

Synway SMG-B Series Analog Gateway

SMG1004B SMG1008B SMG1016B4 SMG1032B4

Analog Gateway

User Manual

Version 1.0

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



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Chapter 1 Product Introduction

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

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

Module	Amount of FXS Port	Amount of FXO Port
SMG1004B-4S	4	0
SMG1004B-2S2O	2	2
SMG1004B-4O	0	4
SMG1008B-8S	8	0
SMG1008B-4S4O	4	4
SMG1008B-8O	0	8
SMG1016B4-16S	16	0
SMG1016B4-8S8O	8	8
SMG1016B4-16O	0	16
SMG1032B4-32S	32	0
SMG1032B4-24S8O	24	8
SMG1032B4-16S16O	16	16
SMG1032B4-32O	0	32

See below table for the modules of SMG-B series analog gateway:

Table 1 Model List

Note: The modules written in black are supported, while those in gray are not yet supported.



1.1 Typical Application



Figure 1-1 Typical Application

1.2 Feature List

Basic Features	Description
TDM Call Call initiated from TDM to IP, via routing and number manipulation to o called IP address.	
<i>IP Call</i> Call initiated from IP to TDM, via routing and number manipulation to obtain destination.	
Number Manipulation	Peels off some digits of a phone number from left/right, or adds a prefix/suffix to a phone number.
Call Forward	Three options available: Unconditional, Busy and No Reply.
Call Waiting	When an FXS channel receives another call while it is in conversation, it will have the newly received call keep waiting. Once the current call is finished, the new one will ring the FXS channel and wait for its answer.



Auto Dial	If there is no dialing operation in a designated time period after pickup, the preset auto dial number will be called.			
Do Not Disturb	Rejects all the incoming calls to the channel.			
CID	Displays the CallerID.			
Echo Cancellation	Provides the echo cancellation feature for a call conversation over the FXS channel.			
TDM/VoIP Routing	Sets a routing path: from IP to TDM or from TDM to IP.			
Fax	Provides multiple fax parameters: fax mode, maximum fax rate, fax train mode, error correction mode, etc.			
Send Polarity Reversal Signal	Sends the polarity reversal signal to a corresponding FXS channel when the called party pick-up behavior is detected.			
Simultaneous Register to Multiple Servers	Registers the gateway to a master registrar server and a spare registrar server simultaneously.			
IMS Network	Registers the gateway to a server under IMS network.			
Group Ringing	Rings all the idle FXS ports in a port group.			
Ringing by Turns	Rings the FXS ports in a port group by turns according to the <i>Rule for Ringing by Turns</i> .			
Preemptive Answer	When a channel in a port group is ringing, another channel in the same port group can press the preemptive answer keyboard shortcut to transfer the call from the ringing channel to the current channel.			
Signaling & Protocol	Description			
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261.			
Voice	CODEC G.711A, G.711U, G.729A/B, G.723, G.722, AMR, iLBC DTMF Mode RFC2833, SIP INFO, INBAND			
Network	Description			
Network Protocol	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN.			
Static IP	IP address modification support.			
DHCP	IP address dynamic allocation support.			
DNS	Domain Name Service support.			
Security	Description			
Admin Authentication	Supports admin authentication to guarantee the resource and data security.			
System Monitor	Monitors the running status of the system and the server.			
Maintain & Upgrade Description				
WEB Configuration	Support of configurations through the WEB user interface.			
Language	Chinese, English.			



Software Upgrade	Support of user interface, gateway service, kernel and firmware upgrades based on WEB.	
Tracking Test	Support of Ping and Tracert tests based on WEB.	
SysLog Type	Three options available: ERROR, WARNING, INFO.	

1.3 Hardware Description

1.3.1 SMG-B 4-port Analog Gateway (Unsupported)

The SMG-B 4-port analog gateway has three types: SMG1004B-4S (4 FXS ports), SMG1004B-2S2O (2 FXS ports and 2 FXO ports) and SMG1004B-4O (4 FXO ports). It supports one LAN and adopts an external 12V power supply.

1.3.2 SMG-B 8-port Analog Gateway

The SMG-B 8-port analog gateway has three types: SMG1008B-8S (8 FXS ports), SMG1008B-4S4O (4 FXS ports and 4 FXO ports) and SMG1008B-8O (8 FXO ports). It supports one LAN and adopts an external 12V power supply. See below for product appearance (taking SMG1008B-8S for example).



Figure 1-2 SMG1008B-8S Front View



Figure 1-3 SMG1008B-8S Rear View

1.3.3 SMG-B 32-port Analog Gateway (Unsupported)

The SMG-B 32-port analog gateway has four types: SMG1032B4-32S (32 FXS ports), SMG1032B4-24S8O (24 FXS ports and 8 FXO ports), SMG1032B4-16S16O (16 FXS ports and 16 FXO ports) and SMG1032B4-32O (16 FXO ports). It supports one LAN.





Figure 1-4 SMG1016B4-16S Front View



Figure 1-5 SMG1016B4-16S Rear View



Figure 1-6 SMG1016B4-16S Left View

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

Interface	Description		
	Amount: 1		
	Type: RJ-45		
	Bandwidth: 10/100 Mbps		
LAN	Self-Adaptive Bandwidth Supported		
	Auto MDI/MDIX Supported		
	Built-in Link indicator and ACTIVE indicator. For more details, refer to 1.4 Indicator		
	Info		
	Amount: Depends on the model of SMG-B analog gateway		
EXC.	Type: RJ-11, RJ-45		
FXS	Maximum Transmission Distance: 5000m		
	Charge Mode: Negative Anti-billing Supported		
Console Port	Amount: 1		
	Type: RS-232		
	Baud Rate: 115200bps		
	Connector: RJ45 to DB-9 Connector		
	Data Bits: 8 bits		



	Stop Bit: 1 bit		
	Parity Unsupported		
	Flow Control Unsupported		
External Power Supply	Provide the 12V voltage with positive inside and negative outside, and the current		
Interface	is larger than 3A		
Button	Description		
Deset Detter	Restore the gateway to factory settings by pressing this button persistently for 3		
Reset Button	seconds		
LED	Description		
Power Indicator	Indicates the power state. It lights up when the gateway starts up with the power		
Power Indicator	cord well connected		
Run & Alarm Indicator	Indicates the running status. For more details, refer to <u>1.4 Indicator Info</u> .		
Naturalla India ata r	Indicates the connection status of the network. For more details, refer to 1.4		
Network Indicator	Indicator Info		
	1. When the channel is idle, the LED Lights up;		
Channel Indicator	2. When the channel is off-hook, the LED flashes slowly;		
	3. When the channel is ringing, the LED flashes fast.		

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

1.3.4 SMG-B 16-port Analog Gateway

The SMG-B 16-port analog gateway has three types: SMG1016B4-16S (16 FXS ports), SMG1016B4-8S8O (8 FXS ports and 8 FXO ports) and SMG1016B4-16O (16 FXO ports). It supports one LAN. See below for product appearance (taking SMG1016B4-16S for example).

1.4 Indicator Info

The SMG-B analog gateway is equipped with two indicators denoting the system's running status: Run & Alarm Indicator (bi-color LED) and Network Indicator (bi-color LED). The table below explains the states and meanings of the two indicators.

LED	State	Description		
	Go out	System is not yet started.		
Dura 6 Alarma	Orange LED light up	Device works normal upon system startup.		
Run & Alarm Indicator	Orange LED flash fast	Device works normal upon system startup.		
maicator	Green LED flash slowly	Device works normal during system runtime.		
	Others	Device works abnormal		
	Go out	No network connection		
Network Indicator	Red LED light up and flash	Bandwidth 10 Mbps		
	Orange LED light up and flash	Bandwidth 100Mbps		
	Go out	Network is not yet connected or the network connection is 10Mbps		
LINK Indicator	Green LED light up	Connect to 100Mbps network, and network connection is normal.		
ACTIVE Indicator Go out Connect to		Connect to 10Mbps network: communication is normal; While		



		connect to 100Mbps network: Communication is abnormal.	
	Orange LED light up and flash	Communication is normal.	

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, the run & alarm indicator is orange and lights up or flashes. Then after the gateway service is successfully started and the device begins to work normally, the run & alarm indicator is green and flashes slowly.
- During runtime, if the run & alarm indicator is orange and 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 C 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-B analog gateway in the shortest time.

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

- SMG-B Series Analog Gateway *1
- External 12V Power Adapter *1 (Unnecessary for SMG-16B)
- Network Cable *1
- Warranty Card *1
- Installation Manual *1

Step 2: Connect the network cable.

These series products provide RJ-45 interfaces.

Step 3: Connect the telephone line.

The SMG-B 8-port analog gateway provides RJ11 interfaces. You can use a common telephone line directly or construct a telephone line by yourself according to Figure 2-1. Note that only the middle two cores in the RJ11 jack are valid for use.



Figure 2-1 RJ11 Connection

The SMG-B 16-port analog gateway has four 8-pin RJ45 jacks each of which can be connected to four 2-pin RJ11 jacks via a 4-way hub. Take the first RJ45 jack for example, the matching relationship among the channel number, the pins of the RJ45 jack and the 4-way hub is shown in the table below.

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

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

Step 4: Power on and start the gateway.

To use the SMG-B 8-port analog gateway, you need an external power supply. Insert it to the power interface of the SMG-8B series analog gateway and power it on with 100~240V AC. See the figure below:





Figure 2-2 SMG-B Power Connection

Step 5: Log in the gateway.

Enter the original IP address (192.168.1.101) of the SMG-B analog gateway in the browser to go to the WEB interface of the gateway. The original username and password of the gateway are both 'admin'. For detailed instructions about login, refer to <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>3.9.6 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>3.9.2 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: Make phone calls.

Note: For your easy understanding and manipulation, all examples given in this step do not involve registration, that is, SIP initiates calls in a point-to-point mode.

Situation 1: Call from a station to another (Tel→Tel)

The gateway allows two FXS ports to call each other by default. Just use a station connected with an FXS port to dial the number of the destination FXS port and you can make a Tel \rightarrow Tel call. The default number of an FXS port is 80XX, among which XX represents the corresponding port number. For example, the default number corresponding to Port 1 is 8001, and that corresponding to Port 8 is 8008.

Actually a Tel \rightarrow Tel call on the gateway is accomplished via the routing of Tel \rightarrow IP \rightarrow Tel. For detailed introductions and configuration guide, refer to <u>Q2</u> in Appendix B.

Situation 2: Call from a station to an IP phone (Tel→IP)

Go to 'Advanced Settings → Dialing Rule' on the WEB interface and click the 'Add New' button to add a new dialing rule. Refer to <u>3.5.8 Dialing 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 of 'Index' and are



required not to leave 'Description' empty.

Example: Set Index to 99, fill in Description with test and configure Dial Rule to 123.

Go to 'Port Settings → Port Group' on the WEB interface and click the 'Add New' button to create a new port group and add FXS ports which are connected with stations to it. Refer to <u>3.6.2 Port Group</u> for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

Example: Provided the FXS port which is connected with a station is Port1, check the checkbox before **Port1**, set **Index** to **1**, fill in **Description** with **test**, and keep the default values of other configuration items.

3. Go to 'Route Settings → Tel→IP' on the WEB interface and click the 'Add New' button to add a new routing rule. Refer to <u>3.7.3 Tel→IP</u> for detailed instructions. Select the port group created in Step2 as 'Source Port Group' and fill in 'Destination IP' and 'Destination Port' with the IP address and the Port number you plan to call. You may use the default values of other configuration items and are required not to leave 'Description' empty.

Example: Provided the remote IP address intended to call is 192.168.0.111 and the port is 5060. Set **Index** to **63**, **Source Port Group** to **1**, fill in **Description** with **test**, configure **Destination IP** to **192.168.0.111**, **Destination Port** to **5060**, and keep the default values of other configuration items.

4. Pick up the station and dial the number set in Step1 to ring the remote IP phone. If you have set a particular number in Step 1, only this number you can dial; if you have set a string of 'x's, how many 'x's there are, how many random numbers you can dial.

Example: Pick up the station and dial 123. Then the IP phone with the IP address 192.168.0.111 and the port 5060 will ring.

Situation 3: Call from an IP phone to a station (IP \rightarrow Tel)

 Go to 'Port Settings → Port Group' on the WEB interface and click the 'Add New' button to create a new port group and add FXS ports which are connected with stations to it. Refer to <u>3.6.2 Port Group</u> for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

Example: Provided the FXS port which is connected with a station is Port1, check the checkbox before **Port1**, set **Index** to **1**, fill in **Description** with **test**, and keep the default values of other configuration items.

 Go to 'Route Settings → IP→Tel' on the WEB interface and click the 'Add New' button to add a new routing rule. Refer to <u>3.7.2 IP→Tel</u> for detailed instructions. Fill in 'Source IP' with the IP address which initiates the call and select the port group created in Step1 as 'Destination Port Group'. You may use the default values of other configuration items and required not to leave 'Description' empty.

Example: Provided the IP address of the IP phone which initiates the call is 192.168.0.111. Set **Index** to **63**, **Destination Port Group** to **1**, fill in **Description** with **test**, configure **Source IP** to **192.168.0.111**, and keep the default values of other configuration items.

