





# **OpenVox Communication Co Ltd**



# VS-GWP1600/2120 Gateway User Manual

Version 1.0





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### **Revise History**

Version	Release Date	Description
1.0	21/7/2022	Full text



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<u>OpenVox</u> 1. Overview

# 1.1 What is GWP1600/2120?

VS-GWP1600/2120 Series Wireless Gateway which can be compatible with a series of modules(2G/3G/4G), enabling interconnection between GSM/WCDMA/LTE network and VoIP network safely and efficiently.

There are two models with the new VoxStack series Gateway, VS-GWP1600 and VS-GWP2120 Wireless Gateway. The VS-GWP1600 Wireless Gateway can provide up to 5 plug-in modules that could support from 4 to 20 GSM/WCDMA/LTE channels. The VS-GWP2120 Wireless Gateway can provide up to 11 plug-in modules that could support from 4 to 44 GSM/WCDMA/LTE channels. With the unique design of the VS-GWP1600/2120 Wireless Gateway, it can support hot-swap for SIM cards and GSM/WCDMA/LTE wireless gateway modules. The wireless gateway is designed with a LAN switch board that provides stackability on the hardware upgrade. Users can simply add or remove the modules for hardware expansion or exchange.

The VS-GWP1600/2120 Wireless Gateway can bring excellent HD voice service with multiple codecs, including G.711A, G.711U, G.722, G.726, G.729A, GSM, and also flexible SMS service with SMS API. It uses the standard SIP protocol and is compatible with leading IMS/NGN platforms, IPPBX, and SIP servers supporting most of the VoIP operating platforms such as Asterisk, 3CX, FreeSWITCH SIP server, BroadSoft, etc.

With a friendly GUI and unique modular design, users may easily set up their customized gateway. Also, secondary development can be completed through API. It can provide users with more diverse telecommunication access methods and help users reduce communication costs.





# 1.2 Application



#### Figure 1-2-1 Application Topology

Figure 1-2-2 VS-GWP1600 series product



#### Table 1-2-1 VS-GWP1600 Slot Description

2	3	5
1	CSU-F	4



#### Figure 1-2-3 VS-GWP2120 series product



#### Table 1-2-2 VS-GWP2120 Slot Description

4	7	11
3	6	10
2	5	9
1	CSU-F	8

Status Light	Color	Status	
	Green and Flash	Module Initiating	
	Red and Flash	No SIM Card	
Signal Status LED	Always red	Worst Signal Quality	
	Always yellow	Medium Signal Quality	
	Always green	Best Signal Quality	
	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	

#### Table 1-2-3 Status Light Description





# 1.3 Panel

1. CSU (Core Switch Unit)



### ①ETH Port

②Channel indicator and power status indicator

- 3 Console
- Beset Button

#### 2. WTU (Wireless Trunk Unit)



- 1 Operating status indicator
- 2Power status indicator
- 3Reset button
- $\textcircled{4}{\rm SIM} \ {\rm Card} \ {\rm Slot}$
- (5) Antenna
- 6 SIM card working status indicator
- $\textcircled{O}\mathsf{SIM}$  card signal strength indicator



## 1.4 Main Features

- SIP/IAX2/Wireless Group Management
- Random call interval
- Call Duration Limitation
- Open API Protocol
- Multiple SMS API
- SMSC/SMS/USSD/SMPP
- Gain Adjustment
- PIN Identification
- IMEI Number Automatically Modify
- Band Binding
- Bind Carrier
- Call Waiting
- Call Forwarding (unconditional, no reply, busy, not reachable)
- SMS Bulk Transceiver and Auto Resend
- SMS to Email
- SMS Coding/Detecting Automatically Identification
- SMS Forwarding and Quick Reply
- SMS Remotely Controlling Gateway
- CLID Display & Hide (Need operators' support)
- USSD transceiver

### **1.5 Physical Information**

• Weight:

VS-GWP1600: 4.8kg

- VS-GWP2120: 6.5kg
- VS-GWM420G: 162g



VS-GWM420W: 170g

VS-GWM420L: 178g

• Size:

VS-GWP1600: 434\*330\*44mm

VS-GWP2120: 434\*330\*88mm

• Power:

VS-GWP1600: 22W

VS-GWP2120: 50W

- Operation Temperature: 0~50°C
- Storage Temperature: -20~70°C
- Operation humidity: 10% ~ 90% non-condensing
- WAN: 2\*10/100M

## 1.6 Software

- Default IP: 172.16.98.1
- Username: admin
- Passward: admin

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



# 2. System

## 2.1 Status

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

Module Information									
Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	Module Status	Remain Time
cdma-1.1	attl	-1	CHINA TELECOM	Registered (Home network)	1	0	0	READY	No Limit
cdma-1.2(18002548416)	att	-1	CHINA TELECOM	Registered (Home network)	2	16	100	READY	No Limit
cdma-1.3	att	-1	CHINA TELECOM	Registered (Home network)	0	0	0	READY	No Limit
cdma-1.4	att	-1	CHINA TELECOM	Registered (Home network)	2	3	100	READY	No Limit
cdma-1.5	att	-1	CHINA TELECOM	Registered (Home network)	4	28	100	READY	No Limit
cdma-1.6	aill	-1	CHINA TELECOM	Registered (Home network)	2	4	100	READY	No Limit
cdma-1.7	att	-1	CHINA TELECOM	Registered (Home network)	0	0	0	READY	No Limit
cdma-1.8	att	-1	CHINA TELECOM	Registered (Home network)	0	0	0	READY	No Limit
cdma-1.9	attl	-1		Undetected SIM Card	0	0	0		No Limit
cdma-1.10	atil	-1	CHINA TELECOM	Registered (Home network)	0	0	0	READY	No Limit
cdma-1.11	att	-1	CHINA TELECOM	Registered (Home network)	0	0	0	READY	No Limit
cdma-1.12	att	-1		Undetected SIM Card	0	0	0		No Limit
cdma-1.13	attl	-1		Undetected SIM Card	0	0	0		No Limit
cdma-1.14	att	-1	CHINA TELECOM	Registered (Home network)	0	0	0	READY	No Limit
cdma-1.15	all	-1	CHINA TELECOM	Registered (Home network)	0	0	0	READY	No Limit
cdma-1.16	att	-1	CHINA TELECOM	Registered (Home network)	2	10	100	READY	No Limit

#### Figure 2-1 Systm Status

SIP Information						
Endpoint Name	User Name	Host	Registration	SIP Status		
1234	1234	172.16.80.216	server	OK (12 ms)		
8888	8888	172.16.33.102	none	Unmonitored		
9999	9999	172.16.33.102	client	No Authentication		

IAX2 Information						
Endpoint Name	User Name	Host	Registration	IAX2 Status		
1002	1002	172.16.80.216	server	OK (38 ms)		
1003	1003	172.16.33.102	none	OK (104 ms)		
1004	1004	172.16.33.102	client	OK (103 ms)		

Routing Information							
Rule Name	From	То	Rules				
OUT	sip-1234	grp-ALL					
IN	grp-ALL	custom-playback					
Network Information							
Name	MAC Address	IP Address	Mask	Gateway	RX Packets	TX Packets	

255.255.0.0

172.16.0.1

602327

172.16.6.130

00:E0:4C:36:00:35

LAN



Options	Definition
Port	Number of each ports.
Signal	Display the signal strength of in each channels of gateway.
BER	Bit Error Rate.
Carrier	Display the network carrier of current SIM card.
Registration	Indicates the registration status of current module.
Status	
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. ModuleStatus Show the status of port, include blank space and
	"READY". Black space means it is unavailable here and "Ready" means the
	port is available
Module	Display the status of the port. "Ready" means registering and "READY" means
Status	port is available
Remain	This value is multiplied by to step length is a rest call time.
Time	

### Table 2-1 Description of System Status





# 2.2 Time

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].
Save Data	Save the Modify of the time settings
Sync from NTP	Sync time from NTP server.
Sync from Client	Sync time from local machine.

#### Table 2-2 Description of Time Settings

For example, you can configure like this:

#### Figure 2-2 Time Settings

iys	
System Time:	2017-11-3 14:41:00
Time Zone:	Chongqing •
POSIX TZ String:	CST-8
NTP Server 1:	pool.ntp.org
NTP Server 2:	64.236.96.53
NTP Server 3:	time.nist.gov
Auto-Sync from NTP:	ON

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

Save Data Sync from NTP Sync from Client



# 2.3 Login Settings

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. Normally, the default web login mode is "http and https." For security, you can switch to "only https".

Options	Definition
User Name	Define your username and password to manage your gateway
	Allowed characters "+. < >&0-9a-zA-Z". Length: 1-32 characters.
Password	Allowed characters "+. < >&0-9a-zA-Z". Length: 4-32 characters.
Confirm	Please input the same password as 'Password' above.
Password	
Login Mode	http and https: You can access gateway via link: <u>http://gatewayIP</u> or
	https://gatewayIP
	https: You can only access gateway via link: <a href="https://gatewaylP">https://gatewaylP</a>
Port	Specify the web server port number.

#### Table 2-3 Description of Login Settings

For example, you can configure like this:

#### Figure 2-3 Login Settings

Web Login Settings	
User Name:	
Password:	
Confirm Password:	
Login Mode:	http and https 🔻
Port:	80
SSH Login Settings	
Enable:	ON
User Name:	super
Password:	urjwxxfW8tdlYx4hNY3
Port:	12345





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

## 2.4 General

### 2.4.1 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".

For example:

#### Figure 2-4 Language Settings

Language Settings		
Language:	English •	
Advanced:	ON	
Language Debug:	TURN ON TURN OFF	
Download:	Download selected language package.	Download
Delete:	Delete selected language.	Delete
Add New Language:	New language Package/ 选择文件 未选择任何文件	Add

### 2.4.2 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 Type

Enable:	OFF	
Reboot Type:	By Running Time ▼	
Running Time:	Hour: 0 V	

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



### 2.5 Tools and Information

### 2.5.1 Reboot Tools

You can choose system reboot and asterisk reboot separately.

