OpenVox Communication Co Ltd

iAG801 Series Analog Gateway User Manual

Version 1.0

OpenVox Communication Co Ltd

Address: Room 624, 6/F, Tsinghua Information Port, Book Building, Qingxiang Road, Longhua Street, Longhua District, Shenzhen, Guangdong, China 518109

Tel: +86-755-66630978, 82535461, 82535362

Business Contact: sales@openvox.cn

Technical Support: support@openvox.cn

Business Hours: 09:00-18:00(GMT+8) from Monday to Friday

URL: www.openvox.cn

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Revise History

Version	Release Date	Description
1.0	25/11/2022	First Version

Contents

1. Overview

1.1 What is iAG Series Analog Gateway?

OpenVox iAG801 is a multifunctional analog gateway that provides 8 FXS ports for seamless connection between IPPBX, fax machines, analog phones and operators. In addition, it has excellent full concurrent voice/fax processing capabilities, strong performance and high stability, and provides high-quality call services for operators, enterprises, call centers and residential users in residential communities.

The iAG801 Analog Gateway, a cost-effective product of the iAG Series, is an ideal analog VoIP gateway solution for SMBs and SOHOs. With friendly GUI and unique design, users may easily customize and configure their gateway. Also secondary development can be completed through AMI (Asterisk Management Interface).

The iAG801 Analog Gateway is developed for interconnecting a wide selection of codecs including G.711A, G.711U, G.722, G.726, G.729A, iLBC. iAG801 series use standard SIP protocol and compatible with leading VoIP platform, IPPBX and SIP servers, such as Asterisk, Issabel, 3CX, FreeSWITCH, BroadSoft and VOS VoIP operating platform.

1.2 Sample Application

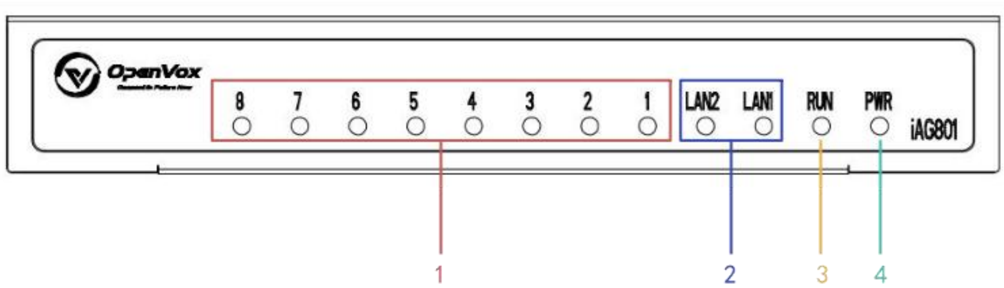
Figure 1-2-1 Topological Graph

1.3 Product Appearance

The picture below is the appearance of iAG801 analog gateway.

Figure 1-3-1 Product Appearance

Figure 1-3-2 Front Panel



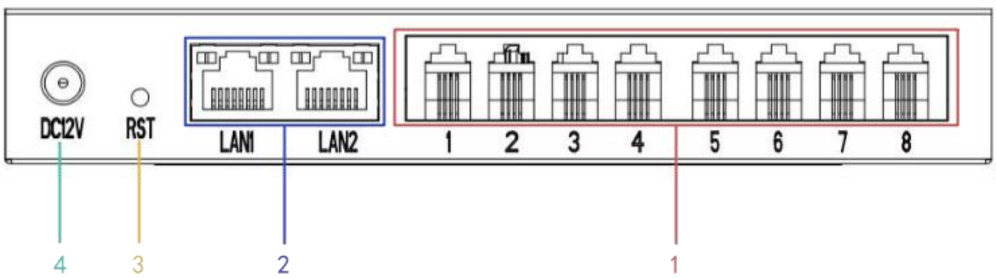
1: Channel Status Indicator

2: Network Status Indicator

3: Running Status Indicator

4: Power Status Indicator

Figure 1-3-3 Back Panel



1: Channel Interface

2: Network Interface

3: Reset button

4: Power Interface

1.4 Main Features

System Features

- NTP time synchronization and client time synchronization
- Support modify username and password for web login
- Update firmware online, backup/restore configuration file
- Abundant Log Info, Automatically Reboot, Call status display
- Language selection (Chinese/English)
- Open API interface (AMI), support for custom scripts, dialplans
- Support SSH remote operation and restore the factory settings

Telephony Features

- Support volume adjustment, gain adjustment, call transfer, call hold, call waiting, call forward, Caller ID display
- Three way calling, call transfer, dial-up matching table
- Support T.38 fax relay and T.30 fax transparent, FSK and DTMF signaling
- Support ring cadence and frequency setting, WMI (Message Waiting Indicator)
- Support echo cancellation, jitter buffer
- Support customizable DISA and other applications

SIP Features

- Support add, modify & delete SIP accounts, batch add, modify & delete SIP accounts
- Support multiple SIP registrations: Anonymous, Endpoint registers with this gateway, This gateway registers with the endpoint
- SIP accounts can be registered to multiple servers

Network

- Network type: Static IP, Dynamic
- Support DDNS, DNS, DHCP, DTMF relay, NAT
- Telnet, HTTP, HTTPS, SSH
- VPN client
- Network Toolbox

1.5 Physical Information

Table 1-5-1 Description of Physical Information

Weight	400g
Size	170*98*26mm
Temperature	-20~70°C (Storage)
	0~50°C (Operation)
Operation humidity	10%~90% non-condensing
Power source	12VDC/2A
Max power	12W

1.6 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.

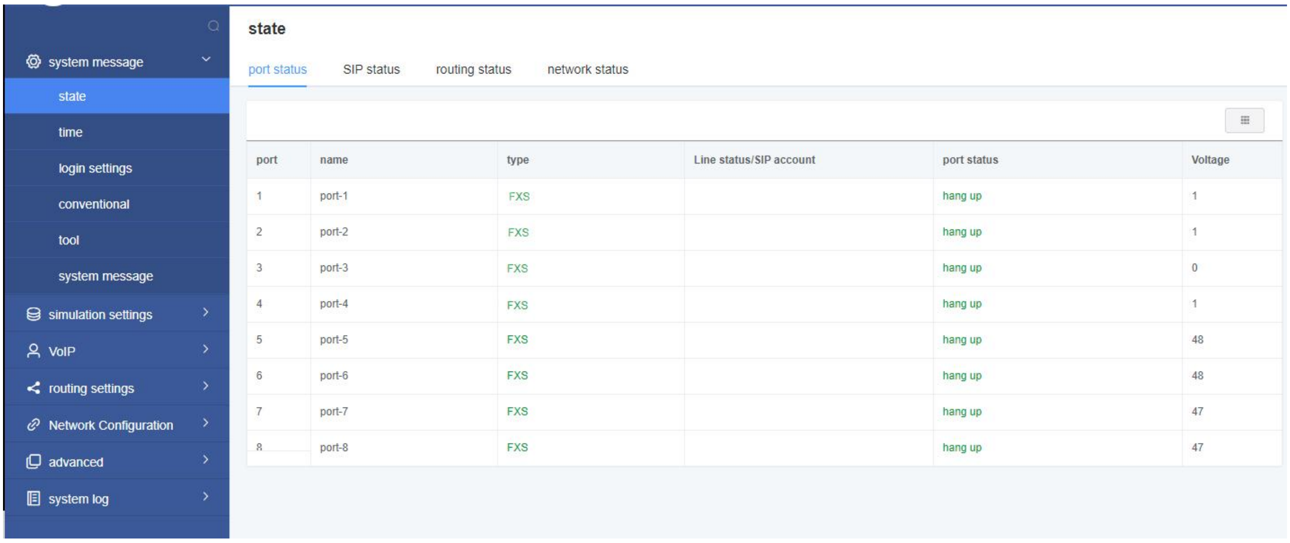
Figure 1-6-1 Login Interface

2. System

2.1 Status

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

Figure 2-1-1 System Status



2.2 Time

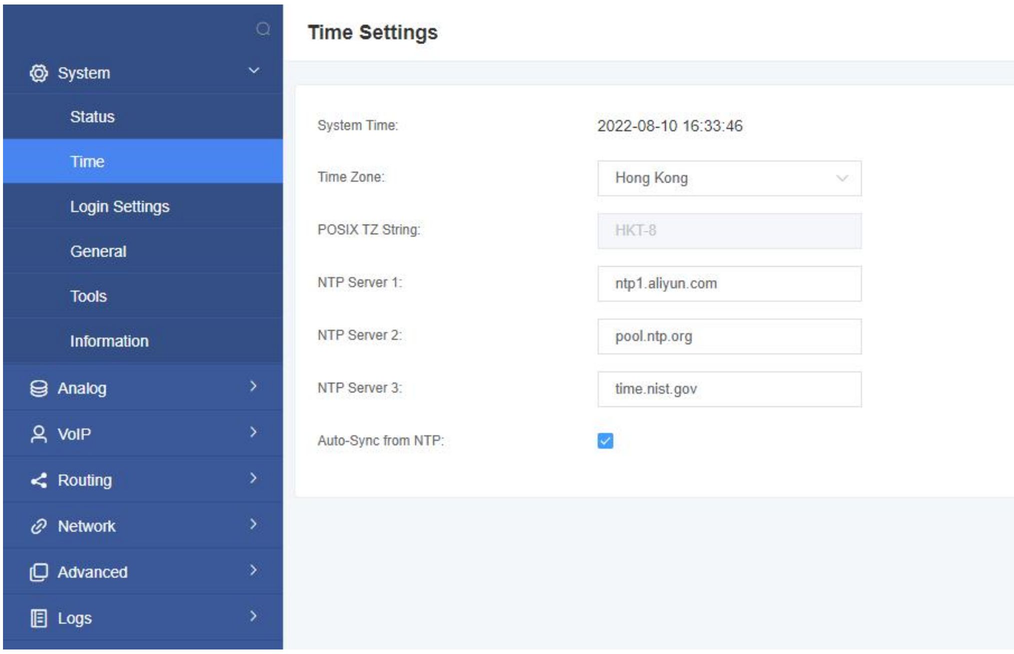
Table 2-2-1 Description of Time Settings

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 is enable, OFF is disable this function.
Sync NTP	Sync time from NTP server.
Sync Client	Sync time from local machine.

For example, you can configure like this:

Figure 2-2-1 Time Settings



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

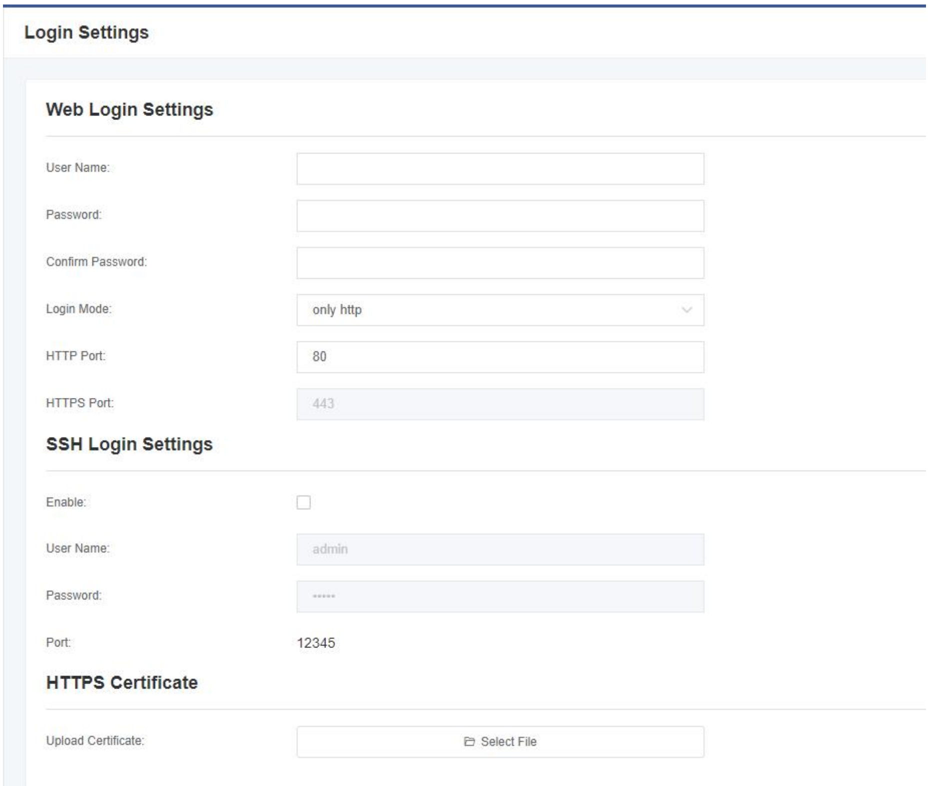
2.3 Login Settings

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 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.

Table 2-3-1 Description of Login Settings

Options	Definition
User Name	Define your username and password to manage your gateway, without space here. Allowed characters "-_+. < >&0-9a-zA-Z". Length: 1-32 characters.
Password	Allowed characters "-_+. < >&0-9a-zA-Z". Length: 4-32 characters.
Confirm Password	Please input the same password as 'Password' above.
Login Mode	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



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

2.4 General

2.4.1 Language Settings

On our gateway products, you can set different languages according to your needs.

First, you need to turn on the "Advanced" mode.

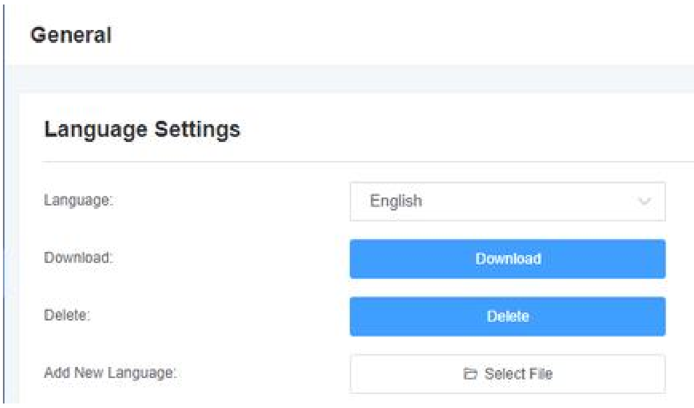
And then "Download" the current language pack of the system.

Then click the "Browse" option.

After importing the language pack you need, click the "Add" button.

And it will take effect without restarting the gateway.

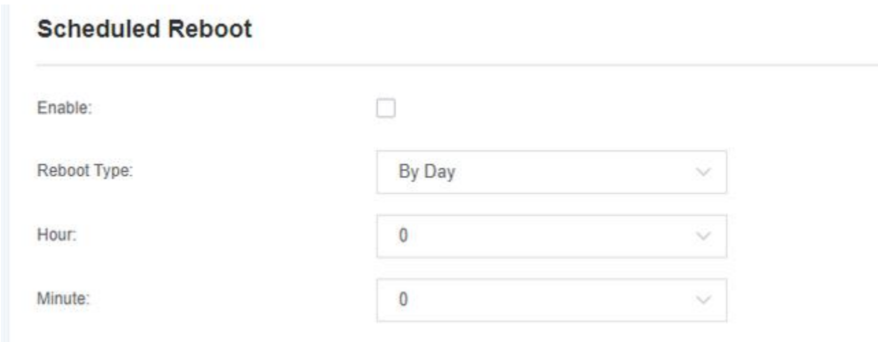
Figure 2-4-1 Language Settings



2.4.2 Scheduled Reboot

You can enable the automatic restart function to make your gateway restart after working for a certain period of time to achieve higher work efficiency.

Figure 2-4-2 Scheduled Reboot

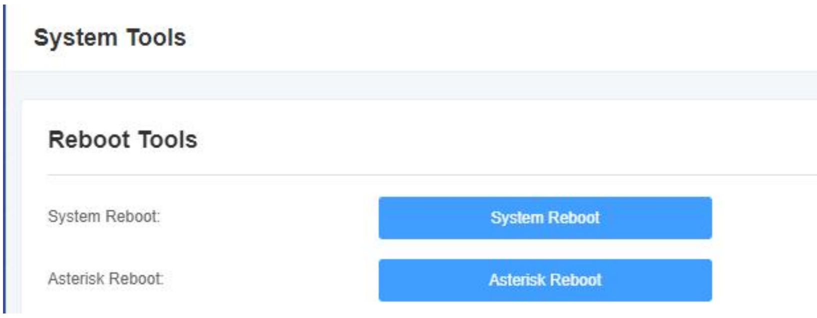


2.5 Tools

In the tool page, users can restart the gateway, upgrade firmware, upload and backup configuration files, and restore factory settings.

