Series GSM/CDMA VoIP Gateway

User Manual

(Version: 2.0)

Content

Chapter 1 Equipment introduction

This chapter mainly introduces functions and structures of GSM/CDMA wireless VoIP gateways, which are named as 2001/2004/2008.

1.1 Introduction

2001/2004/2008 serial is full functions VoIP gateway based on IP and GSM/CDMA wireless network. 200X provides a flexible network configuration, powerful features, and good voice quality. It works for carrier grade, enterprise, SOHO, residential users for cost-effective solution.

1.2 Scenario of applications of products

2001/2004/2008 provides access of GSM/CDMA network.

With the development of users and telecom service, mobile network and fixed network integration will be steadily increasing. 2001/2004/2008 provides high quality VoIP service. 2001/2004/2008 perfectly meets the requirement. This is a scenario shown as figure 1-1

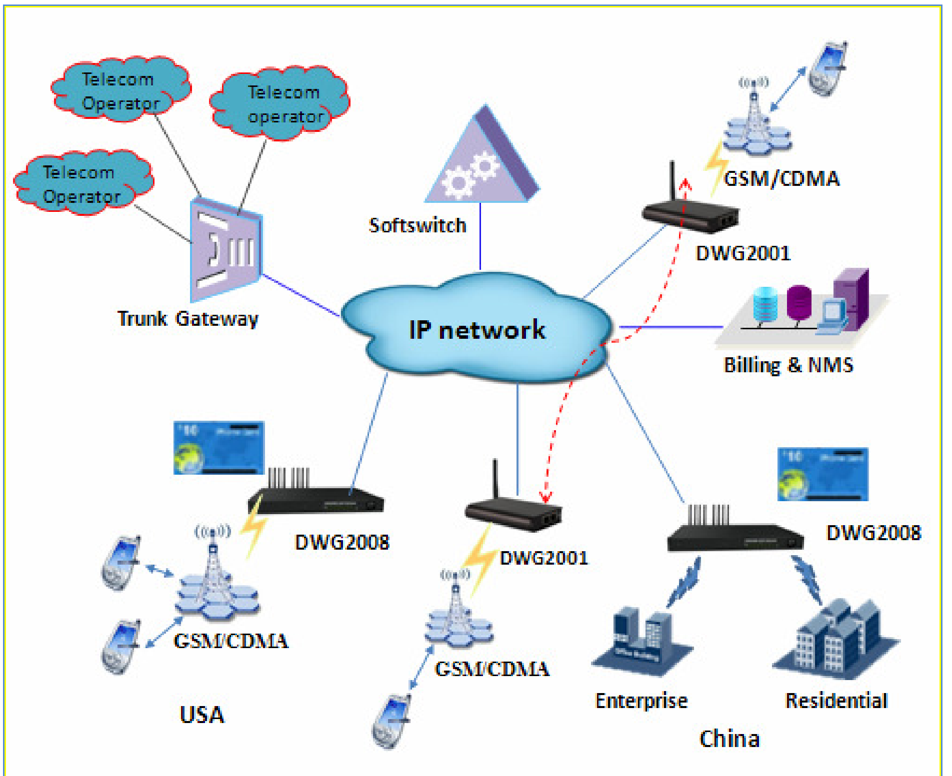


Figure 1-1

2004/2008 can be implemented as VoIP call termination in VoIP operation, the solution shown as figure 1-2

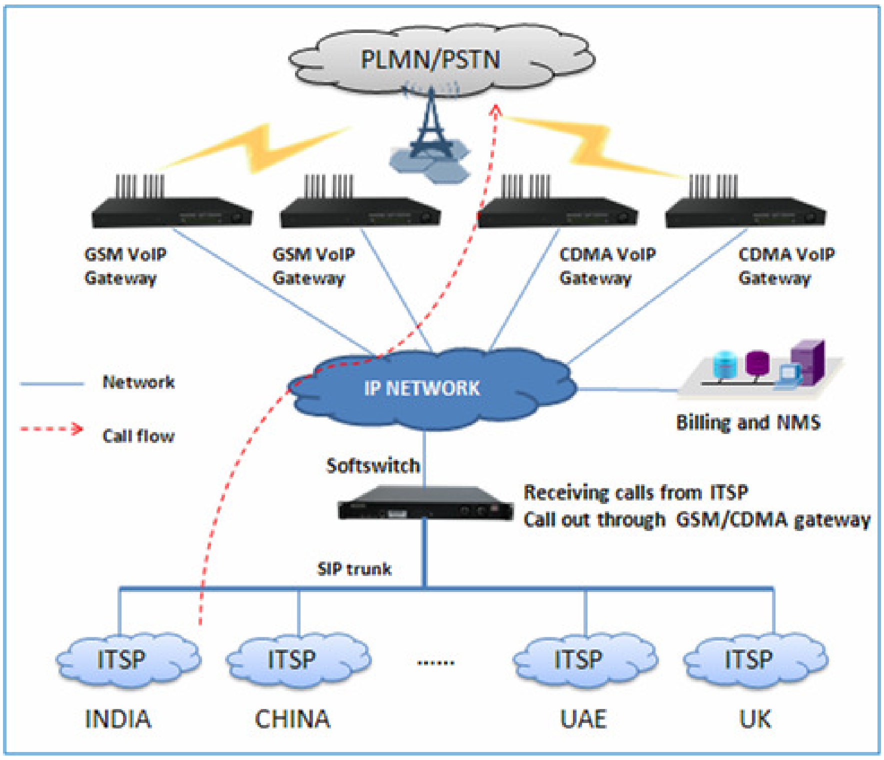


Figure 1-2

1.3 Product appearance

Figure 1-3 Front view of 2004/2008 Figure 1-4 2001

Notes: 2004 equips 4 antennas, 2008equips 8 antennas; both models have the same appearance.

2001/2004/2008 serial supports GSM, CDMA network and frequency supported as below:

GSM: Support quad-band 850/900/1800/1900MHz

CDMA: Support 800MHz

There are 3 models with 200X serial. Three models are shown in table 1-1:

Model	Network Type	Interface
2001	GSM/CDMA	1WAN,1LAN ,1SIM channel
2004	GSM/CDMA	1WAN,3LAN, 4SIM channels
2008	GSM/CDMA	1WAN,3LAN, 8SIM channels

Table 1-1 2001/2004/2008 interface description

1.4 Functions and features

1.4.1 Protocol standard supported:

- Standard SIP and MGCP 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);
- GSM/CDMA;
- ITU-T G.711 α -Law/ μ -Law、G.723.1、G.729AB;

1.4.2 System function

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

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

1.4.3 Industrial Standards supported:

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

1.4.4 General hardware specification

Power Supply:

2001: AC100~240V 50/60HZ DC12V/1A

2004/2008: AC100~240V, 50/60HZ

Temperature: 0~40 ℃ （Operation）, -20~80 ℃（storage）

Humidity: 5%～90%RH,

Power Consumption: 2001: 5W, 2004/2008: 30W

Dimensions:

2001: 112(W) x76(D) x24(H) mm

2004/2008: 440(W) 305(D) x44(H) mm

Net weight: 2001: 0.7kg, 2004/2008: 5.0kg

1.5 Structure of product

The appearance of 2001/2004/2008 serial shows in two views (front view and rear view).

Figure 1-5 Front view of 2004/2008

Figure 1-6 Front view of 2001

1.5.1 Front view

There is variety of indicators on front view of 2001/2004/2008, its names and indications shown as table 1-2:

Indicators	Color	Name	Status	Description
PWR	Green	Power indicator	Lighting	Power is on
			Off	Power failed
RUN	Green	Running indicator	Fast Blinking	SIM cards Registered
			Slow blinking	Unregistered
Occupancy indicator of channels	Red	Indicate that occupancy status of channels	Lighting	Used
			Off	Un-used
Signal indicator	Green	GSM/CDMA signal strength indicator	3 grids	Very strong
			2 grids	Strong
			1 grid	Weak

Table 1-2 Description of indicators on front view of 2001/2004/2008

Note: No signal strength indicator on 2001

1.5.2 Rear view

1) Description of interfaces on rear view of 2004/2008 shows as below:

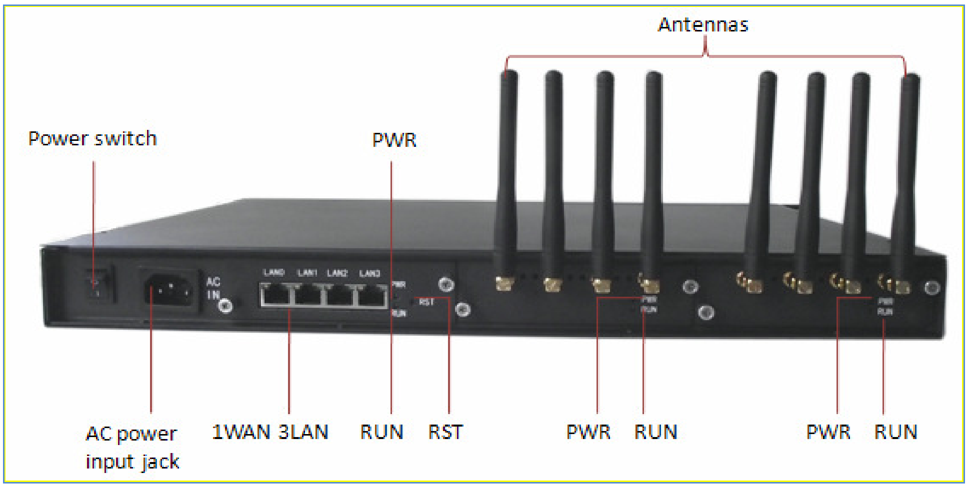


Figure 1-7 Rear view of 2004/2008

2) Description of interfaces on rear view of 2001 shows as below:

Figure 1-8 Rear view of 2001

1.6 Restore factory setting

Push RST button for 6 seconds of 2001/2004/2008, the device will restore factory setting.

Chapter 2 Equipment installation

This chapter mainly introduces 2001/2004/2008 hardware installation and connection of equipment.

2.1 Installation notice

2004/2008 used AC110~220V, 50/60Hz AC power supply, 2001 uses DC12V power. Power supply should ensure the reliability and stability; otherwise, it may damage the SIM card or device. In addition, make sure the power supply connects to ground bar well. With right ground protect connection, that can reduce the surge voltage caused by lightning that damage the equipment, and ensure voice quality (note: when calls with irregular noise occurring, please check the power whether connect ground well). Common measures are as follows:

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

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

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

Network interface of 2001/2004/2008 supports RJ45 standard with 10Mbps or 100Mbps network.

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

To check the network connection, user can monitor the registered status indicator (refer Table 1-2)

2.2 Installation procedure

The outlook of 2001/2004/2008 looks like a 1U chassis; to install hardware the cable is needed. After unpacking the equipment, please do follow the procedure as following steps:

2.2.1 Install SIM card

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

Figure 2-1 Loosen screw of panel

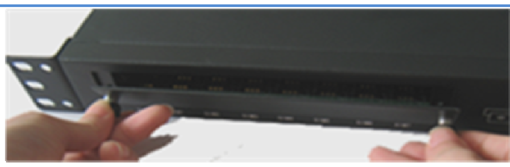


Figure 2-2 Remove panel of SIM slot

Figure 2-3 Install SIM cards

Figure 2-4 Fix panel of SIM slot

below:

2.2.2 Antenna installation

Figure 2-5 shows as below a compelete installation:

Figure 2-5 Antenna installation

2.2.3 Cable connection of equipment

2004/2008 works in Bridge mode

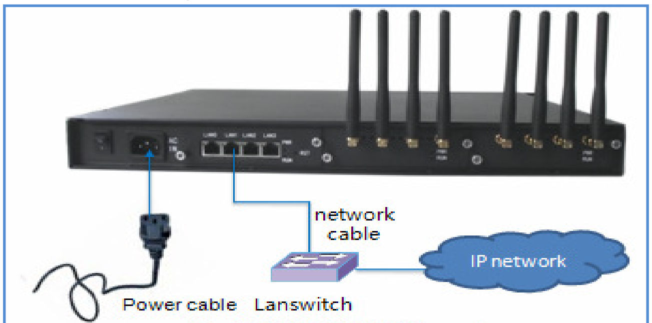


Figure 2-6 WIA2004/WIA2008 connection

2001 works in Route mode

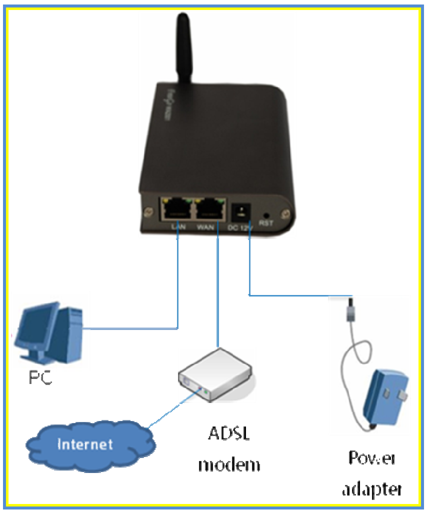


Figure 2-7 2001 connection

2.3 Establishing remote access

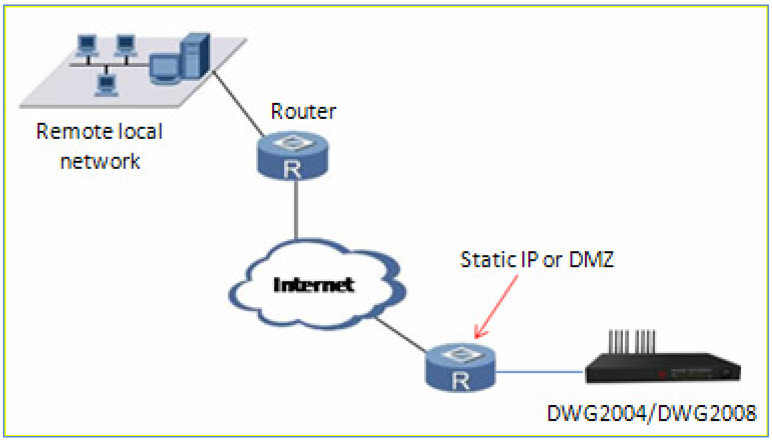


Figure 2-8 Remote login environments

After installation of 2001 /2008, creating a remote access for technical support.