#### Figure 2-6 Reboot Tools

VoxStack	SYSTEM	172.16.6.130 显示: Are you sure to reboot your gateway now? You will lose all data in memory!	× :TWORK	ADVANCED   LOGS
SYSTEM HETANS	Fr	ee Commun I d	ation	OpenVox Solution
Reboot Tools Reboot the gateway and all the current of Reboot the asterisk and all the current of	alls will be dropped.			System Reboot Asterisk Reboot

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

### 2.5.2 Update Firmware

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

If you choose System Online Update, you will see the following information:

#### figure 2-7 Update Firmware

Your current syst	em version is : 1.4.	0		
The latest syste	m version is :2.3.8	В		
3e cautious, ple	ase:			
his might dam	age the structure	of your origin	al configuration files!	
Are you sure to	update your syste	em?		
Warning:				
<mark>Narning:</mark> )0 NOT leave ti vill fail!	his page in the pro	ocess of upda	ting; OTHERWISE syste	em updating
Warning: DO NOT leave ti vill fail!	his page in the pro	ocess of upda	ting; OTHERWISE syste	em updating
Warning: DO NOT leave ti vill fail!	his page in the pro	ocess of upda	ting; OTHERWISE syste	em updating



### 2.5.3 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 2-8 Upload and Backup Configuration

Upload Configuration	
New configuration file: 选择文件 未选择任何文件	File Upload
Backup Configuration	
Current configuration file version: 1.4.0	Download Backup

### 2.5.4 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 2-9 Restore Configuration



### 2.6 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 help information.

#### Figure 2-10 Information



Model Name:	SWG-1016
Modem Description:	800MHz@CDMA 2000
Software Version:	1.4.0
Hardware Version:	1.0
Slot Number:	1
Storage Usage:	516.0K/487.9M (0%)
Memory Usage:	25 884 % Memory Clean
Build Time:	2017-11-07 10:53:01
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:	51
System Time:	2017-11-7 13:56:25
System Uptime:	0 days 01.29.04

### 2.7 User

On the "User" page, webpage accounts can be added via admin user. You can add different accounts with different rights.

### Figure 2-11 Information

Vuser 🗸 🗸 🗸 Vuser				
Username:				
Password:				
Confirm Password:				
Number of logged-in IPS:	1 •			
	All SYSTEM Only View General Status Setting Wizard MODULE Only View	Only View Only View Time Only View Tools Only View	Only View Login Settings Only View Information	Only View Only View
	Module Settings     Simbank     Toolkit	Only View Call Forwarding Only View DTMF Only View Module Update	Only View     Call Waiting       Only View     BCCH       Only View     Call And SMS Limit	Only View Only View Only View Only View
	VOIP Only View	Only View Batch SIP Endpoints Only View Sip Account Security	Only View Advanced SIP Settings Only View	Only View



# 3. MODULE

# **3.1 MODULE Settings**

Port	Carrier	Registration Status	Module Status	Action	ns
cdma-1.1	CHINA TELECOM	Registered (Home network)	READY	2	Ø
cdma-1.2(18002548416)	CHINA TELECOM	Registered (Home network)	READY	0	ø
cdma-1.3	CHINA TELECOM	Registered (Home network)	READY	0	ø
cdma-1.4	CHINA TELECOM	Registered (Home network)	READY	0	ø
cdma-1.5	CHINA TELECOM	Registered (Home network)	READY	2	ø
cdma-1.6	CHINA TELECOM	Registered (Home network)	READY	0	ø
cdma-1.7	CHINA TELECOM	Registered (Home network)	READY	0	ø
cdma-1.8	CHINA TELECOM	Registered (Home network)	READY	2	Ø
cdma-1.9		Undetected SIM Card		2	Ø
cdma-1.10	CHINA TELECOM	Registered (Home network)	READY	0	Ø
cdma-1.11	CHINA TELECOM	Registered (Home network)	READY	2	ø
cdma-1.12		Undetected SIM Card		0	Ø
cdma-1.13		Undetected SIM Card		2	Ø
cdma-1.14	CHINA TELECOM	Registered (Home network)	READY	0	ø
cdma-1.15	CHINA TELECOM	Registered (Home network)	READY	0	Ø
cdma-1.16	CHINA TELECOM	Registered (Home network)	READY	0	Ø

#### Figure 3-1 Module Settings

On this page, you can see your SIM Card information and module status, click action

button

to configure the port.

Port cdma-1.1	
Name:	
Speaker Volume:	50
Microphone Volume:	8
Dial Prefix:	
Pin Code:	. On
Custom AT commands when start:	
CLIR:	OFF
SIM IMSI:	460030237498156
Module IMEI:	0x00A10000530808B9
Module Revision:	+CGMR: 4394B06SIM6320C
Carrier:	CHINA TELECOM
Signal:	21
BER:	-1
Status:	READY

### Figure 3-2 Port Configuration



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

Figure 3-3 PIN Code Application							
Pin Code:	1234	I On					

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 3-4 CLIR Application

17
ON

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 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.
Dial Prefix	The prefix number of outgoing calls from this channel
PIN Code	Personal identification numbers of SIM card. PIN code can
	be modified to prevent SIM card from being stolen.
Custom AT commads	User custom AT commands when start system, use " " to
when start	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.

#### **Table 3-1 Definition of Module Settings**



Module IMEI

Only CDMA module does not support modifying IMEI

### **3.1.1 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 3-5 Single Settings

Call Duration Limit Settings			
Step:	60	Second	
Enable Single Call Duration Limit:	ON		
Single Call Duration Limitation:	2		

**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.



Can Duration Linit Settings			
Step:	60	Second	
nable Single Call Duration Limit:	OFF		
Enable Call Duration Limitation:	ON		
Call Duration Limitation:	20		
Minimum Charging Time:	10	Second	
Alarm Threshold:	3		
Alarm Phone Number:	1860000000		
Alarm Description:	test call limit		
Remain Time:	20	Reset	
Enable Auto Reset:	OFF		



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 3-7 Auto Reset Settings** 

Enable Auto Reset:	ON
Auto Reset Type:	Day(1Day) 🔻
Next Reset Time:	2017-11-03 00:00:00

Options	Definition
Step	Step length value range is 1-999s, step length multiplied by
	time of single call just said a single call duration time allowed.
Enable Single Call	Definite maximum call duration for single call. Example: if Time
Duration Limit	of single call set to 10, the call will be disconnected after
	talking 10*step seconds.
Enable Call	This function is to limit the total call duration of channel. The
Duration Limitation	max call duration is between 1 to 999999 minutes.
Minimum Charging	A single call over this time, Module side of the operators began
Time	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.

#### Table 3-2 Description of Call Duration Limit Settings



Alarm Phone	Receiving alarm phone number, user will received alarm
Number	message from gateway.
Enable Auto Reset	Automatic restore remaining talk time, that is, get total call
	minutes of each 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 3-8 Save to Other Ports

V Save To Other Ports				
Save To Other Ports:	cdma-1.1 cdma-1.5 cdma-1.9 cdma-1.13 All	<pre>cdma-1.2(18002548416) cdma-1.6</pre>	⊂ cdma-1.3 ⊂ cdma-1.7 ⊂ cdma-1.11 ⊄ cdma-1.15	cdma-1.4 cdma-1.8 cdma-1.12 cdma-1.16
Sync All Settings:	Select all settings			

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.

you set.

Your calling status will show on the main interface.

Module Information									
Port	Signal	BER	Carrier	Registration Status	PDD(s)	ACD(s)	ASR(%)	Module Status	Remain Time
:dma-1.1	att	-1	CHINA TELECOM	Registered (Home network)	1	0	0	READY	No Limit
Model IMEI: 0x00A1000053080889 Network Name: CHINA TELECOM Network Status: Registered (Home network) Sional Quality (0.31): 24			CHINA TELECOM	Registered (Home network)	2	16	100	READY	No Limit
		ork)	CHINA TELECOM	Registered (Home network)	0	0	0	READY	No Limit
BER value (0,7): -1 SIM IMSI: 46003023749	18156		CHINA TELECOM	Registered (Home network)	2	3	100	READY	No Limit
SIM SMS Center Number: Own Number: Remain Time: No Limit PDD(s): 1 ACD(s): 0			CHINA TELECOM	Registered (Home network)	4	28	100	READY	No Limit
			CHINA TELECOM	Registered (Home network)	2	4	100	READY	No Limit
ASR(%): 0 State: READY			CHINA TELECOM	Registered (Home network)	0	0	0	READY	No Limit

#### **Figure 3-9 Module Information**



# 3.2 DTMF

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

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	

### Figure 3-10 DTMF Detection Settings

Save

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

Options	Definition
DTMF Normal	It is the difference in power between the row and column
Twist and Reverse	energies. Normal Twist is where the Column energy is greater
Twist	than the Row energy. Reverse Twist is where the Row energy is
	greater.
DTMF Relative	The value is the smaller and the detection is easier. If you lost
Peak Row	some numbers, you can try to put the value down. The
	adjustment range is 0.02 at a time.
DTMF Relative	The value is smaller and the detection is easier. If you lost some
Peak Col	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.