The analog gateway supports individual system restart or Asterisk restart.

Figure 2-5-1 Reboot Tools



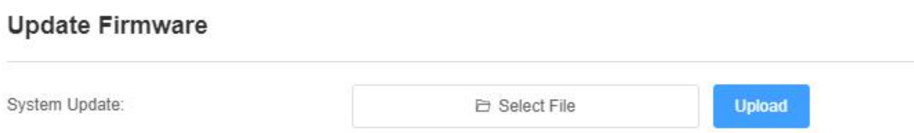
Notice: When you confirm the restart, the system will automatically end all current calls.

Table 2-5-1 Description of Reboots

Options	Definition
System Reboot	This option will restart your gateway and cut off all current sessions.
Asterisk Reboot	This option will restart Asterisk and cut off all current sessions.

The analog gateway provides two firmware upgrade methods, you can choose system upgrade or system online upgrade. To select the system upgrade, you need to download the relevant firmware from the OpenVox website first. The system online upgrade is an easier way with one-click upgrade.

Figure 2-5-2 Update Firmware



After configuring your gateway, you can download the current configuration file. When you need to configure other gateways of the same model or restore the gateway to factory settings, you can choose to upload this backup configuration file without the need to reconfigure the gateway .

Notice: It will take effect only if the version of the configuration file and the current firmware version are the same.

Figure 2-5-3 Upload and Backup Configuration Files

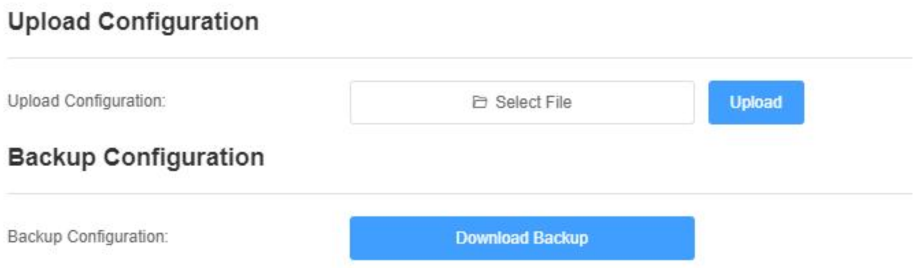


Figure 2-5-4 Voice Record

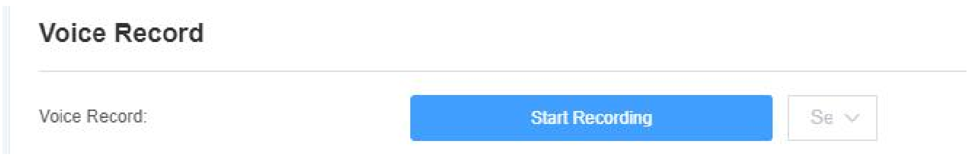


Figure 2-5-5 Restore Configuration File

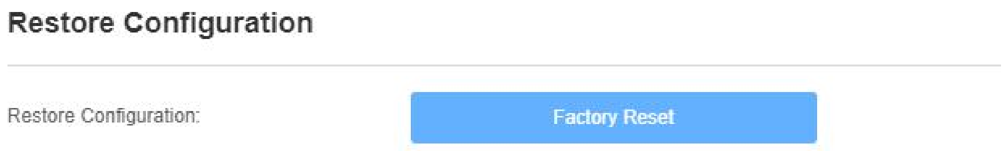
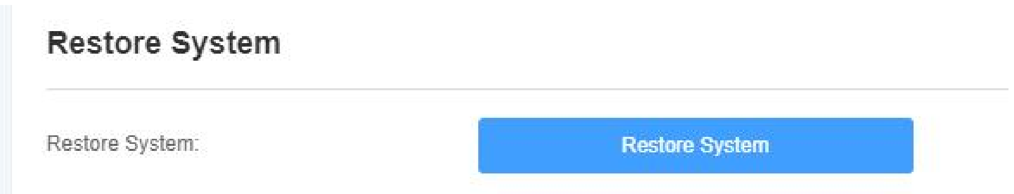


Figure 2-5-6 Restore System

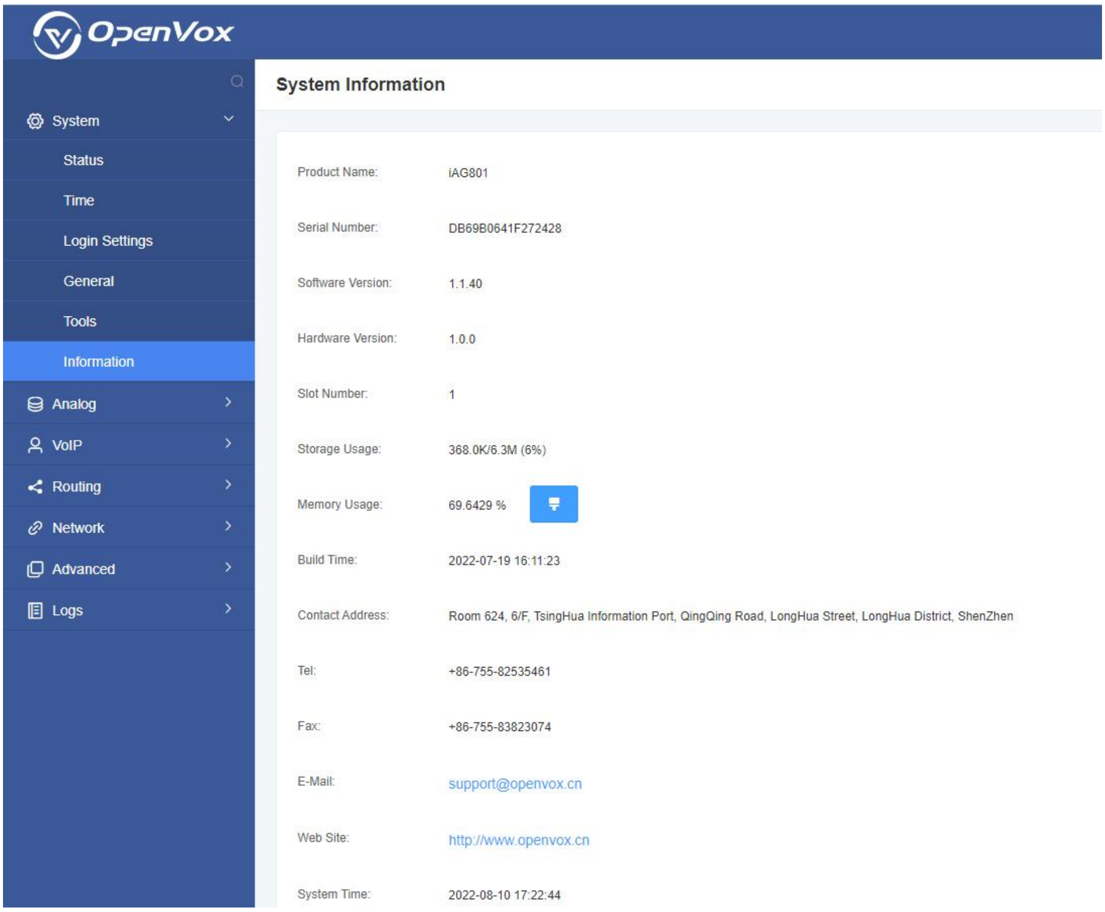


Notice: You can restore the gateway to factory settings by dialing. Connect the phone to the FXS port of the gateway and dial "*1*2*3*4" , then it will restore the gateway to factory settings.

2.6 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.

Figure 2-6-1 System Information

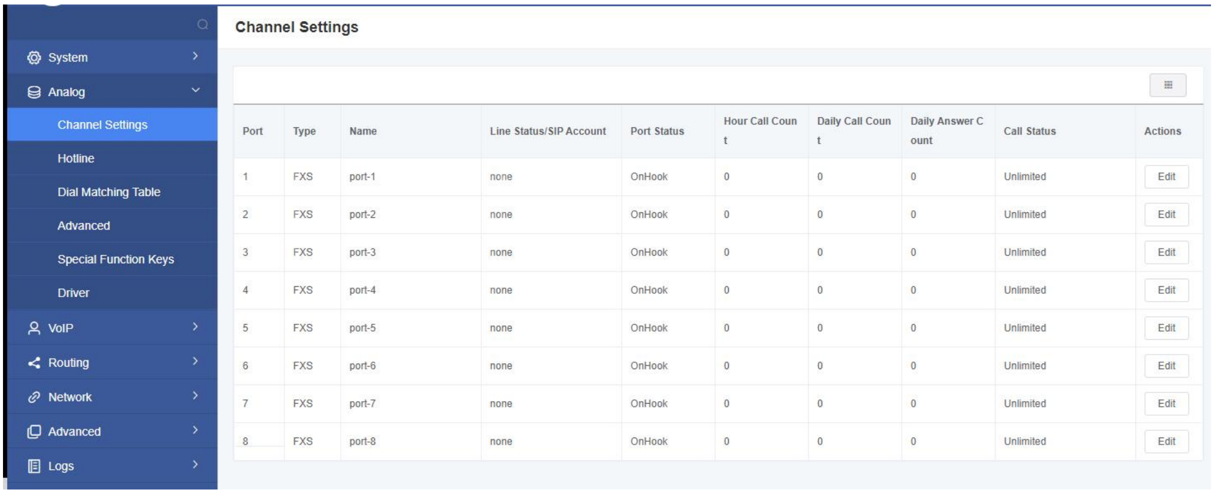


3. Analog

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

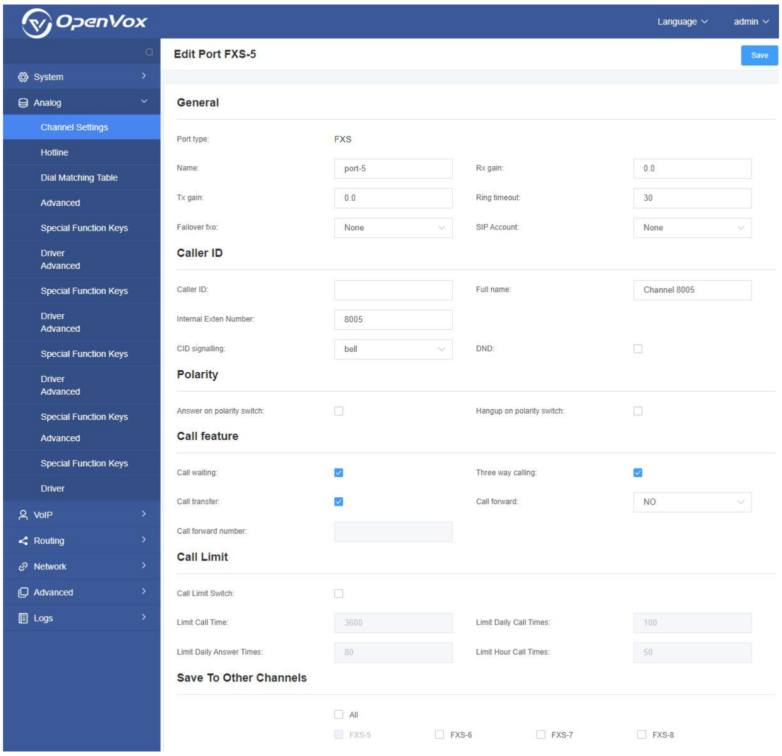
3.1 Channel Settings

Figure 3-1-1 Channel System



Click the button to automatically modify the corresponding port information.

Figure 3-1-2 FXS Port Configure



3.2 Hotline Settings

Call pick-up is a feature used in a telephone system, which allows one phone to answer a call on the another phone. You can set the "Time Out" and "Number" parameters individually or globally for each port. This function is realized by dialing a series of specific numbers, provided that you enable this function and set the "number" parameter correctly.

Figure 3-2-1 Pickup Settings

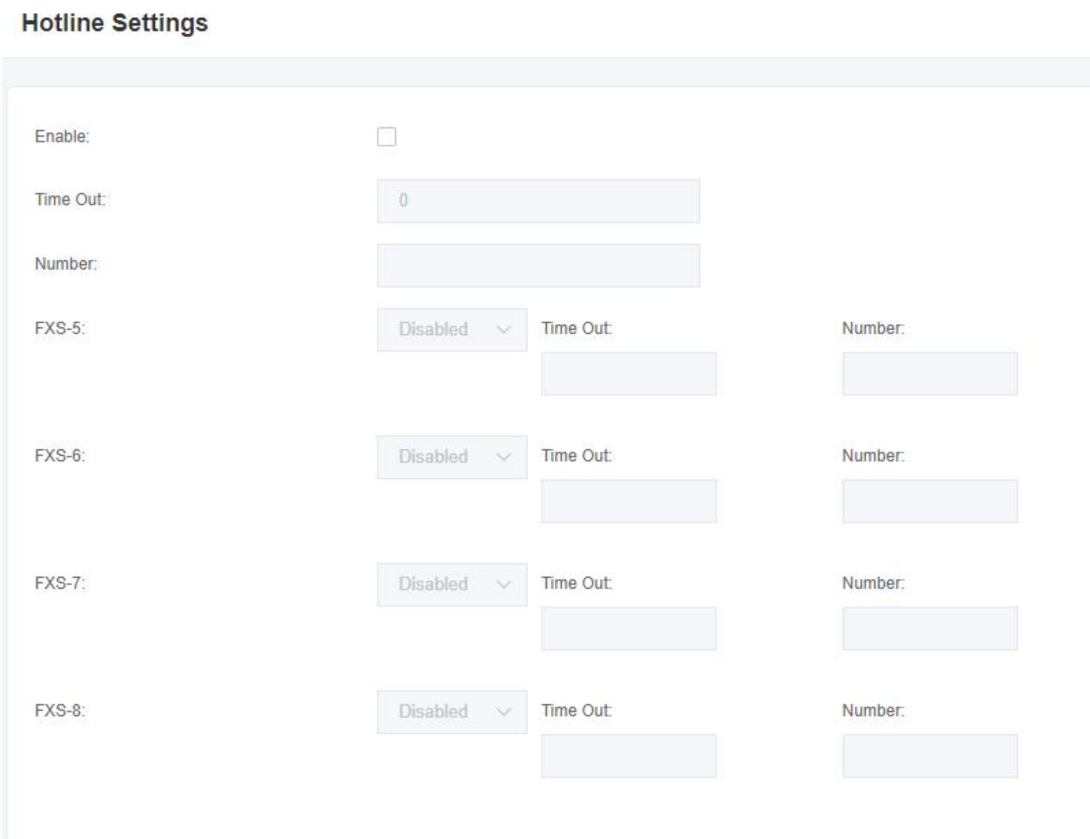


Table 3-2-1 Definition of Pickup

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

3.3 Dial Matching Table

The dial matching table is to effectively judge whether the received number is complete so that it can be sent in time.

The correct use of the dial matching table can help shorten the call establishment time.