3. Pick up the IP phone and call the IP address and port of the SMG-B analog gateway to ring the station.

Example: Provided the IP address of the SMG-B analog gateway is 192.168.0.101 and the port is 5060, use the IP phone to call the IP address 192.168.0.101 and the station connected with Port1 will ring.

Step 8: Enable the auto dial feature. (Skip this step if not necessary.)

Go to the Port Settings interface to enable the auto dial feature and set the parameters 'Auto Dial Number' and 'Wait Time before Auto Dial'. If there is no dialing operation in a time period (i.e. Wait Time before Auto Dial) after pickup, the port will automatically call the preset number (i.e. Auto Dial Number). Refer to <u>3.6.1 FXS</u> for detailed instructions.



Step 9: Enable the DND (do not disturb) feature. (Skip this step if not necessary.)

Go to the Port Settings interface to enable the DND feature. Then, the FXS port will reject all incoming calls. Refer to <u>3.6.1 FXS</u> for detailed instructions.

Step 10: Enable the call waiting feature. (Skip this step if not necessary.)

Go to the Port Settings interface to enable the call waiting feature. Then the corresponding FXS port while in conversation can accept another call from IP and keep it in the waiting state. Once the current conversation is finished and the station hangs up, the call in the waiting state will ring the station and wait for answer. During the time in the waiting state, it will always hear the ringback tone from the FXS port. Refer to <u>3.6.1 FXS</u> for detailed instructions.

Step 11: Perform call forwarding. (Skip this step if not necessary.)

Situation 1: Hook-flash operation



Figure 2-3 Call Forward via Hook-flash

As shown above, Remote A initiates and establishes a call with Station. Then by a hook-flash operation, that is, a rapid clap on the hook or pressing the 'flash' button on the phone set, Station can forward the call to Remote B.

Once a flash is generated, Station will go into the dialing state (the FXS port sends it dialing tones) before it dials the forwarding number.

If the dialing succeeds, the FXS port will send ringback tones to Station. Provided Remote B picks up the call, at this time Station can:

- a) Directly talk with Remote B;
- b) Perform another hook-flash operation to switch the call to either Remote A or Remote B.
- c) Hang up to make Remote A and Remote B go into a direct talk with each other.

If the dialing fails, the FXS port will send busy tones to Station. At this time Station can:

- a) Hang up to go back to the ringing state; then pick up the call again to recover the talk with Remote A.
- b) Perform the hook-flash operation again without hanging up the call to recover the talk with Remote A.

Once Station recovers the call with Remote A, it can forward the call again by a new hook-flash operation.

Situation 2: Automatic call forward

Go to the port setting interface to enable the automatic call forward feature and fill in a forward number. According to what you set, the SMG-B analog gateway can automatically forward the incoming calls on three conditions: unconditional, busy, no reply. Note that this feature is applicable only to a single port, but not to a port group consisting of more than one port. Refer to <u>3.6.1 FXS</u> for detailed instructions.

Special Instructions:

• As the device will gradually heat up while being used, please maintain good ventilation to prevent sudden failure, ensuring that the ventilation holes 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 Security	X				
The server 192. 168. 1. 101 at SMG requires a username and password. Warning: This server is requesting that your username and password be sent in an insecure manner (basic authentication without a secure connection).					
	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>3.9.6 Change Password</u>.

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

Operation Info	8		Systen	ninfo	
System Info			oyoton	1.110	
Channel State		LAN			
Call Count		MAC Address	00:03:19:00:20:01		
		IP Address	192.168.1.101	255.255.255.0	192.168.1.
Quick Config	*	DNS Server	0.0.0.0		
	Č	Receive Packets	All:309	Error:0	Drop:0
S VolP	*	Transmit Packets	All:344	Error:0	Drop:0
		Current Speed	Receive:119 B/s	Transmit:0 B/s	
Advanced	*	Work Mode	100Mb/s Full Duplex		
Dort	*	Runtime	85		
Route	*				
		Current Version			
Num Manipulate	*	WEB	1.0.0_2015051214		
System Tools	*	Gateway	1.0.0_2015051214		
Jojoteni roolo		Serial Num	00001452		
		U-boot	Apr 03 2015-09:48:30		
		Kernel	#179 Mon May 11 13:48	3:32 PDT 2015	
		Product Type	1008B-8S (RJ11)		

Figure 3-2 Main Interface



3.2 Operation Info

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



Figure 3-3 Operation Info

3.2.1 System Info

LAN			
MAC Address	00:03:19:00:20:01		
IP Address	192.168.1.101	255.255.255.0	192.168.1.1
DNS Server	0.0.0.0		
Receive Packets	All:309	Error:0	Drop:0
Transmit Packets	All:344	Error:0	Drop:0
Current Speed	Receive: 119 B/s	Transmit0 B/s	
Work Mode	100Mb/s Full Duplex		
Runtime	8s		
Current Version			
WEB	1.0.0_2015051214		
Gateway	1.0.0_2015051214		
Serial Num	00001452		
U-boot	Apr 03 2015-09:48:30		
Kernel	#179 Mon May 11 13:4	8:32 PDT 2015	
Product Type	1008B-85 (RJ11)		

Refresh

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



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

3.2.2 Channel State

				Channel	State			
Channel	Type	Number	Voltage(v)	State	Direction	CallerID	CalleeID	Reg Status
1	FXS	8001	0		Ci rris k			Unregistered
2	FXS	8002	0		2. <u>111</u> 2			Unregistered
3	FXS	8003	0		· · · · · ·			Unregistered
4	FXS	8004	0		()			Unregistered
5	FXS	8005	0		(1997)			Unregistered
6	FXS	8006	0		20 <u>80.89</u> 3	<u>800</u> 1		Unregistered
7	FXS	8007	0					Unregistered
8	FXS	8008	0					Unregistered

Figure 3-5 Channel State Interface

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

ltem			Description		
Channel	Channel number on the device.				
Туре	Type of the channel on the device. If this item shows, it means this channel is unavailable, that is, the corresponding module to this channel is not inserted or damaged.				
Number	The number cor	The number corresponding to the port.			
Voltage	Line voltage on the channel, calculated by volt (V).				
	Displays the channel state in real time. You can move the mouse onto the channel state icon for detailed state information.				
	State Icon Desc		Description		
	Idle		The channel is available.		
State	Off-hook		The channel picks up the call.		
	Wait Answer		The channel receives the ringback tone and is waiting for the called party to pick up the phone.		
	Ringing		The channel is in the ringing state.		



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	Talking . The channel is in a conversation.			
	Dialing Contract The channel is dialing.			
	Pending The channel is in the pending state.			
	Internal State 🖸 Internal state of the channel.			
	Unusable 🚺 The channel is unavailable.			
Direction	Displays the direction of the call on channel.			
CallerID	Displays the CallerID of the call on channel.			
CalleelD	Displays the CalleeID of the call on channel.			
Reg Status	Displays the registration status of the port.			

3.2.3 Call Count

Call Count								
all Direction	Total Calls	Successful Calls	Busy	No Answer	Call Forward	Routing Failure	Dialing Failure	Unknown Failure
IP->Tel	0	0	0	0	0	0	0	0
Tel->IP	0	0	0	0	0	0	0	0

Figure 3-6 Call Count Interface

See Figure 3-6 for the call count Interface. The above list shows the detailed information about all the calls counted from the startup of the gateway service to the latest open or refresh of this interface. You can click **Refresh** to obtain the current call count information. The table below explains the items shown in Figure 3-6.

ltem	Description
Call Direction	A condition for call count, two options available: $IP \rightarrow Tel$ and $Tel \rightarrow IP$.
Total Calls	Total number of calls in a specified call direction.
Successful Calls	Total number of successful calls in conversation.
Durau	Total number of calls which fail as the called party has been occupied and replies a
Busy	busy message.
	Total number of calls which fail as the called party does not pick up the call in a long
No Answer	time or the calling party hangs up the call before the called party picks it up.
Call Forward	Total number of calls which have been forwarded.
Routing Failure	Total number of calls which fail because no routing rules are matched.
	Total number of calls which fail as the called party number does not conform to the
Dialing Failure	dialing rule or due to dialing timeout.
Unknown Failure	Total number of calls which fail due to unknown reasons.

3.3 Quick Config



Figure 3-7 Quick Config Interface

See Figure 3-7 for the Quick Config interface. Follow the gateway Quick Configuration wizard and



you can easily complete the settings on network, SIP and FXS. The gateway can work normally after configuration.

See Figure 3-8 for the Quick Config-Network Settings interface. Refer to 3.9.2 Network for detailed settings. After configuration, click *Next* to enter the SIP Settings interface.

Network Type:	Static
IP Address (I)	192.168.1.101
Subnet Mask (U)	255.255.255.0
Default Gateway (D)	192.168.1.1
DNS Server (P)	0.0.00
Speed and Duplex Mode	Automatic Detection

Figure 3-8 Quick Config-Network Settings Interface

See Figure 3-9 for the Quick Config-SIP Settings interface. The configuration items on this interface are the same as those on the SIP interface. Refer to <u>3.4.1 SIP</u> for detailed settings. You are required to fill with the information about the registrar if the gateway must be registered. After configuration, click **Back** to go back to the Network Settings interface; click **Next** to enter the FXS Settings interface.

Quick Config-SIP	Settings
Registrar IP Address Registrar Port	
Spare Registrar IP Address Spare Registrar Port	
Registry Validity Period (s)	600
Back	Next

Figure 3-9 Quick Config-SIP Settings Interface

See Figure 3-10 for the FXS Settings interface. The configuration items on this interface are the same as those on the FXS interface. Refer to <u>3.6.1 FXS</u> for detailed settings. After configuration, click **Back** to go back to the SIP Settings interface; click **Next** to enter the Quick Config-Completion interface.



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					FXS Settin	gs						
ort	Туре	SIP Account	Authentication Username	Auto Dial Num	Forbid Outgoing Call	DND	Forward	FWD Type	FWD Number	CID	Call Waiting	Reg Sta
1	FXS	8001			Disable	Disable	Disable	: 		Enable	Disable	Unregiste
2	FXS	8002	3 <u>819</u> 3		Disable	Disable	Disable	2 <u></u> 2		Enable	Disable	Unregiste
3	FXS	8003			Disable	Disable	Disable			Enable	Disable	Unregiste
4	FXS	8004			Disable	Disable	Disable	00		Enable	Disable	Unregist
5	FXS	8005			Disable	Disable	Disable	00		Enable	Disable	Unregist
6	FXS	8006	20123		Disable	Disable	Disable		-	Enable	Disable	Unregist
7	FXS	8007			Disable	Disable	Disable	2 -2		Enable	Disable	Unregist
8	FXS	8008			Disable	Disable	Disable			Enable	Disable	Unregist
1										<i></i>		>
				Q	uick Config-C	omple	tion					

Figure 3-11 Quick Config-Completion Interface

Click **Back** to go back to the FXS Settings interface; click **Finish** to finish the Quick Config wizard and now the gateway can work normally with basic configuration.

3.4 VoIP Settings

VoIP Settings includes five parts: *SIP*, *SIP Compatibility*, *SIP Server*, *NAT Setting* and *Media*. See Figure 3-12. *SIP Settings* is used to configure the general SIP parameters, *SIP Compatibility* is used to set which SIP servers and SIP messages will the gateway be compatible with, *SIP Server* is to set the basic information of the SIP server, *NAT Setting* is used to configure the parameters for NAT, and *Media Settings* is to set the RTP port and the payload type.

📑 VolP	*
SIP	
Sip Compatibility	
NAT Setting	
Media	

Figure 3-12 VoIP Settings



3.4.1 SIP

SIP Port	5060
Sir Poit	5050
Register Status	Unregistered
Register Gateway	Yes 💌
SIP Account	
Password	
Authentication Username	
Registrar IP Address	
Registrar Port	
Spare Registrar Server	Enable
Spare Registrar IP Address	
Spare Registrar Port	
Registry Validity Period (s)	600
Registry validity Period (S)	600
Multi-Registrar Server Mode	Enable
SIP Transport Protocol	
IMS Network	Enable
Externally Bound Address	
Externally Bound Port	5060

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>3.9.9 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-13.

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



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

3.4.2 SIP Compatibility

See Figure 3-14 for the SIP Compatibility interface where you can configure the SIP parameters to determine which SIP servers and SIP messages will the gateway be compatible with. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations.



Sip Compatit	bility
Obtain CalleeID from	"Request" Field 💌
Set CallerID position	Username of From Field 💙
Obtain CallerID from	Username of From Field 😪
Call Transfer Mode	Internal Handling 💌
Call Flash Mode	Platform to Handle SIP II 💌
Hold Music Source	Remote
Two Stage Dialing for SIP Incoming Call	Enable
Maximum Wait Answer Time (s)	60
Set SIP Identifying	Gateway
Call Abnormal Hangup Detection Cycle(s)	✓Enable 0
Server Status Detection Cycle(s)	✓Enable 0
RTP Encryption	Enable
Ignore ACK	Enable
Save	Reset

Figure 3-14 SIP Compatibility Setting Interface

The table below explains the items shown in Figure 3-14.

