DINSTAR

DWG2000-16G User Manual v1.0

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Table of Contents

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1. Equipment Introduction

This chapter mainly introduces functions and structures of DWG2000-16G.

1.1 Introduction

DWG2000-16G 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

DWG2000-16G provides access of GSM network.

With the development of users and telecom service, mobile network and fixed network integration will be steadily increasing. DWG2000-16G 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 DWG2000-16G shows as follow

Figure 1-3-1 Front view of DWG2000-16G

Table 1-3-1 Description of Front view

Inde	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	Used for reset the configuration to factory. Need keep press for 5 seconds
4	RUN	Indicate the status of the device.
5	PWR	Indicate the status of the power connection
6	ANT Interface	Standard antenna interface
7	ANT indicator	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 DWG2000-4G/DWG2000-16G

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Table 1-3-2 Description of Rear view

Inde	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

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

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- 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 ℃（Operation），-20~80 ℃（storage）
- Humidity: 5%～90%RH,
- Power Consumption: 5W
- Dimensions: 112(W) x76(D) x24(H) mm
- Net weight: 0.7kg

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2. Equipment Quickly Installation

This chapter mainly introduces DWG2000-16G hardware installation and connection of equipment.

2.1 Installation Notice

DWG2000-16G 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 DWG2000-16G 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

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

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

Figure 2-2-1 Push in the GSM user board

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2.2.2 Antenna Installation

Figure 2-2-2 Antenna Installation

Connect antenna

2.2.3 Network Cable Connection of Equipment

Figure 2-2-3 DWG2000-16G connection

IP network

Lanswitch

2.2.3 Power Cable Connection of Equipment

Figure 2-2-3 DWG2000-16G connection

Power cable

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3. Network Configuration

In this chapter we will introduce the initial configuration of DWG2000-16G. 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 theat step again.

3.2 General Feature Codes for System Setting

Table 3-3-1 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(≤ 20s), then press # end the
*111#	Restart device

3.3 Static IP Configuration

This chapter introduces IP configuration of DWG2000-16Gthrough calling IVR.

Assuming the IP address of a DWG2000-16G 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

2) Dial the phone number of the SIM card. Dail "*150*1#" after hearing "please dail extension number ". Hang up after hearing "setting successful"

3) 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"

4) 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

2) Dial the phone number of the SIM card. Dail "*150*2#" after hearing "please dail extension number ". That means the DHCP is confirued successfully

3) 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 DWG2000-16G.

4.1 Access the System Through HTTP

Enter IP address of DWG2000-16G in browser. The default IP of LAN port is 192.168.11.1. and the GUI shows as below:

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

DWG2000-16G WEB configuration interface consists of the navigation tree and the detail configuration interfaces.

Figure 4-3-1 WEB introduce

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

Figure 4-4-2 system information

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Table 4-4-1 Description of system information

MAC Address	Displays the current MAC of the gateway, for example: 00-1F-D6-1B-3D-02
Network Mode	DWG2000-16G 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 4-4-3 Mobile information

Table 4-4-2 Description of mobile information

Port	Number of GSM/CDMA ports .
Type	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 connection status of current GSM / CDMA module
Remaining Call Duration	Limite a call duration to the SIM card, when call duration is out of that 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.

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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 4-4-4 SIP information

Displays registration status information with Softswitch platform or SIP Server

Table 4-4-3 Description of SIP information

Port	The number of GSM channel, DWG2000-16G 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"

4.4 Statistics

4.4.1 TCP/UDP

Figure 4-4-4 TCP/UDP Statistics

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

Refresh

4.4.2 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

Table 4-5-1 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 Call History

Port	Incoming Received	Incoming Connected	Incoming Answered	Incoming Failed	Outgoing Attempted	Outgoing Connected	Outgoing Answered	Outgoing Failed
0	3089	3089	3089	0	0	0	0	0
1	3090	3090	3090	0	0	0	0	0
2	3091	3091	3091	0	0	0	0	0
3	3088	3088	3088	0	0	0	0	0
4	3092	3092	3092	0	0	0	0	0
5	3078	3078	3078	0	0	0	0	0
6	3093	3093	3093	0	0	0	0	0
7	3089	3089	3089	0	0	0	0	0
8	3126	3126	3126	0	0	0	0	0
9	3127	3127	3127	0	0	0	0	0
10	2908	2908	2908	0	0	0	0	0
11	3126	3126	3126	0	0	0	0	0
12	3125	3125	3125	0	0	0	0	0
13	3089	3089	3089	0	0	0	0	0
14	3122	3122	3122	0	0	0	0	0
15	3123	3123	3123	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

4.5Network Configuration

4.5.1 Local Network

Figure 4-5-1 Local Network

Note: It must restart the device to take effect.

