OPA Network Speaker User Guide

www.odmpa.com

Contents

1. Preface

1.1 Audience

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

1.2 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
1.0.0	1.0.9	Initial Issue	Jun, 2026

2. Overview

2.1 Product Overview

OPA network speaker include two types of network ceiling speakers and cabinet speaker, three types of network horn speaker and two types of network column speaker, namely: the network ceiling speaker C8215 and C8130 , the network cabinet speaker W8215 and W8130, the network horn speaker H8115, H8130 and H8160, as well as the network column speaker L8130 and L8160. The network ceiling speakers and cabinet speaker are mainly applied to indoor environments, while the H8115, H8130, H8160, L8130 and L8160 can be applied to open outdoor environments.

2.2 Product Specifications

C8215 Network Ceiling Speaker Specifications		
Speaker Components	1 x 6" Woofer+1 x 1" Coaxial driver unit	
Sensitivity	92dB / 1m / 1W	
Max Sound Pressure Level	103dB	
Rated Power	DC: 8Ω 15W or PoE: af-8Ω 8W / at-8Ω 15W	
Frequency Range	150Hz – 20KHz	
Coverage Pattern	135° Conical ，30m²	
Amplifier	Built-in Class D Amplifier	

C8130 Network Ceiling Speaker Specifications

Speaker Components	1 x 8" Woofer+1 x 1" Coaxial driver unit
Sensitivity	95dB / 1m / 1W
Max Sound Pressure Level	115dB
Rated Power	DC: 8Ω 30W or PoE: af-8Ω 8W / at-8Ω 20W
Frequency Range	100Hz – 20KHz
Coverage Pattern	135° Conical ，50m²
Amplifier	Built-in Class D Amplifier

W8215 Network Cabinet Speaker Specifications		
Speaker Components	1x5"Woofer +1x 1" high frequency driver unit	
Sensitivity	93dB / 1 m/ 1W	
Max Sound Pressure Level	104dB	
Rated Power	DC: 8Ω 15W or PoE: af-8Ω 8W / at-8Ω 15W	
Frequency Range	100Hz – 20KHz	
Coverage Pattern	90°H 90°V, 30m²	
Amplifier	Built-in Class D Amplifier	

W8130 Network Cabinet Speaker Specifications		
Speaker Components	1x6.5"Woofer +1x 1" high frequency driver unit	
Sensitivity	95dB / 1 m/ 1W	
Max Sound Pressure Level	109dB	
Rated Power	DC: 8Ω 30W or PoE: af-8Ω 8W / at-8Ω 20W	
Frequency Range	70Hz – 20KHz	
Coverage Pattern	90°H 90°V, 60m²	
Amplifier	Built-in Class D Amplifier	

H8115 Network Horn Speaker Specifications	
Speaker Components	1.3"mid-high frequency driver unit
Sensitivity	113dB / 1m / 1W
Max Sound Pressure Level	124dB
Rated Power	DC: 8Ω 15W or PoE: af-8Ω 8W / at-8Ω 15W
Frequency Range	300Hz – 10KHz
Coverage Pattern	119°H 119°V， effective distance 150m
Amplifier	Built-in Class D Amplifier

H8130 Network Horn Speaker Specifications	
Speaker Components	1.5"mid-frequency driver unit
Sensitivity	115dB / 1m / 1W
Max Sound Pressure Level	129dB
Rated Power	DC: 8Ω 30W or PoE: af-8Ω 8W / at-8Ω 20W
Frequency Range	380Hz – 10KHz
Coverage Pattern	78°H 78°V， effective distance 200m
Amplifier	Built-in Class D Amplifier

H8160 Network Horn Speaker Specifications	
Speaker Components	1.5"mid-frequency driver unit
Sensitivity	116dB / 1m / 1W
Max Sound Pressure Level	133dB
Rated Power	4Ω 60W
Frequency Range	380Hz – 6.5KHz
Coverage Pattern	78°H 78°V， effective distance 300m
Amplifier	Built-in Class D Amplifier

L8130 Network Column Speaker Specifications	
Speaker Components	1x0.75" high frequency +1x 4" full-range driver unit
Sensitivity	90dB / 1W / 1m
Max Sound Pressure Level	104dB
Rated Power	DC: 4Ω 30W or PoE: af-4Ω 8W / at-4Ω 20W
Frequency Range	100Hz – 18KHz
Coverage Pattern	135°， Optimal distance 50m
Amplifier	Built-in Class D Amplifier

L8160 Network Column Speaker Specifications		
Speaker Components	1x0.75" high frequency +2x 4" full-range driver unit	
Sensitivity	90dB / 1W / 1m	
Max Sound Pressure Level	106dB	
Rated Power	2Ω 60W	
Frequency Range	100Hz – 18KHz	
Coverage Pattern	135°, Optimal distance 50m	
Amplifier	Built-in Class D Amplifier	

3. Login the Device

3.1 Accessing the Web GUI

The device 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 default static IP address: 192.168.1.101), you will listen to the device's IP address broadcast at first start up, and enter the IP address in the browser to access the device's Web management interface.

Default username: admin

Default password: admin

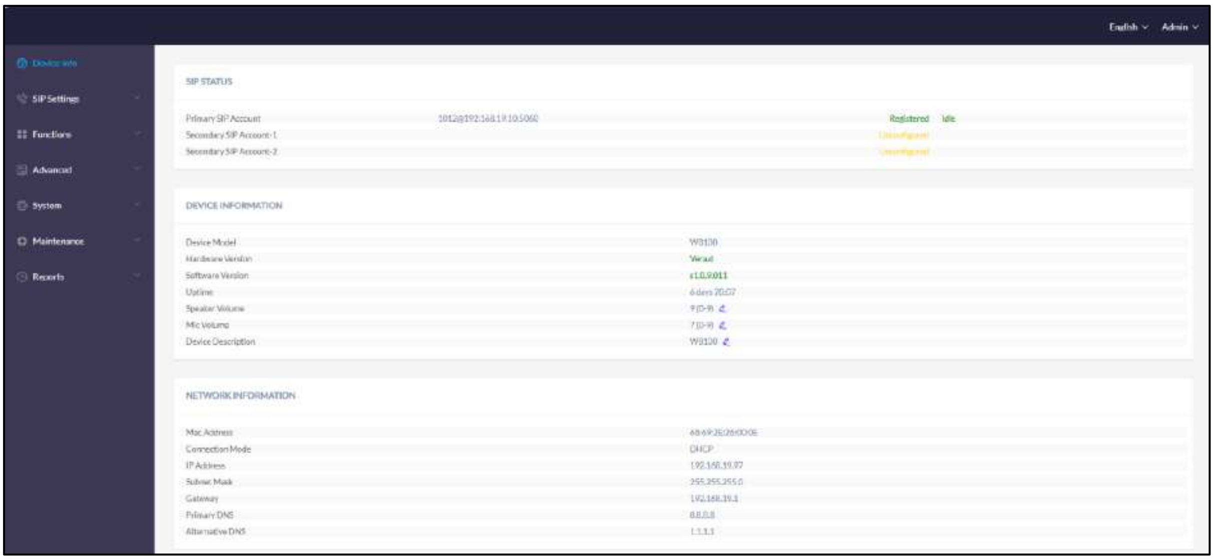
For the safety purpose, it is recommended to change the default password on the first login, please go to **System --> Password Settings** page to change the password.

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.



