

Synway SMG Series Wireless Gateway

SMG4004

SMG4008

SMG4016

SMG4032

Wireless Gateway

User Manual

Version 1.4.0

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



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Chapter 1 Product Introduction

Thank you for choosing Synway SMG Series Wireless Gateway!

The Synway SMG series wireless gateway products (hereinafter referred to as 'wireless gateway'), as a part of the Synway gateway products, works mainly for connecting the wireless network with the VoIP network. It adopts an updated VoIP processor and the wireless module, uses the push-pull SIM card socket for easy replacement of the SIM card, quite advanced in technology. So far, only SMG4008 is available.

Module	Amount of GSM Port	Amount of WCDMA Port	Amount of CDMA Port	Supported Frequency band
4016-16G	16			
4008-8G	8			GSM: 850/900/1800/1900MHz
4004-4G	4			
4008-8W		8		GSM: 900/1800MHz
4004-4W		4		UMTS: 900/2100MHz
4008-8C			8	CDMA:
4004-4C			4	CDMA 2000 800MHz

See below table for the modules of SMG series wireless gateway:

Table 1-1 Model List



1.1 Typical Application



Figure 1-1 Typical Application

1.2 Feature List

Basic Features	Description
TDM Call	Call initiated from TDM to IP, via routing and number manipulation to obtain the called IP address.
IP Call	Call initiated from IP to TDM, via routing and number manipulation to obtain the call destination.
Number Manipulation	Peels off some digits of a phone number from left/right, or adds a prefix/suffix to a phone number.
Call Forward	Three options available: Unconditional, Busy, No Reply and Unreachable.
CID	Displays the CallerID.
Echo Cancellation	Provides the echo cancellation feature for a call conversation over the wireless port.



TDM/VoIP Routing	Sets a routing path: from IP to TDM or from TDM to IP.			
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.			
Custom IVR Recording	Provides the interface to customize the IVR Recording.			
White/Black List	Allows the setting of the white/black list for WEB access.			
Voice Gain Adjust	Supports the gain adjustment for the received or sent voice.			
Receive or Send SMS/USSD	Supports the SMS sending and receiving, as well as the USSD request and response.			
Auto Select Network	Supports the auto identification and selection of the network operator.			
SMS CODEC	Two options available: ASCII and UCS2.			
Signaling & Protocol	Description			
SIP Signaling	Supported protocol: SIP V1.0/2.0, RFC3261.			
Voice	CODEC G.711A, G.711U, G.729A/B, G.723, G.722, AMR, iLBC DTMF Mode RFC2833, SIP INFO, INBAND			
Network	Description			
Network Protocol	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN.			
Network Protocol Static IP	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support.			
Network Protocol Static IP DHCP	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support. IP address dynamic allocation support.			
Network Protocol Static IP DHCP DNS	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support. IP address dynamic allocation support. Domain Name Service support.			
Network Protocol Static IP DHCP DNS Security	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support. IP address dynamic allocation support. Domain Name Service support. Description			
Network Protocol Static IP DHCP DNS Security Admin Authentication	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support. IP address dynamic allocation support. Domain Name Service support. Description Supports admin authentication to guarantee the resource and data security.			
Network ProtocolStatic IPDHCPDNSSecurityAdmin AuthenticationSystem Monitor	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support. IP address dynamic allocation support. Domain Name Service support. Description Supports admin authentication to guarantee the resource and data security. Monitors the running status of the system and the server.			
Network ProtocolStatic IPDHCPDNSSecurityAdmin AuthenticationSystem MonitorMaintain & Upgrade	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support. IP address dynamic allocation support. Domain Name Service support. Description Supports admin authentication to guarantee the resource and data security. Monitors the running status of the system and the server. Description			
Network ProtocolStatic IPDHCPDNSSecurityAdmin AuthenticationSystem MonitorMaintain & UpgradeWEB Configuration	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support. IP address dynamic allocation support. Domain Name Service support. Supports admin authentication to guarantee the resource and data security. Monitors the running status of the system and the server. Support of configurations through the WEB user interface.			
Network ProtocolStatic IPDHCPDNSSecurityAdmin AuthenticationSystem MonitorMaintain & UpgradeWEB ConfigurationLanguage	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support. IP address dynamic allocation support. Domain Name Service support. Description Supports admin authentication to guarantee the resource and data security. Monitors the running status of the system and the server. Description Support of configurations through the WEB user interface. Chinese, English.			
Network ProtocolStatic IPDHCPDNSSecurityAdmin AuthenticationSystem MonitorMaintain & UpgradeWEB ConfigurationLanguageSoftware Upgrade	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support. IP address dynamic allocation support. Domain Name Service support. Description Supports admin authentication to guarantee the resource and data security. Monitors the running status of the system and the server. Description Support of configurations through the WEB user interface. Chinese, English. Support of user interface, gateway service, kernel and firmware upgrades based on WEB.			
Network ProtocolStatic IPDHCPDNSSecurityAdmin AuthenticationSystem MonitorMaintain & UpgradeWEB ConfigurationLanguageSoftware UpgradeTracking Test	Supported protocol: TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN. IP address modification support. IP address dynamic allocation support. Domain Name Service support. Description Supports admin authentication to guarantee the resource and data security. Monitors the running status of the system and the server. Description Support of configurations through the WEB user interface. Chinese, English. Support of user interface, gateway service, kernel and firmware upgrades based on WEB. Support of Ping and Tracert tests based on WEB.			

1.3 Hardware Description

The wireless gateway supports two LANs and adopts an external 12V power supply. See below





Figure 1-4 SMG4016 Front View



Figure 1-5 SMG4016 Rear View

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

Interface	Description
	Amount: 2
	Type: RJ-45
	Bandwidth: 10/100 Mbps
LAN	Self-Adaptive Bandwidth Supported
	Auto MDI/MDIX Supported
	Built-in Link indicator and ACTIVE indicator. For more details, refer to 1.4 Indicator Info
	Amount: 4, 8, 16*4
SIM Card Slot	Network Supported: GSM, WCDMA, CDMA
	Amount: 1
	Type: RS-232
	Baud Rate: 115200bps
	Connector: RJ45 to DB-9 Connector (4004, 4008 series), Mini-USB connecting line (4016
Console Port	series)
	Data Bits: 8 bits
	Stop Bit: 1 bit
	Parity Unsupported
	Flow Control Unsupported
External Power	Provide the 12V voltage with positive inside and negative outside, and the current is larger
Supply Interface	than 3A
Button	Description
Reset Button	Restore the gateway to factory settings by pressing this button persistently for 3 seconds
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 Indicator Info</u> .
Alarm Indicator	Alarms the device malfunction. For more details, refer to <u>1.4 Indicator Info.</u>
Link Indicator	The green LED on the right of LAN, indicating the network connection status.
ACT Indicator	The orange LED on the left of LAN, whose flashing tells the data are being transmitted.
Port Indicator	1. When the port is idle, the LED Lights up in green and keeps on;



2.	When the port is unavailable, the LED Lights up in red and keeps on;
3.	When the port is in use, the LED flashes in green
4.	When the port module is disabled, the LED flashes in red
5.	For SMG4016 series, only the indicator of the card slot in which the SIM card is in using
	lights up and other indicators will go out in the case that there are more than one SIM
	cards inserted in the same channel.

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

1.4 Indicator Info

The wireless 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	State Description		
	Go out	System is not yet started.		
Run Indicator	Light up and flash fast	System is starting.		
	Flash slowly	Device is normal.		
	Go out	Device is normal.		
	1.1.1.	Upon startup: Device is normal.		
Alarm Indicator	Light up	In runtime: Device is abnormal.		
	Flash	Device 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 D 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 wireless gateway in the shortest time.

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

- Wireless Gateway *1
- External 12V Power Adapter *1
- GSM/WCDMA/CDMA Rubber Antenna *4/8/16
- Standard RJ45 to DB-9 Switcher (4004/4008 series) *1, Mini-USB connecting line (4016 series) *1
- 8mm Antenna Wrench *1
- Rubber Foot Pad *4
- Network Cable *1
- Warranty Card *1
- Installation Manual *1

Step 2: Connect the network cable.

This product provides RJ-45 interfaces.

Step 3: Insert the SIM card (standard size) and install the antenna.

The wireless gateway provides a SIM card slot. You are required to insert the SIM card before using it. Take out the rubber antennae from the packing box, install them onto the wireless gateway, screw them up and evenly arrange them.

Step 4: Power on and start the gateway.

To use the wireless gateway, you need an external power supply. Insert it to the power interface of the wireless gateway and power it on with 100~240V AC. See the figure below:







Step 5: Log in the gateway.

Enter the original IP address (192.168.1.101) of the wireless gateway in the browser to go to the WEB interface of the gateway. The original username and password of the gateway are both 'admin'. For detailed instructions about login, refer to 3.1 System Login. We suggest you change the initial username and password via 'System Tools \rightarrow Change Password' on the WEB interface as soon as possible after your first login. For detailed instructions about changing the password, refer to 3.10.5 Change Password. After changing the password, you are required to log in again.

Step 6: Modify IP address of the gateway.

You can modify the IP address of the gateway via 'Advanced Settings \rightarrow Network' on the WEB interface to put it within your company's LAN. Refer to <u>3.5.1 Network</u> for detailed instructions about IP modification. After changing the IP address, you shall log in the gateway again using your new IP address.

Step 7: 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 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>3.5.4 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 the corresponding ports to it. Refer to <u>3.7.2 Port Group</u> for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.

Example: Provided the added port 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>3.8.3 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. Use an external phone to call the number of this SIM card, and then follow the cue tone to 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: The external phone dials the number of this SIM card, and then follows the cue tone to dial 123. Then the IP phone with the IP address 192.168.0.111 and the port 5060 will ring.

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

 Go to 'Port Settings → Port Group' on the WEB interface and click the 'Add New' button to create a new port group and add the corresponding ports which are connected with stations to it. Refer to <u>3.7.2 Port Group</u> for detailed instructions. You may use the default values of other configuration items and are required not to leave 'Description' empty.



Example: Provided the added port 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/IP' on the WEB interface and click the 'Add New' button to add a new routing rule. Refer to <u>3.8.2 IP→Tel</u>/IP 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 wireless gateway to make outgoing calls from the wireless channel.

Example: Provided the IP address of the wireless gateway is 192.168.0.101, the port is 5060, use the IP phone to call the IP address 13529101232@192.168.0.101 and then the first idle wireless port in the port group of step 2 will make an outgoing call to 13529101232.

Special Instructions:

- As the device will gradually heat up while being used, please maintain good ventilation to prevent sudden failure, ensuring that the ventilation holes are never jammed.
- During runtime, if the alarm indicator lights up or flashes, it indicates that the device goes abnormal. If you cannot figure out and solve the problem by yourself, please contact our technicians for help. Otherwise it may lead to a drop in performance or unexpected errors.

Chapter 3 WEB Configuration

3.1 System Login

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

Windows Security	X
The server 192. Warning: This s sent in an insec connection).	168. 1. 101 at SMG requires a username and password. server is requesting that your username and password be sure manner (basic authentication without a secure
	User name Password Remember my credentials
	OK Cancel

Figure 3-1 Login Interface

The gateway only serves one user, whose original username and password are both 'admin'. You can change the username and the password via 'System Tools \rightarrow Change Password' on the WEB interface. For detailed instructions, refer to <u>3.10.5 Change Password</u>.

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

				System	Info	
System Info						
Port State		LAN				
Call Count			MAC Address	00-00-E0-A7-01-00	000 000 000 0	400 400 4
SIP Message Cour	t		IP Address	192.168.1.101	200.200.200.0	192.100.1
en meesage een			DNS Server	0.0.0.0	Email 0	Dennia
Quick Config	*		Receive Packets	All:32441	Error 0	Drop:0
			Current Sneed	Receive:3.0 KB/e	Transmit 1.6 KB/s	Drop.0
VolP	*		Work Mode	100Mb/s Full Duplex	Transmit, 1.0 Kb/s	
Advanced	×		Trontine de	reentere r un papren		
S. Huvanceu	•	Runt	ame	32m 25s		
Wireless	*					
Dert.	~	Curr	ent Version			
Port	•		WEB	1.4.0_2016061312		
Route	*		Gateway	1.4.0_2016061312		
	0		Serial Num	00001560		
Num Manipulate	*		Authorization Code	0x4001		
System Tools	*		FPGA	6.05		
v -			U-boot	Aug 06 2015-15:33:00		
			Kernel	#224 Tue Dec 8 17:17:2	8 CST 2015	
			Device Type	4008-8G		

Figure 3-2 Main Interface



3.2 Operation Info

Operation Info includes four parts: *System Info*, *Port 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

00-00-E0-A7-01-00		
192.168.1.101	255.255.255.0	192.168.1.1
0.0.0.0		
All:32441	Error:0	Drop:0
All:7399	Error:0	Drop:0
Receive:3.9 KB/s	Transmit:1.6 KB/s	
100Mb/s Full Duplex		
32m 25s		
1.4.0_2016061312		
1.4.0_2016061312		
00001560		
0x4001		
6.05		
Aug 06 2015-15:33:00		
#224 Tue Dec 8 17:17:2	28 CST 2015	
4008-8G		
	00-00-E0-A7-01-00 192.168.1.101 0.0.0 All:32441 All:7399 Receive:3.9 KB/s 100Mb/s Full Duplex 32m 25s 1.4.0_2016061312 1.4.0_2016061312 00001560 0x4001 6.05 Aug 06 2015-15:33:00 #224 Tue Dec 8 17:17:2 4008-8G	00-00-E0-A/-01-00 192.168.1.101 255.255.255.0 0.0.0 All:32441 Error:0 All:7399 Error:0 Receive:3.9 KB/s Transmit:1.6 KB/s 100Mb/s Full Duplex 32m 25s 1.4.0_2016061312 1.4.0_2016061312 0.0001560 0x4001 6.05 Aug 06 2015-15:33:00 #224 Tue Dec 8 17:17:28 CST 2015 4008-8G

Figure 3-4 System Info Interface

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

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



Dessive Deskate	The amount of receive packets after the gateway's startup, including three options:					
Receive Packets	All, Error and Drop.					
	The amount of transmit packets after the gateway's startup, including three options:					
Transmit Packets	All, Error and Drop.					
Current Speed	Show the current speed of data receiving and transmitting.					
	Show the work mode of the network, including four modes: 10 Mbps Half Duplex, 10					
work Mode	Mbps Full Duplex, 100 Mbps Half Duplex, 100 Mbps Full Duplex.					
	Time of the gateway keeping running normally after startup, which will be					
Runtime	automatically updated.					
WEB	Current version of the WEB interface.					
Gateway	Current version of the gateway service.					
Serial Num	Unique serial number of a wireless gateway.					
Authorization Code	The authorization codes vary from different wireless modules.					
FPGA	Current version of FPGA.					
U-boot	Current version of Uboot.					
Kernel	Current version of the system kernel on the gateway.					
Device Type	Type of the wireless gateway.					

