EJOIN ACOM516 VoIP Gateway

User Manual

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Chapter Ⅰ Equipment Information

1.1 Product Brief

ACOM516 VoIP Gateway is a multi-functional and high performance product, which is mainly used for call termination (VoIP to Mobile) and origination (Mobile to VoIP). It can enable to make 16 calls simultaneously. It is based on SIP and compatible with Asterisk, 3CX, Elastix, IPPBX, VOS, VPS operating platform.

ACOM516 VoIP Gateway also can be used as a Network SMS modem which supports SMS sending, receiving, group sending. It has flexible HTTP/SIP API for SMS service. Customers can develop SMS server easily by the API.

ACOM516 VoIP Gateway supports GSM, CDMA and WCDMA network (one gateway one network). There are 3 modes in this series gateway: ACOM516-16, ACOM516-64 and ACOM516-128, the SIM capacity in these 3 modes are different.

1.2 Product Application

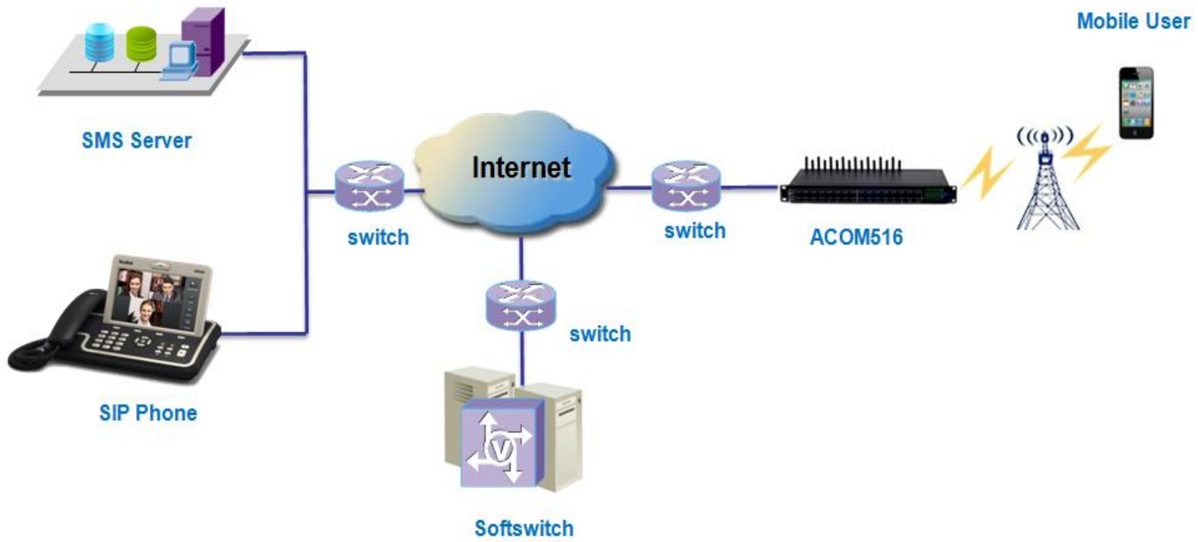


Figure 1.2-1 Product Application

1.3 Product Appearance

Back Panel

Figure 1.3-1 Back Panel

Description of the front panel(from left to right)：

- 1 Ground connection
- 1 reset button (press RST button about 10s will restore to factory settings)
- 1 Power Interface (DC 12V 5A)
- 2 Network Interface (LAN and WAN, RJ45)
- 1 Console Interface (USB to Serial, Baudrate 115200)
- 16 Antenna Connector

Front Panel

Figure 1.3-2 Front Panel

Description of the front panel(from left to right)：

- 64 SIM slots (4 SIM cards per channel)
- 1 Power light (indicate the status of the power connection)
- 64 LED lights (indicate the status of SIM cards)
- 2 fans

LED Status	SIM Status
Blind	No SIM/SIM Ready
Green and No-flash	SIM is calling
Flash(100ms)	SIM is locked by device or operator
Flash(500ms)	SIM is No Balance(when enable Ejoin billing system)
Flash(1000ms)	SIM registered failed

Table 1.3-1 LED Indicators

1.4 Special Features

- Support G729a/b/e,G723.1,G.711 A/U law, iLBC auto-selecting
- EBO (Ejoin Bandwidth Optimization)
- Proxy Encryption Solution for IP Block
- Support SIM Pool
- VPN (PPTP)
- SIM Card Rotating
- Base station intelligent switching/locking
- IMEI modification
- SMS and USSD API
- ERMS (Ejoin Remote Management System)
- Port Inter-Calling
- Fake ringback
- Call waiting
- Support call back
- Auto-recharge
- MNP
- State notification(CDR)
- Call Duration Limitation
- Dial Plan/Prefix Inward Translation/Intelligent Routing
- Web Browser: Firefox/Chrome /IE/Opera

1.5 Specification

Model	ACOM 516-64
Number of Channels	16 Channels
Frequency	GSM: 850/900/1800/1900MHz CDMA:800/1900MHz WCDMA:850/900/1800/1900/2100MHz

SIP Specification	SIP/2.0 RFC3261 Session Timer RFC4028 STUN
Network Protocols	DHCP/PPPoE/VPN(PPTP) NTP Telnet/HTTP/FTP/TFTP Encryption:Ejoin,VOS2000,RC4,XOP.Base64
Telephony Features	Hot-line call ,Dial plan, Speed dial, Phone book, CDR, LCR, White/Black list
Telephony Signaling	DTMF tone detection/generation DTMF relay: in-band, RFC 2833 and SIP info Call forward: unconditional, no answer and busy N-way conferencing Caller ID display/generation, Mobile Number Portability
Voice Capability	Voice codecs:G729a/b/e,G723.1,G.711 A/U law, iLBC Echo cancellation Silence suppression & detection(VAD, CNG) Adaptive jitter buffer Volume adjustable IVR customized
Number of Ports	1 WAN 10/100Base-T ethernet(RJ-45 connector) 1 LAN 10/100Base-T ethernet(RJ-45 connector) 1 Console(USB)
LED	1 Power and 16 groups of card online and running status indicator
Power Supply	100-240VAC, 50 - 60 Hz IN, 12V/5A OUT
Operating Environment	Operating temperature: 0 - 50 ℃ Operating humidity: 10 – 90%RH
Warranty	12 Months

Table 1.5-1

1.6 Mobile Features

- SMS Send, Receive and Forward (GSM/SIP/HTTP)
- SMS Inbox
- AT Command, USSD
- SMS Format: PDU/TXT
- PIN Code Management
- CDMA Delay Answer
- GSM Polarity Reversal
- Carrier Selection
- Caller ID Hidden (need SIM Card support)

Chapter Ⅱ Equipment Installation

2.1 SIM Card Placement

Insert SIM cards like the figure 2.1-1. The SIM cards should be mini-SIM (2FF).

Figure 2.1-1 SIM Card Placement

2.2 Antenna Installation

The external antenna should be installed vertically always on a site with a good wireless signal. It is strongly recommend that you choose the long antenna.

Figure 2.2-1 Antenna Installation

2.3 Network Connection

Plug Ethernet line into gateway WAN port, and then connect the other end of the Ethernet line with switch or router. Note: Do not use LAN port, LAN port is useless.

Figure 2.3-1 Network Connection

2.4 Power Connection

Connect the small end of the power cable to the power input on the back panel, and plug the other end of the cable into a 220V power outlet.

Figure 2.4-1 Power Connection

2.5 Serial Connection

Connect one side of serial cable to the console port on the back panel, another side to computer USB port.(Don't need connect it normally)

Figure 2.5-1 Serial Connection

Chapter Ⅲ Web Settings

3.1 Login

Open the web browser and type the IP address. If it is the first time you login the gateway, please use the default settings below:

IP Address: http://192.168.1.67

Account: root

Password: root

Figure 3.1-1 Login web

3.2 Basic Settings

WAN Settings

There are three types of WAN port IP: Static, Dynamic and PPPoE. (Default static IP is 192.168.1.67). You can also change the wan settings when get a new device. If you want to access in this default IP, your local PC need a same network segment 192.168.1.xxx.

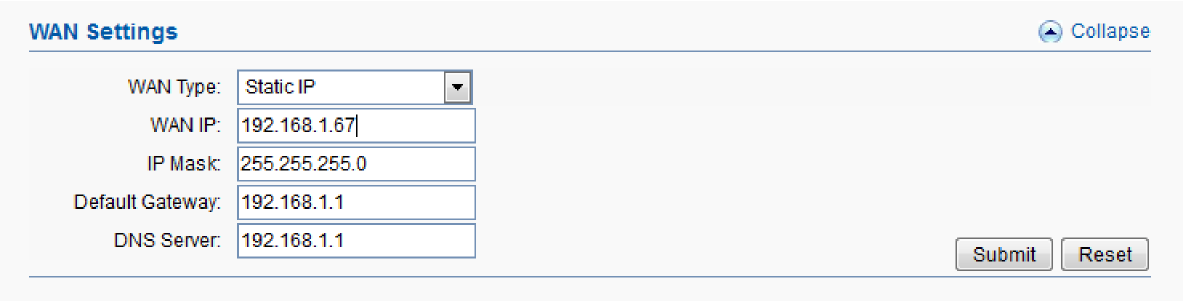


Figure 3.2-1 WAN Settings

Items	Description
WAN Type	Static IP: manually set up gateway IP. Dynamic IP: automatically get IP from local network. PPPoE: need ISP offer the account and password. Use this mode when there is no router in the local network
WAN IP	The WAN IP address of gateway
IP Mask	The subnet mask of gateway
Default Gateway	Default gateway IP address. Example: router IP.
DNS Gateway	Domain name server IP address. Example: 8.8.8.8.

Table 3.2-1

SIP Server Settings

This is the gateway settings for connecting with softswitch or server, such as VOS, VPS, IPPBX and Asterisk.

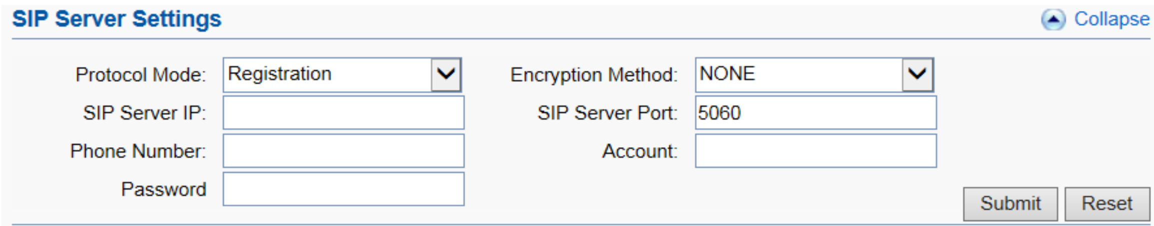


Figure 3.2-2 SIP Server Settings

Items	Description
Protocol Mode	There are two protocol modes: registration and point to point. Note: point to point can be used only when gateway and server in the same LAN or both have public IP.
Encryption Method	There are two encryption methods: EJOIN and VOS2000. (Note: Choose “EJOIN” Encryption need to set proxy server and port first.)
SIP Server IP	The IP or domain name of softswitch which will send traffic to the

	gateway. For example: VOS IP.
SIP Server Port	SIP port of softswitch, default port is 5060.
Phone Number	The caller phone number for SIP client, it can also be regarded as the SIP port number which can be called.
Account	SIP registration account which is provided by softswitch. For example: the routing gateway ID on VOS.
Password	The password of SIP registration account.

Table 3.2-2

3.3 SIP Protocol

Running Parameters

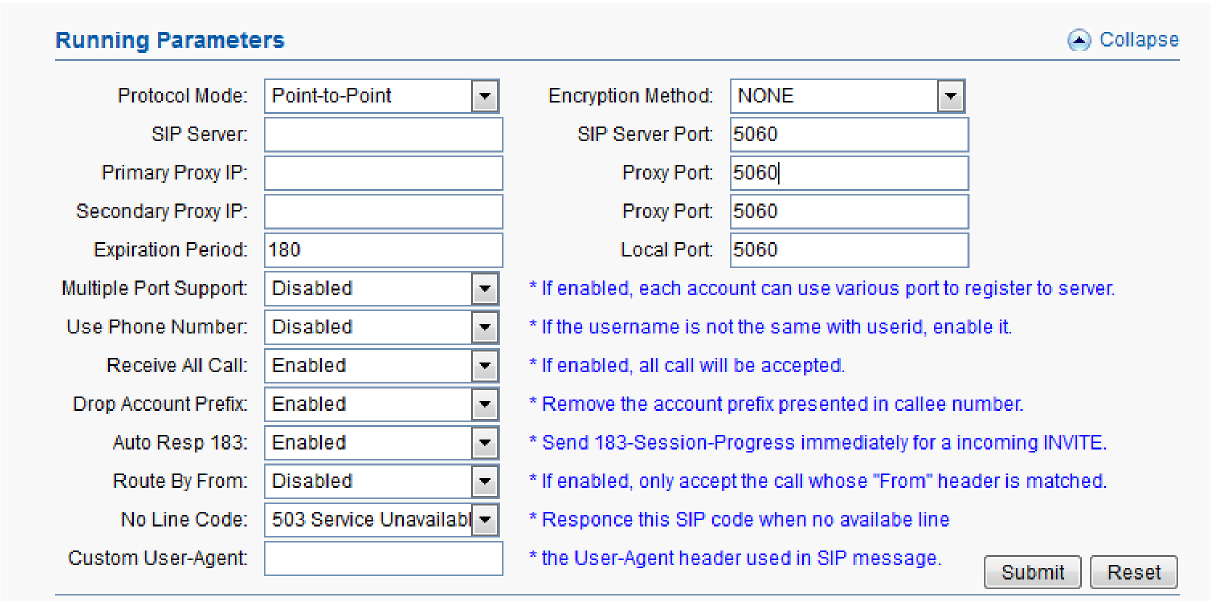


Figure 3.3-1 Running Parameters

Items	Description
Protocol Mode	It is the same as that in Basic Settings. The modification here also apply to Basic Settings page.
Encryption Method	It is the same as that in Basic Settings.
SIP Server	It is the same as that in Basic Settings.
SIP Server Port	It is the same as that in Basic Settings.
Primary Proxy IP	Proxy server will receive requests from client, and make the signaling and media streams are able to penetrate the firewall. It is usually used when gateway can’t registered with the softswitch because of network blockade.

Proxy Port	The proxy server port. Ejoin default proxy port is 25600.
Secondary Proxy IP	It is the same as primary proxy, don't need set it.
Expiration Period	Gateway will send a register request to the softswitch during every expiration period.
Multiple Port Support	Disabled: all 16 ports will be used one SIP account. Enabled: all 16 ports SIP account will be separate.
Use Phone Number	If the username is not the same with user id, enable it.
Receive All Calls	Disabled: only the SIP server address which is type in basic settings or phone book can send traffic to this gateway. Enabled: traffic from any server can send traffic to this gateway (same LAN or both gateway and server have a public IP). It's dangerous when eabled, hackers may send traffic to the gateway then steal SIM balance.
Drop Account Prefix	If it is enabled, it will remove the account prefix presented in callee number.
Auto Resp 183	If it is enabled, gateway will send 183-Session-Progress immediatey for a incoming INVITE.
Route By From	If it is enabled, gateway will only accept the call whose "From" header is matched. Note: if the gateway is just used as call termination, please disable it.
No Line Code	Gateway will send this SIP code as response to SIP server when no available line.
Custom User Agent	The User Agent header which is used in SIP message.

Table 3.3-1

STUN

STUN (Simple Traversal of UDP through NAT) is a protocol for assisting devices behind a NAT firewall or router with their packet routing. If you have the STUN server, enable STUN support, fill the server IP and port (default port is 3478), then it will work.

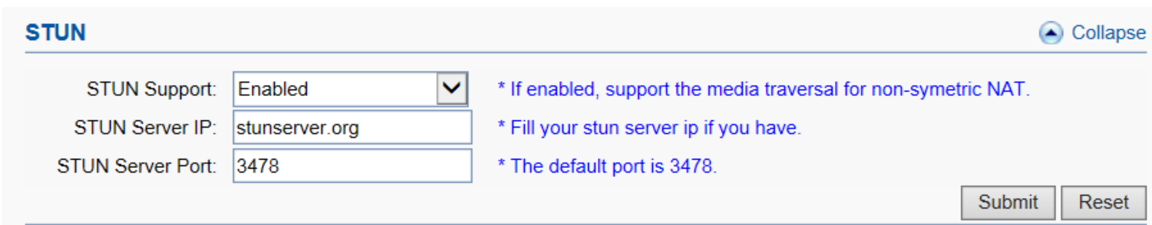


Figure 3.3-2 STUN Settings

MNP

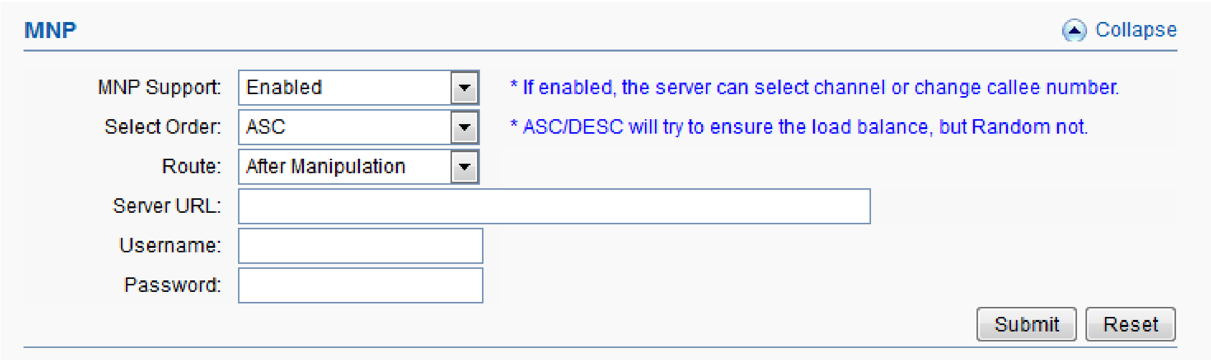


Figure 3.3-3 MNP Settings

Items	Description
MNP support	Mobile Number Portability (MNP) enables mobile telephone users to retain their mobile telephone numbers when changing from one mobile network operator to another.
Select Order	When the traffic send to the gateway, it can select ascending order, descending order or random ports.
Route	There are two choices of route: 1. Route calls after manipulation. 2. Route calls before manipulation. Note: route calls by allow prefix, callee number prefix manipulation by inward translation.
Server URL	MNP server address
Username	MNP server username
Password	Password of the username

Table 3.3-2

SIP Accounts

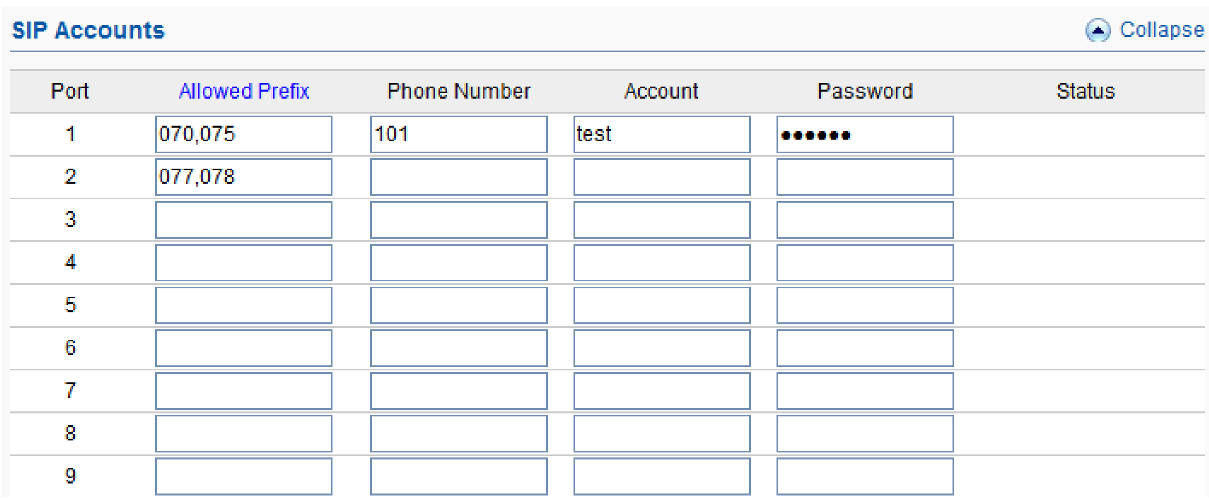


Figure 3.3-4 SIP Accounts Settings

Items	Description
Allowed Prefix	Intelligent routing, gateway will route calls by the allowed prefix. for example: channel 1 is with prefix 070 and 075, this channel will only accept the calls with prefix 070 and 075, others will not be routed to this channel. If allowed prefix is blank, it can accept any calls. If all prefixes don’t match, the call will be rejected.
Phone Number	When enable route by from, the channel will only accept the call which caller ID is input in phone number.
Account	SIP registration account.
Password	The password of SIP registration account.
Status	The status of registration. When gateway is registered with softswitch, it will show ready.

Table 3.3-3

3.4 GoIP Settings

3.4.1 Port Settings

Basic Settings

You can select the frequency band and lock operator here. If SIM is not show in the port status page after inserting, please enable Unnormal SIM Supp.

