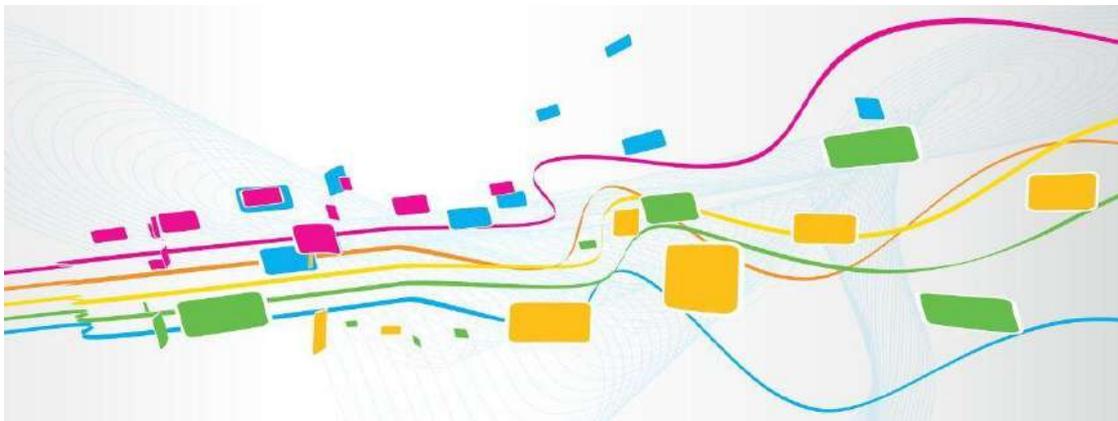




UC2000-VE/F/G GSM/CDMA/WCDMA VoIP Gateway

User Manual



Shenzhen Dinstar Co., Ltd.

Address: 9th Floor, Guoxing Building, Changxing Road, Nanshan District, Shenzhen, China

Postal Code: 518052

Telephone: +86 755 61919966

Fax: +86 755 2645 6659

Emails: sales@dinstar.com, support@dinstar.com

Website: www.dinstar.com

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
V3.3	02/04.23.13.01	Technical Support	2018-08	Match with the firmware version 02/04.23.13.01 or above

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1 Product Description

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

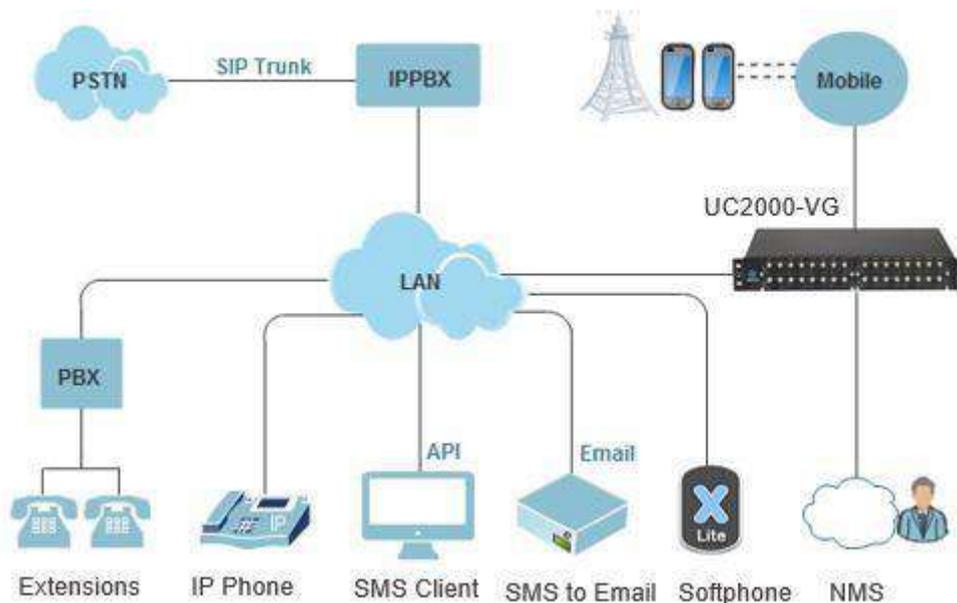
1.1 Overview

UC2000-VE/F/G serials GSM/CDMA/WCDMA/LTE 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/8W/8T



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/8W/8T



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. <ul style="list-style-type: none"> ◆ 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

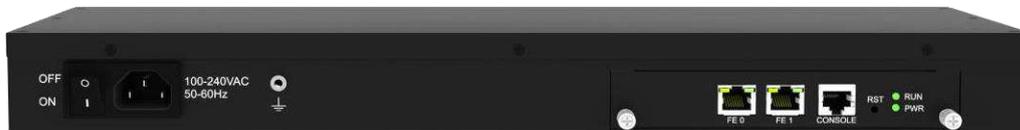


Indicators and connectors

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. <ul style="list-style-type: none"> 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	-	<p>Reset button to restore default IP and password or restore factory setting.</p> <ul style="list-style-type: none"> ◆ Restore IP and Password: hold RST button 3~5 seconds, RUN LED being ON during this time <p>Restore factory setting: Hold RST button 7 seconds, RUN LED being blink</p>
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.711 α -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 °C ~ 45 °C
(Storage): -20 °C ~80 °C
- 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

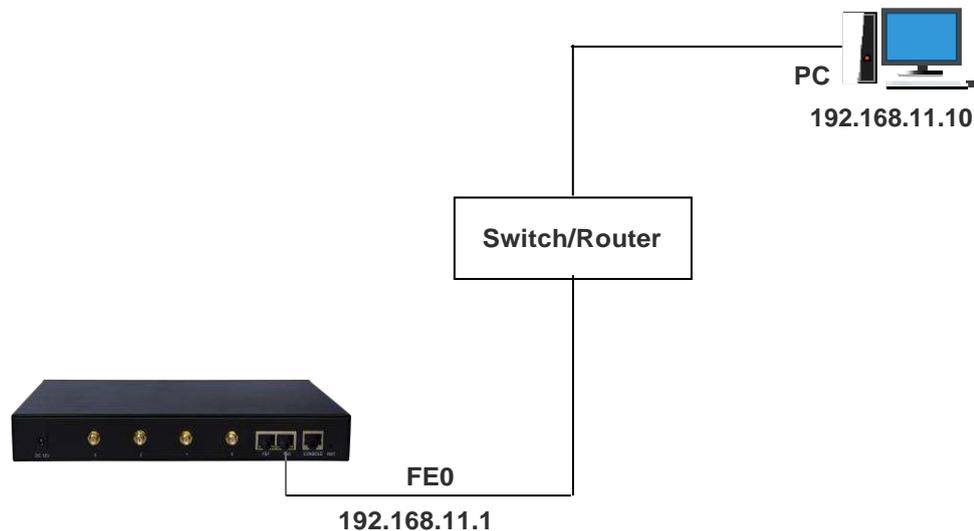
Please pay attention to the following before you install UC2000-VE/F/G/T include:

- 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/T 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

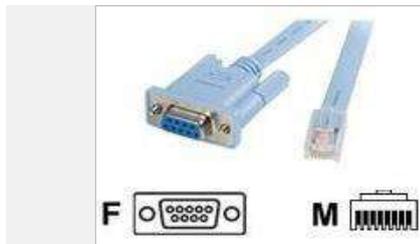
Press RST button for about 7 seconds, then gateway will be rebooted and restored to factory setting.

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);

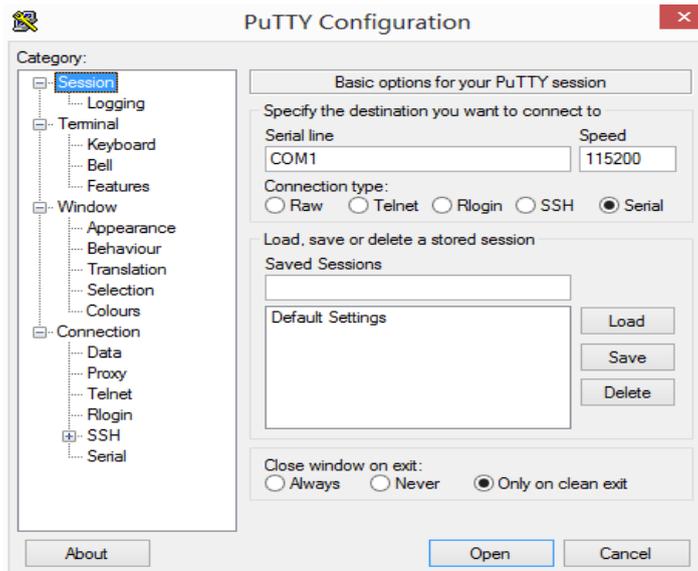


Step 2: Connect the F port of the serial cable with the COM port of PC. If the PC does not have a COM port, please use a USB-to-COM converting tool to connect the serial cable with the PC.

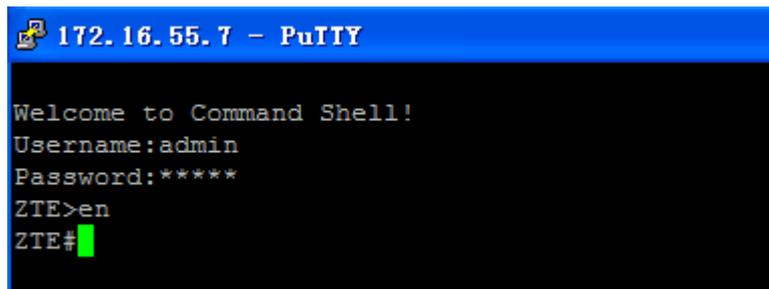
Step 3: Connect the M port of the serial cable with the console port of the device.

Step 4: Conduct configurations on login software.

Herein we take the PuTTY software 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 HTTPS/HTTP protocol. We are strongly recommended 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)*
- ▶ UC2000-VE-4T
- ▶ UC2000-VE-8T
- ▶ UC2000-VF-16T
- ▶ UC2000-VF-8T

4.1 Access UC2000 unit

Enter IP address of UC2000 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 “Login” 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 WEB configuration interface consists of the navigation tree and the detail configuration interfaces.

Figure 4-2-1 WEB introduce

Run Information			
MAC Address	F8-A0-3D-48-99-4B		
Network Mode	Bridge		
Network	172.19.211.120	255.255.0.0	Static
DNS Server	114.114.114.114	0.0.0.0	
Device SN	db27-8310-2400-0171		
Hardware ID	0000-18e7-9dc8		
Cloud Register Status	Not Registered		
License	Basic Function	Enable	
	DBO Advanced	Enable	
System Up Duration	1 m 18 s		
System Time	2018-8-17 16:01:04		
Network Traffic Statistics	Received 76307 Bytes	Sent 360083 Bytes	
Version Information	Device Model	UC2000-VF	
	Package Version	02231301 2018-07-10 17:04:31 official	
	Software Version	02231301 2018-07-10 16:51:31	
	Web Version	02231301	
	Hardware Version	PCB 2	
	Logic Version	LOGIC 0	
	DSP Version	Branch3.0.0.0	
	Userboard 0 Version	B5.3.2.33L51 C.2 DB10-6179-2120-0999	
	Userboard 1 Version	B5.3.2.33L51 C.2 DB10-6179-2120-1117	

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

4.3 System Information

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

4.3.1 System information

Figure 4-3-1 system Information

Run Information			
MAC Address	F8-A0-3D-48-99-4B		
Network Mode	Bridge		
Network	172.19.211.120	255.255.0.0	Static
DNS Server	114.114.114.114	0.0.0.0	
Device SN	db27-8310-2400-0171		
Hardware ID	0000-18e7-9dc8		
Cloud Register Status	Not Registered		
License	Basic Function	Enable	
	DBO Advanced	Enable	
System Up Duration	1 m 18 s		
System Time	2018-8-17 16:01:04		
Network Traffic Statistics	Received 76307 Bytes	Sent 360083 Bytes	
Version Information	Device Model	UC2000-VF	
	Package Version	02231301 2018-07-10 17:04:31 official	
	Software Version	02231301 2018-07-10 16:51:31	
	Web Version	02231301	
	Hardware Version	PCB 2	
	Logic Version	LOGIC 0	
	DSP Version	Branch3.0.0.0	
	Userboard 0 Version	B5.3.2.33L51 C.2 DB10-6179-2120-0999	
	Userboard 1 Version	B5.3.2.33L51 C.2 DB10-6179-2120-1117	

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 status	Its indicates communicate status with SIM Cloud server, there are two type of status: <ul style="list-style-type: none"> ▶ Registered ▶ Not Registered ▶ Need Authentication
License	It indicates device's license status. Contact with support when it displays 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 <ul style="list-style-type: none"> • Device Model: Model name of the device • Package version: 02231301 2018-07-10 17:04:31 official is the version number • Software version: 02231301 2018-07-10 17:04:31 official, 02231301 is the version number • Web version: the version number of web system. The web version must match with software • User board 0 Version: the firmware version of user board slot 0

- User board License ID: Contact with support when it displays asInvalid
- Hardware version/DSP version/ SIM box version

4.3.2 Mobile Information

Figure 4.3-2 Mobile Information

Mobile Information												
Port	Type	Network Mode	IMSI	IMEI	Status	Credits	Operator	Signal	ASR(%)	ACD(s)	PDD(s)	Call Status
0	LTE	LTE	460025169926509	863070016950486	Mobile Registered	No Limit	CMCC		0	0	0	Idle
1	LTE	GSM	460025169926510	860016012523184	Mobile Registered	No Limit	CMCC		0	0	0	Idle
2	LTE			990001005736301		No Limit			0	0	0	Idle
3	LTE			866297033042271		No Limit			0	0	0	Idle
4	LTE			863070012432026		No Limit			0	0	0	Idle
5	LTE			863070016135260		No Limit			0	0	0	Idle
6	LTE			863070018838317		No Limit			0	0	0	Idle
7	LTE			863070012081450		No Limit			0	0	0	Idle

Table 4.3-2 Mobile Information

Parameters	Description
Port	Number of GSM/CDMA ports.
Type	Indicates the current type of module. Such as GSM, CDMA, WCDMA, LTE
Network Mode	Indicates the current type of network. Such as GSM, CDMA, WCDMA, LTE
IMSI	International Mobile Subscriber Identity, it is the uniquely identifies of SIM card
IMEI	Module series NO
Status	Indicates the connection status of current GSM/CDMA /WCDMA/LTE module
Credits	It showing available total call credit or time of SIM card while Call Limit is enabled.
Operator	Displays the network carrier of current SIM card.
Signal	Displays the signal strength of in each channel of GSM/CDMA /WCDMA/LTE

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	<p>Show the Status of port, include idle, active, alert and processing</p> <p><i>Idle</i> means there is no call on this channel</p> <p><i>Processing</i> means call is connecting</p> <p><i>Alerting</i> means destination is ringing</p> <p><i>Active</i> means the call is connected</p> <p><i>Ringing</i> means the gateway is answering incoming call from mobile</p> <p><i>Calling Waiting</i> means the gateway is receiving another call during conversation and implement call waiting service</p> <p><i>Call Hold</i> means the call is hold by extension of IPPBX/SIP Server</p>

4.3.3 SIP Information

Figure 4-3-3 SIP Information

SIP Information							
Port	SIP User ID	Register Status	Port	SIP User ID	Register Status		
0	2018	Unregistered	1	2018	Unregistered		
2	2018	Unregistered	3	2018	Unregistered		
4	2018	Unregistered	5	2018	Unregistered		
6	2018	Unregistered	7	2018	Unregistered		
8	2018	Unregistered	9	2018	Unregistered		
10	2018	Unregistered	11	2018	Unregistered		
12	2018	Unregistered	13	2018	Unregistered		
14	2018	Unregistered	15	2018	Unregistered		
Port Group	SIP User ID	Register Status	Port List	Port Group	SIP User ID	Register Status	Port List
0		Unregistered	0,1,2,3,4,5,6,7,8,9,10,...				

