VI Series Network Intercom

User Guide

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Contents

1. Preface

1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and manage the VI Series Network Intercom. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the VI Series Network Intercom.

1.2 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
1.1.0	1.1.0	Updated operating instructions for software version v1.1.0	Jul, 2025

2. Overview

2.1 Product Overview

The ZYCOO VI Series Network Intercom is a rugged, vandal-proof device, available in three models: the audio model VI-A05, the video model VI-V05, and the dual-button model VI-D05. The VI Series intercom supports PoE and offers flexible installation options, supporting both wall-mount installation (with EA-MB2 mounting box) and flush-mount installation (compatible with standard 2-Gang electrical boxes).

In access control and emergency call scenarios, visitors can establish two-way communication with the operator simply by pressing the call button. The operator can remotely grant access via telephone keypad. Additionally, the VI-V05 provides high-definition video calls, while the VI-D05 is equipped with dual buttons that can be configured for different call destinations to meet diverse communication needs.

All models in the VI Series are equipped with a tamper alarm mechanism. If the device is forcibly removed, a built-in buzzer will sound, which can trigger an alert to security personnel for immediate response.

2.2 Product Specifications

VI Series Network Intercom Specifications		
Sensitivity	96dB ± 3dB/1m/1W	
THD (@1kHz)	<1%	
SNR (20kHz BW)	97dB	

Frequency Response	100Hz - 20KHz	 
Rated Power	4Ω 5W	
Microphone Sensitivity	-36 ± 3dB	

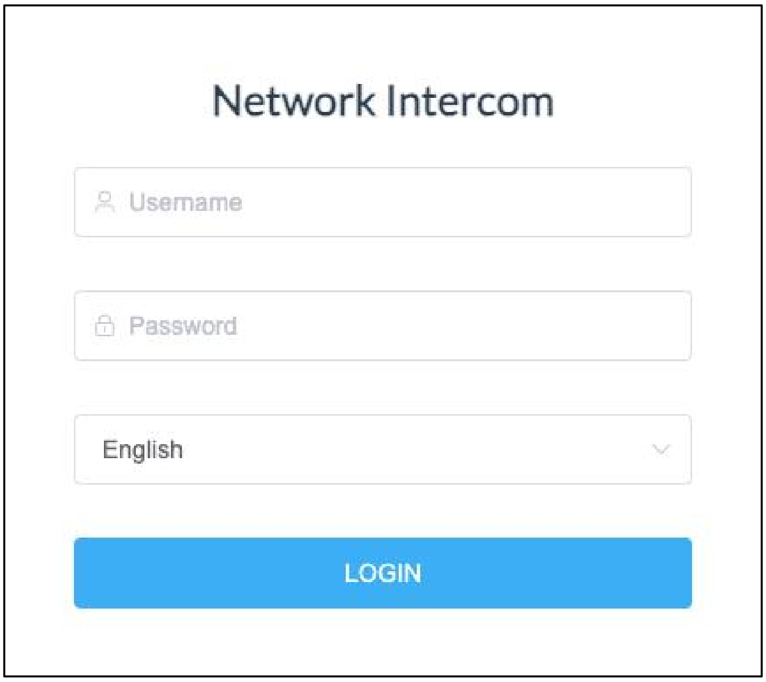
3. Login the Device

3.1 Accessing the Web GUI

VI intercom obtains the IP address through DHCP by default, please ensure that there is an available DHCP server in your LAN (If DHCP fails to obtain an address, it will use a static IP address: 192.168.1.101), enter the IP address in the browser to access the device's Web management interface.

Default username: admin
Default password: admin

For the safety purpose, it is recommended to change the default password on the first login, please go to **System --> Password Settings** page to change the password.

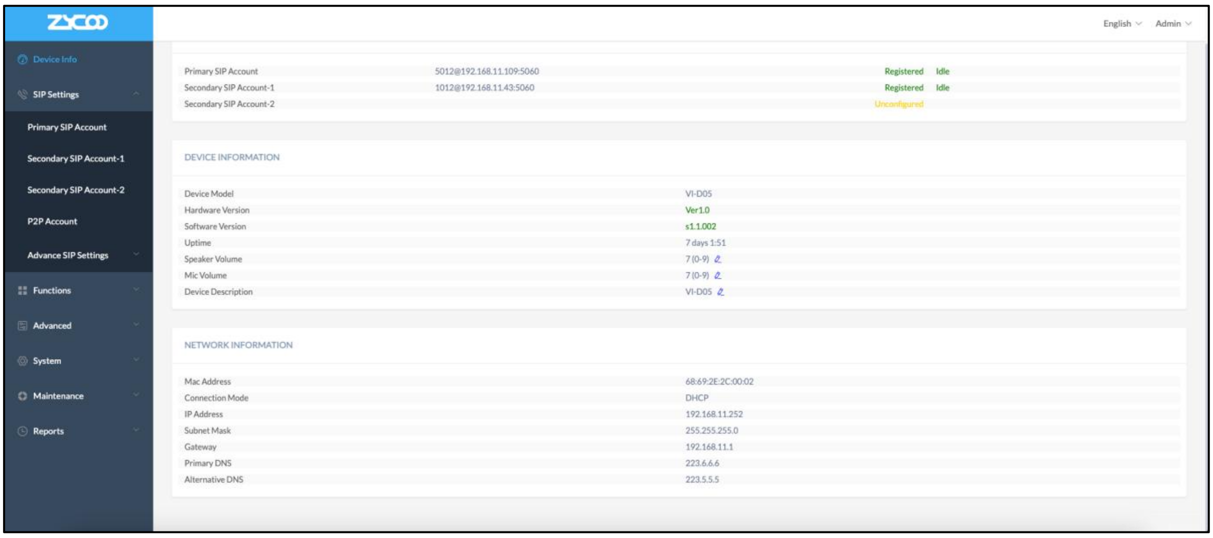


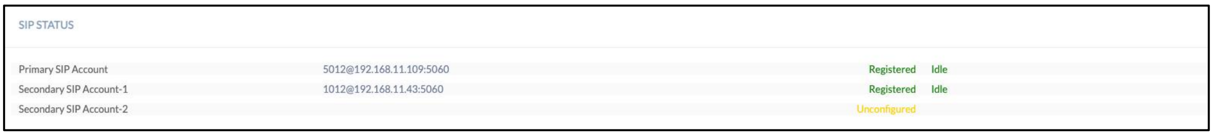
Login Interface

After entering the correct username and password, you can log in to the device's web management interface.

3.2 Device Info

After successful login, you will see the information interface of the device, and you can view the basic information of the device.





SIP STATUS

Primary SIP Account	5012@192.168.11.109:5060	Registered	Idle
Secondary SIP Account-1	1012@192.168.11.43:5060	Registered	Idle
Secondary SIP Account-2		Unconfigured	

SIP Status

- **SIP Account:** Display the SIP number configured on this device.
- **SIP Server:** Display the SIP server (Such as ZYCOO IP Audio Center or IP PBX) address.
- **Register Status:** Display the SIP number registration status.

DEVICE INFORMATION

Device Model	VI-D05
Hardware Version	Ver1.0
Software Version	s1.1.002
Uptime	7 days 1:51
Speaker Volume	7 (0-9)
Mic Volume	7 (0-9)
Device Description	VI-D05

Device Information

- **Device Model:** Displays the model of the device.

- **Hardware Version:** Displays the hardware version number of the device.
- **Software Version:** Display the system version number of the device.
- **UpTime:** Displays the total duration the device has been continuously operating.
- **Speaker Volume:** Displays the current volume of the device.
- **Mic Volume:** Displays the current device microphone input volume.
- **Device Description:** Remark the device information. The description will be displayed in a browser tab. After the Device Description is set, the description will be displayed in the browser tab, which is convenient for distinguishing different terminals when there are many terminal configuration pages.

NETWORK INFORMATION

Mac Address	68:69:2E:2C:00:02
Connection Mode	DHCP
IP Address	192.168.11.252
Subnet Mask	255.255.255.0
Gateway	192.168.11.1
Primary DNS	223.6.6.6
Alternative DNS	223.5.5.5

Network Information

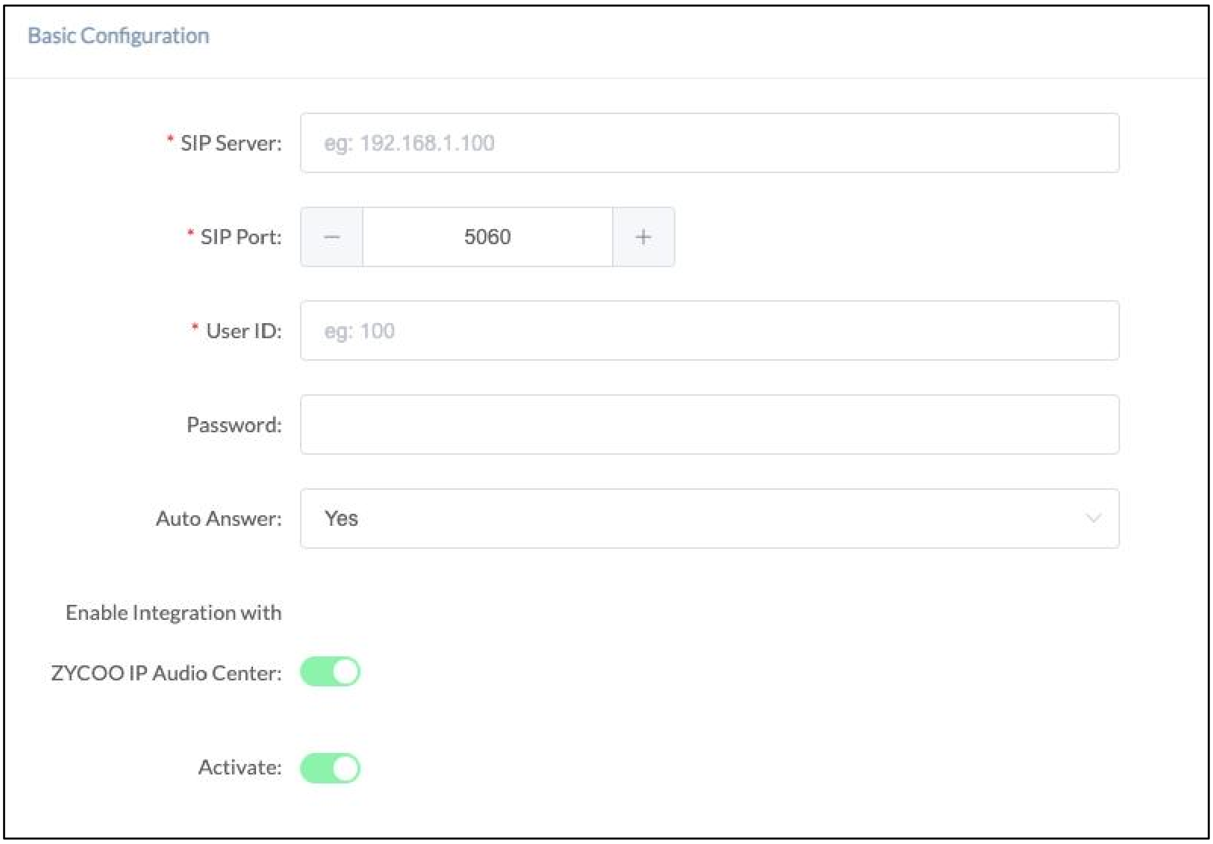
- **Mac Address:** Display the MAC address of the current device.
- **Connection Mode:** Display the network acquisition method of the device, DHCP (dynamic acquisition) or STATIC (static configuration).
- **IP Address:** The current IP address of the device.
- **Subnet Mask:** The current subnet mask of the device.
- **Gateway:** The gateway address currently used by the device.
- **Primary DNS:** The primary domain name server address used by the device.
- **Alternative DNS:** The secondary domain name server address used by the device.

