



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

User Manual



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| | | | | |

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1 Product Description

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

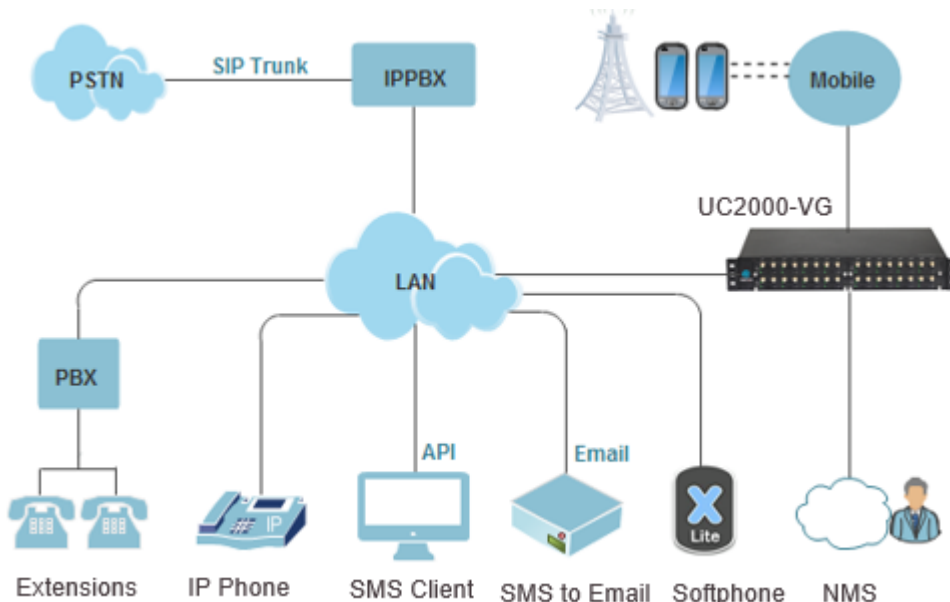
1.1 OVERVIEW

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

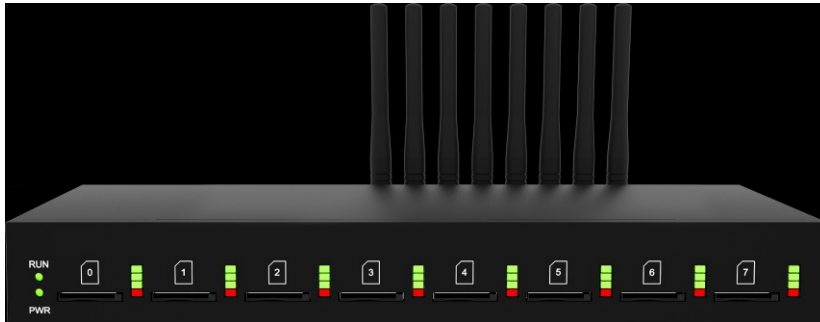


Table 1-3-1 Description of Front view





| 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. <ul style="list-style-type: none"> ◆ Restore IP and Password: hold RST button 3~5 seconds, RUN LED being ON during this time ◆ Restore factory setting: Hold RST button 7 seconds, RUN LED being blink |




1.3.2 Product Appearance of UC2000-VF

Front View

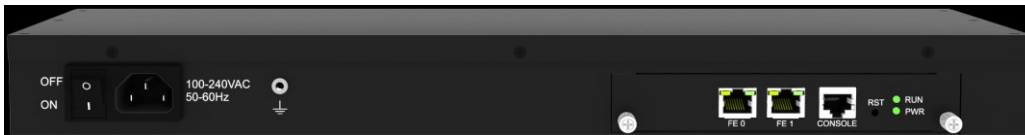


Indicators and connectors




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

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

Back view



Indicators and connectors

| Indicators | Name | Status | Description |
|---|------------------|-----------------|---|
|  | Power switch | - | Power on or power off the device |
|  | Power connect | - | AC Input 110-240V |
| FE0-FE1 | Network | - | Default IP is 192.168.11.1 |
|  | Console | - | RS232 standard, band rate 115200bps |
| RST | RST | - | Reset button to restore default IP and password or restore factory setting. <ul style="list-style-type: none"> Restore IP and Password: hold RST button 3~5 seconds, RUN LED being ON during this time Restore factory setting: Hold RST button 7 seconds, RUN LED being blink |
| PWR | Power indicator | OFF | No power |
| | | ON | Power on |
| RUN | System indicator | Blinking (0.5S) | Device is running normally |

| | | | |
|--|--|-----|--------------------------|
| | | ON | Device is booting up |
| | | OFF | Device is not booting up |

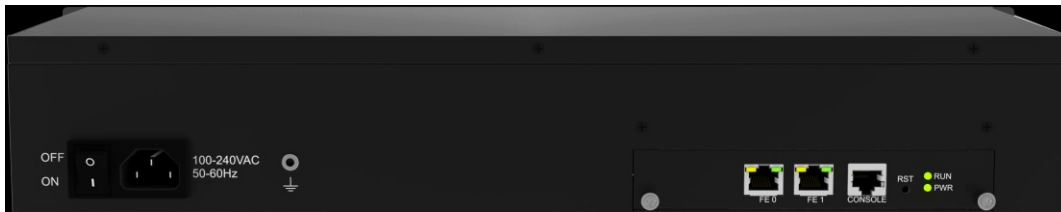
1.3.3 Product appearance of UC2000-VG

Front view




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

Back View



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

| | | | |
|---|------------------|-----------------|---|
|  | Console | - | RS232 standard, band rate 115200bps |
| RST | RST | - | <p>Reset button to restore default IP and password or restore factory setting.</p> <ul style="list-style-type: none"> ◆ Restore IP and Password: hold RST button 3~5 seconds, RUN LED being ON during this time <p>Restore factory setting: Hold RST button 7 seconds, RUN LED being blink</p> |
| PWR | Power indicator | OFF | No power |
| | | ON | Power on |
| RUN | System indicator | Blinking (0.5S) | Device is running normally |
| | | ON | Device is booting up |
| | | OFF | Device is not booting up |

1.4 FUNCTIONS AND FEATURES

1.4.1 Protocols

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

1.4.2 System Function

- PLC: Packet loss concealment
- VAD: Voice activity detection

- CNG: Comfort Noise Generation
- Local/Remote SIM card work mode
- Adjustable gain of port
- DTMF adjustment
- Balance Check
- Lock/unlock SIM/UIM
- Mobile number display rejection
- Sending/receiving SMS
- Customize IVR Recording
- White and black list
- One number access
- Open API for SMS, support USSD
- Echo Cancellation (with ITU-T G.168/165 standard)
- Automatic negotiate network
- Hotline
- BCCH(Support on GSM Gateway only)

1.4.3 Industrial Standards Supported

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

1.4.4 General Hardware Specification

- Power Supply

UC2000-VE:

Input: 100-240V, 50-60Hz

Output: DC12V 4.0A

UC2000-VF/G:

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

- Temperature(Operation): 0 °C ~ 45 °C
(Storage): -20 °C ~80 °C
- Operation Humidity: 10%-90% No Condensation
- Dimension(W/D/H): 250*156*32.5mm
- Weight: 1.069kg
- Package Weight: 2.05kg

2 Quick Installation

2.1 ATTENTIONS BEFORE INSTALLATION

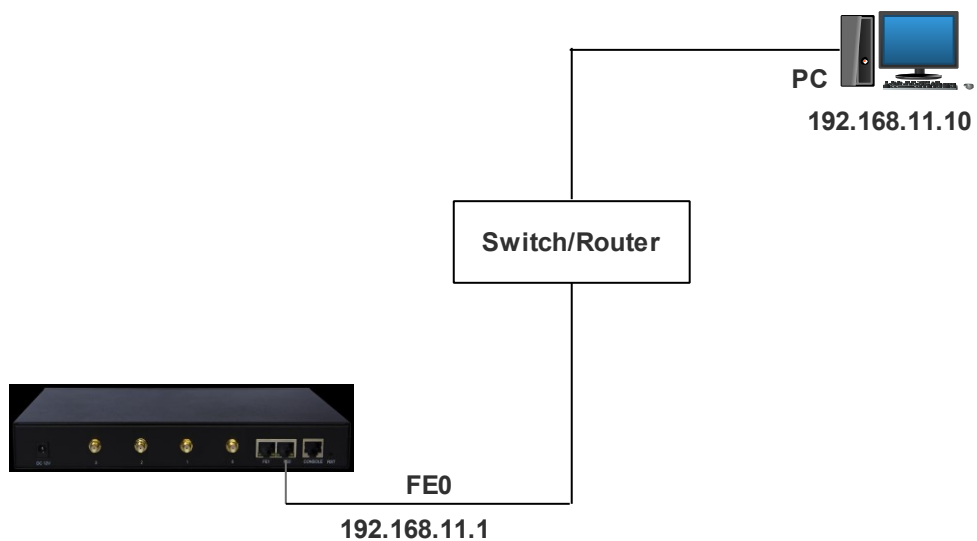
Please pay attention to the following before you install UC2000-VE/F/G include:

- 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

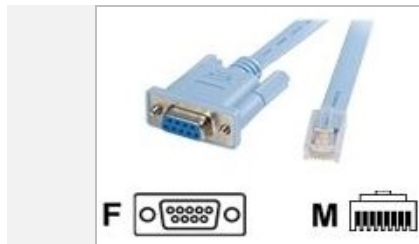
Press RST button for about 7 seconds, then gateway will be rebooted and restored to factory setting.

