

Synway SMG-B Series Analog Gateway

SMG1004B SMG1008B SMG1016B4 SMG1032B4

Analog Gateway

User Manual

Version 1.7.1

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



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Revision History

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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

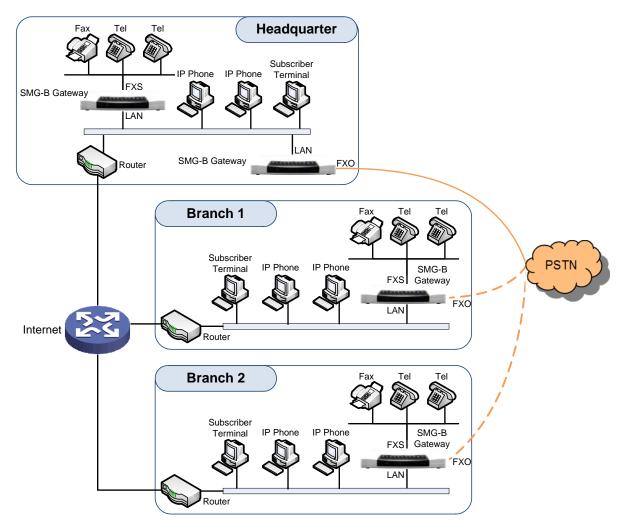


Figure 1-1 Typical Application

1.2 Feature List

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



	auto dial number will be called.		
Do Not Disturb	Rejects all the incoming calls to the channel.		
CID	Displays the CallerID.		
Echo Cancellation	Provides the echo cancellation feature for a call conversation over the FXS channel.		
TDM/VoIP Routing	Sets a routing path: from IP to TDM or from TDM to IP.		
Fax	Provides multiple fax parameters: fax mode, maximum fax rate, fax train mode, error correction mode, etc.		
Communication without Power	Enable a connection of the station which is linked with the FXS port and the trunk which is linked with the FXO port to keep the calls between the FXS port and PSTN uninterrupted during power outage.		
Communication without Network	Automatically routes a call to the proper port according to the configuration in case of network failure or call timeout.		
Send Polarity Reversal Signal	Sends the polarity reversal signal to a corresponding FXS channel when the called party pick-up behavior is detected.		
Detect Polarity Reversal Signal	Turns a corresponding channel into the talking state when the FXO port detects the polarity reversal signal.		
Simultaneous Register to Multiple Servers	Registers the gateway to a master registrar server and a spare registrar server simultaneously.		
IMS Network	Registers the gateway to a server under IMS network.		
SIP Station	Supports a SIP terminal to be registered to the gateway and become a SIP station.		
Group Ringing	Rings all the idle FXS ports in a port group.		
Ringing by Turns	Rings the FXS ports in a port group by turns according to the <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.		
Centralized Manage	The gateway can register to Synway DCMS and accept the management of the platform.		
Signaling & Protocol	Description		
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261.		
Voice	CODEC G.711A, G.711U, G.729A/B, G.723, G.722, AMR, iLBC DTMF Mode RFC2833, SIP INFO, INBAND		
Network	Description		
	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN.		
Network Protocol			
Static IP	IP address modification support.		

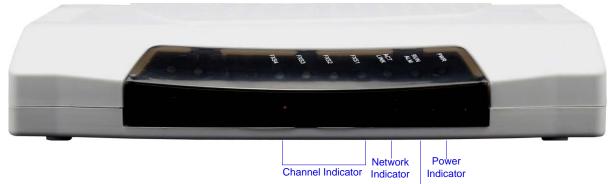


DNS	Domain Name Service support.		
Security	Description		
Admin Authentication	Supports admin authentication to guarantee the resource and data security.		
System Monitor	Monitors the running status of the system and the server.		
Maintain & Upgrade	Description		
WEB Configuration	Support of configurations 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, DEBUG.		

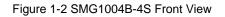
1.3 Hardware Description

1.3.1 SMG-B 4-port Analog Gateway

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. See below for product appearance (taking SMG1004B-4S for example).



Alarm & Run Indicator



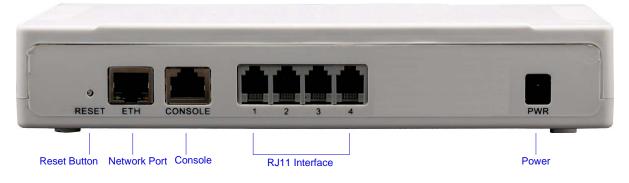




Figure 1-3 SMG1004B-4S Rear View

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).

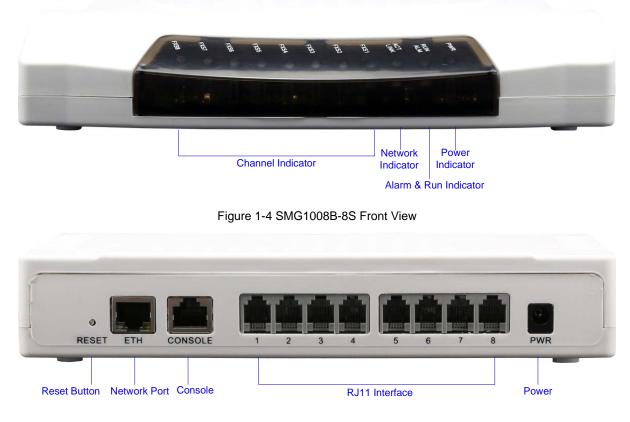


Figure 1-5 SMG1008B-8S Rear View

1.3.3 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).







RJ45 Interface



Grounding Stun 220V AC Power Key

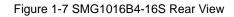




Figure 1-8 SMG1016B4-16S Left View

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		
FXS	Type: RJ-11, RJ-45		
FXS	Maximum Transmission Distance: 5000m		
	Charge Mode: Negative Anti-billing Supported		
	Amount: 1		
	Type: RS-232		
	Baud Rate: 115200bps		
Console Port	Connector: RJ45 to DB-9 Connector		
Console Port	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		
Reset Button	Restore the gateway to factory settings by pressing this button persistently for 3		
Resel Dullon	seconds		
LED	Description		
Power Indicator	Indicates the power state. It lights up when the gateway starts up with the power		



	cord well connected		
Run & Alarm Indicator	Indicates the running status. For more details, refer to <u>1.4 Indicator Info</u> .		
	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 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 (32 FXO ports). It supports one LAN.

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.	
	Orange LED light up	Device works normal upon system startup.	
Run & Alarm Indicator	Orange LED flash fast Device works normal upon system startup.		
	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	
LINK Indicator		10Mbps	
LINK Indicator	Orean LED light up	Connect to 100Mbps network, and network connection is	
	Green LED light up	normal.	
		Connect to 10Mbps network: communication is normal;	
ACTIVE Indicator	Go out	While connect to 100Mbps network: Communication is	
ACTIVE Indicator		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:



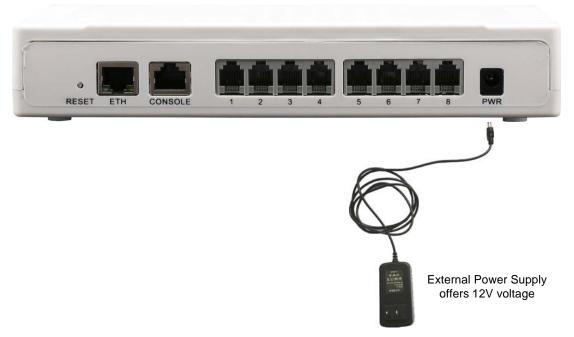


Figure 2-2 SMG-B Power Connection

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>3.9.16 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>3.9.3 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 \rightarrow 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>3.5.9 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>3.6.4 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>3.6.4 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.

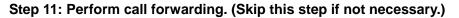


Step 9: Enable the DND (do not disturb) feature. (Skip this step if not necessary.)

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.



Situation 1: Hook-flash operation

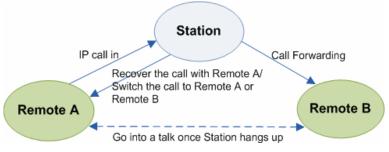


Figure 2-3 Call Forward via Hook-flash

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.

2	中立 English	
Username:		
Login		

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>3.9.16 Change Password</u>.

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

		System	n Info	
	LAN			
		00:00:E0:A7:01:6A		
			255 255 255 0	201.123.115.254
t	IPV6 Address	ff.:1:2:3:4	64	
	DNS Server	0.0.0.0		
*	Receive Packets	All:53492	Error:0	Drop:0
	Transmit Packets	All:72194	Error:0	Drop:0
*	Current Speed	Receive: 1.8 KB/s	Transmit:6.3 KB/s	
*	Work Mode	100Mb/s Full Duplex		
*	Runtime	52m 50s		
*	Current Version			
0		forcommon 171 Pole	2002017042100	
~				
*			38362017042103	
			3 CST 2017	
	Product Type	1008B-4S4O(RJ11)		
	* * * * *	Prov Address DNS Server Receive Packets Transmit Packets Current Speed Work Mode Work Mode Current Version WEB Gateway Serial No. U-boot Kernel	LAN MAC Address 00:00:E0:A7:01:6A IP Address 201.123.115.74 IP V6 Address ft:12:3:4 DNS Server 0.0.0 Receive Packets All:53492 Transmit Packets All:53492 Transmit Packets All:53494 Current Speed Receive:1.8 KB/s Work Mode 100Mb/s Full Duplex Runtime 52m 50s Eurent Version WEB forcommon_1.7.1_Rel/s Gateway forcommon_1.7.1_Rel/s Gateway forcommon_1.7.1_Rel/s Gateway forcommon_1.7.1_Rel/s Gateway forcommon_1.7.1_Rel/s Gateway forcommon_1.7.1_Rel/s Serial No. 00001678 U-boot Mar 02 2016-21.57:12 Kernel #210 Fri Apr 14 09:25:2	MAC Address 00:00:E0:A7:01:6A IP Address 201.123.115.74 255.255.255.0 IP V6 Address 201.123.115.74 64 DNS Server 0.0.0 Receive Packets All:53492 Error.0 Transmit Packets All:72194 Error.0 Current Speed Receive: 1.8 KB/s Transmit.6.3 KB/s Work Mode 100Mb/s Full Duplex Vork Mode Work Mode 100Mb/s Full Duplex Error.0 WeB forcommon_1.7.1_Release2017042109 Gateway forcommon_1.7.1_Release2017042109 Serial No. 0001678 U-boot Mar 02 2016-21:57.12 Kernel #210 Fri Apr 14 09:25:23 CST 2017

Figure 3-2 Main Interface



3.2 Operation Info

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



Figure 3-3 Operation Info

3.2.1 System Info

MAC Address	00:00:E0:A7:01:6A		
IP Address	201.123.115.74	255.255.255.0	201.123.115.254
IPV6 Address	ff::1:2:3:4	64	
DNS Server	0.0.0		
Receive Packets	All:53492	Error:0	Drop:0
Transmit Packets	All:72194	Error:0	Drop:0
Current Speed	Receive: 1.8 KB/s	Transmit:6.3 KB/s	
Work Mode	100Mb/s Full Duplex		
Runtime	52m 50s		
Current Version			
WEB	forcommon_1.7.1_Rel	ease2017042109	
Gateway	forcommon_1.7.1_Rel	ease2017042109	
Serial No.	00001678		
U-boot	Mar 02 2016-21:57:12		
Kernel	#210 Fri Apr 14 09:25:2	23 CST 2017	
Product Type	1008B-4S4O(RJ11)		

Figure 3-4 System Info Interface

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

ltem	Description
MAC Address	MAC address of LAN.
	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.
Receive Packets	The amount of receive packets after the gateway's startup, including three options:



	All, Error and Drop.
	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.
	Show the work mode of the network, including four modes: 10 Mbps Half Duplex, 10
Work Mode	Mbps Full Duplex, 100 Mbps Half Duplex, 100 Mbps Full Duplex.
Denting	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	Туре	Number	Voltage(v)	State	Direction	CallerID	CalleeID	Reg Status	Polarity Reversal Count
1	FXS	8001	0					Unregistered	1000
2	FXS	8002	0		2 <u></u>	2 <u></u> 2	3 <u></u> 1	Unregistered	<u>213</u>
3	FXS	8003	0					Unregistered	
4	FXS	8004	0			()	·	Unregistered	
5	FXO	8005	0	6	-()	-07753	-	Unregistered	
6	FXO	8006	0	6	2 <u>9117</u> 3	2 <u></u> 2	2 <u>322</u> 3	Unregistered	222
7	FXO	8007	0	6	()	()		Unregistered	
8	FXO	8008	0	5				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 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 channe state icon for detailed state information.					
	State	lcon	Description			
State	Idle		The channel is available.			
	Off-hook		The channel picks up the call.			
	Wait Answer		The channel receives the ringback tone and is waiting			
	wait Answer		for the called party to pick up the phone.			



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	<i>Ringing</i> The channel is in the ringing state.
	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.
CalleeID	Displays the CalleeID of the call on channel.
Reg Status	Displays the registration status of the port.
Polarity Reversal Count	The counts of the polarity reversal detected by the FXO port.

3.2.3 Call Count

				Call C	ount			
Direction	Total Calls	Successful Calls	Busy	No Answer	Call Forward	Routing Failure	Dialing Failure	Unknown Failure
P->Tel	0	0	0	0	0	0	0	0
el->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.

Item	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.
Disting Failure	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.



3.2.4 SIP Message Count

				Reques	t				
Request	RE	GISTER	INVITE	ACK	INFO	BYE	CANCEL	NOTIFY	OPTION
Send		0	1	1	0	1	0	0	0
Send Repeatedly		0	0	0	0	0	0	0	0
Receive		0	1	1	0	1	0	0	0
Receive Repeatedly		0	0	0	0	0	0	0	0
				Common Res	ponse				
	7			Common Res	ponse		Å.		
	100 Trying	180 Ringing		Common Res Session Prose	1	200 OK	486 Busy	487 Request Alread	y Terminated
Common Response Send	100 Trying 1	180 Ringing 1			1	200 OK	486 Busy 0	487 Request Alread	y Terminated

Figure 3-7 SIP Message Count Interface

See Figure 3-7 for the SIP Message Count interface. This is used to record the amount of the normal SIP messages that are sent/received or repeatedly sent/received during the period from the startup of the gateway service to the latest open or refresh of the interface. Click **Refresh** to refresh the count of SIP messages, or click **Clear** to clear the current count of SIP messages.

3.3 Quick Config

🕂 Quick Config	*
Quick Config	

Figure 3-8 Quick Config Interface

See Figure 3-8 for the Quick Config interface. Follow the gateway Quick Configuration wizard and you can easily complete the settings on network, SIP and FXS/FXO. The gateway can work normally after configuration.

See Figure 3-9 for the Quick Config-Network Settings interface. Refer to <u>3.9.3 Network</u> for detailed settings. After configuration, click *Next* to enter the SIP Settings interface.



Quick Config-	Network Settings
Network Type:	Static
IP Address (I)	201.123.115.74
Subnet Mask (U)	255.255.255.0
Default Gateway (D)	201.123.115.254
DNS Server (P)	0.0.0.0
IPv6 Address:	☑启用
IPv6 IP Address(I):	ff::1:2:3:4
IPv6 Subnet Mask(U):	64
Default IPv6 Gateway(D):	ff::1:2:3:0
IPv6 DNS Server(P):	ff::1:2:3:1
Speed and Duplex Mode	Automatic Detection

Figure 3-9 Quick Config-Network Settings Interface

See Figure 3-10 for the Quick Config-SIP Settings interface. The configuration items on this interface are the same as those on the SIP interface. Refer to <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 3-10 Quick Config-SIP Settings Interface

See Figure 3-11 for the FXS Settings interface. The configuration items on this interface are the same as those on the FXS interface. Refer to <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 FXO Settings interface.



