UC2000-VE/F/G GSM/CDMA/WCDMA VoIP Gateway

User Manual

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Revision Records

Document Version	Firmware Version	Author	Date	Description
V1.0	02.22/23.08.01	Technical Support	2013-07	First release
V1.1	02.22/23.10.01	Technical Support	2014-05	Support WCDMA Gateway
V2.2	02/04. 23.12.05	Technical Support	2017-02	Match with the firmware version 02/04.23.12.01 or above

Table of Contents

1 Product Description

This chapter mainly introduces functions and structures of UC2000-VE/F/G.

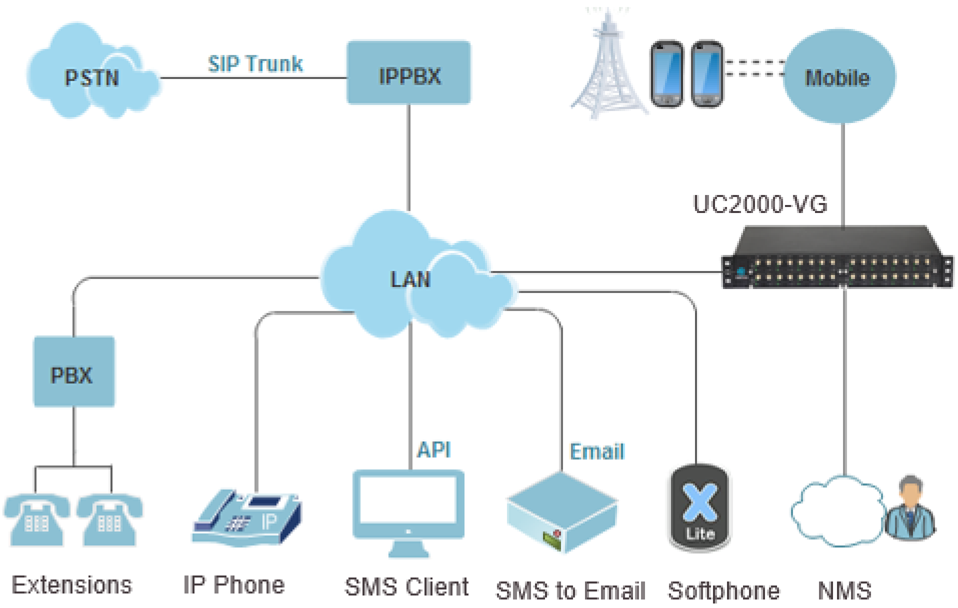
1.1 Overview

UC2000-VE/F/G serials GSM/CDMA/WCDMA VoIP Gateway is full functions VoIP gateway based on IP and Mobile network, which provides a flexible network configuration, powerful features, and good voice quality. It works for carrier grade, enterprise, SOHO, residential users for cost-effective solution.

1.2 Scenario of Application

With the development of users and telecom service, mobile network and fixed network integration will be steadily increasing. UC2000-VE/F/G provides high quality VoIP service which perfectly meets the requirement. This is a scenario shown as figure 1-2-1

Figure 1-2-1 Network scenario



1.3 Product Appearance

1.3.1 Product Appearance of UC2000-VE

The appearance of UC2000-VE shows as follow

Figure 1-3-1 Front view of UC2000-VE-8G/8C

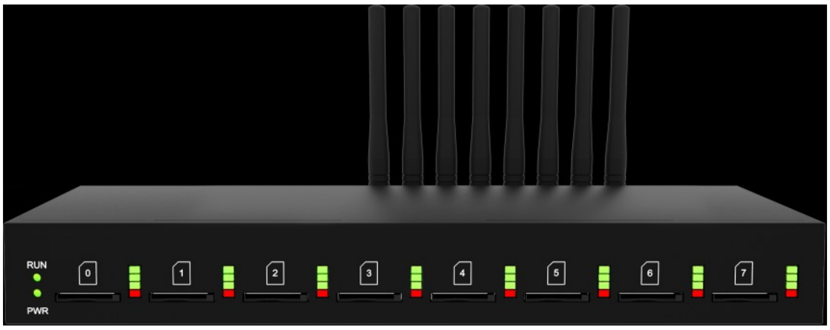


Table 1-3-1 Description of Front view

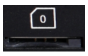
Index	Indicators	Description
1	RUN	On: Starting Off: Abnormal Blinking every 0.5s: Normal status
2	PWR	On: Power on Off: Power off
3	Signal	 Signal strength indicators with green color
4	Channel	 Use/Unuse indicator with Red color, ON is used, Off is unused
5	SIM Slots	 SIM card slot

Figure 1-3-2 Rear view of UC2000-VE-8G/8C

Table 1-3-2 Description of Rear view

Index	Interface	Description
1	Power Connector	 Power connector of DC power. Input: DC12V
2	Antenna Connector	Mark as digits 0 to 7
3	Network	FE0 and FE1, its default IP address **192.168.11.1**
4	Console	RS232 standard, **band rate 115200bps**
5	RST	Reset button to restore default IP and password or restore factory setting. ◆ Restore IP and Password: **hold RST button 3~5 seconds, RUN LED being ON during this time** ◆ Restore factory setting**: Hold RST button 7 seconds, RUN LED being blink**

1.3.2 Product Appearance of UC2000-VF

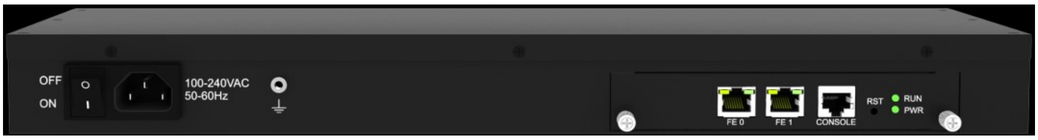
Front View

Indicators and connectors

Indicators	Name	Status	Description

	SIM Card Status Indicator	OFF	Indicates SIM is offline, SIM status may include SIM card not inserted, SIM card not available, SIM card unregistered
		ON	SIM card is in use
		Blinking	SIM card is registered but in IDLE
	Antenna Connector	-	Antenna connect, mark with 0-15
	SIM Card Slot	-	SIM card slot, mark with 0-15

Back view



Indicators and connectors

Indicators	Name	Status	Description
	Power switch	-	Power on or power off the device
	Power connect	-	AC Input 110-240V
FE0-FE1	Network	-	Default IP is 192.168.11.1
	Console	-	RS232 standard, **band rate 115200bps**
RST	RST	-	Reset button to restore default IP and password or restore factory setting. ◆ Restore IP and Password: **hold RST button 3~5 seconds, RUN LED being ON during this time** Restore factory setting: **Hold RST button 7 seconds, RUN LED being blink**
PWR	Power indicator	OFF	No power
		ON	Power on
RUN	System indicator	Blinking (0.5S)	Device is running normally

		ON	Device is booting up
		OFF	Device is not booting up

1.3.3 Product appearance of UC2000-VG

Front view

Indicators	Name	Status	Description
	SIM Card Status Indicator	OFF	Indicates SIM is offline, SIM status may include SIM card not inserted, SIM card not available, SIM card unregistered
		ON	SIM card is in use
		Blinking	SIM card is registered but in IDLE
	Antenna Connector	-	Antenna connect, mark with 0-15
	SIM Card Slot	-	SIM card slot, mark with 0-15

Back View

Indicators	Name	Status	Description
	Power switch	-	Power on or power off the device
	Power connect	-	AC Input 110-240V
FE0-FE1	Network	-	Default IP is 192.168.11.1

	Console	-	RS232 standard, **band rate 115200bps**
RST	RST	-	Reset button to restore default IP and password or restore factory setting. ◆ Restore IP and Password: **hold RST button 3~5 seconds, RUN LED being ON during this time** Restore factory setting**: Hold RST button 7 seconds, RUN LED being blink**
PWR	Power indicator	OFF	No power
		ON	Power on
RUN	System indicator	Blinking (0.5S)	Device is running normally
		ON	Device is booting up
		OFF	Device is not booting up

1.4 FUNCTIONS AND FEATURES

1.4.1 Protocols

- Standard SIP;
- Simple Traversal of UDP over NATs (STUN);
- Point-to-point protocol over Ethernet (PPPoE);
- Hypertext Transfer Protocol (HTTP);
- Dynamic Host Configuration Protocol (DHCP);
- Domain Name System (DNS);
- ITU-T G.711α-Law/μ-Law、G.723.1、G.729AB;
- PPTP(support on 8 channels gateway)

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 Check
- Lock/unlock SIM/UIM
- Mobile number display rejection
- Sending/receiving SMS
- Customize IVR Recording
- White and black list
- One number access
- Open API for SMS, support USSD
- Echo Cancellation (with ITU-T G.168/165 standard)
- Automatic negotiate network
- Hotline
- BCCH(Support on GSM Gateway only)

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

  UC2000-VE:

  Input: 100-240V, 50-60Hz

  Output: DC12V 4.0A

  UC2000-VF/G:

  Input: 100-240VAC, 50-60Hz;

- Temperature(Operation): 0 ℃ ~ 45 ℃

  (Storage): -20 ℃ ~80 ℃

- Operation Humidity: 10%-90% No Condensation
- Dimension(W/D/H): 250*156*32.5mm
- Weight: 1.069kg
- Package Weight: 2.05kg

2 Quick Installation

2.1 ATTENTIONS BEFORE INSTALLATION

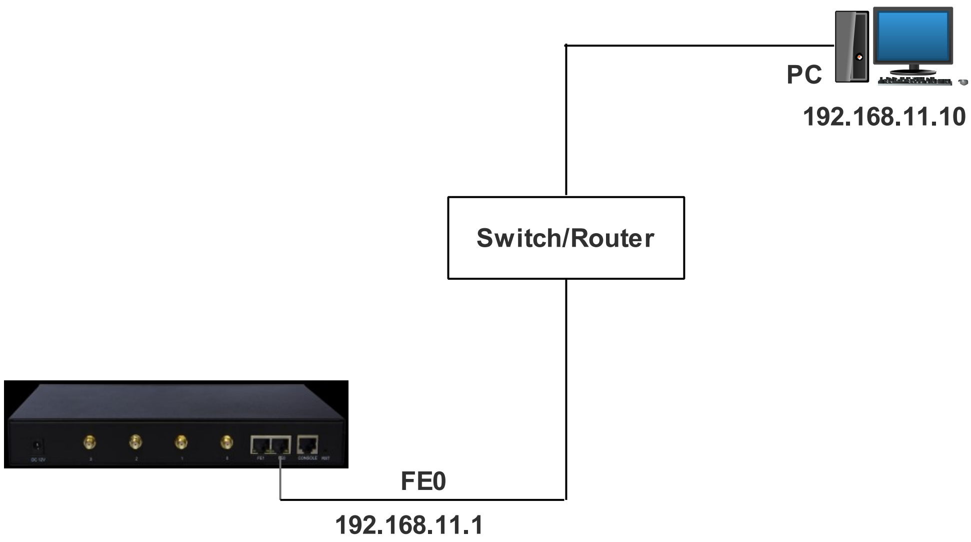
Please pay attention to the following before you install UC2000-VE/F/G include:

- DC power/AC power should be grounded well to ensure reliability and stability
- Network interface should be standard RJ45 with 10Mbps or 100Mbps interfaces
- GSM channels work properly and antennas should be well connected.

2.2 INSTALLATION PROCEDURES

- Connect antennas to the device;
- Connect the power wire to the device;
- Connect network cable to the device;
- Insert SIM cards to SIM slots.

2.3 NETWORK CONNECTION



Note: UC2000-VE/F/G has two Ethernet ports (namely FE0 and FE1). The device can work normally when either of the ports is connected to PC. The IP address of device must be at the same network segment with that of PC.

3 Basic Operation

3.1 Feature Codes

Users can do some basic system setting via dialing feature codes through a telephone.

The device has a built-in IVR navigator for local maintenance. In each step, if you hear an IVR message of “setting succeeds”, it means you have finished this step successfully. However, if you hear “setting fails”, please check and redo that step.

Code	Corresponding Function
150	Dial *150*1 to set the IP address of the gateway as static IP address; dial *150*2 to set the IP address as DHCP IP address
152	Dial *152*192*168*1*10# to set the IP address of the device as 192.168.1.10. (192.168.1.10 is just an example)
156	Dial *156*192*168*1*1# to set the default gateway of the network as 192.168.1.1. (192.168.1.1 is just an example)
153	Dial *153*255*255*0*0*# to set the netmask of the network as 255.255.0.0 (255.255.0.0 is just an example)
*158#	Dial *158 to inquiry IP address of the device
*111#	Dial *111# to restart the device

3.2 Basic Operation

3.2.1 Check IP address

Use a mobile phone to call a SIM card number of the device, then the device will answer and play a voice prompt of ‘dial the extension number’. Press *158# on mobile phone, then the device will report its local IP address automatically.

3.2.2 Restore factory setting via IVR

Use a mobile phone to call a SIM card number of the device, the device will answer and play a voice prompt of ‘dial the extension number’. Press *166*000000# on the mobile phone, then you will hear 'setting succeeds', then the factory setting of the gateway will be restored.

3.2.3 Restore default IP and password

Press RST button for about 3 seconds, then the device will be rebooted and the IP address, username and password will be restored to factory default.

3.2.4 Restore factory setting

Press RST button for about 7 seconds, then gateway will be rebooted and restored to factory setting.

3.3 LOCAL MAINTENANCE THROUGH CONSOLE PORT

To ensure easy maintenance, the device provides a standard RS232 console port, which has a Baud rate of 115200bps. Users can log in the device to carry out maintenance-related configurations through the console port.

- **Example: Log in device via Console Port**

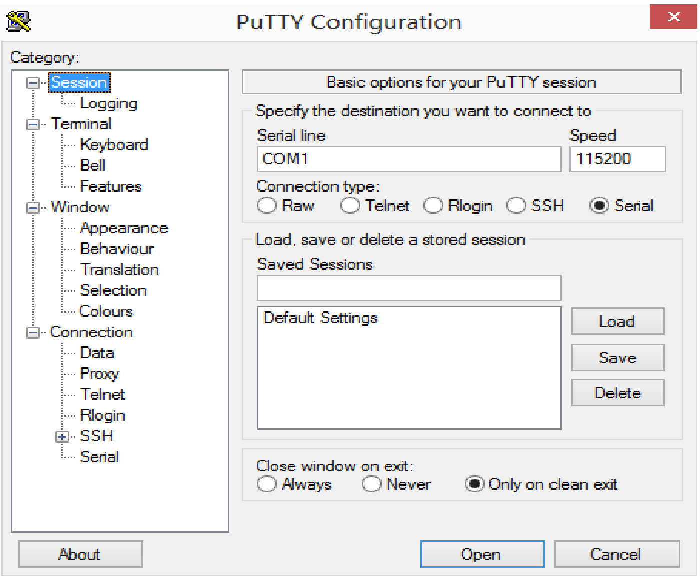
Step 1: Prepare a serial cable as follows (standard RS232, 115200bps);

Step 2: Connect the F port of the serial cable with the COM port of PC. If the PC does not have a COM port, please use a USB-to-COM converting tool to connect the serial cable with the PC.

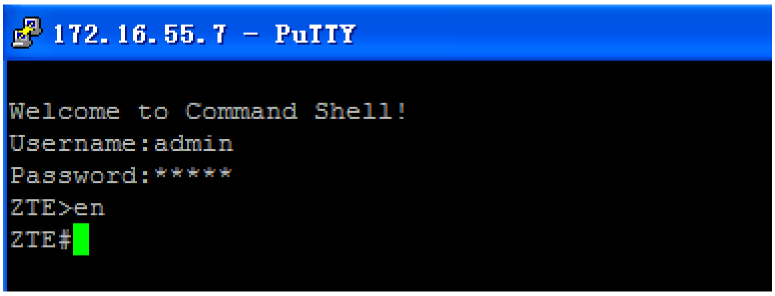
Step 3: Connect the M port of the serial cable with the console port of the device.

Step 4: Conduct configurations on login software.

Herein we take the PuTTY sofeware as an example. Detailed configurations are as follows:



After finishing the above configurations, click the **Open** button to enter the maintenance interface of the console port. The username and password are the same with those of the web interface of device.



Commands for configuring the IP address of the device :

(In the following example, IP address of device needs to be configured as 172.30.66.100, and netmask is 255.255.0.0)

> enable

enable# configure

config# interface ethernet

config-if-br-lan# ip address 172.30.66.100 255.255.0.0

config-if-br-lan# exit

config# ip default-gateway 172.30.0.1

Commands for inquiring the IP address of the device

> enable

enable#show interface

4 WEB Interface Configuration

UC2000-VF/G serials gateway has the same web interface. This chapter describes web configuration of UC2000-VE. The UC2000-VE contains an embedded web server to set parameters by using the HTTP protocol. We are strongly recommend to access device with Google Chrome or Firefox Browser.

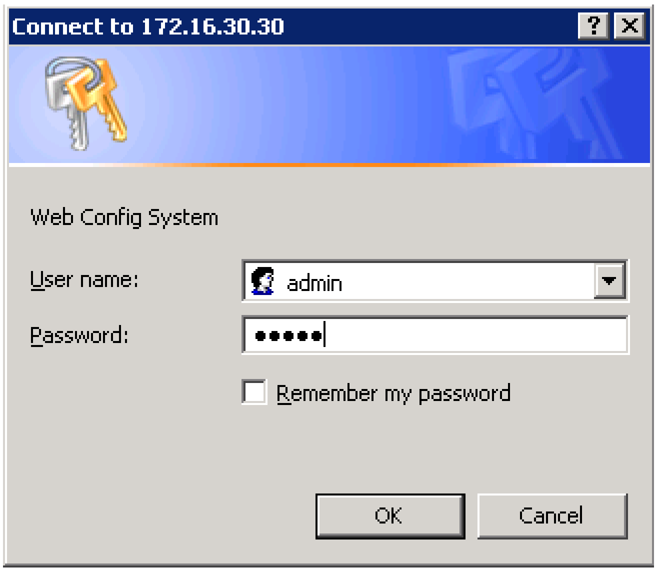
The configuration introduction also suitable for following models:

- UC2000-VE-4G
- UC2000-VE-8G
- UC2000-VF-16G
- UC2000-VF-8G
- UC2000-VF-32G
- UC2000-VE-8C (8 Channels CDMA Gateway)
- UC2000-VE-4C (4 Channels CDMA Gateway)
- UC2000-VF-16C (16 Channels CDMA Gateway)
- UC2000-VF-32C (32 Channels CDMA Gateway)
- UC2000-VE-8W (8 Channels WCDMA Gateway)
- UC2000-VF-16W(16 Channels WCDMA Gateway)
- UC2000-VF-32W(32 Channels WCDMA Gateway)

4.1 ACCESS UC2000-VE UNIT

Enter IP address of UC2000-VE in IE/Google Chrome. The default IP of LAN port is 192.168.11.1. and the GUI shows as below:

Figure 4-1-1 WEB log interface

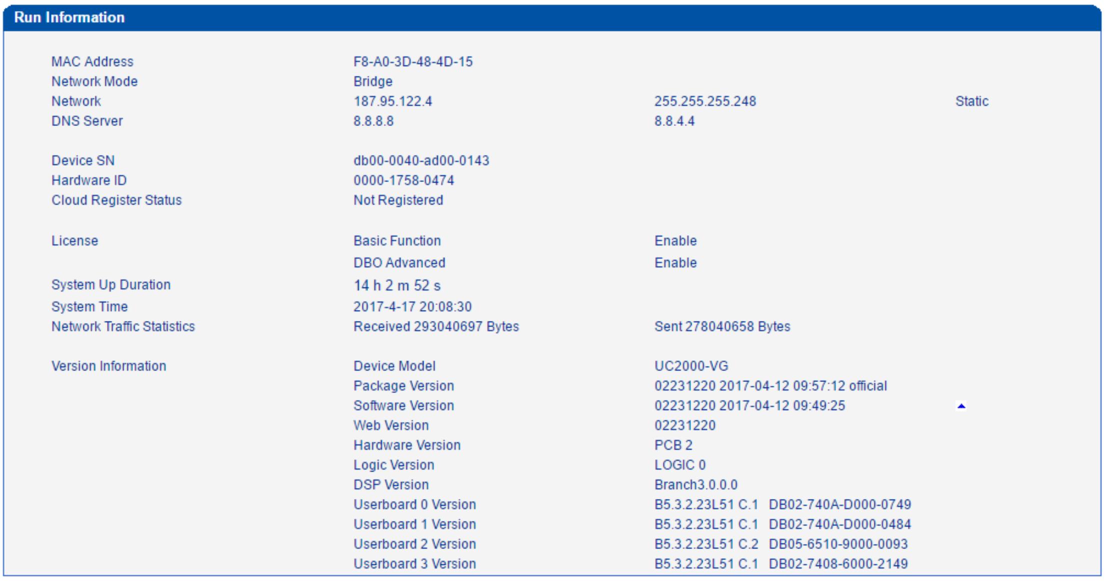


Enter username and password and then click “OK” in configuration interface. The default username and password are “admin/admin”. It is strongly recommended, change the default password to a new password for system security.

4.2 PARAMETERS CONFIGURATION

UC2000-VE WEB configuration interface consists of the navigation tree and the detail configuration interfaces.

Figure 4-2-1 WEB introduce



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

4.3 System Information

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

4.3.1 System Iinformation

Figure 4-3-1 system Information

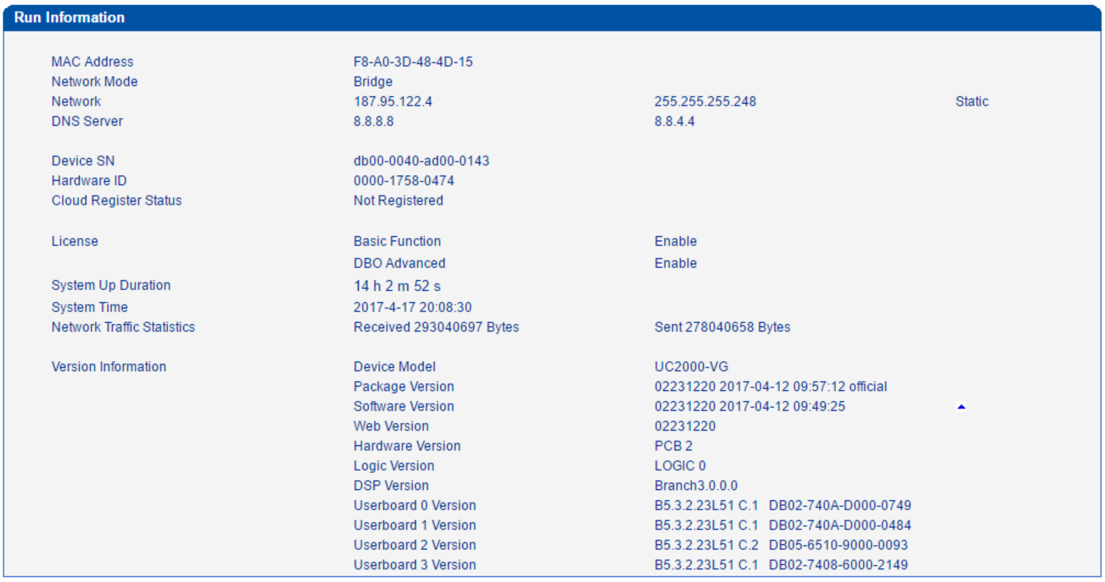


Table 4.3-1 System Information

Parameters	Description
MAC Address	Displays the current MAC of the gateway, for example: 00-1F-D6-1B-3D-02
Network Mode	UC2000-VE works as bridge mode by default
Network	Current IP address and subnet mask of gateway
DNS Server	Displays DNS server IP address in the same network with the gateway
Device ID	A unique device ID which assigned in factory. This device ID to be used as register ID with Dinstar SIM cloud.
Server Register	Its indicates communicate status with SIMCloud server, there are two type of

status	status: ▸ Registered ▸ Not Registered ▸ Need Authentication
License	It indicates device's license status. Contact with support when it display as **Invalid**
System Up Time	Shows the time period of the device running. For example,:1h: 20m, 24s
Traffic Statistics	Calculates the net flow, including the total bytes of message received and sent.
Version info	shows the current firmware version • Device Model: Model name of the device • Package version: 02231220 2016-12-16 17:44:40 official, 02231220 is the version number • Software version: 02231220 2016-12-16 17:36:05, 02231220 is the version number • Web version: the version number of web system. The web version must match with software • Userboard 0 Version: the firmware version of userboard slot 0 • Userboard License ID: Contact with support when it display as Invalid • Hardware version/DSP version/ SIMbox version

4.3.2 Mobile Information

Figure 4.3-2 Mobile Information

Mobile Information

Port	Type	IMSI	IMEI	Status	Remaining Call Duration	Carrier	Signal Quality	BER	ASR(%)	ACD(s)	PDD(s)	Call Status
0	GSM	460004122170725	863070010384344	Mobile Registered	No Limit	CHINA MOBILE		0	0	0	0	Idle
1	GSM	460020187266423	863070010384310	Mobile Registered	No Limit	CHINA MOBILE		0	0	0	0	Idle
2	GSM		863070010385747	No SIM Card	No Limit			0	0	0	0	Idle
3	GSM		863070010385549	No SIM Card	No Limit			0	0	0	0	Idle
4	GSM		863070011821336	No SIM Card	No Limit			0	0	0	0	Idle
5	GSM		863070011821880	No SIM Card	No Limit			0	0	0	0	Idle
6	GSM		863070011899175	No SIM Card	No Limit			0	0	0	0	Idle
7	GSM		863070011764668	No SIM Card	No Limit			0	0	0	0	Idle

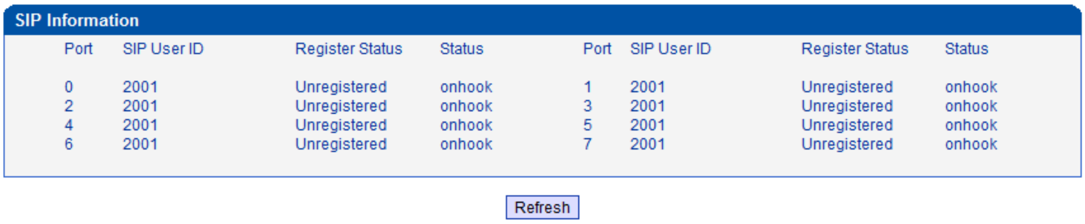
Table 4.3-2 Mobile Information

Parameters	Description
Port	Number of GSM/CDMA ports.
Type	Indicates the current type of network. Such as CDMA or GSM
IMSI	International Mobile Subscriber Identity, it is the uniquely identifies of SIM card
Status	Indicates the connection status of current GSM / CDMA module
Remaining Call Duration	It showing available total call minutes of SIM card while call limitation is enabled.
Carrier	Displays the network carrier of current SIM card.
Signal Quality	Displays the signal strength of in each channels of GSM / CDMA.
BER	Its indicate error rates between mainboard and userboard, Modular and Base station
ASR	Answer Seizure Ratio is a measure of network quality. It's calculated by taking the number of successfully answered calls and dividing by the total number of calls attempted. Since busy signals and other rejections by the called number count as call failures, the ASR value can vary depending on user behavior.
ACD	The Average Call Duration (ACD) is calculated by taking the sum of billable seconds (bill sec) of answered calls and dividing it by the number of these answered calls.
PDD	Post Dial Delay (PDD) is experienced by the originating customer as the time from the sending of the final dialed digit to the point at which they hear ring tone or other in-band information. Where the originating network is required to play an announcement before completing the call then this definition of PDD excludes the duration of such announcements.
Call Status	Show the Status of port, include idle, active, alert and processing *Idle* means there is no call on this channel *Processing* means call is connecting

	Alerting means destination is ringing *Active* means the call is connected *Ringing* means the gateway is answering incoming call from mobile *Calling Waiting* means the gateway is receiving another call during conversation and implement call waiting service *Call Hold* means the call is hold by extension of IPPBX/SIP Server

4.3.3 SIP Information

Figure 4-3-3 SIP Information



SIP Information

Port	SIP User ID	Register Status	Status	Port	SIP User ID	Register Status	Status
0	2001	Unregistered	onhook	1	2001	Unregistered	onhook
2	2001	Unregistered	onhook	3	2001	Unregistered	onhook
4	2001	Unregistered	onhook	5	2001	Unregistered	onhook
6	2001	Unregistered	onhook	7	2001	Unregistered	onhook

Refresh

Displays registration status information with Softswitch platform or SIP Server

Table 4-3-3 SIP information

Parameters	Description
Port	The number of SIP channels, UC2000-VE-8G/C has 8 SIP channels
SIP User ID	SIP registration account which are provided by the Softswitch and SIP server
Register Status	Shows the registration status of VoIP channel, including registered and unregistered.
Status	Show the status of port, Include "onhook" and "offhook"

4.4 STATISTICS

4.4.1 TCP/UDP

Figure 4-4-1 TCP/UDP Statistics

TCP/UDP			
TCP Send Packet	TCP Recv Packet	UDP Send Packet	UDP Recv Packet
1946619	686236	221687	0

Refresh

4.4.2 RTP

Figure 4-4-2 RTP

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

Refresh

Table 4-4-1 Description of RTP Statistics

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

4.4.3 SIP Call History

SIP Call History

SIP Call History

Port	Incoming Received	Incoming Connected	Incoming Answered	Incoming Failed	Outgoing Attempted	Outgoing Connected	Outgoing Answered	Outgoing Failed
0	55	55	55	0	48	0	23	25
1	28	28	28	0	2	0	0	2
2	0	0	0	0	0	0	0	0
3	0	0	0	0	0	0	0	0
4	0	0	0	0	0	0	0	0
5	0	0	0	0	0	0	0	0
6	0	0	0	0	0	0	0	0
7	0	0	0	0	0	0	0	0

Refresh

SIP Call History

Parameters	Description
Port	The port of Call statistics
Incoming Received	The amount of received incoming calls which coming from IP side
Incoming connected	The amount of incoming calls which have connected
Incoming Answered	The amount of incoming calls which answered by GSM/CDMA modular
Incoming Failed	The amount of incoming calls which failed
Outgoing Attempted	The amount of outgoing calls which attempted to IP side
Outgoing Connected	The amount of outgoing calls which have connected
Outgoing Answered	The amount of outgoing calls which answered by IP side
Outgoing Failed	The amount of outgoing calls which failed

4.4.4 IP to GSM Call History

IP to GSM Call History

IP to GSM Call History

Port	Call	Duration	Answered	Call Failed Caused by SIP				Call Failed Caused by GSM				OTHER
				Canceled	Timeout	Not Allowed	Negotiation failed	Busy	NO ANSWER	NO DIALTONE	NO CARRIER	
0	55	2179	16	25	0	0	0	0	0	2	12	0
1	28	1036	6	15	0	0	0	0	0	4	3	0
2	0	0	0	0	0	0	0	0	0	0	0	0
3	0	0	0	0	0	0	0	0	0	0	0	0
4	0	0	0	0	0	0	0	0	0	0	0	0
5	0	0	0	0	0	0	0	0	0	0	0	0
6	0	0	0	0	0	0	0	0	0	0	0	0
7	0	0	0	0	0	0	0	0	0	0	0	0

Refresh Clear

IP to GSM Call History

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

4.4.5 CDR Report

Figure 4-4-5 CDR Report

CDR Report

Enable CDR ○ No ◉ Yes   Save CDR ◉ No ○ Yes   save

Start Date : 2013 Year 6 Month 19 Day   Select Port All   Call Direction ALL

End Date : 2013 Year 6 Month 19 Day   Source   Destination

Min Duration s   Max Duration s   Rtp Loss Rate % to %

CDR Export Export   Refresh   Delete the CDRs in this Report

Port	Start Date	Answer Date	Call Direction	Source	Destination	Status	Duration(s)	Rtp Send	Rtp recv	Rtp loss Rate	jitter(s)
4	2013/06/19 15:14:49	2013/06/19 15:14:59	IP->Gsm	1955555123	01850594108	ANSWERED	39	764	2212	0%	0

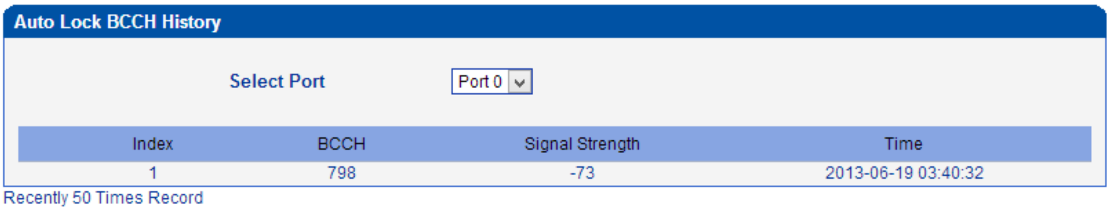
It is support 10000 CDRs on gateway. The CDRs will lost after reboot while save CDR set to No.

Port	Start Date	Answer Date	Call Direction	Source	Destination	Status	Duration(s)	Rtp Send	Rtp recv	Rtp loss Rate	jitter(s)
4	2013/06/19 15:14:49	2013/06/19 15:14:59	IP->Gsm	1955555123	01850594108	ANSWERED	39	764	2212	0%	0
4	2013/06/19 15:15:49		IP->Gsm	1955555123	01746039247	CANCELED	0	83	270	0%	0
2	2013/06/19 15:15:37		IP->Gsm	1955555123	01818910940	CANCELED	0	686	948	0%	0
8	2013/06/19 15:05:36	2013/06/19 15:05:48	IP->Gsm	1955555123	01710663894	ANSWERED	633	20067	31111	0%	0
0	2013/06/19 15:15:12	2013/06/19 15:15:33	IP->Gsm	1955555123	01840283671	ANSWERED	52	1174	3424	0%	0
8	2013/06/19 15:16:35		IP->Gsm	1955555123	019528783740	NO CARRIER	0	198	222	0%	0
8	2013/06/19 15:16:46		IP->Gsm	1955555123	019528783740	CANCELED	0	0	0	0%	0
2	2013/06/19 15:16:19		IP->Gsm	1955555123	01770924823	NOT ANSWERED	0	409	1225	0%	0

Parameters	Description
Port	GSM port number
Start Date/Answer Date	start and end time of calls
Direction	IP to GSM: outbound calls from softswitch/IPPBX to mobile network GSM to IP: incoming calls from mobile network to IPPBX/ Softswitch
Source	Calling number
Destination	Called number
Stauts	Answered: the call was established successful Canceded: the call was canceled by calling party No Carrier: the call was rejected by mobile network Not Answered: no body to answer the call Busy: user busy
Durations	Call duration of the call
RTP send/recv/loss rate	RTP Statistics of the call

4.4.6 Lock BCCH History

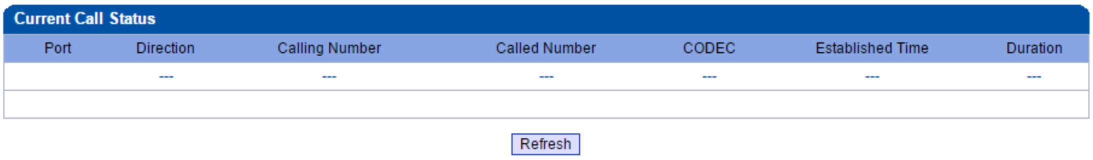
Figure 4-4-6 Auto Lock BCCH History



It is record history of BCCH to help analysis SIM card register status.

