

DWG2000B-16G User Manual V2.0



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Table of Contents

1. Equipment Introduction	4
1.1 Introduction	4
1.2 Scenario of Applications of Products	4
1.3 Product Appearance	5
1.4 Functions and Features	6
1.4.1 Protocol Standard Supported	6
1.4.2 System Function	6
1.4.3 Industrial Standards Supported	7
1.4.4 General Hardware Specification	7
2. Equipment Quickly Installation	8
2.1 Installation Notice	8
2.2 Installation Procedure	8
2.2.1 Install SIM Card	8
2.2.2 Antenna Installation	10
2.2.3 Network Cable Connection of Equipment	10
2.2.4 Power Cable Connection of Equipment	10
3. Network Configuration	11
3.1 Attentions	11
3.2 General Feature Codes for System Setting	11
3.3 Static IP Configuration	12
3.4 DHCP Configuration	12
4. WEB configuration	13
4.1 Access the System Through HTTP	13
4.2 WEB Configuration	14
4.3 System Information	14
4.3.1 System Information	14
4.3.2 Mobile Information	15
4.3.3 SIP Information	16
4.4 Statistics	17
4.4.1 TCP/UDP	17
4.4.2 RTP	17
4.4.3 SIP Call History	
4.4.4 IP to GSM Call History	19
4.5 Network Configuration	19
4.5.1 Local Network	19
4.5.2 VLAN Parameter	21
4.5.3 ARP	22
4.6 Mobile Configuration	23
4.6.1 Basic Configuration	23
4.6.2 Mobile Configuration	24

Figu	re 4-6-2 Mobile State	24
4.6.4	4 PIN Management	
4.6.	5 SMSC	
4.6.	6 SMS	
4.6.	7 USSD	
4.6.3	8 Carrier	
4.6.	9 BCCH	
4.7 Routi	ing Configuration	
4.7.	1 Routing Parameter	
4.7.2	2 IP->Tel Routing	
4.7.	3 Tel->IP Routing	
4.8 Mani	pulaton Configuration	
4.8.	1 IP->Tel Destination Numbers	
4.8.2	2 Tel->IP Source Numbers	
4.8.	3 Tel->IP Destination Numbers	
4.9 Operation	ation	
4.9.	1 IP->Tel Operation	
4.9.2	2 Tel->IP Operation	
4.10 Port	Group Configuration	
4.10).1 Port Group	42
4.11 IP T	runk Configuration	43
4.11	.1 IP Trunk	
4.11	.2 IP Trunk Group	44
4.12 Syst	tem Configuration	45
4.12	2.1 Service Configuration	
4.12	2.2 SIP Configuration	47
4.12	2.3 Port Configuration	
4.13 Digi	it Map	51
4.14 Tool	ls	53
4.14	1 Firmware Upload	53
4.14	.2 Management Parameter	53
4.14	.3 Config Backup	54
4.14	.4 Config Restore	55
4.14	1.5 IVR Voice Prompt Upload	
4.14	.6 Ping Test	56
4.14	.7 Tracert Test	56
4.14	.8 Username & Password	57
4.14	9 Factory Reset	57
4.14	.10 Restart	57
5. FAQ		
6. Glossary		60

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1. Equipment Introduction

This chapter mainly introduces functions and structures of DWG2000B-16G.

1.1 Introduction

DWG2000B-16G 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, SOHO, residential users for cost-effective solution.

1.2 Scenario of Applications of Products

DWG2000B-16G provides access of GSM network.

With the development of users and telecom service, mobile network and fixed network integration will be steadily increasing. DWG2000B-16G provides high quality VoIP service which perfectly meets the requirement. This is a scenario shown as figure 1-2-1

Figure 1-2-1 Network scenario



1.3 Product Appearance

The appearance of DWG2000B-16G shows as follow

Figure 1-3-1 Front view of DWG2000B-16G

Table 1-3-1 Description of Front view

Ind	Sign	Description
1	LAN	Ethernet Interface, 10/100M Base-TX, RJ-45
2	CONSOLE	Serial port is a serial communication physical interface through which information transfers in or out one bit at a time, DB-9 connector
3	RST	If keep press for 3 seconds, RUN lamp keeps light, means restore IP and password; If keep press for 8 seconds, means restore factory.
4	RUN	Indicate the status of the device.
5	PWR	Indicate the status of the power connection
6	ANT	Standard antenna interface
7	ANT	Indicate the status of the SIM card register
8	ANT	An antenna (or aerial) is an electrical device which converts electric currents into radio waves, and vice versa

Figure 1-3-2 Rear view of DWG2000B-16G



Table 1-3-2 Description of Rear view

Ind	Sign	Description
1	Power Switch	Power Switch of the device
2	AC Power Input	110~240VAC,50~60Hz, 1.2A

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;
- PPTP

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:AC100~240V 50/60HZ DC12V/1A
- Temperature: 0~40 °C (Operation), -20~80 °C (storage)
- Humidity: 5% ~90% RH,
- Power Consumption: 35W
- Dimensions: 440(W) x 307(D) x 44(H) mm
- Net weight: 4.2kg

2. Equipment Quickly Installation

This chapter mainly introduces DWG2000B-16G hardware installation and connection of equipment.

2.1 Installation Notice

DWG2000B-16G uses AC power. Power supply should ensure the reliability and stability, otherwise, it may damage the SIM card or device. In addition, make sure the power supply connects to ground bar well. With right ground protect connection, that can reduce the surge voltage caused by lightning that damage the equipment, and ensure voice quality (note: when calls with irregular noise occurring, please check the power whether connect ground well). Common measures are as follows:

Making sure that all devices powered in the buildings are in accordance with NEC (National Electric Code, National Electrical Regulations) Article 250 of manual properly grounded;

Making sure that the panel of building power supply units used high-quality copper wire well connect with the ground wire, copper wire specifications shall comply with NEC Table 250-94/95 relevant provisions of the manual. Grounding cable that buried in the building field, including at least one or several 2.44m deep under the ground, or buried deeply underground at least 0.76m, with a wire around the building (see NEC manual specifications the relevant provisions of the table 250-94/95);

Setting up voltage protector between equipment and ground connected to some other computer equipments (either directly or through other devices), such as terminal or printer must also be plugged into the same surge protector.

