



OpenVox Communication Co Ltd



SWG-2008 Gateway User Manual

Version 1.0



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Revision History

Version	Date	Detail
1.0	2019/04/04	Initial

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1. Overview

1.1 What is SWG-2008?

OpenVox SWG-2008 series wireless gateways include SWG-2008 G/C/L. The SWG-2008 series wireless gateways use the super-performance Intel 4-core CPU and support multiple encodings, such as G.711U, G.711A, GSM, G.722, G.723, G.726, G.729. The SWG-2008 series wireless gateways can help users reduce telecommunications and communication costs. It can be perfectly compatible with SIP servers such as Asterisk, 3CX, FreePBX, FreeSWITCH or VOS VoIP operation platform.

1.2 Application

Figure 1-2-1 SWG 2008 Gateway Product Front



Figure 1-2-2 Front panel

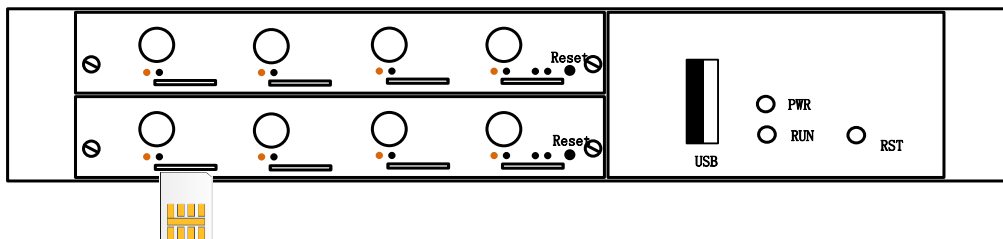


Table 1-2-1 Status light description

Status Light	Color	Status
Signal Status LED	Green and Flash	Module Initiating
	Red and Flash	No SIM Card
	Always red	Worst Signal Quality
	Always yellow	Medium Signal Quality
	Always green	Best Signal Quality
Call Status LED	Flash (0.5s)	Communicating
	Blind	Normal
Network Status LED	Green and Flash	Network Connected
Running Status LED	Green and Flash(0.5s)	Work Normally
Power Indicator	Always Green	Power on

1.3 Main Features

- Support SIP, IAX2 Protocol
- Add, Modify & Delete SIP/IAX2 Trunk
- SIP/IAX2 Registration with Domain
- SIP V2.0 RFC3261 Compliance
- DTMF Mode: RFC2833/Inband/SIP Info
- Multiple SIP/IAX2 Registrations modes
- Abundant Codecs:G.711A, G.711U, G.729, G.722, G.723, G.726, GSM
- IPv4, UDP/TCP, DHCP, TELNET, HTTP/HTTPS, TFTP
- PPTP VPN

- HTTP/SSH (Optical Telnet)
- Ping & Traceroute Command on the Web
- Simple Security Strategy: white list, black list, security rules
- Simple and convenient configuration via Web GUI
- Support maintenance and configuration by SSH
- Support configuration files backup and upload
- Support Chinese and English page
- Firmware Update by HTTP
- Support Web and SSH login password modification
- Restore Factory Settings
- CDR(More than 200,000 Lines CDRs Storage Locally)
- System log
- SIP/IAX2 log
- TCP and SIP capture
- Combine Different SIP/IAX2 Trunk into Group
- CLID Display & Hide (Need operators' support)
- Random call interval
- Call Duration Limitation
- Single Call Duration Limitation
- Real Open API Protocol (based on Asterisk)
- Support DISA
- SMSC/SMS/USSD
- PIN Identification
- Optional Voice Codec
- Ports Group Management
- SMS Remotely Controlling Gateway
- SMS Bulk Transceiver, Sent to Email and Automatically Resend
- SMS Coding/Detecting Automatically Identification

- SMS Forwarding and Quick Reply
- USSD transceiver
- Outbound
- Automatically Reboot
- Support MMP
- Support for custom scripts, dial plans
- Support OpenVox cloud manage

1.4 Physical Information

- Weight: 1.107KG
- Size: 220mm*44mm*192mm
- Operation Temperature:0~40°C
- Storage Temperature: -20~70°C
- Operation humidity:10% ~ 90% non-condensing
- Maximum power: 6W (Without antenna)
- WAN: 1*10/100M

1.5 Login

Default IP: 172.16.98.1

Username: admin

Password: admin

For the first time, you can access SWG-2008 by using default IP 172.16.98.1. Then configure the module as you want.

Figure 1-5-1 Login Interface

The server 172.16.6.108 is asking for your user name and password. The server reports that it is from Openvox-Wireless-Gateway.

admin

Password

OK Cancel

2. System

2.1 Status

On the “Status” page, you will find the information of all Modules, SIP, IAX2, Routing and the Network.

Figure 2-1-1 System Status

Module Information											
Port	Type	Signal	Band	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	Module Status	Remain Time
1.1	GSM			-1		Undetected SIM Card	0	0	0		No Limit
1.2	GSM			-1		Undetected SIM Card	0	0	0		No Limit
1.3	GSM			-1		Undetected SIM Card	0	0	0		No Limit
1.4	GSM			-1		Undetected SIM Card	0	0	0		No Limit
2.5	LTE			-1		Not registered	0	0	0	INIT	No Limit
2.6	LTE			-1		Undetected SIM Card	0	0	0		No Limit
2.7	LTE			-1		Undetected SIM Card	0	0	0		No Limit
2.8	LTE			-1		Undetected SIM Card	0	0	0		No Limit

SIP Information					
Endpoint Name	User Name	Host	Registration	SIP Status	
10028	anonymous	172.16.208.33	none	Unmonitored	
2111	test	(Unspecified)	server	UNKNOWN	

IAX2 Information					
Endpoint Name	User Name	Host	Registration	IAX2 Status	
2133	2133	172.16.208.33	client	UNREACHABLE	

Routing Information				
Rule Name	From	To	Rules	
incoming	grp-all	iax-2133		
gsm-1.12sip	gsm-1.1	sip-10028		
sip2gsm-1.1	sip-10028	gsm-1.1	Dial_pattern (+)[/101]	
sip2lte-2.5	sip-10028	lte-2.5	Dial_pattern (+)[/200]	
outgoing	sip-2111	gsm-1.2		
test	sip-10028	lte-2.6	Dial_pattern (+)[/]	

Network Information						
Name	MAC Address	IP Address	Mask	Gateway	RX Packets	TX Packets
LAN	A0:98:05:0A:2D:F7	172.16.6.108	255.255.0.0	172.16.0.1	1646	147

Table 2-1-1 Description of System Status

Option	Definition
Port	Number of each ports. GSM port starts with "gsm-", Such as: gsm-1.1;4G port starts with "lte-", Such as: lte-2.5.
Signal	Display the signal strength of each channels of gateway.

Band	GSM/ LTE.
BER	Bit Error Rate.
Carrier	Display the network carrier of current SIM card.
Registration Status	Indicates the registration status of current module.
PDD	Post Dial Delay (PDD) is the time from the point at which the final dialed digit is sent until they hear a ringtone or other in-band message. In the case where the originating network needs to play a notification before the call is completed, the definition of the PDD excludes the duration of such notification.
ACD	The Average Call Duration (ACD) is calculated by taking the sum of billable seconds (bill sec) of answered calls and dividing it by the number of these answered calls.
ASR	Answer Seizure Ratio is a measure of network quality. Its calculated by taking the number of successfully answered calls and dividing by the total number of attempted calls. Since busy signals and other rejections by the called number count as call failures, the ASR value can depending on user behavior.
Module Status	Display the status of the port. "A space" means the port is unavailable. "INIT" means registering and "READY" means the port is available.
Remain Time	Multiply this value by the step size as the limit call time.

2.2 Time

Table 2-2-1 Definition of Time Settings

Options	Definition
System Time	Your gateway system time.
Time Zone	The world time zone: Please select the one which is the same or the closest as your city.
POSIX TZ String	Posix time zone strings.
NTP Server 1	Time server domain or hostname. For example: [time.asia.apple.com].
NTP Server 2	The first reserved NTP server. For example: [time.windows.com].
NTP Server 3	The second reserved NTP server. For example: [time.nist.gov].
Automatic Sync from NTP	Whether to enable automatic sync time from NTP. On (enable), OFF (disable).
Sync from NTP	Sync time from NTP server.
Sync from Client	Sync time from local machine.

For example, you can configure like this:

Figure 2-2-1 Time Setting

Time Settings	
System Time:	2019-4-1 12:25:55
Time Zone:	Chongqing ▼
POSIX TZ String:	CST-8
NTP Server 1:	pool.ntp.org
NTP Server 2:	64.236.96.53
NTP Server 3:	ntp1.aliyun.com
Auto-Sync from NTP:	<input checked="" type="checkbox"/> ON

You can synchronize the gateway time in different ways: Sync from NTP or Sync from Client by pressing different buttons.

2.3 Login Settings

The new gateway has no administration privileges, all you can do is reset a username and password to manage your gateway. The username and password have all the permissions to operate the gateway. You can modify “Web Login Settings” and “SSH Login Settings”. If you have changed these settings, you don’t need to log out, just rewriting your new username and password will be OK. Also, you can specify the web server port number. Normally, the default web login mode is "http and https." For security, you can switch to “only https”.

Table 2-3-1 Definition of Login Settings

Options	Definition
User Name	Define your username to manage your gateway. Allowed characters "-_+. < >&0-9a-zA-Z". Length: 1-32 characters.
Password	Define your password to manage your gateway Allowed

	characters "-_+. < > & 0-9a-zA-Z". Length: 1-32 characters.
Confirm Password	Please input the same password as 'Password' above.
Login Mode	http and https: You can access gateway via link: http://gatewayIP or https://gatewayIP https: You can only access gateway via link: https://gatewayIP.
Port	Specify the web server port number.

For example, you can configure like this:

Figure 2-3-1 Login Settings

The screenshot displays two sections of the configuration interface:

- Web Login Settings:**
 - User Name: admin123
 - Password: [masked]
 - Confirm Password: [masked]
 - Login Mode: A dropdown menu with options "http and https" (selected), "http and https", and "only https".
 - Port: [empty]
- SSH Login Settings:**
 - Enable: ON (checkbox checked)
 - User Name: super
 - Password: super
 - Port: 12345

2.4 General

You can choose different languages for your system. If you want to change the language, you can switch "Advanced" on, then "Download" your current language package. After that, you can modify the package with the language you need. Then upload your modified packages, "Choose File" and "Add". For example:

Figure 2-4-1 Language Settings

Language Settings	
Language:	English ▾
Advanced:	<input checked="" type="checkbox"/> ON
Language Debug:	<input type="checkbox"/> TURN ON <input type="checkbox"/> TURN OFF
Download:	Download selected language package. <input type="button" value="Download"/>
Delete:	Delete selected language. <input type="button" value="Delete"/>
Add New Language:	New language Package: <input type="button" value="选择文件"/> 未选择任何文件 <input type="button" value="Add"/>

If you switch it on, you can manage your gateway to reboot automatically as you like. There are four reboot types for you to choose, "By Day, By Week, By Month or By Running Time".

Figure 2-4-2 Reboot Type

Scheduled Reboot	
Enable:	<input type="checkbox"/> OFF
Reboot Type:	By Running Time ▾
Running Time:	Hour: 0 ▾

You can set this enable if you use your system frequently, it can help your system work more efficiently.

2.5 Tools and Information

Reboot Tools

You can choose system reboot or asterisk reboot separately. The new version of the gateway retains two file systems. Clicking "System Switch" will reboot the system and switch to another file system.

Figure 2-5-1 Reboot Tools

WirelessGateway SWG-2008 SYSTEM 来自 172.16.6.108

Are you sure to reboot your gateway now?
You will lose all data in memory!

确定 取消

Free Commun cation OpenVox Solution

Reboot Tools

Reboot the gateway and all the current calls will be dropped.

Reboot the asterisk and all the current calls will be dropped.

Reboot the gateway and all the current calls will be dropped.

If you press "OK", the system will restart and the current call will be hung up. The same as the Asterisk Reboot.

Update Firmware

We offer 2 kinds of update types for you, you can choose System Update or System Online Update.

Figure 2-5-2 Update Firmware

Upload and Backup Configuration

If you want to update your system and remain your previous configuration, you can backup configuration first. Then you can upload configuration directly. It will be very convenient for you.

Figure 2-5-3 Upload and Backup Configuration

Voice Record

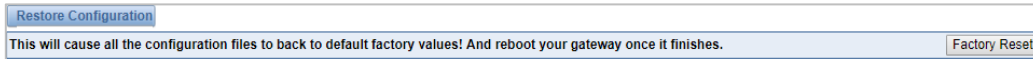
Select a module to record the voice, the maximum duration of the recording is 3 minutes. When the recording time exceeds 3 minutes, the recording file will be automatically downloaded.

Figure 2-5-4 Voice Record

Restore Configuration

Sometimes there is something wrong with your gateway that you don't know how to solve it. Mostly you can select factory reset, you just need to press a button and your gateway will be reset to the factory status.

Figure 2-5-5 Restore Configuration



System Information

On the "Information" page, there shows some basic information about the gateway. You can see software and hardware version, storage usage, memory usage and some other useful information.

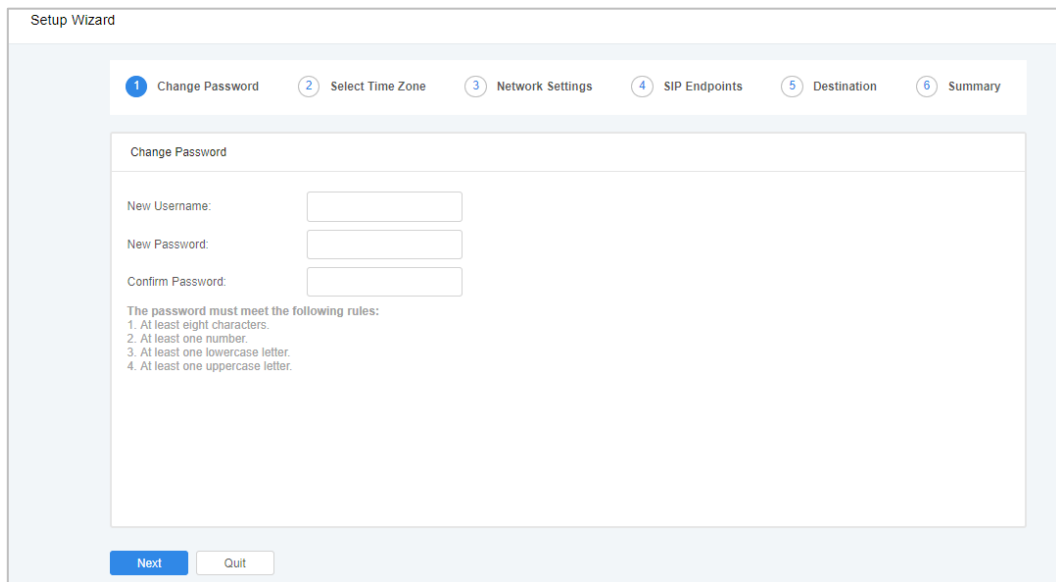
Figure 2-5-6 System Information

Product Name:	SWG-2008
GSM Model Description:	GSM: 850/900/1800/1900MHz
LTE Model Description:	LTE FDD: B1/B3/B5/B8 LTE TDD: B38/B39/B40/B41 TD-SCDMA: B34/B39 CDMA: BCO WCDMA: 900/2100MHz GSM: 900/1800MHz
Software Version:	2.1.0
Hardware Version:	1.0
Slot Number:	1
Storage Usage:	40.5M/487.9M (9%)
Memory Usage:	21.3297 % Memory Clean
Build Time:	2019-03-06 17:19:31
Contact Address:	10/F, Building 6-A, Baoneng Science and Technology Industrial Park, Longhua New District, Shenzhen, Guangdong, China
Tel:	+86-755-82535461
Fax:	+86-755-83823074
E-Mail:	support@openvox.cn
Web Site:	http://www.openvox.cn
Rebooting Counts:	615
System Time:	2019-4-1 14:17:38
System Uptime:	0 days 00:02:20

2.6 Setting Wizard

According to the boot, you can change the password, set the time, network, and create SIP terminals and routes quickly.