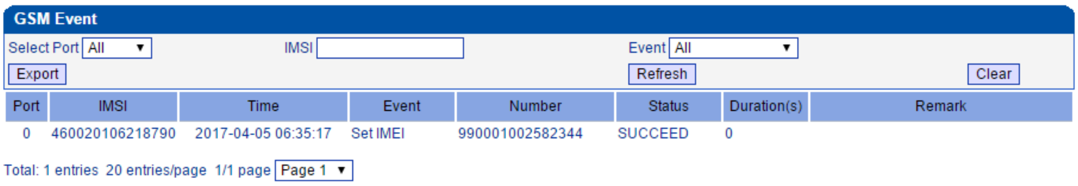
4.4.7 Current call status

On the **Statistics → Current Call Status** interface, status and detail of the current call are shown.



4.4.8 GSM Event

GSM event page will record all the logs of GSM modules such as IMEI change, replace new SIM card to specific port etc.



4.5 Network Configuration

4.5.1 Local Network

Figure 4-5-1 Local Network

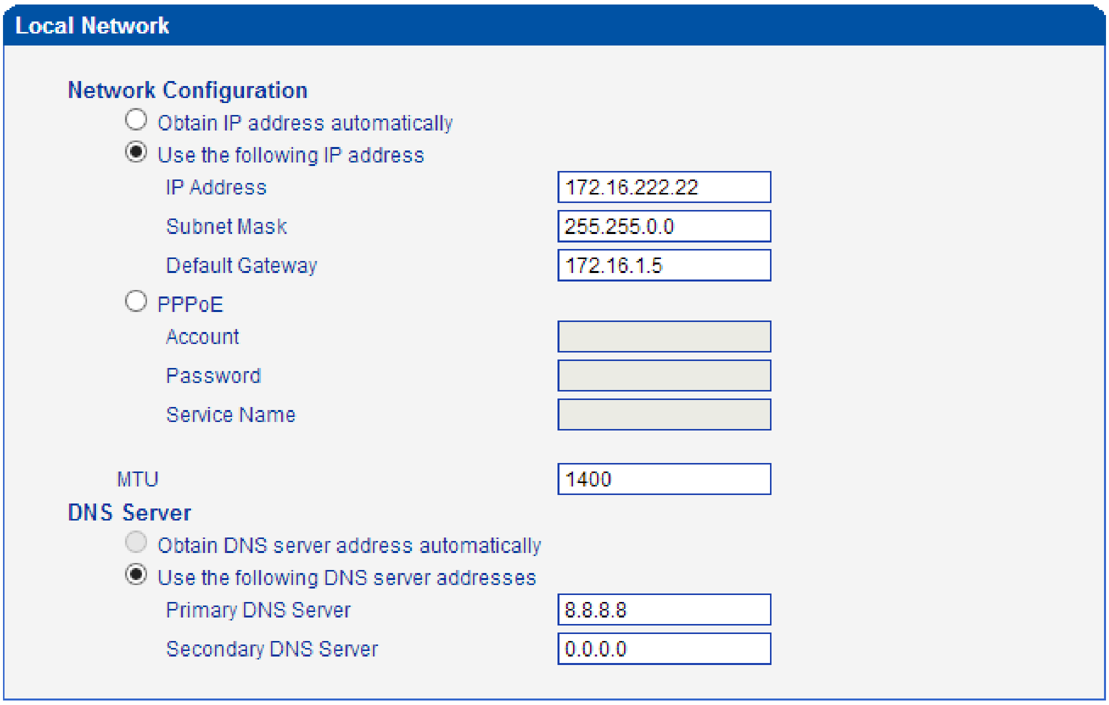


Table 4-5-1 Local network

Parameters	Description
Obtain IP Address Automatically	Enable the device obtain IP Address automatically or not. Default is enabling
Use the Following IP Address	Configure the "IP Address", "Subnet Mask" and "Default Gateway" by manual
PPPoE	Need ISP offer the account and password, Use this mode when there is not router in the local network
MTU	Message transmit unit, default is 1400
Obtain DNS Server Address Automatically	When enable the WAN port option of "Obtain DNS Server Address Automatically", which will be enabled subsequently.
Use the Following DNS Server Addresses	Fill in the IP address of "Primary DNS Server" and "Secondary DNS Server"

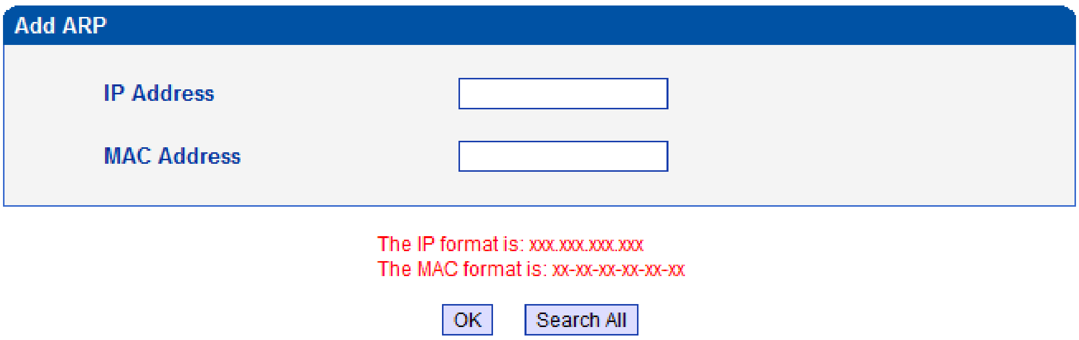
4.5.2 ARP

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

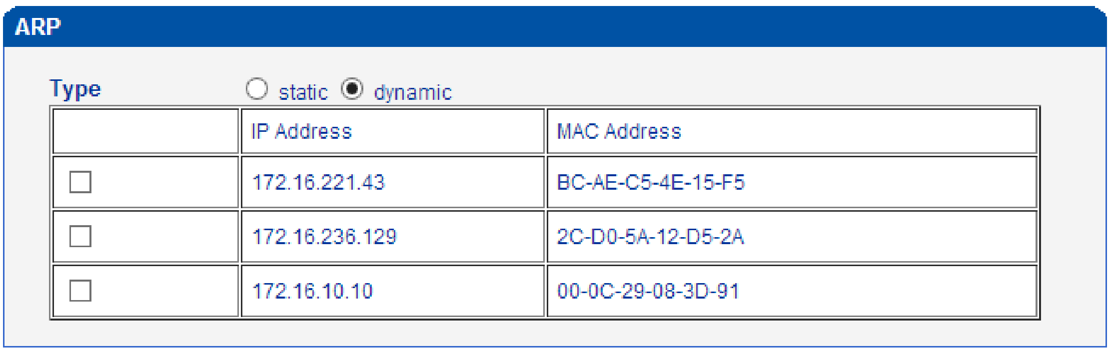
Like other routers, the gateway can automatically find the network device on the same segment. But, sometimes you don't want to use this automatic mapping; you'd rather have fixed (static) associations between an IP address and a MAC address. Gateway provides you the ability to add static ARP entries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

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

Figure 4-5-3 Add ARP



Click *Search All* to check ARP buffer.



4.5.3 VPN Parameter

Figure 4-5-3 VPN Parameter

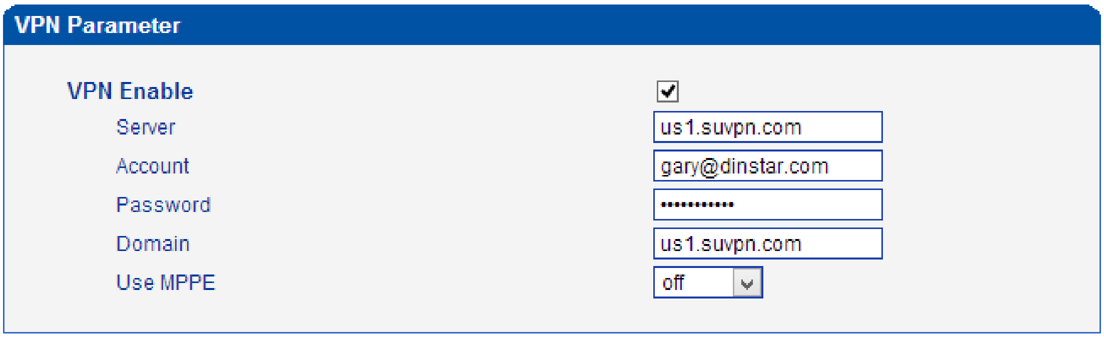


Table 4-5-3 Description of VPN Parameter

Parameters	Description
Server	VPN Server IP or domain name(support PPTP only)
Account	VPN account which provide by server or VPN provider
Password	Password of VPN which provide by server or VPN provider
Domain	Follow VPN setting, can be null
Use MPPE	Encryption parameter, support 40/128 bit, must be match with VPN server

Check VPN connecting status on system information

4.6 Security Center

4.6.1 Access Rules

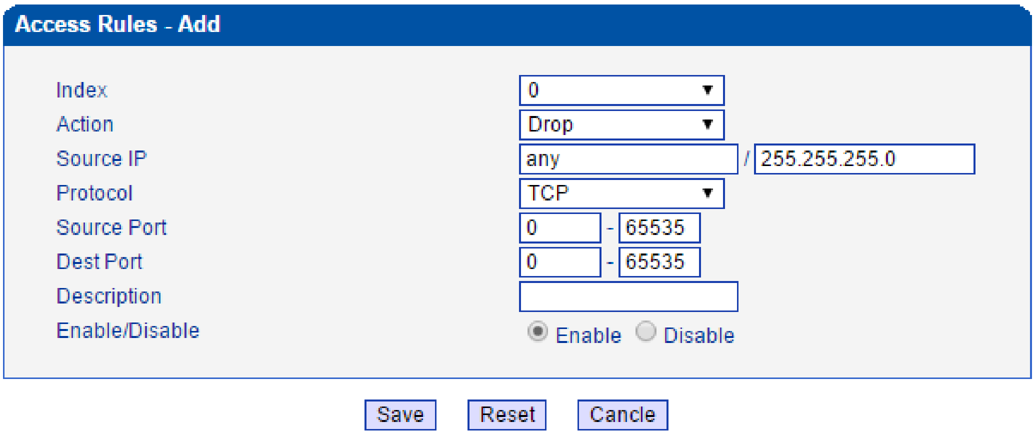
On the **Access Rules** interface, click **Add**, and you can set rules to accept or reject the calls from a specific port, the login of other people via Web or Telnet, or PIN packages.

TCP: accept or reject the login of other people via Web or Telnet;

UDP: accept or reject the calls from a specific port;

ICMP: accept or reject PIN packages.

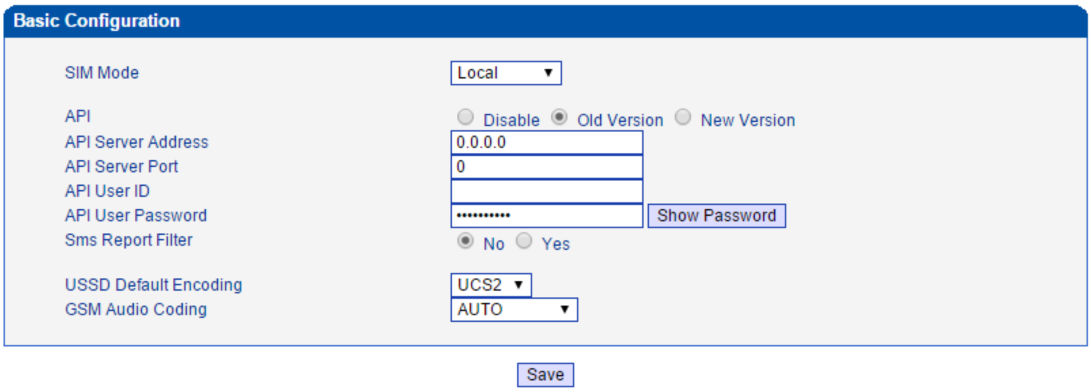
All: accept or reject all the above mentioned items.



4.7 Mobile Configuration

4.7.1 Basic Configuration

Basic Configuration



SIM Mode

Dinstar gateway support two types of SIM card installation, which is local and remote SIM management.

Item	Description
Local	To use local SIM card which install on gateway, this way is most common used by many of users
SIM Box	SIM Box is a small box which use for SIM card storage. It ideal for users who want replace SIM card frequently
SIM Bank	SIM Bank is use for SIM card storage and remote SIM management together with Dinstar SIM Cloud

Introduction to API

The API protocol is used for external applications (for instance: SMS Server) to control the sending and receiving of SMS/USSD on the gateway.

To enable the API function of the GSM gateway, the IP address, port, user ID and password of SMS Sever must be correctly configured, and the TCP Intercept function of the SMS Server must be enabled. Once the connection between the gateway and TCP is established, the gateway will send user ID and password to the SMS Server, and then the SMS Server will send back a message which indicates successful authentication.

The API Server Address, API Server Port, User ID and API User Password on the above interface of Gateway must be the same with the IP Address, Port, Auth ID and Password on the setting interface of SMS Server.

Introduction to GSM Audio Coding

There are eight formats for GSM Audio Coding, including Auto, FR, HR, EFR, AMR_FR, AMR_HR, FR and EFR, EFR and FR.

Auto: it means GSM Audio Coding is automatic.

FR (Full Rate): the first digital speech coding speech standard used in the GSM digital mobile phone system. The bit rate of the codec is 13 kbit/s, or 1.625 bits/audio sample (often padded out to 33 bytes/20 ms or 13.2 kbit/s).

HR (Half Rate): the bit rate of the codec is 6.5 kbit/s. It requires half the bandwidth of the Full Rate codec and network capacity for voice traffic is doubled, at the expense of audio quality. It is recommended to use this codec when the battery is low as it may consume up to 30% less energy.

EFR (Enhanced Full Rate): is a speech coding standard that was developed in order to improve the quite poor quality of Full Rate codec. Working at 12.2 kbit/s, the EFR provides good quality in any noise conditions. The EFR is compatible with the highest AMR mode (both are ACELP). Although the EFR helps to improve call quality, this codec has higher computational complexity, which in a mobile device can potentially result in an increase in energy consumption as high as 5% compared to 'old' FR codec.

AMR (Adaptive Multi-Rate): is an audio compression format optimized for speech coding. AMR speech codec consists of a multi-rate narrowband speech codec that encodes narrowband (200–3400 Hz) signals at variable bit rates ranging from 4.75 to 12.2 kbit/s with toll quality speech starting at 7.4 kbit/s.

There are two modes for the AMR codec in the device:

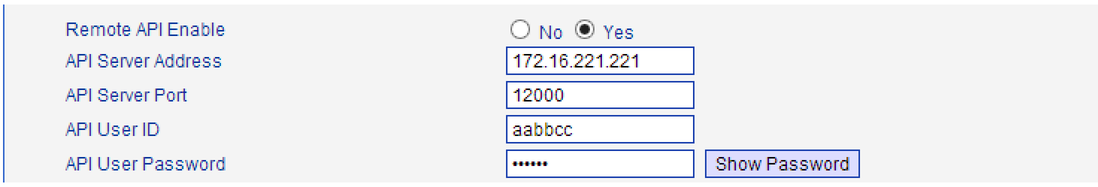
AMR_FR: the AMR codec in a full rate channel (FR)

AMR_HR: the AMR codec in a half rate channel (HR).

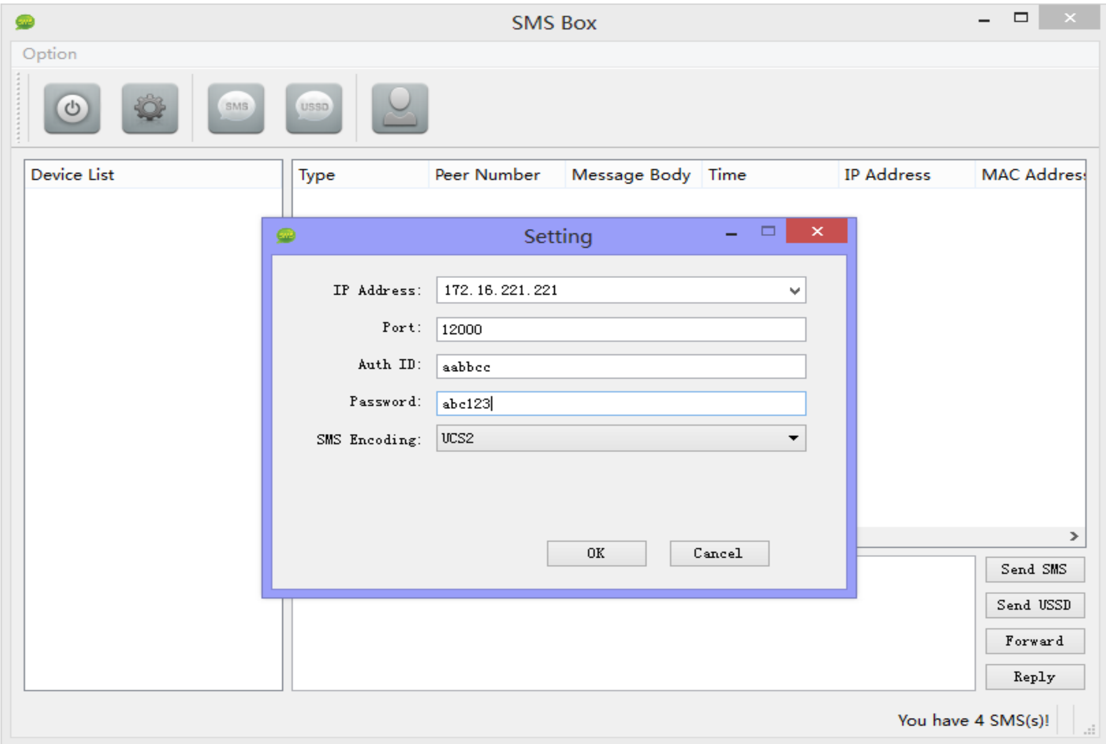
FR and EFR: GSM Audio Coding supports both FR and EFR, but FR is prior to EFR.

EFR and FR: GSM Audio Coding supports both EFR and FR, but EFR is prior to FR.

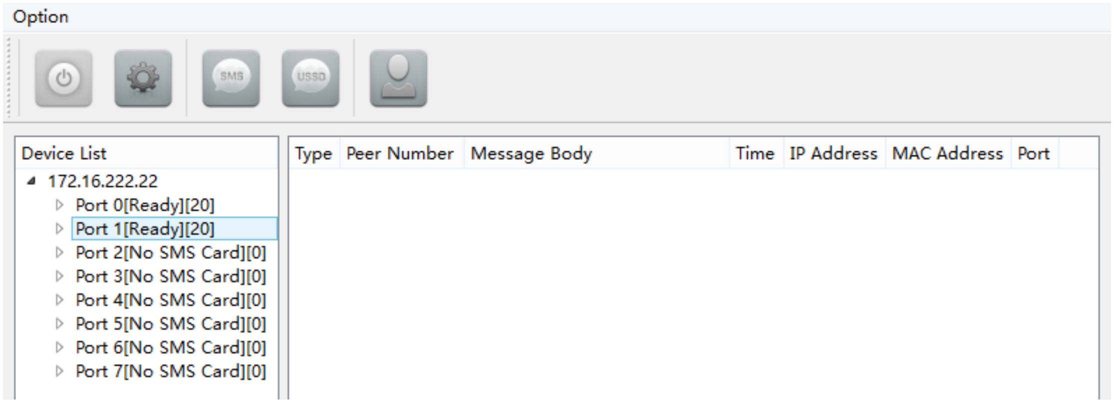
Example:
Configuration between SMS box and gateway
Configure API parameters on gateway



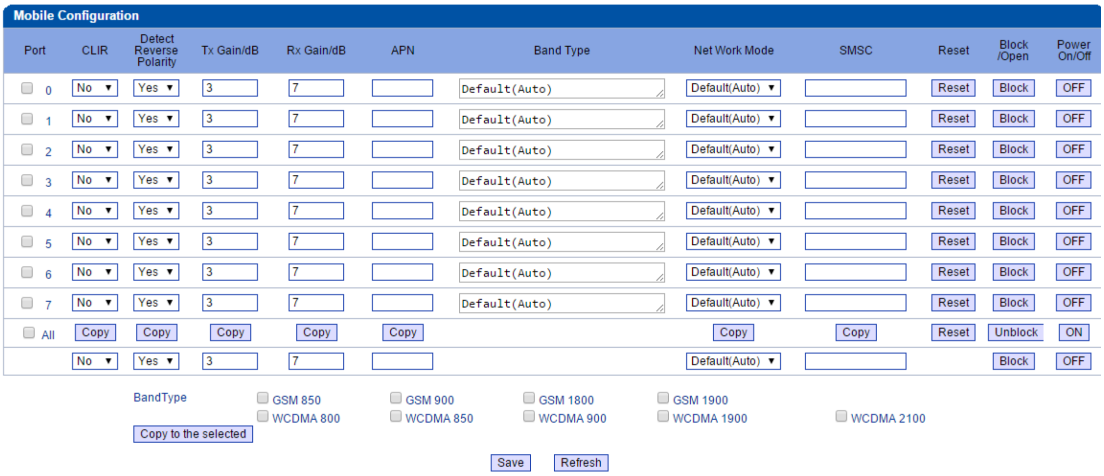
The IP server which installed SMS box software is 172.16.221.221, pre-set Port 12000, User ID aabbcc and password abc123 as example.
Configure SMS box



Then click OK and start service, the gateway IP will be presented in device list of SMS box



4.7.2 Mobile Configuration



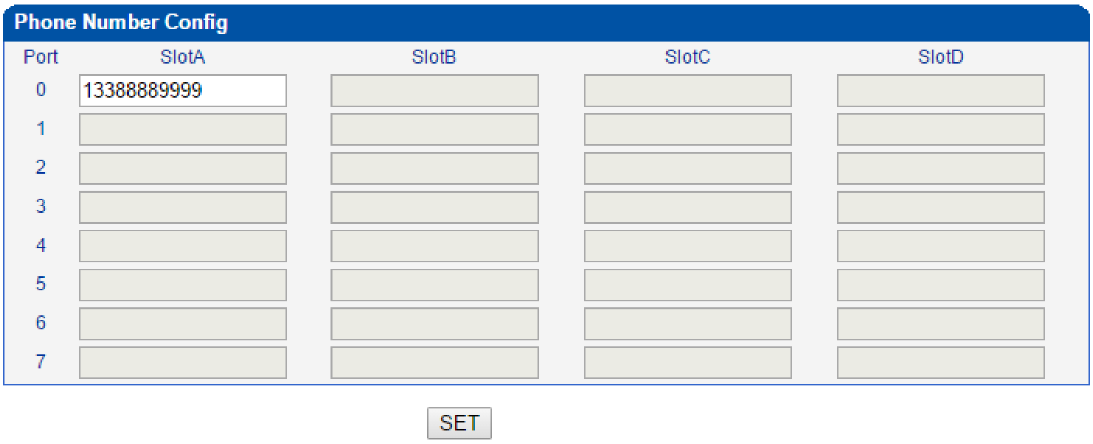
Description of Mobile Configuration

Parameter	**Description**
CLIR	Calling Line Identification Restriction: If the CLIR function is enabled, the phone number of the caller will not be displayed on the called phone.
Detect Reverse Polarity	If the function is enabled, the caller will learn whether the called person has got through the phone.
Tx Gain	Gain of voice sent
Rx Gain	Gain of voice received
Network Mode	Select 2G or 3G
Reset Module	Click **Reset**, and the corresponding module will be reset.
Block/Open Module	Click **Block** or **Unblock**, the corresponding module will turn to the opposite status.
Power On/Off	Click **On** or **Off**, the power of the corresponding module will turn to the opposite status.
Band Type	Choose from GSM850, GSM900, GSM1800, GSM1900, WCDMA800, WCDMA 850, WCDMA900, WCDMA1900, and WCDMA2100

4.7.3 Phone Number Config

On the Phone Number Config interface, you can write a phone number into a specific memory card and SIM Card, and thus the phone number can be called in case that this SIM card has been pulled out and inserted into another port.

Select Yes on the right of ‘Write Phone Number to SIM Card’, enter a phone number and click Submit.

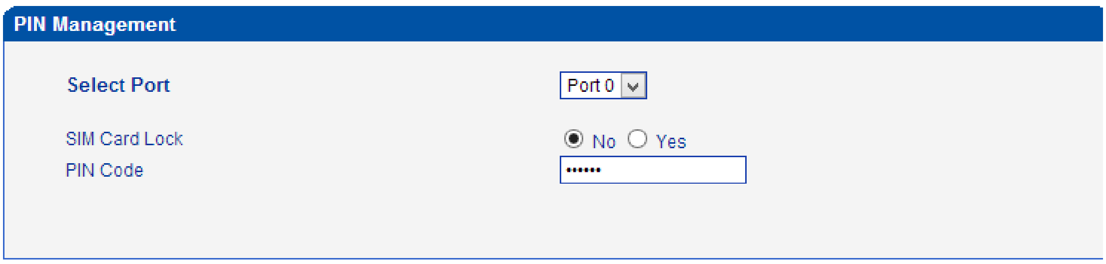


4.7.4 PIN Management

PIN code is a combination of numbers used as an additional password to access the SIM card of the selected port.

On the following interface, you can set a PIN code for the SIM card of the selected port.

PIN Management

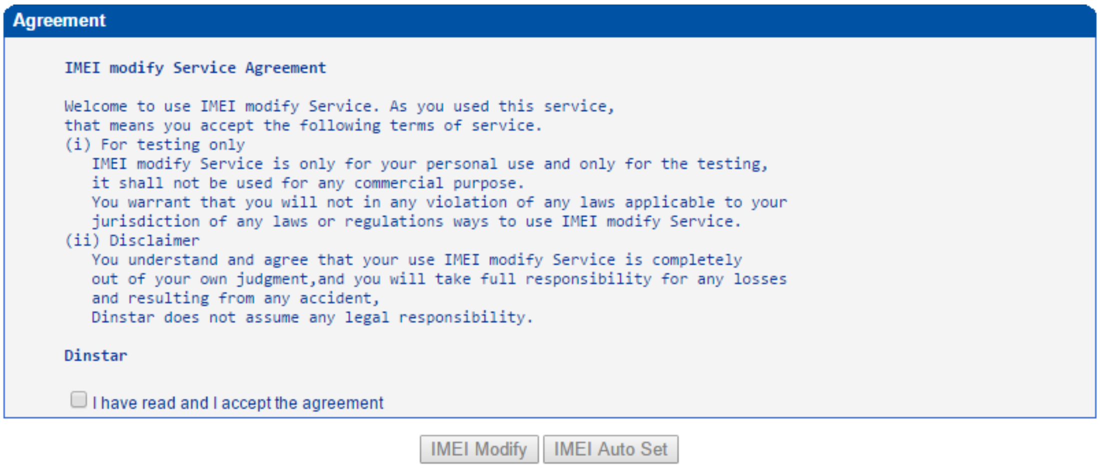


Description of PIN Management

Parameters	Description
PIN	Personal identification number of SIM card. In the status of SIM card locked, PIN can be modified to prevent SIM card from being stolen.

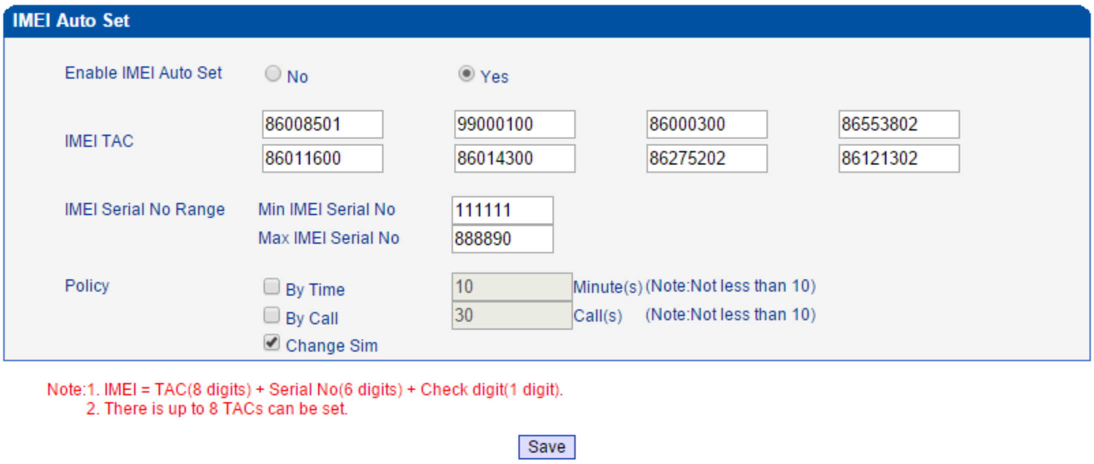
Select Port	Selects the GSM/CDMA channel number

4.7.5 IMEI

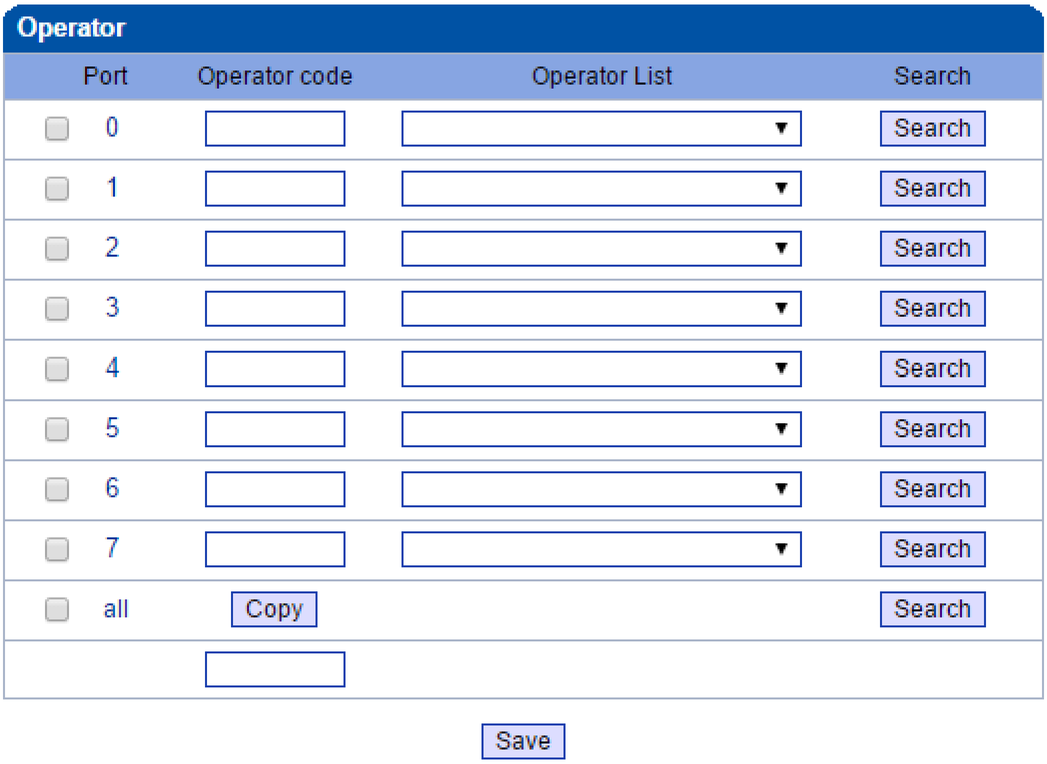


IMEI Modify: to change the IMEI code for specific port/ports

IMEI Auto Set: set some rules to change the IMEI code with some predefinited conditions

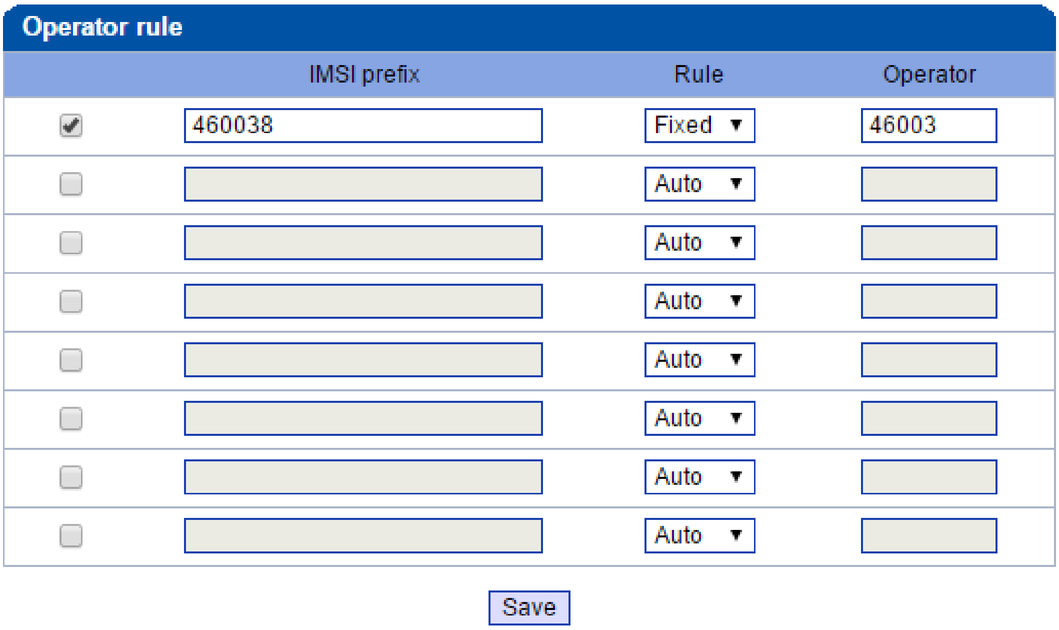


4.7.6 Operator



Click Search button while there is SIM card in that port, after a while, you will see Operator codes list under Operator List dropbox. And then select correct operator code which match with the SIM card insert in the gateway. Finially, save the settting and reboot the device to make SIM card re-register again.

4.7.7 Operator Configuration



Operator configuration aim to set operator code for batch of SIMs. Inserted SIM cards will match with IMSI prefix and register SIM card to the code as per setting.

