Synway SMG Series Analog Gateway

SMG1008

SMG1016

SMG1032

SMG1032A2

SMG1032A4

Analog Gateway

User Manual

Version 1.6.4

Synway Information Engineering Co., Ltd

www.synway.net

Content

Copyright Declaration

Revision History

Version	Date	Comments
Version 1.0	2013-10	Initial publication
Version 1.3.0	2014-03	New revision
Version 1.3.1	2014-06	Add description on the new series SMG1032A2
Version 1.3.2	2014-07	New revision
Version 1.3.3	2014-09	New revision
Version 1.3.5	2014-10	New revision
Version 1.5.0	2014-12	Add description on the new series SMG1032A4
Version 1.5.1	2015-01	New revision
Version 1.5.2	2015-04	New revision
Version 1.5.3	2015-11	New revision
Version 1.6.0	2016-03	New revision
Version 1.6.3	2016-12	New revision
Version 1.6.4	2017-02	New revision

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

Chapter 1 Product Introduction

Thank you for choosing Synway SMG Series Analog Gateway!

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

SMG series analog gateway has five modules:

- SMG1008: 8 FXS/FXO
- SMG1016: 16 FXS/FXO
- SMG1032, SMG1032A2, SMG1032A4: 32 FXS/FXO

1.1 Typical Application

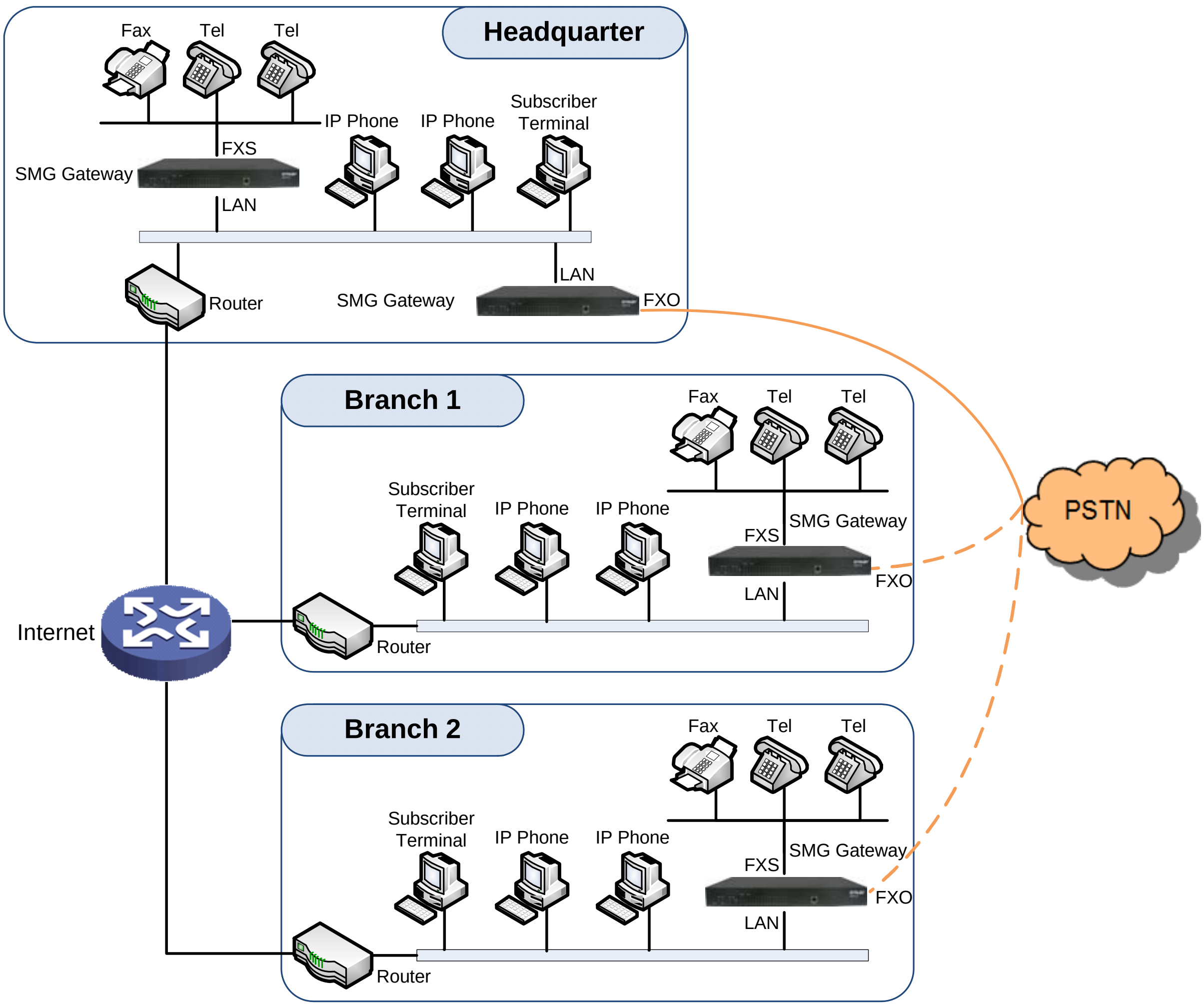


Figure 1-1 Typical Application

1.2 Feature List

Basic Features	Description
TDM Call	Call initiated from TDM to IP, via routing and number manipulation to obtain the called IP address.
IP Call	Call initiated from IP to TDM, via routing and number manipulation to obtain the call destination.
Number Manipulation	Peels off some digits of a phone number from left/right, or adds a prefix/suffix to a phone number.
Call Forward	Three options available: Unconditional, Busy and No Reply.
Call Waiting	When an FXS channel receives another call while it is in conversation, it will have the newly received call keep waiting. Once the current call is finished, the new one will ring the FXS channel and wait for its answer.
Auto Dial	If there is no dialing operation in a designated time period after pickup, the preset auto dial number will be called.
Do Not Disturb	Rejects all the incoming calls to the channel.
CID	Displays the CallerID.
Echo Cancellation	Provides the echo cancellation feature for a call conversation over the FXS/FXO channel.
TDM/VoIP Routing	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	Provides composite modules to enable a direct 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 platform 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, SILK(16K), OPUS(16K), SILK(8K), OPUS(8K) 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.
PPPoE	Virtual dial-up internet access 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.

1.3 Hardware Description

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

Figure 1-2 SMG1032 Front View

Figure 1-3 SMG1032 Rear View

Figure 1-4 SMG1032A2 Front View

Figure 1-5 SMG1032A2 Rear View

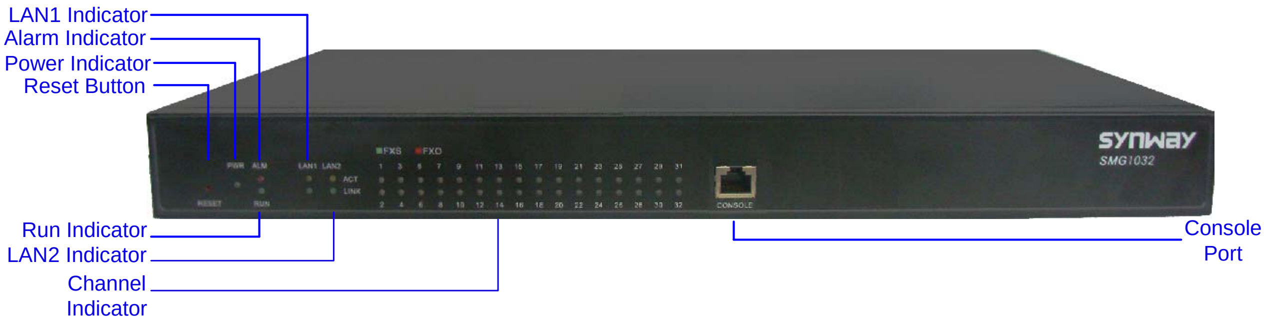


Figure 1-6 SMG1032A4 Front View

Figure 1-7 SMG1032A4 Rear View

Figure 1-8 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, RJ-21, RJ45
	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**
Power Key	Power on/off the SMG analog gateway.
Reset Button	Restore the gateway to factory settings.
LED	**Description**
Power Indicator	Indicates the power state. It lights up when the gateway starts up with the power cord well connected
Run Indicator	Indicates the running status. For more details, refer to 1.4 Alarm Info.
Alarm Indicator	Alarms the device malfunction. For more details, refer to 1.4 Alarm Info.
Link Indicator	The green LED on the left of LAN, indicating the network connection status.

ACT Indicator	The orange LED on the right of LAN, whose flashing tells data are being transmitted.
Channel Indicator	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 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
- 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).

The connection for SMG1008, SMG1016, SMG1032 series products:

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.

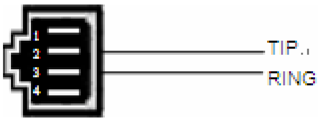


Figure 2-1 RJ11 Connection

The connection for SMG1032A2 series product:

SMG1032A2 adopts two RJ21 interfaces each of which accommodates 16 channels. One corresponds to channels 1 through 16 and the other corresponds to 17 through 32. Each pin in the RJ21 connector functions as follows.

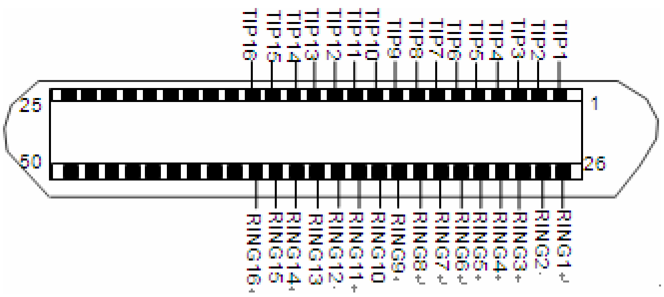


Figure 2-2 RJ21 Pin Layout

The pins Ch1-a/b through Ch16-a/b on the RJ21 interface will be used respectively corresponding to channels 1 through 16.

An RJ21 interface can be converted to 24 RJ11 interfaces through an RJ21-to-RJ11 adapter. See Figure 2-3 for the connection. SMG1032A2 needs two RJ21-to RJ11 adapters of which the first 16 slots will be used.

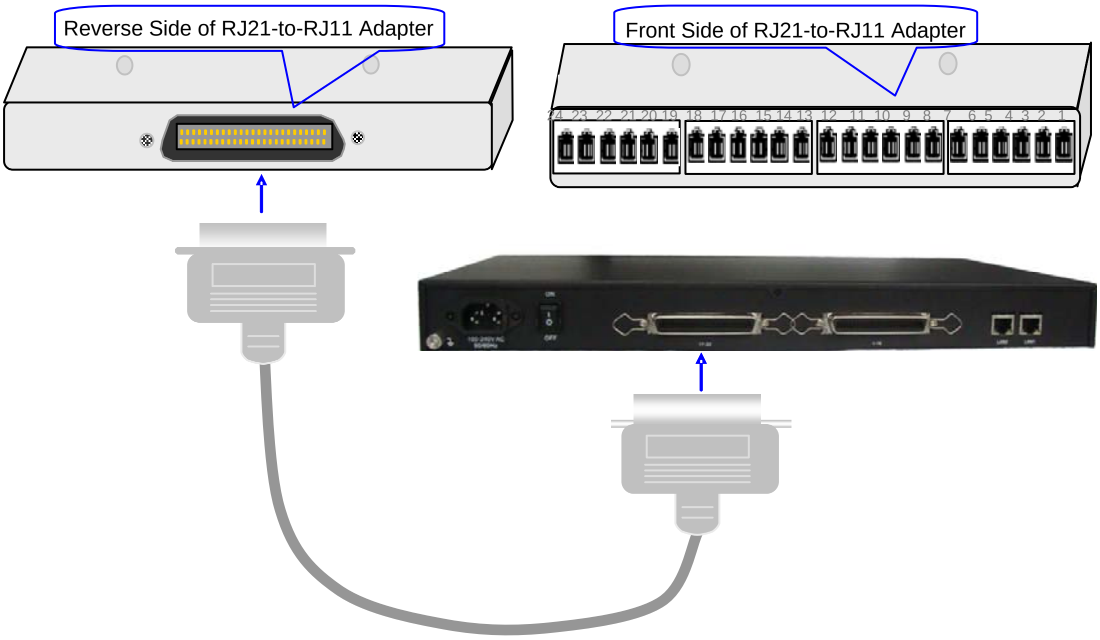


Figure 2-3 RJ21-to-RJ11 Adapter Connection

Users can also use the RJ21 connecting cable directly.

SMG1032A4 has eight 8-pin RJ45 jacks each of which can be connected to four 2-pin RJ11 jacks via a 4-way hub. Take the first RJ45 jack for example, the matching relationship among the channel number, the pins of the RJ45 jack and the 4-way hub is shown in the table below.

Interface	Channel Number	Pins of the RJ45 Jack	4-way Hub
First RJ45 Jack	1	1st and 2nd pins	1st jack
	2	3rd and 4th pins	2nd jack
	3	5th and 6th pins	3rd jack
	4	7th and 8th pins	4th jack

Table 2-1 Matching Relationship among Channel Number, Pins of RJ45 Jack and 4-way Hub

Step 6: Power on and start the gateway.

Step 7: 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 3.9.14 Change Password. After changing the password, you are required to log in again.

Step 8: 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 9: 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 10: 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 11: 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 12: 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 13: Perform call forwarding. (Skip this step if not necessary.)

Situation 1: Hook-flash operation

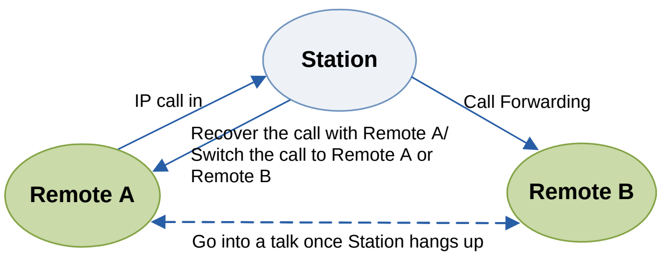


Figure 2-4 Call Forward via Hook-flash

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 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 1-8) 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.

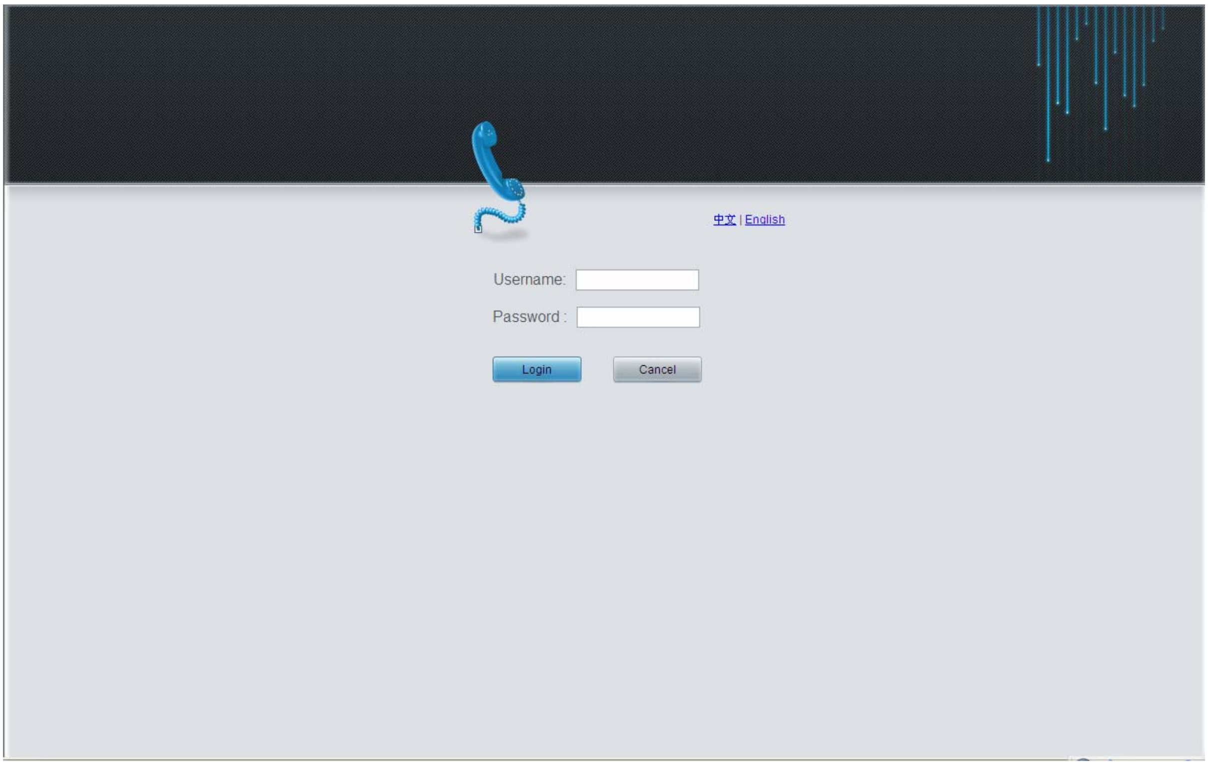


Figure 3-1 Login Interface

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 3.9.14 Change Password.

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

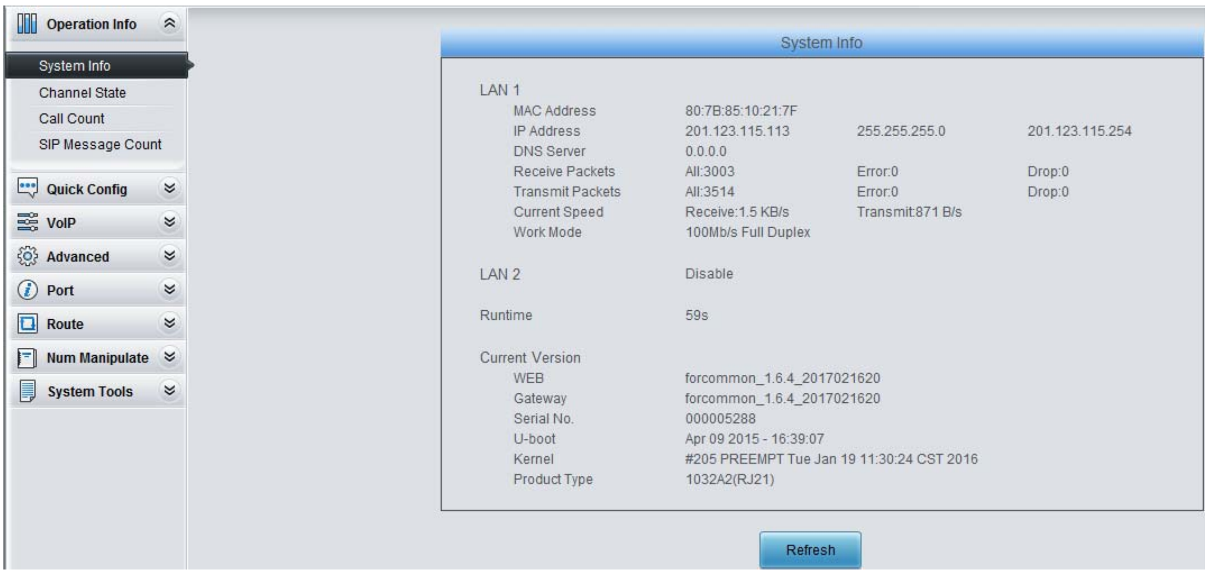


Figure 3-2 Main Interface

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.

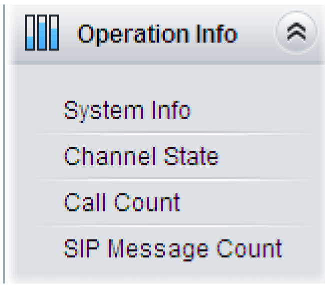


Figure 3-3 Operation Info

3.2.1 System Info

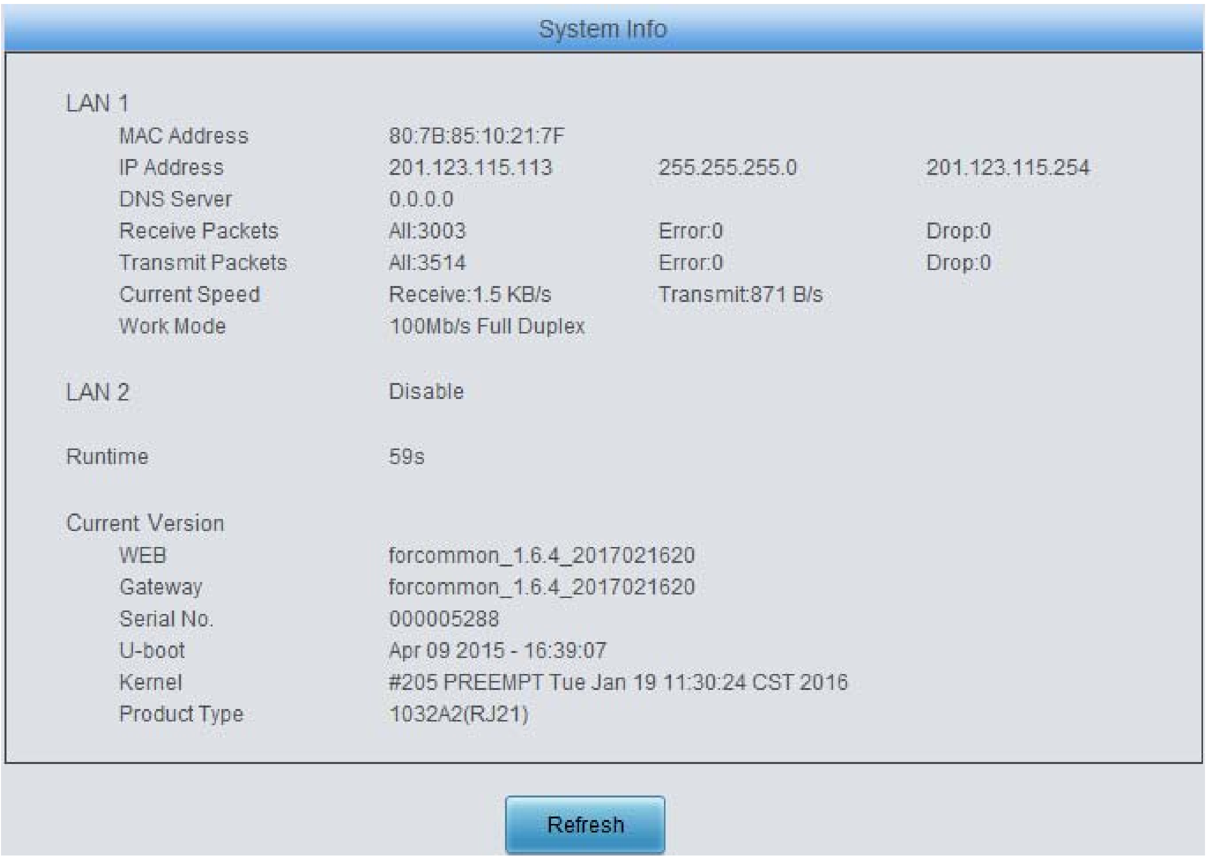


Figure 3-4 System Info Interface

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 1 or LAN 2 (disabled by default).
IP Address	The three parameters from left to right are IP address, subnet mask and default gateway of LAN 1 or LAN 2 (disabled by default).
DNS Server	DNS server address of LAN 1 or LAN 2 (disabled by default).

Receive Packets	The amount of receive packets after the gateway's startup, including three categories: All, Error and Drop.
Transmit Packets	The amount of transmit packets after the gateway's startup, including three categories: All, Error and Drop.
Current Speed	The current speed of data receiving and transmitting.
Work Mode	The work mode of the network, including four options: 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 No.	Unique serial number of an SMG analog gateway.
U-boot	Current version of Uboot.
Kernel	Current version of the system kernel on the gateway. **Note:** The kernel version for the gateways with RJ45/RJ21 interface is different from that for the gateways with RJ11 interface.
Product Type	The type of the analog gateway.

3.2.2 Channel State

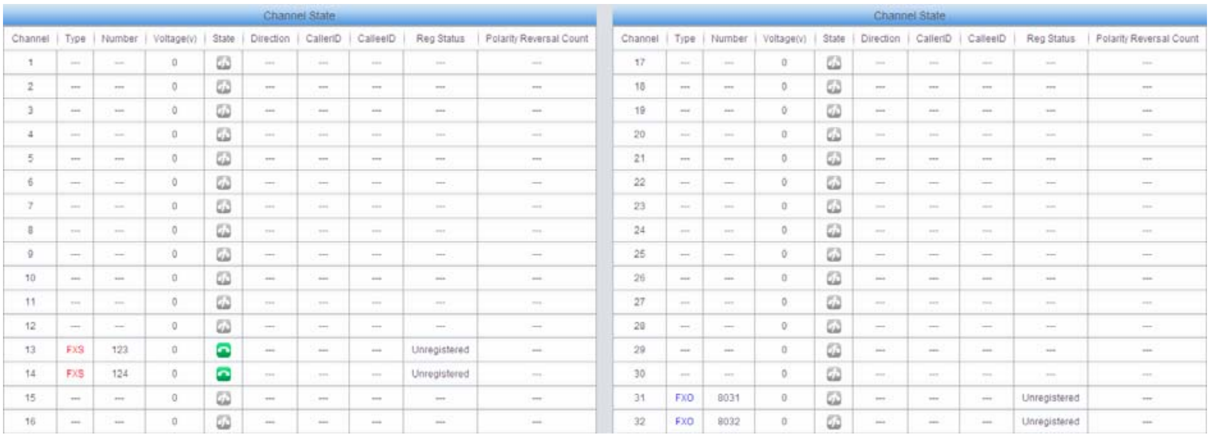


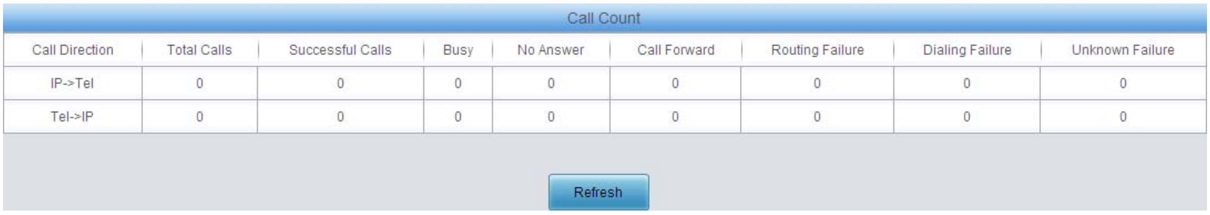
Figure 3-5 Channel State Interface

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: FXS or FXO. If this item shows ---, it means this channel is unavailable, that is, the corresponding module to this channel is not inserted or damaged. **Note:** If the FXO port is unconnected, the channel is unavailable too.
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								
Request	REGISTER	INVITE	ACK	INFO	BYE	CANCEL	NOTIFY	OPTION
Send	0	1	1	0	1	0	0	0
Send Repeatedly	0	0	0	0	0	0	0	0
Receive	0	1	1	0	1	0	0	0
Receive Repeatedly	0	0	0	0	0	0	0	0

Common Response						
Common Response	100 Trying	180 Ringing	183 Session 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.

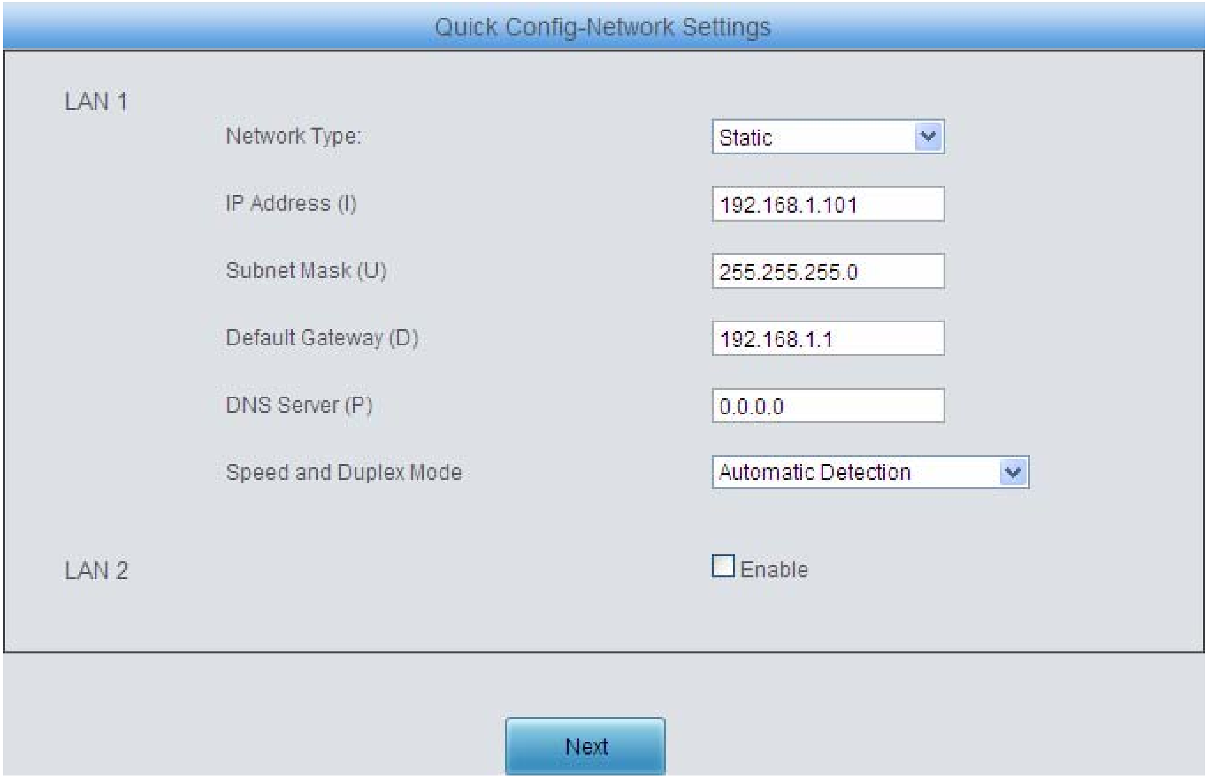


Figure 3-9 Quick Config-Network Settings Interface

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.

