VS-GW1200-4G
User Manual

V2.2

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Contents

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Chapter 1 Overview

1.1 What is VS-GW1200-4G?

There are two GSM gateway models with VoxStack series GSM Gateway, VoxStack is independent development technology by OpenVox, the VS-GW1200-4G and VS-GW1600. There are 4 GSM channels in VS-GW1200-4G. The Modular Design GSM Gateways are ranging from 4 up to 20 GSM channels, developed for interconnecting a wide selection of codecs and signaling protocol, including G.711A, G.711U, G.729, G.722, G.726, GSM & SIP, to the GSM cellular networks to quickly reduce communication expenses and maximize cost-savings. With the unique design of the VoxStack gateway, it can support hot-swap for both SIM cards and GSM gateway modules. Users can simply add or remove the modules for hardware expansion or exchange.

The VoxStack gateway designs with a LAN switch board that provides stack ability on the hardware upgrade. It supports SMS messages sending, receiving, group sending and SMS to E-mail. The GSM gateway will be 100% compatible with such as Asterisk, Elastix, trixbox, 3CX and FreeSWITCH SIP server.

1.2 Sample Application

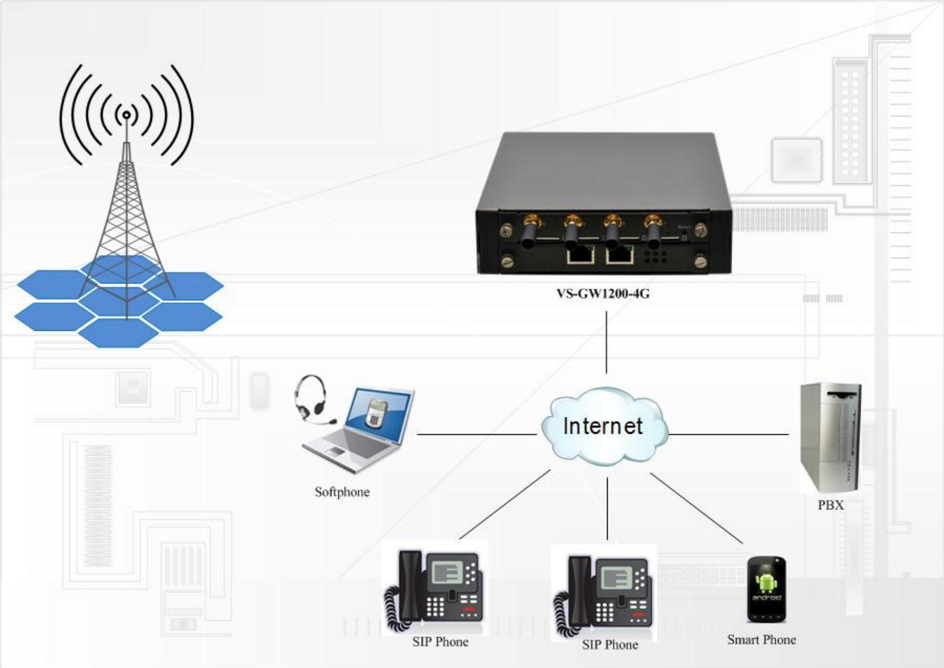


Figure 1 Sample application

1.3 Main Features

- Modular and VoxStack design
- Based on Asterisk®
- 4 default SIP endpoints and simple routings
- Editable Asterisk® configuration file
- Wide selection of codecs and signaling protocol
- SIM cards and modules are all hot-swap
- Support SMS sending, receiving, grouping sending
- Support transferring SMS to E-mail
- Support checking phone number
- Support SIP1.0/2.0， enhanced SIP options
- Support USSD service
- Support IMEI modification
- Be compatible with Asterisk, Elastix, 3CX, FreeSWITCH SIP Server
- Stable performance, flexible dialing, friendly GUI

Chapter 2 Hardware

2.1 Physical Information

Weight: 908g

Size: 15cm*19cm*4.5cm

Frequency: GSM 850/900/1800/1900MHz

Temperature: -40~125 ℃ (Storage)

0~50 ℃ (Operation)

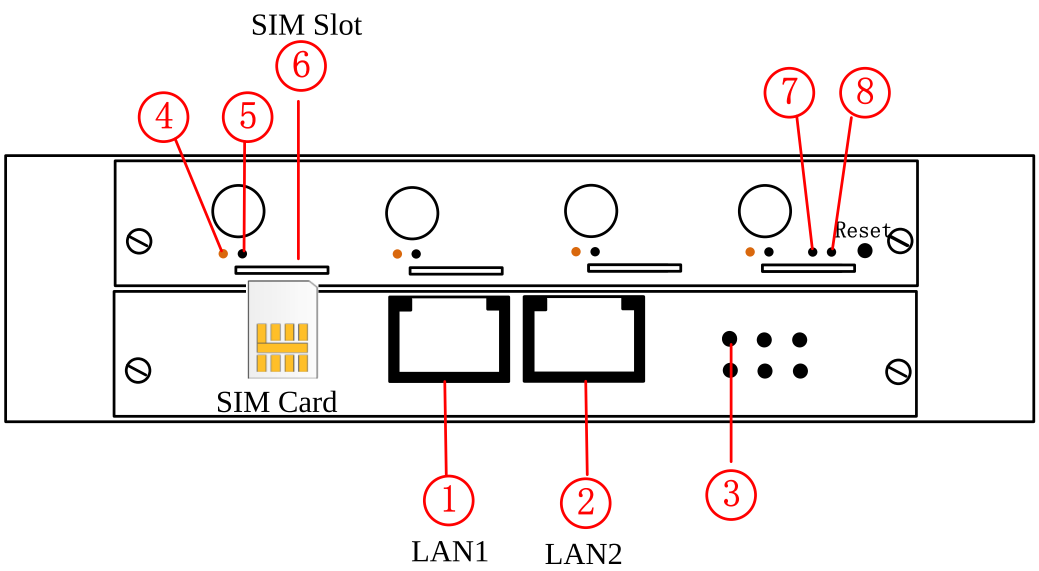
Operation humidity: 10%~90% non-condensing