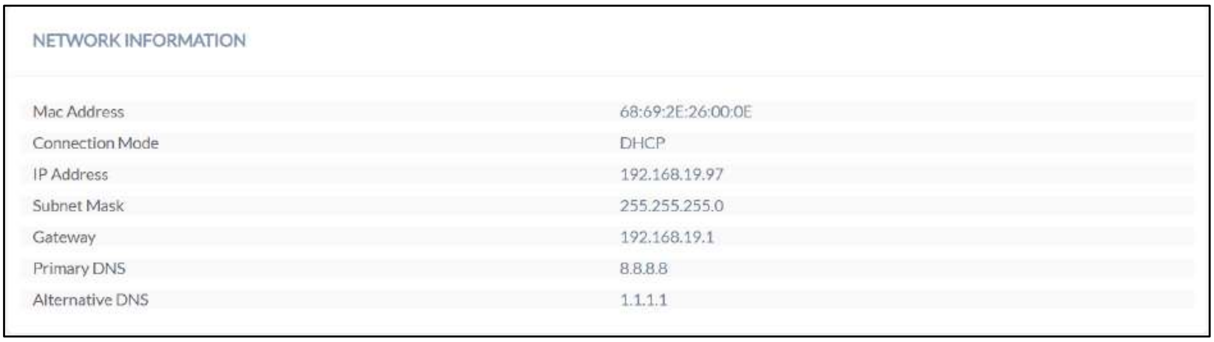
SIP Status

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

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 device's operating time.
- **Speaker Volume:** Displays the current volume of the device.
- **Microphone Volume:** Displays the current input volume of the device's microphone.
- **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:26:00:0E
Connection Mode	DHCP
IP Address	192.168.19.97
Subnet Mask	255.255.255.0
Gateway	192.168.19.1
Primary DNS	8.8.8.8
Alternative DNS	1.1.1.1

Network Information

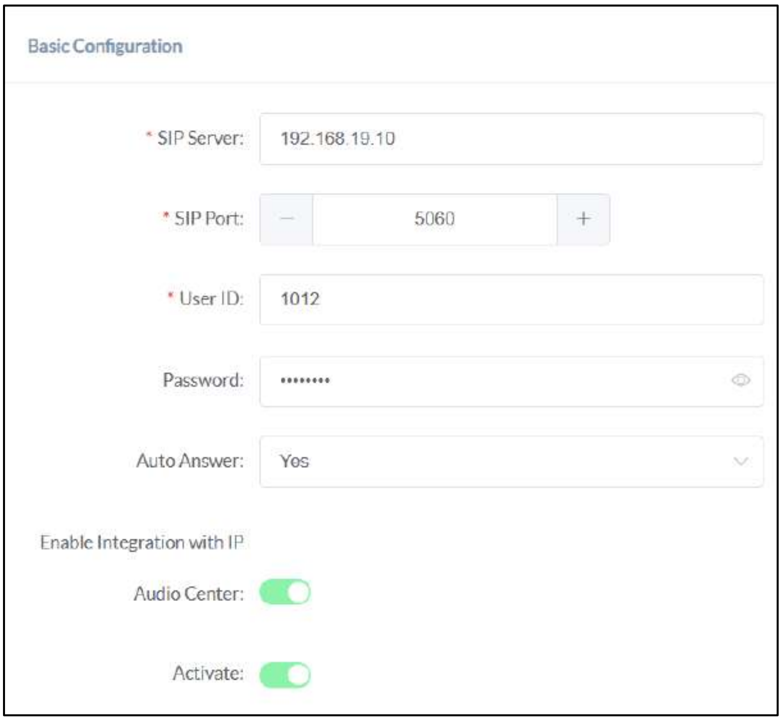
- **Mac Address:** Display the MAC address of the current device.
- **Connection Mode:** Display the network acquisition method of the device, DHCP (dynamic acquisition) or STATIC (static configuration).
- **IP Address:** The current IP address of the device.
- **Subnet Mask:** The current subnet mask 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 Audio Center, please turn on the 'Enable Integration with IP Audio Center' option.

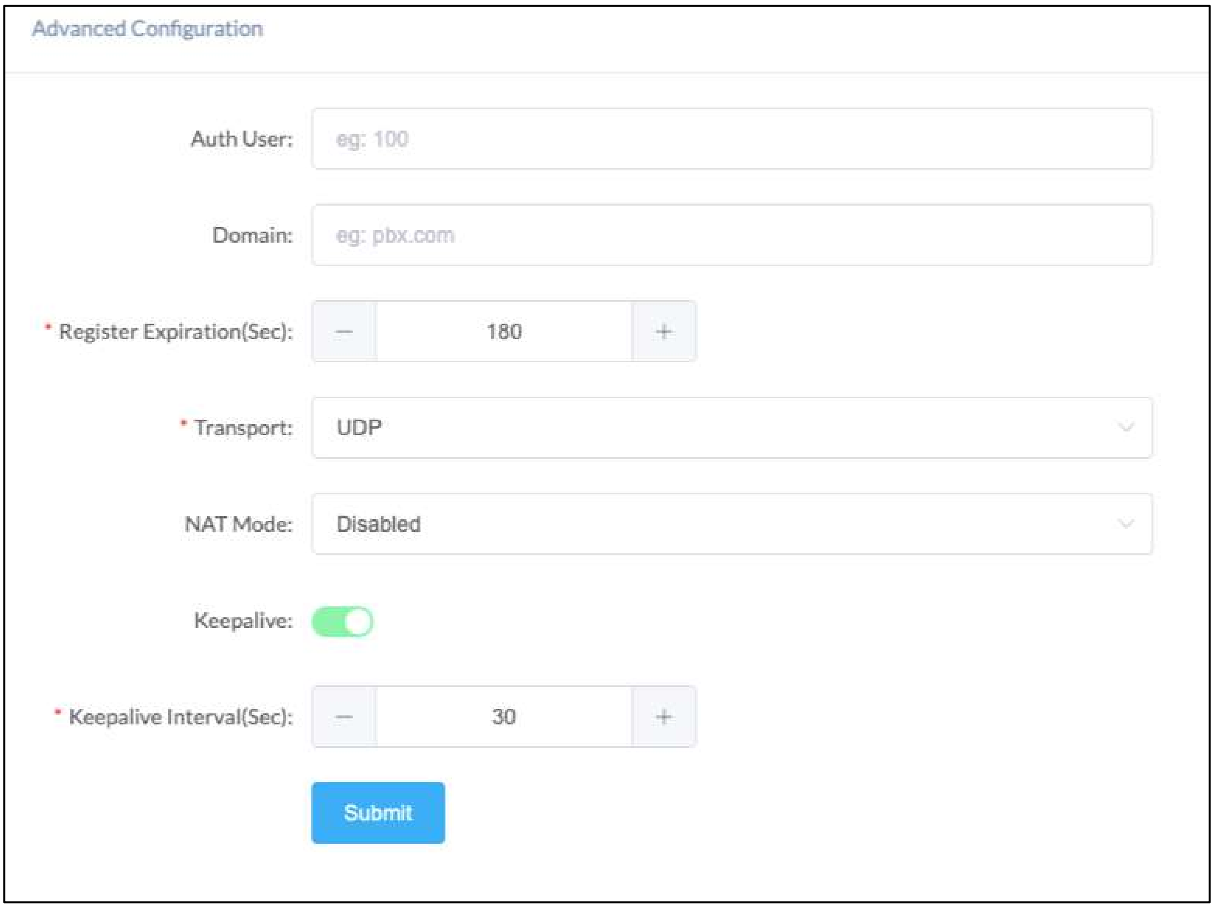
Please go to **SIP Settings --> Primary SIP Account / Secondary SIP Account-1 / Secondary SIP Account-2** page.



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 IP Audio Center:** Disabled by default. Enable this option when connecting to the 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.