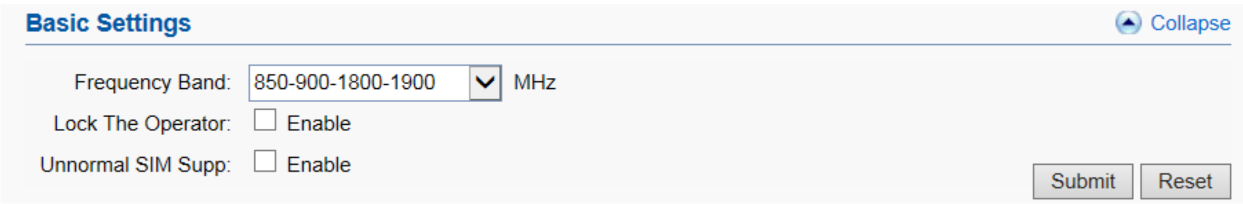


Figure 3.4.1-1 Basic Settings

Hardware Properties

Hardware Properties Collapse

Port	Enable SIM Card	Mobile Base	Provider	Input Vol	Output Vol	IMEI
1	☑A ☑B ☑C ☑D	0	46001	3	12	863835021317644
2	☑A ☑B ☑C ☑D	0	46001	3	12	863835021316851
3	☑A ☑B ☑C ☑D	0	46001	3	12	863835021317149
4	☑A ☑B ☑C ☑D	0	46001	3	12	863835021316166
5	☑A ☑B ☑C ☑D	0	46001	3	12	863835021316331
6	☑A ☑B ☑C ☑D	0	46001	3	12	863835021317537
7	☑A ☑B ☑C ☑D	0	46001	3	12	863835021316927
8	☑A ☑B ☑C ☑D	0	46001	3	12	863835021316653
9	☑A ☑B ☑C ☑D	0	0	3	12	863835021316828
10	☑A ☑B ☑C ☑D	0	0	3	12	863835021316588
11	☑A ☑B ☑C ☑D	0	0	3	12	863835021317172
12	☑A ☑B ☑C ☑D	0	0	3	12	863835021317180
13	☑A ☑B ☑C ☑D	0	0	3	12	863835021313635
14	☑A ☑B ☑C ☑D	0	0	3	12	863835021317685
15	☑A ☑B ☑C ☑D	0	0	3	12	863835021316752
16	☑A ☑B ☑C ☑D	0	0	3	12	863835021316455

Figure 3.4.1-2 Hardware Properties

Items	Description
Port NO.	Gateway channel, starts from 1 to 16.
Enable SIM Card	The SIM is enabled with√, and disabled without √. You can enable or disable SIM by this button.
Mobile Base	The base station of SIM registered.
Input Vol	Input volume of module, unmodifiable value.
Output Vol	Output volume of module. unmodifiable value.
IMEI	International Mobile Equipment Identity of this module. This gateway support IMEI modification, you can do it on IMEI settings page.

Table 3.4.1-1

Port Application Feature

You can see the SIM card number and balance in this page (need enable billing system). “SMS Forward to” and “SMS Center” are the settings of SMS forward by GSM protocol. For example: SMS center number is +8613800755500 (local carrier SMSC), SMS forward To number is +8613715266978, when this port SIM receive a SMS, it will forward to +8613715266978.

Port Applicatioin Feature Collapse

Port No.	Main Access	Check Balance	Card Number	Balance	SMS Forward To	SMS Center
1	☐	☑		0.00	+8613715266978	+8613510844655
2	☐	☑		0.00		
3	☐	☑		0.00		
4	☐	☑		0.00		
5	☐	☑		0.00		
6	☐	☑		0.00		
7	☐	☑		0.00		
8	☐	☑		0.00		
9	☐	☑		0.00		
10	☐	☑		0.00		
11	☐	☑		0.00		
12	☐	☑		0.00		
13	☐	☑		0.00		
14	☐	☑		0.00		
15	☐	☑		0.00		
16	☐	☑		0.00		

Submit Reset

Figure 3.4.1-3 Port Application Feature

3.4.2 Base Stations

Basic Settings

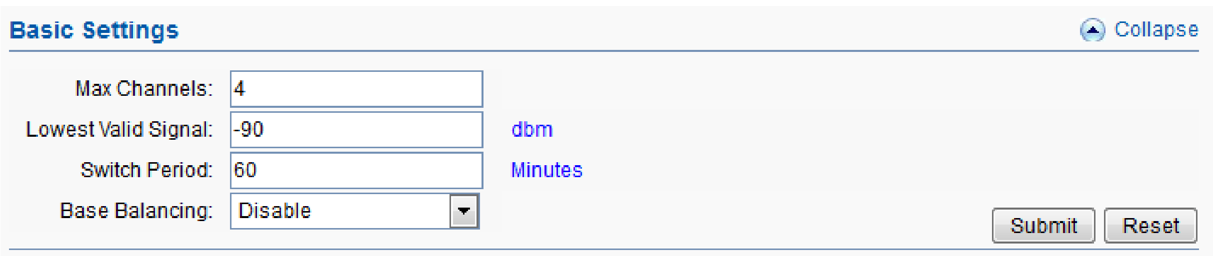
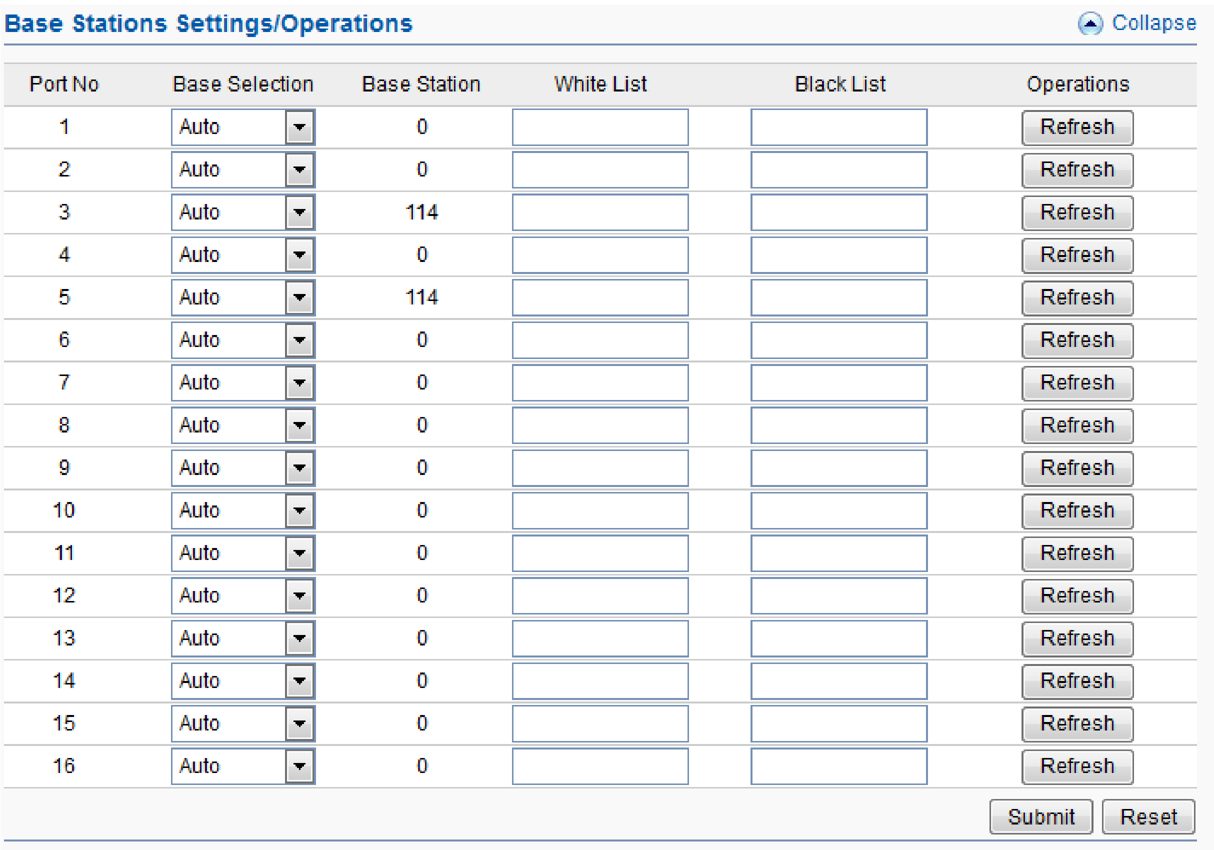


Figure 3.4.2-1 Basic Settings

Items	Description
Max Channels	The maximum number of base station
Lowest Valid Signal	The lowest valid signal of base station, the default value is -90 dbm. SIM card will not register in the base station which signal is lower than the value.
Switch Period	Base station switch period, the default value is 60 minutes. Base station will switch automatically by the period (when base selection is "poll").
Base Balancing	Disable: every channel will select the base station with best signal. We suggest this mode. Enable: every channel will try to select different base station.

Table 3.4.2-1

Base Stations settings/operations.


Base Stations Settings/Operations
Collapse

Port No	Base Selection	Base Station	White List	Black List	Operations
1	Auto	0			Refresh
2	Auto	0			Refresh
3	Auto	114			Refresh
4	Auto	0			Refresh
5	Auto	114			Refresh
6	Auto	0			Refresh
7	Auto	0			Refresh
8	Auto	0			Refresh
9	Auto	0			Refresh
10	Auto	0			Refresh
11	Auto	0			Refresh
12	Auto	0			Refresh
13	Auto	0			Refresh
14	Auto	0			Refresh
15	Auto	0			Refresh
16	Auto	0			Refresh

Submit Reset

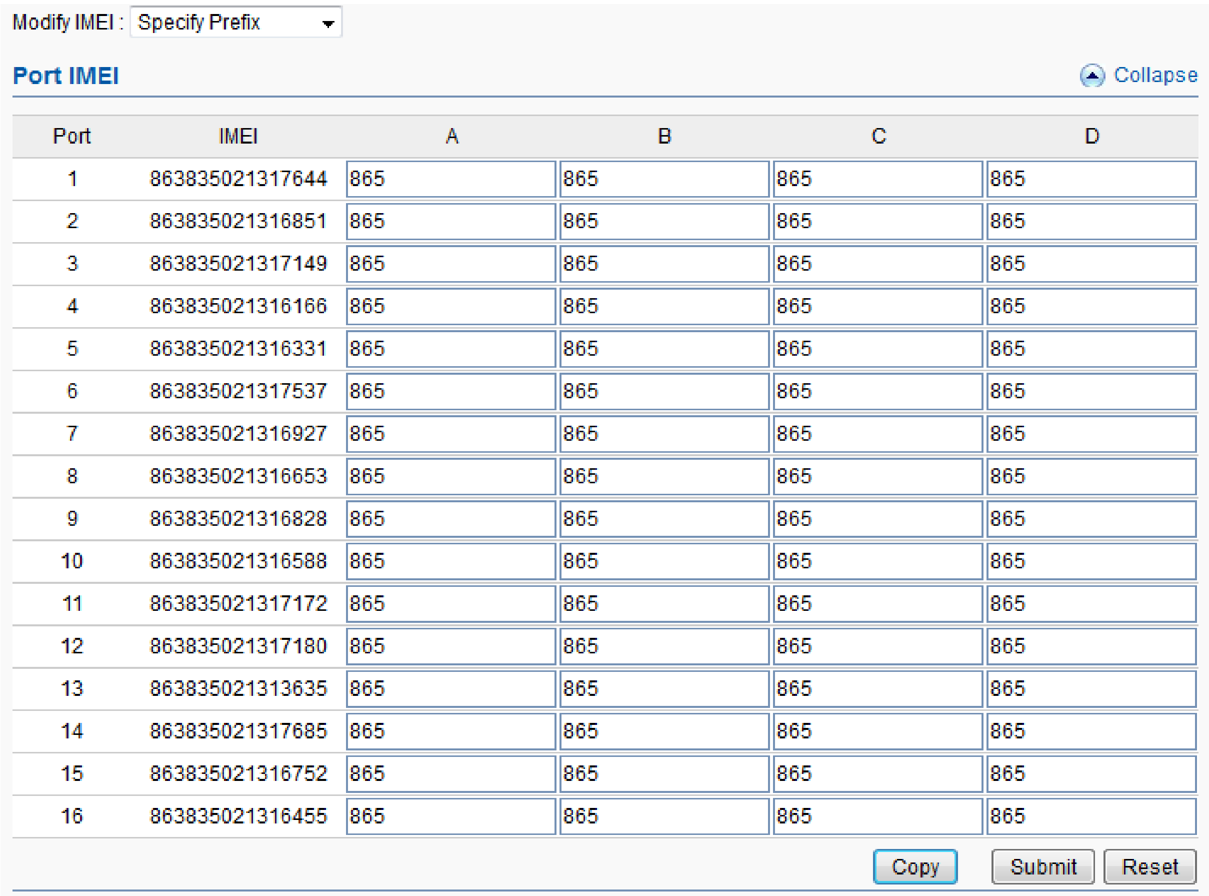
Figure 3.4.2-2 Base Stations Settings

Items	Description
Port NO.	Gateway channel, starts from 1 to 16.
Base Selection	Auto: every channel will select the base station automatically. Poll: base station will switch during every switch period, if set a base station in white list, it will be locked in this channel.
Base station	It will show the base station
White List	The base station white list, if you just put one base here and select “poll”, this channel will lock the base station.
Black List	The base station can’t be used if put in black list.
Operations	Refresh the base station information.

Table 3.4.2-2

3.4.3 IMEI Settings

IMEI means International Mobile equipment Identity, it is a 15-digit number. The gateway can do IMEI modification, it can protect SIM from blocking. With the function, you can do static IMEI or dynamic IMEI.


Modify IMEI : Specify Prefix

Port IMEI

Collapse

Port	IMEI	A	B	C	D
1	863835021317644	865	865	865	865
2	863835021316851	865	865	865	865
3	863835021317149	865	865	865	865
4	863835021316166	865	865	865	865
5	863835021316331	865	865	865	865
6	863835021317537	865	865	865	865
7	863835021316927	865	865	865	865
8	863835021316653	865	865	865	865
9	863835021316828	865	865	865	865
10	863835021316588	865	865	865	865
11	863835021317172	865	865	865	865
12	863835021317180	865	865	865	865
13	863835021313635	865	865	865	865
14	863835021317685	865	865	865	865
15	863835021316752	865	865	865	865
16	863835021316455	865	865	865	865

Copy Submit Reset

Figure 3.4.3-1 IMEI Settings

You can set any different IMEI for every port, just set 14-digit number, the last digit will generate itself. If you need set with special prefix, just click "copy" , you can see the figure as above: set 865 in port 1A, after click "copy", every port will have a IMEI prefix 865.

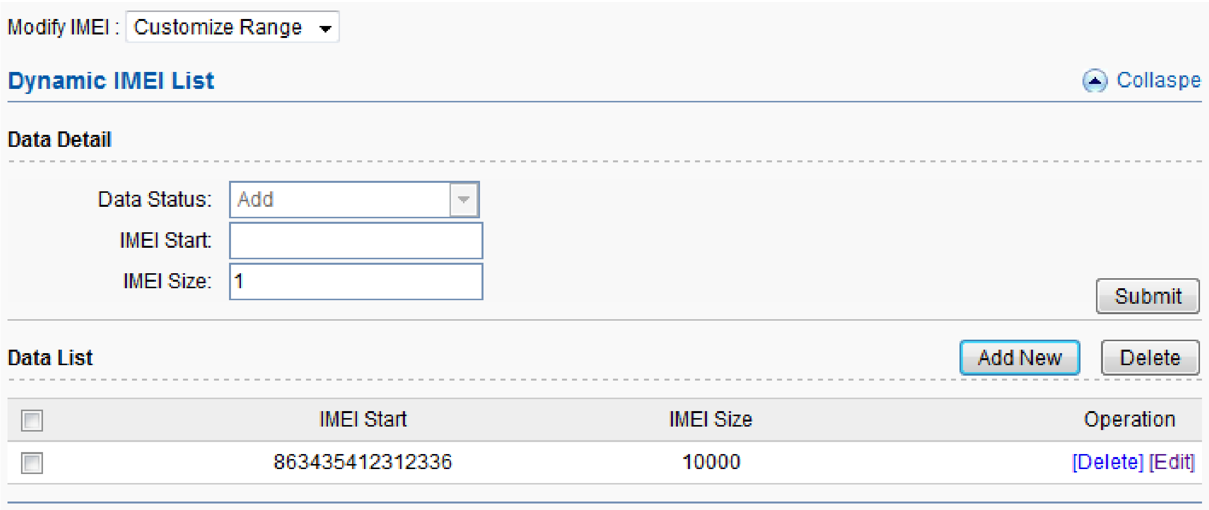


Figure 3.4.3-2 Dynamic IMEI Settings

You can click "Add New" button to add a new dynamic IMEI list, this list includes initial IMEI value of IMEI group and the size of IMEI group. click "Delete" will delete a exist IMEI list, if you want to change the settings of dynamic IMEI list, please click "Edit" button.

3.4.4 PIN Settings

PIN means personal identification number, it just like a password of SIM card, it can help to prevent SIM card from being stolen and improve security. Most SIM cards don't have PIN code. If a SIM card is with PIN, you need input PIN code in corresponding slot and enable "PIN Unblock", then the SIM card will work.

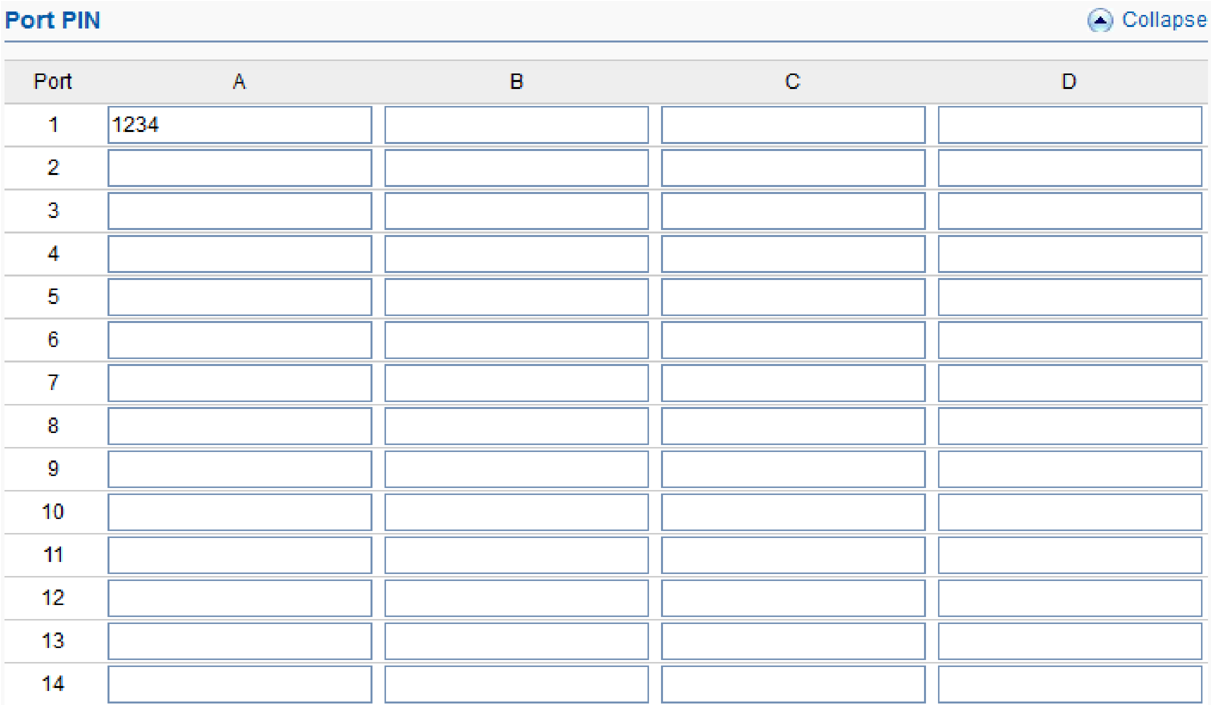


Figure 3.4.4-1 Basic Settings

3.4.5 SMS Send

Basic Settings

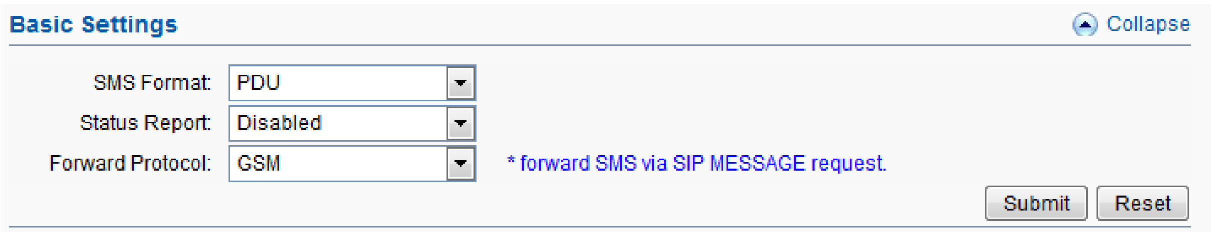


Figure 3.4.5-1 Basic Settings

Items	DesrIPtion
SMS Format	PDU and TXT.
Status Report	SMS status report. If it is enabled, after sending SMS successfully, it will get a status report from operator such as sending successfully.
Forward Protocol	GSM: forward SMS to another mobile by GSM forward protocol. (need set the SMS center number and receiver number) SIP: forward SMS to a server by SIP message request.(need set the server IP) HTTP: forward SMS to a server by http request, you need develop the server follow Ejoin HTTP SMS forwarding API.

