

DWG2000D GSM VoIP Gateway User Manual V1.0



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1. Equipment Introduction

This chapter mainly introduces functions and structures of DWG2000D-32G.

1.1 Introduction

DWG2000D-32G is full functions VoIP gateway based on IP and GSM network, which provides a flexible network configuration, powerful features, and good voice quality. It works for carrier grade, enterprise, residential users for cost-effective solution.

1.2 Applications Scenario

DWG2000D-32G implements smooth transition between PLMN (GSM) and VoIP network.

With the development of users and telecom service, mobile network and fixed network integration will be steadily increasing. DWG2000D-32G provides high quality VoIP service which perfectly meets the requirement. A typical application scenario shown as figure 1-2-1

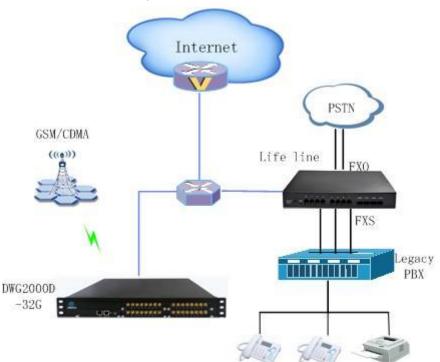


Figure 1-2-1 application scenario

1.3 Product Appearance

The appearance of DWG2000D-32G shows as follow

Figure 1-3-1 Front view of DWG2000D-32G



Table 1-3-1 Description of Front view

Index	Sign	Description
1	ANT Interface	Standard antenna interface
2	ANT indicator	Indicator of SIM card, status: register, unregister
3	LAN	10/100M Base-TX, RJ-45
4	Console	Serial port, it is a serial communication physical interface with RS232
5	RST	Keep press for 7 seconds to restore the factory setting
6	Run	Indicate the status of the device.
7	Power	Indicate the status of the power connection

Figure 1-3-2 Rear view of DWG2000D-32G



1.4 Functions and Features

1.4.1 Protocol Standard Supported

- Standard SIP and MGCP(option) protocol;
- Simple Traversal of UDP over NATs (STUN);
- Point-to-point protocol over Ethernet (PPPoE);
- Hypertext Transfer Protocol (HTTP);
- Dynamic Host Configuration Protocol (DHCP);
- Domain Name System (DNS);
- ITU-T G.711α-Law/μ-Law、G.723.1、G.729AB;
- VLAN and VPN

1.4.2 System Function

- PLC: Packet loss concealment
- VAD: Voice activity detection
- CNG: Comfort Noise Generation
- Local/Remote SIM card work mode
- Adjustable gain of port
- DTMF adjustment
- Balance alarm
- Lock/unlock SIM/UIM
- Mobile number display rejection
- Sending/receiving SMS
- Customize IVR Recording
- White and black list
- One number access
- Open API for SMS, support USSD
- Echo Cancellation (with ITU-T G.168/165 standard)
- Automatic negotiate network
- Hotline
- BCCH

1.4.3 Industrial Standards Supported

- Stationary use environment: EN 300 019: Class 3.1
- Storage environment: EN 300 019: Class 1.2
- Transportation environment: EN 300 019: Class 2.3
- Acoustic noise: EN 300 753
- CE EMC directive 2004/108/EC
- EN55022: 2006+A1:2007
- EN61000-3-2: 2006,
- EN61000-3-3: 1995+A1: 2001+A2: 2005
- EN55024: 1998+A1: 2001+A2: 2003
- Certifications: FCC, CE

1.4.4 General Hardware Specification

- Power Supply: 220VAC
- Temperature: 0~40 °C (Operation), -20~80 °C (storage)
- Humidity: 5%~90%RH
- Power Consumption: 80W
- Dimensions: 440(W) x330 (D) x66 (H) mm
- Net weight: 6.4 kg

2. Equipment Quickly Installation

This chapter mainly introduces DWG2000D-32G hardware installation and connection of equipment.

2.1 Installation Notice

DWG2000D-32G provides standard RJ45 with 10Mbps or 100Mbps interfaces.

For Wireless part, make sure antennas connecting well on device. Inserting SIM cards and GSM channels should work properly.

2.2 Installation Procedure

2.2.1 Install SIM Card

When installing SIM card, loosen the screws on the front panel of device. Procedure shows as below:

- Loosen screw, draw out the user board
- Inset the SIM card to the SIM slot of the back of the user board
- The user board inserted into the device
- Tighten the screws

Figure 2-2-1 SIM card Installation (1)

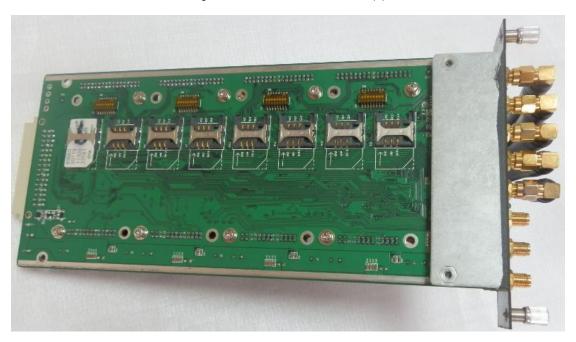


Figure 2-2-2 SIM card Installation (2)

3. Network Configuration

In this chapter we will introduce the initial configuration of DWG2000D-32G. All of the network parameters of the gateway can be configured by IVR guidance.

3.1 Attentions

In each step, if user hears an IVR message of "setting successful", which means that user has finished this step successfully. However, if user hears a "setting failed" message, please re-do that step again.

3.2 General Feature Codes for System Setting

Feature codes	Description
*114#	Play SIP user ID
*150*a#	Set IP address(static/DHCP), a can be digit 1 or 2,*150*1# is static IP address
*152*a*b*c*d#	Configure IP address, a, b, c, d are the four fields of IP address.
*153*a*b*c*d#	Configure subnet mask. a, b, c, d are the four fields of the subnet mask
*156*a*b*c*d#	Configure the device gateway, a, b, c, d are the four fields of the device gateway
*158#	Report LAN port IP address
*159#	Report WAN port IP address
*157	Setting the work mode (route or bridge), * 157 * 0 # is route mode, * 157 * 1 # is
107	bridge mode

Table 3-3-1 Feature codes for system setting

*195#	Play record
*198#	Clear record
*199#	Setting Record. dial*199# start record(≤ 20s), then press # end the recording
*111#	Restart device

3.3 Static IP Configuration

This is an optional configuration step. In case of user forgot the IP address or the device can't obtain IP address from local network properly, IVR guideline may help to fix it.

Assume that DWG2000D-32G IP address to be 172.16.80.89, subnet mask is 255.255.0.0, default

gateway is 172.16.1.1, configure it through IVR as following steps:

- 1) Please make sure SIM card installed well and registered
- Dial the phone number of the SIM card. Press "*150*1#" after heard "dial the extension number ".
 Hang up after heard "setting successful" prompt.
- 3) Dial the phone number of the SIM card. Dial "* 152 * 172 * 16 * 80 * 89 #"after heard "dial the extension number ". Hang up after heard "setting successful".
- Dial the phone number of the SIM card. Dial "*153*255*255*0*0#" after heard "dial the extension number ". Hang up after heard "setting successful"
- 5) Dial the phone number of the SIM card. Dial "*156*172*16*1*1#" after heard "dial extension number". Hang up after heard "setting successful"
- 6) Dial the phone number of the SIM card. Dial "*111#" after heard "dial extension number ", that will restart the device
- 7) Dial the phone number of the SIM card. Dial "*158#" after heard "dial extension number ". It will play report the IP address of LAN port.

3.4 DHCP Configuration

DHCP mode configure as follows:

- 1) Please make sure hardware installation have finished
- Dial the phone number of the SIM card. Dial "*150*2#" after heard "dial extension number ". That means the DHCP is configured successfully
- Restart the device, wait for 30 seconds, and then dial the SIM card telephone number, enter "*158
 #" to query the IP address

Note: If reporting the IP address is 0.0.0.0, which means that the gateway could not obtain a IP address successfully. Please check:

1) Make sure the device have been connected to the network

2) Make sure the DHCP Server is working. If there is no DHCP Server, please set the IP of device to static IP

3) Restart the gateway and try again

4. WEB Configuration

This charpter describes web configuration of DWG2000D-32G.

4.1 Access the System through HTTP

The default IP of LAN port is 192.168.11.1, before web access, make sure the PC is able to ping continuously.

Here the device's IP address is 172.16.80.89, after input this IP, the GUI shows as below:

	30.89:80 requires a username and ver says: GoAhead.
User Name:	admin
Password:	*****

Enter username and password and then click "OK" in configuration interface. The default username and

password are "admin/admin". We are strongly recommend to change the default password for security purpose.

4.2 WEB Configuration

DWG2000D-32G WEB configuration interface consists of the navigation tree and the detail configuration interfaces.

