# DWG2000B-16G User Manual V2.0


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# Revision Records

| | |
|---|---|
| Document version | 2.0 |
| Firmware version | 2.22.01.03 |
| Revised by | Technical Support Department |
| Date | 11/06/2012 |
| Changes | The first version |

# Table of Contents

# 1. Equipment Introduction

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

## 1.1 Introduction

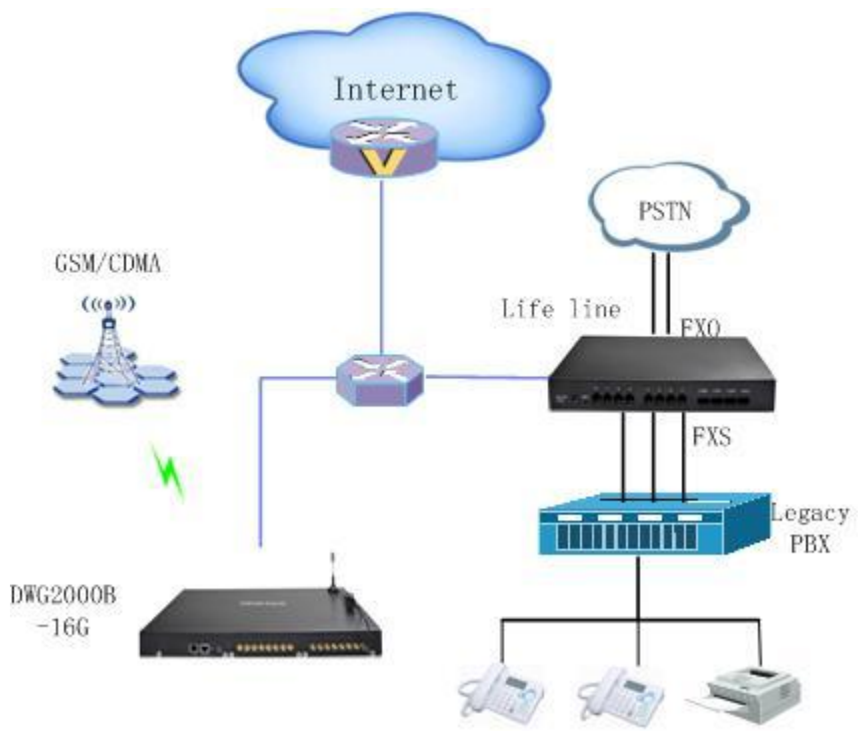
DWG2000B-16G is full functions VoIP gateway based on IP and GSM network, which provides a flexible network configuration, powerful features, and good voice quality. It works for carrier grade, enterprise, SOHO, residential users for cost-effective solution.

## 1.2 Scenario of Applications of Products

DWG2000B-16G provides access of GSM network.

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

Figure 1-2-1 Network scenario



## 1.3 Product Appearance

The appearance of DWG2000B-16G shows as follow

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

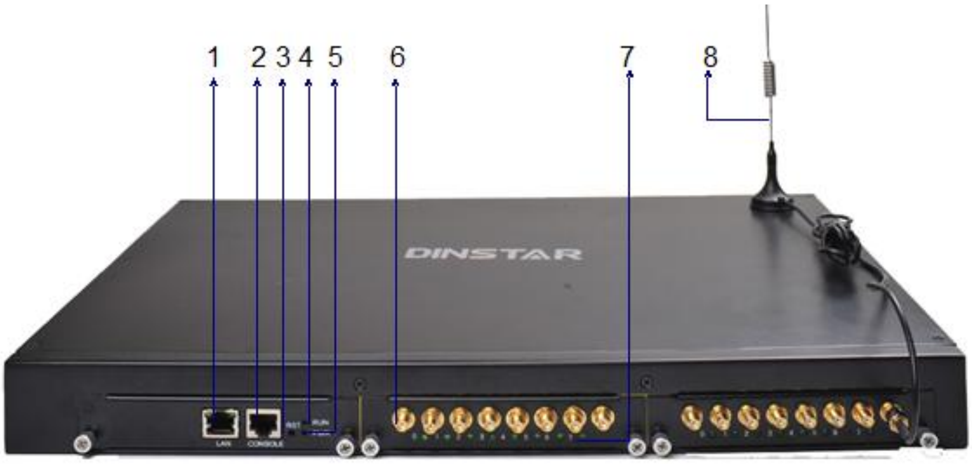



Table 1-3-1 Description of Front view

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

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

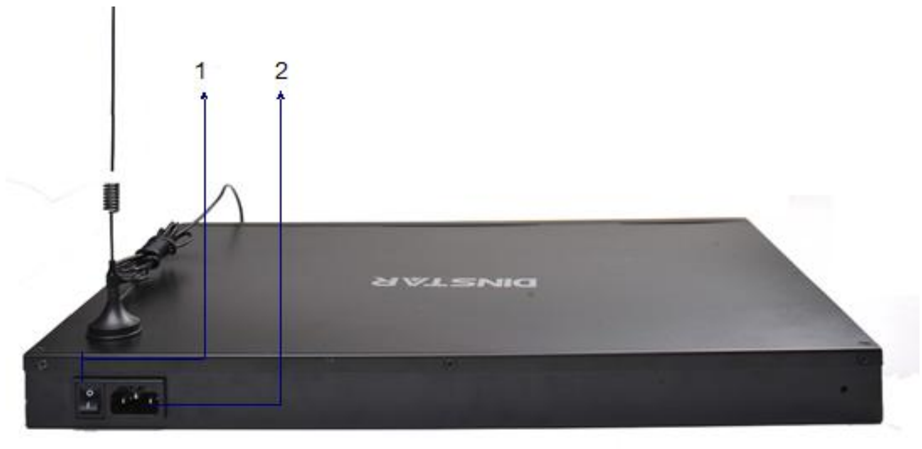



Table 1-3-2 Description of Rear view

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

## 1.4 Functions and Features

### 1.4.1 Protocol Standard Supported

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

### 1.4.2 System Function

- PLC: Packet loss concealment
- VAD: Voice activity detection
- CNG: Comfort Noise Generation
- Local/Remote SIM card work mode
- Adjustable gain of port
- DTMF adjustment
- Balance alarm
- Lock/unlock SIM/UIM
- Mobile number display rejection
- Sending/receiving SMS
- Customize IVR Recording

- White and black list
- One number access
- Open API for SMS, support USSD
- Echo Cancellation (with ITU-T G.168/165 standard)
- Automatic negotiate network
- Hotline
- BCCH

### 1.4.3 Industrial Standards Supported

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

### 1.4.4 General Hardware Specification

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

# 2. Equipment Quickly Installation

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

## 2.1 Installation Notice

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

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

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

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

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

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

## 2.2 Installation Procedure

### 2.2.1 Install SIM Card

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

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

- Push in the GSM user board

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

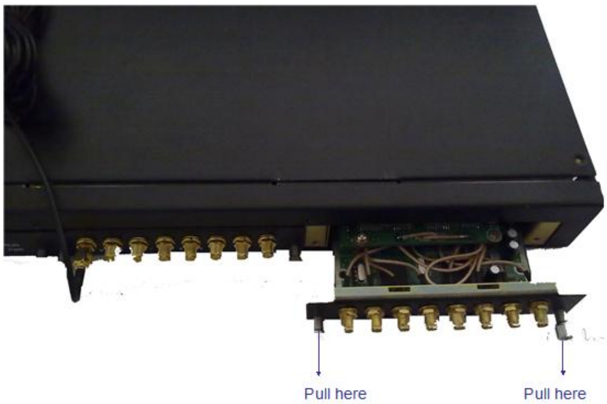



Figure 2-2-2 I Inset the SIM card to the SIM

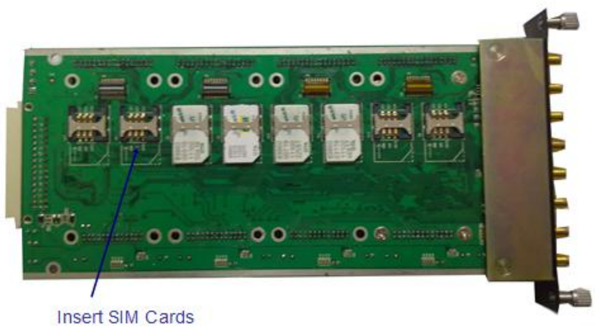



Figure 2-2-3 Push in the GSM user board

### 2.2.2 Antenna Installation

Figure 2-2-4 Antenna Installation


### 2.2.3 Network Cable Connection of Equipment

Figure 2-2-5 DWG2000B-16G connection


### 2.2.4 Power Cable Connection of Equipment

Figure 2-2-6 DWG2000B-16G connection

# 3. Network Configuration

In this chapter we will introduce the initial configuration of DWG2000B-16G. All of the network parameters of the gateway can be configured by IVR guidance.

## 3.1 Attentions

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

## 3.2 General Feature Codes for System Setting

Table 3-2-1 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 DWG2000B-16G through calling IVR.

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

1) Please make sure hardware installation have finished
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

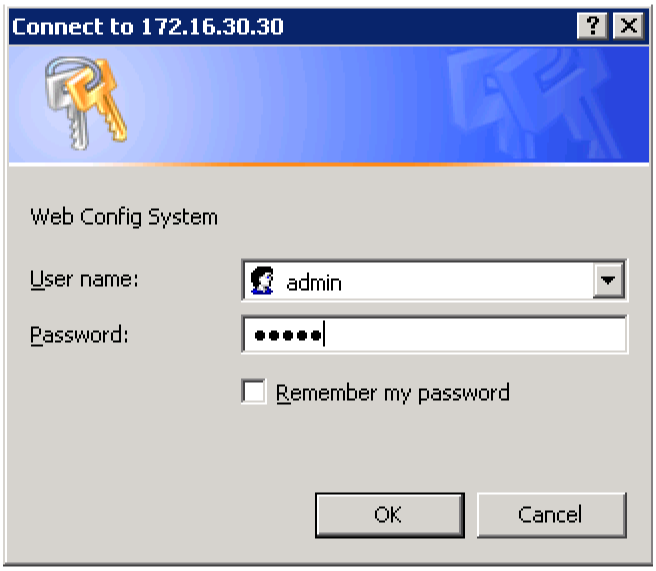
# 4. WEB configuration

This charpter describes web configuration of DWG2000B-16G.

## 4.1 Access the System Through HTTP

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

Figure 4-1-1 WEB log interface

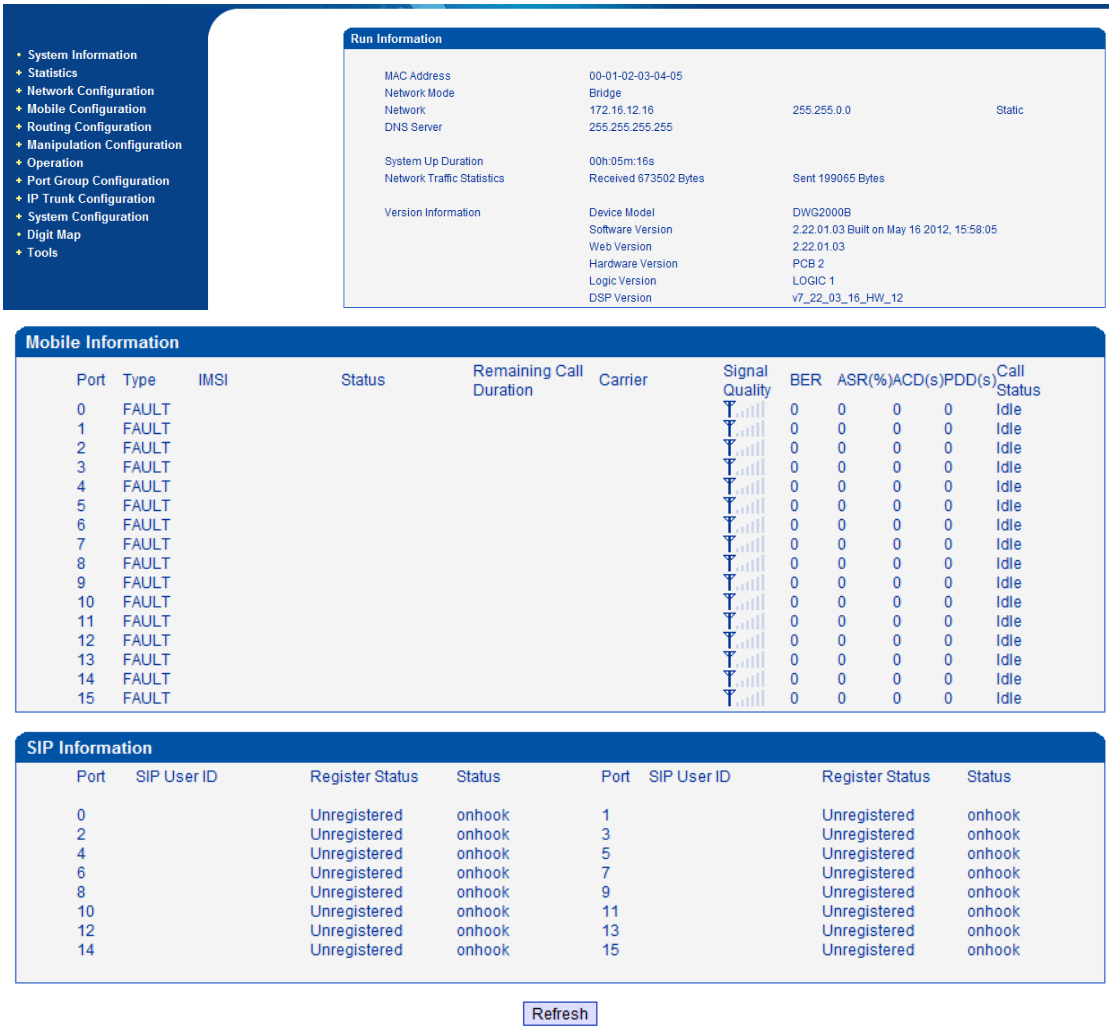



Enter username and password and then click “OK” in configuration interface. The default username and password are “admin/admin”. It is strongly recommended, change the default password to a new password for system security.