Displays registration status information with Softswitch platform or SIP Server

Table 4-3-3 SIP information

Parameters	Description
Port	The number of 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.
Port Group	The number of SIP channels
Port List	The ports of the port groups contain

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 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/CMD/WCDMA/LTE 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 dial tone/ no carrier

4.4.5 CDR Report

Figure 4-4-5 CDR Report

The screenshot shows the CDR Report interface with the following details:

- Filters:**
 - Enable CDR: No Yes
 - Save CDR: No Yes
 - Check Boch: No Yes
 - Start Date: 2018 Year 8 Month 20 Day
 - End Date: 2018 Year 8 Month 20 Day
 - Min Duration: [] s
 - Max Duration: [] s
 - Select Port: All
 - Source: []
 - Destination: []
 - Rtp Loss Rate: [] % to [] %
- Buttons:** Export, Refresh, Delete the CDRs in this Report, save
- Table:**

Port	Start Date	Answer Date	Call Direction	Source	SourceIP	Destination	Hang Side	Reason	Duration(s)	Codec	Rtp Send	Rtp recv	Rtp loss Rate	jitter(s)	BCCH
1	2018/08/20 14:19:28	2018/08/20 14:19:42	IP->Gsm	950811	172.19.1.40	13767243151	Called	NORMAL HANG UP(16)	6	G.729AB	52	213	0%	0	
- Summary:** Total: 1 entries 50 entries/page 1/1 page Page 1

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

Parameters	Description
Port	GSM/CDMA /WCDMA/LTE port number
Start Date/Answer Date	start and end time of calls
Call 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
Source IP	Calling ip address

Destination	Called number
Hang Side	Who hang up the call, calling, called or gateway
Reason	The reason of the call hang up
Duration(s)	Call duration of the call
Codec	The voice code of this call, Include G.723.1/PCMA/PCMU/ G.729AB
RTP send/rcv/loss rate	RTP Statistics of the call
Jitter(s)	Voice jitter
BCCH	Which bcch the call using, first you need enable check bcch

4.4.6 Lock BCCH History

Figure 4-4-6 Auto Lock BCCH History

Index	BCCH	Signal Strength	Time
1	798	-73	2013-06-19 03:40:32

Recently 50 Times Record

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.

Port	Direction	Calling Number	Called Number	CODEC	Established Time	Duration
---	---	---	---	---	---	---

Refresh

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.

GSM Event								
Select Port	All	IMSI		Event	All	Export	Refresh	Clear
Port	IMSI	Time	Event	Number	Status	Duration(s)	Remark	
0	460020106218790	2017-04-05 06:35:17	Set IMEI	990001002582344	SUCCEED	0		

Total: 1 entries 20 entries/page 1/1 page Page 1

4.5 Network Configuration

4.5.1 Local Network

Figure 4-5-1 Local Network

Local Network

Network Configuration

Obtain IP address automatically
 Temporary IP address:
 Temporary Subnet Mask:

Use the following IP address
 IP Address:
 Subnet Mask:
 Default Gateway:

PPPoE
 Account:
 Password:
 Service Name:

MTU:

DNS Server

Obtain DNS server address automatically
 Use the following DNS server addresses
 Primary DNS Server:
 Secondary DNS Server:

Note: It must restart the device to take effect.

Save

Table 4-5-1 Local network

Parameters	Description
Obtain IP Address Automatically	Enable the device obtain IP Address automatically or not.
Temporary IP address and Subnet Mask	When device can't get the ip automatic, it will use the temporary IP

Use the Following IP Address	Configure the "IP Address", "Subnet Mask" and "Default Gateway" by manual, default this is enable, and default ip is 192.168.11.1
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

Add ARP

IP Address

MAC Address

The IP format is: xxx.xxx.xxx.xxx

The MAC format is: xx-xx-xx-xx-xx-xx

Click *Search All* to check ARP buffer.

ARP

Type static dynamic

	IP Address	MAC Address
<input type="checkbox"/>	172.16.221.43	BC-AE-C5-4E-15-F5
<input type="checkbox"/>	172.16.236.129	2C-D0-5A-12-D5-2A
<input type="checkbox"/>	172.16.10.10	00-0C-29-08-3D-91

4.5.3 VPN Parameter

Figure 4-5-3 VPN Parameter

VPN Parameter

VPN Enable

Server

Account

Password

Domain

Use MPPE

Table 4-5-3 Description of VPN Parameter

Parameters	Description
Server	VPN Server IP or domain name (support PPTP only)
Account	VPN account which provides 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

Run Information			
MAC Address	00-12-34-56-78-00		
Network Mode	Bridge		
Network	0.0.0.0	0.0.0.0	Static
DNS Server	8.8.8.8	0.0.0.0	
Device ID	0000-0000-0000-0000		
Server Register Status	Not Registered		
VPN Connection Status	Connecting		
VPN Server	us1.suvpn.com		
VPN Local IP			
VPN Remote IP			

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.

Access Rules - Add

Index	0		
Action	Drop		
Source IP	any	/	255.255.255.0
Protocol	TCP		
Source Port	0	-	65535
Dest Port	0	-	65535
Description	<input style="width: 100%;" type="text"/>		
Enable/Disable	<input checked="" type="radio"/> Enable <input type="radio"/> Disable		

4.7 Mobile Configuration

4.7.1 Basic Configuration

Basic Configuration

Basic Configuration	
SIM Mode	Local
API	<input type="radio"/> Disable <input checked="" type="radio"/> Old Version <input type="radio"/> New Version <input type="radio"/> SMPP
API Server Address	0.0.0.0
API Server Port	0
API User ID	
API User Password	●●●●●●●●●● <input type="button" value="Show Password"/>
Sms Report Filter	<input checked="" type="radio"/> No <input type="radio"/> Yes
USSD Default Encoding	UCS2
GSM Audio Coding	AUTO

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/USDD on the gateway.

Old version:

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.

New version:

The API is based on HTTP and JSON. So please check how to send HTTP request and how to encode/decode JSON data before you write an application with this API. please contact support for the document.

Smpp:

SMPP function support from client after binding to send text messages, length and mass text messages, and support to receive text messages, receive SMS receipt, send the query results, and other functions at present average.

Configured SMPP listener port and the user password, and then restart the equipment. GSM gateway is smpp server, it can connect with smpp client.

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:

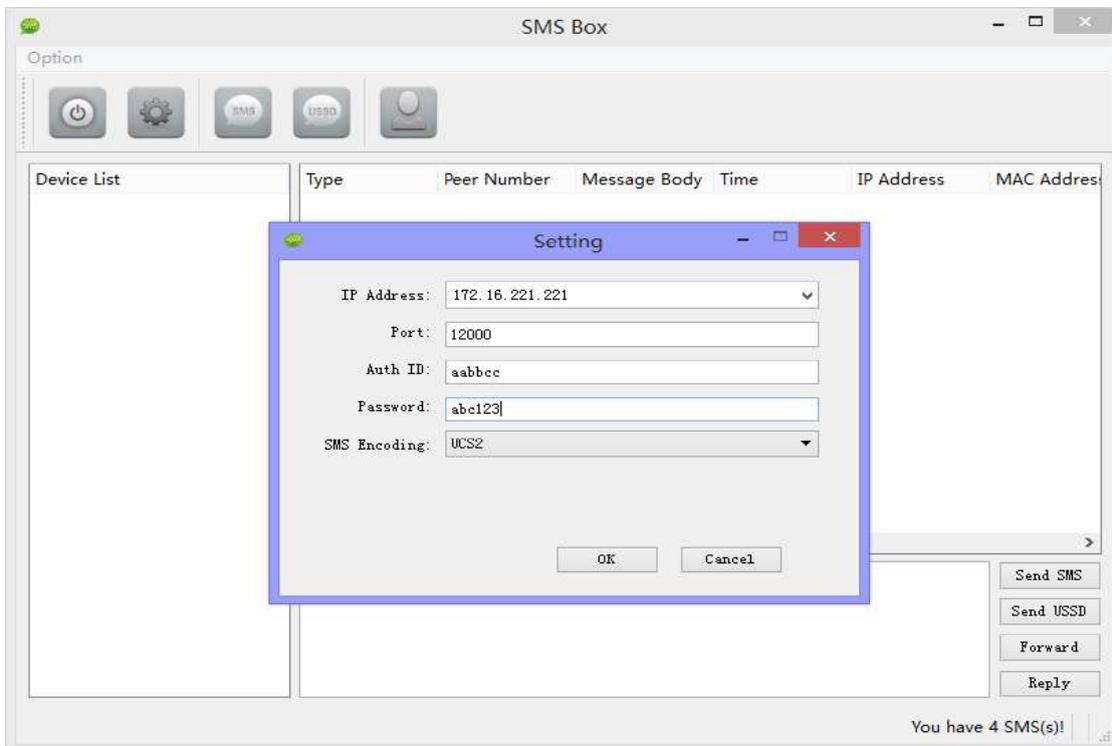
Configuration between SMS box and gateway

Configure API parameters on gateway

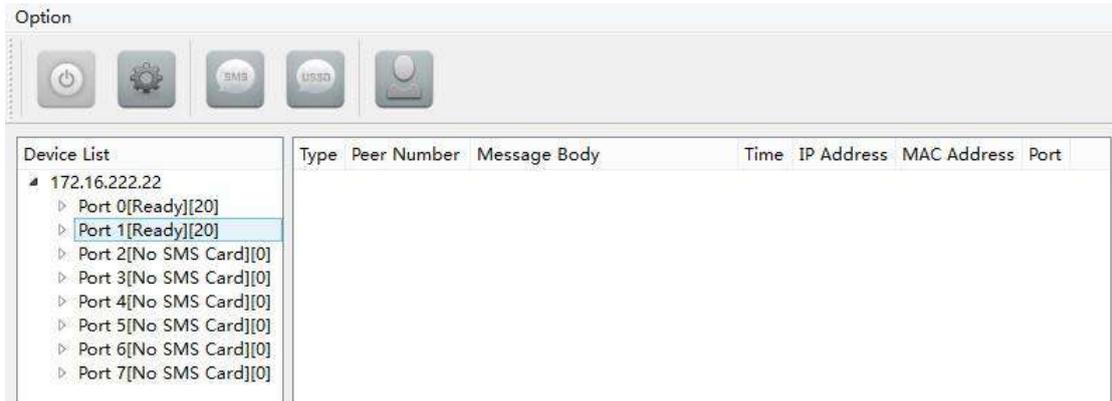
Remote API Enable	<input type="radio"/> No <input checked="" type="radio"/> Yes
API Server Address	172.16.221.221
API Server Port	12000
API User ID	aabbcc
API User Password <input type="button" value="Show Password"/>

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

Port	CLIR	Detect Reverse Polarity	Internet Access	Tx Gain/dB	Rx Gain/dB	APN	APN name	APN PSW	Band Type	Network Mode	SMSC	Reset	Block /Open	Power On/Off
<input type="checkbox"/> 0	No	Yes	No	3	7				Default (Auto)	Default (Auto)	+8613800755500	Reset	Block	OFF
<input type="checkbox"/> 1	No	Yes	No	3	7				Default (Auto)	Default (Auto)	+8613800755500	Reset	Block	OFF
<input type="checkbox"/> 2	No	Yes	No	3	7				Default (Auto)	Default (Auto)		Reset	Block	OFF
<input type="checkbox"/> 3	No	Yes	No	3	7				Default (Auto)	Default (Auto)		Reset	Block	OFF
<input type="checkbox"/> 4	No	Yes	No	3	7				Default (Auto)	Default (Auto)		Reset	Block	OFF
<input type="checkbox"/> 5	No	Yes	No	3	7				Default (Auto)	Default (Auto)		Reset	Block	OFF
<input type="checkbox"/> 6	No	Yes	No	3	7				Default (Auto)	Default (Auto)		Reset	Block	OFF
<input type="checkbox"/> 7	No	Yes	No	3	7				Default (Auto)	Default (Auto)		Reset	Block	OFF

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, this needs support by the operator, if operator not support, device also can't do it.
Detect Reverse Polarity	If the function is enabled, the caller will learn whether the called person has got through the phone.
Internet Access	Allow the sim cards access internet or not. For example, if you want enable auto Internet access, pleaseenable this.
Tx Gain	Gain of voice sent

Rx Gain	Gain of voice received
APN/APN name/APN PSW	APN refers to a network access technology, which is a parameter that must be configured when accessing the Internet via mobile phone.
Band Type	Choose from GSM850, GSM900, GSM1800, GSM1900, WCDMA800, WCDMA 850, WCDMA900, WCDMA1900, and WCDMA2100
Network Mode	Select 2G ,3G or 4G
SMSC	Short Message Service Center
Reset	Click Reset , and the corresponding module will be reset.
Block/Open	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.

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.

Phone Number Config

Port	SlotA	SlotB	SlotC	SlotD
0	<input style="width: 80%;" type="text" value="13388889999"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>
1	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>
2	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>
3	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>
4	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>
5	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>
6	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>
7	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>	<input style="width: 80%;" type="text"/>

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 predefined conditions

IMEI Auto Set			
Enable IMEI Auto Set	<input type="radio"/> No	<input checked="" type="radio"/> Yes	
IMEI TAC	<input type="text" value="86008501"/>	<input type="text" value="99000100"/>	<input type="text" value="86000300"/>
	<input type="text" value="86011600"/>	<input type="text" value="86014300"/>	<input type="text" value="86275202"/>
IMEI Serial No Range	Min IMEI Serial No	<input type="text" value="111111"/>	
	Max IMEI Serial No	<input type="text" value="888890"/>	
Policy	<input type="checkbox"/> By Time	<input type="text" value="10"/>	Minute(s) (Note:Not less than 10)
	<input type="checkbox"/> By Call	<input type="text" value="30"/>	Call(s) (Note:Not less than 10)
	<input checked="" type="checkbox"/> Change Sim		

Note:1. IMEI = TAC(8 digits) + Serial No(6 digits) + Check digit(1 digit).
2. There is up to 8 TACs can be set.

