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
	GSM: 850/900/1800/1900MHz
Fraguanay	CDMA:800/1900MHz
i requeite y	WCDMA:850/900/1800/1900/2100MHz

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

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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).





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

Gateway Administra	ation System
User Login	Account: Password: Login Reset

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.

AN Settings		Collaps
WAN Type:	Static IP	
WAN IP:	192.168.1.67	
IP Mask:	255.255.255.0	
Default Gateway:	192.168.1.1	
DNS Server:	192.168.1.1	Submit

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.

SIP Server Settings	\$					(Collapse
Protocol Mode:	Registration	~	Encryption Method:	NONE	~		
SIP Server IP:			SIP Server Port:	5060			
Phone Number:			Account:				
Password]			Submit	Reset

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

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	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

Running Paramete	ers			Collapse
Protocol Mode:	Point-to-Point	Encryption Method:	NONE	
SIP Server:		SIP Server Port:	5060	
Primary Proxy IP:		Proxy Port:	5060	
Secondary Proxy IP:		Proxy Port:	5060	
Expiration Period:	180	Local Port:	5060	
Multiple Port Support:	Disabled 💌	* If enabled, each acco	ount can use various port to r	register to server.
Use Phone Number:	Disabled 🔻	* If the username is no	t the same with userid, ena	ble it.
Receive All Call:	Enabled 🔹	* If enabled, all call wil	l be accepted.	
Drop Account Prefix:	Enabled 💌	* Remove the account	prefix presented in callee nu	imber.
Auto Resp 183:	Enabled 🔻	* Send 183-Session-P	rogress immediately for a ir	coming INVITE.
Route By From:	Disabled 🔻	* If enabled, only accept	ot the call whose "From" hea	der is matched.
No Line Code:	503 Service Unavailab	* Responce this SIP co	ode when no availabe line	
Custom User-Agent:		* the User-Agent head	er used in SIP message.	Submit Reset

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.

STUN		 Collapse
STUN Support:	Enabled	* If enabled, support the media traversal for non-symetric NAT.
STUN Server IP:	stunserver.org	* Fill your stun server ip if you have.
STUN Server Port:	3478	* The default port is 3478.
		Submit Reset

Figure 3.3-2 STUN Settings

MNP

NP			 Collapse
MNP Support:	Enabled	•	* If enabled, the server can select channel or change callee number.
Select Order:	ASC	•	* ASC/DESC will try to ensure the load balance, but Random not.
Route:	After Manipulation	-	
Server URL:			
Username:			
Password:			
			Submit Reset

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



SIP Accounts

SIP Accou	nts				Collapse
Port	Allowed Prefix	Phone Number	Account	Password	Status
1	070,075	101	test	•••••	
2	077,078				
3					
4					
5					
6					
7					
8					
9					



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.

Basic Settings		6) Collapse
Frequency Band:	850-900-1800-1900 V MHz		
Lock The Operator:			
Unnormal SIM Supp:	Enable	Submit	Reset

Figure 3.4.1-1 Basic Settings

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

Hardware Properties

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 $\sqrt{1}$, and disabled without $\sqrt{1}$. 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 No.	Main Access	Check Balance	Card Number	Balance	SMS Forward To	SMS Center
1		\checkmark		0.00	+8613715266978	+8613510844655
2		\checkmark		0.00		
3		\checkmark		0.00		
4		\checkmark		0.00		
5		\checkmark		0.00		
6		\checkmark		0.00		
7				0.00		
8		\checkmark		0.00		
9		\checkmark		0.00		
10				0.00		
11				0.00		
12		\checkmark		0.00		
13				0.00		
14				0.00		
15		\checkmark		0.00		
16				0.00		

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

Table 3.4.2-1

Base Stations settings/operations.

ort 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

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.

Т	ab	le	3.	4.	2-	2
-	uv	I.	~			_

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	В	С	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
				Сору	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.

Modify IMEI : Customiz	e Range 👻		
Dynamic IMEI List			Collaspe
Data Detail			
Data Status:	Add		
IMEI Start:			
IMEI Size:	1		Submit
Data List			Add New Delete
	IMEI Start	IMEI Size	Operation
	863435412312336	10000	[Delete] [Edit]

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.

