GSM/3G Gateway User Manual

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Full text

The overall layout adjustment

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1. Overview

What is GSM/3G Gateway?

OpenVox GSM/3G Gateway is an open source asterisk-based VoIP Gateway solution for SMBs and SOHOs. With friendly GUI and unique modular design, users may easily setup their customized Gateway. Also secondary development can be completed through AMI (Asterisk Management Interface).

OpenVox GSM/3G Gateways have 4 models: WGW1002G, VS-GW1202, VS-GW1600 and VS-GW2120. VS-GW1002 suports 2 GSM Channels. VS-GW1202 supports 4/8 GSM/3G channels. VS-GW1600 support up to 20 GSM/3G channels. VS-GW2120 supports up to 44 GSM/3G channels. Both GSM and 3G/UMTS gateways are developed for interconnecting the GSM cellular networks with a wide selection of codecs and signaling protocol, including G.711A, G.711U, G.729, G.722, G.723, G.726 and GSM to quickly reduce communication expenses and maximize cost-savings. With the unique design of the VoxStack gateway, it can support hot-swap for both SIM cards and GSM/3G gateway modules. Users can simply add or remove the modules for hardware expansion or exchange.

The VoxStack gateway designs with 2 LAN switch boards to provide stack ability on the hardware upgrade, and each GSM/3G module is independent , so each one has a GUI configuration web. If you connect to ETH1, you can access Board 1 only and access other boards with different port numbers which can avoid IP conflict. Otherwise if you connect to ETH2, you can access different Boards with different IP addresses.

Our products support SMS messages sending, receiving, group sending and SMS to E-mail. The GSM gateway will be 100% compatible with Asterisk, Elastix, trixbox, 3CX, FreeSWITCH SIP server and VOS VoIP operating platform.

Sample Application

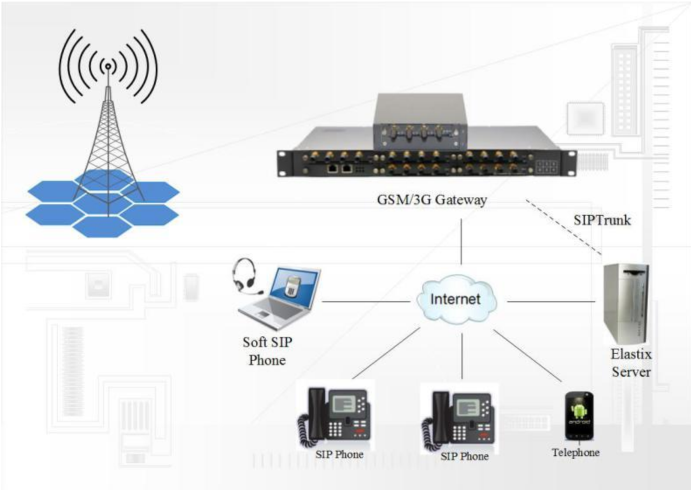


Figure 1-1 TopologicalGraph

Product Appearance

Figure 1-2 Product Appearance of GSM

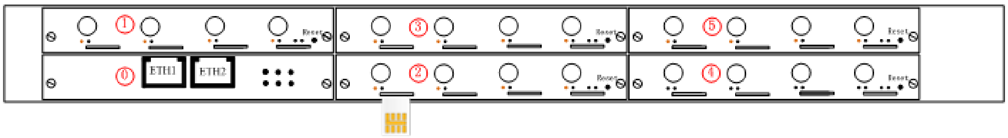


Figure 1-3 Front Panel of GSM

Network Data Switch Board: ETH1, ETH2.

- ETH1: Access Board 1 only, provide proxy access to other boards with different port numbers which can avoid IP conflict.
- ETH2: Access different Boards with different IP address.

Figure 1-4 Product Appearance of 3G

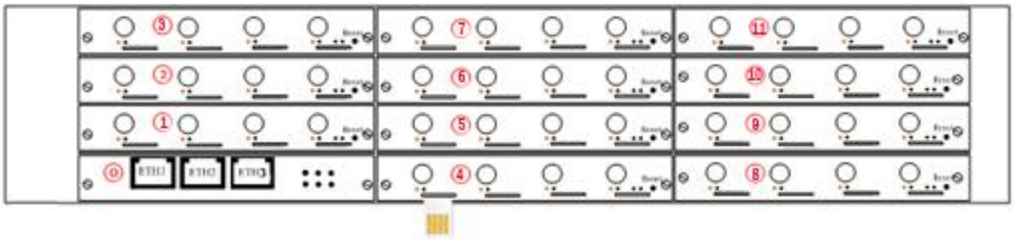


Figure 1-5 Front Panel of 3G

Network Data Switch Board: ETH1, ETH2, ETH3.

- ETH1: Access Board 1 only, provide proxy access to other boards with different port numbers which can avoid IP conflict.
- ETH2: Access different Boards with different IP address.
- ETH3: Access different Boards with different IP address.

VoxStack provides 2 working modes: Stand-alone and Cluster.

- Stand-alone: A single IP address manages one GSM modules (4 ports).

Table 1-1 ETH2 IP Addresses

Slot Num	IP	Username	Password
1	172.16.99.1	admin	admin
2	172.16.99.2	admin	admin
3	172.16.99.3	admin	admin
4	172.16.99.4	admin	admin
5	172.16.99.5	admin	admin
6	172.16.99.6	admin	admin
7	172.16.99.7	admin	admin
8	172.16.99.8	admin	admin
9	172.16.99.9	admin	admin
10	172.16.99.10	admin	admin
11	172.16.99.11	admin	admin

- Cluster: A single IP address manages up to 5 GSM modules (up to 20 ports).

Default IP: 172.16.99.1

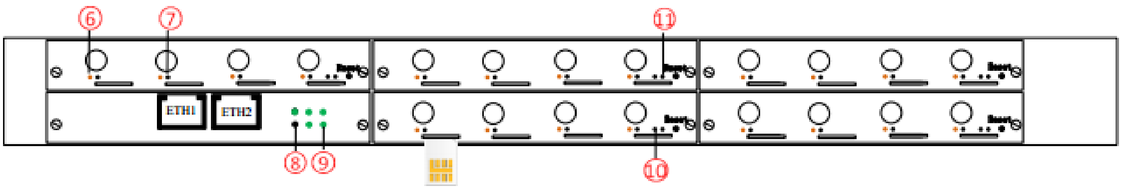


Figure 1-6 Front Panel

LED Indicator	Color	Status
⑥ Signal Status LED	Green and Flash	Module Initiating
	Red and Flash	No SIM Card
	Red and No-flash	Worst Signal Quality
	Yellow and No-flash	Medium Signal Quality
	Green and No-flash	Best Signal Quality
⑦ Call Status LED	Flash (0.25s)	Communicating
	Blind	Normal
⑧ ⑨ Network Status LED	Green and Flash	Network Connected
⑩ Running Status LED	Green and Flash(0.5s)	Work Normally
⑪ Power LED	Always Green	Supply Power

Figure 1-5 LED Indicator

Main Features

- Modular and VoxStack design
- Based on Asterisk®
- Editable Asterisk® configurationfile
- Wide selection of codecs and signaling protocol
- Support SMS sending, receiving, group sending
- Support transferring SMS to E-mail
- Support SMS automatically resend
- Support SMS remotely controlling gateway
- Support USSD service
- Support IMEI modification
- Support PIN identification
- Support unlimited routing rules and flexible routing settings

- SIM cards and modules are all hot-swap
- Stable performance, flexible dialing, friendly GUI
- WCDMA/UMTS: 850/900/1900/2100 MHz
- GSM: 850/900/1800/1900 MHz

Physical Information

- Weight: 4301g(VS-GW1600-20G) 6144g(VS-GW2120-32w)
- Size: 44cm*30cm*4.5cm (VS-GW1600-20G) 44cm*34cm*9cm (VS-GW2120-32w)
- Temperature: -20~70°C (Storage) 0~40°C (Operation)
- Operation humidity: 10%~90% non-condensing
- Max power: 46W(VS-GW1600-20G) 95W(VS-GW2120-32w)
- LAN port: 2(VS-GW1600-20G) 3(VS-GW2120-32w)

Software

Default IP: 172.16.99.1

Username: admin

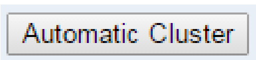
Password: admin

For first time, you can access WGW1002G using default IP 172.16.99.1. Then configure the module as you want.

For VS-GW2102, VS-GW1600, VS-GW2120 series of GSM/3G gateway, every VS-GWM400G/W is independent with each other. There are two RJ45 Network ports: ETH1 and ETH2. They are different.

- If you want each module to work stand-alone and access each of them, choose ETH2 please. Default IP of each module is 172.16.99.X (X is slot number).
- If you want to use one IP to master all the boards, choose ETH1, access board1 using default IP 172.16.99.1 and do the cluster. Then you can access to other boards with different port numbers but the same IP address, this will help to avoid IP conflict. Board1 work as master, and other boards work as slave.

How to do Cluster?

penVox GSM Gateway offers you two ways to cluster your gateway: Automatic Cluster or Manual Cluster. When you first time log in your gateway, you will only see 4 ports of one module. Then you can press 

button, the system will search other modules in the LAN and communicate.

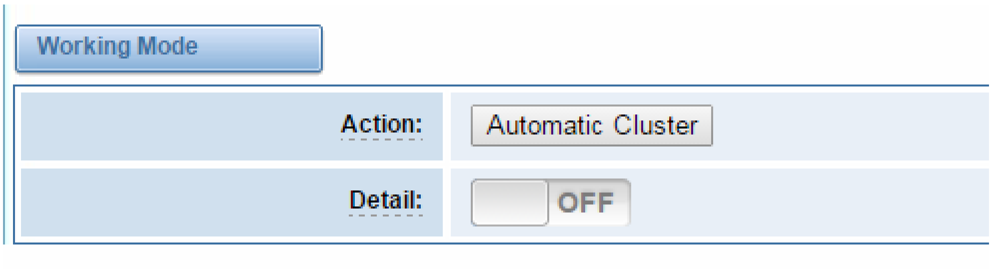
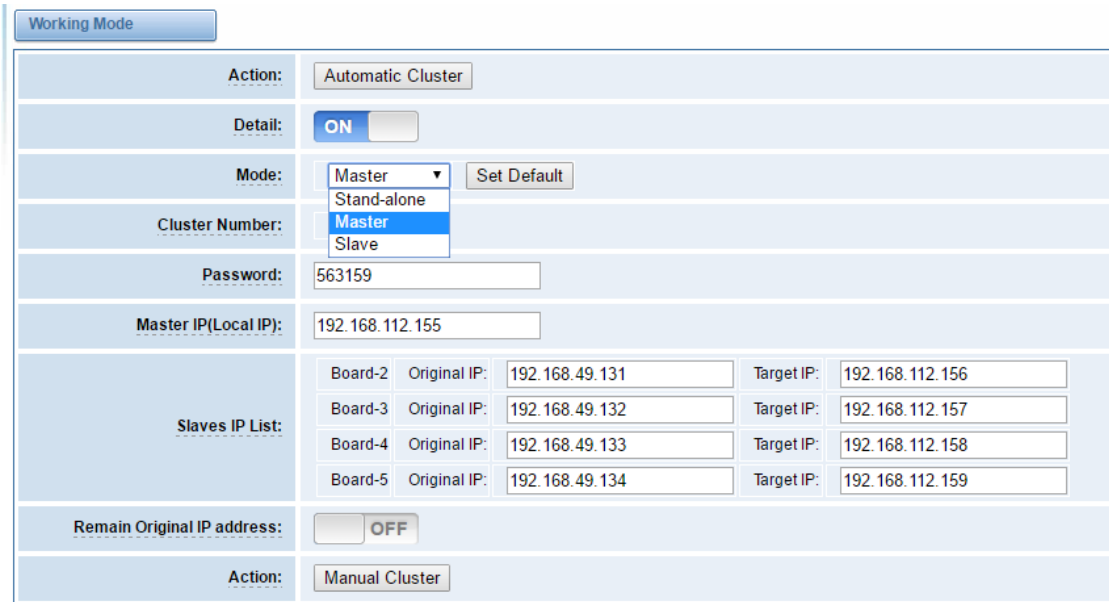


Figure 1-7 Automatic Cluster

If you want to choose **Manual Cluster**, you should switch **Detail** on first.

There are 3 kinds of cluster mode: stand-alone, Master and Slave.

- **Stand-alone Mode:** Run alone, total 4 ports.
- **Master Mode:** Run as master with two different IP, controlling up to 10 slaves. (The master can be accessed by the original IP. The target IP is used to communicate with the slaves.)
- **Slave Mode:** Run as slave with two different IP, controlled by the master. If the original IP is forbidden, the slave can be accessed by the master with inward IP only.



Notice: You can choose Remain Original IP address ON or OFF. If set it on, you can log in your getaway with Original IP and Target IP.

Notice: Log in

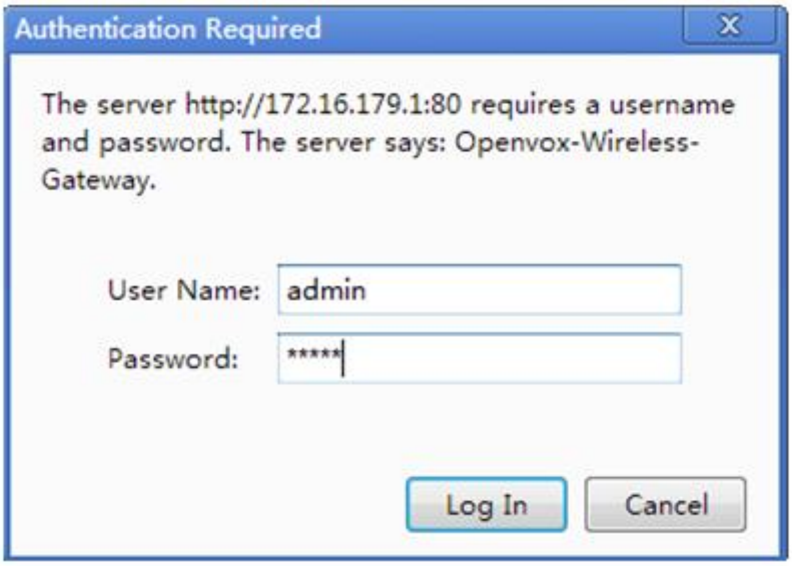


Figure 1-8 LOG Interface

2. System

Status

On the "Status" page, you will find all GSM, SIP, IAX2, Routing, Network information and status.

Module Information

Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	Module Status	Remain Time
gsm-1.1		0		Undetected SIM Card	0	0	0		No Limit
gsm-1.2		0		Undetected SIM Card	0	0	0		No Limit
gsm-1.3		0	CHINA MOBILE	Registered (Home network)	6	292	47	READY	No Limit
gsm-1.4		0	CHINA MOBILE	Registered (Home network)	6	300	40	READY	No Limit
umts-2.1		-1		Undetected SIM Card	0	0	0		No Limit
umts-2.2		-1		Undetected SIM Card	0	0	0		No Limit
umts-2.3		-1	CHINA MOBILE	Registered (Home network)	0	296	40	READY	No Limit
umts-2.4		-1	CHINA MOBILE	Registered (Home network)	0	302	42	READY	No Limit
gsm-3.1		0	CHINA MOBILE	Registered (Home network)	6	298	64	READY	No Limit
gsm-3.2		0	CHINA MOBILE	Registered (Home network)	7	296	72	READY	No Limit
gsm-3.3(66370)		0	CHINA MOBILE	Registered (Home network)	6	297	47	READY	No Limit
gsm-3.4		0	CHINA MOBILE	Registered (Home network)	6	302	62	READY	No Limit
umts-4.1		-1	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
umts-4.2		-1	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
umts-4.3		-1	CHINA MOBILE	Registered (Home network)	0	292	68	READY	No Limit
umts-4.4		-1	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
umts-5.1		-1	CHINA MOBILE	Registered (Home network)	0	0	0	READY	No Limit
umts-5.2		-1		Undetected SIM Card	0	0	0		No Limit
umts-5.3		-1		Not registered	0	0	0	INIT	No Limit
umts-5.4		-1		Not registered	0	0	0	INIT	No Limit

SIP Information

Endpoint Name	User Name	Host	Registration	SIP Status
10027	10027	172.16.2.209	none	Unmonitored
10028	10028	172.16.33.101	none	Unmonitored
test1001	1001	(Unspecified)	server	UNKNOWN
1002	1002	(Unspecified)	server	UNKNOWN

IAX2 Information

Endpoint Name	User Name	Host	Registration	IAX2 Status
1111	1111	172.16.2.209	none	OK (2 ms)
2112	2112	(null)	server	UNKNOWN
2113	2113	172.16.33.111	client	UNREACHABLE
test3001	3001	0.0.0.0	server	UNKNOWN
2111	2111	172.16.33.111	none	OK (2 ms)

Routing Information

Rule Name	From	To	Rules
incoming	grp-all	sip-10028	
outgoing	sip-10027	grp-parts	
test1	sip-1001	iax-2111	Dial_pattern ()+1[/]
test2	sip-1001	iax-2112	Dial_pattern ()+2[/]
test3	sip-1001	iax-2113	Dial_pattern ()+3[/]
test	grp-iax	gsm-1.3 umts-2.3	

Network Information

Name	MAC Address	IP Address	Mask	Gateway	RX Packets	TX Packets
LAN	A0:98:05:01:0B:63	172.16.179.1	255.255.0.0	172.16.0.1	12582903	6435114

Figure 2-1 System Status

Table 2-1 Description of System Status

Options	**Definition**
Port	Number of GSM ports. GSM ports begin with "gsm-", such as gsm-1.1; 3G ports begin with "umts-", such as umts-2.1.
Signal	Display the signal strength of in each channels of GSM.
BER	Bit Error Rate.
Carrier	Display the network carrier of current SIM card.
Registration Status	Indicates the registration status of current GSM module.
PDD	Post Dial Delay (PDD) is experienced by the originating customer as the time from the sending of the final dialed digit to the point at which they hear ring tone or other in-band information.Where the originating network is required to play an announcement before completing the call then this definition of PDD excludes the duration of such announcements.
ACD	The Average Call Duration (ACD) is calculated by taking the sum of billable seconds (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 calls attempted. Since busy signals and other rejections by the called number count as call failures, the ASR value can vary depending on user behavior. GSM Status Show the status of port, include blank space and “READY”. Black space means it is unavailable here and “Ready” means the port is available
Remain Time	This value is multiplied by to step length is a rest call time.

Time

Table 2-2 Description 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].
Auto-Sync from NTP	Whether enable automatically synchronize from NTP server or not. ON is enable, OFF is disable this function.
Sync from NTP	Sync time from NTP server.
Sync from Client	Sync time from local machine.

For example, you can configure like this:

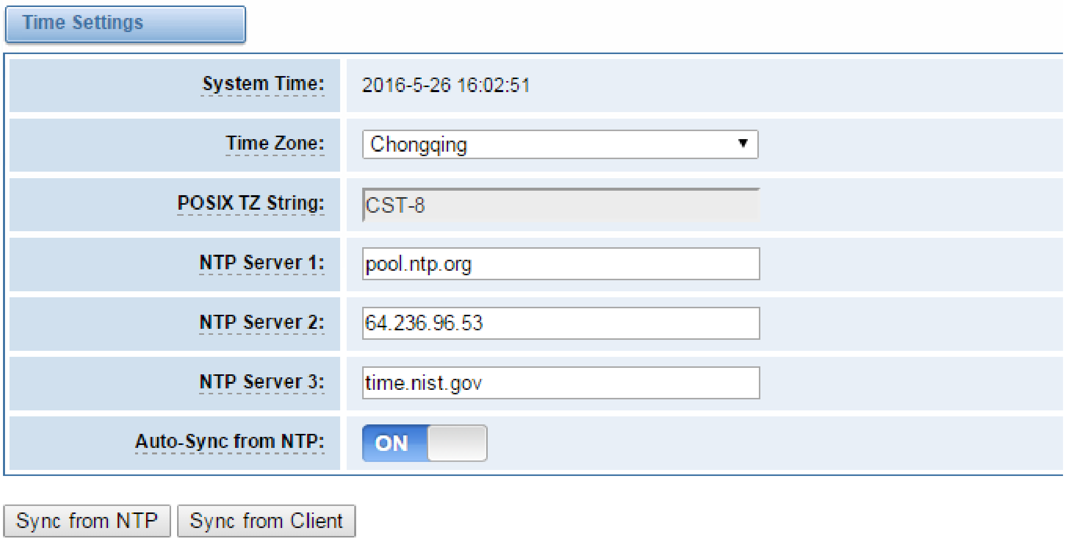


Figure 2-2 Time Settings

You can set your gateway time Sync from NTP or Sync from Client by pressing different buttons.

Login Settings

Your gateway doesn't have administration role. All you can do here is to reset what new username and password to manage your gateway. And it has all privileges to operate your 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 user name and password will be OK. Also you can specify the web server port number.

Table 2-3 Description of Login Settings

Options	**Definition**
User Name	Define your username and password to manage your gateway, without space here. Allowed characters "-_+. < >&0-9a-zA-Z". Length: 1-32 characters.
Password	Allowed characters "-_+. < >&0-9a-zA-Z". Length: 4-32 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.