4.7.6 Operator

Operator			
Port	Operator code	Operator List	Search
<input type="checkbox"/> 0	<input type="text"/>	<input type="text"/>	<input type="button" value="Search"/>
<input type="checkbox"/> 1	<input type="text"/>	<input type="text"/>	<input type="button" value="Search"/>
<input type="checkbox"/> 2	<input type="text"/>	<input type="text"/>	<input type="button" value="Search"/>
<input type="checkbox"/> 3	<input type="text"/>	<input type="text"/>	<input type="button" value="Search"/>
<input type="checkbox"/> 4	<input type="text"/>	<input type="text"/>	<input type="button" value="Search"/>
<input type="checkbox"/> 5	<input type="text"/>	<input type="text"/>	<input type="button" value="Search"/>
<input type="checkbox"/> 6	<input type="text"/>	<input type="text"/>	<input type="button" value="Search"/>
<input type="checkbox"/> 7	<input type="text"/>	<input type="text"/>	<input type="button" value="Search"/>
<input type="checkbox"/> all	<input type="button" value="Copy"/>		<input type="button" value="Search"/>
	<input type="text"/>		

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

4.7.7 Operator Configuration

Operator rule			
	IMSI prefix	Rule	Operator
<input checked="" type="checkbox"/>	<input type="text" value="460038"/>	<input type="text" value="Fixed ▼"/>	<input type="text" value="46003"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text" value="Auto ▼"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text" value="Auto ▼"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text" value="Auto ▼"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text" value="Auto ▼"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text" value="Auto ▼"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text" value="Auto ▼"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>	<input type="text" value="Auto ▼"/>	<input type="text"/>

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

Overview		Whitelist																									
Port	Mode	0			1			2			3			4			5			6			Detail				
		LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm		
0	Default	0XA4F	38950	-51	0XA4F	38950	-51	0XA4F	38950	0	0XA4F	38950	0	0XA4F	38950	-1920	0XA4F	38950	-1920	0XA4F	38950	-1920	0XA4F	38950	-1920		Detail
1	Default	0XA4F	38950	-51	0XA4F	38950	-51	0XA4F	38950	0	0XA4F	38950	0	0XA4F	38950	-1920	0XA4F	38950	-1920	0XA4F	38950	-1920	0XA4F	38950	-1920		Detail
2	Default																										Detail
3	Default																										Detail
4	Default																										Detail
5	Default																										Detail
6	Default																										Detail
7	Default																										Detail
8	Default																										Detail
9	Default																										Detail
10	Default																										Detail
11	Default	0X3614	283	0																							Detail
12	Default																										Detail
13	Default																										Detail
14	Default																										Detail
15	Default																										Detail

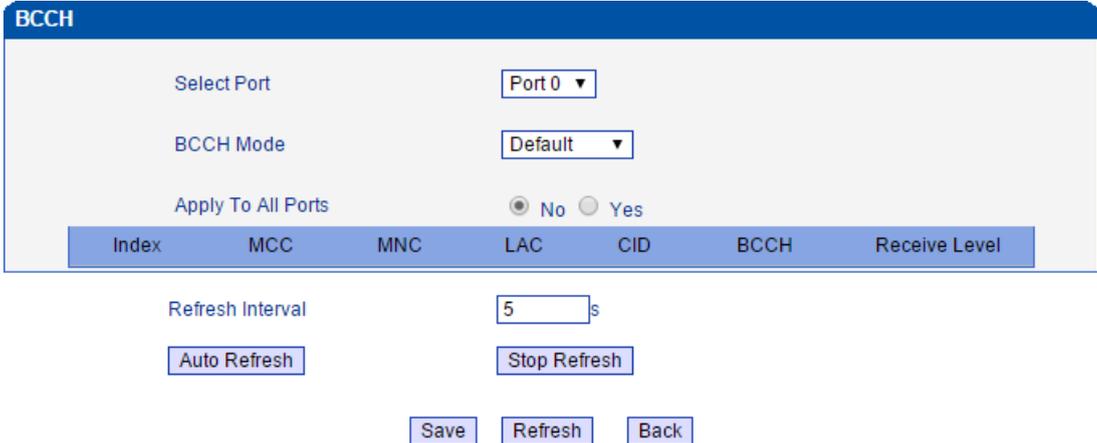
BCCH (Broadcast Control Channel): BCCH is a logical broadcast channel used by the base station in a GSM/WCDMA network to send information about the identity of the network. The information is used by a mobile station to get access to the network. Information includes the Mobile Network Code (MNC), the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

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



BCCH

Select Port: Port 0

BCCH Mode: Default

Apply To All Ports: No Yes

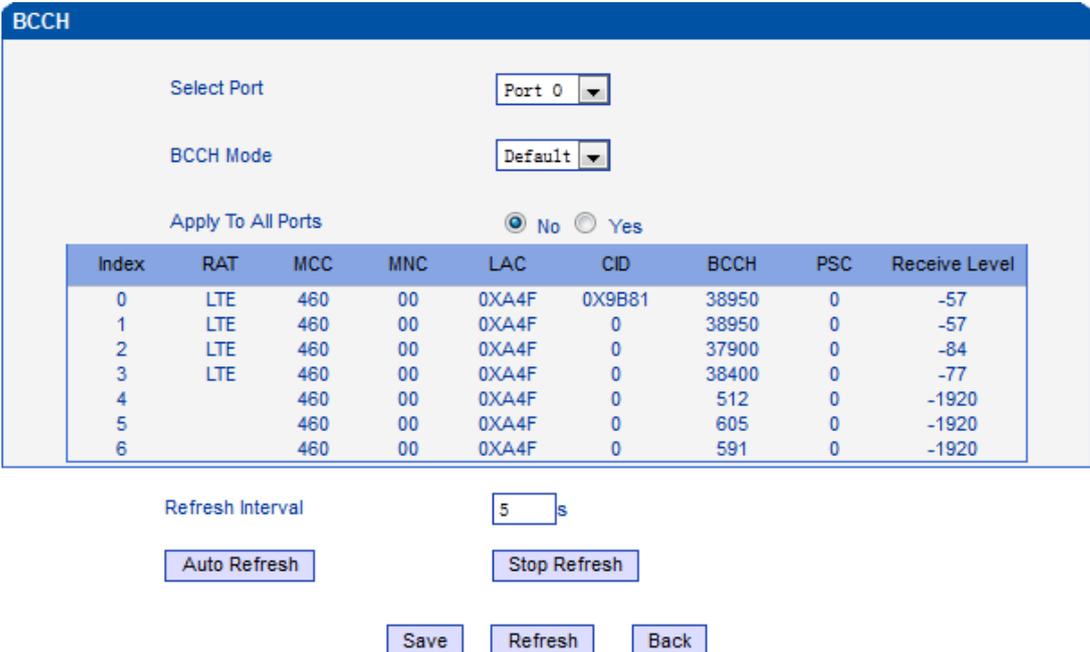
Index	MCC	MNC	LAC	CID	BCCH	Receive Level

Refresh Interval: 5 s

Auto Refresh Stop Refresh

Save Refresh Back

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



BCCH

Select Port: Port 0

BCCH Mode: Default

Apply To All Ports: No Yes

Index	RAT	MCC	MNC	LAC	CID	BCCH	PSC	Receive Level
0	LTE	460	00	0XA4F	0X9B81	38950	0	-57
1	LTE	460	00	0XA4F	0	38950	0	-57
2	LTE	460	00	0XA4F	0	37900	0	-84
3	LTE	460	00	0XA4F	0	38400	0	-77
4		460	00	0XA4F	0	512	0	-1920
5		460	00	0XA4F	0	605	0	-1920
6		460	00	0XA4F	0	591	0	-1920

Refresh Interval: 5 s

Auto Refresh Stop Refresh

Save Refresh Back

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.

BCCH						
Select Port	Port 0 ▼					
BCCH Mode	Fixed ▼					
Frist of BCCH	<input type="text"/>					
Second of BCCH	<input type="text"/>					
Third of BCCH	<input type="text"/>					
Index	MCC	MNC	LAC	CID	BCCH	Receive Level

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.

BCCH							
Select Port	Port 0 ▼						
BCCH Mode	Random ▼						
Minimum Signal Strength allow	<input type="text" value="-90"/>					db	
Auto Period between	<input type="text" value="1"/>	and				<input type="text" value="15"/>	min
Switch BCCH in Calling	<input checked="" type="radio"/> No <input type="radio"/> Yes						
Apply To All Ports	<input checked="" type="radio"/> No <input type="radio"/> Yes						
Index	MCC	MNC	LAC	CID	BCCH	Receive Level	

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.

BCCH							
Select Port	Port 0 ▼						
BCCH Mode	Advanced ▼						
Minimum Signal Strength allow	<input type="text" value="-90"/>					db	
Call Times	<input type="text" value="15"/>	Minimun ASR	<input type="text" value="30"/>				%
Call Failed	<input type="text" value="6"/>						
Apply To All Ports	<input checked="" type="radio"/> No <input type="radio"/> Yes						
Index	MCC	MNC	LAC	CID	BCCH	Receive Level	

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.

Overview
Whitelist

Enable

Auto Generate Whitelist By Scanning

Port:

Number:

IDLE

Operator	LAC	CID	BCCH	LAC	CID	BCCH
0	<input type="text" value="2639"/>	<input type="text" value="ef8"/>	<input type="text" value="595"/>	1	<input type="text" value="2639"/>	<input type="text" value="e88"/>
1	<input type="text" value="46000"/>	<input type="text" value="eb3"/>	<input type="text" value="604"/>	2	<input type="text" value="2639"/>	<input type="text" value="1151"/>
2	<input type="text"/>	<input type="text" value="ef7"/>	<input type="text" value="593"/>	3	<input type="text"/>	<input type="text"/>
3	<input type="text"/>	<input type="text"/>	<input type="text"/>	4	<input type="text"/>	<input type="text"/>
4	<input type="text"/>	<input type="text"/>	<input type="text"/>	5	<input type="text"/>	<input type="text"/>
5	<input type="text"/>	<input type="text"/>	<input type="text"/>	6	<input type="text"/>	<input type="text"/>
6	<input type="text"/>	<input type="text"/>	<input type="text"/>	7	<input type="text"/>	<input type="text"/>
7	<input type="text"/>	<input type="text"/>	<input type="text"/>	8	<input type="text"/>	<input type="text"/>
				9	<input type="text"/>	<input type="text"/>
				10	<input type="text"/>	<input type="text"/>
				11	<input type="text"/>	<input type="text"/>
				12	<input type="text"/>	<input type="text"/>
				13	<input type="text"/>	<input type="text"/>
				14	<input type="text"/>	<input type="text"/>
				15	<input type="text"/>	<input type="text"/>
				16	<input type="text"/>	<input type="text"/>
				17	<input type="text"/>	<input type="text"/>
				18	<input type="text"/>	<input type="text"/>
				19	<input type="text"/>	<input type="text"/>

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

Only GSM module support Fixed/Random/Advance mode, other modules don't support.

4.7.9 Call Forwarding

Calls can be forwarded unconditionally or under certain conditions.

Call Forwarding			
Port	Options	Call forwarding settings	Search
<input type="checkbox"/> 0	<input type="text"/>		<input type="button" value="Search"/>
<input type="checkbox"/> 1	<input type="text"/>		<input type="button" value="Search"/>
<input type="checkbox"/> 2	<input type="text"/>		<input type="button" value="Search"/>
<input type="checkbox"/> 3	<input type="text"/>		<input type="button" value="Search"/>
<input type="checkbox"/> 4	<input type="text"/>		<input type="button" value="Search"/>
<input type="checkbox"/> 5	<input type="text"/>		<input type="button" value="Search"/>
<input type="checkbox"/> 6	<input type="text"/>		<input type="button" value="Search"/>
<input type="checkbox"/> 7	<input type="text"/>		<input type="button" value="Search"/>
<input type="checkbox"/> all	Unconditional ▼	Call Forwarding Unconditional <input type="text"/>	
<input type="button" value="Copy"/>			<input type="button" value="Search"/>

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			Results
Port	Setting		
<input type="checkbox"/> 0	<input type="radio"/> Disable	<input type="radio"/> Enable	No SIM card.
<input type="checkbox"/> 1	<input type="radio"/> Disable	<input type="radio"/> Enable	No SIM card.
<input type="checkbox"/> 2	<input type="radio"/> Disable	<input type="radio"/> Enable	No SIM card.
<input type="checkbox"/> 3	<input type="radio"/> Disable	<input type="radio"/> Enable	No SIM card.
<input type="checkbox"/> 4	<input type="radio"/> Disable	<input type="radio"/> Enable	No SIM card.
<input type="checkbox"/> 5	<input type="radio"/> Disable	<input type="radio"/> Enable	No SIM card.
<input type="checkbox"/> 6	<input type="radio"/> Disable	<input type="radio"/> Enable	No SIM card.
<input type="checkbox"/> 7	<input type="radio"/> Disable	<input type="radio"/> Enable	No SIM card.
<input type="checkbox"/> All	<input type="button" value="Copy"/>		
	<input type="radio"/> Disable	<input type="radio"/> Enable	

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 only takes effective while “Do Not Answer GSM Incoming Call for Hotline” set to Yes.

Call Configuration -> Service Parameter

Do Not Answer GSM Incoming Call for Hotline No Yes

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.

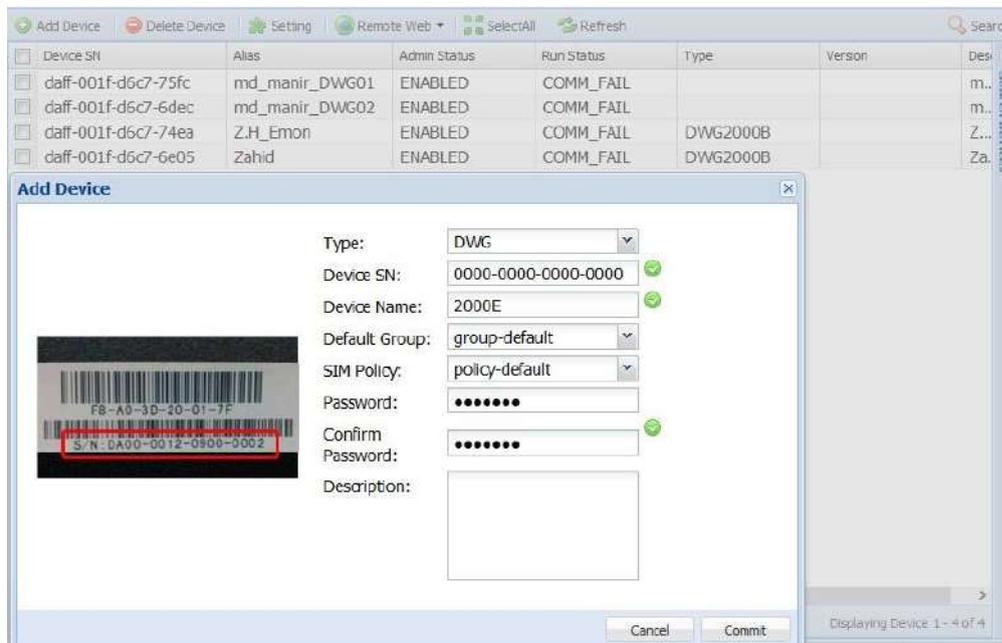
Cloud Server	
Primary Server Domain	<input type="text" value="best.cloud.com"/>
Primary Server Port	<input type="text" value="2020"/>
Secondary Server Domain	<input type="text"/>
Secondary Server Port	<input type="text"/>
Domain Name	<input type="text"/>
Password	<input type="password"/> <input type="button" value="Show Password"/>
LocalPort	<input type="text" value="0"/>
SIM Transport Type	Auto ▼
Port State Control by	Cloud ▼
Anti Call Scanning	<input type="checkbox"/> Enable
Reboot device when register failed over	<input type="text" value="0"/> minutes (0 means disable, value: 2~60)

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 Cloud server.
Password	The password of the sub-domain used by the gateway under the Cloud 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:

Device ID	0000-0000-0000-0000
Server Register Status	Not Registered

4.7.12 MBN Config

MBN Config			
Port	Stat	MBN File	Config
<input type="checkbox"/>	0	<input type="text"/> Down	*VoIte_OpenMkt-Commer ▾ Active Delete
<input type="checkbox"/>	1	<input type="text"/> Down	*VoIte_OpenMkt-Commer ▾ Active Delete
<input type="checkbox"/>	2	<input type="text"/> Down	*OpenMkt-Commercial-C ▾ Active Delete
<input type="checkbox"/>	3	<input type="text"/> Down	*VoIte_OpenMkt-Commer ▾ Active Delete
<input type="checkbox"/>	4	<input type="text"/> Down	*OpenMkt-Commercial-C ▾ Active Delete
<input type="checkbox"/>	5	<input type="text"/> Down	*VoIte_OpenMkt-Commer ▾ Active Delete
<input type="checkbox"/>	6	<input type="text"/> Down	*ROW_Generic_3GPP ▾ Active Delete
<input type="checkbox"/>	7	<input type="text"/> Down	*OpenMkt-Commercial-C ▾ Active Delete
<input type="checkbox"/>	8	<input type="text"/> Down	*OpenMkt-Commercial-C ▾ Active Delete
<input type="checkbox"/>	9	<input type="text"/> Down	*OpenMkt-Commercial-C ▾ Active Delete
<input type="checkbox"/>	10	<input type="text"/> Down	*OpenMkt-Commercial-C ▾ Active Delete
<input type="checkbox"/>	11	<input type="text"/> Down	*OpenMkt-Commercial-C ▾ Active Delete
<input type="checkbox"/>	12	<input type="text"/> Down	*ROW_Generic_3GPP ▾ Active Delete
<input type="checkbox"/>	13	<input type="text"/> Down	*ROW_Generic_3GPP ▾ Active Delete
<input type="checkbox"/>	14	<input type="text"/> Down	*ROW_Generic_3GPP ▾ Active Delete
<input type="checkbox"/>	15	<input type="text"/> Down	*ROW_Generic_3GPP ▾ Active Delete

Select All

NOTE: 1.After activation you need to restart module to take effect!
 2.Don't refresh the page during the operation!
 3.Down,activate,Delete cannot operate at the same time!

