



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

User Manual



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1 Product Description

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

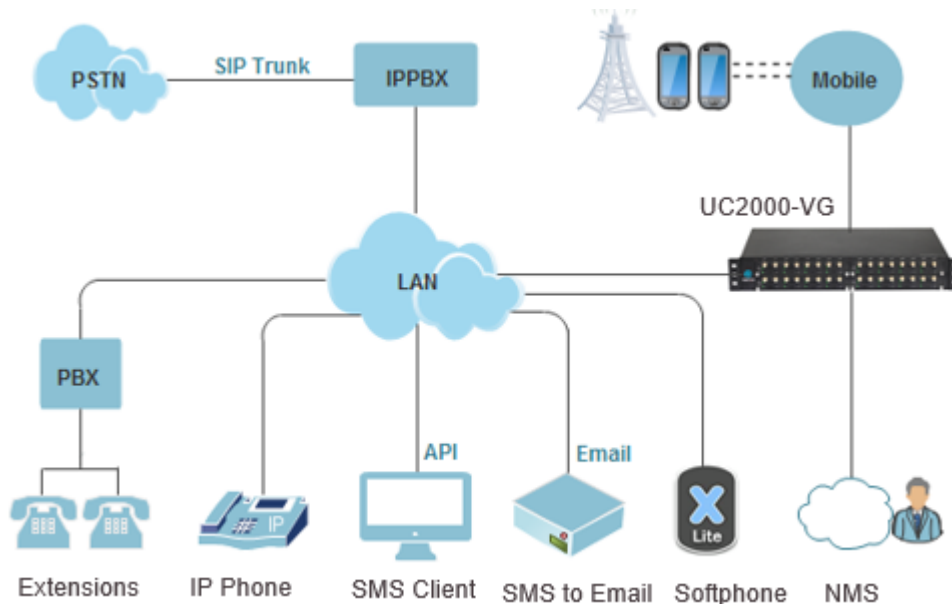
1.1 OVERVIEW

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

1.2 SCENARIO OF APPLICATION

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

Figure 1-2-1 Network scenario



1.3 PRODUCT APPEARANCE

1.3.1 Product Appearance of UC2000-VE

The appearance of UC2000-VE shows as follow

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

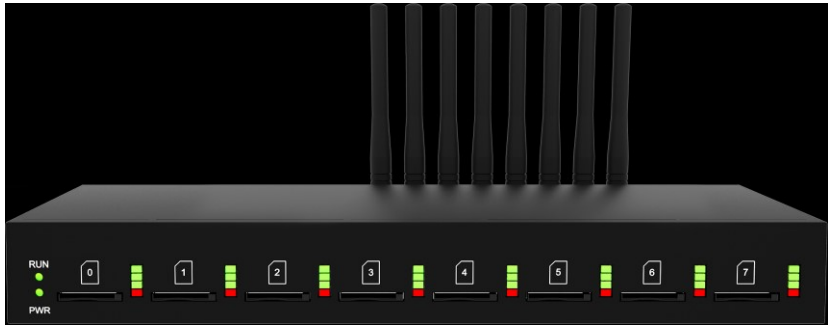


Table 1-3-1 Description of Front view

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

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



Table 1-3-2 Description of Rear view

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

1.3.2 Product Appearance of UC2000-VF

Front View

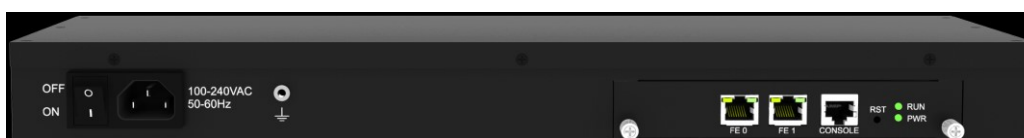


Indicators and connectors

Indicators	Name	Status	Description
------------	------	--------	-------------

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

Back view



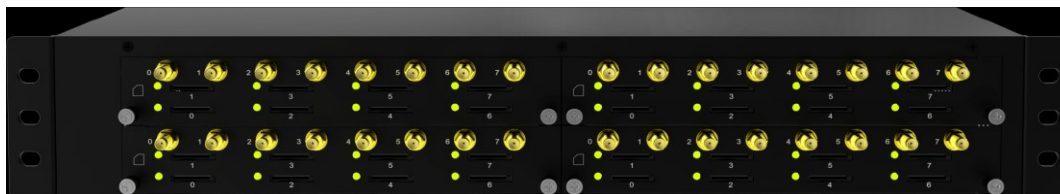
Indicators and connectors

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

		ON	Device is booting up
		OFF	Device is not booting up

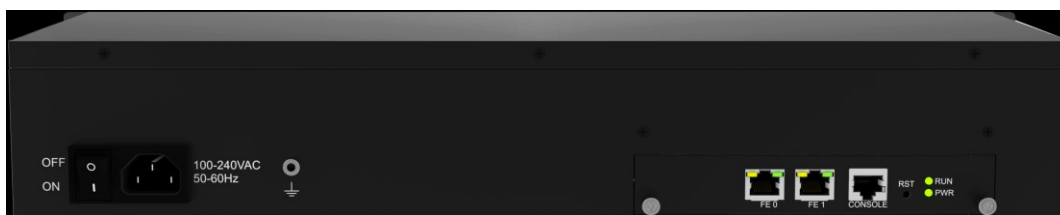
1.3.3 Product appearance of UC2000-VG

Front view



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

Back View



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

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

1.4 FUNCTIONS AND FEATURES

1.4.1 Protocols

- Standard SIP;
- Simple Traversal of UDP over NATs (STUN);
- Point-to-point protocol over Ethernet (PPPoE);
- Hypertext Transfer Protocol (HTTP);
- Dynamic Host Configuration Protocol (DHCP);
- Domain Name System (DNS);
- ITU-T G.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

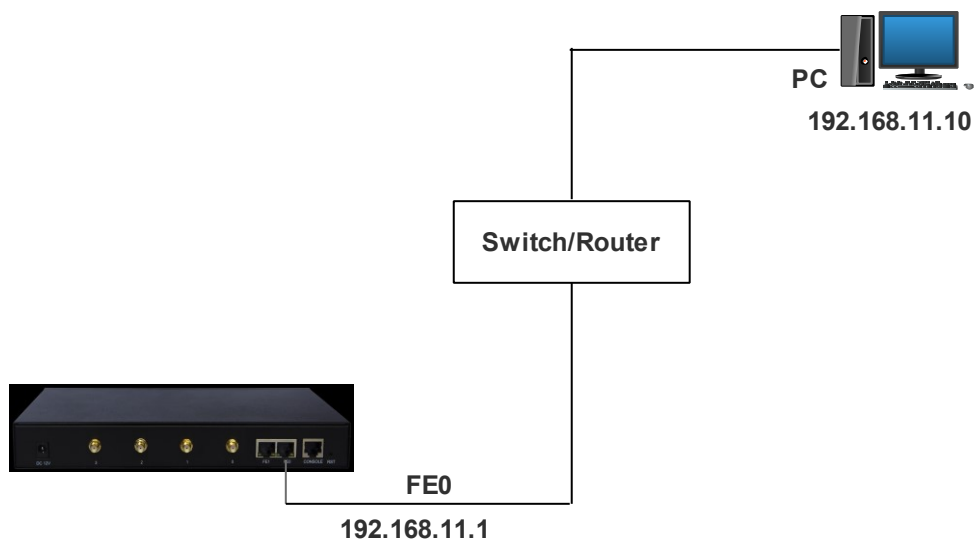
Please pay attention to the following before you install UC2000-VE/F/G include:

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

2.2 INSTALLATION PROCEDURES

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

2.3 NETWORK CONNECTION



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

3 Basic Operation

3.1 FEATURE CODES

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

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

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

3.2 BASIC OPERATION

3.2.1 Check IP address

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

3.2.2 Restore factory setting via IVR

Use a mobile phone to call a SIM card number of the device, the device will answer and play a voice prompt of ‘dial the extension number’. Press *166*000000# on the mobile

phone, then you will hear 'setting succeeds', then the factory setting of the gateway will be restored.

3.2.3 Restore default IP and password

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

3.2.4 Restore factory setting

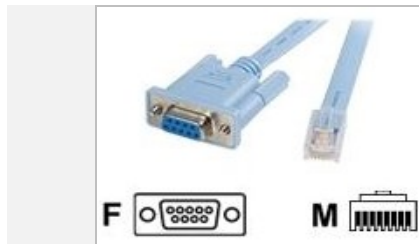
Press RST button for about 7 seconds, then gateway will be rebooted and restored to factory setting.

3.3 LOCAL MAINTENANCE THROUGH CONSOLE PORT

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

➤ Example: Log in device via Console Port

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

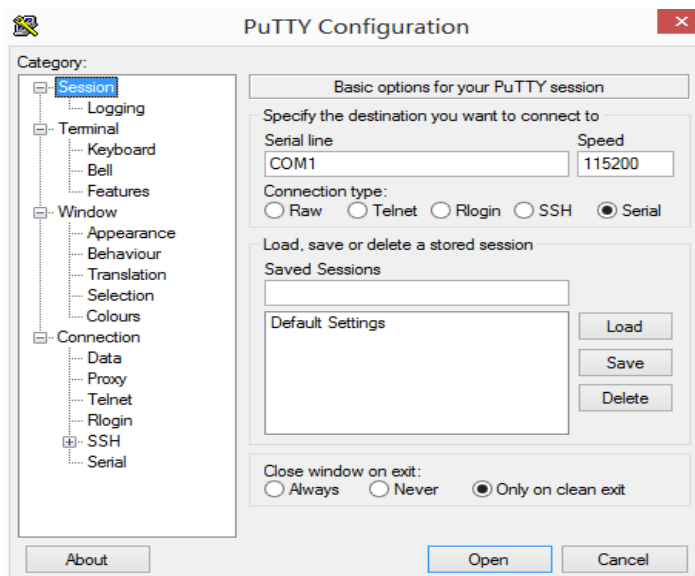


Step 2: Connect the F port of the serial cable with the COM port of PC. If the PC does not have a COM port, please use a USB-to-COM converting tool to connect the serial cable with the PC.