Power source: 12V DC/4A

Max power: 32W

LAN port: 2

2.2 Front Panel



LED Indicator	Color	Status
③Network Status LED	Green and Flash	Network Connected
④Signal Status LED	Green and Flash	Module Initiating
	Red and Flash	No SIM Card
	Red and No-flash	Worst Signal Quality
	Yellow and No-flash	Medium Signal Quality
	Green and No-flash	Best Signal Quality
⑤Call Status LED	Flash (0.25s)	Communicating
	Blind	Normal
⑦Running Status LED	Green and Flash(0.5s)	Work Normally
⑧Power Indicator	Always Green	Supply Power
During reset, all LED indicators flash.		

Chapter 3 Software

Defaut IP: 172.16.99.1
Username: admin
Password: admin
Please enter the default IP in your browser to scan and configure.

3.1 System

3.1.1 Status

In the "Status" page, you will find GSM, SIP, Routing, Network information.

GSM Information

Port	Signal	Carrier	Registration Status	GSM Status
gsm-1		CHINA MOBILE	Registered (Home network)	READY
gsm-2		CHINA MOBILE	Registered (Home network)	READY
gsm-3		CHINA MOBILE	Registered (Home network)	READY
gsm-4		CHINA MOBILE	Registered (Home network)	READY

SIP Information

Endpoint Name	User Name	Host	Registration	SIP Status
1001	1001		server	
1002	1002		server	
1003	1003		server	
1004	1004		server	

Routing Inforamtion

Rule Name	From	To
SIP1_GSM1	1001	gsm-1
GSM1_SIP1	gsm-1	1001
SIP2_GSM2	1002	gsm-2
GSM2_SIP2	gsm-2	1002
SIP3_GSM3	1003	gsm-3

3.1.2 Time

Options	Definition
System Time	Your GSM gateway current 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 timezone strings.
NTP Server 1	Time server domain or hostname. For example, [time.asia.apple.com].
NTP Server 2	The first reserved NTP server. For example, [time.windows.com].
NTP Server 3	The second reserved NTP server. For example, [time.nist.gov].
Auto-Sync from NTP	Whether enable automatically synchronize from NTP server or not. On is enable, OFF is disable this function.
Sync from NTP	Sync time from NTP server.
Sync from Client	Sync time from local machine.

3.1.3 Login Settings

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

Options	Definition
User Name	Input your username without space here. Allowed characters "-_+. < >&0-9a-zA-Z". Length: 1-32 characters.
Password	Allowed characters "-_+. < >&0-9a-zA-Z". Length: 4-32 characters.
Confirm Password	Please input the same password as 'Passoword' above.

3.1.4 Tools, Information

In the “Tools” pages, there are reboot, update, upload, download, reset toolkits.

Reboot Tools

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.

Information

In the “Information” page, there is some basic information about the GSM gateway, such as model name, version, and stack number.

3.2 GSM

3.2.1 GSM Settings

Port	Carrier	Registration Status	GSM Status	Action
gsm-1	CHINA MOBILE	Registered (Home network)	READY	
gsm-2	CHINA MOBILE	Registered (Home network)	READY	
gsm-3	CHINA MOBILE	Registered (Home network)	READY	
gsm-4	CHINA MOBILE	Registered (Home network)	READY	

In this page, you can see your GSM modules’ status, and click action button to configure the port.

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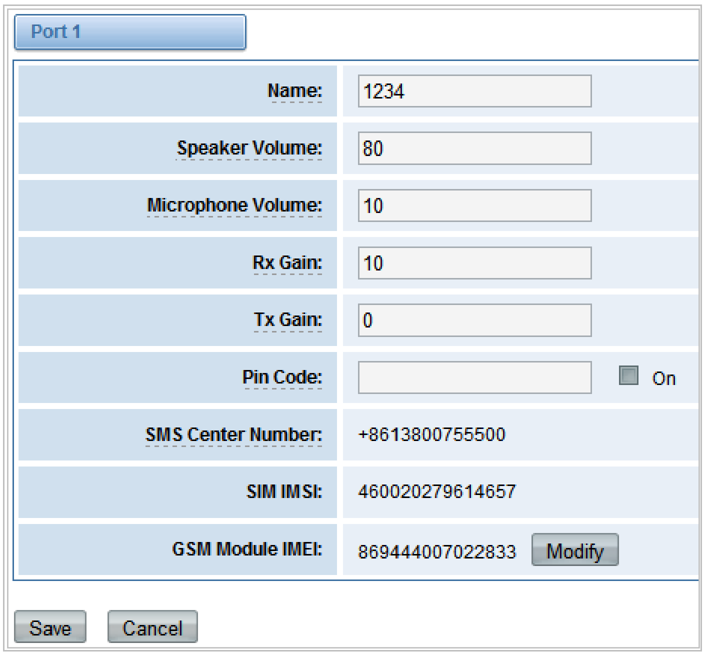
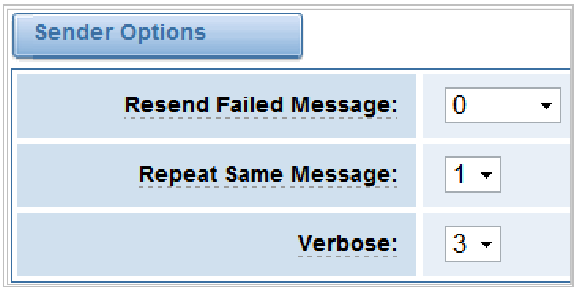


