

SMG2030L

SMG2060L

**Digital Gateway** 

# **User Manual**

Version 1.6.5

Synway Information Engineering Co., Ltd www.synway.net



# Content

Content .		i
Copyright	Declaration	iv
Revision H	listory	v
Chapter 1	Product Introduction	1
1.1 Typic	al Application	1
1.2 Featu	ure List	2
	ware Description	
1.4 Alarn	n Info	
Chapter 2	Quick Guide	6
Chapter 3	WEB Configuration	10
3.1 Syste	em Login	
	ation Info	
3.2.1 S	System Info	
	PSTN Status	
	PCM Info	
	Call Monitor Call Count	
	Varning Info	
	Settings	
	SIP Settings	
	SIP Trunk	
	NP Register	
	SIP Account	
	IP Trunk Group	
	ledia Settings	
	Settings	
	°STN Circuit Maintenance	
	PCM	
	°CM Trunk	
	CM Trunk Group	
	lumber-receiving Rule	
	Reception Timeout	
	STN Forwarding	
	I Settings	
	SDN	
	lumber Parameter	
	Redirecting Number (Hidden item) Sottings	
	Settings	
	Settings ax	
3.7.1 F	αλ	



3.8 Roi	ute Settings	68
3.8.1	Routing Parameters	68
3.8.2	IP to PSTN	68
3.8.3	PSTN to IP	71
3.9 Nui	mber Filter	73
3.9.1	Whitelist	74
3.9.2	Blacklist	
3.9.3	Number Pool	
3.9.4	Filtering Rule	
3.10 Nu	mber Manipulation	
3.10.1	IP to PSTN CallerID	
3.10.2	IP to PSTN CalleeID	
3.10.3	IP to PSTN Original CalleeID	
3.10.4	PSTN to IP CallerID	
3.10.5	PSTN to IP CalleeID	
3.10.6	PSTN to IP Original CalleeID	
3.10.7	CallerID Pool	
	stem Tools	
3.11.1	Network	
3.11.2	Authorization	
3.11.3	Management	
3.11.4	IP Routing Table	
3.11.5	Access Control.	
3.11.6	Centralized Manage	
3.11.7	SIP Account Generator	
3.11.8	Configuration File	
3.11.9	Signaling Capture	
3.11.10		
3.11.11	Signaling Call Track	
-	PING Test	
	TRACERT Test	
	Modification Record	
	Backup & Upload	
	Factory Reset	
	Upgrade	
	Change Password	
	Device Lock	
	Restart	
••••••		
Chapter 4	4 Typical Applications	113
4.1 Apr	plication 1	113
4.1.1		
4.1.2		
4.1.3	Configurations for Branch B	
	plication 2	
	Configurations for Headquarters	
4.2.2	Configurations for Branches	
1.2.2		
Appendix	A Technical Specifications	131
Appendix	x B Troubleshooting	132
Appendix	c C ISDN Pending Cause to SIP Status Code	133
	x D Direction for CDR Use	



Ap	pendix E	Technical/sales	Support		136
AΡ		iecillical/sales	Support	•••••••••••••••••	130



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# **Revision History**

Version Date		Comments		
Version 1.6.5 2017-06		Initial publication.		

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



# **Chapter 1 Product Introduction**

Thank you for choosing Synway SMG Series Digital Gateway!

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

## **1.1 Typical Application**

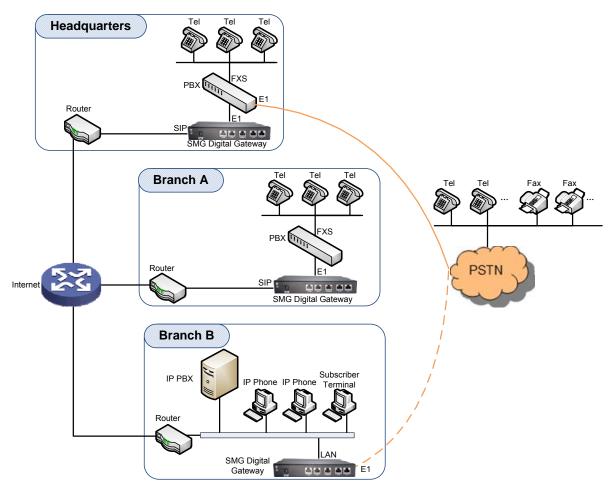


Figure 1-1 Typical Application



## 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			
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.729, G722, G723, iLBC, AMR, SILK(16K), OPUS(16K), SILK(8K), OPUS(8K)           DTMF Mode         RFC2833, SIP INFO, INBAND, RFC2833+Signaling, In-band+Signaling			
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			
Software Upgrade	Support of user interface, gateway service, kernel and firmware upgrades based on WEB			
Tracking Test	Support of Ping and Tracert tests based on WEB			



SysLog Type

Three options available: ERROR, WARNING, INFO

## **1.3 Hardware Description**

The SMG digital gateway integrates embedded LINUX system within the POWERPC+DSP hardware architecture. It has 1/2 E1/T1 ports and 2 100Mb/s Ethernet ports (LAN1 and LAN2) on the chassis.

See below figures for SMG2030L series appearance:



Holes

Figure 1-4 Left View

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

Interface	Description				
	Amount: 2				
	Type: RJ-45				
LAN	Bandwidth: 10/100Mbps				
	Self-Adaptive Bandwidth Supported				
	Auto MDI/MDIX Supported				
<b>F1T1</b>	Amount: 1/2				
E1/T1	Type: RJ-45				

Foot Bracket



	Amount: 1				
	Type: RS-232				
	Baud Rate: 115200 bps				
	Connector: RJ45 (See Figure 1-5 for signal definition)				
Console Port	Data Bits: 8 bits				
	Stop Bit: 1 bit				
	Parity Unsupported				
	Flow Control Unsupported				
External Power Supply					
Interface	Voltage: 12V, positive inside and negative outside; Current: ≥3A.				
Button	Description				
Reset Button	Restore the gateway to factory settings.				
Reset Button LED	Restore the gateway to factory settings.				
Reset Button	Restore the gateway to factory settings. Description				
Reset Button LED	Restore the gateway to factory settings.         Description         Indicates the power state. It lights up when the gateway starts up with the power				
Reset Button LED Power Indicator	Restore the gateway to factory settings.  Description Indicates the power state. It lights up when the gateway starts up with the power cord well connected.				
Reset Button LED Power Indicator Run Indicator	Restore the gateway to factory settings.         Description         Indicates the power state. It lights up when the gateway starts up with the power cord well connected.         Indicates the running status. For more details, refer to <u>1.4 Alarm Info</u> .				
Reset Button LED Power Indicator Run Indicator Alarm Indicator Link Indicator	Restore the gateway to factory settings.         Description         Indicates the power state. It lights up when the gateway starts up with the power cord well connected.         Indicates the running status. For more details, refer to <u>1.4 Alarm Info</u> .         Alarms the device malfunction. For more details, refer to <u>1.4 Alarm Info</u> .				
Reset Button LED Power Indicator Run Indicator Alarm Indicator	Restore the gateway to factory settings.         Description         Indicates the power state. It lights up when the gateway starts up with the power cord well connected.         Indicates the running status. For more details, refer to <u>1.4 Alarm Info</u> .         Alarms the device malfunction. For more details, refer to <u>1.4 Alarm Info</u> .         The green LED on the left of LAN, indicating the network connection status.				
Reset Button LED Power Indicator Run Indicator Alarm Indicator Link Indicator	Restore the gateway to factory settings.         Description         Indicates the power state. It lights up when the gateway starts up with the power cord well connected.         Indicates the running status. For more details, refer to <u>1.4 Alarm Info</u> .         Alarms the device malfunction. For more details, refer to <u>1.4 Alarm Info</u> .         The green LED on the left of LAN, indicating the network connection status.         The orange LED on the right of LAN, whose flashing tells data are being				

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.

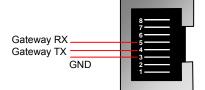


Figure 1-5 Console Port Signal Definition

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

## 1.4 Alarm Info

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

LED	State	Description
	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
- External 12V Power Adapter \*1
- Warranty Card \*1
- Installation Manual \*1

#### Step 2: Connect the power cord.

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

#### Step 3: Connect the network cable.

Step 4: 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.

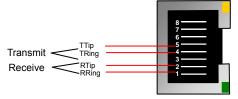


Figure 2-1 Pin Layout for E1 Interface

#### Step 5: 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>3.11.18 Change Password</u>. After changing the password, you are



required to log in again.

#### Step 6: 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>3.11.1 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.

#### Step 7: 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>3.11.20 Restart</u> for detailed instructions.

#### Step 8: Configure signaling protocol parameters.

Further configure the signaling protocol you set in Step 7. Different protocols are configured on different interfaces. See below 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>3.11.20 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>3.11.20 Restart</u> for detailed instructions.

#### Step 9: 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 10: 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 'SIP 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 'SIP 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 → IP→PSTN' for detailed instructions. Select the SIP trunk group set in Step 2 as 'Call Initiator' and the PCM trunk group set in Step 3 as 'Call Destination'. You may use the default values for the other configuration items.

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

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

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

#### Situation 2: PSTN $\rightarrow$ 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 'SIP 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 'SIP 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 → PSTN→IP' for detailed instructions. Select the PCM trunk group set in Step 4 as 'Call Initiator' and the SIP trunk group set in Step 3 as 'Call Destination'. You may use the default values for the other configuration items.

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

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

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

### **Special Instructions:**

- The chassis of the SMG digital gateway must be grounded for safety reasons, according to standard industry requirements. A simple way is earthing with the third pin on the plug or the grounding studs on the machine. No or improper grounding may cause instability in operation as well as decrease in lightning resistance.
- As the device will gradually heat up while being used, please maintain good ventilation to prevent sudden failure, ensuring that the ventilation holes (see Figure 1-4) are never jammed.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Otherwise it may lead to a drop in performance or unexpected errors.



# **Chapter 3 WEB Configuration**

## 3.1 System Login

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

Windows Securit	y X
Warning: This	.123.111 102 at SMG requires a username and password. server is requesting that your username and password be cure manner (basic authentication without a secure
	User name Password Remember my credentials
	OK Cancel

Figure 3-1 Login Interface

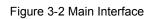
The gateway only serves one user, whose original username and password are both 'admin'. You can change the username and the password via 'System Tools  $\rightarrow$  Change Password' on the WEB interface. For detailed instructions, refer to <u>3.11.18 Change Password</u>.

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



8			System	Info	
(					
JS		LAN 1			
		MAC Address	80:7B:85:10:4D:BF		
		IP Address	192.168.1.101	255.255.255.0	192.168.1.254
		DNS Server	0.0.0.0		
		Receive Packets Transmit Packets	All:0 All:0	Error:0 Error:0	Drop:0 Drop:0
		Current Speed	Receive:0 B/s	Transmit0 B/s	Drop.0
		Work Mode	Disconnected	Transmico Dis	
*					
*		LAN 2			
		MAC Address	80:7B:85:10:4D:C0		
*		IP Address	201.123.111.147	255.255.255.0	201.123.111.254
*		DNS Server	0.0.0		
		Receive Packets	All:57670	Error:0	Drop:21
*		Transmit Packets	All:63008	Error:8	Drop:0
*		Current Speed	Receive:2.4 KB/s	Transmit:573 B/s	
		Work Mode	10Mb/s Full Duplex		
*			(The current mode for th adaptive.)	ie network card is non-	
*		Runtime	39m 17s		
		- Culturine	5511115		
		Operating Mode	ISDN(user)		
		CPU Usage Rate	77%		
		Current RTP Message Data	Packet Loss Rate in Reception:0.00%	Packet Lost in Recepti	ion:0 Total Transmit Packets:0
		DCMS Working Status	Not Enabled		
		Current Version			
		Serial Number	13479(2L)		
		WEB	1.6.5_2017041710		
		Gateway	1.6.5_2017041710		
		Uboot	2.0.7_201701		
		Kernel	#222 PREEMPT Wed Ja	an 4 10:58:42 CST 2017	
		Firmware	1		

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Refresh



## 3.2 Operation Info

Operation Info includes six parts: *System Info*, *PSTN Status*, *PCM Info*, *Call Monitor*, *Call Count* and *Warning Info* showing the current running status of the gateway. See Figure 3-3.

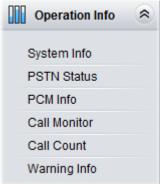


Figure 3-3 Operation Info



## 3.2.1 System Info

LAN 1			
MAC Address	80:7B:85:10:4D:BF	000000000000	100 100 1 07 1
IP Address	192.168.1.101	255.255.255.0	192.168.1.254
DNS Server	0.0.0.0	<b>F</b> 0	-
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	Disconnected		
LAN 2			
MAC Address	80:7B:85:10:4D:C0		
IP Address	201.123.111.147	255.255.255.0	201.123.111.254
DNS Server	0.0.0.0		
Receive Packets	All:57670	Error:0	Drop:21
Transmit Packets	All:63008	Error:8	Drop:0
Current Speed	Receive:2.4 KB/s	Transmit:573 B/s	
Work Mode	10Mb/s Full Duplex		
	(The current mode for th	e network card is non-	
	adaptive.)		
Runtime	39m 17s		
Operating Mode	ISDN(user)		
CPU Usage Rate	77%		
Current RTP Message Data	Packet Loss Rate in Reception:0.00%	Packet Lost in Rece	otion:0 Total Transmit Packets:0
DCMS Working Status	Not Enabled		
Current Version			
Serial Number	13479(2L)		
WEB	1.6.5_2017041710		
Gateway	1.6.5_2017041710		
Uboot	2.0.7_201701		
Kernel	#222 PREEMPT Wed Ja	an 4 10:58:42 CST 2017	
Firmware	1		

#### Figure 3-4 System Info Interface

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

Item	Description
MAC Address	MAC address of LAN 1 or LAN 2.
IP Address	The three parameters from left to right are IP address, subnet mask and default gateway of LAN 1 or LAN 2.
DNS Server	DNS server address of LAN 1 or LAN 2.



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Dessive Deskats	The amount of receiv	ve packets after the gateway's startup, including three								
Receive Packets	categories: All, Error an	d Drop.								
Transmit Destate	The amount of transr	nit 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 I	Mbps Half Duplex, 100 Mbps Full Duplex and Disconnected.								
	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								
On and the state	ISDN(User-side)	The current gateway is configured to be ISDN user-side								
Operating Mode		The current gateway is configured to be ISDN								
	ISDN(Network-side) network-side.									
	SS1	The current gateway is configured to be SS1.								
CPU Temperature	Display the real time ter	mperature of the CPU.								
CPU Usage Rate	Display the real time us	age rate of the CPU.								
Current RTP										
Message Data	Display the receiving ar	nd sending information of the current RTP data.								
DCMS Working										
Status	Display the connecting	status of the gateway and DCMS.								
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	ateway service.								
Uboot	Current version of Uboo	pt.								
Kernel	Current version of the s	ystem kernel on the gateway.								
Firmware	Current version of the fi	rmware on the gateway.								



## 3.2.2 PSTN Status

	d orgin	aling	Status							Frame Sy	nc				-	Signa	ling				Fa	aulty				Un	used	
	Co	lor																						ſ				
Voice Path Status	Idle	Ring	ing	Wait A	nswer	Di	ialing	Tall	king	Pending	) V	Vait M	essag	je	Loca	I Bloc	:k	Rem	note E	Block	В	oth B	lock	Ci	rcuit F	Reset	U	Jnusa
Icon 🕥 😨 🔞 🔇						<b>C</b>					6			<b>1</b>									R			63		
Statistics 59 0 0 0				0	1		0		0				0			0			0			0			0			
Time Slot No. 0	-	2	3	_	5 6	_	8	9	10		13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30
PCM 0																												
PCM 1	and the state of t						_	1000	-			-								_	_	_						
						_	-	-			_					******								-				

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

Figure 3-5 PSTN Status Interface for E1 Lines

	Sync & Sig	naling Statu	5				Frame Sy	nc			Sig	naling			Faulty	,			Unus	ed	
	C	olor																		[	
Voice Path Status	Idle	Ringing	Wait An:	swer	Dialing	Talking	Pending	Wait	Messag	je	Local E	lock	Remote	Block	Both	Block	Cir	rcuit R	eset	Unus	able
Icon			0		3	()	2		6		1		4		6	3		R		G	3
Statistics	45	0	0		0	1	0		0		0		0		(	D		0		(	0
Time Slot No.	(	_	2 3	4	5	6 7	8	9 10	11	12	13		15 16	17	18	19	20	21	22	23	24
Time Slot No.	(	0 1	2 3	4	5		teway1:					14	15 16	17	18	19	20	21	22	23	24
PCM 0						1			_	-	-	-		-	-	-			9		
PCM 0 PCM 1		Talkin	-																-		
16690038		Direct	tion:PST															-	•	-	
16597739		Direct	tion:PST :888611															-	-	-	

Figure 3-6 PSTN Status Interface for T1 Lines

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

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



E1/T1.										
State	Color	Description								
Frame Sync		Frame synchronization normal. The synchronization status is 0x0								
Faulty		status is 0x0. 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 bit1=1: short circuit								
		Other bits: reserved, all remain 0								
For the sign	naling tim	aling time slot, the channel states include:								
State	Color	Description								
Signaling		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. For ISDN, this state indicates 'TEI unassigned', 'assign awaiting TEI', 'establish awaiting TEI', 'TEI assigned',								
Faulty		'awaiting establishment 'or 'awaiting release'. For SS1, this state indicates 'time slot synchronization abnormal'.								
Faulty Unused		For SS1, this state indicates 'time slot synchronization								
Unused	er channe	For SS1, this state indicates 'time slot synchronization abnormal'. This state indicates the signaling time slot on this E1/T1 is not used								
Unused	er channe Icon	For SS1, this state indicates 'time slot synchronization abnormal'. This state indicates the signaling time slot on this E1/T1 is not used.								
<i>Unused</i> ● For the othe		For SS1, this state indicates 'time slot synchronization abnormal'. This state indicates the signaling time slot on this E1/T1 is not used. els, the channel states include:								
Unused For the othe State	lcon	For SS1, this state indicates 'time slot synchronization abnormal'. This state indicates the signaling time slot on this E1/T1 is not used. els, the channel states include: <b>Description</b>								



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	Local Block	1	The channel is blocked by the local application program and cannot receive incoming calls.
	Remote Block	4	The channel is blocked by the specific circuit/circuit group blocking messages sent from the remote PBX and cannot make outgoing calls.
	Both Block	8	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	Ø	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	2	The channel is in the pending state
	Dialing	<b>(-</b>	The channel is dialing.
	Wait Message	<b>6</b>	The channel is waiting for the message from remote PBX.
Statistics	The total amount	of the o	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 114 to the search box, and click the **Search** button, the gateway does a fuzzy search and locates that the ongoing call whose CalledID contains the character 114 occurs on Time Slot No. 1 of PCM 0.

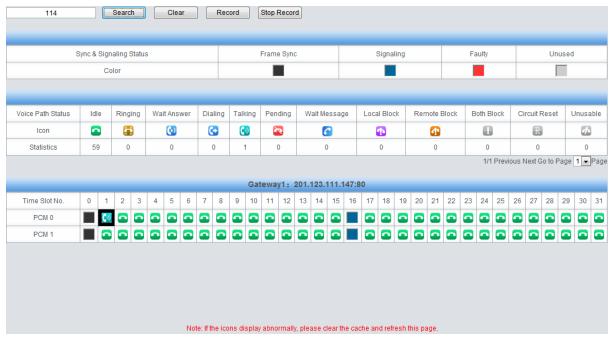


Figure 3-7 Search Calls

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



## 3.2.3 PCM Info

		PCM10	-
PCM Info			
Basic Frame Sync Loss	1		
Duration of Basic Frame Sync Loss Exceeds 100ms	1		
CAS Re-synchronize	0		
CRC Multiframe Sync Loss	0		
Remote Alarm Indicator	0		
Signal Alarm Indicator	0		
Alarm Signal of All-ones on TS16	0		
Signal Loss	1		
Remote MF Alarm	0		
Open Circuit	1		
Short Circuit	0		

Figure 3-8 PCM Info Interface

See Figure 3-8 for the PCM Info interface. It displays the detailed information of E1 lines, facilitating the check on whether the PCM line is stable as well as the troubleshooting. Select a PCM channel via the drop down list on the right top corner. The statistics counters will add 1 each time once the alarm occurs.

## 3.2.4 Call Monitor

				Monitoring Condition					
Monitore	ed CallerID	Monitored Cal	leeID Monitored	Remote Address	Monito	ring LAN Port	LAN1:201.123.111.102	-	set
he monitor f	eature does n	ot work, <u>click here</u> to dowr	nload and install the monitoring	olug-in.	1 Items	Total 50 Items/F	Page Previous Next Go to	Page 1 👻	1 Pages Tot
				Call Info					
				Cour IIIIo					
PCM No.	TS No.	Call Direction	Remote Address	Channel Status	CallerID	CalleelD	Start Time		Duration

Figure 3-9 Call Monitor Interface

See Figure 3-9 for the call monitor Interface. Here you can set a condition for call monitoring. For example, as shown in Figure 3-9, set the CalleeID 114 as the monitoring condition, and after you click the **Set** button, all the calls containing the CalleeID 114 will display in the Call Info list. The table below explains the items shown in Figure 3-9.

Item Description
------------------



Monitored CallerID,	
Monitored CalleelD,	Sets the condition for the call monitoring. You can set to monitor the calls by
Monitored Remote	CallerID, CalleeID or remote address.
Address	
Monitoring LAN Port	Selects the LAN port which is used to monitor the calls.
PCM No.	The number of the PCM, which starts from 0.
TS No.	PCM time slot number in the port.
	The direction of the monitored call, including two options: IP $\rightarrow$ PSTN and
Call Direction	PSTN→IP.
Remote Address	The remote address of the monitored call.
Channel Status	The status of the channel which the monitored call locates at.
CallerID	The CallerID of the monitored call.
CalleelD	The CalleeID of the monitored call.
Start Time	The start time of the monitored call.
Duration	The duration of the monitored call.