4.7.8 BCCH

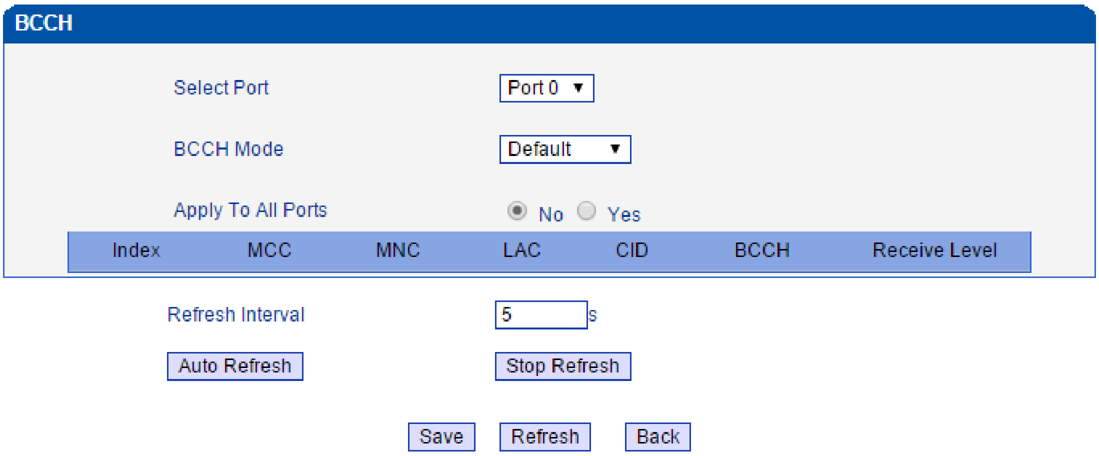
BCCH (Broadcast Control Channel): BCCH is a logical broadcast channel used by the base station in a GSM/WCDMA network to send information about the identity of the network. The information is used by a mobile station to get access to the network. Information includes the Mobile Network Code (MNC), the Location Area Code (LAC) and a list of frequencies used by the neighboring cells.

Configuration Procedures for BCCH:

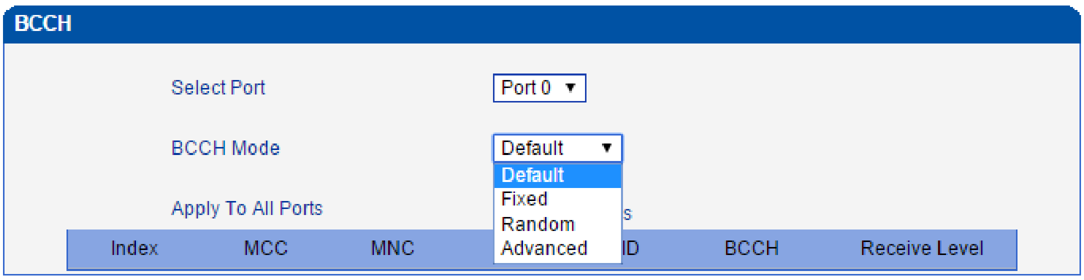
Step 1. In the navigation tree on the left, click **Mobile Configuration → BCCH**.

Step 2. Drag the scroll bar on the bottom of the interface, and you will see Detail buttons.

Click the Detail button of a specific port, and you will see the following interface

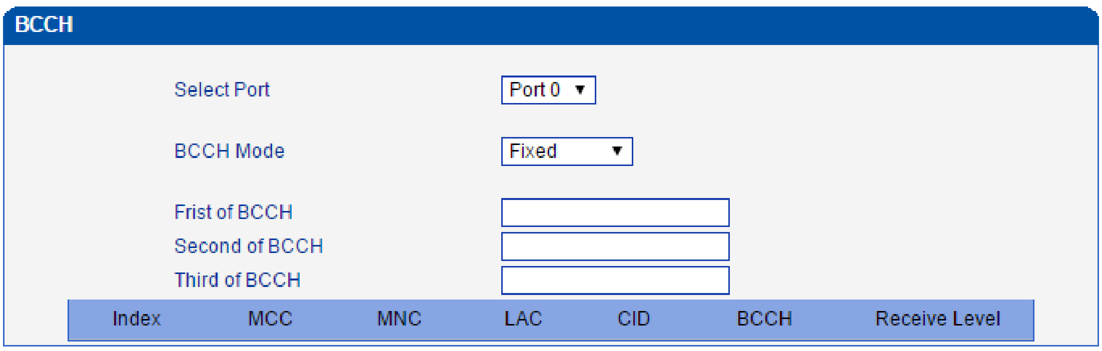


Step 3. Click the drag-down box on the right of **BCCH Mode**, and select a mode.

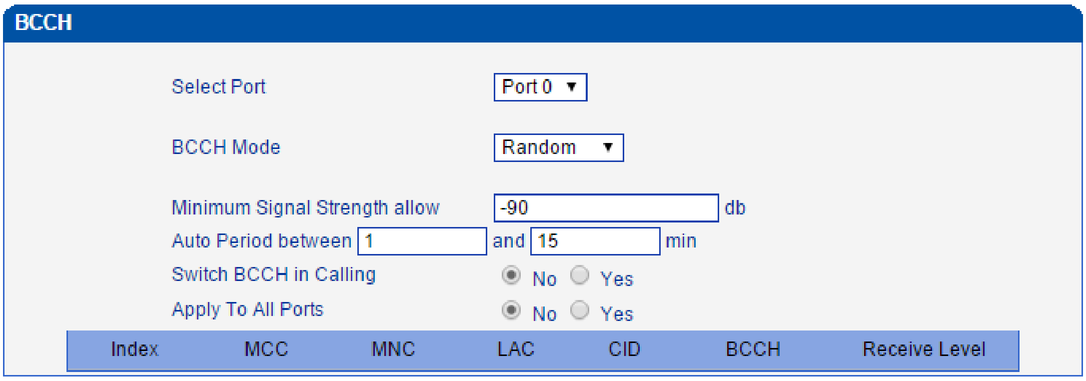


Default: All frequencies will be automatically matched with the gateway.

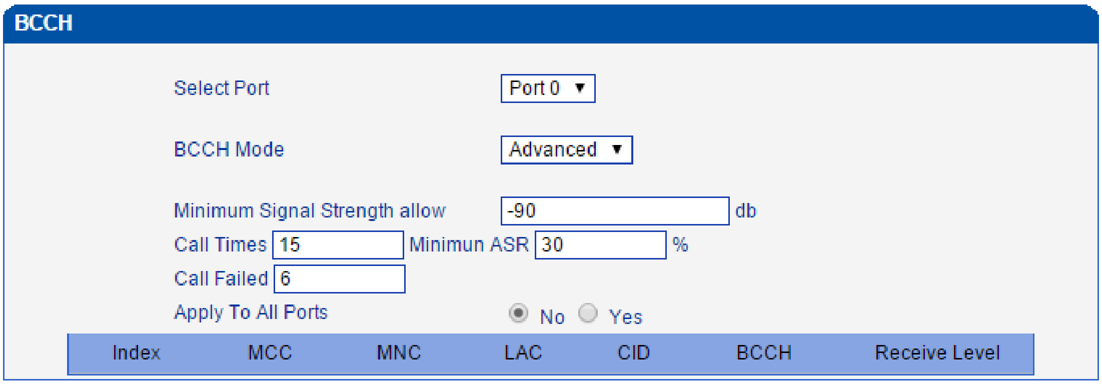
Fixed: You are required to set three fixed frequencies, and the frequencies will be matched with the gateway permanently.



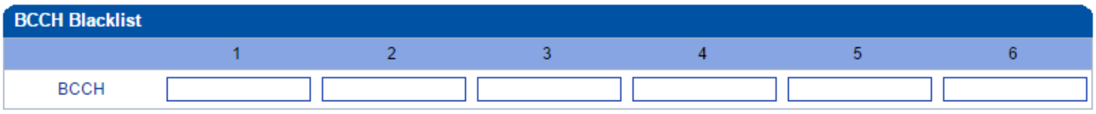
Random: you are required to set some conditions, including minimum signal strength, the period for automatic frequency switch, and whether to switch frequency during calling.



Advanced: you are required to set some conditions, including minimum signal strength, minimum answer-seizure ratio(ASR), number of calls and number of failed calls.



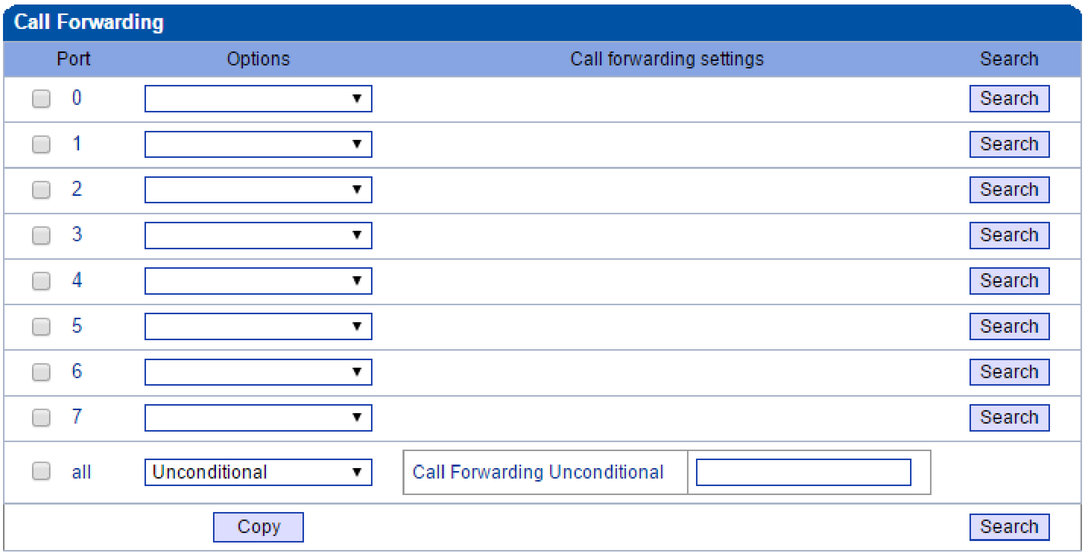
Note: When the actual number of failed calls reaches the set number, frequencies will be switched or when the actual answer-seizure ratio is less than the minimum answer-seizure ratio, frequencies will be switched.



Note: The BCCH Blacklist only works at random mode and advanced mode.

4.7.9 Call Forwarding

Calls can be forwarded unconditionally or under certain conditions.

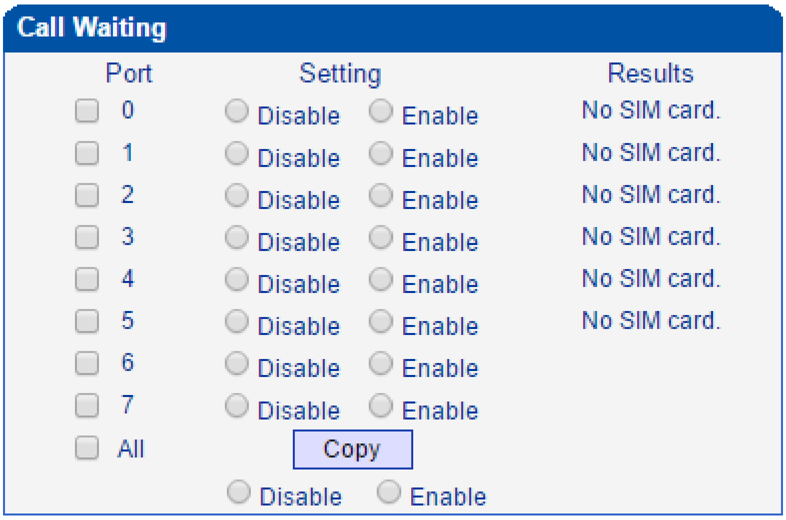


Call forwarding is the same as mobile phone which to activate/deactivate supplementary service of SIM card. For more details of these services, please contact to local providers.

Parameter	Explanation
Call Unconditional	Calls will be forwarded unconditionally
Call Forwarding No Reply	If there is no reply from the called number, calls will be forwarded.
Call Forwarding Busy	If the called number is busy, calls will be forwarded.
Call Forward on Not Reachable	If the called number is not reachable (for example, the called phone is power off), calls will be forwarded.
Cancel All	Calls will not be forwarded.
Call Number	The number where calls will be forwarded.

4.7.10 Call Waiting

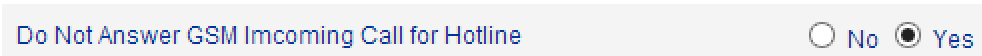
On the **Mobile Configuration →Call Waiting** interface, the call waiting function can be disabled or enabled.



Call waiting is the same as mobile phone which to activate/deactivate supplementary service of SIM card. For more details of these services, please contact to local providers.

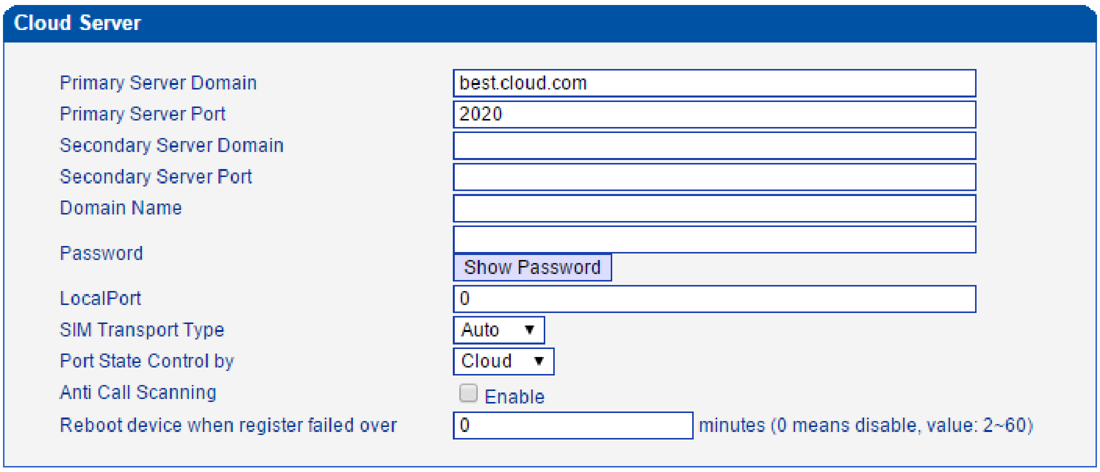
Notes: Call waiting is only take effective while **"Do Not Answer GSM Incoming Call for Hotline"** set to **Yes.**

Call Configuration -> Service Parameter



4.7.11 Cloud Server

Users need to configure the cloud server when the gateway works with SIM Bank or centralized management is required for the gateway.

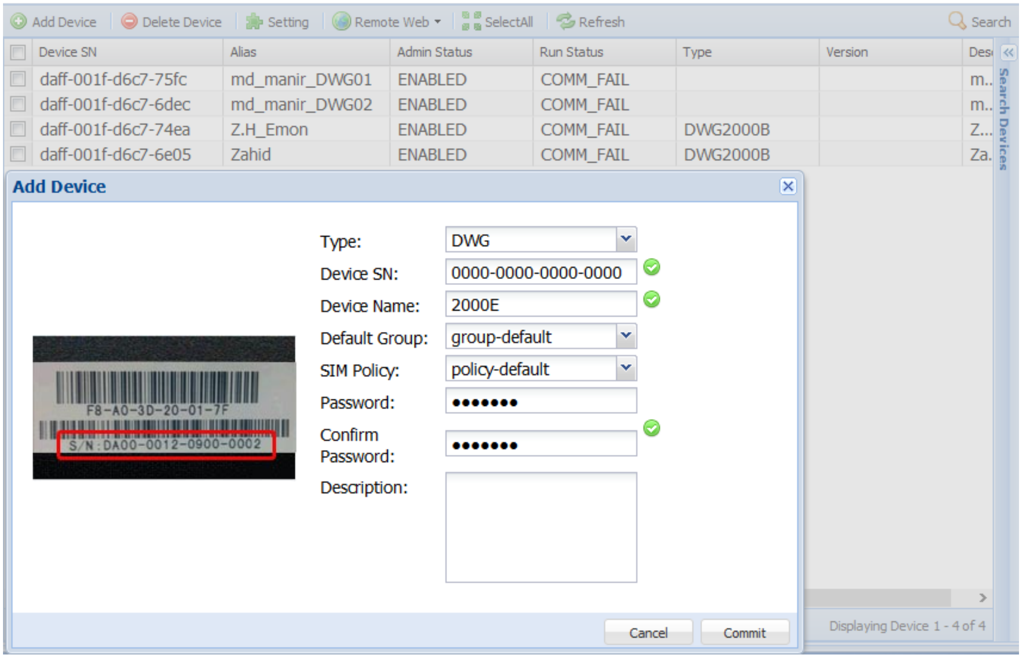


Item	**Description**
Primary Server	The domain name of IP address of the primary Cloud

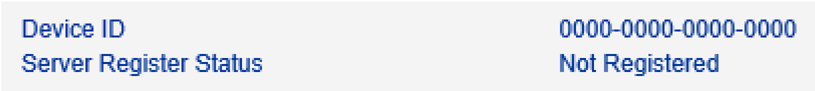
Domain	server
Primary Server Port	The port of the primary Cloud server
Secondary Server Domain	The domain name of IP address of the secondary Cloud server. It can be null.
Secondary Server Port	The port of the secondary Cloud server. It can be null.
Domain Name	The name of the sub-domain used by the gateway under the Could server.
Password	The password of the sub-domain used by the gateway under the Could server.
Local Port	The port of the gateway connected to the Cloud server.
SIM Transport Type	The transmission type of phone numbers of the SIM card.
Port State Control By	The port state is controlled by cloud or the gateway.
Anti Call Scanning	This function must be enabled when the whitelist/blacklist function of the SIM card is enabled.

- **How to register gateway to SIM Cloud?**

Example: add gateway on domain support.cloud.com



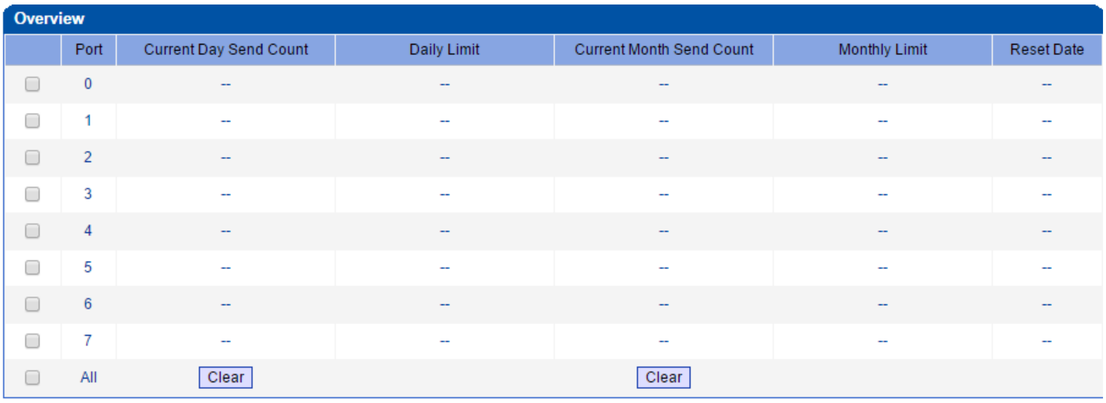
Device S/N is the device ID on gateway, find it on the page **system information**, as below:



4.8 SMS AND USSD

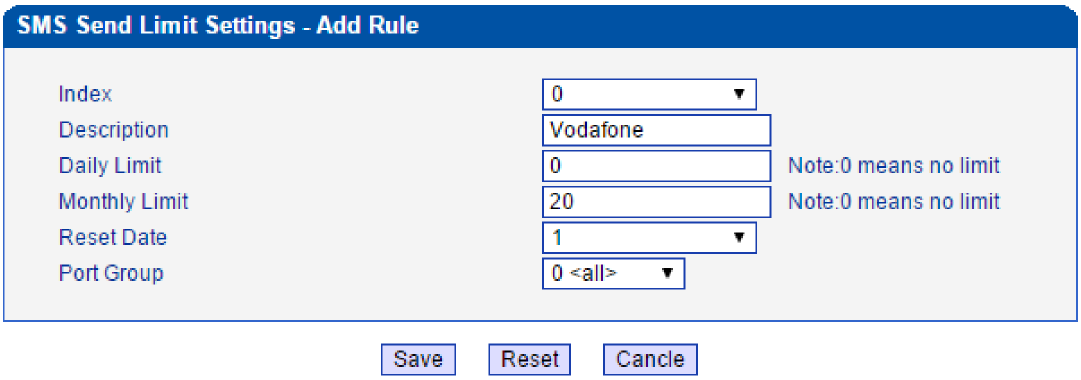
4.8.1 SMS Send Overview

On the SMS Send Overview interface, you can see the number of SMS messages that have been sent via the ports of the gateway, as well as the daily limit and monthly limit of SMS messages that can be sent through the ports of the gateway.



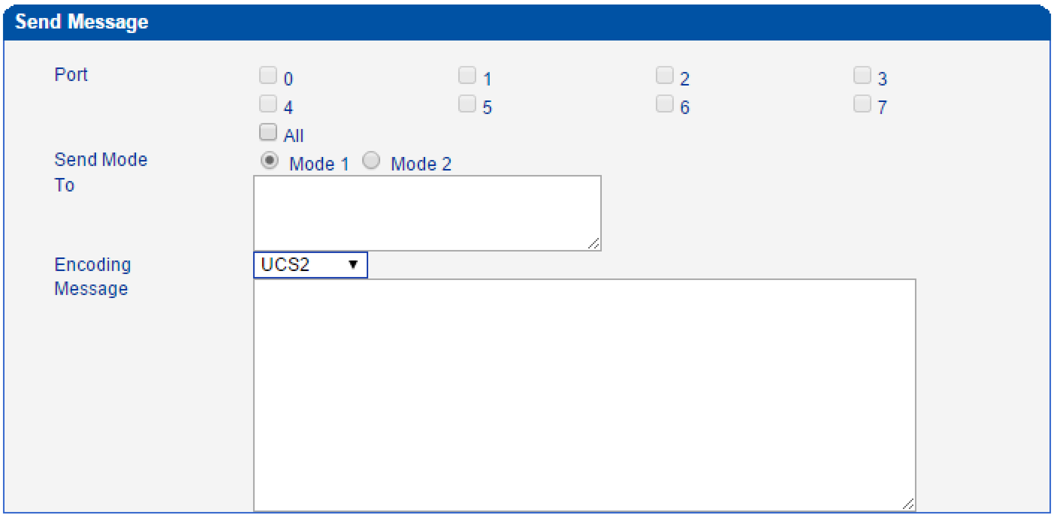
4.8.2 SMS Send Limit Settings

On the SMS Limit Settings interface, click Add, and you can see the following interface.



4.8.3 Send SMS

The GSM gateway can be used to send messages and receive massages.

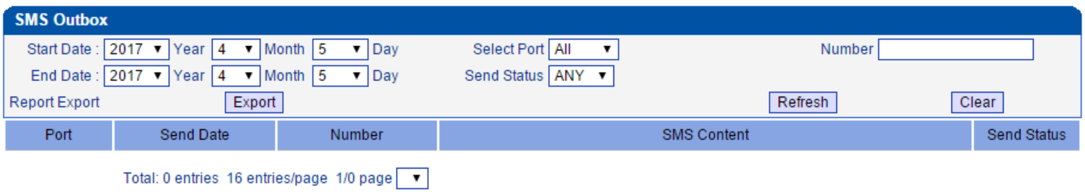


Parameter	Explanation
Port	The port through which SMS messages are sent
To	The number(s) where the SMS message will be sent.
UCS2	UCS2: Support English and Chinese GSM 7bit: Support English only
Message	The content of the message

SMS send report

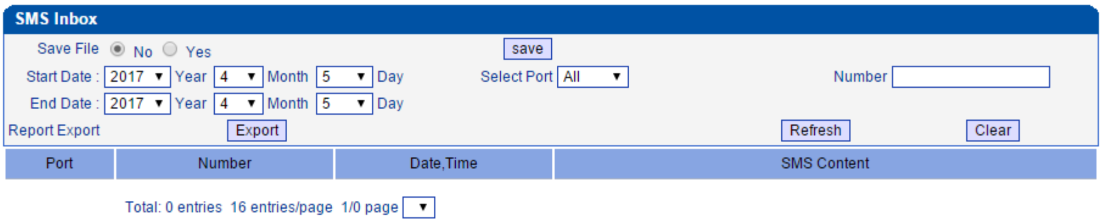
4.8.4 SMS Outbox

On the **SMS Outbox** interface, you can see the detailed information of each SMS message that has been sent, and can export the messages.



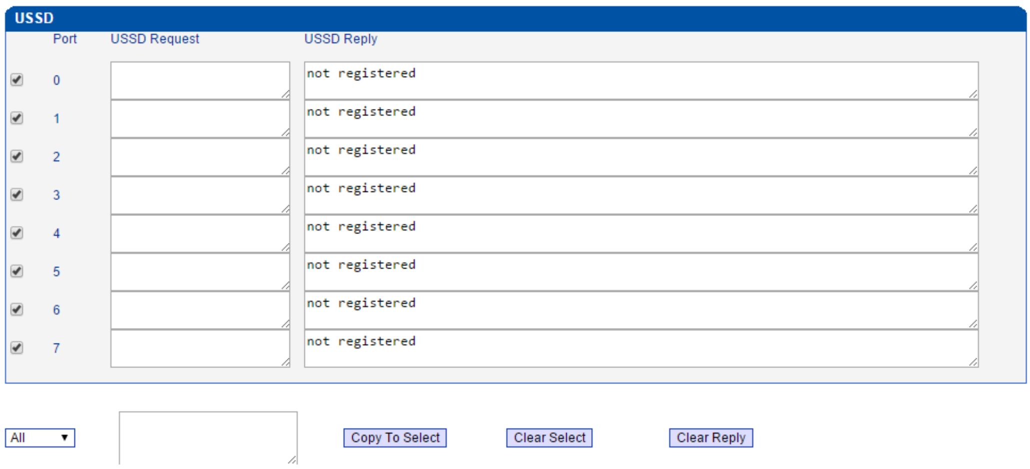
4.8.5 SMS Inbox

On the **SMS Inbox** interface, you can see the detailed information of each SMS message that has been received, and can export the messages.



4.8.6 USSD

USSD (Unstructured Supplementary Service Data): is a service which is provided by a telecom operator and allows GSM/WCDMA mobile phones to interact with the telecom operator's computers. USSD messages travel over GSM/WCDMA signaling channels and are used to query information and trigger services. Unlike similar services (SMS and MMS), which are stored and forwarded, USSD is session-based. It establishes a real-time session between mobile phones and telecom operators' computers or other devices.



4.8.7 Email

How to set Email to SMS

Description

GSM gateway can check the email inbox on time, when have unread email at list and size less 300 chars, will try to read it.

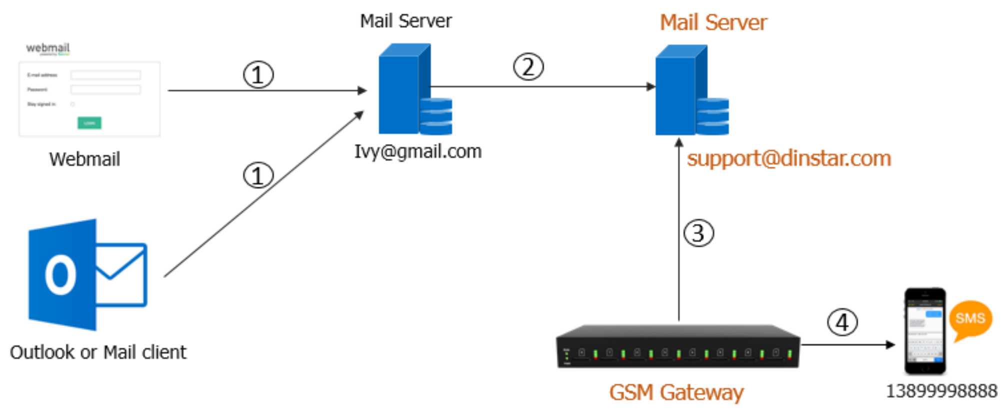
When email use protocol IMAP, if email read successful, will set the email status to read. If read email failed, will try to read again (MAX 3 times). If failed final, the email status will be set to read also.

When email use protocol POP3, if email read successful, will delete it. If email read failed, keep the email status, because the UC2000 will check again at next check time.

After read the email, if the subject matched, will extract the context from key words: “To:”, “Encoding:”, “Message:” as SMS receive number, SMS encoding, SMS context.

If the GSM gateway have not available channel at that time, it will keep the SMS in queue and waiting till have available one. The queue max has 5120 SMS. If the queue full, the UC2000 will stop to check the email.

How does SMS to email works



1) Send Email from webmail or outlook client

   Email Format: Plain text

   Email subject: Test SMS

   Email contents:

   To:13899998888
   Encoding:7Bit
   Message:Hola, this is test SMS from ivy@gmail.com

2) Gmail server forward email to support@dinstar.com
3) GSM gateway check the inbox of support@dinstar.com, find the email subject with ‘Test SMS’
4) GSM gateway send SMS to mobile 13899998888

Notice: Don't set signature at the end of email and make sure the received email is plain text format.

How to configure Email to SMS in GSM gateway

1) Open page **SMS and USSD**>>>>>>>**Email**.
   Email to SMS support both POP3 and IMAP protocol.

   The "Server Domain" means your email services server info, you can get it from your email provider.

   The "TLS Enable" means use Encrypt or not.

   If use TLS, IMAP default server port is 993, POP3 default server port is 995.

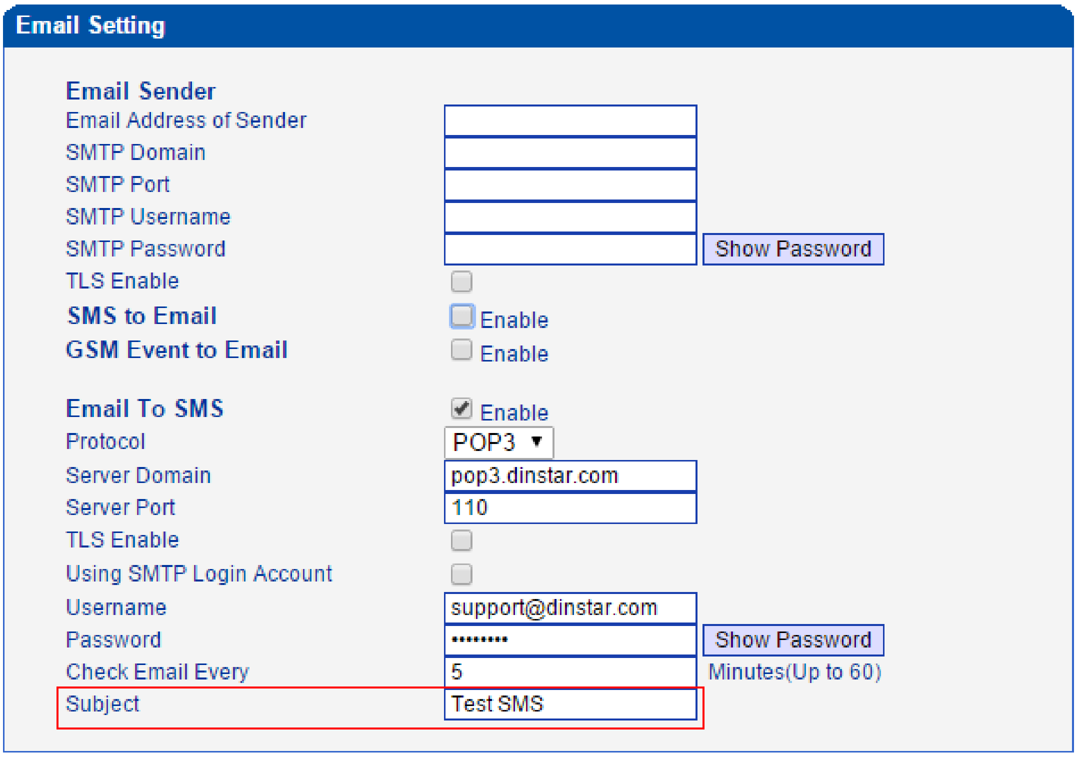
   If not use TLS, IMAP default server port is 143, POP3 default server port is 110.

   The "Using SMTP Login Account" means when you use SMS to Email, you can use the same username and password info for Email to SMS.

   The "Check Email Every" means how long the UC2000 will check the email inbox, the set range is 1-60.

   The "Subject" means when the UC2000 match the email subject, will use that email to SMS.

   Add the Email address info at UC2000 side, like follow pic.



2) Email must use fix format:

Subject: this subject **MUST** be the same as email subject. Example, when you send email with subject “Test SMS”, the Subject s field in GSM gateway must be “Test SMS” also.

Email contents usually include 3 parts:

The “To” means destination number you want send to, this option is obligatory. The format is:

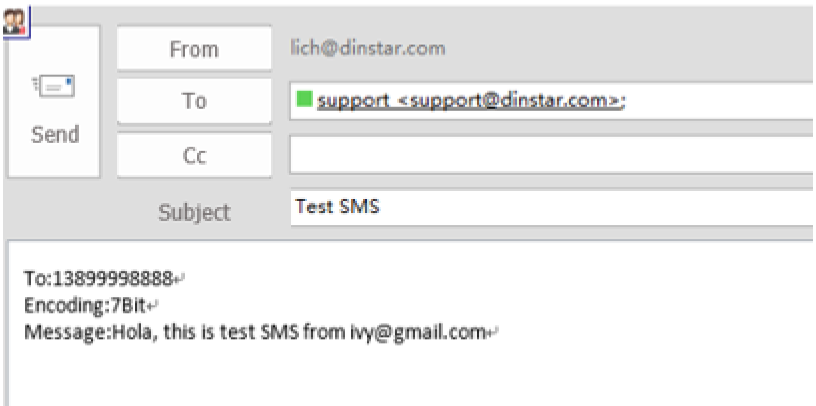
To:xxxxxxxxxxxx

The “Encoding” means which format of SMS used, the format include 7Bit and UCS2, UCS2 is default. This option is Optional. The format is:

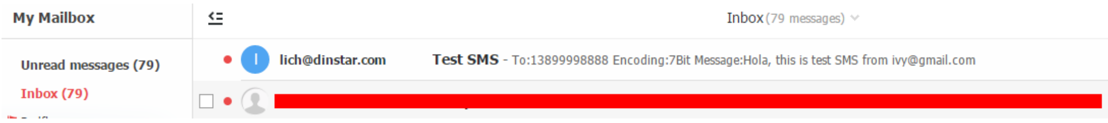
Encoding:7Bit

The “Message” means which content you want send out, this option is obligatory. The content length max 300 chars. The format is:

Message: ………………………………



Received email in the inbox of support@dinstar.com should be

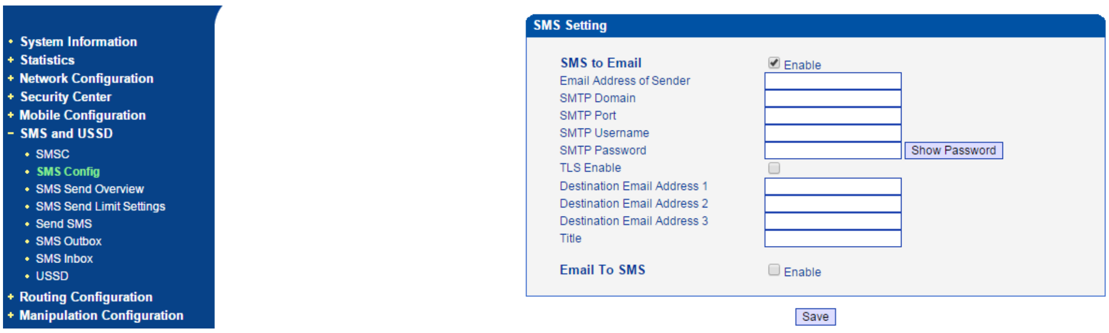


Note:

1) Character set. The UC2000 support character set ASCII and UTF-8 only.
2) Encoding. The email encoding support 8Bit, Base64 and Quoted-Printable only. If the email senders use other encoding, like 7Bit, it will not support.
3) Email size. The email size can't more than 300 chars, if more than it, the UC2000 will not try to read it.