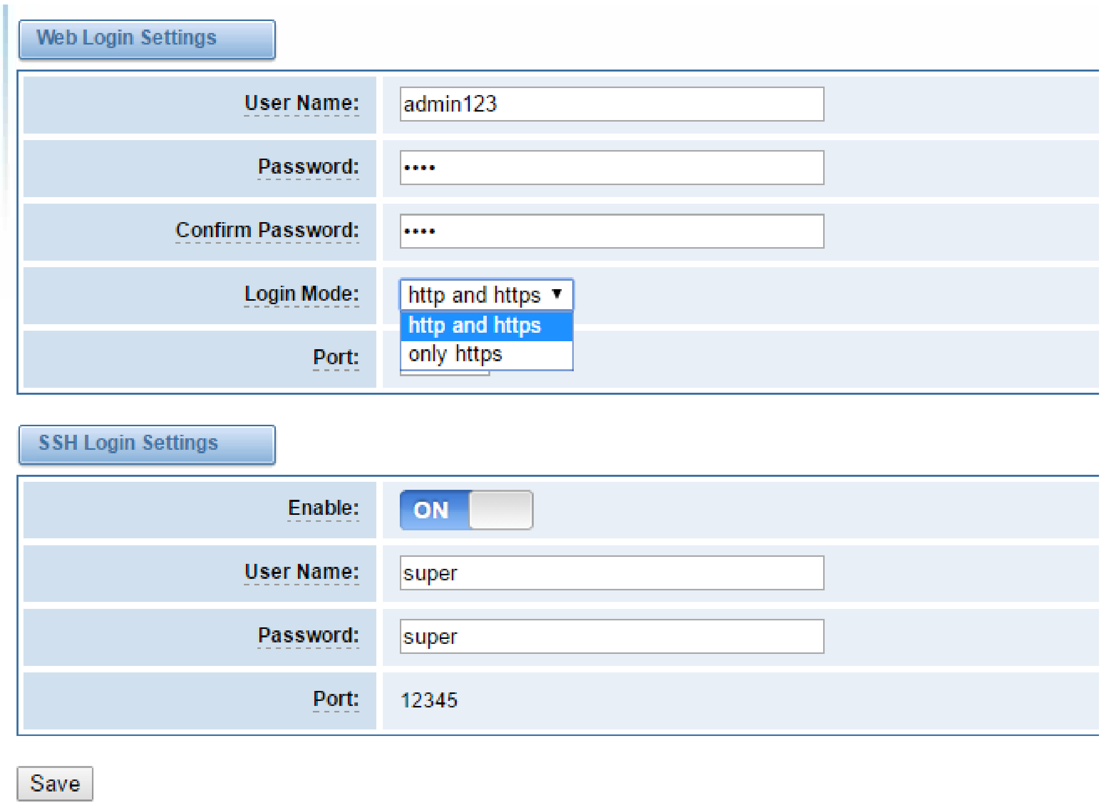


Figure 2-3 Login Settings

Notice: Whenever you do some changes, do not forget to save your configuration.

General

Language Settings

You can choose different languages for your system. If you want to change 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”.

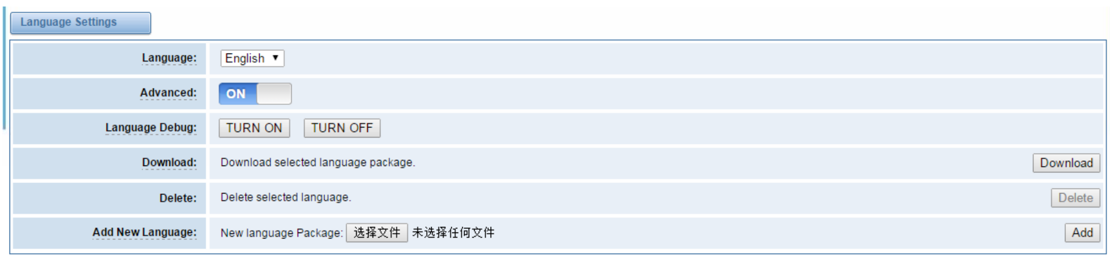


Figure 2-4 Language Settings

Scheduled Reboot

If 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 and By Running Time”.

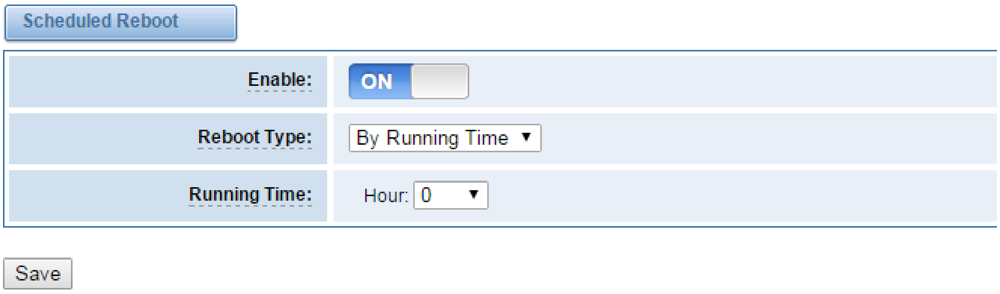
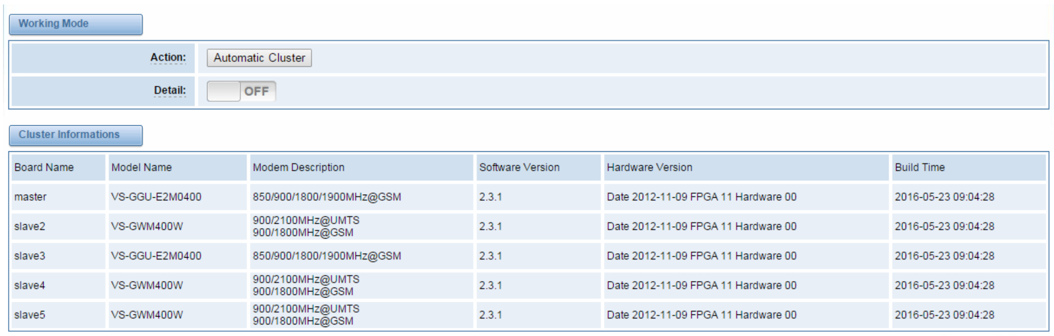


Figure 2-5 Reboot Types

If use your system frequently, you can set this enable, it can helps system work more efficient.

Cluster



Board Name	Model Name	Modem Description	Software Version	Hardware Version	Build Time
master	VS-GGU-E2M0400	850/900/1800/1900MHz@GSM	2.3.1	Date 2012-11-09 FPGA 11 Hardware 00	2016-05-23 09:04:28
slave2	VS-GWM400W	900/2100MHz@UMTS 900/1800MHz@GSM	2.3.1	Date 2012-11-09 FPGA 11 Hardware 00	2016-05-23 09:04:28
slave3	VS-GGU-E2M0400	850/900/1800/1900MHz@GSM	2.3.1	Date 2012-11-09 FPGA 11 Hardware 00	2016-05-23 09:04:28
slave4	VS-GWM400W	900/2100MHz@UMTS 900/1800MHz@GSM	2.3.1	Date 2012-11-09 FPGA 11 Hardware 00	2016-05-23 09:04:28
slave5	VS-GWM400W	900/2100MHz@UMTS 900/1800MHz@GSM	2.3.1	Date 2012-11-09 FPGA 11 Hardware 00	2016-05-23 09:04:28

Figure 2-6 cluster

Tools and Information

Reboot Tools

You can choose system reboot and asterisk reboot separately.

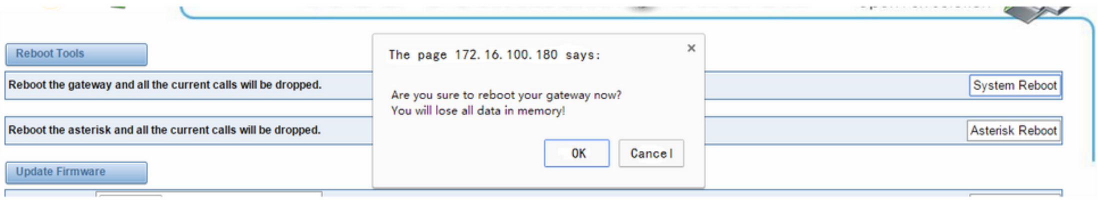


Figure 2-7 Reboot Tools

If you press "OK", your system will reboot and all current calls will be dropped. Asterisk Reboot is the same.

Update Firmware

We offer 2 kinds of update types for you, you can choose System Update or System Online Update. For System Update, you can update all the boards together or only update one of them to make sure all boards have same firmware version.

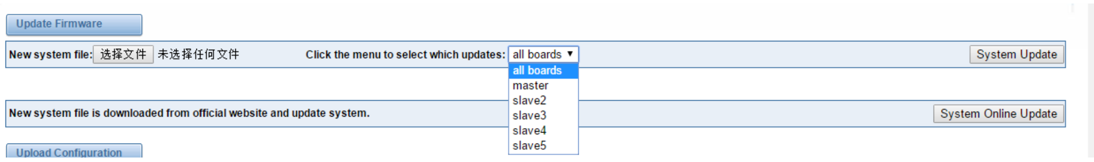


Figure 2-8 Update Firmware

Notice: In SYSTEM ---> Cluster page, you can find which board has different or incompatible firmware version.

Cluster Informations

Board Name	Model Name	Modem Description	Software Version	Hardware Version	Build Time
master	VS-GGU-E2M0400	850/900/1800/1900MHz@GSM	2.3.1	Date 2012-11-09 FPGA 11 Hardware 00	2016-04-22 11:52:12
slave2	VS-GWM400W	900/2100MHz@UMTS 900/1800MHz@GSM	2.3.1	Date 2012-11-09 FPGA 11 Hardware 00	2016-04-22 11:52:12
slave3	VS-GGU-E2M0400	850/900/1800/1900MHz@GSM	2.3.1	Date 2012-11-09 FPGA 11 Hardware 00	2016-04-22 11:52:12
slave4	VS-GWM400W	900/2100MHz@UMTS 900/1800MHz@GSM	2.3.1	Date 2012-11-09 FPGA 11 Hardware 00	2016-04-22 11:52:12
slave5	VS-GWM400W	900/2100MHz@UMTS 900/1800MHz@GSM	2.3.1	Date 2012-11-09 FPGA 11 Hardware 00	2016-04-22 11:52:12

Figure 2-9 Update Firmware

Upload and Backup Configuration

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

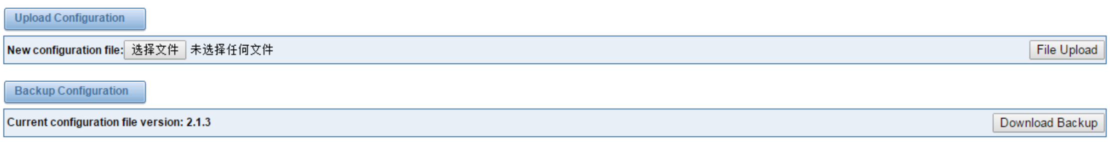


Figure 2-10 Upload and Backup Configuration

Restore Configuration

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

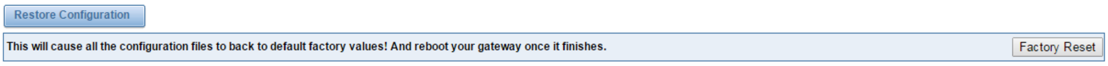


Figure 2-11 Restore Configuration

Information

On the “Information” page, there shows some basic information about the GSM/3G gateway. You can see software and hardware version, storage usage, memory usage and some help information.

Model Name:	VS-GGU-E2M0400
Modem Description:	850/900/1800/1900MHz@GSM
Software Version:	2.3.1
Hardware Version:	Date 2012-11-09 FPGA 11 Hardware 00
Slot Number:	1
Storage Usage:	1.9M/63.5M (3%)
Memory Usage:	86.882 % Memory Clean
Build Time:	2016-05-23 09:04:28
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:	783
System Time:	2016-5-26 16:00:52
System Uptime:	0 days 22:15:49

Figure 2-12 Information

3. MODULE

You can see much information about your SIM cards on this page.

MODULE Settings

Port	Carrier	Registration Status	Module Status	Actions
gsm-1.1		Undetected SIM Card		
gsm-1.2		Undetected SIM Card		
gsm-1.3	CHINA MOBILE	Registered (Home network)	READY	
gsm-1.4	CHINA MOBILE	Registered (Home network)	READY	
umts-2.1		Undetected SIM Card		
umts-2.2		Undetected SIM Card		
umts-2.3	CHINA MOBILE	Registered (Home network)	READY	
umts-2.4	CHINA MOBILE	Registered (Home network)	READY	
gsm-3.1	CHINA MOBILE	Registered (Home network)	READY	
gsm-3.2	CHINA MOBILE	Registered (Home network)	READY	
gsm-3.3(66370)	CHINA MOBILE	Registered (Home network)	READY	
gsm-3.4	CHINA MOBILE	Registered (Home network)	READY	
umts-4.1	CHINA MOBILE	Registered (Home network)	READY	
umts-4.2	CHINA MOBILE	Registered (Home network)	READY	
umts-4.3	CHINA MOBILE	Registered (Home network)	READY	
umts-4.4	CHINA MOBILE	Registered (Home network)	READY	
umts-5.1	CHINA MOBILE	Registered (Home network)	READY	
umts-5.2	CMCCA MOBILE	Registered (Home network)	READY	
umts-5.3		Not registered	INIT	
umts-5.4		Not registered	INIT	

Figure 3-1 GSM/UTMS Settings

On this page, you can see your GSM module status and click action button to configure the port.

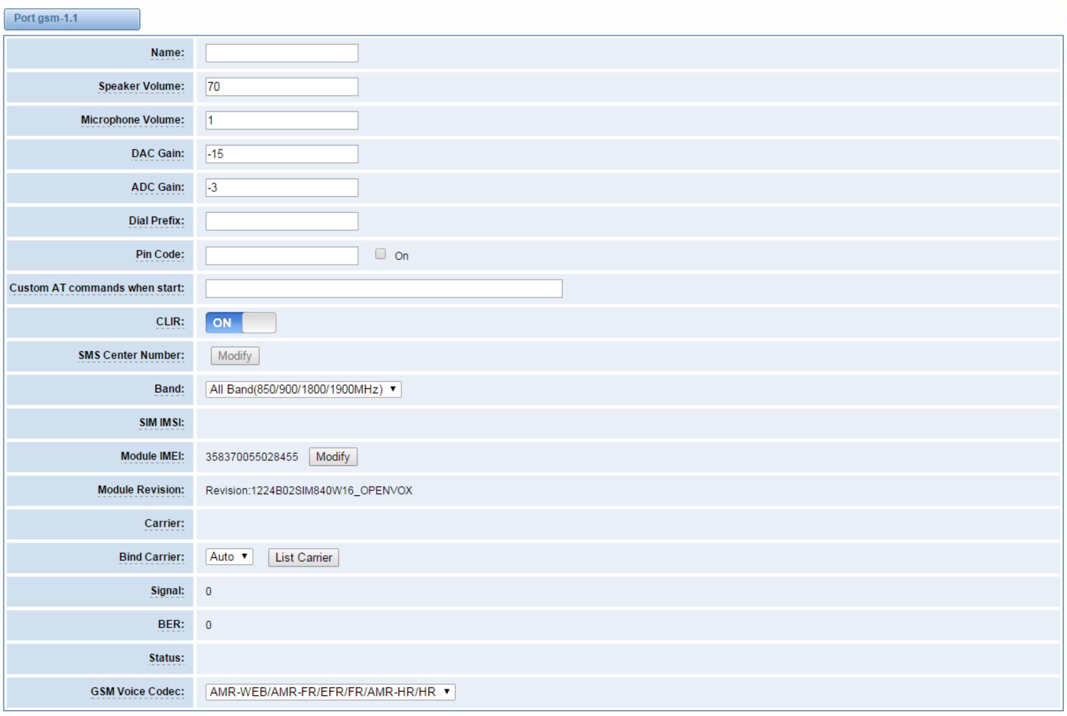


Figure 3-2 Port Configuration

As you see, we have offered **"Band"** option, you can select different bands easily and you have many options.

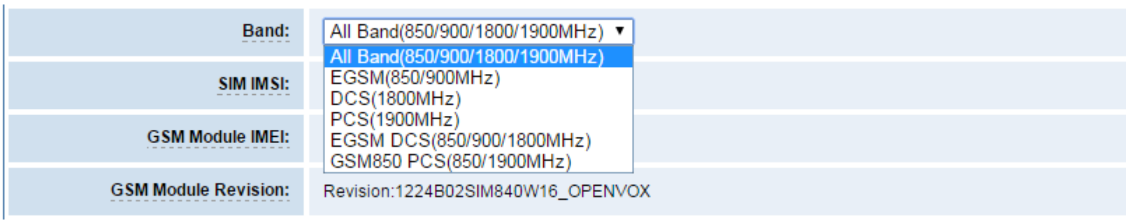


Figure 3-3 Band Binding

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

Figure 3-4 PIN Code Application

Then input your password, system will identify numbers of SIM cards. It can help to prevent SIM card from being stolen and improve security.

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).

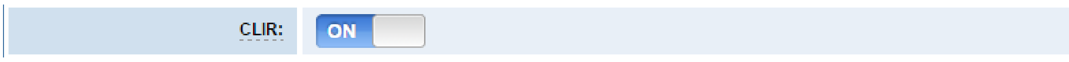


Figure 3-5 CLIR Application

OpenVox GSM/3G gateway support optional GSM Voice Codec. See picture below.

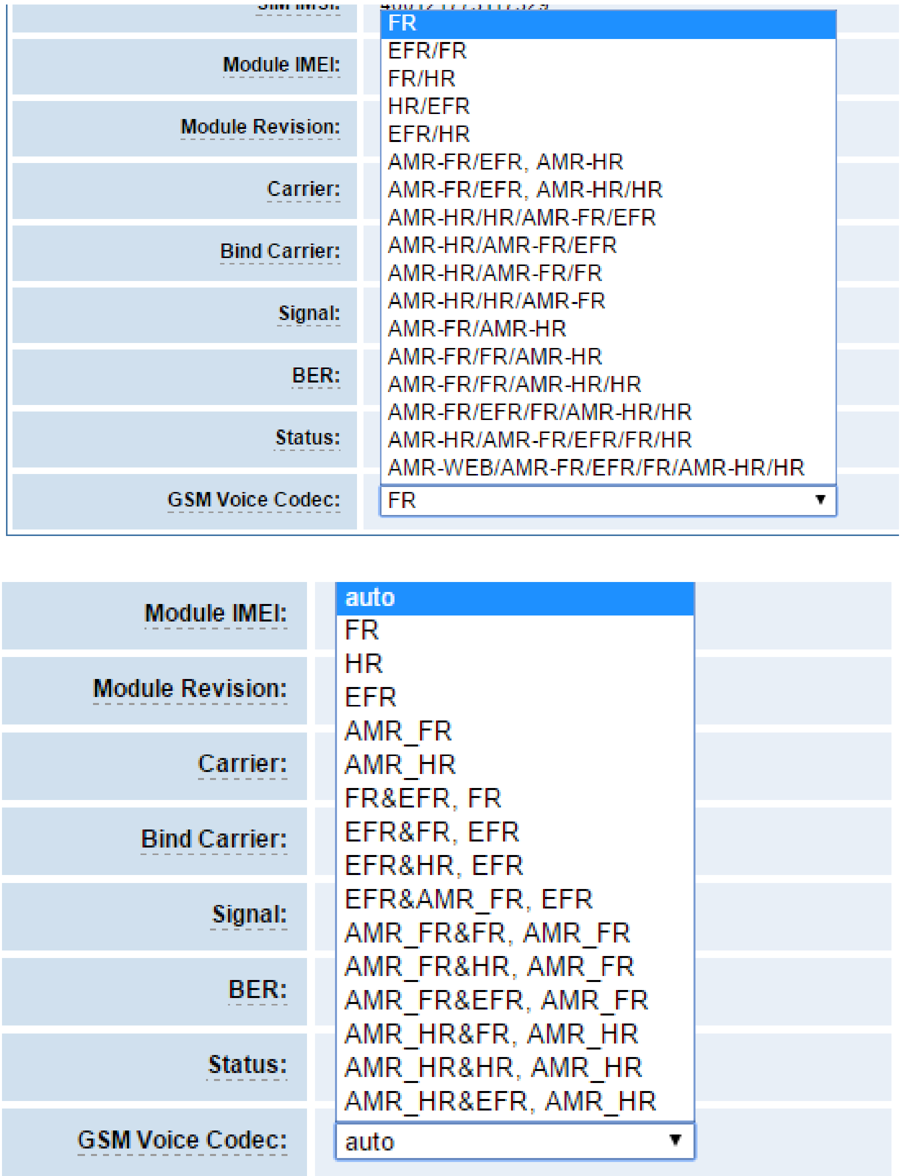


Figure 3-6 GSM Voice Codec

IMEI Modification

One more feature, we offer you IMEI automatically modification

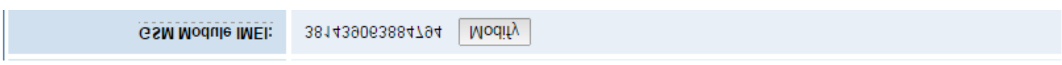
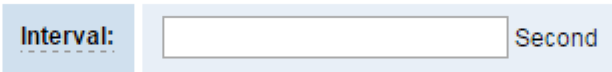


Figure 3-7 Automatically IMEI Modify

We have offered you IMEI modification function. If you want to modify your IMEI number, please do as follows. You can log in your gateway and modify IP address as follows. Input web site below on your browser. http://172.16.99.1/cgi-bin/php/gsm-autoimei.php.

Then you will see the following picture. Don't forget to switch "Enable" to "ON", or you can't change your IMEI numbers.