Click the icon in the channel status column, and you can monitor the call in real-time. If your computer is not installed with the monitoring plug-in, click the icon and you will see a prompt asking you to set the security level. Follow the instructions to configure the IE explorer: Open it and click 'Tools > Internet Options >Security Tab'; then click 'Custom Level' and enable 'Initialize and script ActiveX controls not marked as safe for scripting'. If there is a shadow showing under

the icon, such as ', it means the monitoring goes successful. Click the icon again to cancel the monitoring.

Note: If a channel has been monitored from the very beginning, the monitoring, even if not yet cancelled, will terminate once the channel is removed from the monitor list.



## 3.2.5 Call Count

						Ir	ncoming	SIP C	all Statistics								
SIP Index	Description	SIP Tr	unk Address	1 0	Current	Sun	n   C	onnec	tion Rate	Answeri	ng Rate	Ave	rage Call	Leng	th (s)	INVITE(Ti	mes/s)
1	default	201.	123.112.80		0	0		-	-	-	ā.		0			0	
						0	utaoina		all Statistics								
SIP Inde	x Descrip	tion	SIP Trunk Add	ress		Current		um	1	ction Rate		Answering	Rate		Average	Call Length	1 (S)
1	defau	K.A.C.S.	201.123.112			0		0		-				T	, norugo	0	
					-	-		-									
							PSTN	Call	Statistics								
Trunk No.	Signaling Type	Current Nur	mber of IP->PS	TN	Sum	Connect	ion Rate	Ans	wering Rate	Current	Current Number of PSTN->IP			Cor	nnection Rate	Answe	ring Rate
0	SS7-ISUP	0			0						0		0		1.75	-	77
1	SS7-ISUP	ISUP 0			0	0			-		0						
2	SS7-ISUP					-	-		-		0		0		( <del>-</del>		
3	SS7-ISUP		0		0	5	<b>.</b>			0			0		1.000		-
Total	-		0		0	5	52				0		0		1		-
						Statisti	cs on IP	->PST	N Release	Cause							
Release C	ause Normal D	isconnection	Cancelled	Busy	No	Answer	Route F				nallocate	d Number	Reject	ted	Unspecified	Failed	Others
Amoun	t	0	0	0		0	0		0		0	<u></u>	0		0	0	0
Percenta	ige	-	-	575									-			-	
			1										1.				
						Statisti	cs on PS	STN->	IP Release	Cause							
Relea	se Reason	Norm	al Disconnectio	in	1	Cancelle	d B	usy	No Answe	r R	oute Faile	d	No Idle	Reso	ource	Failed	Others
N	umber		0			0		0	0		0			0		0	0
Per	centage								-					-			-
						F	Reset		Downlo	ad							
				N	ote: Ple	ase do no	t reset th	e Call S	Statistics if the	ere is an on	oing call!						

Figure 3-10 Call Count Interface

See Figure 3-10 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 **Reset** to count the call information again, click **Download** to download all the call logs and ISDN logs. The table below explains the items shown in Figure 3-10.

ltem	Description					
SIP Index	The index of the SIP trunk.					
Description	More information about each SIP trunk group.					
SIP Trunk Address	Address of the SIP trunk, i.e. the IP address or domain name of the remote SIP					
SIF TTUIK AUDIESS	terminal which will establish a call conversation with the gateway.					
Current	The number of the current incoming/outgoing SIP calls.					
Sum	The total number of the incoming SIP calls/ outgoing SIP calls/ IP $ ightarrow$ PSTN calls/					
Sum	$PSTN \rightarrow IP$ calls.					
Composition Data	The percentage of successful calls to total calls by all method. The call methods					
Connection Rate	include SIP Incoming Call, SIP Outgoing Call, IP $\rightarrow$ PSTN call and PSTN $\rightarrow$ IP call.					
Anowaring Bata	The percentage of answered calls to total calls by all methods. The call methods					
Answering Rate	include SIP Incoming Call, SIP Outgoing Call, IP $\rightarrow$ PSTN call and PSTN $\rightarrow$ IP call.					
Average Call Length	e Call Length The average call length for all connected calls.					



INVITE	The number of the invite messages received per second.			
Trunk No.	The number of the PCM trunk, numbered from 0			
Ciama linas Trana	The signaling protocol applied on the digital trunk, including: ISDN User Side, ISDN			
Signaling Type	Network Side and SS1.			
Current Number of				
IP→ PSTN	The number of current calls from IP to PSTN.			
Current Number of				
PSTN → IP	The number of current calls from PSTN to IP.			
Total	Total number and connection rate of calls on all available trunks			
Release Cause	Reason to release the call.			
Normal				
Disconnection	Total number of the calls which are normally cleared.			
Cancelled	Total number of the calls which are cancelled by the calling party.			
Duov	Total number of the calls which fail as the called party has been occupied and			
Busy	replies a busy message.			
No Answer	Total number of the calls which fail as the called party does not pick up the call in a			
NO Aliswei	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.			
No Idle Resource	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.			
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			
Unspecified	unspecified.			
	Total number of the calls which fail as the called party number does not conform to			
Failed	the number-receiving rule or for relative reasons.			
Others	Total number of the calls which fail due to other unknown reasons.			
<b>Percentage</b> The percentage of the calls with a release cause to total calls.				



## 3.2.6 Warning Info

	Warning Log	
2018-02-20 20:17:02	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:19:09	WARNING:State of E1's link 0 changes to 0x0001, Please Check!	
2018-02-20 20:19:44	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:20:33	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	1
2018-02-20 20:21:21	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:21:25	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	E
2018-02-20 20:24:47	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:25:27	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:28:06	WARNING:State of E1's link 0 changes to 0x0001, Please Check!	
2018-02-20 20:28:06	WARNING:State of E1's link 0 changes to 0x0001, Please Check!	
2018-02-20 20:30:45	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:31:24	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:38:34	WARNING:State of E1's link 0 changes to 0x0001, Please Check!	
2018-02-20 20:41:39	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:42:19	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:43:02	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:44:55	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 20:45:39	WARNING:State of E1's link 0 changes to 0x0001, Please Check!	
2018-02-20 20:49:39	WARNING:State of E1's link 0 changes to 0x0011, Please Check!	
2018-02-20 21:14:24	ERROR:Route failed, please check the configuration of 'PSTN->IP' route (srcgroup:0)!	
2018-02-20 21:14:57	ERROR:Route failed, please check the configuration of 'PSTN->IP' route (srcgroup:0)!	-
Note: Only the	Download latest 100 pieces of warning information will be displayed. To check all the information, please	
	click the Download button.	
	Figure 3-11 Warning Information Interface	

Figure 3-11 Warning Information Interface

See Figure 3-11 for the Warning Information interface. All the warning information will be output and displayed on this interface.

## 3.3 SIP Settings

SIP Settings includes five parts: *SIP*, *SIP Trunk*, *SIP Register*, *SIP Account*, *SIP Trunk Group* and *Media*. See Figure 3-12. *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 3-12 SIP Settings



## 3.3.1 SIP Settings



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SIP Settings	
SIP Address of WAN	LAN 2: 201.123.111.20
SIP Signaling Port	5060
Send 183 Message	☑ Enable
Called Number Prefix for 180 Reply (Up to 5 are	
Allowed, Separated by '.') Send 100rel	Enable
Soft-switch to be Connected	
Send 183 Delay Time(ms)	0
183 Send Delay Mode	Mode 1
Hide CallerID	Not Hidden 💌
Obtain CallerID from	Username of From Field 🔹
Obtain/Send CalleelD from	'Request' Field
Asserted Identity Mode	Disable
Send/Obtain Redirecting Number/Original CalleeID from Diversion Field	Enable
NAT Traversal	Enable
SIP Transport Protocol	UDP
SIP Encryption	Enable
RTP Encryption	Enable
RTP Self-adaption	Enable
UDP Header Checksum	Z Enable
Rport	Enable
Filter Out Fake Calls (CallerID is the same as CalleeID)	Enable
Auto Reply of Source Address	Enable
DSCP	Enable
Calls from SIP Trunk Address only	Enable
Switch Signal Port if SIP Registration Failed	Enable
Hang up upon Call Time-out	Enable
Working Period	☑24 Hours
Session Timer	Enable
Early Media	Enable
Early Session	Enable
Not Wait ACK after Sending 200 OK	Enable
The Percentage of Registration Message Sending Cycle to Period of Validity(%)	70
Maximum Wait Answer Time(s)	60
Maximum Wait RTP Time(s)	0
Maximum Wait PSTN Resource Time(ms)	5000
Switch Network Port by Packet Loss Rate	Enable
Add Content to To Field in INVITE Message	©Yes ⊚No
UserAgent Field	
Save Reset Note: Only one SIP Trunk can be configured and its "Local Network Por feature "Switch Network Port by Packet Loss Rate" is enabled.	t" should be set to "Any Lan" once the

#### Figure 3-13 SIP Settings Interface

See Figure 3-13 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>3.11.20 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-13.

Item	Description			
SIP Address of WAN	IP address of WAN for SIP signaling, using LAN 1 by default.			
	Monitoring port of SIP signaling. Range of value: 2000~65535, with the default			
CID Cismoling Dout	value of 5060.			
SIP Signaling Port	Note: The value range of this configuration item and that of the RTP port set in			
	Media Settings cannot be overlapped.			
	Sets whether to send the 183 message instead of 180 to respond to the ringing			
Send 183 Message	tone when the SIP end serves as the called party. By default this feature is			
	enabled.			
	Once the feature "Send 183 Message" is enabled, the gateway will reply the			
	180 message to those calls which have the calleeID with the designated prefix;			
Called Number Prefix	otherwise, it will reply the 183 message. By default, the value is null, that is,			
for 180 Reply	replying the 183 message to all calls. Up to 5 prefixes are allowed to fill in this			
	item, which are separated by ':'			
Send 100rel	Sets whether to send the 100rel field, with the default value of disabled.			
Soft-switch to be	Sets the soft telephony device which will be connected to the gateway, including			
Connected	Others and VOS two options, with the default value of Others.			
	Sets the delay time for sending the 183 message. Range of value: 0~10000,			
Sand 192 Dalay Time	with the default value of 0.			
Send 183 Delay Time	Note: It is valid only when the configuration item Soft-switch to be Connected is			
	set to VOS.			
	Sets the delay mode for sending the 183 message, including two options: Mode			
	1 and Mode 2, with the default value of Mode 1.			
	Mode 1: The PSTN side will send the IAM message and wait for the ACM			
	message once it receives an Invite message from vos. If the ACM message isn't			
	received within the preset-time, the SIP side will reply the 183 message; if the			
	PSTN side receives the ACM message later, the SIP side will send the 183			
183 Send Delay Mode	message once again. If the ACM message is received within the preset-time,			
	the SIP side will reply the 183 message only once.			
	Mode 2: The SIP side will send the 183 message only once upon timeout; it			
	won't send the 183 message if the ACM message is received within the			
	overtime.			
	Note: It is valid only when the configuration item Soft-switch to be Connected is			
	set to VOS.			
Hide CallerID	Sets whether to hide the CallerID, with the default value of Not Hidden.			
	There are two optional ways to obtain the calling party number: from Username			
Obtain CallerID from	of "From" Field or from Displayname of "From" Field. The default value is from			
	Username of "From" Field.			



Obtain/Cand CallealD	There are five entired wave to obtain an acad the called party numbers from
Obtain/Send CalleeID	There are two optional ways to obtain or send the called party number: from
from	"To" Field or from "Request" Field. The default value is from "Request" Field.
	Sets whether to have the invite message include some header information, two
Asserted Identity Mode	options available now: P-Asserted-Identity and P-Preferred-Identity. The default
	value is disabled.
	Once this feature is enabled, the callerID in the From field will not be
Number in From Field	manipulated, with the default value of <i>disabled</i> .
not Manipulated	Note: It is valid only when the configuration item Asserted Identity Mode is
	enabled.
Send/Obtain	
Redirecting	Sets whether to enable the feature of sending or obtaining the Redirecting
Number/Original	Number/Original CalleeID from Diversion Field. By default, the feature is
CalleeID from Diversion	disabled.
Field	
NAT Traversal,	Sets whether to enable the feature of NAT Traversal. By default, the feature is
Traversal Type	disabled. There is only one optional traversal type: Port Mapping.
	The mapping address of the LAN1 and LAN2 in case the NAT traversal is
	enabled. If the port mapping is selected as the traversal type, you are required
LAN1 Mapping Address,	to set the mapping address on the router and fill in the corresponding
LAN2 Mapping Address	information here as well. By default, only the IP address need be filled in, and
	the port value is just the same as the SIP signaling port.
	Once this feature is enabled, the gateway will be enforced to use the mapping
Always Use Mapping	address set in the above configuration item to initiate calls. By default it is
Address	disabled.
	There are two modes UDP and TCP available for running the SIP protocol. The
SIP Transport Protocol	default value is <i>UDP</i> .
	Once this feature is enabled, you can encrypt the SIP signal following selecting
SIP Encryption	an encryption criterion and setting a key. By default it is <i>disabled</i> .
	The criterion used to encrypt the SIP signal. At present only VOS1.1 is
Encryption Criterion	supported.
Кеу	The key to encrypt the SIP signal.
,	Once this feature is enabled, you can encrypt the RTP package. By default it is
RTP Encryption	disabled.
	When this feature is enabled, the RTP reception address or port carried by the
	signaling message from the remote end, if not consistent with the actual state,
RTP Self-adaption	will be updated to the actual RTP reception address or port. By default, this
	feature is <i>disabled</i> .
UDP Header Checksum	When this feature is enabled, the gateway will automatically calculate the check sum of the UDP header during RTP transmission.
	When this feature is enabled, a corresponding Rport field will be added to the
Rport	Via message of SIP. By default, it is <i>disabled</i> .
	via message of SIF. by default, it is disabled.



Filter Out Fake Calls				
(CallerID is the same as	Once this feature is enabled, those outgoing calls from PSTN whose callerID is			
CalleelD)	the same as calleeID will be forbidden. The default value is <i>disabled</i> .			
Auto Reply of Source	Once this feature is enabled, the gateway will reply the source address in the			
Address	invite message. The default value is <i>disabled</i> .			
	Sets whether to enable the DSCP differentiated services code point. By default,			
DSCP	it is <i>disabled</i> .			
	Sets the priority of the voice media for DSCP. The voice media with a bigger			
Voice Media	value has a higher priority. The value range is 0~63, with the default value of 46.			
	Sets the priority of the signal control for DSCP. The signal control with a bigger			
Signal Control	value has a higher priority. The value range is 0~63, with the default value of 26.			
Calls from SIP Trunk	Once this feature is enabled, the gateway will only accept the calls from the IP			
Address only	addresses set in SIP Settings → SIP Trunk. By default, it is <i>disabled</i> .			
Switch Signal Port if SIP	If the SIP registration fails, the SIP signaling port N will switch to N+1 for a new			
Registration Failed	registration. It will continue until the registration succeeds.			
Hang up upon Call	Sets whether to enable the feature to hang up the call once it is time-out, with			
Time-out	the default value of <i>No</i> ,			
Maximum Call Overtime	Sets the maximum overtime for a call. Calculated by minute.			
	The work period for the gateway, You can specify a certain period for the			
Working Period, Period	gateway to make calls. By default, the gateway is allowed to make calls any			
	time in the day (24 Hours).			
	Sets whether to enable the session refresh feature, with the default value of			
Session Timer	disabled. Once this feature is enabled, you are required to enter the minimum			
	time and the timeout value.			
	Sets the minimum time for refreshing the session. Value of range: 90~65535,			
Minimum Time	with the default value of 150.			
	Sets the timeout value for refreshing the session. The value cannot be less than			
Timeout	that of Minimum Time, with the default value of 600.			
Early Martin	Once this feature is enabled, the P-Early-Media field will be included in the			
Early Media	Invite message. The default value is <i>disabled</i> .			
Forthe Constant	Once this feature is enabled, the early-session field will be included in the Invite			
Early Session	message. The default value is <i>disabled.</i>			
Not Wait ACK after	Once this feature is enabled, the gateway does not need to wait the ACK			
Sending 200 OK	message after sending the 200OK message. The default value is <i>disabled</i> .			
The Percentage of				
Registration Message	Sets the percentage of the sending cycle of the SIP registration message to the			
Sending Cycle to Period	validity period. Value of range: 1~200, with the default value of 70.			
of Validity				
	Sets the maximum time for the SIP channel to wait for the answer from the			
Maximum Wait Answer	called party of the outgoing call it initiates. If the call is not answered within the			
Time	specified time period, it will be canceled by the channel automatically. The			
	default value is 60, calculated by s.			



	Sets the maximum time for the SIP channel to wait for the RTP packet. If no
Maximum Wait RTP	RTP packet is received within the specified time period, the channel will enter
Time	the pending state automatically and release the call. The default value is 0,
	calculated by s.
Maximum Wait PSTN	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
Resource Time	value range is 0~10000, calculated by ms, with the default value of 5000.
Curitab Naturally Davit bu	Once this feature is enabled, the gateway will switch to other available network
Switch Network Port by Packet Loss Rate	port once the RTP packet loss rate gets larger than the set value. The default
Packet Loss Rate	value is <i>disabled</i> .
	Sets the RTP packet loss rate which is used as the judgment condition to switch
RTP Packet Loss Rate	the network port, with the default value of 5.
Add Content to To Field	Once this feature is enabled, "user=phone" will be added to the TO field of the
in INVITE Message	INVITE message. The default value is <i>disabled</i> .
Add Content	Sets the content to add to the TO field.
Hoord word Field	Sets the content of the UserAgent field. Currently, it only supports the English
UserAgent Field	uppercase and lowercase letters.

## 3.3.2 SIP Trunk

		_		_	SIP	Trunk			
Check	Index	Description	Remote Address	Remote Port	Local Network Port	Transport Protocol	Outgoing Voice Resource	Incoming Voice Resource	Voice C
	1	default	201.123.111.21	5067	LAN 1(201.123.111.23)	UDP	128	128	G711A,G711U,G
				m					•
Check A		Uncheck All	Inverse	Delete	Clear All				Add New

#### Figure 3-14 SIP Trunk Settings Interface

See Figure 3-14 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 3-15 for the SIP trunk adding interface.



SIP Trunk				
Index: 0 -				
Description: default				
Remote Address:				
Remote Port: 5060				
Local Network Port: LAN1(201.123.111 -				
Display CODEC				
Transport Protocol: UDP -				
Outgoing Voice Resource: 128				
Incoming Voice Resource: 128				
Working Period: 24 Hours				
Save Close				

Figure 3-15 Add New SIP Trunk

The table below explains the items shown in Figure 3-15.

Item	Description				
Index	The unique index of each SIP trunk.				
Description	More information about each SIP trunk group.				
Remote Address	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.				
Local Network Port	The network port where the SIP trunk locates.				
Transport Drata and	SIP transport protocol, providing two modes UDP and TCP. The default value is				
Transport Protocol	UDP.				
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					
Resource	trunk to the gateway.				



Working Period, Period	The work period for the gateway, You can specify a certain period for the gateway to make calls. By default, the gateway is allowed to make calls any time in the day (24 Hours).				
	Supported COE	DECs and their corresponding priorities for the SIP trunk to establish			
	a call conversat	tion. The table below explains the sub-items:			
	Sub-item	Description			
	Priority	Priority for choosing the CODEC in an SIP conversation. The			
		smaller the value is, the higher the priority will be.			
CODEC	-	Seven optional CODECs are supported: G711A, G711U, G729			
	CODEC	G722, G723, iLBC, AMR, SILK(16K), OPUS(16K), SILK(8K),			
		OPUS(8K).			
	See 3.3.6 Media	a Settings for the detailed parameters for each CODEC.			
	The default CO	DDEC for the SIP trunk is the same as that set in 3.3.6 Media			
	<u>Settings</u> .				

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

Click *Modify* in Figure 3-14 to modify a SIP trunk. See Figure 3-16 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					
Description: default					
Remote Address: 201.123.111.21					
Remote Port: 5067					
Local Network Port: LAN1(201.123.111	•				
Display CODEC					
Transport Protocol: UDP 👻					
Outgoing Voice Resource: 128					
Incoming Voice Resource: 128					
Working Period: 24 Hours					
Save Close					

Figure 3-16 Modify SIP Trunk

To delete a SIP trunk, check the checkbox before the corresponding index in Figure 3-14 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 3-14.

# 3.3.3 SIP Register

SIP Register	Operation Info	*		
SIP Trunk Add New SIP Register	VolP	*		No available SIP register!
SIP Register	SIP			-
3IP Account	SIP Trunk			Add New
	SIP Register		•)	
SIP Trunk Group	SIP Account			
	SIP Trunk Group			
Jedia Jedia	Media			

Figure 3-17 SIP Register Configuration Interface

See Figure 3-17 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 3-18.



SIP Register					
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 3-18 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.
Password	Registration password of the gateway. To register the gateway to the SIP server, both
Password	configuration items Username and Password should be filled in.
Register Address	Address of the SIP server to which the SIP trunk is registered.
Register Port	The signaling port of the SIP trunk.
Domain Name	Domain name of the gateway used for SIP registry.



Register Expires	Validity period of the SIP registry. Once the registry is overdue, the gateway should be registered again. Range of value: 10~3600, calculated by s, with the default value of 3600.
IMS Network	Once this feature is enabled, the gateway will send signaling messages to the corresponding externally bound address and port when it registers to the server. Only when this feature is <i>enabled</i> will these items <b>Externally Bound Address</b> , <b>Externally Bound Port</b> and <b>Authentication Username</b> be shown.
Externally Bound Address	Externally bound IP address for registration.
Externally Bound Port	Externally bound port for registration.
Authentication Username	Authentication username for registration.

