

Synway SMG Series Analog Gateway

SMG1008 SMG1016 SMG1032 SMG1032A2 SMG1032A4 Analog Gateway

User Manual

Version 1.7.6

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



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Revision History

Version	Date	Comments	
Version 1.0	2013-10	Initial publication	
Version 1.3.0	2014-03	New revision	
Version 1.3.1	2014-06	Add description on the new series SMG1032A2	
Version 1.3.2	2014-07	New revision	
Version 1.3.3	2014-09	New revision	
Version 1.3.5	2014-10	New revision	
Version 1.5.0	2014-12	Add description on the new series SMG1032A4	
Version 1.5.1	2015-01	New revision	
Version 1.5.2	2015-04	New revision	
Version 1.5.3	2015-11	New revision	
Version 1.6.0	2016-03	New revision	
Version 1.6.3	2016-12	New revision	
Version 1.6.4	2017-02	New revision	
Version 1.7.1	2017-05	New revision	
Version 1.7.6	2018-01	New revision	

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



Chapter 1 Product Introduction

Thank you for choosing Synway SMG Series Analog Gateway!

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

SMG series analog gateway has five modules:

- SMG1008: 8 FXS/FXO
- SMG1016: 16 FXS/FXO
- SMG1032, SMG1032A2, SMG1032A4: 32 FXS/FXO

1.1 Typical Application

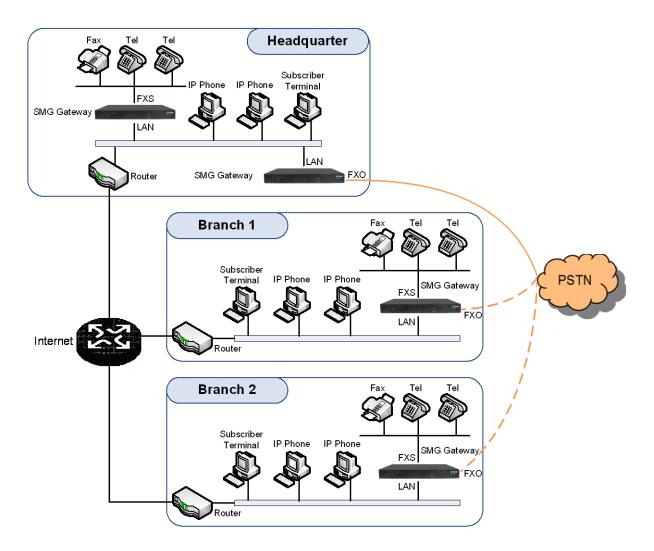


Figure 1-1 Typical Application



1.2 Feature List

Basic Features	Description	
TDM Call	Call initiated from TDM to IP, via routing and number manipulation to obtain the called IP address.	
IP Call	Call initiated from IP to TDM, via routing and number manipulation to obtain th destination.	
Number Manipulation	Peels off some digits of a phone number from left/right, or adds a prefix/suffix 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/FXO 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	Provides composite modules to enable a direct 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	

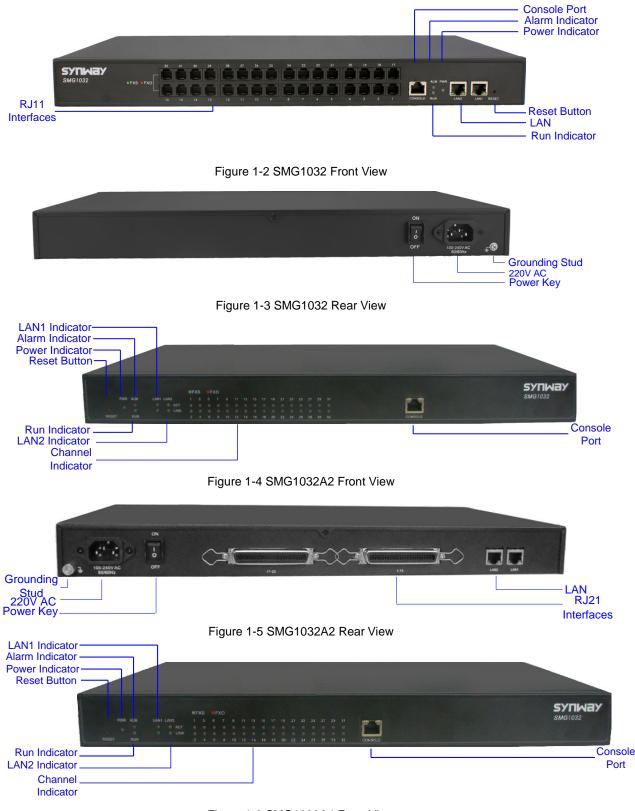


	ringing channel to the current channel.		
Centralized Manage	The gateway can register to Synway DCMS platform and accept the management of the platform.		
Signaling & Protocol	Description		
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261.		
Voice	CODEC G.711A, G.711U, G.729A/B, G.723, G.722, AMR, iLBC, SILK(16K), OPUS(16K), SILK(8K), OPUS(8K) 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.		
PPPoE	Virtual dial-up internet access support.		
DNS	Domain Name Service support.		
Security	Description		
Admin Authentication	Supports admin authentication to guarantee the resource and data security.		
System Monitor	Monitors the running status of the system and the server.		
Maintain & Upgrade	Description		
WEB Configuration	Support of configurations through the WEB user interface.		
Language	Chinese, English.		
Software Upgrade	Support of user interface, gateway service, kernel and firmware upgrades based on WEB.		
Tracking Test	Support of Ping and Tracert tests based on WEB.		
SysLog Type	Three options available: ERROR, WARNING, INFO.		

1.3 Hardware Description

The SMG analog gateway features 1U rackmount design and 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. See below for product appearance.









Ventilation

Holes

Screw Holes for

Foot Bracket



Figure 1-8 Left View

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

Interface	Description
	Amount: 2
	Type: RJ-45
LAN	Bandwidth: 10/100Mbps
	Self-Adaptive Bandwidth Supported
	Auto MDI/MDIX Supported
	Amount: 8/16/32
FXS/FXO	Type: RJ-11, RJ-21, RJ45
FX3/FXU	Maximum Transmission Distance: 1500m
	Charge Mode: Negative Anti-billing Supported
	Amount: 1
	Type: RS-232
	Baud Rate: 115200bps
Console Port	Connector: RJ45 to DB-9 Connector
Console Port	Data Bits: 8 bits
	Stop Bit: 1 bit
	Parity Unsupported
	Flow Control Unsupported
Button	Description
Power Key	Power on/off the SMG analog gateway.
Reset Button	Restore the gateway to factory settings.
LED	Description
Power Indicator	Indicates the power state. It lights up when the gateway starts up with the power
	cord well connected
Run Indicator	Indicates the running status. For more details, refer to <u>1.4 Alarm Info</u> .
Alarm Indicator	Alarms the device malfunction. For more details, refer to <u>1.4 Alarm Info</u> .
Link Indicator	The green LED on the left of LAN, indicating the network connection status.



ACT Indicator	The orange LED on the right of LAN, whose flashing tells data are being			
	transmitted.			
	FXS and FXO channels are respectively marked by green and red LED after power			
	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 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.
Alarm Indicator	Light up	Upon startup: System is normal. 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
- 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).

The connection for SMG1008, SMG1016, SMG1032 series products:

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.

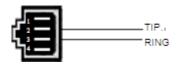
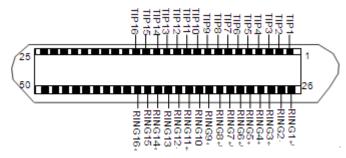
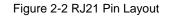


Figure 2-1 RJ11 Connection

The connection for SMG1032A2 series product:

SMG1032A2 adopts two RJ21 interfaces each of which accommodates 16 channels. One corresponds to channels 1 through 16 and the other corresponds to 17 through 32. Each pin in the RJ21 connector functions as follows.





The pins Ch1-a/b through Ch16-a/b on the RJ21 interface will be used respectively corresponding



to channels 1 through 16.

An RJ21 interface can be converted to 24 RJ11 interfaces through an RJ21-to-RJ11 adapter. See Figure 2-3 for the connection. SMG1032A2 needs two RJ21-to RJ11 adapters of which the first 16 slots will be used.

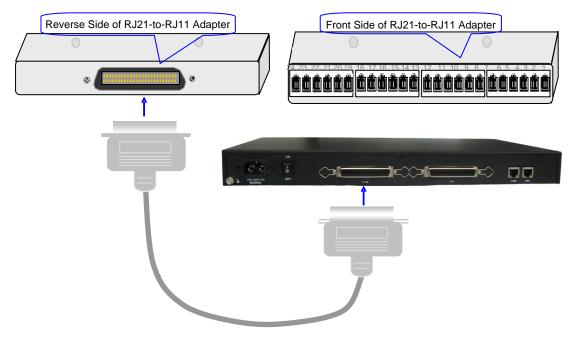


Figure 2-3 RJ21-to-RJ11 Adapter Connection

Users can also use the RJ21 connecting cable directly.

SMG1032A4 has eight 8-pin RJ45 jacks each of which can be connected to four 2-pin RJ11 jacks via a 4-way hub. Take the first RJ45 jack for example, the matching relationship among the channel number, the pins of the RJ45 jack and the 4-way hub is shown in the table below.

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

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

Step 6: Power on and start the gateway.

Step 7: 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 <u>3.1 System Login</u>. We suggest you change the initial username and password via 'System Tools \rightarrow Change Password' on the WEB interface as soon as possible after your first login. For detailed instructions about changing the password, refer to <u>3.9.15 Change Password</u>. After changing the password, you are required to log



in again.

Step 8: Modify IP address of the gateway.

You can modify the IP address of the gateway via 'System Tools \rightarrow Network' on the WEB interface to put it within your company's LAN. Refer to <u>3.9.3 Network</u> for detailed instructions about IP modification. After changing the IP address, you shall log in the gateway again using your new IP address.

Step 9: 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>3.6.4 Port Group</u> for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

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

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

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

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

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

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

1. Go to 'Port Settings → Port Group' on the WEB interface and click the 'Add New' button to create a new port group and add FXS ports which are connected with stations to it. Refer to



<u>3.6.4 Port Group</u> for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

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

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

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

3. Pick up the IP phone and call the IP address and port of the SMG 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 10: 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 11: 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 12: 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 13: Perform call forwarding. (Skip this step if not necessary.)

Situation 1: Hook-flash operation

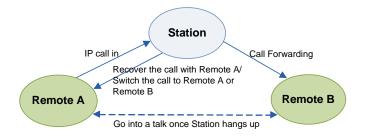


Figure 2-4 Call Forward via Hook-flash

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 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 1-8) 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.

¢_		
r N	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.



			System	Info	
System Info					
Channel State		LAN 1			
Call Count		MAC Address	80:7B:85:10:2E:75		
SID Massage Cours	•	IP Address	201.123.115.141	<u>255.255.255.0</u>	201.123.115.254
SIP Message Count	L	DNS Server	0.0.0		
]		Receive Packets	All:300120	Error:0	Drop:0
🗘 Quick Config	*	Transmit Packets	All:38088	Error:0	Drop:0
🛱 VolP	*	Current Speed	Receive:2.3 KB/s	Transmit:220 B/s	
		Work Mode	100Mb/s Full Duplex		
Advanced	8				
Port	×	LAN 2	Disable		
Route	*	Runtime	5h 54m 3s		
Num Manipulate	*	Current Version			
		WEB	forcommon_1.7.6_20	18011119	
System Tools	8	Gateway	forcommon_1.7.6_20		
		Serial No.	000007422		
		U-boot	Apr 09 2015 - 16:39:0	7	
		Kernel	#205 PREEMPT Tue	Jan 19 11:30:24 CST 2	016
		Product Type	1032A2(RJ21)		

Refresh

Figure 3-2 Main Interface

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3.2 Operation Info

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



Figure 3-3 Operation Info

3.2.1 System Info

MAC Address	80:7B:85:10:2E:75			
	201.123.115.141	255.255.255.0	201.123.115.254	
DNS Server		200.200.200.0	20111201110.201	
Receive Packets		Error:0	Drop:0	
Transmit Packets	All:38088	Error:0	Drop:0	
Current Speed	Receive:2.3 KB/s	Transmit:220 B/s		
Work Mode	100Mb/s Full Duplex			
LAN 2	Disable			
Runtime	5h 54m 3s			
Current Version				
WEB	forcommon_1.7.6_201	8011119		
Gateway	forcommon_1.7.6_2018011119			
Serial No.	000007422			
U-boot	Apr 09 2015 - 16:39:0	7		
Kernel	#205 PREEMPT Tue	Jan 19 11:30:24 CST 2	016	
Product Type	1032A2(RJ21)			

Figure 3-4 System Info Interface

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

ltem	Description



MAC Address	MAC address of LAN 1 or LAN 2 (disabled by default).								
IP Address	The three parameters from left to right are IP address, subnet mask and default								
IF Address	gateway of LAN 1 or LAN 2 (disabled by default).								
DNS Server	DNS server address of LAN 1 or LAN 2 (disabled by default).								
Deserve Destruits	The amount of receive packets after the gateway's startup, including three								
Receive Packets	categories: All, Error and Drop.								
T	The amount of transmit packets after the gateway's startup, including three								
Transmit Packets	categories: All, Error and Drop.								
Current Speed	The current speed of data receiving and transmitting.								
	The work mode of the network, including four options: 10 Mbps Half Duplex, 10								
Work Mode	Mbps Full Duplex, 100 Mbps Half Duplex, 100 Mbps Full Duplex.								
0	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 No.	Unique serial number of an SMG analog gateway.								
U-boot	Current version of Uboot.								
	Current version of the system kernel on the gateway.								
Kernel	Note: The kernel version for the gateways with RJ45/RJ21 interface is different								
	from that for the gateways with RJ11 interface.								
Product Type	The type of the analog gateway.								

3.2.2 Channel State

	Channel State														Chann	el State			
Channel	Туре	Number	Voltage(v)	State	Direction	CallerID	CalleeID	Reg Status	Polarity Reversal Count	Channel	Туре	Number	Voltage(v)	State	Direction	CallerID	CalleeID	Reg Status	Polarity Reversal Count
1			0	6						17			0	53					
2			0	6						18			0	53					
3			0	6						19			0	5					
4			0	6						20			0	5					
5			0	6						21			0	5					
6			0	6						22			0	5					
7			0	6						23			0	53					
8			0	6						24			0	53					
9			0	6						25			0	5					
10			0	6						26			0	5					
11			0	6						27			0	5					
12			0	6						28			0	5					
13	FXS	123	0					Unregistered		29			0	6					
14	FXS	124	0					Unregistered		30			0	6					
15			0							31	FXO	8031	0	6				Unregistered	
16			0							32	FXO	8032	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.

ltem	Description
Channel	Channel number on the device.
	Type of the channel on the device: FXS or FXO. If this item shows, it means this
Туре	channel is unavailable, that is, the corresponding module to this channel is not
	inserted or damaged.