									FXS 5	iettings				FXS Settings						
Port	Type	SIP Account	Display Name	Authentication Username	Auto Diat Num	Forbid Outgoing Call	DND	Forward	FWD Type	FWD Number	CID	Call Waiting	Reg Status	Eche Canceller	Color Ring	Color Ring Index	ServerIndex	Input Gain	Output Gain	Modi
1	F)(\$	8001	-	-		Disattie	Disable	Disable		-	Enable	Disable	Unregistered	Enable	Disable		-	0	0	12
2	F/(\$	8002	-	-	-	Disable	Disable	Disable	-	-	Enable	Disable	Unregistered	Enable	Disable		5 mm	0	0	1
3	FXS	8003	-	(i+-))	-	Disable	Disable	Disable	-	-	Enable	Disable	Unregistered	Enable	Disable		1.000	0	0	Gr
4	F10	8004	- 	3497		Disable	Disable	Disable	-	2	Enable	Disable	Unregistered	Enable	Disable	(##):	-	0	0	12
-																				_
terra	Total 1	ItemsPage 1	11 First Previou	s Next Last Go to Page 1	1 Pages Total														B	atch Mo
									_	-										
									Back	tioit.										

Figure 3-11 FXS Settings Interface

See Figure 3-12 for the FXO Settings Interface. The configuration items on this interface are the same as those on the FXO interface. Refer to <u>3.6.2 FXO</u> for detailed settings. After configuration, click *Back* to back to the FXS Settings interface; click *Next* to enter the Quick Config-Completion interface. See Figure 3-13.

					FXO Settings					
Port	Туре	SIP Account	Display Name	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	F
5	FXO	8005	8005	Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	
6	FXO	8006	8006	Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	
7	FXO	8007	8007	Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	
8	FXO	8008	8008	Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	
tems	Total 1	6 Items/Page 1	1/1 First Previou	s Next Last Go to Page 1 🔹 1 Pages	Total				Batch M	od
				Quick C	onfig-Com	pletion				

Figure 3-13 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 six parts: *SIP*, *SIP Compatibility*, *SIP Station*, *SIP Server*, *NAT Setting* and *Media*. See Figure 3-14. *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 Station* is to set the basic information of the SIP station, *SIP Server* is to set the basic information of the SIP server, *NAT Setting* is used to configure the parameters for NAT, and *Media Settings* is to set the RTP port and the payload type.





Figure 3-14 VoIP Settings

3.4.1 SIP

SI	P Settings
SIP Address	LAN 1 IPv6: ff::1:2:3:4
SIP Port	5060
Register Status	Failed(No Response)
Register Gateway	
SIP Account	Yes
Password	
Authentication Username	
Registrar IP Address	
Registrar Port	
Spare Registrar Server	✓Enable
Spare Registrar IP Address	
Spare Registrar Port	
Registry Validity Period (s)	600
Multi-Registrar Server Mode	Enable
SIP Transport Protocol	UDP
Switch Signal Port if SIP Registrati	ion Failed 🔲 Enable
IMS Network	Enable
Externally Bound Address	
Externally Bound Port	5060

Figure 3-15 SIP Settings Interface



See Figure 3-15 for the SIP settings interface where you can configure the general SIP parameters. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to <u>3.9.17 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-15.

ltem	Description
SIP Address	IP address of SIP signaling, using LAN 1 by default.
	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 Register Gateway is set to No, the value
Register Status	of this item is <i>Unregistered</i> ; 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.
	When the gateway initiates a call to SIP, this item corresponds to the username of
SIP Account	SIP.
Deserver	Registration password of the gateway. To register the gateway to SIP, both
Password	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.
Registry Validity	Validity period of the SIP registry. Once the registry is overdue, the gateway should
Period	be registered again. This configuration item is valid only when <i>Register Gateway</i> is
	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> .
Switch Signal Port if	If the SIP registration fails, the SIP signaling port N will switch to N+1 for a new
SIP Registration	registration. It will continue until the registration succeeds. The default value is
Failed	disabled.
	Once this feature is enabled, the gateway will send signaling messages to the
IMS Network	corresponding externally bound address and port when it registers to the server. By
	default, this feature is disabled. Only when this feature is enabled will these items



	Externally Bound Address, Externally Bound Port and Authentication							
	<i>Username</i> be shown.							
Externally Bound	Externally bound ID address for registration							
Address	Externally bound IP address for registration.							
Externally Bound	Externally bound port for registration.							
Port								

3.4.2 SIP Compatibility

See Figure 3-16 for the SIP Compatibility interface where you can configure the SIP parameters to determine which SIP servers and SIP messages will the gateway be compatible with. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations.



 SIP Compatibility	
Obtain CalleeID from	"Request" Field
Set CallerID position	Username of From Field 💌
Obtain CallerID from	Username of From Field
Use Contact Address	Enable
Call Transfer Mode	Internal Handling
Internal Handle	Match Port Number
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
SIP Station Supported	✓Enable
Set SIP Identifying	Gateway
Maximum Wait RTP Time (s)	15
Call Abnormal Hangup Detection	✓Enable
Cycle(s)	0
Server Status Detection	Enable
Cycle(s)	0
Send Cue Tone	Enable
SIP Encryption	Enable
Encryption Criterion	VOS1.1
Identifier	
Key	
RTP Encryption	Enable
Ignore ACK	Enable
User-defined SIP Code	Enable
Use Iptables	Enable
Save	



Figure 3-16 SIP Compatibility Setting Interface

The table below explains the items shown in Figure 3-16.

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'.
	Sets whether to send the request message according to the content of Contact, with
Use Contact	the default setting of disabled. As it is disabled, if the Contact field indicates an IP
Address	address within the LAN, the request message will be sent according to the source
Addicos	address; if the Contact field indicates an IP address belonging to the WAN, the
	request message will be sent according to this IP address.
Call Transfer Mode	There are two optional ways to deal with call transfer: Internal Handling and
	Platform to Handle SIP Info. The default value is Internal Handling.
Internal Handle	Sets the internal handle mode for the call transfer, including two options: Match Port
	Number and Search Idle FXO Channel. The default value is Match Port Number.
Call Flash Mode	There are two optional ways to deal with call flash: Internal Handling and Platform to
	Handle SIP Info. The default value is Internal Handling.
Hold Music Source	Sets the source of the hold music, with the default value of <i>Remote</i> , This feature
	gets valid only when you choose the mode <i>Platform to Handle SIP Info.</i>
Two Stage Dialing	Once this feature is enabled, the incoming call from SIP should perform the two
for SIP Incoming	stage dialing operation. By default this feature is disabled.
Call	
	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.
SIP Station	Once this feature is enabled, a SIP terminal can be registered to the gateway and
Supported	becomes a SIP station. By default this feature is disabled.
Set SIP Identifying	Sets the SIP identifying content in the SIP call message. The default setting is
	Gateway.
	Sets the maximum time for the SIP channel to wait for the RTP packet. If no RTP
Maximum Wait RTP	packet is received within the specified time period, the channel will enter the
Time	pending state automatically and release the call. The default value is <i>15</i> , calculated
	by s.
Call Abnormal	Sets the interval between checks of the remote end's abnormal hangup, with the
Hangup Detection	default value of 0 (feature disabled), calculated by s. It is suggested to set to 10s if
	this feature is necessary to be used.



Server Status Detection Cycle	The interval of sending a heartbeat packet to detect the master registrar server status, with the default value of 0 (feature disabled), calculated by s. It is suggested to set to 15s if this feature is necessary to be used.
Send Cue Tone	Sets whether to send a cue tone once the server gets disconnected, with the default setting of <i>disabled</i> .
SIP Encryption	Once this feature is enabled, you can encrypt the SIP signal following selecting an encryption criterion and setting a key. By default it is <i>disabled</i> .
Encryption Criterion	The criterion used to encrypt the SIP signal. At present only VOS1.1 is supported.
Identifier	The identifier field of the VOS encryption, which is used to obtain the key of the SIP encryption.
Кеу	The key to encrypt the SIP signal.
RTP Encryption	Once this feature is enabled, you can encrypt the RTP package. By default it is disabled.
Ignore ACK	Once this feature is enabled, the gateway is not necessary to wait for the ACK message after sending the 2000K message to establish a call. By default it is <i>disabled</i> .
User-defined SIP	Once this feature is enabled, you can define a SIP code for the corresponding SIP
Code	status, with the default value of <i>disabled</i> .
Use Iptables	Once this feature is enabled, only the calls from the SIP registration server, the source IP address of the route IP->TEL and these IP addressed set in <u>Access</u> <u>Control</u> interface are permitted.

3.4.3 SIP Station

A SIP terminal can be registered to the gateway and becomes a SIP station. Enable the feature of 'SIP Station Supported' on <u>3.4.2 SIP Compatibility</u>, and you will see the item SIP Station on the VoIP Settings menu. Click 'SIP Station' to go into the SIP Station interface. By default, there is no available SIP station. See Figure 3-17 below.

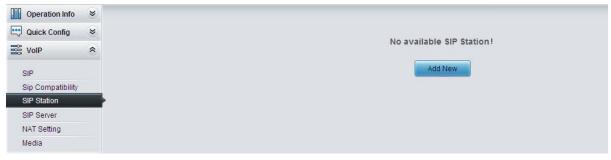


Figure 3-17 SIP Station Setting Interface

Click *Add New* to add SIP stations manually. See Figure 3-18. You can configure basic SIP station information on this interface. The bound port to a SIP station must be an FXO port and unique. The username must be the same as that used to register the SIP terminal to the gateway.



s	IP Station
Number:	0
Username:	
Password:	
Bound Port:	5
Description:	default
Batch Setting:	Enable
Save	Close

Figure 3-18 Add New SIP Station

The table below explains the items shown above:

Item	Description
Number	The logical number for a SIP station to register to the gateway.
Username The username used to register a SIP station to the gateway.	
Password	The password used to register a SIP station to the gateway.
Bound Port	The FXO port which is bound to the SIP station.
Description	It is user-defined, with the default value of <i>default</i> .
Batch Setting	Used to set multiple SIP stations at the same time.

After configuration, click *Save* to save the above settings into the gateway or click *Close* to cancel the settings. See Figure 3-19 for the applied SIP station information.

					SIP Station				
Check	Number	Username	IP Address	Bound Port	Register Status	Register Duration (s)	Voice Channel State	Description	Modify
	0	123		5	Unregistered	-	-	default	
Check All	Uncheck	All Invers	e 🗄 Delet	e Clear A	1				Add Nev

Figure 3-19 SIP Station Interface

Click *Modify* in the above figure to modify the configuration of the SIP station. See Figure 3-20. The configuration items on this interface are the same as those on the *Add New SIP Station* interface.



	SIP Station
Number:	0
Username:	123
Password:	•••
Bound Port:	5 💌
Description:	default
Batch Setting:	Enable
Save	Close

Figure 3-20 SIP Station Modification Interface

To delete a SIP station, check the checkbox before the corresponding index in Figure 3-19 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all SIP stations at a time, click the **Clear All** button in Figure 3-19.

3.4.4 SIP Server

The gateway supports the multi-registrar server feature. Enable the feature of '*Multi-Registrar Server Mode*' on the <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 3-21 below.

Operation Info	*			
Quick Config > Wolp > SIP				
		No Available Registrar Server!		
		Add New		
Sip Compatibility				
SIP Server				
NAT Setting				
Media				

Figure 3-21 SIP Server Interface

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



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 3-22 Add New SIP Server

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

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

After configuration, click **Save** to save the above settings into the gateway or click **Cancel** to cancel the settings. See Figure 3-23 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	<u> </u>		600			

Figure 3-23 SIP Server Management

Click *Modify* in the above figure to modify the configuration of the SIP server. See Figure 3-24.

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



Index	1
Description	defalut
Registrar IP Address	201.123.115.233
Registrar Port	5060
Registry Validity Period (s)	600
IMS Network	Enable
Externally Bound Address	201.123.123.145
Externally Bound Port	5060

Figure 3-24 SIP Server Modification Interface

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

3.4.5 NAT Setting

See Figure 3-25 for the NAT setting interface where you can configure the parameters for NAT. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations.



	NAT Setting	js
Local NAT Trave	irsal	
Meth	od 1:	
	Auto Nat	Enable PMP
	Outer Network Address	Offline
Meth	od 2:	
	STUN Server	C Enable
	NAT Type	Unknown
	STUN Server Address	127.0.0.1
Meth	od 3:	
	Mapping Contact IP	
	Mapping SDP IP	
Meth	od 4:	
	Rport	Enable
	Auto Detect NAT IP	Enable
Help Remote D	evice Complete NAT Traversal	
	RTP Self-adaption	Enable
"Loc "Auto "Map	non-professional person please do not modi al NAT Traversal": Please select one method Nat": It is required to enable the feature of up ping Contact IP": It is required to set the route ping SDP IP": It is required to set the router to	according to your current network environment. oon or pmp for the router. er to map the SIP port to the gateway.

Figure 3-25 NAT Setting Interface

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

Item	Description				
Auto Not	Sets whether to enable the Auto Nat feature. Three options are available:				
Auto Nat	DisableAutoNat, Enable PMP and Enable UPNP, with the default value of Auto Nat.				
Outer Network	The address of the outer network acquired automatically once the PMP or UPNP				



Address	feature is enabled.
STUN Server	Sets whether to enable the STUN server for NAT traversal. By default the STUN
	server is disabled.
	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.
Mapping Contact IP	The IP filled in here will be used in the Contact field of the SIP message.
Mapping SDP IP	The IP filled in here will be used in the SDP field of the SIP message.
Drawt	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 and
Auto Detect NAT IP	port in the message returned by Rport so as to use them for the following
AULO DELECLINAT IP	communication. By default, this feature is disabled.
	Note: This feature gets valid only when Rport is enabled.
	When this feature is enabled, the RTP reception address or port carried by the
DTD Calf adaption	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.



3.4.6 Media

		Media Pa	rameters	
	DTMF Transmit	Mode	RFC2833	v
	RFC2833 Paylo	ad	101	
	RTP Port Rang	e	50000,5076	7
	Silence Suppre	ssion	Disable	~
	JitterMode		Static Mode	×
	JitterBuffer(ms)		20	
	Voice Gain Out	out from IP (dB)	0	
	AGC		☑Enable	
	Target Energy 1	hreshold (dB)	0	
	Maximum Gain	Threshold (dB)	48	
	Maximum Atten	uation Threshold (dB)	0	
	Minimum Input	Energy (dB)	-60	
CODEC P	riority			
Check	Priority	CODEC	Packing Time	Bit Rate (kbs)
	1	G711A 💌	20 💌	64 💌
	2	G711U 🗸	20 💌	64 🗙
	3 4	G729 🗸	20 💙	8 💌
	5	G722 V	30 🗸	64 🗸
	6	AMR 💌	20 💌	4.75 💌
	7	iLBC 💌	30 💌	13.3 💌
4		Save	Reset	

Figure 3-26 Media Settings Interface

See Figure 3-26 for the media settings interface where you can configure the RTP port and payload type depending on your requirements. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to <u>3.9.17 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-26.

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.
DEC2822 Dayland	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> .
litter Me de	Sets the mode for the Jitter buffer. The optional mode is Static Mode and Adaptive
JitterMode	Mode, with the default value of Static Mode.
	Acceptable jitter for data packets transmission over IP, which indicates the buffering
	capacity. A larger JitterBuffer means a higher jitter processing capability but as well
litter Duffer	as an increased voice delay, while a smaller JitterBuffer means a lower jitter
JitterBuffer	processing capability but as well as a decreased voice delay. Range of value:
	20~200, calculated by ms, with the default value of 20.
	Note: This is only valid if the Jitter Mode is set to Static Mode.
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	If the AGC (Automatic Gain Control) feature is enabled, the gateway will automatically adjust the input signal amplitude, increasing that of small signals and
-	
	automatically adjust the input signal amplitude, increasing that of small signals and
AGC	automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals.
AGC Target Energy	automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals. Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the
AGC Target Energy Threshold	automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals. Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the default value of 0.
AGC Target Energy Threshold Maximum Gain	automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals. Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the default value of 0. Set the maximum gain threshold that will be applied to the signal. Range of value: 0~48, calculated by dB, with the default value of 48.
AGC Target Energy Threshold Maximum Gain Threshold	 automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals. Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the default value of 0. Set the maximum gain threshold that will be applied to the signal. Range of value: 0~48, calculated by dB, with the default value of 48. Set the maximum attenuation that will be applied to the signal. Range of value:
AGC Target Energy Threshold Maximum Gain Threshold Maximum	automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals. Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the default value of 0. Set the maximum gain threshold that will be applied to the signal. Range of value: 0~48, calculated by dB, with the default value of 48.
AGC Target Energy Threshold Maximum Gain Threshold Maximum Attenuation Threshold	 automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals. Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the default value of 0. Set the maximum gain threshold that will be applied to the signal. Range of value: 0~48, calculated by dB, with the default value of 48. Set the maximum attenuation that will be applied to the signal. Range of value:
AGC Target Energy Threshold Maximum Gain Threshold Maximum Attenuation	automatically adjust the input signal amplitude, increasing that of small signals and decreasing that of large signals. Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the default value of 0. Set the maximum gain threshold that will be applied to the signal. Range of value: 0~48, calculated by dB, with the default value of 48. Set the maximum attenuation that will be applied to the signal. Range of value: -42~0, calculated by dB, with the default value of 0.