4. SIP Settings

4.1 SIP Account Settings

There are three (3) SIP accounts under the SIP Settings, one (1) primary and two (2) secondary for the use of different SIP accounts to proceed with various tasks. If the current device needs to cooperate with the ZYCOO IP Audio Center, please turn on the 'Enable Integration with ZYCOO IP Audio Center' option.

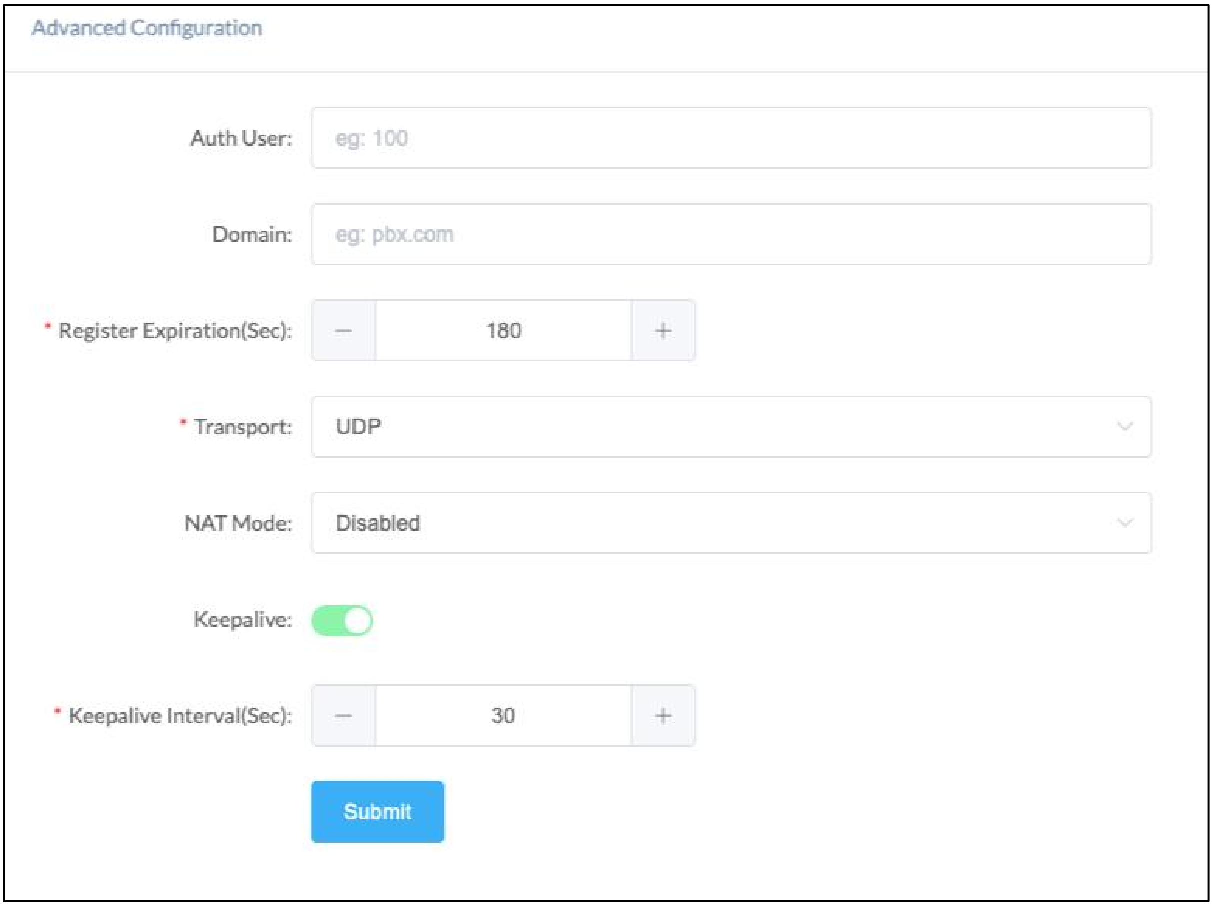
Please go to **SIP Settings --> Primary SIP Account / Secondary SIP Account-1 / Secondary SIP Account-2** page.



SIP Account - Basic Configuration

- **SIP Server:** Enter the IP address or domain name of the SIP server.
- **SIP Port:** The default SIP port is 5060. If your SIP server uses a different port, update this setting accordingly.
- **User ID:** Enter the SIP account number provided by your SIP server.

- **Password:** Enter the password for authorizing the SIP account.
- **Auto Answer:** Options include Yes, No, or Answer Delay. The default setting is 'Yes.'
- **Enable Integration with ZYCOO IP Audio Center:** Disabled by default. Enable this option when connecting to the ZYCOO IP Audio Center. This option is available only for the primary SIP account.
- **Activate:** Once enabled, the account will be activated and registered with the SIP server.



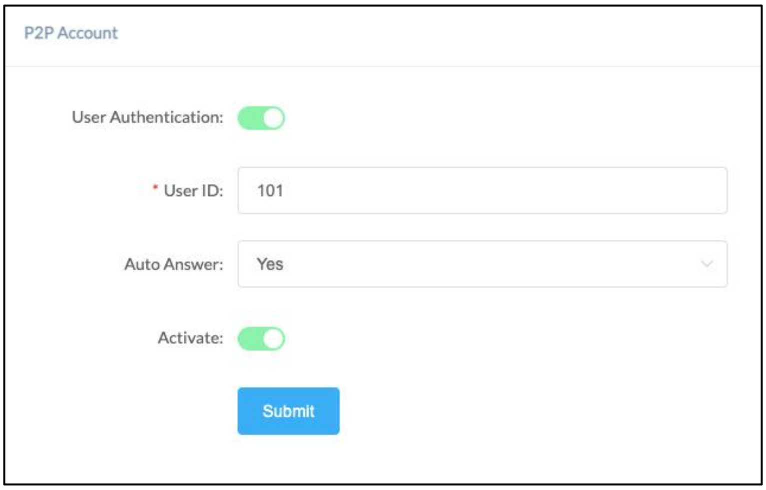
SIP Account - Advanced Configuration

- **Auth User:** Enter the authorized username for the SIP account.
- **Domain:** Enter the SIP Domain.
- **Register Expiration (sec):** Set the SIP registration expiration time, with a default of 180 seconds.

- **Transport:** Choose the transport protocol: UDP, TCP, or TLS.
- **NAT Mode:** Select the NAT mode and provide the necessary details. Supports STUN, TURN, and ICE modes.
- **Keepalive:** Enable the SIP keepalive function to maintain an active connection.
- **Keepalive Interval(Sec):** Set the interval for SIP keepalive messages.

4.2 P2P Account Settings

P2P stands for Peer to Peer. In a P2P network, the peers are connected to each other via the Internet, files can share, or peers can call each other directly between systems on the network without the need for a central server.
Please go to **SIP Settings --> P2P Account Settings** page to configure the P2P settings first. After configuring the P2P account, it can be used with the Outgoing Call feature in **Basic Settings --> I/O Settings**, or use the Outgoing API in **Basic Settings ---> API Settings** to make a P2P call.