	Note: If the FXO	port is	unconnected, the channel is unavailable too.								
Number	The number corr	espond	ing to the port.								
Voltage	Line voltage on t	he chan	nnel, calculated by volt (V).								
	Displays the channel state in real time. You can move the mouse onto the channel state icon for detailed state information.										
	State	Description									
	Idle		The channel is available.								
	Off-hook	<u>د</u>	The channel picks up the call.								
State	Wait Answer		The channel receives the ringback tone and is waiting for the called party to pick up the phone.								
	Ringing		The channel is in the ringing state.								
	Talking		The channel is in a conversation.								
	Dialing	(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 regi	istration	status of the port.								
Polarity Reversal	The country of the		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 Count													
Call 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	>IP 0 0 0		0	0	0	0 0							
	Refresh												
				- teno									

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: $IP \rightarrow TeI$ and $TeI \rightarrow IP$.						
Total Calls	Total number of calls in a specified call direction.						
Successful Calls	Total number of successful calls in conversation.						
Durau	Total number of calls which fail as the called party has been occupied and replies a						
Busy	busy message.						
	Total number of calls which fail as the called party does not pick up the call in a long						
No Answer	time or the calling party hangs up the call before the called party picks it up.						



Call Forward Total number of calls which have been forwarded.							
Routing Failure	Total number of calls which fail because no routing rules are matched.						
Dialing Failura	Total number of calls which fail as the called party number does not conform to the						
Dialing Failure	dialing rule or due to dialing timeout.						
Unknown Failure	Total number of calls which fail due to unknown reasons.						

3.2.4 SIP Message Count

						Reques	t							
Request		REGISTER		INVITE		ACK	INFO		BYE		CANCEL		NOTIFY	OPTION
Send		0		1		1	0		1		0		0	0
Send Repeatedly		0		0		0	0		0		0		0	0
Receive			0	1		1	0		1		0		0	0
Receive Repeatedly			0			0	0		0		0	0		0
					(Common Res	ponse							
Common Response	100) Trying 180 Ringing			183 Session Prosess			20	200 OK 4		486 Busy		487 Request Already Terminated	
Send		1	1		0				2	0		0		
Receive		1	1		0				2		0		0	
					Ref	resh	Clear							

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



Figure 3-8 Quick Config Interface

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

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



	Quick Config-Network	Settings
LAN 1		
	Network Type:	Static
	IP Address (I)	192.168.1.101
	Subnet Mask (U)	255.255.255.0
	Default Gateway (D)	192.168.1.1
	DNS Server (P)	0.0.0.0
	Speed and Duplex Mode	Automatic Detection
LAN 2		Enable
	Next	

Figure 3-9 Quick Config-Network Settings Interface

See Figure 3-10 for the Quick Config-SIP Settings interface. The configuration items on this interface are the same as those on the SIP interface. Refer to <u>3.4.1 SIP</u> for detailed settings. You are required to fill with the information about the registrar if the gateway must be registered. After configuration, click **Back** to go back to the Network Settings interface; click **Next** to enter the FXS Settings interface.

Quick Config-SIP Setting	js
SIP Address	LAN 1: 192.168.1.101
Registrar IP Address Registrar Port	
Spare Registrar IP Address Spare Registrar Port	
Registry Validity Period (s)	600
Back	

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.



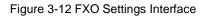
Synway Information Engineering Co., Ltd

																		-
Port	Туре	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	Mod
1	FXS	8001			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	
2	FXS	8002			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	
3	FXS	8003			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	
4	FXS	8004			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	
5	FXS	8005			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	
6	FXS	8006			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	
7	FXS	8007			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	1
8	FXS	8008			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	1
9	FXS	8009			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	6
10	FXS	8010			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	0
11	FXS	8011			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	1
12	FXS	8012			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	1
13	FXS	8013			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	1
14	FXS	8014			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	
15	FXS	8015	-	-	Disable	Disable	Disable		-	Enable	Disable	Unregistered	Enable	Disable	-	0	0	0
16	FXS	8016			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	
										111								
Items	Total	16 Itome/Page	1/2 Eiret Previou	in Next Lent C	o to Page 🚺 💌 2 Pages	Tatal												atch M

Figure 3-11 FXS Settings Interface

See Figure 3-12 for 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.

							-						
ort	Туре	SIP Account	Display Name	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	Polarity Reversal Detection	Input Gain	Output Gain	Мос
17	FXO	8017		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	0
18	FXO	8018		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
19	FXO	8019		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
20	FXO	8020		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
21	FXO	8021		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
22	FXO	8022		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
23	FXO	8023		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
24	FXO	8024		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
25	FXO	8025		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
26	FXO	8026		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
27	FXO	8027		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
28	FXO	8028		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
29	FXO	8029		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
30	FXO	8030		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
31	FXO	8031		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
32	FXO	8032		Two Stages Dialing for Incoming Call		Disable	Disable	Unregistered	Enable	Disable	0	0	6
		1				III						1	



Back Next

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 FXO 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* 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 Settings	
SIP Address	LAN 1: 192.168.1.101 📼
SIP Port	5060
Register Status	Unregistered
Register Gateway	Yes 💌
SIP Account	
Password	
Authentication Username	
Desister ID Address	
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 Bart if SID Degistration Failed	Enable
Switch Signal Port if SIP Registration Failed	
IMS Network	Enable
Externally Bound Address	
Externally Bound Port	5060
Save Reset	

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 service, 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				
SIP Address	IP address of SIP signaling, using LAN 1 by default.				
	Monitoring port of SIP signaling. The value range of it must be greater than 1024				
SIP Port	and 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 <i>Failed</i> or <i>Registered</i> .					
	Sets whether to register the gateway as a whole. The default value is No. Only					
Register Gateway	when this configuration is set to Yes can you see the configuration items SIP					
	Account and Password.					
	When the gateway initiates a call to SIP, this item corresponds to the username of					
SIP Account	SIP.					
Decement	Registration password of the gateway. To register the gateway to SIP, both					
Password	configuration items SIP Account and Password should be filled in.					
Authentication	Authoritization upper and for registration					
Username	Authentication username for registration.					
Registrar IP Address	Address of the registry server for the gateway to register.					
Registrar Port	Signaling port of the registry server.					
Spare Registrar	Check the enable checkbox to enable the spare registrar server. By default, it is					
Server	disabled.					
	Address of the spare registry server for the gateway to register. The gateway will					
Spare Registrar IP	enable the spare registrar server if the master registrar server has no reply, or the					
Address	master server is detected with no response in case the item Detection Server					
	<i>Cycle</i> is enabled.					
Spare Registrar Port	Signaling port of the spare registry server.					
Registry Validity	Validity period of the SIP registry. Once the registry is overdue, the gateway should					
Period	be registered again. This configuration item is valid only when <i>Register Gateway</i> is					
	set to Yes. Range of value: 10~3600, calculated by s, with the default value of 600.					
Multi-Registrar	Tick the checkbox before to enable the multi-registrar server mode. By default, it is					
Server Mode	disabled.					
SIP Transport	There are two modes UDP and TCP available for running the SIP protocol. The					
Protocol	default value is <i>UDP</i> .					
Switch Signal Port if	If the SIP registration fails, the SIP signaling port N will switch to N+1 for a new					
SIP Registration	registration. It will continue until the registration succeeds. The default value is					
Failed	disabled.					
	Once this feature is enabled, the gateway will send signaling messages to the					
	corresponding externally bound address and port when it registers to the server. By					
IMS Network	default, this feature is <i>disabled</i> . Only when this feature is <i>enabled</i> will these items					
	Externally Bound Address, Externally Bound Port and Authentication					
	Username be shown.					
Externally Bound	Externally bound IP address for registration.					
Address						
Externally Bound	Externally bound port for registration.					
Port						

3.4.2 SIP Compatibility

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



configurations.



SIP 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 Handle	Internal Handling Match Port Number
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 Cycle(s)	✓Enable 0
Server Status Detection Cycle(s) Send Cue Tone	<pre>✓Enable 0 Enable</pre>
SIP Encryption Encryption Criterion Identifier Key	VOS1.1
RTP Encryption	Enable
Ignore ACK	Enable
User-defined SIP Code	Enable
Use lptables	Enable
Save	Reset



Figure 3-16 SIP Compatibility Setting Interface

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

Item	Description					
Obtain CalleeID	There are two optional ways to obtain the called party number: from "To" Field and					
from	from "Request" Field. The default value is "Request" Field.					
	There are two options to set the position of the calling party number: "Displayname					
Set CallerID Position	of From Field" and "Username of From Field". The default value is "Username of					
	From Field".					
	There are two optional ways to obtain the calling party number: from "Displayname					
Obtain CallerID from	of From Field" and from "Username of From Field". The default value is "Username					
	of From Field".					
	Sets whether to send the request message according to the content of Contact, with					
Use Contact	the default setting of disabled. As it is disabled, if the Contact field indicates an IP					
	address 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.					
Coll Tropofor Mode	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.					
Internal Handle	Sets the internal handle mode for the call transfer, including two options: Match Port					
internal handle	Number and Search Idle FXO Channel. The default value is Match Port Number.					
Call Flagh Made	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.					
Hold Music Source	Sets the source of the hold music, with the default value of Remote, This feature					
Hold Music Source	gets valid only when you choose the mode Platform to Handle SIP Info.					
Two Stage Dialing	Once this facture is applied, the incoming call from SID should perform the two					
for SIP Incoming	Once this feature is enabled, the incoming call from SIP should perform the two stage dialing operation. By default this feature is <i>disabled</i> .					
Call						
	Sets the maximum time for the SIP channel to wait for the answer from the called					
Maximum Wait	party of the outgoing call it initiates. If the call is not answered within the specified					
Answer Time	time period, it will be canceled by the channel automatically. The default value is 60,					
	calculated by s.					
SIP Station	Once this feature is enabled, a SIP terminal can be registered to the gateway and					
Supported	becomes a SIP station. By default this feature is disabled.					
Set SIP Identifying	Sets the SIP identifying content in the SIP call message. The default setting is					
Set Sir 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					
Cycle	this feature is necessary to be used.					



Server Status Detection Cycle	The interval of sending a heartbeat packet to detect the master registrar server status, with the default value of 0 (feature disabled), calculated by s. It is suggested to set to 15s if this feature is necessary to be used.
Send Cue Tone	Sets whether to send a cue tone once the server gets disconnected, with the default setting of <i>disabled</i> .
SIP Encryption	Once this feature is enabled, you can encrypt the SIP signal following selecting an encryption criterion and setting a key. By default it is <i>disabled</i> .
Encryption Criterion	The criterion used to encrypt the SIP signal. At present only VOS1.1 is supported.
Identifier	The identifier field of the VOS encryption, which is used to obtain the key of the SIP encryption.
Key	The key to encrypt the SIP signal.
RTP Encryption	Once this feature is enabled, you can encrypt the RTP package. By default it is <i>disabled</i> .
Ignore ACK	Once this feature is enabled, it is not necessary for the gateway to wait for the ACK message after sending the 200OK message to establish a call. By default it is <i>disabled</i> .
User-defined SIP Code	Once this feature is enabled, you can define a SIP code for the corresponding SIP status, with the default value of <i>disabled</i> .
Use Iptables	Once this feature is enabled, only the calls from the SIP registration server, the source IP address of the route IP->TEL and these IP addressed set in <u>Access</u> <u>Control</u> interface are permitted.

3.4.3 SIP Station

A SIP terminal can be registered to the gateway and becomes a SIP station. Enable the feature of 'SIP Station Supported' on <u>3.4.2 SIP Compatibility</u> interface, 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	۲
•••	Quick Config	۲
	VolP	*
	SIP	
	Sip Compatibility	
:	SIP Station	
	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.



	SIP Station
Number:	0
Username:	
Password:	
Bound Port:	29
Description:	default
Batch Setting:	Enable
Save	Close

Figure 3-18 Add New SIP Station

The table below explains the items shown above:

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

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

SIP Station												
Check	Number	Username	IP Address	Bound Port	Register Status	Register Duration (s)	Voice Channel State	Description	Modify			
	0	120	-	29	Unregistered	-		default				
Check All E Uncheck All E Inverse E Delete E Clear All												
1 Items Tota	Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 🗸 1 Pages 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.



SIP Station		
Number:	0	
Username:	120	
Password:	•••	
Bound Port:	29 💌	
Description:	default	
Batch Setting:	Enable	
Save	Close	

Figure 3-20 SIP Station Modification Interface

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

3.4.4 SIP Server

The gateway supports the multi-registrar server feature. Enable the feature of '*Multi-Registrar Server Mode*' on the <u>SIP</u> interface (see <u>3.4.1 SIP</u>) and you will see the item SIP Server under the VoIP Settings menu. Click '*SIP Server*' to go into the SIP Server interface. By default, there is no available SIP server. See Figure 3-21 below.

	Operation Info	8
•••	Quick Config	*
	VoIP	No Available Registrar Serv
	SIP	
	Sip Compatibility	
	SIP Station	
	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.



Add New SIP S	Server
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
Save	Cancel
Jave	Cancer

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.

p Modif								
Check All E Uncheck All E Inverse E Delete E Clear All Add New								

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.



Modify SIP	Server
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
Save	Cancel

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 Туре	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 "Auto Dete	professional person please do not modify the configurat T Traversal": Please select one method according to you It is required to enable the feature of upon or pmp for th Contact IP": It is required to set the router to map the SIF SDP IP": It is required to set the router to map the RTP p ect NAT IP": It is valid only when the feature "Rport" is en- ort range to the gateway.	ur current network environment. ne router. P port to the gateway. port range to the gateway.
	Save Reset	
	i i i i i i i i i i i i i i i i i i i	

Figure 3-25 NAT Setting Interface

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

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



Address	feature is enabled.			
STUN Server	Sets whether to enable the STUN server for NAT traversal. By default the STUN			
	server is disabled.			
	Detected NAT (Network Address Translation) type. The gateway will return the NAT			
	type automatically in case STUN Server is enabled. It includes 9 types: unknown;			
NAT Type	no NAT; ConeNat; RestrictedNat; PortRestrictedNat; Symmetric NAT; Symmetric			
	NAT with firewall; can't detect over (fail to send detect message) and fail to detect			
	(No reply from the stun server).			
STUN Server				
Address	Address of the server for STUN traversal.			
Mapping Contact IP	The IP filled in here will be used in the Contact field of the SIP message.			
Mapping SDP IP	The IP filled in here will be used in the SDP field of the SIP message.			
Drawt	When this feature is enabled, a corresponding Rport field will be added to the Via			
Rport	message of SIP. The default value is enabled.			
	When this feature is enabled, the gateway will parse the corresponding address and			
Auto Detect NAT IP	port in the message returned by Rport so as to use them for the following			
Auto Detect NAT IP	communication. By default, this feature is <i>disabled</i> .			
	Note: This feature gets valid only when Rport is enabled.			
	When this feature is enabled, the RTP reception address or port carried by the			
DTD Calf adaption	signaling message from the remote end, if not consistent with the actual state, will			
RTP Self-adaption	be updated to the actual RTP reception address or port. By default, this feature is			
	disabled.			



3.4.6 Media

		Media Par	ameters		
	DTMF Transmit	Mode	RFC2	833 💌	
	RFC2833 Paylo	ad	101		
	RTP Port Range)	6000,	10000	
	Silence Suppres	ssion	Disab	le 💌	
	Auto Noise Red	uction	Disab	le 💌	
	JitterMode		Static	Mode 💌	
	JitterBuffer(ms)		100		
	JitterUnderrunL	ead(ms)	200		
	JitterOverrunLea	ad(ms)	200		
	Voice Gain Outp	out from IP (dB)	0		
CODEC Pr	iority				
Check	Priority 1 2 3 4 5 6 7 8 9 10 11	CODEC G711A • G711U • G729 • G723 • G722 • AMR • iLBC • SILK(16K) • OPUS(16K) • SILK(8K) •	Packing Time 20 20 20 20 30 20 20 20 20 20 20 20 20 20 2	Bit Rat 64 64 8 6.3 64 6.70 15.2 20 20 20 12 12	
		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 service, 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.