	Supported COD	ECs and their corresponding	priority for the IP end to establish a
	call conversation	n. The table below explains the	e sub-items:
	Sub-item	De	escription
	Priority	Priority for choosing the Co smaller the value is, the hig	ODEC in an SIP conversation. The her the priority will be.
	CODEC	Three optional CODECs G729A/B, G723, G722, AM	are supported: <i>G711A</i> , <i>G711U</i> , R and <i>iLBC</i> .
	Packing Time	Time interval for packing an	RTP packet, calculated by ms.
	Bit Rate	The number of thousand bit are conveyed per second.	s (excluding the packet header) that
CODEC Priority	G729A/B, G723	, G722, AMR and iLBC by pric	oported and ordered G711A, G711U, prity from high to low. ferent CODECs are listed in the table
		lues in bold face are the defau	
	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
	iLBC	30 / 60	13.3 / 15.2

3.5 Advanced Settings

Advanced Settings includes fourteen parts: *FXS*, *FXO*, *Tone Detector*, *Tone Generator*, *DTMF*, *Ringing Scheme*, *Fax*, *Function Key*, *Dialing Rule*, *Dialing Timeout*, *Cue Tone*, *Color Ring*, *QoS* and *Action URL*. See Figure 3-27. *FXS* is used to configure the general properties of the FXS port; *FXO* is used to configure the general properties of the FXO port; *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. *Action URL* is used to designate the server path to report the on-hook or off-hook state of the FXS channel.





Figure 3-27 Advanced Settings

3.5.1 FXS

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
Handling of Call from Internal Station	Platform Handling
Light Up Mode for Voice Message	Not Light Up
Echo Canceller	
Work Mode	Near-end cancella
Non-linear Processing	Enable
Fixed Window Size (Near-end, Narrowband 8kHz)	16ms

Figure 3-28 FXS Configuration Interface

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

Item	Description
Hook-flash Detection	Sets whether to enable the hook-flash detection feature or not, with the default
HOOK-HASH Delection	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
	operation. Range of value: 80~ Maximum Time, 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.
CID Transmit Made	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.
Sand Palarity	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 <i>disabled</i> .
Handling of Call from	Sets the handling mode for the calls from station to station, two options available:
Handling of Call from Internal Station	Internal Handling and Platform Handling, with the default value of Platform
	Handling.
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.
	Sets the work mode for the echo canceller. There are two options: Near-end
Work Mode	cancellation and Both near-end and far-end cancellation, with the default value of
	Near-end cancellation.
Non-linear	Sets whether to enable the mode of non-linear processing. By default, this feature is
Processing	enabled.
Fixed Window Size	Sets the size of the window for the fixed cancellation.
Moving Window Size	Sets the size of the window for the moving cancellation.

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 system, do it immediately to apply the changes. Refer to <u>3.9.17 Restart</u> for detailed instructions.



3.5.2 FXO

FXO	
Calling Party Detection Time (s)	10
Silence Detection(FXO will Hang up the Call upon Detecting the Silence.)	Enable
Energy Threshold of Silence (dB)	-30
Time Threshold of Silence (s)	60
Time Threshold to Avoid on-line Voice Dithering (ms)	48
Incoming Call from PSTN	
Rapid Release	Enable
FSK Standard	GR-30(North America, China) 💌
FSK Detection Parameters	Manual
Minimum Energy Threshold for FSK Detection (0.1db)	-380
Minimum Numbers of FSK Seizure Bits	100
Minimum Numbers of FSK Mark Bits on on- hook	40
Minimum Numbers of FSK Mark Bits on off- hook	50
Reception Interval of DTMF CallerID (ms)	250
Delay for Two Stages Dialing (s)	0
Outgoing Call to PSTN	
Flash Time (ms)	100
Delay after Dial (ms)	1000
FXO Pick-up Delay after INVITE Received at IP Side (s	0
Maximum Wait Answer Time (s)(Valid when Polarity Reversal Enabled)	60
Communicate without Network	Enable
Communicate without Network Mode	Auto search idle c 💌
Two Stage Dialing Mode	Enable
Delay to Send 200 OK to IP Side (Invalid if Polarity Reversal is enabled)	Enable
Open Session In-Advance	Enable
Avoid Being Detected as Flash Signal by PBX	Enable
Echo Cancellor	
Work Mode	Near Far Ending 💌
Non Linear Process	✓Enable
Fixed Window Size(Near,8kHz)	8ms 📼
Moving Window Size(Far,8kHz)	8ms 💌
Select Area for Module Parameters	USA 👻
	Land
Save Reset	

Figure 3-29 FXO Configuration Interface

The table below explains the particular configuration items for FXO.

Item	Description



Calling Party	The maximum waiting time for the detection of the calling party number from FXO
Detection Time	port. Range of value: 1~20, calculated by s, with the default value of <i>10</i> .
Detection Time	Used to detect whether the line is silent or not according to the energy threshold
Silence Detection	and time threshold of silence. FXO will hang up the call automatically if these
Sherice Detection	conditions are satisfied. The default setting is being disabled.
Energy Threshold of	The energy threshold to judge whether the line is silent or not. The signal with the
Energy Threshold of	energy less than this set value will be determined to be silence. Range of value:
Silence	-86~6, calculated by s, with the default value of -30.
	Note: This item will be valid only when Silence Detection is enabled.
Time Threshold of	The time threshold to judge whether the line is silent or not, calculated by s, with the
Silence	default value of 60.
	Note: This item will be valid only when Silence Detection is enabled.
Time Threshold to	Once this feature is enabled, the on-line voice will be determined to be dithering if
Avoid On-line Voice	the voice duration is less than the set value here. Range of value: 24~1000,
Dithering	calculated by ms, with the default value of <i>48</i> ,
	Once this feature is enabled, the FXO port will release the source rapidly and go to
Rapid Release	the idle state when a call from PSTN to soft-terminal via FXO port is rejected by the
	IP soft-terminal.
	Standard for sending FSK formatted CallerID, which varies in different countries and
FSK Standard	districts. The optional values are: ETSI (Europe), GR-30 (North America, China)
	and <i>NIT (Japan)</i> , with the default value of <i>GR-30</i> .
	Sets the configuration mode of the FSK parameters, two options available: Default
FSK Detection	and Manual, with the default value of <i>Default</i> . In the Default mode, the FSK
Parameters	parameters use default values and cannot be modified. To modify the parameters,
	please select the Manual mode.
Minimum Energy	Sets the minimum energy threshold for the FSK detection. Range of value: -1125~0,
Threshold for FSK	calculated by 0.1dB, with the default value of -380.
Detection	
Minimum Number of	The FSK seizure bits (0x55) under the on-hook mode. As the protocol provides, it
FSK Seizure Bits	shall be 300 consecutive bit groups of 0 and 1. Range of value: 0~32768, with the
	default value of 100.
Minimum Number of	The number of FSK marking signals (0xFF) under the on-hook mode. As the
FSK Mark Bits on	protocol provides, it shall be 180 consecutive bits of 1. Range of value: 0~32768,
on-hook	with the default value of 40.
Minimum Number of	The numbers of FSK marking signals (0xFF) under the off-hook mode. As the
FSK Mark Bits on	protocol provides, it shall be 80 consecutive bits of 1. Range of value: 0~32768, with
off-hook	the default value of 50.
Reception Interval of	The time interval between digits of the DTMF CallerID from FXO port, calculated by
DTMF CallerID	ms, with the default value of 250.
Dolou for Two Storios	If the feature of two-stages dialing mode is enabled and an incoming call occurs, the
Delay for Two Stages	FXO port will have a delay set by this configuration item before going into the
Dialing	two-stages dialing process,



Flash TimeSets the time for generating a flash signal on the analog trunk. Range of value: 32-1000, calculated by ms, with the default value of 100.Delay after DialSets the delay to send the CalleeID to PBX after you pick up and dial. Range of value: 200-2000, calculated by ms, with the default value of 1000.FXO Pick-up Delay after INVITE Received at IP SideOnce this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message.Maximum Wait Answer TimeThe maximum time to wait the answer of the remote side for an outgoing call from FXO port. This item is valid only when Polarity Reversal is enabled. It is calculated by s, with the default value of 60.Communication without NetworkAutomatically routes a call to the proper port according to the configuration in case of network failure or call timeout. The default value is <i>enabled</i> .Communicate without NetworkSets the mode for the communications without network, two options available: Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of TeI->IP route.Two Stages Dialing ModeSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is <i>enabled</i> .Open Session In AdvanceOnce this feature is enabled, after hanging up a call, the FX
Delay after Dialvalue: 200–2000, calculated by ms, with the default value of 1000.FXO Pick-up Delay after INVITEOnce this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message.Maximum Wait Answer TimeThe maximum time to wait the answer of the remote side for an outgoing call from FXO port. This item is valid only when Polarity Reversal is enabled. It is calculated by s, with the default value of 60.Communication without NetworkAutomatically routes a call to the proper port according to the configuration in case of network failure or call timeout. The default value is enabled.Communicate without NetworkSets the mode for the communications without network, two options available: Auto Search Idle Channel and Use Current Route Setting, with the default value of Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port. By default this feature is disabled.Two Stages Dialing ModeSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Delay to Send 200 Of the IP side.Once this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled.Avoid Being Detected as Flash Sianal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
Delay after Dialvalue: 200–2000, calculated by ms, with the default value of 1000.FXO Pick-up Delay after INVITEOnce this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message.Maximum Wait Answer TimeThe maximum time to wait the answer of the remote side for an outgoing call from FXO port. This item is valid only when Polarity Reversal is enabled. It is calculated by s, with the default value of 60.Communication without NetworkAutomatically routes a call to the proper port according to the configuration in case of network failure or call timeout. The default value is enabled.Communicate without NetworkSets the mode for the communications without network, two options available: Auto Search Idle Channel and Use Current Route Setting, with the default value of Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port. By default this feature is disabled.Two Stages Dialing ModeSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Delay to Send 200 Of the IP side.Once this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is enabled.Avoid Being Detected as Flash Sianal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
FXO Pick-up Delay after INVITEOnce this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message.Maximum Wait Answer TimeThe maximum time to wait the answer of the remote side for an outgoing call from FXO port. This item is valid only when Polarity Reversal is enabled. It is calculated by s, with the default value of 60.Communication without NetworkAutomatically routes a call to the proper port according to the configuration in case of network failure or call timeout. The default value is enabled.Communicate without NetworkSets the mode for the communications without network, two options available: Auto Search Idle Channel and Use Current Route Setting, with the default value of Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of TeI>IP route.Two Stages Dialing ModeSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message. This item is valid only when Polarity Reversal is enabled. The default value is enabled.Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
after INVITE Received at IP SideOnce this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message.Maximum Wait Answer TimeThe maximum time to wait the answer of the remote side for an outgoing call from FXO port. This item is valid only when Polarity Reversal is enabled. It is calculated by s, with the default value of 60.Communication without NetworkAutomatically routes a call to the proper port according to the configuration in case of network failure or call timeout. The default value is enabled.Communicate without NetworkSets the mode for the communications without network, two options available: Auto Search Idle Channel and Use Current Route Setting, with the default value of Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of Tel->IP route.Two Stages Dialing ModeSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is enabled.Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
Received at IP Sidethe IP side receives the INVITE message.Maximum Wait Answer TimeThe maximum time to wait the answer of the remote side for an outgoing call from FXO port. This item is valid only when Polarity Reversal is enabled. It is calculated by s, with the default value of 60.Communication without NetworkAutomatically routes a call to the proper port according to the configuration in case of network failure or call timeout. The default value is enabled.Communicate without NetworkSets the mode for the communications without network, two options available: Auto Search Idle Channel and Use Current Route Setting, with the default value of Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of Tel->IP route.Two Stages Dialing ModeSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is enabled.Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
Maximum Wait Answer TimeThe maximum time to wait the answer of the remote side for an outgoing call from FXO port. This item is valid only when Polarity Reversal is enabled. It is calculated by s, with the default value of 60.Communication without NetworkAutomatically routes a call to the proper port according to the configuration in case of network failure or call timeout. The default value is enabled.Communicate without NetworkSets the mode for the communications without network, two options available: Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of TeI->IP route.Two Stages Dialing ModeSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is enabled.Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
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Communicate without Network ModeSearch Idle Channel and Use Current Route Setting, with the default value of Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of Tel->IP route.Two Stages Dialing ModeSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Delay to Send 200 OK to IP SideOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is <i>enabled</i> .Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
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without Network Modesearch an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of Tel->IP route.Two Stages Dialing ModeSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Delay to Send 200 OK to IP SideOnce this feature is enabled, the gateway will delay to send 200 OK message to the IP side. The default value is <i>disabled</i> .Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is <i>enabled</i> .Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
Modemode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of Tel->IP route.Two Stages DialingSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Delay to Send 200Once this feature is enabled, the gateway will delay to send 200 OK message to the IP side. The default value is disabled.Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is enabled.Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
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Two Stages Dialing ModeSets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO port. By default this feature is disabled.Delay to Send 200 OK to IP SideOnce this feature is enabled, the gateway will delay to send 200 OK message to the IP side. The default value is <i>disabled</i> .Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is <i>enabled</i> .Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be avoid the pick-up signal being detected as a flash signal by the PBX. The default
Moderemote end via an FXO port. By default this feature is disabled.Delay to Send 200Once this feature is enabled, the gateway will delay to send 200 OK message to the IP side. The default value is disabled.Ok to IP SideIP side. The default value is disabled.Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is enabled.Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be avoid the pick-up signal being detected as a flash signal by the PBX. The default
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Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is <i>enabled</i> .Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be avoid the pick-up signal being detected as a flash signal by the PBX. The default
Open Session In Advanceport is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is <i>enabled</i> .Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be avoid the pick-up signal being detected as a flash signal by the PBX. The default
Advanceport is making an outgoing call; otherwise, it will reply the 180 message. This item is valid only when Polarity Reversal is enabled. The default value is <i>enabled</i> .Avoid Being Detected as Flash Signal by PBXOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
Avoid BeingOnce this feature is enabled, after hanging up a call, the FXO channel will be compelled to stay idle for a while before making a new call outside, which helps avoid the pick-up signal being detected as a flash signal by the PBX. The default
Avoid BeingDetected as FlashSignal by PBX
Detected as Flash Signal by PBX
<i>Signal by PBX</i> avoid the pick-up signal being detected as a flash signal by the PBX. The default
Signal by PBX
value is <i>disabled</i> .
Sets the work mode for the echo canceller. There are two options: Near-end
Work Mode cancellation and Both near-end and far-end cancellation, with the default value of
Near-end cancellation.
<i>Non-linear</i> Sets whether to enable the mode of non-linear processing. By default, this feature is
Processing enabled.
<i>Fixed Window Size</i> Sets the size of the window for the fixed cancellation.
<i>Moving Window Size</i> Sets the size of the window for the moving cancellation.
Select Area for
Module Parameters Select an area for hardware parameters of the FXO chip.

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 system, do it immediately to apply the changes. Refer to <u>3.9.17 Restart</u> for detailed instructions.



3.5.3 Tone Detector

Tone Detector										
Check	Index	Tone	Туре	The 1st Mid-frequency	The 2nd Mid-frequency	Duration at ON State	Duration at OFF State	Period Count	Energy	Modify
	0	Dial Tone	Continuous Tone	450	0	1500	0	0	8	
	1	Busy Tone	Periodic Tone	450	0	350	350	2	0	2
	2	Ringback Tone	Periodic Tone	450	0	1000	4000	1	0	

Figure 3-30 Tone Parameters Setting Interface

See Figure 3-30 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 3-30 to modify the tone parameter. See Figure 3-31 for the tone parameter modification interface.