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);

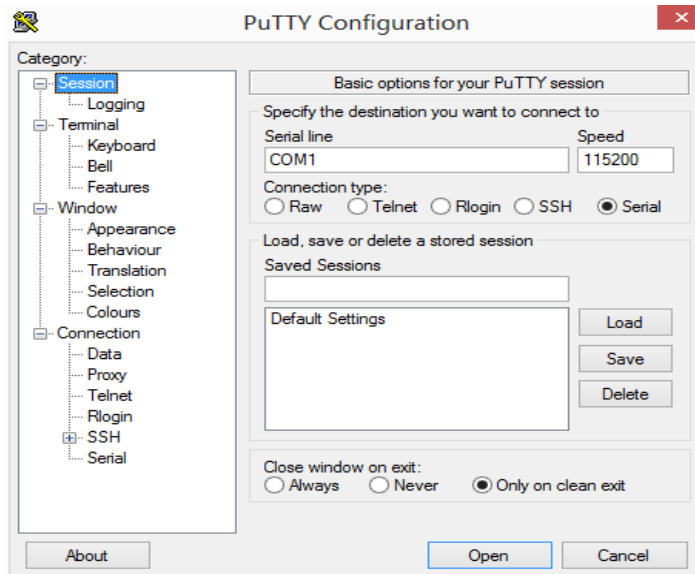


Step 2: Connect the F port of the serial cable with the COM port of PC. If the PC does not have a COM port, please use a USB-to-COM converting tool to connect the serial cable with the PC.

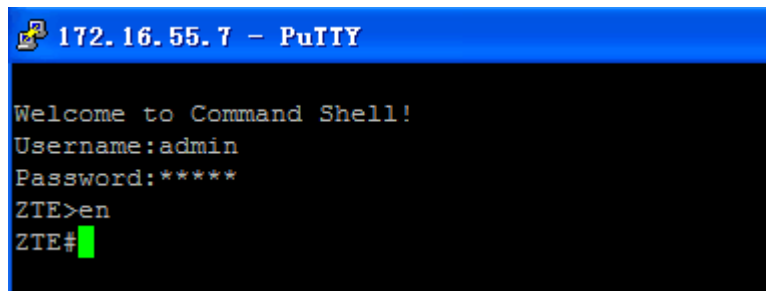
Step 3: Connect the M port of the serial cable with the console port of the device.

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.

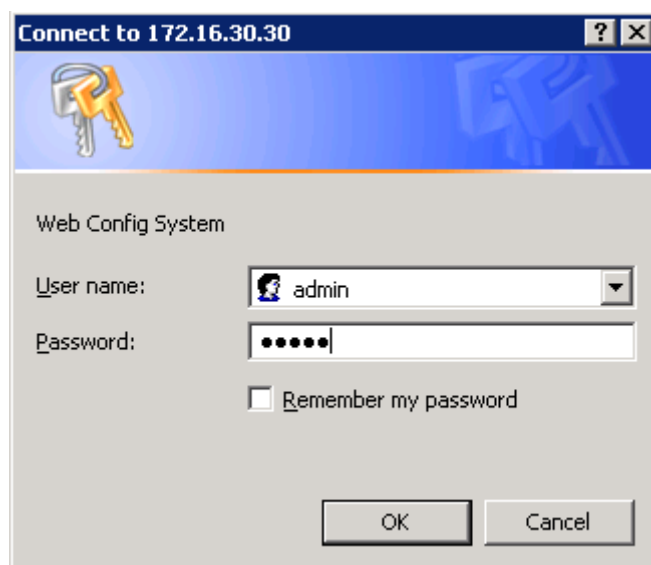
The configuration introduction also suitable for following models:

- ▶ 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: <ul style="list-style-type: none"> ▶ 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 <ul style="list-style-type: none"> • Device Model: Model name of the device • Package version: 02231220 2016-12-16 17:44:40 official, 02231220 is the version number • Software version: 02231220 2016-12-16 17:36:05, 02231220 is the version number • Web version: the version number of web system. The web version must match with software • Userboard 0 Version: the firmware version of userboard slot 0 • Userboard License ID: Contact with support when it display as Invalid • Hardware version/DSP version/ SIMbox version |

4.3.2 Mobile Information

Figure 4.3-2 Mobile Information

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

Table 4.3-2 Mobile Information

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

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

4.3.3 SIP Information

Figure 4-3-3 SIP Information

| SIP Information | | | | | | | |
|-----------------|-------------|-----------------|--------|------|-------------|-----------------|--------|
| Port | SIP User ID | Register Status | 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 |

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 | 0 | 2 | 12 | 0 |
| 1 | 28 | 1036 | 6 | 15 | 0 | 0 | 0 | 0 | 0 | 0 | 4 | 3 | 0 |
| 2 | 0 | 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 | 0 |
| 4 | 0 | 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 | 0 |
| 6 | 0 | 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 | 0 |

IP to GSM Call History

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

4.4.5 CDR Report

Figure 4-4-5 CDR Report

| CDR Report | | | | | | | | | | | | |
|--|---------------------|---------------------|--|------------|-------------|--|-------------|----------|---|---------------|-----------|--|
| Enable CDR <input type="radio"/> No <input checked="" type="radio"/> Yes | | | Save CDR <input checked="" type="radio"/> No <input type="radio"/> Yes | | | <input type="button" value="save"/> | | | | | | |
| Start Date: 2013 Year 6 Month 19 Day | | | Select Port: A11 | | | Call Direction: ALL | | | Destination: | | | |
| End Date: 2013 Year 6 Month 19 Day | | | Source: | | | Rtp Loss Rate: % to % | | | | | | |
| Min Duration: s | | | Max Duration: s | | | <input type="button" value="Refresh"/> | | | <input type="button" value="Delete the CDRs in this Report"/> | | | |
| CDR Export <input type="button" value="Export"/> | | | | | | | | | | | | |
| Port | Start Date | Answer Date | Call Direction | Source | Destination | Status | Duration(s) | Rtp Send | Rtp rcv | 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 rcv | 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/rcv/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 | SetIMEI | 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 | |
| <input type="radio"/> Obtain IP address automatically | |
| <input checked="" type="radio"/> Use the following IP address | |
| IP Address | 172.16.222.22 |
| Subnet Mask | 255.255.0.0 |
| Default Gateway | 172.16.1.5 |
| <input type="radio"/> PPPoE | |
| Account | |
| Password | |
| Service Name | |
| MTU | 1400 |
| DNS Server | |
| <input type="radio"/> Obtain DNS server address automatically | |
| <input checked="" type="radio"/> 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

Add ARP

IP Address

MAC Address

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

Click *Search All* to check ARP buffer.

ARP

Type static dynamic

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

4.5.3 VPN Parameter

Figure 4-5-3 VPN Parameter

| VPN Parameter | |
|---------------|-------------------------------------|
| VPN Enable | <input checked="" type="checkbox"/> |
| Server | us1.suvpn.com |
| Account | gary@dinstar.com |
| Password | ***** |
| Domain | us1.suvpn.com |
| Use MPPE | off |

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.