Figure 2-6-1 Setting Wizard



The screenshot shows the 'Setup Wizard' interface. At the top, there is a progress bar with six steps: 1. Change Password (highlighted), 2. Select Time Zone, 3. Network Settings, 4. SIP Endpoints, 5. Destination, and 6. Summary. Below the progress bar, the 'Change Password' section contains three input fields: 'New Username:', 'New Password:', and 'Confirm Password:'. Below these fields, there is a list of password rules: 'The password must meet the following rules: 1. At least eight characters. 2. At least one number. 3. At least one lowercase letter. 4. At least one uppercase letter.' At the bottom of the wizard, there are two buttons: 'Next' (highlighted in blue) and 'Quit'.

3. Module

You can see the information of SIM card on this page.

3.1 Module Settings

Figure 3-1-1 Module Settings

Port	Type	Carrier	Registration Status	Module Status	Actions
1.1	GSM		Undetected SIM Card		
1.2	GSM		Undetected SIM Card		
1.3	GSM		Undetected SIM Card		
1.4	GSM		Undetected SIM Card		
2.5	LTE		Not registered	INIT	
2.6	LTE		Undetected SIM Card		
2.7	LTE		Undetected SIM Card		
2.8	LTE		Undetected SIM Card		


On this page, you can see the information of the module status, click the  button to configure the port.

Figure 3-1-2 Port Configuration

Port gsm-1.1

Name:

Speaker Volume:

Microphone Volume:

Dial Prefix:

Pin Code: On

Custom AT commands when start:

STK flag: OFF

CLIR: OFF

SMS Center Number:

Band: All Band(850/900/1800/1900MHz) ▼

SIM IMSI:

Module IMEI:

Module Revision:

Carrier:

Bind Carrier: Auto ▼

Signal: -1

BER: -1

Status:

GSM Voice Codec: ERROR ▼

CLCC: ON

AT Timeout: s

You can choose different bands in the options.

Figure 3-1-3 Band binding

Band:	All Band(850/900/1800/1900MHz) ▾
SIM IMSI:	All Band(850/900/1800/1900MHz)
Module IMEI:	EGSM(850/900MHz) DCS(1800MHz) PCS(1900MHz)
Module Revision:	EGSM DCS(850/900/1800MHz) GSM850 PCS(850/1900MHz)

If you have set your Pin Code, you can check on like this:

Figure 3-1-4 PIN Code Application

Pin Code:	123456	<input checked="" type="checkbox"/> On
------------------	--------	--

Then enter the password and the system will recognize the number of the SIM card. It can help you prevent the SIM card from being stolen and increase the safety factor. If you want to hide your number when you call out, you can just switch CLIR “ON” (Of course you need your operator’s support).

Figure 3-1-5 CLIR Application

CLIR:	<input checked="" type="checkbox"/> ON
--------------	--

OpenVox GSM/3G Gateway supports optional GSM voice codec. For more details, you can see the picture below:

Figure 3-1-6 GSM Voice Codec

Band:	auto (Hz) ▾
SIM IMSI:	FR
Module IMEI:	HR
Module Revision:	EFR
Carrier:	AMR_FR
Bind Carrier:	AMR_FR
Signal:	FR&EFR, FR
BER:	EFR&FR, EFR
Status:	EFR&HR, EFR
GSM Voice Codec:	EFR&AMR_FR, EFR
	AMR_FR&FR, AMR_FR
	AMR_FR&HR, AMR_FR
	AMR_FR&EFR, AMR_FR
	AMR_HR&FR, AMR_HR
	AMR_HR&HR, AMR_HR
	AMR_HR&EFR, AMR_HR

IMEI Modification

We also provide IMEI automatically modified performance.

Figure 3-1-7 Automatic IMEI modification

Module IMEI: 352840039592814

If you want to modify the IMEI number, log in to the gateway to modify the IP address as follows. Input <http://gatewayIP/cgi-bin/php/gsm-autoimei.php> in your browser. Then log in to the web page and you will see the setting as "Enable" as "ON" as shown below. Otherwise, the IMEI number cannot be modified.

Figure 3-1-8 IMEI Modification

Automatic Change IMEI

Port: lte-1.1 lte-1.2 lte-1.3 lte-1.4
 gsm-2.5 gsm-2.6 gsm-2.7 gsm-2.8
 All

Enable: ON

Interval: 1800 Second

Immediately: modify IMEI immediately

Force: Modify IMEI no matter whether the channel state is ready or not.

Auto-IMEI Advanced

You can choose to modify one, more ports or all ports, and you can set the automatic modification interval by filling in the required time.

Figure 3-1-9 Time Interval

Interval: 1800 Second

If you select "Modify IMEI now", the changes will take effect immediately. If you select "Force", the system will hang up the current call and modify the IMEI.

Click the button to set it. There are two ways to modify the IMEI: manual modification or automatic generation.

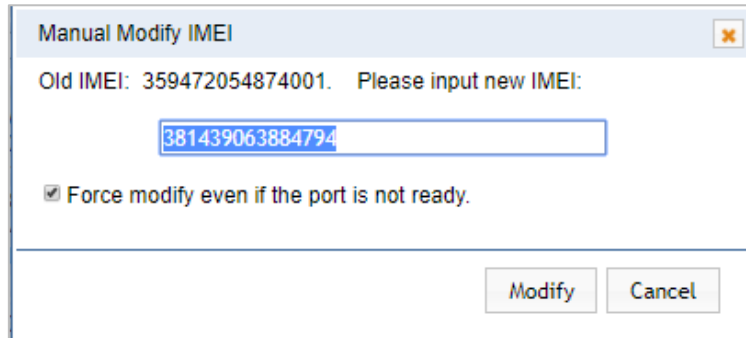
Figure 3-1-10 Advanced Setting

IMEI Number Setting	TAC(6 digit)	FAC(2 digit)	SNR(6 digit)	SP(1 digit)	Current IMEI	Action
Set to All	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	None	<input type="button" value="Set to All"/>
gsm-1.1	<input type="text" value="35xxxx"/>	<input type="text" value="0x"/>	<input type="text" value="xxxxxx"/>	<input type="text" value="Autogeneration"/>	352840039592814	<input type="button" value="Manual"/>
gsm-1.2	<input type="text" value="35xxxx"/>	<input type="text" value="0x"/>	<input type="text" value="xxxxxx"/>	<input type="text" value="Autogeneration"/>	352794018702565	<input type="button" value="Manual"/>
gsm-1.3	<input type="text" value="35xxxx"/>	<input type="text" value="0x"/>	<input type="text" value="xxxxxx"/>	<input type="text" value="Autogeneration"/>	357720013455078	<input type="button" value="Manual"/>
gsm-1.4	<input type="text" value="35xxxx"/>	<input type="text" value="0x"/>	<input type="text" value="xxxxxx"/>	<input type="text" value="Autogeneration"/>	358415039164809	<input type="button" value="Manual"/>
lte-2.5	<input type="text" value="35xxxx"/>	<input type="text" value="0x"/>	<input type="text" value="xxxxxx"/>	<input type="text" value="Autogeneration"/>		<input type="button" value="Manual"/>
lte-2.6	<input type="text" value="35xxxx"/>	<input type="text" value="0x"/>	<input type="text" value="xxxxxx"/>	<input type="text" value="Autogeneration"/>	359472054874001	<input type="button" value="Manual"/>
lte-2.7	<input type="text" value="35xxxx"/>	<input type="text" value="0x"/>	<input type="text" value="xxxxxx"/>	<input type="text" value="Autogeneration"/>	352018070983487	<input type="button" value="Manual"/>
lte-2.8	<input type="text" value="35xxxx"/>	<input type="text" value="0x"/>	<input type="text" value="xxxxxx"/>	<input type="text" value="Autogeneration"/>	867732033402132	<input type="button" value="Manual"/>

As you can see, each port is set to any number. "X" means any number from 0-9. All you have to do is fill in all the lines and click "Set All". Click "Save", the current IMEI will change, this is automatically generated.

If you want to set the IMEI to a specific value, click on "Modify" and enter a new IMEI as required.

Figure 3-1-11 Manually Set IMEI



After the configuration is completed, click "Back Home" to return to the gateway interface.

Table 3-1-1 Definition of Module Settings

Options	Definition
Name	The alias of the each port. Input name without space here. Allowed characters: "-_+.<>&0-9a-zA-Z".Length: 1-32 characters.
Speaker Volume	The range of the speaker volume level is 0-100. This will adjust the loud speaker volume level by an AT command.
Microphone Volume	The range of the microphone volume is: 0-15. This will change the microphone volume level by an AT command.
Dial Prefix	The number prefix of the outbound call through the GSM channel.
PIN Code	Personal identification numbers of SIM card. PIN code can be modified to prevent SIM card from being stolen.
Custom AT	Use " " to split AT command when user use custom AT commands

commands when start	to start system.
CLIR	Caller ID restriction, this function is used to hide the caller ID of the SIM card number. The gateway will add '#31#' in front of the mobile number. This function must be operated by a professional operator.
SMS Center Number	Your SMS center number of your local carrier.
Module IMEI	Press the Modify button to automatically modify.

Save settings to other ports:

Figure 3-1-12 Save to port

Your call status will be displayed on the main screen.

Figure 3-1-13 Module Information

Port	Type	Signal	Band	BER	Carrier
1.1	GSM			-1	
Model IMEI: 352840039592814 Network Name: Network Status: Undetected SIM Card Signal Quality (0,31): -1 BER value (0,7): -1 SIM IMSI: SIM SMS Center Number: Own Number: Phone Number: Remain Time: No Limit PDD(s): 0 ACD(s): 0 ASR(%): 0 State:				-1	

3.2 Call Forwarding

Sometimes it's not convenient for you to answer the call. If you don't want to lose some important calls, you can choose call forwarding. There are different kinds of call types for you to choose, Such as call forwarding unconditional, call forwarding no reply, call forwarding busy and call forward on not reachable. If you want to cancel the call forwarding settings, you can choose to cancel all.

Figure 3-2-1 Call Forwarding

Port	Select	Call Type	Call Number	Status
gsm-1.1	<input type="radio"/>	Call Forwarding Unconditional	<input type="text"/>	
	<input type="checkbox"/>	Call Forwarding No Reply	<input type="text"/>	
	<input type="checkbox"/>	Call Forwarding Busy	<input type="text"/>	
	<input type="checkbox"/>	Call Forward on Not Reachable	<input type="text"/>	
	<input type="radio"/>	Cancel All		
gsm-1.2	<input type="radio"/>	Call Forwarding Unconditional	<input type="text"/>	
	<input type="checkbox"/>	Call Forwarding No Reply	<input type="text"/>	
	<input type="checkbox"/>	Call Forwarding Busy	<input type="text"/>	
	<input type="checkbox"/>	Call Forward on Not Reachable	<input type="text"/>	
	<input type="radio"/>	Cancel All		
gsm-1.3	<input checked="" type="radio"/>	Call Forwarding Unconditional	<input type="text"/>	
	<input type="checkbox"/>	Call Forwarding No Reply	<input type="text"/>	
	<input type="checkbox"/>	Call Forwarding Busy	<input type="text"/>	
	<input type="checkbox"/>	Call Forward on Not Reachable	<input type="text"/>	
	<input type="radio"/>	Cancel All		
gsm-1.4	<input type="radio"/>	Call Forwarding Unconditional	<input type="text"/>	
	<input type="checkbox"/>	Call Forwarding No Reply	<input type="text"/>	
	<input type="checkbox"/>	Call Forwarding Busy	<input type="text"/>	
	<input type="checkbox"/>	Call Forward on Not Reachable	<input type="text"/>	
	<input type="radio"/>	Cancel All		
lte-2.5	<input type="radio"/>	Call Forwarding Unconditional	<input type="text"/>	
	<input type="checkbox"/>	Call Forwarding No Reply	<input type="text"/>	
	<input type="checkbox"/>	Call Forwarding Busy	<input type="text"/>	
	<input type="checkbox"/>	Call Forward on Not Reachable	<input type="text"/>	
	<input type="radio"/>	Cancel All		

3.3 Call Waiting

You can switch the Call Waiting Function of the SIM card on/off in the port on this page.

Figure 3-3-1 Call Waiting

<input type="checkbox"/>	Port	<input type="radio"/> ON <input type="radio"/> OFF (Call Waiting Function)	Status
<input type="checkbox"/>	gsm-1.1	<input type="radio"/> ON <input type="radio"/> OFF	
<input type="checkbox"/>	gsm-1.2	<input type="radio"/> ON <input type="radio"/> OFF	
<input type="checkbox"/>	gsm-1.3	<input type="radio"/> ON <input type="radio"/> OFF	
<input type="checkbox"/>	gsm-1.4	<input type="radio"/> ON <input type="radio"/> OFF	
<input type="checkbox"/>	lte-2.5	<input type="radio"/> ON <input type="radio"/> OFF	
<input type="checkbox"/>	lte-2.6	<input type="radio"/> ON <input type="radio"/> OFF	
<input type="checkbox"/>	lte-2.7	<input type="radio"/> ON <input type="radio"/> OFF	
<input type="checkbox"/>	lte-2.8	<input type="radio"/> ON <input type="radio"/> OFF	

Table 3-3-1 Definition of Call Waiting

Options	Definition
Switch on call waiting	Choose the port you want to set, switch to "On" and click the "Settings" button.
Switch off call waiting	Choose the port you want to set, switch to "Off" and click the "Settings" button.
Status	It will display the results of your execution.
Settings	Used to switch the call waiting function on or off.
Query	Used to query the status of call waiting, observe whether the port that sets the call waiting is open or closed.

3.4 DTMF

You can do some DTMF Detection Settings if you choose "MODULE -> DTMF".

Figure 3-4-1 DTMF Detection Settings

DTMF Detection Settings	
DTMF Detect Flag:	<input checked="" type="checkbox"/> ON
Reference Value:	Custom
Relax DTMF Normal Twist:	6.31 8.00dB
Relax DTMF Reverse Twist:	3.98 5.99dB
DTMF Relative Peak Row:	6.3 7.99dB
DTMF Relative Peak Col:	6.3 7.99dB
DTMF Hits Begin:	2
DTMF Misses End:	3

Notice: You don't have to modify these settings if you don't have special need. You can just choose "Default".

Table 3-4-1 Definition of DTMF Detection Settings

Options	Definition
DTMF Normal Twist and Reverse Twist	It is the power difference between the row and the column energies. Normal Twist is where the Column energy is greater than the Row energy. Reverse Twist is where the Row energy is greater.
DTMF Relative Peak Row	The smaller the value, the easier the detection. If you lost some numbers, you can try to put the value down. The adjustment range is 0.02 at a time.
DTMF Relative Peak Col	The smaller the value, the easier the detection. If you lost some numbers, you can try to put the value down. The adjustment range is 0.1 at a time.
DTMF Hits Begin	Simple match value, choose 2 or 3.
DTMF Misses End	The time interval between the two numbers you entered. Adjust the speed of input. The smaller value represents the shorter intervals.

3.5 BCCH

Figure 3-5-1 BCCH Settings

Port	Mode	0			1			2			3			4			5			6			Status	Detail
		LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm		
gsm-1.1	default																							Detail
gsm-1.2	default																							Detail
gsm-1.3	default																							Detail
gsm-1.4	default																							Detail

Get Current State Search Cell

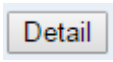
The BCCH mode can be changed by clicking the  button.

Figure 3-5-2 BCCH Mode

gsm-1.1

Port: gsm-1.1

BCCH Mode: **Default** (dropdown menu open showing: Default, Fixed, Random, Advanced)

Apply To All Ports:

Index	MCC	MNC	LAC	CID	BCCH	Receive Level	Lock

Get Current State Search Cell Save Apply Cancel

3.6 Toolkit

You can get USSD information, send AT command or check the number of this module. The AT command is very helpful for you when debugging a module.

Figure 3-6-1 Function Options

Function: **Get USSD** (dropdown menu open showing: Get USSD, Send AT Command, Check Number)

Action:

Port	Input	Output
<input type="checkbox"/> gsm-1.1	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> gsm-1.2	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> gsm-1.3	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> gsm-1.4	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> lte-2.5	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> lte-2.6	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> lte-2.7	<input type="text"/>	<input type="text"/>
<input type="checkbox"/> lte-2.8	<input type="text"/>	<input type="text"/>

Table 3-6-1 Description of Definition of Functions

Options	Definition
Get USSD	Enter a specific USSD number (For example,*142# to check your SIM card's balance. This USSD number is might be different from different carriers) to get the USSD information. The gateway will try to get by AT commands.
AT Command	To perform some specific AT commands. It is useful when you have a debug of the modem. For example, perform [AT+CSQ] to check what signal qualify it is. In AT commands, there is no difference between "a" and "A" (case insensitive).
Check Number	Enter a known number (like your mobile phone number) to check what number it is of the SIM card. Click "Execute", then the gateway will dial to the number you already input. It only rings for one time and hangs up at once. Not generating telephone charge during this procedure.