Tone Parameters				
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			
Energy:	8			
Save	Close			

Figure 3-31 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 of ON State	The duration of tones at on state. The default setting: Dial Tone is 1500ms, Busy
Duration at ON State	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.
Devied Count	Sets the count of periods as the condition to determine a periodic tone. The default
Period Count	setting: Dial Tone is 0, Busy Tone is 2, Ringback Tone is 1.
	Sets the energy threshold for the tone detector to detect the on-line tone. To
Energy	increase the accuracy, you can adjust the value according to the tone volume on the
	line. Range of value: -18~11, calculated by dB. The default value is 0.

To delete a piece of tone, check the checkbox before the corresponding index in Figure 3-30 and click the '*Delete*' button. *Check All* means to select all available items on the current page; *Uncheck All* means to cancel all selections on the current page; *Inverse* means to uncheck the selected items and check the unselected. To clear all tone at a time, click the *Clear All* button in Figure 3-30.

3.5.4 Tone Generator

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 most 5 different toneunits and 4 different frequencies, but the frequency and	
Call Wait Tone	450/200,0/600,450/200,0/1000	duration of the first toneunit cannot be 0. Frequency being 0 means the toneunit is a piece of silence.	
	Save	Reset	

Figure 3-32 Tone Generator Setting Interface

See Figure 3-32 for the Tone Generator Setting interface. By default, there are four 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 -18 \sim 11, calculated by dB, with the default value of 0.

3.5.5 DTMF

DTMF Detect	tor
Minimum Energy Threshold (dB)	-45
Maximum Threshold of Signal Twist (dB)	3
Input Signal Gain (dB)	0
Voice Path Delay (ms)	20
Minimum Duration at ON (ms)	28
DTMF Display via Channel Status	Enable
ABCD Detection	Enable
DTMF Genera	ator
DTMF Energy (dB)	0
Duration at ON (ms)	100
Duration at OFF (ms)	32
Save	Reset

Figure 3-33 DTMF Detector Configuration Interface

See Figure 3-33 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.
lanat Oissal Osia	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.
Vaine Rath Dalay	Once this feature is enabled, the DTMF in the voice data will be clamped, Range of
Voice Path Delay	value: 0 \sim 20, calculated by ms. The default value is 20.
Minimum Duration	Set the minimum duration at ON for the DTMF signal. Range of value: 10 ${\sim}60,$
at ON	calculated by ms. The default value is 28.
DTMF Display via	Once this feature is enabled, the received/sent DTMF will be displayed upon you
Channels Status	putting the mouse on the icon of channel status. The default value is disabled.



APCD Detection	Once this feature is enabled, the gateway can detect the DTMF digits A, B, C and D	
ABCD Detection	(Case-insensitive). The default value is disabled.	
DTME Enormy	Energy of the DTMF signal sent by the gateway. Range of value: -18~11, calculated	
DTMF Energy	by dB, with the default value of 0.	
Duration at ON	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.	
Dume tier at OFF	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. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to <u>3.9.17 Restart</u> for detailed instructions. Click *Reset* to restore the configurations.

3.5.6 Ringing Scheme

Ringing Sc	heme
Matching Scheme	CallerID Matching
Scheme 1	
CallerID	
Ringing Mode	
Scheme 2	
CallerID	
Ringing Mode	
Scheme 3	
CallerID	
Ringing Mode	
Scheme 4	
CallerID	
Ringing Mode	
Save	Reset

Figure 3-34 Ringing Scheme Configuration Interface

See Figure 3-34 for the Ringing Scheme Configuration interface. The gateway can execute different ringing schemes according to the CallerID or Alert-Info.

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

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



	The gateway will match the CallerID set in this item to that of the incoming call. If
	they are matched, the current ringing scheme will be executed; otherwise, the
CallerID	default ringing scheme (1 sec on and 4 sec off) will work.
	The rule to fill in the CallerID is the same as that of <u>3.5.9 Dialing Rule</u> . Multiple
	CallerIDs are supported; they should be separated by ","
	The gateway will match the alert-info value set in this item to that of the incoming
Alert-Info Value	call. If they are matched, the current ringing scheme will be executed; otherwise,
	the default ringing scheme (1 sec on and 4 sec off) will work
	The ringing scheme can be "1,X,Y" or "2,X,Y,M,N", in which, the number 1 or 2
	denotes one group or two groups; X, M denote the duration at on state while Y, N $$
	denote the duration at off state.
Ringing Scheme	Note: The duration at ON or OFF cannot be greater than 12000ms, the total
	duration at ON and OFF cannot be greater than 16000ms, and N - the last duration
	at OFF cannot be less than 1800ms if the item "Occasion to Send FSK CallerID" is
	set to After the first ring.

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

3.5.7 Fax

Fax Para	ameters
Fax Mode	Disable
Save	Reset

Figure 3-35 Fax Configuration Interface (Disable by default)

See Figure 3-35 for the default fax mode configuration. The table below explains the items shown in the above figure.

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

See Figure 3-36 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 3-36 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. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to <u>3.9.17 Restart</u> for detailed instructions. click **Reset** to restore the configurations. The table below explains the configuration items in Figure 3-36.

ltem	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 3-37.



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

Figure 3-37 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.

3.5.8 Function Key

See Figure 3-38 for the Function Key Configuration interface. Here you can set a cluster of combination keys to query a related number.



Function	Enable	Function Key	Mode
Device Function			
Query LAN		*11*	Default
Query Phone Number	V	*20*	Default
Phone Test		*30*	Default
Set LAN		*61*	Default 💌
Query WEB Port		*70*	Default
Reboot		*#88921532*#	Default
Query Missed Call Number		*71*	Default 🗨
Service Available			
Blind Transfer	V	*010*	Default 💌
Call Forward Unconditional Activate		*030*	Default 👻
Call Forward Unconditional Deactivate		*031*	Default 👻
Call Forward Busy Activate		*040*	Default 💌
Call Forward Busy Deactivate		*041*	Default 👻
Call Forward No Reply Activate		*050*	Default 👻
Call Forward No Reply Deactivate		*051*	Default 👻
Do Not Disturb Activate		*060*	Default 👻
Do Not Disturb Deactivate		*061*	Default 💌
Register		*020*	Default
Unregister		*021*	Default 💌
Query Register Status		*022*	Default 💌
Conference		*070*	Default

Figure 3-38 Function Key Configuration Interface

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

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

When the **Blind Transfer** feature is enabled, set a corresponding function key in the box behind. After you transfer a call by rapidly clapping on the hook switch, dial the set function key for **Blind Transfer** and then the called party number. After that, hang up the call once hearing the howler tone to let the subsequent call procedure go out of your control.

3.5.9 Dialing Rule

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



Check	Index	Dialing Rule	Description	Modify
	81	400xxxxxxx	default	
	82	40[1-9]xxxxx	default	
	83	4[1-9]xxxxxx	default	
	84	800xxxxxxx	default	
	85	80[1-9]xxxxx	default	
	86	8[1-9]xxxxxx	default	
	87	[2-3,5-7]x000000	default	
	88	1[3-5,7-8])000000000	default	
	89	100xx	default	
	90	95хох	default	
	91	123xx	default	
	92	111xx	default	
	93	11[0,2-9]	default	
	94	120	default	
	95	0[3-9]xxxxxxxxxxxx	default	
	96	02хоососох	default	
	97	010xxxxxxxx	default	
	98	01[3-5,7-8]x00000000	default	
	99	24	default	

Figure 3-39 Dialing Rule Configuration Interface (Standard)

See Figure 3-39 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 3-40 for the dialing rule adding interface.

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

Figure 3-40 Add New Dialing Rule

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

Item	Description		
Index	The unique index of each dialing rule, which denotes its priority. A dialing rule with a		
Index	smaller index value has a higher priority and will be checked earlier while matching.		
Description	Remarks for the dialing rule. It can be any information, but can not be left empty.		
Dialing Rule	Up to 100 dialing rules can be configured in the gateway, and the maximum length of		



-	rule and regard your dia	aling as finished upon receiving '#' or di	
imeout.			
Character	Description		
"0"~"9"	Digits 0~9.		
"A"~"D"	Letters A~D.		
"X"	1	string of 'x's represents several rando e, 'xxx' denotes 3 random numbers.	
" " •	'.' indicates a randor after it.	m amount (including zero) of characte	
"[]"	can be digits '0~9',	ne range for a number. Values within it on punctuations '-' and ','. For exampl / one of the numbers 1, 2, 3, 6, 8.	
"_"	'-' is used only in '[number between thes	l' between two numbers to indicates a e two numbers.	
((9) 9	',' is used to separate alternatives.	numbers or number ranges, representir	
"*"	Only represents symb	ool "*".	
	Only set it at the beginning of the string, representing symbol		
"#"	Only set it at the beg "#".	ginning of the string, representing symb	
There are 19 o below for detail	"#". dialing rules already co led information.	nfigured on the gateway for easy use.	
There are 19 o below for detail Priority	"#". dialing rules already co	nfigured on the gateway for easy use. Description	
There are 19 o below for detail	"#". dialing rules already co led information.	nfigured on the gateway for easy use. Description Any number in any length.	
There are 19 o below for detail Priority	"#". dialing rules already co led information.	nfigured on the gateway for easy use. Description Any number in any length.	
There are 19 o below for detail Priority 99	"#". dialing rules already co led information. Dialing Rule	nfigured on the gateway for easy use. Description Any number in any length. Any 12-digit number starting with 0	
There are 19 o below for detail Priority 99 98	"#". dialing rules already co led information. Dialing Rule 01[3-5,7-8]xxxxxxxxx.	Description Description Any number in any length. Any 12-digit number starting with 0 014, 015, 017 or 018	
There are 19 o below for detail Priority 99 98 98 97	"#". dialing rules already co led information. Dialing Rule 01[3-5,7-8]xxxxxxxxx 010xxxxxxx	Description Description Any number in any length. Any 12-digit number starting with 0 014, 015, 017 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02	
There are 19 o below for detail Priority 99 98 98 97 96	"#". dialing rules already co led information. Dialing Rule 01[3-5,7-8]xxxxxxxxx 010xxxxxxxx 02xxxxxxxx	Description Any number in any length. Any 12-digit number starting with 0 014, 015, 017 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03,	
There are 19 o pelow for detail Priority 99 98 97 96 95	"#". dialing rules already co led information. Dialing Rule 01[3-5,7-8]xxxxxxxx 010xxxxxxxx 010xxxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxxx	Description Any number in any length. Any 12-digit number starting with 0 014, 015, 017 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 05, 06, 07, 08 or 09 Number 120.	
There are 19 o pelow for detail Priority 99 98 97 96 95 95 94	 "#". dialing rules already colled information. Dialing Rule 01[3-5,7-8]xxxxxxxx 010xxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxx 120 	Description Any number in any length. Any 12-digit number starting with 0 014, 015, 017 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 05, 06, 07, 08 or 09 Number 120. Number 110, 112, 113, 114, 115, 116, 1	
There are 19 o pelow for detail Priority 99 98 97 96 95 95 94 93	"#". dialing rules already co led information. Dialing Rule 01[3-5,7-8]xxxxxxxx 010xxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxx 120 11[0,2-9]	Description Any number in any length. Any 12-digit number starting with 0 014, 015, 017 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 02 Any 12-digit number starting with 03, 05, 06, 07, 08 or 09 Number 120. Number 110, 112, 113, 114, 115, 116, 1 118 or 119	
There are 19 o pelow for detail Priority 99 98 97 96 95 94 93 92	 "#". dialing rules already colled information. Dialing Rule 01[3-5,7-8]xxxxxxxx 010xxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxx 120 11[0,2-9] 111xx 	Description Any number in any length. Any number in any length. Any 12-digit number starting with 0 014, 015, 017 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 0 05, 06, 07, 08 or 09 Number 120. Number 110, 112, 113, 114, 115, 116, 1 118 or 119 Any 5-digit number starting with 111	
There are 19 o pelow for detail Priority 99 98 97 96 95 94 93 93 92 91 90	 "#". dialing rules already colled information. Dialing Rule 01[3-5,7-8]xxxxxxxx 010xxxxxxx 010xxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxx 120 11[0,2-9] 111xx 123xx 95xxx 	DescriptionAny number in any length.Any 12-digit number starting with 0014, 015, 017 or 018Any 11-digit number starting with 010Any 11-digit number starting with 02Any 12-digit number starting with 03, 05, 06, 07, 08 or 09Number 120.Number 110, 112, 113, 114, 115, 116, 1118 or 119Any 5-digit number starting with 123Any 5-digit number starting with 123	
There are 19 o pelow for detail Priority 99 98 97 96 95 94 93 93 92 91	 "#". dialing rules already colled information. Dialing Rule 01[3-5,7-8]xxxxxxxx 010xxxxxxx 02xxxxxxxx 0[3-9]xxxxxxxx 120 11[0,2-9] 111xx 123xx 	Description Any number in any length. Any 12-digit number starting with 0 014, 015, 017 or 018 Any 11-digit number starting with 010 Any 11-digit number starting with 02 Any 12-digit number starting with 03, 0 05, 06, 07, 08 or 09 Number 120. Number 110, 112, 113, 114, 115, 116, 1 118 or 119 Any 5-digit number starting with 111 Any 5-digit number starting with 123	



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 3-39 to modify the dialing rules. See Figure 3-41 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 3-41 Modify Dialing Rule

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

See Figure 3-42 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 string.	
Don't forget to save the configuration after your modification! 400xxxxxxx default	~
40[1-9]xxxx default	
4[1-9]xxxxx default	
800xxxxxx default	
80[1-9]xxxx default	
8[1-9]xxxxx 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	×
20 Items Total	
Save	

Figure 3-42 Dialing Rule Configuration Interface (Character)

3.5.10 Dialing Timeout

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

Figure 3-43 Dialing Timeout Info Interface

See Figure 3-43 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
Inter Divit Times ut	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.
Description	More information about the configuration item Inter Digit Timeout, such as the
Description	reason for adopting the current value.

Click *Modify* in Figure 3-43 to modify the dialing timeout info. See Figure 3-44 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 T	imeout
Description:	example
Inter Digit Timeout (s):	6
Save	Close

Figure 3-44 Modify Dialing Timeout Info

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

3.5.11 Cue Tone

	Upload	
Upload a file of cue tone	File of cue tone for IVR	Browse, Upload
	e a wav file with 8000Hz sampling rate, 16)-bit mono, A-law formatted, and less than

Figure 3-45 Cue Tone Interface

See Figure 3-45 for the Cue Tone interface. The table below explains the items shown in the above figure.

Item	Description
Upload a file of cue	Uploads a user-defined cue tone file to the gateway.
tone	opioads a user-defined cue tone file to the gateway.

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



3.5.12 Color Ring

Operation Info	*		
Quick Config	*		
B VolP	*		No available
Advanced	*		Uplo
FXS			
FXO			
Tone Detector			
Tone Generator			
DTMF Detector			
Ringing Scheme			
Fax			
Function Key			
Dialing Rule			
Dialing Timeout			
Cue Tone			
Color Ring			

Figure 3-46 Coloring Ring Interface

By default, there is no available color ring on the gateway. See Figure 3-46. Click **Upload** to upload a new color ring manually. Follow Figure 3-47 to upload the required color ring file to the gateway.

scription	default		
lor Ring			(Browse)
Note: The file should be a 200KB in size.	a wav file with 8000Hz samı	pling rate, 16-bit mono, A-I	aw formatted, and less thar

Figure 3-47 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 3-48 for the Color Ring Management interface after the upload.



		Color Ring Manage		
Check	Index	Color Ring	Port	Modify
	1	ringtone1		(2)
Check All E Uncheck All E	Inverse E Delete E I	Clear All		Upload

Figure 3-48 Color Ring Management Interface

Click *Modify* in Figure 3-48 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
1
ringtone1
Save Cancel

Figure 3-49 Color Ring Modification Interface

To delete a color ring, check the checkbox before the corresponding index in Figure 3-48 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 3-49.

3.5.13 QoS

 QoS		
QoS	🗹 Enable	
Media Premium QoS	46	
Control Premium QoS	26	

Figure 3-50 Differentiated Services Setting Interface

See Figure 3-50 for the Differentiated Services setting interface. Using this technology, the gateway can meet various application requirements under a limited bandwidth and ensure neither delay nor discard for important services so as to improve its quality of services.