Figure 4-1-1 WEB log interface

			_									
C	41		F	un Information								
System Informa Statistics • TCP/UDP • RTP • SIP Call Histo				MAC Address Network Mode Network DNS Server	Br 17	-01-02-03-04-(dge 2.16.12.20 5.255.255.255		25	5.255.0	.0		Static
 IP to GSM Ca letwork Config Nobile Configure 	l History uration			System Up Duration Network Traffic Statistics	01	h:00m:35s ceived 318604		9	ant 3808	94 Bytes		
outing Configu				Network Haine Statistics		0011000-	io Dytes	00	5000	of Dyteo		
Manipulation Co Operation Port Group Con P Trunk Config System Configu	liguration uration			Version Information	So W Ha	vice Model ftware Version b Version rdware Versio gic Version	n	2.: 2.: PC LC	22.01.04 CB 2 OGIC 1	I Built on Ma		2, 17:48:51
Digit Map Tools					Da	P Version		۷/	_22_03	_16_HW_1	2	
lobile Info	rmation											
Port	Туре	IMSI	Status	Remaining Cal Duration	l Carrie	r	Signal Quality	BER	ASR	.(%)ACD	(s)PDI	O(s) ^{Call} Status
0	GSM		No SIM Ca		~ …		Tati	0	0	0	0	Idle
1	GSM	460021180311883			CHIN	MOBILE	Tall	0	0	0	0	Idle
2	GSM GSM		No SIM Ca No SIM Ca				Tail	0	0	0	0	ldle Idle
4	GSM		No SIM Ca				Tail	0	0	0	ő	Idle
5	GSM		No SIM Ca				Ť.	ŏ	ŏ	ŏ	ŏ	Idle
6	GSM	460002561376808		gistered No Limit	CHIN	MOBILE	Lat	ŏ	ŏ	ŏ	ŏ	Idle
7	GSM		No SIM Ca				Tall	ō	ō	ō	0	Idle
8	GSM		No SIM Ca				Ť	0	0	0	0	Idle
9	GSM	460002921115169	Mobile Re	gistered No Limit	CHIN	MOBILE	Ťall	0	0	0	0	Idle
10	GSM		No SIM Ca	rd No Limit			Taill	0	0	0	0	Idle
11	GSM		No SIM Ca	rd No Limit			Tatt	0	0	0	0	Idle
12	GSM		No SIM Ca				Tall	0	0	0	0	Idle
13	GSM		No SIM Ca				Tatl	0	0	0	0	Idle
14	GSM	460021180311884			CHIN	MOBILE	Taul	0	0	0	0	Idle
15 16	GSM GSM		No SIM Ca No SIM Ca				Latt	0	0	0	0	ldle Idle
10	GSM	460002561377342			CHIN	MOBILE		0	0	0	0	Idle
18	GSM	400002301377342	No SIM Ca		CHIN	WOBILE	Tail	0	ŏ	0	ŏ	Idle
19	GSM		No SIM Ca				Ť.	ŏ	ŏ	ŏ	ŏ	Idle
20	GSM		No SIM Ca				Ť.	ŏ	ŏ	ŏ	ŏ	Idle
21	GSM		No SIM Ca				Ť.	õ	õ	õ	õ	Idle
22	GSM	460023127139358		gistered No Limit	CHIN	MOBILE	Ťail	0	0	0	0	Idle
23	GSM		No SIM Ca	-			Taill	0	0	0	0	Idle
24	GSM		No SIM Ca				Tatt	0	0	0	0	Idle
25	GSM		No SIM Ca				Tatt	0	0	0	0	Idle
26	GSM	460003270439138			CHIN	MOBILE	Tatt	0	0	0	0	Idle
27	GSM		No SIM Ca				Tall	0	0	0	0	Idle
28	GSM	40000040005 (700	No SIM Ca				Tall	0	0	0	0	Idle
29 30	GSM GSM	460020102654729	No SIM Ca	-	CHIN	MOBILE	‡ail	0	0	0	0	ldle Idle
30	GSM	460002171979652			CHIN	MOBILE	Tail	0	0	0	ő	Idle
								-	-	-	-	
SIP Inform	ation											
Port	SIP Us	er ID Re	gister Statu	s Status	Port	SIP User	ID	F	Regist	ter Statu	S S	Status
0	99	Uni	registered	onhook	1	99		1	Unrea	istered		onhook
2	99		registered	onhook	3	99				istered		onhook
4	99		registered	onhook	5	99				istered		onhook
6	99	Uni	registered	onhook	7	99		L. L	Unreg	istered		onhook
8	99		registered	onhook	9	99				istered		onhook
10	00	Lini	harateina	onbook	11	00			Inrea	ictorod		nnhook

Figure 4-2-1 WEB introduce

Unregistered Unregistered Unregistered Unregistered Unregistered Unregistered 10 12 14 16 18 20 22 24 26 28 30 99 99 99 99 99 99 99 99 99 onhook 11 13 15 17 19 21 23 25 27 29 31 99 99 99 99 99 99 99 99 99 onhook onhook onhook onhook onhook Unregistered onhook Unregistered Unregistered Unregistered onhook onhook onhook onhook 99 onhook 99 onhook

Refresh

4.3 System Information

System information interface shows the basic information of status information, Mobile information and SIP information.

4.3.1 System Information

n Information			
MAC Address	00-01-02-03-04-05		
Network Mode	Bridge		
Network	172.16.12.20	255.255.0.0	Static
DNS Server	255.255.255.255		
System Up Duration	01h:00m:35s		
Network Traffic Statistics	Received 3186040 Bytes	Sent 389894 Bytes	
Version Information	Device Model	DWG2000D	
	Software Version	2.22.02.01 Built on May 2	3 2012, 17:48:51
	Web Version	2.22.01.04	
	Hardware Version	PCB 2	
	Logic Version	LOGIC 1	
	DSP Version	v7_22_03_16_HW_12	

Figure 4-3-1 system information

Table 4-3-1 Description of system information	-3-1 Description of system in	formation
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MAC Address	Displays current MAC of device, for example: 00-00-00-00-00-00
Network Mode	DWG2000D-32G works on bridge mode
Network	Shows IP address and subnet mask
DNS Server	Displays DNS server IP address in the same network with the gateway
System Up Duration	shows the time period of the device running. For example,:1h: 20m, 24s
Traffic Statistics	Calculates the netflow, including the total bytes of message received and sent.
Version information	shows the current firmware version

4.3.2 Mobile Information

							_	_	_		
bile Info	ormation										
Port	Туре	IMSI	Status	Remaining Call	Carrier	Signal	BER	ASR	%)ACD	(s)PDD(Call
				Duration		Quality					Status
0	GSM		No SIM Card	No Limit		Tall	0	0	0	0	Idle
1	GSM	460021180311883			CHINA MOBILE	Tall	0	0	0	0	Idle
2	GSM		No SIM Card	No Limit		Tatt	0	0	0	0	Idle
3	GSM		No SIM Card	No Limit		Tall	0	0	0	0	Idle
4	GSM		No SIM Card	No Limit		Tatt	0	0	0	0	Idle
5	GSM		No SIM Card	No Limit		Taill	0	0	0	0	Idle
6	GSM	460002561376808	Mobile Registered	l No Limit	CHINA MOBILE	Tatl	0	0	0	0	Idle
7	GSM		No SIM Card	No Limit		Taul	0	0	0	0	Idle
8	GSM		No SIM Card	No Limit		Tatt	0	0	0	0	Idle
9	GSM	460002921115169	Mobile Registered	l No Limit	CHINA MOBILE	Tail	0	0	0	0	Idle
10	GSM		No SIM Card	No Limit		Tatt	0	0	0	0	Idle
11	GSM		No SIM Card	No Limit		Tatt	0	0	0	0	Idle
12	GSM		No SIM Card	No Limit		Tatt	0	0	0	0	Idle
13	GSM		No SIM Card	No Limit		Ťatl	0	0	0	0	Idle
14	GSM	460021180311884	Mobile Registered	l No Limit	CHINA MOBILE	Tail	0	0	0	0	Idle
15	GSM		No SIM Card	No Limit		Tail	0	0	0	0	Idle
16	GSM		No SIM Card	No Limit		Ťall	0	0	0	0	Idle
17	GSM	460002561377342	Mobile Registered	l No Limit	CHINA MOBILE	Ťail	0	0	0	0	Idle
18	GSM		No SIM Card	No Limit		Tall	0	0	0	0	Idle
19	GSM		No SIM Card	No Limit		Ť	0	0	ō	Ō	Idle
20	GSM		No SIM Card	No Limit		Ť	0	ō	ō	0	Idle
21	GSM		No SIM Card	No Limit		Ϋ́.	0	Ō	ō	Ō	Idle
22	GSM	460023127139358			CHINA MOBILE	Tall	ŏ	ŏ	ŏ	ŏ	Idle
23	GSM		No SIM Card	No Limit		T.	õ	õ	õ	õ	Idle
24	GSM		No SIM Card	No Limit		Ť	ŏ	õ	ŏ	ŏ	Idle
25	GSM		No SIM Card	No Limit		Ť.	ŏ	ŏ	ŏ	ŏ	Idle
26	GSM	460003270439138	Mobile Registered		CHINA MOBILE	Tail	ŏ	ŏ	ŏ	ŏ	Idle
27	GSM		No SIM Card	No Limit	or intrancolle	Ŧ	ŏ	õ	ŏ	ŏ	Idle
28	GSM		No SIM Card	No Limit		Ŧ	ŏ	ŏ	õ	ŏ	Idle
20	GSM	460020102654729			CHINA MOBILE	Tall	õ	õ	õ	ŏ	Idle
30	GSM	400020102034729	No SIM Card	No Limit	OF INA MODILE	T 1111	0	ŏ	ŏ	0	Idle
30	GSM	460002171979652			CHINA MOBILE	Tall	0	0	0	0	Idle
31	GOM	400002171979052	wobile Registered	INO LIMIT	CHINA WOBILE	1 all ll	U	0	0	U	luie

Figure 4-3-2 Mob	ile information
------------------	-----------------

Table 4-3-2 Description of mobile information

Port	GSM channel number, it is range from 0 to 31
Туре	Indicates the current type of network. Such as CDMA or GSM
IMSI	International Mobile Subscriber Identity, it is the uniquely identifies of SIM card
Status	Indicates the register status of current GSM module
Remaining Call	Limited call duration of SIM card, when call duration is out of that duration, the call
Duration	would be disconnected. This option shows remaining talk time.
Carrier	Displays the network carrier of current SIM card.
Signal Quality	Displays the signal strength of in each channels of GSM
BER	Bit error rate, internal parameter
ASR	Answer Seizure Ratio is a measure of network quality. Its calculated by taking the
	number of successfully answered calls and dividing by the total number of calls
	attempted. Since busy signals and other rejections by the called number count as call
	failures, the ASR value can vary depending on user behavior.
ACD	The Average Call Duration (ACD) is calculated by taking the sum of billable seconds
	(billable) of answered calls and dividing it by the number of these answered calls.

