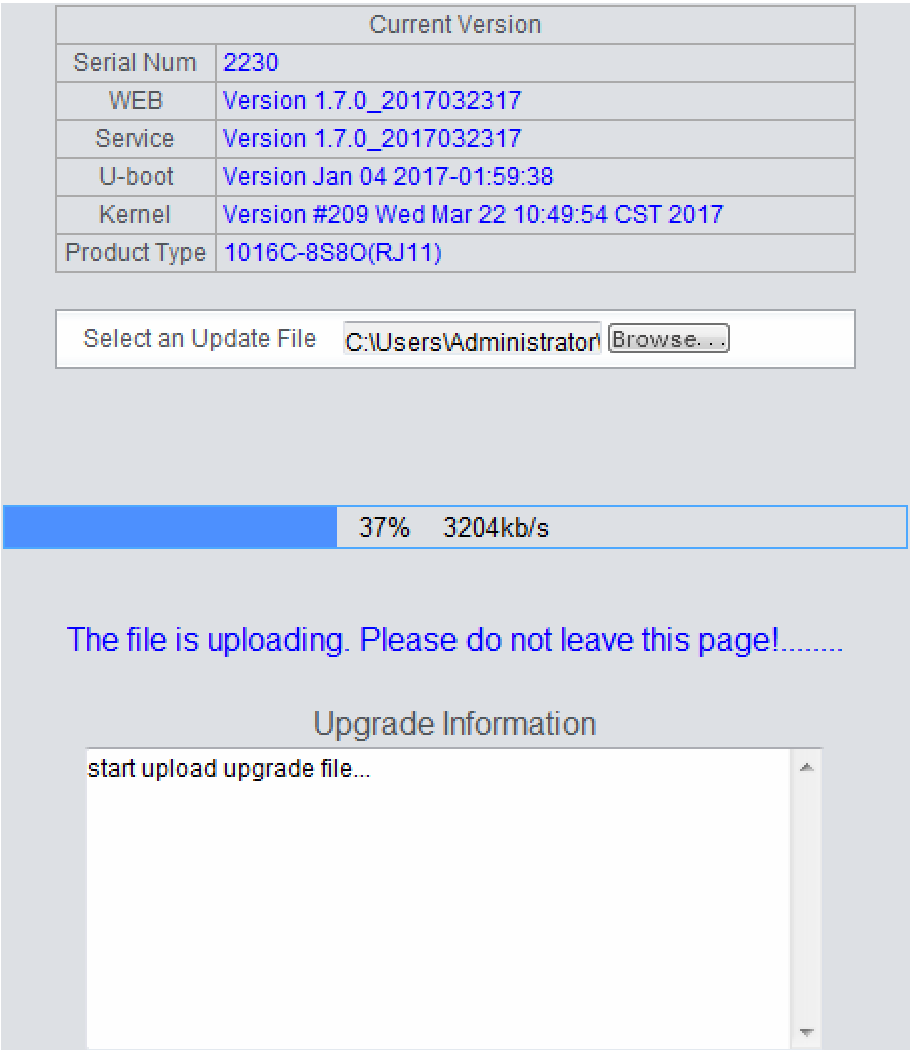


Figure 3-89 File Uploading Interface

After a successful uploading of the file, the gateway will start to upgrade the system. See Figure 3-90 and you can learn the detailed upgrading information from the upgrade information box at the bottom.

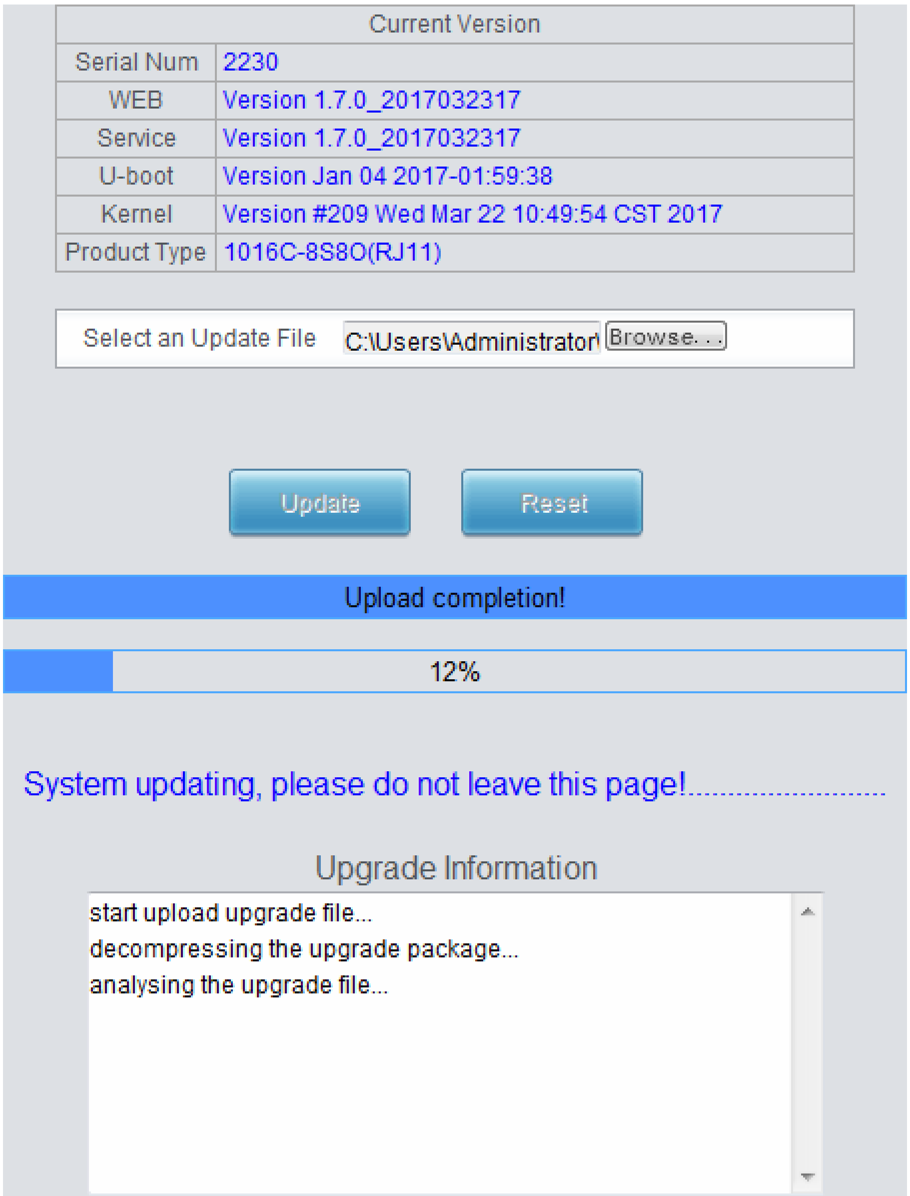


Figure 3-90 System Upgrading Interface

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.