Port PIN				🔿 Collapse
Port	А	В	С	D
1	1234			
2				
3				
4				
5				
6				
7				
8				
9				
10				
11				
12				
13				
14				

Figure 3.4.4-1 Basic Settings

3.4.5 SMS Send

Basic Settings

Basic Settings			Collapse
SMS Format:	PDU 💌		
Status Report:	Disabled 💌		
Forward Protocol:	GSM 💌	* forward SMS via SIP MESSAGE request.	
			Submit Reset

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.



Scheduled Sending

Scheduled Sendir	Collapse			
Content:	maximum 300 ASCII charactor	rs!		.н
Recipients:	maximum 255 digits, including	g the semi-colon!		* Semi-colon can be used to separate multiple receivers.
	Send To Local SIM By Duration: By Consecutive Failed Calls By Consecutive Calls	Minimum Minutes Failure Count: Call Count:	60 0 0	Maximum Minutes: 1440
	By Call Duration	Call Duration:	0	Minutes Submit Reset

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.

Send SMS		 Collapse
Please Select Port:	Image: All Image: 01 million 02 million 03 million 04 million 05 million 06 million 07] 08] 16
Receiver List	8613510087456	* Semi-colon can be used to separate multiple receivers.
SMS Content:	Hello!	Send
Successful SMS: Failed SMS:	0 Clear 0 Clear	

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.

SMS C	ontent			Collapse
SMS Lis	st		Refresh	Clear
Port	Sender	Time	Content	Operations
1A				Details(0)
2D	13510956503	07-17 11:07	ACOM516 VoIP Gateway is a multi-functional and high performance product, which is mainly used for call termination (VoIP to Mobile)	Details(1)
3D				Details(0)
4A				Details(0)
5D				Details(0)
6A				Details(0)



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.

SMS	Detai	ls			🕒 Collapse
Ple Pl	ease Se ease S	elect Port: 2 elect SIM: D		▼ ▼	
SMS	List			Back Refresh	Clear Delete
	Port	Sender	Time	Content	Operations
	2D	13510956503	07-17 11:07	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	Reply Delete
Tota	l: 1 1/	1 Pages Page1	▼ 10/Pa	age →	

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.

Basic Settings			Collapse
SMS Warning:	Enabled	•	
SMS Receiver for Warning:	+8613715266978	St	ubmit Reset

Figure 3.4.7-1 Basic Settings

Conditions for Locking Card

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

Conditions for Locking) Card	 Collapse
SIM Online Time Checking		
Enable or Not:	Enable	
Accumulated Call Duration (Checking	
Enable or Not:	Enable	
Accumulated Connected Ca	lls Checking	
Enable or Not:	Enable	
Reset When Switching:	Enable	* Reset the condition when switching to next SIM card.
Connected Calls:	5	
Locking Duration:	-1	* Seconds, 0 means no lock while -1 means permanent lock.
Accumulated Calls Checking	9	
Enable or Not:	Enable	

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.

If the SIM card is locked by gateway, it will show Θ , 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.

Basic Settings		Collapse
Port Inter-Calling:	Enabled 💌	* If enabled, device will enable the feature by following conditions.
Send SMS:	Disabled 💌	* If enabled, the callee will send a SMS to caller before inter-calling.
Min Call Duration:	60	* Seconds
Max Call Duration:	120	* Seconds
		Submit Reset

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).

Conditions Settings			Collapse
By Device Online Time: Min Interval: Max Interval:	Enable 60 120	* Minutes * Minutes	
Consecutive Failed Calls:	Enable		
By Consecutive Calls:	🔲 Enable		

Figure 3.4.8-2 Conditons Settings

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

SMS List		Collapse
Data List	Add N	lew Delete
	SMS Content	Operation
	please call me!	[Delete] [Edit]
	call me right now!	[Delete] [Edit]
	plz call me when u're free.	[Delete] [Edit]

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.

Auto Settings			🕒 Collapse
Auto-Get LocNum:	USSD		
USSD Command:	*134*2#	Get Now	
Number Keywords:	number	* The prefix keywords of the SIM number in USSD response.	
Prefix Translation:	930	> The prefix to be added	
		Subm	it Reset

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.

Auto Settings			Collapse
Auto-Get LocNum:	SMS 💌		
SMS Content:		Get Now	
Service Num:			
Number Keywords:		* The prefix keywords of the SIM number in SMS response.	
Prefix Translation:	930	> The prefix to be added	
		Submit	Reset

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.