Access Rules - Add

| | | | |
|----------------|---|--|---------------|
| Index | 0 | | 255.255.255.0 |
| Action | Drop | | |
| Source IP | any | | |
| Protocol | TCP | | |
| Source Port | 0 - 65535 | | |
| Dest Port | 0 - 65535 | | |
| Description | | | |
| Enable/Disable | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | | |

4.7 MOBILE CONFIGURATION

4.7.1 Basic Configuration

Basic Configuration

Basic Configuration

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

SIM Mode

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

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

Introduction to API

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

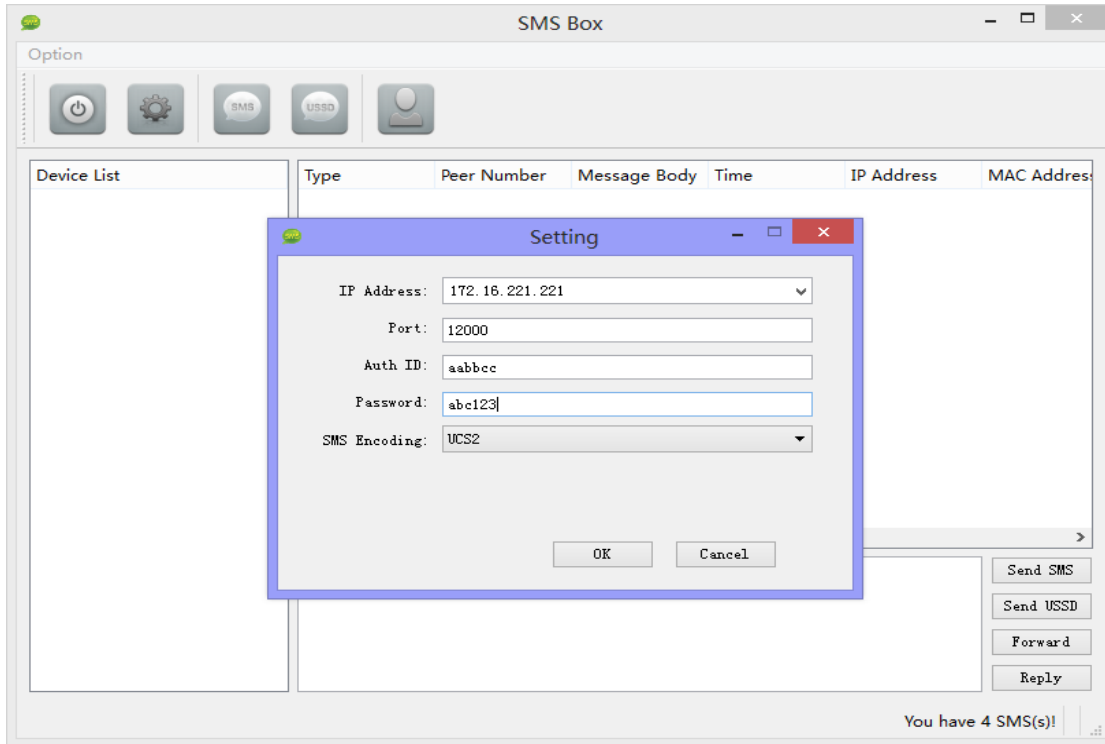
Configuration between SMS box and gateway

Configure API parameters on gateway

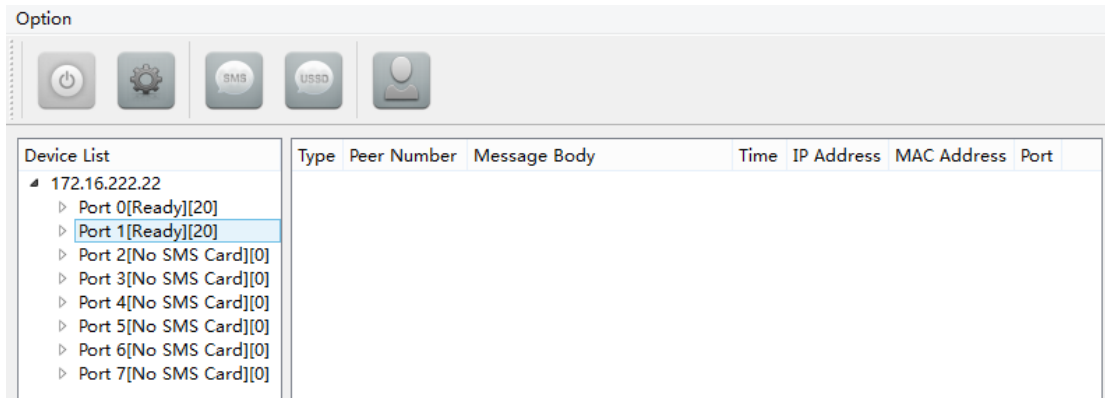
| | |
|--------------------|---|
| Remote API Enable | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| API Server Address | <input type="text" value="172.16.221.221"/> |
| API Server Port | <input type="text" value="12000"/> |
| API User ID | <input type="text" value="aabbcc"/> |
|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

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

Configure SMS box



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



4.7.2 Mobile Configuration

| Port | CLIR | Detect Reverse Polarity | Tx Gain/dB | Rx Gain/dB | APN | Band Type | Net Work Mode | SMSC | Reset | Block /Open | Power On/Off |
|------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|---------------|-------------------------------------|-------------------------------------|--------------------------------------|--|-----------------------------------|
| <input type="checkbox"/> 0 | No | Yes | 3 | 7 | | Default(Auto) | Default(Auto) | | Reset | Block | OFF |
| <input type="checkbox"/> 1 | No | Yes | 3 | 7 | | Default(Auto) | Default(Auto) | | Reset | Block | OFF |
| <input type="checkbox"/> 2 | No | Yes | 3 | 7 | | Default(Auto) | Default(Auto) | | Reset | Block | OFF |
| <input type="checkbox"/> 3 | No | Yes | 3 | 7 | | Default(Auto) | Default(Auto) | | Reset | Block | OFF |
| <input type="checkbox"/> 4 | No | Yes | 3 | 7 | | Default(Auto) | Default(Auto) | | Reset | Block | OFF |
| <input type="checkbox"/> 5 | No | Yes | 3 | 7 | | Default(Auto) | Default(Auto) | | Reset | Block | OFF |
| <input type="checkbox"/> 6 | No | Yes | 3 | 7 | | Default(Auto) | Default(Auto) | | Reset | Block | OFF |
| <input type="checkbox"/> 7 | No | Yes | 3 | 7 | | Default(Auto) | Default(Auto) | | Reset | Block | OFF |
| <input type="checkbox"/> All | <input type="button" value="Copy"/> | <input type="button" value="Copy"/> | <input type="button" value="Copy"/> | <input type="button" value="Copy"/> | <input type="button" value="Copy"/> | | <input type="button" value="Copy"/> | <input type="button" value="Copy"/> | <input type="button" value="Reset"/> | <input type="button" value="Unblock"/> | <input type="button" value="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

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

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. |

| | |
|-------------|-------------------------------------|
| Select Port | Selects the GSM/CDMA channel number |
|-------------|-------------------------------------|

4.7.5 IMEI

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: to change the IMEI code for specific port/ports

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

IMEI Auto Set

Enable IMEI Auto Set No Yes

| | | | | |
|----------|---------------------------------------|---------------------------------------|---------------------------------------|---------------------------------------|
| IMEI TAC | <input type="text" value="86008501"/> | <input type="text" value="99000100"/> | <input type="text" value="86000300"/> | <input type="text" value="86553802"/> |
| | <input type="text" value="86011600"/> | <input type="text" value="86014300"/> | <input type="text" value="86275202"/> | <input type="text" value="86121302"/> |

IMEI Serial No Range

Min IMEI Serial No

Max IMEI Serial No

Policy

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

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

Change Sim

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

4.7.6 Operator

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

Click Search button while there is SIM card in that port, after a while, you will see Operator codes list under Operator List 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 |
| <input checked="" type="checkbox"/> | <input type="text" value="460038"/> | <input type="text" value="Fixed ▼"/> | <input type="text" value="46003"/> |
| <input type="checkbox"/> | <input type="text"/> | <input type="text" value="Auto ▼"/> | <input type="text"/> |
| <input type="checkbox"/> | <input type="text"/> | <input type="text" value="Auto ▼"/> | <input type="text"/> |
| <input type="checkbox"/> | <input type="text"/> | <input type="text" value="Auto ▼"/> | <input type="text"/> |
| <input type="checkbox"/> | <input type="text"/> | <input type="text" value="Auto ▼"/> | <input type="text"/> |
| <input type="checkbox"/> | <input type="text"/> | <input type="text" value="Auto ▼"/> | <input type="text"/> |
| <input type="checkbox"/> | <input type="text"/> | <input type="text" value="Auto ▼"/> | <input type="text"/> |
| <input type="checkbox"/> | <input type="text"/> | <input type="text" value="Auto ▼"/> | <input type="text"/> |

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

4.7.8 BCCH

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

The screenshot shows the 'BCCH' configuration page. At the top, there is a blue header with the text 'BCCH'. Below the header, there are several configuration options: 'Select Port' with a dropdown menu showing 'Port 0', 'BCCH Mode' with a dropdown menu showing 'Default', and 'Apply To All Ports' with radio buttons for 'No' (selected) and 'Yes'. Below these options is a table with columns: Index, MCC, MNC, LAC, CID, BCCH, and Receive Level. Under the table, there is a 'Refresh Interval' field set to '5' with a unit 's', and two buttons: 'Auto Refresh' and 'Stop Refresh'. At the bottom, there are three buttons: 'Save', 'Refresh', and 'Back'.