P2P Account

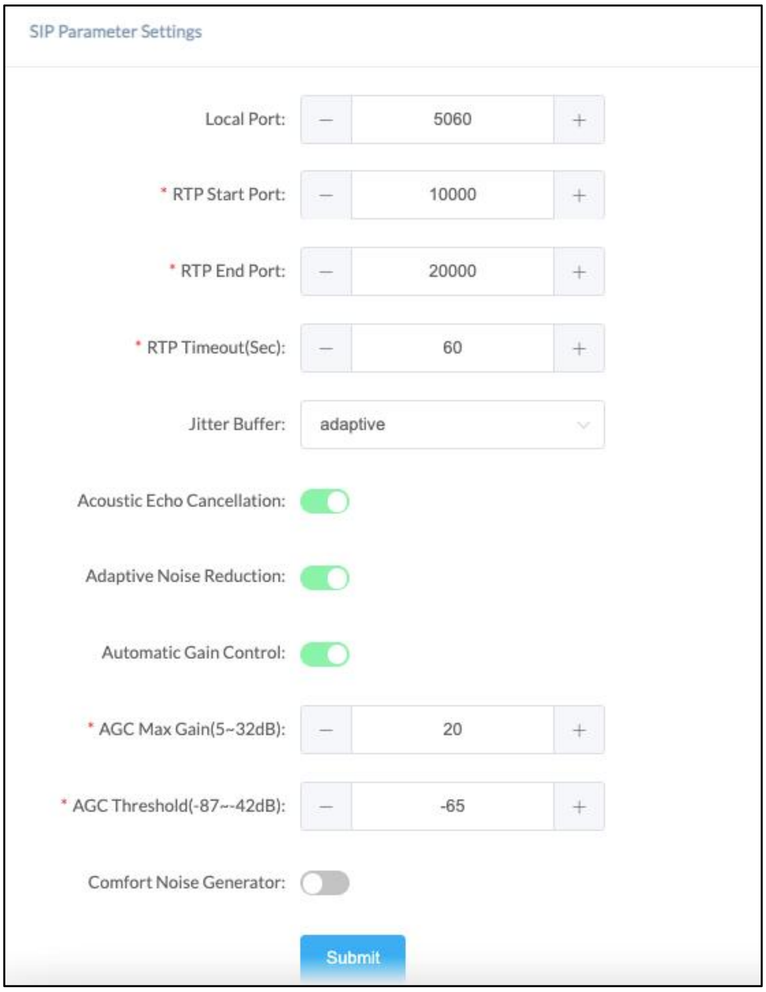
- **User Authentication:** Enable/Disable P2P authentication. If disabled, you can directly enter this device's IP address in the target field of the peer device. If enabled, you must use the following format in the target field of the peer device: This device's P2P User ID + IP address (e.g., 101@192.168.1.101).

- **User ID:** The User ID will be displayed as the outgoing number when calling out, or the number that peer device needs to dial. You must use the following format in the target field of the peer device: This device's P2P User ID + IP address (e.g., 101@192.168.1.101).
- **Auto Answer:** Options include Yes, No, or Answer Delay. The default setting is 'Yes.'
- **Activate:** Enable/Disable the P2P feature.

4.3 Advance SIP Settings

To configure some advanced parameters of the SIP protocol, please go to **SIP Settings --> Advance SIP Settings** page.

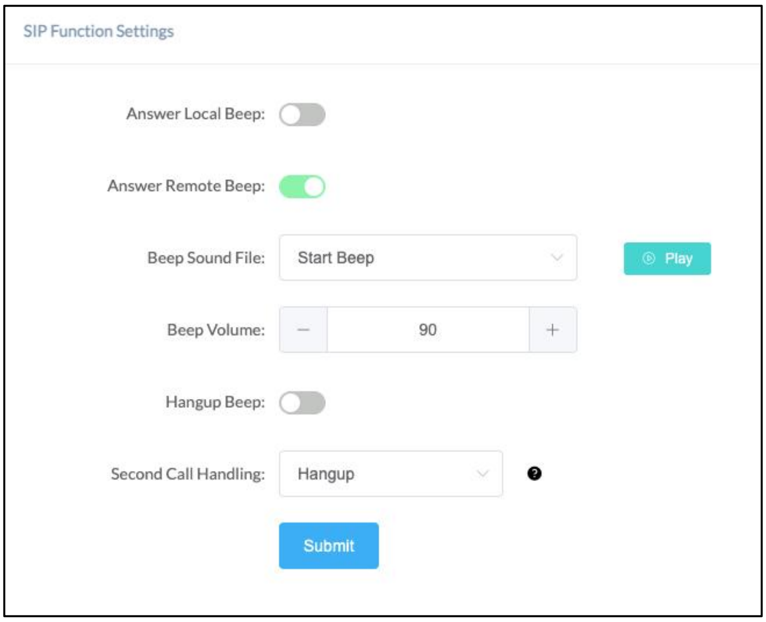
4.3.1 SIP Parameter Settings



SIP Parameter Settings

- **Local Port:** This setting represents the port used to receive SIP packets.
- **RTP Start Port:** This setting represents the starting RTP port that will use for media sessions.
- **RTP End Port:** This setting represents the end RTP port that the system will use for media sessions.
- **RTP Timeout (sec):** This setting means that within a specific time range, if the system does not receive the RTP stream, the call will end.
- **Jitt Buffer:** This setting represents the Jitter buffer where voice packets can be collected, stored, and sent to the voice processor in even intervals. Three options are provided, off/adaptive/fixed. A fixed jitter buffer adds a fixed delay to voice packets. An adaptive jitter buffer can adjust based on the delays in the network.
- **Acoustic Echo Cancellation**: After enabling this feature, echo noise can be suppressed through algorithms.
- **Adaptive Noise Reduction:** After enabling this feature, algorithms can suppress environmental noise collected by microphones.
- **Automatic Gain Control:** After enabling this feature, the voice signal can be automatically enhanced according to the distance and size of the voice source.
- **Comfort Noise Generator:** After enabling this feature, comfortable white noise can be added during calls.

4.3.2 SIP Function Settings



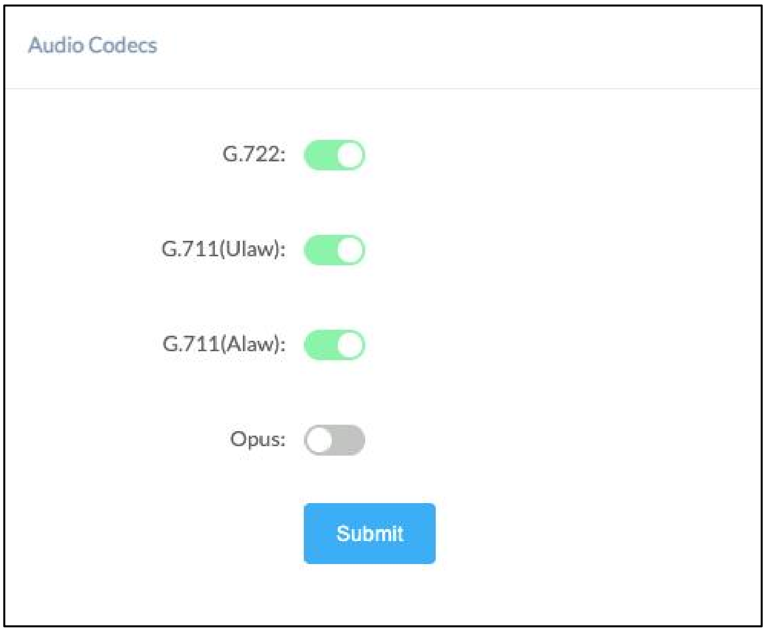
SIP Function Settings

- **Answer Local Beep:** If this setting is enabled, the selected beep sound will be played first on the local device side after the SIP session is answered.
- **Beep Sound File:** Select a specific beep sound file. Click the Play button, you could listen to this audio file.
- **Beep Volume:** Set the volume of the beep.
- **Answer Remote Beep:** If this setting is enabled, the selected beep sound will be played first on the remote device side after the SIP session is answered.
- **Hangup Beep:** If this setting is enabled, the selected beep sound will be played on the local device side before the SIP session is completely hung up.
- **Second Call Hanging:** Set the volume of the beep.Options for handling the second call:

Hangup: Directly hang up the second call. Hold: Hold the first call and automatically resume it after ending the second call. Merge: Join the second call into the first call, allowing all parties to speak simultaneously.

4.3.3 Audio Codecs

VI supports 4 audio codecs: G.722 (wideband codec), G.711(Ulaw), G.711(Alaw), and Opus.

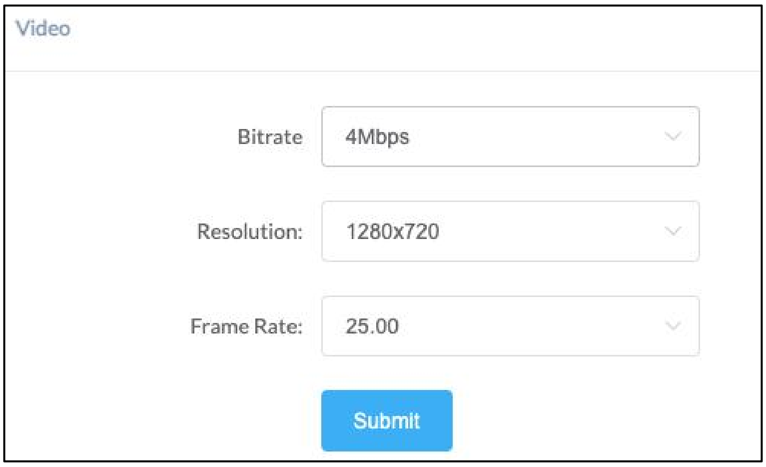


Audio Codecs

Please keep at least one codec enabled and supported by the SIP server, otherwise, SIP paging will not work.

4.3.4 Video Settings (VI-V05 Only)

The VI-V05 is equipped with a built-in HD camera. You can configure the camera parameters on this page to optimize video performance based on your network conditions and application scenarios.



Video

- **Bitrate:** Select the desired video bitrate to balance video quality and bandwidth usage. Available options: 512 kbps, 1 Mbps, 1.6 Mbps, 2 Mbps, 4 Mbps, 6 Mbps, 8 Mbps.
- **Resolution:** Choose from the following resolution settings depending on your video quality requirements: 640 × 360, 640 × 480, 1024 × 576, 1280 × 720, 1920 × 1080, 2560 × 1440.
- **Frame Rate:** Adjust the frame rate to optimize video smoothness and performance: 30 fps, 25 fps, 15 fps.