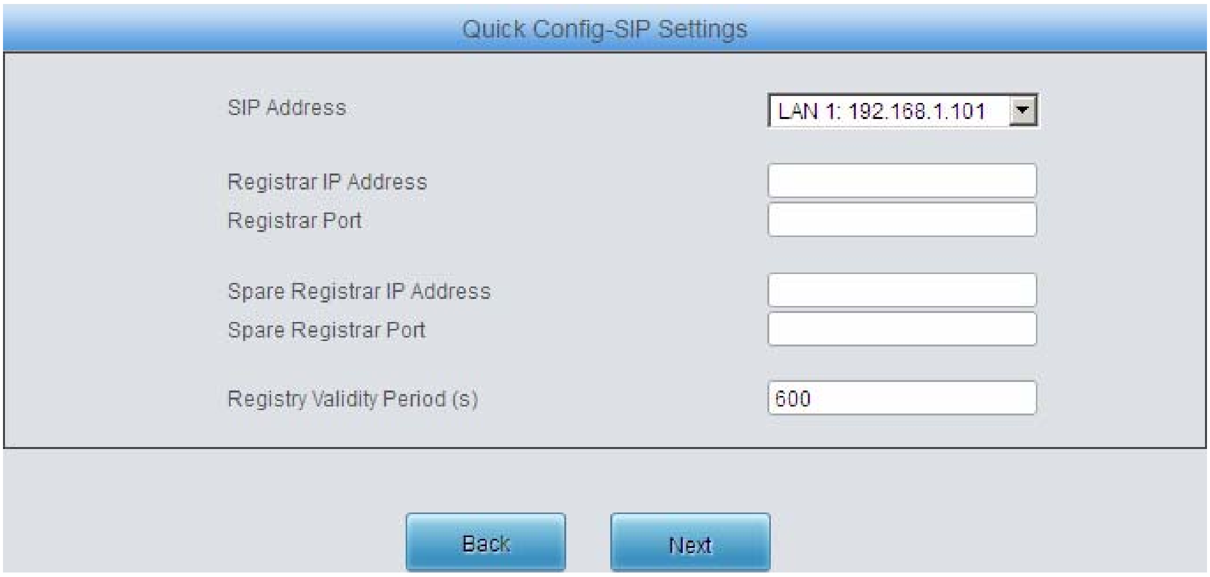


Figure 3-10 Quick Config-SIP Settings Interface

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	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	
9	FXS	8009	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	0	0	
10	FXS	8010	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	0	0	
11	FXS	8011	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	0	0	
12	FXS	8012	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	0	0	
13	FXS	8013	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	0	0	
14	FXS	8014	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	0	0	
15	FXS	8015	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	0	0	
16	FXS	8016	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	0	0	

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

Back Next

Figure 3-11 FXS Settings Interface

See Figure 3-12 for 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	Polarity Reversal Detection	Input Gain	Output Gain	Modify
17	FXO	8017	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
18	FXO	8018	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
19	FXO	8019	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
20	FXO	8020	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
21	FXO	8021	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
22	FXO	8022	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
23	FXO	8023	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
24	FXO	8024	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
25	FXO	8025	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
26	FXO	8026	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
27	FXO	8027	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
28	FXO	8028	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
29	FXO	8029	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
30	FXO	8030	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
31	FXO	8031	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	
32	FXO	8032	—	Two Stages Dialing for Incoming Call	—	Disable	Disable	Unregistered	Enable	Disable	0	0	

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

Back Next

Figure 3-12 FXO Settings Interface

Quick Config-Completion

The configuration is finished. Please click 'Finish' to quit the Quick Config!

Note: the gateway will restart the system after you click 'Finish'. Please log in the gateway again using your new IP address.

Back Finish

Figure 3-13 Quick Config-Completion Interface

Click ***Back*** to go back to the FXO 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*** 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

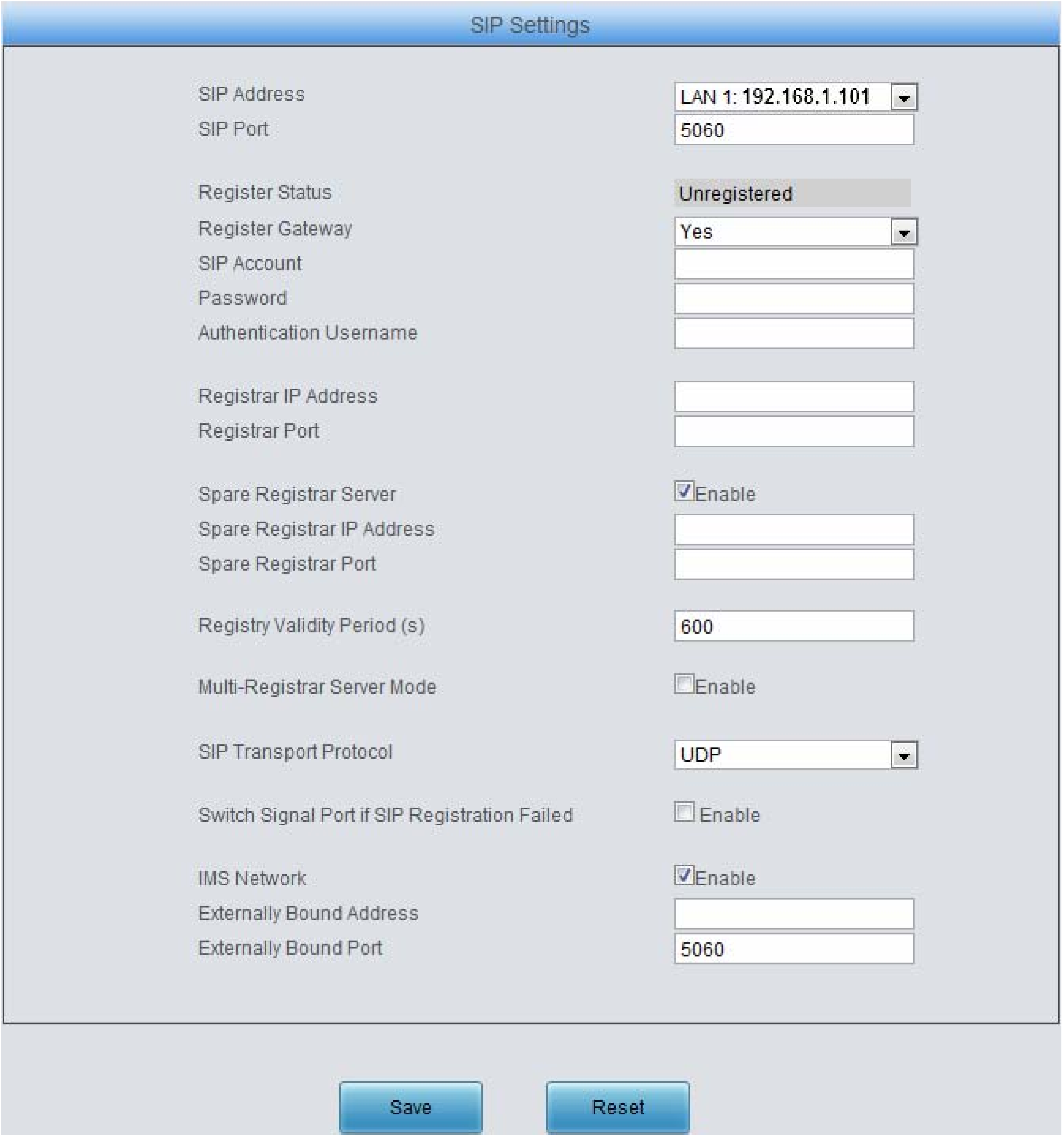


Figure 3-15 SIP Settings Interface

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 service, do it immediately to apply the changes. Refer to 3.9.15 Restart for detailed instructions. The table below explains the items shown in Figure 3-15.

Item	Description
SIP Address	IP address of SIP signaling, using LAN 1 by default.
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.

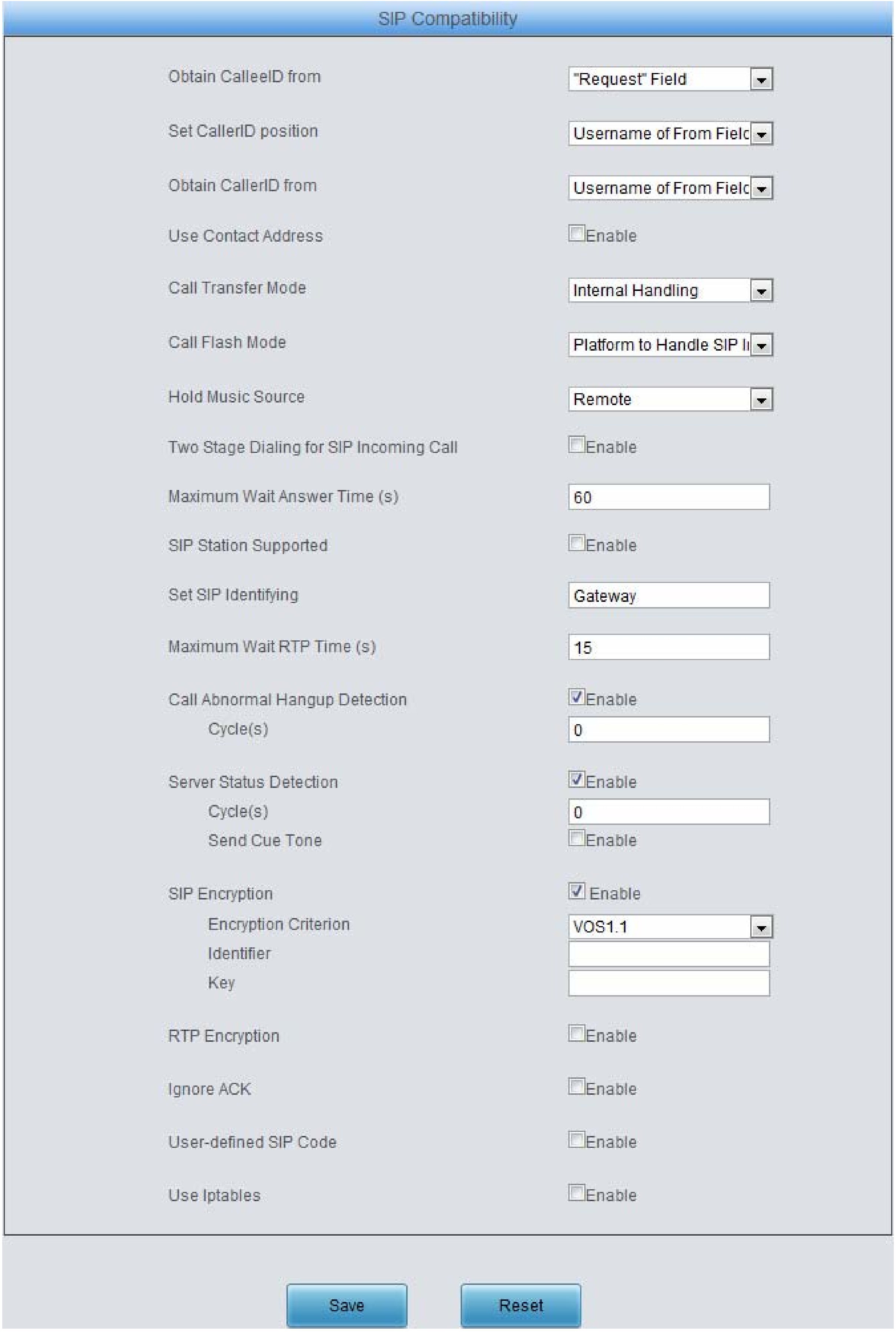


Figure 3-16 SIP Compatibility Setting Interface

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 Cycle	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, it is not necessary for the gateway to wait for the ACK message after sending the 200OK message to establish a call. By default it is *disabled*.
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 interface, and you will see the item SIP Station on the VoIP Settings menu. Click '***SIP Station***' to go into the SIP Station interface. By default, there is no available SIP station. See Figure 3-17 below.

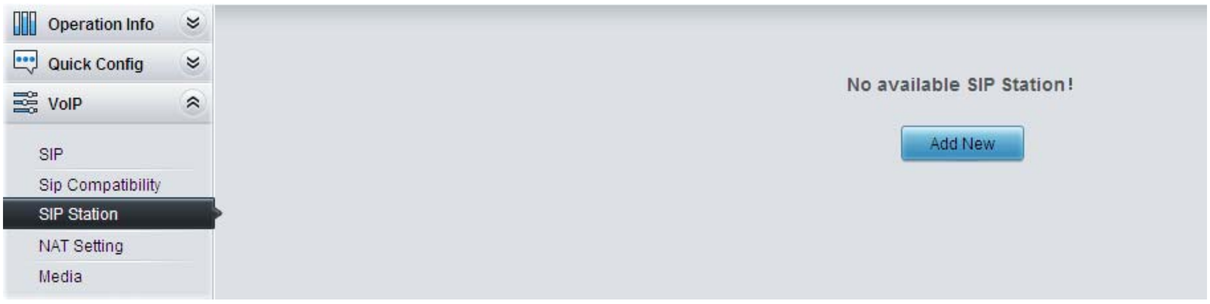


Figure 3-17 SIP Station Setting Interface

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.

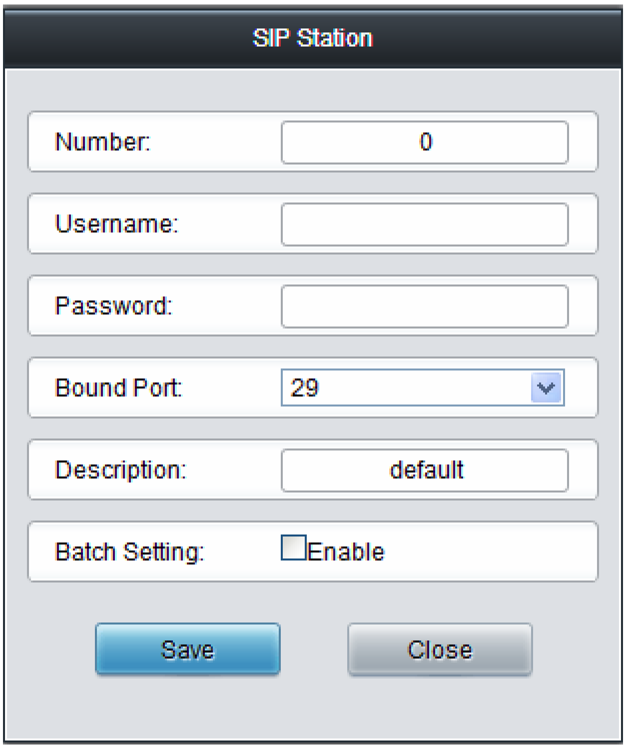


Figure 3-18 Add New SIP Station

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.

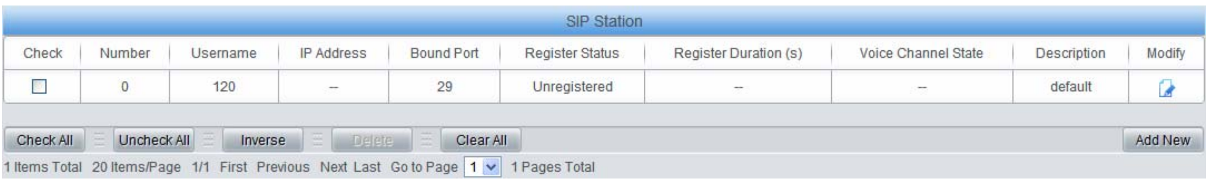


Figure 3-19 SIP Station Interface

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.

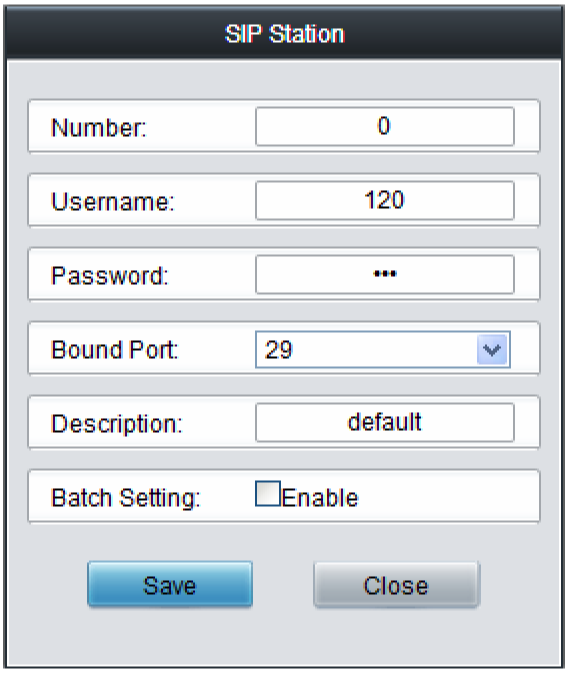


Figure 3-20 SIP Station Modification Interface

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.

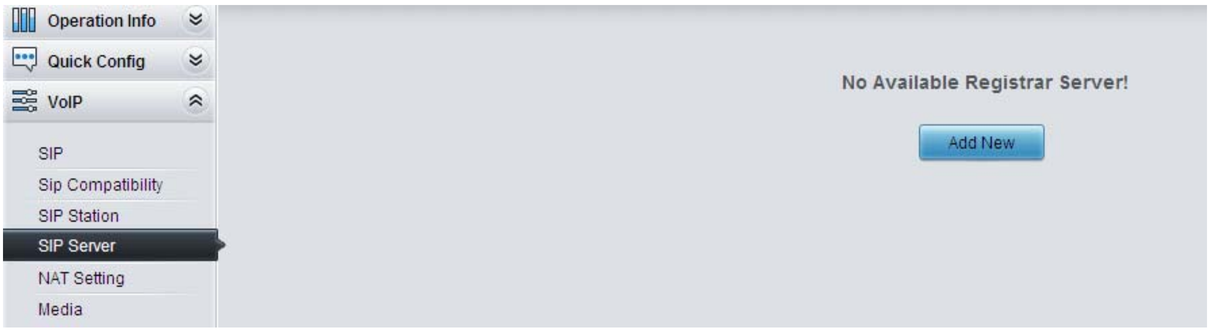


Figure 3-21 SIP Server Interface

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

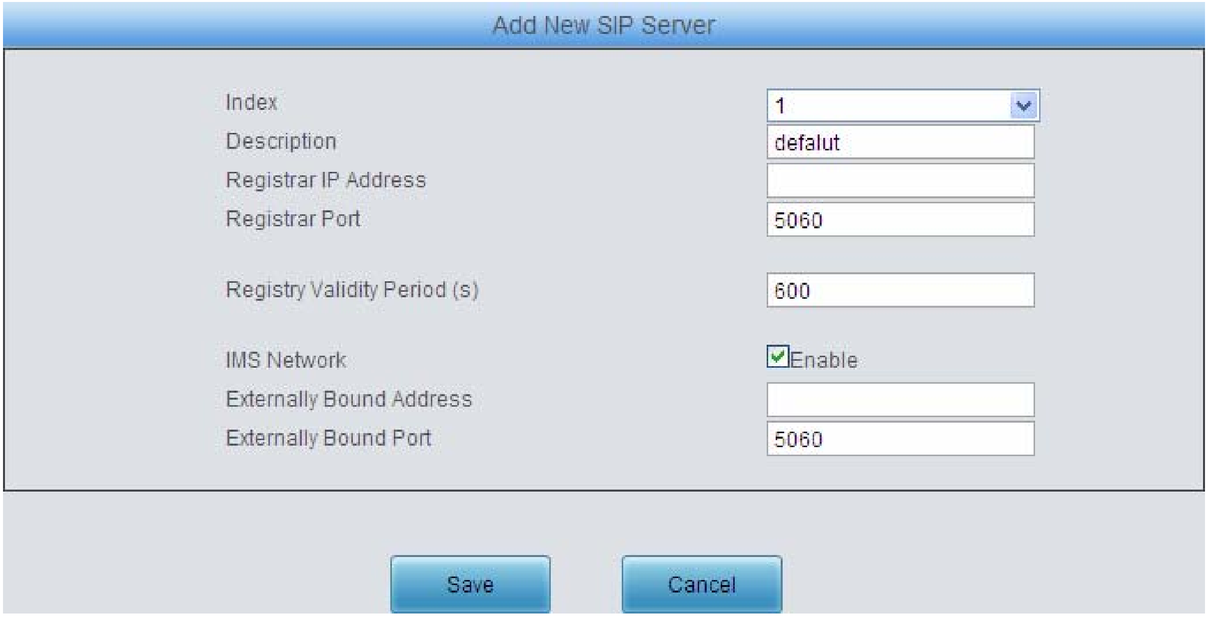


Figure 3-22 Add New SIP Server

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.

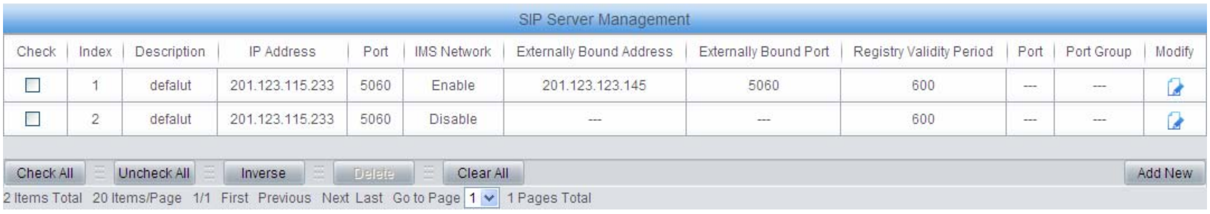


Figure 3-23 SIP Server Management

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.

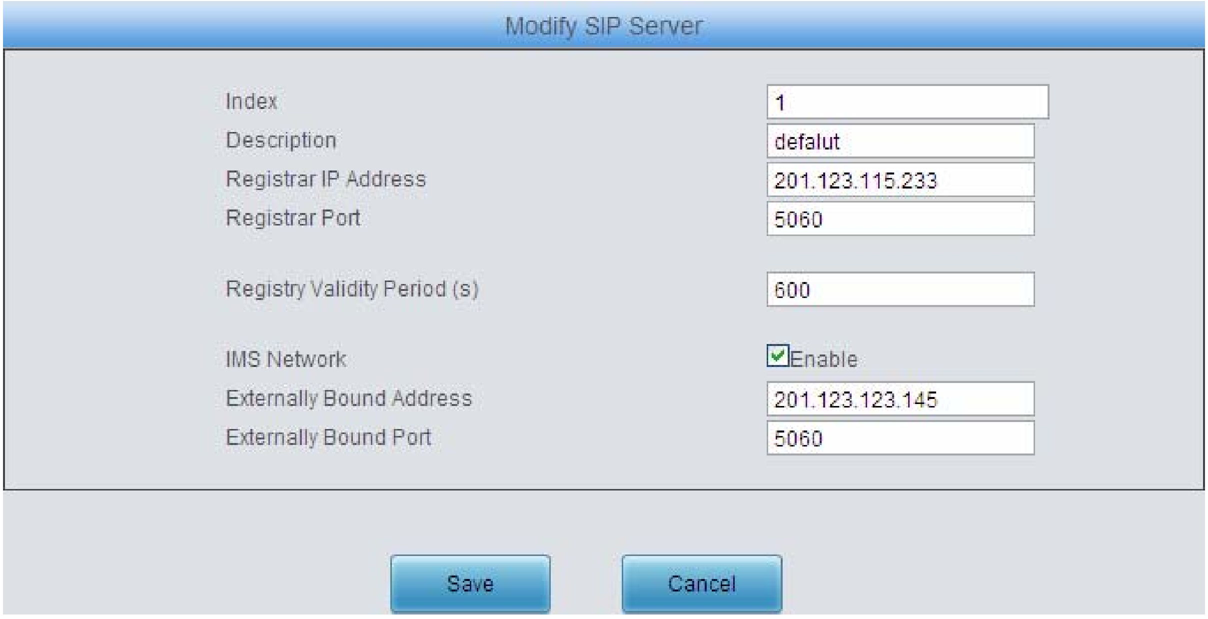


Figure 3-24 SIP Server Modification Interface

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.

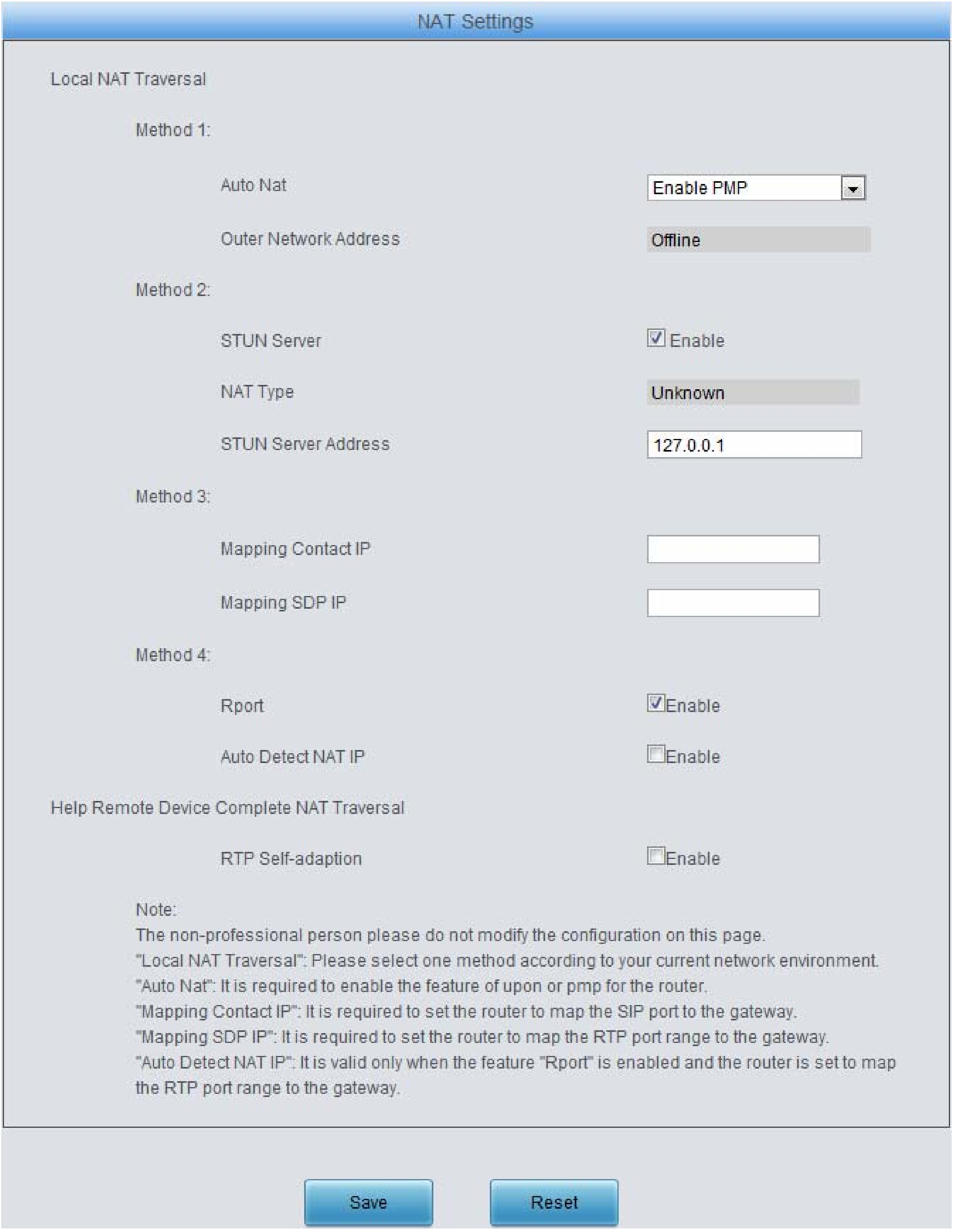


Figure 3-25 NAT Setting Interface

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

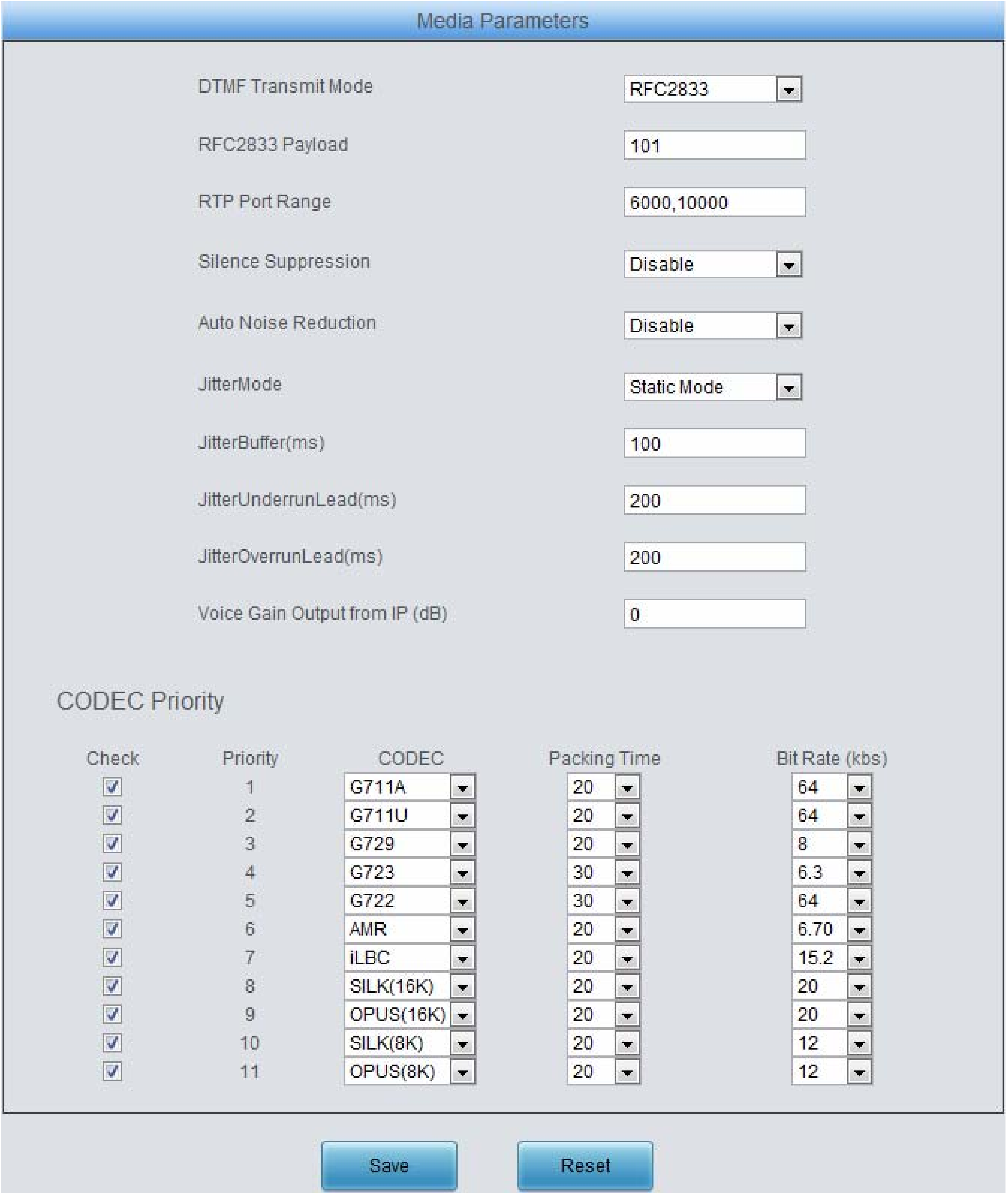


Figure 3-26 Media Settings Interface

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 service, do it immediately to apply the changes. Refer to 3.9.15 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 2000 and the upper limit of 60000 and the difference between larger than 480. The default value is 6000-10000.
Silence Suppression	Sets whether to send comfort noise packets to replace RTP packets or never to send RTP packets to reduce the bandwidth usage when there is no voice signal throughout an IP conversation. The optional values are *Enable* and *Disable*, with the default value of *Disable*.
Auto Noise Reduction	Once this feature is enabled, the volume of the noise accompanied with the line will be reduced automatically. By default, the feature is *disabled*.
JitterMode	Sets the working mode of JitterMode. The optional values are *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: 0~280, calculated by ms, with the default value of 100.
JitterUnderrunLead	Sets the initial delay of packets if they are received later than JitterBuffer. Range of value: 0~280, calculated by ms, with the default value of *200*, **Note:** Only when JitterMode is set to *Static Mode* will this item be shown.
JitterOverrunLead	Sets the initial lead inserted if packets are received earlier than 300-JitterBuffer. Range of value: 0~280, calculated by ms, with the default value of *200*, **Note:** Only when JitterMode is set to *Static Mode* will this item be shown.
JitterMin	Sets the minimum delay that can be set by the adaptive jitter function. It can not be larger than the value set in JitterBuffer. Range of value: 0~280, calculated by ms, with the default value of *10*. **Note:** Only when JitterMode is set to *Adaptive Mode* will this item be shown.
JitterDecreaseRatio	Sets the rate for delay reduction under the adaptive mode. It defines the maximum percentage of silence that can be removed for delay reduction. Range of value: 0~100, with the default value of *50*, **Note:** Only when JitterMode is set to *Adaptive Mode* will this item be shown.
JitterIncreaseMax	Sets the maximum delay that can be increased during a silence period. Range of value: 0~280, calculated by ms, with the default value of *50*, **Note:** Only when JitterMode is set to *Adaptive Mode* will this item be shown.
Voice Gain Output from IP	Adjusts the gain of the voice output from IP. Range of value: -24~24, calculated by dB, with the default value of 0.

