Synway SMG Series Digital Gateway

SMG2030

SMG2060

SMG2120

SMG3008

SMG3016

# Digital Gateway User Manual

Version 1.6.0

Synway Information Engineering Co., Ltd

www.synway.net

# Content

# Copyright Declaration

# Revision History

| Version | Date | Comments |
|---|---|---|
| Version 1.3.0 | 2014-06 | Initial publication. |
| Version 1.3.1 | 2014-08 | New revision |
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| Version 1.6.0 | 2015-03 | 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 Digital Gateway!

The Synway SMG series digital gateway products (hereinafter referred to as ‘SMG digital gateway’) are mainly used for connecting PSTN or enterprise PBX 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.

The SMG series digital gateway has five models:

- SMG2030: 1 E1/T1 interface (30 digital ports)
- SMG2060: 2 E1/T1 interfaces (60 digital ports)
- SMG2120: 4 E1/T1 interfaces (120 digital ports)
- SMG3008: 8 E1/T1 interfaces (240 digital ports)
- SMG3016: 16 E1/T1 interfaces (480 digital ports)

## 1.1 Typical Application

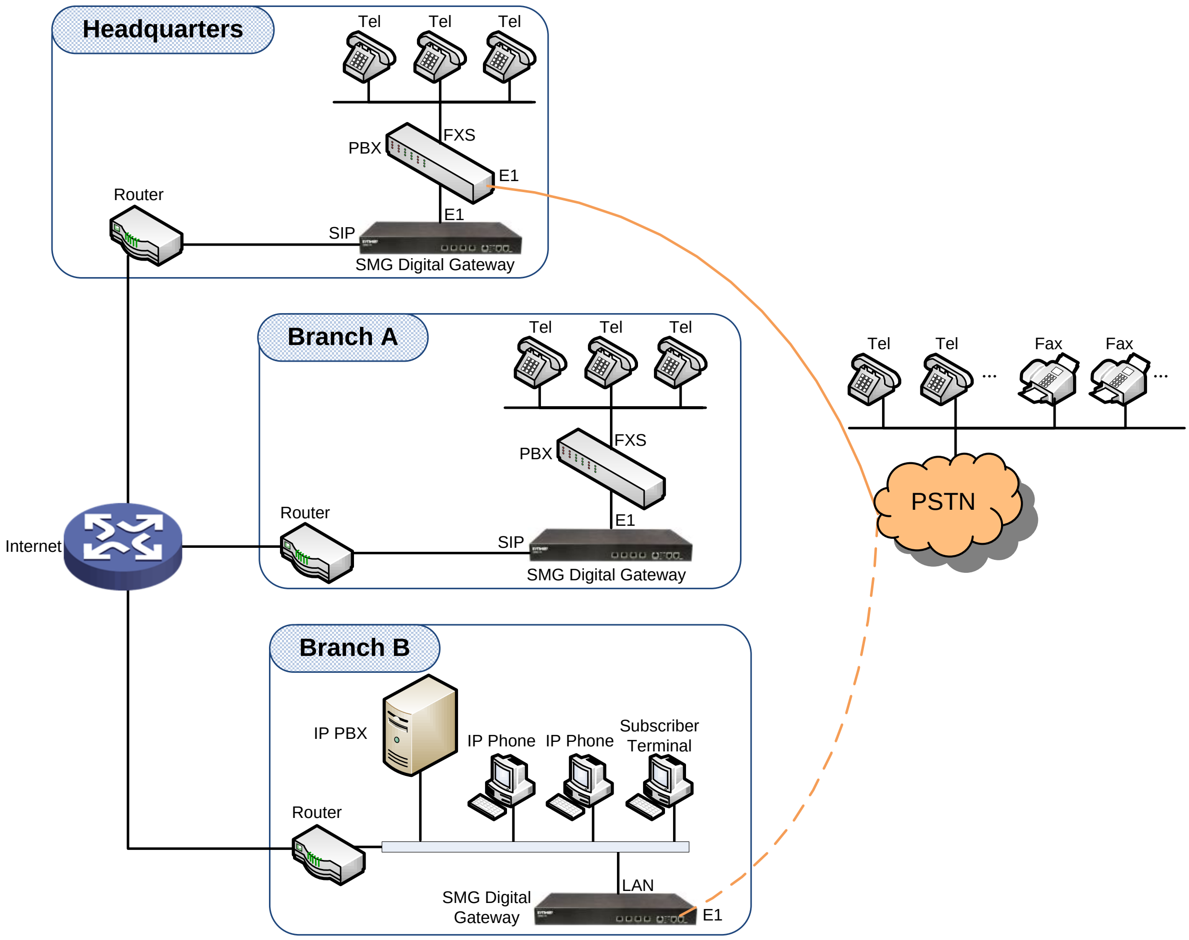



Figure 1-1 Typical Application

## 1.2 Feature List

| Basic Features | Description |
|---|---|
| ***PSTN Call*** | Call initiated from PSTN to a designated SIP trunk, via routing and number manipulation. |
| ***IP Call*** | Call initiated from IP to a designated PCM trunk, via routing and number manipulation. |
| ***Number Manipulation*** | Peels off some digits of a phone number from left/right, or adds a prefix/suffix to a phone number. |
| ***PSTN/ VoIP Routing*** | Routing path: from IP to PSTN or from PSTN to IP. |
| ***Fax*** | Multiple fax parameters: fax mode, maximum fax rate, fax train mode, error correction mode, etc. |
| ***Echo Cancellation*** | Provides the echo cancellation feature for a call conversation. |
| **Signaling & Protocol** | **Description** |
| ***SS7*** | SS7-TUP, SS7-ISUP |
| ***ISDN*** | ISDN User Side, ISDN Network Side |
| ***SS1*** | SS1 Signaling |
| ***SIP Signaling*** | Supported protocol: SIP V1.0/2.0, RFC3261 |
| ***Voice*** | CODEC: G.711A, G.711U, G.729A/B, G723, G722, AMR, iLBC<br>DTMF Mode: RFC2833, SIP INFO, INBAND |
| ***Fax*** | Fax Mode: T.38, Pass-Through<br>Baud Rate: 14400bps, 9600bps, 4800bps |
| **Network** | **Description** |
| ***Network Protocol*** | Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN |
| ***Static IP*** | IP address modification support |
| ***DNS*** | Domain Name Service support |
| **Security** | **Description** |
| ***Admin Authentication*** | Support admin authentication to guarantee the resource and data security |
| **Maintain & Upgrade** | **Description** |
| ***WEB Configuration*** | Support of configurations through the WEB user interface |
| ***Language*** | Chinese, English |
| ***Software Upgrade*** | Support of user interface, gateway service, kernel and firmware upgrades based on WEB |
| ***Tracking Test*** | Support of Ping and Tracert tests based on WEB |

| | |
|---|---|
| ***SysLog Type*** | Three options available: ERROR, WARNING, INFO |

## 1.3 Hardware Description

The SMG digital gateway features 1U rackmount design and integrates embedded LINUX system within the POWERPC+DSP hardware architecture. It has 1/2/4/8/16 E1/T1 ports and 2 Kilomega-Ethernet ports (LAN1 and LAN2) on the chassis.

（a） See below figures for SMG20000 series appearance：


Figure 1-2 Front View


Figure 1-3 Rear View


Figure 1-4 Left View

（b） See below figures for SMG3000 series appearance：


Figure 1-5 Front View

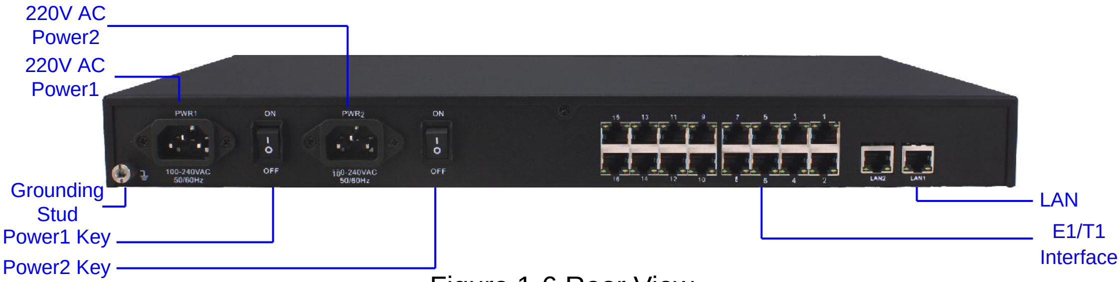



Figure 1-6 Rear View

**Note:** The left view for SMG3000 series is same as that for SMG2000 series, refer to Figure 1-4.

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

| Interface | Description |
|---|---|
| ***LAN*** | Amount: 2 |
| | Type: RJ-45 |
| | Bandwidth: 10/100/1000Mbps |
| | Self-Adaptive Bandwidth Supported |
| | Auto MDI/MDIX Supported |
| ***E1/T1*** | Amount: 1/2/4/8/16 |
| | Type: RJ-45 |
| ***Console Port*** | Amount: 1 |
| | Type: RS-232 |
| | Baud Rate: 115200 bps |
| | Connector: RJ45 (See Figure 1-7 for signal definition) |
| | Data Bits: 8 bits |
| | Stop Bit: 1 bit |
| | Parity Unsupported |
| | Flow Control Unsupported |
| **Button** | **Description** |
| ***Power Key*** | Power on/off the SMG digital gateway. You can turn on the two power keys at the same time to have the power supply working in the hot-backup mode. |
| ***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. |
| ***E1/T1 Indicators*** | The green LED on the right of E1/T1 interface lights up and keeps on after the E1/T1 module is successfully synchronized. |

| | |
|---|---|
| ***Channel Indicators*** | Indicates the synchronization status of E1/T1 channels. It will light up and keep on if E1/T1 is synchronized; otherwise, it will go out. |

Note: The console port is used for debugging. While connection, the transmitting and receiving lines of the gateway and the remote device should be cross-linked. That is, connect the transmitting line of the gateway to the receiving line of the remote device, and vice verse. The figure below illustrates the signal definition of the console port on the gateway.

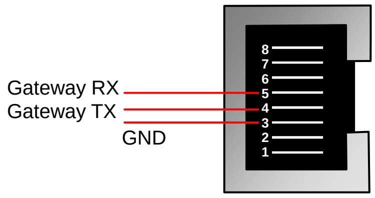



Figure 1-7 Console Port Signal Definition

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

## 1.4 Alarm Info

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

| LED | State | Description |
|---|---|---|
| ***Run Indicator*** | Go out | System is not yet started. |
| | Light up | System is starting. |
| | Flash | Device is running normally. |
| ***Alarm Indicator*** | Go out | Device is working normally. |
| | Light up | Upon startup: Device is running normally.<br>In runtime: Device goes 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 E 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 digital gateway in the shortest time.

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

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

**Step 2: Properly fix the SMG digital 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.

**Note:** Each SMG digital gateway has two power interfaces to meet the requirement for power supply hot backup. As long as you properly connect and turn on these two power keys, either power supply can guarantee the normal operation of the gateway even if the other fails.

**Step 4: Connect the network cable.**

**Step 5: Connect the E1/T1 trunk. Connect the E1/T1 interface of the digital gateway to that of the remote device by E1/T1 trunk. After connection, check if the synchronization indicator (green LED) is lit and keeps on, which indicates that the E1/T1 trunk is well connected and the E1/T1 module is successfully synchronized.**

For the 75Ω-unbalanced coaxial cable, in consideration of various line conditions, each PCM on the digital gateway is equipped with two grounding jumpers which respectively control the grounding of the transmitting and the receiving end. Under normal condition, that is, the chassis of the gateway is well grounded, the grounding jumpers at the receiving end should be disconnected and the ones at the transmitting end should be short-circuited. This configuration is the factory default setting and applicable in most situations so that there is usually no need to change it. For the 120Ω-balanced twisted pair cable, the grounding jumpers at both ends should be disconnected.

You can construct an E1 trunk according to Figure 2-1. Prevent reverse connection of the transmitting and receiving lines. The state of the receiving line can be checked by the synchronization indicator (green LED) of the E1 interface. When the receiving line is in a normal state, the indicator is lit and keeps on. If the indicator is off or flashing, it means that the connection of the receiving line may probably be reversed. However, the state of the transmitting line can only be examined by the opposite terminal. The synchronization indicator starts working only after the device is powered on and successfully initialized.

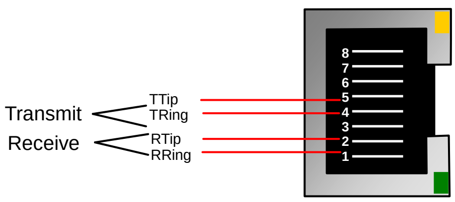


Figure 2-1 Pin Layout for E1 Interface

**Step 6: Log in the gateway.**

Enter the original IP address (LAN 1: 192.168.1.101 or LAN 2: 192.168.0.101) of the SMG digital gateway in the browser to go to the WEB interface. 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 3.12.15 Change Password. After changing the password, you are required to log in again.

**Step 7: Modify IP address of the gateway.**

You can modify the IP address of the gateway via 'System Tools → Network' on the WEB interface to put it within your company's LAN. Refer to 3.12.1 Network for detailed instructions about IP modification. After changing the IP address, you shall log in the gateway again using your new IP address.

**Step8: Set PCM.**

On your initial use of the SMG digital gateway, you shall enter the PCM interface and set the configuration items 'Signaling Protocol' and 'Interface'. These items must be in conformity with the physical connection. You may use the default values of other configuration items. Refer to 3.4.3 PCM for detailed instructions about PCM Settings.

**Note:** You shall restart the service to validate the settings in this step. Refer to 3.12.17 Restart for detailed instructions.

**Step 9: Configure signaling protocol parameters.**

Further configure the signaling protocol you set in Step 8. Different protocols are configured on different interfaces. See below for detailed instructions.

- **SS7-ISUP:**

**Note: For your easy understanding and manipulation, this step does not involve the ISUP quasi-associated mode configuration and the dual gateway feature. For descriptions about these configurations, refer to 3.5 SS7 Settings.**

The configuration interfaces related to SS7-ISUP include: SS7, ISUP and SS7 Server.

On your initial use of the SMG digital gateway, you may adopt the default values of the configuration items on the SS7 and ISUP interfaces. Note that the SS7 Server interface must be configured properly. Otherwise, the PSTN trunks may be unavailable. Follow the instructions here to configure the SS7 Server:

Step 1: Set OPC, Server IP and Signaling Point Code Standard. The OPC is generally allocated by the central office. The Server IP is the IP address of the SS7 server and you may use its default value. The Signaling Point Code Standard, which varies on the PBX model, can be set to 24 or 14. After modification, click the 'Modify' button on the right to save the settings.

Step 2: Modify the current link or click the 'Add New' button below the signaling link list to add a new link. Enter the physical address of the actually used signaling PCM (E1 interface) and click 'Save' to save the modification. If only one PCM is used for signaling in the gateway, you need just configure one signaling link.

Step 3: Modify the current linkset or click the 'Add New' button below the signaling linkset list to add a new linkset. You shall select the link configured in Step 2 for 'Link' and use the default values for the other configuration items. After modification, click 'Save'.

Step 4: Modify the current DPC or click the 'Add New' button below the DPC list to add a new DPC. Fill in 'SP Code' with the signaling point code of the remote end (i.e. signaling destination), select the linkset configured in Step 3 for 'Linkset' and use the default values for the other configuration items. After modification, click 'Save'.

Step 5: Modify the current CIC routing rule or click the 'Add New' button below the ISUP_CIC routing rule list to add a new CIC routing rule. Select the DPC configured in Step 4 for 'DPC', fill in 'CIC_PCM' according to the actual allocation and use the default values for the other configuration items. After modification, click 'Save'. Note that if multiple PCMs in the gateway are used for voice transmission, they should be configured with multiple CIC routing rules accordingly.

**Note:** After configuring SS7-ISUP related interfaces, you shall restart the service to validate the settings. Refer to 3.12.17 Restart for detailed instructions.

- **SS7-TUP:**

**Note: For your easy understanding and manipulation, this step does not involve the TUP quasi-associated mode configuration and the dual gateway feature. For descriptions about these configurations, refer to 3.5 SS7 Settings.**

The configuration interfaces related to SS7-TUP include: SS7, TUP and SS7 Server.

On your initial use of the SMG digital gateway, you may adopt the default value of the configuration items on the SS7 and TUP interfaces. Note that the SS7 Server interface must be configured properly. Otherwise, the PSTN trunks may be unavailable. Follow the instructions here to configure the SS7 Server:

Step 1: Set OPC, Server IP and Signaling Point Code Standard. The OPC is generally allocated by the central office. The Server IP is the IP address of the SS7 server and you may use its default value. The Signaling Point Code Standard, which varies on the PBX model, can be set to 24 or 14. After modification, click the 'Modify' button on the right to save the settings.

Step 2: Modify the current link or click the 'Add New' button below the signaling link list to add a new link. Enter the physical address of the actually used signaling PCM (E1 interface) and click 'Save' to save the modification. If only one PCM is used for signaling in the gateway, you need just configure one signaling link.

Step 3: Modify the current linkset or click the 'Add New' button below the signaling linkset list to add a new linkset. You shall select the link configured in Step 2 for 'Link' and use the default values for the other configuration items. After modification, click 'Save'.

Step 4: Modify the current DPC or click the 'Add New' button below the DPC list to add a new DPC. Fill in 'SP Code' with the signaling point code of the remote end (i.e. signaling destination), select the linkset configured in Step 3 for 'Linkset' and use the default values for the other configuration items. After modification, click 'Save'.

Step 5: Modify the current CIC routing rule or click the 'Add New' button below the TUP_CIC routing rule list to add a new CIC routing rule. Select the DPC configured in Step 4 for 'DPC', fill in 'CIC_PCM' according to the actual allocation and use the default values for the other configuration items. After modification, click 'Save'. Note that if multiple PCMs in the gateway are used for voice transmission, they should be configured with multiple CIC routing rules accordingly.

**Note:** After configuring SS7-TUP related interfaces, you shall restart the service to validate the settings. Refer to 3.12.17 Restart for detailed instructions.

- **ISDN User Side/Network Side:**

The configuration interface related to ISDN User Side/Network Side is ISDN. On your initial use of the SMG digital gateway, you may adopt the default value of the configuration items on this interface.

**Note:** After configuring the ISDN interface, you shall restart the service to validate the settings. Refer to 3.12.17 Restart for detailed instructions.

- **SS1:**

The configuration interface related to SS1 is SS1. On your initial use of the SMG digital gateway, you may adopt the default value of the configuration items on this interface.

**Note:** After configuring the SS1 interface, you shall restart the service to validate the settings. Refer to 3.12.17 Restart for detailed instructions.

**Step 10: Check the PSTN status.**

After the configuration of signaling protocols, you can check the status of the PSTN trunks via 'Operation Info → PSTN Status'. Refer to 3.2.2 PSTN Status for detailed introductions. When Time Slot 0 shows 'Frame Synchronized', the signaling time slot is in the state of 'Signaling Channel' and all the other channels are 'Idle', it indicates the PCM is well configured. If Time Slot 0 or the signaling time slot shows 'Faulty' or the other channels are in the state of 'Unavailable', there may be errors in the signaling protocol configurations and we suggest you return to Step 9 for check.

**Step 11: Set routing rules for calls.**

Note: For your easy understanding and manipulation, all examples given in this step do not involve registration.

**Situation 1: IP → PSTN**

Step 1: Configure the IP address of the remote SIP terminal which can establish conversations with the gateway so that the calls from other terminals will be ignored. Refer to 'VOIP Settings → SIP Trunk' for detailed instructions. Fill in 'Remote IP' and 'Remote Port' with the IP address and port of the remote SIP terminal which will initiate calls to the gateway. You may use the default values for the other configuration items.

**Example:** Provided the IP address of the remote SIP terminal is 192.168.0.111 and the port is 5060. Add **SIP Trunk 0**; set **Remote IP** to **192.168.0.111** and **Remote Port** to **5060**.

Step 2: Add the IP address of the remote SIP terminal configured in Step 1 into the corresponding SIP trunk group. Refer to 'VoIP Settings → SIP Trunk Group' for detailed instructions. Select the SIP trunk configured in Step 1 as 'SIP Trunks'. You may use the default values for the other configuration items.

**Example:** Add **SIP Trunk Group 0**. Check the checkbox before **0** for **SIP Trunks** and keep the default values for the other configuration items.

Step 3: Add PCM into the corresponding PCM Group. Refer to 'PCM Settings → PCM Trunk Group' for detailed instructions. Select the PCM used for call conversation as 'PCM'. You may use the default values for the other configuration items.

**Example:** Provided the PCM used for call conversation is PCM[1]. Add **PCM Trunk Group 0**, check the checkbox before **PCM[1]** and keep the default values for the other configuration items.

Step 4: Add routing rules. Refer to 'Route Settings → IP→PSTN' for detailed instructions. Select the SIP trunk group set in Step 2 as 'Call Initiator' and the PCM trunk group set in Step 3 as 'Call Destination'. You may use the default values for the other configuration items.

**Example:** Select **SIP Trunk Group[0]** as **Call Initiator** and **PCM Trunk Group[0]** as **Call Destination.** Keep the default values for the other configuration items.

Step 5: Initiate a call from the SIP terminal configured in Step 1 to the IP address and port of the SMG digital gateway. Thus you can establish a call conversation via PCM[1] with the PSTN terminal. (Note: The format used for calling an IP address via SIP trunk is as follows: username@IP address, in which, 'username' is a called party number which conforms to the number-receiving rule of the remote device.)

**Example:** Provided the IP address of the SMG digital gateway is 192.168.0.101 and the port is 5060. Provided 123 is a number which conforms to the number receiving rule of the remote device. Initiate a call from SIP terminal 0 to the IP address 192.168.0.101 (in the format: 123@192.168.0.101) and you can establish a call conversation via PCM[1] to the number 123.

**Situation 2: PSTN → IP**

Step 1: Configure the called party numbers which are received from PSTN and will be processed by the gateway. Refer to 'Advanced Settings → Number-receiving 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 for 'Index'.

**Example:** Set **Index** to **99** and configure **Dial Rule** to **123**.

Step 2: Set the IP address of the SIP terminal to be called by the gateway. Refer to 'VOIP Settings → SIP Trunk' for detailed instructions. Fill in 'Remote IP' and 'Remote Port' with the IP address and port of the SIP trunk. You may use the default values for the other configuration items.

**Example:** Provided the IP address of the SIP trunk to be called is 192.168.0.111 and the port is 5060. Add **SIP Trunk 0**; set **Remote IP** to **192.168.0.111** and **Remote Port** to **5060**.

Step 3: Add the IP address of the remote SIP terminal configured in Step 2 into the corresponding SIP trunk group. Refer to 'VoIP Settings → SIP Trunk Group' for detailed instructions. Select the SIP trunk configured in Step 2 as 'SIP Trunks'. You may use the default values for the other configuration items.

**Example:** Add **SIP Trunk Group 0**. Check the checkbox before **0** for **SIP Trunks** and keep the default values for the other configuration items.

Step 4: Add PCM into the corresponding PCM Group. Refer to 'PCM Settings → PCM Trunk Group' for detailed instructions. Select the PCM used for call conversation as 'PCM'. You may use the default values for the other configuration items.

**Example:** Provided the PCM used for call conversation is PCM[1]. Add **PCM Trunk Group 0**, check the checkbox before **PCM[1]** and keep the default values for the other configuration items.

Step 5: Add routing rules. Refer to 'Route Settings → PSTN→IP' for detailed instructions. Select the PCM trunk group set in Step 4 as 'Call Initiator' and the SIP trunk group set in Step 3 as 'Call Destination'. You may use the default values for the other configuration items.

**Example:** Select **PCM Trunk Group[0]** as **Call Initiator** and **SIP Trunk Group[0]** as **Call Destination**. Keep the default values for the other configuration items.

Step 6: Once PCM[1] receives a call from PSTN and the called party number conforms to the number-receiving rules set in Step 1, it can establish a call conversation with the remote SIP terminal via the gateway.

**Example:** Once PCM[1] receives a call from PSTN with the called party number 123, it will route the call to SIP Trunk 0 of the gateway.

## Special Instructions:

- The chassis of the SMG digital 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 1-4) are never jammed.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our

technicians for help. Otherwise it may lead to a drop in performance or unexpected errors.

# Chapter 3 WEB Configuration

## 3.1 System Login

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


Figure 3-1 Login Interface

The gateway only serves one user, whose original username and password are both ‘admin’. You can change the username and the password via ‘System Tools → Change Password’ on the WEB interface. For detailed instructions, refer to 3.12.15 Change Password.

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


Figure 3-2 Main Interface

## 3.2 Operation Info

Operation Info includes four parts: ***System Info***, ***PSTN Status***, ***SS7 Server*** and ***Call Count***, showing the current running status of the gateway. See Figure 3-3.

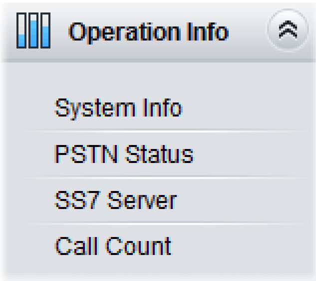



Figure 3-3 Operation Info

### 3.2.1 System Info


Figure 3-4 System Info Interface

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

<table>
<tr><th>Item</th><th>Description</th></tr>
<tr><td>MAC Address</td><td>MAC address of LAN 1 or LAN 2.</td></tr>
<tr><td>IP Address</td><td>The three parameters from left to right are IP address, subnet mask and default gateway of LAN 1 or LAN 2.</td></tr>
<tr><td>DNS Server</td><td>DNS server address of LAN 1 or LAN 2.</td></tr>
<tr><td>Receive Packets</td><td>The amount of receive packets after the gateway's startup, including three categories: All, Error and Drop.</td></tr>
<tr><td>Transmit Packets</td><td>The amount of transmit packets after the gateway's startup, including three categories: All, Error and Drop.</td></tr>
<tr><td>Current Speed</td><td>The current speed of data receiving and transmitting.</td></tr>
<tr><td>Work Mode</td><td>The work mode of the network, including five options: 10 Mbps Half Duplex, 10 Mbps Full Duplex, 100 Mbps Half Duplex, 100 Mbps Full Duplex and 1000 Mbps Full Duplex.</td></tr>
<tr><td>Runtime</td><td>Time of the gateway keeping running normally after startup. This parameter updates every 2s.</td></tr>
<tr><td>Operating Mode</td><td>The operating mode of the gateway includes:
<table>
<tr><th>Operating Mode</th><th>Description</th></tr>
<tr><td>Master Server</td><td>The current gateway applies the SS7 protocol and is used for both signaling and voice transmission. If the dual gateway feature is enabled, the current gateway serves as the master server.</td></tr>
<tr><td>Slave Server</td><td>The current gateway applies the SS7 protocol and is used for both signaling and voice transmission. This operating mode works only when the dual gateway feature is enabled and the current gateway serves as the slave server.</td></tr>
<tr><td>Client</td><td>The current gateway applies the SS7 protocol and is only used for voice transmission.</td></tr>
<tr><td>ISDN(User-side)</td><td>The current gateway is configured to be ISDN user-side..</td></tr>
<tr><td>ISDN(Network-side)</td><td>The current gateway is configured to be ISDN network-side.</td></tr>
<tr><td>SS1</td><td>The current gateway is configured to be SS1.</td></tr>
</table></td></tr>
<tr><td>Serial Number</td><td>Unique serial number of an SMG digital gateway.</td></tr>
<tr><td>WEB</td><td>Current version of the WEB interface.</td></tr>
<tr><td>Gateway</td><td>Current version of the gateway service.</td></tr>
<tr><td>Uboot</td><td>Current version of Uboot.</td></tr>
<tr><td>Kernel</td><td>Current version of the system kernel on the gateway.</td></tr>
<tr><td>Firmware</td><td>Current version of the firmware on the gateway.</td></tr>
</table>

### 3.2.2 PSTN Status

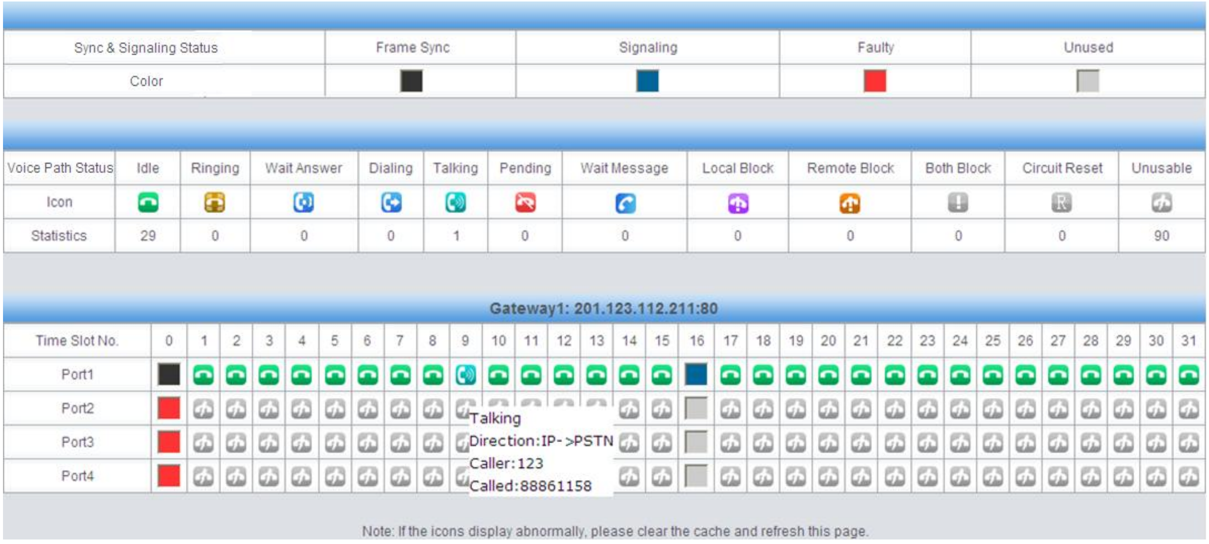



Figure 3-5 PSTN Status Interface for E1 Lines

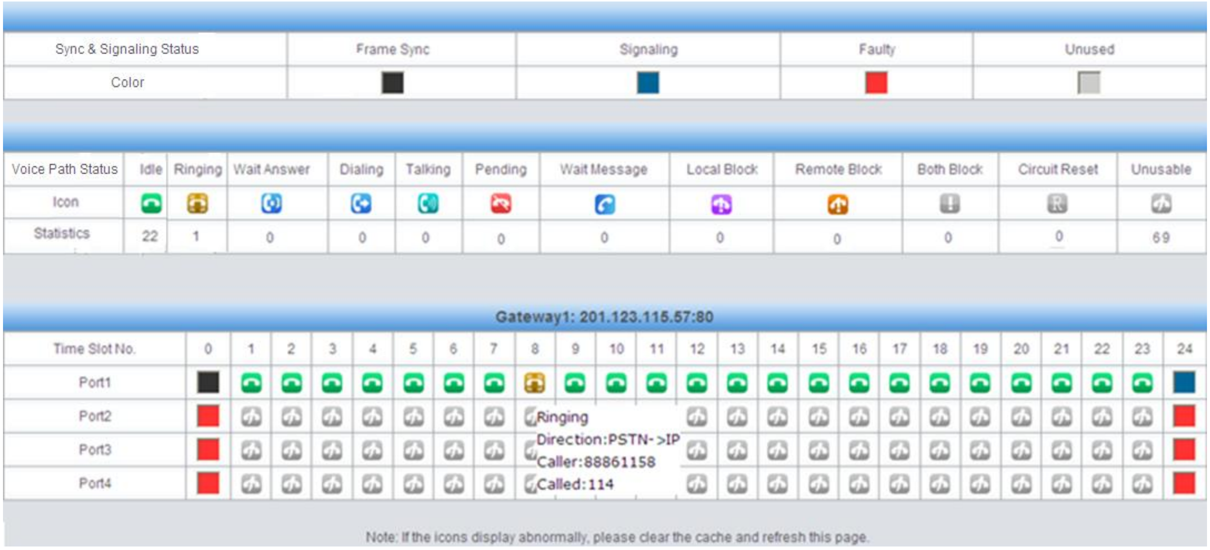



Figure 3-6 PSTN Status Interface for T1 Lines

See Figure 3-5 and Figure 3-6 for the PSTN status interface which shows the real-time status of each PCM on the gateway, including line synchronization, signaling link information and channel states.

| Item | Description |
|---|---|
| ***Port*** | Serial number of the E1/T1 port on the device. |
| ***Time Slot No.*** | PCM time slot number in the port. |
| ***State*** | Displays the channel state in real time. You can move the mouse onto the channel state icon for detailed information about the channel and the call, such as: call direction, calling party number and called party number.<br>● For Time Slot 0, the channel state indicates the synchronization status of E1/T1.<br>**State** \| **Color** \| **Description** |

| | | |
|---|---|---|
| *Frame Sync* | | Frame synchronization normal. The synchronization status is 0x0. |
| *Faulty* | | Configuration errors or hardware failure.<br>You can move the mouse onto the icon for the hexadecimal value for synchronization status which consists of 16 bits and bit 0 is the lowest valid bit. If the bit value is equal to 0, it indicates that the synchronization status is normal; if the bit value is equal to 1, see below for details:<br>bit0=1: basic frame synchronization loss<br>bit1=1: duration of the basic frame synchronization loss exceeds 100ms<br>bit2=1: CAS re-synchronization<br>bit3=1: CRC re-synchronization<br>bit4=1: remote alarm indication<br>bit5=1: signal alarm indication<br>bit6=1: all-ones alarm signal of time slot 16<br>bit7=1: signal loss<br>bit9=1: MF alarm from the remote end<br>bit10=1: open circuit<br>bit11=1: short circuit<br>Other bits: reserved, all remain 0 |

- For the signaling time slot, the channel states include:

| State | Color | Description |
|---|---|---|
| *Signaling* | | For SS7, this state indicates ‘SS7 in service’.<br>For ISDN, this state indicates ‘multiple frames established’ or ‘timer recovery’.<br>For SS1, this state indicates ‘time slot synchronization normal’. |
| *Faulty* | | Configuration errors or hardware failure.<br>For SS7, this state indicates ‘SS7 out of service’, ‘initial alignment’, ‘aligned ready’, ‘aligned not ready’ or ‘processor outage’.<br>For ISDN, this state indicates ‘TEI unassigned’, ‘assign awaiting TEI’, ‘establish awaiting TEI’, ‘TEI assigned’, ‘awaiting establishment ‘or ‘awaiting release’.<br>For SS1, this state indicates ‘time slot synchronization abnormal’. |
| *Unused* | | This state indicates the signaling time slot on this E1/T1 is not used. |

- For the other channels, the channel states include:

| State | Icon | Description |
|---|---|---|
| *Unusable* | | The channel is unavailable. |

| | | | |
|---|---|---|---|
| | *Circuit Reset* | | The circuit is being reset. |
| | *Idle* | | The channel is available. |
| | *Local Block* | | The channel is blocked by the local application program and cannot receive incoming calls. |
| | *Remote Block* | | The channel is blocked by the specific circuit/circuit group blocking messages sent from the remote PBX and cannot make outgoing calls. |
| | *Both Block* | | The channel is blocked by the local end so as not to receive incoming calls, meanwhile, it is blocked by the remote PBX so as not to make outgoing calls either. |
| | *Wait Answer* | | The channel receives the ringback tone and is waiting for the called party to pick up the phone. |
| | *Ringing* | | The channel is in the ringing state. |
| | *Talking* | | The channel is in a conversation. |
| | *Pending* | | The channel is in the pending state |
| | *Dialing* | | The channel is dialing. |
| | *Wait Message* | | The channel is waiting for the message from remote PBX. |
| ***Statistics*** | The total amount of the channels for the corresponding status. | | |

**Note:** The gateway provides the fuzzy search feature on this interface. After you click any characters on Figure 3-5, Figure 3-6, and press the ‘F’ button, the search box will emerge on the right top of this page. Then you can input the key characters and the gateway will locate the channel on which there is an ongoing call that conforms to the fuzzy search condition.

Take an example: As shown in Figure 3-7, after we input the character 888 to the search box, and click the ***Search*** button, the gateway does a fuzzy search and locates that the ongoing call whose CallerID contains the character 888 occurs on Channel 9.

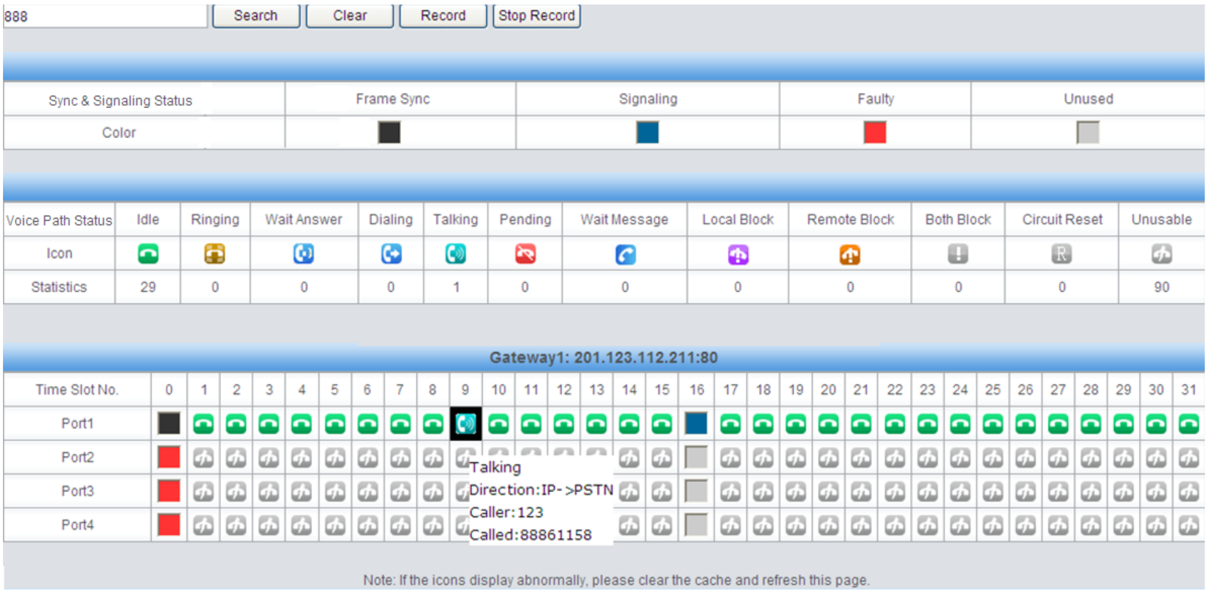



Figure 3-7 Search Calls

**Note:** Click ***Record*** to start recording on the matched channel. If more than one channels match a condition, only the channel with the largest number among them will be recorded.

## 3.2.3 SS7 Server

Users can see the SS7 Server option in the menu only when the configuration item ***Signaling Protocol*** on the PCM settings interface is set to *SS7-TUP* or *SS7-ISUP*.

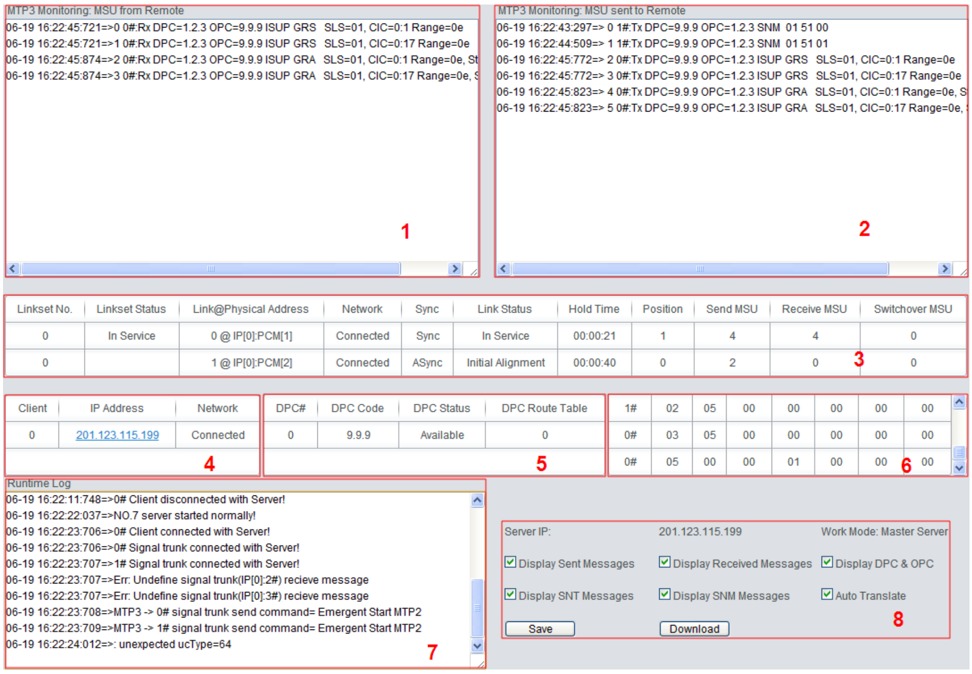



Figure 3-8 SS7 Server Info Interface

See Figure 3-8 for the SS7 server info interface. This interface contains 7 status bars (Status Bar 1~7 in the above figure) and a configuration region (Region 8 in the above figure). Below are the detailed introductions.

- **Status Bar 1 & 2: Receive/transmit message list**

The receive/transmit message lists display the received and sent messages respectively, used for gateway debugging. The display content in these lists can be set by the configuration items in Region 8.

- **Configuration Region 8: Properties configuration for receive/transmit message list**

The table below explains the items in Configuration Region 8.

| Item | Description |
|---|---|
| ***Server IP*** | IP address of the SS7 server, this item can be configured on the SS7 interface. |
| ***Work Mode*** | Work mode of the SS7 server which includes three modes: Master Server, Slave Server and Client. |
| ***Display Sent Messages*** | If this item is ticked, the transmit message list will display the message sent to the remote end. |
| ***Display Received Messages*** | If this item is ticked, the receive message list will display the message received from the remote end. |
| ***Display DPC & OPC*** | If this item is ticked, the receive/transmit message list will display DPC and OPC. |
| ***Display SNT Messages*** | If this item is ticked, the receive/transmit message list will display the SNT messages. |
| ***Display SNM Messages*** | If this item is ticked, the receive/transmit message list will display the SNM messages. |

| | |
|---|---|
| ***Auto Translate*** | If this item is ticked, the received/sent messages displayed on this interface will be translated automatically in the following format:<br>Date Time Total number Signaling link number# SIO Content<br>For the TUP messages, SIO is just 'TUP' (0x84), followed by the message content. It is usually in the following format:<br>Title code CIC=PCM:TS Message body<br>If this item is not ticked, the received/sent messages displayed on this interface will be hexadecimal raw data. |

Users can configure the display content of the receive/transmit message list via the checkbox before each configuration item. After modification, click ***Save*** to apply the configurations. The changes will be shown in the list in real time. Click ***Download*** and you can download the log information of the SS7 server.

- **Status Bar 3: Linkset/signaling link information**

This region displays the information about signaling links and linksets. The table below explains the information items in Status Bar 3.

| Item | Description |
|---|---|
| ***Linkset No.*** | Linkset number. |
| ***Linkset Status*** | Working state of the linkset, including *In service* and *Out of service*. A signaling linkset will go into the state *In service* as long as one link in it is at the state of *In service*. |
| ***Link@Physical Address*** | Signaling link number and its physical position. For example, '0 @ IP[0]:PCM[0]' means the physical position of Link 0 in this gateway is the E1 with the local PCM numbered 0 on Client 0. |
| ***Network*** | Whether the signaling link is registered to the gateway, including two states: *Connected* and *Disconnected* (or no display). The signaling link can be used normally only in the state of *Connected*. |
| ***Sync*** | Basic frame synchronization (Time Slot 0), including two states: *Sync* and *Async*. The signaling link can be used only in the state of *Sync*. |
| ***Link Status*** | Working state of the signaling link, including *In service* and *Initial alignment*. You can refer to 'Status Bar 6: Link information' for detailed information about link status. |
| ***Hold Time*** | Duration since the last time the signaling link enters into the state of *In service*. |
| ***Position*** | Times of positioning that occurs on the signaling link since the program starts. |
| ***Send MSU*** | Total number of messages sent on the signaling link since the program starts. |
| ***Receive MSU*** | Total number of messages received on the signaling link after the program starts. |
| ***Switchover MSU*** | Total number of messages switched over on the signaling link since the program starts. |

- **Status Bar 4: Client information**

This region displays the information about client IP address and connection state. The table below explains the information items in Status Bar 4.

| Item | Description |
|---|---|
| ***Client*** | Client number. |

| | |
|---|---|
| ***IP Address*** | IP address of the client. You can click the link of the IP address to visit the WEB interface of the client. |
| ***Network*** | Whether the client has been successfully connected to the gateway, including two states: *Connected* and *Disconnected* (or no display). |

- **Status Bar 5: DPC Information**

This region displays the information about DPC. The table below explains the information items in Status Bar 5.

| Item | Description |
|---|---|
| ***DPC#*** | DPC number which starts from 0. |
| ***DPC Code*** | Destination point code which is usually allocated by the central office. |
| ***DPC Status*** | Indicates whether the route to this DPC is available, involving two states *Available* and *Unavailable*. The message can be sent to the DPC only when the route to this DPC is at the state of *Available*. The DPC will turn into the state of *Available* as long as one of the linksets reaching the DPC is at the state of *In Service*. |
| ***DPC Route Table*** | Route to the DPC, i.e. linkset number. |

- **Status Bar 6: Link information**

This status bar displays the detailed information on the state of all signaling links, usually used for searching the cause of service interrupt on a signaling link.

| Link# | STA | L2 | POC | LSC | FSN | ERR | CHO |
|---|---|---|---|---|---|---|---|
| Link Number | Link States 0-6 | Link Failure Causes (interrupt) | Processor Failures 0-3 | Live Communication Server Service 0-1 | Forward Sequence Number | spare | spare |
| | 0: uploaded but not started | 0: normal | 0: normal | 0: service is unavailable | | | |
| | 1: service interrupt | 1: BSNR illegal | 1: the local end processor failure | 1: service is available | | | |
| | 2: initial positioning | 2: FIBR illegal | 2: the remote end processor failure | | | | |
| | 3: positioned/ ready | 3: T2 timeout | 3: both ends processor failure | | | | |
| | 4: positioned/ not ready | 4: T6 timeout, the remote end busy | | | | | |

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| | 5: service on | 5: L3 sends a command to stop | | | | | |
| | 6: processor failure | 6: signaling error rate too high | | | | | |
| | | 7: during the course of initial positioning, fail to enter a normal position | | | | | |
| | | 8: Timer 1 timeout | | | | | |
| | | 9: positioned and ready, receive the interrupt signal of the remote end | | | | | |
| | | 10: positioned but not ready, receive the interrupt signal of the remote end | | | | | |
| | | 11: in the state of Service On, receive the interrupt signal of the remote end | | | | | |
| | | 12: in a processor failure, receive the interrupt signal of the remote end | | | | | |

- **Status Bar 7: Runtime Log**

Runtime log records all MTP3 commands and error information that pops up during the operation. This status bar displays all the log records generated after the digital gateway starts.

### 3.2.4 Call Count

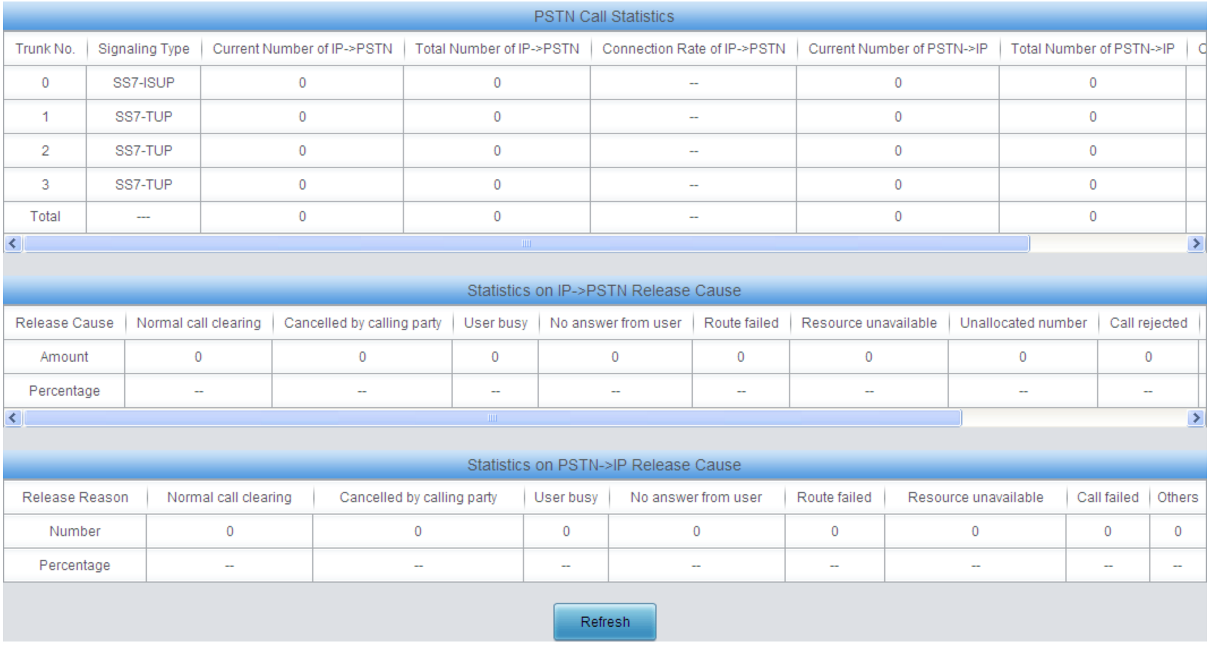



Figure 3-9 Call Count Interface

See Figure 3-9 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. This interface includes three parts: PSTN Call Statistics, Statistics on PSTN Release Cause and Statistics on Sip Release Cause. You can click ***Refresh*** to obtain the latest call count information. The table below explains the items shown in Figure 3-9.

| Item | Description |
|---|---|
| ***Trunk No.*** | The number of the PCM trunk, numbered from 0. |
| ***Signaling Type*** | The signaling protocol applied on the digital trunk, including: *ISDN User Side*, *ISDN Network Side*, *SS7-TUP*, *SS7-ISUP*, and *SS1*. |
| ***Current Number of IP→PSTN*** | The number of current calls from IP to PSTN. |
| ***Total Number of IP→PSTN*** | The total number of current calls from IP to PSTN. |
| ***Connection Rate of IP→PSTN*** | The percentage of successful IP→ PSTN calls to total IP→ PSTN calls. |
| ***Current Number of PSTN →IP*** | The number of current calls from PSTN to IP. |
| ***Total Number of PSTN →IP*** | The total number of current calls from PSTN to IP. |
| ***Connection Rate of PSTN →IP*** | The percentage of successful PSTN → IP calls to total PSTN → IP calls. |
| ***Total*** | Total number and connection rate of calls on all available tunks |
| ***Release Cause*** | Reason to release the call. |
| ***Normal call clearing*** | Total number of the calls which are normally cleared. |

| | |
|---|---|
| ***Cancelled by calling party*** | Total number of the calls which are cancelled by the calling party. |
| ***User busy*** | Total number of the calls which fail as the called party has been occupied and replies a busy message. |
| ***No answer from user*** | Total number of the calls which fail as the called party does not pick up the call in a long time or the calling party hangs up the call before the called party picks it up. |
| ***Routing failed*** | Total number of the calls which fail because no routing rules are matched. |
| ***Resource unavailable*** | Total number of the calls which fail because no voice channel is available. |
| ***Unallocated number*** | Total number of the calls which fail as the called party number is unallocated. |
| ***Call rejected*** | Total number of the calls which fail as the called party replies a rejection message. |
| ***Normal unspecified*** | Total number of the calls which fail as the called party number is normal but unspecified. |
| ***Call failed*** | Total number of the calls which fail as the called party number does not conform to the number-receiving rule or for relative reasons. |
| ***Others*** | Total number of the calls which fail due to other unknown reasons. |
| ***Percentage*** | The percentage of the calls with a release cause to total calls. |

## 3.3 VoIP Settings

VoIP Settings includes five parts: ***SIP***, ***SIP Trunk***, **SIP Register**, ***SIP Account***, ***SIP Trunk Group*** and ***Media***. See Figure 3-10. ***SIP*** is used to configure the general SIP parameters; ***SIP Trunk*** is used to set the basic and register information of the SIP trunk; ***SIP Register*** is used for the registration of SIP; ***SIP Account*** is used for registering SIP accounts to the SIP server; ***SIP Trunk Group*** is to manage SIP trunks by group; and ***Media*** is to set the RTP port and the payload type.


