

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



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# **Revision History**

Version	Date	Comments	
Version 1.3.0	2014-06	Initial publication.	
Version 1.3.1	2014-08	New revision	
Version 1.3.2	2014-10	New revision	
Version 1.5.0	2014-12	Add description on the new series SMG3016	
Version 1.5.1	2015-01	Add description on the new series SMG3008	
Version 1.6.0	2015-03	New revision	

Note: Please visit our website <u>http://www.synway.net</u> 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**

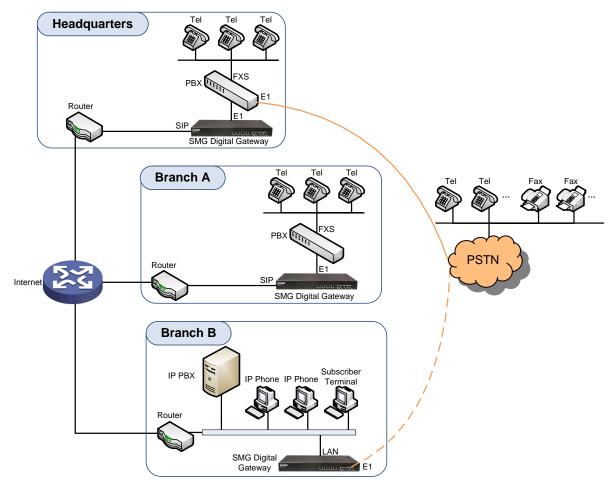


Figure 1-1 Typical Application



# 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           DTMF Mode         RFC2833, SIP INFO, INBAND		
Fax	Fax ModeT.38, Pass-ThroughBaud Rate14400bps, 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		
	Support of user interface, gateway service, kernel and firmware upgrades based on WEB		
Software Upgrade			

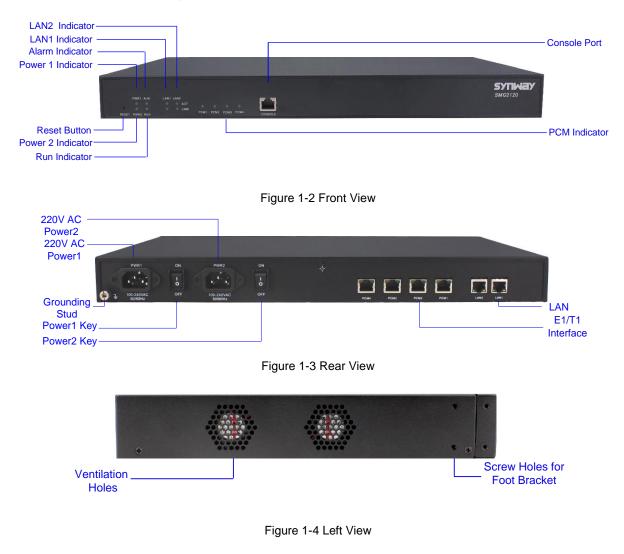


SysLog Type

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:

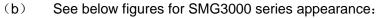




Figure 1-5 Front View



220V AC Power2 220V AC Power1		
Grounding Stud		AN
Power1 Key		E1/T1
Power2 Key —		nterface
F	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		
	Amount: 2		
	Type: RJ-45		
LAN	Bandwidth: 10/100/1000Mbps		
	Self-Adaptive Bandwidth Supported		
	Auto MDI/MDIX Supported		
F4/T4	Amount: 1/2/4/8/16		
E1/T1	Type: RJ-45		
	Amount: 1		
	Type: RS-232		
	Baud Rate: 115200 bps		
	Connector: RJ45 (See Figure 1-7 for signal definition)		
Console Port	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		
Power Rey	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 <u>1.4 Alarm Info</u> .		
Alarm Indicator	Alarms the device malfunction. For more details, refer to <u>1.4 Alarm Info</u> .		
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
Channel Indicators	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.

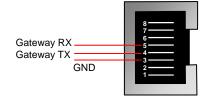


Figure 1-7 Console Port Signal Definition

For other hardware parameters, refer to <u>Appendix A Technical Specifications</u>.

#### 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	
	Go out	System is not yet started.	
Run Indicator	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. 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 <u>Appendix E Technical/sales Support</u> 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  $75\Omega$ -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 120 $\Omega$ -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.

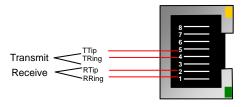




Figure 2-1 Pin Layout for E1 Interface

#### 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 <u>3.1 System Login</u>. We suggest you change the initial username and password via 'System Tools  $\rightarrow$  Change Password' on the WEB interface as soon as possible after your first login. For detailed instructions about changing the password, refer to <u>3.12.15 Change Password</u>. 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  $\rightarrow$  Network' on the WEB interface to put it within your company's LAN. Refer to <u>3.12.1 Network</u> 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 <u>3.4.3</u> <u>PCM</u> for detailed instructions about PCM Settings.

**Note:** You shall restart the service to validate the settings in this step. Refer to <u>3.12.17 Restart</u> 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 <u>3.5 SS7 Settings</u>.

The configuration interfaces related to SS7-ISUP include: <u>SS7</u>, <u>ISUP</u> and <u>SS7 Server</u>.

On your initial use of the SMG digital gateway, you may adopt the default values of the configuration items on the <u>SS7</u> and <u>ISUP</u> interfaces. Note that the <u>SS7 Server</u> 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 <u>3.12.17 Restart</u> 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 <u>3.5 SS7 Settings</u>.

The configuration interfaces related to SS7-TUP include: <u>SS7, TUP</u> and <u>SS7 Server</u>.

On your initial use of the SMG digital gateway, you may adopt the default value of the configuration items on the <u>SS7</u> and <u>TUP</u> interfaces. Note that the <u>SS7 Server</u> 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 <u>3.12.17 Restart</u> for detailed instructions.

#### • ISDN User Side/Network Side:

The configuration interface related to ISDN User Side/Network Side is <u>ISDN</u>. 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 <u>3.12.17 Restart</u> for detailed instructions.

#### • SS1:

The configuration interface related to SS1 is <u>SS1</u>. 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 <u>3.12.17 Restart</u> 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  $\rightarrow$  PSTN Status'. Refer to <u>3.2.2 PSTN Status</u> 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 $\rightarrow$ 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 → <u>SIP Trunk</u>' 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 → <u>SIP Trunk Group</u>' 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 → <u>PCM Trunk</u> <u>Group</u>' 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  $\rightarrow$  <u>IP $\rightarrow$ PSTN</u>' 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 $\rightarrow$ IP

Step 1: Configure the called party numbers which are received from PSTN and will be processed by the gateway. Refer to 'Advanced Settings → <u>Number-receiving Rule</u>' 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 → <u>SIP Trunk</u>' 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 → <u>SIP Trunk Group</u>' 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 → <u>PCM Trunk</u> <u>Group</u>' 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 → <u>PSTN→IP</u>' 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.

ndows Secur	ity 🚾
The server 20	1.123.112.211 at SMG requires a username and password.
	is server is requesting that your username and password be secure manner (basic authentication without a secure
	User name
1/2	Password
	Remember my credentials
	OK Cancel

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  $\rightarrow$  Change Password' on the WEB interface. For detailed instructions, refer to <u>3.12.15 Change Password</u>.

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

eration Info 🔗					
		System Info			
System Info					
PSTN Status	LAN 1				
SS7 Server	MAC Address	0E:12:9A:10:32:01	055 055 055 0	004 400 440 054	
Call Count	IP Address DNS Server	201.123.112.211 0.0.0.0	255.255.255.0	201.123.112.254	
	Receive Packets	All:4825196	Error:0	Drop:0	
VolP 🛛 😸	Transmit Packets	All:318190	Error:0	Drop:0	
	Current Speed	Receive: 1.9 KB/s	Transmit:0 B/s	Drop.0	
PCM 🛛 👻	Work Mode	100Mb/s Full Duplex	Transmico bis		
SS7 🛛 📚					
	LAN 2				
Fax 🛛 😸	MAC Address	0E:12:9A:10:32:02			
	IP Address	192.168.0.101	111.111.111.111	192.168.0.254	
Route	DNS Server	0.0.0.0			
Number Filter 🛛 🗧	Receive Packets	All:0	Error:0	Drop:0	
	Transmit Packets	AII:0	Error:0	Drop:0	
Num Manipulate 🛛 👻	Current Speed	Receive:0 B/s	Transmit:0 B/s		
System Tools 🛛 🗧	Work Mode	10Mb/s Half Duplex			
	Runtime	1d 17h 14m 34s			
	Operating Mode	Master Server			
	Current Version				
	Serial Number	T10000086(4)			
	WEB	1.6.0_2015022715			
	Gateway	1.6.0_2015022715			
	Uboot	2.0.6_201407			
	Kernel	#206 Fri Dec 26 17:20:4	42 CST 2014		
	Firmware	18			

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

LAN 1			
MAC Address	0E:12:9A:10:32:01		
IP Address	201.123.112.211	255.255.255.0	201.123.112.254
DNS Server	0.0.00		
Receive Packets	All:4825196	Error:0	Drop:0
Transmit Packets	All:318190	Error:0	Drop:0
Current Speed	Receive:1.9 KB/s	Transmit:0 B/s	
Work Mode	100Mb/s Full Duplex		
LAN 2			
MAC Address	0E:12:9A:10:32:02		
IP Address	192.168.0.101	111.111.111.111	192.168.0.254
DNS Server	0.0.00		
Receive Packets	All:0	Error:0	Drop:0
Transmit Packets	All:0	Error:0	Drop:0
Current Speed	Receive:0 B/s	Transmit:0 B/s	
Work Mode	10Mb/s Half Duplex		
Runtime	1d 17h 14m 34s		
Operating Mode	Master Server		
Current Version			
Serial Number	T10000086(4)		
WEB	1.6.0_2015022715		
Gateway	1.6.0_2015022715		
Uboot	2.0.6_201407		
Kernel	#206 Fri Dec 26 17:20:4	2 CST 2014	
Firmware	18		

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.

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



#### 3.2.2 PSTN Status

Sync & S	Sync & Signaling Status Frame Sync									Signaling								Faulty						Unused							
	Color																														
											-																				
pice Path Status	Idle	Ring	ing	Wai	it Ans	wer	Dia	aling	Та	lking	Pe	nding	Wait	Mess	age	Lo	cal B	lock	F	temo	te Blo	ck	Bot	th Blo	ck	Circuit Reset				Jnusa	bl
Icon		6	3		0			•	(	->>	1	2		6			•		1							R				53	
												122					0				0									90	
Statistics	29	0			0			0		1		0		0			0		2		0		0	0			0		10	90	
Statistics	29	0			0			0		1		o eway1:	201.1		12.2	11:80					0			0			0		, ,,,	90	
Statistics Time Slot No.	29	0	2	3	0	5	6	0	8	9					<b>12.2</b> 15	<b>11:80</b>		18	19	20	21	22	23	24	25	26	27	28	29	30	
				-	4	5	6	7	-	9	Gat 10	eway1:	2 13	123.1			)	_	_	20	21	-	-	24	-		27	-	-		
Time Slot No.		1	2		4		6	7		9	Gat 10	eway1:	2 13	123.1	15		17	_	_	20	21	-	-	24	-		27	-		30	(
Time Slot No. Port1		1	2		4		6	7		9 3 3 5 7 7 7	Gat 10	eway1: 11 12	2 13	123.1 14	15		17			20	21			24			27			30	

Note: If the icons display abnormally, please clear the cache and refresh this page. 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.

ltem	Description
Port	Serial number of the E1/T1 port on the device.
Time Slot No.	PCM time slot number in the port.
	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
0(= (=	direction, calling party number and called party number.
State	• For Time Slot 0, the channel state indicates the synchronization status of
	E1/T1.
	State Color Description



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Frame Sync		Frame synchronization normal. The synchronization status is 0x0.
Faulty		Configuration errors or hardware failure. 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: bit0=1: basic frame synchronization loss bit1=1: duration of the basic frame synchronization loss exceeds 100ms bit2=1: CAS re-synchronization bit3=1: CRC re-synchronization bit4=1: remote alarm indication bit5=1: signal alarm indication bit6=1: all-ones alarm signal of time slot 16 bit7=1: signal loss bit9=1: MF alarm from the remote end bit10=1: open circuit
For the sign		Other bits: reserved, all remain 0
		e slot, the channel states include:
<b>State</b> Signaling	Color	Description For SS7, this state indicates 'SS7 in service'. For ISDN, this state indicates 'multiple frames established' or 'timer recovery'.
		For SS1, this state indicates 'time slot synchronization normal'. Configuration errors or hardware failure.
Faulty		For SS7, this state indicates 'SS7 out of service', 'initial alignment', 'aligned ready', 'aligned not ready' or 'processor outage'. For ISDN, this state indicates 'TEI unassigned', 'assign awaiting TEI', 'establish awaiting TEI', 'TEI assigned', 'awaiting establishment 'or 'awaiting release'. 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 othe	er channe	els, the channel states include:
State	lcon	Description
Unusable	<b>か</b>	The channel is unavailable.



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	Circuit Reset	R	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	٥	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	<b>C</b>	The channel is dialing.
	Wait Message	6	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.

888				Se	arch		Cle	ear		Rec	ord	_][s	top F	lecor	₫																		
Sync & Sign	aling	Statu	IS	_	_		_	Fran	ne Sy	nc		_		_		Sig	naling			_		_	Fai	ulty	_		_	_	U	nuse	d	_	
Co	lor																																
/oice Path Status	Idl	e	Ring	ing	Wa	it Ans	wer	Di	aling	Та	alking	) P	endi	ng	Wait	Mes	sage	l	_ocal E	Block	F	Remo	te Blo	ock	Во	th Blo	ock	Cir	cuit R	eset	U	Inusa	able
Icon	6	)	6			0			•											R			<i>•</i>										
Statistics	29	)	0			0			0		1		0			0			0				0			0			0			90	
												Ga	tew	ay1:	201.1	123.'	112.2	11:8	30														
Time Slot No.		0	1	2	3	4	5	6	7	8	9	10	11	12	2 13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
Port1																																	6
Port2			6	<i>f</i>	6	6	6	6	¢,	6	6	Calkir				6					6	6	6	<i>7</i>	6	6	5	6	6	6	6	6	6
Port3			6	<i>7</i>	6	6	6				6	Direc	tion									6	6	6	6								
Port4			6	<i>7</i>	6	6	5	6	6	6		Caller Caller			158	6	<i>i</i>			6	6	6	6	<b></b>	6	6		6	<b></b>	6	6	6	ý
								No	te: If th	he ico					ally, ple	ase	clear t	he ca	ache a	nd ref	resh	this p	age.										

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*.

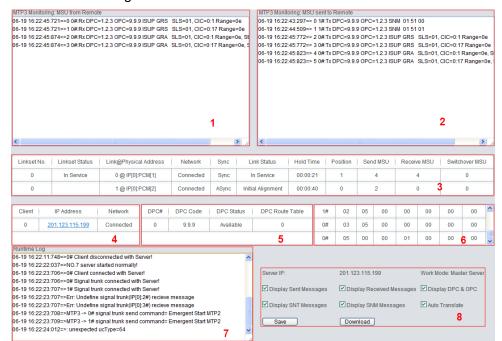


Figure 3-8 SS7 Server Info Interface

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 <u>SS7</u> interface.
Work Mode	Work mode of the SS7 server which includes three modes: Master Server, Slave Server and Client.
Display Sent	If this item is ticked, the transmit message list will display the message sent to the
Messages	remote end.
Display Received	If this item is ticked, the receive message list will display the message received from
Messages	the remote end.
Display DPC & OPC	If this item is ticked, the receive/transmit message list will display DPC and OPC.
Display SNT	If this item is ticked, the receive/transmit message list will display the SNT
Messages	messages.
Display SNM	If this item is ticked, the receive/transmit message list will display the SNM
Messages	messages.