## 4.2 WEB Configuration

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

Figure 4-2-1 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 4-3-1 system information

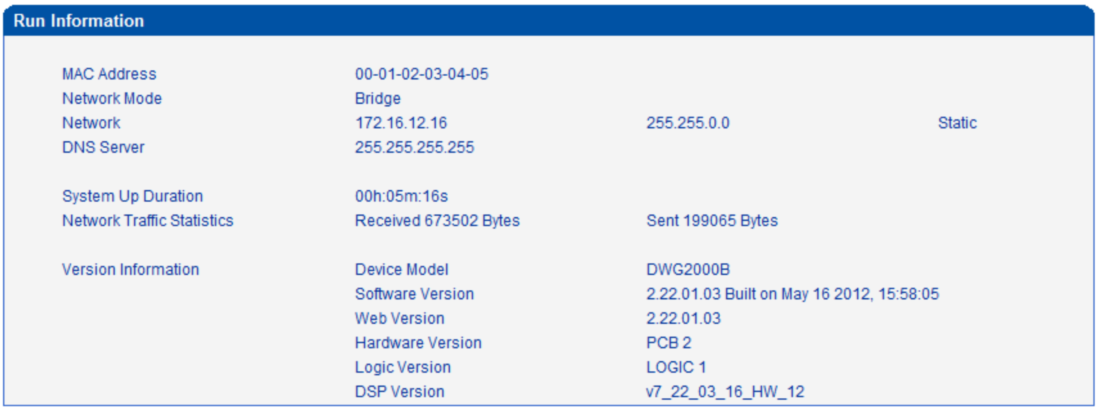
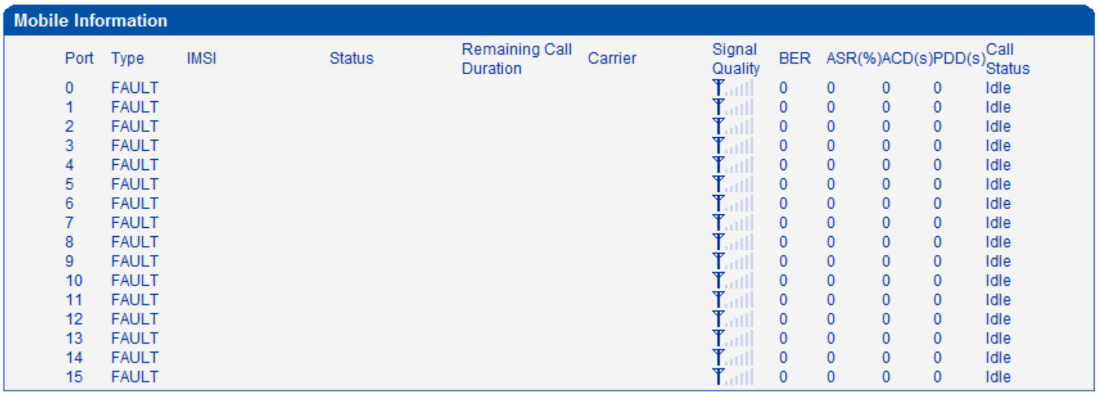



Table 4-3-1 Description of system information

| | |
|---|---|
| MAC Address | Displays the current MAC of the gateway, for example: 00-1F-D6-1B-3D-02 |
| Network Mode | DWG2000B-16G works on bridge mode |
| Network | Shows IP address and subnet mask |
| DNS Server | Displays DNS server IP address in the same network with the gateway |
| System Up Time | Shows the time period of the device running. For example,:1h: 20m, 24s |
| Traffic Statistics | Calculates the netflow, including the total bytes of message received and sent. |
| Version info | shows the current firmware version |

### 4.3.2 Mobile Information

Figure 4-3-2 Mobile information



Mobile Information

| Port | Type | IMSI | Status | Remaining Call Duration | Carrier | Signal Quality | BER | ASR(%) | ACD(s) | PDD(s) | Call Status |
|---|---|---|---|---|---|---|---|---|---|---|---|
| 0 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 1 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 2 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 3 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 4 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 5 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 6 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 7 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 8 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 9 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 10 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 11 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 12 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 13 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 14 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |
| 15 | FAULT | | | | | | 0 | 0 | 0 | 0 | Idle |

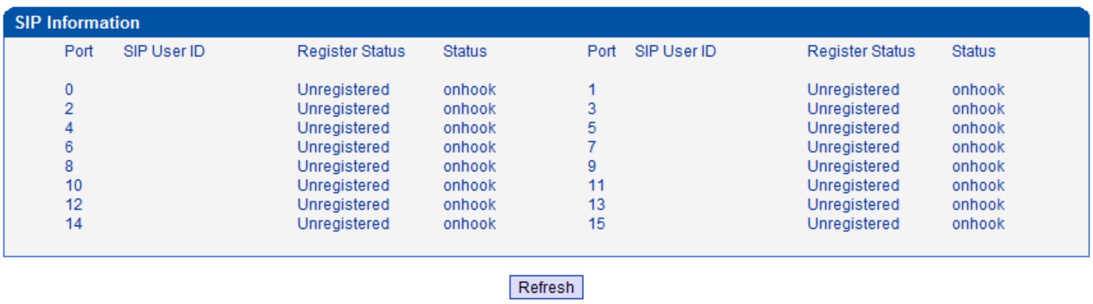
Table 4-3-2 Description of mobile information

| | |
|---|---|
| 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. |
| 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<br>"Idle" means there is no call on this channel<br>"Active" means the call is |

### 4.3.3 SIP Information

Figure 4-3-3 SIP information



SIP Information

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

Refresh

Displays registration status information with Softswitch platform or SIP Server

Table 4-3-3 Description of SIP information

| | |
|---|---|
| Port | The number of GSM channel, DWG2000B-16G has 16 ports |
| SIP User ID | SIP registration account which are provided by the Softswitch and SIP server |
| Register Status | Shows the registration status of VoIP channel, including registered and unregistered. |
| Status | Show the status of port, Include "onhhok" and "offhook" |

## 4.4 Statistics

### 4.4.1 TCP/UDP

Figure 4-4-1 TCP/UDP Statistics

TCP/UDP

| TCP Send Packet | TCP Recv Packet | UDP Send Packet | UDP Recv Packet |
|---|---|---|---|
| 1946619 | 686236 | 221687 | 0 |

Refresh

### 4.4.2 RTP

Figure 4-4-2 RTP

RTP

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

Refresh

Table 4-4-1 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 SIP Call History

Figure 4-4-3 SIP Call History

SIP Call History

| Port | Incoming Received | Incoming Connected | Incoming Answered | Incoming Failed | Outgoing Attempted | Outgoing Connected | Outgoing Answered | Outgoing Failed |
|---|---|---|---|---|---|---|---|---|
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 1 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 2 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 3 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 4 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 5 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 6 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 7 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 8 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 9 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 10 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 11 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 12 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 13 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 14 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 15 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |

Refresh

Table 4-4-2 SIP Call History

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

### 4.4.4 IP to GSM Call History

Figure 4-4-4 IP to GSM Call History

IP to GSM Call History

| Port | Call | Duration | Answered | Call Failed Caused by SIP | | | | Call Failed Caused by GSM | | | | OTHER |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| | | | | Canceled | Timeout | Not Allowed | Negotiation failed | Busy | NO ANSWER | NO DIALTONE | NO CARRIER | |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 1 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 2 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 3 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 4 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 5 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 6 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 7 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 8 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 9 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 10 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 11 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 12 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 13 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 14 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| 15 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |