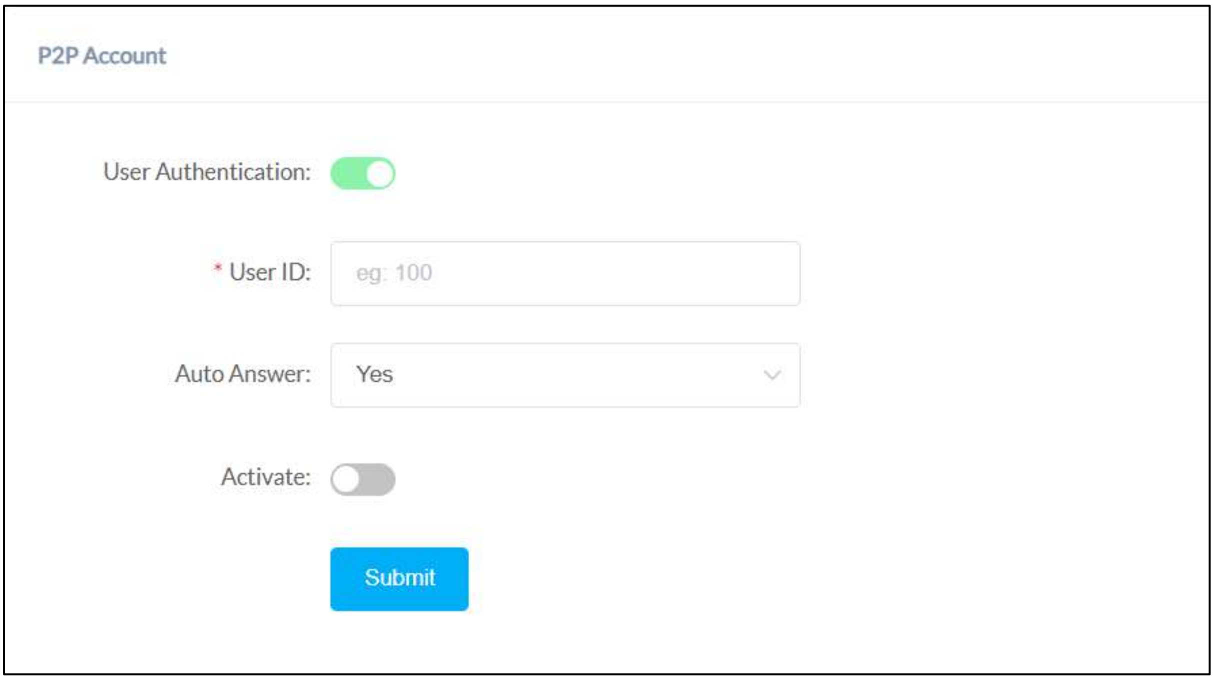
SIP Account - Advanced Configuration

- **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

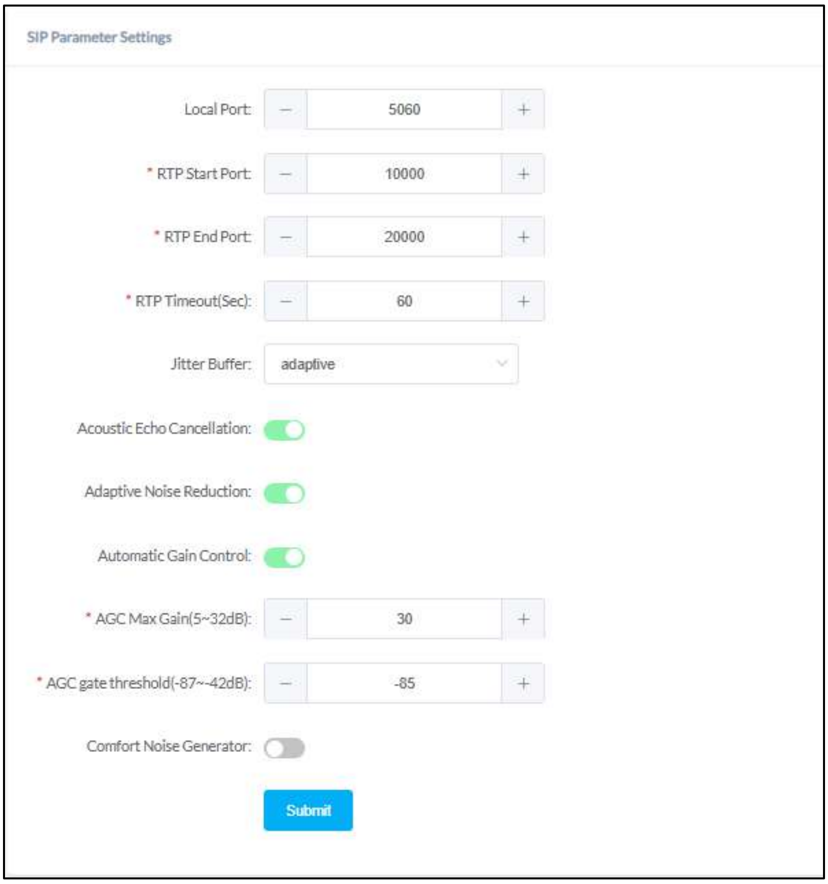
- **User Authentication:** Enable/Disable P2P authentication. If disabled, you can directly enter this device's IP address in the target field of the peer device. If enabled, you must use the following format in the target field of the peer device: This device's P2P User ID + IP address (e.g., 101@192.168.1.101).

- **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

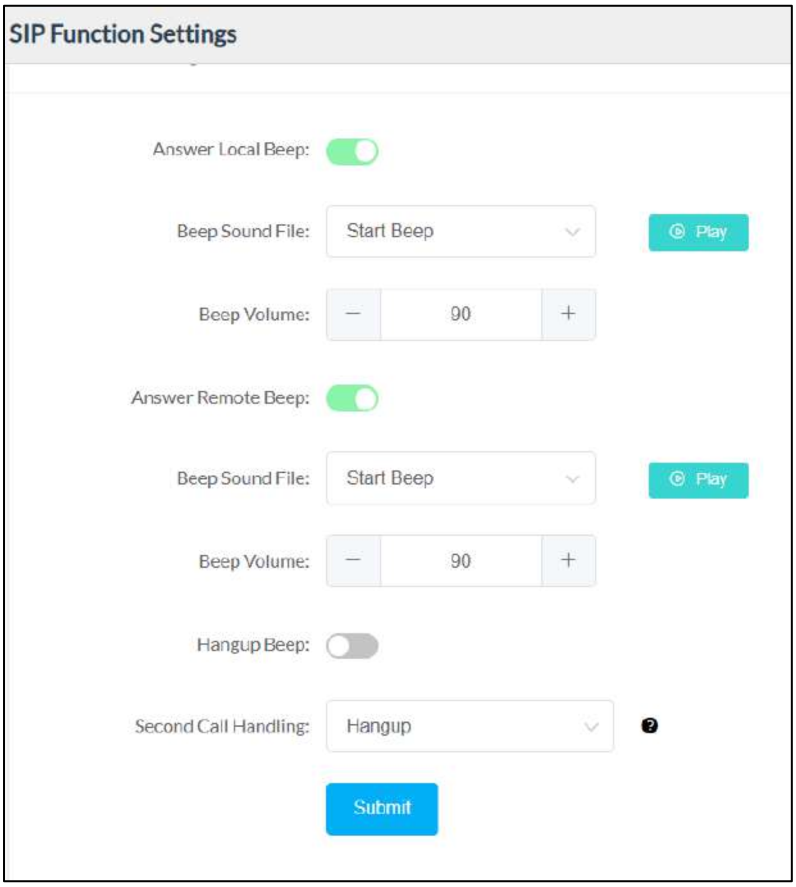
- **Local Port:** This setting represents the port used to receive SIP packets.

- **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:** suppressed through algorithms.
- **Adaptive Noise Reduction:** After enabling this feature, echo noise can be 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. After optimization through the AGC, the effective pickup distance of our equipment can reach a maximum of more than 10 meters.

  AGC Max Gain： can adjust from 5 to 32dB.

  AGC gate threshold: can adjust from -87 to -42dB.
- **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 Handling:** Set the Option 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

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

Audio Codecs

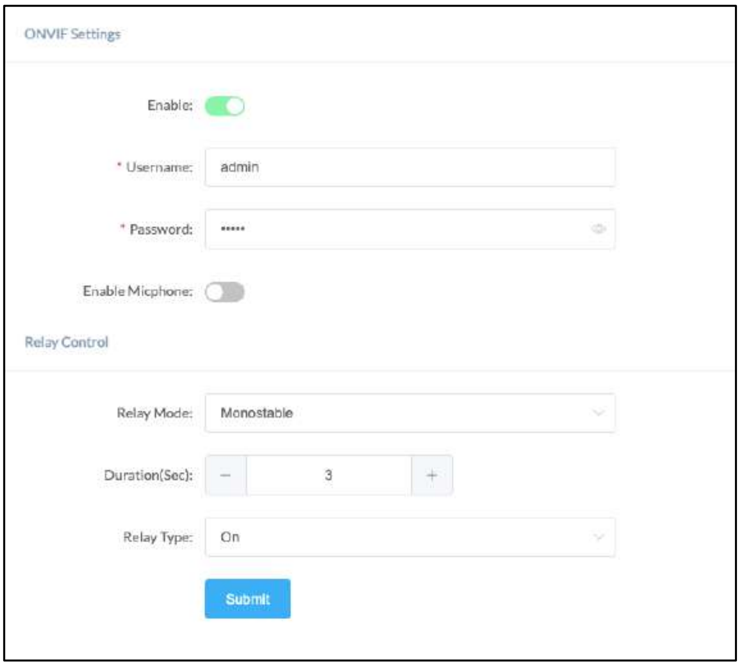
Please keep at least one codec enabled and supported by the SIP server, otherwise, SIP paging will not work.

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 OPA network devices that support ONVIF on it for operation.

