

Synway SMG-C Series Analog Gateway

SMG1008C SMG1016C SMG1032C

Analog Gateway

User Manual

Version 1.7.0

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-C Series Analog Gateway!

The Synway SMG-C series analog gateway products (hereinafter referred to as 'SMG-C 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.

1.1 Typical Application

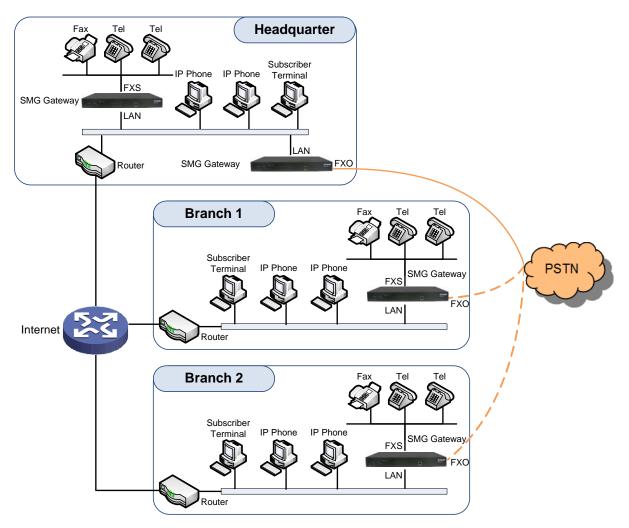


Figure 1-1 Typical Application for SMG-C Series Gateway

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		
Network Protocol	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN.		
Static IP	IP address modification support.		
DHCP	IP address dynamic allocation support.		
DNS	Domain Name Service support.		
Security	Description		
Admin Authentication	Supports admin authentication to guarantee the resource and data security.		
System Monitor	Monitors the running status of the system and the server.		
Maintain & Upgrade	Description		
WEB Configuration	Support of configurations through the WEB user interface.		
Language	Chinese, English.		
Software Upgrade	Support of user interface, gateway service, kernel and firmware upgrades based on WEB.		
Tracking Test	Support of Ping and Tracert tests based on WEB.		
SysLog Type	Three options available: ERROR, WARNING, INFO, DEBUG.		

1.3 Hardware Description

The SMG C-type analog gateway integrates embedded LINUX system within the POWERPC+DSP hardware architecture. It has 8/16/32 voice ports (FXS/FXO) and 2 LANs on the chassis. Each voice port can be configured on demand to serve as an FXS or FXO interface; however, the respective amount of FXS and FXO interfaces must be multiples of 2. The SMG-8C analog gateway adopts an external 12V power supply. See below for product appearance.



Figure 1-2 SMG1008C Front View





Figure 1-3 SMG1008C Rear View







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

Interface	Description
	Amount: 2
	Type: RJ-45
LAN	Bandwidth: 10/100Mbps
	Self-Adaptive Bandwidth Supported
	Auto MDI/MDIX Supported
	Amount: 8/16/32
	Type: RJ-11
FXS/FXO	Maximum Transmission Distance: 1500m
	Charge Mode: Negative Anti-billing Supported
Console Port	Amount: 1
	Type: RS-232
	Baud Rate: 115200bps
	Connector: RJ45 to DB-9 Connector
	Data Bits: 8 bits
	Stop Bit: 1 bit
	Parity Unsupported



	Flow Control Unsupported		
Button	Description		
Reset Button	Restore the gateway to factory settings.		
LED	Description		
Power Indicator	Indicates the power state. It lights up when the gateway starts up with the power cord well connected		
Run Indicator	Indicates the running status. For more details, refer to <u>1.4 Alarm Info</u> .		
Alarm Indicator	Alarms the device malfunction. For more details, refer to <u>1.4 Alarm Info</u> .		
Link Indicator	tor The green LED, indicating the network connection status.		
ACT Indicator The orange LED, whose flashing tells data are being transmitted.			
FXS and FXO channels are respectively marked by green and red LED after on. Channel Indicator 1. When the channel is idle, the LED Lights up; 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>.

1.4 Alarm Info

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

LED	State	Description	
	Go out	System is not yet started.	
Run Indicator	Light up and flash fast	System is starting.	
	Flash slowly	System is normal.	
	Go out	System is normal.	
		Upon startup: System is normal.	
Alarm Indicator	Light up	In runtime: System is abnormal.	
	Flash	System is abnormal.	

Note:

- The startup process consists of two stages: System Booting and Gateway Service Startup. The system booting costs about 1 minute and once it succeeds, both the run indicator and the alarm indicator light up. Then after the gateway service is successfully started and the device begins to work normally, the run indicator flashes and the alarm indicator goes out.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Go to <u>Appendix 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 analog gateway in the shortest time.

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

- SMG Series Analog Gateway *1
- Angle Bracket *2, Rubber Foot Pad *4, Screw for Angle Bracket *8
- 220V Power Cord *1, External 12V Power Adapter *1 for SMG-8C gateway
- Warranty Card *1
- Installation Manual *1

Step 2: Properly fix the SMG analog gateway.

If you do not need to place the gateway on the rack, simply fix the 4 rubber foot pads. Otherwise, you should first fix the 2 angle brackets onto the chassis and then place the chassis on the rack.

Step 3: Connect the power cord.

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

Step 4: Connect the network cable.

Step 5: Connect the telephone line. The line from PSTN should be connected to FXO port (port with red LED flashing); the line from station should be connected to FXS port (port with green LED flashing).

These series products provide 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.

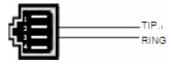


Figure 2-1

Step 6: Log in the gateway.

Enter the original IP address (LAN1: 192.168.1.101) of the SMG 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 3.1 System Login. 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 3.9.15 Change Password. After changing the password, you are required to log in again.

Step 7: Modify IP address of the gateway.

You can modify the IP address of the gateway via 'System Tools \rightarrow Network' on the WEB interface to put it within your company's LAN. Refer to <u>3.9.2 Network</u> for detailed instructions about IP modification. After changing the IP address, you shall log in the gateway again using your new IP address.

Step 8: 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 \rightarrow 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 32 is 8032.

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>3.6.3 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>3.6.3 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 analog gateway to ring the station.

Example: Provided the IP address of the SMG 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 9: 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 10: 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 11: Enable the call waiting feature. (Skip this step if not necessary.)

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

Step 12: Perform call forwarding. (Skip this step if not necessary.)

Situation 1: Hook-flash operation

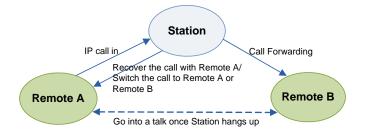


Figure 2-2 Call Forward via Hook-flash

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 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:

- The chassis of the SMG-C analog gateway must be grounded for safety reasons, according to standard industry requirements. A simple way is earthing with the third pin on the plug or the grounding studs on the machine. No or improper grounding may cause instability in operation as well as decrease in lightning resistance.
- As the device will gradually heat up while being used, please maintain good ventilation to prevent sudden failure, ensuring that the ventilation holes (see Figure 1-6) 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	中文 English	
Username: Password :		
Login Cancel		

Figure 3-1 Login Interface

The gateway only serves one user, whose original username and password are both 'admin'. You can change the username and the password via 'System Tools \rightarrow Change Password' on the WEB interface. For detailed instructions, refer to <u>3.9.15 Change Password</u>.