SIM Nun	nber			Collapse
Port	A	В	С	D
1				
2				
3				
4				
5				
6				
7				
8				
9				
10				
11				
12				
13				
14				



3.4.10 AT Command

Module Operations

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

Module Operations		🕥 Collapse
Please Select Module:	▼ Restart Stop Start	

Figure 3.4.10-1 Module Operations

Command Operation

omman	d Operati	ons	Collapse
Please	Select Port:	All 00 01 02 03 04 05 06 07 0 09 01 01 12 01 04 05] 08] 16
Ma AT (USSD (nually Call: Command: Command:	+8613715266978 Start at+cpin? Send Send Query All	
esponse	Data		Clear Resfresh
Port	SIM Statu	S Content	Operation
1 A			
2A	•	at+cpin? +CPIN: READY	
ЗA			
4A			
5A			

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.

USS	D Lis	t				Collapse
USSE) Comi	mand		Сору	Show Current	Show All SIM Clear Data Send
	Port	Status	Command		Response	Operations
	1A					
	2D					Send
	3D					Send
	4A					
	5D					Send
	6A					
	7A					
	8A					
	9A					
	10A					
	11A					
	12A					
	13A					
	14A					

Figure 3.4.11-1 USSD List

Items	Description
Сору	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.

Basic Settings		 Collapse
Billing:	Disbaled 💌	
Hangup The Call:	Disabled 💌	* When the balance is not enough.
Auto Query Balance:	Disabled 💌	* When the balance decrease to warning value, query the balance.
Auto Query Balance:	0	* Minutes, get balance periodically. O means no query.

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

Provid	der List								e) Collapse
Inde	ex Operator	ID Operator	Name	Query	Method	С	aution Balances		Invalid Bal	ances
1	46001	CHINA UNIC	OM GSM	USSD	-	0.	00]	0.00	
									Submit	Reset
USSD	Query Keyw	ord List							e) Collapse
Inde	ex Operator	ID Query Cor	nmand	Balance	Keywords	Invalio	d Balance Keywo	rds	Invalid SIM K	eywords
1	46001									
							Inquiry No	w	Submit	Reset
SMS (Query Keywo	ord List							6) Collapse
Index	Operator ID	Service Num	Query	Cmd	Balance	Keys	Invalid Bal I	Keys	Invalid S	SIM Keys
1	46001									
							Inquiry No	w	Submit	Reset

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 ONO 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.

Tariff List			Collapse
Data Detail			
Data Status:	Edit	\checkmark	
Destination Prefix:	X		
Tariff:	0.0010	/ 60	Submit
Data List			Add New Delete
	Destination Prefix	Tariff	Operation
	Y	0.0010/60	

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.

Call Duration Setti	ngs	 Collapse
Use Global Settings:	Enabled 💌	All Channels use the same call duration control.
Total Max Duration:	0 means no limit	Minutes
Daily Max Duration:	0 means no limit	Minutes, to use this feature, please set the NTP server.
Min Duration Unit:	60	Seconds
DropCall If Expired:	Enabled 💌	Drop the call if the MCD expired.
		Submit Reset

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".

Call D	uration	Statistics					\land Collapse
Data Li	st				Show Current	Show All SIM	Batch Reset
	Port	Status	Total Duration	Remain Duration	Daily Duration	Daily Rem Dur.	Operations
	1A						
	2D	•	00:00:00	:	00:00:00	;	Reset
	3D	•	00:00:00	:	00:00:00	;	Reset
	4A						
	5D	•	00:00:00	:	00:00:00	;	Reset
	6A						
	7A						
	8A						
	9A						
	10A						
	11A						
	12A						
	13A						



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.

Port Settin	Port Settings Collapse					
Port	Total Max Duration	Daily Max Duration	Min Duration Unit	DropCall-If-Expired		
1	0	0	60	\checkmark		
2	0	0	60	\checkmark		
3	0	0	60			
4	0	0	60	\checkmark		

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.

Data Detail				
Data Status: Remote Gateway ID: Gateway IP: Gateway Port:	Edit ejoin 119.81.127.122 5060			Submit
Data List				Add New Delete
Remot	e Gateway ID	Gateway IP	Gateway Port	Operation
	ejoin	119.81.127.122	5060	[Delete] [Edit]

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.