3.2.2 Port State

	Port State											
Port	Туре	State	Voice Type	Direction	CallerID	CalleeID	SIM Card Used	Cell Phone No.	Connection	Signal	SIP Reg Status	
1	GSM	💽 Idle	3.7773				📟 📟 📟	13023634112	Connect	atl	Unregistered	
2	GSM	🖸 Idle	(<u>)</u>					13023634 <mark>1</mark> 83	Connect	atl	Unregistered	
3	GSM	🚮 Unusable							Disconnect	all	Unregistered	
4	GSM	🚮 Unusable	1.0000						Disconnect	aff	Unregistered	
5	GSM	🚮 Unusable	85770			-		1000	Disconnect	all	Unregistered	
6	GSM	M Unusable							Disconnect	all	Unregistered	
7	GSM	🚮 Unusable							Disconnect	atl	Unregistered	
8	GSM	🚮 Unusable	0.0000	();					Disconnect	all	Unregistered	
9	GSM	🚮 Unusable	85770						Disconnect	all	Unregistered	
10	GSM	🚮 Unusable							Disconnect	all	Unregistered	
11	GSM	🚮 Unusable							Disconnect	atl	Unregistered	
12	GSM	🚮 Unusable	1.000	();					Disconnect	all	Unregistered	
13	GSM	🚮 Unusable	87770						Disconnect	all	Unregistered	
14	GSM	🚮 Unusable	10000			-			Disconnect	all	Unregistered	
15	GSM	🚮 Unusable							Disconnect	atl	Unregistered	
16	GSM	🚮 Unusable	(.)	();					Disconnect	atl	Unregistered	

Figure 3-5 Channel State Interface

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

ltem	Description								
Port	Port number on the device.								
Туре	Port type on the device. So far, only GSM, WCDMA and CDMA types are supported.								
State	Displays the port state in real time. You can move the mouse onto the port state icon for detailed state information.								



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	State	lcon	Description				
	Idle		The port is available.				
	Off-hook	2	The port picks up the call.				
	Wait Answer		The port receives the ringback tone and is waiting for the called party to pick up the phone.				
	Ringing		The port is in the ringing state.				
	Talking		The port is in a conversation.				
	Dialing	C>	The port is dialing.				
	Pending	2	The port is in the pending state.				
	Internal State		Internal state of the port.				
Unusable 🕢 The port is unavailable.							
Voice Type	Displays the voice type of the current call.						
Direction	Displays the direction of the call on port.						
CallerID	Displays the Cal	lerID of	the call on port.				
CalleeID	Displays the Cal	leeID of	the call on port.				
SIM Card	ard Displays the real-time state of the SIM card. Move the mouse onto the corresponding icon and you can find the exact state of the SIM card.						
	Note: This item is	s unavai	ilable for SMG4004 and SMG4008 series.				
Cell Phone No	Displays the number of the SIM card inserted in current port. For SMG4016 series,						
	the number is the	at of the	SIM card which is in using.				
Connection	Displays the con	nection	status between the SIM card and the base station.				
Signal	Displays the sigr	nal inten	sity of the wireless module.				
SIP Reg Status	Displays the regi	istration	status of the port.				

3.2.3 Call Count

	Call Count										
Call Direction	Total Calls	Successful Calls	Busy	No Answer	Routing Failure	Dialing Failure	Unknown				
IP->Tel	2	2	0	0	0	0	0				
Tel->IP	1	0	0	0	0	1	0				
	di secolo di		c.t. l	, , , , , , , , , , , , , , , , , , ,		A					
	Refresh										

Figure 3-6 Call Count Interface

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

Item	Description
Call Direction	A condition for call count, two options available: $IP \rightarrow Tel$ and $Tel \rightarrow IP$.
Total Calls	Total number of calls in a specified call direction.
Successful Calls	Total number of successful calls in conversation.
	Total number of calls which fail as the called party has been occupied and replies a
Busy	busy message.



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

3.2.4 SIP Message Count

				Reques	t.				
Request	RE	GISTER	INVITE	ACK	INFO	BYE	CANCEL	NOTIFY	OPTION
Send		0	1	1	0	1	0	0	0
Send Repeatedly		0	0	0	0	0	0	0	0
Receive		0	1	1	0	1	0	0	0
Receive Repeatedly	-	0	0	0	0	0	0	0	0
	100 Toing	180 Ringing	183	Common Res Session Prose	ponse	200 OK	486 Busy	487 Request Alread	
	100 Toing	190 Dinging	102	Common Res	ponse	200 OK	496 Puer	197 Pequest Alread	
Common Response	100 Hying	reertinging						tor requeet hous,	y Terminated
Common Response	1	1		0		2	0	0	y Terminated
Common Response	100 Hying	locitunging					-	tor requeet mode.	y Terminated

Figure 3-7 SIP Message Count Interface

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

3.3 Quick Config



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 Port. The gateway can work normally after configuration.

See Figure 3-9 for the Quick Config-Network Settings interface. Refer to 3.5.1 Network for detailed settings. After configuration, click *Next* to enter the SIP Settings interface.



Network Type:	Static 👻
IP Address (I)	192.168.1.101
Subnet Mask (U)	255.255.255.0
Default Gateway (D)	192.168.1.1
DNS Server (P)	0.0.0.0
Speed and Duplex Mode	Automatic Detection

Figure 3-9 Quick Config-Network Settings Interface

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

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

Figure 3-10 Quick Config-SIP Settings Interface

See Figure 3-11 for the Port Settings interface. The configuration items on this interface are the same as those on the Port interface. Refer to <u>3.7.1 Port</u> for detailed settings. After configuration, click **Back** to go back to the SIP Settings interface; click **Next** to enter the Quick Config-Completion interface.

					1. mix seminar								
Port Typ	e SIP Account	Authentication Username	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	Echo Canceller	Color Ring	Color Ring Index	Server Index	Modify
1 GS	M 8001		Static Binding	180	Disable	Disable	Failed	Enable	Enable	Disable			0
2 GS	M 182		Static Binding	8003	Disable	Disable	Unregistered	Enable	Enable	Disable			12
3 GS	M 8003	(Two Stage Dialing Mode		Disable	Disable	Unregistered	Enable	Enable	Disable	80°	-	12
4 GS	M 8004		Two Stage Dialing Mode	228	Disable	Disable	Unregistered	Enable	Enable	Disable	222	12	Q.
5 GS	M 8005		Two Stage Dialing Mode		Disable	Disable	Unregistered	Enable	Enable	Disable	÷.		2
6 GS	M 8005		Two Stage Dialing Mode		Disable	Disable	Unregistered	Enable	Enable	Disable			12
7 GS	44 8007		Two Stage Dialing Mode	-	Disable	Disable	Unregistered	Enable	Enable	Disable			Q
8 GS	M 8008	())	Two Stage Dialing Mode		Disable	Disable	Unregistered	Enable	Enable	Disable	-		0
				-	Dack	Heat							



Figure 3-11 Port Settings Interface

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

Figure 3-12 Quick Config-Completion Interface

Click **Back** to go back to the Port 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 3-13. *SIP Settings* is used to configure the general SIP parameters, *SIP Compatibility* is used to set which SIP servers and SIP messages will the gateway be compatible with, *SIP Station* is to set the basic information of the SIP station, *SIP Server* is to set the basic information of the SIP server, *NAT Setting* is used to configure the parameters for NAT, and *Media Settings* is to set the RTP port and the payload type.

🚉 VolP	*
SIP	
SIP Compatibility	
SIP Station	
SIP Server	
NAT Setting	
Media	

Figure 3-13 VoIP Settings



3.4.1 SIP

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

Figure 3-14 SIP Settings Interface

See Figure 3-14 for the SIP settings interface where you can configure the general SIP parameters. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to <u>3.10.8 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-14.

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



SIP Account	When the gateway initiates a call to SIP, this item corresponds to the username of			
Sir Account	SIP.			
Password	Registration password of the gateway. To register the gateway to SIP, both			
Passworu	configuration items SIP Account and Password should be filled in.			
Authentication				
Username				
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.			
Devietary Velidity	Validity period of the SIP registry. Once the registry is overdue, the gateway should			
Registry validity	be registered again. This configuration item is valid only when <i>Register Gateway</i> is			
Period	set to Yes. Range of value: 10~3600, calculated by s, with the default value of 600.			
Multi-Registrar	Tick the checkbox before to enable the multi-registrar server mode. By default, it is			
Server Mode	disabled.			
SIP Transport	There are two modes UDP and TCP available for running the SIP protocol. The			
Protocol	default value is <i>UDP</i> .			
	Once this feature is enabled, the gateway will send signaling messages to the			
	corresponding externally bound address and port when it registers to the server. By			
IMS Network	default, this feature is <i>disabled</i> . Only when this feature is <i>enabled</i> will these items			
	Externally Bound Address, Externally Bound Port and Authentication			
	<i>Username</i> be shown.			
Externally Bound	Externally bound ID address for registration			
Address				
Externally Bound	Externally bound port for registration			
Port				

3.4.2 SIP Compatibility

See Figure 3-15 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.



ompatibility	SIP
"Request" Field	Obtain CalleeID from
Username of From Field 🛩	Set CallerID position
Username of From Field	Obtain CallerID from
Enable	Use Contact Address
Call Enable	Two Stage Dialing for SIP Incomin
60	Maximum Wait Answer Time (s)
	SIP Station Supported
Gateway	Set SIP Identifying
0	Call Hangup when RTP Timeout(s
Enable	Ignore ACK
✓Enable	Abnormal Call Hangup Detection Cycle(s)
Enable	Server Status Detection
0	Cycle(s)
Immediately	Occasion to Reply 183
After pickup	Occasion to Reply 200 Ok
© Enable 0 Immediately After pickup	Server Status Detection Cycle(s) Occasion to Reply 183 Occasion to Reply 200 Ok

Figure 3-15 SIP Compatibility Setting Interface

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

ltem	Description
Obtain CalleelD	There are two optional ways to obtain the called party number: from "To" Field and
from	from "Request" Field. The default value is "Request" Field.
	There are two options to set the position of the calling party number: "Displayname
Set CallerID Position	of From Field" and "Username of From Field". The default value is "Username of
	From Field".
	There are two optional ways to obtain the calling party number: from "Displayname
Obtain CallerID from	of From Field" and from "Username of From Field". The default value is "Username
	of From Field".
Use Contact	Sets whether to send the request message according to the content of Contact, with
Address	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; if the Contact field indicates an IP address belonging to the WAN, the
	request message will be sent according to this IP address.
Two Stage Dialing	Once this facture is eachlad, the increasing call form OID should perform the two
for SIP Incoming	Once this feature is enabled, the incoming call from SIP should perform the two
Call	stage dialing operation. By default this feature is disabled.
	Sets the maximum time for the SIP channel to wait for the answer from the called
Maximum Wait	party of the outgoing call it initiates. If the call is not answered within the specified
Answer Time	time period, it will be canceled by the channel automatically. The default value is 60,
	calculated by s.
SIP Station	Once this feature is enabled, a SIP terminal can be registered to the gateway to
Supported	become a SIP station. By default this feature is disabled.
	Sets the SIP identifying content in the SIP call message. The default setting is
Set SIP Identifying	Gateway.
	Sets the maximum time for the SIP channel to wait for the RTP packet. If no RTP
Maximum Wait RTP	packet is received within the specified time period, the channel will enter the
Time	pending state automatically and release the call. The default value is 0 (disabled),
	calculated by s.
	Once this feature is enabled, it is not necessary for the gateway to wait for the ACK
Ignore ACK	message after sending the 2000K message to establish a call. By default it is
	disabled.
	Sets the interval between checks of the remote end's abnormal hangup, with the
Abnormal Call	default value of 0 (feature disabled), calculated by s. It is suggested to set to 10s if
Hangup Detection	this feature is necessary to be used.
_	The interval of sending a heartbeat packet to detect the master registrar server
Server Status	status, with the default value of 0 (feature disabled), calculated by s. It is suggested
Detection	to set to 15s if this feature is necessary to be used.
Occasion to Reply	Sets the occasion to reply the 183 message. Two options including: Immediately
183	and After ringing, with the default value of <i>Immediately</i>
Occasion to Renky	Sets the occasion to reply 200 OK. Two options including: After pickup and After
200 Ok	ringing with the default value of After nickun

3.4.3 SIP Station

A SIP terminal can be registered to the gateway to become a SIP station. Tick the option 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 3-16 below.



Quick Config 👻 VoIP 🔗 No available SIP Static	Operation Info	*	
VolP	🕂 Quick Config	*	
	😤 VolP	*	No available SIP Station!
SIP Add New	SIP		Add New
Sip Compatibility	Sip Compatibility		
SIP Station	SIP Station		
NAT Setting	NAT Setting		
Media	Media		

Figure 3-16 SIP Station Setting Interface

Click *Add New* to add SIP stations manually. See Figure 3-17. You can configure basic SIP station information on this interface. The bound port to a SIP station must be a wireless 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:	1					
Description:	default					
Batch Setting:	Enable					
Save	Close					

Figure 3-17 Add New SIP Station

The table below explains the items shown above:

ltem	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 wireless 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 3-18 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	-	1	Unregistered	-	-	default	
Check All E Uncheck All I Inverse I Delete I Clear All Add New									
Item Total	20 Items/Page	e 1/1 First Previ	ous Next Last (Go to Page 1 🗸	Pages Total				

Figure 3-18 SIP Station Interface

Click *Modify* in the above figure to modify the configuration of the SIP station. See Figure 3-19. The configuration items on this interface are the same as those on the *Add New SIP Station* interface.

٤	SIP Station
Number:	0
Username:	120
Password:	•••
Bound Port:	1
Description:	default
Batch Setting:	Enable
Save	Close

Figure 3-19 SIP Station Modification Interface

To delete a SIP station, check the checkbox before the corresponding index in Figure 3-18 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 3-18.

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 3-20 below.

Operation Info	*	
🕂 Quick Config	*	
VolP	8	No Available Registrar Server!
SIP		Add New
Sip Compatibility		
SIP Server		
NAT Setting		
Media		



Figure 3-20 SIP Server Interface

Click *Add New* to add SIP servers manually. See Figure 3-21. You can configure basic SIP server information on this interface.

Index	1 💌
Description	default
Registrar IP Address	
Registrar Port	5060
Registry Validity Period (s)	600
IMS Network	Enable
Externally Bound Address	2
Externally Bound Port	5060

Figure 3-21 Add New SIP Server

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

ltem	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 3-22 for the SIP server management interface.

							or server managemen	IL.				
Check	Index	Description	IP Addre	SS	Port	IMS Network	Externally Bound Address	Externally Bound Port	Registry Validity Period	Port	Port Group	Modify
	1	default	201.123.11	5.233	5060	Disable		-	600	-		
Check A		Uncheck All	Inverse		Delete	Clear A	н					Add New
1 Item Tota	al 20 Ite	ms/Page 1/1	First Previous	s Next	Last Go	to Page 1 V	1 Pages Total					

Figure 3-22 SIP Server Management

Click *Modify* in the above figure to modify the configuration of the SIP server. See Figure 3-23.