Item	Description
Obtain CalleelD	There are two optional ways to obtain the called party number: from "To" Field and
from	from "Request" Field. The default value is "Request" Field.
	There are two options to set the position of the calling party number: "Displayname
Set CallerID Position	of From Field' and "Username of From Field'. The default value is "Username of
	From Field'.
	There are two optional ways to obtain the calling party number: from "Displayname
Obtain CallerID from	of From Field' and from "Username of From Field'. The default value is "Username
	of From Field".
Os II Turn of an Marke	There are two optional ways to deal with call transfer: Internal Handling and
Call Transfer Mode	Platform to Handle SIP Info. The default value is Internal Handling.



Call Flash Mode	There are two optional ways to deal with call flash: Internal Handling and Platform to
	Handle SIP Info. The default value is Internal Handling.
Hald Maria Carman	Sets the source of the hold music, with the default value of <i>Remote</i> , This feature
Hold Music Source	gets valid only when you choose the mode <i>Platform to Handle SIP Info</i> .
Two Stage Dialing	
for SIP Incoming	Once this feature is enabled, the incoming call from SIP should perform the two
Call	stage dialing operation. By default this feature is disabled.
	Sets the maximum time for the SIP channel to wait for the answer from the called
Maximum Wait	party of the outgoing call it initiates. If the call is not answered within the specified
Answer Time	time period, it will be canceled by the channel automatically. The default value is 60,
	calculated by s.
	Sets the SIP identifying content in the SIP call message. The default setting is
Set SIP Identifying	Gateway.
A	Sets the interval between checks of the remote end's abnormal hangup, with the
Abnormal Hangup	default value of 0 (feature disabled), calculated by s. It is suggested to set to 10s if
Detection Cycle	this feature is necessary to be used.
	The interval of sending a heartbeat packet to detect the master registrar server
Server Detection	status, with the default value of 0 (feature disabled), calculated by s. It is suggested
Cycle	to set to 15s if this feature is necessary to be used.
	Once this feature is enabled, you can encrypt the RTP package. By default it is
RTP Encryption	disabled.
	Once this feature is enabled, the gateway is not necessary to wait for the ACK
Ignore ACK	message after sending the 200OK message to establish a call. By default it is
	disabled.

3.4.3 SIP Server

The gateway supports the multi-registrar server feature. Enable the feature of '*Multi-Registrar* **Server Mode**' on the <u>SIP</u> interface (see <u>3.4.1 SIP</u>) and you will see the item SIP Server under the VoIP Settings menu. Click '*SIP Server*' to go into the SIP Server interface. By default, there is no available SIP server. See Figure 3-15 below.

Operation Info	*	
🕂 Quick Config	*	
VolP	۲	No Available Registrar Server
SIP		Add New
Sip Compatibility		
SIP Server		
NAT Setting		
Media		

Figure 3-15 SIP Server Interface

Click *Add New* to add SIP servers manually. See Figure 3-16. You can configure basic SIP server information on this interface.



Index	1 💌
Description	defalut
Registrar IP Address	
Registrar Port	5060
Registry Validity Period (s)	600
IMS Network	Enable
Externally Bound Address	
Externally Bound Port	5060

Figure 3-16 Add New SIP Server

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

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

After configuration, click **Save** to save the above settings into the gateway or click **Cancel** to cancel the settings. See Figure 3-17 for the SIP server management interface.

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

Figure 3-17 SIP Server Management

Click *Modify* in the above figure to modify the configuration of the SIP server. See Figure 3-18.

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



Ind	ex	1	
De	scription	defalut	
Re	gistrar IP Address	201.123.115.233	
Re	gistrar Port	5060	
Re	gistry Validity Period (s)	600	
IMS	S Network	Enable	
Ext	ernally Bound Address	201.123.123.145	
Ext	ernally Bound Port	5060	

Figure 3-18 SIP Server Modification Interface

To delete a SIP server, check the checkbox before the corresponding index in Figure 3-17 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all SIP servers at a time, click the **Clear All** button in Figure 3-17.

3.4.4 NAT Setting

See Figure 3-19 for the NAT setting interface where you can configure the parameters for NAT. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations.

NAT S	ettings
STUN Server	Enable
NAT Type	Unknown
STUN Server Address	127.0.0.1
Mapping Address	
RTP Self-adaption	Enable
Rport	⊡Enable
Auto Detect NAT IP	Enable
Note: Auto Detect NAT IP:This feature only wo router.	rks cooperatively with the port mapping setting on
Save	Reset



Figure 3-19 NAT Setting Interface

The table below explains the items shown in Figure 3-19.

ltem	Description			
	Sets whether to enable the STUN server for NAT traversal. By default the STUN			
STUN Server	server is disabled.			
	Detected NAT (Network Address Translation) type. The gateway will return the NAT			
	type automatically in case STUN Server is enabled. It includes 9 types: unknown;			
NAT Type	no NAT; ConeNat; RestrictedNat; PortRestrictedNat; Symmetric NAT; Symmetric			
	NAT with firewall; can't detect over (fail to send detect message) and fail to detect			
	(No reply from the stun server).			
STUN Server				
Address	Address of the server for STUN traversal.			
	It should be filled in when there exists NAT or other mapping relationships which			
	leads to the failure of direct communication between the gateway and the			
	destination address, so as to ask the remote end to send signaling messages or			
Mapping Address	voice data to it during the signaling or voice communication between the gateway			
	and the destination.			
	Note: Once this item is filled out, it will be used as the first choice even if Rport and			
	NAT IP are enabled.			
	When this feature is enabled, the RTP reception address or port carried by the			
DTD Colf odertien	signaling message from the remote end, if not consistent with the actual state, will			
RTP Self-adaption	be updated to the actual RTP reception address or port. By default, this feature is			
	disabled.			
Pnort	When this feature is enabled, a corresponding Rport field will be added to the Via			
Rport	message of SIP. The default value is enabled.			
	When this feature is enabled, the gateway will parse the corresponding address			
Auto Dotoot NAT ID	and port in the message returned by Rport so as to use them for the following			
Auto Detect NAT IP	communication. By default, this feature is <i>disabled</i> .			
	Note: This feature gets valid only when Rport is enabled.			



3.4.5 Media

DTMF Transmit Mode	
RFC2833 Payload 101	
RTP Port Range 50000,50767	
Silence Suppression Disable	
JitterBuffer 20	
Voice Gain Output from IP (dB)	
AGC V Target Energy Threshold (dB) 0 Maximum Gain Threshold (dB) 48 Maximum Attenuation Threshold (dB) 0 Minimum Input Energy (dB) -60	
CODEC Priority	(libe)
Check Priority CODEC Packing Time Bit Rate I 1 G711A 20 64 I 2 G711U 20 64 I 3 G729 20 64 I 3 G723 30 6.3 I 5 G722 30 64 I 6 AMR 20 4.75 I 6 AMR 20 15.2	× × × × ×
Save Reset	

Figure 3-20 Media Settings Interface

See Figure 3-20 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>3.9.9 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-20.

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



	Supported RTP port range for the IP end to establish a call conversation, with the			
RTP Port Range	lower limit of 10000 and the upper limit of 60000 and the difference between larger			
	than 480. The default value is 50000-50767.			
	Sets whether to send comfort noise packets to replace RTP packets or never to			
Silence	send RTP packets to reduce the bandwidth usage when there is no voice signal			
Suppression	throughout an IP conversation. The optional values are Enable and Disable, with			
	the default value of <i>Disable</i> .			
	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:			
	20~200, calculated by ms, with the default value of 20.			
Voice Gain Output	Adjusts the gain of the voice output from IP. Range of value: -24~12, calculated by			
from IP	dB, with the default value of 0.			
	If the AGC (Automatic Gain Control) feature is enabled, the gateway will			
AGC	automatically adjust the input signal amplitude, increasing that of small signals and			
	decreasing that of large signals.			
Target Energy	Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the			
Threshold	default value of 0.			
Maximum Gain	Set the maximum gain threshold that will be applied to the signal. Range of value:			
Threshold	0~48, calculated by dB, with the default value of 48.			
Maximum	Pot the maximum attenuation that will be explicit to the sized. Dense of weber			
Attenuation	Set the maximum attenuation that will be applied to the signal. Range of value:			
Threshold	-42~0, calculated by dB, with the default value of 0.			
	Set the minimum threshold for the energy processed by AGC. Signals below this			
Minimum Input	threshold will not be processed by AGC. Range of value: -60~ -25, calculated by			
Energy	dB, with the default value of -60.			



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

3.5 Advanced Settings

Advanced Settings includes twelve parts: *FXS*, *Tone Detector*, *Tone Generator*, *DTMF*, *Ringing Scheme*, *Fax*, *Function Key*, *Dialing Rule*, *Dialing Timeout*, *Cue Tone*, *Color Ring* and *QoS*. See Figure 3-21. *FXS* is used to configure the general properties of the FXS port; *Tone Detector* is used to configure some properties of detected tones; *Tone Detector* is used to configure some properties of detected tones; *Tone Detector* is used to configure some properties of tones sent from gateway; *DTMF* is used to set the properties related to DTMF; *Ringing Scheme* is used to set the ringing scheme for the FXS port; *Fax* is used to configure multiple fax parameters; *Function Key* is used to set a cluster of combination keys for you to query a related number; *Dialing Rule* and *Dialing Timeout* are used to set the judging conditions for dialing; *Cue Tone* is used to set the gateway language for playing voice and the voice file used for the two-stage dialing; *Color Ring* is used to upload the color ring file which can be set as a ringback tone for an incoming call from IP to FXS port; *QoS* uses the differentiated services technology to increase the gateway's service quality.





Figure 3-21 Advanced Settings

3.5.1 FXS

FXS	
Voice Gain Output from FXS (dB)	0
Hook-flash Detection	Enable
Minimum Time (ms)	80
Maximum Time (ms)	700
CID Transmit Mode	FSK
Occasion to Send FSK CallerID	After the first ring 💌
Send Polarity Reversal Signal	Enable
Light Up Mode for Voice Message	Not Light Up 🛛 🖌
Save	leset

Figure 3-22 FXS Configuration Interface

See Figure 3-22 for the FXS configuration interface. The table below explains the items shown in the above figure.

Item	Description		
Voice Gain Output	Adjusts the gain of the voice output from the FXS port. Range of value: -24~12,		
from FXS	calculated by dB, with the default value of 0.		
Usel fleet Detection	Sets whether to enable the hook-flash detection feature or not, with the default		
Hook-flash Detection	setting of being disabled.		
	Time length for judging a flash operation. Only a hook-flash operation which lasts a		
Minimum Time	time more than the value of this configuration item will be regarded as a valid flash		
Minimum Time	operation. Range of value: 80~ <i>Maximum Time</i> , calculated by ms, with the default		
	value of 80.		



	Time length for judging a flash operation. Only a hook-flash operation which lasts a
	time less than the value of this configuration item will be regarded as a valid flash
Maximum Time	operation. Those lasting a time longer than the value of this configuration item will
	be regarded as hangup operations. Range of value: 32~2000, calculated by ms,
	with the default value of 700.
Minimum Time	The minimum time length for detecting whether the phone is on-hook or not. Range
Length of On-hook	of value: 64~2000, calculated by ms, with the default value of 64.
Detection	Note: This item is valid only when the item Hook-flash Detection is disabled.
	The mode adopted by the FXS port to send the CallerID. The optional values are
CID Transmit Mode	FSK and DTMF, with the default value of FSK.
Occasion to Send	Sets when to send the CallerID, before rings or after the 1 st Ring. The default value
FSK CallerID	is after 1 st Ring.
	Once this feature is enabled, the gateway will send the polarity reversal signal to a
Send Polarity	corresponding FXS channel when it detects the called party pick-up behavior. By
Reversal Signal	default, this feature is disabled.
Light Up Mode for	Sets the light up mode for the voice message of the phone, There are two options:
Voice Message	Not Light Up and Light Up by FSK, with the default value of Not Light Up.

After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to <u>3.9.9 Restart</u> for detailed instructions.

3.5.2 Tone Detector

Check	Index	Tone	Туре	The 1st Mid-frequency	The 2nd Mid-frequency	Duration at ON State	Duration at OFF State	Period Count	Modify
	0	Dial Tone	Continuous Tone	450	0	1500	0	0	
	1	Busy Tone	Periodic Tone	450	0	350	350	2	
	2	Ringback Tone	Periodic Tone	450	0	1000	4000	1	1

Figure 3-23 Tone Parameters Setting Interface

See Figure 3-23 for the Tone Parameters setting interface. At most three pieces of tone parameters are allowed to set. By default, there are already three pieces of tone parameters on the gateway which you can modify or delete according to your actual requirement.

Click *Modify* in Figure 3-23 to modify the tone parameter. See Figure 3-24 for the tone parameter modification interface.