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

						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	123	201.123.115.107	5060	-	3600	Failed	No	
					III					
Check Al		Uncheck All	Inverse	E Delete	Clear All					Additiew

#### Figure 3-19 SIP Register Information List

Click *Modify* in Figure 3-19 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:	123				
Password:	•••				
Register Address:	201.123.115.107				
Register Port:	5060				
Domain Name:					
Register Expires (s):	3600				
IMS Network:	No				
Save	Close				

Figure 3-20 SIP Register Modification Interface

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

## 3.3.4 SIP Account

				SIP Account				
Check	Index	SIP Trunk No.	Username	Authentication Username	Register Expires (s)	Register Status	Description	Modify
	0	0	120		3600	Failed	default	
Check All	Unche	ck All Inverse	E Delete	Clear All				Add Net

Figure 3-21 SIP Account Settings Interface

See Figure 3-21 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 3-22 for the SIP account adding interface.



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

Figure 3-22 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.
	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.
Decouvered	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 3-21 to modify a SIP account. See Figure 3-23 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:	120			
Password:	•••			
Register Expires (s):	3600			
Description:	default			
Save	Close			

Figure 3-23 Modify SIP Account

To delete a SIP account, check the checkbox before the corresponding index in Figure 3-21 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 3-21.

# 3.3.5 SIP Trunk Group

				SIP Trunk Group			
Check	Index	SIP Trunks	SIP Trunk Select Mode	Outgoing Call Restriction	Incoming Call Restriction	Description	Modify
	0	0	Increase	No	Yes	default	
Check All	Unchec	k All 🚊 🛛 Inverse		lear All			Add New
Items Total	20 Items/Pa	ge 1/1 First Previo	ous Next Last Go to Page 1	1 Pages Total			

Figure 3-24 SIP Trunk Group Settings Interface

See Figure 3-24 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 3-25 for the SIP trunk group adding interface.



Index:	1
Description:	default
SIP Trunk Select Mode:	Increase
Outgoing Call Restriction	No
Incoming Call Restriction	No
SIP Trunks:	Check All
0	<b>1</b>

Figure 3-25 Add New SIP Trunk Group

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

ltem	Description		
Index	The unique index of each SIP trunk group, which is mainly used in t		
Index	of routing rules and number manipulation rules to correspond to SIP trunk groups.		
Description	More information abou	t each SIP trunk group.	
	When the SIP trunk gr	roup receives a call, it will choose a SIP trunk based on the	
	select mode set by th	is configuration item to ring. The optional values and their	
	corresponding meaning	gs are described in the table below.	
	Option	Description	
	1	Search for an idle SIP trunk in the ascending order of the	
	Increase	SIP trunk number, starting from the minimum.	
SIP Trunk Select	Deereese	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.	
Outroing/Inconstruct	Sets whether to restric	t the number of channels for the outgoing/incoming calls, with	
Outgoing/Incoming	the default value of No	b. If you select 'Yes', you are required to input the number of	
Call Restriction	restricted channels.		
	The SIP trunks in the S	SIP trunk group. If the checkbox before a SIP trunk is grey, it	
SIP Trunks	indicates that the SIP t	runk has been occupied. The ticked SIP trunks herein will be	
	displayed in the colum	n 'SIP Trunks' in Figure 3-24.	

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



settings.

Click *Modify* in Figure 3-24 to modify a SIP trunk group. See Figure 3-26 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	y SIP Trunk Group
Index:	0
Description:	default
SIP Trunk Select Mode:	Increase
Outgoing Call Restriction	No
Incoming Call Restriction	No
SIP Trunks:	Check All
☑ 0	
Save	Cancel

Figure 3-26 Modify SIP Trunk Group

To delete a SIP trunk group, check the checkbox before the corresponding index in Figure 3-24 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 3-24.



### 3.3.6 Media Settings

	Media Paran	neters	
DTMF Tran	nsmit Mode	RFC2833	•
RFC2833	Payload	101	
RTP Port F	Range	6000,10000	
Silence Su	ppression	Disable	_
Noise Red			
Noise Rec	uction	Enable	•
JitterMode		Static Mode	•
JitterBuffer	(ms)	100	
JitterUnde	rrunLead(ms)	100	
JitterOvern	unLead(ms)	50	
Voice Gain	Output from IP(dB)	0	
CODEC Setting Gateway Negotiation Coding Sequence	ult Priority	•	
Priority 1 2 3 4 5 6 7 8 9 10 11	CODEC G711A • G711U • G729 • G722 • G722 • G723 • iLBC • AMR • SILK(16K) • OPUS(16K) • SILK(8K) • OPUS(8K) •	Packing Time(ms) 20 20 20 20 20 30 20	Bit Rate (kbps) 64 64 8 64 63 15.2 12.20 20 20 12 12 12 12 12 12 12 12
	Save	Reset	

Figure 3-27 Media Settings Interface

See Figure 3-27 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>3.11.20 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-27.



# Synway Information Engineering Co., Ltd

Item	Description
DTMF Transmit	Sets the mode for the IP channel to send DTMF signals. The optional values are
Mode	RFC2833, In-band, Signaling, RFC2833+Signaling and In-band+Signaling, with the
wode	default value of <i>RFC2833</i> .
RFC2833 Payload	Payload of the RFC2833 formatted DTMF signals on the IP channel. Range of
KFC2055 Fayloau	value: 90~127, with the default value of <i>101</i> .
	Supported RTP port range for the IP end to establish a call conversation. Range of
RTP Port Range	value: 5000~60000, with the lower limit of 6000 and the upper limit of 10000 and the
	difference between larger than 512.
	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 Enable and Disable, with
Suppression	the default value of <i>Disable</i> .
	Note: When G723 is selected as CODEC, this configuration setting will turn to
	Enable automatically.
Noise Reduction	Once this feature is enabled, the volume of the noise accompanied with the line will
	be reduced automatically. The default setting is <i>Enable</i> .
JitterMode	Sets the working mode of JitterBuffer. The optional values are Static Mode and
Jillermode	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
	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 receive packets upon accepting packets later than
JitterUnderrunLead	the expected value set in JitterBuffer Item. Rnage of value: 0~280, calculated by
Siller Onder an Lead	ms, with the default value of 100,
	Note: Only when JitterMode is to Static Mode 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:
SillerOverrunzeau	0~280, calculated by ms, with the default value of <i>50</i> ,
	Note: Only when JitterMode is to Static Mode 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,
Jillenwin	with the default value of 80.
	Note: Only when JitterMode is to Adaptive Mode 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.
JillerDecreaseRalio	Rnage of value: 0~100, with the default value of 50,
	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 <i>30</i> ,
	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.				
	Sets CODECs for the IP end to establish a call conversation. The table be				
	explains the sub	b-items:			
	Sub-item	Sub-item Description			
	Gateway Negotiation Coding Sequence		ce, including two options: <i>Default Priority</i> , with the default value of		
	Priority		ODEC in an SIP conversation. The		
	CODEC		re supported: G711A, G711U, G729, SILK(16K), OPUS(16K), SILK(8K),		
	Packing Time	Time interval for packing ar	n RTP packet, calculated by ms.		
	Bit Rate	The number of thousand bi are conveyed per second.	ts (excluding the packet header) that		
	-		pported and ordered G711A, G711U, OPUS(16K), SILK(8K), OPUS(8K) by		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim	723, iLBC, AMR, SILK(16K), gh to low. The CODECs set he trunks. ne and bit rate supported by di	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim	723, iLBC, AMR, SILK(16K), h to low. The CODECs set he trunks. he and bit rate supported by di alues in bold face are the defa	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values.		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b>	723, iLBC, AMR, SILK(16K), th to low. The CODECs set he trunks. he and bit rate supported by di alues in bold face are the defa <b>Packing Time (ms)</b>	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va	723, iLBC, AMR, SILK(16K), yh to low. The CODECs set he trunks. he and bit rate supported by di alues in bold face are the defa <b>Packing Time (ms)</b> 10 / <b>20</b> / 30 / 40 / 50 / 60	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps)		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b> <i>G711A</i>	723, iLBC, AMR, SILK(16K), th to low. The CODECs set he trunks. he and bit rate supported by di alues in bold face are the defa <b>Packing Time (ms)</b>	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps) 64		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b> <i>G711A</i> <i>G711U</i>	723, iLBC, AMR, SILK(16K), h to low. The CODECs set he trunks. he and bit rate supported by di alues in bold face are the defa <b>Packing Time (ms)</b> 10 / <b>20</b> / 30 / 40 / 50 / 60 10 / <b>20</b> / 30 / 40 / 50 / 60	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps) 64 64		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b> <i>G711A</i> <i>G711U</i> <i>G729</i>	723, iLBC, AMR, SILK(16K), the low. The CODECs set he trunks. a and bit rate supported by di alues in bold face are the defa <b>Packing Time (ms)</b> 10 / <b>20</b> / 30 / 40 / 50 / 60 10 / <b>20</b> / 30 / 40 / 50 / 60	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps) 64 64 8		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b> <i>G711A</i> <i>G711U</i> <i>G729</i> G722	<ul> <li>723, iLBC, AMR, SILK(16K), yh to low. The CODECs set he trunks.</li> <li>and bit rate supported by dialues in bold face are the defa</li> <li>Packing Time (ms)</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> </ul>	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps) 64 64 64 8 64		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b> <i>G711A</i> <i>G711U</i> <i>G729</i> G722	<ul> <li>723, iLBC, AMR, SILK(16K), yh to low. The CODECs set he trunks.</li> <li>and bit rate supported by di alues in bold face are the defa</li> <li>Packing Time (ms)</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> </ul>	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps) 64 64 64 64 5.3 / 6.3		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b> <i>G711A</i> <i>G729</i> G722 <i>G723</i>	<ul> <li>723, iLBC, AMR, SILK(16K), if to low. The CODECs set he trunks.</li> <li>and bit rate supported by dialues in bold face are the defa</li> <li>Packing Time (ms)</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40</li> <li>30 / 60</li> <li>20 / 40</li> </ul>	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps) 64 64 64 8 64 5.3 / 6.3 15.2		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b> <i>G711A</i> <i>G729</i> G722 <i>G723</i>	<ul> <li>723, iLBC, AMR, SILK(16K), yh to low. The CODECs set he trunks.</li> <li>and bit rate supported by di alues in bold face are the defa</li> <li>Packing Time (ms)</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40</li> <li>30 / 60</li> <li>20 / 40</li> <li>30</li> </ul>	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps) 64 64 64 8 64 5.3 / 6.3 15.2 13.3 13.3 / 15.2 4.75 / 5.15 / 5.90 / 6.70 / 7.40 /		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b> <i>G711A</i> <i>G711U</i> <i>G729</i> G722 <i>G723</i> <i>iLBC</i> AMR	<ul> <li>723, iLBC, AMR, SILK(16K), if to low. The CODECs set he trunks.</li> <li>and bit rate supported by dialues in bold face are the defa</li> <li>Packing Time (ms)</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>20 / 40</li> <li>30</li> <li>60</li> <li>20 / 40 / 60</li> </ul>	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps) 64 64 64 64 64 5.3 / 6.3 15.2 13.3 13.3 / 15.2 4.75 / 5.15 / 5.90 / 6.70 / 7.40 / 7.95 / 10.20 / 12.20		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b> G711A G729 G722 G723 <i>iLBC</i> AMR <i>SILK(16K)</i>	<ul> <li>723, iLBC, AMR, SILK(16K), if to low. The CODECs set he trunks.</li> <li>and bit rate supported by dialues in bold face are the defa</li> <li>Packing Time (ms)</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>20 / 40</li> <li>30</li> <li>60</li> <li>20 / 40 / 60</li> <li>20 / 40 / 60</li> </ul>	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps) 64 64 64 8 64 5.3 / 6.3 15.2 13.3 13.3 / 15.2 4.75 / 5.15 / 5.90 / 6.70 / 7.40 / 7.95 / 10.20 / 12.20 20		
CODEC Setting	G729, G722, G priority from hig new added SIP The packing tim below. Those va <b>COEDC</b> <i>G711A</i> <i>G711U</i> <i>G729</i> G722 <i>G723</i> <i>iLBC</i> AMR	<ul> <li>723, iLBC, AMR, SILK(16K), if to low. The CODECs set he trunks.</li> <li>and bit rate supported by dialues in bold face are the defa</li> <li>Packing Time (ms)</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>10 / 20 / 30 / 40 / 50 / 60</li> <li>20 / 40</li> <li>30</li> <li>60</li> <li>20 / 40 / 60</li> <li>20 / 40 / 60</li> </ul>	OPUS(16K), SILK(8K), OPUS(8K) by ere will be the default CODEC for the fferent CODECs are listed in the table ult values. Bit Rate (kbps) 64 64 64 64 64 5.3 / 6.3 15.2 13.3 13.3 / 15.2 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 *PSTN Forwarding*. See Figure 3-28.



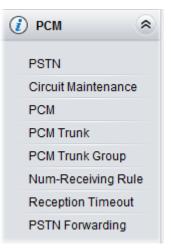


Figure 3-28 PCM Settings



## 3.4.1 PSTN

 PSTN Configuration	
Interface	E1 💌
Encoding Format	A-law
Echo Canceller	Enable
Busy Tone Detection	Enable
Frequency 1(Hz)	450
Frequency 2(Hz)	0
Cycle(ms)	700
Ignore Busy Tone during Call	Enable
Ringback Tone for PSTN->IP call	Enable
PSTN Call Barring	Enable
ISDN 01 Message Contains Progress Indicator	0x82
Ringback Tone Volume (dB)	-25
Voice Gain Output from PSTN (dB)	0
Hot Back-up for E1	✓Enable
Gateway IP for Hot Back-up	
Limited Length of E1 Outgoing CalleeID	0
Time Limit for E1 Outgoing Calls per Month	Enable
Mode Selection	By Minute
Time Limit (min)	100000
PSTN Call Forwarding	Disable
Save Reset	

Figure 3-29 PSTN Settings Interface

See Figure 3-29 for the PSTN Settings interface. The table below explains the items shown in the above figure.



# Synway Information Engineering Co., Ltd

Item	Description
Interfece	Actual type of the line connected with the E1/T1 interface on the gateway.
Interface	Currently, only E1/T1 is supported.
Encoding Format	Sets the voice data encoding format for the voice channels on the digital trunk.
Encoding Format	The optional values are <i>A-law</i> and <i>u-law</i> , with the default value of <i>A-law</i> .
	Sets whether to enable the echo cancellation feature for call conversations over
Echo Canceller	the digital trunk. By default, this feature is enabled and the effect can reach
	128ms.
Busy Tone Detection	Once this feature is enabled, the IP side will reply the 486 message once the E1
Busy Tone Detection	side detects the busy tone. The default value is <i>disabled</i> .
Frequency 1, Frequency 2	Sets the first and second center frequency for the busy tone, calculated by HZ.
	The default value of Frequency 1 is 450 and that of Frequency 2 is 0.
	Sets the busy tone cycle, calculated by ms. 4 different cycles can be added at the
Cycle	same time, sequencing from small to large and separated by ',' (e.g.
	700,1400,2000,3200). Range of value: 25-5000, with the default value of 700,
Ignore Busy Tone during	Once this feature is enabled, the gateway will not hang up the call when detecting
Call	the busy tone during the call. The default value is <i>enabled</i> .
Ringback Tone for	Sets whether to enable the E1 end to provide the ringback tone, with the default
PSTN →IP Call	value of <i>disable</i> .
PSTN Call Barring	Once this feature is enabled, you can set how many outgoing calls will be started
	to the same calledID, with the default value of <i>disable</i> .
Access Threshold for	Sets the maximum times for starting outgoing calls to the same CalledID.
Called Number	
Cycle	Sets the cycle for outgoing calls.
SIP Respond Code	Define the SIP code returned from PSTN to SIP when the times of outgoing calls
	exceed the threshold value.
ISDN 01 Message Contain	Sets the value of the progress indicator within the ISDN 01 message. Value of
Progress Indicator	range: $0x80 \sim 0xff$ , with the default value of $0x82$ . The value $0x0$ means the ISDN
	01 message does not contain the progress indicator.
Ringback Tone Volume	Sets the volume of the ringback tone. Range of value: -35~-2, calculated by dB,
	with the default value of -25.
Voice Gain Output from	Adjusts the voice gain of call from PSTN to the remote end. The value must be a
PSTN	multiple of 3. Range of value: -24~24, calculated by dB, with the default value of 0.
Hot Back-up for E1	Sets whether to enable the feature of hot back-up for E1, with the default value of
	disable.
Gateway IP for Hot	Set the IP of the gateway for the hot back-up for E1.
Back-up	
Limited Length of E1	Limits the CalleeID length of the outgoing calls from PSTN side. The calleeID will
Outgoing CalleeID	be divided into two parts if its length is greater than the value set in this item.
	Range of value: 0~50. The default value is 0, not limited.



	Sets whether to forward the call back to the PSTN side as it fails to start from
PSTN Call Forwarding	PSTN to IP, including three options: Disable, SIP call forwarding unavailable and
	Enable call forwarding immediately, with the default value of disable.
Number of Local SIP Trunk	Sets the local SIP trunk group No. used for forwarding the PSTN incoming call
Group	when it cannot get through.
	Sets the maximum times of the PSTN incoming calls which cannot get through.
Max No-Answer Times	The calls will not be forwarded until the times exceed the set value.

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>3.11.20 Restart</u> for detailed instructions.

# **3.4.2 Circuit Maintenance**

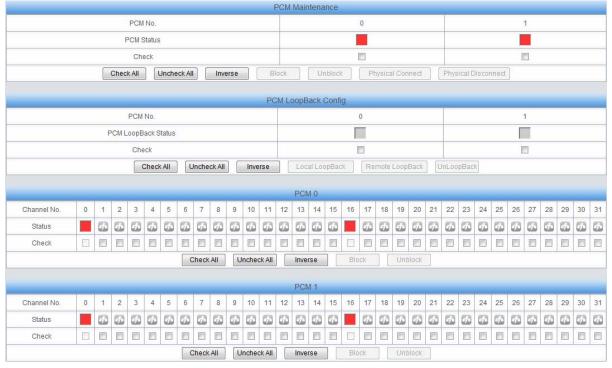


Figure 3-30 Circuit Maintenance Interface

See Figure 3-30 for the Circuit Maintenance interface. You can block, unblock, physical connect or disconnect PCMs, ports and channels on this interface. You can set the loopback feature of trunks for diagnoses or debugging. *Local LoopBack* means the transmitted data loop back from the LIU transmitter to the LIU receiver; *Remote LoopBack* means the transmitted data loop back to the LIU transmitter after being decoded in the LIU receiver. *UnLoopBack* is used to disable the features of local loopback and remote loopback.

**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	CRC-4	Sip Trunk No.	Modify
0	ISDN User Side	Line-synchronization	16	-	Twisted Pair Cable	Enable	-1	
1	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	



#### Figure 3-31 PCM Settings Interface

See Figure 3-31 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 Sideand SS1 in E1, and only includes ISDN User Side, ISDN Network Side		
Signaling Protocol	in T1.		
	Note: 1, Changing the interface type from E1 to T1 will forbid those non-ISDN		
	signaling modes in E1. And in such case, the gateway will by default set this item to		
	ISDN User Side.		
Clock	The clock mode for the digital trunk, including Line-synchronization, Free-run and		
CIOCK	Slave.		
	Sets the time slot used for signaling transmission on the digital trunk. If the		
Circuline: Time Clat	configuration item Signaling Protocol is set to ISDN and SS1, 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 Type	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.		
	Sets 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 <i>Signaling Protocol</i> is set to SS1.		
CRC-4	Sets whether to enable the CRC-4 verification feature. By default, this feature is		
UKU-4	Enabled.		
	The bound SIP trunk No. used to send the option notify message once the status of		
SIP Trunk No.	the PCM trunk changes.		

Click *Modify* in Figure 3-31 to modify a PCM. See Figure 3-32 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.:	0			
Signaling Protocol:	ISDN User Side 💌			
Signaling Time Slot 1:	16 💌			
Clock:	Line-synchronizatio 💌			
Connection Line:	Twisted Pair Cable 💌			
Sip Trunk No.:	1			
Enable CRC-4				
Apply to All PCMs				
Save	Close			

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

Item	Description
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	*
DCM	*
PSTN	
Circuit Maintena	nce
PCM	
PCM Trunk	
PCM Trunk Grou	ip
Num-Receiving	Rule
Reception Time	out
Number Attributi	on

Figure 3-33 PCM Trunk Configuration Interface

See Figure 3-33 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 3-34, Figure 3-35.

Figure 3-32 Modify PCM



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 3-34 Add PCM Trunk Interface

	PCM Trunk	Batch Add	
Including F	PCM:	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]
		(	
S	ave	Clos	e

Figure 3-35 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	1

Figure 3-36 PCM Trunks List

Click *Modify* in Figure 3-36 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	Clo	se

Figure 3-37 PCM Trunk Modification Interface

To delete a PCM trunk, 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 trunks at a time, click the **Clear All** button in Figure 3-36.



# 3.4.5 PCM Trunk Group

Check	Index	PCM Trunks	PCM Trunk Select Mode	Backup Trunk Group	Description	Modify
		7				
	0	0	Increase	None	default	1

Figure 3-38 PCM Trunk Group Settings

See Figure 3-38 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 3-39 for the PCM trunk group adding interface.

PCM Trunk	Group
Index:	1
Description:	default
PCM Trunk Select Mode:	Increase 💌
Backup Trunk Group:	None 💌
PCM Trunks:	Check All
0 E	1
Save	Close

Figure 3-39 Add New PCM Trunk Group

The table below explains the items shown in Figure 3-39.