Refresh Clear

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

## 4.5 Network Configuration

### 4.5.1 Local Network

Figure 4-5-1 Local Network

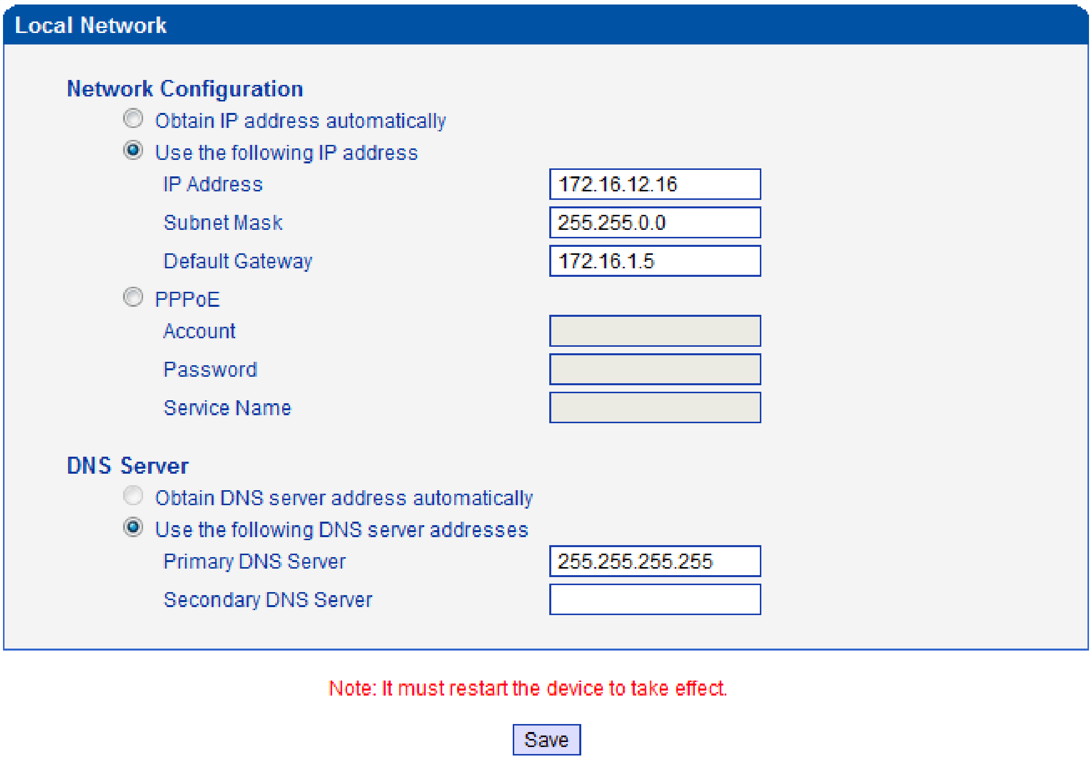



Table 4-5-1 Description of Local network

| | |
|---|---|
| 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" |

### 4.5.2 VLAN Parameter

Figure 4-5-2 VLAN Parameter

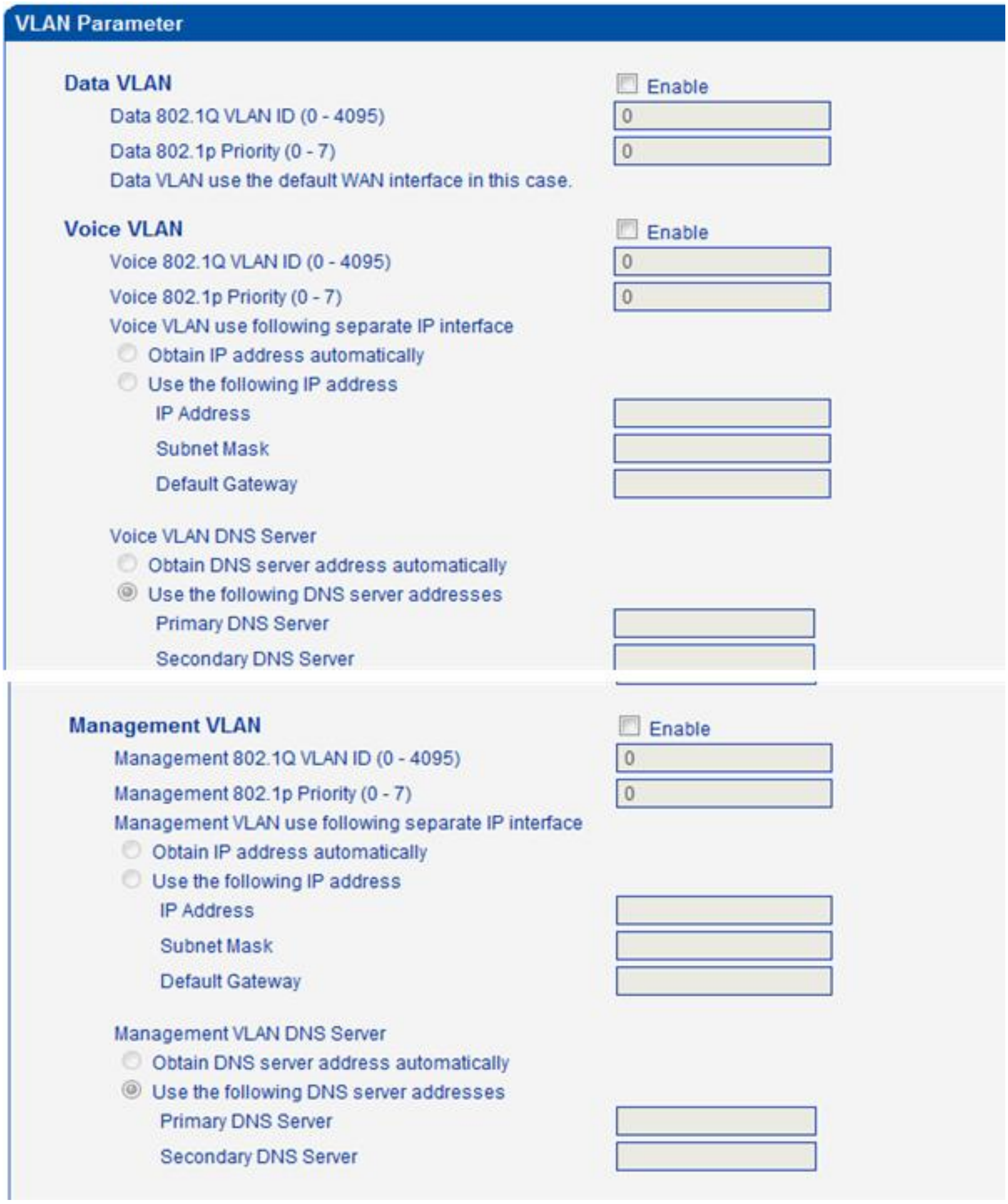


Table 4-5-2 Description of VLAN Parameter

| | | |
|---|---|---|
| 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 |

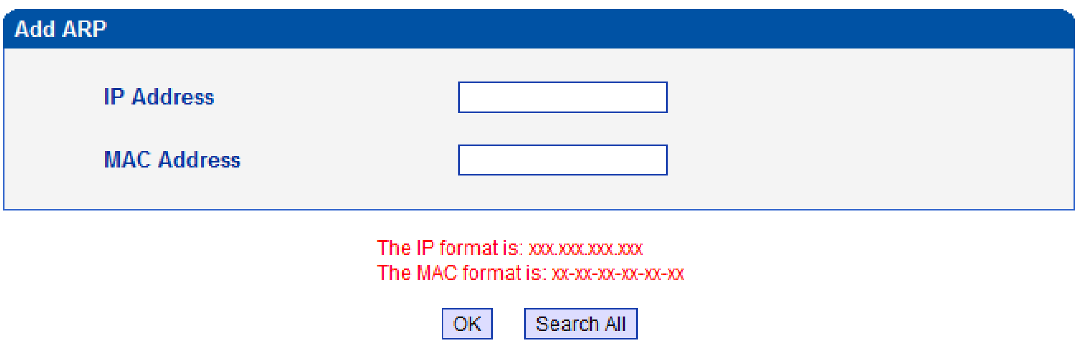
### 4.5.3 ARP

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

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

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

Figure 4-5-3 Add ARP



## 4.6 Mobile Configuration

### 4.6.1 Basic Configuration

Figure 4-6-1Basic Configuration

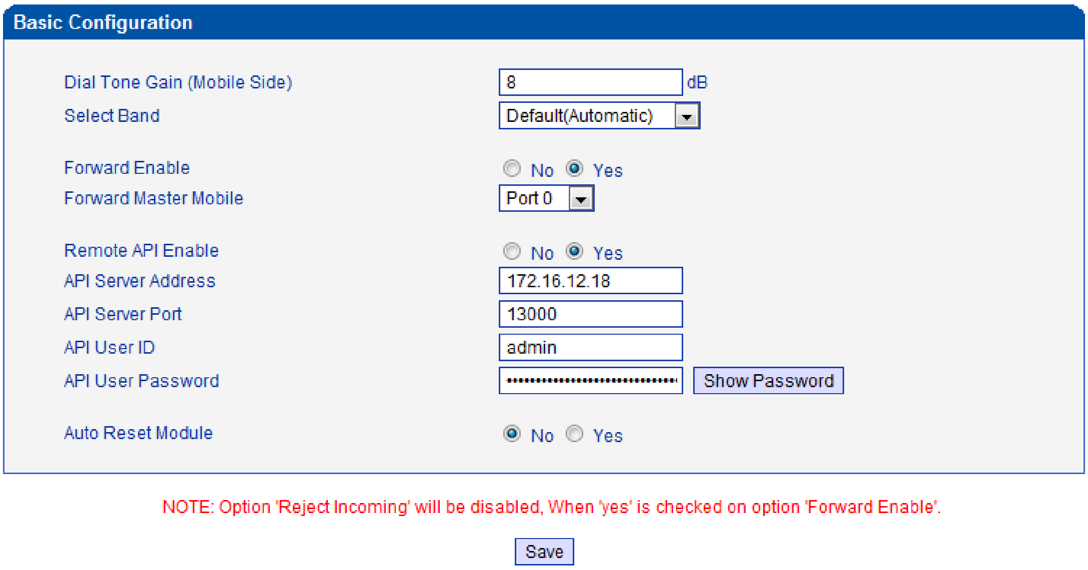



Table 4-6-1 Description of Basic Configuration

| | |
|---|---|
| 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:<br>GSM: 850/900/1800/1900 MHz; CDMA: 800 MHz |
| Forward Enable | When port occupied whether allow call forwarding |
| Forward Master Mobile | Choose the port allowed call transfer. |
| 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 reset GSM/CDMA module when SIM card can not register |

### 4.6.2 Mobile Configuration

Figure 4-6-2 Mobile State

**Mobile State**

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

Figure 4-6-3 Mobile Configuration

**Mobile Configuration**

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

NOTE: 1.If the duration of a call is less than 'Minimum Charging Time', it will be not included in 'Call Duration'.
2.Check the anti-pole signal is only effective on the CDMA.

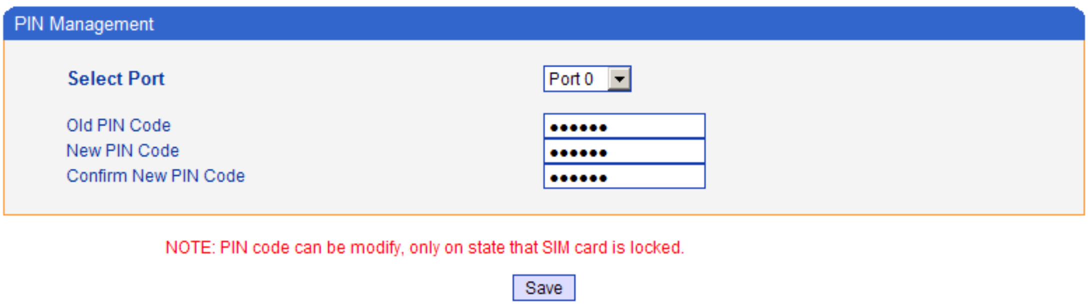
Table 4-6-2 Description of Mobile Configuration

| | |
|---|---|
| Mobile Number | SIM card number of the channel. That must be configured when “Forward” funciton enable. |
| Step | Step length value range is 1-120 s, step length multiplied by time of single call just said a single call duration time allowed. |
| Enable Call Duration Limitation single call | Definite maximum call duration for single call. Example: if Time of single call set to 10, the call will be disconnected after talking 10*step seconds |
| Time of single time | The value of limitation single call, this value range is 1-65535. step length multiplied by time of single call just said a single call duration time allowed. |
| Enable Call Duration Limitation | This function is to limit the max call duration of channel. The max call duration is between 1 to 65535 minutes. |
| Auto reset | Set a time make device reboot |
| Maximum Call Duration | Defines a value by users. That will limit the SIM/UIM card’s total call duration. After the call duration excesses this value, no call will be made from this channel. The value range is 1-65535. If user doesn’t configure this value, Default is no max call duration limits for this channel. |
| Minimum Charging Time | A minimum charging time (in seconds) is defined during which no charging is done at carrier side. If the conversation time is even shorter, the total call duration will not decrease. |
| Alarm Threshold（via 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. |
| 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. |
| 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. |

| | |
|---|---|
| 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, DWG2000B-16G will connect soon. It doesn't wait mobile side answer. |

### 4.6.4 PIN Management

Figure 4-6-4 PIN Management



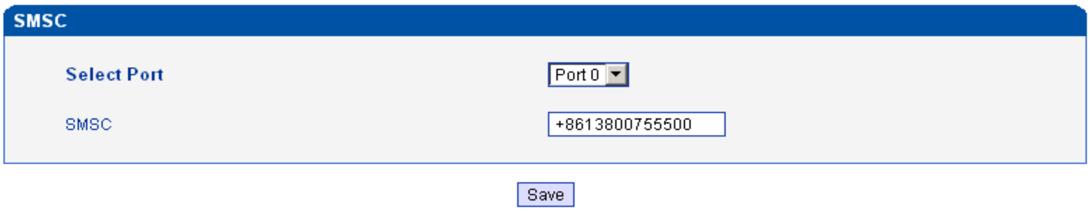

**Detailed description as below:**

Table 4-6-3 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

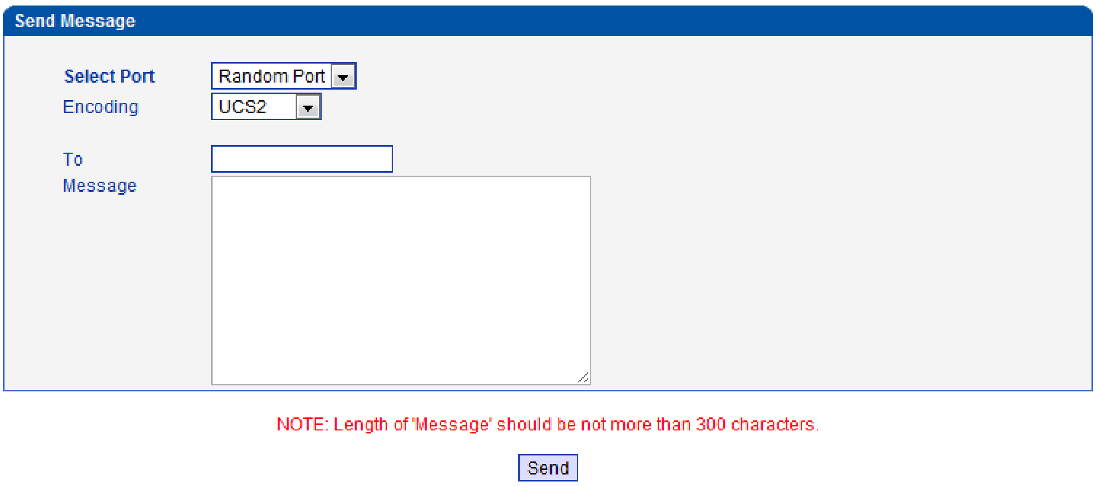



Table 4-6-5 Description of SMS sending

| | |
|---|---|
| Select Port | Users can select a defined channel or random channel to send SMS. Input the recevier's mobile phone No to send SMS. |
| Encoding | Two kinds of message encoding under PDU models, 7-bit code used to sent ordinary ASCII characters; UCS2 coding used to send Unicode characters. |
| To | 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

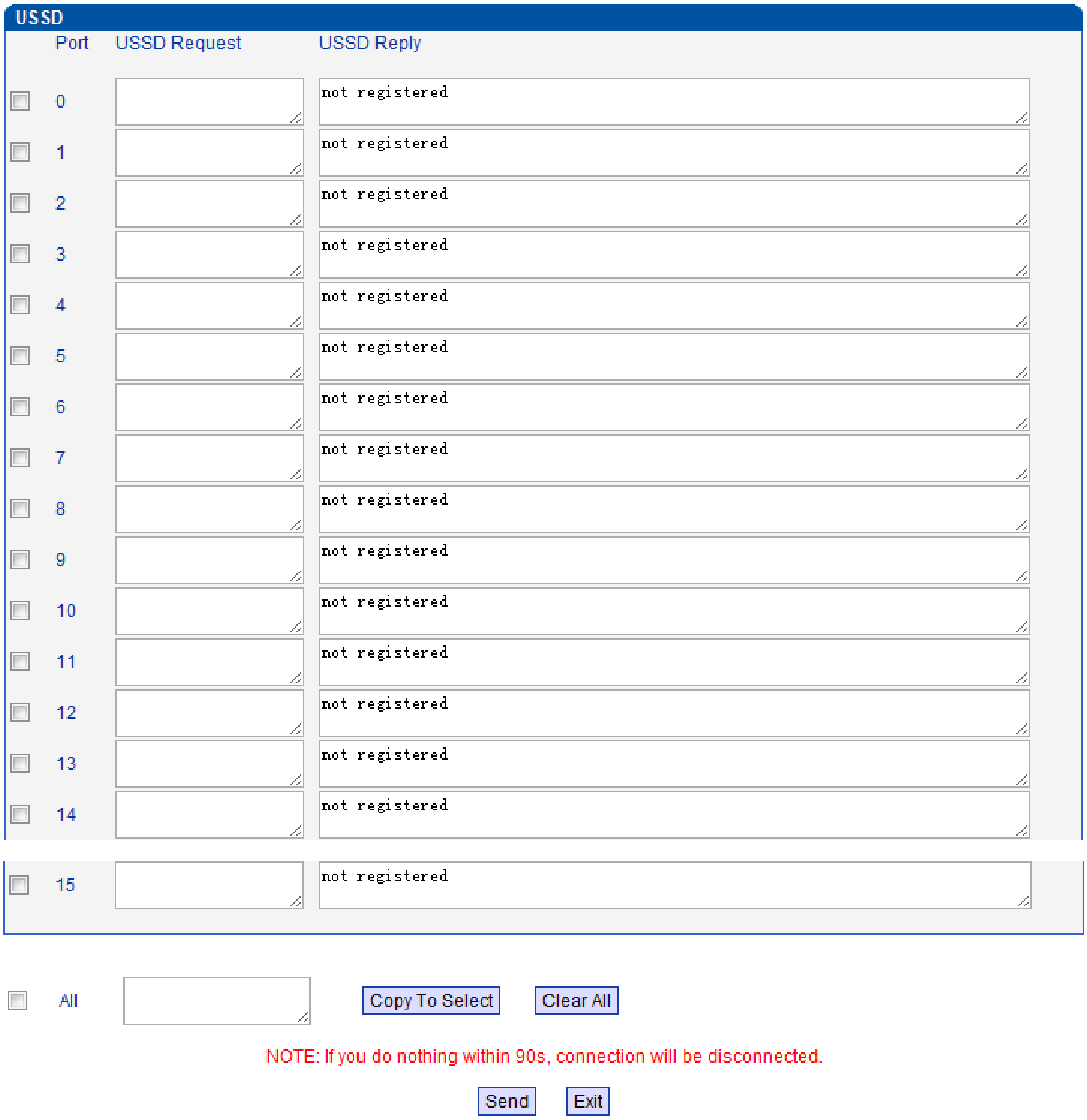



Table 4-6-6 Description of USSD

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

### 4.6.8 Carrier

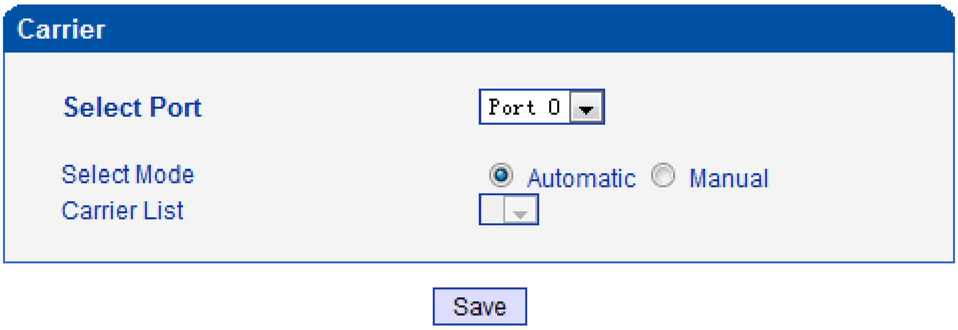



Figure 4-6-8 Select Carrier

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 modes to select carrier automatic and manual. Automatic mode can be automatically search operators. Manual mode can choose operators from the carrier list. |
| Carrier List | If you select manual mode, you can select carrier from carrier list. |