#### Table 3-3 Description of DTMF Detection Settings





# 3.3 Toolkit

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

#### **Figure 3-11 Function Options**

Functi	on: Get USSD V	
Acti	on: Send AT Command Check Number	Copy to Selected Clear All Execute
Port	Input	Output
cdma-1.1		

Options	Definition
Check	Enter a known number (like your mobile phone) to check what
Number	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 modem. e.g. perform [ AT+CSQ ] to check what signal
	qualify it is. In AT commands, there is no difference between "a" and
	"A"

#### Table 3-4 Description of Definition of Functions

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 3-12 AT Command Example



	Function: Send AT Command V			
Action: AT+CSQ		AT+CSQ	Copy to Selected Clear All Execute	
	Port	Input	1	Output
	cdma-1.1	AT+	230	+CSQ: 19.99
	cdma-1.2(18002548416)	AT+0	// 590	HCSQ: 20,99
	cdma-1.3	AT+CSQ		+CSQ: 21,99 0K
	cdma-1.4	AT+0		oK
٠	cdma-1.5	AT+0	29Q	+CSQ: 25,99 0K
	cdma-1.6	AT+C SQ		+CSQ: 23,99 OK
	cdma-1.7	AT+0	25Q //	+CSQ: 22,99 0K
	cdma-1.8	AT+0	59Q //	+CSQ: 22,99 0K
	cdma-1.9	AT+	CSQ //	+CSQ: 16,99 0K
	cdma-1.10	AT+0	CSQ //	+CSQ: 13,99 0K
	cdma-1.11	AT+0	CSQ //	+CSQ: 21,99 0K
	cdma-1.12	AT+0	C9Q	+CSQ: 16,99 0K
	cdma-1.13	AT+0	C3Q //	+CSQ: 22,99 0K
	cdma-1.14	AT+0	C3Q //	+CSQ: 22,99 OK
	cdma-1.15	AT+0	CSQ //	+CSQ: 23,99 OK
	cdma-1.16	AT+0	592	+CSQ: 22,99 0K



## 4.1 Switch

This page displays the status information of each card slot, you can see the corresponding status of each card slot.



#### Figure 4-1 Status of each card slot

Turn on the card strategy switch and set the card switching strategy. You can switch cards in ascending or descending order according to the set sim card registration time, use time, outgoing time, outgoing times, and SMS sending times.

Suggetty (new)		
Switch:	ON	
Sim Policy:	Asc 🗸	
Registration Time:	120	Second
Using Time:	0	Minute
Callout Time:	0	Minute
Callout Count:	0	
SMS Count:	0	

#### Figure 4-2 Strategy of switching sim cards among four cards



# 4.2 Limit

### 4.2.1 Call Limit Times

You can limit the number of daily calls, daily connections calls and hourly calls of the selected channel.

#### Figure 4-3 call limit times

V IAX2 Encryption	
Encryption:	No 🔻
Force Encryption:	No 🔻

### 4.2.2 Call limit time

We provide two types of call time limits, "single call time limit" and "channel total call time limit". You can choose one to control your call time. The call time limit set here will be applied to each call. First, you need to turn on the "call duration limit" switch, then you can set "single length" and "single call duration limit". When you make a call through this port and call duration is equal to "single length" × "single call duration limit", the sim card will be limited to make any calls.

If you set "single call duration", the system will hang up the call when the call time exceeds the set value.

Call Time Limit Switch:	ON		
Step:	60	Second	
Enable Single Call Duration Limit:	ON		
Single Call Duration Limitation:	20		

#### Figure 4-4 call limit time



# 4.3 Lock

The card lock detection switch is a switch for the card lock function. After it is turned on, you need to set the lock card condition parameter. After the card lock condition is reached, the sim card will be disabled and cannot be allocated for use, unless the card is removed and inserted, the gateway is restarted, and the card is manually unlocked (duration time restrictions need to be reset manually), turn off the card lock function operations.

Lock Sim	
Lock Detect Switch:	ON
Mark Switch:	ON
Call Failed Mark Count:	2
Call Failed Lock Switch:	ON
Call Failed Lock Count:	3
SMS Send Detection Switch:	ON
SMS Send Detection Count:	1
Send Sms Number:	10086
Sms Message:	уе
Testing SMS report:	ON

Figure 4-8 Lock sim card

Table 4-9 Instructions of locking sim card
--------------------------------------------

Options	Definition
Call Failed Mark Count	when call failed times reaches the value, the sim card will
	be marked
Call Failed Lock Count	when call failed times reaches the value, the sim card will
	be locked and you can not call successfully via this card.
SMS Send Detection	when it's enable and call failed times reaches value set,
Switch	gateway will send SMS automatically to detect if the sim
	card is available. If SMS is sent successfully, then gateway
	will set call failed time to 0. Unless, the sim card will be
	locked.



Testing SMS report	when it's disable and SMS be sent successfully, it indicates
	the sim card is available; When it's enable, it indicates the
	sim card is available when SMS be sent successfully or
	receiving SMS report.

# 4.4 SMS Limit

SMS Limit Switch	ON
SMS Limit Success Flag	ON
Day Limit SMS Count	10
Month Limit SMS Count	300
SMS Clean Date	

#### Figure 4-10 SMS limit

#### Table 4-11 Instructions of SMS limit

Options	Definition
SMS Limit Success Flag	When it's closed, no matter SMS is sent successfully or not,
	the SMS will be counted. When it's opened, only when SMS
	is sent successfully, the SMS will be counted.
Day Limit SMS Count	value of daily limit SMS count, 0 means no limit.
Month Limit SMS	value of monthly limit SMS count, 0 means no limit.
Count	
SMS Clean Date	Automatically clear the number of sent messages every
	month at 0:00:00 on the set date.



# 4.5 Call Stats

Count the number of calls, the number of answers, the number of consecutive call failures, the duration of calls, the number of calls, the duration of calls, and the duration of use of all ports.

T Call Statistics									
Port-SIM	Call Limit				strategy				
	Hour Call Count	Daily Call Count	Daily Answer Count	Call Failed Count	Call Duration	Callout Count	Callout Time	SMS Count	Using Time
1-1	0	0	0	0	0	0	0	0	0
1-2	0	0	0	0	0	0	0	0	0
1-3	0	0	0	0	0	0	0	0	0
1-4	0	0	0	0	0	0	0	0	0
2-1	0	0	0	0	0	0	0	0	0
2-2	0	0	0	0	0	0	0	0	0
2-3	0	0	0	0	0	0	0	0	0
}-4	0	0	0	0	0	0	0	0	0
2-3 2-4	0	0	0	0	0	0	0	0	0

#### Figure 4-12 call statistics

## 4.6 SMS Stats

#### Figure 4-13 SMS sent statistics

gr.SMS Sending Statistics							
Port-SIM	SMS count of the day	Daily limit	SMS count of the month	Monthly limit	Monthly recovery date		
1-1	0	0	0	0	0		
1-2	0	0	0	0	0		
1-3	0	0	0	0	0		
1-4	0	0	0	0	0		
2-1	0	0	0	0	0		
2-2	0	0	0	0	0		
2-3	0	0	0	0	0		
2-4	0	0	0	0	0		

### 4.7 Pin Code

When the SIM card is set with a pin code, you need to enter the pin code to make a successful call

Figure 4-14 Pin code



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Port-SIM	Pin Code	Action
1-1		8
1-2		
1-3		B
1-4		5
2-1		
2-2		
2-3		
2-4		


# 5. VOIP

## 5.1 VOIP Endpoints

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

#### SIP En Registration Credentials Actions Endpoint N 1234 1234 2 🗙 server 8888 none 8888@172.16.33.102 2 🗙 client 9999@172.16.33.102 2 × 9999 Add New SIP Endpoint IAX2 Endpoint Credentials Actions Endpoint Nam Registration 2 🗙 1002 serve 1002 1003 1003@172.16.33.102 2 🗙 none 1004@172.16.33.102 1004 client 2 🗙

### Figure 5-1 SIP&IAX2 Endpoints

Add New IAX2 Endpoint

### 5.1.1 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.)



Name:	8888
User Name:	8888 Anonymous
Password:	
Registration:	None T
Hostname or IP Address:	172.16.33.102
Transport:	UDP T
NAT Traversal:	Yes

Figure 5-2 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	5-3	Server
--------	-----	--------

Main Endpoint Settings	
Name:	2000
User Name:	2000 Anonymous
Password:	
Registration:	Server •
Hostname or IP Address:	dynamic
Transport:	UDP V
NAT Traversal:	Yes
Advanced:Registration Options     Call Settings	
Save Apply Cancel	



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

Wain Endpoint Settings	
Name:	9999
User Name:	9999 Anonymous
Password:	••••
Registration:	Client •
Hostname or IP Address:	172.16.33.102
Transport:	UDP •
NAT Traversal:	Yes
Advanced:Registration Options     Call Settings	
Save Apply Cancel	

### Figure 5-4 Client

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;
	Server When register as this type, it means the gateway acts
	as a SIP server, and SIP endpoints register to the gateway;
	<b>Client</b> When register as this type, it means the gateway acts as
	a client, and the endpoint should be register to a SIP server;
Hostname or IP	IP address or hostname of the endpoint or 'dynamic' if the
Address	endpoint has a dynamic IP address. This will require registration.
Transport	This sets the possible transport types for outgoing. Order of

### Table 5-1 Definiton of SIP Options



	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	<b>No</b> 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

Figure 5-5 Advanced	<b>Registration Options</b>
---------------------	-----------------------------

Advanced:Registration Options	
Authentication User:	
Register Extension:	Modify
From User:	Modify
From Domain:	
Remote Secret:	
Port:	
Qualify:	No v
Qualify Frequency:	60
Outbound Proxy:	



Options	Definition
Authentication	A username to use only for registration.
User	
Register	When Gateway registers as a SIP user agent to a SIP proxy
Extension	(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.