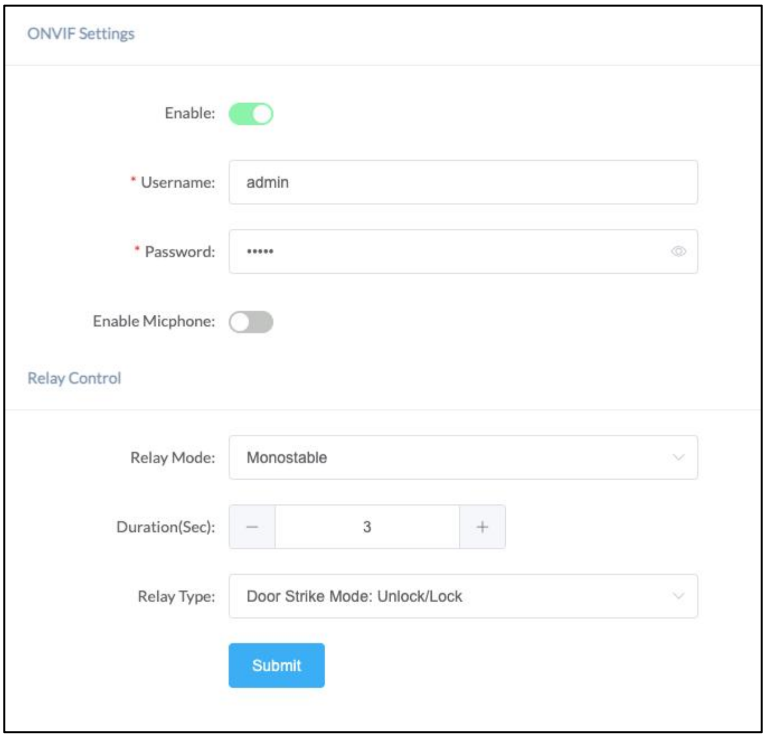
Note: To ensure smooth video streaming, choose a suitable bitrate, resolution, and frame rate according to your network environment. Inappropriate settings may cause issues when accessing video feeds.

5. Function Settings

5.1 ONVIF Settings

ONVIF provides and promotes standardize interfaces for effective interoperability of IP-based physical security products. If the user has installed a VMS that supports ONVIF, they can register ZYCOO network devices that support ONVIF on it for operation.

Please go to **Functions ---> ONVIF Settings** to configure the ONVIF settings.



ONVIF & Relay Control Settings

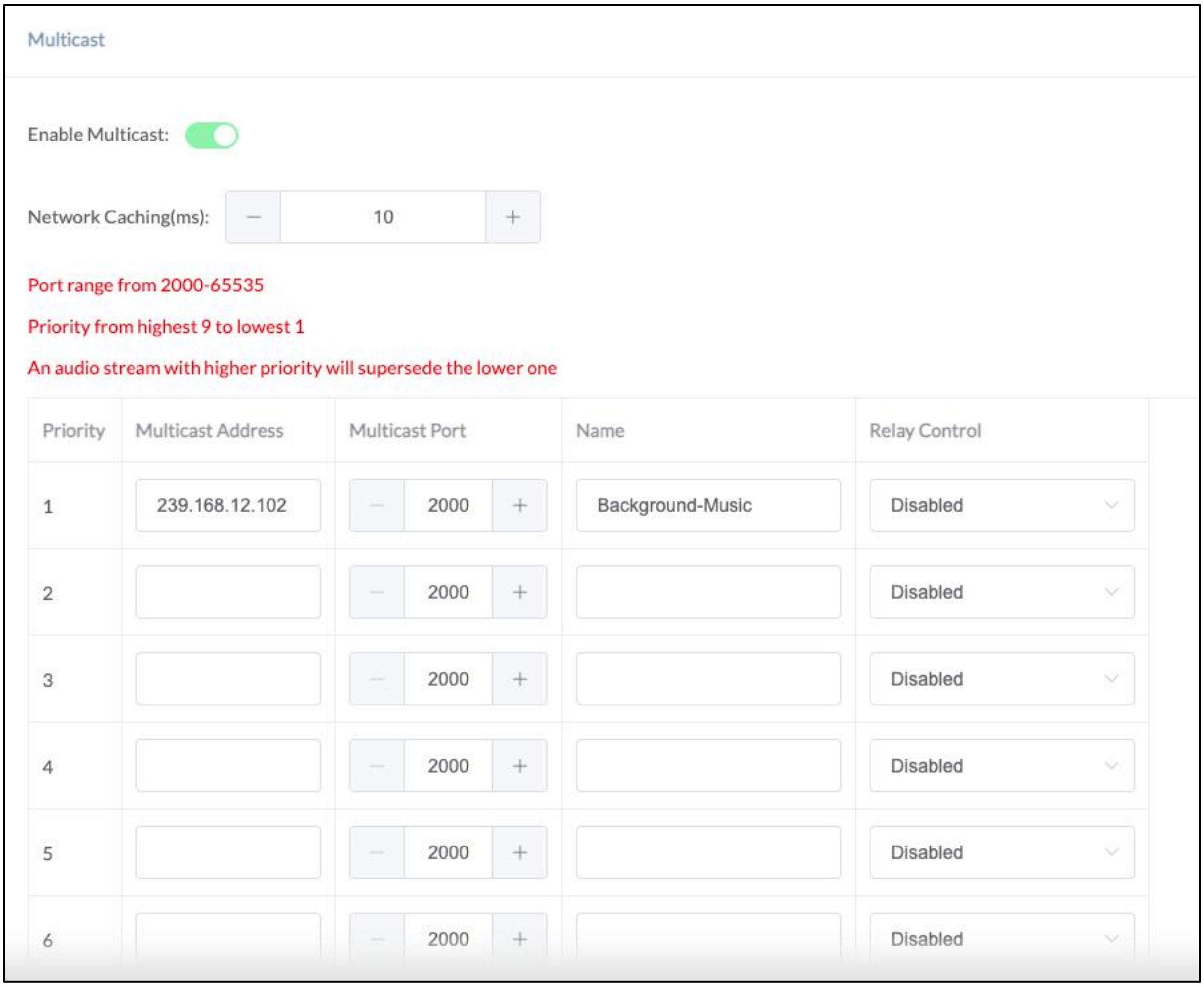
- **Enable:** Enable/Disable ONVIF integration for compatibility with ONVIF-supported VMS platforms.
- **Username:** Enter an account username with matching credentials for adding devices to the VMS platform.
- **Password:** Enter a matching password for the account to add devices to the VMS platform.
- **Enable Microphone:** Enable/Disable the microphone function.

- **Relay Mode:** Set relay control to monostable or bistable. In monostable mode, you can specify the activation duration.
- **Duration(Sec):** Set the activation duration in monostable mode.
- **Relay Type:** Choose a relay response to triggers: ‘Door Strike Mode: Unlock/Lock’, ‘Strobe Light Mode: Fast Flashing’, or ‘Strobe Light Mode: Slow Flashing’.

5.2 Multicast

The multicast settings are used to configure the parameter settings of the multicast function on the intercom. It can be configured to monitor up to 9 different levels of multicast addresses, the audio streams with a higher priority will interrupt the playback of the lower priority audio streams.

Please go to **Functions ---> Multicast** page to enable the multicast feature.



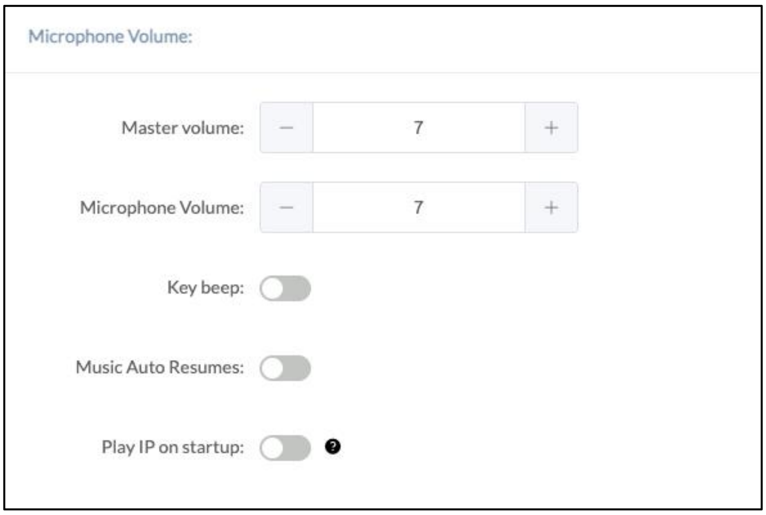
Multicast

- **Priority:** Priority from highest 9 to lowest 1.
- **Multicast Address:** The multicast address range is 224.0.0.0 – 239.255.255.255.
- **Multicast Port:** The multicast port range is 2000 – 65535.
- **Name:** Customize the name of the multicast address.
- **Relay Control:** Options to choose from are ‘Disabled’, ‘On’, ‘Fast Flashing’, ‘Slow Flashing’.

6. Advanced Settings

6.1 Volume Settings

To set the volume of the VI intercom, please go to **Advanced --> Volume** page to configure.



Volume Settings

- **Master Volume:** The default speaker volume is 7, adjustable range is 0 ~ 9.
- **Microphone Volume:** The default microphone volume is 7, adjustable range is 0 ~ 9.
- **Key Beep:** Enable/Disable the beep sound from the key button.
- **Music Auto Resumes:** When the device restarts or reconnects to the network, the previous music tasks will be automatically restored.
- **Play IP on Startup:** When the device starts, it automatically broadcasts its IP address once.

6.2 Audio Priority Settings

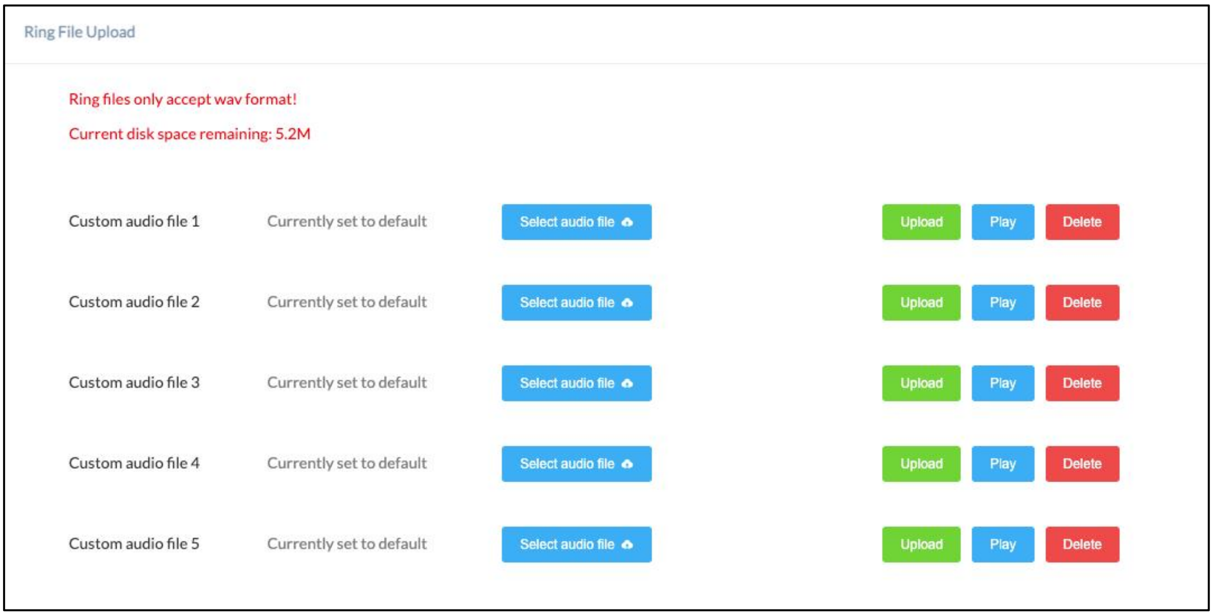
The audio priority can be set according to different applications(such as SIP, ONVIF, MULTICAST, BROADCAST...). Please go to **Advanced ---> Audio Priority** to set the priority.