Dial Pattern Settin	gs	
Pattern List		Collapse
Data Detail		
Data Status :) Pattern:	ldd V	Submit
Data List	Add Ne	w Delete
	Pattern	Operation
	No Data	

Figure 3.5.2-1 Dial Pattern Settings

3.5.3 Number Translation

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

Prefix Translation	List				Collapse
Data Detail					
Data Status:	Edit	-			
Ports:	All	•			
Original Prefix:	[2-9]		x means	all input number, [0-9] mean	s all digits
Translated Prefix:	0755x		x means	the corresponding digit of or	ignal prefix from right to le Submit
Data List					Add New Delete
	Ports	Origina	al Prefix	Translated Prefix	Operation
		[2-	-9]	0755x	[Delete] [Edit]

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.

Data Details				
Data Calle	a Status: Edit e Prefix: 2567	* Asterisk	neans match all digits	
Digits S	Stripped: 3	^ U means	not stripping prefix	
Digita				Subbmit
Data List				Add New Delete
	Callee Prefix	Digits Stripped	Digits Added	Operation
	2567	3	0	[Delete] [Edit]

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.)

CallerId Hidden		 Collapse
CallerId Hidden:	Disabled 🗸	
Dial Prefix:		Submit Reset

Figure 3.5.4-2 CallerId Hidden

3.5.5 Incoming Black List

You can forbid some calls by incoming black list.

Black Li	st			🔶 Collapse
Data Deta	iils			
I	Data Status:	Edit	-	
C	Callee Prefix:	016	'x' means all number	
Ca	allee Length:	12	**' means the callee length is unlimited	Submit
Data List				Add New Delete
		Callee Prefix	Callee Length	Operation
		016	12	[Delete] [Edit]

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.

Incoming White	List Settings		
White List			Collapse
Data Details			
Data Status:	Add	\checkmark	
Callee Prefix:	0167	'x' means all number	
Callee Length:	12	** means the callee length is unlimited	Submit
Data List			Add New Delete
	Callee Prefix	Callee Length	Operation
		No Data	

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.

Basic Settings			🕞 Collap
SIM Pool:	Enable	•	
Registration:	Enable	•	* If connect directly to a SIM pool device, disable the registration.
Server Address:	203.186.75.167		* Add ":port" to specify a special port.
Username:	test.user		
Password:	•••••		
Status:			Submit Rese
Other Settings			 Collap
Use Local Policy:	Disable	•	* If enabled, the policy of page Lock/Switch Card will be used.
			Submit Rese

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.

Basic Settings		 Collapse
Auto Recharge:	Enable	
Server Address:	203.186.75.167	* Add ":port" to specify a special port.
Username:	Goip16	
Password:	•••••	
Status:		Submit Reset
Other Settings		Collapse

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.



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).

Basic Settings			Collapse
State Notification:	Enabled 💌		
Server Address:		* Add ":port" to specify a special port.	
Username:			
Password:			
Registration Status:			Submit Reset

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 registeration 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.

PPTP-VPN Setting	5		Collapse
VPN Support:	Enabled	* Support the PPTP-VPN	
Server Address:			
Username:			
Password:			
Local IP:	0.0.0.0		
Remote IP:	0.0.0.0		Submit Reset

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.

Network Manageme	ent Settings) Collapse
Web Port:	80		
Telnet Port:	23		
		Submit	Reset

Figure 3.6.1-2 Network Management Settings

3.6.2 Port Settings

Port Set	ttings					Collaps
Port	Туре	Disable	Hot-line	Unconditional Forward	NoAnswer Forward	Busy Forward
1	GSM					
2	GSM					
3	GSM					
4	GSM					
5	GSM					
6	GSM					
7	GSM					
8	GSM					
9	GSM					
10	GSM					
11	GSM					
12	GSM					
13	GSM					
14	GSM					
15	GSM					
16	GSM					
	Disable a	ll port 🔲				Submit Reset

Figure 3.6.2-1 Port Settings

Items	Description	
Туре	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

Voice Setings				🕒 Co
Voice Volume:				
Input Volume:	15	Output Volume:	15	
DTMF Volume:	15			
Dial Tone				
High Frequency:	0	Low Frequency:	450	
On Duration:	5000	Off Duration:	0	
Ringback Tone				
High Frequency:	0	Low Frequency:	450	
On Duration:	1000	Off Duration:	4000	
Busy Tone				
High Frequency:	0	Low Frequency:	450	
On Duration	350	Off Duration:	350	

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.

