DWG2000C GSM VoIP Gateway

User Manual V1.0

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Table of Contents

1. Equipment Introduction

This chapter mainly introduces functions and structures of DWG2000C-4/8G.

1.1 Introduction

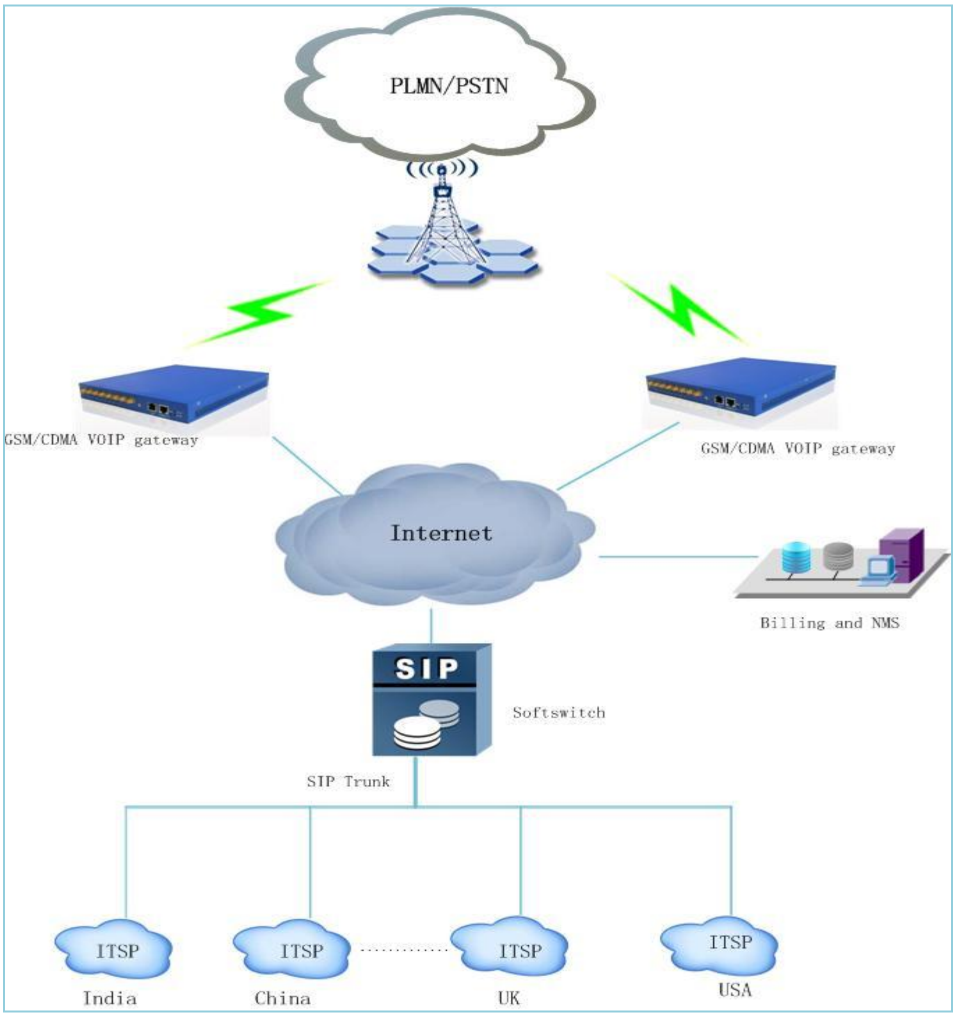
DWG2000C-4/8G is full functions VoIP gateway based on IP and GSM network, which provides a flexible network configuration, powerful features, and good voice quality. It works for carrier grade, enterprise, SOHO, residential users for cost-effective solution.

1.2 Applications Scenario

DWG2000C-4/8G implements smooth transition between PLMN (GSM) and VoIP network.

With the development of users and telecom service, mobile network and fixed network integration will be steadily increasing. DWG2000C-4/8G provides high quality VoIP service which perfectly meets the requirement. A typical application scenario shown as figure 1-2-1

Figure 1-2-1 application scenario



1.3 Product Appearance

The appearance of DWG2000C-4/8G shows as follow

Figure 1-3-1 Front view of DWG2000C-4/8G

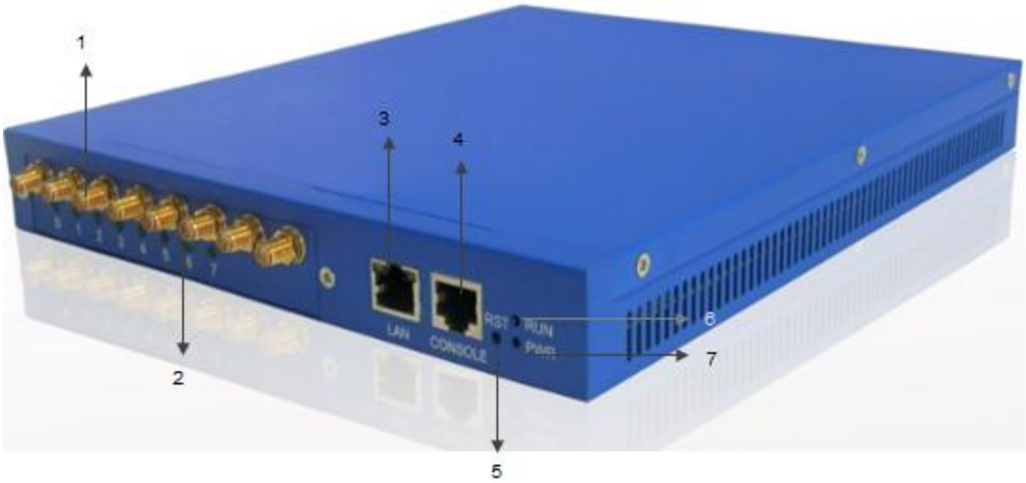


Table 1-3-1 Description of Front view

Index	Sign	Description
1	ANT	Standard antenna interface
2	ANT	Indicator of SIM card, status: register, unregister
3	LAN	10/100M Base-TX, RJ-45
4	Console	Serial port, it is a serial communication physical interface with rs232 standard
5	RST	Keep press for 7 seconds to restore the factory setting
6	Run	Indicate the status of the device.
7	Power	Indicate the status of the power connection

Figure 1-3-2 Rear view of DWG2000C-4/8G

As the picture shown, SIM card slots are located at the bottom of box. The compact shape, coupled with ocean blue brilliant colors make it stylish and practical.

1.4 Functions and Features

1.4.1 Protocol Standard Supported

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

1.4.2 System Function

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

1.4.3 Industrial Standards Supported

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

1.4.4 General Hardware Specification

- Power Supply: DC12V/4A
- Temperature: 0~40 ℃（Operation）, -20~80 ℃（storage）
- Humidity: 5%～90%RH
- Power Consumption: 5W
- Dimensions: 255(W) x220(D) x30(H) mm
- Net weight: 1.48kg

2. Equipment Quickly Installation

This chapter mainly introduces DWG2000C-4/8G hardware installation and connection of equipment.

2.1 Installation Notice

DWG2000C-4/8G adapts 12VDC. Power adapter, make sure AC power supply grounded well to ensure the reliability and stability;

Notes: incorrect power connection may damage power adapter and device.

DWG2000C-4/8G provides standard RJ45 with 10Mbps or 100Mbps interfaces.

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

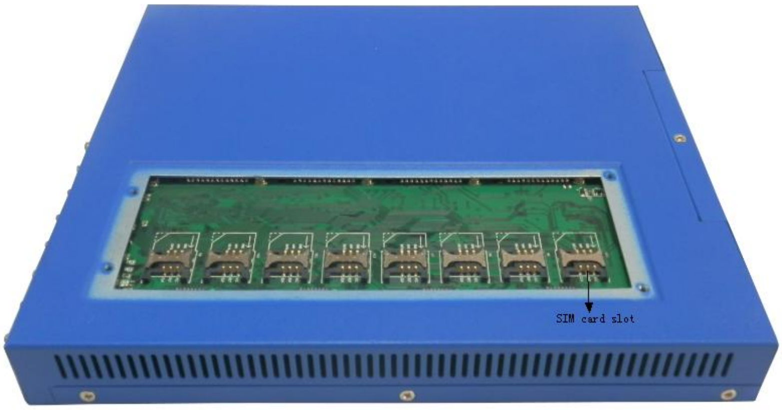
2.2 Installation Procedure

2.2.1 Install SIM Card

When installing SIM card, loosen the screws on the back of a small piece of blue backplane. Procedure shows as below:

- Open the blue backplane
- Inset the SIM card to the SIM slot
- Cover the backplane
- Tighten the screws

Figure 2-2-1 SIM card Installation

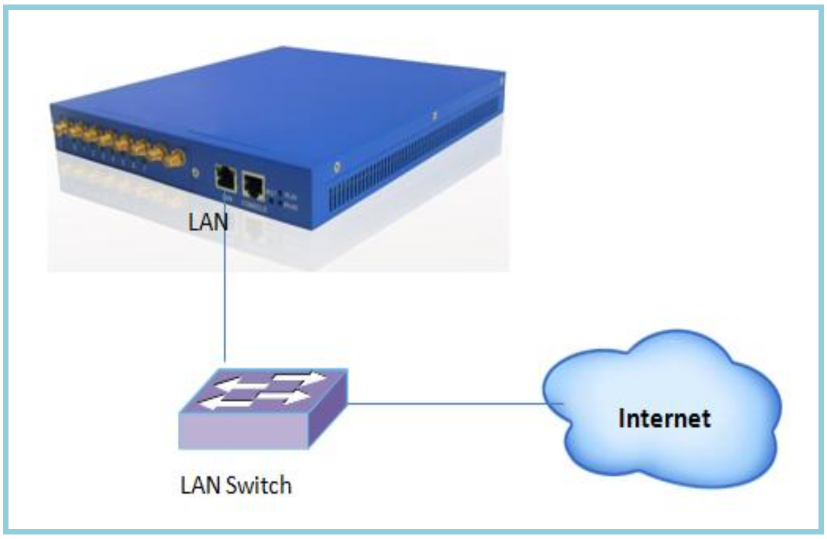


2.2.2 Antenna Installation

Figure 2-2-2 Antenna Installation

2.2.3 Network Cable Connection of Equipment

Figure 2-2-3 DWG2000C-4/8G connection



3. Network Configuration

In this chapter we will introduce the initial configuration of DWG2000C-4/8G. All of the network parameters of the gateway can be configured by IVR guidance.

3.1 Attentions

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

3.2 General Feature Codes for System Setting

Table 3-3-1 Feature codes for system setting

Feature codes	Description
*114#	Play SIP user ID
*150*a#	Set IP address(static/DHCP), a can be digit 1 or 2,*150*1# is static IP address
*152*a*b*c*d#	Configure IP address, a, b, c, d are the four fields of IP address.
*153*a*b*c*d#	Configure subnet mask. a, b, c, d are the four fields of the subnet mask
*156*a*b*c*d#	Configure the device gateway, a, b, c, d are the four fields of the device gateway
*158#	Report LAN port IP address
*159#	Report WAN port IP address
*157	Setting the work mode (route or bridge), * 157 * 0 # is route mode, * 157 * 1 # is bridge mode
*195#	Play record
*198#	Clear record
*199#	Setting Record. dial*199# start record(≤ 20s), then press # end the recording
*111#	Restart device

3.3 Static IP Configuration

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

Assume that DWG2000C-4/8G IP address to be 172.16.80.89, subnet mask is 255.255.0.0, default gateway is 172.16.1.1, configure it through IVR as following steps:

1) Please make sure SIM card installed well and registered
2) Dial the phone number of the SIM card. Press "*150*1#" after heard "dial the extension number ". Hang up after heard “setting successful" prompt.
3) Dial the phone number of the SIM card. Dial "* 152 * 172 * 16 * 80 * 89 #"after heard “dial the

extension number ". Hang up after heard “setting successful".

4) Dial the phone number of the SIM card. Dial "*153*255*255*0*0#" after heard "dial the extension number ". Hang up after heard "setting successful"
5) Dial the phone number of the SIM card. Dial "*156*172*16*1*1#" after heard "dial extension number ". Hang up after heard "setting successful"
6) Dial the phone number of the SIM card. Dial "*111#" after heard "dial extension number ", that will restart the device
7) Dial the phone number of the SIM card. Dial "*158#" after heard “dial extension number ". It will play report the IP address of LAN port.

3.4 DHCP Configuration

DHCP mode configure as follows:

1) Please make sure hardware installation have finished
2) Dial the phone number of the SIM card. Dial "*150*2#" after heard "dial extension number ". That means the DHCP is configured 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

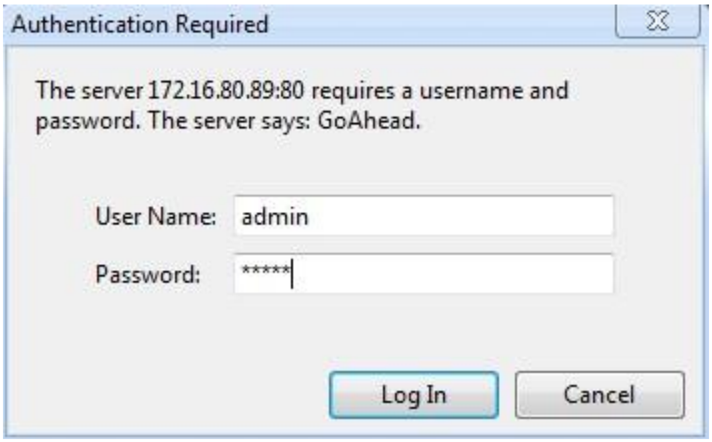
This charpter describes web configuration of DWG2000C-4/8G.

4.1 Access the System through HTTP

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

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

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”. We are strongly recommend to change the default password for security purpose.