CODEC Priority	Supported CODECs and their corresponding priority for the IP end to establish a call conversation. The table below explains the sub-items:

Sub-item	Description
Priority	Priority for choosing the CODEC in an SIP conversation. The smaller the value is, the higher the priority will be.
CODEC	Three optional CODECs are supported: *G711A*, *G711U*, *G729A/B*, *G723*, G722, *AMR, iLBC, SILK(16K), OPUS(16K), SILK(8K) and OPUS(8K).*
Packing Time	Time interval for packing an RTP packet, calculated by ms.
Bit Rate	The number of thousand bits (excluding the packet header) that are conveyed per second.

By default, all of the eleven CODECs are supported and ordered *G711A, G711U, G729A/B, G723, G722, AMR, iLBC, SILK(16K), OPUS(16K), SILK(8K)* and *OPUS(8K)* 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.

COEDC	Packing Time (ms)	Bit Rate (kbps)
G711A	5 / 10 / **20** / 30 / 40 / 50 / 60	**64**
G711U	5 / 10 / **20** / 30 / 40 / 50 / 60	**64**
*G729A/*B	**20**	**8**
G723	**30** / 60 / 90	5.3 / **6.3**
G722	5 / 10 / 20 / **30** / 40	**64**
AMR	**20** / 40 / 60 / 80 / 100	4.75 / 5.15 / 5.90 / **6.70** / 7.40 / 7.95 / 10.20 / 12.20
iLBC	**20** / 40	**15.2**
	30	13.3
	60	13.3 / 15.2
SILK(16K)	**20** /40 / 60	**20**
OPUS(16K)	10 / **20** / 40 / 60	**20**
SILK(8K)	**20** /40 / 60	**12**
OPUS(8K)	10 / **20** / 40 / 60	**12**

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 analog voice ports, such as the conditions for sending the caller party information. ***Tone Detector*** is used to configure some properties of detected tones. ***Tone Generator*** is used to configure some properties of generated tones. ***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.

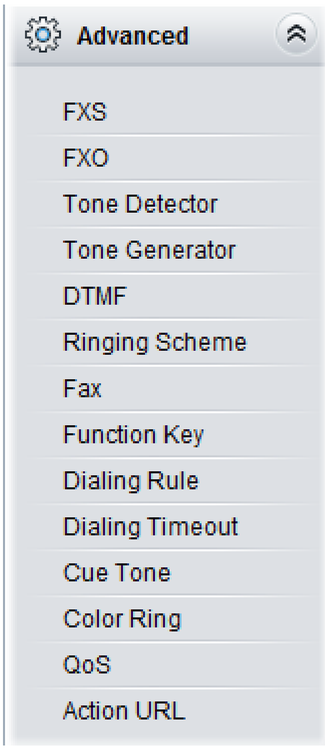


Figure 3-27 Advanced Settings

3.5.1 FXS

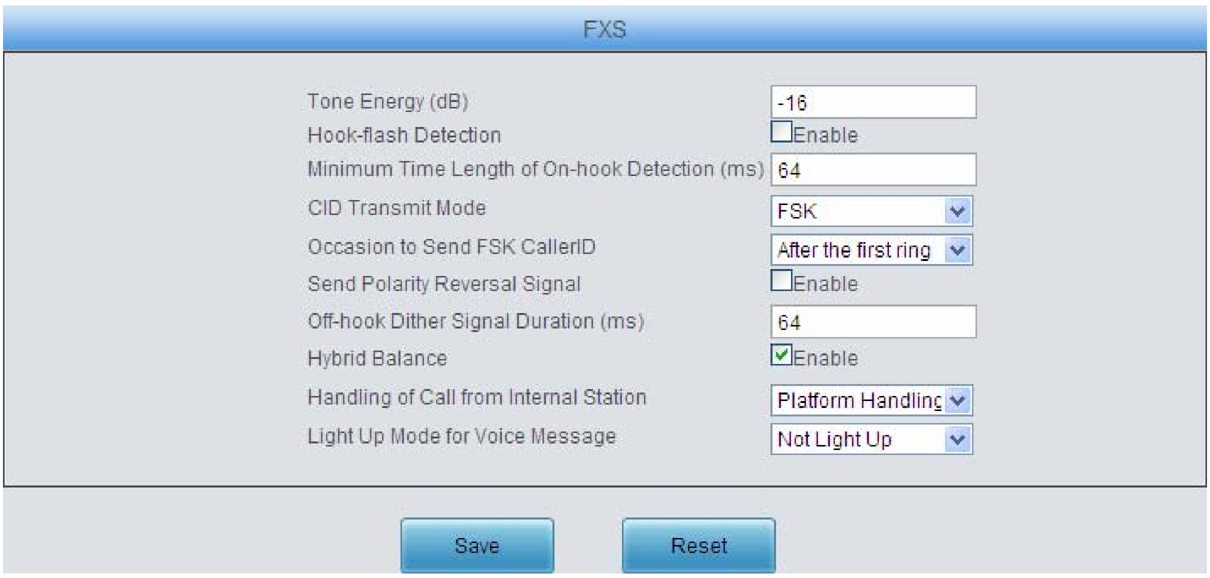


Figure 3-28 FXS Configuration Interface

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

Item	Description
Tone Energy	Energy of the tone signal sent by the gateway. Range of value: -35~15, calculated by dB, with the default value of -16.

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. **Note:** This item appears only when Hook-flash Detection is enabled.
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. **Note:** This item appears only when the hook-flash detection is enabled.
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 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.
Off-hook Dither Signal Duration	The minimum duration of the off-hook signal, calculated by millisecond (ms), which must be the multiple of 16. The less value indicates the larger sensitivity. And the default value is 64.
Hybrid Balance	Sets whether to enable the hybrid balance feature or not. The default setting is being enabled.
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 leaving a voice message on the phone, two options available: Not Light Up and Light Up by FSK, with the default value of Not Light Up.

After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to 3.9.15 Restart for detailed instructions.

3.5.2 FXO

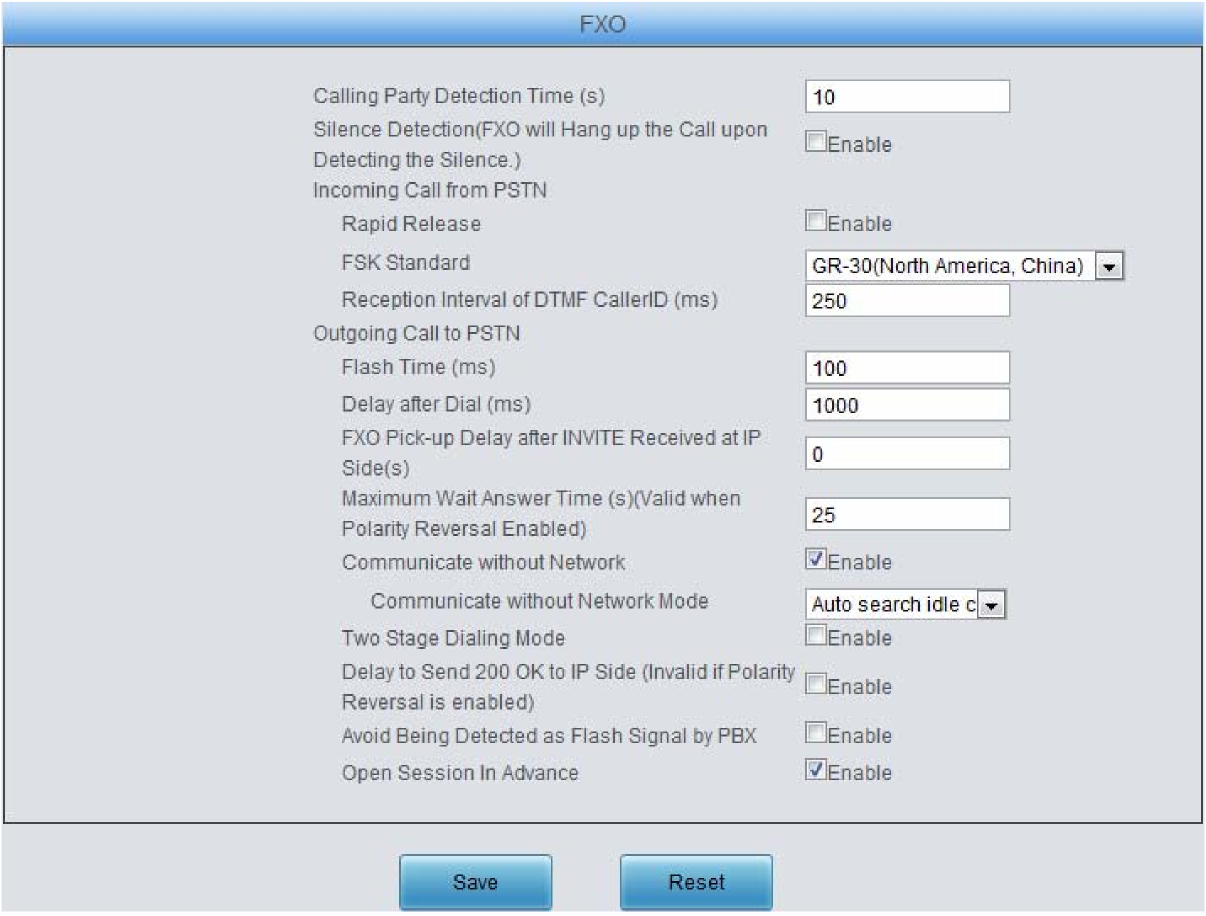


Figure 3-29 FXO Configuration Interface

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: -86~5, calculated by s, with the default value of -34. **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.
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*.
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.
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*.
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*.
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*.

After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to 3.9.15 Restart for detailed instructions.

3.5.3 Tone Detector

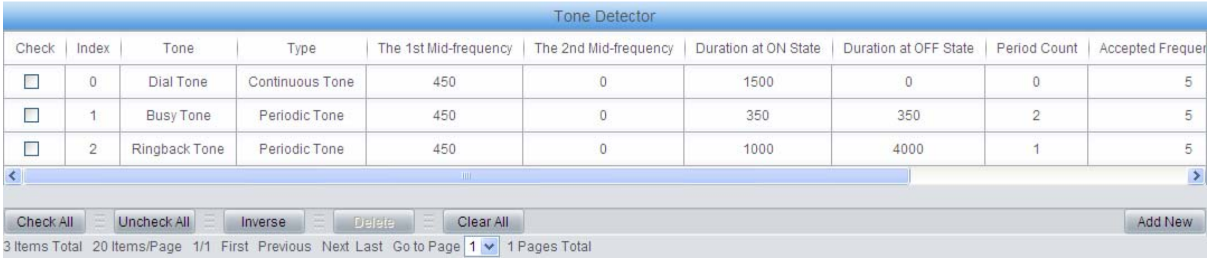


Figure 3-30 Tone Parameters Setting Interface

See Figure 3-30 for the Tone Parameters setting interface. By default, there are three pieces of tone parameters on the gateway. Click ***Add New*** to add tone parameters manually, see Figure 3-31.

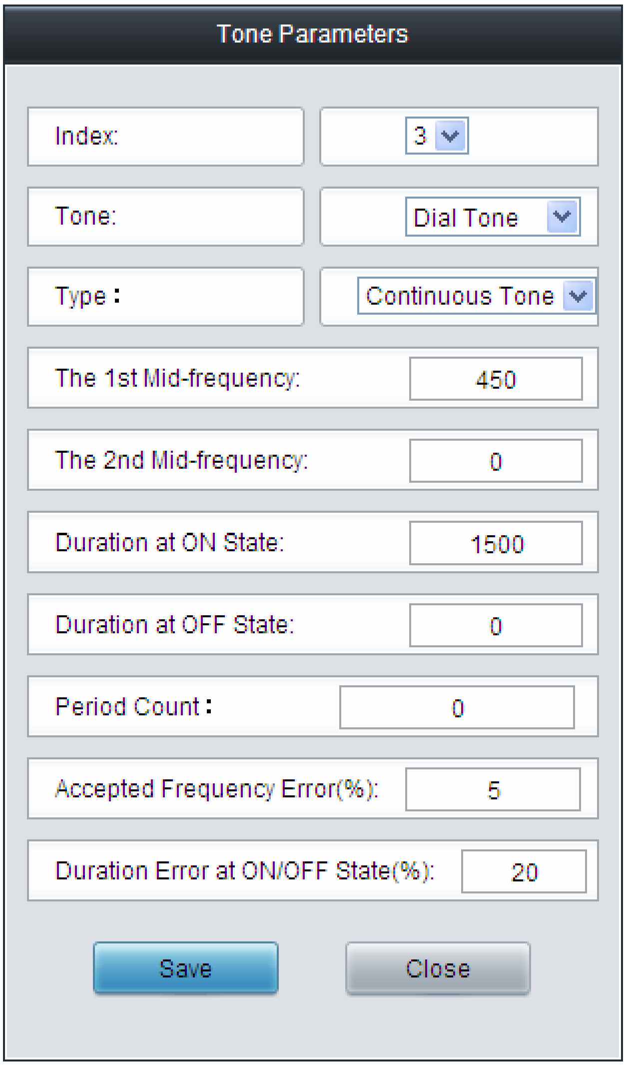


Figure 3-31 Add New Tone Parameter Interface

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	Set 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.
Accepted Frequency Error	Allowable error of the center frequency. Range of value: 1~5, calculated by %, with the default value of 5.
Duration Error at ON/OFF State	The accepted maximum error at on/off state. Range of value: 0~100, calculated by %, with the default value of 20.

After configuration, click ***Save*** to save the above settings into the gateway or click ***Close*** to cancel the settings. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to 3.9.15 Restart for detailed instructions.

Click ***Modify*** in Figure 3-30 to modify the tone parameter. See Figure 3-32 for the tone parameter modification interface. The configuration items on this interface are the same as those on the ***Add New Tone Parameter*** interface.

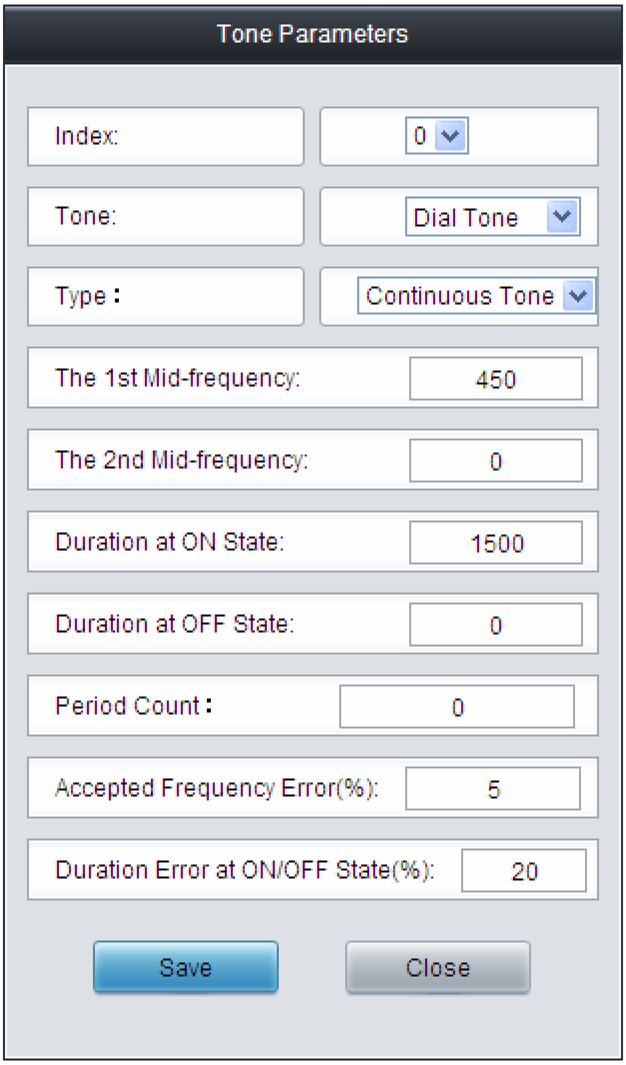


Figure 3-32 Modify Tone Parameter

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

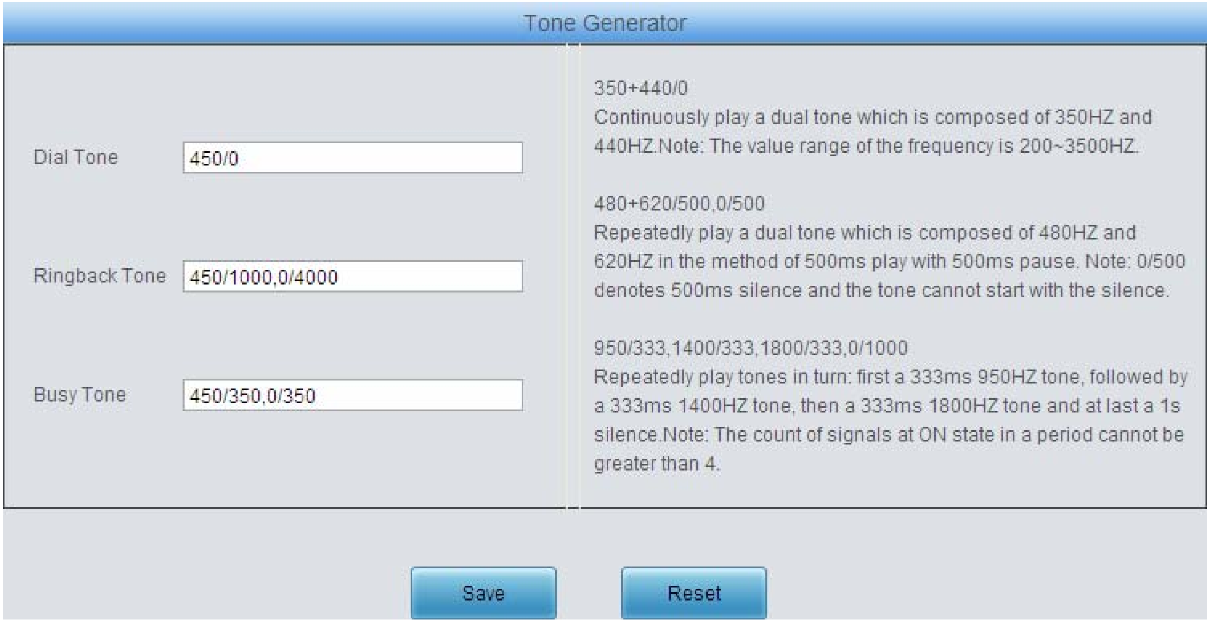


Figure 3-33 Tone Generator Setting Interface

See Figure 3-33 for the Tone Generator Setting interface. By default, there are three tones on it: Dial Tone—a continuous single tone with 450HZ frequency; Ringback Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 1s play and 4s pause; Busy Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 350ms play and 350ms pause. You can configure the tone generator manually. The exact explanation about the format and the meaning is described on the right of the interface.

3.5.5 DTMF

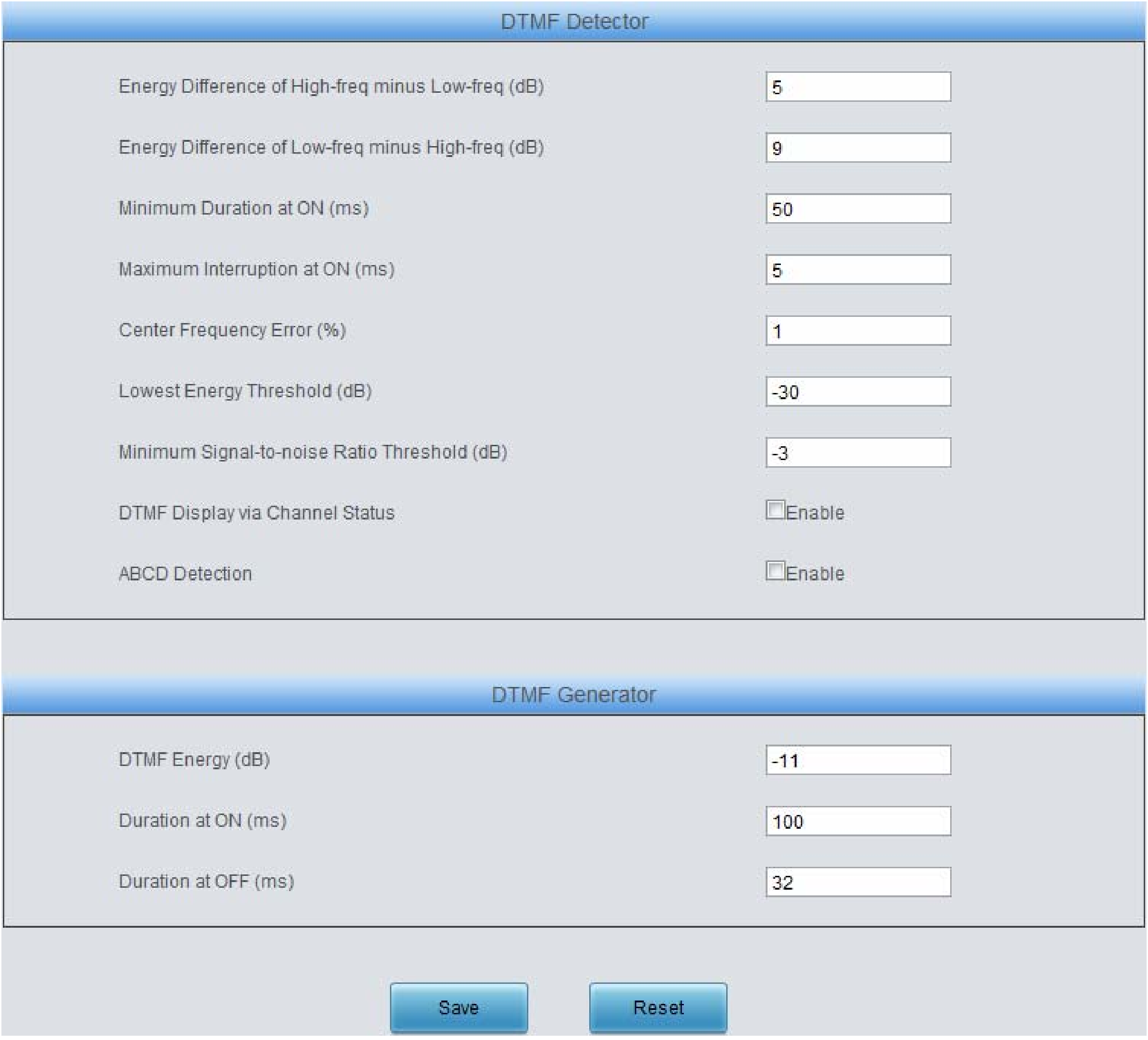


Figure 3-34 DTMF Detector Configuration Interface

See Figure 3-34 for the DTMF detector configuration. The table below explains the items shown in the above figure.

Item	Description
Energy Difference of High-freq minus Low-freq	The allowed difference in dB for the DTMF high frequency energy level to surpass the low frequency energy level. Range of value: 0~24. The default value is 5.
Energy Difference of Low-freq minus High -freq	The allowed difference in dB for the DTMF low frequency energy level to surpass the high frequency energy level. Range of value: 0~24. The default value is 9.
Minimum Duration at ON	The shortest time that a valid tone has to last at ON state. Range of value: 10～2000, calculated by ms. The default value is 50.
Maximum Interruption at ON	The longest time for a valid tone to stay interrupted at ON state. Range of value: 0～20, calculated by ms. The default value is 5.

Center Frequency Error	The error threshold of the center frequency at ON state in the DTMF tone, with the default value of 1.
Lowest Energy Threshold	The energy threshold to trigger the DTMF detection. Range of value: -40～-9, calculated by dB. The default value is -30.
Minimum Signal-to-noise Ratio Threshold	The signal-to-noise ratio threshold to trigger the DTMF detection. Range of value: -9～0, calculated by dB. The default value is -3.
DTMF Display via Channel Status	Once this feature is enabled, the received/transmitted DTMF will be displayed as you put the mouse on the icon of channel status.
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: -35~15, calculated by dB, with the default value of -11.
Duration at ON	The duration of DTMF signal at on state. Range of value: 0~16383, calculated by ms, with the default value of 100.
Duration at OFF	The duration of 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 or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to 3.9.15 Restart for detailed instructions.

3.5.6 Ringing Scheme

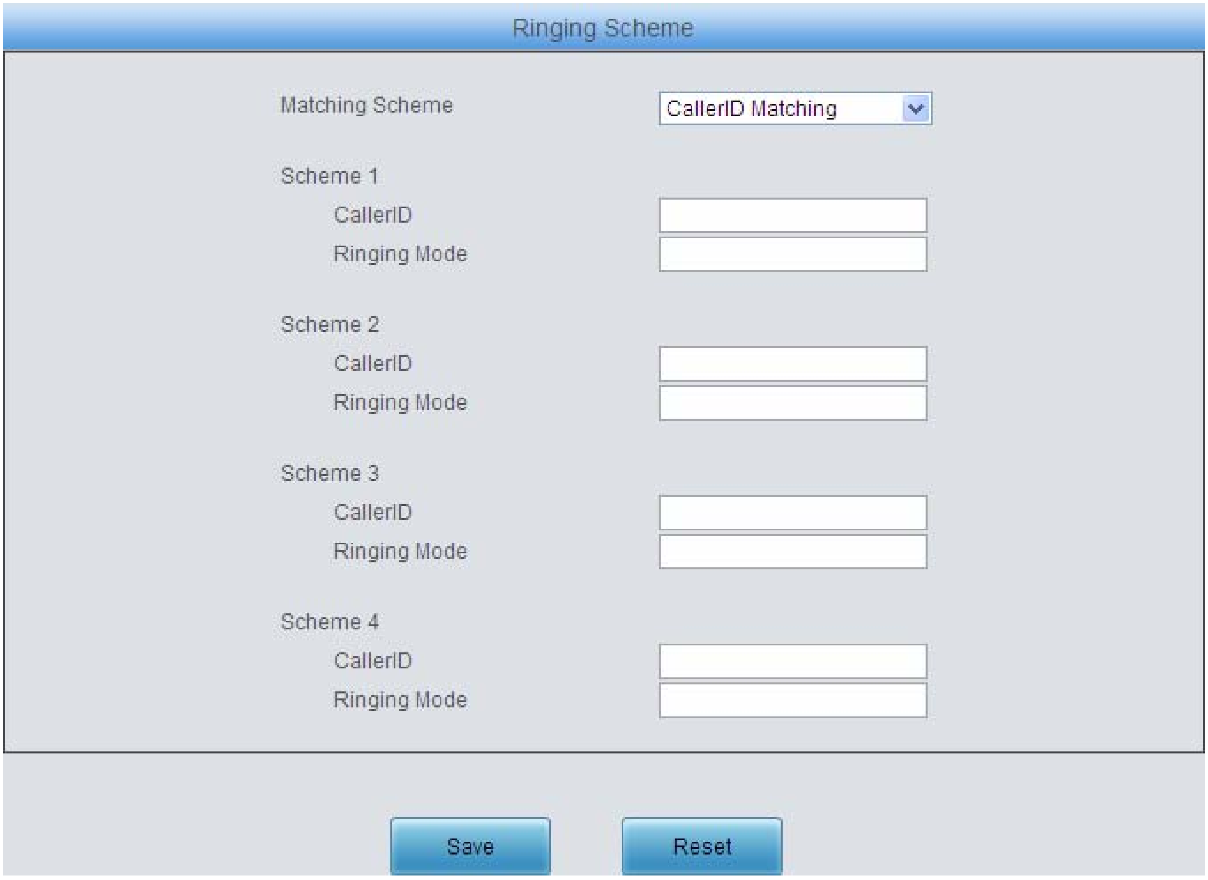


Figure 3-35 Ringing Scheme Configuration Interface

See Figure 3-35 for the Ringing Scheme interface. The gateway will execute different ringing schemes according to the callerID or Alter-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 ","
Alter-Info Value	The gateway will match the alter-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

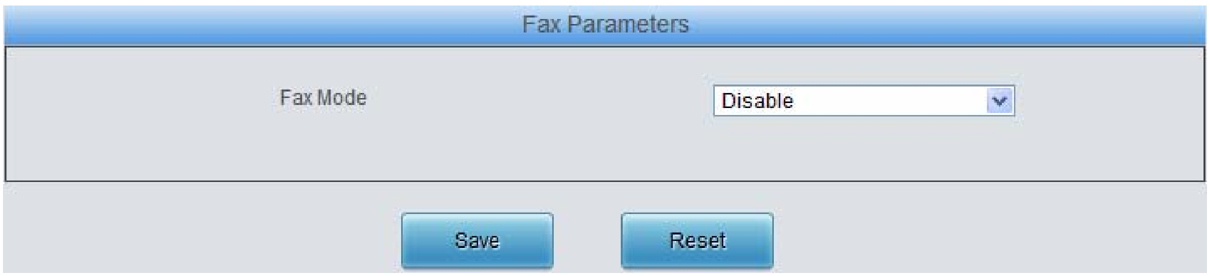


Figure 3-36 Fax Configuration Interface (Disable by default)

See Figure 3-36 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 3-37 for the fax configuration under the T.38 mode.

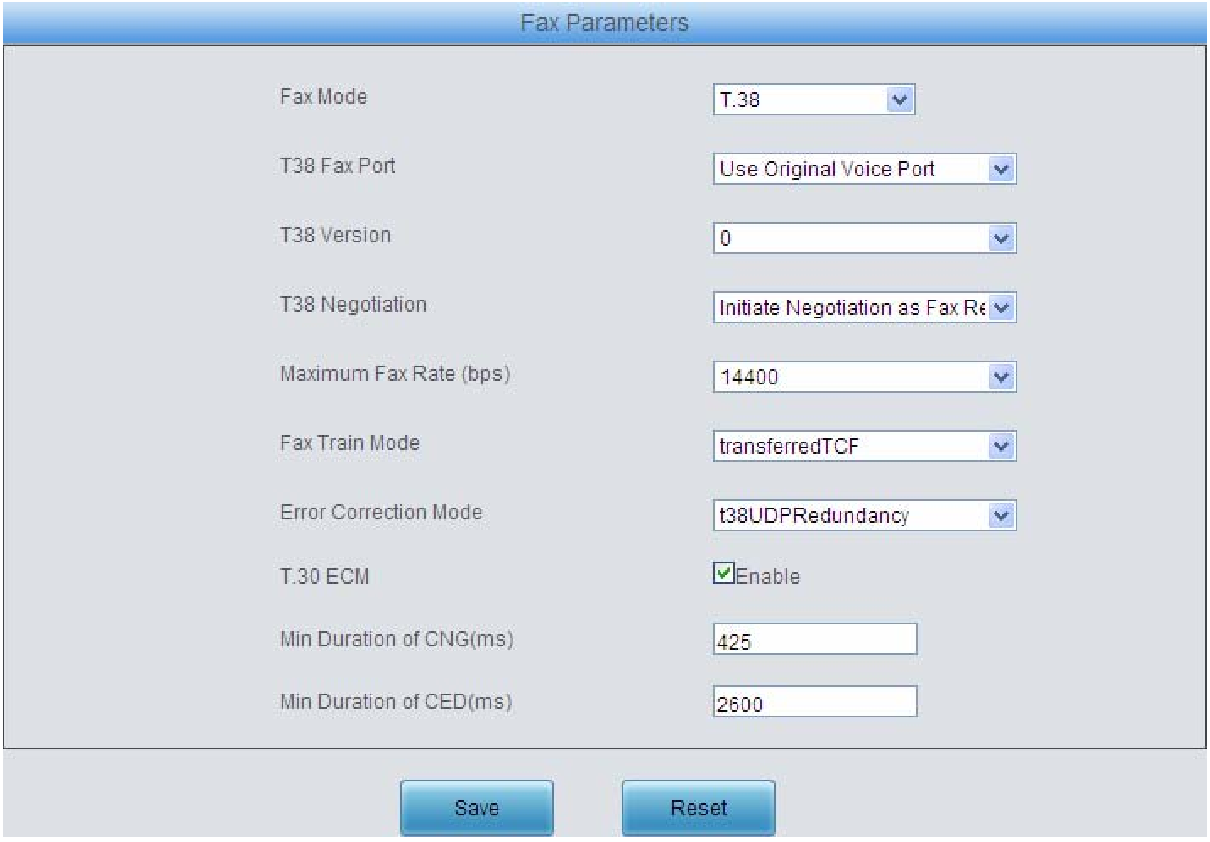


Figure 3-37 Fax Configuration Interface (T.38 Mode)

Users can configure the general fax parameters via this interface. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to 3.9.15 Restart for detailed instructions. The table below explains the configuration items in Figure 3-37.