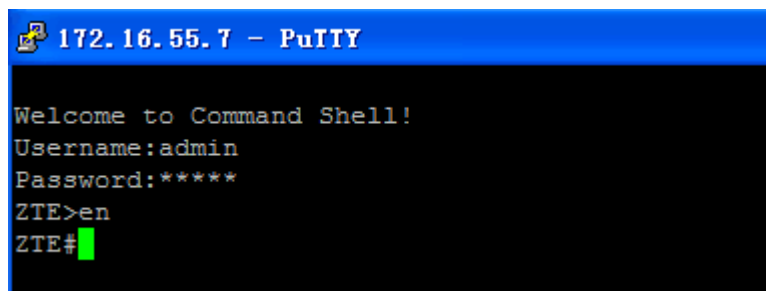
Step 3: Connect the M port of the serial cable with the console port of the device.

Step 4: Conduct configurations on login software.

Herein we take the PuTTY 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 HTTP protocol. We are strongly recommend to access device with Google Chrome or Firefox Browser.

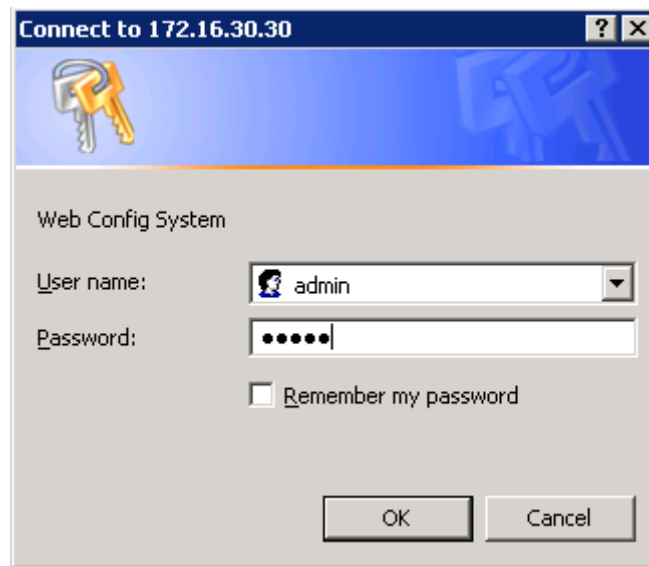
The configuration introduction also suitable for following models:

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

4.1 ACCESS UC2000-VE UNIT

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

Figure 4-1-1 WEB log interface



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

4.2 PARAMETERS CONFIGURATION

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

Figure 4-2-1 WEB introduce

Run Information			
MAC Address	F8-A0-3D-48-4D-15		
Network Mode	Bridge		
Network	187.95.122.4	255.255.255.248	Static
DNS Server	8.8.8.8	8.8.4.4	
Device SN	db00-0040-ad00-0143		
Hardware ID	0000-1758-0474		
Cloud Register Status	Not Registered		
License	Basic Function	Enable	
	DBO Advanced	Enable	
System Up Duration	14 h 2 m 52 s		
System Time	2017-4-17 20:08:30		
Network Traffic Statistics	Received 293040697 Bytes	Sent 278040658 Bytes	
Version Information	Device Model	UC2000-VG	
	Package Version	02231220 2017-04-12 09:57:12 official	
	Software Version	02231220 2017-04-12 09:49:25	
	Web Version	02231220	
	Hardware Version	PCB 2	
	Logic Version	LOGIC 0	
	DSP Version	Branch3.0.0.0	
	Userboard 0 Version	B5.3.2.23L51 C.1	DB02-740A-D000-0749
	Userboard 1 Version	B5.3.2.23L51 C.1	DB02-740A-D000-0484
	Userboard 2 Version	B5.3.2.23L51 C.2	DB05-6510-9000-0093
	Userboard 3 Version	B5.3.2.23L51 C.1	DB02-7408-6000-2149

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-48-4D-15		
Network Mode	Bridge		
Network	187.95.122.4	255.255.255.248	Static
DNS Server	8.8.8.8	8.8.4.4	
Device SN	db00-0040-ad00-0143		
Hardware ID	0000-1758-0474		
Cloud Register Status	Not Registered		
License	Basic Function	Enable	
	DBO Advanced	Enable	
System Up Duration	14 h 2 m 52 s		
System Time	2017-4-17 20:08:30		
Network Traffic Statistics	Received 293040697 Bytes	Sent 278040658 Bytes	
Version Information	Device Model	UC2000-VG	
	Package Version	02231220 2017-04-12 09:57:12 official	
	Software Version	02231220 2017-04-12 09:49:25	
	Web Version	02231220	
	Hardware Version	PCB 2	
	Logic Version	LOGIC 0	
	DSP Version	Branch3.0.0.0	
	Userboard 0 Version	B5.3.2.23L51 C.1	DB02-740A-D000-0749
	Userboard 1 Version	B5.3.2.23L51 C.1	DB02-740A-D000-0484
	Userboard 2 Version	B5.3.2.23L51 C.2	DB05-6510-9000-0093
	Userboard 3 Version	B5.3.2.23L51 C.1	DB02-7408-6000-2149

Table 4.3-1 System Information

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

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

4.3.2 Mobile Information

Figure 4.3-2 Mobile Information

Mobile Information												
	Port	Type	IMSI	IMEI	Status	Remaining Call Duration	Carrier	Signal Quality	BER	ASR(%)	ACD(s)/PDD(s)	Call Status
	0	GSM	460004122170725	863070010384344	Mobile Registered	No Limit	CHINA MOBILE		0	0	0	Idle
	1	GSM	460020187266423	863070010384310	Mobile Registered	No Limit	CHINA MOBILE		0	0	0	Idle
	2	GSM		863070010385747	No SIM Card	No Limit			0	0	0	Idle
	3	GSM		863070010385549	No SIM Card	No Limit			0	0	0	Idle
	4	GSM		863070011821336	No SIM Card	No Limit			0	0	0	Idle
	5	GSM		863070011821880	No SIM Card	No Limit			0	0	0	Idle
	6	GSM		863070011899175	No SIM Card	No Limit			0	0	0	Idle
	7	GSM		863070011764668	No SIM Card	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 network. Such as CDMA or GSM
IMSI	International Mobile Subscriber Identity, it is the uniquely identifies of SIM card
Status	Indicates the connection status of current GSM / CDMA module
Remaining Call Duration	It showing available total call minutes of SIM card while call limitation is enabled.
Carrier	Displays the network carrier of current SIM card.
Signal Quality	Displays the signal strength of in each channels of GSM / CDMA.
BER	Its indicate error rates between mainboard and userboard, Modular and Base station
ASR	Answer Seizure Ratio is a measure of network quality. It's calculated by taking the number of successfully answered calls and dividing by the total number of calls attempted. Since busy signals and other rejections by the called number count as call failures, the ASR value can vary depending on user behavior.
ACD	The Average Call Duration (ACD) is calculated by taking the sum of billable seconds (bill sec) of answered calls and dividing it by the number of these answered calls.
PDD	Post Dial Delay (PDD) is experienced by the originating customer as the time from the sending of the final dialed digit to the point at which they hear ring tone or other in-band information. Where the originating network is required to play an announcement before completing the call then this definition of PDD excludes the duration of such announcements.
Call Status	Show the Status of port, include idle, active, alert and processing <i>Idle</i> means there is no call on this channel <i>Processing</i> means call is connecting

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

4.3.3 SIP Information

Figure 4-3-3 SIP Information

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

[Refresh](#)

Displays registration status information with Softswitch platform or SIP Server

Table 4-3-3 SIP information

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

4.4 STATISTICS

4.4.1 TCP/UDP

Figure 4-4-1 TCP/UDP Statistics

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

4.4.2 RTP

Figure 4-4-2 RTP

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

Table 4-4-1 Description of RTP Statistics

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

4.4.3 SIP Call History

SIP Call History

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

[Refresh](#)

SIP Call History

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

4.4.4 IP to GSM Call History

IP to GSM Call History

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

Refresh Clear

IP to GSM Call History

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

4.4.5 CDR Report

Figure 4-4-5 CDR Report

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

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

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

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

4.4.6 Lock BCCH History

Figure 4-4-6 Auto Lock BCCH History

Auto Lock BCCH History			
Select Port		Port 0 ▾	
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.

Current Call Status						
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	
☒ Use the following IP address	
IP Address	172.16.222.22
Subnet Mask	255.255.0.0
Default Gateway	172.16.1.5
☐ PPPoE	
Account	
Password	
Service Name	
MTU	1400
DNS Server	
☐ Obtain DNS server address automatically	
☒ Use the following DNS server addresses	
Primary DNS Server	8.8.8.8
Secondary DNS Server	0.0.0.0

Table 4-5-1 Local network

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

4.5.2 ARP

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

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

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

Figure 4-5-3 Add ARP

The IP format is: xxx.xxx.xxx.xxx
The MAC format is: xx-xx-xx-xx-xx-xx

OK Search All

Click *Search All* to check ARP buffer.

Type ☐ static ☒ dynamic

Type	IP Address	MAC Address
☐	172.16.221.43	BC-AE-C5-4E-15-F5
☐	172.16.236.129	2C-D0-5A-12-D5-2A
☐	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	us1.suvpn.com
Account	gary@dinstar.com
Password	*****
Domain	us1.suvpn.com
Use MPPE	off v

Table 4-5-3 Description of VPN Parameter

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

Check VPN connecting status on system information

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.

4.7 MOBILE CONFIGURATION

4.7.1 Basic Configuration

Basic Configuration

SIM Mode

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

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

Introduction to API

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

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

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

Introduction to GSM Audio Coding

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

Auto: it means GSM Audio Coding is automatic.