PDD	Post Dial Delay (PDD) is experienced by the originating customer as the time from the
	sending of the final dialled digit to the point at which they hear ring tone or other
	in-band information. Where the originating network is required to play an
	announcement before completing the call then this definition of PDD excludes the
	duration of such announcements.
Call Status	Show the status of call, its include 3 type of status :
	Idle: the GSM channel is free. It is ready to receive the call
	Processing: the call is delivering to mobile network
	Active: the call is established

4.3.3 SIP Information

Port	SIP User ID	Register Status	Status	Port	SIP User ID	Register Status	Status
0	99	Unregistered	onhook	1	99	Unregistered	onhook
2	99	Unregistered	onhook	3	99	Unregistered	onhook
4	99	Unregistered	onhook	5	99	Unregistered	onhook
6	99	Unregistered	onhook	7	99	Unregistered	onhook
8	99	Unregistered	onhook	9	99	Unregistered	onhook
10	99	Unregistered	onhook	11	99	Unregistered	onhook
12	99	Unregistered	onhook	13	99	Unregistered	onhook
14	99	Unregistered	onhook	15	99	Unregistered	onhook
16	99	Unregistered	onhook	17	99	Unregistered	onhook
18	99	Unregistered	onhook	19	99	Unregistered	onhook
20	99	Unregistered	onhook	21	99	Unregistered	onhook
22	99	Unregistered	onhook	23	99	Unregistered	onhook
24	99	Unregistered	onhook	25	99	Unregistered	onhook
26	99	Unregistered	onhook	27	99	Unregistered	onhook
28	99	Unregistered	onhook	29	99	Unregistered	onhook
30	99	Unregistered	onhook	31	99	Unregistered	onhook

Refresh

Displays registration status information with Softswitch platform or SIP Server

Table 4-3-3 Description of SIP information

Port	The number of SIP channel, DWG2000D-32G has 32 ports
SIP User ID	SIP registration account which are provided by the Softswitch and SIP server
Register Status	Shows the registration status of VoIP channel, including registered and unregistered.
Status	Show the status of port, include "onhhok" and "offhook"

4.4 Statistics

4.4.1 TCP/UDP

Figure 4-4-1 TCP/UDP Statistics

TCP/UDP			
TCP Send Packet	TCP Recv Packet	UDP Send Packet	UDP Recv Packet
1946619	686236	221687	0
1946619	686236	221687	U

Refresh

4.4.2 RTP

Figure 4-4-2 PRI Sta	atistics
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RTP										
Port	Payload Type	Packet Period	Local Port	Peer IP	Peer Port	Send Packet	Recv Packet	Loss Packet	Jitter	Duration Time(s)

Refresh

Table 4-4-2 Description of RTP Statistics

Port	The port of RTP statistics
Payload Type	The voice code of this channel, Include G.723.1/PCMA/PCMU/G.729AB
Packet Period	Time of packaging
Local Port	Local port of transmitting RTP packages
Peer IP	End of equipment IP address
Peer Port	Peer port of receiving RTP packages
Send Packet	Total of sending RTP packages
Recv Packet	Total of receiving RTP packages
Loss Packet	Total of losing RTP packages
Jitter	Length of delay jitter
Duration Time(s)	Both ends of the call time

4.4.3 SIP Call History

Figure 4-4-3 SIP Call History

ry							
Incoming Received	Incoming Connected	Incoming Answered	Incoming Failed	Outgoing Attempted	Outgoing Connected	Outgoing Answered	Outgoing Failed
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
	Incoming Received 0 0 0 0 0 0 0	Incoming ReceivedIncoming Connected00000000000000000000	Incoming Received Incoming Connected Incoming Answered 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0	Incoming Received Incoming Connected Incoming Answered Incoming Failed 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0	Incoming ReceivedIncoming ConnectedIncoming AnsweredIncoming FailedOutgoing Attempted00	Incoming ReceivedIncoming ConnectedIncoming AnsweredIncoming FailedOutgoing AttemptedOutgoing Connected00	Incoming Received Incoming Answered Incoming Answered Incoming Failed Outgoing Attempted Outgoing Connected Outgoing Answered 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

Refresh

Port	The port of Call statistics
Incoming Received	The amount of received incoming calls which coming from IP part
Incoming connected	The amount of incoming calls which have connected
Incoming Answered	The amount of incoming calls which answered by IP part
Incoming Failed	The amount of incoming calls which failed
Outgoing Attempted	The amount of outgoing calls which attempted to IP part
Outgoing Connected	The amount of outgoing calls which have connected
Outgoing Answered	The amount of outgoing calls which answered by IP part
Outgoing Failed	The amount of outgoing calls which failed

Table of 4.4.3 SIP Call History

4.4.4 IP to GSM Call History

Figure 4-4-4 IP to GSM Call History

IP to GSM Call History												
				Cal	l Failed C	aused by	SIP	Cal	l Failed Ca	aused by	GSM	
Port	Call	Duratio n	Answer ed	Cancel ed	Timeout	Not Allowed	Negotiat ion failed	Busy	NO ANSWE R	NO DIALTO NE	NO CARRIE R	OTHER
0	0	0	0	0	0	0	0	0	0	0	0	0
1	0	0	0	0	0	0	0	0	0	0	0	0
2	0	0	0	0	0	0	0	0	0	0	0	0
3	0	0	0	0	0	0	0	0	0	0	0	0
4	0	0	0	0	0	0	0	0	0	0	0	0
5	0	0	0	0	0	0	0	0	0	0	0	0
6	0	0	0	0	0	0	0	0	0	0	0	0
7	0	0	0	0	0	0	0	0	0	0	0	0
8	0	0	0	0	0	0	0	0	0	0	0	0
9	0	0	0	0	0	0	0	0	0	0	0	0
10	0	0	0	0	0	0	0	0	0	0	0	0
11	0	0	0	0	0	0	0	0	0	0	0	0
12	0	0	0	0	0	0	0	0	0	0	0	0
13	0	0	0	0	0	0	0	0	0	0	0	0
14	0	0	0	0	0	0	0	0	0	0	0	0
15	0	0	0	0	0	0	0	0	0	0	0	0
16	0	0	0	0	0	0	0	0	0	0	0	0
17	0	0	0	0	0	0	0	0	0	0	0	0
18	0	0	0	0	0	0	0	0	0	0	0	0
19	0	0	0	0	0	0	0	0	0	0	0	0

20	0	0	0	0	0	0	0	0	0	0	0	0
21	0	0	0	0	0	0	0	0	0	0	0	0
22	0	0	0	0	0	0	0	0	0	0	0	0
23	0	0	0	0	0	0	0	0	0	0	0	0
24	0	0	0	0	0	0	0	0	0	0	0	0
25	0	0	0	0	0	0	0	0	0	0	0	0
26	0	0	0	0	0	0	0	0	0	0	0	0
27	0	0	0	0	0	0	0	0	0	0	0	0
28	0	0	0	0	0	0	0	0	0	0	0	0
29	0	0	0	0	0	0	0	0	0	0	0	0
30	0	0	0	0	0	0	0	0	0	0	0	0
31	0	0	0	0	0	0	0	0	0	0	0	0
1												

Refresh Clear

Table of 4.4.4 IP to GSM Call History

Port	The port of Call statistic	2S					
Call	The number of IP->GS	M call					
Duration	Call duration						
Answered	Response statistics	Response statistics					
Call Failed	Canceled The number of cancellation caused by SIP						
Caused by	Timeout The number of timeout caused by SIP						
SIP	Not Allowed The number of banned call caused by SIP						
	Negotiation failed	By SIP signaling negotiation fails cause calls for failure					
Call Failed	Busy	The number of call failed caused by busy					
Caused by	No Answer	The number of call failed caused by no answer					
GSM	No Dialtone	The number of call failed caused by no dialtone					
	No Carrier	The number of call failed caused by no find carrier					
Other	The number of call faile	ed by other					

4.5Network Configuration

4.5.1 Local Network

l Network	
Network Configuration	
Obtain IP address automatically	
Use the following IP address	
IP Address	172.16.12.20
Subnet Mask	255.255.0.0
Default Gateway	172.16.1.5
© PPPoE	
Account	12565463456535353
Password	**
Service Name	
DNS Server	
Obtain DNS server address automatically	
Use the following DNS server addresses	
Primary DNS Server	255.255.255.255
Secondary DNS Server	

Figure 4-5-1 Local Network

Note: It must restart the device to take effect.

Save

Obtain IP Address	Enable the device obtain IP Address automatically by DHCP or not.
Automatically	Default is enabling
Use the Following IP	Configure the "IP Address", "Subnet Mask" and "Default Gateway" by
Address	manual
	Both of account and password are provided by ISP. Use this mode when
PPPoE	there is not router in the local network.
Obtain DNS Server	When enable the WAN port option of "Obtain DNS Server Address
Address Automatically	Automatically", which will be enabled subsequently.
Use the Following DNS	Fill in the IP address of "Primary DNS Server" and "Secondary DNS
Server Addresses	Server"

Table 4-5-1 Description of Local network

4.5.2 VLAN Parameter

VLAN Parameter	
Data VLAN Data 802.1Q VLAN ID (0 - 4095) Data 802.1p Priority (0 - 7) Data VLAN use the default WAN interface in this case.	Enable
Voice VLAN Voice 802.1Q VLAN ID (0 - 4095) Voice 802.1p Priority (0 - 7) Voice VLAN use following separate IP interface Obtain IP address automatically	Enable
 Use the following IP address IP Address Subnet Mask Default Gateway 	192.168.2.5 255.255.255.0
Voice VLAN DNS Server Obtain DNS server address automatically Use the following DNS server addresses Primary DNS Server Secondary DNS Server	
Management VLAN Management 802.1Q VLAN ID (0 - 4095) Management 802.1p Priority (0 - 7) Management VLAN use following separate IP interface © Obtain IP address automatically	Enable 5 0
 Use the following IP address IP Address Subnet Mask Default Gateway 	
Management VLAN DNS Server Obtain DNS server address automatically Use the following DNS server addresses Primary DNS Server Secondary DNS Server	

Figure 4-5-2 VLAN Parameter

Table 4-5-2 Description of VLAN Parameter

Data VLAN	Data 802.1Q VLAN ID	Under standard VLAN protocol set VLAN ID. "0" is
		used to management VLAN, and can't be used to
		service configure.
	Data 802.1p Priority (0-7)	Under 802.1q protocol users can set VLAN priority
Voice VLAN	Voice 802.1Q VLAN ID	Under standard VLAN protocol set VLAN ID
	Voice 802.1p Priority (0-7)	Under 802.1q protocol users can set VLAN priority
	IP address	Users can set DHCP or static IP address
	Voice VLAN DNS Server	Users can set DHCP or static DNS server IP address
Management	Management 802.1Q VLAN	Under standard VLAN protocol set VLAN ID. "0" is
VLAN	ID	used to management VLAN, and can't be used to
		service configure.
	Management 802.1p Priority	Under 802.1p protocol users can set VLAN priority
	(0-7)	
	IP address	Users can set DHCP or static IP address
	Management VLAN DNS	Users can set DHCP or static DNS server IP address
	Server	

4.5.3 ARP

The ARP function mainly used to query and add the map of IP and MAC. There are static or dynamic ARP entries.

Like other routers, the gateway can automatically find the network device on the same segment. But, sometimes you don't want to use this automatic mapping; you'd rather have fixed (static) associations between an IP address and a MAC address. Gateway provides you the ability to add static ARP entries to:

- Protect your network against ARP spoofing
- Prevent network confusion as a result of misconfigured network device

Figure 4-5-4 Add ARP

Add ARP	
IP Address	
MAC Address	
	The IP format is: xxx.xxx.xxxx The MAC format is: xx-xx-xx-xx-xx-xx
	OK Search All

4.6 Mobile Configuration

This is Mobile Configuration menu.