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

This screenshot is similar to the previous one, but the 'BCCH Mode' dropdown menu is open, showing a list of options: 'Default' (highlighted), 'Fixed', 'Random', and 'Advanced'. The rest of the page content remains the same.

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.

The screenshot shows the 'BCCH' configuration page with 'BCCH Mode' set to 'Fixed'. Below this, there are three input fields labeled 'First of BCCH', 'Second of BCCH', and 'Third of BCCH'. The table below the fields has columns: Index, MCC, MNC, LAC, CID, BCCH, and 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 | <input checked="" type="radio"/> No <input type="radio"/> Yes | | | | | |
| Apply To All Ports | <input checked="" type="radio"/> No <input type="radio"/> Yes | | | | | |
| Index | MCC | MNC | LAC | CID | BCCH | Receive Level |

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

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

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

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

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

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

4.7.10 Call Waiting

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

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

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

Notes: Call waiting 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.

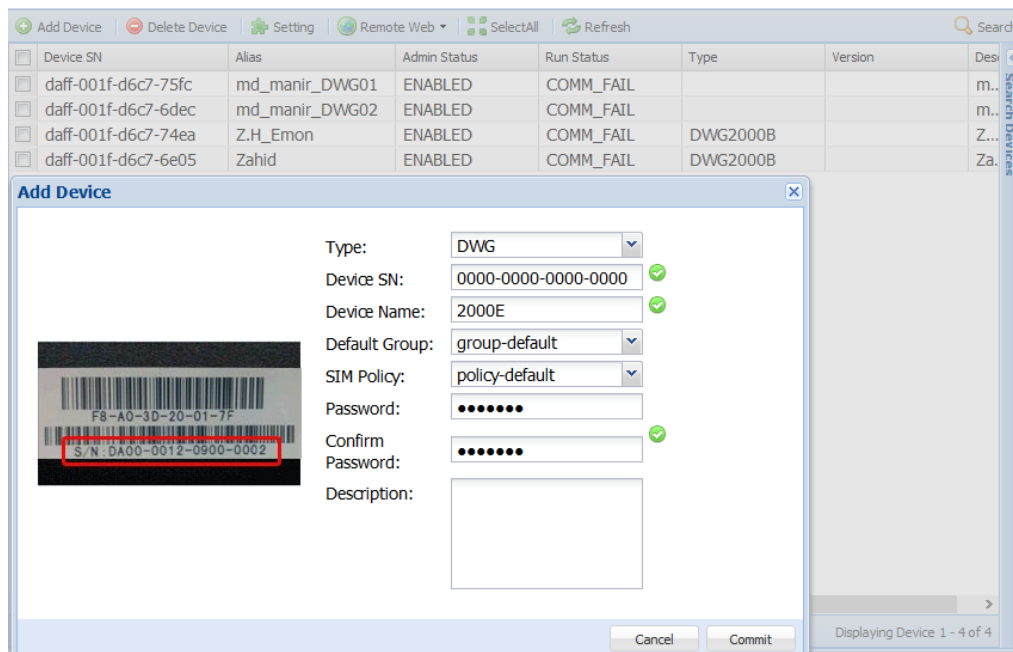
| Cloud Server | |
|---|---|
| Primary Server Domain | <input type="text" value="best.cloud.com"/> |
| Primary Server Port | <input type="text" value="2020"/> |
| Secondary Server Domain | <input type="text"/> |
| Secondary Server Port | <input type="text"/> |
| Domain Name | <input type="text"/> |
|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

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

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

► How to register gateway to SIM Cloud?

Example: add gateway on domain support.cloud.com



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

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

4.8 SMS AND USSD

4.8.1 SMS Send Overview

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

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

4.8.2 SMS Send Limit Settings

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

SMS Send Limit Settings - Add Rule

| | | | |
|---------------|----------|--|-----------------------|
| Index | 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> | | |

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

| Port | Send Date | Number | SMS Content | Send Status |
|--|-----------|--------|-------------|-------------|
| Total: 0 entries 16 entries/page 1/0 page <input type="button" value="▼"/> | | | | |

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.

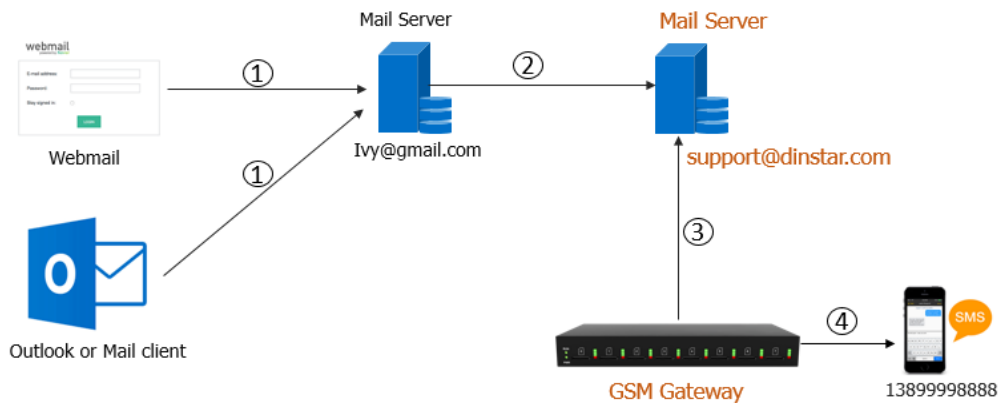
When email use protocol IMAP, if email read successful, will set the email status to read. If read email failed, will try to read again (MAX 3 times). If failed final, the email status will be set to read also.

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 | <input type="text"/> |
| SMTP Domain | <input type="text"/> |
| SMTP Port | <input type="text"/> |
| SMTP Username | <input type="text"/> |
| SMTP Password | <input type="password"/> <input type="button" value="Show Password"/> |
| TLS Enable | <input type="checkbox"/> |
| SMS to Email | <input type="checkbox"/> Enable |
| GSM Event to Email | <input type="checkbox"/> Enable |
| Email To SMS | <input checked="" type="checkbox"/> Enable |
| Protocol | POP3 ▼ |
| Server Domain | pop3.dinstar.com |
| Server Port | 110 |
| TLS Enable | <input type="checkbox"/> |
| Using SMTP Login Account | <input type="checkbox"/> |
| Username | support@dinstar.com |
| Password | ***** <input type="button" value="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:

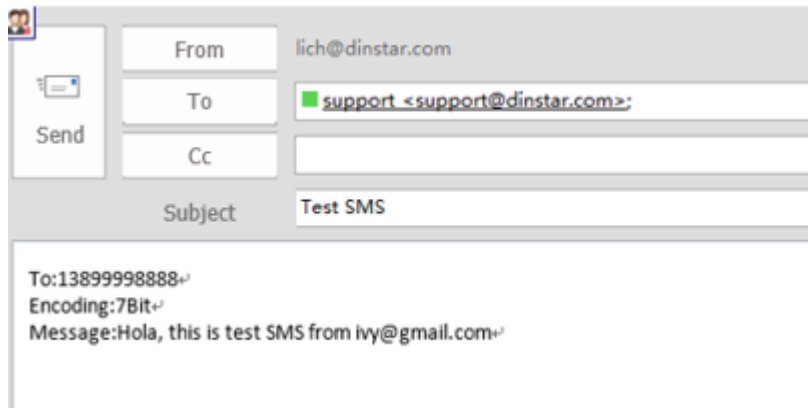
To:xxxxxxxxxxxx

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

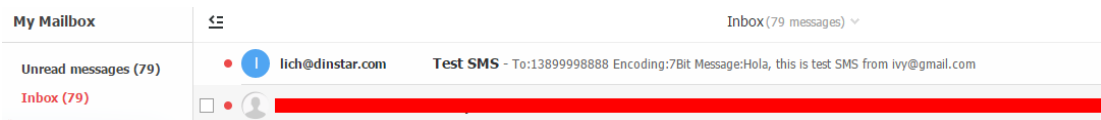
Encoding:7Bit

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

Message:



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

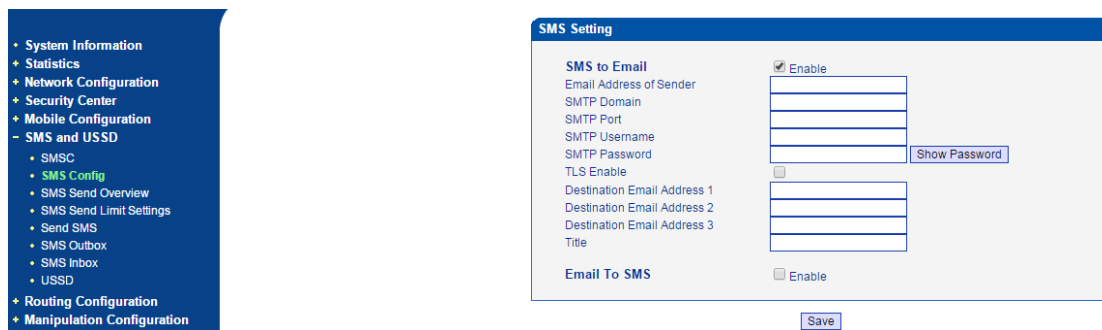


Note:

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

How to set SMS to Email

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



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

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

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

SMTP Username: Enter your e-mail address username.