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

Quality in Info			Syster	n Info	
System Info		LAN			
		MAC Address	00:00:E0:A7:04:FF		
Call Count		IP Address	201.123.115.215	255.255.255.0	201.123.115.254
SIP Message Cou	nt	DNS Server	8.8.8.8		
		Receive Packets	All:34754	Error:0	Drop:0
😲 Quick Config	*	Transmit Packets	All:21180	Error:0	Drop:0
⁶⁹		Current Speed	Receive:2.2 KB/s	Transmit:116 B/s	
S VolP	*	Work Mode	100Mb/s Full Duplex		
Advanced	*				
i) Port	*	Runtime	23m 57s		
Route	*	Current Version			
		WEB	forcommon_1.7.0_201	7032214	
Num Manipulate	*	Gateway	forcommon_1.7.0_201	7032214	
	*	Serial No.	2230		
System Tools	*	U-boot	Jan 04 2017-01:59:38		
		Kernel	#209 Wed Mar 22 10:4	9:54 CST 2017	
		Product Type	1016C-8S80(RJ11)		

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

LAN			
MAC Address	00:00:E0:A7:04:FF		
IP Address	201.123.115.215	255.255.255.0	201.123.115.254
DNS Server	8.8.8.8		
Receive Packets	All:34754	Error:0	Drop:0
Transmit Packets	All:21180	Error:0	Drop:0
Current Speed	Receive:2.2 KB/s	Transmit:116 B/s	
Work Mode	100Mb/s Full Duplex		
Runtime	23m 57s		
Current Version			
WEB	forcommon_1.7.0_201	17032214	
Gateway	forcommon_1.7.0_201	17032214	
Serial No.	2230		
U-boot	Jan 04 2017-01:59:38		
Kernel	#209 Wed Mar 22 10:4	9:54 CST 2017	
Product Type	1016C-8S80(RJ11)		

Figure 3-4 System Info Interface

Refresh

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

Item	Description
MAC Address	MAC address of LAN.
IP Address	The three parameters from left to right are IP address, subnet mask and default 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.					
Transmit Packets	The amount of transmit packets after the gateway's startup, including three options:					
Transmit Packets	All, Error and Drop.					
Current Speed	Show the current speed of data receiving and transmitting.					
	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-C 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

					Chann	el State			
Channel	Туре	Number	Voltage(v)	State	Direction	CallerID	CalleeID	Reg Status	Polarity Reversal Count
1	FXS	170	0					Failed(403)	
2	FXS	171	0					Failed(403)	
3	FXS	8003	0					Unregistered	
4	FXS	8004	0					Unregistered	
5	FXS	8005	0					Unregistered	
6	FXS	8006	0					Unregistered	
7	FXS	8007	0					Unregistered	
8	FXS	8008	0					Unregistered	
9	FXO	8009	0	5				Unregistered	
10	FXO	8010	38					Unregistered	
11	FXO	8011	0	4				Unregistered	
12	FXO	8012	0	1				Unregistered	
13	FXO	8013	17					Unregistered	
14	FXO	8014	0	5				Unregistered	
15	FXO	8015	0	5				Unregistered	
16	FXO	8016	0	6				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.

Item	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).
State	Displays the channel state in real time. You can move the mouse onto the channel



	state icon for det	ailed sta	ate information.
	State	lcon	Description
	Idle		The channel is available.
	Off-hook	2	The channel picks up the call.
	Off-hook Wait Answer Ringing Talking Dialing Pending	B	The channel receives the ringback tone and is waiting
	Walt Answer		for the called party to pick up the phone.
	Ringing	:	The channel is in the ringing state.
	Talking		The channel is in a conversation.
	Dialing	C	The channel is dialing.
	Pending	2	The channel is in the pending state.
	Internal State		Internal state of the channel.
	Unusable		The channel is unavailable.
Direction	Displays the dire	ction of	the call on channel.
CallerID	Displays the Cal	lerID of	the call on channel.
CalleelD	Displays the Cal	leeID of	the call on channel.
Reg Status	Displays the reg	istration	status of the port.
Polarity Reversal	The country of the	o nolo-:+	w reversel detected by the EVO part
Count	The counts of the	e polarit	y reversal detected by the FXO port.

3.2.3 Call Count

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

Figure 3-6 Call Count Interface

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

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.
Busy	Total number of calls which fail as the called party has been occupied and replies a busy message.
No Answer	Total number of calls which fail as the called party does not pick up the call in a long time or the calling party hangs up the call before the called party picks it up.
Call Forward	Total number of calls which have been forwarded.
Routing Failure	Total number of calls which fail because no routing rules are matched.
Dialing Failure	Total number of calls which fail as the called party number does not conform to the 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

REC	GISTER	INVITE	ACK	INFO	BYE	CANCEL	NOTIFY	OPTION	
	0	1	1	0	1	0	0	0	
	0	0	0	0	0	0	0	0	
	0	1	1	0	1	0	0	0	
	0	0	0	0	0	0	0	0	
100 Trying	180 Ringing	183	Session Prose	SS	200 OK	486 Busy	487 Request Already	y Terminated	
1	1		0		2	0	0		
						0			
		0	0 1 0 0 1 0 1	REGISTER INVITE ACK 0 1 1 0 0 0 0 0 0 0 0 0 0 1 1 0 0 0 0 0 0 0 0 0 0 180 Ringing 183 Session Procession	0 1 1 0 0 0 0 0 0 0 1 1 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 Common Response 100 Trying 180 Ringing 183 Session Prosess	REGISTER INVITE ACK INFO BYE 0 1 1 0 1 0 0 0 0 0 0 0 0 0 0 0 0 1 1 0 1 1 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 Common Response 100 Trying 180 Ringing 183 Session Prosess 200 OK 1	REGISTER INVITE ACK INFO BYE CANCEL 0 1 1 0 1 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 <td>REGISTER INVITE ACK INFO BYE CANCEL NOTIFY 0 1 1 0 1 0 1 0 <td< td=""></td<></td>	REGISTER INVITE ACK INFO BYE CANCEL NOTIFY 0 1 1 0 1 0 1 0 <td< td=""></td<>	

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 3.9.2 Network for detailed settings. After configuration, click *Next* to enter the SIP Settings interface.



Network Type:	Static
IP Address (I)	201.123.115.215
Subnet Mask (U)	255.255.255.0
Default Gateway (D)	201.123.115.254
DNS Server (P)	8.8.8.8
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.

Registrar IP Address	201.123.112.212
Registrar Port	5060
Spare Registrar IP Address	
Spare Registrar Port	
Registry Validity Period (s)	60
	L

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.



Synway Information Engineering Co., Ltd

						FXS Settings						
Port	Туре	SIP Account	Display Name	Talkback Function	Bind Number	Auto Dial Num	Forbid Outgoing Call	DND	Forward	FWD Type	FWD Number	CID
1	FXS	8001	-	Disable	-		Disable	Disable	Disable			Enable
2	FXS	8002		Disable			Disable	Disable	Disable	5 <u>228</u> 3		Enable
3	FXS	8003	1	Disable			Disable	Disable	Disable			Enable
4	FXS	8004		Disable			Disable	Disable	Disable	()		Enable
5	FXS	8005		Disable		-	Disable	Disable	Disable			Enable
6	FXS	8006		Disable			Disable	Disable	Disable	5. <u>568</u> 3		Enable
7	FXS	8007	7	Disable			Disable	Disable	Disable			Enable
8	FXS	8008		Disable	-		Disable	Disable	Disable	(- (Enable
(5		III								

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	P
9	FXO	8009		Two Stages Dialing for Incoming Call	(Disable	Enable	Unregistered	Enable	
10	FXO	8010		Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	
11	FXO	8011	-	Two Stages Dialing for Incoming Call	·	Disable	Enable	Unregistered	Enable	
12	FXO	8012		Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	
13	FXO	8013	88	Two Stages Dialing for Incoming Call	0.000	Disable	Disable	Unregistered	Enable	
14	FXO	8014		Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	
15	FXO	8015		Two Stages Dialing for Incoming Call	()	Disable	Enable	Unregistered	Enable	
16	FXO	8016		Two Stages Dialing for Incoming Call	-	Disable	Enable	Unregistered	Enable	
	1			III	1					F.

8 Items Total 16 Items/Page 1/1 First Previous Next Last Go to Page 1 💌 1 Pages Total

Batch Modify

Figure 3-12 FXO Settings Interface

Next

Back

Quick Config-Completion
The configuration is finished. Please click 'Finish' to quit the Quick Config!
Note: the gateway will restart the system after you click 'Finish'. Please log in the gateway again using your new IP address.
Back Finish

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

SIP Port	5060
Register Status	Unregistered
Register Gateway	Yes
SIP Account	
Password	
Authentication Username	
Registrar IP Address	
Registrar Port	
Spare Registrar Server	✓Enable
Spare Registrar IP Address	
Spare Registrar Port	
Registry Validity Period (s)	600
Multi-Registrar Server Mode	Enable
SIP Transport Protocol	UDP 💌
Switch Signal Port if SIP Registration 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>3.9.16 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-15.