Table 3.4.5-1

Scheduled Sending

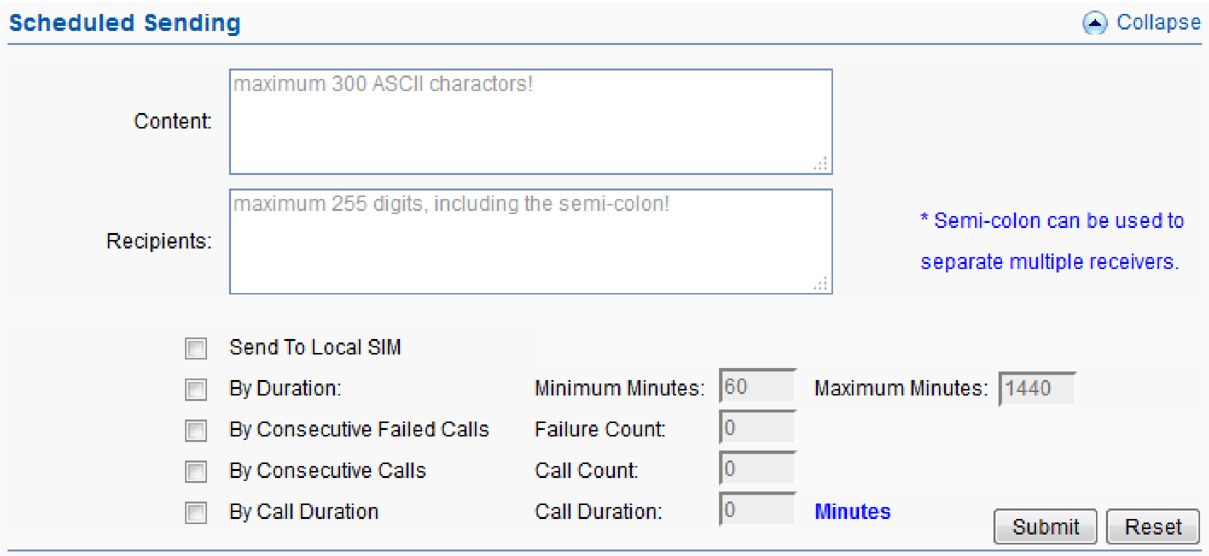


Figure 3.4.5-2 Scheduled Sending

Items	Description
Content	SMS content. The length is limited to 300 ASCII characters.
Recipients	The phone number of receiver. Semi-colon can be used to separate multiple receivers.
Send To Local SIM	Enable this button. Gateway will do inter-port SMS sending (need set SIM number in every channel first), it's random and by the condition below. For example: channel 1 sends SMS to port 3.
By Duration	SMS sending by device online time, and the time between minimum minutes and maximum minutes.
By Consecutive Failed Calls	SMS sending by consecutive failed calls.
By Consecutive Calls	SMS sending by consecutive calls.
By Call Duration	SMS sending by SIM call duration.

Table 3.4.5-2

Send SMS

You can select one or more ports to send SMS to different receiver. Successful and failed SMS records will be show below.

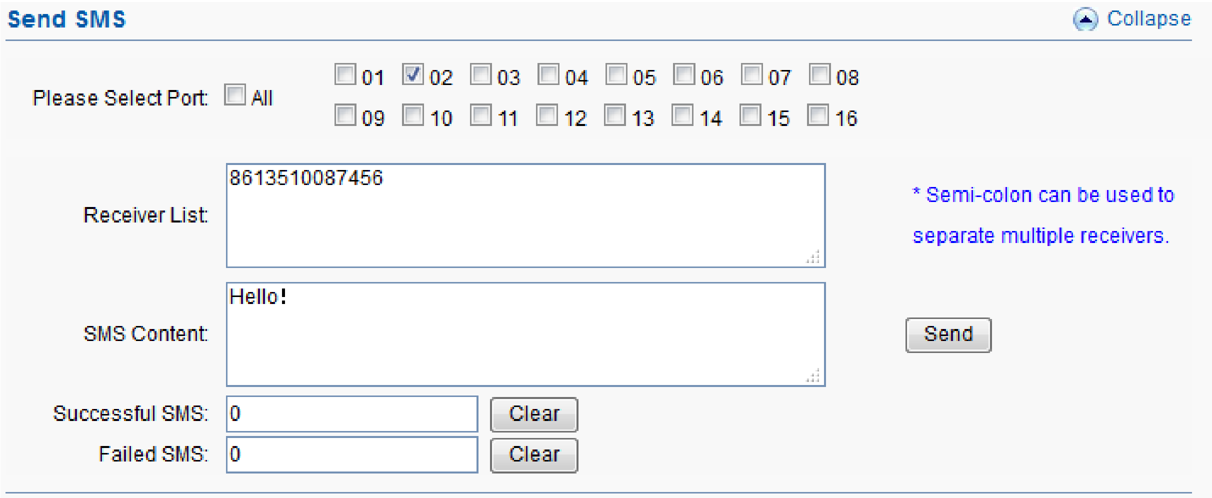


Figure 3.4.5-3 Send SMS

3.4.6 SMS Receive

You can check the latest SMS content and clean up all the SMS content on this page.

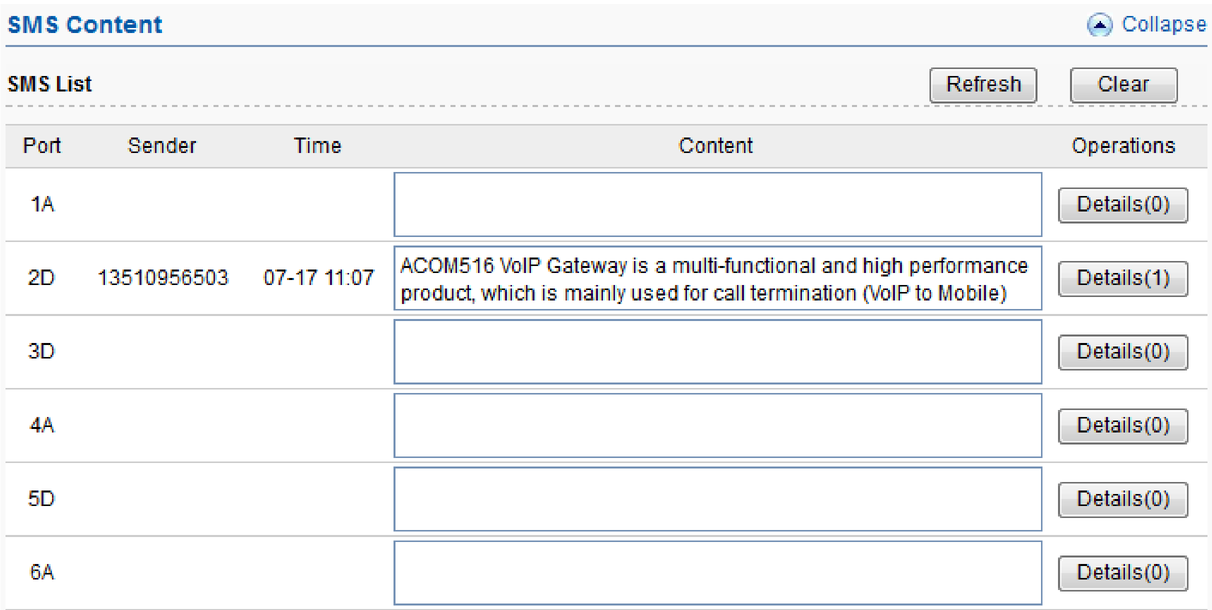


Figure 3.4.6-1 SMS Content

If you want to check more SIM content of this SIM, please click "Details(3)" button.

Then you will see the page below. You can know the SMS details in different port and SIM, reply and delete SMS here.

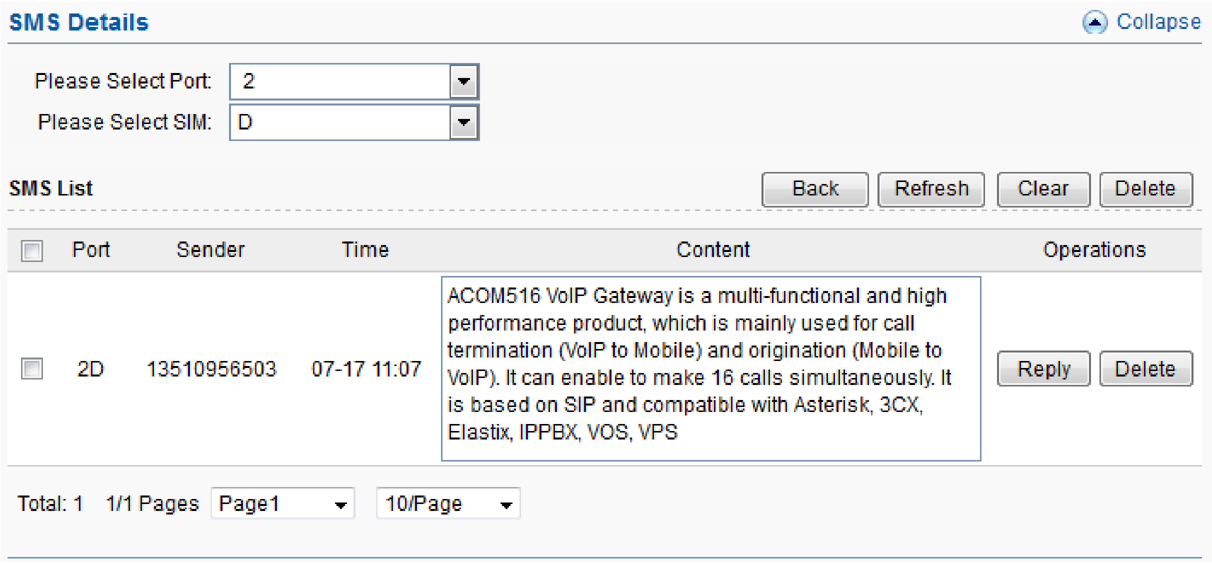


Figure 3.4.6-2 SMS Details

ACOM516-64 gateway has 16 channels; every channel has 4 SIM slots. If one channel is full with 4 SIM cards, only 1 SIM card is active, others are inactive. The inactive cards are used for switching; it may protect SIM cards from blocking.

3.4.7 Lock/Switch Card

Basic Settings

You can enable SMS warning on this page, which means you will get a SMS when SIM card is locked by device.

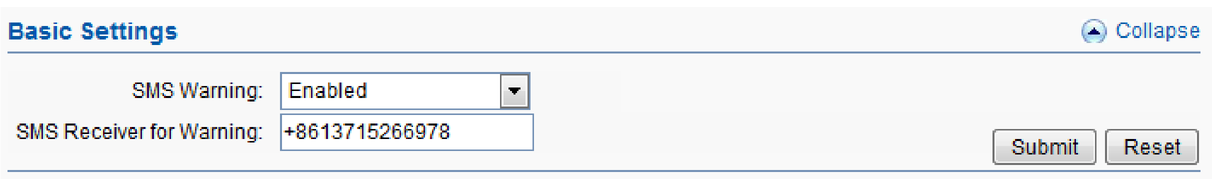


Figure 3.4.7-1 Basic Settings

Conditions for Locking Card

When the SIM excesses any condition below, gateway will lock/switch it.

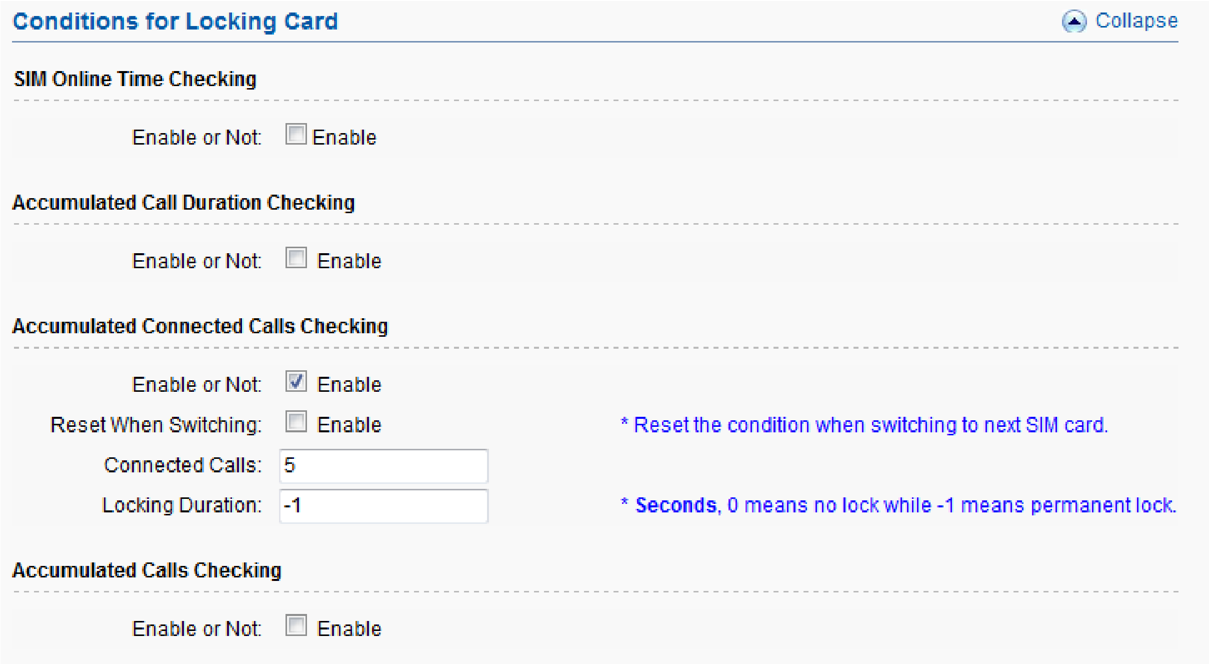


Figure 3.4.7-2 Locking Card Conditions

We take "consecutive failed calls checking" for example to explain the lock/switch function.

Items	Description
Enable or Not	If it is enabled, the consecutive failed calls will be used as a condition for system to check.
Reset When Switching	This condition will be recalculated next time when it is switched by other conditions. For example:
USSD Query	After switch to next SIM, the next SIM will send USSD query command first.
Failed Calls	The maximum number of consecutive failed calls on this SIM card. If the number of consecutive failed calls excesses this value, the card will be locked if this condition is enabled.
Locking duration	The duration of locking. 0 means no lock while -1 means permanent lock.

Table 3.4.7-1

If the SIM card is locked by gateway, it will show , it means locked by device. And you will also see the Description on running status >> call status page.

Lock/switch card conditions	Description on call status page
SIM Online Time Checking	Switch timer fired
Accumulated Call Duration Checking	Talk dur expired
Accumulated Connected Calls Checking	Talk num expired
Accumulated Calls Checking	Call num expired
Consecutive Failed Calls Checking	Failed call num expired
Consecutive No-Alert Calls Checking	Noalert num expired
Consecutive No-Answer Calls Checking	Noanswer num expired
Consecutive No Carrier Calls Checking	Nocarrier num expired
Consecutive Short-Duration Calls Checking	Shortdur num expired
Accumulated SMS Count checking	SMS num expired
Accumulated Failed SMS Count Checking	Failed SMS num expired
Consecutive Failed SMS Count Checking	Con-failed SMS num expired

Table 3.4.7-2

3.4.8 Port Inter-Calling

Port inter-calling is a good solution for protecting SIM from blocking. It's a human behavior feature.

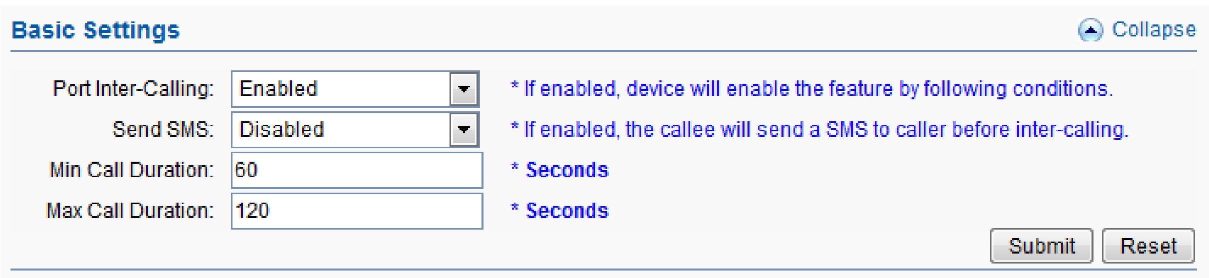


Figure 3.4.8-1 Basic Settings

Items	Description
Port Inter-Calling	The function will work if it is enabled. (need to set SIM number for every port first).
Send SMS	If it is enabled, the callee will send a SMS to caller before inter-calling
Min Call Duration	The minimum call duration when do port inter calling
Max Call Duration	The maximum call duration when do port inter calling. the call duration will between minimum and maximum duration.

Table 3.4.8-1

When enable this function, after excessing the condition below, the idle port will call each other random (need to set the SIM number for every port first).

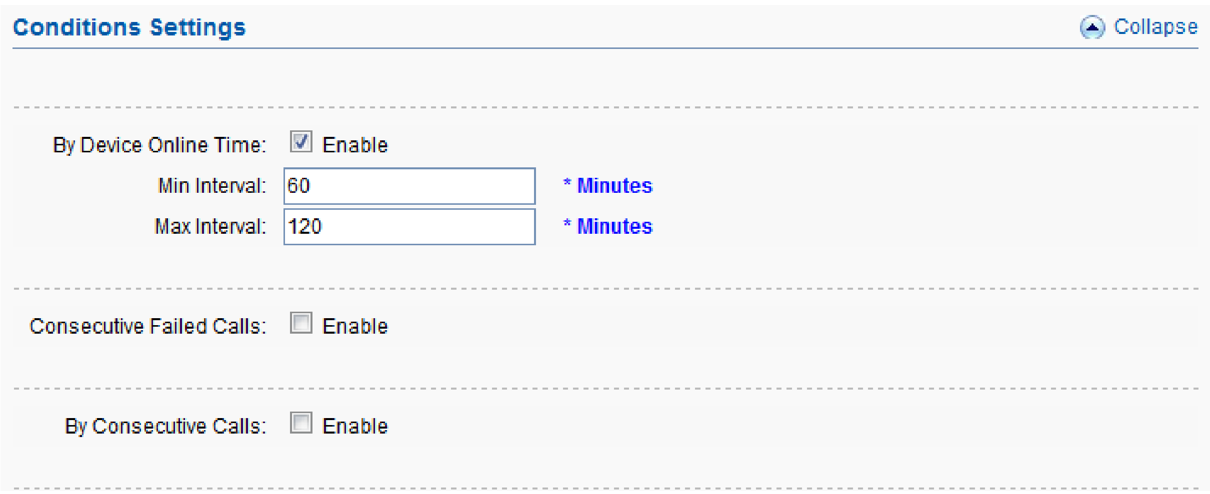


Figure 3.4.8-2 Conditons Settings

If you enable "Send SMS", you will see the page below.

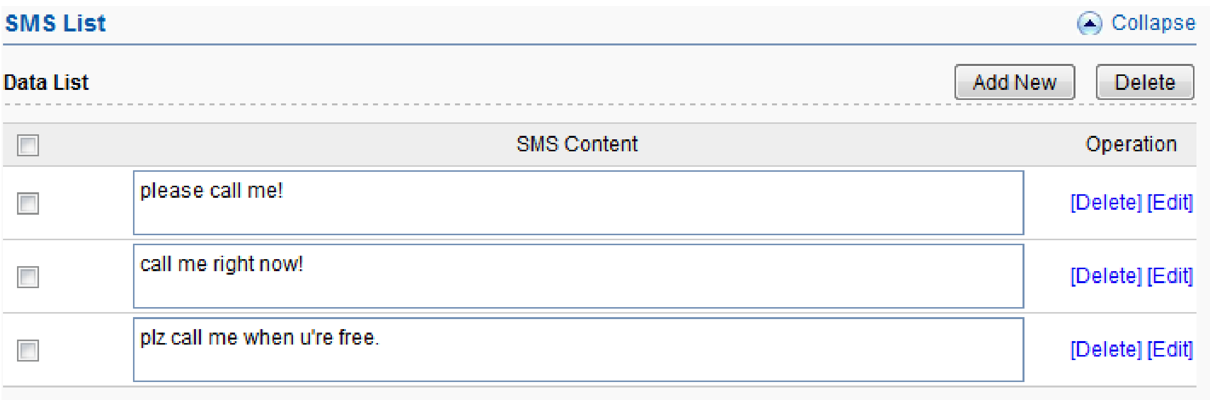


Figure 3.4.8-3 SMS List

The callee will select a SMS content first, then send to caller before inter calling, you can click "Add New" button to add new SMS content and delete or edit the SMS content.

3.4.9 SIM Num Settings

You should set the SIM number first before enable the inter port calling/SMS-sending. You can get SIM number by USSD or SMS automatically.

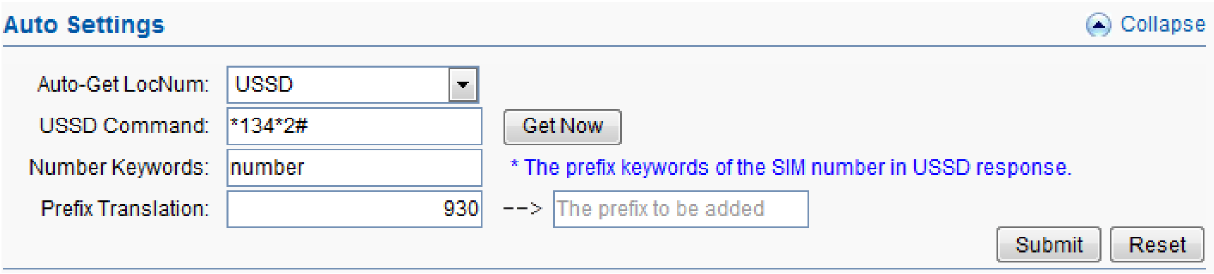


Figure 3.4.9-1 USSD Auto-Get LocNum Settings

Items	Description
Auto-Get LocNum	When choose USSD, the gateway will get the SIM number by USSD
USSD Command	The USSD command for querying SIM number.
Number Keywords	The prefix keywords of the SIM number in USSD response. For example: the USSD response is your SIM number 923345556978, then keyword is number, it is usually the word before SIM number.
Prefix Translation	If you get the number is 923345556978, but you don't need a country code, you can do prefix translation, delete 923 then add 0.

Table 3.4.9-1

The page below shows the setting of getting number by SMS, it is same as USSD, you should send the SMS content to the operator to get the SIM number.