Figure 3-10 VoIP Settings

### 3.3.1 SIP Settings

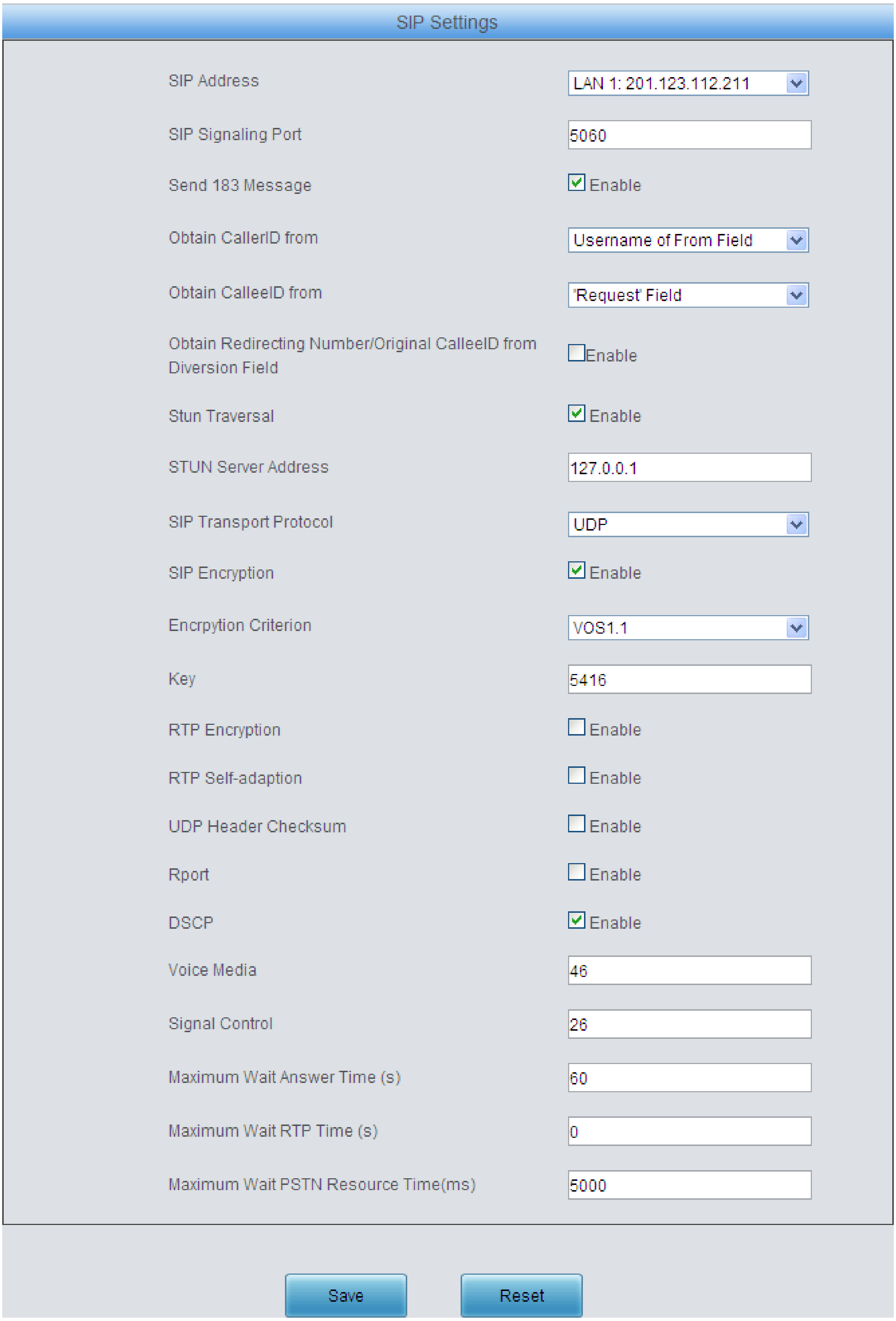


Figure 3-11 SIP Settings Interface

See Figure 3-11 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 3.12.17 Restart for detailed instructions. The table below explains the items shown in Figure 3-11.

| Item | Description |
|---|---|
| ***SIP Address*** | IP address for SIP signaling, using LAN 1 by default. |
| ***SIP Port*** | Monitoring port of SIP signaling. Range of value: 1024~65535, with the default value of 5060. |
| ***183 Message Behavior*** | Sets whether to send the 183 message instead of 180 to respond to the ringing tone when the SIP end serves as the called party. By default this feature is enabled. |
| ***Obtain CallerID from*** | There are two optional ways to obtain the calling party number: from *Username of "From" Field* or from *Displayname of "From" Field*. The default value is from *Username of "From" Field*. |
| ***Obtain CalleeID from*** | There are two optional ways to obtain the called party number: from *"To" Field* or from *"Request" Field*. The default value is from *"Request" Field*. |
| ***Obtain Redirecting Number/Original CalleeID from Diversion Field*** | Sets whether to enable the feature of obtaining the Redirecting Number/Original CalleeID from Diversion Field. By default, the feature is disabled. |
| ***STUN Traversal*** | Sets whether to enable the STUN server for NAT traversal. By default the STUN server is disabled. |
| ***STUN Server Address*** | Address of the server for STUN traversal. |
| ***SIP Transport Protocol*** | There are two modes *UDP* and *TCP* available for running the SIP protocol. The default value is *UDP*. |
| ***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. |
| ***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*. |
| ***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*. |
| ***UDP Header Checksum*** | When this feature is enabled, the gateway will automatically calculate the check sum of the UDP header during RTP transmission. |
| ***Rport*** | When this feature is enabled, a corresponding Rport field will be added to the Via message of SIP. By default, it is *disabled*. |
| ***DSCP*** | Sets whether to enable the DSCP differentiated services code point. By default, it is *disabled*. |

| | |
|---|---|
| ***Voice Media*** | Sets the priority of the voice media for DSCP. The voice media with a bigger value has a higher priority. The value range is 0~63, with the default value of *46*. |
| ***Signal Control*** | Sets the priority of the signal control for DSCP. The signal control with a bigger value has a higher priority. The value range is 0~63, with the default value of *26*. |
| ***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. |
| ***Maximum Wait RTP Time*** | Sets the maximum time for the SIP channel to wait for the RTP packet. If no RTP packet is received within the specified time period, the channel will enter the pending state automatically and release the call. The default value is *0*, calculated by s. |
| ***Maximum Wait PSTN Resource Time*** | Sets the maximum wait time to search the idle PSTN resource for the incoming call from IP. The call will be failed if no channel is found during this time. The value range is 0~10000, calculated by ms, with the default value of *5000*. |

## 3.3.2 SIP Trunk

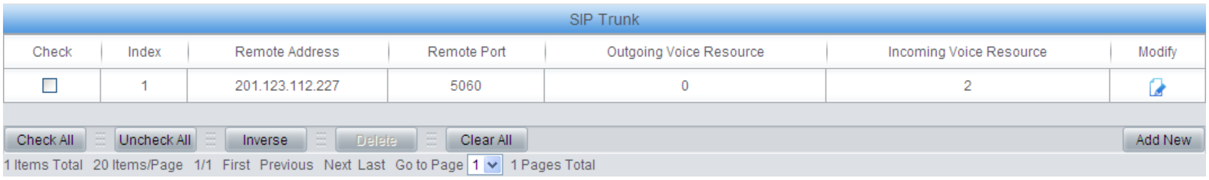



Figure 3-12 SIP Trunk Settings Interface

See Figure 3-12 for the SIP trunk settings interface. A new SIP trunk can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-13 for the SIP trunk adding interface.

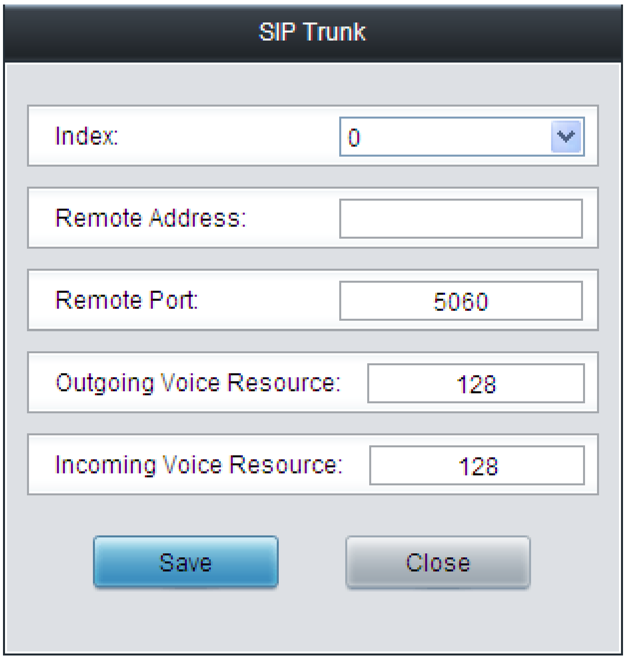



Figure 3-13 Add New SIP Trunk

The table below explains the items shown in Figure 3-13.

| Item | Description |
|---|---|
| ***Index*** | The unique index of each SIP trunk. |
| ***Remote Address*** | Address of the SIP trunk, i.e. the IP address or domain name of the remote SIP terminal which will establish call conversation with the gateway. |
| ***Remote Port*** | Port of the SIP trunk. |
| ***Outgoing Voice Resource*** | Maximum number of voice channels for the outgoing calls allocated by the SIP trunk to the gateway. |
| ***Incoming Voice Resource*** | Maximum number of voice channels for the Incoming calls allocated by the SIP trunk to the gateway. |

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

Click ***Modify*** in Figure 3-12 to modify a SIP trunk. See Figure 3-14 for the SIP trunk modification interface. The configuration items on this interface are the same as those on the ***Add New SIP Trunk*** interface.

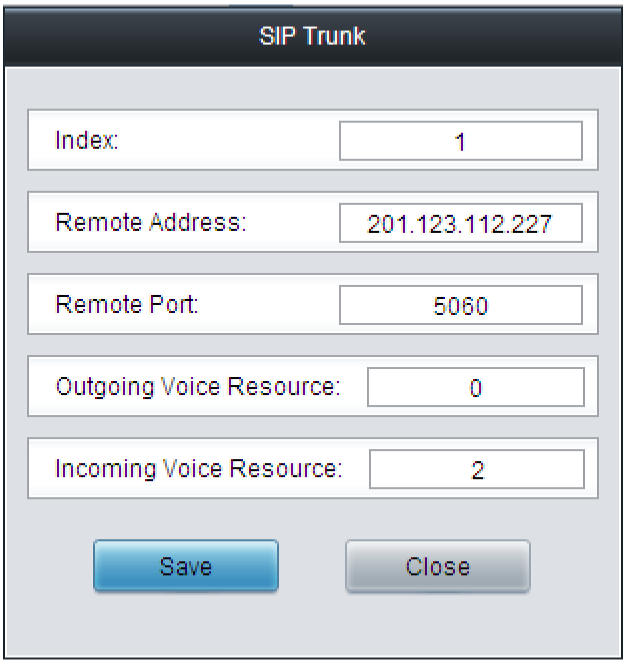



Figure 3-14 Modify SIP Trunk

To delete a SIP trunk, check the checkbox before the corresponding index in Figure 3-12 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 trunks at a time, click the ***Clear All*** button in Figure 3-12.

## 3.3.3 SIP Register

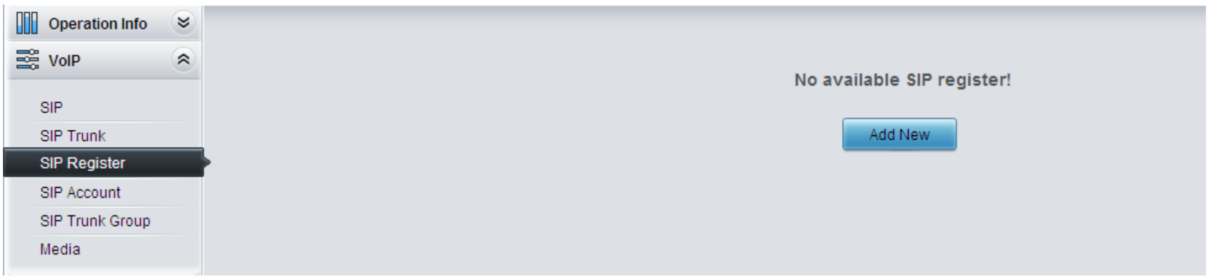


Figure 3-15 SIP Register Configuration Interface

See Figure 3-15 for the SIP Register Configuration interface. By default, there is no SIP register available on the gateway. Click ***Add New*** to add them manually. See Figure 3-16.

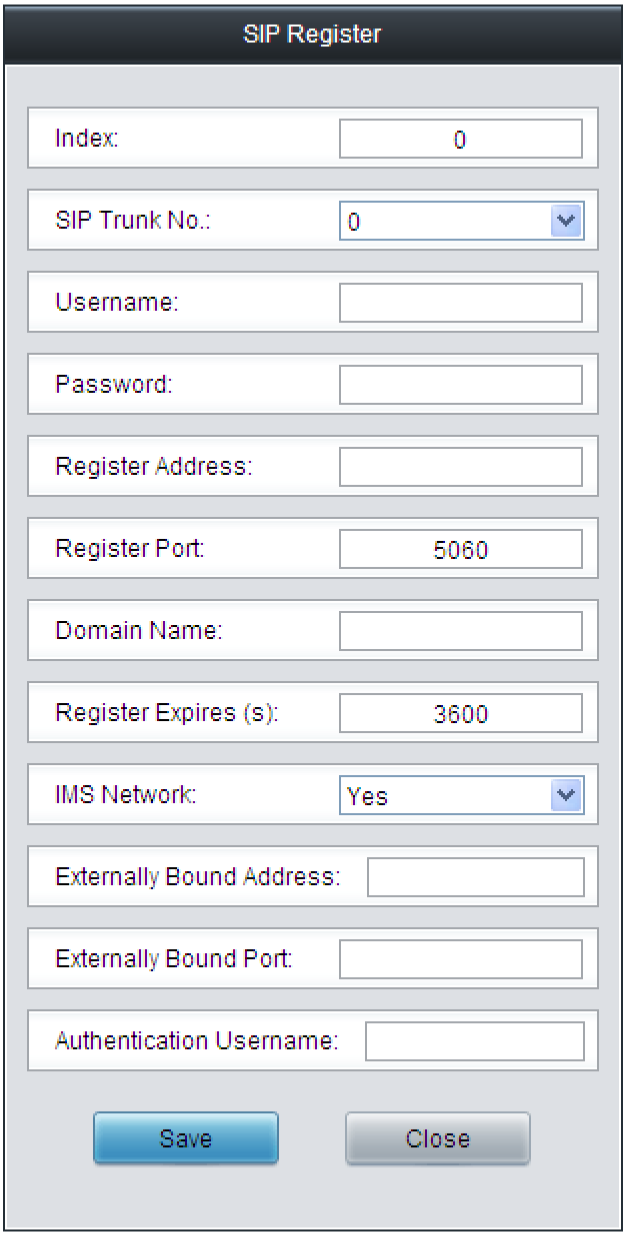



Figure 3-16 Add SIP Register Interface

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

| Item | Description |
|---|---|
| ***Index*** | The unique index of each SIP register. |
| ***SIP Trunk No.*** | The number of the SIP trunk which registers to the SIP server. |
| ***Username*** | When the gateway initiates a call to SIP, this item corresponds to the username of SIP; when the gateway initiates a call to PSTN, this item corresponds to the displayed CallerID. |
| ***Password*** | Registration password of the gateway. To register the gateway to the SIP server, both configuration items ***Username*** and ***Password*** should be filled in. |
| ***Register Address*** | Address of the SIP server to which the SIP trunk is registered. |

| | |
|---|---|
| ***Register Port*** | The signaling port of the SIP trunk. |
| ***Domain Name*** | Domain name of the gateway used for SIP registry. |
| ***Register Expires*** | Validity period of the SIP registry. Once the registry is overdue, the gateway should be registered again. Range of value: 10~3600, calculated by s, with the default value of 3600. |
| ***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. 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. |
| ***Authentication Username*** | Authentication username for registration. |

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

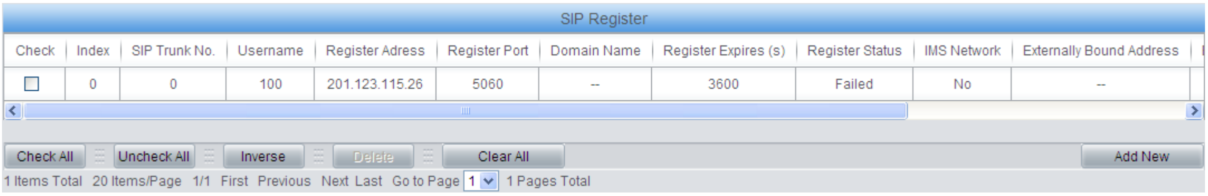



Figure 3-17 SIP Register Information List

Click ***Modify*** in Figure 3-17 to modify a SIP register. The configuration items on the SIP Register Modification Interface are the same as those on the ***Add New SIP Register*** interface.

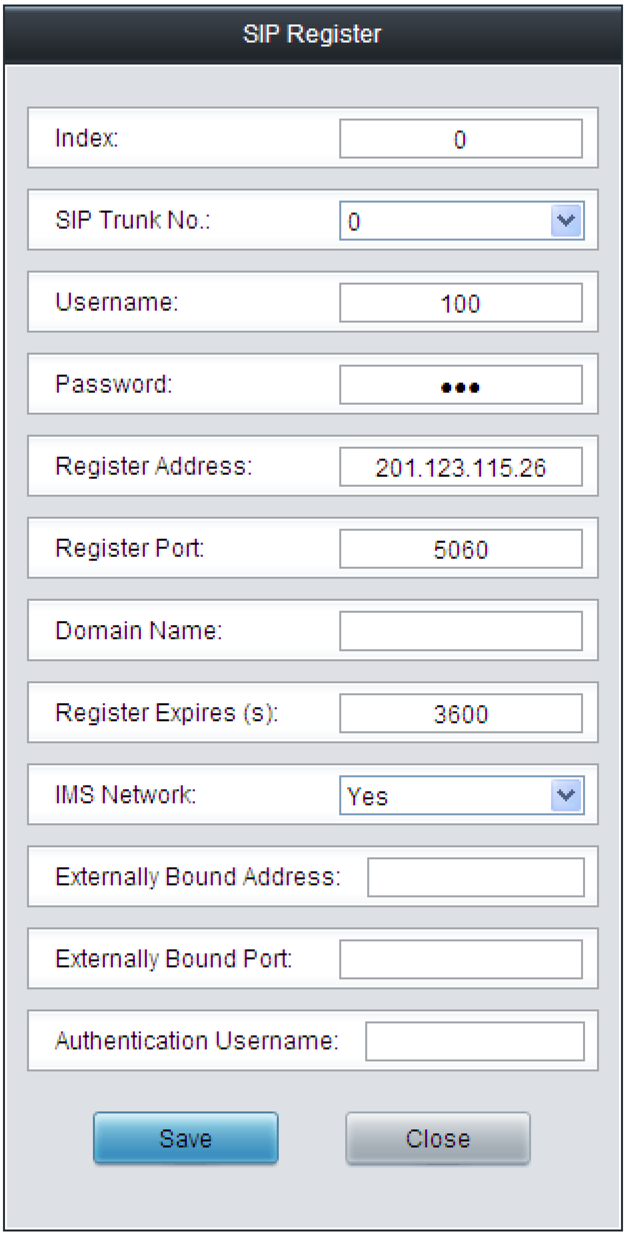



Figure 3-18 SIP Register Modification Interface

To delete a SIP register, check the checkbox before the corresponding index in Figure 3-17 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 registers at a time, click the ***Clear All*** button in Figure 3-17.

## 3.3.4 SIP Account

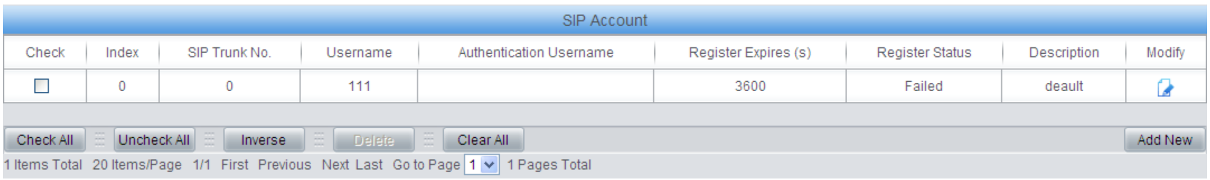



Figure 3-19 SIP Account Settings Interface

See Figure 3-19 for the SIP account settings interface. A new SIP account can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-20 for the

SIP account adding interface.

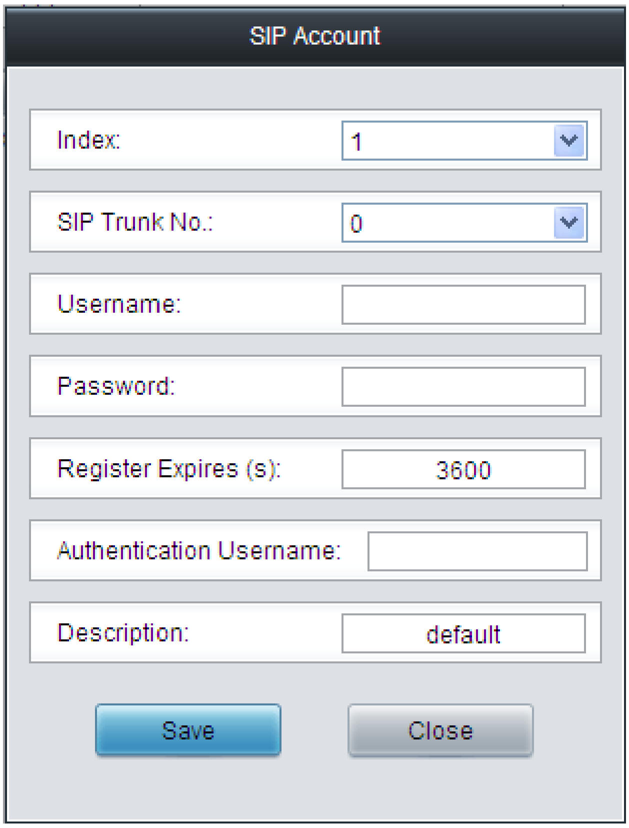



Figure 3-20 Add New SIP Account

The table below explains the items shown in above figures.

| Item | Description |
|---|---|
| ***Index*** | The unique index of each SIP account. |
| ***SIP Trunk No.*** | The number of the SIP trunk to which the SIP account is registered. |
| ***Username*** | The registration username of the SIP account. Once the SIP account is successfully registered, the SIP server can initiate calls to the gateway via ***Username***. |
| ***Password*** | The registration password of the SIP account. To register the SIP account to the SIP trunk, both configuration items ***Username*** and ***Password*** should be filled in. |
| ***Register Expires*** | The validity period of the SIP account registry. Once the registry is overdue, the SIP account should be registered again. Range of value: 10~3600, calculated by s, with the default value of 3600. |
| ***Register Status*** | The registration status of the SIP account. It is either *Registered* or *Failed*. |
| ***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 on the SIP trunk corresponding to this SIP account.** |
| ***Description*** | More information about each SIP account. |

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

Click ***Modify*** in Figure 3-19 to modify a SIP account. See Figure 3-21 for the SIP account modification interface. The configuration items on this interface are the same as those on the ***Add***

***New SIP Account*** interface.

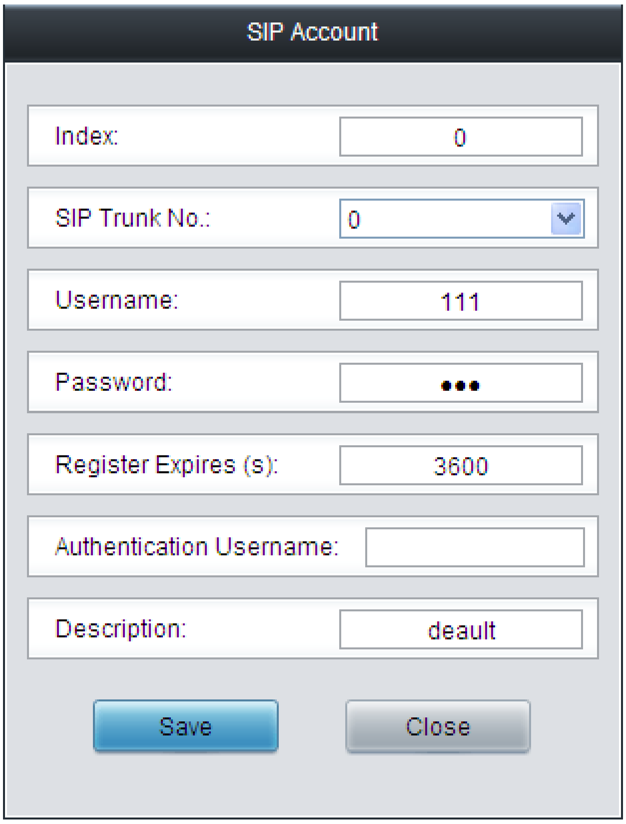



Figure 3-21 Modify SIP Account

To delete a SIP account, 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 accounts at a time, click the ***Clear All*** button in Figure 3-19.

## 3.3.5 SIP Trunk Group

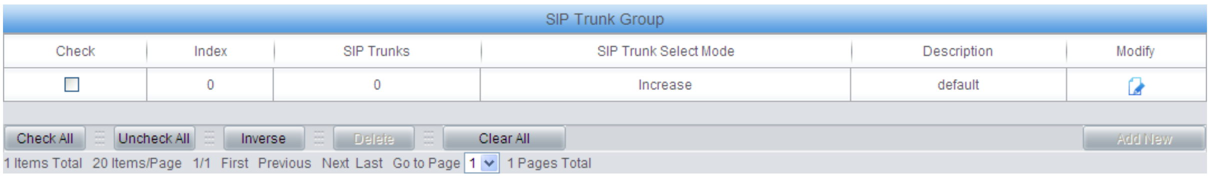



Figure 3-22 SIP Trunk Group Settings Interface

See Figure 3-22 for SIP trunk group settings interface. A new SIP trunk group can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-23 for the SIP trunk group adding interface.

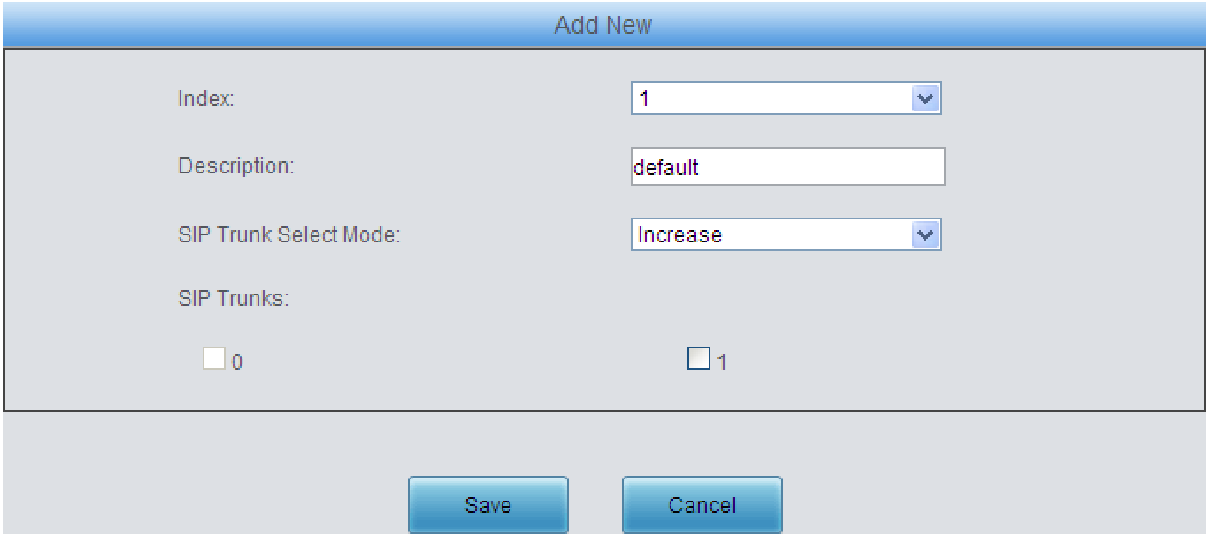



Figure 3-23 Add New SIP Trunk Group

The table below explains the items shown in Figure 3-23.

<table>
<tr><th>Item</th><th>Description</th></tr>
<tr><td>Index</td><td>The unique index of each SIP trunk group, which is mainly used in the configuration of routing rules and number manipulation rules to correspond to SIP trunk groups.</td></tr>
<tr><td>Description</td><td>More information about each SIP trunk group.</td></tr>
<tr><td>SIP Trunk Select Mode</td><td>When the SIP trunk group receives a call, it will choose a SIP trunk based on the select mode set by this configuration item to ring. The optional values and their corresponding meanings are described in the table below.
<table>
<tr><th>Option</th><th>Description</th></tr>
<tr><td>Increase</td><td>Search for an idle SIP trunk in the ascending order of the SIP trunk number, starting from the minimum.</td></tr>
<tr><td>Decrease</td><td>Search for an idle SIP trunk in the descending order of the SIP trunk number, starting from the maximum.</td></tr>
<tr><td>Cyclic Increase</td><td>Provided SIP Trunk N is the available SIP trunk found last time. Search for an idle SIP trunk in the ascending order of the SIP trunk number, starting from SIP Trunk N+1.</td></tr>
<tr><td>Cyclic Decrease</td><td>Provided SIP Trunk N is the available SIP trunk found last time. Search for an idle SIP trunk in the descending order of the SIP trunk number, starting from SIP Trunk N-1.</td></tr>
</table></td></tr>
<tr><td>SIP Trunks</td><td>The SIP trunks in the SIP trunk group. If the checkbox before a SIP trunk is grey, it indicates that the SIP trunk has been occupied. The ticked SIP trunks herein will be displayed in the column 'SIP Trunks' in Figure 3-22.</td></tr>
</table>

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

Click ***Modify*** in Figure 3-22 to modify a SIP trunk group. See Figure 3-24 for the SIP trunk group modification interface. The configuration items on this interface are the same as those on the ***Add New SIP Trunk Group*** interface.

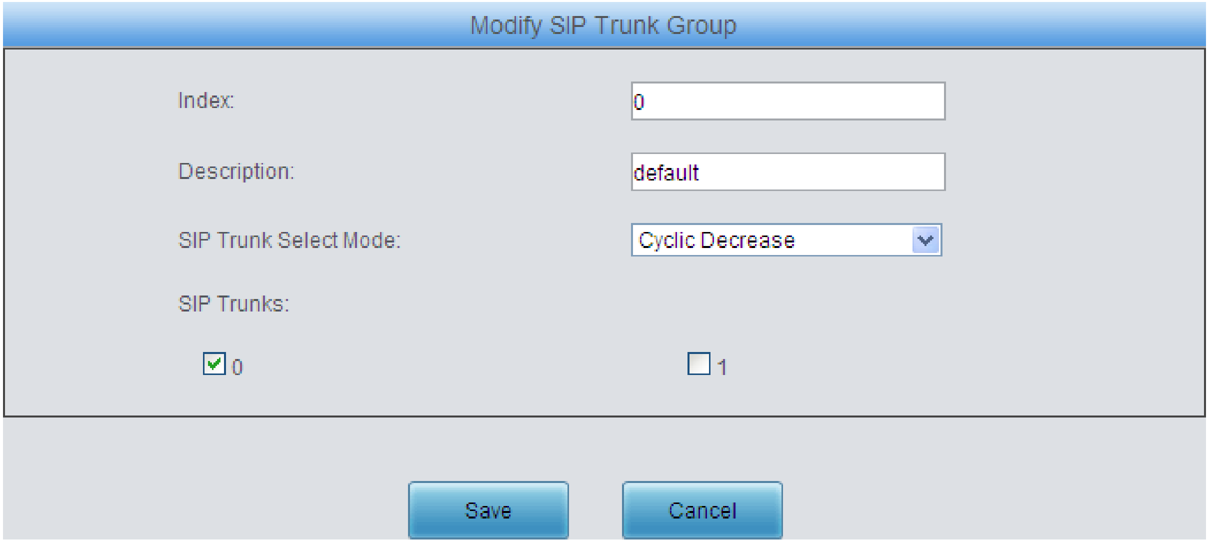



Figure 3-24 Modify SIP Trunk Group

To delete a SIP trunk group, check the checkbox before the corresponding index in Figure 3-22 and click the ***Delete*** button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all SIP trunk groups at a time, click the ***Clear All*** button in Figure 3-22.

### 3.3.6 Media Settings

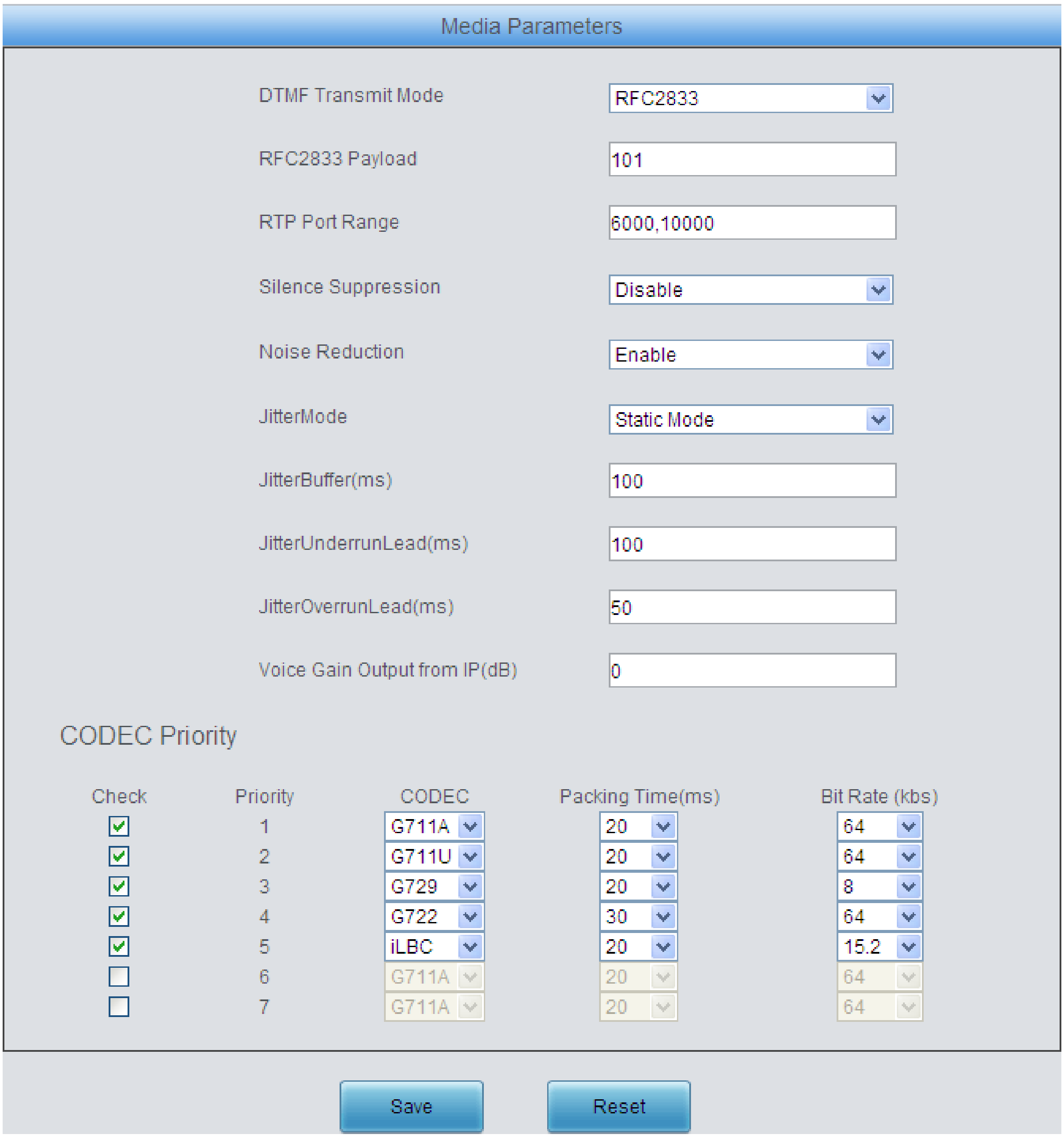


Figure 3-25 Media Settings Interface

See Figure 3-25 for the media settings interface where you can configure the RTP port and payload type depending on your requirements. After configuration, click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to 3.12.17 Restart for detailed instructions. The table below explains the items shown in Figure 3-25.

| Item | Description |
|---|---|
| ***DTMF Transmit Mode*** | Sets the 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 512. 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*.<br>**Note:** When G723 is selected as CODEC, this configuration setting will turn to Enable automatically. |
| ***Noise Reduction*** | Once this feature is enabled, the volume of the noise accompanied with the line will be reduced automatically. The default setting is *Enable*. |
| ***JitterMode*** | Sets the working mode of JitterBuffer. 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 applied to received packets upon accepting packets later than the expected value set in JitterBuffer Item. Rnage of value: 0~280, calculated by ms, with the default value of *100*,<br>**Note:** Only when JitterMode is to *Static Mode* will this item be shown. |
| ***JitterOverrunLead*** | Sets the beforehand time inserted if receiving packets is ahead of time (the time of receiving is earlier than 300 minus the value set in JitterBuffer). Rnage of value: 0~280, calculated by ms, with the default value of *50*,<br>**Note:** Only when JitterMode is 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. Rnage of value: 0~280, calculated by ms, with the default value of 80.<br>**Note:** Only when JitterMode is to *Adaptive Mode* will this item be shown. |
| ***JitterDecreaseRatio*** | Sets the rate of the delay that can be reduced under the adaptive mode. It defines the maximum percentage of silence that can be removed if reducing the delay. Rnage of value: 0~100, with the default value of *50*,<br>**Note:** Only when JitterMode is to *Adaptive Mode* will this item be shown. |
| ***JitterIncreaseMax*** | Sets the maximum delay can be increased during one silence period. Rnage of value: 0~280, calculated by ms, with the default value of *30*,<br>**Note:** Only when JitterMode is to *Adaptive Mode* will this item be shown. |
| ***Voice Gain Output from IP*** | Adjusts the voice gain of call from IP to the remote end. The value must be a multiple of 3. 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:<br>**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* | Seven optional CODECs are supported: *G711A*, *G711U*, *G729AB*, *G723*, *G722*, *AMR* and *iLBC*. |
| *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 seven CODECs are supported and ordered G711A, G711U, G729AB, G723, G722, AMR and iLBC by priority from high to low.

The packing time and bit rate supported by different CODECs are listed in the table below. Those values in bold face are the default values.

| 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** |
| *G729AB* | **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 |

## 3.4 PCM Settings

PCM Settings includes eight parts: ***PSTN***, ***Circuit Maintenance***, ***PCM***, ***PCM Trunk***, ***PCM Trunk Group***, ***Number-Receiving Rule***, ***Reception Timeout*** and ***Number Attribution***. See Figure 3-26.


Figure 3-26 PCM Settings

### 3.4.1 PSTN

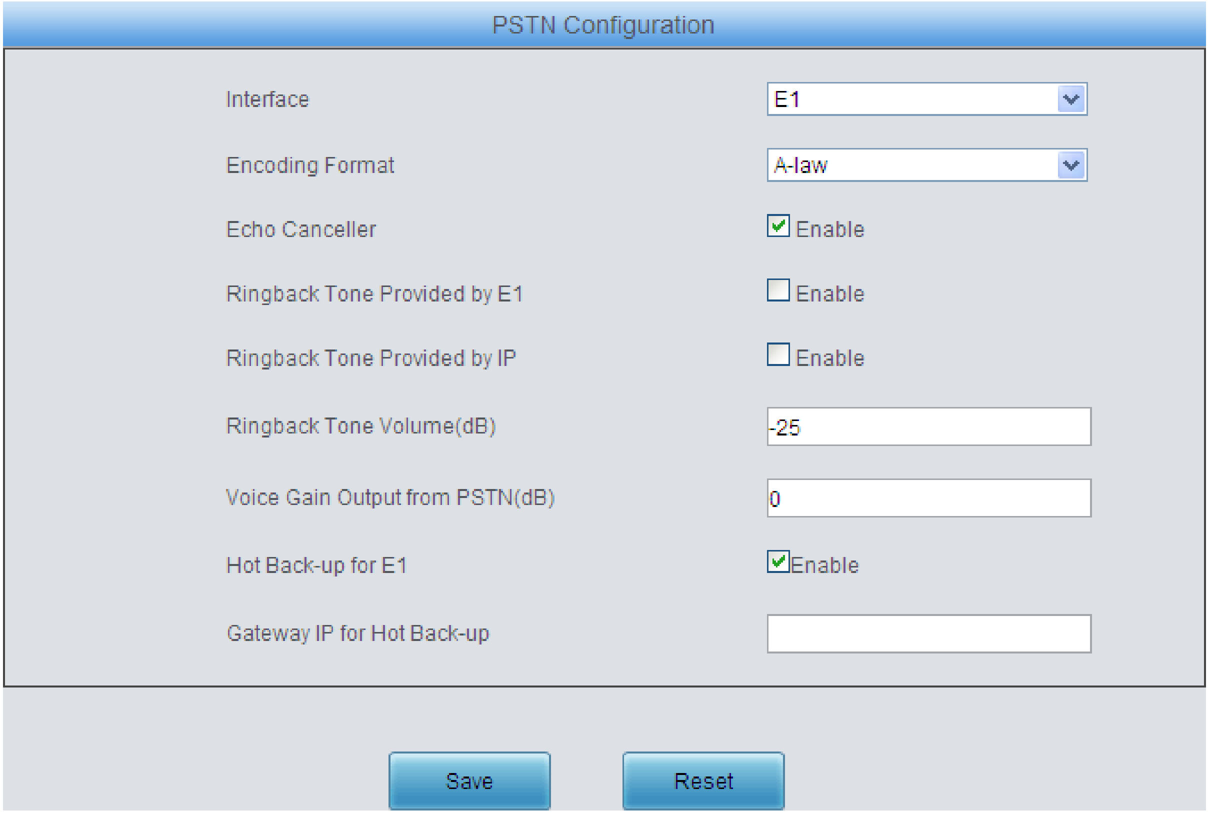



Figure 3-27 PSTN Settings Interface

See Figure 3-27 for the PSTN Settings interface. The table below explains the items shown in the above figure.

| Item | Description |
|---|---|
| ***Interface*** | Actual type of the line connected with the E1/T1 interface on the gateway. Currently, only E1/T1 is supported. |
| ***Encoding Format*** | Sets the voice data encoding format for the voice channels on the digital trunk. The optional values are *A-law* and *u-law*, with the default value of *A-law*. |
| ***Echo Canceller*** | Sets whether to enable the echo cancellation feature for call conversations over the digital trunk. By default, this feature is enabled and the effect can reach 128ms. |
| ***Ringback Tone Provided by E1*** | Sets whether to enable the E1 end to provide the ringback tone, with the default value of *disable*. |
| ***Ringback Tone Provided by IP*** | Sets whether to enable the IP end to provide the ringback tone, with the default value of *disable*. |
| ***Ringback Tone Volume*** | Sets the volume of the ringback tone. Range of value: -35~-2, calculated by dB, with the default value of -25. |
| ***Voice Gain Output from PSTN*** | Adjusts the voice gain of call from PSTN to the remote end. The value must be a multiple of 3. Range of value: -24~24, calculated by dB, with the default value of *0*. |
| ***Hot Back-up for E1*** | Sets whether to enable the feature of hot back-up for E1, with the default value of *disable*. |
| ***Gateway IP for Hot Back-up*** | Set the IP of the gateway for the hot back-up for E1. |

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 3.12.17 Restart for detailed instructions.

## 3.4.2 Circuit Maintenance

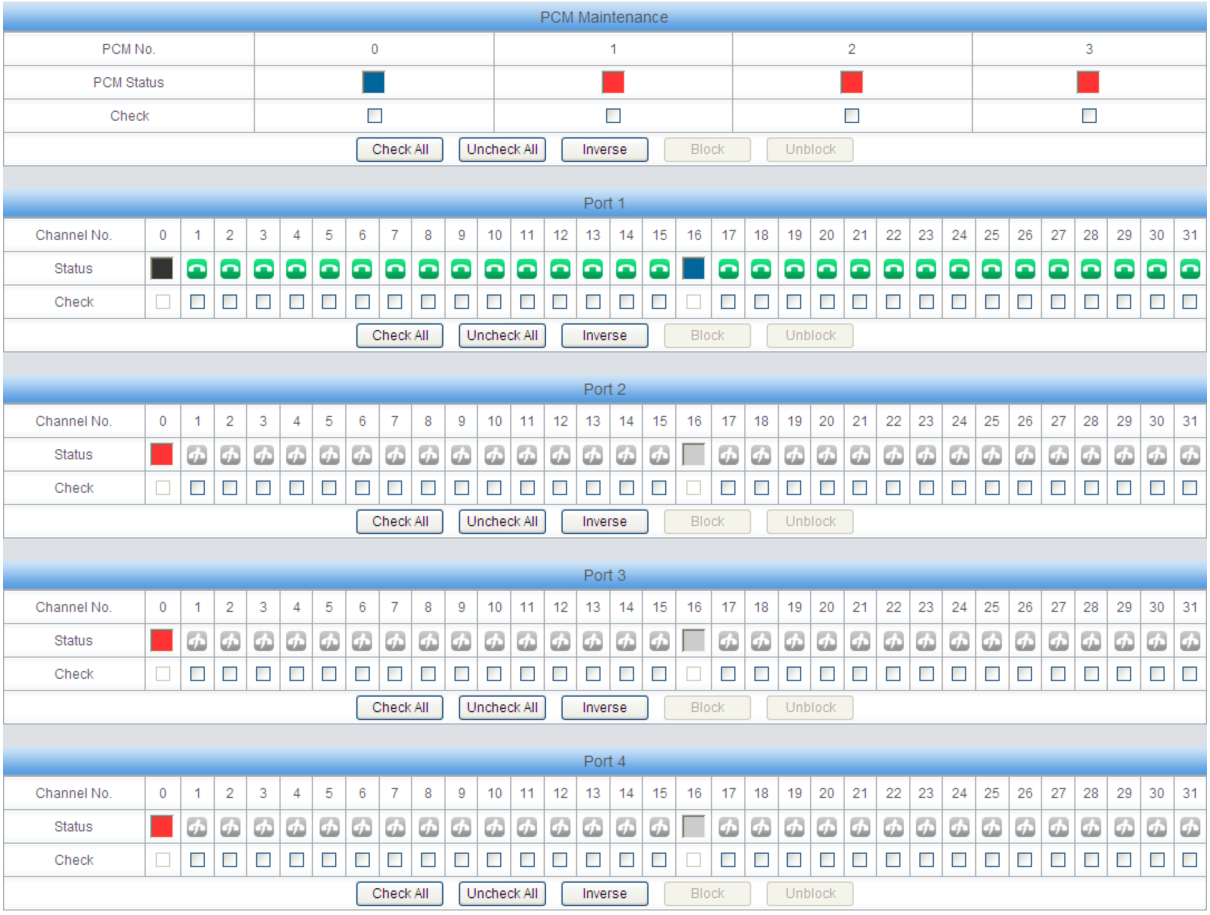
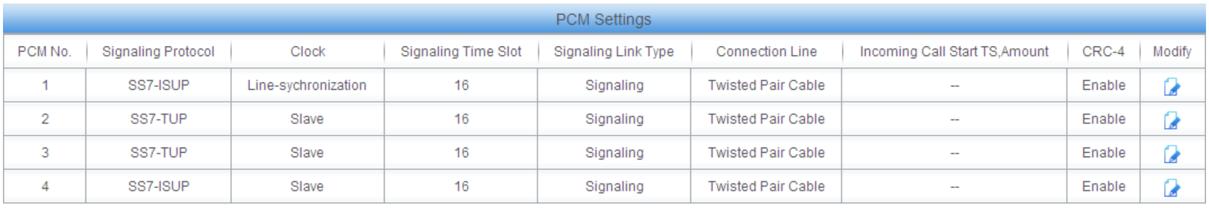



Figure 3-28 Circuit Maintenance Interface

See Figure 3-28 for the Circuit Maintenance interface. You can block or unblock PCMs, ports and channels on this interface. ***Check All*** means to select all available items for the current port; ***Uncheck All*** means to cancel all selections for the current port; ***Inverse*** means to uncheck the selected items and check the unselected.

