DWG2000-8G User Manual v1.0

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Table of Contents

1. Equipment Introduction

This chapter mainly introduces functions and structures of DWG2000-8G.

1.1 Introduction

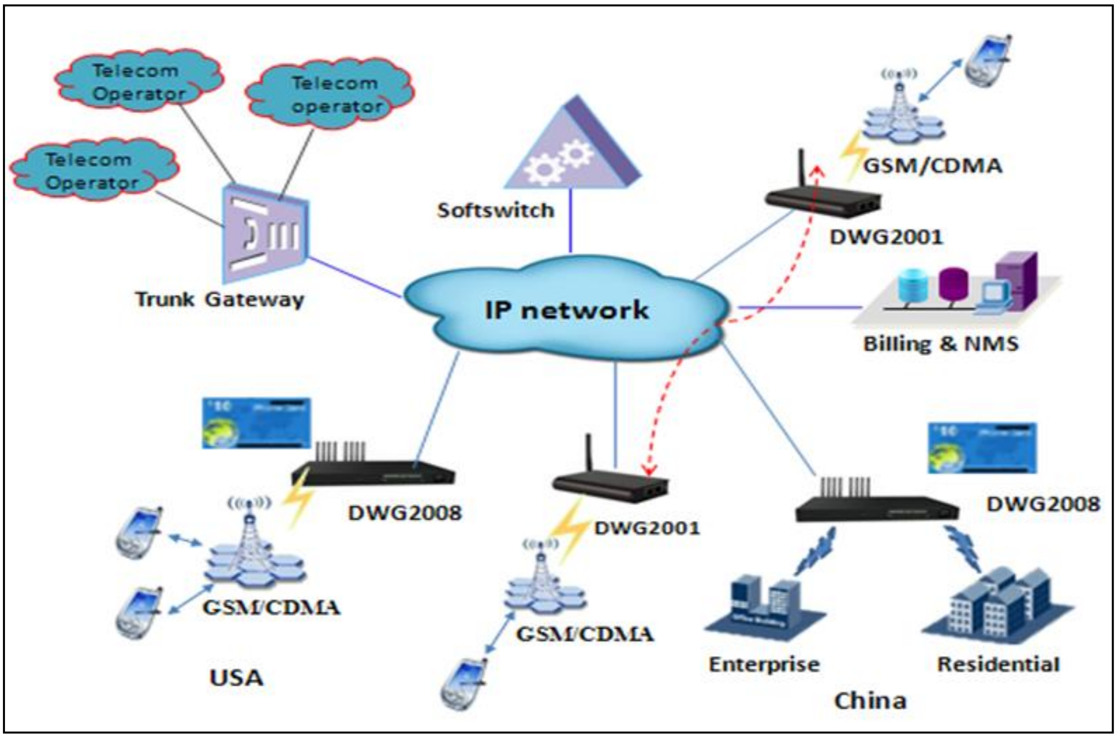
DWG2000-8G is full functions VoIP gateway based on IP and GSM wireless network, which provides a flexible network configuration, powerful features, and good voice quality. It works for carrier grade, enterprise, SOHO, residential users for cost-effective solution.

1.2 Scenario of Applications of Products

DWG2000-8G provides access of GSM network.

With the development of users and telecom service, mobile network and fixed network integration will be steadily increasing. DWG2000-8G 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 DWG2001 shows as follow

Figure 1-3-1 Front view of DWG2000-8G

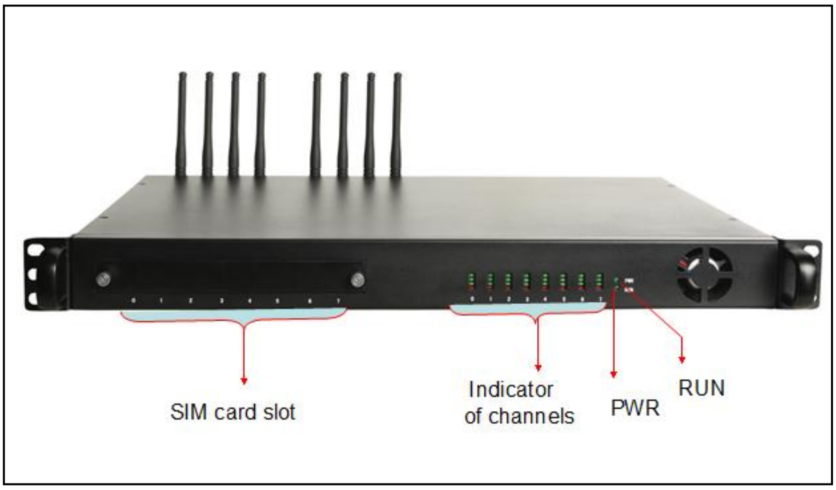


Table 1-3-1 Description of Front view

SIM Card Slot	8 SIM channel
Indicator of Channels	Red: Indicate that occupancy status of channels Green: Indicate GSM/CDMA signal strength indicator
PWR	Indicate the power status of SIM cards
RUN	Indicate the run status of SIM cards

Figure 1-3-2 Rear view of DWG2000-4G/DWG2000-8G

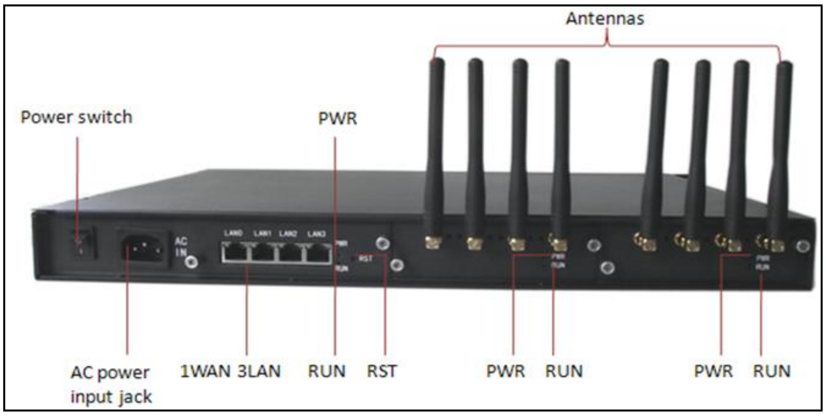


Table 1-3-2 Description of Rear view

Power Switch	Turn on or turn off the device
AC Power Input jack	8 SIM channel
WAN	WAN interface
LAN	LAN interface
RUN	Indicate the run status of module
POWER	Indicate the Power status of module
RST	Press it to reset the device to factory
Antennas	Antennas of CDMA/GSM module

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;

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
- 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

2. Equipment Installation

This chapter mainly introduces DWG2000-8G hardware installation and connection of equipment.

2.1 Installation Notice

DWG2000-8G 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 DWG2000-8G supports RJ45 standard with 10Mbps or 100Mbps network.

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

2.2 Installation Procedure

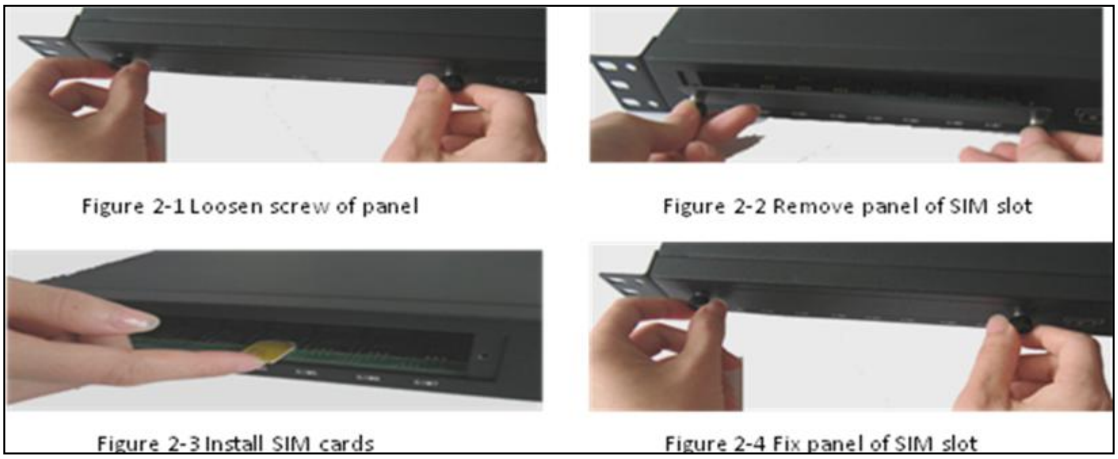
The outlook of DWG2000-8G looks like a 1U chassis; to install hardware the cable is needed. After unpacking the equipment, please do follow the procedure as following steps:

2.2.1 Install SIM Card

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

- Push down the yellow button,the SIM slot will popup;
- Inset the SIM card to the SIM slot.

Figure 2-2-1 Installing SIM card



2.2.2 Antenna Installation

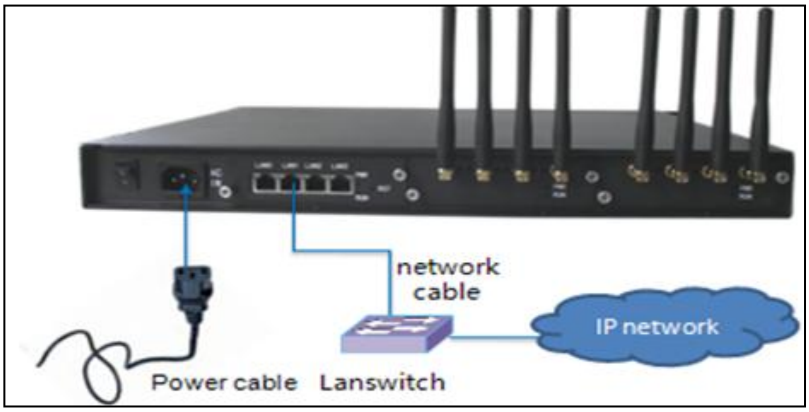
Take antenna connected in antenna interface of DWG which sign of "ANT" on

Figure 2-2-2 Antenna Installation

2.2.3 Cable Connection of Equipment

DWG2001 works in Route mode

Figure 2-2-3 DWG2001 connection



3. Network Configuration

In this chapter we will introduce the initial configuration of DWG2001 gateway. All of the network parameters of the gateway can be configured by IVR guidance.

3.1 Preparation

Please ensure the following stepes are done properly before IVR setting:

- Prepare an analog telephone or mobile phone
- Make sure the gateway is power on
- Make sure the gateway is connected with the network
- Completed the SIM installation
- Make sure that the current mobile network is working

3.2 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.

DWG2001 can work in two modes: route mode and bridge mode. when the gateway is under bridging mode, user should configure network parameters of WAN port; when the gateway is under the route mode, user should configure LAN port.

3.3 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

3.4 Static IP Configuration

This chapter introduces IP configuration of DWG2001 through calling IVR.

Assuming the IP address of a DWG2001 device is 172.16.0.100, subnet mask is 255.255.0.0, IP of gateway is 172.16.0.1, configured as follows:

Insert a SIM card into the DWG2001 gateway

1) The configuration mode: Dial the phone number of this SIM card. hear a message, then enter "*150*1#", hang up when hear " setting successful" message;
2) Configure IP address: Dial the phone number of this SIM card, hear a message, enter "* 152 * 172 * 16 * 0 * 100 #" hang up when hear "setting successful" message;
3) configure subnet mask: Dial the SIM card phone number, enter "*153*255*255*0*0#" hang up when hear "setting successful" message;
4) Configure gateway: Dial the SIM card phone number, enter "*156*172*16*0*1#" hang up when hear "setting successful" message;
5) Please wait about ten seconds when finishing the operations, restart device. dial the SIM card phone number, enter "*158#"to check the Static IP address;

3.5 DHCP Configuration

DHCP mode configure as follows:

- Insert a SIM card into a slot, dial the SIM card number. When hearing a hint message, then enter "*150*1#", if hearing " setting successful" message, which 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 0.0.0.0, which means that the gateway could not obtain a IP address successfully. Please check:

- Make sure the device have been connected to the network;
- Make sure the DHCP Server is working. If there is no DHCP Server, please set the IP of device to static IP .

4. WEB configuration

This charpter describes web configuration of DWG2001.