This tool used to help update LTE module MBN file.

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.

Overview						
	Port	Current Day Send Count	Daily Limit	Current Month Send Count	Monthly Limit	Reset Date
<input type="checkbox"/>	0	--	--	--	--	--
<input type="checkbox"/>	1	--	--	--	--	--
<input type="checkbox"/>	2	--	--	--	--	--
<input type="checkbox"/>	3	--	--	--	--	--
<input type="checkbox"/>	4	--	--	--	--	--
<input type="checkbox"/>	5	--	--	--	--	--
<input type="checkbox"/>	6	--	--	--	--	--
<input type="checkbox"/>	7	--	--	--	--	--
<input type="checkbox"/>	All	<input type="button" value="Clear"/>		<input type="button" value="Clear"/>		

4.8.2 SMS Send Limit Settings

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

SMS Send Limit Settings - Add Rule

Index	<input type="text" value="0"/>	
Description	<input type="text" value="Vodafone"/>	
Daily Limit	<input type="text" value="0"/>	Note:0 means no limit
Monthly Limit	<input type="text" value="20"/>	Note:0 means no limit
Reset Date	<input type="text" value="1"/>	
Port Group	<input type="text" value="0 <all>"/>	

4.8.3 SMS Routing

You can set sms routing if you use gateway as sms terminal.

SMS Routing Rule					
	Source	Destination Number	Destination Port	Digits to be Deleted	Prefix to Add
<input type="checkbox"/>	SMPP	<input type="text"/>	port-0	<input type="text"/>	<input type="text"/>
<input type="checkbox"/>	API	<input type="text"/>	port-group-0 <all>	<input type="text"/>	<input type="text"/>
<input type="checkbox"/>	SMPP	<input type="text"/>	Any	<input type="text"/>	<input type="text"/>
<input type="checkbox"/>	SMPP	<input type="text"/>	Any	<input type="text"/>	<input type="text"/>

4.8.4 SMSC Switch Setting

Every operator can set 8 SMSC, it will switchover by the number you set successful or fail.

For example:

Successful send set 5, fail send set 1,

When the port send 5 sms successful, it will switch to next SMSC.

When the port send 1 sms fail, it will switch to next SMSC.

SMSC Switch Setting										
	Operator	SMSC0	SMSC1	SMSC2	SMSC3	SMSC4	SMSC5	SMSC6	SMSC7	
<input type="checkbox"/> 0	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	
<input type="checkbox"/> 1	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	
<input type="checkbox"/> 2	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	
<input type="checkbox"/> 3	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	
<input type="checkbox"/> 4	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	
<input type="checkbox"/> 5	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	
<input type="checkbox"/> 6	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	
<input type="checkbox"/> 7	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	
<input type="checkbox"/> All										
Policy:		Successful send					Fail Send			
		<input type="text" value="0"/>					<input type="text" value="0"/>			
<input type="button" value="Save"/>										

4.8.5 Send SMS

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

Send Message	
Port	<input type="checkbox"/> 0 <input type="checkbox"/> 1 <input type="checkbox"/> 2 <input type="checkbox"/> 3 <input type="checkbox"/> 4 <input type="checkbox"/> 5 <input type="checkbox"/> 6 <input type="checkbox"/> 7 <input type="checkbox"/> All
Send Mode	<input checked="" type="radio"/> Mode 1 <input type="radio"/> Mode 2
To	<input type="text"/>
Encoding Message	UCS2 ▼ <input type="text"/>

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

Detail Report		
Port	Destination Number	Result

4.8.6 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.

SMS Outbox						
Start Date :	2010	Year	1	Month	1	Day
End Date :	2018	Year	8	Month	22	Day
IMSI :						
Select Port	A11		Number			
Send Status	ANY					
Report Export	Export		Refresh		Clear	
Port	IMSI	Send Date	Number	SMS Content		Send Status
Total: 0 entries 16 entries/page 1/0 page						

4.8.7 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.

SMS Inbox						
Save File	<input checked="" type="radio"/> No <input type="radio"/> Yes		save		Number	
Start Date :	2010	Year	1	Month	1	Day
End Date :	2018	Year	8	Month	22	Day
IMSI :						
Select Port	A11		Number			
Report Export	Export		Refresh		Clear	
Port	IMSI	Number	Date,Time	SMS Content		
Total: 0 entries 16 entries/page 1/0 page						

4.8.8 USSD

USSD (Unstructured Supplementary Service Data): is a service which is provided by a telecom operator and allows GSM/WCDMA/LTE mobile phones to interact with the telecom operator's computers. USSD messages travel over GSM/WCDMA/LTE 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.

USSD		
Port	USSD Request	USSD Reply
<input checked="" type="checkbox"/> 0		not registered
<input checked="" type="checkbox"/> 1		not registered
<input checked="" type="checkbox"/> 2		not registered
<input checked="" type="checkbox"/> 3		not registered
<input checked="" type="checkbox"/> 4		not registered
<input checked="" type="checkbox"/> 5		not registered
<input checked="" type="checkbox"/> 6		not registered
<input checked="" type="checkbox"/> 7		not registered

All

4.8.9 Email

Email Setting

SMS to Email Enable

Destination Email Address 1

Destination Email Address 2

Destination Email Address 3

Title

Email to SMS Enable

Check Email Every Minutes(Up to 60)

Subject

[Email Account Setting](#)

Notes: 1. "Email To SMS" only supports character sets of UTF8 and ASCII.
 It is recommended to use the "Plain Text Mode" (if there exists) when you intends to send an email, or to check whether you have chosen "using UNICODE to send email" in the mailbox setting.
 2. It is suggested that the text length of an email should be no more than 300 characters. Email download may fail in case of excessive text length.

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.

Check Email Every: how long time to check the mailbox.

Subject: Mail subject, gateway will check mailbox subject, if same, it will forward to sms.

Email Setting

Email Account

E-mail: support@dinstar.com

Username: support@dinstar.com

Password: ●●●●●●●●

Outgoing(SMTP)

Server: smtp.dinstar.com

Port: 25

TLS Enable:

Incoming

Protocol: POP3

Server: pop3.dinstar.com

Port: 110

TLS Enable:

Notes: 1. "Email To SMS" only supports character sets of UTF8 and ASCII.
It is recommended to use the "Plain Text Mode" (if there exists) when you intends to send an email, or to check whether you have chosen using UNICODE to send email.
2. It is suggested that the text length of an email should be no more than 300 characters. Email download may fail in case of excessive text length.

Email Account: Configure one e-mail address, which will be used for sending the SMS to destination e-mail.

Outgoing: 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.

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

Incoming: set the outgoing protocol, server, and port, different e-mail address server have different server addresses, please confirm this with your e-mail provider or search from Internet.

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

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.

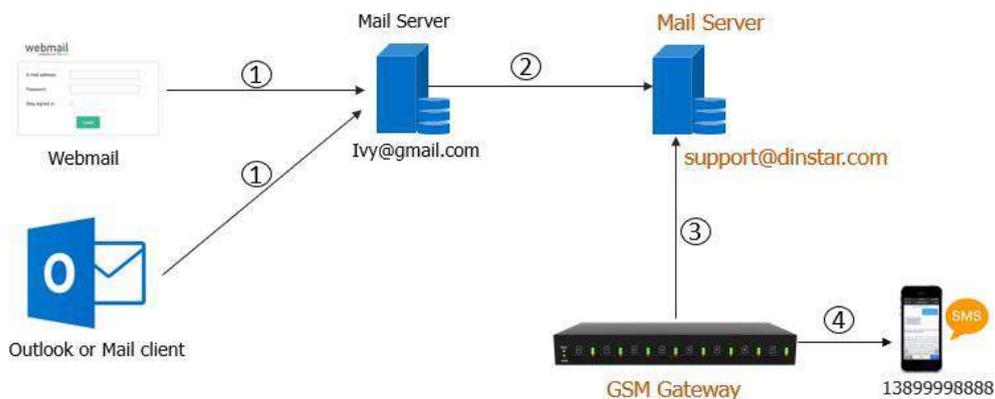
When email use protocol IMAP, if email read successful, will set the email status to read. If read email failed, will try to read again (MAX 3 times). If failed final, the email status will be set to read also.

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



刘 刘晓云	Test SMS - To:1008611 Encoding:7Bit Message:Hello this is test SMS from phoebe
刘 刘晓云	Test SMS - To:13670200513 Encoding:7Bit Message:Hola, this is test SMS from phoebe@dinstar.com

SMS Outbox							
Start Date:	2010	Year	1	Month	1	Day	
End Date:	2018	Year	8	Month	22	Day	
IMSI:						Number	<input type="text"/>
Report Export	Export		Refresh		Clear		
Port	IMSI	Send Date	Number	SMS Content		Send Status	
1	460025169926510	2018/08/22 21:21:48	1008611	Hello this is test SMS from phoebe		Sent	

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

1) Send Email

Email Format: Plain text

Email subject: Test SMS

Email contents:

To:1008611

Encoding:7Bit

Message: Hello this is test SMS from phoebe

- 2) mail 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 1008611

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" 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 25, POP3 default server port is 110.

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.

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:xxxxxxxxxxx

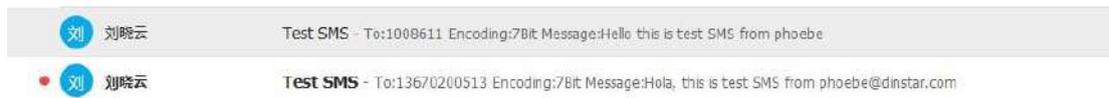
The “Encoding” means which format of SMS used, the format include 7Bit and UCS2, UCS2 is default. This option is Optional. The format is:

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 screenshot shows the 'SMS Inbox' interface. At the top, there are filters for 'Save File' (No/Yes), 'Start Date' (2010 Year, 1 Month, 1 Day), 'End Date' (2018 Year, 8 Month, 22 Day), 'Select Port' (All), and 'Number'. Below the filters are 'Report Export' and 'Export' buttons, along with 'Refresh' and 'Clear' buttons. The main area is a table with columns: Port, IMSI, Number, Date, Time, and SMS Content. The first row shows a message from +86136 at 2018/08/22 21:39:21 with content 'hi, this is test message from phoebe.'. The second row shows a message from 1008611 at 2018/08/22 21:21:37 with Chinese content. The third row shows a message from +8613670200513 at 2018/08/22 17:59:59 with content 'hi'. Below the table is a pagination control: 'Total: 3 entries 16 entries/page 1/1 page Page 1'. Below the table is a detailed view of the first message with fields: 发件人: support@dinstar.com, 收件人: phoebe@dinstar.com, 抄送: (empty), 主题: 11. The message content is: From IP:172.19.211.120 MAC:F8-A0-3D-48-99-4B Port:0, Phone Number:+86136, Time:2018-08-22 21:39:21 0800 hi,this is test message from phoebe.

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-->Email** page, enable SMS to Email function, and configure the other parameters needed.

4.9 Call Configuration

4.9.1 SIP Configuration

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

► Configure SIP server and Outbound Proxy server

The screenshot shows the configuration interface for SIP Proxy and Outbound Proxy. Under 'SIP Proxy', there are fields for 'SIP Server Address' (empty), 'SIP Server Port(default: 5060)' (5060), and 'Check Net Status' (radio buttons for No and Yes, with No selected). Under 'Outbound Proxy', there are fields for 'Outbound Proxy Address' (empty) and 'Outbound Proxy Port' (5060).

▶ 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 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.

All Ports Register Used Same User ID	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use Same Local Sip Port	<input type="radio"/> No <input checked="" type="radio"/> Yes
Use Random Port	<input checked="" type="radio"/> No <input type="radio"/> Yes
Local SIP Port	<input type="text" value="5060"/>

▶ Random

The gateway will generate 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	Authenticate Password	Local SIP Port	Register to	Tx Gain	Rx Gain	
<input type="checkbox"/>	0	<input type="text" value="1000"/>	<input type="text" value="1000"/>	<input type="text" value="●●●●"/>	<input type="text"/>	Sip_Proxy	-1dB	+6dB

► **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	Authenticate Password	Local SIP Port	Register to	Tx Gain	Rx Gain
<input type="checkbox"/> 0	1000	1000	●●●●	5060	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 1	1001	1001	●●●●	5061	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 2	1002	1002	●●●●	5062	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 3	1003	1003	●●●●	5063	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 4	1004	1004	●●●●	5064	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 5	1005	1005	●●●●	5065	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 6	1006	1006	●●●●	5066	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 7	1007	1007	●●●●	5067	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 8	1008	1008	●●●●	5068	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 9	1009	1009	●●●●	5069	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 10	1010	1010	●●●●	5070	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 11	1011	1011	●●●●	5071	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 12	1012	1012	●●●●	5072	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 13	1013	1013	●●●●	5073	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 14	1014	1014	●●●●	5074	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> 15	1015	1015	●●●●	5075	Sip Proxy	-1dB	+6dB
<input type="checkbox"/> All	<input type="text" value="1"/> <input type="button" value="Increment"/>	<input type="button" value="Copy"/>	<input type="button" value="Copy"/>	<input type="button" value="Copy"/>			
	<input type="text" value="1000"/>	<input type="text" value="1000"/>	<input type="text" value="●●●●"/>	<input type="text" value="5060"/>	Sip Proxy	-1dB	+6dB
<input type="button" value="Save"/>							

► **Auto set SIP Account and Router**

The gateway will generate sip account and router auto.

After set, It must restart the device to take effect.

Auto SIP Account No Yes

SIP User ID
 Authenticate ID
 Authenticate Password
 To VOIP Hotline
 To PSTN Hotline

Auto IP->TEL Route No Yes

Source
 Source Prefix
 Destination Prefix
 Prefix to Add
 Digits to be Deleted
 Number of Digits Reserved

Port
 Empty
 Empty
 Empty
 Empty

Iccid

► Auto SIP Account: Prefix+Port/Iccid/IMSI

Auto SIP Account No Yes

SIP User ID + ▼

Authenticate ID + ▼

Authenticate Password + ▼

To VOIP Hotline + ▼

To PSTN Hotline + ▼

Port	SIP User ID	Authenticate ID	Authenticate Password	Local SIP Port	Register to	Tx Gain	Rx Gain	To VOIP Hotline	To PSTN Hotline
<input type="checkbox"/> 0	<input type="text" value="1000"/>	<input type="text" value="200600216990"/>	<input type="text" value="●●●●●●"/>	<input type="text"/>	Sip Proxy	-1dB	+6dB	<input type="text" value="10"/>	<input type="text" value="24802216990"/>
<input type="checkbox"/> 1	<input type="text" value="1001"/>	<input type="text" value="200600216990"/>	<input type="text" value="●●●●●●"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text" value="11"/>	<input type="text" value="24802216990"/>
<input type="checkbox"/> 2	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 3	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 4	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 5	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 6	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 7	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 8	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 9	<input type="text" value="1009"/>	<input type="text" value="2006011090210"/>	<input type="text" value="●●●●●●"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text" value="19"/>	<input type="text" value="2480110902100"/>
<input type="checkbox"/> 10	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 11	<input type="text" value="10011"/>	<input type="text" value="2006011000120"/>	<input type="text" value="●●●●●●"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text" value="111"/>	<input type="text" value="2480110001200"/>
<input type="checkbox"/> 12	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 13	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 14	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 15	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> All	<input type="text" value="1"/> <input type="button" value="Increment"/>	<input type="text" value="1"/> <input type="button" value="Increment"/>	<input type="text" value="0"/> <input type="button" value="Increment"/>	<input type="text" value="1"/> <input type="button" value="Increment"/>	<input type="button" value="Copy"/>				
	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	-1dB	+6dB	<input type="text"/>	<input type="text"/>

► Auto IP->TEL Route: Source/Destination + Iccid/IMSI/Number

Auto IP->TEL Route No Yes

Source ▼

Source Prefix + ▼

Destination Prefix

Prefix to Add

Digits to be Deleted

Number of Digits Reserved

Index	Description	Source	Destination	Call Restriction	Source Prefix	Destination Prefix	Prefix to Add	Digits to be Deleted	Number of Digits Reserved
<input type="checkbox"/> 0	port-0	SIP Server	Port-0	Allow	8884600251...	--	--	--	--
<input type="checkbox"/> 1	port-1	SIP Server	Port-1	Allow	8884600251...	--	--	--	--
<input type="checkbox"/> 11	port-11	SIP Server	Port-11	Allow	8884600378...	--	--	--	--
<input type="checkbox"/> All									

► Register Interval and DNS SRV

Register Interval(range: 1 - 3600s)	<input type="text" value="60"/>	s
DNS query type	<input type="text" value="A query"/>	
DNS refresh interval (range:0 - 60,000min, 0 means disable)	<input type="text" value="0"/>	min

► Register Interval

This field specifies the value that the gateway will send in the Expires header of the REGISTER message. Its value ranges 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, Ranges from 0 to 60000 mins, 0 means disable defaultvalue is disable.