### Table 5-2 Definition of Registration Options

### **Call Settings**

### Figure 5-6 Call Settings

Call Settings	
DTMF Settings	
DTMF Mode:	RFC2833 •
Caller ID Settings	
Trust Remote-Party-ID:	No T
Send Remote-Party-ID:	No T
Caller ID Presentation:	Allowed,passed screen
Maximum Channels	
Call Limit:	



### Table 5-3 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:——Signaling Settings

### Figure 5-7 Signaling Settings

V Advanced: Signaling Settings	
Progress Inband:	Yes T
Append user=phone to URI:	No V
Add Q.850 Reason Headers:	No T
Honor SDP Version:	Yes T
Allow Transfers:	Yes T
Allow Promiscuous Redirects:	No T
Max Forwards:	70
Send TRYING on REGISTER:	No T
L	



Options	Definition
	Whether there is ringing tone.
	Never: Indicates that incoming calls are never
	applicable.
	Optional values: yes / no / never. Default: yes
Annond upper-phone to LIPI	Whether or not to Add 'user = phone' to UPIS to
Append user=phone to URI	include a valid phone number in the URI.
Add O 950 Dessen Headers	If it is available, Whether or not to add a reason header
Add Q.050 Reason Headers	and use it.
Honor SDP Version	Whether or not to display Caller ID.
	Whether or not to globally enable transfers. Choosing
Allow Transfers	'no' will disable all transfers (unless enabled in peers or
	users). Default is enabled.
	Whether or not to allow 302 or REDIR to non-local SIP
Allow Promiscuous	address. Note that promiscredir when redirects are
Redirects	made to the local system will cause loops since this
	gateway is incapable of performing a "hairpin" call.
	Setting for the SIP Max-Forwards header (loop
Max Forwards	prevention). Send TRYING on REGISTER Send a 100
	Trying when the endpoint registers.
	A proxy to which the gateway will send all outbound
Outbound Proxy	signaling instead of sending signaling directly to
	endpoints.





### Advanced——Timer Settings

### Figure 5-8 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 T

### Table 5-5 Definition of Timer Options

Options	Definition
	This timer is used primarily in INVITE transactions. The
	default for Timer T1 is 500ms or the measured run-trip time
Default 11 Timer	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 feature operates in the following three
Cassion Timoro	modes: originate, Request and run session-timers always;
Session Timers	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	Maximum session refresh interval in seconds. Defaults to
Interval	
Session Refresher	The session refresher, uac or uas. Defaults to uas.



### 5.1.2 Add New IAX2 Endpoint

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.

The Main Endpoint Settings	
Name:	1003
User Name:	1003
Password:	
Registration:	None •
Hostname or IP Address:	172.16.33.102
Auth:	md5 T
Transfer:	No •
Trunk:	No 🔻
Advanced:Registration Options	
IAX2 Encryption	
IAX2 Trunk settings	
Save Apply Cancel	

#### Figure 5-9 None Registrarion

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 5-10 Server

Main Endpoint Settings	
Nam	e: 1003
User Nam	e: 1003
Passwor	d:
Registratio	n: Server T
Hostname or IP Addres	s: dynamic
Aut	h: md5 ▼
Transfe	n: No V
Trun	k: No T

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

▼ Main Endpoint Settings	
Name:	1003
User Name:	1003
Password:	
Registration:	Client •
Hostname or IP Address:	172.16.33.102
Auth:	md5 •
Transfer:	No 🔻
Trunk:	No •
Advanced:Registration Options	
IAX2 Encryption	
IAX2 Trunk settings	
Save Apply Cancel	

### Figure 5-11 Client



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 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 gateway acts as a IAX2 client, and the
	endpoint should be register to a IAX2 server;
Hostname or	IP address or hostname of the endpoint or 'dynamic' if the
IP Address	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

Figure 5-12 Registration Options

Advanced:Registration Options	
Qualify:	Yes •
Qualify Smothing:	Yes •
Qualify Freq Ok:	6000
Qualify Freq Not Ok:	6000
Port:	4569
Require Call Token:	Yes •

Table 5-7 Definition of	f Registration Options
-------------------------	------------------------

Options	Definition
Qualify, Qualify Freq	The qualify, qualifyfreqok and qualifyfreqnotok settings are used
Ok, Qualify Freq	to determine the status availability of an IAX peer. If a peer is
Not Ok	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

### Figure 5-13 IAX2 Encryption

VIAX2 Encryption	
Encryption:	No 🔻
Force Encryption:	No 🔻

#### **Table 5-8 Definition of Encrytion 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

### Figure 5-14 IAX2Trunk Settings

VIAX2 Tr	unk settings	
	Trunk Max Size:	128000
	Trunk MTU:	0
	Trunk Frequency:	20
	Trunk Time Stamps:	No 🔻
	Min. RegExpire:	60
	Max. RegExpire:	60



### Table 5-9 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).	



## 5.2 Batch SIP Endpoints

In this page, you can generate multiple SIP Extentations at the same time

ID	User Name	Password	Hostname or IP Address	Port	Register Mode
					client 🔻
1					client <b>T</b>
2					client •
3					client <b>T</b>
4					client •
5					client •
6					client •
7					client <b>T</b>
8					client V
9					client <b>T</b>
10					client •
11					client <b>T</b>
12					client •
13					client •
14					client •
15					client •
16					client 🔻

Figure 5-15 Multiple SIP Extentations Settings

Save Cancel Batch & AutoPassword

You can fill in the user name, password, domain name or IP address, port, and registration mode on the firt line and select the number of SIPs to be created. You can create up to the same number of SIP endpoints as the number of device ports at a time. After the above configuration, click Batch Setup and save it to create SIP endpoints in batches.

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;
	Server When register as this type, it means the gateway acts
	as a SIP server, and SIP endpoints register to the gateway;

	Table 5-10	Definition	of Multin	ole SIP	Extentations
--	------------	------------	-----------	---------	--------------



	Client When register as this type, it means the gateway acts
	as a client, and the endpoint should be register to a SIP server;
Hostname or IP	IP address or hostname of the endpoint or 'dynamic' if the
Address	endpoint has a dynamic IP address. This will require
	registration.
AutoPassword	Tick - Automatically increments based on the password
	entered in the first lineDo not check - All SIP endpoints have
	the same password as the first one.

### 5.3 Advanced SIP Settings

### 5.3.1 Networking

Networking General

### Figure 5-16 Networking General

General	
UDP Bind Port:	5060
Enable TCP:	No 🔻
TCP Bind Port:	5060
TCP Authentication Timeout:	
TCP Authentication Limit:	
Enable Hostname Lookup:	No 🔻
Enable Internal SIP Call:	No 🔻
Internal SIP Call Prefix:	



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.
TOD	The maximum number of seconds a client has to authenticate.
	If the client does not authenticate before this timeout expires,
Authentication Timeout	the client will be disconnected.(default value is: 30 seconds).
ТСР	The maximum number of unauthenticated sessions that will
Authentication Limit	be allowed to connect at any given time (default is: 50).
	Enable DNS SRV lookups on outbound calls Note: the gateway
	only uses the first host in SRV records Disabling DNS SRV
Enable	lookups disables the ability to place SIP calls based on domain
Hostname Lookup	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	Whether enable the internal SIP calls or not when you select
SIP Call	the registration option "Endpoint registers with this gateway".
Internal SIP Call Prefix	Specify a prefix before routing the internal calls.

### Table 5-11 Definition of Networking General Optiongs

### NAT Settings

### Figure 5-17 NAT Settings

NAT Settings		
Local Network:	Add	
Local Network List:	IP Range	Action
Subscribe Network Change Event:	No 🔻	
Match External Address Locally:	No T	
Dynamic Exclude Static:	No T	
Externally Mapped TCP Port:		
External Address:		
External Hostname:		
Hostname Refresh Interval:		



Table 5-12 Definition of NAT	<b>F</b> Settings Options
------------------------------	---------------------------

Options	Definition
	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
Local Network	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.
	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
Subseribe Network Change	will renew all outbound registrations when the monitor
	detects any sort of network change has occurred. By
Event	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	Only substitute the externaddr or externhost setting if it
Locally	matches.
	Disallow all dynamic hosts from registering as any IP
Dunamic Evoludo Static	address used for statically defined hosts. This helps avoid
Dynamic Exclude Static	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.
	How often to perform a hostname lookup. This can be
	useful when your NAT device lets you choose the port
Hostname Refresh Interval	mapping, but the IP address is dynamic. Beware, you
	might suffer from service disruption when the name
	server resolution fails.

### **RTP Settings**

### Figure 5-18 RTP Settings

RTP Settings	
Start of RTP Port Range:	10000
End of RTP port Range:	20000
RTP Timeout:	120