4.2 WEB Configuration

DWG2000C-4/8G WEB configuration interface consists of the navigation tree and the detail configuration interfaces.

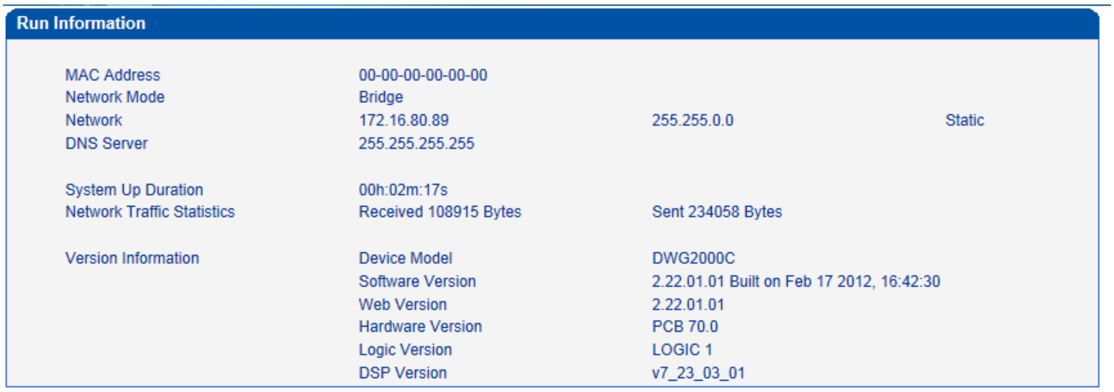
Figure 4-2-1 WEB introduce

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



Run Information

MAC Address	00-00-00-00-00-00		
Network Mode	Bridge		
Network	172.16.80.89	255.255.0.0	Static
DNS Server	255.255.255.255		
System Up Duration	00h:02m:17s		
Network Traffic Statistics	Received 108915 Bytes	Sent 234058 Bytes	
Version Information	Device Model	DWG2000C	
	Software Version	2.22.01.01 Built on Feb 17 2012, 16:42:30	
	Web Version	2.22.01.01	
	Hardware Version	PCB 70.0	
	Logic Version	LOGIC 1	
	DSP Version	v7_23_03_01	

Table 4-3-1 Description of system information

MAC Address	Displays current MAC of device, for example: 00-00-00-00-00-00
Network Mode	DWG2000C-4/8G 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	GSM	460021180311884	Mobile Registered	No Limit	CHINA MOBILE		0	0	0	0	Idle
1	GSM		No SIM Card	No Limit			0	0	0	0	Idle
2	GSM		No SIM Card	No Limit			0	0	0	0	Idle
3	GSM	460003270439138	Mobile Registered	No Limit	CHINA MOBILE		0	0	0	0	Idle
4	GSM		No SIM Card	No Limit			0	0	0	0	Idle
5	GSM		No SIM Card	No Limit			0	0	0	0	Idle
6	GSM		No SIM Card	No Limit			0	0	0	0	Idle
7	GSM		No SIM Card	No Limit			0	0	0	0	Idle

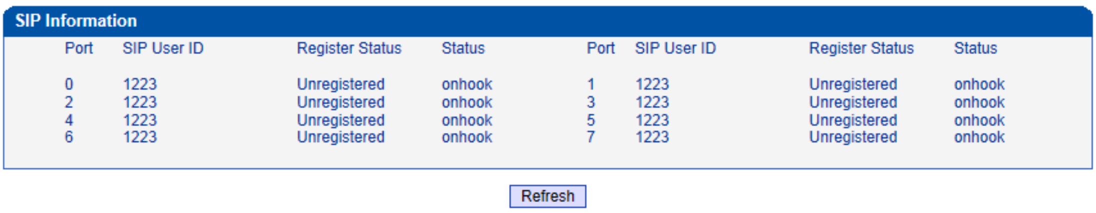
Table 4-3-2 Description of mobile information

Port	GSM channel number, it is range from 0 to 7
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 register status of current GSM module
Remaining Call Duration	Limited call duration of SIM card, when call duration is out of that duration, the call would be disconnected. This option shows remaining talk time.
Carrier	Displays the network carrier of current SIM card.
Signal Quality	Displays the signal strength of in each channels of GSM

BER	Bit error rate, internal parameter
ASR	Answer Seizure Ratio is a measure of network quality. Its calculated by taking the number of successfully answered calls and dividing by the total number of calls attempted. Since busy signals and other rejections by the called number count as call failures, the ASR value can vary depending on user behavior.
ACD	The Average Call Duration (ACD) is calculated by taking the sum of billable seconds (billable) of answered calls and dividing it by the number of these answered calls.
PDD	Post Dial Delay (PDD) is experienced by the originating customer as the time from the sending of the final dialled digit to the point at which they hear ring tone or other in-band information. Where the originating network is required to play an announcement before completing the call then this definition of PDD excludes the duration of such announcements.
Call Status	Show the status of call, its include 3 type of status : Idle: the GSM channel is free. It is ready to receive the call Processing: the call is delivering to mobile network Active: the call is established

4.3.3 SIP Information

Figure 4-3-3 SIP information



Displays registration status information with Softswitch platform or SIP Server

Table 4-3-3 Description of SIP information

Port	The number of SIP channel, DWG2000C-4/8G has 4/8 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 PRI Statistics

RTP

Port	Payload Type	Packet Period	Local Port	Peer IP	Peer Port	Send Packet	Recv Packet	Loss Packet	Jitter	Duration Time(s)
	---	---	---	---	---	---	---	---	---	---

Refresh

Table 4-4-2 Description of RTP Statistics

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

4.4.3 SIP Call History

Figure 4-4-3 SIP Call History

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

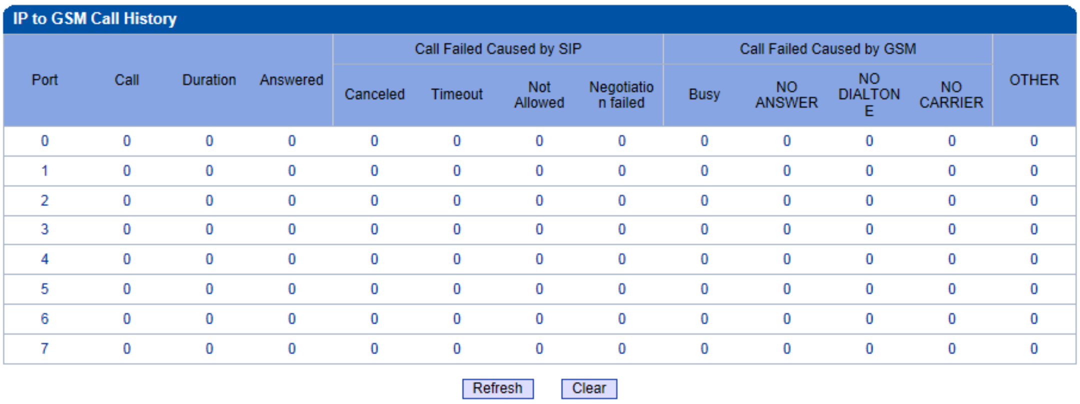
Refresh

Table of 4.4.3 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

Refresh Clear

Table of 4.4.4 IP to GSM Call History

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

4.5Network Configuration

4.5.1 Local Network

Figure 4-5-1 Local Network

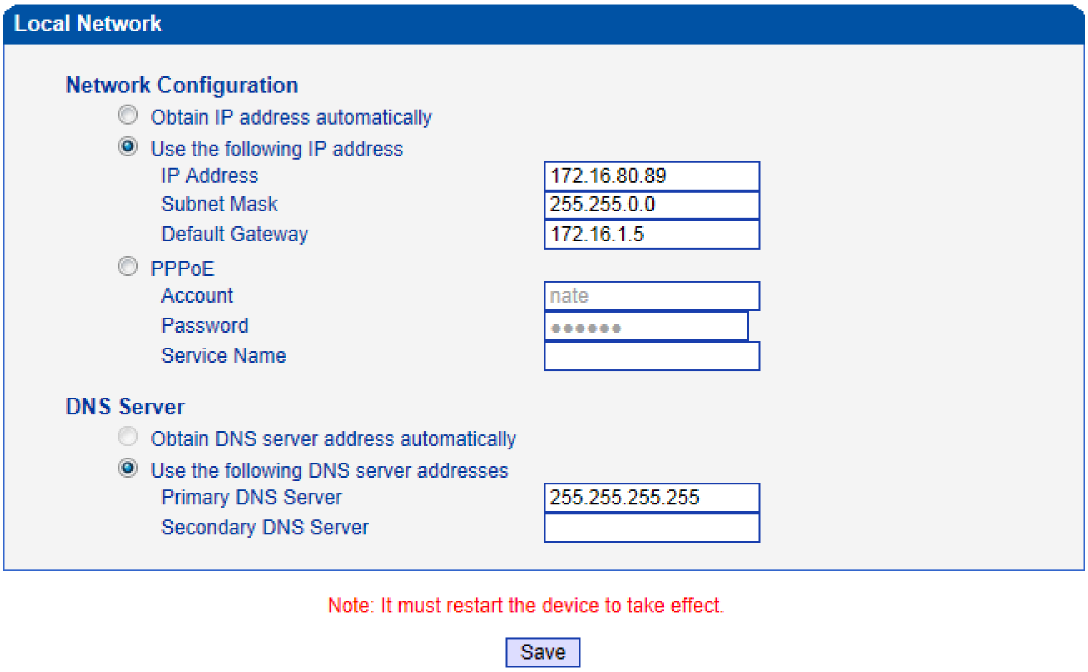


Table 4-5-1 Description of Local network

Obtain IP Address Automatically	Enable the device obtain IP Address automatically by DHCP or not. Default is enabling
Use the Following IP Address	Configure the "IP Address", "Subnet Mask" and "Default Gateway" by manual
PPPoE	Both of account and password are provided by ISP. 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

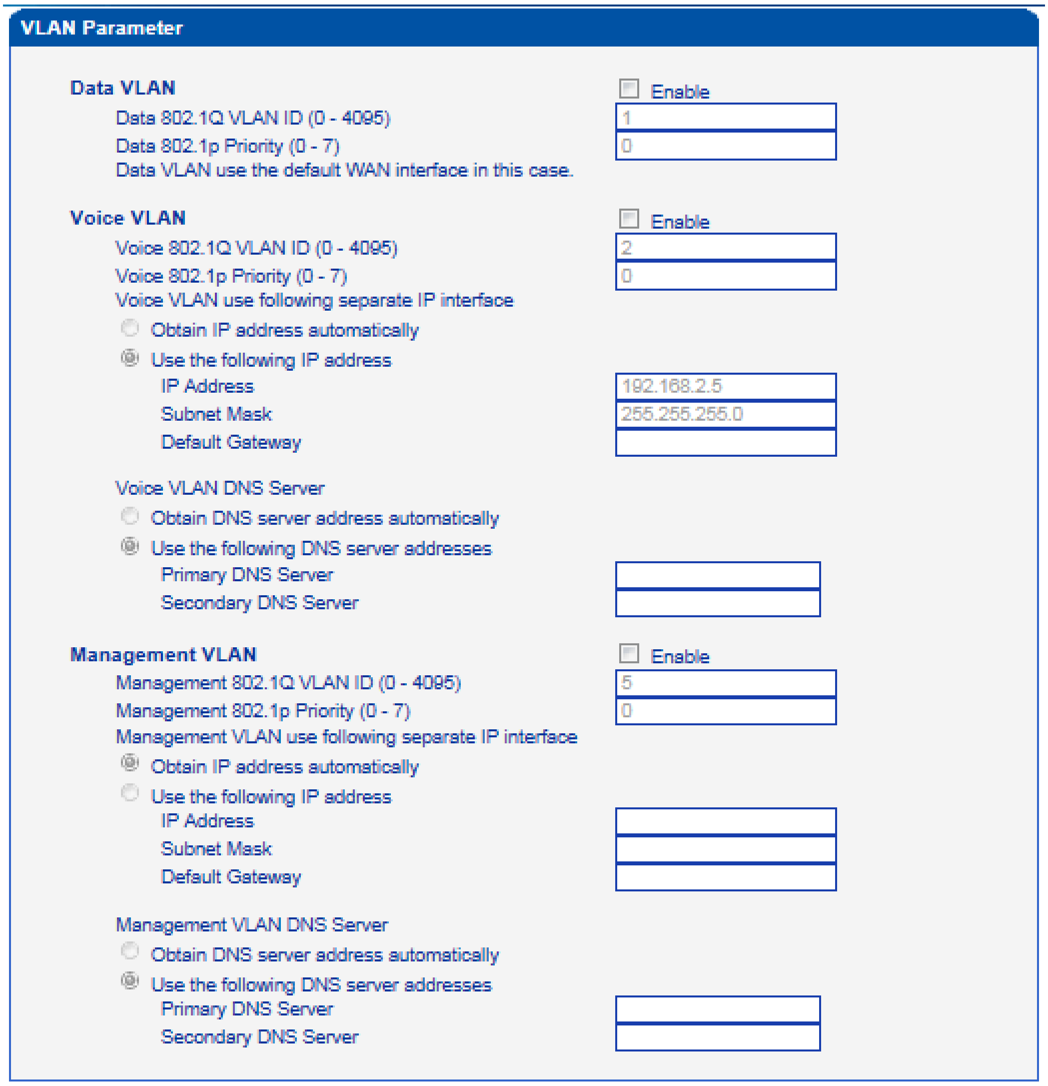


Table 4-5-2 Description of VLAN Parameter

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	Under 802.1p protocol users can set VLAN priority