How to set SMS to Email

The UC2000 series gateway support to send the SMS received on the gateway to user's mail box. Login device's web, go to **SMS and USSD-->SMS Config** page, enable SMS to Email function, and configure the other parameters needed.Here is the configuration page:



Email Address of Sender: Configure one e-mail address, which will be used for sending the SMS to destination e-mail user configured.

SMTP Domain: Configure the SMTP server domain here, different e-mail address server have different server addresses, please confirm this with your e-mail provider or search from Internet.

SMTP Port: configure the SMTP port, usually 25, please also confirm this with your e-mail address provider.

SMTP Username: Enter your e-mail address username.

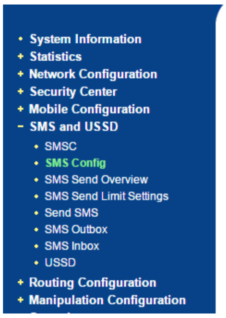
SMTP Password: Enter the password of the e-mail address you configured, please make sure it is correct.

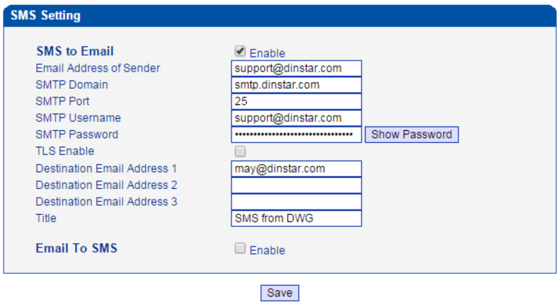
TLS Enable: Enable the TLS or not. If your e-mail address server requires TLS, please enable it.

Destination Email Address 1/2/3: Enter the e-mail address to receive the SMS content.

Title: Configure the title of the e-mail, which will be used as the e-mail title when send the SMS to destination mail.

Here is one example:





4.9 CALL CONFIGURATION

4.9.1 SIP Configuration

This section is describes how to configure SIP server and SIP parameters.

- **Configure SIP server and Outbound Proxy server**

- **SIP Server Address and Port**

  Used for configure SIP server address and port, the address can be IP Address, also can be a domain name which can be resolved by DNS server

- **Check NET Status**

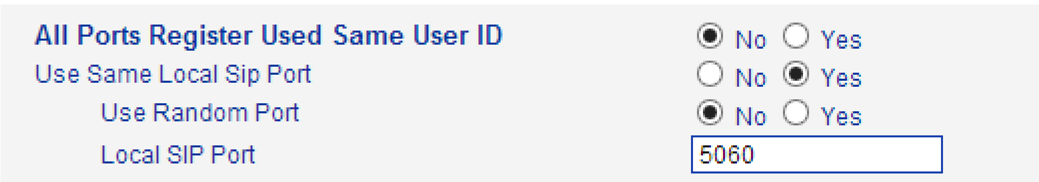
  Default is No. if it set to Yes, the gateway will send SIP OPTION periodic to check health status between gateway and SIP server.

- **Outbound Proxy**

  Outbound proxy, it mainly used in firewall / NAT environment. That make the signaling and media streams are able to penetrate the firewall.

Local SIP Port Configuration

In order to work different application scenarios, the gateway provides flexible configuration with local SIP port.



- **Random**

  The gateway will generates SIP port after each reboot by random. It is commonly used while 5060 is blocked or conflict with other devices.

- **Use the same SIP port**

It is mostly used to SIP trunk interworking with SIP server so that the gateway able to deal with high performance concurrent calls.

Use the same local SIP port and SIP User ID

Port List

Port	SIP User ID	Authenticate ID	Tx Gain	Rx Gain	To VOIP Hotline	To PSTN Hotline	Auto-Dial Delay Time(s)	Detail
0	1000	1000	2	6	s		3	Detail

- **Use the separate SIP port**

Each channel has separate SIP port so that they could be handle SIP call separately.

After *Use Same Local SIP Port* set to *No*

All Ports Register Used Same User ID	◉ No ○ Yes
Use Same Local Sip Port	◉ No ○ Yes

The Local SIP port will be changed on *Port Parameter* page.

Port List

Port	SIP User ID	Authenticate ID	Local Sip Port	Tx Gain	Rx Gain	To VOIP Hotline	To PSTN Hotline	Auto-Dial Delay Time(s)	Detail
0	1000	1000	5060	2	6	s		3	Detail
1			5062	2	6			0	Detail
2			5064	2	6			0	Detail
3			5066	2	6			0	Detail
4			5068	2	6			0	Detail
5			5070	2	6			0	Detail
6			5072	2	6			0	Detail
7			5074	2	6			0	Detail

- **Register Interval and DNS SRV**

Is Register	○ No ◉ Yes
Register Interval(range: 1 - 3600s)	1800 s
DNS query type	A query
DNS refresh interval (range:0 - 60,000min, 0 means disable)	0 min

- **Is Register**

Default set yes, if you want the device can make a call without register, set No, Also enable the "Allow Call from IP to PSTN without Registration" and "Allow Call from PSTN to IP without Registration" function

- **Register Interval**

This field specifies the value that the gateway will send in the Expires header of the REGISTER message. Its value range from 1-3600s. But in fact, the gateway will get 200OK response from SIP server after REGISTER request, and an Expires header will be included in 200 OK message body. This value in the 200OK determines the time, in seconds, after which the registration expires. The gateway will refresh the registration Timer Register Delta seconds before the end of this interval.

- **DNS query type**

The DNS query type defines the type of information that will be requested from DNS server

- **DNS refresh interval**

The interval of DNS refresh, Range from 0 to 60000 mins, 0 means disable default value is disable.

- **Configuring SIP Timers**

T1	500	ms
T2	4000	ms
T4	5000	ms
TMAX	32000	ms
Keepalive Interval(range:32 - 3600s)	32	s
Keepalive SIP ID		
Keepalive Retry Count(range:1 - 10)	3	
Enable 100rel	◉ no ○ yes	

- **T1**

This field specifies the lowest value, in milliseconds, of the retransmission timer for SIP messages. Default specifies 500.

- **T2**

This field specifies retransmission timer for T1 timeout of SIP message, in milliseconds. Default specifies 4000.

- **T4**

This field specifies retransmission timer for T2 timeout of SIP message, in milliseconds. Default specifies 5000.

- **TMAX**

This field specifies maximum timeout value for SIP message. The SIP message will be dropped after TMAX. Default value is 32000

- **Keepalive Interval**

The gateway can monitor the status of SIP server by sending periodic SIP OPTION messages. This field specifies transmission timer of OPTION message. Its range from 32-3600s.

- **Keepalive SIP ID**

This field specifies SIP ID of OPTION. The format would be <xxx@host.com >, example:

***OPTIONS** sip:heartbeat@172.16.0.8:2080 SIP/2.0*

Via: SIP/2.0/UDP 172.16.222.22;branch=z9hG4bK45c4f8d2026d9eed8a0adcd533161efd;

*From: **<sip:heartbeat@172.16.222.22:2080>**;tag=6d48f0a169d33fe7b032c0fd895084fd*

*To: **<sip:heartbeat@172.16.0.8:2080>***

Call-ID: 8874a4e49f11af243c6b717c05a16e35@172.16.222.22

CSeq: 1804289386 OPTIONS

Contact: <sip:31@172.16.222.22>

Max-Forwards: 70

Accept: application/sdp

Content-Length: 0

- **Keepalive Retry Count**

This field specifies retransmission times for OPTION message. Its value range from 1-10 times.

Configuring Caller ID and 183 Mode

From Mode when Caller ID Is Available	Tel/User
From Mode when Caller ID Is Unavailable	Anonymous
Answer Mode	Answered
183 Mode	Immediately
Called Number Parse	Request-Line

- **From Mode when Caller ID Is Available**

Used to configure "From" Mode when Caller ID Is Available when call from GSM to VoIP

Tel/User: *From: Caller ID <sip:3001@host.com>;tag=51088abb*

User/User: *From: 3001 <sip:3001@host.com>;tag=51088abb*

Tel/Tel: *From: Caller ID <sip: Caller ID@host.com>;tag=51088abb*

User/Tel: *From: 3001 <sip: Caller ID @host.com>;tag=51088abb*

- **From Mode when Caller ID Is Unavailable**

Used to configure "From" Mode when Caller ID Is Unavailable

Anonymous : *From: <sip: Anonymous @host.com>;tag=51088abb*

Username : *From: <sip: Username @host.com>;tag=51088abb*

- **Answer Mode**

Answered: Gateway will send SIP message "200 OK" to SIP Server after GSM/CDMA users answered the call.

Alerted: Gateway will send SIP message ‘200 OK’ to SIP Server immediately after 183 Ringing. In this situation, the called party possibly still in ringing status.

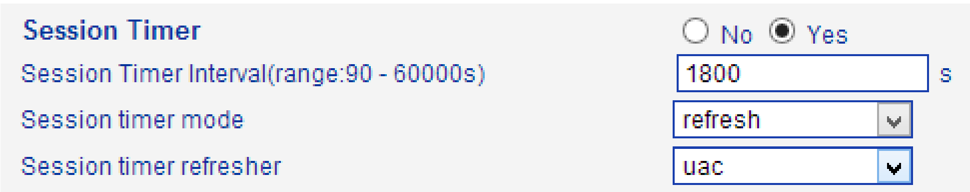
- **183 Mode**

  Immediately: Gateway will send "183 RING" immediately to SIP Server while it receive “INVITE". In this situation, the called party possibly still not in ringing status.

  Alerted: Gateway will send "183 RING" after received exact ringing signal from GSM/CDMA network. In this situation, the called party is definitely in ringing status.

Session Timer

SIP Session Timers which is an extension of SIP RFC 4028 that allows a periodic refreshing of a SIP session using the RE-INVITE/UPDATE message. The refreshing allows both the user agent and proxy to determine if the SIP session is still active. The SIP Session Timer is a keep alive mechanism for SIP sessions that allow User Agents (UA) or proxies to determine the status of a session and to release it if it is not active, even if a BYE has not been received.



- **Session timer Interval**

  The initial INVITE request establishes the duration of the session and may include a Session-Expires header and a Min-SE header. These headers indicate the session timer value required by the user agent (UAC). A receiving user agent server (UAS) or proxy can lower the session timer value, but not lower than the value of the Min-SE header. If the session timer duration is lower than the configured minimum, the proxy or UAS can also send out a 422 response message. If the UAS or proxy finds that the session timer value is acceptable, it copies the Session-Expires header into the 2xx class response.

  A UAS or proxy can insert a Session-Expires header in the INVITE if the UAC did not include one. Thus a UAC can receive a Session-Expires header in a response even if none was present in the request. Its value range from 90-60000s.

- **Session Timer Refresher**

  It specifies refresher which including in SIP message body, user agent client (UAC) or user agent server (UAS).

  UPDATE sips:bob@192.0.2.4 SIP/2.0

Via: SIP/2.0 pc33.atlanta.example.com;branch=z9hG4bKnashds12

Route: sips:p1.atlanta.example.com;lr

Supported: timer

*Session-Expires: 4000;**refresher=uac***

Max-Forwards: 70

To: Bob <sips:bob@biloxi.example.com>;tag=9as888nd

From: Alice <sips:alice@atlanta.example.com>;tag=1928301774

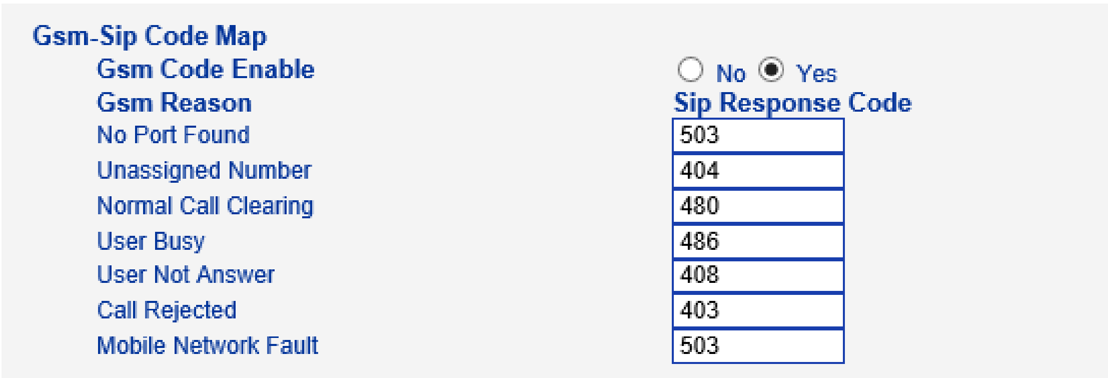
Call-ID: a84b4c76e66710

CSeq: 314162 UPDATE

Contact: <sips:alice@pc33.atlanta.example.com>

Configuring GSM-SIP Mapping Code

This part specifies response codes between GSM cause reason and SIP response code.

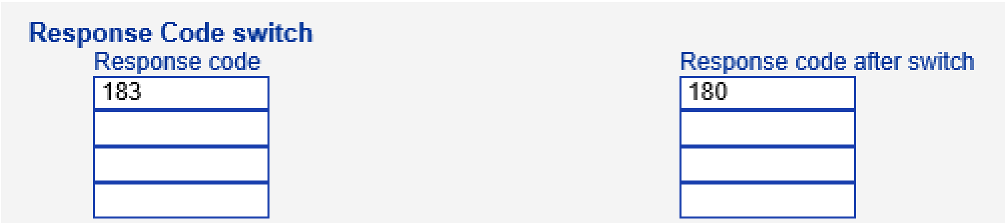


SIP Response

404	Not Found
408	Request Timeout
403	Forbidden
486	Busy Here
480	Temporarily unavailable Resource unavailable
503	Service Unavailable

Response Code switch

This part specifies response codes of SIP between gateway and SIP server. Refer to table *SIP Response,* the SIP server possibly need some specific SIP response from the gateway. Example, SIP server need SIP response *180 Ringing* instead of *183 Ringing,* the configuration should be as below:



4.9.2 SIP Trunk Configuration

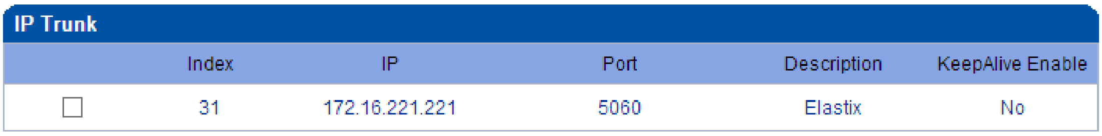


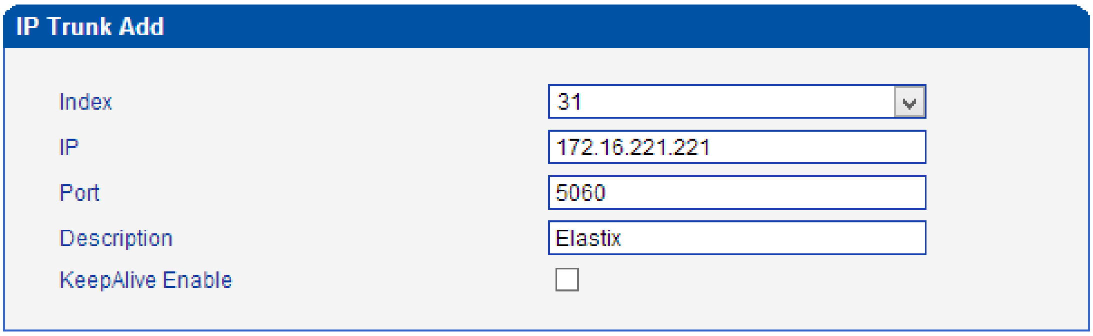
Table 4-11-1 Description of IP Trunk

Parameters	Description
SIP Trunk	Add remote IP of Softswitch, SIP server which will send call traffics to gateway.
Index	It uniquely identifies a trunk. Its value is assigned globally, ranging from 0 to 31.
Description	It describes the trunk for the ease of identification. Its value is character string
IP	It is an interworking parameter between the remote Softswitch and the SIP server. It specifies the IP address of the peer equipment.
Port	It is an interworking parameter between the remote Softswitch and the SIP server. It specifies the SIP port number of the peer equipment

Keep alive	Send OPTION to Softswitch/IPPBX to detect health status

Example

To add a remote IP of Softswitch, SIP trunk index is 31, SIP port number "5060"



4.9.3 SIP Trunk Group

Figure 4-11-3 IP Trunk Group

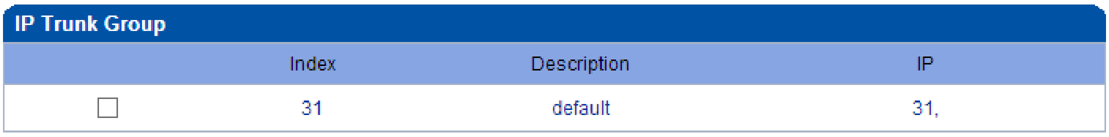


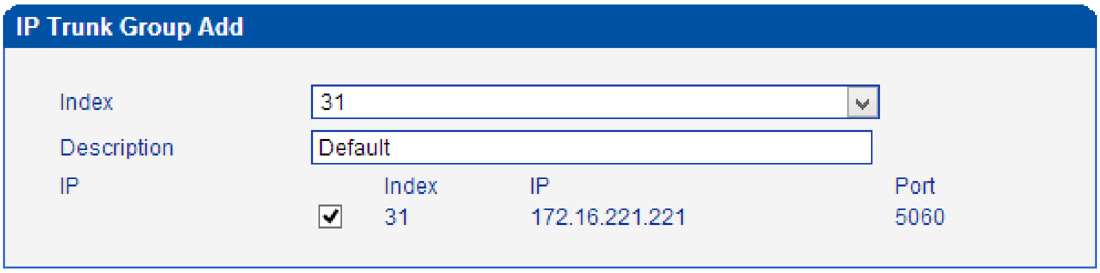
Table 4-11-2 Description of IP Trunk Group

Parameters	Description
IP Trunk Group	This configuration is optional, and is used to add the IP that have the same attributes to an IP group. The IP group will referenced by IP->Tel routing and number manipulation.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.
Description	It describes the route for the ease of identification. Its value is character string
IP	It specifies the IP will add to IP group

Example

To add an IP group, set IP "10, 14, 17" to IP group 18

Figure 4-11-4 IP Trunk group modify



4.9.4 Port Configuration

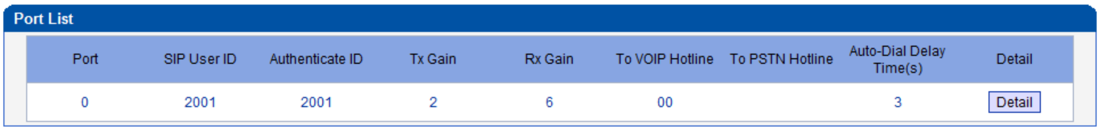


Figure 4-12-4 Port Configuration

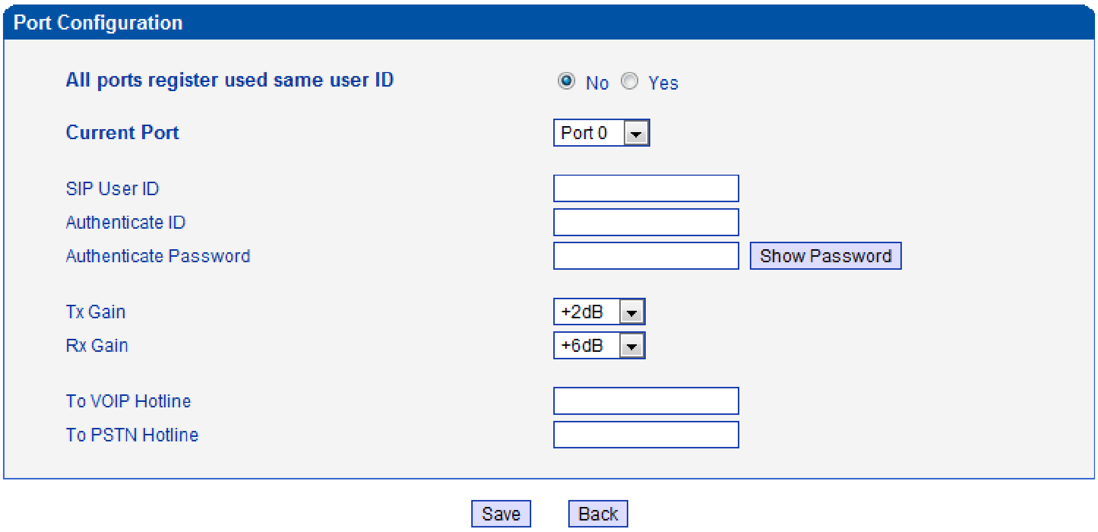
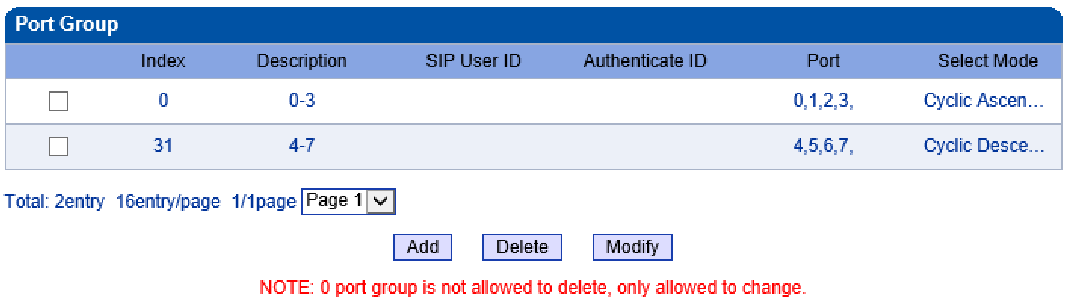


Table 4-12-3 Description of Port Configuration

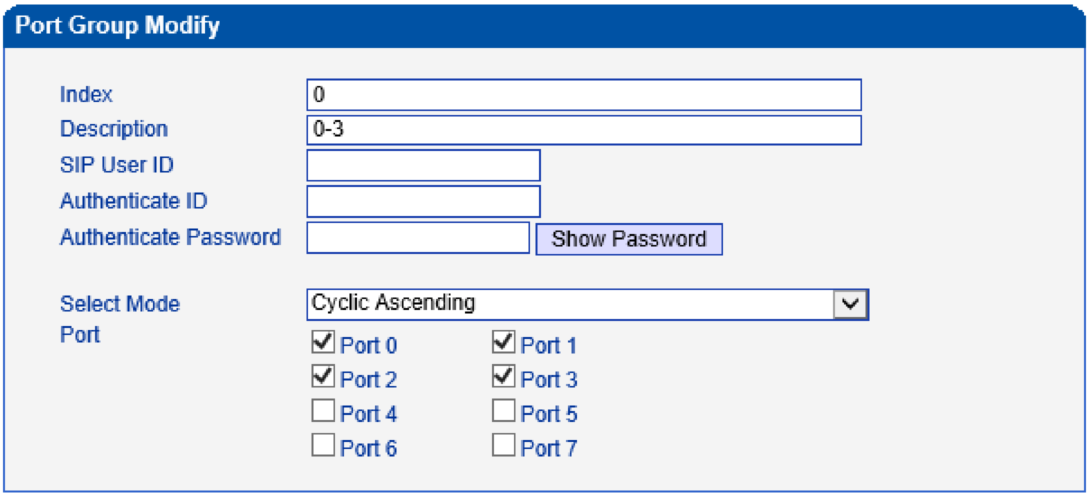
Parameters	Description
Port Configuration	Used to configure ports' gain, Auto-Dial, etc.
ALL ports register used same user ID	The default is no. If set to "yes" ,all the ports will use the same user ID to register to SIP server

SIP User ID	It is the account used for registration which provide by SIP server, equipment port's unique identifier
Authenticate ID	The Authentication ID is used for authentication purposes. The SIP user ID is usually the phone number you received from the service provider. Often, the Authentication ID is the same as the user ID
Password	Password of SIP User ID which provide by SIP server
Tx Gain	Tx Gain value of chipset. Adjusting it will effect volume on GSM side.
Rx Gain	Rx Gain value of chipset. Adjusting it will effect volume on IP side.
To VoIP Hotline	When mobile / fixed line users make call to this port, gateway will auto forward to dedicate number. The hotline could be DID / Ring Group / Extension of SIP server / IP-PBX. *Note: Please configure ***Tel->IP Operation*** if you need this function.
To PSTN Hotline	When VoIP users make calls to this port, gateway will auto forward to dedicate number. The Hotline number could be mobile / fixed line number. Leave it blank if you don’t need this function. *Note: Please configure ***IP->Tel Operation*** if you need this function.
Auto-Dial Delay Time	The auto-dial delay time of hotline , the range is 0-10 seconds

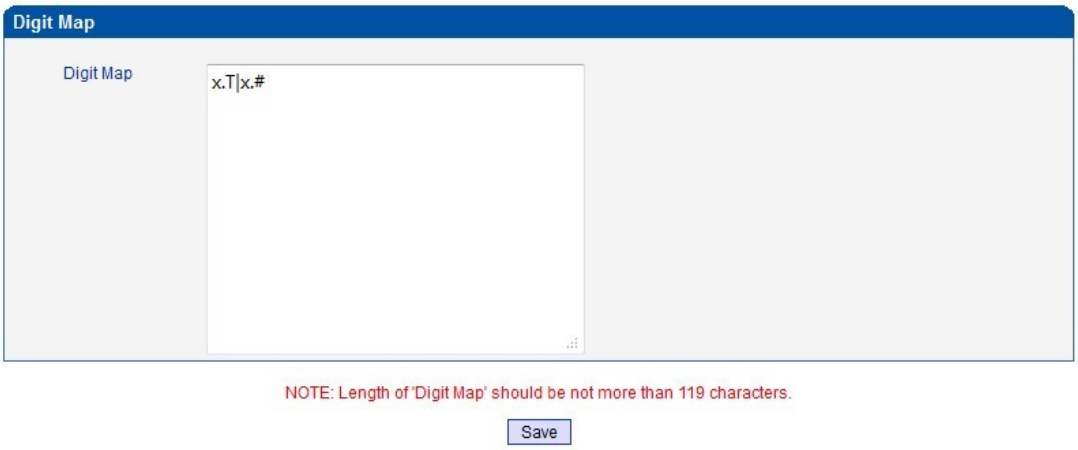
4.9.5 Port Group Configuration



Select ports for definited port group.



4.9.6 Digitmap



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.

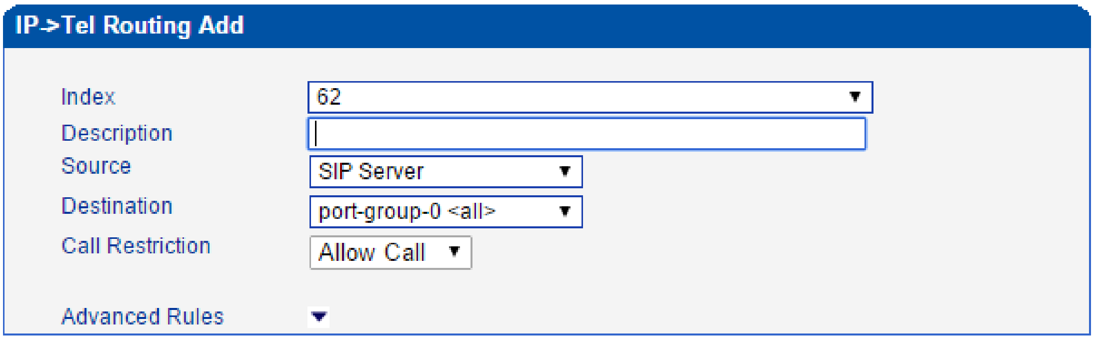
4.9.7 IP->Tel Routing

IP->Tel Routing

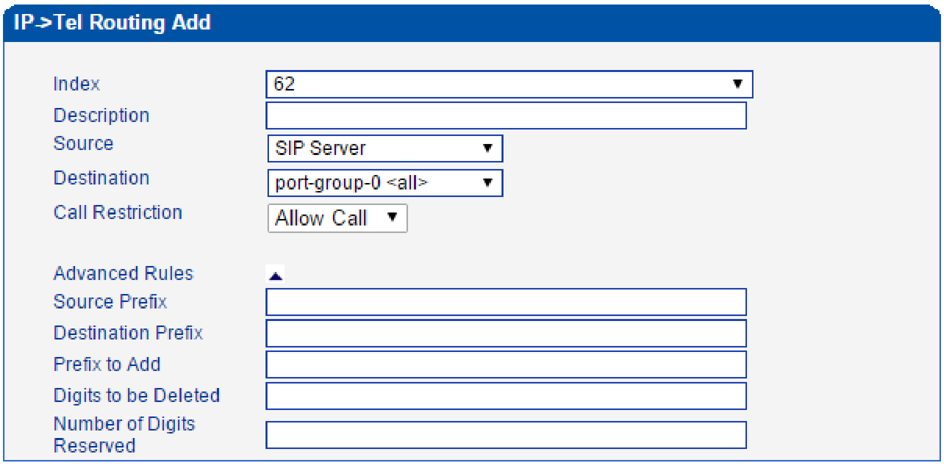
	Index	Description	Source	Destination	Call Restriction	Source Prefix	Destination Prefix	Prefix to Add	Digits to be Deleted	Number of Digits Reserved
☐	63	default	SIP Server	Port Group-0	Allow	--	--	--	--	--
☐	All									

Total: 1 entries 16 entries/Page 1 / 1Page Page 1

Add a new outgoing route rule, click Add button



Click ▼ to set caller and called prefix



Source: indicates call from which SIP server or SIP trunk

Destination: indicates call to which port or port group

Call Restriction: allow or forbid to call out

Source Prefix: to match with prefix of caller number

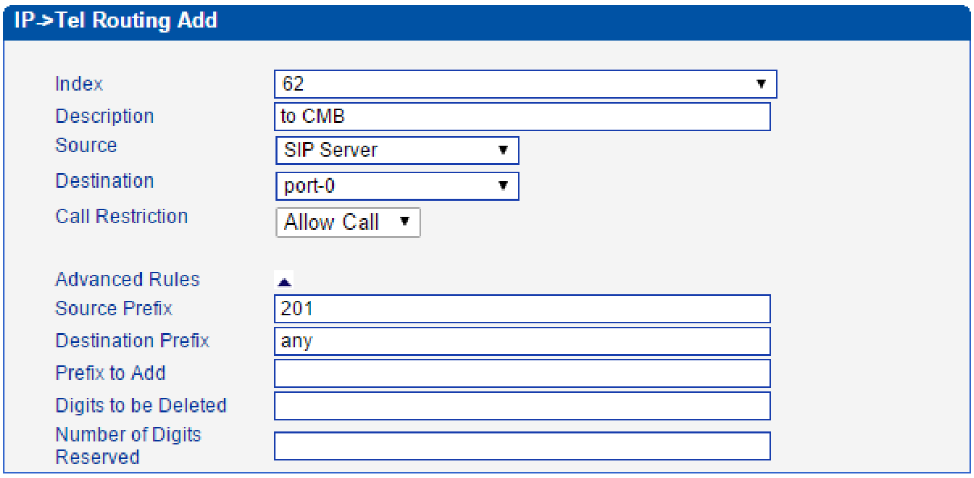
Destination Prefix: to match with prefix of called number

Prefix to add: to add a prefix in front of called number

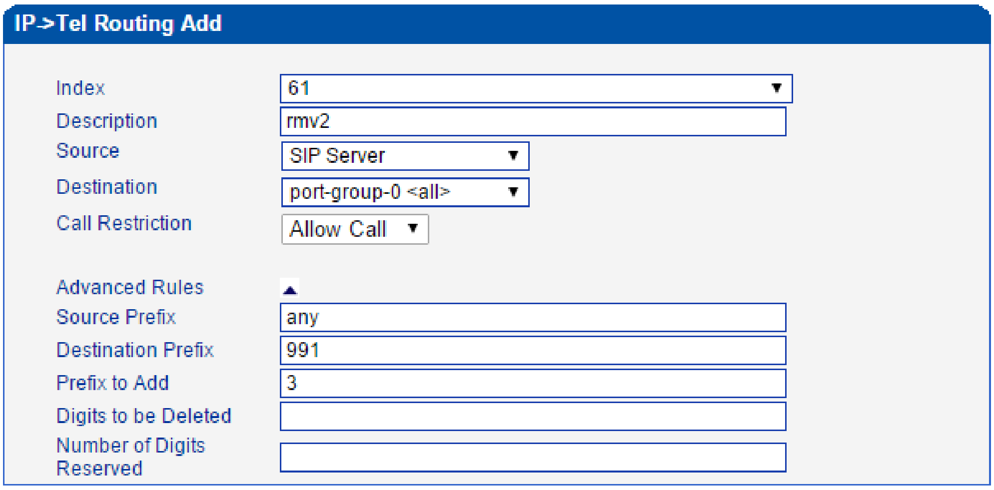
Digits to be deleted: indicates how many digits to be deleted for called number

Number of digits reserved: to definite the number of length of called number

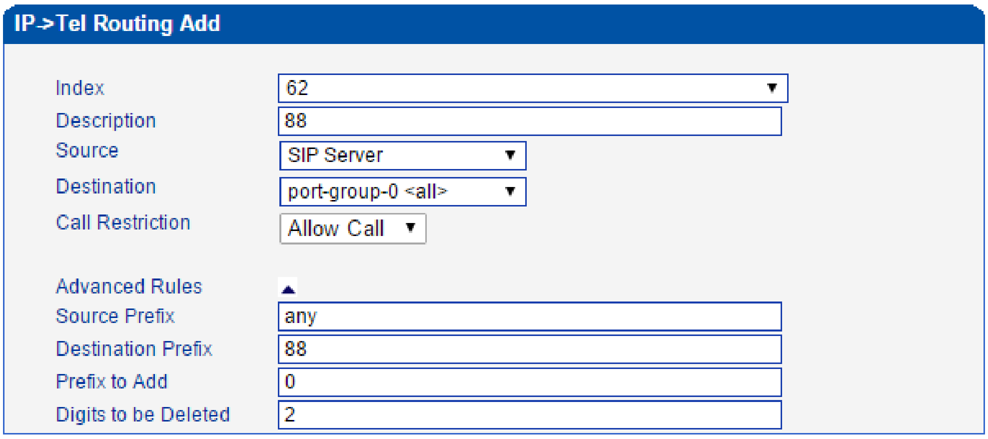
Examples:



Caller number 201 dial any number which will route to port 0.