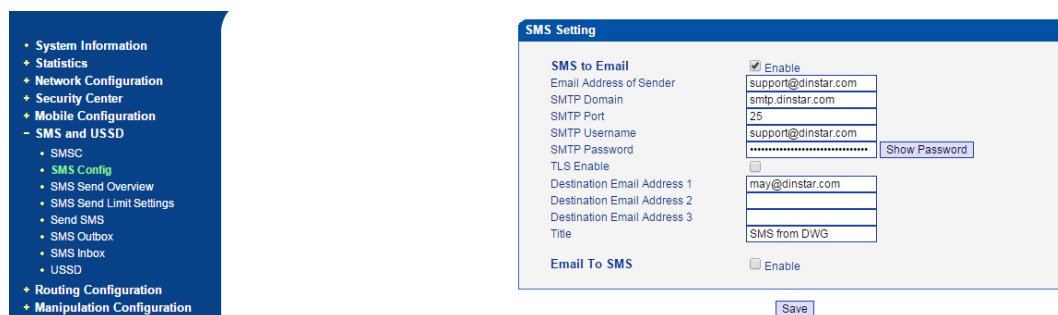
SMTP Password: Enter the password of the e-mail address you configured, please make sure it is correct.

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

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

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

Here is one example:



4.9 CALL CONFIGURATION

4.9.1 SIP Configuration

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

► Configure SIP server and Outbound Proxy server

| | |
|--------------------------------|---|
| SIP Proxy | |
| SIP Server Address | <input type="text"/> |
| SIP Server Port(default: 5060) | <input type="text" value="5060"/> |
| Check Net Status | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Outbound Proxy | |
| Outbound Proxy Address | <input type="text"/> |
| Outbound Proxy Port | <input type="text" value="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 | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Use Same Local Sip Port | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Use Random Port | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Local SIP Port | <input type="text" value="5060"/> |

▶ **Random**

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

► T1

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

▶ **T2**

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

▶ **T4**

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

▶ **TMAX**

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

▶ **Keepalive Interval**

The gateway can monitor the status of SIP server by sending periodic SIP OPTION messages. This field specifies transmission timer of OPTION message. Its range from 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 | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Session Timer Interval(range:90 - 60000s) | 1800 s |
| Session timer mode | refresh |
| Session timer refresher | uac |

▶ Session timer Interval

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

A UAS or proxy can insert a Session-Expires header in the INVITE if the UAC did not include one. Thus a UAC can receive a Session-Expires header in a response even if none was present in the request. Its value 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 | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Gsm Reason | Sip Response Code |
| No Port Found | 503 |
| Unassigned Number | 404 |
| Normal Call Clearing | 480 |
| User Busy | 486 |
| User Not Answer | 408 |
| Call Rejected | 403 |
| Mobile Network Fault | 503 |

► SIP Response

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

► Response Code switch

This part specifies response codes of SIP between gateway and SIP server. Refer to table *SIP Response*, the SIP server possibly 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 | Response code after switch |
|--------------------------|---------------|----------------------------|
| <input type="checkbox"/> | 183 | 180 |
| <input type="checkbox"/> | | |
| <input type="checkbox"/> | | |
| <input type="checkbox"/> | | |

4.9.2 SIP Trunk Configuration

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

Table 4-11-1 Description of IP Trunk

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

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

Example

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

IP Trunk Add

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

4.9.3 SIP Trunk Group

Figure 4-11-3 IP Trunk Group

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

Table 4-11-2 Description of IP Trunk Group

| Parameters | Description |
|----------------|--|
| IP Trunk Group | This configuration is optional, and is used to add the IP that have the same attributes to an IP group. The IP group will 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

Description

| | Index | IP | Port |
|-------------------------------------|-------|----------------|------|
| <input checked="" type="checkbox"/> | 31 | 172.16.221.221 | 5060 |

4.9.4 Port Configuration

| Port List | | | | | | | | |
|-----------|-------------|-----------------|---------|---------|-----------------|-----------------|-------------------------|------------------------|
| Port | SIP User ID | Authenticate ID | 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

SIP User ID

Authenticate ID

Authenticate Password

Tx Gain

Rx Gain

To VOIP Hotline

To PSTN Hotline

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 |
| <input type="checkbox"/> | 0 | 0-3 | | | 0,1,2,3, | Cyclic Ascen... |
| <input type="checkbox"/> | 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 | <input style="width: 95%;" type="text" value="0"/> | | | | | | | | | |
| Description | <input style="width: 95%;" type="text" value="0-3"/> | | | | | | | | | |
| SIP User ID | <input style="width: 95%;" type="text"/> | | | | | | | | | |
| Authenticate ID | <input style="width: 95%;" type="text"/> | | | | | | | | | |
| Authenticate Password | <input style="width: 95%;" type="password"/> | <input type="button" value="Show Password"/> | | | | | | | | |
| Select Mode | <input style="width: 95%;" type="text" value="Cyclic Ascending"/> ▼ | | | | | | | | | |
| Port | <table style="width: 100%; border: none;"> <tr> <td><input checked="" type="checkbox"/> Port 0</td> <td><input checked="" type="checkbox"/> Port 1</td> </tr> <tr> <td><input checked="" type="checkbox"/> Port 2</td> <td><input checked="" type="checkbox"/> Port 3</td> </tr> <tr> <td><input type="checkbox"/> Port 4</td> <td><input type="checkbox"/> Port 5</td> </tr> <tr> <td><input type="checkbox"/> Port 6</td> <td><input type="checkbox"/> Port 7</td> </tr> </table> | | <input checked="" type="checkbox"/> Port 0 | <input checked="" type="checkbox"/> Port 1 | <input checked="" type="checkbox"/> Port 2 | <input checked="" type="checkbox"/> Port 3 | <input type="checkbox"/> Port 4 | <input type="checkbox"/> Port 5 | <input type="checkbox"/> Port 6 | <input type="checkbox"/> Port 7 |
| <input checked="" type="checkbox"/> Port 0 | <input checked="" type="checkbox"/> Port 1 | | | | | | | | | |
| <input checked="" type="checkbox"/> Port 2 | <input checked="" type="checkbox"/> Port 3 | | | | | | | | | |
| <input type="checkbox"/> Port 4 | <input type="checkbox"/> Port 5 | | | | | | | | | |
| <input type="checkbox"/> Port 6 | <input type="checkbox"/> Port 7 | | | | | | | | | |

4.9.6 Digitmap

Digit Map

| | |
|-----------|--|
| Digit Map | <div style="font-family: monospace; font-size: 1.2em;">x.T x.#</div> |
|-----------|--|

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 | |
| <input type="checkbox"/> | 63 | default | SIP Server | Port Group-0 | Allow | -- | -- | -- | -- | -- |
| <input type="checkbox"/> | 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 | <input style="width: 90%;" type="text" value="62"/> |
| Description | <input style="width: 90%;" type="text"/> |
| Source | <input style="width: 90%;" type="text" value="SIP Server"/> |
| Destination | <input style="width: 90%;" type="text" value="port-group-0 <all>"/> |
| Call Restriction | <input style="width: 90%;" type="text" value="Allow Call"/> |
| Advanced Rules | <input style="width: 90%;" type="text"/> |

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 | |
| <input type="checkbox"/> | 63 | default | Any | SIP Server | Allow | -- | -- | -- | -- | |
| <input type="checkbox"/> | 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 | |

Save Reset Cancel

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 | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Enable GSM Incoming Configuration | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Auto Outgoing Routing Type | Polling <input type="button" value="v"/> |
| IP to GSM One Stage Dialing | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Answer Delay | <input type="text" value="5"/> s |
| Redirect Call When All Ports Busy | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Play Voice Prompt for GSM Incoming Calls | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| RTP Detected Enable | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Period without RTP Packet | <input type="text" value="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 | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| User ID Is Phone Number | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Only Accept Calls from SIP Server | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Allow Call from GSM to IP without Registration | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Allow Call from IP to GSM without Registration | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Reject Anonymous Call from IP to GSM | <input checked="" type="radio"/> No <input type="radio"/> Yes |
| Use # as End Key | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| No Answer Timeout | <input type="text" value="55"/> Sec(s) |
| Interdigit Timeout | <input type="text" value="4"/> Sec(s) |
| Reset ASR after SIM Switching | <input checked="" type="radio"/> no <input type="radio"/> yes |

► Enable Private Service

To enable local services like *158# etc.