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

Item	Description
QoS	Sets whether to enable the OoS differentiated services. By default, it is disabled.



Media Premium QoS	Sets the priority of the media premium for QoS. A media premium QoS with a bigger
	value has a higher priority. The value range is 0~63, with the default value of 46.
	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.

3.5.14 Action URL

	Channel State Report Settings
Channel State	Report States to URL
Channel Pick up	
Channel Hang up	
	Save

Figure 3-51 Channel State Report Settings Interface

See Figure 3-51 for the Action URL interface, which is used to designate the server patch to report the on-hook or off-hook state of the FXS channel. You are allowed to designate two different server paths. After setting, the state will be reported to the designated server once any of the FXS channel hangs up or picks up a call. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations.

3.6 Port Settings

Port Settings includes five parts: *FXS, FXO, FXO Port Timer, Port Group* and *Advanced FXO Settings*. See Figure 3-52.

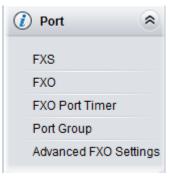


Figure 3-52 Port Settings

3.6.1 FXS

									1765	Settings										
Port	Type	SIP Account	Display Name	Authentication Username	Auto Dial Num	Forbid Outgoing Call	DND	Forward	FWD Type	FWO Number	CO	Call Walting	Reg Status	Echo Canceller	Color Ring	Color Ring Index	Server Index	Input Gam	Output Gain	Mod
1	FX3	8001	-	-	-	Disable	Disable	Disatle		-	Enable	Disable	Unregistered	Enable	Disable		-	D	0	2
2	FXS	8002	<u> </u>	-	1.00	Disable	Disable	Disable	1.000	-	Enable	Disable	Unregistered	Enable	Disable		-	0	0	12
з	17:5	8003			8 m.	Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable	-	22	0	0	02
4	FX8	1004	1 Q		S = .	Disable	Disable	Disable	-		Enable	Disable	Unregistered	Enable	Disable	-	2	0	0	12
									-											

Figure 3-53 FXS Settings Interface



See Figure 3-53 for the FXS settings interface. The list in the above figure shows the feature and properties of each FXS port. Click *Modify* in Figure 3-53 to modify the properties of the corresponding port. See Figure 3-54 for the FXS modification interface.

Port1TypeFXSRegister PortYesSIP Account8001Display Name8001Display Name1Password1Authentication Username1Display Name Preferred1Server Index1:201.123.115.233 •Auto Dial Number0Wait Time before Auto Dial (s)0Input Gain (dB)0Output Gain (dB)0Color CancellerØEnableForbid Outgoing CallEnableColor CancellerØEnableColor Not Disturb)EnableCall WaitingEnableForward TypeUnconditionalForward NumberIColor RingØEnableColor Ring Index1TalkbackØEnableBound NumberIEnableIDisplay Name PreferredIColor Ring IndexITalkbackØEnableServer IndexIServer Inde	Type Register Port SIP Account Display Name Password Authentication Username Display Name Preferred	FXS Yes 8001 Bool Enable 1:201.123.115.233 0
Type FXS Register Port Yes SIP Account 8001 Display Name	Type Register Port SIP Account Display Name Password Authentication Username Display Name Preferred	FXS Yes 8001 Bool Enable 1:201.123.115.233 0
SIP Account 8001 Display Name	SIP Account Display Name Password Authentication Username Display Name Preferred	8001 Enable 1:201.123.115.233 • 0
SIP Account 8001 Display Name	SIP Account Display Name Password Authentication Username Display Name Preferred	8001 Enable 1:201.123.115.233 • 0
Display Name Password Authentication Username Display Name Preferred Enable Server Index 1:201.123.115.233 Auto Dial Number Wait Time before Auto Dial (s) Input Gain (dB) Output Gain (dB) Output Gain (dB) Echo Canceller Forbid Outgoing Call Enable CiD Call Waiting Enable CiD ND (Do Not Disturb) Enable Call Waiting Forward Type Unconditional Forward Type Forward Number Color Ring Color Ring Color Ring Index Advanced Configuration Talkback Bound Number	Display Name Password Authentication Username Display Name Preferred	Enable 1:201.123.115.233 0 0
Password	Password Authentication Username Display Name Preferred	1:201.123.115.233 • 0
Authentication Username Enable Display Name Preferred 1:201.123.115.233 • Server Index 1:201.123.115.233 • Auto Dial Number 0 Wait Time before Auto Dial (s) 0 Input Gain (dB) 0 Output Gain (dB) 0 Echo Canceller ØEnable Forbid Outgoing Call Enable CiD ØEnable DND (Do Not Disturb) Enable Call Waiting ØEnable Forward Type Unconditional • Forward Number • Color Ring ØEnable Color Ring Index 1 Advanced Configuration ØEnable Bound Number •	Authentication Username Display Name Preferred	1:201.123.115.233 • 0
Display Name Preferred Enable Server Index 1:201.123.115.233 • Auto Dial Number Wait Time before Auto Dial (s) 0 Input Gain (dB) 0 Output Gain (dB) 0 Output Gain (dB) 0 Echo Canceller ØEnable Forbid Outgoing Call Enable CID ØEnable Call Waiting Enable Call Waiting Enable Call Forward ØEnable Forward Type Unconditional • Forward Type Unconditional • Forward Number 1 Color Ring Menable Color Ring Index 1 Advanced Configuration ØEnable Forable Bound Number	Display Name Preferred	1:201.123.115.233 • 0
Server Index 1:201.123.115.233 Auto Dial Number Wait Time before Auto Dial (s) Unput Gain (dB) Output Gain (dB) Forward Type Forward Number Color Ring Matenced Configuration Yenable Bound Number		1:201.123.115.233 • 0
Auto Dial Number Wait Time before Auto Dial (s) Input Gain (dB) Output Gain (dB) Output Gain (dB) Echo Canceller Forbid Outgoing Call ClD Call Waiting Call Waiting Call Waiting Call Forward Call Forward Call Forward Color Ring Color Ring Colo		0
Wait Time before Auto Dial (s)0Input Gain (dB)0Output Gain (dB)0Echo CancellerØEnableForbid Outgoing CallEnableCIDØEnableCall WaitingEnableDND (Do Not Disturb)EnableCall ForwardØEnableForward TypeUnconditionalForward Number1Color RingØEnableColor Ring Index1Advanced ConfigurationØEnableBound NumberI		0
Wait Time before Auto Dial (s)OInput Gain (dB)OOutput Gain (dB)OOutput Gain (dB)OEcho CancellerInableForbid Outgoing CallEnableCIDInableCall WaitingEnableDND (Do Not Disturb)EnableCall ForwardInableForward TypeInconditionalForward NumberInconditionalColor RingInableColor Ring Index1Advanced ConfigurationInableTalkbackInableBound NumberInable	Auto Dial Number	0
Input Gain (dB) 0 Output Gain (dB) 0 Echo Canceller Instantion I		0
Output Gain (dB)Image: Constant of the state		
Echo CancellerImage: CancellerImage: CancellerForbid Outgoing CallEnableCIDImage: CancellerCall WaitingEnableDND (Do Not Disturb)EnableCall ForwardImage: CancellerForward TypeUnconditionalForward NumberImage: CancellerColor RingImage: CancellerColor Ring Index1Advanced ConfigurationImage: CancellerTalkbackImage: CancellerBound NumberImage: Canceller	Input Gain (dB)	0
Forbid Outgoing Call Enable CID Image: Call Waiting Call Waiting Enable DND (Do Not Disturb) Enable Call Forward Image: Call Forward Type Forward Type Unconditional Forward Number Image: Color Ring Color Ring Index 1 Advanced Configuration Image: Call Forward Rumber Talkback Image: Call Forward Rumber	Output Gain (dB)	
CID CID Call Waiting Enable Call Waiting Enable DND (Do Not Disturb) Enable Call Forward Vienable Call Forward Vienable Call Forward Vienable Color Ring Forward Number Color Ring Index 1 Color Ring Index 1 Advanced Configuration I Enable Bound Number I Enable	Echo Canceller	✓Enable
Call Waiting Enable DND (Do Not Disturb) Enable Call Forward Orge Venable Forward Type Dunconditional I Forward Number Venable Color Ring Index 1 Advanced Configuration Venable Talkback Venable Bound Number	Forbid Outgoing Call	Enable
DND (Do Not Disturb) Enable Call Forward Image: Color Ring Forward Number Image: Color Ring Index Advanced Configuration Image: Color Ring Talkback Image: Color Ring Bound Number Image: Color Ring	CID	Enable
Call Forward Type Forward Type Forward Number Color Ring Color Ring Index Advanced Configuration Talkback Bound Number	Call Waiting	Enable
Forward Type Forward Number Color Ring Color Ring Index Advanced Configuration Talkback Bound Number	DND (Do Not Disturb)	Enable
Forward Number Color Ring Color Ring Index Advanced Configuration Talkback Bound Number		☑Enable
Forward Number Color Ring Color Ring Index Advanced Configuration Talkback Bound Number	Forward Type	Unconditional
Color Ring Index 1 Advanced Configuration Talkback Bound Number		
Color Ring Index 1 Advanced Configuration Talkback Bound Number	Color Ring	✓Enable
Talkback Bound Number		1
Talkback Bound Number	Advanced Configuration	✓Enable
Bound Number		
		·
Made Diel Musehad and die offent and diele and diele and diele and diele a fisient Time haften Arts Diel.		
Note: 'Auto Dial Number' goes into effect only if no dialing occurs during 'Wait Time before Auto Dial'.	Note: 'Auto Dial Number' goes into effect only if no d	lialing occurs during Wait Time before Auto Dial".
	Modify	Cancel

Figure 3-54 FXS Modification

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

	ltem	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.
	When this item is set to <i>No</i> , the item <i>Reg Status</i> on the FXS settings interface
Register Port	(Figure 3-53) 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.
	Set the content of the displayname field of the SIP message. If it doesn't set with
Display Name	any value, the displayname field will by default display the content of callerid.
	Registration password of the port. To register a port to the SIP server, both items
Password	SIP Account and Password must be filled in.
	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 or Multi-Registrar Server is
ocername	enabled.
	In case this feature is enabled and the port group or the whole gateway is
	registered, if the display name set by the port are different from that set by the port
	group, the displayname in the sent SIP message will be the one set by the port. In
Display Name	case this feature is disabled, if the port group is registered, the displayname in the
Preferred	sent SIP message will be the display name set by the port group; if the whole
	gateway is registered, the displayname in the sent SIP message will be the
	displayname of the gateway.
Server Index	The index of the SIP server which will be quoted by the current FXS port.
Auto Dial Number,	
Wait Time before	The FXS port will dial the Auto Dial Number if there is no dialing operation after
Auto Dial	pickup within a designated time period (i.e. <i>Wait Time before Auto Dial</i>).
Input Gain, Output	Adjusts the gain of the voice input to/ output from the FXS port. Range of value:
Gain	-24~12, calculated by dB, with the default value of 0.
	The echo cancellation feature for a call conversation over the FXS channel. By
Echo Canceller	default, this feature is enabled and the effect can reach 128ms.
Forbid Outgoing	If this feature is enabled, the FXS port will be forbidden to call out. The default
Call	setting is <i>disabled</i> .
	CallerID. If this feature is enabled, the FXS port will send the CallerID of the
010	incoming IP call together with the ringing tone to the corresponding station. The
CID	default setting is enabled. CallerID displays digits only and will filter out any other
	characters if exist.
	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
Call Waiting	



DND	Do Not Disturb.	If this feature is enabled, the FXS port will reply the 403 message to			
	reject all incomir	ng calls. The default setting is <i>disabled</i> .			
	The automatic o	all forward feature for the FXS port. Once this feature is enabled,			
Call Forward	the FXS port w	ill forward incoming IP calls according to FWD Type. Note: To			
Call Fol ward	enable this feature, do not put the FXS port into a port group with other ports. The				
	default setting is	disabled.			
	Forward condition	ons for the FXS port to forward incoming IP calls. The optional			
	values are:				
	Option	Description			
		The FXS port will forward all incoming IP calls to the preset			
	Unconditional	FWD Num immediately when it receives them.			
		The FXS port will forward incoming IP calls to the preset FWD			
FWD Type	Busy	<i>Num</i> if it is busy upon receiving them.			
		The FXS port will forward incoming IP calls to the preset FWD			
	* * *	<i>Num</i> if the corresponding station does not answer them in a			
	No Reply	designated time period (i.e. <i>Time for No Reply Forward</i>). Only			
	when this forward condition is selected does the configuration				
		item <i>Time for No Reply Forward</i> become valid.			
	This item is valid only when <i>Call Forward</i> is set to <i>Enable</i> .				
		which the incoming IP call is forwarded. If the Call Forward feature			
FWD Num		tem can not be left empty.			
		enable the color ring feature or not, with the default setting of being			
Color Ring	disabled.				
3	Note: Only when	n there are available color rings will this item appear.			
Color Ring Index		color ring which will be quoted by the current FXS port.			
		e enabled and a number bound, the port can talkback to its bound			
		, they can start a call with each other as soon as picking up the			
Talkback		ult setting is <i>disabled</i> .			
	-	are is only used in the case of channel registration.			
Bound Number		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 3-55 below for the FXS batch modification interface. The configuration items on this interface are the same as those on the FXS modification interface (Figure 3-54).



Starting Port	1
Ending Port	4
Register Port	Yes
Starting SIP Account	
Starting Display Name	
Starting Authentication Password	
Starting Authentication Username	
Display Name Preferred	Enable
Server Index	1.201.123.115.233 💌
SIP Account Batch Rule	Increase
SIP Account Batch Step Size	1
Display Name Batch Rule	Increase
Display Name Batch Step Size	1
Authentication Password Batch Rule	Increase
Authentication Password Batch Step Size	1
Authentication Username Batch Rule	Increase
Authentication Username Batch Step Size	1
Auto Dial Number	⊡Enable
Auto Dial Number	
Wait Time before Auto Dial (s)	0
Input Gain (dB)	0
Output Gain (dB)	0
CID	Enable
Echo Canceller	✓Enable
Forbid Outgoing Call	Enable
Call Waiting	Enable
DND (Do Not Disturb)	Enable
Call Forward	I Enable
Forward Type	Unconditional
Forward Number	
Color Ring	Enable
Color Ring Index	1
te: 'Auto Dial Number' goes into effect only if no	dialing occurs during 'Wait Time before Auto Dial'.

Figure 3-55 FXS Batch Modification

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



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

Port	Туре	SIP Account	Display Name	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	F
5	FXO	8005	8005	Two Stages Dialing for Incoming Call	()	Disable	Enable	Unregistered	Enable	
6	FXO	8006	8006	Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	
7	FXO	8007	8007	Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	
8	FXO	8008	8008	Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	
				III						•

Figure 3-56 FXO Settings Interface

See Figure 3-56 for the FXO Settings interface. The list in the above figure shows the feature and properties of each FXO port. Click *Modify* in Figure 3-56 to modify the properties of the corresponding port. See Figure 3-57 for the FXO Modification interface.



Port	5
Туре	FXO
Register Port	Yes 💌
SIP Account	8005
Display Name	
Password	
Authentication Username	
Display Name Preferred	Enable
Server Index	1:201.123.115.233 💌
Connection Method	Two Stages Dialing 🚽
Input Gain at Offhook(dB)	0
Output Gain at Offhook(dB)	0
Input Gain at Onhook(dB)	0
Output Gain at Onhook(dB)	0
Echo Canceller	Enable
Forbid Outgoing Call	Enable
Caller ID Detection	Enable
Polarity Reversal Detection	Enable

Figure 3-57 FXO Modification

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

Item	Description
Port	Serial number of the FXO port on the device.
Туре	Type of the port on the device (FXO). This item is not configurable.
	Sets whether to register the port to the SIP server.
De vie (en De ví	When this item is set to No, the item Reg Status on the FXO settings interface
Register Port	(Figure 3-56) shows Unregistered; when this item is set to Yes, the item Reg Status
	shows Failed or Registered.
	Registration account of an FXO port. The default SIP account is 80XX among which
SIP Account	XX represents the corresponding port number. For example, the default SIP
	account corresponding to Port 1 is 8001, and that corresponding to Port 32 is 8032.
Dian Inc. Manua	Set the content of the displayname field of the SIP message. If it doesn't set with
Display Name	any value, the displayname field will by default display the content of callerid.
Deserved	Registration password of the port. To register a port to the SIP server, both items
Password	SIP Account and Password must be filled in.