Remove prefix 991 of called number.



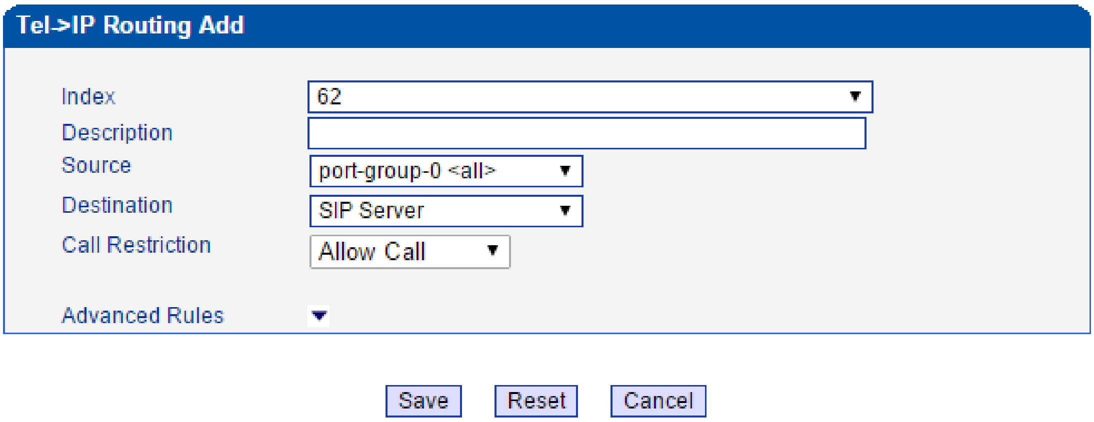
Remove prefix 88 and then add 0 in front of called number

4.9.8 Tel->IP Routing

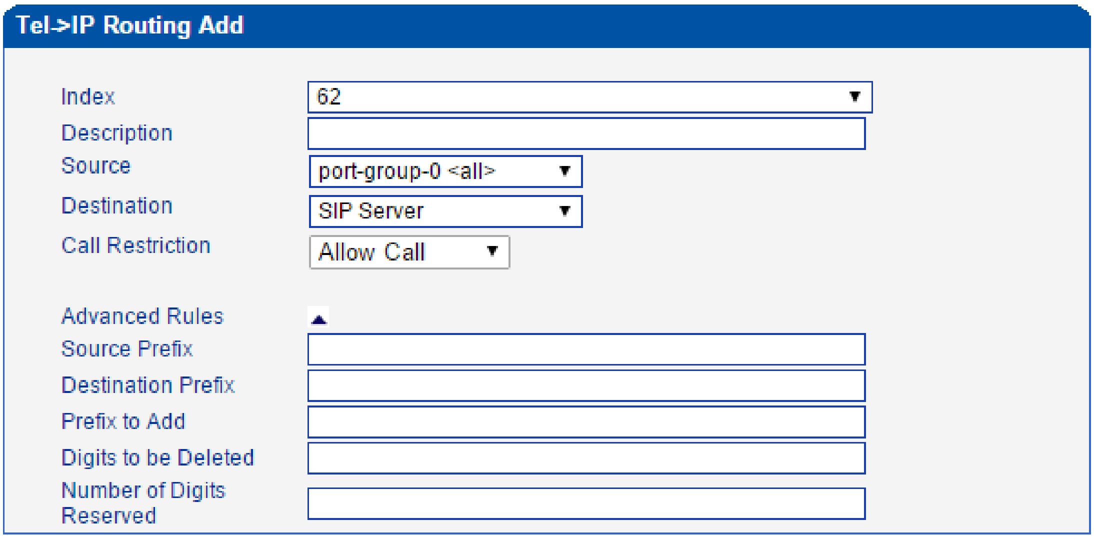
Tel->IP Routing

	Index	Description	Source	Destination	Call Restriction	Source Prefix	Destination Prefix	Prefix to Add	Digits to be Deleted	Number of Digits Reserved
	63	default	Any	SIP Server	Allow	--	--	--	--	--
	All									

Add a new incoming route rule, click Add button



Click ▼ to set caller and called prefix



Source: indicates call from which SIP server or SIP trunk

Destination: indicates call to which port or port group

Call Restriction: allow or forbid to call out

Source Prefix: to match with prefix of caller number

Destination Prefix: to match with prefix of called number

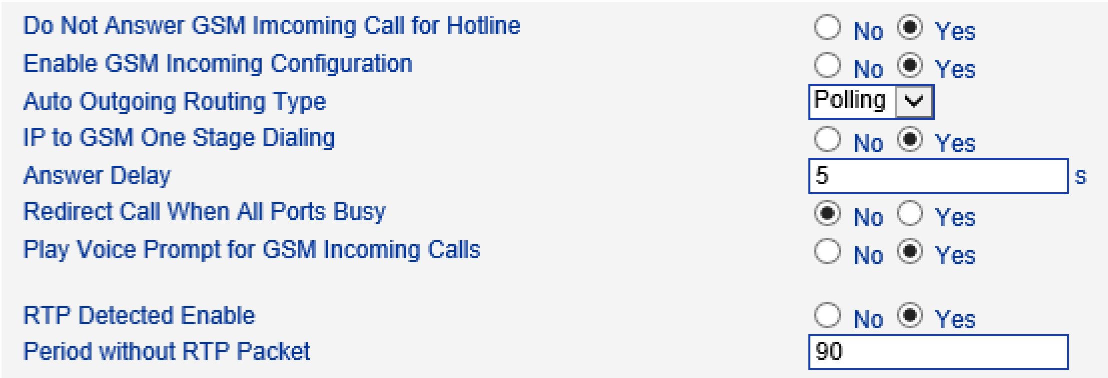
Prefix to add: to add a prefix in front of called number

Digits to be deleted: indicates how many digits to be deleted for called number

Number of digits reserved: to definite the number of length of called number

4.9.9 Service parameter

- **To configure dialing mode parameters**



- **Do Not Answer GSM Incoming Call for Hotline**

When the gateway get incoming call from mobile network, the modular will answer the call then start to DTMF or route to destination hotline number. While this option enabled, the modular won't answer the call but routing to destination hotline number till it getting answer.

Notes: Refer to *Port Parameter* page for *Hotline* configuration.

- **Enable PSTN Incoming Configuration**

Means when call from mobile side, you can dial the feature codes (***Chapter 3 Basic Operation***) to configure IP address and so on

- **Enable Auto Outgoing Routing**

Means when call out, whether by ordinal or polling pick to Select a Channel, this feature are generally used when use the same SIP User ID to register.

- **IP to PSTN One Stage Dialing**

The GSM/CDMA gateway support two dialing mode, one stage and two stage dialing. One stage dialing will obtain called number from *INVITE* message body, either *Request line* or *To* <SIP:xxxxx@host.com> field. Then deliver called number to GSM/CDMA directly.

But for two stage dialing, the SIP server must be dial the SIP channel account and then to generate DTMF to mobile network.

- **Answer Delay**

In most instances, Most of CDMA operators don't offer answer signal. The gateway doesn’t response SIP 200 OK to SIP server in case of missing answer signal from CDMA network. Answer delay is to fix this issue and generate SIP 200 OK to SIP server after answer delay timeout. Default value is 5 seconds. Moreover, it is available for CDMA gateway only.

- **Redirect Call when All Port Busy**

When the gateway is running heavy traffic and not possible to call out, the call will redirect to specific destination route as configuration.

IP and Port: destination gateway or IPPBX to be redirect

- **Play Voice Prompt for PSTN Incoming Calls**

Default setting is Yes. when the gateway receive incoming from mobile, it will play default/customized voice prompt to caller party. Default voice prompt is "Please dial the extension "; if set to No, the device will play dial tone instead of voice prompt.

- **RTP Detect**

This option is to disconnect call when there is no RTP received. Default value is 90s

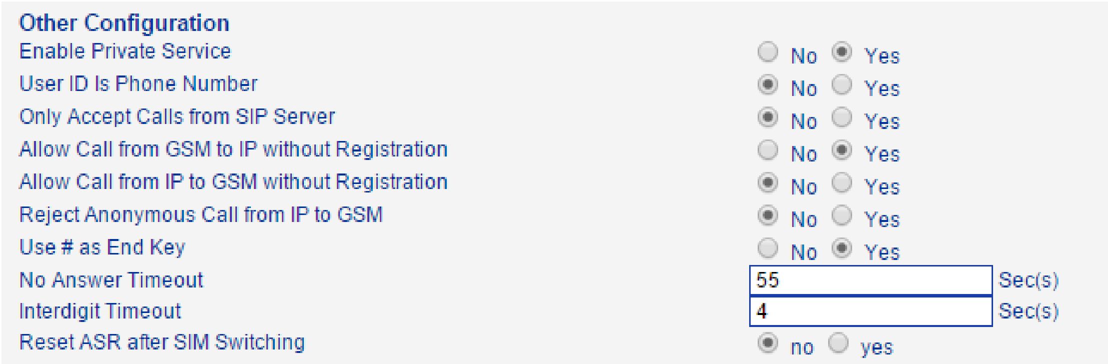
Nat Traversal

NAT Traversal	STUN	
Refresh Interval	0	Sec(s)
STUN Server IP		
STUN Server Port	3478	

Include Static NAT, Dynamic NAT and STUN

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

Other configuration



Enable Private Service

To enable local services like *158# etc.

User ID Is Phone Number

Default is No. user=phone will be added in SIP message body when this option enabled.

Only Accept Calls from SIP Server

Default is No. All calls will be rejected except calls from SIP server. IP Trunk will not work when this option enabled.

Allow Call from PSTN to IP without Registration

Refer to "SIP Configuration" -> "Is register”. If "Is register" setting is no, this option need set Yes, to avoid that the devices can not call in

Allow call from IP to PSTN without Registration

Refer to "SIP Configuration" -> "Is register”. If "Is register" setting is no, this option need set Yes ,to avoid that the devices can not call out

- **Reject Anonymous call from IP to PSTN**

  The incoming anonymous calls will be rejected

- **Use # as End Key**

  In General, SIP phones are based on # as the end, if this option is set to No, the dial-up will end expires dial-up time

- **Inter-digit Timeout**

  Timeout without dialing

4.9.10 Media parameter

Local Start RTP Port	8000
Enable Silence Suppression	○ No ◉ Yes
Call Progress Tone	USA ▼
Ring Back Tone	440,280,480,280,2000,40C
Busy Tone	480,330,620,330,500,500,(
Dial Tone	350,260,440,260,0,0,0,0

- **Local Start RTP Port**

  Means the initial port when RTP voice stream transmit in the IP network, in general, using the factory default values. When there are several DINSTAR units are deployed and they are in the same network or behind the same NAT, user can try to change it to avoid NAT traversal issue;

- **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

  USA Standard:

Ringback Tone: 440,280,480,280,2000,4000,0,0 frequency: 440/480Hz on:2000ms off:4000ms

Busy Tone: 480, 330, 620, 330, 500, 500, 0, 0 frequency: 480/620Hz, on: 500ms off: 500ms

- **DTMF Parameter**

DTMF Parameter	
DTMF Method	RFC2833 ▼
RFC2833 Payload Type	101
DTMF Volume	0dB ▼
DTMF Interval	200 ms

UC2000-VE/F/G support RFC2833 and SIGNAL two ways. DTMF INTERVAL range is 50 ~ 800ms, DTMF VOLUME can use the default Configuration

- **System IVR**

IVR Parameter	
Play IVR for GSM Incoming Calls	○ No ◉ Yes
IVR Play Duration	25 Sec(s)
Play IVR Voice Prompt from	◉ Default ○ Custom

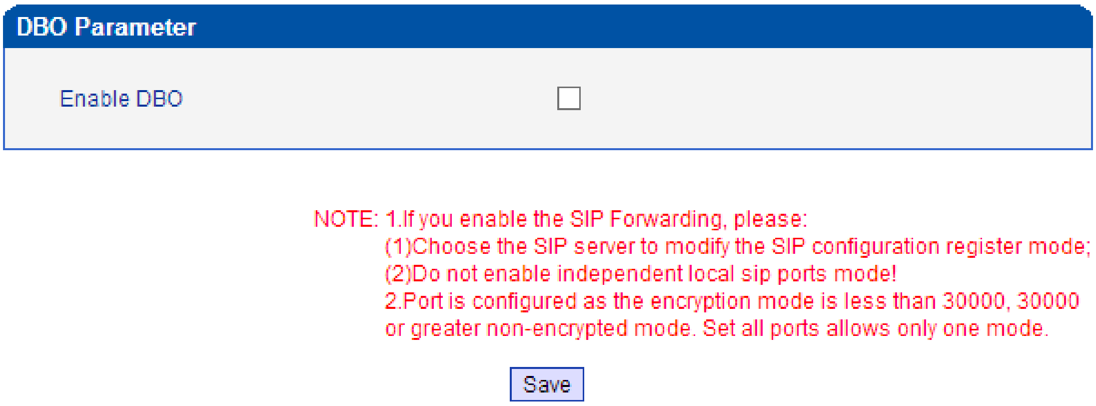
While you make call to SIM card of GSM gateway, you will hear default IVR prompts or customized IVR.

- **Configure codec list**

	Coder Name	Payload Type	Packetization Time(ms)	Rate(kbps)
1	G.729AB ▼	18	20 ▼	8
2	PCMU ▼	0	20 ▼	64
3	PCMA ▼	8	20 ▼	64
4	G.723.1 ▼	4	60 ▼	6.3

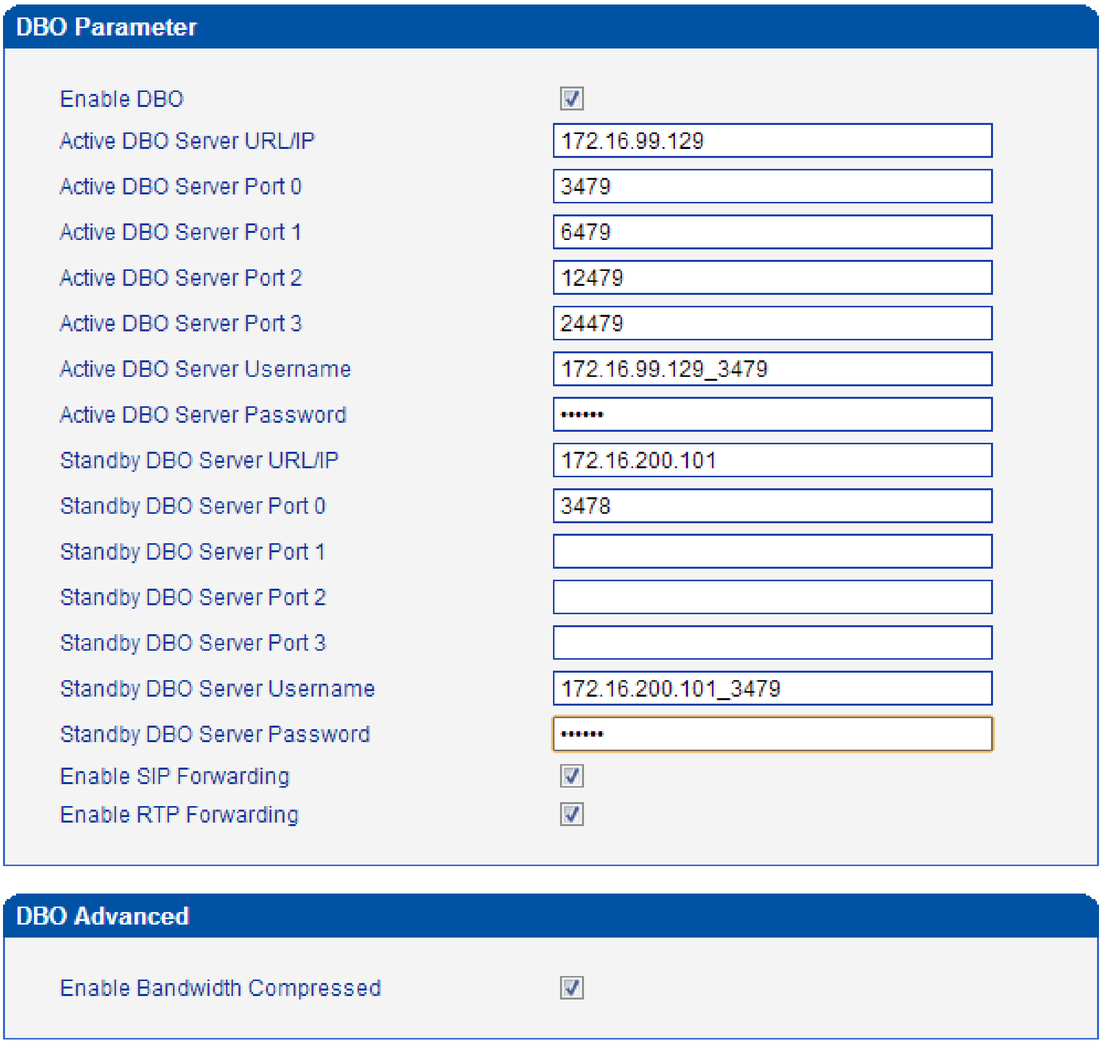
4.9.11 DBO Parameter

Enable DBO service



Configure DBO parameter

More parameter showing on the interface after enable DBO, the main interface as below:



Parameter Description:

Parameters	Description
Active DBO Server URL/IP	Primary DBO server IP or domain for traffics
Active DBO Server Port	DBO service ports that dedicate by DBO server. There are 4 ports definite in the DBO server by default, 3479, 6479, 12479 and 24479, any one of this 4 ports will work with the DBO server.
Active DBO Server Username	The authenticate username which provide by DBO server. The gateway will not allow to pass the traffics if the username and password doesn't match with the server. The username with the format as x.x.x.x_3479 by default. x.x.x.x is the IP of DBO server.
Active DBO Server Password	The authenticate password which provide by DBO server. The gateway will not allow to pass the traffics if the username and password match with the server.
Standby DBO Server URL/IP	Secondary DBO server IP or domain.
Standby DBO Server Port	DBO service ports that dedicate by DBO server. There are 4 ports definite in the DBO server by default, 3479, 6479, 12479 and 24479, any one of this 4 ports will work with the DBO server.
Standby DBO Server Username	The authenticate username which provide by DBO server. The gateway will not allow to pass the traffics if the username and password match with the server. The username with the format as x.x.x.x_3479 by default. x.x.x.x is the IP of DBO server.
Standby DBO Server Password	The authenticate password which provide by DBO server. The gateway will not allow to pass the traffics if the username and password match with the server.
Enable SIP Forwarding	Enable SIP signaling encryption and forward by DBO server. The SIP signaling will forward by

	DBO server after this option enable.
Enable RTP Forwarding	Enable RTP encryption and forward by DBO server. The RTP will forward by DBO server after this option enable.
Enable Bandwidth Compressed	Enable bandwidth saving function. This feature works after uploading proper license.

4.10 HUMAN BEHAVIOR

4.10.1 Overview

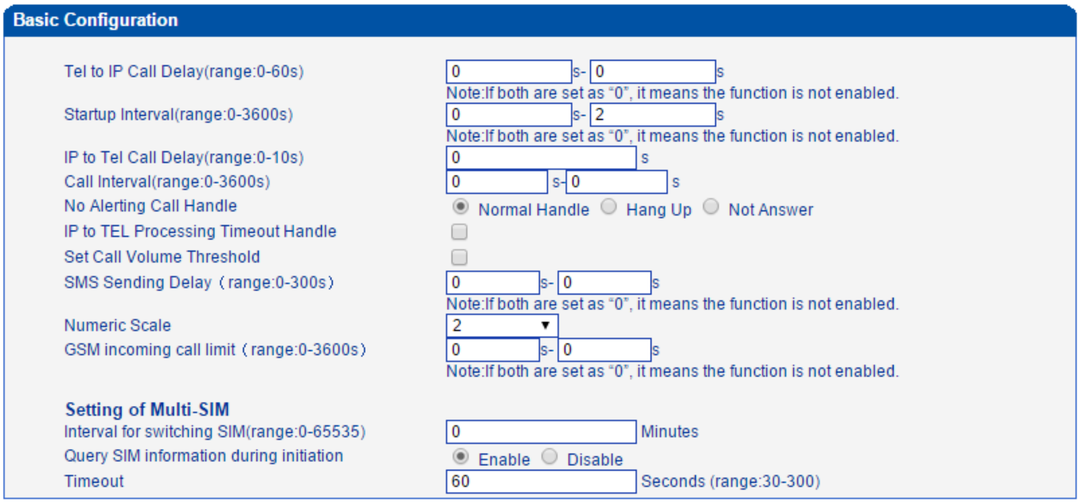
On the **Overview** interview, you can see the number, last matched balance (the balance that is assigned last time), calculated balance (the remaining balance), remaining total credits and remaining daily credits of a SIM card.

Overview

	SIM	Phone Number	Last Matched Balance	Calculated Balance	Total Credits	Daily Credits	Daily Calls
☐	0	546464546	---	---	---	---	---
☐	1		---	---	---	---	---
☐	2		---	---	---	---	---
☐	3		---	---	---	---	---
☐	4		---	---	---	---	---
☐	5		---	---	---	---	---
☐	6		---	---	---	---	---
☐	7		---	---	---	---	---
☐	All		Clear	Clear	Reset	Reset	Reset

4.10.2 Basic Configuration

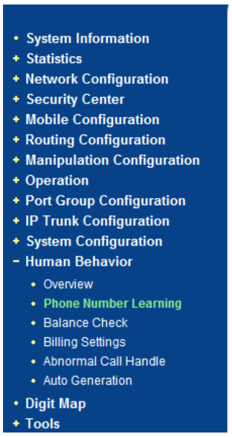
On the **Basic Configuration** interface, you can set how long an IP →Tel call or a Tel→IP call will be delayed, as well as call interval. The 'set call volume threshold function' is mainly used for anti-blocked (such as some operators launched special call testing for the detection of the VoIP equipment, call volume may is mute or great noise) .

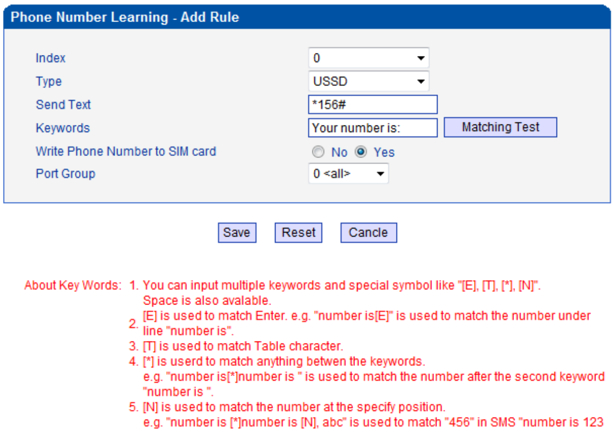


4.10.3 Phone Number Learning

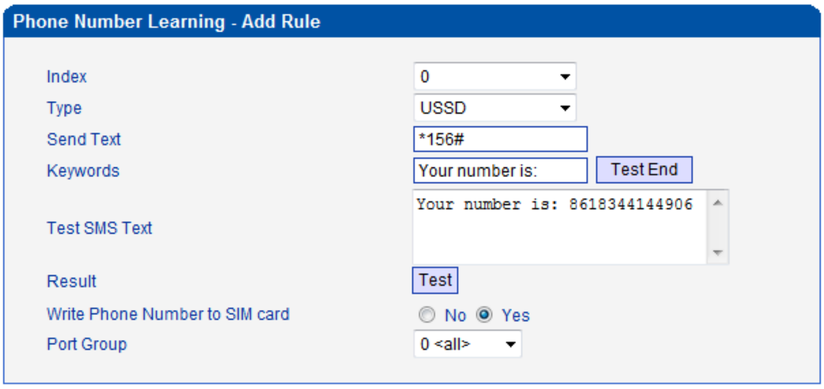
If you want to learn the SIM card number and used for auto call. The GSM gateway provide 3 modes to learn SIM card number: USSD/SMS/Call.

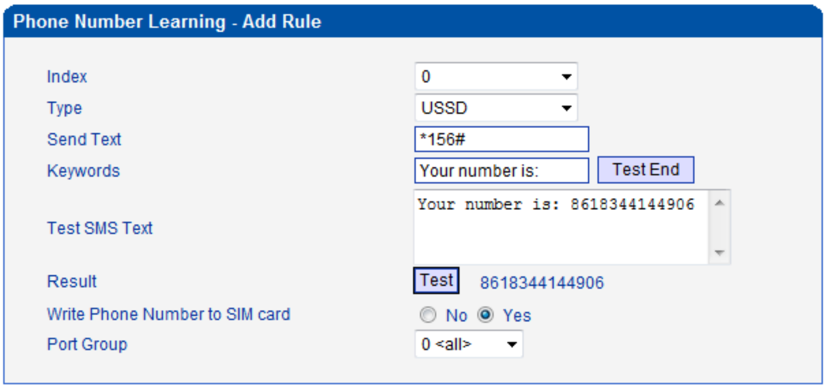
1) USSD. Send USSD to carrier and get the response. For example, send ***156#**, get response: "Your number is: 8618344144906". So, configured the Keywords to "Your number is:", the gateway will take the number **8618344144906**.





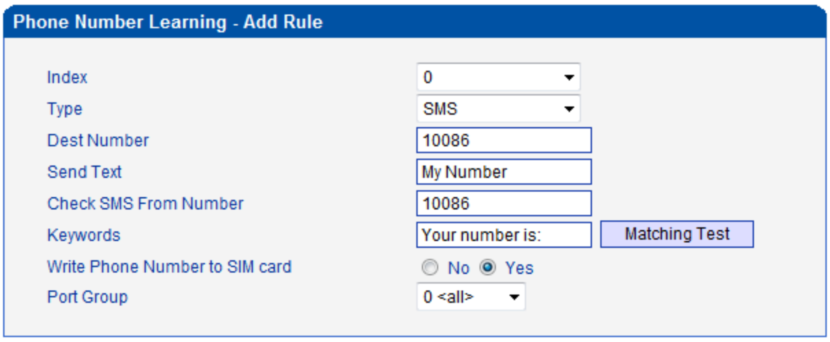
For make sure the configuration work, we can use the Matching Test. Input the "Your number is: 8618344144906" at Test SMS Text, press the Test, you will get the match result.



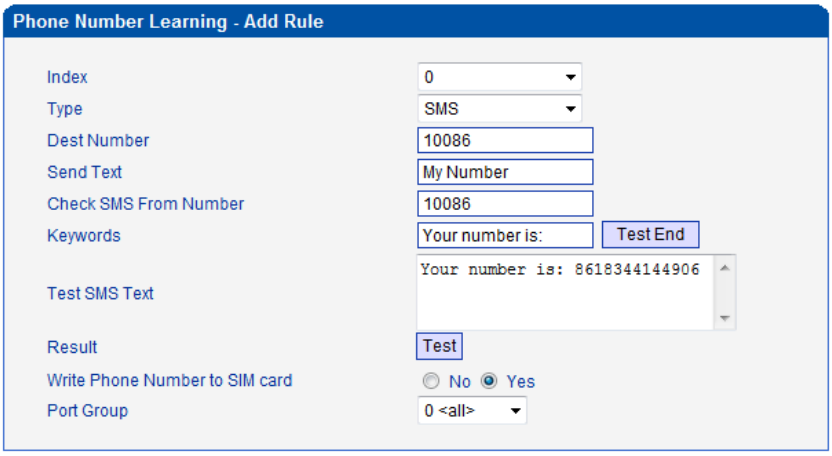


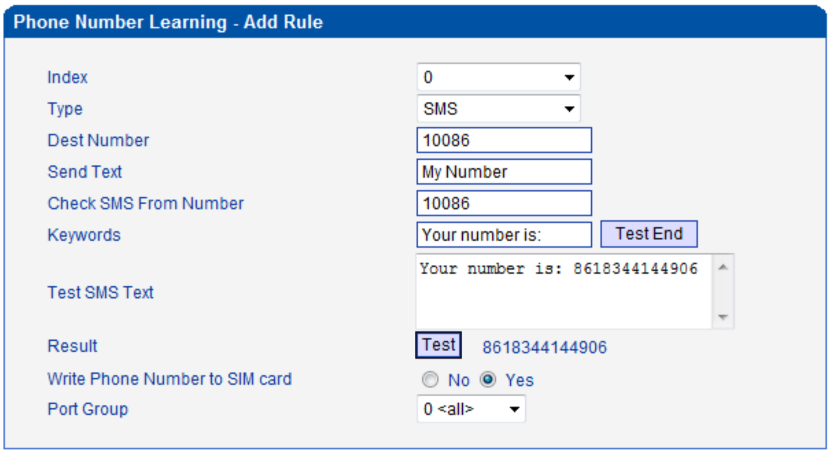
2) SMS.

Send SMS to carrier and get the response. For example, send SMS “My Number” to **10086**, the carrier reply SMS: “Your number is: 8618344144906”. So, configured the Dest Number to **10086**, the Send Text to “My Number”, the Check SMS From Number to **10086**, the Keywords to “Your number is**:”**, the gateway will take the number **8618344144906**.



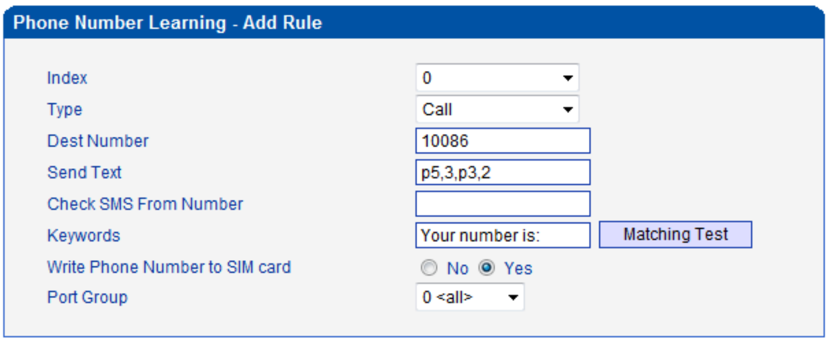
For make sure the configuration work, we can use the Matching Test. Input the “Your number is: 8618344144906” at Test SMS Text, press the Test, you will get the match result.



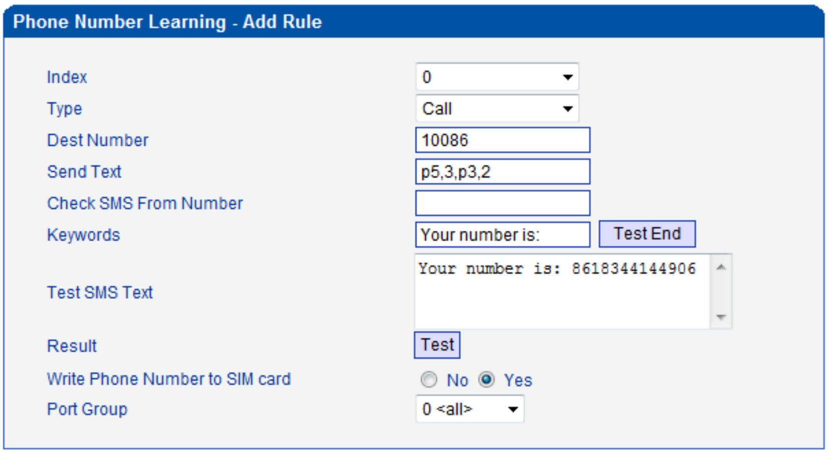


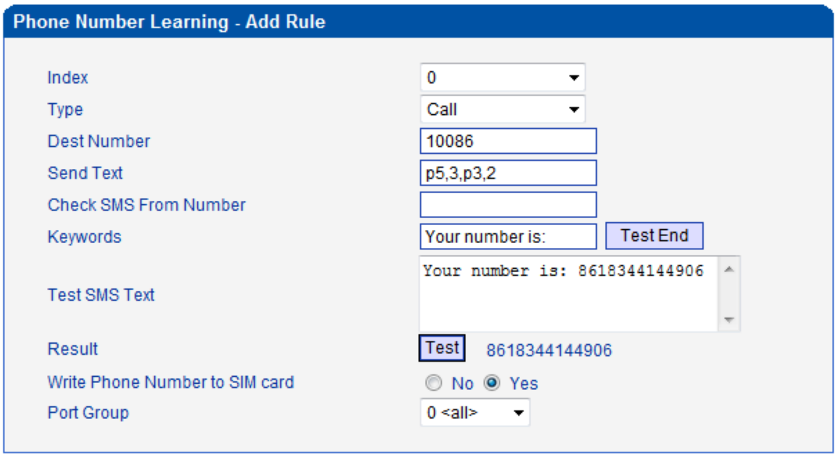
3) Call.

Call to carrier and get the response. For example, call **10086**, after call connected, it will play IVR “welcome to use China Mobile, recharge, press 1; check balance, press 2; other services, press 3 ...” press **3**, it will play IVR “check current package, press 1; check phone number, press 2;…”, press **2**, the carrier reply MSG: “Your number is: 8618344144906”. So, configured the Dest Number to **10086**, the Send Text to **p5,3,p3,2** that means after call connected wait 5s, then press 3, then wait 3s, then press 2. the Check SMS From Number to Null, the Keywords to “Your number is:”, the gateway will take the number **8618344144906**.



For make sure the configuration work, we can use the Matching Test. Input the “Your number is: 8618344144906” at Test SMS Text, press the Test, you will get the match result.