► User ID Is Phone Number

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

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

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

▶ Configure codec list

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

4.9.11 DBO Parameter

Enable DBO service

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

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

Save

Configure DBO parameter

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

| DBO Parameter | |
|-----------------------------|--|
| Enable DBO | <input checked="" type="checkbox"/> |
| Active DBO Server URL/IP | <input type="text" value="172.16.99.129"/> |
| Active DBO Server Port 0 | <input type="text" value="3479"/> |
| Active DBO Server Port 1 | <input type="text" value="6479"/> |
| Active DBO Server Port 2 | <input type="text" value="12479"/> |
| Active DBO Server Port 3 | <input type="text" value="24479"/> |
| Active DBO Server Username | <input type="text" value="172.16.99.129_3479"/> |
| Active DBO Server Password | <input type="password" value="*****"/> |
| Standby DBO Server URL/IP | <input type="text" value="172.16.200.101"/> |
| Standby DBO Server Port 0 | <input type="text" value="3478"/> |
| Standby DBO Server Port 1 | <input type="text"/> |
| Standby DBO Server Port 2 | <input type="text"/> |
| Standby DBO Server Port 3 | <input type="text"/> |
| Standby DBO Server Username | <input type="text" value="172.16.200.101_3479"/> |
| Standby DBO Server Password | <input type="password" value="*****"/> |
| Enable SIP Forwarding | <input checked="" type="checkbox"/> |
| Enable RTP Forwarding | <input checked="" type="checkbox"/> |

| DBO Advanced | |
|-----------------------------|-------------------------------------|
| Enable Bandwidth Compressed | <input checked="" type="checkbox"/> |

Parameter Description:

| Parameters | Description |
|-----------------------------|--|
| Active DBO Server URL/IP | Primary DBO server IP or domain for traffics |
| Active DBO Server Port | <p>DBO service ports that dedicate by DBO server.</p> <p>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.</p> |
| Active DBO Server Username | <p>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.</p> <p>The username with the format as x.x.x.x_3479 by default. x.x.x.x is the IP of DBO server.</p> |
| Active DBO Server Password | <p>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.</p> |
| Standby DBO Server URL/IP | Secondary DBO server IP or domain. |
| Standby DBO Server Port | <p>DBO service ports that dedicate by DBO server.</p> <p>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.</p> |
| Standby DBO Server Username | <p>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.</p> <p>The username with the format as x.x.x.x_3479 by default. x.x.x.x is the IP of DBO server.</p> |
| Standby DBO Server Password | <p>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.</p> |
| Enable SIP Forwarding | <p>Enable SIP signaling encryption and forward by DBO server. The SIP signaling will forward by</p> |

| | |
|-----------------------------|---|
| | 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 |
| <input type="checkbox"/> | 0 | 546464546 | --- | --- | --- | --- | --- |
| <input type="checkbox"/> | 1 | | --- | --- | --- | --- | --- |
| <input type="checkbox"/> | 2 | | --- | --- | --- | --- | --- |
| <input type="checkbox"/> | 3 | | --- | --- | --- | --- | --- |
| <input type="checkbox"/> | 4 | | --- | --- | --- | --- | --- |
| <input type="checkbox"/> | 5 | | --- | --- | --- | --- | --- |
| <input type="checkbox"/> | 6 | | --- | --- | --- | --- | --- |
| <input type="checkbox"/> | 7 | | --- | --- | --- | --- | --- |
| <input type="checkbox"/> | All | | <input type="button" value="Clear"/> | <input type="button" value="Clear"/> | <input type="button" value="Reset"/> | <input type="button" value="Reset"/> | <input type="button" value="Reset"/> |

4.10.2 Basic Configuration

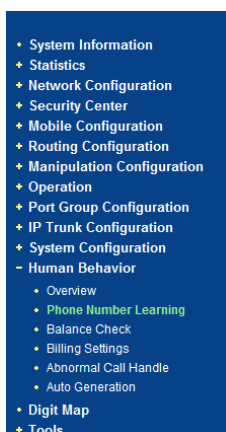
On the **Basic Configuration** interface, you can set how long an IP →Tel call or a Tel→IP call will be delayed, as well as call interval. The 'set call volume threshold function' is mainly used for anti-blocked (such as some operators launched special call testing for the detection of the VoIP equipment,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

| Basic Configuration | |
|---|--|
| Tel to IP Call Delay(range:0-60s) | <input type="text" value="0"/> s- <input type="text" value="0"/> s Note:If both are set as "0", it means the function is not enabled. |
| Startup Interval(range:0-3600s) | <input type="text" value="0"/> s- <input type="text" value="2"/> s Note:If both are set as "0", it means the function is not enabled. |
| IP to Tel Call Delay(range:0-10s) | <input type="text" value="0"/> s |
| Call Interval(range:0-3600s) | <input type="text" value="0"/> s- <input type="text" value="0"/> s |
| No Alerting Call Handle | <input checked="" type="radio"/> Normal Handle <input type="radio"/> Hang Up <input type="radio"/> Not Answer |
| IP to TEL Processing Timeout Handle | <input type="checkbox"/> |
| Set Call Volume Threshold | <input type="checkbox"/> |
| SMS Sending Delay (range:0-300s) | <input type="text" value="0"/> s- <input type="text" value="0"/> s Note:If both are set as "0", it means the function is not enabled. |
| Numeric Scale | <input type="text" value="2"/> |
| GSM incoming call limit (range:0-3600s) | <input type="text" value="0"/> s- <input type="text" value="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) | <input type="text" value="0"/> Minutes |
| Query SIM information during initiation | <input checked="" type="radio"/> Enable <input type="radio"/> Disable |
| Timeout | <input type="text" value="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 | <input type="text" value="0"/> |
| Type | <input type="text" value="USSD"/> |
| Send Text | <input type="text" value="*156#"/> |
| Keywords | <input]="" matching="" td="" test<="" type="text" value="Your number is:"/> |
| Write Phone Number to SIM card | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Port Group | <input type="text" value="0 <all>"/> |
| <input type="button" value="Save"/> <input type="button" value="Reset"/> <input type="button" value="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 [N]number is [N], abc" is used to match "456" in SMS "number is 123

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: <input type="button" value="Test End"/> |
| Test SMS Text | Your number is: 8618344144906 |
| Result | <input type="button" value="Test"/> |
| Write Phone Number to SIM card | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Port Group | 0 <all> |

Phone Number Learning - Add Rule

| | |
|--------------------------------|---|
| Index | 0 |
| Type | USSD |
| Send Text | *156# |
| Keywords | Your number is: <input type="button" value="Test End"/> |
| Test SMS Text | Your number is: 8618344144906 |
| Result | <input type="button" value="Test"/> 8618344144906 |
| Write Phone Number to SIM card | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| 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: <input type="button" value="Matching Test"/> |
| Write Phone Number to SIM card | <input type="radio"/> No <input checked="" type="radio"/> 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: <input type="button" value="Test End"/> |
| Test SMS Text | Your number is: 8618344144906 |
| Result | <input type="button" value="Test"/> |
| Write Phone Number to SIM card | <input type="radio"/> No <input checked="" type="radio"/> 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: <input type="button" value="Test End"/> |
| Test SMS Text | Your number is: 8618344144906 |
| Result | <input type="button" value="Test"/> 8618344144906 |
| Write Phone Number to SIM card | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| 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: <input type="button" value="Matching Test"/> |
| Write Phone Number to SIM card | <input type="radio"/> No <input checked="" type="radio"/> 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 SMS Text: Your number is: 8618344144906
 Result:
 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 SMS Text: Your number is: 8618344144906
 Result: 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.

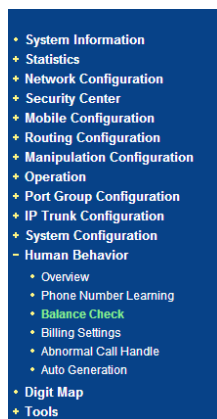
If you want to check balance automatically and block SIM card when it is low balance. The UC2000 have 3 modes to check balance: USSD/SMS/Call.