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

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

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

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

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

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

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

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

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

Example:

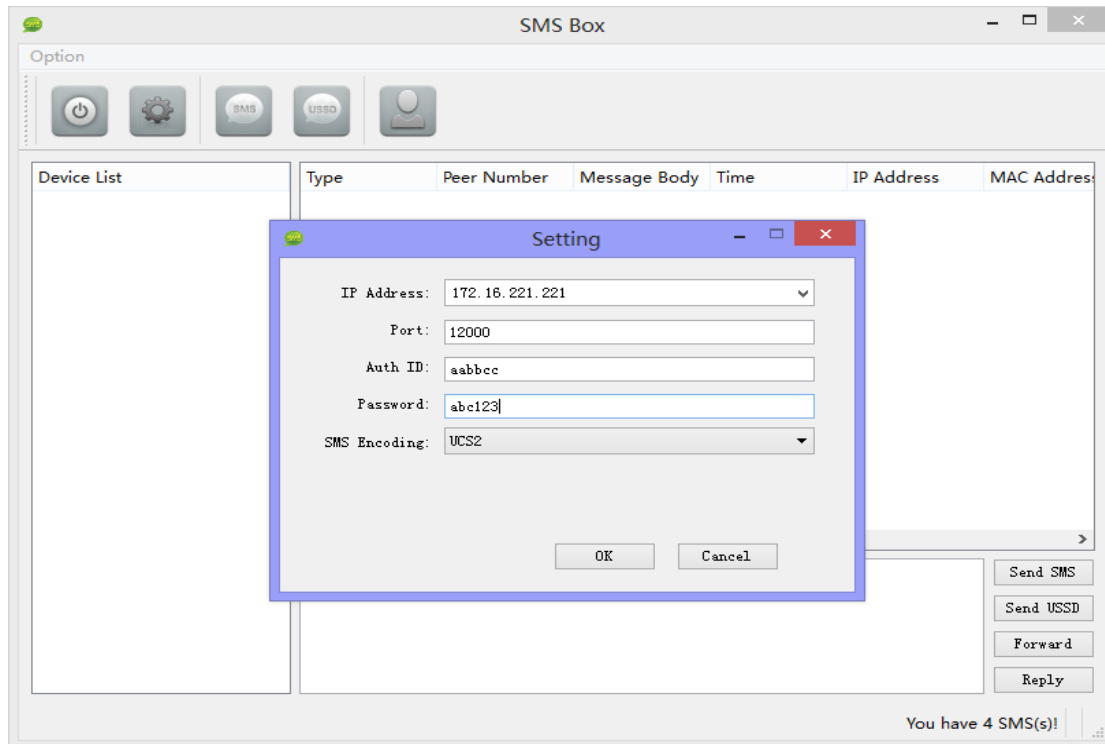
Configuration between SMS box and gateway

Configure API parameters on gateway

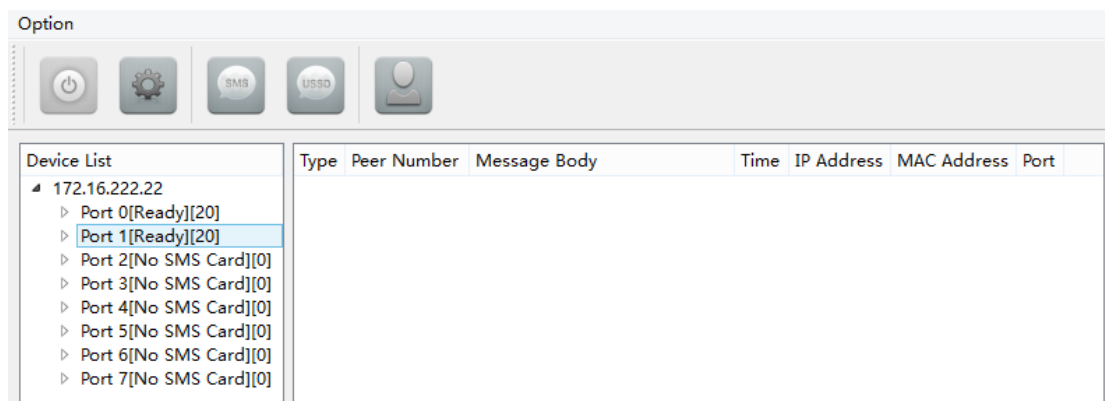
Remote API Enable	☐ No ☒ Yes
API Server Address	172.16.221.221
API Server Port	12000
API User ID	aabbcc
API User Password	***** Show Password

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	Tx Gain/dB	Rx Gain/dB	APN	Band Type	Net Work Mode	SMSC	Reset	Block /Open	Power On/Off
☐ 0	No ▾	Yes ▾	3	7		Default(Auto)	Default(Auto) ▾		Reset	Block	OFF
☐ 1	No ▾	Yes ▾	3	7		Default(Auto)	Default(Auto) ▾		Reset	Block	OFF
☐ 2	No ▾	Yes ▾	3	7		Default(Auto)	Default(Auto) ▾		Reset	Block	OFF
☐ 3	No ▾	Yes ▾	3	7		Default(Auto)	Default(Auto) ▾		Reset	Block	OFF
☐ 4	No ▾	Yes ▾	3	7		Default(Auto)	Default(Auto) ▾		Reset	Block	OFF
☐ 5	No ▾	Yes ▾	3	7		Default(Auto)	Default(Auto) ▾		Reset	Block	OFF
☐ 6	No ▾	Yes ▾	3	7		Default(Auto)	Default(Auto) ▾		Reset	Block	OFF
☐ 7	No ▾	Yes ▾	3	7		Default(Auto)	Default(Auto) ▾		Reset	Block	OFF
☐ All	Copy	Copy	Copy	Copy	Copy		Copy	Copy	Reset	Unblock	ON
	No ▾	Yes ▾	3	7			Default(Auto) ▾			Block	OFF

BandType ☐ GSM 850 ☐ GSM 900 ☐ GSM 1800 ☐ GSM 1900 ☐ WCDMA 800 ☐ WCDMA 850 ☐ WCDMA 900 ☐ WCDMA 1900 ☐ WCDMA 2100

[Copy to the selected](#) [Save](#) [Refresh](#)

Description of Mobile Configuration

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

4.7.3 Phone Number Config

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

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

Port	SlotA	SlotB	SlotC	SlotD
0	13388889999			
1				
2				
3				
4				
5				
6				
7				

SET

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

PIN Management

Select Port: Port 0 ▼

SIM Card Lock: ☒ No ☐ Yes

PIN Code:

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.

Agreement

IMEI modify Service Agreement

Welcome to use IMEI modify Service. As you used this service, that means you accept the following terms of service.

(i) For testing only

IMEI modify Service is only for your personal use and only for the testing, it shall not be used for any commercial purpose.

You warrant that you will not in any violation of any laws applicable to your jurisdiction of any laws or regulations ways to use IMEI modify Service.

(ii) Disclaimer

You understand and agree that your use IMEI modify Service is completely out of your own judgment, and you will take full responsibility for any losses and resulting from any accident,

Dinstar does not assume any legal responsibility.

Dinstar

☐ I have read and I accept the agreement

IMEI Modify

IMEI Auto Set

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

IMEI Auto Set

Enable IMEI Auto Set

☐ No
 ☒ Yes

IMEI TAC

86008501

86011600

99000100

86014300

86000300

86275202

86553802

86121302

IMEI Serial No Range

Min IMEI Serial No

111111

Max IMEI Serial No

888890

Policy

☐ By Time
 ☐ By Call
 ☒ Change Sim

10

Minute(s) (Note:Not less than 10)

30

Call(s) (Note:Not less than 10)

Note:1. IMEI = TAC(8 digits) + Serial No(6 digits) + Check digit(1 digit).

2. There is up to 8 TACs can be set.

Save

4.7.6 Operator

Operator				
	Port	Operator code	Operator List	Search
☐	0		▼	Search
☐	1		▼	Search
☐	2		▼	Search
☐	3		▼	Search
☐	4		▼	Search
☐	5		▼	Search
☐	6		▼	Search
☐	7		▼	Search
☐	all	Copy		Search

Click Search button while there is SIM card in that port, after a while, you will see Operator codes list under Operator List dropbox. And then select correct operator code which match with the SIM card insert in the gateway. 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
☒	460038	Fixed ▼	46003
☐		Auto ▼	
☐		Auto ▼	
☐		Auto ▼	
☐		Auto ▼	
☐		Auto ▼	
☐		Auto ▼	
☐		Auto ▼	

Save

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

4.7.8 BCCH

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

Configuration Procedures for BCCH:

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

Step 2. Drag the scroll bar on the bottom of the interface, and you will see 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	MCC	MNC	LAC	CID	BCCH	Receive Level

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

Second of BCCH

Third of BCCH

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	-90 db					
Auto Period between	1	and	15	min		
Switch BCCH in Calling	☒ No ☐ Yes					
Apply To All Ports	☒ No ☐ 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	-90 db					
Call Times	15	Minimum ASR	30	%		
Call Failed	6					
Apply To All Ports	☒ No ☐ 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.

BCCH Blacklist						
	1	2	3	4	5	6
BCCH						

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

4.7.9 Call Forwarding

Calls can be forwarded unconditionally or under certain conditions.

Call Forwarding			
Port	Options	Call forwarding settings	Search
☐ 0			Search
☐ 1			Search
☐ 2			Search
☐ 3			Search
☐ 4			Search
☐ 5			Search
☐ 6			Search
☐ 7			Search
☐ all	Unconditional	Call Forwarding Unconditional	Search
Copy			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		
☐ 0	☐ Disable	☐ Enable	No SIM card.
☐ 1	☐ Disable	☐ Enable	No SIM card.
☐ 2	☐ Disable	☐ Enable	No SIM card.
☐ 3	☐ Disable	☐ Enable	No SIM card.
☐ 4	☐ Disable	☐ Enable	No SIM card.
☐ 5	☐ Disable	☐ Enable	No SIM card.
☐ 6	☐ Disable	☐ Enable	
☐ 7	☐ Disable	☐ Enable	
☐ All	☐ Disable ☐ 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 is only take 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.