	If this item is ticked, the received/sent messages displayed on this interface will be
	translated automatically in the following format:
	Date Time Total number Signaling link number# SIO Content
Auto Translate	For the TUP messages, SIO is just 'TUP' (0x84), followed by the message content.
Auto Translate	It is usually in the following format:
	Title code CIC=PCM:TS Message body
	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.

ltem	Description
Linkset No.	Linkset number.
	Working state of the linkset, including In service and Out of service. A signaling
Linkset Status	linkset will go into the state In service as long as one link in it is at the state of In
	service.
Link@Physical	Signaling link number and its physical position. For example, '0 @ IP[0]:PCM[0]'
Address	means the physical position of Link 0 in this gateway is the E1 with the local PCM
Address	numbered 0 on Client 0.
	Whether the signaling link is registered to the gateway, including two states:
Network	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.
Sync	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 <i>In service</i> .
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
Switchover MSU	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
IP Address	interface of the client.
Network	Whether the client has been successfully connected to the gateway, including two
Network	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.				
	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 Status	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	СНО
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: servio	e 5: L3 sends a			
on	command to stop			
6:				
processor	6: signaling			
failure	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

	PSTN Call Statistics														
Trunk No.	Signa	aling Type	Current N	Number of IP->PSTN Total Number of IP->		otal Number of IP->PSTN Connection Rate of IP->PSTN C		Current Number of PSTN->IP		ſN->IP	Total Number of PSTN->IF		I->IP │ C		
0	SS	7-ISUP		0		0			-		0			0	
1	SS	87-TUP		0		0			-		0			0	
2	SS	87-TUP		0		0			-		0			0	
3	SS	37-TUP		0		0			-		0		0		
Total				0		0					0			0	
<															>
Release Ca	ause	Normal ca	II clearing	Cancelled by calling		atistics of		STN Release wer from user	Cause Route failed	Resource un	available	Unalloc	ated numbe	er   Call re	ejected
Amoun		0	-	0	g party 00	0			0	0	avanabic	onalioe	0		0
Percenta	ge									-				-	-
<															>
					St	atistics o	on PSTN	->IP Release	Cause						
Release F	Reason	Norm	al call clear	ing Cancelled	by calling pa	arty U	Jser busy	No answe	er from user	Route failed	Resou	rce unava	ailable	Call failed	Others
Numb	ber		0 0		0		0		0	0		0		0	0
Percent	tage		· · · · · · · · · · · · · · · · · · ·												
	Refresh														

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 Tuna	The signaling protocol applied on the digital trunk, including: ISDN User Side, ISDN					
Signaling Type	Network Side, SS7-TUP, SS7-ISUP, and SS1.					
Current Number of	The number of surrent cells from ID to DOTN					
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 $\rightarrow$ PSTN calls to total IP $\rightarrow$ PSTN calls.					
Current Number of	The number of current calls from PSTN to IP.					
PSTN → 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 $\rightarrow$ IP calls to total PSTN $\rightarrow$ 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	Total number of the calls which are cancelled by the calling party.				
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	Total number of the calls which fail as the called party does not pick up the call in a				
user	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.				
	Total number of the calls which fail as the called party number is normal but				
Normal unspecified	unspecified.				
0-11 (-11-1	Total number of the calls which fail as the called party number does not conform to				
Call failed	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

SIP Settings	
SIP Address	LAN 1: 201.123.112.211
SIP Signaling Port	5060
Send 183 Message	Enable
Obtain CallerID from	Username of From Field
Obtain CalleeID from	'Request' Field 💌
Obtain Redirecting Number/Original CalleelD from Diversion Field	Enable
Stun Traversal	✓ Enable
STUN Server Address	127.0.0.1
SIP Transport Protocol	UDP
SIP Encryption	✓ Enable
Encrpytion Criterion	V0S1.1
Кеу	5416
RTP Encryption	Enable
RTP Self-adaption	Enable
UDP Header Checksum	Enable
Rport	Enable
DSCP	✓ Enable
Voice Media	46
Signal Control	26
Maximum Wait Answer Time (s)	60
Maximum Wait RTP Time (s)	0
Maximum Wait PSTN Resource Time(ms)	5000
Save	

#### 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 <u>3.12.17 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-11.

ltem	Description
SIP Address	IP address for SIP signaling, using LAN 1 by default.
SID Down	Monitoring port of SIP signaling. Range of value: 1024~65535, with the default
SIP Port	value of 5060.
183 Message	Sets whether to send the 183 message instead of 180 to respond to the ringing tone
Behavior	when the SIP end serves as the called party. By default this feature is enabled.
	There are two optional ways to obtain the calling party number: from Username of
Obtain CallerID from	"From" Field or from Displayname of "From" Field. The default value is from
	Username of "From" Field.
Obtain CalleelD	There are two optional ways to obtain the called party number: from "To" Field or
from	from "Request" Field. The default value is from "Request" Field.
Obtain Redirecting	
Number/Original	Sets whether to enable the feature of obtaining the Redirecting Number/Original
CalleeID from	CalleeID from Diversion Field. By default, the feature is disabled.
Diversion Field	
STUN Traversal	Sets whether to enable the STUN server for NAT traversal. By default the STUN
	server is disabled.
STUN Server	Address of the server for STUN traversal.
Address	
SIP Transport	There are two modes UDP and TCP available for running the SIP protocol. The
Protocol	default value is <i>UDP</i> .
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 <i>disabled</i> .
Encryption Criterion	The criterion used to encrypt the SIP signal. At present only VOS1.1 is supported.
Кеу	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.
	When this feature is enabled, the RTP reception address or port carried by the
RTP Self-adaption	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	When this feature is enabled, the gateway will automatically calculate the check
Checksum	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 <i>disabled.</i>
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 $0 \sim 63$ , with the default value of <i>46</i> .
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 <i>60</i> , 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 <i>0</i> , 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 <i>5000</i> .

#### 3.3.2 SIP Trunk

	SIP Trunk						
Check	Index	Remote Address	Remote Port	Outgoing Voice Resource	Incoming Voice Resource	Modify	
	1	201.123.112.227	5060	0	2		
Check All = Uncheck All = Inverse = Delete = Clear All Add 1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 v 1 Pages Total							

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.

SIP Trui	nk
Index:	0
Remote Address:	
Remote Port:	5060
Outgoing Voice Resource:	128
Incoming Voice Resource:	128
Save	Close

Figure 3-13 Add New SIP Trunk

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



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

SIP Trunk						
Index:	1					
Remote Address:	201.123.112.227					
Remote Port:	5060					
Outgoing Voice Resource:	0					
Incoming Voice Resource:	2					
Save	Close					

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

Operation Info	*
🛱 VolP	*
SIP	
SIP Trunk	
SIP Register	
SIP Account	
SIP Trunk Group	
Media	



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.

SIP Reg	ister
Index:	0
SIP Trunk No.:	0
Username:	
Password:	
Register Address:	
Register Port:	5060
Domain Name:	
Register Expires (s):	3600
IMS Network:	Yes 💌
Externally Bound Address	
Externally Bound Port:	
Authentication Username	:
Save	Close

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.				
	When the gateway initiates a call to SIP, this item corresponds to the username of				
Username	SIP; when the gateway initiates a call to PSTN, this item corresponds to the				
	displayed CallerID.				
	Registration password of the gateway. To register the gateway to the SIP server,				
Password	both configuration items <b>Username</b> and <b>Password</b> 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.		
	Validity period of the SIP registry. Once the registry is overdue, the gateway should		
Register Expires	be registered again. Range of value: 10~3600, calculated by s, with the default		
	value of 3600.		
	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.		
IMS Network	Only when this feature is <i>enabled</i> will these items <i>Externally Bound Address</i> ,		
	Externally Bound Port and Authentication Username be shown.		
Externally Bound	Externally bound IP address for registration.		
Address			
Externally Bound	Externally bound port for registration.		
Port			
Authentication	Authentication username for registration.		
Username			

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

	SIP Register									
Check	Index	SIP Trunk No.	Username	Register Adress	Register Port	Domain Name	Register Expires (s)	Register Status	IMS Network	Externally Bound Address
	0	0	100	201.123.115.26	5060		3600	Failed	No	
<										
Check A	<b>II</b> 🗄 🛛	Uncheck All 🗄	Inverse	🗄 Delete 🗄	Clear All					Add New
1 Items To	I Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 🗸 1 Pages Total									

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.



SIP Register					
Index:	0				
SIP Trunk No.:	0				
Username:	100				
Password:	•••				
Register Address:	201.123.115.26				
Register Port:	5060				
Domain Name:					
Register Expires (s):	3600				
IMS Network:	Yes 💌				
Externally Bound Address:					
Externally Bound Port:					
Authentication Username:					
Save	Close				

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

SIP Account									
Check	Index	SIP Trunk No.	Username	Authentication Username	Register Expires (s)	Register Status	Description	Modify	
	0	0	111		3600	Failed	deault		
Check All = Uncheck All = Inverse = Delete = Clear All Add									
1 ltems Total 20 ltems/Page 1/1 First Previous Next Last Go to Page 1 💌 1 Pages Total									

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.

SIP Account			
Index:	1		
SIP Trunk No.:	0		
Username:			
Password:			
Register Expires (s):	3600		
Authentication Usernam	e:		
Description:	default		
Save	Close		

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.
11000000000	The registration username of the SIP account. Once the SIP account is successfully
Username	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
Password	trunk, both configuration items <b>Username</b> and <b>Password</b> should be filled in.
	The validity period of the SIP account registry. Once the registry is overdue, the SIP
Register Expires	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 of a port, used to register the port to the SIP server when
Authentication	IMS network is enabled.
Username	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.

SIP Account		
Index:	0	
SIP Trunk No.:	0	
Username:	111	
Password:	•••	
Register Expires (s):	3600	
Authentication Username		
Description:	deault	
Save	Close	

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

SIP Trunk Group					
Check	Index	SIP Trunks	SIP Trunk Select Mode	Description	Modify
	0	0	Increase	default	
Check All = Uncheck All = Inverse = Delete = Clear All Add Trew Add Trew Add Trew Add Trew Add Trew					

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.



Add New			
Index:	1		
Description:	default		
SIP Trunk Select Mode:	Increase		
SIP Trunks:			
0	□1		
Save	Cancel		

Figure 3-23 Add New SIP Trunk Group

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

Item	Description			
Index	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.			
Index				
Description	More information about	ut each SIP trunk group.		
	When the SIP trunk g	When the SIP trunk group receives a call, it will choose a SIP trunk based on the		
	select mode set by the	his configuration item to ring. The optional values and their		
	corresponding meanir	ngs are described in the table below.		
	Option	Description		
	,	Search for an idle SIP trunk in the ascending order of the		
	Increase	SIP trunk number, starting from the minimum.		
SIP Trunk Select		Search for an idle SIP trunk in the descending order of		
Mode	Decrease	the SIP trunk number, starting from the maximum.		
		Provided SIP Trunk N is the available SIP trunk found last		
	Cyclic Increase	time. Search for an idle SIP trunk in the ascending order		
		of the SIP trunk number, starting from SIP Trunk N+1.		
		Provided SIP Trunk N is the available SIP trunk found last		
	Cyclic Decrease	time. Search for an idle SIP trunk in the descending order		
		of the SIP trunk number, starting from SIP Trunk N-1.		
	The SIP trunks in the	The SIP trunks in the SIP trunk group. If the checkbox before a SIP trunk is grey, it		
SIP Trunks	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.			

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.



Modify SIP Trunk Group		
Index:	0	
Description:	default	
SIP Trunk Select Mode:	Cyclic Decrease	
SIP Trunks:		
0	1	
Save	Cancel	

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

Media Parameters				
	DTMF Trans	smit Mode	RFC2833	~
	RFC2833 P	ayload	101	
	RTP Port Ra	ange	6000,10000	
	Silence Sup	pression	Disable	×
	Noise Redu	iction	Enable	×
	JitterMode		Static Mode	×
	JitterBuffer(i	ms)	100	
	JitterUnderr	unLead(ms)	100	
	JitterOverru	nLead(ms)	50	
	Voice Gain (	Output from IP(dB)	0	
CODEC Priorit	y			
Check	Priority 1 2 3 4 5 6 7	CODEC G711A G711U G729 G722 iLBC G711A G711A G711A	Packing Time(ms) 20 • 20 • 20 • 20 • 20 • 20 • 20 • 20 •	Bit Rate (kbs) 64 64 8 64 15.2 64 64 64 64 64 64 64 64
		Save	Reset	

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 <u>3.12.17 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-25.

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



	Supported RTP port range for the IP end to establish a call conversation, with the
RTP Port Range	lower limit of 2000 and the upper limit of 60000 and the difference between larger
	than 512. The default value is 6000-10000.
	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
Silence	throughout an IP conversation. The optional values are <i>Enable</i> and <i>Disable</i> , with
Suppression	the default value of <i>Disable</i> .
	Note: When G723 is selected as CODEC, this configuration setting will turn to
	Enable automatically.
	Once this feature is enabled, the volume of the noise accompanied with the line will
Noise Reduction	be reduced automatically. The default setting is <i>Enable</i> .
	Sets the working mode of JitterBuffer. The optional values are Static Mode and
JitterMode	Adaptive Mode, with the default value of Static Mode.
	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
JitterBuffer	as an increased voice delay, while a smaller JitterBuffer means a lower jitter
onterBuiler	processing capability but as well as a decreased voice delay. Range of value:
	0~280, calculated by ms, with the default value of 100.
	-
	Sets the initial delay applied to received packets upon accepting packets later than the expected value set in littler Ruffer Item. Prage of value: 0, 280, calculated by
JitterUnderrunLead	the expected value set in JitterBuffer Item. Rnage of value: 0~280, calculated by
	ms, with the default value of 100,
	Note: Only when JitterMode is to <i>Static Mode</i> will this item be shown.
	Sets the beforehand time inserted if receiving packets is ahead of time (the time of
JitterOverrunLead	receiving is earlier than 300 minus the value set in JitterBuffer). Rnage of value:
	0~280, calculated by ms, with the default value of <i>50</i> ,
	Note: Only when JitterMode is to <i>Static Mode</i> will this item be shown.
	Sets the minimum delay that can be set by the adaptive jitter function. It can not be
JitterMin	larger than the value set in JitterBuffer. Rnage of value: 0~280, calculated by ms,
	with the default value of 80.
	<b>Note:</b> Only when JitterMode is to <i>Adaptive Mode</i> will this item be shown.
	Sets the rate of the delay that can be reduced under the adaptive mode. It defines
JitterDecreaseRatio	the maximum percentage of silence that can be removed if reducing the delay.
	Rnage of value: 0~100, with the default value of <i>50</i> ,
	Note: Only when JitterMode is to Adaptive Mode will this item be shown.
	Sets the maximum delay can be increased during one silence period. Rnage of
JitterIncreaseMax	value: 0~280, calculated by ms, with the default value of 30,
	Note: Only when JitterMode is to Adaptive Mode will this item be shown.
Voice Gain Output	Adjusts the voice gain of call from IP to the remote end. The value must be a
from IP	multiple of 3. Range of value: -24~24, calculated by dB, with the default value of 0.
	Supported CODECs and their corresponding priority for the IP end to establish a
CODEC Priority	call conversation. The table below explains the sub-items:



Priority	, , ,	DEC in an SIP conversation. The
	smaller the value is, the high	er the priority will be.
CODEC	Seven optional CODECs	are supported: G711A, G711U,
	G729AB, G723, G722, AMR	and <i>iLBC</i> .
Packing Time	Time interval for packing an	RTP packet, calculated by ms.
Bit Rate	The number of thousand bits	e (excluding the packet header) that
Dil Rale	are conveyed per second.	
By default, all o	of the seven CODECs are sup	ported and ordered G711A, G711L
G729AB, G723,	G722, AMR and iLBC by priori	ty from high to low.
The packing tim	e and bit rate supported by diffe	erent CODECs are listed in the tabl
	e and bit rate supported by diffe	
		lt values.
below. Those va	alues in bold face are the defaul	lt values. Bit Rate (kbps)
below. Those va	lues in bold face are the defaul Packing Time (ms)	lt values. Bit Rate (kbps) 64
below. Those va COEDC G711A	alues in bold face are the defaul <b>Packing Time (ms)</b> 5 / 10 / <b>20</b> / 30 / 40 / 50 / 60	lt values. Bit Rate (kbps) 64
below. Those va COEDC G711A G711U	alues in bold face are the defaul Packing Time (ms) 5 / 10 / 20 / 30 / 40 / 50 / 60 5 / 10 / 20 / 30 / 40 / 50 / 60 20	lt values. Bit Rate (kbps) 64 64
below. Those va COEDC G711A G711U G729AB	alues in bold face are the defaul Packing Time (ms) 5 / 10 / 20 / 30 / 40 / 50 / 60 5 / 10 / 20 / 30 / 40 / 50 / 60 20	lt values. Bit Rate (kbps) 64 64 8
below. Those va COEDC G711A G711U G729AB G723 G722	Alues in bold face are the defaul Packing Time (ms) 5 / 10 / 20 / 30 / 40 / 50 / 60 5 / 10 / 20 / 30 / 40 / 50 / 60 20 30 / 60 / 90 5 / 10 / 20 / 30 / 40	It values. Bit Rate (kbps) 64 64 8 5.3 / 6.3
below. Those va COEDC G711A G711U G729AB G723	alues in bold face are the defaul Packing Time (ms) 5 / 10 / 20 / 30 / 40 / 50 / 60 5 / 10 / 20 / 30 / 40 / 50 / 60 20 30 / 60 / 90	t values. Bit Rate (kbps) 64 64 64 8 5.3 / 6.3 64
below. Those va COEDC G711A G711U G729AB G723 G722	Alues in bold face are the defaul Packing Time (ms) 5 / 10 / 20 / 30 / 40 / 50 / 60 5 / 10 / 20 / 30 / 40 / 50 / 60 20 30 / 60 / 90 5 / 10 / 20 / 30 / 40	lt values. Bit Rate (kbps) 64 64 64 5.3 / 6.3 64 4.75 / 5.15 / 5.90 / 6.70 / 7.40 /
below. Those va COEDC G711A G711U G729AB G723 G722	Alues in bold face are the defaul Packing Time (ms) 5 / 10 / 20 / 30 / 40 / 50 / 60 5 / 10 / 20 / 30 / 40 / 50 / 60 20 30 / 60 / 90 5 / 10 / 20 / 30 / 40 20 / 40 / 60 / 80 / 100	Bit Rate (kbps) 64 64 8 5.3 / 6.3 64 4.75 / 5.15 / 5.90 / 6.70 / 7.40 / 7.95 / 10.20 / 12.20

# 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.

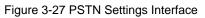


Figure 3-26 PCM Settings



# 3.4.1 PSTN

PSTN Configuration				
Interface	E1			
Encoding Format	A-law			
Echo Canceller	✓ Enable			
Ringback Tone Provided by E1	Enable			
Ringback Tone Provided by IP	Enable			
Ringback Tone Volume(dB)	-25			
Voice Gain Output from PSTN(dB)	0			
Hot Back-up for E1	Enable			
Gateway IP for Hot Back-up				
Save				



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,
Interface	only E1/T1 is supported.
Encoding Format	Sets the voice data encoding format for the voice channels on the digital trunk. The
Encoding Format	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
Echo Canceller	digital trunk. By default, this feature is enabled and the effect can reach 128ms.
Ringback Tone	Sets whether to enable the E1 end to provide the ringback tone, with the default
Provided by E1	value of <i>disable</i> .
Ringback Tone	Sets whether to enable the IP end to provide the ringback tone, with the default
Provided by IP	value of <i>disable</i> .
Ringback Tone	Sets the volume of the ringback tone. Range of value: -35~-2, calculated by dB, with
Volume	the default value of -25.
Voice Gain Output	Adjusts the voice gain of call from PSTN to the remote end. The value must be a
from PSTN	multiple of 3. Range of value: -24~24, calculated by dB, with the default value of 0.
Hot Pook up for 51	Sets whether to enable the feature of hot back-up for E1, with the default value of
Hot Back-up for E1	disable.
Gateway IP for Hot	Set the ID of the asternov for the bet healt up for E1
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 <u>3.12.17 Restart</u> for detailed instructions.