3.9.4 Signaling Capture

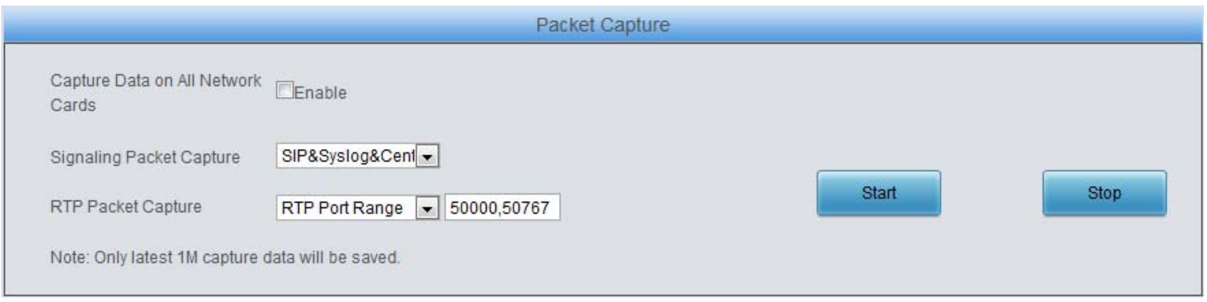


Figure 3-91 Signaling Capture Interface

See Figure 3-91 for the Signaling Capture interface. Packet capture contains Signaling Packet Capture and RTP Packet Capture. You can select either of them to start the capture according to your requirement. Once the configuration item “Capture Data on All Network Cards” is enabled, the gateway will capture the data on all kinds of network cards, including eth0, lo (local loopback) and veth0 (virtual network card); otherwise, it will only capture the data on eth0. Click ***Start*** to start capturing packets. Click ***Stop*** to stop the capture and download the captured packets.

3.9.5 Data Recording

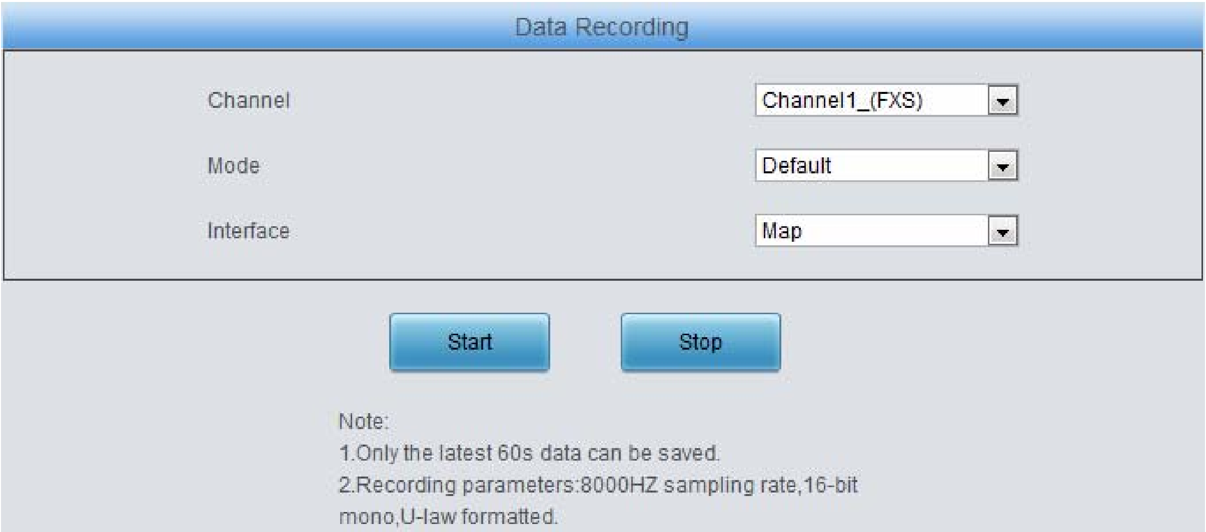


Figure 3-92 Data Recording Interface

See Figure 3-92 for the Debug & Record interface. You can select a channel and the recording mode to start the data recording. Click ***Start*** to start the corresponding recording. Click ***Stop*** to stop the recording and download the recorded file.