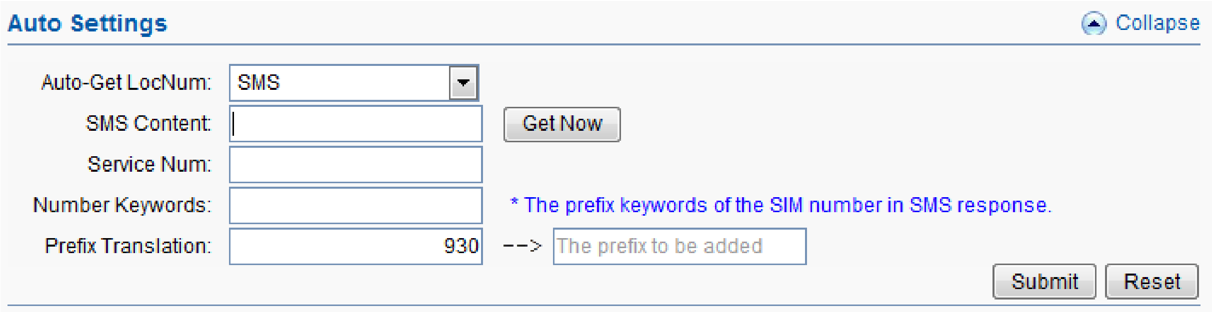


Figure 3.4.9-2 SMS Auto-Get LocNum Settings

If you can’t get the SIM number by USSD or SMS, you need set the SIM number manually.

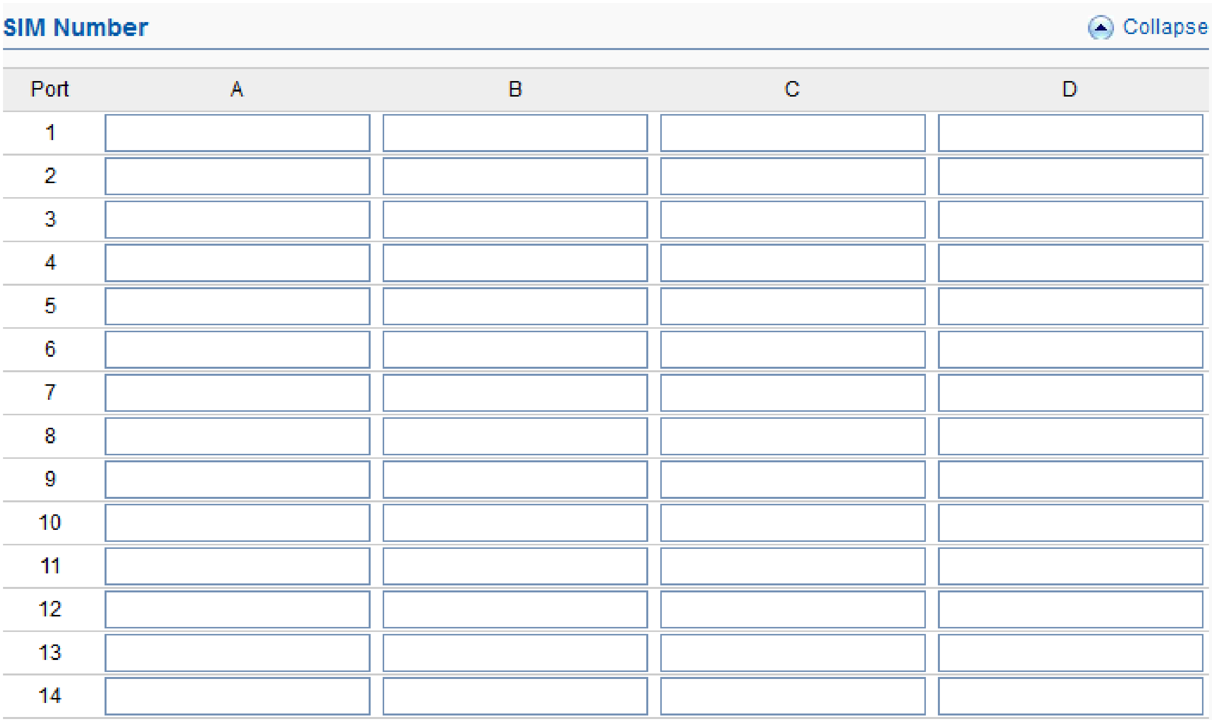


Figure 3.4.9-3 SIM Number

3.4.10 AT Command

Module Operations

You can select different module and do the operations of restart, stop and start.

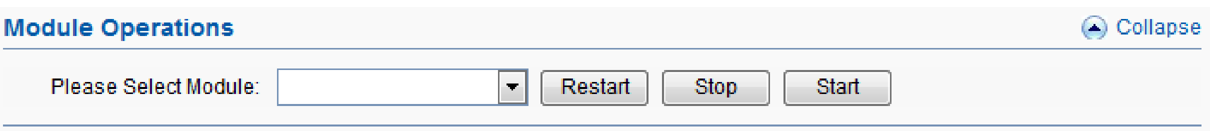


Figure 3.4.10-1 Module Operations

Command Operation

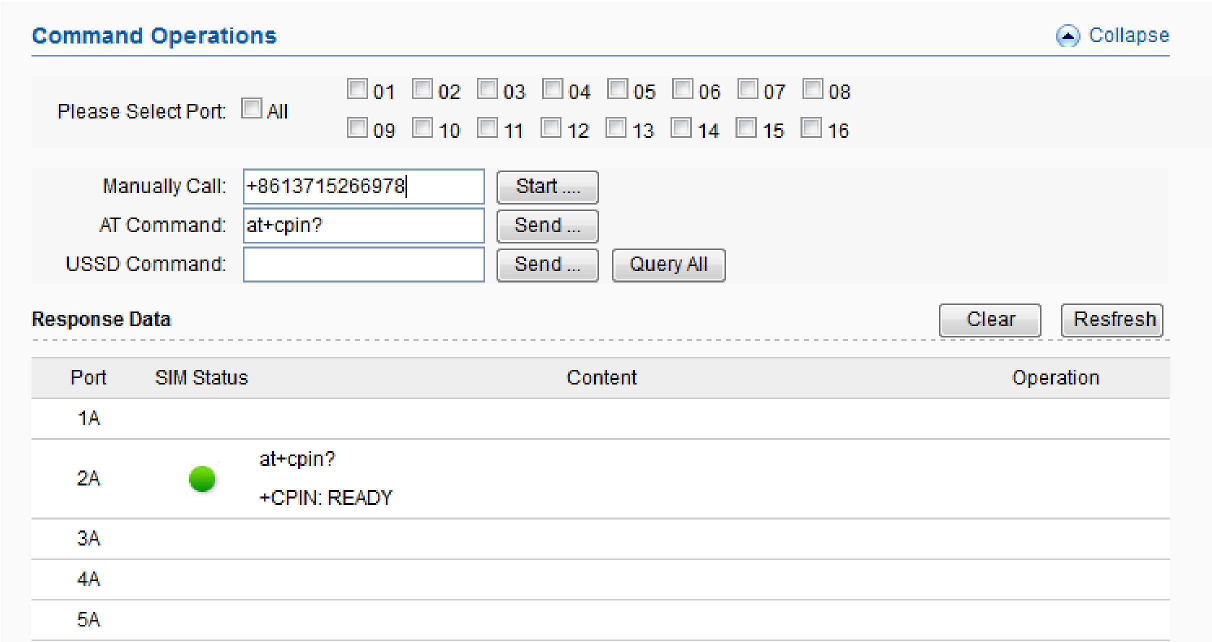


Figure 3.4.10-1 Command Operations

Items	Description
Select port	Select port to do command operations.
Manually call	Check the SIM can send a call or not.
AT Command	AT command to check SIM status.
USSD command	It's for querying balance, number and recharge etc.
SIM status	Display the SIM status.
Content	The response after sending USSD/AT command.

Table 3.4.10-1

3.4.11 USSD Command

On this page, you can send USSD command and get USSD response more convenient.

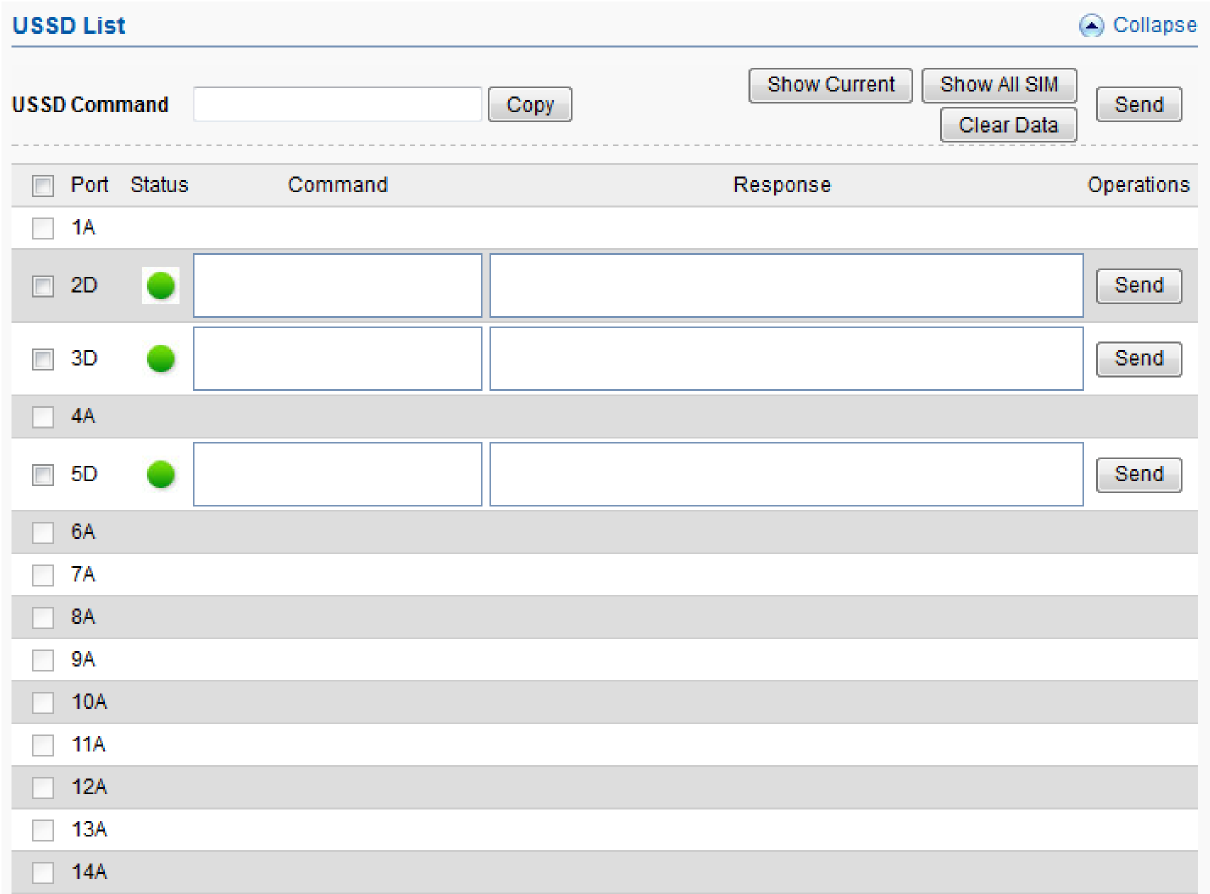


Figure 3.4.11-1 USSD List

Items	Description
Copy	Copy the USSD command to other channel.
Show Current	Display the active SIM cards.
Show ALL SIM	Display all SIM cards.
Clear Data	Clear the USSD response.
Send	Execute the USSD command.

Table 3.4.11-1

3.4.12 Billing Settings

This is the billing system page, this billing system is widely used in querying balance automatically which can remind customers to recharge or replace the no balance SIM cards. The theory of this billing system: every SIM card will get an accurate balance from USSD or SMS response, then the system will deduct money in every billing period by tariff which you set, so it may take some deviation.

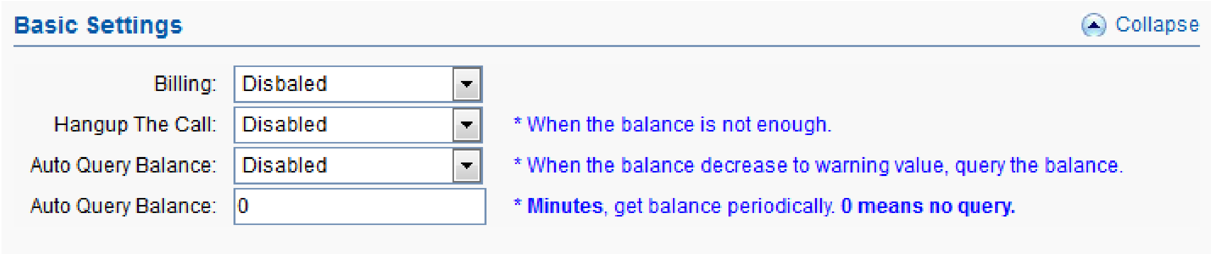


Figure 3.4.12-1 Basic Settings

Items	Description
Billing	Enable it, the billing system will be up.
Hangup The Call	If it is enabled, the call will be hang up when the balance is lower than invalid balance value.
Auto Query Balance	If it is enabled, it will query the balance when lower than caution balance value.
Auto Query Balance	Get balance periodically, it may be more accurate.

Table 3.4.12-1

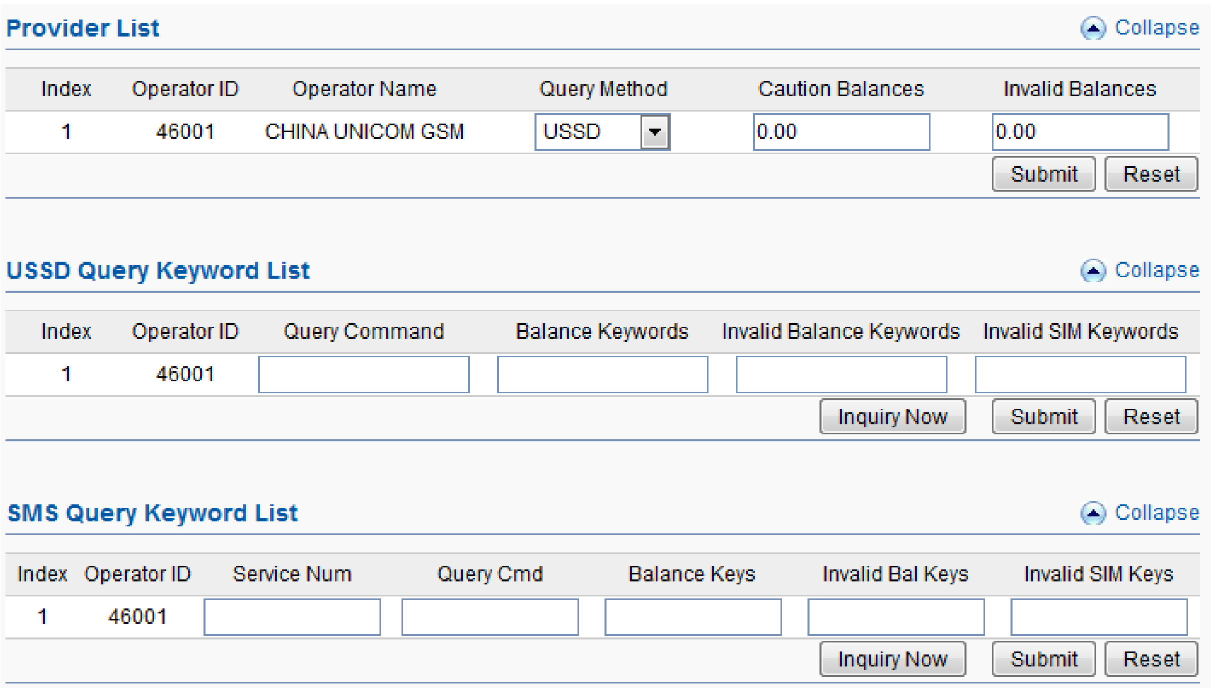


Figure 3.4.12-2 Related Settings

Items	Description
Query Method	USSD or SMS for querying balance
Caution Balances	When the balance is lower than caution balance value, the billing system will send a USSD or SMS to recalibrate balance.
Invalid Balances	The SIM can’t be used if it is lower than invalid balance value and it will show No Balance
Query Command	The HTTP or SMS command for querying balance
Balance Keywords	The balance keywords in USSD or SMS response. For example: your credit balance is AED 45.82. then AED can be the keywords
Invalid Balance Keywords	Can’t get balance from invalid balance keywords.
Invalid SIM Keywords	If the SIM is blocked by operator, it may get another response like: sorry, your SIM is blocked now. then you can set blocked as a invalid SIM keywords. The card will show
Service Num	The operator number, it will send SMS back to you.
Query Cmd	SMS command for querying balance
Balance Keys	Same as Balance keywords.
Invalid Bal Keys	Same as USSD.
Invalid SIM Keys	Same as USSD.

Table 3.4.12-2

Click “Add New” button, you can set a tariff list with different destination prefix. ”x ” means for all prefix. You can also do the operations of delete and edit here.

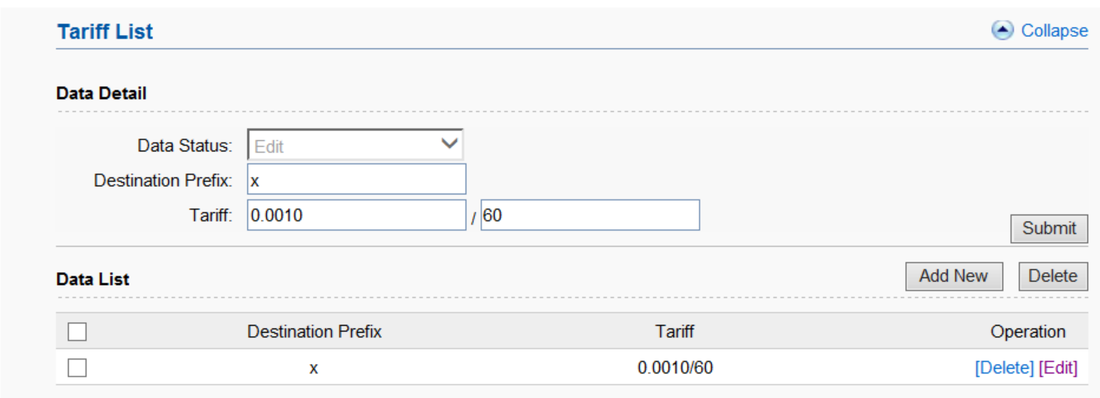


Figure 3.4.12-3 Tariff List

3.4.13 Call Dur. Control

Call duration control is for users to control the SIM using time. And the data will not flush even you restart the device or pull off the SIM.

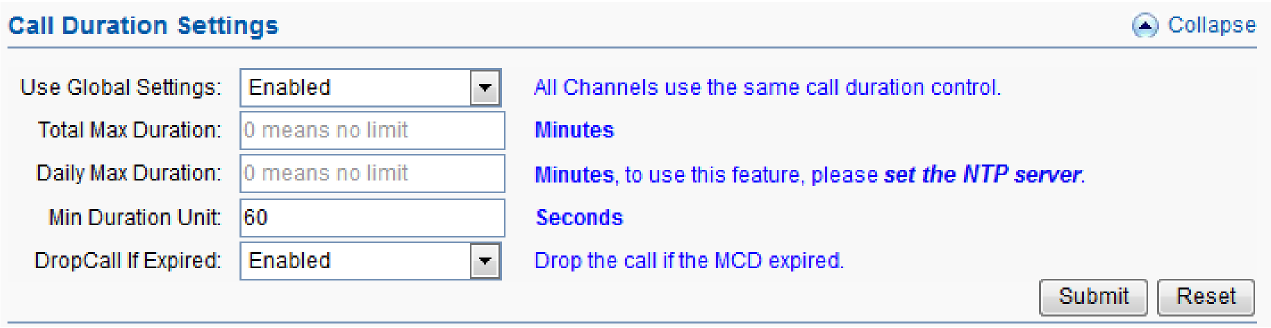


Figure 3.4.13-1 Call Duration Settings

Items	Description
Use Global Settings	Enable: all channels use same call duration limitation. Disable: you can set different call duration limitation for single channel.
Total Max Duration	The value of limitation. After the call duration excesses this value, the SIM will be locked by device. 0 means no limit.
Daily Max Duration	The value of limitation. After the daily call duration excesses this value, the SIM will be locked by device. 0 means no limit.
Min Duration Unit	Operator charging time, when the call is over this time, operator will collect fees. For example: china mobile charge per minute, the min duration unit will be 60 seconds.
Drop Call If Expired	Enabled: calls will be dropped after the SIM excesses call duration time. Disabled: calls will not drop.

Table 3.4.13-1

You can scan more details about the call duration control on the page below. Once the SIM is used up, it will be locked by gateway. If you still want to use it, you need to click “Reset”.

