DVG-6001G User Manual v1.0

Contents

1. Equipment Introduction

This chapter mainly introduces functions and structures of DVG-6001G.

1.1 Overview

DVG-6001G is a full function GSM wireless gateway based on IP. It is able to offer stable network configuration, powerful function characteristics, excellent voice quality and provide affordable VoIP solutions for operators, enterprise, SOHO, home users.

To ensure that existing in structured cabling and ensure the safety of the existing network normal operation, it supports IEEE 802.3af POE.

1.2 Application solutions

DVG-6001G provides wireless access service for custom. The following is a typical scheme network diagram.

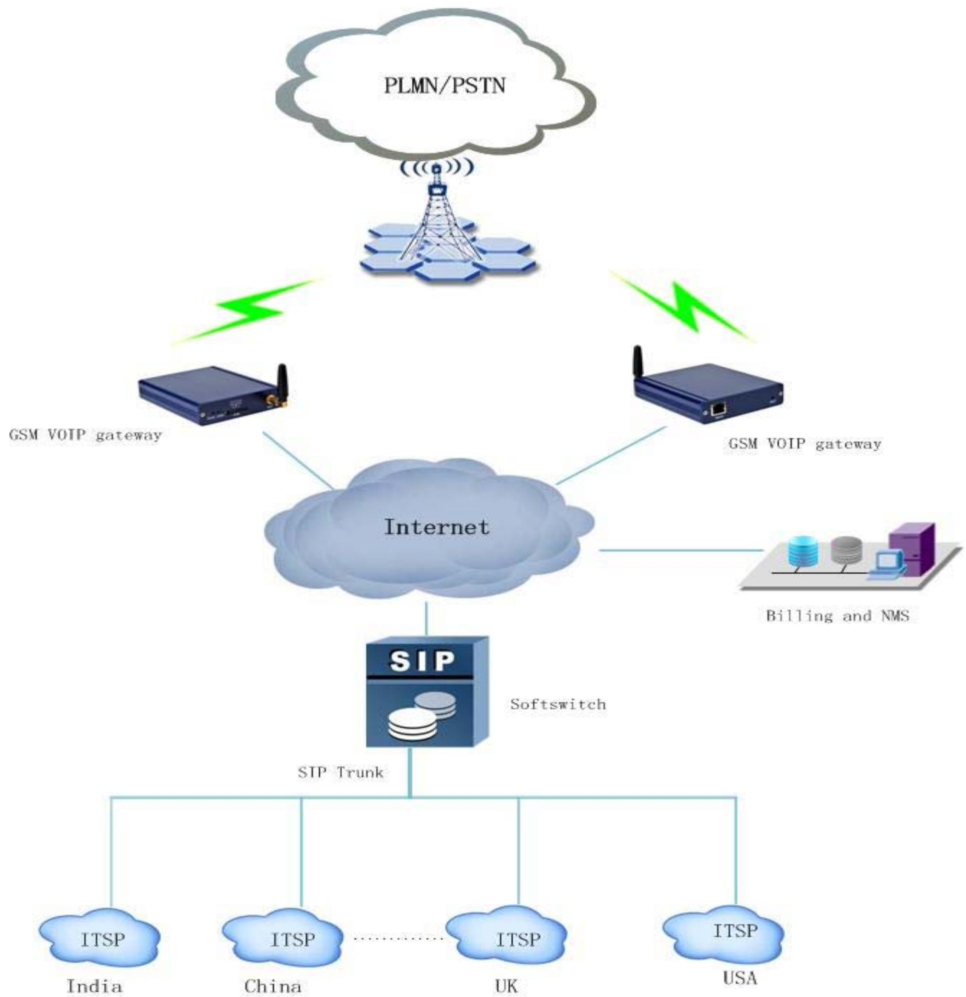


Figure 1-2-1 DVG-6001G application solution

1.3 Product appearance

Figure 1-3-1 DVG-6001G View before and after

Table 1-3-1 DVG-6001G interface introduction

Interface	Discription
WAN	When connecting normal the WAN indicator will flash, while not register, please check whether the network connection is normal firstly. WAN used as a LAN under bridge mode. WAN port in transmits data at the same time the power is transmitted also, for this device provide DC power supply.
RST	Long click this button factory default recovery
SIM Slot	SIM card slot, please press the yellow button at the edge of slot,
ANT	External rubber lining the antenna, gain is 3dBi

Table 1-3-2 DVG-6001G indicator light

LED	Color	State	Description
RUN	Green	Slowly flash	SIP account does not have registered
		Fastly flash	SIP account have registered

GSM	Green	Destroy	SIP account does not have
		Fastly flash	SIP account have registered

1.4 Function characteristics

1.4.1 Protocol

- Standard SIP
- NAT
- PPPoE、HTTP、DHCP、DNS
- ITU-T G.711a-Law/μ-Law、G.723.1、G.729AB;

1.4.2 System Function

- PLC、VAD、CNG
- POE
- Support local and remote SIM card register
- Adjustment port gain
- Many DTMF mode
- Single talk time limit
- Total call time limit
- Talk time remaining reset
- Balance alarm
- SIM/UIM card encryption
- SMS sending and receiving
- Customized IVR
- Black/White list
- SMS、USSD Open API interface
- Echo Cancellation (with ITU-T G.168/165 standard)
- Automatic negotiate network
- Hotline
- Automatic restart module
- Lgeneration dial

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： 44-57V, 550mA MAX
- Temperature: 0~40℃（Operation）, -20~80℃（storage）
- Humidity: 5%～90%RH
- Power Consumption: 6.5-13W
- Dimensions: 120(W) x90(D) x24(H) mm
- Net weight: 0.8kg

2. Equipment Installation

This chapter mainly introduces DVG-6001G hardware installation and connection of equipment.

2.1 Installation Notice

1. DVG-6001G used POE for power to make sure equipment stable power supply
2. DVG-6001G interfaces support RJ45 的 10/100Mbps
3. Directly into the SIM card, GSM channel can be started work

2.2 Installation Procedure

2.2.1 Install SIM Card

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

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

2.2.2 Antenna Installation

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

2.2.3 Cable Connection of Equipment

DVG-6001G works in bridge mode:

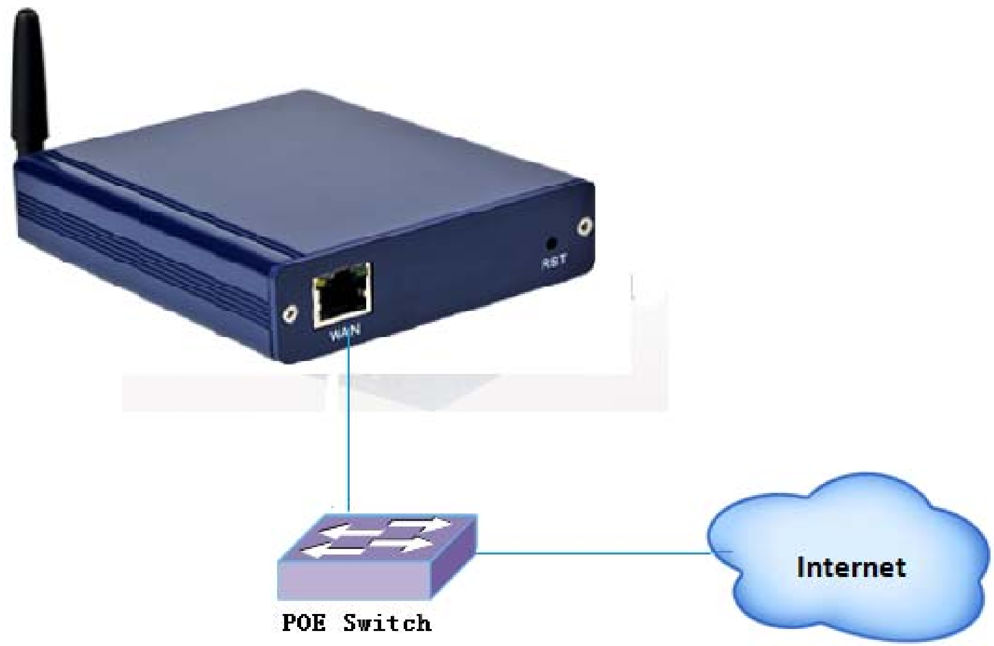


Figure 2-2-1 DVG-6001G connection

3. Network Configuration

In this chapter we will introduce the initial configuration of DVG-6001G gateway. All of the network parameters of the gateway can be configured by IVR guidance.

3.1 Preparation

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

1. Prepare an analog telephone or mobile phone
2. Make sure the gateway is connected with the network
3. Completed the SIM installation
4. Make sure that the current mobile network is working

3.2 Attentions

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

DVG-6001G can work in bridge mode user should configure network parameters of WAN port.

3.3 General Feature Codes for System Setting

Table 3-3-1 Feature codes for system setting

Dial numbers	Features
*114#	Play the phone NO.
*150*a#	Set IP address(static/DHCP), a can be digit 1 or 2,*150*1# is static IP address mode, *150*2# is DHCP mode
*152*a*b*c*d#	Set IP address(static/DHCP), a can be digit 1 or 2,*150*1# is
*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 the IP address
*111#	Restart

3.4 Static IP Configuration

Assuming the IP address of a DVG-6001G device is:

IP：172.16.0.100 subnet mask： 255.255.0.0 gateway: 172.16.1.1

configured as follows:

1. Insert a SIM card into the DVG-6001G gateway
2. The configuration mode: Dial the phone number of this SIM card. hear a message, then enter "*150*1#", hang up when hear " setting successful" message;

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

3.5 3.5 DHCP Configuration

DHCP mode configure as follows:

1. Insert a SIM card into a slot, dial the SIM card number. When hearing a hint message, then enter "*150*2#", if hearing " setting successful" message, which means the DHCP is confirued successfully;
2. 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 .

4. WEB Configuration

This charpter describes web configuration of DVG-6001G.

4.1 Preparing

DVG-6001G connected to the local area network of PC used logged in with LAN.

4.2 WEB Landing Interface

The default IP of DVG-6001G is 192.168.11.1/ 255.255.255.0. Also can modify equipment IP by IVR. Before landing please ensure that with equipment connected PC's IP address and the IP address of the equipment in the same subnet. Enter IP address of DVG-6001G in browser.

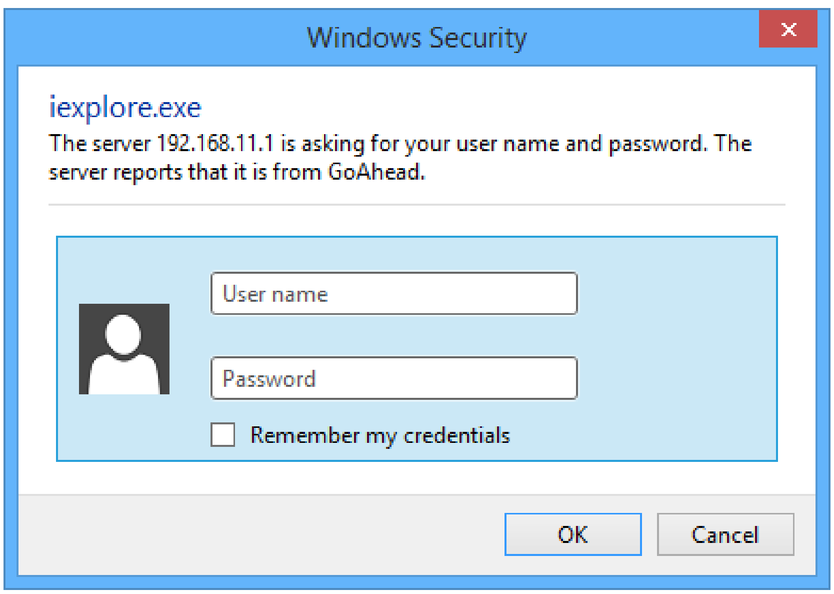


Figure 4-2-1 web landing interface

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

4.3 WEB Configuration

DVG-6001G web configuration interface consists of the navigation tree and the detail configuration interfaces.