Tone Par	ameters
Index:	0 🗸
Tone:	Dial Tone 💌
Туре:	Continuous Tone 🗸
The 1st Mid-frequency:	450
The 2nd Mid-frequency:	0
Duration at ON State:	1500
Duration at OFF State:	0
Period Count :	0
Save	Close

Figure 3-24 Modify Tone Parameter

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

Item	Description
Index	The unique index of each group of tone detectors.
Tone	There are three options: <i>Dial Tone</i> , <i>Busy Tone</i> and <i>Ringback Tone</i> .
Туре	There are two options: Continuous Tone and Periodic Tone.
The 1 st	The 1 st center frequency. Range of value: 300~3400, calculated by Hz. The default
Mid-frequency	value is 450.
The 2 nd	The 2 nd center frequency. Range of value: 0 or 300~3400, calculated by Hz. The
Mid-frequency	default value is 0.
Duration at ON State	The duration of tones at on state. The default setting: Dial Tone is 1500ms, Busy
	Tone is 350ms, Ringback Tone is 1000ms.
Duration at OFF	The duration of tones at off state. The default setting: Dial Tone is 0ms, Busy Tone is
State	350ms, Ringback Tone is 4000ms.
Period Count	Set the count of periods as the condition to determine a periodic tone. The default
	setting: Dial Tone is 0, Busy Tone is 2, Ringback Tone is 1.

To delete a piece of tone, check the checkbox before the corresponding index in Figure 3-23 and click the '*Delete*' button. *Check All* means to select all available items on the current page; *Uncheck All* means to cancel all selections on the current page; *Inverse* means to uncheck the selected items and check the unselected. To clear all tone at a time, click the *Clear All* button in Figure 3-23.


3.5.3 Tone Generator

Tone Generator		
Tone Energy (dB)	0	
Dial Tone 450/1500	FreqA/TimeA,FreqB+FreqC/TimeB Repeatedly play tones in turn: first, TimeA, a single tone with FreqA, then, Time B, a dual tone composed of FreqB and FreqC.	
Ringback Tone 450/1000,0/4000	FreqA+FreqB+FreqC/TimeA,FreqD/TimeB Repeatedly play tones in turn: first, TimeA, a triple tone composed of FreqA, FreqB and FreqC, then, TimeB, a single tone with FreqD.	
Busy Tone 450/350, 0/350	Note: The play time is calculated by ms and cannot be larger than 16383ms for each toneunit. A tone is allowed to contain at	
Call Wait Tone 450/200, 0/600, 450/200, 0/1000	most 5 different toneunits and 4 different frequencies, but the frequency and duration of the first toneunit cannot be 0. Frequency being 0 means the toneunit is a piece of silence.	
Save	Reset	

Figure 3-25 Tone Generator Setting Interface

See Figure 3-25 for the Tone Generator Setting interface. By default, there are three tones on it: Dial Tone—a single tone with 450HZ frequency, plays continuously; Ringback Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 1s play and 4s pause; Busy Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 350ms play and 350ms pause. Call Wait Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 200ms play and 600ms pause, then 200ms play and 1s pause. You can configure the tone generator manually. The exact explanation about the format and the meaning is described on the right of the interface. The value range of the tone energy herein above is -12~17, calculated by dB, with the default value of 0.



3.5.4 DTMF

DTMF Detect	tor
Minimum Energy Threshold (dB) Maximum Threshold of Signal Twist (dB)	-45
Input Signal Gain (dB)	0
DTMF Genera	ator
DTMF Energy (dB)	0
Duration at ON (ms)	100
Duration at OFF (ms)	32
Save	Reset

Figure 3-26 DTMF Detector Configuration Interface

See Figure 3-26 for the DTMF configuration, including two parts: DTMF Detector and DTMF Generator. The table below explains the items shown in the above figure.

Item	Description
Minimum Energy	Set the minimum energy threshold of the DTMF signal. Range of value: -96~-1. The
Threshold	default value is -45.
Maximum Threshold	Set the maximum threshold of the DTMF signal twist. Range of value: 0~12. The
of Signal Twist	default value is 3.
	Set the input gain of the DTMF signal. Range of value: -24 \sim 24, calculated by dB.
Input Signal Gain	The default value is 0.
	Energy of the DTMF signal sent by the gateway. Range of value: -12~17, calculated
DTMF Energy	by dB, with the default value of 0.
	Set the duration of the DTMF signal at ON state. Range of value: 0~16383,
Duration at ON	calculated by ms, with the default value of 100.
Dum tion of OEE	Set the duration of the DTMF signal at OFF state. Range of value: 0~16383,
Duration at OFF	calculated by ms, with the default value of 32.

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



3.5.5 Ringing Scheme

Operation Info	*	
🕂 Quick Config	*	
😤 VolP	*	No available ringing scheme
O Advanced	*	Add New
FXS		
FXO		
Tone Detector		
DTMF Detector		
Ringing Scheme		
Fax		
Function Key		
Dialing Rule		
Dialing Timeout		
Cue Tone		
QoS		

Figure 3-27 Ringing Scheme Configuration Interface

By default, there is no available ringing scheme on the gateway. See Figure 3-27. Click *Add New* to add a ringing scheme manually, see Figure 3-28.

Ringing Scheme	
Duration at ON:	1000
Duration at OFF (s):	4000
CallerID (separated by	.):
Save	Close

Figure 3-28 Add Ringing Scheme Interface

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

ltem	Description	
Duration at ON	The duration of the tone at ON state, with the default value of 1000ms.	
Duration at OFF	The duration of the tone at OFF state, with the default value of 4000ms.	
	After this setting, different ringing schemes will be executed on the FXS port	
CallerID	according to the set CallerID.	

After configuration, click *Save* to save the above settings into the gateway or click *Close* to cancel the settings. See Figure 3-29 for the saved ringing scheme.

		Ringing Scheme		
Check	Duration at ON (ms)	Duration at OFF (ms)	CallerID	Modify
	1000	4000	1234	
Check All Unche	eck All 🗄 Inverse 🗄 Delete 🗄 Clear All			Add Nev

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Figure 3-29 Ringing Scheme List

Click *Modify* in Figure 3-29 to modify the ringing scheme. See Figure 3-30 for the Ringing Scheme Modification interface. The configuration items on this interface are the same as those on the *Add New Ringing Scheme* interface.

Ringing Scheme		
Duration at ON:	1000	
Duration at OFF (s):	4000	
CallerID (separated by ','):		
12	34	
Save	Close	

Figure 3-30 Ringing Scheme Modification Interface

To delete a piece of ringing scheme, check the checkbox before the corresponding index in Figure 3-29 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 ringing scheme at a time, click the *Clear All* button in Figure 3-29.

3.5.6 Fax

Fax Parameters	
Fax Mode	Disable
Save	Reset

Figure 3-31 Fax Configuration Interface (Disable by default)

See Figure 3-31 for the default fax mode configuration. The table below explains the items shown in the above figure.

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

See Figure 3-32 for the fax configuration under the T.38 mode.



Fax Parameters	
Fax Mode	T.38
T38 Fax Port	Use Original Voice Port
T38 Version	0
T38 Negotiation	Initiate Negotiation as Fax Re 💌
Maximum Fax Rate (bps)	14400
Fax Train Mode	transferredTCF
Error Correction Mode	t38UDPRedundancy
Save	set

Figure 3-32 Fax Configuration Interface (T.38 Mode)

Users can configure the general fax parameters via this interface. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. The table below explains the configuration items in Figure 3-32.

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

If you set *Fax Mode* to *Pass-through*, you can see the interface shown as Figure 3-33.



Fax Para	meters
Fax Mode	Pass-Through
Pass-through Payload	102
Save	Reset

Figure 3-33 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.

3.5.7 Function Key

See Figure 3-34 for the function key configuration interface. Here you can set a cluster of combination keys to query a related number.

	Functio	AI 1391	
Function	Enable	Function Key	Mode
Device Function			
Query LAN		*11*	Default 🔽
Query Phone Number		*20*	Default 🔽
Phone Test		*30*	Default 🔽
Set LAN		*61*	Default
Service Available			
Blind Transfer		*010*	Default
Register		*020*	Default
Unregister		*021*	Default 💌
Query Register Status		*022*	Default 💌
	G		
	Save		

Figure 3-34 Function Key Configuration Interface

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

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

When the **Blind Transfer** feature is enabled, set a corresponding function key in the box behind. After you transfer a call by rapidly clapping on the hook switch, dial the set function key for **Blind**



Transfer and then the called party number. After that, hang up the call once hearing the howler tone to let the subsequent call procedure go out of your control.

3.5.8 Dialing Rule

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

Check	Index	Dialing Rule	Description	Modify
	81	400xxxxxxx	default	
	82	40[1-9]cocox	default	
	83	4[1-9]2000000	default	
	84	800xxxxxxx	default	
	85	80[1-9]xxxxx	default	
	86	8[1-9]x00000x	default	
	87	[2-3,5-7]xxxxxxxx	default	
	88	1[3-5,7-8]x0000000x	default	
	89	100xx	default	
	90	95xxx	default	
	91	123xx	default	
	92	111xx	default	
	93	11[0,2-9]	default	
	94	120	default	
	95	0[3-9]0000000000	default	
	96	0.2хосососох	default	
	97	010xxxxxxxxx	default	
	98	01[3-5,7-8]x00000000	default	
	99		default	

Figure 3-35 Dialing Rule Configuration Interface (Standard)

See Figure 3-35 for the Dialing Rule Configuration interface under the standard mode. The list in the above figure shows the dialing rules with their priorities and description, which can be added by the *Add New* button on the bottom right corner. See Figure 3-36 for the dialing rule adding interface.



Dialing Rule		
Index: 98 🗸]	
Description:]	
Dialing Rule:		
Save Close		

Figure 3-36 Add New Dialing Rule

The table below explains the items shown in Figure 3-36.

ltem		Description			
Indox	The unique ind	lex of each dialing rule, which denotes its priority. A dialing rule with a			
Index	smaller index v	value has a higher priority and will be checked earlier while matching.			
Description	Remarks for th	Remarks for the dialing rule. It can be any information, but can not be left empty.			
	Up to 99 dialing	g rules can be configured in the gateway, and the maximum length of			
	each dialing ru	le is 127 characters. See below for the meaning of each character in			
	the dialing rule	the dialing rule. The gateway will do instant matching for your dialing number based			
	on the dialing	rule and regard your dialing as finished upon receiving '#' or dialing			
	timeout.				
	Character	Description			
	"0"~"9"	Digits 0 \sim 9.			
	"A"~"D"	Letters A~D.			
	"x"	A random number. A string of 'x's represents several random			
	~	numbers. For example, 'xxx' denotes 3 random numbers.			
	""	'.' indicates a random amount (including zero) of characters after it.			
Dialing Rule		'[]' 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.			
(1, 3) 3		',' is used to separate numbers or number ranges, representing alternatives.			
		Only represents symbol "*".			
	"#"	Only set it at the beginning of the string, representing symbol "#".			
	There are 19	dialing rules already configured on the gateway for easy use. See			
	below for detai	led information.			
	Priority	Dialing Rule Description			



99	•	Any number in any length.
98	01[3-5,7-8]xxxxxxxx.	Any 12-digit number starting with 013, 014, 015, 017 or 018
97	010xxxxxxx	Any 11-digit number starting with 010
96	02xxxxxxxx	Any 11-digit number starting with 02
95	0[3-9]xxxxxxxxx	Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09
94	120	Number 120。
93	11[0,2-9]	Number 110, 112, 113, 114, 115, 116, 117, 118 or 119
92	111xx	Any 5-digit number starting with 111
91	123xx	Any 5-digit number starting with 123
90	95xxx	Any 5-digit number starting with 95
89	100xx	Any 5-digit number starting with 100
88	1[3-5,7-8]xxxxxxxx	Any 11-digit number starting with 13, 14, 15, 17 or 18
87	[2-3,5-7]xxxxxxx	Any 8-digit number starting with 2, 3, 5, 6 or 7
86	8[1-9]xxxxx	Any 8-digit number starting with 81, 82, 83, 84, 85, 86, 87, 88 or 89
85	80[1-9]xxxxx	Any 8-digit number starting with 801, 802, 803, 804, 805, 806, 807, 808 or 809
84	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

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

Click *Modify* in Figure 3-35 to modify the dialing rules. See Figure 3-37 for the dialing rule modification interface. The configuration items on this interface are the same as those on the *Add New Dialing Rule* interface.



Dialing Rule		
Index:	99 🗸	
Description:	test	
Dialing Rule:	XXX	
Save	Close	

Figure 3-37 Modify Dialing Rule

To delete a dialing rule, check the checkbox before the corresponding index in Figure 3-35 and click the '*Delete*' button. *Check All* means to select all available items on the current page; *Uncheck All* means to cancel all selections on the current page; *Inverse* means to uncheck the selected items and check the unselected. To clear all dialing rules at a time, click the *Clear All* button in Figure 3-35.

See Figure 3-38 for the Dialing Rule Configuration interface under the Character mode. You can edit the dialing rule list to add a new one or modify an old one. The exact meaning of each rule element is described on the page.