### Table 5-13 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
RTPTimeout	RTP Timeout retransmission time



### 5.3.2 Paesing and Compatibility

General               final interpretation            Strict RFC Interpretation               Interpretation            Strict RFC Interpretation               Interpretation            Strict RFC Interpretation               Interpretation            Strict RFC Interpretation               Reference            Strict RFC Interpretation               AK            Strict RFC Interpretation               AK            Strict RFC Interpretation               AK            Strict RFC Interpretation               AK            Nortin               AK            Nortin               AK            Nortin               AK            Nortin               AK            PREARC               AK            Nortin               AK            Nortin               AK            Reference             Nortin              Strict Areits              Nortin	Parsing and Compatibility	
Strict RFC Interpretation       Yes         Strict RFC       No         Refer R       No         Strict RFC       No         Strict RFC       No         Strict Relistrict RFC       No         Strict Relistrict RFC       No         Strict RFC       No         Strict RFC       No         Strict RFC       No         Strict Relistrict RFC       No         Strict Relistrict RFC       No         Strict RFC       No         Strict Relistrict RFC       Strict RFC         Strict Relistrict RElistri	General	
See Compact Heeders:       No         SDP Owner:	Strict RFC Interpretation:	Yes •
SDP Owner:	Send Compact Headers:	No •
SIP Methods         ACK         ACK         BYE         CANCEL         INFO         INTE         INTER         OPTIONS         PRACK         PPULISH-         OPTIONS         PRACK         PPULISH-         REGISTER         SUBSCRUE         UPATE         UPATE         UPATE         UPATE         Tere Configuration         Maximum Registration Expir;         Image Stration Expir;         Default Registration Expir;         OPTicumer Registration         Registration Timecut;         Q         Number of Registration Timecut;	SDP Owner:	
ACK	SIP Methods	
Bis		ACK 🗆
CANCEL       IN TO         IN TO       INTE         INTE       INTE         MESSAGE       INTE         INTEY       INTE         PRODUCTIONS       PRODUCTIONS         PRACK       PRODUCTIONS         Mangup Cause Code       503 Service Unavailable         V       Image Station Expire         Manmun Registration Expire       Image Station Expire         Minmum Registration Expire       Image Station Expire         Cubcound Registration Expire       Image Station Expire         Cubcound Registration Expire       Image Station Expire         Minum Registration Expire       Image Station Expire		BYE
INFO     INFO       OPTIONE     OPTIONE       PRACE     OPTIONE       PRACE     INFO       INFO     INFO		
Invite     Invite		INFO 🗆
bisalioved SIP Methods     MESSAGE       NTTF*     OTTONS       PRACK     PUBLISH       REFER     REFER       REGISTER     SUBSINIBE       UPDATE     OTTON       Torr Configuration     Sol Service Unavailable       Torr Configuration Expiry:     Image Sistation Expiry:       Default Registration Expiry:     Image Sistation Expiry:       Sinder Gegistration Expiry:     Image Sistation Expiry:       Default Registration Expiry:     Image Sistation Expiry:		INVITE 🗉
Disaliowed SIP Methods     NoTIFY [       OPTIONS [     PROLED       PROLED     PROLED       PUBLISH [     REFER [       REGISTER [     SUBSCRIEE [       UDDTE [     OPTOTON [       Braugu Cause Code     503 Service Unavailable [       Strink Caller Die     Sono [       Minimum Registration Expiry:     [       Default Registration Expiry:     [       Outcound Registration Expiry:     [       Default Registration Expiry:     [       Outcound Registration Expiry:     [       Default Registration Expiry:     [       Outcound Registration Expiry:     [		MESSAGE
Optimized on methods     Options       PRACK     PRACK       PREFER     REGISTER       SUBSCRIBE     UPDATE       UPDATE     503 Service Unavailable       UPDATE     503 Service Unavailable       Timer Configuration     No        Maximum Registration Expiry:     Imminum Registration Expiry:       Optiound Registration Expiry:     Imminum Registration Expiry:       Outbound Registration Expiry:     Imminum Registration Expiry:       Segistration Timeout:     20	Disallowed SID Methods	NOTIFY
PRACK       PRACK         PUBLISH       PUBLISH         REFER       REFER         REOISTER       SUBSCRIBE         UPDATE       UPDATE         Caler ID       Shrink Caller ID         No       Image Cause Code		OPTIONS 📃
PUBLISH       PUBLISH       REFER		PRACK
REFR       REFR         REGISTER       REGISTER         REGISTER       SUBSCRIBE         UPDATE       UPDATE         UPDATE       Service Unavailable         Caller ID       N         Shrink Caller ID       N         Immum Registration Expiry:       Immum Registration Expiry:         Default Registration Expiry:       Immum Registration Expiry:         Registration Expiry:       Immum Registration Expiry:         Number of Registration Timeout;       Immum Registration Expiry:		PUBLISH 🔲
REGISTER       REGISTER         SUBSCRIEE       UPDATE         UPDATE       000000000000000000000000000000000000		REFER
SUBSCRIBE     UPDATE       UPDATE     UPDATE       Imagup Cause Code:     503 Service Unavailable       Caller ID     No       Shrink Caller ID:     No       Timer Configuration     Imaguar Code:       Maximum Registration Expiry:     Imaguar Code:       Default Registration Expiry:     Imaguar Code:       Outbound Registration Expiry:     Imaguar Code:		REGISTER 🗌
UPDATE         Hangup Cause Code:       503 Service Unavailable •         Caller ID         Shrink Caller ID:       No •         Timer Configuration         Maximum Registration Expiry:		SUBSCRIBE
Hangup Cause Code:       503 Service Unavailable         Caller ID         Shrink Caller ID:       No         Timer Configuration         Maximum Registration Expiry:		UPDATE
Caller ID Caller ID Shrink Caller ID No No Configuration Timer Configuration Maximum Registration Expiry: Default Registration Expiry: Coutbound Registrati	Hangup Cause Code:	503 Service Unavailable 🔻
Shrink Caller ID: No •     Timer Configuration   Maximum Registration Expiry:   Outbound Registration Expiry:     Outbound Registration S   Registration Timeout:   20   Number of Registration Attempts:	Caller ID	
Simile Caller ID     IV V	Shrink Callor ID:	No. •
Timer Configuration       Maximum Registration Expiry:       Minimum Registration Expiry:       Default Registration Expiry:       Outbound Registrations       Registration Timeout       20       Number of Registration Attempts:       0	Similik Caller ID.	
Maximum Registration Expiry:	Timer Configuration	
Minimum Registration Expiry:	Maximum Registration Expiry:	
Default Registration Expiry:	Minimum Registration Expiry:	
Default Registration Expiry:           Outbound Registration Timeout:           20           Number of Registration Attempts:           0		
Outbound Registrations       Registration Timeout:       20       Number of Registration Attempts:	Default Registration Expiry:	
Registration Timeout:     20       Number of Registration Attempts:     0	Outbound Registrations	
Number of Registration Attempts: 0	Registration Timeout:	20
	Number of Registration Attempts:	0

### Figure 5-19 Paesing 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
	Allows you to change the username filed in the SDP owner
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

#### Table 5-14 Instruction of Parsing and Compatibility



	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	Maximum allowed time of incoming registrations and	
Expiry	subscriptions (seconds).	
Minimum Registration	Minimum longth of registrations (subseriations (default CO	
Expiry	winning in 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.	

### 5.3.3 Security

### Figure 5-20 Security Settings

V Security	
Authentication Settings	
Match Auth Username:	No
Realm:	
Use Domain as Realm:	No •
Always Auth Reject:	No •
Authenticate Options Requests:	No •
Guest Calling	
Allow Guest Calling:	No •



### Table 5-15 Instruction of Security

Options	Definition
	If available, match user entry using the 'username' field from
Match Auth Osername	the authentication line instead of the 'from' field.
	Realm for digest authentication. Realms MUST be globally
Realm	unique according to RFC 3261. Set this to your host name or
	domain name.
	Use the domain from the SIP Domains setting as the realm.
	In this case, the realm will be based on the request 'to' or
Use Domain as Realm	'from' header and should match one of the domain.
	Otherwise, the configured 'realm' value will be used.
	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
Always Auth Reject	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	Enabling this option will authenticate OPTIONS requests just
Requests	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.





### 5.3.4 Media

### Figure 5-22 Media Settings

Media	
QoS/ToS	
TOS for SIP Packets:	
TOS for RTP Packets:	

### Table 5-16 Instruction of Media

Options	Definition
	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
Dromatura Madia	before we have call progress (meaning the SIP channel
Premature Media	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

### 5.3.5 Codec Settings

Select codecs from the list below.



Figure 4-22 Codec Settings

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

## 5.4 Advanced IAX2 Settings

### **5.4.1 General Settings**

### Figure 5-23 General Settings

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





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Table 5-17	Instruction	of General
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Options	Definition
Bind Port	Bind port and bindaddr may be specified
Enable	More than once to bind to multiple addresses, but the first will be
IAXCompat	the default.
Enable	Set iaxcompat to yes if you plan to use layered switches or some
Nochecksums	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	Disable UDP checksums (if no checksums is set, then no checksums
Reject	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.



### 5.4.2 Music on Hold

### Figure 5-24 Music on Hold Settings

Wusic On Hold	
Mohsuggest:	default 🔻
Mohinterpret:	default

#### Table 5-18 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)

### 5.4.3 Instruction of Codec Settings

### Figure 5-25 Codec Settings

V Codec Settings	
Band Width:	low T
Disallow:	all 🔻
Allow:	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:	host •



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 controls the codec negotiation of an inbound IAX2
Codec Priority	call. This option is inherited to all user entity separately which
	will override the setting in general.

### Table 5-19 Instruction of Codec Settings

### 5.4.4 Jitter Buffer Settings

### Figure 5-26 Jitter Buffer

V Jitter Buffer Settings	
Jitter Buffer:	No T
Force Jitter Buffer:	No T
Max Jitter Buffers:	
Resyncthreshold:	Resyncing can be disabled by setting this parameter to -1.
Max Jitter Interps:	
Jitter Target Extra:	

#### Table 5-20 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
	When the jitter buffer notice a significant change in delay that
	continue over a few frames, it will resync, assuming that the
Resyncthreshold	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.

### 5.4.5 Misc Settings

### Figure 5-27 Misc Settings

Wisc Settings	
IAX2 Thread Count:	
IAX2 Max Thread Count:	
Max Call Number:	
MaxCallNumbers_Nonvalidated:	



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.				

### 5.4.6 Quality of Service

Figure	4-28	Quality	of of	Service
--------	------	---------	-------	---------

<b>V</b> Quality of Service	
tos:	High Reliability
cos:	



### Table 4-22 Instruction of Quality of Service

Options	Definition
Tos	Type of service
Cos	Class of service



				Figu	re 6-1 Routing R	ules	
	Move	Order	Rule Name	From	То	Rules	Actions
	\$	1	OUT	sip-1234	grp-ALL		2 🗱
	•	2	IN	grp-ALL	custom-playback		2 🗙
	•	3	test	sip-2000	cdma-1.1		2 🗙
You ar routing	e al g rul	lowe	ed to set up nove rules' o	o new routi order by pul	ng rule by Ne	w Call Routing Rule , ar	nd after setting tton to edit the
routing	g and		to delete	it. Finally clic	k the Save Orde	button to save what y	vou set.
Call Ro	utin	g Rul	e:				
You can	click	Ne	w Call Routing	Rule button	n to set up your rou	tings.	