Figure 3-3-1 Dial Matching Table

3.4 Global Settings

Figure 3-4-1 General Configuration

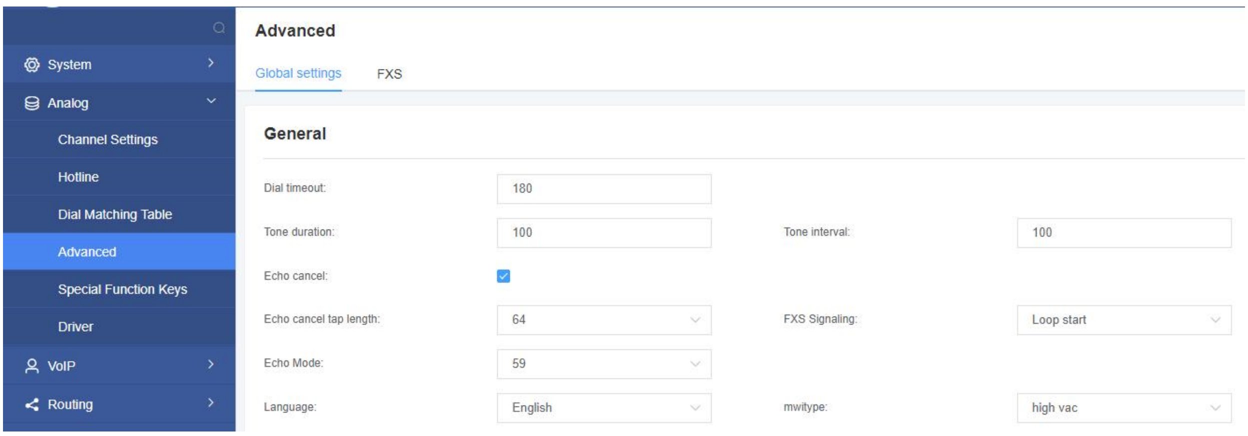


Table 3-4-1 Instruction of General

Options	Definition
Dial timeout	Specifies the number of seconds we attempt to dial the specified devices.
Tone duration	How long generated tones (DTMF and MF) will be played on the channel. (in milliseconds).
Tone interval	How long between tone and tone will be played on the channel. (in milliseconds).
Echo cancel	Echo cancel
Echo cancel tap length	Hardware echo canceler tap length.
FXS Signaling	Default Loop start, busy tone is generated, Kewlstart, power is off, no busy tone is generated

Figure 3-4-2 Fax Configuration

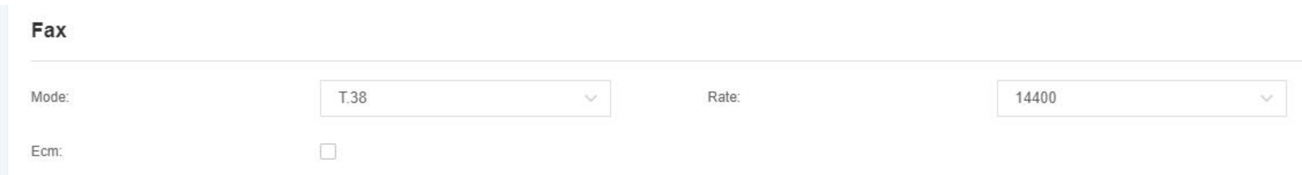


Table 3-4-2 Definition of Fax Option

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 3-4-3 Country Configuration

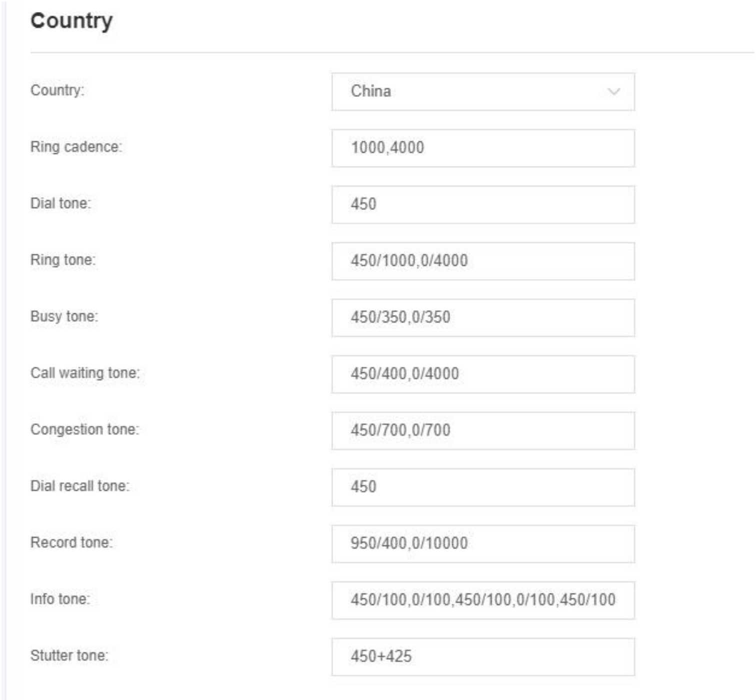
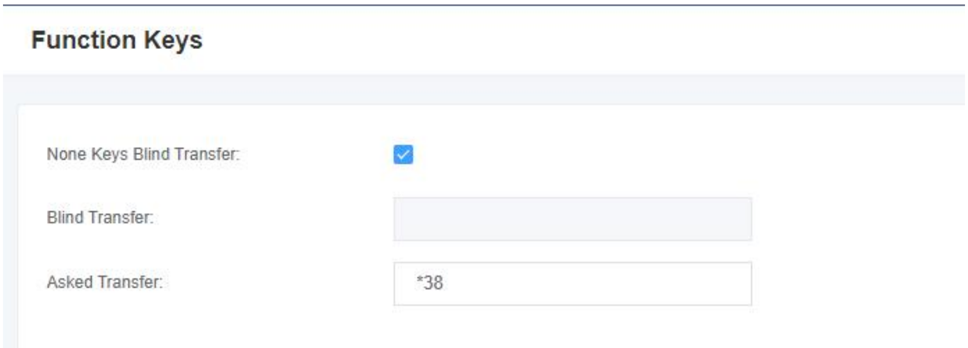


Table 3-4-3 Definition of Country Settings

Options	**Definition**
Country	Set the signal tone standard of the country where the gateway is located.
Ring cadence	List of duration the physical bell rings.
Dial tone	Set the off-hook dial tone.
Ring tone	Set the prompt tone to the caller when ringing.
Busy tone	Set the prompt tone when busy.
Call waiting tone	Set the background prompt tone to play when entering the call waiting.
Congestion tone	Set the prompt tone to be played when congested.
Dial recall tone:	Set the prompt tone for the second dialing after pressing the flash key.
Record tone	Set the prompt tone for the recording process.
Special message tone	Set the prompt tone for playing special information. (for example: the dialed number is not in the service area)

3.5 Special Function Keys

Figure 3-5-1 Function keys



3.6 Driver

Figure 3-6-1 General

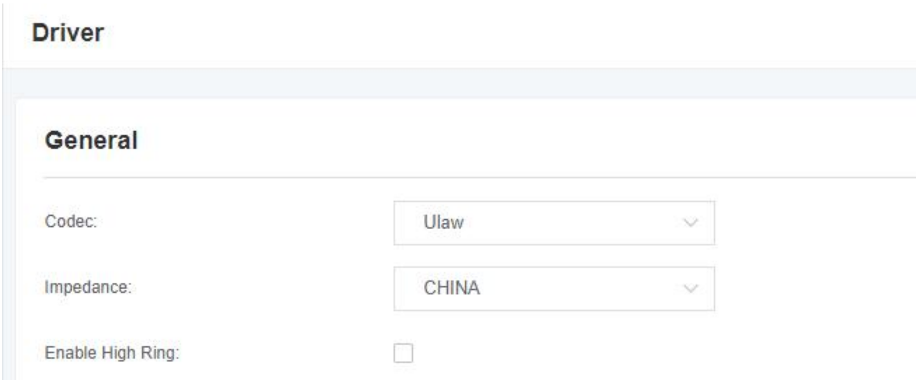


Table 3-6-1 Definition of General

Options	Definition
Codec	Set the global encoding: mulaw, alaw
Impedance	Configuration for impedance.
Enable High Ring	High ring enable help.

Figure 3-6-2 CallerID Detect

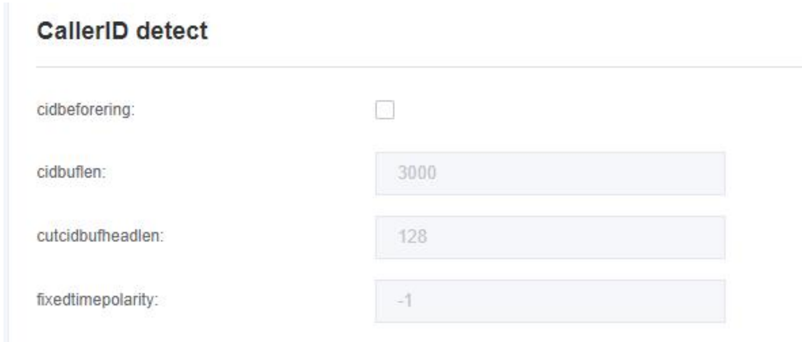


Table 3-6-2 Definition of CallerID Detect

Options	Definition
cidbeforering	Switch to handle irregular CID function.
cidbuflen	CID media stream length byte size.
cutcidbufheadlen	CID media stream header length byte size.
fixedtimepolarity	Transmit polarity line reversal signal delay time.

Figure 3-6-3 Hardware Gain

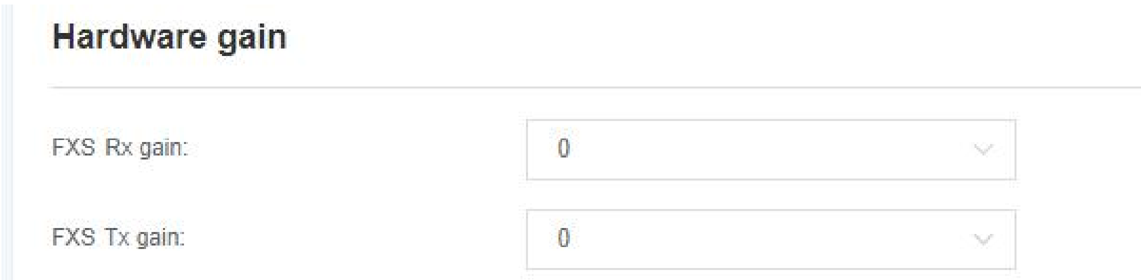


Table 3-6-3 Definition of Hardware Gain

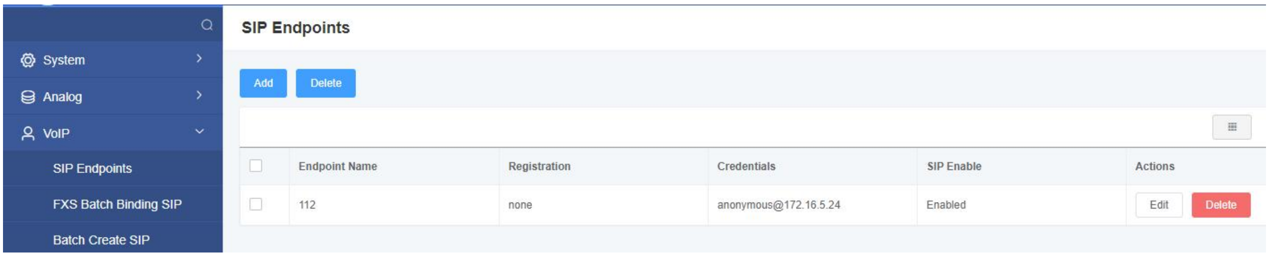
Options	Definition
FXS Rx gain	Set FXS to IP gain. Range: -35, 0 or 35. the default is 0.
FXS Tx gain	Set FXS to terminal gain. Range: -35, 0 or 35. the default is 0.

4. VoIP

4.1 SIP Endpoint

On this page, the status information about the SIP account is displayed.

Figure 4-1-1 SIP Endpoints



Click the edit button to modify the corresponding SIP information.

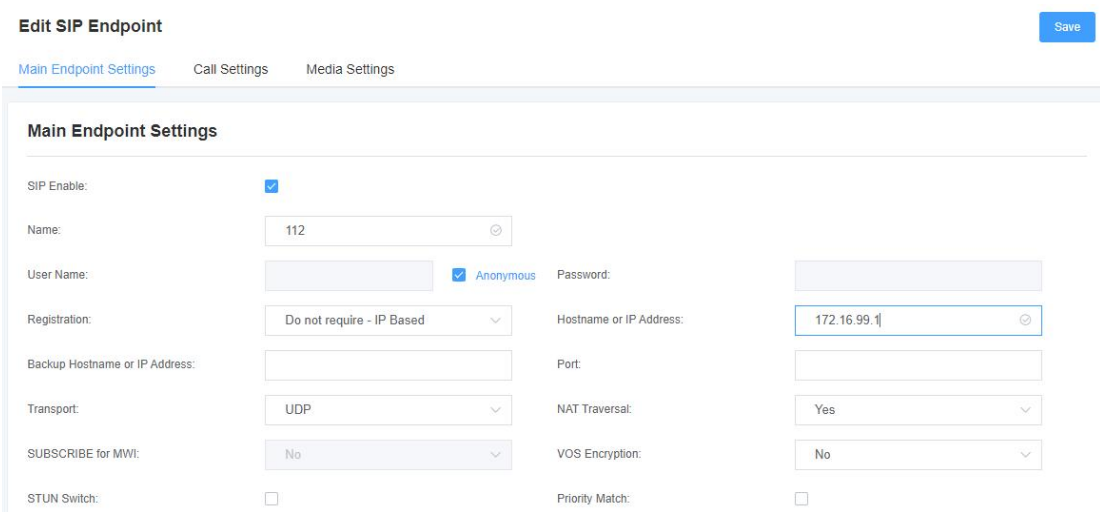
4.1.1 Main Endpoint Settings

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

You can configure as follows:

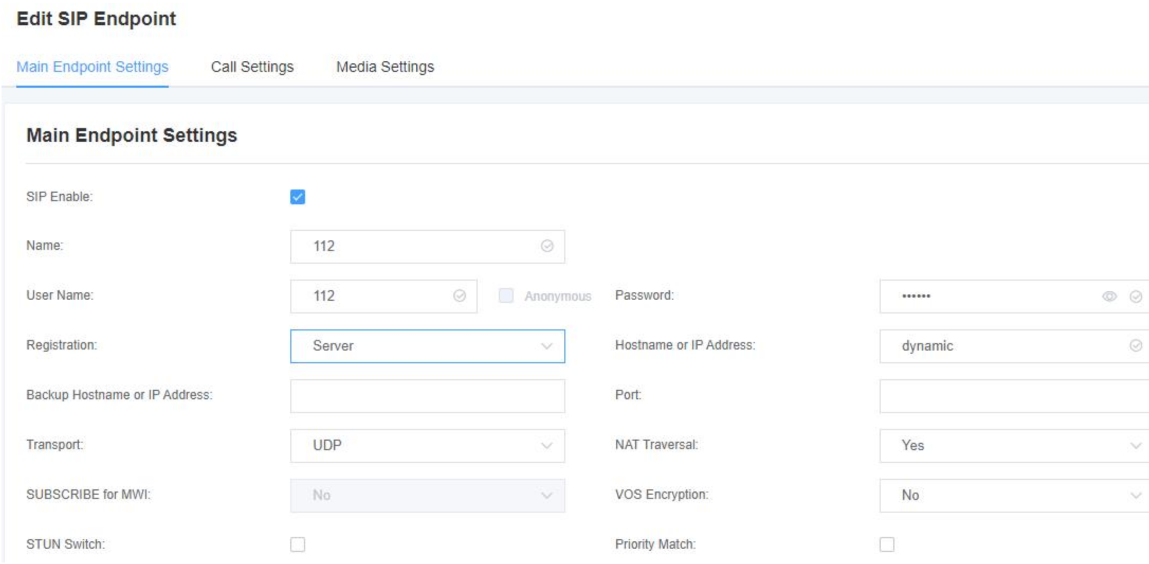
If you set up a SIP endpoint by registration "None" to a server, then you can't register other SIP endpoints to this server. (If you add other SIP endpoints, this will cause Out-band Routes and Trunks confused.)

Figure 4-1-2 Main Endpoint Settings



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 Main Endpoint Settings



When "Gateway is registered to the endpoint", you need to fill in the username and password, and you can register multiple SIP endpoints to the server. Due to the difference in usernames and passwords, there will be no confusion between routing and Trunks.