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



ONVIF & Relay Control Settings

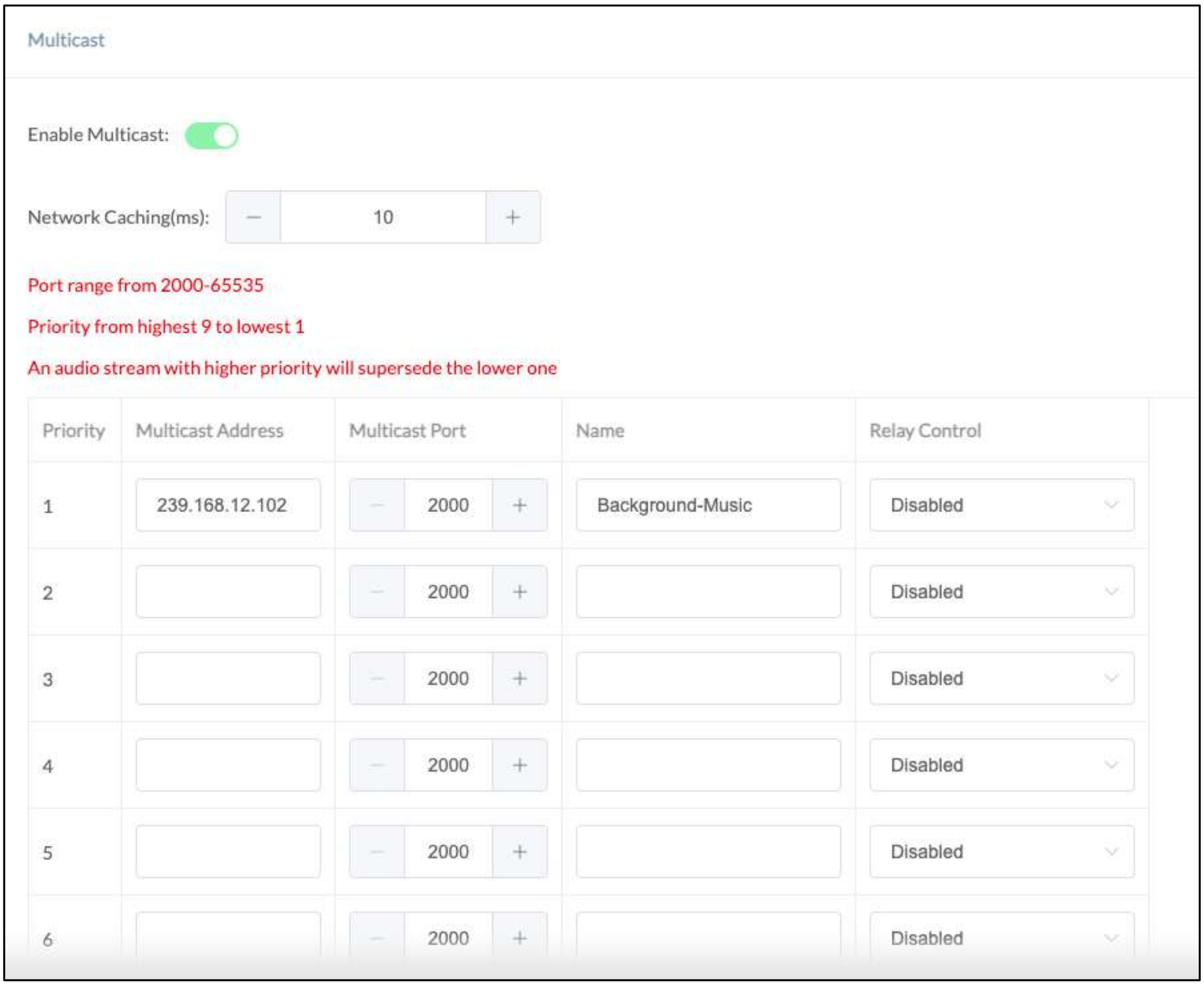
- **Enable:** Enable/Disable ONVIF integration for compatibility with ONVIF-supported VMS platforms.
- **Username:** Enter an account username with matching credentials for adding devices to the VMS platform.
- **Password:** Enter a matching password for the account to add devices to the VMS platform.
- **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: 'On', 'Fast Flashing', or 'Slow Flashing'.

5.2 Multicast

The multicast settings are used to configure the parameter settings of the multicast function on the SIP Safety 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.

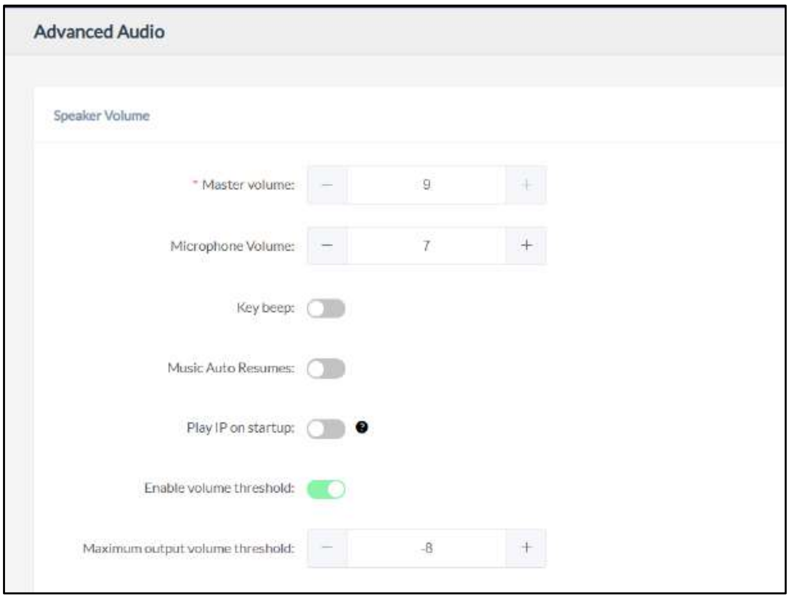
- **Multicast Port:** The multicast port range is 2000 – 65535.
- **Name:** Customize the name of the multicast address.
- **Relay Control:** Options to choose from are ‘Disabled’, ‘On’, ‘Fast Flashing’, ‘Slow Flashing’.

6. Advanced Settings

6.1 Volume Settings

To set the volume of the device, please go to **Advanced --> Advanced Audio** page to configure.

You can adjust the main volume as well as individual volume adjustments for each application (e.g. SIP, multicast, ONVIF...).

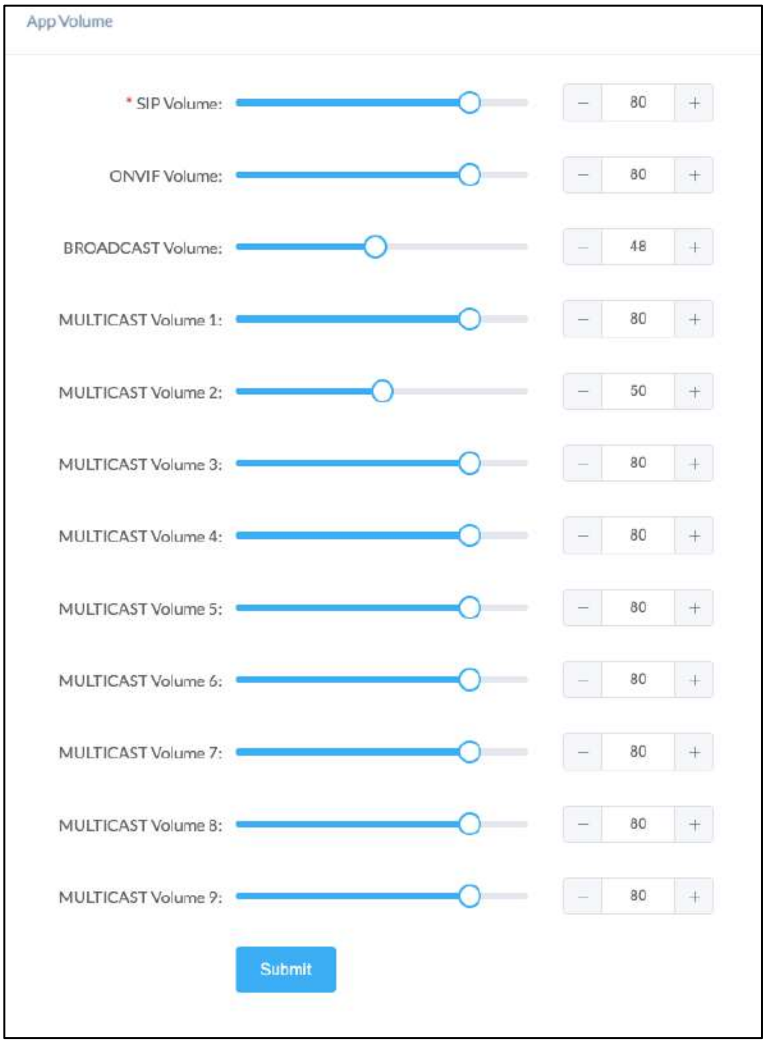


Speaker Volume Settings

- **Master Volume:** Set the speaker master volume. The default volume is 7 and the adjustable range is 0~9.