Authentication Username Display Name	Authentication username of a port, used to register the port to the SIP server when IMS network is enabled. Note: This item appears only when IMS Network or Multi-Registrar Server is enabled. In case this feature is enabled and the port group or the whole gateway is registered, if the display names set by the port are different from that set by the port group, the displayname in the sent SIP message will be the one set by the port. In case this feature is disabled, if the port group is registered, the displayname in the				
Preferred	sent SIP message will be the display name set by the port group; if the whole gateway is registered, the displayname in the sent SIP message will be the displayname of the gateway.				
Server Index	The index of the SIP server which will be quoted by the current FXO port.				
Connection Method	FXO connection methods include:OptionDescriptionStaticBind the number which corresponds to an FXS port to an FXO port. The number will be listed in the Bound Number column. This helps to achieve the corresponding binding between an FXO port and an FXS port (two-way).Two StagesUnder this mode, an incoming call from an FXO port will go into the IVR system. Then IVR will play a speech prompt "Please dial the extension number". If you fail to input the correct target station number before IVR finishes the third repeat of the prompt, the FXO will hang up the call automatically; otherwise, the corresponding station will ring.Note:Both items Connection Method and Bound Number will be hidden if the SIP Station feature is enabled on the SIP Settings interface.				
Input Gain at Offhook/Onhook, Output Gain at Offhook/Onhook	Adjusts the gain of the voice input to/ output from the FXO port when it is offhook or onhook. Range of value: -24~12, calculated by dB, with the default value of 0.				
Echo Canceller	The echo cancellation feature for a call conversation over the FXO channel. By default, this feature is enabled and the effect can reach 128ms.				
Forbid Outgoing	If this feature is enabled, the FXO port will be forbidden to call out. The default				
Call	setting is <i>disabled</i> .				
Caller ID Detection	If this feature is enabled, the FXO port will detect the caller IDs from the incoming calls. The default setting is <i>enabled</i> .				
Polarity Reversal Detection	Once this feature is enabled, only when the FXO port detects the polarity reversal signal will the corresponding channel go into the talking state. The default setting is <i>disabled</i> . Note: This feature and the <i>Two Stages Dialing</i> feature cannot be enabled at the same time.				

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

Or you can click **Batch** to modify several pieces of FXO settings at the same time. See Figure



3-58 below for the FXO Batch Modification interface. The configuration items on this interface are the same as those on the FXO Modification interface (Figure 3-57).

 FXO-Batch Mo	dify
Starting Port	5
Ending Port	
Ending For	8
Register Port	Yes
Starting SIP Account	
Starting Display Name	
Starting Authentication Password	
Starting Authentication Username	
Display Name Preferred	Enable
Server Index	1:201.123.115.233 💌
SIP Account Batch Rule	Increase 🗸
SIP Account Batch Step Size	1
Display Name Batch Rule	Increase 💌
Display Name Batch Step Size	1
Authentication Password Batch Rule	Increase 💌
Authentication Password Batch Step Size	1
Authentication Username Batch Rule	Increase
Authentication Username Batch Step Size	1
Connection Method	Two Stages Dialing 🖵
Input Gain at Offhook(dB)	0
Output Gain at Offhook(dB)	0
Input Gain at Onhook(dB)	0
Output Gain at Onhook(dB)	0
Echo Canceller	Enable
Forbid Outgoing Call	Enable
Caller ID Detection	Enable
Polarity Reversal Detection	Enable

Figure 3-58 FXO Batch Modification

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

Item	Description
Starting Port	The starting serial number of the FXO port on the device in the batch setting.
Ending Port	The ending serial number of the FXO port on the device in the batch setting.
Starting SIP Account	The starting SIP account in the batch setting.
Starting Display Name	The starting displayname in the batch setting.

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Starting Authentication Password	The starting authentication password in the batch setting.
Starting Authentication Username	The starting authentication username in the batch setting.
SIP Account Batch Rule	The rule for batch setting the SIP account, including <i>Increase</i> and <i>Decrease</i> two 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 *Save* to save the settings into the gateway, or click *Cancel* to cancel the settings.

3.6.3 FXO Port Timer

Check	Port	Туре	Unit	Max Time(Single Call)	Max Time(Total Calls)	Used Call Time	Clear Call Time	Modif
	5	FXO	60s	Unlimited	Unlimited		_	
	6	FXO	60s	Unlimited	Unlimited	2 <u>000-0</u> 0		
	7	FXO	60s	Unlimited	Unlimited			
F	8	FXO	60s	Unlimited	Unlimited			12

Figure 3-59 FXO Port Timer Interface

See Figure 3-59 for the FXO Port Timer interface, which displays such information as the max call time limit for a single call, the max call time limit for the total calls on each FXO port, as well as the timer clear cycle. Click Modify for each port in Figure 3-59 to modify the timer settings. See Figure 3-60.



Timii	ng
Port	5 💌
Unit	60s 💌
Time Limit on a Single Call Max Call Time	ODisable OEnable
Time Limit on Total Calls Max Call Time	ODisable OEnable
Timing Cycle Clear Set Spent Call Time	Month ▼ 1st ▼ 00 ▼ 00 ▼
SIP Code Reply	486
Apply to Other Ports	 ● Port ● Port Group ✓ 05 □ 06 □ 07 □ 08
Note: The Port Timer feature will g	et valid only when the Polarity Reversal service is enabled.
Modify	Return

Figure 3-60 FXO Port Timing Setting Interface

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

Item	Description
Port	Serial number of the FXO port on the device.
	Sets the timing unit for the call. The actual call time will be calculated as the integral
Unit	multiple of the setting time. Take an example: supposed the setting time is 30s and
	the actual call time is 72s, thus, the gateway will consider the call time as 90s.
Time Limit on a Single Call	Sets whether to enable the time limit on a single call.
Max Call Time	Sets the maximum time length of a call.
Time Limit on Total Calls	Sets whether to enable the time limit on all calls at the port.
Timing Cycle	Sets the time count cycle for the port.
Clear	Sets the time node to clear the time count.
Set Spent Call Time	Sets the spent call time length of the port.
OID On the Daraha	The FXO port cannot make outgoing call once the spent call time reaches to the
SIP Code Reply	total time limit. And the gateway will reply the designated SIP code to the IP side.
Apply to Other Ports	Sets whether to apply above settings to other ports or port groups.

Click *Modify* to save the settings into the gateway, click *Return* to cancel the settings.



3.6.4 Port Group

Check	Index	Description	SIP Account	Display Name	Authentication Username	Ports	Port Select Mode	Rule for Ringing by Turns	Timeout for Ringing by Tums (s)	Preemptive Answer Keyboard Shortcut	Authentication Mode	Register Status	Server Index	Modify
12	1	default	-	192		1	increase		++	7	Do Not Register	Unregistered	-	0
-														

Figure 3-61 Port Group Settings Interface

See Figure 3-61 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 3-62 for the port group adding interface. Note that a port which has been occupied by one port group cannot be chosen by others.

Index	2
Description	default
Register Port Group	YES
SIP Account	
Display Name	
Password	
Authentication Username	
Server Index	1:201.123.115.233
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 Reused by Multiple Groups	
Port	Port 1(FXS) Port 2(FXS) Port 3(FXS) Port 4(FXS) Port 5(FXO) Port 6(FXO) Port 7(FXO) Port 8(FXC)
	Check All Inverse Check All FXO Ports Check All FXS Ports

Figure 3-62 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 in	When the port group initiates a call to SIP, this item corresponds to the username of			
SIP Account	SIP.				
Diamina Mana	Set the content of the	Set the content of the displayname field of the SIP message. If it doesn't set with			
Display Name	any value, the displayr	name field will by default display the content of callerid.			
	Registration password	of the port group. To register the port group to the SIP server			
Password	both configuration item	is SIP Account and Password should be filled in.			
	Authentication usernar	ne of a port, used to register the port to the SIP server wher			
Authentication	IMS network is enable	IMS network is enabled.			
Username	Note: This item appears only when IMS Network or Multi-Registrar Server is				
	enabled.				
Server Index	The index of the sip se	The index of the sip server which will be quoted by the current FXS port.			
	Sets the way for SIP to make outgoing calls (Tel \rightarrow IP) on the gateway.				
	Option	Description			
	Do Not Register (default)	SIP initiates a call in a point-to-point mode.			
Authomáis stis a		SIP initiates a call with the registered SIP account and			
Authentication Mode	Register Gateway	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.			
		SIP initiates a call with the registered SIP account and			
	Register Port	password of the port.			
	Registration status of t	the port group. When Register Port Group is set to No, the			
Register Status	value of this item is L	Inregistered; when Register Port Group is set to Yes, the			
	value of this item may be Failed or Registered.				



	When the port group r	receives a call, it will choose a port based on the select mode		
	set by this configuration item to ring or to connect. The optional values and their			
	corresponding meanings are described in the table below.			
	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		
		number, starting from the maximum. If no match is found,		
	Decrease	search repeatedly until finding a port which is allowed to		
		enter the call waiting state.		
		Provided Port N is the available port found last time.		
		Search for an idle port in the ascending order of the port		
	Cyclic Increase	number, starting from Port N+1. If no match is found,		
Port Select Mode		search repeatedly until finding a port which is allowed to		
		enter the call waiting state.		
		Provided Port N is the available port found last time.		
		Search for an idle port in the descending order of the port		
	Cyclic Decrease	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.		
		Ring the ports in this port group according to the Rule for		
		Ringing by Turns which can be user-defined. Refer to the		
		format of the rule in Figure 3-62. By default, the ringing		
	Ringing by Turns	will be carried out in the ascending order of the port		
		number. Timeout for Ringing by Turns is used to set the		
		overtime for ringing. Range of value: 15~60, calculated by		
		s, with the default value of 20.		
	When a channel in a	port group is ringing, another channel in the same port group		
Preemptive Answer	can press the keyboar	rd shortcut set by this item to transfer the call from the ringing		
Keyboard Shortcut	channel to the current channel.			
Reyboard Shoricul	Note: This item will be	come invalid if the gateway works under the port select mode		
	Group Ringing or Ringing by Turns.			
Port Reused by	Once this feature is an	nabled, a port can be added to different port groups.		
Multiple Groups		asion, a port our de added to unreferit port groups.		
	The ports in the port g	roup. If the checkbox before a port is grey, it indicates that the		
	port is not available	or has been occupied. Once the feature "Port Reused by		
Port	Multiple Groups" is er	nabled, a port which has been occupied is still available for		
	other port groups. All selected ports for a port group will be displayed in the Ports			
	column in Figure 3-61. 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 FXO Ports** means to select all available FXO ports on the current page; **Check All FXS Ports** means to select all available FXS ports on the current page.

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

	Port Group-Modify
Index	1
Description	default
Description	Uelauit
Register Port	Yes
SIP Account	
Display Name	
Password	
Authentication Username	-1
Server Index	1:201.123.115.233
Authentication Mode	Do Not Register
Port Select Mode	Increase
Preemptive Answer Keyboard Shortcut	
Port Reused by Multiple Groups	
Port	Port 1(FXS) Port 2(FXS) Port 3(FXS) Port 4(FXS) Port 5(FXO) Port 6(FXO) Port 7(FXO) Port 8(FXO)
	Check All Inverse Check All FXO Ports Check All FXS Ports
Modify	Reset

Figure 3-63 Modify Port Group

To delete a port group, check the checkbox before the corresponding index in Figure 3-61 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 3-61.



3.6.5 Advanced FXO Settings

	Advanced FXO Settings
Mailbox Settings	
Mailbox Account	2251582804@qq.com
Password	••••••
Outgoing(SMTP)	smtp.qq.com Port 25
SSL	
Recipient	2251582804@qq.com
Subject	Warning: gateway port disconn
Content	gateway:[devinfo],port[port]
Content	disconnection
Sendir	ing test
(Enclosing)	2000
Note:1,Multiple recipi	ents must be separated by ','
2,In subject and cont	
	ents the device information i.e. device type and serial number.
2.2,[port] indicates	the port number.
FXO Off-line Alarm	
Port	
	Port 1(FXS) Port 2(FXS) Port 3(FXS) Port 4(FXS)
	Port 5(FXO) Port 6(FXO) Port 7(FXO) Port 8(FXO)
	Check All Inverse
Blacklist of FXO Incomir	ng
Calls	
	1223,33213
Blacklist	
December 11 and	
Processing Mode	Hang up after pick-u 💌
Hang-up Delay (ms)	2000
Note:1 Multiple black	lists must be separated by ','
	ection" feature should be enabled for the port to activate the blacklist
feature.	
	Save Reset
_	

Figure 3-64 Advanced FXO Settings Interface

See Figure 3-64 for the Advanced FXO Settings interface. The table below explains the configuration items on the Email Setting interface.

Item	Description



Mailbox Account, Password	Sets the account and password of the mailbox.
Outgoing (SMTP), Port	Sets the server address and port for Email sending.
SSL	Sets whether to encrypt the sending/receiving mails via SSL.
Recipient	Sets the address of the recipient.
Subject	Sets the mail subject.
Content	Sets the mail content.
FXO Off-line Alarm	After selecting the ports, the gateway will send the alarm email when the selected ports are off-line.
Blacklist of FXO Incoming	Sets the blacklist of the FXO incoming calls.
Processing Mode	Sets the processing mode for the blacklist, including two options: Hang up after pick-up and Hang up after ringing. The default value is Hang up after pick-up.
Hang-up Delay	Sets the delay to hang up the call after the pick-up.

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

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 3-65.

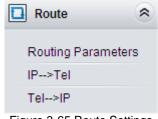


Figure 3-65 Route Settings

3.7.1 Routing Parameters

IP->TEL	Route before Number Manipulate
EL->IP	Route before Number Manipulate
Route Detection Cycle (s)	0

Figure 3-66 Routing Parameters Configuration Interface

See Figure 3-66 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*. The gateway



will send the option message to detect whether the TEL->IP routing is valid or not after setting the Route Detection Cycle. If the remote address doesn't respond this option message within the set cycle, this routing will be regarded as invalid and the outgoing calls won't be routed to this TEL->IP routing.

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	*	
	*	No available routing rule!
() Port	*	Add New
Route	*	
Routing Paramete	ers	
IP>Tel		•
Tel>IP		

Figure 3-67 IP→Tel Routing Rule Configuration Interface (Standard)

See Figure 3-67 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 3-68. You may use the default values of all the configuration items herein.

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

Figure 3-68 Add New Routing Rule (IP→Tel)

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

Item	Description
Index	The unique index of each routing rule, which denotes its priority. A routing rule with



	T
	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 IP	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 $0 \sim 9$, $\sqrt{"[*]"}$, "#" 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
Davida hu Numahar	will be routed to can be set via the item <i>SIP Account</i> on the <u>FXS</u> or <u>FXO</u> Settings
Route by Number	interface. In such case, the configuration item Call Destination goes invalid and
	shows Route by Number on the routing rule configuration interface. The default
	setting is <i>disabled</i> .
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 3-69 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.

				IF	P->Tel Routing Rule			
Check	Index	Source IP	Ca	IlerID Prefix	CalleeID Prefix	Call Destination	Description	Modify
	63	*		*	×	Route by Number	default	
Check All	Uncheck All	Inverse	Deleter	Clear All				Add Nev

Figure 3-69 IP→Tel Routing Rule Configuration Interface

Click **Modify** in Figure 3-69 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\rightarrowTel)** interface. Note that the item **Index** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-69 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the **Clear All** button in Figure 3-69.

See Figure 3-70 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
The priority decreases Symbol * in Source	ases from top to bot ce IP, CallerID Prefix	ns such fields as Source IP, CallerID Prefix, CalleeID Prefix, Route by Number, Destination Port Group and Description. ottom; adjacent fields are separated by a space x and CalleeID Prefix indicates any IP address or string; When Route by Number is set to 1, the Destination Port Group will be enabled. after your modification!
*** 0 0 defa	LUIT	
1 Items Total		Save

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

3.7.3 Tel to IP

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

Figure 3-71 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-71 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 3-72. You may use the default values of all the configuration items herein except for *Destination IP* and *Destination Port*.