Figure 4-1-4 Register to Server

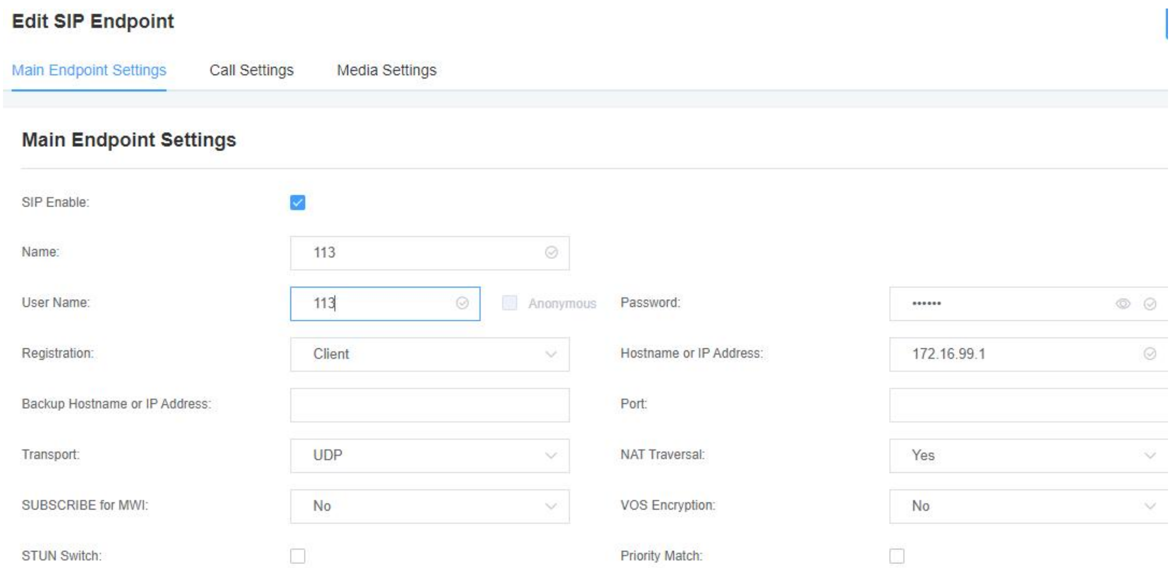


Table 4-1-1 Definition of Endpoint Settings Options

Options	**Definition**
Name	A name which is able to read. And it's only used for user's reference.
Username	Username for authentication between the endpoint and the gateway.
Password	The password for authentication between the endpoint and the gateway, allowing letters.
Registration	**None**---Anonymous registration; **Endpoint registers with this gateway**---The gateway is used as a server, and the SIP endpoint is registered to the gateway; **This gateway registers with the endpoint**---The gateway is used as a client, and the SIP terminal needs to be registered on the server.
Domain name or IP address	IP address or domain name of the endpoint or 'dynamic' .（if the endpoint has a dynamic IP address.）This needs to register.
Alternate domain name or IP address	Same as above. After filling this, it is equivalent to that this account initiates registration to two domain names or IP addresses at the same time. When the account of the primary domain name or IP address expires, it will switch to the account of the alternate domain name or IP address
Transmission method	Set possible transmission types and order of use for outgoing transmissions. When you use various transport protocols: UDP, TCP, TLS, the transmission type enabled for the first time is only used for outgoing messages until registration occurs. If the endpoint requires another transmission type during the registration process, the first transmission type may be changed to another transmission type.
NAT Traversal	Issues related to NAT addresses when incoming SIP or media sessions.
VOS encryption	When the endpoint is a VOS server, the encryption item needs to be used, and the parameters need to be turned on at this time

4.1.2 Advanced: Registration Options

Figure 4-1-5 Registration Options

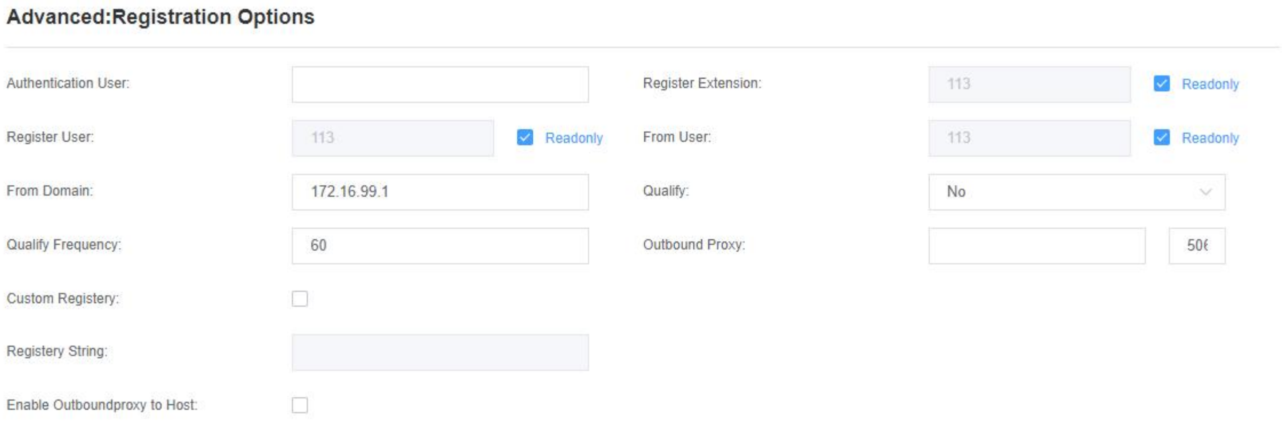


Table 4-1-2 Definition of Registration Options

Options	**Definition**
Authentication User	A username to use only for registration.
Register Extension	When Gateway registers as a SIP user agent to a SIP proxy (provider), calls from this provider connect to this local extension.
Registered user name	The registered username, is the user in "register => user[:secret[:authuser]]@host[:port][/extension]"
User source	A username to identify the gateway to this endpoint.
From Domain	A domain to identify the gateway to this endpoint.
Port	The port number the gateway will connect to at this endpoint.
Quality	To check the endpoint's connection status whether or not.
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.
Customized registration	After opening, customers can customize the registration form by

switch	themselves.
Enable Outboundproxy to Host	

4.1.3 Call Settings

Table 4-1-3 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).
Call Limit	Set a call limit, the maximum number of calls that can be allowed at the same time.
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.
Endpoint 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.

4.1.4 Advanced: Signaling Settings

Table 4-1-4 Definition of Signaling Options

Options	**Definition**
Inbound In-band Signaling	Whether to generate an incoming ring tone. Never: indicates that the incoming signal is never applicable; Optional values: yes, no, never; Default value: 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 reasonable header and to use it if it is available.
SDP Version Header	By default, the gateway will add a session version number to the SDP packet and if the SDP version number is modified, it will only modify the SDP session. Turning off this option will force the gateway to ignore this SDP version number and treat all SDP data as new data. This is necessary for a device that sends non-standard SDP packets. It is turned on by default.
Allow Transfers	Whether or not to globally enable transfers. Choosing 'no' will disable all transfers (unless enabled in endpoints or users). Default is enabled.
Allow Promiscuous Redirection	Whether or not to allow 302 or REDIR to non-local SIP address. **Notice:** Redirecting to the local system will cause a loop call, which Asterisk does not support.
Maximum Forward	Setting for the SIP Max-Forwards header (loop prevention).
Send TRYING on REGISTER	Send a 100 Trying when the endpoint registers.

4.1.5 Advanced: Timer Settings

Table 4-1-5 Definition of Timer Options

Options	**Definition**
Default T1 Timer	This timer is used primarily in INVITE transactions. The default for Timer. The default T1 clock is 500 milliseconds or if you have qualify=yes it will measure the round-trip time between the running gateway and the device.
Call Setup Timer	If no provisional response is received within this period of time, the call will be automatically blocked. The default value is 64*T1.
Session Timers	There are three modes to choose from: Proactively initiate, request and run the session timer; Only accept or run the session timer when requested by other user agents; Refuse, do not run session timers in any case.
Minimum Session Refresh Interval	The minimum session refresh interval (in seconds). The default is 90secs.
Maximum Session Refresh Interval	The maximum session refresh interval (in seconds). The default is 1800secs.
Session Refresher	Session refresher, user agent client or user agent server. The default is the user agent server.

4.1.6 Media Settings

Table 4-1-6 Definition of Media Settings

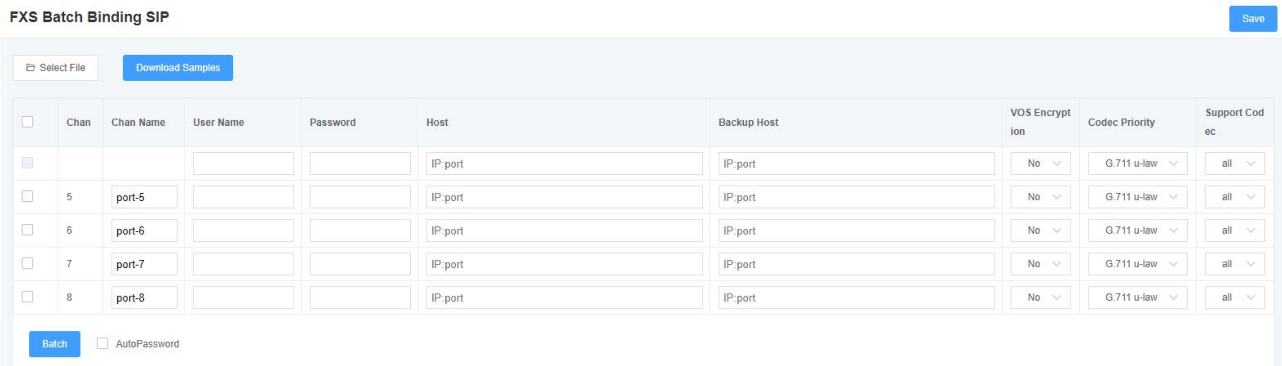
Options	**Definition**
Media Settings	Select codec from the drop down list. Different encoding priorities choose different encoding methods.

4.2 FXS Batch Binding SIP Accounts

If you want to bind SIP accounts in batches on the FXS port, you can configure this page.

Notice: It is only available in the "This gateway registers with the endpoint" working mode.

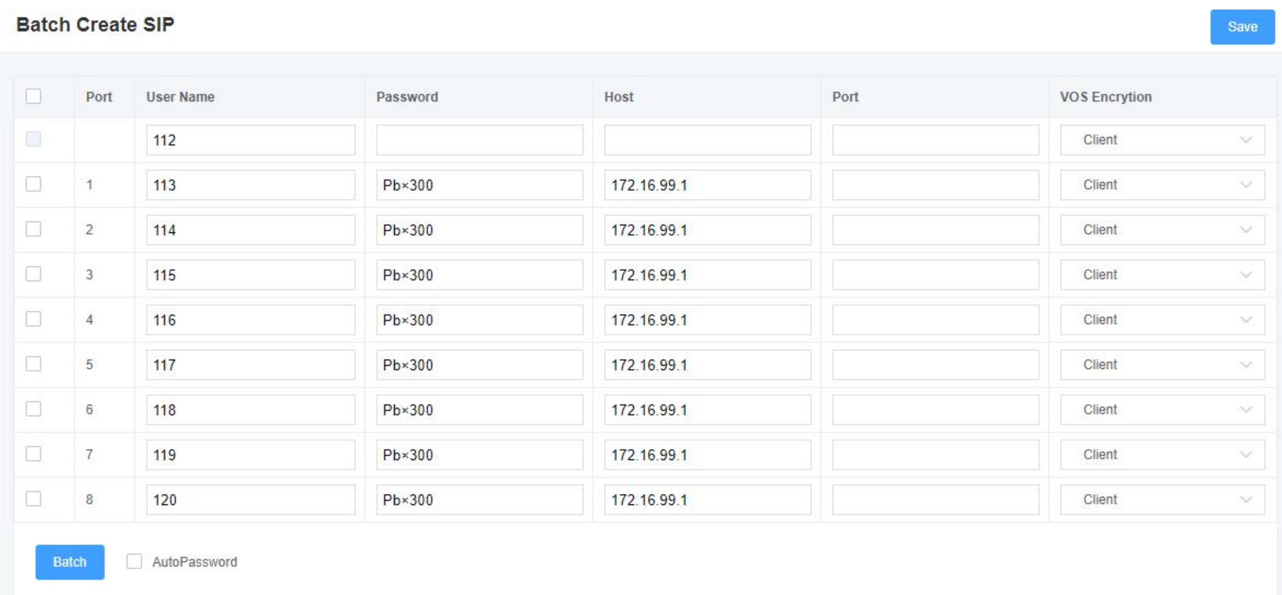
Figure 4-2-1 FXS Batch Binding SIP



4.3 Batch Create SIP

On this interface, users can create multiple SIP accounts at one time. You can choose any registration mode.

Figure 4-3-1 Batch Create SIP



4.4 Advanced SIP Settings

4.4.1 Networking

Table 4-4-1 Regular Choice

Options	**Definition**
UDP Bind Port	Choose a port on which to listen for UDP traffic.
Enable TCP	Enable request server for incoming TCP link (default is no).
TCP Bind Port	Choose a port on which to listen for TCP traffic.
TCP Authentication Timeout	The maximum number of seconds for client link verification. If the client is not authenticated before the time expires, the client will be disconnected. (Default value: 30 seconds)
TCP Authentication Limit	The maximum number of simultaneous links allowed in a given time. (Default value: 50 seconds)
Enable Hostname Lookup	Open the DNS SRV lookup for outbound calls. **Notice:** The gateway is only the first host in the SRV record. This function can be used in dial-up activation to dial SIP calls on the Internet through the domain name.

4.4.2 NAT Settings

Table 4-4-2 Definition of 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 NAT network. This gateway will replace the internal IP address in SIP and SDP messages with the external IP address when a NAT exists between the gateway and other endpoints.
Local Network List	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 external address or domain name if it matches.
Dynamic and Static Selection	Dynamic hosts are not allowed to register with the IP address of static hosts. This will avoid registration errors with the same IP.
External TCP Port Mapping	When the gateway is behind a static NAT or PAT, the TCP port is externally mapped.
External IP Address	The external address of the NAT (and optional TCP port). External IP Address = hostname[:port] specifies a static address[:port] to be used in SIP and SDP messages. Examples: External IP Address = 12.34.56.78 External IP Address = 12.34.56.78:9900
External IP Hostname	The external hostname (and optional TCP port) of the NAT.
Hostname Refresh Interval	It will show how often to perform the hostname lookup. You can also configure a domain name. The gateway will perform a DNS query (This method is not recommended). Try to use IP and configure "externip".

4.4.3 STUN Settings

Table 4-4-3 Definition of STUN Settings Options

Options	**Definition**
Start	Turn on function
Server Port	Default port 3478
Refresh Request Interval	Time interval in seconds, default 30 seconds
Server IP Address/Domain Name	Server address or domain name

4.4.4 RTP Settings

Table 4-4-4 Definition of RTP Settings Options

Options	**Definition**
Start of RTP Port	Start range of port numbers to be used for RTP.
End of RTP port	End range of port numbers to be used for RTP.
RTP Timeout	RTP Timeout

4.4.5 Parsing and Compatibility

Table 4-4-5 Instruction of Parsing and Compatibility

Options	**Definition**
Strict RFC Interpretation	Check header tags, character conversion in URIs, and multiline headers for strict SIP compatibility (default is yes).
Send Compact Headers	Send compact SIP headers
SDP Owner	Allows you to change the domain of the SDP username. £¨o=£©

	This filed **MUST NOT** contain spaces.
Disallowed SIP Methods	When speaking back to other SIP peers, the other peers should include an "Allow" header to tell us the implementation of the SIP method. However, some peers do not include "Allow" headers or forge the methods they implement. In this case, the gateway will assume that the peer supports all known SIP methods. If you know that your SIP peer does not provide support for a specific method, then you may need to provide a list of methods that the peer does not implement in "disallowed methods". **Notice:** If your peer is real, then there is no need to set this option
Shrink Caller ID	The function can 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 allowable time for incoming registration and subscription (seconds).
Minimum Registration Expiry	The minimum length of registration and subscription (default 60).
Default Registration Expiry	Default length of incoming/outgoing registration.
Registration Timeout	How long will it take to re-register the extension (Default 20 seconds).
Number of Registration Attempts	Number of registration attempts before giving up.

4.4.6 Security and Media

Table 4-4-5 Instruction of Security and Media

Options	**Definition**
Match Auth Username	If available, use the user name field of the authentication line to match instead of using the user name of the user name field.
Realm	For authentication domains, all domains must be globally unique according to the RFC3261 standard. Generally can be set to host name or domain name.
Use Domain as Realm	Use the SIP domain as the boundary of the domain.
Always Auth Reject	When an "INVITE" or "REGISTER" request is rejected for any reason, the same reason will always be used. The username is legal but the password is incorrect. It does not tell the requester whether there is this "user" or "peer", which reduces the possibility of an attacker scanning the SIP account (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 customer calls (enabled by default, allowed). If your gateway is connected to an external network and allows customers to call, you want to check which services are provided for everyone and enable it in the default "context".
Premature Media	Some SDN links will 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 medias 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 'no answer' 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

4.5 Sip Account Security

This analog gateway support TLS protocol 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 provided by the server.