## 3.4.3 PCM



PCM Settings

| PCM No. | Signaling Protocol | Clock | Signaling Time Slot | Signaling Link Type | Connection Line | Incoming Call Start TS,Amount | CRC-4 | Modify |
|---|---|---|---|---|---|---|---|---|
| 1 | SS7-ISUP | Line-sychronization | 16 | Signaling | Twisted Pair Cable | -- | Enable | |
| 2 | SS7-TUP | Slave | 16 | Signaling | Twisted Pair Cable | -- | Enable | |
| 3 | SS7-TUP | Slave | 16 | Signaling | Twisted Pair Cable | -- | Enable | |
| 4 | SS7-ISUP | Slave | 16 | Signaling | Twisted Pair Cable | -- | Enable | |

Figure 3-29 PCM Settings Interface

See Figure 3-29 for the PCM settings interface. The above list shows the detailed information and configurations of each PCM. The table below explains the items shown in the above figure.

| Item | Description |
|---|---|
| ***PCM No.*** | The number of the PCM, numbered from 0. This item is not configurable. |

| | |
|---|---|
| ***Signaling Protocol*** | The signaling protocol applied on the digital trunk. It includes *ISDN User Side*, *ISDN Network Side*, *SS7-TUP*, *SS7-ISUP*, and *SS1* in E1, and only includes *ISDN User Side*, *ISDN Network Side* in T1.<br>**Note:** 1, Changing the interface type from E1 to T1 will forbid those non-ISDN signaling modes in E1. And in such case, the gateway will by default set this item to *ISDN User Side*.<br>2, For SMG3008, a single gateway can be configured with two different signaling modes simultaneously.<br>3, For SMG3016, a single gateway can be configured with three different signaling modes simultaneously. |
| ***Clock*** | The clock mode for the digital trunk, including *Line-synchronization*, *Free-run* and *Slave*. |
| ***Signaling Time Slot*** | Sets the time slot used for signaling transmission on the digital trunk. If the configuration item ***Signaling Protocol*** is set to *ISDN* and *SS1*, the signaling time slot is Time Slot 16 in E1 or Time Slot 24 in T1 (SS1 not supported in T1 by far), which cannot be modified. |
| ***Signaling Link Type*** | Indicates whether the PCM is used as a signaling link or a voice link. If no time slot is used to transmit signaling, the PCM is a voice link. |
| ***Connection Line*** | Physical connection line type. |
| ***Incoming Call Start TS, Amount*** | ets a certain amount of channels which starts from a certain TS to process the incoming calls and others on the PCM to process outgoing calls. This is valid only when the configuration item ***Signaling Protocol*** is set to *SS1.* |
| ***CRC-4*** | Sets whether to enable the CRC-4 verification feature. By default, this feature is Enabled. |

Click ***Modify*** in Figure 3-29 to modify a PCM. See Figure 3-30 for the PCM modification interface. Most configuration items on this interface are the same as those on the ***PCM Settings*** interface.

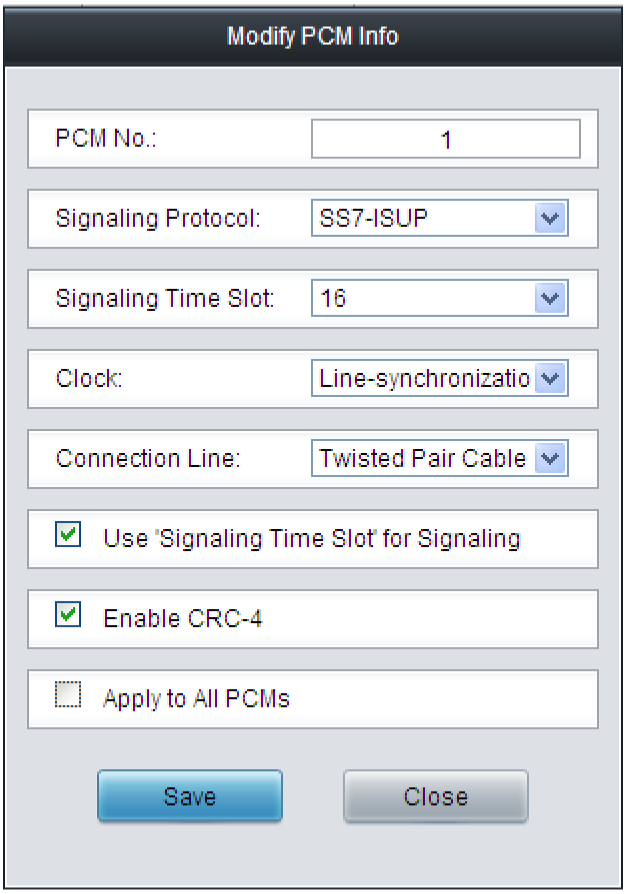



Figure 3-30 Modify PCM

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

| Item | Description |
|---|---|
| ***Use 'Signaling Time Slot' for Signaling*** | If this item is checked, it indicates that the signaling time slot configured in ***Signaling Time Slot*** is used for signaling transmission. You can see this item only when the configuration item ***Signaling Protocol*** is set to *SS7-TUP* or *SS7-ISUP*. |
| ***Apply to All PCMs*** | Check this item to apply the above settings (excluding ***Clock***) to all PCMs. |

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

## 3.4.4 PCM Trunk

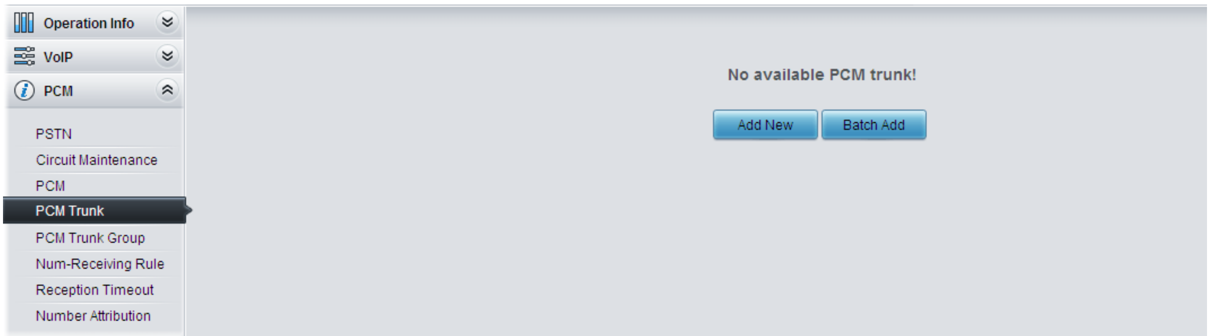



Figure 3-31 PCM Trunk Configuration Interface

See Figure 3-31 for the PCM Trunk Configuration interface. By default, there is no PCM trunk available on the gateway. Click ***Add New*** or ***Batch Add*** to add them manually. See Figure 3-32,

Figure 3-33.

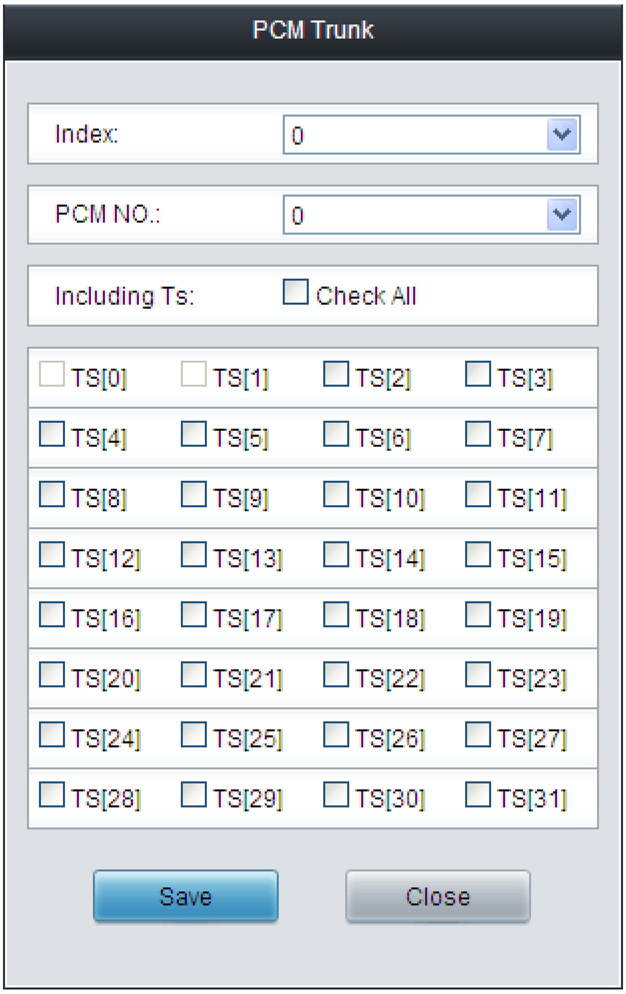



Figure 3-32 Add PCM Trunk Interface

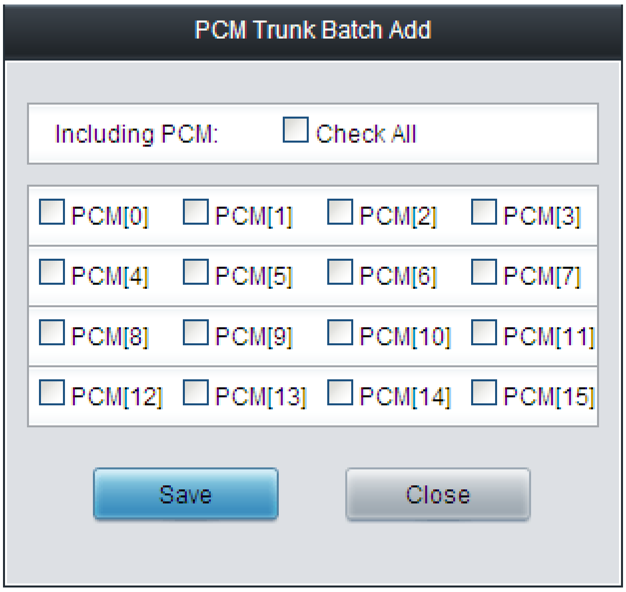



Figure 3-33 PCM Trunk Batch Add Interface

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

| Item | Description |
|---|---|
| ***Index*** | The unique index of each PCM trunk |

| | |
|---|---|
| ***PCM NO.*** | The number of the PCM, numbered from 0. |
| ***Including Ts*** | Sets the TS included in this PCM which can make incoming/outgoing calls. |
| ***Including PCM*** | Sets the PCM included in the PCM trunk. |

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

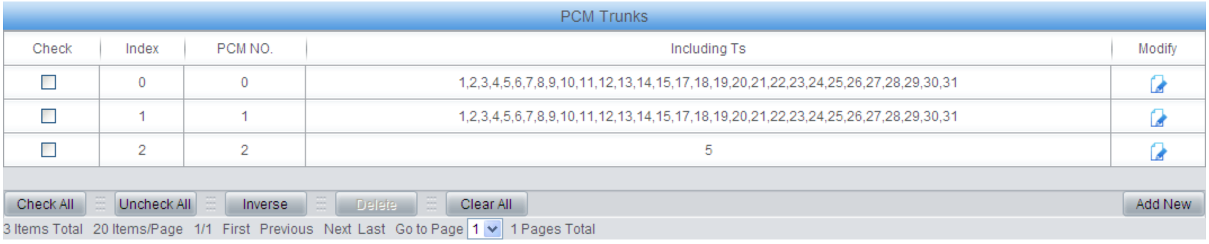



Figure 3-34 PCM Trunks List

Click ***Modify*** in Figure 3-34 to modify a PCM trunk. The configuration items on the PCM Trunk Modification Interface are the same as those on the ***Add PCM Trunk*** interface.

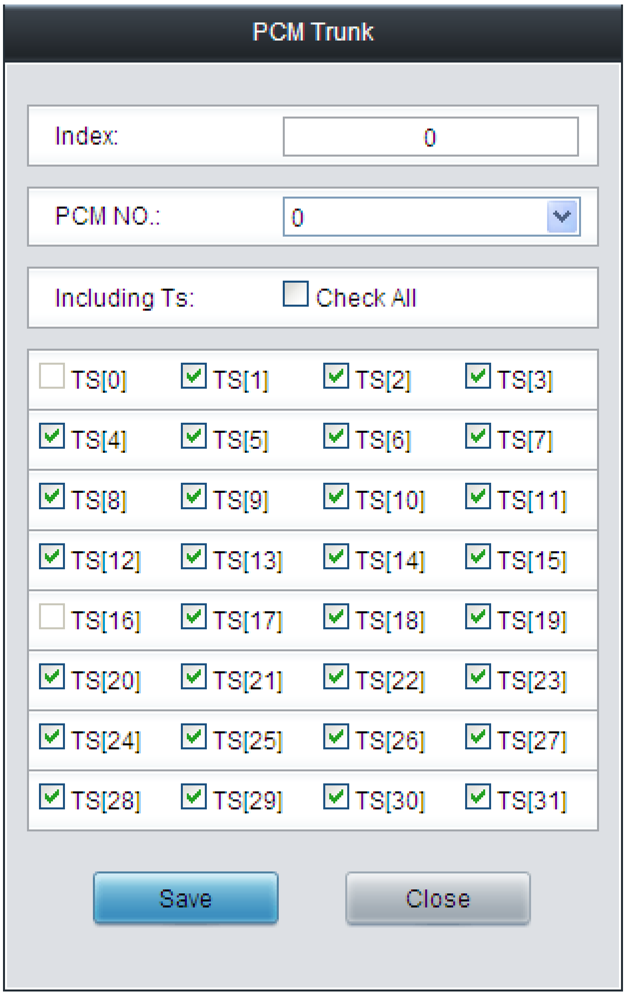



Figure 3-35 PCM Trunk Modification Interface

To delete a PCM trunk, check the checkbox before the corresponding index in Figure 3-34 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 PCM trunks at a time, click the ***Clear All*** button in Figure 3-34.

### 3.4.5 PCM Trunk Group

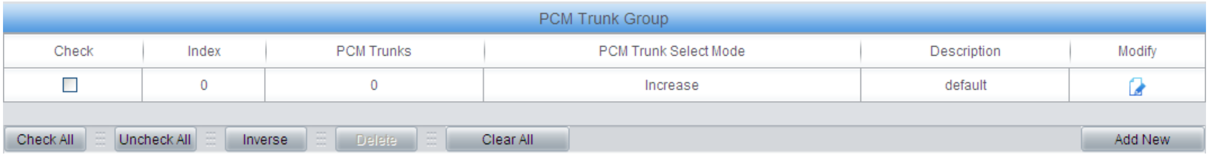



Figure 3-36 PCM Trunk Group Settings

See Figure 3-36 for the PCM trunk group settings interface. A new PCM trunk group can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-37 for the PCM trunk group adding interface.

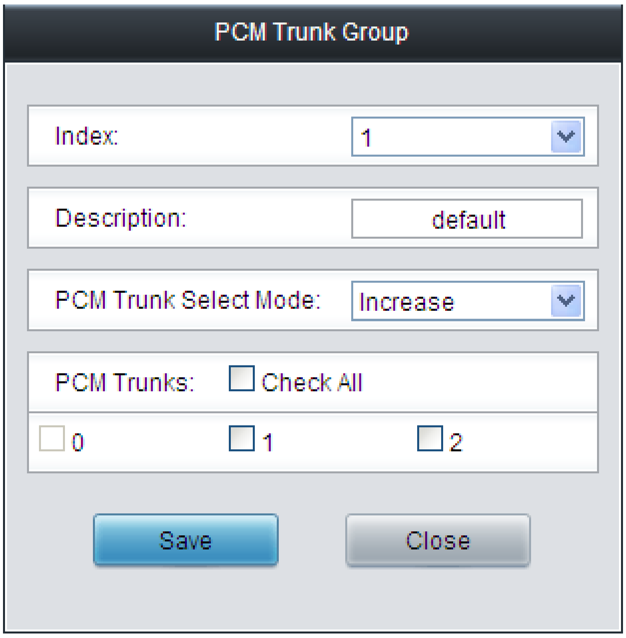



Figure 3-37 Add New PCM Trunk Group

The table below explains the items shown in Figure 3-37.

| Item | Description |
|---|---|
| ***Index*** | The unique index of each PCM trunk group, which is mainly used in the configuration of routing rules and number manipulation rules to correspond to PCM trunk groups. |
| ***Description*** | More information about each PCM trunk group. |

<table>
<tr><td rowspan="5">PCM Trunk Select Mode</td><td colspan="2">When the PCM trunk group receives a call, it will choose a PCM trunk based on the select mode set by this configuration item to ring. The optional values and their corresponding meanings are described in the table below.</td></tr>
<tr><td>Option</td><td>Description</td></tr>
<tr><td>Increase</td><td>Search for an idle PCM trunk in the ascending order of the PCM number, starting from the minimum.</td></tr>
<tr><td>Decrease</td><td>Search for an idle PCM trunk in the descending order of the PCM number, starting from the maximum.</td></tr>
<tr><td>Cyclic Increase</td><td>Provided PCM Trunk N is the available PCM trunk found last time. Search for an idle PCM trunk in the ascending order of the PCM number, starting from PCM Trunk N+1.</td></tr>
<tr><td></td><td>Cyclic Decrease</td><td>Provided PCM Trunk N is the available PCM trunk found last time. Search for an idle PCM trunk in the descending order of the PCM number, starting from PCM trunk N-1.</td></tr>
<tr><td>PCM Trunks</td><td colspan="2">The PCM trunks in the PCM trunk group. If the checkbox before a PCM trunk is grey, it indicates that the PCM trunk has been occupied. The ticked PCM trunks herein will be displayed in the column ‘PCM Trunks’ in Figure 3-36.</td></tr>
</table>

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

Click ***Modify*** in Figure 3-36 to modify a PCM trunk group. See Figure 3-38 for the PCM trunk group modification interface. The configuration items on this interface are the same as those on the ***Add New PCM Trunk Group*** interface.

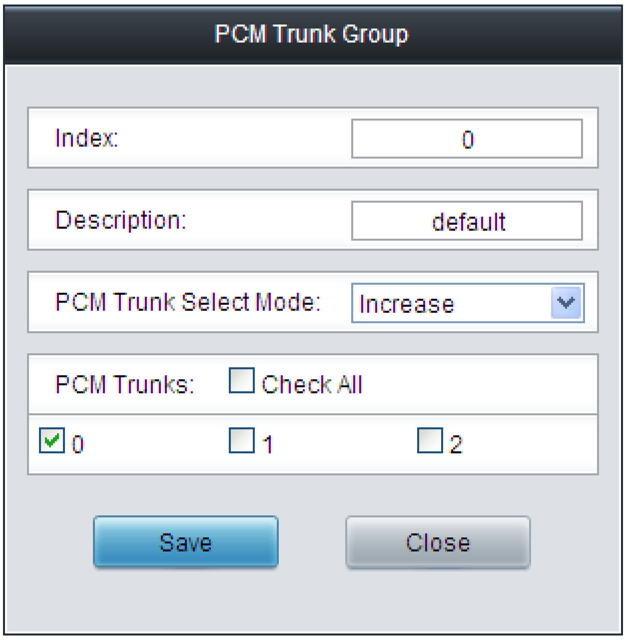



Figure 3-38 Modify PCM Trunk Group

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

### 3.4.6 Number-receiving Rule

The gateway uses a number-receiving plan to filter the numbers received from PSTN. Only those numbers which match the plan will be processed. The number-receiving plan consists of multiple number-receiving rules, each of which has a priority in sequence to avoid conflict.

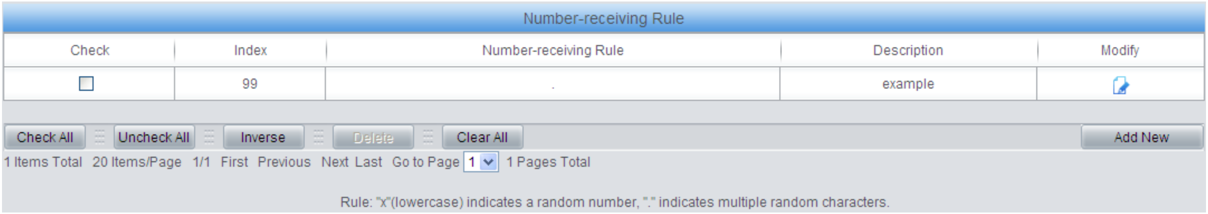



Figure 3-39 Number-Receiving Rule Configuration Interface

See Figure 3-39 for the Number-receiving Rule Configuration interface. The list in the above figure shows the number-receiving rules with their priorities and description. A new number-receiving rule can be added by the ***Add New*** button on the bottom right corner. See Figure 3-40 for the number-receiving rule adding interface.

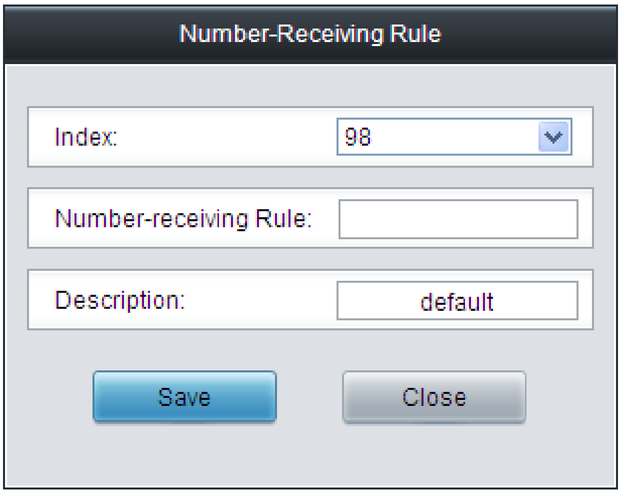



Figure 3-40 Add New Number-Receiving Rule

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

| Item | Description |
|---|---|
| ***Index*** | The unique index of each number-receiving rule, which denotes its priority. A number-receiving rule with a smaller index value has a higher priority and will be checked earlier while matching. |

***Number-Receiving Rule***

Up to 99 number-receiving rules can be configured in the gateway, and the maximum length of each number-receiving rule is 127 characters. See below for the meaning of each character in the number-receiving rule. The gateway will do instant matching for your receiving number based on the number-receiving rule and regard your receiving as finished upon receiving ‘#’ or reception timeout.

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

By default, there is only one rule configured on the gateway. The table below lists 20 rules as example for your easy use and understanding. See below for detailed information.

| Priority | Dialing Rule | Description |
|---|---|---|
| 99 | . | Any number in any length. |
| 98 | 01[3,5,8]xxxxxxxxx. | Any 12-digit number starting with 013, 015 or 018 |
| 97 | 010xxxxxxxx | Any 11-digit number starting with 010 |
| 96 | 02xxxxxxxxx | Any 11-digit number starting with 02 |
| 95 | 0[3-9]xxxxxxxxxx | Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 |
| 94 | 120 | Number 120 |
| 93 | 11[0,2-9] | Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 |
| 92 | 111xx | Any 5-digit number starting with 111 |
| 91 | 123xx | Any 5-digit number starting with 123 |
| 90 | 95xxx | Any 5-digit number starting with 95 |
| 89 | 100xx | Any 5-digit number starting with 100 |
| 88 | 1[3-5,8]xxxxxxxxx | Any 11-digit number starting with 13, 14, 15 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 |
| | 80 | 8xxx | Any 4-digit number starting with 8 |
| ***Description*** | Remarks for the number-receiving rule. It can be any information, but can not be left empty. | | |

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

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

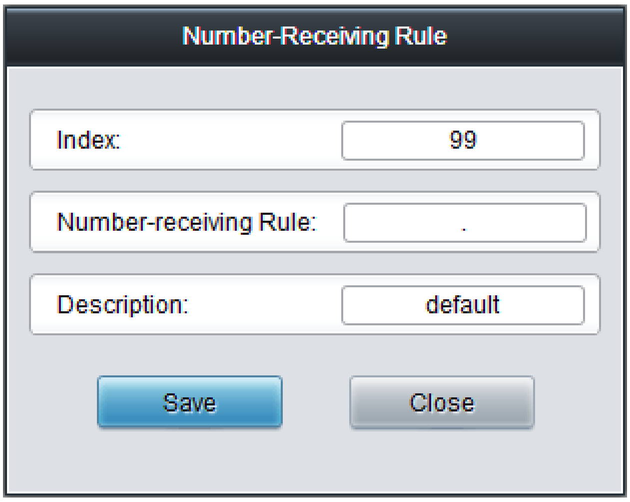



Figure 3-41 Modify Number-receiving Rule

To delete a number-receiving rule, check the checkbox before the corresponding index in Figure 3-39 and click the '***Delete***' button. ***Check All*** means to select all available items on the current page; ***Uncheck All*** means to cancel all selections on the current page; ***Inverse*** means to uncheck the selected items and check the unselected. To clear all number-receiving rules at a time, click the ***Clear All*** button in Figure 3-39.

## 3.4.7 Reception Timeout

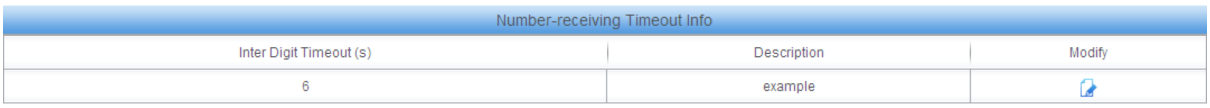



Figure 3-42 Number-receiving Timeout Info Interface

See Figure 3-42 for the number-receiving 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 receiving number. Range of value: 1~10, calculated by s, with the default value of 6. In case your number-receiving rules do not include "**.**", the call will fail if there is no digit received or no number-receiving rule matched during this interval; in case your number-receiving rules include "**.**", the gateway will wait until this interval ends and match to the number-receiving rule "**.**" if there is no digit received or no other number-receiving 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 3-42 to modify the number-receiving timeout info. See Figure 3-43 for the number-receiving timeout info modification interface. The configuration items on this interface are the same as those on the ***Number-receiving Timeout Info Interface***.

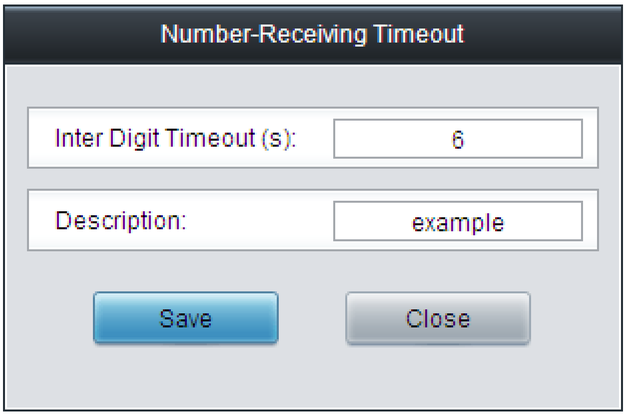



Figure 3-43 Modify Number-receiving Timeout Info

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

### 3.4.8 Number Attribution

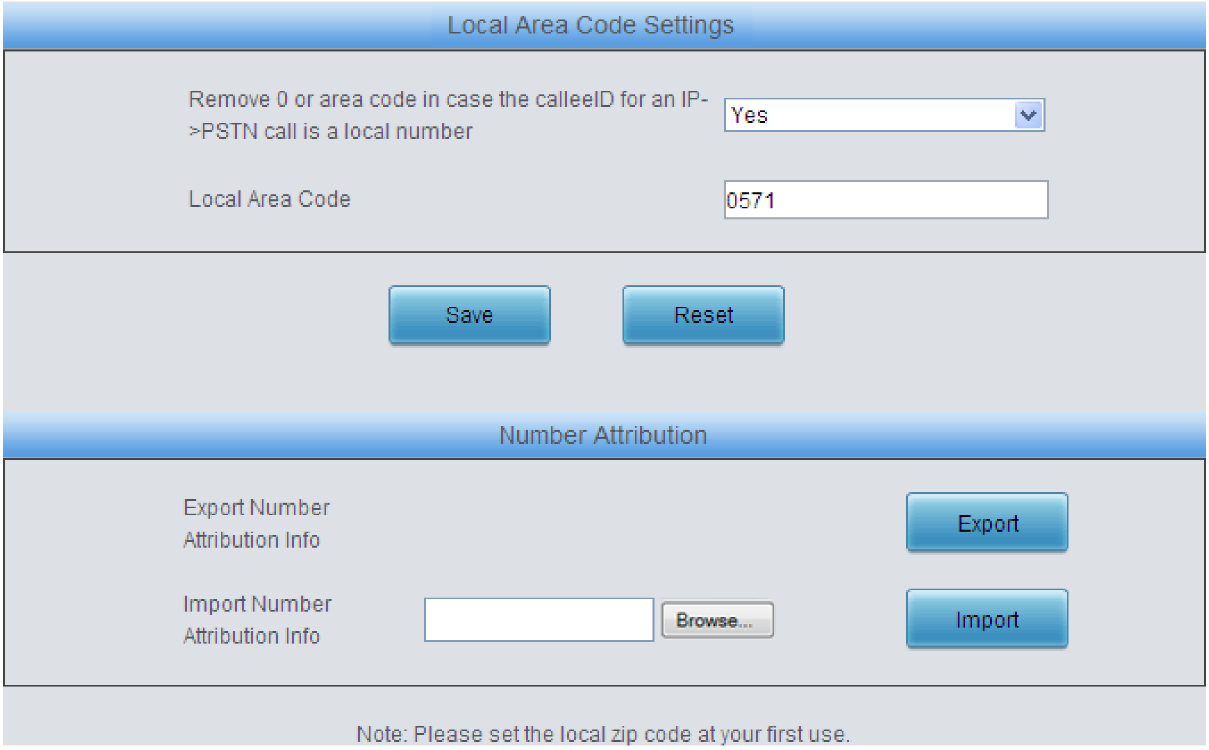



Figure 3-44 Number Attribution Setting Interface

See Figure 3-44 for the Number Attribution Setting interface, which is used to set whether to remove 0 or the area code from CalleeID. Click ***Save*** to save your settings into the gateway or click ***Reset*** to restore the configurations. Click ***Export*** to export and check the information about the number attribution; or select the required number attribution file via ***Browse…*** and click ***Import*** to import it into the gateway.

**Note:** By far only those numbers with the start 13x, 14x, 15x, 17x, or 18x can be imported.

## 3.5 SS7 Settings

Users can see the SS7 option in the menu only when the configuration item ***Signaling Protocol*** on the PCM settings interface is set to *SS7-TUP* or *SS7-ISUP*. SS7 Settings includes eight parts: ***SS7***, ***TUP***, ***TUP Number Param***, ***ISUP***, ***Number Param***, ***Original CalleeID Pool***, ***Redirecting Number Pool*** and ***SS7 Server***. See Figure 3-45.

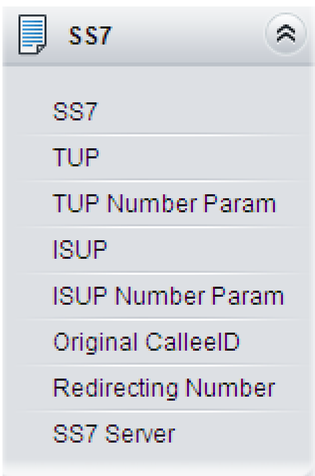


Figure 3-45 SS7 Settings

### 3.5.1 SS7

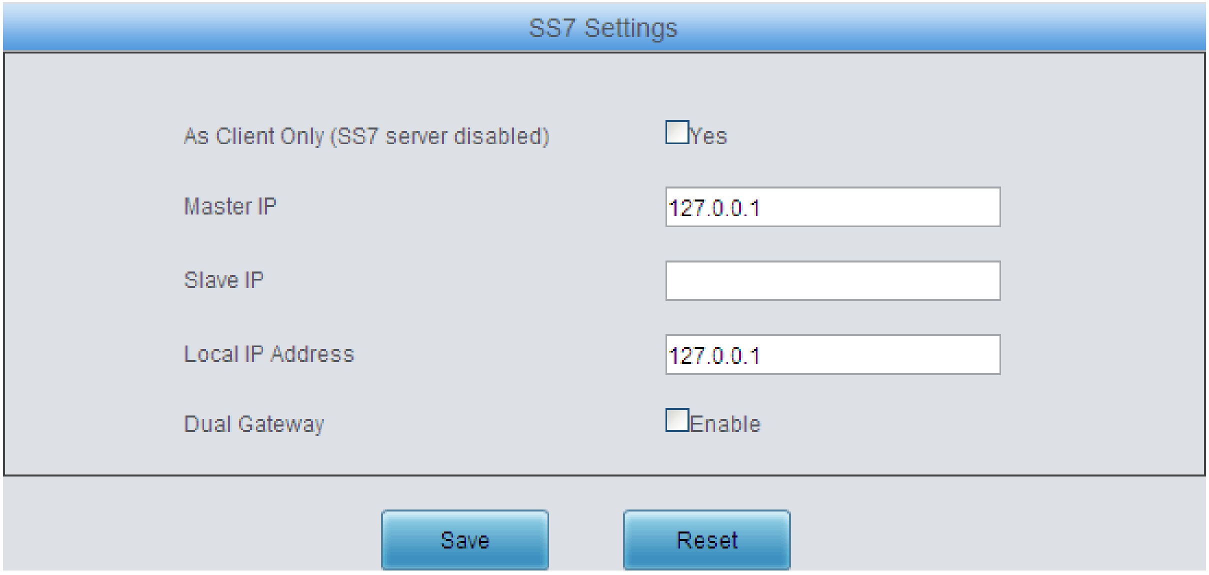



Figure 3-46 SS7 Settings Interface

See Figure 3-46 for the SS7 settings interface where you can configure the general SS7 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 3.12.17 Restart for detailed instructions. The table below explains the items shown in Figure 3-46.

| Item | Description |
|---|---|
| ***As Client Only*** | Sets whether the gateway serves as Client only or not. If it is set to *No* (default), the SS7 server will be disabled. |
| ***Master IP*** | Sets the IP address of the master SS7 server, with the default value of 127.0.0.1, which indicates that there is only one SS7 server available. |
| ***Slave IP*** | Sets the IP address of the slave SS7 server. Only when the item ***Dual Gateway*** is ticked can this item be configured. |
| ***Local IP Address*** | Sets the IP address of the local PC, with the default value of 127.0.0.1. |
| ***Dual Gateway*** | If this feature is enabled, two SS7 servers are used at the same time in the system. The configuration items ***Master IP*** and ***Slave IP*** are respectively used to set the IP addresses of the master and slave servers. |

### 3.5.2 TUP

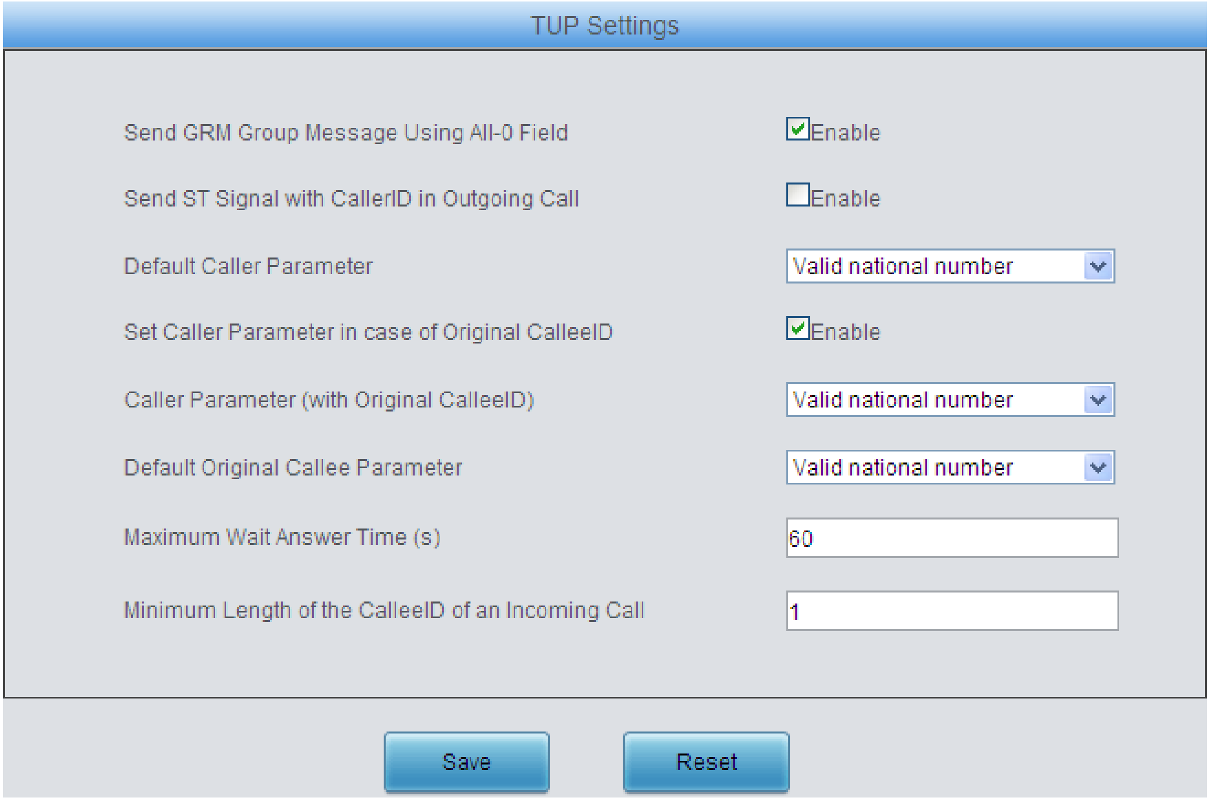



Figure 3-47 TUP Settings Interface

See Figure 3-47 for the TUP settings interface. Users can see this interface and configure the general TUP parameters only when the configuration item ***Signaling Protocol*** on the PCM settings interface is set to *SS7-TUP*. 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 3.12.17 Restart for detailed instructions. The table below explains the items shown in Figure 3-47.

| Item | Description |
|---|---|
| ***Send GRM Group Message Using All-0 Field*** | If this configuration item is enabled, when the local driver sends the circuit group message to the remote PBX, this message covers all time slots TS1~31. By default this item is enabled. |
| ***Send ST Signal with CallerID in Outgoing Call*** | If this configuration item is enabled, the calling party number string sent by the gateway contains the ST signal in the outgoing call. By default this item is disabled. |
| ***Default Caller Parameter*** | Sets the address indicator in the calling line identification field in the IAI message. The optional values are: *Local subscriber number*, *Spare national number*, *Valid national number* and *International number*, with the default value of *Valid national number*. |
| ***Set Caller Parameter in case of Original CalleeID*** | Once this feature is enabled, if the IP end carries the original CalleeID in a call from IP to PSTN, you shall set a separate value for the address indicator in the calling line identification field in the IAI message, i.e. ***Caller Parameter ( with Original CalleeID)***. By default this configuration item is disabled. |

| | |
|---|---|
| ***Caller Parameter (with Original CalleeID)*** | This item is valid only when ***Set Caller Parameter in case of Original CalleeID*** is enabled. It sets the address indicator in the calling line identification field in the IAI message when the IP end carries the original CalleeID in a call from IP to PSTN. The optional values are: *Local subscriber number*, *Spare national number*, *Valid national number* and *International number*, with the default value of *Valid national number*. |
| ***Default Original Callee Parameter*** | Sets the address indicator in the original called party address field of the IAI message. The optional values are: *Local subscriber number*, *Spare national number*, *Valid national number* and *International number*, with the default value of *Valid national number*. |
| ***Maximum Wait Answer Time (s)*** | Sets the maximum time to wait for the answer from the called party of an outgoing call. 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. |
| ***Minimum Length of the CalleeID of an Incoming Call*** | Sets the minimum length of the CalleeID under the fixed-length mode. The value range is 1≤n≤40. Provided it is set to n, that is, the local end has received all the n digits of the called party number of the incoming call, the number reception will be regarded as finished. |

## 3.5.3 TUP Number Parameter

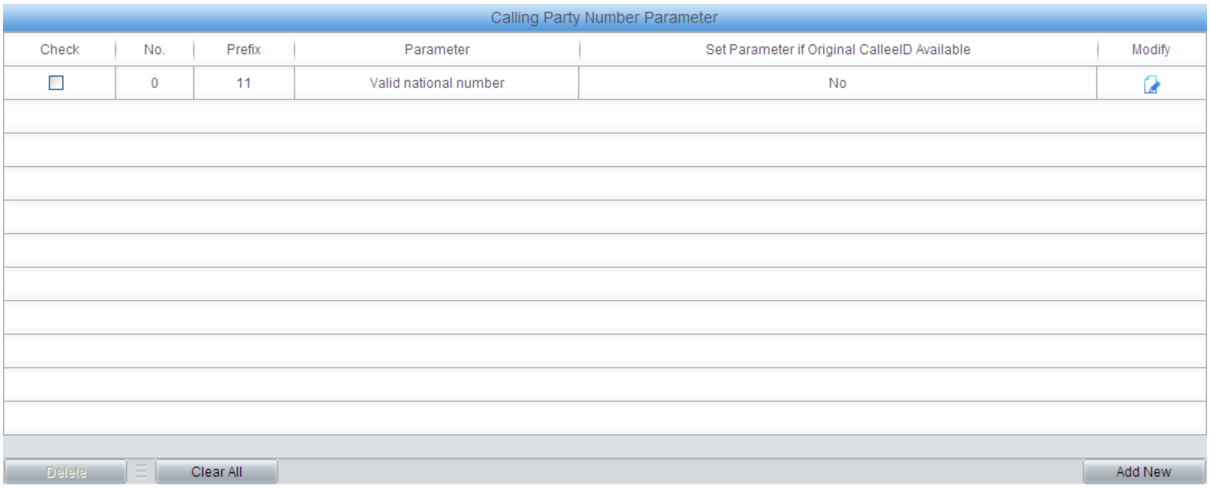



Figure 3-48 TUP Number Parameter Configuration Interface

See Figure 3-48 for the TUP Number Parameter Configuration interface, which is used to set the corresponding parameters for the calling party number in TUP.

A new TUP number parameter can be added by the ***Add New*** button. See Figure 3-49 for the calling party number adding interface.

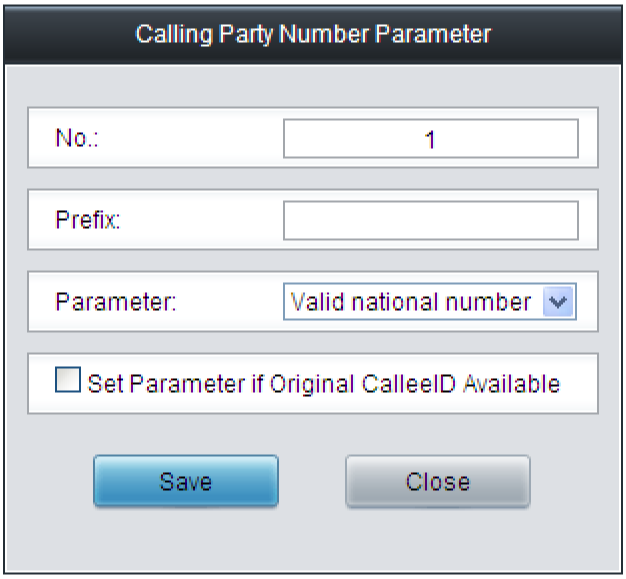



Figure 3-49 Add New Calling Party Number Parameter

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

| Item | Description |
|---|---|
| ***No.*** | The corresponding number for a calling party number parameter, which starts from 0. |
| ***Prefix*** | A string of numbers at the beginning of a calling party number. |
| ***Parameter*** | Sets the parameter for a calling party number. |
| ***Set Parameter if Original CalleeID Available*** | Set whether to enable the feature of setting this parameter only if the original CalleeID is available. |

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

Click ***Modify*** in Figure 3-48 to modify the calling party number parameter. See Figure 3-50 for the calling party number parameter modification interface. The configuration items on this interface are the same as those on the ***Add New Calling Party Number Parameter*** interface.

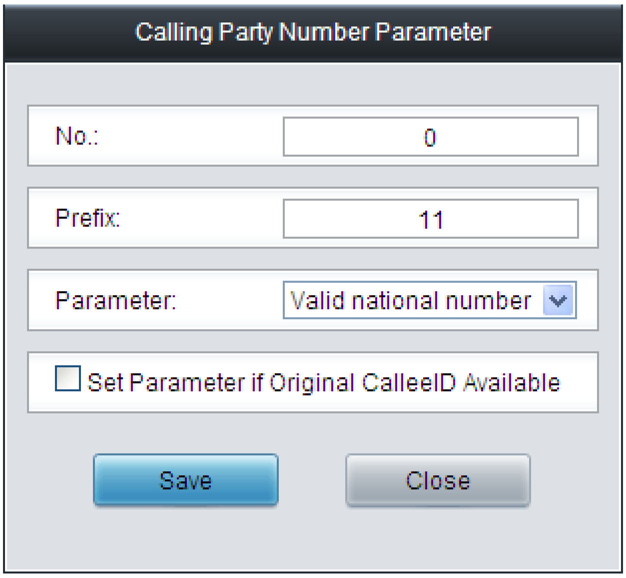



Figure 3-50 Modify Calling Party Number Parameter

To delete a calling party number parameter, check the checkbox before the corresponding index in Figure 3-48 and click the '***Delete***' button. To clear all calling party number parameters at a time, click the ***Clear All*** button in Figure 3-48.

**Note:** If there are two or more calling party numbers with the same prefix, the one numbered the smallest is valid and all the others become invalid.