Options	**Definition**
Name	The alias of the GSM port. Input the name without space here. Allowed characters "-_+.<>&0-9a-zA-Z".Length: 1-32 characters.
Speaker Volume	The speaker volume level, the range is 0-100. This will adjust the loud speaker volume level by an AT command.
Microphone Volume	The microphone volume, range is: 0-15. This will change the microphone gain level by an AT command.
RX Gain	The Asterisk received gain. The range is: -100 to +100 (accepted range). It's recommended to never put the values to more than -11 to +11. And in some case anything outside of the -5 and +5 range will actually cause audio loss.
TX Gain	The Asterisk sent gain. The range is: -100 to +100. The same as "Rx Gain".
PIN Code	Personal identification numbers of SIM card. PIN code can be modified to prevent SIM card from being stolen.
SMS Center Number	Your SMS center number of your local carrier.
GSM Module IMEI	You can click “Modify” button and input a new to modify it.

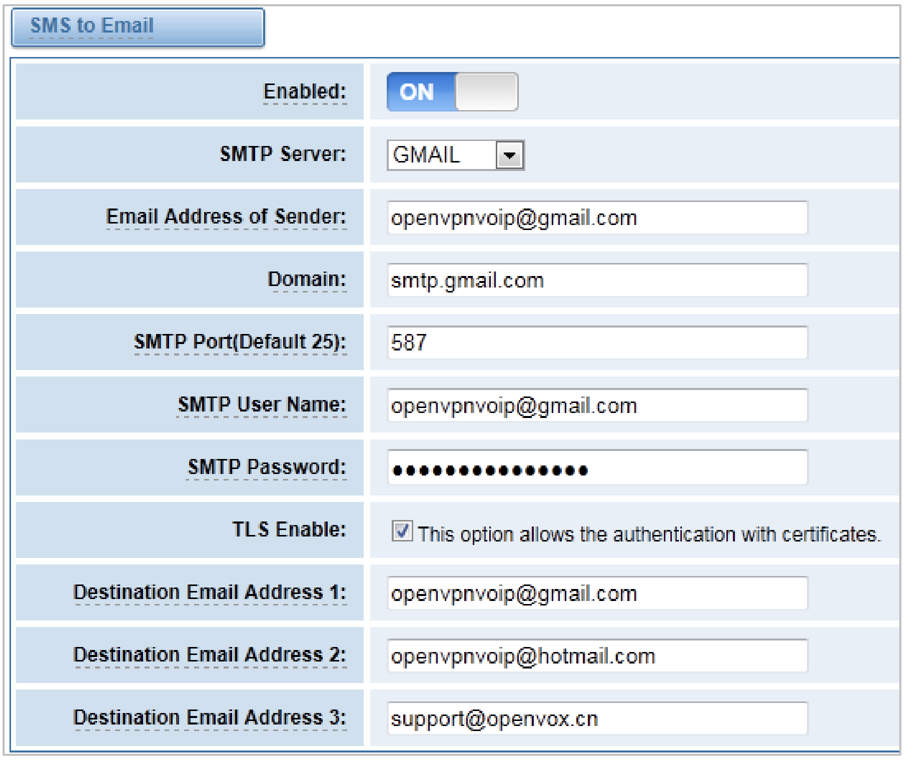
3.2.2 SMS Settings

Sender Options



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

SMS to E-mail: This is a tool which gets the help of your available email account to transmit the inbox SMS to other email boxes. The following settings realize that received SMS through openvpnvoip@gmail.com transmit to openvpnvoip@yahoo.com.cn, openvpnvoip@hotmail.com and support@openvox.cn

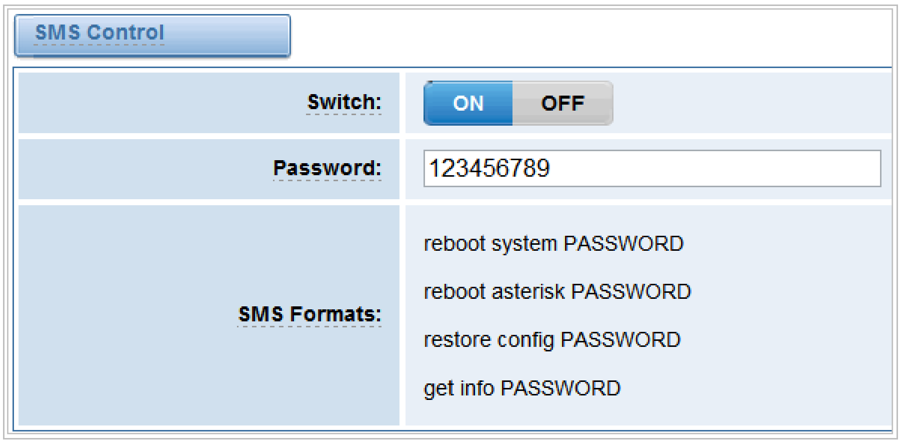


E-mail Box Type	SMTP Server	SMTP Port	SMTP Security Connectivity
Gmail	smtp.gmail.com	587	✓
HotMail	smtp.live.com	587	✓
Yahoo!	smtp.mail.yahoo.co.in	587	✗
Other: 163 free e-mail	smtp.163.com	25	✗

Options	Definition
Enable	When you choose on, the following options are available, otherwise, they are unavailable.
Email Address of Sender	To set the email address of an available email account. For example, openvpnvoip@gmail.com.
Domain	To set outgoing mail server. e.g. smtp.gmail.com
SMTP Port	To set port number of outgoing mail server. (Default is 25)