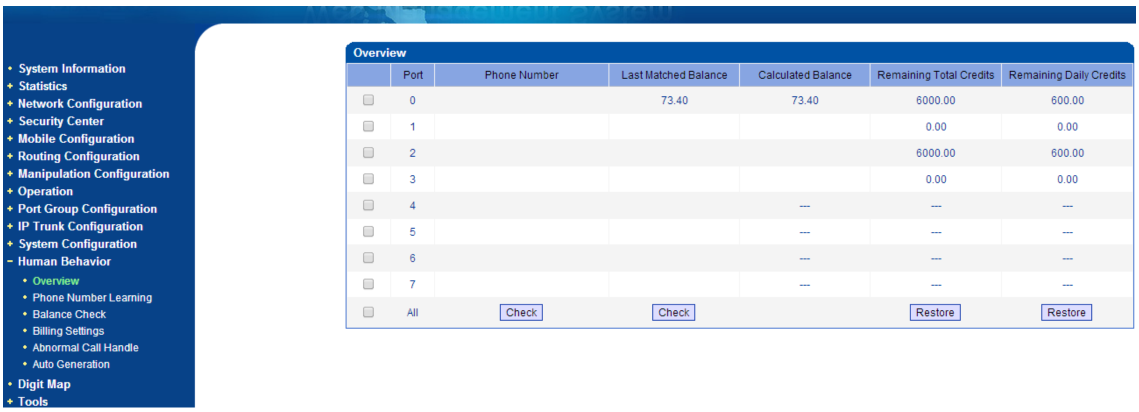




4.10.4 Balance Check

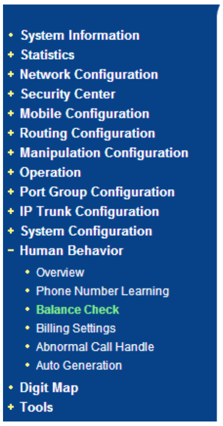
On the **Balance Check** interface, you can check the balance of a SIM card.

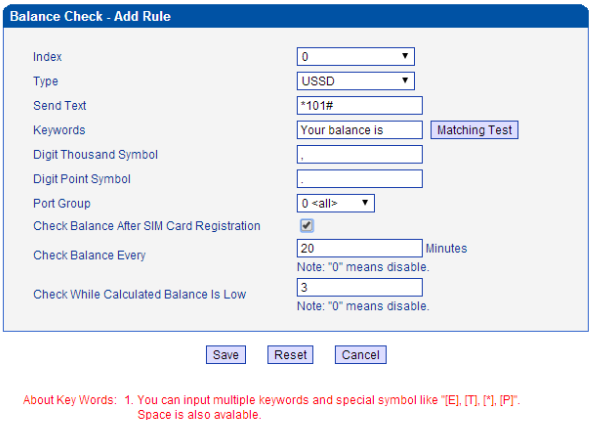
If you want to check balance automatically and block SIM card when it is low balance. The UC2000 have 3 modes to check balance: USSD/SMS/Call.



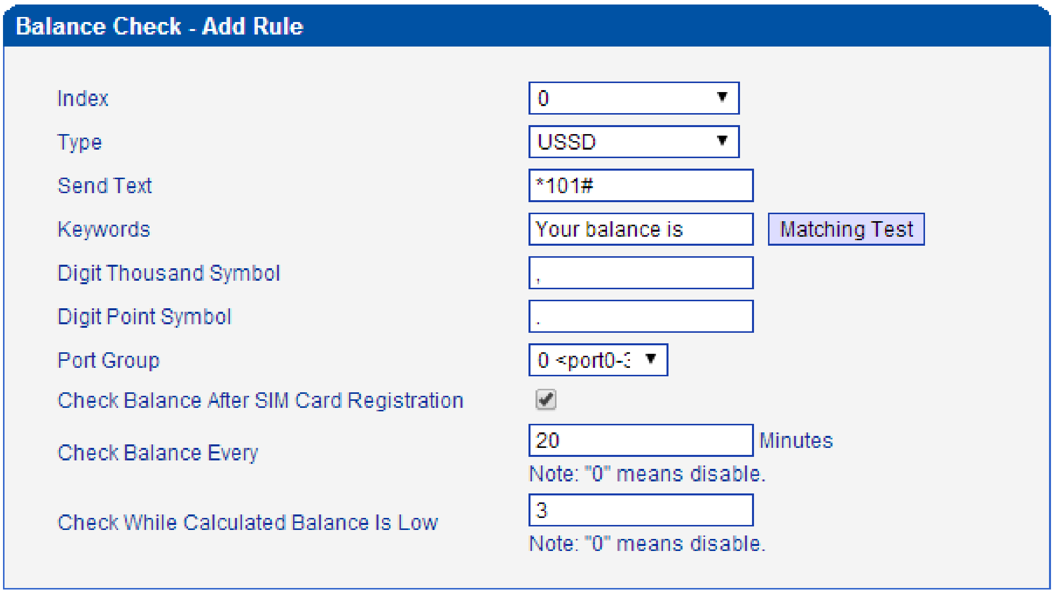
1) Check balance by USSD

Send USSD to carrier and get the response. For example, send *101#, get response: "Your balance is 73.40$". So configured the Keywords to "Your balance is", the gateway will take the number 73.40.

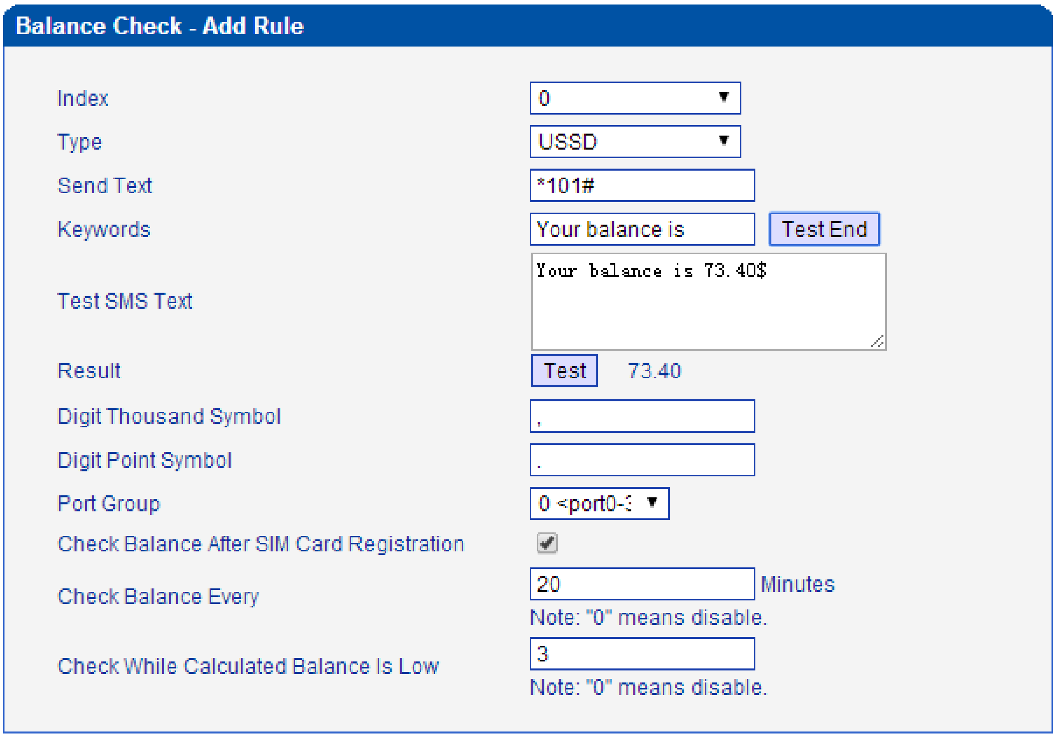




For make sure the configuration work, we can use the Matching Test. Input the "Your balance is 73.40$" at Test SMS Text, press the Test, you will get the match result.

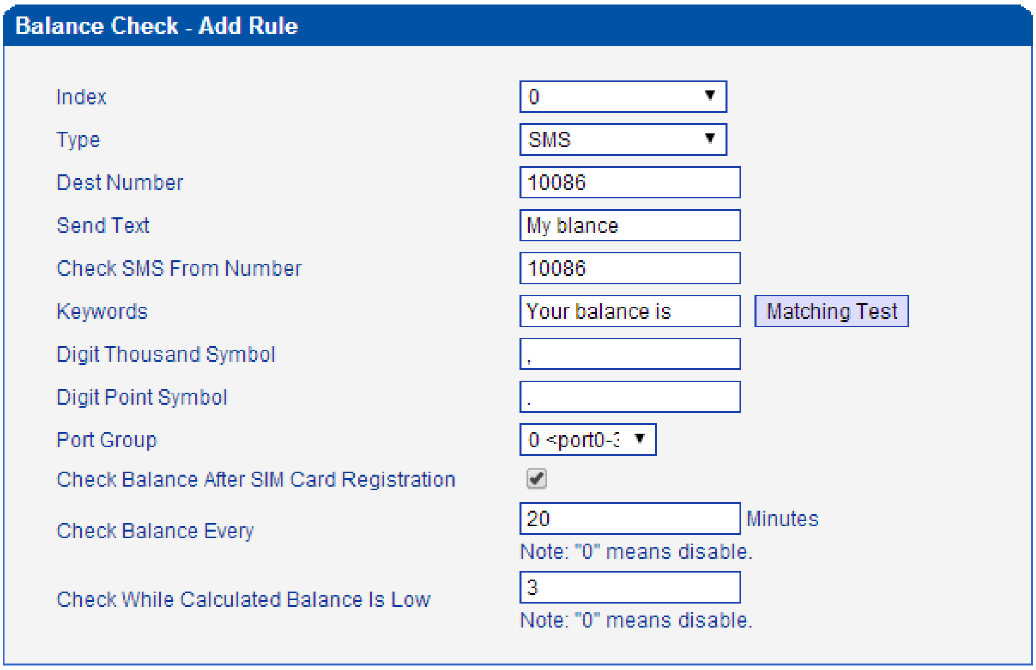


(next page)

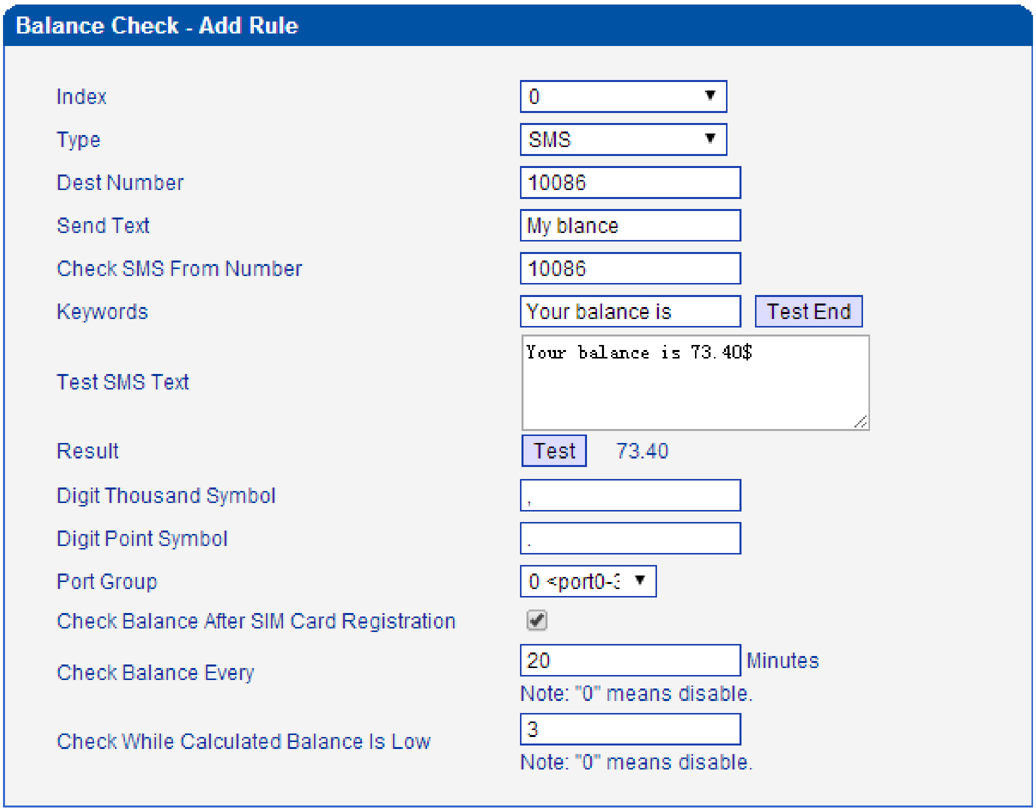


2) Check balance by SMS.

Send SMS to carrier and get the response. For example, send SMS "My balance" to **10086**, the carrier reply SMS: "Your balance is 73.40$". So configured the Dest Number to 10086, the Send Text to "My balance", the Check SMS From Number **10086**, the Keywords to "Your balance is**"**, the gateway will take the number 73.40.

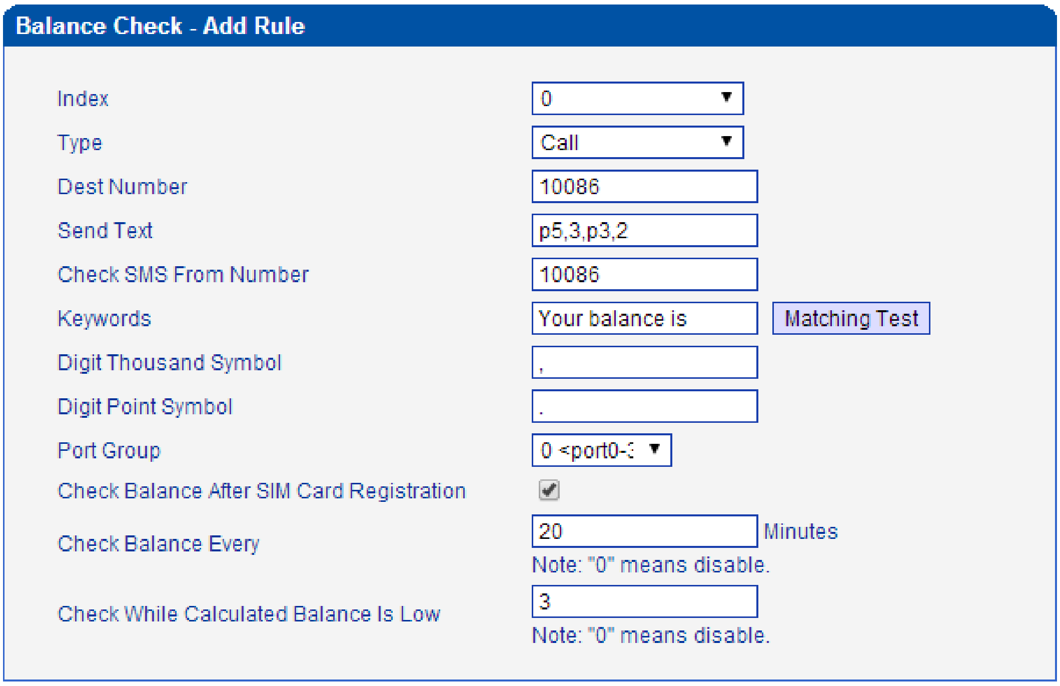


For make sure the configuration work, we can use the Matching Test. Input the "Your balance is 73.40$" at Test SMS Text, press the Test, you will get the match result.

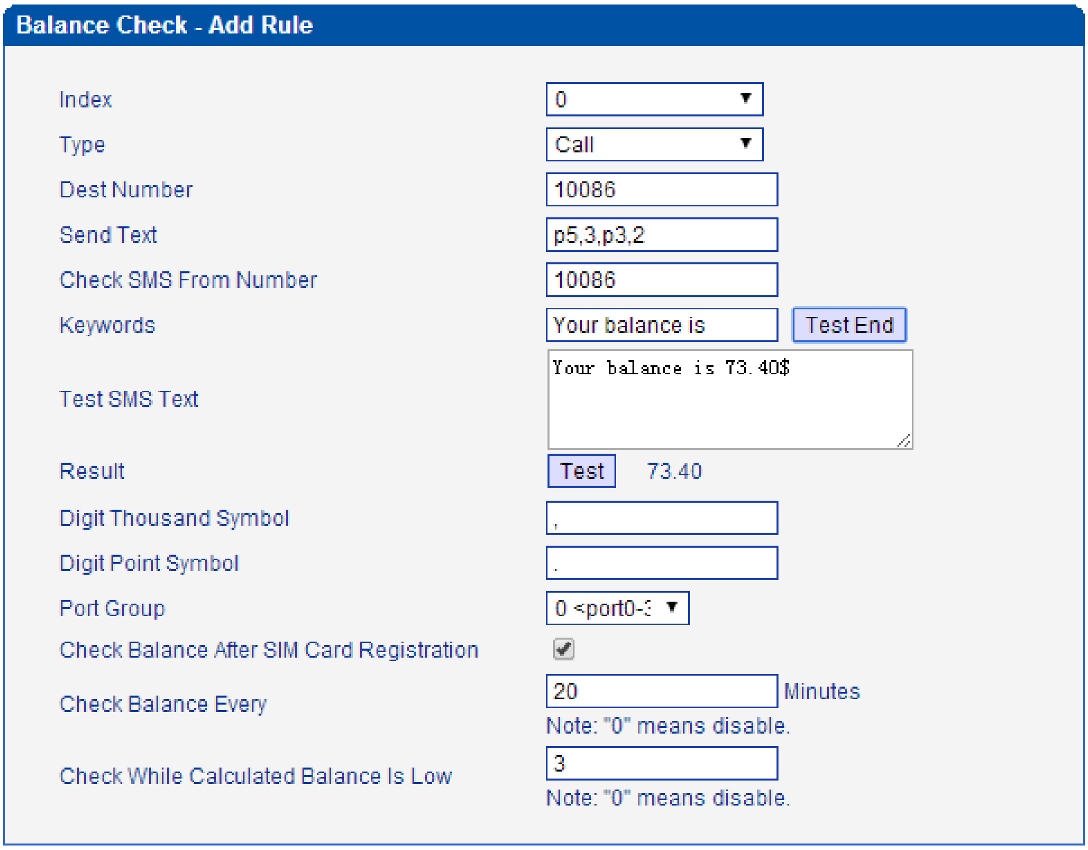


3) Check balance by Call.

Call to carrier and get the response. For example, call **10086**, after call connected, it will play IVR "welcome to use China Mobile, recharge, press 1; check phone number, press 2; other services, press 3 ..." press **3**, it will play IVR "check current package, press 1; check balance, press 2;…", press **2**, the carrier reply MSG: "Your balance is 73.40$". So, configured the Dest Number to **10086**, the Send Text to **p5,3,p3,2** that means after call connected wait 5s, then press 3, then wait 3s, then press 2. the Check SMS From Number to Null, the Keywords to "Your balance is**"**, the gateway will take the number 73.40.

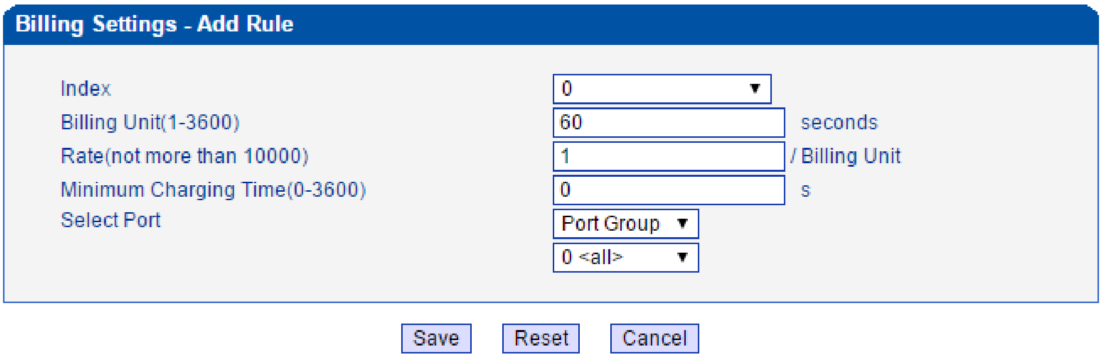


For make sure the configuration work, we can use the Matching Test. Input the "Your balance is 73.40$" at Test SMS Text, press the Test, you will get the match result.



4.10.5 Billing setting

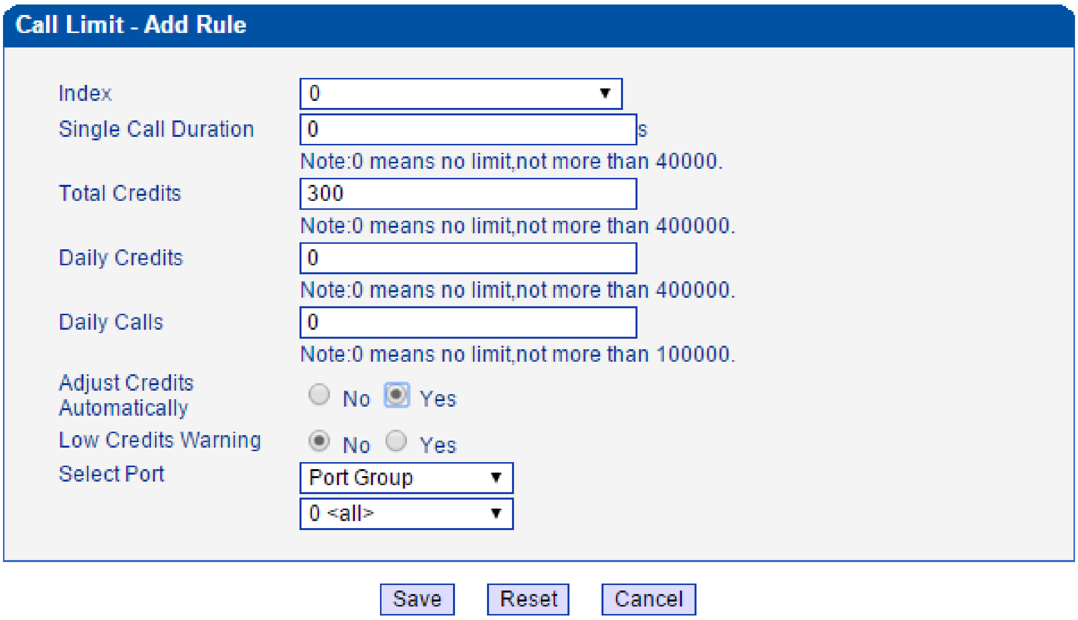
Billing setting mainly use to limit call time of SIM cards, see also call limit.



Minimum Charging Time: set minimum charging time, some operator does not charge if the call is less than some seconds when call is connected, user can set that value here. If the operator starts billing once the call is connected, please set 0 here.

In this example: set 1$ per 60s for port group 0.

4.10.6 Call limit



Single Call Credits: set single call credits, it defines the maximum credit every single call can take, 0 means no limit. If you set 2, it means every call can use 2 credits at most, and call will be disconnected if gets the limit.

Total Credits: set total credits, it defines the maximum credit the port can use, 0 means no limit. If you set 600, it means the port can use 600 credits at most.

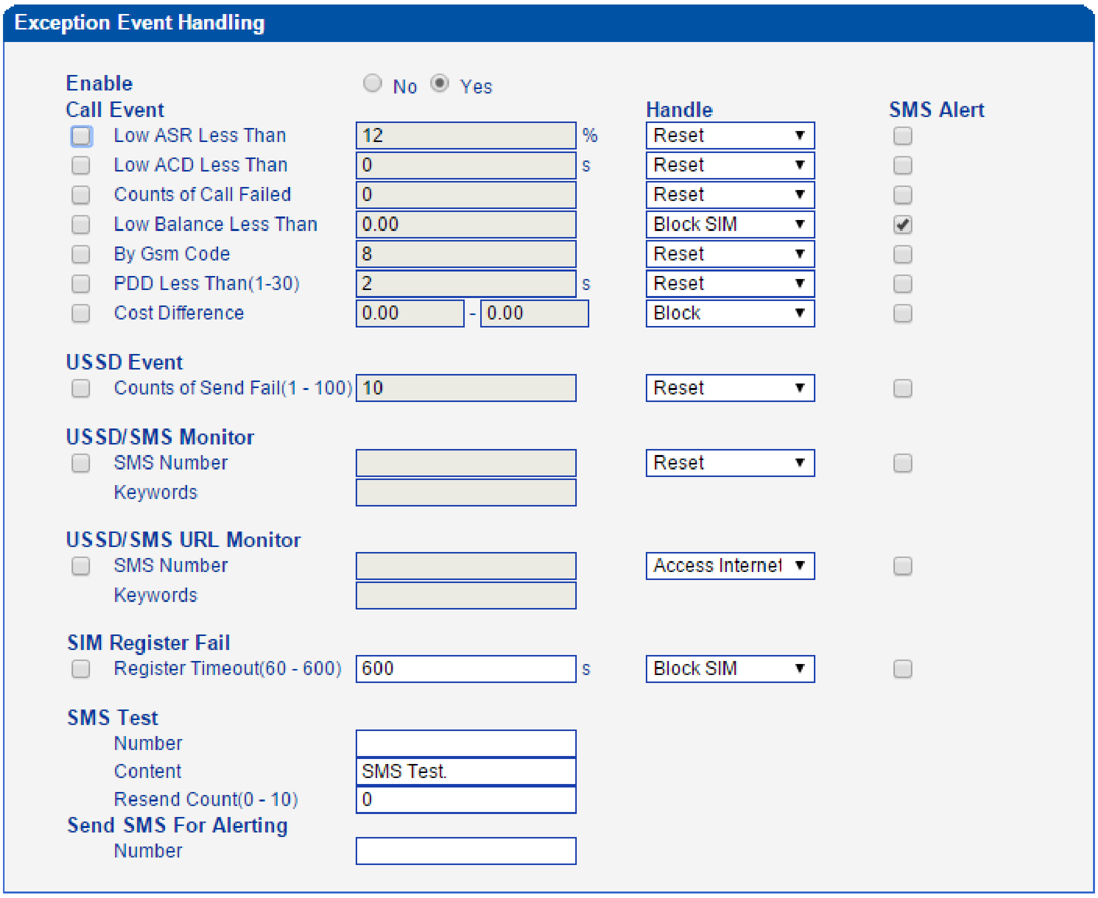
Daily Credits: set daily credits, it defines the maximum credit the port can use in one day, 0 means no limit. If you set 60, it means the port can use 60 at most one day, and the data will be cleared at 0'clock of everyday.

Adjust Credits Automatically: If enable adjust credits automatically or not, Yes means enable, No means disable. This option is used to work together with Balance Check function, when enable both balance check function and billing, the gateway will automatically regulate the balance.

In this case, billing unit = 1$/60s, total credits = 300

Call limitation = 300/1 = 300 minutes

4.10.7 Exception Event Handling



Call Event

1. Definitions

For the purpose of the present document, the following terms and definitions apply:

ACD: The **Average Call Duration** is a measurement in telecommunications that reflects an average length of telephone calls transmitted on telecommunication networks.

ASR: The **Answer-seizure ratio** is a call success rate in telecommunications; it is the percentage of answered telephone calls with respect to the total call volume.

CDR: The **Call Detail Record** is a data record produced by a telephone exchange or other telecommunication equipment that documents the details of a telephone call that passes through the facility or the device.

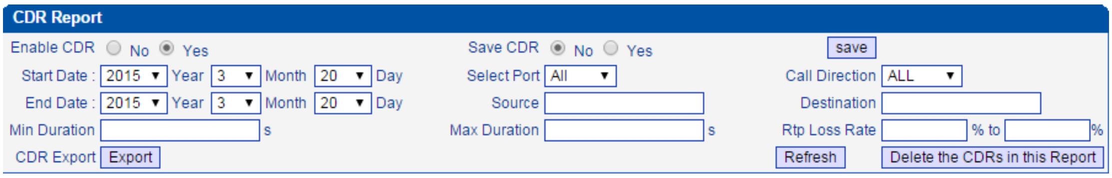
2. Configurations

Low ASR Handling

The ASR is equal to: the answered call, divided by the total attempts of calls. That is, ASR = answered call/total attempts of calls. To calculate the ASR, the gateway checks the CDRs. Because the CDRs on the gateway is disabled by default, you need to enable the CDR before you apply the Low ASR handling.

3. Enable CDRs on the gateway

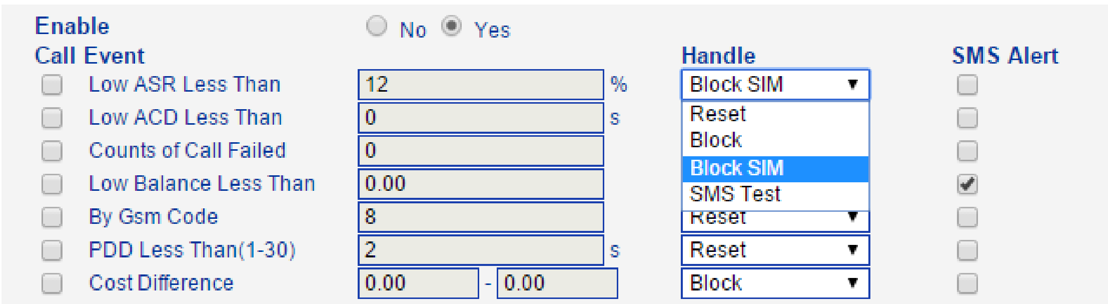
Open the web of the gateway, and then click “Statistics” and “CDR Report”. Then enable the CDR as the below figure shows:



Don’t forget to click “save” after selecting “Yes” on Enable CDR.

4. Configure the Low abnormal call handle

Click “Human Behavior” and “exception event handle”, then select “yes”, the configuration page will be displayed:

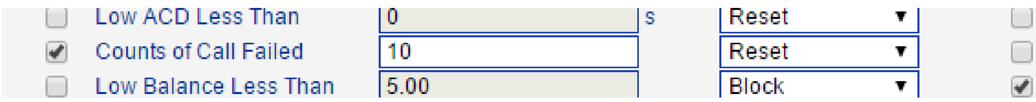


Low ASR Less Than: This value is the threshold of the ASR, once the exact ASR is lower than this value, the UC2000 port will be considered to be low ASR.

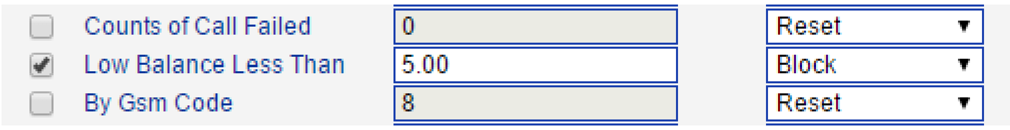
Low ACD Less Than: define the low ACD value threshold once the exact ACD is lower than this value, the UC2000 port will be considered to be low ACD.

Counts of Recent Call: This value define how many recent calls will be counted to calculate the ASR/ACD.

Counts of failed calls: This value define how many failed calls. This feature is used to detect the failure calls, once there are certain counts of call failure consecutively, the gateway port will be considered abnormal.



Low Balance Than: define the low balance value threshold .To apply the Low Balance Handle, it is required to configure the Balance check properly; please refer to the FAQ of balance check for more details.



GSM Network side error code handle

When the gateway makes an outgoing call, GSM network side will respond a code which indicates the cause of the call of failure; gateway will record these error codes until the gateway was restarted.

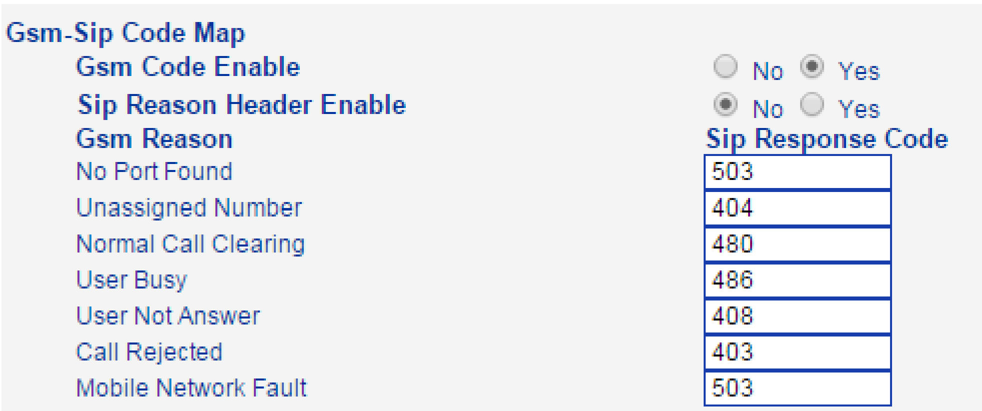
The error code 8, meaning of “Operator determined barring”, indicates precisely that the SIM was blocked by operator; so we provide this feature to detect the error code and then blocked gateway module.

Follow these steps to use this feature:

a) Enable the error code record

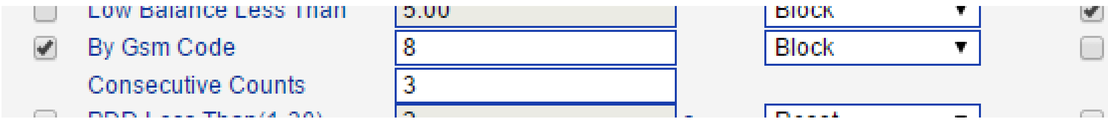
The GSM network side error code record is disabled by default, you need to enable the record before you use this feature.

Click “System Configuration” and “SIP Parameter”, then select “yes” for “GSM-Sip Code Map GSM Code Enable”.



Don’t forget to save the configuration.

b) Configure the GSM code monitor

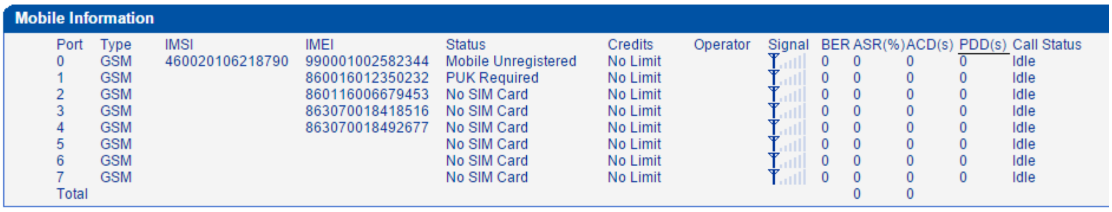


By GSM Code: The GSM network side error code

Counts of consecutive GSM Code: The counts of the error code consecutively.

As the figure shows above, once the GSM error code 8 is detected, the gateway will blocked the gateway module.

PDD Less Than(1-30): define the value of abnormal PDD. You can check PDD value under system information page.



Mobile Information

Port	Type	IMSI	IMEI	Status	Credits	Operator	Signal	BER	ASR(%)	ACD(s)	PDD(s)	Call Status
0	GSM	460020106218790	990001002582344	Mobile Unregistered	No Limit			0	0	0	0	Idle
1	GSM		860016012350232	PUK Required	No Limit			0	0	0	0	Idle
2	GSM		860116006679453	No SIM Card	No Limit			0	0	0	0	Idle
3	GSM		863070018418516	No SIM Card	No Limit			0	0	0	0	Idle
4	GSM		863070018492677	No SIM Card	No Limit			0	0	0	0	Idle
5	GSM			No SIM Card	No Limit			0	0	0	0	Idle
6	GSM			No SIM Card	No Limit			0	0	0	0	Idle
7	GSM			No SIM Card	No Limit			0	0	0	0	Idle
Total									0	0		

Handle abnormal event

Once one of the above abnormal condition is detected, gateway could:

Reset the specified GSM module

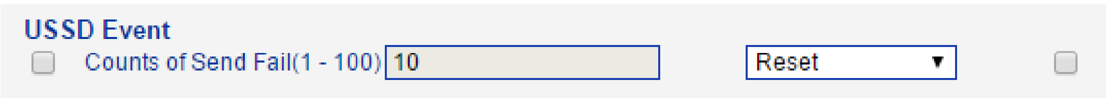
Block the specified GSM module

Block the SIM, this setting only available while remote SIM mode is in using or multiple SIM device.