Save

Table 4-5-1 Description of Local network

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

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4.5.2 VLAN Parameter

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 VLAN	Management 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.
	Management 802.1p Priority (0-7)	Under 802.1q protocol users can set VLAN priority
	IP address	Users can set DHCP or static IP address
	Management VLAN DNS Server	Users can set DHCP or static DNS server IP address

4.5.3 VPN Parameter

A virtual private network (VPN) is a network that uses primarily public telecommunication infrastructure, such as the Internet, to provide remote offices or traveling users access to a central organizational network.

VPNs typically require remote users of the network to be authenticated, and often secure data with encryption technologies to prevent disclosure of private information to unauthorized parties.

Figure 4-5-3 VPN Parameter

4.5.4 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 ARP

Figure 4-5-5 Add ARP

4.6 Mobile Configuration

4.6.1 Basic Configuration

Figure 4-6-1Basic Configuration

Table 4-6-1 Description of Basic Configuration

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: GSM: 850/900/1800/1900 MHz; CDMA: 800 MHz
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 "Yes" under "remote API enable"
Auto reset module	Used to reser GSM/CDMA module when SIM card can not register

4.6.2Mobile

Figure 4-6-2 Mobile Configuration

Table 4-6-2 Description of Mobile Configuration

Mobile Number	SIM card number of the channel. That must be configured when "Forward" funciton enable.
Enable Call Duration Limitation	This function is to limit the max call duration of channel. The max call duration is between 1 to 65535 minutes.
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.
Mobile Number （Receiving Alarm）	The mobile phone No. which used to receive the alarm SMS. Users can get SMS report of SIM/UIM card status(SIM Remain Time) in DWG.

Alarm Threshold（via SMS）	When the SIM remain time is or less than this value, DWG will send the alarm SMS to remind the users of the SIM remain time.
Port Description for Alarm	It is the identification mark of SIM/UIM card in the SMS report. The mobile phone No. of the SIM/UIM card is recommended to use as the port description for alarm, or any other string.
SIM Remain Time	Indicates the current sim remain time. It can't modified
Restore time	Recovers the SIM remain time to initial value, the Maximum Call 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, DWG2000-16G will connect soon. It doesn't wait mobile side answer.

4.6.3 SIM/UIM Card Lock

Figure 4-6-3 Configuration of SIM/UIM Card Lock

Table 4-6-3 Description of Configuration of SIM/UIM Card Lock

Select Port	Select the Channel No. which need to be locked.
SIM Card Lock	SIM card lock or unlock. Default is "No".
PIN Code	Correct PIN code is needed to lock or unlock the SIM card.

4.6.4 PIN Management

Figure 4-6-4 PIN Management

Detailed description as below:

Table 4-6-4 Description of PIN Management

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

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 sending

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.
Addressee	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.

Figure 4-6-7   USSD

Table 4-6-6 Description of USSD

Port	Select the GSM channel to send USSD
Display	Display the result of sending USSD
Input	The area to input USSD code

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4.6.8 Carrier

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 mode to select carrier,automatic and manual.
Carrier List	If you select manual mode,you can select carrier from carrier list.

4.7 Routing Configuration

4.7.1 Routing Parameter
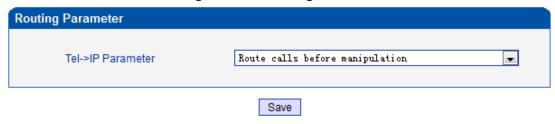
Figure   4-7-1 Routing Parameter

Table 4-7-1Description of Routing Parameter

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

4.7.2 IP->Tel Routing

Figure 4-7-2 IP to Tel Routing

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

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
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

4.7.3 Tel->IP Routing

Figure 4-7-3 Tel to IP Routing

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

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 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
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
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Destination	Its specifies destination Port or Port Group

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.