Figure 4-6-1Basic Configuration

-	Mobile C	onfiguration
	Basic C	onfiguration
	Mobile	Configuration
	 PIN Ma 	nagement
	SMSC	
	 SMS 	
	 USSD 	
	Carrier	
	BCCH	

4.6.1 Basic Configuration

Basic Configuration	
Dial Tone Gain (Mobile Side)	8 dB
Select Band	Default(Automatic)
Forward Enable	🔘 No 🖲 Yes
Forward Master Mobile	Port 0
Remote API Enable	🔘 No 🔘 Yes
API Server Address	0.0.0.0
API Server Port	0
API User ID	
API User Password	Show Password
Auto Reset Module	🔘 No 🖲 Yes
Counts of NO CARRIER to reset	5
Counts of NO DIALTONE to reset	3

Figure 4-6-2Basic Configuration

NOTE: Option 'Reject Incoming' will be disabled, When 'yes' is checked on option 'Forward Enable'.

Save

Dial Tone Gain	It is the dial tone volume of call waiting, dial tone of mobile module when call
	out. default value is 8 dB.
Select Band	Acording to carrier's band standards. Standards are as belows:
	GSM: 850/900/1800/1900 MHz; CDMA: 800 MHz
Remote API Enable	API is mainly for third party software which developed based-on Dinstar API
	protocol. Its help to provide bulk SMS/ SMS/USSD over IP service.
API Server Address	It is the remote IP address who uses API. This is an option when selecting "Yes"
	under 'remote API enable"
API Server Port	It is the port number of IP transmission. This is an option when selecting "Yes"
	under "remote API enable". The port cannot conflict with the other application
	software. The default value is 12000
Auto Reset Module	Reset modular by automatically while some special errors happened, such as
	No Carrier, No Dial tone
Counts of No	A kind of the error, continuously N times will reboot the modular.
CARRIER to reset	N is range from 3 to 255.
Counts of No DIAL	A kind of the error, continuously N times will reboot the modular.
TONE to reset	N is range from 3 to 255.

Table 4-6-1 Description of Basic Configuration

4.6.2Mobile Configuration

Mobile State		5				
Port	Single Call Limitation	Call Limitation	Tx Gain	Rx Gain	Reset Module	Detail
0	No	No	3	7	Reset Module	Detail
1	No	No	3	7	Reset Module	Detail
2	No	No	3	7	Reset Module	Detail
3	No	No	3	7	Reset Module	Detail
4	No	No	3	7	Reset Module	Detail
5	No	No	3	7	Reset Module	Detail
6	No	No	3	7	Reset Module	Detail
7	No	No	3	7	Reset Module	Detail
8	No	No	3	7	Reset Module	Detail
9	No	No	3	7	Reset Module	Detail
10	No	No	3	7	Reset Module	Detail
11	No	No	3	7	Reset Module	Detail
12	No	No	3	7	Reset Module	Detail
13	No	No	3	7	Reset Module	Detail
14	No	No	3	7	Reset Module	Detail
15	No	No	3	7	Reset Module	Detail
16	No	No	3	7	Reset Module	Detail
17	No	No	3	7	Reset Module	Detail
18	No	No	3	7	Reset Module	Detail
19	No	No	3	7	Reset Module	Detail
20	No	No	3	7	Reset Module	Detail
21	No	No	3	7	Reset Module	Detail
22	No	No	3	7	Reset Module	Detail
23	No	No	3	7	Reset Module	Detail
24	No	No	3	7	Reset Module	Detail
25	No	No	3	7	Reset Module	Detail
26	No	No	3	7	Reset Module	Detail
27	No	No	3	7	Reset Module	Detail
28	No	No	3	7	Reset Module	Detail
29	No	No	3	7	Reset Module	Detail
30	No	No	3	7	Reset Module	Detail
31	No	No	3	7	Reset Module	Detail

Figure 4-6-2 Mobile State

Mobile Configuration			
Select Port	Port 0		
Mobile Number			
Enable Call Duration Limitation of single call	No O Yes		
Enable Call Duration Limitation	No O Yes		
CLIR	No O Yes		
Mobile Tx Gain	3	dB	
Mobile Rx Gain	4	dB	
Reset Module			

Figure 4-6-3 Mobile Configuration

NOTE: 1.If the duration of a call is less than 'Minimum Charging Time', it will be not included in 'Call Duration'. 2.Check the anti-pole signal is only effective on the CDMA. 3.Please enable NTP if you want to auto reset Toltal Call Time.

Back Save

Table 4-6-2 Description of Mobile Configuration		
Mobile Number	Phone number of current SIM card	
Enable Call Duration	Definite maximum call duration for single call. Example: if Time of single	
Limitation of single call	call set to 10, the call will be disconnected after talking 10*step seconds	
Step	Step length value range is 1-120 s, step length multiplied by time of single	
	call just said a single call duration time allowed.	
Time of single call	The value of limitation single call, this value range is 1-65535. step	
	length multiplied by time of single call just said a single call duration	
	time allowed.	
Enable Call Duration	This function is to limit the total call duration of GSM channel. The max call	
Limitation	duration is between 1 to 65535 minutes.	
Auto Reset	Automatic restore remaining talk time, that is, get total call minutes of GSM	
	channel	
Reset Period	Reset call minutes by date, by week, by month	
Next Reset time	Defined next reset date, system will count start from that date and work as	
	Reset Period setting	
Minimum Charging Time	A single call over this time, GSM side of the operators began to collect	
	fees, unit for seconds.	
Alarm Threshold(via SMS)	Define a threshold value of call minutes, while the call minutes less than	
	this value, the gateway will send alarm information to designated phone	

	number via SMS.
Mobile Number (Receiving	Receiving alarm phone number, user will received alarm message from
Alarm)	gateway.
Port Description for Alarm	Alarm port information description, which will be sent to user mobile phone
	with alarm information.
SIM Remain Time	This value is multiplied by to step length is a rest call time
Restore Time	Restore the rest of the SIM card talk time to the maximum call duration
CLIR	Caller ID restriction, this function is used to hidden caller ID of SIM card
	number. The gateway will add "#31#" in front of mobile number. This
	function must support by Operator.
Mobile Tx Gain	Control IP to GSM side of call the gain. Default is +6 dB.
Mobile Rx Gain	Control GSM to IP side of call the gain. Default is +6 dB.

4.6.3 PIN Management

Figure 4-6-4 PIN Management

PIN Management Select Port	Port 0 💌
SIM Card Lock PIN Code	◉ No [©] Yes

Save

Detailed description as below:

Table 4-6-4 Description of PIN Management

Select Port	Selects the GSM channel No.
SIM Card Lock	Whether to allow lock the SIM card
PIN Code	Personal identification number of SIM card. In the status of SIM card locked, PIN
	can be modified to prevent SIM card from being stolen.

4.6.4 SMSC

SMSC			
Select Port	Port 0 💌		
SMSC	+8613800755500		
	Save		

Figure 4-6-5 SMSC

SMS center of mobile, in most places, the celluar modular will automatically detect the SMSC number. This configurable option is used in a situation that the SMSC number could not detected by celluar modular. When such case happens, please contact with mobile service provider to identify the SMSC number and then add SMSC number in SMSC configurable web interface.

4.6.5 SMS

Figure 4-6-6 SMS Message		
Send Message		
Select Port	Random Port 💌	
Encoding	UCS2 💌	
	UCS2	
То	GSM 7bit	
Message		

Figure 4-6-6 SMS Message

NOTE: Length of 'Message' should be not more than 300 characters.
Send

Select Port	Users can select a defined channel or random channel to send SMS. Input the
	recevier's mobile phone number to send SMS.
Encoding	Two kinds of message encoding under PDU models, 7-bit code used to sent
	ordinary ASCII characters; UCS2 coding used to sent Unicode characters.
То	Mobile phone No. of the receiver
Message	Content of the SMS. The length is limited to 300 characters.

4.6.6 USSD

USSD (Unstructured Supplementary Service Data) is a Global System for Mobile(GSM) communication technology that is used to send text between a mobile phone and an application program in the network. Applications may include prepaid roaming or mobile chatting. USSD can cluster and group of charge

0.5	SD		
		USSD Request	USSD Reply
	0		not registered
	1		not registered
	2		not registered
	3		not registered
	4		not registered
	5		not registered
	6		not registered
	7		not registered
	8		not registered
	9		not registered
	10		not registered
	11		not registered
	12		not registered
	13		not registered
	14		not registered
	15		not registered
	16	<i>,</i>	not registered
	17	,	not registered
	18	,	not registered
	19	/	not registered
	20		not registered
	21		not registered
	22	<i>,</i>	not registered
	23		not registered
	24		not registered
	25	<i>,</i>	not registered
	26	<u> </u>	not registered
			not registered
	27	ļ,	

Figure 4-6-7 USSD

28	not registered
29	not registered
30	not registered
31	not registered
All	Copy To Select Clear All

NOTE: If you do nothing within 90s, connection will be disconnected.

Send Exit

Table 4-6-6 Description of USSD

Port	Select the GSM channel to send USSD
USSD Reply	Display the state of USSD
USSD Request	Display the result of sending USSD

4.6.7 Carrier

Figure 4-6-8 Select Carrier

Carrier	
Select Port	Port 0
Select Mode Carrier List	Automatic Manual CHINA MOBILE

Save

This function is used to select carrier.

Table 4-6-6 Description of select Carrier

Select Port	Select GSM channel, default Port 0
Select Mode	There are two modes to select carrier automatic and manual. Automatic mode can be automatically search operators. Manual mode can choose operators from the carrier list.
Carrier List	If you select manual mode, you can select carrier from carrier list.