Priority 1 is the highest. You can drag the arrow on the right side to adjust the priority. The execution of a high-priority audio application will interrupt the current low-priority audio application.

Audio Priority Settings

6.3 Ring File

The Ring files section allows users to self-upload up to 5M of audio files to the endpoint and use it as a ringtone or Play API audio file. Please click on the 'Select audio file' button to select and upload the local audio file, then click on the 'upload' button to upload it. Click on the 'play' to test and play the audio file and the 'delete' button for deleting the audio file. Please go to **Advanced ---> Audio Files** to manage the audio files.

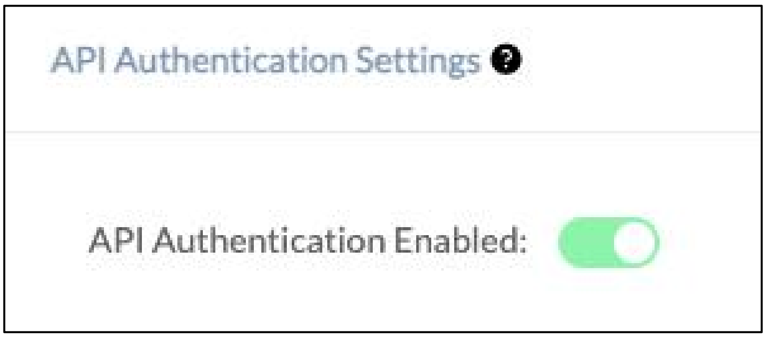


Audio Files

6.4 API Settings

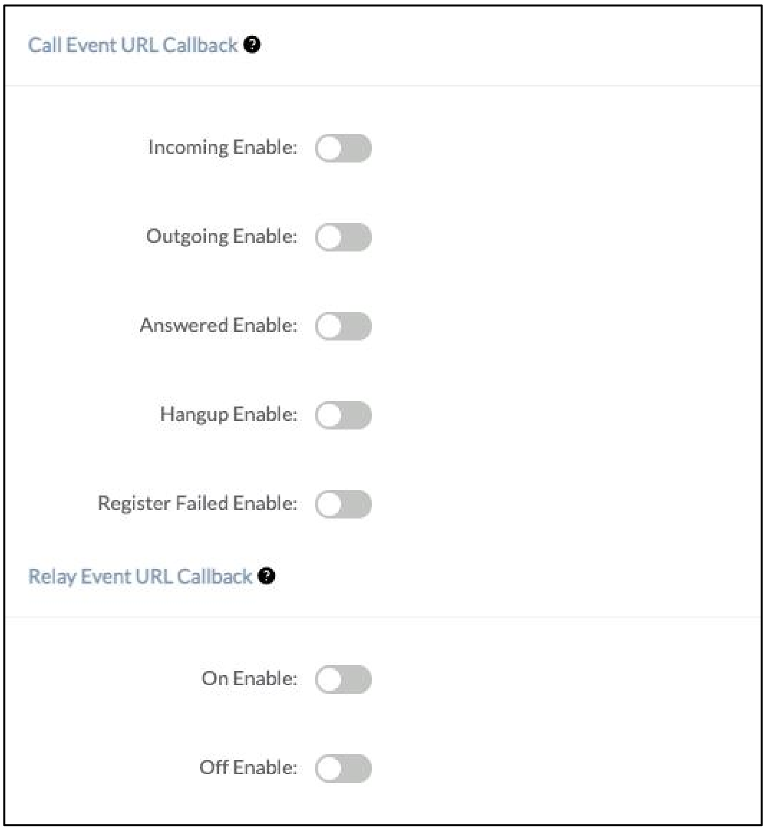
This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the changing status of the call and/or relay.

Please go to **Advanced --> API Settings** page to enable API settings.



API Authentication Settings

- **API Authentication Enabled:** Once enabled, all API requests to this device will require authentication.



Call Event URL Callback & Relay Event URL Callback

When the call status changes, it will trigger an HTTP GET request to call a URL address. Within the URL address, you may use variables to identify some current information.

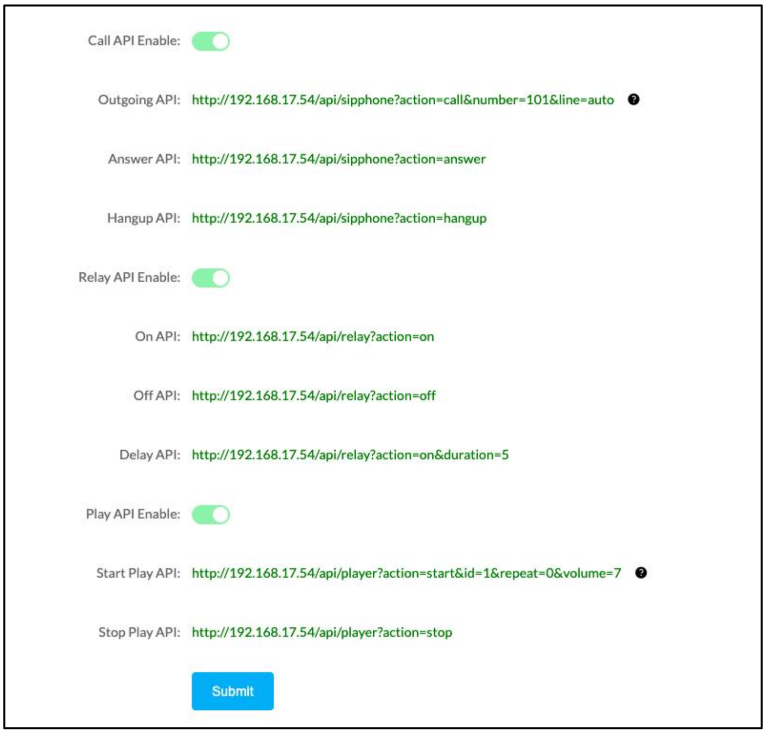
For example:

${ip}:	The current IP address of the device
${mac}:	The current MAC address of the device
${ua}:	The account of the current call
${number}:	The number of the current call

When the relay status changes, it will trigger an HTTP GET request to call a URL address. Within the URL address, you may use variables to identify some current information.

For example:

${ip}:	The current IP address of the device
${mac}:	The current MAC address of the device



API Settings

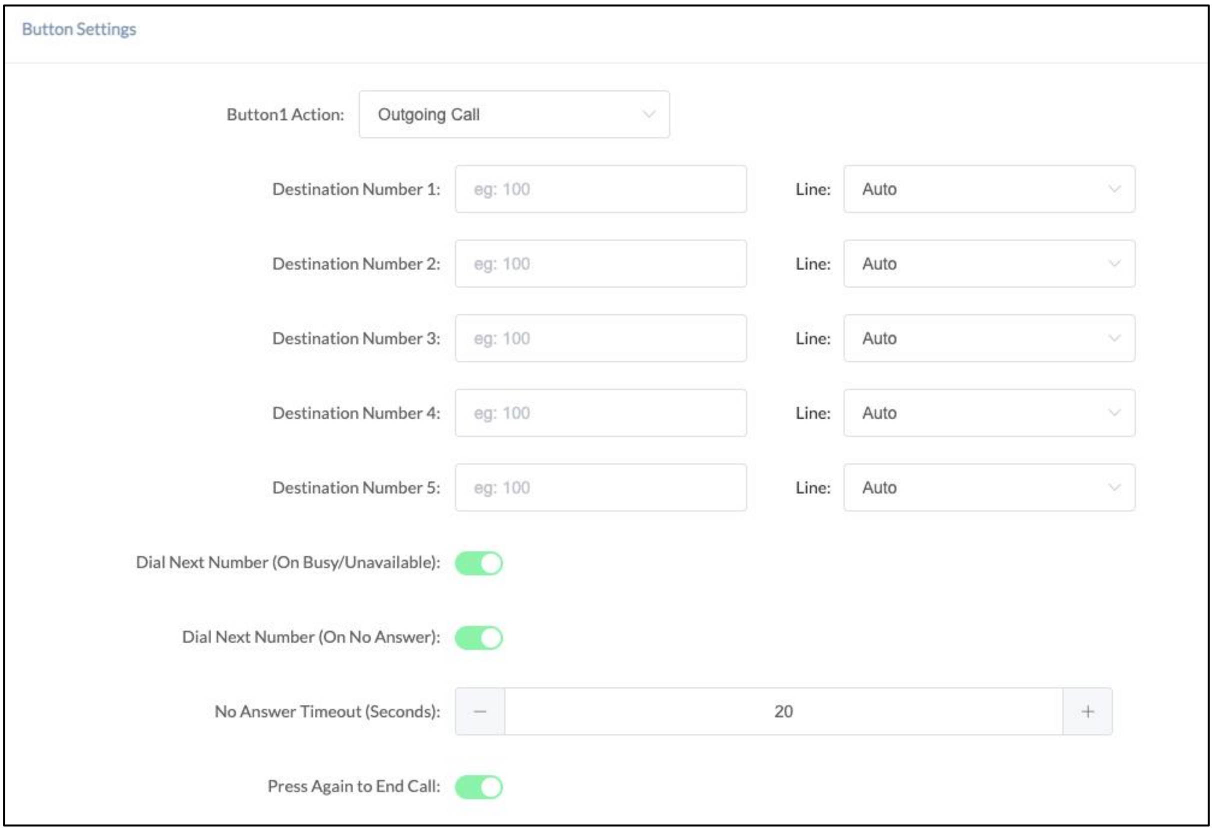
Using the API interface to realize features such as device linkage, call control, relay control, and play sound by the systems.