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".

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

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

	Index	Description	Source IP	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
☐	0	safcom	IP Group 31	any	2547	Port Group...	3	0	0	---	---

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

Add   Delete   Modify

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

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

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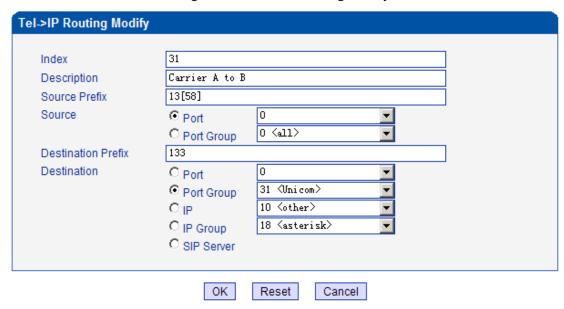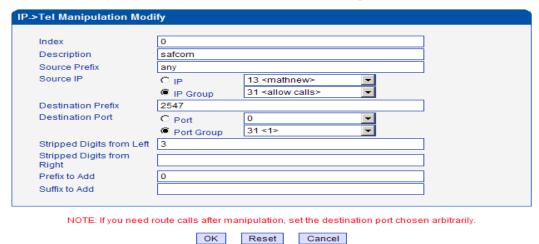
Figure 4-8-2 IP->Tel destination numbers manipulation modify

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

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

Tel->IP destination numbers manipulation	It is an optional configuration item, and is used to add IP->Tel number change data. 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 31.
Description	It describes the rule for the ease of identification. Its value is character string

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

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

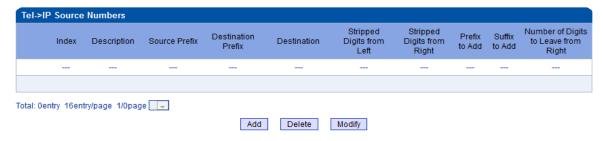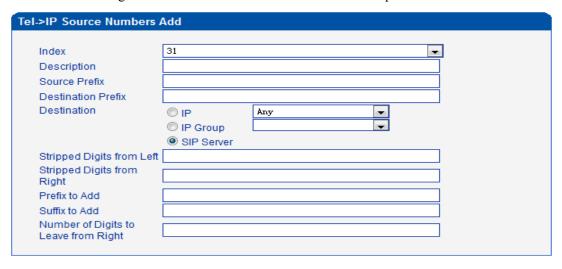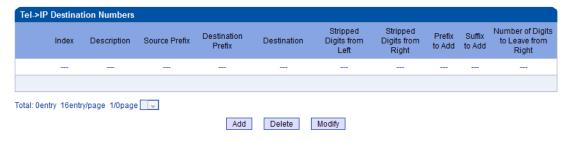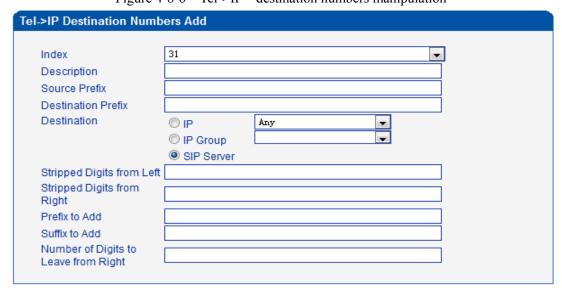Figure 4-8-5 Tel->IP destination numbers manipulation

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

Tel->IP destination numbers manipulation	It is an optional configuration item, and is used to add IP->Tel number change data. 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 31.
Description	It describes the route for the ease of identification. Its value is character string
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

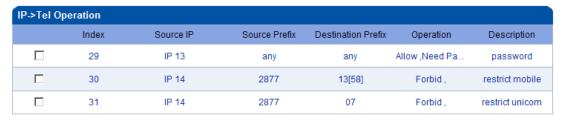Figure 4-8-6　Tel->IP　destination numbers manipulation