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



						
	when this configuration is set to Yes can you see the configuration items SIP					
	Account and Password.					
SIP Account	When the gateway initiates a call to SIP, this item corresponds to the username of SIP.					
Password	Registration password of the gateway. To register the gateway to SIP, both configuration items <i>SIP Account</i> and <i>Password</i> 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.					
Spare Registrar IP Address	Address of the spare registry server for the gateway to register. The gateway will enable the spare registrar server if the master registrar server has no reply, or the master server is detected with no response in case the item <i>Detection Server Cycle</i> is enabled.					
Spare Registrar Port	Signaling port of the spare registry server.					
Registry Validity Period	Validity period of the SIP registry. Once the registry is overdue, the gateway should 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 Protocol	There are two modes <i>UDP</i> and <i>TCP</i> available for running the SIP protocol. The 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.					
IMS Network	Once this feature is enabled, the gateway will send signaling messages to the corresponding externally bound address and port when it registers to the server. By default, this feature is <i>disabled</i> . Only when this feature is <i>enabled</i> will these items <i>Externally Bound Address, Externally Bound Port</i> and <i>Authentication Username</i> be shown.					
Externally Bound Address	Externally bound IP address for registration.					
Externally Bound Port	Externally bound port for registration.					

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 Compa	atibility
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
Call Flash Mode	Platform to Handle SIP I
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	Reset

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					
	the default setting of <i>disabled</i> . As it is disabled, if the Contact field indicates an IP					
Use Contact	address within the LAN, the request message will be sent according to the source					
Address	address; if the Contact field indicates an IP address belonging to the WAN, the					
	request message will be sent according to this IP address.					
	There are two optional ways to deal with call transfer: Internal Handling and					
Call Transfer Mode	Platform to Handle SIP Info. The default value is Internal Handling.					
	There are two optional ways to deal with call flash: Internal Handling and Platform to					
Call Flash Mode	Handle SIP Info. The default value is Internal Handling.					
	Sets the source of the hold music, with the default value of <i>Remote</i> , This feature					
Hold Music Source	gets valid only when you choose the mode <i>Platform to Handle SIP Info</i> .					
Two Stage Dialing	Once this facture is eachlad, the increasing call faces OID should perform the two					
for SIP Incoming	Once this feature is enabled, the incoming call from SIP should perform the two					
Call	stage dialing operation. By default this feature is disabled.					
	Sets the maximum time for the SIP channel to wait for the answer from the called					
Maximum Wait	party of the outgoing call it initiates. If the call is not answered within the specified					
Answer Time	time period, it will be canceled by the channel automatically. The default value is 60,					
	calculated by s.					
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.					
Sot SID Idontifying	Sets the SIP identifying content in the SIP call message. The default setting is					
Set SIP Identifying	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 15, 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	The interval of sending a heartbeat packet to detect the master registrar server					
Detection Cycle	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 disabled.					
Encryption Criterion	The criterion used to encrypt the SIP signal. At present only VOS1.1 is supported.					
lele máifie r	The identifier field of the VOS encryption, which is used to obtain the key of the SIP					
Identifier	encryption.					
Кеу	The key to encrypt the SIP signal.					
	Once this feature is enabled, you can encrypt the RTP package. By default it is					
RTP Encryption	disabled.					
	Once this feature is enabled, the gateway is not necessary to wait for the ACK					
Ignore ACK	message after sending the 200OK message to establish a call. By default it is					
	disabled.					
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> .					
	Once this feature is enabled, only the calls from the SIP registration server, the					
Use Iptables	source IP address of the route IP->TEL and these IP addressed set in Access					
	Control 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.

Operation Info	*	
Quick Config	*	
S VolP	*	No available SIP Station!
SIP		Add New
Sip Compatibili	ty	
SIP Station		
SIP Server		
NAT Setting		
Media		

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	SIP Station
Number:	0
Username:	
Password:	
Bound Port:	9
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	1	User	name	IP Ad	dress	Boun	d Port	Register Status	Register Duration (s)	Voice Channel State	Description	Modify
	0		13	23	8	-	1	9	Unregistered	-	1	default	
Check All	Unch	eck All		Inverse	E	Delete		Clear All					Add New
1 Item Total	20 Items/Pa	age 1/	/1 Fi	rst Previou	s Ne	d Last Go	to Pag	e 1 🖵 1 P	ages Total				

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.



S	IP Station
Number:	0
Username:	123
Password:	•••
Bound Port:	9
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.

peration Info	*	
Quick Config	*	
🛱 VolP	*	No Available Registrar Server!
SIP		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			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 Settings	
Local NAT Traversal		
Method 1:		
	Auto Nat	Enable PMP
	Outer Network Address	Offline
Method 2:		
	STUN Server	🗹 Enable
	NAT Type	Unknown
	STUN Server Address	127.0.0.1
Method 3:		
	Mapping Contact IP	
	Mapping SDP IP	
Method 4:		
	Rport	Enable
	Auto Detect NAT IP	Enable
Help Remote Device	Complete NAT Traversal	
	RTP Self-adaption	Enable
"Local NA" "Auto Nat" "Mapping "Mapping	rofessional person please do not modify the configura T Traversal": Please select one method according to yo It is required to enable the feature of upon or pmp for Contact IP": It is required to set the router to map the SI SDP IP": It is required to set the router to map the RTP ect NAT IP": It is valid only when the feature "Rport" is er ort range to the gateway.	our current network environment. the router. IP port to the gateway. port range to the gateway.

Figure 3-25 NAT Setting Interface

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

Item	Description
Acces No.	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			
STON Server	server is disabled.			
	Detected NAT (Network Address Translation) type. The gateway will return the NAT			
	type automatically in case STUN Server is enabled. It includes 9 types: unknown;			
NAT Type	no NAT; ConeNat; RestrictedNat; PortRestrictedNat; Symmetric NAT; Symmetric			
	NAT with firewall; can't detect over (fail to send detect message) and fail to detect			
	(No reply from the stun server).			
STUN Server				
Address	Address of the server for STUN traversal.			
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.			
Draw	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			
Auto Delect NAT 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 Solf 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	0		
	Maximum Gain	Threshold (dB)	48	48		
	Maximum Atten	uation Threshold (dB)	0	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 💙 G723 🗸	20 💙	8 × 6.3 ×		
	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>3.9.16 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> .
	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
JitterBuffer	as an increased voice delay, while a smaller JitterBuffer means a lower jitter
JitterBuiler	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	automatically adjust the input signal amplitude, increasing that of small signals and
	decreasing that of large signals.
Target Energy	
Target Energy Threshold	decreasing that of large signals.
	decreasing that of large signals. Set the target energy of the AGC, range of value: -50~0, calculated by dB, with the
Threshold	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.
Threshold Maximum Gain	 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.
Threshold Maximum Gain Threshold	 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:
Threshold Maximum Gain Threshold Maximum	 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.
Threshold Maximum Gain Threshold Maximum Attenuation Threshold	 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:
Threshold Maximum Gain Threshold Maximum Attenuation	 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 CODECs and their corresponding priority for the IP end to establish a		
CODEC Priority	call conversation. The table below explains the sub-items:		
	Sub-item	De	scription
	Priority	Priority for choosing the CO smaller the value is, the high	DEC in an SIP conversation. The er the priority will be.
	CODEC	Three optional CODECs G729A/B, G723, G722, AMR	are supported: <i>G711A</i> , <i>G711U</i> , and <i>iLBC</i> .
	Packing Time	Time interval for packing an RTP packet, calculated by ms.	
	Bit Rate	The number of thousand bits (excluding the packet header) that are conveyed per second.	
	By default, all of the seven CODECs are supported and ordered G711A, G711U, G729A/B, G723, G722, AMR and iLBC by priority from high to low.		
	The packing time and bit rate supported by different CODECs are listed in the table		
	below. Those values in bold face are the default values.		
	COEDC	Packing Time (ms)	Bit Rate (kbps)
	G711A	10 / 20 / 30 / 40 / 60	64
	G711U	5 / 10 / 20 / 30 / 40 / 50 / 60	64
	G729A/B	10 / 20	8
	G723	30 / 60	5.3 / 6.3
	G722	5 / 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.
	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 <i>after 1st Ring</i> .
Sand Delavity	Once this feature is enabled, the gateway will send the polarity reversal signal to a
Send Polarity	corresponding FXS channel when it detects the called party pick-up behavior. By
Reversal Signal	default, this feature is disabled.
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
Internal Station	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	

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>3.9.16 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 Minimum Numbers of FSK Mark Bits on off	40
Minimum Numbers of FSK Mark Bits on off- hook	50
Reception Interval of DTMF CallerID (ms)	250
Outgoing Call to PSTN	
Flash Time (ms)	100
Delay after Dial (ms)	1000
FXO Pick-up Delay after INVITE Received at IP Side (s Maximum Wait Answer Time (s)(Valid when Polarity) 0
Reversal Enabled)	25
Communicate without Network	Enable
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 🔹
Tone Detector Prior to DTMF Detector	✓Enable

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 10.		