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. Range of value: 0~3, with the default value of 0.
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*.
T.30 ECM	Sets whether to enable the T.30 error correction mode. By default this feature is enabled.

Min Duration of CNG	As stipulated in the standard FAX CNG, the minimum duration of CNG is 500ms ± 15%, calculated by ms, with the default value of 425. **Note: Usually there is no need to modify it; please contact our technicians if necessary.**
Min Duration of CED	As stipulated in the standard FAX CED, the minimum duration of CED is 2600~4000ms, calculated by ms, with the default value of 2600. **Note: Usually there is no need to modify it; please contact our technicians if necessary.**

If you set ***Fax Mode*** to *Pass-through*, you can see the interface shown as Figure 3-38.

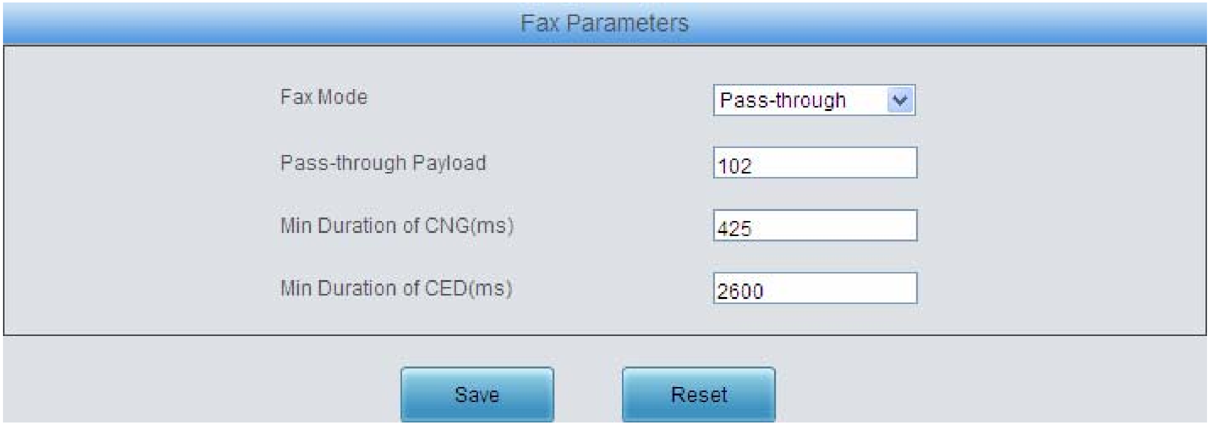


Figure 3-38 Fax Configuration Interface (Pass-through Mode)

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 3-39 for the function key configuration interface. Here you can set a cluster of combination keys to query a related number.

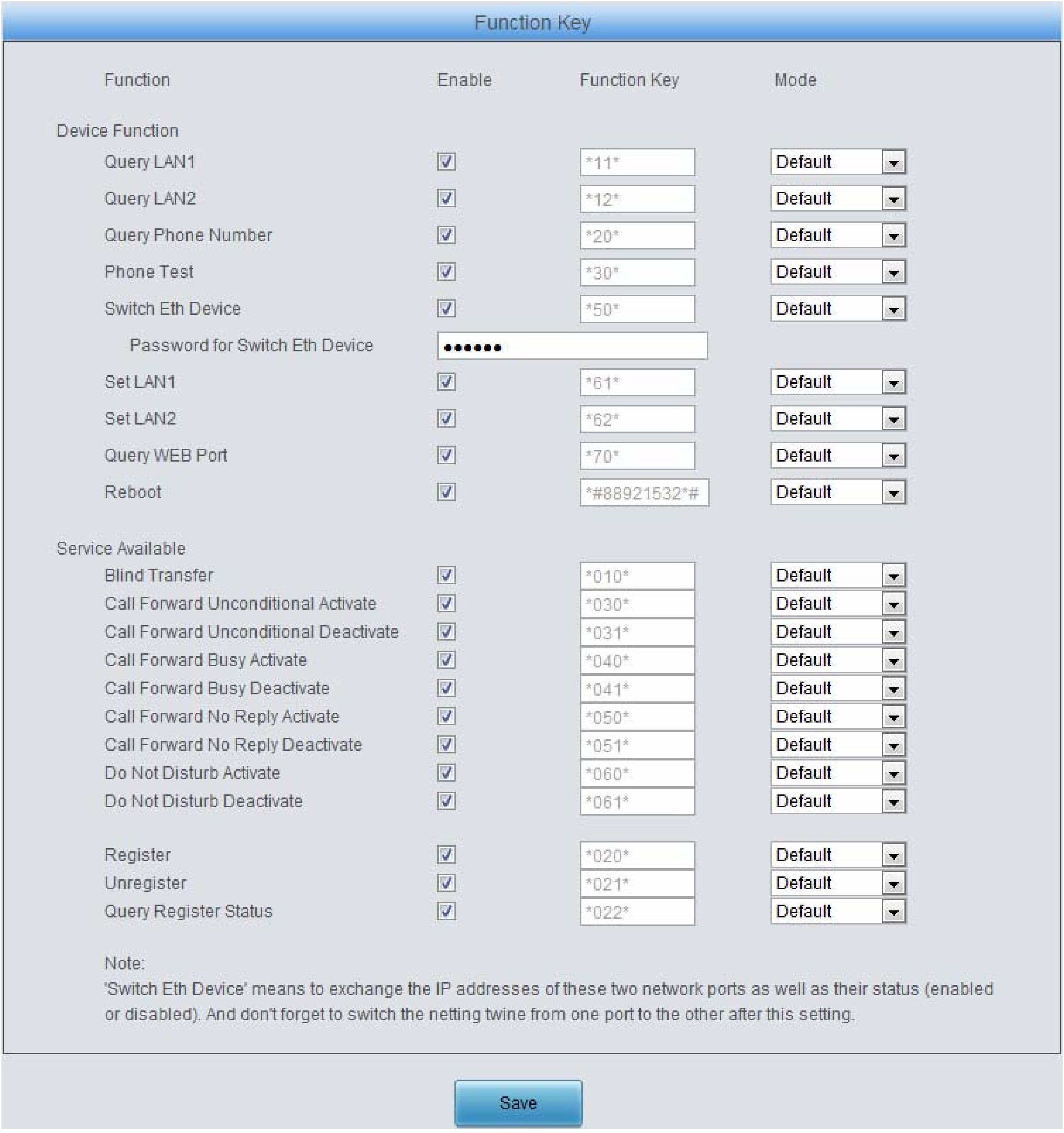


Figure 3-39 Function Key Configuration Interface

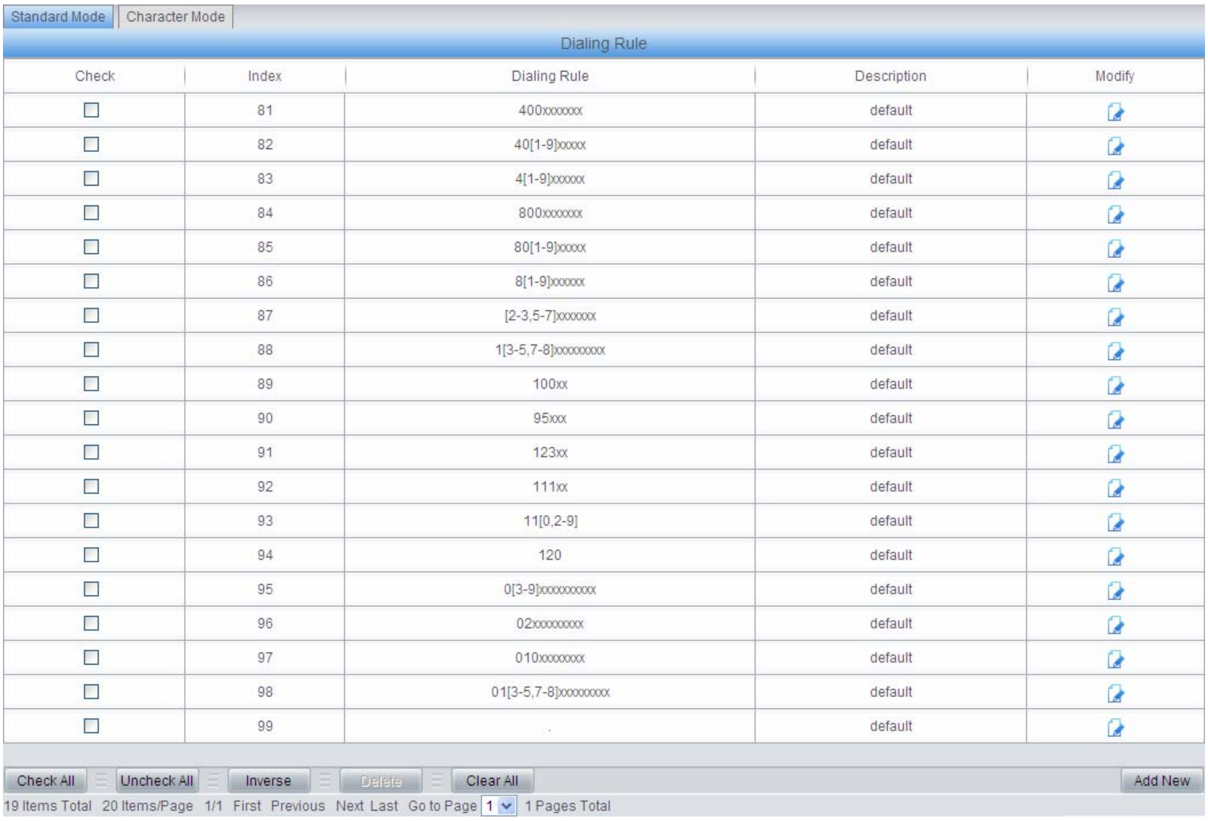
Click “Enable” to enable the corresponding function key. The gateway will use the default function keys when the mode is set to default; and it will allow you to set new function keys when the mode is set to user-defined. Click ***Save*** to save your settings into the gateway.

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.



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	

Figure 3-40 Dialing Rule Configuration Interface (Standard)

See Figure 3-40 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 3-41 for the dialing rule adding interface.

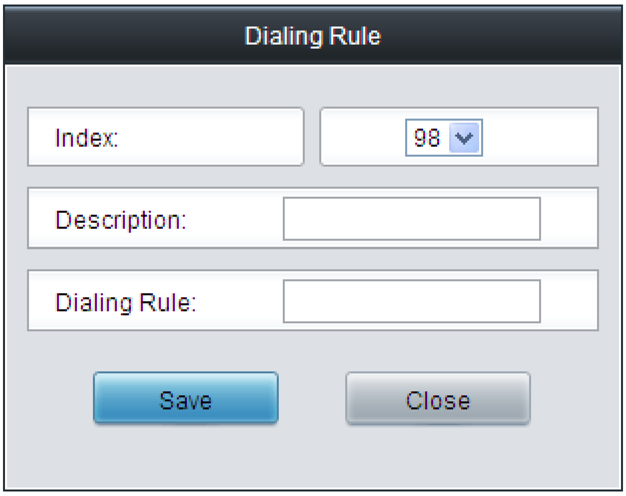


Figure 3-41 Add New Dialing Rule

The table below explains the items shown in Figure 3-41.

<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>
<tr><td>99</td><td>.</td><td>Any number in any length.</td></tr>
<tr><td>98</td><td>01[3-5,7-8]xxxxxxxxx.</td><td>Any 12-digit number starting with 013, 014, 015, 017 or 018</td></tr>
<tr><td>97</td><td>010xxxxxxxx</td><td>Any 11-digit number starting with 010</td></tr>
<tr><td>96</td><td>02xxxxxxxxx</td><td>Any 11-digit number starting with 02</td></tr>
<tr><td>95</td><td>0[3-9]xxxxxxxxxx</td><td>Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09</td></tr>
<tr><td>94</td><td>120</td><td>Number 120。</td></tr>
<tr><td>93</td><td>11[0,2-9]</td><td>Number 110, 112, 113, 114, 115, 116, 117, 118 or 119</td></tr>
<tr><td>92</td><td>111xx</td><td>Any 5-digit number starting with 111</td></tr>
<tr><td>91</td><td>123xx</td><td>Any 5-digit number starting with 123</td></tr>
</table>
</td></tr>
</table>

	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 3-40 to modify the dialing rules. See Figure 3-42 for the dialing rule modification interface. The configuration items on this interface are the same as those on the ***Add New Dialing Rule*** interface.

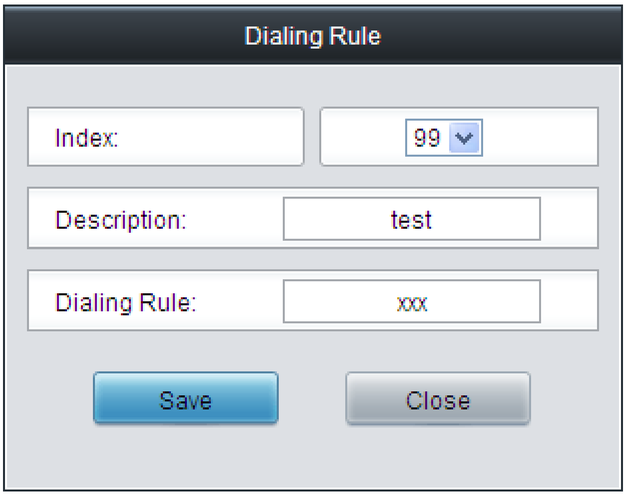


Figure 3-42 Modify Dialing Rule

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

See Figure 3-43 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.

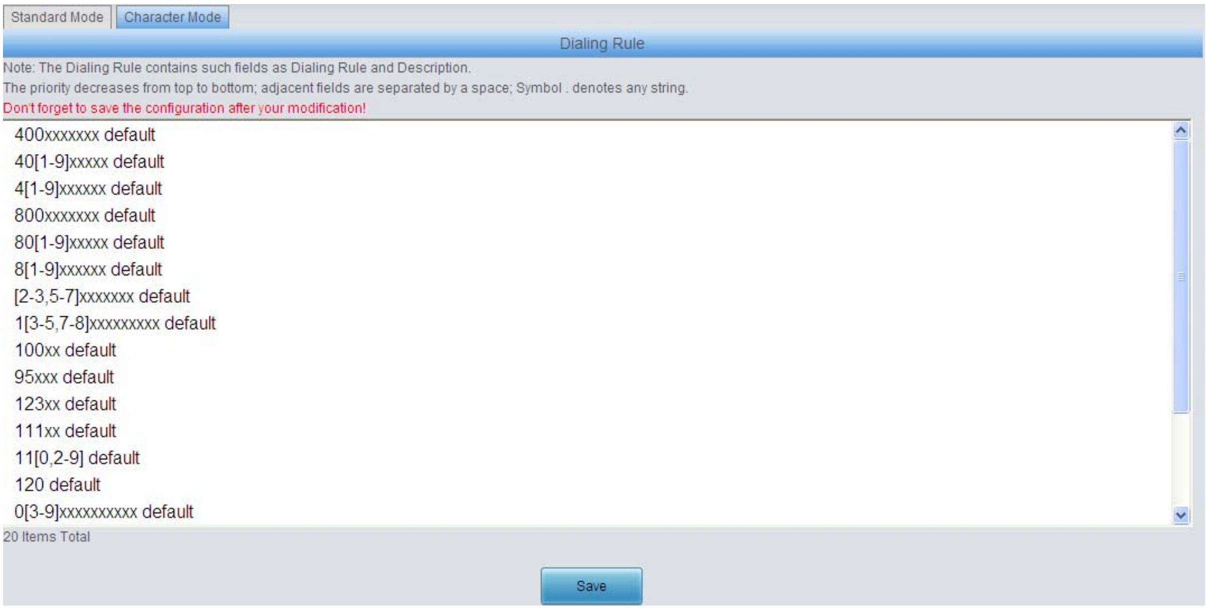


Figure 3-43 Dialing Rule Configuration Interface (Character)

3.5.10 Dialing Timeout

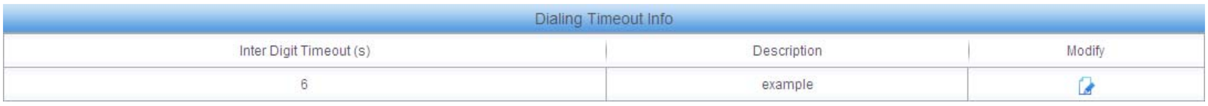


Figure 3-44 Dialing Timeout Info Interface

See Figure 3-44 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 3-44 to modify the dialing timeout info. See Figure 3-45 for the dialing timeout info modification interface. The configuration items on this interface are the same as those on the ***Dialing Timeout Info Interface***.

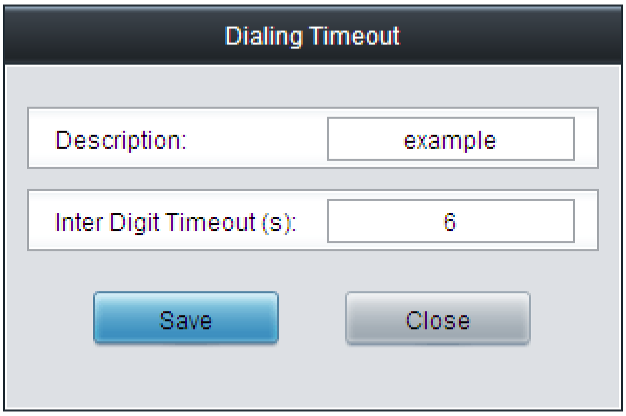


Figure 3-45 Modify Dialing Timeout Info

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

3.5.11 Cue Tone

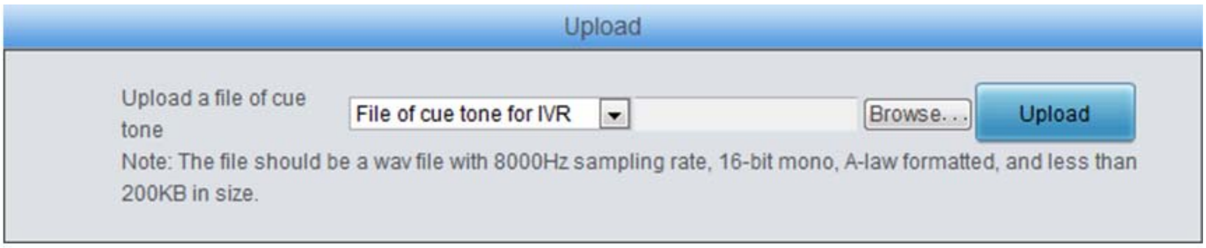


Figure 3-46 Cue Tone Interface

See Figure 3-46 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, including two options: *Cue Tone for IVR* and *Cue Tone for Call Waiting*.

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

3.5.12 Color Ring

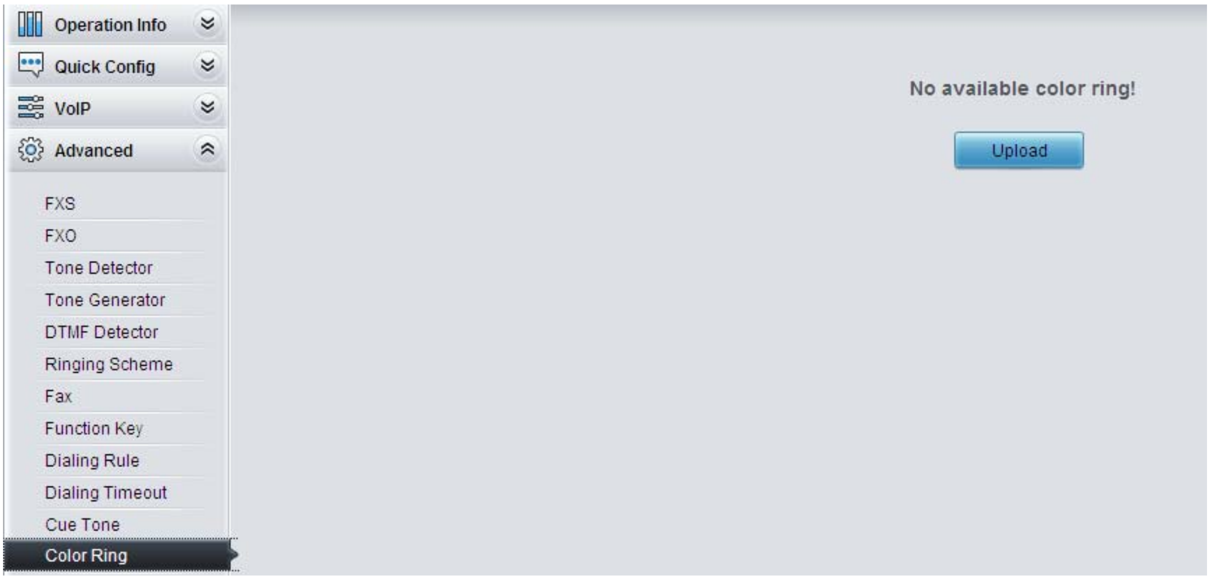


Figure 3-47 Color Ring Interface

By default, there is no available color ring on the gateway. See Figure 3-47. Click ***Upload*** to upload a new color ring manually. See Figure 3-48. You can upload the required color ring file to the gateway following this interface.

Figure 3-48 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. After upload, the color ring will appear on the color ring manage interface, see Figure 3-49.

Figure 3-49 Color Ring Manage Interface

Click ***Modify*** in Figure 3-49 to modify the configuration of the color ring or tick the ***Upload*** checkbox to change the old color ring file. 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.

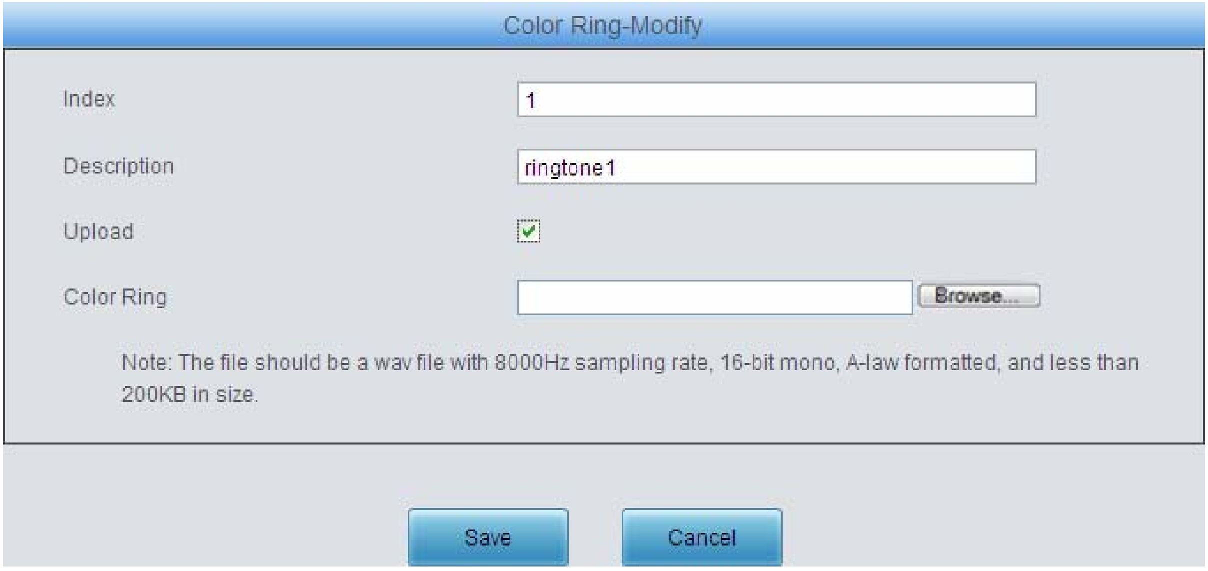


Figure 3-50 Color Ring Modification Interface

To delete a color ring, check the checkbox before the corresponding index in Figure 3-49 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 3-50.

3.5.13 QoS

Figure 3-51 Differentiated Services Setting Interface

See Figure 3-51 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

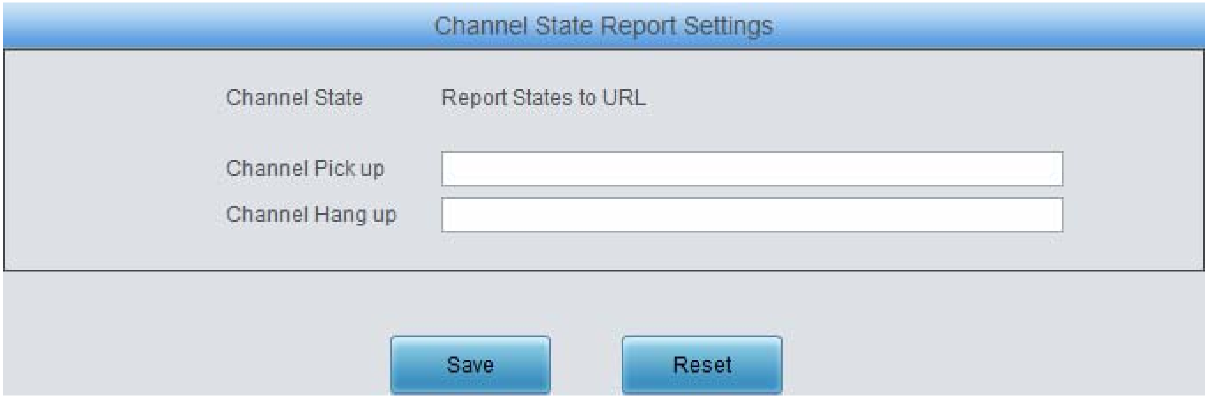


Figure 3-52 Channel State Report Settings Interface

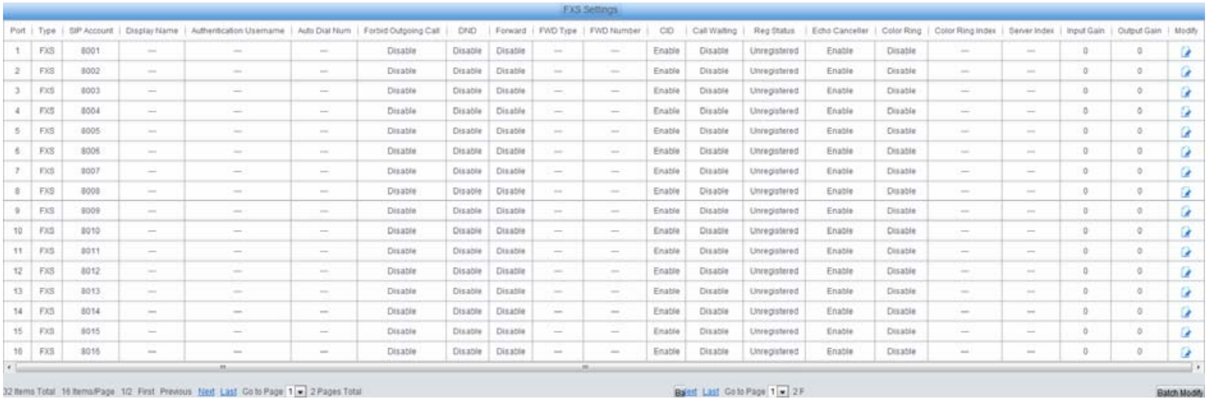
See Figure 3-52 for the Action URL interface, which is used to designate the server patch to report the on-hook or off-hook state of the FXS channel. You are allowed to designate two different server paths. After setting, the state will be reported to the designated server once any of the FXS channel 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 3-53.

Figure 3-53 Port Settings

3.6.1 FXS



FXS Settings

Port	Type	SIP Account	Display Name	Authentication Username	Auto Dial Num	Forbid Outgoing Call	DND	Forward	FWD Type	FWD Number	CID	Call Waiting	Reg Status	Echo Canceller	Color Ring	Color Ring Index	Server 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	
9	FXS	8009	—	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	—	0	0	
10	FXS	8010	—	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	—	0	0	
11	FXS	8011	—	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	—	0	0	
12	FXS	8012	—	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	—	0	0	
13	FXS	8013	—	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	—	0	0	
14	FXS	8014	—	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	—	0	0	
15	FXS	8015	—	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	—	0	0	
16	FXS	8016	—	—	—	Disable	Disable	Disable	—	—	Enable	Disable	Unregistered	Enable	Disable	—	—	0	0	

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

Figure 3-54 FXS Settings Interface

See Figure 3-54 for the FXS settings interface. The list in the above figure shows the feature and properties of each FXS port. Click ***Modify*** in Figure 3-54 to modify the properties of the corresponding port. See Figure 3-55 for the FXS modification interface.

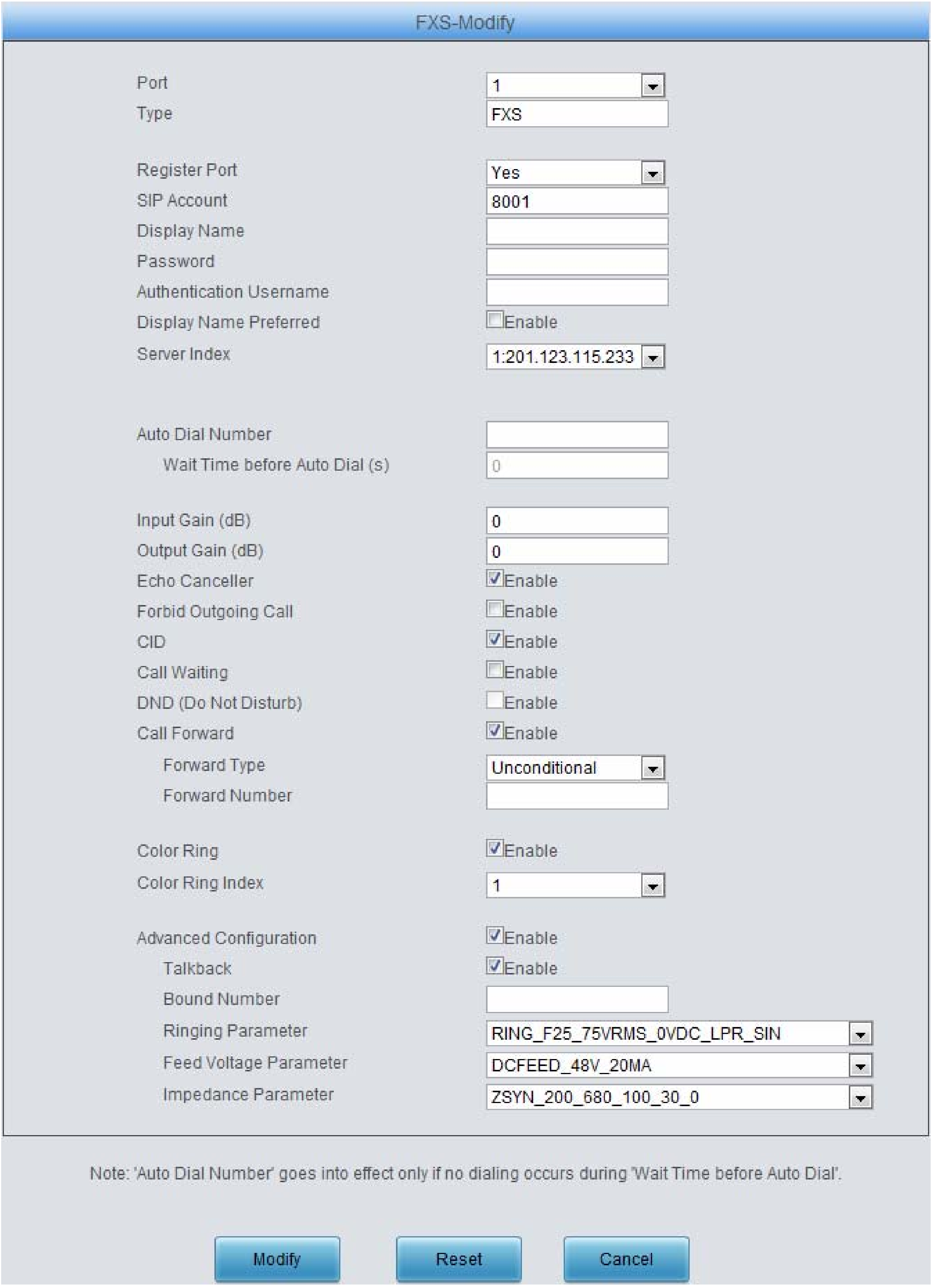


Figure 3-55 FXS Modification

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 3-54) 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 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.
Authentication Username	Authentication username of a port, used to register the port to the SIP server when IMS network is enabled. **Note: This item appears only when IMS Network is enabled.**
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. The value must be multiples of 3. Range of value: -24~24, calculated by dB, with the default value of 0.
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 appear this item.</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>
<tr><td>Ringing Parameter</td><td>Sets the ringing parameter for the FXS module. The default value is RING_F25_75VRMS_0VDC_LPR_SIN
Note: Usually there is no need to modify it; please contact our technicians if necessary.</td></tr>
<tr><td>Feed Voltage Parameter</td><td>Sets the feed voltage parameter for the FXS module. The default value is DCFEED_48V_20MA.
Note: Usually there is no need to modify it; please contact our technicians if necessary.</td></tr>
<tr><td>Impedance Parameter</td><td>Sets the impedance for the FXS module. The default value is ZSYN_200_680_100_30_0.
Note: Usually there is no need to modify it; please contact our technicians if necessary.</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

3-56 below for the FXS batch modification interface. The configuration items on this interface are the same as those on the FXS modification interface (Figure 3-55).