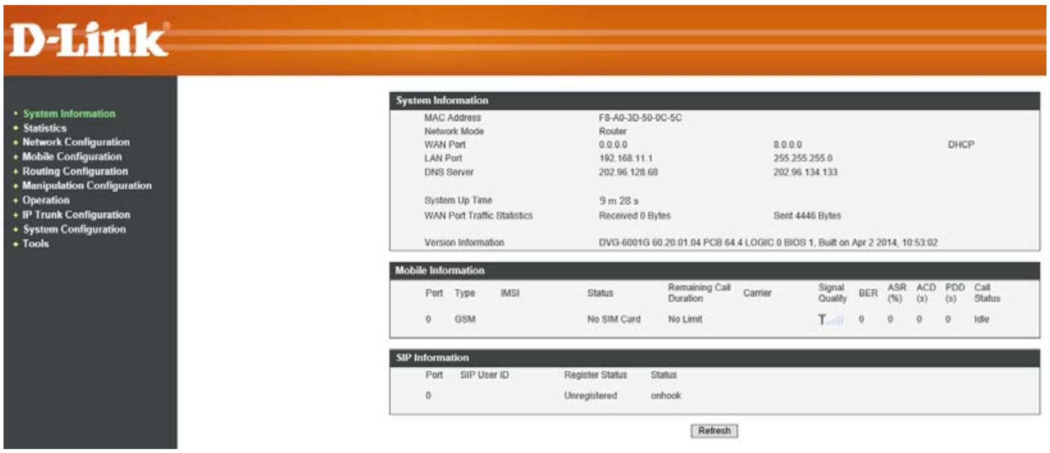


Figure 4-3-1 Web introduce

4.4 System

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

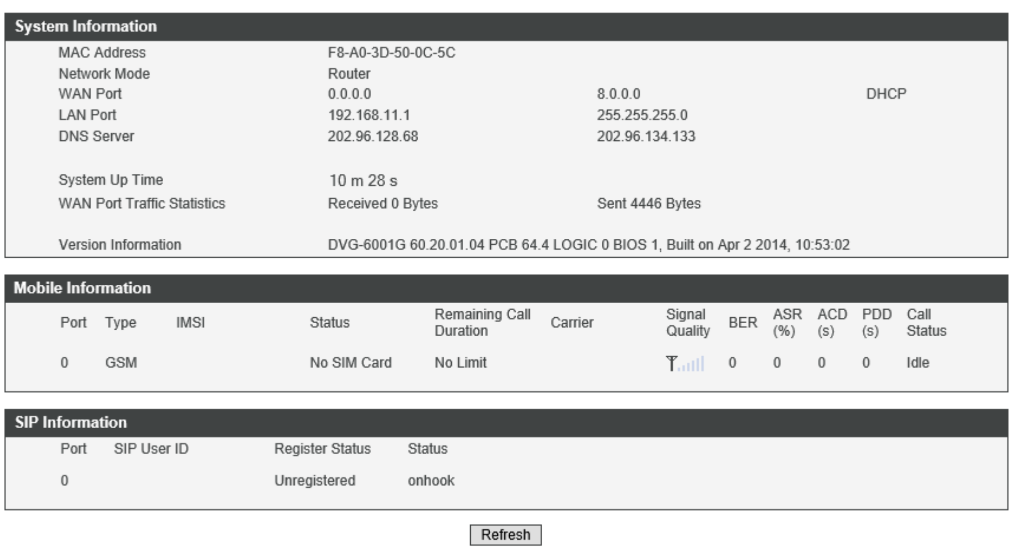


Figure 4-4-1 System Information

4.4.1 System Information

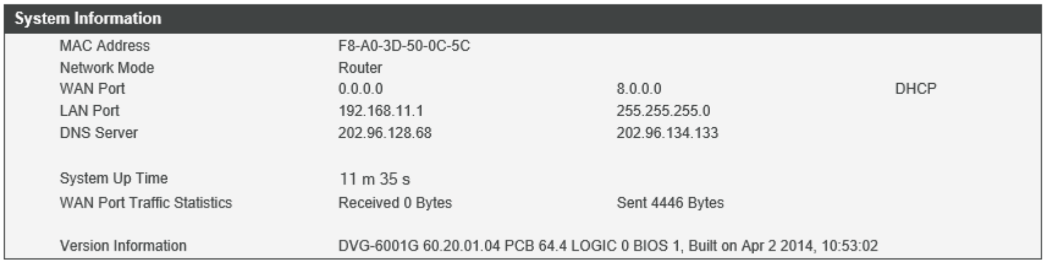


Figure 4-4-2 System Information

Table 4-4-1 System Information Describe

MAC Address	Display the current MAC of the gateway, for example: 00-01-0C-03-A4-2F
Network Mode	DVG-6001G only support “bridge” mode, please reference network configure
Network	Display IP, subnet mask and the way of obtain IP address
DNS Server	Display DNS server IP address
System Up Time	Display the time period of the device running
Network Traffic Statistics	Calculates the netflow, including the total bytes of message received and sent
Version information	Display the version information, include: product model, firm ware version, hardware version and date compiled.

4.4.2 Mobile Information

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

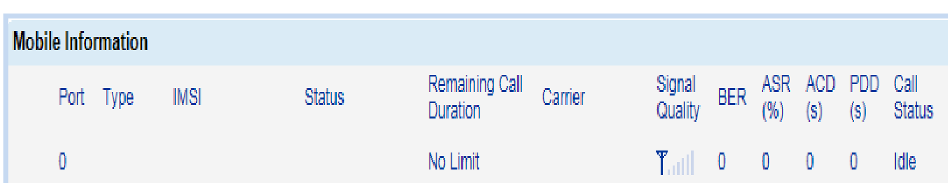


Figure 4-4-3 Mobile Information

Table 4-4-2 Mobile Information

Port	Numbers of ports of GSM
Type	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 module
Remaining Call Duration(min)	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	Display the network carrier of current SIM card.
Signal Quality	Displays the signal strength of in each channels of GSM
BER	Bit erro rate
ASR（%）	Average connection rate
ACD	Average call duration
PDD	Delay between call sent out and call connected
Call Status	Show the status of call, include idle, handle, hangup, call such status

4.4.3 SIP Information

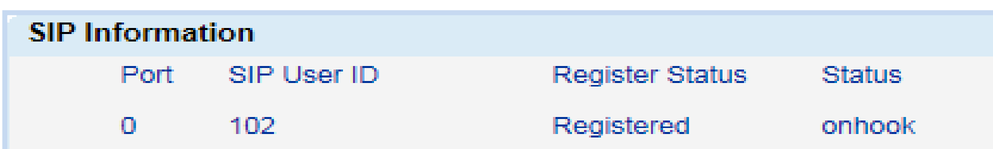


Figure 4-4-4 SIP Information

Table 4-4-3 SIP Information

Port	The corresponding GSM port, DVG-6001G has only 1 port
SIP User ID	SIP registration account of the Softswitch and SIP server provided
Register Status	Show the registration status of VoIP channel, including registered and unregistered.
Status	The status of Off-hook and hang up

4.5 Network Configuration

4.5.1 Local network

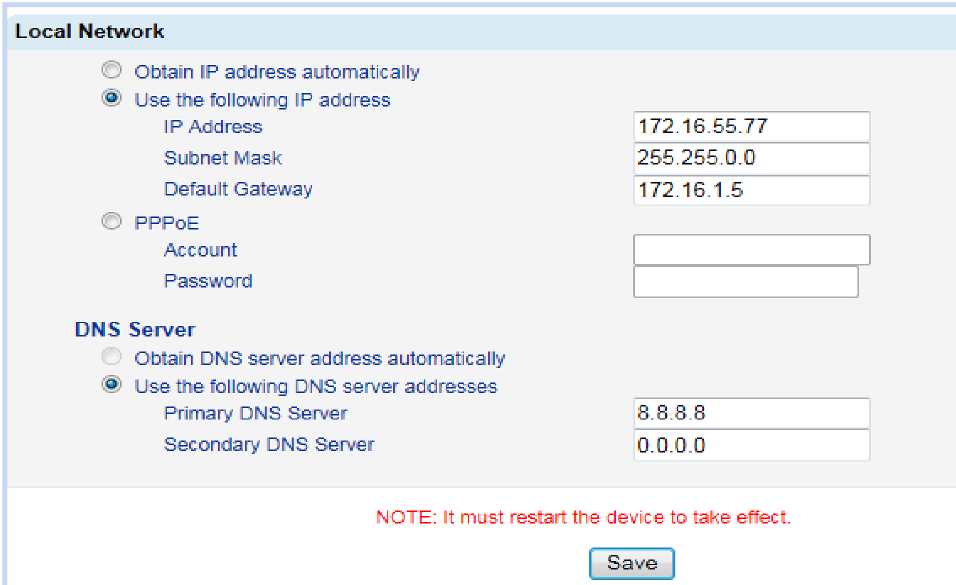


Figure 4-5-1 Local network

Table 4-5-1 Local network

Work mode	Only Bridge mode
Obtain IP Address Automatically	After used, the IP address obtained from DHCP server
Use the following IP address	After used, need to manually add IP address, subnet mask and default gateway
PPPoE	When adopt PPPoE dial-up Internet, need to fill in account and password offered by ISP
Obtain DNS Server Address Automatically	DNS server complete the analytical between domain name and IP address. When enable “Obtain DNS Server Address Automatically”, which will be automatically get DNS server address.
Use the Following DNS Server	Fill in the IP address of "Primary DNS Server" and

Addresses	"Secondary DNS Server"

4.6 Mobile Configuration

4.6.1 Basic Configuration

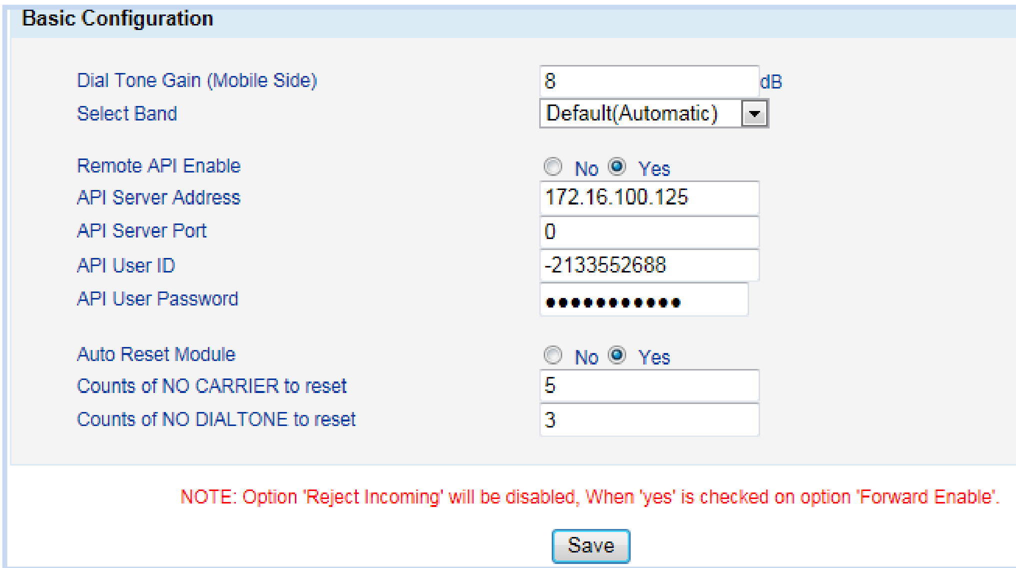


Figure 4-6-1Basic Configuration

Table 4-6-1 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 below: PGSM900, DCS1800, PCS1900, EGSM900/DCS1800, GSM850/PCS1900
Remote API Enable	API is provided interface for third party development with DLL and IAD components. Includes SMS/USSD sending and receiving. If want to use the client to send text message, please open API.
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". The user can defined a not overlap with the other application port of the port number, the proposal value is 12000
API User ID	Remote API user account. This is an option when selecting "Yes" under "remote API enable".
API User password	Remote API user password. This is an option when selecting "Yes" under "remote API enable".
Auto Reset Module	Open the function, in the following case module can be reboot
Counts of No CARRIER to reset	Continuously n times can't find operators, equipment to restart. N is 3-255.
Counts of No DIALTONE to reset	Continuously n times no dialtone, module to restart. N is 3-255.