If you want to send an AT command, enter the command, select a port first and then select "Copy to selected ", and finally click "Execute".

Figure 3-6-2 AT Command Example

Function: <input type="text" value="Send AT Command"/>		
Action: <input type="text" value="AT+CSQ"/> <input type="button" value="Copy to Selected"/> <input type="button" value="Clear All"/> <input type="button" value="Execute"/>		
<input type="checkbox"/> Port	Input	Output
<input checked="" type="checkbox"/> gsm-1.1	<input type="text" value="AT+CSQ"/>	
<input checked="" type="checkbox"/> gsm-1.2	<input type="text" value="AT+CSQ"/>	
<input type="checkbox"/> gsm-1.3	<input type="text"/>	
<input type="checkbox"/> gsm-1.4	<input type="text"/>	
<input type="checkbox"/> lte-2.5	<input type="text"/>	
<input type="checkbox"/> lte-2.6	<input type="text"/>	
<input type="checkbox"/> lte-2.7	<input type="text"/>	
<input type="checkbox"/> lte-2.8	<input type="text"/>	

3.7 Module Update

You can update the module version or update the microcontroller by yourself on this page.

Figure 3-7-1 Module Update

Module Update

Port: gsm-1.1 gsm-1.2 gsm-1.3 gsm-1.4
 lte-2.5 lte-2.6 lte-2.7 lte-2.8
 All Show All

Module Update file: 未选择任何文件

MCU Update

Port: gsm-1 & gsm-2 gsm-3 & gsm-4 lte-5 & lte-6 lte-7 & lte-8
 All Show All

MCU Update file: 未选择任何文件

3.8 Call and SMS Limit

Click the edit button to set call restrictions, lock cards, and SMS restrictions. You can also view statistics on call/sms sent here.

Figure 3-8-1 Call and SMS Limit

Port	Type	Call Status	Lock Status	Mark Status	SMS Status	Actions
1.1	GSM	Unlimited	Unlocked	Unmarked	Unlimited	
1.2	GSM	Unlimited	Unlocked	Unmarked	Unlimited	
1.3	GSM	Unlimited	Unlocked	Unmarked	Unlimited	
1.4	GSM	Unlimited	Unlocked	Unmarked	Unlimited	
2.5	LTE	Unlimited	Unlocked	Unmarked	Unlimited	
2.6	LTE	Unlimited	Unlocked	Unmarked	Unlimited	
2.7	LTE	Unlimited	Unlocked	Unmarked	Unlimited	
2.8	LTE	Unlimited	Unlocked	Unmarked	Unlimited	

Port	Type	Hour Call Count	Daily Call Count	Daily Answer Count	Call Failed Count	Call Duration
1.1	GSM	0	0	0	0	0
1.2	GSM	0	0	0	0	0
1.3	GSM	0	0	0	0	0
1.4	GSM	0	0	0	0	0
2.5	LTE	0	0	0	0	0
2.6	LTE	0	0	0	0	0
2.7	LTE	0	0	0	0	0
2.8	LTE	0	0	0	0	0

SMS Sending Statistics						
Port	Type	SMS count of the day	Daily limit	SMS count of the month	Monthly limit	Monthly recovery date
<input type="checkbox"/> 1.1	GSM	0	0	0	0	0
<input type="checkbox"/> 1.2	GSM	0	0	0	0	0
<input type="checkbox"/> 1.3	GSM	0	0	0	0	0
<input type="checkbox"/> 1.4	GSM	0	0	0	0	0
<input type="checkbox"/> 2.5	LTE	0	0	0	0	0
<input type="checkbox"/> 2.6	LTE	0	0	0	0	0
<input type="checkbox"/> 2.7	LTE	0	0	0	0	0
<input type="checkbox"/> 2.8	LTE	0	0	0	0	0
<input type="button" value="Clear zero"/>			<input type="button" value="Clear zero"/>			

Call Limit

You can limit the number of daily calls, number of days connected, and number of hours of calls for the selected channel.

Figure 3-8-2 Call Limit

Call Limit (gsm-1.1)	
Call Limit Switch:	<input checked="" type="checkbox"/> ON
Limit Daily Call Times:	<input type="text" value="0"/>
Limit Daily Answer Times:	<input type="text" value="0"/>
Limit Hour Call Times:	<input type="text" value="0"/>

Call Limit Time

Now we offer you two types of call duration limit, you can choose “Single Call Duration Limit” or “Call Duration Limitation” to control your calling time.

This will limit the time of each call. First you need to switch “Enable” on, then you can set “Step” and “Single Call Duration Limitation” any digits you want. When you make a call through this port, your calling time will be limited.

- **Single Call Duration Limitation:** If your calling time exceeds the set value, the system will hang up the call. Multiplying the step size by a single call time is to allow a single call duration.

Figure 3-8-3 Single Settings

Call Limit Time	
Call Time Limit Switch:	<input checked="" type="checkbox"/> ON
Step:	<input type="text" value="60"/> Second
Enable Single Call Duration Limit:	<input checked="" type="checkbox"/> ON
Single Call Duration Limitation:	<input type="text" value="0"/>
Enable Call Duration Limitation:	<input type="checkbox"/> OFF

- **Call Duration Limitation:** This will limit your total calling time of this port. If remain time is 0, it will not send calls through this port.

Figure 3-8-4 Call Duration Limitation Settings

Call Limit Time	
Call Time Limit Switch:	<input checked="" type="checkbox"/> ON
Step:	60 Second
Enable Single Call Duration Limit:	<input checked="" type="checkbox"/> ON
Single Call Duration Limitation:	0
Enable Call Duration Limitation:	<input checked="" type="checkbox"/> ON
Call Duration Limitation:	10
Minimum Charging Time:	30 Second
Alarm Threshold:	2
Alarm Phone Number:	18610001000
Alarm Description:	test
Remain Time:	0 <input type="button" value="Reset"/>
Enable Auto Reset:	<input checked="" type="checkbox"/> ON
Auto Reset Type:	Week(7Day) ▼
Next Reset Time:	2019-04-01 16:29:35

The same as the algorithm for the single call duration limit, the total call time of the port cannot exceed the product of the "Step" and the "Call Duration Limitation". The Minimum Billing Time value must be less than the step size.

You can set a value for the Alarm Threshold. When the remaining call duration of the port reaches the set value, the gateway will send alarm info to the designated phone. You can also turn on enable auto reset, which can be selected from one day, one week or one month. After the automatic reset time is reached, the remaining call duration is reset to the set total call duration.

Figure 3-8-5 Auto Reset Settings

Enable Auto Reset:	<input checked="" type="checkbox"/> ON
Auto Reset Type:	Week(7Day) ▼
Next Reset Time:	2019-04-01 16:29:35

Table 3-8-1 Description of Call Duration Limit Settings

Options	Definition
Step	The value range of the step length is 1-999s. Multiplying the step size by a single call time is to allow a single call duration.
Enable Single Call Duration Limit	The value of the single call duration limit, ranging from 1-999999. The step size multiplied by the single talk time is the single call duration.

Enable Call Duration Limitation	The value of the total call duration limit, in the range of 1-999999. The step size multiplied by the total talk time is the total call duration.
Minimum Charging Time	After a period of conversation, the operator starts billing, in seconds. This value must be less than the step size.
Alarm Threshold	The threshold for the duration of the call. When the duration of the call is equal or less than this value, the gateway will send an alert message to the specified phone number via SMS (only once before the total call duration is reset). This value must be less than the call duration limit.
Alarm Phone Number	User will received alarm message from the gateway when receiving alarm phone number.
Alarm Description	The alarm port information description sends the alarm information to the user's phone.
Remaining call duration	Multiplying this value by the step size is the remaining call duration.
Enable Auto Reset	The remaining talk time is automatically restored, that is, the total number of minutes of the call is restored.
Auto Reset Type	Reset call minutes by date, by week, by month.
Next Reset Time	Define the next reset date. The system will count from that date and be set as a reset period.

Lock Card

The lock card detection switch is a switch function of the lock card. After opening, the call failure lock card condition parameter needs to be set. After the lock card condition is reached, the SIM card is disabled and cannot be allocated for use unless the card is removed, the gateway is restarted, and the manual is unlocked (Duration limit requires manual reset), turn off the lock card function, etc.

Figure 3-8-6 Lock Card

Lock Sim	
Lock Detect Switch:	<input checked="" type="checkbox"/> ON
Mark Switch:	<input checked="" type="checkbox"/> ON
Call Failed Mark Count:	<input type="text" value="3"/>
Call Failed Lock Switch:	<input checked="" type="checkbox"/> ON
Call Failed Lock Count:	<input type="text" value="3"/>
SMS Send Detection Switch:	<input checked="" type="checkbox"/> ON
SMS Send Detection Count:	<input type="text" value="10"/>
Send Sms Number:	<input type="text" value="13810001000"/>
Sms Message:	<input type="text" value="test s"/>
Testing SMS report:	<input checked="" type="checkbox"/> ON

Table 3-8-2 Description of Lock Card

Options	Definition
Call Failed Mark Count	The number of consecutive calls failed to reach the number of settings, making the port.
Called Failed Lock Count	Lock the port after the number of successive call failures reaches the set number, restrict the port expiration, and no longer select the port expiration in the group policy.
SMS Send Detection Switch	After opening, when the number of successive call failures reaches the set value, send message to check whether the port is available; if the message send successfully, it will clear the number of successive call failures; if the message fails, it will limit the port's outgoing.
Testing SMS report	When closed, the successful sending of message indicates that the port is available; when opened, the successful

	<p>sending of message and the receipt of message report indicate that the port is available.</p>
--	--

SMS Limit

Figure 3-8-7 SMS Limit

SMS Limit	
SMS Limit Switch	<input checked="" type="checkbox"/> ON
SMS Limit Success Flag	<input checked="" type="checkbox"/> ON
Day Limit SMS Count	<input type="text" value="0"/>
Month Limit SMS Count	<input type="text" value="5"/>
SMS Clean Date	<input type="text" value="1"/>

Table 3-8-3 Definition of SMS Limit

Options	Definition
SMS Limit Success Flag	<p>When closed, no matter whether the message is sent successfully or not, the number of messages is counted ;</p> <p>When opened, the number of messages is counted when the message is sent successfully.</p>
Day Limit SMS Count	<p>Limit the number of SMS sent daily, the default value is 0, indicating unlimited.</p>
Month Limit SMS Count	<p>Limit the number of SMS sent per month, the default value is 0, indicating unlimited.</p>
SMS Clean Date	<p>The number of messages sent monthly is automatically cleared at 0 points, 0 minutes and 0 seconds on the set date.</p>

4. VOIP

4.1 VOIP Endpoints

This page shows everything about your SIP&IAX2, you can see the status of each SIP&IAX2.

Figure 4-1 SIP&IAX2 Endpoints

SIP Endpoint				
<input type="checkbox"/>	Endpoint Name	Registration	Credentials	Actions
<input type="checkbox"/>	10028	none	anonymous@172.16.208.33	
<input type="checkbox"/>	2111	server	test	

Add New SIP Endpoint

IAX2 Endpoint				
<input type="checkbox"/>	Endpoint Name	Registration	Credentials	Actions
<input type="checkbox"/>	2133	client	2133@172.16.208.33	

Add New IAX2 Endpoint

4.1.1 Add New SIP Endpoint

Main SIP Endpoint Settings:

You can click button to add a new SIP endpoint, and if you

want to modify existed endpoints, you can click button.

There are 3 kinds of registration types for you to choose: None, Server or Client. You can configure as follows:

If you set up a SIP endpoint by registration “None” to a server, then you can’t register other SIP endpoints to this server. (If you add other SIP endpoints, this will cause Out-band Routes and Trunks confused.)

Figure 4-1-1 None Registration

Main Endpoint Settings	
Name:	<input type="text" value="6666"/>
User Name:	<input type="text"/> <input checked="" type="checkbox"/> Anonymous
Password:	<input type="text"/>
Registration:	<input type="text" value="None"/>
Hostname or IP Address:	<input type="text" value="172.16.200.20"/>
Transport:	<input type="text" value="UDP"/>
NAT Traversal:	<input type="text" value="Yes"/>

For convenience, we have designed a method that you can register your SIP endpoint to your gateway, thus your gateway just work as a server.

Figure 4-1-2 Server Registration

The screenshot shows the 'Main Endpoint Settings' form. The 'Registration' dropdown menu is highlighted with a red box and is set to 'Server'. Other fields include Name: 10027, User Name: 10027, Password: masked with asterisks, Hostname or IP Address: dynamic, Transport: UDP, and NAT Traversal: Yes.

Also you can choose registration by “This gateway registers with the endpoint”, it’s the same with “None”, except the name and the password.

Figure 4-1-3 Client Registration

The screenshot shows the 'Main Endpoint Settings' form. The 'Registration' dropdown menu is highlighted with a red box and is set to 'Client'. Other fields include Name: 10027, User Name: 10027, Password: masked with asterisks, Hostname or IP Address: 172.16.80.16, Transport: UDP, and NAT Traversal: Yes.

Table 4-1-1 Definition of SIP Options

Options	Definition
Name	A name is easier to understand, only for reference, does not participate in SIP authentication.
Username	Register name in your SIP server.
Password	Authenticating with the gateway and characters are allowed.
Registration	Whether the terminal registers with the gateway or the gateway registers with the terminal.

	<p>None --- Not registering;</p> <p>Server --- When register as this type, it means the gateway acts as a SIP server, and SIP endpoints register to the gateway;</p> <p>Client --- When register as this type, it means the gateway acts as a client, and the endpoint should be register to a SIP server;</p>
<p>Hostname or IP Address</p>	<p>IP address or hostname of the endpoint or 'dynamic' if the endpoint has a dynamic IP address. This will require registration.</p> <p>Notice: If you entered the domain name and changed the DNS, you must restart asterisk.</p>
<p>Transport</p>	<p>This sets the possible transport types for outgoing. Order of usage, when the respective transport protocols are enabled, is UDP, TCP, TLS.</p> <p>The first enabled transport type is only used for outbound messages until a Registration takes place. During the peer registration the transport type may change to another supported type if the peer requests so.</p>
<p>NAT Traversal</p>	<p>No --- Use report if the remote needs it.</p> <p>Force Report on—Force report on.</p> <p>Yes --- Force the report to open and perform commedia RTP processing.</p> <p>Report if requested and commedia --- If the remote needs it, use report and perform commedia RTP processing.</p>

Advanced: Registration Options

Figure 4-1-4 Advanced Registration Options

The screenshot shows a configuration interface for 'Advanced:Registration'. It contains the following fields and options:

- Authentication User: 108
- Register Extension: 108 (with a 'Modify' checkbox)
- Register User: 108 (with a 'Modify' checkbox)
- Contact User: (with a 'Modify' checkbox)
- From User: (with a 'Modify' checkbox)
- From Domain: (empty)
- Port: 5060
- Qualify: No (dropdown menu)
- Qualify Frequency: 60
- Outbound Proxy: (empty)
- Custom Registry: OFF (checkbox)
- Enable Outboundproxy to Host: OFF (checkbox)

Table 4-1-2 Definition of Registration Options

Options	Definition
Authentication User	Just a username used when registering.
Register Extension	When Gateway registers as a SIP user agent to a SIP proxy (provider), calls from this provider connect to this local extension.
Register User	Register user name, it is the user of register => user[:secret[:authuser]]@host[:port][/extension].
Contact User	e.g. When the Contact User is 402 Contact: <sip:402@172.16.6.123:5060;transport=UDP>
From User	A username to identify the gateway to this endpoint.
From Domain	A domain to identify the gateway to this endpoint.
Port	The port number used by the gateway to link the peer.

Qualify	Whether to check the link status of the peer.
Qualify Frequency	How often, in seconds, to check the endpoint's connection status.
Outbound Proxy	A proxy to which the gateway will send all outbound signaling instead of sending signaling directly to endpoints.
Custom Registry	After opening, customers can customize their own registry.
Registry String	Customers can enter their own registration information. The format is:: user[:secret[:authuser]]@host[:port][[/extension] e.g.: 1001@sip.com:pbx122@172.16.6.122/1001.
Enable Outbound proxy to Host	Outbound proxy to Host On / Off.