DTMF Transmit	Sets the transmit mode for the IP channel to send DTMF signals. The optional			
Mode	values are <i>RFC2833</i> , <i>In-band</i> and <i>Signaling</i> , with the default value of <i>RFC2833</i> .			
RFC2833 Payload	Payload of the RFC2833 formatted DTMF signals on the IP channel. Range of			
	value: 90~127, with the default value of 101.			
	Supported RTP port range for the IP end to establish a call conversation, with the			
RTP Port Range	lower limit of 2000 and the upper limit of 60000 and the difference between larger			
	than 480. The default value is 6000-10000.			
	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> .			
Auto Noise	Once this feature is enabled, the volume of the noise accompanied with the line will			
Reduction	be reduced automatically. By default, the feature is disabled.			
litter Mede	Sets the working mode of JitterMode. The optional values are Static Mode and			
JitterMode	Adaptive Mode, with the default value of Static Mode.			
	Acceptable jitter for data packets transmission over IP, which indicates the buffering			
	capacity. A larger JitterBuffer means a higher jitter processing capability but as well			
JitterBuffer	as an increased voice delay, while a smaller JitterBuffer means a lower jitter			
	processing capability but as well as a decreased voice delay. Range of value:			
	0~280, calculated by ms, with the default value of 100.			
	Sets the initial delay of packets if they are received later than JitterBuffer. Range of			
JitterUnderrunLead	value: 0~280, calculated by ms, with the default value of 200,			
	Note: Only when JitterMode is set to Static Mode will this item be shown.			
	Sets the initial lead inserted if packets are received earlier than 300-JitterBuffer.			
JitterOverrunLead	Range of value: 0~280, calculated by ms, with the default value of 200,			
	Note: Only when JitterMode is set to Static Mode will this item be shown.			
	Sets the minimum delay that can be set by the adaptive jitter function. It cannot be			
	larger than the value set in JitterBuffer. Range of value: 0~280, calculated by ms,			
JitterMin	with the default value of <i>10</i> .			
	Note: Only when JitterMode is set to Adaptive Mode will this item be shown.			
	Sets the rate for delay reduction under the adaptive mode. It defines the maximum			
	percentage of silence that can be removed for delay reduction. Range of value:			
JitterDecreaseRatio	0~100, with the default value of <i>50</i> ,			
	Note: Only when JitterMode is set to <i>Adaptive Mode</i> will this item be shown.			
	Sets the maximum delay that can be increased during a silence period. Range of			
JitterIncreaseMax	value: $0 \sim 280$, calculated by ms, with the default value of 50,			
	Note: Only when JitterMode is set to <i>Adaptive Mode</i> will this item be shown.			
Voice Gain Output	Adjusts the gain of the voice output from IP. Range of value: -24~24, calculated by			
from IP	dB, with the default value of 0.			
	1			



	Supported COD	ECs and their corresponding	priority for the IP end to establish a				
	call conversation	call conversation. The table below explains the sub-items:					
	Sub-item	De	scription				
	Duria urite e	Priority for choosing the CC	Priority for choosing the CODEC in an SIP conversation. The				
	Priority	smaller the value is, the high	smaller the value is, the higher the priority will be.				
		Three optional CODECs are	supported: G711A, G711U,				
	CODEC	G729A/B, G723, G722, AMF	R, iLBC, SILK(16K), OPUS(16K),				
		SILK(8K) and OPUS(8K).					
	Packing Time	Time interval for packing an	RTP packet, calculated by ms.				
	Bit Rate	The number of thousand bits are conveyed per second.	(excluding the packet header) that				
	By default, all of	the eleven CODECs are sup	ported and ordered G711A, G711U,				
	G729A/B, G723	3, G722, AMR, iLBC, SILK	(16K), OPUS(16K), SILK(8K) and				
	OPUS(8K) by pr	iority from high to low.					
	The packing time	e and bit rate supported by diffe	erent CODECs are listed in the table				
CODEC Priority	below. Those val	lues in bold face are the defau	It values.				
	COEDC	Packing Time (ms)	Bit Rate (kbps)				
	G711A	5 / 10 / 20 / 30 / 40 / 50 / 60	64				
	G711U	5 / 10 / 20 / 30 / 40 / 50 / 60	64				
	G729A/B	10 / 20 / 30 / 40 / 50 / 60	8				
	G723	30 / 60 / 90	5.3 / 6.3				
	G722	5 / 10 / 20 / 30 / 40	64				
		20 / 40 / 60 / 80 / 100	4.75 / 5.15 / 5.90 / 6.70 / 7.40 /				
	AMR		7.95 / 10.20 / 12.20				
		20 / 40	15.2				
	iLBC	30	13.3				
		60	13.3 / 15.2				
	SILK(16K)	20 /40 / 60	20				
	OPUS(16K)	10 / 20 / 40 / 60	20				
	SILK(8K)	20 /40 / 60	12				
	OPUS(8K)	10 / 20 / 40 / 60	12				

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 analog voice ports, such as the conditions for sending the caller party information. Tone Detector is used to configure some properties of detected tones. Tone Generator is used to configure some properties of generated tones. 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

	FXS	
Minimum (ms) CID Tran Occasion Send Pol Off-hook Hybrid Ba Handling Light Up I	sh Detection Time Length of On-hook Detection smit Mode to Send FSK CallerID arity Reversal Signal Dither Signal Duration (ms)	-16 □Enable 64 FSK After the first ring □Enable 64 ✓ Platform Handling Not Light Up □Enable
	Save	

Figure 3-28 FXS Configuration Interface

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



ltem	Description				
Tono Enormy	Energy of the tone signal sent by the gateway. Range of value: -35~15, calculated				
Tone Energy	by dB, with the default value of -16.				
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				
	time more than the value of this configuration item will be regarded as a valid flash				
Minimum Time	operation. Range of value: 80~ Maximum Time, calculated by ms, with the default				
	value of 80.				
	Note: This item appears only when Hook-flash Detection is enabled.				
	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.				
	Note: This item appears only when the hook-flash detection is enabled.				
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 Hook-flash Detection is disabled.				
CID Transmit Mode	The mode adopted by the FXS port to send the CallerID. The optional values are				
	FSK and DTMF, with the default value of FSK.				
Occasion to Send	Sets when to send the CallerID, before rings or after the 1 st Ring. The default value				
FSK CallerID	is after 1 st Ring.				
Send Polarity	Once this feature is enabled, the gateway will send the polarity reversal signal to a				
Reversal Signal	corresponding FXS channel when it detects the called party pick-up behavior. By				
	default, this feature is disabled.				
Off-hook Dither	The minimum duration of the off-hook signal, calculated by millisecond (ms), which				
Signal Duration	must be the multiple of 16. The less value indicates the larger sensitivity. And the				
	default value is 64.				
Hybrid Balance	Sets whether to enable the hybrid balance feature or not. The default setting is				
	being enabled.				
Handling of Call from	Sets the handling mode for the calls from station to station, two options available:				
Internal Station	Internal Handling and Platform Handling, with the default value of Platform				
	Handling.				
Light Up Mode for	Sets the light up mode for leaving a voice message on the phone, two options				
Voice Message	available: Not Light Up and Light Up by FSK, with the default value of Not Light Up.				
Open Session In	Sets whether to reply 183 for an incoming FXS call.				
Advance					

After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to <u>3.9.16 Restart</u> for detailed instructions.



3.5.2 FXO

FXO	
Calling Party Detection Time (s) Silence Detection(FXO will Hang up the Call upon Detecting the Silence.)	10 Enable
Incoming Call from PSTN Rapid Release	Enable
FSK Standard Reception Interval of DTMF CallerID (ms) Delay for Two Stages Dialing (s)	GR-30(North America, China) 250 0
Outgoing Call to PSTN Flash Time (ms)	100
Delay after Dial (ms) FXO Pick-up Delay after INVITE Received at IP	1000
Side(s) Maximum Wait Answer Time (s)(Valid when Polarity Reversal Enabled)	0 60
Communicate without Network Communicate without Network Mode Two Stage Dialing Mode	Enable Auto search idle c Enable
Delay to Send 200 OK to IP Side (Invalid if Polarity Reversal is enabled)	Enable
Avoid Being Detected as Flash Signal by PBX Open Session In Advance	Enable Enable
Save	

Figure 3-29 FXO Configuration Interface

The table below explains the particular configuration items for FXO.

ltem	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	-86~5, calculated by s, with the default value of -34.
	Note: This item will be valid only when Silence Detection is enabled.
Time Thus she ld of	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.
	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 districts. The optional values are: ETSI (Europe), GR-30 (North America, China) and NIT (Japan), with the default value of GR-30. Reception Interval of DTMF CallerID The interval between digits of the DTMF CallerID from FXO port, calculated by ms, with the default value of 250. Delay for Two Stages Dialing If the feature of two-stages dialing mode is enabled and an incoming call occurs, the FXO port will have a delay set by this configuration item before going into the two-stages dialing process. Flash Time Sets the time for generating a flash signal on the analog trunk. Range of value: 32-1000, calculated by ms, with the default value of 100. Delay after Dial Sets the delay to send the CalleeID to PBX after you pick up and dial. Range of value: 200-2000, calculated by ms, with the default value of 1000. FXO Pick-up Delay after INUTE 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 by s, with the default value of 60. Communication Automatically routes a call to the proper port according to the configuration in case of network failure or call timeout. Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network, its disconnected; in the mode of Use Current Route Setting, will bead on the 200 OK message to the IP side. The default value is disabled. Mode Sets whether it is necessary to perform the two-stages dialing operation to call t		
and NIT (Japan), with the default value of GR-30. Reception Interval of DTMF CallerID The time interval between digits of the DTMF CallerID from FXO port, calculated by ms, with the default value of 250. Delay for Two Stags Dialing If the feature of two-stages dialing mode is enabled and an incoming call occurs, the FXO port will have a delay set by this configuration item before going into the Sets the time for generating a flash signal on the analog trunk. Range of value: 32-1000, calculated by ms, with the default value of 100. Delay after Dial Sets the delay to send the CalleelD to PBX after you pick up and dial. Range of value: 200-2000, calculated by ms, with the default value of 1000. FXO Pick-up Delay after IN/ITE Received at IP Side Once this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the IN/ITE message. Received at IP Side 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 by s, with the default value of 60. Communication Automatically routes a call to the proper port according to the configuration in case without Network 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 Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, with the default value of Automation was a FXO port. By default this feature is disabled. <		
Reception Interval of DTMF CallerID The time interval between digits of the DTMF CallerID from FXO port, calculated by ms, with the default value of 250. Delay for Two Stages Dialing If the feature of two-stages dialing mode is enabled and an incoming call occurs, the FXO port will have a delay set by this configuration item before going into the two-stages dialing process, Flash Time Sets the time for generating a flash signal on the analog trunk. Range of value: 32–1000, calculated by ms, with the default value of 100. Delay after Dial Sets the delay to send the CalleeID to PBX after you pick up and dial. Range of value: 200–2000, calculated by ms, with the default value of 1000. FXO Pick-up Delay after INVITE Once this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message. Maximum Wait Answer Time 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 by s, with the default value of 60. Communicate without Network 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 Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port. By default this feature is disabled. Two Stages Dialing Mode Once this feature is enabled, the gateway will delay to send the 200 OK message to the IP side. The default value is <i>disabled</i> . Two Stages Dialing Mode	FSK Standard	
DTMF CallerID ms, with the default value of 250. Delay for Two Stages Dialing If the feature of two-stages dialing mode is enabled and an incoming call occurs, the FXO port will have a delay set by this configuration item before going into the two-stages dialing process, Flash Time Sets the time for generating a flash signal on the analog trunk. Range of value: 32-1000, calculated by ms, with the default value of 100. Delay after Dial Sets the delay to send the CalleelD to PBX after you pick up and dial. Range of value: 200-2000, calculated by ms, with the default value of 1000. FXO Pick-up Delay after INVITE Once this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message. Maximum Wait Answer Time 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 by s, with the default value of 60. Communication without Network 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 Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port. By default this feature is disabled. Mode Sets whether it is necessary to perform the two-stages dialing operation to call the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of TeI>IP route. Two Stages Dialing Mode Once this feature is enabled, the gatewa	Personation Interval of	
Delay for Two Stages If the feature of two-stages dialing mode is enabled and an incoming call occurs, the FXO port will have a delay set by this configuration item before going into the two-stages dialing process, Flash Time Sets the time for generating a flash signal on the analog trunk. Range of value: 32–1000, calculated by ms, with the default value of 100. Delay after Dial Sets the delay to send the CalleelD to PBX after you pick up and dial. Range of value: 200–2000, calculated by ms, with the default value of 1000. FXO Pick-up Delay after Dial Sets the delay to send the CalleelD to PBX after you pick up and dial. Range of value: 200–2000, calculated by ms, with the default value of 1000. FXO Pick-up Delay after INITE Once this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message. Maximum Wait Answer Time 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 by s, with the default value of 60. Communicate Mutomatically routes a call to the proper port according to the configuration in case of network failure or call timeout. Communicate Sets the mode for the communications without network, two options available: Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an escaping channel according to the settings of Tel->IP route. Two Stages Dialing Sets whether it is necessary to perform the two-stages dialing operation to call the remote end via an FXO	•	
Delay for Two Stages FXO port will have a delay set by this configuration item before going into the two-stages dialing process, Flash Time Sets the time for generating a flash signal on the analog trunk. Range of value: 32–1000, calculated by ms, with the default value of 100. Delay after Dial Sets the delay to send the CalleeID to PBX after you pick up and dial. Range of value: 200–2000, calculated by ms, with the default value of 1000. FXO Pick-up Delay after INVITE Sets the delay to send the CalleeID to PBX after you pick up the call after the IP side receives the INVITE message. Maximum Wait Once this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message. Communication 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 by s, with the default value of 60. Communicate Automatically routes a call to the proper port according to the configuration in case of network failure or call timeout. Sets the mode for the communications without network, two options available: Auto Search Idle Channel. In the mode of Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an elseconing to the settings of TeI>IP route. Two Stages Dialing 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 2000 Once this feature is enabled, the gateway will de	DIMF Callerid	
Dialing two-stages dialing process, Flash Time Sets the time for generating a flash signal on the analog trunk. Range of value: 32–1000, calculated by ms, with the default value of 100. Delay after Dial Sets the delay to send the CalleeID to PBX after you pick up and dial. Range of value: 200–2000, calculated by ms, with the default value of 1000. FXO Pick-up Delay after INVITE Sets the delay to send the CalleeID to PBX after you pick up and dial. Range of the IP side receives the INVITE message. Maximum Wait Once this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message. Maximum Wait 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 by s, with the default value of 60. Communication Automatically routes a call to the proper port according to the configuration in case of network failure or call timeout. Mode Sets the mode for the communications without network, two options available: Auto Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of TeI>IP route. Two Stages Dialing 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 Once thi	Delay for Two Stages	
Flash TimeSets the time for generating a flash signal on the analog trunk. Range of value: 32-1000, calculated by ms, with the default value of 100.Delay after DialSets the delay to send the CalleeID to PBX after you pick up and dial. Range of value: 200-2000, calculated by ms, with the default value of 1000.FXO Pick-up Delay after INVITE Received at IP SideOnce this feature is enabled, the FXO port will be delayed to pick up the call after the IP side receives the INVITE message.Maximum Wait Answer TimeThe maximum time to wait the answer of the remote side for an outgoing call from FXO port. This item is valid only when Polarity Reversal is enabled. It is calculated by s, with the default value of 60.Communication without NetworkAutomatically routes a call to the proper port according to the configuration in case of network failure or call timeout.Communicate without NetworkSets the mode for the communications without network, two options available: Auto Search Idle Channel and Use Current Route Setting, with the default value of <i>Auto</i> Search Idle Channel. In the mode of Auto Search Idle Channel, the gateway will search an idle FXO port to route the call once the network is disconnected; in the mode of Use Current Route Setting, the gateway will search an escaping channel according to the settings of Tel->IP route.Two Stages Dialing Detacted as Flash Signal by PBXOnce this feature is enabled, the gateway will delay to send the 200 OK message to the IP side. The default value is <i>disabled</i> .Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing cal; otherwise, it will reply the 180 message. This item is	Dialing	
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Open Session In AdvanceOnce this feature is enabled, the gateway will reply the 183 message when the FXO port is making an outgoing call; otherwise, it will reply the 180 message. This item is	Signal by PBX	
<i>Open Session In</i> port is making an outgoing call; otherwise, it will reply the 180 message. This item is		Once this feature is enabled, the gateway will reply the 183 message when the FXO
Advance	-	
Valid Only when Folding Reversal is enabled. The default value is enabled.	Advance	valid only when Polarity Reversal is enabled. The default value is <i>enabled</i> .