Dialing Rule	
Note: The Dialing Rule contains such fields as Dialing Rule and Description. The priority decreases from top to bottom; adjacent fields are separated by a space; Symbol . denotes any	otring
Don't forget to save the configuration after your modification!	sung.
400xxxxxx default	2
40[1-9]xxxxx default	
4[1-9]xxxxxx default	
800xxxxxx default	
80[1-9]xxxxx default	
8[1-9]xxxxxx default	
[2-3,5-7]xxxxxxx default	
1[3-5,7-8]xxxxxxxx default	
100xx default	
95xxx default	
123xx default	
111xx default	
11[0,2-9] default	
120 default	
0[3-9]xxxxxxxx default	s
20 Items Total	

Figure 3-38 Dialing Rule Configuration Interface (Character)

3.5.9 Dialing Timeout

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

Figure 3-39 Dialing Timeout Info Interface

See Figure 3-39 for the dialing 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 dialing number. Range of value:	
	1~10, calculated by s, with the default value of 6. In case your dialing rules do not	
	include ".", the call will fail if there is no digit dialed or no dialing rule matched during	
Inter Digit Timeout	this interval; in case your dialing rules include ".", the gateway will wait until this	
	interval ends and match to the dialing rule "." if there is no digit dialed or no other	
	dialing rule matched during this interval.	
More information about the configuration item Inter Digit Timeout,		
Description	reason for adopting the current value.	

Click *Modify* in Figure 3-39 to modify the dialing timeout info. See Figure 3-40 for the dialing timeout info modification interface. The configuration items on this interface are the same as those on the *Dialing Timeout Info Interface*.

Dialing Timeout		
Description:	example	
Inter Digit Timeout (s):	6	
Save	Close	

Figure 3-40 Modify Dialing Timeout Info

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

3.5.10 Cue Tone

	Cue	e Tone		
Language	[Chinese 💌		Save
	U	pload		
Upload a file of cue	File of oue tage for IVP		Browse	Upload
tone	File of cue tone for IVR	ampling rate 16.	bit mono	Browse
100KB in size.	e a wav file with 8000Hz s	ampling rate, 16-i	bit mono, A-law forn	natted,

Figure 3-41 Cue Tone Interface

See Figure 3-41 for the Cue Tone interface. The table below explains the items shown in the above figure.

Item	Description
Language	Sets the language for the gateway to play voice, including two options Chinese and



	English. The default setting is Chinese.
Upload a file of cue	Uploads a user-defined cue tone file to the gateway.
tone	opioaus a user-uenneu cue tone me to the gateway.

Click Save to save the above settings into the gateway.

3.5.11 Color Ring

Operation Info	*
🖳 Quick Config	*
VolP	×
Or Advanced	*
FXS	
FXO	
Tone Detector	
Tone Generator	
DTMF Detector	
Ringing Scheme	
Fax	
Function Key	
Dialing Rule	
Dialing Timeout	
Cue Tone	
Color Ring	

Figure 3-42 Coloring Ring Interface

By default, there is no available color ring on the gateway. See Figure 3-42. Click **Upload** to upload a new color ring manually. Follow Figure 3-43 to upload the required color ring file to the gateway.

Description	default
Color Ring	Browse.,,
Note: The file should be	a wav file with 8000Hz sampling rate, 16-bit mono, A-law formatted, and less th
100KB in size.	

Figure 3-43 Color Ring Upload Interface

The table below explains the items shown above:

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



After configuration, click **Upload** to upload the color ring file to the gateway or click **Return** to cancel the upload. See Figure 3-44 for the Color Ring Management interface after the upload.

		Color Ring Manage		
Check	Index	Color Ring	Port	Modify
	1	ringtone1		
Check All Uncheck All	E Inverse E Delete E Cl	lear All		Upload

Figure 3-44 Color Ring Management Interface

Click *Modify* in Figure 3-44 to modify the configuration of the color ring. See below for the color ring modification interface. The configuration items on this interface are the same as those on the *Color Ring Upload* interface.

	Color Ring-Modify
Index	1
Description	ringtone1
Upload	
2	
	Save Cancel

Figure 3-45 Color Ring Modification Interface

To delete a color ring, check the checkbox before the corresponding index in Figure 3-44 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all color rings at a time, click the **Clear All** button in Figure 3-45.

3.5.12 QoS

QoS	
QoS	Enable
Media Premium QoS	46
Control Premium QoS	26
Save	Reset

Figure 3-46 Differentiated Services Setting Interface

See Figure 3-46 for the Differentiated Services setting interface. Using this technology, the gateway can meet various application requirements under a limited bandwidth and ensure neither delay nor discard for important services so as to improve its quality of services.

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

Item	Description



QoS	Sets whether to enable the OoS differentiated services. By default, it is disabled.		
Media Premium QoS	Sets the priority of the media premium for QoS. A media premium QoS with a bigger		
Media Premium Q03	value has a higher priority. The value range is 0~63, with the default value of 46.		
	Sets the priority of the control premium for QoS. A control premium QoS with a		
Control Premium QoS	bigger value has a higher priority. The value range is 0~63, with the default value of		
	26.		

3.6 Port Settings

Port Settings includes two parts: FXS and Port Group. See Figure 3-47.

🚺 Port	*
FXS	
Port Group	

Figure 3-47 Port Settings

3.6.1 FXS

					FXS Settin	gs						
Port	Туре	SIP Account	Authentication Username	Auto Dial Num	Forbid Outgoing Call	DND	Forward	FWD Type	FWD Number	CID	Call Waiting	Reg Sta
1	FXS	8001	1		Disable	Disable	Disable			Enable	Disable	Unregis
2	FXS	8002	3 <u>818</u> 3		Disable	Disable	Disable			Enable	Disable	Unregis
3	FXS	8003			Disable	Disable	Disable			Enable	Disable	Unregis
4	FXS	8004	()		Disable	Disable	Disable			Enable	Disable	Unregis
5	FXS	8005		-	Disable	Disable	Disable	0.000		Enable	Disable	Unregis
6	FXS	8006	2 <u>011</u> 3		Disable	Disable	Disable			Enable	Disable	Unregis
7	FXS	8007			Disable	Disable	Disable			Enable	Disable	Unregis
8	FXS	8008			Disable	Disable	Disable			Enable	Disable	Unregis
				100						64		

Figure 3-48 FXS Settings Interface

See Figure 3-48 for the FXS settings interface. The list in the above figure shows the feature and properties of each FXS port. Click *Modify* in Figure 3-48 to modify the properties of the corresponding port. See Figure 3-49 for the FXS modification interface.



FX	S-Modify
Port	
	1
Туре	FXS
Register Port	Yes
SIP Account	8001
Password	
Authentication Username	
Server Index	1:201.123.115.233 💌
Auto Dial Number	
Wait Time before Auto Dial (s)	0
Echo Canceller	✓Enable
Forbid Outgoing Call	Enable
CID	Enable
Call Waiting	Enable
DND (Do Not Disturb)	Enable
Call Forward	Enable
Forward Type	Unconditional
Forward Number	
Advanced Configuration	✓Enable
Talkback	Enable
Bound Number	
Note: 'Auto Dial Number' goes into effect only if	f no dialing occurs during "Wait Time before Auto Dial".
Modify	Reset

Figure 3-49 FXS Modification

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

Item	Description	
Port	Serial number of the FXS port on the device.	
Туре	Type of the port on the device (FXS). This item is not configurable.	
	Sets whether to register the port to the SIP server.	
Dogiotor Dort	When this item is set to No, the item Reg Status on the FXS settings interface	
Register Port	(Figure 3-48) shows Unregistered; when this item is set to Yes, the item Reg Status	
	shows Failed or Registered.	
	When the port initiates a call to SIP, this item corresponds to the username of SIP.	
	The default SIP account is 80XX among which XX represents the corresponding	
SIP Account	port number. For example, the default SIP account corresponding to Port 1 is 8001,	
	and that corresponding to Port 8 is 8008.	



Password		ssword of the port. To register a port to the SIP server, both items d Password must be filled in.		
Authentication	Authentication username of a port, used to register the port to the SIP server when IMS network is enabled.			
Username	Note: This item	appears only when IMS Network is enabled.		
Auto Dial Number, Wait Time before Auto DialThe FXS port will dial the Auto Dial Number if there is no dialing pickup within a designated time period (i.e. Wait Time before Auto I				
Echo Canceller		ellation feature for a call conversation over the FXS channel. By ure is enabled and the effect can reach 128ms.		
Forbid Outgoing Call	If this feature is setting is disable	enabled, the FXS port will be forbidden to call out. The default		
CID	CallerID. If this feature is enabled, the FXS port will send the CallerID of the incoming IP call together with the ringing tone to the corresponding station. The default setting is <i>enabled</i> . CallerID displays digits only and will filter out any other characters if exist.			
Call Waiting	IP and keep it in station hangs u	If this feature is enabled, the FXS port in conversation can accept another call from IP and keep it in the waiting state. Once the current conversation is finished and the station hangs up, the call in the waiting state will ring the station and wait for answer. The default setting is <i>disabled</i> .		
DND	Do Not Disturb. If this feature is enabled, the FXS port will reply the 403 m			
Call Forward	The automatic call forward feature for the FXS port. Once this feature is enabled, the FXS port will forward incoming IP calls according to <i>FWD Type</i> . Note: To enable this feature, do not put the FXS port into a port group with other ports. The default setting is <i>disabled</i> .			
	Forward conditive values are:	ons for the FXS port to forward incoming IP calls. The optional		
	Option	Description		
	Unconditional	The FXS port will forward all incoming IP calls to the preset <i>FWD Num</i> immediately when it receives them.		
FWD Type	Busy	The FXS port will forward incoming IP calls to the preset <i>FWD</i> <i>Num</i> if it is busy upon receiving them.		
	The FXS port will forward incoming IP calls to the preset <i>FV</i> <i>Num</i> if the corresponding station does not answer them in <i>No Reply</i> designated time period (i.e. <i>Time for No Reply Forward</i>). O when this forward condition is selected does the configurati item <i>Time for No Reply Forward</i> become valid.			
	This item is valid	d only when Call Forward is set to <i>Enabl</i> e.		
FWD Num		which the incoming IP call is forwarded. If the <i>Call Forward</i> feature tem can not be left empty.		



	Sets whether to enable the color ring feature or not, with the default setting of being
Color Ring	disabled.
	Note: Only when there are available color rings will this item appear.
Color Ring Index	The index of the color ring which will be quoted by the current FXS port.
Server Index	The index of the sip server which will be quoted by the current FXS port.
	With this feature enabled and a number bound, the port can talkback to its bound
-	number. That is, they can start a call with each other as soon as picking up the
Talkback	phone. The default setting is <i>disabled</i> .
	Note: This feature is only used in the case of channel registration.
Bound Number	Sets the bound number for talkback.

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

Or you can click **Batch** to modify several pieces of FXS settings at the same time. See Figure 3-50 below for the FXS batch modification interface. The configuration items on this interface are the same as those on the FXS modification interface (Figure 3-49).



FXS-Batch Modify					
Starting Port	1				
Ending Port	8				
Register Port	Yes				
Starting SIP Account					
Starting Authentication Password					
Starting Authentication Username					
Server Index	1.201.123.115.233 💌				
SIP Account Batch Rule	Increase				
SIP Account Batch Step Size	1				
Authentication Password Batch Rule	Increase				
Authentication Password Batch Step Size	1				
Authentication Username Batch Rule	Increase				
Authentication Username Batch Step Size	1				
	Enable				
CID Echo Canceller	I Enable I Enable				
Forbid Outgoing Call					
Call Waiting					
DND (Do Not Disturb)	Enable				
Call Forward					
Forward Type	Unconditional				
Forward Number					



Figure 3-50 FXS Batch Modification

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

Item	Description			
Starting Port	The starting serial number of the FXS port on the device in the batch setting.			
Ending Port	The ending serial number of the FXS port on the device in the batch setting.			
Starting SIP Account	The starting SIP account in the batch setting.			
Starting Authentication	The starting authentication password in the batch setting.			
Password				
Starting Authentication				
Username	The starting authentication username in the batch setting.			
	The rule for batch setting the SIP account, including Increase and Decrease two			
SIP Account Batch Rule	options.			



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

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

3.6.2 Port Group

					P	ort Group Settings			
Check	Index	Description	SIP Account	Authentication Usernam	e Ports	Port Select Mode	Rule for Ringing by Turns	Timeout for Ringing by Turns (s)	Preemptive Answe
	1	default	900	9000	1,2,3,4	Increase			-
								1	2
Check All		Uncheck All	Inverse	E Delete E C	ear All				Add New

Figure 3-51 Port Group Settings Interface

See Figure 3-51 for the port group settings interface. A port group is a set containing single or multiple ports, used to specify such properties as *Port Selection* and *Authentication Mode* for all the ports in it. A new port group can be added by the *Add New* button on the bottom right corner of the above list. See Figure 3-52 for the port group adding interface. Note that a port which has been occupied by one port group cannot be chosen by others.



	Port Group-Add	
Index	2	*
Description	default	
Register Port Group	YES	~
SIP Account		
Password		
Authentication Username		
Server Index	1:201.123.112.233	~
Authentication Mode	Do Not Register	~
Port Select Mode	Ringing by Turns	~
Rule for Ringing by Turns	1,2,3,4,5,5	
Timeout for Ringing by Turns (s)	20	
Port	Port 1(FXS) Port 2(FXS) Port 3(FXS) Port 5(FXS) Port 6(FXS) Port 7(FXS)	
	Check All Inverse Check All FXS Ports	

Figure 3-52 Add New Port Group

The table below explains the items in the above figure.