	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
	conditions are satisfied. The default setting is being disabled.
	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.
T	The time threshold to judge whether the line is silent or not, calculated by s, with the
Time Threshold of	default value of 60.
Silence	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	
Threshold for FSK	Sets the minimum energy threshold for the FSK detection. Range of value: -1125~0,
Detection	calculated by 0.1dB, with the default value of -380.
	The FSK seizure bits (0x55) under the on-hook mode. As the protocol provides, it
Minimum Number of	shall be 300 consecutive bit groups of 0 and 1. Range of value: 0~32768, with the
FSK Seizure Bits	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 <i>50</i> .	
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.
	Sets the time for generating a flash signal on the analog trunk. Range of value:
Flash Time	32~1000, calculated by ms, with the default value of 100.
	Sets the delay to send the CalleeID to PBX after you pick up and dial. Range of
Delay after Dial	value: 200~2000, calculated by ms, with the default value of 1000.



FXO Pick-up Delay after INVITE Received at IP Side Maximum Wait Answer Time	Once this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message. The 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
Communication without Network	by s, with the default value of 25. Automatically 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 Network Mode	Sets the mode for the communications without network, two options available: Auto Search Idle Channel and Use Current Route Setting, with the default value of <i>Auto Search Idle Channel</i> . 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 Mode	Sets 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 Side	Once 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 Advance	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 <i>enabled</i> .
Avoid Being Detected as Flash Signal by PBX	Once 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 value is <i>disabled</i> .
Work Mode	Sets the work mode for the echo canceller. There are two options: <i>Near-end cancellation</i> and <i>Both near-end and far-end cancellation</i> , with the default value of <i>Near-end cancellation</i> .
Non-linear	Sets whether to enable the mode of non-linear processing. By default, this feature is
Processing Fixed Window Size	enabled. Sets the size of the window for the fixed cancellation.
Moving Window Size	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.
Tone Detector Prior to DTMF Detector	Sets the priorities of the Tone detector and the DTMF detector.

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>3.9.16 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	Ringback Tone	Periodic Tone	450	0	1000	4000	1	0	2
Check A		Uncheck All	Inverse 🚊 D	elete 🗄 🗄 Clear All	1				A	Id New

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

	1	Fone 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 Detector	
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 Genera DTMF Energy (dB)	ator 0
	vo.vo.
	Minimum Energy Threshold (dB) Maximum Threshold of Signal Twist (dB) Input Signal Gain (dB) Voice Path Delay (ms) Minimum Duration at ON (ms)

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.
Innut Signal Cain	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.
Valas Dath Datas	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>3.9.16 Restart</u> for detailed instructions. Click **Reset** to restore the configurations.

3.5.6 Ringing Scheme

Ringing Scheme		
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.

|--|



	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 Parameters		
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>3.9.16 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 Parameters	
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 💌
Service Available			
Blind Transfer		*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

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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]000000	default	
	84	800xxxxxxxx	default	
	85	80[1-9]xxxx	default	
	86	8[1-9]xxxxxxx	default	
	87	[2-3,5-7]0000000	default	
	88	1[3-5,7-8]x000000000	default	
	89	100xx	default	
	90	95xxx	default	
	91	123xx	default	
	92	111xx	default	
	93	11[0,2-9]	default	
	94	120	default	
	95	0[3-9]000000000	default	
	96	02хооооооо	default	
	97	010x0000000	default	
	98	01[3-5,7-8]xxxxxxxxxxx	default	
	99		default	

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

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.



Dialiı	ng 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.

ltem		Description
Indox	The unique ind	ex of each dialing rule, which denotes its priority. A dialing rule with a
Index	smaller index v	alue has a higher priority and will be checked earlier while matching.
Description	Remarks for th	e dialing rule. It can be any information, but can not be left empty.
	Up to 100 dialir	ng rules can be configured in the gateway, and the maximum length of
	each dialing ru	le is 127 characters. See below for the meaning of each character in
	the dialing rule	. The gateway will do instant matching for your dialing number based
	on the dialing	rule and regard your dialing as finished upon receiving '#' or dialing
	timeout.	,
	Character	Description
	"0"~"9"	Digits 0 \sim 9.
	"A"~"D"	Letters A~D.
	"x"	A random number. A string of 'x's represents several random
	^	numbers. For example, 'xxx' denotes 3 random numbers.
	"" •	· indicates a random amount (including zero) of characters after it.
Dialing Rule		'[]' is used to define the range for a number. Values within it only
	"[]"	can be digits '0~9', punctuations '-' and ','. For example,
	-	[1-3,6,8] indicates any one of the numbers 1, 2, 3, 6, 8.
	"_"	'-' is used only in '[]' between two numbers to indicates any number between these two numbers.
	(; 1) 3	',' is used to separate numbers or number ranges, representing alternatives.
		Only represents symbol "*".
	"#"	Only set it at the beginning of the string, representing symbol "#".
	There are 19	dialing rules already configured on the gateway for easy use. See
		ed information.
	Priority	Dialing Rule Description



$\begin{array}{c c c c c c c c c c c c c c c c c c c $
$\begin{array}{ c c c c c c c c c c c c c c c c c c c$
96 $02xxxxxxx$ Any 11-digit number starting with 0295 $0[3-9]xxxxxxx$ Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 0994120Number 120.93 $11[0,2-9]$ Number 110, 112, 113, 114, 115, 116, 117, 118 or 11992111xxAny 5-digit number starting with 1239095xxxAny 5-digit number starting with 9589100xxAny 5-digit number starting with 10088 $1[3-5,7-8]xxxxxxx$ Any 11-digit number starting with 13, 14, 15, 16, 17, 15, 17 or 1887 $[2-3,5-7]xxxxxxx$ Any 8-digit number starting with 2, 3, 5, 6
95 0[3-9]xxxxxxx Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09 94 120 Number 120. 93 11[0,2-9] Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 92 111xx Any 5-digit number starting with 111 91 123xx Any 5-digit number starting with 123 90 95xxx Any 5-digit number starting with 95 89 100xx Any 5-digit number starting with 100 88 1[3-5,7-8]xxxxxxx Any 11-digit number starting with 13, 14, 15, 17 or 18 87 [2-3,5-7]xxxxxxx Any 8-digit number starting with 2, 3, 5, 6
95 0[3-9]xxxxxx 05, 06, 07, 08 or 09 94 120 Number 120. 93 11[0,2-9] Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 92 111xx Any 5-digit number starting with 111 91 123xx Any 5-digit number starting with 123 90 95xxx Any 5-digit number starting with 95 89 100xx Any 5-digit number starting with 100 88 1[3-5,7-8]xxxxxxx Any 11-digit number starting with 13, 14, 15, 17 or 18 87 [2-3,5-7]xxxxxxx Any 8-digit number starting with 2, 3, 5, 6
93 11[0,2-9] Number 110, 112, 113, 114, 115, 116, 117, 118 or 119 92 111xx Any 5-digit number starting with 111 91 123xx Any 5-digit number starting with 123 90 95xxx Any 5-digit number starting with 95 89 100xx Any 5-digit number starting with 100 88 1[3-5,7-8]xxxxxxx Any 11-digit number starting with 13, 14, 15, 17 or 18 87 [2-3,5-7]xxxxxxx Any 8-digit number starting with 2, 3, 5, 6
9311[0,2-9]118 or 11992111xxAny 5-digit number starting with 11191123xxAny 5-digit number starting with 1239095xxxAny 5-digit number starting with 9589100xxAny 5-digit number starting with 100881[3-5,7-8]xxxxxxxAny 11-digit number starting with 13, 14, 15, 17 or 1887[2-3,5-7]xxxxxxAny 8-digit number starting with 2, 3, 5, 6
91123xxAny 5-digit number starting with 1239095xxxAny 5-digit number starting with 9589100xxAny 5-digit number starting with 100881[3-5,7-8]xxxxxxxAny 11-digit number starting with 13, 14, 15, 17 or 1887[2-3,5-7]xxxxxxAny 8-digit number starting with 2, 3, 5, 6
9095xxxAny 5-digit number starting with 9589100xxAny 5-digit number starting with 100881[3-5,7-8]xxxxxxxAny 11-digit number starting with 13, 14, 15, 17 or 1887[2-3,5-7]xxxxxxAny 8-digit number starting with 2, 3, 5, 6
89100xxAny 5-digit number starting with 100881[3-5,7-8]xxxxxxxAny 11-digit number starting with 13, 14, 15, 17 or 1887[2-3,5-7]xxxxxxxAny 8-digit number starting with 2, 3, 5, 6
881[3-5,7-8]xxxxxxxAny 11-digit number starting with 13, 14, 15, 17 or 1887[2-3,5-7]xxxxxxxAny 8-digit number starting with 2, 3, 5, 6
88 1[3-5,7-8]xxxxxxx 15, 17 or 18 Any 8-digit number starting with 2, 3, 5, 6 87 [2-3,5-7]xxxxxx
87 [2-3.5-7]xxxxxx
OI 7
86 8[1-9]xxxxx Any 8-digit number starting with 81, 82, 83, 84, 85, 86, 87, 88 or 89
85 80[1-9]xxxxx Any 8-digit number starting with 801, 802, 803, 804, 805, 806, 807, 808 or 809
84 800xxxxxx Any 10-digit number starting with 800
83 4[1-9]xxxxx Any 8-digit number starting with 41, 42, 43, 44, 45, 46, 47, 48 or 49.
82 40[1-9]xxxx 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.