3.9.6 Call Log

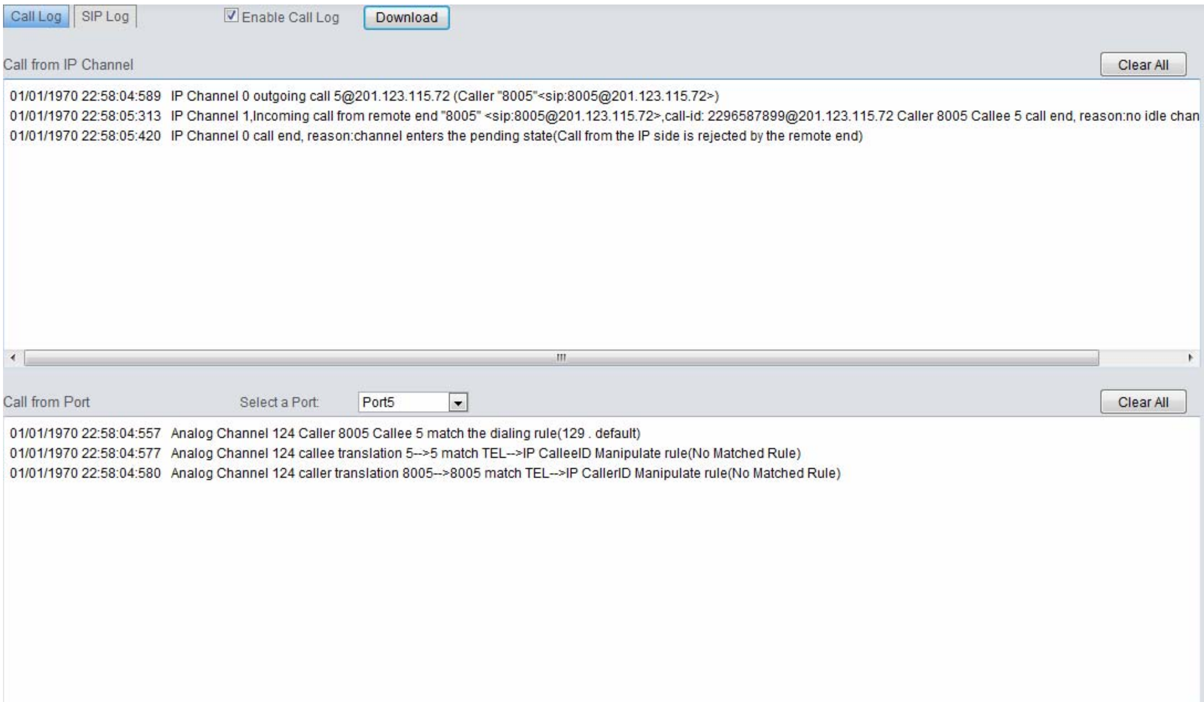


Figure 3-93 Call Log Interface

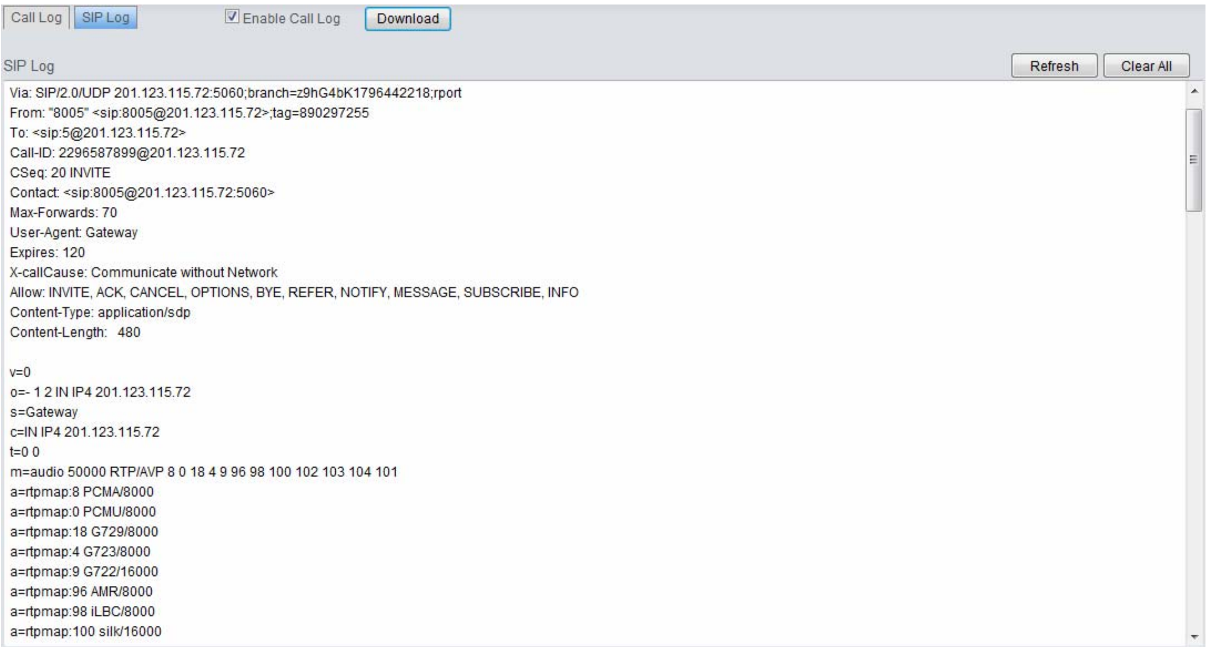


Figure 3-94 SIP Log Interface

See Figure 3-93, Figure 3-94 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***.

3.9.7 Operation Log

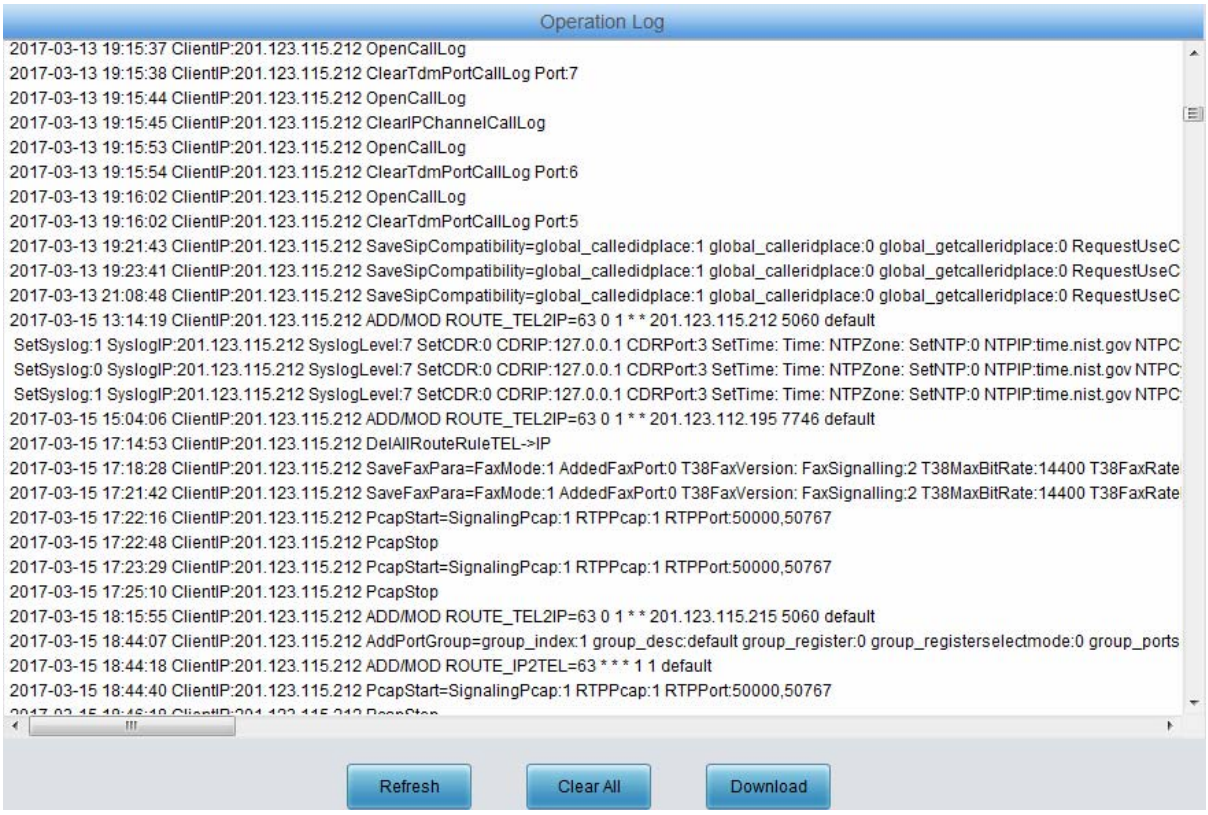


Figure 3-95 Operation Log Interface

See Figure 3-95 for the Operation Log interface, which is used to check the operation records on WEB. Click ***Refresh*** to refresh the log; click ***Clear All*** to clear all the operation logs and click ***Download*** to download the logs.