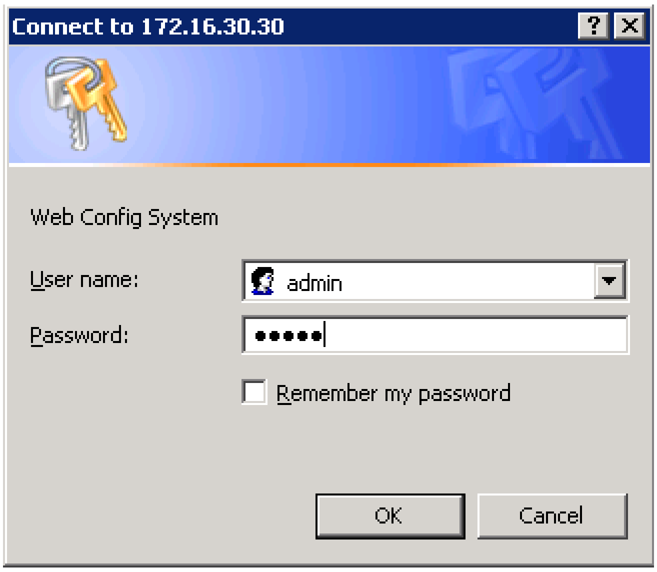
4.1 Preparing

WEB configuration includes the following components: network configuration, system information, mobile configuration and system configuration.

4.2 Access the System Through HTTP

Enter IP address of DWG2001 in browser. The default IP of LAN port is 192.168.11.1. and the GUI shows as below:

Figure 4-2-1 WEB log interface

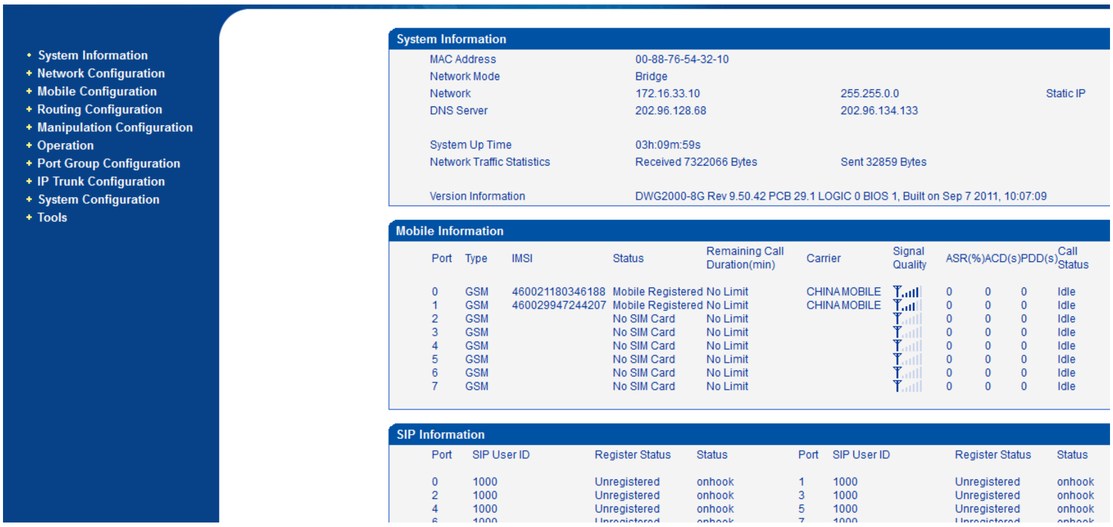


Enter username and password and then click “OK” in configuration interface. The default username and password are “admin/admin”. It is strongly recommended, change the default password to a new password for system security.

4.3 WEB Configuration

DWG2001 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.

4.4 System Information

Figure 4-4-1 system information

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

4.4.1 System Information

Figure 4-4-2 system information

System Information			
MAC Address	00-88-76-54-32-10		
Network Mode	Bridge		
Network	172.16.33.10	255.255.0.0	Static IP
DNS Server	202.96.128.68	202.96.134.133	
System Up Time	03h:09m:59s		
Network Traffic Statistics	Received 7322066 Bytes	Sent 32859 Bytes	
Version Information	DWG2000-8G Rev 9.50.42 PCB 29.1 LOGIC 0 BIOS 1, Built on Sep 7 2011, 10:07:09		

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	DWG2001 support two types network mode, which is bridge and route modes
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

4.4.2 Mobile Information

Figure 4-4-3 Mobile information

Port	Type	IMSI	Status	Remaining Call Duration(min)	Carrier	Signal Quality	ASR(%)	ACD(s)	PDD(s)	Call Status
0	GSM	460021180346188	Mobile Registered	No Limit	CHINA MOBILE		0	0	0	Idle
1	GSM	460029947244207	Mobile Registered	No Limit	CHINA MOBILE		0	0	0	Idle
2	GSM		No SIM Card	No Limit			0	0	0	Idle
3	GSM		No SIM Card	No Limit			0	0	0	Idle
4	GSM		No SIM Card	No Limit			0	0	0	Idle
5	GSM		No SIM Card	No Limit			0	0	0	Idle
6	GSM		No SIM Card	No Limit			0	0	0	Idle
7	GSM		No SIM Card	No Limit			0	0	0	Idle

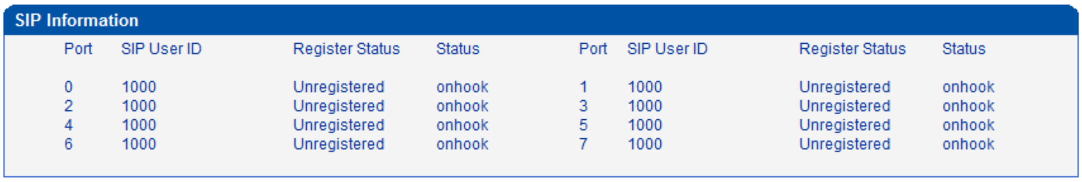
Display GSM / CDMA channel and network status information, detailed shown as below:

Table 4-4-2 Description of mobile information

Port	Numbers of ports of GSM/CDMA.
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.

4.4.3 SIP Information

Figure 4-4-4 SIP information



SIP Information

Port	SIP User ID	Register Status	Status	Port	SIP User ID	Register Status	Status
0	1000	Unregistered	onhook	1	1000	Unregistered	onhook
2	1000	Unregistered	onhook	3	1000	Unregistered	onhook
4	1000	Unregistered	onhook	5	1000	Unregistered	onhook
6	1000	Unregistered	onhook	7	1000	Unregistered	onhook

Displays registration status information with Softswitch platform or SIP Server

Table 4-4-3 Description of SIP information

Port	He corresponding number of GSM channel, DWG2001 has only 1 port.
SIP User ID	SIP registration account of the Softswitch and SIP server provided
SIP User ID	Shows the registration status of VoIP channel, including registered and unregistered.

4.5 Network Configuration

The navigation tree of the route mode and bridge mode as below:

Figure 4-5-1 bridge

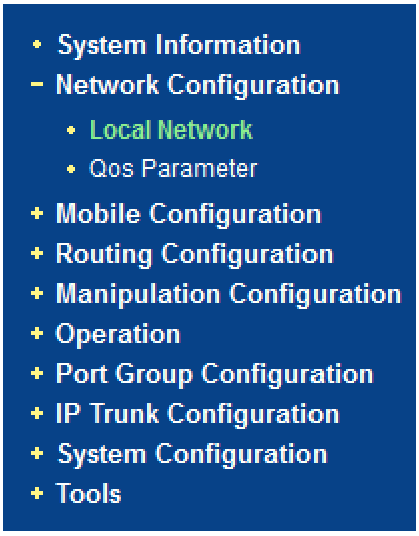
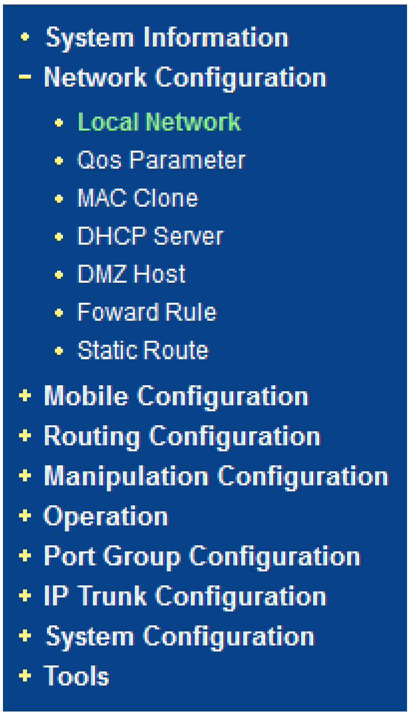


Figure 4-5-2 route mode



In the navigation tree of route mode, will have extra items of “MAC Clone”,”DHCP Server” ,”DMZ Host” ,”forward Rule”,”Static Route” .

4.5.1 Local Network

Under the route mode, WAN port connects with ADSL modem, and LAN port connects with local network. It will be used as a small switch when working in brigde mode. In this situation, user just need to configure the WAN parameter and DNS. User also need configure the LAN port if working in route mode. The web interface as bellows:

Figure 4-5-3 WEB interface of bridge mode

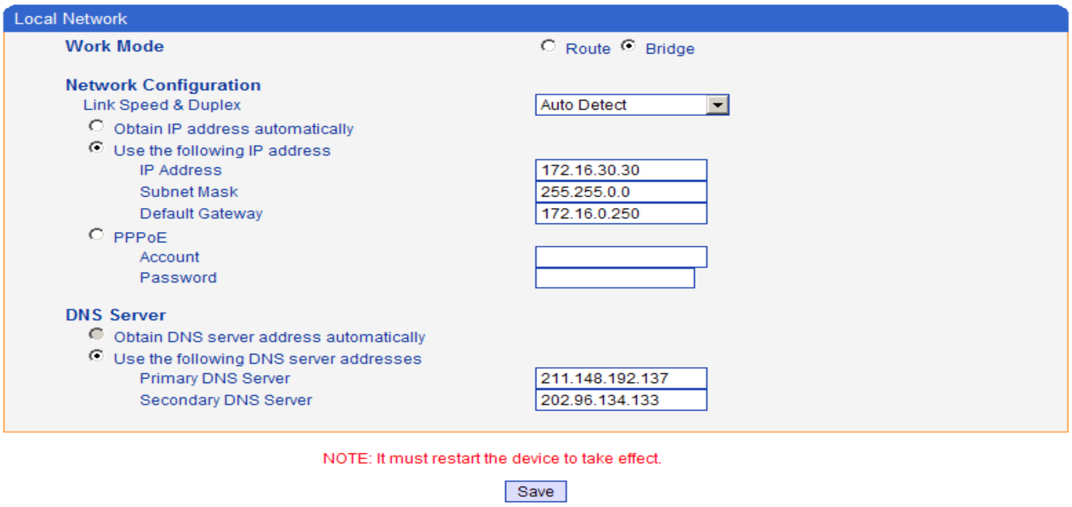


Figure 4-5-4 WEB interface of Route mode

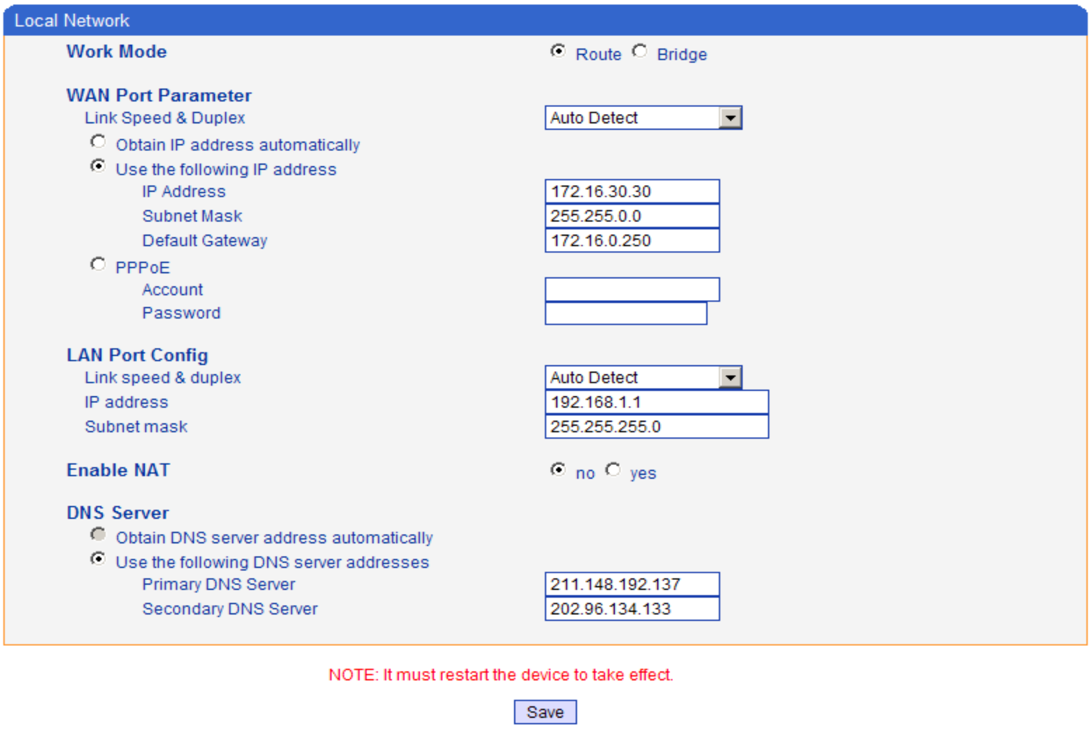
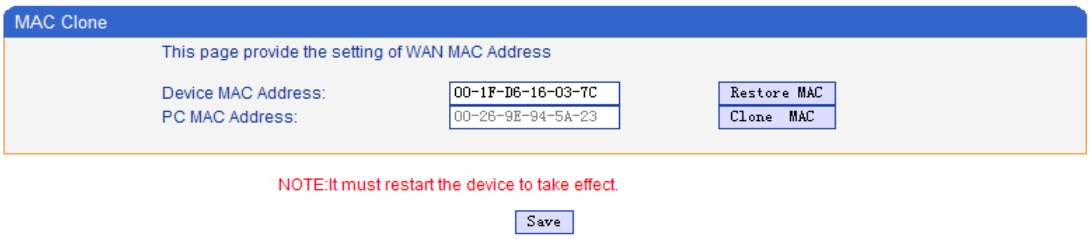


Table 4-5-1 Description of Local network

Work Mode	Two options of route mode and bridge mode, default is bridge mode
Link Speed & Duplex	The 5 options are “Auto Detect”,” 10Mbps/Half Duplex”,”10Mbps/Full Duplex”, “100Mbps/Half Duplex” and “100Mbps/Full Duplex”. Default is “Auto Detect”
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 have 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"
WAN /LAN Port	WAN port used for connecting the external network , LAN port used for connecting local network

4.5.2 MAC Clone

Figure4-5-5 Configuration of Mac Address Clone



This function can prevent the device against blocking by the carriers. Enable this function on the “router mode”, the device can be anti-blocked when the carrier to limit the online users by scanning the MAC address.

4.5.3 DHCP Server

Under “route mode”, DWG works as a router. Config DHCP serive to enable the DHCP service funcition of DWG, then DWG will works as a DHCP server.

Figure 4-5-6 Configuration of DHCP service

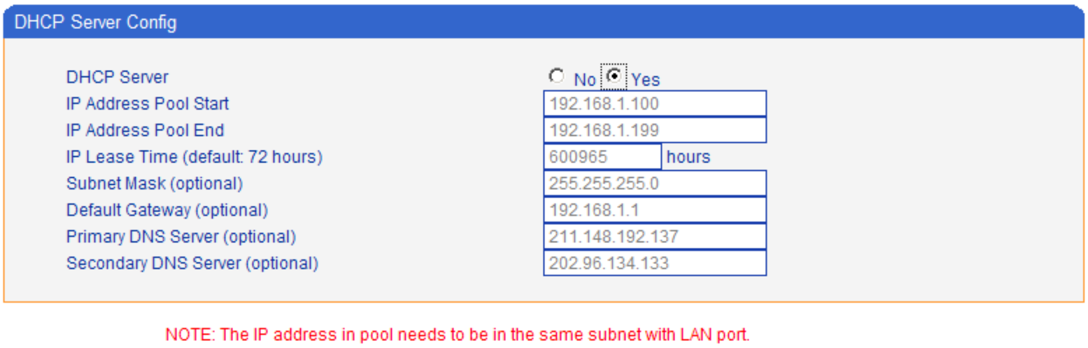


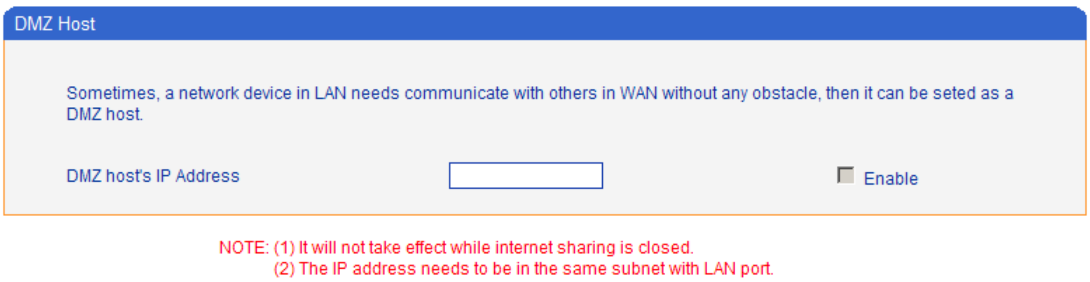
Table 4-5-2 Description of DHCP Server

IP address Pool	Determines the range of IP addess of other devices in this network
IP Lease Time	Sets the duration for how long the IP works with the specific IP. If the duration is out of the duration, the IP would be invalid
The subnet mask, gateway	DNS info will also be allocated to network devices automatically by DHCP protocol. Generally, there is no need to configure those items.

4.5.4 DMZ Host

In some conditions, certain devices in LAN network need to do two-way communication with WAN network(e.g. certain computer in LAN network need to provide multiple services to WAN network). In this situation, Configure this device as the DMZ host.

Figure 4-5-7 Configuration of DMZ Host



4.5.5 Forward Rules

In some conditions, certain devices in LAN network need to provide channel communication with WAN network(e.g. certain computer in LAN network need to provide FTP service of channel 21 to WAN network). In this situation, Configuring forwarding rules to this device is necessary.

The difference between forwarding rules and DMZ host is, DMZ host provides several consecutive channels and communication of all protocols, while forwarding rules provides single or several channels communication based on certain protocol(TCP or UDP). If DMZ host and forwarding rules have conflicts, will be determined by forwarding rules configurations.

Figure 4-5-8 Configuration of forwarding rules

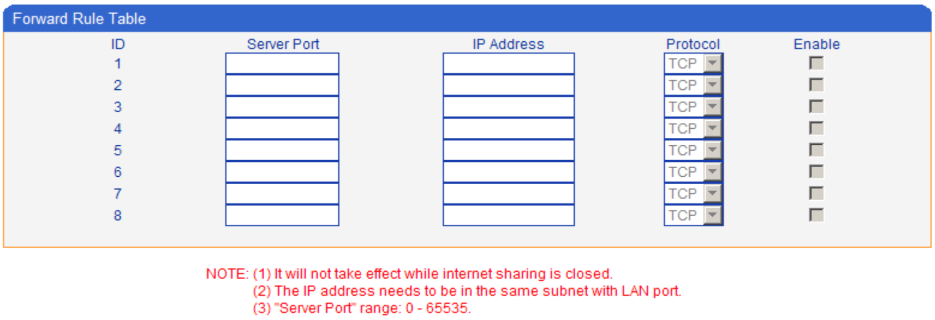


Table 4-5-3 Description of Forward rules

Service Port	The Service channel that should be provided to WAN network
IP address	IP address is the one of devices in LAN net work
Protocol	The service protocol(TCP or UDP)

4.5.6 Static Route

Static route is the route rules in IP communication. Generally speaking, no need to config static route. Configuring static route is necessary in such conditions: when several network segaments exist in LAN network and there’s certain application between these network segament. Please cancel “internet sharing” under “ Network configuration” first, then configure the “static route”.In commly use, please don’t configure static route. If static rule is wrong, the devices may not work.

Figure 4-5-9 Configuration of Static route

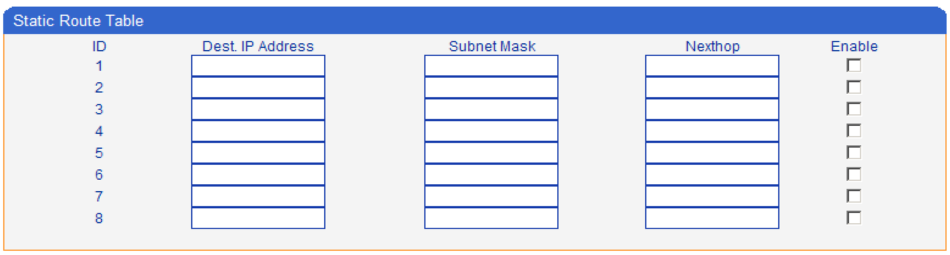
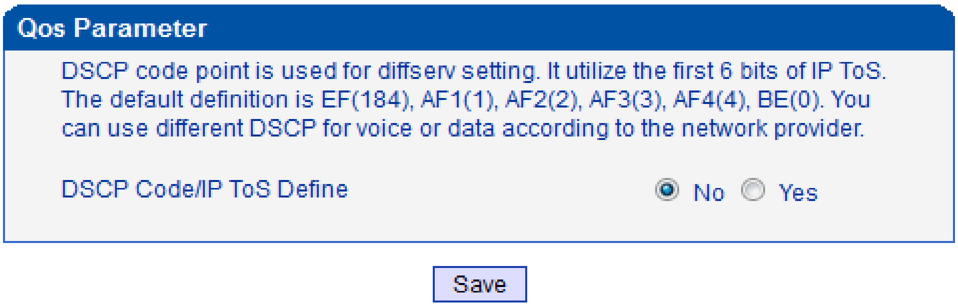


Table 4-5-4 Description of Static Route

Dest IP Address	Destination IP address the date packets will be sended
Subnet Mask	The Subnet Mask of Destination IP address
Nexthop	The Next Hop IP address if want to arrive the Destination IP addres

4.5.7 Qos Parameter

Figure 4-5-10 Qos parameter



If you want to use Qos,please select yes,and click save

4.6 Mobile Configuration

4.6.1 Basic Configuration

Figure 4-6-1Basic Configuration

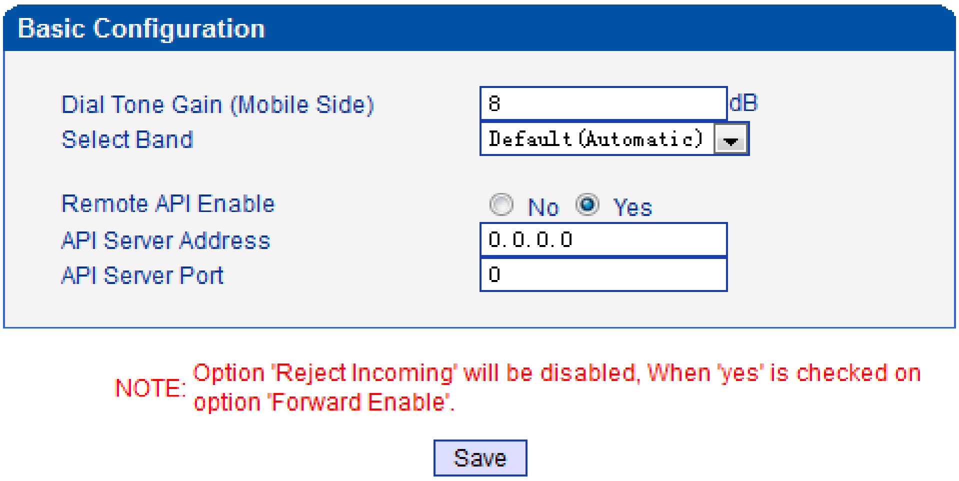


Table 4-6-1 Description of Basic Configuration

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"