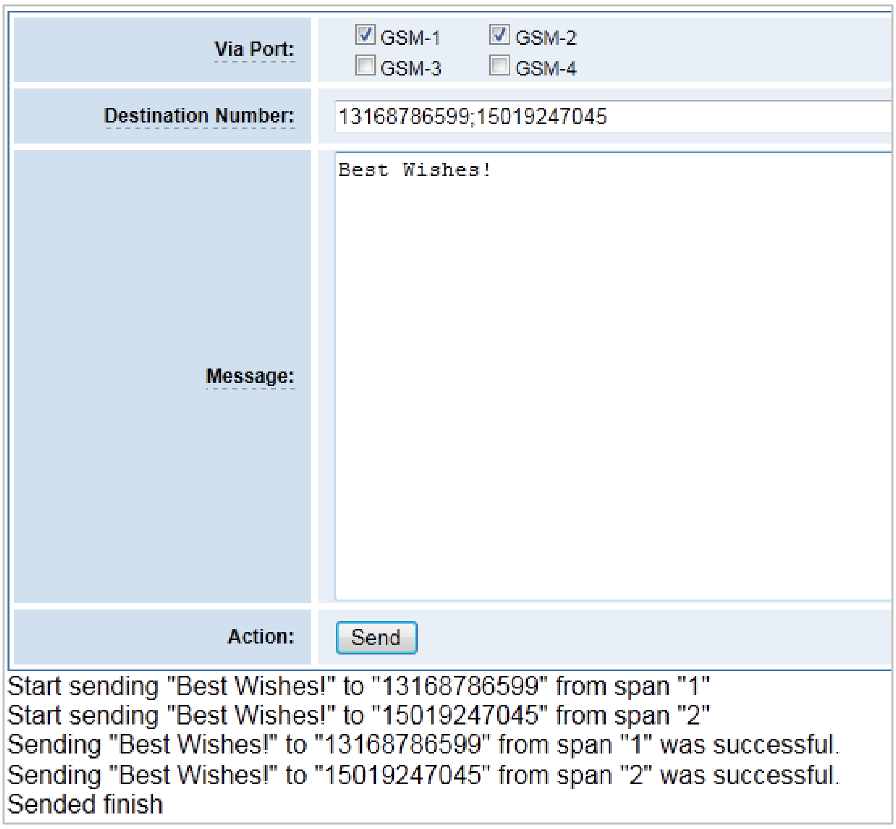
SMTP User Name	The login name of your existing email account. This option might be different from your email address. Some email client doesn't need the email postfix.
SMTP Password	The secret to login your existing email.
TLS Enable	When you choose Yahoo and 163 free e-mails, this option is not available.
SMTP Server	To set outgoing mail server. e.g. mail.openvox.cn.
Destination Email Address	The email address to receive the inbox message.

SMS Control: Allowing endpoints to send some specific KEY WORDS and corresponding PASSWORD to operate the gateway. In default, this function is disabled.



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

3.2.3 SMS Sender

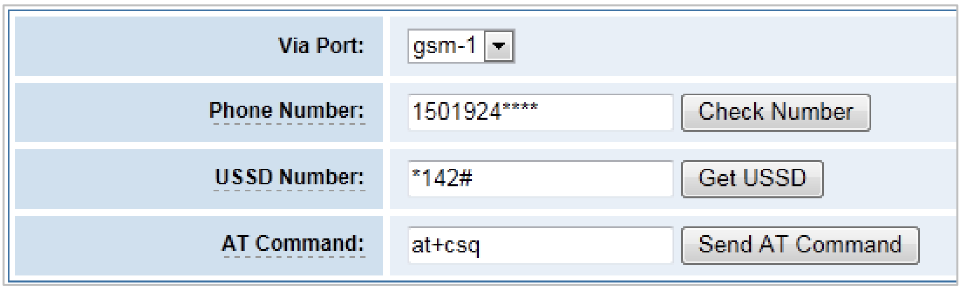


Choose one or more ports to send SMS to the destination number, different number should be separated by symbols: '\r', '\n', space character, semicolon, and comma.

3.2.4 SMS Inbox

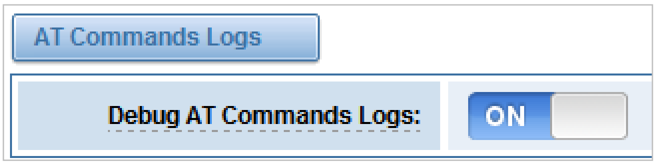
In this page, you are allowed to scan, delete, clean up, and export each port’s received SMS. Also you are allowed to check messages by port, phone number, and time order.

3.2.5 Toolkit



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

Goto the "Logs→AT Commands" for checking result if you have set like the following in "Logs→AT Commands Logs".