Item	Description
Index	The unique index of each port group, which is mainly used in the configuration of
Index	routing rules and number manipulation rules to correspond to port groups.
Description	More information about each port group, with default value of default.
	To register the port group to the SIP server. Only when this configuration item is set
Register Port Group	to Yes can you see the configuration items SIP Account and Password.
	When the port group initiates a call to SIP, this item corresponds to the username of
SIP Account	SIP.
Decouvered	Registration password of the port group. To register the port group to the SIP server,
Password	both configuration items SIP Account and Password should be filled in.
Authoritication	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.
Server Index	The index of the sip server which will be quoted by the current FXS port.



	Sets the way for SIP to make outgoing calls (TeI \rightarrow IP) on the gateway.				
	Option	Description			
	Do Not Register (default)	SIP initiates a call in a point-to-point mode.			
Authentication Mode	Register Gateway	SIP initiates a call with the registered SIP account and password of the whole gateway. (Refer to <u>3.4.1 SIP</u> for gateway registration.)			
	Register Port Group	SIP initiates a call with the registered SIP account and password of the port group.			
	Register Port	SIP initiates a call with the registered SIP account and password of the port.			
	Registration status of the port group. When Register Port Group is set to No, the				
Register Status	value of this item is U	Inregistered; when Register Port Group is set to Yes, the			
	value of this item may b	be Failed or Registered.			



		eceives a call, it will choose a port based on the select mode on item to ring or to connect. The optional values and their			
	corresponding meanings are described in the table below.				
	Option	Description			
		Search for an idle port in the ascending order of the port number, starting from the minimum. If no match is found,			
	Increase (default)	search repeatedly until finding a port which is allowed to enter the call waiting state.			
		Search for an idle port in the descending order of the port			
	Decrease	number, starting from the maximum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.			
Port Select Mode	Cyclic Increase	Provided Port N is the available port found last time. Search for an idle port in the ascending order of the port number, starting from Port N+1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.			
	Cyclic Decrease	Provided Port N is the available port found last time. Search for an idle port in the descending order of the port number, starting from Port N-1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.			
	Group Ringing	Ring all the idle FXS ports in this port group.			
	Ringing by Turns	Ring the ports in this port group according to the <i>Rule for</i> <i>Ringing by Turns</i> which can be user-defined. Refer to the format of the rule in Figure 3-52. By default, the ringing will be carried out in the ascending order of the port number. <i>Timeout for Ringing by Turns</i> is used to set the overtime for ringing. Range of value: 15~60, calculated by s, with the default value of 20.			
	When a channel in a	port group is ringing, another channel in the same port group			
Preemptive Answer Keyboard Shortcut	can press the keyboard shortcut set by this item to transfer the call from the ringing channel to the current channel.				
	Note: This item will be Group Ringing or Ring	come invalid if the gateway works under the port select mode ging by Turns.			
	The ports in the port g	roup. If the checkbox before a port is grey, it indicates that the			
Dort	port is not available or	has been occupied. All selected ports for a port group will be			
Port	displayed in the <i>Ports</i> column in Figure 3-51. Note: When a port group contains multiple ports, the automatic call forward feature is invalid.				

After configuration, click *Save* to save the settings into the gateway, click *Reset* to restore the configurations, or click *Cancel* to cancel the settings. *Check All* means to select all available ports on the current page; *Inverse* means to uncheck the selected items and check the unselected. *Check All FXS Ports* means to select all available FXS ports on the current page.



Click *Modify* at the end of the list in **Port Group Settings Interface** to modify the properties of a port group. See Figure 3-53 for the port group modification interface. The configuration items on this interface are the same as those on the *Add New Port Group* interface.

Index	1	
Description	default	
Register Port	Yes	~
SIP Account	900	
Password	•••	
Authentication Username	9000	
Server Index	1:201.123.112.233	~
Authentication Mode	Do Not Register	~
Port Select Mode	Increase	~
Preemptive Answer Keyboard Shortcut		
Port	Port 1(FXS) Port 2(FXS) Port 3(FXS) Port 5(FXS) Port 6(FXS) Port 7(FXS)	
	Check All Inverse Check All FXS Ports	

Figure 3-53 Modify Port Group

To delete a port group, check the checkbox before the corresponding index in Figure 3-51 and click the '*Delete*' button. *Check All* means to select all available items on the current page; *Uncheck All* means to cancel all selections on the current page; *Inverse* means to uncheck the selected items and check the unselected. To clear all port groups at a time, click the *Clear All* button in Figure 3-51.

3.7 Route Settings

Route Settings is used to specify the routing rules for calls on two directions: $IP \rightarrow Tel$ and $Tel \rightarrow IP$. See Figure 3-54.

Route	*
Routing Paramete	rs
IP>Tel	
Tel>IP	
Figure 3-54 Route Set	tinas



3.7.1 Routing Parameters

P> TEL	Route before Number Manipulate
TEL> IP	Route before Number Manipulate
TEL> IP	Route before Number Manipulate 💉

Figure 3-55 Routing Parameters Configuration Interface

See Figure 3-55 for the routing parameters configuration interface. On this interface, you can set the routing rules for calls respectively on two directions $IP \rightarrow Tel$ and $Tel \rightarrow IP$ 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.

3.7.2 IP to Tel

Operation Info	*	Standard Mode Character Mode
Quick Config	*	
VolP	*	
ố Advanced	*	No available routing rule!
() Port	*	Add New
Route	*	
Routing Paramete	ers	
IP>Tel		•
Tel>IP		

Figure 3-56 IP→Tel Routing Rule Configuration Interface (Standard)

See Figure 3-56 for the IP \rightarrow Tel routing rule configuration interface. By default, there is no available routing rule on the gateway. The IP \rightarrow Tel routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click *Add New* to add them manually. See Figure 3-57. You may use the default values of all the configuration items herein.



IP->Tel Routing Rule		
Index:	63 🗸	
Description:	default	
Source IP:	*	
CallerID Prefix:	*	
CalleeID Prefix:	*	
Route by Number	Enable	
Call Destination:	1 🗸	
Save	Close	

Figure 3-57 Add New Routing Rule (IP→Tel)

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



Call Destination Port group to which the call will be routed.

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

See Figure 3-58 for the IP→Tel routing rule configuration interface after your configuration. There is a rule displayed with Index 63 and Call Destination 'Route by Number', having no restriction on Source IP, CallerID Prefix and CalleeID Prefix, which indicates the gateway will route a call from any IP address to a corresponding port based on its number.

Press the Add New button on the bottom right corner of the list to add a new routing rule.

						IP->Tel Routing Rule			
Check	Index	Source	IP	Cal	IerID Prefix	CalleeID Prefix	Call Destination	Description	Modify
	63	*			*	*	Route by Number	default	
Check All	Uncheck All	Inverse	Ξ	Delete	E Clear All				Add Nev

Figure 3-58 IP→Tel Routing Rule Configuration Interface

Click **Modify** in Figure 3-58 to modify a routing rule. The configuration items on the $IP \rightarrow Tel$ routing rule modification interface are the same as those on the **Add New Routing Rule** ($IP \rightarrow Tel$) interface. Note that the item **Index** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-58 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 3-58.

See Figure 3-59 for the IP \rightarrow Tel Routing Rule Configuration Interface under the Character mode. You can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

Standard Mode Character Mode
IP->Tel Routing Rule
Note: The routing information contains such fields as Source IP, CallerID Prefix, CalleeID Prefix, Route by Number, Destination Port Group and Description. The priority decreases from top to bottom; adjacent fields are separated by a space Symbol * in Source IP, CallerID Prefix and CalleeID Prefix indicates any IP address or string; When Route by Number is set to 1, the Destination Port Group will be enabled. Don't forget to save the configuration after your modification!
*** 0 0 default
1 Items Total

Figure 3-59 IP→Tel Routing Rule Configuration Interface (Character)



3.7.3 Tel to IP

Operation Info	*	Standard Mode Character Mode
Quick Config	*	
VolP	*	
र्ट्टे Advanced	*	No available routing rule!
() Port	*	Add New
Route	*	
Routing Paramet	ers	
IP>Tel		
Tel>IP		

Figure 3-60 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-60 for the Tel \rightarrow IP routing rule configuration interface. By default, there is no available routing rule on the gateway. The Tel \rightarrow IP routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click *Add New* to add them manually. See Figure 3-61. You may use the default values of all the configuration items herein except for *Destination IP* and *Destination Port*.

Tel->IP Ro	outing Rule
Index:	63 🔽
Description:	default
Source Port Group:	
CallerID Prefix:	*
CalleeID Prefix:	*
Destination IP:	
Destination Port:	5060
Save	Close

Figure 3-61 Add New Routing Rule (Tel→IP)

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

Item	Description
	The unique index of each routing rule, which denotes its priority. A routing rule with
Index	a smaller index value has a higher priority. If a call matches several routing rules, it
	will be processed according to the one with the highest priority.
Description	More information about each routing rule, with the default value of <i>default</i> .



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

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

See Figure 3-62 for the Tel→IP routing rule configuration interface after your configuration. There is a rule displayed with Index 63, Destination IP '192.168.1.101' and Destination Port '5060' (i.e. default IP address and port of the gateway), having no restriction on Call Initiator, CallerID Prefix and CalleeID Prefix, which indicates all the outgoing calls from Tel which conform to the dialing rule will be routed to the gateway.

Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Destination IP	Destination Port	Description	Modify
-		e en maaren		Control Print				
	63	×	*	×	192.168.1.101	5060	default	

Figure 3-62 Tel→IP Routing Rule Configuration Interface

Click **Modify** in Figure 3-62 to modify a routing rule. The configuration items on the Tel \rightarrow IP routing rule modification interface are the same as those on the **Add New Routing Rule (Tel\rightarrowIP)** interface. Note that the item **Index** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-62 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the **Clear All** button in Figure 3-62.

See Figure 3-63 for the Tel \rightarrow IP Routing Rule Configuration Interface under the Character mode. You can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.



Standard Mode Character Mode
Tel->IP Routing Rule
Note: The routing information contains such fields as Source Port Group, CallerID Prefix, CalleeID Prefix, Destination IP, Destination Port and Description The priority decreases from top to bottom; adjacent fields are separated by a space CallerID Prefix, CalleeID Prefix, Destination IP Symbol * indicates any character; Source Port Group set to 0 denotes any port group. Don't forget to save the configuration after your modification!
0 *** 0 default
1 Items Total
Save

Figure 3-63 Tel→IP Routing Rule Configuration Interface (Character)

3.8 Number Manipulation

Number Manipulation includes four parts: $IP \rightarrow Tel CallerID$, $IP \rightarrow Tel CalleeID$, $Tel \rightarrow IP CallerID$ and $Tel \rightarrow IP CalleeID$. See Figure 3-64.



Figure 3-64 Number Manipulation

3.8.1 IP to Tel CallerID

			1		IP->Tel CallerID Numb	er Manipulation Rule				
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	D
	63	*	*	*	0	0	0			
										3

Figure 3-65 IP→Tel CallerID Manipulation Interface (Standard)

See Figure 3-65 for the IP \rightarrow Tel CallerID manipulation interface under the Standard mode. A new number manipulation rule can be added by the **Add New** button on the bottom right corner of the list in the above figure. See Figure 3-66 for the IP \rightarrow Tel CallerID manipulation rule adding interface. You may use the default values of all the configuration items herein.



IP->Tel Callerl	D
Index:	63 💌
Description:	default
Call Initiator:	*
CallerID Prefix:	*
CalleeID Prefix:	*
Stripped Digits from Left:	0
Stripped Digits from Right:	0
Reserved Digits from Right:	0
Prefix to Add:	
Suffix to Add:	
Save	Close

Figure 3-66 Add IP→Tel CallerID Manipulation Rule

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

Item	Description
	The unique index of each number manipulation rule, which denotes its priority. A
Index	number manipulation rule with a smaller index value has a higher priority. If a call
Index	matches several number manipulation rules, it will be processed according to the
	one with the highest priority.
Decemination	More information about each number manipulation rule, with the default value of
Description	default.
	IP address from where the call is initiated. This item can be set to a specific IP
Call Initiator	address or "*" which indicates any IP address.



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

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 3-65 to modify a number manipulation rule. See Figure 3-67 for the IP \rightarrow Tel CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the **Add IP\rightarrowTel CallerID Manipulation Rule** interface. Note that the item **Index** cannot be modified.



IP->Tel Calleri	D
Index:	63 💌
Description:	test
Call Initiator:	*
CallerID Prefix:	*
CalleeID Prefix:	*
Stripped Digits from Left:	0
Stripped Digits from Right:	0
Reserved Digits from Right:	0
Prefix to Add:	
Suffix to Add:	
Save	Close

Figure 3-67 Modify IP→Tel CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-65 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 3-65.

See Figure 3-68 for the IP \rightarrow Tel CallerID Manipulation Interface under the Character mode. You can edit the number manipulation rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.