Item	Description
la da c	The unique index of each PCM trunk group, which is mainly used in the configuration
Index	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 th	is configuration item to ring. The optional values and their
	corresponding meaning	s are described in the table below.
	Option	Description
PCM Trunk Select Mode	Increase	Search for an idle PCM trunk in the ascending order of the PCM number, starting from the minimum.
	Decrease Cyclic Increase	Search for an idle PCM trunk in the descending order of the PCM number, starting from the maximum.
		Provided PCM Trunk N is the available PCM trunk found last time. Search for an idle PCM trunk in the ascending order of the PCM number, starting from PCM Trunk N+1.
	Cyclic Decrease	Provided PCM Trunk N is the available PCM trunk found last time. Search for an idle PCM trunk in the descending order of the PCM number, starting from PCM trunk N-1.
Backup Trunk Group	A trunk group used as t	he backup one.
PCM Trunks	The PCM trunks in the PCM trunk group. If the checkbox before a PCM trunk is g indicates that the PCM trunk has been occupied. The ticked PCM trunks herein w displayed in the column 'PCM Trunks' in Figure 3-38.	

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

Click *Modify* in Figure 3-38 to modify a PCM trunk group. See Figure 3-40 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:	default				
PCM Trunk Select Mode: Increase					
Backup Trunk Group: None 💌					
PCM Trunks:	Check All				
<b>⊻</b> 0	]1				
Close					

Figure 3-40 Modify PCM Trunk Group

To delete a PCM trunk group, check the checkbox before the corresponding index in Figure 3-38 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 3-38.

### 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.

Check	Index	Number-receiving Rule	Description	Modify
	99		example	
1 [		- [		l .
eck All Unched	k All 🗄 Inverse 🗏 De	ete 🗄 Clear All		Add Ne

Figure 3-41 Number-Receiving Rule Configuration Interface

See Figure 3-41 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 3-42 for the number-receiving rule adding interface.

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

Figure 3-42 Add New Number-Receiving Rule

The table below explains the items shown in Figure 3-42.

ltem	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 200 nu	mber-receiving rules	can be configured in the gateway, and the			
	maximum leng	th of each number-rec	umber-receiving rule is 64 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 0~9.				
	"X"		A string of 'x's represents several random ble, 'xxx' denotes 3 random numbers.			
	""	om amount (including zero) of characters				
		after it. '[]' is used to define the range for a number. Value				
	"[]"	can be digits '0~9	', punctuations '-' and ','. For example,			
			ny one of the numbers 1, 2, 3, 6, 8.			
	<i>"</i> <b>1</b>	'-' is used only in '[	]' between two numbers to indicates any			
	" <u>"</u>	number between the	ese two numbers.			
	- "" · ""	',' is used to separat alternatives.	e numbers or number ranges, representing			
	By default, there is only one rule configured on the gateway. The table belo					
	-	-	e and understanding. See below for detailed			
Number Dessiving	information.	ipic for your cuby use	and understanding. See below for detailed			
Number-Receiving Rule	Priority	Dialing Rule	Description			
Rule	99		Any number in any length.			
	98	01[3,5,8]xxxxxxxx.	Any 12-digit number starting with 013, 015 or 018			
	97	010xxxxxxx	Any 11-digit number starting with 010			
	96	02xxxxxxxx	Any 11-digit number starting with 02			
	95	0[3-9]xxxxxxxxxx	Any 12-digit number starting with 03, 04,			
			05, 06, 07, 08 or 09			
	94	120	Number 120			
	93	11[0,2-9]	Number 110, 112, 113, 114, 115, 116, 117, 118 or 119			
	92	111xx	Any 5-digit number starting with 111			
	91	123xx	Any 5-digit number starting with 123			
	90	95xxx	Any 5-digit number starting with 95			
	89	100xx	Any 5-digit number starting with 100			
	88	1[3-5,8]xxxxxxxxx	Any 11-digit number starting with 13, 14, 15 or 18			
	87	[2-3,5-7]xxxxxxx	Any 8-digit number starting with 2, 3, 5, 6 or 7			
	86	8[1-9]xxxxx	Any 8-digit number starting with 81, 82, 83, 84, 85, 86, 87, 88 or 89			



	85	80[1-9]xxxx	Any 8-digit number starting with 801, 802, 803, 804, 805, 806, 807, 808 or 809
	84	800xxxxxx	Any 10-digit number starting with 800
	83	4[1-9]xxxxx	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 empty.	the number-receiving	rule. It can be any information, but can not be left

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

Click *Modify* in Figure 3-41 to modify the number-receiving rules. See Figure 3-43 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 3-43 Modify Number-receiving Rule

To delete a number-receiving rule, check the checkbox before the corresponding index in Figure 3-41 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 3-41.

### **3.4.7 Reception Timeout**

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

Figure 3-44 Number-receiving Timeout Info Interface

See Figure 3-44 for the number-receiving timeout info interface. The table below explains the items shown in the above figure.



Item	Description
	Sets the largest interval between two digits of a receiving number. Range of value:
	0~10, calculated by s, with the default value of 1. 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.
Deservicien	More information about the configuration item Inter Digit Timeout, such as the
Description	reason for adopting the current value.

Click *Modify* in Figure 3-44 to modify the number-receiving timeout info. See Figure 3-45 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				
Inter Digit Timeout (s):	1			
Description:	example			
Save	Close			

Figure 3-45 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 PSTN Forwarding

	PSTN Forwarding Number Table							
Check		No.		CallerID	CalleeID	Original CalleeID		Modify
Delete E Clea	ar All							Add New

Figure 3-46 PSTN Forwarding Number Table Interface

See Figure 3-47 for the PSTN Forwarding Number Table interface. This interface will be displayed only when the feature of PSTN Call Forwarding in the <u>PSTN</u> setting interface is enabled. It is used to set the corresponding number for the call from PSTN to IP which fails and is forwarded back to PSTN. Click **Add New** to add them manually.



PSTN Forwarding Number Table				
No.:	0			
CallerID:	*			
CalleeID:				
Original CalleeID:				
Save	Close			

Figure 3-47 PSTN Forwarding Number Table Adding Interface

The table below explains the items shown in above figures.

Item	Description			
<b>No.</b> The corresponding number for the call to be forwarded.				
CallerID The CallerID of the PSTN→IP incoming call.				
CalleeID The CalleeID of the IP→PSTN outgoing call.				
Original CalleelD	The original CalleeID of the PSTN→IP incoming call.			

After configuration, click **Save** to save the settings into the gateway or click **Close** to cancel the settings. See Figure 3-48 for the PSTN Forwarding Number Table.

			PSTN Forwarding Number Tabl	e	
Check	No.	CallerID	CalleelD	Original CalleeID	Modify
	0	#i	78742544	88861456	
Delete 🗄 🗄 Clear	All				Add New

#### Figure 3-48 PSTN Forwarding Number Table

Click *Modify* in Figure 3-48 to modify the number table. See Figure 3-49 for the PSTN forwarding number table modification interface. The configuration items on this interface are the same as those on the *Add PSTN Forwarding Number Table* interface. Note that the item *No.* cannot be modified.



PSTN Forwa	arding Number Table
No.:	0
CallerID:	*
CalleelD:	78742544
Original CalleeID:	88861456
Save	Close

Figure 3-49 PSTN Forwarding Number Table Modification Interface

To delete a piece of number table, check the checkbox before the corresponding index in Figure 3-48 and click the *Delete* button. To clear all forwarding number tables at a time, click the *Clear All* button in Figure 3-48.

# 3.5 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 3-50.

ISDN 🖉	
ISDN	
Number Parameter	
Figure 3-50 ISDN Settings	3



# 3.5.1 ISDN

CRC Ch	Contraction of the second s	CODEC	Default Caller Type	Default Callee Type	Ch Identification	TEI	Logical PCM No.	Link No.
	Enable -	▼ A-Law ▼	<ul> <li>National number (0X21)</li> </ul>		Number -	0	0	User Side: 0
	Enable 👻	✓ A-Law ✓	<ul> <li>National number (0X21)</li> </ul>	National number (0XA1)	Number -	0	1	User Side: 1
V	Enable -	▼ A-Law ▼	<ul> <li>National number (0X21)</li> </ul>	National number (0XA1)	Number -	0	2	User Side: 2
	Enable 👻		✓ National number (0X21)	National number (0XA1)	Number -	0	3	User Side: 3
ecting Num	Iller Type (with Redirect	lum) Call				INo.	-	
							0	
Ŧ							1	
	Tradonar number		National number		L		5	User olde, 5
				ROGRESS' Message			🔲 Ente	
			▼ nsmitted(01) ▼	(s) 60 Ing Call 1 ed to present(00) vided by users, checked and tra	ne for Called Party's Pick up( of the CalleelD of an Incomin erty Present Indicator Allow erty Shielding Indicator Prov	im Wait Tim m Length of Party Prope Party Prope	Enter Maximu Minimu Calling Calling	
			▼ nsmitted(01) ▼	(s) 60 Ing Call 1 ed to present(00) vided by users, checked and tra	ne for Called Party's Pick up( of the CalleeID of an Incomin erty Present Indicator Allow	im Wait Tim m Length of Party Prope Party Prope	Ente Maximu Minimu Calling Calling Default	
				(s) 60 Ig Call 1 ed to present(00) vided by users, checked and tra imber(0X21)	ne for Called Party's Pick up( of the CalleelD of an Incomin erty Present Indicator Allow erty Shielding Indicator Prov g Number Type National nu	m Wait Tim m Length of Party Prope Party Prope Redirecting	Ente Maximu Minimu Calling Calling Default ISDN User Side	
			Wait Confirm Time (T310) (s)	(s) 60 Ig Call 1 ed to present(00) vided by users, checked and tra imber(0X21)	ne for Called Party's Pick up( of the CalleeID of an Incomin erty Present Indicator Allow erty Shielding Indicator Prov g Number Type National nu et Party Number Complete' I	m Wait Tim m Length of Party Prope Party Prope Redirecting d the 'Caller	Ente Maximu Calling Calling Default ISDN User Side	
		ytes		(s) 60 Ig Call 1 ed to present(00) vided by users, checked and tra imber(0X21)	ne for Called Party's Pick up( of the CalleelD of an Incomin erty Present Indicator Allow erty Shielding Indicator Prov g Number Type National nu	m Wait Tim m Length of Party Prope Party Prope Redirecting d the 'Caller	Ente Maximu Calling Calling Default ISDN User Side	
		ytes	Wait Confirm Time (T310) (s)	(s) 60 Ig Call 1 ed to present(00) vided by users, checked and tra imber(0X21)	ne for Called Party's Pick up( of the CalleeID of an Incomin erty Present Indicator Allow erty Shielding Indicator Prov g Number Type National nu et Party Number Complete' I	m Wait Tim m Length of Party Prope Party Prope Redirecting d the 'Called d Channel I	Ente Maximu Calling Calling Default ISDN User Side	
		ytes	Wait Confirm Time (T310) (s)	(s) 60 Ing Call 1 ed to present(00) vided by users, checked and tra imber(0X21) ▼ Parameter	ne for Called Party's Pick up( of the CalleeID of an Incomin erty Present Indicator Allow erty Shielding Indicator Prov g Number Type National nu et Party Number Complete' I	im Wait Tim m Length of Party Prope Party Prope Redirecting d the 'Caller d Channel I ide	Ente Maximu Minimu Calling Default ISDN User Side Sen V Sen ISDN Network S	
er er	Iller Type (with Red National numbe National numbe National numbe	lum) Call	Callee Type (with Redirecting National number National number National number National number	a case of Redirecting Num	( I Voice •	tings r Capability		Link No. ser Side: 0 ser Side: 1 ser Side: 2 ser Side: 3

Figure 3-51 ISDN Settings Interface

See Figure 3-51 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>3.11.20 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-51.

Item	Description
	Terminal Equipment Identifier, which is used to identify the service access point in the
TEI	point-to-point data link connection. Range of value: 0~63, with the default value of 0.
	Note: The TEI values at the corresponding user side and the network side must be the
	same.
Ch Identification	Sets the way to represent channel identification messages on the digital trunk. The
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
	SETUP message during the outgoing call. The optional values are: National number,
Default Callee Type	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
	SETUP message during the outgoing call. The optional values are: National number,
Default Caller Type	International number, Network number, Subscriber number and Unknown, with the
	default value of National number.
00050	Sets the voice CODEC used on the digital trunk. The optional values are A-Law and
CODEC	<i>u-Law</i> , with the default value of <i>A-Law</i> .



	I
Auto Link Building	Sets whether to send the message of automatic link building for the ISDN at ISDN user side or network side. By default this feature is enabled.
CRC Check	Sets whether to enable the feature of CRC check for the digital trunk at ISDN user side or network side. By default this feature is enabled.
Set Caller/Callee Type in case of Redirecting Num	Once this feature is enabled, if the IP end carries the redirecting number in a call from IP to PSTN, you shall set separate values for the type of number and numbering scheme for the calling and called party numbers in the SETUP message, i.e. <b>Callee Type (with Redirecting Num)</b> and <b>Caller Type (with Redirecting Num)</b> . By default this configuration item is disabled.
Callee Type (with Redirecting Num)	This item is valid only when <b>Set Caller/Callee Type in case of Redirecting Num</b> is enabled. It sets the type of number and numbering scheme for the called party numbers in the SETUP message when the IP end carries the redirecting number in a call from IP to PSTN. The optional values are: National number, International number, Network number, Subscriber number and Unknown, with the default value of <i>National</i> <i>number</i> .
Caller Type (with Redirecting Num)	This item is valid only when <b>Set Caller/Callee Type in case of Redirecting Num</b> is enabled. It sets the type of number and numbering scheme for the calling party numbers in the SETUP message when the IP end carries the redirecting number in a call from IP to PSTN. The optional values are: National number, International number, Network number, Subscriber number and Unknown, with the default value of <i>National</i> <i>number</i> .
Transfer Capability	Sets the 'Transfer Capability' filed in the signaling message. The optional values are <i>Voice</i> and <i>3.1k Audio</i> , with the default value of <i>Voice</i> .
Enter Auto Alert State upon Reception of 'CALL PROCEEDING' Message	If this item is checked, the system will go into the state of auto alert when it receives the 02 (CALL PROCEEDING) message and the progress indicator turns to be 8 or 1. By default this item is disabled.
Enter Auto Alert State upon Reception of 'PROGRESS' Message	If this item is checked, the system will go into the state of auto alert when it receives the 03 (PROGRESS) message and the progress indicator turns to be 8 or 1. By default this item is disabled.
Maximum Wait Time for Called Party's Pick up	The maximum time waiting for the called party to pick up the call after the channel state turns to 'WaitAnswer' during an outgoing call. The default value is 60, calculated by s.
Minimum Length of the 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.
Calling Party Property Present Indicator	Sets the calling party property present indicator, including four options: Allowed to present, Restricted to present, Fail to provide numbers due to intercommunication and Reserved, with the default value of <i>Allowed to present</i> .
Calling Party Property Shielding Indicator	Sets the calling party property shielding indicator, including three options: Provide by users, unchecked; Provide by users, checked and transmitted; Provide by network. The default value is <i>Provide by users, checked and transmitted</i> .



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	Sets the number type and numbering scheme for the redirecting number in the SETUP
Default Redirecting	message during the outgoing call, The optional values are: National number,
Number Type	International number, Network number, Subscriber number and Unknown, with the
	default value of National number.
	Sets whether the channel identification message is included in the corresponding reply
Send Channel	message (such as CALL PROCEEDING, ALERT, etc.) after the local end receives the
Identification Message	SETUP message from the remote PBX during an incoming call. By default this item is
	checked.
	Sets the maximum time that the local end waits for the remote end to send back the
Wait Confirm Time	acknowledgement message in an outgoing call. If no acknowledgement message is
(T310)	received within the specified time period, the local end will disconnect the call
(1310)	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	Cate whether to include or not the 'Called Number Complete' perspector in the CETUD
Number Completed'	Sets whether to include or not the 'Called Number Complete' parameter in the SETUP message during an outgoing call.
Parameter	······································
Set Cause Value Length	Once this feature is enabled, the cause field in such messages as status (0x7d),
to 2 bytes	release (0x4d), disconnect (0x45) will be 2 bytes. By default this item is disabled (3 bytes).

## 3.5.2 Number Parameter

Judge C	Judge CallerID/CalleeID Prefix before Number Manipulation. Enable													
	Calling Party Number Type										Called Part	y Numb	er Type	
Check	No.	CallerID Prefix	CalleeID Prefix	Туре	Set if Redirecting Number Available	Modify		Check	No.	CallerID Prefix	CalleeID Prefix	Туре	Set if Redirecting Number Available	Modify
	0	666	888	0x21	No				0	666	888	0xa1	No	
							-							
							-							
							-							
Del	leite	Clea	ar All		Add	New		Del	ete	Clea	ir All		Add	New

Figure 3-52 ISDN Number Parameter Configuration Interface

See Figure 3-52 for the ISDN 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 Type
No.: 1
CallerID Prefix:
CalleeID Prefix:
Type: National number(0X21)
Set if Redirecting Number Available: 🔲 Enable
Close

Figure 3-53 Add New Calling Party Number Parameter

Called Pa	nty Number Type
No.:	1
CallerID Prefix:	
CalleeID Prefix:	
Type N	ational number(0XA1) 🛛 🗸
Set if Redirecting Nu	ımber Available: 🔲 Enable
Save	Close

Figure 3-54 Add New Called Party Number Parameter

The table below explains the items shown in above figures.

Item	Description
CallerID/CalleeID Prefix	Sets whether to judge the prefix of the CallerID/CalleeID which hasn't been
before Number	manipulated, with the default value of disabled, that is, only judge the prefix of
Manipulation	the CallerID/CalleeID which has been manipulated.
No.	The corresponding number for a calling/called party number parameter, which starts from 0.
CallerID/CalledIDPrefix	A string of numbers at the beginning of a calling/called party number.



Set if Redirecting	Set whether to enable the feature of setting this parameter only if the
Number Available	Redirecting Number 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 Type			
No.:	0		
CallerID Prefix:	666		
CalleeID Prefix:	888		
Type: N	lational number(0X21) 💽		
Set if Redirecting Number Available: 🔲 Enable			
Save Close			

Figure 3-55 Modify Calling Party Number Parameter

Called Party Number Type			
No.:	0		
CallerID Prefix:	666		
CalleeID Prefix:	888		
Туре N	ational number(0XA1) 💌		
Set if Redirecting Number Available: 🔲 Enable			
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 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.

## 3.5.3 Redirecting Number (Hidden item)

Judge CallerID before Number			Set			
			Redirec	ting Number Pool		
Check	No.	CallerID Prefix	CalleeID Prefix	Redirecting Number Range	PCM Trunk No.	Modify
	0	888	114	667669	1	
Delete 🗄	Clear All					Add New

Figure 3-57 Redirecting Number Pool Interface

After you enter http://the IP address of your gateway/gfhmc.php in the address column of the browser, the Redirecting Number Pool for ISDN will appear on the web. See Figure 3-57 for the Redirecting Number Pool interface. A new redirecting number can be added by the Add New button. See Figure 3-58 for the redirecting number adding interface.

Redirecting Number					
No.:	1				
CallerID Prefix:	*				
CalleeID Prefix:	CalleelD Prefix: *				
Starting Redirecting	Starting Redirecting Number:				
PCM Trunk:	Check All				
0	1				
Save	Close				

Figure 3-58 Add New Redirecting Number

The table below explains the items shown in above figures.



No.	The corresponding number for an added redirecting number. The value range is		
<i>N</i> O.	0~99.		
	A string of numbers at the beginning of a calling party number, which can be		
CallerID Prefix	numbers or "*" (indicating any string).		
CalleelD Prefix	A string of numbers at the beginning of a called party number, which can be		
	numbers or "*" (indicating any string).		
Starting Redirecting	The range of the redirecting number in the Redirecting Number Pool. It must be filled		
Number	in with numbers and can not be left empty.		
PCM Trunk	Sets the PCM included in the Redirecting Number Pool.		

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 redirecting number parameter. See Figure 3-59 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.

Redirecting Number			
No.:		0	
CallerID Prefix:		888	
CalleeID Prefix:		114	
Starting Redirecting	667		
		669	
PCM Trunk:	Che	eck All	
0	<b>V</b> 1		
Save Close			

Figure 3-59 Modify Redirecting Number

To delete a redirecting number parameter, check the checkbox before the corresponding index Figure 3-57 and click the '*Delete*' button. To clear all redirecting number parameters at a time, click the *Clear All* button in Figure 3-57.

**Note:** If there are two or more calling/called party numbers with the same prefix, the Starting Redirecting Number will increase to be 1 plus the previous one, starting from that with the smallest number.



# 3.6 SS1 Settings

SS1 Settings			
Country	CHINA		
ABCD Duration Timeout (ms)	0		
Max MFC Waiting Time (s)	10		
Receive CallerID	☑ Enable		
Advanced Setting for Incoming Calls			
KB Setting Timeout (s)	3		
KD Wait Time (ms)	60		
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			

Figure 3-60 SS1 Settings Interface

See Figure 3-60 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>3.11.20 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-60.

Item	Description			
Country	Sets the country to use SS1, with the default value of CHINA.			
	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			
ABCD Duration	default value of 0. Only when the on-line ABCD signaling codes vary and the			
Timeout	new value keeps for more than the time specified by this configuration item will			
	the gateway confirm the change of ABCD codes, Otherwise, the driver will			
	believe there are undesired dithering signals on the line.			
Max MEC Waiting Time	Sets the maximum waiting time, i.e. the timer T2 for the SS1 state machine,			
Max MFC Waiting Time	calculated by second, with the default value of 10.			
Receive CallerID	Sets whether to receive the calling party number. The default value is <i>enabled</i> .			
	Sets the maximum time to wait for the application to configure the KB signal,			
KB Setting Timeout	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			
KD Wait Time	timer T3) in the SS1 channel state machine, calculated by second, with the			
	default value of 60.			