# 3.4.2 Circuit Maintenance

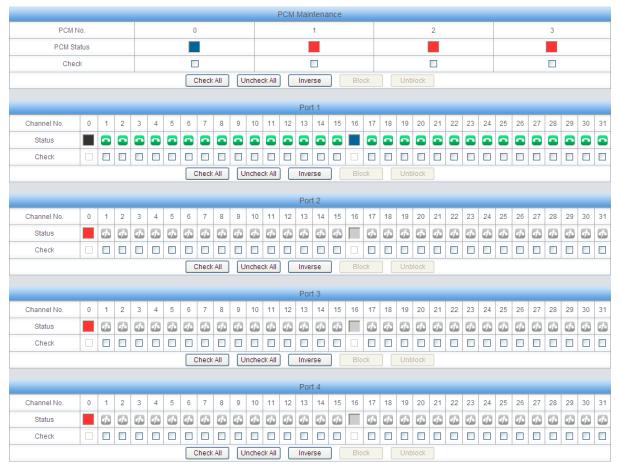


Figure 3-28 Circuit Maintenance Interface

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.

ltem	Description
PCM No.	The number of the PCM, numbered from 0. This item is not configurable.



	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.
	Note: 1, Changing the interface type from E1 to T1 will forbid those non-ISDN
Signaling Protocol	signaling modes in E1. And in such case, the gateway will by default set this
Signaling Frotocor	item to ISDN User Side.
	2, For SMG3008, a single gateway can be configured with two different
	signaling modes simultaneously.
	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
CIUCK	Slave.
	Sets the time slot used for signaling transmission on the digital trunk. If the
Signaling Time Slat	configuration item <i>Signaling Protocol</i> is set to <i>ISDN</i> and <i>SS1</i> , the signaling time
Signaling Time Slot	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 Trees	Indicates whether the PCM is used as a signaling link or a voice link. If no time slot
Signaling Link Type	is used to transmit signaling, the PCM is a voice link.
Connection Line	Physical connection line type.
	ets a certain amount of channels which starts from a certain TS to process the
Incoming Call Start	incoming calls and others on the PCM to process outgoing calls. This is valid only
TS, Amount	when the configuration item <b>Signaling Protocol</b> is set to SS1.
CRC-4	Sets whether to enable the CRC-4 verification feature. By default, this feature is
UKU-4	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.



Modify PCM Info				
PCM No.:	1			
Signaling Protocol:	SS7-ISUP			
Signaling Time Slot:	16 💌			
Clock:	Line-synchronizatio 🗸			
Connection Line:	Twisted Pair Cable 🔽			
Use 'Signaling Time	e Sloť for Signaling			
Enable CRC-4				
Apply to All PCMs				
Save	Close			

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 <b>Signaling Time Slot</b> is used for signaling transmission. You can see this item only when the configuration item <b>Signaling Protocol</b> is set to SS7-TUP or SS7-ISUP.
Apply to All PCMs	Check this item to apply the above settings (excluding <i>Clock</i> ) 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

	Operation Info	*
	😤 VolP	*
(	🚺 РСМ	*
I	PSTN	
	Circuit Maintenand	e
_	PCM	
	PCM Trunk	
	PCM Trunk Group	
	Num-Receiving R	
	Reception Timeou	it
	Number Attribution	ı

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 3-30 Modify PCM



#### Figure 3-33.

	PCM	Trunk	
Index:	0		~
PCM NO.:	0		~
Including	Ts:	Check All	
TS[0]	TS[1]	TS[2]	TS[3]
TS[4]	TS[5]	TS[6]	TS[7]
TS[8]	TS[9]	TS[10]	TS[11]
TS[12]	TS[13]	TS[14]	TS[15]
TS[16]	TS[17]	TS[18]	TS[19]
TS[20]	TS[21]	TS[22]	TS[23]
TS[24]	TS[25]	TS[26]	TS[27]
TS[28]	TS[29]	TS[30]	TS[31]
	Save	Clo	se

Figure 3-32 Add PCM Trunk Interface

	PCM Trunk	Batch Add				
Including F	CM:	Check All				
PCM[0]	PCM[1]	PCM[2]	PCM[3]			
PCM[4]	PCM[5]	PCM[6]	PCM[7]			
PCM[8]	PCM[9]	PCM[10]	PCM[11]			
PCM[12]	PCM[13]	PCM[14]	PCM[15]			
Close						

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.

	PCM Trunks					
Check	Index	PCM NO.	Including Ts	Modify		
	0	0	1,2,3,4,5,6,7,8,9,10,11,12,13,14,15,17,18,19,20,21,22,23,24,25,26,27,28,29,30,31			
	1	1	1,2,3,4,5,6,7,8,9,10,11,12,13,14,15,17,18,19,20,21,22,23,24,25,26,27,28,29,30,31			
	2	2	5			
	Check All 🗄 Uncheck All 🗄 Inverse 🗄 Delete 🗄 Clear All Add New					
3 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 💌 1 Pages Total						

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.

PCM Trunk				
Index:		0		
PCM NO.:	0		~	
Including	Ts:	Check All		
TS[0]	✓ TS[1]	✓ TS[2]	✓ TS[3]	
✓ TS[4]	✓ TS[5]	✓ TS[6]	✓ TS[7]	
▼TS[8]	✓ TS[9]	TS[10]	TS[11]	
TS[12]	✓ TS[13]	TS[14]	TS[15]	
TS[16]	TS[17]	TS[18]	TS[19]	
TS[20]	✓ TS[21]	✓ TS[22]	TS[23]	
TS[24]	✓ TS[25]	✓ TS[26]	TS[27]	
✓ TS[28]	✓ TS[29]	▼ TS[30]	✓ TS[31]	
Save Close				

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

PCM Trunk Group					
Check	Index	PCM Trunks	PCM Trunk Select Mode	Description	Modify
	0	0	Increase	default	
Check All E Uncheck All E Inverse E Delete E Clear All Add New					

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.

PCM Trunk Group				
Index:		1	*	
Description:		default		
PCM Trunk Se	PCM Trunk Select Mode: Increase			
PCM Trunks:				
PCM Trunks:	Ш Спеск /	All		
0	1	2		
Save Close				

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.			



	When the PCM trunk g	roup 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.		
	Option	Description	
	1	Search for an idle PCM trunk in the ascending order of	
	Increase	the PCM number, starting from the minimum.	
PCM Trunk Select	Destroop	Search for an idle PCM trunk in the descending order of	
Mode	Decrease	the PCM number, starting from the maximum.	
		Provided PCM Trunk N is the available PCM trunk found	
	Cyclic Increase	last time. Search for an idle PCM trunk in the ascending	
		order of the PCM number, starting from PCM Trunk N+1.	
		Provided PCM Trunk N is the available PCM trunk found	
	Cyclic Decrease	last time. Search for an idle PCM trunk in the descending	
		order of the PCM number, starting from PCM trunk N-1.	
	The PCM trunks in the PCM trunk group. If the checkbox before a PCM		
PCM Trunks	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.		

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.

PCM Trunk Group					
Index:		0			
Description:					
Description.	Description: default				
PCM Trunk Se	PCM Trunk Select Mode: Increase				
PCM Trunks:	PCM Trunks: Check All				
<b>⊻</b> 0	1	2			
Save Close					

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.

Number-receiving Rule						
Check	Index	Number-receiving Rule	Description	Modify		
	99		example			
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						
Rule: "X"(lowercase) indicates a random number, "," indicates multiple random characters,						

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.

Number-Receiving Rule				
Index:	98	~		
Number-receiving Rule:				
Description:	default			
Save	Close			

Figure 3-40 Add New Number-Receiving Rule

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

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



	Up to 99 num	nber-receiving rules o	can be configured in the gateway, and the
	maximum leng	th of each number-rece	eiving rule is 127 characters. See below for the
	•		nber-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 $\sim$ 9.	
	"X"		A string of 'x's represents several random ole, 'xxx' denotes 3 random numbers.
	<i>u m</i>	'.' indicates a rando	om 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 ar	ny one of the numbers 1, 2, 3, 6, 8.
		'-' is used only in '[	]' between two numbers to indicates any
	"" "	number between the	
	     		e numbers or number ranges, representing
	"" "	alternatives.	
	By default, the	re is only one rule confi	igured on the gateway. The table below lists 20
	-	-	and understanding. See below for detailed
Number-Receiving	information.		
_		!	· · · · · · · · · · · · · · · · · · ·
Rule	Priority	Dialing Rule	Description
Rule	Priority 99	Dialing Rule	
Rule		Dialing Rule 01[3,5,8]xxxxxxxx.	Any number in any length. Any 12-digit number starting with 013,
Rule	99 98	• 01[3,5,8]xxxxxxxx.	Any number in any length. Any 12-digit number starting with 013, 015 or 018
Rule	99 98 97	•	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010
Rule	99 98	• 01[3,5,8]xxxxxxxx.	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02
Rule	99 98 97 96	01[3,5,8]xxxxxxxx 010xxxxxxx 02xxxxxxxx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04,
Rule	99 98 97	01[3,5,8]xxxxxxxxx 010xxxxxxx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09
Rule	99 98 97 96	01[3,5,8]xxxxxxxx 010xxxxxxx 02xxxxxxxx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04,
Rule	99 98 97 96 95	01[3,5,8]xxxxxxxxx 010xxxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxxx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117,
Rule	99 98 97 96 95 94 93	01[3,5,8]xxxxxxxx 010xxxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxxx 120 11[0,2-9]	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117, 118 or 119
Rule	99 98 97 96 95 94 93 92	01[3,5,8]xxxxxxxx     010xxxxxxx     02xxxxxxx     0[3-9]xxxxxxxx     120     11[0,2-9]     111xx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 Any 5-digit number starting with 111
Rule	99 98 97 96 95 94 93 92 91	01[3,5,8]xxxxxxxx 010xxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxxx 120 11[0,2-9] 111xx 123xx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 Any 5-digit number starting with 111 Any 5-digit number starting with 123
Rule	99 98 97 96 95 94 93 93 92 91 90	01[3,5,8]xxxxxxxx 010xxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxxx 120 11[0,2-9] 111xx 123xx 95xxx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 Any 5-digit number starting with 111 Any 5-digit number starting with 123 Any 5-digit number starting with 95
Rule	99 98 97 96 95 94 93 92 91	01[3,5,8]xxxxxxxx 010xxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxxx 120 11[0,2-9] 111xx 123xx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 Any 5-digit number starting with 111 Any 5-digit number starting with 123 Any 5-digit number starting with 95 Any 5-digit number starting with 100
Rule	99 98 97 96 95 94 93 93 92 91 90	01[3,5,8]xxxxxxxx 010xxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxxx 120 11[0,2-9] 111xx 123xx 95xxx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 Any 5-digit number starting with 111 Any 5-digit number starting with 123 Any 5-digit number starting with 123 Any 5-digit number starting with 100 Any 11-digit number starting with 13, 14,
Rule	99 98 97 96 95 94 93 92 91 90 89	01[3,5,8]xxxxxxxx 010xxxxxxx 02xxxxxxx 0[3-9]xxxxxxxx 120 11[0,2-9] 111xx 123xx 95xxx 100xx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 Any 5-digit number starting with 111 Any 5-digit number starting with 123 Any 5-digit number starting with 95 Any 5-digit number starting with 100 Any 11-digit number starting with 13, 14, 15 or 18
Rule	99 98 97 96 95 94 93 92 91 90 89	01[3,5,8]xxxxxxxx 010xxxxxxx 02xxxxxxx 0[3-9]xxxxxxxx 120 11[0,2-9] 111xx 123xx 95xxx 100xx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 Any 5-digit number starting with 111 Any 5-digit number starting with 123 Any 5-digit number starting with 123 Any 5-digit number starting with 100 Any 11-digit number starting with 13, 14, 15 or 18 Any 8-digit number starting with 2, 3, 5, 6
Rule	99 98 97 96 95 94 93 92 91 90 89 88	O1[3,5,8]xxxxxxxx     O10xxxxxxx     O2xxxxxxx     O[3-9]xxxxxxxx     120     11[0,2-9]     111xx     123xx     95xxx     100xx     1[3-5,8]xxxxxxxx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 Any 5-digit number starting with 111 Any 5-digit number starting with 123 Any 5-digit number starting with 123 Any 5-digit number starting with 95 Any 5-digit number starting with 95 Any 5-digit number starting with 13, 14, 15 or 18 Any 8-digit number starting with 2, 3, 5, 6 or 7
Rule	99 98 97 96 95 94 93 92 91 90 89 88	O1[3,5,8]xxxxxxxx     O10xxxxxxx     O2xxxxxxx     O[3-9]xxxxxxxx     120     11[0,2-9]     111xx     123xx     95xxx     100xx     1[3-5,8]xxxxxxxx	Any number in any length. Any 12-digit number starting with 013, 015 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 Number 120 Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 Any 5-digit number starting with 111 Any 5-digit number starting with 123 Any 5-digit number starting with 123 Any 5-digit number starting with 100 Any 11-digit number starting with 13, 14, 15 or 18 Any 8-digit number starting with 2, 3, 5, 6



	85	80[1-0]vvvvv	Any 8-digit number starting with 801, 802,		
	85	80[1-9]xxxxx	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	400xxxxxx	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 le		
Description	empty.	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.

Number-Receiving Rule				
Index:	99			
Number-receiving Rule:	· .			
Description:	default			
Save	Close			

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**

Number-receiving Timeout Info		
Inter Digit Timeout (s)	Description	Modify
6	example	

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.

ltem	Description
	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
Inter Digit Timeout	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.
Decemination	More information about the configuration item Inter Digit Timeout, such as the
Description	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*.

Number-Receiving Timeout		
lates Digit Time out (a)		
Inter Digit Timeout (s):	6	
Description:	example	
Save	Close	

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

Local Area Code Settings			
Remove 0 or area code in case the calleeID for an IP- >PSTN call is a local number Local Area Code 057			
Save Reset Number Attribution			
Export Number Attribution Info Import Number Attribution Info	Export		
Note: Please set the local zip code at you	r first use.		

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 CalleelD Pool, Redirecting Number Pool* and *SS7 Server*. See Figure 3-45.





Figure 3-45 SS7 Settings

#### 3.5.1 SS7

SS7 Settings			
As Client Only (SS7 server disabled)	⊡Yes		
Master IP	127.0.0.1		
Slave IP			
Local IP Address	127.0.0.1		
Dual Gateway	Enable		
Save	Reset		

Figure 3-46 SS7 Settings Interface

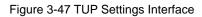
See Figure 3-46 for the SS7 settings interface where you can configure the general SS7 parameters. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to <u>3.12.17 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-46.

ltem	Description		
As Client Only	Sets whether the gateway serves as Client only or not. If it is set to No (default), the		
As Client Only	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,		
Master IP	which indicates that there is only one SS7 server available.		
Clave ID	Sets the IP address of the slave SS7 server. Only when the item <b>Dual Gateway</b> is		
Slave IP	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.		
	If this feature is enabled, two SS7 servers are used at the same time in the system.		
Dual Gateway	The configuration items <i>Master IP</i> and <i>Slave IP</i> are respectively used to set the IP		
	addresses of the master and slave servers.		



# 3.5.2 TUP

TUP Settings	
Send GRM Group Message Using All-0 Field	✓Enable
Send ST Signal with CallerID in Outgoing Call	Enable
Default Caller Parameter	Valid national number
Set Caller Parameter in case of Original CalleeID	Enable
Caller Parameter (with Original CalleeID)	Valid national number
Default Original Callee Parameter	Valid national number
Maximum Wait Answer Time (s)	60
Minimum Length of the CalleeID of an Incoming Call	1
Save	



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 <u>3.12.17</u> <u>Restart</u> 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 mean           The optional values are: Local subscriber number, Spare national number         The optional 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. <i>Caller Parameter ( with Original CalleeID)</i> . By default this configuration item is disabled.		