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

Figure 4-9-1 IP->Tel Operation

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

IP->Tel Operation	It is an optional configuration item. Operation configuration essentially 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 31.
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
Operation	Its specifies number analysis rule • Forbid call • Allow call • Auto call • Password authenticate
Description	It describes the route for the ease of identification. Its value is character string

Example

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

Figure 4-9-2 IP->Tel Operation Modify

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

4.9.2 Tel->IP Operation

Figure 4-9-4 Tel->IP Operation
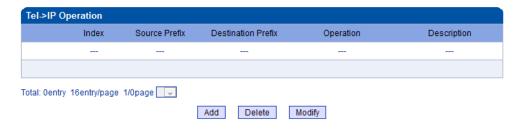

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Table 4-9-2 Description of Tel->IP Operation

Tel->IP Operation	It is an optional configuration item. Operation configuration essentially 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 31.
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
Operation	Its specifies number analysis rule • Forbid call • Allow call • Auto call • Password authenticate
Description	It describes the route for the ease of identification. Its value is character string

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4.10 Port Group Configuration

4.10.1 Port Group

Figure 4-10-1 Port Group

Port Group				
	Index	Description	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 Page 1 ▾

Add   Delete   Modify

Figure 4-10-2 Port Group Modify

Port Group Modify

Index `0`

Description `all`

Select Mode `Cyclic 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

OK   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

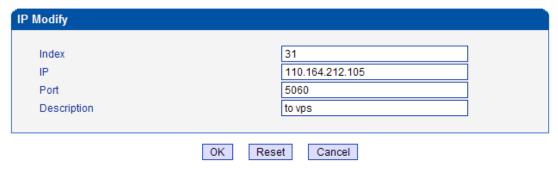| IP Trunk	Add remote IP of softswitch, SIP server which will send call traffics to gateway.
Index	It uniquely identifies a trunk . Its value is assigned globally, ranging from 0 to 31.
Description	It describes the trunk for the ease of identification. Its value is character string
IP	It is an interworking parameter between the remote Softswitch and the SIP server. It specifies the IP address of the peer equipment.
Port	It is an interworking parameter between the remote Softswitch and the 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	31
IP	110.164.212.105
Port	5060
Description	to vps

OK   Reset   Cancel

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4.11.2 IP Trunk Group
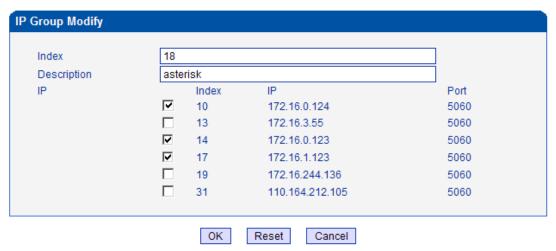
Figure 4-11-3 IP Trunk Group

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 IP "10, 14, 17" to IP group 18

Figure 4-11-4 IP Trunk group modify

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
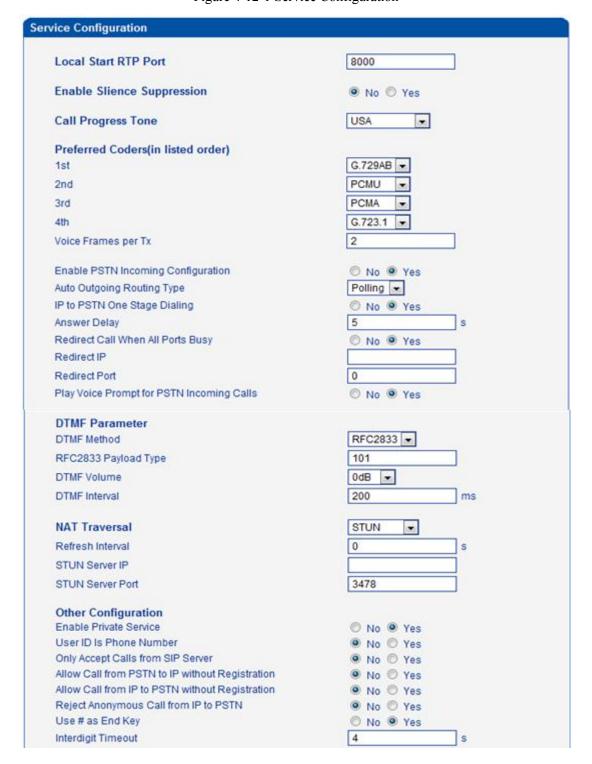
Figure 4-12-1 Service Configuration