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

3.9.8 Backup & Upload

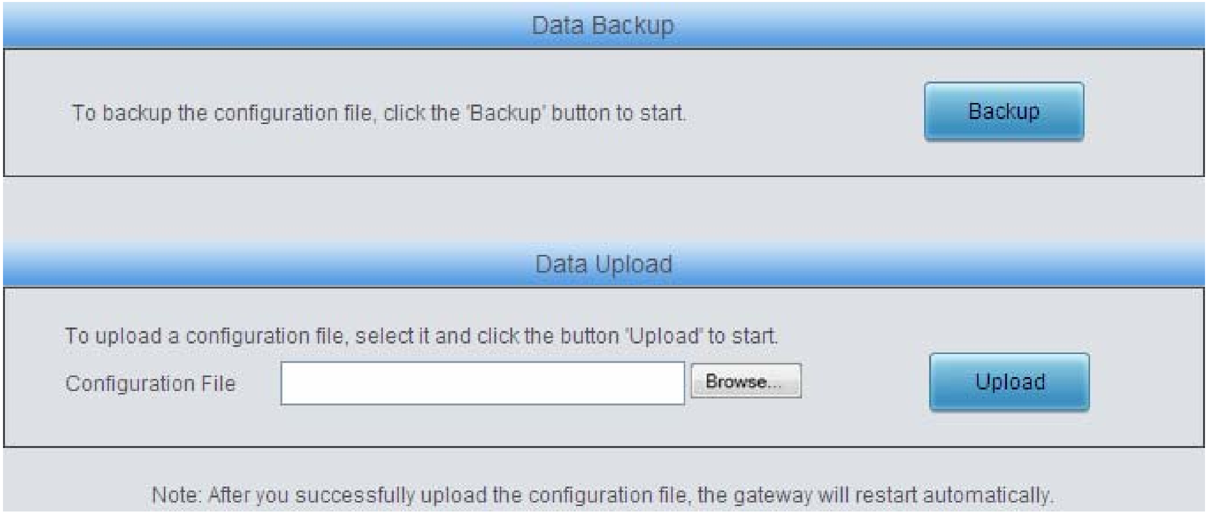


Figure 3-96 Backup & Upload Interface

See Figure 3-96 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***.

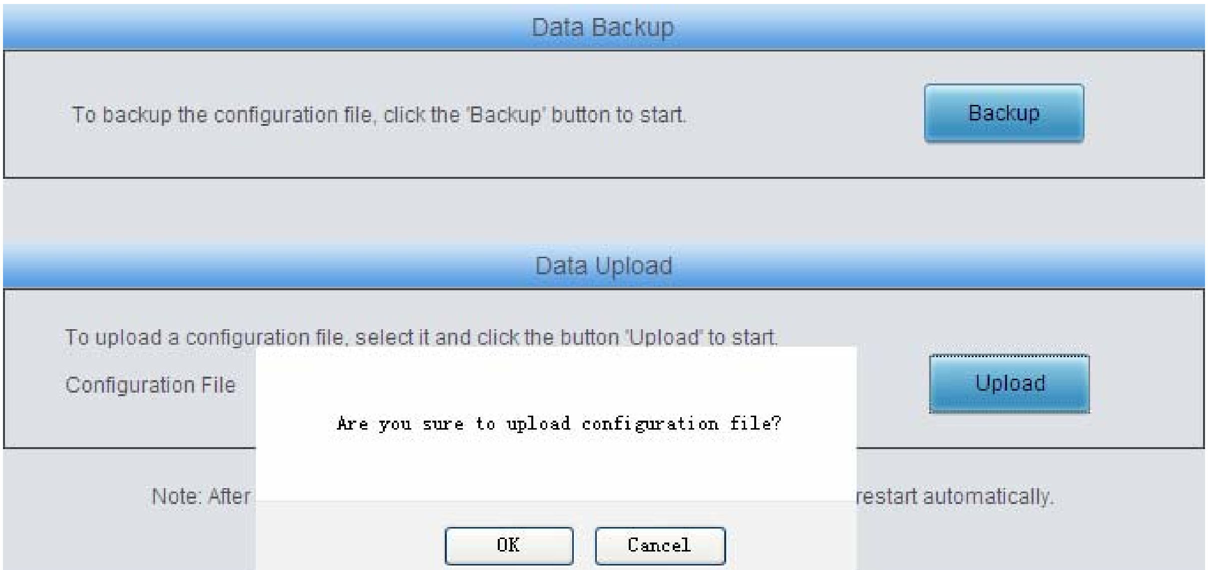


Figure 3-97 Backup & Upload & Prompt Interface

Click ***OK*** on the prompt box (Figure 3-97) to upload the configuration file to the gateway. Now the prompt information 'System is rebooting, please do not leave this page' appears. See Figure 3-98. The gateway will overwrite the current configurations with the uploaded data after restart. Click ***Cancel*** to cancel this upload directly.

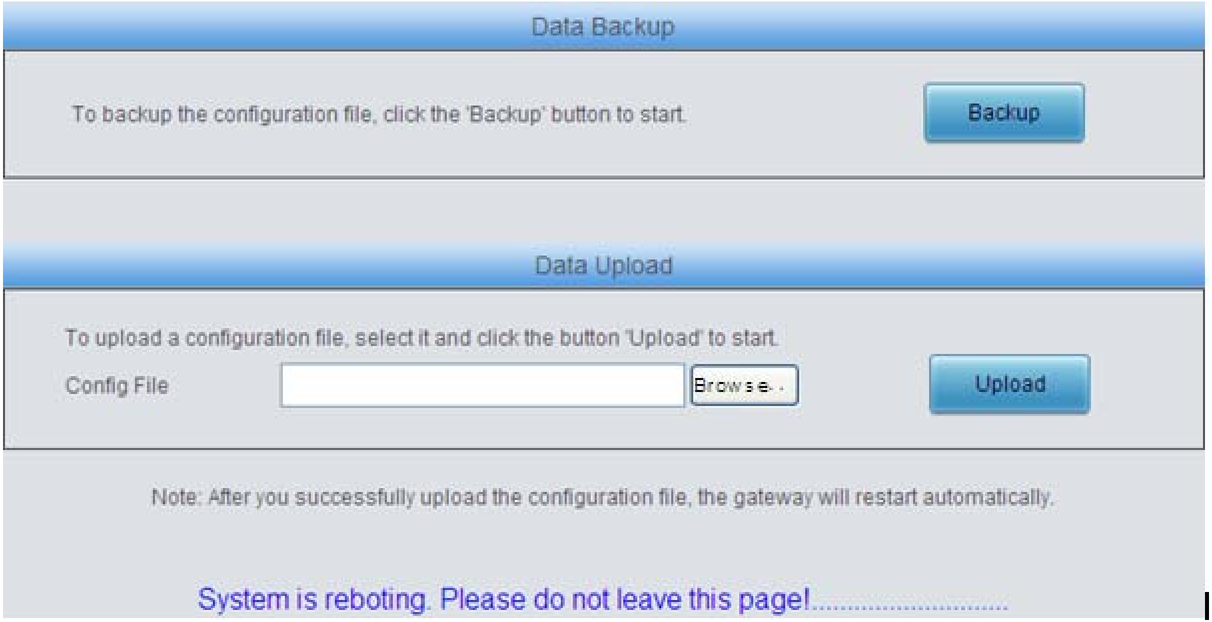


Figure 3-98 Configuration File Uploading Interface

3.9.9 Factory Reset

Figure 3-99 Factory Reset Interface

See Figure 3-99 for the factory reset interface. Click ***Reset*** to restore all configurations on the gateway to factory settings.