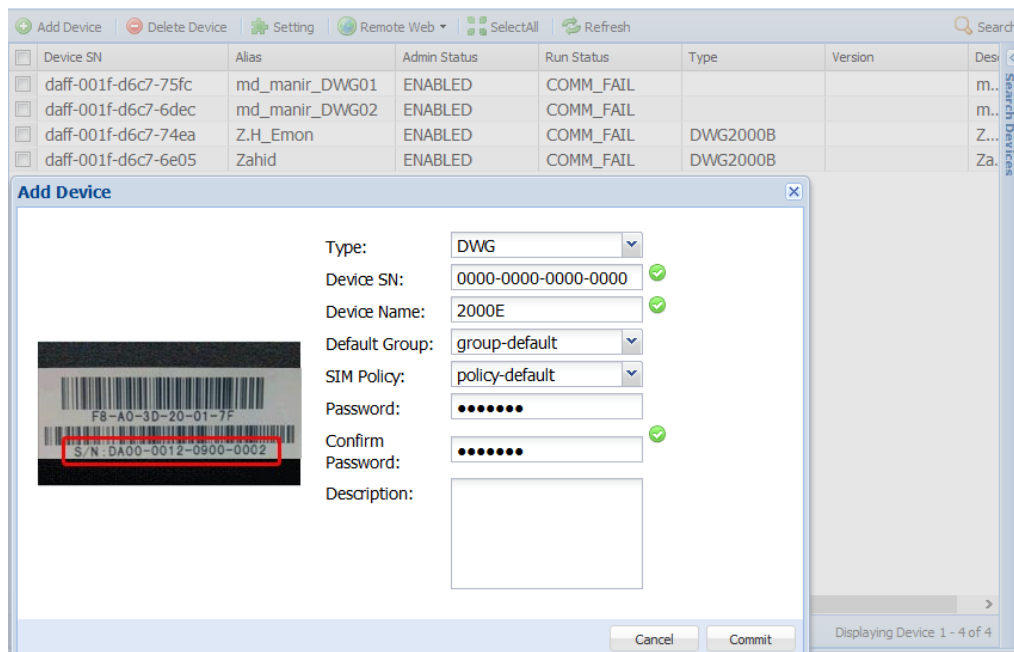
Cloud Server	
Primary Server Domain	best.cloud.com
Primary Server Port	2020
Secondary Server Domain	
Secondary Server Port	
Domain Name	
Password	Show Password
LocalPort	0
SIM Transport Type	Auto
Port State Control by	Cloud
Anti Call Scanning	☐ Enable
Reboot device when register failed over	0 minutes (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

4.8 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
☐	0	--	--	--	--	--
☐	1	--	--	--	--	--
☐	2	--	--	--	--	--
☐	3	--	--	--	--	--
☐	4	--	--	--	--	--
☐	5	--	--	--	--	--
☐	6	--	--	--	--	--
☐	7	--	--	--	--	--
☐	All	Clear		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	0 ▼
Description	Vodafone
Daily Limit	0 Note:0 means no limit
Monthly Limit	20 Note:0 means no limit
Reset Date	1 ▼
Port Group	0 <all> ▼
Save Reset Cancel	

4.8.3 Send SMS

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

Send Message

Port

☐ 0
☐ 1
☐ 2
☐ 3
☐ 4
☐ 5
☐ 6
☐ 7
☐ All

Send Mode

☒ Mode 1
☐ Mode 2

To

Encoding Message

UCS2

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

4.8.4 SMS Outbox

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

SMS Outbox

Start Date : 2017 Year 4 Month 5 Day

Select Port All

Number

End Date : 2017 Year 4 Month 5 Day

Send Status ANY

Report Export

Export

Refresh

Clear

Port	Send Date	Number	SMS Content	Send Status
------	-----------	--------	-------------	-------------

Total: 0 entries 16 entries/page 1/0 page

4.8.5 SMS Inbox

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

4.8.6 USSD

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

4.8.7 Email

How to set Email to SMS

Description

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

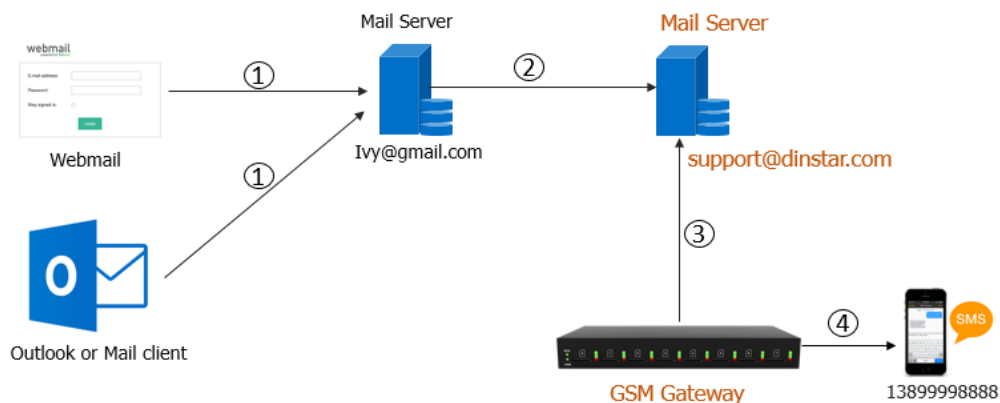
When email use protocol IMAP, if email read successful, will set the email status to read. If read email failed, will try to read again (MAX 3 times). If failed final, the email status will be set to read also.

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

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

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

How does SMS to email works



- 1) Send Email from webmail or outlook client
Email Format: Plain text

Email subject: Test SMS

Email contents:

To:13899998888
Encoding:7Bit
Message:Hola, this is test SMS from ivy@gmail.com
- 2) Gmail server forward email to support@dinstar.com
- 3) GSM gateway check the inbox of support@dinstar.com, find the email subject with 'Test SMS'
- 4) GSM gateway send SMS to mobile 13899998888

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

How to configure Email to SMS in GSM gateway

- 1) Open page **SMS and USSD>>>>>>Email**.

Email to SMS support both POP3 and IMAP protocol.

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

The "TLS Enable" means use Encrypt or not.

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

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

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

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

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

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

Email Setting	
Email Sender	
Email Address of Sender	
SMTP Domain	
SMTP Port	
SMTP Username	
SMTP Password	Show Password
TLS Enable	☐
SMS to Email	☐ Enable
GSM Event to Email	☐ Enable
Email To SMS	☒ Enable
Protocol	POP3 ▼
Server Domain	pop3.dinstar.com
Server Port	110
TLS Enable	☐
Using SMTP Login Account	☐
Username	support@dinstar.com
Password	***** Show Password
Check Email Every	5 Minutes(Up to 60)
Subject	Test SMS

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:xxxxxxxxxxxxx

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:

From: lich@dinstar.com

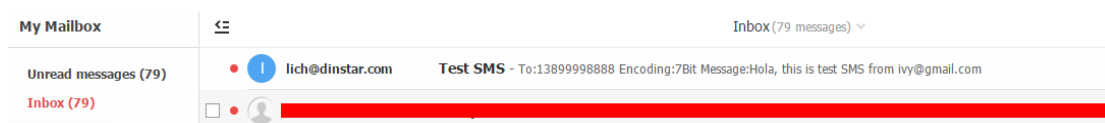
To: support <support@dinstar.com>

Cc:

Subject: Test SMS

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

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



Note:

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

How to set SMS to Email

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

SMS Setting

SMS to Email

☒ Enable

Email Address of Sender

SMTP Domain

SMTP Port

SMTP Username

SMTP Password

TLS Enable

☐

Destination Email Address 1

Destination Email Address 2

Destination Email Address 3

Title

Email To SMS

☐ Enable

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

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

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

SMTP Username: Enter your e-mail address username.

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

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

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

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

Here is one example:

SMS Setting

SMS to Email

☒ Enable

Email Address of Sender: support@dinstar.com

SMTP Domain: smtp.dinstar.com

SMTP Port: 25

SMTP Username: support@dinstar.com

SMTP Password: ***** [Show Password](#)

TLS Enable: ☐

Destination Email Address 1: may@dinstar.com

Destination Email Address 2:

Destination Email Address 3:

Title: SMS from DWG

Email To SMS

☐ Enable

[Save](#)

4.9 CALL CONFIGURATION

4.9.1 SIP Configuration

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

► Configure SIP server and Outbound Proxy server

SIP Proxy	
SIP Server Address	
SIP Server Port(default: 5060)	5060
Check Net Status	☒ No ☐ Yes
Outbound Proxy	
Outbound Proxy Address	
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 are able to penetrate the firewall.

► Local SIP Port Configuration

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

All Ports Register Used Same User ID	☒ No ☐ Yes
Use Same Local Sip Port	☐ No ☒ Yes
Use Random Port	☒ No ☐ Yes
Local SIP Port	5060

► Random

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

► Use the same SIP port

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

Use the same local SIP port and SIP User ID

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

► Use the separate SIP port

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

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

All Ports Register Used Same User ID ☒ No ☐ Yes
Use Same Local Sip Port ☒ No ☐ Yes

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

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

► Register Interval and DNS SRV

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

► Is Register

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

► Register Interval

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

► DNS query type

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

► DNS refresh interval

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

► Configuring SIP Timers

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

► T1

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

► **T2**

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

► **T4**

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

► **TMAX**

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

► **Keepalive Interval**

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

► **Keepalive SIP ID**

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

OPTIONS sip:heartbeat@172.16.0.8:2080 SIP/2.0

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

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

To: <sip:heartbeat@172.16.0.8:2080>

Call-ID: 8874a4e49f11af243c6b717c05a16e35@172.16.222.22

CSeq: 1804289386 OPTIONS

Contact: <sip:31@172.16.222.22>

Max-Forwards: 70

Accept: application/sdp

Content-Length: 0

► Keepalive Retry Count

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

► Configuring Caller ID and 183 Mode

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

► From Mode when Caller ID Is Available

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

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

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

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

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

► From Mode when Caller ID Is Unavailable

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

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

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

► Answer Mode

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

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

► 183 Mode

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

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

► Session Timer

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

Session Timer

☐ No ☒ Yes

Session Timer Interval(range:90 - 60000s) s

Session timer mode ▼

Session timer refresher ▼

► Session timer Interval

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

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

► Session Timer Refresher

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

UPDATE sips:bob@192.0.2.4 SIP/2.0

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

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

Supported: timer

Session-Expires: 4000;refresher=uac

Max-Forwards: 70

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

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

Call-ID: a84b4c76e66710

CSeq: 314162 UPDATE

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

► Configuring GSM-SIP Mapping Code

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

Gsm-Sip Code Map	
Gsm Code Enable	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 need some specific SIP response from the gateway. Example, SIP server need SIP response *180 Ringing* instead of *183 Ringing*, the configuration should be as below:

Response Code switch	Response code after switch
183	180

4.9.2 SIP Trunk Configuration

IP Trunk					
	Index	IP	Port	Description	KeepAlive Enable
☐	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	31
IP	172.16.221.221
Port	5060
Description	Elastix
KeepAlive Enable	☐

4.9.3 SIP Trunk Group

Figure 4-11-3 IP Trunk Group

IP Trunk Group			
	Index	Description	IP
☐	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 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: 31

Description: Default

IP	Index	IP	Port
☒	31	172.16.221.221	5060

4.9.4 Port Configuration

Port List								
Port	SIP User ID	Authenticate ID	Tx Gain	Rx Gain	To VOIP Hotline	To PSTN Hotline	Auto-Dial Delay Time(s)	Detail
0	2001	2001	2	6	00		3	Detail

Figure 4-12-4 Port Configuration

Port Configuration

All ports register used same user ID: ☒ No ☐ Yes

Current Port: Port 0

SIP User ID:

Authenticate ID:

Authenticate Password: [Show Password](#)

Tx Gain: +2dB

Rx Gain: +6dB

To VOIP Hotline:

To PSTN Hotline:

[Save](#) [Back](#)

Table 4-12-3 Description of Port Configuration

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

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

4.9.5 Port Group Configuration

Port Group						
	Index	Description	SIP User ID	Authenticate ID	Port	Select Mode
☐	0	0-3			0,1,2,3,	Cyclic Ascen...
☐	31	4-7			4,5,6,7,	Cyclic Desce...