### 4.6.9 BCCH

Figure 4-6-9 BCCH

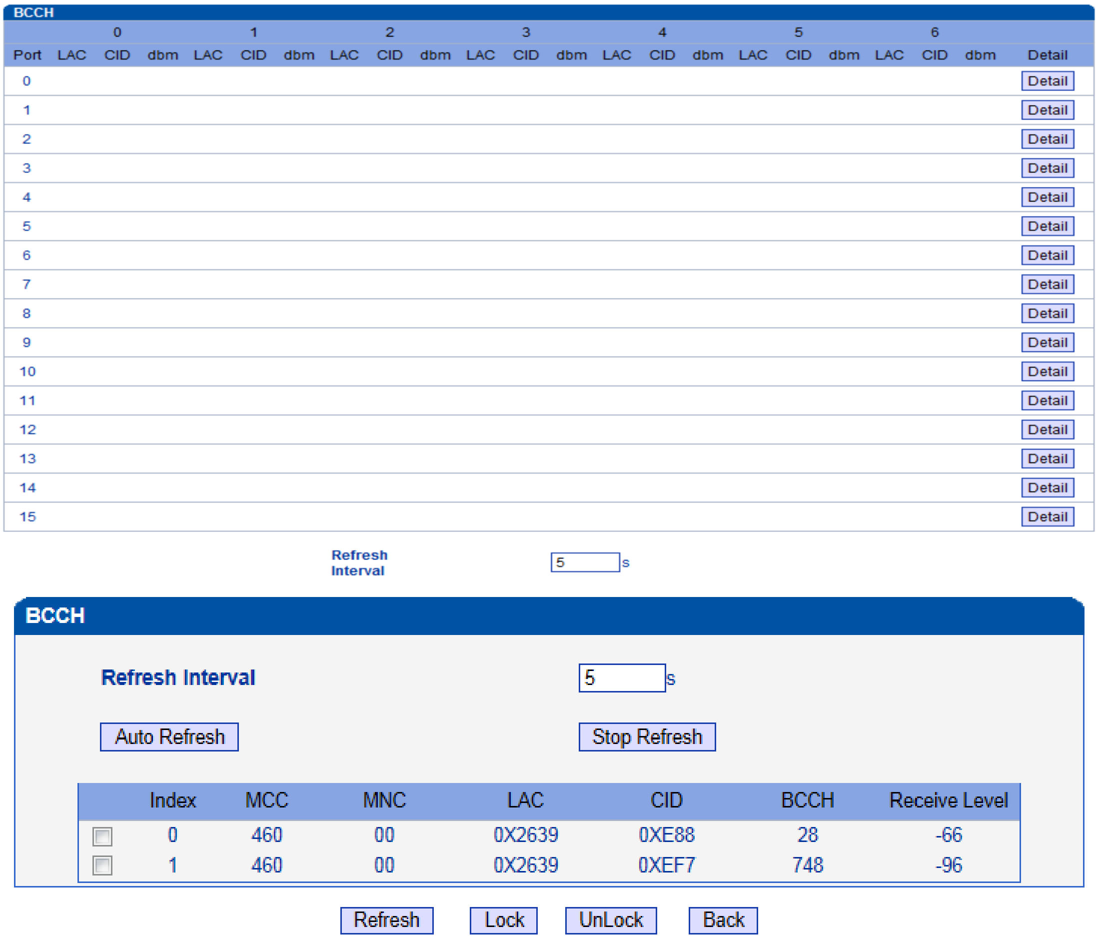


Table 4-6-7 Description of BCCH

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

Choose a frequency to lock the operations.

## 4.7 Routing Configuration

### 4.7.1 Routing Parameter

Figure 4-7-1 Routing Parameter

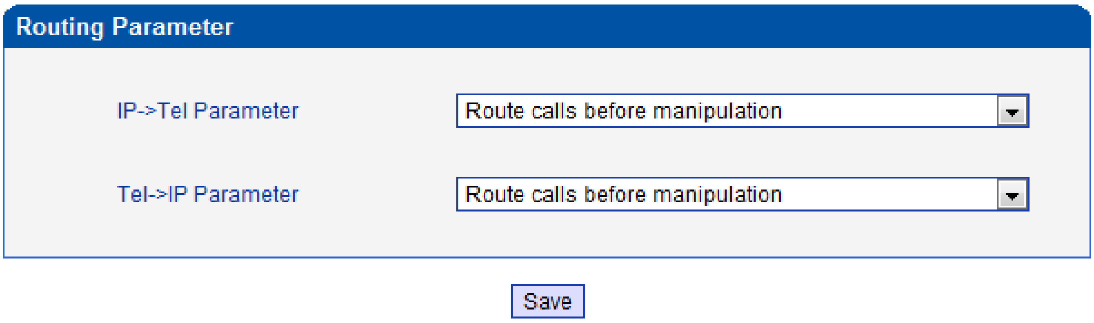
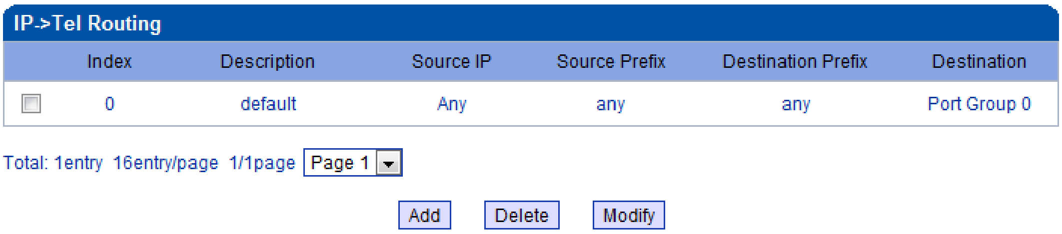



Table 4-7-1Description of Routing Parameter

| | |
|---|---|
| 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



IP->Tel Routing

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

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

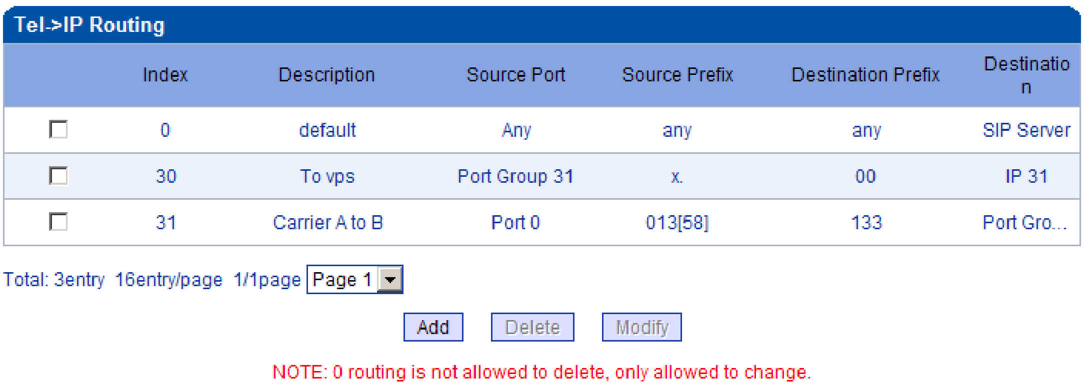
Add Delete Modify

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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>• 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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>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



Tel->IP Routing

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

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

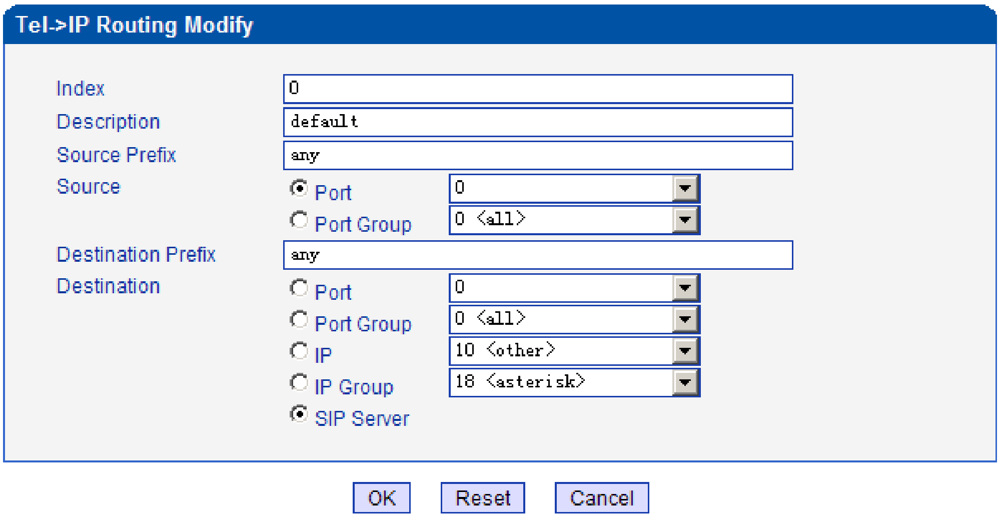
Add Delete Modify

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

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

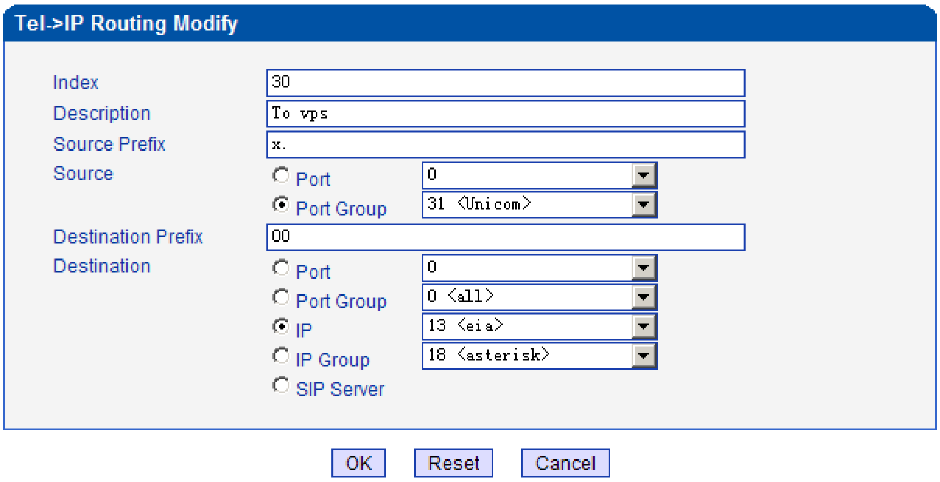
| | |
|---|---|
| Tel -> IP Routing | This item uses to configure incoming call routes which can be used for recieve the calls from the GSM. |
| Index | It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.The route preferentially match the rules which the value of index 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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>• 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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186 |
| Destination | Its specifies destination Port or Port Group |

Figure 4-7-4 Tel to IP routing Modify



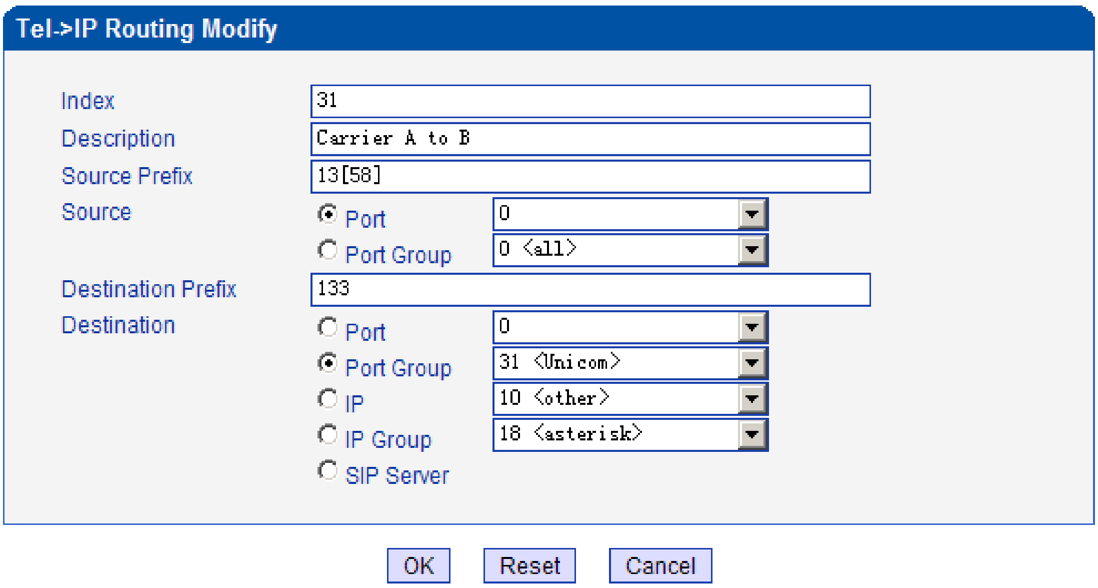

It's a default route configured in gateway. It allows any number from source port 0 send call to SIP server with any prefix.