  Note: It is recommended the speaker volume level setting not exceed 7 under POE power supply mode, otherwise it may cause the device to restart!
- **Microphone Volume:** Set the microphone volume. The default volume is 7 and the adjustable range is 0~9, applicable to C8215、W8215、C8130 and W8130.
- **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.

- **Enable Volume Threshold:** Enable the threshold control to prevent device restarts under PoE power mode.
- **Maximum Output Volume Threshold:** Set the maximum output volume in the range of-50 to 0.



App Volume Settings

- **SIP Volume:** Set the specific volume for SIP application.
- **ONVIF Volume:** Set the specific volume for ONVIF application.
- **Broadcast Volume:** Set the specific volume for broadcast application.
- **Multicast Volume 1:** Set the specific volume for multicast 1 application.
- **Multicast Volume 2:** Set the specific volume for multicast 2 application.
- **Multicast Volume 3:** Set the specific volume for multicast 3 application.
- **Multicast Volume 4:** Set the specific volume for multicast 4 application.

- **Multicast Volume 5:** Set the specific volume for multicast 5 application.
- **Multicast Volume 6:** Set the specific volume for multicast 6 application.
- **Multicast Volume 7:** Set the specific volume for multicast 7 application.
- **Multicast Volume 8:** Set the specific volume for multicast 8 application.
- **Multicast Volume 9:** Set the specific volume for multicast 9 application.

6.2 Audio Priority Settings

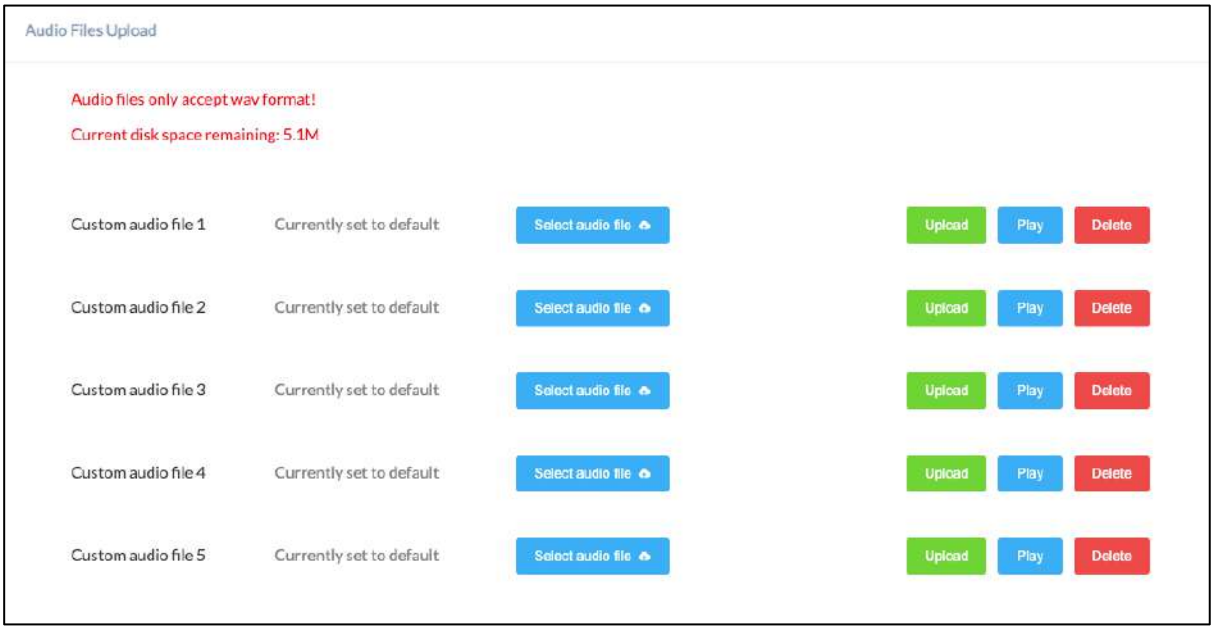
The audio priority can be set according to different applications(such as SIP, ONVIF, MULTICAST, BROADCAST...). Please go to **Advanced ---> Audio Priority** to set the priority.

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 Audio Files

The Audio 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.

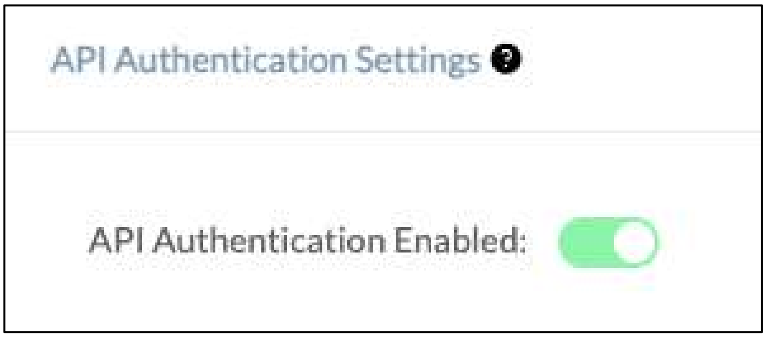


Audio Files

6.4 API Settings

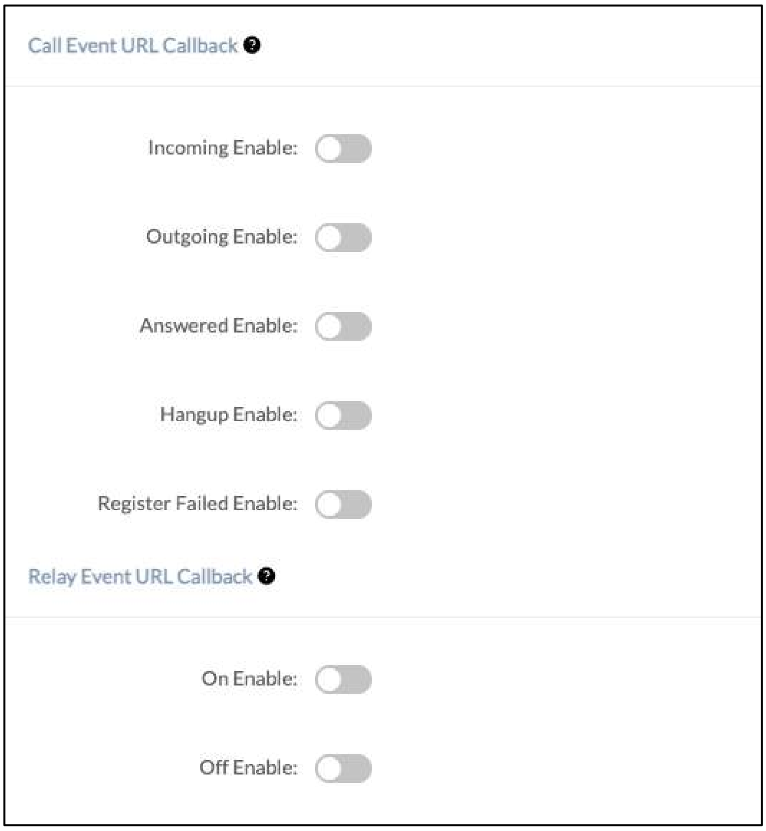
This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the changing status of the call and/or relay.