ACK Wait Timeout	Sets the value of the timer T5, calculated by second, with the default value of 60.	
Calling Party's	Sets the KA signal (calling party's category at the local end) sent in an outgoing	
Category (KA Signal)	call. The value range is 1~10, with the default value of 1 (ordinary/regular).	
KB Wait Timeout	Sets the maximum time to wait for the KB signal from the remote PBX, calculated	
	by second, with the default value of 60.	
Originating Service	Sets the originating service type, i.e. KD, for an outgoing call. The value range is	
Type (KD Signal)	1~6, with the default value of 3 (local call).	

# 3.7 Fax Settings

See Figure 3-61 for the Fax Settings interface which is used to modify the special fax configurations.



Figure 3-61 Fax Settings

## 3.7.1 Fax

Fax Parameters			
Fax Mode	T.38	•	
T38 Version	0	•	
T38 Negotiation	Initiate Negotiation as Fax Re	2	
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	set		

Figure 3-62 Fax Configuration Interface (T.38 Mode)

See Figure 3-62 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>3.11.20 Restart</u> for detailed instructions. The table below explains the configuration items in Figure 3-62.



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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 the default
136 Version	value of 0.
T20 Magadiatian	Sets the Negotiation mode of T.38, including: Unsupported, Initiate Negotiation as
T38 Negotiation	Fax Sender and Initiate Negotiation as Fax Receiver.
Mawimum Fay 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.
Four Train Made	Sets the train mode for T.38 fax. The optional values are transferredTCF and
Fax Train Mode	<i>localTCF</i> , with the default value of <i>transferredTCF</i> .
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.30 Ecm	Sets whether to enable the T.30 error correction mode. By default this feature is
	enabled.
	As stipulated in the standard FAX CNG, the minimum duration of CNG is 500ms $\pm$
Min Duration of CNG	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 3-63.

Fax Para	meters
Fax Mode	Pass-through
Pass-through Payload	102
Min Duration of CNG(ms)	425
Min Duration of CED(ms)	2600
Save	Reset

Figure 3-63 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.



# 3.8 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 3-64.

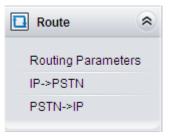


Figure 3-64 Route Settings

## 3.8.1 Routing Parameters

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

Figure 3-65 Routing Parameters Configuration Interface

See Figure 3-65 for the routing parameters configuration interface. On this interface, you can set the routing rules for calls respectively on two directions  $IP \rightarrow PSTN$  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.

## 3.8.2 IP to PSTN

					Routing Rules				
Check	Index	Call Initiator		CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
	255	SIP Trunk Group [0	0] 3	333[1,3]:444[6,9]	*	none	PCM Trunk Group [0]	default	
Check All	Unchec	k All Inverse	E Dele	e Cle	ar All			_	Add New

Figure 3-66 IP→PSTN Routing Rule Configuration Interface

See Figure 3-66 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 3-67 for the IP $\rightarrow$ PSTN routing rule adding interface.



IP->PSTN Routing Rule				
Index:	255			
Call Initiator:	SIP Trunk Group [0]			
CallerId Prefix:	*			
CalleeID Prefix:	*			
Call Destination:	PCM Trunk Group [0]			
Number Filter:	none			
Description:	default			
Save	Close			

Figure 3-67 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.	
	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.	



	can be set to	nbers at the beginning of the calling/called party number. This item o a specific string or "*" which indicates any string. These two ems together with <b>Call Initiator</b> can specify the calls which apply to a on:			
	Character				
	"0" <b>~</b> "9"	Digits 0 $\sim$ 9.			
		[]' is used to define the range for a number. Values within it only			
CallerID Prefix,	"[]"	can be digits '0~9', punctuations '-' and ','. For example,			
CalleeID Prefix	: : :	[1-3,6,8] indicates any one of the numbers 1, 2, 3, 6, 8.			
		'-' is used only in '[]' between two numbers to indicates any			
	-	number between these two numbers.			
	در ۶۶ ۲	',' is used to separate numbers or number ranges, representing alternatives.			
	Example: Rule "0[0-3,7][6-9]" denotes the prefix is 006, 016, 026, 036, 00				
	027, 037, 008,	018, 028, 038, 009, 019, 029, 039, 076, 077, 078, 079.			
	Note: Multiple	rules are supported for CallerID/CalleeID prefix. They are separated			
	by ":".				
Call Destination	PCM trunk gro	up to which the call will be routed.			
	Number filter r	ule which will be applicable to this route. It is set in <b>Number Filter</b> .			
Number Filter	See 3.9.4 Filter	See <u>3.9.4 Filtering Rule</u> for details.			
Description	More information	on 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 3-66 to modify a routing rule. See Figure 3-68 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->PSTN Routing Rule				
Index:	255			
Call Initiator:	SIP Trunk Group [0]			
CallerId Prefix:	333[1,3]:444[6,9]			
CalleeID Prefix:	*			
Call Destination:	PCM Trunk Group [0]			
Number Filter:	none			
Description:	default			
Save	Close			

Figure 3-68 Modify Routing Rule (IP→PSTN)

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-66 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the **Clear All** button in Figure 3-66.

## 3.8.3 PSTN to IP

				Routing Rules				
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
	255	PCM Trunk Group [0]	*	*	none	SIP Trunk Group [0]	default	
Check All	Uncheck	k All 👘 Inverse 👘	Deleter 🗄 🗧 Clear All					Add New

Figure 3-69 PSTN→IP Routing Rule Configuration Interface

See Figure 3-69 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 3-70 for the PSTN $\rightarrow$ IP routing rule adding interface.



PSTN->IP Routing Rule				
Index:	254			
Call Initiator:	PCM Trunk Group [0]			
CallerID Prefix:	*			
CalleeID Prefix:	*			
Call Destination:	SIP Trunk Group [0]			
Number Filter:	none			
Description:	default			
Save	Close			

Figure 3-70 Add New Routing Rule (PSTN→IP)

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

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with
	a smaller index value has a higher priority. If a call matches several routing rules, it
	will be processed according to the one with the highest priority.
Call Initiator	PCM trunk group from which the call is initiated. This item can be set to a specific
	PCM trunk group or PCM Trunk Group [ANY] which indicates any PCM trunk group.
	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
CollorID Brofix	configuration items together with Call Initiator can specify the calls which apply to a
CallerID Prefix, CalleeID Prefix	routing rule.
CalleelD Frenx	See the rule explanation of CallerID/CalleeID Prefix in <u>IP to PSTN</u> .
	Note: Multiple rules are supported in callerID/calleeID prefix. They should be
	separated by ":".
Call Destination	SIP trunk group to which the call will be routed.
Number Filter	Number filter rule which will be applicable to this route. It is set in Number Filter.
	See <u>3.9.4 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 3-69 to modify a routing rule. See Figure 3-71 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->IP Routing Rule		
Index:	255	
Call Initiator:	PCM Trunk Group [0]	
CallerID Prefix:	*	
CalleeID Prefix:	*	
Call Destination:	SIP Trunk Group [0]	
Number Filter:	none	
Description:	default	
Save	Close	

Figure 3-71 Modify Routing Rule (PSTN→IP)

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-69 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 3-69.

# 3.9 Number Filter

Number Filter includes four parts: *Whitelist*, *Blacklist*, *Number Pool* and *Filtering Rule*. See Figure 3-72.



Figure 3-72 Number Filter Interface



# 3.9.1 Whitelist

		CallerID Whitelist					CalleeID Whitelist		
Check	Group No.	No. in Group	CallerID	Modify	Check	Group No.	No. in Group	CalleeID	Modify
	0	0	111			0	0	222	
					-				
					-				
					-				
	Clear A				10	Clear			

Figure 3-73 Whitelist Setting Interface

See Figure 3-73 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 3-74, Figure 3-75 for CallerID/CalleeID whitelist adding interface.

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

Figure 3-74 Add New CallerIDs in Whitelist Interface

Calle	CalleelDs in Whitelist		
Group:	0		
No. in Group:	1		
CalleeID:			
Save	Close		



#### Figure 3-75 Add New CalleeIDs in Whitelist Interface

The table below explains the items shown in above figures.

Item		Description	
Group	The corresponding Group ID for CallerIDs/CalleeIDs in the whitelist. The value range is 0~7.		
No. in Group	The corresponding No. for different CallerIDs/CalleeIDs in a same group.		
	CallerID in the Rule explanation	whitelist, which can not be left empty. on:	
	Character	Description	
		indicating any string	
	"0"~"9"	Digits 0~9.	
	"x"	A random number. A string of 'x's represents several random	
CallerID	X	numbers. For example, 'xxx' denotes 3 random numbers.	
Callerid		'[]' is used to define the range for a number. Values within it only	
	"[]"	can be digits '0~9', punctuations '-' and ','. For example,	
		[1-3,6,8] indicates any one of the numbers 1, 2, 3, 6, 8.	
		'-' is used only in '[]' between two numbers to indicates any	
		number between these two numbers.	
	<i>"</i> ""	',' is used to separate numbers or number ranges, representing	
	,	alternatives.	
0-11	CalleeID in the	whitelist, which can not be left empty. The rules are the same as that	
CalleelD	of CallerID.		

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

Click **Modify** in Figure 3-73 to modify the CallerID or CalleeID whitelist. See Figure 3-76, Figure 3-77 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.

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

Figure 3-76 Modify CallerIDs in Whitelist



Calle	eelDs in W	hitelist	
Group:	0	~	
No. in Group:	[	0	
CalleeID:		101	
Save		Close	

Figure 3-77 Modify CalleeIDs in Whitelist

To delete a CallerIDs/CalleeIDs in the whitelist, check the checkbox before the corresponding index in Figure 3-73 and click the '*Delete*' button. To clear all CallerIDs/CalleeIDs in the whitelist at a time, click the *Clear All* button in Figure 3-73.

**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. The total amount of numbers in both whitelist and blacklist cannot exceed 5000.

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

# 3.9.2 Blacklist

Figure 3-78 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 3-78. The configuration items on this interface are the same as those on the Whitelist Setting interface (Figure 3-74, Figure 3-75).



#### 3.9.3 Number Pool

		Number Pool		
Check	Group No.	No. in Group	Number Range	Modify
	1	0	200201	6
			· · · · · · · · · · · · · · · · · · ·	
Delete	Clear All			Add New

Figure 3-79 Number Pool Setting Interface

See Figure 3-79 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 3-80 for the Number Pool adding interface.

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

Figure 3-80 Add New Number Pool

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

Item	Description
Crown	The corresponding Group ID for numbers in the number pool. The value range is
Group	0~15.
No. in Group	The corresponding No. for different numbers in a same group. It supports up to 100
No. in Group	number s in one group.
	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 3-79 to modify the number pool. See Figure 3-81 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 Deserve	
Number Range:	200
	201
Save	Close

Figure 3-81 Modify Number Pool Interface

To delete a number pool, check the checkbox before the corresponding index in Figure 3-79 and click the '*Delete*' button. To clear all number pools at a time, click the *Clear All* button in Figure 3-79.

# 3.9.4 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 Ca
	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	
										>

Figure 3-82 Filtering Rule Setting Interface

See Figure 3-82 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 3-83 for the Filtering Rule Adding interface.



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

Figure 3-83 Add New Filtering Rule

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

Item	Description
No.	The corresponding number for a filtering rule. The value range is 0~99.
CallerID Whitelist	The Group No. of CallerIDs saved on the whitelist setting interface.
CalleeID Whitelist	The Group No. of CalleeIDs saved on the whitelist setting interface.
CallerID Blacklist	The Group No. of CallerIDs saved on the blacklist setting interface.
CalleeID Blacklist	The Group No. of CalleeIDs saved on the blacklist setting interface.
CallerID Pool in	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 3-82 to modify the filtering rule. See Figure 3-84 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								
CalleeID 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								

Figure 3-84 Modify Filtering Rule Interface

To delete a filtering rule, check the checkbox before the corresponding index in Figure 3-82 and



click the '*Delete*' button. To clear all filtering rules at a time, click the *Clear All* button in Figure 3-82.

# 3.10 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 3-85.

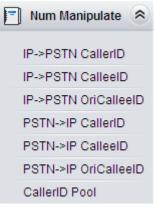


Figure 3-85 Number Manipulation

## 3.10.1 IP to PSTN CallerID

Check Index	Call Initiator										
	Gali illuatoi	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
255	SIP Trunk Group [0]			No	0	0	20			default	1
[					.m.						_

Figure 3-86 IP→PSTN CallerID Manipulation Interface

See Figure 3-86 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 3-87 for the IP $\rightarrow$ PSTN CallerID manipulation rule adding interface.



IP->PSTN CallerID Manipulation								
Index: 2	255 💌							
Call Initiator:	SIP Trunk Group [0]							
CallerID Prefix:	*							
CalleeID Prefix:	*							
With Original CalleelD	No							
Stripped Digits from Le	eft: 0							
Stripped Digits from R	ight: 0							
Reserved Digits from F	Right: 20							
Prefix to Add:								
Suffix to Add:								
Description:	default							
Save	Close							

Figure 3-87 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
maex	matches several number manipulation rules, it will be processed according to the
	one with the highest priority.
Coll 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. This item
	can be set to a specific string or "*" which indicates any string. These two
CallerID Prefix,	configuration items together with Call Initiator and With Original CalleeID can
CalleeID Prefix	specify the calls which apply to a number manipulation rule.
	Note: Multiple CallerID/CalleeID prefixes can be added simultaneously. They are
	separated by ":".



With Original CalleeID	If this item is set to Yes, it indicates that the number manipulation rule is only applicable to the calls with original CalleeID/redirecting number. The default value is <i>No</i> .
Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.
Description	More information about each number manipulation rule.

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

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

Click **Modify** in Figure 3-86 to modify a number manipulation rule. See Figure 3-88 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 M	lanipulation
Index: 255	•
Call Initiator: SIP Tr	unk Group [0] 🔹
CallerID Prefix:	*
CalleeID Prefix:	*
With Original CalleeID:	No
Stripped Digits from Left:	0
Stripped Digits from Right:	0
Reserved Digits from Right:	20
Prefix to Add:	
Suffix to Add:	
Description:	default
Save	Close

Figure 3-88 Modify IP→PSTN CallerID Manipulation Rule

To delete a number manipulation 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 number manipulation rules at a time, click the **Clear All** button in Figure 3-86.

## 3.10.2 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 3-89 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 3-86).

						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
	255	SIP Trunk Group [0]			No	0	0	20			default	2
						m						

Figure 3-89 IP→PSTN CalleeID Manipulation Interface



## 3.10.3 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 3-90 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 3-86).

Call Initiator	OallardD Deefer								
ounningeron	CallerID Prefix	CalleeID Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	Description	Modify
Trunk Group [0]	(*)	*	0	0	20			default	
				m				_	
eck All Inve	rse 🗄 De	ele Clea	ar All					Add	New
	ck All 🚊 🛛 Inve	ck All _ Inverse _ De	ckAll Inverse I Delete Clea		ck All = Inverse = Delete = Clear All	ck All = Inverse = Delete = Clear All	ck All = Inverse = Delete = Clear All	ck All = Inverse = Deleto = Clear All	ck All = Inverse = Deleto = Clear All Add

Figure 3-90 IP→PSTN Original CalleeID Manipulation Interface

# 3.10.4 PSTN to IP CallerID

Check         Index.         CallerID Prefix         CalleeID Prefix         With Original CalleeID         Stripped Digits from Right         Reserved Digits from Right         Prefix to Add         Suffix to Add         Description           255         SIP Trunk Group [0]         *         *         No         0         0         20         default						Number Manipulati	on Rules						
255         SIP Trunk Group [0]         *         No         O         O         20         default	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
		255	SIP Trunk Group [0]	*	*	No	0	0	20			default	
Check All Uncheck All Inverse Delta Clear All Add													New

Figure 3-91 PSTN→IP CallerID Manipulation Interface

See Figure 3-91 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 3-92 for the PSTN $\rightarrow$ IP CallerID manipulation rule adding interface.



PSTN->IP Ca	allerID Manipulation
Index:	254 💌
Call Initiator:	PCM Trunk 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: 20
Prefix to Add:	
Suffix to Add:	
Description:	default
Save	Close

Figure 3-92 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
maex	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
	can be set to a specific string or "*" which indicates any string. These two
CallerID Prefix,	configuration items together with Call Initiator and With Original CalleeID can
CalleeID Prefix	specify the calls which apply to the number manipulation rule.
	Note: Multiple CallerID/CalleeID prefixes can be added simultaneously. They are
	separated by ":".



With Original CalleeID	If this item is set to Yes, it indicates that the number manipulation rule is only applicable to the calls with original CalleeID/redirecting number. The default value is <i>No</i> .
Stripped Digits from Left	The amount of digits to be deleted from the left end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Stripped Digits from Right	The amount of digits to be deleted from the right end of the number. If the value of this item exceeds the length of the current number, the whole number will be deleted.
Reserved Digits from Right	The amount of digits to be reserved from the right end of the number. Only when the value of this item is less than the length of the current number will some digits be deleted from left; otherwise, the number will not be manipulated.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.
Description	More information about each number manipulation rule.

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

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

Click **Modify** in Figure 3-91 to modify a number manipulation rule. See Figure 3-93 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 \rightarrow IP** CallerID Manipulation Rule interface. Note that the item **Index** cannot be modified.



PSTN->IP Ca	llerID Manipulation
Index:	255 🔹
Call Initiator:	PCM Trunk Group [0] 💌
CallerID Prefix:	*
CalleeID Prefix:	*
With Original Calleell	D: No 💌
Stripped Digits from I	Left: 0
Stripped Digits from I	Right: 0
Reserved Digits from	Right: 20
Prefix to Add:	
Suffix to Add:	
Description:	default
Save	Close

Figure 3-93 Modify PSTN→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-91 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 3-91.

# 3.10.5 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 3-94 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 3-91).

						Number	Manipulation Rules					
Check	Index	Call Initiator	CallerID Prefix	CalleelD Prefix	With Original CalleeID	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	Description	Modify
	255	SIP Trunk Group [0]			No	0	0	20			default	1
Check A		Uncheck All Inve	erse   =   De	Clea	or All	m					Add	Now

Figure 3-94 PSTN→IP CalleeID Manipulation Interface



#### 3.10.6 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 3-95 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 3-91).

						Number Ma	anipulation Rules				
Check   In	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	Description	Modify
2	255	SIP Trunk Group [0]	*	*	0	0	20			default	
		ð		11. v		m				2	

Figure 3-95 PSTN→IP Original CalleeID Manipulation Interface

# 3.10.7 CallerID Pool

Manipulate I	P->PSTN	CallerIDs with Designated	d Prefix: * Starting Date:	2	016-01-11	Usage Cycle (Day) 0
IP->PSTN O	utbound C	alls with Designated Call		ignation Mode: SIP Side F Ind Calls with Designated C Cycle set to 0 mean	allerID set to 0 me	reans the feature is disabled; Usage
		IP->	PSTN Manipulated CallerID Pool			PSTN->IP Manipulated CallerID Pool
Check	No.	CallerID	Outgoing Call Resource	Destination PCM	Modify	Check No. CallerID Outgoing Call Resource Modify
	0	1001110027	10	PCM[0]		
Delete	Cle	ear All			Add New	Delete 🗄 Clear All Add Ne

Figure 3-96 CallerID Pool Interface

See Figure 3-96 for the CallerID Pool interface, including two parts: PSTN $\rightarrow$ IP Manipulated CallerID Pool and IP $\rightarrow$ PSTN Manipulated 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. If it is set to manipulate IP $\rightarrow$ PSTN CallerIDs with the designated prefix, only those calls with the CallerID prefix set in the CallerID pool meeting the requirement can be able to go out. The item *Manipulate* IP $\rightarrow$ PSTN CallerIDs with Designated Prefix can not be left empty. By default it is set to "\*", that is, calls with any CallerID prefix can go out. A new CallerID can be added by the Add New button. See Figure 3-97 for the CallerID adding interface.



CallerID
No.: 1
Outgoing Call Resource:
Destination PCM: Any
CallerID:
-
Save Close

Figure 3-97 Add New CallerID Interface

The table below explains the items shown in above figures.

Item	Description
IP-→PSTN Outbound	
Calls with	Sets the times of the outbound calls for the numbers in IP $\rightarrow$ PSTN CallerID Pool.
Designated CallerID	
Starting Data	Sets the starting time to start the IP $ ightarrow$ PSTN Outbound Calls with Designated
Starting Date	CallerID.
User Orale	Sets the execution cycle when the feature of IP $ ightarrow$ PSTN Outbound Calls with
Usage Cycle	Designated CallerID is enabled.
	Sets a mode for an IP $\rightarrow$ PSTN outbound call after all the IP $\rightarrow$ PSTN outbound calls
IP→PSTN	within the Usage Cycle reach the designated times, two options available: Sip Side
Designation Mode	Reject and Designated CallerID.
	Sets the space CallerId for an outbound call.
Set Spare CallerID	Note: This item is only valid when IP $\rightarrow$ PSTN Designation Mode is set to
	Designated CallerID.
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.
Outgoing Call	Cate the merimum number of the outgoing calls for each Caller
Resource	Sets the maximum number of the outgoing calls for each CallerID.
Destination PCM	The calls outgoing from the PCM designated in this item will do the manipulation.
CallerID	Sets the range of the CallerID used for an outgoing call.