Call Settings

Figure 4-1-5 Call Settings

Table 4-1-3 Description of DTMF Detection Settings

Options	Definition
DTMF Mode	Set default DTMF Mode for sending DTMF. Default:

	rfc2833. Other options: 'info', SIP INFO message(application/dtmf-relay); 'Inband', Inband audio (require 64kbit codec -alaw, ulaw).
Trust Remote-Party-ID	Whether or not the Remote-Party-ID header should be trusted.
Send Remote-Party-ID	Whether or not to send the Remote-Party-ID header.
Remote-Party-ID Format	How to set the Remote-Party-ID header: from Remote-Party-ID or from P-Asserted-Identity.
Caller ID Presentation	Whether or not to display the Caller ID.
Call Limit	Usually used when this sip works as a trunk. Limit the maximum number of channels supported by sip trunks.

Advanced: Signaling Settings

Figure 4-1-6 Signaling Settings

▼ Advanced: Signaling

Progress Inband:	<input type="text" value="Never"/>
Append user=phone to URI:	<input type="text" value="No"/>
Add Q.850 Reason Headers:	<input type="text" value="No"/>
Ignor SDP Version:	<input type="text" value="Yes"/>
Directmedia:	<input type="text" value="Yes"/>
Allow Transfers:	<input type="text" value="Yes"/>
Allow Promiscuous Redirects:	<input type="text" value="No"/>
Max Forwards:	<input type="text" value="70"/>
Send TRYING on REGISTER:	<input type="text" value="No"/>

Table 4-1-4 Definition of Signaling Options

Options	Definition
Progress Inband	<p>Whether the tone is ringing.</p> <p>Never: Indicates that the incoming calls are never applied.</p> <p>Option values: yes, no, never.</p> <p>Default: never.</p>
Append user=phone to URI	<p>Whether or not to add 'user=phone' to URIs that contain a valid phone number.</p>
Add Q.850 Reason Headers	<p>Whether or not to add a reason header and use it if is available.</p>
Ignore SDP Version	<p>By default, the gateway will honor the session version number in SDP packets and will only modify the SDP session if the version number changes.</p> <p>Switch this option off to force the gateway to ignore the SDP session version number and treat all SDP data as new data.</p> <p>This is required for devices that send non-standard SDP packets (observed with Microsoft OCS).</p> <p>By default this option is on.</p>
Direct media	<p>The value of parameter direct media is one of them (no, yes, nonat, update, outgoing).</p> <p>Default value is yes.</p>
Allow Transfers	<p>Whether or not to globally enable transfers.</p> <p>Choosing 'no' will disable all transfers (unless enabled in peers or users).</p> <p>Default is enabled.</p>

Allow Promiscuous Redirects	Whether or not to allow 302 or REDIR to non-local SIP address. Notice: Redirecting the local system causes a loop call, which is not supported by asterisk.
Max Forwards	Setting for the SIP Max-Forwards header (loop prevention).
Send TRYING on REGISTER	Send a 100 Trying when the endpoint registers.

Advanced: Timer Settings

Figure 4-1-7 Timer Settings

Advanced:Timer Settings

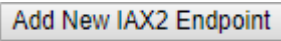
Default T1 Timer:	500
Call Setup Timer:	32000
Session Timers:	Accept ▼
Minimum Session Refresh Interval:	90
Maximum Session Refresh Interval:	1800
Session Refresher:	UAS ▼

Table 4-1-5 Definition of Timer Options

Options	Definition
Default T1 Timer	This timer is used primarily in INVITE transactions. The default for Timer T1 is 500 ms or the measured run-trip time between the gateway and the device if you have qualify=yes for the device.
Call Setup Timer	If a provisional response is not received in this amount of time, the call will auto congest. The default is 64*T1.

Session Timers	There are three modes for you to choose: originate, request and run session-timers always; Accept or run the session timer only when requested by another user agent; Refuse, Do not run session timers in any case.
Minimum Session Refresh Interval	Minimum session refresh interval in seconds. The default is 90 secs.
Maximum Session Refresh Interval	Maximum session refresh interval in seconds. The default is 1800 secs.
Session Refresher	The session refresher, UAC or UAS. The default is UAS.

4.1.2 Add New IAX2 Endpoint

You can click  button to add a new IAX2 endpoint and if you

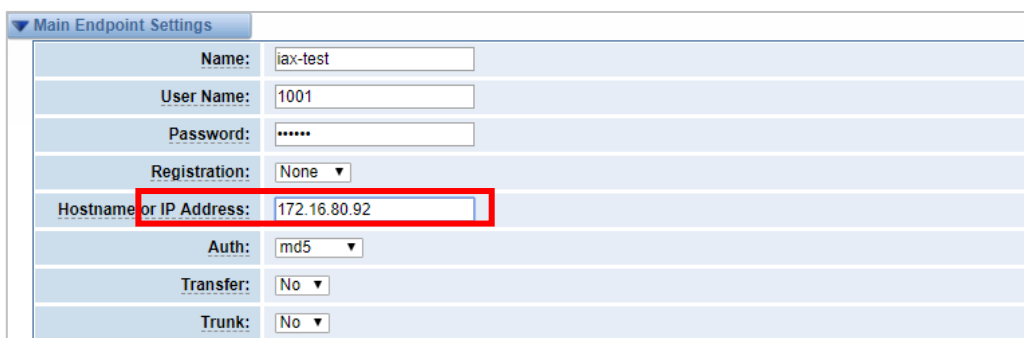
want to modify existed endpoints, you can click  button.

There are three registration types: None, Server or Client.

You can configure as follows:

If you set up an IAX2 endpoint by registration “None” to a server, then you can’t register other IAX2 endpoints to this server, just authenticate the username and password.

Figure 4-1-8 None Registration



The screenshot shows a web form titled "Main Endpoint Settings" with the following fields:

- Name: lax-test
- User Name: 1001
- Password: *****
- Registration: None (dropdown menu)
- Hostname or IP Address: 172.16.80.92 (text input field, highlighted with a red box)
- Auth: md5 (dropdown menu)
- Transfer: No (dropdown menu)
- Trunk: No (dropdown menu)

For convenience, you can choose registration by “Server”, thus your gateway just work as a server.

Figure 4-1-9 Server Registration

The screenshot shows the 'Main Endpoint Settings' form. The 'Registration' dropdown menu is highlighted with a red box and is set to 'Server'. Other fields include Name: iax-test, User Name: 1001, Password: masked with asterisks, Hostname or IP Address: dynamic, Auth: md5, Transfer: No, and Trunk: No.

Also you can choose registration by “Client”, it will work as a Client.

Figure 4-1-10 Client Registration

The screenshot shows the 'Main Endpoint Settings' form. The 'Registration' dropdown menu is highlighted with a red box and is set to 'Client'. Other fields include Name: iax-test, User Name: 1001, Password: masked with asterisks, Hostname or IP Address: 172.16.80.92, Auth: md5, Transfer: No, and Trunk: No.

Table 4-1-6 Definition of IAX2 Options

Options	Definition
Name	A name which is able to read by human. And it's only used for user's reference.
Username	The endpoint will use username to authenticate with the gateway.
Password	The endpoint will use password to authenticate with the gateway.
Registration	Whether this endpoint will register to this gateway or this gateway to the endpoint.

	<p>None --- Not registering;</p> <p>Endpoint registers with this gateway --- When register as this type, it means the gateway acts as a IAX2 server, and IAX2 endpoints register to the gateway;</p> <p>This gateway registers with the endpoint --- When register as this type, it means the gateway acts as an IAX2 client, and the endpoint should be register to an IAX2 server.</p>
<p>Hostname or IP Address</p>	<p>IP address or hostname of the endpoint or 'dynamic' if the endpoint has a dynamic IP address. This will require registration.</p> <p>Notice: If you entered the domain name and changed the DNS, you must restart asterisk.</p>
<p>Auth</p>	<p>There are three authentication methods that are supported: md5, plaintext and rsa. The least secure is "plaintext", which sends passwords clear text across the net. Uses a challenge/response md5 sum arrangement, but still requires both ends have plain text access to the secret. "rsa" allows unidirectional secret knowledge through public/private keys.</p>
<p>Transfer</p>	<p>Disable or not IAX2 native transfer.</p>
<p>Trunk</p>	<p>Use IAX2 trunking with this host.</p>

Advanced: Registration Options

Figure 4-1-11 Registration Options

Advanced:Registration Options	
Qualify:	Yes ▾
Qualify Smoothing:	Yes ▾
Qualify Freq Ok:	6000
Qualify Freq Not Ok:	6000
Port:	4569
Require Call Token:	Yes ▾

Table 4-1-7 Definition of Registration Options

Options	Definition
Qualify	Whether or not to check the endpoint's connection status.
Qualify Smoothing	Use the average of the last two connection results to reduce the lag end of the misdetection. The default is no.
Qualify Freq Ok	How frequently to ping the peer when everything seems to be OK, in milliseconds.
Qualify Freq Not Ok	How frequently to ping the peer when it is either, LAGGED or UNAVAILABLE, in milliseconds.
port	The port number the gateway will connect to at this endpoint. Default is 4569.

IAX2 Encryption

Figure 4-1-12 IAX2 Encryptions

The screenshot shows a configuration window titled 'IAX2 Encryption'. It contains two rows of settings:

- Encryption:** A dropdown menu set to 'No'.
- Force Encryption:** A dropdown menu set to 'No'.

Table 4-1-8 Definition of Encryption Options

Options	Definition
Encryption	Enable IAX2 encryption. The default is no.
Force Encryption	Force encryption insures no connection is established unless both sides support encryption. By switching this option on, encryption is automatically switch on as well. The default is no.

IAX2 Trunk Settings

Figure 4-1-13 IAX2 Trunk Settings

The screenshot shows a configuration window titled 'IAX2 Trunk settings'. It contains several rows of settings:

- Trunk Max Size:** 128000
- Trunk MTU:** 0
- Trunk Frequency:** 20
- Trunk Time Stamps:** No
- Min. RegExpire:** 60
- Max. RegExpire:** 60

Table 4-1-9 Definition of Trunk Options

Options	Definition
Trunk Max Size	The default is 128000 bytes, which supports up to 800 calls of ulaw at 20ms a frame.
Trunk MTU	Set the maximum transmission unit for IAX2 UDP trunk.

Trunk Frequency	How frequently to send trunk messages in msec. This is 20ms by default.
Trunk Time Stamps	Ensure that frame timestamps get sent end-to-end properly.
Min. RegExpire	Minimum amounts of time that IAX2 peers can request as a registration expiration interval in seconds.
Max. RegExpire	Maximum amounts of time that IAX2 peers can request as a registration expiration interval in seconds.

4.2 Batch SIP Endpoints

On this page, you can create multiple SIP Endpoints at the same time.

Figure 4-2-1 Multiple SIP Endpoints Settings

<input type="checkbox"/>	ID	User Name	Password	Hostname or IP Address	Port	Register Mode
<input type="checkbox"/>		<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	client ▼
<input type="checkbox"/>	1	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	client ▼
<input type="checkbox"/>	2	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	client ▼
<input type="checkbox"/>	3	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	client ▼
<input type="checkbox"/>	4	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	client ▼
<input type="checkbox"/>	5	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	client ▼
<input type="checkbox"/>	6	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	client ▼
<input type="checkbox"/>	7	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	client ▼
<input type="checkbox"/>	8	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	client ▼

You should select the number of SIPs to be created after you filling in the username, password, domain name or IP address, port and registration mode in the first line. You can create up to the same number of SIP endpoints as the number of device ports at a time; Then choose whether to tick “AutoPassword” button. After the above configuration is completed, click Batch Settings, and then save to create SIP endpoints in batches.

Table 4-2-1 Definition of Multiple SIP Endpoints

Options	Definition
Username	The display name is the same as the registered name on the SIP server by default.
Password	Authenticating with the gateway and characters are allowed.
Hostname or IP Address	IP address or hostname of the endpoint or 'dynamic' if the endpoint has a dynamic IP address. This will require registration.
Port	Port number registration.
Registration	<p>None --- Not registering;</p> <p>Server --- When register as this type, it means the gateway acts as a SIP server, and SIP endpoints register to the gateway;</p> <p>Client --- When register as this type, it means the gateway acts as a client, and the endpoint should be register to a SIP server.</p>
AutoPassword	<p>Tick - Automatically increments based on the password entered in the first line.</p> <p>Do not tick- All SIP endpoints have the same password as the first one.</p>

4.3 Advanced SIP Settings

Networking

Figure 4-3-1 Networking General

The screenshot shows a configuration window titled 'Networking' with a sub-tab 'General'. The settings are as follows:

- UDP Bind Port: 5060
- Enable TCP: No
- TCP Bind Port: 5060
- TCP Authentication Timeout: (empty)
- TCP Authentication Limit: (empty)
- Enable Hostname Lookup: No
- Enable Internal SIP Call: No
- Internal SIP Call Prefix: (empty)

Table 4-3-1 Definition of Networking General Options

Options	Definition
UDP Bind Port	Choose a port on which to listen for UDP traffic.
Enable TCP	Enable server for incoming TCP connection (default is no).
TCP Bind Port	Choose a port on which to listen for TCP traffic.
TCP Authentication Timeout	The maximum number of seconds a client has to authenticate. If the client does not authenticate before this timeout expires, the client will be disconnected (default value is: 30 seconds).
TCP Authentication Limit	The maximum number of unauthenticated sessions that will be allowed to connect at any given time (default is: 50).
Enable Hostname Lookup	Enable DNS SRV lookups on outbound calls. Notice: The gateway uses the first host in the SRV

	record. This function can be used in dial-up activation to dial SIP calls on the Internet via domain name.
Enable Internal SIP Call	Whether enable the internal SIP calls or not when you select the registration option "Endpoint registers with this gateway".
Internal SIP Call Prefix S	Specify a prefix before routing the internal calls.

Figure 4-3-2 NAT Settings

Table 4-3-2 Definition of NAT Settings Options

Options	Definition
Local Network	Format:192.168.0.0/255.255.0.0 or 172.16.0.0./12 A list of IP address or IP ranges which are located inside a NAT network. This gateway will replace the internal IP address in SIP and SDP messages with the external IP address when a NAT exists between the gateway and other endpoints.
Local Network List	Add local IP address list.
Subscribe Network	Through the use of the test_stun_monitor module, the

<p>Change Event</p>	<p>gateway has the ability to detect when the perceived external network address has changed. When the <code>stun_monitor</code> is installed and configured, <code>chan_sip</code> will renew all outbound registrations when the monitor detects any sort of network change has occurred. By default this option is enabled, but <code>res_stun_monitor</code> only takes effect once. If <code>res_stun_monitor</code> is enabled and you don't want changes on the network to cause an outbound registration, use the option below to disable this feature.</p>
<p>Match External Address Locally</p>	<p>Only substitute the <code>extern</code> address or <code>extern host</code> setting if it matches.</p>
<p>Dynamic Exclude Static</p>	<p>Dynamic hosts are not allowed to register with the static host's IP address, which will avoid the same IP registration error.</p>
<p>Externally Mapped TCP Port</p>	<p>The externally mapped TCP port, when the gateway is behind a static NAT or PAT.</p>
<p>External Address</p>	<p>The external address (and optional TCP port) of the NAT. External Address = <code>hostname[:port]</code>. Specifies a static <code>address[:port]</code> to be used in SIP and SDP messages. Examples: External Address = <code>12.34.56.78</code> External Address = <code>12.34.56.78:9900</code>.</p>
<p>Hostname Refresh Interval</p>	<p>How often to perform a hostname lookup. This can be useful when your NAT device lets you choose the port mapping, but the IP address is dynamic. Beware, you</p>

	might suffer from service disruption when the name server resolution fails.
--	---

Figure 4-3-3 RTP Settings

RTP Settings	
Start of RTP Port Range:	<input type="text" value="10000"/>
End of RTP port Range:	<input type="text" value="20000"/>
RTP Timeout:	<input type="text" value="120"/>

Table 4-3-3 Definition of RTP Settings Options

Options	Definition
Start of RTP Port Range	Start of port numbers to be used for RTP.
End of RTP port Range	End of port numbers to be used for RTP.