Standard Mode	Character Mode	
	IP->Tel CallerID Number Manipulation Rule	
Suffix and Descrip The priority decrea Adjacent fields an	er Manipulation Rule contains such fields as Call Initiator, CallerID Prefix, CalleeID Prefix, Delete Digits from Left, Delete Digits from Right, Reserve Digits from Right, Add Pre ption sases from top to bottom; by default, the rule will be inserted to the end after you click 'Add'. If you want to increase its priority, please copy it to the corresponding position. re separated by a space; Symbol * in Call Initiator, CallerID Prefix and CalleeID Prefix indicates any string; Symbol <@#> in Add Prefix and Add Suffix denotes not to add. we the configuration after your modification!	fix, Ado
***000<@	@#> <@#> default	
1ltems Total	Save	

Figure 3-68 IP→Tel CallerID Manipulation Interface (Character)

3.8.2 IP to Tel CalleeID

The number manipulation process for IP \rightarrow Tel CalleeID is almost the same as that for IP \rightarrow Tel CallerID; only the number to be manipulated changes from CallerID to CalleeID. See,Figure 3-70 for IP \rightarrow Tel CalleeID manipulation interface. The configuration items on this interface are the same as those on *IP\rightarrowTel CallerID Manipulation Interface* (Figure 3-65).

Oheelu	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Obien and Disits frame Laft	Objected Digits from Dight	Dessented Disits from Disht	Deaffy to Add	Outfin to Andre	D
Check	Index	Call Initiator	Callend Pretix	CalleelD Pretix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	D
	63	*		*	0	0	0			
										2

Figure 3-69 IP→Tel CalleeID Manipulation Interface(Standard)

Standard Mode Character Mode
IP->Tel CalleeID Number Manipulation Rule
Interview Cell Callector Mainpulation Rule Note: The Number Mainpulation Rule contains such fields as Call Initiator, CallerID Prefix, CallecID Prefix, Delete Digits from Left, Delete Digits from Right, Reserve Digits from Right, Add Prefix, Add Suffix and Description The priority decreases from top to bottom; by default, the rule will be inserted to the end after you click: Add. If you want to increase its priority, please copy it to the corresponding position. Adjacent fields are separated by a space; Symbol* in Call initiator, CallerID Prefix and CalleeID Prefix indicates any string; Symbol *@#> in Add Prefix and Add Suffix denotes not to add. Don't forget to save the configuration after your modification! **** 0 0 0 <@#> <@#> default
1ltems Total



Figure 3-70 IP→Tel CalleeID Manipulation Interface (Character)

3.8.3 Tel to IP CallerID

					Tel->IP CallerID Numb	er Manipulation Rule				
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	D
	63	*	*	*	0	0	0			
										1

Figure 3-71 Tel→IP CallerID Manipulation Interface (Standard)

See Figure 3-71 for the Tel \rightarrow IP CallerID manipulation interface under the Standard mode. A new number manipulation rule can be added by the *Add New* button on the bottom right corner of the list in the above figure. See Figure 3-72 for the Tel \rightarrow IP CallerID manipulation rule adding interface. You may use the default values of all the other configuration items herein.

Tel->IP CallerID		
Index:	63 💌	
Description:	default	
Source Port Group:	*	
CallerID Prefix:	*	
CalleeID Prefix:	*	
Stripped Digits from Left:	0	
Stripped Digits from Right:	0	
Reserved Digits from Right:	0	
Prefix to Add:		
Suffix to Add:		
Save	Close	

Figure 3-72 Add Tel→IP CallerID Manipulation Rule

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

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



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

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 3-71 to modify a number manipulation rule. See Figure 3-73 for the Tel \rightarrow IP CallerID manipulation rule modification interface. The configuration items on this interface are the same as those on the **Add Tel** \rightarrow IP **CallerID Manipulation Rule** interface. Note that the item **Index** cannot be modified.



Tel->IP CallerID		
Index:	63 💌	
Description:	test	
Source Port Group:	1	
CallerID Prefix:	*	
CalleeID Prefix:	*	
Stripped Digits from Left:	0	
Stripped Digits from Right:	0	
Reserved Digits from Right:	0	
Prefix to Add:		
Suffix to Add:		
Save	Close	

Figure 3-73 Modify Tel→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-71 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 3-71.

See Figure 3-74 for the Tel \rightarrow IP CallerID Manipulation Interface under the Character mode. You can edit the number manipulation rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.


Standard Mode Character Mode
Tel->IP CallerID Number Manipulation Rule
Note: The Number Manipulation Rule contains such fields as Source Port Group, CallerID Prefix, CalleeID Prefix, Delete Digits from Left, Delete Digits from Right, Reserve Digits from Right, Add Prefix, Add Suffix and Description The priority decreases from top to bottom; Adjacent fields are separated by a space. Symbol * In Call Initiator, CallerID Prefix and CalleeID Prefix indicates any string; Symbol <@#> In Add Prefix and Add Suffix denotes not to add. Don't forget to save the configuration after your modification!
0 * * 0 0 0 <@#> <@#> default
1 Items Total
Save

Figure 3-74 Tel→IP CallerID Manipulation Interface (Character)

3.8.4 Tel to IP CalleeID

The number manipulation process for Tel \rightarrow IP CalleeID is almost the same as that for Tel \rightarrow IP CallerID; only the number to be manipulated changes from CallerID to CalleeID. See Figure 3-75, Figure 3-76 for the Tel \rightarrow IP CalleeID manipulation interface. The configuration items on this interface are the same as those on **Tel\rightarrowIP CallerID Manipulation Interface** (Figure 3-71).

					Tel->IP CalleeID Numb	er Manipulation Rule				
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	D
	63	*	*	*	0	0	0			
										1

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

Standard Mode Character Mode
Tel->IP CalleeID Number Manipulation Rule
Tet->IP CalleeID Number Manipulation Rule Note: The Number Manipulation Rule contains such fields as Source Port Group, CallerID Prefix, CalleeID Prefix, Delete Digits from Left, Delete Digits from Right, Reserve Digits from Right, Add Prefix, Add Suffix and Description The priority decreases from top to bottom; Adjacent fields are separated by a space. Symbol * in Call Initiator, CallerID Prefix and CalleeID Prefix indicates any string; Symbol <@#> in Add Prefix and Add Suffix denotes not to add. Dontforget to save the configuration after your modification! 0 * * 0 0 0 < @#> <@#> default
1 Items Total



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

3.9 System Tools

System Tools is mainly for gateway maintenance. It provides such features as IP modification, data backup and connectivity check. See Figure 3-77 for details.

System Tools 🗧						
Management						
Network						
Upgrade						
Signaling Capture						
Call Log						
Change Password						
Backup & Upload						
Factory Reset						
Restart						
System Monitor						
SNMP Config						
PING Test						
TRACERT Test						
Figure 3-77 System Tools						



3.9.1 Management

NEB Mai	nagement	
	WEB Port	80
	Access Setting	Allow All IPs
SYSLOG	Parameters	
	SYSLOG	⊙ Yes O No
	Server Address	201.123.115.20
	SYSLOG Level	INFO.
Time Para	ameters	
	NTP	⊙Yes ONo
	NTP Server Address	127.0.0.1
	Synchronizing Cycle	3600
	Daily Restart	Oyes ONo
	Restart Time/td>	0 💌 h 0 💌 m
	System Time	Modify 2014-11-20 13:34:43
	Time Zone	GMT+8:00 (Beijing, Singapore, Taipei, Kual 🗸

Figure 3-78 Management Parameters Setting Interface

See Figure 3-78 for the Management Parameters Setting interface. The table below explains the items shown in the above figure.

ltem	Description		
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.		
	Sets the IP addresses which can access the gateway via WEB. By default, all IPs		
	are allowed. You can set an IP whitelist to allow all IPs within it to access the		
Access Setting	gateway freely. Also can set an IP blacklist to forbid all IPs within it to access the		
	gateway.		
SVSI OC	Sets whether to enable SYSLOG. It is required to fill in SYSLOG Server Address		
SYSLOG	and SYSLOG Level in case SYSLOG is enabled. By default, SYSLOG is disabled.		
Server Address Sets the SYSLOG server address for log reception.			
	Sets the SYSLOG level. There are three options: ERROR, WARNING, INFO and		
SYSLOG Level	DEBUG. The default value is INFO.		
	Sets whether to enable the NTP time synchronization feature. It is required to fill in		
NTP	NTP Server Address, Synchronizing Cycle and Time Zone in case NTP is		
	enabled. By default, <i>NTP</i> is disabled.		
NTP Server Address	Sets the Server address for NTP time synchronization.		
Synchronizing Cycle	Sets the cycle for NTP time synchronization. The default value is 3600.		
Doily Rootort	Sets whether to restart the gateway regularly every day at the preset Restart Time.		
Daily Restart	By default, this feature is disabled.		



Restart Time Sets the time to restart the gateway regularly.					
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.				

3.9.2 Network

Network S	Settings
Network Type:	Static
IP Address (I)	192.168.1.101
Subnet Mask (U)	255.255.255.0
Default Gateway (D)	192.168.1.1
DNS Server (P)	0.0.0.0
Speed and Duplex Mode	Automatic Detection
Save	Reset
Note: The service will be restarted automatically after sa new IP address if the IP add	

Figure 3-79 Network Settings Interface

See Figure 3-79 for the network settings interface. A gateway has only one LAN, which can be configured with network type, IP address, subnet mask, default gateway and DNS server. Network Type has two options: Static and DHCP.

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



3.9.3 Upgrade

Current Version						
Serial Num 00001452						
WEB	Version 1.0.0_2015051214					
Service	Version 1.0.0_2015051214					
U-boot	Version Apr 03 2015-09:48:30					
Kernel	Version #179 Mon May 11 13:48:32 PDT 2015					
Product Type	Product Type 1008B-8S(RJ11)					
Select an U	Select an Update File Browse					
	Update Reset					

Figure 3-80 Upgrade Interface

See Figure 3-80 for the upgrade interface where you can upgrade the WEB, gateway service, kernel and firmware to new versions. Select the upgrade package "*.tar.gz" (The gateway will do MD5 verification before upgrading and will not start to upgrade until it passes the verification.) via **Browse...** and click **Update**. Then the file uploading interface will appear. See Figure 3-81.

	Current Version						
	Serial Num 00001452						
	WEB Version 1.0.0_2015051214						
	Service Version 1.0.0_2015051214						
	U-boot Version Apr 03 2015-09:48:30						
	Kernel Version #179 Mon May 11 13:48:32 PDT 2015						
	Product Type 1008B-8S(RJ11)						
r							
	Select an Up	odate File C:\Documents and Sett Browse					
l							
		52%					
	The file is	uploading. Please do not leave this page!					
	The life is	uploading. Thease do notheave this pages					
		Lingrada Information					
		Upgrade Information					
	start upload	upgrade file					





After a successful uploading of the file, the gateway will start to upgrade the system. See Figure 3-82 and you can learn the detailed upgrading information from the upgrade information box at the bottom.

Current Version								
Serial	Serial Num 00001452							
WE	WEB Version 1.0.0_2015051214							
Serv	Service Version 1.0.0_2015051214							
U-bo	oot	Version Apr 03 2015-09:48:30						
Kerr	Kernel Version #179 Mon May 11 13:48:32 PDT 2015							
Product	Product Type 1008B-8S(RJ11)							
Selec	Select an Update File C:\Documents and Sett Browse							
Update Reset								
		Upload completion!						
		25%						
System updating, please do not leave this page! Upgrade Information								
1070								
do kil	do killall SMGSvr							
DbgSvr pid=894								
896	896							
897								
898								
	llall Db							
сору	раска	ge file to storage 💌						

Figure 3-82 System Upgrading Interface

Note that clicking *Reset* can only delete the selected update file but not cancel the operation of *Update*.

Note: Please contact our technicians if you need to downgrade the gateway to an old version. An improper operation may cause unexpected problems.

3.9.4 Signaling Capture

Packet Capture						
Signaling Packet Capture	SIP&Syslog 💽					
RTP Packet Capture	RTP Port Range 💌 50000,50767	Start	Stop			
channel Data Recording	×					



Figure 3-83 Signaling Capture Interface

See Figure 3-83 for the Signaling Capture interface. Packet capture contains Signaling Packet Capture, RTP Packet Capture and Channel Data Recording. You can select either of them to start the capture according to your requirement. Click *Start* to start capturing packets. Click *Stop* to stop the capture and download the captured packets.