Diali	ng 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 strin Don't forget to save the configuration after your modification!	y.
400xxxxxx default	<u>^</u>
40[1-9]xxxxx default	
4[1-9]xxxxxx default	
800xxxxxx default	
80[1-9]xxxxx default	
8[1-9]xxxxxx default	
[2-3,5-7]xxxxxxx default	
1[3-5,7-8]xxxxxxxx default	
100xx default	
95xxx default	
123xx default	
111xx default	
11[0,2-9] default	
120 default	
0[3-9]xxxxxxxx default	~
20 Items Total	

Figure 3-42 Dialing Rule Configuration Interface (Character)

3.5.10 Dialing Timeout

Dialing	Timeout 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
	include ".", the call will fail if there is no digit dialed or no dialing rule matched during
Inter Digit Timeout	this interval; in case your dialing rules include ".", the gateway will wait until this
	interval ends and match to the dialing rule "." if there is no digit dialed or no other
	dialing rule matched during this interval.
	More information about the configuration item Inter Digit Timeout, 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 a file of cue	File of cue tone for IVR	Browse Upload
tone		

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		
tone	Uploads 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	×
📸 VolP	*
	*
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.

Description		default	
Color Ring			Browse
Note: The file sh 200KB in size.	ould be a wav file v	with 8000Hz sampling rate, 16-bit	t mono, A-law formatted, and less than
200100 111 5120.			

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 Uncheck All	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 three parts: FXS, FXO and Port Group. See Figure 3-52.

🚺 Port	۶
FXS	
FXO	
Port Group	

Figure 3-52 Port Settings

3.6.1 FXS

FXS Settings																		
Port	Type	SIP Account	Display Name	Auto Dial Num	Forbid Outgoing Call	DND	Forward	FWD Type	FWD Number	CID	Call Waiting	Reg Status	Echo Canceller	Color Ring	Color Ring Index	Input Gain	Output Gain	Modif
1	FXS	8001			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	12
2	FXS	8002		-	Disable	Disable	Disable	<u> </u>	-	Enable	Disable	Unregistered	Enable	Disable	A (0	0	0
3	FXS	8003			Disable	Disable	Disable	<u>21</u>	-	Enable	Disable	Unregistered	Enable	Disable		0	0	12
4	FXS	8004		-	Disable	Disable	Disable	-	-	Enable	Disable	Unregistered	Enable	Disable		0	0	2
6	FXS	8005			Disable	Disable	Disable	-	-	Enable	Disate	Unregistered	Enable	Disable	÷	0	0	2
6	FX8	8006			Disable	Disable	Disable		(11)	Enable	Disable	Unregistered	Enable	Disable) + (0	0	0
7	FXS	8007		1	Disable	Disable	Disable	<u>V 1</u>	-	Enable	Disable	Unregistered	Enable	Disable	Ш.	0	0	12
8	FXS	8008		-	Disable	Disable	Disable	-		Enable	Disable	Unregistered	Enable	Disable		0	0	G
(I												. 101						

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.

Port	1
Туре	FXS
Register Port	Yes
SIP Account	8001
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)	0
Input Gain (dB)	0
Output Gain (dB)	0
Echo Canceller	Enable
Forbid Outgoing Call	Enable
CID	Enable
Call Waiting	Enable
DND (Do Not Disturb)	Enable
Call Forward	Enable
Forward Type	Unconditional 📼
Forward Number	
Color Ring	Enable
Color Ring Index	1
Advanced Configuration	IVEnable
Talkback	Enable
Bound Number	
ta: 'Auto Dial Number' goas into effect only	if no dialing occurs during 'Wait Time before Auto Dia

Figure 3-54 FXS Modification

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

Item	Description			
Port Serial number of the FXS port on the device.				
Туре	Type of the port on the device (FXS). This item is not configurable.			



	Sets whether to register the port to the SIP server.
Register Port	When this item is set to No, the item Reg Status on the FXS settings interface
-	(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.
SIP Account	The default SIP account is 80XX among which XX represents the corresponding
	port number. For example, the default SIP account corresponding to Port 1 is 8001,
	and that corresponding to Port 8 is 8008.
Display Name	Set the content of the displayname field of the SIP message. If it doesn't set with
	any value, the displayname field will by default display the content of callerid.
Password	Registration password of the port. To register a port to the SIP server, both items
1 435 WOLU	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
	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 <i>Auto Dial Number</i> 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
	incoming IP call together with the ringing tone to the corresponding station. The
CID	default setting is <i>enabled</i> . 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	station hangs up, the call in the waiting state will ring the station and wait for
	answer. The default setting is <i>disabled</i> .
	Do Not Disturb. If this feature is enabled, the FXS port will reply the 403 message to
DND	reject all incoming calls. The default setting is <i>disabled</i> .
	rejeor an inconting cans. The deladit setting is disabled.



	The automatic c	all forward feature for the FXS port. Once this feature is enabled,			
o " = i	the FXS port will forward incoming IP calls according to FWD Type. Note: To				
Call Forward	enable this featu	ire, 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			
		Num if the corresponding station does not answer them in a			
	No Reply	designated time period (i.e. Time for No Reply Forward). Only			
		when this forward condition is selected does the configuration			
		item Time for No Reply Forward become valid.			
	This item is valid	l only when <i>Call Forward</i> is set to <i>Enable</i> .			
	The number to which the incoming IP call is forwarded. If the Call Forward feature				
FWD Num	is enabled, this item can not be left empty.				
	Sets whether to	enable the color ring feature or not, with the default setting of being			
Color Ring	disabled.				
	Note: Only when there are available color rings will this item appear.				
Color Ring Index	The index of the	color ring which will be quoted by the current FXS port.			
	With this feature enabled and a number bound, the port can talkback to its bound				
T = 11 + 1 = 1	number. That is	, they can start a call with each other as soon as picking up the			
Talkback	phone. The defa	ult setting is disabled.			
	Note: This featu	ure is only used in the case of channel registration.			
Bound Number	Sets the bound r	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	Type	SIP Account	Display Name	Connection Method	Bound Number	Ferbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	Polarity Reversal Detection	Input Gain at Offhook	Output Gain at Offhook	Input Gain at Onhook	Output Gain at Onhook	Mode
9	FX0	8009	-	Two Stages Dialing for Incoming Call	(m)	Disatie	Enable	Unregistered	Enable	Disable	0	0	0	0	1
10	#X0	8010	-	Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	Disable	0	0	0	0	2
11	FXD	8011	-	Two Stages Dialing for incoming Call		Disable	Enable	Unregistered	Enable	Disable	0	0	02	(Ø)	2
12	FXQ.	8012	-	Two Stages Dialing for Incoming Call	-	Disatle	Enable	Unregistered	Enable	Disable	0	0	0	0.0	2
13	FX0	8013	-	Two Stages Dialing for Incoming Call	1000	Disable	Enable	Unregistered	Enable	Disable	0	0	0	0	02
14	FxO	8014	5	Two Stages Dialing for Incoming Call	-	Disable	Enable	Unregistered	Enable	Disable	a	0	0	0	2
15	FXD	8015	-	Two Stages Dialing for incoming Call		Disatie	Enable	Unregistered	Enable	Disable	0	0	0	0	R
16	810	8016	-	Two Stages Dialing for incoming Call		Disable	Enable	Unregistered	Enable	Disable	0	0	0	0	2

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.