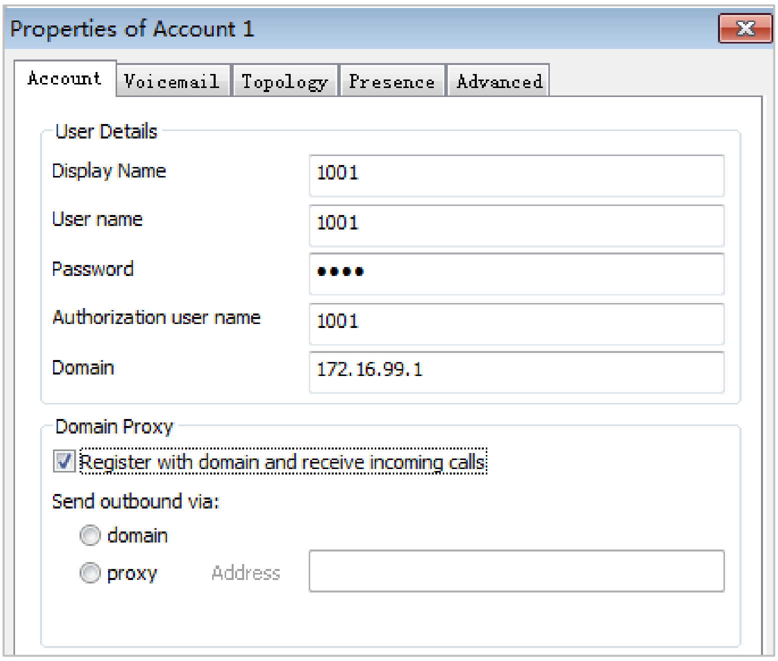


3.3 SIP

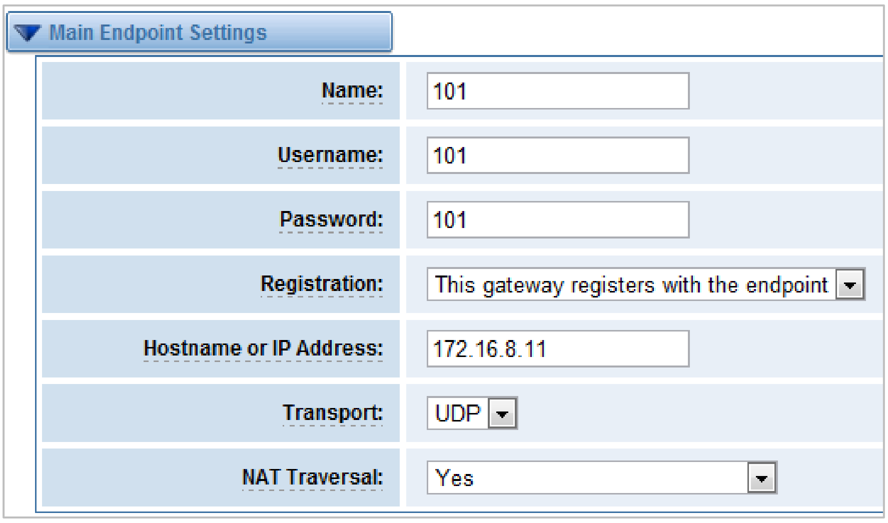
3.3.1 SIP Endpoints

Endpoint Name	Registration	Credentials	Actions
1001	server	1001	
1002	server	1002	
1003	server	1003	
1004	server	1004	

In default, there are four SIP endpoints, if you want to use them, please register with your SIP software. For example, click  bottom, and then open your SIP software such as Xlite



Click the "Add New SIP Endpoint"



Options	**Definition**
Name	Display name.
Username	Register name in your SIP server.
Password	Authenticating with the gateway and characters are allowed.
Registration	**None**---Not registering; **Endpoint registers with this gateway**---When register as this type, it means the GSM gateway acts as a SIP server, and SIP endpoints register to the gateway; **This gateway registers with the endpoint**---When register as this type, it means the GSM gateway acts as a client, and the endpoint should be register to a SIP server;
Hostname or IP Address	IP address or hostname of the endpoint or 'dynamic' if the endpoint has a dynamic IP address. This will require registration.
NAT Traversal	**No**---Use rport if the remote side says to use it. **Force rport on**---Force rport to always be on. **Yes**---Force rport to always be on and perform comedia RTP handling. **Rport if requested and comedia**---Use rport if the remote side says to use it and perform comedia RTP handling.

Advanced: Registration Options

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.

Call Settings

Options	Definition
DTMF Mode	Set default dtmf mode for sending DTMF. Default: rfc2833. Other options: 'info', SIP INFO message (application/dtmf-relay); 'inband', Inband audio (require 64kbit codec -alaw, ulaw).
Trust Remote-Party-ID	Whether or not the Remote-Party-ID header should be trusted.
Send Remote-Party-ID	Whether or not to send the Remote-Party-ID header.
Remote Party ID Format	How to set the Remote-Party-ID header: from Remote-Party-ID or from P-Asserted-Identity.
Caller ID Presentation	Whether or not to display Caller ID.

Advanced: Signaling Settings

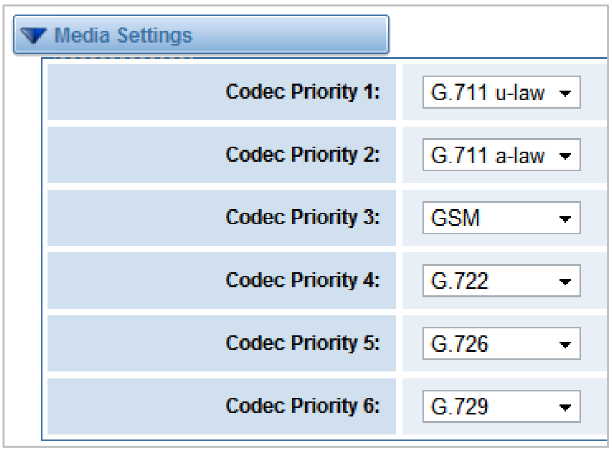
Options	Definition
Progress Inband	Set default dtmf mode for sending DTMF. Default: rfc2833. Other options: 'info', SIP INFO message (application/dtmf-relay); 'inband', Inband audio (require 64kbit codec -alaw, ulaw).
Allow Overlap Dialing	Whether or not the Remote-Party-ID header should be trusted.
Append user=phone to URI	Whether or not to send the Remote-Party-ID header.
Add Q.850 Reason Headers	How to set the Remote-Party-ID header: from Remote-Party-ID or from P-Asserted-Identity.
Honor SDP Version	Whether or not to display Caller ID.
Allow Transfers	Whether or not to globally enable transfers. Choosing 'no' will disable all transfers (unless enabled in peers or users). Default is enabled.
Allow Promiscuous Redirects	Whether or not to allow 302 or REDIR to non-local SIP address. Note that promiscredir when redirects are made to the local system will cause loops since this gateway is incapable of performing a "hairpin" call.
Max Forwards	Setting for the SIP Max-Forwards header (loop prevention).
Send TRYING on REGISTER	Send a 100 Trying when the endpoint registers.
Outbound Proxy	A proxy to which the gateway will send all outbound signaling instead of sending signaling directly to endpoints.