	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
Caller Parameter (with	message when the IP end carries the original CalleeID in a call from IP to PSTN.
Original CalleelD)	The optional values are: Local subscriber number, Spare national number, Valid
	national number and International number, with the default value of Valid national
	number.
	Sets the address indicator in the original called party address field of the IAI
Default Original Callee	message. The optional values are: Local subscriber number, Spare national
Parameter	number, Valid national number and International number, with the default value of
	Valid national number.
Maximum Wait Answer	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
Time (s)	by the channel automatically. The default value is 60, calculated by s.
Minimum Longth of the	Sets the minimum length of the CalleeID under the fixed-length mode. The value
Minimum Length of the	range is $1 \le n \le 40$ . Provided it is set to n, that is, the local end has received all the n
CalleelD of an Incoming	digits of the called party number of the incoming call, the number reception will be
Call	regarded as finished.

## 3.5.3 TUP Number Parameter

Calling Party Number Parameter					
Check	No.	Prefix	Parameter	Set Parameter if Original CalleeID Available	Modify
	0	11	Valid national number	No	
				·	
Delete		Clear All			Add New

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.



Calling Party Number Parameter		
No.:	1	
Prefix:		
Parameter:	Valid national number 💌	
Set Parameter if	Original CalleeID Available	
Save	Close	

Figure 3-49 Add New Calling Party Number Parameter

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

ltem	Description		
No.	The corresponding number for a calling party number parameter, which starts from		
NO.	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 CalleelD	Set whether to enable the feature of setting this parameter only if the original		
Available	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.

Calling Part	y Number Parameter
No.:	0
Prefix:	11
Parameter:	Valid national number 💌
Set Parameter if	Original CalleelD Available
Save	Close

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

ISUP Settings	
Calling Party's Category	Ordinary calling subscriber
Default Caller Parameter	Subscriber number 🔍 0x1301
Default Callee Parameter	National number 🔍 0x1003
Set Caller/Callee Parameter in case of Original CalleeID	Enable
Send Generic Number	Enable
Transmission Medium Requirement	Speech
Obtain First Called Party Number from	Original calleelD/Redirecting nur 🗸
Auto Reply INF upon Reception of Remote INR	✓Enable
Reset Circuit upon Service Start before Entering Idle State	✓Enable
Information on First Two Bytes of Redirecting Number	0x1001
Maximum Wait Answer Time (s)	180
Minimum Length of the CalleeID of an Incoming Call	1
Forward Call Indicator	0x0040
Nature of Connection Indicator	0x00
User Service Information	0x80,0x90,0xa3
Optional Forward Call Indicator	0x00 Enable
Save Reset	

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 <u>3.12.17</u> <u>Restart</u> 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.



	Sets the calling party number parameter field in the IAM message. The optional
Default Caller Parameter	values are: Subscriber number, National number, and International number, with the
	default value of Subscriber number.
	Sets the called party number parameter field in the IAM message. The optional
Default Callee Parameter	values are: Subscriber number, National number, and International number, with the
	default value of National number.
Set Caller/Callee	Once this feature is enabled, if the IP end carries the original CalleeID in a call from
Parameter in case of	IP to PSTN, you shall set separate values for the caller and callee parameters in the
Original CalleelD	IAM message, i.e. Caller Parameter (with Original CalleeID) and Callee
	Parameter (with Original CalleeID). By default this configuration item is disabled.
	This item is valid only when Set Caller/Callee Parameter in case of Original
Caller Parameter (with	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.
Original CalleeID)	The optional values are: Subscriber number, National number, and International
	number, with the default value of Subscriber number.
	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
Callee Parameter (with	message when the IP end carries the original CalleeID in a call from IP to PSTN.
Original CalleeID)	The optional values are: Subscriber number, National number, and International
	number, with the default value of National number.
	Sets the generic number parameter in IAM message, with the default value of
Send Generic Number	disabled.
	Sets the transmission medium requirement parameter in the IAM message. The
	optional values are: Speech, 64 kb/s unrestricted, 3.1khz audio, Alternative: speech
Transmission Medium	(service 2)/ 64kbit/s unrestricted (service 1) (Spare), Alternative: 64kbit/s
Requirement	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.
	Sets where the first called party number is obtained from. The optional values are:
Obtain First Called Party	Only original CalleeID and Original CalleeID/ Redirecting number, with the default
Number from	value of Only original CalleeID.
Auto Reply INF upon	If this feature is enabled, once the INR message is received from the remote PBX in
Reception of Remote	an outgoing call, the driver will automatically reply it with the INF message. By
INR	default this feature is enabled.
Reset Circuit upon	
Service Start before	If this feature is enabled, the circuit will send a circuit reset message before entering
Entering Idle State	the idle state after the ISUP service is enabled. By default this feature is enabled.
Information on First Two	Sets the first two bytes of the redirecting number in the IAM message, including the
Bytes of Redirecting	nature of address indicator, numbering plan indicator and address presentation
Number	restricted indicator, with the default value of 0x1001.
Maximum Wait Answer	Sets the maximum time to wait for the answer from the called party of an outgoing
Time (s)	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 CalleelD of an Incoming Call	Sets the minimum length of the CalleeID under the fixed-length mode. The value range is $1 \le n \le 40$ . 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	Sets the nature of connection indicator in the IAM message, with the default value of		
Indicator	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

Calling Party Number Parameter								Called Pa	arty Number Parameter			
Check	No.	Prefix	Parameter	Set Parameter if Original CalleeID Available	Modify		Check	No.	Prefix	Parameter	Set Parameter if Original CalleeID Available	Modify
	0	1	0x1003	No				0	6	0x1004	No	
De	lete		Clear All	Add	New		De	leite		Clear All	Add	New
				Note: You shall restart	the service	e to	validate th	e settin	as on this	s page!		

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.



Calling Party Number Parameter						
No.:	1					
Prefix:						
Parameter:	Subscriber number 🛛 👻					
	0x1301					
Set Parameter	if Original CalleeID Available					
Save	Close					

Figure 3-53 Add New Calling Party Number Parameter

Called Party Number Parameter						
No.:	1					
Prefix:						
Parameter:	Subscriber number					
	0x1001					
Set Parameter if (	Original CalleeID Available					
Save	Close					

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			
No.	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 CalleelD	Set whether to enable the feature of setting this parameter only if the original			
Available	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.

Calling Party Number Parameter							
No.:	0						
Prefix:	1						
Parameter:	National number						
	0x1003						
Set Parameter if Original CalleeID Available							
Save	Close						

Figure 3-55 Modify Calling Party Number Parameter

Called Party Number Parameter						
No.:	0					
Prefix:	6					
Parameter:	International number 💌					
	0x1004					
Set Parameter if Original CalleeID Available						
Save	Close					

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

	Original CalleelD Pool						
Check	No.	CallerID Prefix	CalleeID Prefix	Original CalleelD Range	Modify		
	0	*	*	100000000000010000000000000001			
				·			
Delete III	Clear All				Add New		

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.

Origir	nal CalleelD
No.:	1
CallerID Prefix:	*
CalleeID Prefix:	*
Original CalleeID Ra	nge:
Save	Close

Figure 3-58 Add New Original CallerID

The table below explains the items shown in above figures.

ltem	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 CalleelD	The range of the original CalleeID in the Original CalleeID Pool. It must be filled in
Range	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.

Original Ca	lleelD
No.:	0
CallerID Prefix:	*
CalleeID Prefix:	*
Original CalleelD Range:	100000000000000
-	100000000000001
Save	Close

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

				Redirecting Number Pool		
Check	No.	CallerID Prefix	CalleeID Prefix	Redirection Information	Redirecting Number Range	Modify
	0	*	*	0x0321	12345678901234561234567890123457	
Delete	🗄 Clea	ar All				Add New

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.

Redirecting Number					
No.:	1				
CallerID Prefix:	*				
CalleeID Prefix:	×				
Redirection Information:	0x0321				
Redirecting Number Ran	ge:				
Save	Close				

Figure 3-61 Add New Redirecting Number

The table below explains the items shown in above figures.

ltem	Description	
No	The corresponding number for an added redirecting number. The value range is	
No.	0~99.	
Coller D. Drofix	A string of numbers at the beginning of a calling party number, which can be	
CallerID Prefix	numbers or "*" (indicating any string).	
	A string of numbers at the beginning of a called party number, which can be	
CalleelD Prefix	numbers or "*" (indicating any string).	
	Sets the redirection information field in the IAM message. The parameter type of the	
Redirecting	redirection information field is 0x13, which contains 2 bytes. By default, it is set to	
Information	0x0321, i.e. call forwarding on no answer. Refer to the ISUP protocol standard for	
	the detailed description of each byte.	
Redirecting Number	The range of the redirecting number in the Redirecting Number Pool. It must be filled	
Range	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.



Redirec	cting Number
No.:	0
CallerID Prefix:	*
CalleeID Prefix:	*
Redirection Informati	ion: 0x0321
Redirecting Number	Range: 12345678901234
	12345678901234
Save	Close

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.

Server 1	IP:	201.123	.112.211	Serv	rer 2 IP:			Sigr	aling P	oint Code	Standard	24	~	Su	bservice Code:	Nationa	al network	~
Send SL	TM:	Enable			Save					1								
		Client			c	ignaling Link			Sig	inaling I	inkeet		ו		DPC Se	ttings		
Check	No.	IP Address	Modify	Check	No.	Physical Address	Modify	Signaling Linkset				DPC Settings Check No. STP SP Code Linkset Mod					Modi	
	0	201.123.112.211			0	IP[0]:PCM[0]			0	0	1.2.3			0	Associated Mode	9.9.9	0	
																		1.05
		:	2									3					4	
	] ]				] (				]					]	(			
Delete	-	Clear All	Add New	Delete		Clear All	Add New	Delete	-	Clear A	All	Add New	Delete	-	Clear All			Add Ne
UP_CIC	Route	ISUP_CIC Route																
CI	neck	No.	DPC		(	CIC_PCM	(	CIC Range			Loca	PCM		SP	Code	STP	Mo	dify
																5		
Delete		Clear All															Ad	d New

# 3.5.8 SS7 Server

Note: You shall restart the service to validate the settings on this page!

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.

ltem	Description			
Comren 4 ID	Sets the IP address for the master SS7 server. If only one server is used in the			
Server 1 IP	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	The value of this item varies on the PBX model. The optional values are 14 and 24,			
Code Standard	with the default value of 24. The China SS7 uses 24.			
	Sets the SS7 subservice code. The optional values are: International network,			
Subservice Code	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			
Send SLIW	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.

Client IP Settings				
No.:	1			
IP Address:				
Save	Close			

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					
No.	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.

Signaling Link				
No.:		1		
	-			
Client:	0		×	
PCM:				
Save		Close		

Figure 3-65 Add New Signaling Link

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

ltem	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 number. This configuration item together with PCM determines the physical	
Client	address of the E1 interface of the signaling link. Each physical address maps a	
	signaling link.	
РСМ	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.



Signaling Linkset				
No.:	0			
Link:				
Link 0				
OPC:	Decimal 💌			
	1.2.3			
Save	Close			

Figure 3-66 Add New Signaling Linkset

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

Item		Description		
No	The unique index of each signaling linkset, which is mainly used in the configuration			
No.	of DPC to correspond to the signaling linkset, numbered from 0.			
Link	The signaling links in the linkset. If the checkbox before a link is grey, it indicates that the link has been occupied.			
OPC	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:			
		14 bit	24 bit	
	Decimal (a.b.c)	a, c: 0~7, b: 0~255	a, b, c: 0~255	
	Hexadecimal	a, c: 3-digit hexadecimal number,	a, b, c: hexadecimal	
	(abc)	b: 8-digit hexadecimal number	number inbetween 00~ff	

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.



	DPC
No.:	0
Associated Mode	OQuasi-Associated Mode
SP Code:	Decimal 💌
	9.9.9
Linkset:	Link 0
Save	Close

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						
<i>N</i> 0.	TUP_CIC Route or ISUP_CIC Route to correspond to the DPC, numbered from 0.						
	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						
Associated Mode/	network is predetermined and fixed within a certain period of time, is called						
Quasi-associated	Quasi-associated Mode. These two concepts are vividly illustrated below.						
Mode	SP						
	(a) Associated Mode (b) Quasi-associated Mode						
SP Code	Signaling point code of the DPC, usually allocated by the central office.						
	Sets the first STP (signaling transport point) the signaling message reaches during						
STP	the transmission under the quasi-associated mode. Only when you select the						
	quasi-associated mode can this item be seen and configured.						
	The linkset which is used to transmit signaling messages. For the associated mode,						
Linkset	this item sets the signaling linksets between the OPC and the DPC. For the						
LIIKSEL	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.

	CIC Route
No.:	0
DPC:	0
CIC_PCM:	0
CIC_PCM Range:	0-31
Client:	0
PCM:	
Consecutively add	1 CIC_PCM for this DPC
Save	Close

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 <b>PCM</b> determines the local PCM in the CIC routing rule.
РСМ	PCM number on the client.



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

CIC Route							
No.:	1						
DPC:	0						
CIC_PCM:	1						
CIC Range:	Default V 32-63						
Client:	User-define						
PCM:	2						
Save Close							

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 <u>3.12.17 Restart</u> 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.



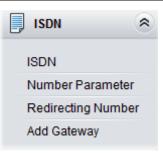


Figure 3-70 ISDN Settings

### 3.6.1 ISDN

					ISDN Settings				
Link No. User Side: 0 User Side: 1 User Side: 2 User Side: 3	Logical PCM No. 1 2 3 4	TEI 0 0 0	Ch Identificat Number Number Number Number	ion V	Default Callee Type National number (0XA1) National number (0XA1) National number (0XA1) National number (0XA1)	> > >	Default Caller Type National number (0X21) National number (0X21) National number (0X21) National number (0X21)	CODEC           •           A-Law           •           A-Law           •           A-Law	CRC Check V V V
Link No. User Side: 0 User Side: 1 User Side: 2 User Side: 3	Logical PCM No. 1 2 3 4	Se	t Caller/Callee Type in ca	se of Re		Nation Nation	(with Redirecting Num) al number v al number v al number v al number v	Caller Type (with Red National numbe National numbe National numbe	r 🗸
	Enter Auto Maximum Wai	Alert State Alert State t Time for (	e V upon Reception of 'CALL upon Reception of 'PRO Called Party's Pick up(s) CalleelD of an Incoming (	GRESS'I 60					
	ISDN Network Side	inel Identif I Channel	ication Message		Sen Wait Co	d the 'C nfirm Ti	ime (T310) (s) <mark>15 and 15 and </mark>		
				Save	Reset				

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 <u>3.12.17 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-71.

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



	Sets the type of number and numbering scheme for the called party numbers in the
Default Callee Type	SETUP message during the outgoing call. The optional values are: <i>National number</i> ,
	International number, Network number, Subscriber number and Unknown, with the
	default value of National number.
	Sets the type of number and numbering scheme for the calling party numbers in the
Default Caller Type	SETUP message during the outgoing call. The optional values are: National number,
Delaun Ganer Type	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
CODEC	u-Law, with the default value of A-Law.
	Sets whether to enable the feature of CRC check for the digital trunk at ISDN user
CRC Check	side or network side. By default this feature is enabled.
	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
Set Caller/Callee Type in	numbering scheme for the calling and called party numbers in the SETUP message,
case of Redirecting Num	i.e. Callee Type (with Redirecting Num) and Caller Type (with Redirecting
	<i>Num</i> ). By default this configuration item is disabled.
	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
Callee Type (with	numbers in the SETUP message when the IP end carries the redirecting number in
Redirecting Num)	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.
	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
Caller Type (with	numbers in the SETUP message when the IP end carries the redirecting number in
Redirecting Num)	a call from IP to PSTN. The optional values are: National number, International
<b>J</b>	number, Network number, Subscriber number and Unknown, with the default value
	of National number.
	Sets the 'Transfer Capability' filed in the signaling message. The optional values are
Transfer Capability	Voice and 3.1k Audio, with the default value of Voice.
Enter Auto Alert State	
upon Reception of	If this item is checked, the system will go into the state of auto alert when it receives
'CALL PROCEEDING'	the 02 (CALL PROCEEDING) message and the progress indicator turns to be 8 or
Message	1. By default this item is disabled.
Enter Auto Alert State	If this item is checked, the system will go into the state of auto alert when it receives
upon Reception of	the 03 (PROGRESS) message and the progress indicator turns to be 8 or 1. By
'PROGRESS' Message	default this item is disabled.
	The maximum time waiting for the called party to pick up the call after the channel
Maximum Wait Time for	state turns to 'WaitAnswer' during an outgoing call. The default value is 60,
Called Party's Pick up	calculated by s.
	Calculated by 5.



Minimum Length of the CalleelD of an Incoming Call	Sets the minimum length of the CalleeID under the fixed-length mode. The value range is $1 \le n \le 40$ . 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

	Calling Party Number Type									Ca	alled Party Number Type	
Check	No.	Prefix	Туре	Set Parameter if Redirecting Number Available	Modify		Check	No.	Prefix	Туре	Set Parameter if Redirecting Number Available	Modify
	0	6	0x11	Yes				0	1	0x91	Yes	
						ŀ						
						ŀ						
						ŀ						
						ŀ						
						ŀ						
						L						
Del	leie		Clea	r All Add	New		Del	ele		Clea	r All Add	New

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

Redirecting Number Pool								
Check	No.	CallerID Prefix	CalleeID Prefix	Redirecting Number Range	Modify			
	0	*	*	1001010019				
	Clear All				Add New			

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).