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

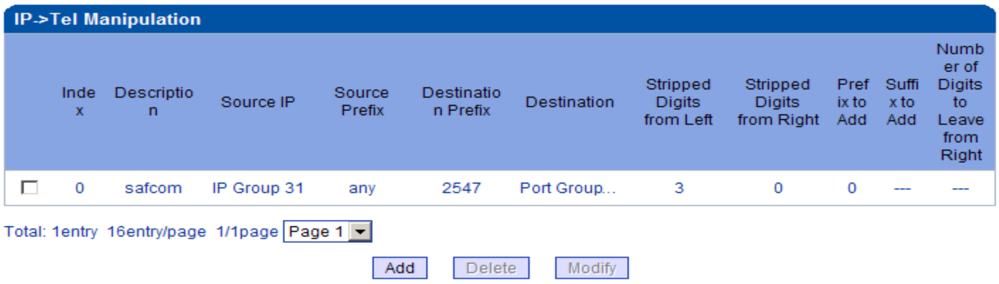
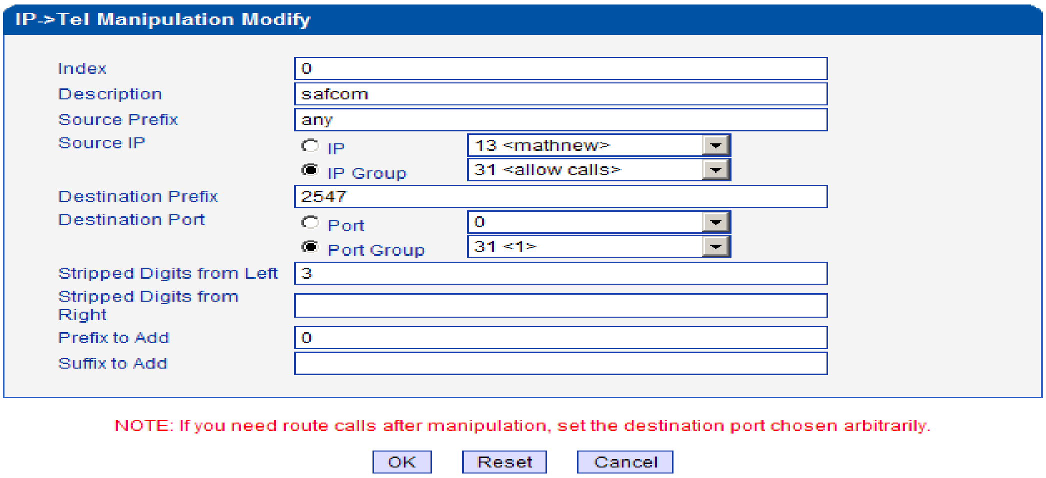



Table 4-8-1 Description of IP->Tel destination numbers manipulation

| | |
|---|---|
| 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<br>• Any: any IP address<br>• IP: specific an IP address<br>• 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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>• 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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>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 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

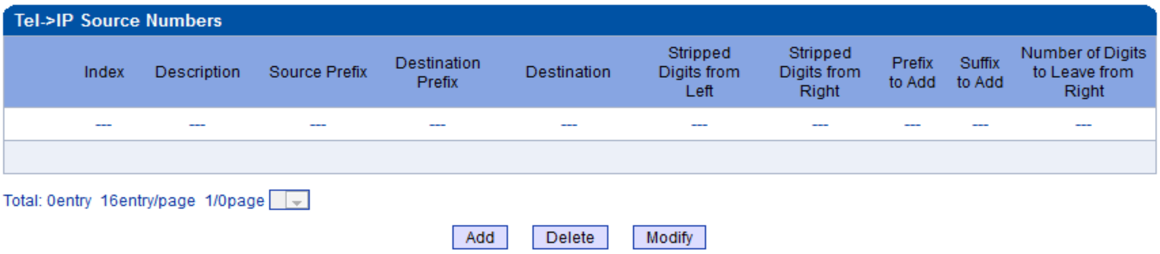



Table 4-8-2 Description of Tel->IP destination numbers manipulation

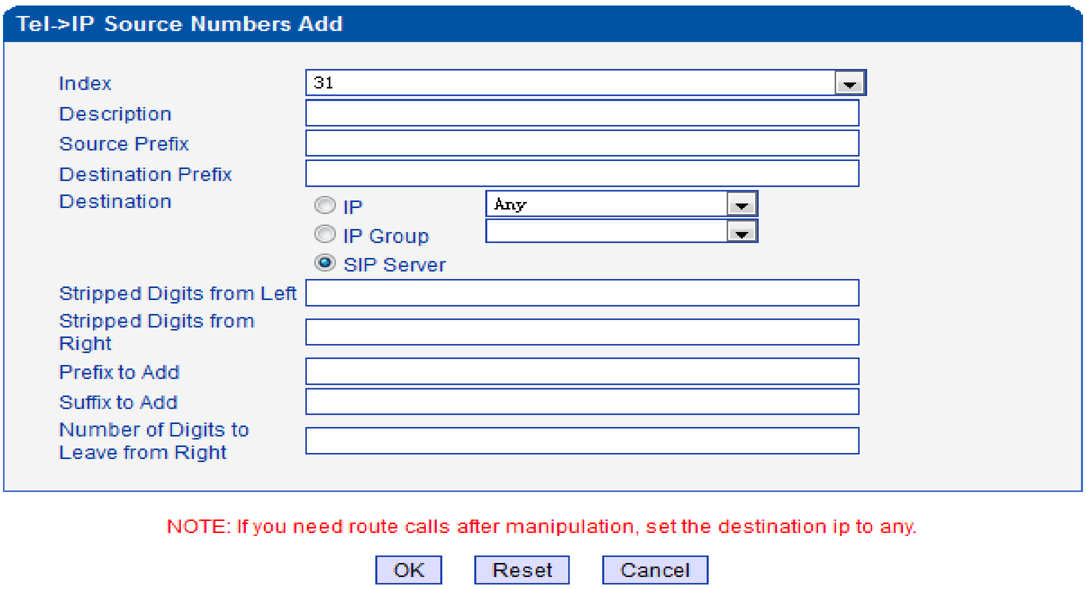
| | |
|---|---|
| Tel->IP destination numbers manipulation | It is an optional configuration item, and is used to add IP->Tel number change data.<br>The IP->Tel Manipulation defined the rules of add, and deletion of called numbers, which are referenced by IP->Tel routing. |
| Index | It uniquely identifies 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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>• 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<br>• Any: include anonymous, 0xxxx, 1[2-9] xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>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

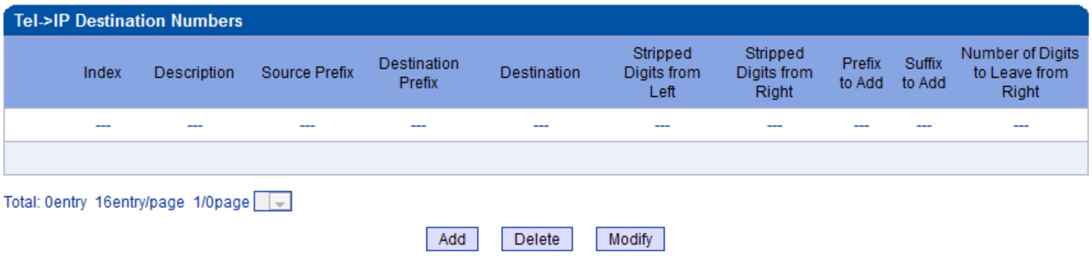


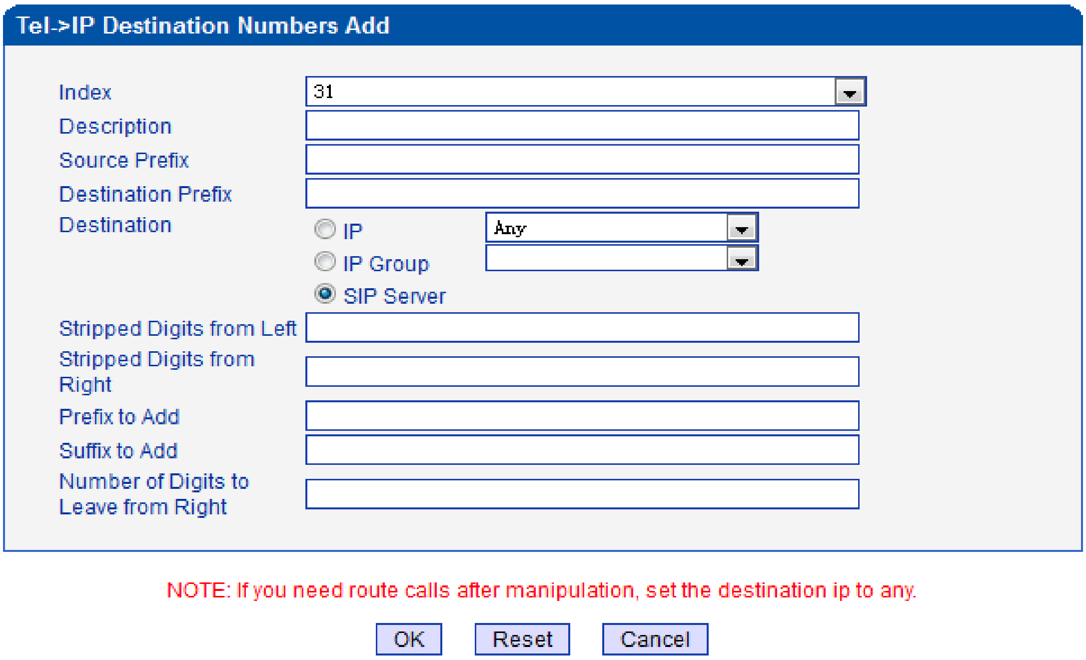

Table 4-8-3 Description of Tel->IP destination numbers manipulation

| | |
|---|---|
| Tel->IP destination numbers manipulation | It is an optional configuration item, and is used to add IP->Tel number change data.<br>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<br>• Any: include anonymous, 0xxxx, 1[2-9] xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>• 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<br>• Any: include anonymous, 0xxxx, 1[2-9] xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>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

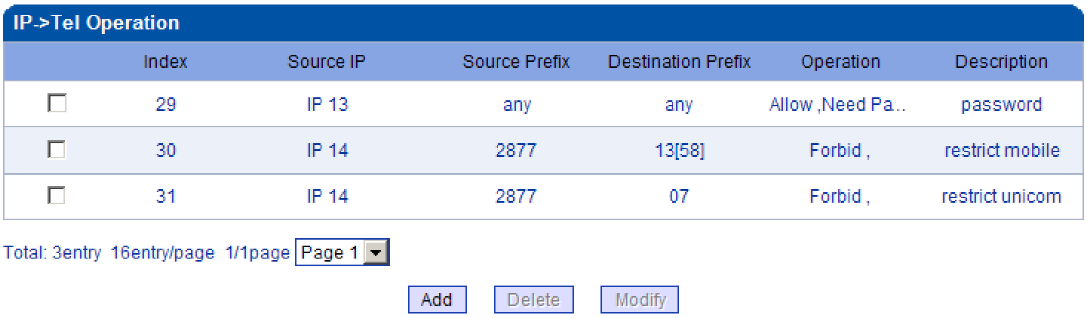



It indicates that calls coming from IP Group will match the prefix ”any”, and the called nubmer which match with the prefix ”2547” will delete 3 digits in front of it and replace it by digit ”0”.

## 4.9 Operation

### 4.9.1 IP->Tel Operation

Figure 4-9-1 IP->Tel Operation



IP->Tel Operation

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

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

Add Delete Modify

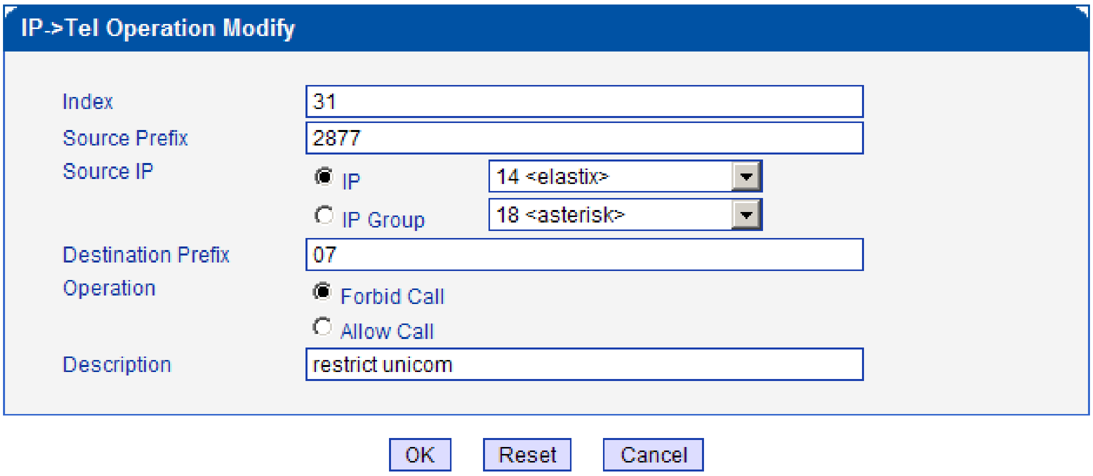
Table 4-9-1 Description of IP->Tel Operation

| | |
|---|---|
| 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<br>• Any: any IP address<br>• IP: specific an IP address<br>• 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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>• 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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186 |
| Operation | Its specifies number analysis rule<br>• Forbid call<br>• Allow call<br>• Auto call<br>• 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>