Chapter 3 Network configuration

In this chapter we will introduce the initial configuration of 2001/2004/200 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 power on
3) Make sure the gateway is connected with the network
4) Completed the SIM installation
5) 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 theatstep again.

2001/2004/2008 can work in two modes: route mode and bridge mode. when the gateway is under bridging mode, user should configure network parameters of WAN port; when the gateway is under the route mode, user should configure LAN port.

3.3 General feature codes for system setting

Dial numbers	Features
*114#	Play the phone NO.
*115#	Check the TT number of gateway (using just when the device interconnects with other series of DINSTAR products)
*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#	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 the 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

Table 3-1 Feature codes for system setting

3.4 Static IP Configuration

This chapter introduces IP configuration of 2001/2004/2008 through calling IVR. Assuming the IP address of a 2001/2004/2008 device is 172.16.0.100, subnet mask is 255.255.0.0, IP of gateway is 172.16.0.1, configured as follows:

1) Insert a SIM card into the 2001/2004/2008 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 DHCP configuration

DHCP mode configure as follows:

1) Insert an SIM in one of the port, Dial the SIM card phone number. When hearing a message, then enter "*150*1#", hang up when hearing " setting successful" message;
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 .

Chapter 4 WEB configuration

This charpter describes web configuration of 2001/2004/2008.

4.1 Preparing

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

Dial "* 158 #" to get the IP address of the gateway, log on the web interface by a browser.

4.2 Access the system through http

Enter IP address of 2001/2004/2008 in browser, and the GUI shows as below:

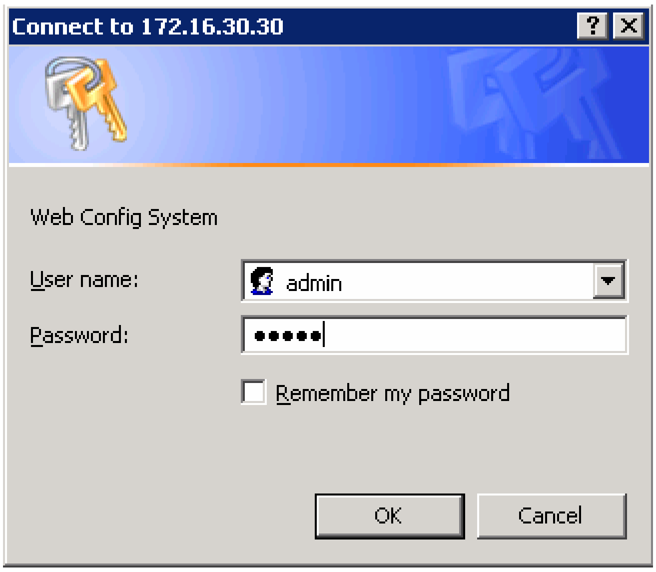


Figure 4-2-1 WEB log interface

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

4.3 WEB configuration

4.3.1 WEB Operation Guide

2001/2004/2008 WEB configuration interface consists of the navigation tree and the detail configuration interfaces.

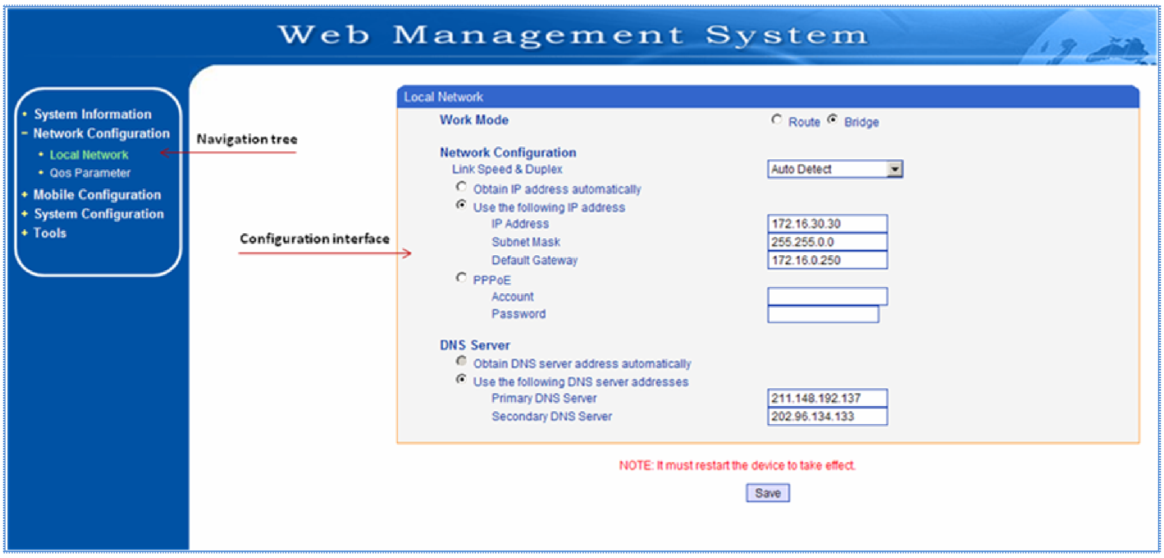


Figure 4-3-1 WEB introduce

Go through navigation tree, user can check, view modify, and set the device configuration on the right of configuration interface.

4.4 System Information

Figure 4-3-2 system information

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

4.4.1 System information

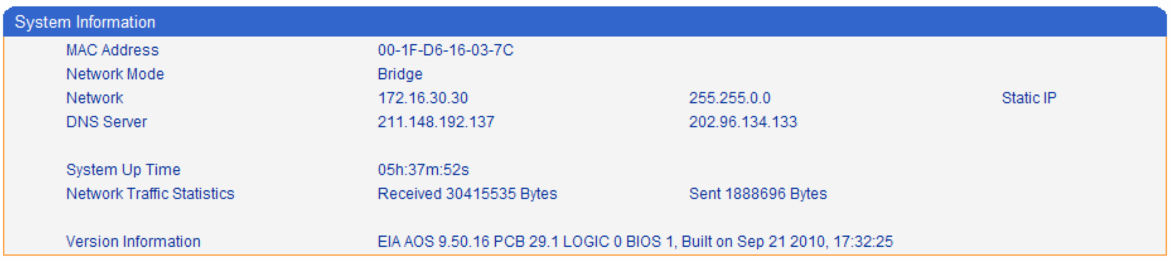


Figure 4-3-3 system information

【MAC Address】Displays the current MAC of the gateway, for example: 00-1F-D6-16-O3-7C;

【Network Mode】2001/2004/2008 support two types network mode, which is bridge and route modes;

【Network】shows IP address and subnet mask.

【DNS Server】Displays DNS server IP address in the same network with the gateway.

【System Up Time】shows the time period of the device running. For example,:1h:

20m, 24s;

【Network Traffic Statistics】Calculates the netflow, including the total bytes of message received and sent.

【Version information】shows the current firmware version, for example EIA AOS 9.50.16 PCB 29.1 LOGIC 0 BIOS 1, Built on Sep 21 2010, 17:32:25;

4.4.2 Mobile information

Mobile Information

Port	Type	IMSI	Status	Remaining Call Duration(min)	Carrier	Signal Quality
0	GSM		No SIM Card	19999		
1	GSM		No SIM Card	No Limit		
2	GSM		No SIM Card	No Limit		
3	GSM		No SIM Card	No Limit		
4	GSM		No SIM Card	No Limit		
5	GSM		No SIM Card	No Limit		
6	GSM		No SIM Card	No Limit		
7	GSM		No SIM Card	No Limit		

Figure 4-3-4 mobile information

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

【Port】Numbers of ports of GSM/CDMA.

【Type】Indicates the current type of network. Such as CDMA or GSM

【IMSI】International Mobile Subscriber Identity, it is the uniquely identifies of SIM card.

【Status】indicates the connection status of current GSM / CDMA module.

【Remaining Call Duration】Limite a call duration to the SIM card, when call duration is out of that duration, the call would be discontinued. This option shows remaining talk time.

【Carrier】Displays the network carrier of current SIM card.

【Signal Quality】Displays the signal strength of in each channels of GSM / CDMA

4.4.3 SIP Information

SIP Information

Port	SIP User ID	Status	Port	SIP User ID	Status
0	20313229	Registered	1	20313230	Registered
2	20313231	Registered	3	20313232	Registered
4	555	Registered	5	556	Registered
6	557	Registered	7		Unregistered

Figure 4-3-5 SIP information

Displays registration status information with Softswitch platform or SIP Server

【Port】the corresponding number of GSM/CDMA channel, identify with 0-9 figure

【SIP User ID】SIP registration account of the Softswitch and SIP server provided

【Registered】 Shows the registration status of VoIP channels, including 2 state of registered and unregistered

4.5 Network Configuration

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

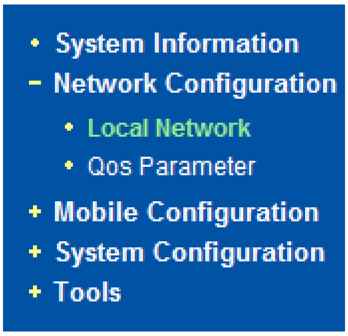


Figure 4-3-6 bridge mode

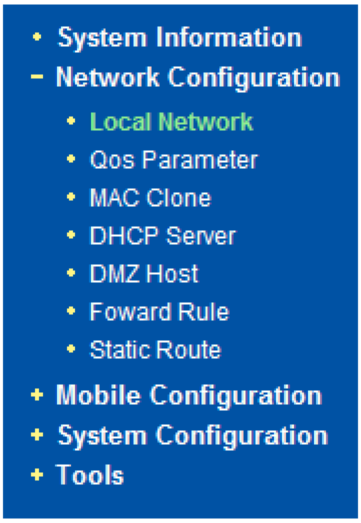


Figure 4-3-7 route mode

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

4.5.1 Local network

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

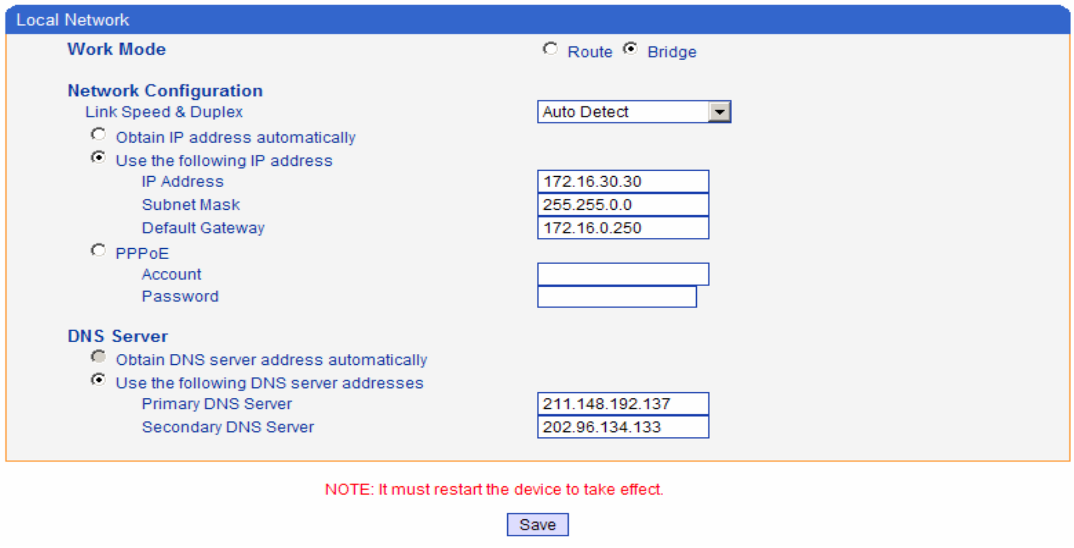


Figure 4-3-8 WEB interface of bridge mode

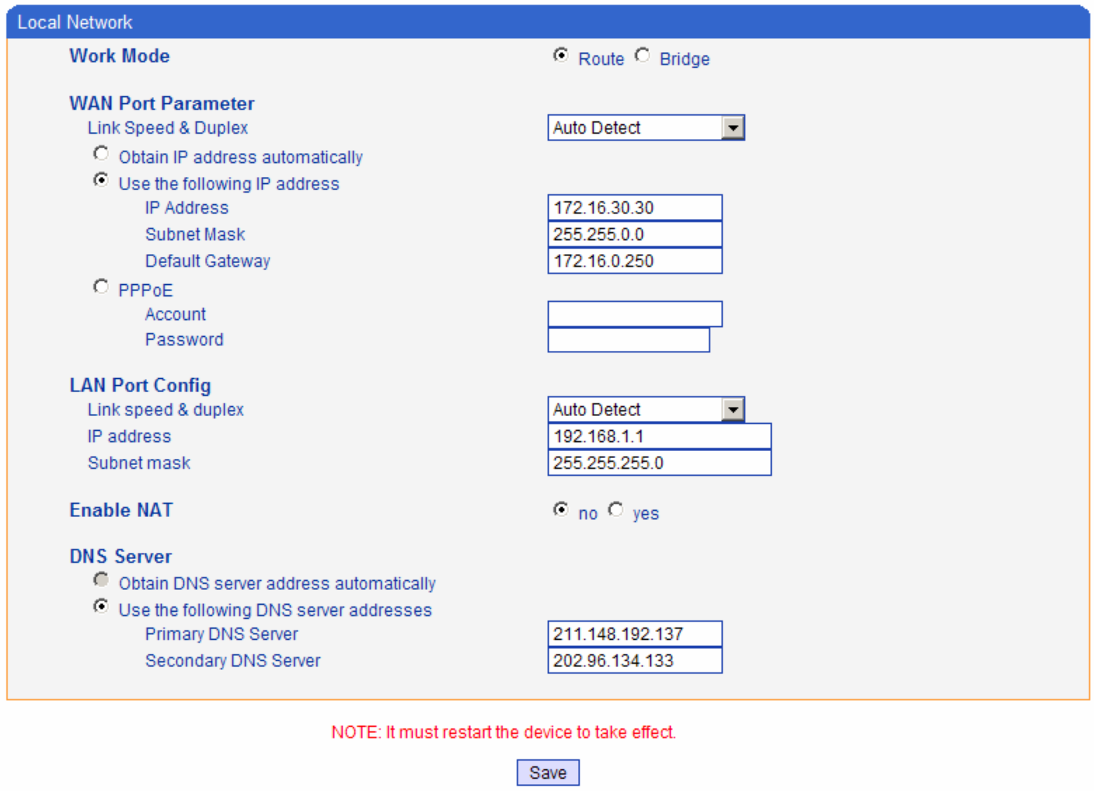


Figure 4-3-9 WEB interface of Route mode

Detailed parameters described as below:

【Work mode】two options of route mode and bridge mode, default is bridge mode.