4.6.2 Mobile Configuration

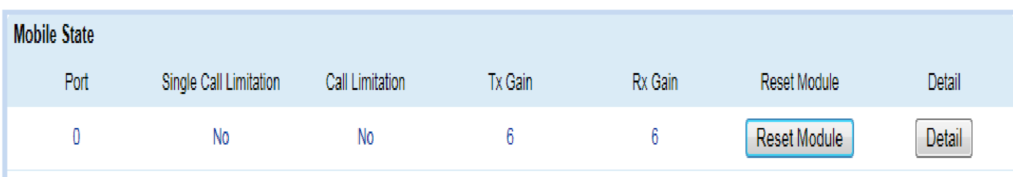


Figure 4-6-2 Mobile Configuration

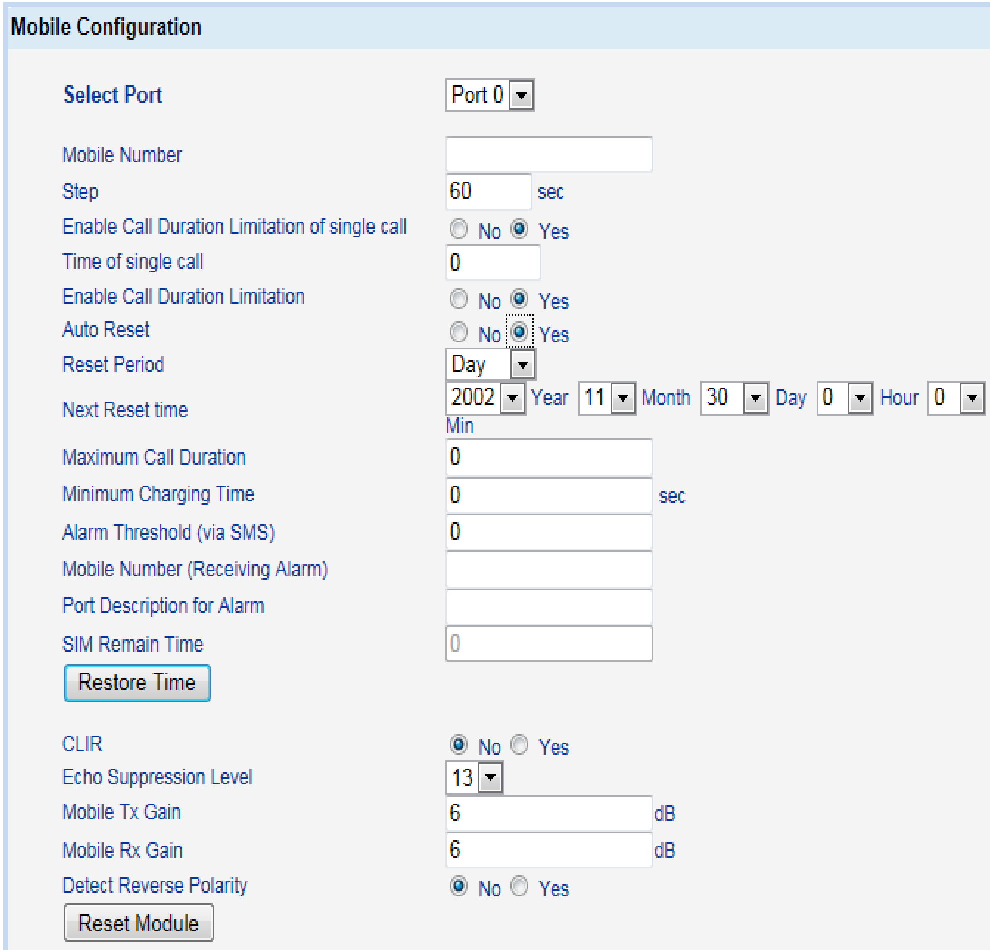


Figure 4-6-3 Mobile Configuration

Table 4-6-2 Mobile Configutation

Mobile Number	Corresponding port SIM card number
Enable Call Duration Limitation of single call	This function is to limit the max call duration of channel. Users can customize the SIM card on the single call duration; if more than the duration, call will be take out stitches. If select "Yes", then need to set the following two options.
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 max call duration of channel. The max call duration is between 1 to 65535 minutes.
Auto Reset	Automatic reset talk time remaining, Let remaining call time is equal to the maximum call duration.
Reset Period	User defined daily, weekly or monthly reset SIM card information, that is, remaining call time is equal to the maximum call duration and start counting.
Next Reset time	The user defined when to begin to reset, then from the date according to reset period reset.
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)	Talk time remaining is equal to or lower than the value, the gateway to the alarm information by SMS messages to the designated mobile phone number.
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	This function is used to GSM side exhale hidden SIM card number. Adding a “#31#” infront of mobile phone number can realize the function. This function need operators support.
Echo Suppression Level	Control echo of call process. The higher the level, the more powerful the echo suppression.

Mobile Tx Gain	Control IP to GSM side of call the gain. Default is 6dB.
Mobile Rx Gain	Control GSM to IP side of call the gain. Default is 6dB.User can adjust the two gain to adjust the size of the voice.
Detect Reverse Polarity	To GSM module is invalid, in the role of CDMA module, the local CDMA network support open when the extremely. When not open this function, use a overtime time to report a fake the extremely, overtime time for response time delay, see business configuration parts.

4.6.3 PIN Management

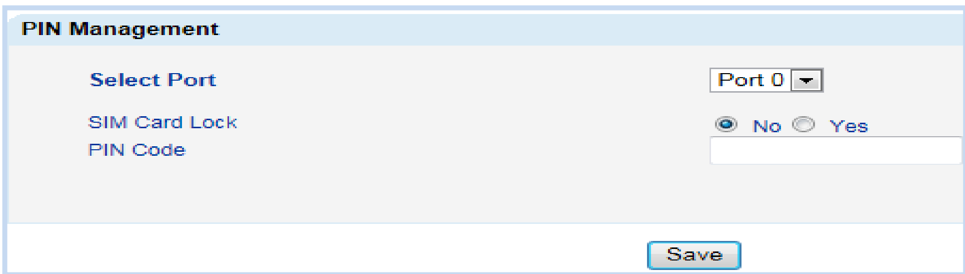


Figure 4-6-3 PIN Management

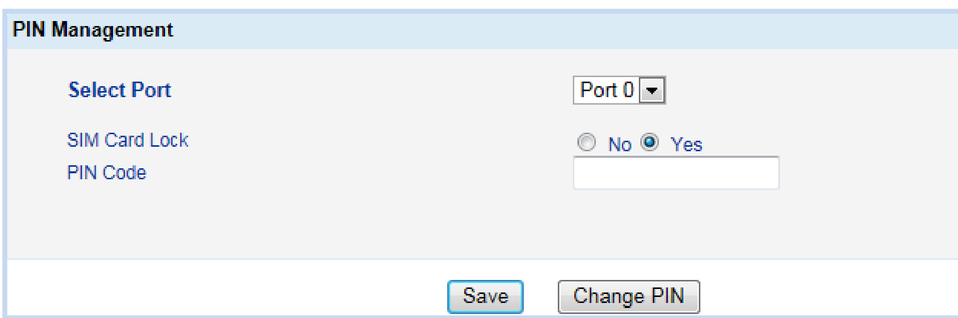


Figure 4-6-4 PIN Management

Table 4-6-3 PIN Code Management

Select Port	Selection need locked channel number
SIM Card Lock	To prevent the SIM card is the use of others, user can lock SIM card.
PIN Code	Locked or unlocked SIM card need to input PIN code
Change PIN	Click this button to modify PIN

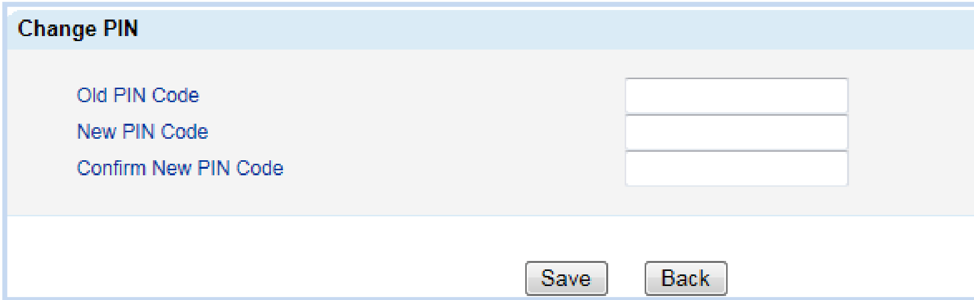


Figure 4-6-5 Change PIN Code

PIN is Personal Identification Number of SIM card. Here the PIN code changed.

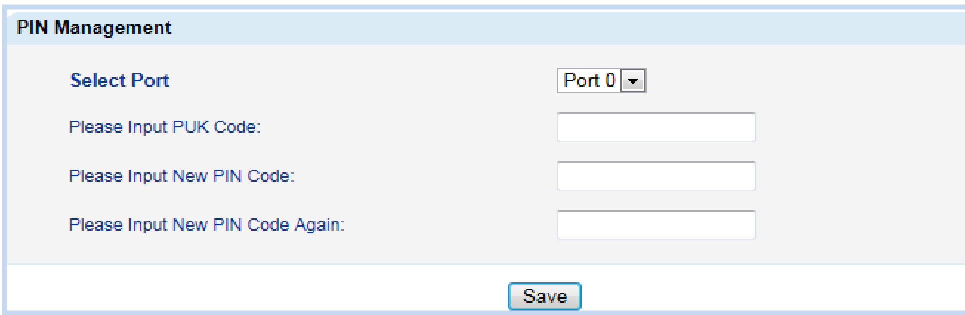


Figure 4-6-6 PIN code to unlock

When the PIN code three consecutively input error, system will tip input PUK yards, and reinstall new PIN code.

Table 4-6-5 PIN code to unlock

Select Port	Select GSM port needed input PUK code
PUK Code	PIN Unlocking Key is the PIN code unlock code. PIN code three consecutive input error, SIM card will be locked, need to unlock the PUK yards. PUK yards of input opportunity is 10 times, 10 times all lose correctly, SIM card will be locked to the permanent, that is discarded.
Please input new PIN code	Set a new PIN code
Please input new PIN code again	Again confirmed the new PIN code.

4.6.4 SMSC

Figure 4-6-7 SMSC

Mobile phone text message center, in theory the wireless module can automatically detect the SMS center number. But when wireless module can't automatically detect the SMSC number, please contact mobile network operators, and manual Settings SMSC number.

4.6.5 Send Message

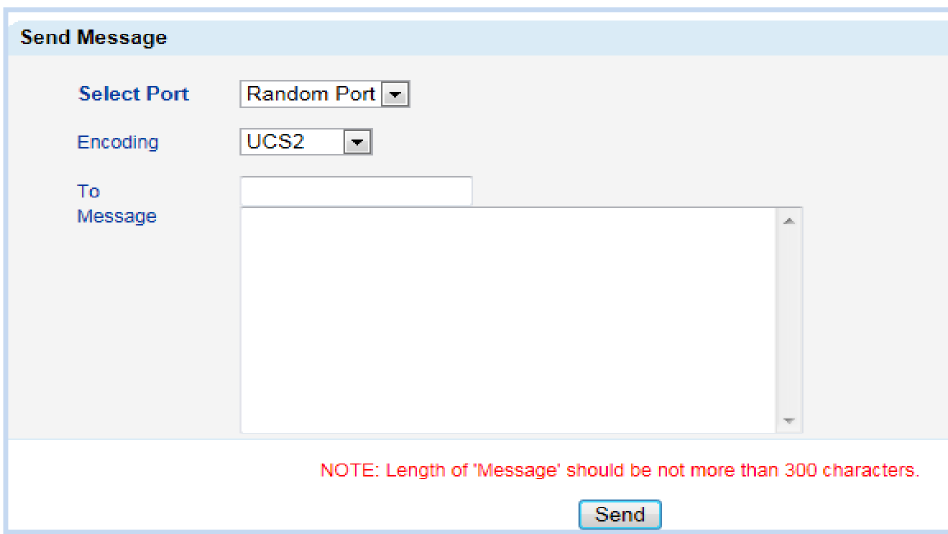


Figure 4-6-8 Send Message

Table 4-6-6 Send Message

Select Port	From the designated port can send, also can choose random ports to send
Encoding	SMS code can be used in two ways, UCS2 and GSM 7bit. Editor pure English short message can use GSM 7 bit, otherwise, use UCS2.
To	Mobile phone number received SMS

Message	The content of the messages, the length is not more than 300 characters.