Callback	Settings	 Collapse
Port	Enable	Callback Numbers (* means all, supports up to 32 numbers seperated by comma)
1		
2		
3		
4		
5		
6		
7		
8		
9		
10		
11		
12		
13		
14		
15		



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.

Call Waiti	 Collapse 		
Port	SIM Status E	nabled	Status
1			
2			Deactived
3			Deactived
4			
5			Deactived
6			
7			
8			



3.6.6 Other Settings

Application Featur	re) Collapse		
Caller ID Display:	Enable Silence Suppression: Enable					
Adaptive Jitter Buffer:	Enable IP TOS: Enable					
Don't send # to PSTN:	Enable Append # to PSTN: Enable					
Carry PSTN Caller ID:	Enable					
Forbid GSM Call:	Enable		* excluding white list number	ers		
White Number List:			* Seperated by comma			
DTMF Pre-Act Time:	1					
DTMF Activity Time:	3					
Max Alerting Time:	120	* Seconds				
Max Ringback Time:	120	* Seconds				
Max Call Duration:	0	* Seconds, 0 means no lim	iit			
RTP Inactivity Time:	60	* Seconds				
Auto Alerting Time:	0	* Seconds				
Stop Pseudo Alert:	Enable	* Stop the pseudo alert whe	en callee is alerting.			
GSM AutoAnswer:	Enable	AutoAnswer Time:	0 - 0	* Secs		
VoIP AutoAnswer:	Enable	AutoAnswer Time:	0	* Secs		
DTMF Mode:	RFC2833 🔹	RFC2833 Payload Type:	101			
RTP Ptime:	20 💌	RTP Start Port:	16868			
			Submit	Reset		

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.

User List			Collapse
Data Detail			
Data status Account: Password: Privilege:	Add Add Admin		Submit
Data List			Add New Delete
	Account root	Privilege Admin	Operation [Edit]

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.

Basic Settings			Collapse
Device Alias:			
Auto Reboot:	0	* After running specified times(hours)	
Scheduled Reboot:	Disabled 🗸		Submit Reset

Figure 3.7.2-1 Basic Settings

Date and Time

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

Date And Time			🕟 Collapse
Time Zone:	+8		
Time Server:	pool.ntp.org	* NTP Server's host or IP address.	Submit Reset



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.

Remote Managem	\land Collapse			
Enable ERM:	enabled	-		
ERM Server IP:	203.186.75.167			
ERM Server Port:	50000			
Account:	hello.user		No account? Register now!	
Password:	•••••			
Status:	ОК			Submit Reset

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 accourt		
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.

File List						Collapse
Seq.	Dirname	Filename	Modification Time	Туре	Size	Operations
1	/opt/ejoin/var/log	acomapp.log	2015-07-17 09:56:15	log	106648	Delete Export

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 Upgr	ading		Collapse		
Port	Туре	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.

Import File		Collapse
File Type: File Name:	Firmware ▼ 浏览··· 未选择文件。	Submit Cancel

Figure 3.7.5-1 Import File

Export Configuration

Click "Export" button to export the configuration files.

Export Configuration	\land Collapse
Click 'Export' button to export the configuration.	Export

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).

Restore To Factory	🔺 Collapse
Click 'Restore' button will restore system to factory settings.	Restore

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.

Auto Ping		 Collapse
Auto Ping:	Enabled 💌	
IP Address:		
Packet Size:	0	
Duration:	0	* Seconds, 0 means always ping
Normal Loss Rate:	0	* The system will rebooting when packet loss rate is less than it.
		Submit Reset

Figure 3.8.1-1 Auto Ping

Manual Ping			💫 Collapse
IP Address:	203.186.75.167]	
Packet Size:		* Default is 56 bytes	
Packet Count:		* Default is 4, 0 means always ping	
Result		_	Start
PING 203.186.75.16 64 bytes from 203.13 64 bytes from 203.13 203.186.75.167 p 2 packets transmitte round-trip min/avg/m	7 (203.186.75.167): 56 dat 86.75.167: seq=0 ttl=50 tim 86.75.167: seq=1 ttl=50 tim ing statistics id, 2 packets received, 0% p nax = 14.916/15.754/16.592	a bytes ne=16.592 ms ne=14.916 ms packet loss 2 ms	

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.