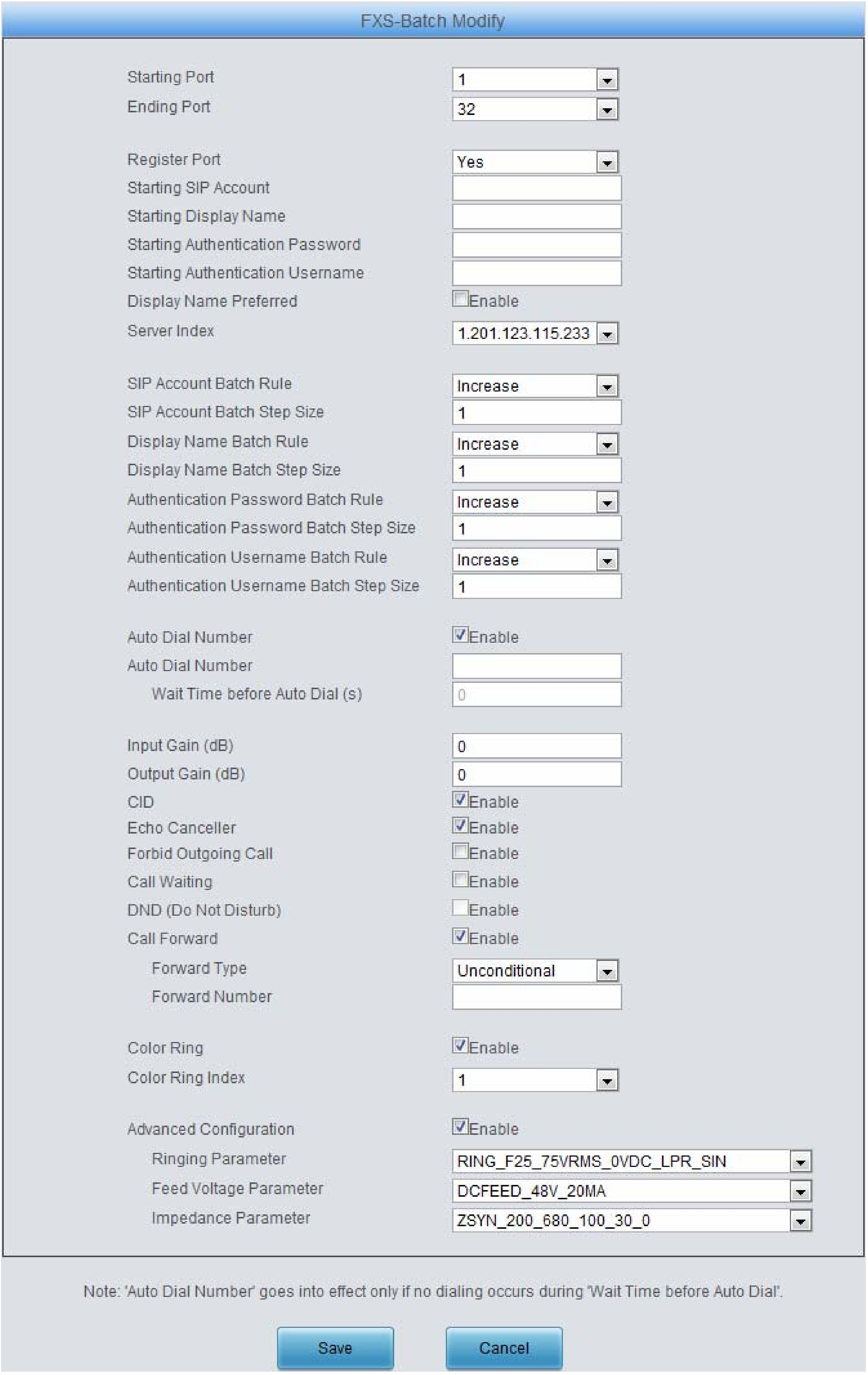

FXS-Batch Modify
Starting Port 1
Ending Port 32
Register Port Yes
Starting SIP Account
Starting Display Name
Starting Authentication Password
Starting Authentication Username
Display Name Preferred Enable
Server Index 1.201.123.115.233
SIP Account Batch Rule Increase
SIP Account Batch Step Size 1
Display Name Batch Rule Increase
Display Name Batch Step Size 1
Authentication Password Batch Rule Increase
Authentication Password Batch Step Size 1
Authentication Username Batch Rule Increase
Authentication Username Batch Step Size 1
Auto Dial Number Enable
Auto Dial Number
Wait Time before Auto Dial (s) 0
Input Gain (dB) 0
Output Gain (dB) 0
CID Enable
Echo Canceller Enable
Forbid Outgoing Call Enable
Call Waiting Enable
DND (Do Not Disturb) Enable
Call Forward Enable
Forward Type Unconditional
Forward Number
Color Ring Enable
Color Ring Index 1
Advanced Configuration Enable
Ringing Parameter RING_F25_75VRMS_0VDC_LPR_SIN
Feed Voltage Parameter DCFEED_48V_20MA
Impedance Parameter ZSYN_200_680_100_30_0
Note: 'Auto Dial Number' goes into effect only if no dialing occurs during 'Wait Time before Auto Dial'.
Save
Cancel

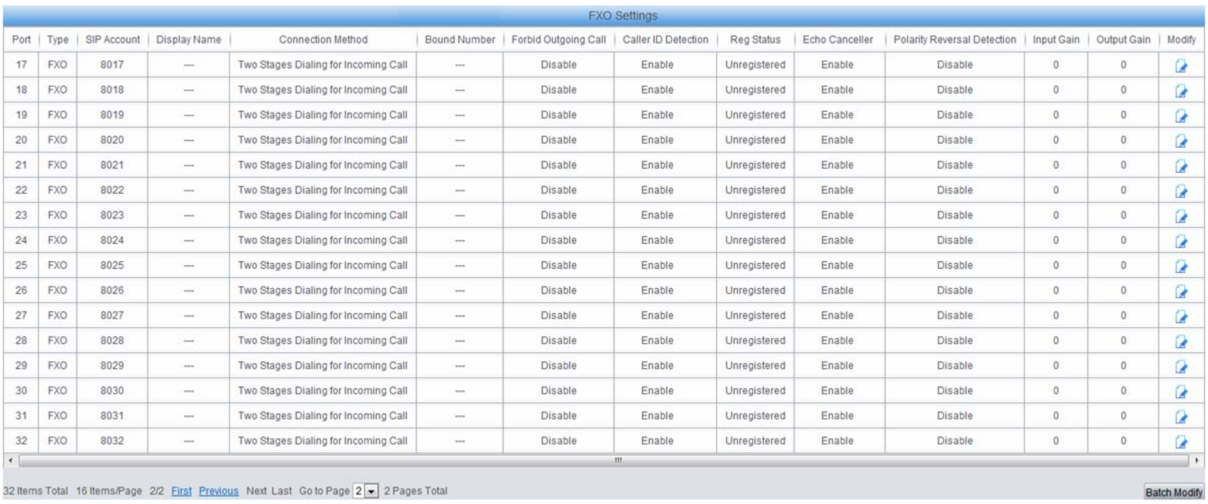
Figure 3-56 FXS Batch Modification

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


FXO Settings

Port	Type	SIP Account	Display Name	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	Polarity Reversal Detection	Input Gain	Output Gain	Modify
17	FXO	8017	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
18	FXO	8018	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
19	FXO	8019	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
20	FXO	8020	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
21	FXO	8021	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
22	FXO	8022	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
23	FXO	8023	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
24	FXO	8024	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
25	FXO	8025	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
26	FXO	8026	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
27	FXO	8027	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
28	FXO	8028	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
29	FXO	8029	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
30	FXO	8030	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
31	FXO	8031	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	
32	FXO	8032	---	Two Stages Dialing for Incoming Call	---	Disable	Enable	Unregistered	Enable	Disable	0	0	

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

Batch Modify

Figure 3-57 FXO Settings Interface

See Figure 3-57 for the FXO settings interface. The list in the above figure shows the feature and properties of each FXO port. Click ***Modify*** in Figure 3-57 to modify the properties of the corresponding port. See Figure 3-58 for the FXO modification interface.

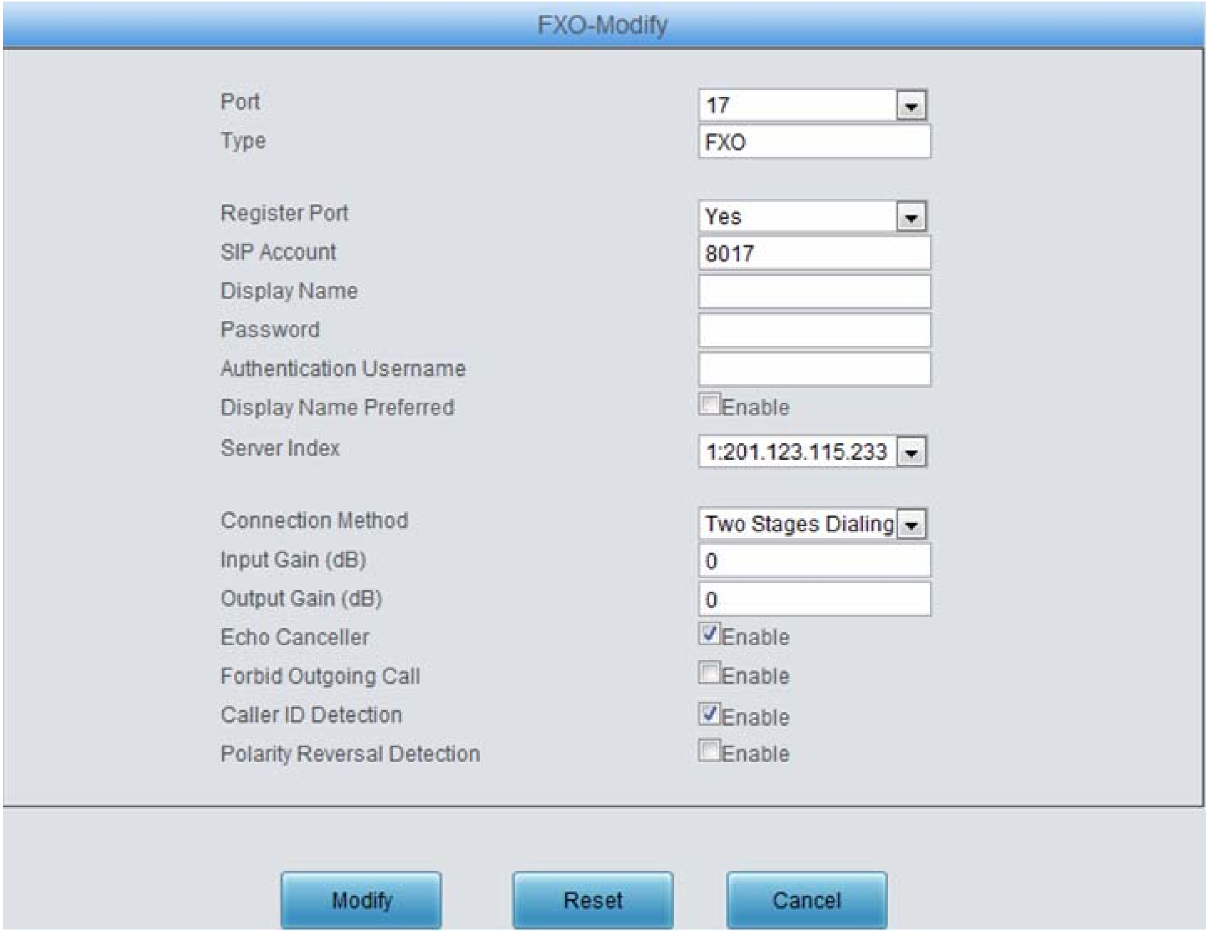


Figure 3-58 FXO Modification

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 3-57) 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.
Authentication Username	Authentication username of a port, used to register the port to the SIP server when IMS network is enabled. **Note: This item appears only when IMS Network is enabled.**

<table>
<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, Output Gain</td><td>Adjusts the gain of the voice input to/ output from the FXO port. The value must be multiples of 3. Range of value: -24~24, 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 3-59 below for the FXO batch modification interface. The configuration items on this interface are the same as those on the FXO modification interface (Figure 3-58).

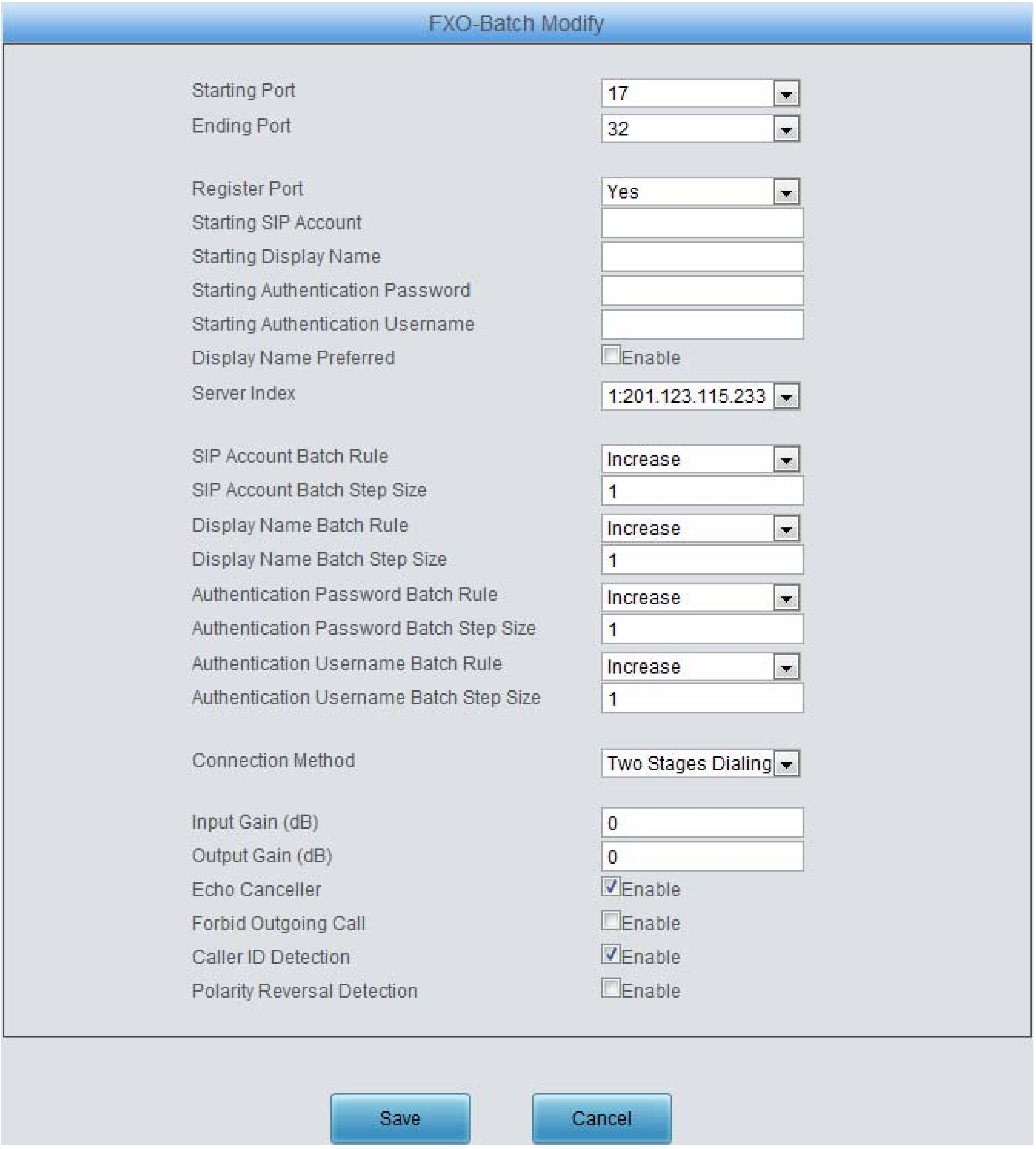


Figure 3-59 FXO Batch Modification

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

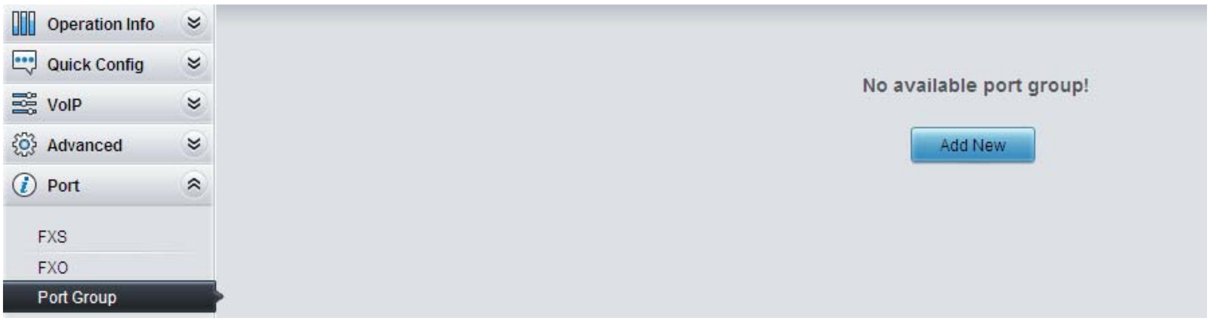


Figure 3-60 Port Group Setting Interface

See Figure 3-60 for the Port Group Settings interface. By default, there is no available port group on the gateway. A port group is a set containing one or more than one port, having such properties as ***Port Selection*** and ***Authentication Mode*** the same 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 3-61 for the port group adding interface. Note that a port which has been occupied by one port group cannot be chosen by others.

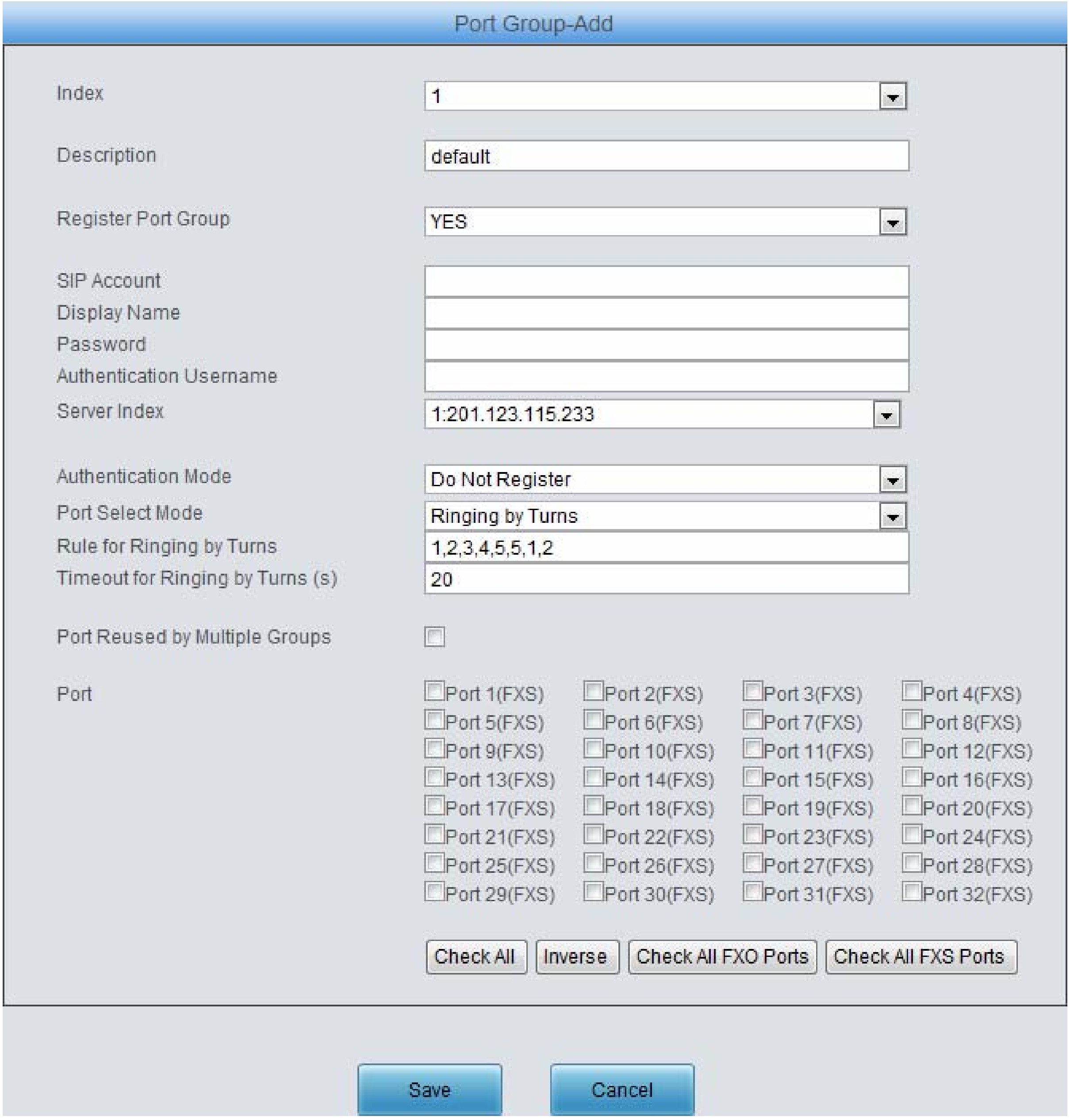


Figure 3-61 Add New Port Group

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.

<table>
<tr><td>Password</td><td>Registration password of the port group. To register the port group to the SIP server, both configuration items SIP Account and Password should be filled in.</td></tr>
<tr><td>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 is enabled.</td></tr>
<tr><td>Server Index</td><td>The index of the SIP server which will be quoted by the current port group.</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 3-61. 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 3-62. 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. See Figure 3-62 for the port group list with saved configurations.

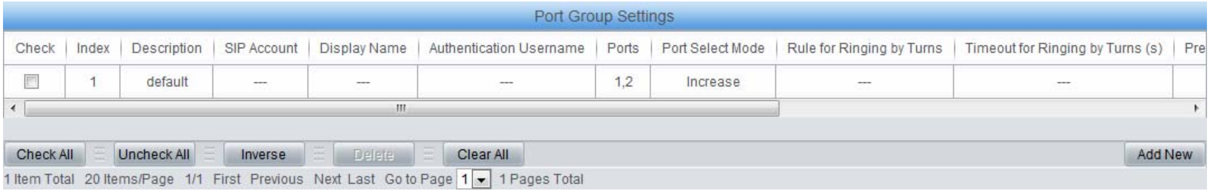


Figure 3-62 Port Group List

Click ***Modify*** at the end of the list in Figure 3-62 to modify the properties of a port group. See Figure 3-63 for the port group modification interface. The configuration items on this interface are the same as those on the ***Add New Port Group*** interface.

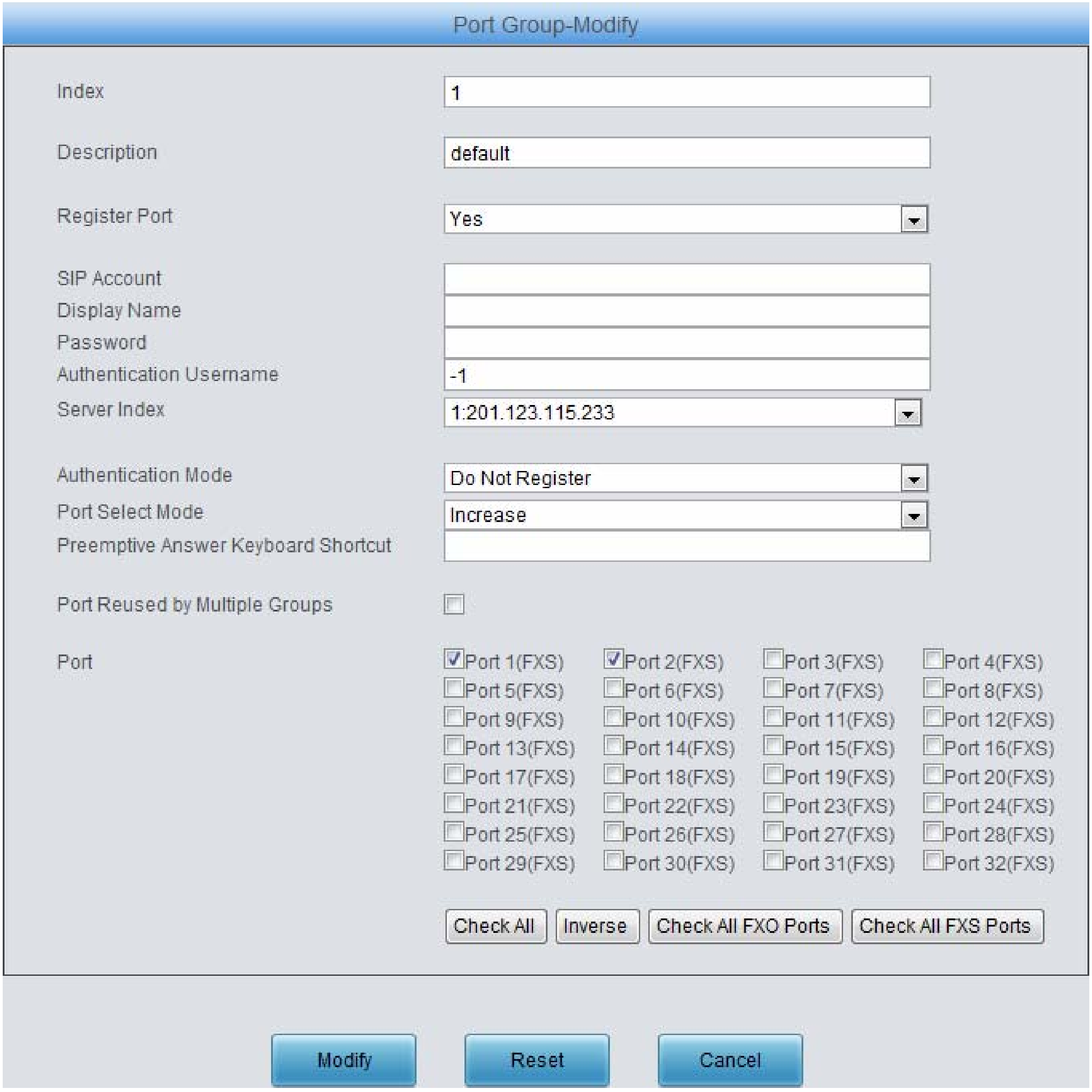


Figure 3-63 Modify Port Group

To delete a port group, check the checkbox before the corresponding index in Figure 3-62 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 3-62.

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 3-64.

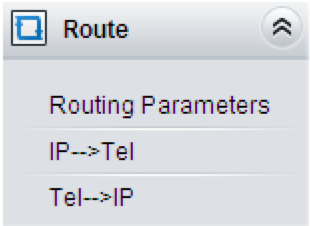


Figure 3-64 Route Settings

3.7.1 Routing Parameters

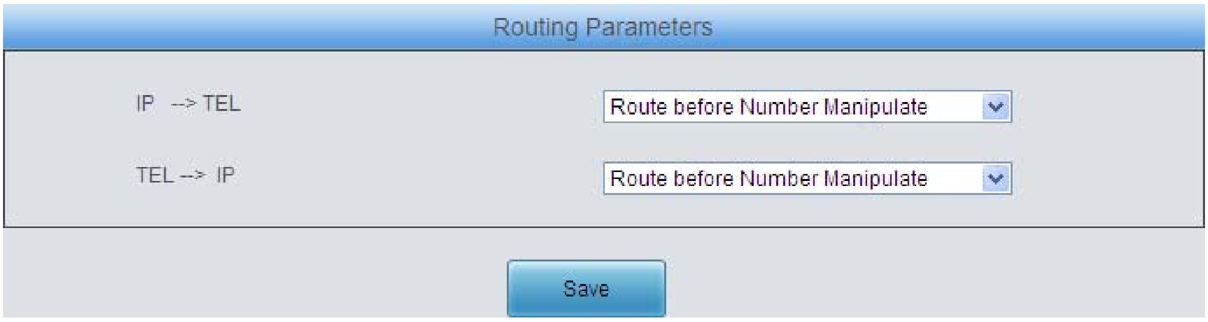


Figure 3-65 Routing Parameters Configuration Interface

See Figure 3-65 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

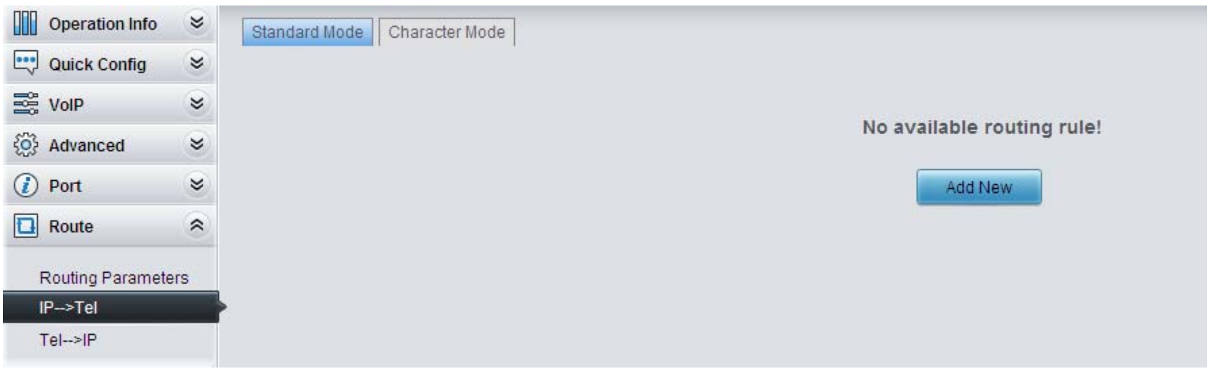


Figure 3-66 IP→Tel Routing Rule Configuration Interface (Standard)

See Figure 3-66 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 3-67. You may use the default values of all the configuration items herein.