Figure 4-5-1 TLS settings

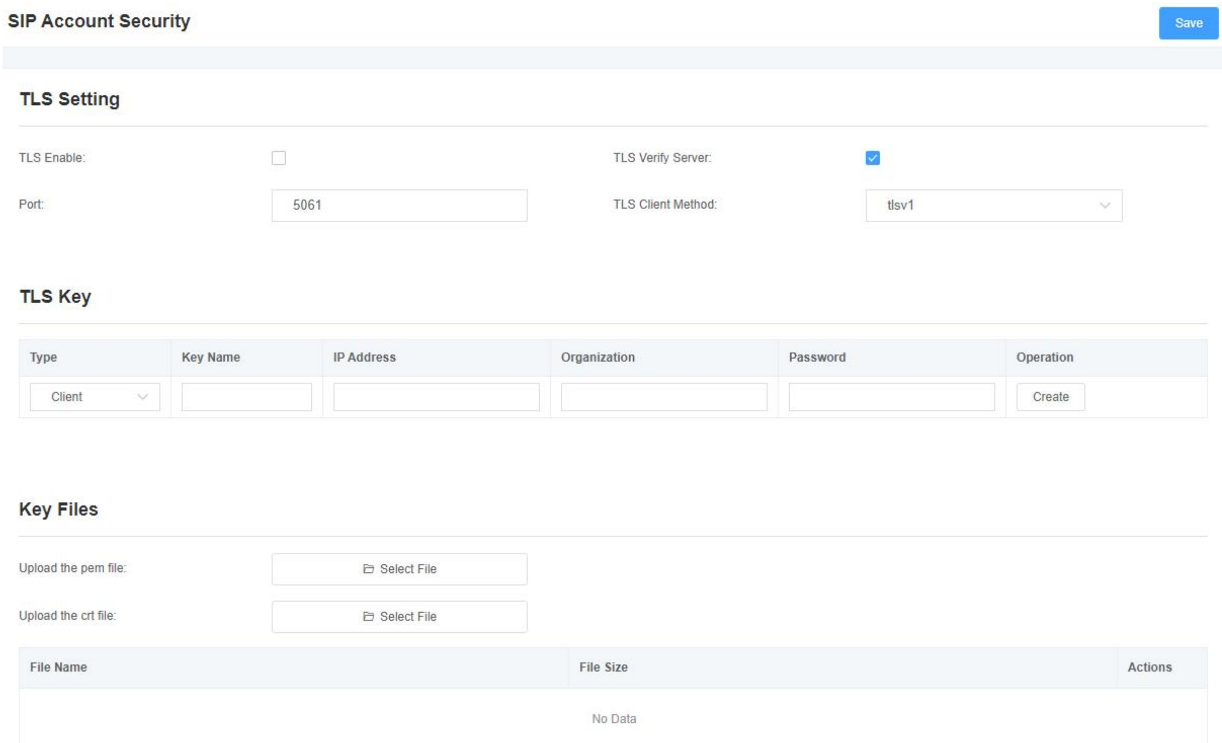


Table 4-5-1 Instruction of TLS

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).

5. Routing

The gateway has a friendly user interface and very flexible settings. It supports up to 512 routing rules and each routing rule supports up to 100 pairs of calling/called number filtering and conversion operations. It supports DID function (the use of DID function: how to use the T1/E1 gateway DID function of China Telecom). The gateway supports trunk group and trunk priority management.

5.1 Call Routing Rules

Figure 5-1-1 Routing Rules

Click "Add" , you can set up a new routing rule. Click "Edit" to modify the routing rule, and click "Delete" to delete the routing rule.

Figure 5-1-2 Example of Setup Routing Rule

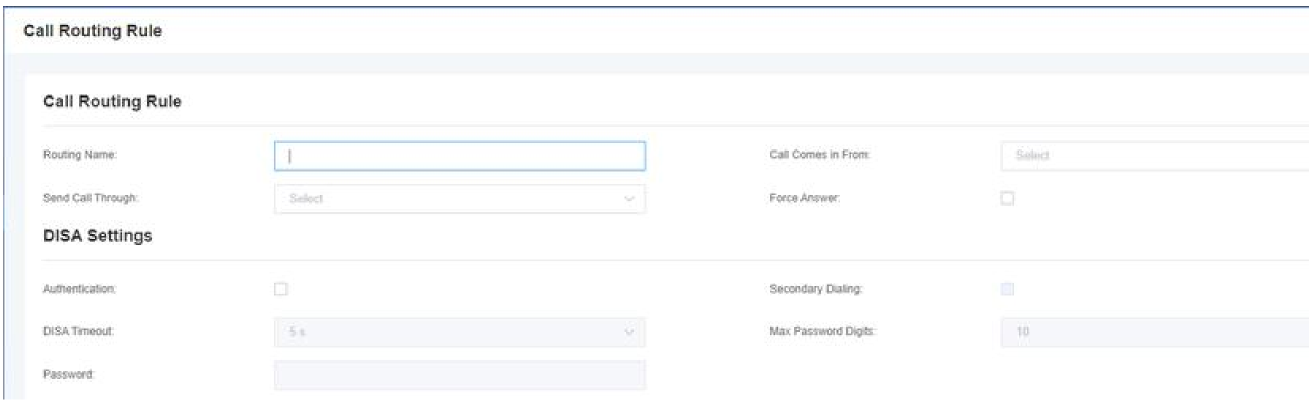


Table 5-1-1 Definition of Call Routing Rule

Options	Definition
Routing Name	This is a rule name. The type of match usually used to describe (for example, 'sip1 TO port1' or 'port1 TO sip1').
Call Is From	Source of the call.
Call Delivery	The destination to receive the incoming calls.
DISA Timeout	The specific setting time of DISA timeout.
Maximum Number of Digits In Password	Set the maximum number of password digits
Password	Set a password within the specified range

Figure 5-1-3 Advance Routing Rule

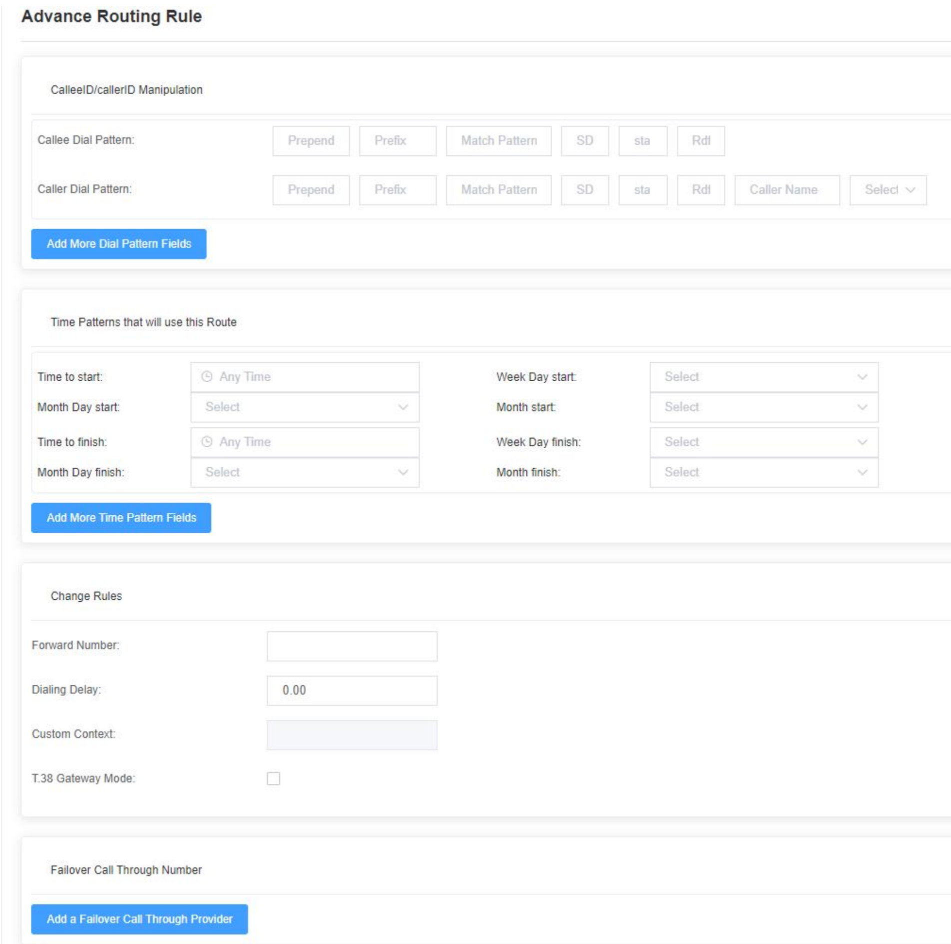


Table 5-1-2 Definition of Advanced Routing Rule

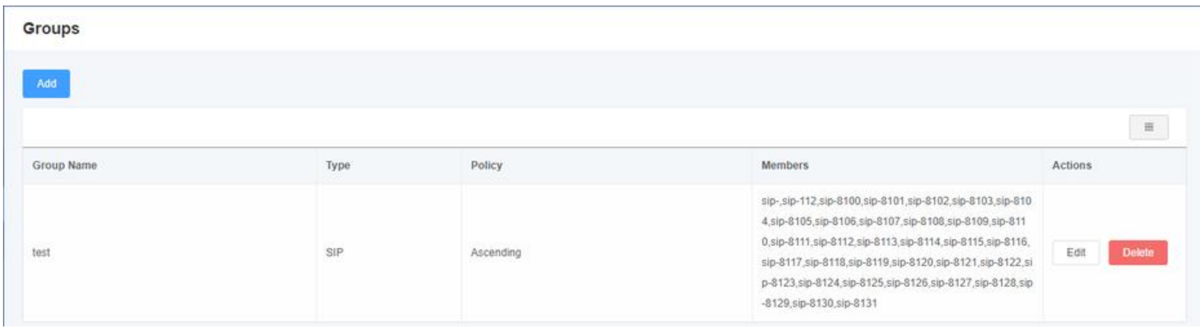
Options	**Definition**
CalleeID/callerID Manipulation	A Dial Pattern is a unique set of digits that will select this route and send the call to the designated trunks. If a dialed pattern matches this route, no subsequent routes will be tried. If Time Groups are enabled, subsequent routes will be checked 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) ***matches one or more digits** Prepend<add prefix>: The number added when the pattern matches successfully. If the dialed number matches the pattern specified in the subsequent column, the number will be added before being sent to the trunk. Prefix: Removed when the pattern is matched successfully. The dialed number is matched with the pattern specified in the subsequent column. Once the match is successful, the prefix will be removed from the number before being sent to the trunk. Match Pattern: The dialed number will be compared with the number in the "prefix +" this matching pattern. Once the match is successful, the matched pattern part of the dial will be sent to the trunks. SDfR<Delete digits from the right>: The number of digits to be deleted from the right end of the number. If this value of this item exceeds the length of the current number, the entire number will be deleted. RDfR<Reserved digits from the right>: The reserved digits from the right. StA<Add Suffix>: Add this number from the right end of the current number. Caller Name <caller display name>: Set your favorite caller name before sending this call to the terminal, allowing the use of local languages, such as Chinese and Latin.

Time Patterns that will use this Route	Time mode setting of routing rules.
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 by the order you specified.

5.2 Groups

Sometimes you want to make a call through one port, but you don't know if it is available, so you have to check which port is free. That would be troublesome. But with our product, you don't need to worry about it. You can combine many 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



You can click the "Add" button to set up a new group, if you want to modify an existing group, you can click the "Edit" button.

Figure 5-2-2 Create a Group

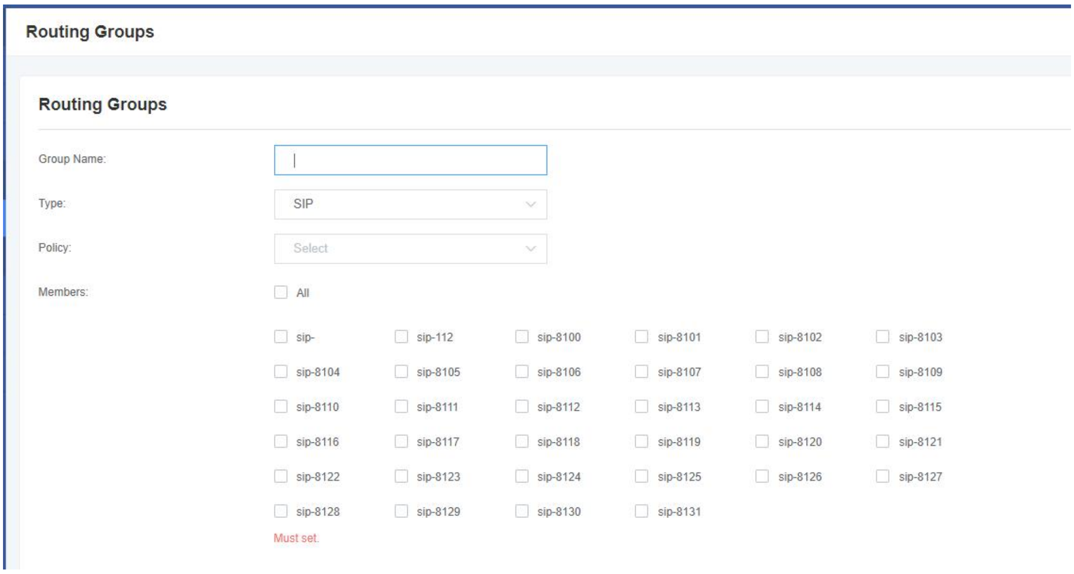


Figure 5-2-3 Modify a Group

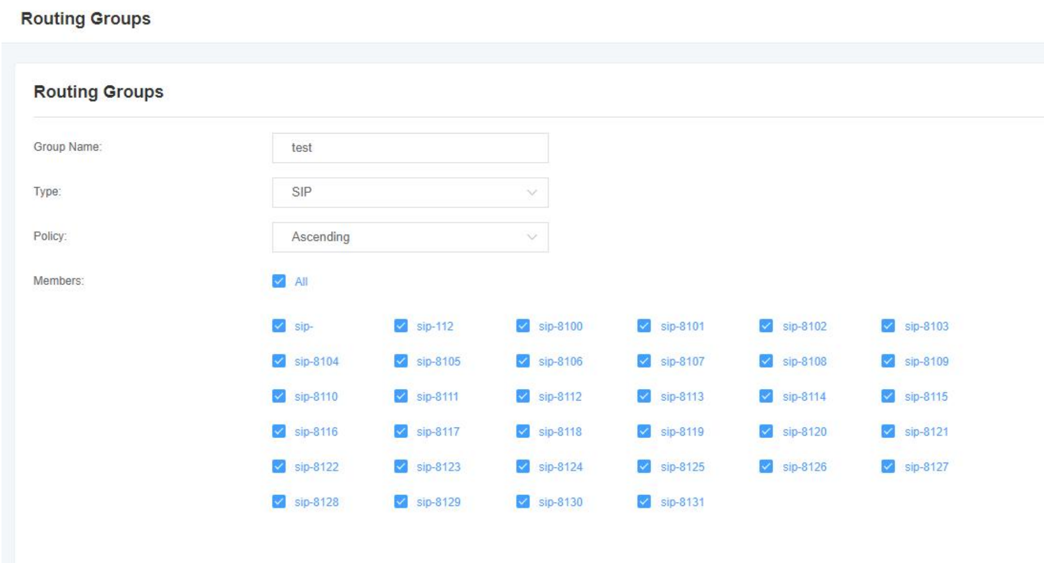


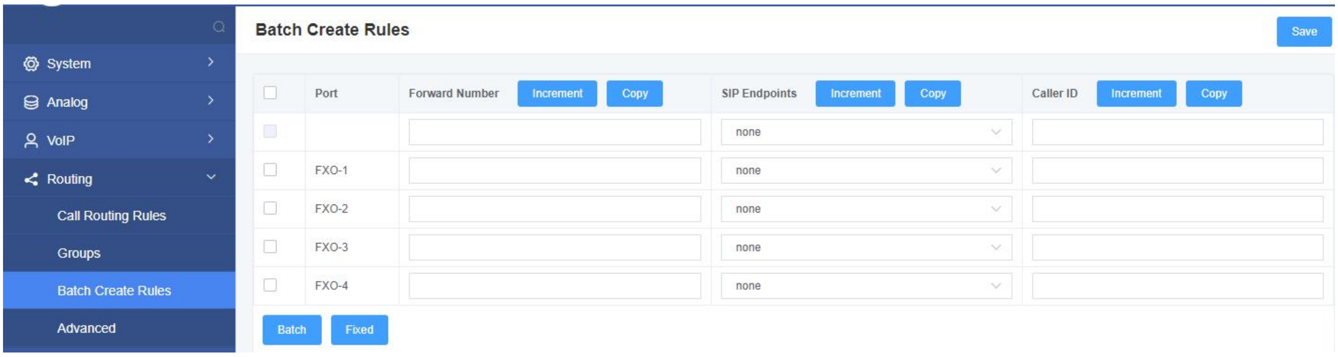
Table 5-2-1 Definition of Routing Groups

Options	Definition
Group description	The name of the route, used to describe the type of this call route, for example, sip1 TO port1 or port1 TO sip2.