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 CallerID information. See Figure 3-98 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
Outgoing Call Reso	urce: 10
Destination PCM:	PCM[0]
CallerID:	10011
-	10027
Save	Close

Figure 3-98 Modify CallerID Interface

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

# 3.11 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 3-99 for details.



<u>8</u> 28	System Tools	~
Ν	Vetwork	
A	Authorization	
Ν	lanagement	
П	P Routing Table	
A	ccess Control	
C	Centralized Manage	
8	BIP Account General	tor
C	Config File	
8	Signaling Capture	
8	Signaling Call Test	
8	Signaling Call Track	
F	PING Test	
Т	RACERT Test	
Ν	Iodification Record	
E	Backup & Upload	
F	actory Reset	
U	Jpgrade	
C	Change Password	
C	Device Lock	
F	Restart	

Figure 3-99 System Tools



## 3.11.1 Network

	Network Se	ettings
LAN 1		
	Network Type (M):	Static
	IP Address (I):	192.168.1.101
	Subnet Mask (U):	255.255.255.0
	Default Gateway (D)	192.168.1.254
	DNS Server (P)	0.0.0.0
LAN 2		
	Network Type (M):	Static
	IP Address (I):	201.123.111.147
	Subnet Mask (U):	255.255.255.0
	Default Gateway (D):	201.123.111.254
	DNS Server (P)	0.0.0.0
ARP Mode		
	Default Mode:	1
BOND Settin	9	
	BOND:	OYes  No

Figure 3-100 Network Settings Interface

See Figure 3-100 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: 1. The two configuration items IP Address and Default Gateway cannot be the same for NET 1 and NET 2.

2. By default, *Speed and Duplex Mode* is hidden, set to Automatic Detection, you can click 'F' to let it display. We suggest you do not modify it because the non-automatic



#### detection may cause abnormity in network interface.

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.

#### 3.11.2 Authorization

Authorization Information					
Serial Number	13479				
SS7 Supported	Yes				
Please select an authorization file: Browse					
			1		
	Update	Reset			

Figure 3-101 Authorization Interface

See Figure 3-101 for the Authorization interface. The SS7 signaling (ISUP and TUP included) can be supported by uploading an authorization file which is provided by our company and cannot be modified by users.

Click **Browse** to selset an authorization file, then click the **Update** button to upload it. Click **Reset** to restore the configurations.



# 3.11.3 Management

	Manageme	nt Parameters
WEB Manage	ment	
	WEB Port	80
	Access Setting	IPs in Whitelist
		201.123.115,201.123.3
	IP Address	<ul> <li>Control Control of Control Contro</li></ul>
		IP addresses are
	Time to Log out	separated by ','
	Time to Log out	1800s
Remote Data (	Capture Config	
	Remote Data Capture	
FTP Config		
5	FTP	©Yes ⊙No
Telnet Config		
	Telnet	©Yes ⊙No
Watchdog Set	ting	
	Enable Watchdog	©Yes ⊖No
SYSLOG Para	ameters	
	SYSLOG	●Yes <sup>O</sup> No
	Server Address	127.0.0.1
	SYSLOG Level	ERROR
CDR Paramete	ers	
	Send CDR	
	Server Address	127.0.0.1
	Server Port	3
	Send CDR Info of Failure Calls	
NAT Paramete		
	Monitor Self-adaption	©Yes ⊙No
Time Paramet		
	NTP	
	NTP Server Address	127.0.0.1
	Synchronizing Cycle	3600 s
	Daily Restart	●Yes ONo
	Restart Time	7 💌 h 13 💌 m
	System Time	Modify 1970-01-01 11:34:43
	Time Zone	GMT+8:00 (Beijing, Singapore, Taipei, Kua 💌



#### Figure 3-102 Management Parameters Setting Interface

See Figure 3-102 for the Management Parameters Setting interface. The table below explains the items shown in the above figure.

Item	Description
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.
Access Setting	Sets the IP addresses which can access the gateway via WEB. By default, all IPs are allowed. You can set an IP whitelist to allow all the IPs within it to access the gateway freely. Also you can set an IP blacklist to forbid all the IPs within it to access the gateway freely.
Time to Log Out	The gateway will log out automatically if it is not operated during a time longer than the value of this item, calculated by s, with the default value of 1800.
Remote Data Capture	After this feature is enabled, you can obtain the gateway data via a remote capture tool. The default value is No.
Capture RTP	Sets whether to capture RTP. Once this feature is enabled, the RTP package will also be captured by the selected network.
FTP	Sets whether to enable the FTP server, with the default value of Yes.
Telnet	Sets whether to enable the Telnet feature, with the default value of Yes. <b>Note:</b> By default, this configuration item is hidden. To display or hide it, you should click any part of the interface and press the "F" button.
Enable Watchdog	Sets whether to enable the watchdog feature, with the default value of Yes.
SYSLOG	Sets whether to enable SYSLOG. It is required to fill in <b>SYSLOG Server Address</b> and <b>SYSLOG Level</b> in case SYSLOG is enabled. By default, <b>SYSLOG</b> 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.
Send CDR	Sets whether to enable the feature of sending CDR. It is required to fill in <i>Server Address</i> and <i>Server Port</i> in case Send CDR is enabled. By default, <i>Send CDR</i> is disabled.
Server Address	The address of the server to receive CDR.
Server Port	The port of the server to receive CDR.
Send CDR Info of Failure Calls	Once this feature is enabled, the gateway will send the CDR for unsuccessful calls; otherwise, it will only send the CDR data for successful calls.
Monitor Self-adaption	Enable the NAT stun between the gateway and the monitor tool. By default, it is disabled.
NTP	Sets whether to enable the NTP time synchronization feature. It is required to fill in <i>NTP Server Address</i> , <i>Synchronizing Cycle</i> and <i>Time Zone</i> in case NTP is enabled. By default, <i>NTP</i> is disabled.
NTP Server Address	Sets the Server address for NTP time synchronization.
Synchronizing Cycle	Sets the cycle for NTP time synchronization.
Daily Restart	Sets whether to restart the gateway regularly every day at the preset <b>Restart Time</b> . By default, this feature is disabled.



Restart Time	Sets the time to restart the gateway regularly.
System Time	The system time. Check the checkbox before <i>Modify</i> and change the time in the edit box.
Time Zone	The time zone of the gateway.

# 3.11.4 IP Routing Table

IP Routing Table is used to set the route for the LAN port when two network ports both transport SIP. Thus, the LAN can access some IPs in other different network segment. By default, there is no routing table available on the gateway, click *Add New* to add them manually. See Figure 3-103.

Routing Table		
No.:	0	
Destination:		
Subnet Mask:		
Network Port:	NET 1(192.168.1.101)	
Save	Close	

Figure 3-103 Routing Table Adding Interface

The table below explains the items shown in above figures.

ltem	Description
No.	The number of the routing for the LAN in routing table.
Destination	The network segment the in which the IP address is accessible for the network port.
Subnet Mask	The subnet mask of the network segment.
Network Port	The corresponding network port of the routing.

After configuration, click *Save* to save the settings into the gateway or click *Close* to cancel the settings. See Figure 3-104 for the Routing Table List.



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			IP Routing Table		
Check	No.	Destination	Subnet Mask	Network Port	Modify
	0	201.123.112.0	255.255.255.0	NET 1(192.168.1.101)	
			· · · · · · · · · · · · · · · · · · ·		
5					
Delete 🗄 (	Clear All				Add New

Figure 3-104 Routing Table List

Click *Modify* in Figure 3-104 to modify a routing. See Figure 3-105 for the routing table modification interface. The configuration items on this interface are the same as those on the *Add Routing Table* interface. Note that the item *No.* cannot be modified.

Routing Table		
No.:	0	
Destination:	201.123.112.0	
Subnet Mask:	255.255.255.0	
Network Port:	NET 1(192.168.1.101) ¥	
Save	Close	

Figure 3-105 Routing Table Modification Interface

To delete a routing, check the checkbox before the corresponding index in Figure 3-104 and click the *Delete* button. To clear all routing tables at a time, click the *Clear All* button in Figure 3-104.

# 3.11.5 Access Control

Check	Index	Command		Modify
	0	iptables -I INPUT -s 123.45.6.7 -j	DROP	
heck All 📃 Unched	ck All Inverse	Delete Clear All	Apply	Add New

Figure 3-106 Access Control List Interface

See Figure 3-106 for the Access Control List interface. Once you add a piece of command to ACL, the network flow will be restricted, only the particular devices allowed to visit the gateway and only the data packages on the designated ports be forwarded. Click *Add New* to add a new piece of command. See Figure 3-107.



	Access Control Command
Index:	1
Command:	
	Close

Figure 3-107 Add Access Control Command Interface

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

Click *Modify* in Figure 3-106 to modify a command. See Figure 3-108 for the Access Control Command Modification interface. The configuration items on this interface are the same as those on the *Add Access Control Command* interface. Note that the item *Index* cannot be modified.

	Access Control Command
Index:	0
Command:	iptables -I INPUT -s 123.45.6.7 -j DROP
	Close

Figure 3-108 Access Control Command Modification Interface

To delete an Access Control Command, check the checkbox before the corresponding index in Figure 3-106 and click the **Delete** button, and then click the **Apply** button to make the deleted command invalid. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all access control commands at a time, click the **Clear All** button in Figure 3-106.

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

2, When you add or modify or delete commands manually, don't forget to click the *Apply* button to make your settings valid. However, when the gateway restarts or the configuration is leading-in, you need not click the *Apply* button and the commands will get valid automatically.



# 3.11.6 Centralized Manage

Centralized Manage				
Centralized Manage	🔽 Enable			
Auto Change Default Gateway:	Enable			
Management Platform:	DCMS			
Company Name:				
Gateway Description:				
Centralized Management Protocol:	SNMP -			
SNMP Version:	V2 •			
SNMP Server Address:	127.0.0.1			
C Monitoring Port	162			
Community String:	public			
Working Status:	Not Enabled			
Save	Download MIB			

Figure 3-109 Centralized Manage Setting Interface

See Figure 3-109 for the Centralized Manage Setting interface. The gateway can register to a centralized management platform and accept the management of the platform. The table below explains the items shown in above figures.

Item	Description
Auto Channe Default	Once this feature is enabled, the gateway will connect the DCMS via another
Auto Change Default	network port automatically once the connected network cable is loosen or drawn
Gateway	out. The default value is disabled.
Management Plarform	Select a management platform for the gateway to register.
Company Name	The company name used to register the gateway to DCMS, only valid when
	DCMS is selected.
Gateway Description	The description displayed on DCMS after the gateway is registered to DCMS,
	giving an easy identification of the gateway in device grouping. This item is only
	valid when DCMS is selected.
Centralized	Cate the controlized measurement protocol. It asks comparts CNMD currently
Management Protocol	Sets the centralized management protocol. It only supports SNMP currently.
SNMP Version	Sets the version of SNMP, three options available: V1, V2 and V3, with the default
	value of V2.
SNMP Server Address	IP address of SNMP.



Monitoring Port	Monitoring Port for SNMP on the gateway.			
Community String	Community string used for information acquisition.			
Account	The account of SNMP, only valid when the SNMP version is set to V3.			
Grade	The grade of SNMP, three options available: Neither authenticated nor encrypted,			
	Authenticated but not encrypted and Authenticated and encrypted, with the default			
	value of Neither authenticated nor encrypted. It is only valid when the SNMP			
	version is set to V3.			
Authentication	The authentication password required to enter when the item Grade is set to			
Password	Authenticated but not encrypted or Authenticated and encrypted.			
Encryption Password	The encryption password required to enter when the item Grade is set to			
	Authenticated and encrypted.			
Working Status	The status of the connection between the gateway and the centralized			
	management server. It is only valid when DCMS is selected.			

## 3.11.7 SIP Account Generator

SIP Trunk No.: 0 *Please save and uplo	Registration Validity Period (s): vad again after modification!*	Addres	ims zi chinamohile ( Description d	lefault Save
			Upload	
	To upload a file, select i Please upload a file	t and click the butto	n "Upload" on the right to start. Browse · · · ·	Upload
			DownLoad	
	File To Be Backup	SIP Account File	Please click the right mouse butt back up files to your computer!	on to Download

Figure 3-110 SIP Account Generator Interface

See Figure 3-110 for the SIP Account Generator interface. The gateway allows to transform the common SIP account and password to the specific format it supports, upload a file containing the SIP account and password, and modify the SIP Trunk No., Registration Validity Period, Registration Address and Description according to your requirement. Click **Save** to save your settings and upload the SIP account source file again. Then the SIP account in the format that the gateway supports will be generated. Click **Download** to check the generated SIP account.

Note: As to the upload file, only the txt. format is supported at present, and the SIP account and password must be separated by ",".



# 3.11.8 Configuration File

SMGC	onfig.ini	•
Config File		
[monitor]		
LocalAddress=127.0.0.1		<u></u>
LocalPort=1002		_
AutoExec=1		
UpgradeExecPath=/usr/local/apache/htdocs/RecUpgrade		=
IniFilePath=/mnt/flash		
[DigitsMapRulesInfo]		
DigitsMapRulesNum=1		
[NetConfig]		
BondFlag=0		
arpMode=1		
IpAddr1=201.123.111.20		
Subnet1=255.255.255.0		
Gateway1=201.123.111.254		
DNS1=0.0.0		
CheckNet1=0		
lpAddr2=192.168.0.101		
Subnet2=255.255.255.0		
Gateway2=192.168.0.254		
DNS2=0.0.0		
CheckNet2=0		
Mode1=0		
Mode2=0		
EnableBond=0		
[SysInfo]		
RunTime=12685		
NtpStatus=1		
RecvNetFlow=12346		
SendNetFlow=58469		
SetNTP=0		
NTPIP=127.0.0.1		
NTPIP2=127.0.0.2		
NTPCycle=3600		
NTPZone=2000		
SetReboot=0		
Debeet lour-7		Ŧ
Save Reset		
Note: You shall restart the service or system to validate the modified configuration file!		

Figure 3-111 Configuration File Interface

See Figure 3-111 for the Configuration File interface, including three files: SMGConfig.ini, ShConfig.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. 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.



### 3.11.9 Signaling Capture

	Data Capture				
Choose a network interface to capture Data Please designate the calling number to capture RTP! Destination address for syslog	LAN 1(192.168.1		Star		
TS Recording					
Choose a PCM and TS to record data	PCM 0	▼ E1 Time Slot 0(T1 T ▼	Start	Stop	
Choose a PCM and TS to record data	PCM 0	▼ E1 Time Slot 16 ▼	Start	Stop	
	E1	Two-way Recording			
Choose a PCM and TS to record data	PCM 0	E1 Time Slot 1(T1 T	Start	Stop	

Figure 3-112 Signaling Capture Interface

See Figure 3-112 for the Signaling Capture interface. Data Capture is used to capture data on the network interface you choose. Click *Start* to start capturing data (up to 400M for SMG2000 series; up to 800M for SMG3000 series) on the corresponding network interface. SIP, ISDN 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. Once the option Capture RTP is ticked, you are required to input the calling number of the RTP to be captured.

Data Recording (one-way) and E1 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.

Click *Clean Data* to clean all the recording files and captured packages. Click *Download Log* to download such logs as core files, configuration files, error information and so on.



### 3.11.10 Signaling Call Test

		Signaling Call Test	
	Test Type	IP->PSTN -	
	SIP Trunk Group No.	SIP Trunk Group[0]	
	CallerID		
	CalledID		
	Original CalleeID/Redirecting Number		
Signalir	Start ng Trace	Clear	
Transelat CtiAutoDi chid=051 GWS_OU chid=000 chid=051 GWS_OU chid=000 GWS_IN_ chid=051 GWS_OU chid=051 GWS_IN_ chid=051 GWS_CA chid=000 chid=051	al(514,888222) successed. 4,chid=0003,777111->88822 IT_REAL_MAKE_CALL>GW 3,chid=0514,111->222 CALL 4,chid=0003,777111->88822 IT_WAIT_CALL_RESULT>G 3,chid=0514,111->222 CALL SEND_RING>GWS_IN_W 4,chid=0003,777111->88822 IT_WAIT_CONNECT>GWS_ 3,chid=0514,111->222 CALL WAIT_OUT_CONNECT>GWS_ 3,chid=0514,111->222 CALL WAIT_OUT_CONNECT>GWS_ 4,chid=0003,777111->88822 LL_FINISHED>GWS_CALL 3,chid=-001,-> CALL_ID= sta 4,chid=-001,-> CALL_ID= sta	77111, Calledid: 222>888222 2 CALL_ID= stat change: /S_OUT_WAIT_CALL_RESULT _ID= stat change: GWS_IDLE>GWS_IN_SEND_RING 2 CALL_ID= stat change: GWS_OUT_WAIT_CONNECT _ID= stat change: AIT_OUT_CONNECT 2 CALL_ID= stat change: _CALL_ID= stat change: _ID= stat change: WS_CALL_CLEAR 2 CALL_ID= stat change:	E
		at change: GWS_WAIT_TO_IDLE>GWS_IDLE	

### Figure 3-113 Signaling Call Test Interface

See Figure 3-113 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	
Taoé Tumo	The source trunk type for signaling call test. There are three options: IP→PSTN,	
Test Type	PSTN→IP, PSTN Call Out and IP Call Out.	
	The SIP trunk group number you are required to select if choosing IP→PSTN or IP	
SIP Trunk Group No.	Call Out in Test Type.	



	The PCM trunk group number you are required to select if choosing <b>PSTN→IP</b> in
PCM Trunk Group No.	Test Type.
CallerID	The CallerID for the signaling call test.
CalleelD	The CalleeID for the signaling call test.
Original	
CalleeID/Redirecting	The original CalleeID/Redirecting Number for the signaling call test.
Number	
DOM Dani	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.
	You are required to select the PCM channel if choosing PSTN Call Out in Test
PCM Channel	Туре.
	Note: This item will appear only if you choose PSTN Call Out in Test Type.
Orand Oranania Manakan	Sets whether the IAM message will send the generic number or not.
Send Generic Number	Note: This item will appear only if you choose PSTN Call Out in Test Type.
O an antia Namakan Duan anta	Sets the generic number for the IAM message, This configuration item is valid only
Generic Number Property	when the feature of Send Generic Number is enabled.
	You can select this item to send DTMFs after the establishment of call conversation
DTME	on the channel for call test, if choosing PSTN Call Out or IP Call Out in Test Type.
DTMF	Note: This item will appear only if you choose PSTN Call Out or IP Call Out in Test
	Type, and RFC2833 is unsupported for IP Call Out.
Add Invite Header Field	You can add the invite header and its corresponding content if choosing IP Call Out
Add Invite Header, Field	in Test Type.
Name, Field Content	Note: This item will appear only if you choose IP Call Out in Test Type.
Cirmoling Trees	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 gateway can stop the testing only when the Test Type is set to PSTN Call Out; otherwise, the call test will not terminate until the called party ends it.



## 3.11.11 Signaling Call Track

Call 1	Frack	
● Filter CallerID	0	
O Filter Callend	0	
O Filter CalleerD		
O Filter None		
Start Stop Filter	r Clear Download	
Track Message		

### Figure 3-114 Call Track Interface

See Figure 3-114 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.



### 3.11.12 PING Test

Ping Tes	t
Source IP Address	LAN 1: 201.123.111.102
Destination Address	127.0.0.1
Ping Count (1-100)	4
Package Length (56-1024 bytes)	56
Start	End
Info	<u>~</u>
	<u>×</u>

### Figure 3-115 Ping Test Interface

See Figure 3-115 for the Ping Test interface. A Ping test can be initiated from the gateway on a designated IP address to check the connection status between them. The table below explains the configuration items shown in the above figure.

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

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



### 3.11.13 TRACERT Test

Tracert T	fest
Source IP Address	LAN 1: 201.123.111.102
Destination Address	127.0.0.1
Maximum Jumps (1-255)	30
Start	End
Info	V
	<b>×</b>

### Figure 3-116 Tracert Test Interface

See Figure 3-116 for the Tracert Test interface. A Tracert test can be initiated from the gateway on a designated IP address to check the routing status between them. The table below explains the configuration items shown in the above figure.

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

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



### 3.11.14 Modification Record

1.123.115.107	2
70-01-01 08:34:36 Mod:Config/SMGConfig.ini-NetConfig-Mode2:4>0 from 201.123.111.5	
70-01-01 08:53:49 Mod:Config/SMGConfig.ini-GWGLOBAL-global_PSTNCallForWarding:0>1 from 201.123.115.107	
70-01-01 09:41:28 Mod:Config/SMGConfig.ini-NetConfig-Mode2:0>1 from 201.123.111.5	
70-01-01 09:41:38 Add:Config/SMGConfig.ini-GWGLOBAL-global_getcalleridplace:-1>2 from 201.123.111.183	
70-01-01 09:42:49 Mod:Config/SMGConfig.ini-NetConfig-Mode2:1>3 from 201.123.111.5	
70-01-01 10:22:34 Mod:Config/SMGConfig.ini-SysInfo-Language:1>0 from 201.123.115.107	
70-01-01 11:16:54 Mod:/Config/SMGConfig.ini-SysInfo-Language:0>1 from 201.123.115.107	
70-01-01 11:23:15 Add:Config/SMGConfig.ini-ISDN-CalleeParamNum:-1>1 from 201.123.115.107	
70-01-01 11:23:15 Add:Config/SMGConfig.ini-ISDN-CallerIdPre4Callee0:-1>666 from 201.123.115.107	
70-01-01 11:23:16 Add:Config/SMGConfig.ini-ISDN-CalleeIdPre0:-1>888 from 201.123.115.107	
70-01-01 11:23:16 Add:Config/SMGConfig.ini-ISDN-CalleeParam0:-1>0xa1 from 201.123.115.107	
70-01-01 11:23:17 Add:Config/SMGConfig.ini-ISDN-RedirectingNumber CalledEnable0:-1>0 from 201.123.115.107	
70-01-01 11:23:37 Add:Config/SMGConfig.ini-ISDN-CallerParamNum:-1>1 from 201.123.115.107	
70-01-01 11:23:38 Add:Config/SMGConfig.ini-ISDN-CallerIdPre0:-1>666 from 201.123.115.107	
70-01-01 11:23:38 Add:Config/SMGConfig.ini-ISDN-CalleeldPre4Caller0:-1>888 from 201.123.115.107	
70-01-01 11:23:39 Add:Config/SMGConfig.ini-ISDN-CallerParam0:-1>0x21 from 201.123.115.107	
70-01-01 11:23:39 Add:Config/SMGConfig.ini-ISDN-RedirectingNumber_CallerEnable0:-1>0 from 201.123.115.107	
70-01-01 11:52:39 Mod:Config/SMGConfig.ini-SysInfo-Language:1>0 from 201.123.115.107	-
70-01-01 12:23:26 Mod:/Config/SMGConfig.ini-SysInfo-Language:0>1 from 201.123.115.107	
70-01-01 12:24:33 Mod:Config/SMGConfig.ini-SysInfo-Language:1>0 from 201.123.115.107	
70-01-01 12:32:49 Mod:Config/SMGConfig.ini-NetConfig-Mode2:3>0 from 201.123.111.5	
70-01-01 08:05:10 Mod:/Config/SMGConfig.ini-SysInfo-Language:0>1 from 201.123.111.5	
70-01-01 08:06:34 Mod:Config/SMGConfig.ini-NetConfig-Mode2:0>2 from 201.123.111.5	
70-01-01 08:07:56 Mod:Config/SMGConfig.ini-SysInfo-Language:1>0 from 201.123.115.107	1
NG 2013년 1927년 2월 2013년 2월 2014년 2월 21일 전 2014년 전 2014년 2월 2015년 7월 2014년 2월 2014년 2월 2014년 2월 2014년 2월 2014년 2	

Figure 3-117 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 3-117. Click *Download* to download the record file.