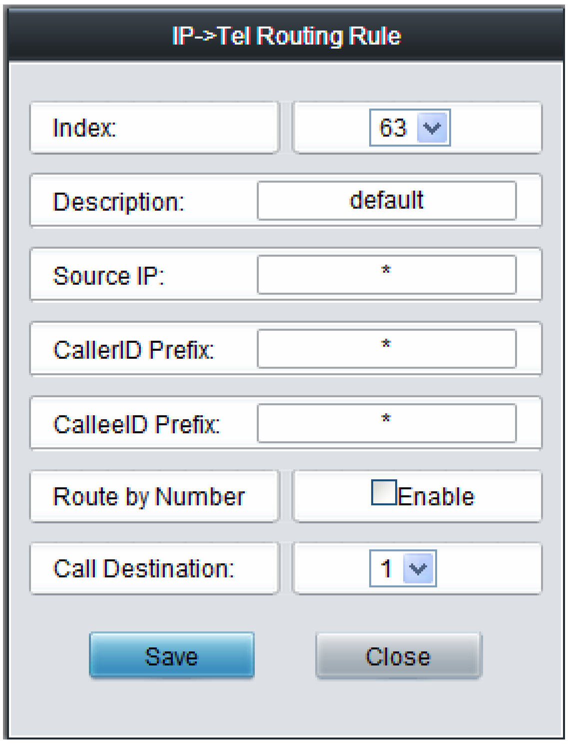


Figure 3-67 Add New Routing Rule (IP→Tel)

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 3-68 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.

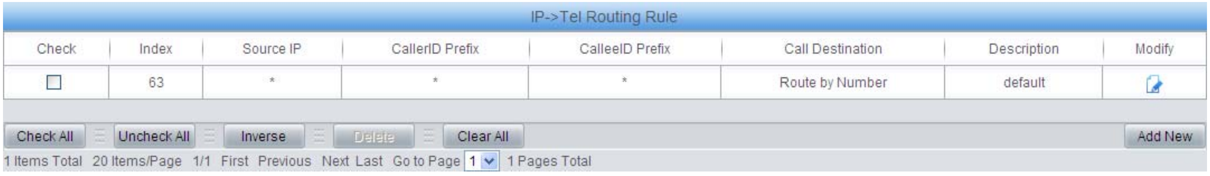


Figure 3-68 IP→Tel Routing Rule Configuration Interface

Click ***Modify*** in Figure 3-68 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 3-68 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 3-68.

See Figure 3-69 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.

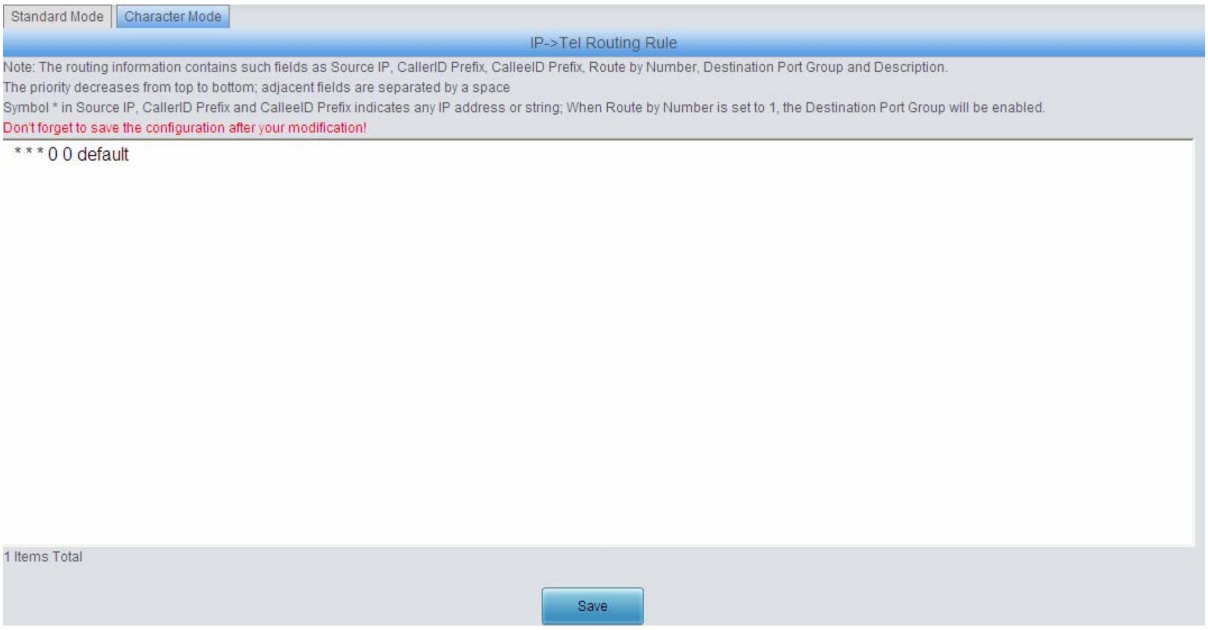


Figure 3-69 IP→Tel Routing Rule Configuration Interface (Character)

3.7.3 Tel to IP

Figure 3-70 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-70 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 3-71. You may use the default values of all the configuration items herein except for ***Destination IP*** and ***Destination Port***.

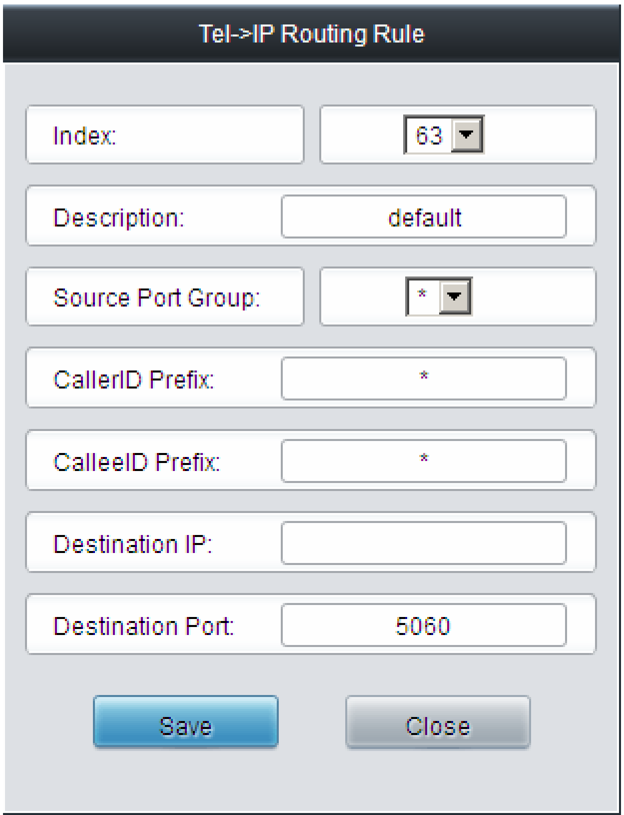


Figure 3-71 Add New Routing Rule (Tel→IP)

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 3-72 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.

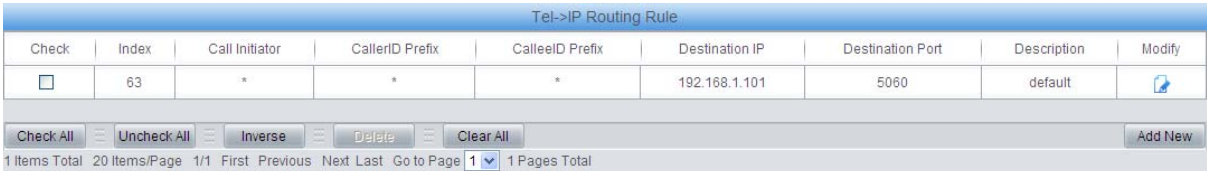


Figure 3-72 Tel→IP Routing Rule Configuration Interface

Click ***Modify*** in Figure 3-72 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 3-72 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 3-72.

See Figure 3-73 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.

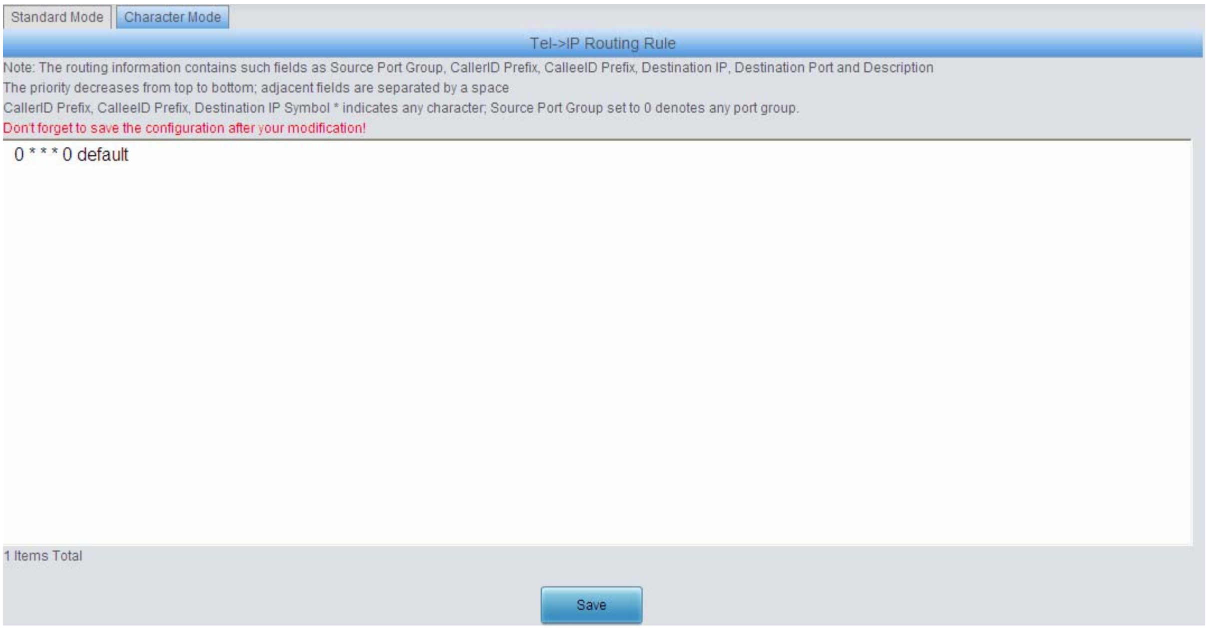


Figure 3-73 Tel→IP Routing Rule Configuration Interface (Character)

3.8 Number Manipulation

Number Manipulation includes four parts: **IP→Tel CallerID**, **IP→Tel CalleeID**, **Tel→IP CallerID** and **Tel→IP CalleeID**. See Figure 3-74.

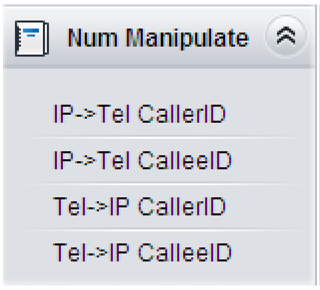


Figure 3-74 Number Manipulation

3.8.1 IP to Tel CallerID

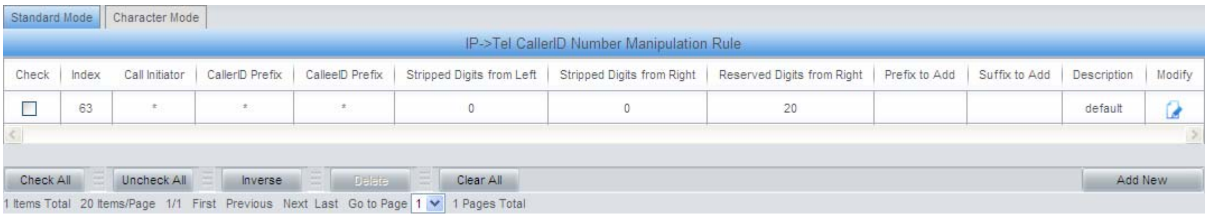


Figure 3-75 IP→Tel CallerID Manipulation Interface (Standard)

See Figure 3-75 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 3-76 for the IP→Tel CallerID manipulation rule adding interface. You may use the default values of all the configuration items herein.

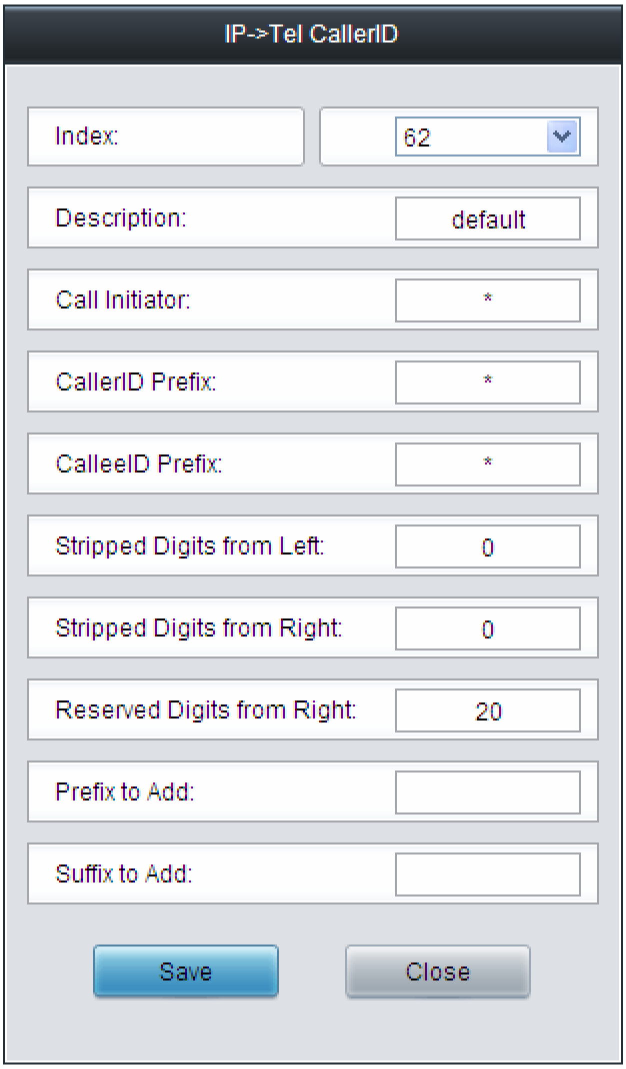


Figure 3-76 Add IP→Tel CallerID Manipulation Rule

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 3-75 to modify a number manipulation rule. See Figure 3-77 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.

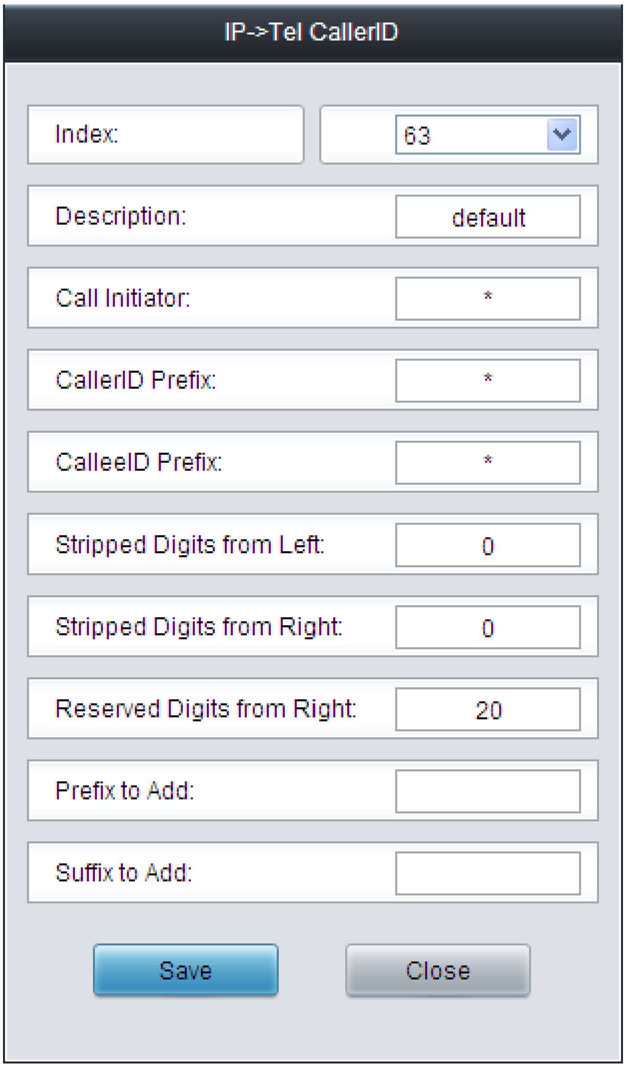


Figure 3-77 Modify IP→Tel CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-75 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 3-75.

See Figure 3-78 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.

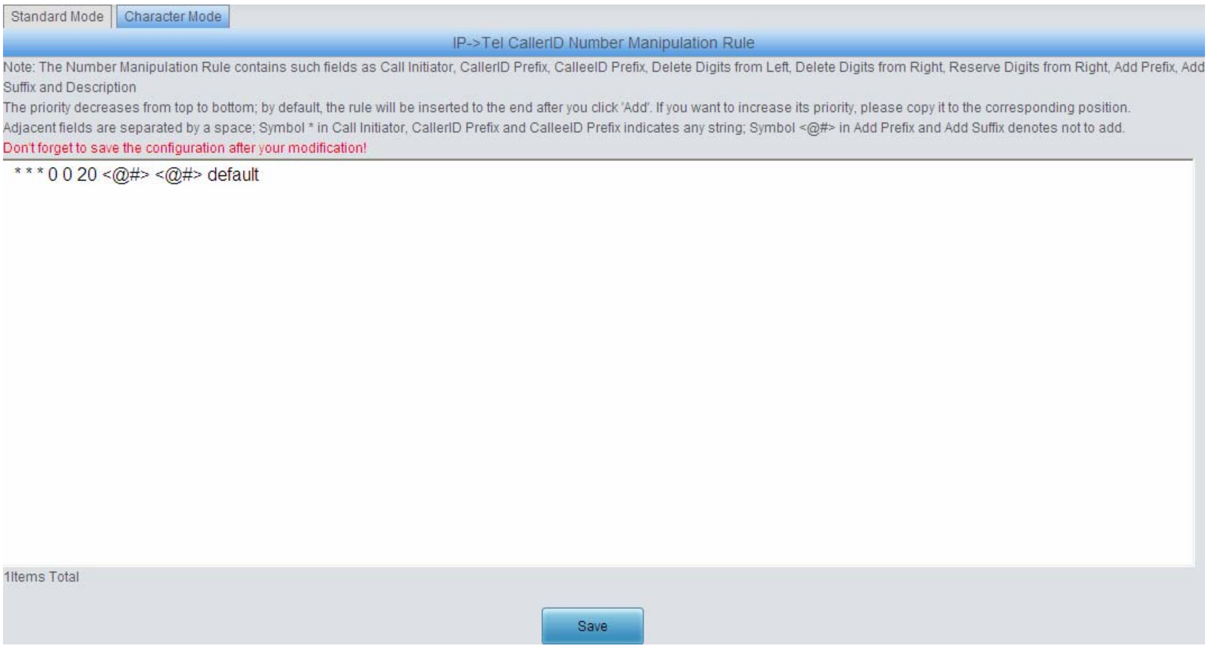


Figure 3-78 IP→Tel CallerID Manipulation Interface (Character)

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 3-80 for IP→Tel CalleeID manipulation interface. The configuration items on this interface are the same as those on ***IP→Tel CallerID Manipulation Interface*** (Figure 3-75).

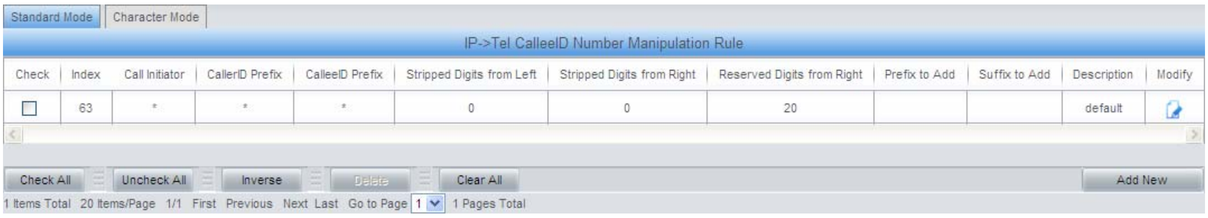


Figure 3-79 IP→Tel CalleeID Manipulation Interface(Standard)

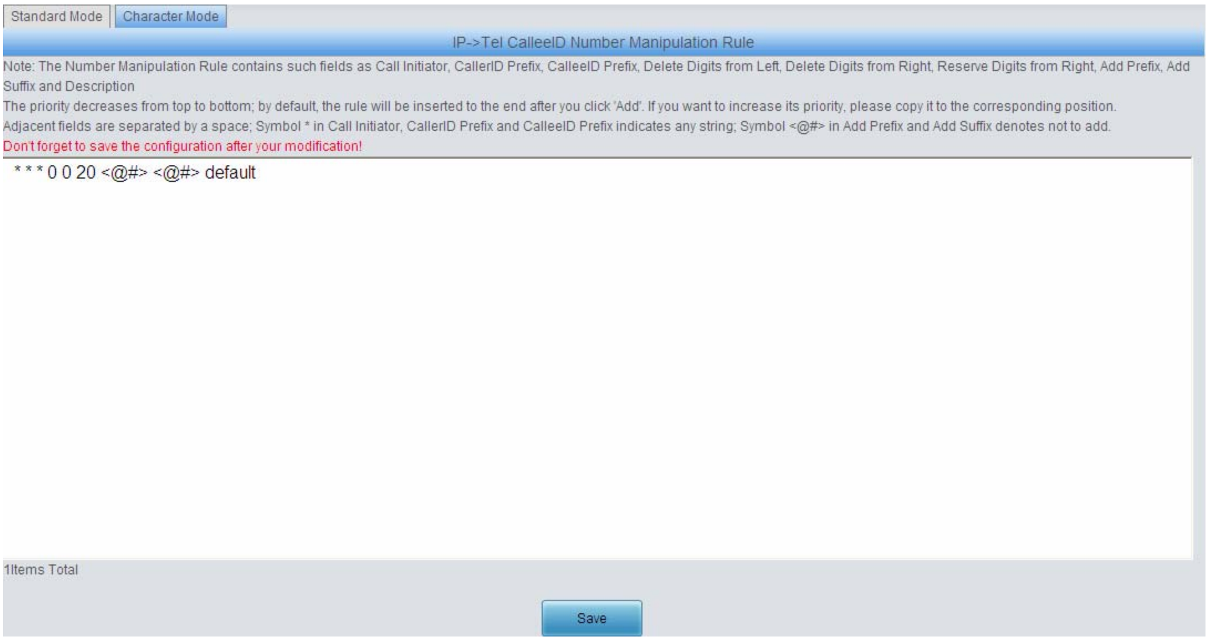


Figure 3-80 IP→Tel CalleeID Manipulation Interface (Character)

3.8.3 Tel to IP CallerID

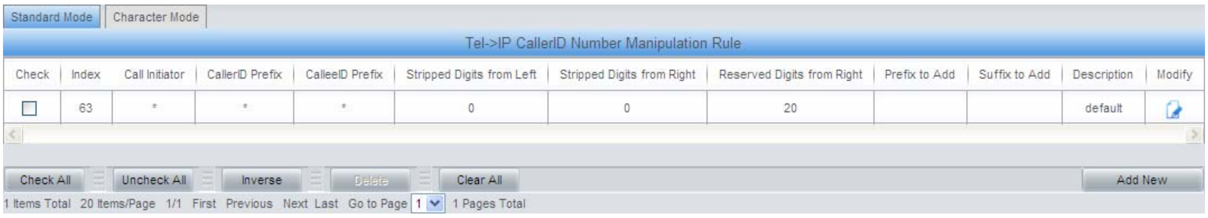


Figure 3-81 Tel→IP CallerID Manipulation Interface (Standard)

See Figure 3-81 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 3-82 for the Tel→IP CallerID manipulation rule adding interface. You may use the default values of all the other configuration items herein.

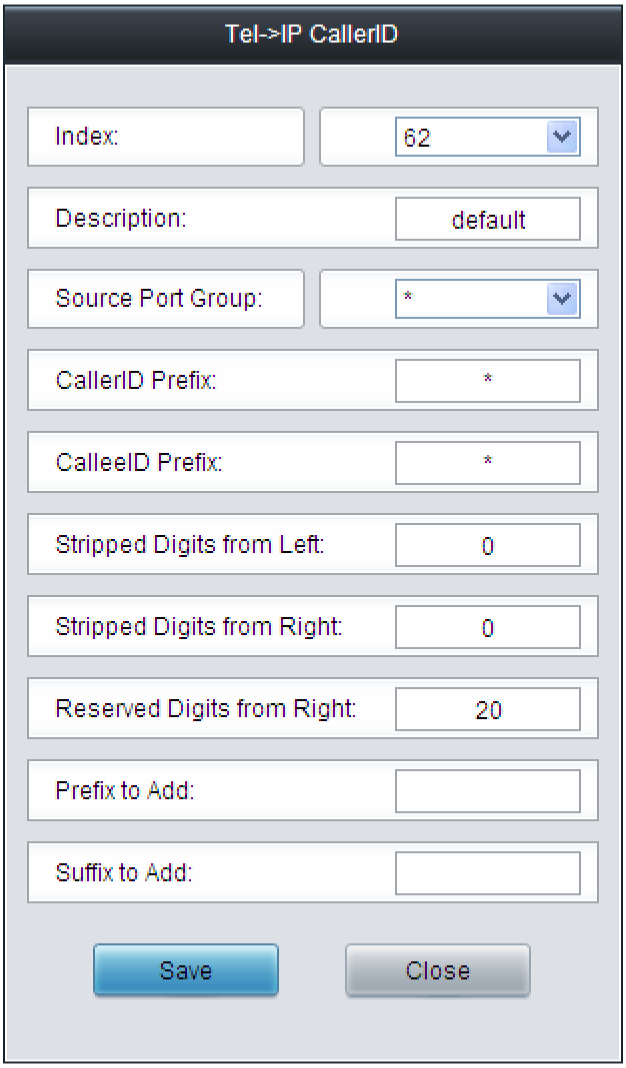


Figure 3-82 Add Tel→IP CallerID Manipulation Rule

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 3-81 to modify a number manipulation rule. See Figure 3-83 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.

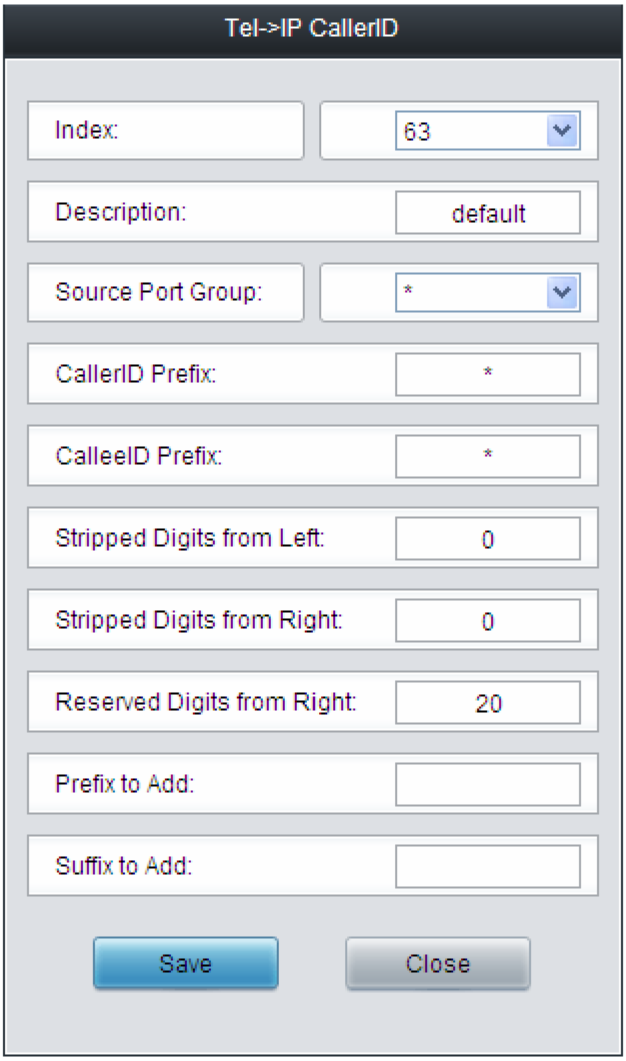


Figure 3-83 Modify Tel→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-81 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 3-81.

See Figure 3-84 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.

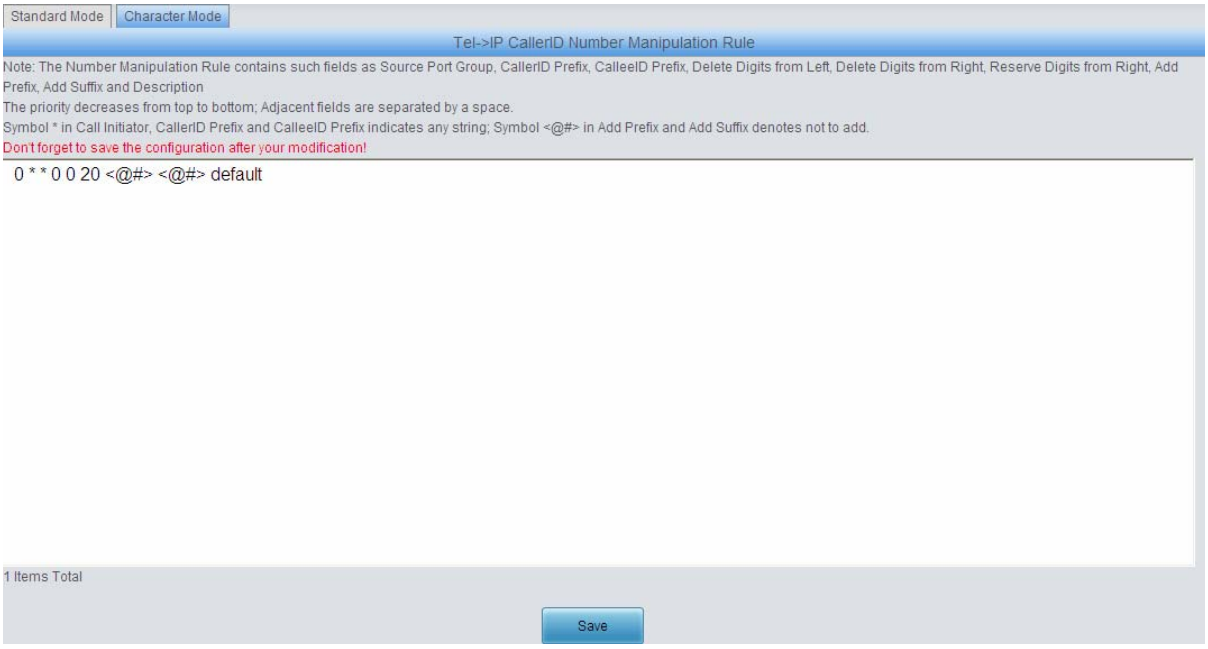


Figure 3-84 Tel→IP CallerID Manipulation Interface (Character)

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 3-85, Figure 3-86 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 3-81).