4.6.8 BCCH

BCCI	H																					
Port	LAC	0 CID	dbm	LAC	1	dhm	140	2	dhm	140	3	dhm	140	4 CID	dhm	LAC	5	dbm	LAC	6 CID	dbm	Detail
0	LAC	CID	ubili	LAU	CID	ubili	LAC	CID	ubili	LAC	CID	dbm	LAC	CID	dbm	LAC	CID	doni	LAC	CID	ubili	Detail
1	0X2	0X	-109																			Detail
2	0742																					Detail
3																						Detail
4																						Detail
5																						Detail
6	0X2	0X	-56	0X2	0X1	-69	0X2	0X	-81	0X2	0XE	-81	0X2	0X1	-82	0X2	0XE	-82	0X2	0X1	-98	Detail
7																						Detail
8																						Detail
9	0X2	0X	-77	0X2	0X	-93	0X2	0X	-102	0X2	0X1	-103	0X2	0XE	-104	0X2	0X1	-250				Detail
10																						Detail
11																						Detail
12																						Detail
13																						Detail
14	0X2	0X	-62	0X2	0X1	-84	0X2	0X	-86	0X2	0X1	-93	0X2	0XE	-99							Detail
15																						Detail
16																						Datail
	020	07	57	020	07	00	020	01/1	05	070	OVE	05	020	OVE	0.0	020	01/4	01				Detail
17 18	0X2	UX	-57	0X2	UX	-63	UAZ	0X1	-60	0	0XE	-60	UAZ	0XE	-80	υλ2	0X1	-91				Detail Detail
19																						Detail
20																						Detail
21																						Detail
22	0X2	0X	-45	0X2	0X	-76	0X2	0X1	-82	0X2	0XE	-90	0X2	0X1	-94	0X2	0XE	-98	0X2	0XE	-99	Detail
23																						Detail
24																						Detail
25																						Detail
26	0X2	0X	-59	0X2	0X1	-80	0X2	0X	-88	0X2	0XE	-88	0X2	0X1	-91	0X2	0XE	-94	0X2	0XE	-106	Detail
27																						Detail
28																						Detail
29	0X2	0X	-57	0X2	0X1	-81	0X2	0X	-86	0X2	0XE	-90	0X2	0XE	-90	0X2	0X1	-93				Detail
30																						Detail
31	0X2	0X	-50	0X2	0X1	-71	0X2	0X	-74	0X2	0XE	-79	0X2	0X1	-83	0X2	0XE	-83				Detail

Figure 4-6-9 BCCH

Figure 4-6-10 BCCH

Auto Refresh Stop Refresh

Refresh

ССН								
	Ref	resh inte	rval		5	s		
	Au	to Refrest	١			Stop Refresh		
		Index	MCC	MNC	LAC	CID	BCCH	Receive Level
[0	460	00	0X2639	0XE88	28	-66
[1	460	00	0X2639	0XEF7	748	-96
				Refresh	Lock	nLock Back	(

	Table 4-0-7 Description of Beerr
Refresh Interval	Set frequency detection refresh time
Auto Refresh/Stop Refresh	Choose whether to refresh frequency
Index	Serial number
MCC	Mobile country code, China is 460
MNC	Mobile network code, used to distinguish between different network
	operators
LAC	Location area codes
CID	Village identification number
BCCH	Public radio channel
Receive Level	Receiving signal strong strength

Table 4-6-7 Description of BCCH

Choose a frequency to lock the operations.

4.7 Routing Configuration

4.7.1 Routing Parameter

Figure 4-7-1 Routing Parameter

Routing Parameter		
IP->Tel Parameter	Route calls before manipulation	•
Tel->IP Parameter	Route calls before manipulation	▼

Save

Tel->IP Parameter	Globle parameters, it will take effect while number manipulation configured
Route calls after	The parameters indicate that the gateway will select Tel->IP routes after number
manipulation	manipulation completed
Route calls before	The parameters indicate that the gateway will select Tel->IP routes before
manipulation	number manipulation completed

Table 4-7-1Description of Routing Parameter

4.7.2 IP->Tel Routing

Figure 4-7-2 IP to Tel Routing

Index	Description	Source IP	Source Prefix	Destination Prefix	Destination
0	default	Any	any	any	Port Group 0

Figure 4-7-3 IP to Tel Routing Add

IP->Tel Routing Add		
Index	31	
Description		
Source Prefix		
Source IP	© IP Any ▼	
	© IP Group	
	SIP Server	
Destination Prefix		
Destination	© Port 0	
	Port Group O <all></all>	
	•	
	OK Reset Cancel	

IP ->Tel Routing	This item uses to configure outgoing call routes which can be used for recieve the calls
	from the GSM
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.
	The route preferentially match the rules which the value of index is smaller
Description	It describes the route for the ease of identification. Its value is character string
Source IP	It specifies the IP of the caller
	All the caller number must match the source prefix. It specifies the source prefix allow to
	send call out
Source Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.
	Oxxxx: consist of some digits such as 015,08,09
	• 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186
	All the called number must match the destination prefix, the call prefix indicates the
	connected number
Destination Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.
	Oxxxx: consist of some digits such as 015,08,09
	1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186
Destination	Its specifies destination Port or Port Group

4.7.3 Tel->IP Routing

Index	Description	Source Port	Source Prefix	Destination Prefix	Destinatio n
0	default	Any	any	any	SIP Serve

Figure 4-7-3 Tel to IP Routing

NOTE: 0 routing is not allowed to delete, only allowed to change.

Tel -> IP Routing	This item uses to configure incoming call routes which can be used for				
	recieve the calls from the GSM.				
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to				
Index	31. The route preferentially match the rules which the value of index is smaller				
Description	It describes the route for the ease of identification. Its value is character string				
Source Port	It specifies the Port or Port Group which will receive the calls from PLMN				
	All the caller number must match the source prefix. It specifies the source				
	prefix allow to send call out				
Source Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.				
	Oxxxx: consist of some digits such as 015,08,09				
	• 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186				
	All the called number must match the destination prefix, the call prefix				
	indicates the connected number				
Destination Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.				
	Oxxxx: consist of some digits such as 015,08,09				
	1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186				
Destination	Its specifies destination Port or Port Group				

Table 4-7-3 Description of Tel to IP Routing

Tel->IP Routing Modify			
Index	0		
Description	default		
Source Prefix	any		
Source	Port	0	
	C Port Group	0 <all></all>	
Destination Prefix	any		
Destination	C Port	0	
	C Port Group	0 ⟨all⟩ ▼	
	OIP	10 <other></other>	
	C IP Group	18 <asterisk></asterisk>	
	SIP Server		
	ОК	Reset Cancel	

Figure 4-7-4 Tel to IP routing Modify

It's a default route configured in gateway. It allows any number from source port 0 send call to SIP server with any prefix.

Index	30		
Description	To vps		=
Source Prefix	х.		_
Source	O Port	0	
	Port Group	31 (Unicom)	
Destination Prefix	00		
Destination	O Port	0 🔽	
	C Port Group	0 <all></all>	
	Θ _{IP}	13 <eia></eia>	
	C IP Group	18 <asterisk></asterisk>	
	C SIP Server		

Figure 4-7-5 Tel to IP routing Modify

Add a GSM to VoIP route. It indicates that the calls coming from Port Group 31<Unicom> will match the prefix "x.", "x." is a wildcard string which will match any prefix except "anonymous" calls. Meanwhile sending the calls destination IP 13<eia> if called number match with destination prefix "00".

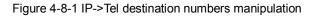
	-	
Tel->IP Routing Modify		
Index	31	
Description	Carrier A to B	
Source Prefix	13[58]	
Source	Port	0
	O Port Group	0 <all></all>
Destination Prefix	133	
Destination	O Port	0
	Port Group	31 (Unicom)
	OIP	10 <other></other>
	C IP Group	18 <asterisk></asterisk>
	SIP Server	
	OK	Reset Cancel

Figure 4-7-6 Tel to IP routing Modify

Add GSM to GSM route, its mainly used for saving the cost between two carriers. It indecates that calls coming from Port 0 will match the prefix 13[58], "13[58]" include prefix 135 and 138, caller number can't match prefix 135 and 138 will reject by gateway. Meanwhile sending the calls to Port Group 31<Unicom> if called number match with prefix 133.

4.8 Manipulaton Configuration

4.8.1 IP->Tel Destination Numbers



Inde	Description	Source	Source Prefix	Destination Prefix	Destination	Stripped Digits from Left	Stripped Digits from Right	Prefix to Add	Suffix to Add	Number of Digits to Leave from Right

Total: 0entry 16entry/page 1/0page

Add Delete Modify

IP->Tel destination	It is an optional configuration item, and is used to add a rule for changing
numbers manipulation	number
	It uniquely identifies a route. Its value is assigned globally, ranging from 0
Index	to 31. The route preferentially match the rules which the value of index is
	smaller
Description	It describes the rule for the ease of identification. Its value is character
Description	string
Source	It specifies the source IP which will send the calls to gateway

Table 4-8-1 Description of IP->Tel destination numbers manipulation

	Any: any IP address			
	IP: specific an IP address			
	IP Group: specific an IP group			
	All the caller number must match the source prefix. It specifies the source			
	prefix allow to send call out			
Source Prefix	 Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. 			
	0xxxx: consist of some digits such as 015,08,09			
	• 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186			
	All the called number must match the destination prefix, the call prefix			
	indicates the connected number			
Destination Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.			
	Oxxxx: consist of some digits such as 015,08,09			
	1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186			
Destination	Its specifies destination Port or Port Group			
Stripped Digits from	It specifies the length of the digits to be deleted from left			
Left	is specifies the length of the digits to be deleted from left			
Stripped Digits from	It specifies the length of the digits to be delated from right			
Right	It specifies the length of the digits to be deleted from right			
Prefix to Add	Add the new digits in front of the original number			
Suffix to Add	Add the new digits at the end of the original number			
Number of Digits to	It specifies the length of the digits to be deleted from left			
leave from right				

Add an IP->Tel Manipulation, to change the called number from 25478888888 to 078888888

Figure 4-8-2 IP->Tel destination numbers manipulation modify

Tel Manipulation Mod	ify		
Index	0		
Description	safcom		
Source Prefix	any		
Source IP	O IP	13 <mathnew></mathnew>	
	IP Group	31 <allow calls=""></allow>	
Destination Prefix	2547		
Destination Port	C Port	0	
	Port Group	31 <1>	
Stripped Digits from Left	3		
Stripped Digits from Right			
Prefix to Add	0		
Suffix to Add			
NOTE: If you need r	oute calls after ma	nipulation, set the destination port chose	n arbitrarily.
	OK	Reset Cancel	

It indicates that calls coming from IP Group will match the prefix "any", and the called nubmer which

match with the prefix "2547" will delete 3 digits in front of it and replace it by digit "0".