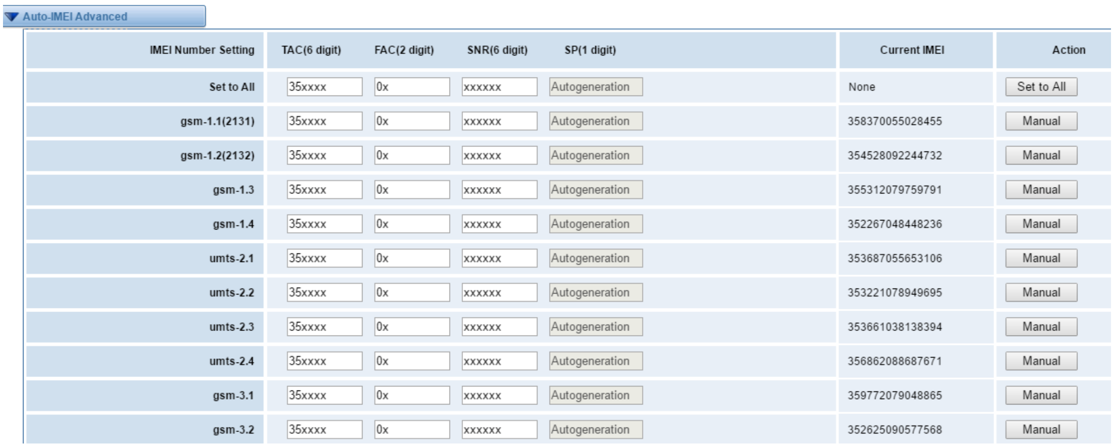
Figure 3-8 IMEI Modification

Also you can choose to modify one or more certain ports or all ports. You can set automatic modification interval by filling in the time you want. 

If you choose **"Immediately"**, then the ports you have chosen will modify IMEI numbers at once. On the contray, system will keep time from now until the time of next modification. And If you choose **"Force"**, System will hang up all your current calls, then modify IMEI.

You can press 

to do some settings. We offer you two ways to modify your IMEI. You can choose Autogeneration or Manual.



IMEI Number Setting	TAC(6 digit)	FAC(2 digit)	SNR(6 digit)	SP(1 digit)	Current IMEI	Action
Set to All	35xxxx	0x	xxxxxx	Autogeneration	None	Set to All
gsm-1.1(2131)	35xxxx	0x	xxxxxx	Autogeneration	358370055028455	Manual
gsm-1.2(2132)	35xxxx	0x	xxxxxx	Autogeneration	354528092244732	Manual
gsm-1.3	35xxxx	0x	xxxxxx	Autogeneration	355312079759791	Manual
gsm-1.4	35xxxx	0x	xxxxxx	Autogeneration	352267048448236	Manual
umts-2.1	35xxxx	0x	xxxxxx	Autogeneration	353687055653106	Manual
umts-2.2	35xxxx	0x	xxxxxx	Autogeneration	353221078949695	Manual
umts-2.3	35xxxx	0x	xxxxxx	Autogeneration	353661038138394	Manual
umts-2.4	35xxxx	0x	xxxxxx	Autogeneration	356862088687671	Manual
gsm-3.1	35xxxx	0x	xxxxxx	Autogeneration	359772079048865	Manual
gsm-3.2	35xxxx	0x	xxxxxx	Autogeneration	352625090577568	Manual

Figure 3-9 Advanced Settings

As you can see, you can set any number you wan for every port. "X" means any digits from 0 to 9.

You just need to fill in "Set to All ", then press "Set to All", you can see the interface as above. Don't forget to press "Save". Then "Current IMEI" will change. That means Autogeneration. If you want to set a certain number as your IMEI, you can press "Manual". Then you will be required to input a new IMEI.

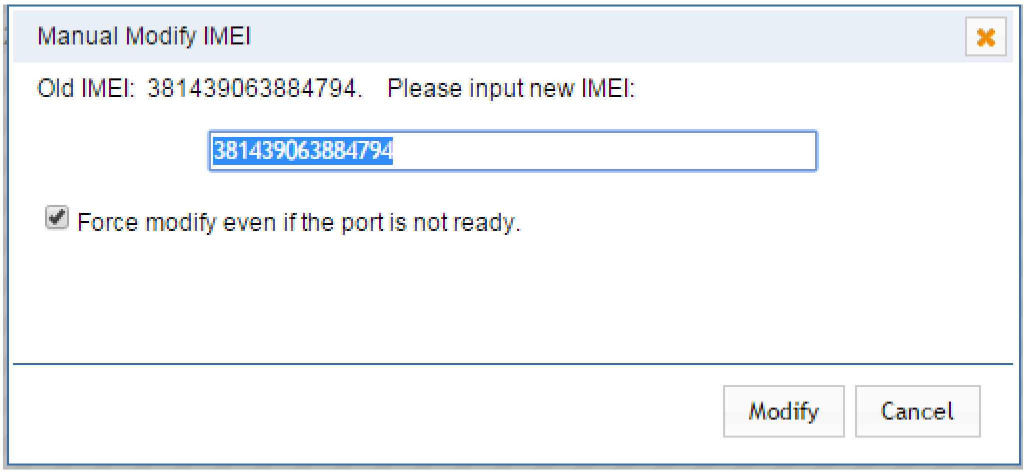


Figure 3-10 Manual

After configuration, you can press “Back Home” to return your gateway interface.

Table 3-1 Definition of GSM Settings

Options	**Definition**
Name	The alias of the GSM port. Input name without space here. Allowed characters "-_+.<>&0-9a-zA-Z".Length: 1-32 characters.
Speaker Volume	The speaker volume level, the range is 0-100. This will adjust the loud speaker volume level by an AT command.
Microphone Volume	The microphone volume, range is: 0-15. This will change the microphone gain level by an AT command.
DAC Gain	The range is: -42 to +20
ADC Gain	The range is: -42 to +20
Dial Prefix	The prefix number of outgoing calls from this GSM channel
PIN Code	Personal identification numbers of SIM card. PIN code can be modified to prevent SIM card from being stolen.

Custom AT commads when start	User custom AT commands when start system, use “|” to split AT command.
CLIR	Caller ID restriction, this function is used to hidden caller ID of SIM card number. The gateway will add ‘#31#’ in front of mobile number. This function must support by Operator.
SMS Center Number	Your SMS center number of your local carrier.
GSM Module IMEI	You can click “Modify” button and automatically modify it.

Call Duration Limit Settings

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

- **Single Call Duration Limit:** 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 by this port, it will limit your calling time within the product of

Step * Single Call Duration Limitation

And if your calling time overtops the value above, the system will hang up this call.

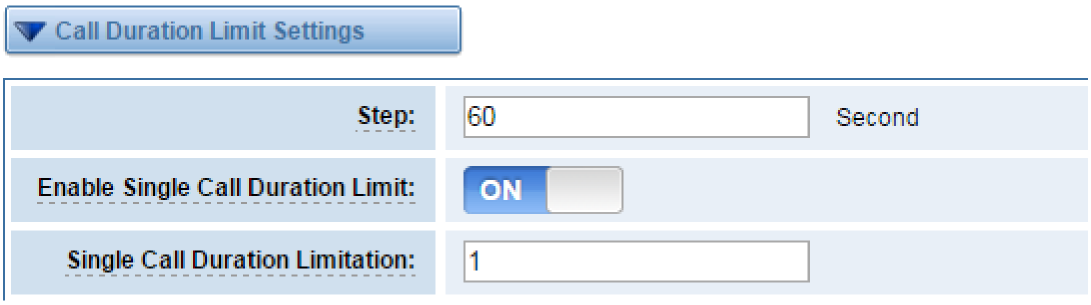


Figure 3-11 Single Settings

- **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.

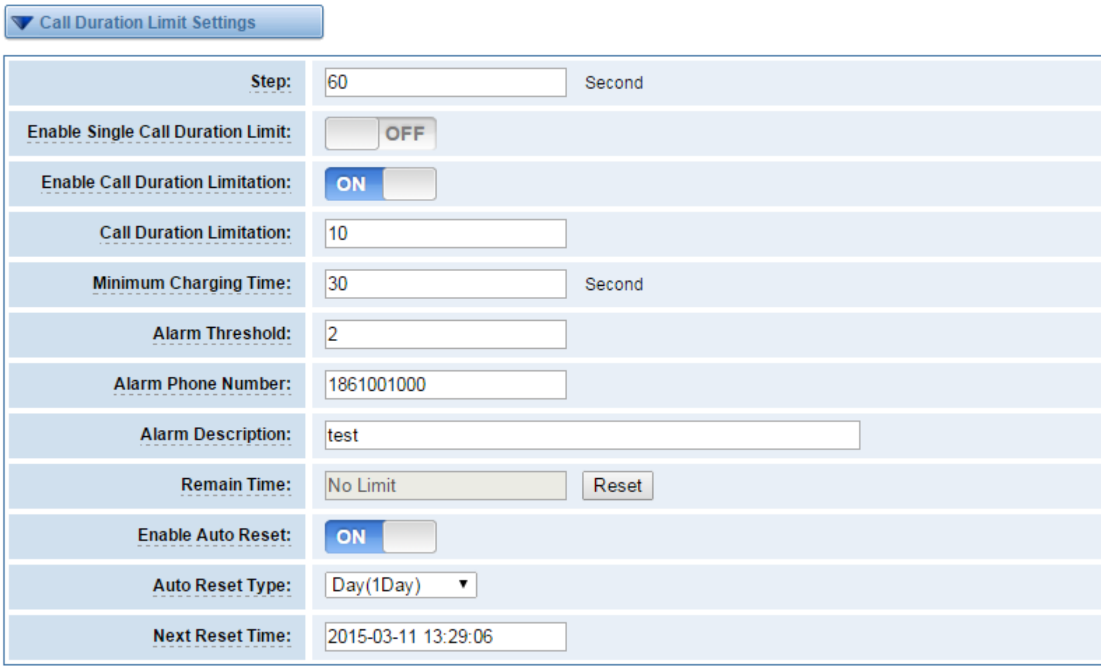


Figure 3-12 Call Duration Limitation Settings

The same algorithm with single time limitation, the total calling time of this port can't beyond the product of "Step" and "Call Duration Limitation".

If the duration of a call is less than "Minimum Charging Time", it will be not included in "Call Duration".

You can set a digit for "Alarm Threshold", when the call minutes less than this value, the gateway will send alarm info to designated phone.

You can enable your Auto Reset, then choose by day, by week, or by month.

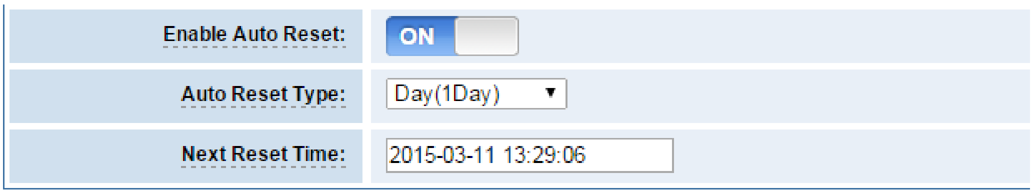


Figure 3-13 Auto Reset Settings

Table3-2 Description of Call Duration Limit Settings

Options	Definition
Step	Step length value range is 1-120 s, step length multiplied by time of single call just said a single call duration time allowed.
Enable Single Call Duration Limit	Definite maximum call duration for single call. Example: if Time of single call set to 10, the call will be disconnected after talking 10*step seconds.

Enable Call Duration Limitation	This function is to limit the total call duration of GSM channel. The max call duration is between 1 to 65535 minutes.
Call Duration Limitation	The value of limitation single call, this value range is 1-65535. Step length multiplied by time of single call just said a single call duration time allowed.
Minimum Charging Time	A single call over this time, GSM side of the operators began to collect fees, unit for seconds.
Alarm Threshold	Define a threshold value of call minutes, while the call minutes less than this value, the gateway will send alarm information to designated phone.
Alarm Description	Alarm port information description, which will be sent to user mobile phone with alarm information.
Alarm Phone Number	Receiving alarm phone number, user will received alarm message from gateway.
Remain Time	This value is multiplied by to step length is a rest call time.
Enable Auto Reset	Automatic restore remaining talk time, that is, get total call minutes of GSM channel.
Auto Reset Type	Reset call minutes by date, by week, by month.
Next Reset Time	Defined next reset date, system will count start from that date and work as Reset Period setting

You can save your configuration to other ports.

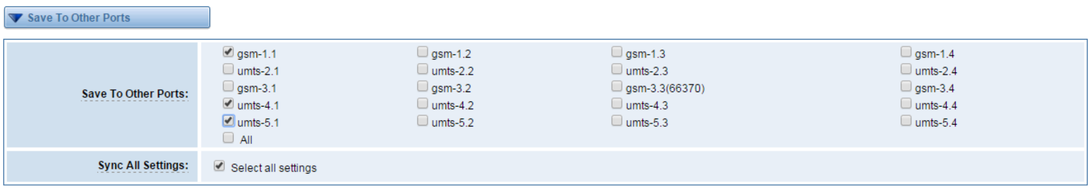


Figure 3-14 Save To Other Ports

If you have set like this, you will see many  on the Web GUI, you can set whether to check.

Notice: When you do some changes, you need to Save and Apply, then “Remain Time” will show as you set.

Your calling status will show on the main interface.

Figure 3-15 GSM/UMTS Information

Call Forwarding

Sometimes it’s not convenient for you to answer a call, if you don’t want to lose some important calls, you can choose Call Forwarding. You can choose Call Forwarding Unconditional, Call Forwarding No Reply, Call Forwarding Busy or Call Forwarding on Not Reachable. If want to cancel your call forwarding settings, you can choose Cancel All.

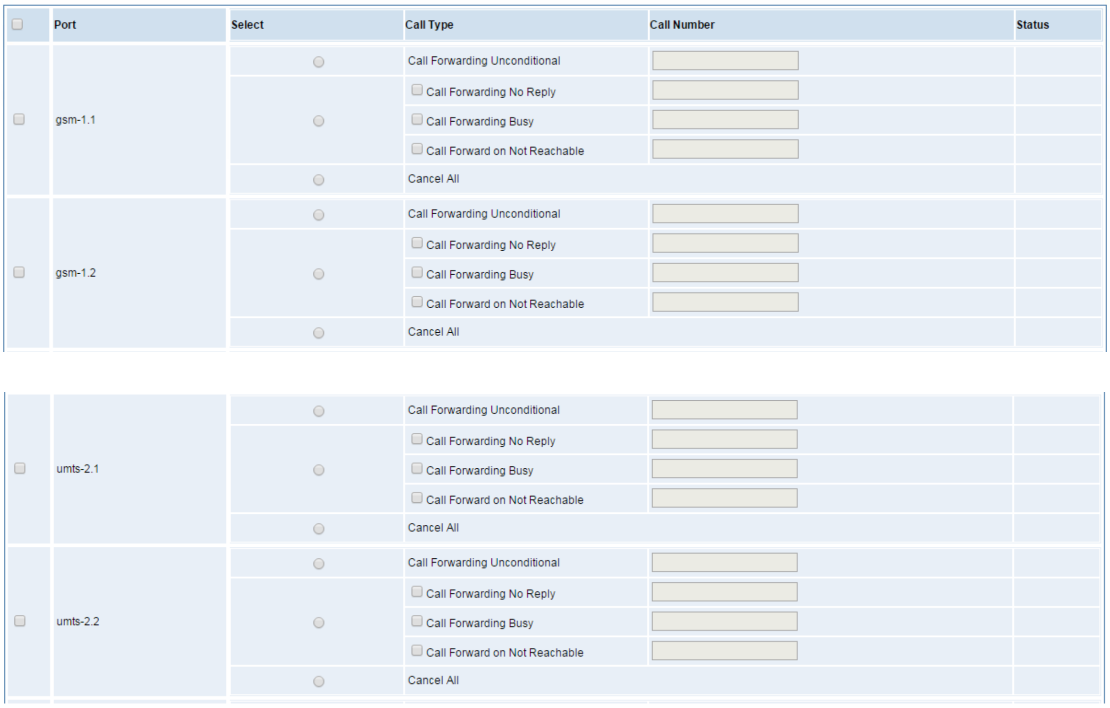


Figure 3-16 Call Forwarding

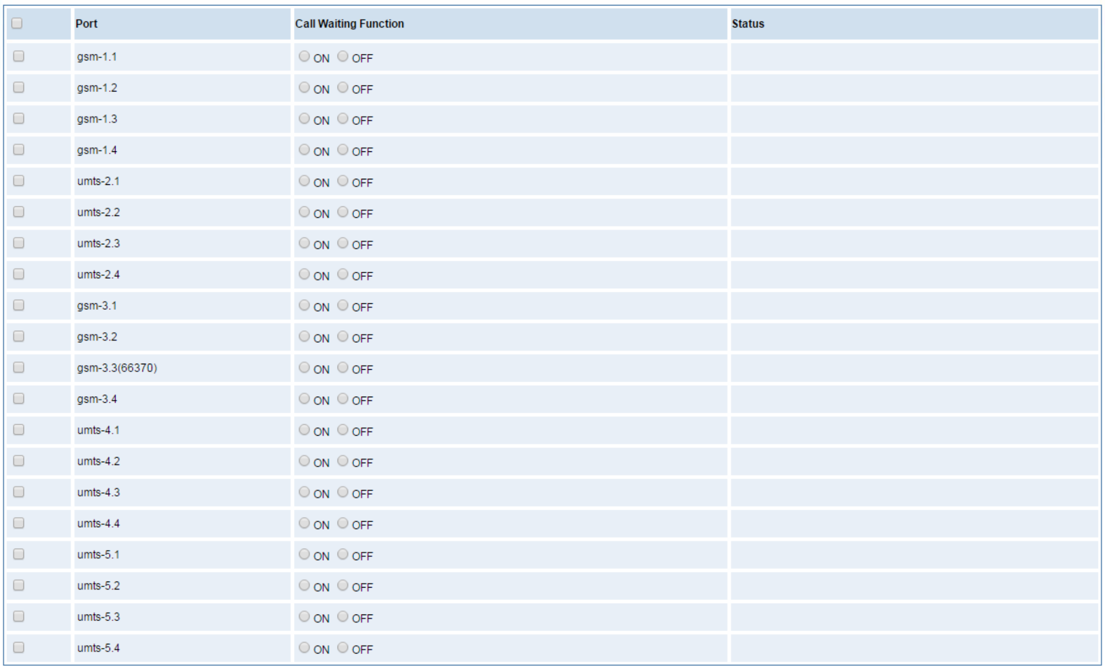
Notice: Don’t forget to save your settings. Please first press button, then press button..

Call Waiting

You can turn on/off the call waiting function of the sim cards in the ports on this page.

Table3-3 Description of Call Waiting

Options	**Definition**
Turn on call waiting	choose the ports you want to set, select "On", then click "Settings" button.
Turn off call waiting	choose the ports you want to set, select "Off", then click "Settings" button.
Status	It will show yo executing result.
Settings	used to turn on/off call waiting function
Query	used to query the status of call waiting. To see call waiting function of the ports you select is on or off.



	Port	Call Waiting Function	Status
	gsm-1.1	ON OFF	
	gsm-1.2	ON OFF	
	gsm-1.3	ON OFF	
	gsm-1.4	ON OFF	
	umts-2.1	ON OFF	
	umts-2.2	ON OFF	
	umts-2.3	ON OFF	
	umts-2.4	ON OFF	
	gsm-3.1	ON OFF	
	gsm-3.2	ON OFF	
	gsm-3.3(66370)	ON OFF	
	gsm-3.4	ON OFF	
	umts-4.1	ON OFF	
	umts-4.2	ON OFF	
	umts-4.3	ON OFF	
	umts-4.4	ON OFF	
	umts-5.1	ON OFF	
	umts-5.2	ON OFF	
	umts-5.3	ON OFF	
	umts-5.4	ON OFF	

Figure 3-17 Call Waiting

DTMF

You can do some DTMF Detection Settings if you choose “**GSM -> DTMF**”.

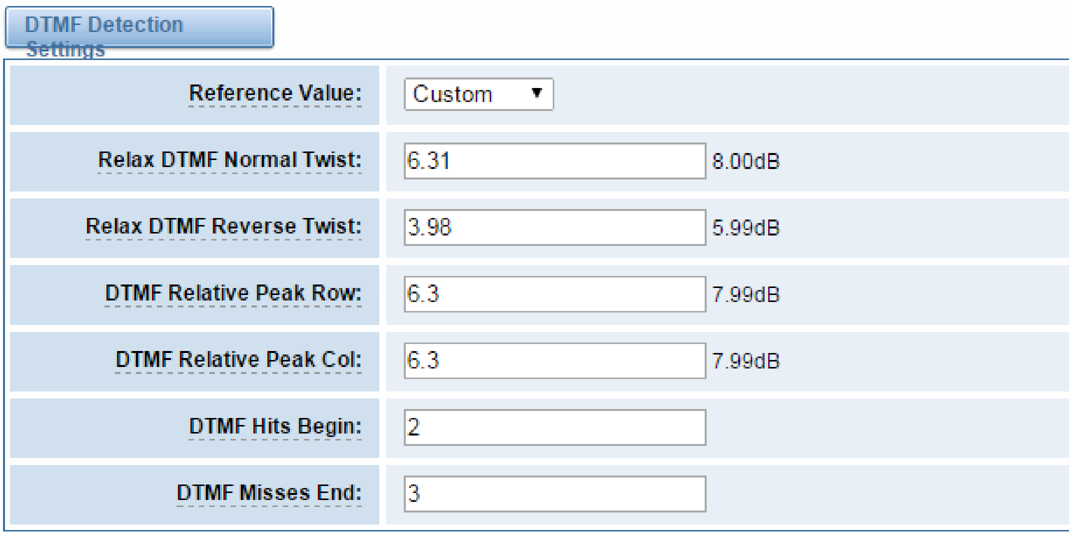


Figure 3-18 DTMF Detection Settings

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

Table3-4 Description of DTMF Detection Settings

Options	**Definition**
DTMF Normal Twist and Reverse Twist	It is the difference in power between the row and 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 value is the smaller and the detection is easier. 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 value is smaller and the detection is easier. 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	Sampling matching value. You can choose 2 or 3.
DTMF Misses End	The time interval between the two digits you input. Adjust the speed of input. The smaller value represents the shorter intervals.