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

Figure 3-72 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,	ID address and part number of the remate and to which the call will be routed
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.



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See Figure 3-73 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.

			Tel->IP Routing	Rule			
Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Destination IP	Destination Port	Description	Modify
63	*	*	*	192.168.1.101	5060	default	
Uncheck /	All E Inverse	E Delete E Cle	ar All				Add Nev
	63	63 *	63 * *	Index Call Initiator CallerID Prefix CalleeID Prefix 63 * * * *	63 * * * 192.168.1.101	Index Call Initiator CallerID Prefix CalleeID Prefix Destination IP Destination Port 63 * * * 192.168.1.101 5060	Index Call Initiator CallerID Prefix CalleeID Prefix Destination IP Destination Port Description 63 * * * 192.168.1.101 5060 default

Figure 3-73 Tel→IP Routing Rule Configuration Interface

Click **Modify** in Figure 3-73 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 3-73 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 3-73.

See Figure 3-74 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.

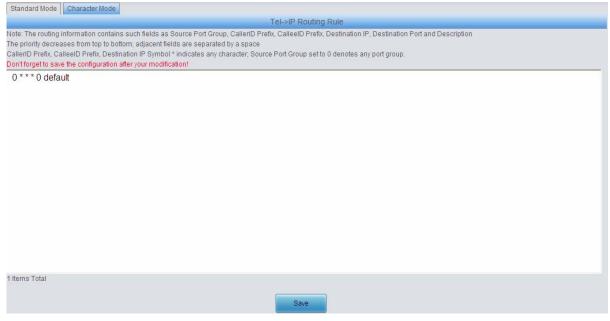


Figure 3-74 Tel→IP Routing Rule Configuration Interface (Character)

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 3-75.



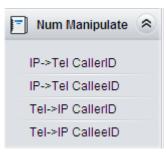


Figure 3-75 Number Manipulation

3.8.1 IP to Tel CallerID

					IP->Tel Calle	erID Number Manipulation	n 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	Description	Modify
	63	8	2	ż	0	0	20			default	2
									5		

Figure 3-76 IP→Tel CallerID Manipulation Interface (Standard)

See Figure 3-76 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 3-77 for the IP \rightarrow Tel CallerID manipulation rule adding interface. You may use the default values of all the configuration items herein.



IP->Tel CallerI)
Index:	62 💌
Description:	default
Call Initiator:	*
CallerID Prefix:	*
CalleeID Prefix:	*
Stripped Digits from Left:	0
Stripped Digits from Right:	0
Reserved Digits from Right:	20
Prefix to Add:	
Suffix to Add:	
Save	Close

Figure 3-77 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.
Description	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.



	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
CallerID Prefix,	indicates any character between these two characters. ',' is used to separate
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	this item exceeds the length of the current number, the whole number will be
Left	deleted. The default value is 0.
	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. The default value is 0.
	The amount of digits to be reserved from the right end of the number. Only when the
Reserved Digits	value of this item is less than the length of the current number will some digits be
from Right	deleted from left; otherwise, the number will not be manipulated. The default value
	is 20.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

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

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

Click **Modify** in Figure 3-76 to modify a number manipulation rule. See Figure 3-78 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 Calle	erlD
Index:	63 💌
Description:	default
Call Initiator:	*
CallerID Prefix:	*
CalleeID Prefix:	*
Stripped Digits from Left:	0
Stripped Digits from Right:	0
Reserved Digits from Right:	20
Prefix to Add:	
Suffix to Add:	
Save	Close

Figure 3-78 Modify IP→Tel CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-76 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 3-76.

See Figure 3-79 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
Note: The Number 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 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 20 <@#> <@#> default
1items Total
Save

Figure 3-79 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 3-81 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 3-76).

					IP->Tel Calle	elD Number Manipulation	n 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	Description	Modify
	63		*		0	0	20			default	2

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

Standard Mode Character Mode
IP->Tel CalleelD Number Manipulation Rule
Note: The Number 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 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 20 <@#> <@#> default
1Items Total
Save



Figure 3-81 IP→Tel CalleeID Manipulation Interface (Character)

3.8.3 Tel to IP CallerID

Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	Description	Modify
	63	.*.:	2	2	0	0	20			default	0

Figure 3-82 Tel→IP CallerID Manipulation Interface (Standard)

See Figure 3-82 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 3-83 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 Callerl	D
Index:	62 💙
Description:	default
Source Port Group:	*
CallerID Prefix:	*
CalleeID Prefix:	*
Stripped Digits from Left:	0
Stripped Digits from Right:	0
Reserved Digits from Right:	20
Prefix to Add:	
Suffix to Add:	
Save	Close

Figure 3-83 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
Index	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
Description	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 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
CallerID Prefix,	indicates any character between these two characters. ',' is used to separate
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
	Initiator specify a number manipulation rule for calls.
	Note: "[*]" represents DTFM symbol *, while "*" represents any string.
Chrisper of Distitute from	The amount of digits to be deleted from the left 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
Left	deleted. The default value is 0.
Chrisper of Distitute from	The amount of digits to be deleted from the right end of the number. If the value of
Stripped Digits from	this item exceeds the length of the current number, the whole number will be
Right	deleted. The default value is 0.
	The amount of digits to be reserved from the right end of the number. Only when the
Reserved Digits	value of this item is less than the length of the current number will some digits be
from Right	deleted from left; otherwise, the number will not be manipulated. The default value
	is 20.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

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

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

Click **Modify** in Figure 3-82 to modify a number manipulation rule. See Figure 3-84 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 Calle	rID
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:	20
Prefix to Add:	
Suffix to Add:	
Save	Close

Figure 3-84 Modify Tel→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-82 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 3-82.

See Figure 3-85 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 20 <@#> <@#> default
1 Items Total
Save

Figure 3-85 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 3-86, Figure 3-87 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 3-82).

					Tel->IP Calle	elD Number Manipulation	n 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	Description	Modify
	63		*	2	0	0	20			default	2
			5				h				

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

Standard Mode Character Mode
Tel->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 *: 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 20 <@#> <@#> default
1 Items Total



Figure 3-87 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 3-88 for details.

System Tools 📚
Management
Config File
Network
Upgrade
Signaling Capture
Data Recording
Call Log
Operation Log
Backup & Upload
Factory Reset
Access Control
System Monitor
Centralized Manage
PING Test
TRACERT Test
Change Password
Restart

Figure 3-88 System Tools



3.9.1 Management

	nagement	
	WEB Port	80
	Access Setting	Allow All IPs
SYSLOG	Parameters	
	SYSLOG	⊙ Yes ◯ No
	Server Address	201.123.115.20
	SYSLOG Level	INFO 💌
CDR Para	meters	
	Send CDR	⊙Yes ONo
	Server Address	127.0.0.1
	Server Port	3
Time Para	imeters	
	NTP	⊙Yes ONo
	NTP Server Address	time.nist.gov
	Synchronizing Cycle	3600
	Daily Restart	OYes ONo
	Restart Time	0 💌 h 0 💌 m
	System Time	Modify 2015-11-09 15:20:54
	Time Zone	GMT+8:00 (Beijing, Singapore, Taipei, Kual 💙

Figure 3-89 Management Parameters Setting Interface

See Figure 3-89 for the Management Parameters Setting interface. The table below explains the items shown in the above figure.

Item	Description
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.
	Sets the IP addresses which can access the gateway via WEB. By default, all IPs
Access Soffing	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 feature of sending CDR. It is required to fill in Server
Send CDR	Address and Server Port in case Send CDR is enabled. By default, Send CDR is
	disabled.



.		
Server Address	The address of the server to receive CDR.	
Server Port	The port of the server to receive CDR.	
	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 enabled.	
	Sets the Server address for NTP time synchronization. By default, the address is	
NTP Server Address	time.nist.gov	
	Sets the cycle for NTP time synchronization, calculated by s, with the default value	
Synchronizing Cycle	of 3600.	
	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.	
	The system time. Check the checkbox before <i>Modify</i> and change the time in the	
System Time	edit box when NTP is disabled.	
Time Zone	The time zone of the gateway.	



3.9.2 Configuration File

	SMGConfig.ini	•
Config File		
[Version]		
GWSvrV=1.2.12_20130719		
WebV=1.2.11_20130719		
CpldV=0.3		=
[DbgLog]		
LogLevel=3		
LogCreatePeriod=24		
LogMaxPeriod=4		
LogMaxPeriodSaved=5		
LogOverWrite=1		
LogDirectory=/dev/shm/shcti		
LogType=3		
[WebCtrl]		
LocalAddress=127.0.0.1		
LocalPort=1001		
MailAlarm=1		
MailUser=2251582804@qq.com		
MailPassword=wb110923		
SMTPAddress=smtp.qq.com		
MailReceivers=2251582804@qq.com		
ChinfoPath=/usr/local/SMG/ch.data.php		
PopInfoPath=/usr/local/SMG/pop.data.php		
SMTPPort=25		
EnableSSL=1		
MailTitle=Warning: gateway port disconnection		
MailBody=gateway:[devinfo],port[port] disconnection		
[UserInfo]		
UserName=BqtTPNLUr/23x1wC/w		
Pwd=BqtTPNLUr/23x1wC/w		
[Monitor]		
LocalAddress=127.0.0.1		
LocalPort=1002		
AutoExec=1		
ExecPath=E:\recorder\Recorder\Output\BIN\		
UpgradeExecPath=/usr/local/apache/htdocs/RecUpgrade		-
Save Reset Note: You shall restart system to validate the modified configuration file	!	
Figure 3-90 Configuration File Interface		

Figure 3-90 Configuration File Interface

See Figure 3-90 for the Configuration File interface, including two files: SMGConfig.ini and ShConfig.ini. You can check and modify the items in these configuration files through this interface. Configurations about the gateway server, such as route rules, number manipulation and so on, are included in SMGConfig.ini; configurations about the board are included in ShConfig.ini. You can modify these configurations on the interface directly, and then click **Save** to save the above settings into the gateway or click **Reset** to restore the configurations.



3.9.3 Network

Ner	twork Settings
Network Type:	Static
IP Address (I):	201.123.115.74
Subnet Mask (U):	255.255.255.0
Default Gateway (D):	201.123.115.254
DNS Server (P):	0.0.0.0
IPv6 Address:	Enable
IPv6 IP Address(I):	ff::1:2:3:4
IPv6 Subnet Prefix Length(U):	88
Default IPv6 Gateway(D):	ff::1:2:3:0
IPv6 DNS Server(P):	ff::1:2:3:1
Speed and Duplex Mode:	Automatic Detection
Save	Reset

Figure 3-91 Network Settings Interface

See Figure 3-91 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 three options: Static, DHCP and PPPoE. If PPPoE is used, it is necessary to enter the username and the password of the network.

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.



3.9.4 Upgrade

	Current Version		
Serial Num	00001678		
WEB	Version 1.7.1_Release2017042711		
Service	Version 1.7.1_Release2017042711		
U-boot	Version Mar 02 2016-21:57:12		
Kernel	Version #210 Fri Apr 14 09:25:23 CST 2017		
Product Type	1008B-4S4O(RJ11)		
Select an U	odate File Browse		
	Update Reset		

Figure 3-92 Upgrade Interface

See Figure 3-92 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 3-93.

Serial Num	00001678	}		
WEB	Version 1.	7.1_Release2017042711		
Service	Version 1.			
U-boot	Version M	ar 02 2016-21:57:12		
Kernel	Version #2	210 Fri Apr 14 09:25:23 CST 2017		
Product Type	1008B-4S	40(RJ11)		
Select an Up	pdate File	C:\Users\Administrator\ Browse		
		32%		
		32%		
		32%		
The file is	uploadin	32% ng. Please do not leave this page		
The file is	uploadin		el	
The file is		ng. Please do not leave this page		
	U	ng. Please do not leave this page) 	
The file is start upload	U	ng. Please do not leave this page		
	U	ng. Please do not leave this page		
	U	ng. Please do not leave this page		
	U	ng. Please do not leave this page		
	U	ng. Please do not leave this page		
	U	ng. Please do not leave this page	_	



Figure 3-93 File Uploading Interface

After a successful uploading of the file, the gateway will start to upgrade the system. See Figure 3-94 and you can learn the detailed upgrading information from the upgrade information box at the bottom.

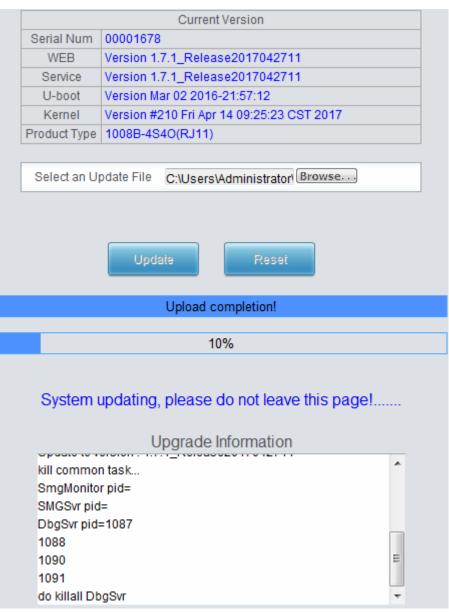


Figure 3-94 System Upgrading Interface

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.



3.9.5 Signaling Capture

	Packet Capture		
Capture Data on All Network Cards	Enable		
Signaling Packet Capture	SIP&Syslog&Cent -		
RTP Packet Capture	RTP Port Range 💌 50000,50767	Start	Stop

Figure 3-95 Signaling Capture Interface

See Figure 3-95 for the Signaling Capture interface. Packet capture contains Signaling Packet Capture and RTP Packet Capture. You can select either of them to start the capture according to your requirement. Once the configuration item "Capture Data on All Network Cards" is enabled, the gateway will capture the data on all kinds of network cards, including eth0, lo (local loopback) and veth0 (virtual network card); otherwise, it will only capture the data on eth0. Click *Start* to start capturing packets. Click *Stop* to stop the capture and download the captured packets.

3.9.6 Data Recording

Data Re	cording
Channel	Channel1_(FXS)
Mode	Default
Interface	Мар
Start Note: 1.Only the latest 60s data ca 2.Recording parameters:80 mono,U-law formatted.	

Figure 3-96 Data Recording Interface

See Figure 3-96 for the Debug & Record interface. You can select a channel and the recording mode to start the data recording. Click *Start* to start the corresponding recording. Click *Stop* to stop the recording and download the recorded file.



3.9.7 Call Log

Call Log	SIP Log	Enable Call Log	Download				
Call from IF	Channel						Clear All
01/01/197	0 22:58:05:313	IP Channel 0 outgoing call 5@ IP Channel 1,incoming call fro IP Channel 0 call end, reason:	m remote end "8005"	<pre><sip:8005@201.123.115.72< pre=""></sip:8005@201.123.115.72<></pre>	>,call-id: 2296587899@	Callee 5 call end, re	ason:no idle chan
•				. 111.			
Call from F	ort	Select a Port	Port5				Clear All
			Fig	ure 3-97 Call Log	g Interface		
Call Log	SIP Log	Call Log	Download				
SIP Log						Refresh	Clear All
From: "80 To: <sip:5 Call-ID: 2 CSeq: 20</sip:5 	05" <sip:8005@ @201.123.115 296587899@20 INVITE ssip:8005@201</sip:8005@ 						E

soniacc -sip.0003@201.123.113.12.3000-	
Aax-Forwards: 70	
Jser-Agent: Gateway	
Expires: 120	
(-callCause: Communicate without Network	
NIOW: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO	
Content-Type: application/sdp	
Content-Length: 480	
=0	
p=- 1 2 IN IP4 201.123.115.72	
s=Gateway	
≔IN IP4 201.123.115.72	
=0 0	
n=audio 50000 RTP/AVP 8 0 18 4 9 96 98 100 102 103 104 101	
a=rtpmap:8 PCMA/8000	
a=rtpmap:0 PCMU/8000	
a=rtpmap:18 G729/8000	
a=rtpmap:4 G723/8000	
a=rtpmap:9 G722/16000	
a=rtpmap:96 AMR/8000	
i=rtpmap:98 iLBC/8000	
=rtpmap:100 silk/16000	

Figure 3-98 SIP Log Interface

See Figure 3-97, Figure 3-98 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**.