Table 4-12-1 Description of Service Configuration

LOCAL RTP PORT Channel	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
Enable Silence Suppression	Enable the "silence suppression" almost no impact on call quality, and can save 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 standards, users can select the area standard from here .
Preferred Coders	Means the code format when Voice transfer on IP network, support PCMA, PCMU, G.723.1 andG.729AB.
Enable PSTN Incoming Configuration	Means when call from PSTN side, you can dial the function keys for checking number, setting IP and so on
Enable Auto Outgoing Routing	Means when call out , whether by ordinal or polling pick to Select a Channel, this feature are generally used when use the same SIP User ID to register
IP to PSTN One Stage Dialing	The User ID will be sent directly to PSTN, for example: the user calls 6715, the device will sent 6715 User ID to PSTN
Play Voice Prompt for PSTN Incoming Calls	Setting is yes, when through the PSTN calls to the Channel, the device will with the clew tone, the default is "Please dial the extension User ID"; setting to No, the device will play dial tone
DTMF	DWG2001/DWG2004/DWG2000-16G 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 PSTN without Registration	Refer to "SIP Configuration" -> "Is register" . If "Is register" setting is no, this option need set Yes ,to avoid that the devices can not call out
Allow Call from PSTN to IP without Registration	Refer to "SIP Configuration" -> "Is register" . If "Is register" setting is no, this option need set Yes ,to avoid that the devices can not call in
Reject Anonymous call from IP to PSTN	The incoming anonymous calls will be rejected
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
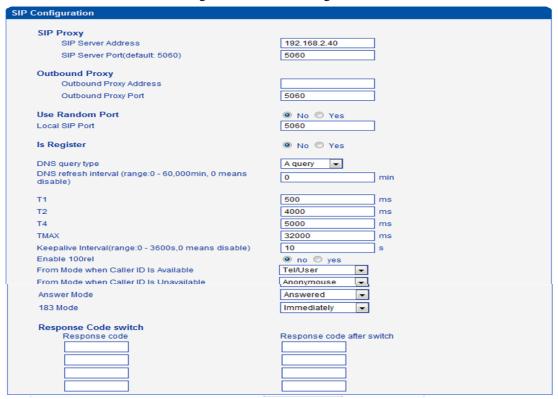
Figure 4-12-2 SIP Configuration

Table 4-12-2 SIP Configuration

SIP Server Address	Used for configure SIP server address and port, the address can be IP 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 interval	The interval of DNS refresh, Range from 0 to 60000 mins, 0 means disable 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 Interval	Used to keep communicate between equipment and the SIP server that make the device is available . In general, using the factory default values
From Mode when Caller ID Is Available	Used to config "From" Mode when Caller ID Is Available when call from GSM to VoIP Tel/User: From: caller number <sip:3001@IP>;tag=51088abb User/User: From: 3001 <sip:3001@IP>;tag=51088abb Tel/Tel: From: caller number <sip: caller number @IP>;tag=51088abb User/Tel: From: 3001 <sip: caller number @IP>;tag=51088abb
From Mode when Caller ID Is Unavailable	Used to config "From" Mode when Caller ID Is Unavailable Anonymous : From: <sip: Anonymous @IP>;tag=51088abb Username : From: <sip: Username @IP>;tag=51088abb
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 switch	Used to config the compatibility of SIP Response Code , Fill the response code in the front , and Fill the switch code in the behind

4.12.3 Port Configuration

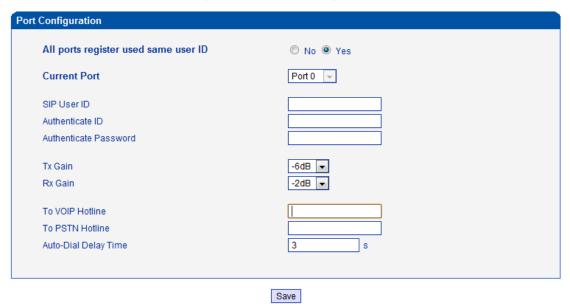Figure 4-12-3 Port Configuration

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
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

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 | 13xxxxxxxxx

    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)xxxxxxxxx

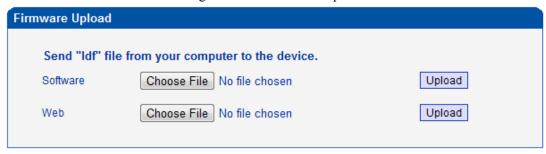   Means that first is "13","15" or "18", followed by 8 digits.