Parsing and compatibility

Figure 4-3-4 Parsing and Compatibility

▼ Parsing and Compatibility

General

Strict RFC Interpretation: ▼

Send Compact Headers: ▼

SDP Owner:

Ring 183 Mode: ▼

SIP Methods

Disallowed SIP Methods

- ACK
- BYE
- CANCEL
- INFO
- INVITE
- MESSAGE
- NOTIFY
- OPTIONS
- PRACK
- PUBLISH
- REFER
- REGISTER
- SUBSCRIBE
- UPDATE

Hangup Cause Code: ▼

Notify Unlimited: OFF

Caller ID	
Shrink Caller ID:	No ▾
Caller ID:	SIP From ▾ Number ▾
SIP From:	Tel/Tel ▾
Callee ID	
SIP To:	Tel/Tel ▾
Callee ID:	EXTEN ▾
Permit Dialing Letters:	<input type="checkbox"/> OFF
Timer Configuration	
Maximum Registration Expiry:	<input type="text"/>
Minimum Registration Expiry:	<input type="text"/>
Default Registration Expiry:	<input type="text"/>
Outbound Registrations	
Registration Timeout:	20 <input type="text"/>
Number of Registration Attempts:	0 <input type="text"/>
Transform Local Port	
Client Auto Flag:	<input type="checkbox"/> OFF

Table 4-3-4 Instruction of Parsing and Compatibility

Options	Definition
Strict RFC Interpretation	Check header tags, character conversion in URIs, and multiline headers for strict SIP compatibility (default is yes).
Send Compact Headers	Send compact SIP headers.
SDP Owner	Allows you to change the username filed in the SDP owner string. This filed MUST NOT contain spaces
Disallowed SIP Methods	When a dialog is started with another SIP endpoint, the other endpoint should include an Allow header telling us what SIP methods the endpoint implements. However, some endpoints either do not include an Allow header or lie about what methods they implement. In the former case, the gateway makes the assumption that the endpoint supports all known SIP methods. If you know

	<p>that your SIP endpoint does not provide support for a specific method, then you may provide a list of methods that your endpoint does not implement in the disallowed methods option. Note that if your endpoint is truthful with its Allow header, then there is no need to set this option.</p>
Shrink Caller ID	<p>The Shrink Caller Id function removes '(', ' ', ')', non-trailing '.', and '-' not in square brackets. For example, the caller id value 555.5555 becomes 5555555 when this option is enabled. By default this option is on.</p>
Caller ID	<p>Default: SIP From and Number.</p> <p>For example: When selecting SIP From, Name is Peter and Number is 402.</p> <p>The From mode is: "Peter"<sip:402@172.16.6.239;transport=UDP>;tag=bd481672.</p>
Callee ID	<p>Default: EXTEN</p> <p>For example: When selecting SIP To, Name is Jason and Number is 401.</p> <p>The mode is: "Jason"<sip:401@172.16.6.239;transport=UDP>.</p>
Maximum Registration Expiry	<p>Maximum allowed time of incoming registrations and subscriptions (seconds).</p>
Minimum Registration	<p>Minimum length of registrations/subscriptions (default 60).</p>

Expiry	
Default Registration Expiry	Default length of incoming/outgoing registration.
Registration Timeout	How often, in seconds, to retry registration calls. Default 20 seconds.
Number of Registration	Number of registration attempts before we give up.

Security

Figure 4-3-5 Security Settings

The screenshot shows a 'Security' settings panel. Under 'Authentication Settings', there are five rows: 'Match Auth Username' (No), 'Realm' (text input), 'Use Domain as Realm' (No), 'Always Auth Reject' (No), and 'Authenticate Options Requests' (No). Under 'Guest Calling', there is one row: 'Allow Guest Calling' (No).

Table 4-3-5 Instruction of Security

Options	Definition
Match Auth Username	If available, match user entry using the 'username' field from the authentication line instead of the 'from' field.
Realm	Realm for digest authentication. All realms must be globally unique according to the RFC3261 standard and can generally be set to hostname or domain name.

Use Domain as Realm	Use the domain from the SIP Domains setting as the realm.
Always Auth Reject	When an INVITE or REGISTER request is rejected for any reason, the same reason is always used, the username is legal but the password is incorrect. It does not tell the requester whether there is this user or peer, which will reduce the possibility of the attacker scanning the SIP account. This parameter is enabled by default.
Authenticate Options Requests	Enabling this option will authenticate OPTIONS requests just like INVITE requests are. By default this option is disabled.
Allow Guest Calling	Allow or reject customer calls (default yes, allowed). If your gateway is connected to the Internet and allows customers to call, you want to check which services are available to everyone, by enabling them in the default context.

Media

Figure 4-3-6 Media Settings



Table 4-3-6 Instruction of Media

Options	Definition
TOS for SIP Packets	Sets type of service for SIP packets.

TOS for RTP Packets	Sets type of service for RTP packets.
---------------------	---------------------------------------

Codec Settings

Select codecs from the list below:

Figure 4-3-7 Codec Settings

▼ Codec Settings	
Codec Priority 1:	G.711 u-law ▼
Codec Priority 2:	G.711 a-law ▼
Codec Priority 3:	GSM ▼
Codec Priority 4:	G.722 ▼
Codec Priority 5:	G.723 ▼
Codec Priority 6:	G.726 ▼
Codec Priority 7:	G.729 ▼

4.4 Advanced IAX2 Settings

General Settings

Figure 4-4-1 General Settings

▼ General Settings	
Bind Port:	4569
Bind Address:	0.0.0.0
Enable IAXCompat:	No ▼
Enable Nochecksums:	No ▼
Enable Delay Reject:	No ▼
ADSI:	No ▼
SRV Loopup:	No ▼
AMA Flags:	default ▼
Auto Kill:	Yes ▼
Lauguage:	English ▼
Account Code:	
Call Token Optional:	
Description:	

Table 4-4-1 Instruction of General

Options	Definition
Bind Port	Bind port and bindaddr may be specified
Enable IAXCompat	Set laxcompat to yes if you need to use a tiered switch, or if some of the latency may occur when performing a lookup in a dial plan, otherwise enabling it will result in a small performance conflict.
Enable Nochecksums	Disable UDP checksums (if Nochecksums is set, then Nochecksums will be calculated/checked on systems supporting this feature).
ADIS	ADSI (Analog Display Services Interface) can be enabled if you have (or may have) ADSI compatible CPE equipment.
SRV Lookup	Whether or not to perform an SRV lookup on outbound calls.
AMA Flags	You may specify a global default AMA flag for IAX internal calls. These flags are used in the generation of call detail records.
Auto Kill	If we don't get ACK to our NEW within 2000ms, and Auto Kill is set to yes, then we cancel the whole thing (that's enough time for one retransmission only). This is used to keep things from stalling for a long time for a host that is not available, However, this function is not friendly in the case of poor connection status.
Language	You may specify a global default language for users. This can be specified also on a per-user basis. If you ignore this item, it will use the default English language.

Account Code	You may specify a default account for Call Detail Records (CDRs) in addition to specifying on a per-user basis.
--------------	---

Music on Hold

Figure 4-4-2 Music on Hold Settings

The screenshot shows a configuration panel titled "Music On Hold". It contains two rows of settings:

- Mohsuggest:** A dropdown menu currently set to "default".
- Mohinterpret:** A dropdown menu currently set to "default".

Table 4-4-2 Instruction of Music on Hold

Options	Definition
Mohsuggest	Specifies the type of waiting music that plays while the channel is on hold. It can specify global settings or be specified by the user or each peer.
Mohinterpret	Set the content of the channel playback when the phone is suspended. If it is "default", the music specified in the dialing rules will be played in the channel when the phone hangs. If it is "transfer", it will be replaced by a message signal.

Instruction of Codec Settings

Figure 4-4-3 Codec Settings

The screenshot shows a configuration panel titled "Codec Settings". It contains several rows of settings:

- Band Width:** A dropdown menu set to "low".
- Disallow:** A dropdown menu set to "all".
- Allow:** A section containing six priority dropdown menus:
 - Priority 1: GSM
 - Priority 2: G.711 u-law
 - Priority 3: G.711 a-law
 - Priority 4: G.722
 - Priority 5: G.723
 - Priority 6: G.729
- Codec Priority:** A dropdown menu set to "host".

Table 4-4-3 Instruction of Codec Settings

Options	Definition
Band Width	Specify bandwidth of low, medium, or high to control which codes are used in general.
Disallow	Adjust the type of codec to be switched off.
Allow	Adjust the type of codec to be switched on and the priority.
Codec Priority	Codec priority controls the codec negotiation of an inbound IAX2 call. This option is inherited to all user entities. It can also be defined in each user entity separately which will override the setting in general.

Jitter Buffer Settings

Figure 4-4-4 Jitter Buffer



The screenshot shows a configuration window titled "Jitter Buffer Settings". It contains several parameters:

- Jitter Buffer:** A dropdown menu set to "No".
- Force Jitter Buffer:** A dropdown menu set to "No".
- Max Jitter Buffers:** An empty text input field.
- Resyncthreshold:** An empty text input field with a tooltip that reads "Resyncing can be disabled by setting this parameter to -1."
- Max Jitter Interps:** An empty text input field.
- Jitter Target Extra:** An empty text input field.

Table 4-4-4 Instruction of Jitter Buffer

Options	Definition
Jitter Buffer	Whether to set a global buffer should be enabled by an unstable network environment, generally you do not need. Because the terminal device buffers the jitter processing. The default is "no".

Force Jitter Buffer	In the ideal world, when we bridge VoIP channels we don't want to jitter buffering on the switch, since the endpoints can each handle this. However, some endpoints may have poor jitter buffers themselves, so this option will force to always jitter buffer, even in this case. The default is "no".
Max Jitter Buffers	A maximum size for the jitter buffer.
Resyncthreshold	When the jitter buffer notices a significant change in delay that continues over a few frames, it will resync, assuming that the change in delay was caused by a timestamping mix-up. The threshold for noticing a change in delay is measured as twice the measured jitter plus this resync threshold. Resyncing can be disabled by setting this parameter to -1.
Max Jitter Interps	The maximum number of interpolation frames the jitter buffer should return in a row.
Jitter Target Extra	Number of milliseconds by which the new jitter buffer will pad its size. The default is 40, so without modification, the new jitter buffer will set its size to the jitter value plus 40 milliseconds. Increasing this value may help if your network normally has low jitter, but occasionally has spikes.

Misc Settings

Figure 4-4-5 Misc Settings

▼ Misc Settings	
IAX2 Thread Count:	<input type="text"/>
IAX2 Max Thread Count:	<input type="text"/>
Max Call Number:	<input type="text"/>
MaxCallNumbers_Nonvalidated:	<input type="text"/>

Table 4-4-5 Instruction of Misc Settings

Options	Definition
IAX Thread Count	Establishes the number of IAX helper threads to handle I/O
IAX2 Max Thread Count	Establishes the number of extra dynamic threads that may be spawned to handle I/O.
Max Call Number	The 'Max Call Numbers' option limits the amount of call numbers allowed for each individual remote IP address. Once an IP address reaches its call number limit, no more new connections are allowed until the previous ones close. This option can be used in a peer definition as well, but only takes effect for the IP of a dynamic peer after it completes registration.
MaxCallNumbers_Nonvalidated	The 'MaxCallNumbers_Nonvalidated' is used to set the combined number of call numbers that can be allocated for connections where call token validation has been disabled. Unlike the 'Max Call Number' option, this limit is not separate for each individual IP address. Any connection resulting in a non-call token validated call number being allocated

	<p>contributes to this limit. For use cases, see the call token user guide. This option's default value of 8192 should be sufficient in most cases.</p>
--	---

Quality of Service

Figure 4-4-6 Quality of Service

The screenshot shows a configuration panel for Quality of Service. It has a dropdown menu for 'Quality of Service' and two input fields: 'tos' with a dropdown menu set to 'High Reliability', and 'cos' with an empty text input box.

Table 4-4-6 Instruction of Quality of Service

Options	Definition
TOS	Type of service.
COS	Class of service.

4.5 SIP Account Security

The SIP account security module greatly enhances the security of the system. The wireless gateway supports TLS encrypted calls (sip), which requires sip phone support.

Figure 4-5-1 SIP Account Security

The screenshot shows a configuration panel for SIP Account Security. It is divided into three sections:

- TLS Setting:** Includes 'TLS Enable' (ON), 'TLS Verify Server' (ON), 'Port' (5061), and 'TLS Client Method' (tisv1).
- TLS keys:** A table with columns: Type (client), Key Name, IP Address, Organization, Password, and Operation (Create).
- Key Files:** Includes 'Upload the pem file' and 'Upload the crt file' (both with file selection buttons), a 'File Upload' button, and a table with columns: File Name, File Size, and Operation.

You need to understand its rules when using TLS. The following table shows the setting parameters of TLS.

Table 4-5-1 Parameters of TLS settings

Options	Definition
TLS Enable	Enable or disable DTLS-SRTP support.
TLS Verify Server	If set it off, there is no need to verify the server certificate when the client is operating. If you do not have a server CA certificate, you can disable authentication to connect without the need for a TLS CA file.
Port	TLS port SIP registration, default port 5061.
TLS Client Method	Values include tlsv1, sslv3, sslv2, which specify the protocol for the outbound client connection. The default is sslv2.

Enter the key name, IP address, organization and password in the above settings to create a ca certificate. After mutual authentication between the client and the server, it can be accessed with a certificate.

We need to do a few basic steps:

1. Your asterisk server requires an authentication. We must create or add a certificate on the asterisk server. We need to create a digital key for our server, the server key is the key.pem file, and the certificate request is request.pem.
2. Add some configuration settings to the sip.conf file .
3. Configure the client to use TLS.





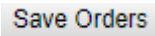
5. Routing

5.1 Call Routing Rules

Figure 5-1-1 Call Routing Rules

	Move	Order	Rule Name	From	To	Rules	Actions
<input type="checkbox"/>		1	incoming	grp-all	iax-2133		
<input type="checkbox"/>		2	outgoing	sip-2111	lte-1.2		
<input type="checkbox"/>		3	test	sip-10028	gsm-2.6	Dial_pattern (+1[0-9])	

New Call Routing Rule Delete Save Orders

You are allowed to set up new call routing rules by , and after setting routing rules, move rules' order by pulling  up and down, click  button to edit the routing and  to delete it. Finally click the  button to save what you set.

Call Routing Rules

You can click  button to set up your routings

Figure 5-1-2 Call Routing Rules

Call Routing Rule

Routing Name: incoming

Call Comes in From: all

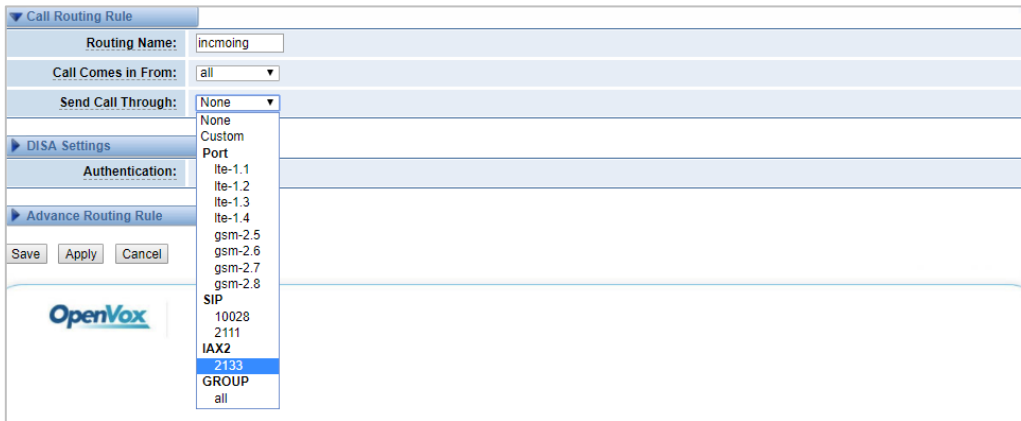
Send Call Through: None
Any SIP
Port
lte-1.1
lte-1.2
lte-1.3
lte-1.4
gsm-2.5
gsm-2.6
gsm-2.7
gsm-2.8
SIP
8000
8001
401
GROUP
all

DISA Settings

Authentication:

Advance Routing Rule

Save Apply Cancel

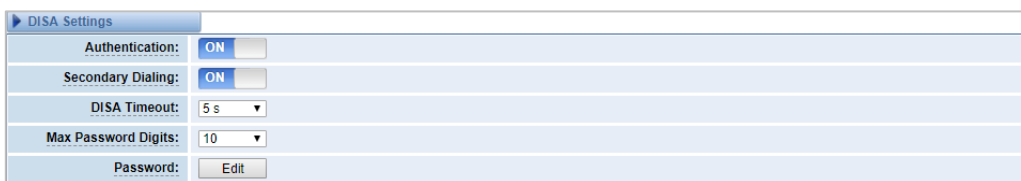


The figure above shows that all the phones in the group ALL are transferred to the IAX2-2133. When the "incoming source" is GSM/LTE port, "prepend", "prefix" and "match pattern" in the "Advanced Routing Rules" are invalid, and only the "CallerID" option is available.