3.9.10 System Monitor

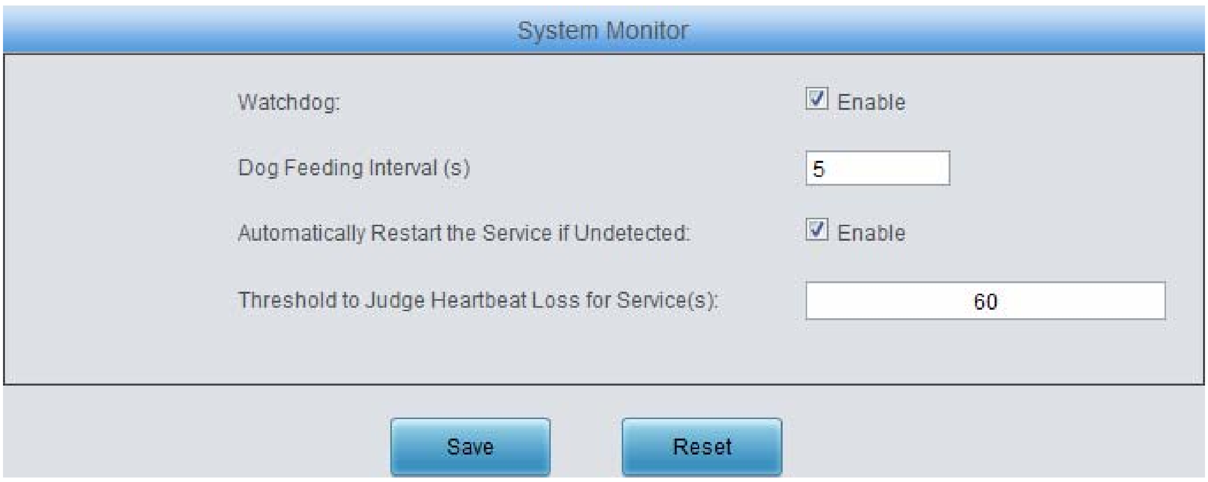


Figure 3-100 System Monitor Configuration Interface

See Figure 3-100 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. Threshold to Judge Heartbeat Loss for Service is used to judge whether the gateway receives the heartbeat packets from the service during the set time, if not, it is considered that the gateway service has been disconnected. It is calculated by s, with the value range of 20~120s and the default value of 60s.

3.9.11 Centralized Manage

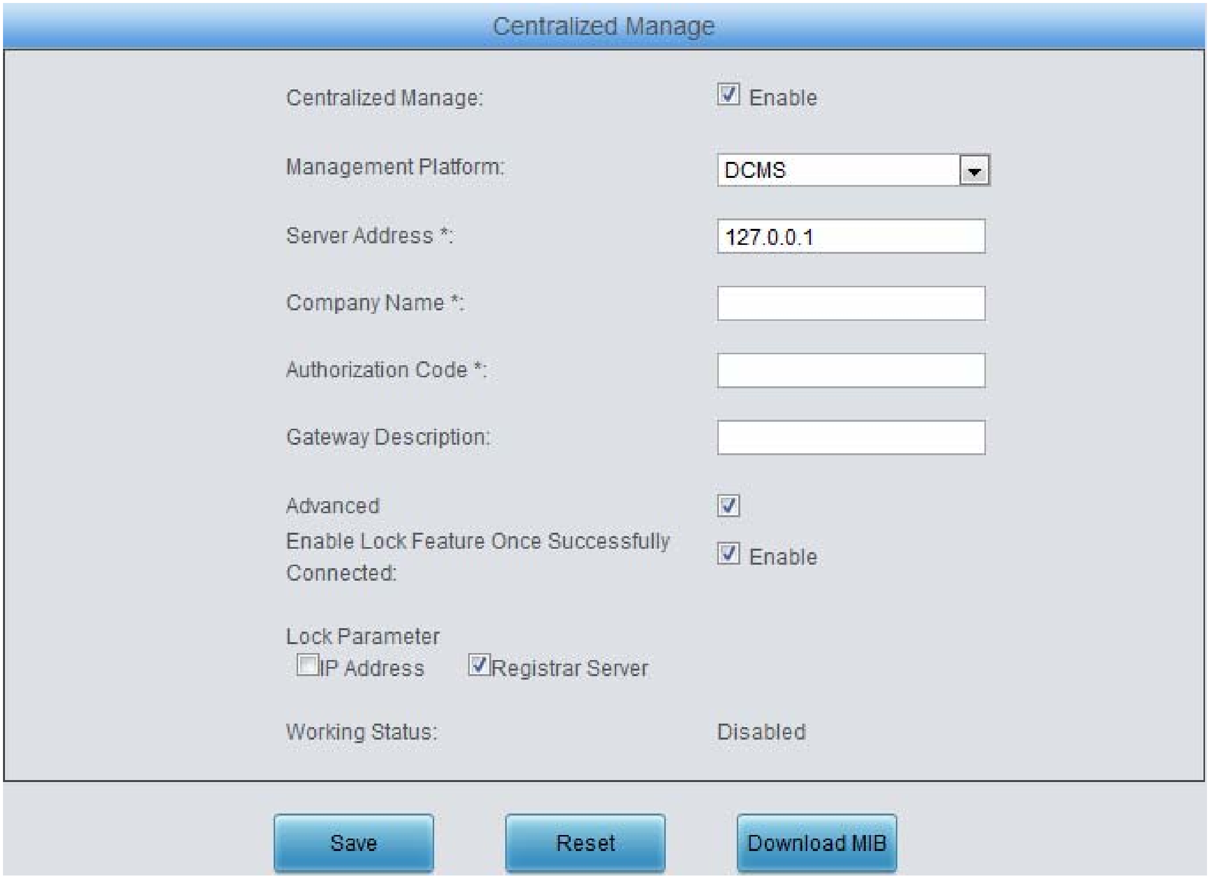


Figure 3-101 Centralized Manage Setting Interface

See Figure 3-101 for the Centralized Manage Setting interface. The gateway can register to a centralized management platform and accept the management of the platform. The table below explains the items shown in above figures.