4.6.6 USSD

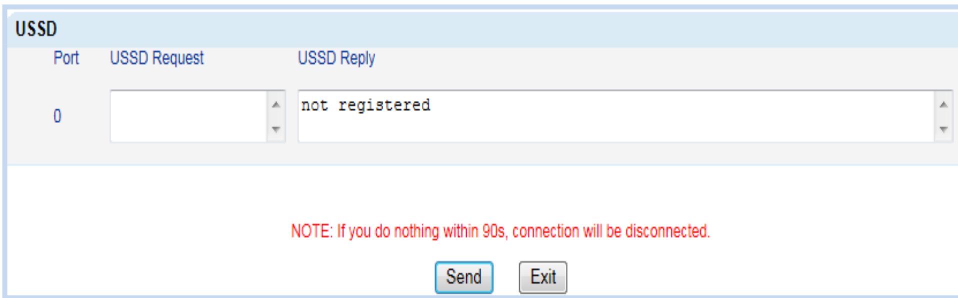


Figure 4-6-9 USSD

USSD（Unstructured Supplementary Service Data）is a new type of based on GSM network interactive data business. When using a mobile phone keyboard input some prescribed number or symbols such as * #, etc, then press the dial-up key , mobile phone will send an instruction to network. According to instructions, network choice special services to you.

USSD technology used alone or in combination with the current short message technology, General Packet Radio Service GPRS (General Packet Radio Service) technology combined to provide various value-added services, such as Mobile Banking, Financial stock trading, Mobile phone calls inquires, Meteorological information prediction and query, Send and Receive Email, Flight Track, Booking Tickets Online etc.

4.6.7 Carrier

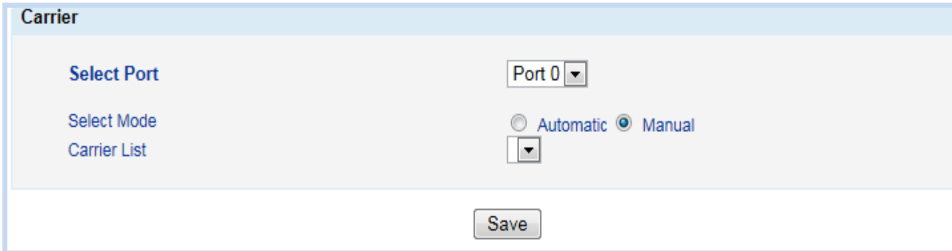


Figure 4-6-10 Carrier

Table 4-6-7 Carrier

Select Port	Select a SIM card
Select Mode	There are automatic and manial two mode. Automatic mode can detect carrier automaticly; manual mode will select carrier from drap-dowm list.
Carrier List	Here will list all detected operators

4.6.8 BCCH

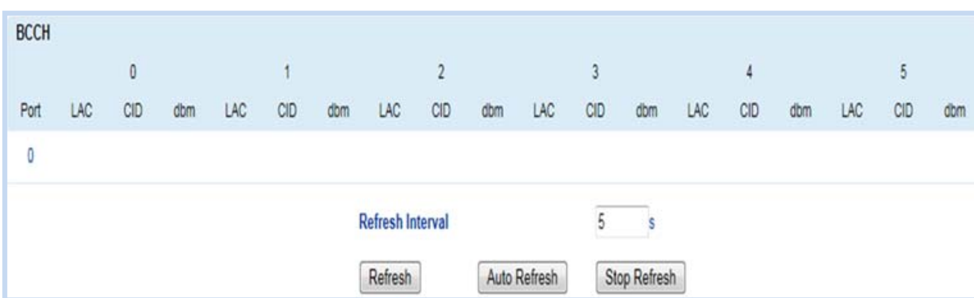


Figure 4-6-11 BCCH

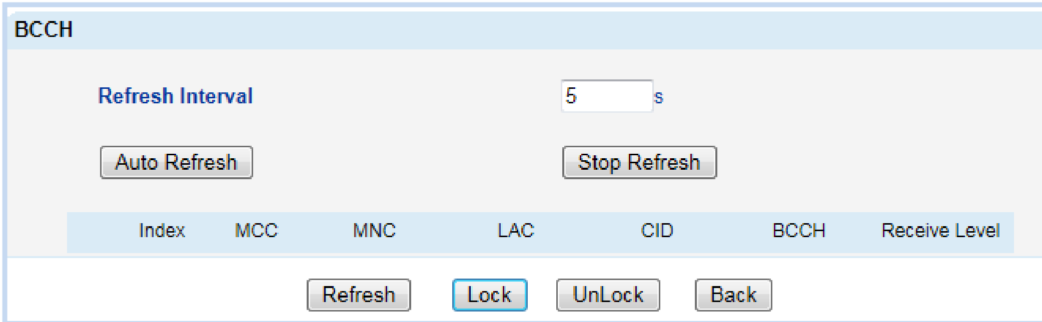


Figure 4-6-12 BCCH

Table 4-6-8 BCCH Description

Refresh Interval	Set BCCH parameters automatically refresh time
Index	Base station parameters numbers
MCC	Mobile Country Code, China is 460
MNC	Mobile Network Code, used to distinguish between different network operators.
LAC	Location area number, in order to determine the position of the mobile station, each GSM PLMN coverage area is divided into many location area, location area codes (LAC) is used to identify the different location area.
CID	To the only to express the GSM PLMN every community, network operators should be assigned to the network of all the village a code, that is, CI, CI and LAI yards combined, used to identify each of the network and its coverage of the village BTS.
BCCH	Broadcast Control Channel, general information transmission, used for mobile measurement signal strength and identify district mark etc.
Receive Level	The base level received signal from BTS
Lock	Signal can be locked in a few signal good base station, selected to lock base station, click on the lock. If base station signal is very

	poor that has locked, signal will also automatically switch to other stations.
Unlock	Unlock the base station that has locked.

4.7 Routing Configuration

4.7.1 Routing Parameter

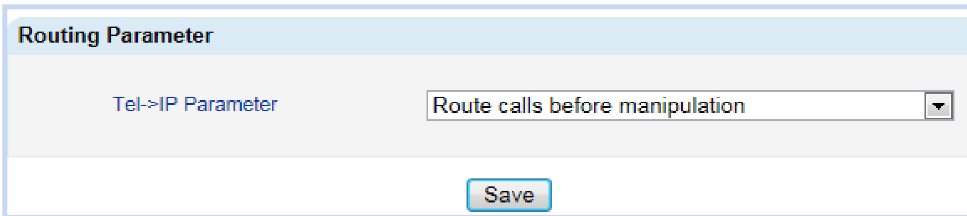


Figure 4-7-1 Rout Parameter

Table 4-7-1 Rout Parameter

Tel->IP Parameter	Routing parameter from GSM to IP
Route calls before manipulation	First routing, after number transformation
Route calls after manipulation	First number transformation, after routing

4.7.2 Tel->IP Routing

Figure 4-7-2 Tel ->IP Routing

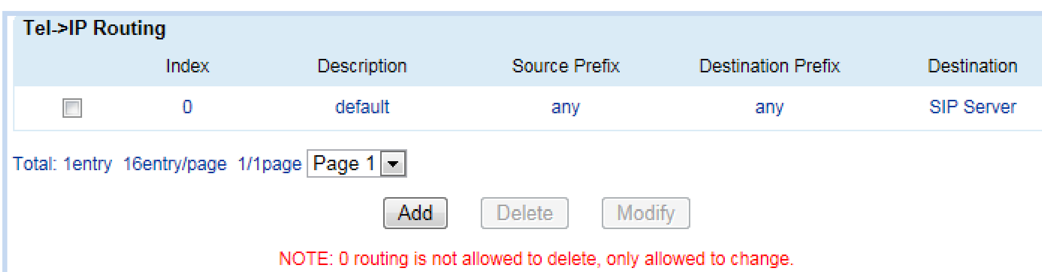


Table 4-7-2 Tel ->IP Routing

Tel ->IP Routing	This item uses to configure incoming call routes which can be used for recieve the calls from the GSM.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from

	0 to 31.
Description	It describes the route for the ease of identification. Its value is character string
Source Prefix	When calling number matching the prefix, this routing will take effect. Any: indicates any number 0xxxx: All of the number of begin to 0
Destination Prefix	When callee number matching the prefix, this routing will take effect.
Destination	Specify the specific IP, IP group and SIP Server

Click “Add” or “Modify” enter the following interface.

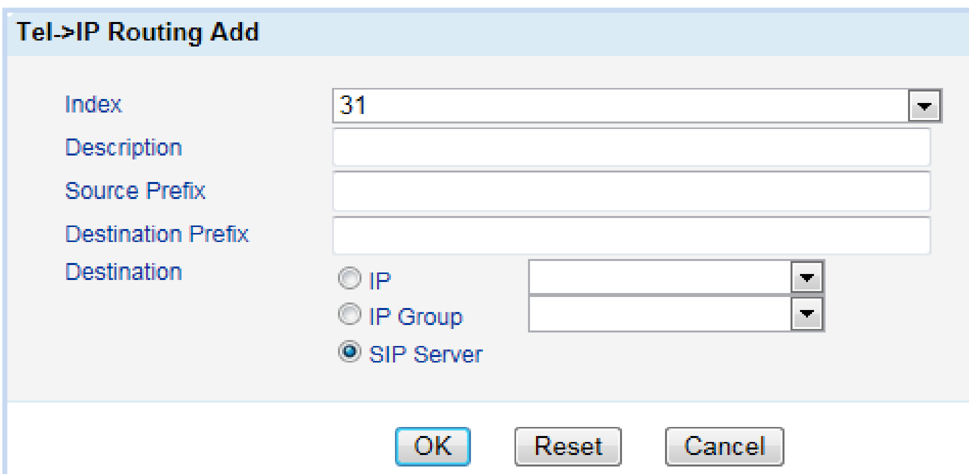


Figure 4-7-3 Tel->IP Routing Modify

4.8 Manipulation Configuration

4.8.1 IP->Tel Destination Numbers

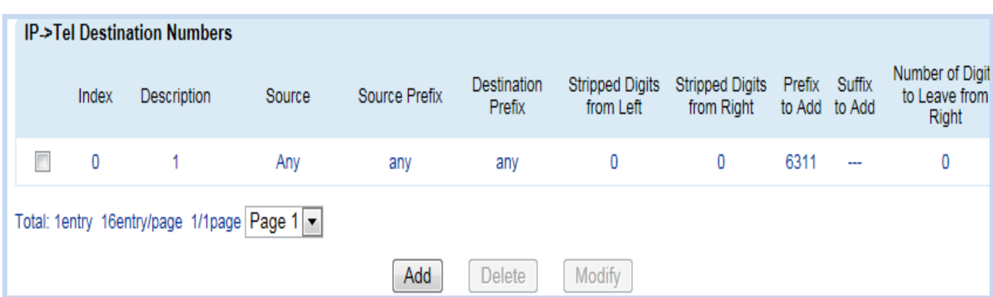


Figure 4-8-1 IP->Tel Destination Numbers

Table 4-8-1 IP->Tel Destination Number

Manipulation	This option can modify the Lord called number pass by gateway
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.
Description	It describes the route for the ease of identification. Its value is character string
Source	It specifies the source IP which will send the calls to gateway
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
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 digit to leave from right	From the right side began to retain the digits

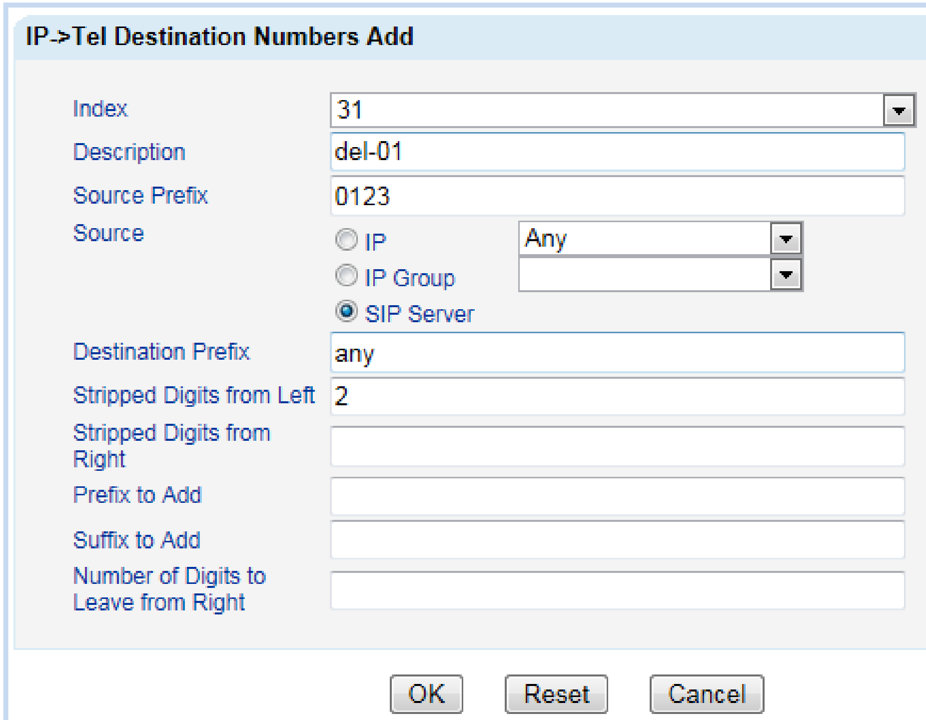


Figure 4-8-2 IP->Tel Destination Number Add

For example, source prefix is 0123, after transform, the prefix become 23.