Network interface of DWG2000B-16G supports RJ45 standard with 10Mbps or 100Mbps network.

Wireless section, inserting SIM card directly, GSM channel should work properly.

2.2 Installation Procedure

2.2.1 Install SIM Card

When installing SIM card, opening blank panel of SIM slot, procedure shows as below:

- Pull out the GSM user board
- Inset the SIM card to the SIM slot

• Push in the GSM user board



Figure 2-2-1 Pull out the GSM user board

Figure 2-2-2 I Inset the SIM card to the SIM



Figure 2-2-3 Push in the GSM user board



2.2.2 Antenna Installation



2.2.3 Network Cable Connection of Equipment



Lanswitch

2.2.4 Power Cable Connection of Equipment

Figure 2-2-6 DWG2000B-16G connection



3. Network Configuration

In this chapter we will introduce the initial configuration of DWG2000B-16G. 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 check redo that step again.

3.2 General Feature Codes for System Setting

	• •
Dial numbers	Features
*114#	Play the phone NO.
*115#	Check the TT number of gateway (using just when the device interconnects
*150*a#	Set IP address(static/DHCP), a can be digit 1 or 2,*150*1# is static IP
*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
*158#	Report the IP address
*157	Setting the work mode (route or bridge), * 157 * 0 # is route mode, * 157 *
*195#	Play record
*198#	Clear record
*199#	Setting Record. dial*199# start record(≤ 20 s), then press # end the
*111#	Restart device

3.3 Static IP Configuration

This chapter introduces IP configuration of DWG2000B-16G through calling IVR.

Assuming the IP address of a DWG2000B-16G device is 192.168.1 200, subnet mask is 255.255.255.0, IP of gateway is 192.168.1.1, configured as follows:

- 1) Please make sure hardware installation have finished
- Dial the phone number of the SIM card. Dail "*150*1#" after hearing "please dail extension number ". Hang up after hearing "setting successful"
- Dial the phone number of the SIM card. Dail "* 152 * 192 * 168 * 1 * 200 #"after hearing "please dail extension number". Hang up after hearing "setting successful"
- Dial the phone number of the SIM card. Dail "*153*255*255*255*0#" after hearing "please dail extension number ". Hang up after hearing "setting successful"
- 5) Dial the phone number of the SIM card. Dail "*156*192*168*1*1#" after hearing "please dail extension number ". Hang up after hearing "setting successful"
- 6) Dial the phone number of the SIM card. Dail "*111#" after hearing "please dail extension number ", that will restart the device
- 7) Dial the phone number of the SIM card. Dail "*158#" after hearing "please dail extension number ". It will play IVR about the IP of the device

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. Dail "*150*2#" after hearing "please dail extension number ". That means the DHCP is confirued 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, 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 DWG2000B-16G.

4.1 Access the System Through HTTP

Enter IP address of DWG2000B-16G in browser. The default IP of LAN port is 192.168.11.1. and the GUI shows as below:

Connect to 172.16.3	30.30 ? ×
	G
Web Config System	
<u>U</u> ser name:	🖸 admin 💌
Password:	•••••
	Remember my password
	OK Cancel

Figure 4-1-1 WEB log interface

Enter username and password and then click "OK" in configuration interface. The default username and password are "admin/admin". It is strongly recommended, change the default password to a new password for system security.

4.2 WEB Configuration

DWG2000B-16G WEB configuration interface consists of the navigation tree and the detail configuration interfaces.

	_													
				Run Information										
 System Statistic 	Informa cs	tion		MAC Address	5	00-01-02-03-	04-05							
+ Networ	k Config	uration		Network Mod	le	Bridge								
+ Mobile	Configu	ration		Network		172.16.12.16			255.25	5.0.0			Static	
+ Routing	J Configu	iration		DNS Server		255.255.255.	255							
+ Operati		onngurauon		System Up F	Juration	00h:05m:16s								
+ Port Gr	oup Con	figuration		Network Traf	fic Statistics	Received 673	3502 Bytes		Sent 19	99065 Byt	es			
+ IP Trun	k Config	uration												
+ System	Configu	ration		Version Infor	mation	Device Model	l i		DWG2	000B				
• Digit M	ар					Software Vers	sion		2.22.01	1.03 Built	on May 16 2	2012, 15:58	05	
+ Tools						Web Version	raian		2.22.01	1.03				
						Lonic Version	151011		10602	1				
						DSP Version			v7_22	03_16_H	W_12			
											-			
Mobi	le Info	rmation												
	Port	Туре	IMSI	Status	Remaining Call Duration	Carrier		Signal Quality	BER	ASR(%)ACD	(s)PDD(⁵⁾ Status	
	0	FAULT			Daration			Taul	0	0	0	0	Idle	
	1	FAULT						Ť.	0	0	0	0	Idle	
	2	FAULT						Ť	0	0	0	0	Idle	
	3	FAULT						Ťall	0	0	0	0	Idle	
	4	FAULT						Tatt	0	0	0	0	Idle	
	5	FAULT						Tatt	0	0	0	0	Idle	
	6	FAULT						Tatt	0	0	0	0	Idle	
	7	FAULT						Tatt	0	0	0	0	Idle	
	8	FAULT						Tatt	0	0	0	0	Idle	
	9	FAULT						Tatt	0	0	0	0	Idle	
	10	FAULT						Tatt	0	0	0	0	Idle	
	11	FAULT						Tatt	0	0	0	0	Idle	
	12	FAULT						Tatt	0	0	0	0	Idle	
	13	FAULT						Tatt	0	0	0	0	Idle	
	14	FAULT						Tatt	0	0	0	0	Idle	
	15	FAULT						Tattl	0	0	0	0	Idle	
cin i														
SIP I	mornin	atton												
	Port	SIP Use	rID	Register Status	Status	Port S	SIP User II	D	F	Registe	er Status	s St	atus	
	0			Unregistered	onhook	1			U	Unregis	stered	on	hook	
	2			Unregistered	onhook	3			l	Jnregis	stered	on	hook	
	4			Unregistered	onhook	5			U	Jnregis	stered	on	hook	
	6			Unregistered	onhook	7			l	Jnregis	stered	on	hook	
	8			Unregistered	onhook	9			l	Jnregis	stered	on	hook	
	10			Unregistered	onhook	11			l	Jnregis	stered	on	hook	
	12			Unregistered	onhook	13			l	Jnregis	stered	on	hook	
	14			Unregistered	onhook	15			U	Jnregis	stered	on	hook	

Figure 4-2-1 WEB introduce

Refresh

Go through navigation tree, user can check, view modify, and set the device configuration on the right of configuration interface.