Item	Description
Management Platform	Select a management platform for the gateway to register, including two options: DCMS and Others.
Server Address	The address of the server in which the management platform locates, It can be IP or a domain name, valid only when DCMS is selected. Note: To configure the domain name, the DNS should be already configured and the corresponding domain name must be analyzable.
Company Name	The name used to register the gateway to Synway DCMS, valid only when DCMS is selected.

Authorization Code	The authorization code is used for the connection verification. A device can connect to the DCMS successfully only after it passes the verification. Only valid when DCMS is selected.
Gateway Description	The description displayed on Synway DCMS after the gateway is registered to Synway DCMS, giving an easy identification of the gateway in device grouping. This item is valid only when DCMS is selected.
Enable Lock Feature Once Successfully Contected	Once this feature is enabled, you can lock the device according to the corresponding parameters. This item is valid only when DCMS is selected.
IP Address	Once this feature is enabled, you are required to fill in the authorization code while modifying the information related to the IP address in the Network interface. This item is valid only when DCMS is selected.
Registrar Server	Once this feature is enabled, you are required to fill in the authorization code while modifying the address and port of the registrar server in the SIP Settings interface. This item is valid only when DCMS is selected.
Working Status	The status of the connection between the gateway and the centralized management server. This item is valid only when DCMS is selected.
Centralized Management Protocol	Set the centralized management protocol. It only supports SNMP currently.
SNMP Version	Set the version of SNMP, three options available: V1, V2 and V3, with the default value of V2. This item is valid only when Others is selected.
Monitoring Port	Monitoring Port for SNMP on the gateway. This item is valid only when Others is selected.
Community String	Community string used for information acquisition.
Account	The account of SNMP, valid only when the SNMP version is set to V3.
Grade	The grade of SNMP, three options available: Neither authenticated nor encrypted, Authenticated but not encrypted and Authenticated and encrypted, with the default value of *Neither authenticated nor encrypted*. It is valid only when the SNMP version is set to V3.
Authentication Password	The authentication password required to enter when the item Grade is set to Authenticated but not encrypted or Authenticated and encrypted.
Encryption Password	The encryption password required to enter when the item Grade is set to Authenticated and encrypted.

3.9.12 Access Control

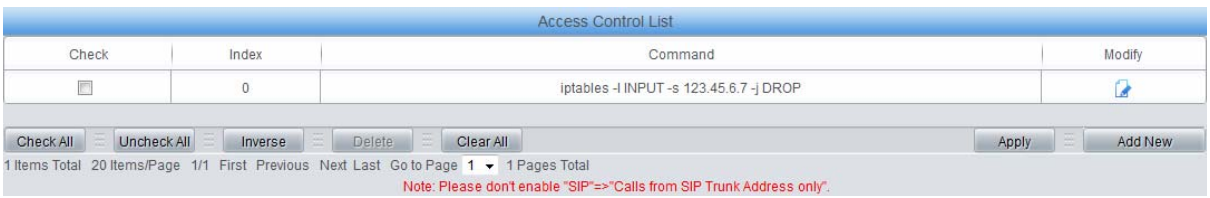


Figure 3-102 Access Control List Interface

See Figure 3-102 for the Access Control List interface. Once you add a piece of command to ACL, the network flow will be restricted: only the particular devices are allowed to visit the gateway and only the data packages on the designated ports can be forwarded. Click ***Add New*** to add a new piece of command. See Figure 3-103.

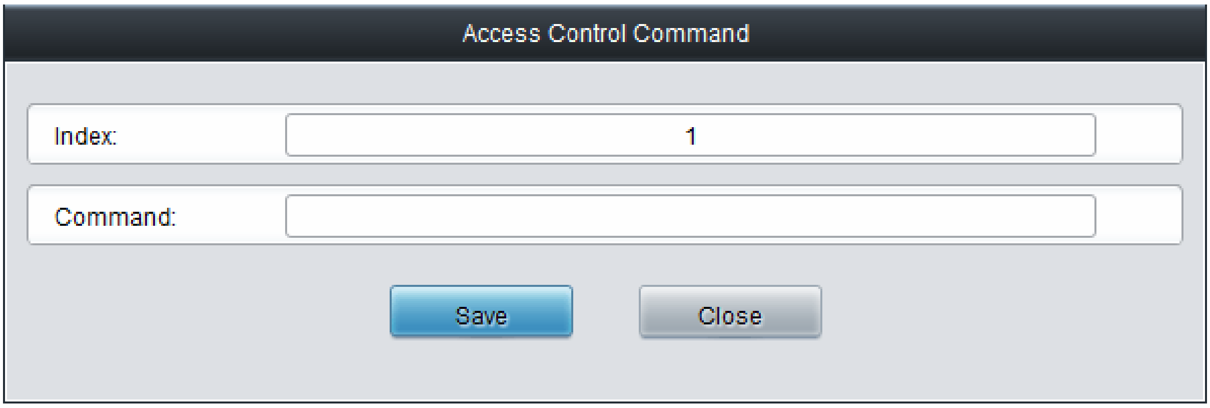


Figure 3-103 Add Access Control Command Interface

Input a piece of command into the Command item and click ***Save*** to save the settings to the gateway. Click ***Close*** to cancel your settings. After that, click ***Apply*** to make the new command valid.

Click ***Modify*** in Figure 3-102 to modify a command. See Figure 3-104 for the Access Control Command Modification interface. The configuration items on this interface are the same as those on the ***Add Access Control Command*** interface. Note that the item ***Index*** cannot be modified.