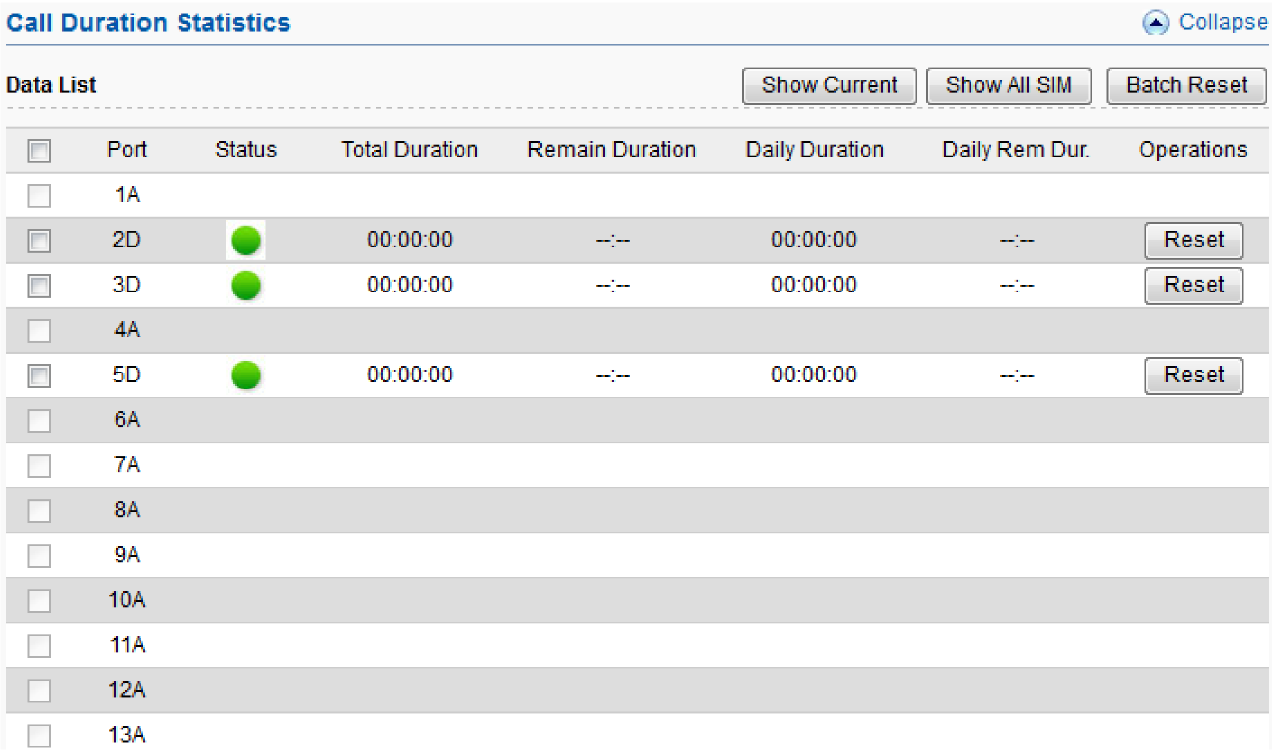


Figure 3.4.13-2 Call Duration Statistics

Items	Description
Total Duration	The value of total duration
Remain Duration	Indicates the current SIM remain time.
Daily Duration	The value of Daily Duration
Daily Rem Dur.	Indicates the current SIM daily remain time
Reset	The call duration will reset to the initial value. (daily cal duration will reset every day)

Table 3.4.13-2

If you need every channel has different call duration (single call duration control), please disable use global settings, and then you will see the page below.

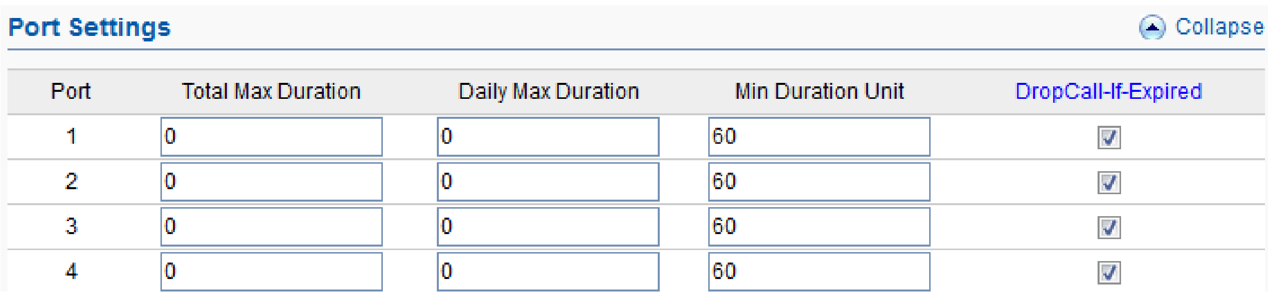


Figure 3.4.13-3 Port Settings

3.5 Application Settings

3.5.1 Phone Book

When you need other SIP server to send traffic to this gateway, you can add server details in phone book. But make sure it's the point to point mode. Click "Add New" button, setting the server details here. You can also delete and edit phone book list.

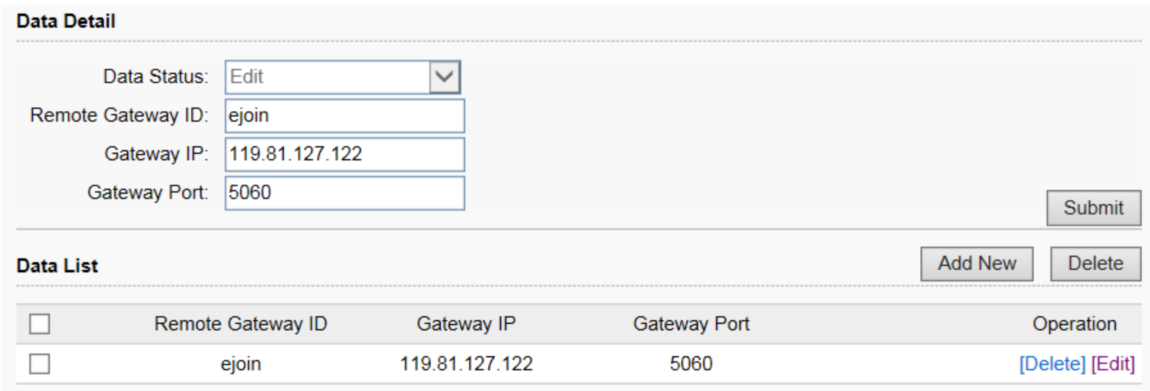


Figure 3.5.1-1 Phone Book List

3.5.2 Dial Plan

The dial pattern string is a normal regular expression. For example: The pattern 90[1-4] means the dialed number start with 90 and end with anyone of 1/2/3/4. So like the input 901,902,903 or 904 all can be accepted.

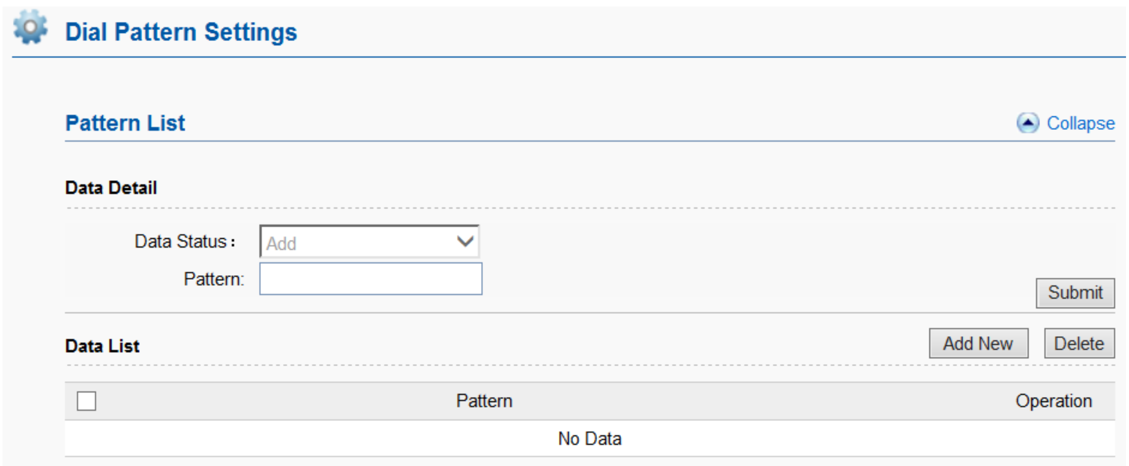


Figure 3.5.2-1 Dial Pattern Settings

3.5.3 Number Translation

Number translation is apply to GSM ->> IP calls.

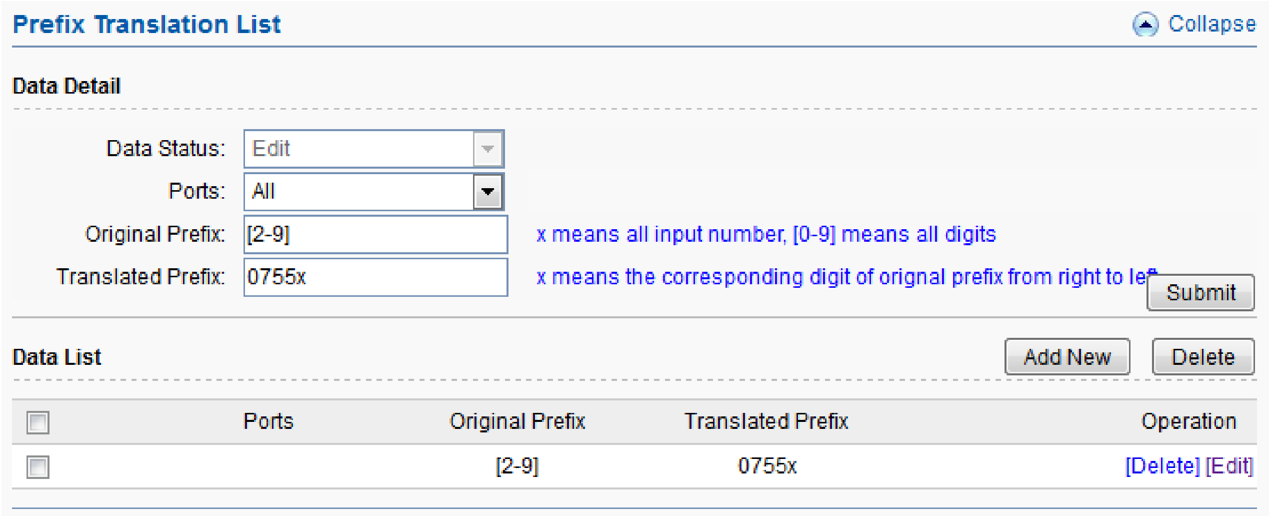


Figure 3.5.3-1 Prefix Translation List

Taking the figure above as an example, calling the SIM in gateway, you will hear an IVR: please dial a number, if you dial 85245166, it will be translated to 075585245166.

3.5.4 Incoming Translation

Incoming translation is apply to IP->>GSM calls. When you send traffic to the gateway, you can do the callee number translation here.

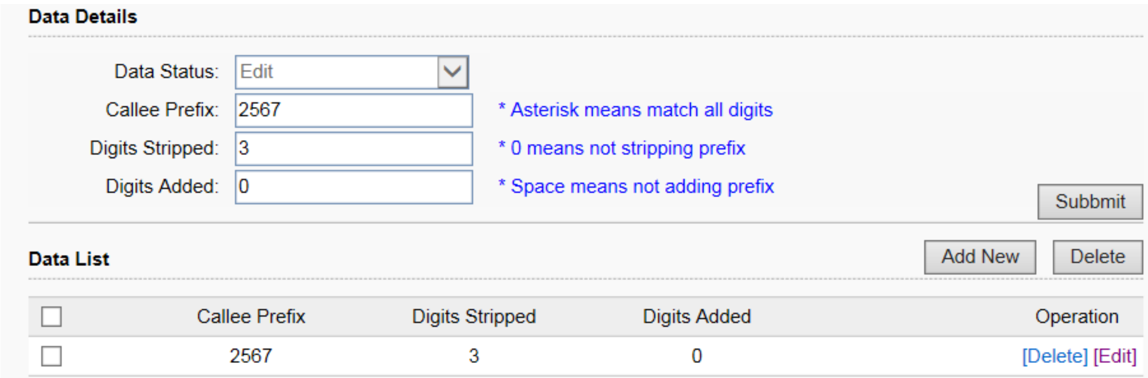


Figure 3.5.4-1 Translation List

Taking the figure above as an example, the callee number is 25670123456, it is with prefix 2567, the system will stripped 3 digits, then add 0, the callee number will be translated to 070123456.

Caller ID Hidden

If you want to hide caller ID, please enable caller ID hidden then input the dial prefix. (Note: Need operators support with this function.)

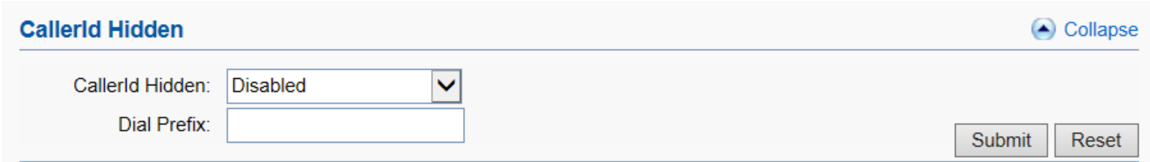


Figure 3.5.4-2 CallerId Hidden

3.5.5 Incoming Black List

You can forbid some calls by incoming black list.

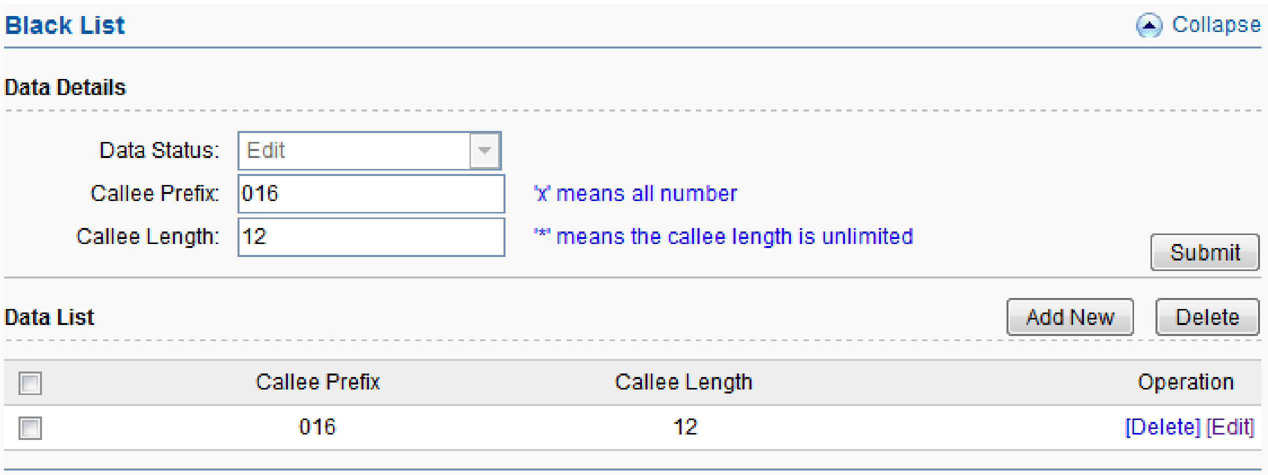


Figure 3.5.5-1 Incoming Black List Settings

Taking the figure above as an example, if the callee id like 016xxxx, and the length is 12 digits, these calls will be rejected by the gateway.

3.5.6 Incoming White List

Incoming white list is base on black list.

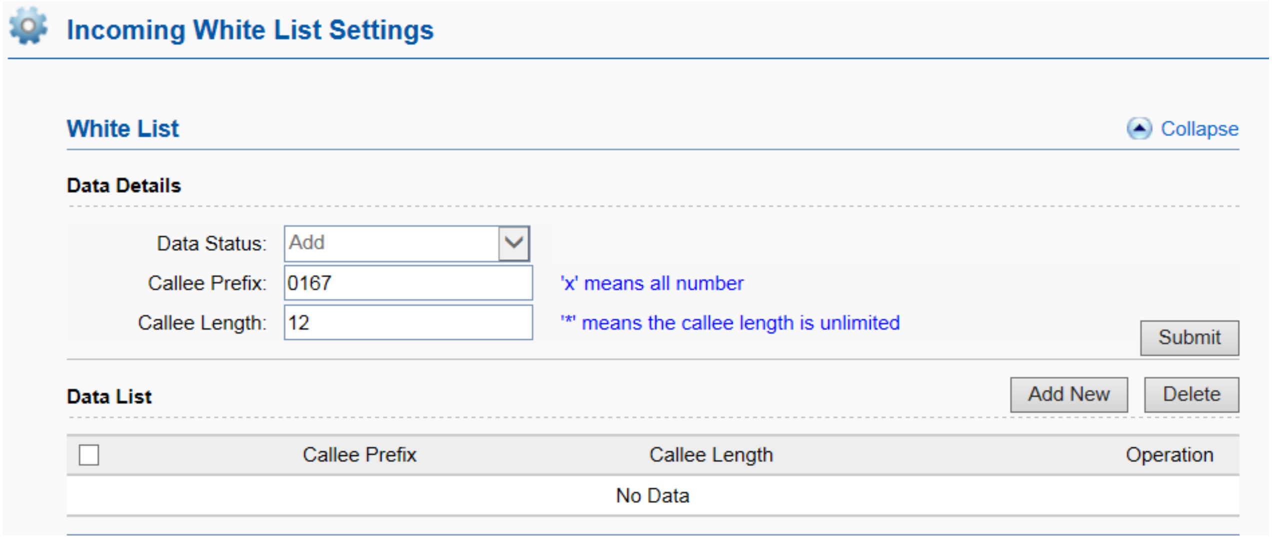


Figure 3.5.6-1 Incoming White List Settings

Taking the figure above as an example, just these number begain with 0167 and long 12 digits can through this gateway when you set the "Incoming Black List" like above.

3.5.7 SIM Pool Settings

When you want to manage SIM cards remotely or intensively, you can use this function.

Figure 3.5.7-1 SIM Pool Settings

Items	Description
SIM Pool	When you enable it, cards on gateway will be disabled, it can just use these cards on SIM Pool.
Registration	Enable: connect to SIM center. Disable: connect directly to SIM pool.
Server Address	SIM center or SIM pool address.
Username	The GOIP account in SIM center
Password	The password of GOIP account in SIM center.
Status	Show the gateway registration status.
Use Local Policy	If it is enabled, the policy of page lock/switch card can be used in SIM Pool.

Table 3.5.7-1

3.5.8 Auto Recharge

Auto recharge is based on billing system, if you want to do auto recharge, please configure billing system first.

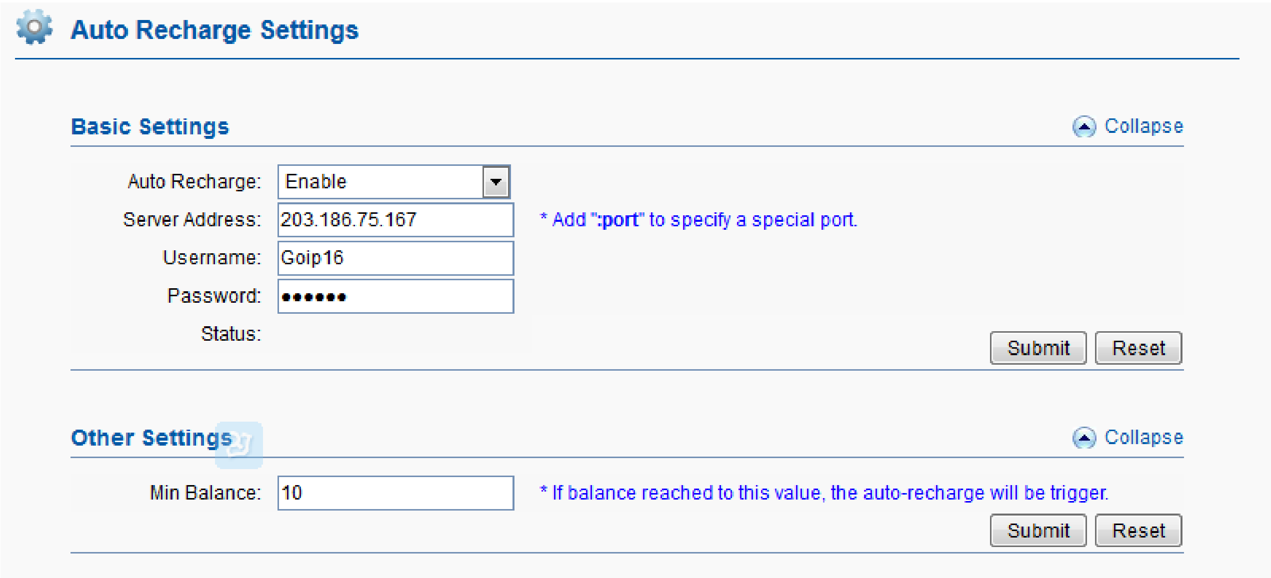


Figure 3.5.8-1 Auto Recharge Settings

Items	Description
Auto Recharge	Auto recharge will work when enable it.
Server Address	The auto recharge server address. (the server with EJOIN ear system)
Username	It is created in EJOIN ear system.
password	It is created in EJOIN ear system.
status	Show the registration status.
Min balance	If the balance is lower than the value, the ear system will do auto recharge.

Figure 3.5.8-1

3.5.9 State Notification

With this function, device will send state notification which includes registration status, SIM status and CDR to the server (Ejoin ein system).

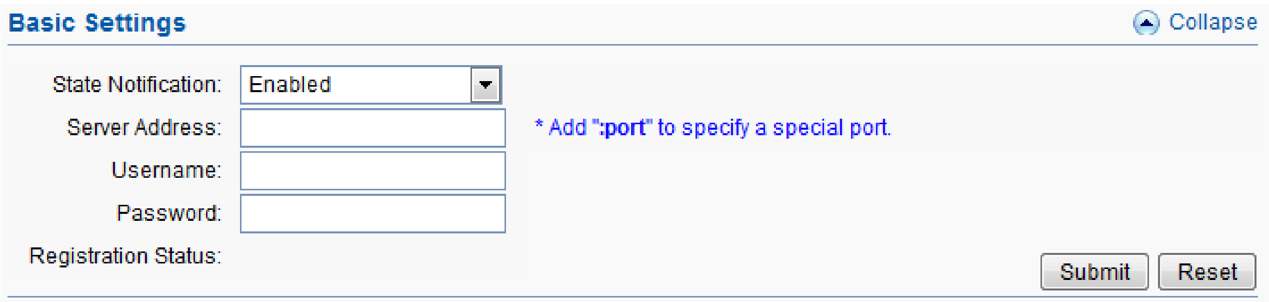


Figure 3.5.9-1 Basic Settings

Items	Description
State notifcation	If it is enabled, device will send state notification to the server.
Server address	The server which can get state notification.(need install EJOIN ein system)
Username	The device account in ein system.
password	The password of account in ein system.
Registration Status	Show the registesration status.