- Network Configuration

【Link Speed & Duplex】the 5 options are "Auto Detect"," 10Mbps/Half Duplex","10Mbps/Full Duplex", "100Mbps/Half Duplex" and "100Mbps/Full Duplex". Default is "Auto Detect"

【Obtain IP Address Automatically】enable the device obtain IP Address automatically or not. Default is enabling

【Use the Following IP Address】configure the "IP Address"," Subnet Mask" and "Default Gateway" by manual

【PPPoE】need ISP offer the account and password. Use this mode when have not router in the local network.

- LAN port configuration

This option works when "route mode" is selected.

【Link Speed & Duplex】 the 5 options are "Auto Detect"," 10Mbps/Half Duplex","10Mbps/Full Duplex", "100Mbps/Half Duplex" and "100Mbps/Full Duplex". Default is "Auto Detect"

【IP Address】set IP address by manual configuring. Default is 192.168.1.1

【Subnet Mask】set the lan port Subnet Mask by manual configuring, Default is 255.255.255.0

- DNS Server

【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"

Note: all the options in this interface **can** take effect after restarting the device

4.5.2 MAC Clone (configure this only when necessary)

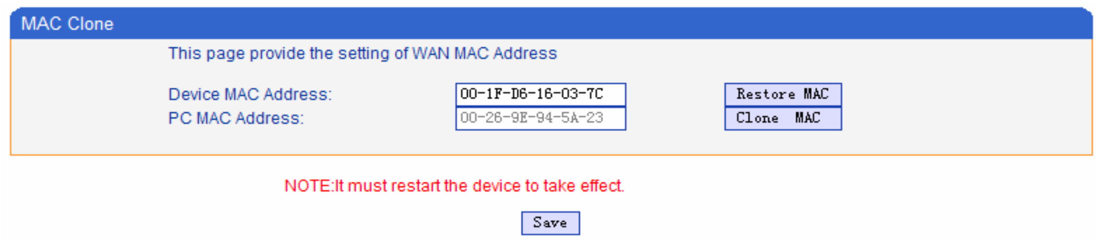


Figure4-3-10 Configuration of Mac Address Clone

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

4.5.3 DHCP Server

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

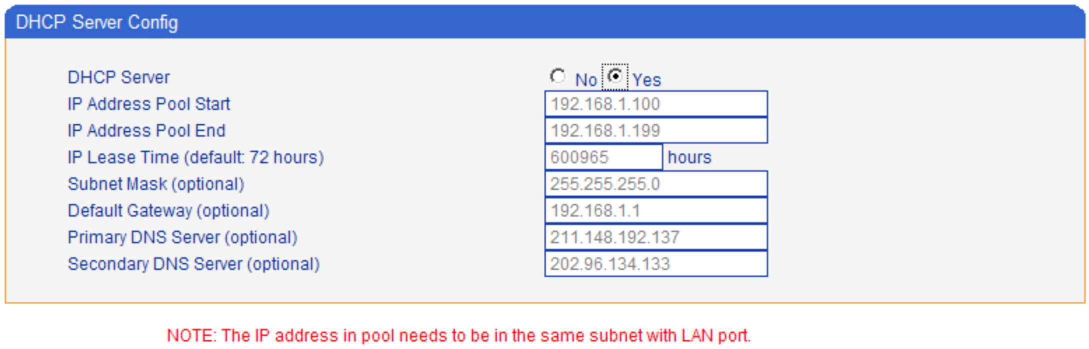


Figure 4-3-11 Configuration of DHCP service

IP address Pool determines the range of IP addess of other devices in this network; IP Lease Time sets the duration for how long the IP works with the specific IP. If the duration is out of the duration, the IP would be invalid;

The subnet mask, gateway, DNS info will also be allocated to network devices automatically by DHCP protocol. Generally, there is no need to configure those items.

NOTE: Please configure the start and end IP addressed in IP address pool, subnet mask and gateway in the same network segment. Other devices in this network may not work when obtaining the IP address.

4.5.4 DMZ Host (Option under "route mode")

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

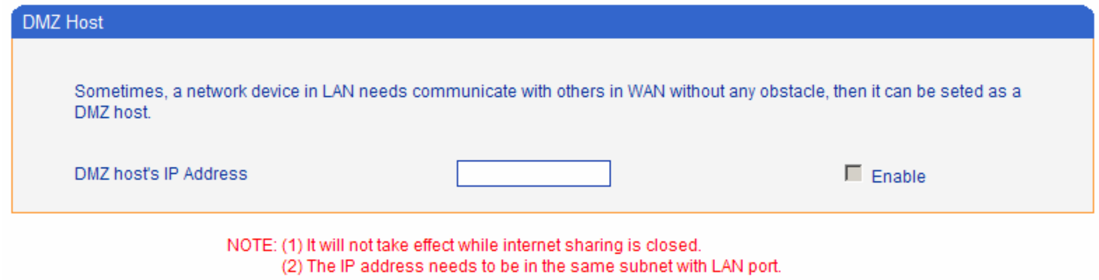


Figure 4-3-12 Configuration of DMZ Host

4.5.5 Forward rules (Option under "route mode")

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

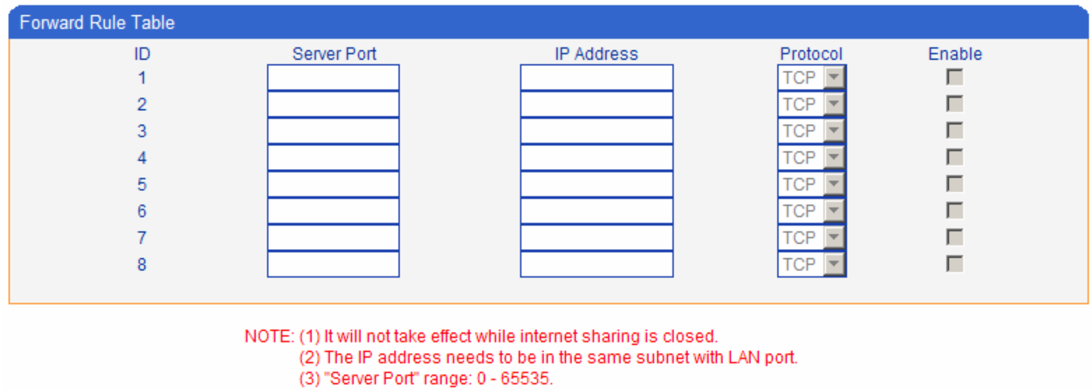


Figure 4-3-13 Configuration of forwarding rules

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

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

4.5.6 Static route(Option under "route mode")

Static route is the route rules in IP communication. Generally speaking, no need to config static route. Configuring static route is necessary in such conditions: when several network segaments exist in LAN network and there's certain application between these network segament. Please cancel “internet sharing” under “ Network configuration” first, then configure the “static route”.

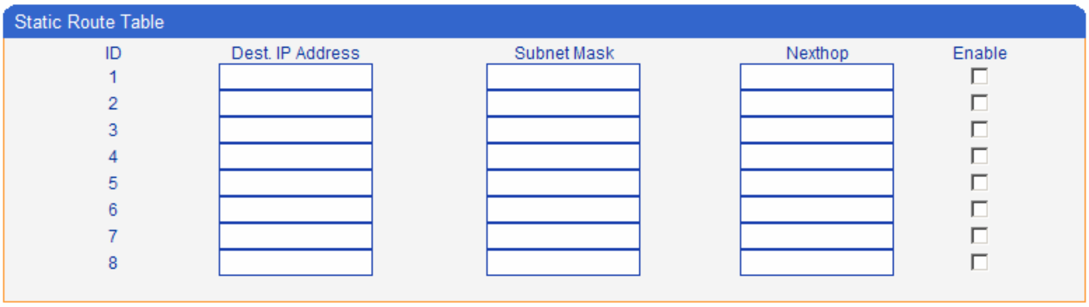


Figure 4-3-14 Configuration of Static route

In commly use, please don’t configure static route. If static rule is wrong, the devices may not work.

4.6 Mobile configuration

4.6.1 Basic configuration

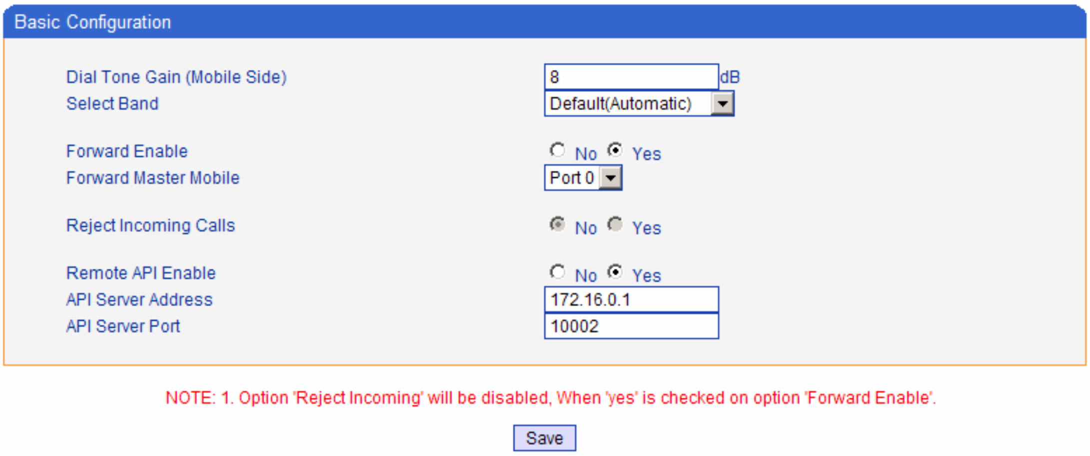


Figure 4-3-15 Basic Configuration

Configuration details are as below:

【Dial Tone Gain（Mobile Side）】it is the dial tone volume of call waiting, dial tone of mobile module when call out. Usually adopt the default configuration.

【Select Band】according to carrier’s band standards. Standards are as belows:

GSM：850/900/1800/1900 MHz ; CDMA：800 MHz;

【Forward enable】enable the function of trunk call in for mobile phone user ID, that is, choosing one user ID of the SIM channels as the access No of device. When several calls use the user ID at the same time, gateway will do call transfer to other available channels. *Configuring the corresponding user ID in each channel is necessary.*

【Forward master mobile】select this channel's mobile phone No. as the one access No.

【Reject Incoming Calls】when enable this function, this channel can only be used to call out, can't used to call in.

【Remote API Enable】 API is provided for third party development with DLL and IAD components. The API includes SMS sending and receiving, USSD sending and receiving. The default is "No"

【API Server Address】it is the remote IP address who uses API. This is an option when selecting "Yes" under "remote API enable".

【API Server Port】it is the remote channel No. who uses API. This is an option when selecting "Yes" under "remote API enable".

4.6.2 Mobile

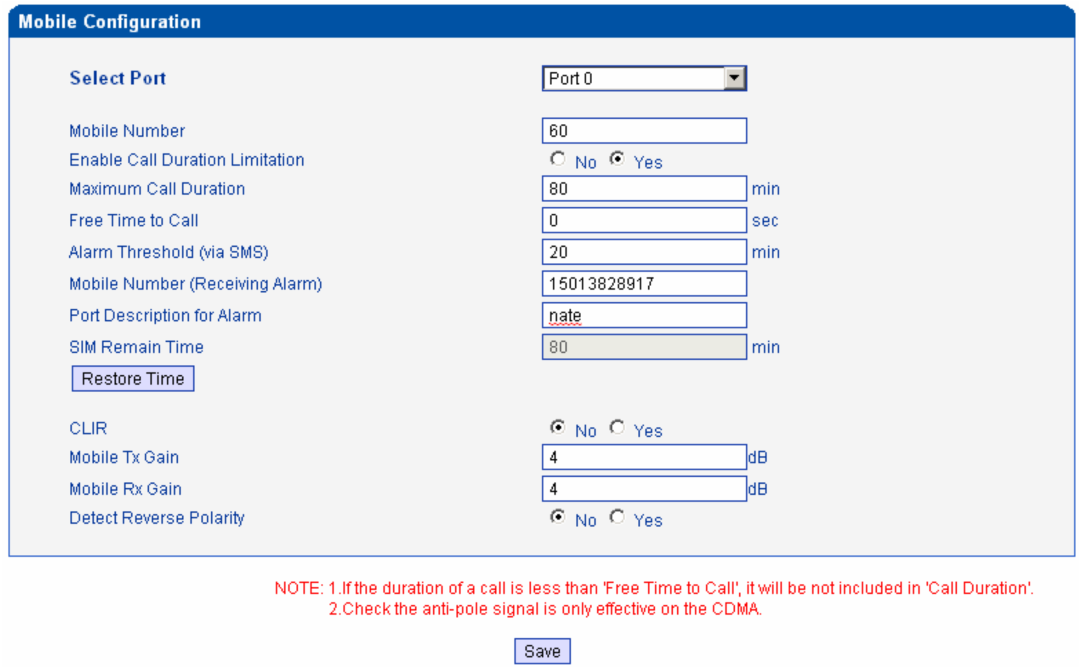


Figure 4-3-16 Mobile Configuration

Configuration details are as below:

【Mobile Number】SIM card user ID of the channel. That must be configured when "one access No." funciton enable.