4.8.2 Tel->IP Source Numbers

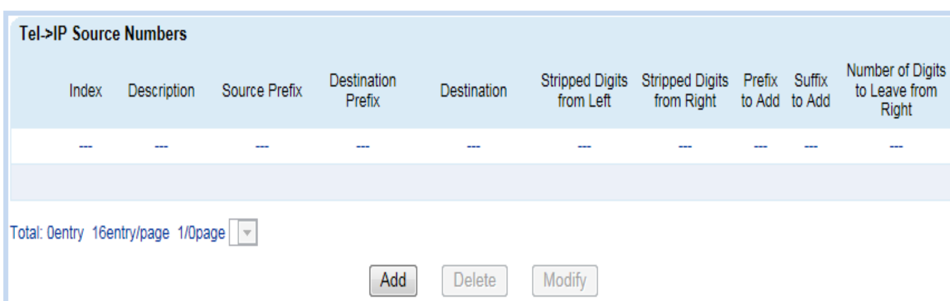


Figure 4-8-3 Tel->IP Source Numbers

Table 4-8-3 Tel->IP Source Numbers

Index	It is the number of the transformation and said number transformation rule label the priority. Value range is 0-31.
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	Appoint number destination: IP, IP group or SIP server
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

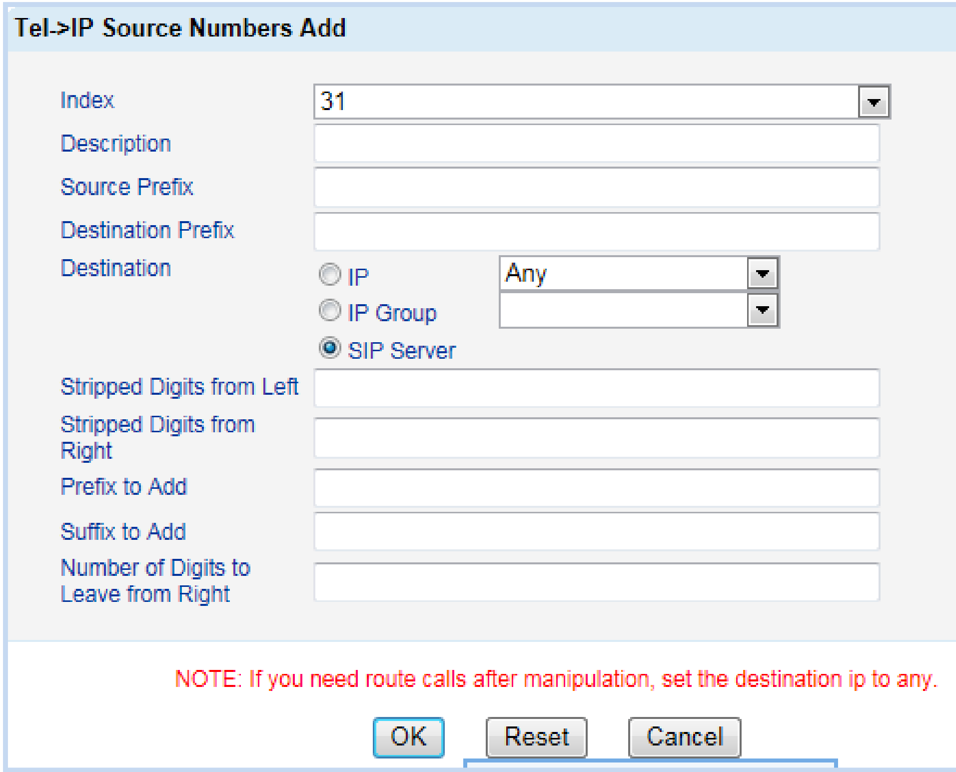


Figure 4-8-4   Tel->IP Source Numbers Add

4.8.3 Tel->IP Destination Numbers

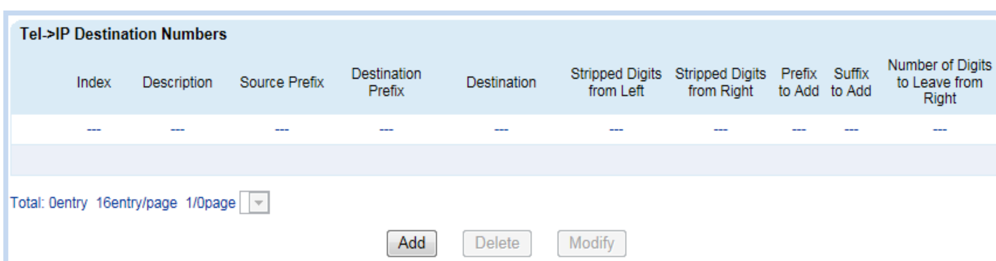


Figure 4-8-5 Tel->IP Destination Numbers

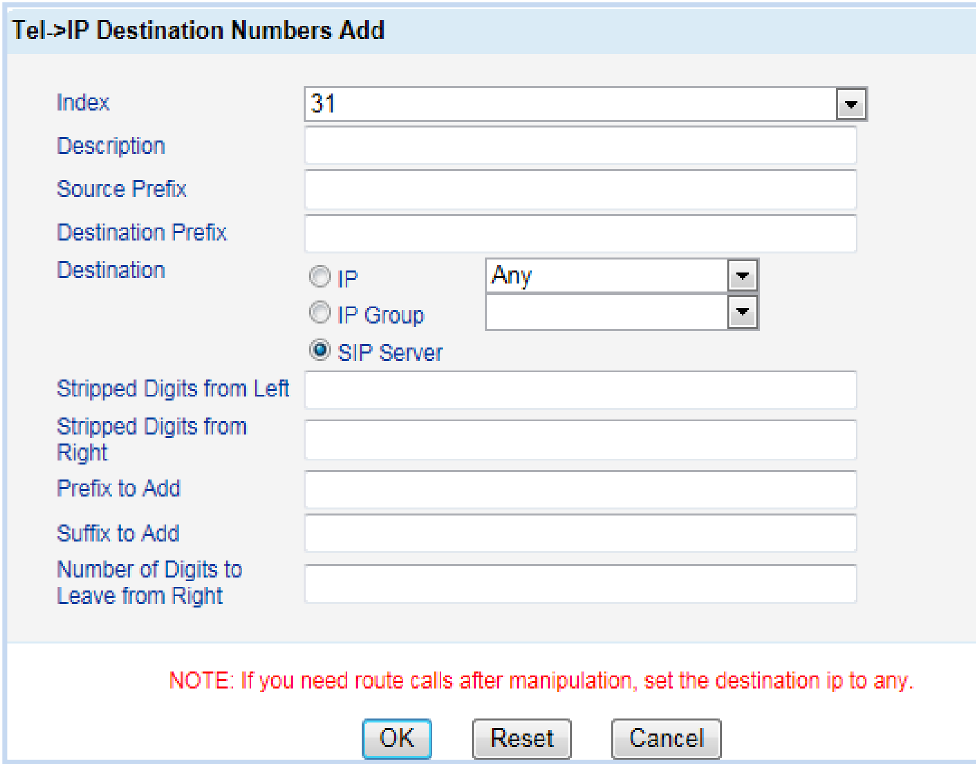


Figure 4-8-6 Tel->IP Destination Numbers Add

Please reference Tel->IP Source Numbers. Matching rules completely the same.

4.9 Option

4.9.1 IP->Tel Option

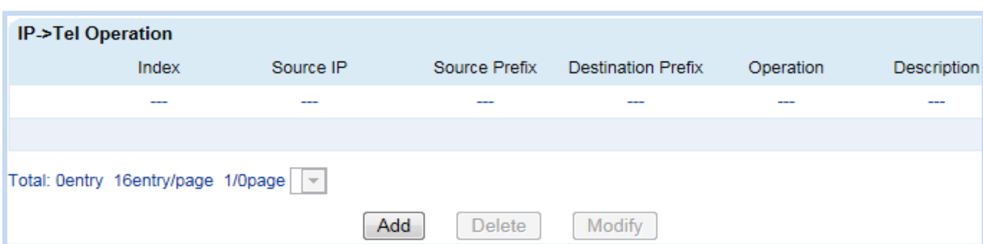


Figure 4-9-1 IP->Tel Option

Table 4-9-1 IP->Tel Option

IP->Tel Operation	This is an optional configuration items, when using the hotline, this item must be configured. Include: forbid call, call allowance, auto call, and password authentication.

Index	Number, value range from 0-31.
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

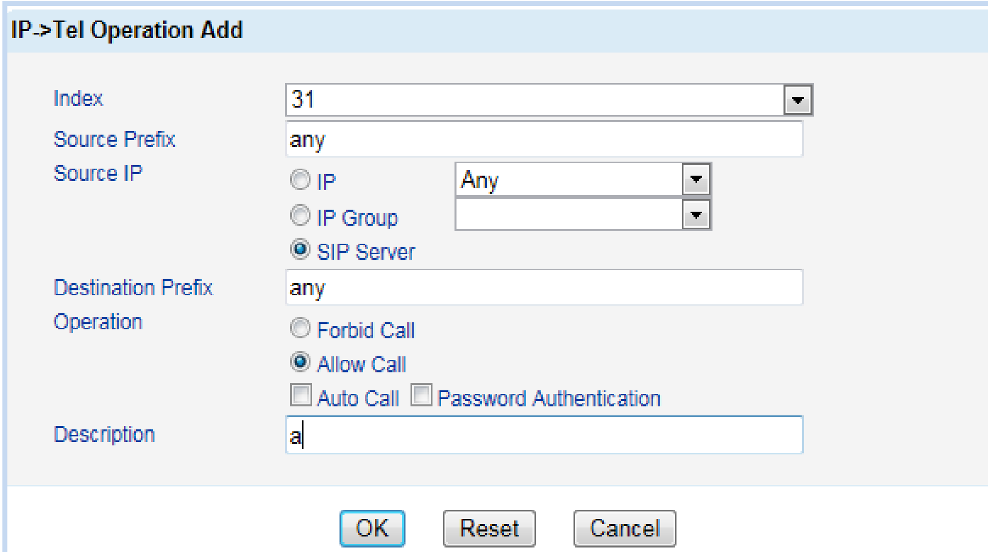


Figure 4-9-2 IP->Tel Operation Add

For example: The above configuration said: Allow all calls from SIP server.

4.9.2 Tel->IP Operation

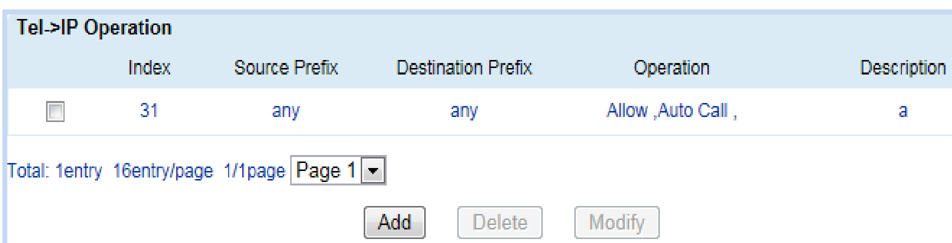


Figure 4-9-3 Tel->IP Operation

Table 4-9-2 Tel->IP Operation

Index	Number, value range from 0-31
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

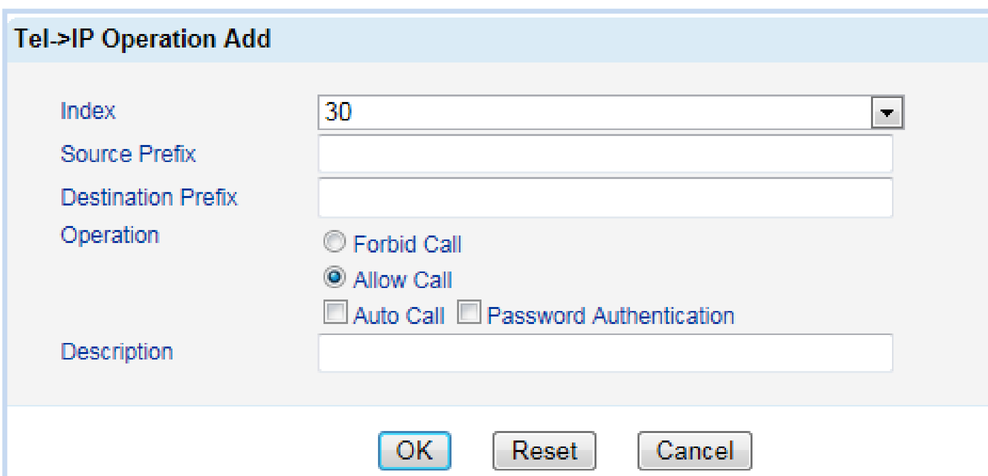


Figure 4-9-4 Tel->IP Operation

Pictured above, allow all call generation from port dial a number to IP side.