After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to <u>3.9.16 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	Accepted Freque
	0	Dial Tone	Continuous Tone	450	0	1500	0	0	5
	1	Busy Tone	Periodic Tone	450	0	350	350	2	5
	2	Ringback Tone	Periodic Tone	450	0	1000	4000	1	5
<								>	
Check All 🗄 Uncheck All 🗄 Inverse 🗄 Delete 🗮 Clear All Add New									
3 Items To	3 Items Total 20 Items/Page 1/1 First Previous Next Last Goto Page 1 🛩 1 Pages Total								

Figure 3-30 Tone Parameters Setting Interface

See Figure 3-30 for the Tone Parameters setting interface. By default, there are three pieces of tone parameters on the gateway. Click *Add New* to add tone parameters manually, see Figure 3-31.

Tone Pa	rameters
Index:	3 🗸
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
Accepted Frequency Err	ror(%): 5
Duration Error at ON/OF	F State(%): 20
Save	Close

Figure 3-31 Add New Tone Parameter Interface

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.
Period Count	Set the count of periods as the condition to determine a periodic tone. The default
	setting: Dial Tone is 0, Busy Tone is 2, Ringback Tone is 1.
Accepted Frequency	Allowable error of the center frequency. Range of value: 1~5, calculated by %, with
Error	the default value of 5.
Duration Error at	The accepted maximum error at on/off state. Range of value: 0~100, calculated
ON/OFF State	by %, with the default value of 20.

After configuration, click *Save* to save the above settings into the gateway or click *Close* to cancel the settings. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to <u>3.9.16 Restart</u> for detailed instructions.

Click *Modify* in Figure 3-30 to modify the tone parameter. See Figure 3-32 for the tone parameter modification interface. The configuration items on this interface are the same as those on the *Add New Tone Parameter* 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		
Accepted Frequency Error(%): 5		
Duration Error at ON/OFF State(%): 20		
Save	Close	

Figure 3-32 Modify Tone Parameter

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



3.5.4 Tone Generator

Tone Generator		
Dial Tone 450/0	350+440/0 Continuously play a dual tone which is composed of 350HZ and 440HZ.Note: The value range of the frequency is 200~3500HZ.	
Ringback Tone 450/1000,0/4000	480+620/500,0/500 Repeatedly play a dual tone which is composed of 480HZ and 620HZ in the method of 500ms play with 500ms pause. Note: 0/500 denotes 500ms silence and the tone cannot start with the silence.	
Busy Tone 450/350,0/350	950/333,1400/333,1800/333,0/1000 Repeatedly play tones in turn: first a 333ms 950HZ tone, followed by a 333ms 1400HZ tone, then a 333ms 1800HZ tone and at last a 1s silence.Note: The count of signals at ON state in a period cannot be greater than 4.	
Save	Reset	

Figure 3-33 Tone Generator Setting Interface

See Figure 3-33 for the Tone Generator Setting interface. By default, there are three tones on it: Dial Tone—a continuous single tone with 450HZ frequency; 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. You can configure the tone generator manually. The exact explanation about the format and the meaning is described on the right of the interface.



3.5.5 DTMF

DTMF Detector	
Energy Difference of High-freq minus Low-freq (dB)	5
Energy Difference of Low-freq minus High-freq (dB)	9
Minimum Duration at ON (ms)	50
Maximum Interruption at ON (ms)	5
Center Frequency Error (%)	1
Lowest Energy Threshold (dB)	-30
Minimum Signal-to-noise Ratio Threshold (dB)	-3
DTMF Display via Channel Status	Enable
ABCD Detection	Enable
DTMF Generator	
DTMF Energy (dB)	-11
Duration at ON (ms)	100
Duration at OFF (ms)	32
Save	t

Figure 3-34 DTMF Detector Configuration Interface

See Figure 3-34 for the DTMF detector configuration. The table below explains the items shown in the above figure.

Item	Description	
Energy Difference of High-freq minus Low-freq	The allowed difference in dB for the DTMF high frequency energy level to surpass the low frequency energy level. Range of value: 0~24. The default value is 5.	
Energy Difference of Low-freq minus High -freq	The allowed difference in dB for the DTMF low frequency energy level to surpase the high frequency energy level. Range of value: 0~24. The default value is 9.	
Minimum Duration at ON	The shortest time that a valid tone has to last at ON state. Range of value: 10 \sim 2000, calculated by ms. The default value is 50.	
Maximum Interruption at ON	The longest time for a valid tone to stay interrupted at ON state. Range of value: 0 \sim 20, calculated by ms. The default value is 5.	



Center Frequency	The error threshold of the center frequency at ON state in the DTMF tone, with the	
Error	default value of 1.	
Lowest Energy	The energy threshold to trigger the DTMF detection. Range of value: -40 \sim -9,	
Threshold	calculated by dB. The default value is -30.	
Minimum Signal-to-noise Ratio Threshold	The signal-to-noise ratio threshold to trigger the DTMF detection. Range of value: -9 ~0, calculated by dB. The default value is -3.	
DTMF Display via Channel Status	Once this feature is enabled, the received/transmitted DTMF will be displayed as you put the mouse on the icon of channel status.	
ABCD Detection	Once this feature is enabled, the gateway can detect the DTMF digits A, B, C and D (Case-insensitive). The default value is disabled.	
DTMF Energy	Energy of the DTMF signal sent by the gateway. Range of value: -35~15, calculated by dB, with the default value of -11.	
Duration at ON	The duration of DTMF signal at on state. Range of value: 0~16383, calculated by ms, with the default value of 100.	
Duration at OFF	The duration of DTMF signal at off state. Range of value: 0~16383, calculated by ms, with the default value of 32.	

After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to <u>3.9.16 Restart</u> for detailed instructions.

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 3-35 Ringing Scheme Configuration Interface

See Figure 3-35 for the Ringing Scheme interface. The gateway will execute different ringing schemes according to the CallerID or Alert-Info.

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

Item	Description	
	The gateway will match the CallerID set in this item to that of the incoming call. If	
	they are matched, the current ringing scheme will be executed; otherwise, the	
CallerID	default ringing scheme (1 sec on and 4 sec off) will work.	
	The rule to fill in the CallerID is the same as that of <u>3.5.9 Dialing Rule</u> . Multiple	
	CallerIDs are supported; they should be separated by ","	
	The gateway will match the Alert-info value set in this item to that of the incoming	
Alert-Info Value	call. If they are matched, the current ringing scheme will be executed; otherwise,	
	the default ringing scheme (1 sec on and 4 sec off) will work	
	The ringing scheme can be "1,X,Y" or "2,X,Y,M,N", in which, the number 1 or 2	
denotes one group or two groups; X, M denote the duration at on state wh		
	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 3-36 Fax Configuration Interface (Disable by default)

See Figure 3-36 for the default fax mode configuration. The table below explains the items shown in the above figure.

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

See Figure 3-37 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	
T.30 ECM	Enable	
Min Duration of CNG(ms)	425	
Min Duration of CED(ms)	2600	
Save	set	

Figure 3-37 Fax Configuration Interface (T.38 Mode)

Users can configure the general fax parameters via this interface. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the service, do it immediately to apply the changes. Refer to <u>3.9.16 Restart</u> for detailed instructions. The table below explains the configuration items in Figure 3-37.

ltem	Description		
T20 Fax Dart	The port for T.38 faxing, providing two options: Use Original Voice Port and Use		
T38 Fax Port	New Port. The default setting is Use Original Voice Port.		
TOON	Version of T.38 which is defined by ITU-T. Range of value: 0~3, with the default		
T38 Version	value of 0.		
	The Negotiation mode of T.38, providing two options: Initiate Negotiation as Fax		
T38 Negotiation	Sender and Initiate Negotiation as Fax Receiver. The default value is Initiate		
	Negotiation as Fax Receiver.		
	Sets the maximum faxing rate for both receiving and transmitting. Range of value:		
Maximum Fax Rate	14400, 9600 and 4800, calculated by bps, with the default value of 14400.		
	Sets the train mode for T.38 fax. The optional values are transferredTCF and		
Fax Train Mode	localTCF, with the default value of transferredTCF.		
	Sets the error correction mode for T.38 fax. The optional values are		
Error Correction	t38UDPRedundancy (Redundancy Error Correction) and t38UDPFEC (Forward		
Mode	Error Correction), with the default value of t38UDPRedundancy.		
	Sets whether to enable the T.30 error correction mode. By default this feature is		
T.30 ECM	enabled.		



	As stipulated in the standard FAX CNG, the minimum duration of CNG is 500ms \pm
	15%, calculated by ms, with the default value of 425.
Min Duration of CNG	Note: Usually there is no need to modify it; please contact our technicians if
	necessary.
	As stipulated in the standard FAX CED, the minimum duration of CED is
Min Duration of CED	2600~4000ms, calculated by ms, with the default value of 2600.
Min Duration of CED	Note: Usually there is no need to modify it; please contact our technicians if
	necessary.

If you set *Fax Mode* to *Pass-through*, you can see the interface shown as Figure 3-38.

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

Figure 3-38 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 3-39 for the function key configuration interface. Here you can set a cluster of combination keys to query a related number.



Function			
1 difedon	Enable	Function Key	Mode
Device Function			
Query LAN1		*11*	Default 💌
Query LAN2		*12*	Default 💌
Query Phone Number		*20*	Default
Phone Test		*30*	Default
Switch Eth Device	V		
		50	Default
Password for Switch Eth Device	•••••		
Set LAN1		*61*	Default 💌
Set LAN2		*62*	Default 💌
Query WEB Port		*70*	Default 💌
Reboot		*#88921532*#	Default 💌
Query Missed Call Number		*71*	Default 👻
Service Available			
Blind Transfer		*010*	Default 💌
Call Forward Unconditional Activate		*030*	Default 💌
Call Forward Unconditional Deactivate		*031*	Default 💌
Call Forward Busy Activate	V	*040*	Default 💌
Call Forward Busy Deactivate	V	*041*	Default 💌
Call Forward No Reply Activate	V	*050*	Default 💌
Call Forward No Reply Deactivate	V	*051*	Default 💌
Do Not Disturb Activate	V	*060*	Default 💌
Do Not Disturb Deactivate		*061*	Default 💌
Register		*020*	Default
Unregister	V	*021*	Default 👻
Query Register Status		*022*	Default 💌
Note: 'Switch Eth Device' means to exchange or disabled). And don't forget to switch t			
	Sav	re	

Figure 3-39 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.

tandard Mode Characte	er Mode	Dialing Rule		
Check	Index	Dialing Rule	Description	Modify
	81	400xxxxxx	default	
	82	40[1-9]xxxxx	default	
	83	4[1-9]xxxxxx	default	
	84	800xxxxxx	default	
	85	80[1-9]xxxxx	default	
	86	8[1-9]x0000x	default	
	87	[2-3,5-7]xxxxxxx	default	
	88	1[3-5,7-8]xxxxxxxxx	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]xxxxxxxxx	default	
	96	0.2xxxxxxxxx	default	
	97	010xxxxxxxx	default	
	98	01[3-5,7-8]000000000	default	
	99		default	
heck All 🗮 Uncheck		Delete E Clear All lext Last Go to Page 1 🗸 1 Pages Total		Add N

Figure 3-40 Dialing Rule Configuration Interface (Standard)

See Figure 3-40 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 3-41 for the dialing rule adding interface.

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





The table below explains the items shown in Figure 3-41.

Item		Description			
Index	The unique in	The unique index of each dialing rule, which denotes its priority. A dialing rule with a			
Index	smaller index	value has a higher priorit	ty and will be checked earlier while matching.		
Description	Remarks for t	Remarks for the dialing rule. It can be any information, but cannot be left empty.			
	Up to 100 dial	Up to 100 dialing rules can be configured in the gateway, and the maximum length of			
	each dialing r	each dialing rule is 127 characters. See below for the meaning of each character in			
	the dialing rule	the dialing rule. The gateway will do instant matching for your dialing number based			
	on the dialing	rule and regard your dia	aling 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 exampl	e, 'xxx' denotes 3 random numbers.		
	""	'.' indicates a rando	m amount (including zero) of characters		
	"[]"	'[]' 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.			
Dialing Rule	" " 7	',' is used to separate numbers or number ranges, representing alternatives.			
	"*"	Only represents symb	00 "*".		
	"#"	"#" Only set it at the beginning of the string, representing symbol "#".			
		dialing rules already co iled information.	onfigured on the gateway for easy use. See		
	Priority	Dialing Rule	Description		
	99		Any number in any length.		
	98	01[3-5,7-8]xxxxxxxxx.	Any 12-digit number starting with 013, 014, 015, 017 or 018		
	97	010xxxxxxx	Any 11-digit number starting with 010		
	96	02xxxxxxxx	Any 11-digit number starting with 02		
	95	0[3-9]xxxxxxxxx	Any 12-digit number starting with 03, 04, 05, 06, 07, 08 or 09		
	94	120	Number 120。		
	93	11[0,2-9]	Number 110, 112, 113, 114, 115, 116, 117, 118 or 119		
	92	111xx	Any 5-digit number starting with 111		
	91	123xx	Any 5-digit number starting with 123		

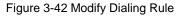


90	95xxx	Any 5-digit number starting with 95
89	100xx	Any 5-digit number starting with 100
88	1[3-5,7-8]xxxxxxxx	Any 11-digit number starting with 13, 14, 15, 17 or 18
87	[2-3,5-7]xxxxxxx	Any 8-digit number starting with 2, 3, 5, 6 or 7
86	8[1-9]xxxxx	Any 8-digit number starting with 81, 82, 83, 84, 85, 86, 87, 88 or 89
85	80[1-9]xxxxx	Any 8-digit number starting with 801, 802, 803, 804, 805, 806, 807, 808 or 809
84	800xxxxxx	Any 10-digit number starting with 800
83	4[1-9]xxxxx	Any 8-digit number starting with 41, 42, 43, 44, 45, 46, 47, 48 or 49.
82	40[1-9]xxxxx	Any 8-digit number starting with 401, 402, 403, 404, 405, 406, 407, 408 or 409
81	400xxxxxx	Any 10-digit number starting with 400

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

Click *Modify* in Figure 3-40 to modify the dialing rules. See Figure 3-42 for the dialing rule modification interface. The configuration items on this interface are the same as those on the *Add New Dialing Rule* interface.