4.3 System Information

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

4.3.1 System Information

-

Run Information			
MAC Address	00-01-02-03-04-05		
Network Mode	Bridge		
Network	172.16.12.16	255.255.0.0	Static
DNS Server	255.255.255.255		
System Up Duration	00h:05m:16s		
Network Traffic Statistics	Received 673502 Bytes	Sent 199065 Bytes	
Version Information	Device Model	DWG2000B	
	Software Version	2.22.01.03 Built on May 16	2012, 15:58:05
	Web Version	2.22.01.03	
	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

MAC Address	Displays the current MAC of the gateway, for example: 00-1F-D6-1B-3D-02
Network Mode	DWG2000B-16G 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 Time	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 info	shows the current firmware version

4.3.2 Mobile Information

Figure 4-3-2 Mobile information

Mobile Information											
Port	Туре	IMSI	Status	Remaining Call Duration	Carrier	Signal Quality	BER	ASR	(%)ACD	(s)PDE	O(s) ^{Call} Status
0	FAULT					Tatt	0	0	0	0	Idle
1	FAULT					Taill	0	0	0	0	Idle
2	FAULT					Taill	0	0	0	0	Idle
3	FAULT					Taill	0	0	0	0	Idle
4	FAULT					Taill	0	0	0	0	Idle
5	FAULT					Tattl	0	0	0	0	Idle
6	FAULT					Tattl	0	0	0	0	Idle
7	FAULT					Taill	0	0	0	0	Idle
8	FAULT					Tattl	0	0	0	0	Idle
9	FAULT					Tatt	0	0	0	0	Idle
10	FAULT					Tattl	0	0	0	0	Idle
11	FAULT					Tattl	0	0	0	0	Idle
12	FAULT					Tatt	0	0	0	0	Idle
13	FAULT					Tattl	0	0	0	0	Idle
14	FAULT					Taill	0	0	0	0	Idle
15	FAULT					Tattl	0	0	0	0	Idle

|--|

Port	Number of GSM/CDMA ports .
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

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Status	Indicates the connection status of current GSM / CDMA module
Remaining	Limite a call duration to the SIM card, when call duration is out of that duration,
Call Duration	the call would be discontinued. 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 / CDMA.
ASR	Answer Seizure Ratio is a measure of network quality. Its calculated by taking
	the number of sucessfully 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 behaviour.
ACD	The Average Call Duration (ACD) is calculated by taking the sum of billable
	seconds (billsec) 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 port, include idle and active
	"Idle" means there is no call on this channel
	"Active" means the call is

4.3.3 SIP Information

Figure 4-3-3 SIP information

SIP Information									
Port	SIP User ID	Register Status	Status	Port SIP User ID	Register Status	Status			
0		Unregistered	onhook	1	Unregistered	onhook			
2		Unregistered	onhook	3	Unregistered	onhook			
4		Unregistered	onhook	5	Unregistered	onhook			
6		Unregistered	onhook	7	Unregistered	onhook			
8		Unregistered	onhook	9	Unregistered	onhook			
10		Unregistered	onhook	11	Unregistered	onhook			
12		Unregistered	onhook	13	Unregistered	onhook			
14		Unregistered	onhook	15	Unregistered	onhook			

Refresh

Displays registration status information with Softswitch platform or SIP Server

	Tuble + 5 5 Description of Sir Information
Port	The number of GSM channel, DWG2000B-16G has 16 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"

Table 4-3-3 Description of SIP information

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
	Ref	resh	

4.4.2 RTP

Figure 4-4-2 RTP

RTP										
Port	Payload Type	Packet Period	Local Port	Peer IP	Peer Port	Send Packet	Recv Packet	Loss Packet	Jitter	Duration Time(s)
0	G.723.1	30	8000	172.30.50.177	12646	332	0	1	0	13
1	G.723.1	30	8004	172.30.50.177	12642	332	0	1	0	13
2	PCMU	10	8008	172.30.50.177	12656	999	0	1	0	13
3	G.729AB	20	8012	172.30.50.177	12652	499	0	1	0	13
4	PCMU	10	8016	172.30.50.177	12638	998	0	1	0	13
5	G.723.1	60	8020	172.30.50.177	12716	166	0	1	0	13
6	G.729AB	40	8024	172.30.50.177	12644	249	0	1	0	13
7	G.729AB	20	8028	172.30.50.177	12762	500	0	1	0	13
8	G.723.1	30	8032	172.30.50.177	12664	332	0	1	0	13
9	PCMU	20	8036	172.30.50.177	12660	499	0	1	0	13
11	G.729AB	20	8044	172.30.50.177	12622	499	0	1	0	13
12	PCMU	10	8048	172.30.50.177	12648	998	0	1	0	13
13	PCMA	30	8052	172.30.50.177	12610	332	0	1	0	13
14	G.723.1	30	8056	172.30.50.177	12680	332	0	1	0	13
15	G.729AB	20	8060	172.30.50.177	12662	499	0	1	0	13

Refresh

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

SIP Call Histor	y							
Port	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
1	0	0	0	0	0	0	0	0
2	0	0	0	0	0	0	0	0
3	0	0	0	0	0	0	0	0
4	0	0	0	0	0	0	0	0
5	0	0	0	0	0	0	0	0
6	0	0	0	0	0	0	0	0
7	0	0	0	0	0	0	0	0
8	0	0	0	0	0	0	0	0
9	0	0	0	0	0	0	0	0
10	0	0	0	0	0	0	0	0
11	0	0	0	0	0	0	0	0
12	0	0	0	0	0	0	0	0
13	0	0	0	0	0	0	0	0
14	0	0	0	0	0	0	0	0
15	0	0	0	0	0	0	0	0