4.10 IP Trunk

4.10.1 IP Trunk

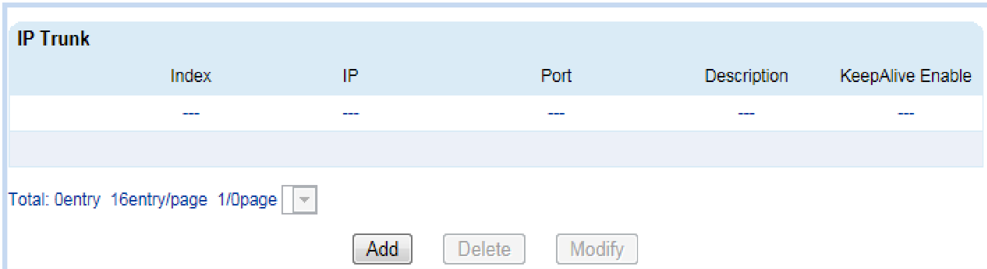


Figure 4-10-1 IP Trunk

Table 4-10-1 IP Trunk

IP Trunk	When device connected to softswitch or SIP server, equipment and the soft switch exchange information through the IP trunk.
Index	Number, value range from 0-31
Description	It describes the route for the ease of identification. Its value is character string
IP	To end the soft switch or SIP server IP
Port	To end the soft switch or SIP server port

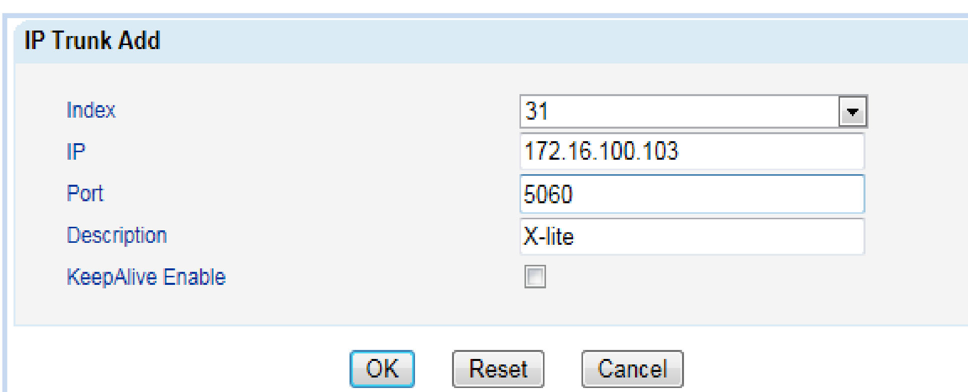


Figure 4-10-2 IP Trunk Modify

For example, No.31 trunk to connect to SIP server of 172.16.100.103:5060.

4.10.2 IP Trunk Group

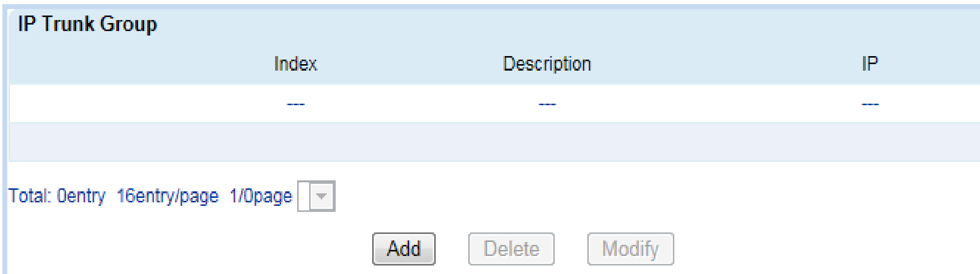


Figure 4-10-3 IP Trunk Group

Table 4-10-2 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	Number, value range from 0-31
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. When building the trunk later can choose some trunk form a trunk group.

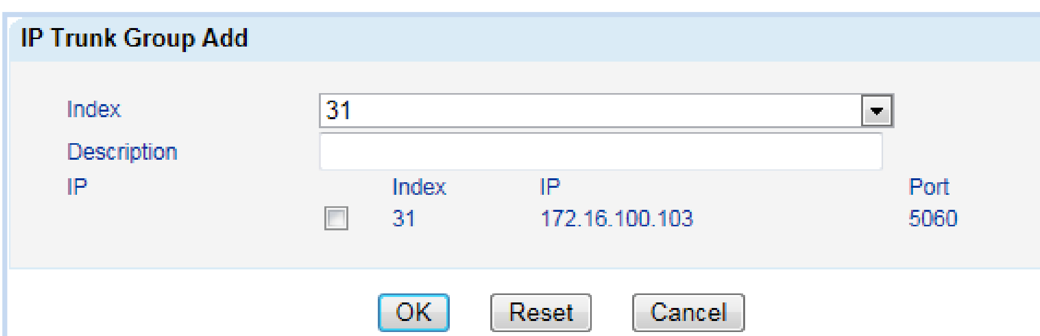


Figure 4-10-4 IP Trunk Group Add

For example, NO.31 trunk group has only one trunk NO.31, a trunk group can have multiple relay, a relay can only belong to a trunk group.

4.11 System Configuration

4.11.1 System Configuration

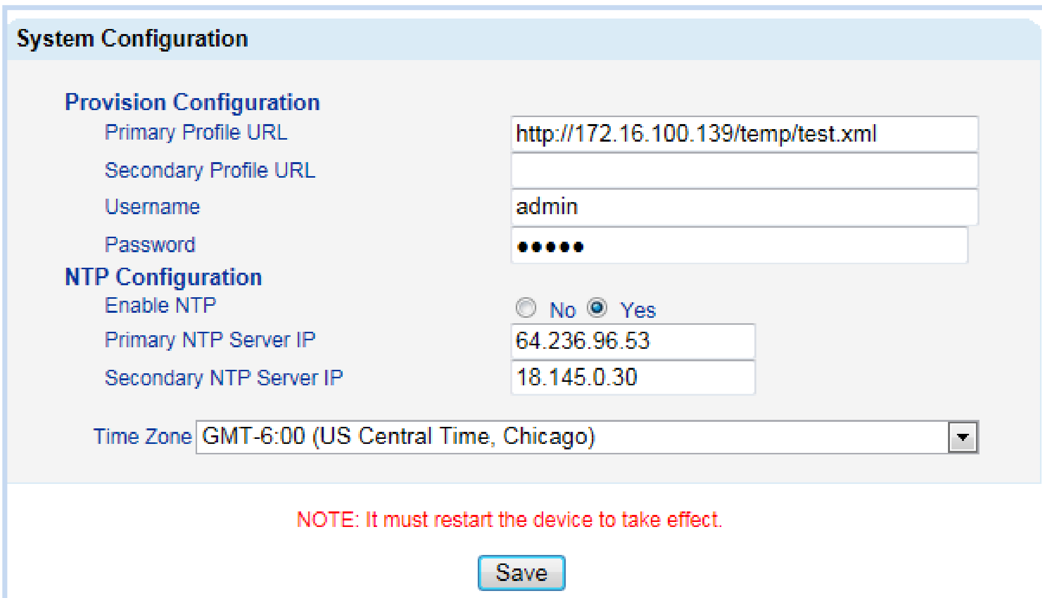


Figure 4-11-1 System Configuration

Network Time Protocol（NTP）is a protocol used to make computer time synchronization. It can make the computer to the server or clock source do synchronization, provide high precision accuracy of time correction.

4.11.2 Service Configuration

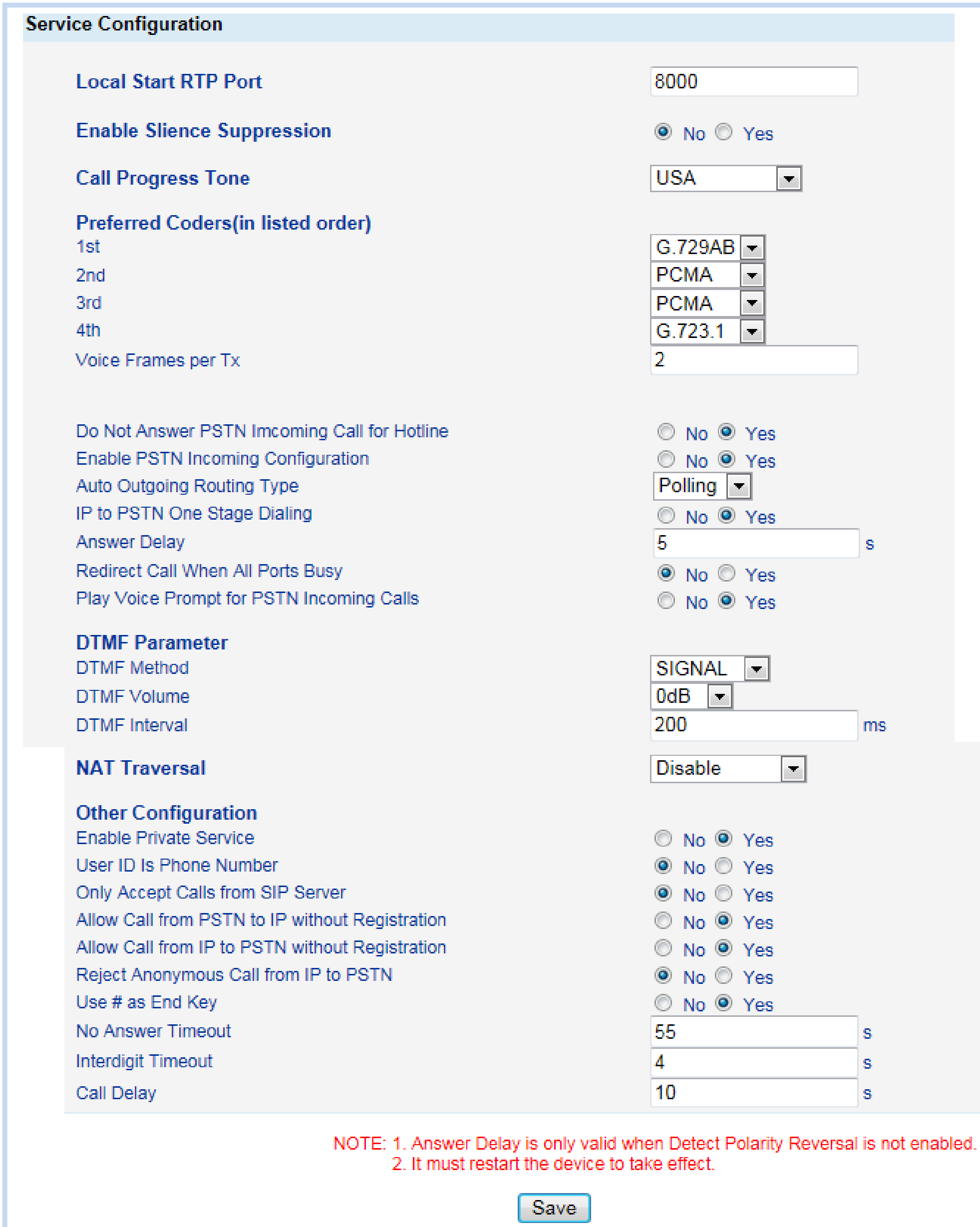


Figure 4-11-2 Service Configuration

Table 4-11-2 Service Configuration

LOCAL Start RTP PORT	Real-time Transport Protocol details the standard packet format on the Internet to deliver audio and video. The initial allocation of Channel when RTP voice stream transmit in the IP