SMS Test, send a SMS through specific port to verify if the SIM card works properly

Sending SMS to a phone number for alerting, this is optional.

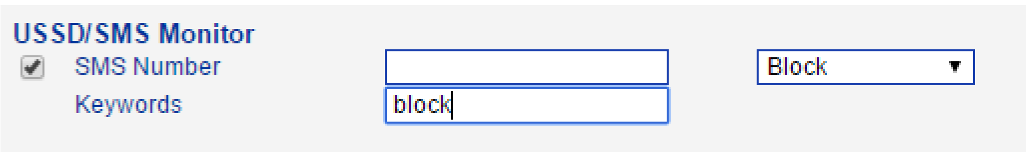
USSD Event



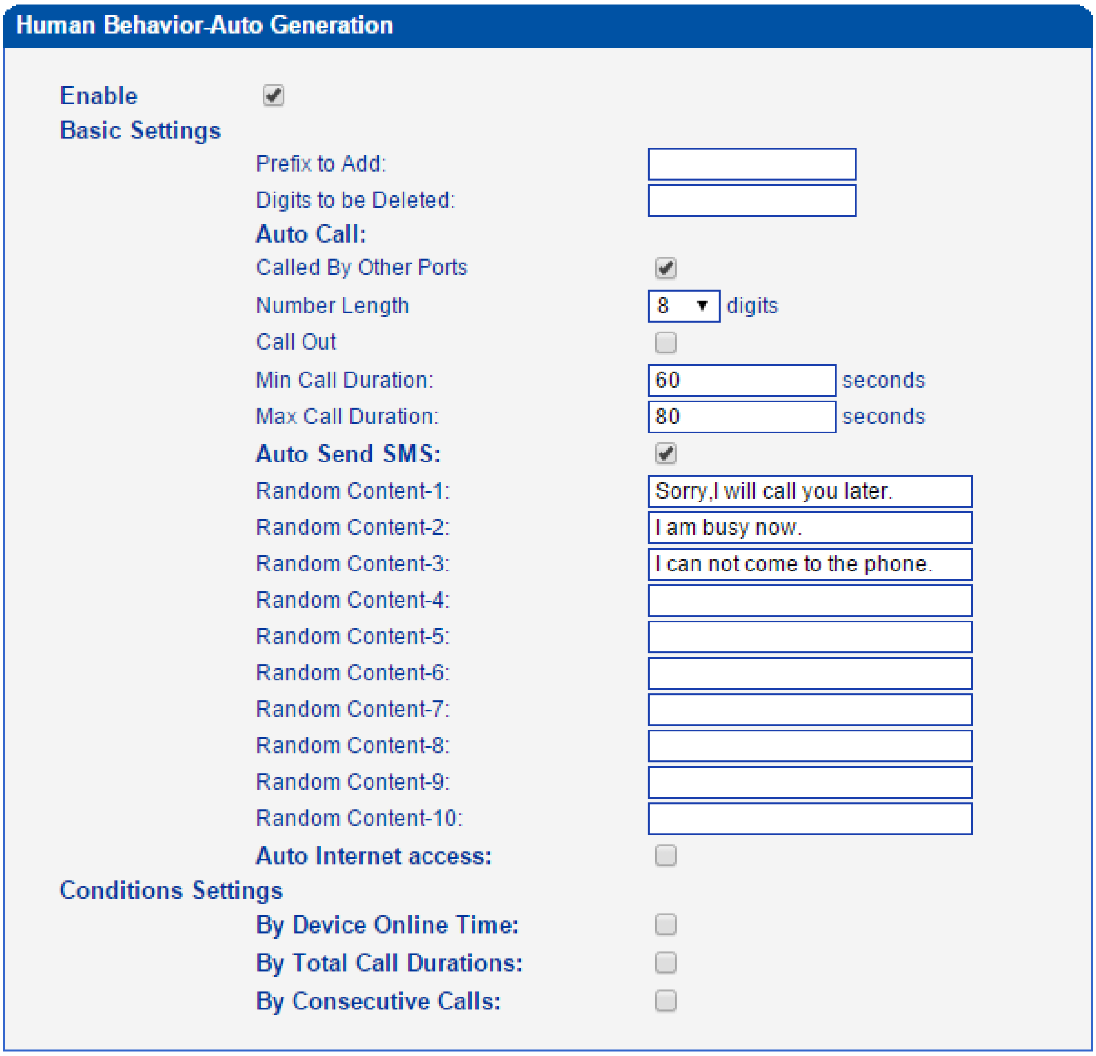
Reset module/block Port/Block SIM card in case of USSD failed more than defined value threshold.

USSD/SMS Monitor

This parameter be used to Monitor SMS/USSD response contents, which helps gateway to know SIM card is blocked.



4.10.8 Auto generation



Auto Generation mainly used to make calls and SMS between SIM cards which in same device, also you can make call or send SMS to other numbers.

Why need Auto Generation?

Because the device used to call out as a landing, a large number of outgoing easily be detected abnormality, so we need auto generation incoming calls, outgoing calls which between different operators.

Basic Settings:

Auto Call:

Called by other ports： Auto call between the same device ports

Note:Auto Generation between SIM cards must learn number at first,please refer to Learn

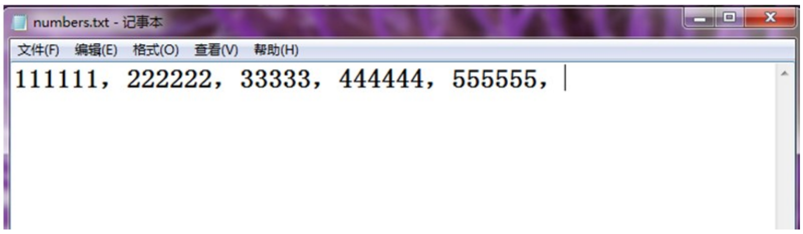
SIM Card Number section.

Call Out: We can set to call fixed numbers

Import Numbers: Choose the file, then save the text file as .txt format.

How to make the txt file?

The length of each number up to 22 digits, use "," separated, we can only input 600 digits in one file (Include commas)

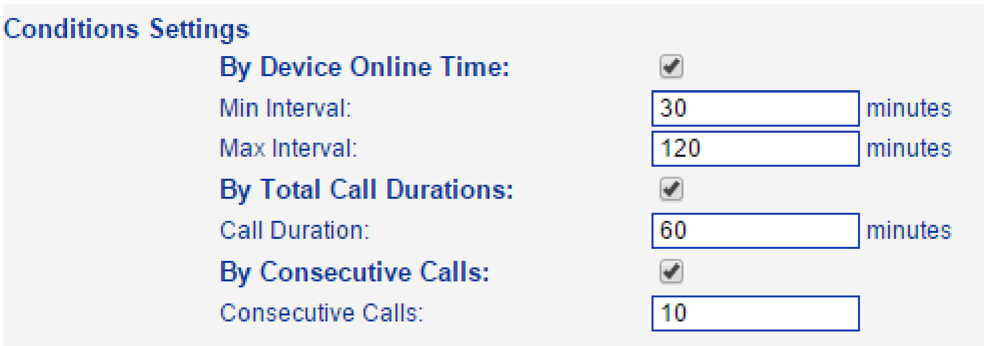


Number of retries after call failure: After the automatic call failure, whether to retry.

Call Duration: you can set any time you want, Automatic call duration will between the Min and Max.

Auto Send SMS: Auto SMS between the same device ports

Conditions Settings: define the value when auto SMS/Call generation start to work



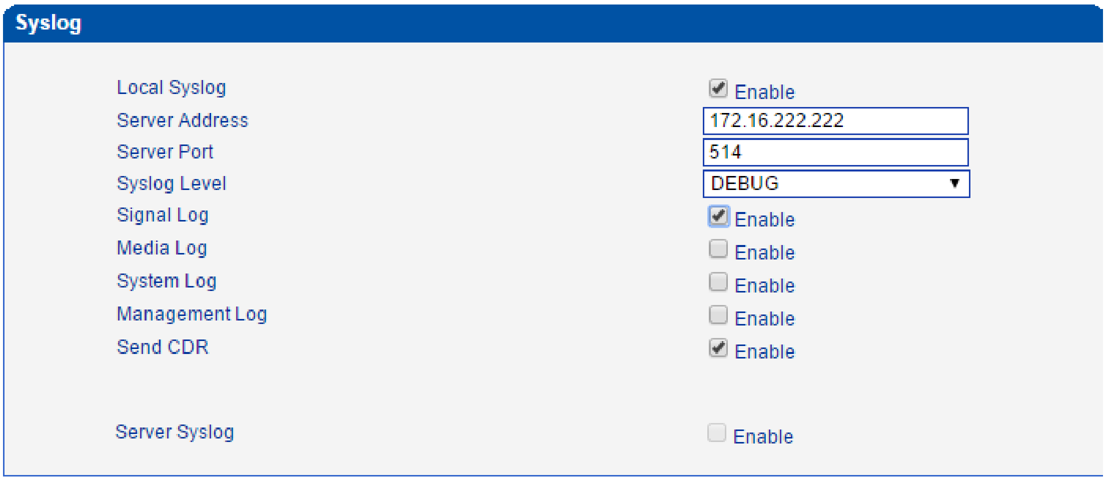
1)By Device Online Time: SIM cards register in device time, every 30-120mins, it will make call or send SMS, Random intervals between 30-120minutes.

2)By Total Call Durations: When call out time reach 60mins, there will generate an automatic call or SMS.

3)**By Consecutive Calls:** There are 20 consecutive outgoing calls, there will generate an automatic call or SMS. But if there are 19 consecutive outgoing calls, the SIM card receive an incoming call, it will be re-count.

4.11 DIAGNOSTIC

4.11.1 Syslog



Syslog is a standard for network device data logging. It allows separation of the software that generates messages from the system that stores them and the software that reports and analyzes them. It also provides devices which would otherwise be unable to communicate a means to notify administrators of problems or performance. There are 5 levels of syslog, Including NONE, DEBUG, NOTICE, WARNING and ERROR.

The Signal Log is include following traces which defined in system by default

- *SD, hardware debug*
- *SIP, SIP signaling trace*
- *STUN, STUN logs*
- *ECC, detail information of call control modular*
- *RE, the common communication modular for SCP and SIM*
- *SCP, the communication protocol between gateway and cloud server*

The media log is include following traces which defined in system by default

- *RTP, RTP stream info collection*
- *SIM, to output traces between gateway and remote SIM cards*

The System Log is include following traces which mainly used by developer

- *SYS, system log*
- *TIMER, system process*
- *TASK, system task process*
- *CFM, system process*
- *NTP*

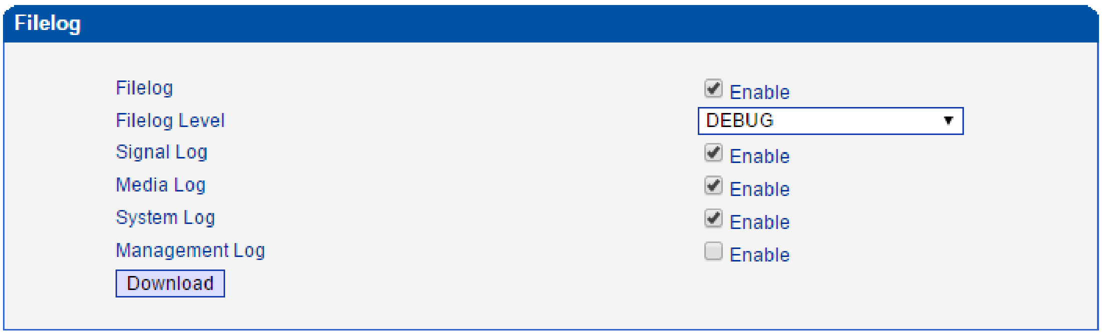
The Management Log is include following traces which defined in system by default

- *CLI, command line*
- *TEL,*
- *LOAD, firmware upload*
- *SNMP*
- *WEBS, embedded web server*
- *PROV, provisioning*

Server Syslog:

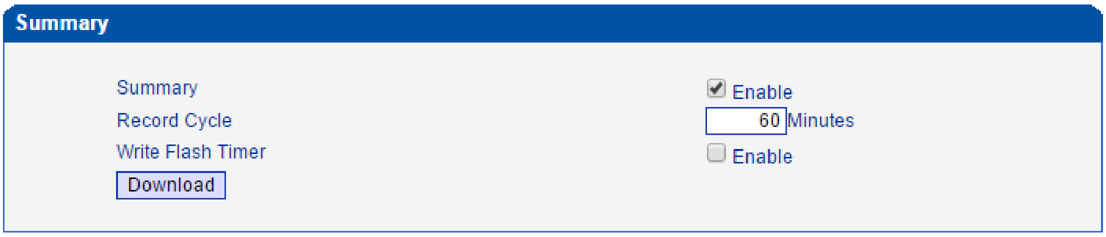
When the gateway register to SIM Cloud server, the option will be changed to un-configurable and all logs to be storage on server.

4.11.2 Filelog



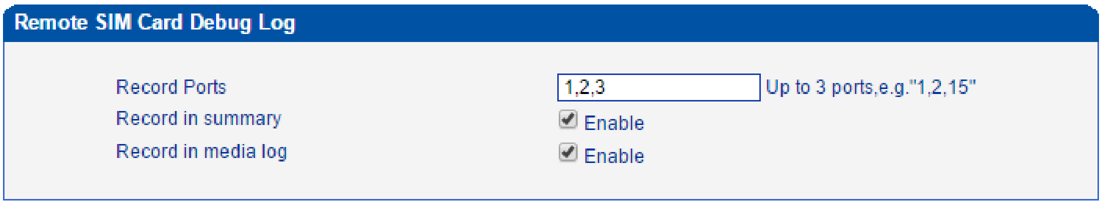
The filelog includes signal log, media log and system log, you can enable it if you want to do some troubleshooting. Click download button to save the filelog.

4.11.3 Summary



Summary file is enabled by default. Just click download button in case of some of system error happened.

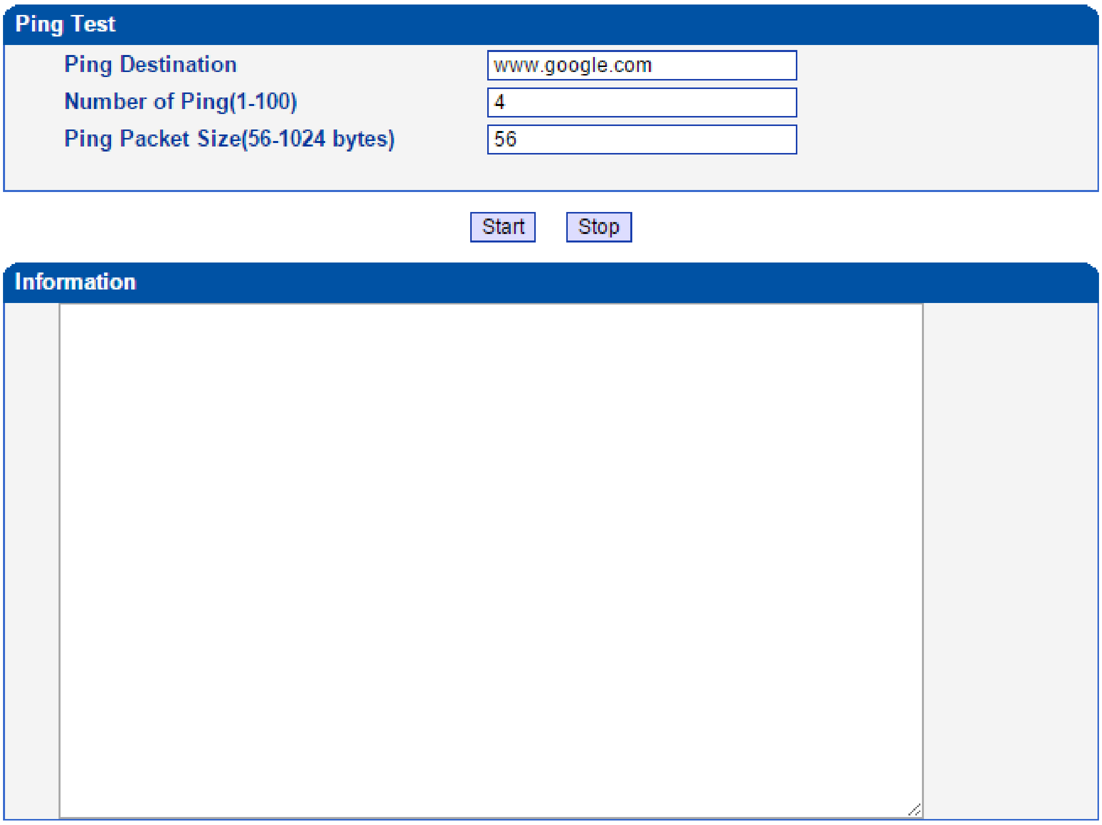
4.11.4 SIM card debug



Enable trace while remote SIM card used in this device.

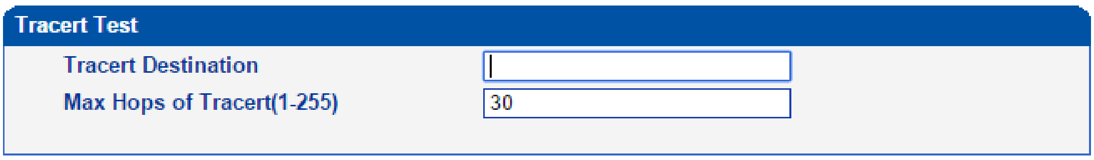
4.11.5 Ping test

you can use Ping to check whether the network is working or not.



4.11.6 Tracert Test

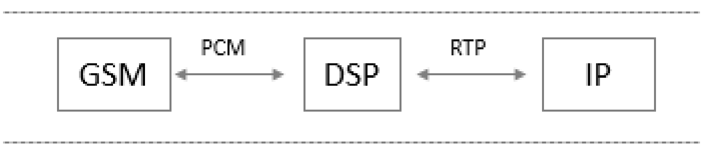
You can check the routes of the tracert destination.



4.11.7 Network Capture

Network capture is a very important diagnostic tool for maintenance. This section is describes how to enable network capture.

Voice stream transmit path of the gateway as below:

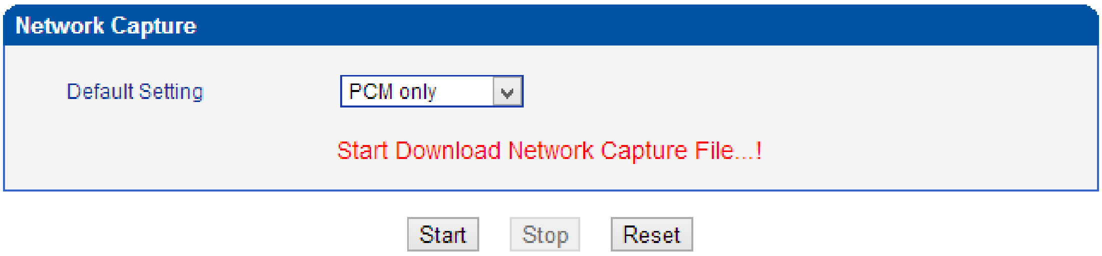


Getting start to PCM capture

PCM capture is help to analysis voice stream between GSM/CDMA modular and DSP chipset.

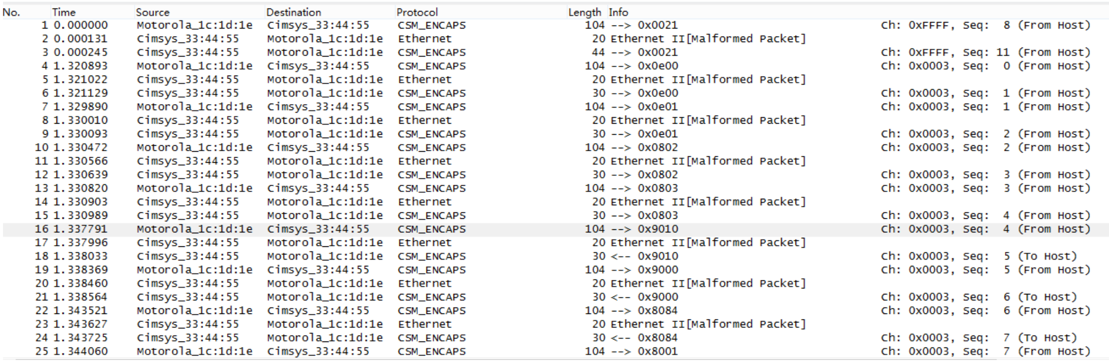
To enable PCM capture

- Select ‘PCM only’ on Network Capture page



- Click “*Start’* to enable PCM capture
- Dialing out through gateway, start talking a short while then hangup the call.
- Click ‘*Stop’* to disable network capture
- Save the capture file to local computer

The capture is named to ‘capture(x).pcap’, x is serial number of capture and will be added 1 in next time. The sample of PCM capture as below:

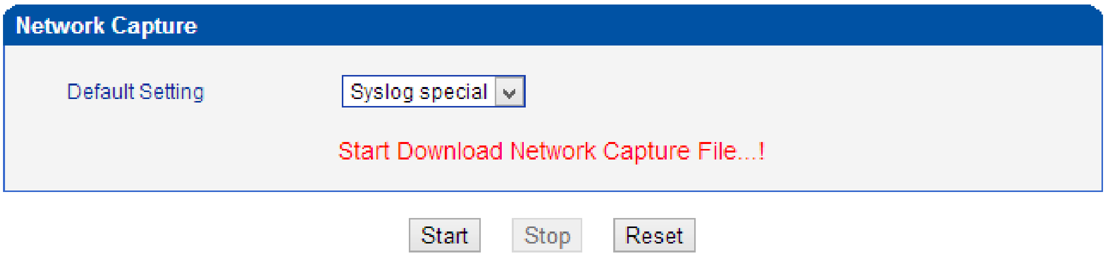


No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0021 Ch: 0xFFFF, Seq: 8 (From Host)
2	0.000131	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
3	0.000245	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	44	--> 0x0021 Ch: 0xFFFF, Seq: 11 (From Host)
4	1.320893	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0e00 Ch: 0x0003, Seq: 0 (From Host)
5	1.321022	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
6	1.321129	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	--> 0x0e00 Ch: 0x0003, Seq: 1 (From Host)
7	1.329890	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0e01 Ch: 0x0003, Seq: 1 (From Host)
8	1.330010	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
9	1.330093	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	--> 0x0e01 Ch: 0x0003, Seq: 2 (From Host)
10	1.330472	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0802 Ch: 0x0003, Seq: 2 (From Host)
11	1.330566	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
12	1.330639	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	--> 0x0802 Ch: 0x0003, Seq: 3 (From Host)
13	1.330820	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0803 Ch: 0x0003, Seq: 3 (From Host)
14	1.330903	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
15	1.330989	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	--> 0x0803 Ch: 0x0003, Seq: 4 (From Host)
16	1.337791	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x9010 Ch: 0x0003, Seq: 4 (From Host)
17	1.337996	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
18	1.338033	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x9010 Ch: 0x0003, Seq: 5 (To Host)
19	1.338369	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x9000 Ch: 0x0003, Seq: 5 (From Host)
20	1.338460	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
21	1.338564	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x9000 Ch: 0x0003, Seq: 6 (To Host)
22	1.343521	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x8084 Ch: 0x0003, Seq: 6 (From Host)
23	1.343627	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
24	1.343725	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x8084 Ch: 0x0003, Seq: 7 (To Host)
25	1.344060	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x8001 Ch: 0x0003, Seq: 7 (From Host)

Getting start to Syslog capture

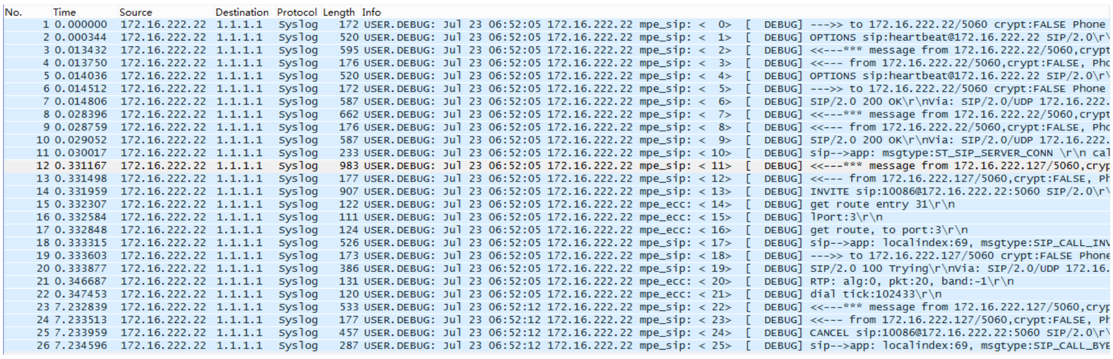
Syslog capture is another way to obtain syslog which the same as remote syslog server and filelog. The capture file is save as pcap format so that it can be opened in some of capture software like Wireshark, Ethereal software etc.

- **To enable syslog capture**
  - Select Syslog special only on Network Capture page



  - Click "Start' to enable syslog capture
  - Dialing out through gateway, start talking a short while then hangup the call.
  - Click 'Stop' to disable syslog capture
  - Save the capture to local computer

The capture is named to 'capture(x).pcap', x is serial number of capture and will be added 1 in next time. The sample of syslog capture as below:

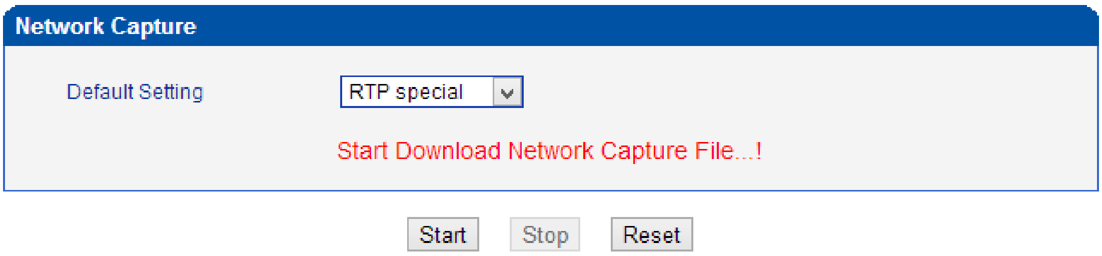


No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	172.16.222.22	1.1.1.1	Syslog	172	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 0> [DEBUG] --->> to 172.16.222.22/5060 crypt:FALSE Phone
2	0.000344	172.16.222.22	1.1.1.1	Syslog	520	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 1> [DEBUG] OPTIONS sip:heartbeat@172.16.222.22 SIP/2.0\r\
3	0.013432	172.16.222.22	1.1.1.1	Syslog	595	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 2> [DEBUG] <<---*** message from 172.16.222.22/5060,crypt
4	0.013750	172.16.222.22	1.1.1.1	Syslog	176	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 3> [DEBUG] <<--- from 172.16.222.22/5060,crypt:FALSE, Pho
5	0.014036	172.16.222.22	1.1.1.1	Syslog	520	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 4> [DEBUG] OPTIONS sip:heartbeat@172.16.222.22 SIP/2.0\r\
6	0.014512	172.16.222.22	1.1.1.1	Syslog	172	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 5> [DEBUG] --->> to 172.16.222.22/5060 crypt:FALSE Phone
7	0.014806	172.16.222.22	1.1.1.1	Syslog	587	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 6> [DEBUG] SIP/2.0 200 OK\r\nVia: SIP/2.0/UDP 172.16.222.
8	0.028396	172.16.222.22	1.1.1.1	Syslog	662	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 7> [DEBUG] <<---*** message from 172.16.222.22/5060,crypt
9	0.028759	172.16.222.22	1.1.1.1	Syslog	176	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 8> [DEBUG] <<--- from 172.16.222.22/5060,crypt:FALSE, Pho
10	0.029052	172.16.222.22	1.1.1.1	Syslog	587	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 9> [DEBUG] SIP/2.0 200 OK\r\nVia: SIP/2.0/UDP 172.16.222.
11	0.030017	172.16.222.22	1.1.1.1	Syslog	233	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 10> [DEBUG] sip-->app: msgtype:ST_SIP_SERVER_CONN \r\n cal
12	0.331167	172.16.222.22	1.1.1.1	Syslog	983	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 11> [DEBUG] <<---*** message from 172.16.222.127/5060,cryp
13	0.331498	172.16.222.22	1.1.1.1	Syslog	177	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 12> [DEBUG] <<--- from 172.16.222.127/5060,crypt:FALSE, Ph
14	0.331959	172.16.222.22	1.1.1.1	Syslog	907	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 13> [DEBUG] INVITE sip:10086@172.16.222.22:5060 SIP/2.0\r\
15	0.332307	172.16.222.22	1.1.1.1	Syslog	122	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_ecc: < 14> [DEBUG] get route entry 31\r\n
16	0.332584	172.16.222.22	1.1.1.1	Syslog	111	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_ecc: < 15> [DEBUG] lPort:3\r\n
17	0.332848	172.16.222.22	1.1.1.1	Syslog	124	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_ecc: < 16> [DEBUG] get route, to port:3\r\n
18	0.333315	172.16.222.22	1.1.1.1	Syslog	526	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 17> [DEBUG] sip-->app: localindex:69, msgtype:SIP_CALL_INV
19	0.333603	172.16.222.22	1.1.1.1	Syslog	173	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 18> [DEBUG] --->> to 172.16.222.127/5060 crypt:FALSE Phone
20	0.333877	172.16.222.22	1.1.1.1	Syslog	386	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 19> [DEBUG] SIP/2.0 100 Trying\r\nVia: SIP/2.0/UDP 172.16.
21	0.346687	172.16.222.22	1.1.1.1	Syslog	131	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_ecc: < 20> [DEBUG] RTP: alg:0, pkt:20, band:-1\r\n
22	0.347453	172.16.222.22	1.1.1.1	Syslog	120	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_ecc: < 21> [DEBUG] dial tick:102433\r\n
23	7.232839	172.16.222.22	1.1.1.1	Syslog	533	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 22> [DEBUG] <<---*** message from 172.16.222.127/5060,cryp
24	7.233513	172.16.222.22	1.1.1.1	Syslog	177	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 23> [DEBUG] <<--- from 172.16.222.127/5060,crypt:FALSE, Ph
25	7.233959	172.16.222.22	1.1.1.1	Syslog	457	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 24> [DEBUG] CANCEL sip:10086@172.16.222.22:5060 SIP/2.0\r\
26	7.234596	172.16.222.22	1.1.1.1	Syslog	287	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 25> [DEBUG] sip-->app: localindex:69, msgtype:SIP_CALL_BYE

- **Getting start to RTP capture**

PCM capture is help to analysis voice stream between gateway and remote IPPBX/SIP Server.

- **To enable RTP capture:**
  - Select RTP special on Network Capture page



  - Click Start to enable RTP capture

- Dialing out through gateway, start talking a short while then hangup the call.
- Click Stop to disable RTP capture
- Save the capture to local computer

The capture is named to 'capture(x).pcap', x is serial number of capture and will be added 1 in next time. The sample of RTP capture as below:

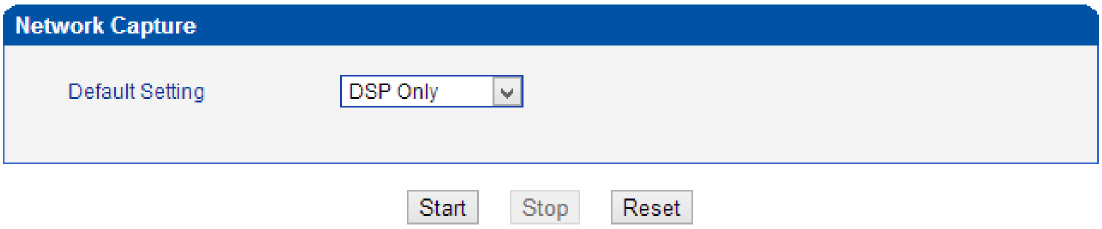
No.	Time	Source	Destination	Protocol	Length	Info
176	7.020000	172.16.221.228	116.204.105.50	SIP	565	Request: REGISTER sip:116.204.105.50 |
178	7.030000	116.204.105.50	172.16.221.228	SIP	411	Status: 200 OK (1 bindings) |
244	11.610000	172.16.221.228	58.56.64.101	SIP/SDP	814	Request: INVITE sip:201@58.56.64.101 |
248	11.710000	58.56.64.101	172.16.221.228	SIP	480	Status: 100 Trying |
249	11.710000	58.56.64.101	172.16.221.228	SIP/SDP	733	Status: 183 Session Progress |
250	11.710000	58.56.64.101	172.16.221.228	SIP/SDP	719	Status: 200 OK |
252	11.720000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
253	11.720000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
254	11.720000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1000, Time=160, Mark
255	11.720000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
256	11.730000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
257	11.730000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
258	11.740000	172.16.221.228	58.56.64.101	SIP	434	Request: ACK sip:201@58.56.64.101:5060 |
259	11.740000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1001, Time=320
261	11.770000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1002, Time=480
263	11.780000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1003, Time=640
264	11.810000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1004, Time=800
265	11.830000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1005, Time=960
266	11.840000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1006, Time=1120
267	11.870000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1007, Time=1280
268	11.890000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1008, Time=1440
270	11.900000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1009, Time=1600
271	11.930000	172.16.221.228	58.56.64.101	RTP	74	PT=ITU-T G.729, SSRC=0x43455AA6, Seq=31521, Time=1806312883
273	11.930000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1010, Time=1760
274	11.940000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1011, Time=1920
275	11.950000	172.16.221.228	58.56.64.101	RTP	74	PT=ITU-T G.729, SSRC=0x43455AA6, Seq=31522, Time=1806313043
277	11.970000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6D15, Seq=1012, Time=2080
278	11.970000	172.16.221.228	58.56.64.101	RTP	74	PT=ITU-T G.729, SSRC=0x43455AA6, Seq=31523, Time=1806313203

Getting start to DSP capture

DSP capture is help to analysis voice stream inside DSP chipset. The DSP chipset will handle RTP from IP network as well as voice stream from GSM/CDMA modular.