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

- ## Call 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:

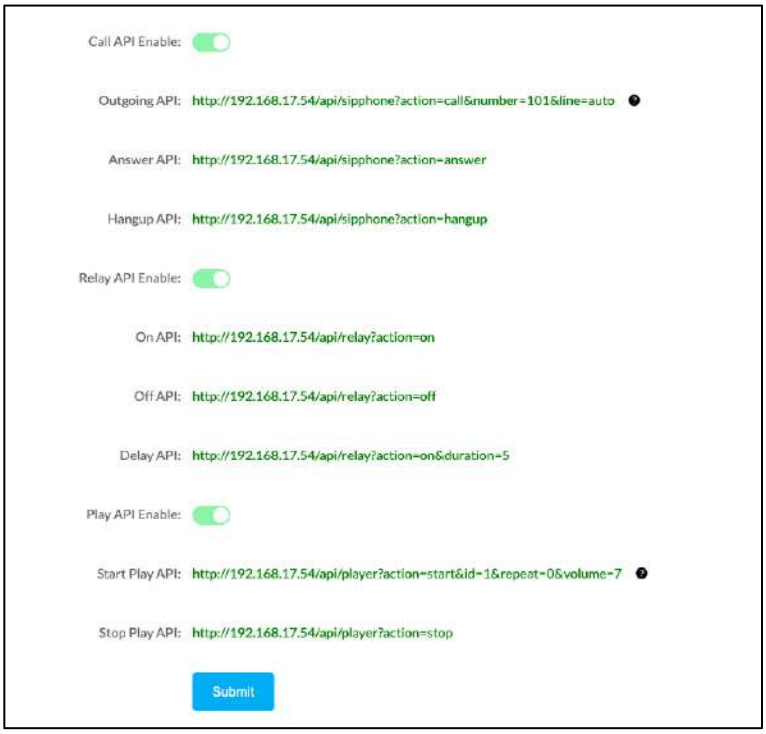
${ip}:	The current IP address of the device
${mac}:	The current MAC address of the device
${ua}:	The account of the current call
${number}:	The number of the current call

- ## Relay Event URL Callback

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

- **API Settings**

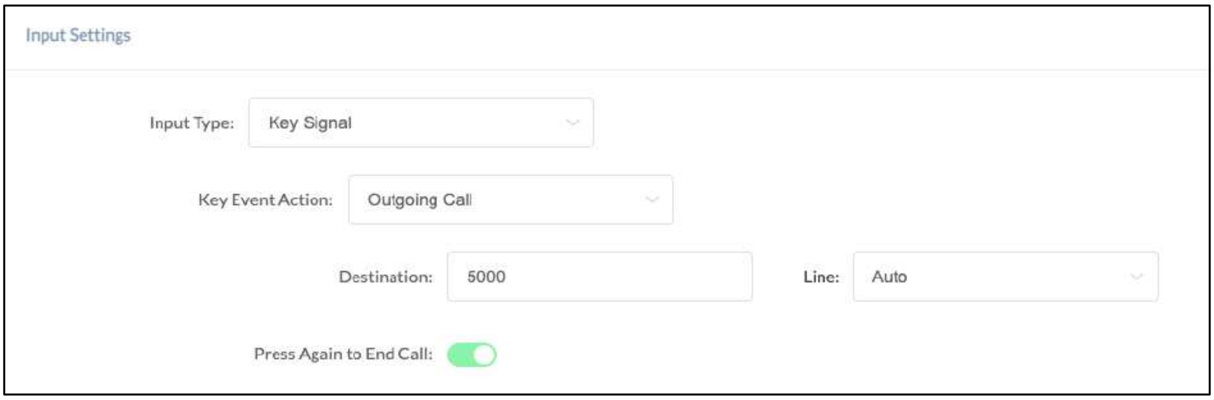
Using the API interface to realize features such as device linkage, call control, relay control, and play sound by the systems.

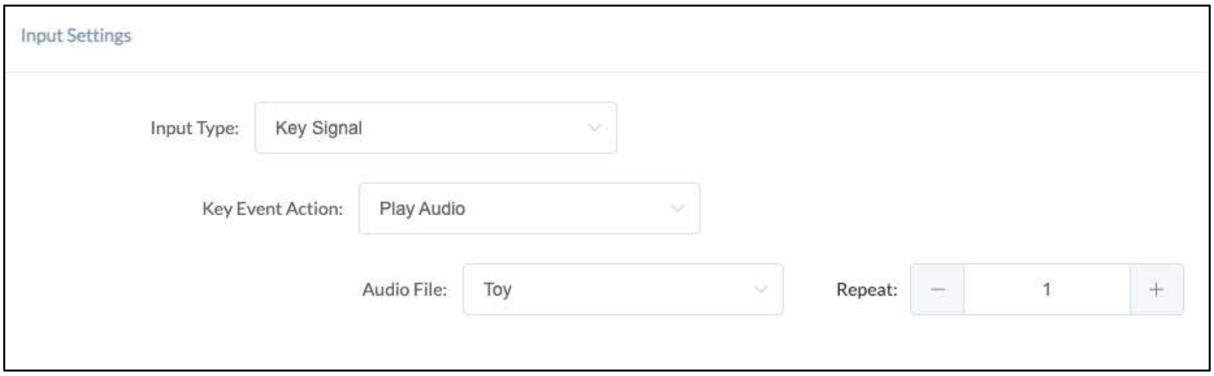
Note: Authentication and encryption are not used in the API interface, so please pay attention to the security of the network environment when opening and using these API interfaces.

6.5 I/O Settings

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

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



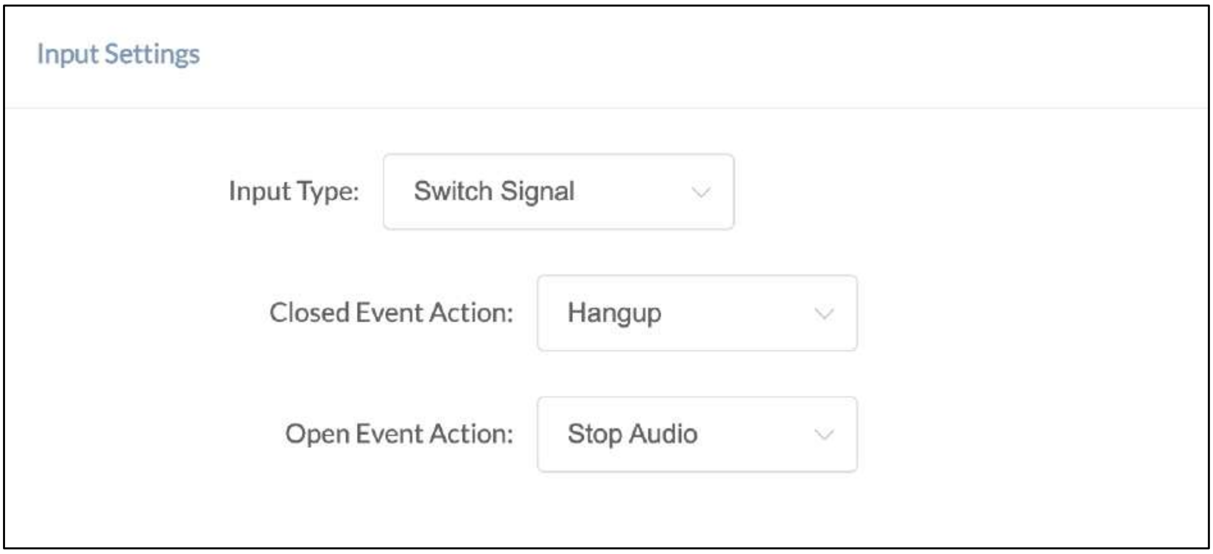


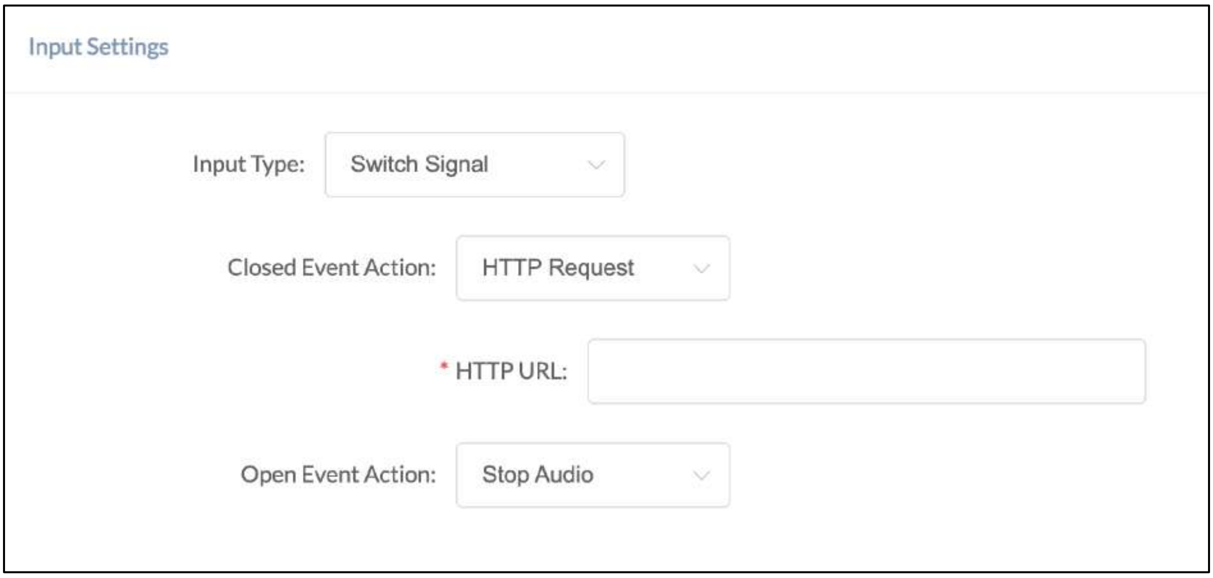
Key Signal Input Settings

- **Input Type:** Choose the Key Signal or Switch Signal to trigger events.
- **Key Event Action:** Choose different event linkage including Outgoing Call, HTTP Request and Play Audio.
- **Outgoing Call:** Make a call to the destination.
- **Destination:** This setting represents the response device's number when the external button is pressed.
- **Line:** This setting represents the corresponding line for making outgoing calls.