### Figure 6-2 Example of Set up Routing Rule

Routing Name:	IN		
Call Comes in From:	ALL	T	
Send Call Through:	cdma-1.8 cdma-1.9 cdma-1.10		
Settings	cdma-1.11 cdma-1.12 cdma-1.13		
Authentication:	cdma-1.14 cdma-1.15		
Secondary Dialing:	cdma-1.16 SIP		
DISA Timeout:	8888 9999		
Max Password Digits:	2000 IAX2		
Password:	1002 1003		
nce Routing Rule	GROUP		





Call Routing Rule	
Routing Name:	IN
Call Comes in From:	ALL
Send Call Through:	1234
	Custom
	Port
DISA Setungs	cdma-1.2(18002548416)
Authentication:	cdma-1.3 cdma-1.4
Secondary Dialing:	cdma-1.5 cdma-1.6
DISA Timeout:	cdma-1.7 cdma-1.8
Max Password Digits:	cdma-1.9 cdma-1.10
Password:	cdma-1.11 cdma-1.12 cdma-1.13
	cdma-1.13
	cdma-1.15
Advance Routing Rule	cdma-1.16
	SIP
Save Apply Cancel	1234
DISA Settings	
Authentica	tion: ON
Secondary Dia	ling: OFF
DISA Tim	eout: 5 s
Max Password D	igits: 10 V
Passv	word: Edit
Advance Routing Rule	
Save Apply Cancel	

The figure above shows that all the phones in the group ALL are transferred to the SIP-1234 terminal.

Options	Definition
	The name of this route. Should be used to describe what types
Routing Name	of calls this route matches (for example, 'SIP2CDMA' or
	'CDAM2SIP').
Call Comes in From	The launching point of incoming calls.
Send Call Through	The destination to receive the incoming calls.



Options	Definition			
	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.			
Dial Dattorns that will	prepend: Digits to prepend to a successful match			
Use this Poute	If the dialed number matches the patterns specified by the			
use this Route	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.			

### Table 6-2 Description of Advanced Routing Rule



	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	What caller ID name would you like to set before sending		
ID Name to	this call to the endpoint.		
Forward Number	What destination number will you dial? This is very useful		
	when you have a transfer call.		
Custom Context	User-defined dialing rules		
Failover Call Through	The gateway will attempt to send the call out each of these		
	in the order you specify. You can create various time routes		
Number	and use these time conditions to limit some specific calls.		

#### Figure 6-3 Time Patterns that will use this Route

Time Patterns that will use this Route					
Time to start: - 🔻 : - 🔻	Week Day start: -	Month Day start: - 🔻	Month start: -	•	~
Time to finish: - 🔻 : - 🔻	Week Day finish: -	Month Day finish: - 🔻	Month finish: -	۲	*
+ Add More Time Pattern Fields					

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 6-4 Failover Call Through Number

Failover Call Through Number			
Failover Call Through Number 1:	None	Ŧ	
Add a Failover Call Through Provider			

You can add one or more "Failover Call Through Numbers".



### 6.1 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.

Routing Groups		
Group Name:	ALL	
Туре:	MODULE T	
Policy:	Roundrobin	
Members	NO.       All         1       ✓ cdma-1.1         2       ✓ cdma-1.2(18002548416)         3       ✓ cdma-1.3         4       ✓ cdma-1.4         5       ✓ cdma-1.4         6       ✓ cdma-1.6         7       ✓ cdma-1.6         7       ✓ cdma-1.7         8       ✓ cdma-1.8         9       ✓ cdma-1.10         11       ✓ cdma-1.10         11       ✓ cdma-1.11         12       ✓ cdma-1.12         13       ✓ cdma-1.13         14       ✓ cdma-1.15         16       ✓ cdma-1.16	

#### **Figure 6-5 Routing Group**

Save Apply Cancel





### 6.2 Batch Creating rules

This page can generate multiple routing rules at the same time

Port	Sim Number	Sip Trunk	CallerID
gsm-1.1		None •	
gsm-1.2		None •	
gsm-1.3		None •	
gsm-1.4		None •	
gsm-1.5		None T	
gsm-1.6		None <b>T</b>	
gsm-1.7		None •	
gsm-1.8		None <b>T</b>	
gsm-1.9		None <b>v</b>	
gsm-1.10		None 🔻	
gsm-1.11		None •	
gsm-1.12		None <b>v</b>	
gsm-1.13		None <b>T</b>	
gsm-1.14		None <b>v</b>	
gsm-1.15		None •	
gsm-1.16		None •	

#### Figure 6-6 Batch Creating rules Group

Save Cancel

You can configure the SIM Number, SIP trunk and calling Number for each port. And then, click "save" to batch creating multiple Routing rules. By an attention, the SIP trunk must be configured and the SIM number and calling Number can be empty.

Options	Definition
Forward Number	What destination number will you dial? This is very useful
	when you have a transfer call.
SIP Trunk	Inbound and outbound calls through designated SIP trunks
Set the Caller	What caller ID name would you like to set before sending
ID Name to	this call to the endpoint.

#### Table 6-3 Description of Advanced Routing Rule
### 6.3 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}.

MNP Settings	
MNP Check Enable:	ON
MNP URL:	
MNP Timeout:	
Manipulation Choice:	Route calls after manipulation     Route calls before manipulation

Save

#### Figrue 6-7 MNP Settings



# **7. SMS**

### 7.1 General

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

### Figure 7-1 SMS Settings

General 🔥 Tur	m on SMS Received switch before you enable SMS Local Stored, SMS to Email or SMS to HTTP!
SMS Received:	
SMS Local Stored:	
SMS Status Report:	OFF

### 7.1.1 Sender Options

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

#### Figure 7-2 Sender Options

Sender Options	
Resend Failed Message:	1 •
Repeat Same Message:	2 •
Verbose:	3 •

#### **Table 7-1 Description of Sender Options**

Options	Definition
Posond Failed Mossage	The times that you will attempt to resend your failed
Reserve Falled Message	message.
Repeat Same Message	The times that you will resend the same message.

### 7.1.2 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 <u>openvpnvoip@gmail.com</u> transmit



### to openvpnvoip@yahoo.com.cn, openvpnvoip@hotmail.com and support@openvox.cn

SMS to Email	
Enable:	ON
SMTP Server:	OTHER •
Email Address of Sender:	openvpnvoip@gmail.com
Domain:	smtp.gmail.com
SMTP Port(default 25):	25
SMTP User Name:	openvpnvoip@gmail.com
SMTP Password:	
TLS Enable:	This option allows the authentication with certificates.
Destination Email Address 1:	openvpnvoip@gmail.com
Destination Email Address 2:	openvpnvoip@gmail.com
Destination Email Address 3:	support@openvox.cn
Title:	support
Content:	We can offer you 24 hours' support

Figure 7-3 SMS to Email

### Table 7-2 Types of E-mail Box

E-mail Box Type	SMTP Server	SMTP Port	SMTP Security Connectivity
Gmail	smtp.gmail.com	587	V
HotMail	smtp.live.com	587	V
Yahoo!	smtp.mail.yahoo.co.i n	587	×
e-mail	smtp.163.com	25	×

#### Table 7-3 Definition of SMS to E-mail

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



Email Address	To set the email address of an available email account. For				
of Sender	example, <u>openvpnvoip@gmail.com</u> .				
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.				



### 7.1.3 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.

SMS Control	
Enable:	ON
Password:	123456
SMS Formats:	reboot system PASSWORD reboot asterisk PASSWORD restore config PASSWORD get info PASSWORD
SMS Inbox Auto clean:	ON maxsize: 20MB V

### Figure 7-4 SMS Control

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.

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

#### Table 7-4 Definition of SMS Control



	"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	switch on: When the size of the SMS inbox record file reaches the max				
Auto clean	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				

### 7.1.4 HTTP to SMS

<b>-</b> <sup>1</sup> · · · ·				C
Figure	7-5	HIIP	το	SIVIS

HTTP to SMS				
Enable:	ON			
URL:	http://172.16.6.130:80/sends/	ms?username=xxx&password=xxx&phonenum	ber=xxx&message=xxx&[port=xxx&][report=xxx&]	timeout=xxx]
User Name:	smsuser	Subsect of the second s		
Password:				
Port:	<ul> <li>✓ cdma-1.1</li> <li>✓ cdma-1.5</li> <li>✓ cdma-1.9</li> <li>✓ cdma-1.13</li> <li>All</li> </ul>	<ul> <li>♂ cdma-1.2(18002548416)</li> <li>♂ cdma-1.6</li> <li>♂ cdma-1.10</li> <li>♥ cdma-1.14</li> </ul>	<ul> <li>✓ cdma-1.3</li> <li>✓ cdma-1.7</li> <li>✓ cdma-1.11</li> <li>✓ cdma-1.15</li> </ul>	<ul> <li>✓ cdma-1.4</li> <li>✓ cdma-1.8</li> <li>✓ cdma-1.12</li> <li>✓ cdma-1.16</li> </ul>
Report:	String •			
Advanced:	ON			
Debug:	0			
Timeout:	20	second		
Wait Timeout:	20	second		
GSM Send Timeout:	10	second		
Socket Timeout:	2	second		

### 7.1.5 SMS to HTTP

### Figure 7-6 SMS to HTTP Settings

SMS to HTTP	
Enable:	
URL:	http:// 172.16.80.211 : 80 / receivesms.php ? num =phonenumber & port =port & message = message & time = time & User Defined



### 7.2 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.

Port:	cdma-1.1 cdma-1.5 cdma-1.9	cdma-1.2(18002548416) cdma-1.6 cdma-1.10	cdma-1.3 cdma-1.7 cdma-1.11	cdma-1.4 cdma-1.8 cdma-1.12	
	cdma-1.13	Cdma-1.14	Cdma-1.15	cdma-1.16	
Flash SMS:	OFF				
Load numbers from text file:	选择文件 未选择任何	文件			
Destination Number:	"; semicolon" , "  vertical	Bar",",comma "," blank ",":colon ",".dot " wen	e treated as separators in Destination Number	List	
Message:					
Action:	Send Stop				