## 3.5.4 ISUP

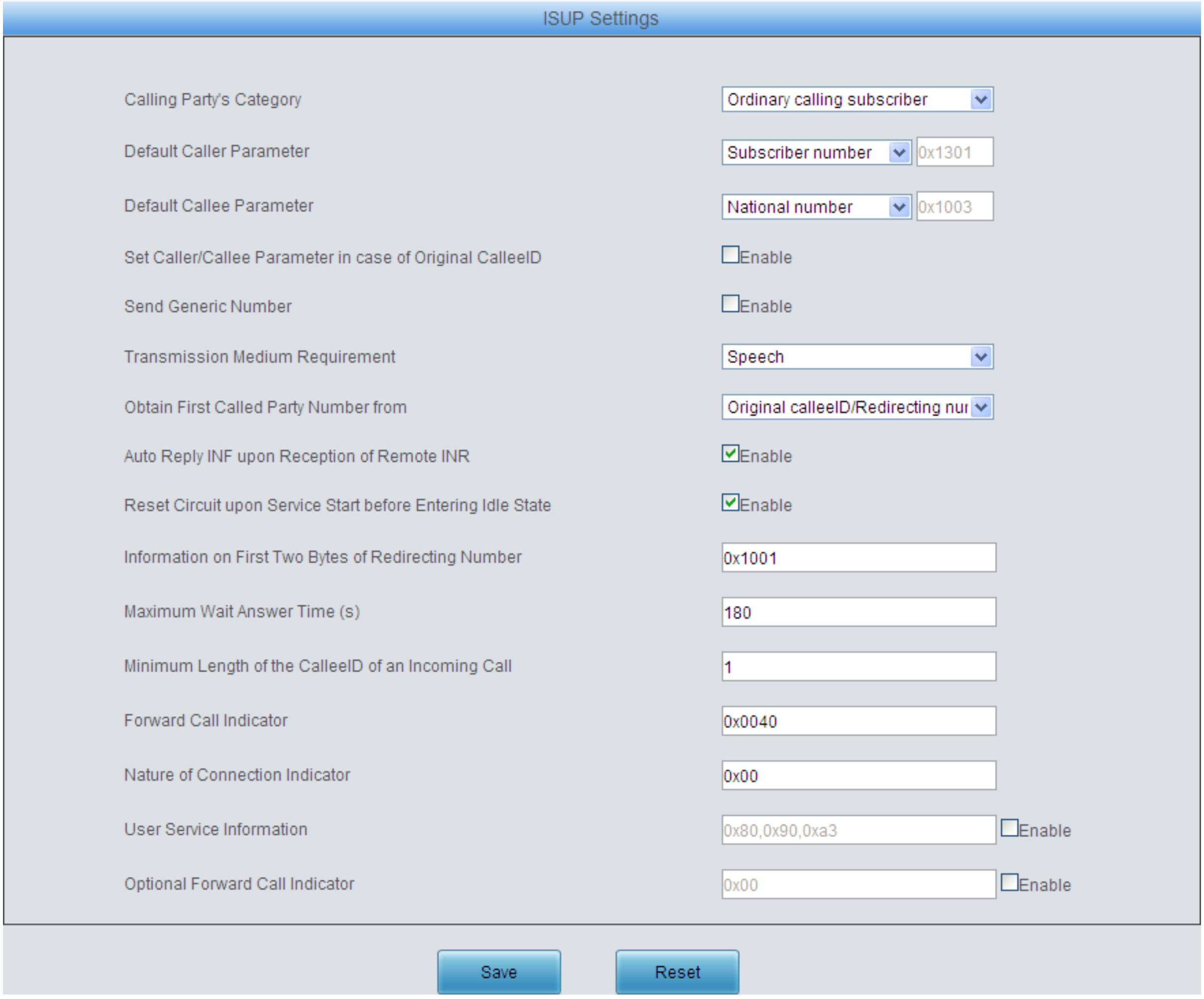



Figure 3-51 ISUP Settings Interface

See Figure 3-51 for the ISUP settings interface. Users can see this interface and configure the general ISUP parameters only when the configuration item ***Signaling Protocol*** on the PCM settings interface is set to *SS7-ISUP*. 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 3.12.17 Restart for detailed instructions. The table below explains the items shown in Figure 3-51.

| Item | Description |
|---|---|
| ***Calling Party's Category*** | Sets the calling party's category indicator in the IAM message. The optional values are: *National operator*, *Ordinary calling subscriber*, *Calling subscriber with priority*, *Data call*, *Test call* and *Payphone/Others*, with the default value of *Ordinary calling subscriber*. |

| | |
|---|---|
| ***Default Caller Parameter*** | Sets the calling party number parameter field in the IAM message. The optional values are: *Subscriber number*, *National number*, and *International number*, with the default value of *Subscriber number*. |
| ***Default Callee Parameter*** | Sets the called party number parameter field in the IAM message. The optional values are: *Subscriber number*, *National number*, and *International number*, with the default value of *National number*. |
| ***Set Caller/Callee Parameter in case of Original CalleeID*** | Once this feature is enabled, if the IP end carries the original CalleeID in a call from IP to PSTN, you shall set separate values for the caller and callee parameters in the IAM message, i.e. ***Caller Parameter (with Original CalleeID)*** and ***Callee Parameter (with Original CalleeID)***. By default this configuration item is disabled. |
| ***Caller Parameter (with Original CalleeID)*** | This item is valid only when ***Set Caller/Callee Parameter in case of Original CalleeID*** is enabled. It sets the calling party number parameter field in the IAM message when the IP end carries the original CalleeID in a call from IP to PSTN. The optional values are: *Subscriber number*, *National number*, and *International number*, with the default value of *Subscriber number*. |
| ***Callee Parameter (with Original CalleeID)*** | This item is valid only when ***Set Caller/Callee Parameter in case of Original CalleeID*** is enabled. It sets the called party number parameter field in the IAM message when the IP end carries the original CalleeID in a call from IP to PSTN. The optional values are: *Subscriber number*, *National number*, and *International number*, with the default value of *National number*. |
| ***Send Generic Number*** | Sets the generic number parameter in IAM message, with the default value of *disabled*. |
| ***Transmission Medium Requirement*** | Sets the transmission medium requirement parameter in the IAM message. The optional values are: *Speech*, *64 kb/s unrestricted*, *3.1khz audio*, *Alternative: speech (service 2)/ 64kbit/s unrestricted (service 1) (Spare)*, *Alternative: 64kbit/s unrestricted (service 1)/ speech (service 2) (Spare)*, *64kb/s preferred*, *2*64kb/s unrestricted*, *384 kb/s unrestricted*, *1920 kb/s unrestricted* and *Spare*, with the default value of *Speech*. |
| ***Obtain First Called Party Number from*** | Sets where the first called party number is obtained from. The optional values are: *Only original CalleeID* and *Original CalleeID/ Redirecting number*, with the default value of *Only original CalleeID*. |
| ***Auto Reply INF upon Reception of Remote INR*** | If this feature is enabled, once the INR message is received from the remote PBX in an outgoing call, the driver will automatically reply it with the INF message. By default this feature is enabled. |
| ***Reset Circuit upon Service Start before Entering Idle State*** | If this feature is enabled, the circuit will send a circuit reset message before entering the idle state after the ISUP service is enabled. By default this feature is enabled. |
| ***Information on First Two Bytes of Redirecting Number*** | Sets the first two bytes of the redirecting number in the IAM message, including the nature of address indicator, numbering plan indicator and address presentation restricted indicator, with the default value of 0x1001. |
| ***Maximum Wait Answer Time (s)*** | Sets the maximum time to wait for the answer from the called party of an outgoing call. If the call is not answered within the specified time period, it will be canceled by the channel automatically. The default value is 180, calculated by s. |

| | |
|---|---|
| ***Minimum Length of the CalleeID of an Incoming Call*** | Sets the minimum length of the CalleeID under the fixed-length mode. The value range is 1≤n≤40. Provided it is set to n, that is, the local end has received all the n digits of the called party number of the incoming call, the number reception will be regarded as finished. |
| ***Forward Call Indicator*** | Sets the forward call indicator in the IAM message, with the default value of 0x0040. |
| ***Nature of Connection Indicator*** | Sets the nature of connection indicator in the IAM message, with the default value of 0x00. |
| ***User Service Information*** | Sets whether the IAM message contains the user service information. By default this feature is disabled. If this feature is enabled, its value is usually determined by the remote PBX, with the default value of 0x80, 0x90, 0xa3. This default value is applicable to Huawei PBXes. |
| ***Optional Forward Call Indicator*** | Sets whether the IAM message contains the optional forward call indicator. By default this feature is disabled. If this feature is enabled, its value is usually determined by the remote PBX, with the default value of 0x00. |

## 3.5.5 ISUP Number Parameter

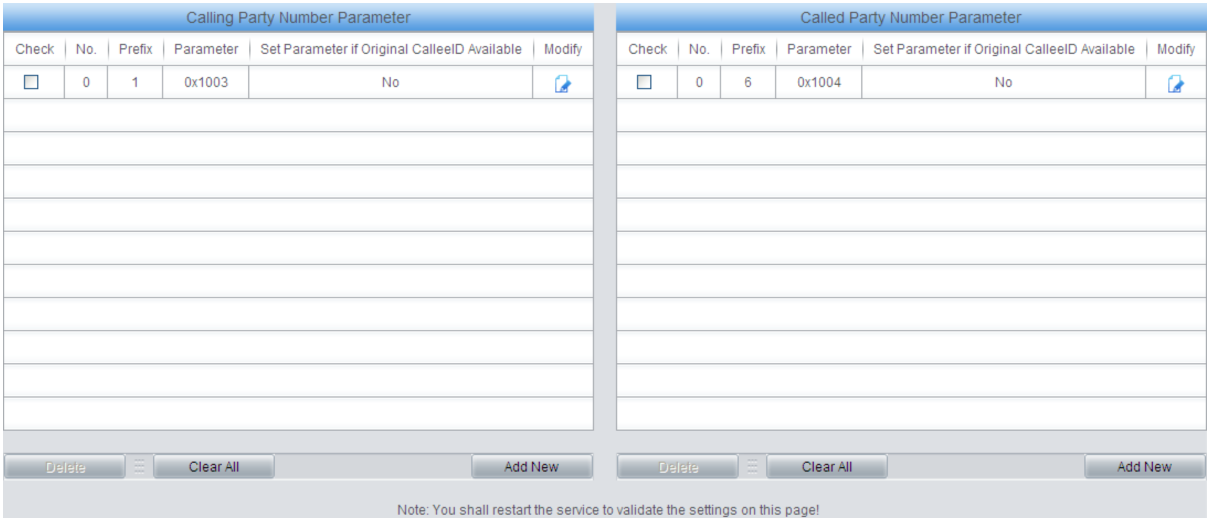



Figure 3-52 ISUP Number Parameter Configuration Interface

See Figure 3-52 for the ISUP Number Parameter Configuration interface, which includes two parts: ***Calling Party Number Parameter*** and ***Called Party Number Parameter***.

A new calling/called party number parameter can be added by the ***Add New*** button. See Figure 3-53, Figure 3-54 for the calling/called party number parameter adding interface.

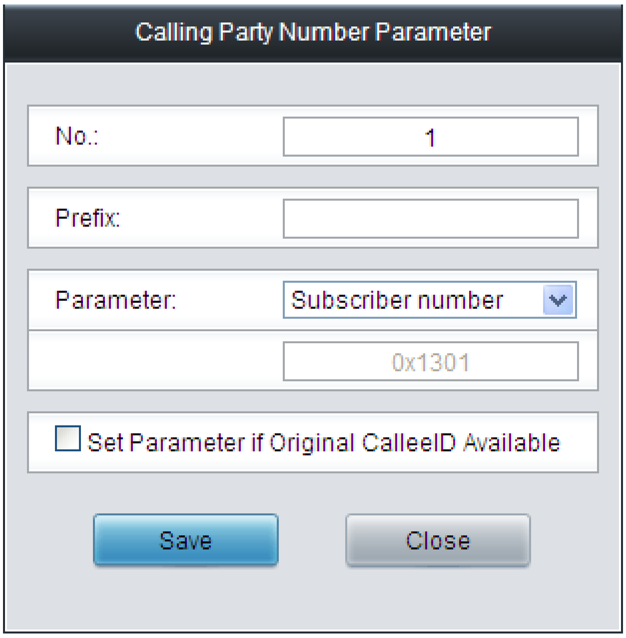



Figure 3-53 Add New Calling Party Number Parameter


Figure 3-54 Add New Called Party Number Parameter

The table below explains the items shown in above figures.

| Item | Description |
|---|---|
| ***No.*** | The corresponding number for a calling/called party number parameter, which starts from 0. |
| ***Prefix*** | A string of numbers at the beginning of a calling/called party number. |
| ***Parameter*** | Sets the parameter for a calling/called party number. |
| ***Set Parameter if Original CalleeID Available*** | Set whether to enable the feature of setting this parameter only if the original CalleeID is available. |

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

Click ***Modify*** in Figure 3-52 to modify the calling/called party number parameter. See Figure 3-55, Figure 3-56 for the calling/called party number parameter modification interface. The configuration items on this interface are the same as those on the ***Add New Calling/Called Party Number Parameter*** interface.

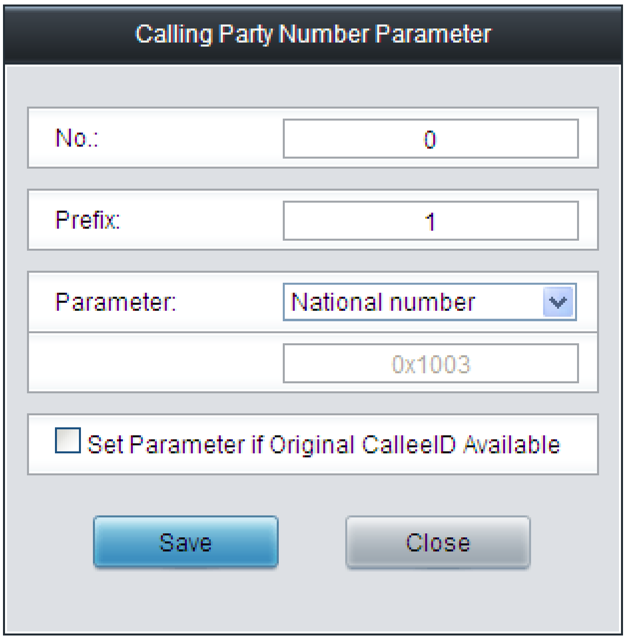



Figure 3-55 Modify Calling Party Number Parameter

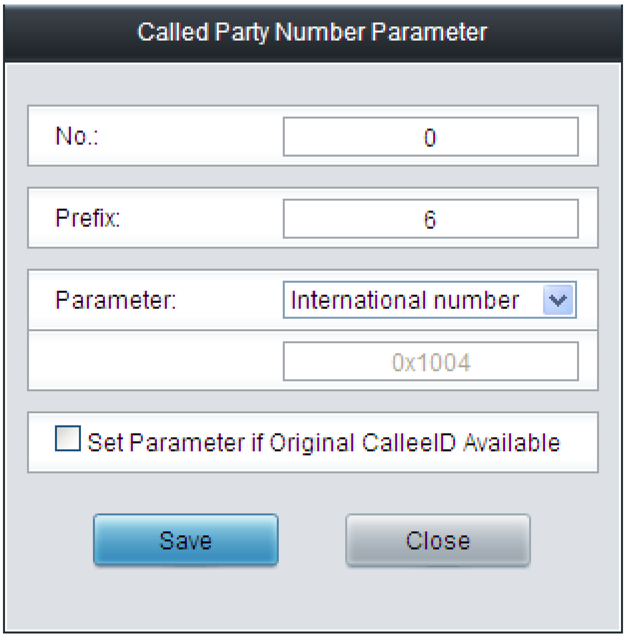



Figure 3-56 Modify Called Party Number Parameter

To delete a calling/called party number parameter, check the checkbox before the corresponding index in Figure 3-52 and click the '***Delete***' button. To clear all calling/called party number parameters at a time, click the ***Clear All*** button in Figure 3-52.

**Note:** If there are two or more calling/called party numbers with the same prefix, the one numbered the smallest is valid and all the others become invalid.

### 3.5.6 Original CalleeID Pool

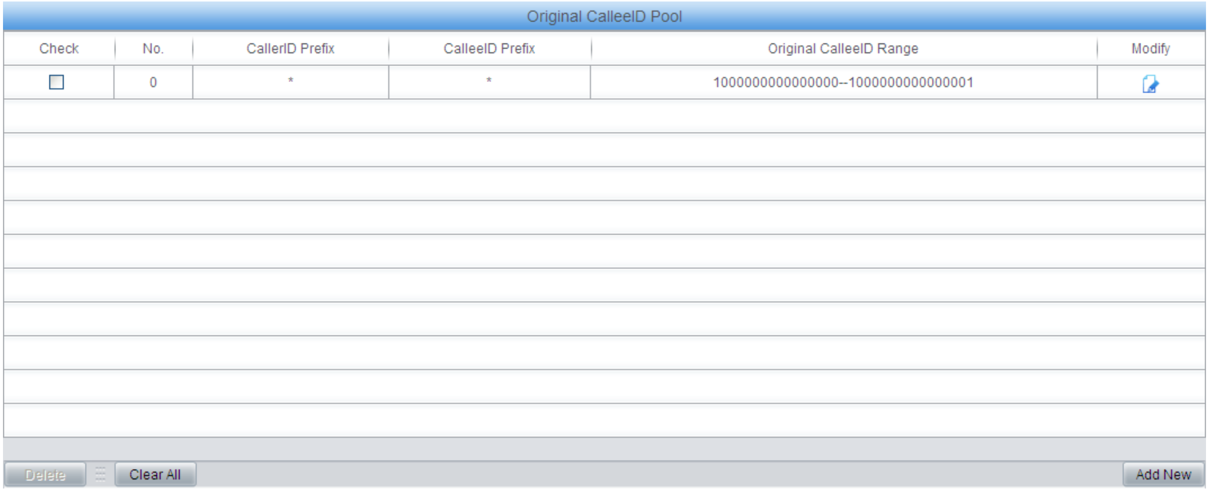



Figure 3-57 Original CalleeID Pool Interface

See Figure 3-57 for the Original CalleeID Pool interface, which is used to add the original CalleeID for all outgoing calls or some special calls which contain the specified calling/called prefix.

A new original CalleeID can be added by the ***Add New*** button. See Figure 3-58 for the original CalleeID adding interface.

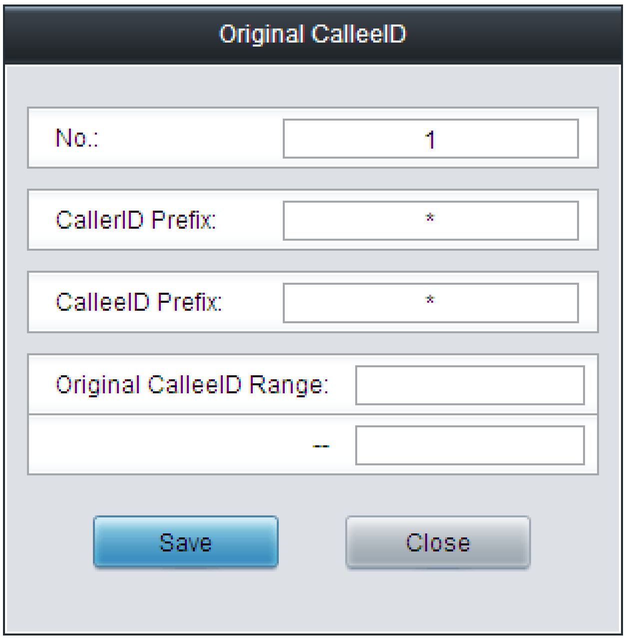



Figure 3-58 Add New Original CallerID

The table below explains the items shown in above figures.

| Item | Description |
| --- | --- |
| ***No.*** | The corresponding number for an added original CalleeID. The value range is 0~99. |
| ***CallerID Prefix*** | A string of numbers at the beginning of a calling party number, which can be numbers or "*" (indicating any string). |
| ***CalleeID Prefix*** | A string of numbers at the beginning of a called party number, which can be numbers or "*" (indicating any string). |

| | |
|---|---|
| ***Original CalleeID Range*** | The range of the original CalleeID in the Original CalleeID Pool. It must be filled in with numbers and can not be left empty. |

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

Click ***Modify*** in Figure 3-57 to modify the calling/called party number parameter. See Figure 3-59, for the original CalleeID modification interface. The configuration items on this interface are the same as those on the ***Add New Original CalleeID*** interface. Note that the item ***No.*** cannot be modified.

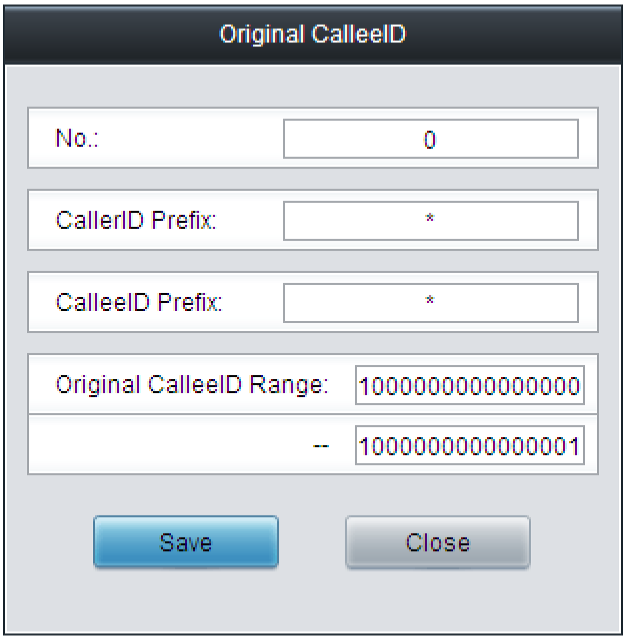



Figure 3-59 Modify Original CalleeID

**Note:** If there are two or more calling/called party numbers with the same prefix, Starting Original CalleeID and Number Amount corresponding to the one numbered the smallest are valid and all the others become invalid.

## 3.5.7 Redirecting Number Pool

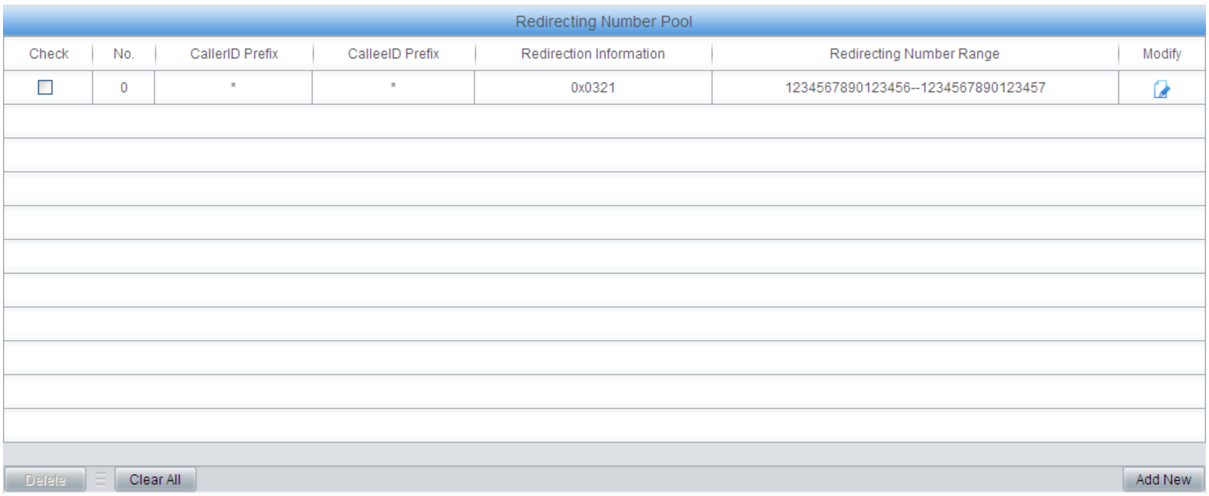



Figure 3-60 Redirecting Number Pool Interface

See Figure 3-60 for the Redirecting Number Pool interface, which is used to set the redirecting number in the setup message for all outgoing calls or some calls which contain a specified calling/called prefix. This feature is only applicable to ISUP calls.

A new redirecting number can be added by the Add New button. See Figure 3-61 for the redirecting number adding interface.

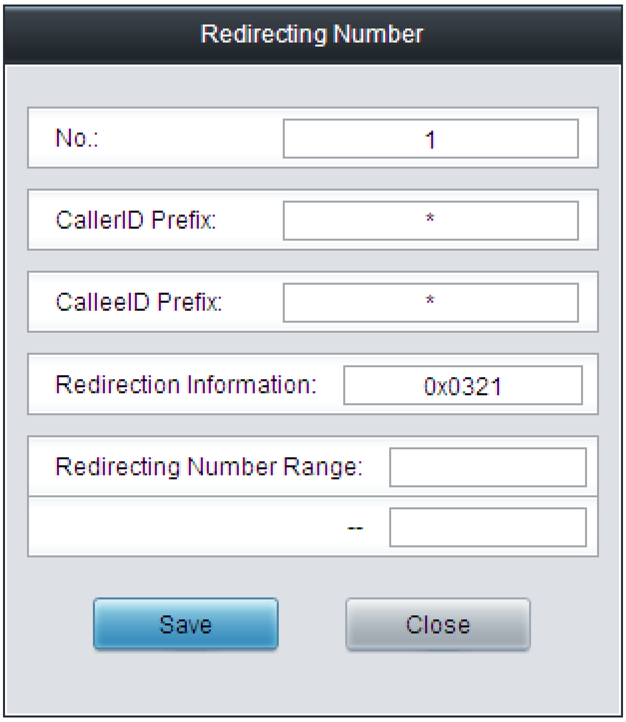



Figure 3-61 Add New Redirecting Number

The table below explains the items shown in above figures.

| Item | Description |
|---|---|
| ***No.*** | The corresponding number for an added redirecting number. The value range is 0~99. |
| ***CallerID Prefix*** | A string of numbers at the beginning of a calling party number, which can be numbers or "*" (indicating any string). |
| ***CalleeID Prefix*** | A string of numbers at the beginning of a called party number, which can be numbers or "*" (indicating any string). |
| ***Redirecting Information*** | Sets the redirection information field in the IAM message. The parameter type of the redirection information field is 0x13, which contains 2 bytes. By default, it is set to *0x0321*, i.e. call forwarding on no answer. Refer to the ISUP protocol standard for the detailed description of each byte. |
| ***Redirecting Number Range*** | The range of the redirecting number in the Redirecting Number Pool. It must be filled in with numbers and can not be left empty. |

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

Click ***Modify*** in Figure 3-60 to modify the redirecting number parameter. See Figure 3-62 for the redirecting number modification interface. The configuration items on this interface are the same as those on the ***Add New Redirecting Number*** interface. Note that the item ***No.*** cannot be modified.

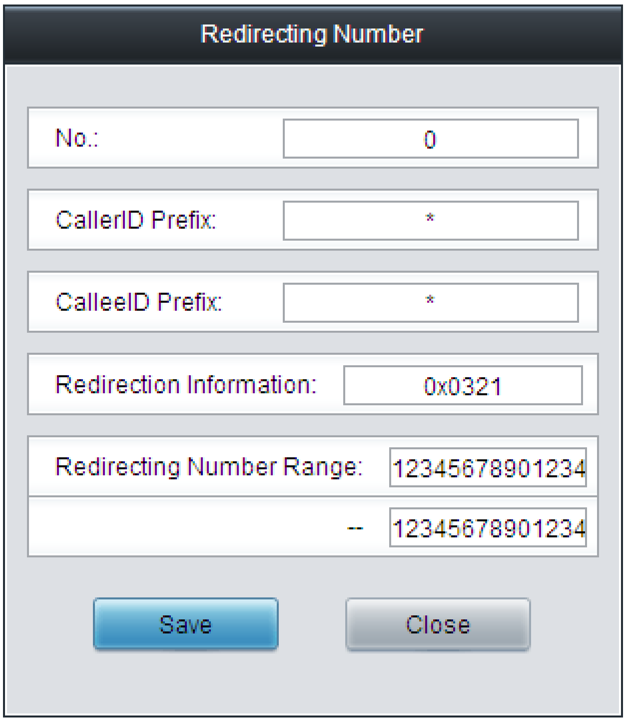



Figure 3-62 Modify Redirecting Number

To delete a redirecting number parameter, check the checkbox before the corresponding index in Figure 3-60 and click the '***Delete***' button. To clear all redirecting number parameters at a time, click the ***Clear All*** button in Figure 3-60.

**Note:** If there are two or more calling/called party numbers with the same prefix, only the one numbered the smallest are valid for Starting Redirecting Number and Number Amount.

## 3.5.8 SS7 Server

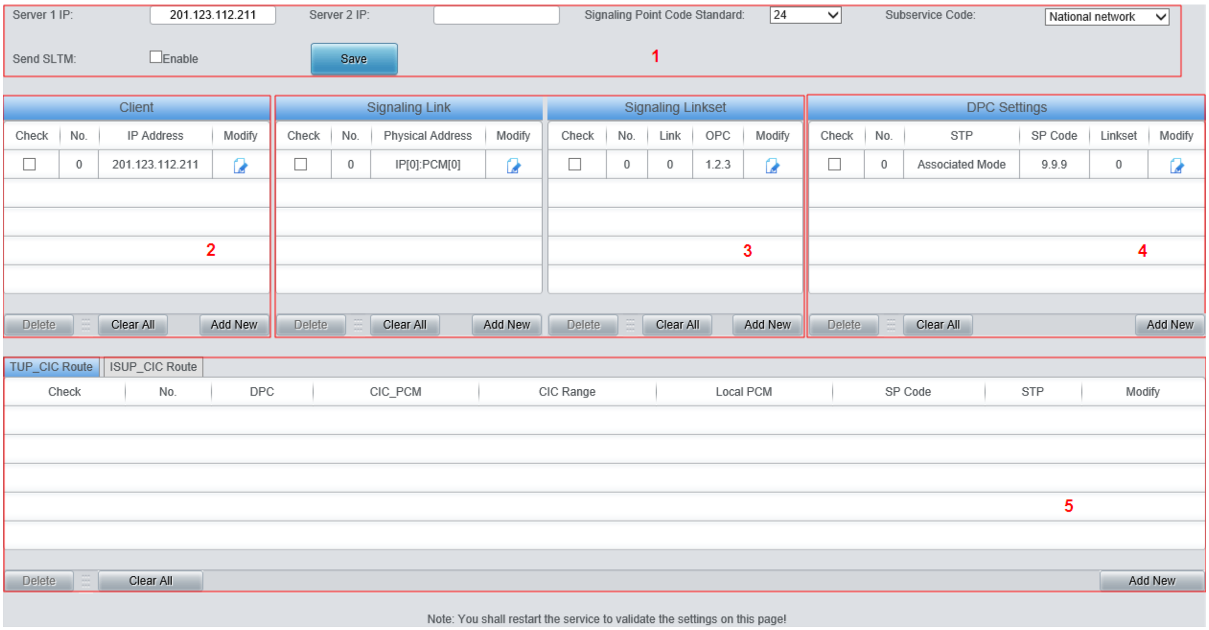



Figure 3-63 SS7 Server Configuration Interface

When the gateway uses the SS7 signaling, it must run the SS7 server first. See Figure 3-63 for the SS7 configuration interface, where you can set the SS7 server configuration file (Ss7server.ini). Follow the instructions below to accomplish the configurations step by step.

**Step 1:** Set Server IP and Signaling Point Code Standard. See Region 1 in Figure 3-63.

The table below explains these configuration items.

| Item | Description |
|---|---|
| ***Server 1 IP*** | Sets the IP address for the master SS7 server. If only one server is used in the system, there is no need to set the configuration item ***Server 2 IP***. |
| ***Server 2 IP*** | Sets the IP address for the slave SS7 server. |
| ***Signaling Point Code Standard*** | The value of this item varies on the PBX model. The optional values are 14 and 24, with the default value of 24. The China SS7 uses 24. |
| ***Subservice Code*** | Sets the SS7 subservice code. The optional values are: *International network*, *Spare international network*, *National network*, *Spare national network*, with the default value of *Spare national network*. |
| ***Send SLTM*** | Sets whether to regularly send the Signaling Link Test Message (SLTM) to the remote PBX. By default it is disabled. |

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

**Step 2:** Configure the client. See Region 2 in Figure 3-63.

A new client can be added by the ***Add New*** button on the bottom right corner of the client list. See Figure 3-64 for the new client adding interface.

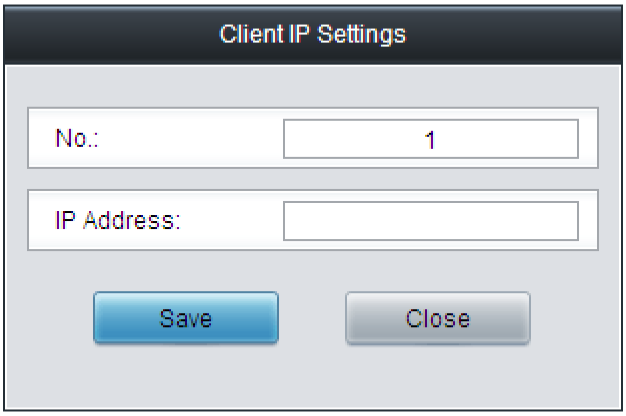



Figure 3-64 Add New Client

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

| Item | Description |
|---|---|
| ***No.*** | The unique index of each client, which is mainly used in the configuration of signaling links to correspond to the client, numbered from 0. |
| ***IP Address*** | IP address of the client. |

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

To modify a client, click ***Modify*** in the client list. The configuration items on the modification interface are the same as those on the ***Add New Client*** interface.

To delete a client, check the checkbox before the corresponding index and click the ***Delete*** button under the list. To clear all clients at a time, click the ***Clear All*** button. Note: If a client is occupied by a signaling link, it cannot be deleted or cleared unless you delete the signaling link first. You can only delete the clients in turn from back to front.

**Step 3:** Configure signaling links and linksets. See Region 3 in Figure 3-63.

The link used to transmit signaling messages between two signaling points is called Signaling

Link. Each signaling link maps a physical address. A new signaling link can be added by the ***Add New*** button on the bottom right corner of the signaling link list. See Figure 3-65 for the new signaling link adding interface.


Figure 3-65 Add New Signaling Link

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

| Item | Description |
|---|---|
| ***No.*** | The unique index of each signaling link, which is mainly used in the configuration of signaling linksets to correspond to the signaling link, numbered from 0. |
| ***Client*** | Client number. This configuration item together with ***PCM*** determines the physical address of the E1 interface of the signaling link. Each physical address maps a signaling link. |
| ***PCM*** | Local PCM number. This configuration item together with ***Client*** determines the physical address of the E1 interface of the signaling link. |

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

To modify a signaling link, click ***Modify*** in the signaling link list. The configuration items on the modification interface are the same as those on the ***Add New Signaling Link*** interface.

To delete a signaling link, check the checkbox before the corresponding index and click the ***Delete*** button under the list. To clear all signaling links at a time, click the ***Clear All*** button. Note: If a signaling link is occupied by a signaling linkset, it cannot be deleted or cleared unless you delete the signaling linkset first. You can only delete the signaling links in turn from back to front.

A group of signaling links used to connect two signaling points directly constitute a signaling linkset. A new signaling linkset can be added by the ***Add New*** button on the bottom right corner of the signaling linkset list. See Figure 3-66 for the new signaling linkset adding interface.


Figure 3-66 Add New Signaling Linkset

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

<table>
<tr><th>Item</th><th>Description</th></tr>
<tr><td>No.</td><td>The unique index of each signaling linkset, which is mainly used in the configuration of DPC to correspond to the signaling linkset, numbered from 0.</td></tr>
<tr><td>Link</td><td>The signaling links in the linkset. If the checkbox before a link is grey, it indicates that the link has been occupied.</td></tr>
<tr><td>OPC</td><td>Originating Point Code for the signaling server which is usually allocated by the central office,. See the table below for the format and the value range:
<table>
<tr><td></td><td>14 bit</td><td>24 bit</td></tr>
<tr><td>Decimal (a.b.c)</td><td>a, c: 0~7, b: 0~255</td><td>a, b, c: 0~255</td></tr>
<tr><td>Hexadecimal (abc)</td><td>a, c: 3-digit hexadecimal number, b: 8-digit hexadecimal number</td><td>a, b, c: hexadecimal number inbetween 00~ff</td></tr>
</table>
</td></tr>
</table>

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

To modify a signaling linkset, click ***Modify*** in the signaling linkset list. The configuration items on the modification interface are the same as those on the ***Add New Signaling Linkset*** interface.

To delete a signaling linkset, check the checkbox before the corresponding index and click the ***Delete*** button under the list. To clear all signaling linkset at a time, click the ***Clear All*** button. Note: If a signaling linkset is occupied by a DPC, it cannot be deleted or cleared unless you delete the DPC first. You can only delete the signaling linksets in turn from back to front.

**Step 4:** Configure DPC. See Region 4 in Figure 3-63.

The signaling point that receives messages is called Destination Point Code (DPC). A new DPC can be added by the ***Add New*** button on the bottom right corner of the DPC list. See Figure 3-67 for the new DPC adding interface.

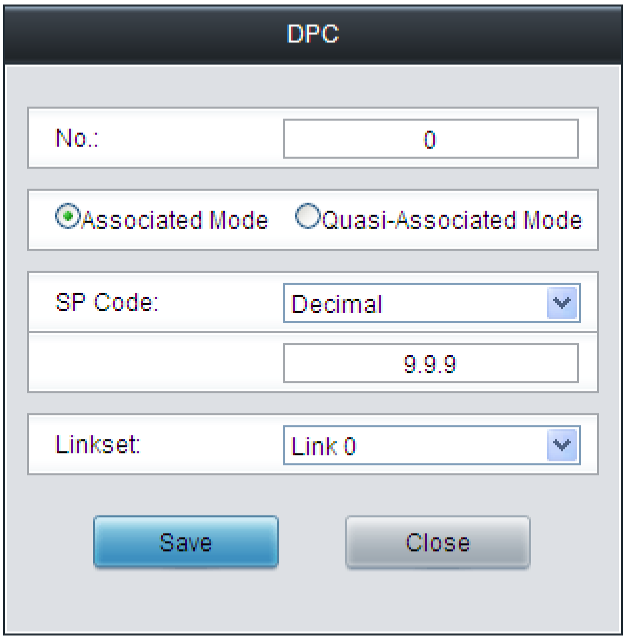
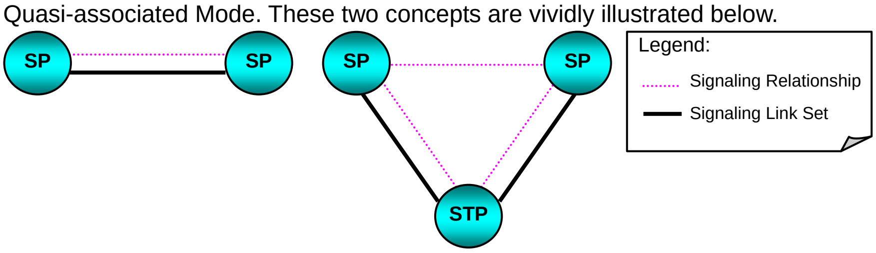



Figure 3-67 Add New DPC

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

| Item | Description |
|---|---|
| ***No.*** | The unique index of each DPC, which is mainly used in the configuration of TUP_CIC Route or ISUP_CIC Route to correspond to the DPC, numbered from 0. |
| ***Associated Mode/ Quasi-associated Mode*** | Sets the way to transmit signaling messages between two signaling points, including *Associated Mode* and *Quasi-associated Mode*. Directly connecting the signaling links between two signaling points to transmit the inbetween signaling messages is called Associated Mode. Connecting two or more than two signaling links serially via one or more than one signaling transport points to transmit signaling messages, provided the path of signaling messages through the signaling network is predetermined and fixed within a certain period of time, is called Quasi-associated Mode. These two concepts are vividly illustrated below.<br> <br>(a) Associated Mode (b) Quasi-associated Mode |
| ***SP Code*** | Signaling point code of the DPC, usually allocated by the central office. |
| ***STP*** | Sets the first STP (signaling transport point) the signaling message reaches during the transmission under the quasi-associated mode. Only when you select the quasi-associated mode can this item be seen and configured. |
| ***Linkset*** | The linkset which is used to transmit signaling messages. For the associated mode, this item sets the signaling linksets between the OPC and the DPC. For the quasi-associated mode, this item sets the signaling linksets between the OPC and the first STP (signaling transport point). |

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

To modify a DPC, click ***Modify*** in the DPC list. The configuration items on the modification interface are the same as those on the ***Add New DPC*** interface.

To delete a DPC, check the checkbox before the corresponding index and click the ***Delete*** button under the list. To clear all DPCs at a time, click the ***Clear All*** button. Note: If a DPC is occupied by a CIC routing rule, it cannot be deleted or cleared unless you delete the routing rule first. You can only delete the DPCs in turn from back to front.

**Step 5:** Configure TUP_CIC or ISUP_CIC Route. See Region 5 in Figure 3-63.

A new TUP_CIC routing rule can be added by the ***Add New*** button on the bottom right corner of the TUP_CIC routing rule list. See Figure 3-68 for the new TUP_CIC routing rule adding interface.

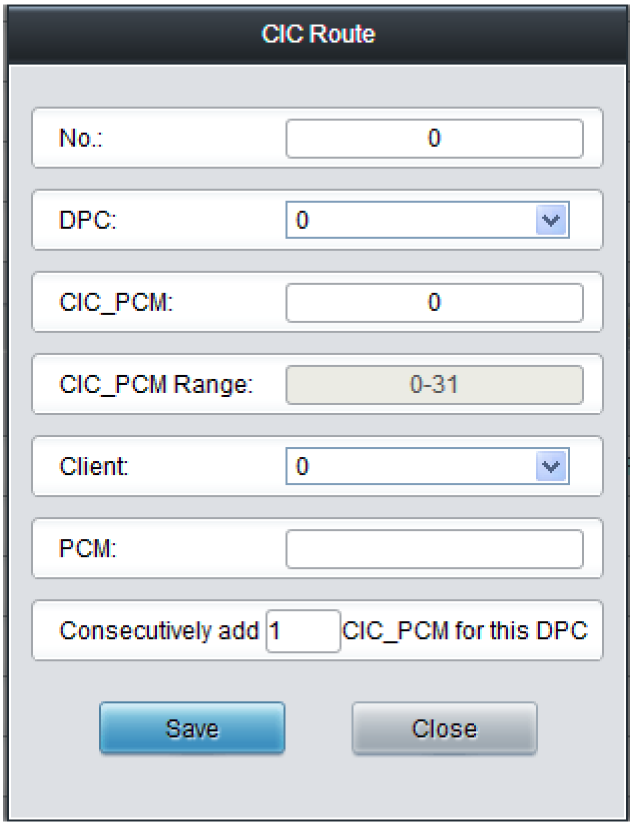



Figure 3-68 Add New TUP_CIC Routing Rule

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

| Item | Description |
|---|---|
| ***No.*** | The unique index of each CIC routing rule, which is numbered from 0. |
| ***DPC*** | DPC used in the routing rule. |
| ***CIC_PCM*** | PCM number in the CIC field and the value is obtained by dividing the initial CIC number from the central office by 32. |
| ***CIC_PCM Range*** | Range of the PCM time slots corresponding to CIC. |
| ***Client*** | Client number. This configuration item together with ***PCM*** determines the local PCM in the CIC routing rule. |
| ***PCM*** | PCM number on the client. |

| | |
|---|---|
| ***Consecutively add _CIC_PCM for this DPC*** | Consecutively adds one or more CIC_PCM routes for a DPC. |

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

To modify a routing rule, click ***Modify*** in the TUP_CIC routing rule list. The configuration items on the modification interface are the same as those on the ***Add New TUP_CIC Routing Rule*** interface.

To delete a routing rule, check the checkbox before the corresponding index and click the ***Delete*** button under the list. To clear all routing rules at a time, click the ***Clear All*** button.

For the ISUP_CIC route settings, click the ISUP_CIC Route tab in Region 5 in Figure 3-63. See Figure 3-69 for the ISUP_CIC route settings interface. The configuration items and operations on this interface are absolutely the same as those in the TUP_CIC route settings interface. Note: Besides the default setting, the CIC Range for ISUP_CIC route can also be user-defined.

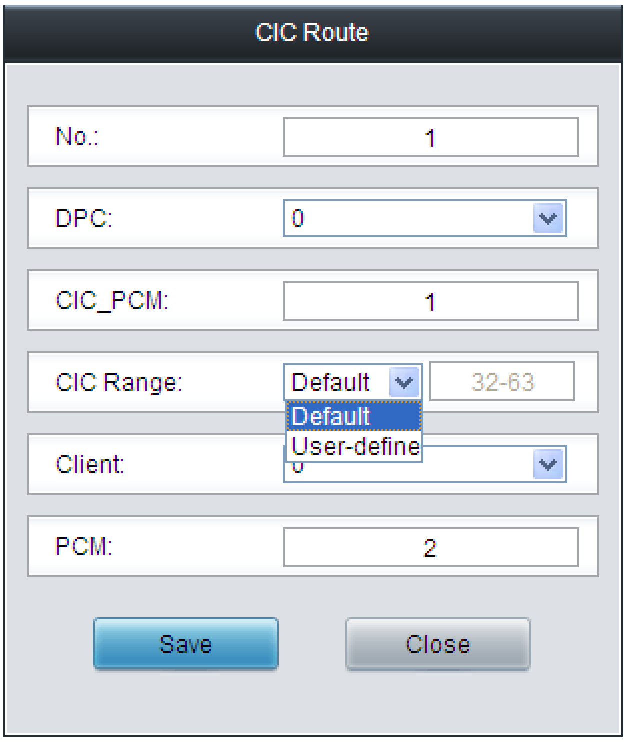



Figure 3-69 ISUP_CIC Route Settings Interface

After completing the configurations on ***SS7 Server Configuration Interface*** (Figure 3-63), you shall restart the service to validate them. Refer to 3.12.17 Restart for detailed instructions.

## 3.6 ISDN Settings

Users can see the ISDN option in the menu only when the configuration item ***Signaling Protocol*** on the PCM settings interface is set to *ISDN User Side* or *ISDN Network Side*. See Figure 3-70.

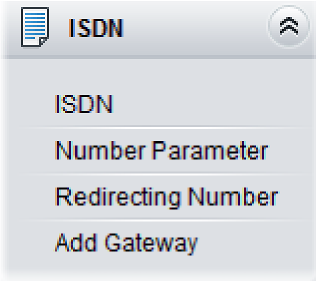



Figure 3-70 ISDN Settings

### 3.6.1 ISDN

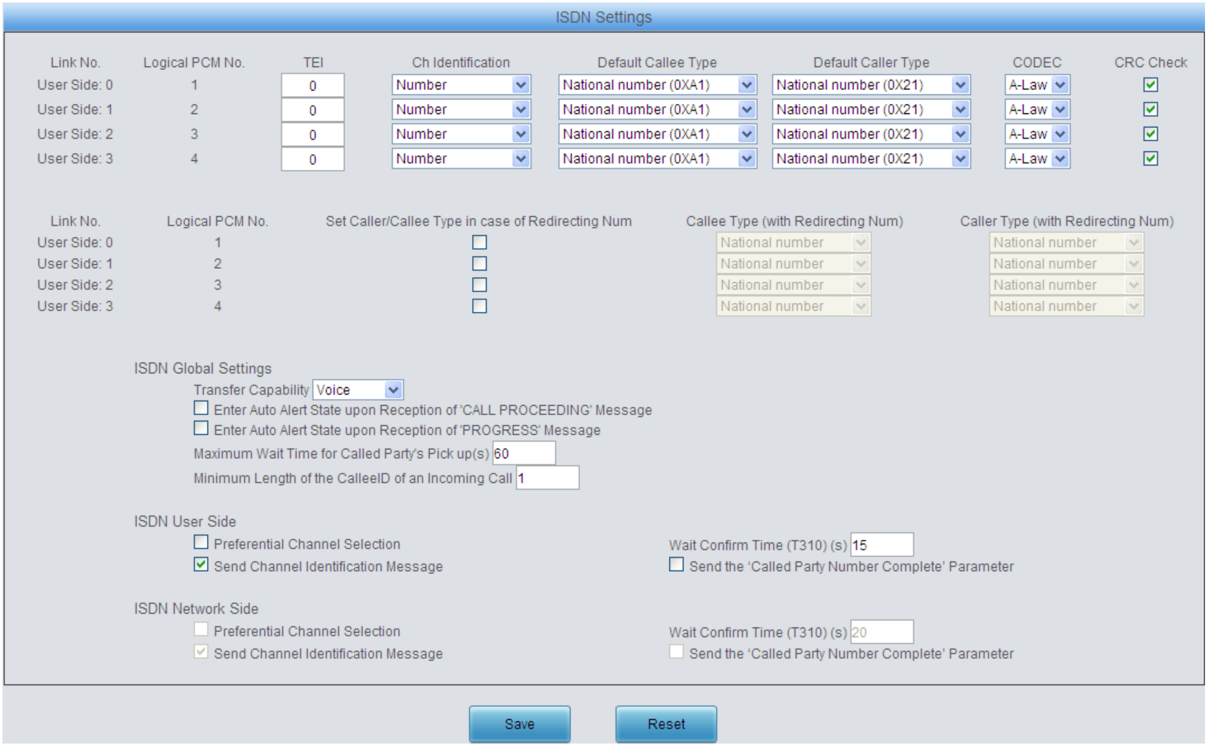



Figure 3-71 ISDN Settings Interface

See Figure 3-71 for the ISDN settings interface where users can configure the general ISDN 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 3.12.17 Restart for detailed instructions. The table below explains the items shown in Figure 3-71.

| Item | Description |
|---|---|
| ***TEI*** | Terminal Equipment Identifier, which is used to identify the service access point in the point-to-point data link connection. Range of value: 0~63, with the default value of 0. Note: The TEI values at the corresponding user side and the network side must be the same. |
| ***Ch Identification*** | Sets the way to represent channel identification messages on the digital trunk. The optional values are: *Number* and *Time slot diagram*, with the default value of *Number*. |

| | |
|---|---|
| ***Default Callee Type*** | Sets the type of number and numbering scheme for the called party numbers in the SETUP message during the outgoing call. The optional values are: *National number*, *International number*, *Network number*, *Subscriber number* and *Unknown*, with the default value of *National number*. |
| ***Default Caller Type*** | Sets the type of number and numbering scheme for the calling party numbers in the SETUP message during the outgoing call. The optional values are: *National number*, *International number*, *Network number*, *Subscriber number* and *Unknown*, with the default value of *National number*. |
| ***CODEC*** | Sets the voice CODEC used on the digital trunk. The optional values are *A-Law* and *u-Law*, with the default value of *A-Law*. |
| ***CRC Check*** | Sets whether to enable the feature of CRC check for the digital trunk at ISDN user side or network side. By default this feature is enabled. |
| ***Set Caller/Callee Type in case of Redirecting Num*** | Once this feature is enabled, if the IP end carries the redirecting number in a call from IP to PSTN, you shall set separate values for the type of number and numbering scheme for the calling and called party numbers in the SETUP message, i.e. ***Callee Type (with Redirecting Num)*** and ***Caller Type (with Redirecting Num)***. By default this configuration item is disabled. |
| ***Callee Type (with Redirecting Num)*** | This item is valid only when ***Set Caller/Callee Type in case of Redirecting Num*** is enabled. It sets the type of number and numbering scheme for the called party numbers in the SETUP message when the IP end carries the redirecting number in a call from IP to PSTN. The optional values are: *National number*, *International number*, *Network number*, *Subscriber number* and *Unknown*, with the default value of *National number*. |
| ***Caller Type (with Redirecting Num)*** | This item is valid only when ***Set Caller/Callee Type in case of Redirecting Num*** is enabled. It sets the type of number and numbering scheme for the calling party numbers in the SETUP message when the IP end carries the redirecting number in a call from IP to PSTN. The optional values are: *National number*, *International number*, *Network number*, *Subscriber number* and *Unknown*, with the default value of *National number*. |
| ***Transfer Capability*** | Sets the 'Transfer Capability' filed in the signaling message. The optional values are *Voice* and *3.1k Audio*, with the default value of *Voice*. |
| ***Enter Auto Alert State upon Reception of 'CALL PROCEEDING' Message*** | If this item is checked, the system will go into the state of auto alert when it receives the 02 (CALL PROCEEDING) message and the progress indicator turns to be 8 or 1. By default this item is disabled. |
| ***Enter Auto Alert State upon Reception of 'PROGRESS' Message*** | If this item is checked, the system will go into the state of auto alert when it receives the 03 (PROGRESS) message and the progress indicator turns to be 8 or 1. By default this item is disabled. |
| ***Maximum Wait Time for Called Party's Pick up*** | The maximum time waiting for the called party to pick up the call after the channel state turns to 'WaitAnswer' during an outgoing call. The default value is 60, calculated by s. |

| | |
|---|---|
| ***Minimum Length of the CalleeID of an Incoming Call*** | Sets the minimum length of the CalleeID under the fixed-length mode. The value range is 1≤n≤40. Provided it is set to n, that is, the local end has received all the n digits of the called party number of the incoming call, the number reception will be regarded as finished. |
| ***Preferential Channel Selection*** | Sets whether to allow the preferential channel selection. By default this item is unchecked. |
| ***Send Channel Identification Message*** | Sets whether the channel identification message is included in the corresponding reply message (such as CALL PROCEEDING, ALERT, etc.) after the local end receives the SETUP message from the remote PBX during an incoming call. By default this item is checked. |
| ***Wait Confirm Time (T310)*** | Sets the maximum time that the local end waits for the remote end to send back the acknowledgement message in an outgoing call. If no acknowledgement message is received within the specified time period, the local end will disconnect the call automatically. For ISDN User Side, the default value is 15; for ISDN Network Side, the default value is 20, calculated by s. |
| ***Send the 'Called Party Number Completed' Parameter*** | Sets whether to include or not the 'Called Number Complete' parameter in the SETUP message during an outgoing call. |

### 3.6.2 Number Parameter

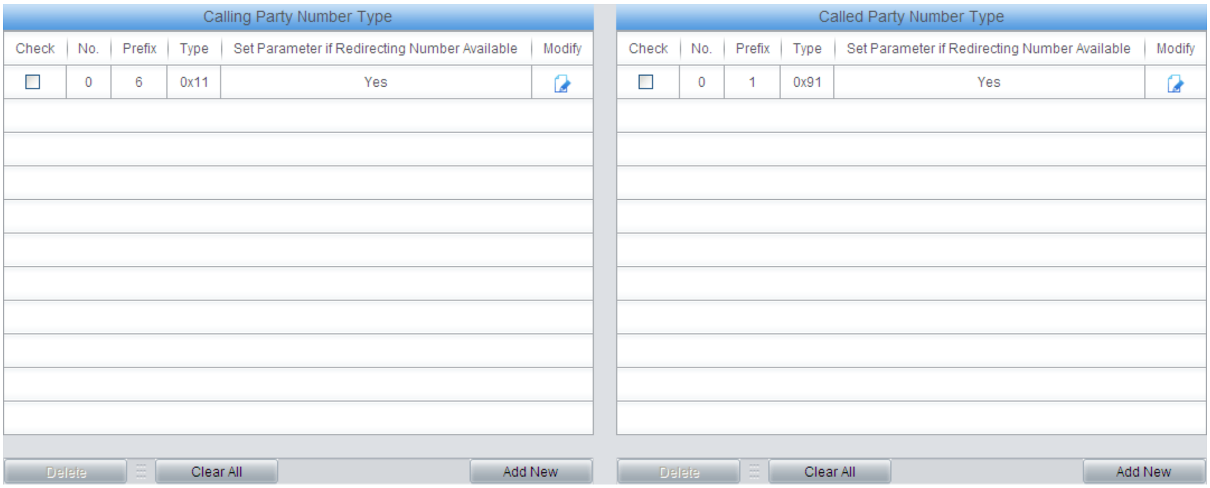



Figure 3-72 Number Parameter Configuration Interface

Number Parameter for ISDN is almost the same as that for SS7; only the calling/called party number changes from SS7 to ISDN; "set parameter if original CalleeID available" changes to "set parameter if redirecting number available" in ISDN. See Figure 3-72 for Number Parameter for ISDN. The configuration items on this interface are the same as those on Number Parameter for SS7 (Figure 3-53, Figure 3-54).