5.3 Batch Create Rules

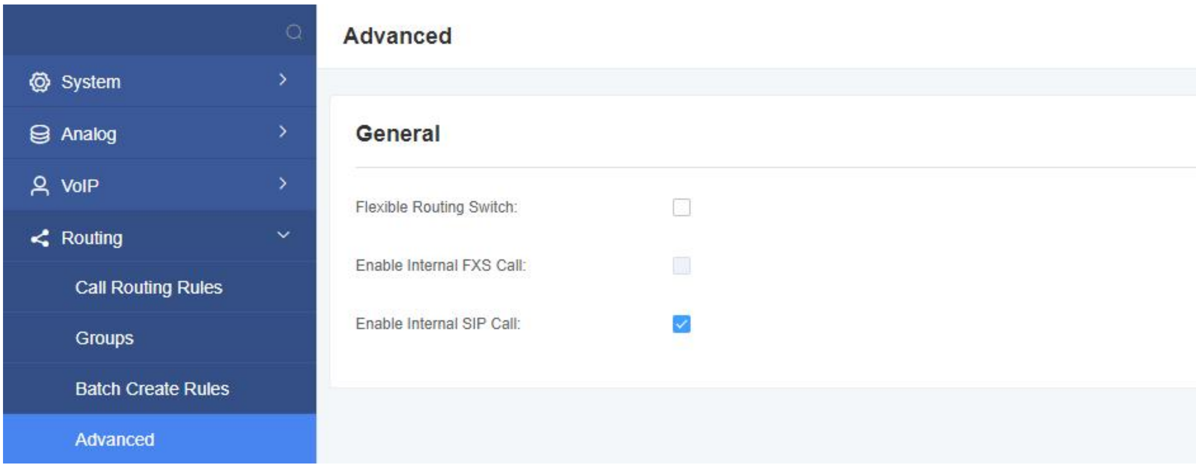
If you bind telephone for each FXO port and want to establish separate call routing for them. For convenience, you can create call routing rules for each FXO port at once in this page in batches.

Figure 5-3-1 Batch Create Rules



5.4 Advanced

Figure 5-4-1 General



6. Network

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

6.1 Basic Settings

There are three types of LAN port IP to choose from: Factory, Static and DHCP. The default type is: factory, the default IP is 172.16.99.1. If you forget the current IP, you can connect the phone to any FXS port of the analog gateway and dial "**" to query the current IP.

Figure 6-1-1 LAN Settings Interface

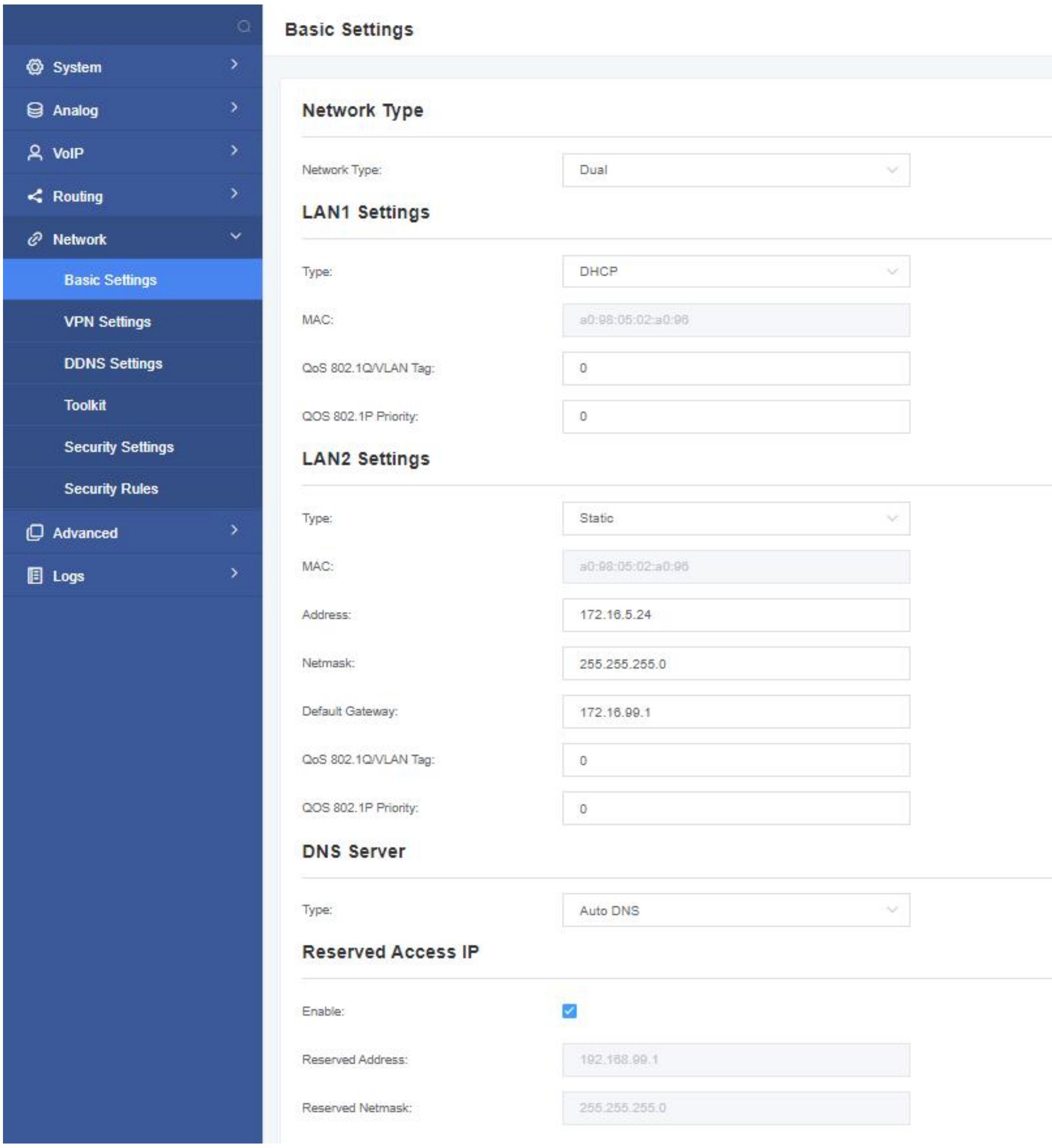


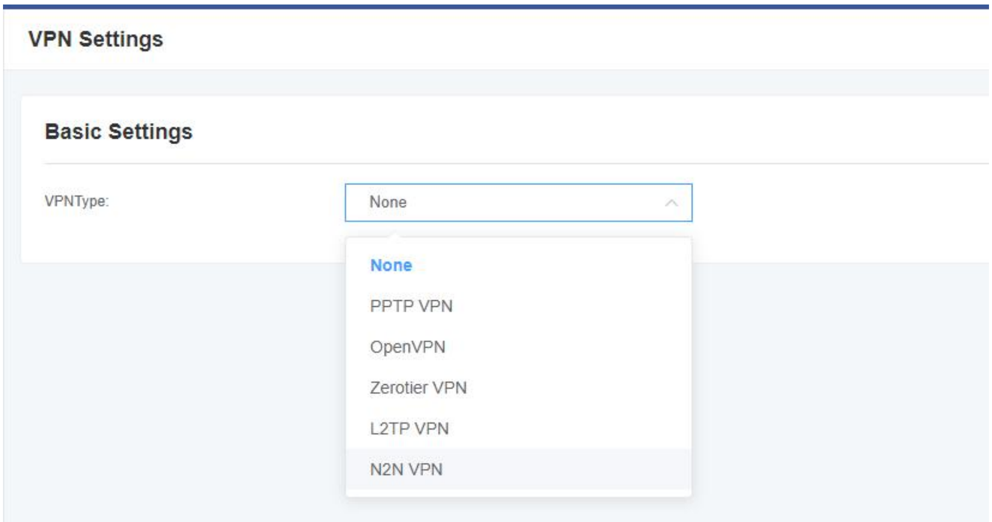
Table 6-1-1 Definition of Network Settings

Options	Definition
Network Type	The name of network interface.
Type	The method to get IP. Static: manually set up your gateway IP. DHCP: dynamically obtain the gateway IP address.
MAC	The physical address of the network interface.
Address	The IP address of your gateway.
Netmask	The subnet mask of your gateway.
Default Gateway	Default getaway IP address.
Reserved Access IP	List of domain name server IP addresses. This information is mainly obtained from the local network service provider.
Enable	Enable or disable the reserved IP address switch. ON(enabled), OFF(disabled)
Reserved Address	The reserved IP address for this gateway.
Reserved Netmask	The subnet mask of the reserved IP address.

6.2 VPN Settings

You can select VPN type and upload OpenVPN client configuration file or fill in PPTP VPN account information. If successful, you can see a VPN virtual network card on the system status page. You can refer to the parameter hints and sample configuration.

Figure 6-2-1 VPN Interface



6.3 DDNS Settings

You can enable or disable DDNS (Dynamic Domain Name Server) according to your needs

Figure 6-3-1 DDNS Interface

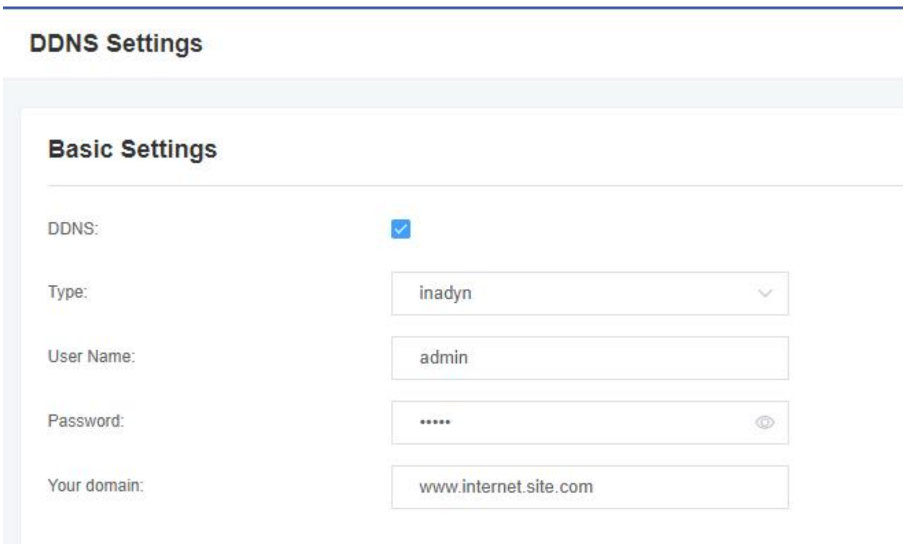


Table 6-3-1 Definition of DDNS Settings

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

6.4 Toolkit

This tool is used to detect the network connection, you can execute the Ping command on the web interface.