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



Index	1
Description	default
Registrar IP Address	201.123.115.233
Registrar Port	5060
Registry Validity Period (s)	600
IMS Network	Enable

Figure 3-23 SIP Server Modification Interface

To delete a SIP server, check the checkbox before the corresponding index in Figure 3-22 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 3-22.

3.4.5 NAT Setting

See Figure 3-24 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 Se	ettings
STUN Server	Enable
NAT Type	Unknown
STUN Server Address	127.0.0.1
Mapping Address	
RTP Self-adaption	Enable
Rport	Enable
Auto Detect NAT IP	Enable
Note: Auto Detect NAT IP:This feature only work router.	ks cooperatively with the port mapping setting on
Save	Reset

Figure 3-24 NAT Setting Interface

The table below explains the items shown in Figure 3-24.



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



3.4.6 Media

		Media Pa	rameters	
	DTMF Transmit	Mode	RFC2833	×
	RFC2833 Paylo	ad	101	
	RTP Port Rang	е	50000,5076	7
	Silence Suppre	ssion	Disable	
	JitterBuffer		20	
	Voice Gain Out	put from IP (dB)	0	
CODEC P	AGC Target Energy T Maximum Gain Maximum Atten Minimum Input riority	Threshold (dB) Threshold (dB) uation Threshold (dB) Energy (dB)	✓Enable 0 48 0 -60	
Check V V V V V V V V V	Priority 1 2 3 4 5 6 7	CODEC G711A V G711U V G729 V G723 V G722 V AMR V iLBC V	Packing Time 20 20 20 30 20 30 20 30 20 30 20 30 20 30 20 30 20 30 20 30 30 30 30 30 30 30 3	Bit Rate (kbs) 64 64 8 6.3 64 4.75 13.3
		Save	Reset	

Figure 3-25 Media Settings Interface

See Figure 3-25 for the media settings interface where you can configure the RTP port and payload type depending on your requirements. After configuration, click **Save** to save your settings into the gateway or click **Reset** to restore the configurations. If a dialog box pops up after you save your settings asking you to restart the system, do it immediately to apply the changes. Refer to <u>3.10.8 Restart</u> for detailed instructions. The table below explains the items shown in Figure 3-25.

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



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



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

3.5 Advanced Settings

Advanced Settings includes eleven parts: *Network, System Param, Service Config, Dialing Rule, Function Key, Cue Tone, Color Ring, QoS, Tone Generator, CDR Query* and *VPN.* See Figure 3-26. *Network* is used to configure the general properties of the network port; *System Param* is used to configure some properties of the system; *Service Config* is used to configure some properties which corresponds to the service; *Dialing Rule* is used to set the judging conditions for dialing; *Function Key* is used to set a cluster of combination keys for you to query or set the network port; *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 wireless port; *QoS* uses the differentiated services technology to increase the gateway's service quality; *Tone Generator* is used to configure some properties of tones sent from gateway. *CDR Query* is used to inquire the detailed call record; VPN makes use of the tunnel technology to transport the data, and the methods of user authentication and data encryption to prevent the data being read and distorted when they are transported on the public network.





Figure 3-26 Advanced Settings

3.5.1 Network

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 💌
	IP Address (I) Subnet Mask (U) Default Gateway (D) DNS Server (P) Speed and Duplex Mode

Figure 3-27 Network Settings Interface

See Figure 3-27 for the network settings interface. A gateway has two LANs which can be configured with the same network type, IP address, subnet mask, default gateway and DNS server to realize the feature of hot backup. There are three options in type: Static, DHCP and PPPoE.

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.



3.5.2 System Param

NEB Mana	gement	
	WEB Port	80
	Access Setting	Allow All IPs
SYSLOG P	arameters	
	SYSLOG Enabled	⊙Yes ONo
	Server Address	201.123.115.36
	SYSLOG Level	INFO
	AT Debug Enabled	⊙Yes ONo
	Echo Mode Enabled	Oyes ONo
	Port	port 1
CDR Param	eters	
	CDR Enabled	⊙Yes ONo
	Server Address	127.0.0.1
	Server Port	3
	Save CDR	⊙Yes ONo
	Amount of Saved CDR	5000
API Parame	ters	
	API Enabled	⊙Yes ONo
	Remote IP Address Allowed to Invoke API	
		(Separated by ', ** denotes all IP addresses)
	Username for API Call	ApiUserAdmin
	Password for API Call	
Time Param	eters	
	Time Calibration	ONTP OSynchronized with Operator OClose
	NTP Server Address	127.0.0.1
	Synchronizing Cycle	3600
	System Time	Modify 2016-03-16 09:44:38
	Time Zone	GMT+8:00 (Beijing, Singapore, Taipei, Kuala Lui 🗸
	Daily Restart	EYes ONo
	Restart Time	0 💌 h 0 💌 m

Figure 3-28 System Parameters Setting Interface

See Figure 3-28 for the System Parameters Setting interface. The table below explains the items shown in the above figure.

Item	Description
WEB Port	The port which is used to access the gateway via WEB. The default value is 80.



Access Setting	Sets the IP addresses which can access the gateway via WEB. By default, all IPs
	are allowed. You can set an IP whitelist to allow all IPs within it to access the
	gateway freely. Also you can set an IP blacklist to forbid all IPs within it to access the
	gateway.
SYSLOG Enabled	Sets whether to enable SYSLOG. It is required to fill in SYSLOG Server Address
	and SYSLOG Level in case SYSLOG is enabled. By default, SYSLOG is disabled.
Server Address	Sets the SYSLOG server address for log reception.
SYSLOG Level	Sets the SYSLOG level. There are three options: ERROR, WARNING, INFO and
	DEBUG. The default value is INFO.
AT Debug Enabled	Sets whether to enable the AT debug feature, with the default value of No. Once this
	feature is enabled, the related information about AT will be output to the SYSLOG.
Echo Mode Enabled	Sets whether to enable the echo mode, with the default value of No. Once this
	feature is enabled, both the sent and received information will be displayed.
Port	Select the port to execute the AT debug.
	Sets whether to enable the feature of CDR. It is required to fill in Server Address
CDR Enabled	and Server Port in case CDR is enabled. By default, CDR is disabled.
Server Address	Sets the server address to receive CDR.
Server Port	Sets the server port to receive CDR.
Save CDR	Sets whether to save CDR, with the default value of NO.
Amount of Saved	Sets the amount of saved CDR. Range of value: 200~10000, with the default value
CDR	of 5000.
	When this feature is enabled, the remote terminal can invoke the API interface. The
API Enabled	default value is <i>No</i> .
Remote IP Address	Sets the remote IP addresses which are allowed to invoke the API interface. Up to 5
allowed to Invoke	addresses can be configured and each of them are separated by ",". "*" denotes all
API	IP addresses are allowed.
Username for API	
Call, Password for	The authorized username and password for calling the API interface.
API Call	
Time Calibratian	Sets the calibration mode for the time. Three options available: NTP, Synchronized
I ime Calibration	with Operator and Close, with the default value of Synchronized with Operator.
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.
System Time	The system time. Check the checkbox before <i>Modify</i> and change the time in the edit
	box if <i>Time Calibration</i> is set to Close.
Time Zone	The time zone of the gateway.
Daily Restart	Sets whether to restart the gateway regularly every day at the preset Restart Time .
	By default, this feature is disabled.
Restart Time	Sets the time to restart the gateway regularly.


3.5.3 Service Config

Service Parameters		O	
	Enable Two Stage Dialing Mode for PSTN Outgoing Calls	Obisable OEnable	
	Maximum wait time for PSTN Outgoing Calls	60	- 5
	Diai Interval	6	S
	Busy Tone Detection Mode	OCommon ODelay	Olgn
Abnormality Handling			
	Communicate without Network	ODisable OEnable	
	IP->Tel Call Failure, Auto Transfer	OEnable	
	Tel->IP Call Failure, Auto SMS Reply	ODisable OEnable	
Echo Canceller			
	Work Mode	Both near-end an	
	Non-linear Processing	✓Enable	
	Fixed Window Size (Near-end, Narrowband 8kHz)	8ms 🗸	
	Moving Window Size (Far-end, Narrowband 8kHz)	8ms 💌	

Figure 3-29 Service Config Interface

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

ltem	Description
Enchla Two Stores	Sets whether to enable the two stage dialing mode for PSTN outgoing calls. Under
Dialing Mode for	this mode, for an outgoing call from a wireless port, the IP side will hear the dial
Dialing Mode for	tone. If you fail to input the number during the schedule time, the wireless port will
	hang up the call automatically; otherwise, it will make an outgoing call to the number.
Calls	The default value is <i>disabled</i> .
Maximum Wait Time for PSTN Outgoing Calls	Sets the maximum wait time waiting for the called party pickup during an outgoing call. Range of value: 10~120, calculated by s, with the default value of 60.
Dial Interval	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 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.
Busy Tone Detection Mode	Sets the busy tone detection mode, three options available: Common (hangup on busy), Delayed (Delayed hangup on busy), Undetected (no busy detection). By default it is set to Common.



Communication	Automatically routes a call to the wireless port in case of network failure or call		
without Network	timeout. The default value is <i>disabled</i> .		
	Sets whether to enable the feature of transferring the call to a designated IP		
IP→Tel Call Failure,	automatically when a call from IP to Tel fails, with the default value of disable. If this		
Auto Transfer	feature is enabled, you are required to enter Target Number (Registered) or Target		
	IP and Target Port (Unregistered).		
	Sets whether to enable the feature of automatic SMS reply when a call from Tel to IP		
	fails, with the default value of <i>disable</i> . The following four options will be available if		
	this feature is enabled. They are Unconnected, No Answer, Rejected, Fail to		
Auto SMS Reply	Connect. You can select any one of them and define the corresponding content to		
	reply.		
	Sets the work mode for the echo canceller. There are two options: Near-end		
Work Mode	cancellation and Both near-end and far-end cancellation, with the default value of		
	Near-end cancellation.		
Non-linear	Sets whether to enable the mode of non-linear processing. By default, this feature is		
Processing	enabled.		
Fixed Window Size	Sets the size of the window for the fixed cancellation.		
Moving Window			
Size	Sets the size of the window for the moving cancellation.		

3.5.4 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 Charact	er Mode	Dialing Rule		
Check	Index	Dialing Rule	Description	Modify
	81	400xxxxxxx	default	R
	82	40[1-9]x000x	default	Q
	83	4[1-9]xxxxxx	default	
	84	800хососох	default	
	85	80[1-9]xxxxx	default	2
	86	8[1-9]xxxxxx	default	6
	87	[2-3,5-7]xxxxxxx	default	
	. 88	1[3-5,7-8]xxxxxxxx	default	
	89	100xx	default	
	90	95xxx	default	
	91	123xx	default	
	92	111xx	default	
	93	11[0,2-9]	default	2
	94	120	default	2
	95	0[3-9]xxxxxxxxxxxx	default	2
	96	02xxxxxxxxx	default	
	97	010xxxxxxxx	default	6
	98	01[3-5,7-8]x00000000	default	6
	99		default	
heck All 📃 Uncheck	All = Inverse = Dei	ere Clear All		Add Ne

Figure 3-30 Dialing Rule Configuration Interface (Standard)

See Figure 3-30 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 3-31 for the dialing rule adding interface.

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

Figure 3-31 Add New Dialing Rule

The table below explains the items shown in Figure 3-31.

ltem	Description		
Indox	The unique index of each dialing rule, which denotes its priority. A dialing rule with a		
Index	smaller index value has a higher priority and will be checked earlier while matching.		
Description	Remarks for the dialing rule. It can be any information, but can not be left empty.		
Dialing Rule	Up to 99 dialing rules can be configured in the gateway, and the maximum length of		



each dialing rule is 127 characters. See below for the meaning of each character in				
the dialing rule. The gateway will do instant matching for your dialing number based				
on the dialing rule and regard your dialing as finished upon receiving '#' or dialing				
timeout.				
Character	er Description			
"0"~"9" Digits 0~9.				
"A"~"D"				
	A random number. A	string of 'x's represents several random		
· ^	numbers. For example	e, 'xxx' denotes 3 random numbers.		
- 66 99	'.' indicates a randor	m amount (including zero) of characters		
· • · ·	after it.			
-	"[]' is used to define th	ne 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 thes	e two numbers.		
""	',' is used to separate	numbers or number ranges, representing		
. *	alternatives.			
- "*"	Only represents symbol "*".			
"#"	Only set it at the beginning of the string, representing symbol "#".			
There are 19 dialing rules already configured on the gateway for easy use. See				
below for detai	led information.			
Priority	Dialing Rule	Description		
99		Any number in any length.		
00		Any 12-digit number starting with 013,		
98	01[3-5,7-8]XXXXXXXXX	014, 015, 017 or 018		
97	010xxxxxxx	Any 11-digit number starting with 010		
96	02xxxxxxxxx	Any 11-digit number starting with 02		
05		Any 12-digit number starting with 03, 04,		
90	0[3-9]XXXXXXXXX	05, 06, 07, 08 or 09		
94	120	Number 120。		
03	11[0 2-0]	Number 110, 112, 113, 114, 115, 116, 117,		
	11[0,2-9]	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,		
	10 0,7 0]77777777	15, 17 or 18		
87	[2-3 5-7]	Any 8-digit number starting with 2, 3, 5, 6		
07	اد-۵,۵-۱]۸۸۸۸۸۵	or 7		



	86 8[1-9]xxxxxx	8[1-0]	Any 8-digit number starting with 81, 82,
		83, 84, 85, 86, 87, 88 or 89	
	05	0014 01-000	Any 8-digit number starting with 801, 802,
	00	00[1-9]xxxxx	803, 804, 805,.806, 807, 808 or 809
	84	800xxxxxx	Any 10-digit number starting with 800
	83 /[1_0]vvvvv	Any 8-digit number starting with 41, 42,	
	03	4[1-9]XXXXXX	43, 44, 45, 46, 47, 48 or 49.
	00	40[4_0].	Any 8-digit number starting with 401, 402,
	02	40[1-9]XXXXX	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 3-30 to modify the dialing rules. See Figure 3-32 for the dialing rule modification interface. The configuration items on this interface are the same as those on the *Add New Dialing Rule* interface.

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

Figure 3-32 Modify Dialing Rule

To delete a dialing rule, 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 dialing rules at a time, click the *Clear All* button in Figure 3-30.