3.9.8 Operation Log

2017-01-23 09:02:07 ClientIP:201.123.115.56 OpenCallLog 2017-01-23 09:02:07 ClientIP:201.123.115.56 ClearIPChannelCallLog 2017-01-23 09:02:09 ClientIP:201.123.115.56 ClearIPChannelCallLog 2017-01-23 09:02:20 ClientIP:201.123.115.56 ClearIPChannelCallLog 2017-01-23 09:02:20 ClientIP:201.123.115.56 ClearSipCallLog 2017-01-23 09:02:20 ClientIP:201.123.115.56 ClearSipCallLog 2017-01-23 09:02:20 ClientIP:201.123.115.56 ClearSipCallLog 2017-01-23 14:04:41 ClientIP:201.123.115.56 ClearSipCallLog 2017-01-02 03:17:27 ClientIP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 03:18:01 ClientIP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 09:08:03 ClientIP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 09:08:03 ClientIP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 09:08:03 ClientIP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 09:08:03 ClientIP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-01 03:05:9 ClientIP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 10:36:59 ClientIP:201.123.115.56 Restart System 1970-01-01 08:02:20 ClientIP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 08:02:20 ClientIP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 08:03:22 ClientIP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 08:03:23 ClientIP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 12:58:33 ClientIP:201.123.115.56 PcapStat=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767 1970-01-01 12:58:33 ClientIP:201.123.115.56 PcapStap 1970-01-01 14:41:30 ClientIP	
2017-01-23 09:02:09 ClientlP:201.123.115.56 OpenCallLog 2017-01-23 09:02:20 ClientlP:201.123.115.56 ClearTdmPortCallLog Port.1 2017-01-23 09:02:20 ClientlP:201.123.115.56 OpenCallLog 2017-01-23 14:04:41 ClientlP:201.123.115.56 ClearSipCallLog 2017-01-23 14:04:41 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 03:17:27 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 03:18:10 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 09:08:19 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 09:08:19 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 09:09:03 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 09:09:03 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 10:36:59 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-01 08:03:29 ClientlP:201.123.115.56 ModFXOPort=port.0 port_type:FX08001 <@#> 0 <@#> 0 1 0 0 <@#> 0 1 0 0 admin 0 1970-01-01 08:03:29 ClientlP:201.123.115.56 Restart System 1970-01-01 08:02:20 ClientlP:201.123.115.56 SaveFXOSet= global_FXODetectCallerlDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold-30 Silence 1970-01-01 08:02:20 ClientlP:201.123.115.56 SaveFXOSet= global_FXODetectCallerlDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold-30 Silence 1970-01-01 08:03:33 ClientlP:201.123.115.56 Restart System 1970-01-01 08:03:33 ClientlP:201.123.115.56 Restart System 1970-01-01 12:58:22 ClientlP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767 1970-01-01 12:58:33 ClientlP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767 1970-01-01 14:41:30 ClientlP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767	
2017-01-23 09:02:10 ClientlP:201.123.115.56 ClearTdmPortCallLog Port.1 2017-01-23 09:02:20 ClientlP:201.123.115.56 OpenCallLog 2017-01-23 14:04:41 ClientlP:201.123.115.56 ClearSipCallLog 2017-01-23 14:04:41 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FXO8005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 03:17:27 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FXO8005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 03:18:01 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FXO8005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 03:09:09:08:19 ClientlP:201.123.115.56 ModFXOPort=port.4 port_type:FXO8005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 09:09:09:09:09:09:09:09:09:09:09:09:09:0	
2017-01-23 09:02:20 ClientlP:201.123.115.56 OpenCallLog 2017-01-23 09:02:20 ClientlP:201.123.115.56 ClearSipCallLog 2017-01-23 14:04:41 ClientlP:201.123.115.56 SaveSipSet= LocalSipPort:5060 global_register:1 global_username:keep12 global_password:123 global_ 1970-01-02 03:17:27 ClientlP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 03:18:01 ClientlP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 09:08:19 ClientlP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 09:08:19 ClientlP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 09:09:03 ClientlP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 10:36:59 ClientlP:201.123.115.56 Restart System 1970-01-01 13:54:15 ClientlP:201.123.115.56 Restart System 1970-01-01 08:03:29 ClientlP:201.123.115.56 Restart System 1970-01-01 08:03:20 ClientlP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 08:03:20 ClientlP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 08:03:23 ClientlP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 08:03:24 ClientlP:201.123.115.56 Restart System 1970-01-01 08:03:23 ClientlP:201.123.115.56 Restart System 1970-01-01 08:03:23 ClientlP:201.123.115.56 Restart System 1970-01-01 08:03:23 ClientlP:201.123.115.56 Restart System 1970-01-01 12:58:22 ClientlP:201.123.115.56 Restart System 1970-01-01 12:58:22 ClientlP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767 1970-01-01 12:58:33 ClientP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767 1970-01-01 14:41:57 ClientlP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767	1
2017-01-23 09:02:20 ClientIP:201.123.115.56 ClearSipCalLog 2017-01-23 14:04:41 ClientIP:201.123.115.56 SaveSipSet= LocalSipPort:5060 global_register:1 global_username:keep12 global_password:123 global_ 1970-01-02 03:17:27 ClientIP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 03:18:01 ClientIP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 09:08:19 ClientIP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 09:09:03 ClientIP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 09:09:03 ClientIP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 10:36:59 ClientIP:201.123.115.56 RodFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-01 03:59 ClientIP:201.123.115.56 RodFXSPort=port:0 port_type:FXS8001 <@#> 0 <@#> 0 1 0 0 <@#> 0 1 0 0 admin 0 1970-01-01 08:03:29 ClientIP:201.123.115.56 Restart System 1970-01-01 08:02:20 ClientIP:201.123.115.56 Restart System 1970-01-01 08:02:20 ClientIP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 08:02:26 ClientIP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 08:03:43 ClientIP:201.123.115.56 Restart System 1970-01-01 08:03:43 ClientIP:201.123.115.56 Restart System 1970-01-01 02:03:108 ClientIP:201.123.115.56 Restart System 1970-01-01 02:03:133 ClientIP:201.123.115.56 Restart System 1970-01-01 12:58:22 ClientIP:201.123.115.56 Restart System 1970-01-01 12:58:33 ClientIP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767 1970-01-01 12:58:33 ClientIP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767 1970-01-01 14:41:57 ClientIP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767	
2017-01-23 14:04:41 ClientIP:201.123.115.56 SaveSipSet= LocalSipPort:5060 global_register:1 global_username:keep12 global_password:123 global_ 1970-01-02 03:17:27 ClientIP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 03:18:01 ClientIP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 09:08:19 ClientIP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 09:08:19 ClientIP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 1 0 admin 0 1970-01-02 09:09:03 ClientIP:201.123.115.56 ModFXOPort=port:4 port_type:FX08005 <@#> 1 <@#> 0 0 0 admin 0 1970-01-02 10:36:59 ClientIP:201.123.115.56 ModFXSPort=port:0 port_type:FX08005 <@#> 0 <@#> 0 1970-01-01 13:54:15 ClientIP:201.123.115.56 ModFXSPort=port:0 port_type:FXS8001 <@#> 0 <@#> 0 1 0 0 <@#> 0 1 0 0 admin 0 1970-01-01 08:03:29 ClientIP:201.123.115.56 Restart System 1970-01-01 08:03:29 ClientIP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 08:02:20 ClientIP:201.123.115.56 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-30 Silence 1970-01-01 08:03:43 ClientIP:201.123.115.56 Restart System 1970-01-01 02:05:31:08 ClientIP:201.123.115.56 Restart System 1970-01-01 02:06:31:08 ClientIP:201.123.115.56 Restart System 1970-01-01 12:58:22 ClientIP:201.123.115.56 Restart System 1970-01-01 12:58:22 ClientIP:201.123.115.56 Restart System 1970-01-01 12:58:33 ClientIP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767 1970-01-01 12:58:33 ClientIP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767	
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1970-01-01 14:41:57 ClientIP:201.123.115.56 PcapStop	
1970-01-01 15:28:36 ClientlP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767	
1970-01-01 15:30:29 ClientIP:201.123.115.56 PcapStop	
1970-01-01 15:34:29 ClientIP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767	
1970-01-01 15:34:42 ClientlP:201.123.115.56 PcapStop	
1970-01-01 15:38:02 ClientlP:201.123.115.56 PcapStart=SignalingPcap:3 RTPPcap:1 RTPPort:50000,50767	
< <u> </u>	•

Figure 3-99 Operation Log Interface

See Figure 3-99 for the Operation Log interface, which is used to check the operation records on WEB. Click **Refresh** to refresh the log; click **Clear All** to clear all the operation logs and click **Download** to download the logs.

Note: The sign <@#> here means the configuration item is unconfigured.

3.9.9 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	d click the button 'Upload' to start.	
Configuration File	Browse	Upload

Figure 3-100 Backup & Upload Interface

See Figure 3-100 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	ation 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 3-101 Backup & Upload & Prompt Interface

Click **OK** on the prompt box (Figure 3-101) to upload the configuration file to the gateway. Now the prompt information 'System is rebooting, please do not leave this page' appears. See Figure 3-102. 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 fi	e, click the 'Backup' button to start.	Backup
	Data Upload	
To upload a configuration file, Config File	Browse.	Upload
	sfully upload the configuration file, the gatewa	
System is rel	oting. Please do not leave this page	L
Fig	ure 3-102 Configuration File Uploading In	terface



3.9.10 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 3-103 Factory Reset Interface

See Figure 3-103 for the factory reset interface. Click *Reset* to restore all configurations on the gateway to factory settings.

3.9.11 System Monitor

System Monitor	ſ
Watchdog:	☑ Enable
Dog Feeding Interval (s)	5
Automatically Restart the Service if Undetected	Enable
Threshold to Judge Heartbeat Loss for Service	e(s): 60
Save	Reset

Figure 3-104 System Monitor Configuration Interface

See Figure 3-104 for the System Monitor Configuration interface. Watchdog is a timing reset system used to avoid application crash. You can set the dog feeding interval when this feature is enabled. The feeding interval is calculated by s, with the value range of 1~15s. By default, this feature is enabled with the default value of 5s. As the feature 'Automatically restart the service if undetected' is enabled, the service application will restart automatically if it is not detected by the gateway guard application. By default, this feature is enabled. Threshold to Judge Heartbeat Loss for Service is used to judge whether the gateway receives the heartbeat packets from the service during the set time, if not, it is considered that the gateway service has been disconnected. It is calculated by s, with the value range of 20~120s and the default value of 60s.



3.9.12 Centralized Manage

Centralized Manage			
Centralized Manage:	☑ Enable		
Management Platform:	DCMS		
Server Address *:	127.0.0.1		
Company Name *:			
Authorization Code *:			
Gateway Description:			
Advanced Enable Lock Feature Once Successfully Connected:	 ✓ Enable 		
Lock Parameter			
Working Status:	Disabled		
Save	Download MIB		

Figure 3-105 Centralized Manage Setting Interface

See Figure 3-105 for the Centralized Manage Setting interface. The gateway can register to a centralized management platform and accept the management of the platform. The table below explains the items shown in above figures.

Item	Description
Management	Select a management platform for the gateway to register, including two options:
Platform	DCMS and Others.
	The address of the server in which the management platform locates, It can be IP or
O a margin A status a s	a domain name, valid only when DCMS is selected.
Server Address	Note: To configure the domain name, the DNS should be already configured and
	the corresponding domain name must be analyzable.
	The name used to register the gateway to Synway DCMS, valid only when DCMS is
Company Name	selected.
	The authorization code is used for the connection verification. A device can connect
Authorization Code	to the DCMS successfully only after it passes the verification. Only valid when
	DCMS is selected.
	The description displayed on Synway DCMS after the gateway is registered to
Gateway	Synway DCMS, giving an easy identification of the gateway in device grouping. This
Description	item is valid only when DCMS is selected.



Enable Lock Feature Once Successfully Contected	Once this feature is enabled, you can lock the device according to the corresponding parameters. This item is valid only when DCMS is selected.	
IP Address	Once this feature is enabled, you are required to fill in the authorization code while modifying the information related to the IP address in the Network interface. This item is valid only when DCMS is selected.	
Registrar Server	Once this feature is enabled, you are required to fill in the authorization code while modifying the address and port of the registrar server in the SIP Settings interface. This item is valid only when DCMS is selected.	
Working Status	The status of the connection between the gateway and the centralized management server. This item is valid only when DCMS is selected.	
Centralized Management Protocol	Set the centralized management protocol. It only supports SNMP currently.	
SNMP Version	Set the version of SNMP, three options available: V1, V2 and V3, with the default value of V2. This item is valid only when Others is selected.	
Monitoring Port	Monitoring Port for SNMP on the gateway. This item is valid only when Others is selected.	
Community String	Community string used for information acquisition.	
Account	The account of SNMP, valid only when the SNMP version is set to V3.	
Grade	The grade of SNMP, three options available: Neither authenticated nor encrypted, Authenticated but not encrypted and Authenticated and encrypted, with the default value of <i>Neither authenticated nor encrypted</i> . It is valid only when the SNMP version is set to V3.	
Authentication	The authentication password required to enter when the item Grade is set to	
Password	Authenticated but not encrypted or Authenticated and encrypted.	
Encryption	The encryption password required to enter when the item Grade is set to	
Password	Authenticated and encrypted.	

3.9.13 Access Control

Check	Index	Command		Modify
	0	iptables -I INPUT -s 123.45.6.7 -j DROP		
	ck All Inverse	Delete Clear All		

Figure 3-106 Access Control List Interface

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



Access Control Command		
Index:	1	
Command:		
	Close	

Figure 3-107 Add Access Control Command Interface

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

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

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

Figure 3-108 Access Control Command Modification Interface

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

Note:

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

2. After you add, modify or delete a command manually, don't forget to click the *Apply* button to make your settings valid. However, in case the gateway restarts or the configuration is leading-in, the command will get valid automatically without the need for you to click the *Apply* button.



3.9.14 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 3-109 Ping Test Interface

See Figure 3-109 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	Destination IP address or domain name on which the Ping test is executed.	
Ping Count	The number of times that the Ping test should be executed. Range of value: 1~100.	
Package Length	Length of the data package used in the Ping test. Range of value: 56~1024 bytes.	
Info	The information returned during the Ping test, helping you to learn the network	
Info	connection status between the gateway and the destination address.	

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



3.9.15 TRACERT Test

Tracert 1	Test
Destination Address	127.0.0.1
Maximum Jumps (1-255)	30
Start	End
Info	
	<u>×</u>

Figure 3-110 Tracert Test Interface

See Figure 3-110 for the Tracert test interface. A Tracert test can be initiated from the gateway on a designated IP address to check the routing status between them. The table below explains the configuration items shown in the above figure.

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

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



3.9.16 Change Password

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

Figure 3-111 Password Changing Interface

See Figure 3-111 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.

3.9.17 Restart

System Restart					
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 3-112 System Restart Interface

See Figure 3-112 for the Restart interface. Click **Restart** to restart the whole gateway system. A dump file will be generated each time you restart the system. Click **Download** and you can download it to help troubleshoot issues.



Appendix A Technical Specifications

Dimensions

SMG1004B, SMG1008B: 220×148×40 mm³ SMG1016B4: 440×44×267 mm³

Weight

SMG1004B, SMG1008B: 0.375 kg SMG1016B4: 2.530 kg

Environment

Operating temperature: 0 °C-45 °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: 4/8/16

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:

SMG1004B, SMG1008B: 12V the direct current bigger than 3A

SMG1016B4: 100~240V AC

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



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>3.5.8 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.

Q10. How to configure the features Communication without Power and Communication without Network for the SMG-B analog gateway?

The feature **Communication without Power** is implemented in hardware. Once the power to the device is cut off, the station which is linked with the FXS port and the trunk which is linked with the FXO port will connect to each other directly and keep the good communications between phones and networks. Currently, this feature is only supported on SMG1004B-2S2O, SMG1008B-4S4O and SMG1016B4-8S8O. The FXS and FXO ports are one-to-one correspondence (Take SMG1016B4-8S8O for example, the phone linked with Channel 1 will be connected to the PSTN line which links with Channel 9.).

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



Refer to $\underline{Q2}$ in this chapter for detailed information.



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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