



OpenVox Communication Co Ltd



Analog Gateway User Manual

Version 1.0





OpenVox Communication Co Ltd

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Revise History

Version	Release Date	Description
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1. Overview

What is Analog Gateway Series?

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

The Modular Design Analog Gateway Series are developed for interconnecting the PSTN networks with a wide selection of codecs and signaling protocol, including G.711A, G.711U, G.729, G.722, G.723, ILBC to quickly reduce communication expenses and maximize cost-savings.

The Analog Gateway Series use standard SIP protocol and compatible with Leading IMS/NGN platform, IPPBX and SIP servers, support most of the VoIP operating platforms such as Asterisk, Elastix, 3CX, FreeSWITCH, Broadsoft etc.

Sample Application







Product Appearance

The picture below is appearance of Analog Series Gateway.

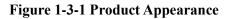
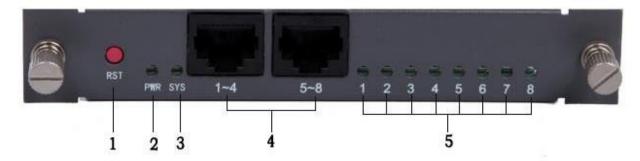




Figure 1-3-2 Front Panel



- 1: Reset button
- 2: Power Indicator
- 3: System LED
- 4: Analog Telephone Interface (2)
- 5: Channels State Indicator

Main Features

Modular design



- Based on Asterisk[®]
- Editable Asterisk[®] configuration file
- Support T.38 fax relay and T.30 fax transparent, can continually fax multiple page
- Echo cancellation and Static jitter buffer
- Wide selection of codecs and signaling protocol
- DTMF relay
- Ring cadence and frequency setting
- MWI(Message waiting indicator)
- > DHCP , DNS/DDNS, NAT Network
- VAG and CNG
- All hot-swap
- Stable performance, flexible dialing, friendly GUI
- Two-year time warranty

Physical Information

Weight	200g	
Size	22cm*2cm*12cm	
Tomporature	-40~125°C (Storage)	
Temperature	0~50°C (Operation)	
Operation humidity	10%~90% non-condensing	
Power source	12V DC/2A	
Max power	11W	

Table 1-5-1 Description of Physical Information

Software

Default IP: 172.16.99.1

Username: admin

Password: admin



Please enter the default IP in your browser to scan and configure the module you want. Now we offer you two RJ45 Network ports to access to your gateway on the board, ETH1 and ETH2. ETH1 only can enter the ip of slot 1, and ETH2 can access every slot. You can choose either of them, that depends on your demand.

Figure 1-6-1 Login Interface

Authentication Required				
The server http://172.16.8.125:80 requires a username and password. The server says: Openvox-Analog- Gateway.				
User Name:	admin			
Password:	****			
	Log In Cancel			



Status

On the "Status" page, you will see Port/SIP/Routing/Network information and status.

Port Information											
Port		Name	Name Type				Line Status/Sip Acco	unt	Port Status		
1		board-1-port1 FXC		FXO	хо		Disconnected		OnHook		
2		board-1-port2		FXO	FXO		Disconnected	Disconnected		OnHook	
3		board-1-port3		FXO	FXO		Disconnected		OnHook		
4		board-1-port4		FXO	FXO		Disconnected		OnHook		
5		board-1-port5		FXS	FXS		8005		OnHook		
6		board-1-port6		FXS			8006		OnHook		
7		board-1-port7		FXS			8007		OnHook		
8		board-1-port8		FXS			8008		OnHook		
SIP Informatio	on										
Endpoint Name)	User Name		Host		Registratio	on SIP Status		Response C	ode	
9001		9001 172.		172.16.8.250	2.16.8.250 server		ок				
9002		9002		172.16.8.250	172.16.8.250 s		ок	ок			
9003		9003		(Unspecified)	specified) server		UNKNOWN	UNKNOWN			
9004		9004		172.16.8.250		server	ок	ок			
9000		9000 17		172.16.208.33 c		client	Registered	Registered		200 OK	
8005		8005		172.16.208.33	2.16.208.33 client		Registered	Registered		200 OK	
8006		8006		172.16.208.33	16.208.33 clier		Registered	Registered		200 OK	
8007		8007		172.16.208.33	2.16.208.33 cli		Registered	Registered		200 OK	
8008	8 8008 172		172.16.208.33	6.208.33 client		Registered		200 OK			
Routing Information											
Rule Name From T		То									
Network Information											
Name	MAC Address	IAC Address IP Address			Mask		iateway	RX Packets		TX Packets	
LAN	A0:98:05:01:51:76 172.16.		172.16.80.16	255.255.0.0		1	72.16.0.1 83463			3704	

Figure 2-1-1 System Status

Time

Table 2-2-1 Description of Time Settings

Options	Definition
System Time	Your gateway system time.
Time Zone	The world time zone. Please select the one which is the same or the



	closest as your city.
POSIX TZ String	Posix time zone strings.
NTP Server 1	Time server domain or hostname. For example, [time.asia.apple.com].
NTP Server 2	The first reserved NTP server. For example, [time.windows.com].
NTP Server 3	The second reserved NTP server. For example, [time.nist.gov].
Auto-Sync from NTP	Whether enable automatically synchronize from NTP server or not. ON
Auto-sync nom NTP	is enable, OFF is disable this function.
Sync from NTP	Sync time from NTP server.
Sync from Client	Sync time from local machine.

For example, you can configure like this:

Figure 2-2-1 Time Settings

Time Settings				
	System Time:	2017-12-20 12:19:56		
	Time Zone:	Hong Kong	T	
PO	SIX TZ String:	нкт-8		
1	NTP Server 1:	pool.ntp.org	<u>A</u>	
1	NTP Server 2:	202.112.29.82		
NTP Server 3:				
Auto-Sy	nc from NTP:	ON		
Sync from NTP S	ync from Client			

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

Login Settings

Your gateway doesn't have administration role. All you can do here is to reset what new username and password to manage your gateway. And it has all privileges to operate your gateway. You can modify both your "Web Login



Г

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.

Options Definition	
	Define your username and password to manage your gateway, without
User Name	space here. Allowed characters
	"+. < >&0-9a-zA-Z". Length: 1-32 characters.
Password	Allowed characters "+. < >&0-9a-zA-Z". Length: 4-32 characters.
Confirm Password	Please input the same password as 'Password' above.
Login Mode	Select the mode of login.
HTTP Port	Specify the web server port number.
HTTPS Port	Specify the web server port number.
Port	SSH login port number.