BCCH

		0			1			2			3			4			5			6				
Port	Mode	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	LAC	BCCH	dbm	Status	Detail
gsm-1.1	default																							Detail
gsm-1.2	default																							Detail
gsm-1.3	default																							Detail
gsm-1.4	default																							Detail
umts-2.1																								Detail
umts-2.2																								Detail
umts-2.3																								Detail
umts-2.4																								Detail
gsm-3.1	default																							Detail
gsm-3.2	default																							Detail
gsm-3.3(66370)	default																							Detail
gsm-3.4	default																							Detail
umts-4.1																								Detail
umts-4.2																								Detail
umts-4.3																								Detail
umts-4.4																								Detail
umts-5.1																								Detail
umts-5.2																								Detail
umts-5.3																								Detail
umts-5.4																								Detail

Figure 3-19 BCCH setting

You can click Detail button, and you can change the BCCH Mode as bellow.

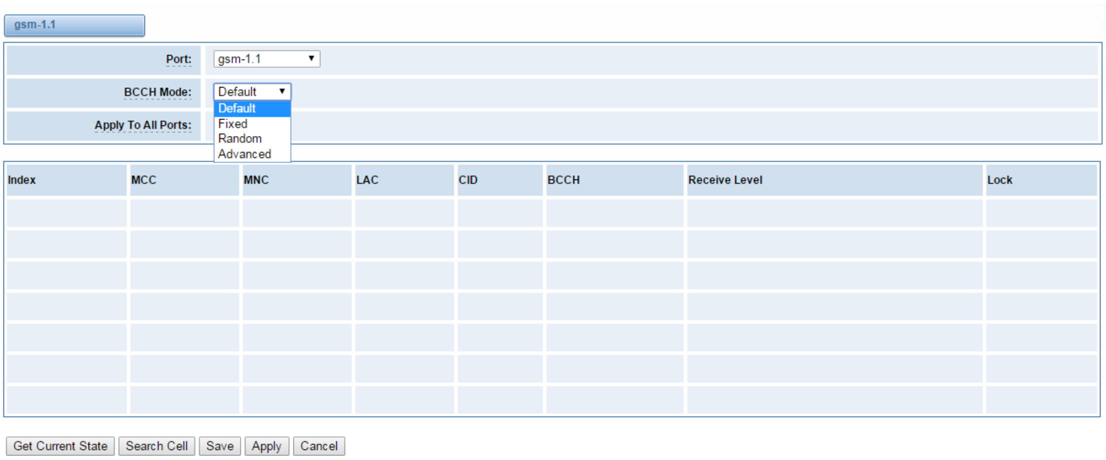


Figure 3-20 BCCH Mode

Toolkit

You can get USSD information, send AT command and check number with this module. When you have a debug of the GSM module, AT command is useful.

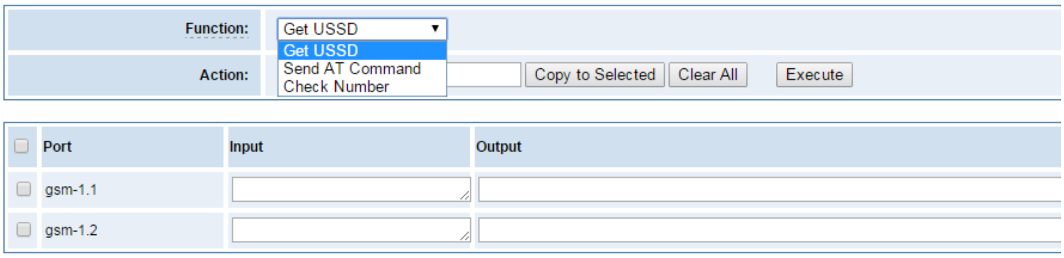


Figure 3-21 Function Options

Table3-5 Description of Definition of Functions

Options	**Definition**
Check Number	Enter a known number (like your mobile phone) 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.
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. This is useful when you have a debug of the GSM modem. e.g. perform [AT+CSQ] to check what signal qualify it is. In AT commands, there is no difference between “a” and “A”

If you want to send AT command, first you should input your command, then select certain ports and choose “**Copy to Selected**”, finally choose “**Execute**”.

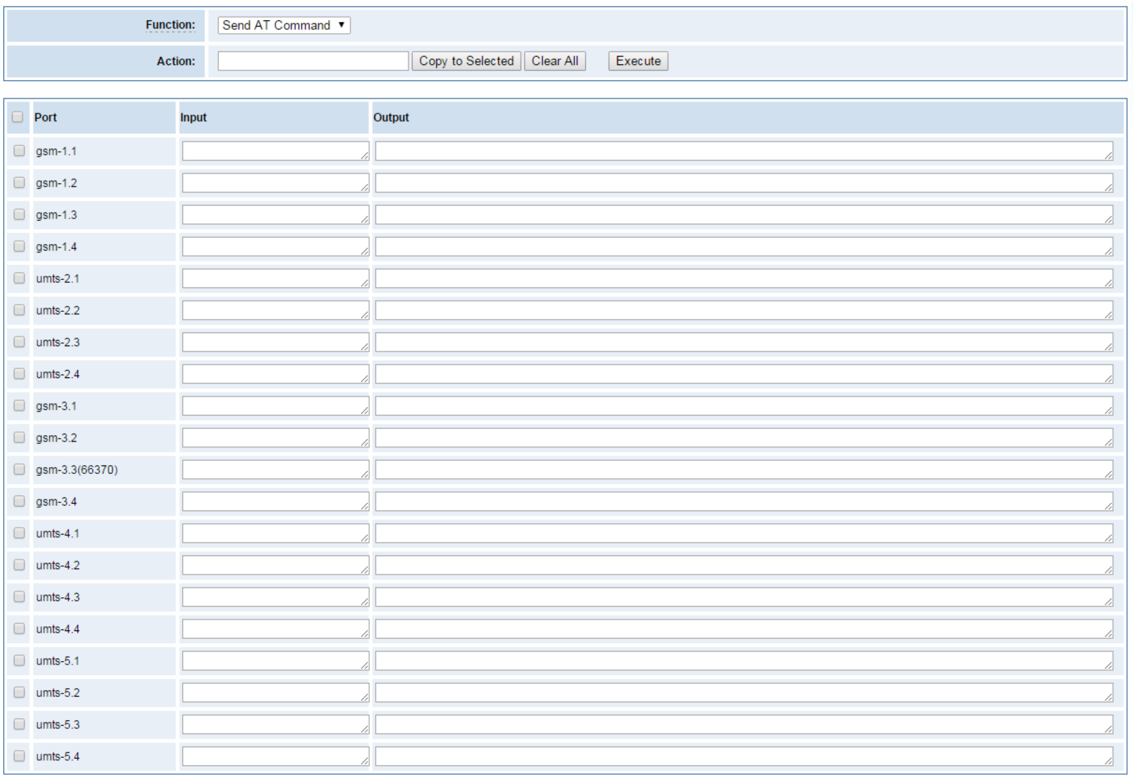
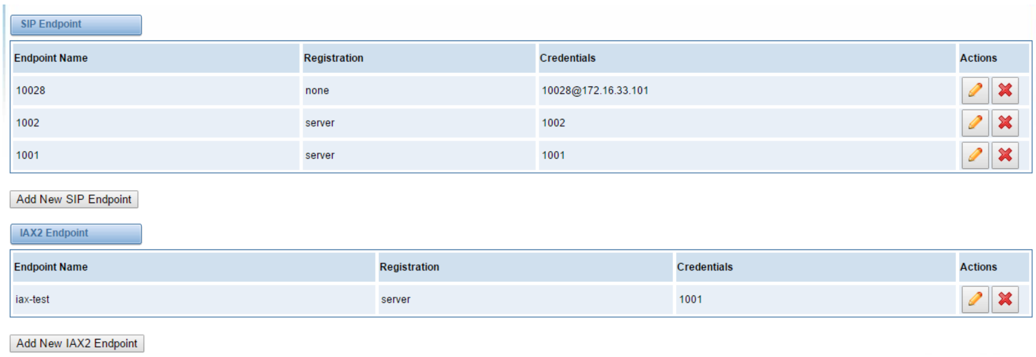


Figure 3-22 AT Command Example

4. VOIP

VOIP Endpoints

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



SIP Endpoint

Endpoint Name	Registration	Credentials	Actions
10028	none	10028@172.16.33.101	
1002	server	1002	
1001	server	1001	

Add New SIP Endpoint

IAX2 Endpoint

Endpoint Name	Registration	Credentials	Actions
iax-test	server	1001	

Add New IAX2 Endpoint

Figure 4-1 SIP&IAX2 Endpoints

Add New SIP Endpoint

Main SIP Endpoint Settings

You can click Add New SIP Endpoint 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 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.)

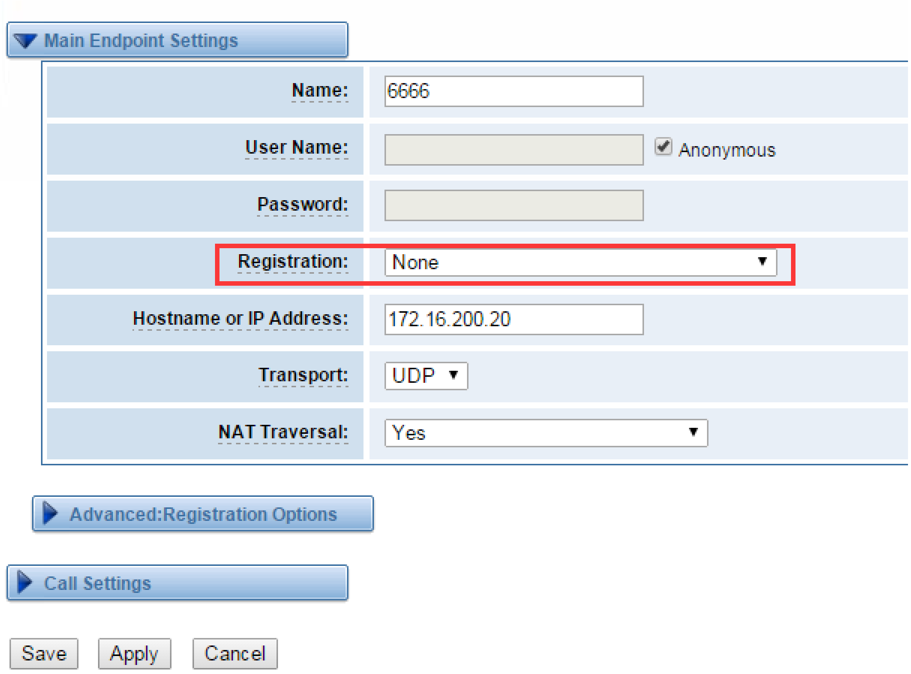


Figure 4-2 None Registration

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.

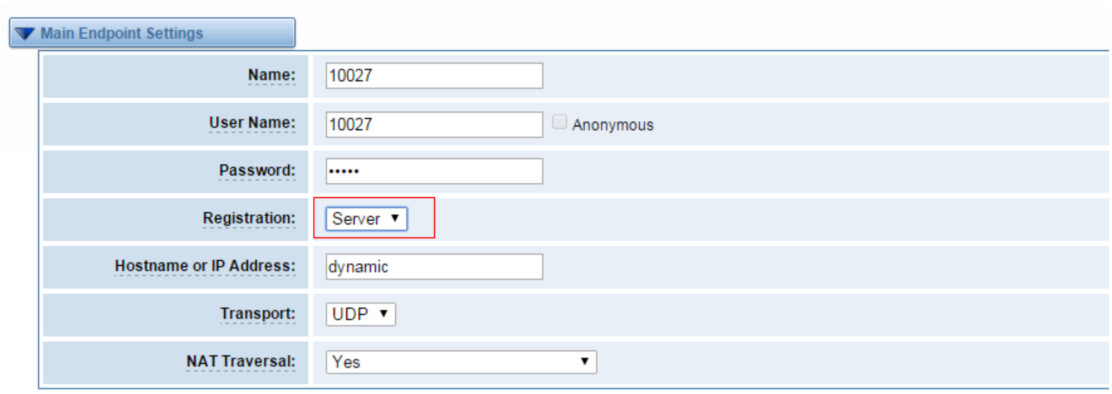


Figure 4-3 Sever

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

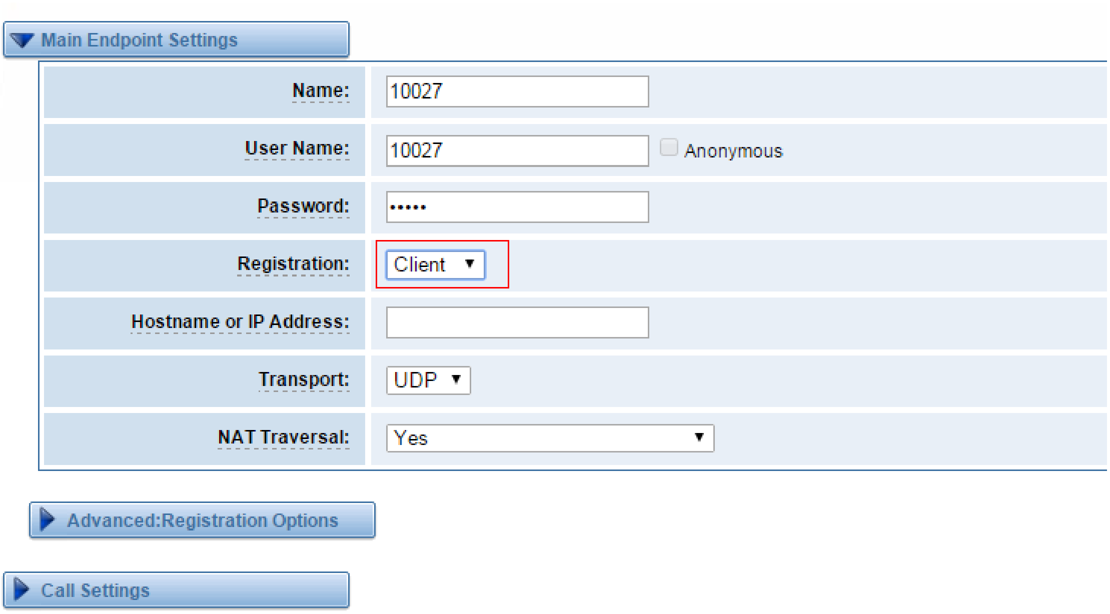


Figure 4-4 Client

table 4-1 Definition of SIP Options

Options	**Definition**
Name	Display name
Username	Register name in your SIP server
Password	Authenticating with the gateway and characters are allowed.
Registration	**None** --- Not registering; **Endpoint registers with this gateway** --- When register as this type, it means the GSM gateway acts as a SIP server, and SIP endpoints register to the gateway; **This gateway registers with the endpoint** --- When register as this type, it means the GSM gateway acts as a client, and the endpoint should be register to a SIP server;
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.
Transport	This sets the possible transport types for outgoing. Order of usage, when the respective transport protocols are enabled, is UDP, TCP,

	TLS. 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.
NAT Traversal	**No** --- Use Rport if the remote side says to use it. **Force Rport on** --- Force Rport to always be on. **Yes** --- Force Rport to always be on and perform comedia RTP handling. **Rport if requested and comedia** --- Use Rport if the remote side says to use it and perform comedia RTP handling.

Advanced -- Registration Options

Table 4-2 Definition of Registration Options

Options	**Definition**
Authentication User	A username to use only for registration.
Register Extension	When Gateway registers as a SIP user agent to a SIP proxy (provider), calls from this provider connect to this local extension.
From User	A username to identify the gateway to this endpoint.
From Domain	A domain to identify the gateway to this endpoint.
Remote Secret	A password which is only used if the gateway registers to the remote side.
Port	The port number the gateway will connect to at this endpoint.
Qualify	Whether or not to check the endpoint's connection status
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 signalling instead of sending signalling dirrectly to endpoints.

Call Settings

Table 4-3 Definition of Call Options

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 Caller ID.
Call Limit	Usually used when this sip work as a trunk. To limit number of maximum channels supported by the sip trunk.

Advanced -- Signalling Settings

Table 4-4 Definition of Signaling Options

Options	Definition
Progress Inband	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).
Allow Overlap Dialing	Whether or not the Remote-Party-ID header should be trusted.
Append user=phone to URI	Whether or not to send the Remote-Party-ID header.
Add Q.850 Reason Headers	How to set the Remote-Party-ID header: from Remote-Party-ID or from P-Asserted-Identity.
Honor SDP Version	Whether or not to display Caller ID.
Allow Transfers	Whether or not to globally enable transfers. Choosing 'no' will disable all transfers (unless enabled in peers or users). Default is enabled.

Allow Promiscuous Redirects	Whether or not to allow 302 or REDIR to non-local SIP address. Note that promiscredir when redirects are made to the local system will cause loops since this gateway is incapable of performing a "hairpin" call.
Max Forwards	Setting for the SIP Max-Forwards header (loop prevention). Send TRYING on REGISTER Send a 100 Trying when the endpoint registers.
Outbound Proxy	A proxy to which the gateway will send all outbound signaling instead of sending signaling directly to endpoints.

Advanced -- Timer Settings

Table 4-5 Definition of Timer Options

Options	**Definition**
Default T1 Timer	This timer is used primarily in INVITE transactions. The default for Timer T1 is 500ms 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. Defaults to 64 times the default T1 timer.
Session Timers	Session-Timers feature operates in the following three modes: originate, Request and run session-timers always; accept, run session-timers only when requested by other UA; refuse, do not run session timers in any case.
Minimum Session	Minimum session refresh interval in seconds. Default is 90secs.
Maximum Session Refresh Interval	Maximum session refresh interval in seconds. Defaults to 1800secs.
Session Refresher	The session refresher, uac or uas. Defaults to uas.

Main IAX2 Endpoint Settings

You can click [Add New IAX2 Endpoint] button to add a new IAX2 endpoint, and if you want to modify existed endpoints, you can click  button.

There are 3 kinds of registration types for choose. You can choose None, Endpoint registers with this gateway(work as a Server) or This gateway registers with the endpoint(work as a Client).

You can configure as follows:

If you set up a 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.

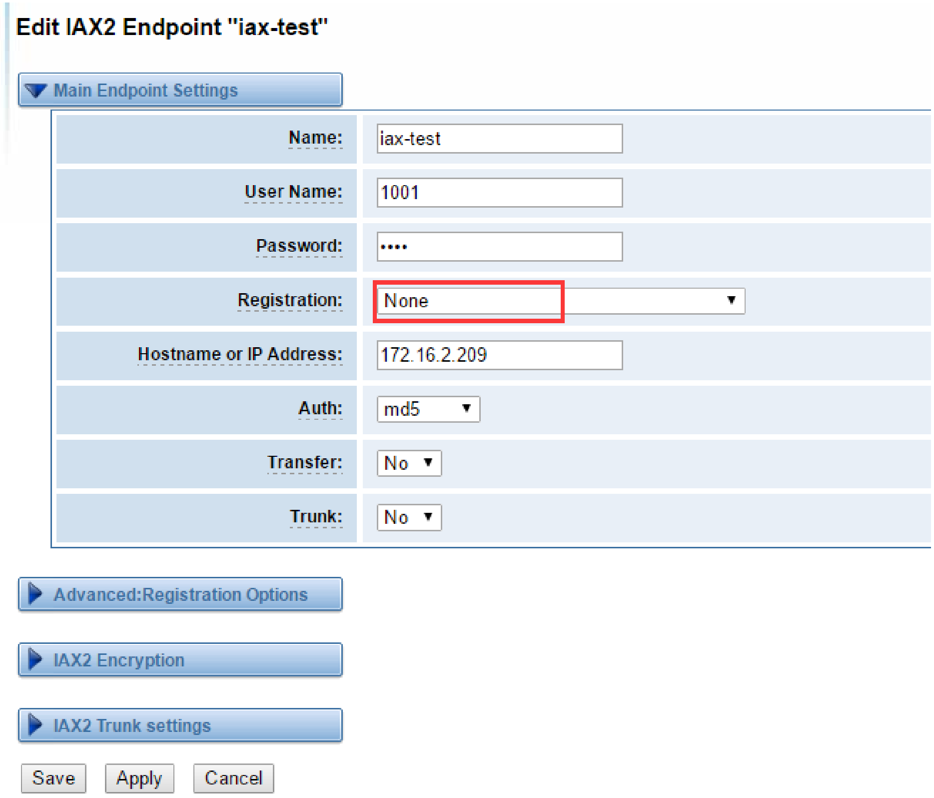


Figure 4-5 None Registration

For convenience, we have designed a method that you can register your IAX2 endpoint to your gateway, thus your gateway just work as a server.