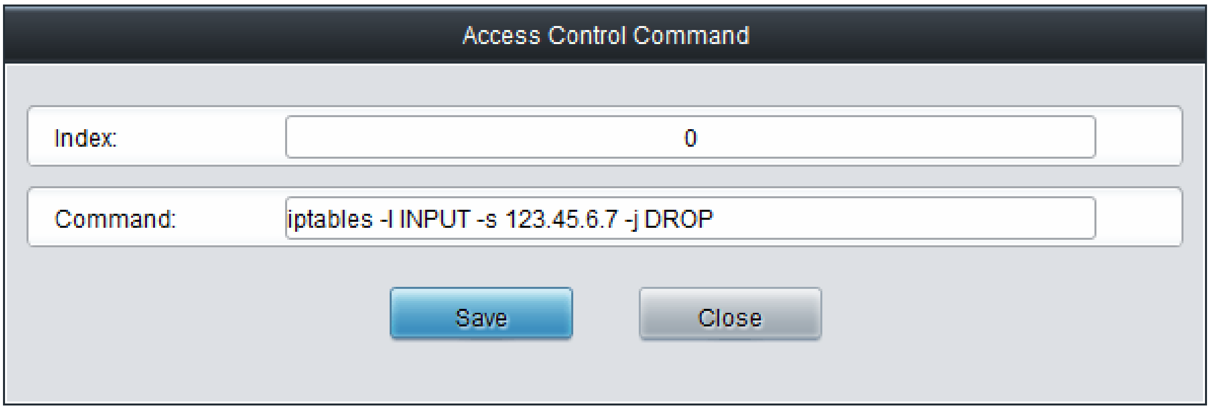


Figure 3-104 Access Control Command Modification Interface

To delete an Access Control Command, check the checkbox before the corresponding index in Figure 3-102 and click the ***Delete*** button, and then click the ***Apply*** button to make the deleted command invalid. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all access control commands at a time, click the ***Clear All*** button in Figure 3-102.

Note:

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

2. After you add, modify or delete a command manually, don't forget to click the ***Apply*** button to make your settings valid. However, in case the gateway restarts or the configuration is leading-in, the command will get valid automatically without the need for you to click the ***Apply*** button.

3.9.13 PING Test

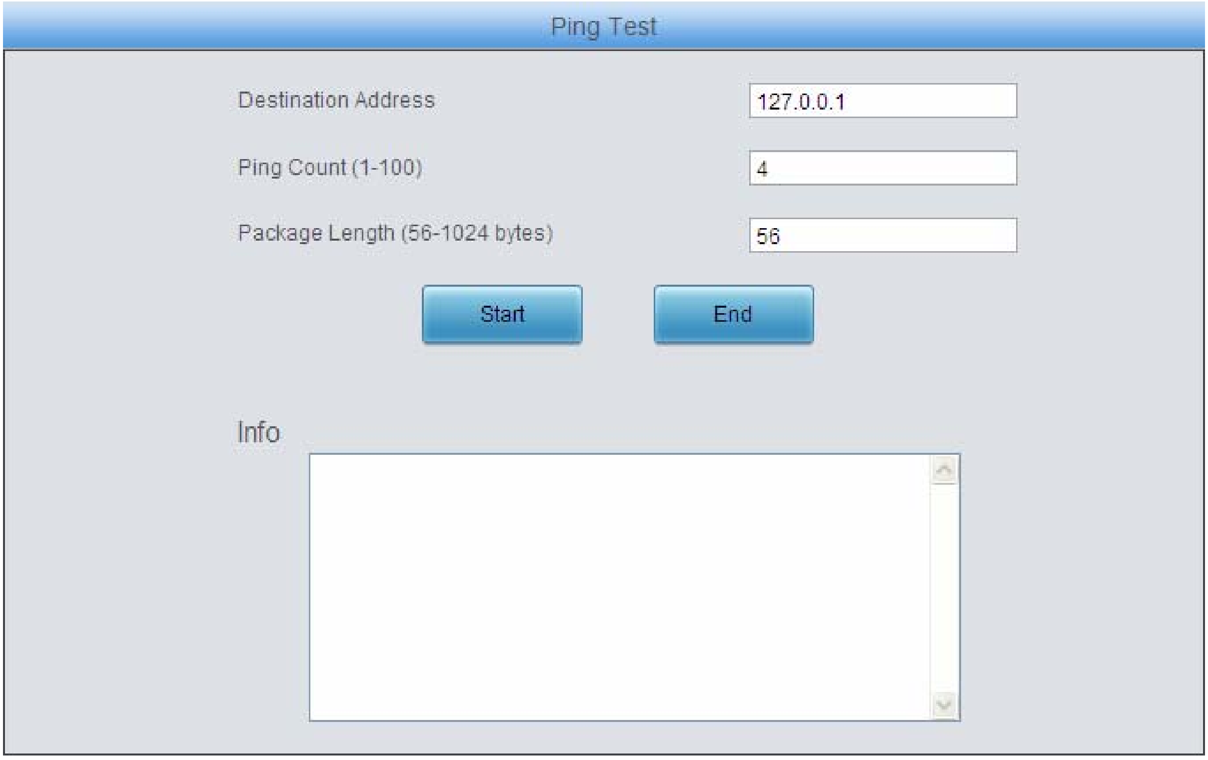


Figure 3-105 Ping Test Interface

See Figure 3-105 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
Destination Address	Destination IP address or domain name on which the Ping test is executed.
Ping Count	The number of times that the Ping test should be executed. Range of value: 1~100.
Package Length	Length of the data package used in the Ping test. Range of value: 56~1024 bytes.
Info	The information returned during the Ping test, helping you to learn the network connection status between the gateway and the destination address.