See Figure 3-33 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	<u>~</u>
40[1-9]xxxxx default	
4[1-9]xxxxx 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]xxxxxxxx default	~
20 Items Total	
Save	

Figure 3-33 Dialing Rule Configuration Interface (Character)

3.5.5 Function Key

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

Function Key					
Function	Enable	Function Key	Mode		
Query LAN		*11*	Default		
Set LAN		*61*	Default		
	Save				

Figure 3-34 Function Key Configuration Interface

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



3.5.6 Cue Tone

	C	ue Tone		
Languag	e	English	~	Save
	l	Jpload		
Upload a file of cue tone Two Stage Dialing	File of cue tone for IVR	v	Browse	Upload
Prompts for PSTN Outgoing Calls	Dial Tone	<u> </u>	Save	
Note: The file should less than 100KB in si	be a wav file with 8000Hz ze.	sampling ra	te, 16-bit mono, A-I	aw formatted, and

Figure 3-35 Cue Tone Interface

See Figure 3-35 for the Cue Tone interface. The table below explains the items shown in the above figure.

ltem	Description				
1	Sets the language for the gateway to play voice, including two options Chinese and				
Language	English. The default setting is <i>English.</i>				
Upload a file of cue	Uploads a user-defined cue tone file to the gateway.				
tone					
Two Stage Dialing	Sets the cue tone of two stage dialing for the PSTN outgoing calls, including two				
for PSTN Outgoing	options: Dial Tone and File Playback. You are required to upload a file for playing if				
Calls Tips	File Playback is selected.				

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

3.5.7 Color Ring

Operation Info	*	
Quick Config	*	
VoIP	*	No availat
	*	Up
Network		
System Param		
Service Config		
Dialing Rule		
Function Key		
Cue Tone		
Color Ring		

Figure 3-36 Color Ring Interface

By default, there is no available color ring on the gateway. See Figure 3-36. Click **Upload** to upload a new color ring manually. Follow Figure 3-37 to upload the required color ring file to the gateway.



Description	default	
Upload		
Color Ring	Browse	
Note: The file should be a 200KB in size.	wav file with 8000Hz sampling rate, 16-bit mono, A-law formatted, and I	ess thar

Figure 3-37 Color Ring Upload Interface

The table below explains the items shown above:

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

After configuration, click **Upload** to upload the color ring file to the gateway or click **Return** to cancel the upload. See Figure 3-38 for the Color Ring Management interface after the upload.

Color Ring Manage							
Check	Index	Color Ring	Modify				
1 1		ringtone1 📿					
Check All 🗧 Uncheck All 🗧	Inverse 🗄 Delete 🗄	Clear All		Upload			
I Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 👻 1 Pages Total							

Figure 3-38 Color Ring Management Interface

Click *Modify* in Figure 3-38 to modify the configuration of the color ring. See below for the color ring modification interface. The configuration items on this interface are the same as those on the *Color Ring Upload* interface.

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

Figure 3-39 Color Ring Modification Interface



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To delete a color ring, check the checkbox before the corresponding index in Figure 3-38 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all color rings at a time, click the **Clear All** button in Figure 3-39.

3.5.8 QoS

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

Figure 3-40 Differentiated Services Setting Interface

See Figure 3-40 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.

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



3.5.9 Tone Generator

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

Figure 3-41 Tone Generator Setting Interface

See Figure 3-41 for the Tone Generator Setting interface. By default, there are three tones on it: Dial Tone—a single tone with 450HZ frequency, plays continuously; Ringback Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 1s play and 4s pause; Busy Tone—a single tone with 450HZ frequency, repeatedly playing in the method of 350ms play and 350ms pause. You can configure the tone generator manually. The exact explanation about the format and the meaning is described on the right of the interface. The value range of the tone energy herein above is -12~17, calculated by dB, with the default value of 0.



3.5.10 CDR Query

CDR Inquire					
Starting Date	2015-10-22				
Ending Date	2015-10-23				
Port	All				
Call Direction	All				
CalleriD					
CalleelD					
Call Duration(s)					
Query					

Figure 3-42 CDR Query Setting Interface

See Figure 3-42 for the CDR Query Setting interface. The table below explains the items shown in the above figure.

Item Description					
Starting Date,					
Ending Date	Sets the starting and ending dates for CDR query.				
Port Sets the port on which CDR query will proceed.					
Call Direction	Sets the call direction for CDR query.				
CallerID, CalleeID	Sets the CallerID/CalleeID for CDR query.				
Call Duration	Sets the minimum/maximum call duration for CDR query.				

Click *Query* to query the CDR information corresponds to the above settings.

CDR Info					CDR Export				
Port	Starting Time	Answer Time	Call Direction	CallerID	CallerIP	CalleeID	Hungup Side	Pending Reason	Call Duration(s)
2	2015-10-23 16:37:47	2015-10-23 16:37:47	Tel->IP	057188861158			Gateway	MATCH_DIALDIGIT_FAILED	13
Delete	a All								
1 Item	Total 20 Items/Page 1/1	First Previous Next La	st Go to Page 1	 1 Pages Total 					

Figure 3-43 CDR Information Interface

Note: This page will appear only when the CDR feature is enabled (set in 3.5.2 System Param).

3.5.11 VPN

	VPN Settings	
Enable OPENVPN	OYes ONo	Save



Figure 3-44 VPN Settings Interface

Thanks to the embedded VPN Client, the wireless gateway can access the VPN network via OPENVPN directly, not requiring extra VPN client, which simplifies the network deployment. Meanwhile, the design of both SIP signaling messages and voice streams transporting via VPN avoids possible problems induced by the SIP protocol in passing through the firewall and NAT. See Figure 3-44 for the VPN Settings interface. The table below gives the explanation to the items shown in the above figure.

Item Description			
	Sets whether to enable the VPN feature, with the default value of No. If this		
Enable OPEN VPN	feature is enabled, the gateway will work as a VPN client.		

You are required to upload the VPN certificate after enabling the VPN feature. See Figure 3-45.

	VPN Settings	
Enable OPENVPN	©Yes ONo	Save
	VPN Certificate	
Upload VPN Certificate		Browse Upload

Figure 3-45 VPN Certificate Upload Interface

Note: Refer to Appendix C VPN Certificate for how to make a VPN certificate.

3.6 Wireless Settings

Wireless Settings includes ten parts: *Basic Param*, *Wireless Param*, *Call Forwarding*, *Short Message*, *IMEI*, *USSD*, *Email*, *Balance*, *SIM Card* and *PIN Manage*. See Figure 3-46.



Figure 3-46 Wireless Settings



3.6.1 Basic Parameters

	Basic Paramete	ers
Voice		
Voice	GSM Voice Encoding	Automotio
	Som voice Encoding	Automatic
DTMF		
	GSM DTMF Send Mode	Voice Playback 💌
	GSM DTMF Receive Mode	Wireless Module Receive 🔽
	DTMF Voltage Detection for GSM	Off 0ms 🔽 On 40ms 👻
SMS		
	SMS Sending Interval(s)	1
	Maximum Pieces of Saved Logs	100
SIP Answer Code		
	Busy/Rejected	486
	No Answer	408
	Other Fault	480

Figure 3-47 Basic Parameters Setting Interface for GSM

	Basic Parameter	S	
Voice			
VOICE	WCDMA Voice Encoding	AMR	~
	the Data the Chevening	AMIX	
Network			
	Network Scan Mode	Automatic	~
	Network Scan Sequence	Automatic	~
DTME			
DTMF	WCDMA DTME Send Mode	Voice Blackeak	
	WCDMA DTMF Receive Mode	Wireless Madula Dessius	
	WODMADTMP Receive mode	Wireless Module Receive	*
SMS			
	SMS Sending Interval(s)	1	
	Maximum Pieces of Saved Logs	100	
SIP Answer Code			
	Busy/Rejected	486	
	No Answer	408	
	Other Fault	480	

Figure 3-48 Basic Parameters Setting Interface for WCDMA



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	CDMA DTMF Send Mode	Voice Playback	c	~
	CDMA DTMF Receive Mode	Chip Receive		~
	Minimum Duration at ON	28 ms		~
SMS				
	SMS Sending Interval(s)	1		
	Maximum Pieces of Saved Logs	100		
Call Forwarding				
	Set/Cancel Service No. for FWD	*72	*720	
	Set/Cancel Service No. for FWD on Busy	*90	*900	
	Set/Cancel Service No. for FWD on No Reply	*92	*920	
	Cancel All Service No.	*730		
	Cancel Service No. for Call Waiting	*740		
SIP Answer Code				
	Busy/Rejected	486		_
	No Answer	408		_
	Other Fault	480		

Figure 3-49 Basic Parameters Setting Interface for CDMA

See Figure 3-47, Figure 3-48, Figure 3-49 for the basic parameters setting interface. The table below explains the items shown in the above figure.

Item	Description
	Sets the mode of the GSM (WCDMA) voice encoding. By default, the voice
GSM (WCDMA) Voice Encoding	encoding for GSM is Automatic and for WCDMA is AMR.
GSM (WCDMA/CDMA) DTMF	Sets the mode to send the GSM (WCDMA/CDMA) DTMF, two options available:
Send Mode	Voice Playback and Remote Transmission. The default value is Voice Playback.
	Sets the mode to receive the GSM (WCDMA/CDMA) DTMF, two options
GSM (WCDMA/CDMA) DTMF	available: Chip Receive and Wireless Module Receive. The default value for
Receive Mode	GSM and WCDMA is Wireless Modulw Receive; The default value for CDMA is
	Chip Receive.
	The shortest time that a valid tone has to last at ON state, calculated by ms. The
Minimum Duration of ON	default value is 28.
Minimum Duration at ON	Note: This configuration item is only valid when the DTMF Receive Mode is set
	to Chip Receive.
DTMF Voltage Detection for	Set the On and off of the DTME detection for CSM
GSM	Set the On and on of the DTMF detection for GSM.
Network Scan Mode	Sets a network for the call, three options available: Automatic, GSM Only and



	WCDMA Only. The default value is Automatic.		
	Sets the priority of the network, three options available: Automatic, GSM prior to		
Network Scan Sequence	WCDMA and WCDMA prior to GSM. The default value is Automatic.		
	Sets the interval to send SMS for each port. Range of value: 1~60, with the		
SMS Sending Interval	default value of 1.		
Maximum Pieces of Saved	Sets the amount of the logs to be saved for each port. Range of value: 50~500,		
Logs	with the default value of 100.		
SIP Answer Code	Sets the sip answer code for each state of the calling party.		
Set/Cancel Service No. for FWD			
Unconditionally, Set/Cancel	Sets or Cancels the service No. for FWD unconditionally, FWD on busy or FWD		
Service No. for FWD on Busy,	on no reply. The former box is used to set the service No, while the latter one is		
Set/Cancel Service No. for FWD	to cancel the service No,.		
on No Reply			
Connect All Commiss No.	Used to cancel all service numbers for FWD unconditional, FWD on busy and		
Cancel All Service No.	FWD on no reply.		
Cancel Service No. for Call			
Waiting	Used to cancel the service number for call waiting.		

Click Save to save the setting into the gateway, click Reset to restore the configurations.

3.6.2 Wireless Param

				Wireless Param				
Check	Port	Cell Phone No.	IP->CDMA Voice Volume	CDMA->IP Voice Volume	IMSI	IMEI	Status	Modify
	1	18143476793	1	2	460030764810073	805589A1	Enable	
	2		1	2			Enable	
	3		1	2			Enable	
	4		1	2		·	Enable	
	5		1	2			Enable	
	6		1	2			Enable	
	7		1	2		3	Enable	
	8		1	2			Enable	

Figure 3-50 Wireless Parameters Configuration Interface

See Figure 3-50 for the Wireless Parameters Configuration interface. Click *Modify* in Figure 3-50 to modify the properties of the corresponding module. See Figure 3-51 for the Wireless Parameters Modification interface.



Wireless Param-I	Modify	
Port	1	
Cell Phone No.	18143476793	
IP>CDMA Voice Volume	1	(Range:0-3)
CDMA>IP Voice Volume	2	(Range:0-7)
Apply to all the modules (Cell Phone No. excluded)		
Modify	Cancel	

Figure 3-51 Wireless Parameters Modification Interface

The table below explains the configuration items on the Wireless Parameters Modification interface.

Item	Description
Port	The number of the port corresponding to the wireless module.
Call Dhama Na	The number of the SIM card corresponding to the wireless module. This number
Cell Phone No.	should be configured manually.
IP->GSM(WCDMA/CDMA)	The volume of the voice from IP to GSM/WCDMA/CDMA. By default, the value for
Voice Volume	GSM is 3; the value for WCDMA is 10000; the value for CDMA is 1.
GSM(WCDMA/CDMA)->IP	The volume of the voice from GSM/WCDMA/CDMA to IP. By default, the value for
Voice Volume	GSM is 40; the value for WCDMA is 3; the value for CDMA is 2.
MC	International Mobile Subscriber Identification Number, the unique identity of the SIM
11/1/21	card.
IMEI	International Mobile Equipment Identity.
0	The operator of the wireless module. It is obtained automatically. This configuration
Operator	is unavailable for CDMA module.
Working Frequency	Displays the working frequency band of the wireless module. This configuration is
Band	unavailable for CDMA module.
Status	Displays the current state of the wireless module.
	Sets whether to apply all the settings except for the cell phone number to all the
Apply to all the modules	modules.

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



3.6.3 Call Forwarding

				Call For	warding			
Check	Port	FWD Unconditionally	FWD on Busy	FWD on No Reply	FWD on Unreachable	FWD Setting Status	FWD Query Status	Modify
	1	Close	Close	Close	+8613800571176		Successful	2
	2	Close	Close	Close	+8613800571176		Successful	1
	3	Close	Close	Close	+8613800571176		Successful	
	4							
	5			- 0 -1				
	6			- 2222				
	7			. .				
	8							
			5 A				V.	

Figure 3-52 Call Forwarding Configuration Interface

See Figure 3-52 for the Call Forwarding Configuration interface. The table below explains the items shown in the above figure.

Item	Description
Port	The number of the port corresponding to the wireless module.
FWD	Sets whether to enable the feature of FWD unconditionally and the FWD number if
Unconditionally	it is enabled.
	Sets whether to enable the feature of FWD on busy and the FWD number if it is
FWD on Busy	enabled.
	Note: Be sure to disable the Call Waiting feature before using it.
FM/D on No Donky	Sets whether to enable the feature of FWD on no reply and the FWD number if it is
FWD on No Reply	enabled.
FWD on	Sets whether to enable the feature of FWD on unreachable and the FWD number if
Unreachable	it is enabled. This configuration is unavailable for CDMA module.
FWD Setting Status	Displays the setting status of the call forwarding service.
	Displays the query status of the FWD settings. This configuration is unavailable for
FWD Query Status	CDMA module.
0	Cancels all the setting on call FWD service. This item will appear if none of the call
Cancel All	FWD is selected.

Click *Modify* in Figure 3-52 to modify the properties of the corresponding port. See Figure 3-53 for the call forwarding modification interface. Then click *Modify* to save the settings into the gateway. It will take some time to apply the settings, and you can check the result in the 'FWD Setting Status' column. Click *Reset* to restore the configurations, or click *Cancel* to cancel the settings.