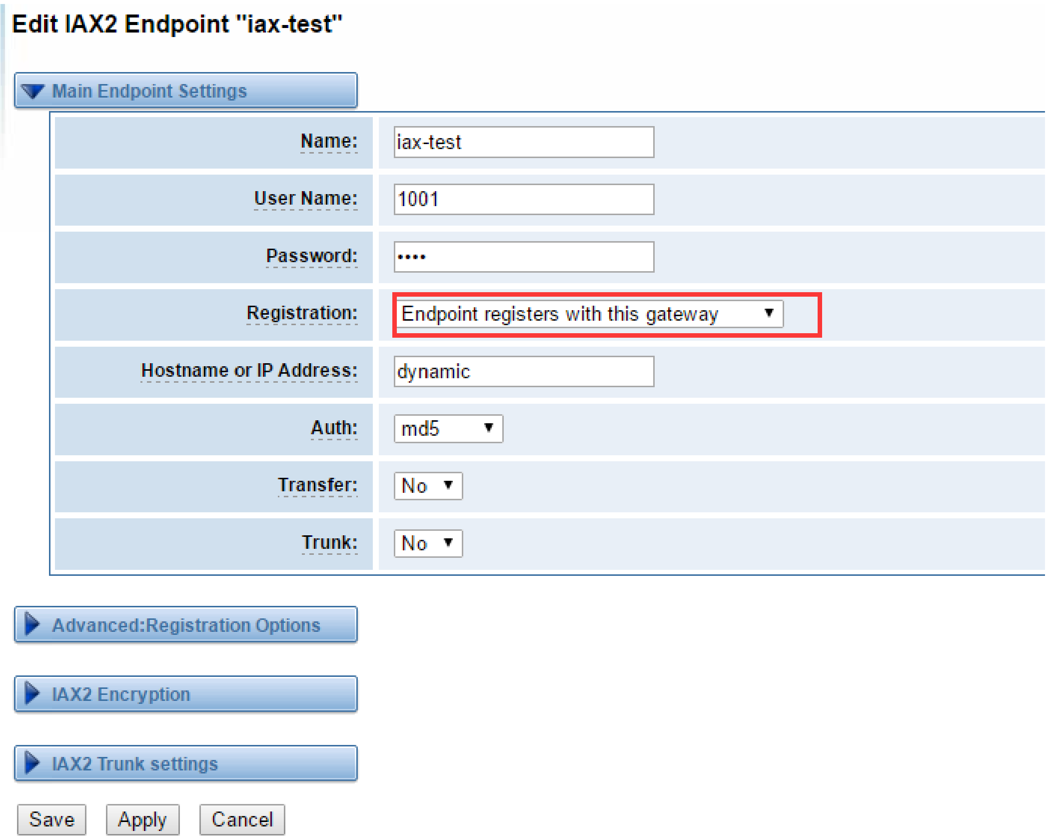


Figure 4-6 Endpoint Register With this Gateway

Also you can choose registration by "This gateway registers with the endpoint", it will work as a Client.

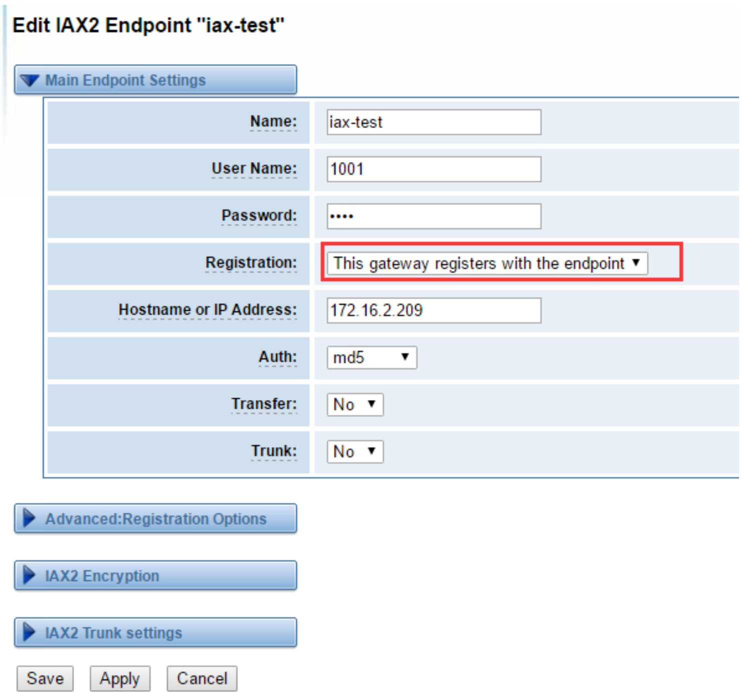


Figure 4-7 This Gateway Register With the Endpoint

Table 4-6 Definition of IAX2 Options

Options	**Definition**
Name	Display name
Username	Authenication name in your IAX2 server
Password	Authenticating with the gateway and characters are allowed.
Registration	**None** --- Not registering; **Endpoint registers with this gateway** --- When register as this type, it means the GSM/3G gateway acts as a IAX2 server, and IAX2 endpoints register to the gateway; **This gateway registers with the endpoint** --- When register as this type, it means the GSM/3G gateway acts as a IAX2 client, and the endpoint should be register to a IAX2 server;
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.
Auth	There are three authentication methods that are supported: md5, plaintext and rsa. The least secure is "plaintext", which sends passwords cleartext across the net. "md5" 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.If "rsa" authentication is used, "inkeys" is a list of acceptable public keys on the local system that can be used to authenticate the remote peer, separated by the ":" character. "outkey" is a single, private key to use to authenticate to the other side.
Transfer	This application allows you to transfer calls.
Trunk	**"trunk=yes"** Purpose: To obtain a better chart of actual bandwidth usage per codec as seen "on-the-wire" when using IAX2 trunking between two Asterisk telephony servers.

Advanced:Registration Options

Table 4-7 Definition of Registration Options

Options	Definition
Qualify, Qualify Freq Ok, Qualify Freq Not Ok	The qualify, qualifyfreqok and qualifyfreqnotok settings are used to determine the status availability of an IAX peer. If a peer is consdered to be in a reachable (OK or LAGGED) state, it is queried for availability every "qualifyfreqok" milliseconds. If it is considered to be in an UNREACHABLE state, it is queried for availability every "qualifyfreqnotok" milliseconds.The qualify= setting turns the qualify system on (if the "yes" or xxx options are used) or off (if qualify=no, which is by default). The millisecond value of the qualify= setting specifies the maximum response time of the availability acknowledgement before the peer is considered to be in a "LAGGED" state.
Qualify Smothing	Use an average of the last two PONG result to reduce falsely detected LAGGED host. The default is 'no'.
Port	The port number the gateway will connect to at this endpoint.

IAX2 Encryption

Table 4-8 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 turning this option on, encryption is automatically; turned on as well. The default is no

IAX2 Trunk Settings

Table 4-9 Definition of Trunk Options

Options	Definition

Trunk Max Size	Defaults to 128000 bytes, which supports up to 800; calls of ulaw at 20ms a frame.
Trunk MTU	With a large amount of traffic on IAX2 trunk, there is a risk of bad voice quality when allowing the Linux system to handle fragmentation of UDP packets. Depending on the side of each payload, allowing the OS to handle fragmentation may not be very efficient. This setting sets the maximum transmission unit for AIX2 UDP trunking. The default is 1240 bytes which means if a trunk's payload is over 1240 bytes for every 20ms it will be broken into multiple 1240 bytes messages. Zero disables this functionality and let's the OS handle fragmentation.
Trunk Frequency	How frequently to send trunk msgs (in ms). This is 20ms by default.
Trunk Time Stamps	Should we send timestamps for the individual sub_frames within trunk frames? There is a small bandwith use for these (less than 1kbps/call), but they ensure that frame timestamps get sent end-to-end properly. If both ends of all your trunks go directly to TDM, _and_your trunkfreq equals the frame length for your codecs, you can probably suppress these. The receiver must also need to have it enabled.
Min. RegExpire	Minimum amounts of time that IAX2 peers can request as a registration interval (in seconds).
Max. RegExpire	Maximum amounts of time that IAX2 peers can request as a registration expiration interval(in seconds).

Advanced SIP Settings

Networking

Table 4-10 Definition of Networking General Options

Options	**Definition**
UDP Bind Port	UDP Bind Port
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 Note: the gateway only uses the first host in SRV records Disabling DNS SRV lookups disables the ability to place SIP calls based on domain names to some other SIP users on the Internet specifying a port in a SIP peer definition or when dialing outbound calls with suppress SRV lookups for that peer or call.
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	Specify a prefix before routing the internal calls.

Table 4-11 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 NATed 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	Local IP address list that you added.
Subscribe Network Change Event	Through the use of the test_stun_monitor module, the gateway has the ability to detect when the perceived external network address has changed. When the stun_monitor is installed and configured, chan_sip will renew all outbound registrations when the monitor detects any sort of network change has occurred. By default this option is enabled, but only takes effect once res_stun_monitor is configured. If res_stun_monitor is enabled and you wish to not generate all outbound registrations on a network change, use the option below to disable this feature.

Match External Address Locally	Only substitute the externaddr or externhost setting if it matches.
Dynamic Exclude Static	Disallow all dynamic hosts from registering as any IP address used for statically defined hosts. This helps avoid the configuration error of allowing your users to register at the same address as a SIP provider.
Externally Mapped TCP Port	The externally mapped TCP port, when the gateway is behind a static NAT or PAT.
External Hostname	The external hostname (and optional TCP port) of the NAT.
Hostname Refresh Interval	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 might suffer from service disruption when the name server resolution fails.

Table 4-12 Definition of RTP Settings Options

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

Paesing and Compatibility

Table 4-13 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	The external hostname (and optional TCP port) of the NAT.
Shrink Caller ID	The shrinkcallerid function removes '(', ' ', ')', non-trailing '.', and '-' not in square brackets. For example, the caller

	id value 555.5555 becomes 5555555 when this option is enabled. Disabling this option results in no modification of the caller id value, which is necessary when the caller id represents something that must be preserved. By default this option is on.
Maximum Registration Expiry	Maximum allowed time of incoming registrations and subscriptions (seconds).
Minimum Registration Expiry	Minimum length of registrations/subscriptions (default 60).
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	Attempts Enter '0' for unlimited Number of registration attempts before we give up. 0 = continue forever, hammering the other server until it accepts the registration. Default is 0 tries, continue forever.

Security

Table 4-14 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. Realms MUST be globally unique according to RFC 3261. Set this to your host name or domain name.
Use Domain as Realm	Use the domain from the SIP Domains setting as the realm. In this case, the realm will be based on the request 'to' or 'from' header and should match one of the domain. Otherwise, the configured 'realm' value will be used.
Always Auth Reject	When an incoming INVITE or REGISTER is to be rejected, for any reason, always reject with an identical response equivalent to valid username and invalid password/hash

	instead of letting the requester know whether there was a matching user or peer for their request. This reduces the ability of an attacker to scan for valid SIP usernames. This option is set to 'yes' 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 guest calls (default is yes, to allow). If your gateway is connected to the Internet and you allow guest calls, you want to check which services you offer everyone out there, by enabling them in the default context.

Media

Table 4-15 Instruction of Media

Options	Definition
Premature Media	Some ISDN links send empty media frames before the call is in ringing or progress state. The SIP channel will then send 183 indicating early media which will be empty - thus users get no ring signal. Setting this to "yes" will stop any media before we have call progress (meaning the SIP channel will not send 183 Session Progress for early media). Default is 'yes'. Also make sure that the SIP peer is configured with progressinband=never. In order for 'noanswer' applications to work, you need to run the progress() application in the priority before the app.
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.

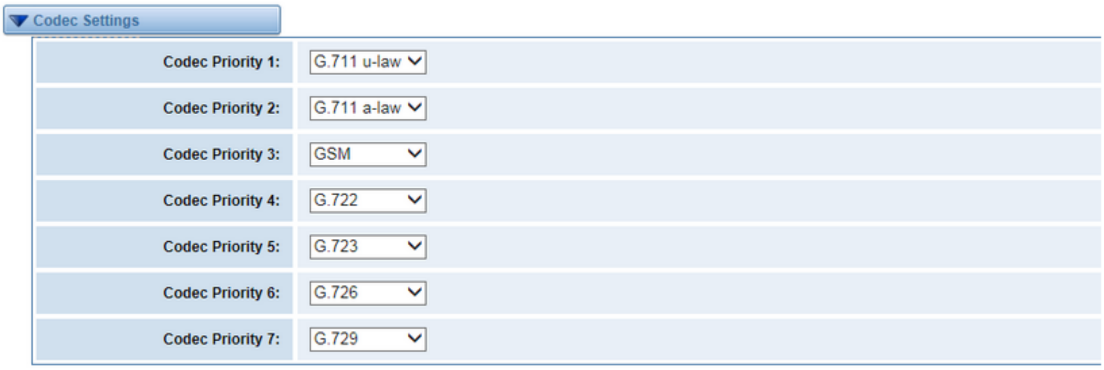


Figure 4-8 Codec Settings

Advanced IAX2 Settings

Table 4-16 Instruction of General

Options	**Definition**
Bind Port	Bind port and bindaddr may be specified
Enable IAXCompat	More than once to bind to multiple addresses, but the first will be the default.
Enable Nochecksums	Set iaxcompat to yes if you plan to use layered switches or some other scenario which may cause some delay when doing a lookup in the dialplan. It incurs a small performance hit to enable it. This option cause Asterisk to spawn a separate thread when it receives an IAX DPREQ (Dialplan Request) instead of blocking while it waits for a response.
Enable Delay Reject	Disable UDP checksums (if no checksums is set, then no checksums will be calculated/checked on system supporting the feature)
ADSI	ADSI (Analog Display Services Interface) can be enable if you have (or may have) ADSI compatible CPE equipment.
SRV Loopup	Whether or not to perform an SRV lookup on outbound calls
AMA Flags	You may specify a global default AMA flag for iaxtel calls. These flags are used in the generation of call detail records.
autokill	If we don’t get ACK to our NEW within 2000ms,and autokill 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 for bad connections.
Language	You may specify a global default language for users. This can be specified also on a per-user basis. If omitted, will fallback to English(en)
Account Code	You may specify a default account for Call Detail Records (CDRs) in addition specifying on a per-user basis.

Table 4-17 Instruction of Music on Hold

Options	**Definition**
Mohsuggest	The 'Mohsuggest' option specifies which music on hold class to suggest to the peer channel when this channel place the peer on hold. It may be specified globally or on a per-user or per-peer basis.
Mohinterpret	You may specify a global default language for users. This can be specified also on a per-user basis. If omitted, will fall back to English(en)

Table 4-18 Instruction of Codec Settings

Options	**Definition**
Band Width	Specify bandwith of low, medium, or high to control which codes are used in general
Disallow	Fine tune codes here using "allow" and "disallow" clause with specific codes
Allow	Fine tune codes here using "allow" and "disallow" clause with specific codes
Codec Priority	Codec priority controls the codec negotiation of an inbound IAX2 call. This option is inherited to all user entity separately which will override the setting in general.

Table 4-19 Instruction of Jitter Buffer

Options	**Definition**
Jitter Buffer	Global default as to whether you want the jitter buffer at all
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.
Max Jitter Buffers	A maximum size for the jitter buffer
Resyncthreshold	When the jitter buffer notice a significant change in delay that

	continue 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.
Max Jitter Interps	The maximum number of interpolation frames the jitter buffer should return in a row. Since some clients do not send CNG/DTX frames to indicate silence, the jitter buffer will assume silence has begun after returning this many interpolations. This prevents interpolating throughout a long silence.
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 may help if your network normally has low jitter, but occasionally has spikes.

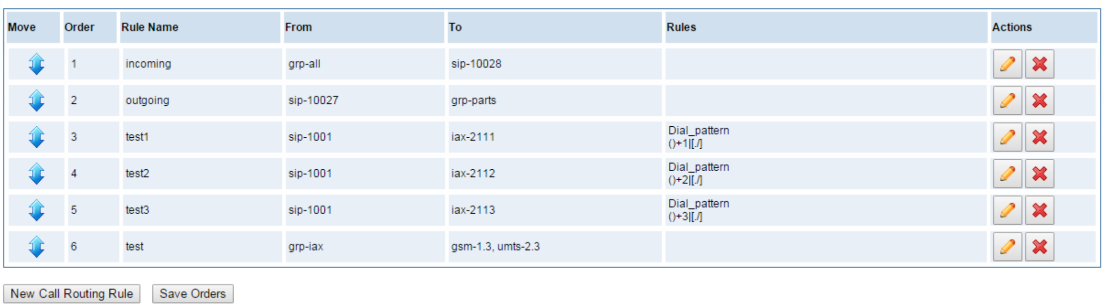
Table 4-20 Instruction of Misc Settings

Options	**Definition**
IAX Thread Count	Establishes the number of iax helper thread to handle I/O
IAX Max Thread Count	Establishes the number of extra dynamic threads that may by spawned to handle I/O
Max Call Number	The 'maxcallnumbers' 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 'maxcallnumbers' option, this limit is not separate for each individual IP address. Any connection resulting in a non-call token validated call number being allocated contributes to this limit. For use cases, see the call should be sufficient in most cases.

Table 4-21 Instruction of Quality of Service

Options	**Definition**
Tos	Type of service
Cos	Class of service

5. Routing



Move	Order	Rule Name	From	To	Rules	Actions
	1	incoming	grp-all	sip-10028		
	2	outgoing	sip-10027	grp-parts		
	3	test1	sip-1001	iax-2111	Dial_pattern ()+1[[.]]	
	4	test2	sip-1001	iax-2112	Dial_pattern ()+2[[.]]	
	5	test3	sip-1001	iax-2113	Dial_pattern ()+3[[.]]	
	6	test	grp-iax	gsm-1.3, umts-2.3		

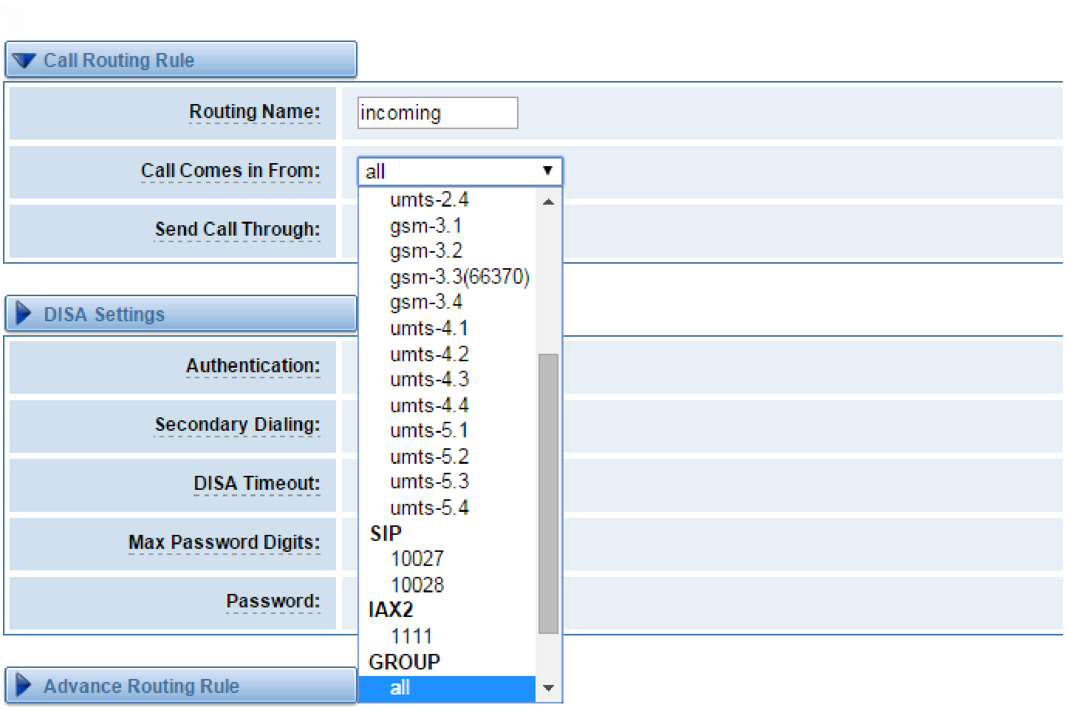
New Call Routing Rule   Save Orders

Figure 5-1 Routing Rules

You are allowed to set up new routing rule by New Call Routing Rule , 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 Save Orders button to save what you set. Rules shows current routing rules. Otherwise you can set up unlimited routing rules.

Call Routing Rule

You can click button to set up your routings.