Advanced: Timer Settings

Options	Definition

Default T1 Timer	This timer is used primarily in INVITE transactions. The default for Timer T1 is 500ms or the measured run-trip time between the gateway and the device if you have qualify=yes for the device.
Call Setup Timer	If a provisional response is not received in this amount of time, the call will auto-congest. Defaults to 64 times the default T1 timer.
Session Timers	Session-Timers feature operates in the following three modes: originate, Request and run session-timers always; accept, run session-timers only when requested by other UA; refuse, do not run session timers in any case.
Minimum Session Refresh Interval	Minimum session refresh interval in seconds. Default is 90secs.
Maximum Session Refresh Interval	Maximum session refresh interval in seconds. Defaults to 1800secs.
Session Refresher	The session refresher, uac or uas. Defaults to uas.

Media Settings: Select codec from the drop down list. Codecs should be different for each Codec Priority.



3.3.2 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 outbound calls with suppress SRV lookups for that peer or call.
Enable Internal SIP Call	Whether enable the internal SIP calls or not when you select the registration option "Endpoint registers with this gateway".
Internal SIP Call Prefix	Specify a prefix before routing the internal calls.
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.

Advanced: NAT Settings

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

Parsing and Compatibility

Options	Definition
Strict RFC Interpretation	Check header tags, character conversion in URIs, and multiline headers for strict SIP compatibility(default is yes)
Send Compact Headers	Send compact SIP headers
SDP Owner	Allows you to change the username filed in the SDP owner string. This filed MUST NOT contain spaces.
Disallowed SIP Methods	The external hostname (and optional TCP port) of the NAT.
Shrink Caller ID	The shrinkcallerid function removes '(', ' ', ')', non-trailing '.', and '-' not in square brackets. For example, the caller id value 555.5555 becomes 5555555 when this option is enabled. Disabling this option results in no modification of the caller id value, which is necessary when the caller id represents something that must be preserved. By default this option is on.
Maximum Registration Expiry	Maximum allowed time of incoming registrations and subscriptions (seconds).
Minimum Registration Expiry	Minimum length of registrations/subscriptions (default 60).
Default Registration Expiry	Default length of incoming/outgoing registration.
Registration Timeout	How often, in seconds, to retry registration calls. Default 20 seconds.
Number of Registration Attempts Enter '0' for unlimited	Number of registration attempts before we give up. 0 = continue forever, hammering the other server until it accepts the registration. Default is 0 tries, continue forever.

Security

Options	**Definition**
Match Auth Username	If available, match user entry using the 'username' field from the authentication line instead of the 'from' field.
Realm	Realm for digest authentication. Realms MUST be globally unique according to RFC 3261. Set this to your host name or domain name.
Use Domain as Realm	Use the domain from the SIP Domains setting as the realm. In this case, the realm will be based on the request 'to' or 'from' header and should match one of the domain. Otherwise, the configured 'realm' value will be used.
Always Auth Reject	When an incoming INVITE or REGISTER is to be rejected, for any reason, always reject with an identical response equivalent to valid username and invalid password/hash instead of letting the requester know whether there was a matching user or peer for their request. This reduces the ability of an attacker to scan for valid SIP usernames. This option is set to 'yes' by default.
Authenticate Options Requests	Enabling this option will authenticate OPTIONS requests just like INVITE requests are. By default this option is disabled.
Allow Guest Calling	Allow or reject guest calls (default is yes, to allow). If your gateway is connected to the Internet and you allow guest calls, you want to check which services you offer everyone out there, by enabling them in the default context.

Media

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.

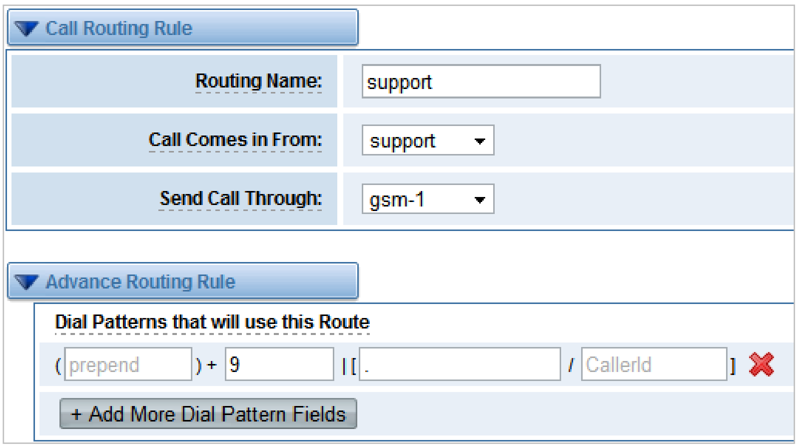
TOS for SIP Packets	Sets type of service for SIP packets
TOS for RTP Packets	Sets type of service for RTP packets

3.4 Routing

Move	Order	Rule Name	From	To	Actions
	1	SIP1_GSM1	1001	gsm-1	
	2	GSM1_SIP1	gsm-1	1001	
	3	SIP2_GSM2	1002	gsm-2	
	4	GSM2_SIP2	gsm-2	1002	
	5	SIP3_GSM3	1003	gsm-3	
	6	GSM3_SIP3	gsm-3	1003	
	7	SIP4_GSM4	1004	gsm-4	
	8	GSM4_SIP4	gsm-4	1004	

There are some default routings, if necessary, you can also add other new routings.