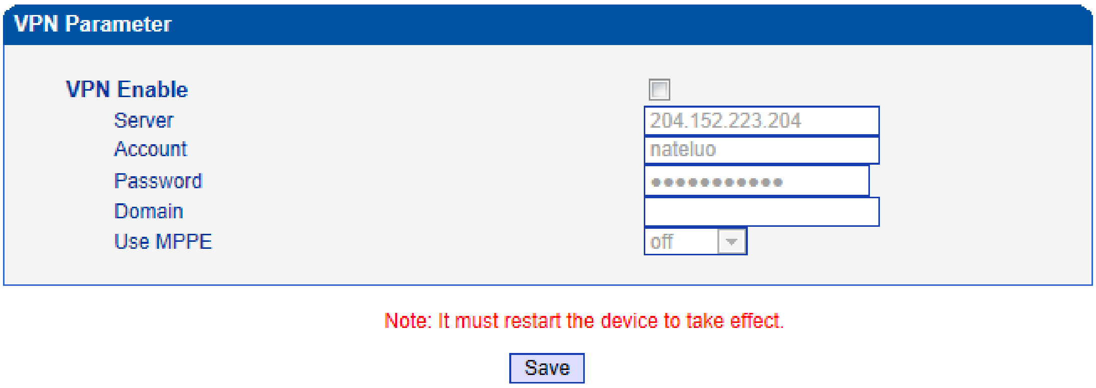
	(0-7)	
	IP address	Users can set DHCP or static IP address
	Management VLAN DNS Server	Users can set DHCP or static DNS server IP address

4.5.3 VPN Parameter

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

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

Figure 4-5-3 VPN Parameter



4.5.4 ARP

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

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

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

Figure 4-5-4 ARP

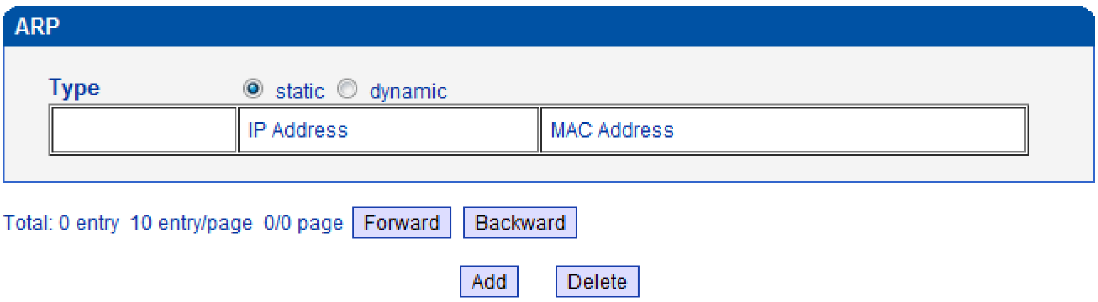
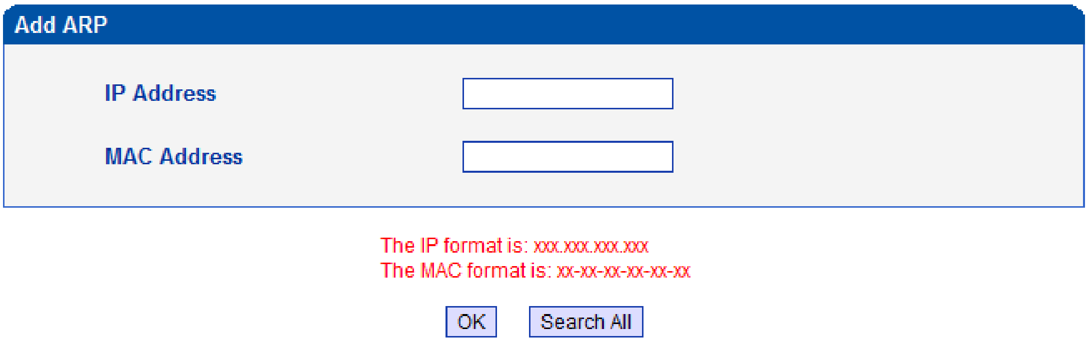


Figure 4-5-5 Add ARP



4.6 Mobile Configuration

This is Mobile Configuration menu.

Figure 4-6-1Basic Configuration

4.6.1 Basic Configuration

Figure 4-6-2Basic Configuration

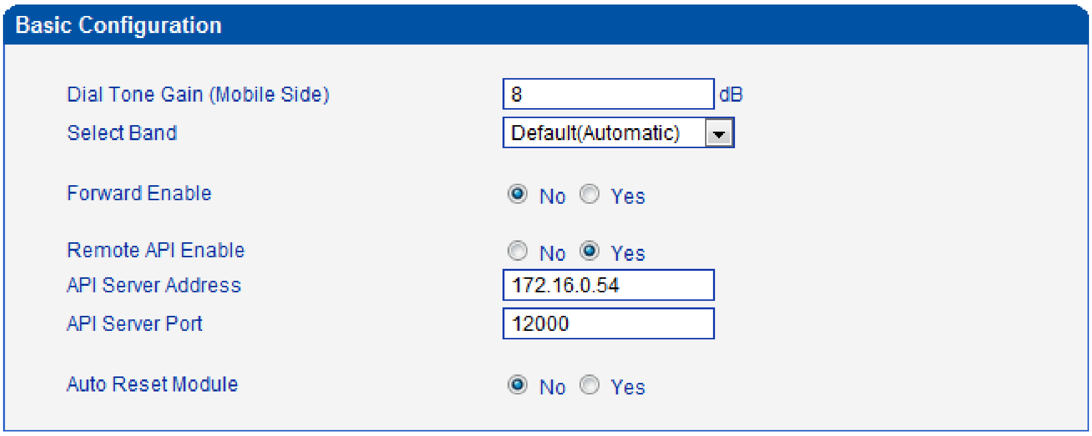


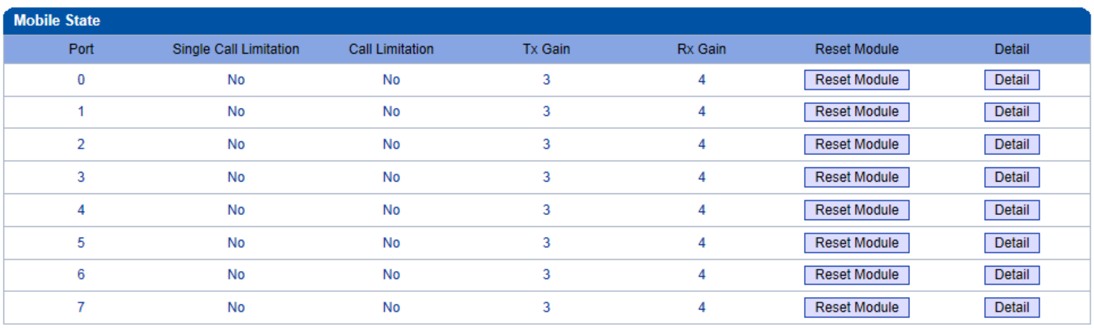
Table 4-6-1 Description of Basic Configuration

Dial Tone Gain	It is the dial tone volume of call waiting, dial tone of mobile module when call out. default value is 8 dB.
Select Band	Acording to carrier’s band standards. Standards are as belows: GSM: 850/900/1800/1900 MHz; CDMA: 800 MHz

Remote API Enable	API is mainly for third party software which developed based-on Dinstar API protocol. Its help to provide bulk SMS/ SMS/USSD over IP service.
API Server Address	It is the remote IP address who uses API. This is an option when selecting "Yes" under 'remote API enable"
API Server Port	It is the port number of IP transmission. This is an option when selecting "Yes" under "remote API enable". The port cannot conflict with the other application software. The default value is 12000
Auto Reset Module	Reset modular by automatically while some special errors happened, such as No Carrier, No Dial tone
Counts of No CARRIER to reset	A kind of the error, continuously N times will reboot the modular. N is range from 3 to 255.
Counts of No DIAL TONE to reset	A kind of the error, continuously N times will reboot the modular. N is range from 3 to 255.

4.6.2Mobile 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	3	4	Reset Module	Detail
1	No	No	3	4	Reset Module	Detail
2	No	No	3	4	Reset Module	Detail
3	No	No	3	4	Reset Module	Detail
4	No	No	3	4	Reset Module	Detail
5	No	No	3	4	Reset Module	Detail
6	No	No	3	4	Reset Module	Detail
7	No	No	3	4	Reset Module	Detail

Figure 4-6-3 Mobile Configuration

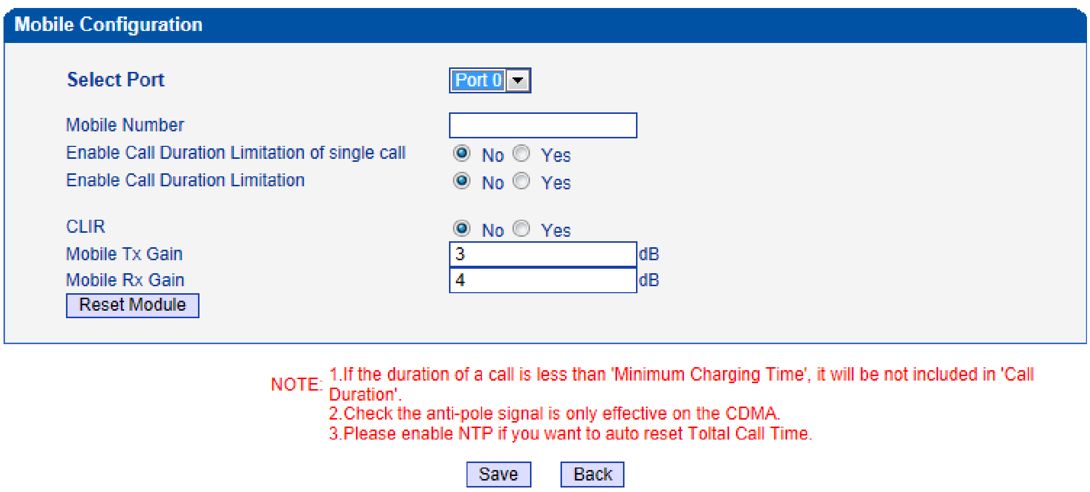


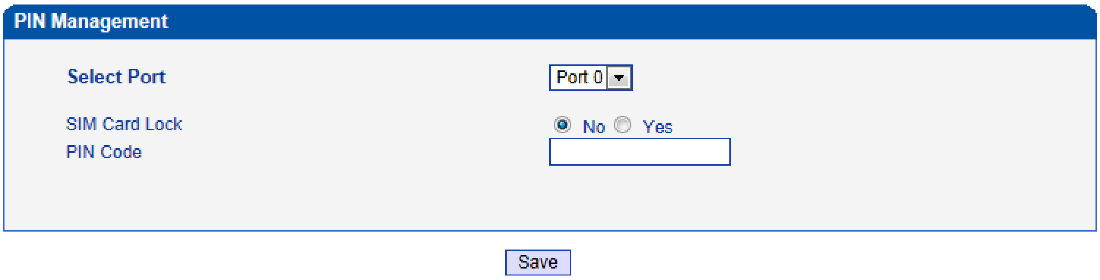
Table 4-6-2 Description of Mobile Configuration

Mobile Number	Phone number of current SIM card
Enable Call Duration Limitation of 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
Step	Step length value range is 1-120 s, step length multiplied by time of single call just said a single call duration time allowed.
Time of single call	The value of limitation single call, this value range is 1-65535. step length multiplied by time of single call just said a single call duration time allowed.
Enable Call Duration Limitation	This function is to limit the total call duration of GSM channel. The max call duration is between 1 to 65535 minutes.
Auto Reset	Automatic restore remaining talk time, that is, get total call minutes of GSM channel
Reset Period	Reset call minutes by date, by week, by month
Next Reset time	Defined next reset date, system will count start from that date and work as Reset Period setting
Minimum Charging Time	A single call over this time, GSM side of the operators began to collect fees, unit for seconds.
Alarm Threshold(via SMS)	Define a threshold value of call minutes, while the call minutes less than this value, the gateway will send alarm information to designated phone number via SMS.
Mobile Number (Receiving Alarm)	Receiving alarm phone number, user will received alarm message from gateway.
Port Description for Alarm	Alarm port information description, which will be sent to user mobile phone with alarm information.
SIM Remain Time	This value is multiplied by to step length is a rest call time
Restore Time	Restore the rest of the SIM card talk time to the maximum call duration
CLIR	Caller ID restriction, this function is used to hidden caller ID of SIM card number. The gateway will add "#31#" in front of mobile number. This function must support by Operator.

Mobile Tx Gain	Control IP to GSM side of call the gain. Default is +6 dB.
Mobile Rx Gain	Control GSM to IP side of call the gain. Default is +6 dB.