4.8.2 Tel->IP Source Numbers

Figure 4-8-3 Tel->IP destination numbers manipulation

Tel->	IP Source	e Numbers								
	Index	Description	Source Prefix	Destination Prefix	Destination	Stripped Digits from Left	Stripped Digits from Right	Prefix to Add	Suffix to Add	Number of Digits to Leave from Right
Total: 0	entry 16er	try/page 1/0pag	ge 🖵							
				Add	Delete	Modify				

	It is an optional configuration item, and is used to add IP->Tel number				
Tel->IP destination	change data.				
numbers manipulation	The IP->Tel Manipulation defined the rules of add, and deletion of called				
	numbers, which are referenced by IP->Tel routing.				
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to				
Index	31.				
Description	It describes the rule for the ease of identification. Its value is character string				
	All the caller number must match the source prefix. It specifies the source				
	prefix allow to send call out				
Source Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.				
	0xxxx: consist of some digits such as 015,08,09				
	• 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186				
	All the called number must match the destination prefix, the call prefix				
	indicates the connected number				
Destination Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.				
	Oxxxx: consist of some digits such as 015,08,09				
	1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186				
Destination	Its specifies destination Port or Port Group				
Stripped Digits from					
Left	It specifies the length of the digits to be deleted from left				
Stripped Digits from					
Right	It specifies the length of the digits to be deleted from right				
Prefix to Add	Add the new digits in front of the original number				
Suffix to Add	Add the new digits at the end of the original number				
Number of Digits to	It specifies the number of Digits to leave from Right				
Leave from Right					

Table 4-8-2 Description of Tel->IP destination numbers manipulation

Example:

Add an IP->Tel Manipulation, to change the called number from 25478888888 to 078888888

>IP Source Numbers	Add			
Index	31		•	
Description				
Source Prefix				
Destination Prefix				
Destination	© IP	Any	-	
	IP Group		-	
	SIP Server			
Stripped Digits from Left				
Stripped Digits from Right				
Prefix to Add				
Suffix to Add				
Number of Digits to Leave from Right				

Figure 4-8-4 Tel ->IP destination numbers manipulation add

NOTE: If you need route calls after manipulation, set the destination ip to any.

OK Reset Cancel

It indicates that calls coming from IP Group will match the prefix "any", and the called nubmer which

match with the prefix "2547" will delete 3 digits in front of it and replace it by digit "0".

4.8.3 Tel->IP Destination Numbers

Figure 4-8-5 Tel->IP	destination	numbers	manipulation

nation Numbers								
C Description	Source Prefix	Destination Prefix	Destination	Stripped Digits from Left	Stripped Digits from Right	Prefix to Add	Suffix to Add	Number of Digits to Leave from Right
	x Description		x Description Source Prefix Destination Prefix	x Description Source Prefix Destination Destination Prefix	Stripped x Description Source Prefix Destination Destination Digits from Prefix Left	Stripped Stripped x Description Source Prefix Destination Destination Digits from Digits from Prefix Left Right	Stripped Stripped Prefix x Description Source Prefix Destination Destination Digits from Digits from to Add Left Right	Stripped Stripped Prefix Suffix x Description Source Prefix Destination Destination Digits from Digits from to Add to Add Left Right

Total: 0entry 16entry/page 1/0page

Add	Delete	Modify
-----	--------	--------

	It is an optional configuration item, and is used to add IP->Tel number change		
Tel->IP destination	data.		
numbers manipulation	The IP->Tel Manipulation defined the rules of add, and deletion of called		
	numbers, which are referenced by IP->Tel routing.		
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to		
Index	31.		
Description It describes the route for the ease of identification. Its value is chara			
	All the caller number must match the source prefix. It specifies the source		
	prefix allow to send call out		
Source Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.		
	Oxxxx: consist of some digits such as 015,08,09		
	• 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186		
Destination Prefix	All the called number must match the destination prefix, the call prefix		
Destination Plenx	indicates the connected number		

Table 4-8-3 Description of Tel->IP destination numbers manipulation

	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.		
	Oxxxx: consist of some digits such as 015,08,09		
	1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186		
Destination	Its specifies destination Port or Port Group		
Stripped Digits from	It specifies the length of the digits to be deleted from left		
Left			
Stripped Digits from	It specifies the length of the digits to be deleted from right		
Right	The specifies the length of the digits to be deleted from right		
Prefix to Add	Add the new digits in front of the original number		
Suffix to Add Add the new digits at the end of the original number			
Number of Digits to	It specifies the number of Digits to leave from Right		
Leave from Right	is specifies the number of Digits to leave from Right		

Example:

Add an IP->Tel Manipulation, to change the called number from 25478888888 to 078888888

Figure 4-8-6 Tel->IP destination numbers manipulation

Tel->IP Destination Numb	el->IP Destination Numbers Add					
Index	31	-				
Description						
Source Prefix						
Destination Prefix						
Destination	© IP	Any				
	IP Group	•				
	SIP Server					
Stripped Digits from Left						
Stripped Digits from						
Right						
Prefix to Add						
Suffix to Add						
Number of Digits to						
Leave from Right						

NOTE: If you need route calls after manipulation, set the destination ip to any.

OK	Reset		Cancel
----	-------	--	--------

It indicates that calls coming from IP Group will match the prefix "any", and the called nubmer which

match with the prefix "2547" will delete 3 digits in front of it and replace it by digit "0".

4.9 Operation

When configure hotline, must configure operation.

4.9.1 IP->Tel Operation

IP->Tel Op	peration					
	Index	Source IP	Source Prefix	Destination Prefix	Operation	Description
	29	IP 13	any	any	Allow ,Need Pa	password
	30	IP 14	2877	13[58]	Forbid ,	restrict mobile
	31	IP 14	2877	07	Forbid ,	restrict unicom

Figure 4-9-1 IP->Tel Operation

Total: 3entry 16entry/page 1/1page Page 1 💌

Add	Delete	Modify	

Г

	It is an optional configuration item. Operation configuration essentially			
IP->Tel Operation	involves allow, barring some IP and IP Group send calls to certain numbers. It			
	includes: forbid call, call allowance, auto call, and password authentication.			
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to			
Index	31.			
	It specifies the source IP which will send the calls to gateway			
Source IP	Any: any IP address			
Source IP	IP: specific an IP address			
	IP Group: specific an IP group			
	All the caller number must match the source prefix. It specifies the source			
	prefix allow to send call out			
Source Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.			
	Oxxxx: consist of some digits such as 015,08,09			
	• 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186			
	All the called number must match the destination prefix, the call prefix			
	indicates the connected number			
Destination Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.			
	Oxxxx: consist of some digits such as 015,08,09			
	1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186			
	Its specifies number analysis rule			
	Forbid call			
Operation	Allow call			
	Auto call			
	Password authenticate			
Description	It describes the route for the ease of identification. Its value is character string			

Table 4-9-1 Description of IP->Tel Operation

Example:

Index 31: barring the certain calling number from IP 14<elastix>

Index	31		
Source Prefix	2877		
Source IP	© IP	14 <elastix></elastix>	
	C IP Group	18 <asterisk></asterisk>	
Destination Prefix	07		
Operation	Forbid Call		
	C Allow Call		
Description	restrict unicom		

It indicates that calling party from IP 14<elastix> matched prefix 2877, and also called party matched prefix 07 are not allowed call out. The calls match this rule will be rejected by gateway.

Index 29: definite a rule for IP 17<FreeSentral> that all the calls must go with valid password authentication.

Figure 4-9-3 IP->Tel Operation Modify

Index	29		
Source Prefix	any		
Source IP	€ IP	17 <freesentral></freesentral>	
	C IP Group	18 <asterisk></asterisk>	
Destination Prefix	any		
Operation	C Forbid Call		
	Allow Call		
	🗖 Auto Call 🗹	Password Authentication	
Authentication Passw	ord •••		
Description	password		

4.9.2 Tel->IP Operation

Figure 4-9-4 Tel->IP Operation

urce Port Sou	rce Prefix Destinati	on Prefix Opera	ation Description
			Description
	1/0page		

	It is an optional configuration item. Operation configuration essentially			
Tel->IP Operation	involves allow, barring some IP and IP Group send calls to certain numbers.			
	It includes: forbid call, call allowance, auto call, and password authentication.			
Index	It uniquely identifies a rule. Its value is assigned globally, ranging from 0 to			
macx	31.			
	It specifies the source IP which will send the calls to gateway			
Source IP	Any: any IP address			
	IP: specific an IP address			
	IP Group: specific an IP group			
	All the caller number must match the source prefix. It specifies the source			
	prefix allow to send call out			
Source Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.			
	Oxxxx: consist of some digits such as 015,08,09			
	• 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186			
	All the called number must match the destination prefix, the call prefix			
	indicates the connected number			
Destination Prefix	Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.			
	Oxxxx: consist of some digits such as 015,08,09			
	1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186			
	Its specifies number analysis rule			
	Forbid call			
Operation	Allow call			
	Auto call			
	Password authenticate			
	It describes the route for the ease of identification. Its value is character			
Description	string			
	-			

Table 4-9-2 Description of Tel->IP Operation

4.10 Port Group Configuration

4.10.1 Port Group

Figure 4-10-1 Port Group

0 all 0,1,2,3,4,5,6,7 Cyclic Ascen		Index	Description	Port	Select Mode
	[[[[]]]	0	all	0,1,2,3,4,5,6,7	Cyclic Ascending
al: 1entry 16entry/page 1/1page Page 1 💌					

Group Add			
Index	31		-
Description			
Select Mode	Ascending		-
Port	Port 0	Port 1	
	Port 2	Port 3	
	Port 4	Port 5	
	Port 6	Port 7	
	Port 8	Port 9	
	Port 10	Port 11	
	Port 12	Port 13	
	Port 14	Port 15	
	Port 16	Port 17	
	Port 18	Port 19	
	Port 20	Port 21	
	Port 22	Port 23	
	Port 24	Port 25	
	Port 26	Port 27	
	Port 28	Port 29	
	Port 30	Port 31	

Figure 4-10-2 Port Group Modify

Reset Cancel

OK

Table 4-10-1 Description port group

Index	Port group priority
Description	Port group decription
Select Mode	Choose the port that composition port group by drop-down list select mode
Port	The selected port

If you have the need for a group of port the same operation, then port group of configuration can help you improve efficiency.