- 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 |
| <input type="checkbox"/> | 0 | | 73.40 | 73.40 | 6000.00 | 600.00 |
| <input type="checkbox"/> | 1 | | | | 0.00 | 0.00 |
| <input type="checkbox"/> | 2 | | | | 6000.00 | 600.00 |
| <input type="checkbox"/> | 3 | | | | 0.00 | 0.00 |
| <input type="checkbox"/> | 4 | | | --- | --- | --- |
| <input type="checkbox"/> | 5 | | | --- | --- | --- |
| <input type="checkbox"/> | 6 | | | --- | --- | --- |
| <input type="checkbox"/> | 7 | | | --- | --- | --- |
| <input type="checkbox"/> | All | | | | <input type="button" value="Restore"/> | <input type="button" value="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.



| Balance Check - Add Rule | |
|--|--|
| Index | 0 |
| Type | USSD |
| Send Text | *101# |
| Keywords | Your balance is <input type="button" value="Matching Test"/> |
| Digit Thousand Symbol | . |
| Digit Point Symbol | . |
| Port Group | 0 <all> |
| Check Balance After SIM Card Registration | <input checked="" type="checkbox"/> |
| Check Balance Every | 20 Minutes <small>Note: "0" means disable.</small> |
| Check While Calculated Balance Is Low | 3 <small>Note: "0" means disable.</small> |
| <input type="button" value="Save"/> <input type="button" value="Reset"/> <input type="button" value="Cancel"/> | |

About Key Words: 1. You can input multiple keywords and special symbol like "[E], [T], [I], [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 <input type="button" value="Matching Test"/> |
| Digit Thousand Symbol | . |
| Digit Point Symbol | . |
| Port Group | 0 <port0-3> |
| Check Balance After SIM Card Registration | <input checked="" type="checkbox"/> |
| Check Balance Every | 20 Minutes <small>Note: "0" means disable.</small> |
| Check While Calculated Balance Is Low | 3 <small>Note: "0" means disable.</small> |

(next page)

| Balance Check - Add Rule | |
|---|---|
| Index | 0 |
| Type | USSD |
| Send Text | *101# |
| Keywords | Your balance is <input type="button" value="Test End"/> |
| Test SMS Text | Your balance is 73.40\$ |
| Result | <input type="button" value="Test"/> 73.40 |
| Digit Thousand Symbol | . |
| Digit Point Symbol | . |
| Port Group | 0 <port0-3 |
| Check Balance After SIM Card Registration | <input checked="" type="checkbox"/> |
| 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 | <input checked="" type="checkbox"/> |
| 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 | <input checked="" type="checkbox"/> |
| 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> |
| Check Balance After SIM Card Registration | <input checked="" type="checkbox"/> |
| 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 <input type="button" value="Test End"/> |
| Test SMS Text | Your balance is 73.40\$ |
| Result | <input type="button" value="Test"/> 73.40 |
| Digit Thousand Symbol | , |
| Digit Point Symbol | . |
| Port Group | 0 <port0-3> |
| Check Balance After SIM Card Registration | <input checked="" type="checkbox"/> |
| Check Balance Every | 20 Minutes Note: "0" means disable. |
| 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 | <input type="radio"/> No <input checked="" type="radio"/> Yes | |
| Low Credits Warning | <input checked="" type="radio"/> No <input type="radio"/> 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 | <input type="radio"/> No <input checked="" type="radio"/> Yes | | |
| Call Event | | | |
| <input type="checkbox"/> Low ASR Less Than | 12 | % | Handle: Reset |
| <input type="checkbox"/> Low ACD Less Than | 0 | s | Handle: Reset |
| <input type="checkbox"/> Counts of Call Failed | 0 | | Handle: Reset |
| <input type="checkbox"/> Low Balance Less Than | 0.00 | | Handle: Block SIM |
| <input type="checkbox"/> By Gsm Code | 8 | | Handle: Reset |
| <input type="checkbox"/> PDD Less Than(1-30) | 2 | s | Handle: Reset |
| <input type="checkbox"/> Cost Difference | 0.00 | - 0.00 | Handle: Block |
| USSD Event | | | |
| <input type="checkbox"/> Counts of Send Fail(1 - 100) | 10 | | Handle: Reset |
| USSD/SMS Monitor | | | |
| <input type="checkbox"/> SMS Number | | | Handle: Reset |
| Keywords | | | |
| USSD/SMS URL Monitor | | | |
| <input type="checkbox"/> SMS Number | | | Handle: Access Internet |
| Keywords | | | |
| SIM Register Fail | | | |
| <input type="checkbox"/> Register Timeout(60 - 600) | 600 | s | Handle: 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 | Value | Unit | Handle | SMS Alert |
|--|-------|--------|-----------|-------------------------------------|
| <input type="checkbox"/> Low ASR Less Than | 12 | % | Block SIM | <input type="checkbox"/> |
| <input type="checkbox"/> Low ACD Less Than | 0 | s | Reset | <input type="checkbox"/> |
| <input type="checkbox"/> Counts of Call Failed | 0 | | Block | <input type="checkbox"/> |
| <input type="checkbox"/> Low Balance Less Than | 0.00 | | Block SIM | <input checked="" type="checkbox"/> |
| <input type="checkbox"/> By Gsm Code | 8 | | SMS Test | <input type="checkbox"/> |
| <input type="checkbox"/> PDD Less Than(1-30) | 2 | s | Reset | <input type="checkbox"/> |
| <input type="checkbox"/> Cost Difference | 0.00 | - 0.00 | Block | <input type="checkbox"/> |

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

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

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

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

GSM Network side error code handle

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

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

Follow these steps to use this feature:

- a) Enable the error code record

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

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

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

Don't forget to save the configuration.

- b) Configure the GSM code monitor

| | | | | |
|-------------------------------------|-----------------------|------|-------|-------------------------------------|
| <input type="checkbox"/> | Low Balance Less Than | 5.00 | Block | <input checked="" type="checkbox"/> |
| <input checked="" type="checkbox"/> | By Gsm Code | 8 | Block | <input type="checkbox"/> |
| <input type="checkbox"/> | Consecutive Counts | 3 | | <input type="checkbox"/> |

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 | 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 | |
|---|--|
| <input type="checkbox"/> Counts of Send Fail(1 - 100) | <input type="text" value="10"/> <input type="button" value="Reset"/> |

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 | |
|--|--------------------------------------|
| <input checked="" type="checkbox"/> SMS Number | <input type="text"/> |
| Keywords | <input type="text" value="block"/> |
| | <input type="button" value="Block"/> |

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 digits

Call Out

Min Call Duration: seconds

Max Call Duration: seconds

Auto Send SMS:

Random Content-1:

Random Content-2:

Random Content-3:

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

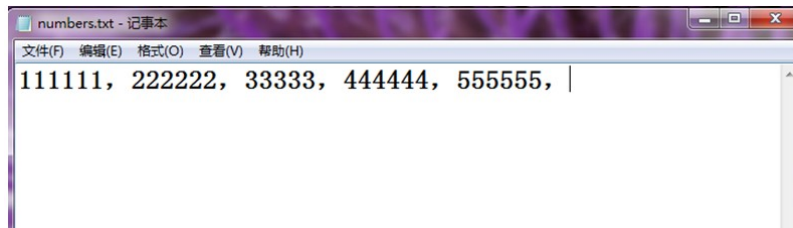
SIM Card Number section.

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: | <input checked="" type="checkbox"/> |
| Min Interval: | <input type="text" value="30"/> minutes |
| Max Interval: | <input type="text" value="120"/> minutes |
| By Total Call Durations: | <input checked="" type="checkbox"/> |
| Call Duration: | <input type="text" value="60"/> minutes |
| By Consecutive Calls: | <input checked="" type="checkbox"/> |
| Consecutive Calls: | <input type="text" value="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

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

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

The Signal Log is 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*

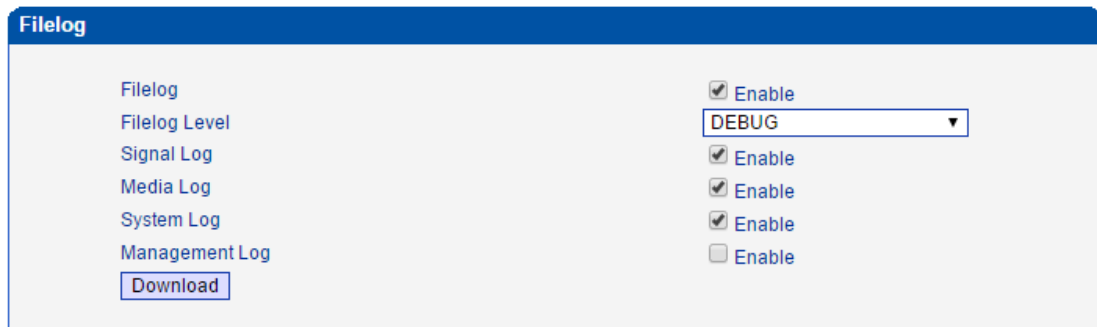
The Management Log is include following traces which defined in system by default

- *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



| | |
|----------------|--|
| Filelog | <input checked="" type="checkbox"/> Enable |
| Filelog Level | DEBUG ▾ |
| Signal Log | <input checked="" type="checkbox"/> Enable |
| Media Log | <input checked="" type="checkbox"/> Enable |
| System Log | <input checked="" type="checkbox"/> Enable |
| Management Log | <input type="checkbox"/> 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 | <input checked="" type="checkbox"/> Enable |
| Record Cycle | <input type="text" value="60"/> Minutes |
| Write Flash Timer | <input type="checkbox"/> Enable |
| <input type="button" value="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 | <input type="text" value="1,2,3"/> Up to 3 ports,e.g."1,2,15" |
| Record in summary | <input checked="" type="checkbox"/> Enable |
| Record in media log | <input checked="" type="checkbox"/> Enable |

Enable trace while remote SIM card used in this device.