【Enable Call Duration Limitation】this function is to limit the max call duration of channel. The max call duration is between 1 to 65535 minutes.

【Maximum Call Duration】defines a value by users. That will limit the SIM/UIM card's total call duration. After the call duration excesses this value, no call will be initiative by this channel. The value range is 1-65535. If user doesn't configure this value, Default is no max call duration limits for this channel.

【Free time to call】a minimum charging time (in seconds) is defined during which no charging is done at carrier side. If the conversation time is even shorter, the total call duration will not decrease.

【Mobile Number（Receiving Alarm）】 the mobile phone No. which used to receive the alarm SMS. Users can get SMS report of SIM/UIM card status(SIM Remain Time) in .

【Alarm Threshold（via SMS）】when the SIM remain time is or less than this value, will send the alarm SMS to remind the users of the SIM remain time.

【Port Description for Alarm】it is the identification mark of SIM/UIM card in the designated SMS report. The mobile phone No. of the SIM/UIM card is recommended to use as the port description for alarm, or any other string.

【SIM Remain Time】indicates the current sim remain time. It can't modified.

【Restore time】recovers the SIM remain time to initial value, the Maximum Call Duration.

【CLIR】caller ID display restrict. This function is used to restrict the mobile phone No. By adding "#31" before the mobile phone ID, this funciton should be supported by carrier.

【Mobile Tx Gain】transits gain of the mobile module, from IP side to PSTN side.

【Mobile Rx Gain】receives gain of the mobile module,from PSTN side to IP side.

【Detect Reverse Polarity】this option for CMDA Reverse Polarity detection. Most CDMA operators don't offer polarity reverse . So VoIP to mobile, 2008 will connect soon. It doesn't wait mobile side answer.

4.6.3 SIM/UIM card lock

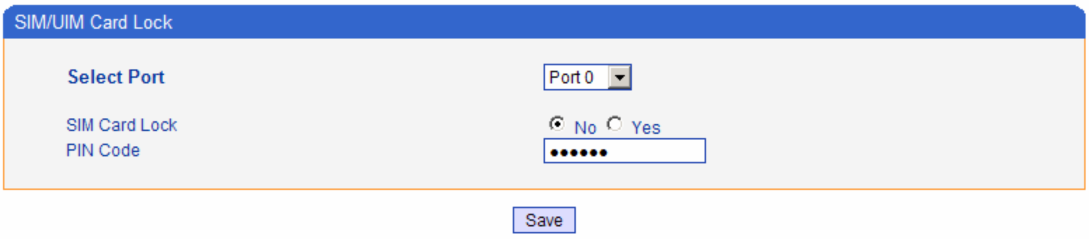


Figure 4-3-17 Configuration of SIM/UIM Card Lock

Configurations are as below:

【Select Port】select the Channel No. which need to be locked.

【SIM Card Lock】SIM card lock or unlock. Default is "No".

【PIN Code】correct PIN code is needed to lock or unlock the SIM card.

4.6.4 PIN Management

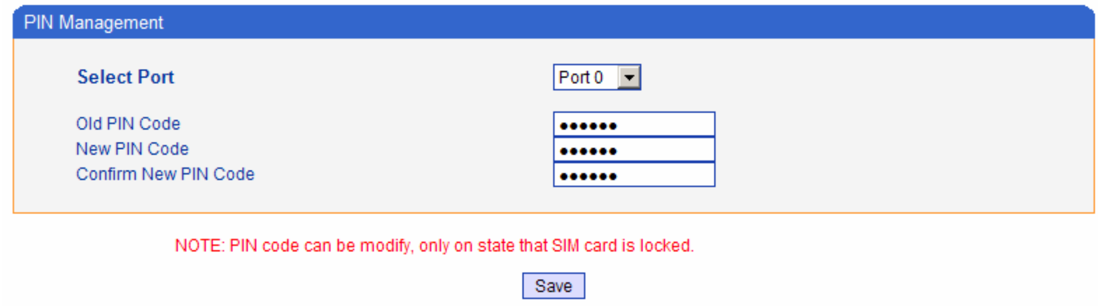


Figure 4-3-18 PIN Management

Detailed description as below:

【PIN management】PIN is the password of SIM card personal identification. In the status of SIM card locked, PIN can be modified to prevent SIM card from being stolen.

【Select Port】selects the GSM/CDMA channel No.

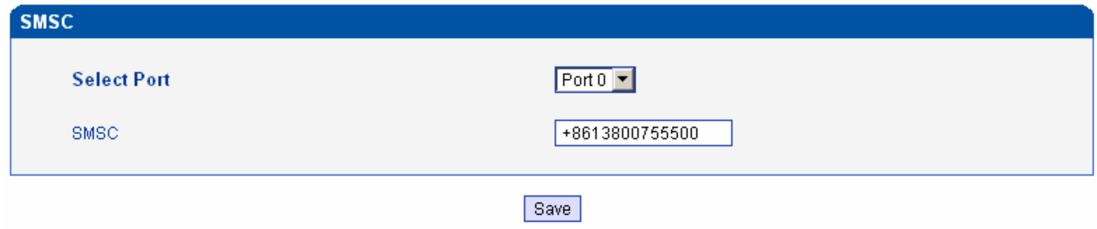
【Old PIN code】the previous PIN code

【New PIN code】inputs a new PIN code

【Confirm New PIN Code】inputs the new PIN code again.

4.6.5 SMSC

SMS center of mobile, theoretically, the celluar modular will automatically detect the SMSC number. This configurable option is used in a situation that the SMSC number could not detected by celluar modular. When such case happens, please contact with mobile service provider to identify the SMSC number and then add SMSC number in SMSC configurable web interface.



4.6.6 SMS

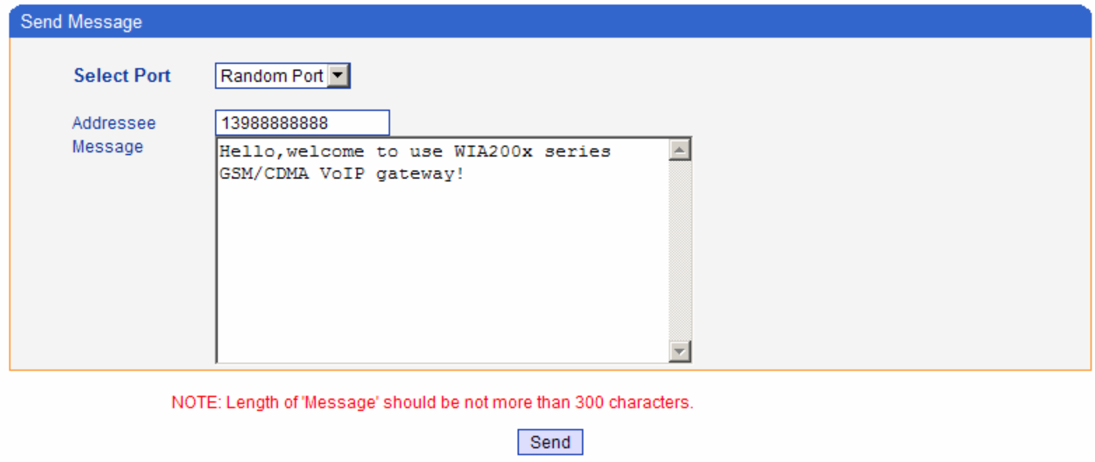


Figure 4-3-19 SMS sending

Configurations are as below:

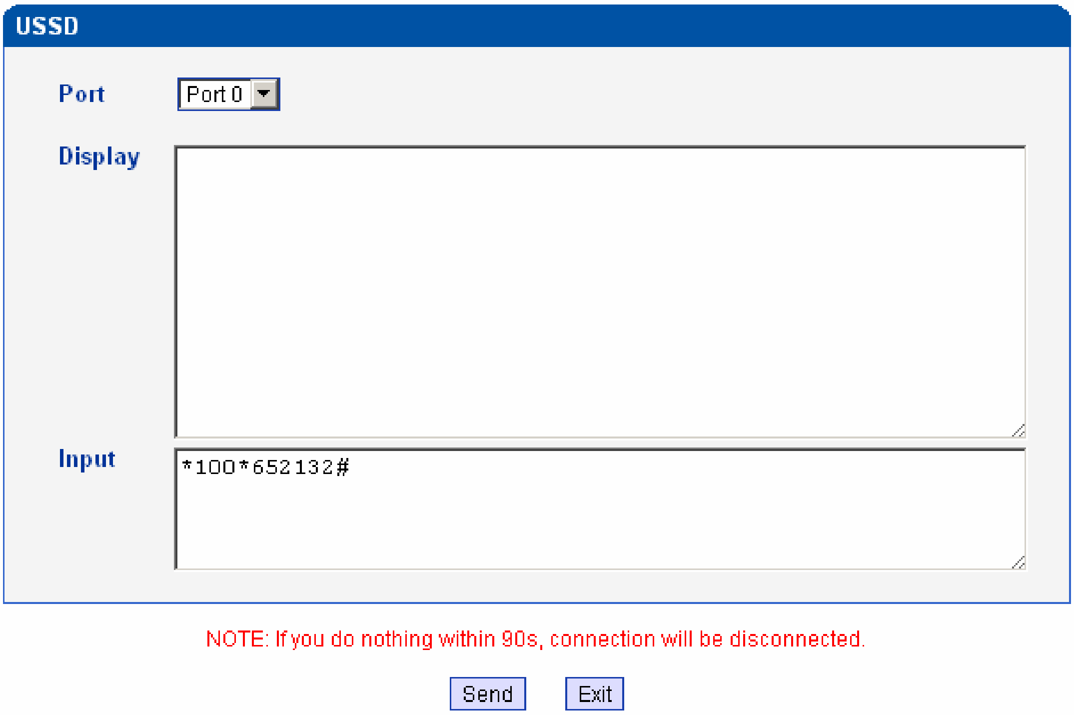
【Select Port】users can select a defined channel or random channel to send SMS. Input the recevier's mobile phone No to send SMS.

【Addressee】mobile phone No. of the receiver

【Message】content of the SMS. The length is limited to 300 characters.

4.6.7 USSD

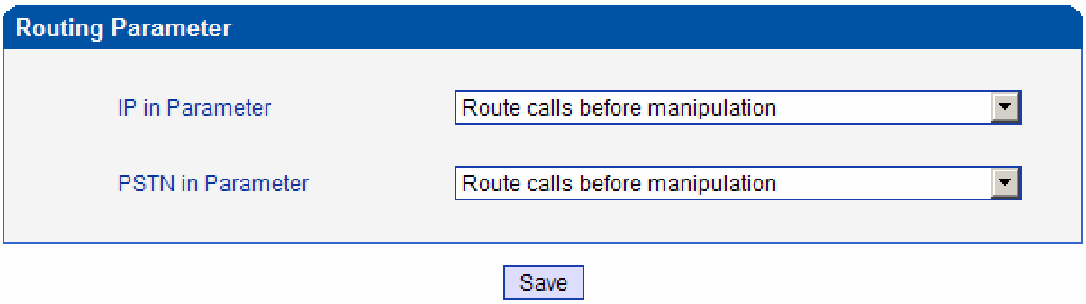
USSD (Unstructured Supplementary Service Data) is a Global System for Mobile(GSM) communication technology that is used to send text between a mobile phone and an application program in the network. Applications may include prepaid roaming or mobile chatting. USSD is similar with Short Messaging Service (SMS), but unlike SMS. USSD transactions occur during the session only. With SMS, messages can be sent to a mobile phone and stored for several days if the phone is not activated or within range.



Notes:please check with local service provider before using this service.

4.7 Routing configuration

4.7.1 Routing parameter



Globle parameters, it will take effect while number manipulation configured

Route calls after manipulation: the parameters indicate that the gateway will select IP->Tel or Tel->IP routes after number manipulation completed.

Route calls before manipulation: the parameters indicate that the gateway will select IP->Tel or Tel->IP routes before number manipulation completed.

4.7.2 IP->Tel Routing

IP->Tel Routing

	Index	Description	Source IP	Source Prefix	Destination Prefix	Destination
☐	0	to CMCC	IP Group 18	any	13[4-9]xxxxxxxx|6xxx|1...	Port Group 31
☐	1	to CMCC	IP 10	any	15[0189]xxxxxxxx|188x...	Port Group 31
☐	2	all	Any	any	any	Port Group 31

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

Add Delete Modify

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

Function:

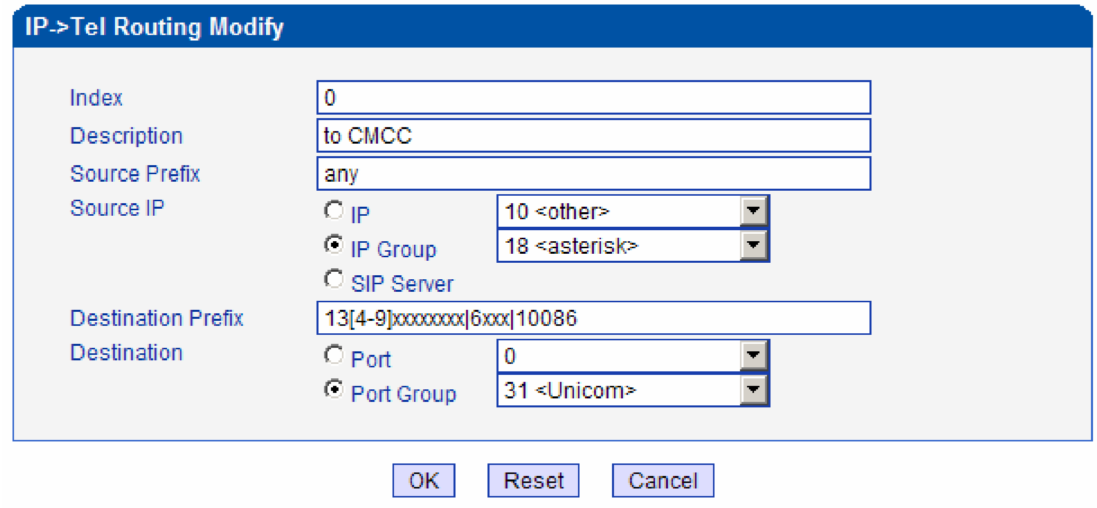
When the softswitch/ sip server send the call traffic to gateway directly, must be added IP->Tel routing.