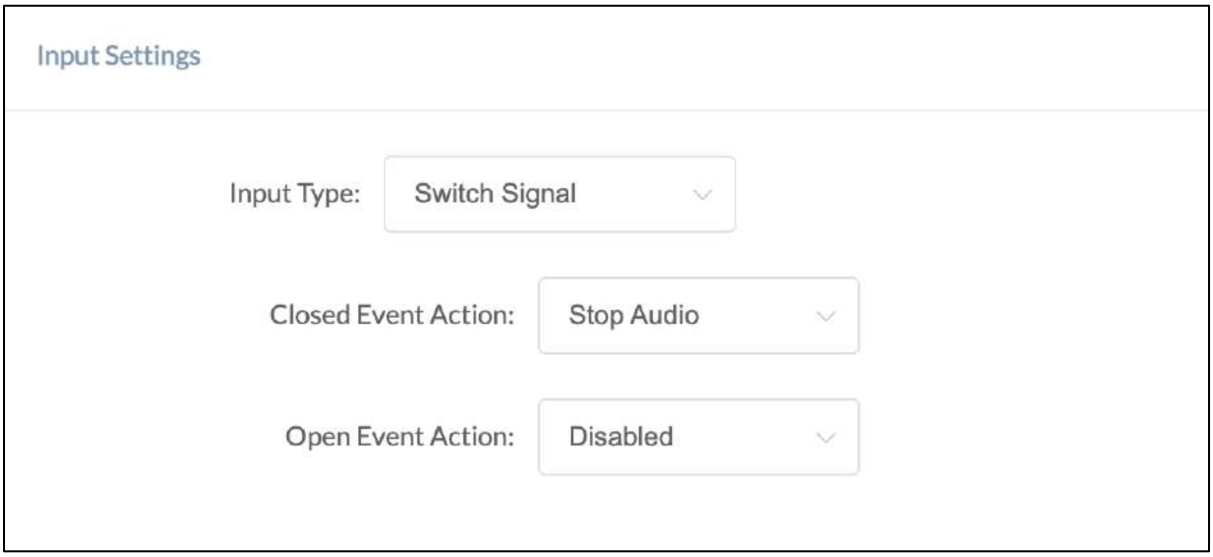
Note: When using the P2P line to call, please specify the device's number + IP address, such as 101@192.168.11.123.

- **Press Again to End Call:** After the call is connected, users can end the call or conversation by pressing the button again.
- **HTTP URL:** Configure the API URL address triggered by linkage.
- **Play Audio:** Play the pre-configured audio file.
- **Audio File:** Configure the audio triggered by linkage.
- **Repeat:** Configure the times of audio repetitions triggered by linkage.




Input Settings
Input Type:
Switch Signal
Closed Event Action:
HTTP Request
* HTTP URL:
Open Event Action:
Stop Audio


Input Settings
Input Type:
Switch Signal
Closed Event Action:
Play Audio
Audio File:
Sweet
Repeat:
0
Open Event Action:
Stop Audio

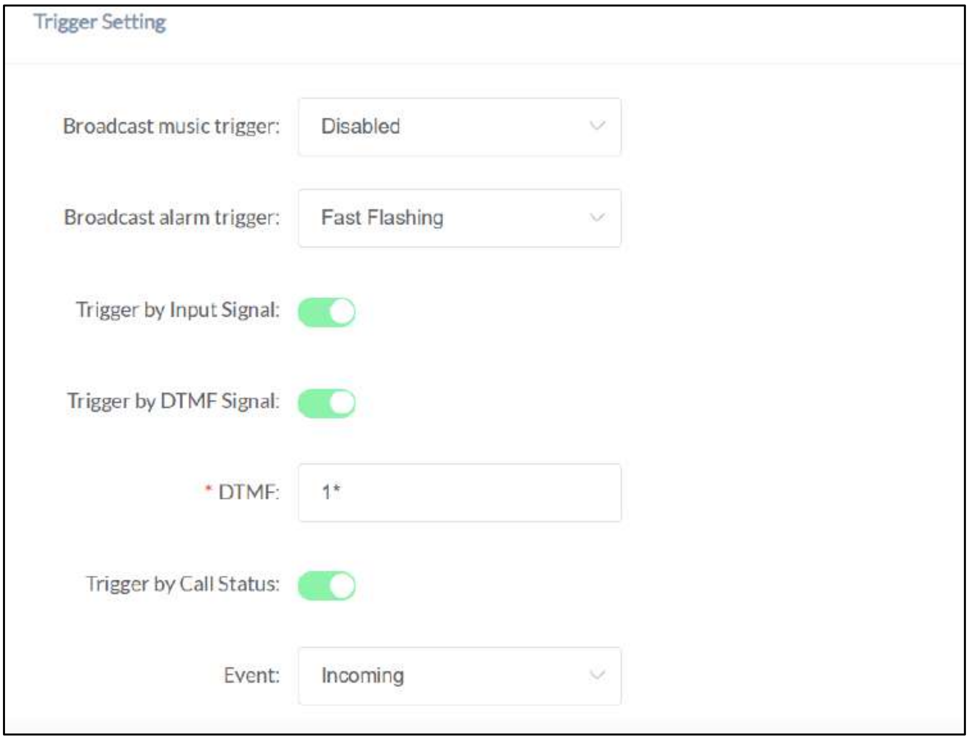

Input Settings
Input Type:
Switch Signal
Closed Event Action:
Stop Audio
Open Event Action:
Disabled

Switch Signal Input Settings

- **Input Type:** Choose the Key Signal or Switch Signal to trigger events.
- **Close/Open Event Action:** Choose different event linkage for the close/open status, including Outgoing Call, Hangout, HTTP Request, Play Audio and Stop Audio.
- **Outgoing Call:** Make a call to the destination.
- **Destination:** This setting represents the response device's number.
- **Line:** This setting represents the corresponding line for making outgoing calls.

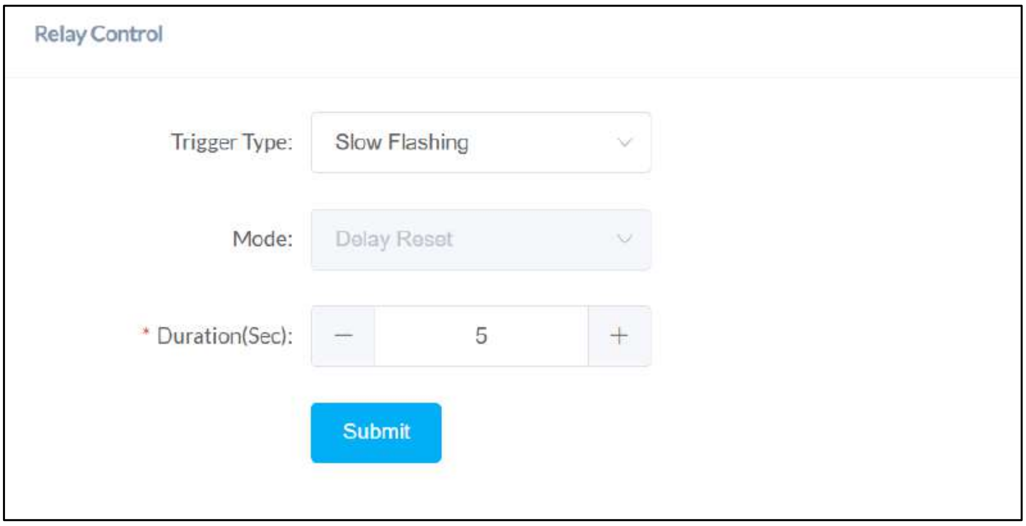
Note: When using the P2P line to call, please specify the device's number + IP address, such as 101@192.168.11.123.