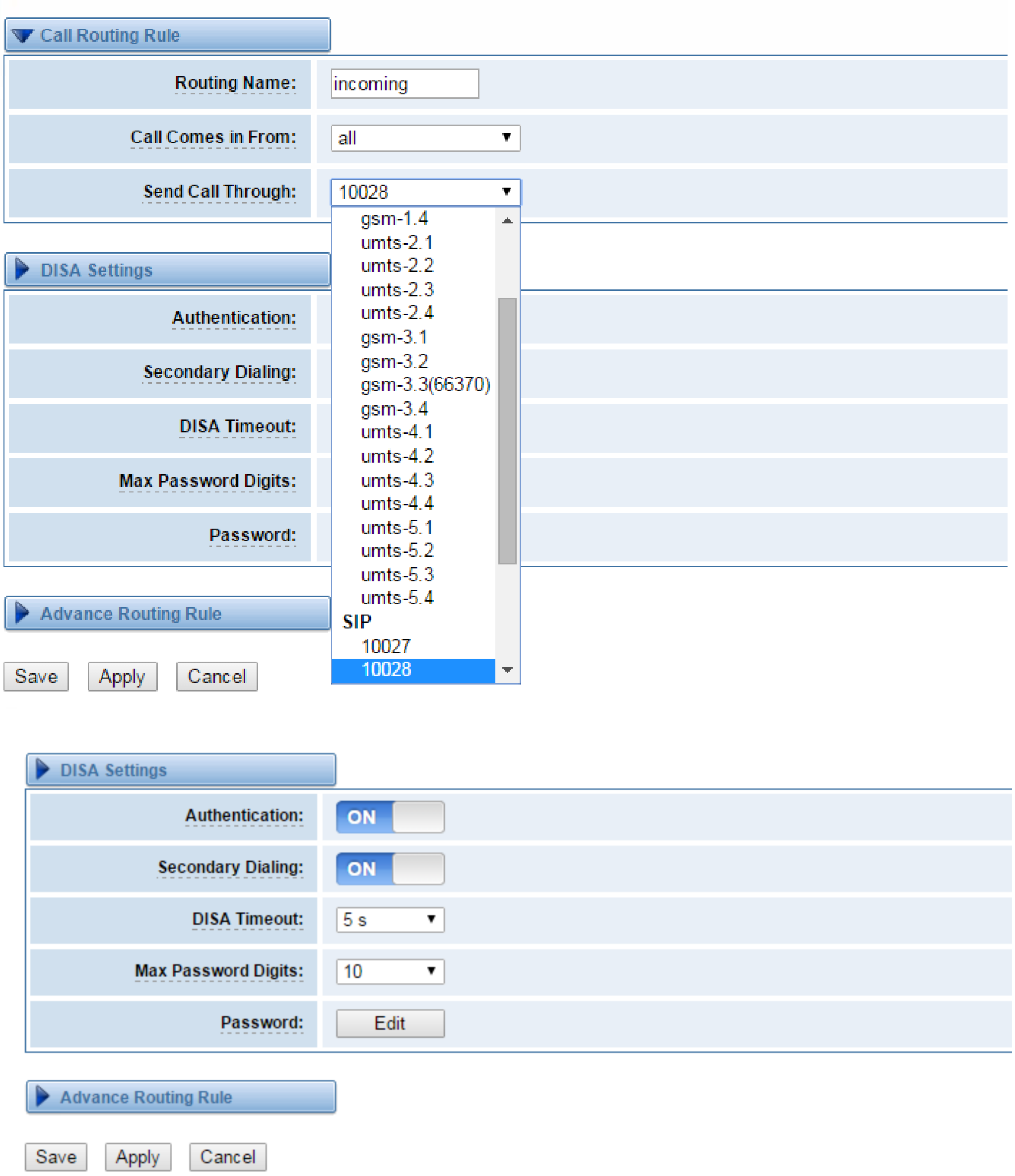


Figure 5-2 Example of Set up Routing Rule

The figure above realizes that calls from “1001” SIP endpoint switch you have registered will be transferred to Port-1. When “Call Comes in From” is T1/E1 Port, “prepend”, “prefix” and “match pattern” in “Advanced Routing Rule” are ineffective, and just “CallerID” option is available.

Table5-1 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, 'SIP2GSM' or 'GSM2SIP').
Call Comes in From	The launching point of incoming calls.
Send Call Through	The destination to receive the incoming calls.

Table5-2 Description of Advanced Routing Rule

Options	Definition
Dial Patterns that will use this Route	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). Rules: **X** matches any digit from 0-9 **Z** matches any digit from 1-9 **N** matches any digit from 2-9 **[1237-9]** matches any digit in the brackets (example: 1,2,3,7,8,9) **. wildcard**: matches one or more dialed digits. **prepend**: Digits to prepend to a successful match If the dialed number matches the patterns specified by the subsequent columns, then this will be prepended before sending to the trunks **prefix**: Prefix to remove on a successful match 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. **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 **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.

	The above special matching sequences can be used for CallerID matching similar to other number matches.
Set the Caller ID Name to	What caller ID name would you like to set before sending this call to the endpoint.
Set the Caller ID Number to	What caller number would you like to set before sending this call to the endpoint.
Forward Number	What destination number will you dial? This is very useful when you have a transfer call.
Failover Call Through Number	The gateway will attempt to send the call out each of these in the order you specify. You can create various time routes and use these time conditions to limit some specific calls.

Figure 5-3 Time Patterns that will use this Route

If you configure like this, then from January to March, from the first day to the last day of these months, from Monday to Thursday, from 00:00 to 02:00, during this time (meet all above time conditions), all calls will follow this route. And the time will synchronize with your Sever time.

Figure 5-4 Failover Call Through Number

You can add one or more “Failover Call Through Numbers”.

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 GSM or SIP to groups. Then if you want to make a call, it will find available port automatically.

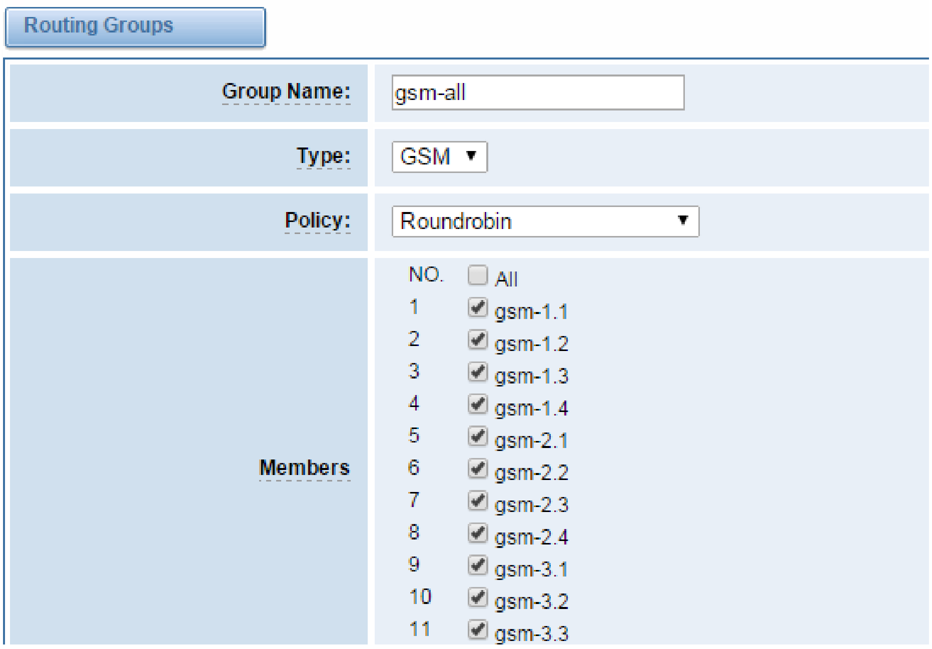


Figure 5-5 Routing Group

MNP Settings

Mobile Number Portability allows switching between mobile phone operators without changing the mobile number. Sounds simple, but there are loads of tasks performed behind the scene at the operator end.

The URL is shown in the password string way. So please type the url in other place such a txt file, check it, then copy it to the gateway. The outgoing number in the url should be replaced by the variables **${num}**.

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, when config the MNP url, should replce it with ${num}. Then it turns to https://s1.bichara.com.br:8181/chkporta.php?user=832700&pwd=sdsfdg&tn=${num}.

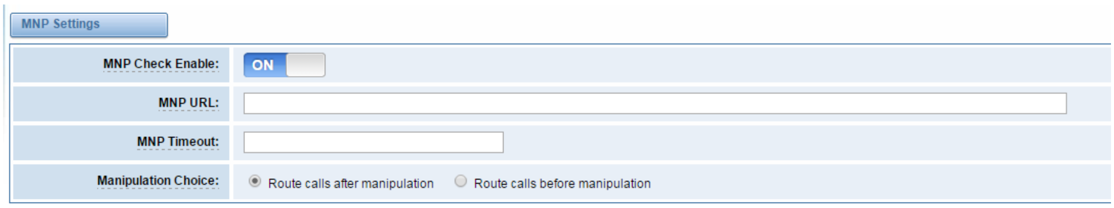


Figure 5-6 MNP Settings

6. SMS

General

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

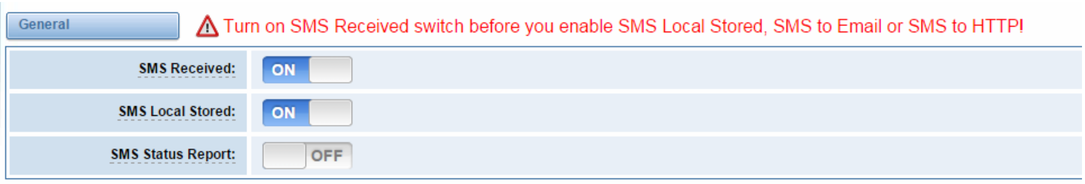


Figure 6-1 SMS Settings

Sender Options

You can change sender options here, include resend, times of resend.

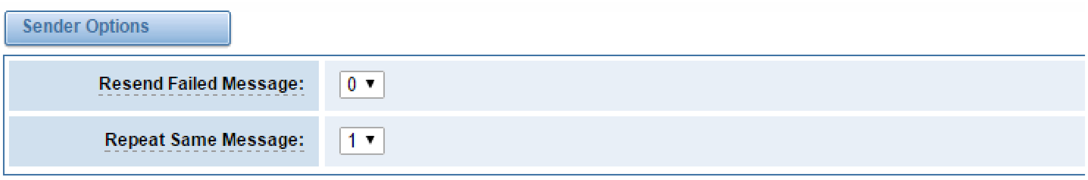


Figure 6-2 Sender Options

Table 6-1 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.

SMS to Email

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

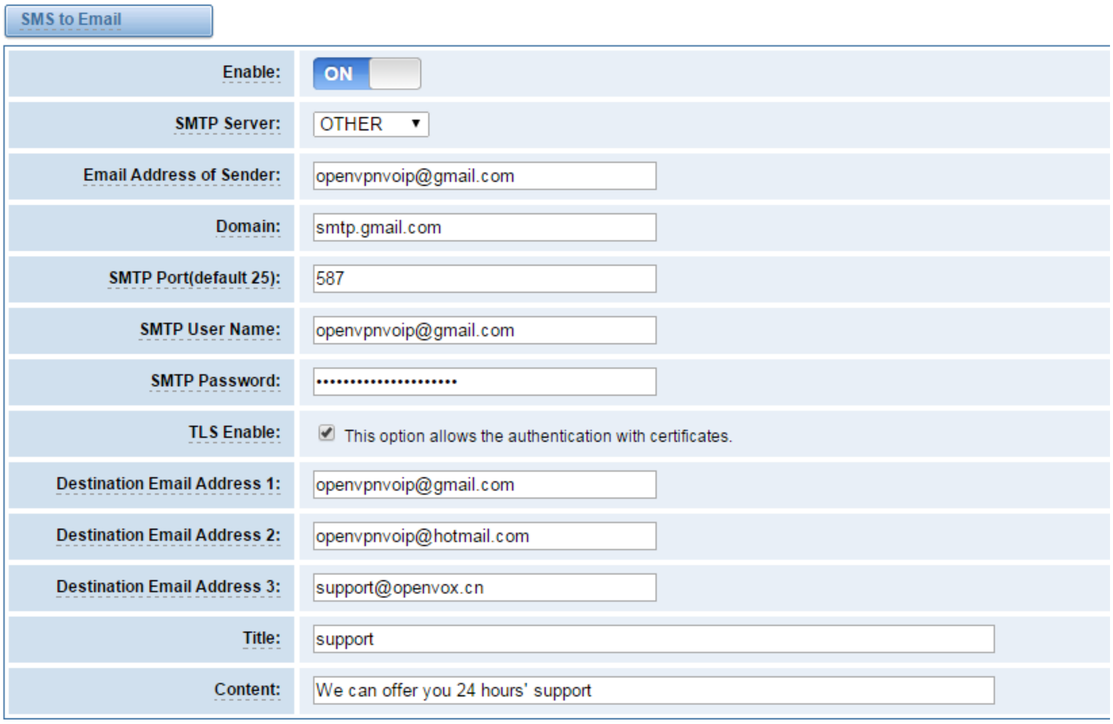


Figure 6-3 SMS to Email

Table6-2 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	×

Table6-3 Definition of SMS to E-mail

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

Email Address of Sender	To set the email address of an available email account. For example, openvpnvoip@gmail.com.
Domain	To set outgoing mail server. e.g. smtp.gmail.com
SMTP Port	To set port number of outgoing mail server. (Default is 25)
SMTP User Name	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. e.g. 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

Allowing endpoints to send some specific KEY WORDS and corresponding PASSWORD to operate the gateway and message is case-sensitive. In default, this function is disabled.

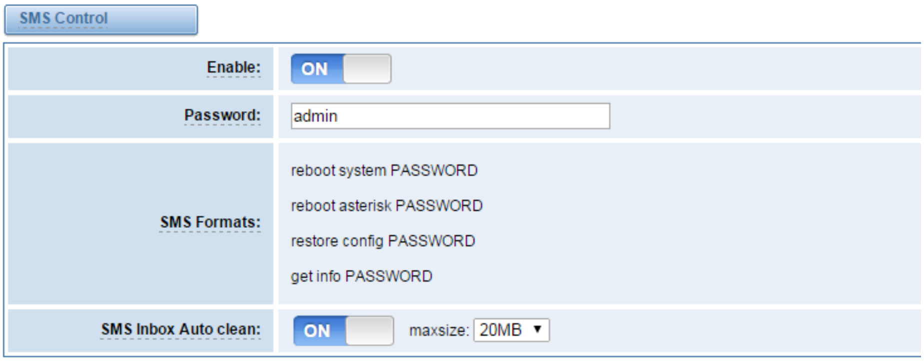


Figure 6-4 SMS Control

For example, SMS control password is 123456789 which has nothing to do with the login password, you can send “get info 123456789” to the GSM module’s phone number to get your gateway’s IP information.

Table6-4 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 configs 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

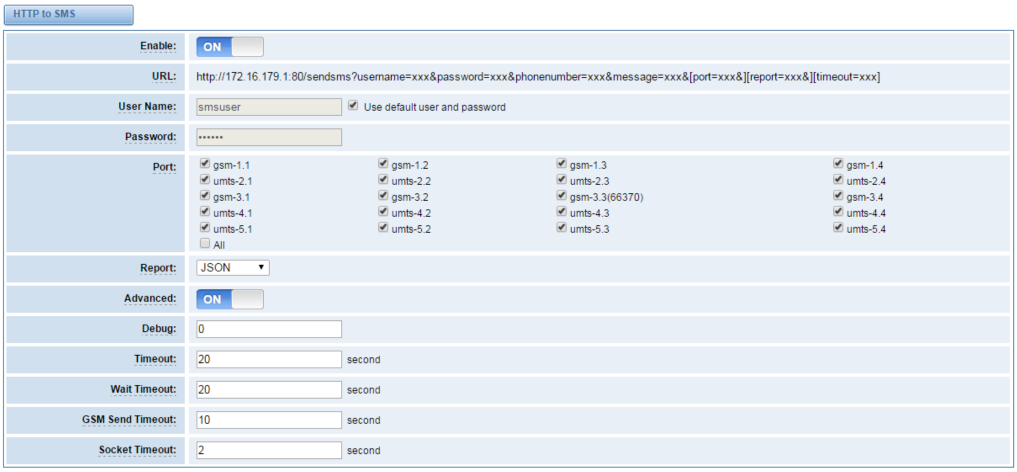


Figure 6-5 HTTP to SMS

SMS to HTTP

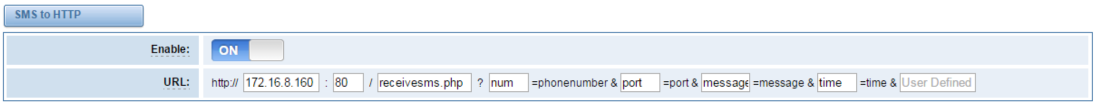


Figure 6-6 SMS to HTTP setting

SMS Sender

You can choose one or more ports to send SMS to the destination number, different numbers should be separated by symbols: '\r', '\n', space character, semicolon and comma. Then you can see much feedback information.

Figure 6-7 SMS Sender

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.

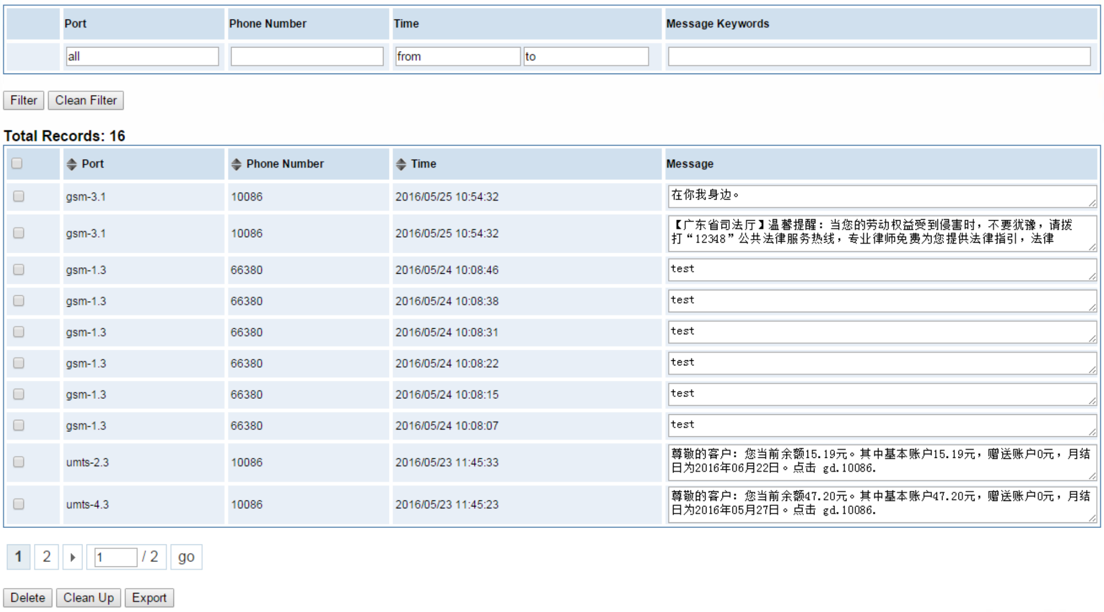


Figure 6-8 SMS Inbox

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.

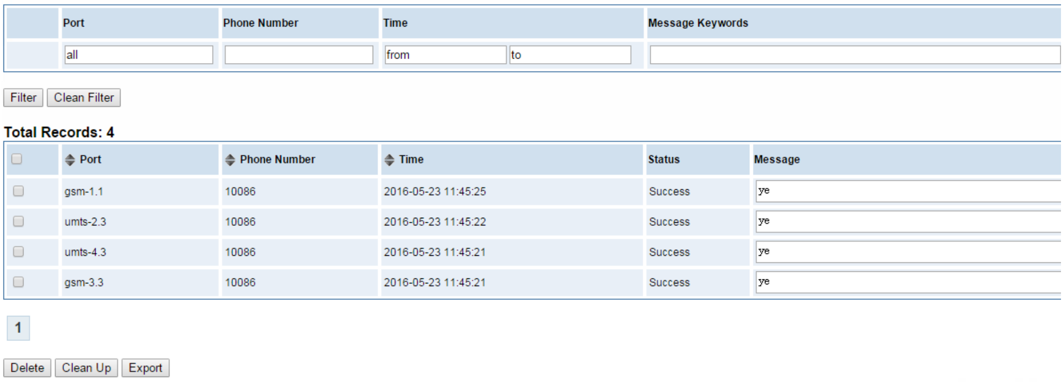


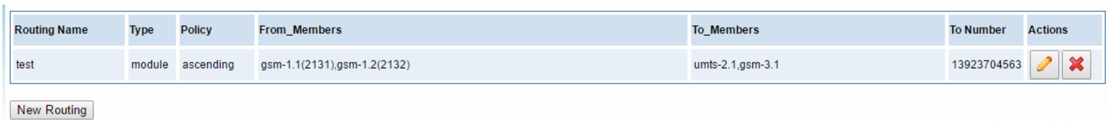
Figure 6-9 SMS Outbox

SMS Forwarding

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

click New Routing button to add new routing.

Such as:



SMS received by gsm-1.1 and gsm-1.2, will be transfered to phone number 13923704563 through port umts-2.1 or gsm-3.1.

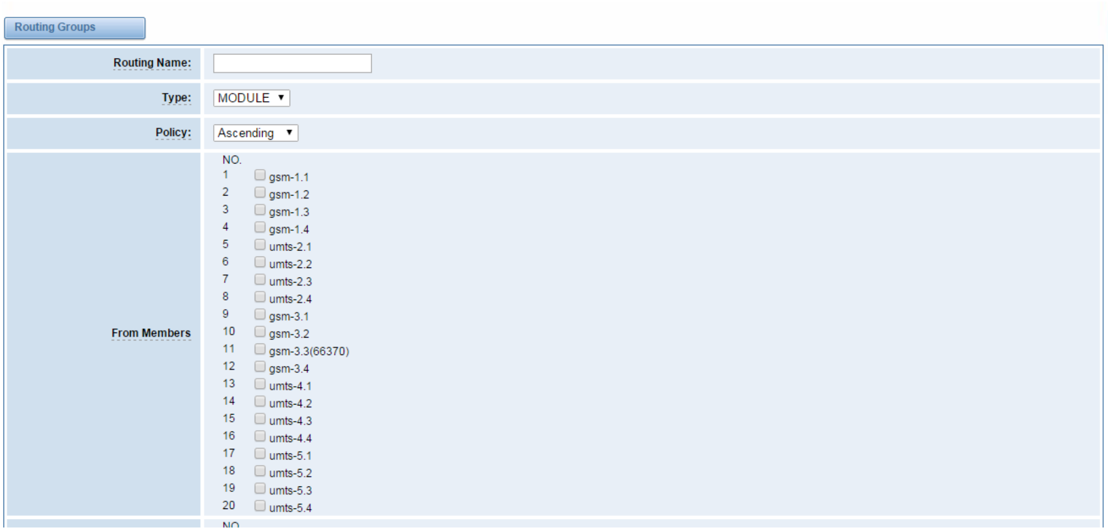


Figure 6-10 Create a routing

For "ascending" Policy, if you choose 2 or more ports members, it will use first available port to transfer sms. For this case, if umts-2.1 is availble, it will always use umts-2.1 to trnasfer sms; Otherwise, it will use gsm-3.1 to transfer sms.