Table 3.5.9-1

3.6 Advanced Setting

3.6.1 Network settings

PPTP-VPN settings

A virtual private network (VPN) extends a private network across a public network, such as the Internet. It enables a computer or network-enabled device to send and receive data across shared or public networks as if it were directly connected to the private network, while benefiting from the functionality, security and management policies of the private network. This device works as VPN(PPTP) client mode only, if you want to use VPN function, please input the VPN parameter on the PPTP-VPN settings page.

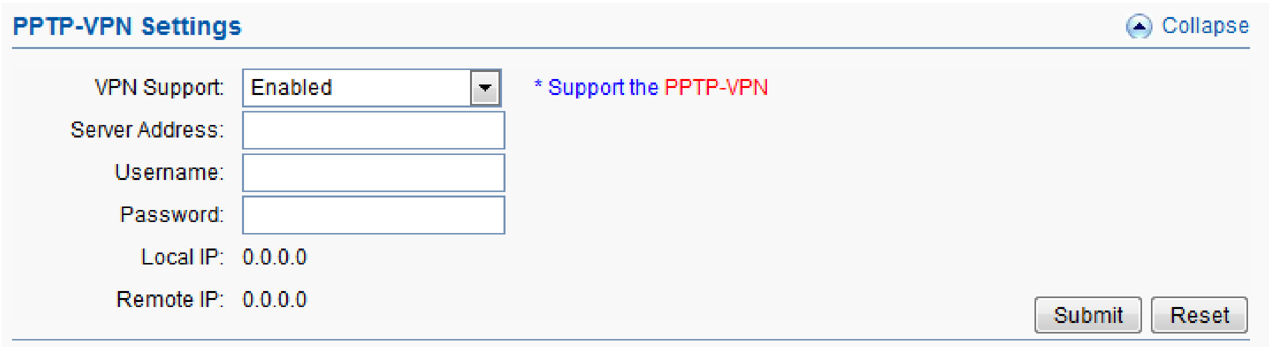


Figure 3.6.1-1 PPTP-VPN Settings

Network Settings

There are three ways to access the device: web, telnet and serial. web default port is 80, telnet is 23 and serial is the com port you insert. Web configuration is widely used in this device. If you want to change web and telnet default port, please input new port on this page.

Figure 3.6.1-2 Network Management Settings

3.6.2 Port Settings

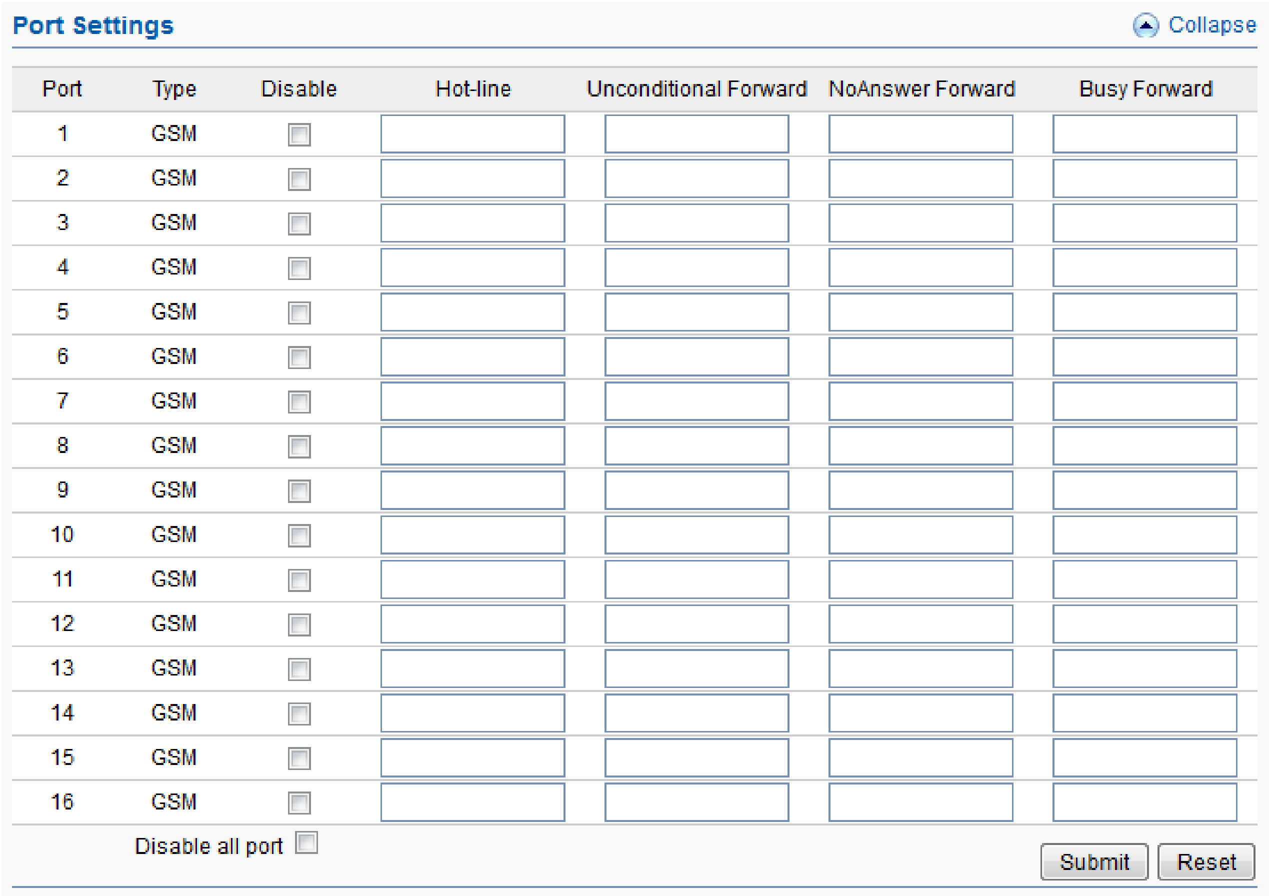


Figure 3.6.2-1 Port Settings

Items	Description
Type	Indicates the current type of network GSM/CDMA/WCDMA..
Disable	If it is disabled, this channel will be locked by gateway.
Hot-line	When GSM part client call to this channel, gateway will auto forward to the hot-line (Mobile to VoIP). Leave it blank if you don't need this function.
Unconditional Forward	When GSM part client call to this channel, gateway will forward the call to another mobile unconditionally.
No Answer Forward	When GSM part client calls to this channel, if this channel is no answer, gateway will forward the call to another mobile.
Busy Forward	When GSM part client call to this channel, if this channel is busy, gateway will forward the call to another mobile.

Table 3.6.2-1

3.6.3 Voice and Codec

Voice and Codec Settings

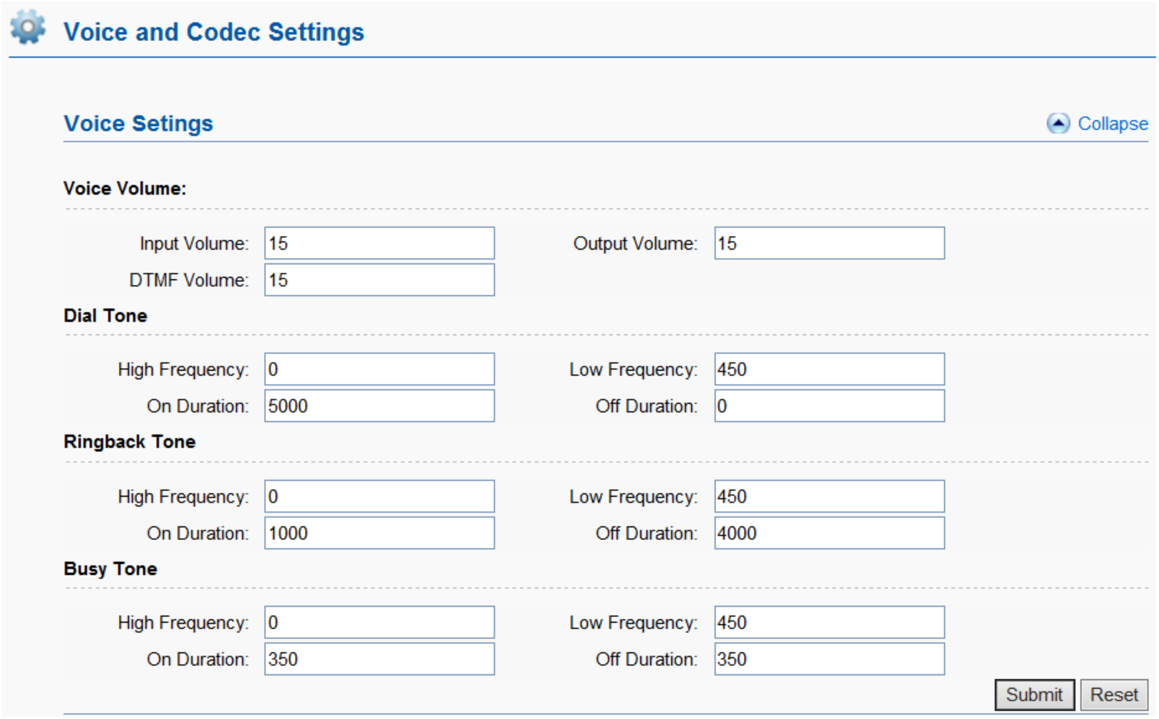


Figure 3.6.3-1 Voice and Codec Settings

Items	Description
Voice Volume	The DSP volume. the value range is 10-40. Input volume is on IP side and output volume is on GSM side. You can adjust volume here.
Dial Tone	The dial tone is sent to a customer or operator to indicate that the receiving end is ready to receive dial pulses or DTMF signals. It is used in all types of dial offices when the customer's or operator's dials produce dial pulses. Usually adopt the default settings.
Ringback Tone	The ring back tone(or ringing tone) is an audible indication that can be heard on caller side while the callee side phone is ringing. Normally, it is a repeated tone, designed to assure the caller that the callee side phone is ringing. Usually adopt the default settings.
Busy Tone	The busy tone indicates that the called customer's line has been reached but that it is busy, being wrong, or on permanent signal. When an operator applies a busy signal, it is sometimes called a busy-back tone. Line Busy Tone is a low tone that is on and off every 0.5 second. Usually adopt the default settings.

Table 3.6.3-1

Voice Codec Priority

You can click “Up” or “Down” to adjust the codec priority.

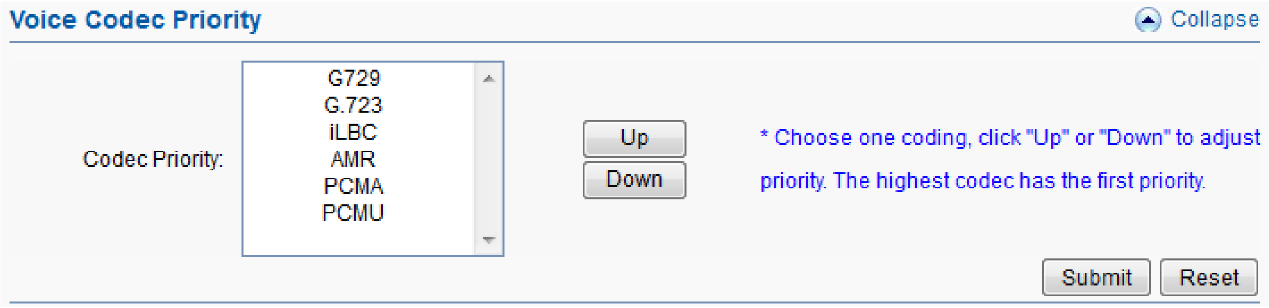


Figure 3.6.3-2 Voice Codec Priority

3.6.4 Callback Settings

Callback function, when you dial the SIM in gateway with mobile phone, it will hang up soon and send a call back to you, after you pick up the call, you can dial a VoIP extension or another phone number. If you want to use this function, please enable it and set the callback numbers.

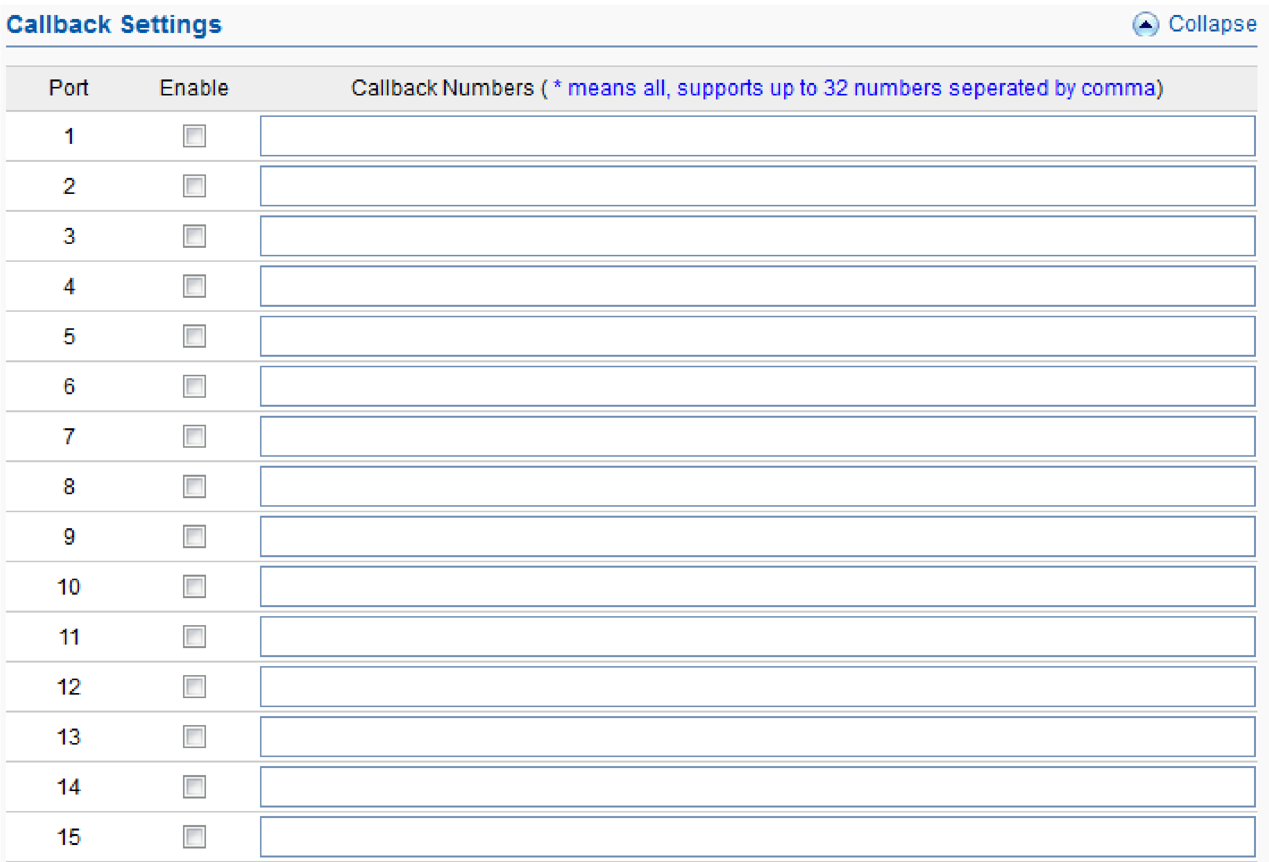
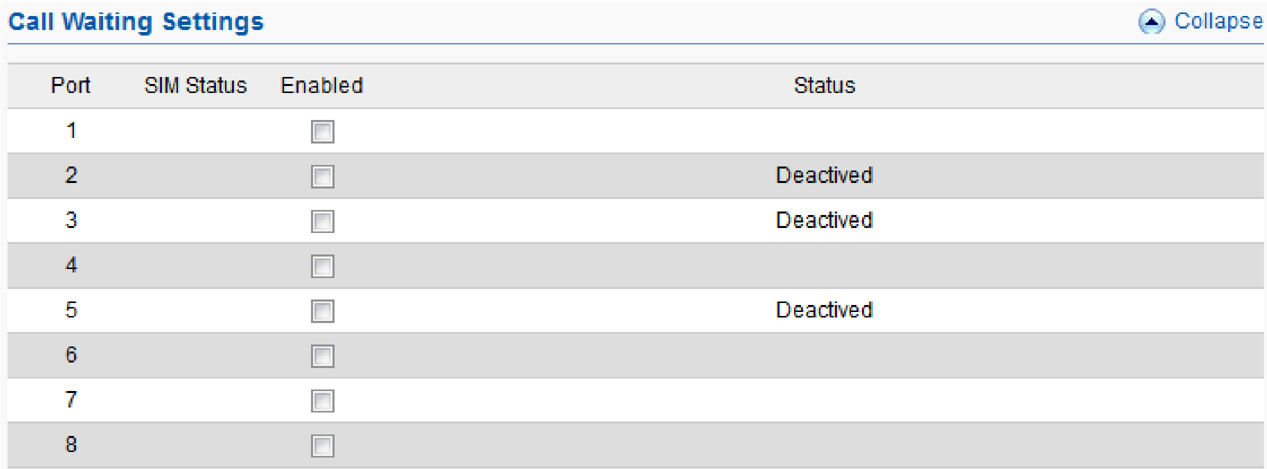


Figure 3.6.4-1 Callback Settings

3.6.5 Callwait Settings

Call waiting is a feature supported by SIM carrier, when there is a second call dialing into this SIM card, there will be waiting tone instead of hang up. You can enable it when you need this feature.



Port	SIM Status	Enabled	Status
1		☐	
2		☐	Deactived
3		☐	Deactived
4		☐	
5		☐	Deactived
6		☐	
7		☐	
8		☐	

Figure 3.6.5-1 Call Waiting Settings

3.6.6 Other Settings

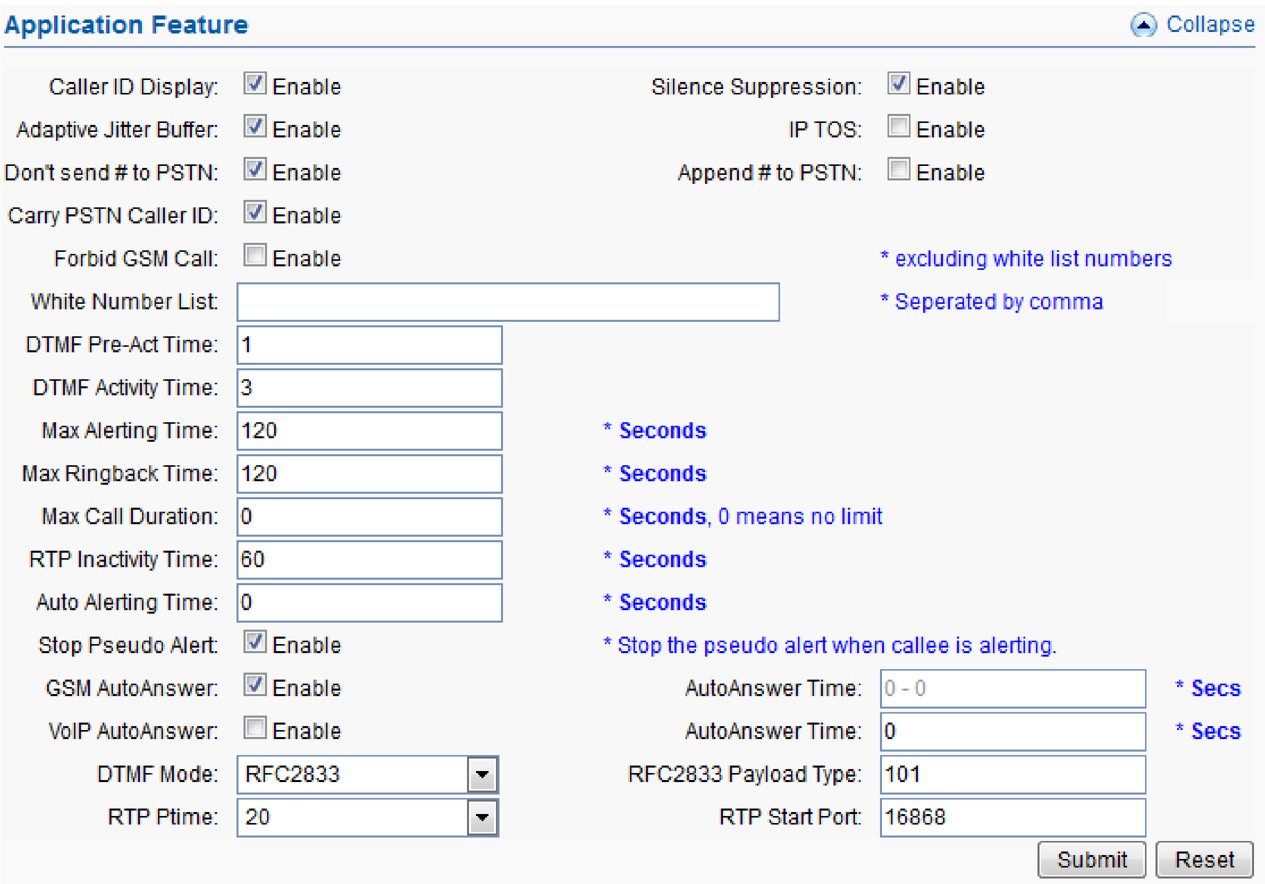


Figure 3.6.6-1 Application Feature

Items	Description
Caller ID Display	If it is disabled, caller ID will not show on “call status” page.
Silence Suppression	If it is enabled, half of the bandwidth will be saved.
Adaptive Jitter Buffer	A jitter buffer is a shared data area where voice packets can be collected, stored, and sent to the voice processor in evenly spaced intervals.
IP TOS	TOS of IP packets.
Don’t send # to PSTN	If it is enabled, the last digit # of callee number will be removed.
Append # to PSTN	If it is enabled, # will be appended in the callee number
Carry PSTN Caller ID	SIP extension will show the mobile number when you call the SIM in gateway.
Forbid GSM call	Calls will be rejected when calling the SIM in gateway.
White Number List	The numbers in white list will not be rejected if forbid GSM call is enabled.
DTMF Pre-Act time	The prepare time until DTMF tone is detected.
DTMF Activity time	The minimum of DTMF activity time.
Max Alerting Time	The maximum time of alerting.
Max Ringback Time	The maximum time of ring back.
Max Call Duration	The maximum duration for every call. System will hang up the call automatically if the call duration excesses this value.
RTP Inactivity Time	The maximum duration of silence from gateway. System will hang up the call automatically if the silence duration excesses this value
Auto Alerting Time	Fake ring back time, gateway will do fake ring back when excesses this value.
Stop Pseudo Time	Stopping fake ring back when the callee is alerting.
GSM Auto Answer	Applying to calls from GSM network. The gateway will answer the incoming calls automatically when excesses the value.
VoIP Call Auto Answer	Applying to calls from IP network. The gateway will answer the calls automatically when excesses the value.
DTMF Mode	RFC2833, SIP INFO and IN-BAND. The default one is RFC2833.
RFC2833 Payload Type	RTP Payload for DTMF, the default is 101.
RTP Ptime	The interval of RTP packages.
RTP Start Port	The initial port when RTP voice stream transmit the IP network.