#### Figure 7-7 SMS Sender

### 7.3 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 7-8 SMS Inbox



#### GWP1600/2120 User Manual

	all		from	to	
er al R	Clean Filter ecords: 180				
	Port	Phone Number	\$ Time		Message
	cdma-1.10	106980008868	2017/11/03 21:09:37		,祝您投资愉快!更多账户信息请徽信关注"国泰基金"。送订回复0X12【国 基金】
	cdma-1.10	106980008868	2017/11/03 21:09:37		尊敬的高小平,您11/2的申购国泰估值优势申请已成功,金额100.00元,单 净值3.024元,份额33.02份。感谢您对本公司的信赖
1	cdma-1.13	106902142205656	2017/11/03 12:20:45		【大街网】您好,我是职业顾问Grace,您很符合光线传媒的人才库标准,现 邀请您加入 d-j.me/DR84CH1 回复TD邀订
	cdma-1.13	@18664565204	2017/11/03 11:43:52		test teststet
	cdma-1.1	18002549645	2017/11/03 11:43:36		test teststet
1	cdma-1.11	@18664565204	2017/11/03 11:43:42		test teststet
	cdma-1.1	18002549645	2017/11/03 11:43:33		test teststet
	cdma-1.2	18002547641	2017/11/03 11:22:43		() )088 !  df
0	cdma-1.2	18002547641	2017/11/03 11:22:40		send\r\n receive send \r\n receive \$`↑↓0,0(∩_
	cdma-1.10	@18664565204	2017/11/03 09:54:43		test sms forwarding 5 1

Delete Clean Up Export

### 7.4 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.

	Port	Phone Number	Time	Message Keyw	ords	
	all		from to			
Filter	Clean Filter					
Total F	Records: 131					
	Port	Phone Number	💠 Time	Status	Message	
	cdma-1.13	18664565204	2017-11-03 11:43:52	Success	test teststet	
	cdma-1.11	18664565204	2017-11-03 11:43:42	Success	test teststet	
	cdma-1.5	18002547641	2017-11-03 11:43:38	Success	test teststet	
	cdma-1.5	18002547641	2017-11-03 11:43:34	Success	test teststet	2
	cdma-1.5	18002547641	2017-11-03 11:39:53	Success	test teststet	
0	cdma-1.1	18002548416	2017-11-03 11:22:44	Success	send\r\n receive send \r\n receive % ↑ ↓ 0, 0(∩_∩)00ģ! "" df	,
0	cdma-1.1	18664565204	2017-11-03 11:22:35	Success	send\r\n receive send \r\n receive % ↑ ↓ 0, 0(∩_∩)0%}! "" df	
	cdma-1.1	18664565204	2017-11-03 10:17:42	Success	test flash sms	
	cdma-1.5	18664565204	2017-11-03 10:14:37	Success	test flash sms	j
	cdma-1.5	18664565204	2017-11-03 10:12:56	Success	test flash sms	

#### **Figure 7-9 SMS Outbox**

1 2 3 4 5 6 7 8 9 10 11 **b** 1 / 14 go

Delete Clean Up Export



### 7.5 SMS Forwarding

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

button to add new routing.

Such as:

#### Figure 7-10 SMS Forwarding Rules

Routing Name	Туре	Policy	From_Members	To_Members	To Number	Actions
test	module	ascending	cdma-1.1,cdma-1.2(18002548416),cdma-1.4	cdma-1.8,cdma-1.10	18664565204	2 🗙
New Pouting						

SMS received by cdma-1.1 and cdma-1.2, cdma-1.4, will be transferred to phone number 18664565204 through port cdma-1.8 or cdma-1.10.

Figure 7-11 Create a Routing



Routing Groups	
Routing Name:	test
Туре:	MODULE *
Policy:	Ascending T
From Members	NO. 1
To Members	NO. 1 cdma-1.1 2 cdma-1.2 3 cdma-1.3 4 cdma-1.4 5 cdma-1.5 6 cdma-1.6 7 cdma-1.7 8 cdma-1.7 8 cdma-1.7 8 cdma-1.8 9 cdma-1.9 10 cdma-1.10 11 cdma-1.11 12 cdma-1.12 13 cdma-1.13 14 cdma-1.15 15 cdma-1.15 16 cdma-1.16
To Number:	18664565204

Save Cancel

For "ascending" Policy, if you choose 2 or more ports members, it will use first available port to transfer sms. For this case, if cdma-1.8 is available, it will always use cdma-1.8 to transfer sms; Otherwise, it will use cdma-1.10 to transfer sms.

8. Network

### 8.1 LAN Settings

There are three types of LAN port IP, Factory, Static and DHCP. Factory is the default type, and it is 172.16.98.1. 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.

LAN IPv4	
Interface:	eth0
Туре:	Static •
MAC:	00:e0:4c:36:00:35
IPv4 Settings	
Address:	172.16.6.130
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:	ON
Reserved Address:	192.168.99.1
Reserved Netmask:	255.255.255.0
Save	

#### **Figure 8-1 LAN Settings**



Options	Definition
Interface	The name of network interface.
	The method to get IP.
	Factory: Getting IP address by Slot Number
Туре	(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.

Table 8-1 Definition of LAN Settings

**DNS Servers:** A list of DNS IP address. Basically this info is from your local network service provider, and you can fill in four DNS servers.

### 8.2 WAN Settings

There are three types of WAN port IP, Disable, Static and DHCP. DHCP is the default type. When you Choose IPv4 type is "Disable" or "DCHP", this page is not editable.

Figure 8-2 WAN Settings

WAN IPv4	
Interface:	eth1
Туре:	Static •
MAC:	6E:C6:41:63:9D:D4
IPv4 Settings	
Address:	
Netmask:	
Default Cateway	



Options	Definition
Interface	The name of network interface.
	The method to get IP.
	Factory: Getting IP address by Slot Number
Туре	(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.

### Table 8-2 Definition of WAN Settings

### 8.3 VPN Settings

VS-GWP1600/2120 series gateways support PPTP VPN.

### Figure 8-3 VPN Settings

VPNType:	PPTP VPN V
P VP <mark>N</mark> Settings	
Server:	172.16.8.136
Account:	
Password:	
Use MPPE:	



Options	Definition	
	None – close VPN	
VPN Туре	PPTP VPN – use PPTP VPN	
server	The server's IP address	
Account	Server account	
Password	The server's password	
Use MPPE	Whether to use MPPE	
Connection Status	Is it successful to connect to the server	

#### **Table 8-3 Definition of VPN Settings**

### 8.4 DDNS Settings

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

### Figure 8-4 DDNS Settings

DDNS Settings	
DDNS	
Туре:	inadyn 🔻
User Name:	admin
Password:	
Your domain:	www.internet.site.com

Save

Options

DDNS

# Definition Enable/Disable DDNS(dynamic domain name server) Set the type of DDNS server.

#### **Table 8-4 Definition of DDNS Settings**

Set the type of DDNS server.
Your DDNS account's login name.
Your DDNS account's password.
The domain to which your web server will belong.



### 8.5 Toolkit

### 8.5.1 Ping and Traceroute

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

#### Figure 8-5 Toolkit

GSM IP: 172.16.6.13	0 🔻	
baidu.com	Ping	
google.com	Traceroute	
Report		
		ping -I 172.16.6.130 -c 4 baidu.com
PING baidu.com (111.13. 64 bytes from 111.13.101 64 bytes from 111.13.101 64 bytes from 111.13.101 64 bytes from 111.13.101 baidu.com ping statisti 4 packets transmitted, 4 p round-trip min/avg/max =	101.208) from 172.16.6.130: 56 data bytes 208: seq=0 ttl=54 time=61.386 ms 208: seq=1 ttl=54 time=61.084 ms 208: seq=2 ttl=54 time=61.023 ms 208: seq=3 ttl=54 time=60.704 ms CS vackets received, 0% packet loss 60.704/61.049/61.386 ms	
		Result
Successfully ping [ baidu.	com].	

### 8.5.2 TCP Capture

You can capture the tcp packets on the page to facilitate locationg problems.

### 

Figure 8-6 TCP Capture



### **Table 8-5 Definition of DDNS Settings**

Options	Definition	
Inferface	You can choose eth0 or eth1	
Source host	Source host IP	
Destination host	Destination host IP	
Port	Which port you want to capture?	
Protocol	Which protocol you want to capture?	

### 8.6 Security Settings

### 8.6.1 Firewall Settings

### Figure 8-7 Firewall Settings

Firewall Settings	
Firewall Enable:	
Ping Enable:	

Options	Definition	
Firowall Englo	If you want to use White/Black List, and security rules,	
Firewall Enale	you must enable this option.	
	To disable ping or not. OFF: disable ping. This gateway will	
Ping Enable	not allow to ping.	

### **Table 8-6 Deginition of Firewall Settings**



### 8.6.2 White/Black List Settings

White List Enbale: To enable white list or not.

List IP Settings: IPs are separated only by "," character.