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



To delete a dialing rule, check the checkbox before the corresponding index in Figure 3-40 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 3-40.

See Figure 3-43 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.



Standard Mode Character Mode	
Dialing Rule	
Note: The Dialing Rule contains such fields as Dialing Rule and Description.	
The priority decreases from top to bottom; adjacent fields are separated by a space; Symbol . denotes any string. Don't forget to save the configuration after your modification!	
400xxxxxx default	Ξ.
40[1-9]xxxxx default	
4[1-9]xxxxxx default	
800xxxxxx default	
80[1-9]xxxxx default	
8[1-9]xxxxx default	
[2-3,5-7]xxxxxx 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]xxxxxxxxx default	~
20 Items Total	
Save	

Figure 3-43 Dialing Rule Configuration Interface (Character)

3.5.10 Dialing Timeout

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

Figure 3-44 Dialing Timeout Info Interface

See Figure 3-44 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.	
Description	More information about the configuration item Inter Digit Timeout, such as the	
	reason for adopting the current value.	

Click *Modify* in Figure 3-44 to modify the dialing timeout info. See Figure 3-45 for the dialing timeout info modification interface. The configuration items on this interface are the same as those on the *Dialing Timeout Info Interface*.



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

Figure 3-45 Modify Dialing Timeout Info

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

3.5.11 Cue Tone

	Upload	
Upload a file of cue tone Note: The file should b 200KB in size.	File of cue tone for IVR	Browse Upload , 16-bit mono, A-law formatted, and less than

Figure 3-46 Cue Tone Interface

See Figure 3-46 for the Cue Tone interface. The table below explains the items shown in the above figure.

Item	Description
Upload a file of cue	Uploads a user-defined cue tone file to the gateway, including two options: Cue
tone	Tone for IVR and Cue Tone for Call Waiting.

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



3.5.12 Color Ring

Operation Info	۲
Quick Config	*
B VolP	*
{ Advanced	*
FXS	
FXO	
Tone Detector	
Tone Generator	
DTMF Detector	
Ringing Scheme	
Fax	
Function Key	
Dialing Rule	
Dialing Timeout	
Cue Tone	
Color Ring	

Figure 3-47 Color Ring Interface

By default, there is no available color ring on the gateway. See Figure 3-47. Click **Upload** to upload a new color ring manually. See Figure 3-48. You can upload the required color ring file to the gateway following this interface.

	Color Ring-Upload
Index	1
Description	ringtone1
Color Ring	Browse
Note: The file should be a wav file with 200KB in size.	8000Hz sampling rate, 16-bit mono, A-law formatted, and less than
Upic	ad Return

Figure 3-48 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. After upload, the color ring will appear on the color ring manage interface, see Figure 3-49.



Color Ring Manage				
Check	Index	Color Ring	Port	Modify
	1	ringtone1		
Check All = Uncheck All = Inverse = Delete = Clear All Upload 1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total				

Figure 3-49 Color Ring Manage Interface

Click *Modify* in Figure 3-49 to modify the configuration of the color ring or tick the *Upload* checkbox to change the old color ring file. See below for the color ring modification interface. The configuration items on this interface are the same as those on the *Color Ring Upload* interface.

	Color Ring-Modify		
Index	1		
Description	ringtone1		
Upload			
Color Ring	Browse		
Note: The file should be a wav file with 8000Hz sampling rate, 16-bit mono, A-law formatted, and less than 200KB in size.			
Sa	Ve Cancel		

Figure 3-50 Color Ring Modification Interface

To delete a color ring, check the checkbox before the corresponding index in Figure 3-49 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 3-50.

3.5.13 QoS

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

Figure 3-51 Differentiated Services Setting Interface

See Figure 3-51 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.	
Control Premium QoS	Sets the priority of the control premium for QoS. A control premium QoS with a 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 u	
	Save

Figure 3-52 Channel State Report Settings Interface

See Figure 3-52 for the Action URL interface, which is used to designate the server patch to report the on-hook or off-hook state of the FXS channel. You are allowed to designate two different server paths. After setting, the state will be reported to the designated server once any of the FXS channel hangs up or picks up a call. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations.

3.6 Port Settings

Port Settings includes five parts: *FXS*, *FXO*, *FXO Port Timer*, *Port Group* and *Advanced FXO Settings*. See Figure 3-53.



Figure 3-53 Port Settings

3.6.1 FXS

									FXS Setting	IS								
Port	Туре	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	Modify
1	FXS	8001			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	
2	FXS	8002			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	
30	FXS	8030			Disable	Disable	Disable			Enable	Disable	Unregistered	Enable	Disable		0	0	

Figure 3-54 FXS Settings Interface



See Figure 3-54 for the FXS settings interface. The list in the above figure shows the feature and properties of each FXS port. Click *Modify* in Figure 3-54 to modify the properties of the corresponding port. See Figure 3-55 for the FXS modification interface.

FXS-N	Modify
Port	1 ~
Туре	FXS
Register Port	Yes V
SIP Account	8001
Display Name	
Password	
Display Name preferred	Enable
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
Advanced Configuration	Enable
Talkback	Enable
Bound Number	
Ringing Parameter	RING_F25_75VRMS_0VDC_LPR_SIN V
Feed Voltage Parameter	DCFEED_48V_20MA
Impedance Parameter	ZSYN_200_680_100_30_0 V
Note: 'Auto Dial Number' goes into effect only if no	dialing occurs during 'Wait Time before Auto Dial'.

Modify Reset Cancel

Figure 3-55 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.



Register Port	Sets whether to register the port to the SIP server. When this item is set to <i>No</i> , the item <i>Reg Status</i> on the FXS settings interface (Figure 3-54) shows <i>Unregistered</i> ; when this item is set to Yes, the item <i>Reg Status</i>
	shows Failed or Registered.
SIP Account	When the port initiates a call to SIP, this item corresponds to the username of SIP. The default SIP account is 80XX among which XX represents the corresponding port number. For example, the default SIP account corresponding to Port 1 is 8001, and that corresponding to Port 32 is 8032.
	Set the content of the displayname field of the SIP message. If it doesn't set with
Display Name	any value, the displayname field will by default display the content of callerid.
	Registration password of the port. To register a port to the SIP server, both items
Password	SIP Account and Password must be filled in.
Display Name Preferred	In case this feature is enabled and the port group or the whole gateway is registered, if the display name set by the port is different from that set by the port group, the displayname in the sent SIP message will be the one set by the port. In case this feature is disabled, if the port group is registered, the displayname in the sent SIP message will be the displayname in the gateway is registered, the displayname in the sent SIP message will be the displayname in the sent SIP message will be the displayname in the sent SIP message will be the displayname in the sent SIP message will be 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 Auto Dial	The FXS port will dial the <i>Auto Dial Number</i> if there is no dialing operation after 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. The value must be
Gain	multiples of 3. Range of value: -24~24, calculated by dB, with the default value of 0.
Echo Canceller	The echo cancellation feature for a call conversation over the FXS channel. By 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> .
CID	CallerID. If this feature is enabled, the FXS port will send the CallerID of the incoming IP call together with the ringing tone to the corresponding station. The default setting is <i>enabled</i> . CallerID displays digits only and will filter out any other characters if exist.
Call Waiting	If this feature is enabled, the FXS port in conversation can accept another call from IP and keep it in the waiting state. Once the current conversation is finished and the station hangs up, the call in the waiting state will ring the station and wait for answer. The default setting is <i>disabled</i> .
DND	Do Not Disturb. If this feature is enabled, the FXS port will reply the 403 message to reject all incoming calls. The default setting is <i>disabled</i> .
Call Forward	The automatic call forward feature for the FXS port. Once this feature is enabled, the FXS port will forward incoming IP calls according to <i>FWD Type</i> . Note: To enable this feature, do not put the FXS port into a port group with other ports. The default setting is <i>disabled</i> .



		ons for the FXS port to forward incoming IP calls. The optional				
	values are: Option	Description				
	Unconditional	The FXS port will forward all incoming IP calls to the preset FWD Num immediately when it receives them.				
FWD Type	Busy	The FXS port will forward incoming IP calls to the preset <i>FWD</i> <i>Num</i> if it is busy upon receiving them.				
	No Reply	The FXS port will forward incoming IP calls to the preset <i>FWD</i> <i>Num</i> if the corresponding station does not answer them in a designated time period (i.e. <i>Time for No Reply Forward</i>). Only when this forward condition is selected does the configuration item <i>Time for No Reply Forward</i> become valid.				
FWD Num	which the incoming IP call is forwarded. If the <i>Call Forward</i> feature tem cannot be left empty.					
Color Ring	Sets whether to enable the color ring feature or not, with the default setting of being <i>disabled</i> . Note: Only when there are available color rings will appear this item.					
Color Ring Index	The index of the	color ring which will be quoted by the current FXS port.				
Talkback	With this feature enabled and a number bound, the port can talk back to its bound number. That is, they can start a call with each other as soon as picking up the phone. The default setting is <i>disabled</i> . Note: This feature is only used in the case of channel registration.					
Bound Number	Sets the bound r	number for talkback.				
Ringing Parameter	Sets the ringing parameter for the FXS module. The default value is <i>RING_F25_75VRMS_0VDC_LPR_SIN</i> Note: Usually there is no need to modify it; please contact our technicians if necessary.					
Feed Voltage Parameter	voltage parameter for the FXS module. The default value is 20MA. here is no need to modify it; please contact our technicians if					
Impedance Parameter	ZSYN_200_680_	bedance for the FXS module. The default value is _100_30_0. here is no need to modify it; please contact our technicians if				
	necessary.					

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 3-56 below for the FXS batch modification interface. The configuration items on this interface are the same as those on the FXS modification interface (Figure 3-55).



Starting Port	1 ~
Ending Port	30 ~
	30 🗸
Register Port	Yes 🗸
Starting SIP Account	
Starting Display Name	
Starting Authentication Password	
Display Name Preferred	Enable
SIP Account Batch Rule	Increase
SIP Account Batch Step Size	1
Display Name Batch Rule	Increase V
Display Name Batch Step Size	1
Authentication Password Batch Rule	Increase V
Authentication Password Batch Step Size	1
Auto Dial Number	Enable
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	Enable
Advanced Configuration	Enable
Ringing Parameter	
Feed Voltage Parameter	RING_F25_75VRMS_0VDC_LPR_SIN DCFEED_48V_20MA
Impedance Parameter	
importance i arameter	ZSYN_200_680_100_30_0

Save

Cancel

Figure 3-56 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 The starting authentication password in the batch setting. Password The starting authentication password 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.			
Display Name Batch Rule	The rule for batch setting the display name, including <i>Increase</i> , <i>Decrease</i> and <i>All Same</i> three options.			
Display Name Batch Step Size	Sets the increase or decrease step size of the display name in the batch setting.			
Authentication Password	The rule for batch setting the authentication password, including Increase,			
Batch Rule	Decrease and All Same three options.			
Authentication Password	Sets the increase or decrease step size of the authentication password 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

						FXU Sei	ungs						(
Port	Type	SIP Account	Display Name	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	Polarity Reversal Detection	Input Gain	Output Gain	Modify
27	27 FX0 8027 Two Stages Dialing for Incoming Call Disable Enable Enable Unregistered Enable Disable 0 0 0												
28	FXO	8028		Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	Disable	0	0	
29	FXO	8029		Two Stages Dialing for Incoming Call		Disable	Enable	Unregistered	Enable	Disable	0	0	
3 Items T	Batch Modify												

Figure 3-57 FXO Settings Interface

See Figure 3-57 for the FXO settings interface. The list in the above figure shows the feature and properties of each FXO port. Click *Modify* in Figure 3-57 to modify the properties of the corresponding port. See Figure 3-58 for the FXO modification interface.



FXO-Modify	
Port Type	27 ∨ FXO
Register Port SIP Account	Yes 8027
Display Name Password	
Display Name preferred	Enable
Connection Method Bound Number	Static Binding for OI
Input Gain (dB) Output Gain (dB) Echo Canceller	0 0 Imable
Forbid Outgoing Call Caller ID Detection	✓Enable
Polarity Reversal Detection	Enable
Modify Reset	Cancel

Figure 3-58 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.
Dominton Domt	When this item is set to No, the item Reg Status on the FXO settings interface (Figure
Register Port	3-57) 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 XX
SIP Account	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.
	Set the content of the displayname field of the SIP message. If it doesn't set with any
Display Name	value, the displayname field will by default display the content of callerid.
	Registration password of the port. To register a port to the SIP server, both items SIP
Password	Account and Password must be filled in.



	In case this feature is enabled and the port group or the whole gateway is registered, if							
	the display names set by the port are different from that set by the port group, the							
Display Name	displayname in the sent SIP message will be the one set by the port. In case this feature							
Preferred	is disabled, if the	port group is registered, the displayname in the sent SIP message will						
	be the display na	ame set by the port group; if the whole gateway is registered, the						
	displayname in th	e sent SIP message will be the displayname of the gateway.						
Server Index	The index of the S	SIP server which will be quoted by the current FXO port.						
	FXO connection r	nethods include:						
	Option	Description						
		Bind the number which corresponds to an FXS port to an FXO						
	Static	port. The number will be listed in the Bound Number column. This						
	Binding	helps to achieve the corresponding binding between an FXO port						
		and an FXS port (two-way).						
		Under this mode, an incoming call from an FXO port will go into						
Connection Method	Two Stages	the IVR system. Then IVR will play a speech prompt "Please dial						
	Dialing	the extension number". If you fail to input the correct target						
	Mode	station number before IVR finishes the third repeat of the prompt,						
	(default)	the FXO will hang up the call automatically; otherwise, the						
	(acidali)	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, Output	Station feature is enabled on the SIP Settings interface.Adjusts the gain of the voice input to/ output from the FXO port. The value must be							
Gain	multiples of 3. Range of value: $-24 \sim 24$, calculated by dB, with the default value of 0.							
Gain	· · · · · · · · · · · · · · · · · · ·							
Echo Canceller	The echo cancellation feature for a call conversation over the FXO channel. By default,							
Forbid Outgoing	this feature is enabled and the effect can reach 128ms.							
Call	If this feature is enabled, the FXO port will be forbidden to call out. The default setting is <i>disabled</i> .							
		nabled the EVO part will detect the caller IDs from the incoming calle						
Caller ID Detection	If this feature is enabled, the FXO port will detect the caller IDs from the incoming calls.							
	The default setting	-						
	Once this feature is enabled, only when the FXO port detects the polarity reversal signal							
Polarity Reversal	-	iding channel go into the talking state. The default setting is <i>disabled</i> .						
Detection	Note: This feature and the <i>Two Stages Dialing</i> feature cannot be enabled at the same							
	time.							