### 3.6.3 Redirecting Number

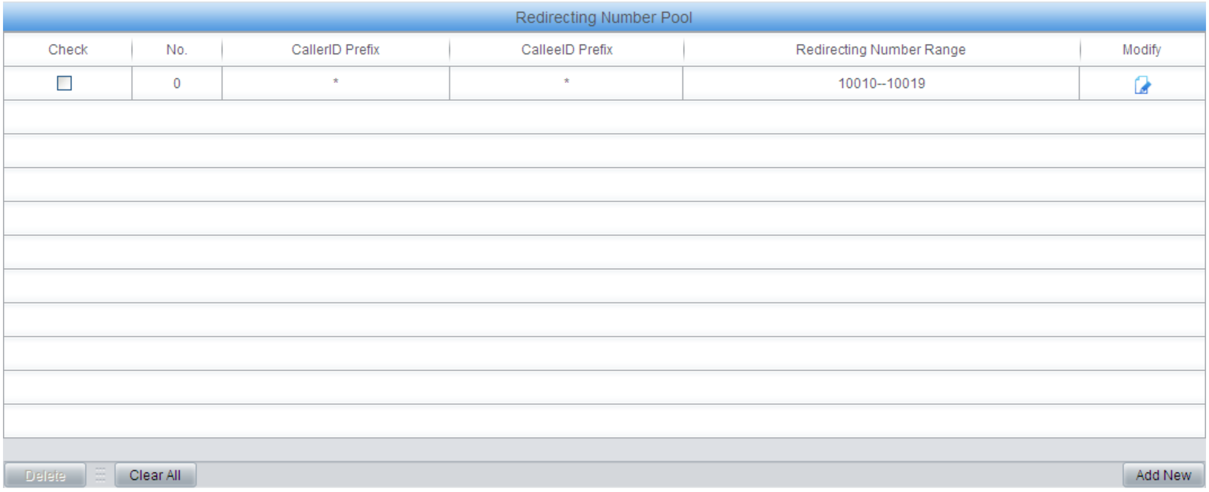



Figure 3-73 Redirecting Number Interface

Redirecting Number Pool for ISDN is almost the same as Original CalleeID Pool for SS7; only the calling/called party number changes from SS7 to ISDN. See Figure 3-73 for Redirecting Number Pool for ISDN. The configuration items on this interface are the same as those on original CalleeID pool for SS7 (Figure 3-58).

### 3.6.4 Add Gateway


Figure 3-74 Add Gateway Interface

See Figure 3-74 for the Add Gateway Interface. A new gateway can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. The information about the added gateway will be displayed under ***Operation Info → PSTN Status***. See Figure 3-75 for the gateway adding interface.

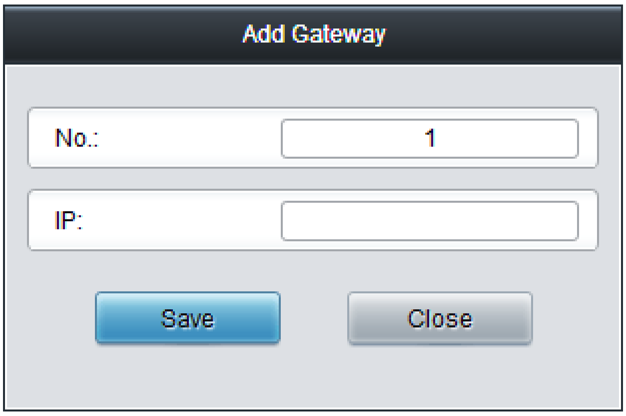



Figure 3-75 Add New Gateway

The table below explains the items shown in above figures.

| Item | Description |
|---|---|
| ***No.*** | The corresponding number for a new gateway, which starts from 0. |
| ***IP*** | The corresponding IP address for the new gateway, which must be in the same network section of the SIP address set via **VoIP→SIP.** |

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

Click ***Modify*** in Figure 3-74 to modify the gateway information. See Figure 3-76 for the gateway modification interface. The configuration items on this interface are the same as those on the ***Add New Gateway*** interface.

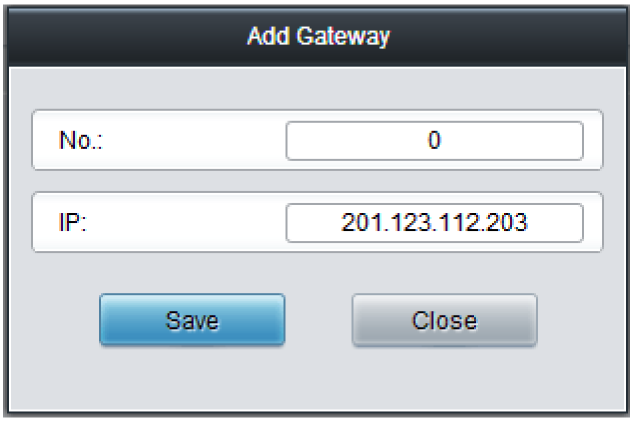



Figure 3-76 Modify Gateway Information

## 3.7 SS1 Settings

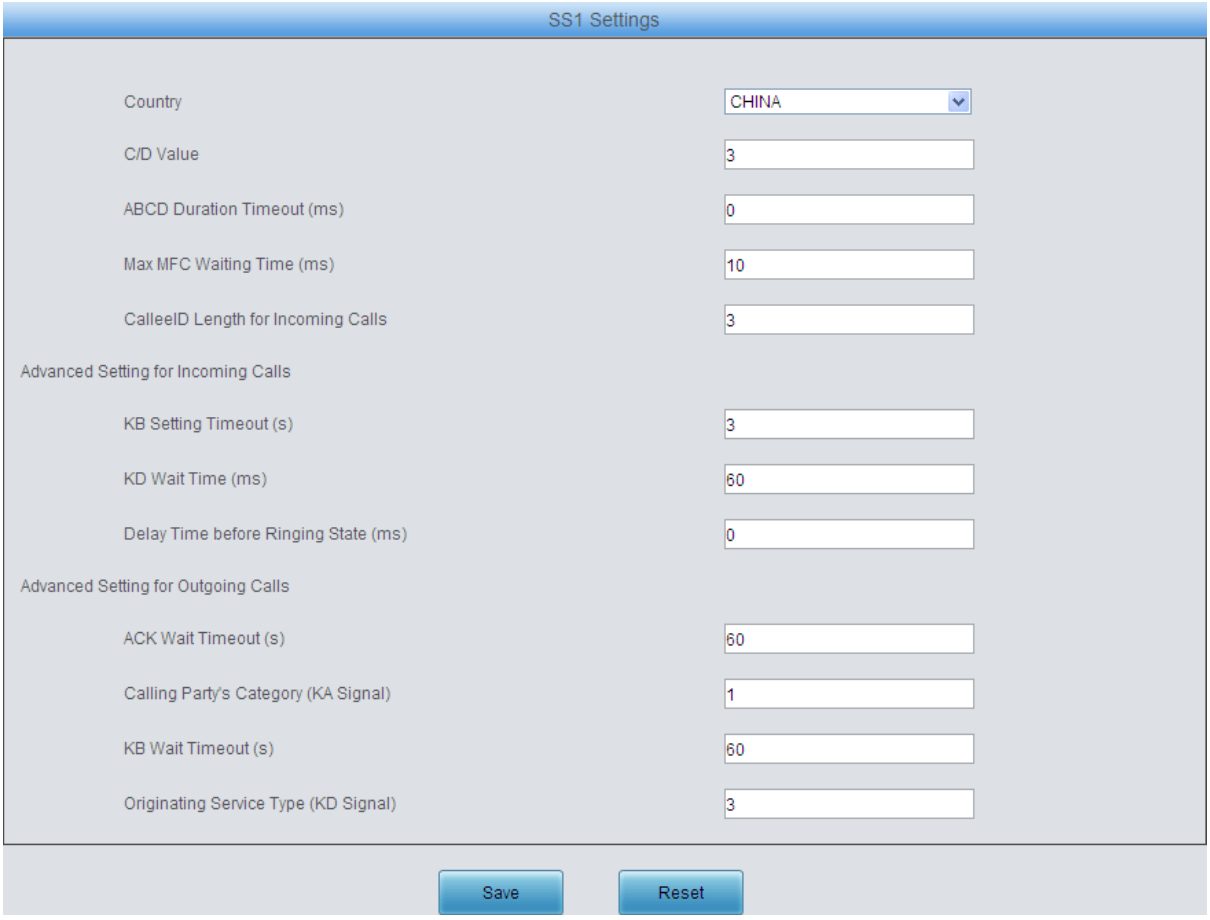



Figure 3-77 SS1 Settings Interface

See Figure 3-77 for the SS1 settings interface. This interface appears only when the configuration item ***Signaling Protocol*** on the PCM settings interface is set to *SS1*. You can set general information of SS1. 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 3.12.17 Restart for detailed instructions. The table below explains the items shown in Figure 3-77.

| Item | Description |
|---|---|
| ***Country*** | Sets the country to use SS1, with the default value of *CHINA*. |
| ***C/D Value*** | Sets the value of CD in the ABCD signaling codes sent by the local end to the remote PBX. The high 6 bits should be set to 0, being reserved; the low 2 bits are C/D signaling codes, Bit1 (Signaling Code C) and Bit0 (Signaling Code D), both with the default value of 1. |
| ***ABCD Duration Timeout*** | Sets the minimum duration of ABCD signaling codes sent out by the remote PBX, calculated by millisecond (ms), which has to be the multiple of 8, with the default value of 0. Only when the on-line ABCD signaling codes vary and the new value keeps for more than the time specified by this configuration item will the gateway confirm the change of ABCD codes, Otherwise, the driver will believe there are undesired dithering signals on the line. |

| | |
|---|---|
| ***Max MFC Waiting Time*** | Sets the maximum waiting time, i.e. the timer T2 for the SS1 state machine, calculated by second, with the default value of *10*. |
| ***CalleeID Length for Incoming Calls*** | Sets the way to receive the number, with the default value of *3 which means receiving all the 3 digits of the called party number of the incoming call will put the local number reception into an end*. |
| ***KB Setting Timeout*** | Sets the maximum time to wait for the application to configure the KB signal, calculated by second, with the default value of *3*. |
| ***KD Wait Time*** | Sets the maximum time to wait for the remote PBX to send the KD signal (i.e. the timer T3) in the SS1 channel state machine, calculated by second, with the default value of *60*. |
| ***Delay Time before Ringing State*** | Sets the delay time, i.e. a period of waiting time before the channel goes into the 'Ringing' state following the reception of the complete called party number in case the SS1 channel on the gateway serves as the incoming end. It is calculated by second, with the default value of 0. |
| ***ACK Wait Timeout*** | Sets the value of the timer T5, calculated by second, with the default value of *60*. |
| ***Calling Party's Category (KA Signal)*** | Sets the KA signal (calling party's category at the local end) sent in an outgoing call. The value range is 1~10, with the default value of *1 (ordinary/regular)*. |
| ***KB Wait Timeout*** | Sets the maximum time to wait for the KB signal from the remote PBX, calculated by second, with the default value of *60*. |
| ***Originating Service Type (KD Signal)*** | Sets the originating service type, i.e. KD, for an outgoing call. The value range is 1~6, with the default value of *3 (local call)*. |

## 3.8 Fax Settings

See Figure 3-78 for the Fax Settings interface which is used to modify the special fax configurations.

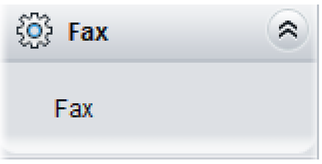



Figure 3-78 Fax Settings

### 3.8.1 Fax

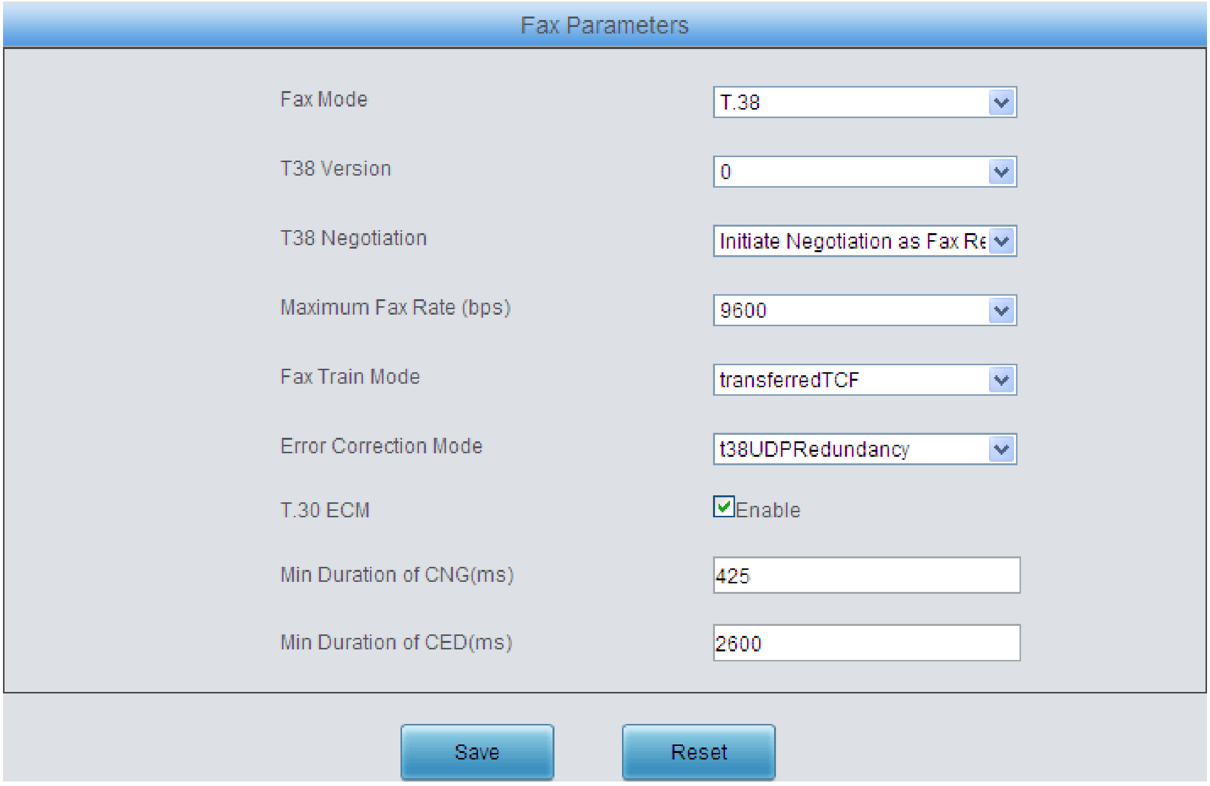



Figure 3-79 Fax Configuration Interface (T.38 Mode)

See Figure 3-79 for the fax configuration interface with all default settings under the T.38 fax 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 3.12.17 Restart for detailed instructions. The table below explains the configuration items in Figure 3-79.

| Item | Description |
|---|---|
| ***Fax Mode*** | The real-time IP fax mode. The optional values are *T.38*, *Pass-through* and *Disable*, with the default value of *T.38*. Setting this item to *Disable* means to disable both T.38 and Pass-through. |
| ***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*** | Sets the Negotiation mode of T.38, including: *Unsupported*, *Initiate Negotiation as Fax Sender* and *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 9600. |
| ***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 3-80.

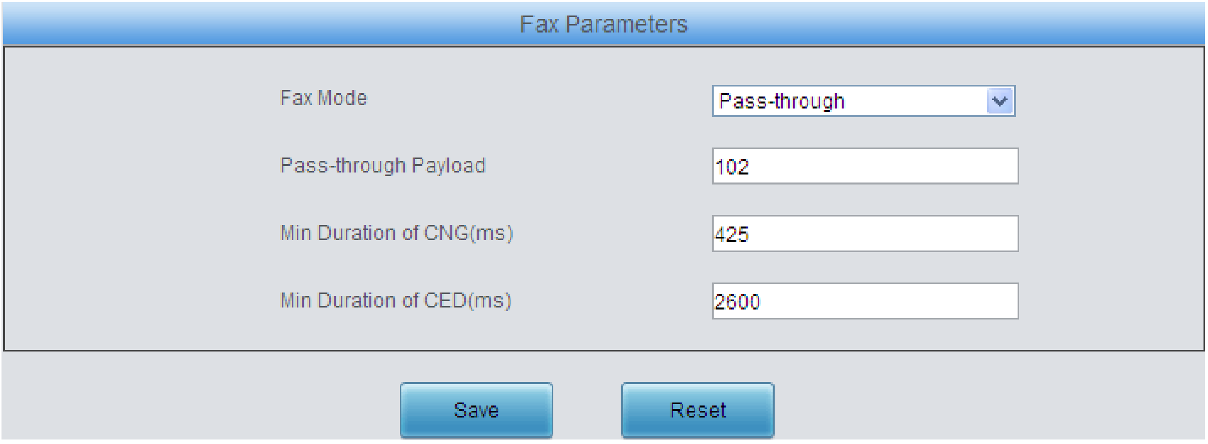



Figure 3-80 Fax Configuration Interface (Pass-through Mode)

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. |

## 3.9 Route Settings

Route Settings is used to specify the routing rules for calls on two directions: IP→PSTN and PSTN→IP. See Figure 3-81.

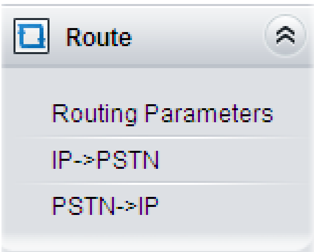



Figure 3-81 Route Settings

### 3.9.1 Routing Parameters

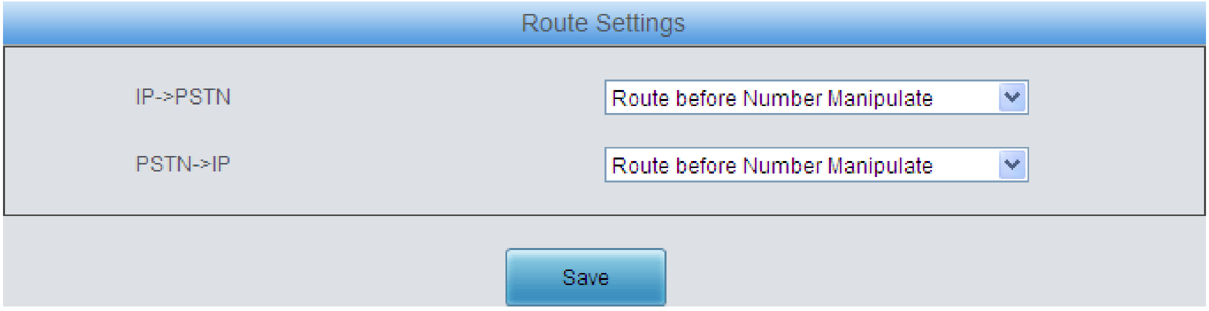



Figure 3-82 Routing Parameters Configuration Interface

See Figure 3-82 for the routing parameters configuration interface. On this interface, you can set the routing rules for calls respectively on two directions IP→PSTN and PSTN→IP to be routing before or after number manipulation. The default value is *Route before Number Manipulate*.

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

### 3.9.2 IP to PSTN


Figure 3-83 IP→PSTN Routing Rule Configuration Interface

See Figure 3-83 for the IP→PSTN routing rule configuration interface. A new routing rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-84 for the IP→PSTN routing rule adding interface.

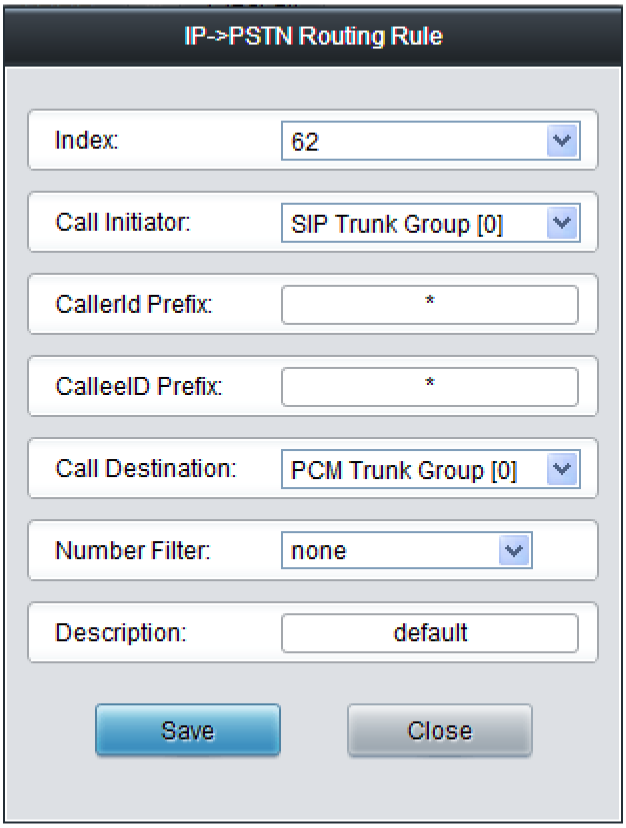


Figure 3-84 Add New Routing Rule (IP→PSTN)

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. |
| ***Call Initiator*** | SIP trunk group from where the call is initiated. This item can be set to a specific SIP trunk group or SIP Trunk Group [ANY] which indicates any SIP trunk group. |
| ***CallerID Prefix, CalleeID Prefix*** | A string of numbers at the beginning of the calling/called party number. This item can be set to a specific string or "*" which indicates any string. These two configuration items together with ***Call Initiator*** can specify the calls which apply to a routing rule.<br>**Note:** Multiple rules are supported for CallerID/CalleeID prefix. They are separated by ":". |
| ***Call Destination*** | PCM trunk group to which the call will be routed. |
| ***Number Filter*** | Number filter rule which will be applicable to this route. It is set in ***Number Filter***. See 3.10.4 Filtering Rule for details. |
| ***Description*** | More information about each routing rule. |

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

Click ***Modify*** in Figure 3-83 to modify a routing rule. See Figure 3-85 for the IP→PSTN routing rule modification interface. The configuration items on this interface are the same as those on the ***Add New Routing Rule (IP→PSTN)*** interface. Note that the item ***Index*** cannot be modified.

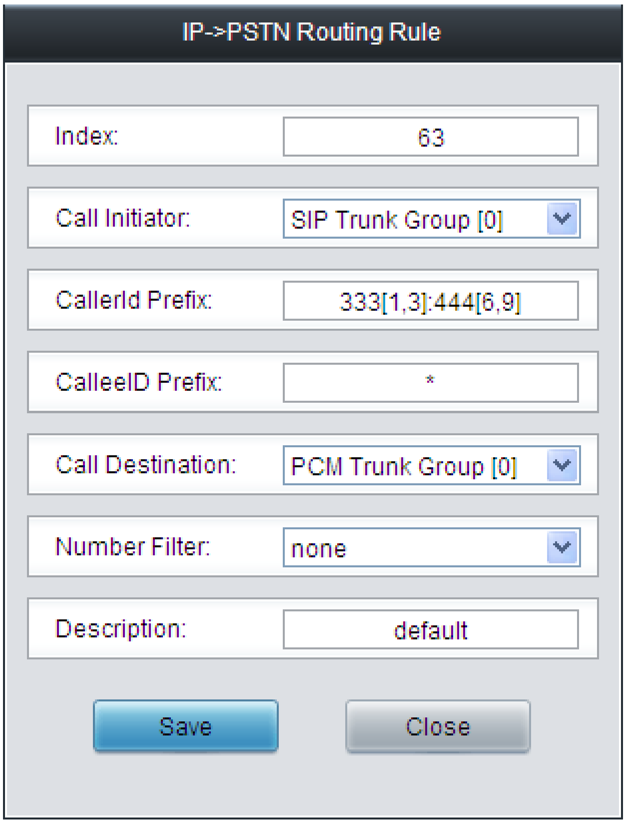



Figure 3-85 Modify Routing Rule (IP→PSTN)

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-83 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 3-83.

## 3.9.3 PSTN to IP

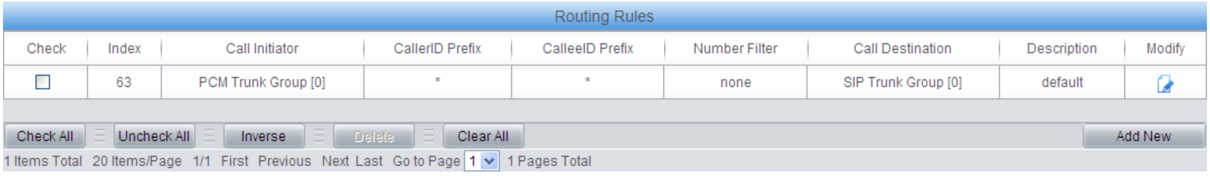



Figure 3-86 PSTN→IP Routing Rule Configuration Interface

See Figure 3-86 for the PSTN→IP routing rule configuration interface. A new routing rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-87 for the PSTN→IP routing rule adding interface.

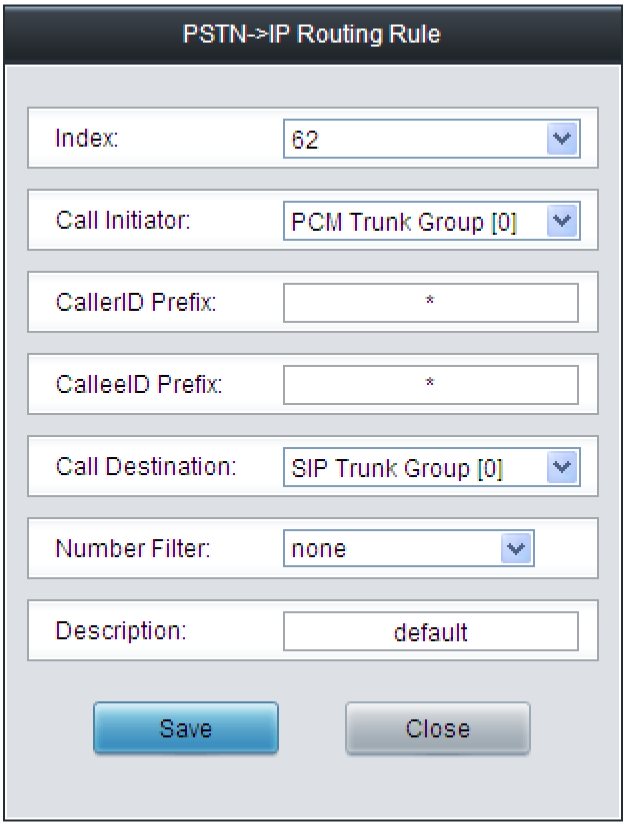



Figure 3-87 Add New Routing Rule (PSTN→IP)

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. |
| ***Call Initiator*** | PCM trunk group from which the call is initiated. This item can be set to a specific PCM trunk group or PCM Trunk Group [ANY] which indicates any PCM trunk group. |

| | |
|---|---|
| ***CallerID Prefix, CalleeID Prefix*** | A string of numbers at the beginning of the calling/called party number. This item can be set to a specific string or "*" which indicates any string. These two configuration items together with ***Call Initiator*** can specify the calls which apply to a routing rule.<br>**Note:** Multiple rules are supported in callerID/calleeID prefix. They should be separated by ":". |
| ***Call Destination*** | SIP trunk group to which the call will be routed. |
| ***Number Filter*** | Number filter rule which will be applicable to this route. It is set in ***Number Filter***. See 3.10.4 Filtering Rule for detailed setting. |
| ***Description*** | More information about each routing rule. |

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

Click ***Modify*** in Figure 3-86 to modify a routing rule. See Figure 3-88 for the PSTN→IP routing rule modification interface. The configuration items on this interface are the same as those on the ***Add New Routing Rule (PSTN→IP)*** interface. Note that the item ***Index*** cannot be modified.

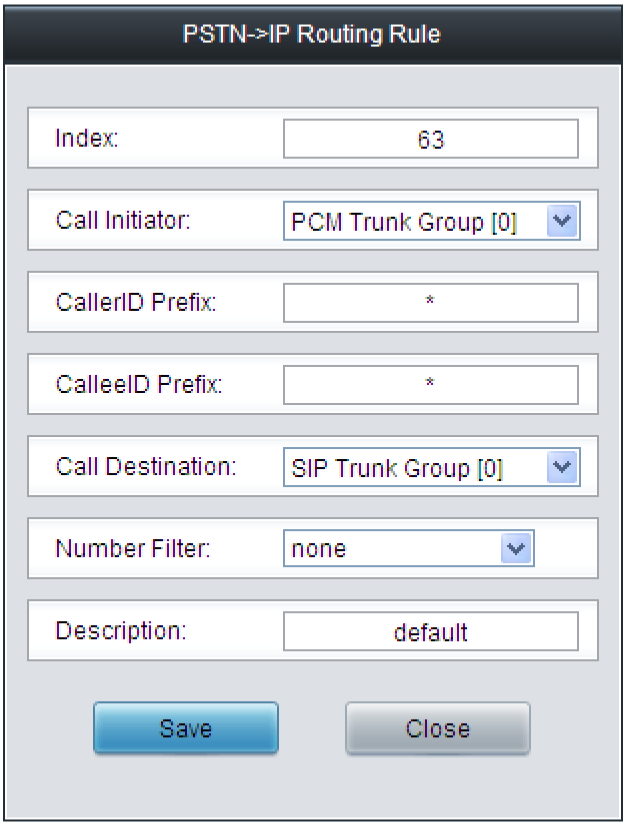



Figure 3-88 Modify Routing Rule (PSTN→IP)

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-86 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 3-86.

## 3.10 Number Filter

Number Filter includes four parts: ***Whitelist***, ***Blacklist***, ***Number Pool*** and ***Filtering Rule***. See Figure 3-89.

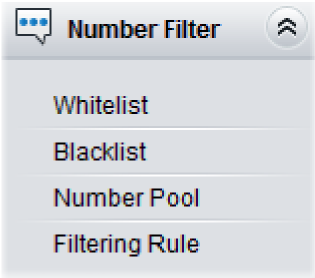



Figure 3-89 Number Filter Interface

## 3.10.1 Whitelist

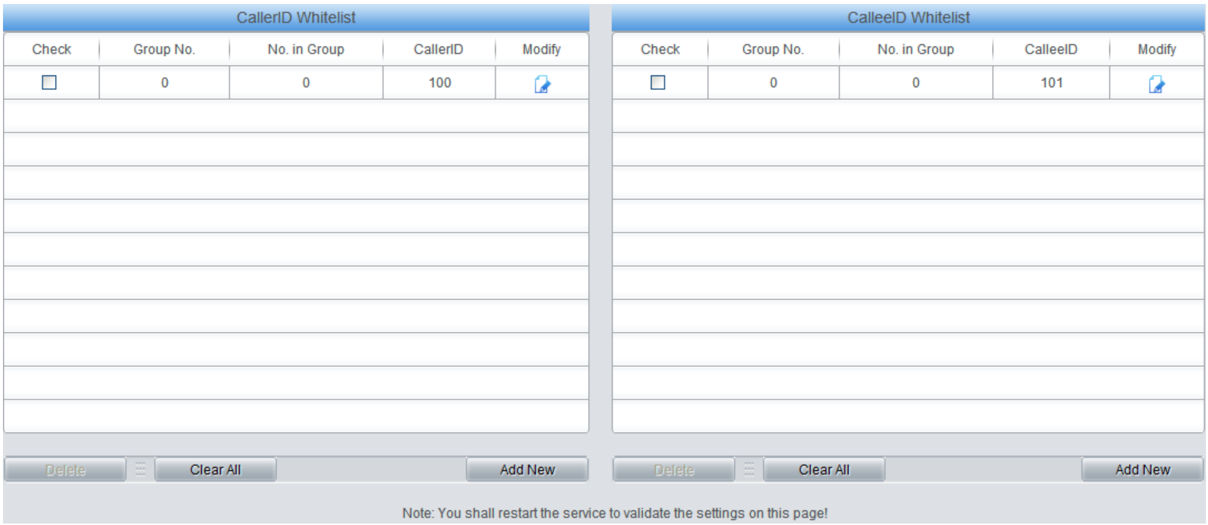



Figure 3-90 Whitelist Setting Interface

See Figure 3-90 for the Whitelist Setting Interface, which includes two parts: ***CallerID Whitelist*** and ***CalleeID Whitelist***.

A new CallerID/CalleeID whitelist can be added by the ***Add New*** button. See Figure 3-91, Figure 3-92 for CallerID/CalleeID whitelist adding interface.


Figure 3-91 Add New CallerIDs in Whitelist Interface


Figure 3-92 Add New CalleeIDs in Whitelist Interface

The table below explains the items shown in above figures.

| Item | Description |
|---|---|
| ***Group*** | The corresponding Group ID for CallerIDs/CalleeIDs in the whitelist. The value range is 0~7. |
| ***No. in Group*** | The corresponding No. for different CallerIDs/CalleeIDs in a same group. It is allowed to set up to 100 numbers in one group. |
| ***CallerID*** | CallerID in the whitelist, which must be filled in with numbers or "*" (indicating any string) and can not be left empty. Example: 135*1 denotes any CallerIDs which start from 135 and end with 1 will be accepted. |
| ***CalleeID*** | CalleeID in the whitelist, which must be filled in with numbers or "*" (indicating any string) and can not be left empty. Example: 135*1 denotes any CalleeIDs which start from 135 and end with 1 will be accepted. |

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

Click ***Modify*** in Figure 3-90 to modify the CallerID or CalleeID whitelist. See Figure 3-93, Figure 3-94 for CallerIDs/CalleeIDs on the Whitelist Modification interface. The configuration items on this interface are the same as those on the ***Add New CallerIDs/CalleeIDs in Whitelist*** interface. The item *Group No.* cannot be modified.

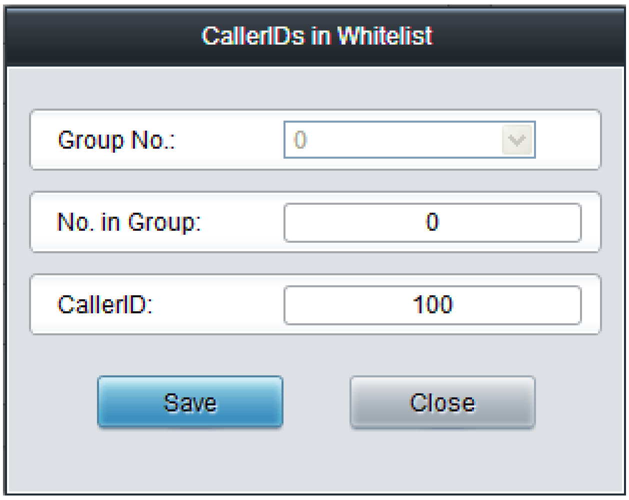



Figure 3-93 Modify CallerIDs in Whitelist

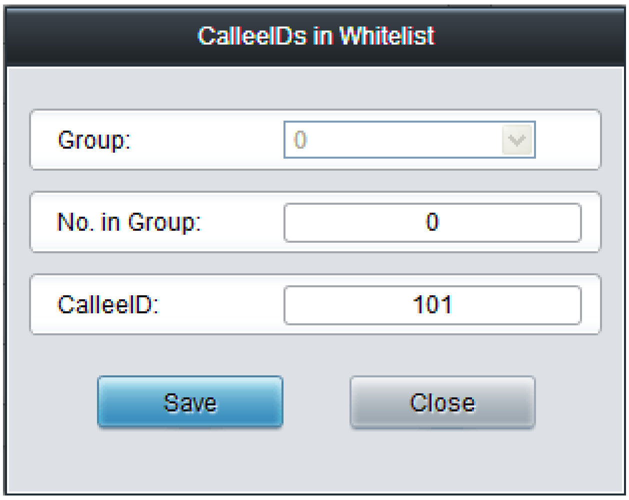



Figure 3-94 Modify CalleeIDs in Whitelist

To delete a CallerIDs/CalleeIDs in the whitelist, check the checkbox before the corresponding index in Figure 3-90 and click the '***Delete***' button. To clear all CallerIDs/CalleeIDs in the whitelist at a time, click the ***Clear All*** button in Figure 3-90.

**Note:** If a CallerID or CalleeID set in the whitelist is the same as one in the blacklist, it will go invalid. That is, the blacklist has a higher priority than the whitelist.

## 3.10.2 Blacklist

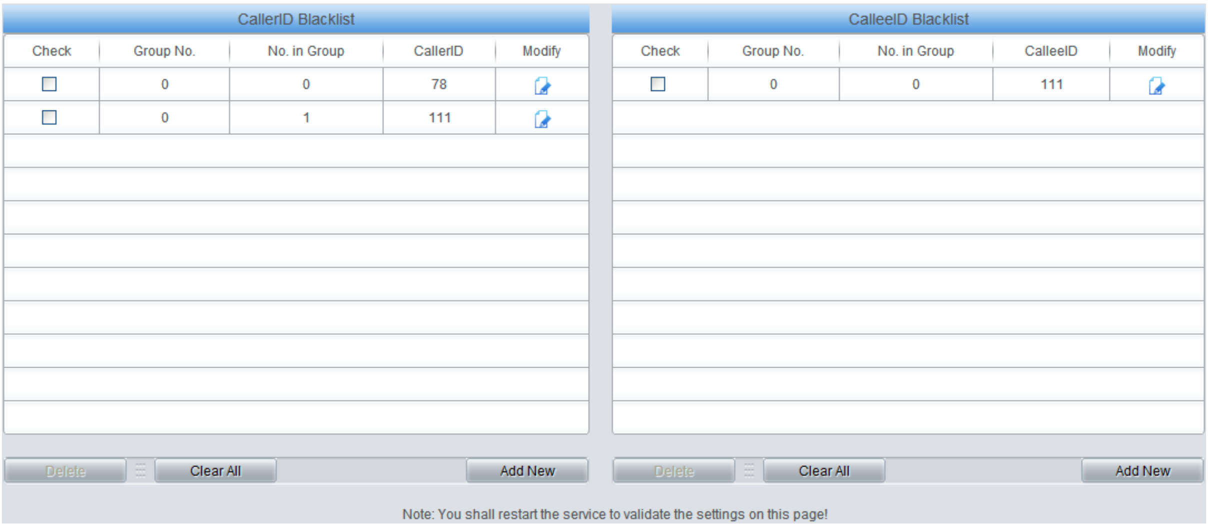



Figure 3-95 Blacklist Setting Interface

The Blacklist Setting interface is almost the same as the Whitelist Setting interface; only the whitelist changes to the blacklist. See Figure 3-95. The configuration items on this interface are the same as those on the Whitelist Setting interface (Figure 3-91, Figure 3-92).

### 3.10.3 Number Pool

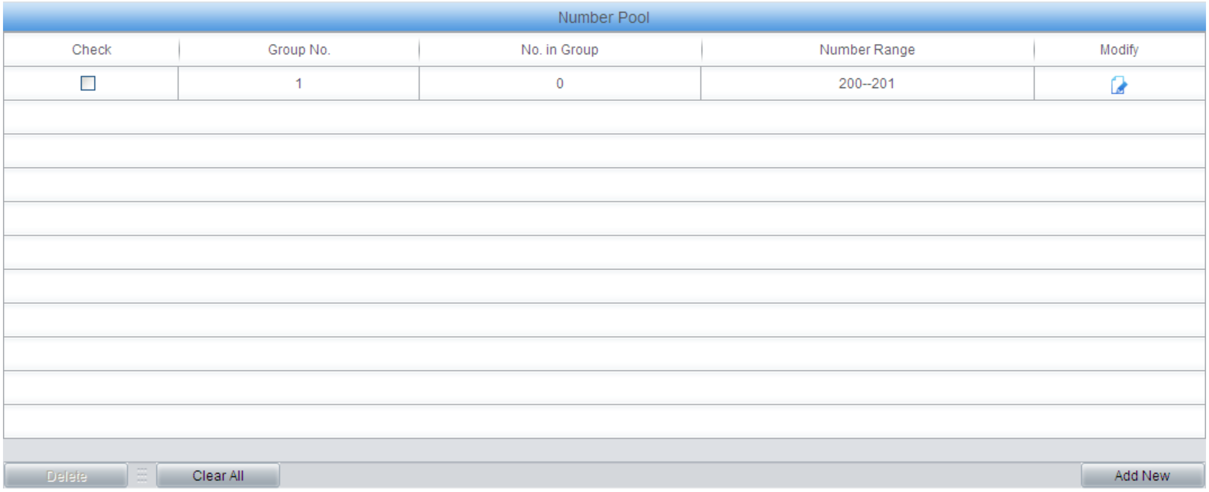



Figure 3-96 Number Pool Setting Interface

See Figure 3-96 for the Number Pool Setting interface. A new number pool can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-97 for the Number Pool adding interface.

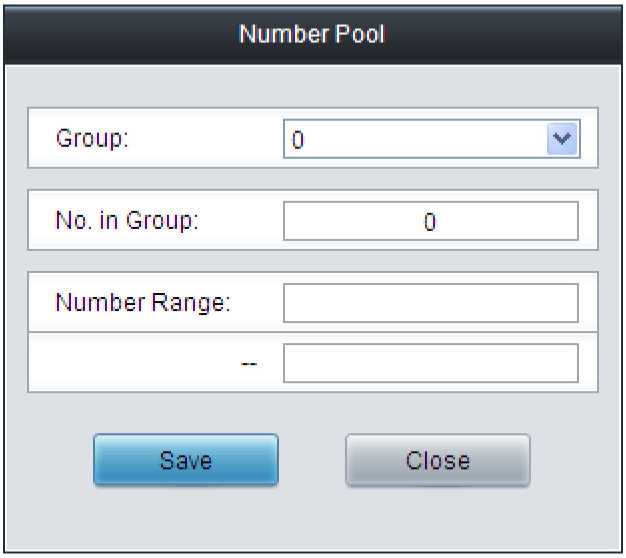



Figure 3-97 Add New Number Pool

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

| Item | Description |
|---|---|
| ***Group*** | The corresponding Group ID for numbers in the number pool. The value range is 0~15. |
| ***No. in Group*** | The corresponding No. for different numbers in a same group. It supports up to 100 number s in one group. |
| ***Number Range*** | The range of the numbers in a number Pool. It must be filled in with numbers and can not be left empty. |

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

Click ***Modify*** in Figure 3-96 to modify the number pool. See Figure 3-98 for the number pool modification interface. The configuration items on this interface are the same as those on the ***Add***

***New Number Pool*** interface.

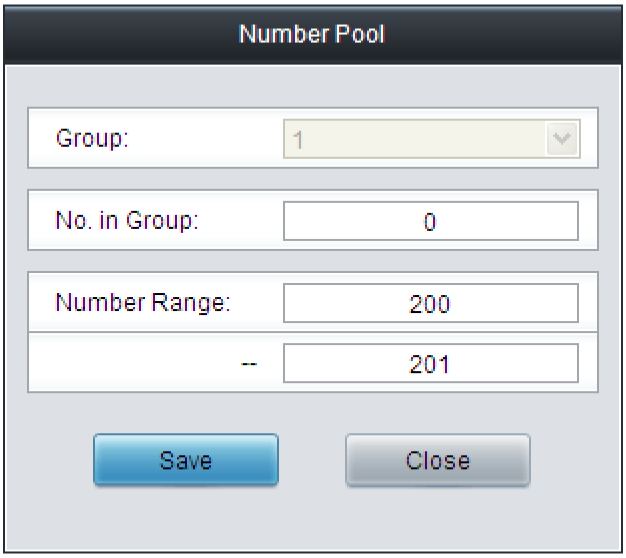
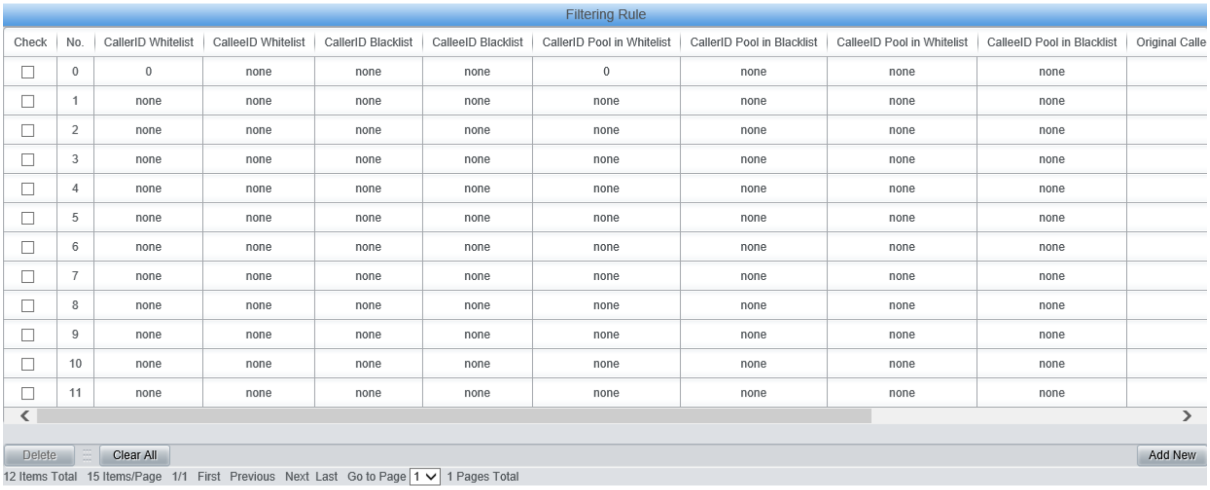



Figure 3-98 Modify Number Pool Interface

To delete a number pool, check the checkbox before the corresponding index in Figure 3-96 and click the ‘***Delete***’ button. To clear all number pools at a time, click the ***Clear All*** button in Figure 3-96.

## 3.10.4 Filtering Rule



Filtering Rule

| Check | No. | CallerID Whitelist | CalleeID Whitelist | CallerID Blacklist | CalleeID Blacklist | CallerID Pool in Whitelist | CallerID Pool in Blacklist | CalleeID Pool in Whitelist | CalleeID Pool in Blacklist | Original Calle |
|---|---|---|---|---|---|---|---|---|---|---|
| ☐ | 0 | 0 | none | none | none | 0 | none | none | none | |
| ☐ | 1 | none | none | none | none | none | none | none | none | |
| ☐ | 2 | none | none | none | none | none | none | none | none | |
| ☐ | 3 | none | none | none | none | none | none | none | none | |
| ☐ | 4 | none | none | none | none | none | none | none | none | |
| ☐ | 5 | none | none | none | none | none | none | none | none | |
| ☐ | 6 | none | none | none | none | none | none | none | none | |
| ☐ | 7 | none | none | none | none | none | none | none | none | |
| ☐ | 8 | none | none | none | none | none | none | none | none | |
| ☐ | 9 | none | none | none | none | none | none | none | none | |
| ☐ | 10 | none | none | none | none | none | none | none | none | |
| ☐ | 11 | none | none | none | none | none | none | none | none | |

Delete Clear All Add New

12 Items Total 15 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

Figure 3-99 Filtering Rule Setting Interface

See Figure 3-99 for the Filtering Rule Setting Interface. A new filtering rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-100 for the Filtering Rule Adding interface.


Figure 3-100 Add New Filtering Rule

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

| Item | Description |
|---|---|
| ***No.*** | The corresponding number for a filtering rule. The value range is 0~99. |
| ***CallerID Whitelist*** | The Group No. of CallerIDs saved on the whitelist setting interface. |
| ***CalleeID Whitelist*** | The Group No. of CalleeIDs saved on the whitelist setting interface. |
| ***CallerID Blacklist*** | The Group No. of CallerIDs saved on the blacklist setting interface. |
| ***CalleeID Blacklist*** | The Group No. of CalleeIDs saved on the blacklist setting interface. |
| ***CallerID Pool in Whitelist*** | Select a Group No. which is set in the whitelist from the number pool as the CallerID pool in whitelist. |
| ***CallerID Pool in Blacklist*** | Select a Group No. which is set in the blacklist from the number pool as the CallerID pool in blacklist. |
| ***CalleeID Pool in Whitelist*** | Select a Group No. which is set in the whitelist from the number pool as the CalleeID pool in whitelist. |

| | |
|---|---|
| ***CalleeID Pool in Blacklist*** | Select a Group No. which is set in the blacklist from the number pool as the CalleeID pool in blacklist. |
| ***Original CalleeID Pool in Whitelist*** | Select a Group No. which is set in the whitelist from the number pool as the original CalleeID pool in whitelist. |
| ***Original CalleeID Pool in Blacklist*** | Select a Group No. which is set in the blacklist from the number pool as the original CalleeID pool in blacklist. |
| ***Description*** | Remarks for the filtering rule. It can be any information, but can not be left empty. |

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

Click ***Modify*** in Figure 3-99 to modify the filtering rule. See Figure 3-101 for the filtering rule modification interface. The configuration items on this interface are the same as those on the ***Add New Filtering Rule*** interface.