#### Figure 8-8 White/Black List Settings

White List Settings	
White List Enable:	
List IP Settings:	172. 16. 8. 160, 172. 16. 2. 6
Black List Settings	
Black List Enable:	
List IP Settings:	172. 16. 6. 134

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 8-9 Firewall Rules Apply**

Firewall Rules Apply		×
Warning:		
Please check your security rules care Wrong rules will cause abnormal beh	efully before apply!!! navior on gateway!	
Apply Tips:		
If your security rules will result in no You can login gateway and check the Otherwise, they will be applied succe	response on web login, all rules will be dea e rules again after 1 minute. essfully.	activated.
Notice:		
58 seconds later, all rules will be de The dialog will close automitically, whether the dialog will close automitically.	activated. hen the time runs out.	
	· · · · · · · · · · · · · · · · · · ·	
	Apply	Close

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



Figure 8-10 Firewall Rules Apply

Firewall Rules Apply	×
All rules are active now!	
Firewall rules list below:	
Chain INPUT (policy ACCEPT) target prot opt source destination ACCEPT all 127.0.0,1 0.0.0.0/0 ACCEPT all 172.16.8.160 0.0.0.0/0 ACCEPT all 172.16.2.6 0.0.0.0/0 DROP all 172.16.6.134 0.0.0.0/0	
Chain FORWARD (policy ACCEPT) target prot opt source destination	
Chain OUTPUT (policy ACCEPT) target prot opt source destination	
	Apply Close

### 8.7 Security Rules

### **Figure 8-11 Security Rules**

Rule Name	Туре	Protocol	IP	Port	Actio	ns	
test1	TCP	ACCEPT	172.16.80.216/255.255.0.0	5060:5060	0		×
test2	UDP	DROP	172.16.80.216/255.255.0.0	1000:2000	0		×

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 8-12 Security Rules Apply







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

**Figure 8-13 Security Rules Apply** 

Firewall Rules Apply	×
All rules are active now!	
Firewall rules list below:	
Chain INPUT (policy ACCEPT) target prot opt source destination ACCEPT all 127.0.0.1.0.0.0/0	
DROP udp 172.16.0.0/16 0.0.0.0/0 udp dpts:1000:2000 ACCEPT tcp 172.16.0.0/16 0.0.0.0/0 tcp dpt:5060 DROP tcp 0.0.0.0/0 0.0.0.0/0 tcp dpt:5060	
Chain FORWARD (policy ACCEPT) target prot opt source destination	
Chain OUTPUT (policy ACCEPT) target prot opt source destination	
	Apply Close

### 8.8 SIP Capture

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

#### Figure 8-14 SIP Capture

SIP Capture	
Interface:	eth0 T
Method-filter:	INVITE OPTIONS REGISTER All

Start Capture

Options	Definition
Inferface	You can choose eth0 or eth1
Method-filter	You can choose INVITE, OPTIONS and REGISTER

#### **Table 8-7 SIP Capture Settings**



## 9. Advances

### 9.1 Asterisk API

6

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

Enable:	
Port:	5038
Manager	
Manager Name:	admin
Manager secret:	admin
Deny:	
Permit:	
Rights	
System:	read: 🗹 write: 🗹
Call:	read: 🗹 write: 🗹
Log:	read: 🖉 write: 🗹
Verbose:	read: 🖉 write: 🗹
Command:	read: 🔲 write: 🖉
Agent:	read: 🗹 write: 🖉
User:	read; 🗹 write; 🖉
Config:	read: 🗹 write: 🗹
DTMF:	read: 🗹 write:
Reporting:	read: 🗹 write: 🗹
CDR:	read: 🗹 write:
Dialplan:	read: 🗹 write:
Originate:	read: 💿 write: 🗹
All:	read: 🗹 write: 🗹

### Figure 9-1 Asterisk API

Save

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&10.0.0/255.0.0.0		
Permit	If you want to permit many hosts or network, use char &		
	as separator. Example: 0.0.0/0.0.0.0 or		
	192.168.1.0/255.255.255.0&10.0.0/255.0.0.0		
	General information about the system and ability to run		
System	system management commands, such as Shutdown,		
	Restart, and Reload.		
Call	Information about channels and ability to set information in		
Call	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.		
Agont	Information about queues and agents and ability to add		
Agent	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.		
Doporting	Ability to get information about the system. CDR Output of cdr,		
Reporting	manager, if loaded.		
CDR	Call records. 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 telnet. 172.16.179.1 is the gateway's IP, and 5038 is its API port.



#### Figure 9-2 Telnet Access Gateway API

[c:\~]\$ telnet	172.16.6.130 5038	
Connecting to 1 Connection estal To escape to lo Asterisk Call M action:login username:admin secret:admin	72.16.6.130:5038 blished. cal shell, press Ctrl+Alt+]. anager/1.1 input this manuall	
Response: Succe Message: Authen	ss tication accepted	

### 9.2 Asterisk CLI

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

Asterisk CLI		
Command:	gsm show spans	Execute
utput:		
SM span 1: Power on, Provis SM span 2: Power on, Provis SM span 3: Power on, Provis SM span 4: Power on, Provis SM span 5: Power on, Provis SM span 6: Power on, Provis SM span 7: Power on, Provis SM span 9: Power on, Provis SM span 10: Power on, Provis SM span 11: Power on, Provis SM span 12: Power on, Provis SM span 12: Power on, Provis SM span 13: Power on, Provis SM span 14: Power on, Provis SM span 15: Power on, Provis SM span 15: Power on, Provis	ioned, Up, Active, Standard ioned, Up, Active, Standard sioned, Up, Active, Standard	

Figure 9-3 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.



### 9.3 Asterisk File Editor

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

Configuration Files	
File Name	File Size
asterisk.conf	275
<u>cdr.conf</u>	572
chan extra.conf	56
dnsmar.conf	245
dsp.conf	1520
extensions.conf	120
extensions custom.conf	278
extensions macro.conf	3354
extensions routing.conf	13440
extra-channels.conf	10780
1 2 3 4 <b>•</b> 1 /4 go	

#### Figure 9-4 Asterisk File Editor

New Configuration File Reload Asterisk

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

### 9.4 Cloud Management

VS-GWP1600/2120 series gateways support OpenVox Cloud Management.

Figure	9-5	Cloud	Management
--------	-----	-------	------------

Cloud	
Enable Cloud Service:	
Choose Service:	America •
Account:	
* Password:	
* Connection Status:	Cloud Service Disconnected
	Save Don't have an account? Sign up

If your device is connected to the cloud management, the SSH and 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.



Options	Definition	
Enable Cloud	Turn on/off cloud management	
Service		
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	Is it currently connected to the cloud management platform?	
Status		

Table 9-2 Definition	of Cloud Management	
	or cloud manufchicht	



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.

System Logs	
System Logs:	ON
Auto clean:	ON maxsize : 1MB 🔻
SIP Logs	
SIP Logs:	
Auto clean:	ON maxsize : 2MB •
IAX2 Logs	
IAX2 Logs:	ON
Auto clean:	ON maxsize : 100KB •
Call Detail Record	
Call Detail Record:	OFF
Append IMEI:	OFF
Auto clean:	ON maxsize : 20MB V

### Figure 10-1 Log Settings

Save

#### Figure 10-2 System Logs

System Logs	
[2017/11/02 14:30:28] 1	Power off
[2017/11/02 14:31:29] 1	Power an
[2017/11/02 14:49:16] ]	Restart asterisk (keeper).
[2017/11/02 12:25:03] 1	Power an
[2017/11/02 18:20:26] 1	Restart asterisk (gsm 1 block).
[2017/11/02 18:28:55] 1	Power off
[2017/11/02 18:29:55] 1	Power an
[2017/11/02 18:31:56] ]	Restore configuration files
[2017/11/02 18:31:57] 1	Power off
[2014/01/09 08:14:37] 1	Auto restore configuration files
[2017/11/02 18: 32: 59] 1	Power on
[2017/11/02 18:35:35] :	Send SMS to 18664565204 by 1 (get ip)
[2017/11/03 09:11:17] ]	Restore configuration files
[2017/11/03 09:11:18] 1	Power off
[2014/01/09 22:53:59] ]	Auto restore configuration files
[2017/11/03 09:12:19] 1	Power an
[2017/11/03 09:13:58] ]	Power off
[2017/11/03 09:14:57] ]	Power an
[2017/11/06 10:50:23] 1	Restart asterisk (keeper).

Refresh Rate: 1s • Refresh Clean Up



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	То	Start Time	Duration	Result
					from to	from	to
Ite	r Clean Filter						
ta	Records: 1120	9					
	Caller ID	Callee ID	From	💠 To	Start Time	Duration	💠 Result
)	18025401526	test	cdma-1.8(IMEI:0x00A100005 3080813)	playback	2017-11-02 14:03:45	00:02:45	ANSWERED
0	18018753460	test	cdma-1.6(IMEI:0x00A100005 30808BA)	playback	2017-11-02 14:03:42	00:02:47	ANSWERED
	18025303830	test	cdma-1.7(IMEI:0x00A100005	playback	2017-11-02 14:03:43	00:02:46	ANSWERED

#### Figure 10-3 CDR Output

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

GSM Outbound									
Port	All Calls	All Durations	Answered	Canceled	Busy	No Answer	No Dialtone	No Carrier	Other
cdma-1.1	0	0	0	0	0	0	0	0	0
cdma-1.2(18002548416)	0	0	0	0	0	0	0	0	0
cdma-1.3	0	0	0	0	0	0	0	0	0
cdma-1.4	0	0	0	0	0	0	0	0	0
cdma-1.5	0	0	0	0	0	0	0	0	0
cdma-1.6	0	0	0	0	0	0	0	0	0
cdma-1.7	0	0	0	0	0	0	0	0	0

### Figure 10-4 Outbound

#### Table 10-1 definition of Logs

Options	Definition					
System Logs	Whether enable or disable system log.					
	switch on : when the size of log file reaches the max size, the					
Auto clean	system will cut a half of the file. New logs will be retained;					
(System Logs)	switch off : logs will remain, and the file size will increase					
	gradually. default on, maxsize=1M.					
SIP Logs	Whether enable or disable SIP log.					
Auto clean (SIP logs)	<b>switch on</b> : 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)	<ul> <li>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.</li> <li>switch off: logs will remain, and the file size will increase gradually. default on, maxsize=100KB.</li> </ul>				
Call Detail Record	Displaying Call Detail Records for each channel.				
	switch on : when the size of log file reaches the max size, the				
Auto clean	system will cut a half of the file. New logs will be retained.				
(CDR logs)	switch off : logs will remain, and the file size will increase				
	gradually. default on, max size=20MB.				