Parameter Description

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

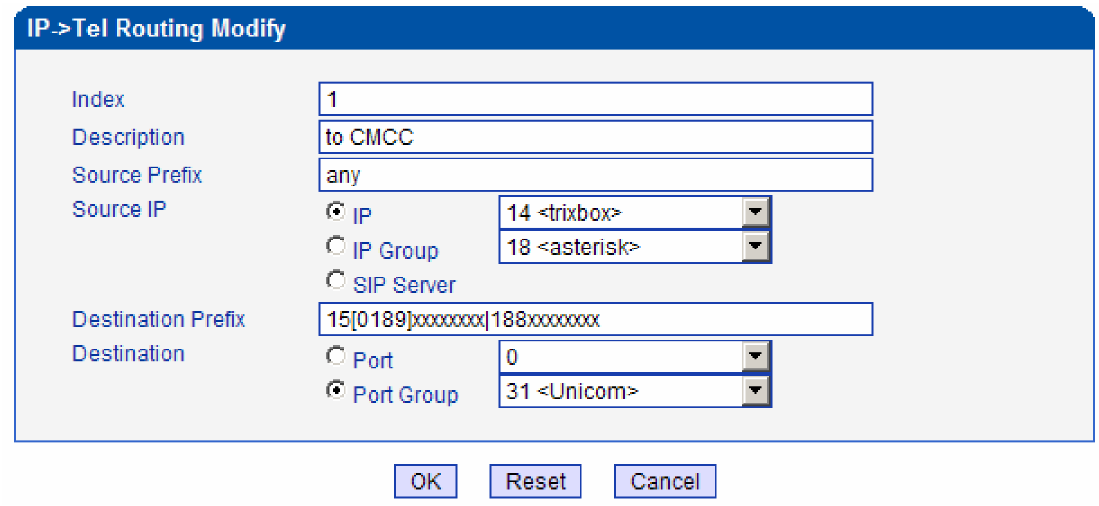
Example:

Index 0 : add a route from VoIP to GSM, the calls coming from IP Group 18<asterisk> will call out thourgh destination Port Group 31<Unicom>.



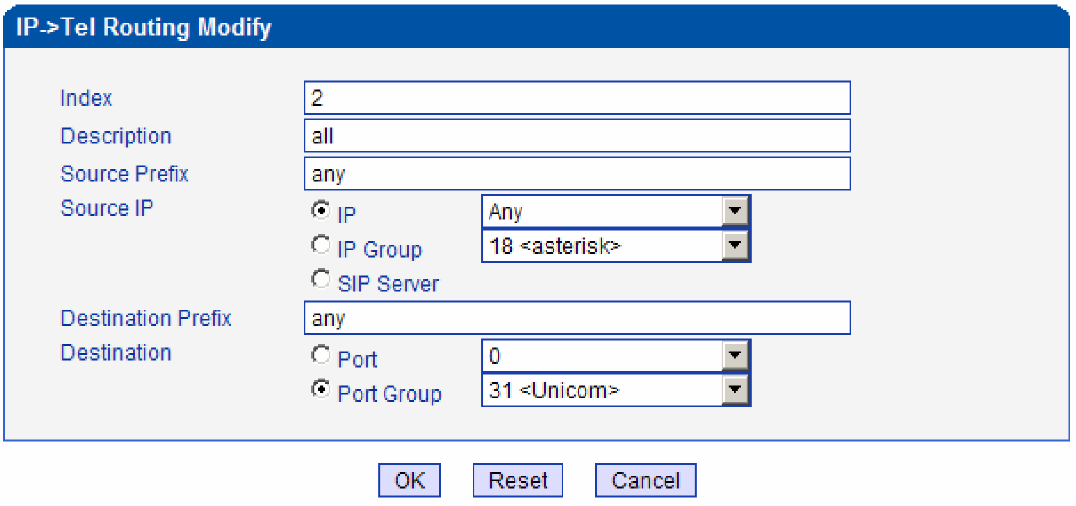
The caller party coming from IP Group 18 will match source prefix "any", "any" is a wildcard string include anonymous calls. At same time, destination prefix must match with " 13[4-9]xxxxxxxx | 6xxx | 10086".

Index 1: add a route from VoIP to GSM, the calls coming from IP 14<tribox> will call out through destination Port Group 31<Unicom>



The caller party coming from IP 14 will match source prefix "any", "any" is a wildcard string which will match prefix anonymous calls. At the same time, destination prefix must match with " 15[0189]xxxxxxxx | 188xxxxxxxx".

Index 2: add a VoIP to GSM route which allowed all the calls. It will receive all calls coming from any IP to destination Port Group 31



Prefix "any" is a wildcard string which will match anonymous calls as well.

4.7.3 Tel->IP Routing


Tel->IP Routing

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

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

Add Delete Modify

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

Function:

This item uses to configure incoming call routes which can be used for recieve the calls from the GSM.

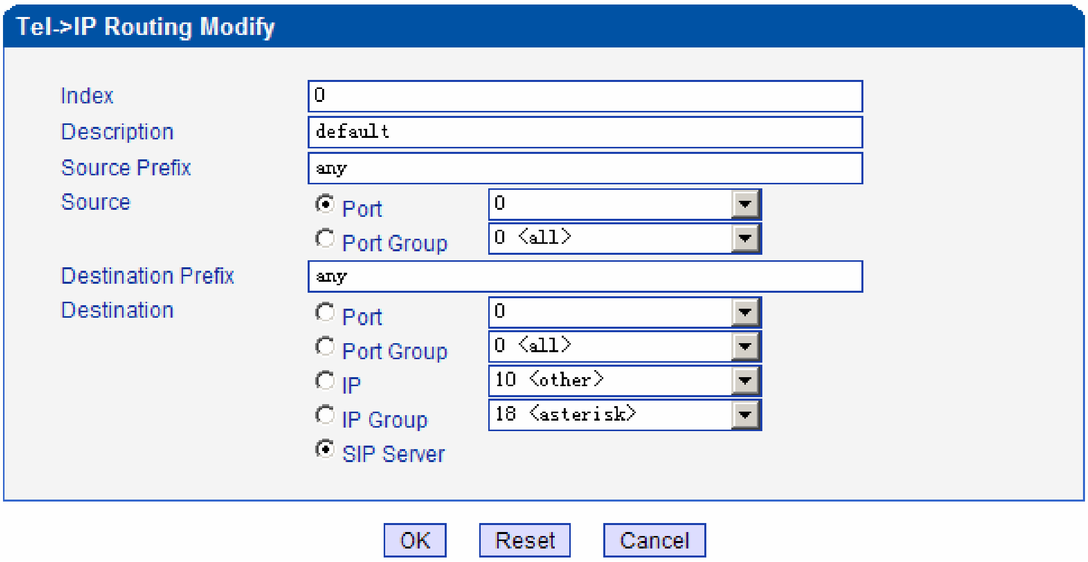
Parameter Description

Parameter	**Parameter Description**
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.
Description	It describes the route for the ease of identification. Its value is character string
Source Port	It specifies the Port or Port Group which will receive the calls from PLMN
Source Prefix	All the caller number must match the source prefix. It specifies the source prefix allow to send call out • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186

Destination Prefix	All the called number must match the destination prefix, the call prefix indicates the connected number • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186
Destination	Its specifies destination Port or Port Group

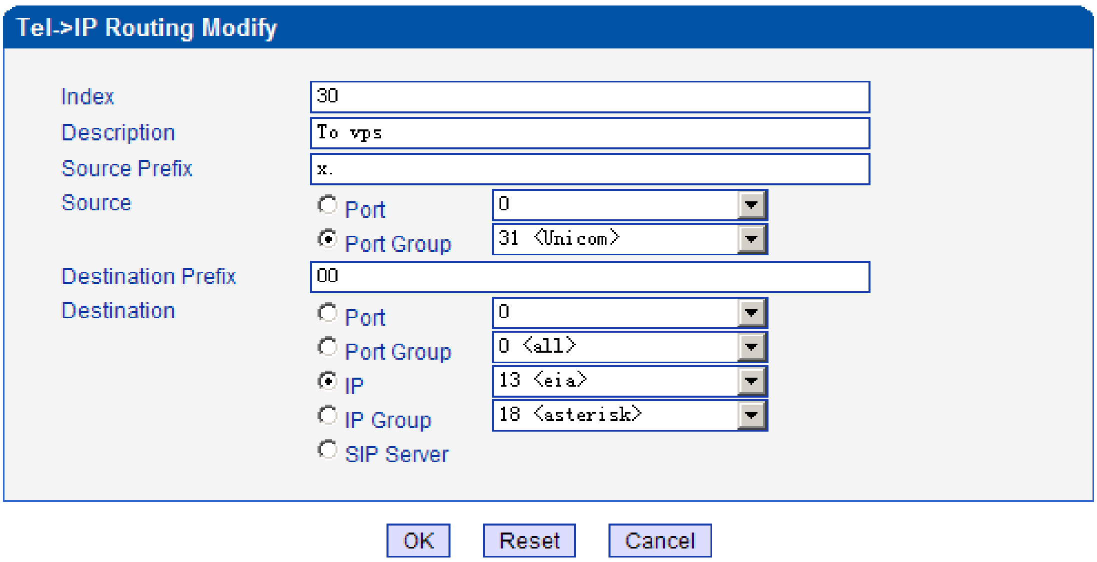
Example:

Index 0: its a default route configured in gateway



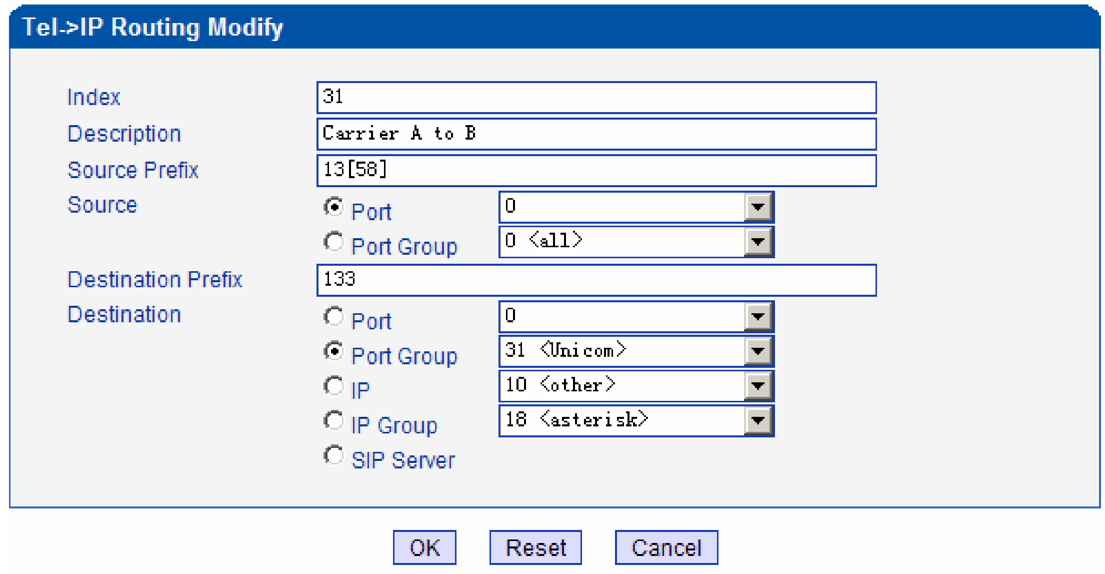
It allows any number from source port 0 send call to SIP server with any prefix.

Index 30: add a GSM to VoIP route



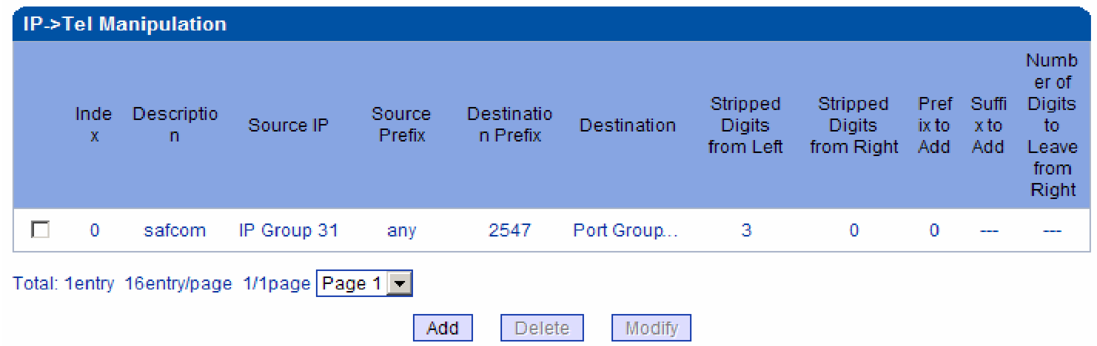
It indicates that the calls coming from Port Group 31<Unicom> will match the prefix "x.", "x." is a wildcard string which will match any prefix except "anonymous" calls. Meanwhile sending the calls destination IP 13<eia> if called number match with destination prefix "00".