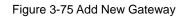
#### 3.6.4 Add Gateway

Gateway			
Check	No.	IP Address	Modify
	0	201.123.112.203	
Delete 🗄 Clear All			Add New

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**  $\rightarrow$  **PSTN Status**. See Figure 3-75 for the gateway adding interface.

Add Gateway	
No.:	1
IP:	
Sa	Close



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	
IF	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.

Add Gateway	
No.:	0
IP:	201.123.112.203
Sa	Close

Figure 3-76 Modify Gateway Information



# 3.7 SS1 Settings

SS1 Settings		
Country	CHINA	
C/D Value	3	
ABCD Duration Timeout (ms)	0	
Max MFC Waiting Time (ms)	10	
CalleeID Length for Incoming Calls	3	
Advanced Setting for Incoming Calls		
KB Setting Timeout (s)	3	
KD Wait Time (ms)	60	
Delay Time before Ringing State (ms)	0	
Advanced Setting for Outgoing Calls		
ACK Wait Timeout (s)	60	
Calling Party's Category (KA Signal)	1	
KB Wait Timeout (s)	60	
Originating Service Type (KD Signal)	3	
Save	Reset	

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 <u>3.12.17 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-77.

ltem	Description
Country	Sets the country to use SS1, with the default value of CHINA.
	Sets the value of CD in the ABCD signaling codes sent by the local end to the remote
0/0 //-/	PBX. The high 6 bits should be set to 0, being reserved; the low 2 bits are C/D
C/D Value	signaling codes, Bit1 (Signaling Code C) and Bit0 (Signaling Code D), both with the
	default value of 1.
	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
ABCD Duration	of 0. Only when the on-line ABCD signaling codes vary and the new value keeps for
Timeout	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.



Sets the maximum waiting time, i.e. the timer T2 for the SS1 state machine, calculated	
by second, with the default value of 10.	
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.	
Sets the maximum time to wait for the application to configure the KB signal, calculated	
by second, with the default value of 3.	
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.	
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.	
Sets the value of the timer T5, calculated by second, with the default value of 60.	
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 <i>1 (ordinary/regular)</i> .	
Sets the maximum time to wait for the KB signal from the remote PBX, calculated by	
second, with the default value of 60.	
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

Fax Parameter	'S
Fax Mode	T.38
T38 Version	0
T38 Negotiation	Initiate Negotiation as Fax Re
Maximum Fax Rate (bps)	9600
Fax Train Mode	transferredTCF
Error Correction Mode	t38UDPRedundancy
T.30 ECM	✓Enable
Min Duration of CNG(ms)	425
Min Duration of CED(ms)	2600
Save	Reset

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 <u>3.12.17 Restart</u> for detailed instructions. The table below explains the configuration items in Figure 3-79.

Item	Description	
	The real-time IP fax mode. The optional values are T.38, Pass-through and Disable,	
Fax Mode	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 value of 0.		
		T29 Negatistics
T38 Negotiation	Fax Sender and Initiate Negotiation as Fax Receiver.	
Marian Far Data	Sets the maximum faxing rate for both receiving and transmitting. Range of value:	
Maximum Fax Rate	14400, 9600 and 4800, calculated by bps, with the default value of 9600.	
<b>F</b> . <b>F</b> . <b>C</b> . <b>M</b> . <b>C</b>	Sets the train mode for T.38 fax. The optional values are transferredTCF and	
Fax Train Mode	localTCF, with the default value of transferredTCF.	
Error Correction	Sets the error correction mode for T.38 fax. The optional values are	
	t38UDPRedundancy (Redundancy Error Correction) and t38UDPFEC (Forward	
Mode	Error Correction), with the default value of t38UDPRedundancy.	
T 00 5	Sets whether to enable the T.30 error correction mode. By default this feature is	
T.30 Ecm	enabled.	



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

Fax Parameters	
Fax Mode	Pass-through
Pass-through Payload	102
Min Duration of CNG(ms)	425
Min Duration of CED(ms)	2600
Save	set

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

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

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

## 3.9 Route Settings

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

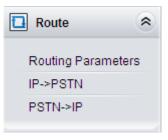


Figure 3-81 Route Settings



#### 3.9.1 Routing Parameters

Route Settings					
IP->PSTN	Route before Number Manipulate				
PSTN->IP	Route before Number Manipulate				
	Save				

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  $IP \rightarrow PSTN$  and  $PSTN \rightarrow IP$  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

	Routing Rules							
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
	63	SIP Trunk Group [0] 333[1,3]:444[6,9]		*	none	PCM Trunk Group [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 3-83 IP→PSTN Routing Rule Configuration Interface

See Figure 3-83 for the IP $\rightarrow$ 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 $\rightarrow$ PSTN routing rule adding interface.

IP->PS1	IN Routing Rule
Index:	62 💌
Call Initiator:	SIP Trunk Group [0]
CallerId Prefix:	*
CalleeID Prefix:	*
Call Destination:	PCM Trunk Group [0]
Number Filter:	none
Description:	default
Save	Close



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

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

Item	Description					
	The unique index of each routing rule, which denotes its priority. A routing rule with					
Index	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					
Call Initiator	SIP trunk group or SIP Trunk Group [ANY] which indicates any SIP trunk group.					
A string of numbers at the beginning of the calling/called party number. Th						
	can be set to a specific string or "*" which indicates any string. These two					
CallerID Prefix,	configuration items together with <i>Call Initiator</i> can specify the calls which apply to a					
CalleeID Prefix	routing rule.					
	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 rule which will be applicable to this route. It is set in <b>Number Filter</b> .					
Number Filter	See <u>3.10.4 Filtering Rule</u> 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 $\rightarrow$ PSTN routing rule modification interface. The configuration items on this interface are the same as those on the *Add New Routing Rule (IP\rightarrowPSTN)* interface. Note that the item *Index* cannot be modified.

IP->PST	N Routing Rule
Index:	63
Call Initiator:	SIP Trunk Group [0]
Callerld Prefix:	333[1,3]:444[6,9]
CalleeID Prefix:	*
Call Destination:	PCM Trunk Group [0]
Number Filter:	none
Description:	default
Save	Close

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

	Routing Rules							
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
	63	3 PCM Trunk Group [0] *		*	none	SIP Trunk Group [0]	default	
Check All	Check All 🗄 Uncheck All 🗄 Inverse 🗄 Delete 🗄 Clear All 🖉 Add New							
1 ltems Total 20 ltems/Page 1/1 First Previous Next Last Go to Page 1 v 1 Pages Total								

Figure 3-86 PSTN→IP Routing Rule Configuration Interface

See Figure 3-86 for the PSTN $\rightarrow$ 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 $\rightarrow$ IP routing rule adding interface.

PSTN->I	PSTN->IP Routing Rule				
Index:	62 💌				
Call Initiator:	PCM Trunk Group [0]				
CallerID Prefix:	*				
CalleeID Prefix:	*				
Call Destination:	SIP Trunk Group [0]				
Number Filter:	none				
Description:	default				
Save	Close				

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

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

Item	Description					
	The unique index of each routing rule, which denotes its priority. A routing rule with					
Index	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.					
	PCM trunk group from which the call is initiated. This item can be set to a specific					
Call Initiator	PCM trunk group or PCM Trunk Group [ANY] which indicates any PCM trunk group.					



	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				
CallerID Prefix,	configuration items together with <i>Call Initiator</i> can specify the calls which apply to a				
CalleeID Prefix	routing rule.				
	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 rule which will be applicable to this route. It is set in <b>Number Filter</b> .				
Number Filter	See <u>3.10.4 Filtering Rule</u> 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 $\rightarrow$ IP routing rule modification interface. The configuration items on this interface are the same as those on the **Add New Routing Rule (PSTN \rightarrowIP)** interface. Note that the item **Index** cannot be modified.

PSTN->I	PSTN->IP Routing Rule				
Index:	63				
Call Initiator:	PCM Trunk Group [0]				
CallerID Prefix:	*				
CalleeID Prefix:	*				
Call Destination:	SIP Trunk Group [0]				
Number Filter:	none				
Description:	default				
Save	Close				

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

		CallerID Whitelist					CalleeID Whitelist		
Check	Group No.	No. in Group	CallerID	Modify	Check	Group No.	No. in Group	CalleeID	Modify
	0	0	100			0	0	101	
	) (						1		
Delete	Clear	All		Add New	Delete	Clear	All		Add New
			Note: You shal	I restart the ser	vice to validate the s	ettings on this page!			

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.

CallerIDs in Whitelist				
Group No.:	0			
No. in Group:	1			
CallerID:				
Save	Close			

Figure 3-91 Add New CallerIDs in Whitelist Interface



Calle	elDs in Wł	nitelist	
Group:	0	*	
No. in Group:		1	
CalleelD:			
Save		Close	

Figure 3-92 Add New CalleeIDs in Whitelist Interface

The table below explains the items shown in above figures.

ltem	Description
Group	The corresponding Group ID for CallerIDs/CalleeIDs in the whitelist. The value
Group	range is 0~7.
No. in Group	The corresponding No. for different CallerIDs/CalleeIDs in a same group. It is
No. in Group	allowed to set up to 100 numbers in one group.
	CallerID in the whitelist, which must be filled in with numbers or "*" (indicating any
CallerID	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 in the whitelist, which must be filled in with numbers or "*" (indicating any
CalleelD	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.

CallerIDs in Whitelist				
Group No.:	0			
No. in Group:	0			
CallerID:	100			
Save	Close			

Figure 3-93 Modify CallerIDs in Whitelist



Calle	eelDs in Whitelist
Group:	0
No. in Group:	0
CalleeID:	101
Save	Close

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.

		CallerID Blacklist					CalleeID Blacklist		
Check	Group No.	No. in Group	CallerID	Modify	Check	Group No.	No. in Group	CalleelD	Modify
	0	0	78			0	0	111	
	0	1	111						

### 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

		Number Pool		
Check	Group No.	No. in Group	Number Range	Modify
	1	0	200201	
Delete	Clear All			Add New

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.

Nu	imber Pool
Group:	0
No. in Group:	0
Number Range:	
-	
Save	Close

Figure 3-97 Add New Number Pool

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

ltem	Description
Group	The corresponding Group ID for numbers in the number pool. The value range is
Group	0~15.
	The corresponding No. for different numbers in a same group. It supports up to 100
No. in Group	number s in one group.
Number Denne	The range of the numbers in a number Pool. It must be filled in with numbers and
Number Range	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.

N	umber Pool
Group:	1 👻
No. in Group:	0
Number Panae:	000
Number Range:	200
	201
Save	Close

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 T	2 Items Total 15 Items/Page 1/1 First Previous Next Last Go to Page 1 V 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.



Filtering Rule
No.: 12
CallerID Whitelist: none
CalleeID Whitelist: none
CallerID Blacklist: none
CalleelD Blacklist: none
CallerID Pool in Whitelist: none
CallerID Pool in Blacklist: none
CalleeID Pool in Whitelist: none
CalleeID Pool in Blacklist: none
Original CalleeID Pool in Whitelist: none 💌
Original CalleelD Pool in Blacklist: none 💌
Description: default
Close

Figure 3-100 Add New Filtering Rule

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

ltem	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	Select a Group No. which is set in the whitelist from the number pool as the CallerID
Whitelist	pool in whitelist.
CallerID Pool in	Select a Group No. which is set in the blacklist from the number pool as the CallerID
Blacklist	pool in blacklist.
CalleeID Pool in	Select a Group No. which is set in the whitelist from the number pool as the CalleeID
Whitelist	pool in whitelist.



CalleeID Pool in	Select a Group No. which is set in the blacklist from the number pool as the CalleeID
Blacklist	pool in blacklist.
Original CalleelD	Select a Group No. which is set in the whitelist from the number pool as the original
Pool in Whitelist	CalleeID pool in whitelist.
Original CalleelD	Select a Group No. which is set in the blacklist from the number pool as the original
Pool in Blacklist	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.

Filtering Rule
No.: 0
CallerID Whitelist: 0
CalleelD Whitelist: none
CallerID Blacklist: none
CalleelD Blacklist: none
CallerID Pool in Whitelist: 0
CallerID Pool in Blacklist: none
CalleelD Pool in Whitelist: none
CalleelD Pool in Blacklist: none
Original CalleeID Pool in Whitelist: 0
Original CalleeID Pool in Blacklist: none 💌
Description: default
Close



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 $\rightarrow$ PSTN CallerID, IP $\rightarrow$ PSTN CalleeID, IP $\rightarrow$ PSTN Original CalleeID, PSTN $\rightarrow$ IP CallerID, PSTN $\rightarrow$ IP CalleeID, PSTN $\rightarrow$ IP Original CalleeID and CallerID Pool. See Figure 3-102.

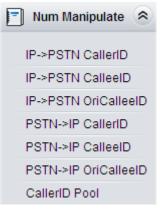


Figure 3-102 Number Manipulation

#### 3.11.1 IP to PSTN CallerID

	Number Manipulation Rules											
Chedk	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 A	Check All Uncheck All Inverse Delete Clear All Add New											
1 Items To	tal 20 Ite	ms/Page 1/1 First Pre	evious Next Last	Go to Page 1 💌	1 Pages Total							

Figure 3-103 IP→PSTN CallerID Manipulation Interface

See Figure 3-103 for the IP $\rightarrow$ 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 $\rightarrow$ PSTN CallerID manipulation rule adding interface.



IP->PSTN Ca	allerID Ma	nipulation	
Index:	62		*
Call Initiator:	SIP Trur	nk Group [0]	~
CallerID Prefix:		*	
CalleeID Prefix:		*	
With Original Calleel	D:	No	~
Stripped Digits from	Left:	0	
Stripped Digits from	Right:	0	
Reserved Digits from	n Right:	0	
Prefix to Add:			
Suffix to Add:			
Description:		default	
Save		Close	

Figure 3-104 Add IP→PSTN CallerID Manipulation Rule

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

Item	Description
	The unique index of each number manipulation rule, which denotes its priority. A
Index	number manipulation rule with a smaller index value has a higher priority. If a call
Index	matches several number manipulation rules, it will be processed according to the
	one with the highest priority.
	SIP trunk group from where the call is initiated. This item can be set to a specific
Call Initiator	SIP trunk group or SIP Trunk Group[ANY] which indicates any SIP trunk group.
	A string of numbers at the beginning of the calling/called party number. This item
CallerID Prefix,	can be set to a specific string or "*" which indicates any string. These two
CalleeID Prefix	configuration items together with Call Initiator and With Original CalleeID can
	specify the calls which apply to a number manipulation rule.
With Original	If this item is set to Yes, it indicates that the number manipulation rule is only
With Original	applicable to the calls with original CalleeID/redirecting number. The default value is
CalleelD	No.



Stripped Digits from	The amount of digits to be deleted from the left end of the number. If the value of
Left	this item exceeds the length of the current number, the whole number will be
Len	deleted.
Stripped Digits from	The amount of digits to be deleted from the right end of the number. If the value of
Stripped Digits from Right	this item exceeds the length of the current number, the whole number will be
Right	deleted.
Decominant Distite	The amount of digits to be reserved from the right end of the number. Only when the
Reserved Digits	value of this item is less than the length of the current number will some digits be
from Right	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 $\rightarrow$ PSTN CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the **Add IP** $\rightarrow$ **PSTN CallerID Manipulation Rule** interface. Note that the item **Index** cannot be modified.



IP->PSTN CallerID Manipulation	
Index: 63	
00	
Call Initiator: SIP Trunk Group [0]	-
CallerID Prefix: 9	
CalleeID Prefix: *	
With Original CalleeID: No	✓
Stripped Digits from Left: 1	
Stripped Digits from Right: 0	
Reserved Digits from Right: 0	
Prefix to Add:	
Suffix to Add:	
Description: default	
Close	

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

63         SIP Trunk Group [0]         •         No         O         O         O         default         @		Number Manipulation Rules											
	Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	With Original CalleelD	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]	•	•	No	0	0	0			default	
	Check All Uncheck All Inverse Option Clear All Add New												

Figure 3-106 IP→PSTN CalleeID Manipulation Interface



#### 3.11.3 IP to PSTN Original CalleeID

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

	Number Manipulation Rules										
Check	Index	Call Initiator	CallerID Prefix	CalleelD Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	Description	Modif
	63	SIP Trunk Group [0]	2	5	1	0	10	666	888	default	
Check All 🗧 Uncheck All 🗧 hverse 🗏 Diskie 🗏 Clear All 🖉 Add New											
Name Tak	-1 20 #-	ma/Daga 1/1 First D	and the Alexandrian	the Option Design of L	4 Dense Tatal						

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 A	Check All Uncheck All I Inverse Delete Clear All Add New											
1 Items To	tal 20 Ite	ms/Page 1/1 First Prev	vious Next Last 0	Bo to Page 🚺 👽 1	Pages Total							

Figure 3-108 PSTN→IP CallerID Manipulation Interface

See Figure 3-108 for the PSTN $\rightarrow$ 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 $\rightarrow$ IP CallerID manipulation rule adding interface.