Click New Call Routing Rule .



Call Routing Rule

The above figure realizes that calls from gsm-1 will transfer to the "support" which you have registered in SIP endpoints. When "Call Comes in From" is gsm, "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.

Advanced Routing Rule

Options	**Definition**
Set the Caller ID Name to	What caller ID name would you like to set before sending this call to the endpoint?
Set the Caller ID Number to	What caller number would you like to set before sending this call to the endpoint?
Forward Number	What destination number will you dial? This is very useful when you have a transfer call.
Failover Call Through Number	The gateway will attempt to send the call out each of these in the order you specify.
Dial Patterns that will use this Route	A Dial Pattern is a unique set of digits that will select this route and send the call to the designated trunks. If a dialed pattern matches this route, no subsequent routes will be tried. If Time Groups are enabled, subsequent routes will be checked for matches outside of the designated time(s). Rules: X matches any digit from 0-9 Z matches any digit from 1-9 N matches any digit from 2-9 [1237-9] matches any digit in the brackets (example: 1,2,3,7,8,9) . wildcard, matches one or more dialed digits. prepend: Digits to prepend to a successful match. If the dialed number matches the patterns specified by the subsequent columns, then this will be prepended before sending to the trunks.

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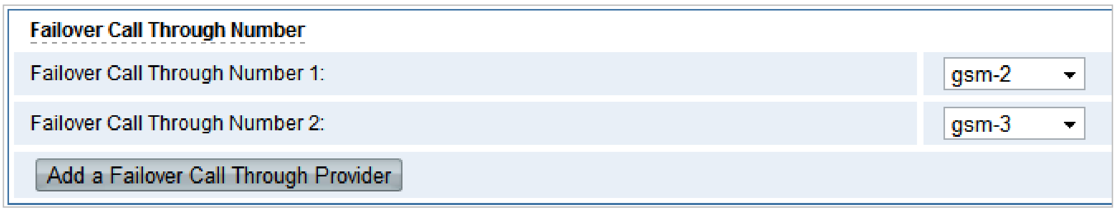
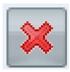
	prefix: Prefix to remove on a successful match. The dialed number is compared to this and the subsequent columns for a match. Upon a match, this prefix is removed from the dialed number before sending it to the trunks. match pattern: The dialed number will be compared against the prefix + this match 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 being transmitted matches this. When extensions make outbound calls, the CallerID will be their extension number and NOT their Outbound CID. The above special matching sequences can be used for CallerID matching similar to other number matches.



After set some routing rules, you are allowed to move rule's order by pulling  up and down, click  button to edit the routing and  to delete it.

Move	Order	Rule Name	From	To	Actions
	1	SIP1_GSM1	1001	gsm-1	
	2	GSM1_SIP1	gsm-1	1001	

Move	Order	Rule Name	From	To	Actions
	1	GSM1_SIP1	gsm-1	1001	
	2	SIP1_GSM1	1001	gsm-1	

Finally click the Save Orders button to save what you set.

3.5 Network

In "Network" page, there are four sub-pages, "LAN Settings", "WAN Settings", "DNS Settings", and "Toolkit".

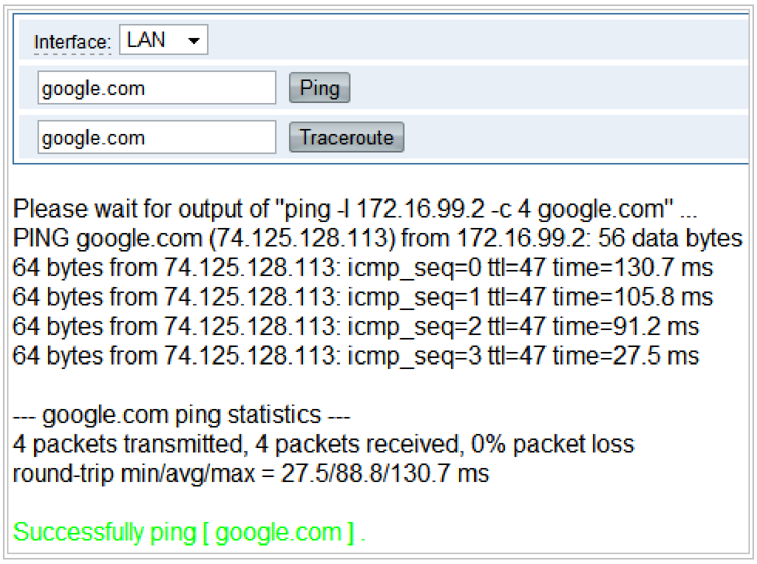
LAN Settings: There are three types of LAN port IP, Factory, static and DHCP. Factory is the default type, and it is 172.16.99.1. When you Choose LAN IPv4 type is "Factory", this page is not editable.

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.

WAN Settings: There are four methods to get WAN port's IP, factory, static, DHCP and PPPoE. Disable is WAN port unavailable.