FXO	Modify
Port	9
Туре	FXO
Register Port	Yes
SIP Account	8009
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
	Type Register Port SIP Account Display Name Password Authentication Username Display Name Preferred Server Index Connection Method Input Gain at Offhook(dB) Output Gain at Offhook(dB) Input Gain at Onhook(dB) Output Gain at Onhook(dB) Echo Canceller Forbid Outgoing Call Caller ID Detection

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.
Deviator Devt	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.
Deserver	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 wh	nen				
Authentication	IMS network is enabled.					
Username	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 p	ort				
	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 t	the				
Preferred	sent SIP message will be the display name set by the port group; if the who	ole				
	gateway is registered, the displayname in the sent SIP message will be t	the				
	displayname of the gateway.					
Server Index	The index of the SIP server which will be quoted by the current FXO port.					
	FXO connection methods include:					
	Option Description					
	Bind the number which corresponds to an FXS port to an FXC	0				
	Static port. The number will be listed in the Bound Number column. Thi	is -				
	Binding helps to achieve the corresponding binding between an FXO pol	rt				
	and an FXS port (two-way).					
Connection Method	Under this mode, an incoming call from an FXO port will go int	o				
Connection method	the IVR system. Then IVR will play a speech prompt "Please dia	al				
	Stages the extension number". If you fail to input the correct target statio	n :				
	number before IVR finishes the third repeat of the prompt, th	e				
	FXO will hang up the call automatically; otherwise, th	e				
	(default) 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,	Adjusts the gain of the voice input to/ output from the FXO port when it is offhook	or				
Output Gain at	onhook. Range of value: -24~12, calculated by dB, with the default value of 0.					
Offhook/Onhook						
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 defa	ault				
Call	setting is <i>disabled</i> .					
Caller ID Detection	If this feature is enabled, the FXO port will detect the caller IDs from the incomi	ing				
	calls. The default setting is <i>enabled.</i>					
	Once this feature is enabled, only when the FXO port detects the polarity rever					
Polarity Reversal	signal will the corresponding channel go into the talking state. The default setting	-				
Detection	disabled. Note: This feature and the Two Stages Dialing 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).

Chading Dad		
Starting Port	9	
Ending Port	16	•
Register Port	Yes	•
Starting SIP Account		
Starting Display Name		
Starting Authentication Password		
Starting Authentication Username		
Display Name Preferred	Enable	
Server Index	1:201.123.115.2	33 🔻
SIP Account Batch Rule	Increase	-
SIP Account Batch Step Size	1	
Display Name Batch Rule	Increase	-
Display Name Batch Step Size	1	hanna
Authentication Password Batch Rule	Increase	-
Authentication Password Batch Step Size	1	- Contract
Authentication Username Batch Rule	Increase	-
Authentication Username Batch Step Size	1	Luna
Connection Method	Two Stages Dia	ing 🔻
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.



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 Port Group

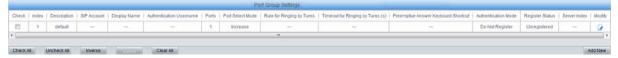


Figure 3-59 Port Group Settings Interface

See Figure 3-59 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 3-60 for the port group adding interface. Note that a port which has been occupied by one port group cannot be chosen by others.



201220	
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(FXS) Port 6(FXS) Port 7(FXS) Port 8(FXS) Port 9(FXO) Port 10(FXO) Port 11(FXO) Port 12(FXO)
	Port 13(FXO) Port 14(FXO) Port 15(FXO) Port 16(FXO)
	Check All Inverse Check All FXO Ports Check All FXS Ports

Figure 3-60 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
mdex	routing rules and number manipulation rules to correspond to port groups.
Description	More information about each port group, with default value of default.
Dania (an Dani Onam	To register the port group to the SIP server. Only when this configuration item is set
Register Port Group	to Yes can you see the configuration items SIP Account and Password.
	When the port group initiates a call to SIP, this item corresponds to the username of
SIP Account	SIP.
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 group. To register the port group to the SIP server,
Password	both configuration items SIP Account and Password should be filled in.



	Authentication usernam	Authentication username of a port, used to register the port to the SIP server when				
Authentication	IMS network is enabled	d.				
Username	Note: This item appe	ars only when IMS Network or Multi-Registrar Server is				
	enabled.	enabled.				
Server Index	The index of the sip se	rver 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				
Authentication Mode	Do Not Register (default)	SIP initiates a call in a point-to-point mode.				
	Register Gateway	SIP initiates a call with the registered SIP account and password of the whole gateway. (Refer to <u>3.4.1 SIP</u> for gateway registration.)				
	Register Port Group	SIP initiates a call with the registered SIP account and password of the port group.				
	Register Port	SIP initiates a call with the registered SIP account and password of the port.				
	Registration status of t	he port group. When <i>Register Port Group</i> is set to <i>No</i> , the				
Register Status	value of this item is U	Inregistered; when Register Port Group is set to Yes, the				
	value of this item may l	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 meanir	ngs 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.		
Port Select Mode		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,		
		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 3-60. 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 Shortcut	Note: This item will be	ecome invalid if the gateway works under the port select mode		
	Group Ringing or Ring	ging by Turns.		
Port Reused by	Once this feature is er	nabled, a port can be added to different port groups.		
Multiple Groups		· · · · ·		
		roup. If the checkbox before a port is grey, it indicates that the		
	-	or has been occupied. Once the feature "Port Reused by		
Port		nabled, a port which has been occupied is still available for		
		selected ports for a port group will be displayed in the <i>Ports</i>		
	_	59. 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 3-61 for the port group modification interface. The configuration items on this interface are the same as those on the *Add New Port Group* interface.

Index	1
Description	default
Register Port	Yes
SIP Account	
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(FXS) Port 6(FXS) Port 7(FXS) Port 8(FXS)
	Port 9(FXO) Port 10(FXO) Port 11(FXO) Port 12(FXO)
	Port 13(FXO) Port 14(FXO) Port 15(FXO) Port 16(FXO)
	Check All Inverse Check All FXO Ports Check All FXS Ports

Figure 3-61 Modify Port Group

To delete a port group, check the checkbox before the corresponding index in Figure 3-59 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 3-59.

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 3-62.



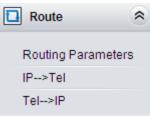


Figure 3-62 Route Settings

3.7.1 Routing Parameters

> TEL	Route before Number Manipulate 🛛 👻
-> IP	Route before Number Manipulate

Figure 3-63 Routing Parameters Configuration Interface

See Figure 3-63 for the routing parameters configuration interface. On this interface, you can set the routing rules for calls respectively on two directions $IP \rightarrow Tel$ and $Tel \rightarrow IP$ to be routing before or after number manipulation. The default value is *Route before Number Manipulate*.

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

3.7.2 IP to Tel

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

Figure 3-64 IP→Tel Routing Rule Configuration Interface (Standard)

See Figure 3-64 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 3-65. You may use the default values of all the configuration items herein.



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

Figure 3-65 Add New Routing Rule (IP→Tel)

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

Item	Description
	The unique index of each routing rule, which denotes its priority. A routing rule with
Index	a smaller index value has a higher priority. If a call matches several routing rules, it
	will be processed according to the one with the highest priority.
Description	More information about each routing rule, with the default value of <i>default</i> .
Courses /D	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
Route by Number	will be routed to can be set via the item <i>SIP Account</i> on the <u>FXS</u> or <u>FXO</u> . Settings
Noule 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 disabled.



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

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

See Figure 3-66 for the IP→Tel routing rule configuration interface after your configuration. There is a rule displayed with Index 63 and Call Destination 'Route by Number', having no restriction on Source IP, CallerID Prefix and CalleeID Prefix, which indicates the gateway will route a call from any IP address to a corresponding port based on its number.

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

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

Figure 3-66 IP→Tel Routing Rule Configuration Interface

Click **Modify** in Figure 3-66 to modify a routing rule. The configuration items on the $IP \rightarrow Tel$ routing rule modification interface are the same as those on the **Add New Routing Rule** ($IP \rightarrow Tel$) interface. Note that the item **Index** cannot be modified.

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

See Figure 3-67 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.

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



3.7.3 Tel to IP

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

Figure 3-68 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-68 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 3-69. You may use the default values of all the configuration items herein except for *Destination IP* and *Destination Port*.