Total: 2entry 16entry/page 1/1page Page 1

Add

Delete

Modify

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

Select ports for definted port group.

Port Group Modify

Index	0									
Description	0-3									
SIP User ID										
Authenticate ID										
Authenticate Password		Show Password								
Select Mode	Cyclic Ascending v									
Port	<table><tr><td>☒ Port 0</td><td>☒ Port 1</td></tr><tr><td>☒ Port 2</td><td>☒ Port 3</td></tr><tr><td>☐ Port 4</td><td>☐ Port 5</td></tr><tr><td>☐ Port 6</td><td>☐ Port 7</td></tr></table>		☒ Port 0	☒ Port 1	☒ Port 2	☒ Port 3	☐ Port 4	☐ Port 5	☐ Port 6	☐ Port 7
☒ Port 0	☒ Port 1									
☒ Port 2	☒ Port 3									
☐ Port 4	☐ Port 5									
☐ Port 6	☐ Port 7									

4.9.6 Digitmap

Digit Map

Digit Map	x.T x.#
-----------	--------------------------------------

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 | 13xxxxxxxxxx

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)xxxxxxxxxx

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	62 ▼
Description	
Source	SIP Server ▼
Destination	port-group-0 <all> ▼
Call Restriction	Allow Call ▼
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 out

Source Prefix: to match with prefix of caller number

Destination Prefix: to match with prefix of called number

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

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

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

4.9.9 Service parameter

► To configure dialing mode parameters

Do Not Answer GSM Incoming Call for Hotline	☐ No ☒ Yes
Enable GSM Incoming Configuration	☐ No ☒ Yes
Auto Outgoing Routing Type	Polling v
IP to GSM One Stage Dialing	☐ No ☒ Yes
Answer Delay	5 s
Redirect Call When All Ports Busy	☒ No ☐ Yes
Play Voice Prompt for GSM Incoming Calls	☐ No ☒ Yes
RTP Detected Enable	☐ No ☒ Yes
Period without RTP Packet	90

► Do Not Answer GSM Incoming Call for Hotline

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

Notes: Refer to [Port Parameter](#) page for [Hotline](#) configuration.

► Enable PSTN Incoming Configuration

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

► Enable Auto Outgoing Routing

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

► IP to PSTN One Stage Dialing

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

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

► Answer Delay

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

► Redirect Call when All Port Busy

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

IP and Port: destination gateway or IPPBX to be redirect

► Play Voice Prompt for PSTN Incoming Calls

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

► RTP Detect

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

Nat Traversal

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

Include Static NAT, Dynamic NAT and STUN

STUN (Simple Traversal of UDP over NATs) is a network protocol. It is allowed to stay behind the NAT (or multiple NAT) client part to identify their clients' public

address, found himself after what Type of NAT and NAT for a particular Channel is bound to a local Internet terminal Channel. This information is used for two host to set up UDP communication behind the same NAT router. The agreement defined by the RFC 3489

► Other configuration

Other Configuration	
Enable Private Service	☐ No ☒ Yes
User ID Is Phone Number	☒ No ☐ Yes
Only Accept Calls from SIP Server	☒ No ☐ Yes
Allow Call from GSM to IP without Registration	☐ No ☒ Yes
Allow Call from IP to GSM without Registration	☒ No ☐ Yes
Reject Anonymous Call from IP to GSM	☒ No ☐ Yes
Use # as End Key	☐ No ☒ Yes
No Answer Timeout	55 Sec(s)
Interdigit Timeout	4 Sec(s)
Reset ASR after SIM Switching	☒ no ☐ 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.

► Only Accept Calls from SIP Server

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

► Allow Call from PSTN to IP without Registration

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

► Allow call from IP to PSTN without Registration

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

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

The incoming anonymous calls will be rejected

► **Use # as End Key**

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

► **Inter-digit Timeout**

Timeout without dialing

4.9.10 Media parameter

Local Start RTP Port	8000
Enable Silence Suppression	☐ No ☒ Yes
Call Progress Tone	USA
Ring Back Tone	440,280,480,280,2000,400
Busy Tone	480,330,620,330,500,500,1
Dial Tone	350,260,440,260,0,0,0,0

► **Local Start RTP Port**

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

► **Enable Silence Suppression**

Enable the "silence suppression" almost no impact on call quality, and can save about half of the bandwidth;

► **Call Progress Tone**

Each country has its different call progress tone required standards, such as busy tone, ring back tones and ring tone standards, users can select the area standard from here

USA Standard:

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

on:2000ms off:4000ms

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

500ms

► DTMF Parameter

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

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

► System IVR

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

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

► Configure codec list

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

4.9.11 DBO Parameter

Enable DBO service

DBO Parameter	
Enable DBO	☐

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	☒
Active DBO Server URL/IP	172.16.99.129
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	172.16.99.129_3479
Active DBO Server Password	*****
Standby DBO Server URL/IP	172.16.200.101
Standby DBO Server Port 0	3478
Standby DBO Server Port 1	
Standby DBO Server Port 2	
Standby DBO Server Port 3	
Standby DBO Server Username	172.16.200.101_3479
Standby DBO Server Password	*****
Enable SIP Forwarding	☒
Enable RTP Forwarding	☒

DBO Advanced	
Enable Bandwidth Compressed	☒

Parameter Description:

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

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

4.10 HUMAN BEHAVIOR

4.10.1 Overview

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

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

4.10.2 Basic Configuration

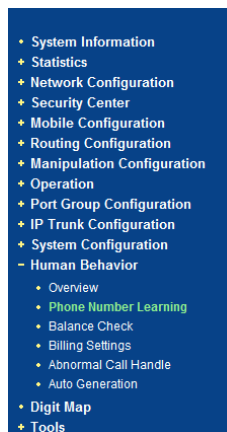
On the **Basic Configuration** interface, you can set how long an IP → Tel call or a Tel → IP call will be delayed, as well as call interval. The 'set call volume threshold function' is mainly used for anti-blocked (such as some operators launched special call testing for the detection of the VoIP equipment,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

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- 2 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)	0 s- 0 s
No Alerting Call Handle	☒ Normal Handle ☐ Hang Up ☐ Not Answer
IP to TEL Processing Timeout Handle	☐
Set Call Volume Threshold	☐
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	
Interval for switching SIM(range:0-65535)	0 Minutes
Query SIM information during initiation	☒ Enable ☐ Disable
Timeout	60 Seconds (range:30-300)

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**.



Phone Number Learning - Add Rule	
Index	0
Type	USSD
Send Text	*156#
Keywords	Your number is: Matching Test
Write Phone Number to SIM card	☐ No ☒ Yes
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 123 456 abc".

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 - Add Rule

Index	0
Type	USSD
Send Text	*156#
Keywords	Your number is: Test End
Test SMS Text	Your number is: 8618344144906
Result	Test
Write Phone Number to SIM card	☐ No ☒ Yes
Port Group	0 <all>

Phone Number Learning - Add 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
Port Group	0 <all>

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**.

Phone Number Learning - Add Rule

Index	0
Type	SMS
Dest Number	10086
Send Text	My Number
Check SMS From Number	10086
Keywords	Your number is: Matching Test
Write Phone Number to SIM card	☐ No ☒ Yes
Port Group	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 - Add Rule

Index	0
Type	SMS
Dest Number	10086
Send Text	My Number
Check SMS From Number	10086
Keywords	Your number is: Test End
Test SMS Text	Your number is: 8618344144906
Result	Test
Write Phone Number to SIM card	☐ No ☒ Yes
Port Group	0 <all>

Phone Number Learning - Add Rule

Index	0
Type	SMS
Dest Number	10086
Send Text	My Number
Check SMS From Number	10086
Keywords	Your number is: Test End
Test SMS Text	Your number is: 8618344144906
Result	Test 8618344144906
Write Phone Number to SIM card	☐ No ☒ Yes
Port Group	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 - Add Rule

Index	0
Type	Call
Dest Number	10086
Send Text	p5,3,p3,2
Check SMS From Number	
Keywords	Your number is: Matching Test
Write Phone Number to SIM card	☐ No ☒ Yes
Port Group	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 - Add 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

Write Phone Number to SIM card: ☐ No ☒ Yes

Port Group: 0 <all>

Phone Number Learning - Add 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

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.

- System Information
- Statistics
- Network Configuration
- Security Center
- Mobile Configuration
- Routing Configuration
- Manipulation Configuration
- Operation
- Port Group Configuration
- IP Trunk Configuration
- System Configuration
- Human Behavior
 - Overview
 - Phone Number Learning
 - Balance Check
 - Billing Settings
 - Abnormal Call Handle
 - Auto Generation
- Digit Map
- Tools

Overview						
	Port	Phone Number	Last Matched Balance	Calculated Balance	Remaining Total Credits	Remaining Daily Credits
☐	0		73.40	73.40	6000.00	600.00
☐	1				0.00	0.00
☐	2				6000.00	600.00
☐	3				0.00	0.00
☐	4			---	---	---
☐	5			---	---	---
☐	6			---	---	---
☐	7			---	---	---
☐	All					

Check
Check
Restore
Restore

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.