PSTN->IP Call	erID Manipulation				
Index:	63 💌				
Call Initiator:	PCM Trunk Group [0]				
CallerID Prefix:	*				
CalleeID Prefix:	*				
With Original CalleelD	: No 💌				
Stripped Digits from Left: 0					
Stripped Digits from R	ight: 0				
Reserved Digits from I	Right: 0				
Prefix to Add:					
Suffix to Add:					
Description:	default				
Save	Close				

Figure 3-109 Add PSTN→IP CallerID Manipulation Rule

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

Item	Description
	The unique index of each number manipulation rule, which denotes its priority. A
Index	number manipulation rule with a smaller index value has a higher priority. If a call
Index	matches several number manipulation rules, it will be processed according to the
	one with the highest priority.
Coll Initiator	PCM trunk group from where the call is initiated. This item can be set to a specific
Call Initiator	PCM trunk group or PCM Trunk Group[ANY] which indicates any PCM trunk group.
	A string of numbers at the beginning of the calling/called party number. This item
CallerID Prefix,	can be set to a specific string or "*" which indicates any string. These two
CalleeID Prefix	configuration items together with Call Initiator and With Original CalleeID can
	specify the calls which apply to the number manipulation rule.
With Original	If this item is set to Yes, it indicates that the number manipulation rule is only
With Original	applicable to the calls with original CalleeID/redirecting number. The default value is
CalleelD	No.



Stripped Digits from	The amount of digits to be deleted from the left end of the number. If the value of				
Left	this item exceeds the length of the current number, the whole number will be				
Len	deleted.				
Stripped Digits from	The amount of digits to be deleted from the right end of the number. If the value of				
Stripped Digits from	this item exceeds the length of the current number, the whole number will be				
Right	deleted.				
December 1 Distin	The amount of digits to be reserved from the right end of the number. Only when the				
Reserved Digits	value of this item is less than the length of the current number will some digits be				
from Right	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 $\rightarrow$ IP CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the **Add PSTN\rightarrowIP CallerID Manipulation Rule** interface. Note that the item **Index** cannot be modified.



PSTN->IP CallerID	Manipulation
Index:	63
Call Initiator: PCM	Trunk Group [0] 🕑
CallerID Prefix:	89
CalleeID Prefix:	*
With Original CalleeID:	No
Stripped Digits from Left:	2
Stripped Digits from Right:	0
Reserved Digits from Righ	t: 0
Prefix to Add:	
Suffix to Add:	
Description:	default
Save	Close

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

	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]	0	9	No	1	0	0			default							
Chedx All Unchedx All Inverse Delete Clear All Add New Items Total 20 Items Page 1/1 First Previous Next Last Go to Page 1 V 1 Pages Total																		
i items i o	tai 20 ite	ems/mage 1/1 mist met	nous ivext Last (	so to rage 1 💌 1	rages rotar							ems Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 🕐 1 Pages Total						

Figure 3-111 PSTN→IP CalleeID Manipulation Interface



#### 3.11.6 PSTN to IP Original CalleeID

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

	Number Manipulation Rules										
Check	Index	Call Initiator	CallerID Prefix	CalleelD Prefix	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]	*	*	0	0	0			default	
Check All Uncheck All I Inverse Delete Clear All Add New											
1 Items Tot	Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 V 1 Pages Total										

Figure 3-112 PSTN→IP Original CalleeID Manipulation Interface

#### 3.11.7 CallerID Pool

IP->PSTN CallerID Pool							PSTN-	>IP CallerID Pool	
Check	No.	CallerID	Outgoing Call Resource	Modify	Check	No.	CallerID	Outgoing Call Resource	Modify
	0	111	1			0	0100	2	
	1	222	1			1	6	1	
	2	333	1						
	3	444	1						
	4	555	1						
Delete	Clea	ar All		Add New	Delete	Clea	ar All		Add New

Figure 3-113 CallerID Pool Interface

See Figure 3-113 for the CallerID Pool interface, including two parts: PSTN $\rightarrow$ IP CallerID Pool and IP $\rightarrow$ 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.

Ca	llerID
No.:	5
CallerID:	
Outgoing Call Resource	ce:
Batch Add:	1
Save	Close



#### 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
No.	priority. A CallerID with a smaller index value has a higher priority.
CallerID	Sets the CallerID used for an outgoing call.
Outgoing Call	Sets the maximum number of the outgoing calls for each CallerID.
Resource	
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.

	CallerID
No.:	0
CallerID:	111
Outgoing Call Res	source: 1
Save	Close

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

	Network Settir	ngs
LAN 1		
	IP Address (I)	201.123.112.211
	Subnet Mask (U)	255.255.255.0
	Default Gateway (D)	201.123.112.254
	DNS Server (P)	0.0.0.0
	Speed and Duplex Mode	Automatic Detection
LAN 2		
	IP Address (I)	192.168.0.101
	Subnet Mask (U)	111.111.111.111
	Default Gateway (D)	192.168.0.254
	DNS Server (P)	0.0.0.0
	Speed and Duplex Mode	Automatic Detection
BOND Setting	l	
	BOND:	Eyes ONo
	BOND Address:	LAN 1
	Save	Reset

Note: After IP address modification, please log in again using your new IP address.

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

	Management Parameters
WEB Management	
WEB Port	1080
Access Setting	IPs in Whitelist
-	201.123.115,201.123.113
IP Address	
	IP addresses are
	separated by ','
SSH Management Config	
SSH	⊙Yes ONo
SSH Port	22
Remote Data Capture Config	Q 27%
Remote Data Captu	re OYes 🖾 No
SYSLOG Parameters	
SYSLOG	⊙Yes ONo
Server Address	127.0.0.1
SYSLOG Level	ERROR
Time Parameters	
NTP	⊙Yes ONo
NTP Server Address	127.0.0.1
Synchronizing Cycle	3600 s
Daily Restart	⊙Yes ONo
Restart Time	7 💌 h 13 💌 m
System Time	Modify 2014-10-14 16:00:25
Time Zone	GMT+8:00 (Beijing, Singapore, Taipei, Kual 🕶

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.

ltem	Description
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.
	Sets the IP addresses which can access the gateway via WEB. By default, all IPs
Assess Ostilizer	are allowed. You can set an IP whitelist to allow all the IPs within it to access the
Access Setting	gateway freely. Also you can set an IP blacklist to forbid all the IPs within it to access
	the gateway.
0011	Sets whether to enable the gateway to be accessed via SSH, with the default value
SSH	of No.



SSH Port	The port which is used to access the gateway via SSH.	
Remote Data	After this feature is enabled, you can obtain the gateway data via a remote capture	
Capture	tool. The default value is <i>No</i> .	
	Sets whether to enable SYSLOG. It is required to fill in SYSLOG Server Address	
SYSLOG	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.	
	Sets whether to enable the NTP time synchronization feature. It is required to fill in	
NTP	NTP Server Address, Synchronizing Cycle and Time Zone in case NTP is	
	enabled. By default, <i>NTP</i> is disabled.	
NTP Server Address	s Sets the Server address for NTP time synchronization.	
Synchronizing Cycle	Sets the cycle for NTP time synchronization.	
Deiles Destart	Sets whether to restart the gateway regularly every day at the preset Restart Time.	
Daily Restart	By default, this feature is disabled.	
Restart Time	Sets the time to restart the gateway regularly.	
Curatam Time	The system time. Check the checkbox before <i>Modify</i> and change the time in the	
System Time	edit box.	
Time Zone	The time zone of the gateway.	

### 3.12.3 SNMP Config

SNMP Configuration				
SNMP Configuration SNMP Server Address Monitoring Port Community String Configuration Read-only Community String	<ul> <li>Enable SNMP</li> <li>127.0.0.1</li> <li>161</li> </ul>			
Save	Reset			

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

Radius Configuration				
Radius:	Enable			
Master Server:	201.123.115.26:1813			
Shared Key:	••••			
Spare Server:	201.123.112.210:1813			
Shared Key:	•••			
Timeout (s):	4			
Retransmission Times:	1			
Call Type (Records Output Required):	<ul> <li>PSTN-&gt;IP</li> <li>IP-&gt;PSTN</li> <li>Conversation Start</li> <li>Access Failure</li> </ul>			
Save				

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.

Item	Description		
Radius	Sets whether to enable Radius or not, with the default setting of disabled.		
Maakan Oamaan	Sets the IP address and port of the master Radius server.		
Master Server	Note: If the port isn't designated, the default port 1813 will be used.		
	Sets the shared key used for the communication encryption between the master		
Sharad Kay	Radius server and the Radius client.		
Shared Key	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.		
	Sets the IP address and the port of the spare Radius server which will be		
Crown Common	automatically started upon the occurrence of malfunction on the communications		
Spare Server	between the gateway and Radius master server.		
	Note: If the port isn't designated, the default port 1813 will be used.		

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



	1			
	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			
Timeout	gateway will s	start the message retransmission mechanism once the charge		
	message sent	from the gateway to the Radius server is timeout without any		
	response.			
Retransmission	Sets the retran	smission times on no response to the Radius message, with the		
Times	default value of	f 3.		
	Sets the type of calls which are required to output call records, including four			
	options: PSTN-	$\rightarrow$ IP, IP $\rightarrow$ PSTN, conversion start and access failure.		
	Туре	Meaning		
		Whether to send the Radius charge message for the calls from		
	PSTN→IP	PSTN to IP		
		Whether to send the Radius charge message for the calls from		
Call Turne (Deserveda	IP→PSTN	IP to PSTN		
Call Type (Records		Whether to send the record of the initial conversion, that is,		
output required)	Conversion	whether to have the gateway send the record information about		
	Start	the initial conversion to the Radius server upon the connection		
		of the conversion.		
		Whether to send the record of the calls in access failure, that is,		
	Access	whether to have the gateway send the record information about		
	Failure	the calls in access failure to the Radius server upon the access		
		failure occurs.		

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



### 3.12.5 Configuration File

	SMGConfig.ini	¥
Config File		
[Version]		^
GWSvrV=1.0.1hehehex		
KernelV=Linux mpc8309som 2.6.34 #85 Thu Dec 6 10:12:49 CST 2012		
WebV=1.0.1		
CpIdV=45621.586		
HWaddr1=00:04:9F:EF:03:02		
HWaddr2=00:04:9F:EF:03:02		
[WebCtrl]		
LocalAddress=127.0.0.1		
LocalPort=1001		
[Monitor]		
LocalAddress=127.0.0.1		
LocalPort=1002		
AutoExec=1		
UpgradeExecPath=/usr/local/apache/htdocs/RecUpgrade		
IniFilePath=/mnt/flash		
[DigitsMapRulesInfo]		
DigitsMapRulesNum=1		
[NetConfig]		
BondFlag=0		
lpAddr1=201.123.112.211		
Subnet1=255.255.255.0		
Gateway1=201.123.112.254		
DNS1=0.0.0		
CheckNet1=0		
lpAddr2=192.168.0.101		
Subnet2=111.111.111.111		
Gateway2=192.168.0.254		
DNS2=0.0.0		
CheckNet2=0		
Mode1=0		
Mode2=0		
EnableBond=0		
BondEth=		
[SysInfo]		~
Save Reset	nurotion file!	
Note: You shall restart the service or system to validate the modified config	juration 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

Data Capture			
Choose a network interface to capture data Destination Address for Syslog	LAN 1	Start	
	Data Recording		
Choose a port and a time slot to record data	ort 1 💌 E1 Time Slot 0(T1 Time Slot 0)	D) V Start	Stop
Choose a port and a time slot to record P	ort 1 💌 E1 Time Slot 16	▼ Start	Stop
	Two-way Recording		
Choose a port and a time slot to record P	ort 1 💌 E1 Time Slot 0(T1 Time Slot 0)	D) V Start	Stop
Choose a port and a time slot to record P data	ort 1 💌 E1 Time Slot 16	▼ Start	Stop
Clean Data Download Log			

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

	Signaling Call Test	
Test Type	IP->PSTN	<b>v</b>
SIP Trunk Group No.	SIP Trunk Group[0]	•
CallerID		
CalledID		
Start	Clear	
GWS_OUT_MAKE_CAL chid=0006,chid=0514,1 GWS_IDLE>GWS_IN_ chid=0514,chid=0006,1 GWS_OUT_WAIT_CALL chid=0006,chid=0514,1	12->2222 CALL_ID= stat char RESULT>GWS_OUT_WAI 12->2222 CALL_ID= stat char	nge: T_CONNECT nge:
chid=0006,chid=0514,1 GWS_IN_WAIT_OUT_C chid=0514,chid=0006,1 GWS_OUT_WAIT_CON	->GWS_IN_WAIT_OUT_CONN 12->2222 CALL_ID= stat char CONNECT>GWS_IN_SEND_ 12->2222 CALL_ID= stat char NECT>GWS_OUT_CONNEC 12->2222 CALL_ID= stat char	nge: PICKUP nge: CTED
GWS_IN_SEND_PICKU ch=514 Rcv DTMF 1 ch=514 Rcv DTMF 1 ch=514 Rcv DTMF 4	JP>GWS_IN_CONNECTED	<b>▼</b>

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

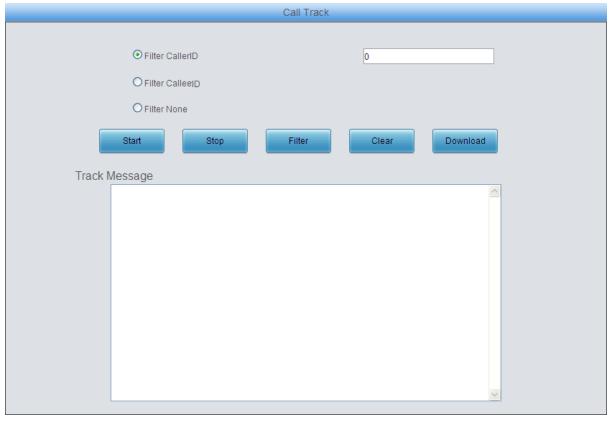


PCM Channel	You are required to select the PCM channel if choosing PSTN Call Out in Test
PCM Channel	Type, Note: This item will appear only if you choose PSTN Call Out in Test Type,
	You can select this item to send DTMFs after the establishment of call conversation
DTMF	on the channel for call test,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

	Ping Te	st
Source	IP Address	LAN 1: 192.168.1.101
Destina	ation Address	127.0.0.1
Ping Co	ount (1-100)	4
Packag	e Length (56-1024 bytes)	56
	Start	End
Info		

#### 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.	
la fa	The information returned during the Ping test, helping you to learn the network	
Info	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

Tracert Test			
Sourc	e IP Address	LAN 1: 192.168.1.101 💌	
Destir	nation Address	127.0.0.1	
Maxim	num Jumps (1-255)	30	
Info	Start	End	

#### 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.

ltem	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

Modification Record	
2014-11-27 T0.29.04 Mod.Comig/SMGComig.mi-retComig-ipAddrit.192.106.1.101201.125.115.196.itom	^
2014-11-27 10:29:54 Mod:Config/SMGConfig.ini-NetConfig-Gateway1:192.168.1.254>201.123.115.254 from	
201.123.115.35	
2014-11-27 10:29:54 Add:Config/SMGConfig.ini-NetConfig-Mode1:-1>0 from 201.123.115.35	
2014-11-27 10:29:54 Add:Config/SMGConfig.ini-NetConfig-Mode2:-1>0 from 201.123.115.35	
2014-11-27 10:29:54 Mod:Config/SMGConfig.ini-GWGLOBAL-global_localaddress:192.168.1.101>201.123.115.198	
from 201.123.115.35	
2014-11-27 10:29:54 Mod:Config/ShConfig.ini-SIP-LocalSiplp:192.168.1.101>201.123.115.198 from 201.123.115.35	
2014-11-27 10:29:54 Mod:Config/ShConfig.ini-BoardId=0-BoardIP:192.168.1.101>201.123.115.198 from	
201.123.115.35	
2014-11-27 10:29:54 Mod:Config/ShConfig.ini-BoardId=0-Gateway:192.168.1.254>201.123.115.254 from	
201.123.115.35	
2014-11-27 10:30:14 Mod:Config/Ss7Server.ini-Ss7SystemConfig-SetClient:1>0 from 201.123.115.35	
2014-11-27 10:30:47 Mod:Config/Ss7Server.ini-Ss7SystemConfig-ServerIP:127.0.0.1>201.123.115.198 from	
201.123.115.35	
2014-11-27 10:30:47 Del:Config/Ss7Server.ini-Ss7SystemConfig-SecondServerIP from 201.123.115.35	
2014-11-27 10:30:59 Mod:Config/ShConfig.ini-SS7-LocalIP:127.0.0.1>201.123.115.198 from 201.123.115.35	
2014-11-27 10:31:10 Mod:Config/ShConfig.ini-SS7-Ss7ServerIP:127.0.0.1>201.123.115.199 from 201.123.115.35	
2014-11-27 10:31:54 Mod:Config/Ss7Server.ini-Ss7ClientInfo-IP[0]:127.0.0.1>201.123.115.198 from 201.123.115.35	
2014-11-27 10:31:55 AllDel:Config/Ss7Server.ini-PCMLINKINFO from 201.123.115.35	
2014-11-27 10:31:55 Add:Config/Ss7Server.ini-PCMLINKINFO-Ss7PcmLink0:-1>0 from 201.123.115.35	
2014-11-27 10:32:39 Mod:Config/ShConfig.ini-SS7-Ss7ServerIP:201.123.115.199>201.123.115.198 from	
201.123.115.35	
2014-11-27 10:32:45 Mod:Config/Ss7Server.ini-Ss7SystemConfig-ServerIP:201.123.115.198>201.123.115.199 from	
201.123.115.35	
2014-11-27 10:32:45 Del:Config/Ss7Server.ini-Ss7SystemConfig-SecondServerIP from 201.123.115.35	
2014-11-27 10:32:52 AllDel:Config/Ss7Server.ini-PCMLINKINFO from 201.123.115.35	
2014-11-27 10:32:52 Add:Config/Ss7Server.ini-PCMLINKINFO-Ss7PcmLink0:-1>0 from 201.123.115.35	
2014-11-27 10:33:00 Mod:Config/Ss7Server.ini-Ss7ClientInfo-IP[0]:201.123.115.198>201.123.115.199 from	
201.123.115.35	
2014-11-27 10:33:00 AllDel:Config/Ss7Server.ini-PCMLINKINFO from 201.123.115.35	
2014-11-27 11:03:29 Mod:Config/Ss7Server.ini-Ss7SystemConfig-ServerIP:201.123.115.198>201.123.115.199 from	
201.123.115.35	
2014-11-27 11:03:29 Del:Config/Ss7Server.ini-Ss7SystemConfig-SecondServerIP from 201.123.115.35	
2014-11-27 11:32:12 Add:Config/ShConfig.ini-SS7-IsupPcm[0]:-1>LocalPcm[0] from 201.123.115.35	
2014-11-27 11:32:12 Mod:Config/ShConfig.ini-SS7-TotallsupPcm:0>1 from 201.123.115.35	
2014-11-27 11:32:18 Add:Config/ShConfig.ini-SS7-IsupPcm[1]:-1>LocalPcm[1] from 201.123.115.35	~
Check Download	