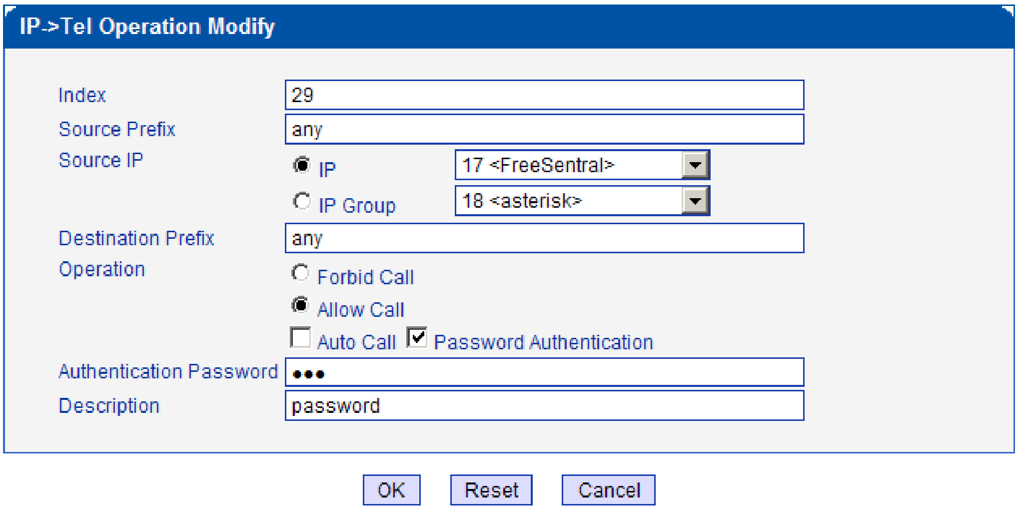
Figure 4-9-2 IP->Tel Operation Modify




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

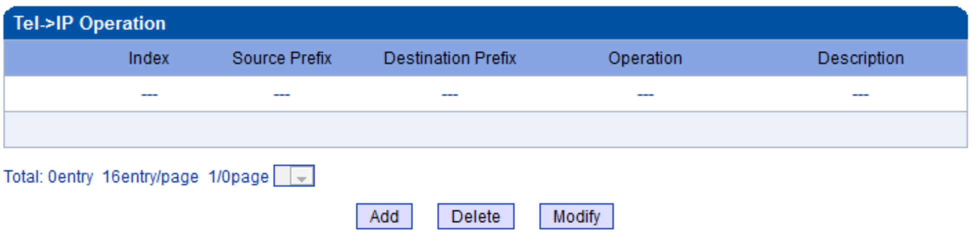


Table 4-9-2 Description of Tel->IP Operation

| 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 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<br>• Any: any IP address<br>• IP: specific an IP address<br>• 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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>• 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<br>• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.<br>• 0xxxx: consist of some digits such as 015,08,09<br>1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186 |
| Operation | Its specifies number analysis rule<br>• Forbid call<br>• Allow call<br>• Auto call<br>• Password authenticate |
| Description | It describes the route for the ease of identification. Its value is character string |

## 4.10 Port Group Configuration

### 4.10.1 Port Group

Figure 4-10-1 Port Group

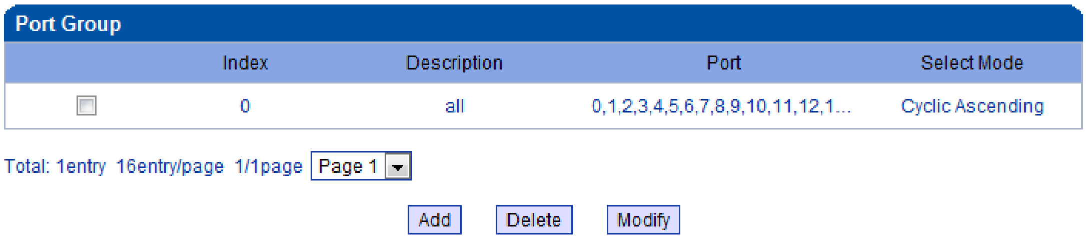
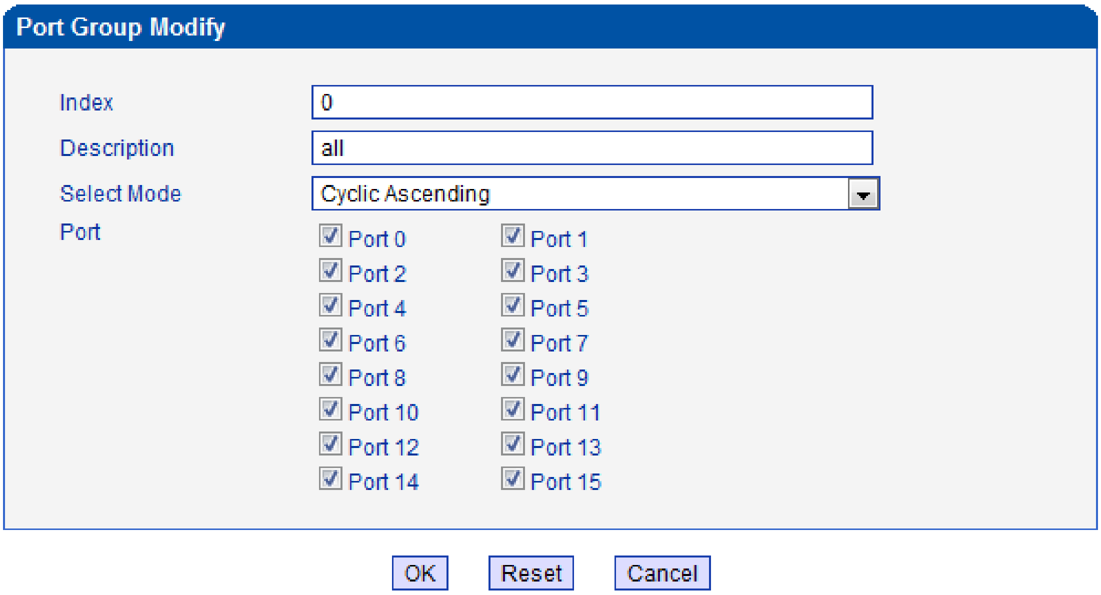



Figure 4-10-2 Port Group Modify



## 4.11 IP Trunk Configuration

### 4.11.1 IP Trunk

Figure 4-11-1 IP Trunk

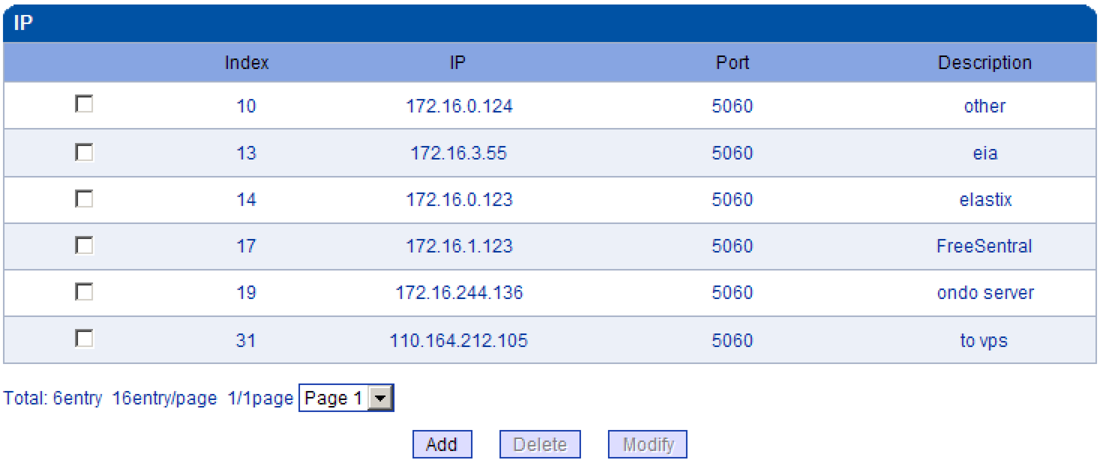


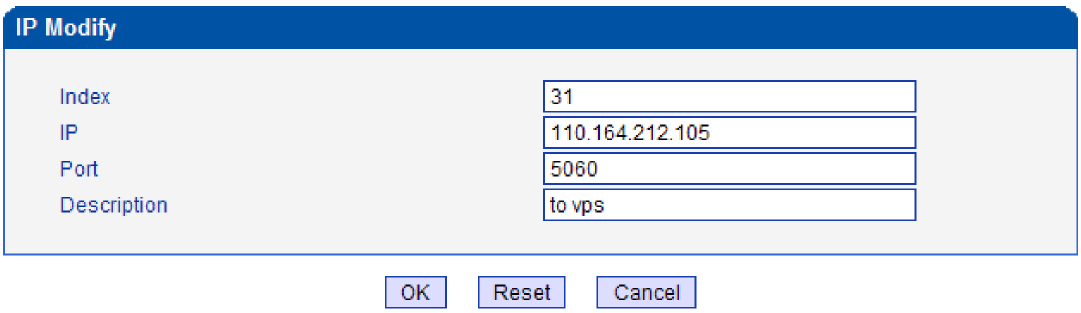

Table 4-11-1 Description of IP Trunk

| | |
|---|---|
| IP Trunk | Add remote IP of softswitch, SIP server which will send call traffics to gateway. |
| Index | It uniquely identifies a trunk . 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



### 4.11.2 IP Trunk Group

Figure 4-11-3 IP Trunk Group

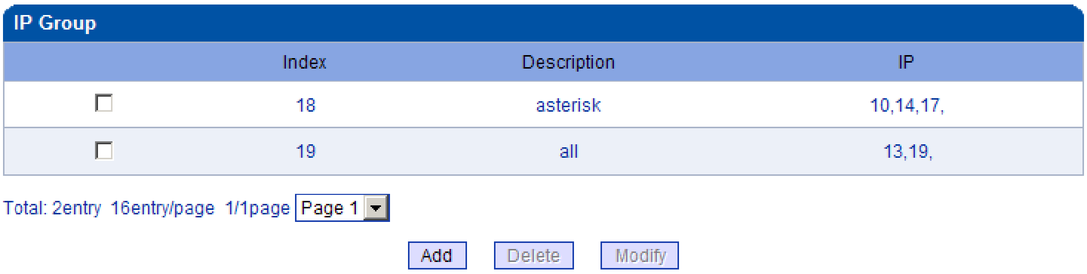


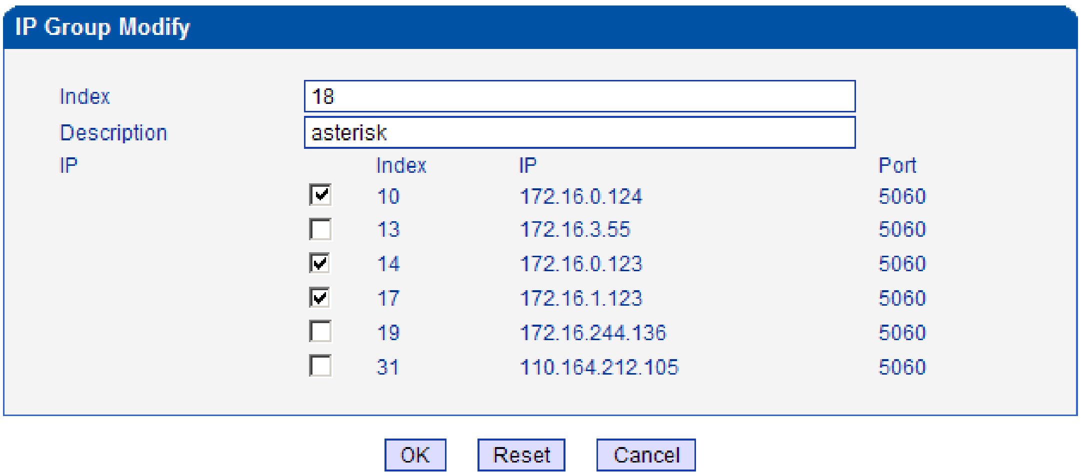

Table 4-11-2 Description of IP Trunk Group

| | |
|---|---|
| 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

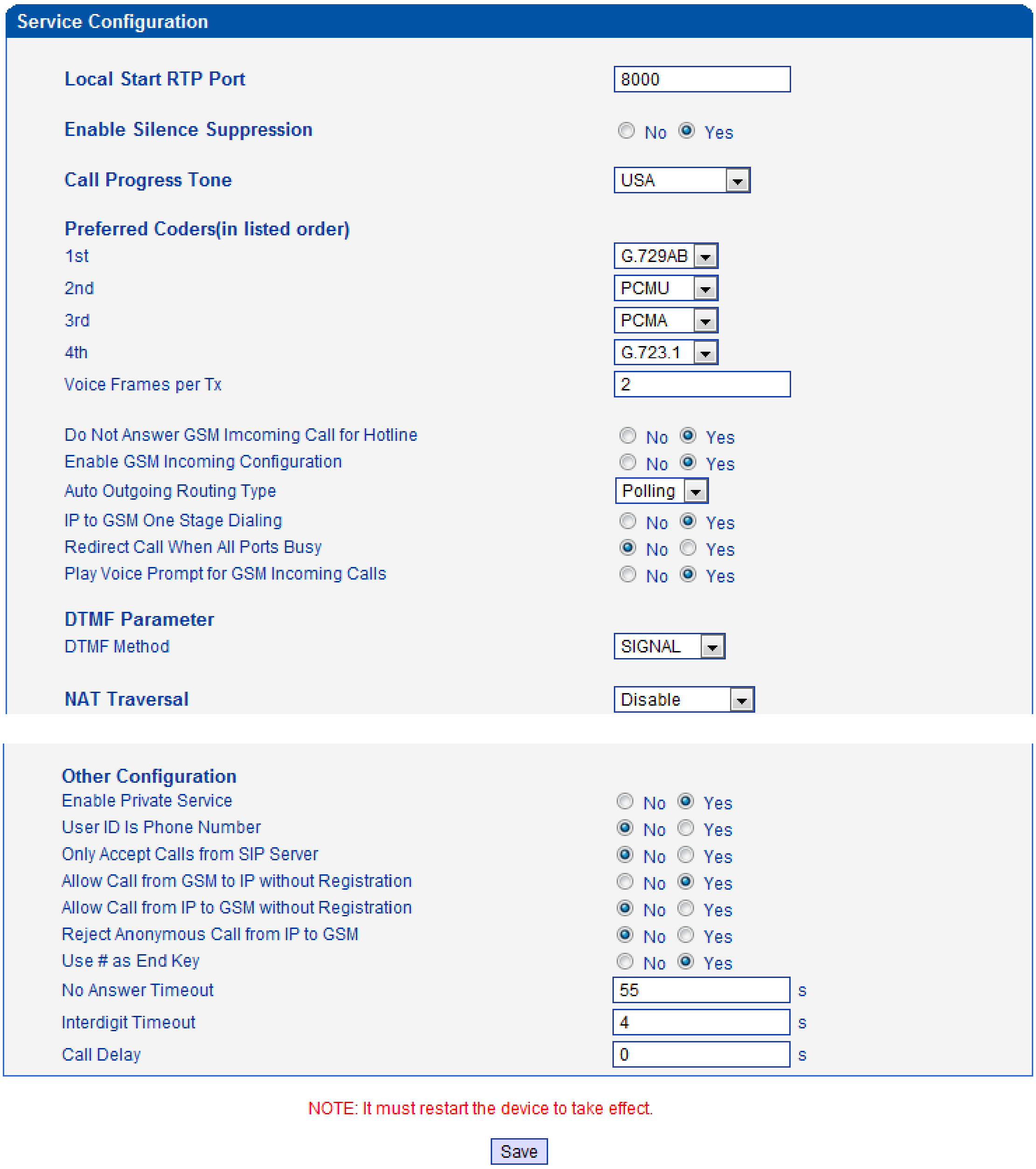


Table 4-12-1 Description of Service Configuration

| | |
|---|---|
| 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/DWG2000B-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