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

3.9.14 TRACERT Test

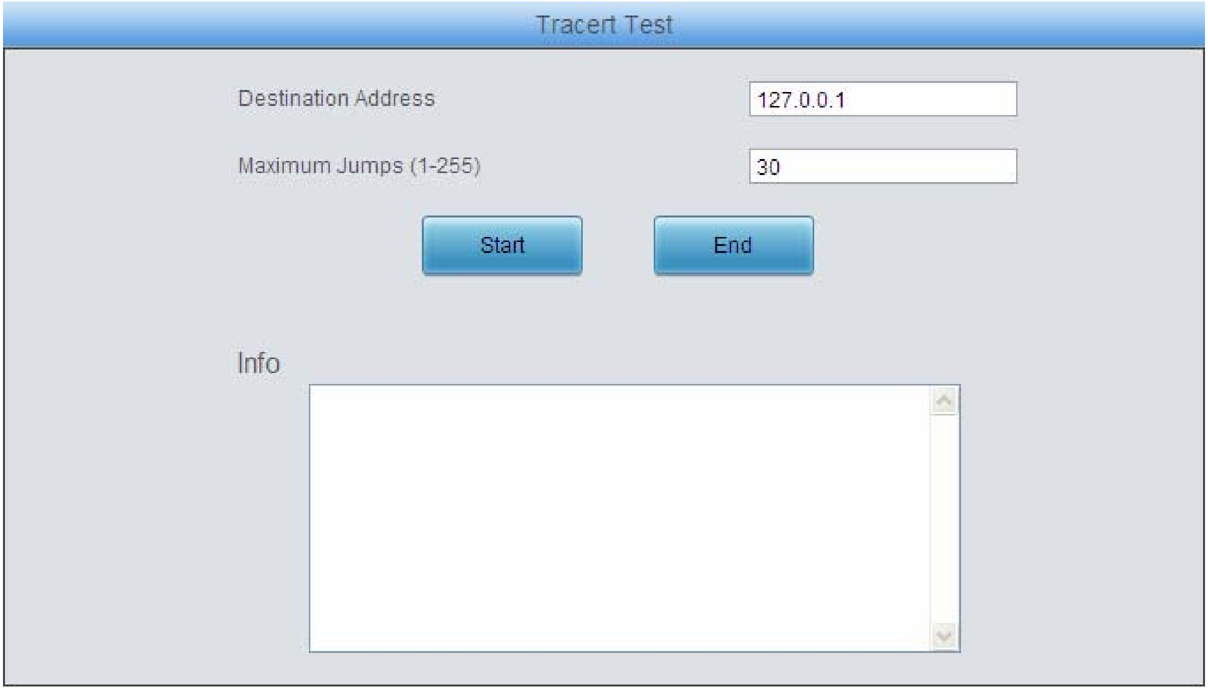


Figure 3-106 Tracert Test Interface

See Figure 3-106 for the Tracert test interface. A Tracert test can be initiated from the gateway on a designated IP address to check the routing status between them. The table below explains the configuration items shown in the above figure.

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

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

3.9.15 Change Password

Figure 3-107 Password Changing Interface

See Figure 3-107 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.

3.9.16 Restart

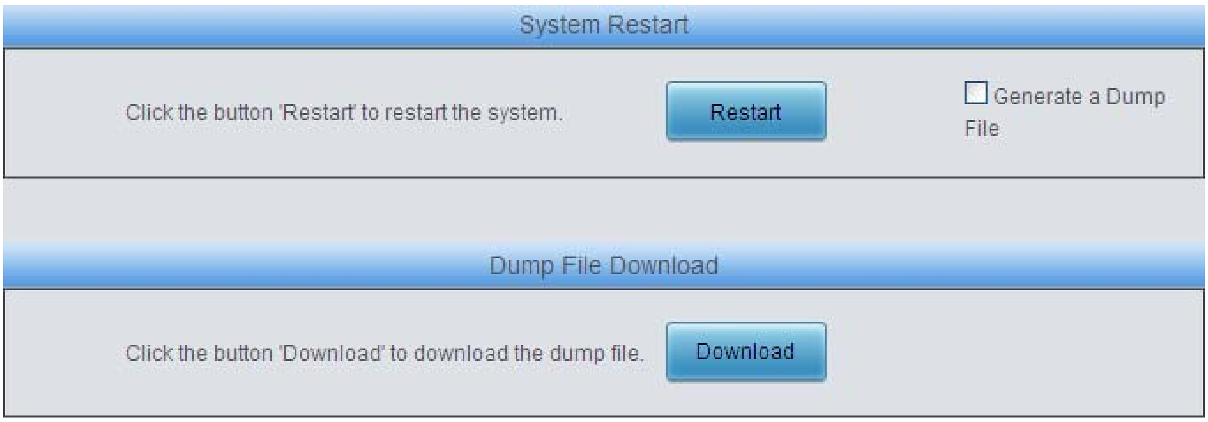


Figure 3-108 System Restart Interface

See Figure 3-108 for the Restart interface. Click ***Restart*** to restart the whole gateway system. A dump file will be generated each time you restart the system. Click ***Download*** and you can download it to help troubleshoot issues.

Appendix A Technical Specifications

Dimensions

SMG1008C: 210×30×150 mm^3

SMG1016C: 440×44×200 mm^3

SMG1032C: 440×44×260 mm^3

Weight

SMG1008C: 0.83 kg

SMG1016C-16S, SMG1016C-16O : 3.13 kg

SMG1032C-32S: 3.18 kg

SMG1032C-32O: 3.01 kg

Environment

Operating temperature: 0℃—45℃

Storage temperature: -20℃—85℃

Humidity: 8%— 90% non-condensing

Storage humidity: 8%— 90% non-condensing

LAN

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

Self-adaptive bandwidth supported

Auto MDI/MDIX supported

FXS Port

Amount: 8/16/32

Type: RJ11

Maximum transmission distance: 5000m

Impedance

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

Console 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:

SMG1008C：12V the direct current bigger than 3A

SMG1016C, SMG1032C： 100～240V AC

Signaling & Protocol

SIP signaling

Supported protocol: SIP V1.0/2.0, RFC3261

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

Appendix B Troubleshooting

Q1. What to do if I forget the IP address of the SMG-C 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) Dial the corresponding function key through an FXS port to query the IP address. See 3.5.8 Function Key for more details.

Q2. The SMG-C 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?

By default, you can make Tel→Tel calls without any routing configuration.

If you need to make Tel→Tel calls in a specific way, try 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. 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. **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. Fill in 'Source IP' with the gateway's IP address (e.g. **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.

Q3. Does call forwarding involve routing and number manipulation?

Case 1: If the forwarding number is the number of the gateway port. There is no need to use routing and number manipulation rules. Because the gateway will find the corresponding number according to the forwarding number and make a call.

Case 2: If the forwarding number is not the number of the gateway port. It is required to use routing and number manipulation rules. 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.

Q4. In what cases can I conclude that the SMG-C 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.

Q5. What to do if I cannot enter the WEB interface of the SMG-C 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.

Q6. How many ports can be rung by turns according to the *Ringing by Turns* rule?

According to the 180s ringing timeout limit in RFC3261 protocol, the time used for ringing all ports by turns cannot exceed 180s. Therefore, based on the minimum timeout 15s for each port in the ringing queue, the maximum number of ports for ringing by turns is 12.

For example, if you set ***Timeout for Ringing by Turns*** to 20s, the maximum number of ports for ringing by turns should be 180s/20s=9; if you set ***Timeout for Ringing by Turns*** to 30s, the maximum number of ports for ringing by turns should be 180s/30s=6.

Q7. Is there any cell-phone APP can make calls to the SMG-C 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,

Q8. Does the SMG-C gateway support fax?

Yes. Currently the SMG-C gateway supports two fax modes: T.38 and Pass-Through.

Q9. Which RTP codecs are supported by the SMG-C gateway?

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

Q10. How to configure the features Communication without Power and Communication without Network for the SMG-C analog gateway?

The feature **Communication without Power** is implemented in hardware. Once the power to the device is cut off, the station which is linked with the FXS port and the trunk which is linked with the FXO port will connect to each other directly and keep the good communications between phones and networks. The FXS and FXO ports are one-to-one correspondence (Take SMG1016C-8S8O for example, the phone linked with Channel 1 will be connected to the PSTN line which links with Channel 9.).

The feature **Communication without Network** is implemented via the WEB management over the analog gateway. It will automatically route a call to the FXO port in case of network failure or call timeout.

Refer to Q2 in this chapter for detailed information.

Appendix C Technical/sales Support

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

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