4.6.3 PIN Management

Figure 4-6-4 PIN Management



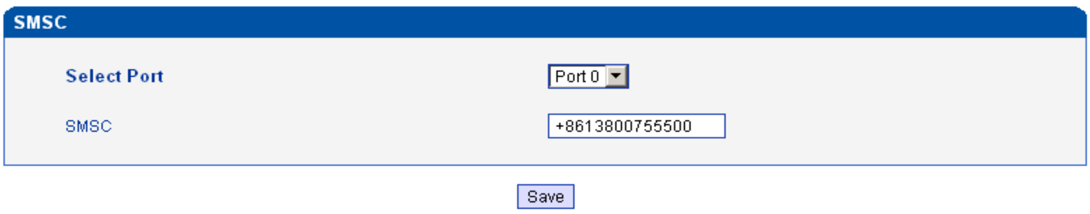
Detailed description as below:

Table 4-6-4 Description of PIN Management

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

4.6.5 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 Message

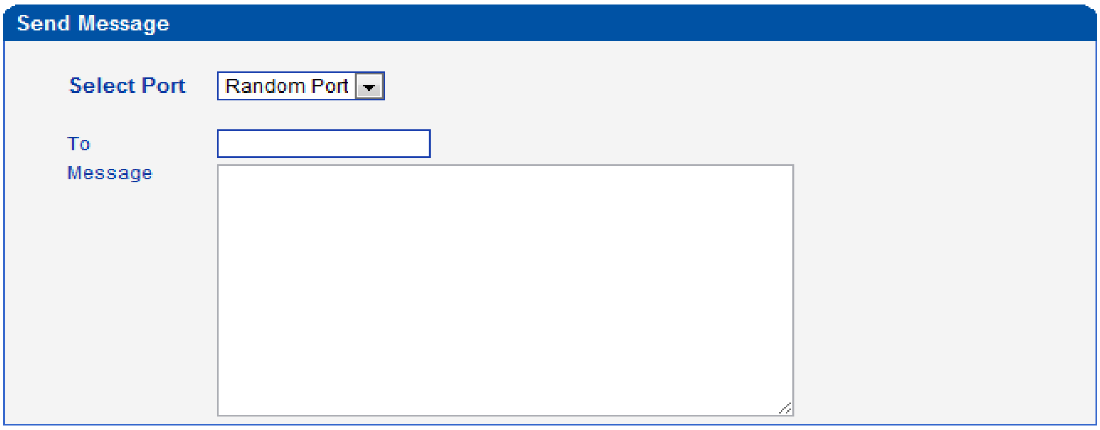


Table 4-6-5 Description of SMS sending

Select Port	Users can select a defined channel or random channel to send SMS. Input the recevier’s mobile phone number to send SMS.
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. USSD can cluster and group of charge

Figure 4-6-7 USSD

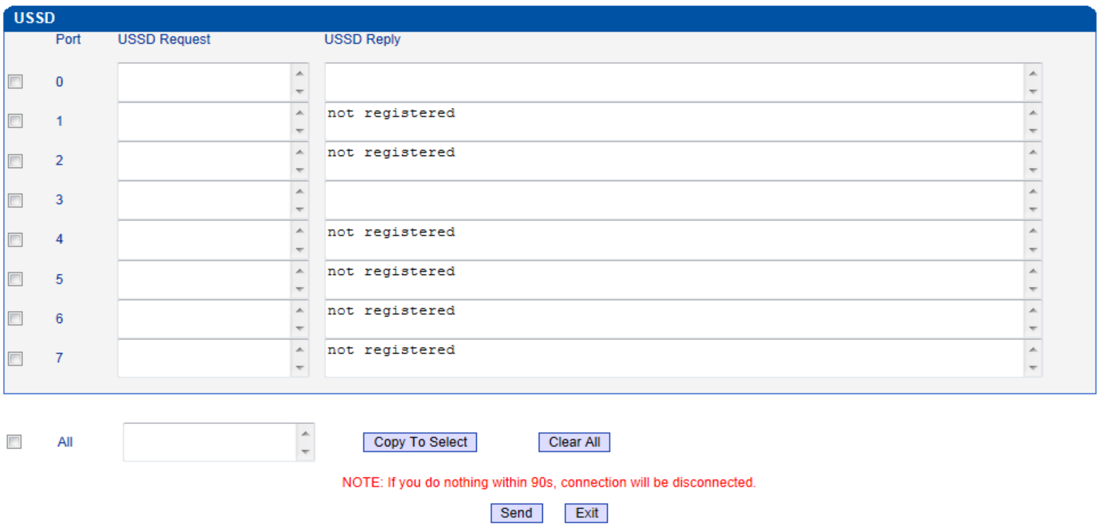


Table 4-6-6 Description of USSD

Port	Select the GSM channel to send USSD
USSD Reply	The area to input USSD code
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

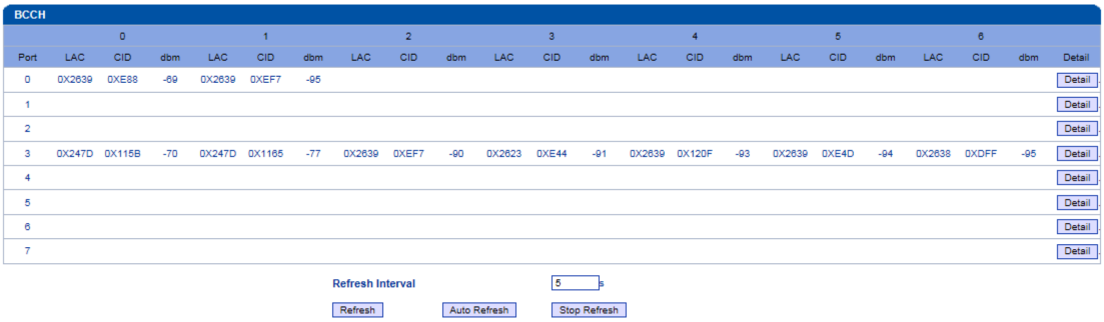


Figure 4-6-10 BCCH

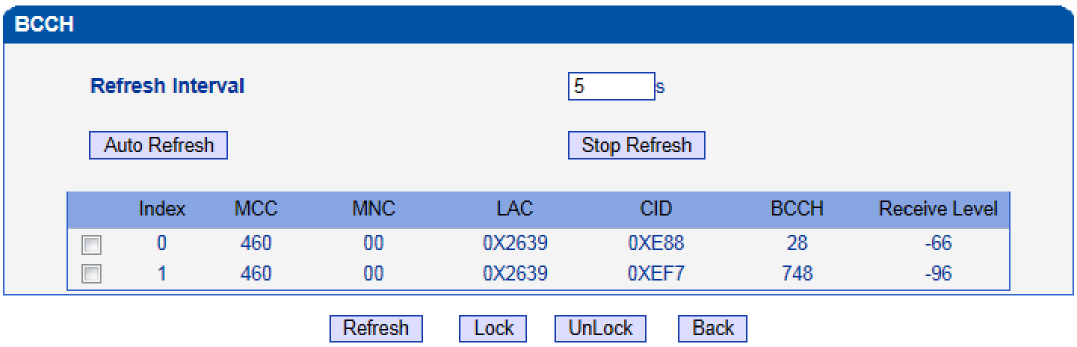


Table 4-6-7 Description of BCCH

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

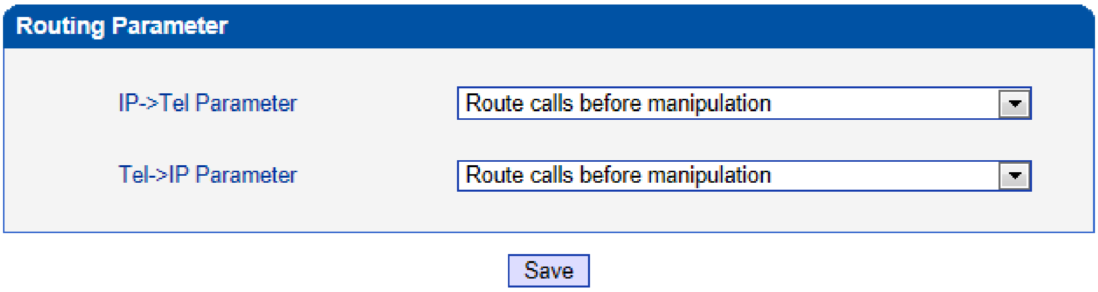


Table 4-7-1Description of Routing Parameter

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

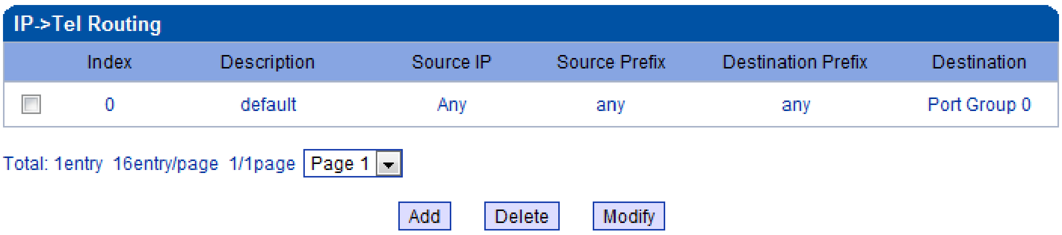


Figure 4-7-3 IP to Tel Routing Add

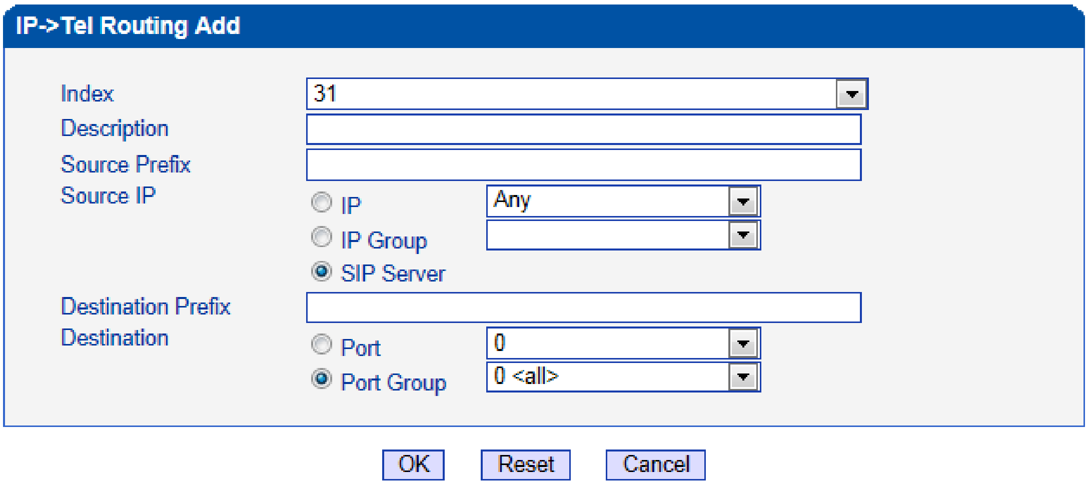


Table 4-7-2 Description of IP to Tel Routing

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

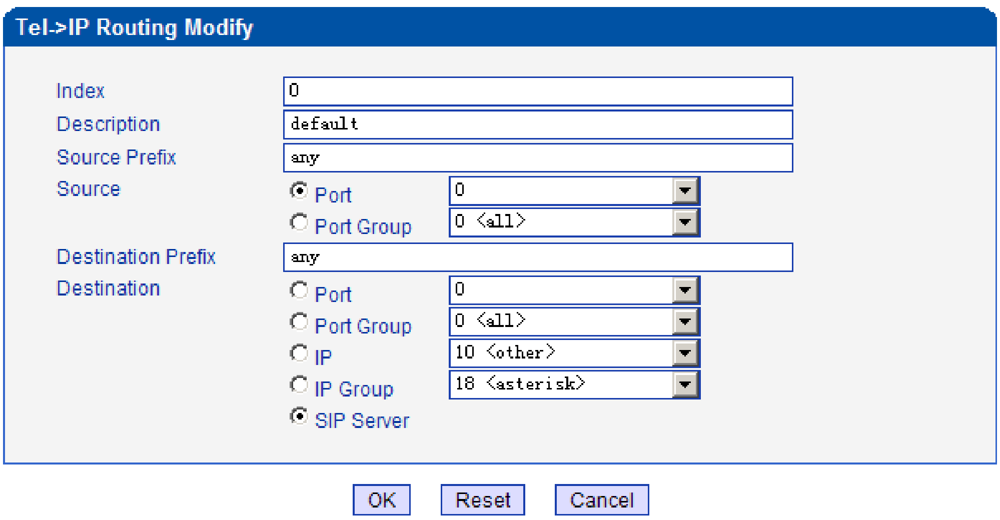
4.7.3 Tel->IP Routing

Figure 4-7-3 Tel to IP Routing

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

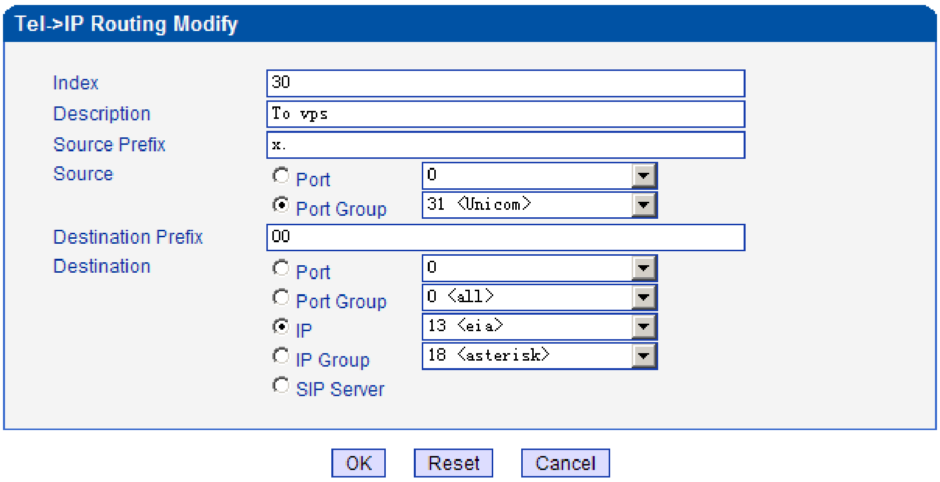
Tel -> IP Routing	This item uses to configure incoming call routes which can be used for recieve the calls from the GSM.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.The route preferentially match the rules which the value of index is smaller
Description	It describes the route for the ease of identification. Its value is character string
Source Port	It specifies the Port or Port Group which will receive the calls from PLMN
Source Prefix	All the caller number must match the source prefix. It specifies the source prefix allow to send call out • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186
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 4-7-4 Tel to IP routing Modify



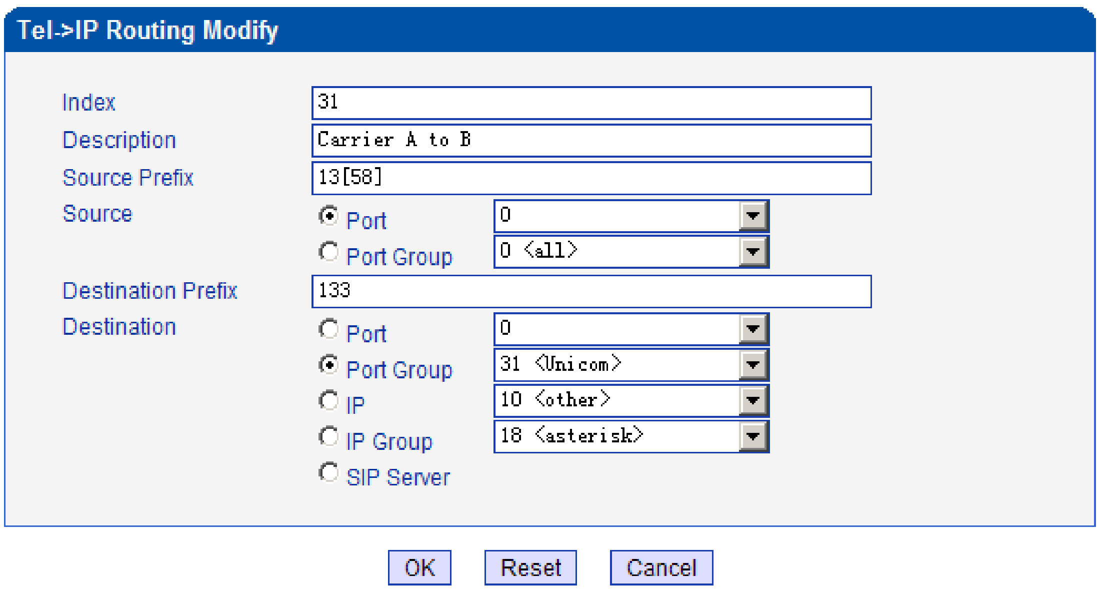
It's a default route configured in gateway. It allows any number from source port 0 send call to SIP server with any prefix.