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

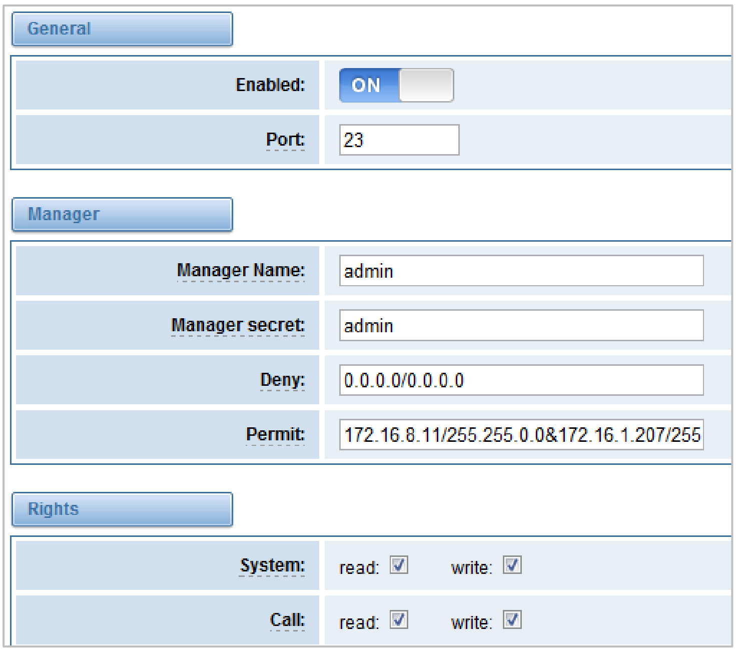
Toolkit: It is used to check network connectivity.



3.6 Advanced

3.6.1 Asterisk API

When you make “Switch” on, this page is available.



Once you set like the above figure, the host 172.16.99.16/255.255.0.0 is allowed to access the gateway API. Please refer to the following figure to access the gateway API by putty. 172.16.99.1 is the gateway’s IP, and 23 is its API port.

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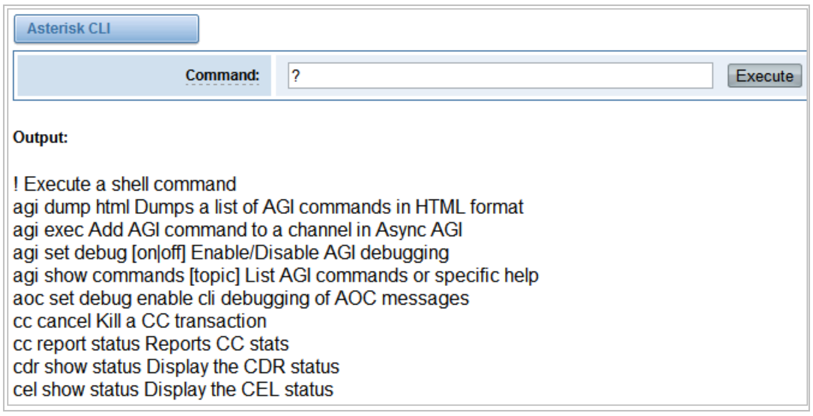
```
[root@centos ~]# telnet 172.16.99.1 23
Trying 172.16.99.1...
Connected to 172.16.99.1 (172.16.99.1).
Escape character is '^]'.
Asterisk Call Manager/1.1
Action: login
Username: admin
Secret: admin

Response: Success
Message: Authentication accepted

Event: FullyBooted
Privilege: system,all
Status: Fully Booted
```

3.6.2 Asterisk CLI

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



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

3.6.3 Asterisk File Editor

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

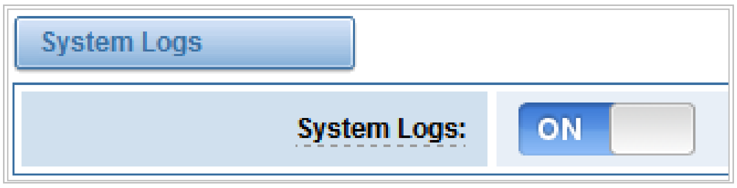
dnsmgr.conf :

```
[general]
;enable=yes              ; enable creation of managed DNS lookups
                         ;   default is 'no'
;refreshinterval=1200    ; refresh managed DNS lookups every <n> seconds
                         ;   default is 300 (5 minutes)
```

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

3.7 Logs

In 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.



Appendix Feature List

System	
Sync Time from NTP and Client	√
Web username and password modification	√
System and Asterisk reboot	√
Online firmware update	√
Configuration files upload	√
Backup function support	√
Factory reset	√
GSM	
Speaker and microphone volume adjustment	√
RX/TX gain adjustment	√
PIN code management	√
Transfer SMS to E-mail box	√
SMS command control	√
SMS receiving, sending, group sending	√
Number check	√
USSD support	√
Polarity-reversal	√
SIP	
Domain name registration	√
Registration	None; Endpoint registers with this gateway; This gateway registers with the endpoint;
SIP trunk delete, add and edit	√
DTMF mode	RFC2833、Inband、Info
SIP timer settings	√
Assigned SIP port	UDP、TCP、RTP
SIP authentication settings	√
Custom SIP method	√
Caller ID Advanced settings	√
Codec	G.711A,G.711U,GSM,G.722， G.726,G.729,LPC10
NAT settings	√
QoS/ToS settings	√

Routing	
Routing calls before/after manipulation	√
TEL->IProuting	√
IP->TELrouting	√
Add/modify/delete routing	√
IP->TEL destination number manipulation	√
TEL->IP destination number manipulation	√
TEL->IP source number manipulation	√
Failover	√
Network	
LAN IP	Factory, static, DHCP
WAN IP	Factory, static, DHCP, PPPoE
DNS settings	√
Ping, traceroute function	√
Advanced	
Opened Asterisk API	√
Executable Asterisk CLI	√
Editable Asterisk files	√
Logs	
System logs support	√
Asterisk logs support	√
SIP logs support	√
AT command logs	√
CDR	√

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