Figure 2-3-1 Login Settings

Web Login Settings		
User Name:	admin	
Password:	····· (9)	
Confirm Password:	····· (9)	
Login Mode:	http and https ▼	
HTTP Port:	80	
HTTPS Port:	443	
SSH Login Settings		
Enable:	ON	
User Name:	admin	
Password:	····· (9)	
Port:	12345	

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



General

Language Settings

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

Figure 2-4-1 Language Settings

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

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 2-4-2 Reboot Types

Scheduled Reboot	
Enable:	ON
Reboot Type:	By Week
Week:	Tue •
Time:	Hour: 14 V Minute: 16 V

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

Tools

On the "Tools" pages, there are reboot, update, upload, backup and restore toolkits.



You can choose system reboot and Asterisk reboot separately.

Figure 2-5-1 Reboot Prompt

Reboot Tools		
	The page at 172.16.179.1 says: 🛛 🗶	
Reboot the gateway and all the current calls will be	Are you sure to reboot your gateway now?	System Reboot
	You will lose all data in memory!	
Reboot the asterisk and all the current calls will be		Asterisk Reboot
	OK Cancel	

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

Options	Definition
System Reboot	This will turn off your gateway and then turn it back on. This will drop all current calls.
Asterisk Reboot	This will restart Asterisk and drop all current calls.

Table 2-5-1 Instruction of reboots

We offer two kinds of update types for you, you can choose System Update or System Online Update. System Online Update is an easier way to update your system.

Figure 2-5-2 Update Firmware

Opdate Firmware		
New system file:	浏览	System Update
New system file is downloaded	I from official website and update system.	System Online Update

If you want to store your previous configuration, you can first backup configuration, then you can upload configuration directly. That will be very convenient for you. Notice, the version of backup and current firmware should be same, otherwise, it would not take effect.

Figure 2-5-3 Upload and Backup

Upload Configuration	
New configuration file: Choose File No file chosen	File Upload
Backup Configuration	
Current configuration file version: 1.0.1	Download Backup

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 2-5-4 Factory Reset

Restore Configuration		
This will cause all the configu	ration files to back to default factory values! And reboot your gateway once it finishes.	Factory Reset



Information

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

Model Name:	VS-AGU-E2M0800	
Software Version:	1.1.14	
Hardware Version:	1.0.0	
Slot Number:	1	
Storage Usage:	1.7W/63.5M (3%)	
Memory Usage:	emory Usage: 60.3877 % Memory Clean	
Build Time:	Build Time: 2017-12-12 16:31:16	
Contact Address:	act Address: 10/F, Building 6-A, Baoneng Science and Technology Industrial Park, Longhua New District, Shenzhen, Guangdong, China 518109	
Tel:	+86-755-82535461	
Fax:	+86-755-83823074	
E-Mail:	support@openvox.cn	
Web Site:	www.openvox.cn	
System Time:	2017-12-20 13:51:31	
System Uptime:	0 days 02:10:41	

Figure 2-6-1 System Information



You can see much information about your ports on this page.

Channel Settings

Port	Туре	Name	Line Status/Sip Account	Port Status	Actions
1	FXO	board1-port1	Disconnected	OnHook	0
2	FXO	board1-port2	Disconnected	OnHook	0
3	FXO	board1-port3	Disconnected	OnHook	0
4	FXO	board1-port4	Disconnected	OnHook	0
5	FXS	board1-port5	8005	OnHook	0
6	FXS	board1-port6	8006	OnHook	0
7	FXS	board1-port7	8007	OnHook	0
8	FXS	board1-port8	8008	OnHook	0

Figure 3-1-1 Channel System

On this page, you can see every port status, and click action



button to configure the port.

V General	
Port type:	FXO
Name:	board1-port1
Rx gain:	3.0
Tx gain:	3.0
Ring timeout:	8
Caller ID	
Use callerid:	ON
Hide callerid:	OFF
CID signalling:	bell T
DND:	OFF
CID start signal:	ring
▼ Polarity	
Answer on polarity switch:	OFF
Hangup on polarity switch:	OFF
Polarity on answer delay:	600
Delay reply 200 OK switch:	OFF

Figure 3-1-2 FXO Port Configure



Figure 3-1-3 FXS Port Configure

General	
Port type:	FXS
Name:	board1-port5
Rx gain:	3.0
Tx gain:	3.0
Ring timeout:	180
Sip Account:	None •
Failover fxo:	None T
Caller ID	
Caller ID:	8005
Full name:	Channel 8005
CID signalling:	bell T
DND:	OFF
Call feature	
Call waiting:	ON
Three way calling:	ON
Call transfer:	ON THE
Call forward:	No
Call forward number:	

Pickup Settings

Call pickup is a feature used in a telephone system that allows one to answer someone else's telephone call. You can set the "Time Out" and "Number" parameters either globally or separately for each port. The feature is accessed by pressing a special sequence of numbers which you set as "Number" parameter on the telephone set when it is enabled this function.

Figure	3-2-1	Pickup	Configure

Status Settings	
Enable:	
Time Out:	3000
Number:	**8006
5	Disabled Time Out Number
6	Disabled Time Out Number
7	Disabled Time Out Number
8	Enabled Time Out Number
Save	



Options	Definition
Enable	ON(enabled),OFF(disabled)
Time Out	Set the timeout, in milliseconds (ms).Note: You can only enter numbers.
Number	Pickup number

Table 3-2-1 Definition of Pickup

Dial Matching Table

Dialing rules is used to effectively judge whether the received number sequence is complete, in order to timely

end receiving number and send out number

The correct use of dial-up rules, helps to shorten the turn-on time of phone call

Figure 3-3-1 Port Configure

_01[358]XXXXXXXXX	Dial Matching rule may be numbers, letters, or combinations
010XXXXXXXX	thereof. If an rule is prefixed by a '' character, it is
_02XXXXXXXXX	interpreted as a pattern rather than a literal. In
_0[3-9]XXXXXXXXXX	patterns, some characters have special meanings:
_11[02-9]	
_111XX	X - any digit from 0-9
_9[56]XXX	Z - any digit from 1-9
_100XX	N - any digit from 2-9
_10[1-9]	[1235-9] - any digit in the brackets (in this example,
_12[0-24-9]	1, 2, 3, 5, 6, 7, 8, 9)
_1[358]XXXXXXXXX	! - wildcard, causes the matching process to complete
_[235-7]XXXXXXX	as soon as ;it can unambiguously determine that no other
_[48] [1-9] XXXXXX	matches are possible
_[48]0[1-9]XXXXX	
_[48]00XXXXXXX	For example, the rule _NXXXXXX would match normal 7 digit
_#XX	dialings, while _1NXXNXXXXX would represent an area code
_*XX	plus phone number preceded by a one.
##	
_X.	