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

Or you can click **Batch** to modify several pieces of FXO settings at the same time. See Figure 3-59 below for the FXO batch modification interface. The configuration items on this interface are the same as those on the FXO modification interface (Figure 3-58).



FXO-Batch	
Starting Port	27 ~
Ending Port	29 ~
Register Port	Yes
Starting SIP Account	
Starting Display Name	
Starting Authentication Password	
Display Name Preferred	
SIP Account Batch Rule	Increase V
SIP Account Batch Step Size	1
Display Name Batch Rule	Increase V
Display Name Batch Step Size	1
Authentication Password Batch Rule	Increase V
Authentication Password Batch Step Size	1
Connection Method	Static Binding for OI V
Starting Bound Number	
Batch Rule of Bound Number	Increase V
Batch Step Size of Bound Number	1
Input Gain (dB)	0
Output Gain (dB)	0 Senable
Echo Canceller	
Forbid Outgoing Call	■Enable
Caller ID Detection	
Polarity Reversal Detection	
Save	Cancel

Figure 3-59 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	The starting authentication password in the batch setting.		
Password			
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.		
Display Name Batch Rule	The rule for batch setting the display name, including <i>Increase</i> , <i>Decrease</i> and <i>All</i>		
	Same three options.		
Display Name Batch Step Size	Sets the increase or decrease step size of the display name in the batch setting.		
Authentication Password	The rule for batch setting the authentication password, including Increase,		
Batch Rule	Decrease and All Same three options.		
Authentication Password	Sets the increase or decrease step size of the authentication password in the batch		
Batch Step Size	setting.		
Batch Rule of Bound	The rule for batch setting the bound number, including Increase, Decrease and		
Number	Use the same number three options.		
Batch Step Size of Bound Number	Sets the increase or decrease step size of the bound number in the batch setting.		

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

3.6.3 FXO Port Timer

FXO Port Timer								
Check	Port	Туре	Unit	Max Time(Single Call)	Max Time(Total Calls)	Used Call Time	Clear Call Time	Modify
	1	FXO	60s	60	60/Month	0	1th day of month 00:00	
	2	FXO	60s	Unlimited	Unlimited			
	5	FXO	60s	Unlimited	Unlimited			
	6	FXO	60s	Unlimited	Unlimited			
Check All Uncheck All Clear reflash								

Figure 3-60 FXO Port Timer Interface

See Figure 3-60 for the FXO Port Timer interface, which displays such information as the max call time limit for a single call, the max call time limit for the total calls on each FXO port, as well as the timer clear cycle. Click Modify for each port in Figure 3-60 to modify the timer settings. See Figure 3-61.



Timir	ng
Port	1
Unit	60s 💌
Time Limit on a Single Call Max Call Time	ODisable enable 60
Time Limit on Total Calls Max Call Time	ODisable Enable 60
Timing Cycle Clear Set Spent Call Time	Month ▼ 1st ▼ 00 ▼ 00 ▼
SIP Code Reply	486
Apply to Other Ports	 ●Port ©Port Group ✓ 01 □ 02 □ 05 □ 06
Modify	Return

Figure 3-61 FXO Port Timing Setting Interface

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

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

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



3.6.4 FXO List Timer

Port Timing							Setting	
Port 5		5	6		7		8	
Index[Already Used]								
Timing Rule						Add		
Check	Index			Matching Rule		Calling Rule	Modify	
	1	123			Prefix Matching	100	0 Minute/Month Clear(1st 00:00)	
Check All Delete Refresh								

Figure 3-62 FXO List Timer Interface

See Figure 3-62 for the FXO List Timer interface, which displays the index information of the FXO port in timing. Click the **Setting** button on the top right corner to set the timer. See Figure 3-63. Click the **Add New** button at the bottom to add the list timing rule. See Figure 3-64.

Set Port Timing Rule						
	Rule Index 0 0	6 8	Rule Index 0 0 0 by ","; 0 means not to use			
Save Reset Return						

Figure 3-63 List Timing Setting Interface

Add	Name List Timing Rule
Index	2 ~
Number	(Separated by ",")
Import Number	Browse Import
Number Matching Rule Max Call Time Timing Cycle Clear	Prefix Matching (Minute) Month 1st 00 <>
Save	Reset

Figure 3-64 List Timing Rule Adding Interface

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

ltem	Description
Rule Index	The index of timing rule, used for the FXO port in list timing.
Set Spent Call Time	The length of the time already used in this rule.
Import Number	Import the matching numbers.



Number Matching	There are two number matching modes: <i>Prefix Matching</i> and <i>Whole Words only</i> .		
Rule			
Max Call Time	The maximum call time in this rule		
Timing Cycle	The timing cycle in this rule		
Clear	The time to clear the timer within the timing cycle in this rule		

Click *Save* to save the settings into the gateway; click *Reset* to restore the configurations; click *Return* to cancel the settings.

3.6.5 Port Group

Operation Info	*
🕂 Quick Config	*
🚟 VolP	*
Advanced	*
i Port	*
FXS	
FXO	
Port Group	

Figure 3-65 Port Group Setting Interface

See Figure 3-65 for the Port Group Settings interface. By default, there is no available port group on the gateway. A port group is a set containing one or more than one port, having such properties as **Port Selection** and **Authentication Mode** the same 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 3-66 for the port group adding interface. Note that a port which has been occupied by one port group cannot be chosen by others.



Index 1 Description default Register Port Group YES SIP Account Display Name Password Authentication Username 1:201.123.115.233 Server Index 1:201.123.115.233 Authentication Mode Do Not Register Port Select Mode Ringing by Turns	
Register Port Group YES SIP Account	
SIP Account Display Name Password Authentication Username Server Index 1:201.123.115.233	
Display Name Password Authentication Username Server Index Authentication Mode Do Not Register	
Password	
Authentication Username Server Index 1:201.123.115.233 Authentication Mode Do Not Register	
Server Index 1:201.123.115.233	
Authentication Mode Do Not Register	
De Net Register	
Port Select Mode Ringing by Turns	
Rule for Ringing by Turns 1,2,3,4,5,5,1,2	
Timeout for Ringing by Turns (s) 20	
Port Reused by Multiple Groups	
Port Port 1(FXS) Port 2(FXS) Port 3(FXS) Port 4(FX) Port 5(FXS) Port 6(FXS) Port 7(FXS) Port 8(FX) Port 9(FXS) Port 10(FXS) Port 11(FXS) Port 12(F) Port 13(FXS) Port 14(FXS) Port 15(FXS) Port 16(F) Port 17(FXS) Port 13(FXS) Port 14(FXS) Port 15(FXS) Port 20(F) Port 17(FXS) Port 17(FXS) Port 22(FXS) Port 23(FXS) Port 24(F) Port 25(FXS) Port 26(FXS) Port 27(FXS) Port 28(F) Port 29(FXS) Port 30(FXS) Port 31(FXS) Port 32(F) Check All Inverse Check All FXO Ports Check All FXS Ports	FXS) (FXS) (FXS) (FXS) (FXS) (FXS) (FXS)
Save	

Figure 3-66 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
maex	routing rules and number manipulation rules to correspond to port groups.
Description	More information about each port group, with default value of default.
	To register the port group to the SIP server. Only when this configuration item is set
Register Port Group	to Yes can you see the configuration items SIP Account and Password.
0/5.4	When the port group initiates a call to SIP, this item corresponds to the username of
SIP Account	SIP.
	Set the content of the displayname field of the SIP message. If it doesn't set with
Display Name	any value, the displayname field will by default display the content of callerid.



	Registration password or	f 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 username	e of a port, used to register the port to the SIP server when			
Authentication	IMS network is enabled.				
Username	Note: This item appears only when IMS Network is enabled.				
Server Index	The index of the SIP ser	ver which will be quoted by the current port group.			
	Sets the way for SIP to r	make outgoing calls (Tel→IP) on the gateway.			
	Option	Description			
	Do Not Register (default)	SIP initiates a call in a point-to-point mode.			
Authentication Mode	Register Gateway	SIP initiates a call with the registered SIP account and password of the whole gateway. (Refer to $3.4.1$ SIP for gateway registration.)			
	Register Port Group	SIP initiates a call with the registered SIP account and password of the port group.			
	Register Port	SIP initiates a call with the registered SIP account and password of the port.			
	Registration status of the port group. When Register Port Group is set to No, the				
Register Status	value of this item is Un	nregistered; when Register Port Group is set to Yes, the			
	value of this item may be	e Failed or Registered.			



	When the port group re	eceives 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						
	corresponding meanings are described in the table below.						
	Option	Description					
		Search for an idle port in the ascending order of the port					
		number, starting from the minimum. If no match is found,					
	Increase (default)	search repeatedly until finding a port which is allowed to					
		enter the call waiting state.					
		Search for an idle port in the descending order of the port					
		number, starting from the maximum. If no match is found,					
	Decrease	search repeatedly until finding a port which is allowed to					
		enter the call waiting state.					
		Provided Port N is the available port found last time.					
		Search for an idle port in the ascending order of the port					
	Cyclic Increase	number, starting from Port N+1. If no match is found,					
Port Select Mode		search repeatedly until finding a port which is allowed to					
		enter the call waiting state.					
		Provided Port N is the available port found last time.					
		Search for an idle port in the descending order of the port					
	Cyclic Decrease	number, starting from Port N-1. If no match is found,					
		search repeatedly until finding a port which is allowed to					
		enter the call waiting state.					
	Group Ringing	Ring all the idle FXS ports in this port group.					
		Ring the ports in this port group according to the <i>Rule for</i>					
		Ringing by Turns which can be user-defined. Refer to the					
		format of the rule in Figure 3-66. 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 p	port group is ringing, another channel in the same port group					
	-	d shortcut set by this item to transfer the call from the ringing					
Preemptive Answer	channel to the current channel.						
Keyboard Shortcut	Note: This item will become invalid if the gateway works under the port select mo						
	Group Ringing or Ring						
Port Reused by							
Multiple Groups	Once this feature is en	abled, a port can be added to different port groups.					
	The ports in the port gr	oup. If the checkbox before a port is grey, it indicates that the					
	port is not available of	or has been occupied. Once the feature "Port Reused by					
	Multiple Groups" is en	abled, a port which has been occupied is still available for					
Port	other port groups. All s	selected ports for a port group will be displayed in the Ports					
	column in Figure 3-6	7. 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. See Figure 3-67 for the port group list with saved configurations.

	Port Group Settings									
Check	Index	Description	SIP Account	Display Name	Authentication Username	Ports	Port Select Mode	Rule for Ringing by Turns	Timeout for Ringing by Turns (s)	Pre
	1	default				1,2	Increase			
•	Image: A state of the state									
Check A	Check All 🗄 Uncheck All 🗄 Inverse 🗏 Delete 🗏 Clear All 🛛 Add New									
1 Item Tot	I Item Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 💌 1 Pages Total									

Figure 3-67 Port Group List

Click *Modify* at the end of the list in Figure 3-67 to modify the properties of a port group. See Figure 3-68 for the port group modification interface. The configuration items on this interface are the same as those on the *Add New Port Group* interface.

	Port Group-Modify
Index	1
Description	default
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	
For 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(FXS) Port 10(FXS) Port 11(FXS) Port 12(FXS)
	Port 13(FXS) Port 14(FXS) Port 15(FXS) Port 16(FXS)
	Port 17(FXS) Port 18(FXS) Port 19(FXS) Port 20(FXS)
	Port 21(FXS) Port 22(FXS) Port 23(FXS) Port 24(FXS)
	Port 25(FXS) Port 26(FXS) Port 27(FXS) Port 28(FXS)
	Port 29(FXS) Port 30(FXS) Port 31(FXS) Port 32(FXS)
	Check All Inverse Check All FXO Ports Check All FXS Ports
Modify	Reset Cancel

Figure 3-68 Modify Port Group



To delete a port group, check the checkbox before the corresponding index in Figure 3-67 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 3-67.



3.6.6 Advanced FXO Settings

	Advanced FXO Settings
Mailbox Settings Mailbox Account Password Outgoing(SMTP) SSL Recipient Subject Content	asd@sfd.com Port 25 Warning: gateway port disconn gateway:[devinfo],port[port] disconnection
Sending	test
Note:1,Multiple recipie 2,In subject and conter	nts must be separated by ',' nt: nts the device information i.e. device type and serial number.
FXO Off-line Alarm Port	
	Port 1(FXO) Port 2(FXO) Port 3() Port 4() Port 5(FXO) Port 6(FXO) Port 7() Port 8() Port 9(FXS) Port 10(FXS) Port 11() Port 12() Port 13(FXS) Port 14(FXS) Port 15() Port 16() Port 17() Port 18() Port 19() Port 20() Port 21() Port 22() Port 23() Port 24() Port 25() Port 26() Port 27() Port 32() Port 29() Port 30() Port 31() Port 32()
Blacklist of FXO Incoming	
Calls	
Blacklist	
Processing Mode	Hang up after pick-u 🗸
Hang-up Delay (ms)	2000
	sts must be separated by ',' tion" feature should be enabled for the port to activate the blacklist
	Save Reset



Figure 3-69 Advanced FXO Settings Interface

See Figure 3-69 for the Advanced FXO Settings interface. The table below explains the configuration items on the interface.

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

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

3.7 Route Settings

Route Settings is used to specify the routing rules for calls on two directions: $IP \rightarrow Tel$ and $Tel \rightarrow IP$. See Figure 3-70.

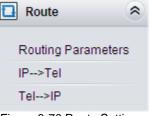


Figure 3-70 Route Settings

3.7.1 Routing Parameters

Routing Parameters					
IP->TEL	Route before Number Manipulate				
TEL->IP	Route before Number Manipulate				
Route Detection Cycle (s)	0				
	Save				



Figure 3-71 Routing Parameters Configuration Interface

See Figure 3-71 for the routing parameters configuration interface. On this interface, you can set the routing rules for calls respectively on two directions IP \rightarrow Tel and Tel \rightarrow IP to be routing before or after number manipulation. The default value is *Route before Number Manipulate*. The gateway will send the option message to detect whether the TEL->IP routing is valid or not after setting the Route Detection Cycle. If the remote address doesn't respond this option message within the set cycle, this routing will be regarded as invalid and the outgoing calls won't be routed to this TEL->IP routing.

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

3.7.2 IP to Tel

Operation Info	۲	Standard Mode Character Mode
Quick Config	*	
S VolP	*	
	*	No available routing rule!
i Port	*	Add New
Route	*	
Routing Paramete	ers	
IP>Tel		
Tel>IP		

Figure 3-72 IP→Tel Routing Rule Configuration Interface (Standard)

See Figure 3-72 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 3-73. You may use the default values of all the configuration items herein.

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

Figure 3-73 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> .		
Course ID	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		
Route by Number	interface. In such case, the configuration item Call Destination goes invalid and		
	shows Route by Number on the routing rule configuration interface. The default		
	setting is <i>disabled</i> .		
Call Destination	Port group to which the call will be routed.		