Figure 4-4-3 SIP Call History

Refresh

Table 4-4-2 SIP Call History

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

4.4.4 IP to GSM Call History

IP to GSM Call History												
			Call Failed Caused by SIP			Call Failed Caused by GSM						
Port	Call	Duration	Answered	Canceled	Timeout	Not Allowed	Negotiatio n failed	Busy	NO ANSWER	NO DIALTONE	NO CARRIER	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

Figure 4-4-4 IP to GSM Call History

Refresh Clear

Port	Device GSM port				
Call	Statistics the number of calls in this port				
Duration	Statistics call total time				
Answered	Statistics response times				
Call Failed Caused by SIP	Statistics cause of call failure from SIP, include:canceled/ timeour				
	not allowed/ Negotiation failed				
Call Failed Caused by GSM	Statistics cause of call failure from GSM, include: Busy/ no				
	answer/ no dialtone/ no carrier				

4.5 Network Configuration

4.5.1 Local Network

Figure 4-5-1 Local Network

Local Network	
Network Configuration	
Obtain IP address automatically	
Ose the following IP address	
IP Address	172.16.12.16
Subnet Mask	255.255.0.0
Default Gateway	172.16.1.5
O PPPoE	
Account	
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	

Note: It must restart the device to take effect.

Save

Obtain IP Address	Enable the device obtain IP Address automatically or not. Default is		
Automatically	enabling		
Use the Following IP	Configure the "IP Address", "Subnet Mask" and "Default Gateway"		
Address	by manual		
DDDoE	Need ISP offer the account and password. Use this mode when there		
PPPOE	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

Figure 4-5-2 VLAN Para	ineter
/LAN Parameter	
Data VLAN	Enable
Data 802.1Q VLAN ID (0 - 4095)	0
Data 802.1p Priority (0 - 7)	0
Data VLAN use the default WAN interface in this case.	
Voice VLAN	Enable
Voice 802.1Q VLAN ID (0 - 4095)	0
Voice 802.1p Priority (0 - 7)	0
Voice VLAN use following separate IP interface	
Obtain IP address automatically	
Use the following IP address	
IP Address	
Subnet Mask	
Default Gateway	
Voice VLAN DNS Server	
Obtain DNS server address automatically	
Use the following DNS server addresses	
Primary DNS Server	
Secondary DNS Server	
Management VLAN	Enable
Management 802.1Q VLAN ID (0 - 4095)	0
Management 802.1p Priority (0 - 7)	0
Management VLAN use following separate IP interface	
Obtain IP address automatically	
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

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

Table 4-5-2 Description	of VLAN Parameter
-------------------------	-------------------

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 4-5-3 Add ARP

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

4.6 Mobile Configuration

4.6.1 Basic Configuration

Basic Configuration	
Dial Tone Gain (Mobile Side)	dB
Select Band	Default(Automatic)
Forward Enable	🔘 No 🔍 Yes
Forward Master Mobile	Port 0 💌
Remote API Enable	🔘 No 🔍 Yes
API Server Address	172.16.12.18
API Server Port	13000
API User ID	admin
API User Password	Show Password
Auto Reset Module	No O Yes

Figure 4-6-1Basic 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. Usually adopt the default configuration.
Select Band	Acording to carrier's band standards. Standards are as belows:
	USM. 850/900/1800/1900 MHZ, CDMA. 800 MHZ
Forward Enable	When port occupied whether allow call forwarding
Forward Master	Choose the port allowed call transfer.
Mobile	
Remote API Enable	API is provided for third party development with DLL and IAD
	components. The API includes SMS sending and receiving, USSD sending
	and receiving. The default is "No".
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 remote channel No. who uses API. This is an option when
	selecting res under remote API enable
Auto reset module	Used to reset GSM/CDMA module when SIM card can not register

Table 4-6-1 Description of Basic Configuration

4.6.2 Mobile Configuration

Mobile State						
Port	Single Call Limitation	Call Limitation	Tx Gain	Rx Gain	Reset Module	Detail
0	No	No	0	0	Reset Module	Detail
1	No	No	0	0	Reset Module	Detail
2	No	No	0	0	Reset Module	Detail
3	No	No	0	0	Reset Module	Detail
4	No	No	0	0	Reset Module	Detail
5	No	No	0	0	Reset Module	Detail
6	No	No	0	0	Reset Module	Detail
7	No	No	0	0	Reset Module	Detail
8	No	No	0	0	Reset Module	Detail
9	No	No	0	0	Reset Module	Detail
10	No	No	0	0	Reset Module	Detail
11	No	No	0	0	Reset Module	Detail
12	No	No	0	0	Reset Module	Detail
13	No	No	0	0	Reset Module	Detail
14	No	No	0	0	Reset Module	Detail
15	No	No	0	0	Reset Module	Detail

Figure 4-6-2 Mobile State

Figure 4-6-3 Mobile Configuration

bile Configuration		
Select Port	Port 0 💌	
Mobile Number		
Step	sec	
Enable Call Duration Limitation of single call	🔘 No 🖲 Yes	
Time of single call		
Enable Call Duration Limitation	🔘 No 🖲 Yes	
Auto Reset	🔘 No 🔘 Yes	
Maximum Call Duration		
Minimum Charging Time		sec
Alarm Threshold (via SMS)		
Mobile Number (Receiving Alarm)		
Port Description for Alarm		
Remain Time		
Restore Time		
CLIR	🔘 No 🖲 Yes	
Mobile Tx Gain		dB
Mobile Rx Gain		dB
Reset Module		

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.

Mobile Number	SIM card number of the channel. That must be configured when
	"Forward" funciton enable.
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.
Enable Call Duration	Definite maximum call duration for single call. Example: if Time of
Limitation single call	single call set to 10, the call will be disconnected after talking 10*step
	seconds
Time of single time	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 max call duration of channel. The max call duration is between 1 to 65535 minutes.
Limitation	
Auto reset	Set a time make device reboot
Maximum Call Duration	Defines a value by users. That will limit the SIM/UIM card's total call
	duration. After the call duration excesses this value, no call will be
	made from this channel. The value range is 1-65535. If user doesn't
	configure this value, Default is no max call duration limits for this
	channel.
Minimum Charging Time	A minimum charging time (in seconds) is defined during which no
	charging is done at carrier side. If the conversation time is even
	shorter, the total call duration will not decrease.
Alarm Threshold (via	When the SIM remain time is or less than this value, DWG will send
SMS)	the alarm SMS to remind the users of the SIM remain time.
Mobile Number	The mobile phone No. which used to receive the alarm SMS. Users
(Receiving Alarm)	can get SMS report of SIM/UIM card status(SIM Remain Time) in
	DWG.
Port Description for	It is the identification mark of SIM/UIM card in the SMS report. The
Alarm	mobile phone No. of the SIM/UIM card is recommended to use as the
	port description for alarm, or any other string.