7. Network

On “Network” page, there are three sub-pages: “LAN Settings”, “DDNS Settings”, and “Toolkit”.

Network Settings

There are three types of WAN/LAN port IP, Factory, Static and DHCP. Factory is the default type, and it is 172.16.99.1. When you Choose LAN IPv4 type is “Factory”, this page is not editable.

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.

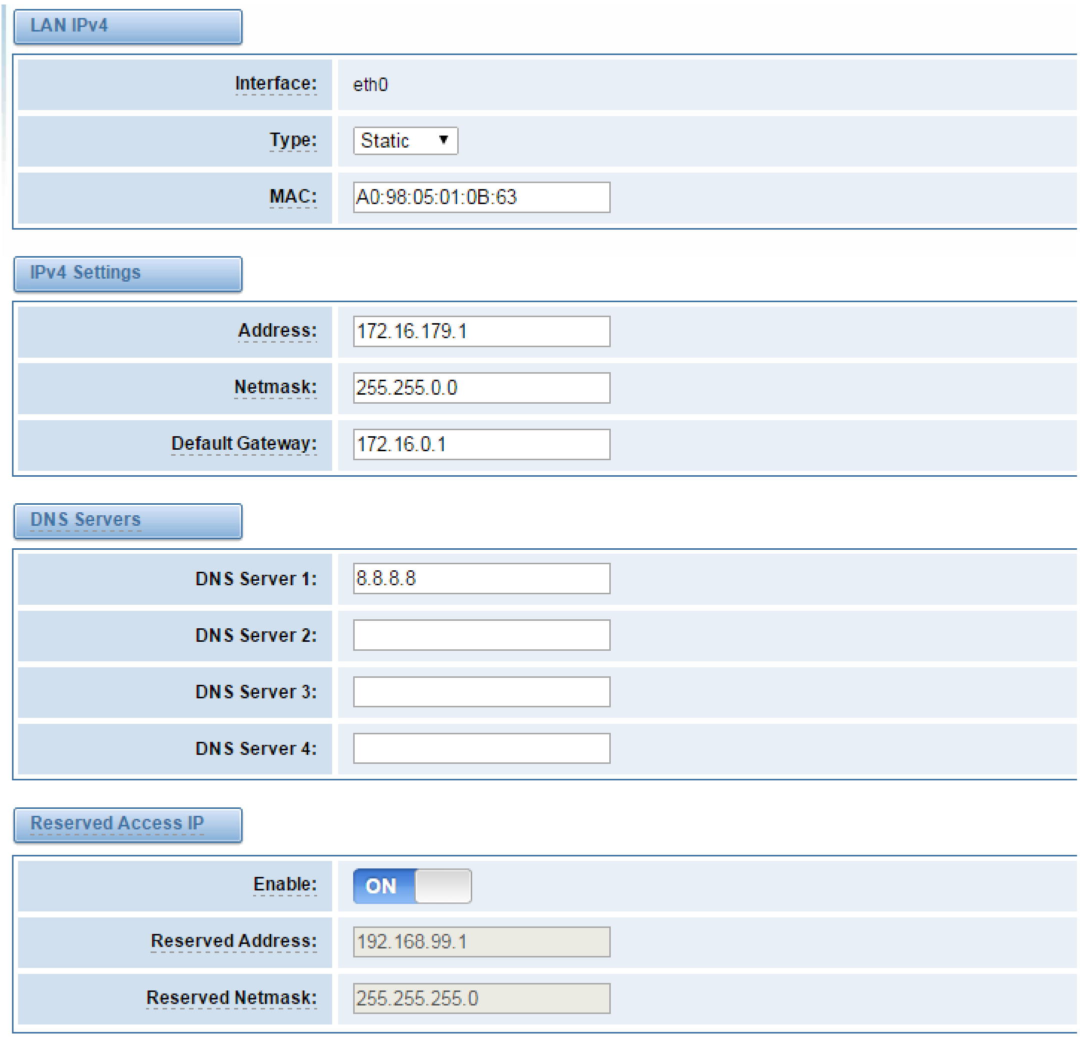


Figure 7-1 LAN Settings

Table7-1 Definition of LAN Settings

Options	Definition
Interface	The name of network interface.
Type	The method to get IP. **Factory**: Getting IP address by Slot Number (System information to check slot number). **Static**: manually set up your gateway IP. **DHCP**: automatically get IP from your local LAN.
MAC	Physical address of your network interface.
Address	The IP address of your gateway.
Netmsk	The subnet mask of your gateway.
Default Gateway	Default getaway IP address.

Basically this info is from your local network service provider, and you can fill in four DNS servers.

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

DDNS Settings

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

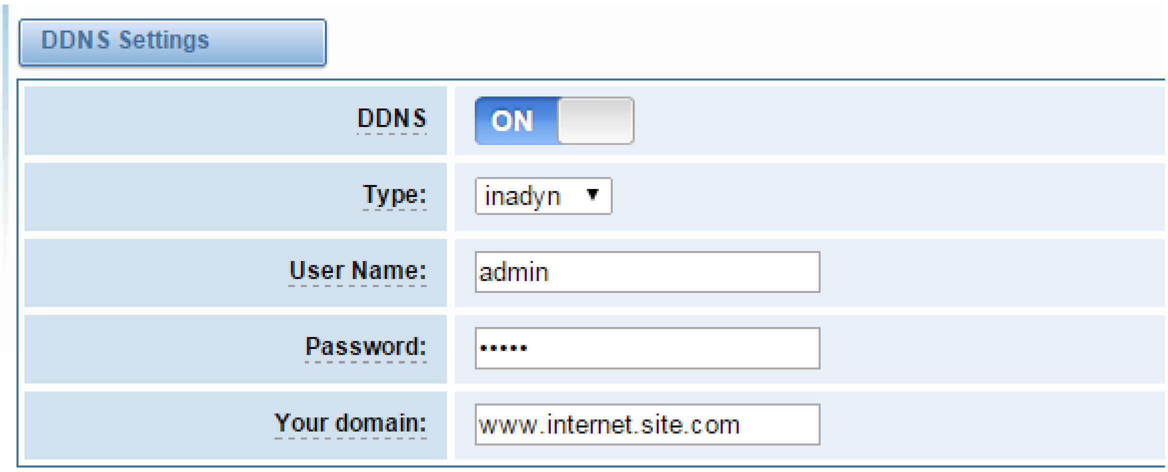


Figure 7-2 DDNS Settings

Table7-2 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.

Toolkit

It is used to check network connectivity. Support Ping command on web GUI.

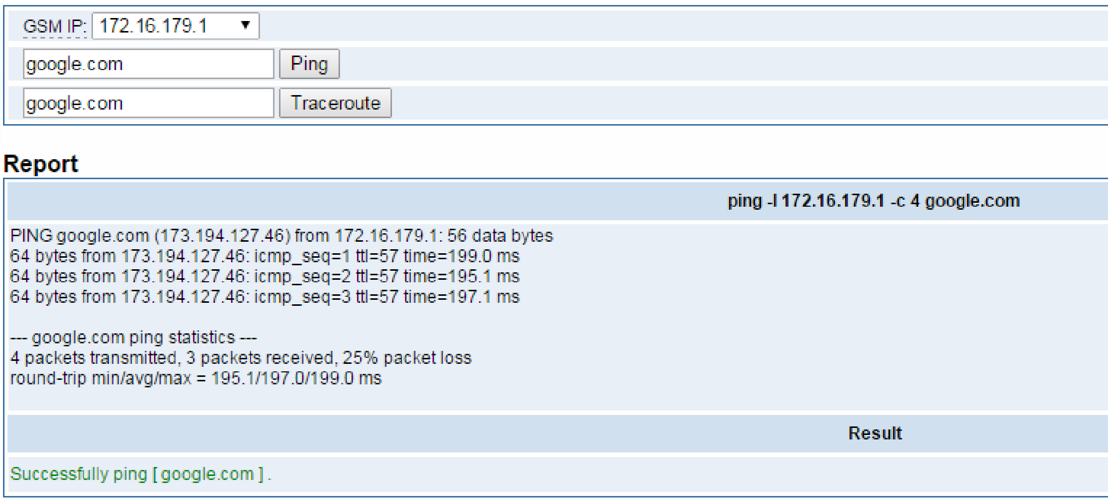


Figure 7-3 Toolkit

Security Settings

Firewall settings

Table7-3 Definition of Firewall settings

Options	Definition
Firewall Enale	If you want to use White/Black List, and security rules, you must enble this option.

Ping Enable	To disable ping or not. OFF: disable ping. This gateway will not allow to ping.

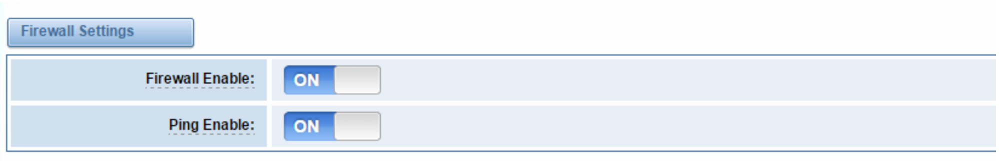
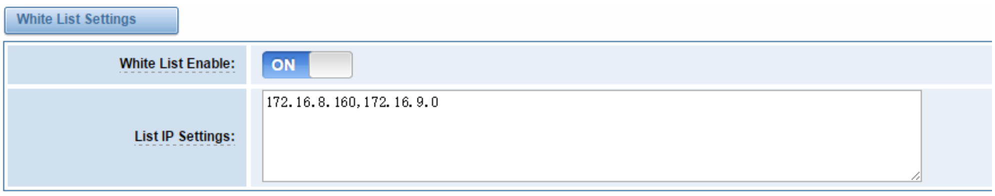


Figure 7-4 Firewall Setting

White List Settings

- White List Enbale: To enable white list or not.
- List IP Settings: IPs are separated only by "," character.



Click "Save" button to save configration; 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.

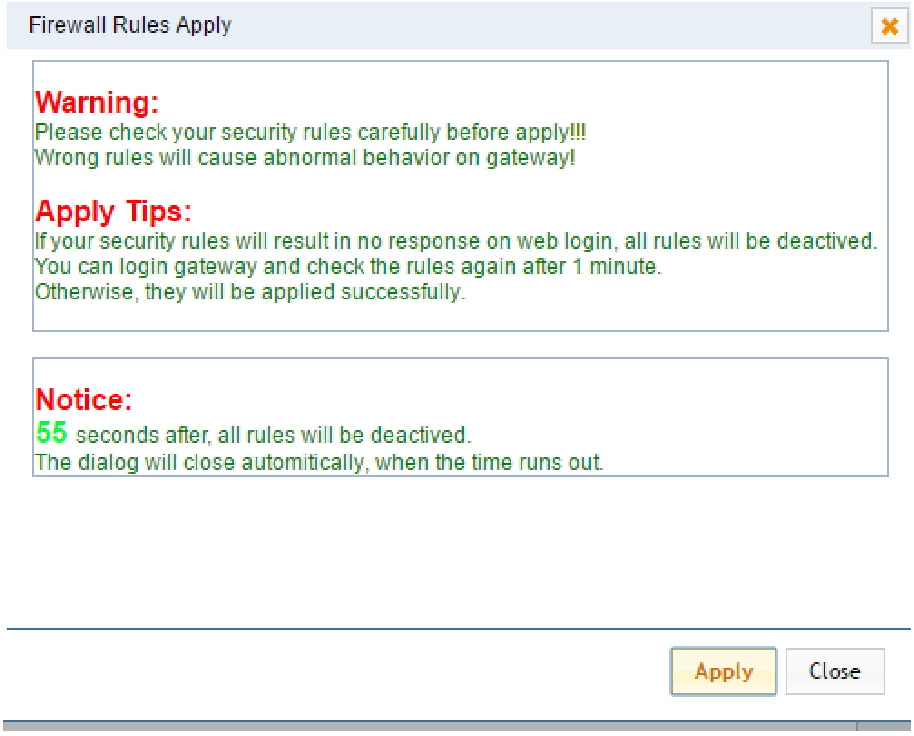


Figure 7-5 Firewall Rules Apply

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

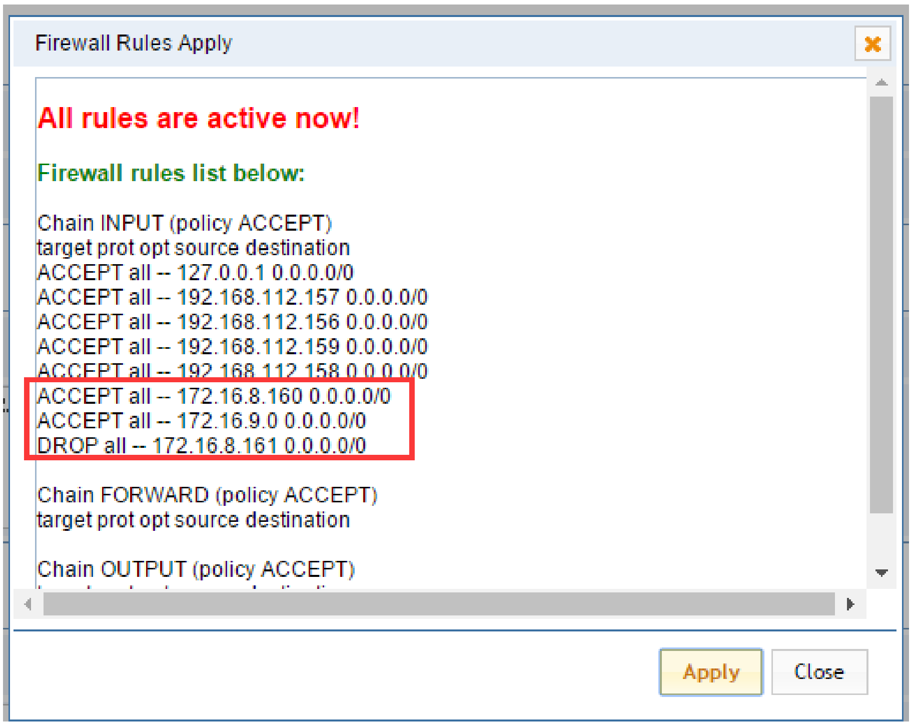
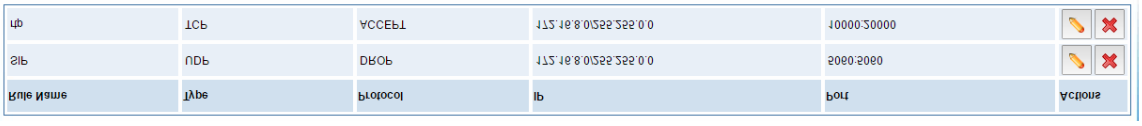


Figure 7-6 Firewall Rules Apply

Security Rules

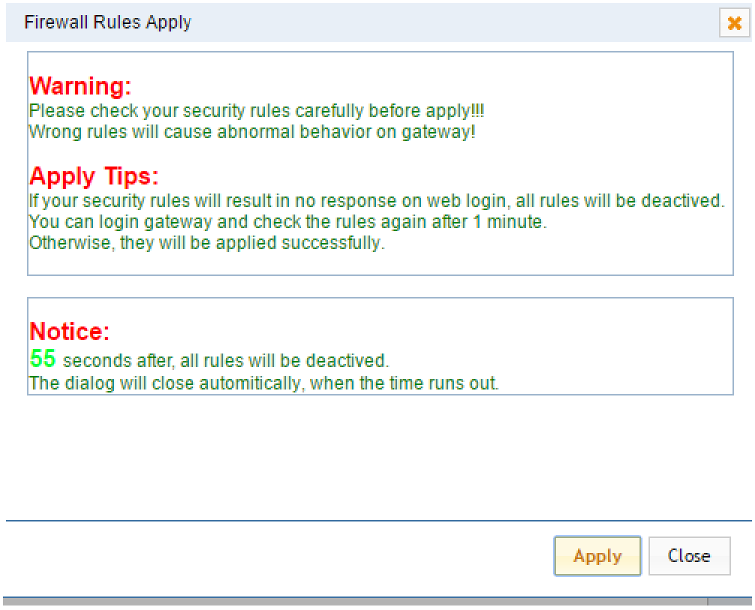


Rule Name	Type	Protocol	Ip	Port	Actions
sip	UDP	DROP	172.16.8.0/255.255.0.0	5060:5060	
rtp	TCP	ACCEPT	172.16.8.0/255.255.0.0	10000:20000	

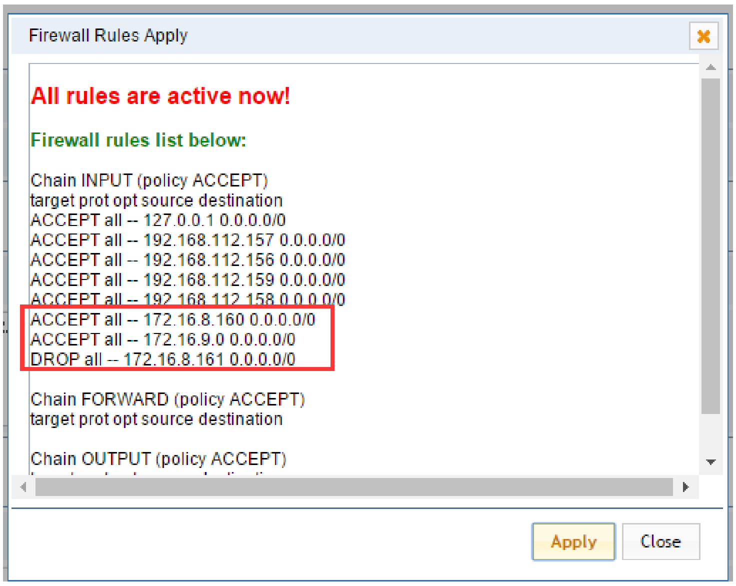
Figure 7-7 Security Rules

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.



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



8. Advanced

Asterisk API

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

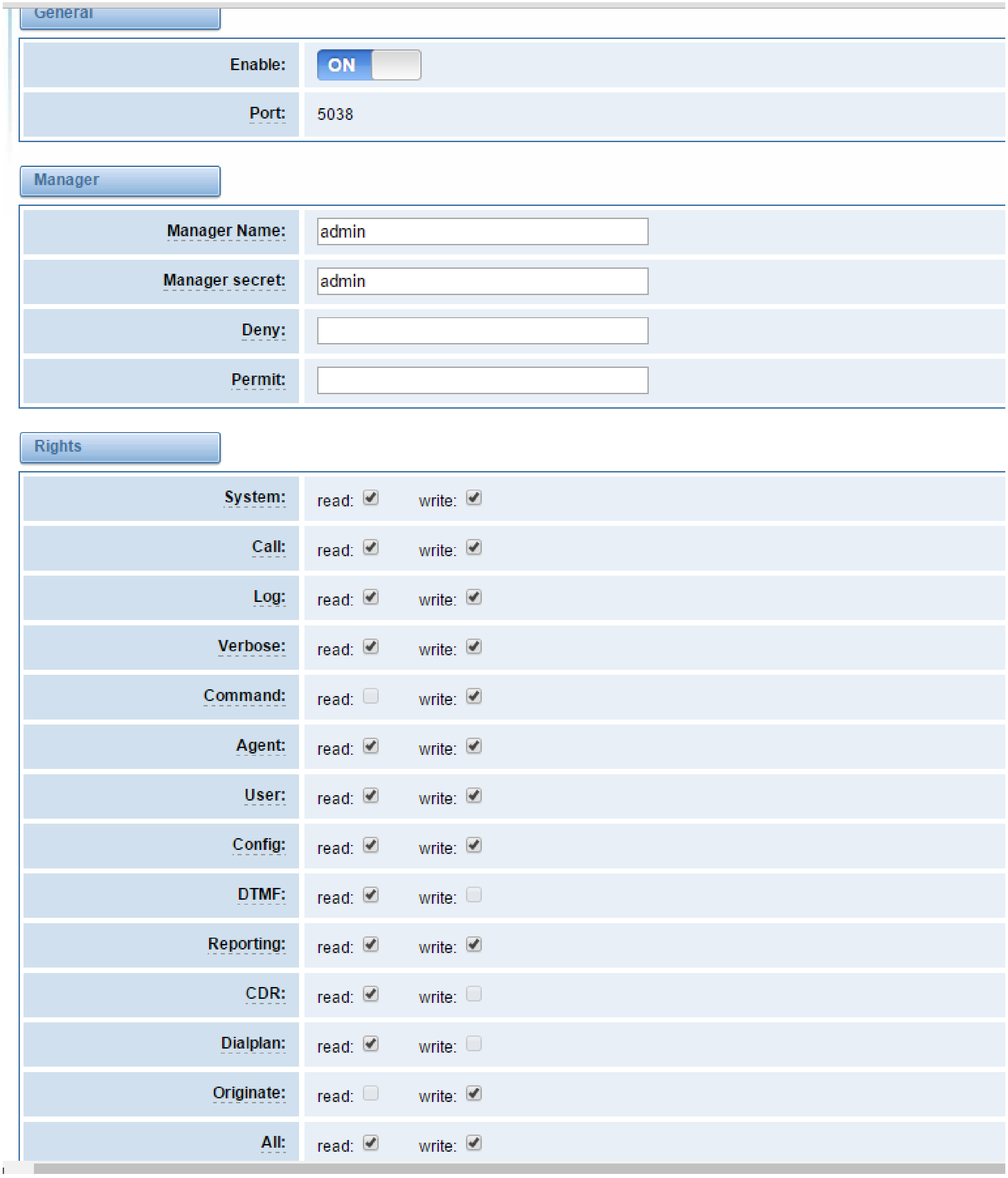


Figure 8-1 Asterisk API Interface

Table8-1 Definition of Asterisk API

Options	Definition

Port	Network port number
Manager Name	Name of the manager without space
Manager secret	Password for the manager. Characters: Allowed characters “-_+.<>&0-9a-zA-Z”. Length:4-32 characters.
Deny	If you want to deny many hosts or networks, use char & as separator. Example: 0.0.0.0/0.0.0.0 or 192.168.1.0/255.255.255.0&10.0.0.0/255.0.0.0
Permit	If you want to permit many hosts or network, use char & as separator. Example: 0.0.0.0/0.0.0.0 or 192.168.1.0/255.255.255.0&10.0.0.0/255.0.0.0
System	General information about the system and ability to run system management commands, such as Shutdown, Restart, and Reload.
Call	Information about channels and ability to set information in a running channel.
Log	Logging information. Read-only. (Defined but not yet used.)
Verbose	Verbose information. Read-only. (Defined but not yet used.)
Command	Permission to run CLI commands. Write-only.
Agent	Information about queues and agents and ability to add queue members to a queue.
User	Permission to send and receive UserEvent.
Config	Ability to read and write configuration files.
DTMF	Receive DTMF events. Read-only.
Reporting	Ability to get information about the system. CDR Output of cdr, manager, if loaded. Read-only.
Dialplan	Receive NewExten and Varset events. Read-only.
Originate	Permission to originate new calls. Write-only.
All	Select all or deselect all.