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

Figure 3-69 Add New Routing Rule (Tel→IP)

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

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



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

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

See Figure 3-70 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			
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Destination IP	Destination Port	Description	Modify
	63	*	*	(* .)	192.168.1.101	5060	default	

Figure 3-70 Tel→IP Routing Rule Configuration Interface

Click **Modify** in Figure 3-70 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 3-70 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 3-70.

See Figure 3-71 for the Tel \rightarrow IP Routing Rule Configuration Interface under the Character mode. You can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.



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

Figure 3-71 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 3-72.



Figure 3-72 Number Manipulation

3.8.1 IP to Tel CallerID

					IP->Tel Calle	arlD 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	*	ż	ż	0	0	20			default	2
							b				

Figure 3-73 IP→Tel CallerID Manipulation Interface (Standard)

See Figure 3-73 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 3-74 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	D
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 3-74 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.
Oficiana d Distitution	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 3-73 to modify a number manipulation rule. See Figure 3-75 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 3-75 Modify IP→Tel CallerID Manipulation Rule

To delete a number manipulation 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 number manipulation rules at a time, click the **Clear All** button in Figure 3-73.

See Figure 3-76 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 3-76 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 3-78 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 3-73).

					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 3-77 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
1ltems Total
Save



Figure 3-78 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 3-79 Tel→IP CallerID Manipulation Interface (Standard)

See Figure 3-79 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 3-80 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 3-80 Add Tel→IP CallerID Manipulation Rule

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

	Item	Description
	la dev	The unique index of each number manipulation rule, which denotes its priority. A
'	ndex	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				
	<i>Initiator</i> specify a number manipulation rule for calls.				
	Note: "[*]" represents DTFM symbol *, while "*" represents any string.				
Chrispered Distite 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.				
Chrispered Distite 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 3-79 to modify a number manipulation rule. See Figure 3-81 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 3-81 Modify Tel→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-79 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 3-79.

See Figure 3-82 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 3-82 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 3-83, Figure 3-84 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 3-79).

					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	*	ż	ż	0	0	20			default	2

Figure 3-83 Tel→IP CalleeID Manipulation Interface (Standard)

Standard Mode Character Mode
Tel->IP CalleelD 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 * (a 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



Figure 3-84 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 3-85 for details.

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

Figure 3-85 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	OYes 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 3-86 Management Parameters Setting Interface

See Figure 3-86 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.
NTP Server Address	Sets the Server address for NTP time synchronization. By default, the address is
NTP Server Address	time.nist.gov
Currenterentering Currente	Sets the cycle for NTP time synchronization, calculated by s, with the default value
Synchronizing Cycle	of 3600.
Daily Dactart	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.
Custom Time	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 Network

Network S	ettings
Network Type:	Static
IP Address (I):	201.123.115.215
Subnet Mask (U):	255.255.255.0
Default Gateway (D):	201.123.115.254
DNS Server (P):	202.101.172.35
Speed and Duplex Mode:	Automatic Detection
Save	Reset
Note: Please log in again using your new IP ad	dress if the IP address has been modified!

Figure 3-87 Network Settings Interface

See Figure 3-87 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.



3.9.3 Upgrade

	Current Version
Serial Num	2230
WEB	Version 1.7.0_2017032317
Service	Version 1.7.0_2017032317
U-boot	Version Jan 04 2017-01:59:38
Kernel	Version #209 Wed Mar 22 10:49:54 CST 2017
Product Type	1016C-8S8O(RJ11)
Select an U	pdate File Browse
	Update Reset

Figure 3-88 Upgrade Interface

See Figure 3-88 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 3-89.



		Current Version	
Serial Num	2230		
WEB	Version 1.	7.0_2017032317	
Service	Version 1.	7.0_2017032317	
U-boot	Version Ja	an 04 2017-01:59:38	
Kernel	Version #2	209 Wed Mar 22 10:49:54 CST 2017	
Product Type	1016C-8S	80(RJ11)	
Select an Up	odate File	C:\Users\Administrator Browse	
		37% 3204kb/s	
		31% 3204KD/S	
The file is a	Inloading	g. Please do not leave this page!	
The file is t	apioading	g. Thease do not leave this page	
		pgrade Information	
start upload	l upgrade fi	le *	

Figure 3-89 File Uploading Interface

After a successful uploading of the file, the gateway will start to upgrade the system. See Figure 3-90 and you can learn the detailed upgrading information from the upgrade information box at the bottom.



	Current Version	
Serial Num	2230	
WEB	Version 1.7.0_2017032317	
Service	Version 1.7.0_2017032317	
U-boot	Version Jan 04 2017-01:59:38	
Kernel	Version #209 Wed Mar 22 10:49:54 CST 2017	
Product Type	1016C-8S8O(RJ11)	
Select an U	odate File C:\Users\Administrator\Browse)	
	Update Reset	
	Upload completion!	
	12%	
System updat	ing, please do not leave this page!	
	Upgrade Information	
start upload	l upgrade file	*
decompres	sing the upgrade package	
analysing th	ne upgrade file	
		Ŧ

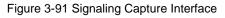
Figure 3-90 System Upgrading Interface

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

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

3.9.4 Signaling Capture

	Packet Capture		
Capture Data on All Network Cards	Enable		
Signaling Packet Capture	SIP&Syslog&Cent		
RTP Packet Capture	RTP Port Range 💌 50000,50767	Start	Stop
Note: Only latest 1M capture of	lata will be saved.		





See Figure 3-91 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.

3.9.5 Data Recording

Data Re	ecording
Channel	Channel1_(FXS)
Mode	Default
Interface	Мар
Note: 1.Only the latest 60s data of 2.Recording parameters:80 mono,U-law formatted.	Stop can be saved. 000HZ sampling rate,16-bit

Figure 3-92 Data Recording Interface

See Figure 3-92 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.

3.9.6 Call Log

Call Log SIP Log	Call Log	Download				
Call from IP Channel						Clear All
01/01/1970 22:58:05:313	IP Channel 1, Incoming call fro	m remote end "800	aller "8005" <sip:8005@201.123.11 5" <sip:8005@201.123.115.72>,ca pending state(Call from the IP side</sip:8005@201.123.115.72></sip:8005@201.123.11 	I-id: 2296587899@201.123.1	15.72 Caller 8005 Callee 5 call end, r)	eason:no idle chan
-						Þ
Call from Port	Select a Port	Port5 💌				Clear All
01/01/1970 22:58:04:577		nslation 5>5 mate	the dialing rule(129 . default) ch TEL->IP CalleeID Manipulate rul 05 match TEL->IP CallerID Manipu			

Figure 3-93 Call Log Interface



NP Log	Refresh Clear Al
/ia: SIP/2.0/UDP 201.123.115.72:5060;branch=z9hG4bK1796442218;rport	
From: "8005" <sip:8005@201.123.115.72>;tag=890297255</sip:8005@201.123.115.72>	
To: <sip:5@201.123.115.72></sip:5@201.123.115.72>	
Call-ID: 2296587899@201.123.115.72	
CSeq: 20 INVITE	
Contact: <sip:8005@201.123.115.72:5060></sip:8005@201.123.115.72:5060>	
Max-Forwards: 70	
User-Agent: Gateway	
Expires: 120	
X-callCause: Communicate without Network	
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO	
Content-Type: application/sdp	
Content-Length: 480	
v=0	
0=- 1 2 IN IP4 201.123.115.72	
s=Gateway	
c=IN IP4 201.123.115.72	
t=0 0	
m=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	
a=rtpmap:98 iLBC/8000	
a=rtpmap:100 silk/16000	

Figure 3-94 SIP Log Interface

See Figure 3-93, Figure 3-94 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**.