Advanced Settings



General	
Tone duration:	100
Dial timeout:	180
Codec:	Ulaw •
Impedance:	China
Echo cancel tap length:	512 •
VAD/CNG:	OFF
Flash/Wink:	ON
Min flash time:	40
Max flash time:	400
"#" as Ending Dial Key:	ON
Checking SIP Status:	OFF

Figure 3-4-1 General Configuration

Table 3-4-1 Instruction of General

Options	Definition
Tone duration	How long generated tones (DTMF and MF) will be
	played on the channel. (in milliseconds)
Dial timeout	Specifies the number of seconds we attempt to dial
	the specified devices.
Codec	Set the global encoding : mulaw, alaw.
Impedance	Configuration for impedance.
Echo cancel tap length	Hardware echo canceler tap length.
VAD/CNG	Turn on/off VAD/CNG.
Flash/Wink	Turn on/off Flash/wink.
Max flash time	Max flash time.(in milliseconds).
"#"as Ending Dial Key	Turn on/off Ending Dial Key.
Checking SIP Status	Turn on/off SIP Account registration status checking.



Figure 3-4-2 Caller ID

Caller ID	
The pattern of sending CID:	send CID after first ring
Waiting time before sending CID:	100
Send polarity reversal(DTMF Only):	OFF
Start code(DTMF Only):	
Stop code(DTMF Only):	
Display extension number	OFF

Table 3-4-2 Instruction of Caller ID

Options	Definition
The pattern of sending CID	Some countries(UK) have ring tones with different ring tones(ring-ring), which means the caller ID needs to be set later on, and not just after the first ring, as per the default(1).
Waiting time before sending CID	How long we will waiting before sending the CID on the channel. (in milliseconds).
Sending polarity reversal(DTMF Only)	Send polarity reversal before sending the CID on the channel.
Start code(DTMF Only)	Start code.
Stop code(DTMF Only)	Stop code.

Figure 3-4-3 Hardware Gain

▼ Hardware gain	
FXO Rx gain:	0
FXO Tx gain:	0
FXS Rx gain:	0 •
FXS Tx gain:	0 •



Options	Definition
FXS Rx gain	Set the FXS port Rx gain. Range: from -150 to 120. Select -35, 0 or 35.
FXS Tx gain	Set the FXS port Tx gain. Range: from -150 to 120. Select -35, 0 or 35.

Table 3-4-3 Instruction of Hardware gain

Figure 3-4-4 Fax Configuration

🔻 Fax		
	Mode:	T.38 •
	Rate:	14400 🔻
	Ecm:	OFF

Table 3-4-4 Definition of Fax

Options	Definition
Mode	Set the transmission mode.
Rate	Set the rate of sending and receiving.
Ecm	Enable/disable T.30 ECM (error correction mode) by default.

Figure 3-4-5 Country Configuration

🔻 Coun	try	
	Country:	China
	Ring cadence:	1000,4000
	Dial tone:	450
	Ring tone:	450/1000,0/4000
	Busy tone:	450/350,0/350
	Call waiting tone:	450/400,0/4000
	Congestion tone:	450/700,0/700
	Dial recall tone:	450
	Record tone:	950/400,0/10000
	Info tone:	450/100,0/100,450/100,0/100,450/100,0/100,450/400,0/400
	Stutter tone:	450+425



Table 3-4-5 Definition of Country

Options	Definition
Country	Configuration for location specific tone indications.
Ring cadence	List of durations the physical bell rings.
Dial tone	Set of tones to be played when one picks up the hook.
Ring tone	Set of tones to be played when the receiving end is ringing.
Busy tone	Set of tones played when the receiving end is busy.
Call waiting tone	Set of tones played when there is a call waiting in the background.
Congestion tone	Set of tones played when there is some congestion.
Dial recall tone	Many phone systems play a recall dial tone after hook flash.
Record tone	Set of tones played when call recording is in progress.
Info tone	Set of tones played with special information messages (e.g.,
	number is out of service.)

Special Function Keys

Figure 3-5-1 Function keys

V Function Keys		
	None Keys Blind Transfer:	ON
	Blind Transfer:	
	Asked Transfer:	*38
Sa	ave Cancel	



4. SIP

SIP Endpoints

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

Figure 4-1-1 SIP Status

Endpoint Name	Registration	Credentials	Actions
8000	client	8000@172.16.80.134	2 🗙
8001	client	8001@172.16.80.134	2 🗙
8002	client	8002@172.16.80.134	2 🗙
8003	client	8003@172.16.80.134	🥖 💥
8004	client	8004@172.16.80.134	2 🗙
8005	client	8005@172.16.80.134	2 🗙
8006	client	8006@172.16.80.134	/ 🗙
8007	client	8007@172.16.80.134	2 🗙

Add New SIP Endpoint Delete

```
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.
```

Main Endpoint Settings

There are 3 kinds of registration types for choose. You can choose "Anonymous, Endpoint registers with this gateway or This gateway registers with the endpoint".

You can configure as follows:

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



Figure 4-1-2 Anonymous Registration

Main Endpoint Settings		
	Name:	301 *Allowed character must be any of [0-9a-zA-Z"~1@#\$%^*()_{:!?+-=.], 1 - 32 characters.
	User Name:	Anonymous
	Password:	()
	Registration:	None
	Hostname or IP Address:	172.16.208.33
	Backup Hostname or IP Address:	
	Transport:	UDP •
	NAT Traversal:	Yes T
	SUBSCRIBE for MWI:	No •
	VOS Encryption:	No •

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

Figure 4-1-3 Register to Gateway

Main Endpoint Settings	
Name:	301 *Allowed character must be any of [0-9a-ZA-Z'~I@#\$%^^()_[]:[?+~=.], 1 - 32 characters.
User Name:	301 Anonymous
Password:	· (9)
Registration:	Endpoint registers with this gateway
Hostname or IP Address:	dynamic
Backup Hostname or IP Address:	
Transport:	UDP V
NAT Traversal:	Yes •
SUBSCRIBE for MWI:	No T
VOS Encryption:	No V

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

Figure 4-1-4 Register to Server

V	Main Endpoint Settings	
	Name:	301 *Allowed character must be any of [0-9a-zA-Z'~I@#\$%^*()_{: ?+-=.], 1 - 32 characters.
	User Name:	301 Anonymous
	Password:	··· •
	Registration:	This gateway registers with the endpoint •
	Hostname or IP Address:	172.16.208.63
	Backup Hostname or IP Address:	
	Transport:	UDP V
	NAT Traversal:	Yes •
	SUBSCRIBE for MWI:	No T
	VOS Encryption:	No V