4.11 IP Trunk Configuration

4.11.1 IP Trunk

IP Trunk								
	Index		IP		Port	Des	cription	KeepAlive Enable
Total: 0entry	16entry/page	1/0page 🔽						
			Add	Delete	Modify			

Figure 4-11-1 IP Trunk

Table 4-11-1 Description of IP Trunk

IP Trunk	Add remote IP of softswitch, SIP server which will send call traffics to gateway.			
Index	It uniquely identifies a trunk, ranging from 0 to 31.			
IP	It is an interworking parameter between the remote Softswitch and the SIP			
IF	server. It specifies the IP address of the peer equipment.			
Port	It is an interworking parameter between the remote Softswitch and the SIP			
FOIL	server. It specifies the SIP port number of the peer equipment			
Description	It describes the trunk for the ease of identification. Its value is character string			
KeepAlive Enable	It is use to detect connection between GSM gateway and remote IP trunk			

Example

To add a remote IP of Softswitch, set "index" to "31", SIP port number "5060"

Figure 4-11-2 IP Trunk Modify

lex	31	•
t		
scription		
epAlive Enable		

4.11.2 IP Trunk Group

Figure 4-11-3 IP Trunk Group

IP Group			
	Index	Description	IP
	18	asterisk	10,14,17,
	19	all	13,19,

Total: 2entry 16entry/page 1/1page Page 1 💌

Add	Delete	Modify

Table 4-11-2 Description of IP Trunk Group

IP Trunk Group	This configuration is optional, and is used to add the IP that have the same attributes to an IP group. The IP group will referenced by IP->Tel routing and number manipulation.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.
Description	It describes the route for the ease of identification. Its value is character string
IP	It specifies the IP will add to IP group

Example

To add an IP group, set index 10,14,17 to IP group 18

Figure 4-11-4 IP Trunk group modify

ndex	18	18			
escription	asteris	k			
Þ		Index	IP	Port	
	V	10	172.16.0.124	5060	
		13	172.16.3.55	5060	
		14	172.16.0.123	5060	
	\checkmark	17	172.16.1.123	5060	
		19	172.16.244.136	5060	
		31	110.164.212.105	5060	

4.12 System Configuration

4.12.1 Service Configuration

Service Configuration is used for configuring voice calls and some small businesses, such as Call

Progress Tone, codec, silence suppression, * service, the second dial and so on Figure 4-12-1 Service Configuration

Service Configuration	
Local Start RTP Port	8000
Eucal start RTP For	8000
Enable Slience Suppression	🖲 No 🖱 Yes
Call Progress Tone	USA
Preferred Coders(in listed order)	
1st	G.729AB 💌
2nd	PCMU 💌
3rd	PCMA 💌
4th	G.723.1 💌
Voice Frames per Tx	2
Do Not Answer GSM Imcoming Call for Hotline	🖱 No 🏽 Yes
Enable GSM Incoming Configuration	🕙 No 🕷 Yes
Auto Outgoing Routing Type	Polling
IP to GSM One Stage Dialing	🖱 No 🕷 Yes
Redirect Call When All Ports Busy	🕷 No 🕙 Yes
Play Voice Prompt for GSM Incoming Calls	🖱 No 📽 Yes
DTMF Parameter	
DTMF Method	SIGNAL
DTMF Volume	OdB 💌
DTMF Interval	200 ms
NAT Traversal	Disable
Other Configuration	
Enable Private Service	🖱 No 🕷 Yes
User ID Is Phone Number	🖲 No 😳 Yes
Only Accept Calls from SIP Server	🕷 No 😳 Yes
Allow Call from GSM to IP without Registration	🖱 No 🕷 Yes
Allow Call from IP to GSM without Registration	O No 🕷 Yes
Reject Anonymous Call from IP to GSM	® No [©] Yes
Use # as End Key	C No 🕷 Yes
No Answer Timeout	55 s
Interdigit Timeout	4 s
Call Delay	0 s

	Table 4-12-1 Description of Service Configuration
LOCAL Start RTP PORT	Means the initial port when RTP voice stream transmit in the IP network , in
	general, using the factory default values. When there are multiple DINSTAR
	series voice products, and the network gateway or router's NAT with
	loopholes user can try changing this item

	loopholes, user can try changing this item
Enable Silence	Enable the "silence suppression" almost no impact on call quality, and can save
Suppression	about half of the bandwidth.
Call Progress Tone	Each country has its different call progress tone required standards, such as
	busy tone, ring back tones and ring tone.
Preferred Coders	Means the code format when Voice transfer on IP network, support PCMA,
	PCMU, G.723.1 andG.729AB.
Enable PSTN Incoming	Means when call from PSTN side, you can dial the function keys for checking
Configuration	number, setting IP and so on
Enable Auto Outgoing	Means when call out , whether by ordinal or polling pick to Select a Channel,

Routing	this feature are generally used when use the same SIP User ID to register
IP to PSTN One Stage	The User ID will be sent directly to PSTN, for example: the user calls 6715, the
Dialing	device will sent 6715 User ID to PSTN
Play Voice Prompt for	Setting is yes, when through the PSTN calls to the Channel, the device will with
PSTN Incoming Calls	the clew tone, the default is "Please dial the extension User ID"; setting to No,
	the device will play dial tone
DTMF	DWG2001/DWG2004/DWG2000D-32G support RFC2833 and SIGNAL two
	ways. DTMF INTERVAL range is 50 ~ 800ms, DTMF VOLUME can use the
	default Configuration
Nat Traversal	Include Static NAT and STUN, NAT's UDP simple cross
STUN	STUN (Simple Traversal of UDP over NATs) is a network protocol. It is allowed
	to stay behind the NAT (or multiple NAT) client part to identify their clients'
	public address, found himself after what Type of NAT and NAT for a particular
	Channel is bound to a local Internet terminal Channel. This information is used
	for two host to set up UDP communication behind the same NAT router. The
	agreement defined by the RFC 3489
Allow call from IP to	Refer to "SIP Configuration" -> "Is register" . If "Is register" setting is no, this
PSTN without	option need set Yes ,to avoid that the devices can not call out
Registration	
Allow Call from PSTN to	Refer to "SIP Configuration" -> "Is register" . If "Is register" setting is no, this
IP without Registration	option need set Yes ,to avoid that the devices can not call in
Reject Anonymous call	The incoming anonymous calls will be rejected
from IP to PSTN	
Use # as End Key	In General, SIP phones are based on # as the end, if this option is set to No, the
	dial-up will end expires dial-up time
Interdigit Timeout	Bit of between the dialing time ,over the time will be seem as end of dial

4.12.2 SIP Configuration

0	_ 0 00ga.a
SIP Configuration	
SIP Proxy	
SIP Server Address	
SIP Server Port(default: 5060)	0
Check Net Status	🖲 No 🗇 Yes
Outbound Proxy	
Outbound Proxy Address	
Outbound Proxy Port	5060
Use Random Port	® No [©] Yes
Local SIP Port	5060
Is Register	In the second
DNS query type	A query 💌
DNS refresh interval (range:0 - 60,000min, 0 means disable)	0 min
T1	500 ms
T2	4000 ms
T4	5000 ms
TMAX	32000 ms
Keepalive Interval(range:1 - 3600s)	10 s
Enable 100rel	🖲 no 🗇 yes
From Mode when Caller ID Is Available	Tel/User
From Mode when Caller ID Is Unavailable	Anonymous
Answer Mode	Answered 💌
183 Mode	Immediately
Response Code switch	
Response code	Response code after switch

Figure 4-12-2 SIP Configuration

Table 4-12-2 SIP Configuration

SIP Server	Used for configure SIP server address and port, the address can be IP Address,
Address	also can be a domain nameWhich can be resolved by DNS server
SIP Proxy Port	Port default setting is 5060. For details, please consult the service provider
Outbound Proxy	Outbound proxy, it mainly used in firewall / NAT environment. That make the signaling and media streams are able to penetrate the firewall
Use Random Port	Set the local monitor SIP port (fixed or random) , random is every time you start the device will random Select a free SIP port For listening
Is Register	Default set yes, if you want the device can make a call without register, set No, Also enable the "Allow Call from IP to PSTN without Registration" and "Allow Call from PSTN to IP without Registration" function
Register Interval	Means how often the equipment will register to the SIP server/proxy
DNS query type	The DNS query type defines the type of information that will be requested from DNS server
DNS refresh	The interval of DNS refresh, Range from 0 to 60000 mins, 0 means disable default
interval	value is disable.

T1	Used to define the SIP protocol T1 timer value, default is 500ms
Т2	Used to defines the SIP protocol timer values, default value is 4000ms
Т3	Used to define the T2 timer value in SIP protocol, the default is 5000ms
Keep alive Interval	Used to keep communicate between equipment and the SIP server that make the
Reep alive Interval	device is available . In general, using the factory default values
	Used to config "From" Mode when Caller ID Is Available when call from GSM to VoIP
From Mode when Caller ID Is	Tel/User: From: caller number <sip:3001@ip>;tag=51088abb</sip:3001@ip>
Available	User/User: From: 3001 <sip:3001@ip>;tag=51088abb Tel/Tel: From: caller number <sip: @ip="" caller="" number="">;tag=51088abb</sip:></sip:3001@ip>
	User/Tel: From: 3001 <sip: @ip="" caller="" number="">;tag=51088abb</sip:>
From Mode when	Used to config "From" Mode when Caller ID Is Unavailable
Caller ID Is	Anonymous : From: <sip: @ip="" anonymous="">;tag=51088abb</sip:>
Unavailable	Username : From: <sip: @ip="" username="">;tag=51088abb</sip:>
Answer Mode	Answered: Gateway answer the IP incoming call (send SIP message "200 OK" to IP part) after GSM part answered Alerted: Gateway answer the IP incoming call after GSM part Alerted
183 Mode	Immediately: Gateway send "183 RING" immediately to IP part while it receive "INVITE" from IP part. Alerted: Gateway send "183 RING" after receive "ring back" from PSTN
Response Code	Used to configure the compatibility of SIP Response Code , Fill the response code
switch	in the front , and Fill the switch code in the behind

4.12.3 Port Parameter

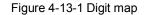
rt List								
Port	SIP User ID	Authenticate ID	Tx Gain	Rx Gain	To VOIP Hotline	To PSTN Hotline	Auto-Dial Delay Time(s)	Detail
0	1223		-2	2			3	Detail
ort Configu	ration							
All por	ts register use	ed same user II)	© N	o 🖲 Yes			
Curren	t Port			Port	0 🔻			
SIP Use	er ID			1223	}			
Authenti	cate ID							
Authenti	cate Password	l						
Tx Gain				-2dB	•			
Rx Gain				+2dE	3 💌			
To VOIP	Hotline							
To PST	N Hotline							
				Save Ba	ck			

Figure 4-12-3 Port Parameter

Table 4-12-3 Description of Port Configuration

Table 4-12-3 Description of Port Configuration				
Port Configuration	Used to configure ports' gain, Auto-Dial, etc.			
ALL ports register used				
same user ID	The default is not. If set to "yes" ,all the port will use user ID			
SIP User ID	It is the account used for registration, equipment port's unique identifier			
Authenticate ID	Used for authenticate			
Password	Its register Password			
1 4350014				
Tx Gain	Its DSP's Tx Gain. Adjusting it will effect volume on GSM side.			
Rx Gain	Its DSP's Tx Gain. Adjusting it will effect volume on IP side.			
	When PSTN part client calls to this port, gateway will auto forward to the			
To VoIP Hotline	hotline User ID. Leave it blank if you don't need this function. *Note: Please			
	config Tel->IP Operation if you need this function.			
	When VolP part client calls to this part. Gatoway will auto forward to the			
	When VoIP part client calls to this port, Gateway will auto forward to the			
To PSTN Hotline	number to PSTN part. Leave it blank if you don't need this function.			
	*Note: Please config IP->Tel Operation if you need this function.			