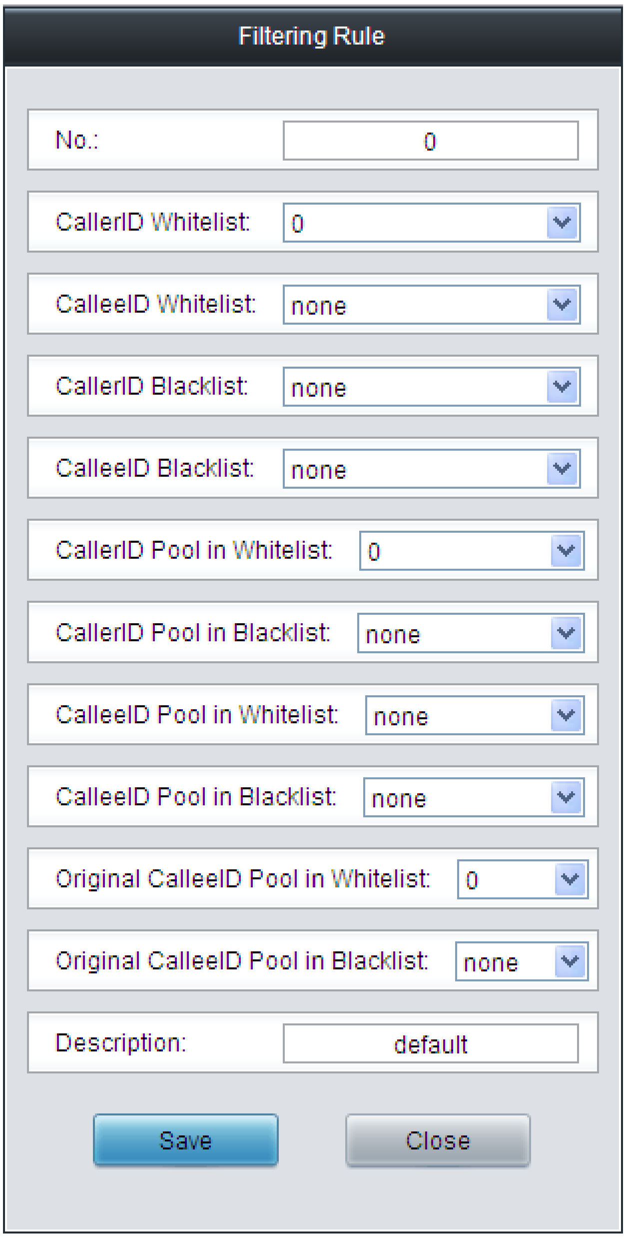



Figure 3-101 Modify Filtering Rule Interface

To delete a filtering rule, check the checkbox before the corresponding index in Figure 3-99 and click the '***Delete***' button. To clear all filtering rules at a time, click the ***Clear All*** button in Figure 3-99.

## 3.11 Number Manipulation

Number Manipulation includes seven parts: **IP→PSTN CallerID**, **IP→PSTN CalleeID**, **IP→PSTN Original CalleeID**, **PSTN→IP CallerID**, **PSTN→IP CalleeID**, **PSTN→IP Original CalleeID** and **CallerID Pool**. See Figure 3-102.

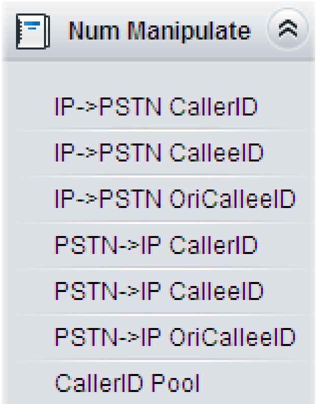
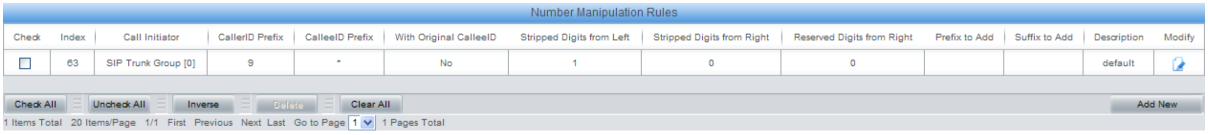



Figure 3-102 Number Manipulation

### 3.11.1 IP to PSTN CallerID



Number Manipulation Rules

| Check | Index | Call Initiator | CallerID Prefix | CalleeID Prefix | With Original CalleeID | Stripped Digits from Left | Stripped Digits from Right | Reserved Digits from Right | Prefix to Add | Suffix to Add | Description | Modify |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| ☐ | 63 | SIP Trunk Group [0] | 9 | * | No | 1 | 0 | 0 | | | default | |

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

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

Figure 3-103 IP→PSTN CallerID Manipulation Interface

See Figure 3-103 for the IP→PSTN CallerID manipulation interface. A new number manipulation rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-104 for the IP→PSTN CallerID manipulation rule adding interface.

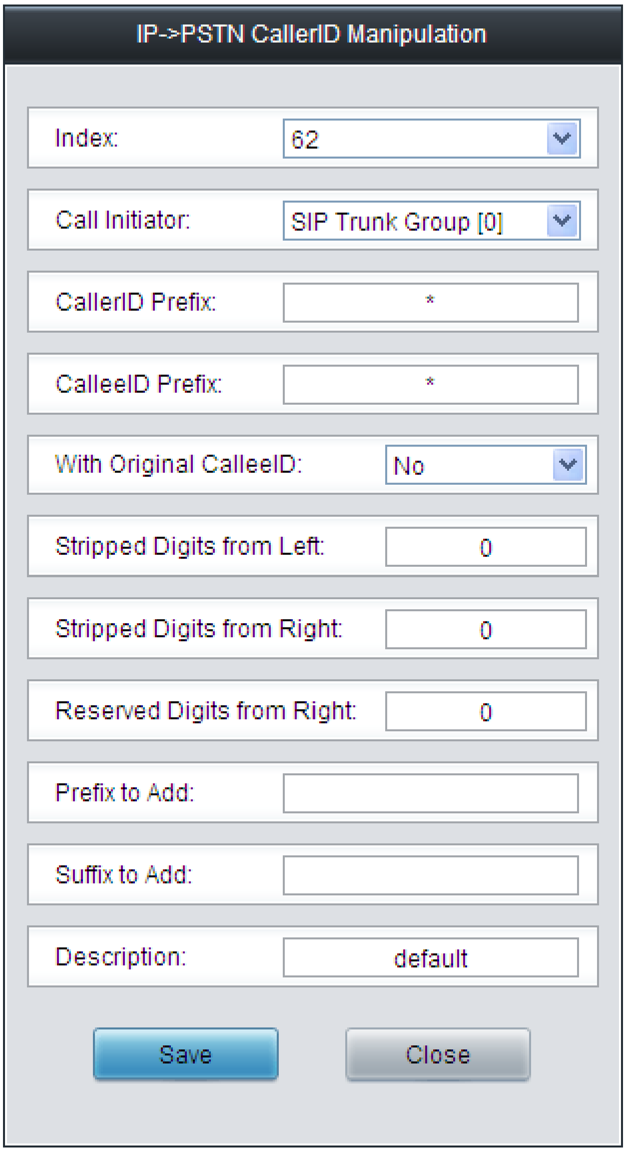



Figure 3-104 Add IP→PSTN CallerID Manipulation Rule

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. |
| ***Call Initiator*** | SIP trunk group from where the call is initiated. This item can be set to a specific SIP trunk group or SIP Trunk Group[ANY] which indicates any SIP trunk group. |
| ***CallerID Prefix, CalleeID Prefix*** | A string of numbers at the beginning of the calling/called party number. This item can be set to a specific string or "*" which indicates any string. These two configuration items together with ***Call Initiator*** and ***With Original CalleeID*** can specify the calls which apply to a number manipulation rule. |
| ***With Original CalleeID*** | If this item is set to *Yes*, it indicates that the number manipulation rule is only applicable to the calls with original CalleeID/redirecting number. The default value is *No*. |

| | |
|---|---|
| ***Stripped Digits from Left*** | The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted. |
| ***Stripped Digits from Right*** | The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted. |
| ***Reserved Digits from Right*** | The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated. |
| ***Prefix to Add*** | Designated information to be added to the left end of the current number. |
| ***Suffix to Add*** | Designated information to be added to the right end of the current number. |
| ***Description*** | More information about each number manipulation rule. |

**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 3-103 to modify a number manipulation rule. See Figure 3-105 for the IP→PSTN CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the ***Add IP→PSTN CallerID Manipulation Rule*** interface. Note that the item ***Index*** cannot be modified.

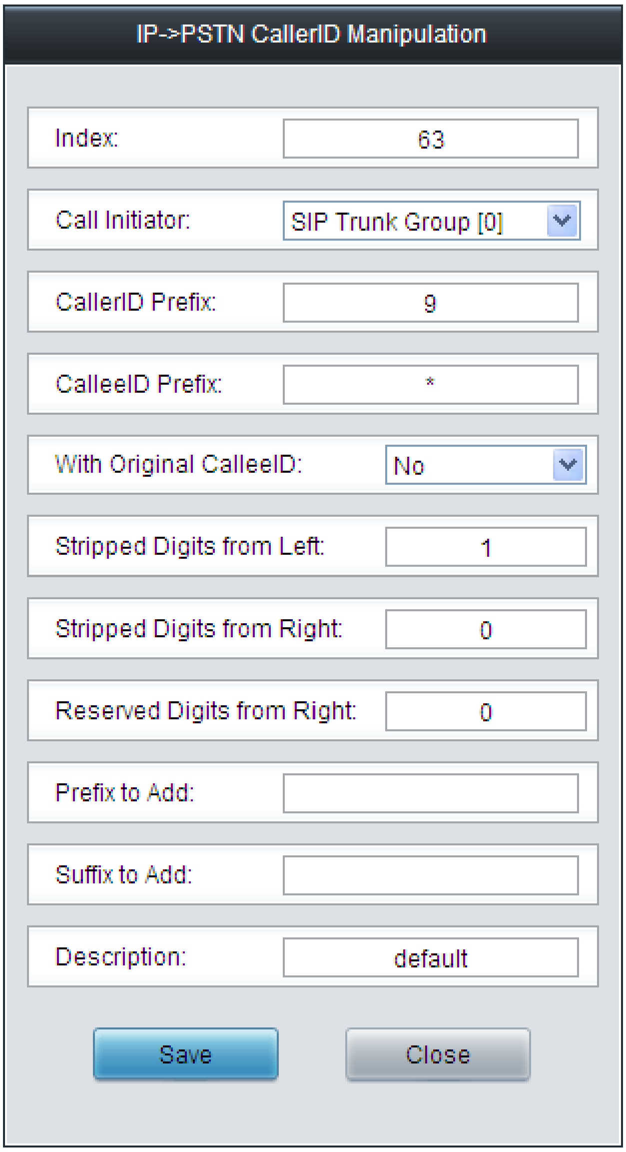



Figure 3-105 Modify IP→PSTN CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-103 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 3-103.

## 3.11.2 IP to PSTN CalleeID

The number manipulation process for IP→PSTN CalleeID is almost the same as that for IP→PSTN CallerID; only the number to be manipulated changes from CallerID to CalleeID. See Figure 3-106 for IP→PSTN CalleeID manipulation interface. The configuration items on this interface are the same as those on ***IP→PSTN CallerID Manipulation Interface*** (Figure 3-103).

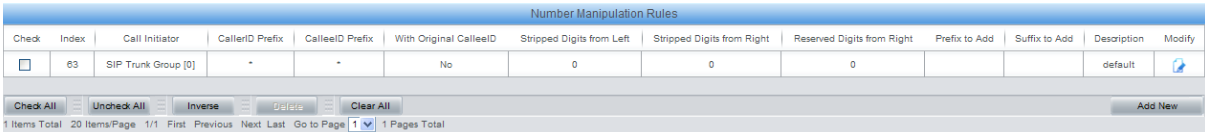



Figure 3-106 IP→PSTN CalleeID Manipulation Interface

### 3.11.3 IP to PSTN Original CalleeID

The number manipulation process for IP→PSTN Original CalleeID is almost the same as that for IP→PSTN CallerID; only the number to be manipulated changes from CallerID to Original CalleeID. See Figure 3-107 for IP→PSTN Original CalleeID manipulation interface. The configuration items on this interface are the same as those on ***IP→PSTN CallerID Manipulation Interface*** (Figure 3-103).

Number Manipulation Rules

| Check | Index | Call Initiator | CallerID Prefix | CalleeID Prefix | Stripped Digits from Left | Stripped Digits from Right | Reserved Digits from Right | Prefix to Add | Suffix to Add | Description | Modify |
|---|---|---|---|---|---|---|---|---|---|---|---|
| ☐ | 63 | SIP Trunk Group [0] | 2 | 5 | 1 | 0 | 10 | 666 | 888 | default | |

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

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

Figure 3-107 IP→PSTN Original CalleeID Manipulation Interface

### 3.11.4 PSTN to IP CallerID

Number Manipulation Rules

| Check | Index | Call Initiator | CallerID Prefix | CalleeID Prefix | With Original CalleeID | Stripped Digits from Left | Stripped Digits from Right | Reserved Digits from Right | Prefix to Add | Suffix to Add | Description | Modify |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| ☐ | 63 | PCM Trunk Group [0] | 89 | * | No | 2 | 0 | 0 | | | default | |

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

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

Figure 3-108 PSTN→IP CallerID Manipulation Interface

See Figure 3-108 for the PSTN→IP CallerID manipulation interface. A new number manipulation rule can be added by the ***Add New*** button on the bottom right corner of the list in the above figure. See Figure 3-109 for the PSTN→IP CallerID manipulation rule adding interface.

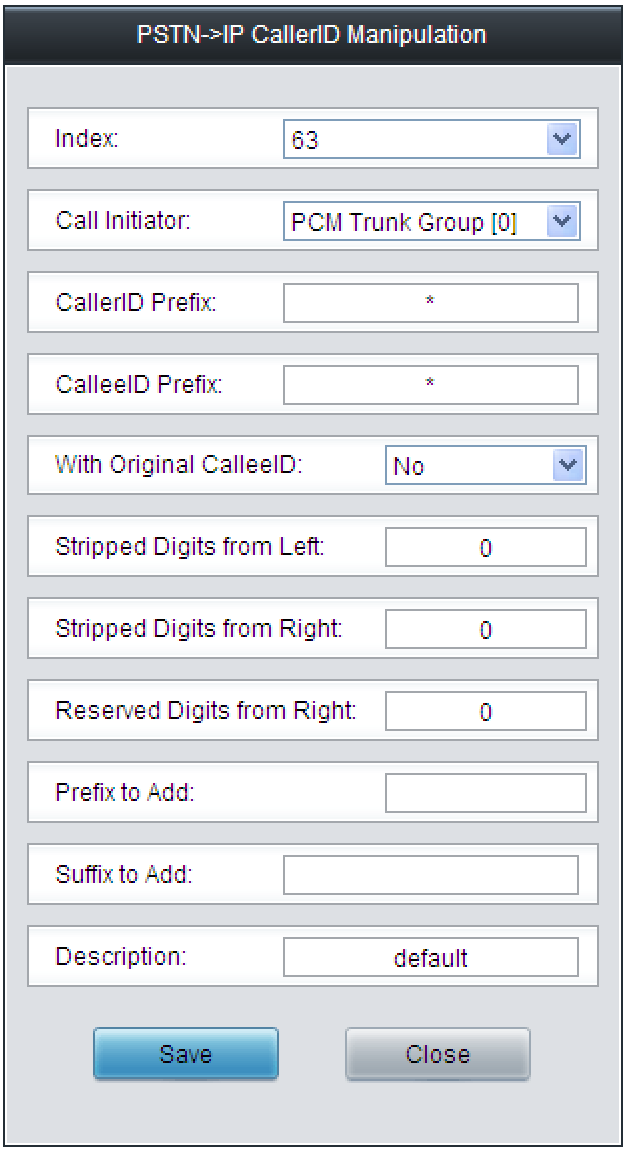



Figure 3-109 Add PSTN→IP CallerID Manipulation Rule

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. |
| ***Call Initiator*** | PCM trunk group from where the call is initiated. This item can be set to a specific PCM trunk group or PCM Trunk Group[ANY] which indicates any PCM trunk group. |
| ***CallerID Prefix, CalleeID Prefix*** | A string of numbers at the beginning of the calling/called party number. This item can be set to a specific string or "*" which indicates any string. These two configuration items together with ***Call Initiator*** and ***With Original CalleeID*** can specify the calls which apply to the number manipulation rule. |
| ***With Original CalleeID*** | If this item is set to *Yes*, it indicates that the number manipulation rule is only applicable to the calls with original CalleeID/redirecting number. The default value is *No*. |

| | |
|---|---|
| ***Stripped Digits from Left*** | The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted. |
| ***Stripped Digits from Right*** | The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted. |
| ***Reserved Digits from Right*** | The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated. |
| ***Prefix to Add*** | Designated information to be added to the left end of the current number. |
| ***Suffix to Add*** | Designated information to be added to the right end of the current number. |
| ***Description*** | More information about each number manipulation rule. |

**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 3-108 to modify a number manipulation rule. See Figure 3-110 for the PSTN→IP CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the ***Add PSTN→IP CallerID Manipulation Rule*** interface. Note that the item ***Index*** cannot be modified.

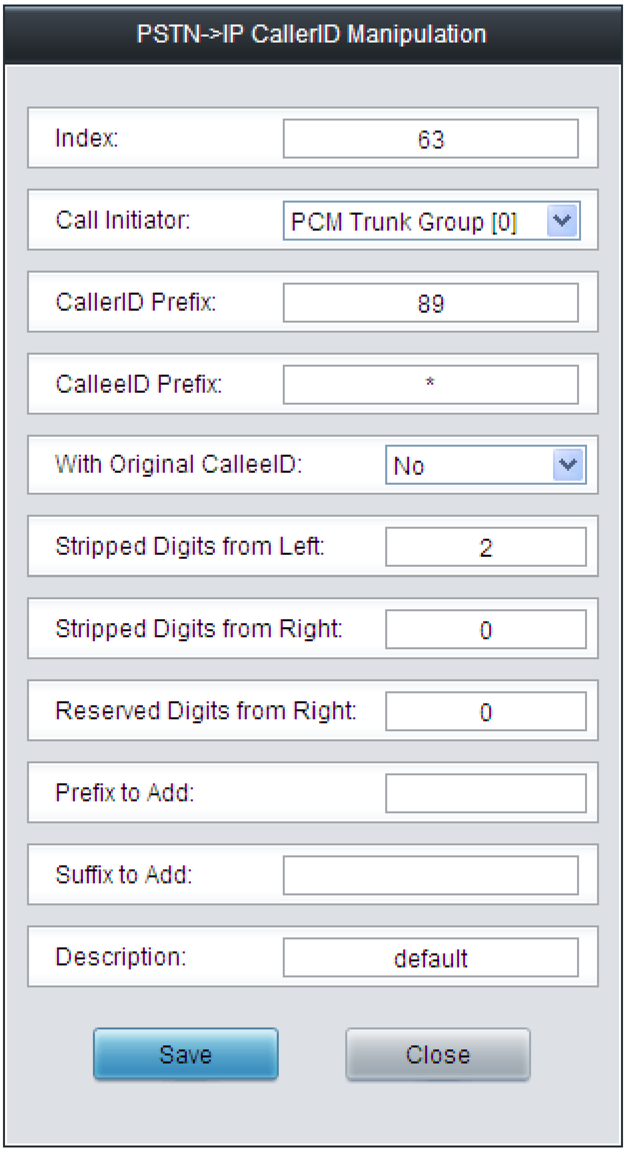



Figure 3-110 Modify PSTN→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-108 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 3-108.

## 3.11.5 PSTN to IP CalleeID

The number manipulation process for PSTN→IP CalleeID is almost the same as that for PSTN→IP CallerID; only the number to be manipulated changes from CallerID to CalleeID. See Figure 3-111 for the PSTN→IP CalleeID manipulation interface. The configuration items on this interface are the same as those on ***PSTN→IP CallerID Manipulation Interface*** (Figure 3-108).

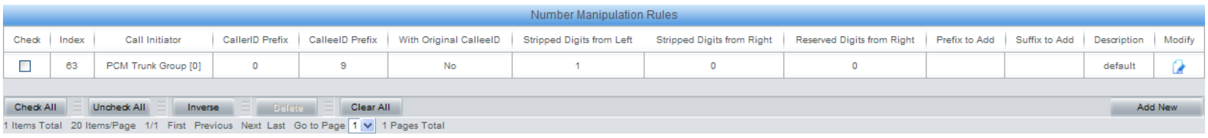



Figure 3-111 PSTN→IP CalleeID Manipulation Interface

## 3.11.6 PSTN to IP Original CalleeID

The number manipulation process for PSTN→IP Original CalleeID is almost the same as that for PSTN→IP CallerID; only the number to be manipulated changes from CallerID to Original CalleeID. See Figure 3-112 for the PSTN→IP Original CalleeID manipulation interface. The configuration items on this interface are the same as those on ***PSTN→IP CallerID Manipulation Interface*** (Figure 3-108).


Figure 3-112 PSTN→IP Original CalleeID Manipulation Interface

## 3.11.7 CallerID Pool

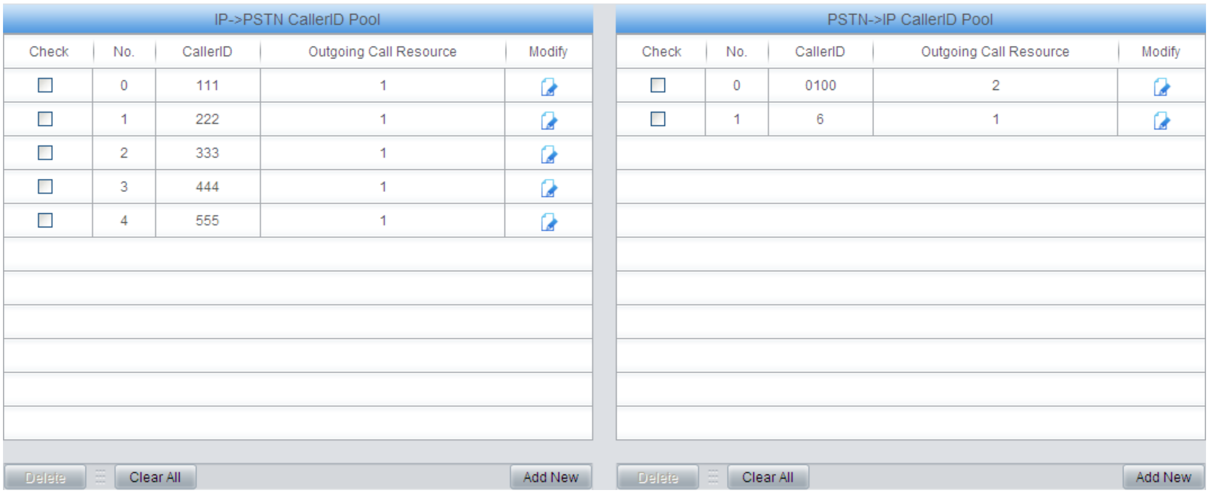



Figure 3-113 CallerID Pool Interface

See Figure 3-113 for the CallerID Pool interface, including two parts: PSTN→IP CallerID Pool and IP→PSTN CallerID Pool. It is used to designate the CallerID for outgoing calls and restrict the call amount for each designated callerID at the same time. A new CallerID can be added by the ***Add New*** button. See Figure 3-114 for the CallerID adding interface.

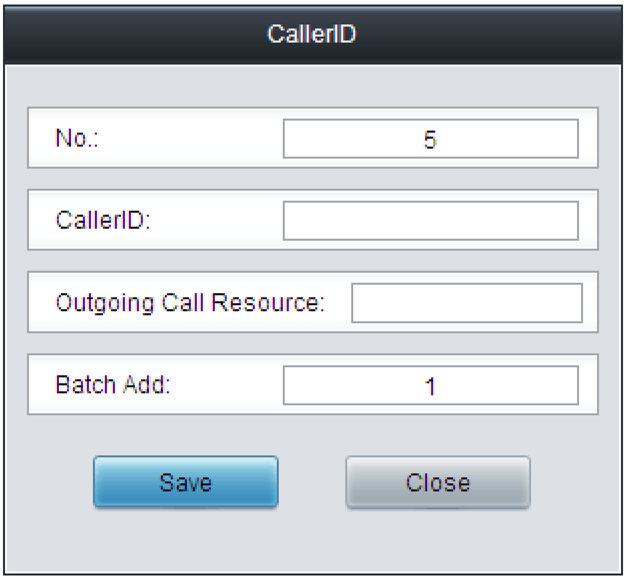


Figure 3-114 Add New CallerID Interface

The table below explains the items shown in above figures.

| Item | Description |
|---|---|
| ***No.*** | The unique index of the CallerID in the pool, which starts from 0 and denotes its priority. A CallerID with a smaller index value has a higher priority. |
| ***CallerID*** | Sets the CallerID used for an outgoing call. |
| ***Outgoing Call Resource*** | Sets the maximum number of the outgoing calls for each CallerID. |
| ***Batch Add*** | Sets the amount of CallerIDs to be batch added. |

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

Click ***Modify*** in Figure 3-113 to modify the CallerID information. See Figure 3-115 for the CallerID modification interface. The configuration items on this interface are the same as those on the ***Add New CallerID*** interface. The item *No.* cannot be modified.

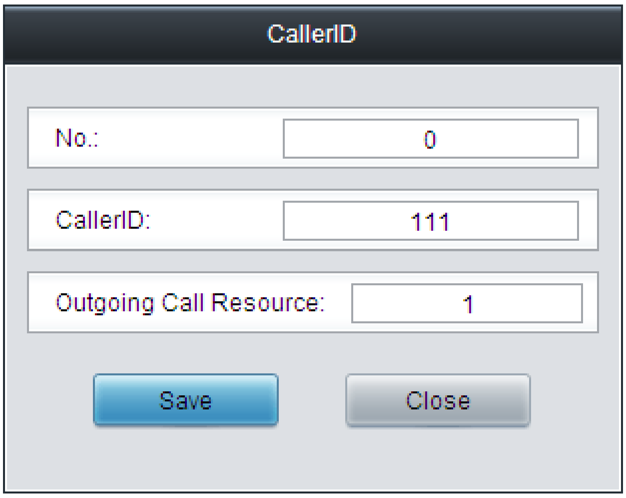



Figure 3-115 Modify CallerID Interface

To delete a CallerID in the pool, check the checkbox before the corresponding index in Figure 3-113 and click the '***Delete***' button. To clear all CallerIDs in the pool at a time, click the ***Clear All*** button in Figure 3-113.

## 3.12 System Tools

System Tools is mainly for gateway maintenance. It provides such features as IP modification, time synchronization, data backup, log inquiry and connectivity check. See Figure 3-116 for details.

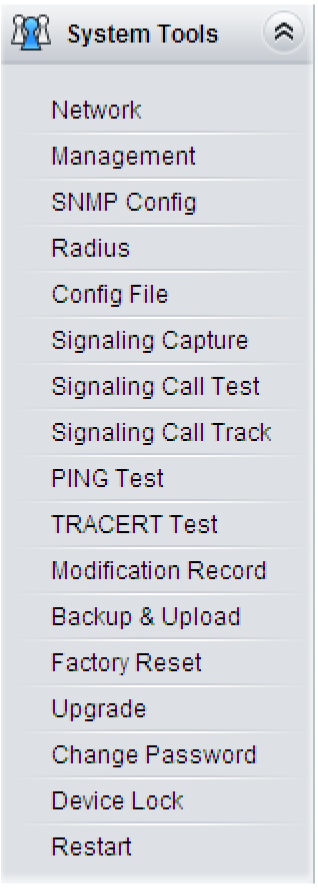



Figure 3-116 System Tools

### 3.12.1 Network

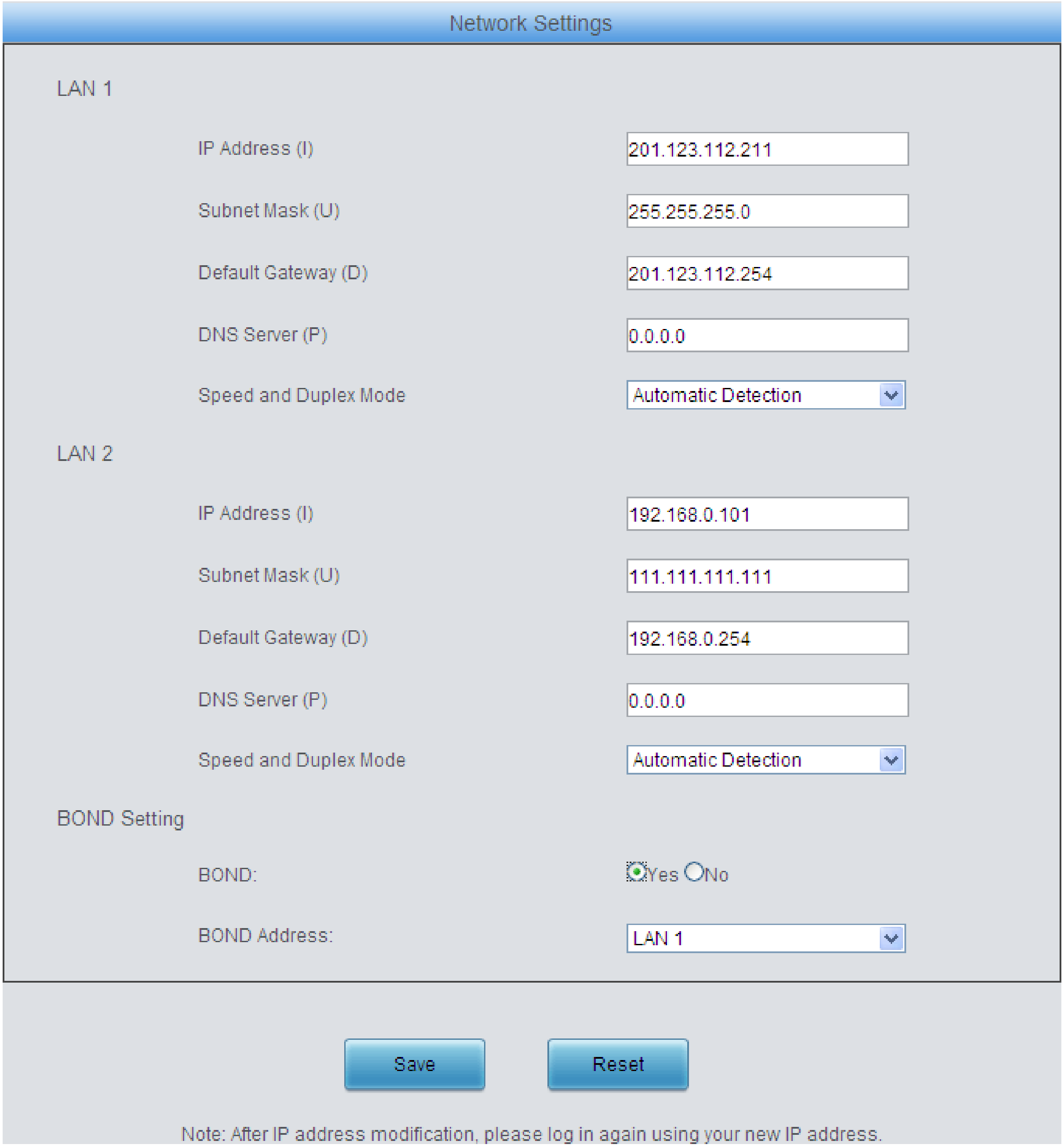



Figure 3-117 Network Settings Interface

See Figure 3-117 for the network settings interface. A gateway has two LANs, each of which can be configured with independent IP address, subnet mask, default gateway and DNS server. The Bond feature when enabled will make the information of LAN1 and LAN2 duplicated and backed up.so as to realize the hot-backup function between LAN1 and LAN2. By default, this feature is *disabled*.

**Note: By default, *Speed and Duplex Mode* is set to Automatic Detection. We suggest you not modify it because the non-automatic detection may cause abnormity in network interface.**

If the Network Detect feature is enabled, a ping test will automatically be initiated from this IP address to the gateway to check the connection status between them. By default, this feature is disabled.

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

## 3.12.2 Management

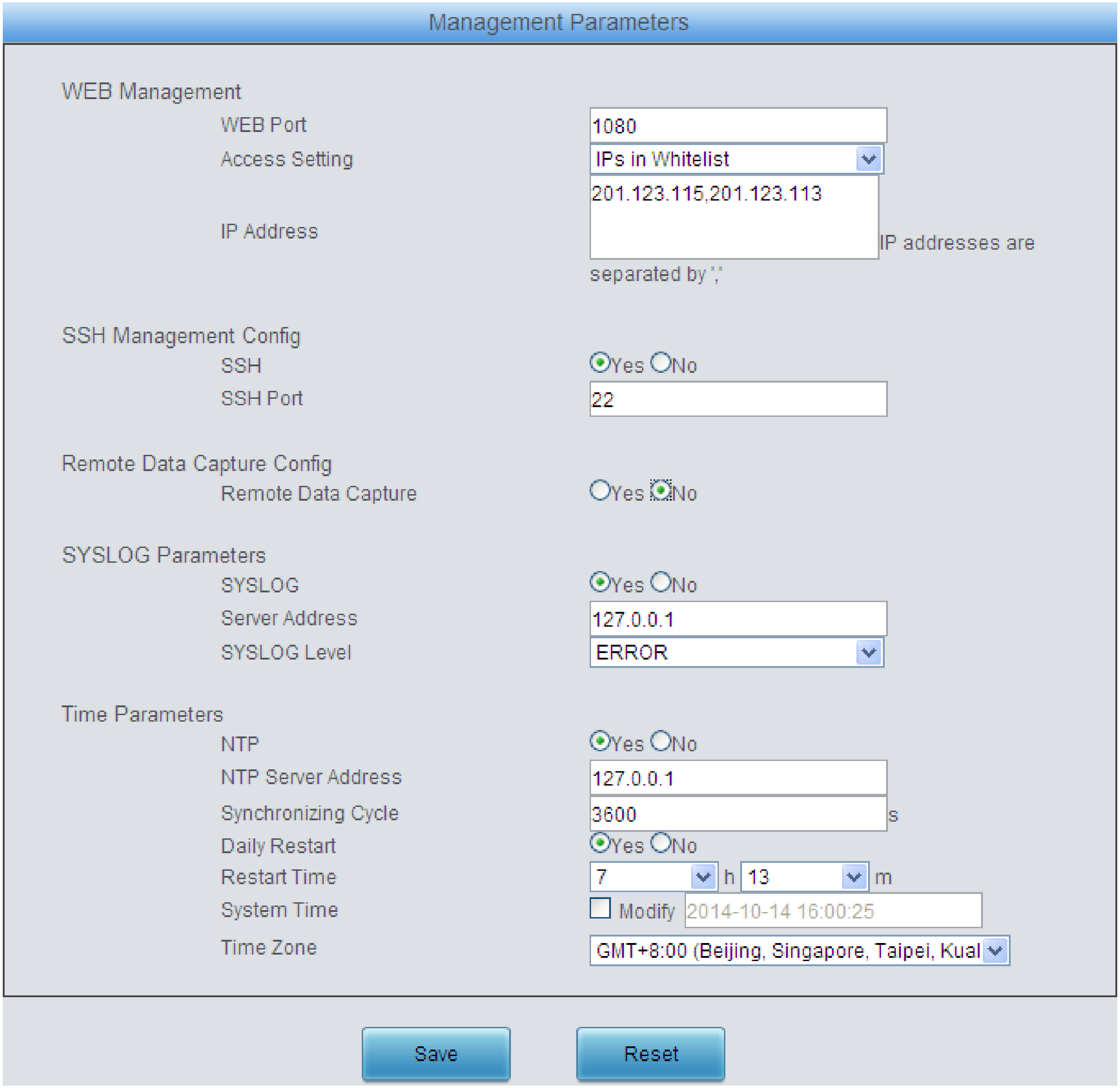



Figure 3-118 Management Parameters Setting Interface

See Figure 3-118 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 the IPs within it to access the gateway freely. Also you can set an IP blacklist to forbid all the IPs within it to access the gateway. |
| ***SSH*** | Sets whether to enable the gateway to be accessed via SSH, with the default value of *No*. |

| | |
|---|---|
| ***SSH Port*** | The port which is used to access the gateway via SSH. |
| ***Remote Data Capture*** | After this feature is enabled, you can obtain the gateway data via a remote capture tool. The default value is *No*. |
| ***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*. |
| ***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. |
| ***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. |

## 3.12.3 SNMP Config

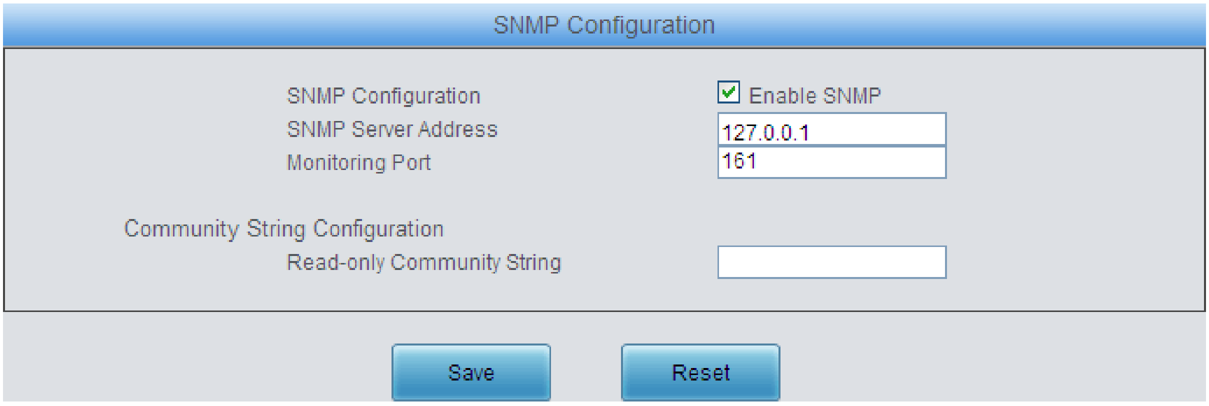



Figure 3-119 SNMP Configuration Interface

See Figure 3-119 for the SNMP configuration interface. If the SNMP feature is enabled, once the gateway receives a request from the SNMP management software, it will collect relevant information and reply to the SNMP management software. By default, the SNMP feature is disabled. The available information includes kernel version, CPU usage, processes, memory usage, startup information, LAN status and etc. Currently, the gateway only provides the community string for information acquisition.

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

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

You can query OID (object identification trees) = .1.3.6.1.4.1.2021.51 at the SNMP Client to obtain the signaling link status and the line synchronization information,

## 3.12.4 Radius

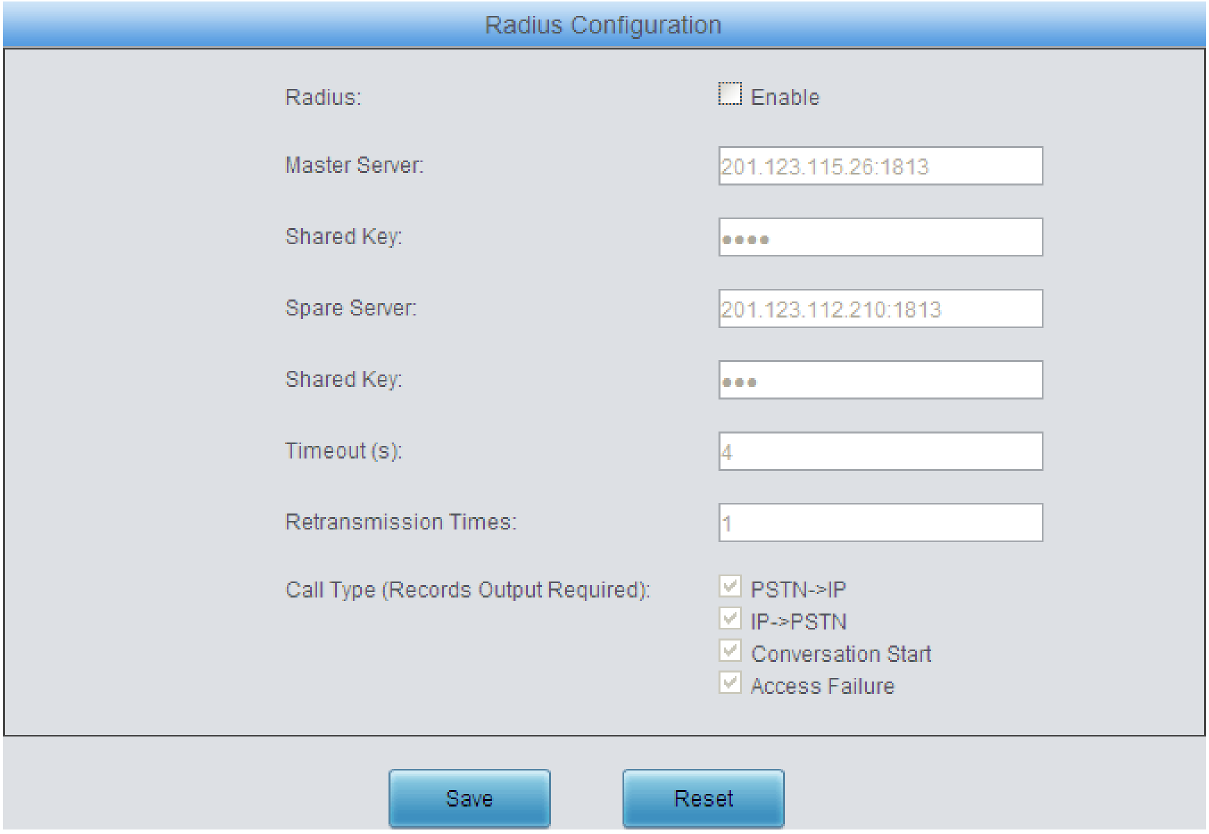



Figure 3-120 Radius Configuration Interface

See Figure 3-120 for the Radius Configuration interface. The Radius feature is supported. Once it is enabled, the gateway will serve as the Radius client and send messages to the Radius server at the start and end of each call to fulfill the charge business.

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

| Item | Description |
|---|---|
| ***Radius*** | Sets whether to enable Radius or not, with the default setting of *disabled.* |
| ***Master Server*** | Sets the IP address and port of the master Radius server.<br>**Note:** If the port isn't designated, the default port 1813 will be used. |
| ***Shared Key*** | Sets the shared key used for the communication encryption between the master Radius server and the Radius client.<br>**Note:** The key should be appointed by both the client and the server end ahead of time, and be configured the same at both sides. |
| ***Spare Server*** | Sets the IP address and the port of the spare Radius server which will be automatically started upon the occurrence of malfunction on the communications between the gateway and Radius master server.<br>**Note:** If the port isn't designated, the default port 1813 will be used. |

<table>
<tr><td>Timeout</td><td>Sets the maximum time to wait for the response after the message is sent out by Radius, with the default value of 3s. To guarantee the accuracy of the charge, the gateway will start the message retransmission mechanism once the charge message sent from the gateway to the Radius server is timeout without any response.</td></tr>
<tr><td>Retransmission Times</td><td>Sets the retransmission times on no response to the Radius message, with the default value of 3.</td></tr>
<tr><td>Call Type (Records output required)</td><td>Sets the type of calls which are required to output call records, including four options: PSTN→IP, IP→PSTN, conversion start and access failure.
<table>
<tr><th>Type</th><th>Meaning</th></tr>
<tr><td>PSTN→IP</td><td>Whether to send the Radius charge message for the calls from PSTN to IP</td></tr>
<tr><td>IP→PSTN</td><td>Whether to send the Radius charge message for the calls from IP to PSTN</td></tr>
<tr><td>Conversion Start</td><td>Whether to send the record of the initial conversion, that is, whether to have the gateway send the record information about the initial conversion to the Radius server upon the connection of the conversion.</td></tr>
<tr><td>Access Failure</td><td>Whether to send the record of the calls in access failure, that is, whether to have the gateway send the record information about the calls in access failure to the Radius server upon the access failure occurs.</td></tr>
</table></td></tr>
</table>

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

### 3.12.5 Configuration File

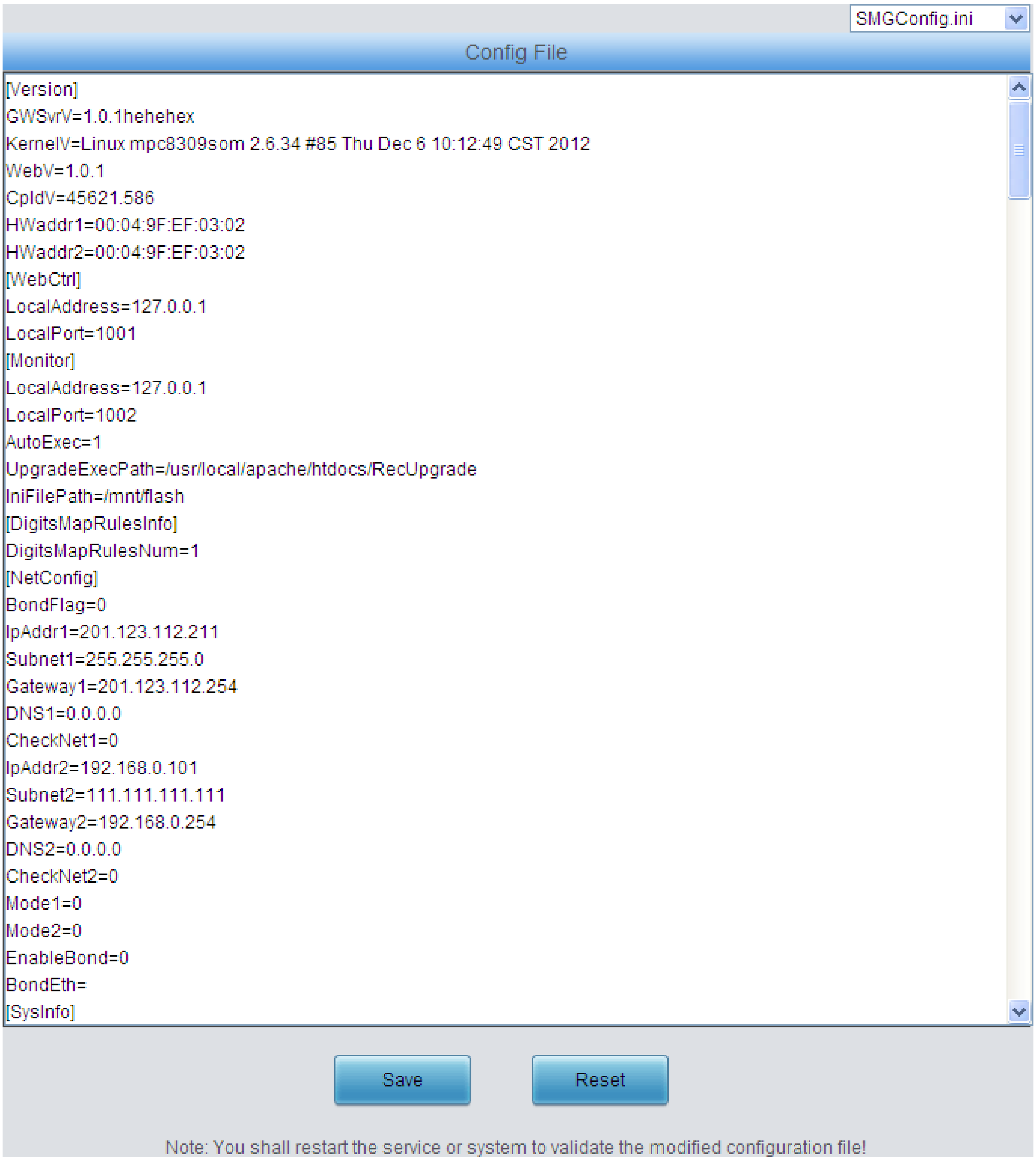



Figure 3-121 Configuration File Interface

See Figure 3-121 for the Configuration File interface, including three files: SMGConfig.ini, ShConfig.ini and Ss7Server.ini. You can check and modify the items in these configuration files through this interface. Configurations about the gateway server, such as route rules, number manipulation, number filter and so on, are included in SMGConfig.ini; Configurations about the board are included in ShConfig.ini; and configurations about the SS7 server are included in Ss7Server.ini. You can modify these configurations on the interface directly, and then click ***Save*** to save the above settings into the gateway or click ***Reset*** to restore the configurations.

### 3.12.6 Signaling Capture

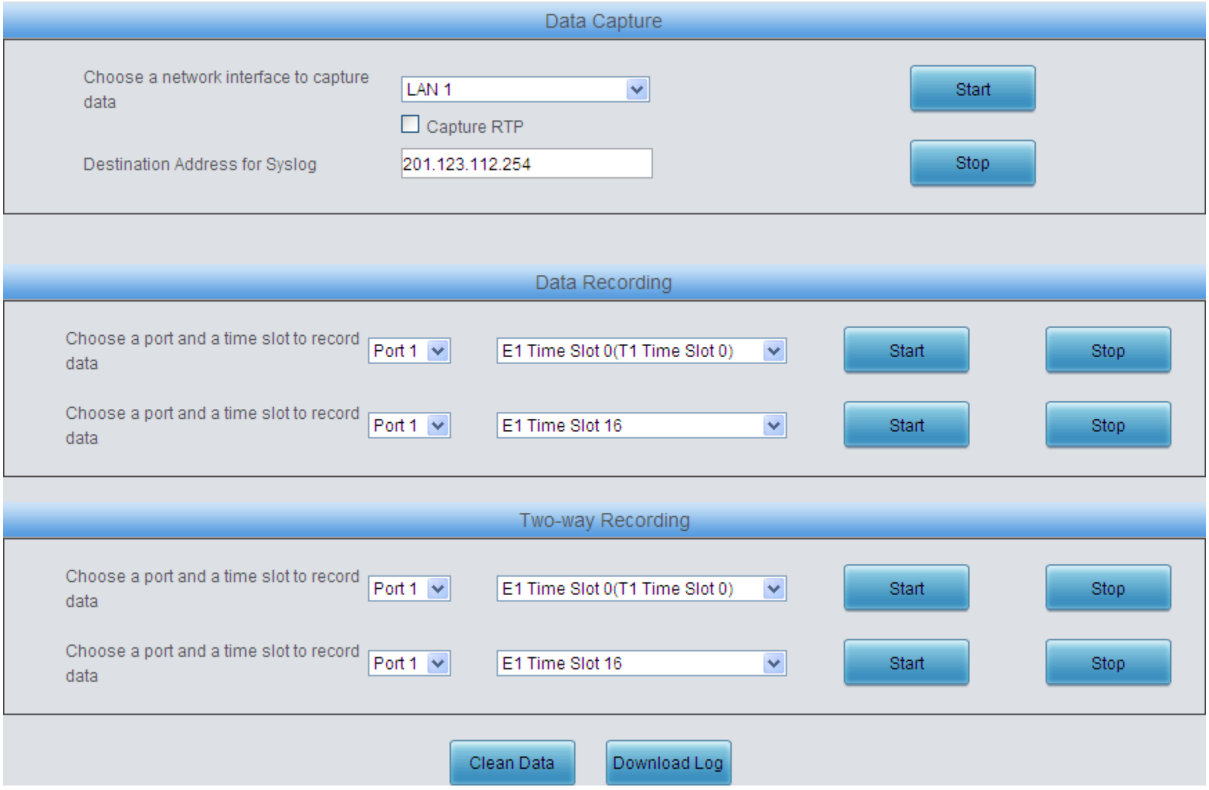



Figure 3-122 Signaling Capture Interface

See Figure 3-122 for the Signaling Capture interface. Data Capture is used to capture data on the network interface you choose. Click ***Start*** to start capturing data (1024000 packets at most) on the corresponding network interface. SIP, ISDN, SS7 and SysLog are supported at present. You can enter the Syslog destination address to send Syslog to wherever required. Click ***Stop*** to stop data capture and download the captured packets.