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

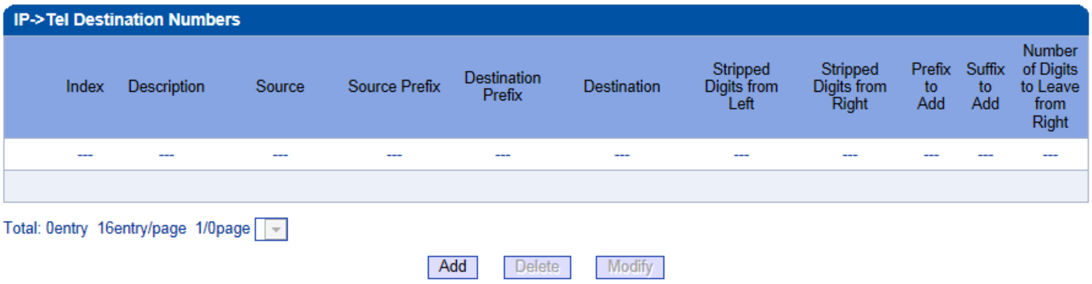
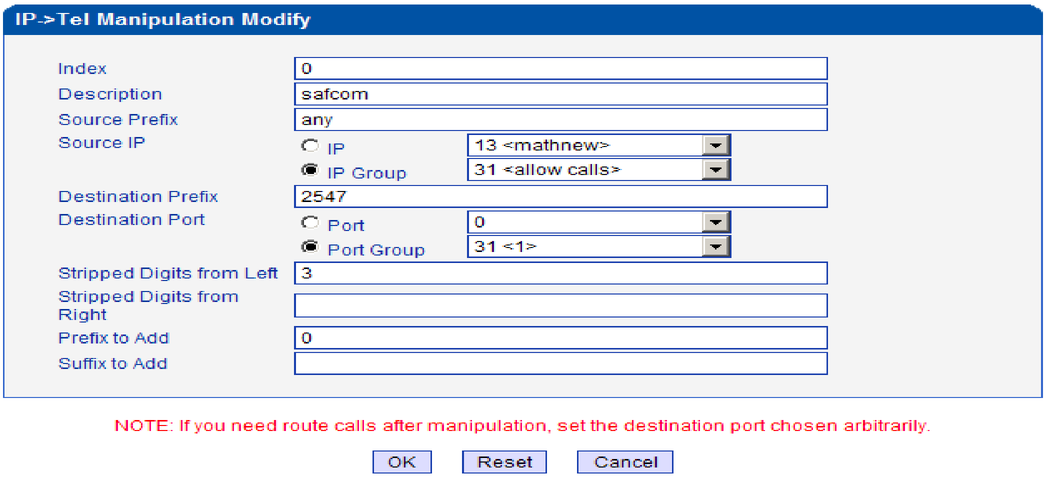


Table 4-8-1 Description of IP->Tel destination numbers manipulation

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	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
Number of Digits to leave from right	It specifies the length of the digits to be deleted from left

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

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



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

4.8.2 Tel->IP Source Numbers

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

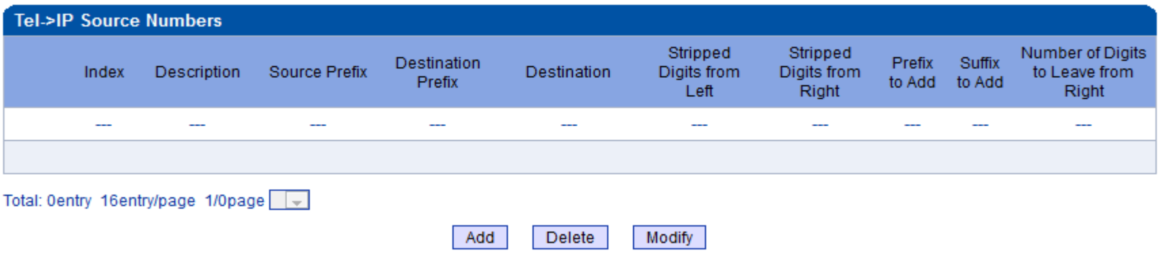


Table 4-8-2 Description of Tel->IP destination numbers manipulation

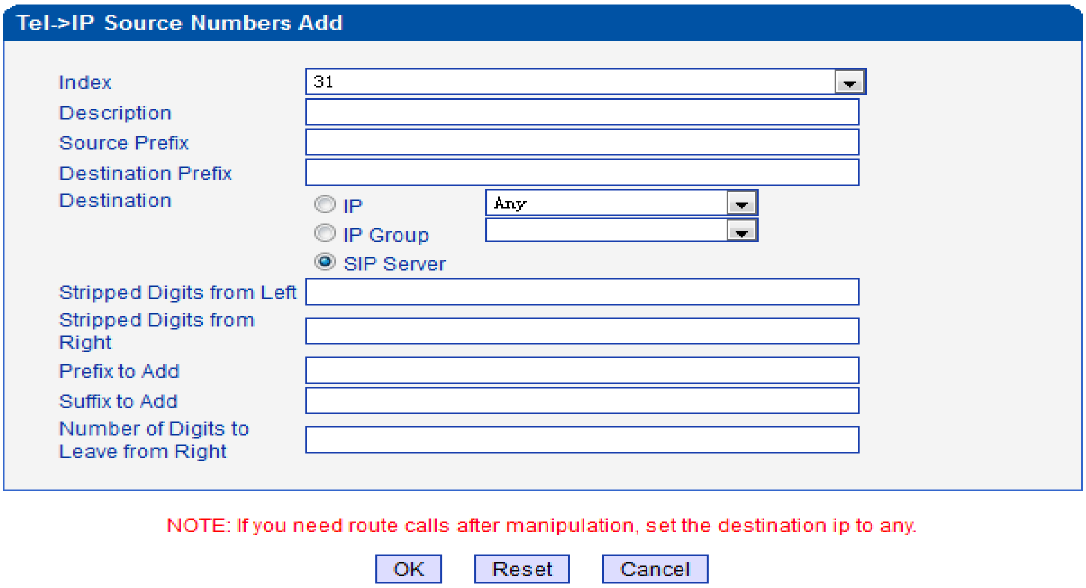
Tel->IP destination numbers manipulation	It is an optional configuration item, and is used to add IP->Tel number change data. The IP->Tel Manipulation defined the rules of add, and deletion of called numbers, which are referenced by IP->Tel routing.
Index	It uniquely identifies a route. Its value is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

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

Example:

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

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



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

4.8.3 Tel->IP Destination Numbers

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

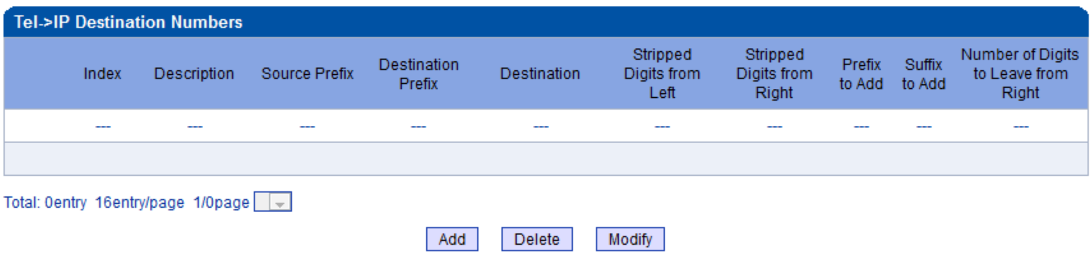


Table 4-8-3 Description of Tel->IP destination numbers manipulation

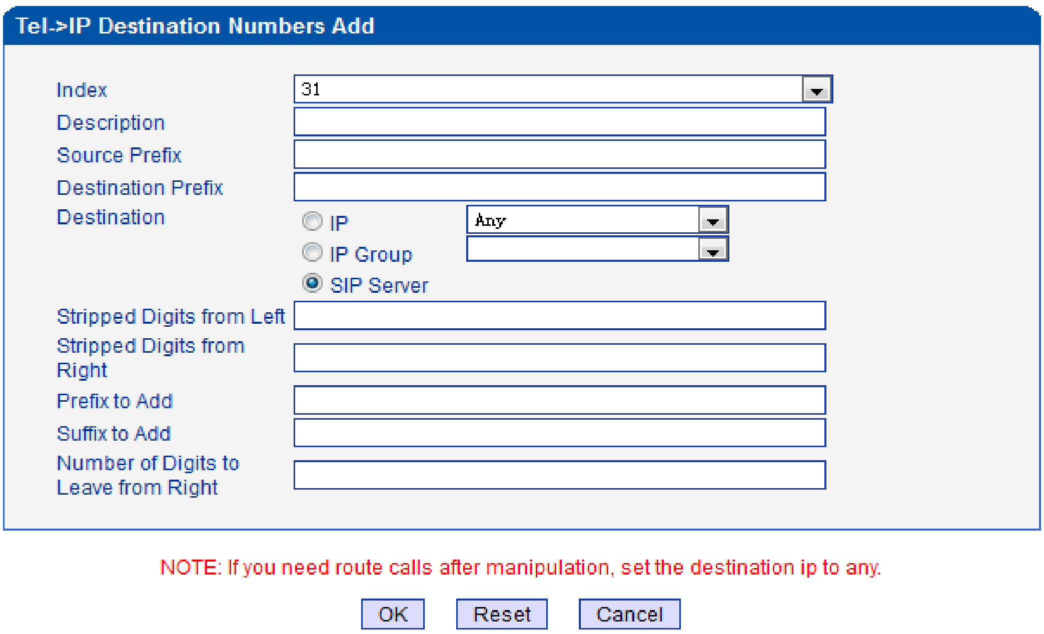
Tel->IP destination	It is an optional configuration item, and is used to add IP->Tel number change

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

Example:

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

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



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

4.9 Operation

When configure hotline, must configure 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

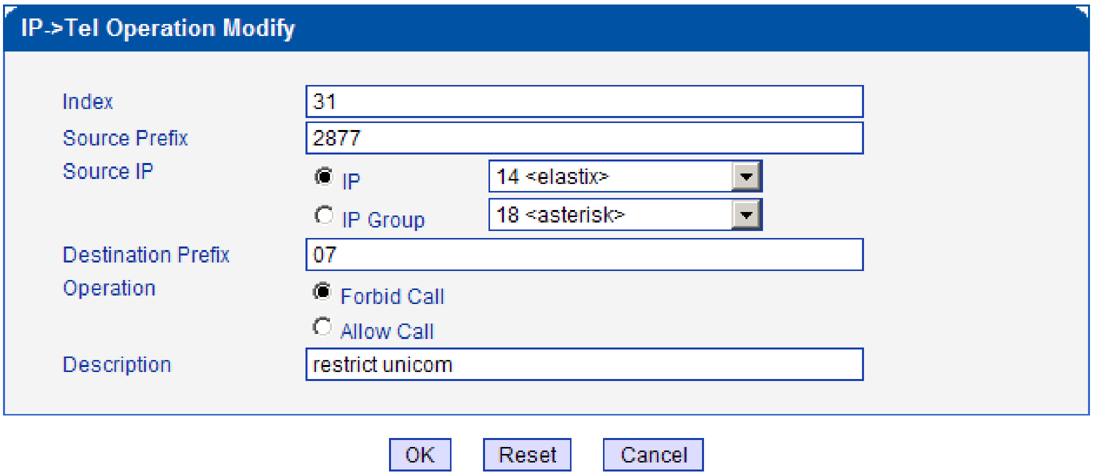
Table 4-9-1 Description of IP->Tel Operation

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 • 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>