Index 31: 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


IP->Tel Manipulation

	Index	Description	Source IP	Source Prefix	Destination Prefix	Destination	Stripped Digits from Left	Stripped Digits from Right	Prefix to Add	Suffix to Add	Number of Digits to Leave from Right
☐	0	safcom	IP Group 31	any	2547	Port Group...	3	0	0	---	---

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

Add Delete Modify

Function

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

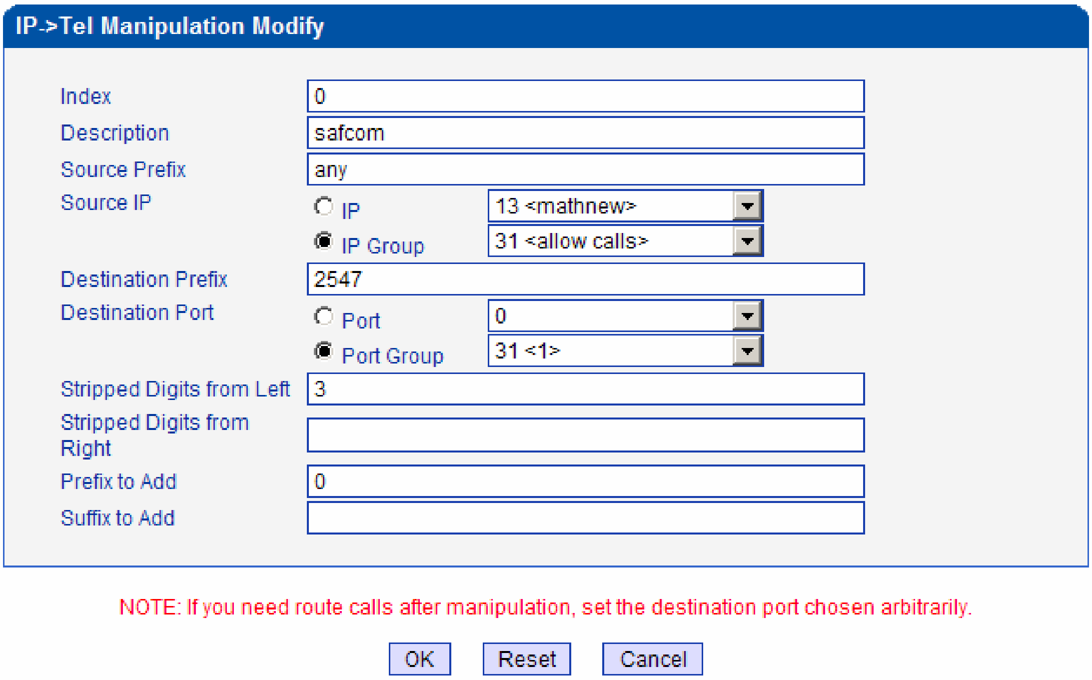
The IP->Tel Manipulation defined the rules of add, and deletion of called numbers, which are referenced by IP->Tel routing.

Parameter Description

Parameter	**Parameter Description**
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.
Description	It describes the route for the ease of identification. Its value is character string
Source IP	It specifies the source IP which will send the calls to gateway • Any: any IP address • IP: specific an IP address • IP Group: specific an IP group
Source Prefix	All the caller number must match the source prefix. It specifies the source prefix allow to send call out • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186
Destination Prefix	All the called number must match the destination prefix, the call prefix indicates the connected number • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186
Destination	Its specifies destination Port or Port Group
Stripped Digits from Left	It specifies the length of the digits to be deleted from left
Stripped Digits from Right	It specifies the length of the digits to be deleted from right
Prefix to Add	Add the new digits in front of the original number
Suffix to Add	Add the new digits at the end of the original number

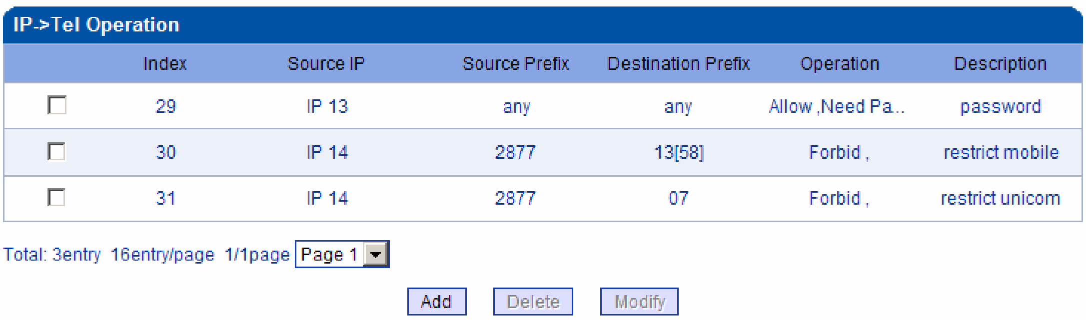
Example

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



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

4.9 Operation



Function

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.

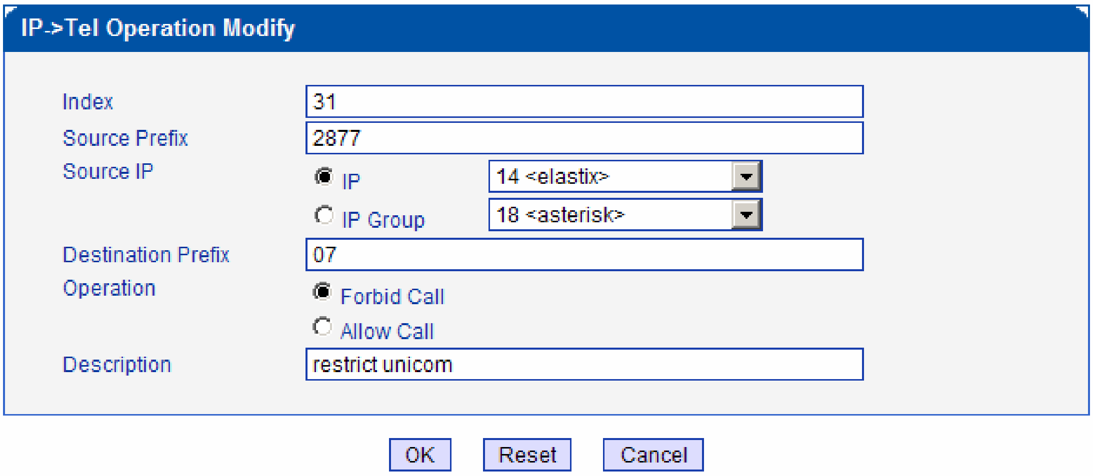
Parameter Description

Parameter	Parameter Description
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

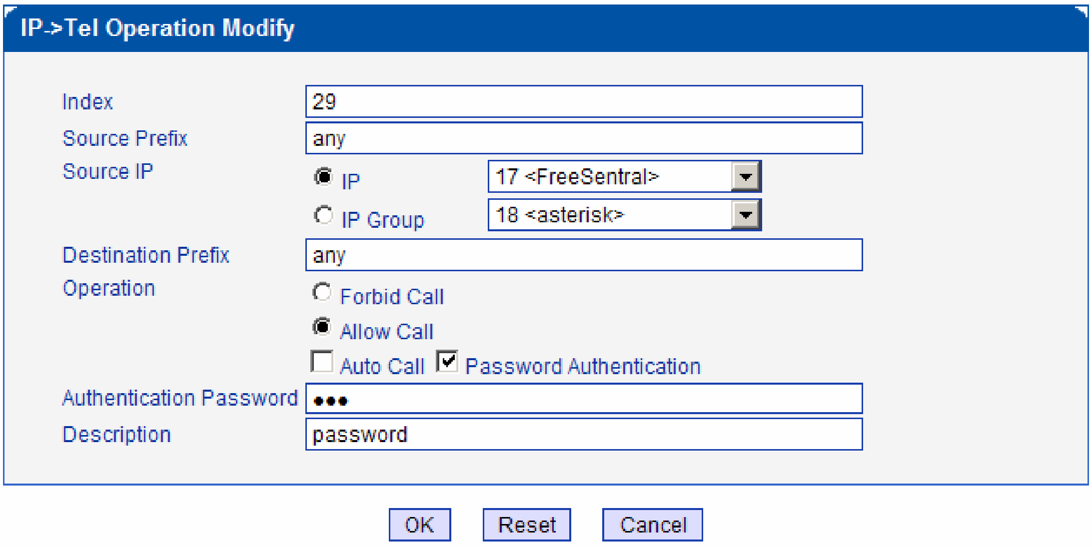
Example

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



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.



4.3.8 Port Group Configuration

Function

This configuration step is necessary, and is used to add port group in gateway. The port group will referenced by IP->Tel routing and number manipulation.

Parameter Description

Parameter	Parameter Description
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
Port	It specifies the Port will add to port group
Select Mode	It specifies the policy for selecting port in a port group • Ascending: the system always selects a port from the minimum number. The preferential selection of the port can be realized through this mode • Cyclic ascending: when system selects ports' Priority, it always begin from the number next to the number selected last time, if the maximum priority number is selected last time, then the next number is the minimum priority number, and move in cycles like this • Descending: when system selects ports' priority, it always begin to

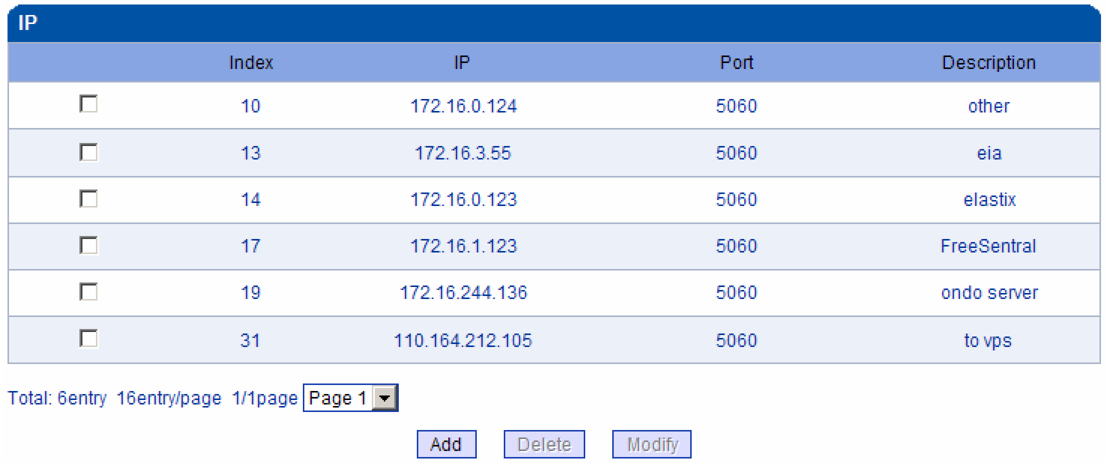
	select from the maximum priority number • Cyclic descending: when system selects ports' Priority, it always begin from the number before to the number selected last time, if the minimum priority number is selected last time, then the next number is the maximum priority number, and move in cycles like this

Example

To add a port group, assume that port 6, port 7 belong to the same carrier, add port 6 and port 7 to port group 30, select mode is cyclic ascending.

4.3.9 IP Group Configuration

1) IP configuration



IP

	Index	IP	Port	Description
☐	10	172.16.0.124	5060	other
☐	13	172.16.3.55	5060	eia
☐	14	172.16.0.123	5060	elastix
☐	17	172.16.1.123	5060	FreeSentral
☐	19	172.16.244.136	5060	ondo server
☐	31	110.164.212.105	5060	to vps

Total: 6entry 16entry/page 1/1page Page 1

Add Delete Modify

Function

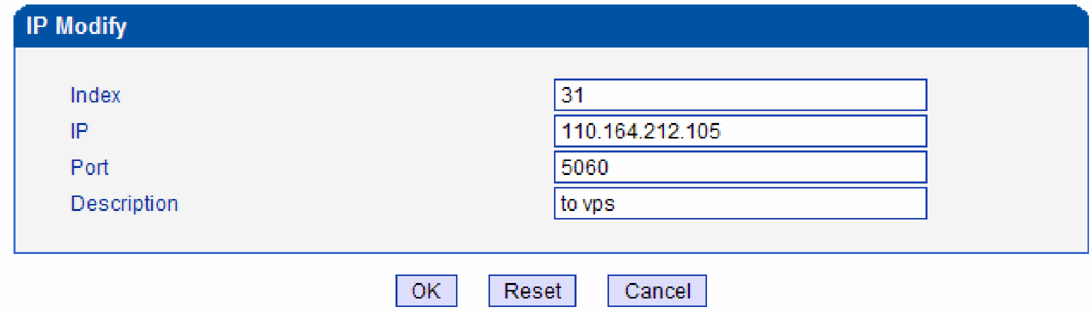
Add remote IP of Softswitch, SIP server which will send call traffics to gateway.

Parameter Description

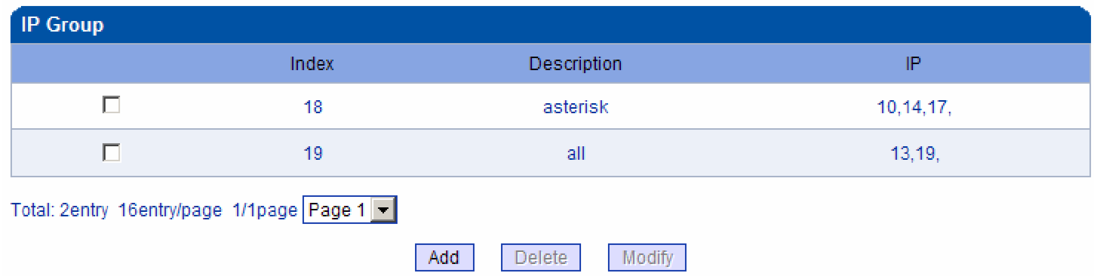
Parameter	Parameter Description
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.
Description	It describes the route for the ease of identification. Its value is character string
IP	It is an interworking parameter between the remote Softswitch and the SIP server. It specifies the IP address of the peer equipment.
Port	It is an interworking parameter between the remote Softswitch and the SIP server. It specifies the SIP port number of the peer equipment

Example

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



2) IP Group



Function

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.