Table 5-1-1 Definition of Routing Options

Options	Definition
Routing Name	The name of this route. Should be used to describe what types of calls this route matches (for example: 'sip1TOport1' or 'port1TOsip1').
Call Comes in From	The launching point of incoming calls.
Send Call Through	The destination to receive the incoming calls.

Figure 5-1-3 DISA Settings



In the DISA Settings, you can set a Password, Authentication or Secondary Dialing.

Figure 5-1-4 Advance Routing Rule Settings

In the Advance Routing Rule Settings, you can set the function of the dial Patterns, Time Patterns, Calling, Forward, Delay and Failover.

Table 5-1-2 Description of Advanced Routing Rule

Options	Definition
Dial Patterns that will use this Route	<p>A Dial Pattern is a unique set of digits that will select this route and send the call to the designated trunks. If a dialed pattern matches this route, no subsequent routes will be tried. If Time Groups are enabled, subsequent routes will be checked for matches outside of the designated time(s).</p> <p>Rules:</p> <p>X matches any digit from 0-9</p> <p>Z matches any digit from 1-9</p> <p>N matches any digit from 2-9</p> <p>[1237-9] matches any digit in the brackets (for example: 1,2,3,7,8,9).</p> <p>Wildcard: matches one or more dialed digits.</p> <p>Prepend: Digits to prepend to a successful match. If the dialed number matches the patterns specified by the subsequent</p>

	<p>columns, then this will be prepended before sending to the trunks.</p> <p>Prefix: Prefix to remove on a successful match.</p> <p>The dialed number is compared to this and the subsequent columns for a match. Upon a match, this prefix is removed from the dialed number before sending it to the trunks.</p> <p>Match pattern: The dialed number will be compared against the prefix + this match pattern. Upon a match, the match pattern portion of the dialed number will be sent to the trunks.</p> <p>CallerID: If CallerID is supplied, the dialed number will only match the prefix + match pattern if the CallerID has been transmitted matches this. When extensions make outbound calls, the CallerID will be their extension number and NOT their Outbound CID.</p> <p>The above special matching sequences can be used for CallerID matching similar to other number matches.</p>
<p>Set the Caller ID Name to</p>	<p>What caller ID name would you like to set before sending this call to the endpoint.</p>
<p>Set the Caller ID Number to</p>	<p>What caller ID number would you like to set before sending this call to the endpoint.</p>
<p>Forward Number</p>	<p>This number will be sent to target PBX or SIP Server as a DID number for inbound rule settings.</p>
<p>Custom Context</p>	<p>Custom the context of dialing rules.</p>
<p>Random Delay</p>	<p>Randomly obtain any value within the threshold interval.</p>

Failover Through Number	Call	The gateway will attempt to send the call out each of these in the order you specify.
----------------------------	------	---

5.2 Groups

Sometimes you want to make a call through one port, but you don't know if it is available, so you have to check which port is free. That would be troublesome. But with our product, you don't need to worry about it. You can combine many Port or SIP to groups. Then if you want to make a call, it will find available port automatically. For the "Ascending" policy, if 2 or more port members are selected, it will use the first available port to call/access the call. In this case, if the gsm-1.1 is available, it will always use the gsm-1.1 call, otherwise, it will use gsm-1.2, and so on.

Figure 5-2-1 Routing Groups

5.3 Batch Creating rules

This page can generate multiple routing rules at the same time.

Figure 5-3-1 Batch Creating rules

Port	Sim Number	Sip Trunk	CallerID
lte-1.1	66001	10028	101
lte-1.2	66002	None	102
lte-1.3	66003	None	200
lte-1.4		None	
gsm-2.5		None	
gsm-2.6		None	
gsm-2.7		None	
gsm-2.8		None	

SIM Number: This number will be sent to target PBX or SIP Server as a DID number for inbound rule settings. If the SIM number is empty, no incoming routing is created.

The SIM number will be sent to the target PBX or SIP server as DID number, and the calling number will be set to the caller ID in the outgoing route.

Figure 5-3-2 Create an incoming route

Call Routing Rule			
Routing Name:	gsm-1.12sip		
Call Comes in From:	lte-1.1		
Send Call Through:	10028		
DISA Settings			
Authentication:	<input type="checkbox"/> OFF		
Advance Routing Rule			
Dial Patterns that will use this Route			
<input type="text" value="(prepend)"/> + <input type="text" value="prefix"/> <input type="text" value="[match pattern]"/> / <input type="text" value="Callerid"/> ✖			
+ Add More Dial Pattern Fields			
Time Patterns that will use this Route			
Time to start:	-	-	✖
Time to finish:	-	-	
Week Day start:	-	-	
Week Day finish:	-	-	
Month Day start:	-	-	
Month start:	-	-	
Month Day finish:	-	-	
Month finish:	-	-	
+ Add More Time Pattern Fields			
Change Rules			
Set the Caller ID Name to	<input type="text"/>		
Set the Caller ID Number to	<input type="text"/>		
Forward Number	66001		
Custom Context	<input type="text"/>		
Random Delay:	<input type="checkbox"/> OFF		
Failover Call Through Number			
+ Add a Failover Call Through Provider			

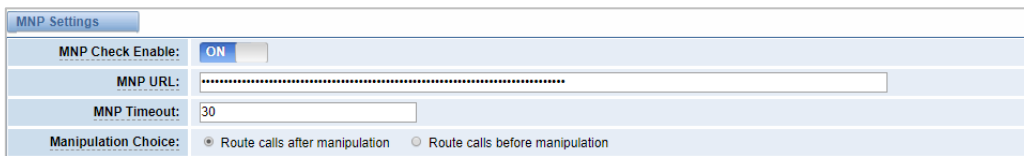
Figure 5-3-3 Create an outgoing route

Call Routing Rule			
Routing Name:	sip2gsm-1.1		
Call Comes in From:	10028		
Send Call Through:	lte-1.1		
DISA Settings			
Authentication:	<input type="checkbox"/> OFF		
Advance Routing Rule			
Dial Patterns that will use this Route			
<input type="text" value="(prepend)"/> + <input type="text" value="prefix"/> <input type="text" value="[match pattern]"/> / <input type="text" value="Callerid"/> ✖			
+ Add More Dial Pattern Fields			
Time Patterns that will use this Route			
Time to start:	-	-	✖
Time to finish:	-	-	
Week Day start:	-	-	
Week Day finish:	-	-	
Month Day start:	-	-	
Month start:	-	-	
Month Day finish:	-	-	
Month finish:	-	-	
+ Add More Time Pattern Fields			

5.4 MNP Settings

Mobile Number Portability allows switching between mobile phone operators without changing the mobile number.

Figure 5-4-1 MNP Settings



MNP Settings	
MNP Check Enable:	<input checked="" type="checkbox"/> ON
MNP URL:
MNP Timeout:	30
Manipulation Choice:	<input checked="" type="radio"/> Route calls after manipulation <input type="radio"/> Route calls before manipulation

The URL is displayed as a password string. So please enter the URL in the format below and check if it is correct. After confirming, copy it to the gateway.

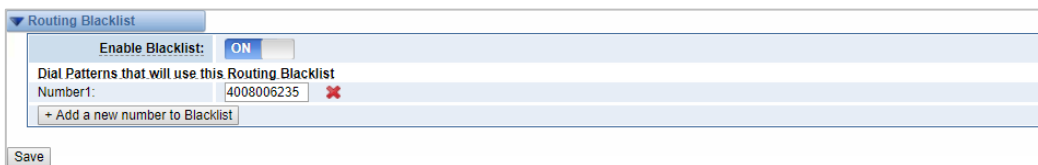
Here is an example of the MNP url: <https://s1.bichara.com.br:8181/chkporta.php?user=832700&pwd=sdsfdg&tn=8388166902>.

The 8388166902 is the outgoing phone number, The outgoing number in the url should be replaced by a variable `${num}`.

When configuring MNP url, it should be replaced with `${num}`, Then paste: `https://s1.bichara.com.br:8181/chkporta.php?user=832700&pwd=sdsfdg&tn=${num}`.

5.5 Routing Blacklist

Figure 5-5-1 Routing Blacklist Settings



Routing Blacklist	
Enable Blacklist:	<input checked="" type="checkbox"/> ON
Dial Patterns that will use this Routing Blacklist	
Number1:	4008006235 ✖
+ Add a new number to Blacklist	
Save	

Blacklists are used to add a phone number to the blacklist or to remove a phone number from the blacklist. After opening the blacklist routing, in the incoming mode, if the calling number is in the blacklist, it will be routed to `hangup()` to end the call.

6. SMS

6.1 General

You can choose enable SMS Received, SMS Local Stored and SMS Status Report or not.

Figure 6-1-1 SMS Settings

The screenshot shows a 'General' tab with a warning message: 'Turn on SMS Received switch before you enable Query Balance, Query Number, SMS Local Stored, SMS to Email or SMS to HTTP!'. Below the warning are three settings, each with a blue 'ON' button and a grey 'OFF' button:

- SMS Received: ON
- SMS Local Stored: ON
- SMS Status Report: ON

Sender Options

You can change sender options here, include resend and the time of resend.

Figure 6-1-2 Sender Options

The screenshot shows a 'Sender Options' tab with three dropdown menus:

- Resend Failed Message: 0
- Repeat Same Message: 1
- Verbose: 3

Table 6-1-1 Description of Sender Options

Options	Definition
Resend Failed Message	The times that you will attempt to resend your failed message.
Repeat Same Message	The times that you will resend the same message.
Verbose	Verbose level of sending message.

SMS to Email

This is a tool that makes it available for you to email account to transmit th

e SMS to other email boxes. The following settings realize that received SMS through openvpnvoip@gmail.com transmit to openvpnvoip@yahoo.com.cn, openvpnvoip@hotmail.com and support@openvox.cn.

Figure 6-1-3 SMS to Email

SMS to Email	
Enable:	<input checked="" type="checkbox"/> ON
SMTP Server:	OTHER
Email Address of Sender:	openvpnvoip@gmail.com
Domain:	smtp.gmail.com
SMTP Port(default 25):	587
SMTP User Name:	openvpnvoip@gmail.com
SMTP Password:	*****
TLS Enable:	<input checked="" type="checkbox"/> This option allows the authentication with certificates.
Destination Email Address 1:	openvpnvoip@yahoo.com.cn
Destination Email Address 2:	openvpnvoip@hotmail.com
Destination Email Address 3:	support@openvox.cn
Title:	support
Content:	we can offer 24 hour's support

Table 6-1-2 Types of E-mail Box

E-mail Box Type	SMTP Server	SMTP Port	SMTP Security Connectivity
Gmail	smtp.gmail.com	587	√
Hotmail	smtp.live.com	587	√
Yahoo!	smtp.mail.yahoo.co.in	587	×
e-mail	smtp.163.com	25	×

Table 6-1-3 Definition of SMS to E-mail

Options	Definition
Enable	When you switch on, the following options are available, otherwise, the following options are unavailable.

Email Address of Sender	To set the email address of an available email account. For example: <u>openvpnvoip@gmail.com</u> .
Domain	To set outgoing mail server. For example:smtp.gmail.com
SMTP Port	To set the port number of outgoing mail server. (Default is 25)
SMTP User Name	The login name of your existing email account. The login name of your existing email account. This option might be different from your email address. Some email client doesn't need the email postfix.
SMTP Password	The password to login your existing email.
TLS Enable	When you choose Yahoo and 163 free e-mails, this option is not available.
SMTP Server	To set outgoing mail server. For example: mail.openvox.cn.
Destination Email Address1	The first email address to receive the inbox message.
Destination Email Address2	The second email address to receive the inbox message.
Destination Email Address3	The third email address to receive the inbox message.

SMS Control

It allows the endpoint to send some specific KEY WORDS and corresponding PASSWORD in response to operate the gateway. SMS is case-sensitive. In default, this function is disabled.

Figure 6-1-4 SMS Control

SMS Control	
Enable:	<input checked="" type="checkbox"/> ON
Password:	<input type="text" value="admin"/>
SMS Formats:	reboot system PASSWORD reboot asterisk PASSWORD restore config PASSWORD get info PASSWORD
SMS Inbox Auto clean:	<input checked="" type="checkbox"/> ON maxsize: 20MB ▾

For example, SMS control password is 123456 which has nothing to do with the login password, you can send “get info 123456” to the module’s phone number to get your gateway’s IP information.

Table 6-1-4 Definition of SMS Control

Options	Definition
Enable	ON (enable), OFF (disable).
Password	The password to confirm that SMS makes the gateway rebooted, shut down, restored configuration files and get info on this gateway.
SMS Format	For example, the message formats: reboot system PASSWORD: To reboot your whole gateway (The PASSWORD is referring to the PASSWORD you set up from option “PASSWORD” above.) reboot asterisk PASSWORD: To restart your gateway core. restore config PASSWORD: To reset the configuration files back to the default factory settings. get info PASSWORD: To get your gateway IP address.
SMS inbox Auto clean	Switch on: When the size of the SMS inbox record file reaches the max size, the system will cut a half of the file. New record will be retained.

	Switch off: SMS record will remain, and the file size will increase gradually. Default on, max size = 20 MB.
--	--

HTTP to SMS

Figure 6-1-5 HTTP to SMS Settings

Table 6-1-5 Definition of HTTP to SMS

Options	Definition
Enable	ON (enable), OFF (disable).
Enable CORS	ON (enable), OFF (disable).
Allow Access Origin Domain	Allow source domain names or IPs that can be accessed across domains.
URL	<p>The URL for send sms.</p> <p>Username: the login username for send sms.</p> <p>Password: the login password for send sms.</p> <p>Phone number: the destination telephone number.</p> <p>Message: the SMS contents.</p> <p>Port: the gsm port for send sms. For example: gsm-1.1, gsm-1.2.</p> <p>Report: the sending result report format.</p> <p>Timeout: how long to wait.</p>

SMS to HTTP

Figure 6-1-6 SMS to HTTP Settings

Table 6-1-6 Definition of SMS to HTTP

Options	Definition	Example
host	HTTP server IP address or domain.	172.16.6.171
port	HTTP server port.	80
param1	File path for receiving and processing the SMS data requests.	smsreceive.php
param2	The name of the parameter used to process the 'port' value.	port
param3	The name of the parameter used to process the 'port name' value.	port name
param4	The name of the parameter used to process the ' message ' value.	message
param5	The name of the parameter used to process the ' time ' value.	time

6.2 SMS Sender

Click the box to select the corresponding port number, fill in the target number and

SMS content .Then you can send it. Different numbers should be separated by symbols: '\r', '\n', space character, semicolon and comma. After sending a text message, if the SMS status report is enabled, a detailed report of the SMS transmission will be displayed below.

Figure 6-2-1 SMS Sender

Port:	<input type="checkbox"/> lte-1.1 <input type="checkbox"/> lte-1.2 <input type="checkbox"/> lte-1.3 <input type="checkbox"/> lte-1.4 <input type="checkbox"/> gsm-2.5 <input type="checkbox"/> gsm-2.6 <input type="checkbox"/> gsm-2.7 <input type="checkbox"/> gsm-2.8 <input type="checkbox"/> All
Flash SMS:	<input type="checkbox"/> OFF
Load numbers from text file:	<input type="button" value="选择文件"/> 未选择任何文件
Destination Number:	<div style="border: 1px solid gray; padding: 5px; min-height: 30px;">10006</div> <small>"; semicolon", " vertical Bar", ", comma ", " blank ", " : colon ", " . dot " were treated as separators in Destination Number List</small>
Message:	<div style="border: 1px solid gray; padding: 5px; min-height: 60px;">ye</div>
Action:	<input type="button" value="Send"/> <input type="button" value="Stop"/>

6.3 SMS Inbox

On this page, you are allowed to scan, delete, clean up, and export each port's received SMS. Also you are allowed to check messages by port, phone number, time order and message keywords.