Options	Definition
Name	A name which is able to read by human. And it's only used for user's reference.
Username	User Name the endpoint will use to authenticate with the gateway.
Password	Password the endpoint will use to authenticate with the gateway. Allowed characters.
Registration	NoneNot registering; Endpoint registers with this gatewayWhen register as this type, it means the GSM gateway acts as a SIP server, and SIP endpoints register to the gateway; This gateway registers with the endpointWhen register as this type, it means the GSM gateway acts as a client, and the endpoint should be register to a SIP server;
Hostname or IP Address	IP address or hostname of the endpoint or 'dynamic' if the endpoint has a dynamic IP address. This will require registration.
Transport	This sets the possible transport types for outgoing. Order of usage, when the respective transport protocols are enabled, is UDP, TCP, TLS. The first enabled transport type is only used for outbound messages until a Registration takes place. During the peer Registration the transport type may change to another supported type if the peer requests so.
NAT Traversal	Addresses NAT-related issues in incoming SIP or media sessions. NoUse Rport if the remote side says to use it. Force Rport onForce Rport to always be on. YesForce Rport to always be on and perform comedia RTP handling. Rport if requested and comediaUse Rport if the remote side says to use it and perform comedia RTP handling.

Table 4-1-1 Definition of SIP Options

Advanced: Registration Options



Options	Definition
Authentication User	A username to use only for registration.
Register Extension	When Gateway registers as a SIP user agent to a SIP proxy (provider), calls from this provider connect to this local extension.
From User	A username to identify the gateway to this endpoint.
From Domain	A domain to identify the gateway to this endpoint.
Remote Secret	A password which is only used if the gateway registers to the remote side.
Port	The port number the gateway will connect to at this endpoint.
Quality	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 signaling instead of sending signaling directly to endpoints.
Custom Registery	Custom Registery On / Off.
Enable Outboundproxy to Host	Outboundproxy to Host On / Off.

Table 4-1-2 Definition of Registration Options

Call Settings

Table 4-1-3 Definition of Call Options

Options	Definition
	Set default DTMF Mode for sending DTMF. Default: rfc2833.
DTMF Mode	Other options: 'info', SIP INFO message (application/dtmf-relay);
	'Inband', Inband audio (require 64kbit codec -alaw, ulaw).
Call Limit	Setting a call-limit will cause calls above the limit not to be accepted.



Trust	Whether or not the Remote-Party-ID header should be trusted.
Remote-Party-ID	
Send	Whether er net to cond the Demote Derty ID beader
Remote-Party-ID	Whether or not to send the Remote-Party-ID header.
Remote Party ID	How to set the Remote-Party-ID header: from Remote-Party-ID or
Format	from P-Asserted-Identity.
Caller ID Presentation	Whether or not to display Caller ID.

Advanced: Signaling Settings

Options	Definition
Progress Inband	If we should generate in-band ringing. Always use 'never' to never use in-band signaling, even in cases where some buggy devices might not render it. Valid values: yes, no never. Default: never.
Allow Overlap Dialing	Allow Overlap Dialing: Whether or not to allow overlap dialing. Disabled by default.
Append user=phone to URI	Whether or not to add '; user=phone' to URIs that contain a valid phone number.
Add Q.850 Reason Headers	Whether or not to add Reason header and to use it if it is available.
Honor SDP Version	By default, the gateway will honor the session version number in SDP packets and will only modify the SDP session if the version number change. Turn this option off to force the gateway to ignore the SDP session version number and treat all SDP data as new data. This is

Table 4-1-4 Definition of Signaling Options



	required for devices that send non-standard SDP packets (observed with Microsoft OCS). By default this option is on.
Allow Transfers	Whether or not to globally enable transfers. Choosing 'no' will disable all transfers (unless enabled in peers or users). Default is enabled.
Allow Promiscuous Redirects	Whether or not to allow 302 or REDIR to non-local SIP address. Note that promiscredir when redirects are made to the local system will cause loops since this gateway is incapable of performing a "hairpin" call.
Max Forwards	Setting for the SIP Max-Forwards header (loop prevention).
Send TRYING on REGISTER	Send a 100 Trying when the endpoint registers.

Advanced: Timer Settings

Options	Definition
Default T1 Timer	This timer is used primarily in INVITE transactions. The default for Timer T1 is 500ms or the measured run-trip time between the gateway and the device if you have qualify=yes for the device.
If a provisional response is not received in this amount of time, the call Call Setup Timer auto-congest. Defaults to 64 times the default T1 timer.	
Session Timers	Session-Timers feature operates in the following three modes: originate, Request and run session-timers always; accept, run session-timers only when requested by other UA; refuse, do not run session timers in any case.
Minimum Session Refresh Interval	Minimum session refresh interval in seconds. Default is 90secs.

Table 4-1-5 Definition of Timer Options



Maximum	
Session Refresh	Maximum session refresh interval in seconds. Defaults to 1800secs.
Interval	
Session Refresher	The session refresher, uac or uas. Defaults to uas.

Media Settings

Options	Definition
Madia Sattings	Select codec from the drop down list. Codecs should be different for each
Media Settings	Codec Priority.

Table 4-1-6 Definition of Media Settings

FXS Batch Binding SIP

If you want binding batch Sip accounts to FXS port, you can configure this page. Look out: this is only used when

"This gateway registers with the endpoint" work mode.

Figure 4-2-1	FXS Batcl	h Binding SIP
--------------	-----------	---------------

							Choose	File
							Do	wnload Samples
Port	Port Name	User Name	Password	Hostname or IP Address	Port	VOS Encryption	Codec Priority	Support Codec
				· · · · · · · · · · · · · · · · · · ·		No 🔻	G.711 u-law ▼	All 🔻
1	board-1-port1					No 🔻	G.711 u-law *	All 🔻
2	board-1-port2					No 🔻	G.711 u-law *	All 🔻
3	board-1-port3					No 🔻	G.711 u-law V	All 🔻
4	board-1-port4					No 🔻	G.711 u-law V	All 🔻
5	board-1-port5			·		No 🔻	G.723 V	All 🔻
6	board-1-port6			·		No 🔻	G.711 u-law ▼	Solo 🔻
7	board-1-port7			·		No 🔻	G.711 a-law ▼	All 🔻
8	board-1-port8			· · · · · · · · · · · · · · · · · · ·		No 🔻	G.711 u-law ▼	All 🔻

Save Cancel Batch CAutoPassword



Batch Create SIP

If you want add batch Sip accounts, you can configure this page. You can choose all the register mode.