- System Information
- Statistics
- Network Configuration
- Security Center
- Mobile Configuration
- Routing Configuration
- Manipulation Configuration
- Operation
- Port Group Configuration
- IP Trunk Configuration
- System Configuration
- Human Behavior
 - Overview
 - Phone Number Learning
 - **Balance Check**
 - Billing Settings
 - Abnormal Call Handle
 - Auto Generation
- Digit Map
- Tools

Balance Check - Add Rule

Index	0	▼	
Type	USSD	▼	
Send Text	*101#		
Keywords	Your balance is		Matching Test
Digit Thousand Symbol	.		
Digit Point Symbol	.		
Port Group	0 <all>	▼	
Check Balance After SIM Card Registration	☒		
Check Balance Every	20		Minutes
	Note: "0" means disable.		
Check While Calculated Balance Is Low	3		
	Note: "0" means disable.		

About Key Words: 1. You can input multiple keywords and special symbol like "[E]", "[T]", "[*]", "[P]". Space is also available.

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 - Add Rule

Index	0	▼	
Type	USSD	▼	
Send Text	*101#		
Keywords	Your balance is		Matching Test
Digit Thousand Symbol	.		
Digit Point Symbol	.		
Port Group	0 <port0-3>	▼	
Check Balance After SIM Card Registration	☒		
Check Balance Every	20		Minutes
	Note: "0" means disable.		
Check While Calculated Balance Is Low	3		
	Note: "0" means disable.		

(next page)

Balance Check - Add Rule	
Index	0 ▼
Type	USSD ▼
Send Text	*101#
Keywords	Your balance is Test End
Test SMS Text	Your balance is 73.40\$
Result	Test 73.40
Digit Thousand Symbol	,
Digit Point Symbol	.
Port Group	0 <port0-3 ▼
Check Balance After SIM Card Registration	☒
Check Balance Every	20 Minutes
	Note: "0" means disable.
Check While Calculated Balance Is Low	3
	Note: "0" means disable.

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 - Add Rule	
Index	0 ▼
Type	SMS ▼
Dest Number	10086
Send Text	My blance
Check SMS From Number	10086
Keywords	Your balance is Matching Test
Digit Thousand Symbol	,
Digit Point Symbol	.
Port Group	0 <port0-3 ▼
Check Balance After SIM Card Registration	☒
Check Balance Every	20 Minutes Note: "0" means disable.
Check While Calculated Balance Is Low	3 Note: "0" means disable.

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 - Add Rule	
Index	0 ▼
Type	SMS ▼
Dest Number	10086
Send Text	My blance
Check SMS From Number	10086
Keywords	Your balance is Test End
Test SMS Text	Your balance is 73.40\$
Result	Test 73.40
Digit Thousand Symbol	,
Digit Point Symbol	.
Port Group	0 <port0-3 ▼
Check Balance After SIM Card Registration	☒
Check Balance Every	20 Minutes Note: "0" means disable.
Check While Calculated Balance Is Low	3 Note: "0" means disable.

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 - Add Rule	
Index	0 ▼
Type	Call ▼
Dest Number	10086
Send Text	p5,3,p3,2
Check SMS From Number	10086
Keywords	Your balance is Matching Test
Digit Thousand Symbol	,
Digit Point Symbol	.
Port Group	0 <port0-3 ▼
Check Balance After SIM Card Registration	☒
Check Balance Every	20 Minutes Note: "0" means disable.
Check While Calculated Balance Is Low	3 Note: "0" means disable.

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 - Add Rule	
Index	0
Type	Call
Dest Number	10086
Send Text	p5,3,p3,2
Check SMS From Number	10086
Keywords	Your balance is
Test SMS Text	Your balance is 73.40\$
Result	Test 73.40
Digit Thousand Symbol	,
Digit Point Symbol	.
Port Group	0 <port0>
Check Balance After SIM Card Registration	☒
Check Balance Every	20 Minutes
Check While Calculated Balance Is Low	3
	Note: "0" means disable.

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	0
Billing Unit(1-3600)	60 seconds
Rate(not more than 10000)	1 /Billing Unit
Minimum Charging Time(0-3600)	0 s
Select Port	Port Group
	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	0 ▼	
Single Call Duration	0	s
	Note: 0 means no limit, not more than 40000.	
Total Credits	300	
	Note: 0 means no limit, not more than 400000.	
Daily Credits	0	
	Note: 0 means no limit, not more than 400000.	
Daily Calls	0	
	Note: 0 means no limit, not more than 100000.	
Adjust Credits Automatically	☐ No ☒ Yes	
Low Credits Warning	☒ No ☐ Yes	
Select Port	Port Group ▼ 0 <all> ▼	

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

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

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

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

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

Call limitation = $300/1 = 300$ minutes

4.10.7 Exception Event Handling

Exception Event Handling				
Enable		☐ No ☒ Yes		
Call Event				
☐ Low ASR Less Than	12	%	Handle	SMS Alert
☐ Low ACD Less Than	0	s	Reset	☐
☐ Counts of Call Failed	0		Reset	☐
☐ Low Balance Less Than	0.00		Block SIM	☒
☐ By Gsm Code	8		Reset	☐
☐ PDD Less Than(1-30)	2	s	Reset	☐
☐ Cost Difference	0.00	- 0.00	Block	☐
USSD Event				
☐ Counts of Send Fail(1 - 100)	10		Reset	☐
USSD/SMS Monitor				
☐ SMS Number			Reset	☐
Keywords				
USSD/SMS URL Monitor				
☐ SMS Number			Access Internet	☐
Keywords				
SIM Register Fail				
☐ Register Timeout(60 - 600)	600	s	Block SIM	☐
SMS Test				
Number				
Content	SMS Test			
Resend Count(0 - 10)	0			
Send SMS For Alerting				
Number				

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	Threshold	Unit	Handle	SMS Alert
☐	Low ASR Less Than	12	%	Block SIM	☐
☐	Low ACD Less Than	0	s	Reset	☐
☐	Counts of Call Failed	0		Block	☐
☐	Low Balance Less Than	0.00		Block SIM	☒
☐	By Gsm Code	8		SMS Test	☐
☐	PDD Less Than(1-30)	2	s	Reset	☐
☐	Cost Difference	0.00 - 0.00		Block	☐

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

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

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

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

Enable	Call Event	Threshold	Unit	Handle	SMS Alert
☐	Low ACD Less Than	0	s	Reset	☐
☒	Counts of Call Failed	10		Reset	☐
☐	Low Balance Less Than	5.00		Block	☒

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.

☐ Counts of Call Failed	0	Reset
☒ Low Balance Less Than	5.00	Block
☐ 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	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

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		

By GSM Code: The GSM network side error code

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

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

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

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

Handle abnormal event

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

Reset the specified GSM module

Block the specified GSM module

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

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

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

USSD Event

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

USSD/SMS Monitor
☒ SMS Number

☐ Keywords

4.10.8 Auto generation

Human Behavior-Auto Generation	
Enable	☒
Basic Settings	
Prefix to Add:	
Digits to be Deleted:	
Auto Call:	
Called By Other Ports	☒
Number Length	8 digits
Call Out	☐
Min Call Duration:	60 seconds
Max Call Duration:	80 seconds
Auto Send SMS:	
Random Content-1:	Sorry,I will call you later.
Random Content-2:	I am busy now.
Random Content-3:	I can not come to the phone.
Random Content-4:	
Random Content-5:	
Random Content-6:	
Random Content-7:	
Random Content-8:	
Random Content-9:	
Random Content-10:	
Auto Internet access:	☐
Conditions Settings	
By Device Online Time:	☐
By Total Call Durations:	☐
By Consecutive Calls:	☐

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

Why need Auto Generation?

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

Basic Settings:

Auto Call:

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

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

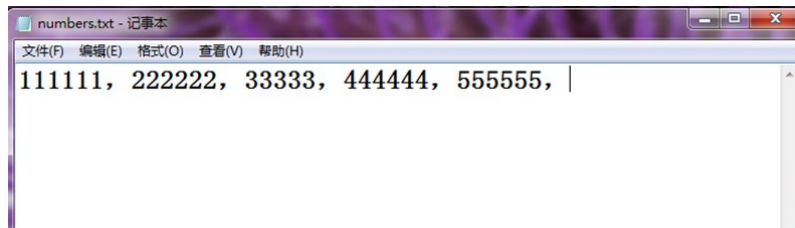
SIM Card Number section.

Call Out: We can set to call fixed numbers

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

How to make the txt file?

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



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

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

Auto Send SMS: Auto SMS between the same device ports

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

Conditions Settings	
By Device Online Time:	☒
Min Interval:	30 minutes
Max Interval:	120 minutes
By Total Call Durations:	☒
Call Duration:	60 minutes
By Consecutive Calls:	☒
Consecutive Calls:	10

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	Value/Status
Local Syslog	☒ Enable
Server Address	172.16.222.222
Server Port	514
Syslog Level	DEBUG ▼
Signal Log	☒ Enable
Media Log	☐ Enable
System Log	☐ Enable
Management Log	☐ Enable
Send CDR	☒ Enable
Server Syslog	☐ 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 include following traces which defined in system by default

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

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

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

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

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

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

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

Server Syslog:

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

4.11.2 Filelog

Log Type	Status
Filelog	☒ Enable
Filelog Level	DEBUG
Signal Log	☒ Enable
Media Log	☒ Enable
System Log	☒ Enable
Management Log	☐ Enable

Download

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	
Summary	☒ Enable
Record Cycle	60 Minutes
Write Flash Timer	☐ Enable
Download	

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	1,2,3 Up to 3 ports,e.g."1,2,15"
Record in summary	☒ Enable
Record in media log	☒ 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	www.google.com
Number of Ping(1-100)	4
Ping Packet Size(56-1024 bytes)	56
Start Stop	

Information	

4.11.6 Tracert Test

You can check the routes of the tracert destination.