Note: Authentication and encryption are not used in the API interface, so please pay attention to the security of the network environment when opening and using these API interfaces.

6.5 I/O Settings

This page is used to configure configuration parameters related to security linkage, such as: button, relay settings and other related configurations.

Please go to **Advanced --> I/O Settings** page to set the specific settings.



Outgoing Call Settings

- **Button Action:** Defines the action to be performed when the button is pressed. You can choose between Outgoing Call (SIP call) or HTTP Request.
- **Destination Number:** Configures the destination numbers to be dialed when the button is triggered. 1~5 number(s) can be defined. The system will call them in order. When multiple numbers are configured, enabling the Dial Next Number option allows the system to automatically try the next number in case of busy, no answer, or unreachable conditions, ensuring a failover mechanism.

Note: Destination 1 is the default call destination and is required. Destinations 2–5 are optional fallback numbers

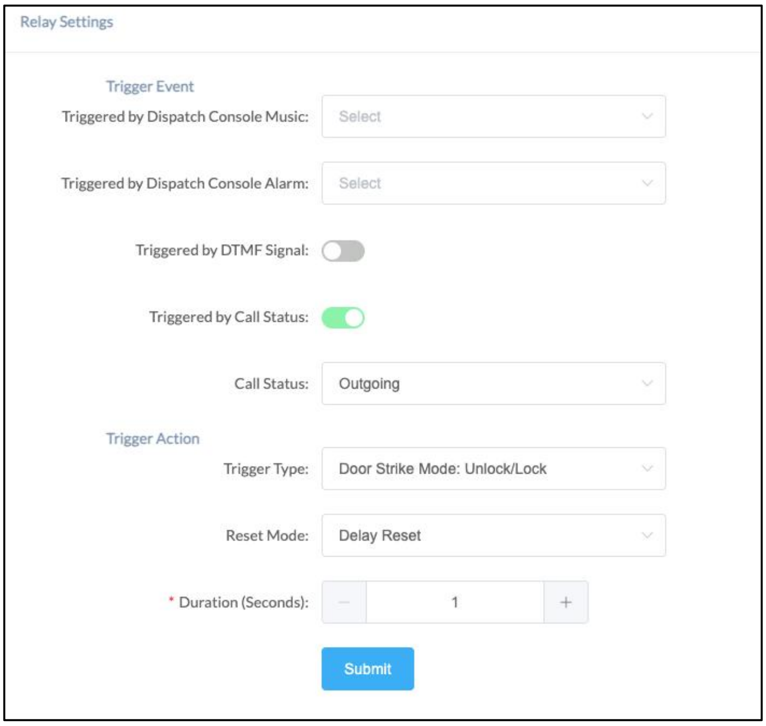
- **Line:** This setting represents the corresponding line for making outgoing calls.

Note: when using the P2P line to call, please specify the device's number + IP address, such as 101@192.168.11.123.

- **Dial Next Number (On Busy/Unavailable):** When the current number is busy or unreachable, proceed to the next number if enabled.
- **Dial Next Number (On No Answer):** If the current number doesn't answer within the timeout, proceed to the next number if enabled.
- **No Answer Timeout (Seconds):** Defines how long (in seconds) to wait for an answer before moving to the next number when the 'Dial Next Number (On No Answer)' option is enabled.
- **Press Again to End Call:** After the call is connected, users can end the call or conversation by pressing the button again.

HTTP Request Settings

- **HTTP URL:** Configure the API URL address triggered by linkage.



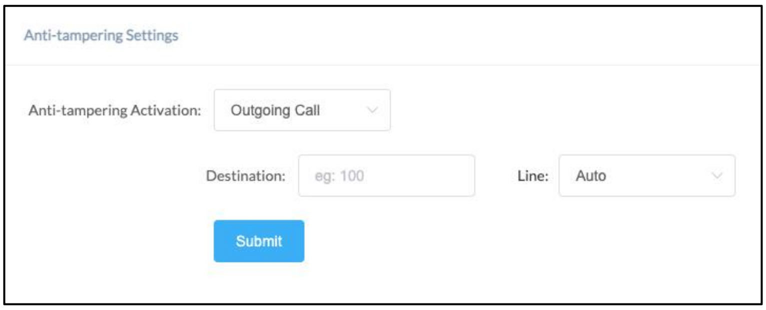
Relay Control

- **Triggered by Dispatch Console Music:** Enable this option to trigger the relay when a user activates the Music function via the IP Audio Dispatch Console.
- **Triggered by Dispatch Console Alarm:** Enable this option to trigger the relay when a user activates the Alarm function via the IP Audio Dispatch Console.
- **Triggered by DTMF Signal:** Enable or disable relay triggering via DTMF signal (RFC2833 only).
- **DTMF Code:** Specify the DTMF number to be dialed to trigger the relay.
- **Triggered by Call Status:** Enable or disable relay triggering based on call status changes.
- **Call Status:** Select the call condition that will trigger the relay: Outgoing, Incoming, Incoming/Outgoing, Answered, Hangup.
- **Trigger Type:** Defines the relay response mode when triggered by DTMF signal or call status. Available options: Door Strike Mode: Unlock / Lock (single trigger), Strobe Light Mode: Fast Flashing (continuous fast trigger), Strobe Light Mode: Slow Flashing (continuous slow trigger).
- **Reset Mode:** Configure how the relay resets after activation: Delay Reset: The relay resets after a specified duration, Hang-up Reset: The relay resets when the call ends, Answered Hang-up Reset: The relay resets after the call is answered and then hung up.
- **Duration (Sec):** This setting is only available when Delay Reset is selected. Defines how long (in seconds) the relay remains active after being triggered.

6.6 Anti-Tamper Settings

All VI Series devices support tamper detection. Once the device is properly installed, any unauthorized removal or tampering will trigger the internal tamper alarm system.

Please go to **Advanced ---> Anti-Tamper Settings** page to set.



Anti-Tamper Setting

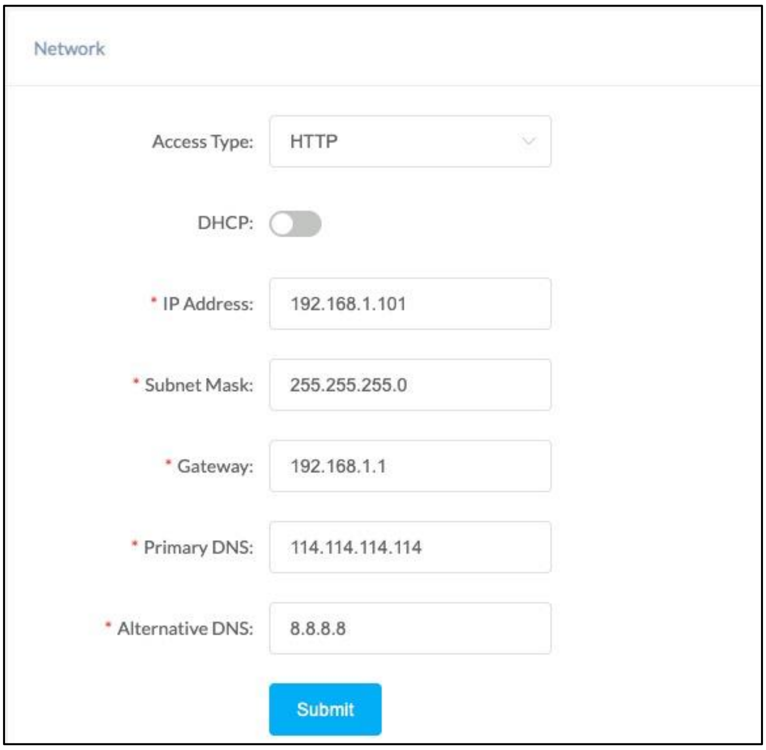
- **Tamper Action:** When a tamper event is detected, the following actions can be executed: Outgoing Call, HTTP Request, Play Audio File.
- **Destination:** Specify the call destination to be dialed when the tamper alarm is triggered.
- **Line:** Select the line to be used for placing the call.
- **Audio File:** Select the audio file to be played when a tamper event is triggered.
- **Repeat:** Set the number of times the selected audio file will be played.

7. System Settings

7.1 Network

VI intercom uses DHCP to dynamically obtain IP addresses by default.

To change the IP assignment from DHCP to Static IP, please go to **System--> Network** page. Turn the DHCP switch button off to show the network parameter settings.



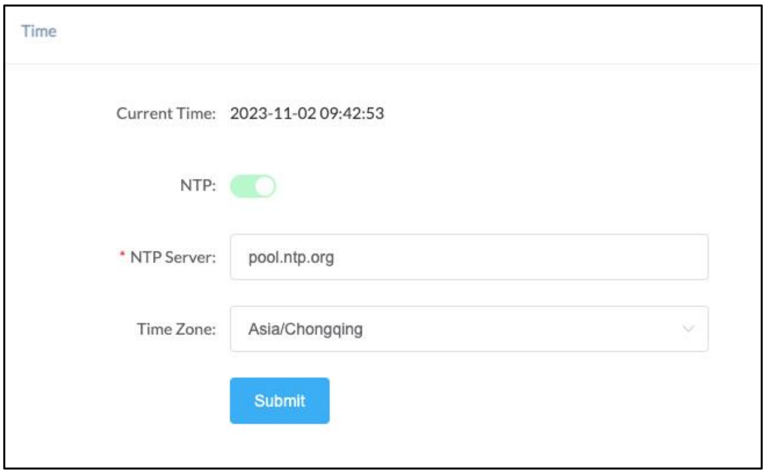
Network Configuration

- **Access Type:** Specify the access method of the website, which currently supports HTTP and HTTPS.
- **IP Address:** Enter a vacant IP address within your LAN.
- **Subnet Mask:** Enter the subnet mask of your LAN.
- **Gateway:** Enter the default gateway of your LAN, this is essential for the device when the IP Audio Center or other SIP server is installed outside the LAN.
- **Primary DNS:** Enter an effective primary DNS server address.