Figure 6-3-1 SMS Inbox

Port	Phone Number	Time	Message Keywords
all		from	to
Filter Clean Filter			
Total Records: 170			
Port	Phone Number	Time	Message
gsm-7.27	10086	2019/04/02 12:03:33	【好网通移动，流量放心用！，猛戳 dx.10086.cn/bx1h05】尊敬的客户：您当前账户余额25.50元，下一个月结日为2019年04月22日。如需充值可点击 http://gd.10086.cn/oz 。【0元领副卡获赠10G流量： http://dx.10086.cn/v/FvY3a 】【中国移动】
gsm-7.25	10086	2019/04/02 12:03:32	【好网通移动，流量放心用！，猛戳 dx.10086.cn/bx1h05】尊敬的客户：您当前账户余额30.00元，下一个月结日为2019年04月05日。如需充值可点击 http://gd.10086.cn/oz 。【0元领副卡获赠10G流量： http://dx.10086.cn/v/FvY3a 】【中国移动】
gsm-7.28	10086	2019/04/02 12:03:32	尊敬的客户：您已欠费4.40元，已被停机。如需充值可点击 http://gd.10086.cn/oz 。【中国移动】
gsm-7.26	10086	2019/04/02 12:03:32	尊敬的客户：您已欠费1.04元，已被停机。如需充值可点击 http://gd.10086.cn/oz 。【中国移动】
lte-4.15	10086	2019/04/02 12:03:30	【好网通移动，流量放心用！，猛戳 dx.10086.cn/bx1h05】尊敬的客户：您当前账户余额23.51元，下一个月结日为2019年04月28日。如需充值可点击 http://gd.10086.cn/oz 。【0元领副卡获赠10G流量： http://dx.10086.cn/v/FvY3a 】【中国移动】
lte-4.14	10086	2019/04/02 12:03:30	【好网通移动，流量放心用！，猛戳 dx.10086.cn/bx1h05】尊敬的客户：您当前账户余额38.20元，下一个月结日为2019年04月28日。如需充值可点击 http://gd.10086.cn/oz 。【0元领副卡获赠10G流量： http://dx.10086.cn/v/FvY3a 】【中国移动】
lte-4.13	10086	2019/04/02 12:03:30	尊敬的客户：您已欠费3.60元，已被停机。如需充值可点击 http://gd.10086.cn/oz 。【中国移动】
lte-4.15	10086055	2019/04/02 09:33:21	尊敬的客户，感谢您一直以来对我的支持，诚邀您体验我司88元畅享套餐尊享计划优惠。此套餐优惠仅需88元，即可获得一年内每月60GB国内通用流量*2000分钟国内通话分钟数。基本相当于只需支付88元/月，即可语音流量任性用。此优惠仅限受邀客户办理，数量有限办完即止，赶快点击 http://dx.10086.cn/XFF088 抢购吧。优惠名额有限，欲购从速。如有疑问，请拨打10086-399。
lte-4.13	10086	2019/04/01 11:34:31	尊敬的客户：截止01日04:56，您的号码账户余额为0元，尚需缴费3.60元，号码现已暂停使用。支付宝自动充可享充值优惠，打开支付宝APP-充值中心-自动充办理，戳此链接马上办理： http://dx.10086.cn/duanxin1 。【中国移动】
lte-4.13	10086	2019/04/01 11:10:54	尊敬的客户：04月01日因您账户余额不足扣取月结日所需扣费用5.00元，月结扣费失败导致欠费停机，为保障您的通讯服务不受影响，请及时充值开机。支付宝自动充可享充值优惠，打开支付宝APP-充值中心-自动充办理，戳此链接马上办理： http://dx.10086.cn/duanxin1 。【中国移动】

6.4 SMS Outbox

On this page, you are allowed to scan, delete, clean up, and export each port’s received SMS. Also you are allowed to check messages by port, phone number, time order and message keywords.

Figure 6-4-1 SMS Outbox

Port	Phone Number	Time	Message Keywords	
all		from	to	
Filter Clean Filter				
Total Records: 105				
Port	Phone Number	Time	Status	Message
gsm-7.27	10086	2019-04-02 14:41:15	DELIVERD	ye
gsm-7.28	10086	2019-04-02 14:41:00	DELIVERD	ye
gsm-7.25	10086	2019-04-02 14:41:00	DELIVERD	ye
gsm-3.9	10086	2019-04-02 14:41:00	DELIVERD	ye
gsm-7.26	10086	2019-04-02 14:40:59	DELIVERD	ye
lte-4.14	10086	2019-04-02 14:40:52	DELIVERD	ye
lte-4.15	10086	2019-04-02 14:40:52	DELIVERD	ye
lte-4.13	10086	2019-04-02 14:40:47	DELIVERD	ye
lte-4.16	10086	2019-04-02 14:40:47	DELIVERD	ye
gsm-7.25	10086	2019-04-02 14:35:55	DELIVERD	ye

6.5 SMS Forwarding



Using this function, you can forward incoming sms to your mobile. You can click

New Routing

button to add new routing.

For example:

Figure 6-5-1 SMS Forwarding

Routing Name	Type	Policy	From_Members	To_Members	To Number	Actions
forward	module	ascending	lte-1.1,lte-1.2	lte-1.3,lte-1.4	13923704563	 

SMS received by lte-1.1 and lte-1.2, will be transferred to phone number 13923704563 through lte-1.3 or lte-1.4.

Figure 6-5-2 Create a Routing

Routing Groups

Routing Name:

Type: MODULE

Policy: Ascending

From Members

NO. lte-1.1
 lte-1.2
 lte-1.3
 lte-1.4
 gsm-2.5
 gsm-2.6
 gsm-2.7
 gsm-2.8

To Members

NO. lte-1.1
 lte-1.2
 lte-1.3
 lte-1.4
 gsm-2.5
 gsm-2.6
 gsm-2.7
 gsm-2.8

To Number:

For "Ascending" Policy, if you choose 2 or more ports members, it will use first available port to transfer sms. For this case, if lte-1.3 is available, it will always use lte-1.3 to transfer sms. Otherwise, it will use lte-1.4 to transfer sms.

7. Network

7.1 LAN Settings

There are three types of LAN port IP: Factory, Static, and DHCP. The Factory is the default type with an IP of 172.16.98.1. This page is not editable when the LAN IPv4 type is selected as "Factory".

DNS Servers: A list of DNS IP address. Basically this info is from your local network service provider.

A reserved IP address to access in case your gateway IP is not available. Remember to set a similar network segment with the following address of your local PC.

Figure 7-1-1 LAN Settings

LAN IPv4	
Interface:	eth0
Type:	Static ▼
MAC:	A0:98:05:0A:2D:F7
IPv4 Settings	
Address:	172.16.6.108
Netmask:	255.255.0.0
Default Gateway:	172.16.0.1
DNS Servers	
DNS Server 1:	8.8.8.8
DNS Server 2:	
DNS Server 3:	
DNS Server 4:	
Reserved Access IP	
Enable:	<input type="checkbox"/> OFF

Table 7-1-1 Definition of LAN Settings

Options	Definition
Interface	The name of the network interface.
Type	The method to get IP: Factory: Getting IP address by Slot Number (System information to check slot number).

	<p>Static: Manually set up your gateway IP.</p> <p>DHCP: Automatically get IP from your local LAN.</p>
MAC Address	Physical address of your network interface.
IP Address	The IP address of your gateway.
Netmask	The subnet mask of your gateway.
Default Gateway	Default gateway IP address.

7.2 VPN Settings

The wireless gateway provides PPTP and N2N VPN connections, which allow users to establish virtual private networks, encrypt communications, and achieve remote access.

Figure 7-2-1 PPTP VPN Settings

The screenshot shows the 'VPN Settings' configuration page. At the top, 'VPNType:' is set to 'PPTP VPN'. Below this, the 'PPTP VPN Settings' section includes:

- Server: 172.16.8.136
- Account: [empty text box]
- Password: [empty text box]
- Use MPPE:
- Connection Status: Failed to connect (in red text)

 A 'Save' button is located at the bottom left of the form.

Figure 7-2-2 N2N VPN Settings

The screenshot shows the 'VPN Settings' configuration page with 'N2N VPN' selected in the 'VPNType:' dropdown menu. The 'N2N VPN Settings' section includes:

- Enable: ON
- Server Address: [empty text box]
- Port: [empty text box]
- Local IP: [empty text box]
- Subnet Mask: [empty text box]
- User Name: [empty text box]
- Password: [empty text box]
- Connection Status: Failed to connect (in red text)

7.3 DDNS Settings

You can enable or disable DDNS (dynamic domain name server).

Figure 7-3-1 DDNS Settings

DDNS Settings	
DDNS	<input checked="" type="checkbox"/> ON
Type:	inadyn ▼
User Name:	admin
Password:
Your domain:	www.internet.site.com

Table 7-3-1 Definition of DDNS Settings

Options	Definition
DDNS	Enable/Disable DDNS (dynamic domain name server).
Type	Set the type of DDNS server.
Username	Your DDNS account’s login name.
Password	Your DDNS account’s password.
Your domain	The domain to which your web server will belong.

7.4 Toolkit

Some tools for checking network connections are provided, Ping commands and route tracking are supported in the web user interface. You can set the source/destination host address, port, and protocol to capture network packets.

Figure 7-4-1 Toolkit

The screenshot shows a web interface for network tools. At the top, there is a 'GSM IP' dropdown set to '172.16.6.108'. Below it are two input fields for 'www.openvox.cn' with 'Ping' and 'Traceroute' buttons. A 'Channel Recording' section contains fields for 'Interface' (eth0), 'Source host', 'Destination host', 'Port', and 'Protocol' (All). A 'Start' button is below. The 'Report' section displays the output of a ping command: 'ping -I 172.16.6.108 -c 4 www.openvox.cn'. The output shows four successful pings with varying response times. A 'Result' section at the bottom states 'Successfully ping [www.openvox.cn]'.

7.5 Firewall Settings

Figure 7-5-1 Firewall Settings

The screenshot shows the 'Firewall Settings' section of the web interface. It contains two rows of settings, each with a label and a toggle switch. The first row is 'Firewall Enable' with a toggle set to 'ON'. The second row is 'Ping Enable' with a toggle set to 'ON'.

Table 7-5-1 Definition of Firewall Settings

Options	Definition
Firewall Enable	If you want to use White/Black List, and security rules, you must enable this option.
Ping Enable	Whether to enable the Ping function. If the status is OFF:

	disable ping, the gateway does not allow ping.
--	--

Figure 7-5-2 White/Black List Settings

White List Settings

White List Enable: ON

List IP Settings:

Black List Settings

Black List Enable: ON

List IP Settings:

Table 7-5-2 Definition of White/Black List Settings

Options	Definition
White/Black List Enable	To enable White/Black list or not.
List IP	IP is separated only by "," character.

7.6 Security Rules

Figure 7-6-1 Security Rules Settings

Rule Name	Type	Protocol	IP	Port	Actions
SIP	UDP	DROP	172.16.8.0/255.255.0.0	5060:5060	
RTP	TCP	ACCEPT	172.16.8.0/255.255.0.0	10000:20000	

Security Rules

Rule Name:

Protocol:

Port: :

IP / MASK: /

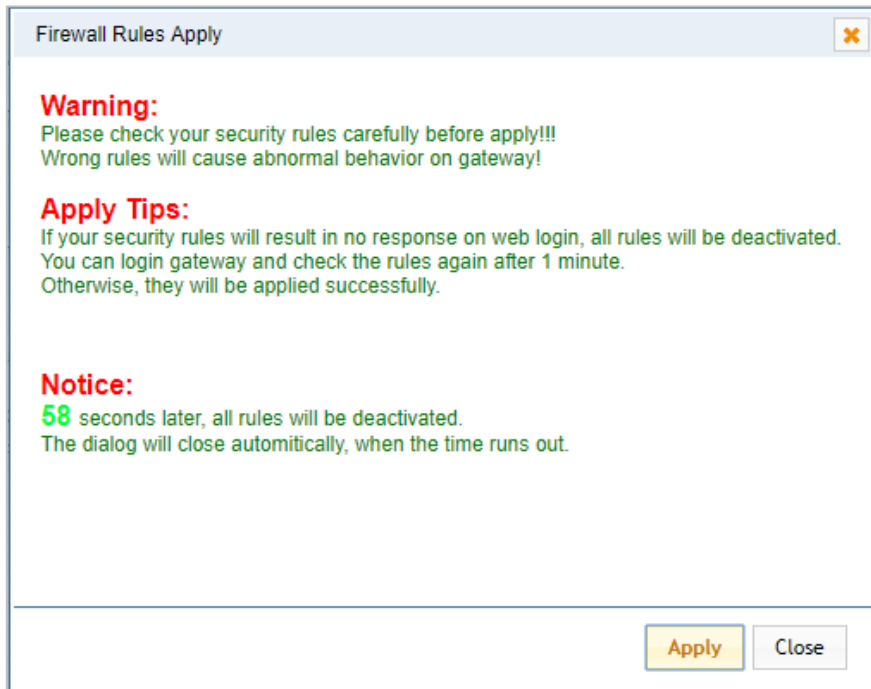
Actions:

Click "submit" button to submit and apply configuration.

If "List IP Settings" has no problem, you will see popup window like below. Please

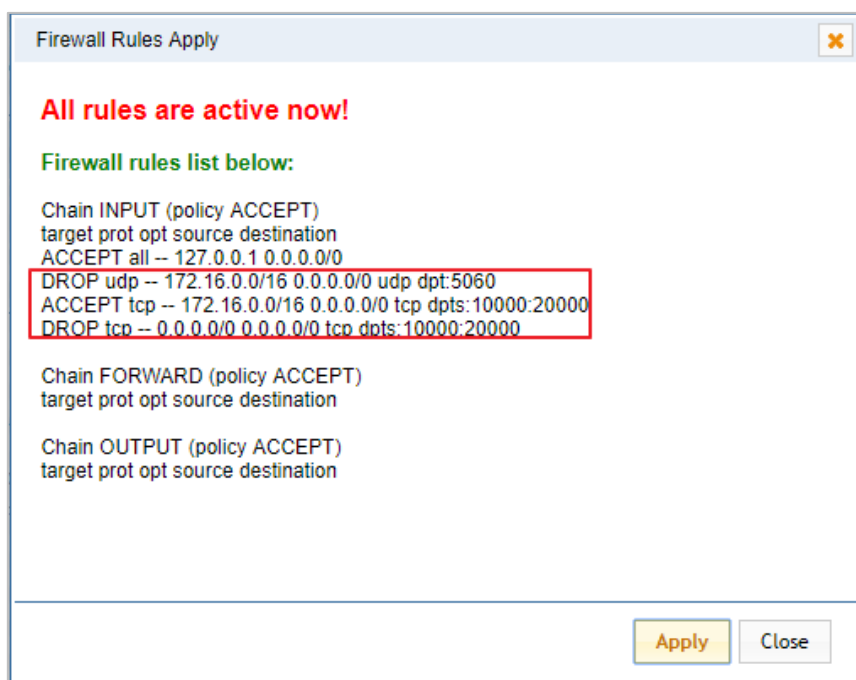
read the warning and tips carefully. And Click "Apply" button in 1 minute. If time runs out, this window will close automatically.

Figure 7-6-2 Security Rules Apply



If you see the windows like below. It means your configuration has been applied successfully.

Figure 7-6-3 Security Rules Apply



7.7 SIP Capture

You can capture the SIP packets on this page to facilitate location problems.

Figure 7-7-1 SIP Capture

Table 7-7-1 SIP Capture Settings

Options	Definition
Interface	You can choose eth0, eth1 or eth0:0.
Method-filter	You can choose INVITE, OPTIONS ,REGISTER or ALL.

8. Advances

8.1 Asterisk API

When you make “Enable” switch to “ON”, this page is available.