### 3.11.15 Backup & Upload

		Data Backup		
Choose a file to backup:	Configuration file	<ul> <li>Click the 'Backup' butto</li> </ul>	on on the right to backup the file.	Backup
		Data Upload		
To upload a file, select it a	and click the button 'Upload' on	n the right to start.		
		•	Browse	Upload

Figure 3-118 Backup & Upload Interface

See Figure 3-118 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.

### 3.11.16 Factory Reset

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

Figure 3-119 Factory Reset Interface

See Figure 3-119 for the Factory Reset interface. Click *Reset* to restore all configurations on the gateway to factory settings.

### 3.11.17 Upgrade

	Current Version
Serial	000002963
Number	
WEB	1.6.5_2017032108
Service	1.6.5_2017032108
Uboot	2.1.5_201509
Kernel	#419 SMP Fri Mar 3 17:00:07 CST 2017
Firmware	18
Select an Up	Browse
	Update Reset
Select an U	

Figure 3-120 Upgrade Interface

See Figure 3-120 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**.



### 3.11.18 Change Password

Change P	assword
Current Username	admin
Current Password	
New Username	
New Password	
Confirm New password	
Save Note: The username and the password can cor	Reset

Figure 3-121 Password Changing Interface

See Figure 3-121 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.

### 3.11.19 Device Lock

Device Lock	
Please select the condition to lock the device (Note: You are required to input the password before you modify any configuration of the selected items.)	
Confirm Password	
Lock Reset	

Figure 3-122 Device Lock Configuration Interface

See Figure 3-122 for the Device Lock Configuration interface. When you select one or more than one conditions to lock the gateway, the configurations of the gateway related to the selected conditions will be locked. That is, to modify any one of those configurations, you are required to input the lock password. Click *Lock* after setting and the device lock interface will be locked. To unlock the interface, enter your password (just the lock password) and click the *Unlock* button.



	Device Lock
Password	
	Unlock Reset

Figure 3-123 Unlock Device Interface

## 3.11.20 Restart

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

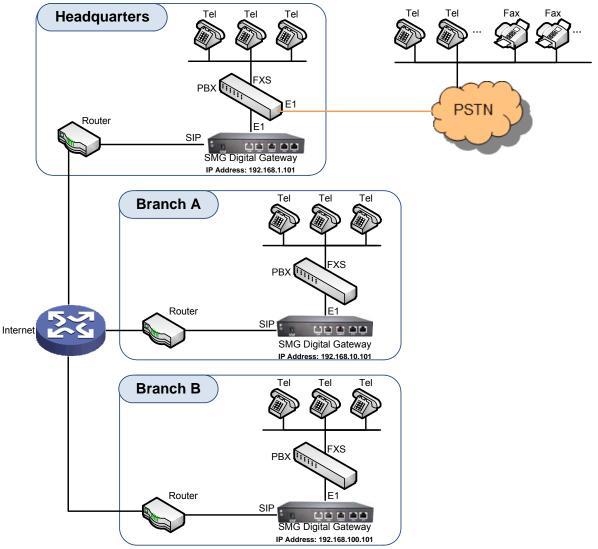
Figure 3-124 Service/System Restart Interface

See Figure 3-124 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.



Operation Info	
	*
SIP	*
SIP	
SIP Trunk SIP Register	
SIP Account	
SIP Trunk Group	
Media	
🚺 РСМ	*
ISDN	*
🔅 Fax	*
Route	*
Number Filter	*
Num Manipulate	*
System Tools	*

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SIP Settings	
SIP Address of WAN	LAN 2: 201.123.111.20
SIP Signaling Port	5060
Send 183 Message	Enable
Called Number Prefix for 180 Reply (Up to 5 are Allowed, Separated by '.')	
Send 100rel	🗆 Enable
Soft-switch to be Connected	VOS
Send 183 Delay Time(ms)	0
183 Send Delay Mode	Mode 1
Hide CallerID	Not Hidden
Obtain CallerID from	Username of From Field 💌
Obtain/Send CalleelD from	'Request Field 🔹
Asserted Identity Mode	Disable
Send/Obtain Redirecting Number/Original CalleeID from Diversion Field	Enable
NAT Traversal	Enable
SIP Transport Protocol	UDP
SIP Encryption	Enable
RTP Encryption	🗆 Enable
RTP Self-adaption	Enable
UDP Header Checksum	Enable
Rport	Enable
Filter Out Fake Calls (CallerID is the same as CalleeID)	Enable
Auto Reply of Source Address	Enable
DSCP	Enable
Calls from SIP Trunk Address only	Enable
Switch Signal Port if SIP Registration Failed	Enable
Hang up upon Call Time-out	Enable
Working Period	24 Hours
Session Timer	Enable
Early Media	Enable
Early Session	Enable
Not Wait ACK after Sending 200 OK	🗆 Enable
The Percentage of Registration Message Sending Cycle to Period of Validity(%)	70
Maximum Wait Answer Time(s)	60
Maximum Wait RTP Time(s)	0
Maximum Wait PSTN Resource Time(ms)	5000
Switch Network Port by Packet Loss Rate	Enable
Add Content to To Field in INVITE Message	OYes @No

Note: Only one SIP Trunk can be configured and its "Local Network Port" should be set to "Any Lan" once the feature "Switch Network Port by Packet Loss Rate" is enabled.



Figure 4-2

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

Operation Info	*	-					SIP 1	irunk			
SIP	*	Check Index Description Remote Address Remote Port Local Network Port Transport Protocol Outgoing Voice Resource Incoming Voice Resource									1
SIP										-	
SIP Trunk			0	default	201.123.112.227	5060	LAN 1(201.123.111.23)	UDP	512	512	G711A,G711U
SIP Register			1	default	201.123.112.147	5060	LAN 1(201.123.111.23)	UDP	512	512	G711A,G711
SIP Account		•			m						,
SIP Trunk Group						1.000	(				(
Media		Check A		Uncheck All	Inverse	Delete	Clear All				Add New

Figure 4-3

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

SIP	*	SIP Trunk Group									
-		Check	Index	SIP Trunks	SIP Trunk Select Mode	Outgoing Call Restriction	Incoming Call Restriction	Description	Modify		
SIP			0	0	Increase	No	No	Branch_A	1		
SIP Trunk		-				No	No	Durach D	0		
SIP Register			1.	. 1	Increase	NO	NO	Branch_B			
SIP Account		-									
SIP Trunk Group		Check All	Unchec	k All Inverse	Delete E Clea	r All					
Media		2 Items Total	20 Items/Pa	ge 1/1 First Previo	ous Next Last Go to Page 1 🛩	1 Pages Total					



### 4. Set PCM.

Operation Info	*									
SIP	*				PCM	Settings				
PCM	*	PCM No.	Signaling Protocol	Clock	Signaling Time Slot	Signaling Link Type	Connection Line	CRC-4	Sip Trunk No.	Modif
e rem		0	ISDN User Side	Line-synchronization	16	-	Twisted Pair Cable	Enable	1	
PSTN		1	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	1
E1 Outgoing Call		2	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	6
Circuit Maintenan	ce					200				
PCM		3	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
PCM Trunk		4	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	
PCM Trunk Group Num-Receiving R		5	ISDN User Side	Slave	16	<b>=</b> :	Twisted Pair Cable	Enable	-1	2
Reception Timeo		6	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	2
ISDN	*	7	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Sol Fax	*	8	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	2
Route	*	9	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	2
Number Filter	*	10	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	2
Num Manipulate		11	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	2
System Tools	*	12	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Ju System 10015	~	13	ISDN User Side	Slave	16	H2	Twisted Pair Cable	Enable	-1	
		14	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	
		15	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	0



### 5. Add PCM trunk

Operation In	fo 👻					
SIP	*	-			PCM Trunks	
(i) PCM	*	Check	Index	PCM NO.	Including Ts	Modify
PSTN	0		0	0	1,2,3,4,5,6,7,9,10,11,13,16,17,19,20,21,23,24,25,27,28,29,30,31	
Circuit Mainter	nance	Check All	Uncheck All	Inverse Dele	ie Clear All	Add Nev
PCM		1 Items Total 20	Items/Page 1/1	First Previous Next Last	Go to Page 1 👽 1 Pages Total	
PCM Trunk						

Figure 4-6

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



Operation Info	*												
SIP	¥		PCM Trunk Group										
DCM	*	Check	Index	PCM Trunks	PCM Trunk Select Mode	Backup Trunk Group	Description	Modify					
PCM	^		0	0	Increase	None	Headquarters						
PSTN				).									
Circuit Maintenar	nce	Check All	Uncheck All	Inverse 🗄 Dele	le 📄 🗧 Clear All			Add New					
PCM		1 Items Total 20	Items/Page 1/1	First Previous Next Last	Go to Page 1 💌 1 Pages Total								
PCM Trunk													
PCM Trunk Group	p	•											

Figure 4-7

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

Operation Info	*		
SIP	*		Route Settings
DCM	*	IP->IP	Route before Number Manipulate 💌
ISDN	*	PSTN->IP	Route before Number Manipulate
중 Fax	*		6
Route	*		Save
Routing Paramet	ers		
IP->PSTN			
PSTN->IP			

Figure 4-8

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

*									
¥					Routing Ru	lles			
	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	12
		62	SIP Trunk Group [1]	*		none	PCM Trunk Group [0]	from_Branch_B	2
	-								
0	_							(j	Add New
ers	2 Items Total	I 20 Items/P	age 1/1 First Previous Ne	ext Last Go to Page 1	<ul> <li>1 Pages Total</li> </ul>				
	» » » » «		Check         Index           Check         Index           Check All         63           Check All         Unche           2 Items Total         20 ItemsP	Check Index Call Initiator Check Index Call Initiator Check All Gal SIP Trunk Group [0] Check All Uncheck All Inverse 2 Items Total 20 Items/Page 1/1 First Previous No		Routing Ru         Check       Index       Call Initiator       CallerID Prefix       CallerID Prefix         Check       63       SIP Trunk Group [0]       *       *         Check All       Gal SiP Trunk Group [1]       *       *         Check All       Uncheck All       Inverse       Dister       Clear All         2 Items Total       20 Items/Page       11 First       Previous       Next Last Go to Page [1]       *	Routing Rules     Check Index Call Initiator CalleriD Prefix CalleeID Prefix Number Filter     63 SIP Trunk Group [0] * * none     62 SIP Trunk Group [1] * none     Check All Uncheck All Inverse Delete Clear All     Zitems 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       Call Destination         Image: Image	Routing Rules         Check       Index       Call Initiator       CallerID Prefix       CalleeID Prefix       Number Filter       Call Destination       Description         Image: State of the stat

Figure 4-9

 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.

	100					Routing Rules				
SIP	*	Check	Index	Call Initiator	CallerID Prefix	CalleelD Prefix	Number Filter	Call Destination	Description	Modify
PCM	*	Спеск	Index	Call Initiator	Callerid Pretix	CalleelD Pretix	Number Filter	Call Destination	Description	Modify
ISDN	*		63	PCM Trunk Group [0]	. *	9	none	SIP Trunk Group [0]	to_Branch_A	
			62	PCM Trunk Group [0]	*	*	none	SIP Trunk Group [1]	to_Branch_B	2
S Fax	*									Les
Route	*	Check All	Unched	k All Inverse	Delete 🗄 Clear Al	1				Add New
Routing Paramet	ters	2 Items Total	20 Items/Pa	age 1/1 First Previous Next	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	*													
SIP	×	-						Number Manipulatio	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	Modif
ISDN	*		63	PCM Trunk Group [0]		8	No	1	0	100			to_Branch_A	12
<ul> <li>Fax</li> </ul>	*		62	PCM Trunk Group [0]		7	No	1	0	100			to_Branch_B	12
Route	*	Check A	1 1	Uncheck All Inv	erse 🗧 😒	Cles	ar All						Add	New
Number Filter	*	2 Items To	al 20 Ite	ems/Page 1/1 First P	revious Next Las	t Go to Page 1 N	1 Pages Total							
Num Manipulate														
IP->PSTN CallerID														
IP->PSTN CallerID														
IP->PSTN CalleeID														

Figure 4-11

## 4.1.2 Configurations for Branch A

1. Configure SIP Settings for Branch A.



Operation Info       Sile         SIP       Sile         SIP Register       Sile         SIP Account       Sile         SIP Trunk Group       Media         ISP Trunk Group       Media         ISDN       Sile         Route       Sile         Number Filter       Sile         System Tools       Sile	828	
SIP SIP Trunk SIP Register SIP Account SIP Trunk Group Media PCM	Operation Info	
SIP Trunk SIP Register SIP Account SIP Trunk Group Media SID Trunk Sroup Media Fax SIP CM SIP SIDN SIP SIDN SIP SIDN SIP SIP SIDN SIP SIP SIP SIP SIP SIP SIP SIP SIP SIP	SIP	*
SIP Register SIP Account SIP Trunk Group Media SIDN % Fax % Route % Number Filter %		
SIP Account SIP Trunk Group Media PCM & SIDN & SIDN & Fax & Route & Number Filter & Number Filter &		
SIP Trunk Group Media     ¥       IsDN     ¥       IsDN     ¥       Rax     ¥       Number Filter     ¥       Istantiation     ¥		
Image: PCM       >         Image:		
ISDN     ≥       ISDN     ≥       Image: Second	Media	
Image: Second	DCM	*
Route     >       Image: Number Filter     >       Image: Num Manipulate     >	ISDN	*
Number Filter     >       Num Manipulate     >		*
🗐 Num Manipulate 😤	Route	*
	1	*
₩ System Tools 🛛 📚		*
	System Tools	*

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SIP Address of WAN	
SIP Address of WAN	LAN 2: 201.123.111.20
SIP Signaling Port	5060
Send 183 Message	Enable
Called Number Prefix for 180 Reply (Up to 5 are Allowed, Separated by ':')	
Send 100rel	Enable
Soft-switch to be Connected	VOS 💌
Send 183 Delay Time(ms)	0
183 Send Delay Mode	Mode 1
Hide CallerID	Not Hidden
Obtain CallerID from	Username of From Field 💌
Obtain/Send CalleeID from	'Request' Field 💌
Asserted Identity Mode	Disable
Send/Obtain Redirecting Number/Original CalleelD from Diversion Field	Enable
NAT Traversal	Enable
SIP Transport Protocol	UDP
SIP Encryption	Enable
RTP Encryption	Enable
RTP Self-adaption	Enable
UDP Header Checksum	Enable
Rport	Enable
Filter Out Fake Calls (CallerID is the same as CalleeID)	Enable
Auto Reply of Source Address	Enable
DSCP	Enable
Calls from SIP Trunk Address only	Enable
Switch Signal Port if SIP Registration Failed	Enable
Hang up upon Call Time-out	Enable
Working Period	24 Hours
Session Timer	Enable
Early Media	Enable
Early Session	Enable
Not Wait ACK after Sending 200 OK	Enable
The Percentage of Registration Message Sending Cycle to Period of Validity(%)	70
Maximum Wait Answer Time(s)	60
Maximum Wait RTP Time(s)	0
Maximum Wait PSTN Resource Time(ms)	5000
Switch Network Port by Packet Loss Rate	Enable
Add Content to To Field in INVITE Message	©Yes ⊛No



Figure 4-12

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

Operation Info							10.000			
SIP 2						SIP T	frunk			
	Check	Index	Description	Remote Address	Remote Port	Local Network Port	Transport Protocol	Outgoing Voice Resource	Incoming Voice Resource	
SIP		0	default	201.123.112.227	5060	LAN 1(201.123.111.23)	UDP	512	512	G711A,G711
SIP Trunk			110 01		10000			2121	202	1
SIP Register		1	default	201.123.112.147	5060	LAN 1(201.123.111.23)	UDP	512	512	G711A,G711
SIP Account	•			m						,
SIP Trunk Group	-				1					-
Media	Check A	and the second second	Uncheck All	Inverse	Delete	Clear All				Add New
	2 Items To	otal 20 lt	ems/Page 1/1	First Previous Ne	xt Last Go to P	age 1 - 1 Pages Total				

Figure 4-13

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

Operation Info 🛛 🗧								
SIP 😞					SIP Trunk Group			_
	Check	Index	SIP Trunks	SIP Trunk Select Mode	Outgoing Call Restriction	Incoming Call Restriction	Description	Modify
SIP		0	0	Increase	No	No	Headquarters	2
SIP Trunk					100			0
SIP Register		1	1	Increase	No	No	Branch_B	
SIP Account				- Antonio - Antonio				
SIP Trunk Group	Check All	Unche			ar All		(i	Add New
Media	2 Items Total	20 Items/Pa	age 1/1 First Prev	ious Next Last Go to Page 1 🛩	1 Pages Total			



### 4. Set PCM.

Operation Info	*									
SIP	*				PCM	Settings				
(i) PCM	*	PCM No.	Signaling Protocol	Clock	Signaling Time Slot	Signaling Link Type	Connection Line	CRC-4	Sip Trunk No.	Modif
U PCM	0	0	ISDN User Side	Line-synchronization	16	-	Twisted Pair Cable	Enable	1	0
PSTN		1	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	0
E1 Outgoing Call		2	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Circuit Maintenan	ice					-				
PCM		3	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
PCM Trunk		4	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
PCM Trunk Group Num-Receiving F		5	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	2
Reception Timeo		6	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	2
ISDN	×	7	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Solar Fax	*	8	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Route	*	9	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	1
Number Filter	*	10	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	
Num Manipulate		11	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
System Tools	*	12	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
- of stand 10013		13	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
		14	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	2
		15	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	

#### Figure 4-15

### 5. Add PCM trunk

Circuit Mainter	nance	Check All	Uncheck All	Inverse E Delete	Clear All	Add Nev
PSTN			0	0	1,2,3,4,5,6,7,9,10,11,13,16,17,19,20,21,23,24,25,27,28,29,30,31	
🚺 РСМ	*					
and the second se		Check	Index	PCM NO.	Including Ts	Modify
SIP	*				PCM Trunks	
Operation In		-			PCM Trunks	

### Figure 4-16

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



Operation In	fo 🛛							
SIP	*				PCM Trunk Group			
DCM	*	Check	Index	PCM Trunks	PCM Trunk Select Mode	Backup Trunk Group	Description	Modify
D PCM	<u> </u>		0	0	Increase	None	Branch_A	
PSTN								
Circuit Mainter	nance	Check All =	Uncheck All	Inverse 🗄 Delet	Clear All			Add New
PCM		1 Items Total 20	Items/Page 1/1	First Previous Next Last	Go to Page 1 💌 1 Pages Total			
PCM Trunk								
PCM Trunk Gr	oup							

Figure 4-17

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

Operation Info	*		
SIP	*		Route Settings
🚺 РСМ	*	IP->IP	Route before Number Manipulate 💌
ISDN	*	PSTN->IP	Route before Number Manipulate
<ii>중 Fax</ii>	*		
Route	*		Save
Routing Paramet	ers		
IP->PSTN			
PSTN->IP			

Figure 4-18

 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.

*									
*	1.0				Routing Ru	ules			
~	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_HQ	1
*		62	SIP Trunk Group [1]	×	*	none	PCM Trunk Group [0]	from_Branch_B	1
*	Check All	Unche	ck All Inverse	Delete Clea	ar All				Add New
rs	2 Items Tota	1 20 Items/P	age 1/1 First Previous Ne	xt Last Go to Page 1	✓ 1 Pages Total				
	* * * *		Check         Index           Check         Index           Check         63           Check All         62		Check         Index         Call Initiator         CallerID Prefix           Check         63         SIP Trunk Group [0]         *           62         SIP Trunk Group [1]         *           Check All         Uncheck All         Inverse         Delete         Clease           2 Items Total         20 Items/Page         1/1         First:         Providues. Next Last:         Goto Page         1	Routing Ru         Check       Index       Call Initiator       CallerID Prefix       CallerID Prefix         Check       63       SIP Trunk Group [0]       *       *         Check       62       SIP Trunk Group [1]       *       *         Check       All       Inverse       Dible       Clear All         2 Items Total       20 Items/Page       11 First       Previous Next Last Go to Page [1]       11 Pages Total	Routing Rules     Check Index Call Initiator CallerID Prefix CalleeID Prefix Number Filter     A 3 SIP Trunk Group [0] * * none     62 SIP Trunk Group [1] * * none     Check All Uncheck All Inverse Delete Clear All     Zitems Total 20 items/Page 1/1 First Previous Next Last Goto Page 1 v 1 Pages Total	Routing Rules         Check       Index       Call Initiator       CallerID Prefix       CalleeID Prefix       Number Filter       Call Destination         Image: State of the state	Routing Rules         Check       Index       Call Initiator       CallerID Prefix       CalleeID Prefix       Number Filter       Call Destination       Description         Image:

Figure 4-19

 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.