► Configuring SIP Timers

T1	<input type="text" value="500"/>	ms
T2	<input type="text" value="4000"/>	ms
T4	<input type="text" value="5000"/>	ms
TMAX	<input type="text" value="32000"/>	ms
Keepalive Interval(range:10 - 3600s)	<input type="text" value="32"/>	s
Keepalive SIP ID	<input type="text"/>	
Keepalive Retry Count(range:1 - 10)	<input type="text" value="3"/>	
Enable 100rel	<input checked="" type="radio"/> no <input type="radio"/> 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 10-3600s.

▶ **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

How many counts will retry if no response. Its value ranges 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
Delay Answer(range:0 - 10s)	0 s
183 Mode	Immediately
Called Number Parse	Request-Line
Caller Number Source	User
Request Line	Default

► 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.

► Delay Answer (range:0 - 10s)

Outgoing call from ip to gsm,when gsm side answered,device will delay the time response 200 OK to ip side.

► 183 Mode

Immediately: Gateway will send "183 RING" immediately to SIP Server while it receives "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 definite in ringing status.

► Called Number Parse

Where get the called number, from Request-Line or To header.

```

Frame 14: 1342 bytes on wire (10736 bits), 1342 bytes captured (10736 bits)
Linux cooked capture
Internet Protocol, Src: 192.168.11.70 (192.168.11.70), Dst: 192.168.11.1 (192.168.11.1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:69818187448@192.168.11.1:5060 SIP/2.0
Method: INVITE
Request-URI: sip:69818187448@192.168.11.1:5060
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 192.168.1.146:5060;branch=z9hg4bk-d87543-480fac190b199e06-1--d87543-;rport
Max-Forwards: 70
Contact: <sip:955@192.168.1.146:5060>
To: "69818187448"<sip:69818187448@192.168.11.1:5060>
From: "955"<sip:955@192.168.11.1:5060>;tag=2f57a741
Call-ID: Y2F1MTMONTIOMDAILY2F1Mzk1ZTZyMDAZZW0ZDVkMjM.
CSeq: 1 INVITE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
Content-Type: application/sdp
User-Agent: X-Lite release 1011s stamp 41150
Content-Length: 737
Message Body

```

► Caller Number Source

Where get the called number, from User name or Display name.

```

Frame 14: 1342 bytes on wire (10736 bits), 1342 bytes captured (10736 bits)
Linux cooked capture
Internet Protocol, Src: 192.168.11.70 (192.168.11.70), Dst: 192.168.11.1 (192.168.11.1)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
  Request-Line: INVITE sip:69818187448@192.168.11.1:5060 SIP/2.0
  Method: INVITE
    Request-URI: sip:69818187448@192.168.11.1:5060
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/UDP 192.168.1.146:5060;branch=z9hG4bK-d87543-480fac190b199e06-1--d87543-;rport
    Max-Forwards: 70
    Contact: <sip:955@192.168.1.146:5060>
    To: "69818187448" <sip:69818187448@192.168.11.1:5060>
    From: "955" <sip:955@192.168.11.1:5060>;tag=2f57a741
    Call-ID: Y2F1MTMONTIOMDA1Y2F1Mzk1ZTQyMDAzZW0ZDZVkmjM.
    CSeq: 1 INVITE
    Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
    Content-Type: application/sdp
    User-Agent: X-Lite release 1011s stamp 41150
    Content-Length: 737
  Message Body

```

► Request Line

The request uri forced to use Remote Contact during the session. normal calling doesn't need set this.

► 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	<input type="radio"/> No <input checked="" type="radio"/> Yes
Session Timer Interval(range:90 - 60000s)	<input type="text" value="1800"/> s
Session timer mode	<input type="text" value="refresh"/> ▼
Session timer refresher	<input type="text" value="uac"/> ▼

► 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 ranges 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.

Gsm-Sip Code Map	
Gsm Code Enable	<input type="radio"/> No <input checked="" type="radio"/> Yes
Gsm Reason	Sip Response Code
No Port Found	503
Unassigned Number	404
Normal Call Clearing	480
User Busy	486
User Not Answer	408
Call Rejected	403
Mobile Network Fault	503

► 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 needs some specific SIP response from the gateway. Example, SIP server needs SIP response *180 Ringing* instead of *183 Ringing*, the configuration should be as below:

Response Code switch	Response code	Response code after switch
	183	180

► Custom Extensions Header

Custom Extensions Header	
Enable IMEI Header	<input checked="" type="radio"/> no <input type="radio"/> yes
Enable IMSI Header	<input checked="" type="radio"/> no <input type="radio"/> yes
Enable Portno Header	<input checked="" type="radio"/> no <input type="radio"/> yes
SIP Encryption	<input checked="" type="radio"/> No <input type="radio"/> Yes
RTP Encryption	<input checked="" type="radio"/> No <input type="radio"/> Yes

► Customer Extensions Header

Send the IMEI/IMSI/Portno info to other in SIP Header.

```

+ Frame 5152: 861 bytes on wire (6888 bits), 861 bytes captured (6888 bits)
+ Ethernet II, Src: f8:a0:3d:48:99:4b (f8:a0:3d:48:99:4b), Dst: 08:23:45:67:21:ca (08:23:45:67:21:ca)
+ Internet Protocol, Src: 172.19.211.120 (172.19.211.120), Dst: 172.19.0.87 (172.19.0.87)
+ User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
- Session Initiation Protocol
  + Request-Line: INVITE sip:4100@172.19.0.87:5060 SIP/2.0
  - Message Header
    + Via: SIP/2.0/UDP 172.19.211.120;branch=z9hg4bkfbdbbc10aa7e359fed54dfe0ae92f3c4;rport
    + From: 13670200513 <sip:9009@172.19.0.87:5060>;tag=68f3fa40df2a91e966b62de89420113a
    + To: <sip:4100@172.19.0.87:5060>
      Call-ID: b60eaac18b58c397be8b78c7ae20ae70@172.19.211.120
    + CSeq: 548901696 INVITE
    + Contact: <sip:9009@172.19.211.120>
      Supported: replaces
    + X-IMEI: 352260056404753
    + X-IMSI: 460025169926509
    + X-PORT-NO: 0
    User-Agent: UC2000-VF-02231301
    Allow: INVITE, ACK, CANCEL, BYE, OPTIONS, INFO, UPDATE, NOTIFY, REFER
    Content-Type: application/sdp
    Max-Forwards: 70
    Content-Length: 193
  - Message Body
    + Session Description Protocol

```

► SIP/RTP Encryption

When you use VOS as sip server, and you want SIP and RTP encryption, please enable this option.

4.9.2 SIP Trunk Configuration

IP Trunk					
	Index	IP	Port	Description	KeepAlive Enable
<input type="checkbox"/>	31	172.16.221.221	5060	Elastix	No

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"

IP Trunk Add

Index	<input style="width: 90%;" type="text" value="31"/>
IP	<input style="width: 90%;" type="text" value="172.16.221.221"/>
Port	<input style="width: 90%;" type="text" value="5060"/>
Description	<input style="width: 90%;" type="text" value="Elastix"/>
KeepAlive Enable	<input type="checkbox"/>

4.9.3 SIP Trunk Group

Figure 4-11-3 IP Trunk Group

IP Trunk Group			
	Index	Description	IP
<input type="checkbox"/>	31	default	31,

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 be 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

IP Trunk Group Add

Index	<input style="width: 90%;" type="text" value="31"/>		
Description	<input style="width: 90%;" type="text" value="Default"/>		
IP	Index	IP	Port
<input checked="" type="checkbox"/>	31	172.16.221.221	5060

4.9.4 Port Configuration

Port List									
Port	SIP User ID	Authenticate ID	Authenticate Password	Local SIP Port	Register to	Tx Gain	Rx Gain	To VOIP Hotline	To PSTN Hotline
<input type="checkbox"/> 0	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 1	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 2	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 3	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 4	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 5	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 6	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 7	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 8	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 9	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 10	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 11	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 12	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 13	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 14	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> 15	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> All	<input type="text" value="1"/> Increment	<input type="text" value="1"/> Increment	<input type="text" value="0"/> Increment	<input type="text" value="1"/> Increment	Copy	Copy	Copy	Copy	Copy
	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	No Register	+0dB	+6dB	<input type="text"/>	<input type="text"/>

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
Authenticate Password	Password of SIP User ID which provide by SIP server
Local SIP Port	The channel sip port
Register to	Register with which sip server
Tx Gain	Tx Gain value of chipset. Adjusting it will affect volume on GSM side.
Rx Gain	Rx Gain value of chipset. Adjusting it will affect 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

Port Group								
Index	Description	SIP User ID	Authenticate ID	Port	Register	Select Mode	To VOIP Hotline	To PSTN Hotline
0	all			0,1,2,3,4,5,6,...	No Regi...	Cyclic Ascending		

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

NOTE: 1. 0 port group is not allowed to delete, only allowed to change.

Select ports for defined port group.

Port Group Modify

Index:

Description:

SIP User ID:

Authenticate ID:

Authenticate Password:

To VOIP Hotline:

To PSTN Hotline:

Register to:

Select Mode:

Port:

<input checked="" type="checkbox"/> Port 0	<input checked="" type="checkbox"/> Port 1
<input checked="" type="checkbox"/> Port 2	<input checked="" type="checkbox"/> Port 3
<input checked="" type="checkbox"/> Port 4	<input checked="" type="checkbox"/> Port 5
<input checked="" type="checkbox"/> Port 6	<input checked="" type="checkbox"/> Port 7
<input checked="" type="checkbox"/> Port 8	<input checked="" type="checkbox"/> Port 9
<input checked="" type="checkbox"/> Port 10	<input checked="" type="checkbox"/> Port 11
<input checked="" type="checkbox"/> Port 12	<input checked="" type="checkbox"/> Port 13
<input checked="" type="checkbox"/> Port 14	<input checked="" type="checkbox"/> Port 15

4.9.6 Digitmap

Digit Map

Digit Map:

NOTE: Length of 'Digit Map' should be not more than 119 characters.

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)xxxxxxxx

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

IP->Tel Routing Add

Index:

Description:

Source:

Destination:

Call Restriction:

Advanced Rules:

Click to set caller and called prefix

IP->Tel Routing Add	
Index	62
Description	
Source	SIP Server
Destination	port-group-0 <all>
Call Restriction	Allow Call
Advanced Rules	▲
Source Prefix	
Destination Prefix	
Prefix to Add	
Digits to be Deleted	
Number of Digits Reserved	

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:

IP->Tel Routing Add	
Index	62
Description	to CMB
Source	SIP Server
Destination	port-0
Call Restriction	Allow Call
Advanced Rules	▲
Source Prefix	201
Destination Prefix	any
Prefix to Add	
Digits to be Deleted	
Number of Digits Reserved	

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

IP->Tel Routing Add

Index: 61
 Description: rmv2
 Source: SIP Server
 Destination: port-group-0 <all>
 Call Restriction: Allow Call

Advanced Rules
 Source Prefix: any
 Destination Prefix: 991
 Prefix to Add: 3
 Digits to be Deleted:
 Number of Digits Reserved:

Remove prefix 991 of called number.

IP->Tel Routing Add

Index: 62
 Description: 88
 Source: SIP Server
 Destination: port-group-0 <all>
 Call Restriction: Allow Call

Advanced Rules
 Source Prefix: any
 Destination Prefix: 88
 Prefix to Add: 0
 Digits to be Deleted: 2

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

Tel->IP Routing Add	
Index	62
Description	
Source	port-group-0 <all>
Destination	SIP Server
Call Restriction	Allow Call
Advanced Rules	

Click  to set caller and called prefix

Tel->IP Routing Add	
Index	62
Description	
Source	port-group-0 <all>
Destination	SIP Server
Call Restriction	Allow Call
Advanced Rules	
Source Prefix	
Destination Prefix	
Prefix to Add	
Digits to be Deleted	
Number of Digits Reserved	

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 in

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 Gsm Calling Config

Modify the caller number of the incoming call.

Gsm Calling Rule		
Calling Prefix	Digit Delete	Prefix Add
<input type="checkbox"/> 136	<input type="text" value="3"/>	<input type="text" value="888"/>
<input type="checkbox"/>	<input type="text" value="0"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="0"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="0"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="0"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="0"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="0"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="0"/>	<input type="text"/>

NOTE: 1.Support regular expression, refer to Digit Map.
 NOTE: 2.The Prefix Add Length add the Digit Delete of Number can't more than 31 .

CDR															
Enable CDR <input type="radio"/> No <input checked="" type="radio"/> Yes			Save CDR <input checked="" type="radio"/> No <input type="radio"/> Yes			Check Bch <input checked="" type="radio"/> No <input type="radio"/> Yes									
Start Date: 2018 Year 8 Month 24 Day			Select Port: A11			Call Direction: ALL									
End Date: 2018 Year 8 Month 24 Day			Source:			Destination:									
Min Duration: s			Max Duration: s			Rtp Loss Rate: % to %									
<input type="button" value="Export"/>			<input type="button" value="Refresh"/>			<input type="button" value="Delete the CDRs in this Report"/>									
Port	Start Date	Answer Date	Call Direction	Source	SourceIP	Destination	Hang Side	Reason	Duration(s)	Codec	Rtp Send	Rtp recv	Rtp loss Rate	jitter(s)	BCCH
0	2018/08/24 08:37:07		Gsm->IP	888702005		4100	Called	REJECTED	0	G.711U	0	0	0%	0	
0	2018/08/24 07:07:41	2018/08/24 07:07:45	Gsm->IP	136702005		4100	Calling	NORMAL HANG UP(31)	7	G.711U	36	301	0%	0	
0	2018/08/24 07:03:49	2018/08/24 07:03:59	Gsm->IP	136702005		4100	Called	NORMAL HANG UP	7	G.711U	83	201	0%	0	
0	2018/08/24 07:03:00		Gsm->IP	136702005		4100	Called	REJECTED	0	G.711U	0	0	0%	0	

Total: 4 entries 50 entries/page 1/1 page

4.9.10 Service parameter

► To configure dialing mode parameters

Do Not Answer GSM Incoming Call for Hotline	<input type="radio"/> No <input checked="" type="radio"/> Yes
Enable GSM Incoming Configuration	<input type="radio"/> No <input checked="" type="radio"/> Yes
Answer Delay	<input type="text" value="5"/> Sec(s)
Ringback Tone	<input type="radio"/> None <input checked="" type="radio"/> GSM Ringback <input type="radio"/> Fake Ringback
RTP Detected Enable	<input type="radio"/> No <input checked="" type="radio"/> Yes
Period without RTP Packet	<input type="text" value="90"/>
Auto CLIP Routing	<input checked="" type="radio"/> No <input type="radio"/> Yes

► 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 GSM 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

▶ **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.