4.6.2Mobile

Figure 4-6-2 Mobile Configuration

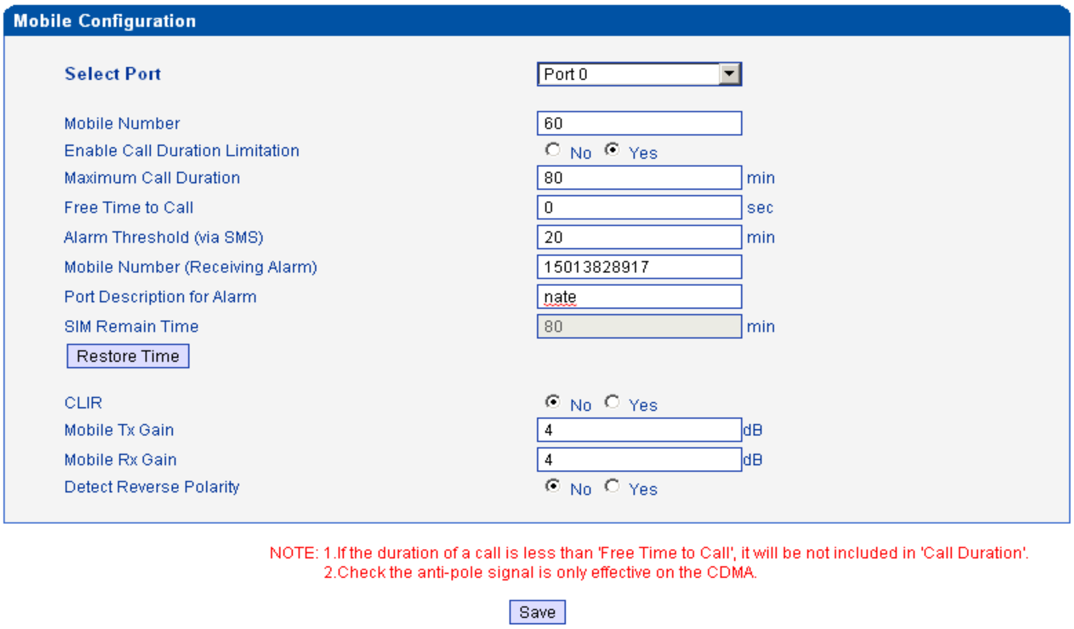


Table 4-6-2 Description of Mobile Configuration

Mobile Number	SIM card user ID of the channel. That must be configured when “one access No.” 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 initiative by 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	When the SIM remain time is or less than this value, DWG will send the alarm

（via SMS）	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 designated 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, DWG2000-8G 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

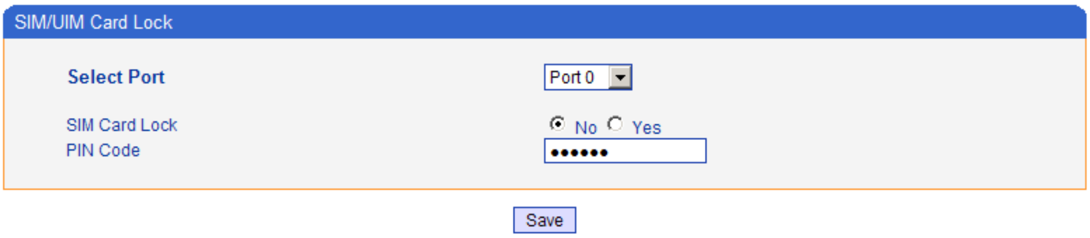
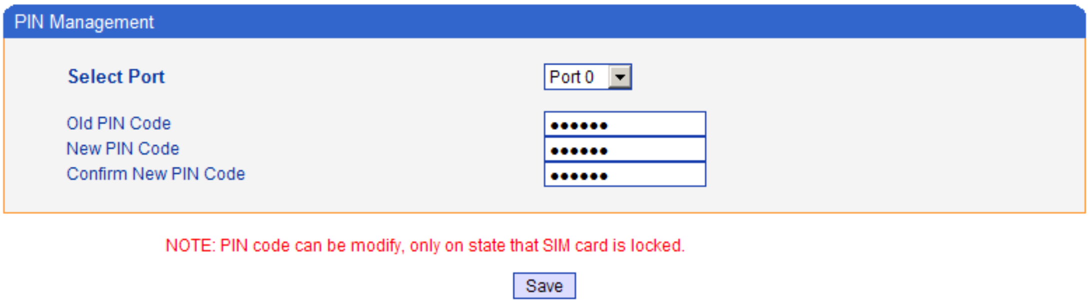


Table 4-6-3 Description of Configuration of SIM/UIM Card Lock

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



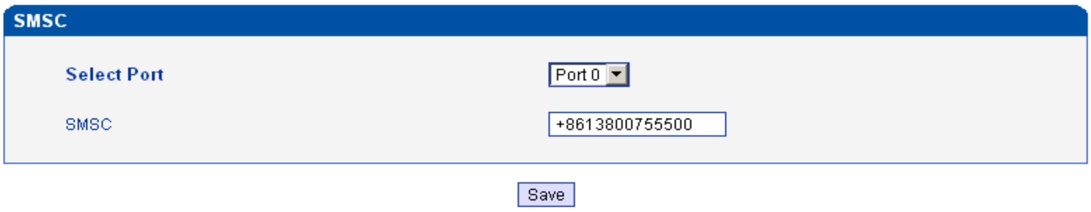
Detailed description as below:

Table 4-6-4 Description of PIN Management

PIN	PIN is the password of SIM card personal identification. 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, theoretically, 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

Configurations are as below:

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. USSD is similar with Short Messaging Service (SMS), but unlike SMS. USSD transactions occur during the session only. With SMS, messages can be sent to a mobile phone and stored for several days if the phone is not activated or within range.

Figure 4-6-7 USSD

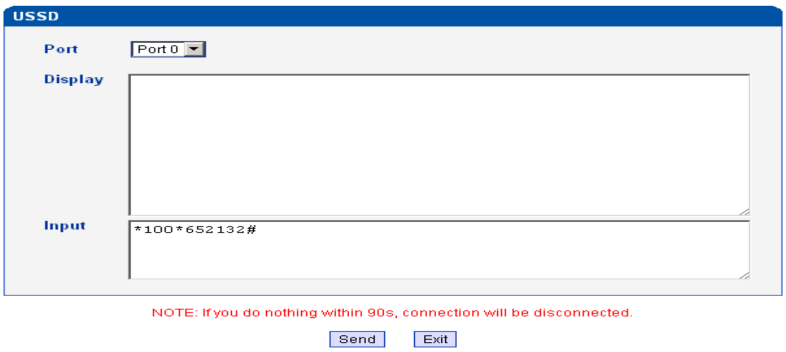
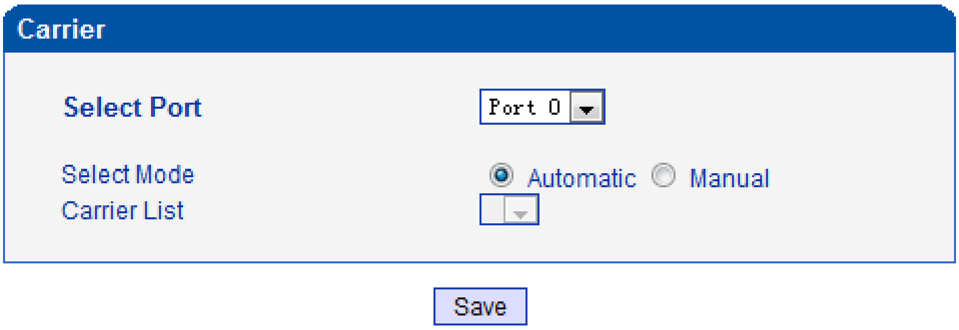


Table 4-6-6 Description of USSD

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

4.6.8 Carrier

Figure 4-6-8select 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

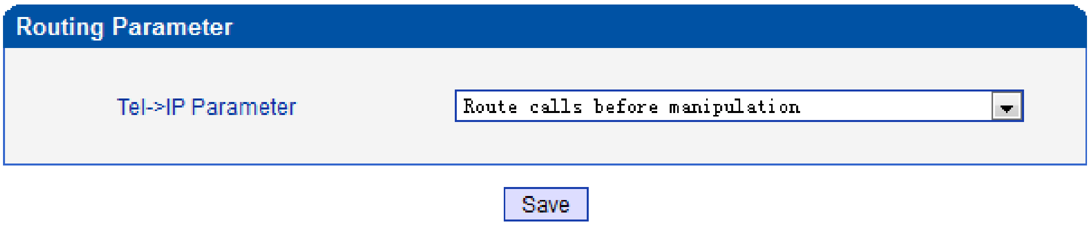


Table 4-7-1Description of Routing Parameter

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 Tel->IP Routing

Figure 4-7-2 Tel to IP Routing

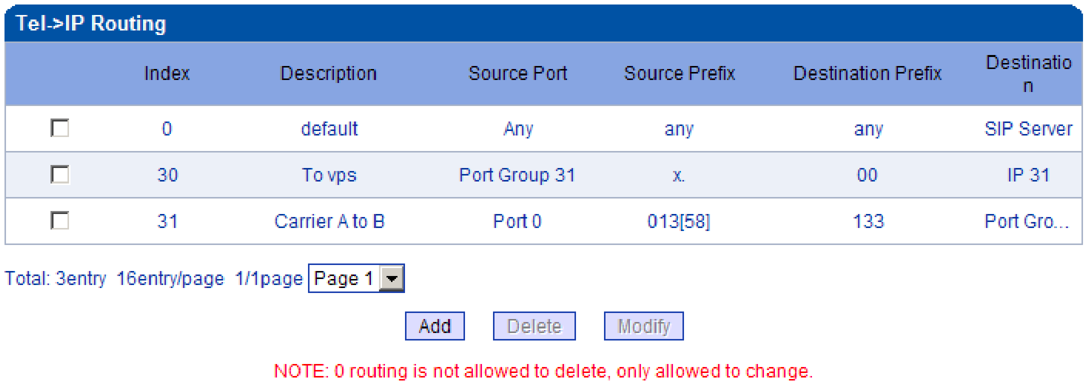
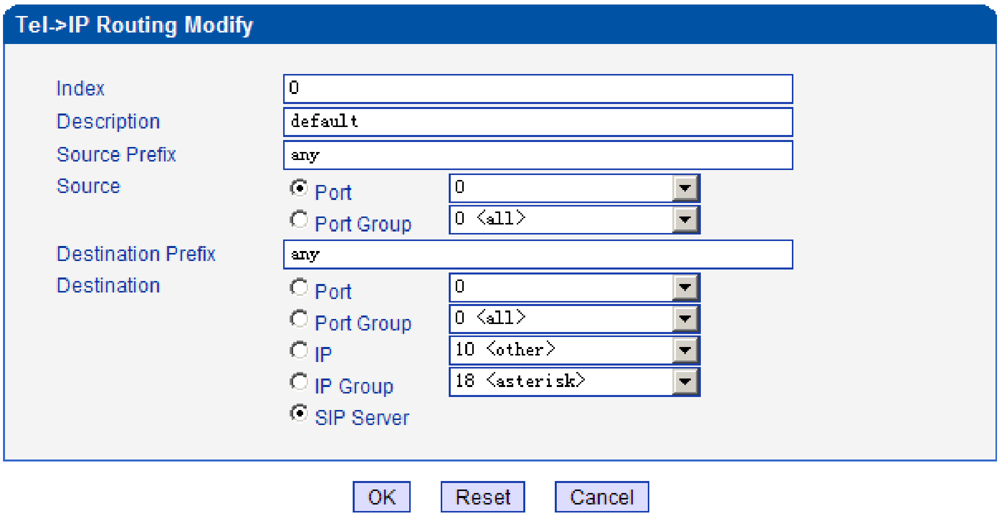


Table 4-7-2 Description of Tel to IP Routing

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.
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