Table 3.6.6-1

3.7 System Settings

3.7.1 User Mgmt

The default username/password of gateway is root/root. You are allowed to change the password and add new users on this page.

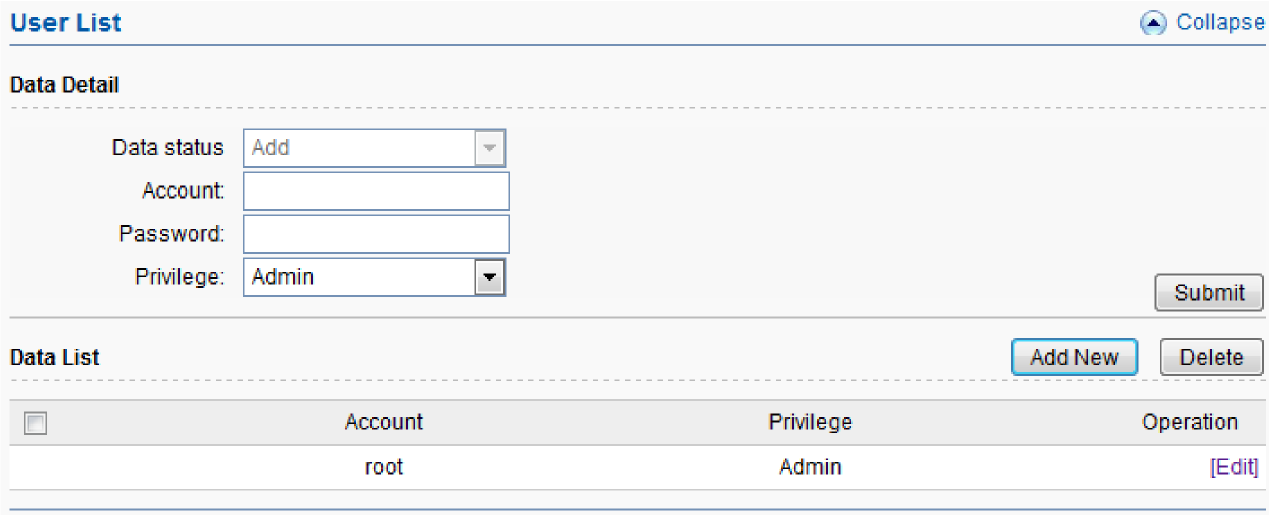


Figure 3.7.1-1 User List

3.7.2 Device Mgmt

Basic Settings

You are allowed to set an alias for device. You can also manage your gateway to reboot automatically as you like. There are two types for you to choose, one is after gateway running specified time, and the other one is scheduled reboot.

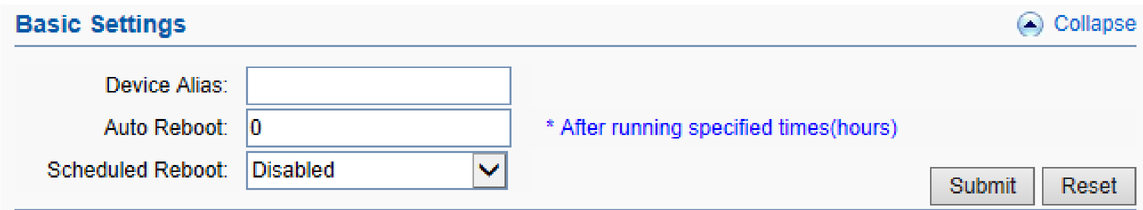


Figure 3.7.2-1 Basic Settings

Date and Time

You can choose your time zone or change the NTP server address here.

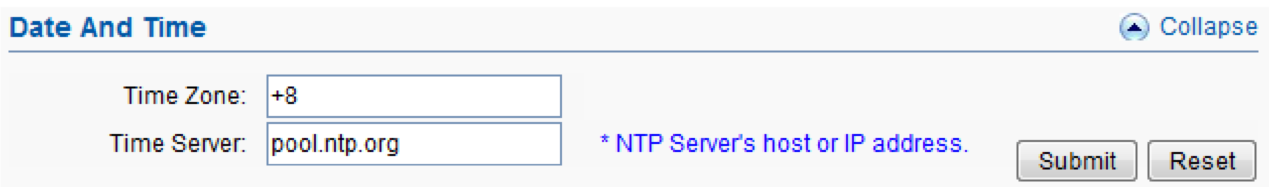


Figure 3.7.2-2 Date And Time

Remote Management System

Remote Management system is used to manage the gateway when it located in other physical locations. Network must be available for the gateway to communicate with ERM Server.

If ERM is enabled and correctly set, the gateway will register to ERM server and set up the connection between itself and ERM server. Administrator can login ERM server and monitor all the registered gateways.

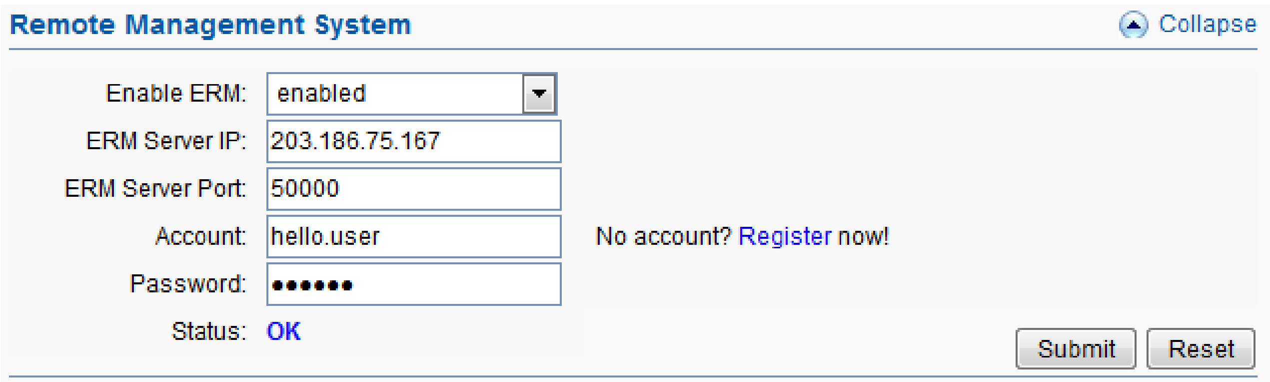


Figure 3.7.2-3 ERMS Settings

Items	Description
Enable ERM	Enable Ejoin remote management system.
ERM Server IP	The ERM server which is installed with Ejoin ERM software.
ERM Server Port	The port of ERM service. Default is 50000
Account	ERM account. You can also click “Register” to create a new account.
Password	Password of ERM account.
status	The Registration status of gateway with ERM server.

Table 3.7.2-1

3.7.3 File Management

File management is used for debugging the device. It has gdb, dying message and call statistics files. You can export or delete the logs from this page.

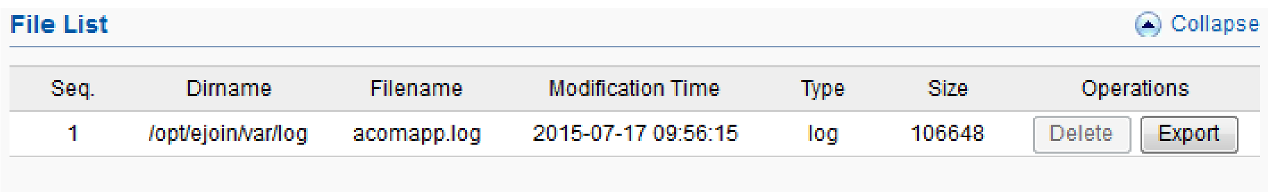
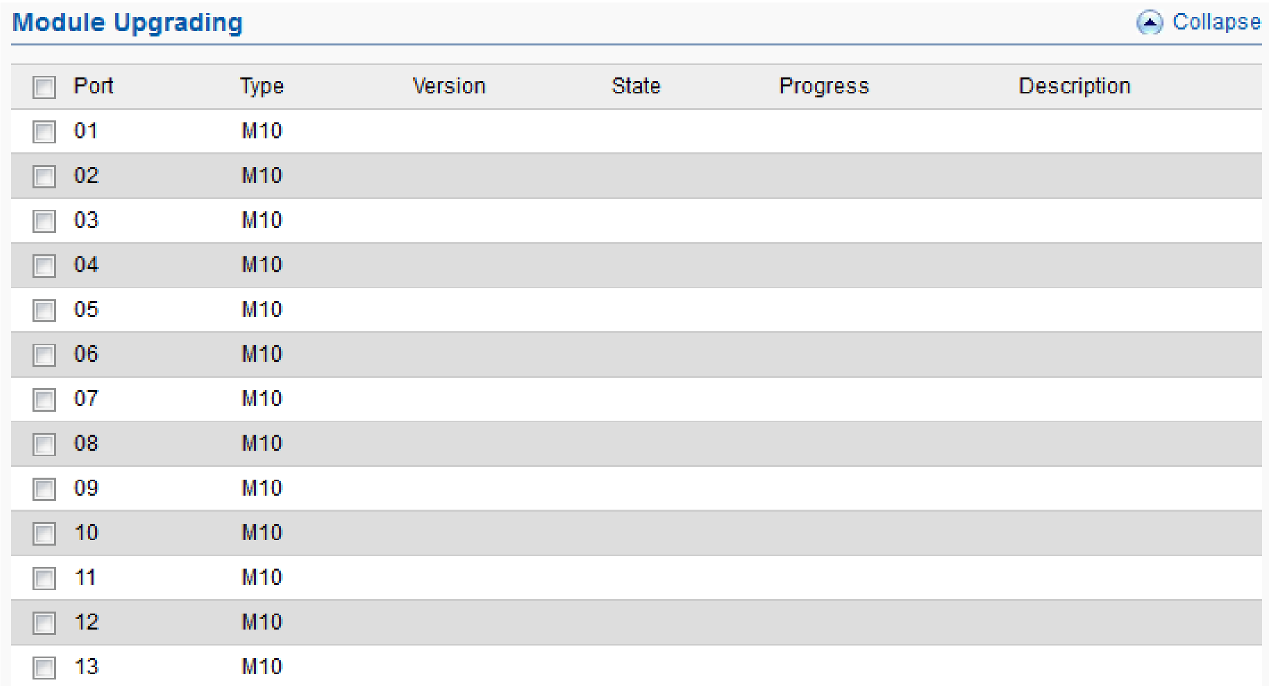


Figure 3.7.3-1 File List

3.7.4 Module Update

On this page, you can update the GSM/CDMA/WCDMA module software for every channel.



Module Upgrading

Port	Type	Version	State	Progress	Description
01	M10				
02	M10				
03	M10				
04	M10				
05	M10				
06	M10				
07	M10				
08	M10				
09	M10				
10	M10				
11	M10				
12	M10				
13	M10				

Figure 3.7.4-1 Module Upgrading

3.7.5 System Update

Import File

On this page, you can update the firmware for device, you can also update other files like kernel, ramfs etc.

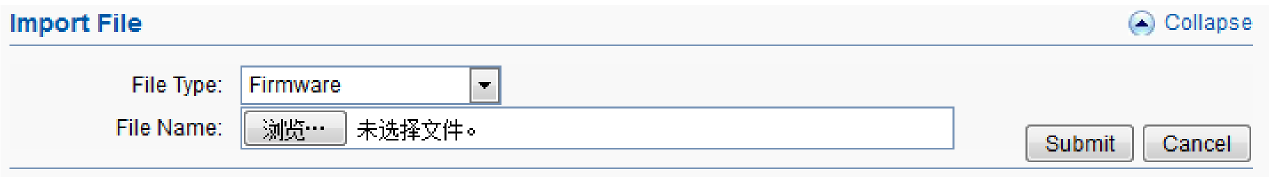


Figure 3.7.5-1 Import File

Export Configuration

Click “Export” button to export the configuration files.

Figure 3.7.5-2 Export Configuration

Restore To Factory

Sometimes there is something wrong with your gateway that you don't know how to solve it, mostly you will reset it. Just click "restore" button, your gateway will be reset to the factory settings.(IP will not change) .

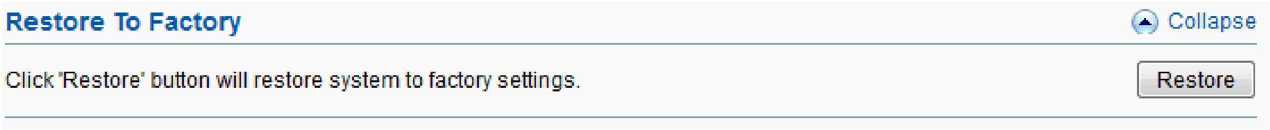


Figure 3.7.5-3 Restore To Factory

3.8 Debugging Tools

3.8.1 Test Network

PING is utility used to test the reachability of a host on IP network and measure the network quality between device to the destination host. There are two types of ping test: one is auto ping, the other one is manual ping.

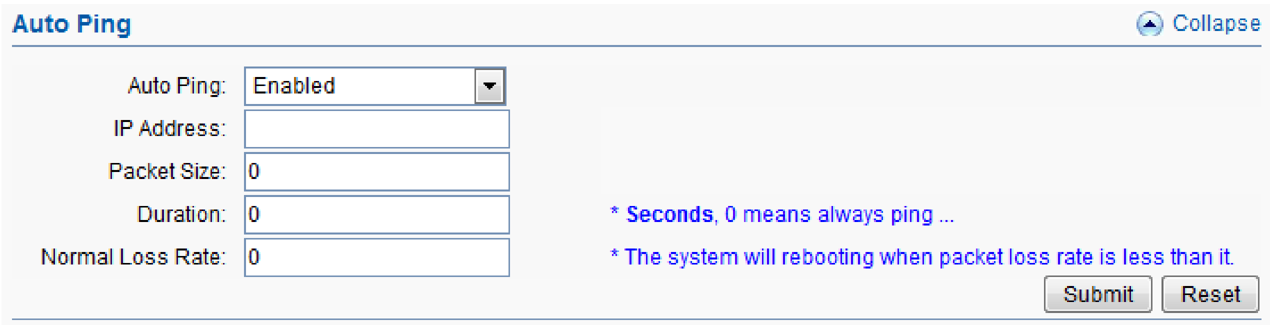


Figure 3.8.1-1 Auto Ping

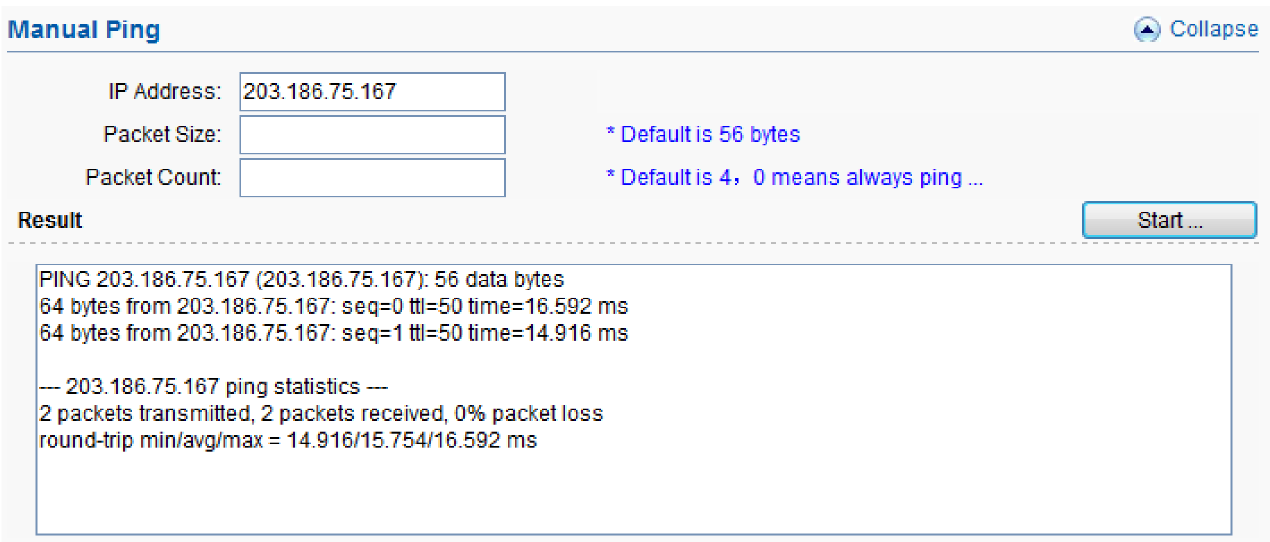


Figure 3.8.1-2 Manual Ping

3.8.2 Log System

You can enable the specific progress module running logs to monitor the device working status, and set the log file size. Device will save 5 logs defaultly , 20 is the maximum of logs num that device can saved in memory.

You can back to System Setting>>File management page to download these log files.

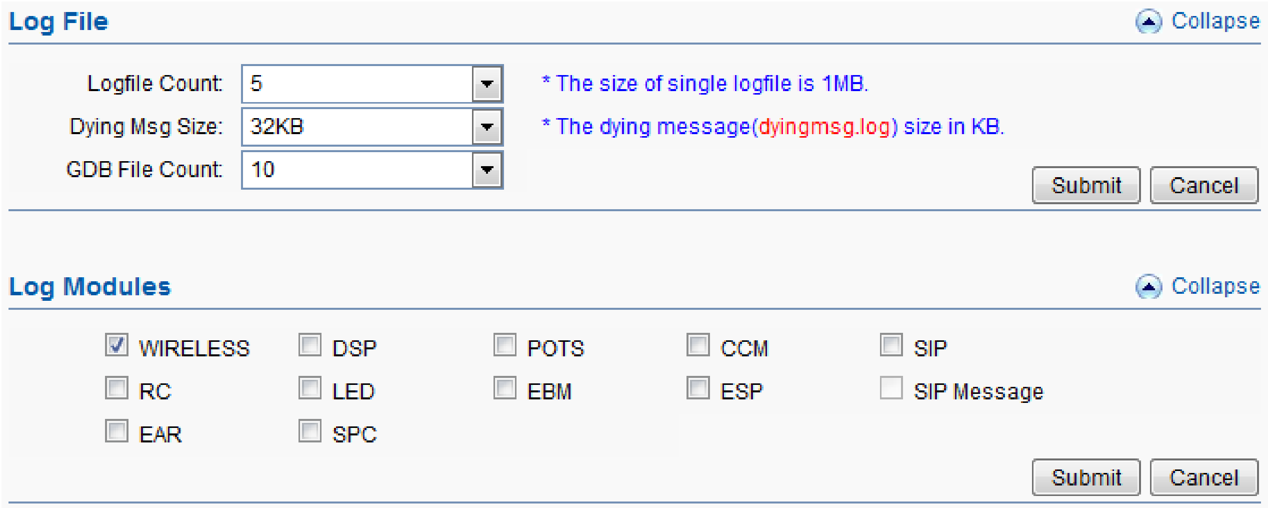


Figure 3.8.2-1 Log System

3.9 Running Status

3.9.1 Port Status

SIP Client Status

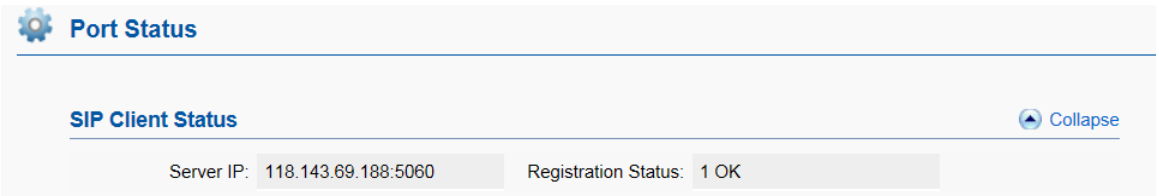


Figure 3.9.1-1 Port Status

SIP Client Status displays the register status while device work with SIP registration mode.

OK means registered to server successfully, ready to receive call from server. Failed means device is not registered on server or device working in a SIP Point-to-Point mode.

Port LED

Port LED display every SIM card status on device.