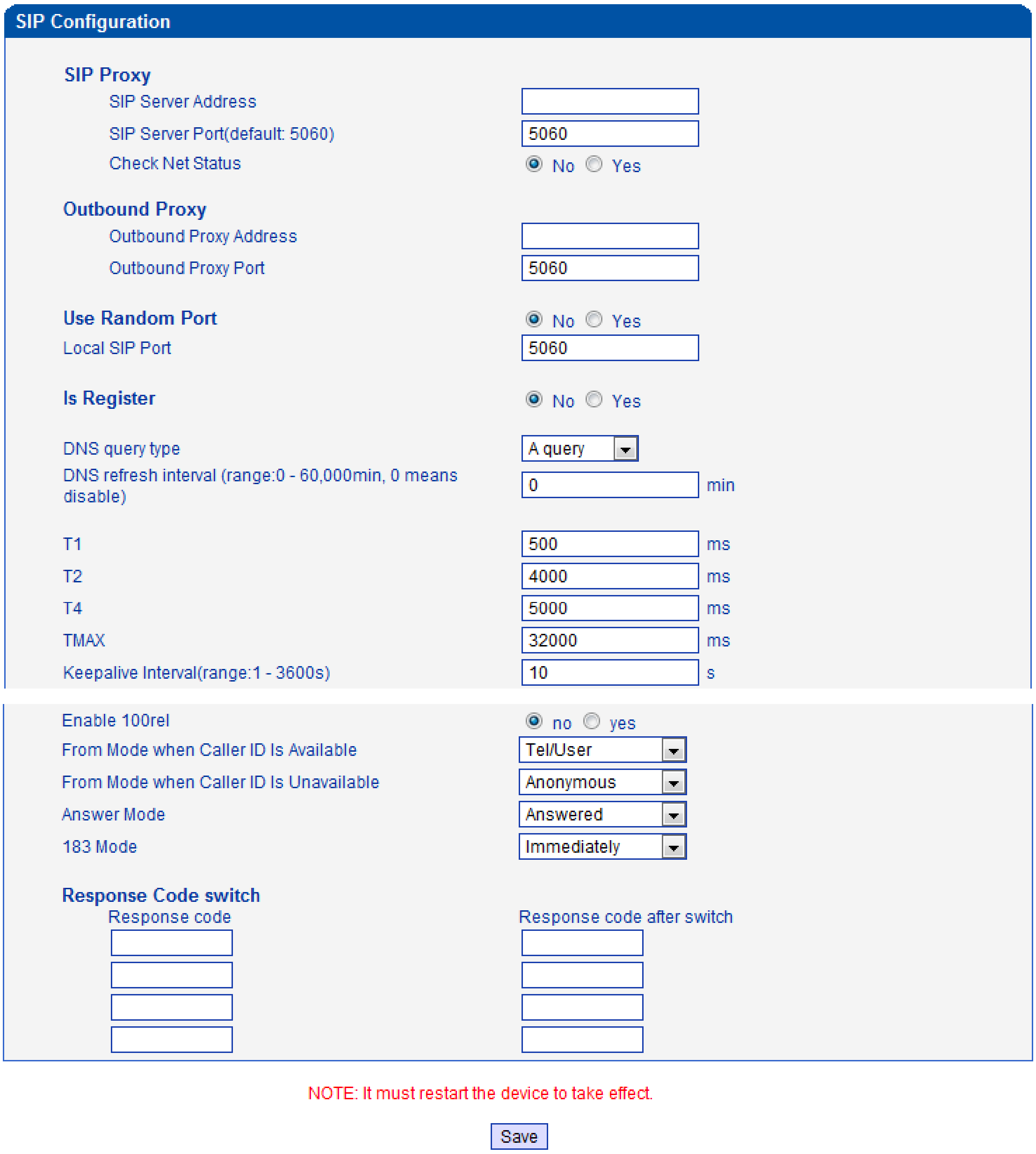


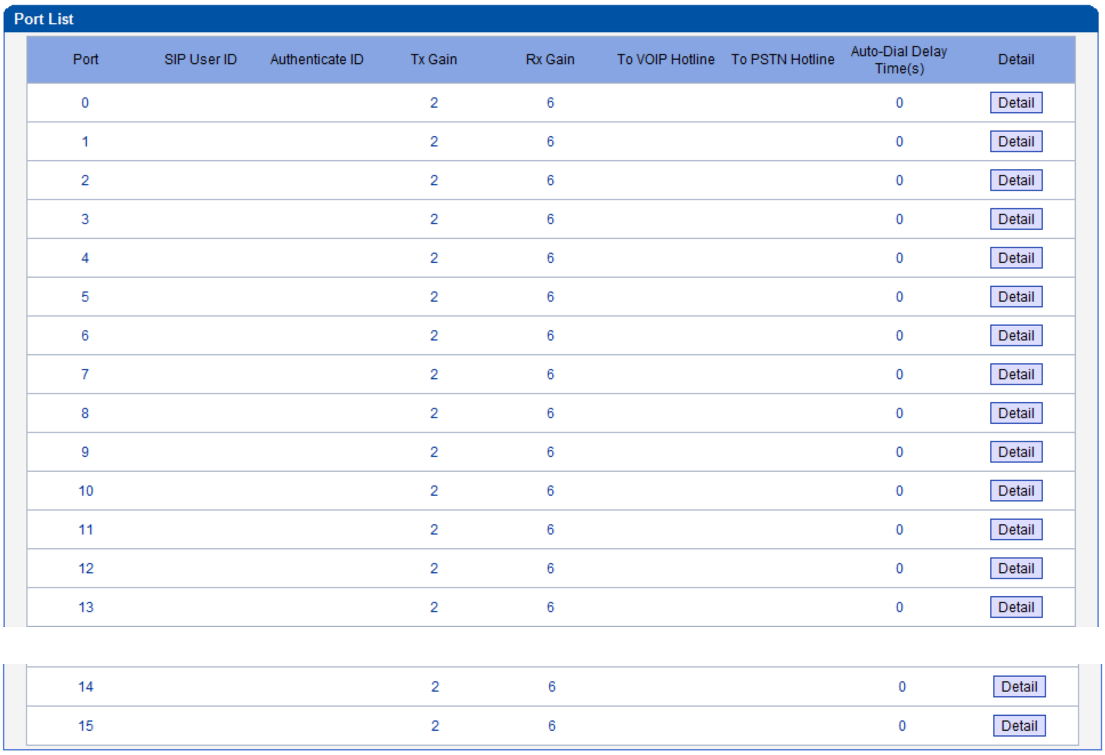

Table 4-12-2 SIP Configuration

| | |
|---|---|
| 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<br>Tel/User: From: caller number <sip:3001@IP>;tag=51088abb<br>User/User: From: 3001 <sip:3001@IP>;tag=51088abb<br>Tel/Tel: From: caller number <sip: caller number @IP>;tag=51088abb<br>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<br>Anonymous : From: <sip: Anonymous @IP>;tag=51088abb<br>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<br>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.<br>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 List


Port List

| Port | SIP User ID | Authenticate ID | Tx Gain | Rx Gain | To VOIP Hotline | To PSTN Hotline | Auto-Dial Delay Time(s) | Detail |
|---|---|---|---|---|---|---|---|---|
| 0 | | | 2 | 6 | | | 0 | Detail |
| 1 | | | 2 | 6 | | | 0 | Detail |
| 2 | | | 2 | 6 | | | 0 | Detail |
| 3 | | | 2 | 6 | | | 0 | Detail |
| 4 | | | 2 | 6 | | | 0 | Detail |
| 5 | | | 2 | 6 | | | 0 | Detail |
| 6 | | | 2 | 6 | | | 0 | Detail |
| 7 | | | 2 | 6 | | | 0 | Detail |
| 8 | | | 2 | 6 | | | 0 | Detail |
| 9 | | | 2 | 6 | | | 0 | Detail |
| 10 | | | 2 | 6 | | | 0 | Detail |
| 11 | | | 2 | 6 | | | 0 | Detail |
| 12 | | | 2 | 6 | | | 0 | Detail |
| 13 | | | 2 | 6 | | | 0 | Detail |
| 14 | | | 2 | 6 | | | 0 | Detail |
| 15 | | | 2 | 6 | | | 0 | Detail |