	network. In general, using the factory default values. When there are multiple D-LINK series voice products, and the network gateway or router's NAT with loopholes, user can try changing this item.
Enable Silence Suppression	Silence suppression technique to ensure that only wehn talking to both sides in call to take up bandwidth and improve the utilization rate of channel. 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.
Do Not Answer PSTN Imcoming Call for Hotline	Inbound hotline immediately pick up, after waiting for VOIP side picked .
Enable PSTN Incoming Configuration	Select "Yes", users can configure the device through dialing feature codes.
Auto Outgoing Routing Type	When adopt two stage dialing, this configuration option takes effect and routing doesn't take effect. Ordinary mode means Minimum port selected. Polling means that according to the port in turn choice.
IP to PSTN One Stage Dialing	This function will be displayed only when select "Enable Auto Outgoing Routing" function, the User ID will be sent directly to PSTN, for example: the user calls 6715, the device will sent 6715 User ID to PSTN
Answer Delay	Only for CDMA effect, when IP to PSTN call and don't open the

	extremely signal detection, then call send out after a few seconds delay to connect.
Redirect Call When All Ports Busy	When all the GSM port occupied, this function will switch the call toanother equipment and need to provide the device IP and port.
Play Voice Prompt for PSTN Incoming Calls	Setting is yes, when through the PSTN calls to the Channel, the device will with the clew tone, the default is "Please dial the extension User ID"; setting to No, the device will with dial tone
DTMF Parameter	Support RFC2833 and SIGNAL two ways. DTMF INTERVAL range is 50 ~ 800ms, DTMF VOLUME can use the default Configuration.
NAT Traversal	Network Address Translation is a private (keep) addresses into legitimate IP address conversion technology. Including three ways:STUN, static NAT and dynamic NAT.
Enable Private Service	Start with "*" the beginning of a local business, such as *158# inquiry IP address.
User ID is Phone Number	SIP compatibility configuration, INVITE news in the head is carry "User = Phone" parameters
Only Accept Calls from SIP Server	Only accept SIP server launch of call and refused to other sources of call.
Allow Call from PSTN to IP without Registration	Reference “Is register” of “SIP Configuration”, if "Is register" setting is no, this option need set Yes ,to avoid that the devices can not call out. This option allows equipment not registered on the phone.
Allow Call from IP to PSTN without Registration	Reference “Is register” of “SIP Configuration”, if "Is register" setting is no, this option need set Yes ,to avoid that the devices can not call out. This option allows equipment not registered on the phone.

Reject Anonymous Call from IP to PSTN	The call from IP to PSTN will be rejected.
Use # as END Key	General dial-up end of the logo has two kinds: 1. # operator as dial-up end, 2. Waiting for a few seconds until dial-up overtime.
No Answer Timeout	Call in or out over a certain time no response, the call off.
Interdigit Timeout	Bit of between the dialing time ,over the time will be seem as end of dial.
Call Delay	When a SIM card to call on the broken, in this delay time will not accept new calls, ensure the call the success rate.

4.11.3 SIP Configuration

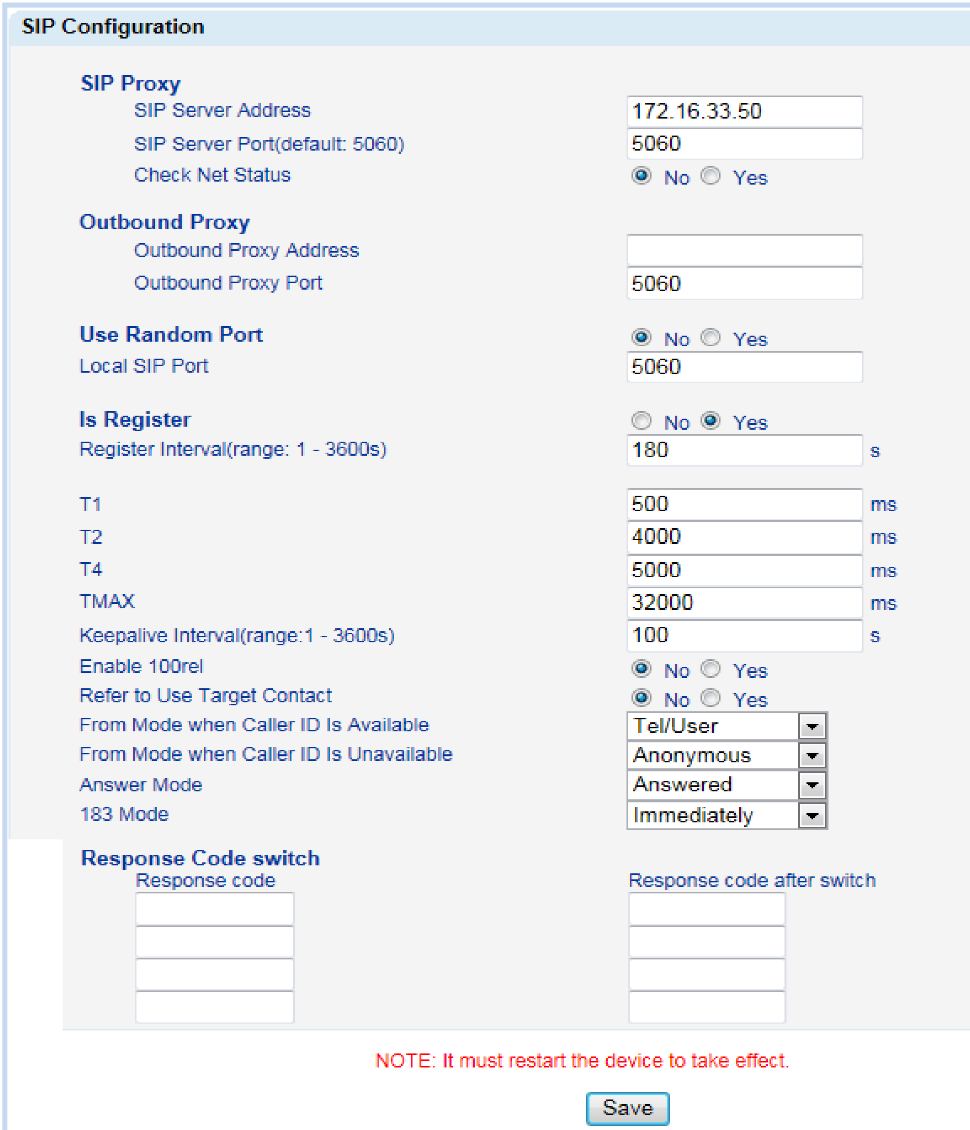


Figure 4-11-3 SIP Configuration

Table 4-11-3 SIP Configuration

SIP Proxy	In SIP a proxy server realize voice over IP based on the exchange. SIP server address can be IP address, can also is a domain name.
Outbound Proxy	Outbound Proxy is ususlly used in network with firewall/NAT. Used for processing signal and help multimedia data pass through the firewall.
Check Net Status	According to Keep alive interval, and constantly to the equipment to send messages, check the network connectivity.

SIP server Port	Local SIP listening socket, can choose the random or fixed. Random is selecting a random port when start device. Fixed port can be specified by customer. Default is 5060.
Is Register	DVG-6001G can work at two work mode: register and unregister. Default is register mode.
Register Interval	Registration time intervals of equipment to SIP server or outbound proxy registration.
T1	T1 timer of SIP protocol, default is 500ms
T2	T2 timer of SIP protocol, default is 4000ms
T4	T4 timer of SIP protocol, default is 5000ms.
TMAX	SIP compatibility configuration, after sending a SIP request, it is overtime if had not received any response in the largest waitting time of response retransmission. The largest waitting time of response retransmission double after response retransmission.
Keep alive Interval	Used for communication between device and SIP server and ensure the state of equipment registered. Often use the factory default.
Enable 100rel	SIP compatibility configuration, Used when the news comes to send 100 to each other PRACK reply.
Refer-to Use Target Contact	SIP compatibility configuration, fill in contact header in “Refer-to” fieldof SIP message.
From Mode when Caller ID is Available	SIP compatibility configuration, FROM field used to transfer Caller ID. 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	SIP compatibility configuration, used for transmission ID when no Call

when Caller ID is Unavailable	ID Numbers FROM field. Anonymous : From: <sip: Anonymous @IP>;tag=51088abb Username : From: <sip: Username @IP>;tag=51088abb
Answer Mode	Response way of IP to PSTN side, includes:Answered and Alerted. If select “Answered”, SIP protocol back to 200 news on the side hook; if select “Alerted”, SIP protocol back to 200 news on the side ringing. Usually keep default Settings.
183 Mode	Reply 183 news after reply100 or ringback. Usually keep default settings.
Response Code switch	SIP compatibility configuration. Response code is SIP news code. Suchas: 183 and 100. User can modify response code in this configuration items.

4.11.4 Port Configuration

Figure 4-11-4 Port Configuration

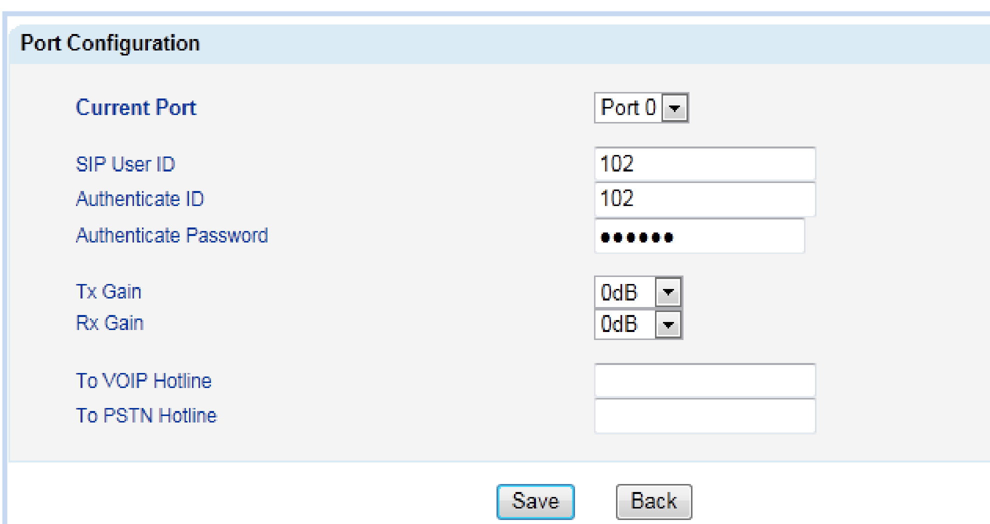


Figure 4-11-5 Port Configuration

Table 4-11-4 Port Configuration

Current Port	Choose the current registration port
SIP User ID	Used to SIP server registered the authentication, SIP registered account number is the phone part of users in SIP address, and is often used as an ID information callers, displayed in SIP software or phone on the LCD. The typical cases, SIP registered account number is a phone number or expanded the number, or a user name.
Authenticate ID	The authentication name is strictly to the authentication purpose, is telephone contact SIP server to verify user identity with.SIP User ID could be the same with authenticate ID, also can not.
Authenticate Password	SIP account register password.
Tx Gain	Gain from PSTN side, default is 0.
Rx Gain	Gain from IP side, default is 0.
To VOIP Hotline	PSTN side calls the port, the port immediately sent hotline number to IP side after hook.
To PSTN Hotline	IP side calls the port, the port immediately sent hotline number to PSTN side after hook.