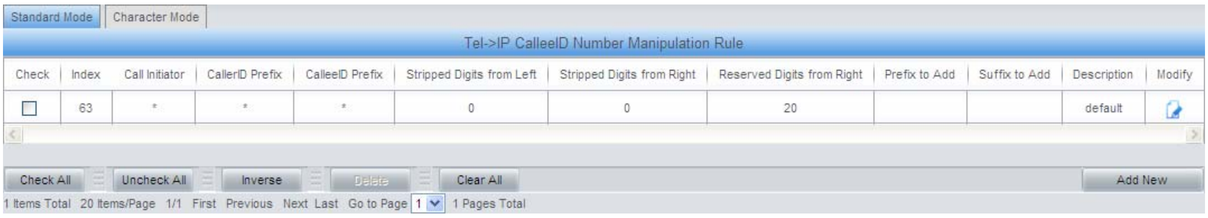


Figure 3-85 Tel→IP CalleeID Manipulation Interface (Standard)

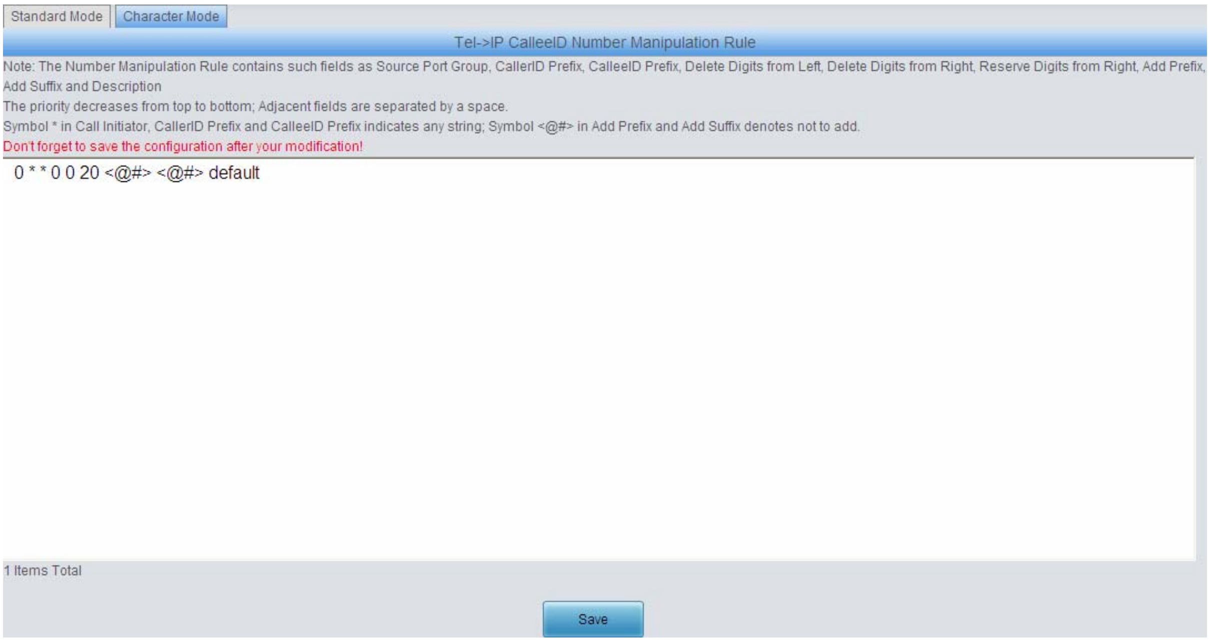


Figure 3-86 Tel→IP CalleeID Manipulation Interface (Character)

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 3-87 for details.

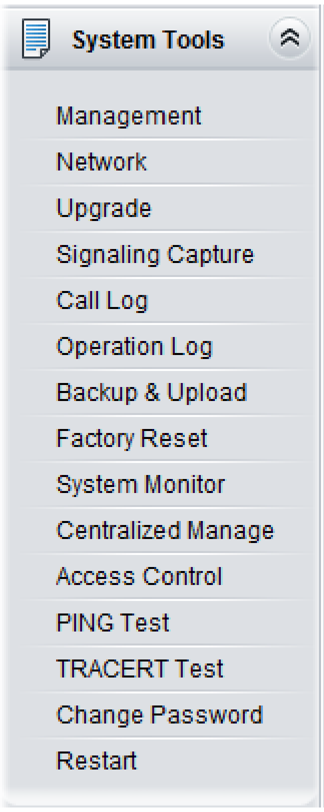


Figure 3-87 System Tools

3.9.1 Management

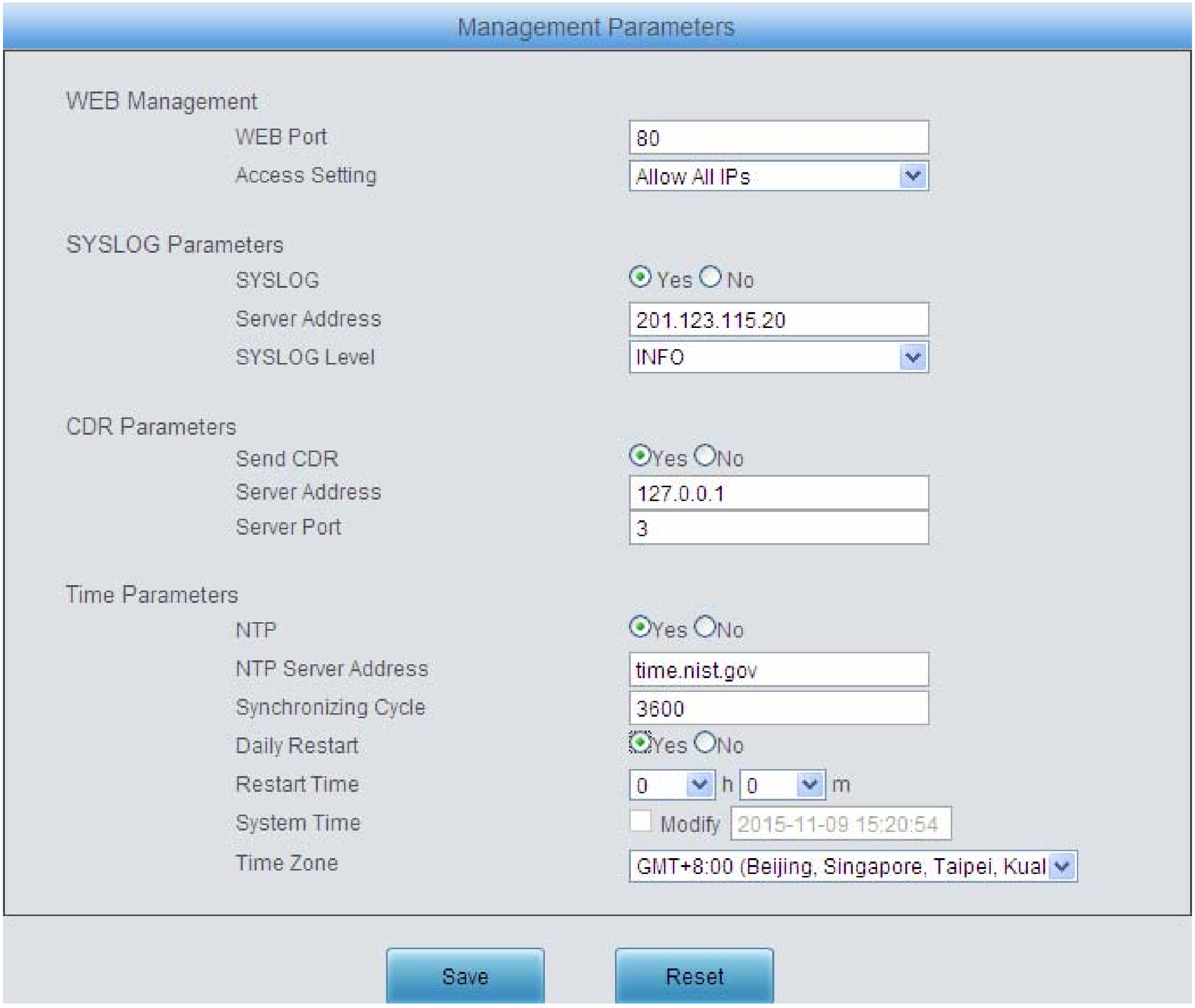


Figure 3-88 Management Parameters Setting Interface

See Figure 3-88 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* and *INFO*. 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 disabled.
NTP Server Address	Sets the Server address for NTP time synchronization.
Synchronizing Cycle	Sets the cycle for NTP time synchronization. The default value is 3600.
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.
Time Zone	The time zone of the gateway.

3.9.2 Network

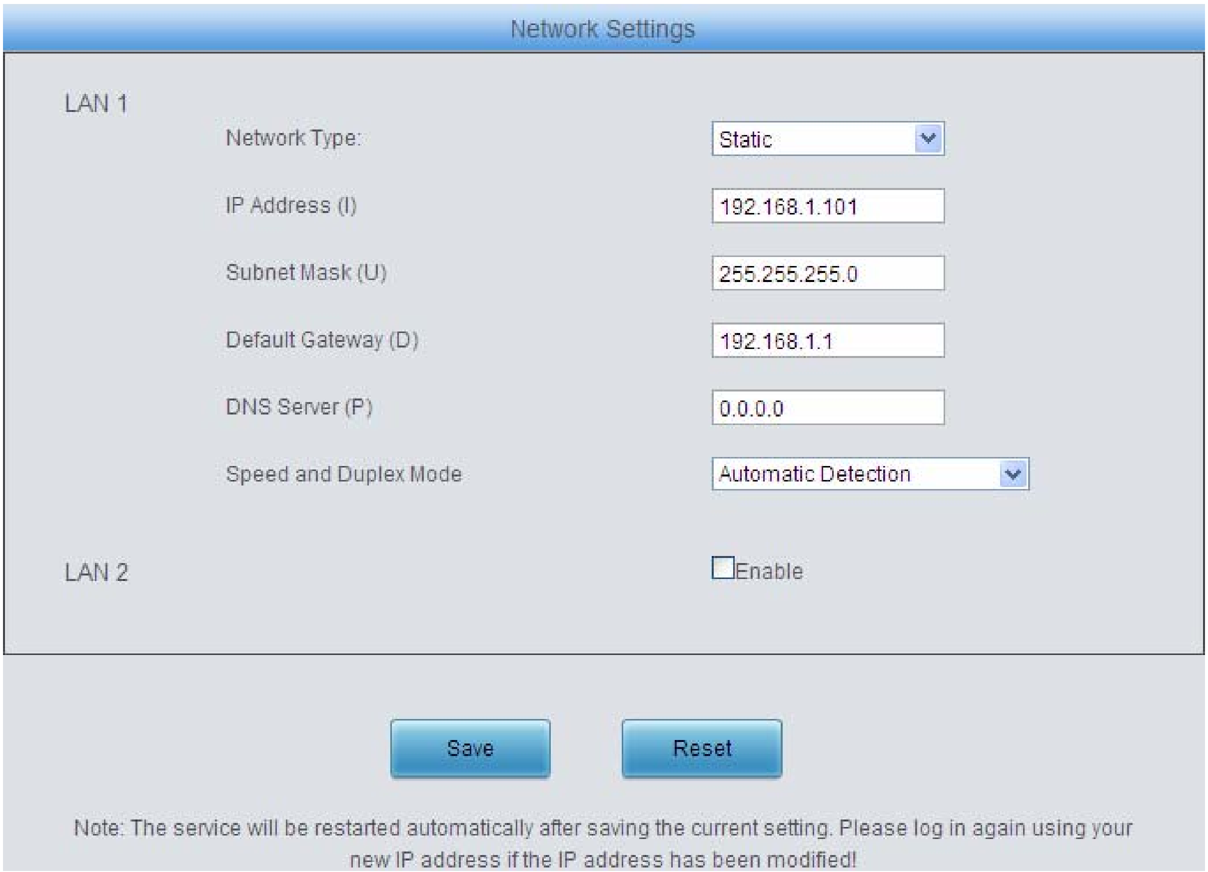


Figure 3-89 Network Settings Interface

See Figure 3-89 for the network settings interface. A gateway has two LANs, each of which can be configured with independent 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. By default, LAN1 is enabled and LAN2 is disabled.

Note:

1. The values of the *IP address*, *Subnet Mask*, *Default Gateway* and *DNS Server* shown in

Figure 3-89 are all factory settings. The IP Address for LAN 1 and that for LAN 2 cannot be in the same segment.

2. **LAN2 is disabled by default for the gateway Version 1.3.3 or above. If you want to use LAN2, please log in the gateway through LAN1 first, and then modify the network settings to enable LAN2.**

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

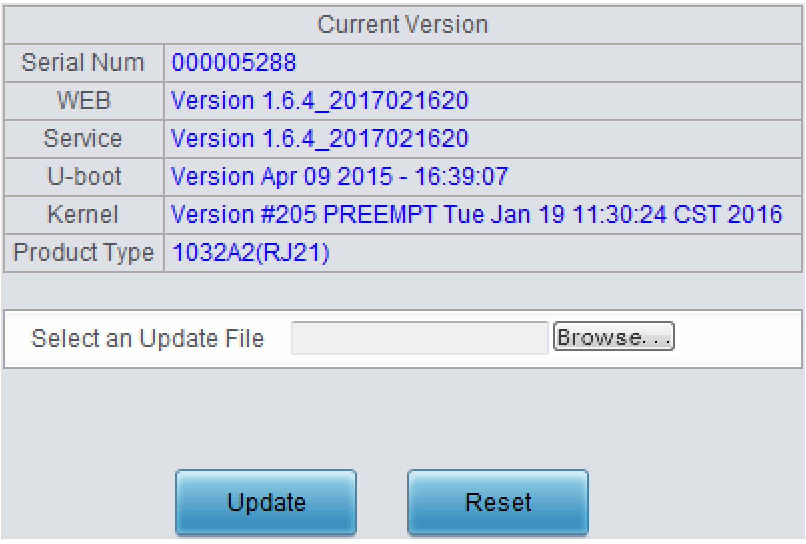


Figure 3-90 Upgrade Interface

See Figure 3-90 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 3-91.

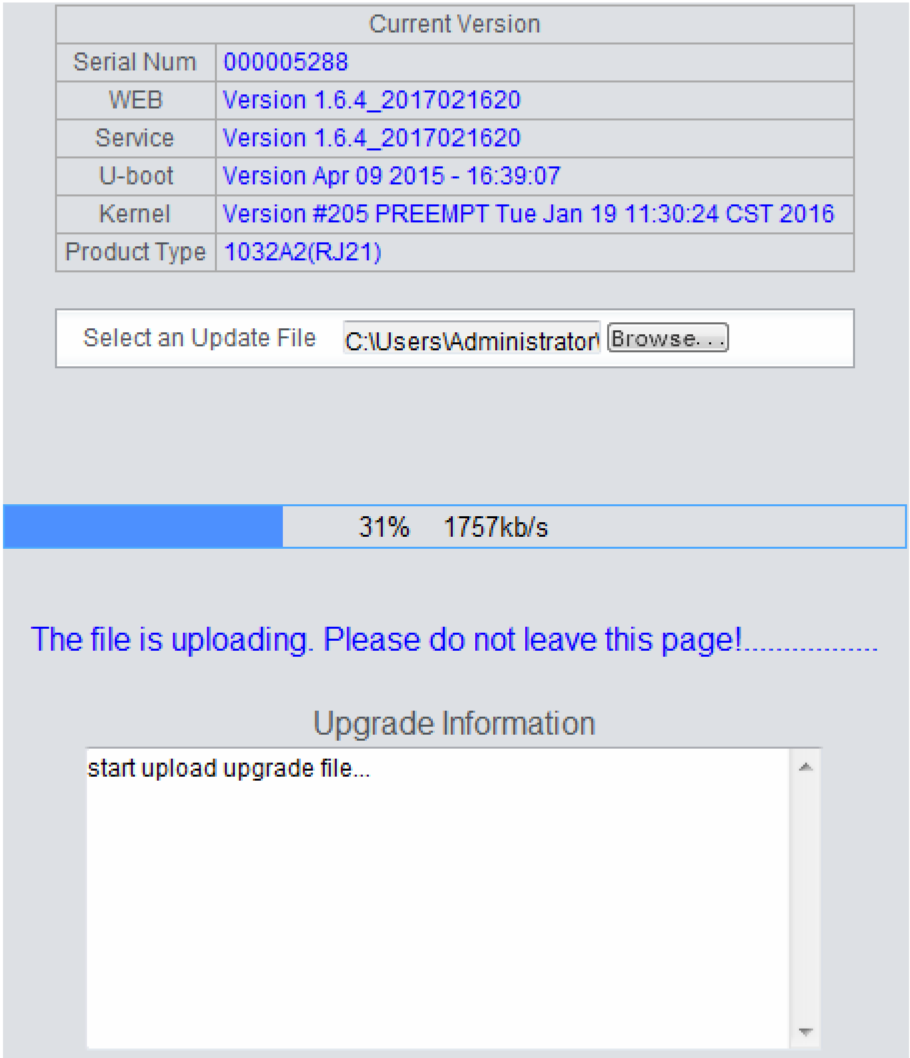


Figure 3-91 File Uploading Interface

After a successful uploading of the file, the gateway will start to upgrade the system. See Figure 3-92 and you can learn the detailed upgrading information from the upgrade information box at the bottom.

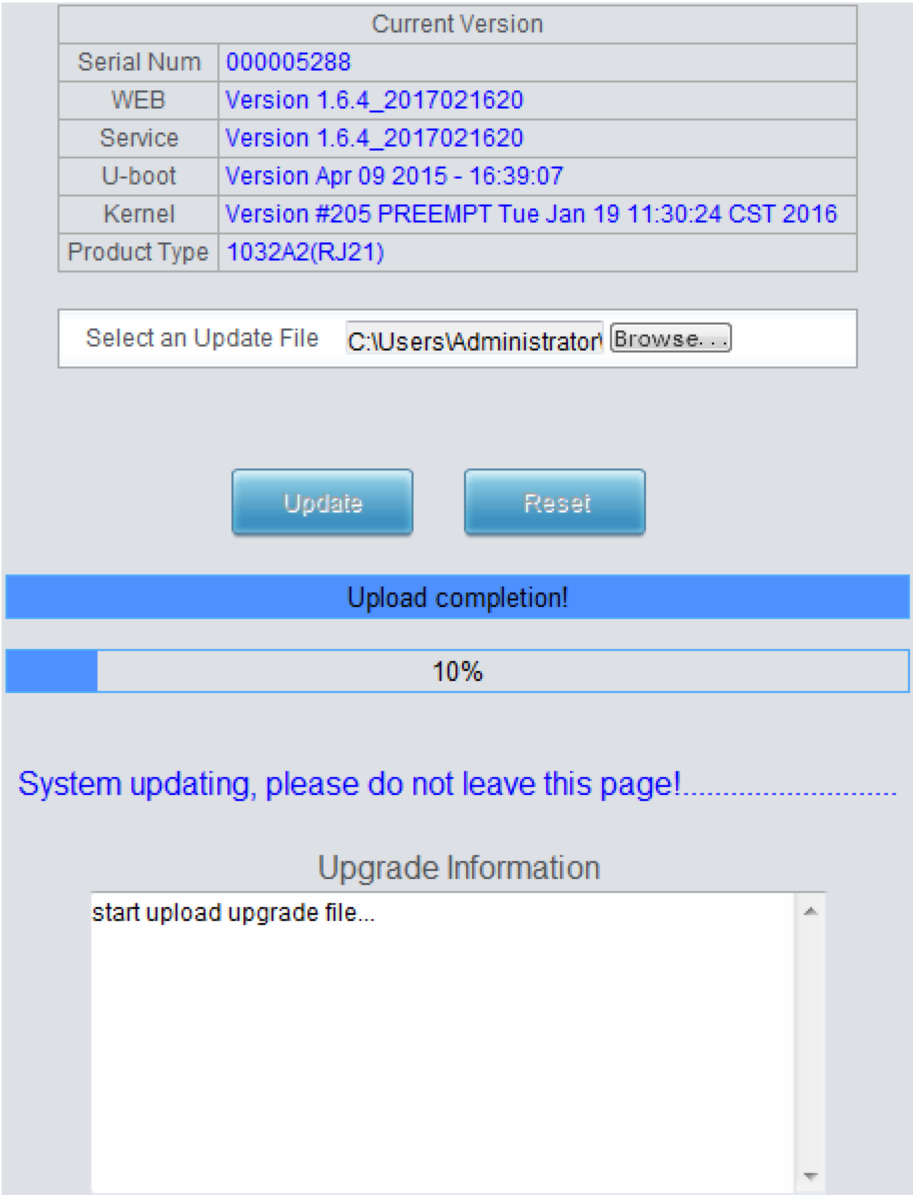


Figure 3-92 System Upgrading Interface

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

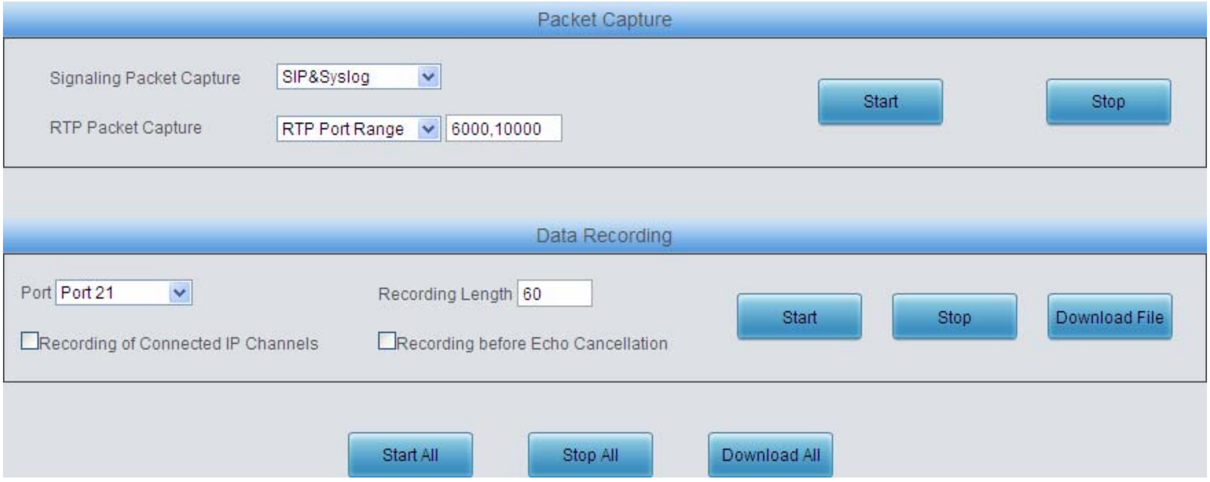


Figure 3-93 Signaling Capture Interface

See Figure 3-93 for the Signaling Capture interface, including two parts: Packet Capture and Data Recording. Packet capture contains Signaling Packet Capture and RTP Packet Capture. You can select either of them to start the capture according to your requirement. Click ***Start*** to start capturing packets. Click ***Stop*** to stop the capture and download the captured packets.

Data Recording will execute the recording task on the set port with the set recording time length. You can choose 'Recording of Connected IP Channels or 'Recording before Echo Cancellation'. Click ***Start*** to start recording data (consecutively recording 300 seconds at most) on the corresponding port with the corresponding time length. Click ***Stop*** to stop the recording and click ***Download File*** to download the recorded data.

3.9.5 Call Log

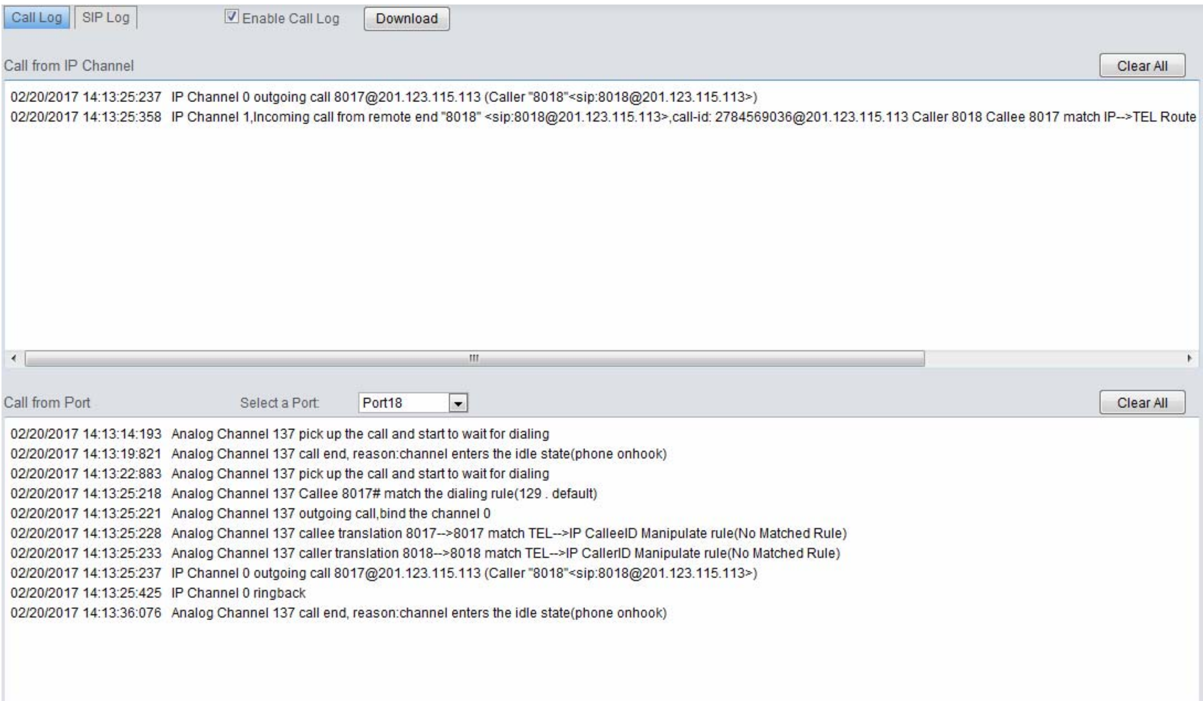


Figure 3-94 Call Log Interface

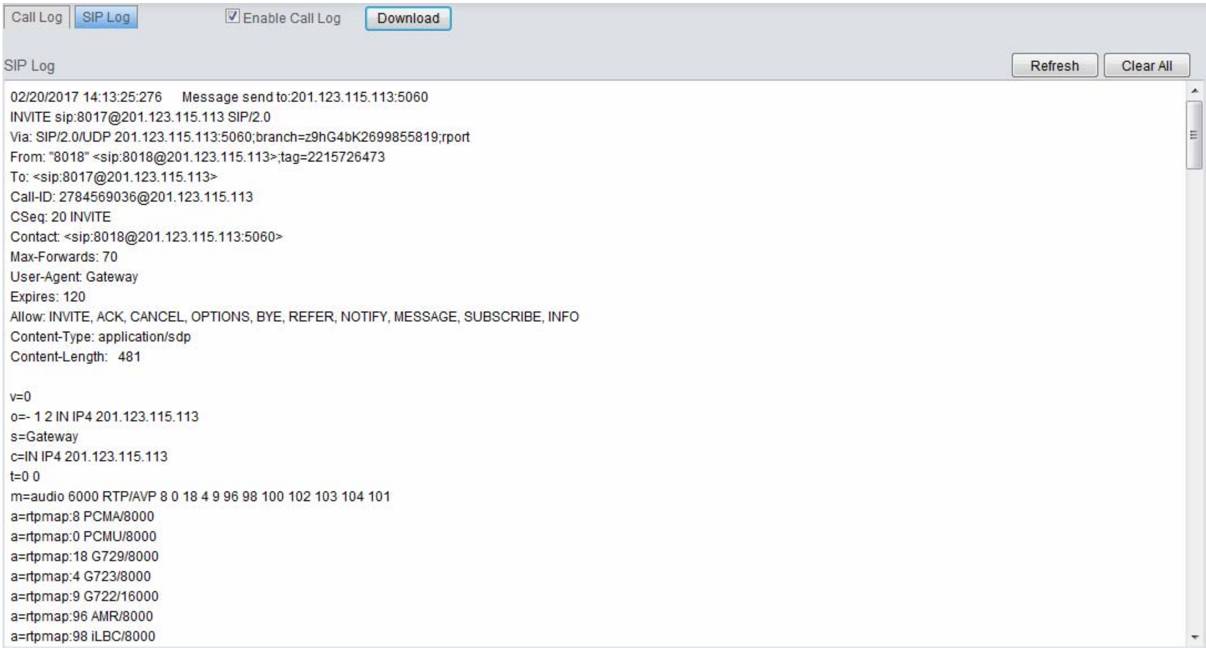


Figure 3-95 SIP Log Interface

See Figure 3-94, Figure 3-95 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***.

3.9.6 Operation Log

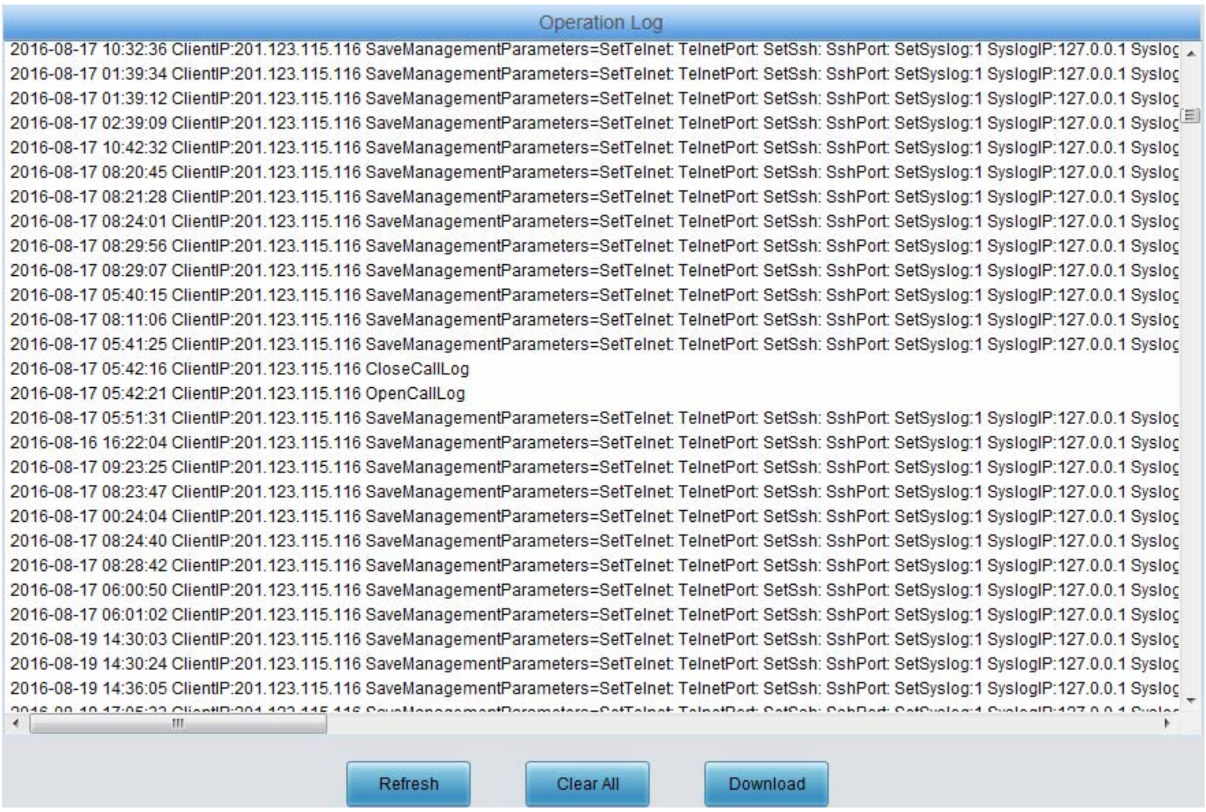


Figure 3-96 Operation Log Interface

See Figure 3-96 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.