Figure 6-4-1 Network Connectivity Checking

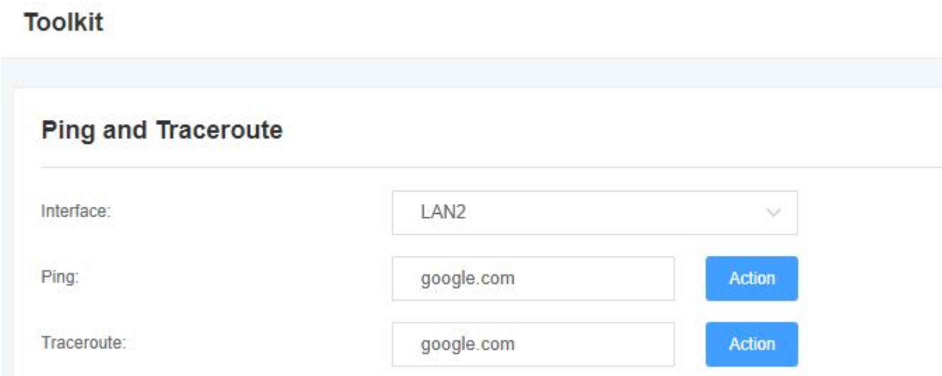


Figure 6-4-2 Channel Recording

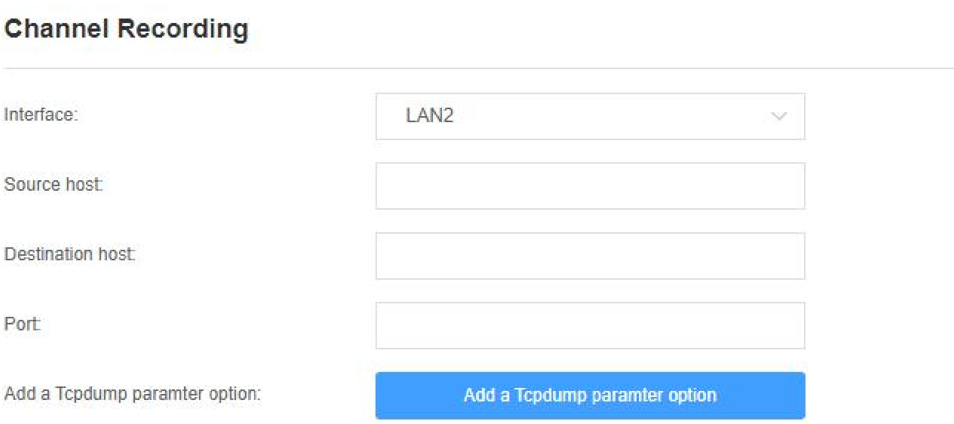


Figure 6-4-3 Capture Network Data

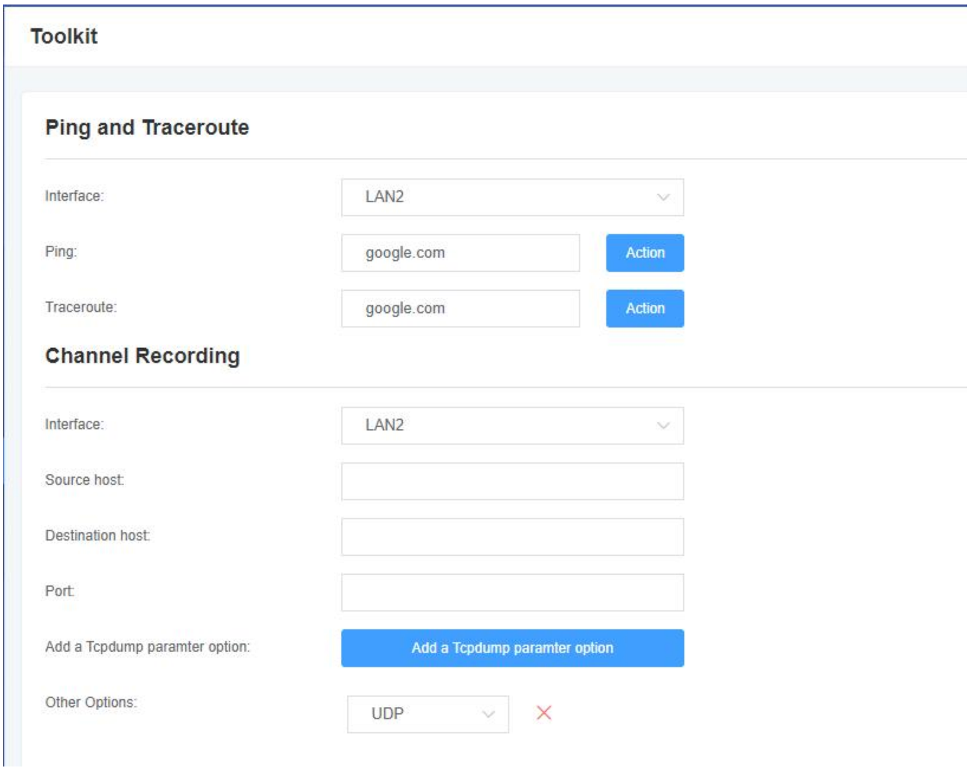
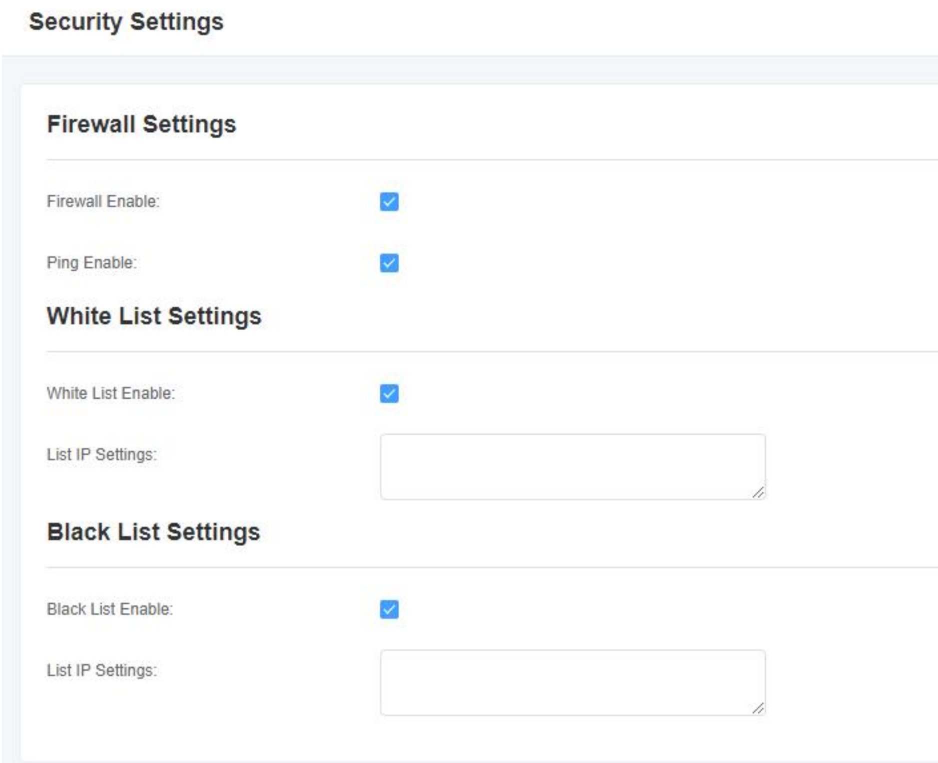


Table 6-4-1 Definition of Channel Recording

Options	**Definition**
Interface	The name of network interface.
Source Host Address	Specify the source address of the data you want to get
Destination Host	Specify the destination address you want to get data from
Port	Specify the port where you want to get data
Channel	Specify the channel number you want to get data
Tcpdump Option Parameter	The tool of tcpdump capture network data by parameter option specified.

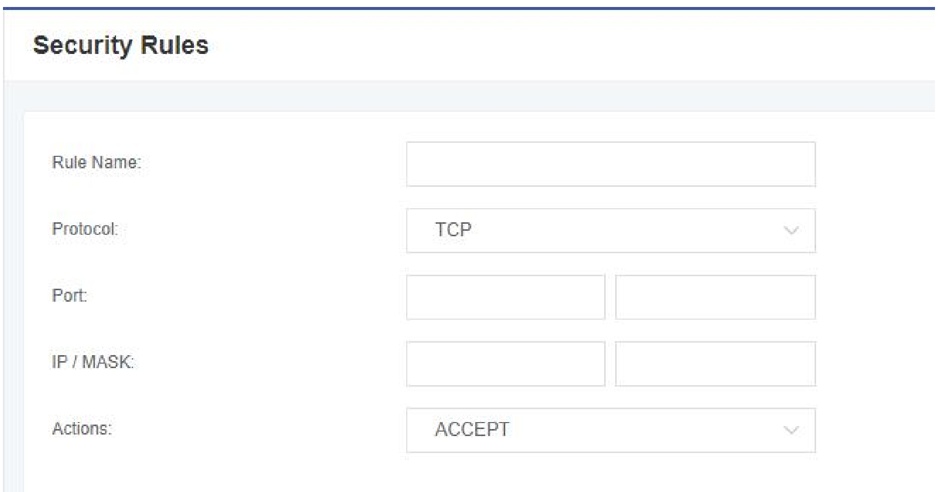
6.5 Security Settings

Figure 6-5-1 Security Settings Interface



6.6 Security Rules

Figure 6-5-1 Security Settings Interface



7. Advanced

7.1 API

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

Figure 7-1-1 API Interface

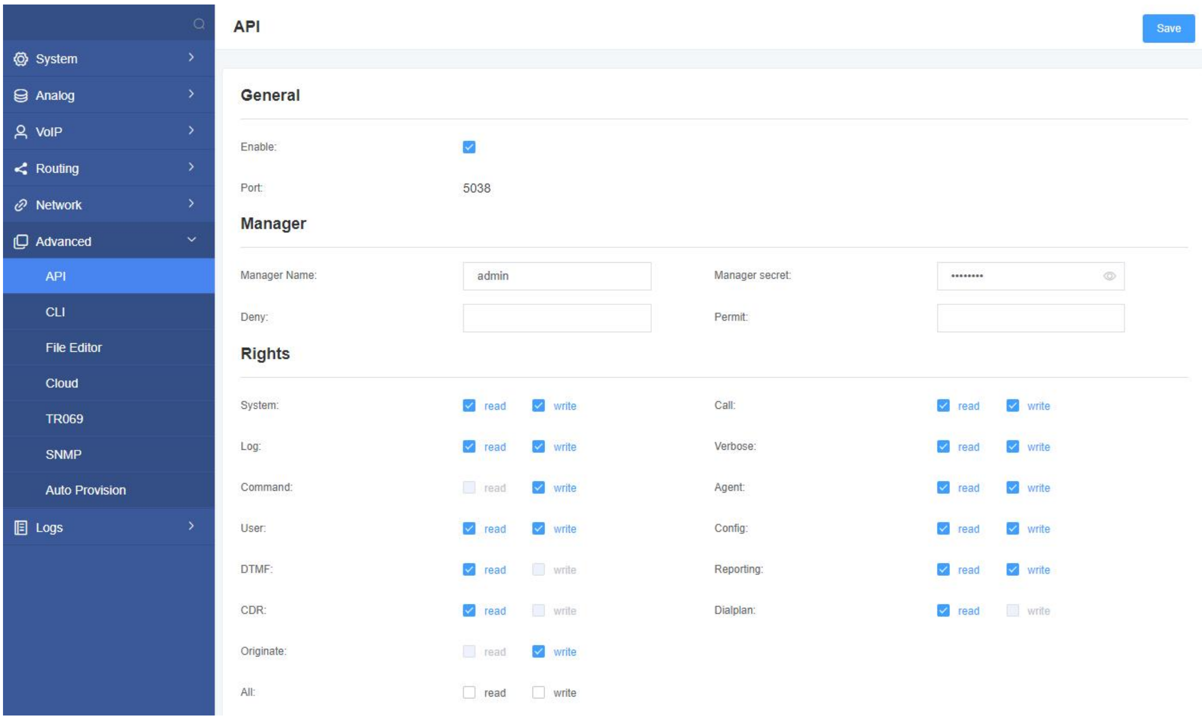


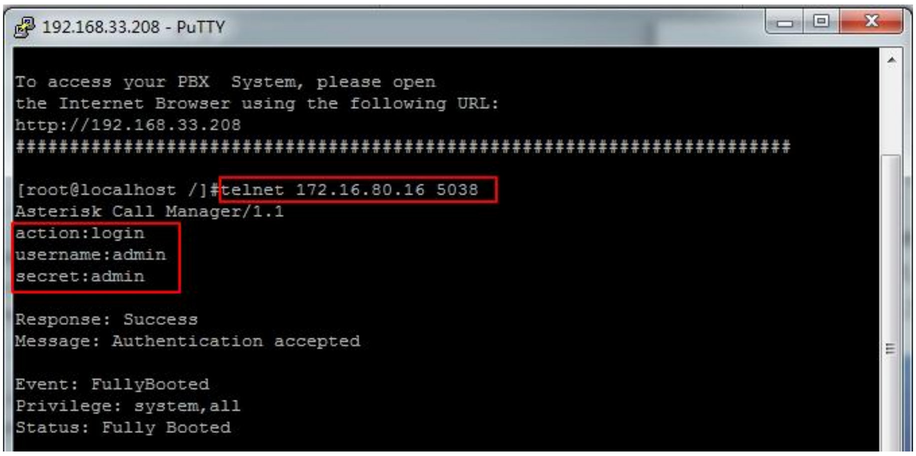
Table 7-1-1 Definition of Asterisk API

Options	**Definition**
Port	Network port number
Manager Name	Name of the manager cannot contain spaces
Manager Secret	Password for the administrator. Characters: Allowed characters "-_+.<>&0-9a-zA-Z". Length: 4-32 characters.
Deny	If you want to deny some hosts or networks, use char & as s

	eparator. Example: 0.0.0.0/0.0.0.0 or 192.168.1.0/255.255.255.0& 10.0.0.0/255.0.0.0
Permit	If you want to permit many hosts or network, use char & as s eparator.Example: 0.0.0.0/0.0.0.0 or 192.168.1.0/255.255.255.0&1 0.0.0.0/255.0.0.0
System	General information about the system and ability to run system management commands, such as Shutdown, Restart, and Reload.
Call	Channel information and setting information of the channel in use.
Log	Logging information. Read-only. (Defined but not yet used.)
Verbose	Verbose information. Read-only. (Defined but not yet used.)
Command	CLI commands allowed to run. (Read-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.

Refer to the above configuration diagram, the host 172.16.80.16/255.255.0.0 has been allowed to enter the gateway API, and the port number is 5038.

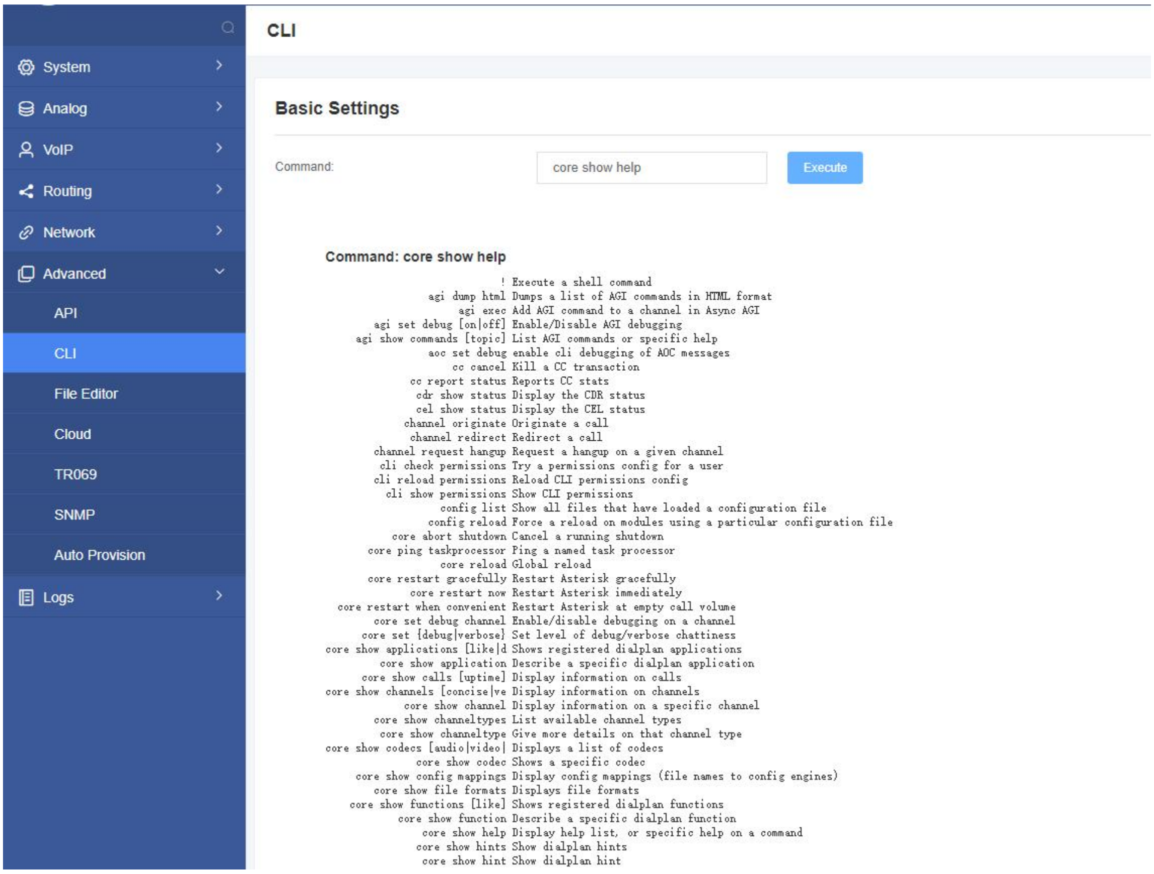
Figure 7-1-2 Putty Display Image



7.2 CLI

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

Figure 7-2-1 Asterisk Command Interface

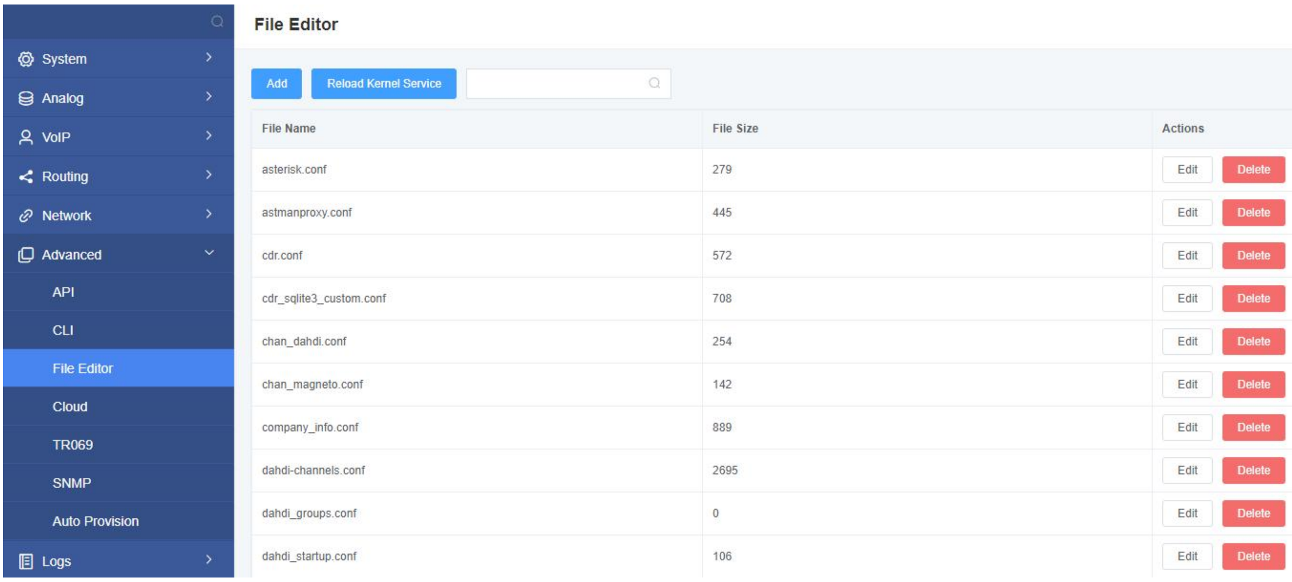


For example: enter "help" or "?" in the command bar, after execution, the page will prompt for executable commands, as shown in the figure above.