Log File					Collapse
Logfile Count:	5	▼ * The size	of single logfile is	1MB.	
Dying Msg Size:	32KB	 The dyir 	ig message(<mark>dying</mark> r	msg.log) size in KB.	
GDB File Count:	10	•			Submit Cancel
Log Modules					Collapse
Log Modules	B DSP	POTS	ССМ	SIP	Collapse
Log Modules WIRELESS RC	B DSP	POTS	CCM	SIP	Collapse
Log Modules WIRELESS RC EAR	B DSP	POTS	CCM	SIP	Collapse

Figure 3.8.2-1 Log System

3.9 Running Status

3.9.1 Port Status

SIP Client Status

0	Port Status				
	SIP Client Status				Collapse
	Server IP:	118.143.69.188:5060	Registration Status:	1 OK	

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 Pointto-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.
0	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
θ	SIM card is lock by device.
8	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	ce							Collapse
Port	1	2	3	4	5	6	7	8
А	106851.00	66898.00	57083.00	91161.00	105321.00	40770.00	88160.00	51200.00
В								
С								
D								
Port	9	10	11	12	13	14	15	16
А	106425.00	42054.00	111232.00	80261.00	85376.00	96739.00	115094.00	30307.00
В								
С								
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	3				Collapse
Port No.	Module Detected	SIM Registered	Provider	Signal Strength	SMS Count
1A	Yes	Yes	45201	23	
2A	Yes	Yes	45201	26	
ЗA	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.

0	Call Status							
	Call Status	s List					Collapse	
	Port No.	Туре	State	Duration	Balance	Description		
	1A	GSM	CONNECTED	00:02:30	71661.00	166->0703509805		
	2A	GSM	HANGUP		48958.00			
	ЗA	GSM	HANGUP		17717.00			
	4A	GSM	HANGUP		58731.00			
	5A	GSM	HANGUP		99801.00			
	6A	GSM	CONNECTED	00:04:54	9283.00	166->0436625362		
	7A	GSM	HANGUP		84710.00			
	8A	GSM	CONNECTED	00:16:24	21597.00	166->0313732294		
	9A	GSM	HANGUP		79515.00			
	10A	GSM	CONNECTED	00:29:14	1606.00	166->0723889136		
	11A	GSM	CONNECTED	00:38:55	69832.00	166->0583816205		
	12A	GSM	CONNECTED	00:14:29	60941.00	166->0573828943		
	13A	GSM	HANGUP		72956.00			
	14A	GSM	CONNECTED	00:28:46	77419.00	166->07113876346		
	15A	GSM	CONNECTED	00:08:01	86114.00	166->0838369900		
	16A	GSM	HANGUP		5467.00			

Figure 3.9.2-1 Call Status

Items	Description
Port No. Number of GSM/CDMA/WCDMA ports.	
Туре	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.

WAN Status				🕒 Collapse
Connection Mode:	Static	Connection Status:	Connected	
IP:	192.168.1.231	Default Gateway	192.168.1.1	
DNS Server IP: 192.168.1.1		MAC Address:	00-30-f1-00-02-09	
LAN Status				🕒 Collapse
IP:	192.167.1.1	IP Mask:	255.255.255.0	
DHCP Server Status:	Enabled			
Other Status				Collapse
ETMS Status:		ERM Status:	ок	
Current Time:	2015-07-17 11:10:00 UTC+8	Running Time:	0 Hr 58 Min 21 Sec	
Hardware Version:	2.0.0.2.4	Firmware Version:	0.4.3	
Software Version:	516-471-812-041-100-000	Released Time:	Jul 14 2015 16:14:32 r2141	

Figure 3.9.3-1 System Status

3.9.4 Call Statistics

Call Stati	istics 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	Stati	stics							🔿 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	
	ЗA								
	4A								
	5A								
	6A								
	7A								
	8A								
	9A								
	10A								
	11A								
	12A								
	13A								
	14A								
	15A								
	Total		0	0	0	0	0	0	
	rotal		U	U	U	U	U	U	
	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							
Port No.	State	Duration	Incoming Calls	Outgoing Calls	Descriptions		
1A	IDLE		0	0			
2A	IDLE		0	0			
ЗA	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.)

Operations					
Select Operation:	Save 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



ERM System: Ejoin Remote Management System

ESP System: Ejoin SIM Pool System

EIN System: Ejoin Information Notification System

EAR System: Ejoin Auto-Recharge System