Call F	orwarding-Modify
Port	1
O FWD Unconditionally C	FWD Conditionally O Cancel All
FWD on Busy FWD on No Reply	
FWD on Unreachable	+8613800571176
Modify	Cancel

Figure 3-53 Wireless Service Modification Interface

3.6.4 Short Message

	Short Message					
Check	Port	Cell Phone No.	SMS Center	SMS Receiving Details	SMS Sending Details	Send SMS
	1	15990156537	8613800571500	<u> N:5</u>	<u>N:56</u>	9
	2	15990150207	8613800571500	🗩 <u>N:0</u>	<u>N:3</u>	9
	3			<u>N:0</u>	<u>N:0</u>	
	4			○ N:0	<u>N:0</u>	
	5			D N:0	<u>N:0</u>	
	6			<u>N:0</u>	<u>N:0</u>	1207
	7			<u>N:0</u>	<u>N:0</u>	-
	8	. ता कहा है		∑ N:0	<u>N:0</u>	ন ন ন ন ন ন ন ন ন ন ন ন ন ন ন ন ন ন ন
Check All	Uncheck A	Clear All				

Figure 3-54 Short Message Interface

See Figure 3-54 for the Short Message interface which displays the related information about the received/sent SMS.

Click **SMS Center** to go into the SMS Center Modification interface. See Figure 3-55. Click **Save** to save the settings into the gateway, click **Close** to cancel the settings.

SMS Center				
Port	1			
SMS Center	8613800571500			
Save	Close			

Figure 3-55 SMS Center Modification Interface

Click **SMS Receiver Details** in Figure 3-54 to go into the SMS Receiver Details interface. See Figure 3-56. Such information as the remote cell phone number, the time and the content will be displayed on this page.



Check	No.	Port	Receive/Send	Remote Phone Number	Time	Content
	1	1	Receive	10010	2015-10-15 16:24:50	82.79
	2	1	Receive Receive	10010	2015-10-15 16:24:56	82.79
	3	1	Receive 💽	10010	2015-10-15 16:28:38	82.79
	4	1	Receive	10010	2015-10-15 16:31:47	<u>82.79</u>
	5	1	Receive	8618668137917	2015-10-19 14:31:56	<u>16</u>

Figure 3-56 SMS Receiving Details Interface

To delete a piece of SMS receiving detail, check the checkbox before the corresponding index in Figure 3-56 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; to clear all SMS receiver details at a time, click the **Clear All** button in Figure 3-56; to go back to the previous page, click **Return**.

Click *Records* in Figure 3-54 to go into the SMS Sending interface. See Figure 3-57. Such information as the receive/send status of the SMS, the remote cell phone number, the time, and the content will be displayed on this page.

Check	No.	Port	Receive/Send	Remote Phone Number	Time	Content	Result	From
	1	1	🚱 Send	135167742561	2015-10-15 09:48:50	<u>coolman</u>	🤣 Successful	WEB
	2	.1	C Send	13516774256	2015-10-15 09:48:58	<u>coolman</u>	🔗 Successful	WEB
	3	1	💽 Send	10010	2015-10-15 16:22:59	102	🥑 Successful	WEB
	4	1	💽 Send	10010	2015-10-15 16:23:04	<u>101</u>	🤣 Successful	WEB
	5	1	🚱 Send	13516774256	2015-10-15 16:24:27	<u>123456</u>	🤣 Successful	WEB
	6	1	🚱 Send	13516774256	2015-10-15 16:24:33	<u>123456</u>	📀 Successful	WEB
	7	1	🚱 Send	13516774256	2015-10-15 16:24:38	<u>123456</u>	🔮 Successful	WEB
	8	1	C Send	13516774256	2015-10-15 16:24:42	123456	🕑 Successful	WEB
	9	1	🚱 Send	13516774256	2015-10-15 16:24:47	<u>123456</u>	🤣 Successful	WEB
	10	1	💽 Send	13516774256	2015-10-15 16:24:52	<u>1111111</u>	🕑 Successful	WEB
	11	1	🚱 Send	13516774256	2015-10-15 16:24:56	<u>1111111</u>	🤡 Successful	WEB
	12	1	💽 Send	13516774256	2015-10-15 16:25:01	<u>1111111</u>	🤣 Successful	WEB
	13	1	🚱 Send	13516774256	2015-10-15 16:25:09	<u>1111111</u>	🤣 Successful	WEB
	14	1	🚱 Send	13516774256	2015-10-15 16:25:15	<u>1111111</u>	📀 Successful	WEB
	15	1	🚱 Send	13516774256	2015-10-15 16:25:19	<u>123456</u>	🔮 Successful	WEB
	16	1	C Send	13516774256	2015-10-15 16:25:24	123456	🔮 Successful	WEB
	17	1	🚱 Send	13516774256	2015-10-15 16:25:30	<u>123456</u>	🔗 Successful	WEB
	18	1	💽 Send	13516774256	2015-10-15 16:25:35	<u>123456</u>	🕑 Successful	WEB
	19	1	💽 Send	13516774256	2015-10-15 16:25:40	<u>123456</u>	🥑 Successful	WEB
	20	1	💽 Send	13516774256	2015-10-15 16:25:44	123456	🥑 Successful	WEB
Check All	Unch	s/Page 1/3	Delete Cle	ar All Return	ides Total			

Figure 3-57 SMS Sending Interface

To delete a piece of record, check the checkbox before the corresponding index in Figure 3-57 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; to clear all records at a time, click the **Clear All** button in Figure 3-56; to go back to the previous page, click **Return**.

Click Send SMS in Figure 3-54 to go into the Send SMS interface. See Figure 3-58.



	Send SMS
Port Number Import	Assignation Port 1 2 3 4 5 6 7 8 Browse Import
Send to	(Separated by ',')
Encoding Format	GSM 7bit
Content	
	Note:1.SMS can be sent to 50 numbers at most. 2.Number file must be *.bt [*] .number separated by ^{*,*} or 'enter'.
	3.The length of SMS cannot exceed 600 characters. Send Clear Result
	Time Port Number Result
Result	

Figure 3-58 Send SMS Interface

The table below explains the configuration items on the Send SMS interface.

ltem	Description
Port	Select a port to send the SMS.
	Click Browse to select the required number file and then click Import to import this
Number Import	file.
Send to Enter the remote number to receive the SMS.	
Encoding Format	The encoding format for the SMS, two options available: GSM 7bit and UCS2.
Content The content of the SMS required to be sent.	
Result	Display the send result of the SMS.

Click **Send** to send out the SMS, click **Clear Result** to clear all results. Click **Reset** to restore the configurations, or click **Return** to go back to the previous



3.6.5 IMEI

IMEI	
 IMEI Modification Service Agreement Welcome to use the IMEI modification service. By using this service you accept all the following terms. (1) Used for test only This service is only provided to individuals for test. (2) Can not be used for any commercial purposes You should use this service under the premise of not violating any laws or regulations. (3) Exemption You are liable for any possible losses in your use of this service, Our company will take no legal responsibility for it. 	
Accept	

Figure 3-59 IMEI Interface

See Figure 3-59 for the IMEI interface. Read the agreement carefully and click *Accept* before you go into the IMEI Modification interface. There are two optional modes for IMEI modification: Manual Modify and Auto Modify. Click Manual Modify to go into the IMEI manual modification interface (Figure 3-60).

Port IM	El	Port IN	IEI	
1	123456785003126	2	123456785003134	
3	123456785003142	4	123456785003159	
5	123456785003167	6	123456785003175	
7	123456785003183	8	123456785003191	
Note:1.1 2.1	MEI is a 15-digit number! The digits from the first to the fo the fifteenth digit is generated a Save	urteenth are v is a check vali Reset	alid. According to them, ue.	

Figure 3-60 IMEI Manual Modification Interface

The default IMEI information will be displayed after clicking Initial Value in Figure 3-60, you can save and use it according to your requirement.

Click Auto Modify to go into the IMEI auto modification interface (Figure 3-61).



Synway Information Engineering Co., Ltd

	IMEI - Auto Modify
IMEI Auto Modification	O Disable Enable
Mode	Based on Time 66 (Minute) Based on Call (Times)
IMEI TAC IMEI Serial Number Range	66666666 500000 800000
Note:1.IMEI = TAC(8 digits) + S 2.Auto-modified IMEI valu cyclically) + Check Bit(Auto cal	Serial Number(6 digits) + Check Bit(1 digit). ue: TAC(Set value) + Serial Number(Value in the range accumulated culated).
S	ave Reset

Figure 3-61 IMEI Auto Modification Interface

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

Note: This configuration is unavailable for CDMA module.



3.6.6 USSD

Figure 3-62 USSD Setting Interface

See Figure 3-62 for the USSD Setting interface. The table below explains the items shown in the above figure.

ltem	Description
Default USSD Encoding	Sets the default encoding format for USSD, two options available: ASCII and UCS2.
Port	Sets the port used to send the USSD request.
Request	Inputs the content of the USSD request.
Respond	Displays the result of the USSD respond.



All

Selects all the available ports to send the same USSD request.

Click *Send* in Figure 3-62 to send out the USSD request. Click *Clear Data* to clear all data. **Note:** This configuration is unavailable for CDMA module.

3.6.7 Email

Mailbox Account	husidonotest@sanhuid.co	m	
Password	•••••		
Outgoing(SMTP)	201.123.116.240	Port 25	Send test
Incoming(POP3)	201.123.116.240	Port 110	Receive test
Conversion between Email & SMS	Show Log		
Convert SMS to Email	Enable		
Target Address			(Separated by ',')
Subject	SMStoEmail		
Convert Email to SMS	Enable		
Receiving Cycle	1	Minute(Ra	inge:1~60)
Subject	EmailtoSMS		
SMS Sending Port	Automatic 💌		
Return Receipt	Successful	Failed	
Note:1,Only UTF-8 and	ASCII Formatted mails are su	pported to covert t	o SMS.
2,The pure text m	ode and Unicode(UTF-8) are r	ecommended.	
3.Mails exceedin	g 300 characters may fail to be	converted.	
4,Mails have san	te subject as the settings can t	pe converted (Cas	e Insensitive). ZEpdi:(Coose Inc.
S,Email Format. (XXX:Send Tarr	et Number XXX SMS coding (Bi	11[End] [303]22	ZIERUJ, Case inse SMS content
voccord raig	erritering (b	(1° 01 0 002),222.	Sino content

Figure 3-63 Email Setting Interface

See Figure 3-63 for the Email Setting interface. The table below explains the configuration items on the Email Setting interface.

ltem	Description
Mailbox Account,	Cote the account and account of the melliner
Password	Sets the account and password of the malibox.
Outgoing (SMTP),	Coto the common address and nort for Empile and in r
Port	Sets the server address and port for Email sending.
Incoming (POP3),	Onto the second a difference on the ent for Example and in a
Port	Sets the server address and port for Email receiving.
Show Log	Click it to display the log which contains the Email to SMS converted information.
Convert SMS to	CMC can be converted to Empile if this feature is enabled
Email	SWIS can be converted to Emails II this feature is enabled.



Target Address	The target address to which the Email converted by SMS will be sent.
Subject	Sets the subject for the Email converted by SMS.
	When this feature is enabled, the mails in a designated format (See Note 4 and 5 in
Covert Email to SMS	Figure 3-63) can be converted to SMS.
Deservices Oracle	Sets the cycle to receive mails. Range of value: 1~60, calculated by minute, with
Receiving Cycle	the default value of 5.
SMS Sending Port	Sets the port from which the SMS will be sent out. The default value is automatic .
Return Receipt	Sets whether to receive a return receipt telling the mail is sent successfully or not.

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

3.6.8 Balance

			Balance Query		
Check	Port	Cell Phone No.	Time	Balance	Modify
	1	13023634185			6
	2	13082814738			6
	3	15990152395	2016-03-16 14:29:17	20.52	()
	4	15990150759	2016-03-16 14:29:18	82.19	2
	5	15990150207	2016-03-16 14:29:19	96.15	
	6	15990119352			
	7				
	8				
Check All E Und	heck All Query	Refresh		1	

Figure 3-64 Balance Query Interface

See Figure 3-64 for the Balance Query interface. You can query the balance for a designated cell phone number. Click Modify in Figure 3-64 to modify the query mode. See Figure 3-65.

Modify Que	ry Mode
Port	1
Query Mode Destination Number Content to Send Keywords to Match	SMS
Query after SIM Card Registered Query Regularly	No (Minute,0:disabled)
Apply to Other Ports	 ●Port OPort Group ✓ 1 2 3 4 5 6 7 8
Modify	Cancel

Figure 3-65 Query Mode Modification Interface

The table below explains the configuration items on the Query Mode Modification interface.



Item	Description
Query Mode	Sets the mode to query the balance.
Destination Number	Sets the destination number to query the balance
Content to Send	Sets the content to query the balance.
Keywords to Match	The balance matching the keywords will be displayed.
Query after SIM Sets whether to query the balance automatically once the SIM card is regi	
Card Registered	the base station.
Query Regularly	Sets the time to query the balance regularly.
Apply to Other Ports	Sets whether to apply these query conditions to other ports or port groups.

Click *Modify* to save the above settings into the gateway or click *Reset* to restore the configurations. Click *Cancel* to cancel the modification.

3.6.9 SIM Card

	SIM Card List							
Port	Card A	Card B	Card C	Card D	Mobile Phone Number	Auto Switch to Available SIM Card	Switch Strategy for SIM Card	Modify
1	🔛 Using	Exist	Empty	Empty	13750845226	Enable	Disable	
2	Empty	Empty	Empty	Empty		Enable	Disable	2
3	Empty	Empty	Empty	Empty		Enable	Disable	
4	Empty	Empty	Empty	Empty		Enable	Disable	
5	Empty	Empty	Empty	Empty		Enable	Disable	
6	Empty	Empty	Empty	Empty		Enable	Disable	
7	Empty	Empty	Empty	Empty		Enable	Disable	
8	Empty	Empty	Empty	Empty		Enable	Disable	2
9	Empty	Empty	Empty	Empty		Enable	Disable	
10	Empty	Empty	Empty	Empty		Enable	Disable	2
11	Empty	Empty	Empty	Empty		Enable	Disable	
12	Empty	Empty	Empty	Empty		Enable	Disable	2
13	Empty	Empty	Empty	Empty		Enable	Disable	
14	Empty	Empty	Empty	Empty		Enable	Disable	2
15	Empty	Empty	Empty	Empty		Enable	Disable	
16	Empty	Empty	Empty	Empty		Enable	Disable	2

Figure 3-66 SIM Card List Interface

See Figure 3-66 for the SIM Card List interface, which displays the states of each SIM card and the strategy to switch the SIM, etc.. Click the SIM card in Exist state to set it to Using state, at the same time, the SIM card which is ever in Using state at first will switch to Exist state. Click Modify to modify the parameters. See Figure 3-67.