SIP       SIP       Collection       Called D Prefix       Number Filter       Call Destination       Description         I SDN       Sip       63       PCM Trunk Group [0]       *       9       none       SIP Trunk Group [0]       to_HQ         Sip       Fax       Sip       Fax       Sip       PCM Trunk Group [0]       *       9       none       SIP Trunk Group [0]       to_Branch_B
Check       Index       Call Initiator       CalleID Prefix       CalleeID Prefix       Number Filter       Call Destination       Description         Ison       Image: State St
ISDN         63         PCM Trunk Group [0]         *         9         none         SIP Trunk Group [0]         to_HQ           (b) Fax         62         PCM Trunk Group [0]         *         7*         none         SIP Trunk Group [1]         to_Branch_B
Comparison         Compari
Porte a 61 PCM Trunk Group [0] * 0 none SIP Trunk Group [0] to_PSTN
Route

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	*													
SIP	*	-						Number Manipulatio	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	Modif
and the second se			63	PCM Trunk Group [0]		9	No	1	0	100			to_HQ	0
ISDN	*		62	PCM Trunk Group [0]		7	No	1	0	100			to_Branch_B	12
Fax	*											6		1 10
Route	*	Check A	8	Uncheck All Inv	erse - De	Cle	ar All						Add	d New
🕂 Number Filter	×	2 Items To	tal 20 It	ems/Page 1/1 First P			1 Pages Total							
Num Manipulate	^													
Num Manipulate IP->PSTN CallerID	*													
IP->PSTN CallerID														
IP->PSTN CallerID IP->PSTN CalleeID														

Figure 4-21

## 4.1.3 Configurations for Branch B

1. Configure SIP Settings for Branch B.



Operation Info	
SIP	*
SIP	
SIP Trunk	
SIP Register	
SIP Account	
SIP Trunk Group	
Media	
DCM	*
ISDN	*
Fax	*
Route	*
•••• Number Filter	*
Num Manipulate	*
System Tools	*

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SIP Address of WAN	LAN 2: 201.123.111.20	•
SIP Signaling Port	5060	
Send 183 Message	Enable	
Called Number Prefix for 180 Reply (Up to 5 are Allowed, Separated by ':')		
Send 100rel	Enable	
Soft-switch to be Connected	VOS	•
Send 183 Delay Time(ms)	0	
183 Send Delay Mode	Mode 1	]
Hide CallerID	Not Hidden	•
Obtain CallerID from	Username of From Field	•
Obtain/Send CalleeID from	'Request Field	•
Asserted Identity Mode	Disable	•
Send/Obtain Redirecting Number/Original CalleeID	Enable	
from Diversion Field		
NAT Traversal	Enable Enable	
SIP Transport Protocol	UDP	•
SIP Encryption	Enable	
RTP Encryption	Enable	
RTP Self-adaption	Enable	
UDP Header Checksum	Enable	
Rport	Enable	
Filter Out Fake Calls (CallerID is the same as CalleeID)	Enable	
Auto Reply of Source Address	Enable	
DSCP	Enable	
Calls from SIP Trunk Address only	Enable	
Switch Signal Port if SIP Registration Failed	Enable	
Hang up upon Call Time-out	Enable	
Working Period	24 Hours	
Session Timer	Enable	
Early Media	Enable	
Early Session	Enable	
Not Wait ACK after Sending 200 OK	Enable	
The Percentage of Registration Message Sending Cycle to Period of Validity(%)	70	
Maximum Wait Answer Time(s)	60	
Maximum Wait RTP Time(s)	0	
Maximum Wait PSTN Resource Time(ms)	5000	
Switch Network Port by Packet Loss Rate	Enable	
Add Content to To Field in INVITE Message	©Yes ◉No	
UserAgent Field		

Note: Only one SIP Trunk can be configured and its "Local Network Port" should be set to "Any Lan" once the feature "Switch Network Port by Packet Loss Rate" is enabled.



Figure 4-22

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

Operation Info												
SIP #	SIP Trunk											
	Check	Index	Description	Remote Address	Remote Port	Local Network Port	Transport Protocol	Outgoing Voice Resource	Incoming Voice Resource			
SIP		0	default	201.123.112.227	5060	LAN 1(201.123.111.23)	UDP	512	512	G711A,G711L		
SIP Trunk	-		110 01					2002	202			
SIP Register		1	default	201.123.112.147	5060	LAN 1(201.123.111.23)	UDP	512	512	G711A,G711		
SIP Account	•									,		
SIP Trunk Group		1 1			1.000	[				-		
Media	Check A		Uncheck All	Inverse	Delete =	Clear All				Add New		
meana	2 Items To	tal 20 lt	ems/Page 1/1	First Previous Ne	xt Last Go to P	age 1 - 1 Pages Total						

Figure 4-23

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

Operation Info	*								
SIP	*					SIP Trunk Group			
		Check	Index	SIP Trunks	SIP Trunk Select Mode	Outgoing Call Restriction	Incoming Call Restriction	Description	Modify
SIP			0	0	Increase	No	No	Headquarters	12
SIP Trunk		-	1	4	Increase	No	No	Branch A	0
SIP Register			- 1	- 1.	Increase	110	NO		
SIP Account		-							
SIP Trunk Group	Þ	Check All	Unche			ear All			Add New
Media		2 Items Total	20 Items/Pa	age 1/1 First Prev	ious Next Last Go to Page 1 ⊻	1 Pages Total			



### 4. Set PCM.

Operation Info	*									
SIP	*	-			PCM	Settings				
(i) PCM	*	PCM No.	Signaling Protocol	Clock	Signaling Time Slot	Signaling Link Type	Connection Line	CRC-4	Sip Trunk No.	Modif
e rem		0	ISDN User Side	Line-synchronization	16	-	Twisted Pair Cable	Enable	1	2
PSTN		1	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	8
E1 Outgoing Call		2	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Circuit Maintenan	ice									
PCM		3	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	
PCM Trunk		4	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
PCM Trunk Group Num-Receiving F		5	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	2
Reception Timeo		6	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	2
ISDN	*	7	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Service Fax	*	8	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Route	*	9	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	
Number Filter	*	10	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Num Manipulate		11	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
System Tools	*	12	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
un oyacan roois		13	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
		14	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	2
		15	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	2

Figure 4-25

### 5. Add PCM trunk

Operation In	fo ≫ ≫	-			PCM Trunks	
<ul> <li>31P</li> <li>10</li> <li></li></ul>	*	Check	Index	PCM NO.	Including Ts	Modify
PSTN	<u>^</u>		0	0	1,2,3,4,5,6,7,9,10,11,13,16,17,19,20,21,23,24,25,27,28,29,30,31	C2
Circuit Mainte	nance	Check All	Uncheck All	Inverse Delete	Clear All	Add Nev
PCM		1 Items Total 20	Items/Page 1/1	First Previous Next Last Go	to Page 1 🗸 1 Pages Total	

Figure 4-26

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



Operation In	fo 👻							
SIP	*	-			PCM Trunk Group			
DCM	*	Check	Index	PCM Trunks	PCM Trunk Select Mode	Backup Trunk Group	Description	Modify
PCM	0		0	0	Increase	None	Branch_B	6
PSTN								
Circuit Mainter	nance	Check All	Uncheck All	Inverse 🗄 Delet	Clear All			Add New
PCM		1 Items Total 20	Items/Page 1/1	First Previous Next Last	Go to Page 1 💌 1 Pages Total			
PCM Trunk								
PCM Trunk Gr	oup	•						

Figure 4-27

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

Operation Info	*		
📽 SIP	*		Route Settings
🚺 РСМ	*	IP->IP	Route before Number Manipulate
ISDN	*	PSTN->IP	Route before Number Manipulate
🔅 Fax	*		0
Route	*		Save
Routing Paramete	ers		
IP->PSTN			
PSTN->IP			

Figure 4-28

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

Operation Info	*									
SIP	*	-				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_HQ	1
			62	SIP Trunk Group [1]	*	*	none	PCM Trunk Group [0]	from_Branch_A	2
OF Fax	×									
Route	*	Check All	Unche	ck All Inverse	Delete E Clea	ar All				Add New
Routing Paramete	ere	2 Items Total	20 Items/P	age 1/1 First Previous Ne	xt Last Go to Page 1	<ul> <li>1 Pages Total</li> </ul>				
IP->PSTN										

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	*									
SIP	*	-				Routing Rules	;			
(1) PCM	*	Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
	*		63	PCM Trunk Group [0]	7.81	9	none	SIP Trunk Group [0]	to_HQ	
<ul><li>Fax</li></ul>	*		62	PCM Trunk Group [0]	*	8	none	SIP Trunk Group [1]	to_Branch_A	2
Route	*		61	PCM Trunk Group [0]	*	0	none	SIP Trunk Group [0]	to_PSTN	
Routing Paramete		Check All	Unched	x All Inverse	Delete 🗄 Clear A					Add New
IP->PSTN	515	TO BEAU BOARD FOR		age 1/1 First Previous Next						
PSTN->IP										

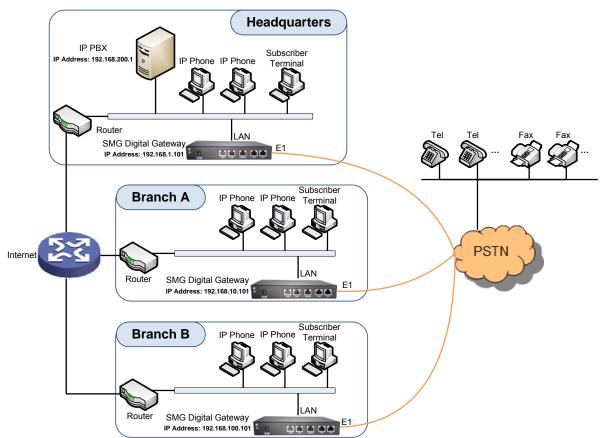
#### Figure 4-30

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



Operation Info	*													
SIP	×	-						Number Manipulatio	in 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	12
<ul> <li>Fax</li> </ul>	*		62	PCM Trunk Group [0]	•	8	No	1	0	100			to_Branch_A	0
Route	×	Check A		Uncheck All Inv		Cles	ar All						Add	New
Number Filter	×	2 Items To	al 20 Ite	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														
	ID													
IP->PSTN OriCallee														
PSTN->IP CallerID														

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.



Operation Info	*
SIP	*
SIP	
SIP Trunk	
SIP Register	
SIP Account SIP Trunk Group	
Media	
DCM	*
ISDN	*
Fax	*
Route	*
Number Filter	*
Num Manipulate	*
System Tools	*

SIP Address of WAN	LAN 2: 201.123.111.20
SIP Signaling Port	5060
Send 183 Message	Enable
Called Number Prefix for 180 Reply (Up to 5 are Allowed, Separated by ':')	
Send 100rel	Enable
Soft-switch to be Connected	VOS
Send 183 Delay Time(ms)	0
183 Send Delay Mode	Mode 1
Hide CallerID	Not Hidden
Obtain CallerID from	Username of From Field
Obtain/Send CalleeID from	'Request Field
Asserted Identity Mode	Disable
Send/Obtain Redirecting Number/Original CalleelD from Diversion Field	Enable
NAT Traversal	Enable
SIP Transport Protocol	UDP
SIP Encryption	Enable
RTP Encryption	Enable
RTP Self-adaption	Enable
UDP Header Checksum	I Enable
Rport	Enable
Filter Out Fake Calls (CallerID is the same as CalleeID)	Enable
Auto Reply of Source Address	Enable
DSCP	Enable
Calls from SIP Trunk Address only	Enable
Switch Signal Port if SIP Registration Failed	Enable
Hang up upon Call Time-out	Enable
Working Period	I 24 Hours
Session Timer	Enable
Early Media	Enable
Early Session	Enable
Not Wait ACK after Sending 200 OK	Enable
The Percentage of Registration Message Sending Cycle to Period of Validity(%)	70
Maximum Wait Answer Time(s)	60
Maximum Wait RTP Time(s)	0
Maximum Wait PSTN Resource Time(ms)	5000
Switch Network Port by Packet Loss Rate	Enable
Add Content to To Field in INVITE Message	©Yes ◉No
UserAgent Field	

Note: Only one SIP Trunk can be configured and its "Local Network Port" should be set to "Any Lan" once the feature "Switch Network Port by Packet Loss Rate" is enabled.



Figure 4-33

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

Operation Info	*										
SIP	*	-					SIP	Trunk			
		Check	Index	Description	Remote Address	Remote Port	Local Network Port	Transport Protocol	Outgoing Voice Resource	Incoming Voice Resource	
SIP		E	0	default	201.123.111.12	5060	LAN 1(201, 123, 111, 23)	UDP	512	512	G711A.G711U.G
SIP Trunk											
SIP Register											
SIP Account		Check A		Uncheck All	Inverse	Delete	Clear All				Add New
SIP Trunk Group		1 Items To	tal 20 lt	ems/Page 1/1	First Previous Ne	at Last Go to P	age 1 - 1 Pages Total				
Media											

Figure 4-34

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

Operation Info	) ×	-				SIP Trunk Group			
SIP	*					SIP TTUIK Group			
		Check	Index	SIP Trunks	SIP Trunk Select Mode	Outgoing Call Restriction	Incoming Call Restriction	Description	Modif
SIP			0	0	Increase	No	No	IP PBX	6
SIP Trunk								1	1.15
SIP Register			-		1-				
SIP Account		Check All	Unched	k All E Inverse	Delete E Clear	r All		1000	Add New

#### Figure 4-35

### 4. Set PCM.

Operation Info	*									
SIP	*				PCM	Settings				
(1) PCM	*	PCM No.	Signaling Protocol	Clock	Signaling Time Slot	Signaling Link Type	Connection Line	CRC-4	Sip Trunk No.	Modif
() PCM	0	0	ISDN User Side	Line-synchronization	16	-	Twisted Pair Cable	Enable	1	0
PSTN		1	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	Q
E1 Outgoing Call	Timer	2	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	_
Circuit Maintenan	се	2	ISDN Oser Side	Slave	10	-	Twisted Pair Cable	Enable	-1	
PCM		3	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
PCM Trunk		4	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
PCM Trunk Group Num-Receiving R		5	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	6
Reception Timeo	ut	6	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
ISDN	*	7	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
E Fax	*	8	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Route	*	9	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Number Filter	*	10	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
Num Manipulate	*	11	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
System Tools	*	12	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
-		13	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	
		14	ISDN User Side	Slave	16		Twisted Pair Cable	Enable	-1	
		15	ISDN User Side	Slave	16	-	Twisted Pair Cable	Enable	-1	0

Figure 4-36

### 5. Add PCM trunk

Operation Info	*					
SIP	*	-			PCM Trunks	
() PCM	*	Check	Index	PCM NO.	Including Ts	Modify
PSTN			0	0	1,2,3,4,5,6,7,9,10,11,13,16,17,19,20,21,23,24,25,27,28,29,30,31	
Circuit Maintenan	се	Check All	Uncheck All	Inverse Del	ete 🚊 Clear All	Add New
PCM		1 Items Total 20	Items/Page 1/1	First Previous Next Las	st Go to Page 1 💌 1 Pages Total	
PCM Trunk		•				

Figure 4-37

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

Operation Info	*							
SIP	*	-			PCM Trunk Gro	lb.		
DCM	*	Check	Index	PCM Trunks	PCM Trunk Select Mode	Backup Trunk Group	Description	Modify
-	0		0	0	Increase	None	Headquarters	
PSTN								
Circuit Maintena	nce	Check All	Uncheck All	E Inverse E Dele	te E Clear All			Add New
PCM		1 Items Total 20	) Items/Page 1/	1 First Previous Next Las	t 🛛 Go to Page 1 💌 1 Pages Total			
PCM Trunk								
PCM Trunk Grou	ID .							

Figure 4-38

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

Operation Info	*		
SIP	*		Route Settings
DCM	*	IP->IP	Route before Number Manipulate
ISDN	*	PSTN->IP	Route before Number Manipulate
Fax	*		
Route	*		Save
Routing Paramet	ers		
IP->PSTN			
PSTN->IP			



 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.

📸 SIP 🛛 👻 🔜				Routing Rules	(j			
PCM      Ch	eck 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	
{ĝ} Fax ♥ Che	k All 🗄 Unched	ck All Inverse	Delete E Clear A	Л				Add New
Route A 1 Items	Total 20 Items/Pa	age 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	*									
📑 SIP	*					Routing Rules				
PCM	*	Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Modify
ISDN	*		63	PCM Trunk Group [0]	*	9	none	SIP Trunk Group [0]	from_PSTN	
Eax Fax	*	Check All	Unched	k All 🗧 Inverse 🚍 🗐	lelete 🗧 Clear Al					Add New
Route	*	1 Items Total	20 Items/Pa	ige 1/1 First Previous Next I	Last Go to Page 1 💌	1 Pages Total				
Routing Paramete	ers									
IP->PSTN										
PSTN->IP		• T								

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.



console port; or it may work abnormally.

# **Appendix A Technical Specifications**

### Dimensions

190×30×120 mm<sup>3</sup>

### Weight

About 0.65 kg

### Environment

Operating temperature: 0 °C—40 °C Storage temperature: -20 °C—85 °C Humidity: 8%— 90% non-condensing Storage humidity: 8%— 90% non-condensing

#### LAN

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

### E1/T1 Port

Amount: 1/2

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 Power Requirements Input voltage: 12V DC ±10% Input power: ≥3A DC Maximum power consumption: ≤8W Signaling & Protocol 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
SILK(16K)	20 <i>kbps</i>
OPUS(16K)	20 <i>kbps</i>
SILK(8K)	20 <i>kbps</i>
OPUS(8K)	20 <i>kbps</i>

### 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 ISDN Pending Cause to SIP

# **Status Code**

ISDN Return Value	Cause	SIP Status Code	Implication
1	1 Unallocated (unassigned) number		Not found
2	2 No route to specified transit network		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
16	Normal call clearing	500	Decline
17	User busy	486	Busy here
132	Network busy (internal definition, only applies to 486		Busy here
21	Call rejected	486	Busy here
18	No user responding	408	Request timeout
19	No answer from user (user alerted)	480	Temporarily unavailable
20	Subscriber absent	480	Temporarily unavailable
31	Normal, unspecified	480	Temporarily unavailable
136	Connection after pickup failed (internal definition, only applies to ISDN)	480	Temporarily unavailable
137	Pickup time out (internal definition, only apply to ISDN)	480	Temporarily unavailable
55 Incoming calls barred within CUG		403	Forbidden
57	Bearer capability not authorized	403	Forbidden
87	User not member of CUG	403	Forbidden
22	Number changed	410	Gone
27	Destination out of order	502	Bad gateway
28	Invalid number format	484	Address incomplete
29 Facility rejected		501	Not implemented
79	79 Service or option not implemented, unspecified		Not implemented
34	No circuit/channel available	503	Service unavailable



38	Network out of order	503	Service unavailable
	Temporary failure Switching equipment congestion	503	Service
41			unavailable
42			Service
	Resource unavailable, unspecified	503	unavailable
47			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
100	ISDN)		unavailable
134	Temporary fault (internal definition, only applies to	503	Service
134	ISDN)	503	unavailable
135	Data link failure (internal definition, only applies to	502	Service
155	ISDN)	503	unavailable
65		400	Not acceptable
65	Bearer capability not implemented	488	here
70	Only restricted digital information bearer capability	100	Not acceptable
70	is available	488	here
102	Recovery on timer expiry	504	Server time-out
100	T303 time out (internal definition, only applies to		
128	ISDN)	504	Server time-out
400	T304 time out (internal definition, only applies to	50.4	
129	ISDN)	504	Server time-out
400	T310 time out (internal definition, only applies to	504	Server time-out
130	ISDN)		
	Protocol error, unspecified	500	Server internal
111			error
	Interworking, unspecified	500	Server internal
127			error
Others	Others	408	Request timeout
	•		•



## Appendix D Direction for CDR Use

CDR is a call detail record. The digital gateway can record the CDR to the memory and send them to the designated server in real time.

Methods:

- 1. By using the TCP protocol, the gateway works as a client to configure a CDR server, and then sends the CDR to the server regularly.
- 2. The gateway sends the CDR to the server every 3 seconds.
- 3. The gateway will connect the CDR server again every 30 seconds if lossing connection from it.
- 4. There are up to 2000 pieces of CDR saved in the server, and the first 100 pieces of the record will be deleted once the pieces exceed 2000.
- 5. Example CDR format:

### Outgoing example:(ip->pstn)

"2014-12-20 14:55:33.345", "2014-12-20 14:57:43.627", "1000", "5551234", "SIP/1000", "Zap/444", "", ""

### Incoming example:(pstn->ip)

"2014-12-20 14:55:33.345", "2014-12-20 14:57:43.627", "5551234", "1000", "Zap/444", "SIP/1000", "1234", ""

#	Field Name	Format	Description
1	Start Time	YYYY-MM-DD HH:MM:SS.mmm	Call start timestamp
2	End Time	YYYY-MM-DD HH:MM:SS.mmm	Call end timestamp
3	Calling Number (A)		Calling Number
4	Dialed Number (B)		Dialed Number
5	Incoming Call Leg		Incoming Call Leg
6	Outgoing Call Leg		Outgoing Call Leg
7	DNIS		DNIS (incoming only)
8	Queue		Queue (incoming only)



# **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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