- **Alternative DNS:** Enter an alternative DNS server address, when the primary DNS fails, alternative DNS will be used.

7.2 Time

VI intercom obtains the time from the network time servers using NTP.
To change the NTP settings, please go to **System --> Time** page.



Time Settings

- **Current Time:** Display the current system time of the device.
- **NTP:** Enable/Disable using NTP to obtain the time.
- **NTP Server:** The network time server used to obtain the time.
- **Time Zone:** Set the time zone used by the device.

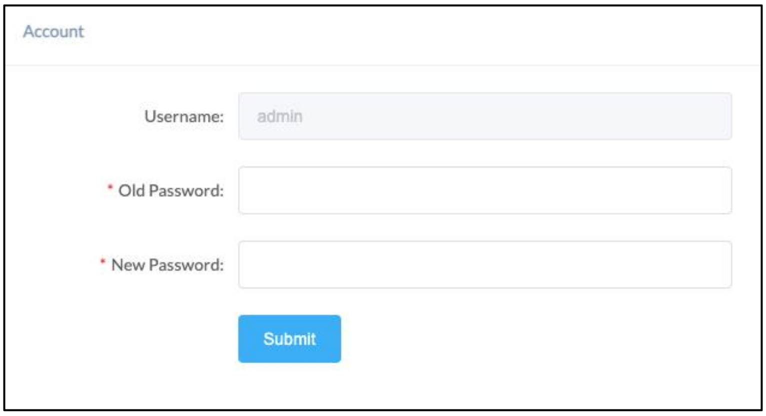
7.3 Prompt Language

The language of local voice prompts, like IP address announcements. Currently, only Chinese and English are provided.
Please go to **System --> Prompt Language** page to set a voice prompt language.

Prompt Language

7.4 Account

For resetting the current device's password, please go to **System --> Account** page.



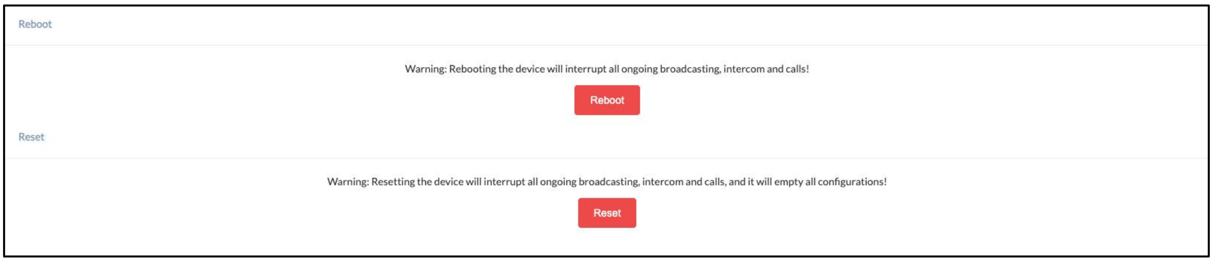
Web Password Settings

- **Old Password:** This setting represents the current user password.
- **New Password:** This setting represents the new password user would like to set up.

7.5 Reboot & Reset

VI intercom can be rebooted and reset from the web management interface.

If you need to reboot or reset the device, please go to **System --> Reboot & Reset** page.



Reboot & Reset Settings

Users can restart the device without power failure on this page. The restart process takes about 10 seconds. After the restart is complete, refresh the page to log in again.

If you need to restore the factory settings of the intercom, you can reset it through this page After restarting, the pop-up window disappears, and the device is restored successfully.

Note: Restoring factory settings will erase all user settings, please operate with caution!

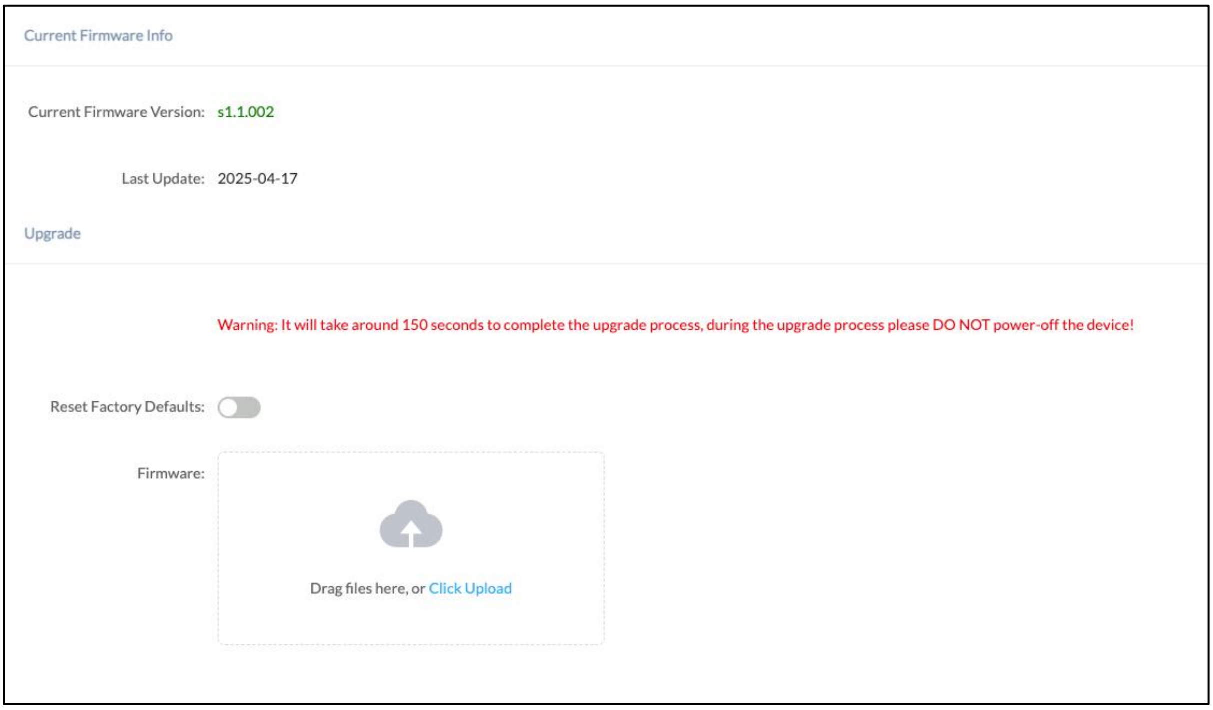
Reboot Schedule

When the Reboot Schedule feature is Enabled, you can set up the automatic reboot daily, weekly, or monthly at a specified time.

8. Maintenance

8.1 Upgrade

To upgrade the device's firmware, please go to **Maintenance --> Upgrade** page.



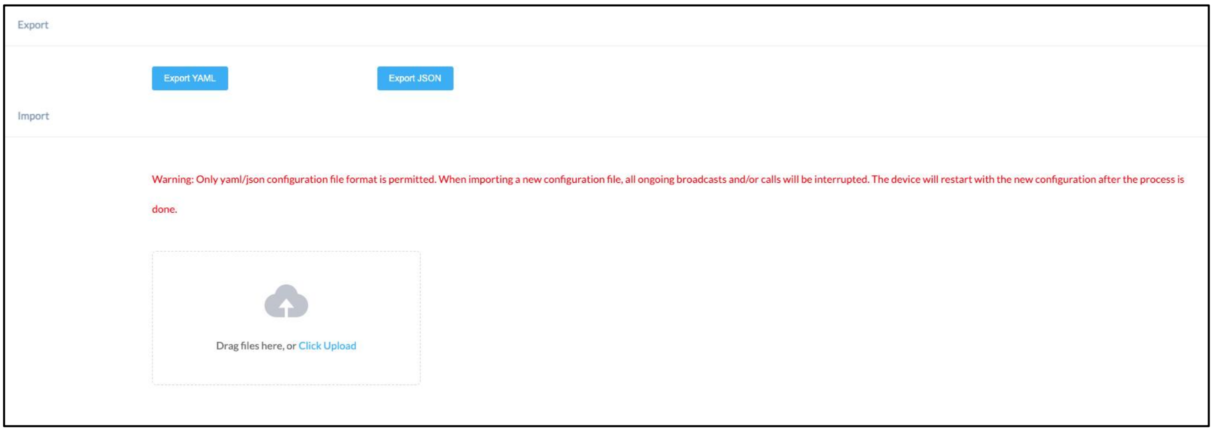
Upgrade Settings

- **Current Firmware Version:** Displays the version currently used by the system.
- **Last Update:** Displays the last system updating time.
- **Reset Factory Defaults:** Specify whether to restore factory settings when upgrading.
- **Firmware:** Click to select the firmware that needs to be used to upgrade the current device.

8.2 Import/Export

This page is used to import and export the current configuration of the device, and you may use this configuration file to backup and/or recover. Both YAML and JSON formats are supported.

Please go to **Maintenance --> Import/Export** page to backup or recover.



Import/Export

8.3 Auto Provisioning

The system is supporting DHCP Option 066 and static TFTP/HTTP two auto provisioning methods.