▶ **Ringback Tone**

Default device forward the Ringback to IP side from GSM operator. But sometimes GSM operator Ringback not clear or other issue,client want device or softswitch play ringback,you can set Fake Ringback or None.

▶ **RTP Detect**

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

▶ **Auto CLIP Routing**

Same callee route to same port, Force means if the port is busy, the call can't call through the device even there is idle port.

Auto CLIP Routing
CLIP Routing is force

No Yes
 No Yes

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

Other Configuration	
Enable Private Service	<input type="radio"/> No <input checked="" type="radio"/> Yes
User ID Is Phone Number	<input checked="" type="radio"/> No <input type="radio"/> Yes
Reject Anonymous Call from IP to GSM	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use # as End Key	<input type="radio"/> No <input checked="" type="radio"/> Yes
No Answer Timeout	55 Sec(s)
Interdigit Timeout	4 Sec(s)
Reset ASR after SIM Switching	<input checked="" type="radio"/> no <input type="radio"/> yes

► 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.

► 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

▶ **No Answer Timeout**

How long time hang up the call if callee no answered.

▶ **Interdigit Timeout**

Timeout without dialing

▶ **Reset ASR after SIM Switching**

Reset ASR or not after SIM Switch

4.9.11 Media parameter

Local Start RTP Port	8000
Enable Silence Suppression	<input type="radio"/> No <input checked="" type="radio"/> Yes
Enable Busy Tone Detect	<input checked="" type="radio"/> No <input type="radio"/> Yes
Call Progress Tone	USA
Ring Back Tone	440,280,480,280,2000,4
Busy Tone	480,330,620,330,500,50
Dial Tone	350,260,440,260,0,0,0,1

▶ **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;

▶ **Enable Busy Tone Detect**

As usual, we detect Reverse Polarity then hang up the call, if gsm don't sendReverse Polarity, you can enable Busy Tone Detect.

▶ **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	<input type="radio"/> No <input checked="" type="radio"/> Yes
IVR Play Duration	25 Sec(s)
Play IVR Voice Prompt from	<input checked="" type="radio"/> Default <input type="radio"/> 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

4.9.12 DBO Parameter

Enable DBO service

DBO Parameter	
Enable DBO	<input type="checkbox"/>

NOTE: 1.If you enable the SIP Forwarding, please:
 (1)Choose the SIP server to modify the SIP configuration register mode;
 (2)Do not enable independent local sip ports mode!
 2.Port is configured as the encryption mode is less than 30000, 30000 or greater non-encrypted mode. Set all ports allows only one mode.

Save

Configure DBO parameter

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

DBO Parameter	
Enable DBO	<input checked="" type="checkbox"/>
DBO Local Port(0 means Random Port)	<input type="text"/>
Active DBO Server URL/IP	54.251.248.30
Active DBO Server Port 0	3479
Active DBO Server Port 1	6479
Active DBO Server Port 2	12479
Active DBO Server Port 3	24479
Active DBO Server Username	54.251.248.30_3479
Active DBO Server Password	••••••
Standby DBO Server URL/IP	<input type="text"/>
Standby DBO Server Port 0	<input type="text"/>
Standby DBO Server Port 1	<input type="text"/>
Standby DBO Server Port 2	<input type="text"/>
Standby DBO Server Port 3	<input type="text"/>
Standby DBO Server Username	<input type="text"/>
Standby DBO Server Password	<input type="text"/>
Enable SIP Forwarding	<input checked="" type="checkbox"/>
Enable RTP Forwarding	<input checked="" type="checkbox"/>
Enable Bandwidth Compressed	<input type="checkbox"/>

Parameter Description:

Parameters	Description
------------	-------------

DBO Local Port (0 means Random Port)	Which port use to connect dbo in device
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, monthly, daily credits and remaining daily, hourly callcounts of a SIM card.

Overview											
	SIM	Port Status	Phone Number	Last Matched Balance	Calculated Balance	Credits			Remain Calls		Daily Connected Counts
						Total	Monthly	Daily	Daily	Hourly	
<input type="checkbox"/>	0		13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	1	Mobile Registered		---	---	---	---	---	---	---	---
<input type="checkbox"/>	2	Mobile Registered		---	---	---	---	---	---	---	---
<input type="checkbox"/>	3		13611100492								
<input type="checkbox"/>	4	Mobile Registered	18312524253	---	---	---	---	---	---	---	---
<input type="checkbox"/>	5	Mobile Registered	13430547595	---	---	---	---	---	---	---	---
<input type="checkbox"/>	6	searching network	13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	7		13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	8		13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	9		13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	10		13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	11		13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	12		13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	13		13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	14		13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	15		13611100492	---	---	---	---	---	---	---	---
<input type="checkbox"/>	All			<input type="button" value="Clear"/>	<input type="button" value="Clear"/>	<input type="button" value="Reset"/>					
						<input type="button" value="Set"/>	<input type="button" value="Set"/>	<input type="button" value="Set"/>			

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) .

Basic Configuration	
Tel to IP Call Delay(range:0-60s)	0 s- 0 s Note:If both are set as "0", it means the function is not enabled.
Startup Interval(range:0-3600s)	0 s- 0 s Note:If both are set as "0", it means the function is not enabled.
IP to Tel Call Delay(range:0-10s)	0 s
Call Interval(range:0-3600s)	10 s- 15 s
No Alerting Call Handle	<input checked="" type="radio"/> Normal Handle <input type="radio"/> Hang Up <input type="radio"/> Not Answer
IP to TEL Processing Timeout Handle	<input type="checkbox"/>
Set Call Volume Threshold	<input type="checkbox"/>
SMS Sending Delay (range:0-300s)	0 s- 0 s Note:If both are set as "0", it means the function is not enabled.
Numeric Scale	2
GSM incoming call limit (range:0-3600s)	0 s- 0 s Note:If both are set as "0", it means the function is not enabled.
Setting of Multi-SIM	
SIM Switching Setting	
Switch SIM after running	0 Minutes(0-65535)
Switch SIM after calling	0 (0-65535)
Switch SIM after calling	0 Minutes(0-65535)
Switch SIM after sending	0 SMS(0-65535)
Query SIM information during initiation	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Timeout	30 Seconds (range:30-300)

▶ Tel to IP Call Delay

Incoming call reach device, device will delay the secs to send to IP side.

▶ Startup Interval

Module power on time interval. when device power on, all module won't power on at same time, they will power on one by one.

▶ IP to Tel Call Delay

Outgoing call reach device, device will delay the secs to send to GSM side.

▶ Call Interval

When one call end, the port will rest the time, if you set 5-120secs, it means the port will rest min 5secs, max 120secs.

▶ No Alerting Call Handle

Outgoing call don't have alerting before receive Reverse Polarity, we can choose Normal Handle, Hang Up or Not Answer.

Normal Handle: Call will normal active.

Hang Up: Call will hang up by device.

Not Answer: Call won't connect, call will timeout or caller cancel it.

▶ **IP to TEL Processing Timeout Handle**

Enable Processing Timeout Handle, you can set timeout time.

▶ **Set Call Volume Threshold**

Enable the Call Volume Threshold, if the Volume is lower or higher than the threshold you set, call will be hanged up by device.

▶ **SMS Sending Delay**

SMS send interval, when one SMS send out, next one will delay send out.

▶ **Numeric Scale**

How many digits displayed after the decimal point in balance.

▶ **GSM incoming call limit**

Limit the incoming call duration.

▶ **Setting of Multi-SIM SIM Switching Setting**

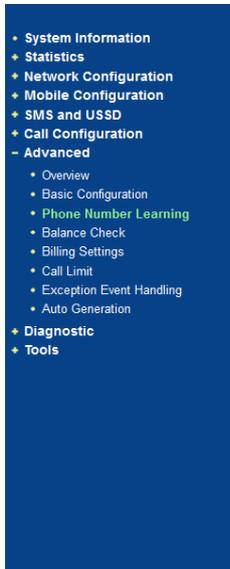
This setting for Multi-SIM device, like 8/32,16/64,32/128, four slots for one module, you can set switchover card by SIM running time, call counts, calltime, and sms counts.

Enable Query SIM information during initiation, when device power on, all slots cards will register one by one to get their info, like sim number, sim balance, if you have set auto balance check and number study.

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, but local number is 18344144906, you need delete the 86



Phone Number Learning - Modify Rule

Index: 0
 Type: USSD
 Send Text: *156#
 Keywords: Your number is [Matching Test]
 Write Phone Number to SIM card: No Yes
 Stripped Digits from Left: 2
 Prefix to Add:
 Port Group: 0 <all>

[Save] [Reset] [Cancel]

About Key Words: 1. You can input multiple keywords and special symbol like "[E], [T], [, [N]". Space is also available.
 2. [E] is used to match Enter. e.g. "number is[E]" is used to match the number under line "number is".
 3. [T] is used to match Table character.
 4. [,] is used to match anything between the keywords. e.g. "number is[]number is " is used to match the number after the second keyword "number is ".
 5. [N] is used to match the number at the specify position. e.g. "number is []number is [N]. abc" is used to match "456" in SMS "number is 123 and number is 456, abcdefg...".
 6. It can match the number after a keyword or in front of a keyword. It is depending on the position of the [N] symbol. If the [N] symbol is follow close behind a keyword, it match a number after the keyword. If a keyword is follow close behind the [N] symbol, the [N] symbol match a number in front of the keyword. Normally, we use [] [N] to match a number in front of the keyword. e.g. "is number 1[] [N] is number 2" is used to match "456" in SMS "123 is number 1 and 456 is number 2".

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.

Phone Number Learning - Modify Rule

Index: 0
 Type: USSD
 Send Text: *156#
 Keywords: Your number is [Test End]
 Test SMS Text: Your number is 8618344144906
 Result: [Test] 8618344144906
 Write Phone Number to SIM card: No Yes
 Stripped Digits from Left: 2
 Prefix to Add:
 Port Group: 0 <all>

[Save] [Reset] [Cancel]

The screenshot shows the configuration interface for SIM cards. The left sidebar has a menu with the following items: System Information, Statistics, Network Configuration, Mobile Configuration, SMS and USSD, Call Configuration, and Advanced. Under 'Advanced', 'Overview' is selected and highlighted with a red box. The main area displays a table titled 'Overview' with the following data:

	SIM	Port Status	Phone Number	Last Matched
<input type="checkbox"/>	0			---
<input type="checkbox"/>	1	Mobile Registered	18344144906	---
<input type="checkbox"/>	2		13611100492	---
<input type="checkbox"/>	3			---
<input type="checkbox"/>	4	Mobile Unregistered	13611100492	---
<input type="checkbox"/>	5	Mobile Registered	13430547595	---
<input type="checkbox"/>	6			---
<input type="checkbox"/>	7		13611100492	---
<input type="checkbox"/>	8	searching network		---
<input type="checkbox"/>	9		13611100492	---
<input type="checkbox"/>	10			---
<input type="checkbox"/>	11		13611100492	---
<input type="checkbox"/>	12		13611100492	---

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**, you can delete or add prefix.

The screenshot shows the 'Phone Number Learning - Modify Rule' configuration page. The form contains the following fields and values:

- Index: 0
- Type: SMS
- Encoding: UCS2
- Dest Number: 10086
- Send Text: My Number
- Check SMS From Number: 10086
- Keywords: Your number is
- Write Phone Number to SIM card: No Yes
- Stripped Digits from Left: 2
- Prefix to Add: (empty)
- Port Group: 0 <all>

A 'Matching Test' button is located to the right of the Keywords field.

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.

Phone Number Learning - Modify Rule	
Index	<input type="text" value="0"/>
Type	<input type="text" value="SMS"/>
Encoding	<input type="text" value="UCS2"/>
Dest Number	<input type="text" value="10086"/>
Send Text	<input type="text" value="My Number"/>
Check SMS From Number	<input type="text" value="10086"/>
Keywords	<input type="text" value="Your number is"/> <input type="button" value="Test End"/>
Test SMS Text	<input type="text" value="Your number is 8618344144906"/>
Result	<input type="button" value="Test"/> 8618344144906
Write Phone Number to SIM card	<input type="radio"/> No <input checked="" type="radio"/> Yes
Stripped Digits from Left	<input type="text" value="2"/>
Prefix to Add	<input type="text"/>
Port Group	<input type="text" value="0 <all>"/>

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**.

Phone Number Learning - Modify Rule	
Index	<input type="text" value="0"/>
Type	<input type="text" value="Call"/>
Dest Number	<input type="text" value="10086"/>
Send Text	<input type="text" value="p5,3,p3,2"/>
Check SMS From Number	<input type="text"/>
Keywords	<input type="text" value="Your number is"/> <input type="button" value="Matching Test"/>
Write Phone Number to SIM card	<input type="radio"/> No <input checked="" type="radio"/> Yes
Stripped Digits from Left	<input type="text" value="2"/>
Prefix to Add	<input type="text"/>
Port Group	<input type="text" value="0 <all>"/>

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.

Phone Number Learning - Modify Rule

Index: 0

Type: Call

Dest Number: 10086

Send Text: p5,3,p3,2

Check SMS From Number:

Keywords: Your number is Test End

Test SMS Text: Your number is 8618344144906

Result: Test 8618344144906

Write Phone Number to SIM card: No Yes

Stripped Digits from Left: 2

Prefix to Add:

Port Group: 0 <all>

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.

	SIM	Port Status	Phone Number	Last Matched Balance	Calculated Balance	Credits		
						Total	Monthly	Daily
<input type="checkbox"/>	0			---	---	---	---	---
<input type="checkbox"/>	1	Mobile Registered	13430547595	73.40	73.40	---	---	---
<input type="checkbox"/>	2			---	---	---	---	---
<input type="checkbox"/>	3			---	---	---	---	---
<input type="checkbox"/>	4			---	---	---	---	---
<input type="checkbox"/>	5			---	---	---	---	---
<input type="checkbox"/>	6			---	---	---	---	---
<input type="checkbox"/>	7			---	---	---	---	---
<input type="checkbox"/>	8			---	---	---	---	---

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.

Balance check condition can be time, balance threshold and call counts.



Balance Check - Modify Rule

Mode: One step

Index: 0

Type: USSD

Send Text: *101#

Check SMS From Number: []

Balance Prefix Keys-1: Your balance is

Balance Prefix Keys-2: []

Matching Test: Test Start

Check Balance After SIM Card Registration:

Check Balance Every: 30 Minutes
Note: "0" means disable.

Check While Calculated Balance Is Low: 10.00
Note: "0" means disable.

Check Balance by Call Count(range:0-100): 5
Note: "0" means disable.

Digit Thousand Symbol: ,

Digit Point Symbol: .

Port Group: 0 <all>

Save Reset Cancel

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.

Balance Check - Modify Rule

Mode: One step

Index: 0

Type: USSD

Send Text: *101#

Check SMS From Number: []

Balance Prefix Keys-1: Your balance is

Balance Prefix Keys-2: []

Matching Test: Test Start

Check Balance After SIM Card Registration:

Check Balance Every: 30 Minutes
Note: "0" means disable.

Check While Calculated Balance Is Low: 10.00
Note: "0" means disable.

Check Balance by Call Count(range:0-100): 5
Note: "0" means disable.

Digit Thousand Symbol: ,

Digit Point Symbol: .

Port Group: 0 <all>

(next page)

Balance Check - Modify Rule	
Mode	One step
Index	0
Type	USSD
Send Text	*101#
Check SMS From Number	
Balance Prefix Keys-1	Your balance is
Balance Prefix Keys-2	
Matching Test	Test End
Test SMS Text	Your balance is 73.40\$
Result	Test 73.40
Check Balance After SIM Card Registration	<input checked="" type="checkbox"/>
Check Balance Every	30 Minutes Note: "0" means disable.
Check While Calculated Balance Is Low	10.00 Note: "0" means disable.
Check Balance by Call Count(range:0-100)	5 Note: "0" means disable.
Digit Thousand Symbol	,
Digit Point Symbol	.
Port Group	0 <all>

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.