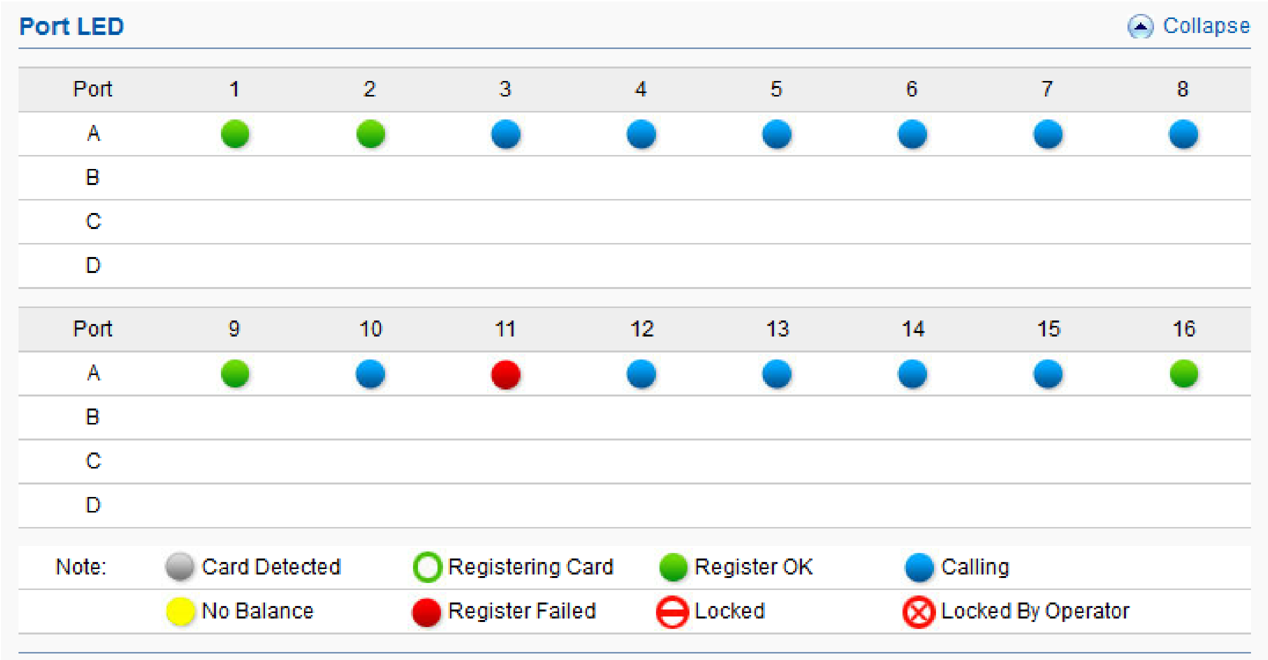


Figure 3.9.1-2 Port LED

Items	Description
	SIM card is detected, but it is not active.
	SIM card is searching operator to register to.
	SIM card is registered.
	SIM card is calling.
	Low balance(lower than the invalid balance when enable billing system)
	SIM card register failed
	SIM card is lock by device.
	SIM card is locked by operator.

Table 3.9.1-1

Port Balance

You can scan the balance for every SIM on this page.

Port Balance Collapse

Port	1	2	3	4	5	6	7	8
A	106851.00	66898.00	57083.00	91161.00	105321.00	40770.00	88160.00	51200.00
B								
C								
D								

Port	9	10	11	12	13	14	15	16
A	106425.00	42054.00	111232.00	80261.00	85376.00	96739.00	115094.00	30307.00
B								
C								
D								

Figure 3.9.1-3 Port Balance

Port Status

Port status display every wireless module detect status, and register operator information, signal value for channels.

Port Status Collapse

Port No.	Module Detected	SIM Registered	Provider	Signal Strength	SMS Count
1A	Yes	Yes	45201	23	
2A	Yes	Yes	45201	26	
3A	Yes	Yes	45201	25	
4A	Yes	Yes	45201	25	
5A	Yes	Yes	45201	22	
6A	Yes	Yes	45201	22	
7A	Yes	Yes	45201	27	
8A	Yes	Yes	45201	21	
9A	Yes	Yes	45201	31	
10A	Yes	Yes	45201	27	
11A	Yes	Yes	45201	27	
12A	Yes	Yes	45201	26	
13A	Yes	Yes	45201	31	
14A	Yes	Yes	45201	31	
15A	Yes	Yes	45201	31	
16A	Yes	Yes	45201	28	

Figure 3.9.1-4 Port Status

Items	Description
Port No.	Number of GSM/CDMA/WCDMA ports.
Module Detected	Indicates whether module is detected or not.
SIM Registered	Indicates whether SIM is registered or not
Provider	Displays the network carrier of current SIM card.
Signal Strength	Displays the signal strength of current SIM card
SMS Count	Displays the SMS count which has been sent since the last start up of system.

Table 3.9.1-2

3.9.2 Call Status

On this page you can monitor every current call on device.

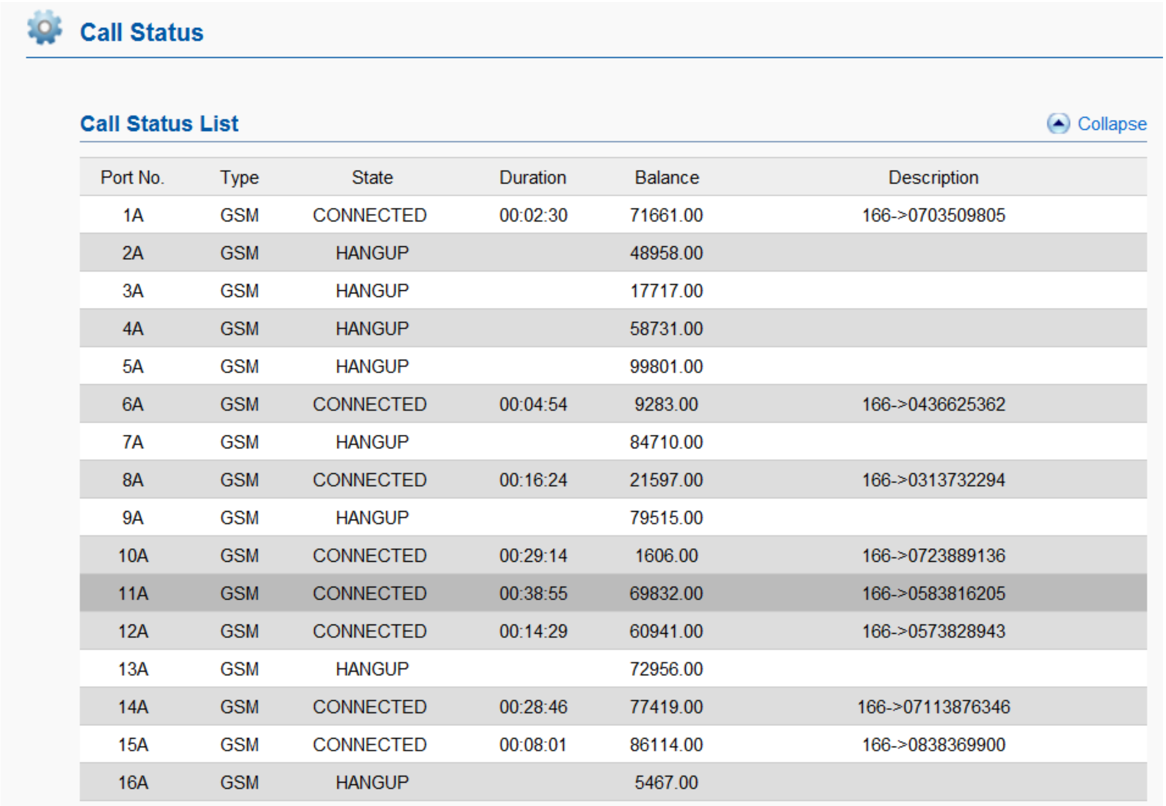


Figure 3.9.2-1 Call Status

Items	Description
Port No.	Number of GSM/CDMA/WCDMA ports.
Type	Indicates the current type of network. GSM/CDMA/WCDMA.
State	call status, it can be hangup, dialing, alerting, connected etc.
Duration	The duration this channel stay in current status.
Balance	The SIM card balance
Description	Display the SIM card status and caller, callee ID.

Table 3.9.2-1

3.9.3 System Status

System status includes WAN status, LAN status and other system information. This page can help you get the system status detail like firmware version, system tim e, running time etc in a fast, simple way.

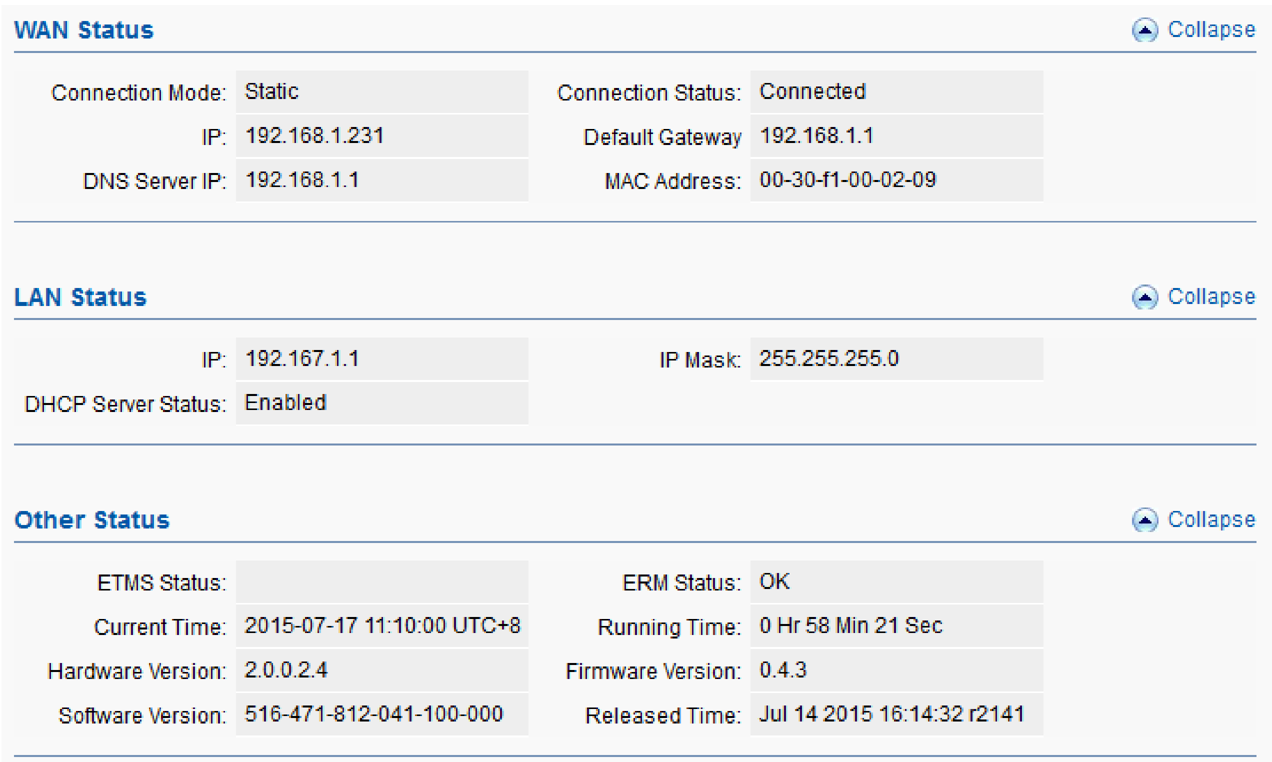


Figure 3.9.3-1 System Status

3.9.4 Call Statistics

Call Statistics List Clear Collapse

Port	Calls	Alerted	Connected	Con Fails	NC	PDD	ACD	ASR	Tot CallDur
Total	1926	1158	764	0	8/462	00:00:05	00:08:01	39%	101:21:00
1	127	61	41	3	1/36	00:00:05	00:07:58	32%	05:27:04
2	99	61	46	0	0/21	00:00:04	00:07:36	46%	05:42:04
3	153	81	54	0	1/50	00:00:05	00:08:13	35%	07:24:09
4	119	73	47	0	1/31	00:00:05	00:07:35	39%	05:49:19
5	117	69	40	0	0/27	00:00:05	00:11:50	34%	07:42:07
6	107	66	45	0	0/25	00:00:05	00:07:55	42%	05:56:32

Figure 3.9.4-1 Call Statistics

Items	Description
Port No.	Number of GSM/CDMA/WCDMA ports.
Calls	The total number of calls that send out from this SIM card.
Alerted	The total calls which is responded alerting message.
Connected	The total answered calls
Consecutive Fails	The consecutive failed calls.
No Carriers	No Carriers times and trying times.
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.
ACD	The Average Call Duration(ACD) is calculated by taking the sum of billable seconds(billsec) of answered calls and dividing it by the number of these answered calls.
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.

Table 3.9.4-1

3.9.5 SMS Statistics

On this page, you can scan the SMS statistics include the total SMS numbers re ceived from wireless network, the total numbers of SMS send out to phone user, the total number of send successfully for every SIM card.

SMS Statistics 收起

Data List Show Cur | Show All | Clear Data

Port	SIM Status	Received	Sent	Sent OK	Send Failed	Con. Failed	Sending	Success Rate
Total		0	0	0	0	0	0	
1A								
2A		0	0	0	0	0	0	
3A								
4A								
5A								
6A								
7A								
8A								
9A								
10A								
11A								
12A								
13A								
14A								
15A								
16A								
Total		0	0	0	0	0	0	

Port Show Cur | Show All | Clear Data

Figure 3.9.5-1 SMS Statistics

3.9.6 Inter-Call Status

When you enable the port-inter calling, you can monitor the executing details on this page. State column show inter calling status, duration display the time stay in related status. Incoming calls count the total calls this SIM card received while outgoing call display the total number of calls that send out from this SIM card. Descriptions show the caller and callee number in a inter call.

Inter-Calling Statistics

Inter-Calling Statistics Collapse

Port No.	State	Duration	Incoming Calls	Outgoing Calls	Descriptions
1A	IDLE		0	0	
2A	IDLE		0	0	
3A	IDLE		0	0	
4A	IDLE		0	0	
5A	IDLE		0	0	
6A	IDLE		0	0	
7A	IDLE		0	0	
8A	IDLE		0	0	
9A	IDLE		0	0	
10A	IDLE		0	0	
11A	IDLE		0	0	
12A	IDLE		0	0	
13A	IDLE		0	0	
14A	IDLE		0	0	
15A	IDLE		0	0	
16A	IDLE		0	0	

Figure 3.9.6-1 Inter-Calling Statistics

3.10 Save and Reboot

Modification will be applied after you saving and rebooting gateway.(All calls will break off when you reboot.)

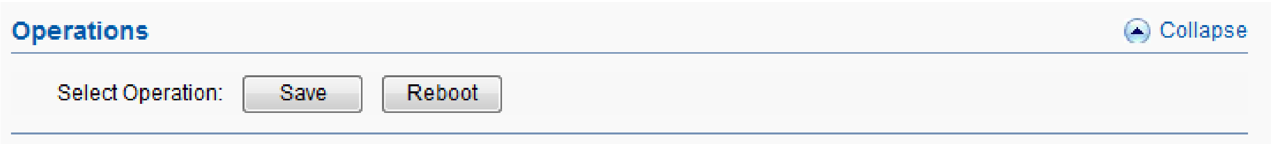


Figure 3.10-1 Save and Reboot

Chapter Ⅳ Typical Used Scenario

4.1 Landing from IP network to Mobile network

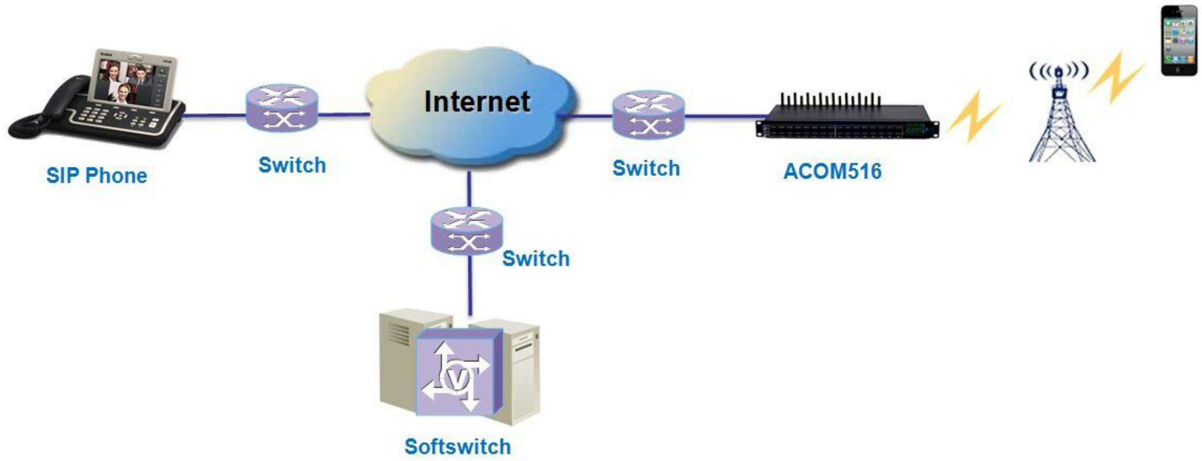


Figure 4.1-1 IP to Mobile

4.2 Accessing from Mobile network to IP network

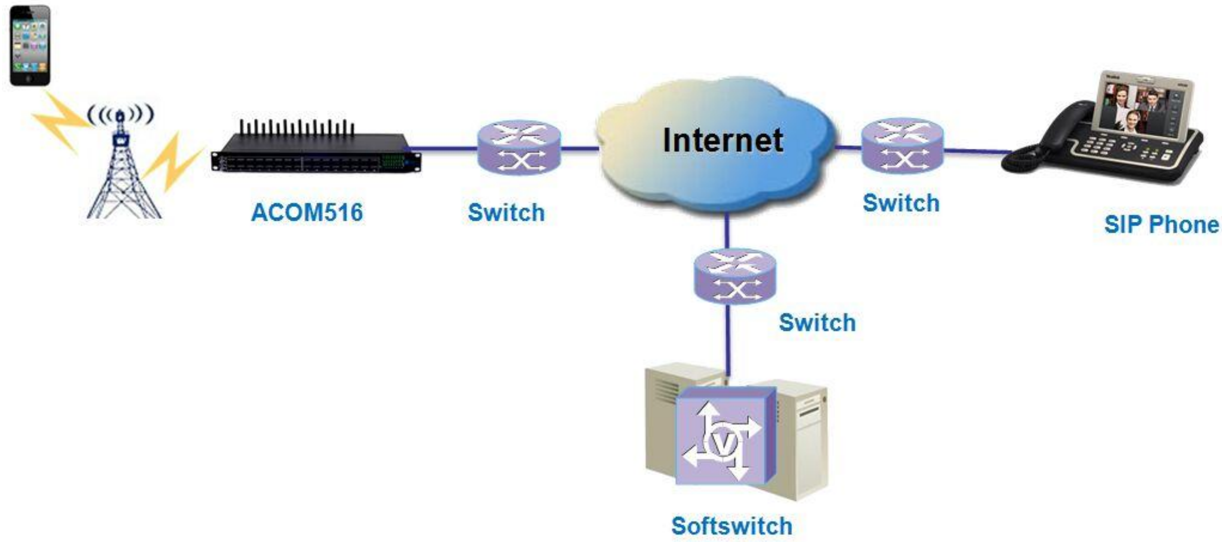


Figure 4.2-2 Mobile to IP

Chapter Ⅴ Ejoin Cloud System

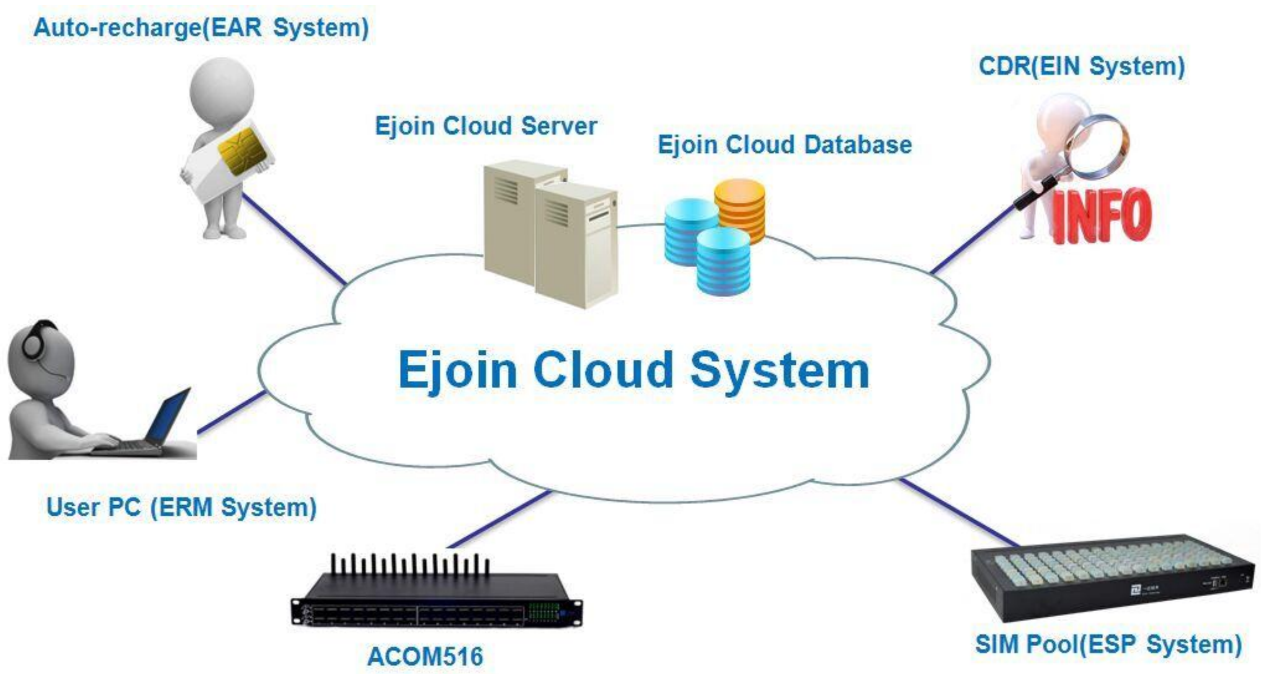


Figure 5-1 Ejoin Cloud System

ERM System: Ejoin Remote Management System

ESP System: Ejoin SIM Pool System

EIN System: Ejoin Information Notification System

EAR System: Ejoin Auto-Recharge System