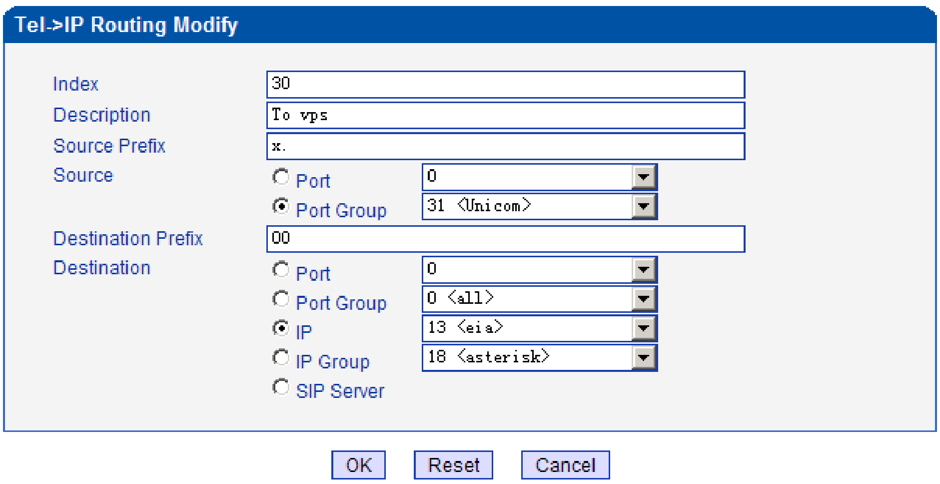
Source Prefix	All the caller number must match the source prefix. It specifies the source prefix allow to send call out • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186
Destination Prefix	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

Figure 4-7-3 Tel to IP routing Modify



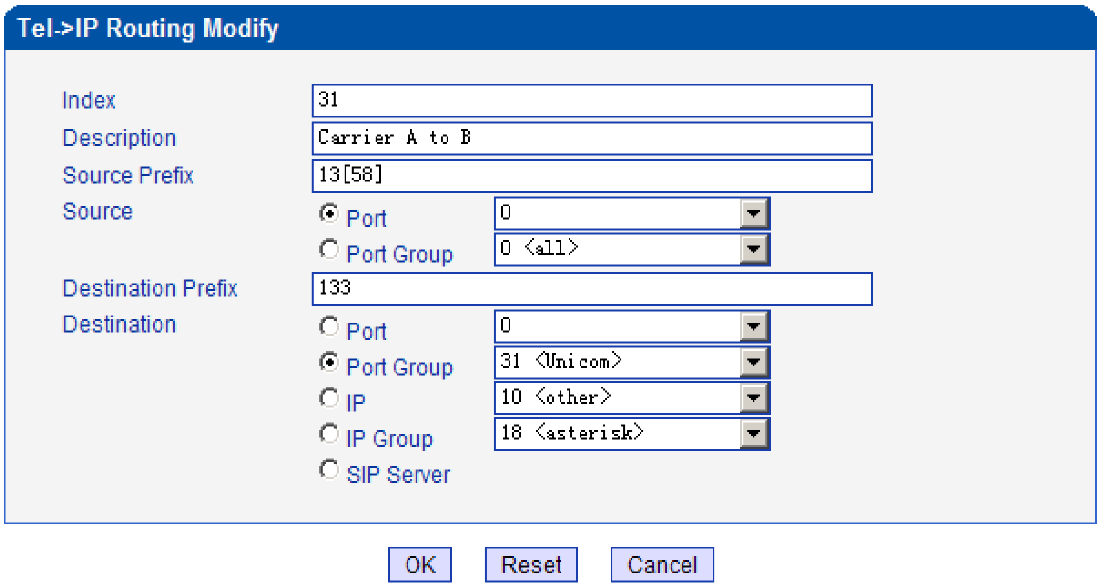
It's a default route configured in gateway. It allows any number from source port 0 send call to SIP server with any prefix.

Figure 4-7-3 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 4-7-3 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

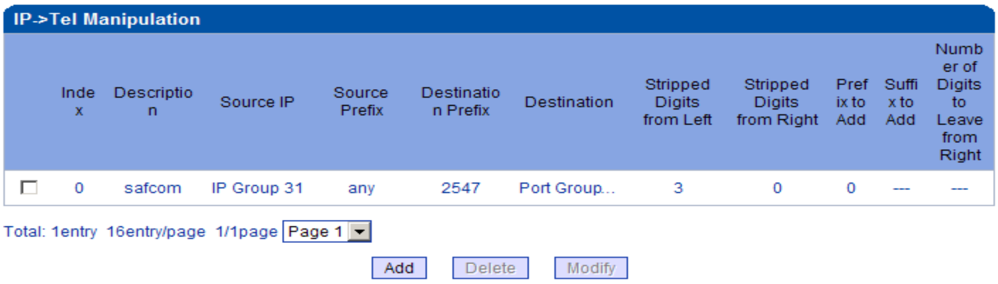
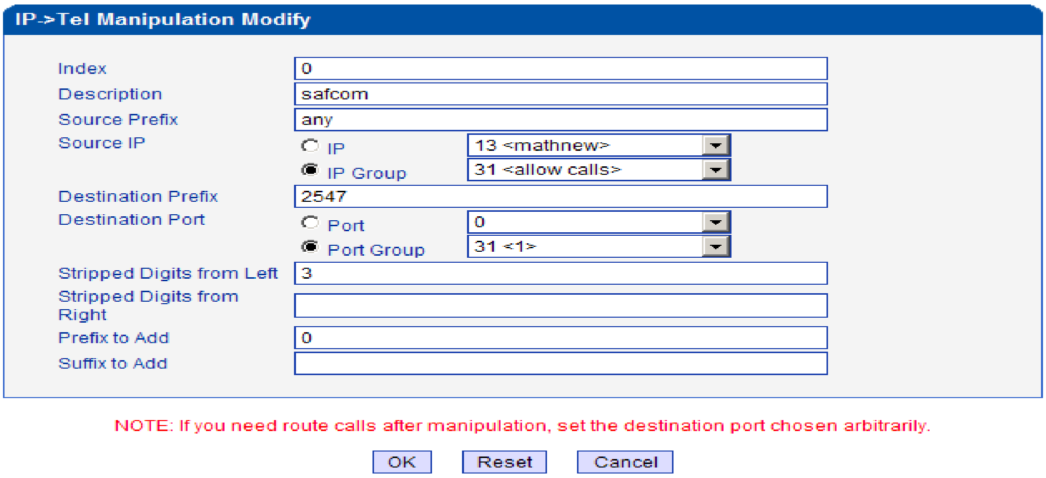


Table 4-8-1 Description of IP->Tel destination numbers manipulation

IP->Tel destination numbers manipulation	It is an optional configuration item, and is used to add 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 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

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 indecates that calls coming from IP Group will match the prefix ”any”, and the called nubmer whom 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

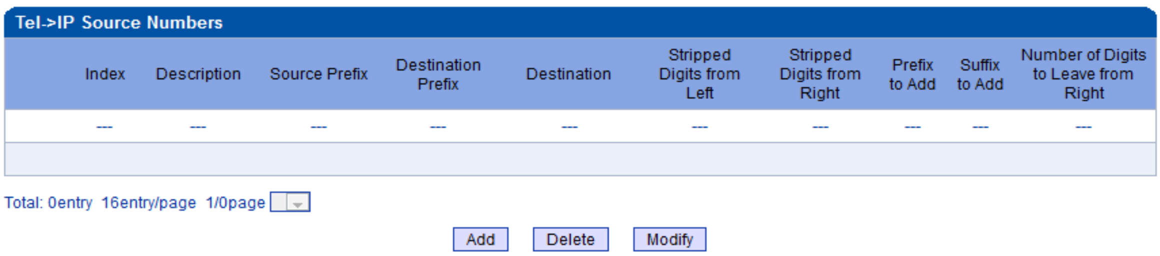


Table 4-8-2 Description of Tel->IP destination numbers manipulation

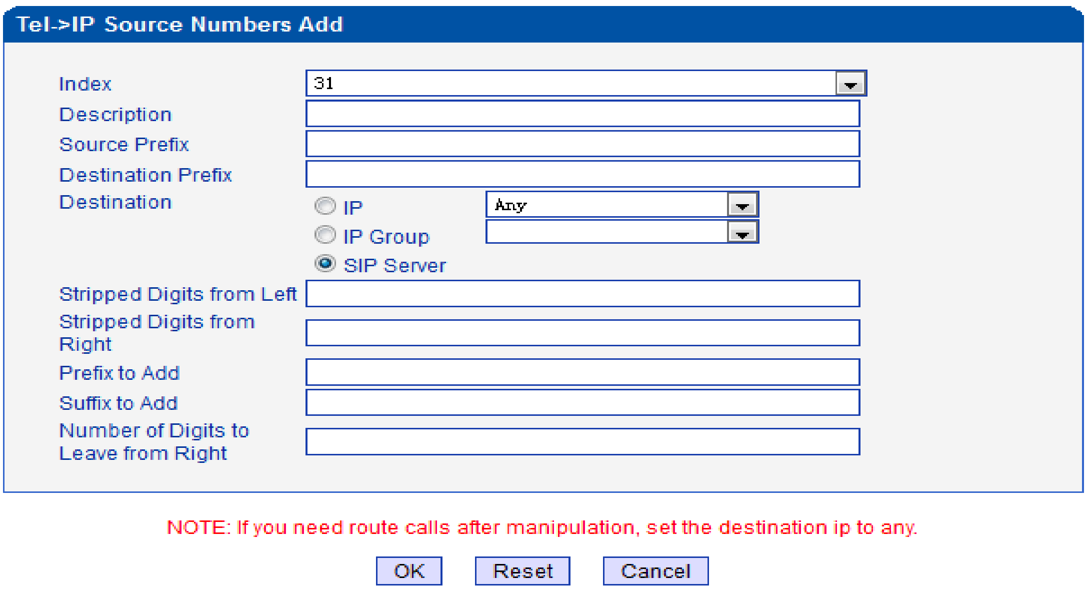
Tel->IP destination numbers manipulation	It is an optional configuration item, and is used to add IP->Tel number change data. The IP->Tel Manipulation defined the rules of add, and deletion of called numbers, which are referenced by IP->Tel routing.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.
Description	It describes the 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 4-8-4 Tel ->IP destination numbers manipulation add



It indecates that calls coming from IP Group will match the prefix ”any”, and the called nubmer whom 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

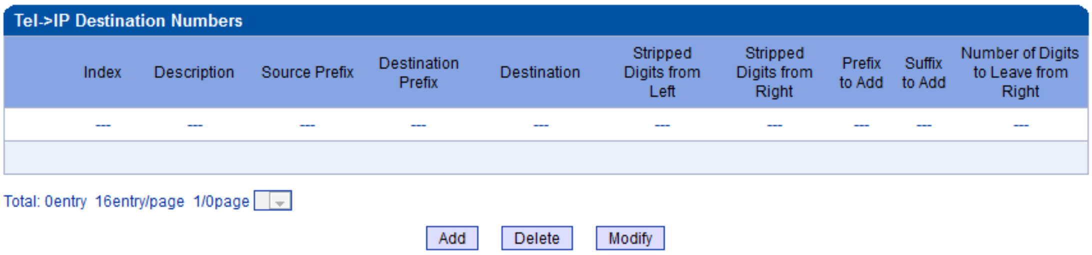


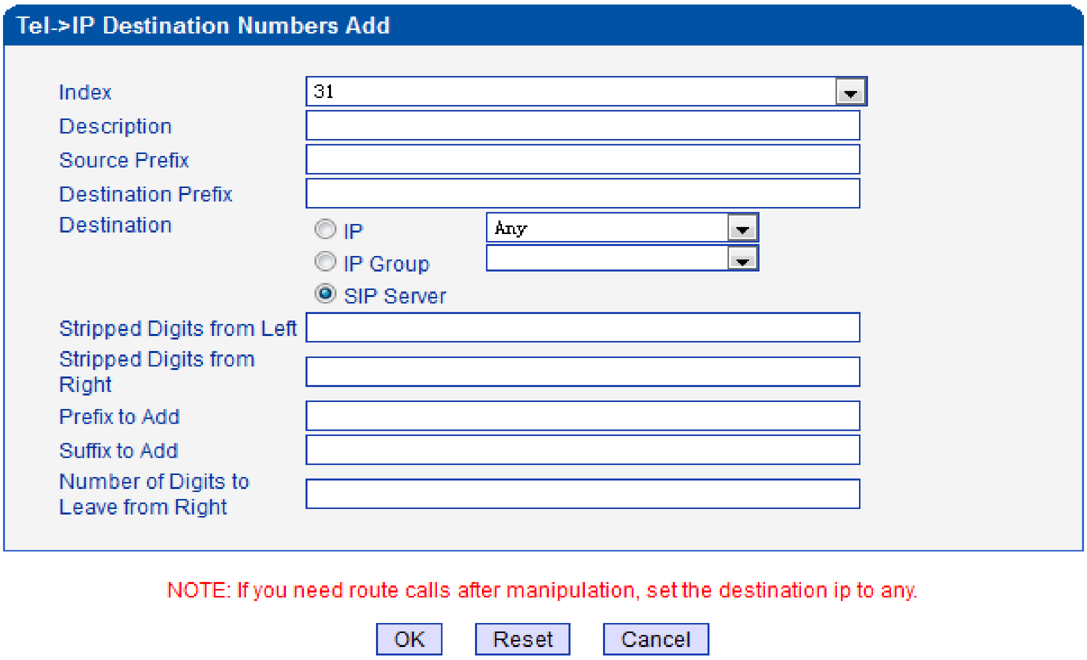
Table 4-8-3 Description of Tel->IP destination numbers manipulation