Tracert Test	
Tracert Destination	
Max Hops of Tracert(1-255)	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:

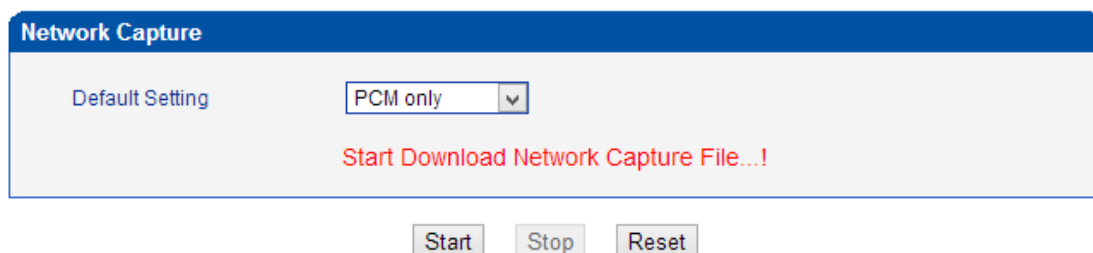


► Getting start to PCM capture

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



- ◆ 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:

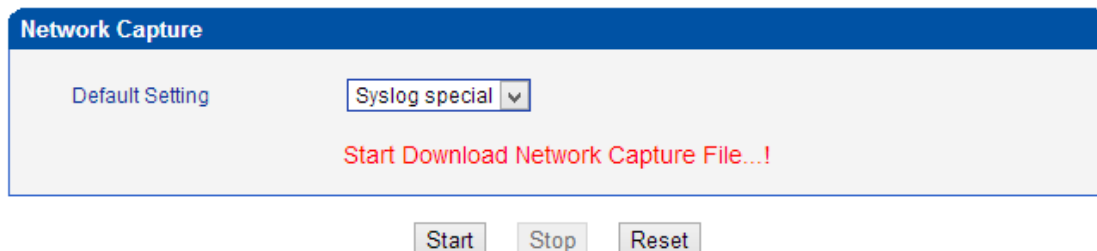
No.	Time	Source	Destination	Protocol	Length	Info	
1	0.000000	Motorola_1c:1d:1e	Cimsys_33:44:55	CSM_ENCABS	104	--> 0x0021	Ch: 0xFFFF, Seq: 8 (From Host)
2	0.000131	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
3	0.000245	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCABS	44	--> 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	1.329890	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	--> 0x8084	Ch: 0x0003, Seq: 6 (From Host)
23	1.343627	Cimsys_33:44:55	Motorola_1c:1d:1e	Ethernet	20	Ethernet II[Malformed Packet]	
24	1.343725	Cimsys_33:44:55	Motorola_1c:1d:1e	CSM_ENCABS	30	--> 0x8084	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 save as pcap format so that it can be opened in some of capture software like Wireshark, Ethereal software etc.

► To enable syslog capture

- ◆ Select Syslog special only on Network Capture page



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

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

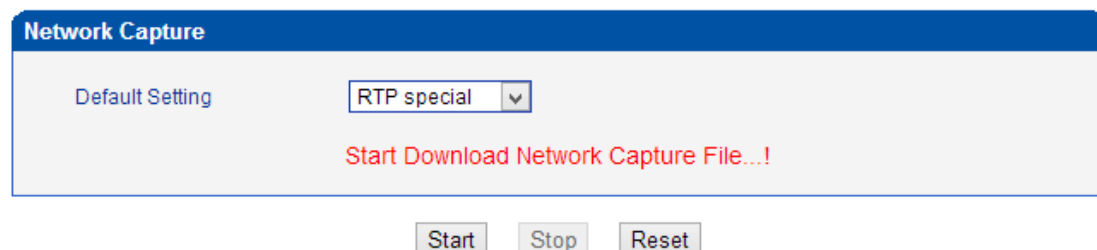
No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000	172.16.222.22	1.1.1.1	Syslog	172	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 0> [DEBUG] ----> to 172.16.222.22/5060 crypt:FALSE Phone
2	0.000344	172.16.222.22	1.1.1.1	Syslog	520	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 1> [DEBUG] OPTIONS sip:heartbeat@172.16.222.22 SIP/2.0\r\n
3	0.013432	172.16.222.22	1.1.1.1	Syslog	595	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 2> [DEBUG] <----- message from 172.16.222.22/5060, crypt
4	0.013750	172.16.222.22	1.1.1.1	Syslog	176	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 3> [DEBUG] <----- from 172.16.222.22/5060, crypt:FALSE, Ph
5	0.014036	172.16.222.22	1.1.1.1	Syslog	520	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 4> [DEBUG] OPTIONS sip:heartbeat@172.16.222.22 SIP/2.0\r\n
6	0.014512	172.16.222.22	1.1.1.1	Syslog	172	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 5> [DEBUG] ----> to 172.16.222.22/5060 crypt:FALSE Phone
7	0.014806	172.16.222.22	1.1.1.1	Syslog	587	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 6> [DEBUG] SIP/2.0 200 OK\r\nVia: SIP/2.0/UDP 172.16.222.
8	0.028396	172.16.222.22	1.1.1.1	Syslog	662	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 7> [DEBUG] <----- message from 172.16.222.22/5060, crypt
9	0.028759	172.16.222.22	1.1.1.1	Syslog	176	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 8> [DEBUG] <----- from 172.16.222.22/5060, crypt:FALSE, Ph
10	0.029052	172.16.222.22	1.1.1.1	Syslog	587	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 9> [DEBUG] SIP/2.0 200 OK\r\nVia: SIP/2.0/UDP 172.16.222.
11	0.030017	172.16.222.22	1.1.1.1	Syslog	233	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 10> [DEBUG] sip->app: msgtype:ST_SIP_SERVER_CONN \r\n ca
12	0.331167	172.16.222.22	1.1.1.1	Syslog	983	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 11> [DEBUG] <----- message from 172.16.222.22/5060, cryp
13	0.331498	172.16.222.22	1.1.1.1	Syslog	177	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 12> [DEBUG] <----- from 172.16.222.22/5060, crypt:FALSE, PF
14	0.331959	172.16.222.22	1.1.1.1	Syslog	907	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 13> [DEBUG] INVITE sip:100868172.16.222.22:5060 SIP/2.0\r\n
15	0.332207	172.16.222.22	1.1.1.1	Syslog	122	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 14> [DEBUG] get route entry 31\r\n
16	0.332584	172.16.222.22	1.1.1.1	Syslog	111	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 15> [DEBUG] lport:3\r\n
17	0.332848	172.16.222.22	1.1.1.1	Syslog	124	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 16> [DEBUG] get route, to port:3\r\n
18	0.333315	172.16.222.22	1.1.1.1	Syslog	526	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 17> [DEBUG] sip->app: localindex:69, msgtype:SIP_CALL_IN
19	0.333603	172.16.222.22	1.1.1.1	Syslog	173	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 18> [DEBUG] ----> to 172.16.222.22/5060 crypt:FALSE Phone
20	0.333877	172.16.222.22	1.1.1.1	Syslog	386	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 19> [DEBUG] SIP/2.0 100 Trying\r\nVia: SIP/2.0/UDP 172.16.
21	0.346687	172.16.222.22	1.1.1.1	Syslog	131	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 20> [DEBUG] RTP: alg:0, pkt:20, band:-1\r\n
22	0.347453	172.16.222.22	1.1.1.1	Syslog	120	USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 21> [DEBUG] dial tick:102433\r\n
23	7.232839	172.16.222.22	1.1.1.1	Syslog	533	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 22> [DEBUG] <----- message from 172.16.222.22/5060, cryp
24	7.233513	172.16.222.22	1.1.1.1	Syslog	177	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 23> [DEBUG] <----- from 172.16.222.22/5060, crypt:FALSE, PF
25	7.233959	172.16.222.22	1.1.1.1	Syslog	457	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 24> [DEBUG] CANCEL sip:100868172.16.222.22:5060 SIP/2.0\r\n
26	7.234596	172.16.222.22	1.1.1.1	Syslog	287	USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 25> [DEBUG] sip->app: localindex:69, msgtype:SIP_CALL_BYE

► Getting start to RTP capture

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

► To enable RTP capture:

- ◆ Select RTP special on Network Capture page



- ◆ Click Start to enable RTP capture

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

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

No.	Time	Source	Destination	Protocol	Length	Info
176	7.020000	172.16.221.228	116.204.105.50	SIP	565	Request: REGISTER sip:116.204.105.50
178	7.030000	116.204.105.50	172.16.221.228	SIP	411	Status: 200 OK (1 bindings)
244	11.610000	172.16.221.228	58.56.64.101	SIP/SDP	814	Request: INVITE sip:201@58.56.64.101
248	11.710000	58.56.64.101	172.16.221.228	SIP	480	Status: 100 Trying
249	11.710000	58.56.64.101	172.16.221.228	SIP/SDP	733	Status: 183 Session Progress
250	11.710000	58.56.64.101	172.16.221.228	SIP/SDP	719	Status: 200 OK
252	11.720000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
253	11.720000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
254	11.720000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, Seq=1000, Time=160, Mark
255	11.720000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
256	11.730000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
257	11.730000	172.16.221.228	58.56.64.101	RTP	66	Unknown RTP version 1
258	11.740000	172.16.221.228	58.56.64.101	SIP	434	Request: ACK sip:201@58.56.64.101:5060
259	11.740000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, Seq=1001, Time=320
261	11.770000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, Seq=1002, Time=480
263	11.780000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, Seq=1003, Time=640
264	11.810000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, Seq=1004, Time=800
265	11.830000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, Seq=1005, Time=960
266	11.840000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, Seq=1006, Time=1120
267	11.870000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, Seq=1007, Time=1280
268	11.890000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, Seq=1008, Time=1440
270	11.900000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, 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=0x497E6015, Seq=1010, Time=1760
274	11.940000	58.56.64.101	172.16.221.228	RTP	74	PT=ITU-T G.729, SSRC=0x497E6015, 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=0x497E6015, 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

Network Capture

Default Setting

DSP Only
▼

Start

Stop

Reset

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

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

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

► Configurable capture options

► Getting start to custom capture

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

Network Capture

Default Setting

Custom

Network Interface

☒ LAN
☐ DSP

Srouce Host

Destination Host

Select Port

None

Protocol(s)

☒ TCP
☐ UDP
☐ RTP
☐ RTCP
☐ ICMP
☐ ARP

Start Download Network Capture File...!

Start Stop Reset

4.11.8 Voice Loopback Test

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

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

Voice stream patch on gateway:



► DSP TDM Test

DSP TDM Test is the loopback of GSM side.