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

Select the software or Web 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.

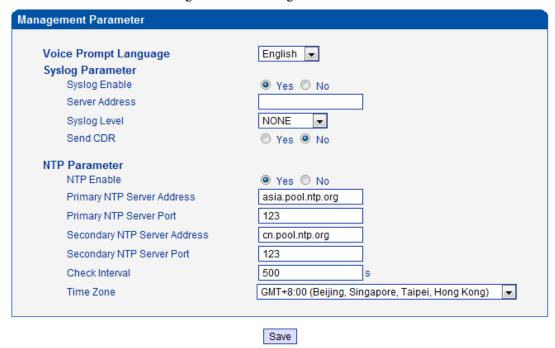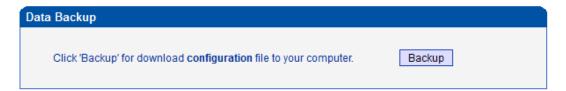### 4.14.2 Management Parameter

Figure 4-14-2 Management Parameter

Table 4-14-1 Management Parameter

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

4.14.3 Data Backup

Figure 4-14-3 Data backup

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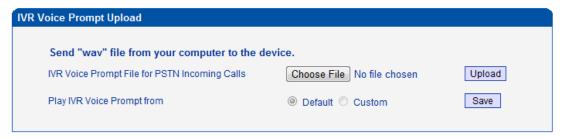
4.14.4 Data Restore

Figure 4-14-4 Data 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.

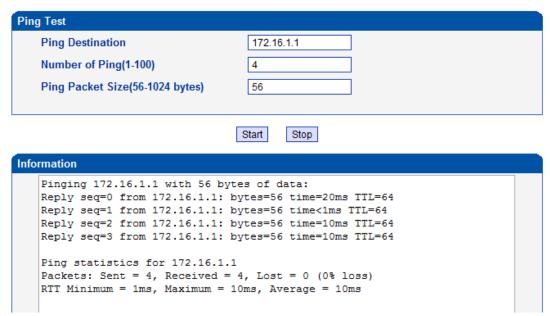Figure 4-14-5 IVR Voice Prompt Upload

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.
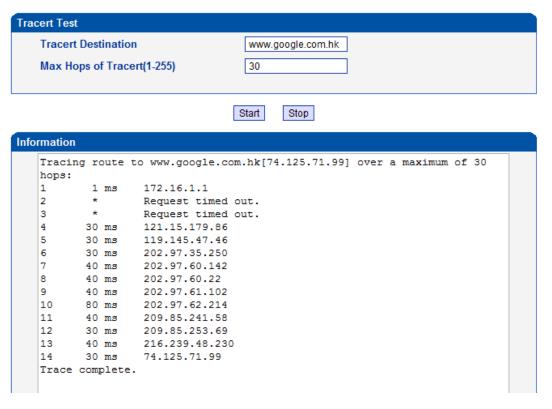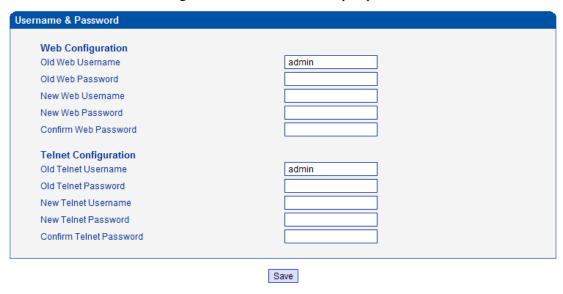
Figure 4-14-6 Ping Test

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

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4.14.8 Login Password

Figure 4-14-8 IVR Voice Prompt Upload

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

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

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

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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