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

See Figure 3-74 for the IP \rightarrow 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 CallerID Prefix CalleeID Prefix Call Destination Description							
	63	* *		*	default			
Observative and a second								
Check All = Uncheck All I Inverse = Delete = Clear All Additional Context All Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 V 1 Pages Total						Add Nev		

Figure 3-74 IP→Tel Routing Rule Configuration Interface

Click **Modify** in Figure 3-74 to modify a routing rule. The configuration items on the IP \rightarrow Tel routing rule modification interface are the same as those on the **Add New Routing Rule (IP\rightarrowTel)** interface. Note that the item **Index** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-74 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 3-74.

See Figure 3-75 for the IP \rightarrow Tel Routing Rule Configuration Interface under the Character mode. You can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

Standard Mode	Character Mode	
		IP->Tel Routing Rule
Note: The routing i	nformation contain	s such fields as Source IP, CallerID Prefix, CalleeID Prefix, Route by Number, Destination Port Group and Description.
		tom; adjacent fields are separated by a space
		and CalleeID Prefix indicates any IP address or string; When Route by Number is set to 1, the Destination Port Group will be enabled.
		after your modification!
* * * 0 0 defa	ult	
1 Items Total		
		Save

Figure 3-75 IP→Tel Routing Rule Configuration Interface (Character)

3.7.3 Tel to IP

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

Figure 3-76 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-76 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 3-77. You may use the default values of all the configuration items herein except for *Destination IP* and *Destination Port*.



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

Figure 3-77 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.



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See Figure 3-78 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	Check Index Call Initiator CallerID Prefix CalleeID Prefix Destination IP Destination Port Description								
	63	*	*	*	192.168.1.101	5060	default		
Check All = Uncheck All = Inverse = Delete = Clear All A								Add New	
1 Items Total	Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 🔽 1 Pages Total								

Figure 3-78 Tel→IP Routing Rule Configuration Interface

Click **Modify** in Figure 3-78 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 \rightarrow IP) interface. Note that the item **Index** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-78 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 3-78.

See Figure 3-79 for the Tel \rightarrow IP Routing Rule Configuration Interface under the Character mode. You can edit the routing rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.

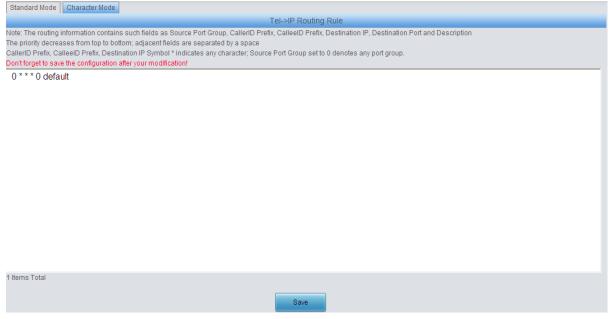


Figure 3-79 Tel→IP Routing Rule Configuration Interface (Character)

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 3-80.





Figure 3-80 Number Manipulation

3.8.1 IP to Tel CallerID

Standard Mode Character Mode											
IP->Tel CallerID Number Manipulation Rule											
Check	Index	Call Initiator	CallerID Prefix	CalleelD Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	Description	Modify
	63	*	*	*	0	0	20			default	
<											
Check All 🚊 Uncheck All 🗏 Inverse 🗏 Delete 🗮 Clear All 🕹 Add New											
Items Tot	al 20 lter	ms/Page 1/1 F	First Previous Ne	ext Last Go to Pa	ge 1 💌 1 Pages Total						

Figure 3-81 IP→Tel CallerID Manipulation Interface (Standard)

See Figure 3-81 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 3-82 for the IP \rightarrow Tel CallerID manipulation rule adding interface. You may use the default values of all the configuration items herein.



IP->Tel CallerID								
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 3-82 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.				
Decerintien	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.				



r	
	A string of characters at the beginning of the caller/called party number. It can be a
	specific string consisting of digits 0~9, "[*]", "#" or character ranges defined by []. '[]'
	represents a character within the range it defines. Values in [] only can be digits
	'0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to
CallerID Prefix,	indicates any character between these two characters. ',' is used to separate
CalleeID Prefix	characters or character ranges, representing alternatives.) For example, 057[1-3,6]
	represents the string 0571, 0572, 0573 or 0576. Also these items can be set to "*"
	which indicates any string. These two configuration items together with Call
	<i>Initiator</i> specify a number manipulation rule for calls.
	Note: "[*]" represents DTFM symbol *, while "*" represents any string.
	The amount of digits to be deleted from the left end of the number. If the value of
Stripped Digits from	this item exceeds the length of the current number, the whole number will be
Left	deleted. The default value is 0.
	The amount of digits to be deleted from the right end of the number. If the value of
Stripped Digits from	this item exceeds the length of the current number, the whole number will be
Right	deleted. The default value is 0.
	The amount of digits to be reserved from the right end of the number. Only when the
Reserved Digits	value of this item is less than the length of the current number will some digits be
from Right	deleted from left; otherwise, the number will not be manipulated. The default value
	is 20.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

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

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

Click **Modify** in Figure 3-81 to modify a number manipulation rule. See Figure 3-83 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 Call	erID
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 3-83 Modify IP→Tel CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-81 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 3-81.

See Figure 3-84 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
1ltems Total
Save

Figure 3-84 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 3-86 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 3-81).

Standard Mode Character Mode											
IP->Tel CalleeID Number Manipulation Rule											
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	Description	Modify
	63	±	±	*	0	0	20			default	
Check All Uncheck All Inverse Delate Clear All Add New Add New Items Total 20 items/Page 1/1 First Previous Next Last Go to Page 1 1 1 Pages Total											

Figure 3-85 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
See.



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

3.8.3 Tel to IP CallerID

Standard Mode Character Mode											
Tel->IP CallerID Number Manipulation Rule											
Check	Index	Call Initiator	CallerID Prefix	CalleelD Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	Description	Modify
	63	*	*	*	0	0	20			default	
Check All I Uncheck All I Inverse Digitie Clear All Add New											
1 Items Tot	al 20 Iter	ms/Page 1/1 F	First Previous No	ext Last Go to Pa	ge 1 💙 1 Pages Total						

Figure 3-87 Tel→IP CallerID Manipulation Interface (Standard)

See Figure 3-87 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 3-88 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 CallerIE)
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 3-88 Add Tel→IP CallerID Manipulation Rule

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

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



matches several number man	ipulation rules, it will be processed according to the
one with the highest priority.	
More information about each	number manipulation rule, with the default value of
Description default.	
Source Port Group Port group from which the cal	I is initiated. This item can be set to a specific port
(Call Initiator) group or '*' which indicates any	<i>i</i> port group.
A string of characters at the be	ginning of the caller/called party number. It can be a
specific string consisting of digi	its 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 betw	een these two characters. ',' is used to separate
CalleelD Prefix characters or character ranges	, representing alternatives.) For example, 057[1-3,6]
represents the string 0571, 05	72, 0573 or 0576. Also these items can be set to "*"
which indicates any string.	These two configuration items together with Call
Initiator specify a number man	nipulation rule for calls.
Note: "[*]" represents DTFM sy	/mbol *, while "*" represents any string.
Stripped Digits from	eted from the left end of the number. If the value of
Left	of the current number, the whole number will be
deleted. The default value is 0.	
Stringed Digits from	eted from the right end of the number. If the value of
Stripped Digits from Bight 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 rese	rved 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, th	e number will not be manipulated. The default value
is 20.	
Prefix to Add Designated information to be a	dded to the left end of the current number.
Suffix to Add Designated information to be a	dded 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 3-87 to modify a number manipulation rule. See Figure 3-89 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 Callerl	D
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 3-89 Modify Tel→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-87 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 3-87.

See Figure 3-90 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. Dont forget to save the configuration after your modification!
0 ** 0 0 20 <@#> <@#> default
1 Items Total

Figure 3-90 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 3-91, Figure 3-92 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 3-87).

Standard I	Mode	Character Mode	9								
					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	±	*	0	0	20			default	
<				1	•	•		1			>
Check A			Inverse	Delete	Clear All					Add I	New

Figure 3-91 Tel→IP CalleeID Manipulation Interface (Standard)

Standard Mode Character Mode
Tel->IP CalleeID Number Manipulation Rule
Note: The Number Manipulation Rule contains such fields as Source Port Group, CallerID Prefix, CalleeID Prefix, Delete Digits from Left, Delete Digits from Right, Reserve Digits from Right, Add Prefi 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. Dont forget to save the configuration after your modification!
0 ** 0 0 20 <@#> <@#> default
1 Items Total
Save



Figure 3-92 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 3-93 for details.

System Tools 🙈
Management
Config File
Network
Upgrade
Signaling Capture
Call Log
Operation Log
Backup & Upload
Factory Reset
System Monitor
Centralized Manage
Access Control
Call Test
PING Test
TRACERT Test
Change Password
Restart
Figure 3-93 System Tools



3.9.1 Management

Managemei	nt Parameters
WEB Management	
WEB Port	80
Access Setting	Allow All IPs V
SYSLOG Parameters	
SYSLOG	• Yes O No
Server Address	127.0.0.1
SYSLOG Level	INFO V
Remote Data Capture Config	
Remote Data Capture	● Yes ○ No
CDR Parameters	
Send CDR	●Yes ONo
Server Address	127.0.0.1
Server Port	3
Time Parameters	
NTP	●Yes ONo
NTP Server Address	time.nist.gov
Synchronizing Cycle	3600
Daily Restart	●Yes ONo
Restart Time	0 🗸 h 0 🗸 m
System Time	Modify 2018-01-17 15:35:19
Time Zone	GMT+8:00 (Beijing, Singapore, Tair 🗸
Save	Reset

Figure 3-94 Management Parameters Setting Interface

See Figure 3-94 for the Management Parameters Setting interface. The table below explains the items shown in the above figure.

ltem	Description
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.
Access Setting	Sets the IP addresses which can access the gateway via WEB. By default, all IPs are allowed. You can set an IP whitelist to allow all IPs within it to access the gateway freely. Also can set an IP blacklist to forbid all IPs within it to access the gateway.



SYSLOG	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 and INFO.
SYSLOG Level	The default value is <i>INFO</i> .
Remote Data	
Capture	是否开启远程抓包服务程序
	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 disabled.
NTP Server Address	Sets the Server address for NTP time synchronization.
Synchronizing Cycle	Sets the cycle for NTP time synchronization. The default value is 3600.
	Sets whether to restart the gateway regularly every day at the preset Restart Time .
Daily Restart	By default, this feature is disabled.
Restart Time	Sets the time to restart the gateway regularly.
	The system time. Check the checkbox before <i>Modify</i> and change the time in the
System Time	edit box.
Time Zone	The time zone of the gateway.



3.9.2 Configuration File

	SMGConfig.ini	•
Config File		
[Version]		•
GWSvrV=1.2.12_20130719		
WebV=1.2.11_20130719		
CpldV=0.3		Ξ
[DbgLog]		
LogLevel=3		
LogCreatePeriod=24		
LogMaxPeriod=4		
LogMaxPeriodSaved=5		
LogOverWrite=1		
LogDirectory=/dev/shm/shcti		
LogType=3		
[WebCtrl] LocalAddress=127.0.0.1		
LocalPort=1001		
MailAlarm=1		
MailUser=asd@sfd.com		
MailPassword=		
SMTPAddress= MailReceivers=		
ChinfoPath=/usr/local/SMG/ch.data.php		
PopInfoPath=/usr/local/SMG/pop.data.php		
[UserInfo]		
UserName=BqtTPNLUr/23x1wC/w		
Pwd=BqtTPNLUr/23x1wC/w		
[Monitor]		
LocalAddress=127.0.0.1		
LocalPort=1002		
AutoExec=1		
CheckSvrThread=1		
ExecPath=E:\recorder\Recorder\Output\BIN\		
UpgradeExecPath=/usr/local/apache/htdocs/RecUpgrade		
[INIPath]		
ShConfig=ShConfig.ini		
[DBPARAM]		+
Save Reset		
Note: You shall restart system to validate the modified configuration file!		

Figure 3-95 Configuration File Interface

See Figure 3-95 for the Configuration File interface, including two files: SMGConfig.ini and ShConfig.ini. You can check and modify the items in these configuration files through this interface. Configurations about the gateway server, such as route rules, number manipulation and so on, are included in SMGConfig.ini; configurations about the board are included in ShConfig.ini. You can modify these configurations on the interface directly, and then click **Save** to save the above settings into the gateway or click **Reset** to restore the configurations.



3.9.3 Network

	Netwo	ork Settings
LAN 1		
	Network Type:	Static
	IP Address (I)	192.168.1.101
	Subnet Mask (U)	255.255.255.0
	Default Gateway (D)	192.168.1.1
	DNS Server (P)	0.0.0.0
	Speed and Duplex Mode	Automatic Detection
LAN 2		Enable
	Save	Reset
Note: Th		er saving the current setting. Please log in again using your P address has been modified!
	new iP address if the P	- audress has been modilied!

Figure 3-96 Network Settings Interface

See Figure 3-96 for the network settings interface. A gateway has two LANs, each of which can be configured with independent 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. By default, LAN1 is enabled and LAN2 is disabled.

Note:

- 1. The values of the *IP address*, *Subnet Mask*, *Default Gateway* and *DNS Server* shown in Figure 3-96 are all factory settings. The IP Address for LAN 1 and that for LAN 2 cannot be in the same segment.
- 2. LAN2 is disabled by default for the gateway Version 1.3.3 or above. If you want to use LAN2, please log in the gateway through LAN1 first, and then modify the network settings to enable LAN2.

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



3.9.4 Upgrade

	Current Version
Serial Num	00000057
WEB	Version 1.7.1_Release2017042109
Service	Version 1.7.1_Release2017042109
U-boot	Version Nov 24 2014 - 09:24:52
Kernel	Version #187 PREEMPT Tue Jan 19 10:35:29 CST 2016
Product Type	1032(RJ11)
Select an U	pdate File Browse
	Update Reset

Figure 3-97 Upgrade Interface

See Figure 3-97 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 3-98.

Serial Num	00000057
WEB	Version 1.7.1_Release2017042109
Service	Version 1.7.1_Release2017042109
U-boot	Version Nov 24 2014 - 09:24:52
Kernel	Version #187 PREEMPT Tue Jan 19 10:35:29 CST 2016
Product Type	1032(RJ11)
Select an Up	odate File C:\Users\Administrator\Browse
	18% 3353kb/s
	18% 3353kb/s
	18% 3353kb/s
The file is u	18% 3353kb/s Iploading. Please do not leave this page!
The file is u	
The file is u	
	ploading. Please do not leave this page!
	ploading. Please do not leave this page! Upgrade Information
	ploading. Please do not leave this page! Upgrade Information
	ploading. Please do not leave this page! Upgrade Information
	ploading. Please do not leave this page! Upgrade Information
	ploading. Please do not leave this page! Upgrade Information
	ploading. Please do not leave this page! Upgrade Information



Figure 3-98 File Uploading Interface

After a successful uploading of the file, the gateway will start to upgrade the system. See Figure 3-99 and you can learn the detailed upgrading information from the upgrade information box at the bottom.