Table 4-6-2 Description of Mobile Configuration

Remain Time	Indicates the current sim remain time. It can't modified
Restore time Recovers the SIM remain time to initial value, the Maxin	
	Duration.
CLIR	Caller ID display restrict. This function is used to restrict the mobile
	phone No. By adding "#31#" before the mobile phone ID, this
	funciton should be supported by carrier.
Mobile Tx Gain	Transits gain of the mobile module, from IP side to PSTN side.
Mobile Rx Gain	Receives gain of the mobile module, from PSTN side to IP side.
Detect Reverse Polarity	This option for CMDA Reverse Polarity detection. Most CDMA
	operators don't offer polarity reverse . So VoIP to mobile,
	DWG2000B-16G will connect soon. It doesn't wait mobile side
	answer.

4.6.4 PIN Management

Figure 4-6-4 PIN Management

PIN Management				
Select Port	Port 0			
Old PIN Code	•••••			
New PIN Code	•••••			
Confirm New PIN Code	•••••			

NOTE: PIN code can be modify, only on state that SIM card is locked.

Save

Detailed description as below:

Table 4-6-3	Description	of PIN	Management
1000 - 0	Description	1011111	wianagemen

PIN	Personal identification number of SIM card. In the status of SIM card	
	locked, PIN can be modified to prevent SIM card from being stolen.	
Select Port	Selects the GSM/CDMA channel No.	
Old PIN code	The previous PIN code	
New PIN code	Inputs a new PIN code	

4.6.5 SMSC

Figure 4-6-5 SMSC

SMSC			
Select Port	Port 0 💌		
SMSC	+8613800755500		
	Save		

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.

4.6.6 SMS

Figure 4-6-6 SMS

Send Message	
Select Port Encoding	Random Port
To Message	
	Å

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

Send

Table 4-6-5 Description of SMS sending

Select Port	Users can select a defined channel or random channel to send SMS. Input	
	the recevier's mobile phone No 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 send Unicode characters.	
То	Mobile phone No. of the receiver	
Message	Content of the SMS. The length is limited to 300 characters.	

4.6.7 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				
	Port	USSD Request	USSD Reply	
	0		not registered	
	1		not registered	
	2		not registered	
	3		not registered	
	4		not registered	
	5	h	not registered	
	6	h	not registered	
	7		not registered	
	8		not registered	
	9		not registered	
	10		not registered	
	11	/	not registered	
	12	/	not registered	
	13	1.	not registered	
	14	1	not registered	
	15		not registered	
	A.II.			
	All			
	NOTE: If you do nothing within 90s, connection will be disconnected.			
	Send Exit			

Figure 4-6-7 USSD

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

4.6.8 Carrier

Carrier	
Select Port	Port 0 💌
Select Mode Carrier List	Automatic

Save

Figure 4-6-8 Select Carrier

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.

4.6.9 BCCH

Figure 4-6-9 BCCH

BUU	1																					
		0			1			2			3			4			5			6		
Port	LAC	CID	dbm	Detail																		
0																						Detail
1																						Detail
2																						Detail
3																						Detail
4																						Detail
5																						Detail
6																						Detail
7																						Detail
8																						Detail
9																						Detail
10																						Detail
11																						Detail
12																						Detail
13																						Detail
14																						Detail
15																						Detail



	1.
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
ВССН	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



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
manipulation	number 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

IP->1	Fel Routing					
	Index	Description	Source IP	Source Prefix	Destination Prefix	Destination
	0	default	Any	any	any	Port Group 0
			_			

Figure 4-7-2 IP to Tel Routing

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

Add Delete Modify

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.
Index	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.
	• 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.
	• 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
Destination	Its specifies destination Port or Port Group

Table 4-7-2 Description of IP to Tel Routing

4.7.3 Tel->IP Routing

Figure 4-7-3 Tel to IP Routing

Index	Description	Source Port	Source Prefix	Destination Prefix	Destinatio n
0	default	Any	any	any	SIP Server
30	To vps	Port Group 31	х.	00	IP 31
31	Carrier A to B	Port 0	013[58]	133	Port Gro

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.
	It uniquely identifies a route. Its value is assigned globally, ranging from
Index	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
Description	string
Carrier Dant	It specifies the Port or Port Group which will receive the calls from
Source Port	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.
	• 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.
	• 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
Destination	Its specifies destination Port or Port Group

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

Figure 4-7-4 Tel to IP routing Modify

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

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			7
Description	To vps			Ī
Source Prefix	х.			Ī
Source	C Port	0	•	_
	Port Group	31 (Unicom)	-	
Destination Prefix	00			
Destination	O Port	0	-	
	C Port Group	0 <all></all>	-	
	ΘIP	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".

D !	170	$T_{-1} + -$	ID.		N/ - 1:C
HIGHTE	4-/-n	Terto	1 1 1 1	$\alpha_{\rm HH} n \sigma$	N/I/MITV
IIZUIU	T = 1 = 0			Julie	IVIOUII V
					/

index	31	
Description	Carrier A to B	
Source Prefix	13[58]	
Source	Port	0
	C Port Group	0 (all)
Destination Prefix	133	
Destination	O Port	0
	Port Group	31 (Unicom)
	O IP	10 <other></other>
	C IP Group	18 <asterisk></asterisk>
	C SIP Server	

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

Figure 4-8-1 IP->Tel destination numbers manipulation

IP->	Tel Ma	anipulation									
	Inde x	Descriptio n	Source IP	Source Prefix	Destinatio n Prefix	Destination	Stripped Digits from Left	Stripped Digits from Right	Pref ix to Add	Suffi x to Add	Numb er of Digits to Leave from Right
	0	safcom	IP Group 31	any	2547	Port Group	3	0	0		
Total:	1entry	16entry/page	e 1/1page Pag	ie 1 💌							
				Ad	d Delete	Modify					

IP->Tel destination numbers manipulation	It is an optional configuration item, and is used to add a rule for changing number
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 rule for the ease of identification. Its value is character string
Source IP	 It specifies the source IP which will send the calls to gateway Any: any IP address IP: specific an IP address IP Group: specific an IP group
Source Prefix	 All the caller number must match the source prefix. It specifies the source prefix allow to send call out 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
Destination Prefix	 All the called number must match the destination prefix, the call prefix indicates the connected number 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
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

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

Dinstar Technologies Co., Ltd.