3.9.5 Call Log

Call Log SIP Log	Enable Call Log	Download					
Call from IP Channel							Clear All
11/30/1999 08:08:59:428	IP Channel 0,Incoming call fro	om remote end "802	9" <sip:8029@192.168< td=""><td>3.1.2>,call-id: 780b455</td><td>if007f0a3d@UEMtMjAxN</td><td>ITA2MjExMTAz Caller 8029 Call</td><td>ee 8027 bind the chan</td></sip:8029@192.168<>	3.1.2>,call-id: 780b455	if007f0a3d@UEMtMjAxN	ITA2MjExMTAz Caller 8029 Call	ee 8027 bind the chan
<			1111				>
Call from Port	Select a Port:	Port28					Clear All
	IP Channel 0,Incoming call fro Analog Channel 146 caller tra					ITA2MjExMTAz Caller 8029 Call	ee 8027 bind the chan
11/30/1999 08:08:59:446	Analog Channel 146 callee tra	anslation 8027>80	27 match IP>TEL Cal				
	Analog Channel 146 ringing,C Analog Channel 146 pickup, b		027				
	Analog Channel 146 call end,		ers the idle state(phon	e onhook)			
<u> </u>							>

Figure 3-84 Call Log Interface

Call Log SIP Log Enable Call Log Download			
SIP Log	Refresh	Clear All	
11/30/1999 08:08:59:001 Message received from:192.168.1.2:5062			^
INVITE sip:8027@192.168.1.3 SIP/2.0			
Via: SIP/2.0/UDP 192.168.1.2:5062;branch=z9hG4bK-d87543-e62d157b0636a518-1d87543-;rport			
Max-Forwards: 70			
Contact: <sip:8029@192.168.1.2:5062></sip:8029@192.168.1.2:5062>			
To: <sip:8027@192.168.1.3></sip:8027@192.168.1.3>			
From: "80299" <sip: 8029@192.168.1.2="">.tag=794b815a</sip:>			
Call-ID: 780b455f007f0a3d@UEMtMJA4MTA2MJEX/ITA2			
CSeq: 1 INVITE Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO			
Allow, HWTE, AUN, CHARLEL, OF HOTS, BTE, REFER, NOTHT, MESSAGE, SOBSCRIBE, NYO			
Content rype, explorations up			
User-Agent evelBeam AudioOnly release 3015c stamp 27106			
Content-Length: 305			
and a second			
v=0			
o=- 183077394 183077479 IN IP4 192.168.1.2			
s=eyeBeam AudioOnly			
c=IN IP4 192.168.1.2			
t=0 0			
m=audio 6164 RTP/AVP 100 6 0 8 3 18 98 97 5 102			
a=alt11:A26AA641C6B19F3E192.168.1.2.6164			
a=rtpmap:100 speex/16000			
a=rtpmap:98 libc/8000			
a=rtpmap:97 speex/8000 a=rtpmap:102116/16000			
a=rpmap.ru2/16/16000 a=sendrecy			
a-serureu			24
			(Y

Figure 3-85 SIP Log Interface

See Figure 3-84, Figure 3-85 for the Call Log interface. Click the checkbox before **Enable Call** Log to enable the call log feature, including **Call Log** and **SIP Log**. **Call from IP Channel** displays the call log information generated on all IP channels, and **Call from Port** displays the call



log information generated on the port you select. All the SIP related information will be displayed in *SIP Log*.

3.9.6 Change Password

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

Figure 3-86 Password Changing Interface

See Figure 3-86 for the password changing interface where you can change username and password of the gateway. Enter the current password, the new username and password, and then confirm the new password. After configuration, click **Save** to apply the new username and password or click **Reset** to restore the configurations. After changing the username and password, you are required to log in again.

3.9.7 Backup & Upload

	Data Backup	
To backup the configuration file, click the '	Backup' button to start.	Backup
	Data Upload	
To upload a configuration file, select it and	click the button 'Upload' to start.	
Configuration File	Browse	Upload

Figure 3-87 Backup & Upload Interface

See Figure 3-87 for the backup and upload interface. To back up the configuration file to your PC, just click *Backup*. To upload a configuration file, select it via *Browse...* and click *Upload*.



	Data Backup	
To backup the configu	ration file, click the 'Backup' button to start.	Backup
	Data Upload	
To upload a configurati Configuration File	on file, select it and click the button "Upload" to start. Are you sure to upload configuration file?	Upload
Note: After	OK Cancel	restart automatically.

Figure 3-88 Backup & Upload & Prompt Interface

Click **OK** on the prompt box (Figure 3-88) to upload the configuration file to the gateway. Now the prompt information 'System is rebooting, please do not leave this page' appears. See Figure 3-89. The gateway will overwrite the current configurations with the uploaded data after restart. Click **Cancel** to cancel this upload directly.

	Data Backup	
To backup the configuration file, o	dick the 'Backup' button to start.	Backup
	Data Upload	
To upload a configuration file, self	ect it and click the button 'Upload' to start. Browse.	Upload
	illy upload the configuration file, the gateway	
System is rebot	ing. Please do not leave this page!	
Figu	e 3-89 Configuration File Uploading Inte	erface



3.9.8 Factory Reset

Factory Reset
Click the button 'Reset' below to restore to factory settings.
Reset Note: After you successfully restore the gateway to factory settings, the gateway will restart automatically and its IP address will be restored to the default one

Figure 3-90 Factory Reset Interface

See Figure 3-90 for the factory reset interface. Click *Reset* to restore all configurations on the gateway to factory settings.

3.9.9 Restart

Service Restart					
Click the button 'Restart' to restart the service.	Restart	Generate a Dump File			
System Res	tart				
Click the button 'Restart' to restart the system.	Restart	Generate a Dump File			
Dump File Download					
Click the button 'Download' to download the dump file.	Download				

Figure 3-91 System Restart Interface

See Figure 3-91 for the restart interface. Click **Restart** under the service restart interface to restart the gateway service or click **Restart** under the system restart interface to restart the whole gateway system. A dump file will be generated each time you restart the service or the system. Click **Download** and you can download it to help troubleshoot issues.



3.9.10 System Monitor

System Monitor	
Watchdog:	C Enable
Dog Feeding Interval (s)	5
Automatically restart the service if undetected:	Enable
Save	

Figure 3-92 System Monitor Configuration Interface

See Figure 3-92 for the System Monitor Configuration interface. Watchdog is a timing reset system used to avoid application crash. You can set the dog feeding interval when this feature is enabled. The feeding interval is calculated by s, with the value range of 1~15s. By default, this feature is enabled with the default value of 5s. As the feature 'Automatically restart the service if undetected' is enabled, the service application will restart automatically if it is not detected by the gateway guard application. By default, this feature is enabled.

3.9.11 SNMP Config

SNMP Configurat	ion
SNMP Configuration	Enable SNMP
SNMP Server Address	127.0.0.1
Monitoring Port	161
Community String Configuration Access Password	
Save	teset

Figure 3-93 SNMP Configuration Interface

See Figure 3-93 for the SNMP configuration interface. If the SNMP feature is enabled, once the gateway receives a request from the SNMP management software, it will collect relevant information and reply them to the SNMP management software. By default, the SNMP feature is disabled. The available information includes kernel version, CPU usage, processes, memory usage, startup information, LAN status and etc. Currently, the gateway only provides the community string for information acquisition. The table below explains the configuration items shown in Figure 3-93.

Item	Description	
SNMP Server		
Address	IP address of SNMP.	
Monitoring Port	Monitoring Port for SNMP on the gateway.	
Access Password Community string used for information acquisition.		



3.9.12 PING Test

Ping Test				
De	stination Address	127.0.0.1		
Pin	g Count (1-100)	4		
Pa	ckage Length (56-1024 bytes)	56		
Inf	O Start En	d		

Figure 3-94 Ping Test Interface

See Figure 3-94 for the Ping test interface. A Ping test can be initiated from the gateway on a designated IP address to check the connection status between them. The table below explains the configuration items shown in the above figure.

Item	Description	
Destination Address	The number of times that the Ping test should be executed. Range of value: 1~100.	
Ping Count		
Package Length		
Info	The information returned during the Ping test, helping you to learn the network	
	connection status between the gateway and the destination address.	

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



3.9.13 TRACERT Test

Tracert Test				
Destination Address	127.0.0.1			
Maximum Jumps (1-255)	30			
Start	End			
Info				
	. 😥			

Figure 3-95 Tracert Test Interface

See Figure 3-95 for the Tracert test interface. A Tracert test can be initiated from the gateway on a designated IP address to check the routing status between them. The table below explains the configuration items shown in the above figure.

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

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



Appendix A Technical Specifications

Dimensions

SMG1008B: 220×148×40 mm³ SMG1016B4: 440×44×267 mm³

Weight

SMG1008B: 0.375 kg SMG1016B4: 2.530 kg

Environment

Operating temperature: $0^{\circ}C$ —55 $^{\circ}C$

Storage temperature: -20 $^\circ\!\!\!C$ —85 $^\circ\!\!\!C$

Humidity: 8%— 90% non-condensing

Storage humidity: 8%— 90% non-condensing

LAN

Amount: 1 (10/100 BASE-TX (RJ-45))

Self-adaptive bandwidth supported

Auto MDI/MDIX supported

FXS Port

Amount: 4/8

Type: RJ11, RJ45

Maximum transmission distance: 5000m

Impedance

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

Console Port

Amount: 1 (RS-232)

Baud rate: 115200bps

Connector: RJ45 to DB-9 Connector

Data bits: 8 bits

Stop bit: 1 bit

Parity unsupported

Flow control unsupported

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

Power Requirements

Input power:

SMG1008B: 12V the direct current bigger than 3A

SMG1016B4: 100~240VAC

Signaling & Protocol

SIP signaling

Supported protocol: SIP V1.0/2.0, RFC3261

Audio Encoding & Decoding

G.711A	64 kbps
G.711U	64 kbps
G.729A/B	8 kbps
G723	5.3/6.3 kbps
G722	64 kbps
AMR	4.75 kbps
iLBC	13.3/15.2 kbps

Sampling Rate

8kHz

Safety

Lightning resistance: Level 4



Appendix B Troubleshooting

Q1. What to do if I forget the IP address of the SMG-B gateway?

There are two ways to get the IP address:

- 1) Long press the Reset button on the gateway to restore to factory settings. The default IP address is 192.168.1.101
- 2) Dial the corresponding function key through an FXS port to query the IP address. See <u>3.5.7 Function Key</u> for more details.

Q2. The SMG-B gateway only supports routing on two directions, i.e. Tel \rightarrow IP and IP \rightarrow Tel. What to do if I want to make a Tel \rightarrow Tel call?

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

If you need to make Tel \rightarrow Tel calls in a specific way, try via the routing of Tel \rightarrow IP \rightarrow IP \rightarrow Tel. See below for detailed introductions.

Provided you are going to initiate a call from Port Group 1 to Port Group 2; the IP address and port number of your gateway are 192.168.1.101 and 5060 respectively.

- a) Add a new routing rule on the Tel→IP routing rule configuration interface. Select a port group (e.g. **Port Group 1**) as 'Source Port Group' to initiate the call and fill in 'Destination IP' and 'Destination Port' with the gateway's IP address (e.g. **192.168.1.101**) and port number (e.g. **5060**). Then the call initiated from the station corresponding to Port Group 1 will be routed to the gateway.
- b) Add a new routing rule on the IP→Tel routing rule configuration interface. Fill in 'Source IP' with the gateway's IP address (e.g. 192.168.1.101) and select a port group (e.g. Port Group 2) as 'Destination Port Group' to be called. Then if the IP end of the gateway calls itself, the station corresponding to Port Group 2 will ring.
- c) Finishing the above configurations, you can perform a Tel→Tel call from Port Group 1 to Port Group 2 simply by the way you make a Tel→IP call.

Q3. Does call forwarding involve routing and number manipulation?

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

Case 2: If the forwarding number is not the number of the gateway port. It is required to use routing and number manipulation rules. A call forward procedure can be regarded as a Tel \rightarrow IP call. It uses the routing rules and number manipulation rules in the same way as the Tel \rightarrow IP call. A complete call forward is performed as follows:

- a) An incoming IP call to the gateway rings the port which matches the IP→Tel routing and number manipulation rules and obtains a new CallerID.
- b) Then the gateway uses the newly obtained CallerID and the call forward number, via the Tel→IP routing and number manipulation rules, to make another call from the port to a remote IP address.

Q4. In what cases can I conclude that the SMG-B gateway is abnormal and turn to Synway's technicians for help?

a) During runtime, the run indicator does not flash or the alarm indicator lights up or flashes,



and such error still exists even after you restart the device or restore it to factory settings.

- b) Voice problems occur during call conversation, such as that one party or both parties cannot hear the voice or the voice quality is unacceptable.
- c) The port of the gateway is well connected, but the channel indicator never lights up after the gateway startup or the color it lights up does not comply with the actual state or port type.

Other problems such as inaccessible calls, failed registrations, incorrect numbers and abnormal dialing operations on the FXS port are probably caused by configuration errors. We suggest you refer to <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.

Q5. What to do if I cannot enter the WEB interface of the SMG-B gateway after login?

This problem may happen on some browsers. To settle it, follow the instructions here to configure your browser. Enter 'Tools > Internet Options >Security Tab', and add the current IP address of the gateway into 'Trusted Sites'. If you changes the IP address of the gateway, add your new IP address into the above settings too.

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

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

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

Q7. Is there any cell-phone APP can make calls to the SMG-B gateway?

Yes. Linphone is a soft SIP phone that is supported by multiple platforms, such as Linux, Windows, iOS, Android, etc. It must be registered to the SIP registrar server before dialing to other SIP devices or PSTN telephones,

Q8. Does the SMG-B gateway support fax?

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

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

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



Appendix C Technical/sales Support

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

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