SIM Card	I Management
Port	1
Auto Switch to Available SIM Card	ODisable OEnable
Switch Strategy for SIM Card	O Based on Time O Based on Call O Fixed Time O Disable (Minute) (Times) O ♥ H 0 ♥ M
Apply to All Ports	
Modify	Reset

Figure 3-67 SIM Card Management Interface

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

Item	Description	
Port	Serial number of the port on the device.	
Auto Ousitale to Austitale la	Once this feature is enabled, it will switch to other available SIM card automatically	
Auto Switch to Available SIM Card	if the current SIM card is drawn out or the corresponding port is unavailable due to	
	the SIM card is damageed. The default value is enable.	
Switch Strategy for SIM	Sets the switch strategy for the SIM card. There are four options: Based on Time,	
Card	Based on Call, Fixed Time and Disable, with the default value of Disable.	
Apply to All Ports	Sets whether to apply the above configurations to all ports.	

Click *Modify* to save the above settings into the gateway or click *Reset* to restore the configurations. Click *Return* to cancel the modification.

3.6.10 PIN Manage

	PIN Manage						
Port	SIM Card State	PIN Required	PUK Required	Setting Status	Modify		
1	Unlocked	No	No				
2							
3					-		
4	Unlocked	No	No				
5							
6		111					
7							
8		. 			5.00		

Figure 3-68 PIN Manage Interface

See Figure 3-68 for the PIN Manage interface, which display the status of the SIM card and the setting status of PIN and PUK. Click Modify to go into the modification interface. See Figure 3-69.



Port	Port1
Lock SIM Card	⊙YesONo
PIN	

Figure 3-69 PIN Manage Modification Interface

Click "Yes" and input the correct PIN to lock the SIM card. The incoming/outgoing calls will not be initiated once the SIM card is locked. See Figure 3-70.

	PIN Manage						
Port	SIM Card State	PIN Required	PUK Required	Setting Status	Modify		
1	Locked	Yes	No	1777			
2			(1.154) (1.154)	()			
3			(****)))	(<u></u>)	(<u>-11</u>)		
4	Unlocked	No	No				
5			1000	1 1 1 1 1 1 1 1 1	1771		
6				(<u></u>)	-		
7			(1000) (1 (1000) (1		(44)		
8							

Figure 3-70 SIM Card Locked PIN Required

Click Modify in Figure 3-70, you are required to input PIN again, see Figure 3-71.

	PIN Manage-Modify
Port	Port1
PIN	
Note: There is a restriction on the	number of input times of PIN and PUK. Please proceed with caution.

Figure 3-71 Input PIN Interface

After the correct PIN is input, the SIM card is still locked but the channel turns idle and allows the initiation of incoming/outgoing calls, see Figure 3-72.

	PIN Manage					
Port	SIM Card State	PIN Required	PUK Required	Setting Status	Modify	
1	Locked	No	No	Successful		
2	(1000				
3					-	
4	Unlocked	No	No			
5	North	30036				
6	(1 <u>111</u>)	(111)				
7	-				-	
8	(1 777)	17 777 1		2000		

Figure 3-72 SIM Card Lcoked Do not Require PIN

Click Modify in Figure 3-72 to unlock the SIM card or modify the PIN, see below figure.



	PIN Manage-Modify		
Port	Port1		
Lock SIM Card	⊙ Yes ◯ No		
Modify PIN	O Yes ⊙ No		
Note: There is a restriction on the nun	nber of input times of PIN and PUK. Please proceed with caution.		
Modify	Reset Cancel		

Figure 3-73 Lock SIM Card or Modify PIN Interface

The SIM card will also be locked and cannot make incoming/outgoing calls if you input a wrong PIN code three times, You are required to input the PUK to reset the PIN, see Figure 3-74.

		PIN	Manage		
Port	SIM Card State	PIN Required	PUK Required	Setting Status	Modify
1	Locked	Yes	Yes		
2	(***)				
3					-
4	Unlocked	No	No		
5					-
6					
7					-
8		(1 777),	1.777		

Figure 3-74 SIM Card Locked Need PIN and PUK

Click Modify in Figure 3-74 to input PUK and reset a new PIN, see Figure 3-75.

PUK New PIN	Port	Port1
New PIN	PUK	
	New PIN	
Confirm New PIN	Confirm New PIN	

Figure 3-75 New PIN setting interface

The SIM card is still locked but do not need PIN and PUK again after inputting the correct PUK and resetting a new PIN. The status of the port displaying in <u>Port State</u> is idle. So the port can make incoming/outgoing calls, Click **Modify** to save the above settings into the gateway or click **Reset** to restore the configurations. Click **Cancel** to cancel the modification.

Note: The SIM card will be locked forever if you input a wrong PUK more than 10 times. You need to insert a new card.

3.7 Port Settings

Port Settings includes two parts: *Port* and *Port Group*. See Figure 3-76.



٢

Figure 3-76 Port Settings

3.7.1 Port

						Port Settings						Batch	Modify
Port	Туре	SIP Account	Authentication Username	Connection Method	Bound Number	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Canceller	Color Ring	Color Ring Index	Server Index	Modify
1	GSM	8001	-	Static Binding	180	Disable	Disable	Failed	Enable	Disable	-		
2	GSM	182		Static Binding	8003	Disable	Disable	Unregistered	Enable	Disable			
3	GSM	8003		Two Stage Dialing Mode		Disable	Disable	Unregistered	Enable	Disable			12
4	GSM	8004		Two Stage Dialing Mode		Disable	Disable	Unregistered	Enable	Disable		()	
5	GSM	8005		Two Stage Dialing Mode	-	Disable	Disable	Unregistered	Enable	Disable			12
6	GSM	8006		Two Stage Dialing Mode		Disable	Disable	Unregistered	Enable	Disable			2
7	GSM	8007		Two Stage Dialing Mode	2.2221	Disable	Disable	Unregistered	Enable	Disable			12
8	GSM	8008		Two Stage Dialing Mode		Disable	Disable	Unregistered	Enable	Disable			1

Figure 3-77 Port Settings Interface

See Figure 3-77 for the Port Settings interface. The list in the above figure shows the feature and properties of each port. Click *Modify* in Figure 3-77 to modify the properties of the corresponding port. See Figure 3-78 for the Port Modification interface.

Port-	Modify
Port	1
Register Port	Yes
SIP Account	8001
Password	••••
Authentication Username	
Connection Method	Static Binding(SIP A 💌
Bound Number	180
Echo Canceller	Enable
Forbid Outgoing Call	Enable
Caller ID Detection	
Color Ring	Enable
Color Ring Index	1
Modify	Set
mouny	Carlos

Figure 3-78 Port Modification

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

ltem	Description			
Port	Serial number of the port on the device.			



	Sets whether to register the port to the SIP server.				
Posiciar Port	When this item is	s set to No, the item Reg Status on the Port Settings interface (Figure			
Register Port	3-77) shows Unregistered; when this item is set to Yes, the item Reg Status shows				
	Failed or Registe	ored.			
	When the port ini	tiates a call to SIP, this item corresponds to the username of SIP. The			
SIP Account	default SIP account is 80XX among which XX represents the corresponding port				
SIF Account	number. For example, the default SIP account corresponding to Port 1 is 8001, and				
	that correspondin	ng to Port 8 is 8008.			
Deserver	Registration pass	sword of the port. To register a port to the SIP server, both items SIP			
Password	Account and Pa	ssword must be filled in.			
	Authentication us	sername of a port, used to register the port to the SIP server when			
Authentication	IMS network is e	nabled.			
Username	Note: This item	appears only when IMS Network is enabled.			
	Port connection r	nethods include:			
	Option Description				
	Static Binding	Bind the number to a wireless port. The number will be listed in the			
	(SIP Account)	Bound Number column.			
		Under this mode, an incoming call from a wireless port will go into			
	_	the IVR system. Then IVR will play a speech prompt "Please dial			
Connection Method	Iwo Stages	the extension number". If you fail to input the correct target			
	Dialing Mode	number before IVR finishes the third repeat of the prompt, the port			
	(default)	will hang up the call automatically; otherwise, the call goes out			
	-	successfully.			
	Note: Both items	s Connection Method and Bound Number will be hidden if the SIP			
	Station feature is	enabled on the SIP Settings interface.			
54.0	The echo cance	llation feature for a call conversation over the wireless channel. By			
Echo Canceller	default, this feature is enabled and the effect can reach 128ms.				
Forbid Outgoing	If this feature is	this feature is enabled, the port will be forbidden to call out. The default setting is			
Call	disabled.				
	If this feature is e	nabled, the port will detect the Caller IDs from the incoming calls. The			
Caller ID Detection	default setting is	disabled.			
	Sets whether to	enable the color ring feature or not, with the default setting of being			
Color Ring	disabled.				
	Note: Only when	there are available color rings will this item appear.			
Color Ring Index	The index of the	The index of the color ring which is guoted by the current wireless port.			

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 port settings at the same time. See Figure 3-79 below for the Port Batch Modification interface.



Port-Batch Mo	dify
0	
Starting Port	1
Ending Port	8
Register Port	Yes
Starting SIP Account	
Starting Authentication Password	
Starting Authentication Username	
SIP Account Batch Rule	Increase
SIP Account Batch Step Size	1
Authentication Password Batch Rule	Increase 🗸
Authentication Password Batch Step Size	1
Authentication Username Batch Rule	Increase 💌
Authentication Username Batch Step Size	1
Connection Method	Static Binding(SIP A 🗸
Bound Number	
Echo Canceller	Enable
Forbid Outgoing Call	Enable
Caller ID Detection	Enable
Color Ring	Enable
Color Ring Index	1

Figure 3-79 Port Batch Modification

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

Item	Description		
Starting Port	The starting serial number of the port on the device in the batch setting.		
Ending Port	The ending serial number of the port on the device in the batch setting.		
Starting SIP Account	The starting SIP account in the batch setting.		
Starting Authentication	The starting outbouties recovered in the batch action		
Password	The starting authentication password in the batch setting.		
Starting Authentication	The starting outherstication upgraphs in the botch acting		
Username	i në starting authentication username in the batch setting.		
SID Account Botch Bulo	The rule for batch setting the SIP account, including <i>Increase</i> and <i>Decrease</i> two		
SIP Account Batch Rule	options.		
SIP Account Batch Step	Sate the increases or decreases atom size of the SID account in the batch actting		
Size	Sets the increase or decrease step size of the SIP account in the batch setting.		



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

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

3.7.2 Port Group

							Port Group Set	tings					
Check	Index	Description	SIP Account	Authenticatio	n Username	Ports	Port Select Mode	Authentication Mode	Register Status	Server Index	Color Ring	Color Ring Index	Modify
	1	default		-	-	1,2,3,4,5,6,7,8	Increase	Do Not Register	Unregistered		Disable		2
<													<u>></u>
Check	All	Uncheck Al	I Inver	se 🗄 🗌	Jelete 🗄	Clear All						P	Add New
1 Item To	tal 20 Ite	ms/Page 1/1	First Previous	Next Last Go	to Page 1 🗙	1 Pages Total							

Figure 3-80 Port Group Settings Interface

See Figure 3-80 for the port group settings interface. A port group is a set containing single or multiple ports, used to specify such properties as *Port Selection* and *Authentication Mode* for all the ports in it. A new port group can be added by the *Add New* button on the bottom right corner of the above list. See Figure 3-81 for the port group adding interface. Note that a port which has been occupied by one port group cannot be chosen by others.

Description	default.	
Description	default	
Register Port Group	YES	
SIP Account		
Password		
Authentication Username		
Server Index	1:201.123.112.12	
Authentication Mode	Do Not Register	
Port Select Mode	Increase	
Port	Port 1(GSM) Port 2(GSM) Port 3(GSM) Port 4	(GSI
	Port 5(GSM) Port 6(GSM) Port 7(GSM) Port 8	(GSI
	Check All Inverse	



Figure 3-81 Add New Port Group

The table below explains the items in the above figure.

Item	Description				
Index	The unique index of each port group, which is mainly used in the configuration of				
Index	routing rules and number manipulation rules to correspond to port groups.				
Description	More information about	t each port group, with default value of <i>default</i> .			
Pagister Port Group	To register the port group to the SIP server. Only when this configuration item is set				
	to Yes can you see the configuration items SIP Account and Password.				
SIP Account	When the port group in	itiates a call to SIP, this item corresponds to the username of			
	SIP.				
Password	Registration password of the port group. To register the port group to the SIP server,				
1 435 1014	both configuration items SIP Account and Password should be filled in.				
Authentication	Authentication username of a port, used to register the port to the SIP server when				
Heornamo	IMS network is enabled.				
Username	Note: This item appears only when IMS Network is enabled.				
Server Index	The index of the sip server which will be quoted by the current port.				
	Sets the way for SIP to make outgoing calls (Tel \rightarrow IP) on the gateway.				
	Option	Description			
	Do Not Register	SIP initiates a call in a point-to-point mode.			
	(default)				
		SIP initiates a call with the registered SIP account and			
Authentication	Register Gateway	password of the whole gateway. (Refer to 3.4.1 SIP for			
Mode		gateway registration.)			
	Desister Part Crown	SIP initiates a call with the registered SIP account and			
	Register Port Group	password of the port group.			
	Pegister Port	SIP initiates a call with the registered SIP account and			
		password of the port.			
	Group Ringing	Ring all the idle wireless ports in this port group.			
	Registration status of t	he port group. See Figure 3-80. When Register Port Group			
Register Status	is set to <i>No</i> , the value of	of this item is Unregistered; when Register Port Group is set			
	to Yes, the value of this item may be Failed or Registered.				



	When the port group	When the port group receives a call, it will choose a port based on the select mode			
	set by this configuration item to ring or to connect. The optional values and their				
	corresponding meanir	ngs are described in the table below.			
	Option	Description			
	Increase (default)	Search for an idle port in the ascending order of the port number, starting from the minimum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.			
Port Select Mode	Decrease	Search for an idle port in the descending order of the port number, starting from the maximum. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.			
	Cyclic Increase	Provided Port N is the available port found last time. Search for an idle port in the ascending order of the port number, starting from Port N+1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.			
	Cyclic Decrease	Provided Port N is the available port found last time. Search for an idle port in the descending order of the port number, starting from Port N-1. If no match is found, search repeatedly until finding a port which is allowed to enter the call waiting state.			
Port	The ports in the port group. If the checkbox before a port is grey, it indicates that the port is not available or has been occupied. All selected ports for a port group will be				
Port	displayed in the Port	s column in Figure 3-80. 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 **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.

Click *Modify* at the end of the list in **Port Group Settings Interface** to modify the properties of a port group. See Figure 3-82 for the Port Group Modification interface. The configuration items on this interface are the same as those on the *Add New Port Group* interface.