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

Data Backup				
Choose a file to backup: Configuration file Click the 'Backup' button on the right to backup the file.			Backup	
		Da	ta Upload	
To upload a file, select it and click the button 'Upload' on the right to start. Choose a file to upload: Configuration file  Upload 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

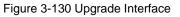
Factory Reset		
Click the button 'Reset' below to restore to factory settings.		
Reset		

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

Current Version		
Serial	000000902	
Number	00000902	
WEB	1.6.0_2015030916	
Service	1.6.0_2015030916	
Uboot	3.0.2_201501	
Kernel	#338 SMP Tue Jan 27 15:28:50 CST 2015	
Firmware	19	
Select an Update File Browse		
Update		



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

Change Password			
Current Username	admin		
Current Password			
New Username			
New Password			
Confirm New password			
Save	Reset		

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

Device Lock			
Please select the condition to lock the device (Note: The device will be locked upon any one of the selected items being modified.)          IP       IP			
	Lock	Reset	

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.

Device Lock			
Password			
	Unlock Reset		

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.

This device	e has been locked successfully!
	OK

Figure 3-134 Device Lock Interface

### 3.12.17 Restart

Service Restart			
Click the button 'Restart' to restart the service.	Restart		
System Restart			
Click the button 'Restart' to restart the system.	Restart		

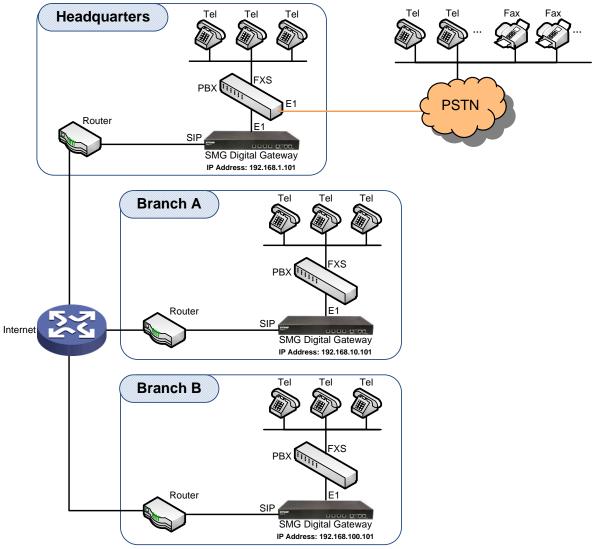
Figure 3-135 Service/System Restart Interface

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.

	*
🖁 VolP	*
SIP	
SIP Trunk	
SIP Register SIP Account	
SIP Trunk Group Media	
) рсм	*
J ISDN	×
Route	*
Number Filter	*
Num Manipulate	
System Tools	×

Figure 4-2

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

Operation Info 🛸												
📑 VolP 🔹		SIP Trunk										
-	Check	Index	Remote Address	Remote Port	Outgoing Voice Resource	Incoming Voice Resource	Modify					
SIP SIP Trunk		0	201.123.112.212	5060	128	2						
SIP Register		1	201.123.112.210	5060	128	128						
SIP Account												
SIP Trunk Group		Uncheck All					Add New					
Media	2 Items Total 2	0 Items/Page	1/1 First Previous Next Last	Go to Page 1 💙 1 Page	es Total							

Figure 4-3

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

000



Operation Info 🛛 😆													
📑 VolP 🛸		SIP Trunk Group											
	Check	Index	SIP Trunks	SIP Trunk Select Mode	Description	Modify							
SIP		0	0	Increase	Branch_A								
SIP Trunk		1	1	Increase	Branch B	-							
SIP Register					biditor_b	LØ							
SIP Account	Check All	Uncheck All	se 🗄 Delete 🗄	Clear All									
SIP Trunk Group			revious Next Last Go to Page 1			MULTISH							
Media	2 Northa Fotor a	to nonion ago internot th	and a more case of the age										

#### Figure 4-4

### 4. Set PCM.

Operation Info 🛛 🗧									
式 VolP 🛛 👻					PCM Settings				
(i) PCM	PCM No.	Signaling Protocol	Clock	Signaling Time Slot	Signaling Link Type	Connection Line	Incoming Call Start TS, Amount	CRC-4	Modif
	0	ISDN Network Side	Line-sychronization	16	Signaling	Twisted Pair Cable	-	Enable	
PSTN	1	ISDN Network Side	Slave	16	Voice	Twisted Pair Cable		Enable	
Circuit Maintenance PCM	2	ISDN Network Side	Slave	16	Signaling	Twisted Pair Cable	-	Enable	
PCM Trunk	3	ISDN Network Side	Slave	16	Signaling	Twisted Pair Cable	-	Enable	
PCM Trunk Group									
Num-Receiving Rule									
Reception Timeout									
Number Attribution									

Figure 4-5

### 5. Add PCM trunk

Operation Info	*					
VolP	*				PCM Trunks	
-	*	Check	Index	PCM NO.	Including Ts	Modify
DCM	~		0	0	1,2,3,4,5,6,7,8,9,10,11,12,13,14,15,17,18,19,20,21,22,23,24,25,26,27,28,29,30,31	
PSTN						
Circuit Maintenan	ice	Check All	Uncheck A	II 🗄 Inverse	🗄 Delete 🗄 Clear All	Add New
PCM		1 Items Total 2	0 Items/Page	1/1 First Previou	is Next Last Go to Page 1 V 1 Pages Total	
PCM Trunk		,				
PCM Trunk Group	0					
Num-Receiving R	Rule					
Reception Timeo	ut					

Figure 4-6

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

Operation Info	*											
B VolP	*	PCM Trunk Group										
DCM	*	Check	Index	PCM Trunks	PCM Trunk Select Mode	Description	Modify					
() PCm			0	0	Increase	Headquarters						
PSTN												
Circuit Maintenan	се	Check All 🗄 U	ncheck All 🗏 🛛 Inv	verse 🗄 Delete 🗄	Clear All							
PCM												
PCM Trunk												
PCM Trunk Group	>											
Num-Receiving R	Rule											
Reception Timeo	ut											
Number Attributio	n											

Figure 4-7

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

Operation Info	*		
*			Route Settings
	*	IP->PSTN	Route before Number Manipulate
SDN	*	PSTN->IP	Route before Number Manipulate
Fax	*		
Route	*		Save
Routing Parameter	rs		
IP->PSTN			
PSTN->IP			



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.

Operation Info	*									
VolP	*					Routing Ru	iles			
PCM	*	Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
	*		63	SIP Trunk Group [0]	*	*	none	PCM Trunk Group [0]	from_Branch-A	
Fax	*		62	SIP Trunk Group [1]	*	ź	none	PCM Trunk Group [0]	from_Branch-B	
Route	*	Check All	- Unche	ck All 🗄 Inverse 🗮	Delete 🗄 Clea	r All				Add New
Routing Paramete	ers	2 Items Total	20 Items/P	age 1/1 First Previous Ne	t Last Go to Page 1	<ul> <li>1 Pages Total</li> </ul>				
IP->PSTN										
PSTN->IP										



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.

Operation Info	*									
VolP	*					Routing Rules				
PCM	*	Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
	*		63	PCM Trunk Group [0]	*	*	none	SIP Trunk Group [0]	to_Branch-A	
_			62	PCM Trunk Group [0]	*	*	none	SIP Trunk Group [1]	to_Branch-B	
🔅 Fax	*								1	
Route	*	Check All	E Unche	ck All 🗄 Inverse 🗄 🛙	)elete 🛛 🗄 🛛 Clear All					Add New
Routing Parameter	rs	2 Items Total	20 Items/Pa	age 1/1 First Previous Next I	Last Go to Page 1 💌	1 Pages Total				
IP->PSTN										
PSTN->IP		•								

#### 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.

Operation Info	*												
🗱 VolP 🔅	*						Number Manipulation	n Rules					
	S 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]	•	8	No	1	0	100			to_Branch-A	
	*	62         PCM Trunk Group [0]         •         7         No         1         0         100         to_B											
	Check	AII E	Uncheck All 📄 Inver	se 🗄 Delet	Clear A							Add	d New
📑 Number Filter 🛛 3	2 Items 1	otal 20 It	ems/Page 1/1 First P	revious Next Last	t Go to Page 1 🗙	1 Pages Total							
💌 Num Manipulate 🤅	*												
IP->PSTN CallerID													
IP->PSTN CalleeID													
IP->PSTN OriCalleeID	)												
PSTN->IP CallerID													
PSTN->IP CalleeID													
PSTN->IP OriCalleeIE	)												
CallerID Pool													

Figure 4-11

### 4.1.2 Configurations for Branch A

1. Configure SIP Settings for Branch A.



	*		
📑 VolP	*	SIP Settings	
SIP		SIP Address	LAN 1: 201.123.112.211
SIP Trunk SIP Register		SIP Signaling Port	5060
SIP Account		Send 183 Message	Enable
SIP Trunk Group Media		Obtain CallerID from	Username of From Field
() PCM	*	Obtain CalleeID from	'Request Field
ISDN	*	Obtain Redirecting Number/Original CalleeID from Diversion Field	Enable
Route	*	Stun Traversal	Enable
Number Filter	*	SIP Transport Protocol	UDP
Num Manipulate		SIP Encryption	Enable
Ystem Tools	×	RTP Encryption	Enable
		RTP Self-adaption	Enable
		UDP Header Checksum	Enable
		Rport	Enable
		DSCP	Enable
		Maximum Wait Answer Time (s)	60
		Maximum Wait RTP Time (s)	0
		Maximum Wait PSTN Resource Time(ms)	5000

Figure 4-12

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

Operation Info 🛛 😆								
式 VolP						SIP Trunk		
	-	Check	Index	Remote Address	Remote Port	Outgoing Voice Resource	Incoming Voice Resource	Modify
SIP SIP Trunk			0	201.123.112.212	5060	128	2	
SIP Register			1	201.123.112.210	5060	128	128	
SIP Account								
SIP Trunk Group		Check All	Uncheck Al					Add New
Media	2	Items Total 20	) Items/Page	1/1 First Previous Next Last	Go to Page 1 🔽 1 Page	is Total		

Figure 4-13

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

Operation Info	*						
式 VolP	*				SIP Trunk Group		
		Check	Index SIP Trunks		SIP Trunk Select Mode	Description	Modify
SIP			0	0	Increase	Headquarters	
SIP Trunk			1 1		Increase	Branch_B	
SIP Register				1	increase	branch_b	ß
SIP Account		Check All 🗄 Und			21 11		
SIP Trunk Group					Clear All		Add New
Media		2 Items Total 20 Items	s/Page 1/1 First F	revious Next Last Go to Page	1 🗙 1 Pages Total		

Figure 4-14

4. Set PCM.



Operation Info 🛛 🗧									
😹 VolP 🛛 😣					PCM Settings				
<ul> <li>(i) PCM</li> <li>(ii) A</li> </ul>	PCM No.	Signaling Protocol	Clock	Signaling Time Slot	Signaling Link Type	Connection Line	Incoming Call Start TS, Amount	CRC-4	Modify
PCM	0	ISDN Network Side	Line-sychronization	16	Signaling	Twisted Pair Cable		Enable	
PSTN	1	ISDN Network Side	Slave	16	Voice	Twisted Pair Cable	-	Enable	
Circuit Maintenance PCM	2	ISDN Network Side	Slave	16	Signaling	Twisted Pair Cable		Enable	
PCM Trunk	3	ISDN Network Side	Slave	16	Signaling	Twisted Pair Cable		Enable	
PCM Trunk Group									
Num-Receiving Rule									
Reception Timeout									
Number Attribution									

Figure 4-15

### 5. Add PCM trunk

Operation Info 🛛 😆					
式 VolP 🛛 😒				PCM Trunks	
<ul> <li>(i) PCM</li> <li>♠</li> </ul>	Check	Index	PCM NO.	Including Ts	Modify
() PCM		0	0	1,2,3,4,5,6,7,8,9,10,11,12,13,14,15,17,18,19,20,21,22,23,24,25,26,27,28,29,30,31	
PSTN					
Circuit Maintenance	Check All	Uncheck A	II 🗄 Inverse	🗄 Delete 🗄 Clear All	Add New
PCM	1 Items Total 2	20 Items/Page	1/1 First Previou	is Next Last Go to Page 1 💌 1 Pages Total	
PCM Trunk	•				
PCM Trunk Group					
Num-Receiving Rule					
Reception Timeout					
Number Attribution					

Figure 4-16

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

Operation Info 🛛 🗧						
式 VolP 🛛 🗧				PCM Trunk Group		
() PCM ≈	Check	Index	PCM Trunks	PCM Trunk Select Mode	Description	Modify
PCM A		0	0	Increase	Branch_A	
PSTN		1	1	1		
Circuit Maintenance	Check All 🗄 Un	check All 🗄 Inve	erse 🗄 Delete 🗄 (	Clear All		
PCM						
PCM Trunk						
PCM Trunk Group	Þ					
Num-Receiving Rule						
Reception Timeout						
Number Attribution						

Figure 4-17

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

Operation Info	*
VolP	*
() PCM	*
ISDN	*
Fax	*
Route	*
Routing Paramete	rs
IP->PSTN	
PSTN->IP	



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.