Figure 4-12-4 Port Configuration

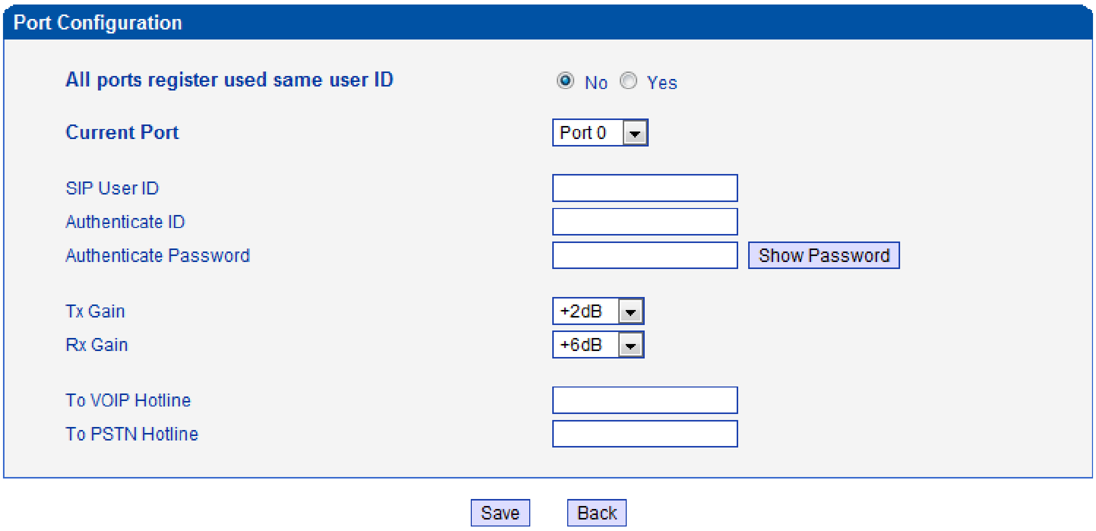


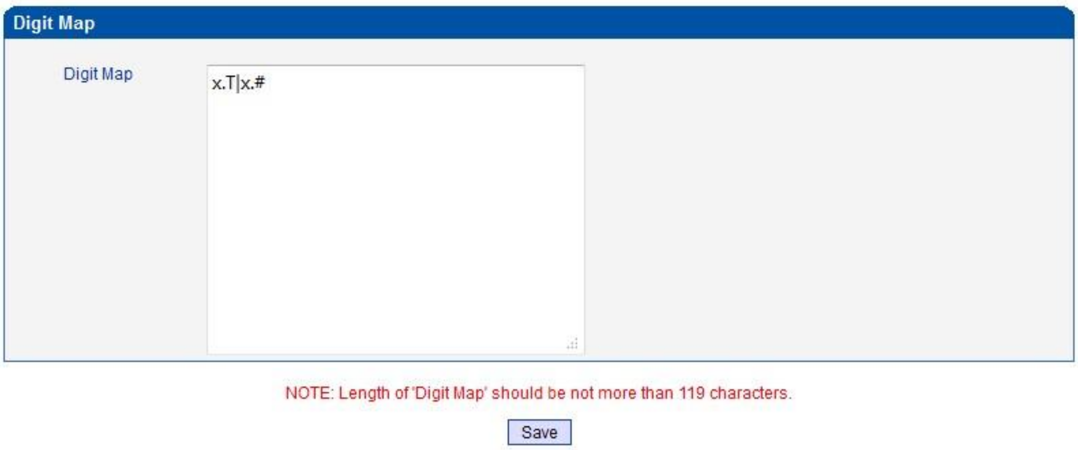

Table 4-12-3 Description of Port Configuration

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

### 4.14.2 Management Parameter

Figure 4-14-2 Management Parameter

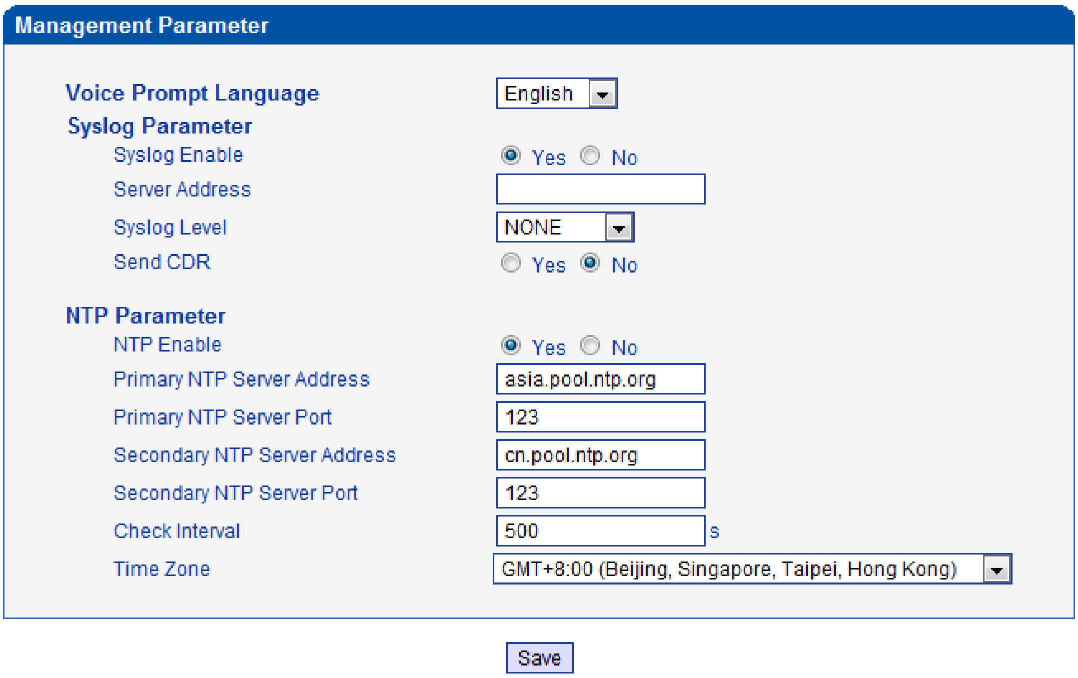
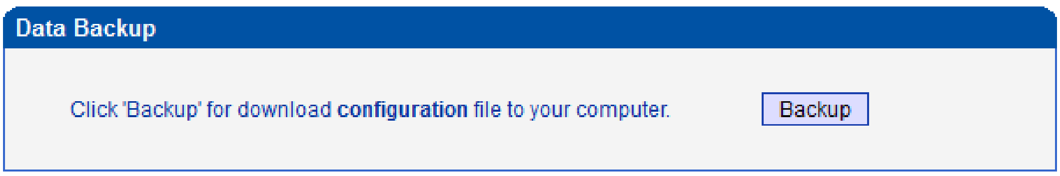


Table 4-14-1 Management Parameter

| | |
|---|---|
| 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.<br>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.<br>User need to fill the NTP Server Address and select Time Zone |

### 4.14.3 Config Backup

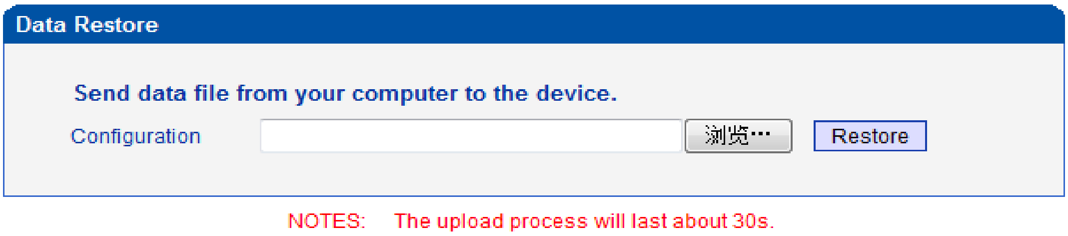
Figure 4-14-3 Data backup




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

**4.14.4 Config Restore**

Figure 4-14-4 Data restore

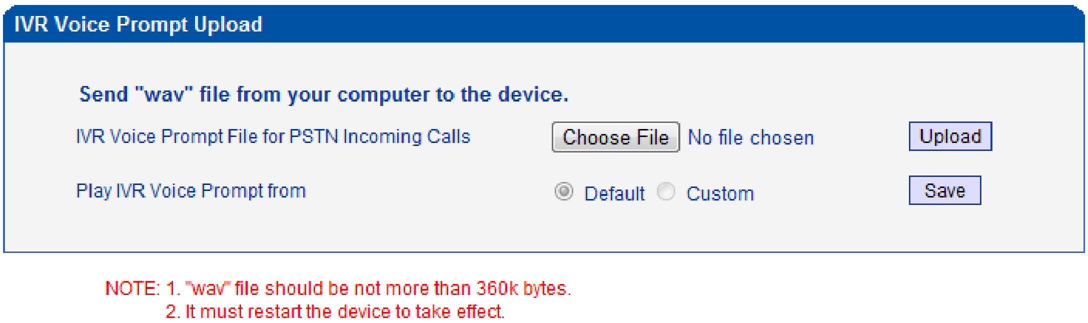



Send data file from your computer to the device

**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

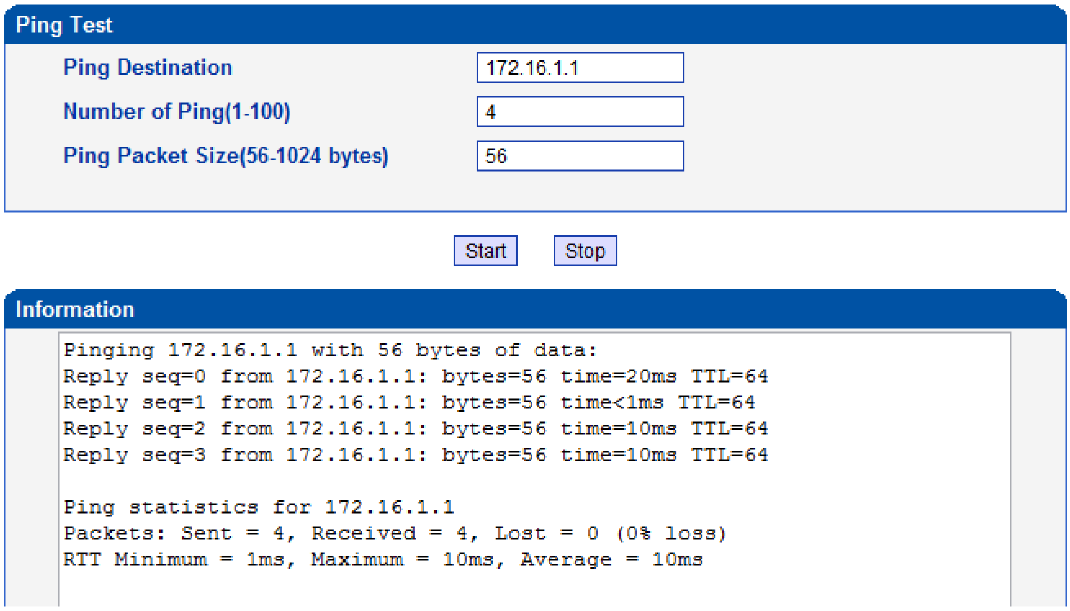



**NOTE:** the customize voice files can be recorded using Windows recording programs, the sound format is 8000Hz, 16 bit sampling in mono, with WAV format, size of files can not exceed 190KB

### 4.14.6 Ping Test

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

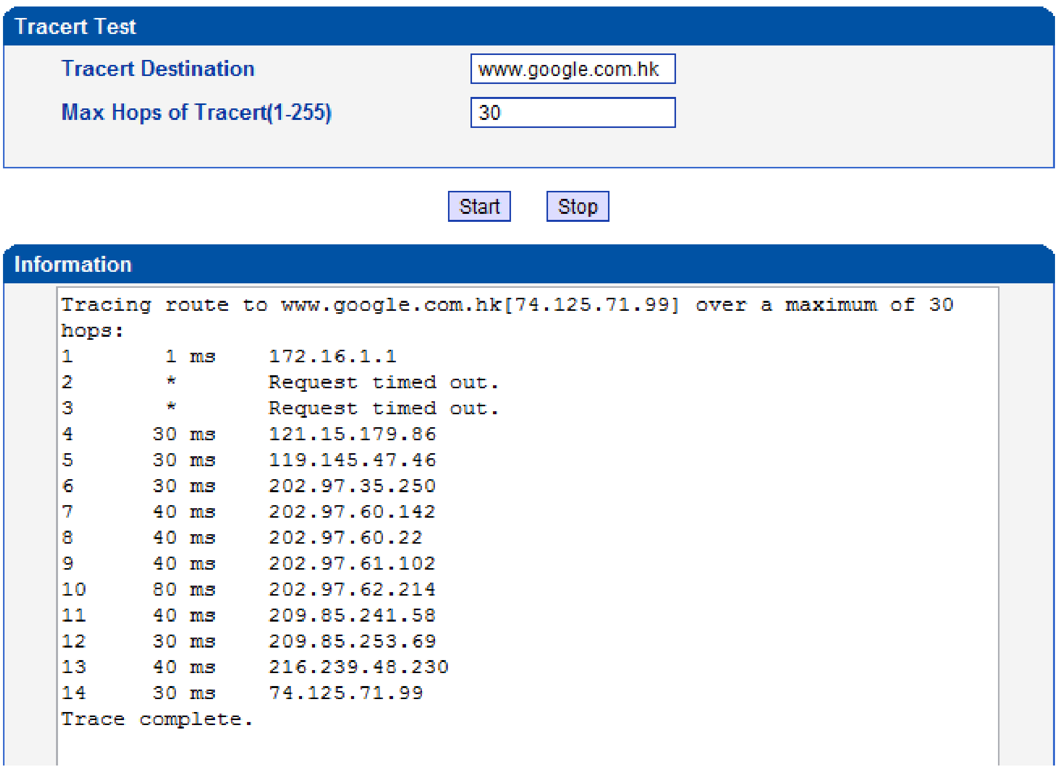
Figure 4-14-6 Ping Test




### 4.14.7 Tracert Test

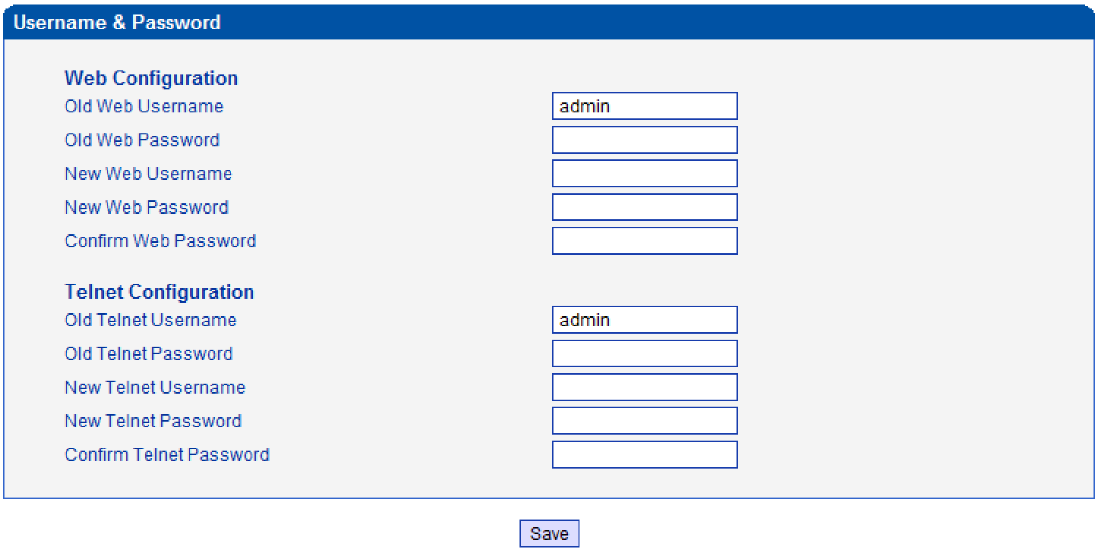
Traceroute is a computer network diagnostic tool for displaying the route (path) and measuring transit delays of packets across an Internet Protocol (IP) network.

Figure 4-14-7 Tracert Test



### 4.14.8 Username & Password

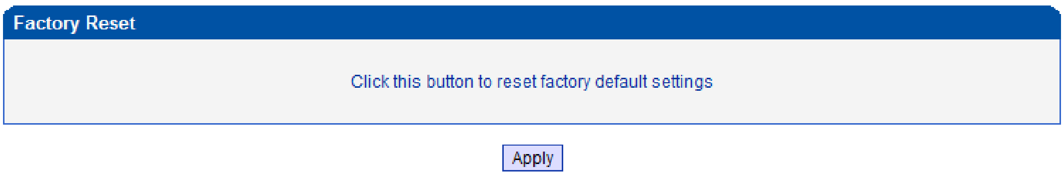
Figure 4-14-8 IVR Voice Prompt Upload




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

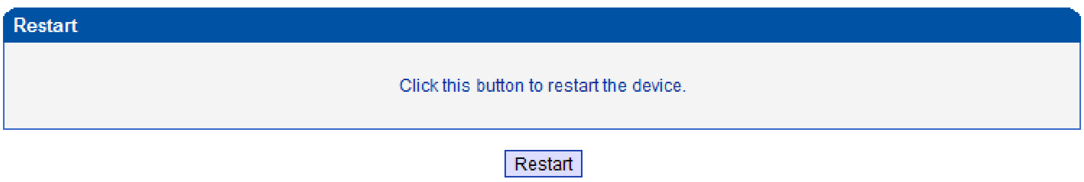
Figure 4-14-9 Factory Reset




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

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