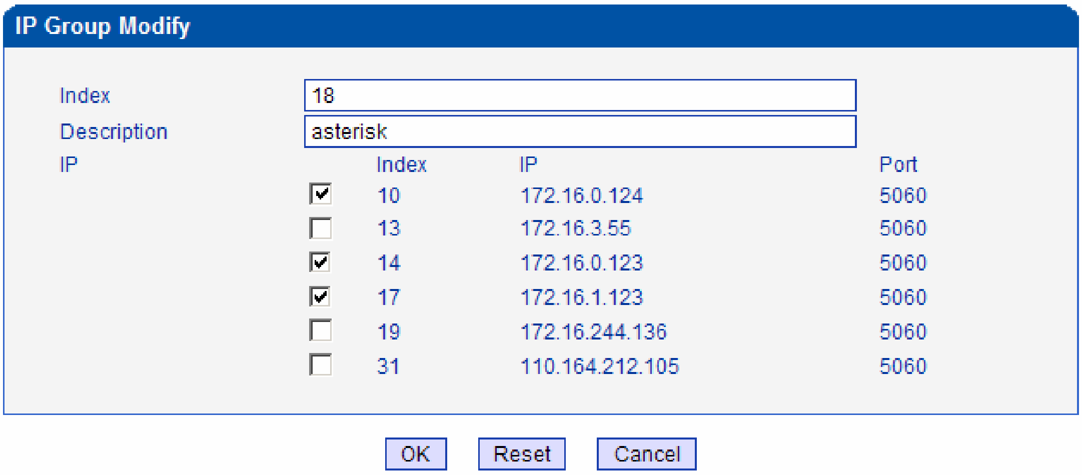
Parameter description

Parameter	Parameter Description
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



1) System configuration

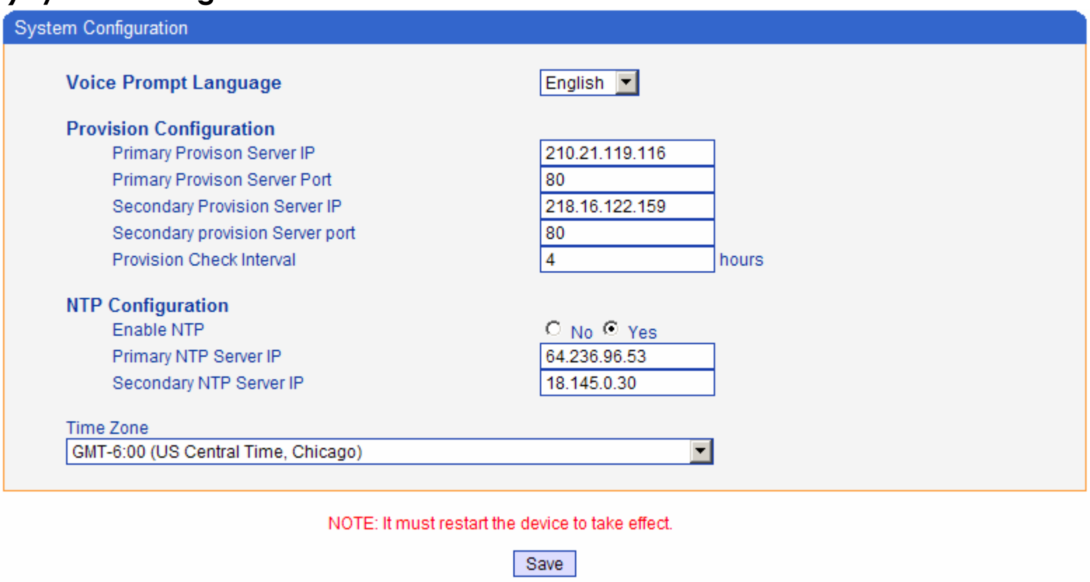


Figure 4-3-20 System Configuration

Configuration details as below:

【Voice Prompt Language】configure the voice prompt of ., e.g. configure voice prompt of IP address success or failure. supports English and Chinese. Users can customize other languages. The default setting is in Chinese.

- Provision configuration

Provision is used to maintain the devices. E.g. Provision can config, update and remote manage the devices in bulk.

【Primary Provision Server IP】this is provided by carrier. Keep the default value if carrier don't provide this value.

【Primary Provision Server Port】this is provided by carrier. Default is 80.

【Secondary Provision Server IP】this is provided by carrier. Keep the default value if carrier don't provide this value.

【Secondary Provision Server Port】this is provided by carrier. Default is 80.

【Provision Check Interval】default is 4 hours.

- NTP Configuration

【Enable NTP】NTP enable switch

【Primary NTP Server IP】can keep the default

【Secondary Provision NTP Server IP】can keep the default

【Time Zone】the default is GMT +8:00, the user can adjusted accordingly according to their area

2) Service configuration (for SIP only)

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

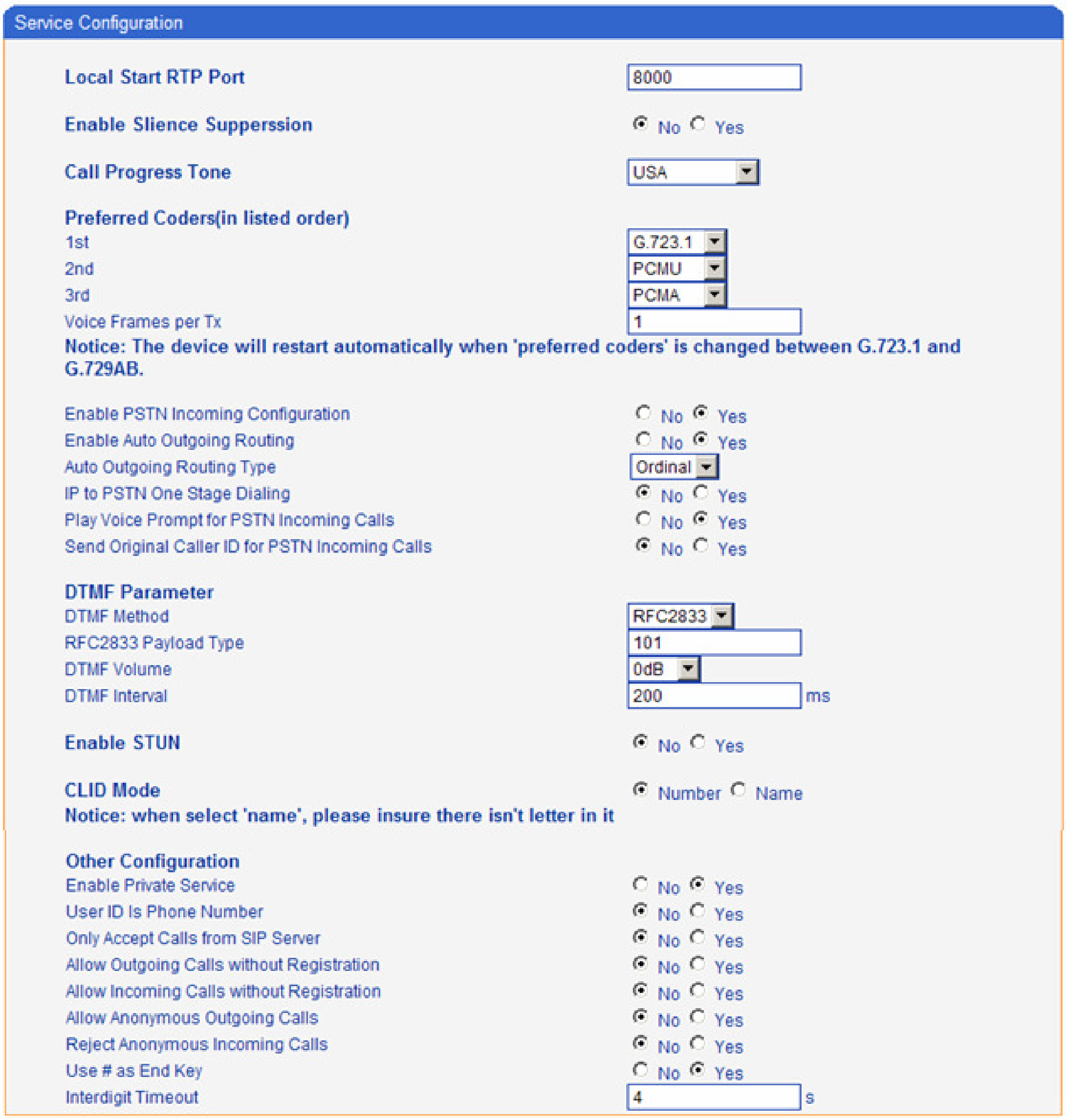


Figure 4-3-21 Service Configuration

Detailed description as follow:

【LOCAL RTP PORT Channel】means the initial allocation of Channel when RTP voice stream transmit in the IP network , in general, using the factory default values. When there are multiple DINSTAR series voice products, and the network gateway or router's NAT with loopholes, user can try changing this item.

【Enable Silence Suppression】enable the "silence suppression" almost no impact on call quality, and can save about half of the bandwidth.

【Call Progress Tone】each country has its different call progress tone required standards, such as busy tone, ring back tones and ring tone standards, users can select the area standard from here .

【Preferred Coders】means the code format when Voice transfer on IP network, support PCMA, PCMU, G.723.1 andG.729AB.

Note: when the preferred codec switch between G.723.1 and G.729AB , System will automatically reset

【Enable PSTN Incoming Configuration】 means when call from PSTN side, you can dial the function keys for check number, setting IP and so on function

【Enable Auto Outgoing Routing 】means when call out , whether by ordinal or polling pick to Select a Channel, this feature are generally used for when use the same SIP User ID to register or use as trunking mode

【IP to PSTN One Stage Dialing 】 this function will be displayed only when select "Enable Auto Outgoing Routing" function, the User ID will be sent directly to PSTN, for example: the user calls 6715, the device will sent 6715 User ID to PSTN

【Play Voice Prompt for PSTN Incoming Calls 】setting is yes, when through the PSTN calls to the Channel, the device will with the clew tone, the default is "Please dial the extension User ID"; setting to No, the device will with dial tone

【Send Original Caller ID for PSTN Incoming】for Example, the phone A from PSTN side call 2001/2004/2008 Channel SIM card corresponding User ID, the Channel's SIP User ID is C, Channel hook and then call B, when "Send Original Caller ID for PSTN Incoming" setting is Yes, the Caller User ID that send to B will be A, when "Send Original Caller ID for PSTN Incoming" setting is No, the caller User ID that sent to B will be C(except for anonymous outgoing)

【DTMF】2001/2004/2008 support RFC2833 and SIGNAL two ways. DTMF INTERVAL range is 50 ~ 800ms, DTMF VOLUME can use the default Configuration

【Enable STUN】(Simple Traversal of UDP over NATs，NAT's UDP simple cross) is a network protocol. It is allowed to stay behind the NAT (or multiple NAT) client part to identify their clients' public address, found himself after what Type of NAT and NAT for a particular Channel is bound to a local Internet terminal Channel. This information is used for two host to set up UDP communication behind the same NAT router. The agreement defined by the RFC 3489

【CLID Mode】Select the name under the special needs , the most common way is use the default User ID

【Other Configuration】:

Enable"*"at the beginning private internal service: the default is enable, that the private internal network-related service refer to Configuration query, recording, adjust the volume, restore the factory default values, etc. ;

SIP request message whether with a user = phone: is a SIP protocol requirement, if the soft switch no require, setting is no;

Only Accept Calls from SIP Server: is whether only accept the call from the SIP PROXY Server , if the setting is Yes, the call through point to point IP ADDRESS will be rejected (not launched from SIP server's address) ;

"Allow Outgoing Calls without Registration"and "Allow Incoming Calls without Registration"need use together , which refer to "SIP Configuration" - "Is register" option, if"Is register" setting is no, this two options need set Yes ,to avoid that the devices can not call out ;

"Allow Anonymous Outgoing Calls" setting to No, the device caller User ID is a registered User ID, setting to Yes, the device Caller User ID as "Anonymous" ;

"Allow Anonymous Inbound" setting to No, the incoming anonymous calls will be rejected;

"Use # as End Key",in General, SIP phones are based on # as the end, if this option is set to No, the dial-up will end expires dial-up time;

"Inter digit Timeout"Bit of between the dialing time ,over the time will be seem as end of dial-up;

【NOTE】close* Business functions , not mean the related business can not be used, users can also through web Configuration the corresponding function

3) SIP Configuration (for SIP only)