Balance Check - Modify Rule	
Mode	One step
Index	0
Type	SMS
Encoding	UCS2
Dest Number	10086
Send Text	My balance
Check SMS From Number	10086
Balance Prefix Keys-1	Your balance is
Balance Prefix Keys-2	
Matching Test	Test Start
Check Balance After SIM Card Registration	<input checked="" type="checkbox"/>
Check Balance Every	30 Minutes Note: "0" means disable.
Check While Calculated Balance Is Low	10.00 Note: "0" means disable.
Check Balance by Call Count(range:0-100)	5 Note: "0" means disable.
Digit Thousand Symbol	,
Digit Point Symbol	.
Port Group	0 <all>

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.

Balance Check - Modify Rule	
Mode	One step
Index	0
Type	SMS
Encoding	UCS2
Dest Number	10086
Send Text	My balance
Check SMS From Number	10086
Balance Prefix Keys-1	Your balance is
Balance Prefix Keys-2	
Matching Test	Test End
Test SMS Text	Your balance is 73.40\$
Result	Test 73.40

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.

Balance Check - Modify Rule	
Mode	One step
Index	0
Type	Call
Dest Number	10086
Send Text	p5,3,p3,2
Check SMS From Number	10086
Balance Prefix Keys-1	Your balance is
Balance Prefix Keys-2	
Matching Test	Test Start
Check Balance After SIM Card Registration	<input checked="" type="checkbox"/>
Check Balance Every	30 Minutes Note: "0" means disable.
Check While Calculated Balance Is Low	10.00 Note: "0" means disable.
Check Balance by Call Count(range:0-100)	5 Note: "0" means disable.
Digit Thousand Symbol	,
Digit Point Symbol	.
Port Group	0 <all>

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.

Balance Check - Modify Rule	
Mode	One step
Index	0
Type	Call
Dest Number	10086
Send Text	p5,3,p3,2
Check SMS From Number	10086
Balance Prefix Keys-1	Your balance is
Balance Prefix Keys-2	
Matching Test	Test End
Test SMS Text	Your balance is 73.40\$
Result	Test 73.40

4.10.5 Billing setting

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

Billing Settings - Add Rule

Index	<input type="text" value="0"/>	
Billing Unit(1-3600)	<input type="text" value="60"/>	seconds
Rate(not more than 10000)	<input type="text" value="1"/>	/ Billing Unit
Minimum Charging Time(0-3600)	<input type="text" value="0"/>	s
Select Port	<input type="text" value="Port Group"/>	
	<input type="text" value="0 <all>"/>	

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

Call Limit - Add Rule

Index	<input type="text" value="1"/>	
Single Call Duration	<input type="text" value="0"/>	s
	<small>Note:0 means no limit,not more than 40000.</small>	
Total Credits	<input type="text" value="300"/>	
	<small>Note:0 means no limit,not more than 400000.</small>	
Monthly Credits	<input type="text" value="0"/>	
	<small>Note:0 means no limit,not more than 400000.</small>	
Daily Credits	<input type="text" value="0"/>	
	<small>Note:0 means no limit,not more than 400000.</small>	
Daily Calls	<input type="text" value="0"/>	
	<small>Note:0 means no limit,not more than 100000.</small>	
Hourly Calls	<input type="text" value="0"/>	
	<small>Note:0 means no limit,not more than 1000.</small>	
Daily Connected Counts	<input type="text" value="0"/>	
	<small>Note:0 means no limit,not more than 100000.</small>	
Adjust Credits Automatically	<input checked="" type="radio"/> No <input type="radio"/> Yes	
Low Credits Warning	<input checked="" type="radio"/> No <input type="radio"/> Yes	
Reset Monthly Date	<input type="text" value="1"/>	
Select Port	<input type="text" value="Port Group"/>	
	<input type="text" value="0 <all>"/>	

Single Call Duration: set single call duration, it defines the maximum duration every single call can take, 0 means no limit. If you set 40, it means every call can last 40secs 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.

Monthly Credits: set monthly credits, it defines the maximum credit the port can use in one day, 0 means no limit. If you set 300, it means the port can use 300 at most one month, and the data will be cleared at Reset Monthly Date.

Daily Credits: set daily credits, it defines the maximum credit the port can use in one day, 0 means no limit. If you set 30, it means the port can use 30 at most one day, and the data will be cleared at 0'clock of everyday.

Daily Calls: set daily calls, it defines the maximum counts the port can use in one day, 0 means no limit. If you set 30, it means the port can call 30 counts at most one day, and the data will be cleared at 0'clock of everyday

Hourly Calls: set hourly calls, it defines the maximum counts the port can use in one hour, 0 means no limit. If you set 10, it means the port can call 10 counts at most one hour, and the data will be cleared next hour.

Daily Connected Counts: set daily Daily Connected Counts, it defines the maximum counts the port can use in one day, 0 means no limit. If you set 20, it means the port can call 20 connect calls 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.

Low Credits Warning: When the total credit reaches the setting, it will send sms to thecell phone number you set.

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

Call limitation = $300/1 = 300$ minutes

4.10.7 Exception Event Handling

Exception Event Handling			
Enable	<input type="radio"/> No <input checked="" type="radio"/> Yes		
Call Event		Handle	Alerting
<input type="checkbox"/> Low ASR Less Than	50 %	Reset	<input type="checkbox"/>
<input checked="" type="checkbox"/> Low ACD Less Than	5 s	Block	<input type="checkbox"/>
Counts of Recent Call	3		
<input type="checkbox"/> Counts of Call Failed	3	Reset	<input type="checkbox"/>
<input type="checkbox"/> Low Balance Less Than	0.00	Block SIM	<input type="checkbox"/>
<input type="checkbox"/> By Gsm Code	8	Reset	<input type="checkbox"/>
<input type="checkbox"/> PDD Less Than(1-30)	1 s	Reset	<input type="checkbox"/>
<input type="checkbox"/> Cost Difference	0.00 - 0.00	Block	<input type="checkbox"/>
USSD Event			
<input type="checkbox"/> Counts of Send Fail(1 - 100)	1	Reset	<input type="checkbox"/>
USSD/SMS Monitor			
<input checked="" type="checkbox"/> SMS Number	121454	Reset	<input checked="" type="checkbox"/>
Keywords	快跑		
Keywords1			
USSD/SMS URL Monitor			
<input type="checkbox"/> SMS Number	13767243151	Access Internet	<input type="checkbox"/>
Keywords	上网		
Keywords1			
SIM Register Fail			
<input type="checkbox"/> Register Timeout(60 - 600)	60 s	Reset	<input type="checkbox"/>
Abnormal BCCH			
<input type="checkbox"/> Check BCCH Every	10 s	Power Down	<input type="checkbox"/>
SMS Test			
Number	1123146		
Content	SMS Test.		
Resend Count(0 - 10)	0		
Alerting Setting	<input checked="" type="radio"/> SMS Alerting <input type="radio"/> Email Alerting		
Number for SMS Alerting	1215664		
Number1 for SMS Alerting			

Save

NOTE: 1.To enable the GSM code monitoring, you need to enable Gsm-Sip Code Map.
 2.To enable to apply other exceptional call event, you need to enable CDR.
 3.The GSM Code up to set 4, and Need to be separated by commas
 4.SMS Test: try sending SMS to the set number, until the success or more than resend count.
 If the test send fail, the module is blocked.
 5.Low balance value range form 0 to 400000.
 6.Email Alerting: Before using this function, please confirm that the "Email Sender" setting in "SMS and USSD - Email Setting" has been set correctly.

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:

Enable Call Event	Value	Unit	Handle	SMS Alert
<input type="checkbox"/> Low ASR Less Than	12	%	Block SIM	<input type="checkbox"/>
<input type="checkbox"/> Low ACD Less Than	0	s	Reset	<input type="checkbox"/>
<input type="checkbox"/> Counts of Call Failed	0		Block	<input type="checkbox"/>
<input type="checkbox"/> Low Balance Less Than	0.00		Block SIM	<input checked="" type="checkbox"/>
<input type="checkbox"/> By Gsm Code	8		SMS Test	<input type="checkbox"/>
<input type="checkbox"/> PDD Less Than(1-30)	2	s	Reset	<input type="checkbox"/>
<input type="checkbox"/> Cost Difference	0.00	- 0.00	Reset	<input type="checkbox"/>
			Block	<input type="checkbox"/>

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 as the 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 as low ACD.

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

Counts of failed calls: This value defines 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.

<input type="checkbox"/>	Low ACD Less Than	0	s	Reset	<input type="checkbox"/>
<input checked="" type="checkbox"/>	Counts of Call Failed	10		Reset	<input type="checkbox"/>
<input type="checkbox"/>	Low Balance Less Than	5.00		Block	<input checked="" type="checkbox"/>

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.

<input type="checkbox"/>	Counts of Call Failed	0	Reset
<input checked="" type="checkbox"/>	Low Balance Less Than	5.00	Block
<input type="checkbox"/>	By Gsm Code	8	Reset

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”.

Gsm-Sip Code Map

Gsm Code Enable No Yes

Sip Reason Header Enable No Yes

Gsm Reason

No Port Found

Unassigned Number

Normal Call Clearing

User Busy

User Not Answer

Call Rejected

Mobile Network Fault

Sip Response Code

503

404

480

486

408

403

503

Don't forget to save the configuration.

- b) Configure the GSM code monitor

Low Balance Less Than 5.00 BLOCK

By Gsm Code 8 Block

Consecutive Counts 3

PDD Less Than (1-30)

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 block 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	0	0	

Handle abnormal event

Once one of the above abnormal conditions 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

USSD Event

Counts of Send Fail(1 - 100)

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/USDD response contents, which helps gateway to know SIM card is blocked.

USSD/SMS Monitor

SMS Number

Keywords

Keywords1

USSD/SMS URL Monitor

This parameter be used to Monitor SMS/USDD response contents, which gets the right keywords, access the internet.

USSD/SMS URL Monitor

SMS Number

Keywords

Keywords1

SIM Register Fail

This parameter be used to Monitor the cards register status.

SIM Register Fail

Register Timeout(60 - 600) s

Abnormal BCCH

This parameter be used to Monitor the BCCH, the bcch isn't in the whitelist, the module will power down.

Abnormal BCCH

Check BCCH Every s

SMS Test

When sim blocked by Call Event Monitor, it will send sms to confirm again.

Send sms success, the sim will be unblocked.

SMS Test	
Number	1123146
Content	SMS Test.
Resend Count(0 - 10)	0

Alerting Setting

Set SMS or Email Alerting send.

Email Alerting: Before using this function, please confirm the setting in "SMS and USSD - Email " has been set correctly.

Alerting Setting	
	<input checked="" type="radio"/> SMS Alerting <input type="radio"/> Email Alerting
Number for SMS Alerting	1215664
Number1 for SMS Alerting	

4.10.8 Auto generation

Auto Generation	
Enable	<input checked="" type="checkbox"/>
Basic Settings	
Prefix to Add:	<input type="text"/>
Digits to be Deleted:	<input type="text"/>
Auto Call:	
Called By Other Ports	<input checked="" type="checkbox"/>
Number Length	10 <input type="text"/> digits
Call Out	<input type="checkbox"/>
Min Call Duration:	5 <input type="text"/> seconds
Max Call Duration:	10 <input type="text"/> seconds
Auto Send SMS:	<input type="checkbox"/>
Auto Internet access:	<input type="checkbox"/>
Conditions Settings	
By Device Online Time:	<input type="checkbox"/>
By Total Call Durations:	<input type="checkbox"/>
By Consecutive Calls:	<input type="checkbox"/>

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, the 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:

Prefix to Add and Deleted: when call the other ports, modify the prefix you want.

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 LearnSIM Card Number section.

Number Length: The number valid digits from right.

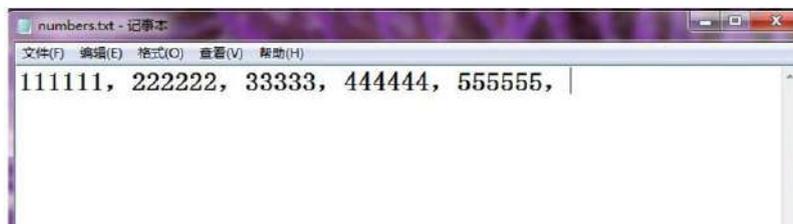
for example, when you call in, device know the number 18612345678 will dial in, but callershow +8618612345678, device will reject the call, if we set the valid number length as 11, it will only check the last 11 digits, the call will allow pass.

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

Auto Internet access: Auto access internet to anti-block.

How to enable Internet access?

Enable Internet Access and set VPN to active this feature, Gateway won't enable this feature successfully if APN is blank or wrong.

YOU can set url as you want, but support https or not decided by the module type, you cancheck access internet success or not under GSM Event.

- System Information
- Statistics
- Network Configuration
- Mobile Configuration
 - Basic Configuration
 - Mobile Configuration
 - Phone Number Config
 - PIN Management
 - IMEI
 - Operator
 - Operator Configuration
 - BCCH
 - Call Forwarding
 - Call Waiting
 - Cloud Server
 - MBN Config
- SMS and USSD
- Call Configuration
- Advanced
- Diagnostic
- Tools

Mobile Configuration									
Port	CLIR	Detect Reverse Polarity	Internet Access	Tx Gain/dB	Rx Gain/dB	APN	APN name	APN PSW	Band Type
0	No	Yes	Yes	3	7	CMNET			Default (Auto)
1	No	Yes	Yes	3	7	CMNET			Default (Auto)
2	No	Yes	Yes	3	7	CMNET			Default (Auto)
3	No	Yes	Yes	3	7	CMNET			Default (Auto)
4	No	Yes	Yes	3	7	CMNET			Default (Auto)
5	No	Yes	Yes	3	7	CMNET			Default (Auto)
6	No	Yes	Yes	3	7	CMNET			Default (Auto)
7	No	Yes	Yes	3	7	CMNET			Default (Auto)

Auto Generation

Enable

Basic Settings

Prefix to Add:

Digits to be Deleted:

Auto Call:

Called By Other Ports

Call Out

Auto Send SMS:

Auto Internet access:

URL-1:

URL-2:

URL-3:

URL-4:

URL-5:

Conditions Settings

By Device Online Time:

Min Interval: minutes

Max Interval: minutes

By Total Call Durations:

By Consecutive Calls:

Consecutive Calls:

GSM Event

Select Port: IMSI: Event:

Send Event via email: Yes No Email Address: Title:

Event Account Setting:

Port	IMSI	Time	Event	Number	Status	Duration(s)	Remark
1	490020106219790	2016-09-07 09:30:11	GSM NET		SUCCESS	0	https://server02.dmicis.com:3000/index

Total: 1 entries 20 entries/page 1/1 page Page 1

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

Conditions Settings

By Device Online Time:

Min Interval: minutes

Max Interval: minutes

By Total Call Durations:

Call Duration: minutes

By Consecutive Calls:

Consecutive Calls:

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

Option	Enable
Local Syslog	<input checked="" type="checkbox"/> Enable
Server Address	172.16.222.222
Server Port	514
Syslog Level	DEBUG
Signal Log	<input checked="" type="checkbox"/> Enable
Media Log	<input type="checkbox"/> Enable
System Log	<input type="checkbox"/> Enable
Management Log	<input type="checkbox"/> Enable
Send CDR	<input checked="" type="checkbox"/> Enable
Server Syslog	<input type="checkbox"/> Enable

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 including 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 including 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 including 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 including 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 registers to SIM Cloud server, the option will be changed toun-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

Remote SIM Card Debug Log	
Record Ports	<input type="text" value="1,2,3"/> Up to 3 ports,e.g."1,2,15"
Record in summary	<input checked="" type="checkbox"/> Enable
Record in media log	<input checked="" type="checkbox"/> Enable

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.