Description	default	
Register Port Group	No	~
Authentication Mode	Do Not Register	~
Port Select Mode	Increase	*
Port	✓Port 1(GSM) ✓Port 2(GSM) ✓Port ✓Port 5(GSM) ✓Port 6(GSM) ✓Port	t 3(GSM) Port 4(GSM) t 7(GSM) Port 8(GSM
	Check All Inverse	

Figure 3-82 Modify Port Group

To delete a port group, check the checkbox before the corresponding index in Figure 3-80 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 3-80.

3.8 Route Settings

Route Settings is used to specify the routing rules for calls on two directions: $IP \rightarrow Tel/IP$ and $Tel \rightarrow IP$. See Figure 3-83.



Figure 3-83 Route Settings

3.8.1 Routing Parameters

ute before Number Manipulate 🛛 💉
ute before Number Manipulate 💌

Figure 3-84 Routing Parameters Configuration Interface



See Figure 3-84 for the routing parameters configuration interface. On this interface, you can set the routing rules for calls respectively on two directions $IP \rightarrow Tel/IP$ and $Tel \rightarrow IP$ to be routing before or after number manipulation. The default value is *Route before Number Manipulate*.

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

3.8.2 IP to Tel/IP

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

Figure 3-85 IP→Tel/IP Routing Rule Configuration Interface (Standard)

See Figure 3-85 for the IP \rightarrow Tel/IP routing rule configuration interface. By default, there is no available routing rule on the gateway. The IP \rightarrow Tel/IP routing rule configuration has two modes: Standard and Character.

Under the Standard mode, click *Add New* to add them manually. See Figure 3-86. You may use the default values of all the configuration items herein.

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

Figure 3-86 Add New Routing Rule (IP→Tel/IP)


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

ltem	Description
	The unique index of each routing rule, which denotes its priority. A routing rule with
Index	a smaller index value has a higher priority. If a call matches several routing rules, it
	will be processed according to the one with the highest priority.
Description	More information about each routing rule, with the default value of <i>default</i> .
0	IP address from where the call is initiated. This item can be set to a specific IP
Source IP	address or "*" which indicates any IP address
	A string of characters at the beginning of the caller/called party number. It can be a
	specific string consisting of digits $0 \sim 9$, $\sqrt{"[*]"}$, "#" 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
Pouto by Number	will be routed to can be set via the item <i>SIP Account</i> on the Port Settings interface.
Route by Number	In such case, the configuration item Call Destination goes invalid and shows
	Route by Number on the routing rule configuration interface. The default setting is
	disabled.
Call Destination	Designate a port group or an IP for the call to route.
Destination Port	
Group	Port group to which the call will be routed.
Destination IP,	The ID address and next to which the cell will be revited
Destination Port	i ne in address and port to which the call will be fouted.

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

See Figure 3-87 for the IP \rightarrow Tel/IP 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.

Standard Mode	Character Mo	de	IF	2->Tel/IP Routing Rule			
Check	Index	Source IP	CallerID Prefix	CalleeID Prefix	Call Destination	Description	Modify
	63	*	*	*	Route by Number	default	
Check All	Uncheck All	Inverse	Delete Clear All	nes Total			Add Nev

Figure 3-87 IP→Tel/IP Routing Rule Configuration Interface

Click *Modify* in Figure 3-87 to modify a routing rule. The configuration items on the IP→Tel/IP



routing rule modification interface are the same as those on the *Add New Routing Rule* $(IP \rightarrow Tel/IP)$ interface. Note that the item *Index* cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-87 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 routing rules at a time, click the **Clear All** button in Figure 3-87.

See Figure 3-88 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 The priority decre Symbol * in Sour Destination Port port must be set Don't forget to sa	information contains such fields as Source IP, CallerID Prefix, CalleeID Prefix, Route by Number, Destination Port Group and Description. ases from top to bottom; adjacent fields are separated by a space ze IP, CallerID Prefix and CalleeID Prefix indicates any IP address or string;When Route by Number is set to 1, the Destination Port Group is enabled; When it is set to 0 and Group is set to 0, the Route by Number is enabled; When it is set to 0 and Destination Port Group is set to 1, the IP is enabled. Besides, If the IP is disabled, the destination IP and to 0. we the configuration after your modification!
* * * 0 0 defa	uut o o
1 Item Total	
	Save

Figure 3-88 IP→Tel/IP Routing Rule Configuration Interface (Character)

3.8.3 Tel to IP

Operation Info	×	Standard Mode Character Mode	
Quick Config	*		
VolP	*		
Advanced	*		No available routing rule!
🔅 Wireless	*		Add New
(i) Port	*		
Route	*		
Routing Param	eters		
IP->Tel			
Tel->IP			

Figure 3-89 Tel→IP Routing Rule Configuration Interface (Standard)

See Figure 3-89 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 3-90. You may use the default values of all the configuration items herein except for *Destination IP* and *Destination Port*.



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

Figure 3-90 Add New Routing Rule (Tel→IP)

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

Item	Description				
	The unique index of each routing rule, which denotes its priority. A routing rule with				
Index	a smaller index value has a higher priority. If a call matches several routing rules, it				
	will be processed according to the one with the highest priority.				
Description	More information about each routing rule, with the default value of <i>default</i> .				
Source Port Group	Port group from which the call is initiated. This item can be set to a specific port				
(Call Initiator)	group or '*' which indicates any port group.				
	A string of characters at the beginning of the caller/called party number. It can be a				
	specific string consisting of digits 0~9, "[*]", "#" or characters ranges defined by [].				
	'[]' represents a character within the range it defines. Values in [] only can be digits				
	'0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to				
CallerID Prefix,	indicates any characters between these two characters. ',' is used to separate				
CalleeID Prefix	characters or characters ranges, representing alternatives.) For example,				
	057[1-3,6] represents the string 0571, 0572, 0573 or 0576. Also these items can be				
	set to "*" which indicates any string. These two configuration items together with				
	Source Port Group (Call Initiator) specify a routing rule for calls.				
	Note: "[*]" represents DTFM symbol *, while "*" represents any string.				
Destination IP,	ID address and part number of the remate and to which the call will be resided				
Destination Port	P address and port number of the remote end to which the call will be routed.				

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



See Figure 3-91 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.

Standard Mor	de Characte	er Mode						
				Tel->IP Routing	g Rule			
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Destination IP	Destination Port	Description	Modify
	63		*	*	192.168.1.101	5060	default	
	- [- (-					
Check All	Uncheck	All = Inverse		ear All				Add New
1 Item Total 2	0 Items/Page	1/1 First Previous	Next Last Go to Page 1	 1 Pages Total 				

Figure 3-91 Tel→IP Routing Rule Configuration Interface

Click **Modify** in Figure 3-91 to modify a routing rule. The configuration items on the Tel \rightarrow IP routing rule modification interface are the same as those on the **Add New Routing Rule (Tel\rightarrowIP)** interface. Note that the item **Index** cannot be modified.

To delete a routing rule, check the checkbox before the corresponding index in Figure 3-91 and click the **Delete** button. **Check All** means to select all available items on the current page; **Uncheck All** means to cancel all selections on the current page; **Inverse** means to uncheck the selected items and check the unselected. To clear all routing rules at a time, click the **Clear All** button in Figure 3-91.

See Figure 3-92 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 3-92 Tel→IP Routing Rule Configuration Interface (Character)

3.9 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 3-93.





Figure 3-93 Number Manipulation

3.9.1 IP to Tel CallerID

Operation Info	*	Standard Mode Character Mode
Quick Config	*	
VolP	*	
Advanced	*	No available number manipulation rule!
ତ Wireless	*	Add New
(i) Port	*	
Route	*	
Num Manipulate	*	
IP->Tel CallerID		
IP->Tel CalleeID		
Tel->IP CallerID		
Tel->IP CalleeID		

Figure 3-94 IP→Tel CallerID Manipulation Interface (Standard)

See Figure 3-94 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 3-95 for the IP \rightarrow Tel CallerID manipulation rule adding interface. You may use the default values of all the configuration items herein.



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

Figure 3-95 Add IP→Tel CallerID Manipulation Rule

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

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



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

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

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

Standard	d Mode	Character Mod	e							
IP->Tel/IP CallerID 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	De
	63	*		*	0	0	0			c
<									>	
Check / 1 Item Tot	Check All Uncheck All Inverse Delete Clear All Item Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 V 1 Pages Total								Add New	

Figure 3-96 IP→Tel CallerID Manipulation Interface (Standard)

Click **Modify** in Figure 3-96 to modify a number manipulation rule. See Figure 3-97 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/IP C	CallerID
Index:	63 🗸
Description:	default
Call Initiator:	*
CallerID Prefix:	*
CalleeID Prefix:	*
Stripped Digits from Left:	0
Stripped Digits from Right	: 0
Reserved Digits from Rig	ht: 0
Prefix to Add:	
Suffix to Add:	
Save	Close

Figure 3-97 Modify IP→Tel CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-94 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 3-94.

See Figure 3-98 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, Ad 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!
U U U <@#> <@#> default
1 Item Total
Save

Figure 3-98 IP→Tel CallerID Manipulation Interface (Character)

3.9.2 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 3-99, Figure 3-100 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 3-94).

Standard	Mode	Character Mod	e							
					IP->Tel/IP CalleeID Num	ber 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	De
	63	*		*	0	0	0			c
<										>
Check / 1 Item Tot	al 20 lte	Uncheck All = ms/Page 1/1	Inverse First Previous N	Delete lext Last Go to Pa	Clear All ge 1 💌 1 Pages Total				Add New	

Figure 3-99 IP→Tel CalleeID Manipulation Interface(Standard)

 Standard Mode
 IP->Tel CalleelD Number Manipulation Rule

 Note: The Number Manipulation Rule contains such fields as Call initiator, CalleriD Prefix, CalleelD Prefix, 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 the configuration after your modification!

 **** 0 0 0 <@#> <@#> default



Figure 3-100 IP→Tel CalleeID Manipulation Interface (Character)

3.9.3 Tel to IP CallerID

lode	Character Mode	e							
				Tel->IP CallerID Numb	er Manipulation Rule				
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	De
63	*	*		0	0	0			4
									>
1000000									
E	Uncheck All	Inverse	Delete	Clear All				Add New	
	Index 63	Index Call Initiator 63 * Uncheck All	ode Character Mode Index Call Initiator CallerID Prefix 63 * * = Uncheck All Inverse	Index Call Initiator CallerID Prefix CalleeID Prefix 63 * * * = Uncheck All Inverse Delete	Index Call Initiator CallerID Prefix CalleeID Prefix Stripped Digits from Left 63 * * * 0 Image: Stripped Digits from Left 63 * * 0 Image: Stripped Digits from Left 63 * * 0 Image: Stripped Digits from Left Image: Stripped Digits from Left	IDE Character Mode Tel->IP CallerID Number Manipulation Rule Index Call Initiator CallerID Prefix CalleeID Prefix Stripped Digits from Left Stripped Digits from Right 63 * * 0 0 Image: Stripped Digits from Left Uncheck All Image: Stripped Digits Image: Stripped Digits from Right Image: Stripped Digits Image: Strippe	IDE CallerID Number Manipulation Rule Index Call Initiator CallerID Prefix CallerID Prefix Stripped Digits from Left Stripped Digits from Right Reserved Digits from Right 63 * * 0 0 0 Image: Stripped Digits from Left Uncheck All Image: Stripped Digits from Left Uncheck All	Tel->IP CallerID Number Manipulation Rule Index Call Initiator CallerID Prefix CalleeID Prefix Stripped Digits from Right Reserved Digits from Right Prefix to Add 63 * * * 0 0 0 Uncheck All Inverse Delete Clear All	IDE CallerID Number Manipulation Rule Index Call Initiator CallerID Prefix CallerID Prefix Stripped Digits from Right Reserved Digits from Right Prefix to Add Suffix to Add 63 * * 0 0 0 0 Image: Stripped Digits from Left Uncheck All Imverse Imverse Imverse Imverse Imverse Add New

Figure 3-101 Tel→IP CallerID Manipulation Interface (Standard)

See Figure 3-101 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 3-102 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 C	allerID
Index:	63 💌
Description:	default
Source Port Group:	*
CallerID Prefix:	*
CalleeID Prefix:	*
Stripped Digits from Left:	0
Stripped Digits from Righ	t: 0
Reserved Digits from Rig	ht: 0
Prefix to Add:	
Suffix to Add:	
Save	Close

Figure 3-102 Add Tel→IP CallerID Manipulation Rule

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

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



	matches several number manipulation rules, it will be processed according to the
	one with the highest priority.
Description	More information about each number manipulation rule, with the default value of
Description	default.
Source Port Group	Port group from which the call is initiated. This item can be set to a specific port
(Call Initiator)	group or '*' which indicates any port group.
	A string of characters at the beginning of the caller/called party number. It can be a
	specific string consisting of digits 0~9, "[*]", "#" or character ranges defined by []. '[]
	represents a character within the range it defines. Values in [] only can be digits
	'0~9', "[*]", "#", punctuations '-' and ','. ('-' is used between two characters to indicate
CallerID Prefix,	any character between these two characters. ',' is used to separate characters or
CalleeID Prefix	character ranges, representing alternatives.) For example, 057[1-3,6] represents
	the string 0571, 0572, 0573 or 0576. Also these items can be set to "*" which
	indicates any string. These two configuration items together with Call Initiator
	specify a number manipulation rule for calls.
	Note: "[*]" represents DTFM symbol *, while "*" represents any string.
Strinned Digits from	The amount of digits to be deleted from the left end of the number. If the value of
	this item exceeds the length of the current number, the whole number will be
Len	deleted.
Strinned Digits from	The amount of digits to be deleted from the right end of the number. If the value of
Bight	this item exceeds the length of the current number, the whole number will be
кіўт	deleted.
Recomined Digits	The amount of digits to be reserved from the right end of the number. Only when the
from Pight	value of this item is less than the length of the current number will some digits be
	deleted from left; otherwise, the number will not be manipulated.
Prefix to Add	Designated information to be added to the left end of the current number.
Suffix to Add	Designated information to be added to the right end of the current number.

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

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

Click **Modify** in Figure 3-101 to modify a number manipulation rule. See Figure 3-103 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 C	allerID
Index:	63 💌
Description:	default
Source Port Group:	*
CallerID Prefix:	*
CalleeID Prefix:	*
Stripped Digits from Left:	0
Stripped Digits from Righ	nt: 0
Reserved Digits from Rig	yht: 0
Prefix to Add:	
Suffix to Add:	
Save	Close

Figure 3-103 Modify Tel→IP CallerID Manipulation Rule

To delete a number manipulation rule, check the checkbox before the corresponding index in Figure 3-101 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 3-101.

See Figure 3-104 for the Tel \rightarrow IP CallerID Manipulation Interface under the Character mode. You can edit the number manipulation rule list to add a new one or modify an old one. The exact meaning of each element of the rule is described on the page.



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

Figure 3-104 Tel→IP CallerID Manipulation Interface (Character)

3.9.4 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 3-105, Figure 3-106 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 3-101).