4.11.5 Digit Map

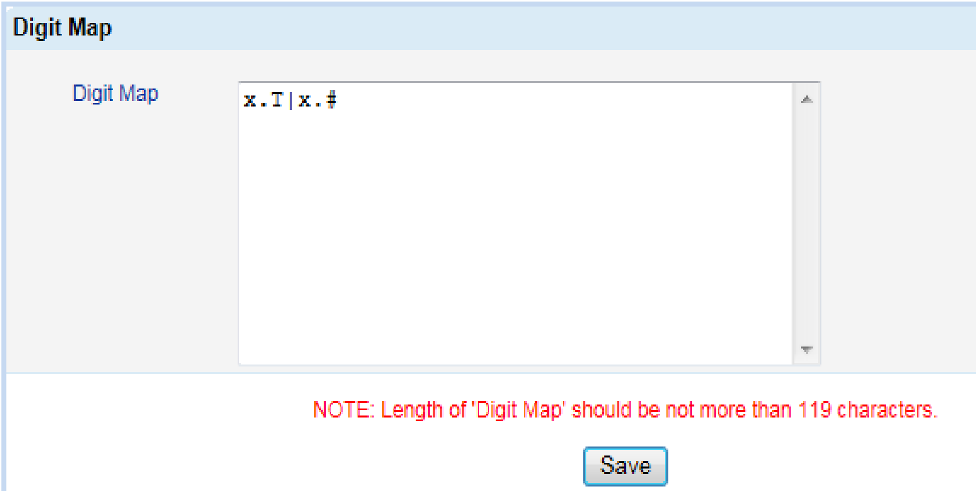


Figure 4-11-6 Digit map

Digit Map Syntax:

1. Supported objects

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

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

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

2. Range []

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

3. Range ()

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

4. Separator

   |: Separated expressions or DTMF symbols.

5. Subrange

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

6. Wildcard

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

7. Modifiers

.: Match 0 or more times.

8. Modifiers

+: Match 1 or more times.

9. Modifiers

?: Match 0 or 1 times.

Example:

Assume we have the following digit maps:

1. xxxxxxx | x11

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

2. [2-8] xxxxxx | 13xxxxxxxxx

Means that first is "2","3","4","5","6","7" or "8", followed by 6 digits; or first is 13, followed by 9 digits.

3. (13 | 15 | 18)xxxxxxxxx

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

4. [1-357-9]xx

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

4.12 Tools

4.12.1 Firmware Upload

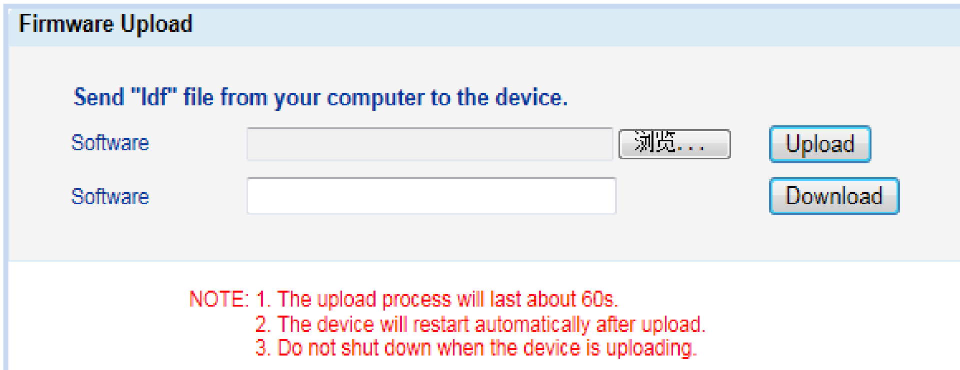


Figure 4-12-1 Firmware Upload

Please consult equipment provider before upgrading, select the appropriate firmware version. Click browse choose appropriate firmware, and then click upload. Uploading please don't shut off the power, otherwise lead to paralysis of equipment.

4.12.2 IVR Voice Prompt Upload

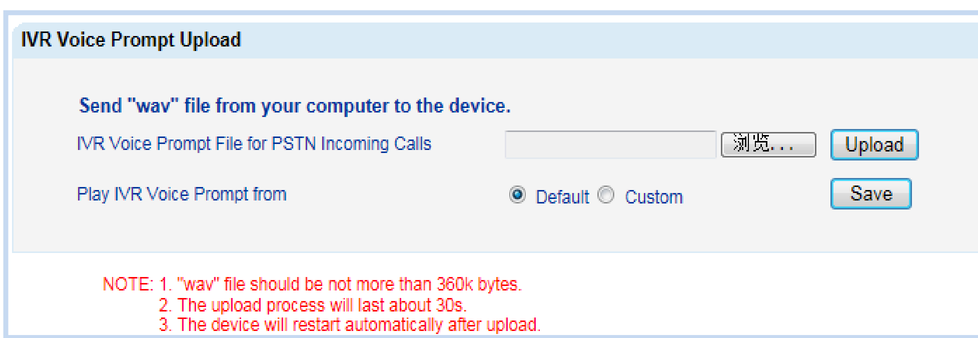
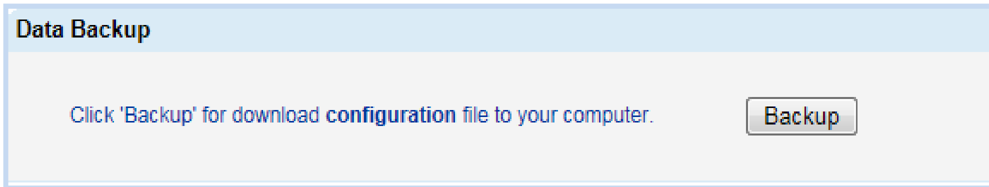


Figure 4-12-2 IVR Voice Prompt Upload

The default is when the telephone call in the PSTN, play is the default IVR "please dial the extension number", users can customize the IVR voice, and through the menu loading. Please note that loaded IVR file format must for 8000 Hz, 16 bit sampling mono wav format, and can't more than 360 KB.

4.12.3 Data Backup

Figure 4-12-3 Data Backup



When a device configuration is finished, please click data backup and saved the configuration file in reliable place. When the equipment malfunction or have other equipment needed to add, through the data restore function rapid configuration a function similar or the same equipment.

4.12.4 Data Restore

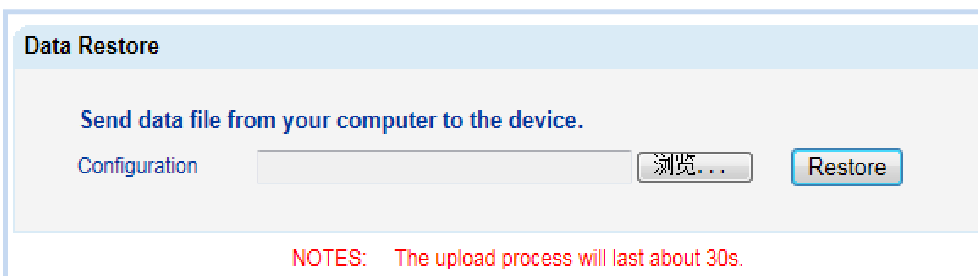


Figure 4-12-4 Data Restore

Importing backup data to equipment can save configuration time. Import will reboot. Network configuration can't through the data recovery.

4.12.5 Syslog Parameter

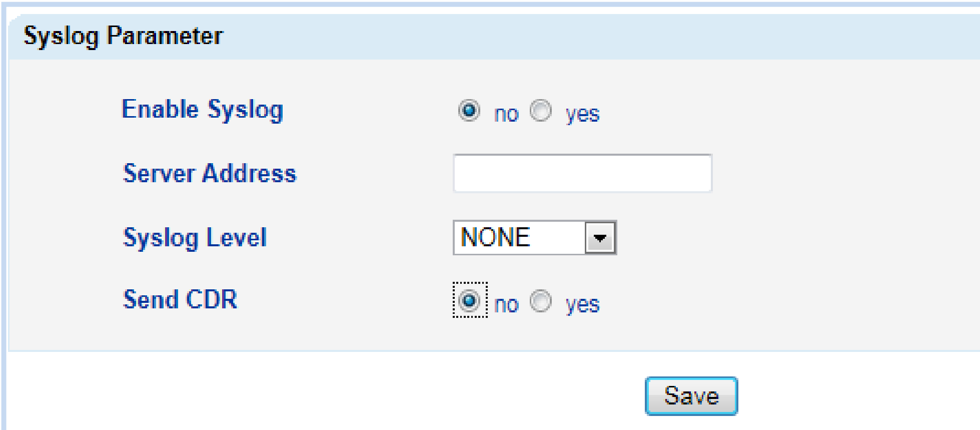


Figure 4-12-5 Syslog Parameter

Table 4-12-1 Syslog Parameter

Server Address	Syslog information will be saved in Syslog server. Fill in Syslog server IP.
Syslog Level	The information contained in the system logs are: NONE、DEBUG、NOTICE、WARNING、ERROR. At present, only NONE and DEBUG level effective.
Send CDR	If choose CDR, Syslog level should be chosen NONE.

4.12.6 Login Password

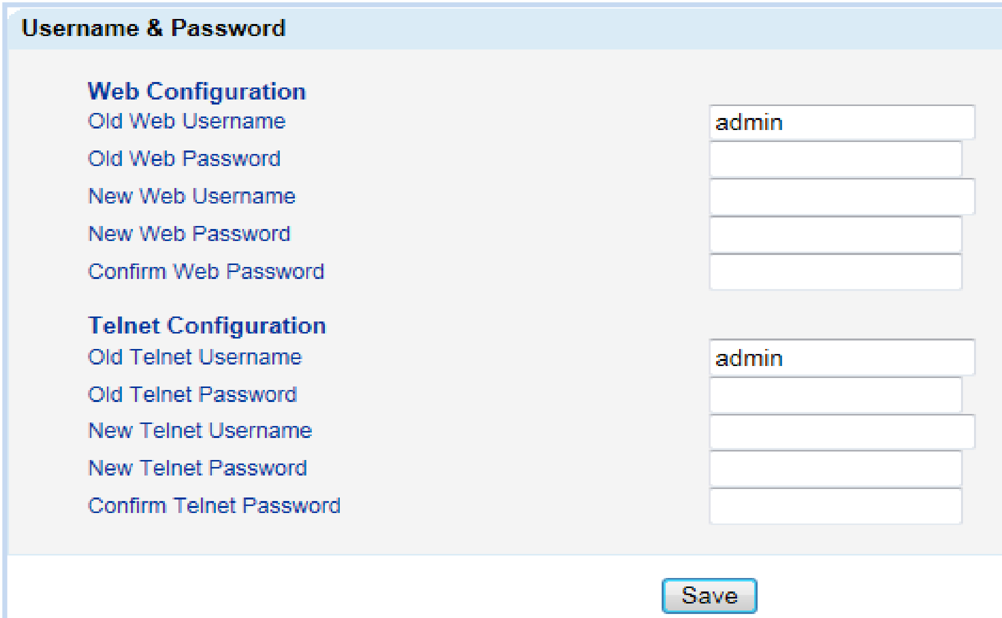


Figure 4-12-6 Username and Password

The default WEB and TELNET user name/password is admin/admin,, if open the remote login please modify user name and password to prevent others to use the default user name and password to land.

4.12.7 Factory Reset

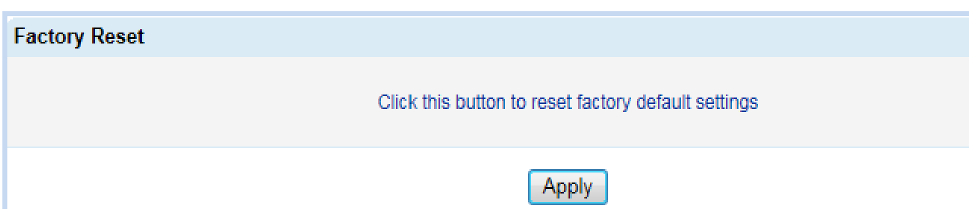


Figure 4-12-7 Factory Reset

Please use caution this operation, the operation will lead to all the parameters recovery factory state, including configuration parameters and network parameters. For safety, before reset factory, please backup configuration files. Pay attention to restore the factory default need to come into force after the restart your device.

4.12.8 Restart

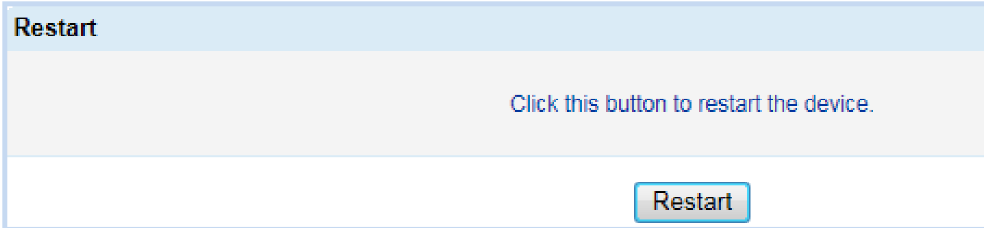


Figure 4-12-8 Restart

By clicking on the restart can remote restart your device when recovery factory defaults, data recovery or modify system parameters need to restart equipment, please try to use the WEB to restart way to restart.

5. 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