Add an IP->Tel Manipulation, to change the called number from 2547888888 to 07888888 Figure 4-8-2 IP->Tel destination numbers manipulation modify

Index	0					
Description	safcom	safcom				
Source Prefix	any					
Source IP	O IP	13 <mathnew></mathnew>				
	IP Group	31 <allow calls=""></allow>				
Destination Prefix	2547					
Destination Port	O Port	0 🗸				
	Port Group	31 <1>				
Stripped Digits from Left	3					
Stripped Digits from Right						
Prefix to Add	0					
Suffix to Add						

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



Tel->IP	Source	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: 0er	ntry 16ent	try/page 1/0pag	je 🚽							
				Add	Delete	Modify				

Tal \ID destination	It is an optional configuration item, and is used to add IP->Tel number
numbers	change data.
manipulation	The IP->Tel Manipulation defined the rules of add, and deletion of
manipulation	called numbers, which are referenced by IP->Tel routing.
Index	It uniquely identifies a route. Its value is assigned globally, ranging
Index	from 0 to 31.
Description	It describes the rule for the ease of identification. Its value is character
Description	string

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

Source Prefix	 All the caller number must match the source prefix. It specifies the source prefix allow to send call out 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
Destination Prefix	 All the called number must match the destination prefix, the call prefix indicates the connected number 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
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 Leave from Right	It specifies the number of Digits to Leave from Right

Example

Add an IP->Tel Manipulation, to change the called number from 2547888888 to 07888888

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

Tel->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				
-				

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

Tel->IP	Destinat	tion Numbers 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: 0er	ntry 16entr	ry/page 1/0page	• -							

Add Delete Modify

Tel->IP destination	It is an optional configuration item, and is used to add IP->Tel number change data.					
manipulation	The IP->Tel Manipulation defined the rules of add, and deletion of called					
manipulation	numbers, which are referenced by IP->Tel routing.					
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					
Description	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.					
	• 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					
Destination	Its specifies destination Port or Port Group					
Stripped Digits from	It specifies the length of the digits to be delated from left					
Left	it 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 eight					
Right	it specifies the length of the digits to be deleted from fight					
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 appoints the number of Divite to Leave from Dight					
Leave from Right	is specifies the number of Dights to Leave from Right					

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

Example

Add an IP->Tel Manipulation, to change the called number from 2547888888 to 07888888

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

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]

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

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

Table 4-9-1 Descrip	ntion of IP->Tel	Operation
	p n o n p n o n p n o n p n o n n o n n o n o n o n o n n o n n o n o n n n o n n n n n o n	operation

	It is an optional configuration item. Operation configuration essentially				
ID > Tal Operation	involves allow, barring some IP and IP Group send calls to certain				
IF->1er Operation	numbers. It includes: forbid call, call allowance, auto call, and password				
	authentication.				
Index	It uniquely identifies a route. Its value is assigned globally, ranging				
Index	from 0 to 31.				
	It specifies the source IP which will send the calls to gateway				
Course ID	• 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.				
	• 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.				
	• 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				
	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				
Description	string				

Example

Index 31: barring the certain calling number from IP 14<elastix> Figure 4-9-2 IP->Tel Operation Modify

Index	31		
Source Prefix	2877		7
Source IP	© IP	14 <elastix></elastix>	
	C IP Group	18 <asterisk></asterisk>	
Destination Prefix	07		7
Operation	Forbid Call		_
	C Allow Call		
Description	restrict unicom		7

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 Passwo	rd 🐽		
Description	password		

4.9.2 Tel->IP Operation

Figure 4-9-4 Tel->IP Operation

Tel->IP Op	eration					
	Index	Source Prefix	Destination Prefix	Operation	Description	
Total: Gentry 16entry/page 1/0page						
rotal. contay	roomajipago	hopago	Add Delete	Modify		
			Add Delete	woully		

	It is an optional configuration item. Operation configuration essentially				
Tal > ID Operation	involves allow, barring some IP and IP Group send calls to certain				
iei->ir Operation	numbers. It includes: forbid call, call allowance, auto call, and				
	password authentication.				
Indox	It uniquely identifies a rule. Its value is assigned globally, ranging from				
Index	0 to 31.				
	It specifies the source IP which will send the calls to gateway				
Course ID	• 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.				
	• 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.				
	• 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				
	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				
Description	string				

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

4.10 Port Group Configuration

4.10.1 Port Group

		Figure	e 4-10-1 Por	t Group	
Port Group					
	Index	Descrip	tion	Port	Select Mode
	0	all		0,1,2,3,4,5,6,7,8,9,10,11,12,	,1 Cyclic Ascending
Total: 1entry 16entry/	page 1/1page P	'age 1 👻			
		Add	Delete	Modify	
		Figure 4-1	0-2 Port Gr	oup Modify	
Port Group Mod	lify				
	_				
Index	0				
Description	al	I			
Select Mode	C	yclic Ascending)		V
Port	V	Port 0	Port 1		
	V	Port 2	Port 3		
	1	Port 4	Port 5		
	v	Port 6	Port 7		
	V	Port 8	Port 9		
	V	Port 10	Port 11	1	
	1	Port 12	Port 13	3	
	\checkmark	Port 14	Port 15	5	
		ОК	Reset	Cancel	

4.11 IP Trunk Configuration

4.11.1 IP Trunk

Figure 4-11-1 IP Trunk

IP				
	Index	IP	Port	Description
	10	172.16.0.124	5060	other
	13	172.16.3.55	5060	eia
	14	172.16.0.123	5060	elastix
	17	172.16.1.123	5060	FreeSentral
	19	172.16.244.136	5060	ondo server
	31	110.164.212.105	5060	to vps