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 indecates that calls coming from IP Group will match the prefix ”any”, and the called nubmer whom 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

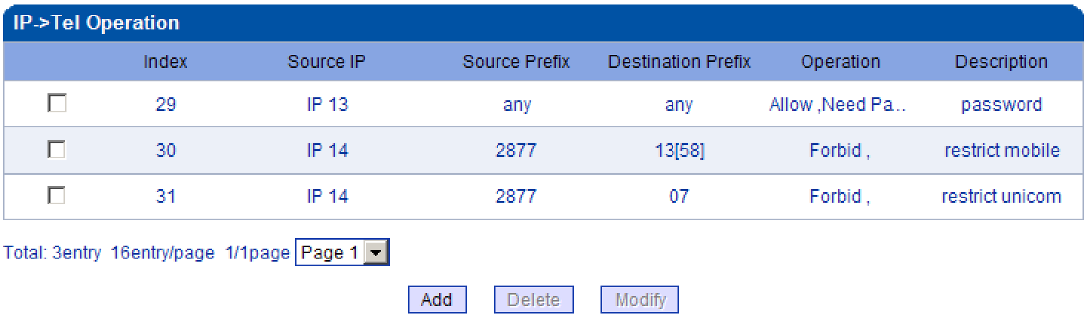


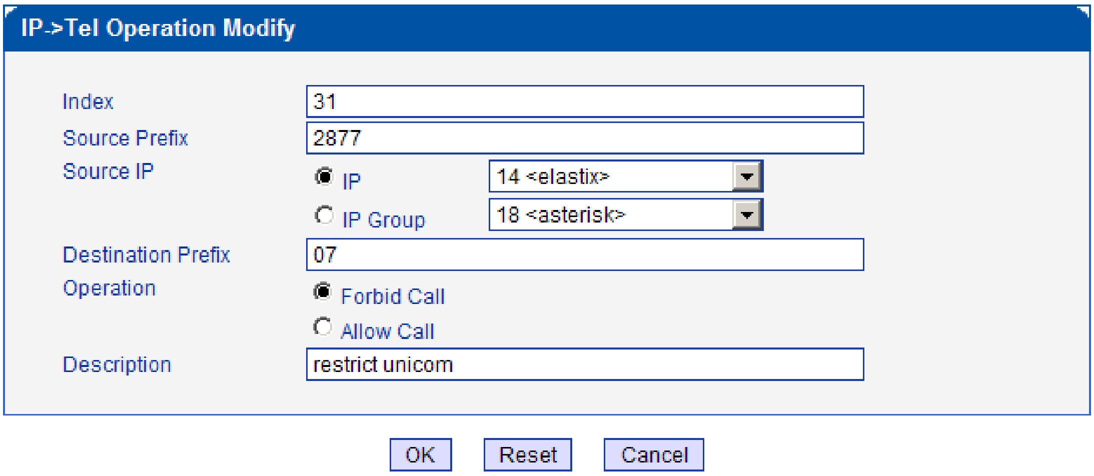
Table 4-9-1 Description of IP->Tel Operation

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>

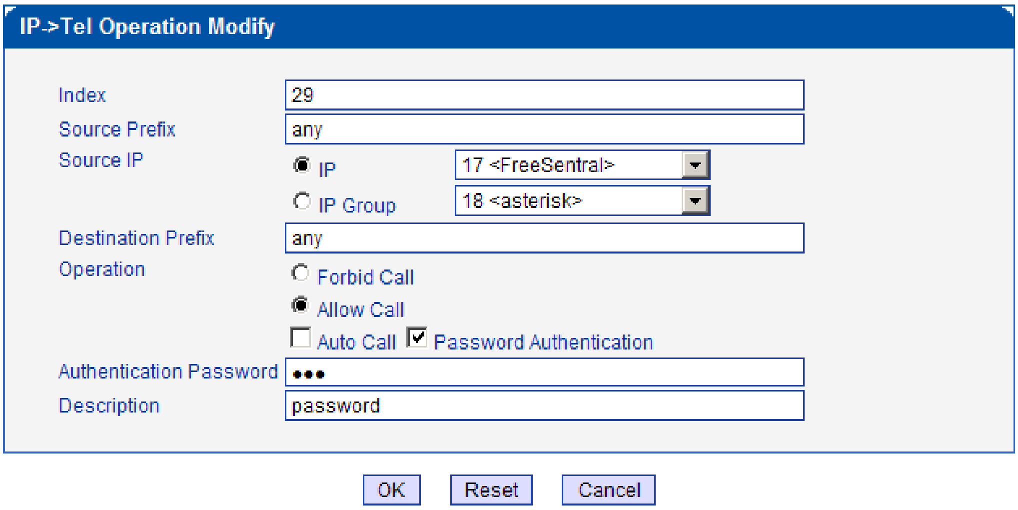
Figure 4-9-2 IP->Tel Operation Modify



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

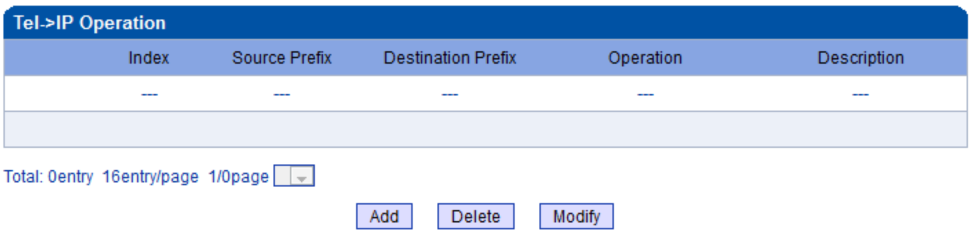


Table 4-9-2 Description of Tel->IP Operation

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 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

4.10 IP Trunk Configuration

4.10.1 IP Trunk

Figure 4-10-1 IP Trunk

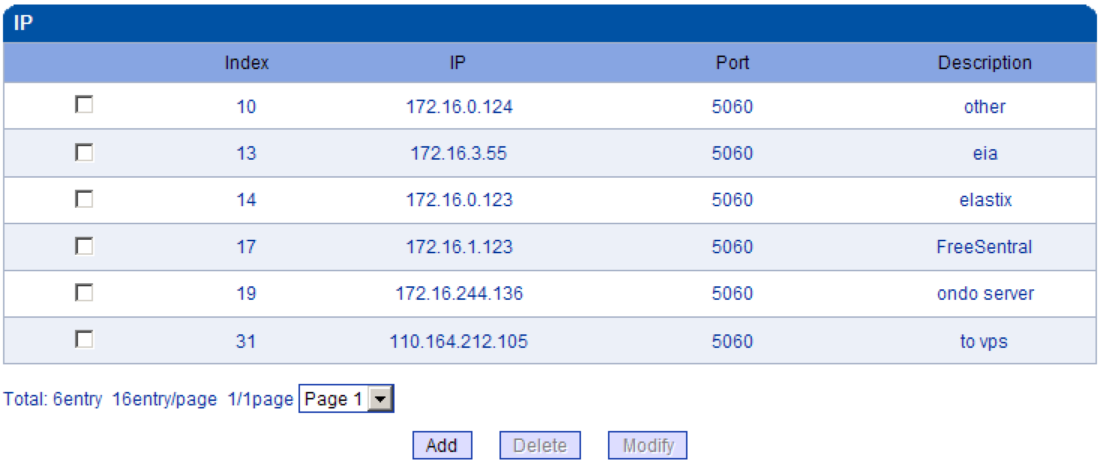


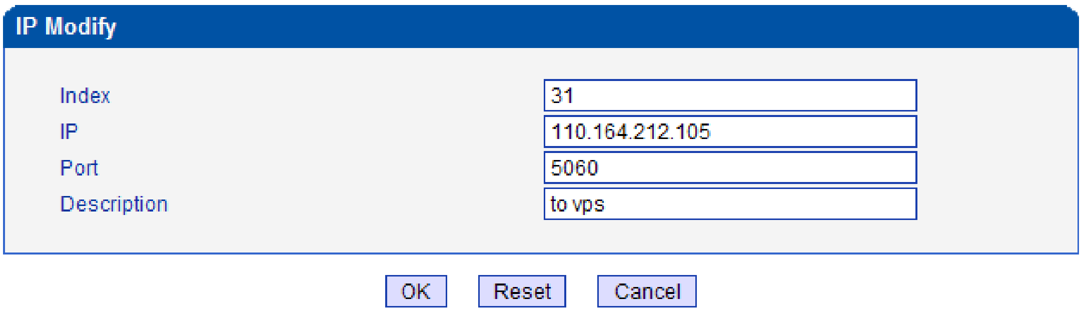
Table 4-10-1 Description of IP Trunk

IP Trunk	Add remote IP of softswitch, SIP server which will send call traffics to gateway.
Index	It uniquely identifies a 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 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 4-10-2 IP Trunk Modify



4.10.2 IP Trunk Group

Figure 4-10-3 IP Trunk Group

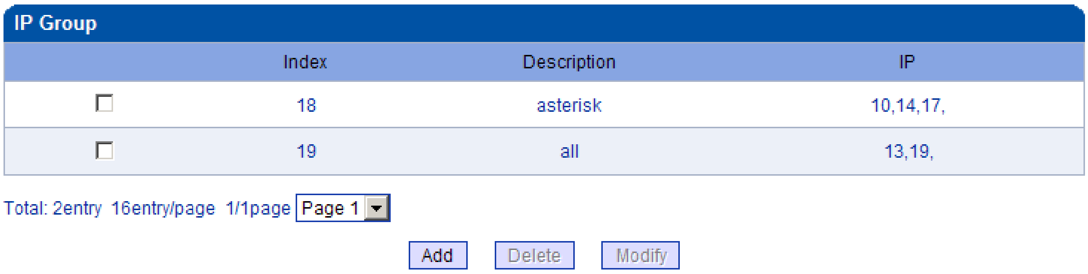


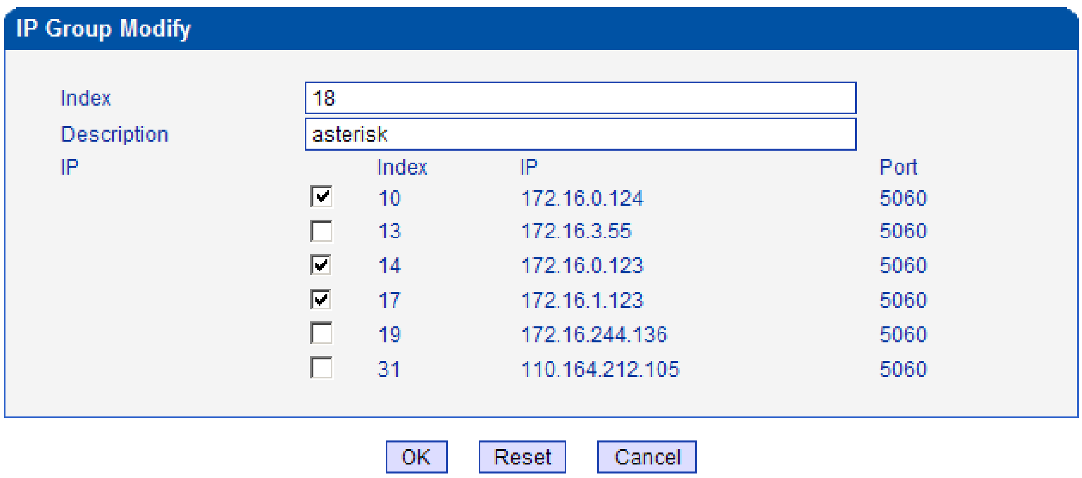
Table 4-10-2 Description of IP Trunk Group