ID	User Name	Password	Hostname or IP Address	Port	Register Mode
	E	P			client •
1		9			client V
2		(P)			client V
3		P			client V
4		P			client V
5		P			client V
6		9			client V
7		(P)			client V
8		() ()			client •

Figure 4-3-1 Batch SIP Endpoints

Save Cancel Batch 🗹 AutoPassword

Advanced SIP Settings

Networking

Options	Definition	
UDP Bind Port	Choose a port on which to listen for UDP traffic.	
Enable TCP	Enable server for incoming TCP connection (default is no).	
TCP Bind Port	Choose a port on which to listen for TCP traffic.	
TCP Authentication Timeout	The maximum number of seconds a client has to authenticate. If the client does not authenticate before this timeout expires, the client will be disconnected.(default value is: 30 seconds).	
TCP Authentication Limit	The maximum number of unauthenticated sessions that will be allowed to connect at any given time(default is:50).	
Enable Hostname Lookup	Enable DNS SRV lookups on outbound calls Note: the gateway only uses the first host in SRV records Disabling DNS SRV lookups disables the ability to place SIP calls based on domain names to some other SIP users on the Internet specifying a port in a SIP peer definition or when dialing	

Table 4-4-1 Definition of Networking Options



outbound calls with suppress SRV lookups for that peer or call.

NAT Settings

Options	Definition
Local Network	Format:192.168.0.0/255.255.0.0 or 172.16.0.0./12. A list of IP address or IP ranges which are located inside a NATed network. This gateway will replace the internal IP address in SIP and SDP messages with the external IP address when a NAT exists between the gateway and other endpoints.
Local Network List	Local IP address list that you added.
Subscribe Network Change Event	Through the use of the test_stun_monitor module, the gateway has the ability to detect when the perceived external network address has changed. When the stun_monitor is installed and configured, chan_sip will renew all outbound registrations when the monitor detects any sort of network change has occurred. By default this option is enabled, but only takes effect once res_stun_monitor is configured. If res_stun_monitor is enabled and you wish to not generate all outbound registrations on a network change, use the option below to disable this feature.
Match External Address Locally	Only substitute the externaddr or externhost setting if it matches
Dynamic Exclude Static	Disallow all dynamic hosts from registering as any IP address. Used for staticly defined hosts. This helps avoid the configuration error of allowing your users to register at the same address as a SIP provider.
Externally Mapped TCP Port	The externally mapped TCP port, when the gateway is behind a static NAT or PAT
External Address	The external address (and optional TCP port) of the NAT. External Address = hostname[:port] specifies a static address[:port] to be used in SIP and SDP messages.Examples: External Address = 12.34.56.78

Table 4-4-2 Definition of NAT Settings



	External Address = 12.34.56.78:9900
External Hostname	The external hostname (and optional TCP port) of the NAT. External Hostname = hostname[:port] is similar to External Address. Examples: External Hostname = foo.dyndns.net
Hostname Refresh Interval	How often to perform a hostname lookup. This can be useful when your NAT device lets you choose the port mapping, but the IP address is dynamic. Beware, you might suffer from service disruption when the name server resolution fails.

RTP Settings

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 range of port numbers to be used for RTP.
RTP Timeout	

Table 4-4-3 Definition of NAT Settings Options

Parsing and Compatibility

Table 4-4-4 Instruction of Parsing and Compatibility			
Options	Definition		
Strict RFC	Check header tags, character conversion in URIs, and multiline		
Interpretation	headers for strict SIP compatibility(default is yes)		
Send Compact	Sand compact SID headors		
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	The external hostname (and optional TCP port) of the NAT.		

Table 4-4-4 Instruction of Parsing and Compatibility



Methods	
Shrink Caller ID	The shrinkcallerid function removes '(', ' ', ')', non-trailing '.', and
	'-' not in square brackets. For example, the caller id value
	555.5555 becomes 5555555 when this option is enabled.
	Disabling this option results in no modification of the caller id
	value, which is necessary when the caller id represents
	something that must be preserved. By default this option is on.
Maximum	Maximum allowed time of incoming registrations and
Registration Expiry	subscriptions (seconds).
Minimum	Minimum length of registrations/subscriptions (default 60).
Registration Expiry	
Default Registration	Default length of incoming/outgoing registration.
Expiry	
Registration	How often, in seconds, to retry registration calls. Default 20
Timeout	seconds.
Number of	Number of registration attempts before we give up 0 - continue
Registration	Number of registration attempts before we give up. 0 = continue
Attempts Enter '0'	forever, hammering the other server until it accepts the
for unlimited	registration. Default is 0 tries, continue forever.

Security

Table 4-4-5 Instruction of Security

Options	Definition
Match Auth Username	If available, match user entry using the 'username' field from the authentication line instead of the 'from' field.
Realm	Realm for digest authentication. Realms MUST be globally unique according to RFC 3261. Set this to your host name or domain name.



Use Domain as Realm	Use the domain from the SIP Domains setting as the realm. In this case, the realm will be based on the request 'to' or 'from' header and should match one of the domain. Otherwise, the configured 'realm' value will be used.
Always Auth Reject	When an incoming INVITE or REGISTER is to be rejected, for any reason, always reject with an identical response equivalent to valid username and invalid password/hash instead of letting the requester know whether there was a matching user or peer for their request. This reduces the ability of an attacker to scan for valid SIP usernames. This option is set to 'yes' by default.
Authenticate Options	Enabling this option will authenticate OPTIONS requests just like INVITE
Requests	requests are. By default this option is disabled.
Allow Guest Calling	Allow or reject guest calls (default is yes, to allow). If your gateway is connected to the Internet and you allow guest calls, you want to check which services you offer everyone out there, by enabling them in the default context.

Media

Table 4-4-6 Instruction of Media

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



	application in the priority before the app.
TOS for SIP Packets	Sets type of service for SIP packets
TOS for RTP Packets	Sets type of service for RTP packets

Sip Account Security

This analog gateway support TLS protocl for encrypting calls. On the one hand, it can worked as TLS server, generate the session keys used for the secure connection. On the other hand, it also can be registered as a client, upload the key files provied by the server.

Figure	4-5-1	TLS	settings
8			500000

	TLS Setting					
	TLS	Enable: ON				
	TLS Verify	Server: ON				
		Port: 5061	<u>ا</u>			
	TLS Client	Method: tlsv1 •				
_						
	TLS keys					
	,-					
	Type Key Name		IP Address	Organization	Password	Operation
			IP Address	Organization	Password	Operation Create
	Type Key Name		IP Address	Organization		
	Type Key Name Client ▼ Key Files Upload the pem file: 选择3	Z(件) 未选择任何文件 年 未选择任何文件	IP Address	Organization		

Table 4-5-1 Instruction of TLS

Options	Definition				
TLS Enable	Enable or disable DTLS-SRTP support.				
TLS Verify Server	Enable or disable tls verify server(default is no).				
Port	Specify the port for remote connection.				
TLS Client Method	Values include tlsv1, sslv3, sslv2, Specify protocol for outbound client				
	connections, default is sslv2.				