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

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

► To start DSP TDM Test:

- ◆ Make a call test through gateway, the call can be 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
- ◆ Hangup

► To start DSP IP Test:

DSP IP Test is the loopback of VoIP side.

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

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

► To start DSP IP Test:

- ◆ Make a call test through gateway, the call can be 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

4.12 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 use 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 actually in the unit cannot be used for a restore operation on the unit.

4.12.4 Management Parameter

Management Parameter	
NTP Parameter	
NTP Enable	☒ Yes ☐ No
Primary NTP Server Address	pool.ntp.org
Primary NTP Server Port	123
Secondary NTP Server Address	
Secondary NTP Server Port	123
Check Interval	180 s
Time Zone	GMT+8:00 (Beijing, Singapore, Taipei, Hong Kong) ▼
WEB Parameter	
WEB Port	80
Telnet Parameter	
Telnet Port	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	☒
Server URL/IP	server02.dmclid.com
Server Port	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.dmcl.com:3000

4.12.6 Email Account Setting

Email Setting	
Email Account	
E-mail	
Username	
Password	Show Password
Outgoing(SMTP)	
Server	
Port	
TLS Enable	☐
Incoming	
Protocol	IMAP ▼
Server	
Port	993
TLS Enable	☒

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	
New Web Username	
New Web Password	
Confirm Web Password	
Telnet Configuration	
Old Telnet Username	admin
Old Telnet Password	
New Telnet Username	
New Telnet Password	
Confirm Telnet Password	

4.12.8 Access control

Access Control

New Password:

Show Password

Confirm Password:

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	☐ Disable ☒ Enable
Cloud Server	☐ Disable ☒ Enable
Routing	☐ Disable ☒ Enable
SIP Trunk and SIP trunk group	☐ Disable ☒ Enable
SIP Configuration	☐ Disable ☒ Enable
Port and Port group	☐ Disable ☒ Enable
Config Restore and Backup	☐ Disable ☒ Enable
Manage(NTP WEB Telnet and Username&Password)	☐ Disable ☒ Enable
Software Upload	☐ Disable ☒ Enable
Factory Reset	☐ Disable ☒ 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

Click this button to reset factory default settings

Notes:The device must restart to take effect.

Apply

4.12.10 Auto Restart

Configure auto restart at pre-defined HH/MM

Auto Restart

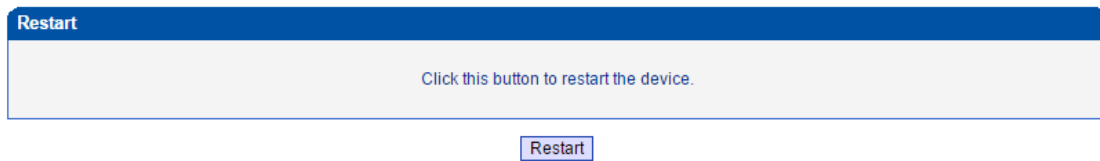
Auto Restart Enable

☒ Yes
 ☐ No

Restart Time

:

4.12.11 Restart



5 Troubleshooting and Command Line

5.1 LOGIN UC2000 & GENERAL KNOWLEDGE OF UC2000 COMMAND

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

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

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

```
C:\Users\Administrator>telnet 172.16.101.142
```

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

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

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

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

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

5.2. COMMANDS IN "ROS#" MODE

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

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

5.2.1 SUMMARIZE OF COMMANDS IN "ROS#" MODE

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

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

5.2.2 GENERAL PURPOSE COMMANDS IN "ROS#" MODE

► Show IP address (show int)

```
ROS#
ROS#sho int

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

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

ROS#_
```

► Show Time (show clock)

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

► Show version (show version)

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

► Show sip Information (show sip config)

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

► Show memory status (show memory detail)

```

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

```

► Show SIP port status (show sip all)

```

ROS#
ROS#sho sip all

```

Index	UserId	State	Expire(s)	RemainTime
0	30	OK	1800	976
1	31	OK	1800	976
2	33	OK	1800	976

```

ROS#

```

► Show Current calls (sh ecc call)

```

ROS#
ROS#sho ecc call

```

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

```

ROS#

```

► Show RTP session (sho rtp se)

```

ROS#
ROS#sho rtp se
RTP Information:
  RTP System TimeStamp 1586900(ms)
  MBUF Waiting for Playing 0, MBUF Discarded 0
EIA RTP Session List:
  PT-Payload Type, PP-Packet Period, PL-Packet Length,
  SP-Sample Period, SL-Sample Length, P/S-PP/SP, LR-NetLostRate, RLR-RealLostRate

```

RTPNo	Mode	PT	Send/ToDsp	LR/RLR	Local IP: Port	Peer IP: Port	PP	PL	SP	SL	P/S	P2P	silence
0	STD	18	9250/9205	0/0	LocalHost: 8000	66.152.170.74:10562	20	20	20	20	1	NO	0
2	STD	18	6499/6227	0/0	LocalHost: 8004	66.152.170.74:10658	20	20	20	20	1	NO	3
4	STD	18	56225/56145	0/0	LocalHost: 8008	66.152.170.74: 9558	20	20	20	20	1	NO	0
8	STD	18	13300/13201	0/0	LocalHost: 8016	66.152.170.74:10498	20	20	20	20	1	NO	1
10	STD	4	7253/14451	0/0	LocalHost: 8020	64.15.152.90: 6042	60	48	60	48	1	NO	1
14	STD	18	11745/11599	0/0	LocalHost: 8028	66.152.170.74:10522	20	20	20	20	1	NO	0
16	STD	18	248/210	0/0	LocalHost: 8032	66.152.170.74:10766	20	20	20	20	1	NO	0
18	STD	18	31800/31747	0/0	LocalHost: 8036	66.152.170.74:10186	20	20	20	20	1	NO	1
20	STD	18	10499/10322	0/0	LocalHost: 8040	66.152.170.74:10554	20	20	20	20	1	NO	3
24	STD	18	30028/29901	0/0	LocalHost: 8048	66.152.170.74:10198	20	20	20	20	1	NO	1
26	STD	18	29614/6065	0/0	LocalHost: 8052	64.15.152.90:11854	20	20	20	20	1	NO	1
28	STD	18	71018/70690	0/0	LocalHost: 8056	66.152.170.74: 9138	20	20	20	20	1	NO	1

ROS#

► Show ASR/ACD statistics (show ecc state)

```

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

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

```

ROS#

5.3 COMMANDS IN "CONFIG" MODE

5.3.1 SUMMARIZE OF COMMANDS IN "CONFIG" MODE

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

```

ROS#
ROS#^config
ROS<config>#
ROS<config>#

```

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

```

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

```

5.3.2 GENERAL PURPOSE COMMANDS IN "CONFIG" MODE

► **Set time (clock set)**

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

► **Save the configuration (save)**

```
ROS<config>#  
ROS<config>#save  
ROS<config>#
```

► **Restart device (reset eia)**

```
ROS<config>#  
ROS<config>#reset  
Are you sure to reset? <y/n>:y  
ROS<config>#
```

► **Enable debug**

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

```
ROS<config>#  
ROS<config>#deb port 0  
Succ! Debug PortNo:0  
  
ROS<config>#
```

To enable all ports debug, with the command "deb port all"

```
ROS<config>#deb port all  
Debug All!?.  
  
ROS<config>#
```

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

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

```
ROS<config>#  
ROS<config>#deb sip msg all  
ROS<config>#
```

5.4 HOW TO TRACE SIP LOGS

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

```
Welcome to Command Shell!  
  
Username:admin  
  
Password:*****  
  
ROS>en  
  
ROS#  
  
ROS#^config  
  
ROS(config)#deb sip msg all  
  
ROS(config)#ex  
  
ROS#  
  
ROS#^ada  
  
ROS(ada)#ADA CONNECTED ...,WELCOME!  
  
ROS(ada)#  
  
ROS(ada)#turnon 71  
  
Disable sip trace:  
  
ROS(ada)#turnoff 71
```

5.5 HOW TO TRACE ECC LOGS (CALL DETAILS)

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

```
ROS#
ROS#^config
ROS(config)#deb port all
    Debug All!.
//enable trace on all port
ROS(config)#
ROS(config)#deb port 0
Succ! Debug PortNo:0
// enable trace port 0
ROS(config)#
ROS(config)#no deb port all
ROS(config)#
ROS(config)#ex
ROS#^ada
ROS(ada)#ADA CONNECTED ...,WELCOME!
ROS(ada)#turnon 84
Disable trace:
ROS(ada)#turnoff 84
```

5.6 HOW TO TRACE MODULAR LOGS

```
Welcome to Command Shell!
Username:admin
Password:*****
ROS>en
ROS#^ada
```

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

ROS(ada)#cmd 53 19 0 0 1

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

ROS(ada)#cmd 53 19 0 0 0

//disable modular trace

6 Glossary

GSM: Global System for Mobile Communications

CDMA: Code Division Multiple Access

FMC: Fixed Mobile Convergence

SIP: Session Initiation Protocol

MGCP: Media Gateway Control Protocol

DTMF: Dual Tone Multi Frequency

USSD: Unstructured Supplementary Service Data

PSTN: Public Switched Telephone Network

STUN: Simple Traversal of UDP over NAT

IVR: Interactive Voice Response

IMSI: International Mobile Subscriber Identification Number

IMEI: International Mobile Equipment Identity

DMZ: Demilitarized Zone

API: Application programming Interface

BCCH: Broadcast Control Channel

LAC: Location Area Code

CID: Cell ID

BTS: Base Transceiver Station

DTMF: Dual-Tone Multifrequency

IVR: Interactive Voice Response

NAT: Network Address Translation

RTP: Real-time Transport Protocol

VoIP: Voice over IP

Warning:

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation. Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

NOTE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation.

If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit

different

from that to which the receiver is connected.

-- Consult the dealer or an experienced radio/TV technician for help.

NOTE: This device and its antenna(s) must not be co-located or operation in conjunction with any other antenna or transmitter.

RF Exposure Statement

To maintain compliance with FCC's RF Exposure guidelines, This equipment should be installed and operated with minimum distance of 20cm the radiator your body.

This device and its antenna(s) must not be co-located or operation in conjunction with any other antenna or transmitter

Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Warning:

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation. changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

NOTE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

NOTE: This device and its antenna(s) must not be co-located or operation in conjunction with any other antenna or transmitter.