7.3 File Editor

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

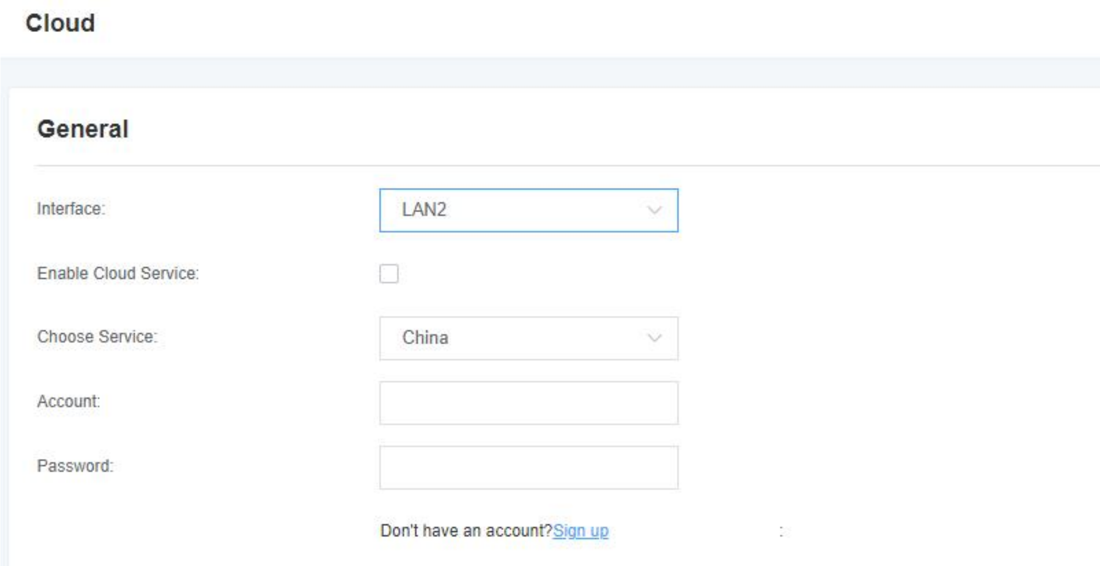
Figure 7-3-1 Configuration Files List



Notice: After modifying the configuration file, Asterisk needs to be reloaded.

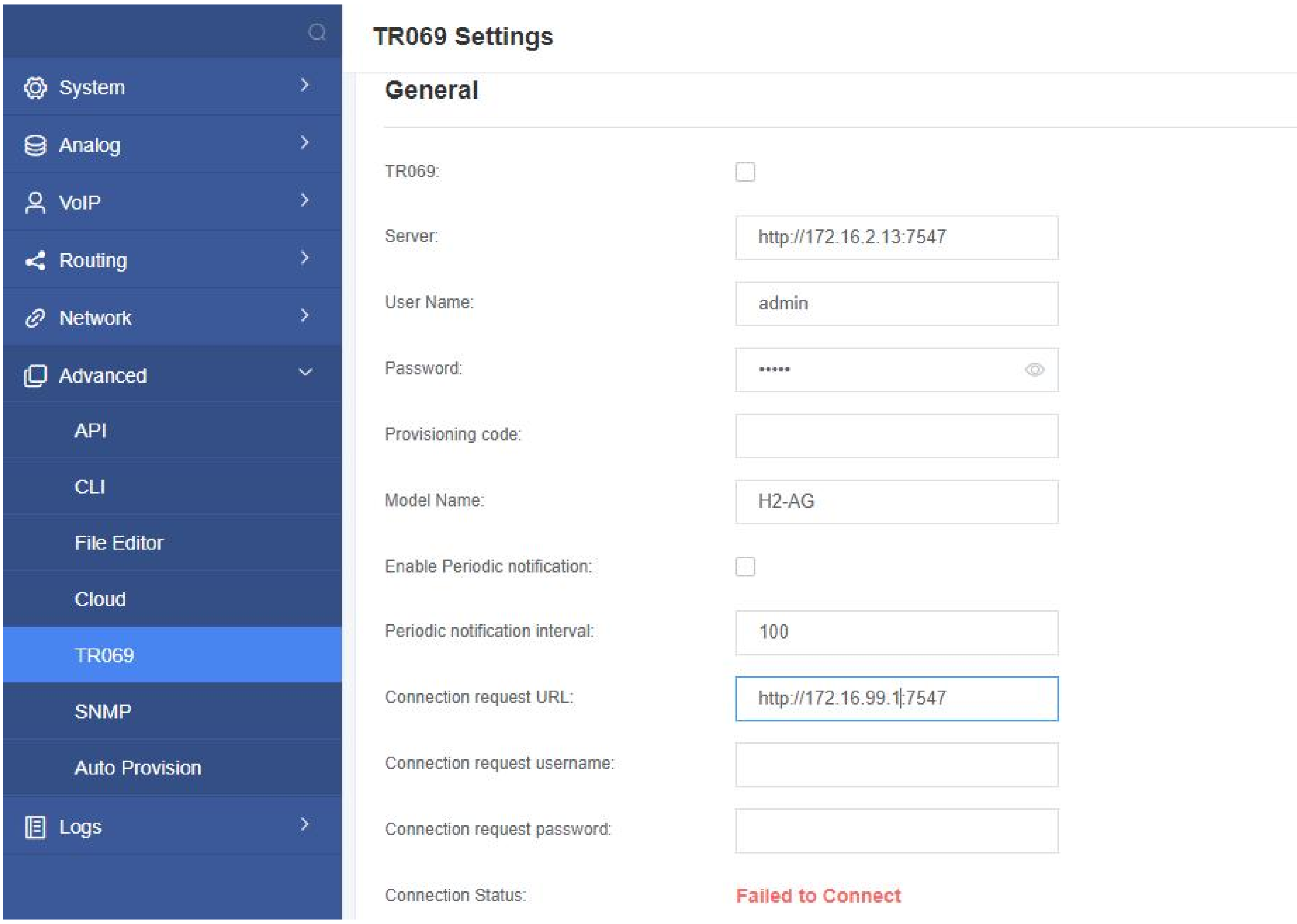
7.4 Cloud Management

Figure 7-4-1 Cloud Management Interface



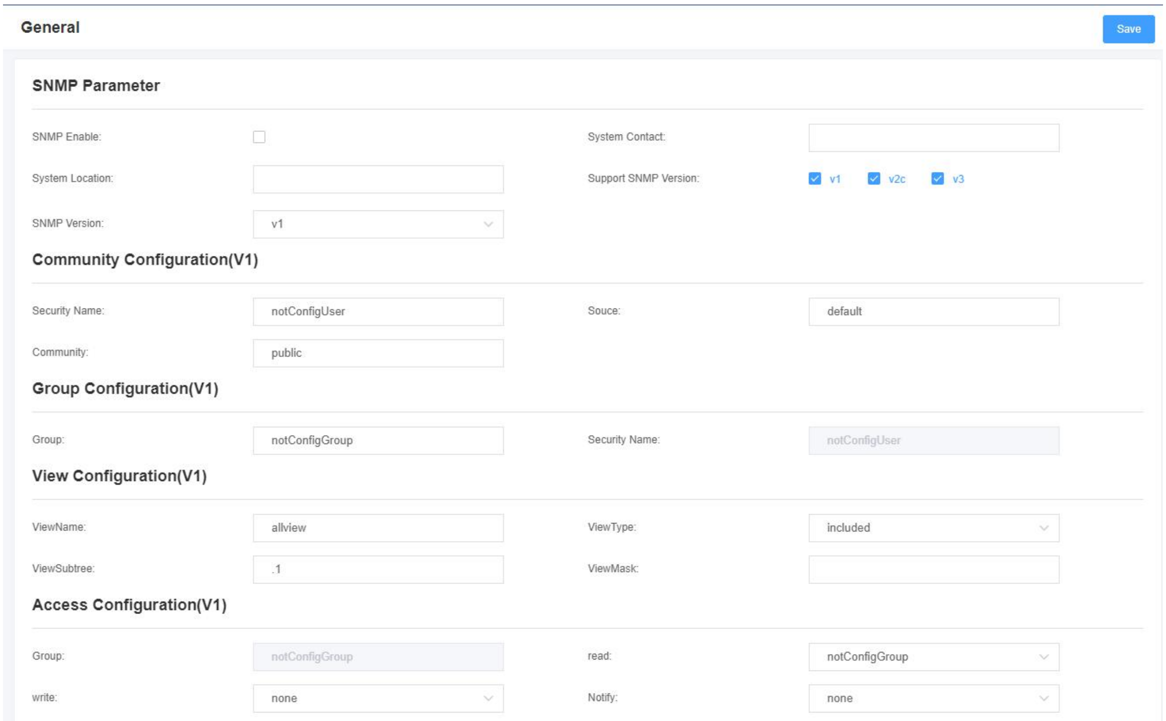
7.5 TR069

Figure 7-5-1 TR069 Interface



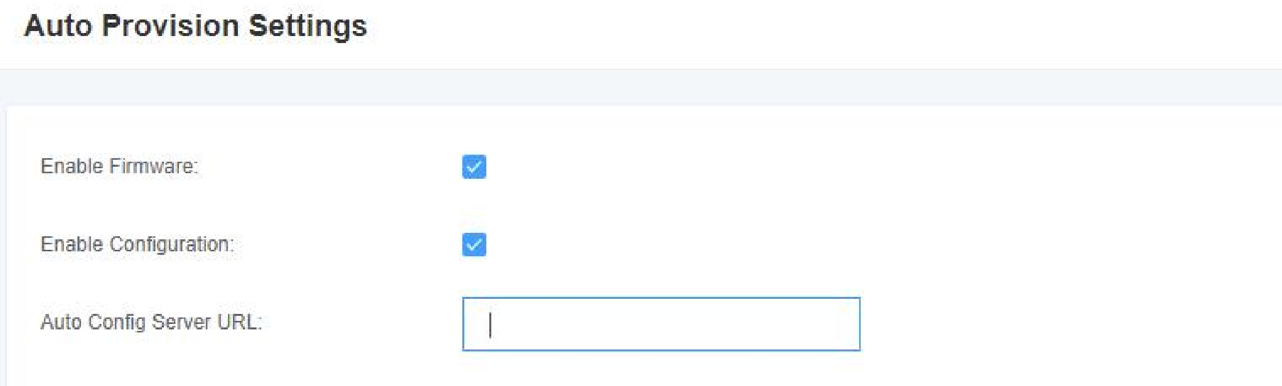
7.6 SNMP

Figure 7-6-1 SNMP Interface



7.7 Auto Provision Settings

Figure 7-7-1 Auto Provision Settings



8. Logs

8.1 Log Settings

In the log setting interface, open the corresponding log option, and you can view different logs in the corresponding interface. Take the system log as an example.

Figure 8-1-1 Logs Settings

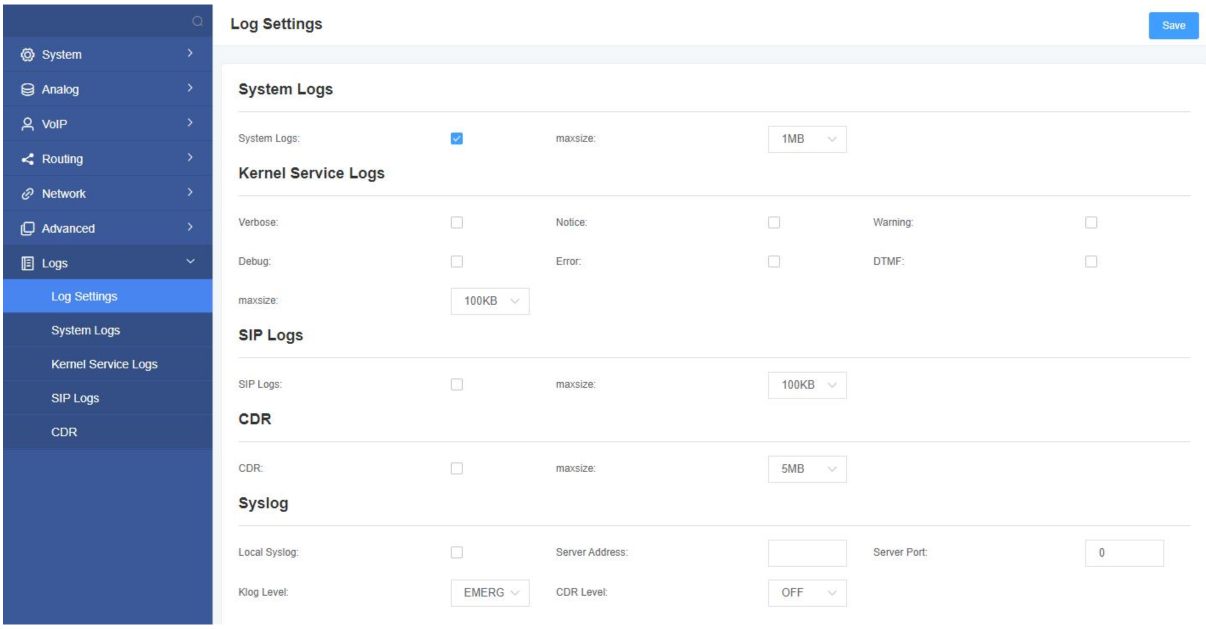


Figure 8-1-2 System Log Output

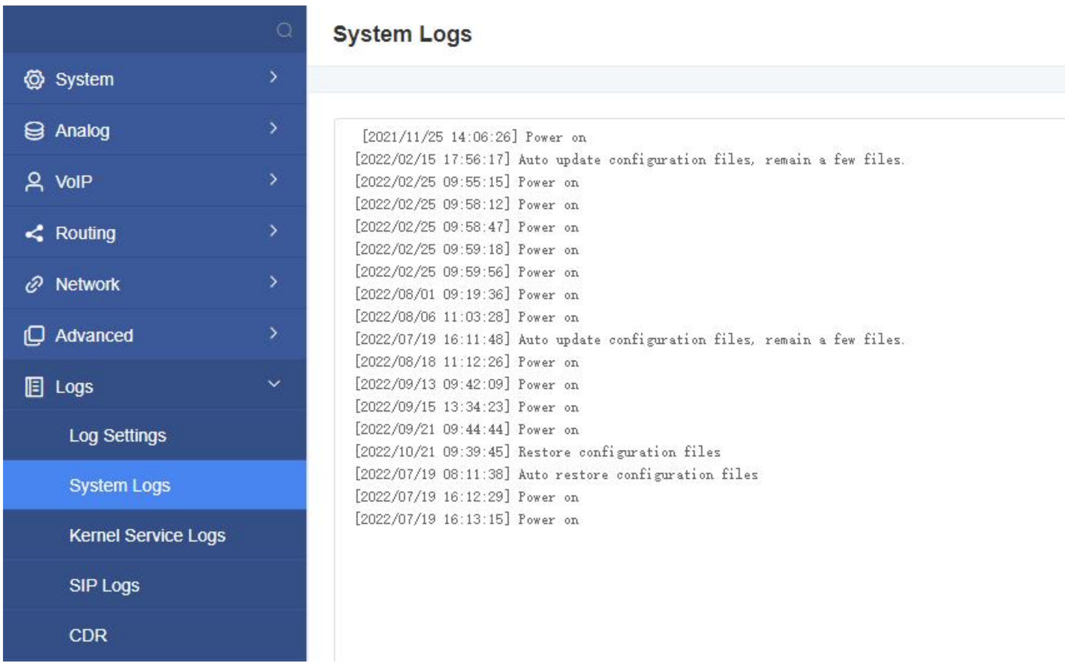


Table 8-1-1 Definition of Log Option

Options	Definition
System Logs	Whether to open the 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)	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.
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.

8.2 CDR

You can browse the details of each call record on this page. If you need to search for a specific record, you can use the filter function.

Figure 8-2-1 Call Detail Record