Data Recording (one-way) and Two-way Recording (two-way) are used to record data on the time slot you choose. Click ***Start*** to start recording data (maximum consecutively recording time: data recording is100 minutes and two-way recording is 1 minutes) on the corresponding port and time slot. Click ***Stop*** to stop data recording and download the recorded data.

### 3.12.7 Signaling Call Test

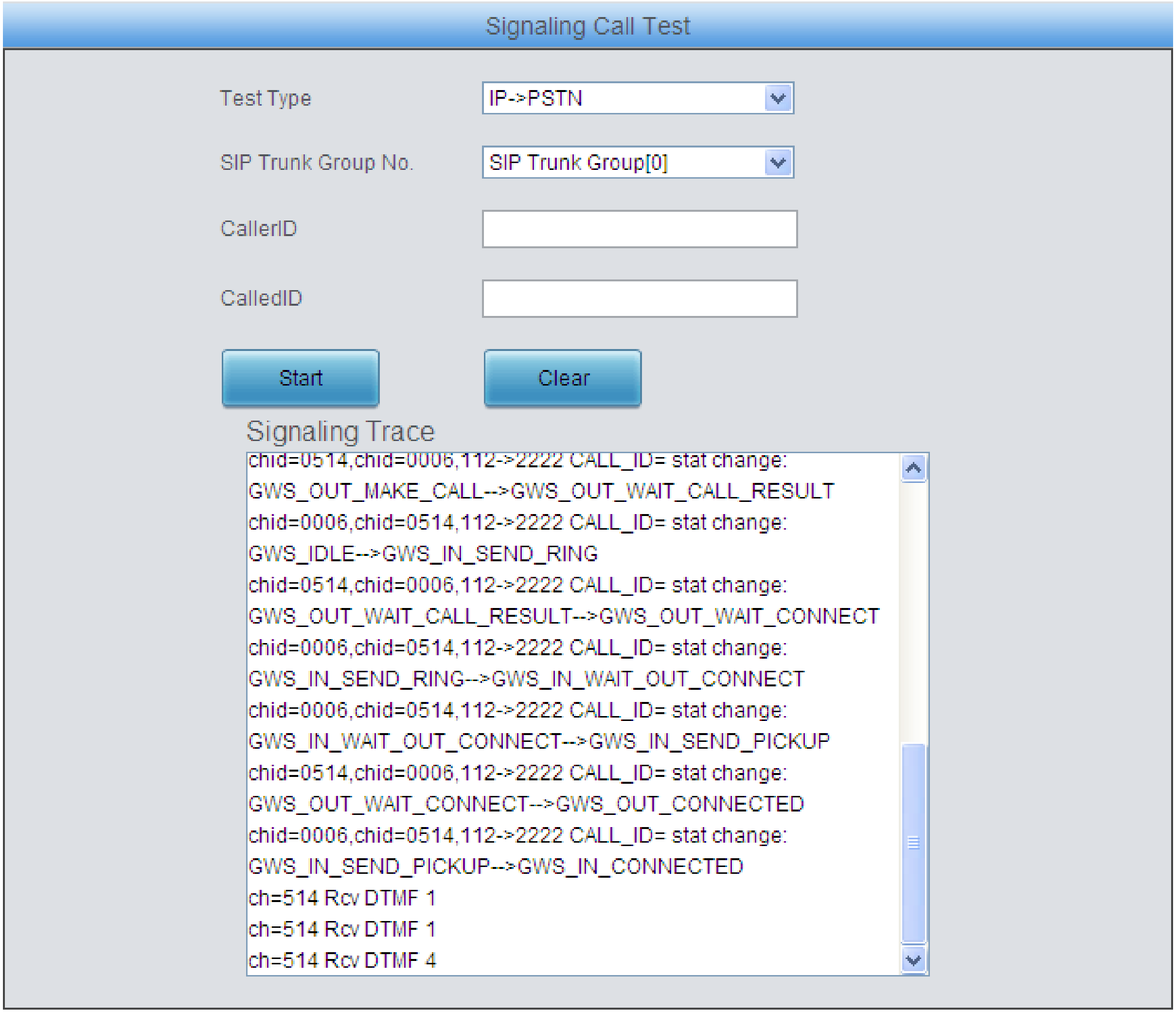



Figure 3-123 Signaling Call Test Interface

See Figure 3-123 for the Signaling Call Test interface. This feature can help to test whether the route and the number manipulation already configured are proper or not, and whether the call can succeed or not.

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

| Item | Description |
|---|---|
| ***Test Type*** | The source trunk type for signaling call test. There are three options: **IP→PSTN**, **PSTN→IP** and **PSTN Call Out** |
| ***SIP Trunk Group No.*** | The SIP trunk group number you are required to select if choosing ***IP→PSTN*** in ***Test Type***, |
| ***PCM Trunk Group No.*** | The PCM trunk group number you are required to select if choosing ***PSTN→IP*** in ***Test Type***, |
| ***CallerID*** | The CallerID for the signaling call test. |
| ***CalleeID*** | The CalleeID for the signaling call test. |
| ***PCM Port*** | You are required to select the PCM port if choosing ***PSTN Call Out*** in ***Test Type***,<br>**Note:** This item will appear only if you choose ***PSTN Call Out*** in ***Test Type***, |

| | |
|---|---|
| ***PCM Channel*** | You are required to select the PCM channel if choosing ***PSTN Call Out*** in ***Test Type***, **Note:** This item will appear only if you choose ***PSTN Call Out*** in ***Test Type***, |
| ***DTMF*** | You can select this item to send DTMFs after the establishment of call conversation on the channel for call test, if choosing ***PSTN Call Out*** in ***Test Type***,<br>**Note:** This item will appear only if you choose ***PSTN Call Out*** in ***Test Type***, |
| ***Signaling Trace*** | The information returned during the signaling call test, helping you to learn the detailed information about the test call. |

After configuration, click ***Start*** to execute the signaling call test; click ***Clear*** to clear the signaling trace information.

**Note:** The call test will be finished only if the called party ends it. That is, the gateway can not stop the testing.

### 3.12.8 Signaling Call Track

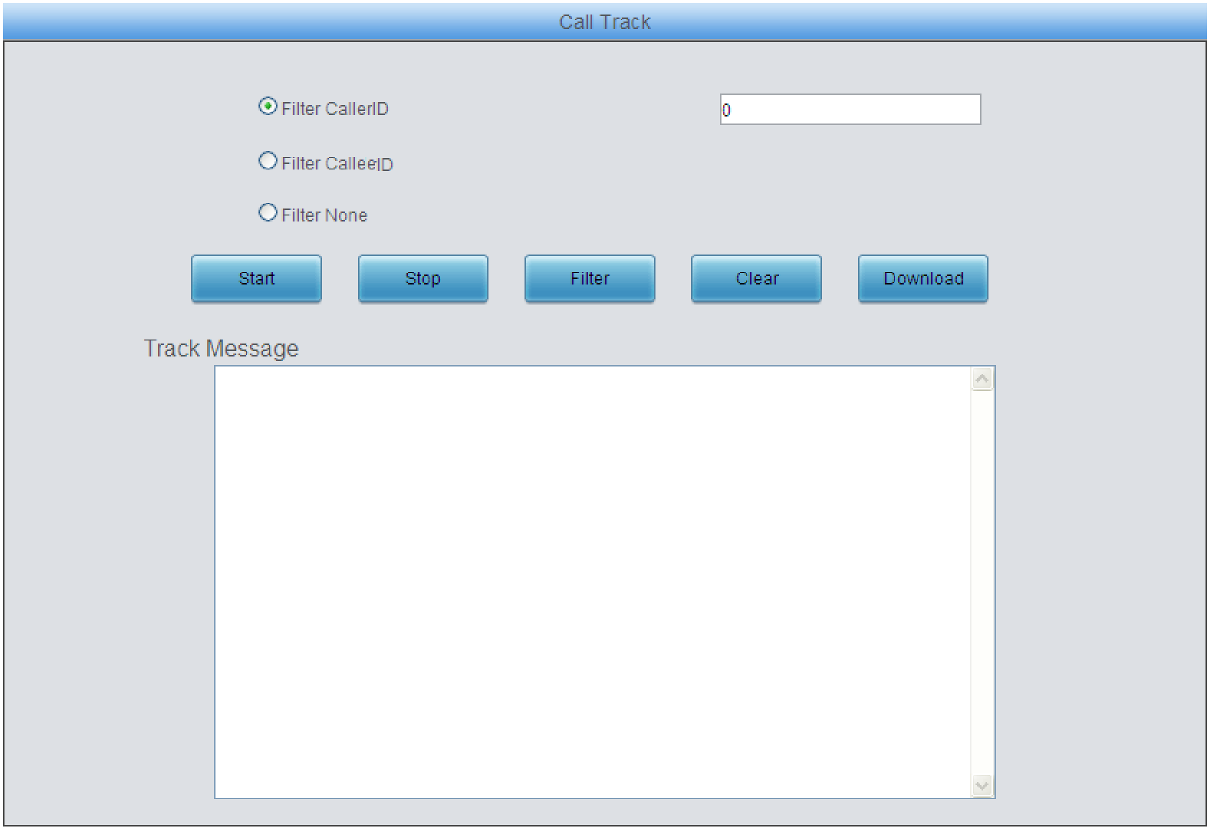



Figure 3-124 Call Track Interface

See Figure 3-124 for the Call Track Interface, including three modes: Filter CallerID, Filter CalleeID and Filter None. This is mainly used to output and save call information, facilitating call trace and problem debugging. Click ***Start*** to track calls, and the trace logs will be shown in the “Track Message” field; click ***Stop*** to stop the call track; click ***Filter*** to filter the trace logs according to the condition you set; click ***Clear*** to clear all trace logs; click ***download*** to download trace logs.

### 3.12.9 PING Test

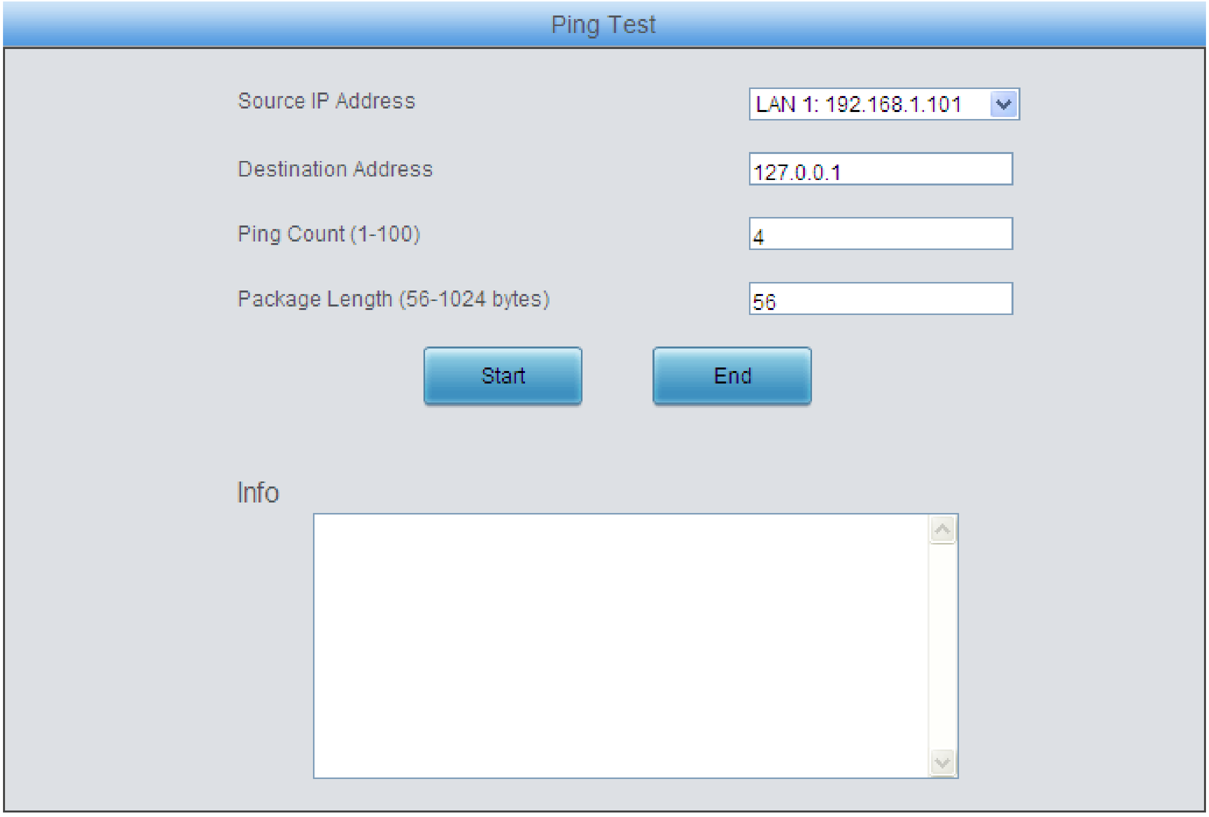



Figure 3-125 Ping Test Interface

See Figure 3-125 for the Ping Test interface. A Ping test can be initiated from the gateway on a designated IP address to check the connection status between them. The table below explains the configuration items shown in the above figure.

| Item | Description |
|---|---|
| ***Source IP Address*** | Source IP address where the Ping test is initiated. |
| ***Destination Address*** | Destination IP address on which the Ping test is executed. |
| ***Ping Count*** | The number of times that the Ping test should be executed. Range of value: 1~100. |
| ***Package Length*** | Length of a 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.

### 3.12.10 TRACERT Test

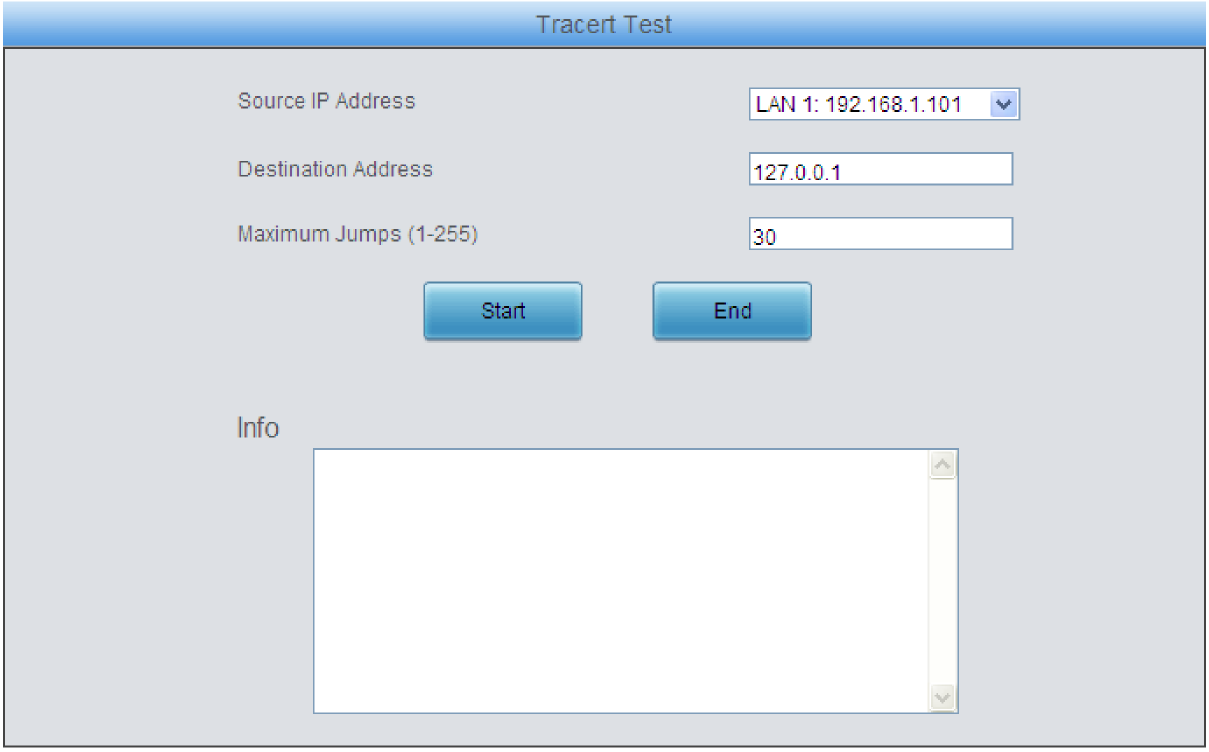



Figure 3-126 Tracert Test Interface

See Figure 3-126 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 can be returned in 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.

### 3.12.11 Modification Record

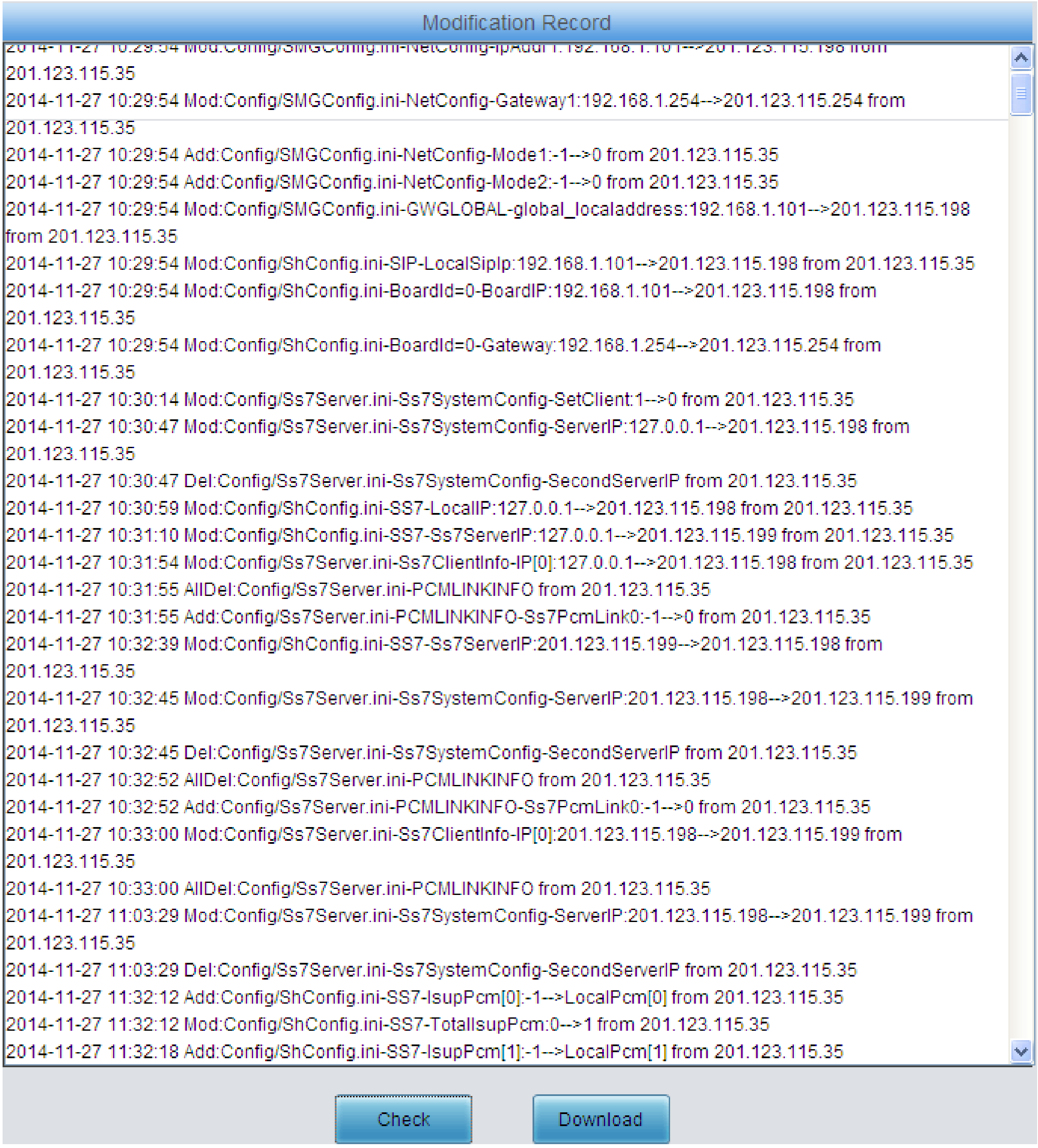



Figure 3-127 Modification Interface

The Modification Record interface is used to check the modification record on the web configuration. Click ***Check*** and the modification record will be shown on the dialog box. See Figure 3-127. Click ***Download*** to download the record file.

### 3.12.12 Backup & Upload

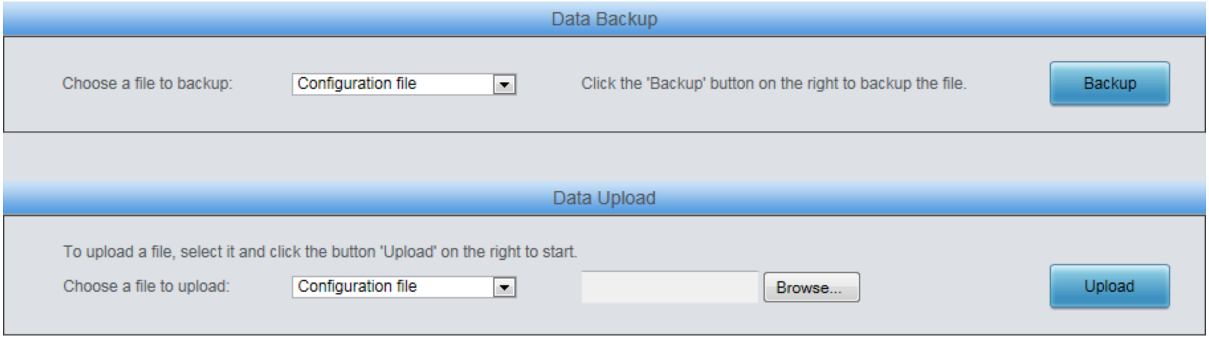



Figure 3-128 Backup & Upload Interface

See Figure 3-128 for the Backup and Upload interface. To back up data to your PC, you shall first choose the file in the pull-down list and then click ***Backup*** to start. To upload a file to the gateway, you shall first choose the file type in the pull-down list, then select it via ***Browse…***, and at last click ***Upload***. The gateway will automatically apply the uploaded data to overwrite the current configurations.

### 3.12.13 Factory Reset


Figure 3-129 Factory Reset Interface

See Figure 3-129 for the Factory Reset interface. Click ***Reset*** to restore all configurations on the gateway to factory settings.

### 3.12.14 Upgrade

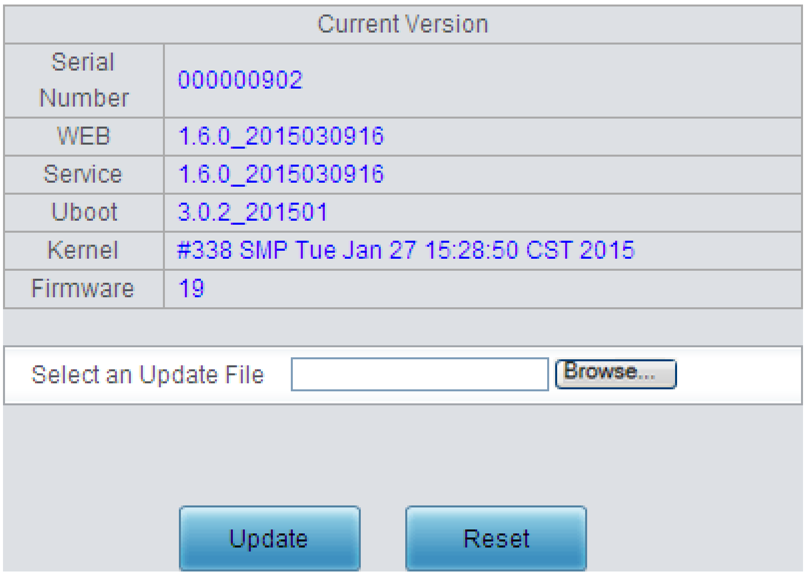



Figure 3-130 Upgrade Interface

See Figure 3-130 for the upgrade interface where you can upgrade the WEB, gateway service, kernel and firmware to new versions. Select the upgrade package “*.tar.gz” via ***Browse…*** and click ***Update*** (The gateway will do MD5 verification before upgrading and will not start to upgrade until it passes the verification). Wait for a while and the gateway will finish the upgrade automatically. Note that clicking ***Reset*** can only delete the selected update file but not cancel the operation of ***Update***.

### 3.12.15 Change Password

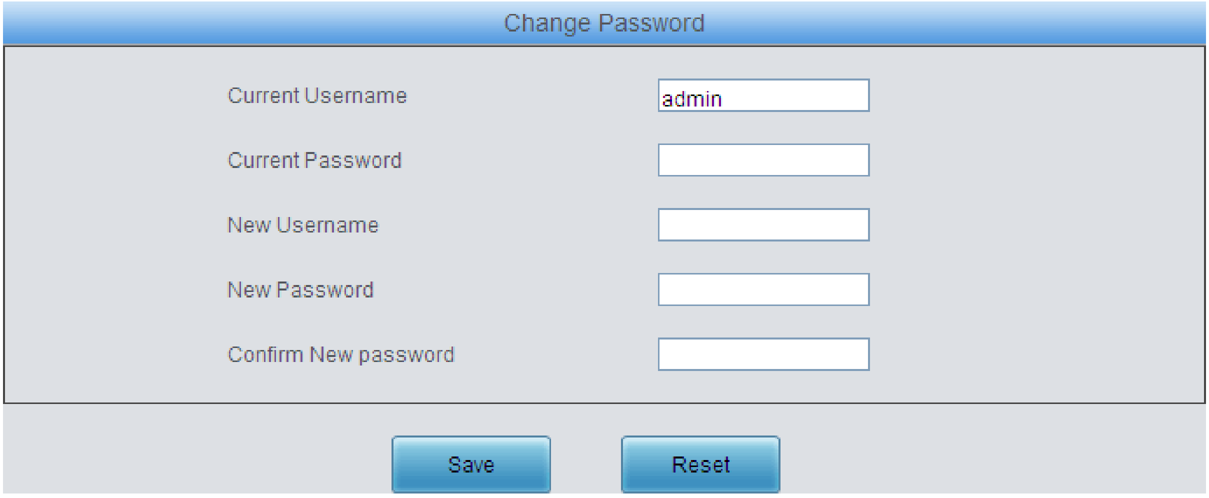



Figure 3-131 Password Changing Interface

See Figure 3-131 for the Password Changing interface where you can change username and password of the gateway. Enter the current password, the new username and password, and then confirm the new password. After configuration, click ***Save*** to apply the new username and password or click ***Reset*** to restore the configurations. After changing the username and password, you are required to log in again.

### 3.12.16 Device Lock

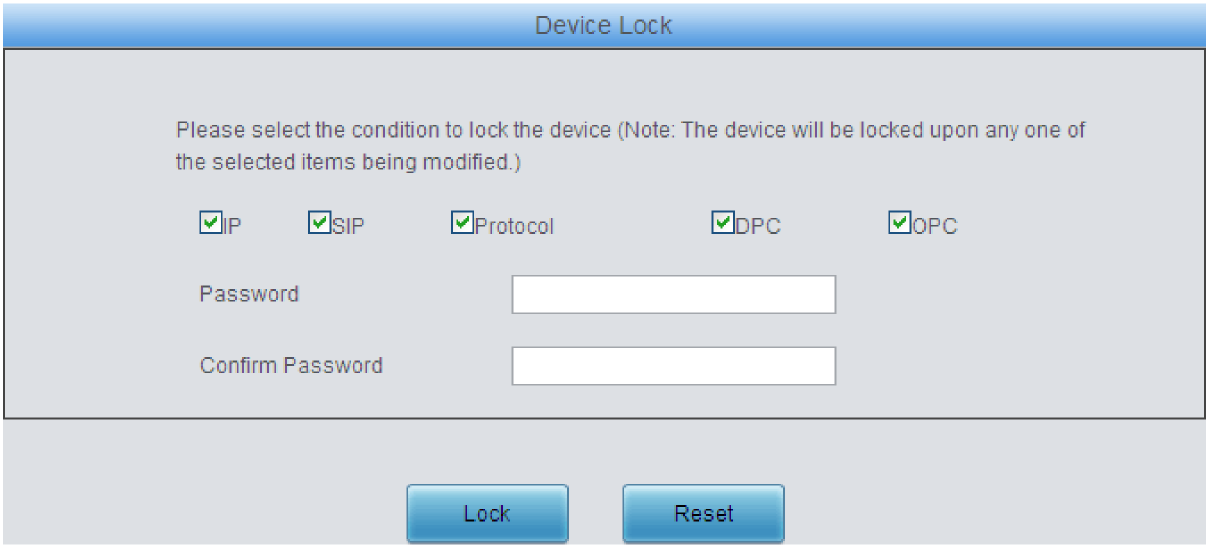



Figure 3-132 Device Lock Configuration Interface

See Figure 3-132 for the Device Lock Configuration interface. You can select at least one item as the condition to judge whether to lock the gateway or not, that is, as long as an item in the selected list is modified, the gateway will be locked. You shall enter the password which is necessary for device unlock. After your setting, click ***Lock*** and the device lock interface will be locked. See Figure 3-133. To unlock the interface, enter your password and click the ***Unlock*** button.

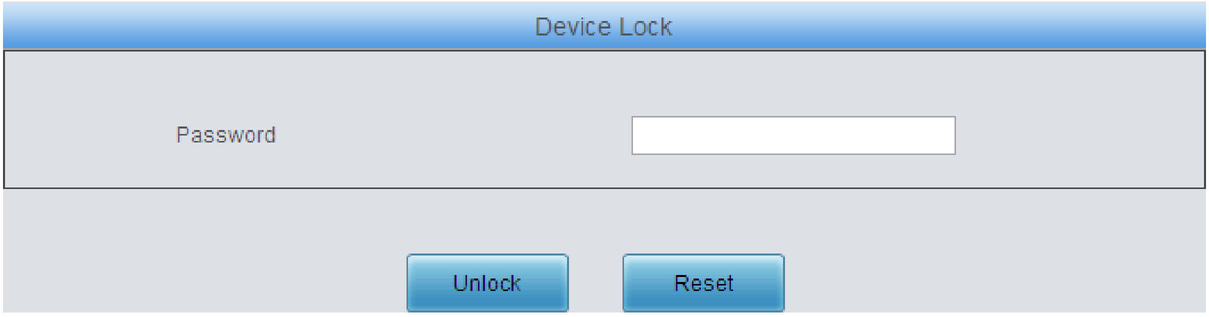



Figure 3-133 Unlock Device Interface

As long as an item in the selected list in Figure 3-132 is modified, the gateway will be locked. See Figure 3-134. In such case, only five pages including *system info*, *network setting*, *change password*, *device lock* and *restart* are available. Calls on both directions (from IP to PSTN and from PSTN to IP) will all be rejected. (The exception is, when the device is locked by Protocol, DPC or OPC being changed, calls will not be rejected until you restart the service.) Enter the device unlock interface (Figure 3-133) and input your password to unlock the device.


Figure 3-134 Device Lock Interface

### 3.12.17 Restart

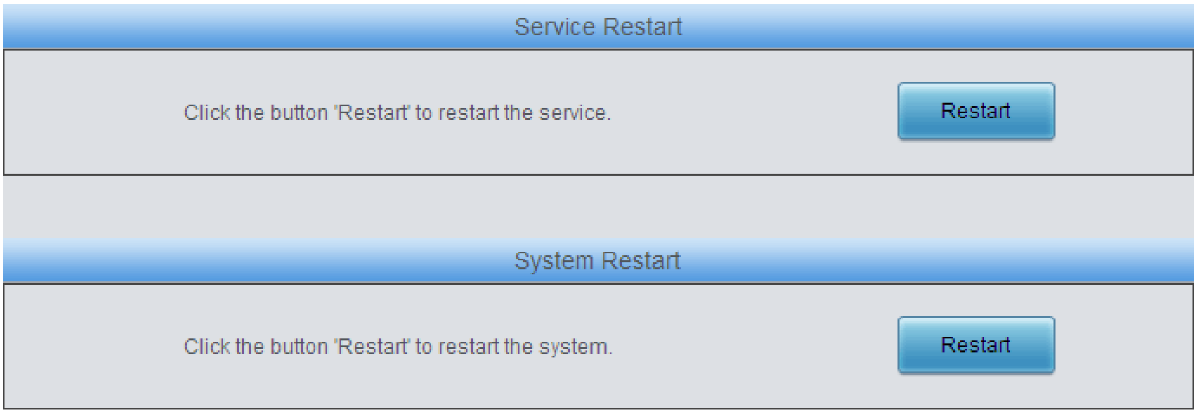
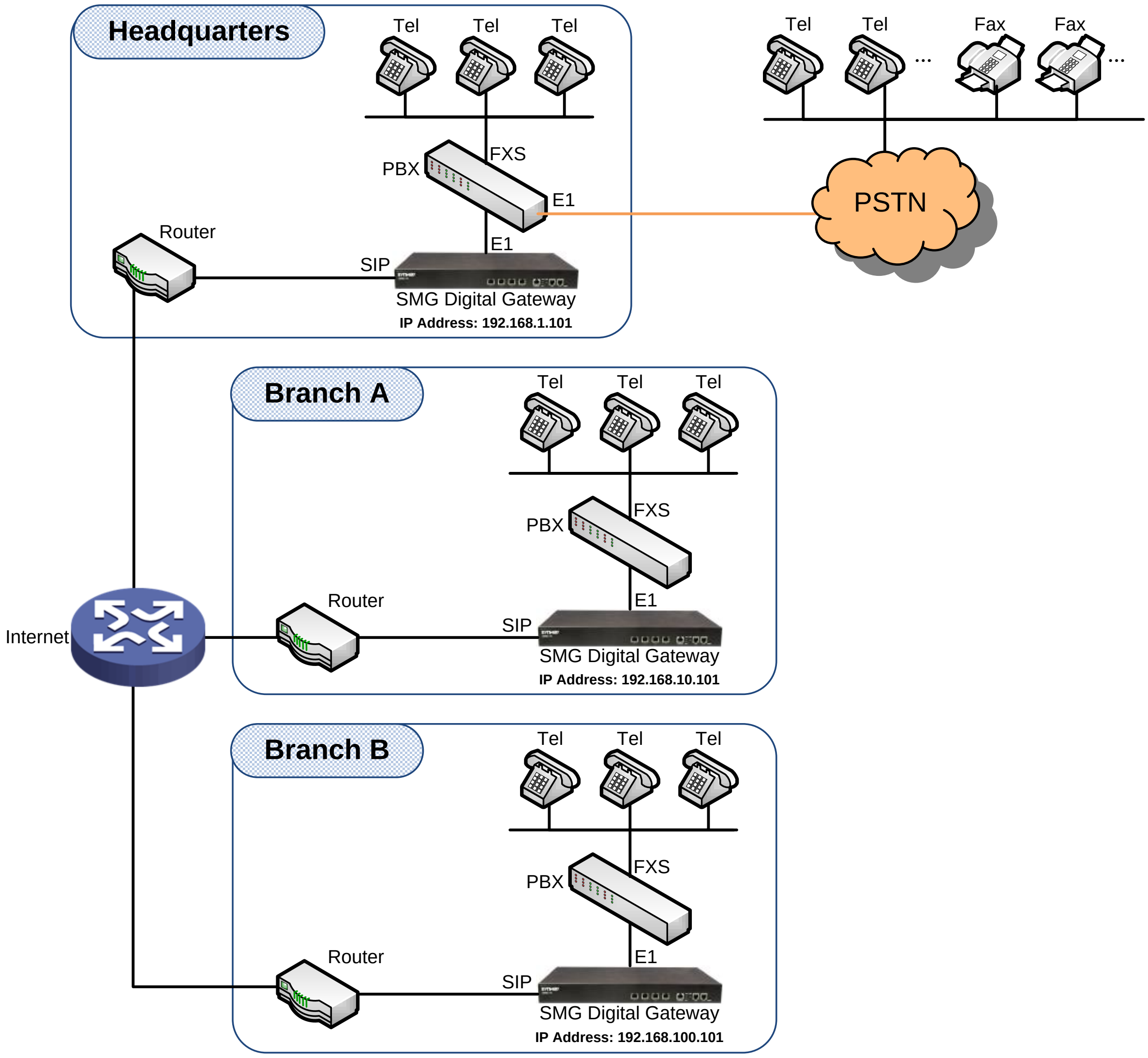



Figure 3-135 Service/System Restart Interface

See Figure 3-135 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.

# Chapter 4 Typical Applications

## 4.1 Application 1




Note: In this application, we assume that Branch A, Branch B and the headquarter have established VLAN using VPN technology.

Figure 4-1 Application 1

In this application, calls within the enterprise, i.e. calls among the headquarters, Branch A and Branch B, are all carried via SIP without PSTN. Outbound calls from the enterprise are all processed by the PBX at the headquarters. This application provides an enterprise with a unified interface for outbound call communications, and facilitates their call recording management as well.

This section takes SMG2120 as an example and introduces the configurations for the gateway application with the following dialing plan:

Call from the headquarters to Branch A: 8+EXT (extension number)

Call from the headquarters to Branch B: 7+EXT

Make an outbound call from the headquarters: 0+Number

Call from Branch A to the headquarters: 9+EXT

Call from Branch A to Branch B: 7+EXT

Make an outbound call from Branch A: 0+Number

Call from Branch B to the headquarters: 9+EXT

Call from Branch B to Branch A: 8+EXT

Make an outbound call from Branch B: 0+Number

### 4.1.1 Configurations for Headquarters

1. Configure SIP Settings for the headquarters.

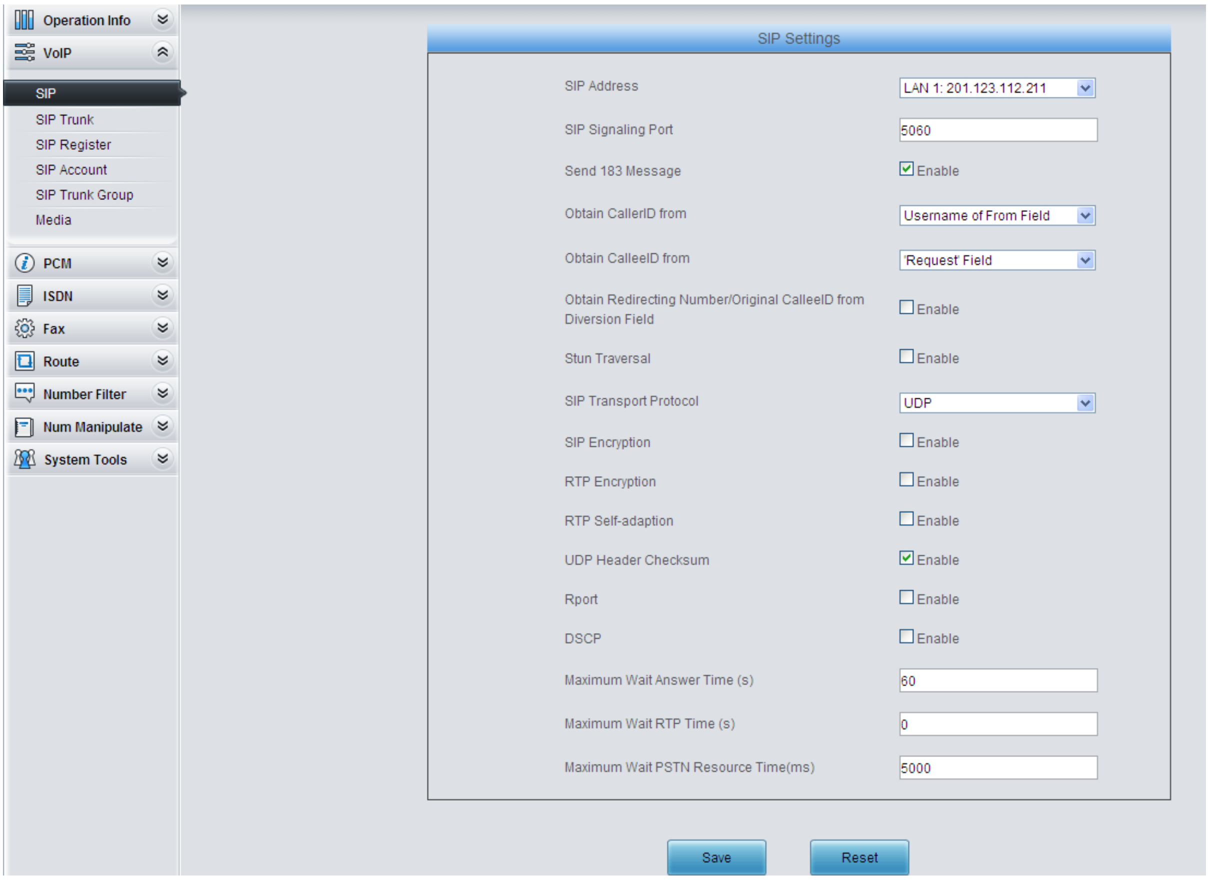



Figure 4-2

2. Add the IP addresses of the gateways at Branch A and Branch B into the SIP trunks.

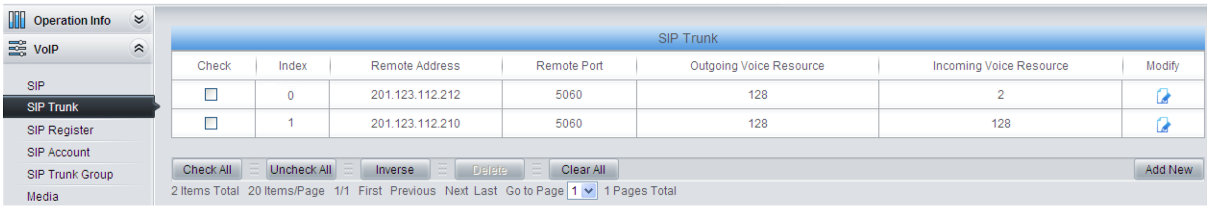



Figure 4-3

3. Add the SIP trunks at Branch A and Branch B into the corresponding SIP trunk groups.

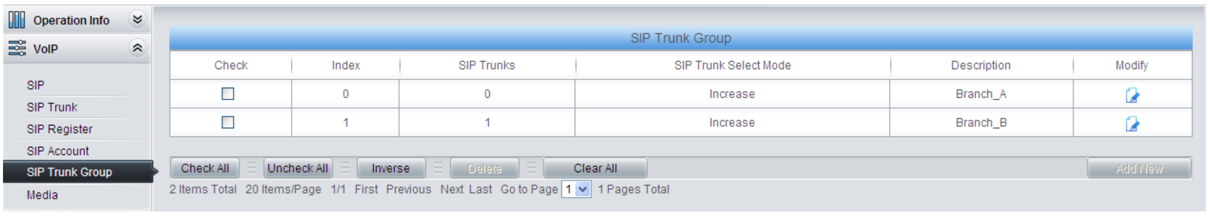



Figure 4-4

4. Set PCM.

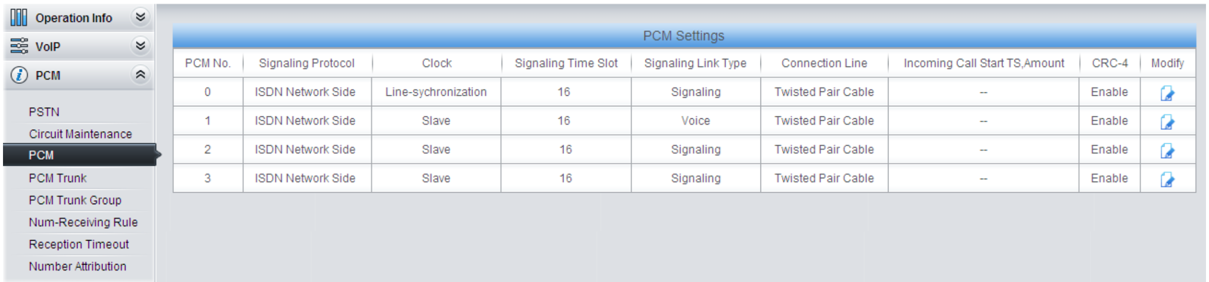



Figure 4-5

5. Add PCM trunk

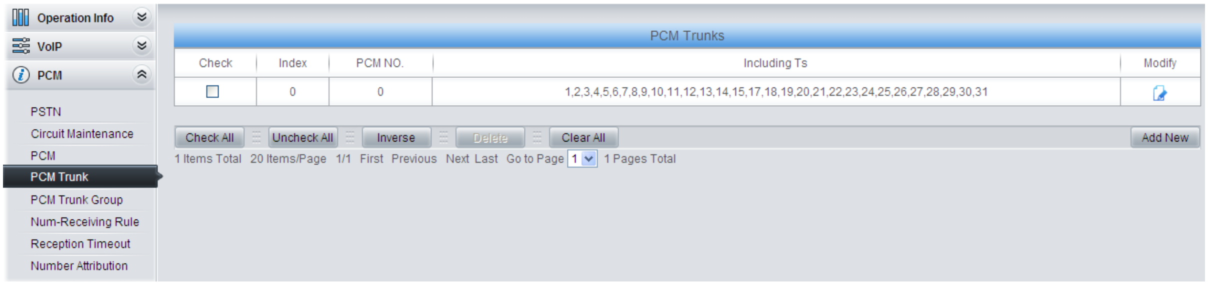



Figure 4-6

6. Add PCM trunk into the corresponding PCM trunk group.


Figure 4-7

7. Set routing parameters. You may adopt the default value 'Route before Number Manipulate' for both configuration items.

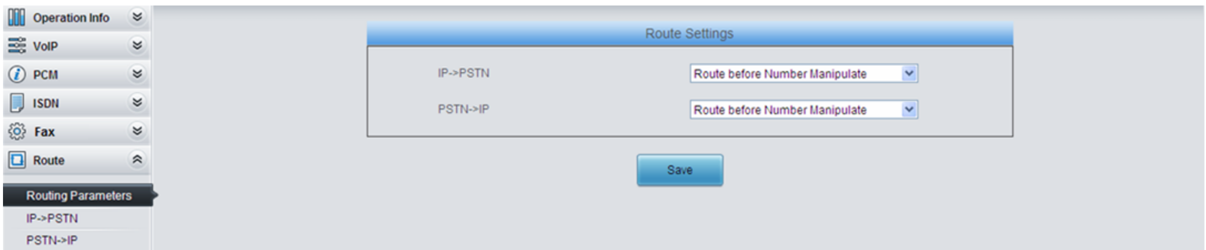



Figure 4-8

8. Set IP→PSTN routing rules to route calls from different SIP trunk groups to the corresponding

PCM trunk groups. In this step, all incoming IP calls will be routed to PCM Trunk Group 0 regardless of the CalleeID prefix.

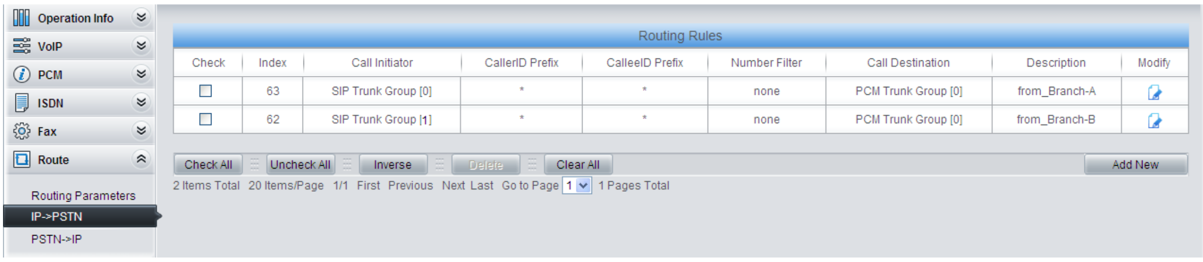



Figure 4-9

9. Set PSTN→IP routing rules to route calls from different PCM trunk groups to the corresponding SIP trunk groups. In this step, those calls with the CalleeID prefix 8 will be routed to SIP Trunk Group 0 while those with the CalleeID prefix 7 will be routed to SIP Trunk Group 1.


Figure 4-10

10. Set number manipulation rules. When the gateway receives a call from PSTN, it will first check the CalleeID prefix. If the CalleeID prefix is 7 or 8, the gateway will delete it before routing the call to the corresponding SIP trunk group.