Ping Test	
Ping Destination	<input type="text" value="www.google.com"/>
Number of Ping(1-100)	<input type="text" value="4"/>
Ping Packet Size(56-1024 bytes)	<input type="text" value="56"/>

Information	

4.11.6 Tracert Test

You can check the routes of the tracert destination.

Tracert Test	
Tracert Destination	<input type="text"/>
Max Hops of Tracert(1-255)	<input type="text" value="30"/>

4.11.7 Network Capture

Network capture is a very important diagnostic tool for maintenance. This section 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

Network Capture	
Default Setting	<input type="text" value="PCM only"/>
Select Port	<input type="text" value="Port 0"/>

- ◆ 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:

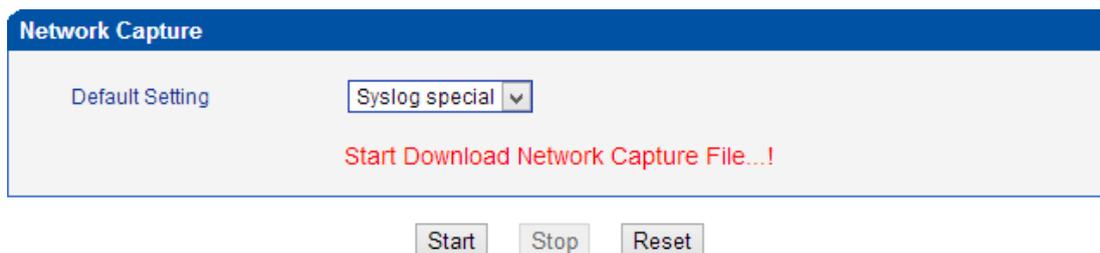
No.	Time	Source	Destination	Protocol	Length	Info	
1	0.000000	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	104	--> 0x0021	Ch: 0xFFFF, Seq: 8 (From Host)
2	0.000131	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
3	0.000245	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCABS	44	--> 0x0021	Ch: 0xFFFF, Seq: 11 (From Host)
4	1.320893	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	30	--> 0x0e00	Ch: 0x0003, Seq: 1 (From Host)
7	1.320890	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	30	--> 0x0e01	Ch: 0x0003, Seq: 2 (From Host)
10	1.330472	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	30	--> 0x0802	Ch: 0x0003, Seq: 3 (From Host)
13	1.330820	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	30	--> 0x0803	Ch: 0x0003, Seq: 4 (From Host)
16	1.337791	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	30	<-- 0x9010	Ch: 0x0003, Seq: 5 (To Host)
19	1.338369	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	30	<-- 0x9000	Ch: 0x0003, Seq: 6 (To Host)
22	1.343521	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	104	--> 0x0804	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_ENCABS	30	<-- 0x0804	Ch: 0x0003, Seq: 7 (To Host)
25	1.344060	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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 saving 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:

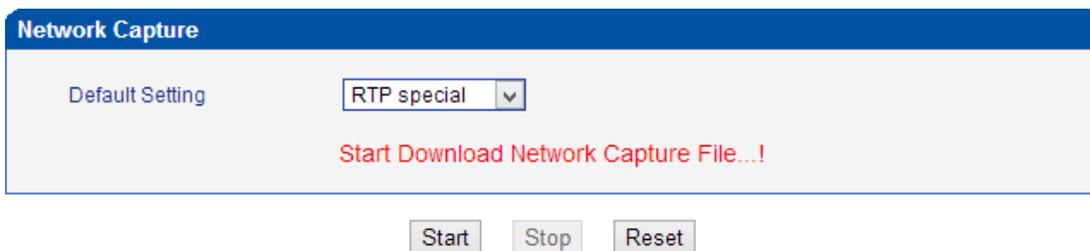
No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	172.16.222.22	1.1.1.1	Syslog	172	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 0>
2	0.000344	172.16.222.22	1.1.1.1	Syslog	520	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 1>
3	0.013432	172.16.222.22	1.1.1.1	Syslog	595	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 2>
4	0.013750	172.16.222.22	1.1.1.1	Syslog	176	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 3>
5	0.014059	172.16.222.22	1.1.1.1	Syslog	320	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 4>
6	0.014312	172.16.222.22	1.1.1.1	Syslog	172	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 5>
7	0.014806	172.16.222.22	1.1.1.1	Syslog	587	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 6>
8	0.028306	172.16.222.22	1.1.1.1	Syslog	662	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 7>
9	0.028759	172.16.222.22	1.1.1.1	Syslog	176	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 8>
10	0.029052	172.16.222.22	1.1.1.1	Syslog	587	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 9>
11	0.020017	172.16.222.22	1.1.1.1	Syslog	223	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 10>
12	0.331167	172.16.222.22	1.1.1.1	Syslog	983	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 11>
13	0.331498	172.16.222.22	1.1.1.1	Syslog	177	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 12>
14	0.331959	172.16.222.22	1.1.1.1	Syslog	907	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 13>
15	0.332307	172.16.222.22	1.1.1.1	Syslog	122	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 14>
16	0.332584	172.16.222.22	1.1.1.1	Syslog	111	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 15>
17	0.332848	172.16.222.22	1.1.1.1	Syslog	124	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 16>
18	0.333315	172.16.222.22	1.1.1.1	Syslog	526	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 17>
19	0.333603	172.16.222.22	1.1.1.1	Syslog	173	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 18>
20	0.333877	172.16.222.22	1.1.1.1	Syslog	386	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 19>
21	0.346687	172.16.222.22	1.1.1.1	Syslog	121	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 20>
22	0.347453	172.16.222.22	1.1.1.1	Syslog	120	USER, DEBUG: Jul 23 06:52:05 172.16.222.22 npe_sip: < 21>
23	0.332839	172.16.222.22	1.1.1.1	Syslog	533	USER, DEBUG: Jul 23 06:52:12 172.16.222.22 npe_sip: < 22>
24	0.233513	172.16.222.22	1.1.1.1	Syslog	177	USER, DEBUG: Jul 23 06:52:12 172.16.222.22 npe_sip: < 23>
25	0.233959	172.16.222.22	1.1.1.1	Syslog	457	USER, DEBUG: Jul 23 06:52:12 172.16.222.22 npe_sip: < 24>
26	0.234596	172.16.222.22	1.1.1.1	Syslog	287	USER, DEBUG: Jul 23 06:52:12 172.16.222.22 npe_sip: < 25>

▶ 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	365	Request: REGISTER sip:116.204.105.50
178	7.030000	116.204.105.50	172.16.221.228	SIP	411	Status: 200 OK (4 bindings)
244	11.610000	172.16.221.228	58.56.64.101	SIP/SDP	814	Request: INVITE sip:201858.56.64.101
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:201858.56.64.101:5060
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_ENCABS	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_ENCABS	44	--> 0x0021 Ch: 0xFFFF, Seq: 5 (From Host)
4	2.994581	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	44	--> 0x0021 Ch: 0xFFFF, Seq: 6 (From Host)
7	5.992790	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	44	--> 0x0021 Ch: 0xFFFF, Seq: 7 (From Host)
10	7.691428	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	30	<-- 0x9010 Ch: 0x0003, Seq: 1 (To Host)
13	7.701379	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	30	<-- 0x9000 Ch: 0x0003, Seq: 2 (To Host)
16	7.709662	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	30	<-- 0x8084 Ch: 0x0003, Seq: 3 (To Host)
19	7.710238	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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_ENCABS	30	<-- 0x8001 Ch: 0x0003, Seq: 4 (To Host)
22	7.716241	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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	7.716485	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCABS	30	<-- 0x8018 Ch: 0x0003, Seq: 5 (To Host)
25	7.716711	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	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.

Network Capture

Default Setting: Custom

Network Interface: LAN DSP

Source Host:

Destination Host:

Select Port: None

Protocol(s): TCP UDP RTP RTCP ICMP ARP

Start Download Network Capture File...!

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	<input type="button" value="Dsp Tdm Test"/>	<input type="button" value="Dsp IP Test"/>	<input type="button" value="Recover"/>
1	<input type="button" value="Dsp Tdm Test"/>	<input type="button" value="Dsp IP Test"/>	<input type="button" value="Recover"/>
2	<input type="button" value="Dsp Tdm Test"/>	<input type="button" value="Dsp IP Test"/>	<input type="button" value="Recover"/>
3	<input type="button" value="Dsp Tdm Test"/>	<input type="button" value="Dsp IP Test"/>	<input type="button" value="Recover"/>
4	<input type="button" value="Dsp Tdm Test"/>	<input type="button" value="Dsp IP Test"/>	<input type="button" value="Recover"/>
5	<input type="button" value="Dsp Tdm Test"/>	<input type="button" value="Dsp IP Test"/>	<input type="button" value="Recover"/>
6	<input type="button" value="Dsp Tdm Test"/>	<input type="button" value="Dsp IP Test"/>	<input type="button" value="Recover"/>
7	<input type="button" value="Dsp Tdm Test"/>	<input type="button" value="Dsp IP Test"/>	<input type="button" value="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 initiated 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
- ◆ Hang up

▶ **To start DSP IP Test:**

DSP IP Test is the loopback of VoIP side.

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

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

▶ **To start DSP IP Test:**

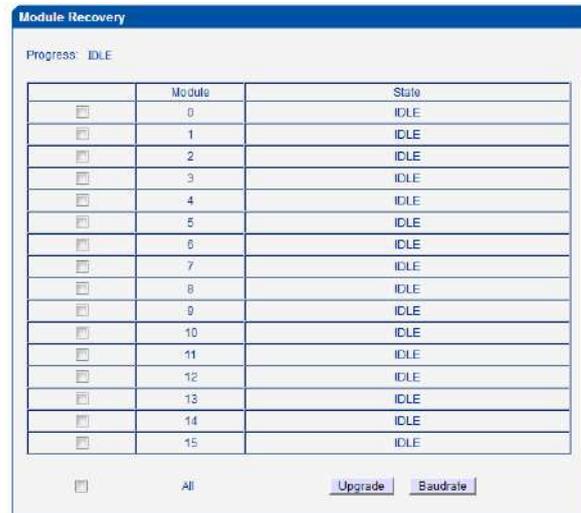
- ◆ Make a call test through gateway, the call can be initiated by IPPHONE. Keep the conversation after call establish
- ◆ Click DSP IP Test to start test
- ◆ Check the voice on both sides. GSM side become silence and echo should be generated on IPPHONE side
- ◆ Hang up

4.11.9 Mobile Network Test

GSM or WCDMA module test the call or register on web.

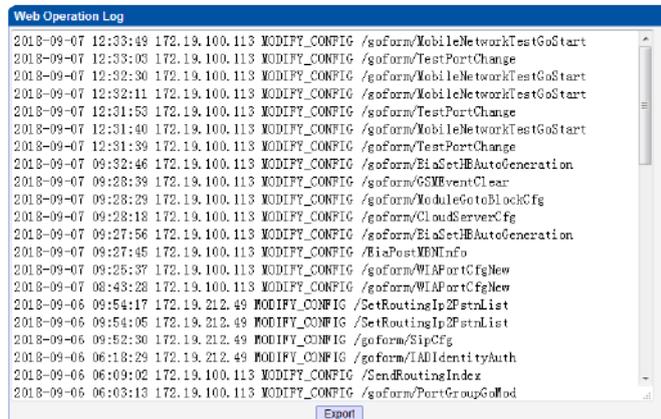
4.11.10 Module Recovery

When module unknown or version lower, we can update directly from network, but not all module version can update, please contact technical support to help you if you need.



4.11.11 Web Operation Log

Show the web operator log.



4.12 tools

4.12.1 File Upload

File Upload

Software Upload ▾

Send package file from your computer to the device.

Software
Choose File
No file chosen
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

UserBoard Upgrade

UserBoard No	Version	Status	Upgrade
0	B5.3.2.25L51 B.1	WORKING	Upgrade

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

4.12.3 Config Restore and Backup

Config Restore and Backup

Send data file from your computer to the device.

Configuration
Choose File
No file chosen
Restore

Data Backup

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

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 actual in the unit cannot be used for a restore operation on the unit.

4.12.4 Management Parameter

Management Parameter	
NTP Parameter	
NTP Enable	<input checked="" type="radio"/> Yes <input type="radio"/> No
Primary NTP Server Address	<input type="text" value="pool.ntp.org"/>
Primary NTP Server Port	<input type="text" value="123"/>
Secondary NTP Server Address	<input type="text"/>
Secondary NTP Server Port	<input type="text" value="123"/>
Check Interval	<input type="text" value="180"/> s
Time Zone	GMT+8:00 (Beijing, Singapore, Taipei, Hong Kong) ▼
WEB Parameter	
WEB Port	<input type="text" value="80"/>
Telnet Parameter	
Telnet Port	<input type="text" value="23"/>

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

Remote Server	
Enable	<input checked="" type="checkbox"/>
Server URL/IP	<input type="text" value="server02.dmclid.com"/>
Server Port	<input type="text" value="3100"/>

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.dmclid.com:3000

4.12.6 Email Account Setting

Email Setting	
Email Account	
E-mail	<input type="text"/>
Username	<input type="text"/>
Password	<input type="password"/> <input type="button" value="Show Password"/>
Outgoing(SMTP)	
Server	<input type="text"/>
Port	<input type="text"/>
TLS Enable	<input type="checkbox"/>
Incoming	
Protocol	IMAP ▾
Server	<input type="text"/>
Port	993
TLS Enable	<input checked="" type="checkbox"/>

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.

Username & Password	
Web Configuration	
Old Web Username	admin
Old Web Password	<input type="password"/>
New Web Username	<input type="text"/>
New Web Password	<input type="password"/>
Confirm Web Password	<input type="password"/>
Telnet Configuration	
Old Telnet Username	admin
Old Telnet Password	<input type="password"/>
New Telnet Username	<input type="text"/>
New Telnet Password	<input type="password"/>
Confirm Telnet Password	<input type="password"/>

4.12.8 Access control

Access Control	
New Password:	<input type="password"/> <input type="button" value="Show Password"/>
Confirm Password:	<input type="password"/> <input type="button" value="Show Password"/>

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.

Access Control		
Function	Permission	
Network Configuration	<input type="radio"/> Disable	<input checked="" type="radio"/> Enable
Cloud Server	<input type="radio"/> Disable	<input checked="" type="radio"/> Enable
Routing	<input type="radio"/> Disable	<input checked="" type="radio"/> Enable
SIP Trunk and SIP trunk group	<input type="radio"/> Disable	<input checked="" type="radio"/> Enable
SIP Configuration	<input type="radio"/> Disable	<input checked="" type="radio"/> Enable
Port and Port group	<input type="radio"/> Disable	<input checked="" type="radio"/> Enable
Config Restore and Backup	<input type="radio"/> Disable	<input checked="" type="radio"/> Enable
Manage(NTP WEB Telnet and Username&Password)	<input type="radio"/> Disable	<input checked="" type="radio"/> Enable
Software Upload	<input type="radio"/> Disable	<input checked="" type="radio"/> Enable
Factory Reset	<input type="radio"/> Disable	<input checked="" type="radio"/> Enable

Input the password: [Setting Password.](#)

4.12.9 Factory Reset

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

Factory Reset
<p>Click this button to reset factory default settings</p> <p>Notes:The device must restart to take effect.</p> <p><input type="button" value="Apply"/></p>

4.12.10 Auto Restart

Configure auto restart at pre-defined HH/MM

Auto Restart
<p>Auto Restart Enable <input checked="" type="radio"/> Yes <input type="radio"/> No</p> <p>Restart Time <input type="text" value="0"/> : <input type="text" value="0"/></p>

4.12.11 Restart

Restart
<p>Click this button to restart the device.</p> <p><input type="button" value="Restart"/></p>

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 commands 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
-----
  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           0           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 switches
  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 these 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