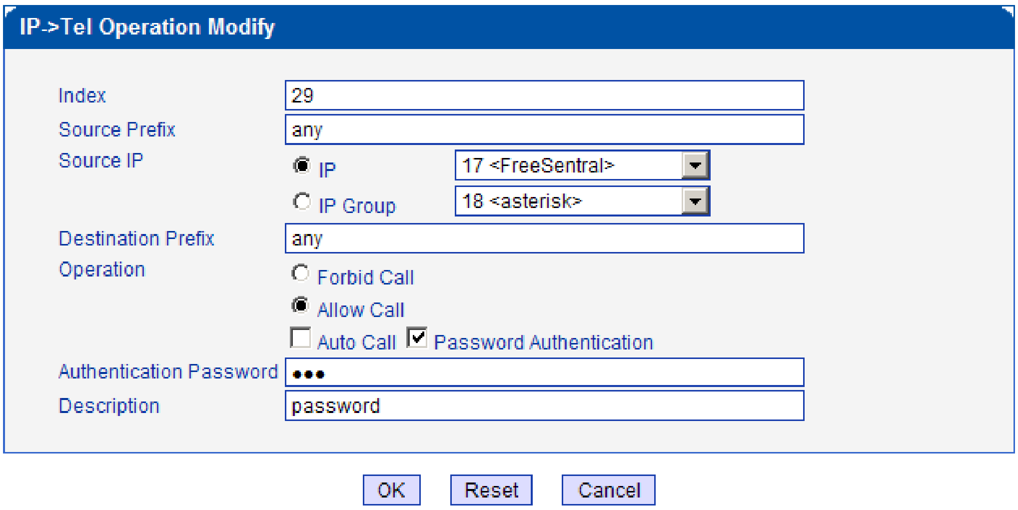
Figure 4-9-2 IP->Tel Operation Modify



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

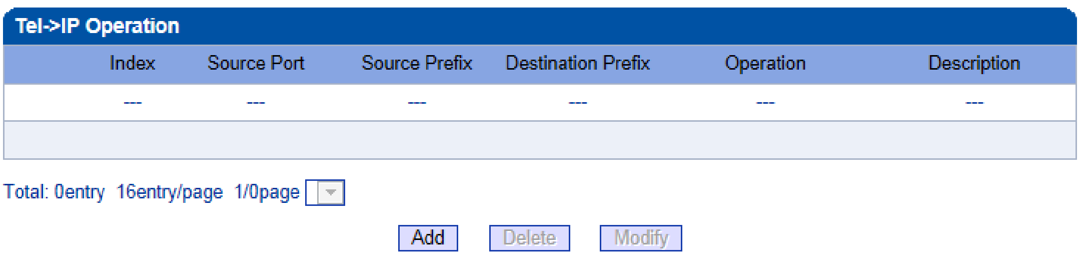


Table 4-9-2 Description of Tel->IP Operation

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

4.10 Port Group Configuration

4.10.1 Port Group

Figure 4-10-1 Port Group

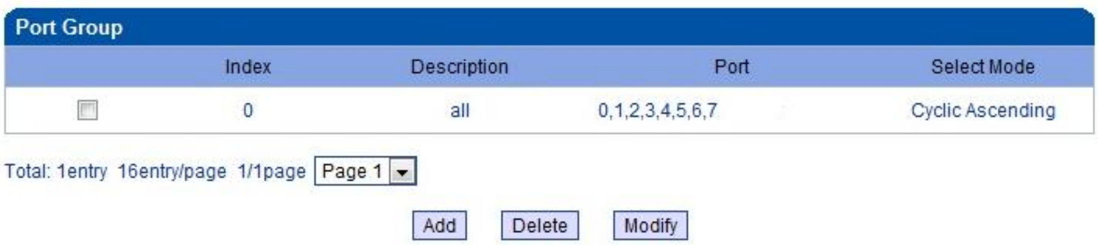


Figure 4-10-2 Port Group Modify

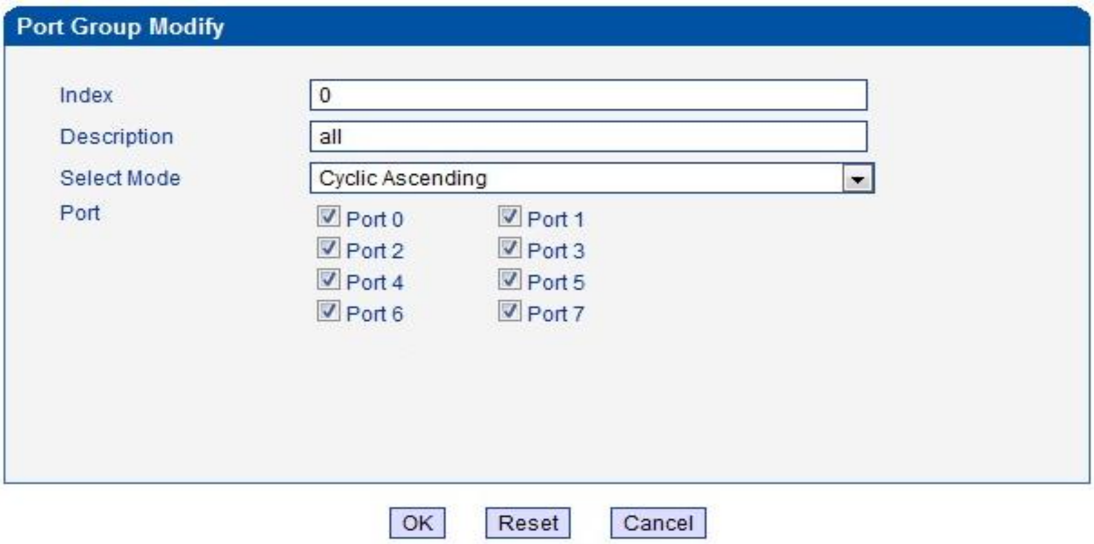


Table 4-10-1 Description port group

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

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

4.11 IP Trunk Configuration

4.11.1 IP Trunk

Figure 4-11-1 IP Trunk

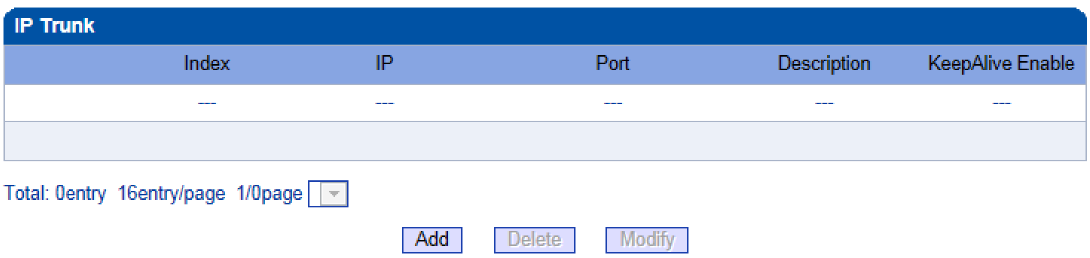


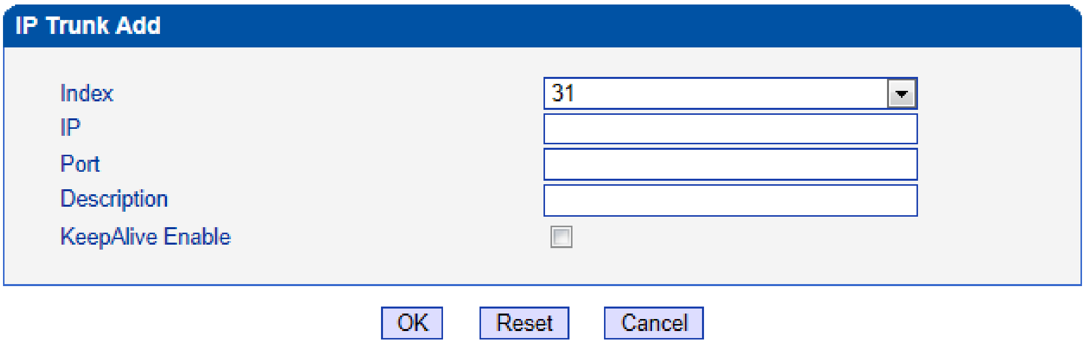
Table 4-11-1 Description of IP Trunk

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

Example

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

Figure 4-11-2 IP Trunk Modify



4.11.2 IP Trunk Group

Figure 4-11-3 IP Trunk Group

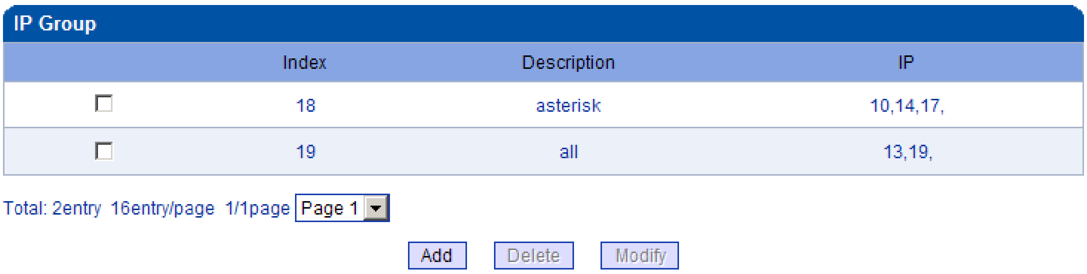


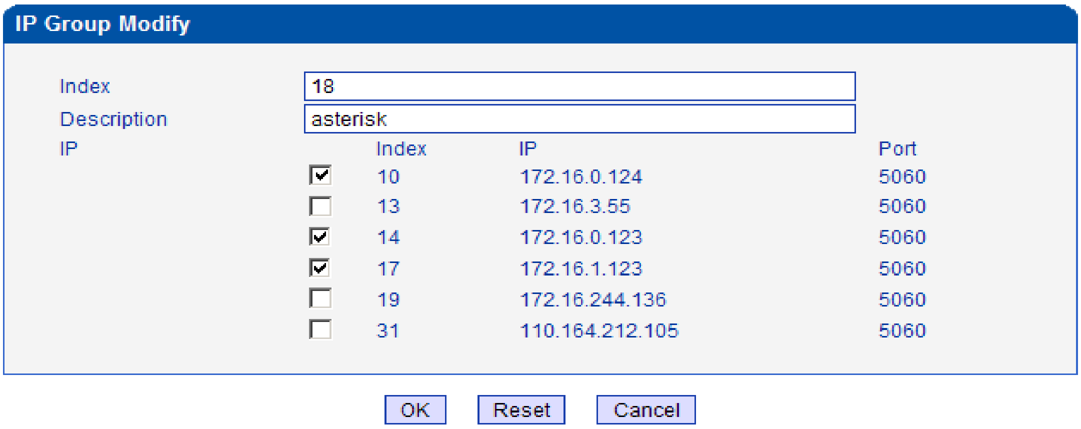
Table 4-11-2 Description of IP Trunk Group

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

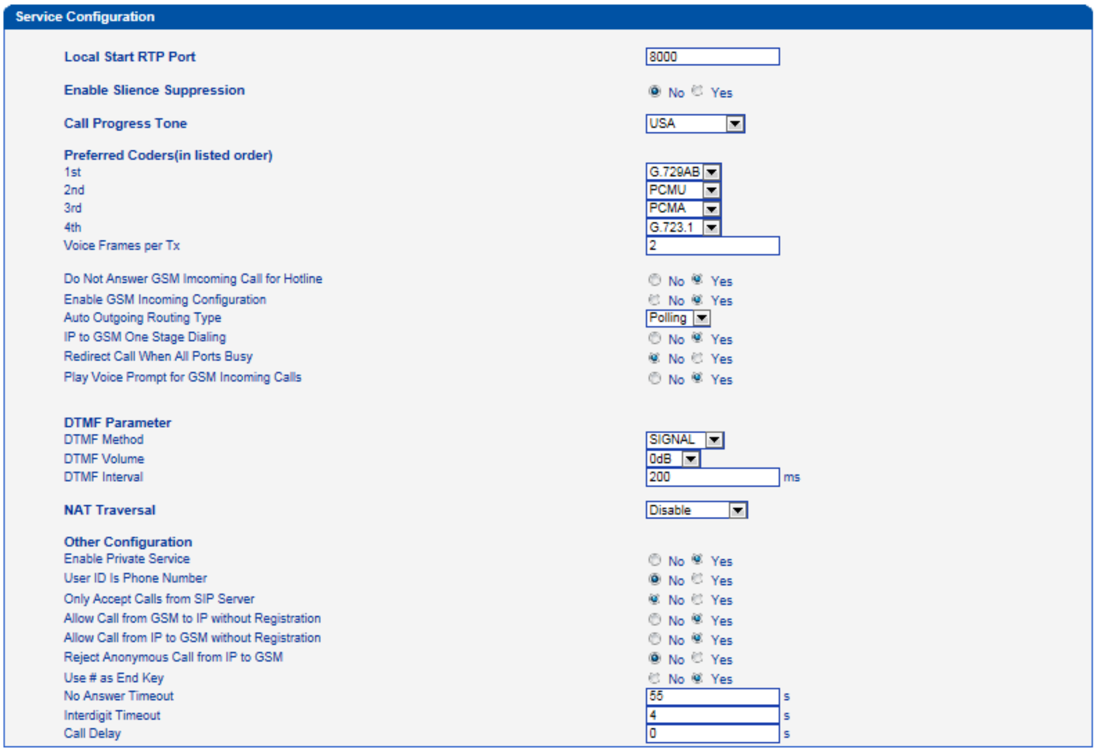


Table 4-12-1 Description of Service Configuration

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.
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/DWG2000C-4/8G 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	The incoming anonymous calls will be rejected

from IP to PSTN	
Use # as End Key	In General, SIP phones are based on # as the end, if this option is set to No, the dial-up will end expires dial-up time
Interdigit Timeout	Bit of between the dialing time ,over the time will be seem as end of dial