Figure 8-1-1 Asterisk API

General		
Enable:	<input checked="" type="checkbox"/> ON	
Port:	5038	
Manager		
Manager Name:	admin	
Manager secret:	NS1hPs2d	
Deny:		
Permit:		
Rights		
System:	read: <input checked="" type="checkbox"/>	write: <input checked="" type="checkbox"/>
Call:	read: <input checked="" type="checkbox"/>	write: <input checked="" type="checkbox"/>
Log:	read: <input checked="" type="checkbox"/>	write: <input checked="" type="checkbox"/>
Verbose:	read: <input checked="" type="checkbox"/>	write: <input checked="" type="checkbox"/>
Command:	read: <input type="checkbox"/>	write: <input checked="" type="checkbox"/>
Agent:	read: <input checked="" type="checkbox"/>	write: <input checked="" type="checkbox"/>
User:	read: <input checked="" type="checkbox"/>	write: <input checked="" type="checkbox"/>
Config:	read: <input checked="" type="checkbox"/>	write: <input checked="" type="checkbox"/>
DTMF:	read: <input checked="" type="checkbox"/>	write: <input type="checkbox"/>
Reporting:	read: <input checked="" type="checkbox"/>	write: <input checked="" type="checkbox"/>
CDR:	read: <input checked="" type="checkbox"/>	write: <input type="checkbox"/>
Dialplan:	read: <input checked="" type="checkbox"/>	write: <input type="checkbox"/>
Originate:	read: <input type="checkbox"/>	write: <input checked="" type="checkbox"/>
All:	read: <input checked="" type="checkbox"/>	write: <input checked="" type="checkbox"/>

Table 8-1-1 Definition of Asterisk API

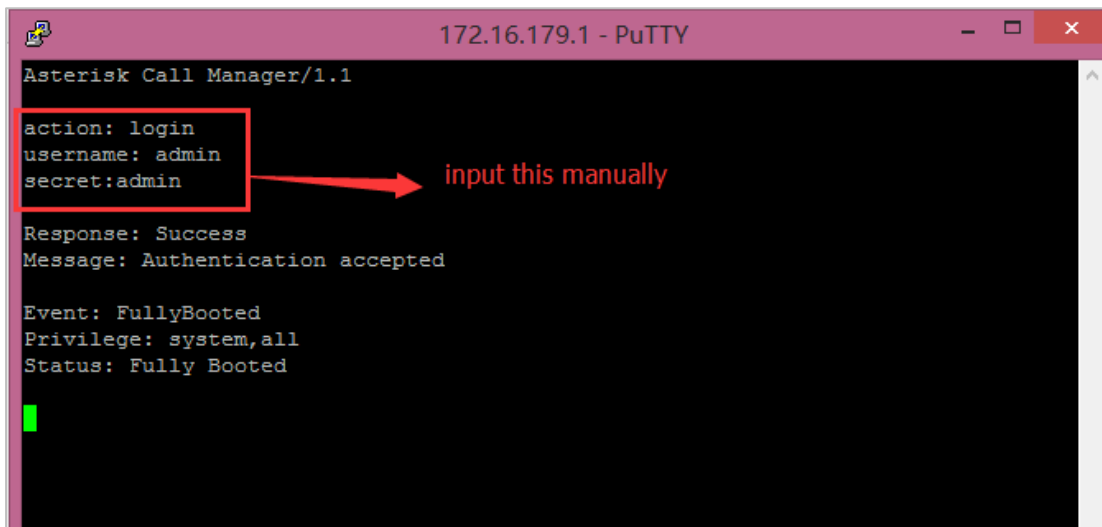
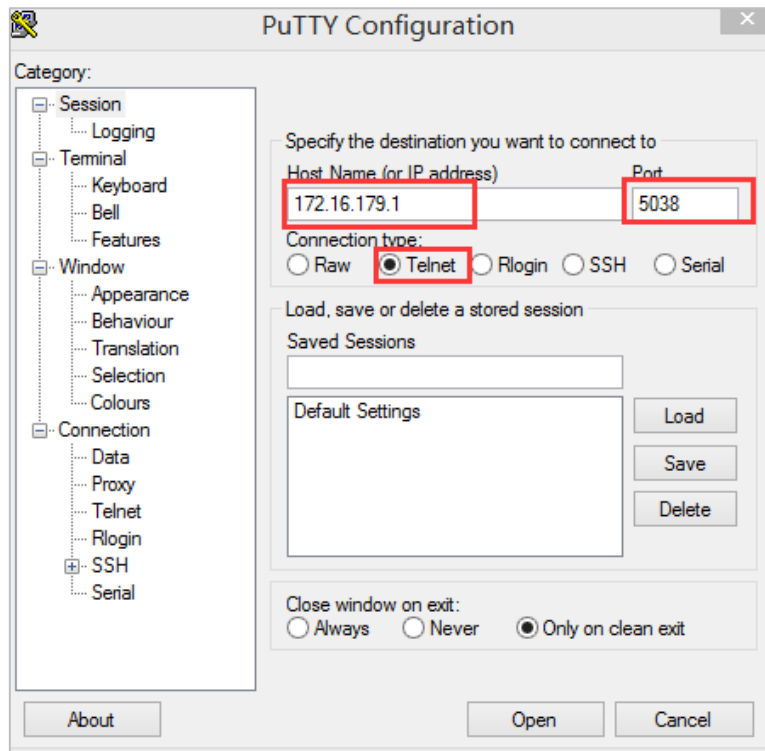
Options	Definition
Port	Network port number.
Manager Name	Name of the manager without space
Manager secret	Password for the manager. Allowed characters: “- _ + . < > & 0-9a-zA-Z”. Length: 4-32 characters.

Deny	<p>If you want to deny some hosts or networks, you can use char & as separator. For example:</p> <p>192.168.1.0/255.255.255.0&10.0.0.0/255.0.0.0</p>
Permit	<p>If you want to permit some hosts or networks, you can use char & as separator. For example:</p> <p>192.168.1.0/255.255.255.0&10.0.0.0/255.0.0.0</p>
System	<p>General information about the system and ability to run system management commands, such as Shutdown, Restart, and Reload.</p>
Call	<p>The information about channels and the setting information of channel in use.</p>
Log	<p>Logging information.</p>
Verbose	<p>Verbose information.</p>
Command	<p>Permission to run CLI commands. Write-only.</p>
Agent	<p>The information about queues and agents and the ability to add queue members to a queue.</p>
User	<p>Permission to send and receive User Event.</p>
Config	<p>Ability to read and write configuration files.</p>
DTMF	<p>Receive DTMF events. Read-only.</p>
Reporting	<p>The ability to get information about the system.</p>
CDR	<p>Call records. Read-only.</p>
Dialplan	<p>Receive NewExten and Varset events. Read-only.</p>

Originate	Permission to initiate new calls. Write-only.
All	Select all or deselect all.

Please refer to the following figure to access the gateway API by telnet. 172.16.179.1 is the gateway's IP, and 5038 is its API port.

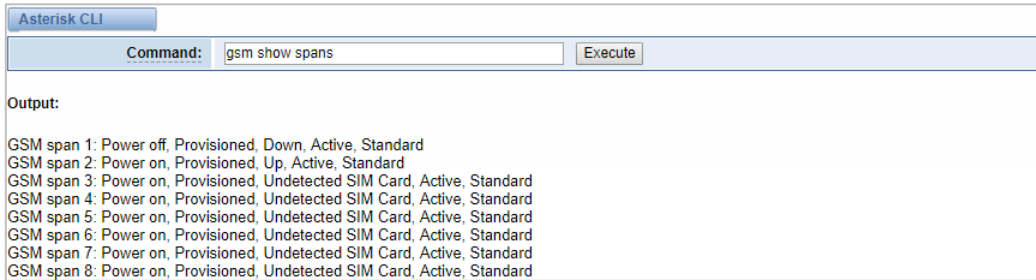
Figure 8-1-2 Telnet Access Gateway API



8.2 Asterisk CLI

On this page, you are allowed to run Asterisk commands.

Figure 8-2-1 Asterisk CLI



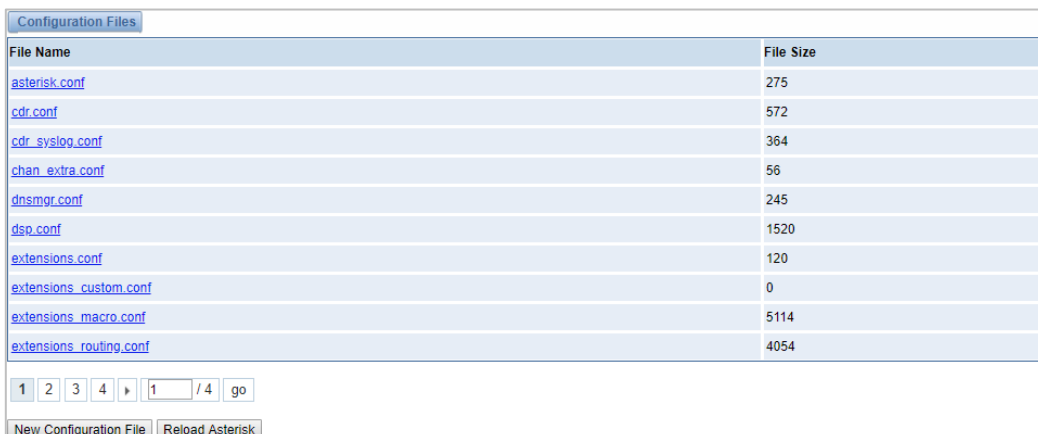
Command: Type your Asterisk CLI commands here to check or debug your gateway.

Notice: If you type “help” or “?” and execute it, the page will show you the shell command.

8.3 Asterisk File Editor

On this page, you are allowed to edit and create configuration files. Click the file to edit.

Figure 8-3-1 Asterisk File Editor



Click [New Configuration File](#) to create a new configuration file. After editing or

creating, you should reload Asterisk.

8.4 Internet

In order to meet the requirements of some operators Package consumption flow, we provide Internet access which can be set in batches (This function is only available on the LTE module currently. Other hardware does not support the Internet function at this time).

Figure 8-4-1 Asterisk Internet

ID	Open	APN User Name	APN Password	APN	URL	MAX(MB)	USED	Time	Save
	No								
lte-1.1	No								
lte-1.2	No								
lte-1.3	No								
lte-1.4	No								

8.5 Cloud Management

SWG and VoxStack series gateways both support OpenVox Cloud Management.

Figure 8-5-1 OpenVox Cloud Management

If your device is connected to the cloud management, the SSH and the web pages of the gateway can be accessed through the cloud management, and it can be monitored whether the device is connected to the cloud management platform. On the cloud management platform, you can also count your device model, quantity, distribution area, and so on which can provide you with efficient and excellent service and experience.

Table 8-5-1 Definition of Cloud Management

Options	Definition
Enable Cloud ervice	Turn on/off the cloud management.
Choose Service	Currently supports two servers, one is China and the other is the United States.
Account	Registered account or email on the cloud management platform.
Password	The password of the account registered on the cloud management platform.
Connection Status	Whether currently connected to the cloud management platform or not.

8.6 Balance

The wireless gateway provides a balance inquiry function for users who use the SIM card in batches, it can automatically send a short message to the operator to query the current balance on the SIM card, which is convenient, fast and accurate.

Figure 8-6-1 Balance









Port	Query Type	Destination Number	Receive Number	Send Message	Matching Key	Balance	Actions
lte-1.1	SMS	10086	10086	ye	余额		
lte-1.2	SMS	10086	10086	ye	余额		
lte-1.3	SMS	10086	10086	ye	余额		
lte-1.4	SMS	10086	10086	ye	余额		
gsm-2.5	SMS	10086	10086	ye	余额		
gsm-2.6	SMS	10086	10086	ye	余额		
gsm-2.7	SMS	10086	10086	ye	余额		
gsm-2.8	SMS	10086	10086	ye	余额		

Figure 8-6-2 Balance Settings

Port Ite-1.1

Query Swith:	<input type="checkbox"/> ON
Query Type:	SMS
Destination Number:	10086
Receive Number:	10086
Send Message:	ye
Matching Key:	余额
Registered Query:	<input type="checkbox"/> OFF
Interval:	0 Minute
Call Count Query:	0
Decimal mark:	.
Kilo mark:	.

The matching test

Message content:	【好网选移动，流量放心用！，猛戳 dx.10086.cn/bz1h05 】尊敬的客户：您当前账户余额30.00元，下一个月结日为2019年04月05日。如需充值请点击：//gd.10086.cn/CZ。【0元领副卡获赠10G流量：http://dx.10086.cn/yJFvT3q 】【中国移动】
Matching results:	Test 30.00

Save To Other Ports

8.7 Phone Number

The wireless gateway also provides a number inquiry function, which can automatically send a short message to the operator to query the number of the SIM card that you use currently.

Figure 8-7-1 Phone Number

Port	Query Type	Destination Number	Receive Number	Send Message	Matching Key	PhoneNumber	Actions
Ite-1.1	SMS	10086	10086	BJ	号码		
Ite-1.2	SMS	10086	10086	BJ	号码		
Ite-1.3	SMS	10086	10086	BJ	号码		
Ite-1.4	SMS	10086	10086	BJ	号码		
gsm-2.5	SMS	10086	10086	BJ	号码		
gsm-2.6	SMS	10086	10086	BJ	号码		
gsm-2.7	SMS	10086	10086	BJ	号码		
gsm-2.8	SMS	10086	10086	BJ	号码		

Figure 8-7-2 Phone Number

Query Swith:	<input type="checkbox"/> ON
Query Type:	SMS
Destination Number:	10086
Receive Number:	10086
Send Message:	BJ
Matching Key:	号码

The matching test

Message content:	
Matching results:	Test

Save To Other Ports

9. Logs

On the “Log Settings” page, you should set the related logs on to scan the responding logs page. For example, set “System Logs” on like the following, then you can turn to “System” page for system logs. Otherwise, system logs are unavailable. Other log pages are the same.

Figure 9-1 Logs Settings

Table 9-1 Definition of Logs

Options	Definition
System Logs	Whether to enable the system logs or not
Auto clean (System Logs)	<p>Switch on: when the size of log file reaches the max size, the system will cut a half of the file and the new logs will be retained;</p> <p>Switch off: logs will remain, and the file size will increase gradually.</p> <p>The default is on, the default maxsize is 1MB.</p>
SIP Logs	Whether to open the sip logs or not

Auto clean (SIP logs)	<p>Switch on: when the size of log file reaches the max size, the system will cut a half of the file and the new logs will be retained;</p> <p>Switch off: logs will remain, and the file size will increase gradually.</p> <p>The default is on, the default maxsize is 100KB.</p>
IAX Logs	Whether to open the IAX logs or not
Auto clean (IAX logs)	<p>Switch on: when the size of log file reaches the max size, the system will cut a half of the file and the new logs will be retained;</p> <p>Switch off: logs will remain, and the file size will increase gradually.</p> <p>The default is on, the default maxsize is 100KB.</p>
Call Detail Record	Show Call Detail Records for each channel.
Append IMEI	<p>Switch on: IMEI will be appended to the CDR gsm channel in 'From' or 'To'.</p> <p>Switch off: No appended IMEI.</p> <p>The default is off.</p>
Auto clean (Call Detail Record)	<p>Switch on: when the size of log file reaches the max size, the system will cut a half of the file and the new logs will be retained;</p> <p>Switch off: logs will remain, and the file size will increase gradually.</p> <p>The default is on, the default maxsize is 10MB.</p>

With the Syslog software, the gateway's logs and CDRs can be monitored and stored locally on the PC.

Figure 9-2 Logs Settings

Figure 9-3 System Logs

You can easily browse your CDRs on the Web GUI, you also can delete, clean or export your CDR information.

Figure 9-4 CDR

Caller ID	Callee ID	From	To	Start Time	Duration	Result	
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	from <input type="text"/> to <input type="text"/>	from <input type="text"/> to <input type="text"/>	All <input type="text"/>	
Filter Clean Filter							
Total Records: 33385							
<input type="checkbox"/>	Caller ID	Callee ID	From	To	Start Time	Duration	Result
<input type="checkbox"/>	66376	1028@172.16.33.102	gsm-1.1	1028	1970-01-01 08:04:06	00:01:55	ANSWERED
<input type="checkbox"/>	66389	1028@172.16.33.102	gsm-2.2	1028	1970-01-01 08:01:50	00:01:00	ANSWERED
<input type="checkbox"/>	66390	1028@172.16.33.102	gsm-1.3	1028	1970-01-01 08:02:21	00:00:25	ANSWERED

In the new version we have enriched the LOGS display which you can see each port of the call outbound clearly.

Figure 9-5 GSM Outbound

GSM Outbound									
Port	All Calls	All Durations	Answered	Canceled	Busy	No Answer	No Dialtone	No Carrier	Other
lte-1.1	0	0	0	0	0	0	0	0	0
lte-1.2	0	0	0	0	0	0	0	0	0
lte-1.3	0	0	0	0	0	0	0	0	0
lte-1.4	0	0	0	0	0	0	0	0	0
gsm-2.5	0	0	0	0	0	0	0	0	0
gsm-2.6	0	0	0	0	0	0	0	0	0
gsm-2.7	0	0	0	0	0	0	0	0	0
gsm-2.8	0	0	0	0	0	0	0	0	0
Total	0	0	0	0	0	0	0	0	0