Total: 6entry 16entry/page 1/1page Page 1

Add Delete Modify

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 . Its value is assigned globally, ranging from
muex	0 to 31.
Description	It describes the trunk for the ease of identification. Its value is character
Description	string
ID	It is an interworking parameter between the remote Softswitch and the
IL	SIP server. It specifies the IP address of the peer equipment.
Dort	It is an interworking parameter between the remote Softswitch and the
FOIL	SIP server. It specifies the SIP port number of the peer equipment

Example

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

Figure 4-11-2 IP Trunk Modify

IP Modify	
Index	24
index	31
IP	110.164.212.105
Port	5060
Description	to vps
	OK Reset Cancel

4.11.2 IP Trunk Group

IP Group			
	Index	Description	IP
	18	asterisk	10,14,17,
	19	all	13,19,
Total: 2entry 16entry/pag	e 1/1page Page 1 💌	Add Delete Modify	

Figure 4-11-3 IP Trunk Group

Table 4-11-2 Description	of IP Trunk	Group
--------------------------	-------------	-------

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

Example

To add an IP group, set IP "10, 14, 17" to IP group 18 Figure 4-11-4 IP Trunk group modify

idex	18			
escription	asteris	k		
þ		Index	IP	Port
	v	10	172.16.0.124	5060
		13	172.16.3.55	5060
	V	14	172.16.0.123	5060
	v	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
Enable Silence Suppression	🔘 No 🖲 Yes
0 H D T	
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 Incoming Call for Hotline	© No ● Yes
Auto Outgoing Douting Tuno	No Ves
Auto Outgoing Routing Type	
P to GSM One Stage Draining Redirect Call When All Parts Rusy	O No 🔍 Yes
Play Voice Prompt for GSM Incoming Calls	NO Yes
	0 140 0 163
DTMF Parameter	
DTMF Method	SIGNAL
NAT Traversal	Disable
Other Configuration	
Licor ID Is Phone Number	© No ♥ Yes
Only Accent Calls from SIP Server	● No ○ Yes
Allow Coll from CSM to IP without Pagistration	● No O Yes
Allow Call from IP to CSM without Registration	© No ♥ Yes
Reject Anonymous Call from IP to GSM	
Use # as End Key	No Ves
No Answer Timeout	55 s
Interdiait Timeout	4
Call Delay	
Call Delay	v s

NOTE: It must restart the device to take effect.

Save

Table 4-12-1	Description	of Service	Configuration
--------------	-------------	------------	---------------

	Means the initial port when RTP voice stream transmit in the IP network,				
LOCAL RTP PORT	in general, using the factory default values. When there are multiple				
Channel	DINSTAR series voice products, and the network gateway or router's				
	NAT with loopholes, user can try changing this item				
Enable Silence	Enable the "silence suppression" almost no impact on call quality, and can				
Suppression	save about half of the bandwidth.				
	Each country has its different call progress tone required standards, such				
Call Progress Tone	as busy tone, ring back tones and ring tone standards, users can select the				
	area standard from here .				
	Means the code format when Voice transfer on IP network, support				
Preferred Coders	PCMA, PCMU, G.723.1 and G.729AB.				
Enable PSTN	Means when call from PSTN side, you can dial the function keys for				
Incoming	checking number, setting IP and so on				
Configuration					
Enable Auto Outgoing	Means when call out , whether by ordinal or polling pick to Select a				
Routing	Channel, this feature are generally used when use the same SIP User ID				
Routing	to register				
ID to DSTN One Stage	The User ID will be sent directly to PSTN, for example: the user calls				
Dialing	6715, the device will sent 6715 User ID to PSTN				
Disc Vaira Durant fam	Setting is yes, when through the PSTN calls to the Channel, the device				
Play voice Prompt for	will with the clew tone, the default is "Please dial the extension User				
PSTN Incoming Calls	ID"; setting to No, the device will play dial tone				
	DWG2001/DWG2004/DWG2000B-16G support RFC2833 and SIGNAL				
DTMF	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				

Dinstar Technologies Co., Ltd.

	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						
STUN	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							
PSTN without	Refer to "SIP Configuration" -> "Is register". If "Is register" setting is no,						
1 5 1 Without	this option need set Yes ,to avoid that the devices can not call out						
Registration							
Allow Call from PSTN	Pafor to "SID Configuration" > "Is register". If "Is register" sotting is no						
to IP without	Refer to SIF Configuration -> is register . If is register setting is no,						
	this option need set Yes ,to avoid that the devices can not call in						
Registration							
Reject Anonymous call	The incoming anonymous calls will be rejected						
from IP to PSTN	The incoming anonymous cans will be rejected						
	In General, SIP phones are based on # as the end, if this option is set to						
Use # as End Key	The Scherar, on phones are based on a as the end, it this option is set to						
	ino, the diai-up will end expires diai-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

Figure 4-12-2 SIP Configuration

SIP Configuration	
SIP Proxy	
SIP Server Address	
SIP Server Port(default: 5060)	5060
Check Net Status	No
Outbound Proxy	
Outbound Proxy Address	
Outbound Proxy Port	5060
Use Random Port	◎ No [©] Yes
Local SIP Port	5060
Is Register	◉ No ♡ Yes
DNS query type	A query
DNS refresh interval (range:0 - 60,000min, 0 means disable)	0 min
T1	500 ms
T2	4000 ms
Τ4	5000 ms
ТМАХ	32000 ms
Keepalive Interval(range:1 - 3600s)	10 s
Enable 100rel	● no ◎ ves
From Mode when Caller ID Is Available	Tel/User
From Mode when Caller ID Is Unavailable	Anonymous
AnswerMode	Answered
183 Mode	Immediately 📼
Response Code switch	
Response code	Response code after switch

NOTE: It must restart the device to take effect.