OpenVox 5. Routing

The gateway embraces the flexible and friendly routing settings for user. It supports up to 512 routing rules and about 100 pairs of calleeID/callerID manipulations can be set in a rule. It support DID function The gateway support trunk group and trunk priority management.

Call Routing Rules

Mo	ove	Order	Rule Name	From	То	Actions
	¢	2	test	8006	4	2 🗙
	¢	1	outbound	soft phone	9000	2 🗙
	¢	3	out_test	9004	8005	2 🗙
	¢	4	inbound	9000	soft phone	2 🗙
	•	ting Rule	Delete Save Orders	2000		

Figure 5-1-1 Routing Rules

You are allowed to set up new routing rule by	New Call Routing Rule	, and after setting routing rules, move
rules' order by pulling up and down, click	button to edit the routi	ing and 🗱 to delete it. Finally click
the Save Orders button to save what you	set. Routing Informatio	will show current routing rules.

Otherwise you can set up unlimited routing rules.

There is an example for routing rules number conversion, it transform calling, called number at the same time. Suppose you want eleven numbers start at 159 to call the eleven numbers of start at 136. Calling transform delete the three numbers from left, then writing number 086 as prefix, delete the last four numbers, and then add number 0755 at the end, it will show caller name is China Telecom. Called transform adds 086 as prefix, and Change the last two number to 88.

			8				
processing rules	prepend	prefix	Match pattern	SdfR	StA	RdfR	Caller Name
Calling Transformation	086	159	ххххххх	4	0755		China telecom
Called transformation	086	136	хххххх	2	88		N/A

Figure	5-1	-1
--------	-----	----



You can click

New Call Routing Rule

button to set up your routings.

Figure 5-1-2 Example of Setup Routing Rule

Create a Call Routing Rule	
Call Routing Rule	
Routing Name:	support
Call Comes in From:	fxo-1 T
Send Call Through:	soft phone T
DISA Settings	
Authentication:	OFF
Advance Routing Rule	
Cid Number Settings	
Cid Number	
·	

Save Cancel

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

Options	Definition
Routing Name	The name of this route. Should be used to describe what types of calls this route matches (for example, 'SIP2GSM' or 'GSM2SIP').
Call Comes in From	The launching point of incoming calls.
Send Call Through	The destination to receive the incoming calls.

Table 5-1-2 Definition of Call Routing Rule

Figure 5-1-3 Advance Routing Rule

Advance Routing Rule								
CalleelD/callerID Manipulation								
Callee_Dial_pattern Prepend	+ Prefix	Match Pattern] (- SDfR	+ StA) RdfR			~
Caller_Dial_pattern Prepend	+ Prefix	Match Pattern] (- SDfR	+ StA) RdfR	Caller Name	Modify_CallerID <	~
+ Add More Dial Pattern Fields								
Time Patterns that will use this Rou	ute							
Time to start: - 🔻 : -	•	Week Day start: -	*		Month Day start: - 🔻	Month start: -	۲	~
Time to finish: - 🔻 : -	•	Week Day finish: -	•		Month Day finish: - •	Month finish: -	•	~
+ Add More Time Pattern Fields								
Change Rules								
_								
Forward Number								
Failover Call Through Number								
Add a Failover Call Through Pro	vider							

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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).
	X matches any digit from 0-9
	Z matches any digit from 1-9
	N matches any digit from 2-9
	[1237-9]matches any digit in the brackets (example: 1,2,3,7,8,9)
	. wildcard, matches one or more dialed digits
	Prepend: Digits to prepend to a successful match. If the dialed number matches the
	patterns specified by the subsequent columns, then this will be prepended before
	sending to the trunks.
	Prefix: Prefix to remove on a successful match. The dialed number is compared to this
CalleeID/callerID	and the subsequent columns for a match. Upon a match, this prefix is removed from
Manipulation	the dialed number before sending it to the trunks.
	Mach 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.
	SDfR(Stripped Digits from Right): The amount of digits to be deleted from the right
	end of the number. If the value of this item exceeds the length of the current number,
	the whole number will be deleted.
	RDfR(Reserved Digits from Right): The amount of digits to be resevered from the right
	end of the number. If the value of this item under the length of the current number,
	the whole number will be reserverd.
	StA(Suffix to Add): Designated information to be added to the right end of the current
	number.
	Caller Name: What caller name would you like to set before sending this call to the

Table 5-1-3 Definition of Advance Routing Rule



	endpoint.
	Disabled Caller Number Change : Disable the caller number change, and fixed caller
	number match pattern.
Time Patterns	
that will use this	Time Patterns that will use this Route help
Route	
Forward	What destination number will you dial?
Number	This is very useful when you have a transfer call.
Failover Call	
Through	The gateway will attempt to send the call out each of these in the order you specify.
Number	

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

Figure 5-2-1 Group Rules

Group Name	Туре	Policy	Members	Actions
all	fxo	roundrobin	1, 2, 3, 4, 5, 6, 7, 8	2 🗙
SIP	sip	roundrobin	987, 1002, 980, 981, 982, 983, 984, 985, 986	2 🗙
New Group				
u can click	N	lew Gro		
			button to set new group, and if you want to modify exist	ed grou



Figure 5-2-2 Create a Group

Create a Group

Group Name:	23
Туре:	SIP •
Policy:	Reverse Roundrobin
Members	NO. All 1 sip-8005 2 sip-8006 3 sip-8007 4 sip-8008 5 sip-9001 6 sip-9002 7 Ø sip-9003 8 sip-9004 9 sip-9000

Figure 5-2-3 Modify a Group

Modify a Group

Routing Groups	
Group Name:	all
Туре:	FXO T
Policy:	Least Recent(*experiment) ▼
Members	NO. All 1

Save Cancel

Table 5-2-1 Definition	of Routing Groups
------------------------	-------------------

Options	Definition
Group Name	The mean of this route. Should be used to describe what types of calls
	this route match (for example, 'sip1 TO port1' or 'port1 To sip2').

Batch Create Rules

If you bind telephone for each FXO port and want to establish separate call routings for them. For convenience,

you can batch create call routing rules for each FXO port at once in this page.