[o			
	Current Version				
	Serial Num	00000057			
	WEB Version 1.7.1_Release2017042109				
	Service Version 1.7.1_Release2017042109				
	U-boot Version Nov 24 2014 - 09:24:52				
	Kernel Version #187 PREEMPT Tue Jan 19 10:35:29 CST 2016				
	Product Type	1032(RJ11)			
r					
	Select an Up	odate File C:\Users\Administrator\ Browse			
Update Reset					
		Upload completion!			
		10%			
System updating, please do not leave this page!					
		Upgrade Information			
	start upload	l upgrade file 🔺			
	decompres	sing the upgrade package			
		-			

Figure 3-99 System Upgrading Interface

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

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



3.9.5 Signaling Capture

Packet Capture	
Signaling Packet Capture SIP&Syslog RTP Packet Capture RTP Port Range 6000,10000	Start Stop
Data Recording	
Port Port 21 Recording Length 60 Recording of Connected IP Channels Recording before Echo Cancellation	Start Stop Download File
Start All Stop All	Download All

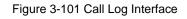
Figure 3-100 Signaling Capture Interface

See Figure 3-100 for the Signaling Capture interface, including two parts: Packet Capture and Data Recording. Packet capture contains Signaling Packet Capture and RTP Packet Capture. You can select either of them to start the capture according to your requirement. Click **Start** to start capturing packets. Click **Stop** to stop the capture and download the captured packets.

Data Recording will execute the recording task on the set port with the set recording time length. You can choose 'Recording of Connected IP Channels or 'Recording before Echo Cancellation'. Click **Start** to start recording data (consecutively recording 300 seconds at most) on the corresponding port with the corresponding time length. Click **Stop** to stop the recording and click **Download File** to download the recorded data.

3.9.6 Call Log

Call Log SIP Log Call Log Download	
Call from IP Channel	Clear All
04/24/2017 15:17:47:191 IP Channel 0,Incoming call from remote end "100" <sip:100@201.123.115.75>, call-id: 144c5668db359444@d3N3YW5raW5nLVBD Caller 100 Callee 8009</sip:100@201.123.115.75>	bind the chann
4 M	•
Call from Port Select a Port Port9	Clear All
04/24/2017 15:17:47:191 IP Channel 0,Incoming call from remote end "100" <sip:100@201.123.115.75>,call-id: 144c5668db359444@d3N3YW5raW5nLVBD Caller 100 Callee 8009 04/24/2017 15:17:47:198 Analog Channel 128 caller translation 100->100 match IP->TEL CallerID Manipulate rule(No Matched Rule)</sip:100@201.123.115.75>	bind the chann
04/24/2017 15:17:47:198 Analog Channel 128 callee translation 8009>8009 match IP>TEL CalleeID Manipulate rule(No Matched Rule)	
04/24/2017 15:17:47:204 Analog Channel 128 ringing,Caller 100,Callee 8009	
4	





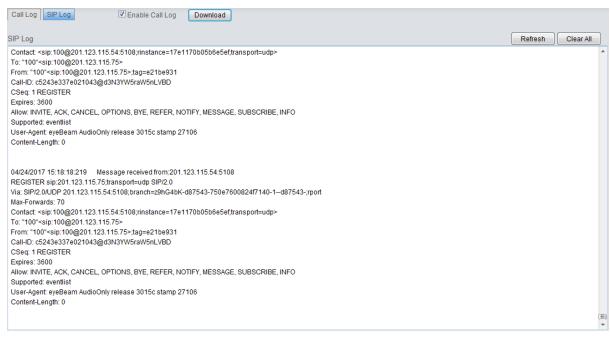
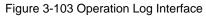


Figure 3-102 SIP Log Interface

See Figure 3-101, Figure 3-102 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
10 MaxWaitAutoDialAnswerTime:60 global_netoffescape:1 global_netoffescapeType:0 global_delaysend200ok:0 global_delaysend200oktime: global_fxodel; 2017-04-12 13:37:15 ClientIP:201.123.115.54 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-34 SilenceTim 2017-04-12 13:39:06 ClientIP:201.123.115.54 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-34 SilenceTim 2017-04-12 13:39:51 ClientIP:201.123.115.54 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-34 SilenceTim 2017-04-12 13:39:51 ClientIP:201.123.115.54 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-34 SilenceTim 2017-04-12 13:40:56 ClientIP:201.123.115.54 SaveFXOSet= global_FXODetectCallerIDTime:10 SilenceDetectFlag:0 SilenceEnergyThreshold:-34 SilenceTim 2017-04-12 13:43:31 ClientIP:201.123.115.54 SoftUpdate=filename:/usr/local/apache/htdocs/install.tar.gz 2017-04-14 14:42:03 ClientIP:201.123.115.54 SoftUpdate=filename:/usr/local/apache/htdocs/install.tar.gz 2017-04-14 14:42:03 ClientIP:201.123.115.166 SaveSipSet=LocalSipIp:201.123.115.75 LocalSipPort:5060 global_register:1 global_username:100 global_p: 2017-04-14 14:42:35.6 ClientIP:201.123.115.54 Factory Reset
2017-04-14 15:26:09 ClientIP:201.123.115.54 SaveNetWork=IPC1:0:201.123.115.75:255.255.255.0:201.123.115.254:0.0.0:0:IPC2:3 [CallConfig] Unit_ch0:6 SingleMaxTime_ch0:60 AllMaxTime_ch0:60 [Set used time] 0(30) 2017-04-14 15:31:54 ClientIP:201.123.115.54 Factory Reset
2017-04-14 15:33:04 ClientIP:201.123.115.54 SaveNetWork=IPC1:0:201.123.115.75:255.255.255.255.0:201.123.115.254:0.0.0:0:IPC2:3 2017-04-14 15:34:44 ClientIP:201.123.115.54 SoftUpdate=filename:/usr/local/apache/htdocs/install.tar.gz [CallConfig] Unit_ch0:6 SingleMaxTime_ch0:60 AllMaxTime_ch0:60 [Set used time]
2017-04-14 15:39:11 ClientIP:201.123.115.54 Factory Reset [CallConfig] Unit_ch0:6 SingleMaxTime_ch0:60 AllMaxTime_ch0:60 [Set used time]
2017-04-14 15:47:39 ClientlP:201.123.115.166 SaveSipSet=LocalSipIp:192.168.1.101 LocalSipPort:5060 global_register:1 global_username:1000 global_p: 2017-04-14 15:57:00 ClientlP:201.123.115.166 SaveSipSet=LocalSipIp:192.168.1.101 LocalSipPort:5060 global_register:1 global_username:1000 global_p: 2017-04-14 16:01:42 ClientlP:201.123.115.166 SaveSipSet=LocalSipIp:192.168.1.101 LocalSipPort:5060 global_register:1 global_username:1000 global_p: 2017-04-14 16:10:48 ClientlP:201.123.115.166 SaveSipSet=LocalSipIp:192.168.1.101 LocalSipPort:5060 global_register:1 global_username:1000 global_p: 2017-04-14 16:10:48 ClientlP:201.123.115.166 SaveSipSet=LocalSipIp:192.168.1.101 LocalSipPort:5060 global_register:1 global_username:1000 global_p: SetSyslog:1 SyslogIP:201.123.115.166 SyslogLevel:6 SetCDR:0 CDRIP:127.0.0.1 CDRPort:3 SetTime: Time: NTPZone: SetNTP:1 NTPIP:time.nist.gov NTPC 2017-04-14 16:44:19 ClientlP:201.123.115.166 SaveSipSet=LocalSipIp:192.168.1.101 LocalSipPort:5060 global_register:0 global_username: global_passw
Refresh Clear All Download





See Figure 3-103 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

	Data Backup	
To backup the configu	ration file, click the 'Backup' button to start.	Backup
	Data Upload	
To upload a configurati Config File	on file, select it and click the button 'Upload' to start. Browse	Upload

Figure 3-104 Backup & Upload Interface

See Figure 3-104 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 configuration file, click the 'Backup' button to start.	Backup
Data Upload	
To upload a configuration file, select it and click the button 'Upload' to start.	
Config File C:\Users\Administrator\Desktop\hasp\readme Browse	Upload
Message from webpage	
The gateway service will automatically restart after you upload the config file.Are you sure to upload?	
OK Cancel	

Figure 3-105 Backup & Upload & Prompt Interface

Click **OK** on the prompt box (Figure 3-105) to upload the configuration file to the gateway. Now the prompt information 'The gateway service is restarting, please do not leave this page' appears. See Figure 3-106. The gateway will overwrite the current configurations with the uploaded data after restart. Click **Cancel** to cancel this upload directly.



	Data Backup				
To backup the c	configuration file, click the 'Backup' button to start.		Backup		
	Data Upload				
To upload a cont	To upload a configuration file, select it and click the button 'Upload' to start.				
Config File	C:\Users\Administrator\Desktop\hasp\readme	Browse	Upload		
1	The gateway service is restarting. Please do not	leave this pag	je		

Figure 3-106	Configuration	File U	oloading	Interface

3.9.9 Factory Reset

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

Figure 3-107 Factory Reset Interface

See Figure 3-107 for the factory reset interface. Click *Reset* to restore all configurations on the gateway to factory settings.

3.9.10 System Monitor

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

Figure 3-108 System Monitor Configuration Interface

See Figure 3-108 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	je
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 Reset	Download MIB

Figure 3-109 Centralized Manage Setting Interface

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

Item	Description	
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 DCMS locates, It can be IP or a domain	
DCMS Server name valid only when DCMS is selected.		
Address Note: To configure the domain name, the DNS should be already configu		
	the corresponding domain name must be analyzable.	
Commonly Name	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. This item is valid only	
	when DCMS is selected.	



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

3.9.12 Access Control

Access Control List				
Check	Index	Command		Modify
	0	iptables -I INPUT -s 123.45.6.7 -j DROP		
Check All 🗮 Uncheck	Check All Inverse Delete Clear All Apply Add New			
1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 🔹 1 Pages Total				
Note: Please don't enable "SIP"=>"Calls from SIP Trunk Address only".				

Figure 3-110 Access Control List Interface

See Figure 3-110 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 3-111.



Access Control Command		
Index:	1	
Command:		
	Close	

Figure 3-111 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 3-110 to modify a command. See Figure 3-112 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:	ommand: iptables -I INPUT -s 123.45.6.7 -j DROP		
	Close		

Figure 3-112 Access Control Command Modification Interface

To delete an Access Control Command, check the checkbox before the corresponding index in Figure 3-110 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 3-110.

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 Call Test

	Call Test	
Test Type	IP Call out	
Local Alias		
Local SIP Account		
Remote Alias		
Remote SIP Account		
Called IP Address		
Called Port		
DTMF(RFC2833 Unsupporte	ed)	Send
Add or Modify Invite Header Field	Field Name	Field Content
Enable		
Start	Stop	Clear
Signaling Trace		

Figure 3-113 Call Test Interface

See Figure 3-113 for the Call Test interface. A call 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



Test Type	There are two types of call tests: PSTN Call out and IP Call out .	
Channel	The channel on which the call test will be performed.	
CalledID	The called party number of the call from the PSTN channel.	
Delay after Dial	The time from the dial behavior on the PSTN channel to the call's going out.	
Local Alias	The content of displayname in the from field of the invite message during the call out from the IP channel.	
Local SIP Account	The content of username in the from field of the invite message during the call out from the IP channel.	
Remote Alias	The content of displayname in the to field of the invite message during the call out from the IP channel.	
Remote SIP Account	The content of username in the to field of the invite message during the call out from the IP channel.	
Called IP Address	The called IP address of the call out from the IP channel.	
Called Port	The called port of the call out from the IP channel.	
DTMF	The DTMF digits sent by the IP channel after starting a call.	
Add or Modify Invite Header Field	The field name and content added or modified in the message header during the call out from the IP channel.	
Signaling Trace	Displays the call test process.	

After configuration, click *Start* to execute the call test; click *Stop* to terminate it immediately; click *Clear* to clear the records of call tests.



3.9.14 PING Test

Ping Test			
Source IP Address	LAN 1: 192.168.1.101		
Destination Address	127.0.0.1		
Ping Count (1-100)	4		
Package Length (56-1024 bytes)	56		
Start	End		
Info			
	.::		

Figure 3-114 Ping Test Interface

See Figure 3-114 for the Ping Test interface. A Ping test can be initiated from the gateway on a designated IP address to check the connection status between them. The table below explains the configuration items shown in the above figure.

Item	Description	
Source IP Address	Source IP address where the Ping test is initiated.	
Destination Address	Destination IP address 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.



3.9.15 TRACERT Test

Tracert Test	
Source IP Address	LAN 1: 192.168.1.101
Destination Address	127.0.0.1
Maximum Jumps (1-255)	30
Start	End
Info	
	:

Figure 3-115 Tracert Test Interface

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

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

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



3.9.16 Change Password

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

Figure 3-116 Password Changing Interface

See Figure 3-116 for the Password Changing interface where you can change your username and password of the gateway. Enter your current password, your new username and new password, and then confirm your new password. After configuration, click **Save** to apply your new username and password or click **Reset** to restore the configurations. After changing the username and password, you are required to log in again.

3.9.17 Restart

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

Figure 3-117 Service/System Restart Interface

See Figure 3-117 for the Restart interface. Click **Restart** on the service restart interface to restart the gateway service or click **Restart** on the system restart interface to restart the whole gateway system. A dump file will be generated each time you restart the service or the system. Click **download** and you can download it to help troubleshoot issues.



Appendix A Technical Specifications

Dimensions

440×44×267 mm³

Weight

About 4 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/FXO Port

Amount: 8/16/32

Type: RJ11, RJ21, RJ45

Maximum transmission distance: 1500m

Impedance

Input impedance:

 $\geq 1M\Omega/500V DC; \geq 10k\Omega/1000V AC$

Insulation resistance of telephone line from PC:

≥2*M*Ω/500V DC

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: 100~240V AC

Maximum power consumption: ≤50W

Signaling & Protocol

SIP signaling

Supported protocol: SIP V1.0/2.0, RFC3261

Audio Encoding & Decoding

G.711A	64 kbps
G.711U	64 kbps
G.729A/B	8 kbps
G723	5.3/6.3 kbps
G722	64 kbps
AMR	4.75/5.15/5.90/6.70/7.40/7.9 5/10.20/12.20 kbps
iLBC	13.3/15.2 kbps
SILK(16K)	20 kbps
OPUS(16K)	20 kbps
SILK(8K)	12 kbps
OPUS(8K)	12 kbps

Sampling Rate

8kHz



Appendix B Troubleshooting

Q1. What to do if I forget the IP address of the SMG 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 as follows:

LAN1: 192.168.1.101

LAN2 (disabled by default): 192.168.0.101

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 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. LAN1: **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. LAN1: **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 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 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 to configure the features Communication without Power and Communication without Network for the SMG analog gateway?

The feature **Communication without Power** is implemented with the help of composite modules equipped in the gateway. Once the power to the device is cut off, the station which is linked with the FXS port on the composite module and the trunk which is linked with the FXO port on the same module will connect to each other directly and keep the good communications between phones and networks. What you need to do is just to configure the composite module properly at your purchase of our gateway.

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

Refer to <u>Q2</u> in this chapter for detailed information.

Q7. 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.

Q8. Is there any cell-phone APP can make calls to the SMG 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,

Q9. Does the SMG gateway support fax?

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



Q10. Which RTP codecs are supported by the SMG gateway?

At present, the supported RTP codecs are: G.711A, G.711u, G.729, G.723, G.722, AMR, iLBC, SILK(16K), OPUS(16K), SILK(8K) and OPUS(8K).



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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