4.12.2 SIP Configuration

Figure 4-12-2 SIP Configuration

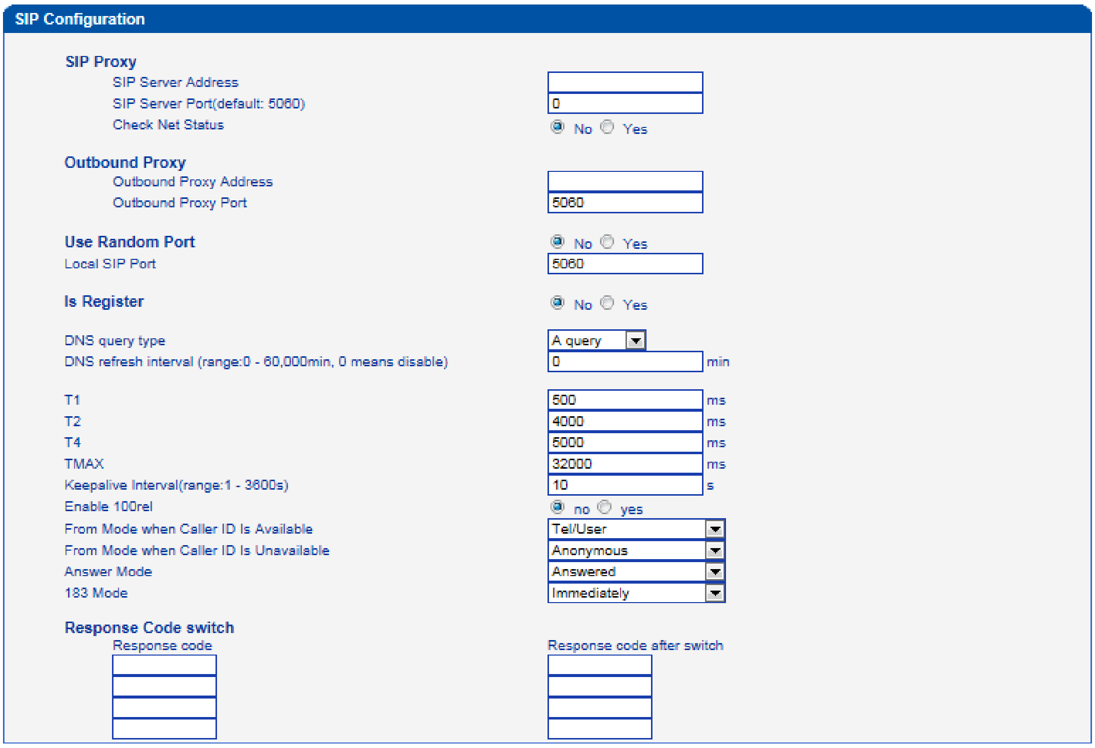


Table 4-12-2 SIP Configuration

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

DNS query type	The DNS query type defines the type of information that will be requested from DNS server
DNS refresh interval	The interval of DNS refresh, Range from 0 to 60000 mins, 0 means disable default value is disable.
T1	Used to define the SIP protocol T1 timer value, default is 500ms
T2	Used to defines the SIP protocol timer values, default value is 4000ms
T3	Used to define the T2 timer value in SIP protocol, the default is 5000ms
Keep alive Interval	Used to keep communicate between equipment and the SIP server that make the device is available . In general, using the factory default values
From Mode when Caller ID Is Available	Used to config "From" Mode when Caller ID Is Available when call from GSM to VoIP Tel/User: From: caller number <sip:3001@IP>;tag=51088abb User/User: From: 3001 <sip:3001@IP>;tag=51088abb Tel/Tel: From: caller number <sip: caller number @IP>;tag=51088abb User/Tel: From: 3001 <sip: caller number @IP>;tag=51088abb
From Mode when Caller ID Is Unavailable	Used to config "From" Mode when Caller ID Is Unavailable Anonymous : From: <sip: Anonymous @IP>;tag=51088abb Username : From: <sip: Username @IP>;tag=51088abb
Answer Mode	Answered: Gateway answer the IP incoming call (send SIP message "200 OK" to IP part) after GSM part answered Alerted: Gateway answer the IP incoming call after GSM part Alerted
183 Mode	Immediately: Gateway send "183 RING" immediately to IP part while it receive “INVITE" from IP part. Alerted: Gateway send "183 RING" after receive "ring back" from PSTN
Response Code switch	Used to configure 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 Parameter

Figure 4-12-3 Port Parameter

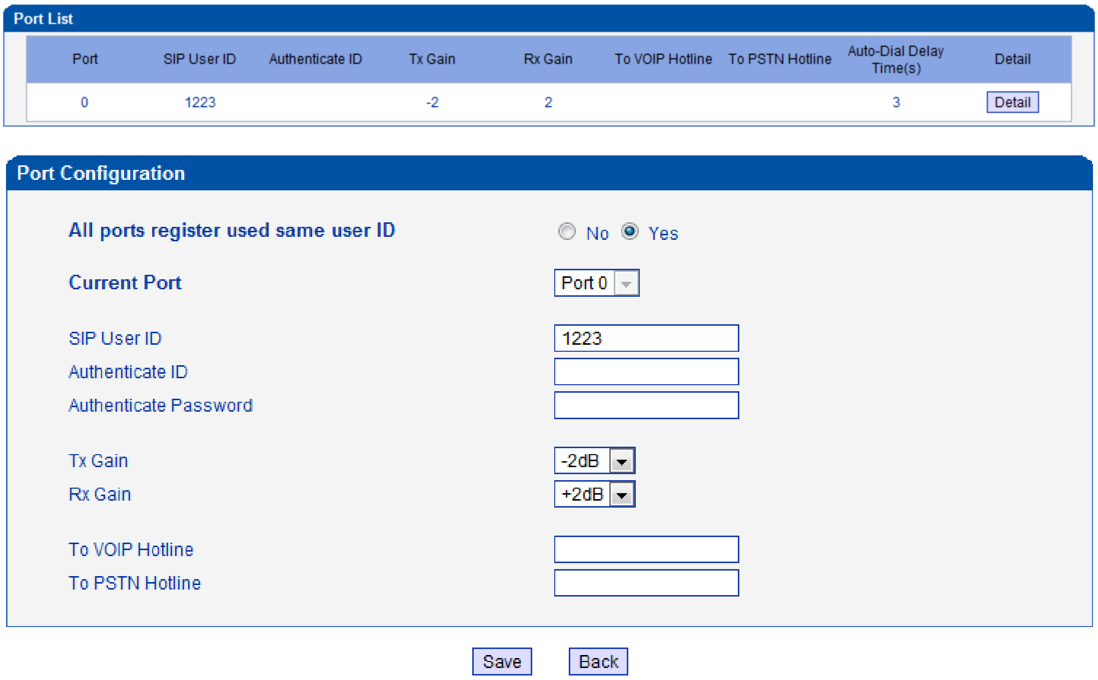
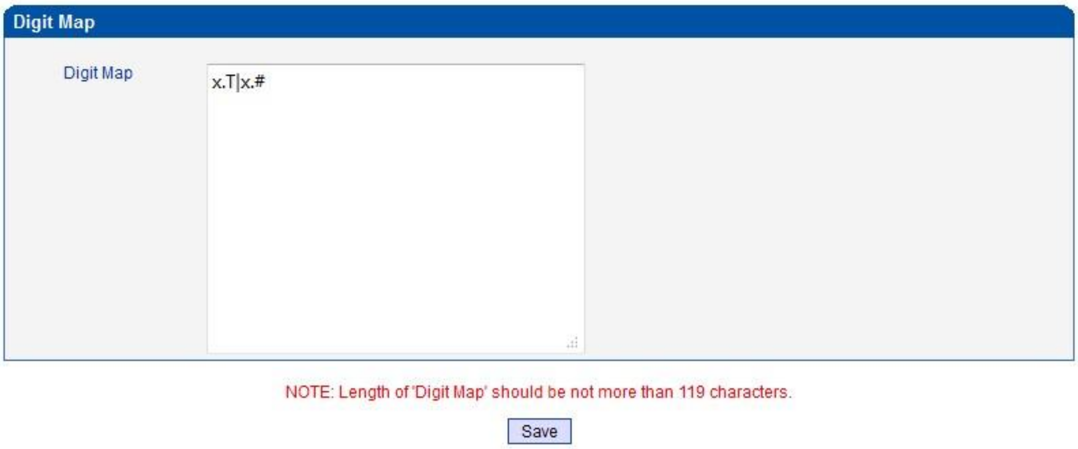


Table 4-12-3 Description of Port Configuration

Port Configuration	Used to configure ports' gain, Auto-Dial, etc.
ALL ports register used same user ID	The default is not. If set to "yes" ,all the port will use user ID
SIP User ID	It is the account used for registration, equipment port's unique identifier
Authenticate ID	Used for authenticate
Password	Its register Password
Tx Gain	Its DSP's Tx Gain. Adjusting it will effect volume on GSM side.
Rx Gain	Its DSP's Tx Gain. Adjusting it will effect volume on IP side.
To VoIP Hotline	When PSTN part client calls to this port, gateway will auto forward to the hotline User ID. Leave it blank if you don't need this function. *Note: Please config Tel->IP Operation if you need this function.
To PSTN Hotline	When VoIP part client calls to this port, Gateway will auto forward to the number to PSTN part. Leave it blank if you don't need this function. *Note: Please config IP->Tel Operation if you need this function.
Auto-Dial Delay Time	The auto-dial delay time of hotline , the range is 0-10 seconds

4.13 Digit Map

Figure 4-13-1 Digit map



Digit Map Syntax:

1. Supported objects

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

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

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

2. Range []

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

3. Range ()

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

4. Separator

   |: Separated expressions or DTMF symbols.

5. Subrange

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

6. Wildcard

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

7. Modifiers

   .: Match 0 or more times.

8. Modifiers

   +: Match 1 or more times.

9. Modifiers

   ?: Match 0 or 1 times.

Example:

Assume we have the following digit maps:

1. xxxxxxx | x11

   and a current dial string of "41". Given the input "1" the current dial string becomes "411". We have a partial match with "xxxxxxx", but a complete match with "x11", and hence we send "411" to the Call Agent.

2. [2-8] xxxxxx | 13xxxxxxxxx

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

or first is 13, followed by 9 digits.

3. (13 | 15 | 18)xxxxxxxxx

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

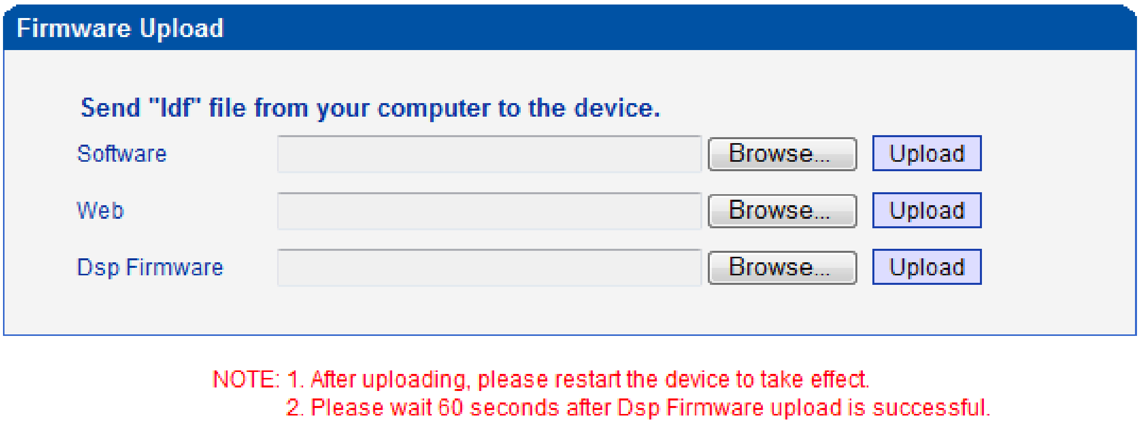
4. [1-357-9]xx

Means that first is "1","2","3" or "5" or "7","8","9", followed by 2 digits.