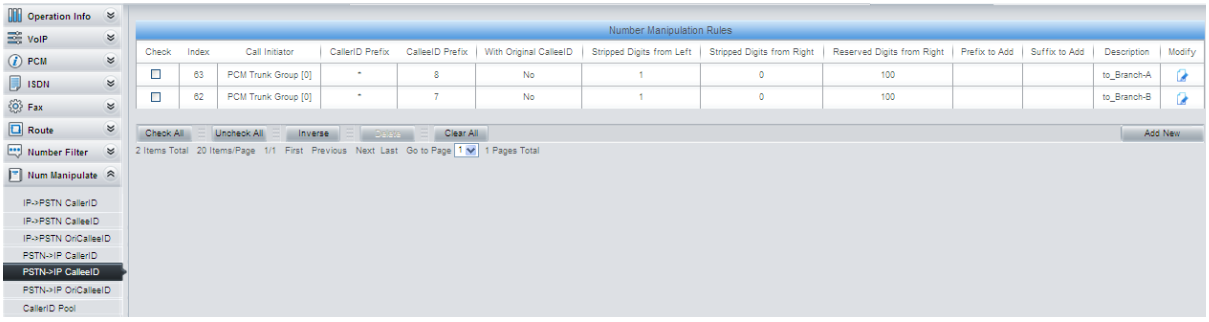



Figure 4-11

## 4.1.2 Configurations for Branch A

1. Configure SIP Settings for Branch A.

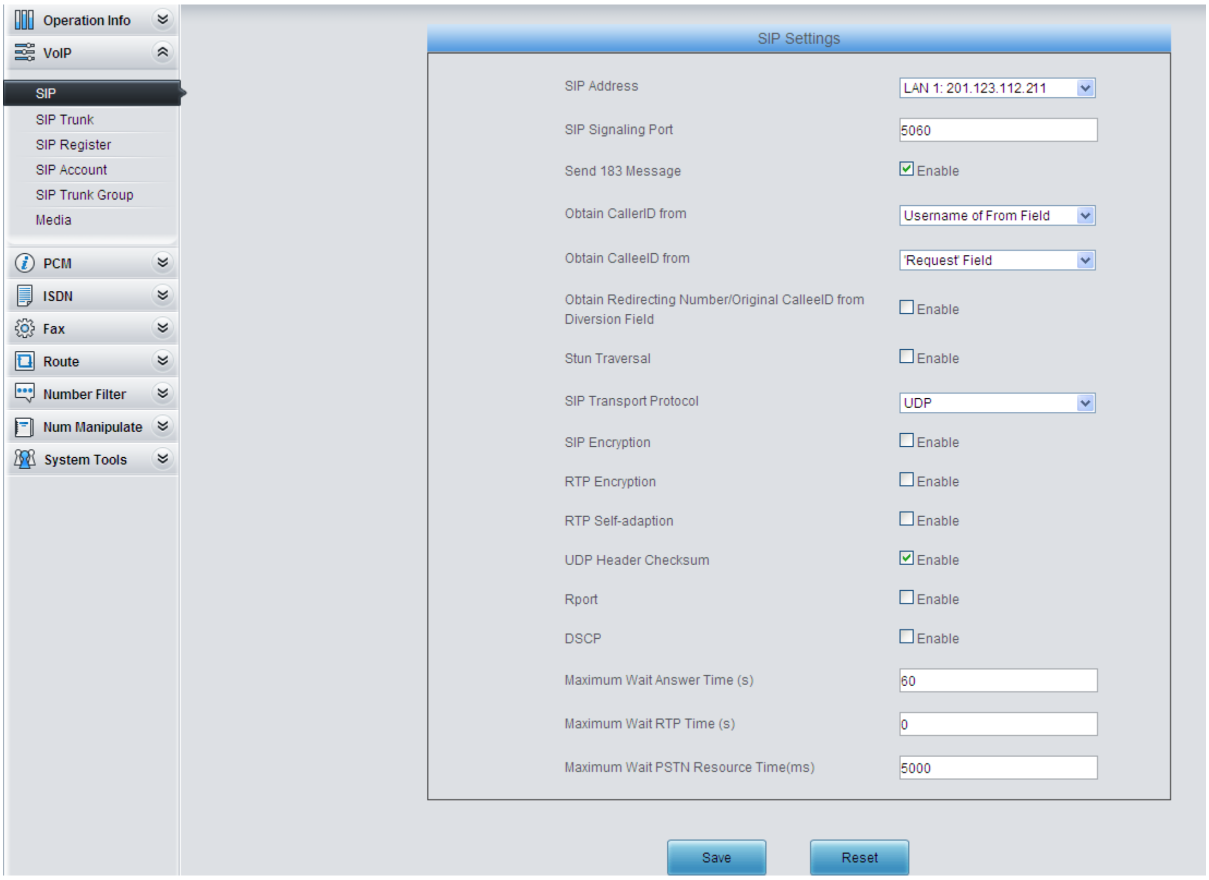



Figure 4-12

2. Add the IP addresses of the gateways at the headquarters and Branch B into the SIP trunks.

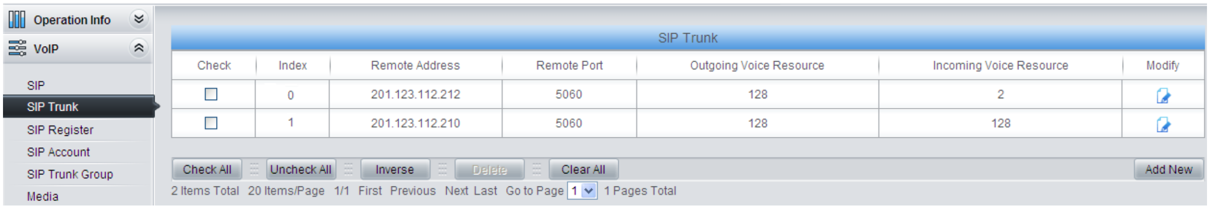



Figure 4-13

3. Add the SIP trunks at the headquarters and Branch B into the corresponding SIP trunk groups.

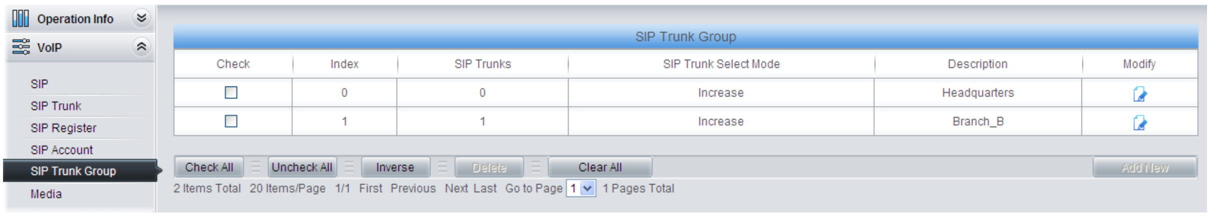



Figure 4-14

4. Set PCM.

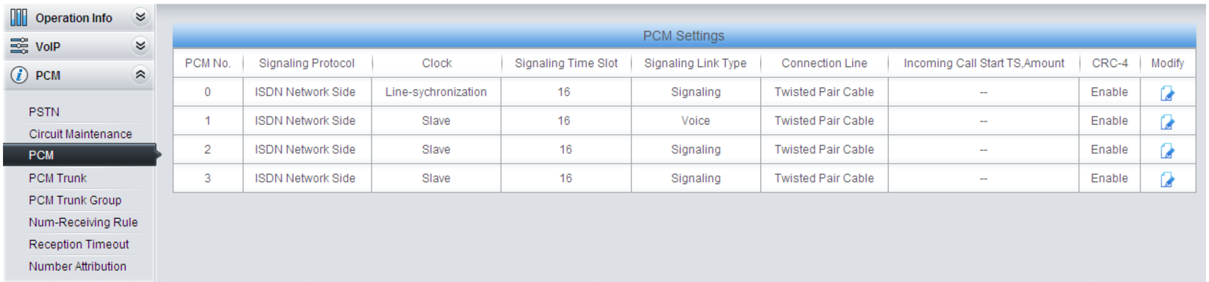



Figure 4-15

5. Add PCM trunk

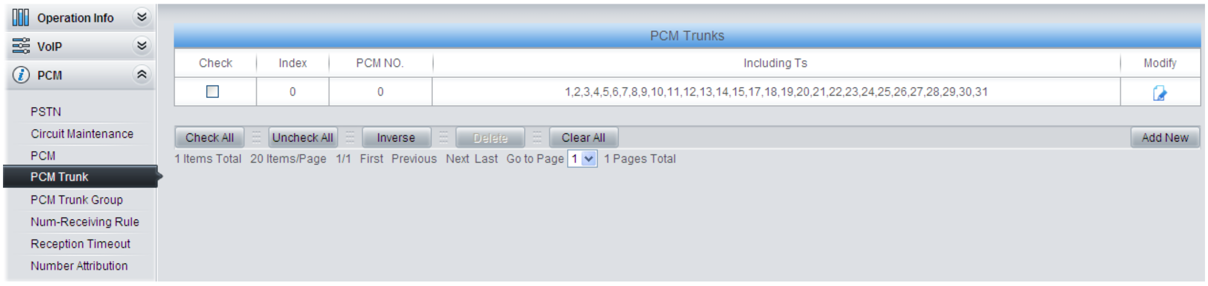



Figure 4-16

6. Add PCM trunk into the corresponding PCM trunk group.

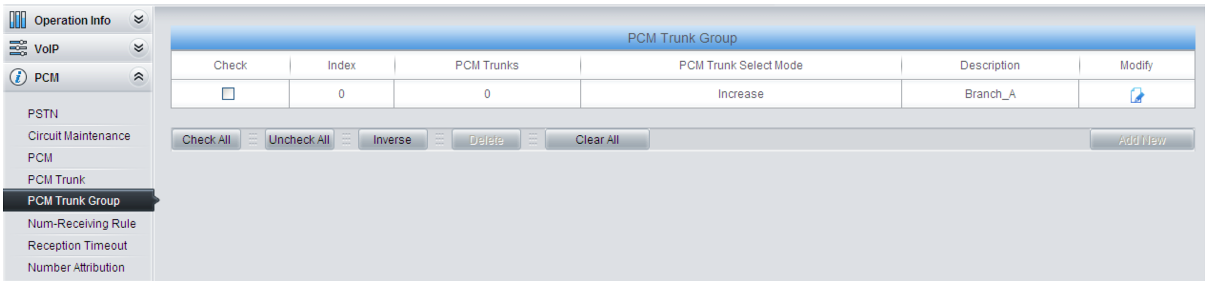



Figure 4-17

7. Set routing parameters. You may adopt the default value ‘Route before Number Manipulate’ for both configuration items.

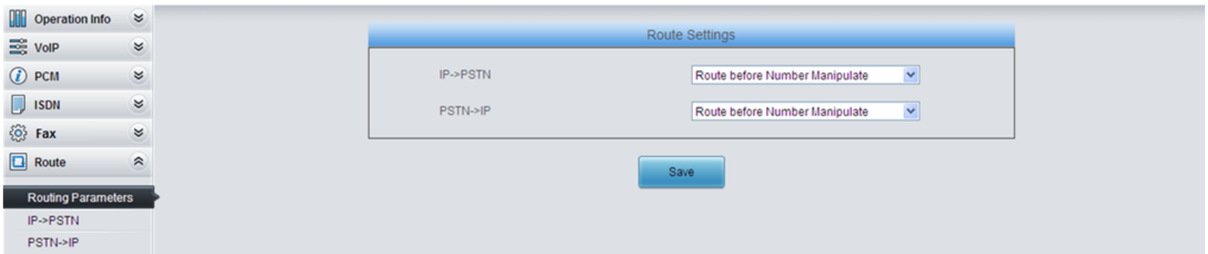



Figure 4-18

8. Set IP→PSTN routing rules to route calls from different SIP trunk groups to the corresponding PCM trunk groups. In this step, all incoming IP calls will be routed to PCM Trunk Group 0 regardless of the CalleeID prefix.


Figure 4-19

9. Set PSTN→IP routing rules to route calls from different PCM trunk groups to the corresponding SIP trunk groups. In this step, those calls with the CalleeID prefix 9 or 0 will be routed to SIP Trunk Group 0 while those with the CalleeID prefix 7 will be routed to SIP Trunk Group 1.


Figure 4-20

10. Set number manipulation rules. When the gateway receives a call from PSTN, it will first check the CalleeID prefix. If the CalleeID prefix is 9 or 7, the gateway will delete it before routing the call to the corresponding SIP trunk group.

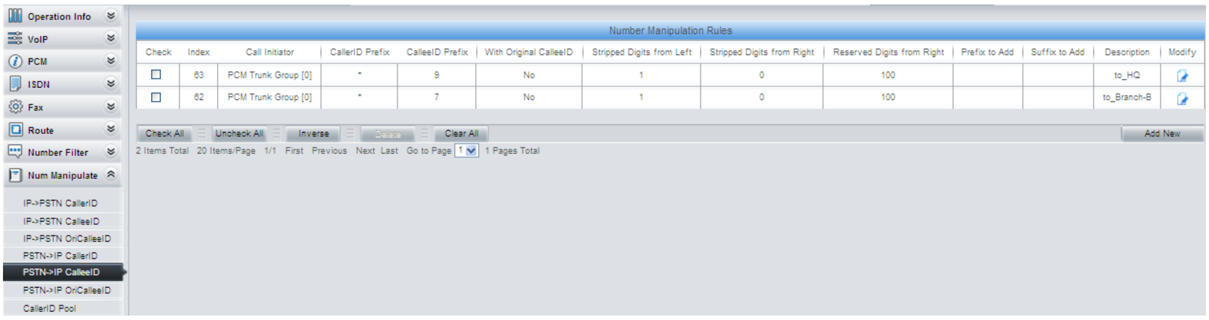



Figure 4-21

## 4.1.3 Configurations for Branch B

1. Configure SIP Settings for Branch B.

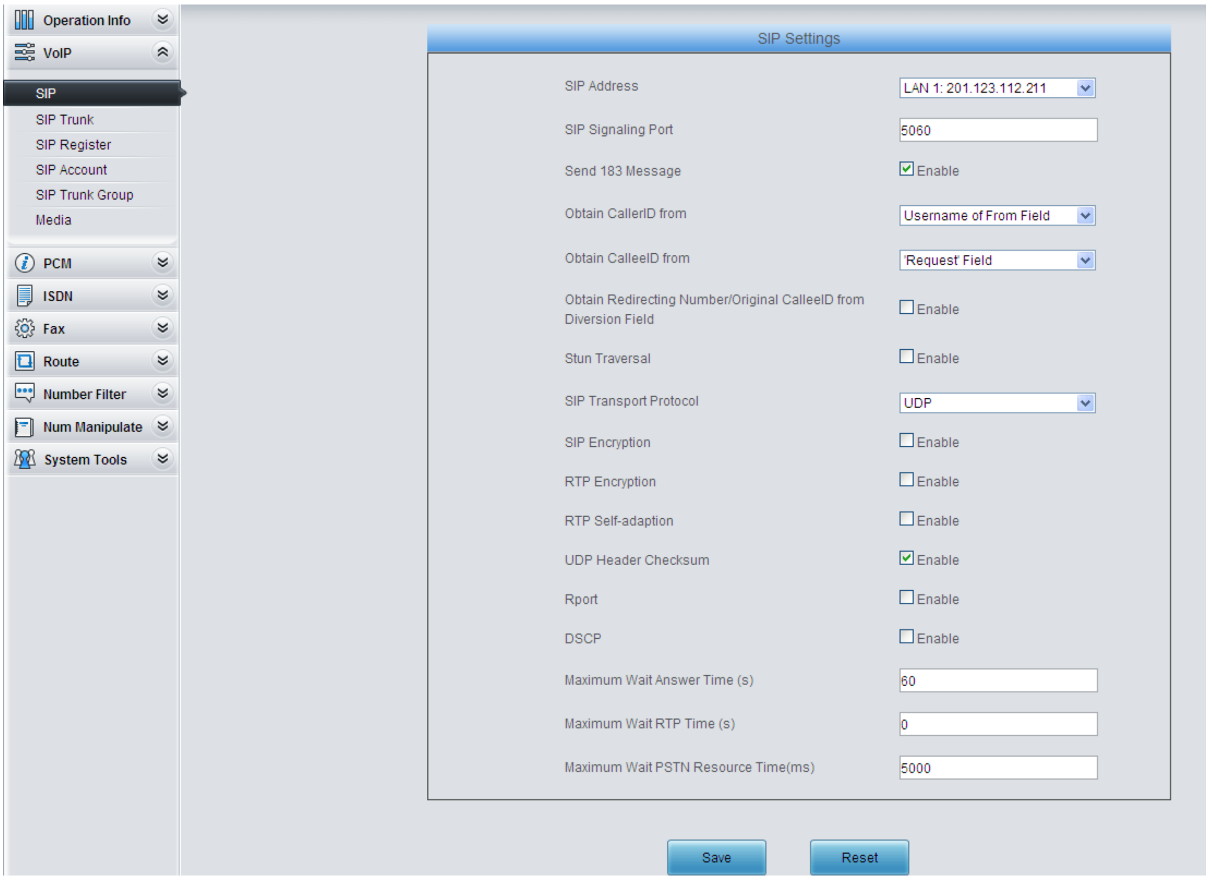



Figure 4-22

2. Add the IP addresses of the gateways at the headquarters and Branch A into the SIP trunks.

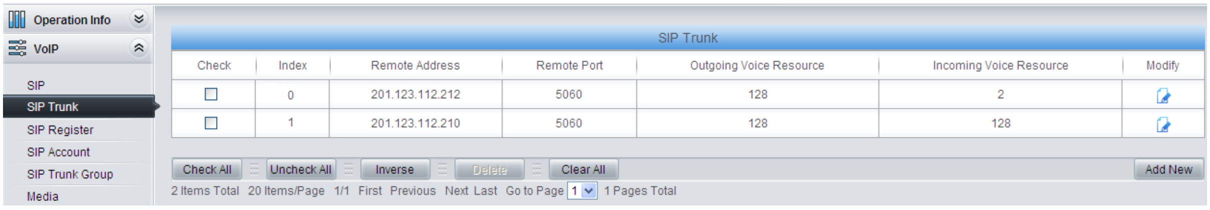



Figure 4-23

3. Add the SIP trunks at the headquarters and Branch A into the corresponding SIP trunk groups.

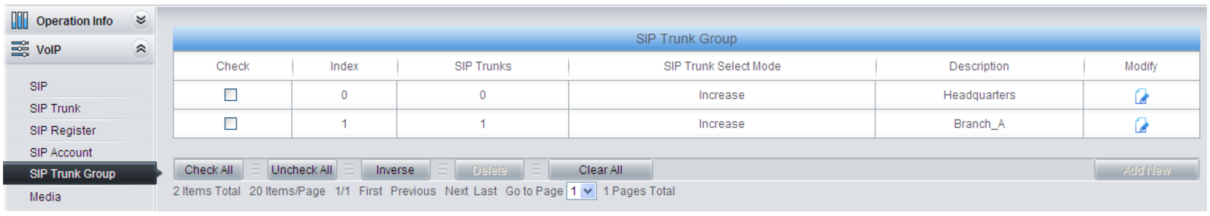



Figure 4-24

4. Set PCM.

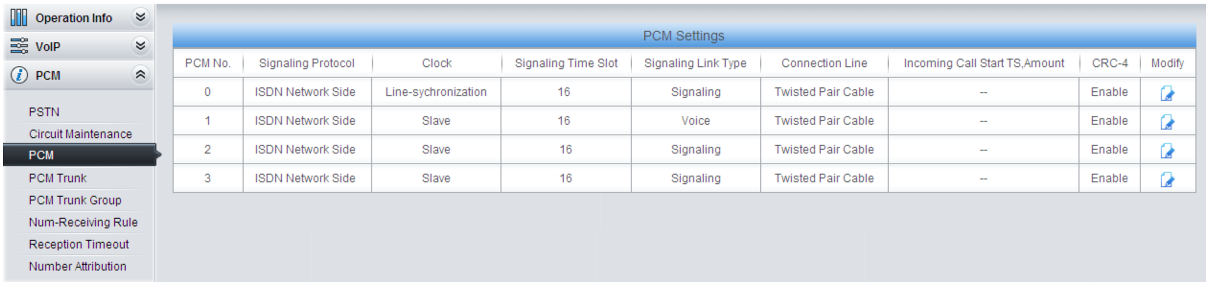



Figure 4-25

5. Add PCM trunk

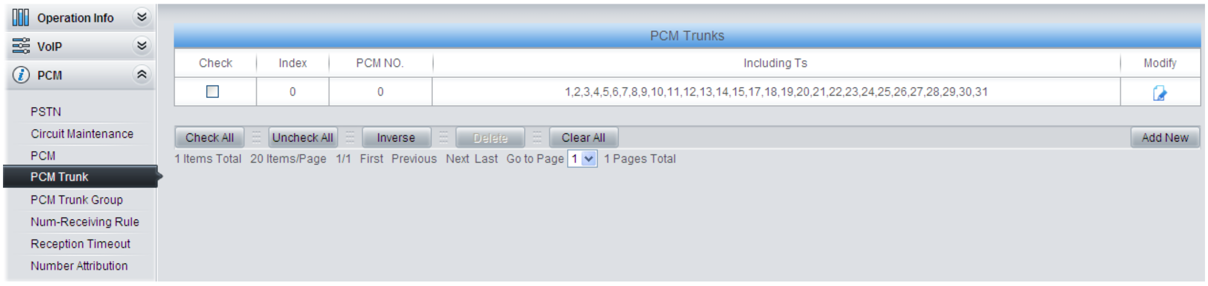



Figure 4-26

6. Add PCM trunk into the corresponding PCM trunk group.

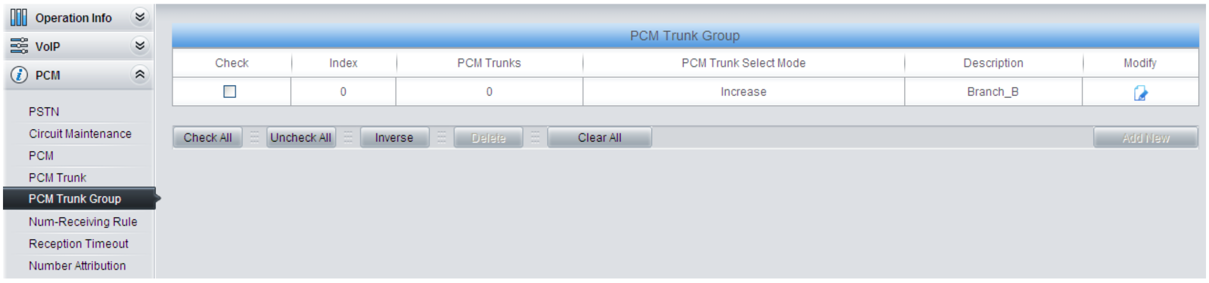



Figure 4-27

7. Set routing parameters. You may adopt the default value ‘Route before Number Manipulate’ for both configuration items.

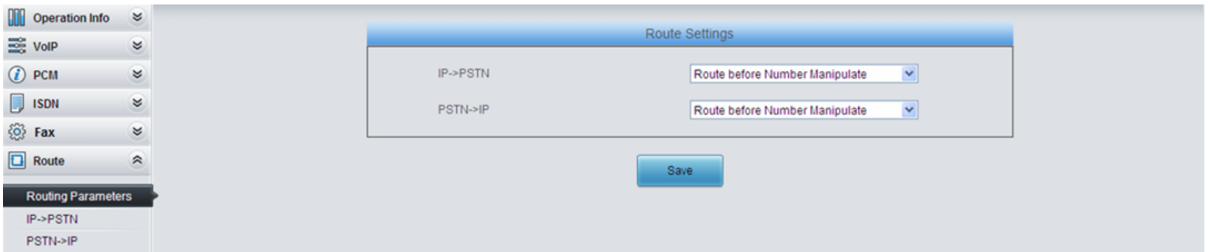



Figure 4-28

8. Set IP→PSTN routing rules to route calls from different SIP trunk groups to the corresponding PCM trunk groups. In this step, all incoming IP calls will be routed to PCM Trunk Group 0 regardless of the CalleeID prefix.

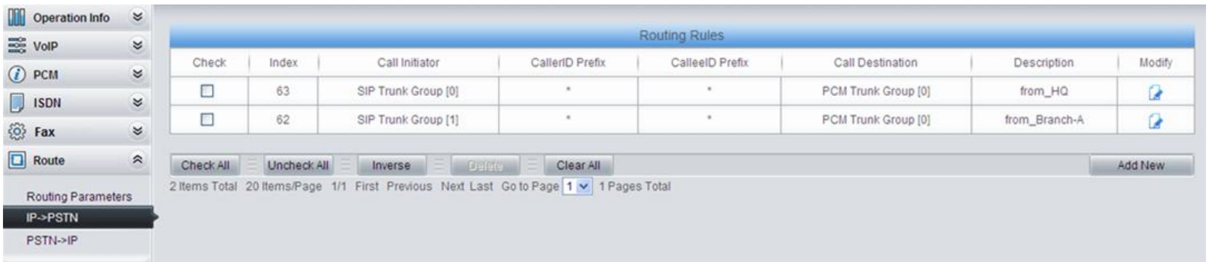



Figure 4-29

9. Set PSTN→IP routing rules to route calls from different PCM trunk groups to the corresponding SIP trunk groups. In this step, those calls with the CalleeID prefix 9 or 0 will be routed to SIP Trunk Group 0 while those with the CalleeID prefix 8 will be routed to SIP Trunk Group 1.


Figure 4-30

10. Set number manipulation rules. When the gateway receives a call from PSTN, it will first check the CalleeID prefix. If the CalleeID prefix is 9 or 8, the gateway will delete it before routing the call to the corresponding SIP trunk group.

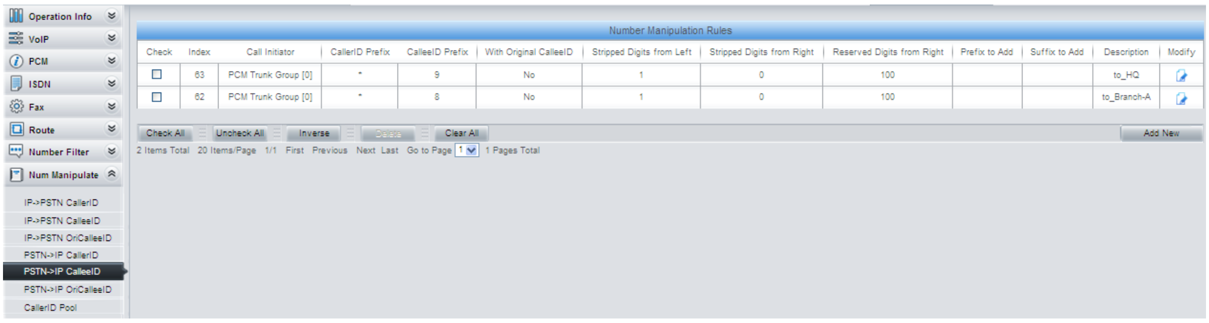



Figure 4-31

## 4.2 Application 2

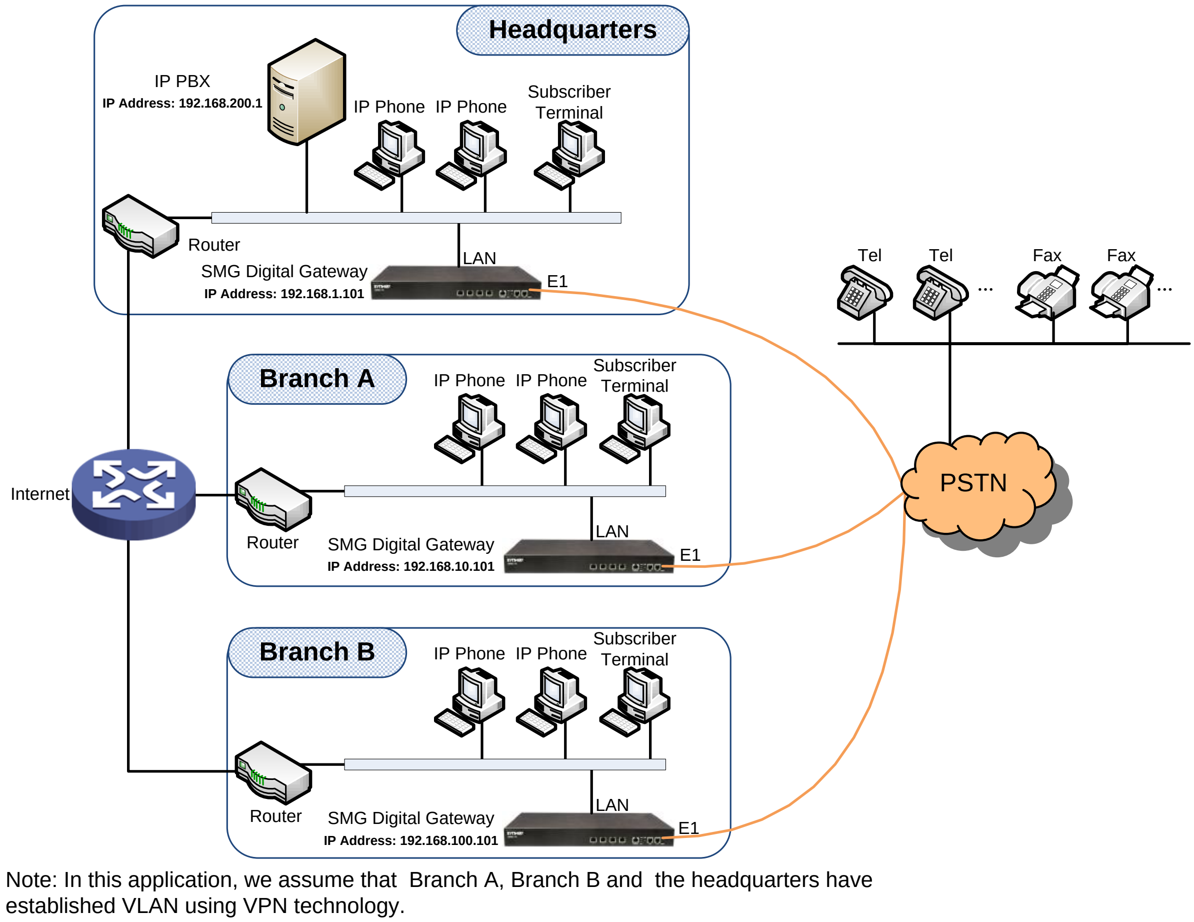



Figure 4-32 Application 2

In this application, the headquarters, Branch A and Branch B all have their own independent digital gateways to connect with the PSTN. Calls within the enterprise are all carried via SIP. Outbound calls to PSTN can be allocated to different gateways by the IP PBX. This application makes a full use of each E1/T1 trunk, helps an enterprise to eliminate the single point failure caused by device or network malfunction and enhance the stability of the IP telephony network.

This section takes SMG2120 as an example and introduces the configurations for the gateway application with the following dialing plan:

Make an outbound call from the headquarters: 0+Number

Make an outbound call from Branch A or Branch B: 0+Number

### 4.2.1 Configurations for Headquarters

1. Configure SIP Settings for the headquarters.

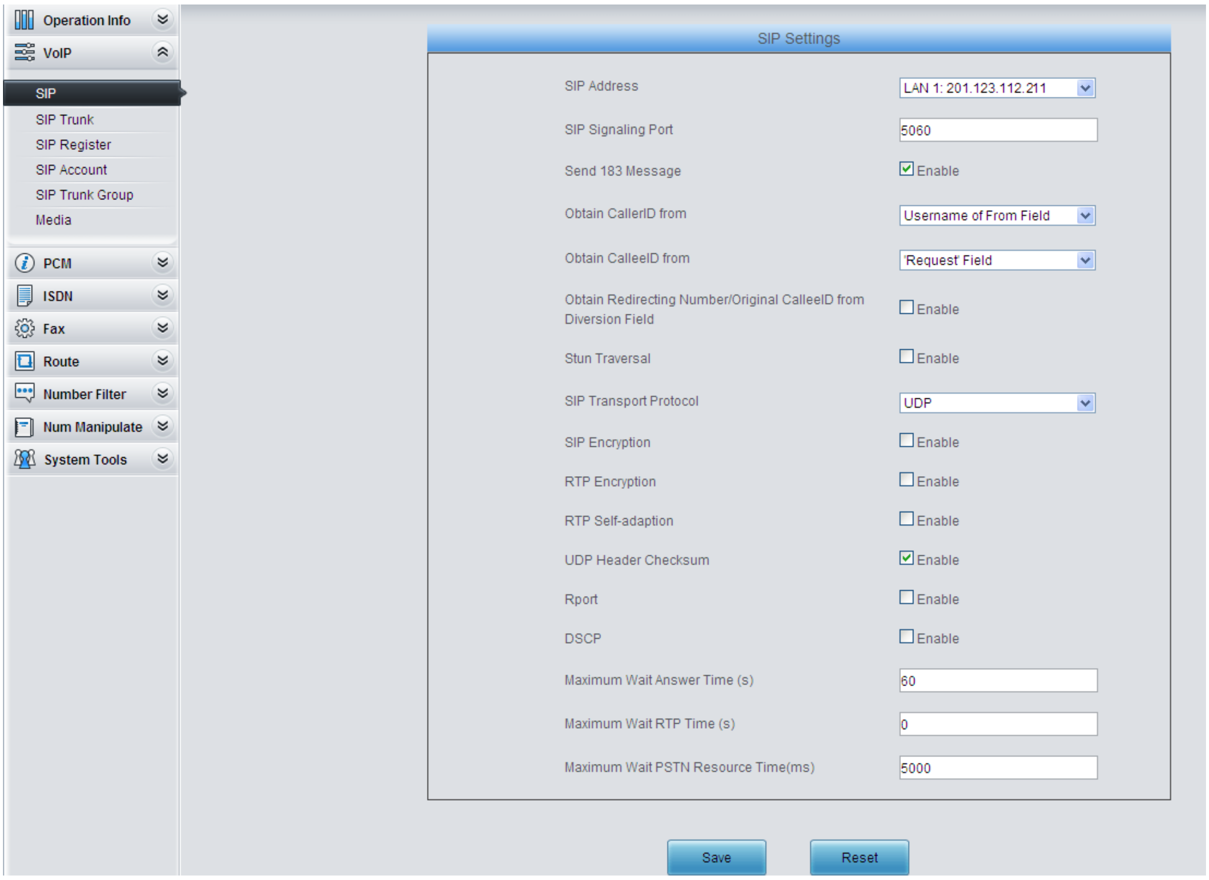



Figure 4-33

2. Add the IP address of the IP PBX into the SIP trunk.

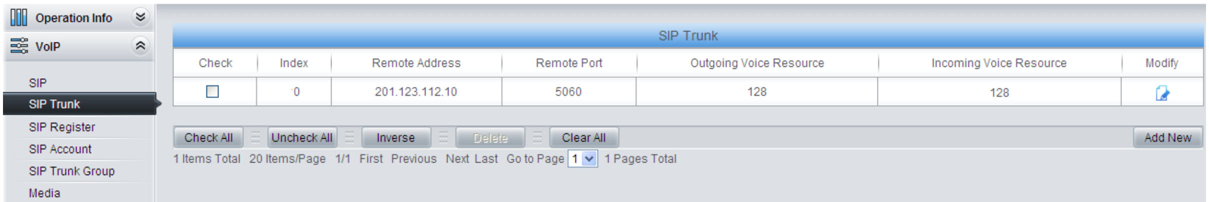



Figure 4-34

3. Add the SIP trunk into the corresponding SIP trunk group.

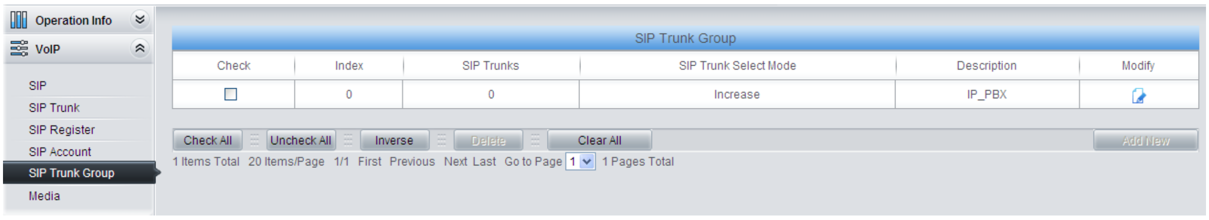



Figure 4-35

4. Set PCM.

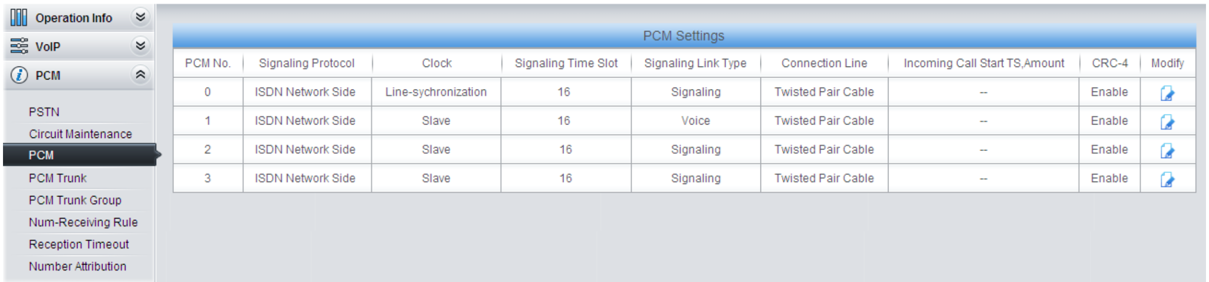



Figure 4-36

5. Add PCM trunk

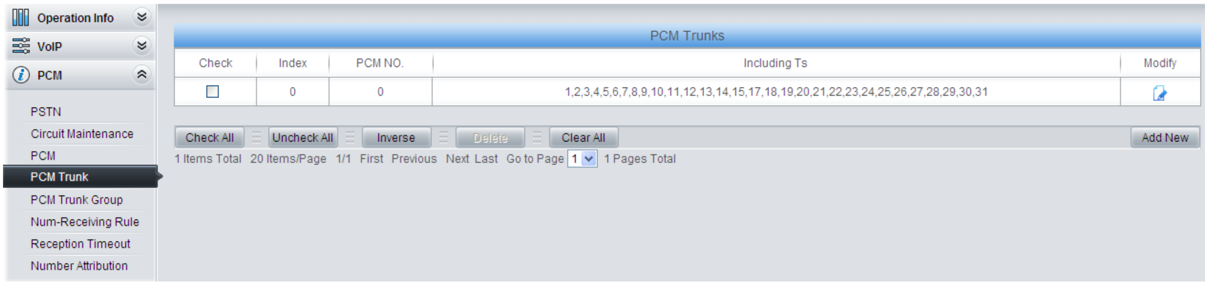



Figure 4-37

6. Add PCM trunk into the corresponding PCM trunk group.

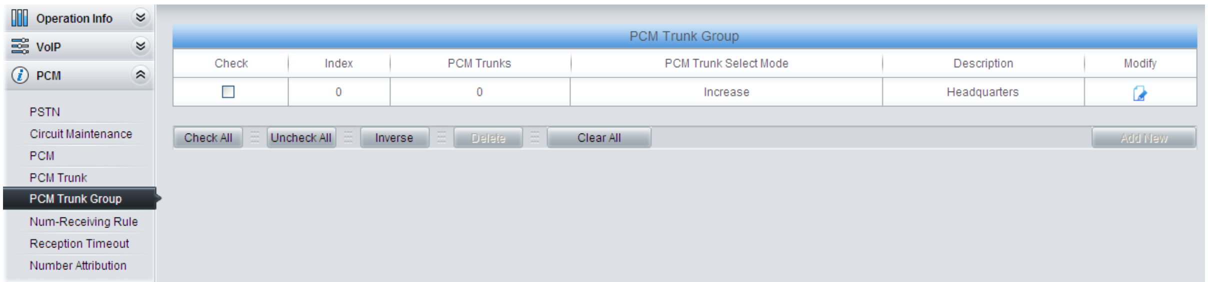



Figure 4-38

7. Set routing parameters. You may adopt the default value 'Route before Number Manipulate' for both configuration items.

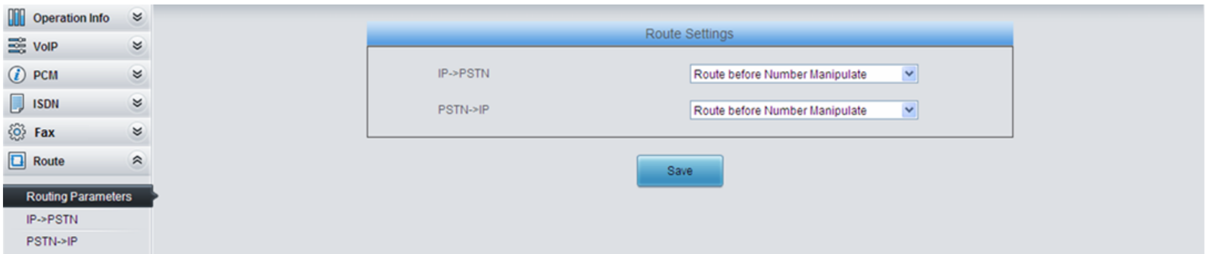



Figure 4-39

8. Set IP→PSTN routing rules to route calls from different SIP trunk groups to the corresponding PCM trunk groups. In this step, all incoming IP calls will be routed to PCM Trunk Group 0 regardless of the CalleeID prefix.

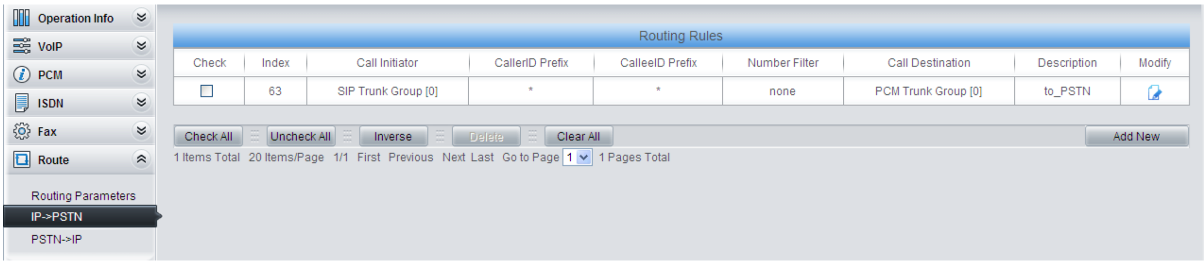



Figure 4-40

9. Set PSTN→IP routing rules to route calls from different PCM trunk groups to corresponding SIP trunk groups. In this step, all incoming calls from PSTN will be routed to SIP Trunk Group 0 regardless of the CalleeID prefix.

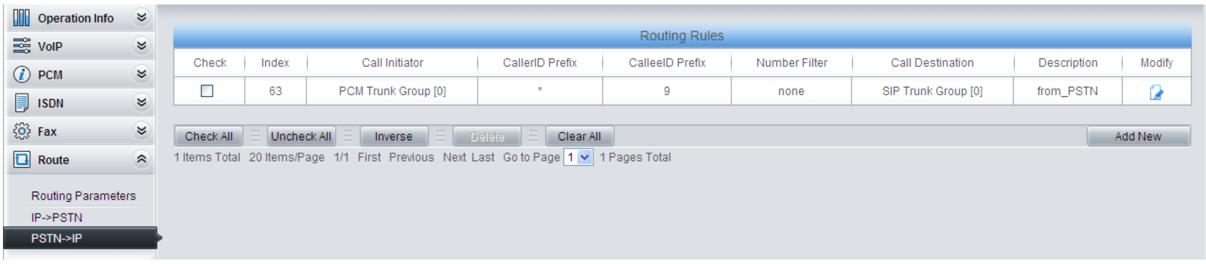



Figure 4-41

**Note:** In this application, the number manipulation feature is implemented by the IP PBX. That is, when a subscriber at the headquarters makes an outbound call dialing “0+Number”, the IP PBX will delete the prefix 0 before rooting it to the gateway. Therefore, it is not necessary to configure the number manipulation rules on the gateway. However, you shall add to the IP PBX the number manipulation rule of deleting the CalleeID prefix 0.

## 4.2.2 Configurations for Branches

For the gateways at Branch A and Branch B, you shall fill in their actual IP addresses to the configuration item ‘SIP Address’. All the other configurations are the same as those for the headquarters.

# Appendix A Technical Specifications

## Dimensions

*440×44×267 mm$^3$*

## Weight

*About 3.1 kg*

## Environment

*Operating temperature: 0 ℃—55 ℃*

*Storage temperature: -20 ℃—85 ℃*

*Humidity: 8%— 90% non-condensing*

*Storage humidity: 8%— 90% non-condensing*

## LAN

*Amount: 2 (10/100/1000 BASE-TX (RJ-45))*

*Self-adaptive bandwidth supported*

*Auto MDI/MDIX supported*

## E1/T1 Port

*Amount: 1/2/4/8/16*

*Type: RJ45*

## Console Port

*Amount: 1 (RS-232)*

*Baud rate: 115200bps*

*Connector: RJ45 （See Hardware Description for signal definition）*

*Data bits: 8 bits*

*Stop bit: 1 bit*

*Parity unsupported*

*Flow control unsupported*

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

## Power Requirements

*Input power: 100~240V AC*

*Maximum power consumption：*

*SMG2000 series: ≤12W*

*SMG3000 series: ≤22W*

## Signaling & Protocol

*SS7: TUP, ISUP*

*ISDN: ISDN User Side, ISDN Network Side*

*SS1: SS1 Signaling*

*SIP signaling: 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 |

## Sampling Rate

*8kHz*

## Safety

*Lightning resistance: Level 4*

# Appendix B Troubleshooting

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

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

LAN1: 192.168.1.101

LAN2: 192.168.0.101

**2. In what cases can I conclude that the SMG digital 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 E1/T1 trunk of the gateway is well connected, but the E1/T1 indicators never light up after the gateway startup or their indications do not comply with the actual state.

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

**3. What to do if I cannot enter the WEB interface of the SMG digital 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 change the IP address of the gateway, add your new IP address into the above settings.

# Appendix C ISUP (ISDN) Pending Cause to SIP Status Code

| ISUP (ISDN) Return Value | Cause | SIP Status Code | Implication |
|---|---|---|---|
| 1 | Unallocated (unassigned) number | 404 | Not found |
| 2 | No route to specified transit network | 404 | Not found |
| 3 | No route to destination | 404 | Not found |
| 26 | Non-selected user clearing | 404 | Not found |
| 16 | Normal call clearing (and the failure reason is that Waiting for off-hook signal from called party is overtime) | 603 | Decline |
| 17 | User busy | 486 | Busy here |
| 132 | Network busy (internal definition, only applies to ISDN) | 486 | Busy here |
| 21 | Call rejected | 486 | Busy here |
| 18 | No user responding | 408 | Request timeout |
| 19 | No answer from user (user alerted) | 480 | Temporarily unavailable |
| 20 | Subscriber absent | 480 | Temporarily unavailable |
| 31 | Normal, unspecified | 480 | Temporarily unavailable |
| 136 | Connection after pickup failed (internal definition, only applies to ISDN) | 480 | Temporarily unavailable |
| 137 | Pickup time out (internal definition, only apply to ISDN) | 480 | Temporarily unavailable |
| 55 | Incoming calls barred within CUG | 403 | Forbidden |
| 57 | Bearer capability not authorized | 403 | Forbidden |
| 87 | User not member of CUG | 403 | Forbidden |
| 22 | Number changed | 410 | Gone |
| 27 | Destination out of order | 502 | Bad gateway |
| 28 | Invalid number format | 484 | Address incomplete |
| 29 | Facility rejected | 501 | Not implemented |
| 79 | Service or option not implemented, unspecified | 501 | Not implemented |
| 34 | No circuit/channel available | 503 | Service unavailable |
| 38 | Network out of order | 503 | Service |

| | | | |
|---|---|---|---|
| | | | unavailable |
| 41 | Temporary failure | 503 | Service unavailable |
| 42 | Switching equipment congestion | 503 | Service unavailable |
| 47 | Resource unavailable, unspecified | 503 | Service unavailable |
| 58 | Bearer capability not presently available | 503 | Service unavailable |
| 88 | Incompatible destination | 503 | Service unavailable |
| 133 | Circuit restarted (internal definition, only applies to ISDN) | 503 | Service unavailable |
| 134 | Temporary fault (internal definition, only applies to ISDN) | 503 | Service unavailable |
| 135 | Data link failure (internal definition, only applies to ISDN) | 503 | Service unavailable |
| 65 | Bearer capability not implemented | 488 | Not acceptable here |
| 70 | Only restricted digital information bearer capability is available | 488 | Not acceptable here |
| 102 | Recovery on timer expiry | 504 | Server time-out |
| 128 | T303 time out (internal definition, only applies to ISDN) | 504 | Server time-out |
| 129 | T304 time out (internal definition, only applies to ISDN) | 504 | Server time-out |
| 130 | T310 time out (internal definition, only applies to ISDN) | 504 | Server time-out |
| 111 | Protocol error, unspecified | 500 | Server internal error |
| 127 | Interworking, unspecified | 500 | Server internal error |
| Others | Others | 500 | Server internal error |

# Appendix D TUP Pending Cause to SIP Status Code

| TUP Return Value | Cause | SIP Status Code | Implication |
|---|---|---|---|
| 11 | SS7 signaling: receives SSB message from remote PBX | 486 | Busy here |
| 12 | SS7 signaling: receives SLB message from remote PBX | 486 | Busy here |
| 13 | SS7 signaling: receives STB message from remote PBX | 486 | Busy here |
| 67 | TUP: receives CBK message from remote PBX | 403 | Forbidden |
| 21 | SS7 signaling: receives ACB message from remote PBX | 403 | Forbidden |
| 18 | SS7 signaling: receives CFL message from remote PBX | 403 | Forbidden |
| 14 | SS7 signaling: receives UNN message from remote PBX | 488 | Not acceptable here |
| 16 | SS7 signaling: receives CGC message from remote PBX | 406 | Not acceptable |
| 17 | SS7 signaling: receives NNC message from remote PBX | 406 | Not acceptable |
| 19 | SS7 signaling: receives LOS message from remote PBX | 406 | Not acceptable |
| 20 | SS7 signaling: receives SST message from remote PBX | 406 | Not acceptable |
| 22 | SS7 signaling: receives DPN message from remote PBX | 406 | Not acceptable |
| 23 | SS7 signaling: receives EUM message from remote PBX | 406 | Not acceptable |
| 24 | SS7 signaling: receives ADI message from remote PBX | 484 | Address incomplete |

# Appendix E Technical/sales Support

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

## Headquarters

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