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 4-10-4 IP Trunk group modify



4.11 System Configuration

4.11.1 System Configuration

Figure 4-11-1 System Configuration

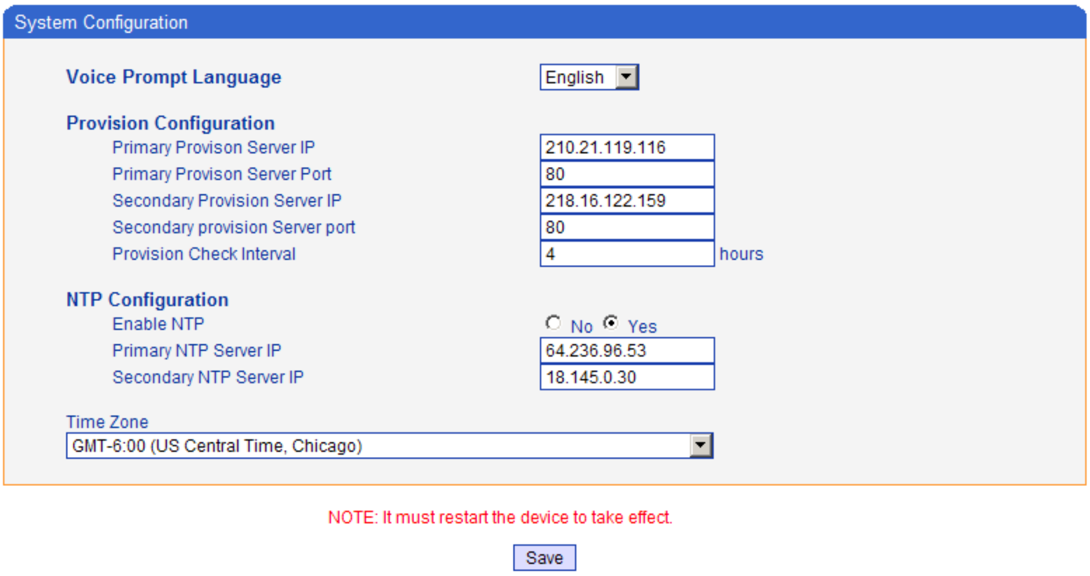


Table 4-11-1 Description of System Configuration

Voice Prompt Language	Configure the voice prompt of DWG., e.g. configure voice prompt of IP address success or failure. DWG supports English and Chinese. Users can customize other languages. The default setting is in Chinese.
Provision configuration	Provision is used to maintain the devices. E.g. Provision can config, update and remote manage the devices in bulk.
Primary Provision Server IP	This is provided by carrier. Keep the default value if carrier don't provide this value.
Primary Provision Server Port	This is provided by carrier. Default is 80.
Secondary Provision Server IP	This is provided by carrier. Keep the default value if carrier don't provide this value.
Secondary Provision Server Port	This is provided by carrier. Default is 80.

Provision Check Interval	Default is 4 hours.
Enable NTP	NTP enable switch
Primary NTP Server IP	Can keep the default
Secondary Provision NTP Server IP	Can keep the default
Time Zone	The default is GMT +8:00, the user can adjusted accordingly according to their area

4.11.2 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 4-11-2 Service Configuration

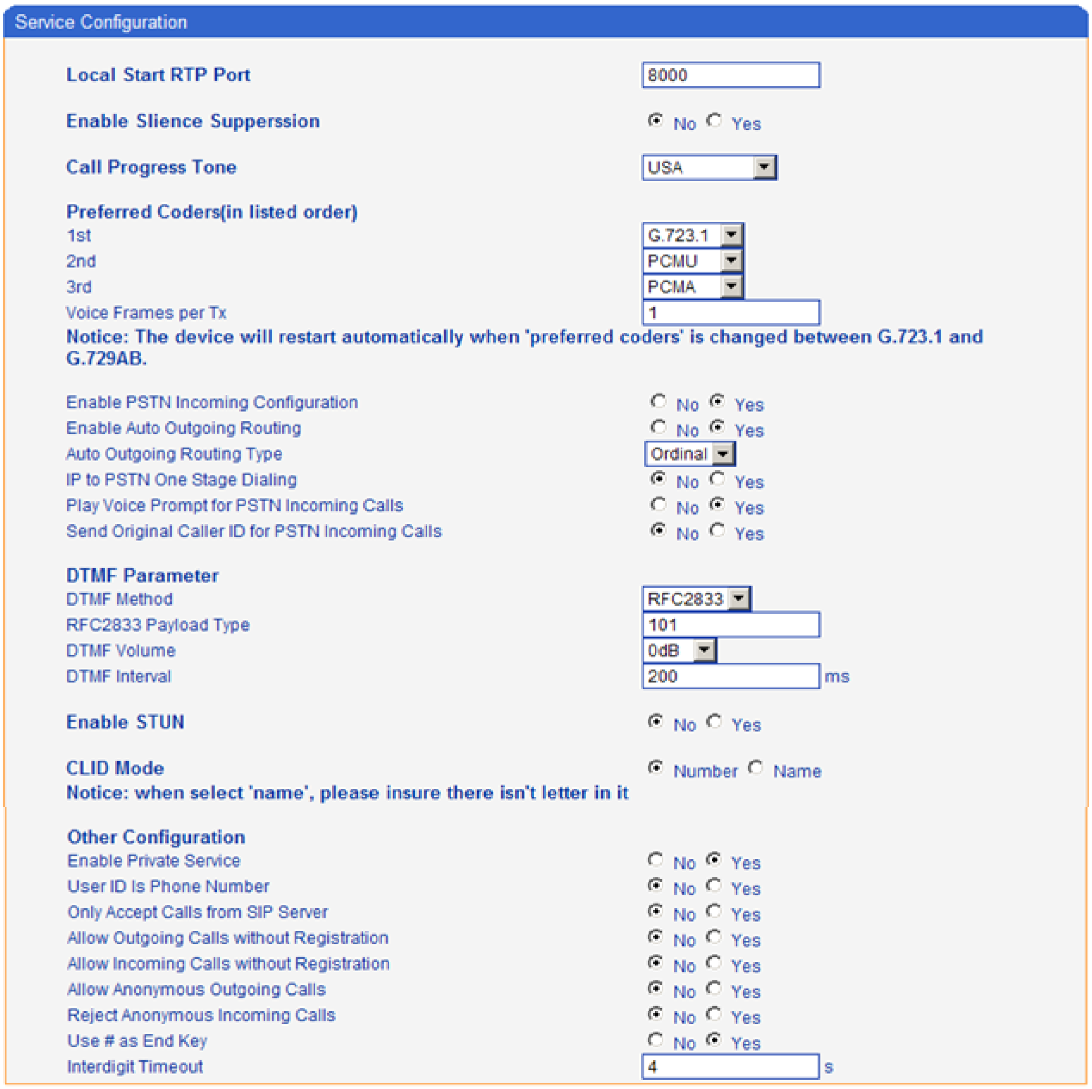


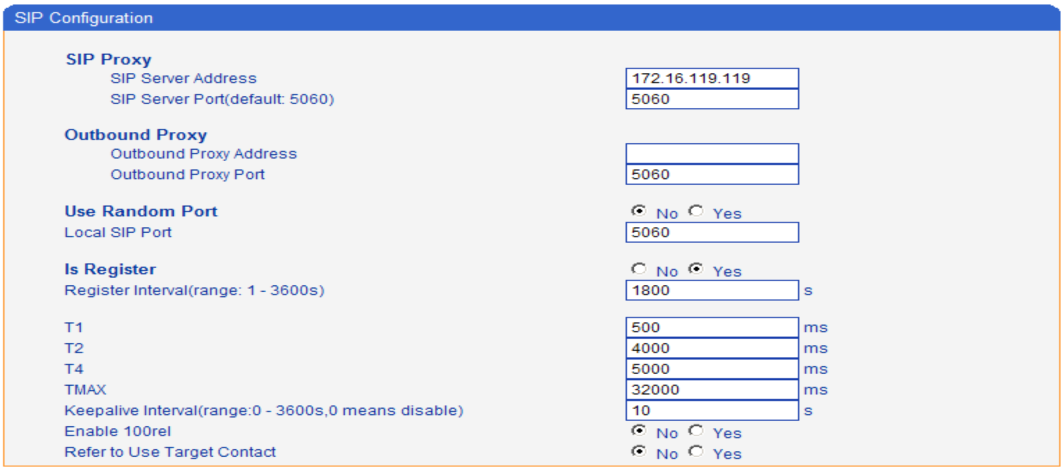
Table 4-11-2 Description of Service Configuration

LOCAL RTP PORT Channel	Means the initial allocation of Channel 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. Note: when the preferred codec switch between G.723.1 and G.729AB , System will automatically reset
Enable PSTN Incoming Configuration	Means when call from PSTN side, you can dial the function keys for check number, setting IP and so on function
Enable Auto Outgoing Routing	Means when call out , whether by ordinal or polling pick to Select a Channel, this feature are generally used for when use the same SIP User ID to register or use as trunking mode
IP to PSTN One Stage Dialing	This function will be displayed only when select "Enable Auto Outgoing Routing" function, 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 with dial tone
Send Original Caller ID for PSTN Incoming	For Example, the phone A from PSTN side call DWG2001/DWG2000-4G/DWG2000-8G Channel SIM card corresponding User ID, the Channel's SIP User ID is C, Channel hook and then call B, when "Send Original Caller ID for PSTN Incoming" setting

	is Yes, the Caller User ID that send to B will be A, when "Send Original Caller ID for PSTN Incoming" setting is No, the caller User ID that sent to B will be C(except for anonymous outgoing)
DTMF	DWG2001/DWG2000-4G/DWG2000-8G support RFC2833 and SIGNAL two ways. DTMF INTERVAL range is 50 ~ 800ms, DTMF VOLUME can use the default Configuration
Enable STUN	（Simple Traversal of UDP over NATs，NAT's UDP simple cross) 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
CLID Mode	Select the name under the special needs , the most common way is use the default User ID
Allow Outgoing Calls 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 Incoming Calls 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
Allow Anonymous Outgoing Calls	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
Inter digit Timeout	Bit of between the dialing time ,over the time will be seem as end of dial

4.11.3 SIP Configuration

Figure 4-11-3 SIP Configuration



SIP Configuration	Used for Configuring VoIP channel, add SIP Registry Platform and local SIP Channel, and configure SIP protocol and other related information
SIP Server Address	Used for configure SIP server address and Channel, the address can be IP Address, also can be a domain name (DNS should to be able to resolution), the details please advisory the service provider
SIP Proxy Port	Port default setting is 5060. For details, please consult the service provider
Outbound Proxy Address	Outbound proxy, it mainly used in firewall / NAT environment. That make the signaling and media streams are able to penetrate the firewall, the details please advisory the service provider
Outbound Proxy Port	Outbound proxy port number, the details please advisory the service provider
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 Monitor
Is Register	Default set yes, if you want the device can make a call without register, set No, Also enable the "Allow Outgoing Calls without Registration" and "Allow Incoming Calls without Registration" function
Register Interval	Means how often the equipment will register once to the SIP server/proxy
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 in the best available Registered. In general, using the factory default values

4.11.4 Port Configuration

Port Configuration is used to configure ports' gain, Off-hook Auto-Dial, etc.