Standard	d Mode	Character Mod	e							
					Tel->IP CalleeID Numb	er Manipulation Rule				
Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Stripped Digits from Left	Stripped Digits from Right	Reserved Digits from Right	Prefix to Add	Suffix to Add	De
	63	*	*	*	0	0	0			c
<										>
Check A 1 Item Tot	All 20 Ite	Uncheck All ms/Page 1/1	Inverse First Previous N	Delete ext Last Go to Pa	Clear All ge 1 💙 1 Pages Total				Add New	

Figure 3-105 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 Rig Prefix, Add Suffix and Description The priority decreases from top to bottom; Adjacent fields are separated by a space. Symbol * in Call Initiator, CallerID Prefix and CalleeID Prefix indicates any string; Symbol <@#> in Add Prefix and Add Suffix denotes not to add. Don't forget to save the configuration after your modification!	ht, Reserve Digits from Right, Add
0 ** 0 0 0 <@#> <@#> default	
1 item Total	
Save	



Figure 3-106 Tel→IP CalleeID Manipulation Interface (Character)

3.10 System Tools

System Tools is mainly for gateway maintenance. It provides such features as IP modification, data backup and connectivity check. See Figure 3-107 for details.

System Tools 🔗
Upgrade
Signaling Capture
Data Recording
Call Log
Change Password
Backup & Upload
Factory Reset
Restart
System Monitor
SNMP Config
PING Test
TRACERT Test
Wireless Test

Figure 3-107 System Tools

3.10.1 Upgrade

	Current Version
Serial Num	00001560
WEB	Version 1.4.0_2016061312
Service	Version 1.4.0_2016061312
FPGA	Version 6.05
U-boot	Version Aug 06 2015-15:33:00
Kernel	Version #224 Tue Dec 8 17:17:28 CST 2015
Device Type	4008-8G
Select an U	pdate File Browse
	Update Reset

Figure 3-108 Upgrade Interface

See Figure 3-108 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 3-109.



	Current version
Serial Num	00001560
WEB	Version 1.4.0_2016061312
Service	Version 1.4.0_2016061312
FPGA	Version 6.05
U-boot	Version Aug 06 2015-15:33:00
Kernel	Version #224 Tue Dec 8 17:17:28 CST 2015
Device Type	4008-8G
The file is	uploading. Please do not leave this page!
	Upgrade Information
start upload u	Upgrade Information

Figure 3-109 File Uploading Interface

After a successful uploading of the file, the gateway will start to upgrade the system. See Figure 3-110 and you can learn the detailed upgrading information from the upgrade information box at the bottom.



	Current Version
Serial Num	00001560
WEB	Version 1.4.0_2016061312
Service	Version 1.4.0_2016061312
FPGA	Version 6.05
U-boot	Version Aug 06 2015-15:33:00
Kernel	Version #224 Tue Dec 8 17:17:28 CST 2015
Device Type	4008-8G
	·
	Upload completion!
	2%

System updating, please do not leave this page!.....

Upgrade Information

start upload upgrade file...

Figure 3-110 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.

3.10.2 Signaling Capture

	Packet Capture	
Signaling Packet Capture	SIP&Syslog 💌 RTP Port Range 💌 50000,50767	Start Download
	Note: Only 10,000 pieces of capture data will b	be saved.

Figure 3-111 Signaling Capture Interface

See Figure 3-111 for the Signaling Capture interface. Packet capture contains Signaling Packet Capture, 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. Click *Download* to download the captured packets.



3.10.3 Data Recording

		Data Recording	_	
Recording Port	Port1	*	Start	Download
	3			

Figure 3-112 Data Recording Interface

See Figure 3-112 for the Data Recording interface. Click *Start* to start the recording. Click *Stop* to stop the recording. Click *Download* to download the recorded data.

3.10.4 Call Log

Call Log SIP Log	Enable Call Log	Download			
Call from IP Channel					Clear All
03/21/2016 10:30:26:764	IP Channel 0,Incoming call fro	om remote end "180" <sip:180< td=""><td>@201.123.112.212>,call-id: e5</td><td>1e3517f2704a4e@V0IOLTVST</td><td>UpDRzZIQkE0 Caller 180 Callee unknown match</td></sip:180<>	@201.123.112.212>,call-id: e5	1e3517f2704a4e@V0IOLTVST	UpDRzZIQkE0 Caller 180 Callee unknown match
<		III			>
Call from Port	Select a Port.	Port1 💌			Clear All
03/21/2016 10:30:26:764	IP Channel 0,Incoming call fro	om remote end "180" <sip:180< td=""><td>@201.123.112.212>,call-id: e5</td><td>1e3517f2704a4e@V0IOLTVST</td><td>UpDRzZIQkE0 Caller 180 Callee unknown match</td></sip:180<>	@201.123.112.212>,call-id: e5	1e3517f2704a4e@V0IOLTVST	UpDRzZIQkE0 Caller 180 Callee unknown match
03/21/2016 10:30:37:037	Analog Channel 32 callee tran	nslation 681>681 match IP>	TEL/IP CalleeID Manipulate ru	le()	
03/21/2016 10:30:37:041 03/21/2016 10:30:37:042	Analog Channel 32 outgoing Analog Channel 32 outgoing	call(AutoDial) 681 call(two stages dialing)			
03/21/2016 10:30:44:628	Analog Channel 32 call end, r	eason:channel enters the pen	ding state(No carrier)		
1					X
		110			2

Figure 3-113 Call Log Interface



Call Log SIP Log Download			
SIP Log	Refresh	Clear All	
03/21/2016 10:30:26:151 Message received from:201.123.115.36:5064 INVITE sip:unknown@201.123.115.177:5060 SIP/2.0 Via: SIP/2.0/UDP 201.123.115.36:5064;branch=z9hG4bK-d87543-727ee778b90f404e-1d87543-;rport Max-Forwards: 70			~
Contact <sip:180@201.123.115.36:5064> To: <sip:unknown@201.123.115.177:5060> From: 180°<sip:180@201.123.112.212>tag=d43bf877</sip:180@201.123.112.212></sip:unknown@201.123.115.177:5060></sip:180@201.123.115.36:5064>			
Call D: e51e3517/270444e@V0IOL1VS10DDR2/QRE0 CSeq: 1 INVITE Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO Content-Type: application/sdp Supported: eventlist User-Agent: eyeBeam AudioOnly release 3015c stamp 27106 Content-Length: 279			
v=0 o=- 5374020 5374036 IN IP4 201.123.115.36 s=eyeBeam AudioOnly c=IN IP4 201.123.115.36 t=0 0 m=audio 8650 RTP/AVP 0 8 3 18 102 101 a=att 1 1: BC0EF2C4 DE47B706 201.123.115.36 8650 a=tmtp:101 10-15 a=tmtp:102116/16000 a=tmtp:102116/16000 a=tmtp:101 telephone-event/8000 a=sendrecv			
			*

Figure 3-114 SIP Log Interface

See Figure 3-113, Figure 3-114 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**.

3.10.5 Change Password

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

Figure 3-115 Password Changing Interface

See Figure 3-115 for the password changing interface where you can change username and password of the gateway. Enter the current password, the new username and password, and then confirm the new password. After configuration, click **Save** to apply the new username and password or click **Reset** to restore the configurations. After changing the username and password, you are required to log in again.



3.10.6 Backup & Upload

	Data Backup	
To backup the configuration file,	, click the 'Backup' button to start.	Backup
	Data Upload	
To upload a configuration file, s	elect it and click the button 'Upload' to start.	

Note: After you successfully upload the configuration file, the gateway will restart automatically.

Figure 3-116 Backup & Upload Interface

See Figure 3-116 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
	Bata Helead	
To upload a Configuration	Are you sure to upload configuration file?	to start.
Not	OK Cancel	he gateway will restart automatically.

Figure 3-117 Backup & Upload & Prompt Interface

Click **OK** on the prompt box (Figure 3-117) to upload the configuration file to the gateway. Now the prompt information 'System is rebooting, please do not leave this page' appears. See Figure 3-118. The gateway will overwrite the current configurations with the uploaded data after restart. Click **Cancel** to cancel this upload directly.



	Data Backup	
To backup the configuration	file, click the 'Backup' button to start.	Backup
	Data Upload	
To upload a configuration file	e, select it and click the button "Upload' to start.	Linioad
	Divise	
Note: After you succ	cessfully upload the configuration file, the gatewa	y will restart automatically.
Svs	tem is reboting. Please do not leave th	is page!

Figure 3-118 Configuration File Uploading Interface

3.10.7 Factory Reset

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

Figure 3-119 Factory Reset Interface

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

3.10.8 Restart



Figure 3-120 System Restart Interface

See Figure 3-120 for the restart interface. Click Restart under the service restart interface to



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

3.10.9 System Monitor

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

Figure 3-121 System Monitor Configuration Interface

See Figure 3-121 for the System Monitor Configuration interface. Watchdog is a timing reset system used to avoid application crash. You can set the dog feeding interval when this feature is enabled. The feeding interval is calculated by s, with the value range of 1~15s. By default, this feature is enabled with the default value of 5s. As the feature 'Automatically restart the service if undetected' is enabled, the service application will restart automatically if it is not detected by the gateway guard application. By default, this feature is enabled.

3.10.10 SNMP Config

SNMP Configuration	LI Enable SNMP
SNMP Server Address	127.0.0.1
Monitoring Port	161
Community String Configuration Access Password	

Figure 3-122 SNMP Configuration Interface

See Figure 3-122 for the SNMP configuration interface. If the SNMP feature is enabled, once the gateway receives a request from the SNMP management software, it will collect relevant information and reply them to the SNMP management software. By default, the SNMP feature is disabled. The available information includes kernel version, CPU usage, processes, memory usage, startup information, LAN status and etc. Currently, the gateway only provides the community string for information acquisition. The table below explains the configuration items shown in Figure 3-122.

Item	Description
SNMP Server	
Address	



Monitoring Port	Monitoring Port for SNMP on the gateway.
Access Password	Community string used for information acquisition.

3.10.11 PING Test

Ping Test			
Destin	ation Address	127.0.0.1	
Ping C	ount (1-100)	4	
Packa	ge Length (56-1024 bytes)	56	
Info	Start	End	

Figure 3-123 Ping Test Interface

See Figure 3-123 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.

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

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



3.10.12 TRACERT Test

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

Figure 3-124 Tracert Test Interface

See Figure 3-124 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.

ltem	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.



3.10.13 Wireless Network Test

Port 1	
Called Number	
Conversation Time Length (s) 5	
Call Times	
Start Stop	

Figure 3-125 Wireless Network Test Interface

See Figure 3-125 for the Wireless Network Test interface. This test is to check whether the SIM card inserted in the gateway port can make normal calls. The table below gives the explanation to the configuration items shown in the above figure.

ltem	Description
Port	The port used for the test
Called Number	The called party number which will be dialed for the test
Conversion Time Length	The time length of the conversion
Call Times	The times of the testing call

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



Appendix A Technical Specifications

Dimensions

4004/4008 series: 260×153×30 mm³ 4016 series: 440×44×200 mm³

Weight

4004/4008 series Net: 1.2 kg 4016 series Net: 3.5 kg

Environment

Operating temperature: 0 $^\circ\!C\!-\!45\,^\circ\!C$

Storage temperature: -20 °C—85 °C

Humidity: 8%— 90% non-condensing

Storage humidity: 8%— 90% non-condensing

LAN

Amount: 2 (10/100 BASE-TX (RJ-45))

Self-adaptive bandwidth supported

Auto MDI/MDIX supported

Console Port

Amount: 1 (RS-232)

Baud rate: 115200bps

Connector: RJ45 to DB-9 Connector (4004/4008 series), Mini-USB connecting line (4016 series)

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: 12V DC ±10% Input Current: ≥3A DC Signaling & Protocol SIP signaling Supported protocol: SIP V1.0/2.0, RFC3261 **Network Protocol** IP v4, UDP/TCP, PPPoE, DHCP, FTP/TFTP ARP. RARP. NTP. HTTP. Telnet Audio Encoding & Decoding G.711A 64 kbps G.711U 64 kbps G.729A/B 8 kbps 5.3/6.3 kbps G723 G722 64 kbps AMR 4.75 kbps iLBC 13.3/15.2 kbps

Sampling Rate

8kHz

Wireless Feature

GSM Frequency band: 850/900/1800/1900MHz WCDMA Frequency band: GSM 900/1800MHz, UMTS 900/2100MHz CDMA Frequency band: CDMA 2000 800MHz SMS CODEC: ASCII/UCS2



Appendix B Troubleshooting

Q1. What to do if I forget the IP address of the wireless gateway?

There are two ways to get the IP address:

- 1) Long press the Reset button on the gateway to restore to factory settings. The default IP address is 192.168.1.101
- 2) Make a call to any wireless port and press the function key to query the IP address. See <u>3.5.5 Function Key</u> for more details.

Q2. In what cases can I conclude that the wireless 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 with the antenna and has a SIM card properly inserted, but the port indicator never lights up after the gateway startup or the color it lights up does not comply with the actual port state or port type.

Other problems such as inaccessible calls, failed registrations, incorrect numbers are probably caused by configuration errors. We suggest you refer to Chapter 3 WEB Configuration for further examination. If you still cannot figure out or solve your problems, please feel free to contact our technicians.

Q3. What to do if I cannot enter the WEB interface of the 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.

Q4. Is there any cell-phone APP can make calls to the 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,

Q5. Which RTP codecs are supported by the gateway?

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



Appendix C VPN Certificate

The steps to make a VPN certificate;

Step 1 Get the file of client.ovpn from the VPN server and rename it to "client.conf".

Step 2 Examine or add the following content in/to the file.

The file should contain the following content, in which the black part is fixed while the red part shall change according to the note.

dev tap (Note: Fill in tap or tun according to the VPN server's requirement. Currently, only tap is supported.)

persist-tun

persist-key

cipher AES-128-CBC

tls-client

tls-auth ta.key 1 (Note: It is used to enable the feature of TLS encryption, and should be consistent with that of the server.)

client

remote 192.168.143.235 1194 udp (Note: Fill in the IP address and the port number of the VPN server.)

tls-remote yfadmin

comp-lzo

passtos

<ca>-----BEGIN CERTIFICATE-----

Note: Fill in the key copied from the file of ca.crt.

-----END CERTIFICATE-----

</ca><cert>----BEGIN CERTIFICATE-----

Note: Fill in the key copied from the file of client.crt, that is, the content inbetween "-----BEGIN CERTIFICATE-----" and "-----ENDCERTIFICATE-----"

-----END CERTIFICATE-----

</cert><key>-----BEGIN RSA PRIVATE KEY-----

Note: Fill in the key copied from the file of client.key

-----END RSA PRIVATE KEY-----

</key>

<tls-auth>

Note: Fill in the key copied from the file of ta.key

</tls-auth>

Step 3 Save the file after your examination or supplement and upload it to the device. Note that the suffix of the file must be .conf.



Appendix D 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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