Operation Info	*								
VolP	*	-				Routing Rules			
D PCM	×	Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Call Destination	Description	Modify
J ISDN	*		63	SIP Trunk Group [0]			PCM Trunk Group [0]	from_HQ	6
<ul> <li>Fax</li> </ul>	*		62	SIP Trunk Group [1]		•	PCM Trunk Group [0]	from_Branch-B	0
Route	*	Check All	Uncheck A	I Inverse I Opp	Clear All				Add New
Routing Paramete	ers	2 Items Total 20	) Items/Page	1/1 First Previous Next Last		s Total			

#### 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.

Operation Info	*				Ro	uting Rules			
		Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Call Destination	Description	Modify
PCM ISDN	× ×		63	PCM Trunk Group [0]	•	9	SIP Trunk Group [0]	to_HQ	0
6 Fax	*		62	PCM Trunk Group [0]	•	7	SIP Trunk Group [1]	to_Branch-B	1
Route	*		61	PCM Trunk Group [0]	•	0	SIP Trunk Group [0]	to_PSTN	G
Route Routing Paramet		Check All	Uncheck All	Inverse Ocide	Clear All				Add New
IP->PSTN		3 items Total	20 Items/Page	1/1 First Previous Next Last G	to to Page 1 V 1 Pages To	tal			

#### 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.

Operation Info	*													
VolP	*							Number Manipulation	n Rules					
() PCM	*	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
ISDN	*		63	PCM Trunk Group [0]	•	9	No	1	0	100			to_HQ	
<ul> <li>Fax</li> </ul>	*		62	PCM Trunk Group [0]	•	7	No	1	0	100			to_Branch-B	
-9-	*	Check A		Uncheck All Inver	se 🗄 Deist	Clear A								New
_				ems/Page 1/1 First Pi										INEN
Num Manipulate	*													
IP->PSTN CallerID														
IP->PSTN CalleeID														
IP->PSTN OriCallee														
PSTN->IP CallerID														
PSTN->IP CalleeID														
PSTN->IP OriCallee	ID													
CallerID Pool														

Figure 4-21

### 4.1.3 Configurations for Branch B

1. Configure SIP Settings for Branch B.



Operation Info	*		
🚟 VolP	*	SIP Settings	
SIP		SIP Address	LAN 1: 201.123.112.211
SIP Trunk SIP Register		SIP Signaling Port	5060
SIP Account		Send 183 Message	Enable
SIP Trunk Group Media		Obtain CallerID from	Username of From Field
🚺 PCM	*	Obtain CalleelD from	'Request' Field
ISDN	*	Obtain Redirecting Number/Original CalleelD from Diversion Field	Enable
Route	*	Stun Traversal	Enable
Number Filter	*	SIP Transport Protocol	UDP
Num Manipulate		SIP Encryption	Enable
M System Tools	*	RTP Encryption	Enable
		RTP Self-adaption	Enable
		UDP Header Checksum	Enable
		Rport	Enable
		DSCP	Enable
		Maximum Wait Answer Time (s)	60
		Maximum Wait RTP Time (s)	0
		Maximum Wait PSTN Resource Time(ms)	5000

Figure 4-22

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

Operation Info 🛛 😆							
式 VolP					SIP Trunk		
	Check	Index	Remote Address	Remote Port	Outgoing Voice Resource	Incoming Voice Resource	Modify
SIP		0	201.123.112.212	5060	128	2	
SIP Trunk SIP Register		1	201.123.112.210	5060	128	128	
SIP Account							
SIP Trunk Group	Check All	Uncheck Al					Add New
Media	2 Items Total	20 Items/Page	1/1 First Previous Next Last	Go to Page 1 💌 1 Page	es Total		

Figure 4-23

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

Operation Info 🛛 😆						
				SIP Trunk Group		
	Check	Index	SIP Trunks	SIP Trunk Select Mode	Description	Modify
SIP		0	0	Increase	Headquarters	
SIP Trunk		1	4	Increase	Branch_A	~
SIP Register		'	1	Inclease	brancii_A	
SIP Account		) = [ .				
SIP Trunk Group	Check All 🗄 Un			Clear All		Add New
Media	2 Items Total 20 Item	s/Page 1/1 First F	revious Next Last Go to Page	1 💙 1 Pages Total		

Figure 4-24

4. Set PCM.



Operation Info	*									
🚟 VolP	*					PCM Settings				
() PCM	*	PCM No.	Signaling Protocol	Clock	Signaling Time Slot	Signaling Link Type	Connection Line	Incoming Call Start TS, Amount	CRC-4	Modify
PCIII	0	0	ISDN Network Side	Line-sychronization	16	Signaling	Twisted Pair Cable		Enable	
PSTN		1	ISDN Network Side	Slave	16	Voice	Twisted Pair Cable	-	Enable	
Circuit Maintenance PCM		2	ISDN Network Side	Slave	16	Signaling	Twisted Pair Cable	-	Enable	
PCM Trunk		3	ISDN Network Side	Slave	16	Signaling	Twisted Pair Cable	-	Enable	
PCM Trunk Group					1					
Num-Receiving Ru	le									
Reception Timeout	t									
Number Attribution										

Figure 4-25

### 5. Add PCM trunk

Operation Info 🛛 🗧					
📑 VolP 🛛 😒				PCM Trunks	
	Check	Index	PCM NO.	Including Ts	Modify
(i) PCM 😤		0	0	1,2,3,4,5,6,7,8,9,10,11,12,13,14,15,17,18,19,20,21,22,23,24,25,26,27,28,29,30,31	
PSTN					
Circuit Maintenance	Check All	Uncheck A	II 🗄 Inverse	🗄 Delete 🗄 Clear All	Add New
PCM	1 Items Total 2	0 Items/Page	1/1 First Previou	s Next Last Go to Page 1 💌 1 Pages Total	
PCM Trunk	,				
PCM Trunk Group					
Num-Receiving Rule					
Reception Timeout					
Number Attribution					

Figure 4-26

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

Operation Info	*						
VolP	*				PCM Trunk Group		
(i) PCM	*	Check	Index	PCM Trunks	PCM Trunk Select Mode	Description	Modify
Pem	0		0	0	Increase	Branch_B	
PSTN				1			
Circuit Maintenan	ice	Check All 🗮 Und	heck All 🗄 🗌 Inv	erse 🗄 Delete 🗄 🗰	Clear All		
PCM							
PCM Trunk							
PCM Trunk Group	p >						
Num-Receiving F	Rule						
Reception Timeo	ut						
Number Attributio	n						

Figure 4-27

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

Operation Info	*		
VoIP	*		Route Settings
(i) PCM	*	IP->PSTN	Route before Number Manipulate
ISDN	*	PSTN->IP	Route before Number Manipulate
Fax	*		
Route	*		Save
Routing Paramete	rs		
IP->PSTN			
PSTN->IP			



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.



Operation Info	*								
VolP	*					Routing Rules			
DCM	×	Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Call Destination	Description	Modify
ISDN	*		63	SIP Trunk Group [0]	*		PCM Trunk Group [0]	from_HQ	2
			62	SIP Trunk Group [1]			PCM Trunk Group [0]	from_Branch-A	0
🔅 Fax	×							1	
Route	*	Check All	Uncheck A	II E Inverse E Opp	Clear All				Add New
Routing Paramete	ers	2 Items Total	20 Items/Page	1/1 First Previous Next Last	Go to Page 1 💌 1 Page	s Total			
IP->PSTN									
PSTN->IP									

#### 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.

Operation Info	*								
VolP	*				Ro	uting Rules			
PCM	*	Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Call Destination	Description	Modify
ISDN	*		63	PCM Trunk Group [0]		9	SIP Trunk Group [0]	to_HQ	
Fax	*		62	PCM Trunk Group [0]		8	SIP Trunk Group [1]	to_Branch-A	
Route	*		61	PCM Trunk Group [0]		0	SIP Trunk Group [0]	to_PSTN	
Routing Paramete			Uncheck Al						Add New
IP->PSTN PSTN->IP		3 Items Total	20 Items/Page	1/1 First Previous Next Last Go	o to Page 1 💌 1 Pages To	tal			



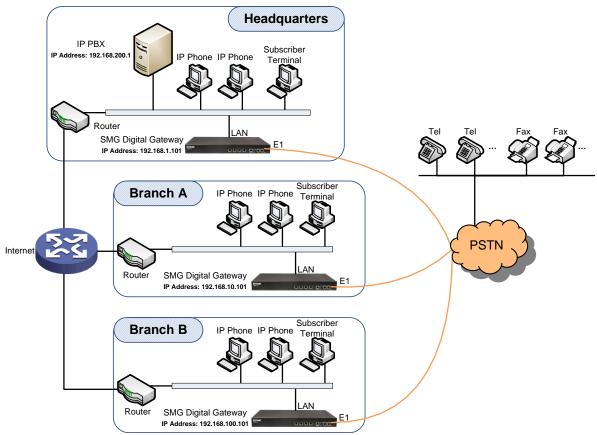
10. Set number manipulation rules. When the gateway receives a call from PSTN, it will first check the CalleeID prefix. If the CalleeID prefix is 9 or 8, the gateway will delete it before routing the call to the corresponding SIP trunk group.

Operation Info	*												
VolP	*						Number Manipulation	n Rules					
PCM	S 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]	-	9	No	1	0	100			to_HQ	
ISDN	× -	62	PCM Trunk Group [0]	-	8	No	1	0	100			to_Branch-A	
Fax	*											-	
Route	S Check	AL	Uncheck All Inver	rse 🗄 Delet	Clear A	1						Add	d New
😳 Number Filter	S 2 Items	Fotal 20 It	ems/Page 1/1 First P	revious Next Las	t Go to Page 1 🗸	1 Pages Total							
Num Manipulate													
IP->PSTN CallerID													
IP->PSTN CalleeID													
IP->PSTN OriCallee	D												
PSTN->IP CallerID	_												
PSTN->IP CalleeID													
PSTN->IP OriCallee	ID												

Figure 4-31



### 4.2 Application 2



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

#### 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.



SIP       SIP Settings         SIP Trunk       SIP Address         SIP Register       SIP Signaling Port         SIP Account       SIP Signaling Port         SIP Trunk Group       Bend 183 Message         Media       Obtain CallerID from         Image: Sing Processing Sing Trunk Group       Obtain CallerID from         Image: Sing Processing Sing Processing Sing Processing Sing Processing Sing Processing Sing CallerID from       Image: Sing Processing Sing Proc
SIP       SIP Address       L         SIP Trunk       SIP Signaling Port       SIP         SIP Account       Send 183 Message       SIP         SIP Trunk Group       Obtain CallerID from       SIP         Media       Obtain CalleeID from       SIP         ISDN       State       SIP         State       State       SIP         ISDN       State       SIP         Obtain CalleeID from       SIP       SIP         State       State       SIP         Route       State       SIP Transport Protocol         Num Manipulate       SIP Encryption       SIP Encryption
SIP Register     SIP Signaling Port     SI       SIP Account     Send 183 Message     Send 183 Message       SIP Trunk Group     Obtain CallerID from     I       Media     Obtain CalleeID from     I       ISDN     SiP Fax     Obtain Redirecting Number/Original CalleeID from       Route     Stun Traversal       Image: Number Filter     SIP Transport Protocol       Image: Num Manipulate     SIP Encryption
SIP Trunk Group       Obtain CallerID from         Media       Obtain CallerID from         I ISDN       Image: Control of CallerID from         I ISDN       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from         Image: Control of CallerID from       Image: Control of CallerID from
Media     Obtain CallerID from       Image: Decimal content of the set of
ISDN       ISDN         ISDN       ISDN         Fax       ISDN         Route       ISDN         Image: Number Filter       Image: Silp Transport Protocol         Image: Num Manipulate       Image: Silp Encryption
Image: Sign of the set of t
Route     Stun Traversal       Image: Stress of the standard stress of the standard stress of the stress of
Image: Simple Police     Simple Police       Image: Simple Police     Simple Police       Image: Simple Police     Simple Police
SIP Encryption
RTP Encryption
RTP Self-adaption
UDP Header Checksum
Rport
DSCP
Maximum Wait Answer Time (s) 6
Maximum Wait RTP Time (s) 0
Maximum Wait PSTN Resource Time(ms) 5

Figure 4-33

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

Operation Info	> >							
VolP	*					SIP Trunk		
		Check	Index	Remote Address	Remote Port	Outgoing Voice Resource	Incoming Voice Resource	Modify
SIP			0	201.123.112.10	5060	128	128	0
SIP Trunk		•						
SIP Register		Check All	Uncheck Al	I II Inverse II Delet	e 🗄 Clear All			Add New
SIP Account				1/1 First Previous Next Last		es Total		Addition
SIP Trunk Group	p	Titems Total 2	o nemon age	In That Trevious Next East	Go to l'age 11 eg	65 10(a)		
Media								

Figure 4-34

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

Operation Info	*						
😤 VolP	*				SIP Trunk Group		
		Check	Index	SIP Trunks	SIP Trunk Select Mode	Description	Modify
SIP			0	0	Increase	IP_PBX	
SIP Trunk		_				-	
SIP Register		Check All 🗄 Uncl	heck All 🗄 Inver	se 🗄 Delete 🗄	Clear All		
SIP Account				evious Next Last Go to Page 1			And Hen
SIP Trunk Group		Titems Total 20 items	wrage wi riist ri	evious Next Last Goto Page	I rages total		

Figure 4-35

4. Set PCM.



Operation Info 🛛 🗧									
😹 VolP 🛛 😣					PCM Settings				
<ul> <li>(i) PCM</li> <li>(ii) A</li> </ul>	PCM No.	Signaling Protocol	Clock	Signaling Time Slot	Signaling Link Type	Connection Line	Incoming Call Start TS, Amount	CRC-4	Modify
PCM	0	ISDN Network Side	Line-sychronization	16	Signaling	Twisted Pair Cable		Enable	
PSTN	1	ISDN Network Side	Slave	16	Voice	Twisted Pair Cable	-	Enable	
Circuit Maintenance PCM	2	ISDN Network Side	Slave	16	Signaling	Twisted Pair Cable		Enable	
PCM Trunk	3	ISDN Network Side	Slave	16	Signaling	Twisted Pair Cable		Enable	
PCM Trunk Group									
Num-Receiving Rule									
Reception Timeout									
Number Attribution									

Figure 4-36

### 5. Add PCM trunk

Operation Info 🛛 😸					
📑 VolP 🛛 👻				PCM Trunks	
	Check	Index	PCM NO.	Including Ts	Modify
👔 PCM 📚		0	0	1,2,3,4,5,6,7,8,9,10,11,12,13,14,15,17,18,19,20,21,22,23,24,25,26,27,28,29,30,31	
PSTN					
Circuit Maintenance	Check All	Uncheck A	II 🗄 Inverse	🗄 Delete 🗄 Clear All	Add New
PCM	1 Items Total 2	0 Items/Page	1/1 First Previou	s Next Last Go to Page 1 💌 1 Pages Total	
PCM Trunk	•				
PCM Trunk Group					
Num-Receiving Rule					
Reception Timeout					
Number Attribution					

Figure 4-37

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

Operation Info	*						
📑 VolP	*	_			PCM Trunk Group		
DCM	*	Check	Index	PCM Trunks	PCM Trunk Select Mode	Description	Modify
			0	0	Increase	Headquarters	
PSTN							
Circuit Maintenand	e	Check All 🗄 U	ncheck All 🗄 🗌 In	verse 🗄 Delete 🗮	Clear All		
PCM							
PCM Trunk							
PCM Trunk Group							
Num-Receiving Ru	lle						
Reception Timeou	t						
Number Attribution							

Figure 4-38

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

Operation Info	*		
VolP	*		Route Settings
() PCM	*	IP->PSTN	Route before Number Manipulate
JISDN	*	PSTN->IP	Route before Number Manipulate
Fax	*		
Route	*		Save
Routing Parameter	rs		
IP->PSTN			
PSTN->IP			



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.



*									
¥	Routing Rules								
	Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
		63	SIP Trunk Group [0]	*	*	none	PCM Trunk Group [0]	to_PSTN	
	Check All \Xi Uncheck All 🚍 Inverse 🗮 Deleto 🗮 Clear All 🖉 Add New							Add New	
~	1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 🛩 1 Pages Total								
Routing Parameters IP->PSTN									
PSTN->IP									
	» » » » » «	Check All	Check Index 63 Check All Unche 1 Items Total 20 Items/P	Check Index Call Initiator           63         SIP Trunk Group [0]           Check All         Uncheck All           Items Total 20 Items/Page 1/1 First Previous Next	Check Index Call Initiator CalleriD Prefix          63       SIP Trunk Group [0]         *       Check All         Interse       Delete         Check All       Inverse         Items Total       20 Items/Page         Items       1/1         First       Previous         Next       Last         Go to Page       Image: Check All	Routing Rules         Check       Index       Call Initiator       CallerID Prefix       CallerID Prefix         63       SIP Trunk Group [0]       *       *         Check All       Uncheck All       Inverse       Clear All         Items Total 20 Items/Page 1/1       First Previous Next Last Go to Page 1 v 1 Pages Total	Routing Rules         Check       Index       Call Initiator       CallerID Prefix       CalleeID Prefix       Number Filter         63       SIP Trunk Group [0]       *       *       none         Check All       Uncheck All       Inverse       Delcte       Clear All         1 Items Total 20 Items/Page       1/1 First Previous Next Last Go to Page       1 Pages Total	Routing Rules         Check       Index       Call Initiator       CallerID Prefix       CalleeID Prefix       Number Filter       Call Destination         63       SIP Trunk Group [0]       *       *       none       PCM Trunk Group [0]         Check All       Uncheck All       Inverse       Defere       Clear All         1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 v       1 Pages Total	Routing Rules         Check Index Call Initiator CallerD Prefix CalleeID Prefix Number Filter Call Destination Description         63       SIP Trunk Group [0]       *       *       none       PCM Trunk Group [0]       to_PSTN         Check All       Uncheck All       Inverse       Destrict       /       /         1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1/1 Pages Total       /       /       /

#### Figure 4-40

 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.

Operation Info	*									
VolP	*	Routing Rules								
DCM	*	Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
-			63	PCM Trunk Group [0]	*	9	none	SIP Trunk Group [0]	from_PSTN	
ISDN ISDN	*					1				
{ŷ} Fax	*	Check All 🗧 Uncheck All 🗧 Inverse 🚍 Delete 🚍 Clear All 🖉 Add New								
Route	~	1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 💌 1 Pages Total								
Routing Paramete										
IP->PSTN	15									
PSTN->IP		,								

#### 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<sup>3</sup>

#### 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 <u>Hardware Description</u> 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	4.75/5.15/5.90/6.70/7.40/7.9 5/10.20/12.20 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 <u>Chapter 3 WEB Configuration</u> 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
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	
41		505	unavailable	
42	Switching equipment congestion	503	Service	
72		505	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	503	Service	
	ISDN)		unavailable	
134	Temporary fault (internal definition, only applies to	503	Service	
	ISDN)		unavailable	
135	Data link failure (internal definition, only applies to	503	Service	
	ISDN)		unavailable	
65	Bearer capability not implemented	488	Not acceptable	
		100	here	
70	Only restricted digital information bearer capability	488	Not acceptable	
	is available	100	here	
102	Recovery on timer expiry	504	Server time-out	
128	T303 time out (internal definition, only applies to	504	Server time-out	
120	ISDN)	504		
129	T304 time out (internal definition, only applies to	504	Server time-out	
123	ISDN)	504		
130	T310 time out (internal definition, only applies to	504	Server time-out	
130	ISDN)	504		
111	Protocol error, unspecified	500	Server internal	
		500	error	
127	Interworking, unspecified	500	Server internal	
121		500	error	
Others	Others	500	Server internal	
Ouleis		500	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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