Figure 4-11-4 Port Configuration

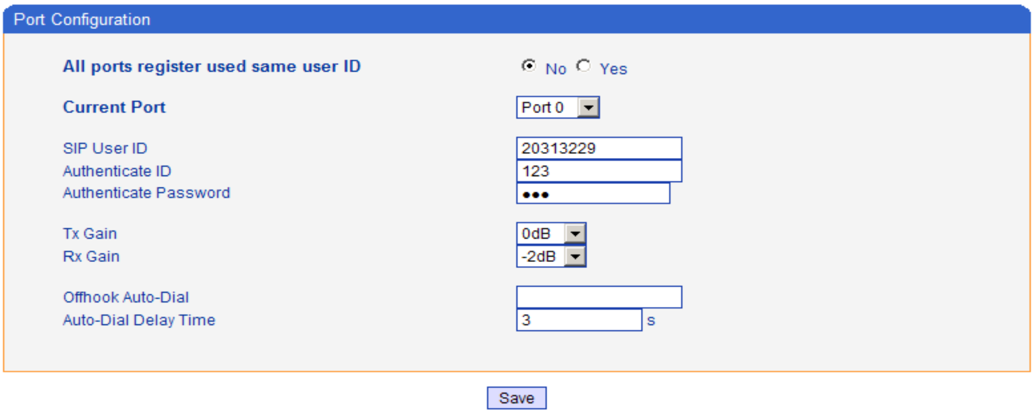
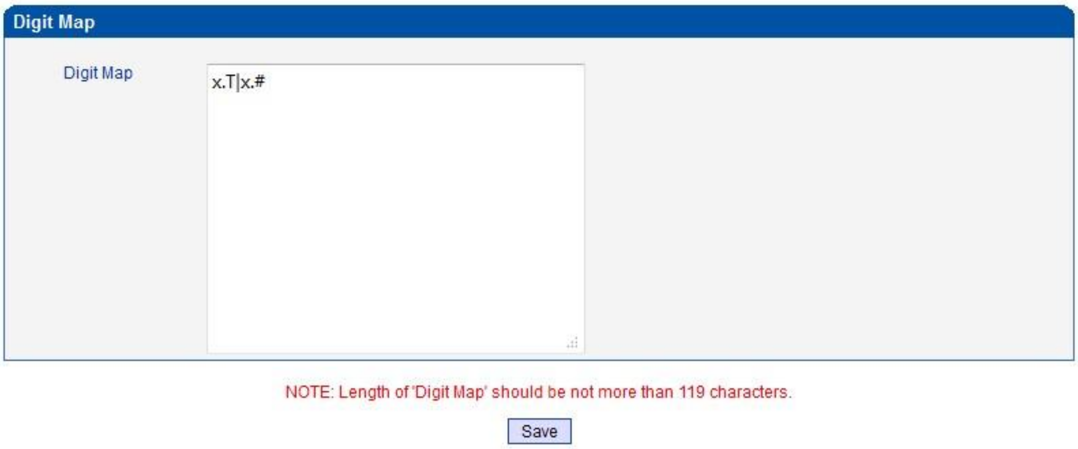


Table 4-11-3Description of Port Configuration

Port Configuration	Used to configure ports' gain, Off-hook Auto-Dial, etc.
ALL ports register used same user ID	The default is not, each SIP account, if set yes ,all the port will use user ID of port0, when the call ring in sequence
SIP User ID	Is the account used for registration, equipment port's unique identifier, "Authenticate ID" is equivalent to show the name, "Password" is register Password, which no passward can no fill, the details please contact the service provider
Tx Gain	Refers to the call volume that from himself during a call to the end users, adjust the "Tx Gain" will affect the voice volume of the end user, the default value is 0
Rx Gain	Refer to the call volume from the remote end user to ourself volume, adjust the "gain acceptance" will affect the voice volume we will heard, the default value is 0.
Offhook Auto-Dial	Hotline service.when PSTN part client calls to this port,will auto forward to the hotline User ID. If no need this feature, just left it blank
Auto-Dial Delay Time	Offhook Auto-Dial delay time, the range is 0-10 seconds

4.11.5 Digit Map

Figure 4-11-5 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.

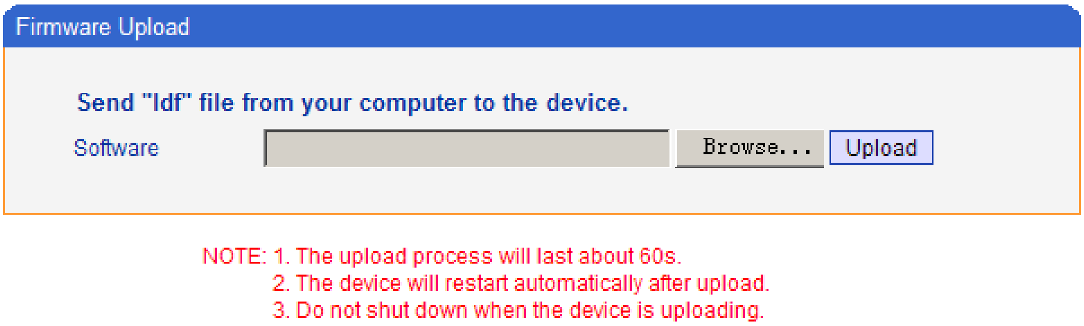
4. [1-357-9]xx
   Means that first is "1","2","3" or "5" or "7","8","9", followed by 2 digits.

4.12 Tools

4.12.1 Firmware Upload

Equipment upgrades can through Dinstar's softswitch platform, when the device disconnects with Dinstar's softswitch platform or some special circumstances. The firmware is able to upload locally.

Figure 4-12-1 Firmware upload



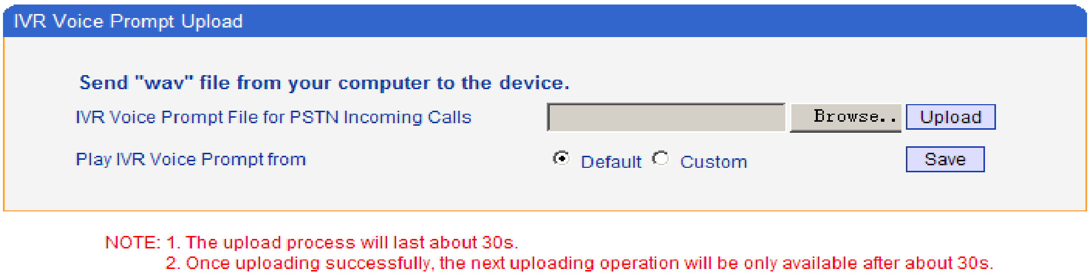
Select the upgrade program under correct directory services, and then click upload will complete upgrade the firmware.

NOTE: during the upgrade process, please do not swtich off the power supply, equipment may paralyze.

4.12.2 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.

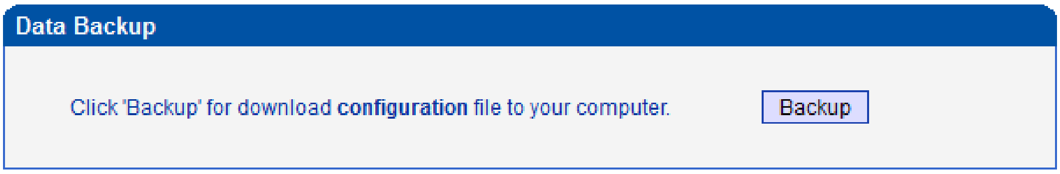
Figure 4-12-2 IVR Voice Prompt Upload



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

4.12.3 Data Backup

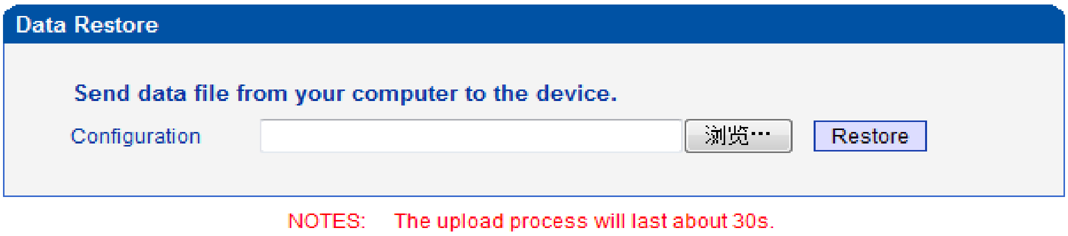
Figure 4-12-3 Data backup



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

4.12.4 Data Restore

Figure 4-12-4 Data restore



Send data file from your computer to the device

4.12.5 Syslog Parameter

Figure 4-12-5 Syslog parameter

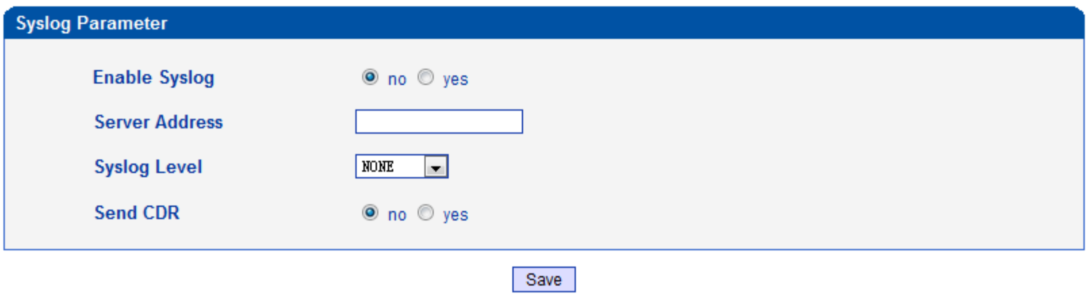
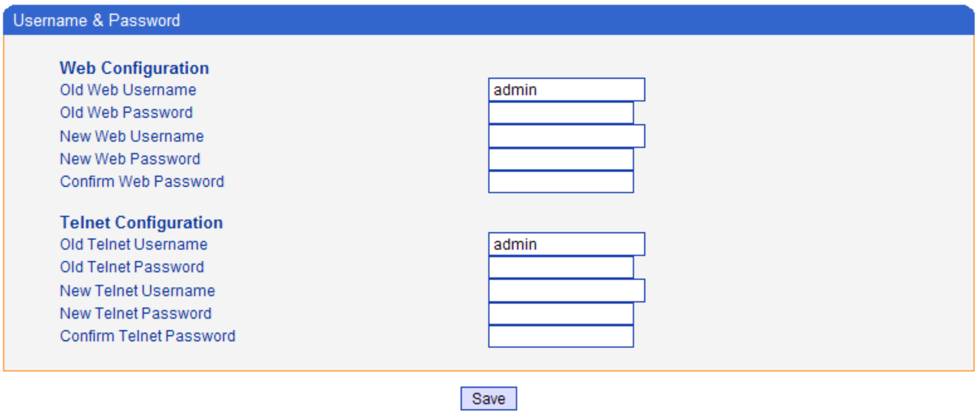


Table 4-12-1 Description of Syslog Parameter

Enable SysLog	select yes to enable syslog client function
Server Address	Fill in the Syslog server IP Address here
Syslog Level	There are five level of syslog:NONE、DEBUG、NOTICE、WARNING、ERROR, we urge you to select DEBUG.
Send CDR	If you select yes, DWG will send CDR to syslog server.

4.12.6 Login Password

Figure 4-12-6 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.

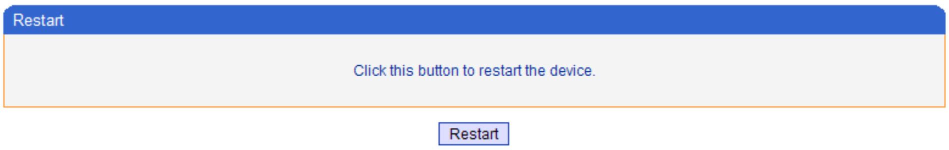
4.12.7 Factory Reset

Figure 4-12-7 Factory Reset

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

4.12.8 Restart

Figure 4-12-8 Restart



When system restarts, user click RESET button on the web.

5. FAQ

5.1 Device have been connected to network physically, but the network cannot be connected or network communication is not normal

1) Make sure the network cable is ok or not , can through view the device WAN port or LAN 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) 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, reference WEB Configuration Interface Description section);
4) 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;
5) 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 when calling in, the caller always hears a busy tone

Make sure Enable DND(Do-not-Disturb) in system

5.5 sudden interruption during a call

1) make sure whether is human error caused the problem
2) Make sure with the account balance or lack of disruption caused the call disconnected
3) Make sure whether there is interference with the fax tone or equipment busy tone, these interference may lead to calls dropped
4) Make sure whether the LAN equipment such as gateway or router fails, user can try to restart the gateway or router

5.6 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