Figure 5-3-1 Batch Create Rules

Port	Forward Number	Sip Endpoint	CallerID
FXO-1	L L	None	
FXO-2		None	
FXO-3		None •	
FXO-4		None •	

Save Cancel



On "Network" page, there are "Network Settings", "VPN Settings", "DDNS Settings", and "Toolkit".

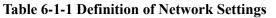
Network Settings

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

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

LAN IPv4	
Interface:	eth0
Туре:	Static •
MAC:	A0:98:05:01:51:76
IPv4 Settings	
Address:	172.16.80.16
Netmask:	255.255.0.0
Default Gateway:	172.16.0.1
DNS Servers	
DNS Server 1:	202.96.134.133
DNS Server 2:	202.96.128.166
DNS Server 3:	8.8.8.8
DNS Server 4:	
Reserved Access IP	
Enable:	ON
Reserved Address:	192.168.99.1
Reserved Netmask:	255.255.255.0

Figure 6-1-1 LAN Settings Interface



Options	Definition
---------	------------

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Interface	The name of network interface.
	The method to get IP.
	Factory: Getting IP address by Slot Number (System $ ightarrow$
Туре	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.
Netmask	The subnet mask of your gateway.
Default Gateway	Default getaway IP address.
	A reserved IP address to access in case your gateway IP
Reserved Access IP	is not available. Remember to set a similar network
	segment with the following address of your local PC.
Enable	A switch to enable the reserved IP address or not.
	ON(enabled), OFF(disabled)
Reserved Address	The reserved IP address for this gateway.
Reserved Netmask	The subnet mask of the reserved IP address.

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

Figure 6-1-2 DNS Interface

DNS Servers	
DNS Server 1:	221.179.38.7
DNS Server 2:	
DNS Server 3:	
DNS Server 4:	



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

Table 6-1-2 Definition of DNS Settings

VPN Settings

You can upload the VPN client configuration, if success, you can see a VPN virtual network card on SYSTEM status

page. About the configure format you can refer to the Notice and Sample configuration.

Figure 6-2-1 VPN Interface

VPN Settings	
VPNType: OpenVPN •	
OpenVPN Settir None: None.	
Upload Configuration: 选择文件 未选择任何文件	File Upload
Notice:	
1. The format of the upload file should be like this xxxx.tar.gr:	
2. The postfix of configuration files should be .conf;	
3. The upload file can not include any directory;	
4. If still confused please download the sample configuration and refer to it;	1.
Sample Configuration	Download Samples

Save

DDNS Settings

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

Figure 6-3-1 DDNS Interface

DDNS Settings	
DDNS:	ON
Туре:	inadyn 💌
Username:	admin
Password:	admin
Your domain:	www.internet.site.com



Table 6-3-1 Definition of DDNS Settings

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

Toolkit

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

Figure 6-4-1 Network Connectivity Checking

Interface: LAN 💌	
google.com	Ping
google.com	Traceroute

Output

	ping -l 172.16.179.1 -c 4 google.com
PING google.com (173.194.72.101) from 172.16.179.1: 56 d 64 bytes from 173.194.72.101: icmp_seq=1 ttl=46 time=596. 64 bytes from 173.194.72.101: icmp_seq=3 ttl=46 time=600. google.com ping statistics 4 packets transmitted, 2 packets received, 50% packet loss round-trip min/avg/max = 596.6/598.5/600.5 ms	6 ms
	Result
Successfully ping [google.com] .	

Figure 6-4-2 Channel Recording

Channel Recording		
In	nterface:	eth0 •
Source	ce host:	
Destinatio	on host:	172.16.208.33
	Port:	5060
<u>c</u>	Channel:	1 •
Tcpdump Option Pa	aramater	UDP Add a Tcpdump paramter option
Start		



Figure	6-4-3	Canture	Network Data
riguit	0-4-5	Capture	

Interface: LAN V		Capture Network Data	
google.com 🗎 Ping		66 65	
google.com Trac	ceroute	00:02	
Channel Recording		The maximum duration of this recording is	
Interface:	eth0 🔻	3 minutes, and the system will stop and	
Source host:		download the recording file automatically when time is up	
Destination host:	172.16.208.33		
Port:	5060		
Channel:	1 •		
Tcpdump Option Paramater	UDP V Add a Tcpdu	Stop Capture	

Table 6-4-1	Definition	of Channel	Recording
	Deminion	or channer	ite cor ang

Options	Definition
Interface	The name of network interface.
Source host	Capture the data of source host you specified
Destination host	Capture the data of destination host you specified
Port	Capture the data of port you specified
Channel	Capture the data of channel you specified
Tcpdump Option Parameter	The tool of tcpdump capture network data by parameter option specified.



7. Advanced

Asterisk API

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

General	
Enabled:	ON
Port:	5038
Manager	
Manager Name:	admin
Manager secret:	admin
Deny:	0.0.0.0/0.0.0.0
Permit:	172.16.123.123/255.255.0.0&192.168.1.0/2
Rights	
System:	read: 🕅 write: 🕅
Call:	read: 🗹 write: 🗹
Log:	read: 🗹 write: 🗹
Verbose:	read: 🔽 write: 🗹

Figure 7-1-1 API Interface

Table 7-1-1 Definition of Asterisk API

Options	Definition
Port	Network port number
Manager Name	Name of the manager without space
Manager secret	Password for the manager. 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 or 192.168.1.0/255.255.0&10.0.0/255.0.0.0

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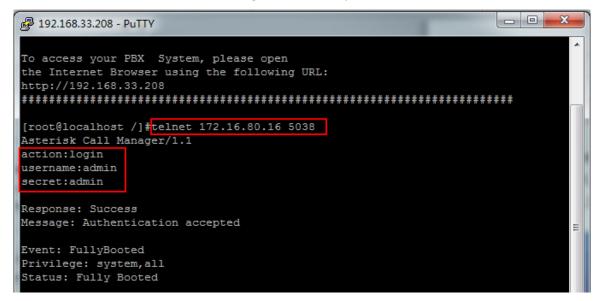


	If you want to permit many hosts or network, use char					
Permit	& as separator.Example: 0.0.0.0/0.0.0.0 or 192.168.1.0/255.					
	255.255.0&10.0.0/255.0.0.0					
	General information about the system and ability to run system					
System	management commands, such as Shutdown, Restart, and					
	Reload.					
Coll	Information about channels and ability to set information in a					
Call	running channel.					
Log	Logging information. Read-only. (Defined but not yet used.)					
Verbose	Verbose information. Read-only. (Defined but not yet used.)					
Command	Permission to run CLI commands. Write-only.					
Agent	Information about queues and agents and ability to add queue					
	members to a queue.					
User	Permission to send and receive UserEvent.					
Config	Ability to read and write configuration files.					
DTMF	Receive DTMF events. Read-only.					
Reporting	Ability to get information about the system.					
CDR	Output of cdr, manager, if loaded. Read-only.					
Dialplan	Receive NewExten and Varset events. Read-only.					
Originate	Permission to originate new calls. Write-only.					
All	Select all or deselect all.					