4.14 Tools

4.14.1 Firmware Upload

Figure 4-14-1 Firmware upload



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

4.14.2 Management Parameter

Figure 4-14-2 Management Parameter

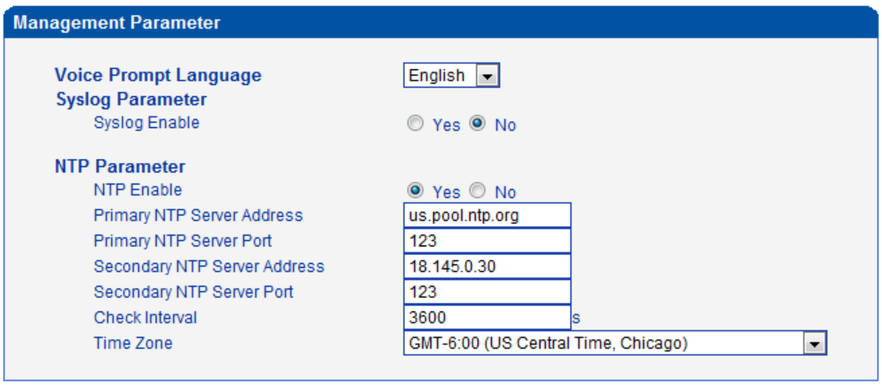
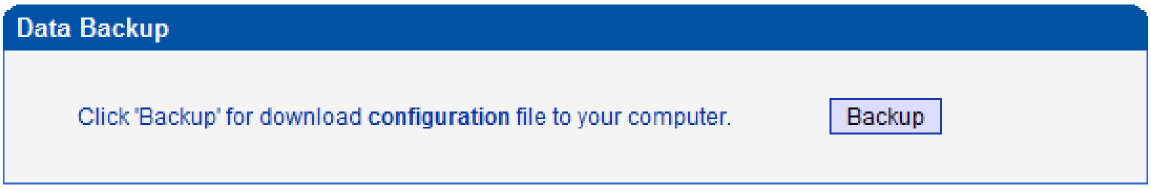


Table 4-14-1 Management Parameter

Voice Prompt Language	Select the language of voice prompt. There are two kind of voice : English and Chinese
Syslog Paremeter	Syslog is a standard for network device data logging. It allows separation of the software that generates messages from the system that stores them and the software that reports and analyzes them. It also provides devices which would otherwise be unable to communicate a means to notify administrators of problems or performance. There are 5 grades of syslog, Including NONE, DEBUG, NOTICE, WARNING and ERROR.
Send CDR	Telephone exchanges generate so called Call Detail Records (CDRs) which contain detailed information about calls originating from, terminating at or passing through the exchange. Not surprisingly CDRs are used for billing. Set to Yes gateway will sne the CDR to the syslog server.
NTP Parameter	The Network Time Protocol (NTP) is a protocol and software implementation for synchronizing the clocks of computer systems over packet-switched, variable-latency data networks. User need to fill the NTP Server Address and select Time Zone

4.14.3 Config Backup

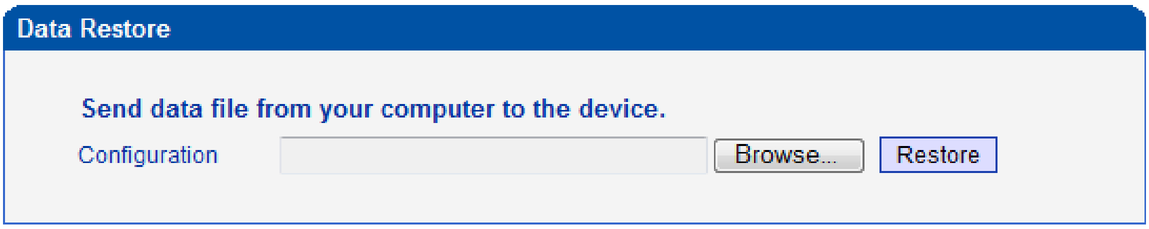
Figure 4-14-3 Config backup



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

4.14.4 Data Restore

Figure 4-14-4 Data restore

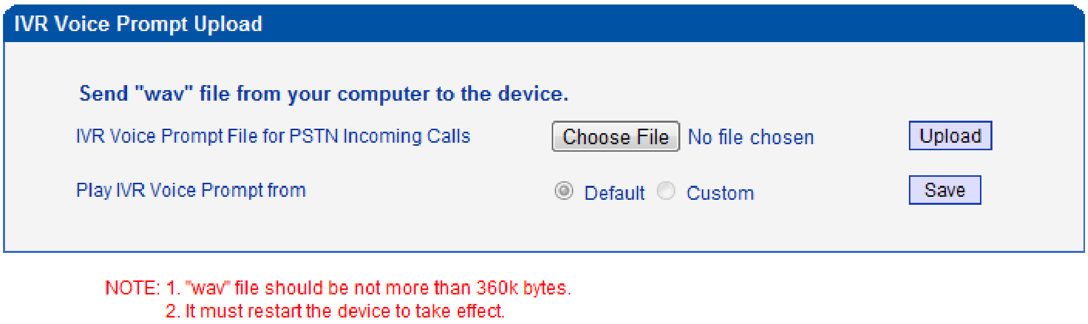


Send data file from your computer to the device

4.14.5 IVR Voice Prompt Upload

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

Figure 4-14-5 IVR Voice Prompt Upload

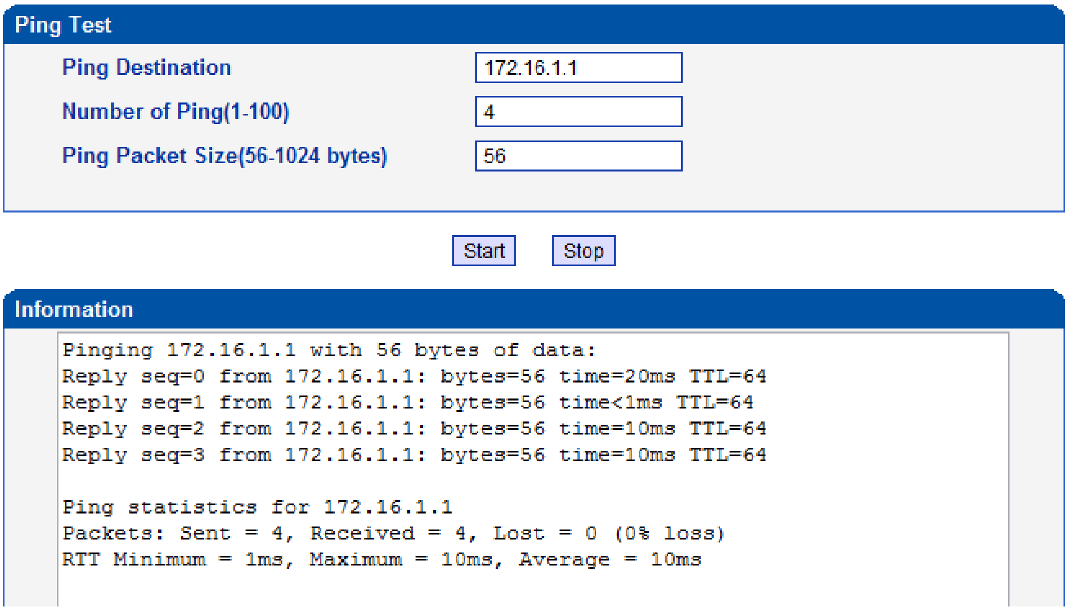


NOTE: the customize voice files can be recorded using Windows recording programs, the sound format is 8000Hz, 16 bit sampling in mono, with WAV format, size of files can not exceed 190KB

4.14.6 PING test

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

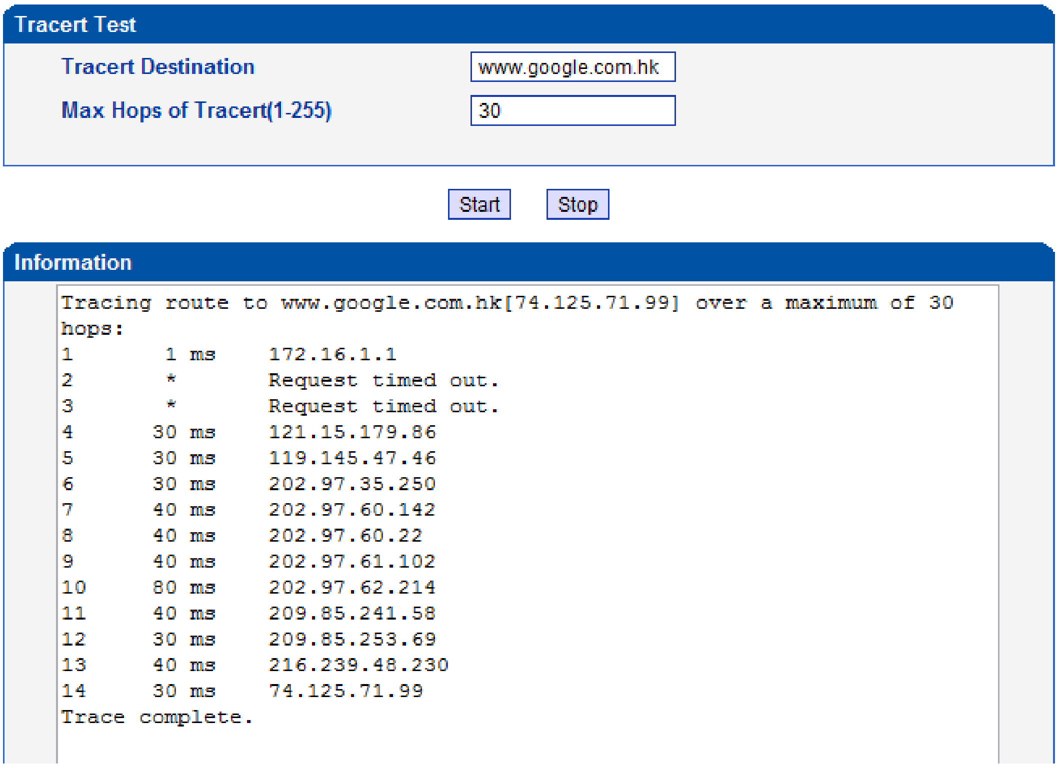
Figure 4-14-6 Ping Test



4.14.7 Tracert Test

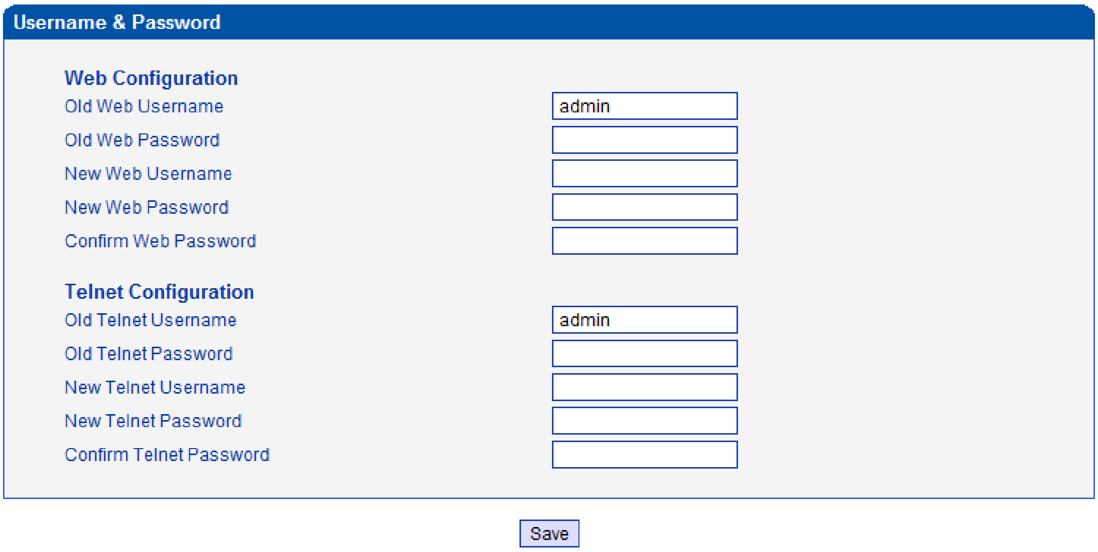
Traceroute is a computer network diagnostic tool for displaying the route (path) and measuring transit delays of packets across an Internet Protocol (IP) network.

Figure 4-14-7 Tracert Test



4.14.8 Login Password

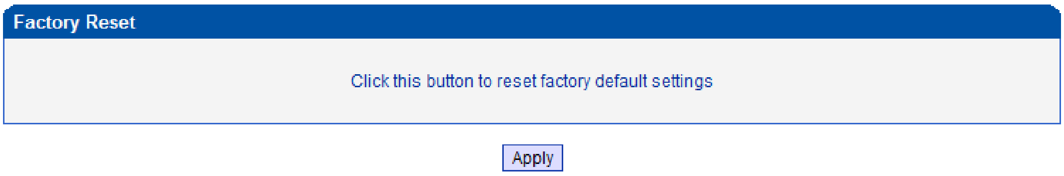
Figure 4-14-8 IVR Voice Prompt Upload



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

4.14.9 Factory Reset

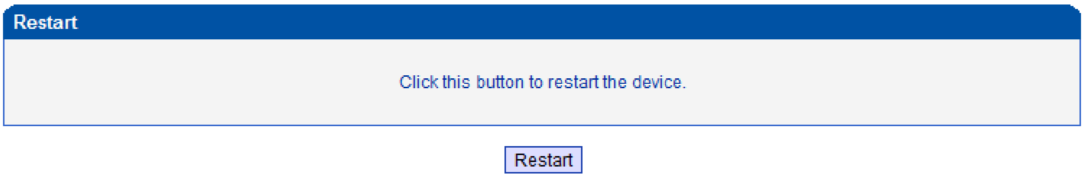
Figure 4-14-9 Factory Reset



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

4.14.10 Restart

Figure 4-14-10 Restart



5. FAQ

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

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

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

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

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

5.2 Equipment can not register

If the Run LED does not flash mean unregistered

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

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

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

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

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

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

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

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

5.4 sudden interruption during a call

1) make sure whether is human error caused the problem
2) Check the balance.
3) Make sure whether the LAN equipment such as gateway or router fails, user can try to restart the gateway or router

5.5 voice single-pass, double-barrier or poor quality
1) Make sure the equipment is working properly with grounded power
2) Check the device network connection is in working status
3) Ask network administrators to open limitation with the equipment's network communications (it is a special equipment, not afraid of virus attacks); (2) try to enable the equipment tunnel (through the WEB for Configuration, Also, please NOTE, open the tunnel will impact voice quality, Please do not enable the tunnel as far as possible, refer WEB Configuration Interface Description section)
4) Make sure the LAN equipment is working, user can try to restart the gateway or router to solve the problem
5) Check whether there is more than one DINSTAR series products in LAN network: some gateways or routers, processing network packet is vulnerable (for example, to multiple network devices or the same protocol network communication, NAT allocated the same conversion communications Channel). If there is such a case, suggest replacing a router or specify each voice gateway with different LOCAL RTP PORT Channel (refer to the base WEB Configuration interface section)
6) Check the equipment network environment for the softswitch platform, monitor the network condiation, make sure the network is solid

6. Glossary

GSM: Global System for Mobile Communications

CDMA: Code Division Multiple Access

FMC: Fixed Mobile Convergence

SIP: Session Initiation Protocol

MGCP: Media Gateway Control Protocol

DTMF: Dual Tone Multi Frequency

USSD: Unstructured Supplementary Service Data

PSTN：Public Switched Telephone Network

STUN: Simple Traversal of UDP over NAT

IVR: Interactive Voice Response

IMSI: International Mobile Subscriber Identification Number

IMEI: International Mobile Equipment Identity

DMZ: Demilitarized Zone