3.9.7 Backup & Upload

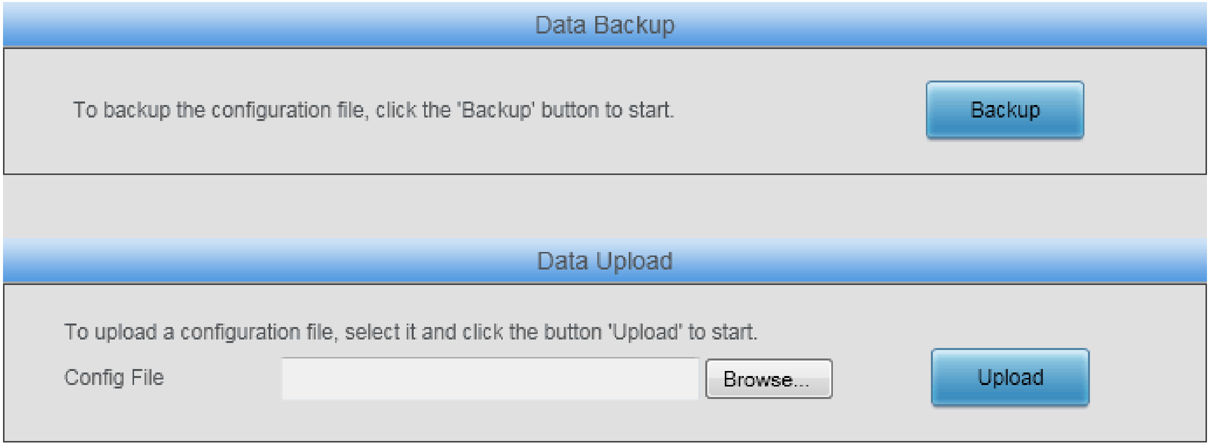


Figure 3-97 Backup & Upload Interface

See Figure 3-97 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***.

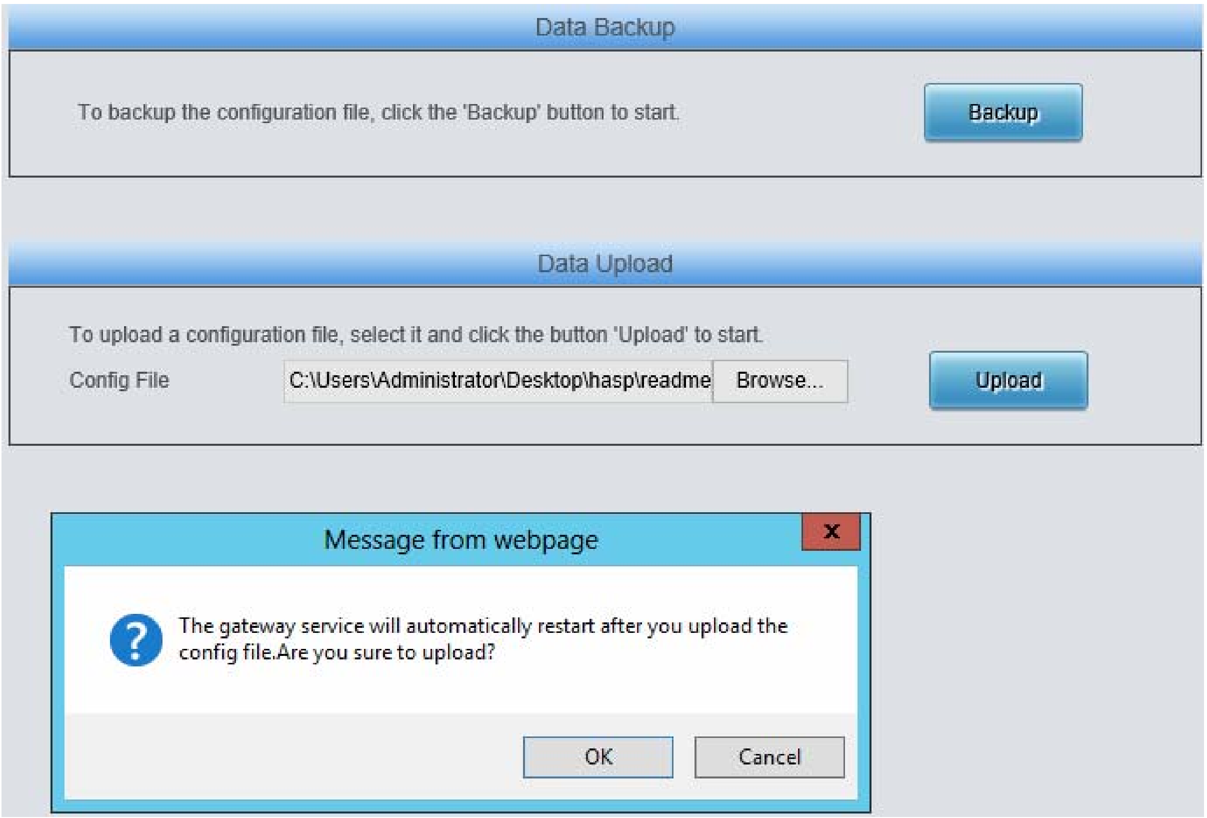


Figure 3-98 Backup & Upload & Prompt Interface

Click ***OK*** on the prompt box (Figure 3-98) to upload the configuration file to the gateway. Now the prompt information 'The gateway service is restarting, please do not leave this page' appears. See Figure 3-99. The gateway will overwrite the current configurations with the uploaded data after restart. Click ***Cancel*** to cancel this upload directly.

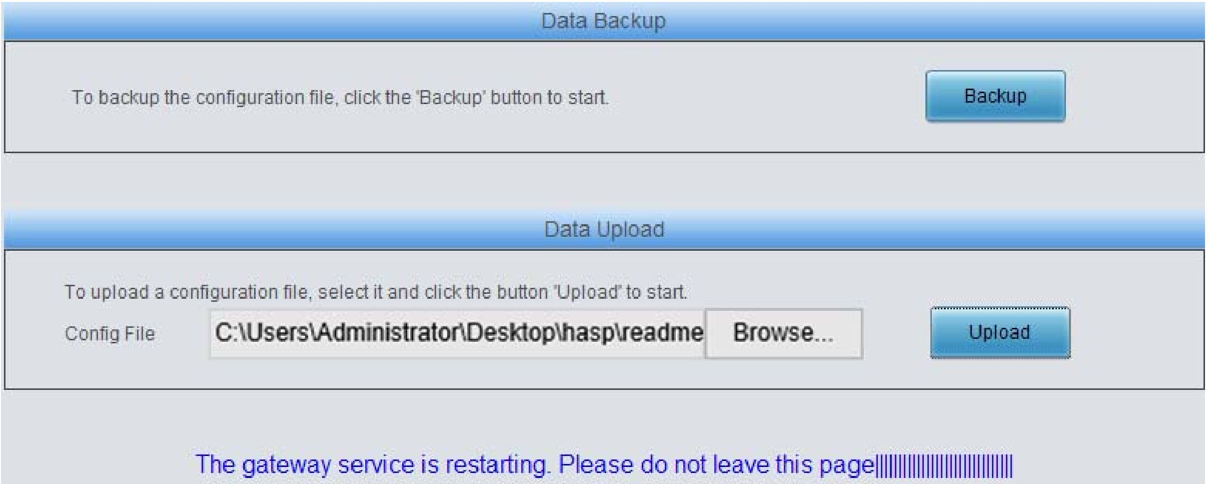


Figure 3-99 Configuration File Uploading Interface

3.9.8 Factory Reset

Figure 3-100 Factory Reset Interface

See Figure 3-100 for the factory reset interface. Click ***Reset*** to restore all configurations on the gateway to factory settings.

3.9.9 System Monitor

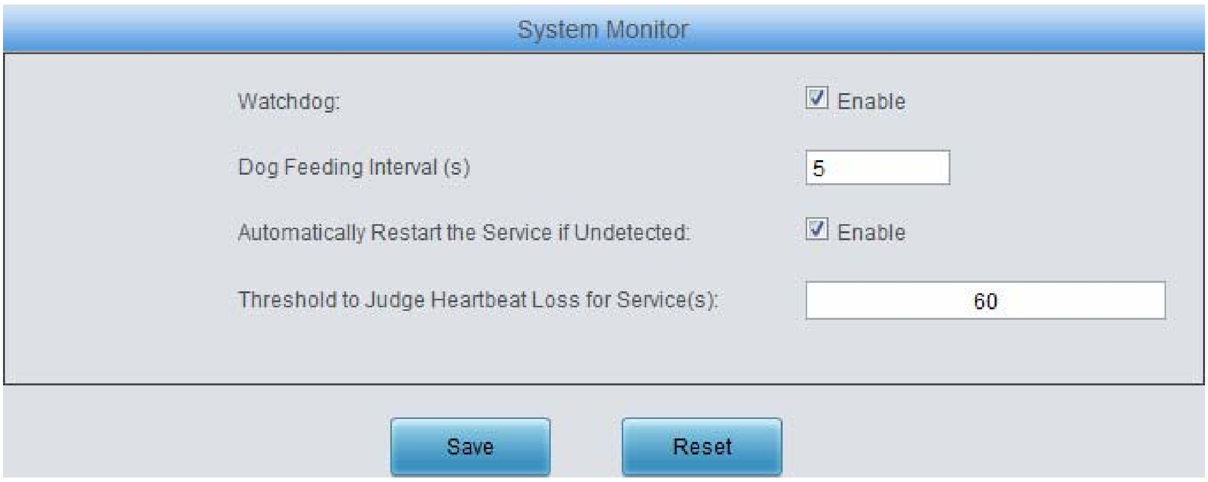


Figure 3-101 System Monitor Configuration Interface

See Figure 3-101 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.

3.9.10 Centralized Manage

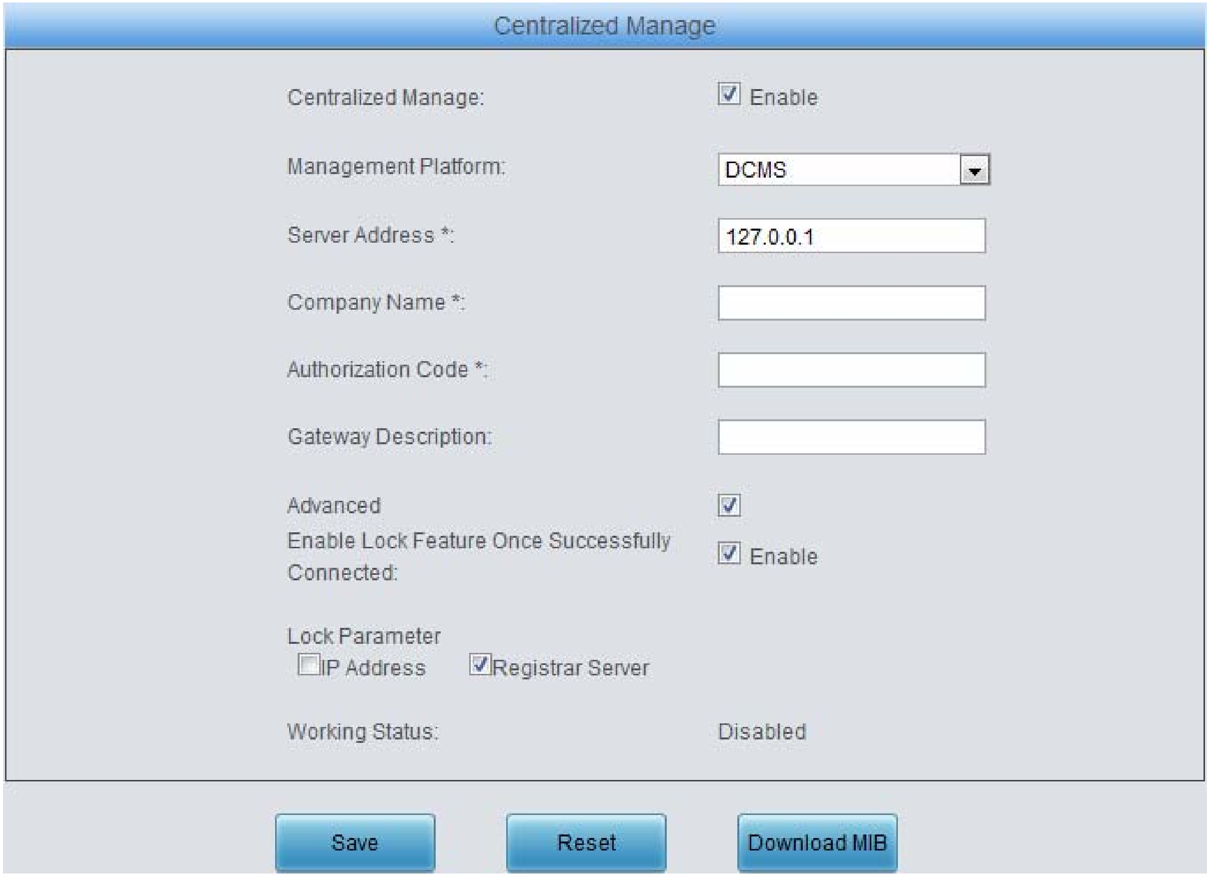


Figure 3-102 Centralized Manage Setting Interface

See Figure 3-102 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.
DCMS Server Address	The address of the server in which the DCMS 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. This item is valid only 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.

3.9.11 Access Control

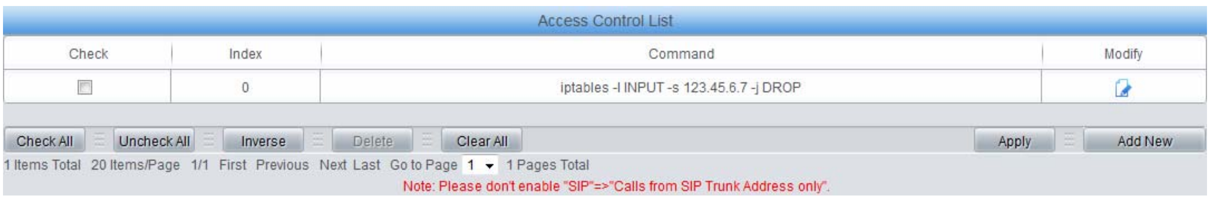


Figure 3-103 Access Control List Interface

See Figure 3-103 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 3-104.

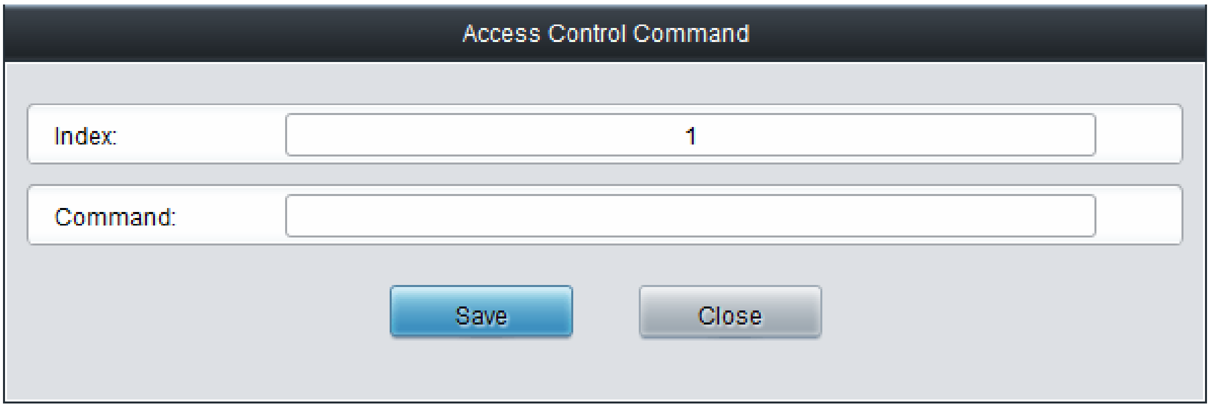


Figure 3-104 Add Access Control Command Interface

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 3-103 to modify a command. See Figure 3-105 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.

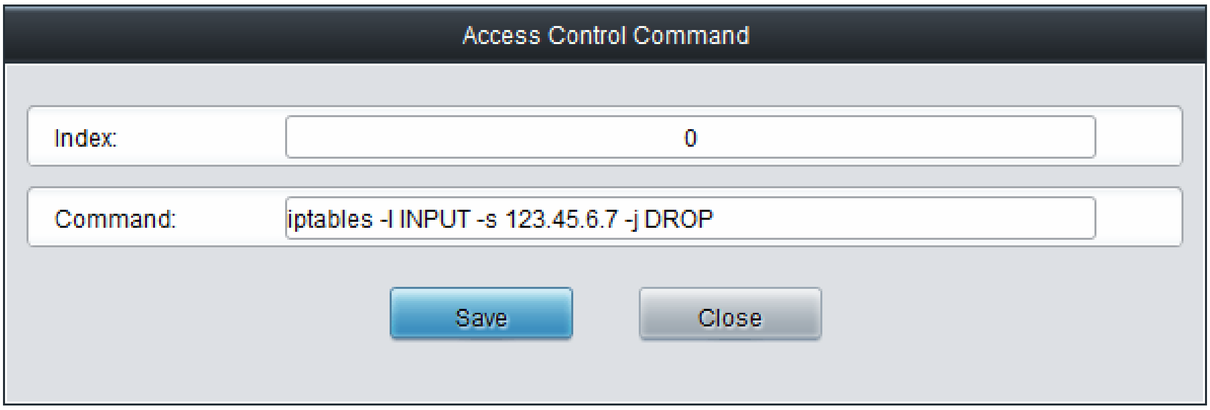


Figure 3-105 Access Control Command Modification Interface

To delete an Access Control Command, check the checkbox before the corresponding index in Figure 3-103 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 3-103.

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.

3.9.12 PING Test

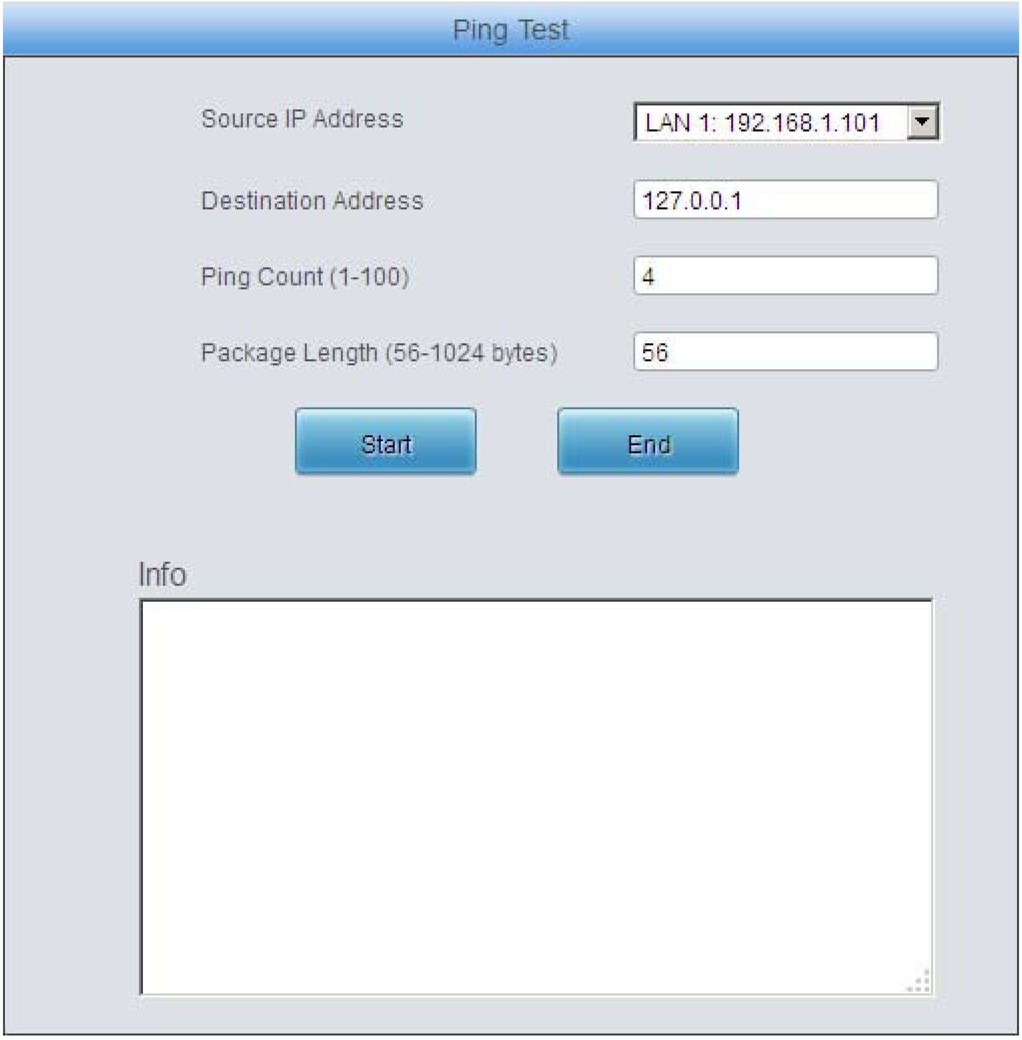


Figure 3-106 Ping Test Interface

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

Item	Description
Source IP Address	Source IP address where the Ping test is initiated.
Destination Address	Destination IP address 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.

3.9.13 TRACERT Test

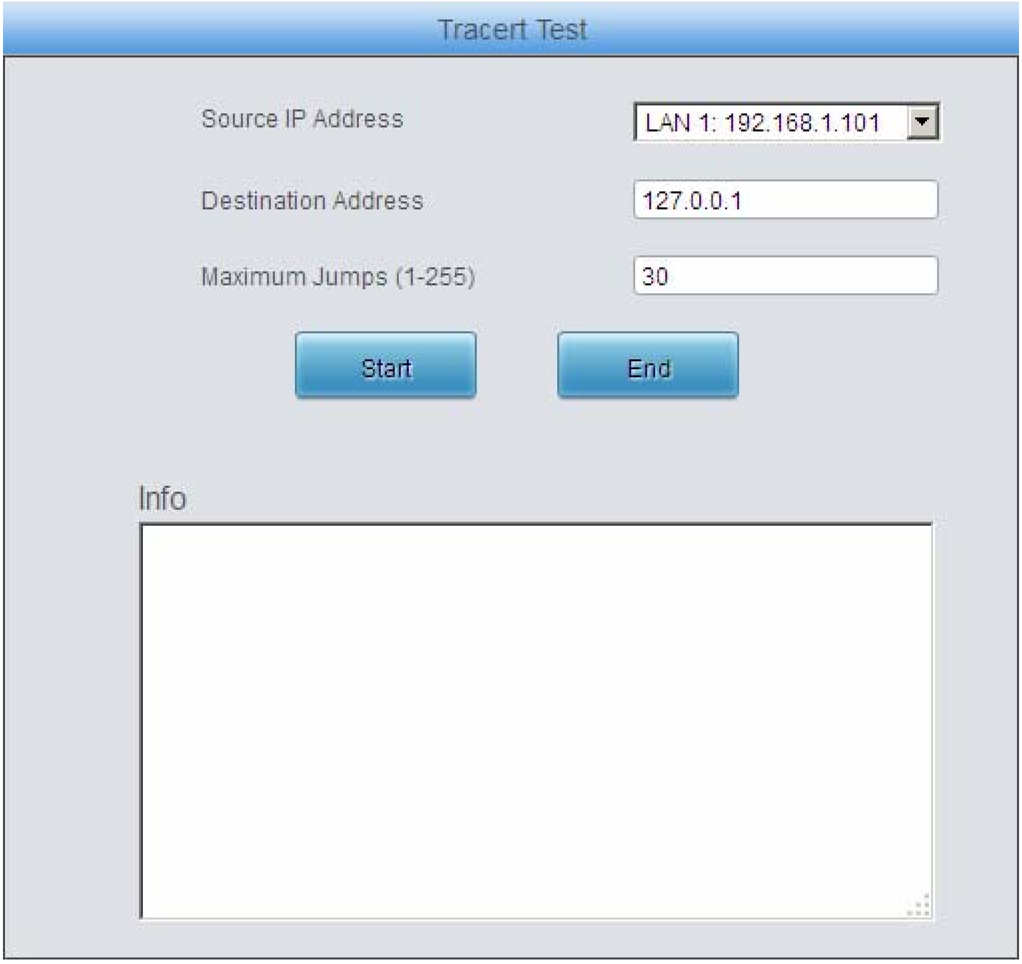


Figure 3-107 Tracert Test Interface

See Figure 3-107 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.

3.9.14 Change Password

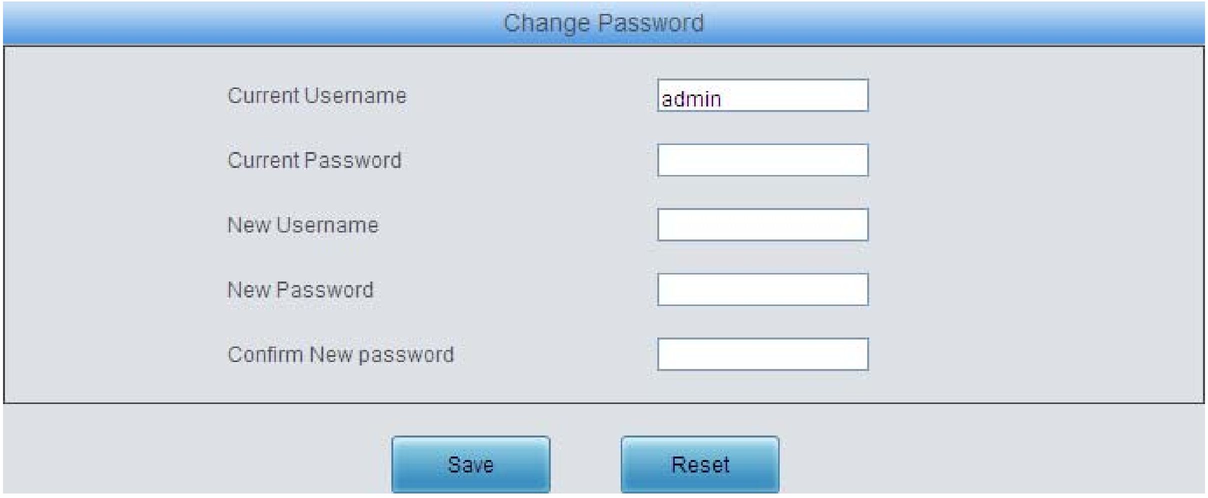


Figure 3-108 Password Changing Interface

See Figure 3-108 for the Password Changing interface where you can change your username and password of the gateway. Enter your current password, your new username and new password, and then confirm your new password. After configuration, click ***Save*** to apply your new username and password or click ***Reset*** to restore the configurations. After changing the username and password, you are required to log in again.

3.9.15 Restart

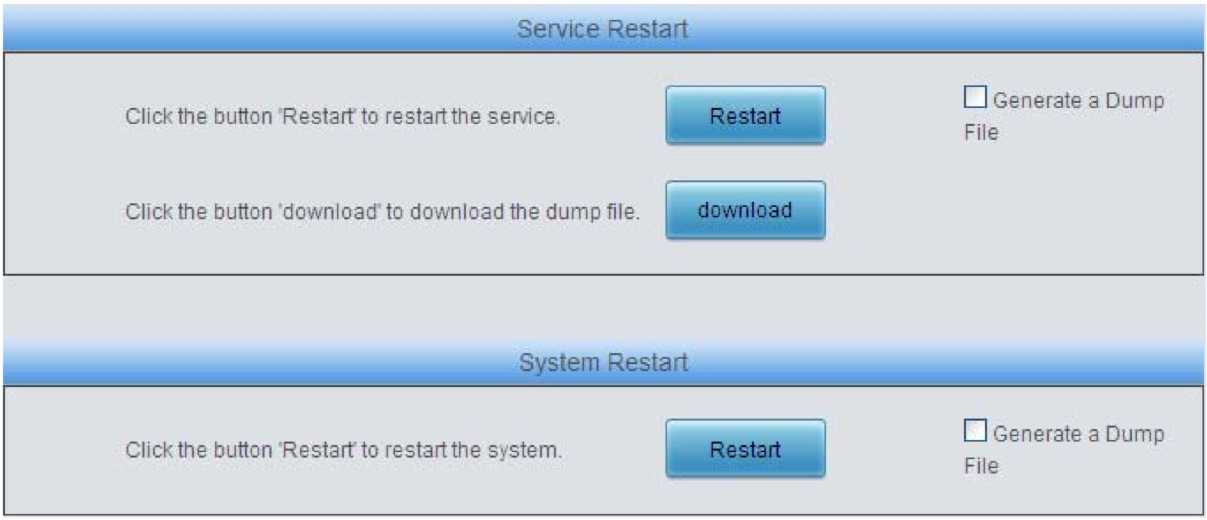


Figure 3-109 Service/System Restart Interface

See Figure 3-109 for the Restart interface. Click ***Restart*** on the service restart interface to restart the gateway service or click ***Restart*** on the system restart interface to restart the whole gateway system. A dump file will be generated each time you restart the service or the system. Click ***download*** and you can download it to help troubleshoot issues.

Appendix A Technical Specifications

Dimensions

440×44×267 mm³

Weight

About 4 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/FXO Port

Amount: 8/16/32

Type: RJ11, RJ21, RJ45

Maximum transmission distance: 1500m

Impedance

Input impedance:

≥1MΩ/500V DC; ≥10kΩ/1000V AC

Insulation resistance of telephone line from PC:

≥2MΩ/500V DC

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

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: 100~240V AC

Maximum power consumption：≤50W

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	*4.75/5.15/5.90/6.70/7.40/7.95/10.20/12.20 kbps*
iLBC	*13.3/15.2 kbps*
SILK(16K)	*20 kbps*
OPUS(16K)	*20 kbps*
SILK(8K)	*12 kbps*
OPUS(8K)	*12 kbps*

Sampling Rate

8kHz

Appendix B Troubleshooting

Q1. What to do if I forget the IP address of the SMG 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 as follows:

   LAN1: 192.168.1.101

   LAN2 (disabled by default): 192.168.0.101

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 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. LAN1: **192.168.1.101**) and port number (e.g. **5060**). Then the call initiated from the station corresponding to Port Group 1 will be routed to the gateway.

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

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

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 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 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 to configure the features Communication without Power and Communication without Network for the SMG analog gateway?

The feature **Communication without Power** is implemented with the help of composite modules equipped in the gateway. Once the power to the device is cut off, the station which is linked with the FXS port on the composite module and the trunk which is linked with the FXO port on the same module will connect to each other directly and keep the good communications between phones and networks. What you need to do is just to configure the composite module properly at your purchase of our gateway.

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

Refer to Q2 in this chapter for detailed information.

Q7. 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.

Q8. Is there any cell-phone APP can make calls to the SMG 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,

Q9. Does the SMG gateway support fax?

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

Q10. Which RTP codecs are supported by the SMG gateway?

At present, the supported RTP codecs are: G.711A, G.711u, G.729, G.723, G.722, AMR, iLBC, SILK(16K), OPUS(16K), SILK(8K) and OPUS(8K).

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.

Headquarters

Synway Information Engineering Co., Ltd

http://www.synway.net/

9F, Synway D&R Center, No.3756, Nanhuan Road, Binjiang District, Hangzhou, P.R.China, 310053

Tel: +86-571-88860561

Fax: +86-571-88850923

Wechat QR Code: Scan the QR code below to add us on Wechat.

Technical Support

Tel: +86-571-88864579

Mobile: +86-18905817070

Email: techsupport@sanhuid.com

Email: techsupport@synway.net

MSN: synway.support@hotmail.com

Sales Department

Tel: +86-571-88860561

Tel: +86-571-88864579

Fax: +86-571-88850923

Email: sales@synway.net