To enable DSP capture:

- Select DSP only on Network Capture page



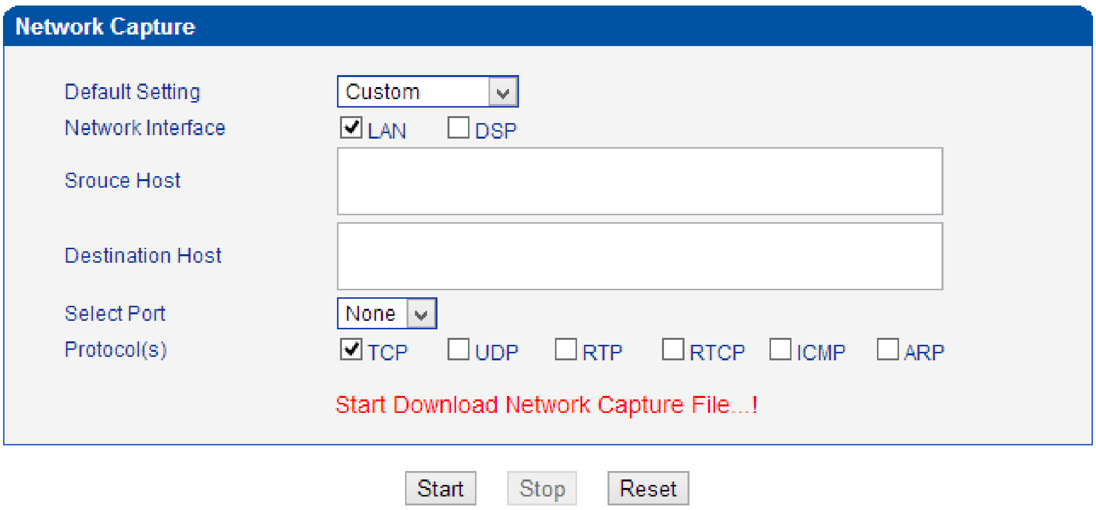
- Click Start to enable DSP capture
- Dialing out through gateway, start talking a short while then hangup the call.
- Click Stop to disable DSP capture
- Save the capture to local computer

The capture is named to 'capture(x).pcap', x is serial number of capture and will be added 1 in next time. The sample of RTP capture as below:

No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0021 Ch: 0xFFFF, Seq: 2 (From Host)
2	0.007246	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
3	0.007260	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	44	--> 0x0021 Ch: 0xFFFF, Seq: 5 (From Host)
4	2.994581	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0021 Ch: 0xFFFF, Seq: 3 (From Host)
5	2.997308	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
6	2.997316	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	44	--> 0x0021 Ch: 0xFFFF, Seq: 6 (From Host)
7	5.992790	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x0021 Ch: 0xFFFF, Seq: 4 (From Host)
8	5.997282	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
9	5.997290	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	44	--> 0x0021 Ch: 0xFFFF, Seq: 7 (From Host)
10	7.691428	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x9010 Ch: 0x0003, Seq: 3 (From Host)
11	7.691552	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
12	7.691715	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x9010 Ch: 0x0003, Seq: 1 (To Host)
13	7.701379	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x9000 Ch: 0x0003, Seq: 4 (From Host)
14	7.701494	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
15	7.701622	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x9000 Ch: 0x0003, Seq: 2 (To Host)
16	7.709662	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x8084 Ch: 0x0003, Seq: 5 (From Host)
17	7.709798	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
18	7.709902	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x8084 Ch: 0x0003, Seq: 3 (To Host)
19	7.710238	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x8001 Ch: 0x0003, Seq: 6 (From Host)
20	7.710328	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
21	7.710496	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x8001 Ch: 0x0003, Seq: 4 (To Host)
22	7.716241	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x8018 Ch: 0x0003, Seq: 7 (From Host)
23	7.716352	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]
24	7.716465	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCAPS	30	<-- 0x8018 Ch: 0x0003, Seq: 5 (To Host)
25	7.716711	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCAPS	104	--> 0x805b Ch: 0x0003, Seq: 8 (From Host)

- **Configurable capture options**
  - **Getting start to custom capture**

This menu provides more options to capture specific packets as actually needs.



4.11.8 Voice Loopback Test

Voice Loopback test should be done on call status. Each call can do one kind of test. After each test, please hang up and call again, refresh web interface and go on the other tests.

Voice Loopback Test

Port	Voice Loopback Test		
0	Dsp Tdm Test	Dsp IP Test	Recover
1	Dsp Tdm Test	Dsp IP Test	Recover
2	Dsp Tdm Test	Dsp IP Test	Recover
3	Dsp Tdm Test	Dsp IP Test	Recover
4	Dsp Tdm Test	Dsp IP Test	Recover
5	Dsp Tdm Test	Dsp IP Test	Recover
6	Dsp Tdm Test	Dsp IP Test	Recover
7	Dsp Tdm Test	Dsp IP Test	Recover

Voice stream patch on gateway:

DSP TDM Test

DSP TDM Test is the loopback of GSM side.

VoIP <-----DSP<----- Modular<------ Mobile

-----> ------> ------->

- **To start DSP TDM Test:**
  - Make a call test through gateway, the call can be initialed by IPPHONE. Keep the conversation after call establish
  - Click DSP TDM Test to start test
  - Check the voice on both sides. VoIP side become silence and echo should be generated on Mobile phone side
  - Hangup
- **To start DSP IP Test:**

DSP IP Test is the loopback of VoIP side.

IPHONE------>VoIP-------> DSP

<------ <-------

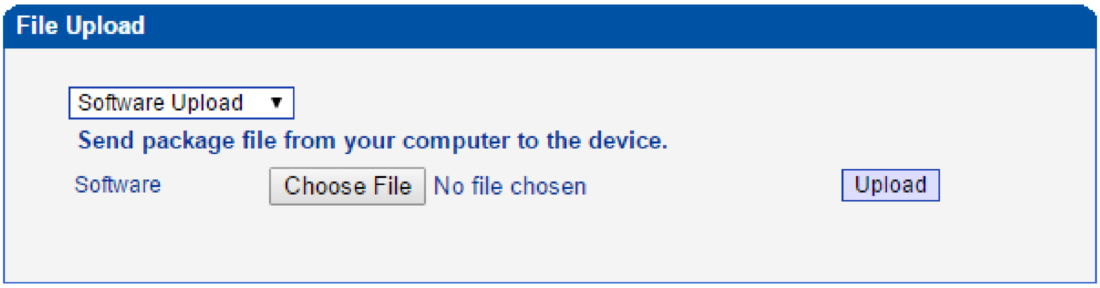
- **To start DSP IP Test:**
  - Make a call test through gateway, the call can be initialed by IPPHONE. Keep the

conversation after call establish

- Click DSP IP Test to start test
- Check the voice on both sides. GSM side become silence and echo should be generated on IPPHONE side
- Hang up

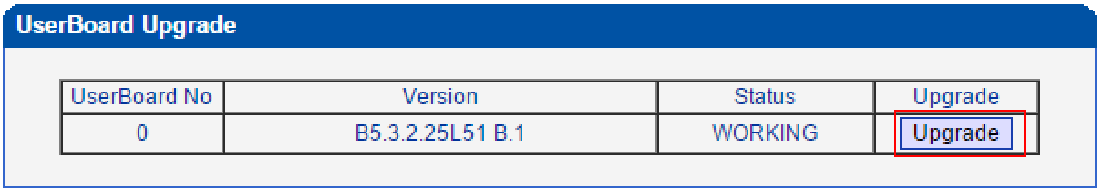
4.12 TOOLS

4.12.1 File Upload



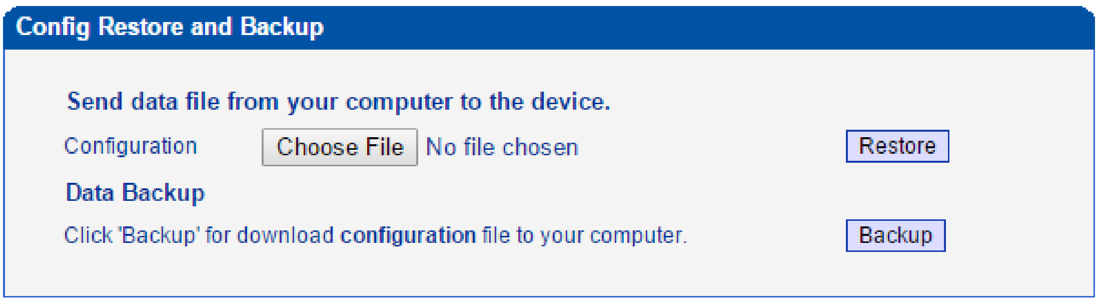
On the **Tools → Firmware Upload**, you can upload a firmware to upgrade the device. But you need to restart the device for the change to take effect.

4.12.2 Userboard Upgrade



Click Upgrade button while Status show as “FAULT”. This page mainly use to reload the userboard firmware.

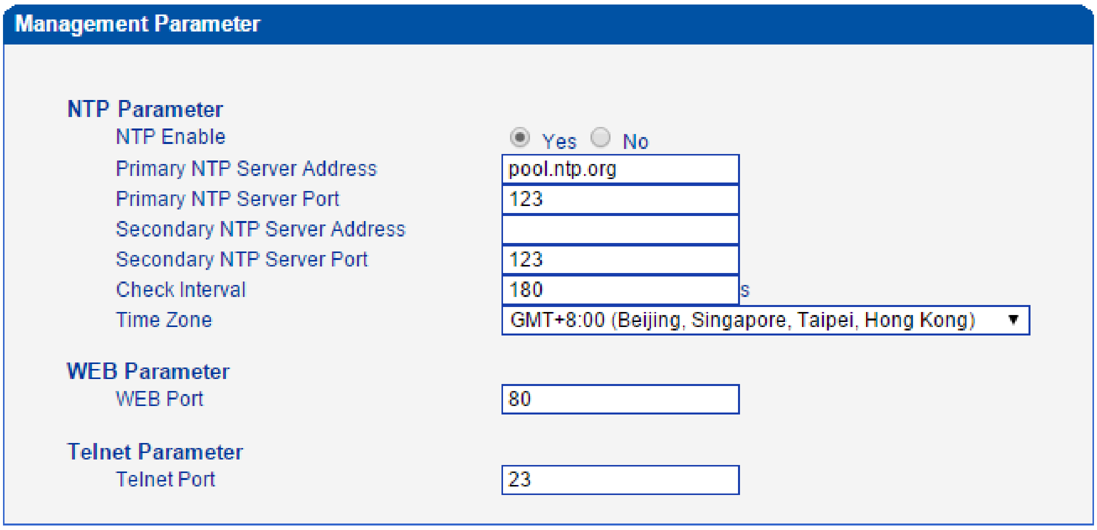
4.12.3 Config Restore and Backup



Backup or restore config file of the device.

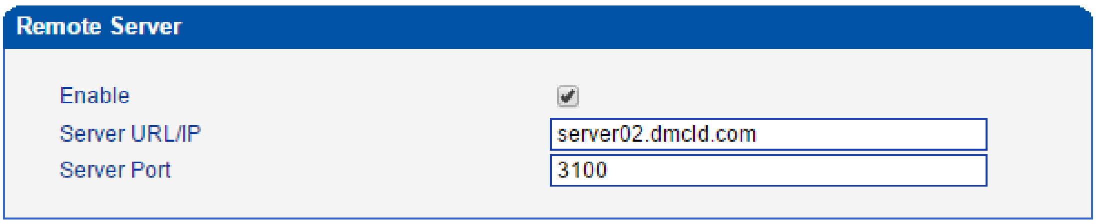
You can restore this configuration in case the unit loses it for any reason or to clone a unit with the configuration of another unit. The configuration backup configurations are in txt format. Please note that you can use a backup file from an older firmware version and use it in a unit with a more recent firmware version. However, a backup file from a newer firmware version than the one actually in the unit cannot be used for a restore operation on the unit.

4.12.4 Management Parameter



Parameters	Description
NTP Parameter	The Network Time Protocol (NTP) is a protocol and software implementation for synchronizing the clocks of computer systems over packet-switched, variable-latency data networks. User need to fill the NTP Server Address and select Time Zone
Web Port	Default is 80
Telnet Port	Default is 23

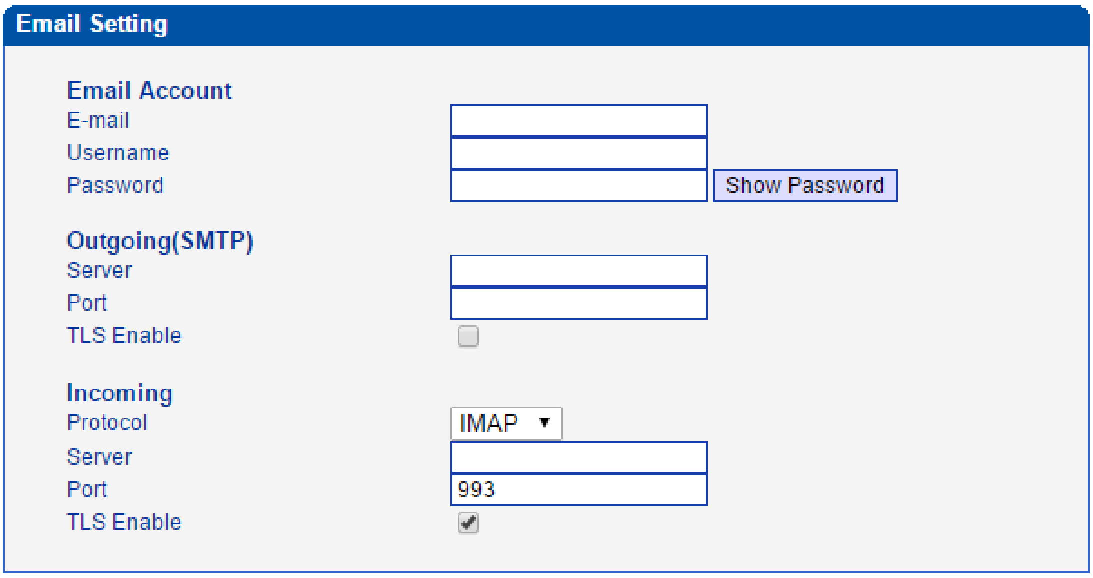
4.12.5 Remote Server



While devices deployed behind router/firewall. Users can't access the device remotely. With the Remote Server, device can register to it and access web/telnet through Remote Server remotely.

Register Remote Server account from web site server02.dmcld.com:3000

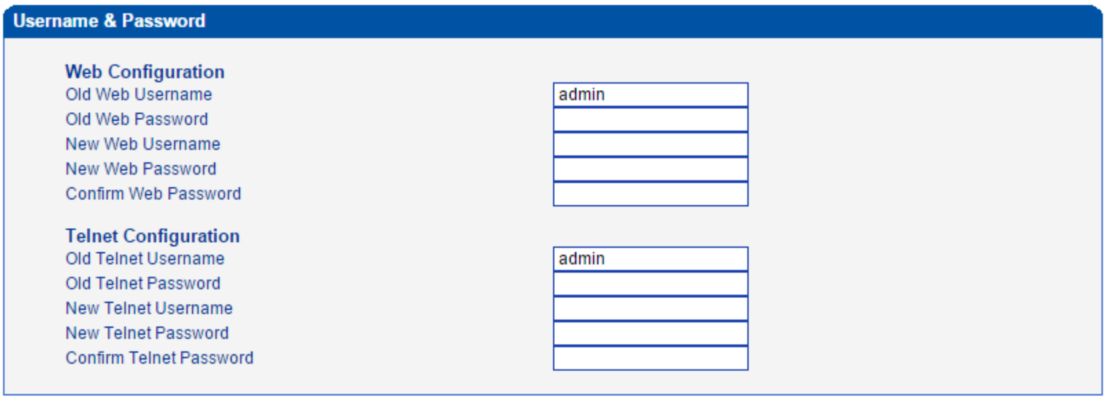
4.12.6 Email Account Setting



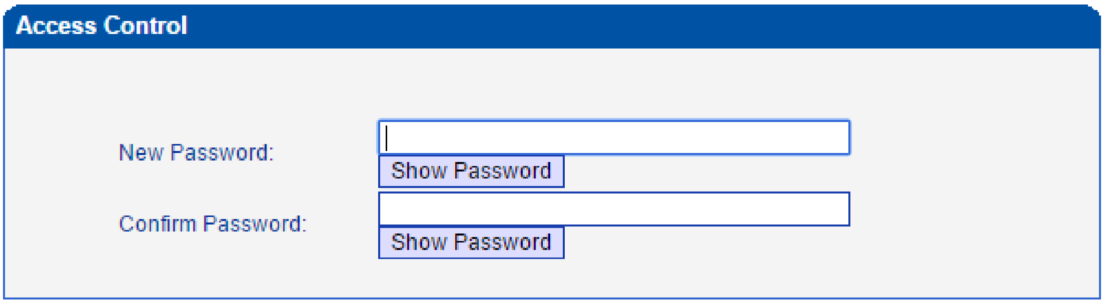
Please refer to **section. SMS and USSD -> Email**.

4.12.7 Username and Password

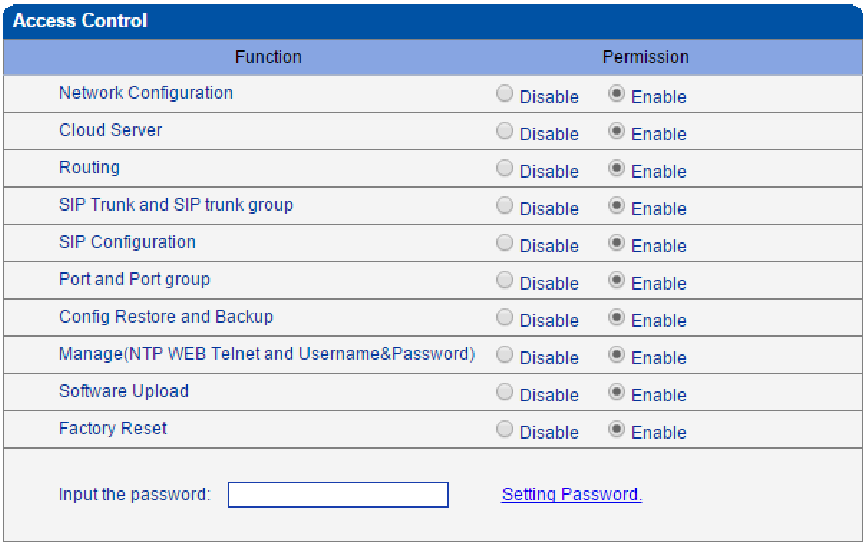
When using web or telnet Configuration, please enter default user name and password. User can modify the login name and password.



4.12.8 Access control



You need to set a new password to control access level of web links. After set password, you can set which page is allow/disallow to access by default user.

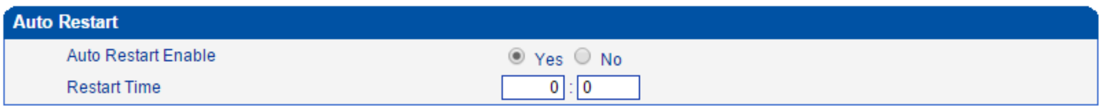


4.12.9 Factory Reset

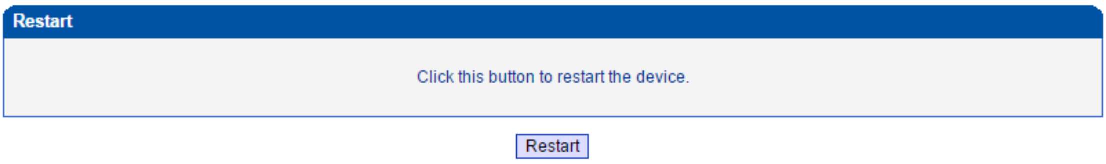
Be careful do this operation, after restore factory setting, all the parameters will be changed to the factory default.

4.12.10 Auto Restart

Configure auto restart at pre-defined HH/MM



4.12.11 Restart



5 Troubleshooting and Command Line

5.1 Login UC2000 & General Knowledge of UC2000 Command

This is a document for some customers who need more details of DINSTAR's products with command lines. To make sure the system runs successfully, we suggest customers setting UC2000 by GUI. In this manual, some topics such as how to check the IP, signaling and call conversation are covered.

Tips: The document is fit for all UC2000-VE/F/G models.

Run system tool Telnet to login UC2000. The default username and password is "admin".

C:\Users\Administrator>telnet 172.16.101.142

```
Welcome to Command Shell!
Username:admin
Password:*****
ROS>
```

Input "?" to show the all commands and its information.

```
ROS>
ROS>?
    enable  Turn on privileged commands
    exit    Exit from the EXEC
    show    Show running system information
ROS>
```

Abbreviation is supported in UC2000 command. For example you can input "en" substitute for "enable", input "sh" substitute for "show", input "cl" substitute for "clock",

```
ROS>
ROS>sho ?
    clock   Display the system clock
    version System hardware and software status
ROS>sho cl
12/14/2011  21:27:56
ROS>
```

5.2. COMMANDS IN "ROS#" MODE

There is only a litter commans in "ROS>" mode. If you need more commands you must enter the "ROS#" mode. Input "enable" to enter "ROS#" mode if you have in the "ROS>" mode.

```
ROS>
ROS>en
ROS#
```

5.2.1 SUMMARIZE OF COMMANDS IN "ROS#" MODE

Input "?" to get the information of all commands in "ROS#" mode.

```
ROS#
ROS#?
    dbg                 Show ada information
    dspconfigure        Configure device parameters
    exit                Exit from privelige mode
    menuconfigure       Configure system parameters
    ntp                 Configure ntp_sntp parameters
    ping                Send echo messages
    show                Show running system information
ROS#
```

5.2.2 GENERAL PURPOSE COMMANDS IN "ROS#" MODE

- **Show IP address (show int)**

```
ROS#
ROS#sho int

Ethernet0/0/0 is up, line protocol is up
MTU is 1500 in bytes, Internet Address is owned, 192.168.11.1/24
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 001F.D6A0.023F


Ethernet0/0/1 is up, line protocol is up
MTU is 1500 in bytes, Internet Address is owned, 172.16.101.142/16
IP Sending Frames' Format is PKTFMT_ETHNT_2, Hardware address is 001F.D6A0.023F

ROS#_
```

Show Time (show clock)

```
ROS#
ROS#sho cl
12/14/2011  21:19:13
ROS#
```

Show version (show version)

```
ROS#sho ver
DWG2000D 2.22.01.04  PCB 2 LOGIC 0 BIOS 1, Built on Jun 19 2012, 15:26:51
ROS#_
```

Show sip Information (show sip config)

```
ROS#
ROS#sho sip config
 local ipaddr    : 172.16.101.142
 keep  alive     : on 10(s)
 message check   : off
 noanswer  time  : 90(s)
 sip currentport: 5060
 T0              : 500(ms)
 T1              : 500(ms)
 T2              : 4000(ms)
 T4              : 5000(ms)
 TMax            : 32000(ms)
 do  not  reg    : off
 100rel          : off
 referto use contact: off
 local port random: off
 client crypt    : off
 firewall  ip    : 172.16.101.142
 firewall  port  : 5060
 dns type        : A Query
 dns refresh time: 0(min)
------------------------
 proxy id        : 0
 proxy domain    : 172.16.0.8
 proxy ip        : 172.16.0.8
 proxy port      : 2080
 reg interval    : 1800
ROS#
ROS#_
```

Show memory status (show memory detail)

```
ROS#
ROS#sho memory detail
Addr(0x) Size        Mpe  Sid(0x)  Tick        Ref  Line  File
4019f004 12          71   0        3607511     1    149   osip_port.c
4019f018 12          71   0        3607511     1    149   osip_port.c
4019f02c 12          71   0        3607511     1    149   osip_port.c
4019f040 12          71   0        3607511     1    149   osip_port.c
4019f054 12          71   0        3607511     1    149   osip_port.c
4019f068 12          71   0        3607511     1    149   osip_port.c
4019f07c 12          71   0        3607511     1    149   osip_port.c
4019f090 12          71   0        3607511     1    149   osip_port.c
4019f0a4 12          71   0        3607511     1    149   osip_port.c
4019f0b8 12          71   0        3607511     1    149   osip_port.c
4019f0cc 12          53   0        2955251     1    337   atchannel.c
4019f0e0 12          53   0        2955472     1    331   atchannel.c
4019f0f4 12          53   0        197         1    1362  atchannel.c
4019f108 12          53   0        2955550     1    331   atchannel.c
4019f180 12          53   0        2955503     1    337   atchannel.c
4019f1a8 12          53   0        2955305     1    337   atchannel.c
4019f1bc 12          53   0        2955518     1    331   atchannel.c
4019f1e4 12          53   0        196         1    1362  atchannel.c
4019f1f8 12          53   0        2955305     1    331   atchannel.c
4019f220 12          53   0        2955472     1    331   atchannel.c
4019f234 12          53   0        2955472     1    337   atchannel.c
4019f25c 12          53   0        2955518     1    337   atchannel.c
 ---- More ( Press CTRL_C to break ) ----
```

Show SIP port status (show sip all)

```
ROS#
ROS#sho sip all
 Index   UserId                             State            Expire(s)  RemainTime
------------------------------------------------------------------------
 0       30                                 OK               1800       976
 1       31                                 OK               1800       976
 2       33                                 OK               1800       976
ROS#
```

Show Current calls (sh ecc call)

```
ROS#
ROS#sho ecc call
 CcbNo  PortNo           Caller       Called       CcbState
 -------------------------------------------------------------------------------------
 2      14               01212043684  01759408567  out_active
 3      9                198257604    01715214621  out_active
 6      5                H3258884     01830573560  out_active
 13     3                bablohath    01710719124  out_active
 16     8                0503298872   01720419701  out_recving
 18     7                Mal106       01745599151  out_active
 19     2                Jahid.2416   01831644239  out_active
 22     0                22336688     01742670956  out_active
 23     1                456789255    01834636875  out_active
 -------------------------------------------------------------------------------------
ROS#
```

- **Show RTP session (sho rtp se)**

```
ROS#
ROS#sho rtp se
RTP Information:
   RTP System TimeStamp 1586900(ms)
   MBUF Waiting for Playing 0, MBUF Discarded 0
EIA RTP Session List:
   PT-Payload Type, PP-Packet Period, PL-Packet Length,
   SP-Sample Period, SL-Sample Length, P/S-PP/SP, LR-NetLostRate, RLR-RealLostRate
---- ---- -----  ----------  ------- ----------------------- ---------------------- ---------------------- -------
RTPNO Mode  PT    Send/ToDsp    LR/RLR          Local IP: Port          Peer IP: Port PP  PL  SP  SL P/S P2P silence
---- ---- -----  ----------  ------- ----------------------- ---------------------- ---------------------- -------
   0  STD  18     9250/9205    0/0        LocalHost: 8000   66.152.170.74:10562  20  20  20  20   1  NO    0
   2  STD  18     6499/6227    0/0        LocalHost: 8004   66.152.170.74:10658  20  20  20  20   1  NO    3
   4  STD  18   56225/56145    0/0        LocalHost: 8008   66.152.170.74: 9558  20  20  20  20   1  NO    0
   8  STD  18   13300/13201    0/0        LocalHost: 8016   66.152.170.74:10498  20  20  20  20   1  NO    1
  10  STD   4    7253/14451    0/0        LocalHost: 8020    64.15.152.90: 6042  60  48  60  48   1  NO    1
  14  STD  18   11745/11599    0/0        LocalHost: 8028   66.152.170.74:10522  20  20  20  20   1  NO    0
  16  STD  18     248/210      0/0        LocalHost: 8032   66.152.170.74:10766  20  20  20  20   1  NO    0
  18  STD  18   31800/31747    0/0        LocalHost: 8036   66.152.170.74:10186  20  20  20  20   1  NO    1
  20  STD  18   10499/10322    0/0        LocalHost: 8040   66.152.170.74:10554  20  20  20  20   1  NO    3
  24  STD  18   30028/29901    0/0        LocalHost: 8048   66.152.170.74:10198  20  20  20  20   1  NO    1
  26  STD  18   29614/6065     0/0        LocalHost: 8052    64.15.152.90:11854  20  20  20  20   1  NO    1
  28  STD  18   71018/70690    0/0        LocalHost: 8056   66.152.170.74: 9138  20  20  20  20   1  NO    1
---- ---- -----  ----------  ------- ----------------------- ---------------------- ---------------------- -------
ROS#
```

- **Show ASR/ACD statistics (show ecc state)**

```
ROS#sho ecc state
PortNo       Call      Cancel    Timeout  NotAllowed   Connected      Busy   NoAnswer  NoDialTone   NoCarrier  SdpNegFailed   CallDelay
er
--------------------------------------------------------------------------------------------------------------------------------------
----
    0         31          5         0         0           8          1         0         11          6          0            0
    1         24          6         0         0           9          0         0          5          4          0            0
    2         28         11         1         0          13          0         0          0          3          0            0
    3         24          5         0         0          12          1         0          0          6          0            0
    4         19          3         2         0          10          1         0          2          1          0            0
    5          0          0         0         0           0          0         0          0          0          0            0
    6         16          5         1         0           8          1         0          0          1          0            0
    7         11          3         0         0           8          0         0          0          0          0            0
    8          0          0         0         0           0          0         0          0          0          0            0
    9         12          3         0         0           7          1         0          0          1          0            0
   10         14          4         1         0           8          1         0          0          0          0            0
   11         24          8         0         0          11          2         0          0          3          0            0
   12         31         10         1         0          14          0         0          0          6          0            0
   13         28          7         3         0          11          2         0          1          4          0            0
   14          8          2         0         0           4          1         0          0          1          0            0
   15          0          0         0         0           0          0         0          0          0          0            0

PortNo    Duration    ASR     ACD  ResetNoCar  ResetNoDil
---------------------------------------------------------------------------------------------------------------------

    0       2836      25     405          0          0
    1       5017      37     627          0          0
    2       1235      46     102          0          0
    3       5419      50     492          0          0
    4       5967      52     596          0          0
    5          0       0       0          0          0
    6       3715      50     530          0          0
    7       7799      72    1114          0          0
    8          0       0       0          0          0
    9       5692      58     948          0          0
   10       5711      57     713          0          0
   11       3199      45     290          0          0
   12       2451      45     188          0          0
   13       2002      39     200          0          0
   14       2592      50     864          0          0
   15          0       0       0          0          0

ROS#_
```

5.3 COMMANDS IN "CONFIG" MODE

5.3.1 SUMMARIZE OF COMMANDS IN "CONFIG" MODE

Input "^config" in the "ROS# " to enter "config" mode.

```
ROS#
ROS#^config
ROS(config)#
ROS(config)#
```

Input "?" to show the all commands and its information.

```
ROS(config)#
ROS(config)#?
    bridge          set software forwarding in device
    clear           clear ip statistics
    clock           Manage the system clock
    config          configuration files handle
    debug           Debugging functions
    default         reset default
    dhs             dhcpserver enable|disable|reboot
    dhsconfig       Configure DHCP server
    dns-server      Configure DNS servers
    ecc             config ecc param
    ethmode         set ethernet workmode
    exit            Exit from configure mode
    host            Add or delete a host's name and IP address
    icmp            Config icmp send and receive redirect packet
    interface       Select an interface to configure
    ip              Config static route
    load            load commands
    mac             mac
    monitor         Copy debug output to the current terminal
    nat             nat cfg cmd
    no              Disable some parameter switchs
    ppp             PPP
    product         Product default config
    reset           Reset the board
    rtp             RTP debug command
    save            save configuration
    sd              sd debug command
    setcustom       set custom
    shutdown        shutdown a user
    sip             config sip informations
    snmp-server     Modify SNMP parameters
    user_timeout    set telnet users timeout
    vlan            vlan route add or delete
    vlanif          vlan interface tagged properties
    webs            web server command
    workmode        network workmode selection:bridge or router
ROS(config)#
ROS(config)#
```

5.3.2 GENERAL PURPOSE COMMANDS IN "CONFIG" MODE

- **Set time (clock set)**

```
ROS(config)#
ROS(config)#clock ?
    set     Set the time and date,07/25/2003 13:25:43
ROS(config)#clock set 12/15/2011 11:46:35
ROS(config)#
```

- **Save the configuration (save)**

```
ROS(config)#
ROS(config)#save
ROS(config)#
```

- **Restart device (reset eia)**

```
ROS(config)#
ROS(config)#reset
Are you sure to reset? (y/n):y
ROS(config)#
```

- **Enable debug**

The command format is deb port + port number, to enable port 0 debug, as below:

```
ROS(config)#
ROS(config)#deb port 0
Succ! Debug PortNo:0

ROS(config)#
```

To enable all ports debug, with the command “deb port all”

```
ROS(config)#deb port all
 Debug All!!.

ROS(config)#
```

Without this steps, no trace logs will display on output window

- **Enable SIP debug (deb sip msg all)**

```
ROS(config)#
ROS(config)#deb sip msg all
ROS(config)#
```

5.4 HOW TO TRACE SIP LOGS

Create telnet session to gateway, the main steps as below:

Welcome to Command Shell!

Username:admin

Password:*****

ROS>en

ROS#

ROS#^config

ROS(config)#deb sip msg all

ROS(config)#ex

ROS#

ROS#^ada

ROS(ada)#ADA CONNECTED ...,WELCOME!

ROS(ada)#

ROS(ada)#turnon 71

Disable sip trace:

ROS(ada)#turnoff 71

5.5 HOW TO TRACE ECC LOGS (CALL DETAILS)

Welcome to Command Shell!

Username:admin

Password:*****

ROS>en

ROS#

ROS#^config

ROS(config)#deb port all

Debug All!!.

//enable trace on all port

ROS(config)#

ROS(config)#deb port 0

Succ! Debug PortNo:0

// enable trace port 0

ROS(config)#

ROS(config)#no deb port all

ROS(config)#

ROS(config)#ex

ROS#^ada

ROS(ada)#ADA CONNECTED ...,WELCOME!

ROS(ada)#turnon 84

Disable trace:

ROS(ada)#turnoff 84

5.6 How to trace Modular logs

Welcome to Command Shell!

Username:admin

Password:*****

ROS>en

ROS#^ada

ROS(ada)#ADA CONNECTED ...,WELCOME!

ROS(ada)#cmd 53 19 0 0 **1**

// enable trace. 0 0 means port range 0 to 0, 0 8 means port range from 0 to 8; **1** means enable modular trace

ROS(ada)#cmd 53 19 0 0 **0**

//disable modular trace

6 Glossary

GSM: Global System for Mobile Communications

CDMA: Code Division Multiple Access

FMC: Fixed Mobile Convergence

SIP: Session Initiation Protocol

MGCP: Media Gateway Control Protocol

DTMF: Dual Tone Multi Frequency

USSD: Unstructured Supplementary Service Data

PSTN： Public Switched Telephone Network

STUN: Simple Traversal of UDP over NAT

IVR: Interactive Voice Response

IMSI: International Mobile Subscriber Identification Number

IMEI: International Mobile Equipment Identity

DMZ: Demilitarized Zone

API: Application programming Interface

BCCH: Broadcast Control Channel

LAC: Location Area Code

CID: Cell ID

BTS: Base Transceiver Station

DTMF: Dual-Tone Multifrequency

IVR: Interactive Voice Response

NAT: Network Address Translation

RTP: Real-time Transport Protocol

VoIP: Voice over IP