Once you set like the above figure, the host 172.16.80.16/255.255.0.0 is allowed to access the gateway API. Please refer to the following figure to access the gateway API by putty. 172.16.80.16 is the gateway's IP, and 5038 is its API port.



Figure 7-1-2 Putty Access



Asterisk CLI

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



Asterisk CLI		
Command:	?	Execute

Output:

! Execute a shell command	
agi dump html Dumps a list of AGI commands in HTML format	
agi exec Add AGI command to a channel in Async AGI	
agi set debug [on off] Enable/Disable AGI debugging	
agi show commands [topic] List AGI commands or specific help	
aoc set debug enable cli debugging of AOC messages	
cc cancel Kill a CC transaction	
cc report status Reports CC stats	
cdr show status Display the CDR status	
cel show status Display the CEL status	
channel request hangup Request a hangup on a given channel	

Table 7-2-1	Definition	of Asterisk API
-------------	------------	-----------------

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



If you type "help" or "?" and execute it, the page will show you the executable commands.

Asterisk File Editor

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

Click the file to edit.

Configuration Files	
File Name	File Size
asterisk.conf	247
astmanproxy.conf	440
capture_channel.conf	0
cdr.conf	572
chan dahdi.conf	240
dahdi-channels.conf	2982
dahdi startup.conf	78
dismar.conf	245
extensions.conf	195
extensions_dialmatchingrules.conf	927
1 2 3 4 ▶ 1 / 4 go New Configuration File Reload Asterisk	

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



Log Settings

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 🔻
Asterisk Logs	
Verbose:	OFF
Notice:	OFF
Warning:	OFF
Debug:	OFF
Error:	OFF
DTMF:	OFF
Auto clean:	ON maxsize : 100KB 🔻
SIP Logs	
SIP Logs:	OFF
Auto clean:	ON maxsize : 100KB 🔻
Call Detail Record	
Call Detail Record:	OFF
Auto clean:	ON maxsize : 20MB 🔻

Figure 8-1-1 Logs Settings



Table 8-1-1 Definition of LOG

Options	Definition
System Logs	Whether enable or disable system log.
Auto clean (System Logs)	switch on : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. switch off : logs will remain, and the file size will increase gradually. default on, max size=1MB.
Verbose	Asterisk console verbose message switch.
Notice	Asterisk console notice message switch.
Warning	Asterisk console warning message switch.
Debug	Asterisk console debug message switch.
Error	Asterisk console error message switch.
DTMF	Asterisk console DTMF info switch.
Auto clean: (asterisk logs)	<pre>switch on : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. switch off : logs will remain, and the file size will increase gradually. default on, max size=100KB.</pre>
SIP Logs:	Whether enable or disable SIP log.
Auto clean: (SIP logs)	switch on : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. switch off : logs will remain, and the file size will increase gradually. default on, default size=100KB.
Call Detail Record	Displaying Call Detail Records for each channel.
Auto clean: (Call Detail Record)	switch on : when the size of log file reaches the max size, the system will cut a half of the file. New logs will be retained. switch off : logs will remain, and the file size will increase gradually. default on, default size=20MB.



System Logs	
[1970/01/01 00:00:08] Auto restore configuration files	
[1970/01/01 07:01:20] Power on	
[2017/12/09 15:47:03] Restart asterisk from crontab.	
[2017/12/12 10:11:43] Power off	
[1970/01/01 07:01:17] Power on	
[2017/12/12 15:51:00] Restore configuration files	
[2017/12/12 15:51:10] Power off	
[1970/01/01 00:00:08] Auto restore configuration files	
[1970/01/01 07:01:19] Power on	
[1970/01/01 07:01:20] Power on	
[1970/01/01 07:26:44] System Update	
[1970/01/01 07:26:59] Power off	
[1970/01/01 07:00:10] Auto update configuration files, remain a few files.	
[1970/01/01 07:01:27] Power on	
[2017/12/13 11:10:05] Power off	
[1970/01/01 07:01:22] Power on	
[2017/12/13]347f3] Restore configuration files [2017/12/13]347f50] Power off	
[2017/12/13 IS:41:00] Fower off [1970/01/01 00:00:08] Auto restore configuration files	
[1970/01/01 07:01:55] Power on	
[2017/12/14 18:29:39] Restore configuration files	
[1970/010 00:00:08] Auto restore configuration files	
[1970/01/01 07:01:33] Power on	
[2017/12/18 15:44:08] Power on	
[2017/12/18 15:53:56] System Update	
[2017/12/18 15:54:15] Power off	
[2017/12/18 15:55:35] Power on	
[2017/12/20 11:42:01] Power on	Ŧ
	1
Refresh Rate: Off ▼ Refresh Clean Up	

Figure 8-2-1 System Logs Output

Notice: The same to Asterisk Logs and SIP Logs.

CDR

You can scan every call detail records in this page. We also provide the filter for you to search some specific records.

Caller ID		Callee ID	From	То	Start Time		Duration		Result			
					from	to	from	to	All			
ilte	r Clean Filter											
ta	Records: 281											
	🔷 Caller ID	🔷 Callee ID	🜲 From	🜲 То	🜲 Start Time		Duratio	n	🜲 Result			
	8888	8008	8008	fxs-8	2017-12-13 17:43	36	00:00:00	00:00:00		00:00:00		
	8888	8005	8005	fxs-5	2017-12-13 17:43	33	00:00:00		NO ANSWER			
	8888	8008	8008	fxs-8	2017-12-13 16:35	:11	00:00:00		NO ANSWER			
	8888	8008	8008	fxs-8	2017-12-13 16:33	40	00:00:00		NO ANSWER			
	8888	8008	8008	fxs-8	2017-12-13 16:31	:51	00:00:00		NO ANSWER			
	8888	9001	fxo-1	9001	2017-12-12 15:45	:16	00:00:00		00:00:00		NO ANSWER	
	8888	9001	fxo-1	9001	2017-12-12 15:44	:57	00:00:00		BUSY			
	12345	8888	12345	9000	2017-12-12 15:43	:11	00:00:01		ANSWERED			
	8888	9002	fxo-2	9002	2017-12-12 15:25	44	00:00:10		ANSWERED			
	8888	9001	fxo-1	9001	2017-12-12 15:25	:36	00:00:00		NO ANSWER			

Delete Clean Up Export