Save

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				
interval	default value is disable.				
T1	Used to define the SIP protocol T1 timer value, default is 500ms				
T2	Used to defines the SIP protocol timer values, default value is 4000ms				
T3	Used to define the T2 timer value in SIP protocol, the default is 5000ms				
Keep alive	Used to keep communicate between equipment and the SIP server that make				
Interval	the device is available. In general, using the factory default values				
	Used to config "From" Mode when Caller ID Is Available when call from				
From Mode	GSM to VoIP				
when Caller ID	Tel/User: From: caller number <sip:3001@ip>;tag=51088abb</sip:3001@ip>				
	User/User: From: 3001 <sip:3001@ip>;tag=51088abb</sip:3001@ip>				
Is Available	Tel/Tel: From: caller number <sip: @ip="" caller="" number="">;tag=51088abb</sip:>				
	User/Tel: From: 3001 <sip: @ip="" caller="" number="">;tag=51088abb</sip:>				
From Mode	Used to config "From" Mode when Caller ID Is Unavailable				
when Caller ID	Anonymous : From: <sip: @ip="" anonymous="">;tag=51088abb</sip:>				
Is Unavailable	Username : From: <sip: @ip="" username="">;tag=51088abb</sip:>				
	Answered: Gateway answer the IP incoming call (send SIP message "200				
Answer Mode	OK" to IP part) after GSM part answered				
	Alerted: Gateway answer the IP incoming call after GSM part Alerted				
	Immediately : Gateway send "183 RING" immediately to IP part while it				
183 Mode	receive "INVITE" from IP part.				
	Alerted: Gateway send "183 RING" after receive "ring back" from PSTN				
Response Code	Used to config the compatibility of SIP Response Code, Fill the response				
switch	code in the front, and Fill the switch code in the behind				

0

Detail

4.12.3 Port Configuration

15

ist								
Port	SIP User ID	Authenticate ID	Tx Gain	Rx Gain	To VOIP Hotline	To PSTN Hotline	Auto-Dial Delay Time(s)	Detail
0			2	6			0	Detail
1			2	6			0	Detail
2			2	6			0	Detail
3			2	6			0	Detail
4			2	6			0	Detail
5			2	6			0	Detail
6			2	6			0	Detail
7			2	6			0	Detail
8			2	6			0	Detail
9			2	6			0	Detail
10			2	6			0	Detail
11			2	6			0	Detail
12			2	6			0	Detail
13			2	6			0	Detail
14			2	6			0	Detail

Figure 4-12-3 Port List

Figure 4-12-4 Port Configuration

6

2

Port Configuration	
All ports register used same user ID Current Port	No O Yes Port 0
SIP User ID	
Authenticate ID	
Authenticate Password	Show Password
Tx Gain Rx Gain	+2dB • +6dB •
To VOIP Hotline	
To PSTN Hotline	

Save Back

Table 4-12-3	Description	of Port	Configu	ration
10010 + 12 J	Description	or r ort	conngu	ration

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	
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.	
To VoIP Hotline	When PSTN part client calls to this port, gateway will auto forward to the hotline User ID. Leave it blank if you don't need this function. *Note: Please config Tel->IP Operation if you need this function.	
To PSTN Hotline	When VoIP part client calls to this port, Gateway will auto forward to the 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.	
Auto-Dial Delay Time	The auto-dial delay time of hotline, the range is 0-10 seconds	

4.13 Digit Map

Figure 4-13-1 Digit map

Distant and		
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 "xxxxxxx", 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)xxxxxxx

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 f	from your computer to the device.	Lipload	
Web	选择文件 未选择文件	Upload	
Dsp Firmware	选择文件】未选择文件	Upload	



Select the software, Web and 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

Management Parameter	
Voice Prompt Language	English
Syslog Parameter Syslog Enable	● Yes ◎ No
Server Address	
Syslog Level	NONE
Send CDR	© Yes ◉ No
NTP Parameter	
NTP Enable	● Yes © No
Primary NTP Server Address	asia.pool.ntp.org
Primary NTP Server Port	123
Secondary NTP Server Address	cn.pool.ntp.org
Secondary NTP Server Port	123
Check Interval	500 s
Time Zone	GMT+8:00 (Beijing, Singapore, Taipei, Hong Kong) 💌
	Save

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.	
Send CDR	Telephone exchanges generate so called Call Detail Records (CDRs)	
	which contain detailed information about calls originating from,	
	terminating at or passing through the exchange. Not surprisingly	
	CDRs are used for billing.	
	Set to Yes gateway will sne the CDR to the syslog server.	
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 Data backup



Click 'Backup' for download configuration file to your computer.

4.14.4 Config Restore

Data Restore	
Send data file f	rom your computer to the device.
Configuration	浏览···· Restore
	NOTES: The upload process will last about 30s.

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.



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

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 reach ability 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	
Information		
Pinging 172.16.1.1 with 56 byte	es of data:	
Reply seq=0 from 172.16.1.1: by	tes=56 time=20ms TTL=64	
Reply seq=1 from 172.16.1.1: by	/tes=56 time<1ms TTL=64	
Reply seq=2 from 172.16.1.1: by	tes=56 time=10ms TTL=64	
Reply seq=3 from 172.16.1.1: by	ytes=56 time=10ms TTL=64	
Ping statistics for 172.16.1.1		
Packets: Sent = 4, Received = 6	H, LOST = 0 (0% LOSS)	
RII MINIMUM - IMS, MAXIMUM = 10	ms, Average - ioms	

4.14.7 Tracert Test

Traceroute 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

acert Test							
Trace	ert Destinati	on www.google.com.hk					
Мах	Hops of Tra	cert(1-255) 30					
		Start Stop					
ormatio	on						
Trac	ing route	to www.google.com.hk[74.125.71.99] over a maximum of 30					
hops	:						
1	1 ms	172.16.1.1					
2	*	Request timed out.					
3	*	Request timed out.					
4	30 ms	121.15.1/9.86					
5	30 ms	119.145.47.40					
2	30 ms	202.97.55.250					
6	40 mg	202.07.60.22					
a a	40 mg	202.97.61.102					
10	80 ms	202.97.62.214					
11	40 ms	209.85.241.58					
12	30 ms	209.85.253.69					
14							
13	40 ms	216.239.48.230					
13 14	40 ms 30 ms	74.125.71.99					

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4.14.8 Username & Password

Figure 4-14-8 IVR Voice Prompt Upload

rname & Password	
Web Configuration	
Old Web Username	admin
Old Web Password	
New Web Username	
New Web Password	
Confirm Web Password	
Talact Configuration	
Old Telnet Username	admin
	aunn
Old Teinet Password	
New Telnet Username	
New Telnet Password	
Confirm Telnet Password	

Save

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



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 have been connected to network physically, but can not 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 condition, 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