3.9.7 Operation Log

Operation Log	
2017-03-13 19:15:37 ClientlP:201.123.115.212 OpenCallLog	
2017-03-13 19:15:38 ClientlP:201.123.115.212 ClearTdmPortCallLog Port:7	
2017-03-13 19:15:44 ClientIP:201.123.115.212 OpenCallLog	
2017-03-13 19:15:45 ClientlP:201.123.115.212 ClearlPChannelCallLog	(E
2017-03-13 19:15:53 ClientlP:201.123.115.212 OpenCallLog	
2017-03-13 19:15:54 ClientlP:201.123.115.212 ClearTdmPortCallLog Port:6	
2017-03-13 19:16:02 ClientlP:201.123.115.212 OpenCallLog	
2017-03-13 19:16:02 ClientlP:201.123.115.212 ClearTdmPortCallLog Port:5	
2017-03-13 19:21:43 ClientlP:201.123.115.212 SaveSipCompatibility=global_calledidplace:1 global_calleridplace:0 global_getcalleridplace:0 RequestUs	eC
2017-03-13 19:23:41 ClientlP:201.123.115.212 SaveSipCompatibility=global_calledidplace:1 global_calleridplace:0 global_getcalleridplace:0 RequestUs	eC
2017-03-13 21:08:48 ClientlP:201.123.115.212 SaveSipCompatibility=global_calledidplace:1 global_calleridplace:0 global_getcalleridplace:0 RequestUs	eC
2017-03-15 13:14:19 ClientIP:201.123.115.212 ADD/MOD ROUTE_TEL2IP=63 0 1 ** 201.123.115.212 5060 default	
SetSyslog:1 SyslogIP:201.123.115.212 SyslogLevel:7 SetCDR:0 CDRIP:127.0.0.1 CDRPort:3 SetTime: Time: NTPZone: SetNTP:0 NTPIP:time.nist.gov NTI	PC
SetSyslog:0 SyslogIP:201.123.115.212 SyslogLevel:7 SetCDR:0 CDRIP:127.0.0.1 CDRPort:3 SetTime: Time: NTPZone: SetNTP:0 NTPIP:time.nist.gov NTI	PC
SetSyslog:1 SyslogIP:201.123.115.212 SyslogLevel:7 SetCDR:0 CDRIP:127.0.0.1 CDRPort:3 SetTime: Time: NTPZone: SetNTP:0 NTPIP:time.nist.gov NTI	PC
2017-03-15 15:04:06 ClientlP:201.123.115.212 ADD/MOD ROUTE_TEL2IP=63 0 1 * * 201.123.112.195 7746 default	
2017-03-15 17:14:53 ClientlP:201.123.115.212 DelAlIRouteRuleTEL->IP	
2017-03-15 17:18:28 ClientlP:201.123.115.212 SaveFaxPara=FaxMode:1 AddedFaxPort:0 T38FaxVersion: FaxSignalling:2 T38MaxBitRate:14400 T38FaxRa	ate
2017-03-15 17:21:42 ClientlP:201.123.115.212 SaveFaxPara=FaxMode:1 AddedFaxPort:0 T38FaxVersion: FaxSignalling:2 T38MaxBitRate:14400 T38FaxRa	ate
2017-03-15 17:22:16 ClientlP:201.123.115.212 PcapStart=SignalingPcap:1 RTPPcap:1 RTPPort:50000,50767	
2017-03-15 17:22:48 ClientlP:201.123.115.212 PcapStop	
2017-03-15 17:23:29 ClientlP:201.123.115.212 PcapStart=SignalingPcap:1 RTPPcap:1 RTPPort:50000,50767	
2017-03-15 17:25:10 ClientlP:201.123.115.212 PcapStop	
2017-03-15 18:15:55 ClientlP:201.123.115.212 ADD/MOD ROUTE_TEL2IP=63 0 1 * * 201.123.115.215 5060 default	
2017-03-15 18:44:07 ClientlP:201.123.115.212 AddPortGroup=group_index:1 group_desc:default group_register:0 group_registerselectmode:0 group_po	rts
2017-03-15 18:44:18 ClientlP:201.123.115.212 ADD/MOD ROUTE_IP2TEL=63 * * * 1 1 default	
2017-03-15 18:44:40 ClientlP:201.123.115.212 PcapStart=SignalingPcap:1 RTPPcap:1 RTPPort:50000,50767	
2047 02 45 40 46-40 OlionUD-204 422 445 242 Deep Olon	+
	1.00
Refresh Clear All Download	

Figure 3-95 Operation Log Interface



See Figure 3-95 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.

3.9.8 Backup & Upload

Backup
t.
Upload
1

Figure 3-96 Backup & Upload Interface

See Figure 3-96 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 configur	ation file, click the 'Backup' button to start.	Backup
	Data Upload	
To upload a configuration 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 3-97 Backup & Upload & Prompt Interface

Click *OK* on the prompt box (Figure 3-97) to upload the configuration file to the gateway. Now the prompt information 'System is rebooting, please do not leave this page' appears. See Figure 3-98. The gateway will overwrite the current configurations with the uploaded data after restart. Click *Cancel* to cancel this upload directly.



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	Data Backup	
To backup the configuration file, clic	k the 'Backup' button to start.	Backup
	Data Upload	
To upload a configuration file, select Config File	it and click the button 'Upload' to start.	Upload
Note: After you successfully	upload the configuration file, the gateway v	will restart automatically.
System is reboting	g. Please do not leave this page!	

Figure 3-98 Configuration File Uploading Interface

3.9.9 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 3-99 Factory Reset Interface

See Figure 3-99 for the factory reset interface. Click *Reset* to restore all configurations on the gateway to factory settings.

3.9.10 System Monitor

Do	itchdog: g Feeding Interval (s)		C Enable	
Do				
			5	
Au	tomatically Restart the Service	if Undetected:	Enable	
Th	reshold to Judge Heartbeat Lo	ss for Service(s):		60

Figure 3-100 System Monitor Configuration Interface



See Figure 3-100 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.

3.9.11 Centralized Manage

Centralized Manag	ge
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 3-101 Centralized Manage Setting Interface

See Figure 3-101 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	
Server Address	a domain name, valid only when DCMS is selected.	
	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.
Cotoway	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	Once this feature is enabled, you can lock the device according to the
Contected	corresponding parameters. This item is valid only when DCMS is selected.
	Once this feature is enabled, you are required to fill in the authorization code while
IP Address	modifying the information related to the IP address in the Network interface. This
	item is valid only when DCMS is selected.
	Once this feature is enabled, you are required to fill in the authorization code while
Registrar Server	modifying the address and port of the registrar server in the SIP Settings interface.
	This item is valid only when DCMS is selected.
	The status of the connection between the gateway and the centralized
Working Status	management server. This item is valid only when DCMS is selected.
Centralized	
Management	Set the centralized management protocol. It only supports SNMP currently.
Protocol	
	Set the version of SNMP, three options available: V1, V2 and V3, with the default
SNMP Version	value of V2. This item is valid only when Others is selected.
Maniferina Dauf	Monitoring Port for SNMP on the gateway. This item is valid only when Others is
Monitoring Port	selected.
Community String	Community string used for information acquisition.
Account	The account of SNMP, valid only when the SNMP version is set to V3.
	The grade of SNMP, three options available: Neither authenticated nor encrypted,
a <i>i</i>	Authenticated but not encrypted and Authenticated and encrypted, with the default
Grade	value of Neither authenticated nor encrypted. 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.

3.9.12 Access Control

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

Figure 3-102 Access Control List Interface

See Figure 3-102 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 3-103.

	Access Control Command
Index:	1
Command:	
	Close

Figure 3-103 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 3-102 to modify a command. See Figure 3-104 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 3-104 Access Control Command Modification Interface

To delete an Access Control Command, check the checkbox before the corresponding index in Figure 3-102 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 3-102.

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.



3.9.13 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 3-105 Ping Test Interface

See Figure 3-105 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	
	connection status between the gateway and the destination address.	

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



3.9.14 TRACERT Test

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

Figure 3-106 Tracert Test Interface

See Figure 3-106 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.



3.9.15 Change Password

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

Figure 3-107 Password Changing Interface

See Figure 3-107 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.

3.9.16 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 3-108 System Restart Interface

See Figure 3-108 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

SMG1008C: 210×30×150 mm³ SMG1016C: 440×44×200 mm³ SMG1032C: 440×44×260 mm³

Weight

SMG1008C: 0.83 kg SMG1016C-16S, SMG1016C-16O : 3.13 kg SMG1032C-32S: 3.18 kg SMG1032C-32O: 3.01 kg

Environment

Operating temperature: 0 ℃—45 ℃ Storage temperature: -20 ℃—85 ℃ Humidity: 8%— 90% non-condensing Storage humidity: 8%— 90% non-condensing

LAN

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

FXS Port

Amount: 8/16/32

Type: RJ11

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: SMG1008C: 12V the direct current bigger than 3A SMG1016C, SMG1032C: 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 64 kbps G722 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-C 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-C 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-C 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-C 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-C 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-C gateway support fax?

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

Q9. Which RTP codecs are supported by the SMG-C 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-C 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. The FXS and FXO ports are one-to-one correspondence (Take SMG1016C-8S80 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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