4.13 Digit Map



jit Map		
Digit Map	x.T x.#	

NOTE: Length of 'Digit Map' should be not more than 119 characters.

Save

Digit Map Syntax:

1. Supported objects

Digit: A digit from "0" to "9".

Timer: The symbol "T" matching a timer expiry.

DTMF: A digit, a timer, or one of the symbols "A", "B", "C", "D", "#", or "*".

2. Range []

One or more DTMF symbols enclosed between square brackets ("[" and "]"), but only one can be selected.

3. Range ()

One or more expressions enclosed between round brackets ("(" and ")"), but only one can be selected.

4. Separator

: Separated expressions or DTMF symbols.

5. Subrange

-: Two digits separated by hyphen ("-") which matches any digit between and including the two. The subrange construct can only be used inside a range construct, i.e., between "[" and "]".

6. Wildcard

x: matches any digit ("0" to "9").

7. Modifiers

.: Match 0 or more times.

- 8. Modifiers
 - +: Match 1 or more times.
- 9. Modifiers

?: Match 0 or 1 times.

Example:

Assume we have the following digit maps:

1. xxxxxxx | x11

and a current dial string of "41". Given the input "1" the current dial

string becomes "411". We have a partial match with "xxxxxx", but a complete match with "x11", and hence we send "411" to the Call Agent.

2. [2-8] xxxxxx | 13xxxxxxxx

Means that first is "2", "3", "4", "5", "6", "7" or "8", followed by 6 digits; or first is 13, followed by 9 digits.

- 3. (13 | 15 | 18)xxxxxxxx Means that first is "13", "15" or "18", followed by 8 digits.
- 4. [1-357-9]xx Means that first is "1","2","3" or "5" or "7","8","9", followed by 2 digits.

4.14 Tools

4.14.1 Firmware Upload

Figure 4-14-1 Firmware upload

Firmware Upload	
Send "Idf" file fi	rom your computer to the device.
Software	Browse Upload
Web	Browse Upload
Dsp Firmware	Browse Upload

NOTE: 1. After uploading, please restart the device to take effect. 2. Please wait 60 seconds after Dsp Firmware upload is successful.

Select the software, Web or DSP firmware program under correct directory services, and then click upload will complete upgrade the firmware. During the upgrade process, please do not swtich off the power supply, equipment may paralyze.

4.14.2 Management Parameter

Figure 4-14-2 Management Parameter

Malaa Baarat Laansaa	To all a l
Voice Prompt Language	English 💌
Syslog Parameter	
Syslog Enable	🔘 Yes 🖲 No
NTP Parameter	
NTP Enable	• Yes O No
Primary NTP Server Address	us.pool.ntp.org
Primary NTP Server Port	123
Secondary NTP Server Address	18.145.0.30
Secondary NTP Server Port	123
Check Interval	3600 s
Time Zone	GMT-6:00 (US Central Time, Chicago)

Voice Prompt Language	Select the language of voice prompt. There are two kind of voice : English and Chinese
Syslog Paremeter	Syslog is a standard for network device data logging. It allows separation
	of the software that generates messages from the system that stores them
	and the software that reports and analyzes them. It also provides devices
	which would otherwise be unable to communicate a means to notify
	administrators of problems or performance. There are 5 grades of syslog,
	Including NONE, DEBUG, NOTICE, WARNING and ERROR.
NTP Parameter	The Network Time Protocol (NTP) is a protocol and software
	implementation for synchronizing the clocks of computer systems over
	packet-switched, variable-latency data networks.
	User need to fill the NTP Server Address and select Time Zone

Table 4-14-1 Management Parameter

4.14.3 Config Backup

Figure 4-14-3 Config backup

I	Data Backup			
	Click 'Backup' for download configuration file to your computer. Backup			
Click 'Backup' for download configuration file to your computer.				
4.1	4.14.4 Data Restore			

Figure 4-14-4 Data restore

Data Restore	
Send data file	from your computer to the device.
Configuration	Browse Restore

Send data file from your computer to the device

4.14.5 IVR Voice Prompt Upload

By default, when PSTN call incoming, the system will play the default IVR, and also the user can load custom IVR.

Figure 4-14-5 IVR Voice Prompt Upload

IVR Voice Prompt Upload				
Send "wav" file from your computer to the device.				
IVR Voice Prompt File for PSTN Incoming Calls	Choose File No file chosen	Upload		
Play IVR Voice Prompt from	Oefault Custom	Save		

NOTE: 1. "wav" file should be not more than 360k bytes. 2. It must restart the device to take effect.

NOTE: the customize voice files can be recorded using Windows recording programs, the sound format is 8000Hz, 16 bit sampling in mono, with WAV format, size of files can not exceed 190KB.

4.14.6 PING test

Ping is utility used to test the reachability of a host on an Internet Protocol (IP) network and to measure the round-trip time for messages sent from the originating host to a destination host.

Figure 4-14-6 Ping Test

Ping Test	
Ping Destination	172.16.1.1
Number of Ping(1-100)	4
Ping Packet Size(56-1024 bytes)	56
	Start Stop

inging 172.16.1.1 with 56 bytes of data:	
eply seq=0 from 172.16.1.1: bytes=56 time=20ms TTL=6	ł
eply seq=1 from 172.16.1.1: bytes=56 time<1ms TTL=64	
eply seq=2 from 172.16.1.1: bytes=56 time=10ms TTL=6	ł
eply seq=3 from 172.16.1.1: bytes=56 time=10ms TTL=6	Ł
Ping statistics for 172.16.1.1	
ackets: Sent = 4, Received = 4, Lost = 0 (0% loss)	
TT Minimum = 1ms, Maximum = 10ms, Average = 10ms	

4.14.7 Tracert Test

Trace route is a computer network diagnostic tool for displaying the route (path) and measuring transit delays of packets across an Internet Protocol (IP) network.

Figure 4-14-7 Tracert Test

Tracert Destination		www.google.com.hk	
Мах	Hops of Trac	cert(1-255) 30	
		Start Stop	
rmatio	on		
	-	to www.google.com.hk[74.125.71.99] over a maximum of 30	
hops			
1	1 ms *	172.16.1.1	
2 3	*	Request timed out.	
-		Request timed out. 121.15.179.86	
5		119.145.47.46	
6		202.97.35.250	
7		202.97.60.142	
8		202.97.60.22	
9		202.97.61.102	
-		202.97.62.214	
		209.85.241.58	
		209.85.253.69	
		216.239.48.230	

4.14.8 Login Password

Figure 4-14-8 IVR Voice Prompt Upload

Username & Password	
Web Configuration	
Old Web Username	admin
Old Web Password	
New Web Username	
New Web Password	
Confirm Web Password	
Telnet Configuration	
Old Telnet Username	admin
Old Telnet Password	
New Telnet Username	
New Telnet Password	
Confirm Telnet Password	
Sa	ve

When using web or telnet Configuration, please enter default user name and password. User can modify the login name and password.

4.14.9 Factory Reset

Figure 4-14-9 Factory Reset

Factory Reset	
	Click this button to reset factory default settings
-	Apply

Be careful do this operation, after restore factory setting, all the parameters will be changed to the factory default.

4.14.10 Restart

Figure 4-14-10 Restart

Restart		
	Click this button to restart the device.	
	Restart	

5. FAQ

5.1 Device has been connected to network physically, but cannot access the gateway

1) Make sure the network cable is ok , can through view the device network port indicator light to determine the physical connection is working or not;

2) Make sure the connected network devices (router, switch or hub) support 10M/100M adaptive, if not, connect the Equipment directly to PC, landing WEB and in the "local connection" Configuration interface Select the correct Ethernet Work Mode;

3) Check the Network Configuration, if the Configuration is incorrect, please re-Configuration. If you are using DHCP mode, check DHCP Server is working properly;

4) Check whether there is a LAN device conflict with the exists IP address.

5.2 Equipment can not register

If the Run LED does not flash mean unregistered

1) Check the network connection is working (see above section), whether the Configuration is correct;

2) Check whether the LAN firewall setting is inappropriate (such whether limit the network communication); If it is, there are two ways to try to resolve;

3) Check whether the Local Network to the SIP PROXY platform network environment is relatively poor or not, and if so, please check Local Network or contact the service provider;

4) if go through those steps, the device still be in trouble, please contact the equipment provider;

5.3 When calling out, the callee's phone shows wrong caller ID:

1) Ask the callee checks whether the device is failure or device battery power is low

2) Make sure the callee has been subscribed called User ID display service

3) If only part of the caller User ID with this problem, please contact the telecom carrier.

5.4 sudden interruption during a call

1) make sure whether is human error caused the problem

2) Check the balance.

3) Make sure whether the LAN equipment such as gateway or router fails, user can try to restart the gateway or router

5.5 voice single-pass, double-barrier or poor quality

1) Make sure the equipment is working properly with grounded power

2) Check the device network connection is in working status

3) Ask network administrators to open limitation with the equipment's network communications (it is a special equipment, not afraid of virus attacks); (2) try to enable the equipment tunnel (through the WEB for Configuration, Also, please NOTE, open the tunnel will impact voice quality, Please do not enable the tunnel as far as possible, refer WEB Configuration Interface Description section)

4) Make sure the LAN equipment is working, user can try to restart the gateway or router to solve the problem

5) Check whether there is more than one DINSTAR series products in LAN network: some gateways or routers, processing network packet is vulnerable (for example, to multiple network devices or the same protocol network communication, NAT allocated the same conversion communications Channel). If there is such a case, suggest replacing a router or specify each voice gateway with different LOCAL RTP PORT Channel (refer to the base WEB Configuration interface section)

6) Check the equipment network environment for the softswitch platform, monitor the network condiation, make sure the network is solid

6. Glossary

GSM: Global System for Mobile Communications

- CDMA: Code Division Multiple Access
- FMC: Fixed Mobile Convergence
- SIP: Session Initiation Protocol
- MGCP: Media Gateway Control Protocol
- DTMF: Dual Tone Multi Frequency
- USSD: Unstructured Supplementary Service Data
- PSTN: Public Switched Telephone Network
- STUN: Simple Traversal of UDP over NAT
- IVR: Interactive Voice Response
- IMSI: International Mobile Subscriber Identification Number
- IMEI: International Mobile Equipment Identity
- DMZ: Demilitarized Zone