Once you set like the above figure, the host 172.16.100.110/255.255.0.0 is allowed to access the gateway API. Please refer to the following figure to access the gateway API by putty. 172.16.179.1 is the gateway's IP, and 5038 is its API port.

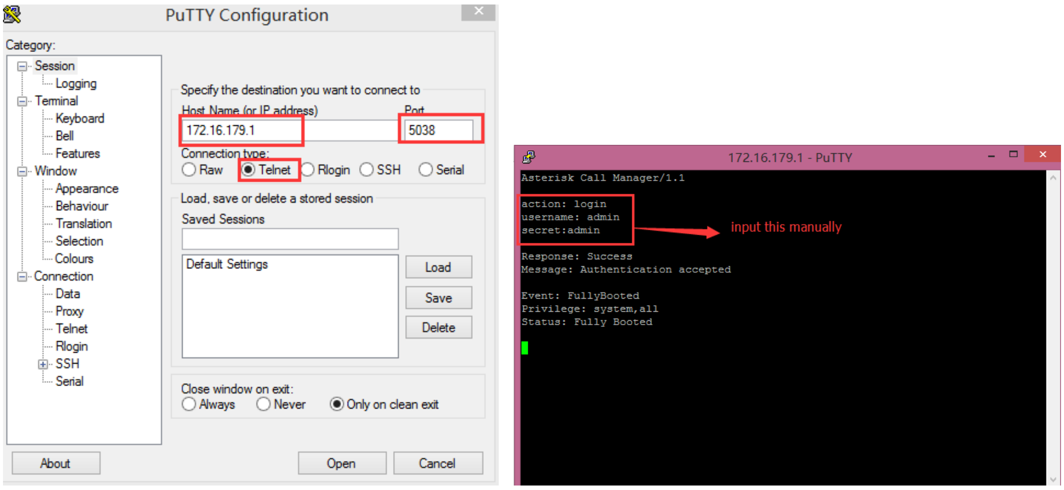


Figure 8-2 Putty Access Gateway API

Asterisk CLI

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

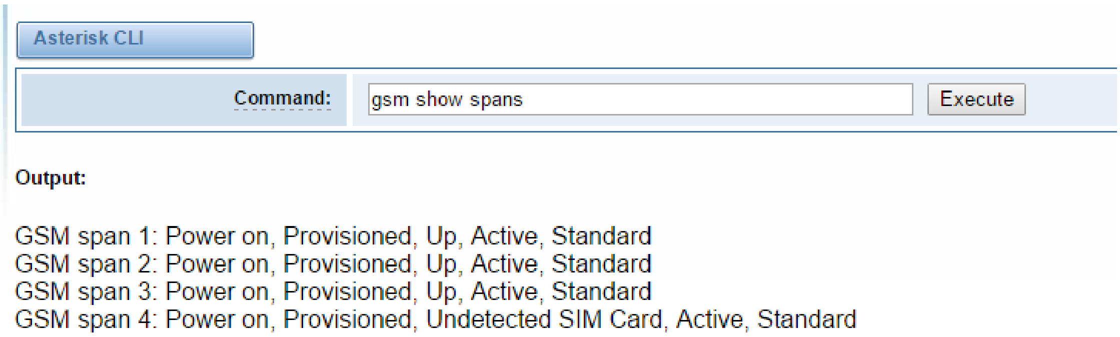


Figure 8-3 Asterisk CLI

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 executable commands.

Asterisk File Editor

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

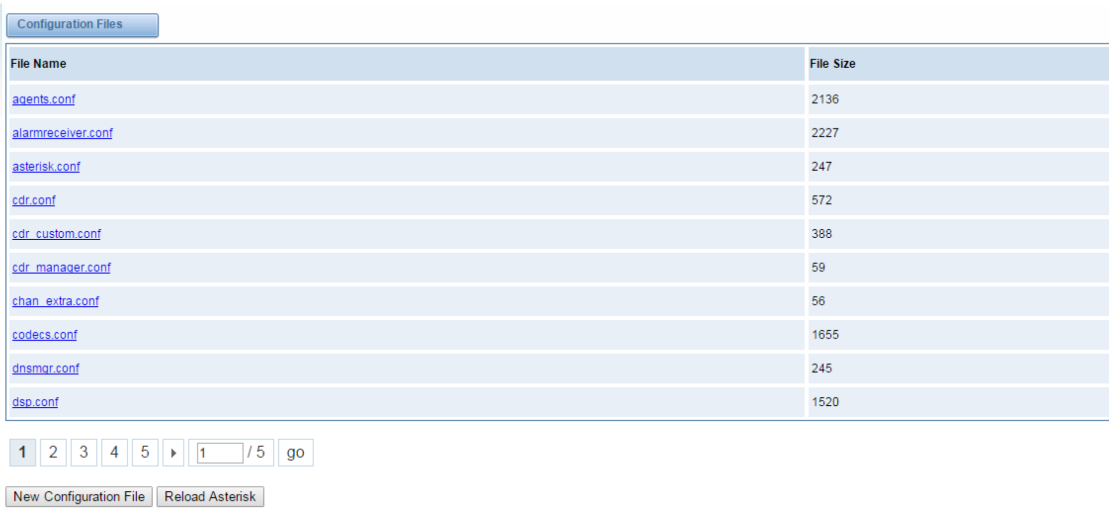


Figure 8-4 Asterisk File Editor

Click “New Configuration File” to create a new configuration file. After editing or creating, please reload Asterisk.

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 is unavailable. And the same with other log pages.

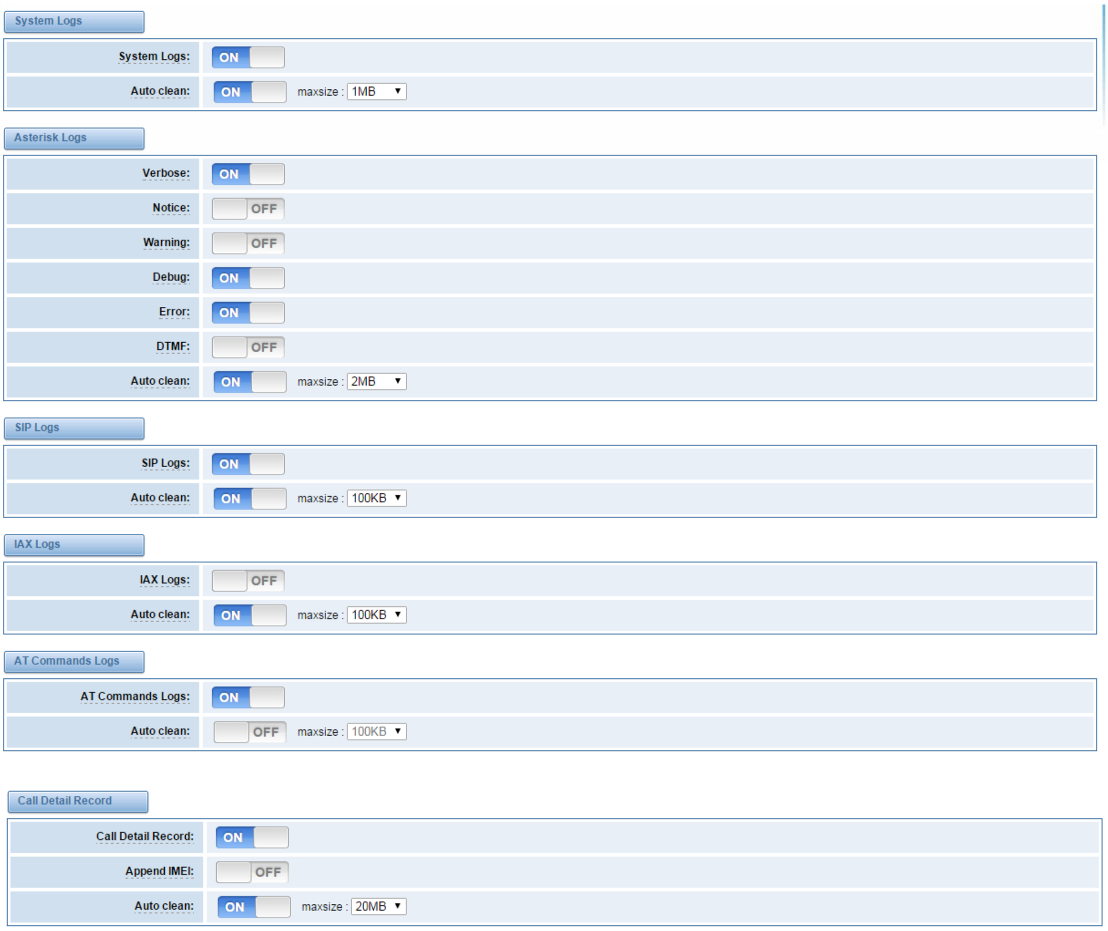


Figure 9-1 Log Settings

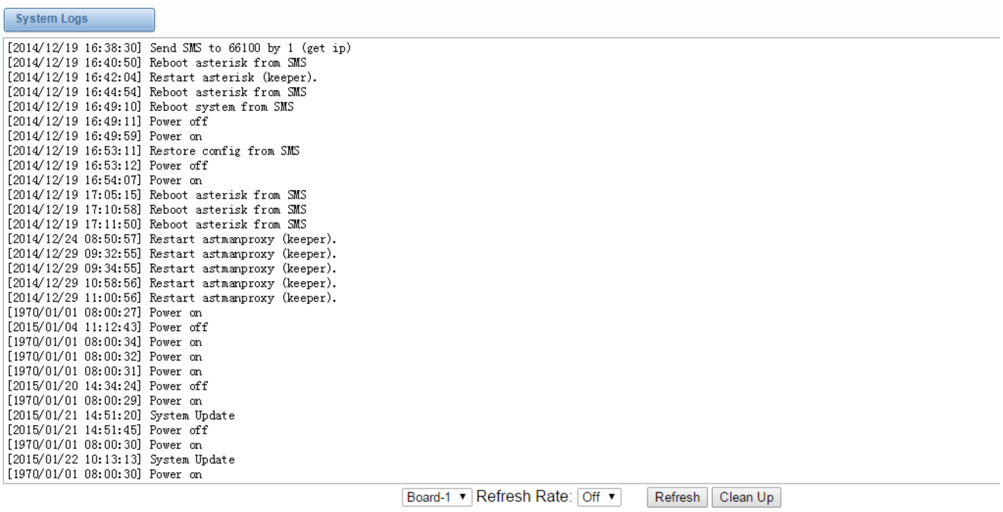


Figure 9-2 System Logs

You can choose one specific board to see it related logs.

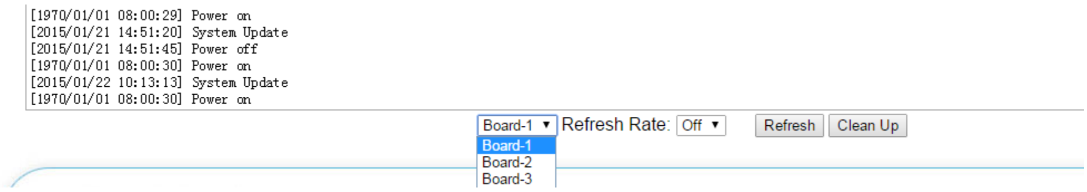
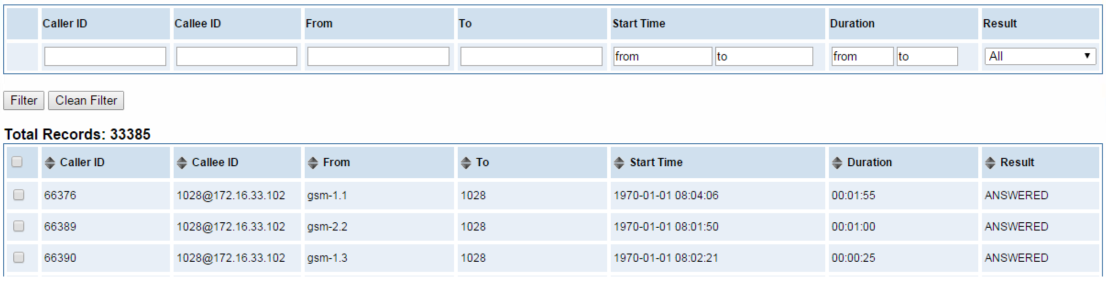


Figure 9-3 Choose One Board

You can scan your CDR easily on web GUI, and also you can delete, clean up or export your CDR information.



	Caller ID	Callee ID	From	To	Start Time	Duration	Result
					from to	from to	All

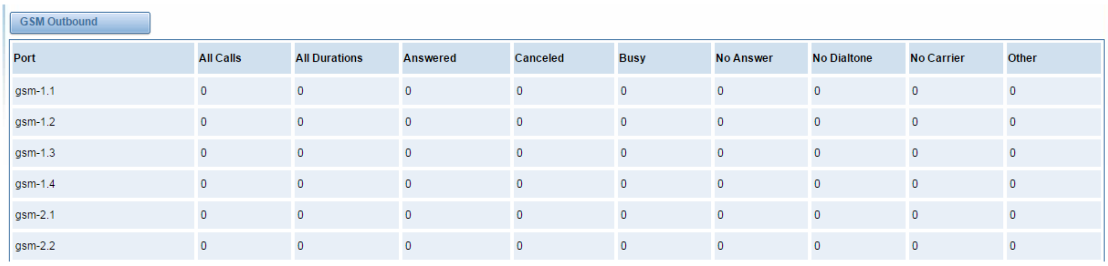
Filter Clean Filter

Total Records: 33385

	Caller ID	Callee ID	From	To	Start Time	Duration	Result
	66376	1028@172.16.33.102	gsm-1.1	1028	1970-01-01 08:04:06	00:01:55	ANSWERED
	66389	1028@172.16.33.102	gsm-2.2	1028	1970-01-01 08:01:50	00:01:00	ANSWERED
	66390	1028@172.16.33.102	gsm-1.3	1028	1970-01-01 08:02:21	00:00:25	ANSWERED

Figure 9-4 CDR Output

Recently we have made our LOGS display richer, you can see your GSM Outbound of every port clearly.



GSM Outbound

Port	All Calls	All Durations	Answered	Canceled	Busy	No Answer	No Dialtone	No Carrier	Other
gsm-1.1	0	0	0	0	0	0	0	0	0
gsm-1.2	0	0	0	0	0	0	0	0	0
gsm-1.3	0	0	0	0	0	0	0	0	0
gsm-1.4	0	0	0	0	0	0	0	0	0
gsm-2.1	0	0	0	0	0	0	0	0	0
gsm-2.2	0	0	0	0	0	0	0	0	0

Figure 9-5 Time Patterns that will use this Route

Table9-1 definition of Logs

Options	**Definition**
System Logs	Whether enable or disable system log.
Auto clean (System Logs)	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained; **switch off** : logs will remain, and the file size will increase gradually. default on, maxsize=1M.
Verbose	Asterisk console verbose message switch.

Notice	Asterisk console notice message switch.
Warning	Asterisk console warning message switch.
Debug	Asterisk console debug message switch.
Error	Asterisk console error message switch.
DTMF	Asterisk console DTMF info switch.
Auto clean (asterisk logs)	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **switch off** : logs will remain, and the file size will increase gradually. default on, maxsize=100KB.
SIP Logs	Whether enable or disable SIP log.
Auto clean (SIP logs)	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **switch off** : logs will remain, and the file size will increase gradually. default on, maxsize=100KB.
IAX Logs	Whether enable or disable IAX log.
Auto clean(IAX logs)	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **switch off** : logs will remain, and the file size will increase gradually. default on, maxsize=100KB.
Debug AT Command Logs	Displaying GSm module AT messages.
Auto clean (AT logs)	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **switch off** : logs will remain, and the file size will increase gradually. default on, maxsize=100KB.
Call Detail Record	Displaying Call Detail Records for each channel. =
Auto clean (CDR logs)	**switch on** : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. **switch off** : logs will remain, and the file size will increase gradually. default on, max size=20MB.

Appendix Feature List

General Info

- Size: VS -GW21(GSM) 48.3cm*33.1cm*8.8cm
  VS-GW1600(GSM) 44cm*4.5cm*30cm
  VS-GW1202(GSM) 15cm*19cm*4.5cm
  WGW1002G 16cm*10.1cm*3.1cm
  VS-GW2120(3G) 44cm*34cm*9cm
  VS-GW1600(3G) 44cm*4.5cm*30cm
  VS-GW1202(3G) 15cm*19cm*4.5cm
- VS-GW1002: 2 GSM channels
  VS-GW1202: 4/8 GSM/UMTS channels
  VS-GW1600: up to 20 GSM/UMTS channels
  VS-GW2120: up to 44 GSM/UMTS channels
- Power: VS-GW2120 88W
  VS-GW1600 46W
  VS-GW1202 18W
  WGW1002G 6W
- Weight: VS-GW2120(GSM) 8624g
  VS-GW1600(GSM) 4301g
  VS-GW1202(GSM) 1300g
  WGW1002G 237g
  VS-GW2120(3G) 6144g
  VS-GW1600(3G) 3682g
  VS-GW1202(3G) 502g
- LAN Port: 2
- SIM Cards: Hot-Swap
- GWM400G/GWM400W Module: Hot-Swap
- Operation Humidity Range:10%~90% non-condensing
- Storage Temperature Range: -20~70°C
- Operation Temperature Range: 0~40°C

GSM/WCDMA Features

- CLID Display & Hide (Need operators' support)
- WCDMA/UMTS: 850/900/1900/2100 MHz
- GSM: 850/900/1800/1900 MHz
- Real Open API Protocol (based on Asterisk)
- Call Duration Limitation
- SMSC/SMS/USSD
- Gain Adjustment

- PIN Identification
- IMEI Number Automatically Modify
- Band Binding
- Bind Carrier
- Optional GSM/UMTS Voice Codec
- Call Waiting
- Call Forwarding (unconditional, no reply, busy, not reachable)
- GSM/UMTS Ports Group Management
- SMS Bulk Transceiver, Sent to Email and Automatically Resend
- SMS Coding/Detecting Automatically Identification
- SMS Remotely Controlling Gateway
- SMS Forwarding and Quick Reply
- USSD transceiver

VOIP Characters

- Support SIP, IAX2 Protocol
- Add, Modify & Delete SIP/IAX2 Trunk
- SIP/IAX2 Registration with Domain
- Combine Different SIP/IAX2 Trunk into Group
- DTMF Mode: RFC2833/Inband/SIPInfo
- SIP V2.0 RFC3261 Compliance
- Multiple SIP/IAX2 Registrations modes: None (No registration, just IP and Password authenication)Endpoint registers with this gateway (work as a SIP Sever)This gateway registers with the endpoint (work as a SIP/IAX2 client)

Network

- IPv4, UDP/TCP, DHCP, TFTP, TELNET, HTTP/HTTPS, SMTP, POP3
- HTTP/SSH (Optical Telnet)
- Ping & Traceroute Command on the Web
- Two Types of IP Access
- Simple Security Strategy: white list, black list, security rules

System Features

- Abundant Codecs:G.711A, G.711U, G.729, G.722, G.723, G.726, GSM
- Simple and convenient configuration via Web GUI
- Firmware Update by HTTP
- Automatically Reboot
- Extensible Automatic Callback and Speed Dial
- TTL Serial Port and Virtual Serial via TCP/IP Protocol
- Support DISA
- Customizable IVR
- Multiple Detailed LOG Output
- Call Status Display
- PDD/ACD/ASR/BER Display
- Mobile number portability (MNP)
- CDR (More than 200,000 Lines CDRs Storage Locally)
- Support configuration files backup and upload
- Support for custom scripts, dialplans
- Least Cost Routing(LCR),according to Time, Port, Calling Number
- Independent System for Each Module
- Restore Factory Settings
- High Equipment Materials Specifications, Suitable for Long Distance Transportation

Application Diagrams