4.11.5 Ping test

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

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

4.11.6 Tracert Test

You can check the routes of the tracert destination.

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

4.11.7 Network Capture

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

Voice stream transmit path of the gateway as below:

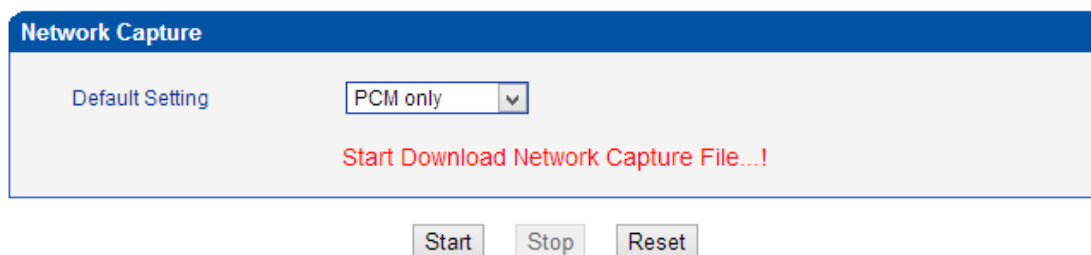


▶ Getting start to PCM capture

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

▶ To enable PCM capture

- ◆ Select 'PCM only' on Network Capture page



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

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

| No. | Time | Source | Destination | Protocol | Length | Info | |
|-----|----------|-------------------|-------------------|------------|--------|-------------------------------|---------------------------------|
| 1 | 0.000000 | Motorola_1c:1d:1e | Cimsys_33:44:55 | CSM_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] | Ch: 0xFFFF, Seq: 11 (From Host) |
| 3 | 0.000245 | Cimsys_33:44:55 | Motorola_1c:1d:1e | CSM_ENCABS | 44 | --> 0x0021 | Ch: 0x0003, Seq: 0 (From Host) |
| 4 | 1.320893 | Motorola_1c:1d:1e | Cimsys_33:44:55 | CSM_ENCABS | 104 | --> 0x0e00 | |
| 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 | 1.330659 | 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

The screenshot shows the 'Network Capture' interface. A dropdown menu is set to 'Syslog special'. Below the menu is a red button that says 'Start Download Network Capture File...!'. At the bottom, there are three buttons: 'Start', 'Stop', and 'Reset'.

- ◆ 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 cal |
| 12 | 0.331167 | 172.16.222.22 | 1.1.1.1 | Syslog | 983 | USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 11> [DEBUG] <<<<<< message from 172.16.222.127/5060, cryp |
| 13 | 0.331498 | 172.16.222.22 | 1.1.1.1 | Syslog | 177 | USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 12> [DEBUG] <<<<<< from 172.16.222.127/5060, crypt:FALSE, 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:100806172.16.222.22:5060 SIP/2.0\r\n |
| 15 | 0.332307 | 172.16.222.22 | 1.1.1.1 | Syslog | 122 | USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_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 172.16.222.127/5060, crypt:FALSE Phone |
| 20 | 0.333877 | 172.16.222.22 | 1.1.1.1 | Syslog | 386 | USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_sip: < 19> [DEBUG] SIP/2.0 100 trying\r\nvia: SIP/2.0/UDP 172.16. |
| 21 | 0.346687 | 172.16.222.22 | 1.1.1.1 | Syslog | 131 | USER.DEBUG: Jul 23 06:52:05 172.16.222.22 mpe_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 172.16.222.127/5060, cryp |
| 24 | 7.233513 | 172.16.222.22 | 1.1.1.1 | Syslog | 177 | USER.DEBUG: Jul 23 06:52:12 172.16.222.22 mpe_sip: < 23> [DEBUG] <<<<<< from 172.16.222.127/5060, crypt:FALSE, 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:100806172.16.222.22:5060 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

The screenshot shows the 'Network Capture' interface. A dropdown menu is set to 'RTP special'. Below the menu is a red button that says 'Start Download Network Capture File...!'. At the bottom, there are three buttons: 'Start', 'Stop', and 'Reset'.

- ◆ Click Start to enable RTP capture

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

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

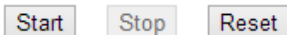
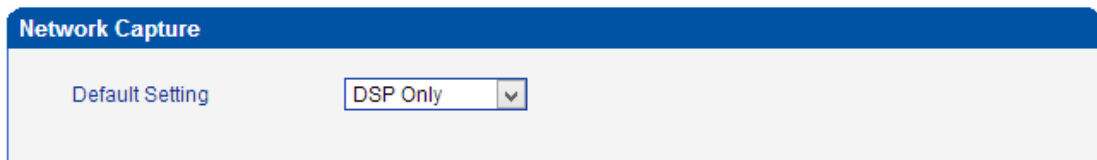
| No. | Time | Source | Destination | Protocol | Length | Info |
|-----|-----------|----------------|----------------|----------|--------|---|
| 176 | 7.020000 | 172.16.221.228 | 116.204.105.50 | SIP | 565 | Request: REGISTER sip:116.204.105.50 |
| 178 | 7.030000 | 116.204.105.50 | 172.16.221.228 | SIP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

▶ Getting start to DSP capture

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

▶ To enable DSP capture:

- ◆ Select DSP only on Network Capture page



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

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

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

► Configurable capture options

► Getting start to custom capture

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

Network Capture

Default Setting: Custom

Network Interface: LAN DSP

Source Host:

Destination Host:

Select Port: None

Protocol(s): TCP UDP RTP RTCP ICMP ARP

Start Download Network Capture File...!

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 No file chosen

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 | <input type="button" value="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 No file chosen

Data Backup

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

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

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

4.12.5 Remote Server

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

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

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

4.12.6 Email Account Setting

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

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

4.12.7 Username and Password

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

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

4.12.8 Access control

Access Control

New Password:

Confirm Password:

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

Access Control

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

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

4.12.9 Factory Reset

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

Factory Reset

Click this button to reset factory default settings

Notes:The device must restart to take effect.

4.12.10 Auto Restart

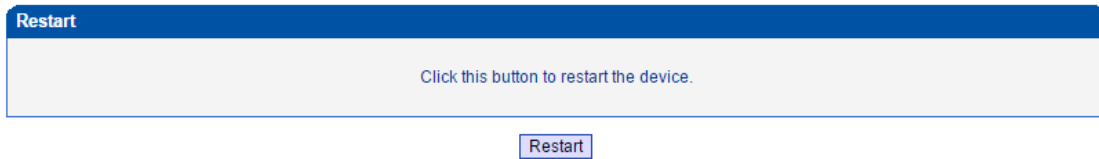
Configure auto restart at pre-defined HH/MM

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 switches
  ppp        PPP
  product     Product default config
  reset       Reset the board
  rtp         RTP debug command
  save        save configuration
  sd          sd debug command
  setcustom   set custom
  shutdown    shutdown a user
  sip         config sip informations
  snmp-server Modify SNMP parameters
  user_timeout set telnet users timeout
  vlan        vlan route add or delete
  vlanif      vlan interface tagged properties
  webs        web server command
  workmode    network workmode selection:bridge or router
ROS<config>#
ROS<config>#

```

5.3.2 GENERAL PURPOSE COMMANDS IN "CONFIG" MODE

▶ Set time (clock set)

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

▶ Save the configuration (save)

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

▶ Restart device (reset eia)

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

▶ Enable debug

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

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

ROS<config>#
```

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

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

ROS<config>#
```

Without 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.