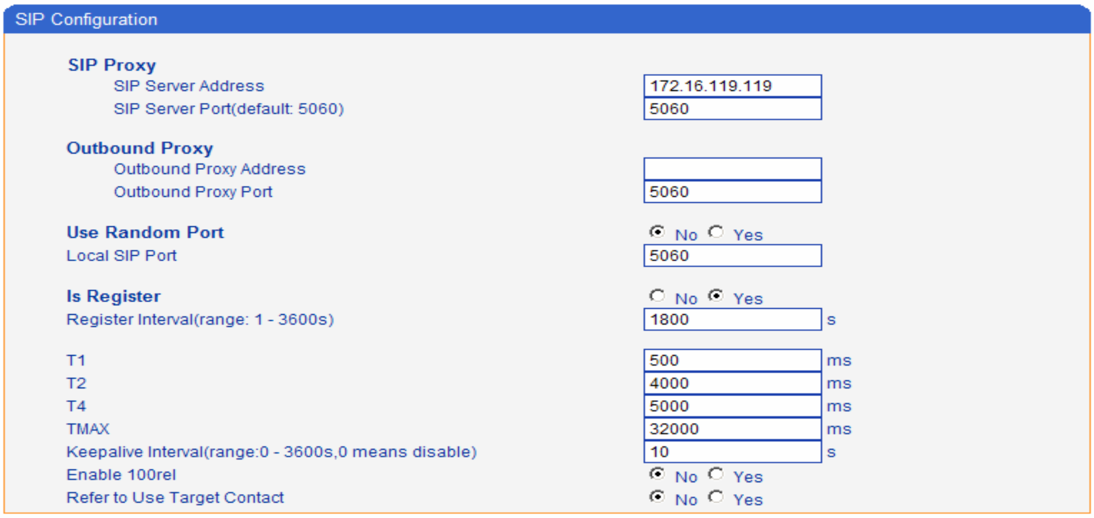


Figure 4-3-22 SIP Configuration

SIP Configuration: used for Configuring VoIP channel, add SIP Registry Platform and local SIP Channel, and configure SIP protocol and other related information

Detailed description as follow:

- SIP PROXY

【SIP Server Address】use for configure SIP server address and Channel, the address can be IP Address, also can be a domain name (DNS should to be able to resolution), the details please advisory the service provider

【SIP Proxy Port】port default setting is 5060. For details, please consult the service provider

- Outbound Proxy

【Outbound Proxy Address】Outbound proxy, it mainly used in firewall / NAT environment. That make the signaling and media streams are able to penetrate the firewall, the details please advisory the service provider

【Outbound Proxy Port】outbound proxy port number, the details please advisory the service provider

【Use Random Port】set the local monitor SIP port(fixed or random), random is every time you start the device will random Select a free SIP port Monitor

【Is Register】 default set yes, if you want the device can make a call without register, set No, Also enable the "Allow Outgoing Calls without Registration" and

"Allow Incoming Calls without Registration" function

【Register Interval】means how often the equipment will register once to the SIP server/proxy

【T1】 used to define the SIP protocol T1 timer value, default is 500ms;

【T2】 used to defines the SIP protocol timer values, default value is 4000ms;

【T3】used to define the T2 timer value in SIP protocol, the default is 5000ms;

【Keep alive Interval】used for keep communicate between equipment and the SIP server that make the device is in the best available Registered. In general, using the factory default values

Note: all parameters refering to defaultt setting of SIP protocol, user can modify that if necessary

4) Port configuration

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

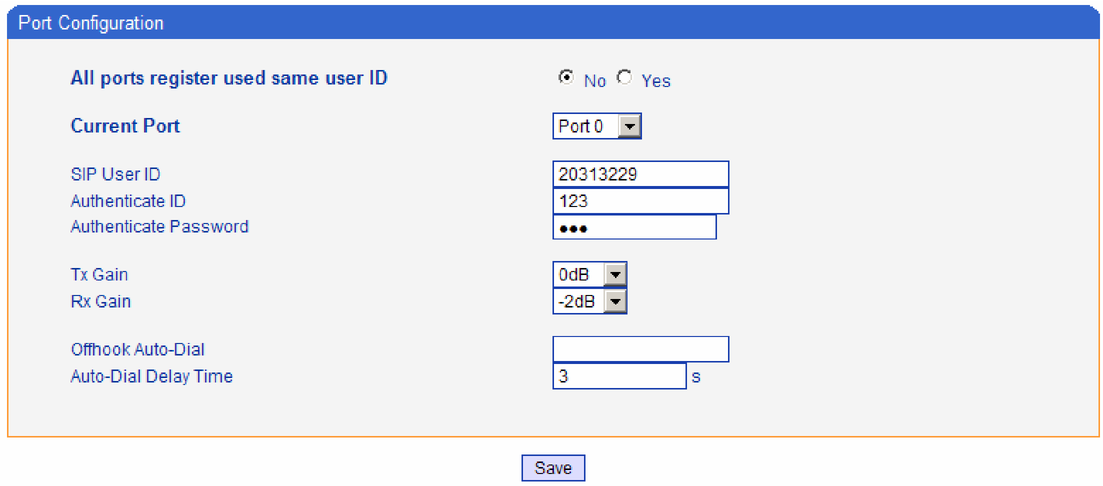


Figure 4-3-22 Port Configuration

Interface describe as following:

【ALL ports register used same user ID】 The default is not, each SIP account, if set yes ,all the port will use user ID of port0, when the call ring in sequence

【SIP User ID】is the account used for registration, equipment port's unique identifier, "Authenticate ID" is equivalent to show the name, "Password" is register Password, which no passward can no fill, the details please contact the service provider

【Tx Gain】refers to the call volume that from himself during a call to the end users, adjust the "Tx Gain" will affect the voice volume of the end user, the default value is 0

【Rx Gain】 refer to the call volume from the remote end user to ourself volume, adjust the "gain acceptance" will affect the voice volume we will heard, the default value is 0.

【Offhook Auto-Dial】 Hotline service.when PSTN part client calls to this port,will auto forward to the hotline User ID. If no need this feature, just left it blank

【Auto-Dial Delay Time】 Offhook Auto-Dial delay time, the range is 0-10 seconds

5) Fax Configuration

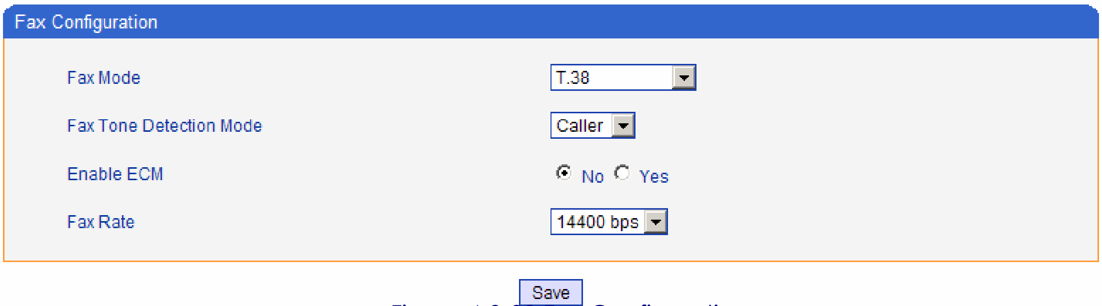


Figure 4-3-23 Fax Configuration

Fax Configuration use for configure fax, and support pass-Through and T.38 fax mode, the maximum baud rate supported 14400bps

Interface describe as following:

【Fax way】 support T.38 and pass-Through；

【Fax Tone Detection Mode】 There are two calling and called

【Enable ECM】 Error Correction Mode. If enabled, when transmiss the fax ,will automatically detects the wrong part of the requested retransmission to eliminate data errors, but will increase the fax time

【Fax Rate】 support fax rate list include：2400bps、4800 bps、7200 bps、9600 bps、12000 bps、14400 bps

4.3.6 Tools

1) Firmware upload

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

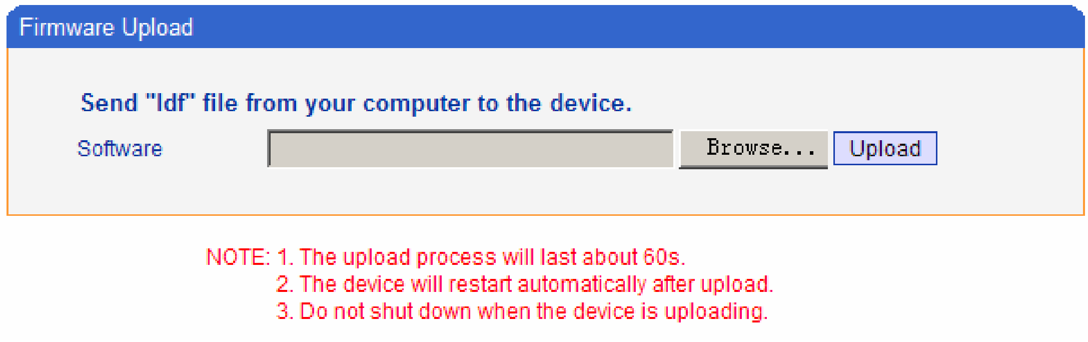


Figure 4-3-24 firmware upload

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

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

2) IVR Voice Prompt Upload

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

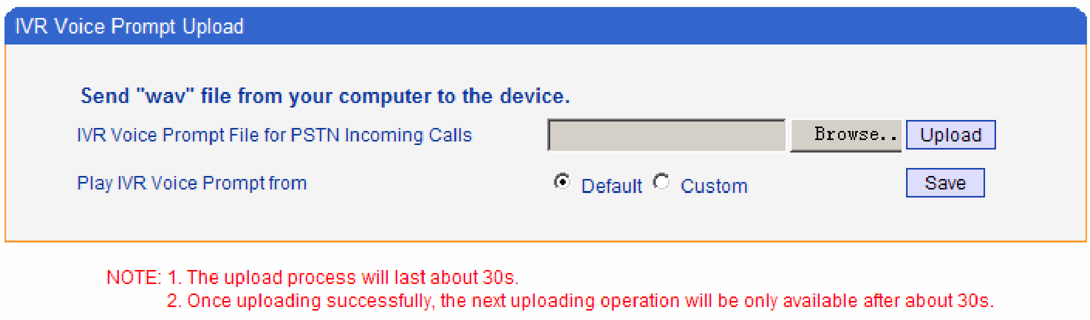


Figure 4-3-25 IVR Voice Prompt Upload

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

3) Login password

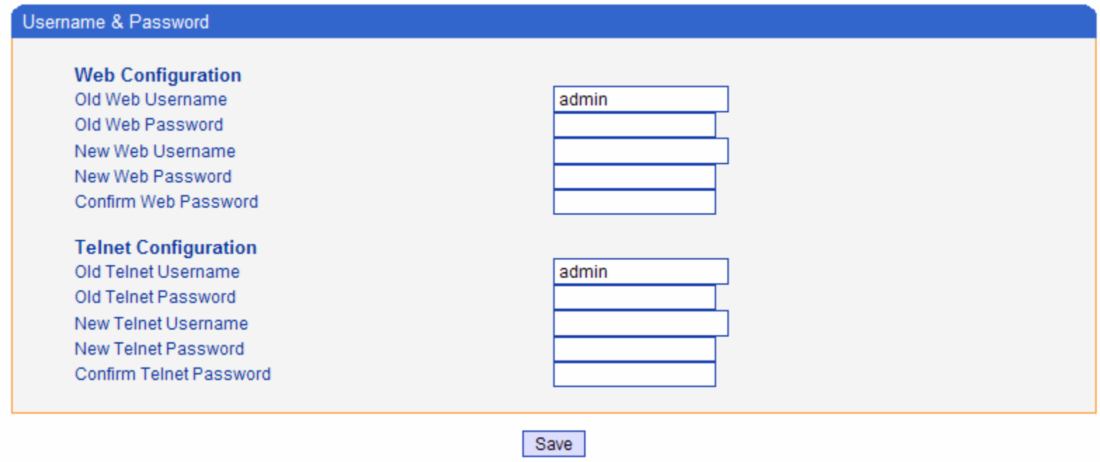


Figure 4-3-25 IVR Voice Prompt Upload

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

4) Factory Reset

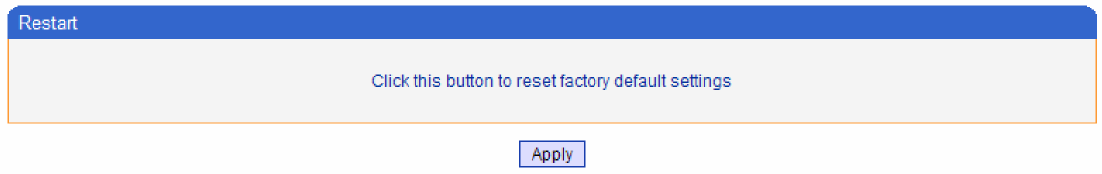


Figure 4-3-26 Factory Reset

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

6) Restart

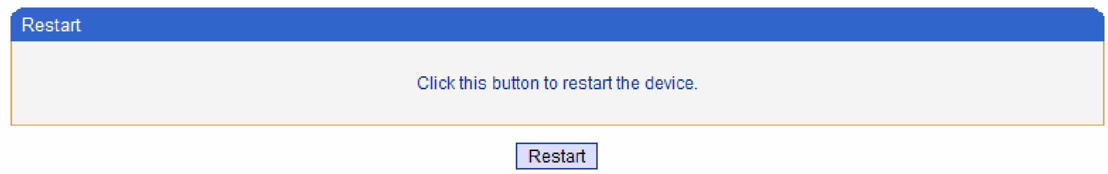


Figure 4-3-27 Restart

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

Chapter 5 FAQ

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

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

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

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

4) Check whether there is a LAN device conflict with the exists IP ADDRESS;

- Equipment cannot register

If the Run LED does not flash mean unregistered

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

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

3) Ask network administrators to open limitation with the equipment's network communications (it is a special equipment, not afraid of virus attacks); (2) try to enable the equipment tunnel (through the WEB for Configuration, Also, please NOTE, open the tunnel will impact voice quality, Please do not enable the tunnel as far as possible, reference WEB Configuration Interface Description section);

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

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

- 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 to identify the problem.

- when calling in, the caller always hears a busy tone

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

- sudden interruption during a call

1) make sure whether is human error caused the problem

2) Make sure with the account balance or lack of disruption caused the call disconnected

3) Make sure whether there is interference with the fax tone or equipment busy tone, these interference may lead to calls dropped

4) Make sure whether the LAN equipment such as gateway or router fails, user can try to restart the gateway or router

- 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

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