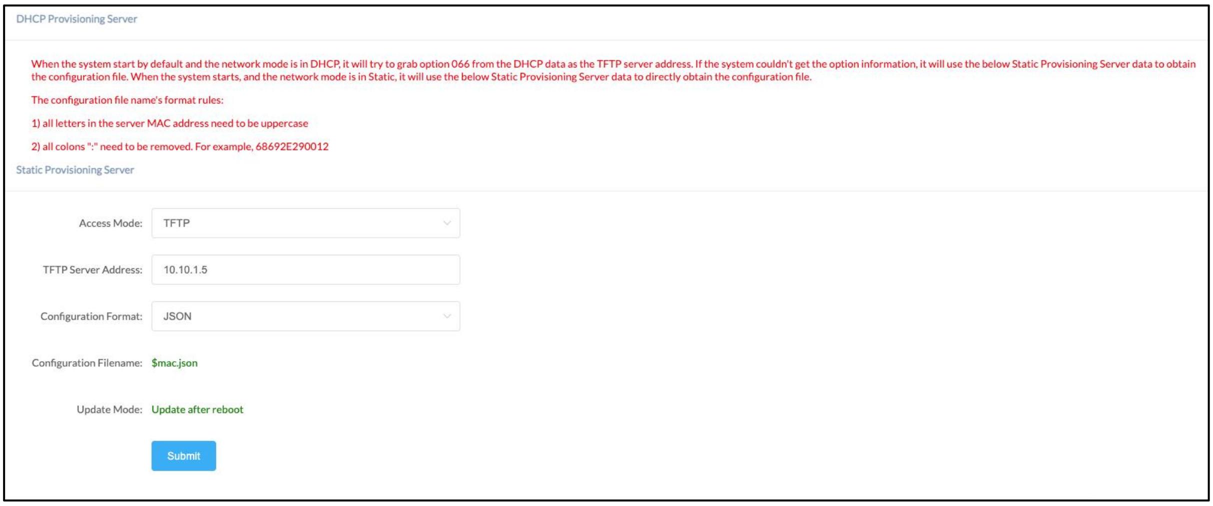
When the system starts by default and the network mode is in DHCP, it will try to grab option 066 from the DHCP data as the TFTP server address. If the system couldn't get the option information, it will use the below Static Provisioning Server data to obtain the configuration file. When the system starts, and the network mode is in Static, it will use the below Static Provisioning Server data to directly obtain the configuration file.

The configuration file name's format rules:

1) all letters in the server MAC address need to be uppercase.

2) all colons ":" need to be removed. For example, 68692E290012.

Please go to **Maintenance --> Auto Provisioning** page to configure static server.



Auto Provisioning

8.4 Diagnostic

Ping is a network administration utility or tool used to test connectivity on an IP network. Input other devices' IP addresses and click on the submit button to trace the network route. Please go to **Maintenance --> Diagnostic** page to execute ping command.

Ping

8.5 Ethernet Capture

The purpose of the Ethernet capture tool is to capture Ethernet network packets and store them in a standard Wireshark-compatible packet capture '.pacp' file for immediate viewing and data analysis.

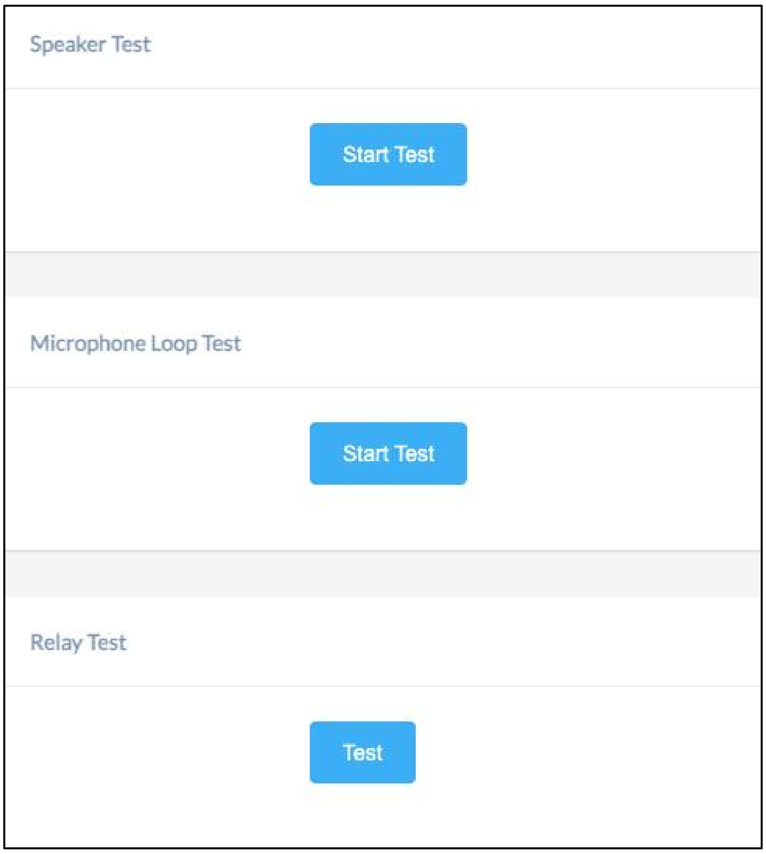
Please go to **Maintenance --> Ethernet Capture** page to operate.

Ethernet Capture

8.6 Test

The detection feature provides an option for the user to check whether the speaker, microphone and relay will work functionally before registering it to the server.

Please go to **Maintenance --> Test** page to test whether the component is working properly.



Test Settings

- **Speaker Test:** Click on the Start Test button, and the speaker will play a ringtone to test whether the speaker is working. If the speaker is working functionally, you should hear the voice back.
- **Microphone Loop Test:** Click on the Start Test button, then start speaking to the device.
- **Relay Test:** Click on the Test button and the device will output signals to the relay for testing.

9. Reports

9.1 Call Logs

Call Logs allows you to check the call related information such as Call Date, Time, Account, Telephone Number, Call Duration, Call Type and Status. Please go to **Reports --> Call Logs** page to view the logs.

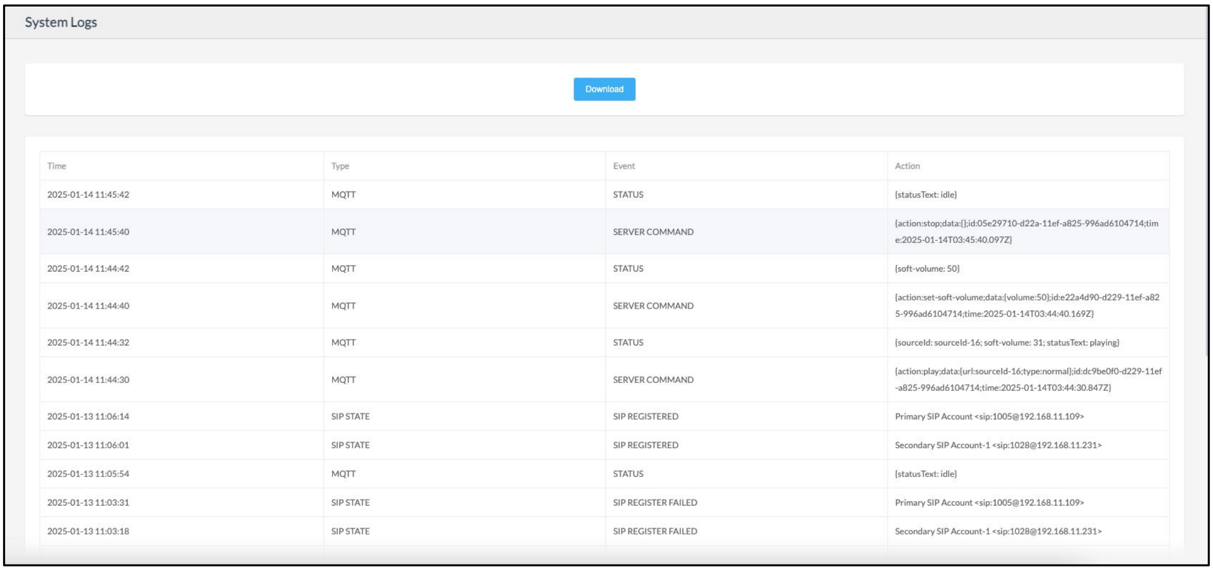
Date	Time	Account	Telephone Number	Duration	Type	Status
2024-03-15	17:18:00	sip:5011@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:47:58	sip:5015@192.168.11.62	5000	00:00:02	Inbound	Answered
2024-03-15	16:38:30	sip:5005@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:38:16	sip:5005@192.168.11.62	5000	00:00:01	Inbound	Answered

Total 4 < 1 >

Call Logs

9.2 System Logs

System Logs allows you to check the event related information such as Operating Time, Operating Type (MQTT, Function, SIP, Multicast...), Event and Action details. Please go to **Reports --> System Logs** page to view the logs. Click the Download button and the .csv log file will be saved on your computer.



System Logs

Download

Time	Type	Event	Action
2025-01-14 11:45:42	MQTT	STATUS	{statusText: idle}
2025-01-14 11:45:40	MQTT	SERVER COMMAND	{action:stop;data:{};id:05e29710-d22a-11ef-a825-996ad6104714;time:2025-01-14T03:45:40.097Z}
2025-01-14 11:44:42	MQTT	STATUS	{soft-volume: 50}
2025-01-14 11:44:40	MQTT	SERVER COMMAND	{action:set-soft-volume;data:{volume:50};id:e22a4d90-d229-11ef-a825-996ad6104714;time:2025-01-14T03:44:40.169Z}
2025-01-14 11:44:32	MQTT	STATUS	{sourceId: sourceId-16; soft-volume: 31; statusText: playing}
2025-01-14 11:44:30	MQTT	SERVER COMMAND	{action:play;data:{url:sourceId-16;type:normal};id:dc9be0f0-d229-11ef-a825-996ad6104714;time:2025-01-14T03:44:30.847Z}
2025-01-13 11:06:14	SIP STATE	SIP REGISTERED	Primary SIP Account <sip:1005@192.168.11.109>
2025-01-13 11:06:01	SIP STATE	SIP REGISTERED	Secondary SIP Account-1 <sip:1028@192.168.11.231>
2025-01-13 11:05:54	MQTT	STATUS	{statusText: idle}
2025-01-13 11:03:31	SIP STATE	SIP REGISTER FAILED	Primary SIP Account <sip:1005@192.168.11.109>
2025-01-13 11:03:18	SIP STATE	SIP REGISTER FAILED	Secondary SIP Account-1 <sip:1028@192.168.11.231>

System Logs

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