- **Hangup:** Hangup the Outgoing Call.
- **HTTP URL:** Configure the API URL address triggered by linkage.
- **Play Audio:** Play the pre-configured audio file.
- **Audio File:** Configure the audio triggered by linkage.
- **Repeat:** Configure the times of audio repetitions triggered by linkage.
- **Stop Audio:** Stop playing the pre-configured audio file.



Trigger Settings

- **Broadcast Music Trigger:** Disabled/On/Fast Flashing/Slow Flashing, enable this option will trigger the relay when there is broadcast music on.
- **Broadcast Alarm Trigger:** Disabled/On/Fast Flashing/Slow Flashing, enable this option will trigger the relay when there is a broadcast alarm on.
- **Trigger by Input Signal:** Enable/Disable, enable this option if need to use the input signal to trigger.
- **Trigger by DTMF Signal:** Enable/Disable, enable this option when need to use DTMF signal to trigger (only RFC2833 supported).
- **DTMF:** This setting represents the number to dial to trigger DTMF.
- **Trigger by Call Status:** Enable/Disable, enable this option will change the call status when triggered.
- **Event:** Set the corresponding call state, you can choose 【Outgoing】,【Incoming】,【Incoming/Outgoing】,【Answered】 and 【Hangup】.



Relay Control

- **Trigger Type:** This setting represents the responses by the triggers, there are ‘On’, ‘Fast Flashing’, and ‘Slow Flashing’ options to choose from.
- **Mode:** This setting represents the reset mode after the trigger is responded, there is only ‘Delay Reset’ options to choose from.

- **Duration (Sec):** This setting is only available if the reply control mode is on Delay Reset, it represents the time duration when the configure interface status changed.

6.6 PTP Settings

PTP (Precision Time Protocol) is a network time protocol used to provide high-precision time synchronization. Please go to **Advanced ---> PTP Settings** page to set. After enabling PTP settings, you can manually set the PTP server to improve the synchronization of the music playback clock.

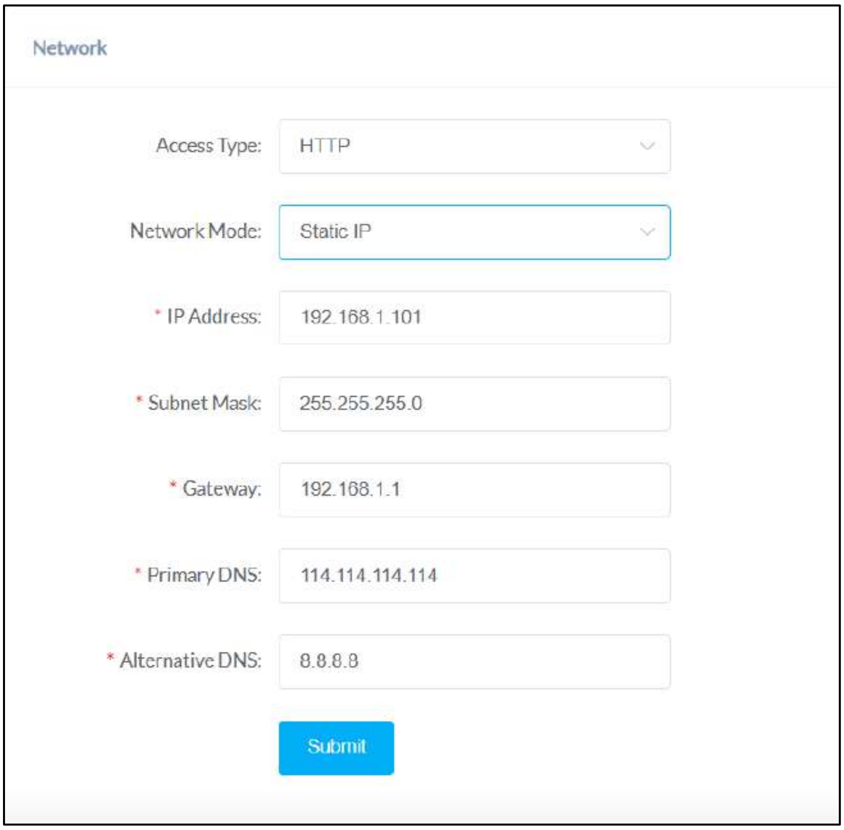
PTP Setting

7. System Settings

7.1 Network

The device 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. Select Static IP to show the network parameter settings.



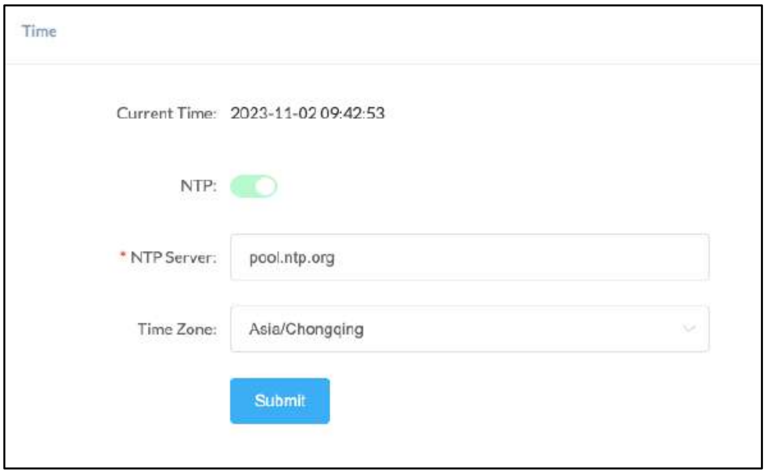
Network Configuration

- **Access Type:** Specify the access method of the website, which currently supports HTTP、HTTPS and HTTP& HTTPS.
- **Network Mode:** Select DHCP or Static IP
- **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

The device 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.

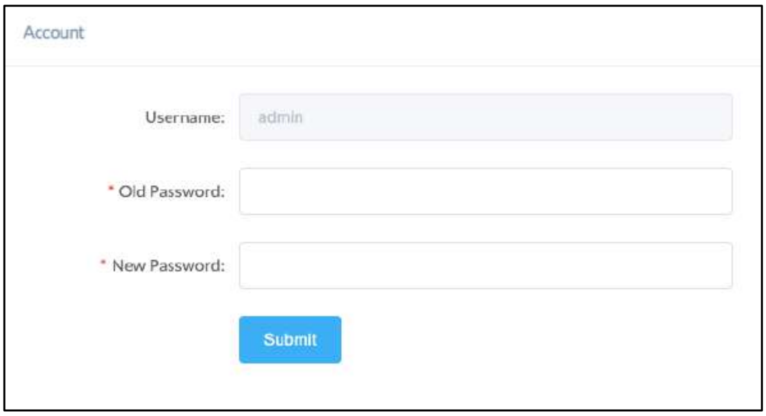
7.3 Prompt Language

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.



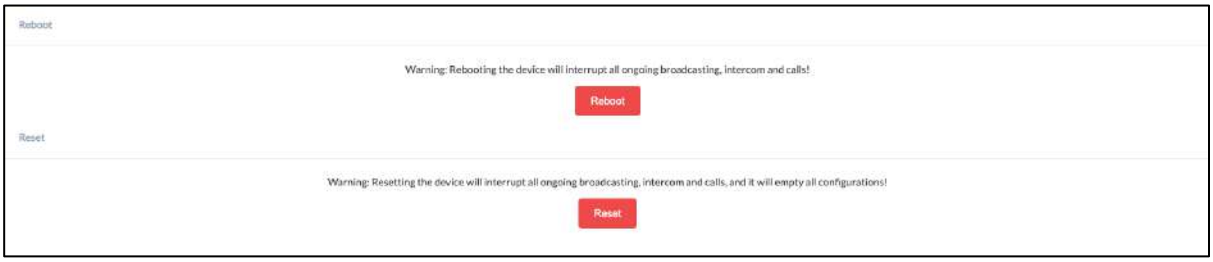
Web Password Settings

- **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

The device 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 device, you can reset it through this page, after restarting, the current IP address of the device will be automatically announced by voice prompts. Users can log in to the device's Web management interface with this IP address.

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

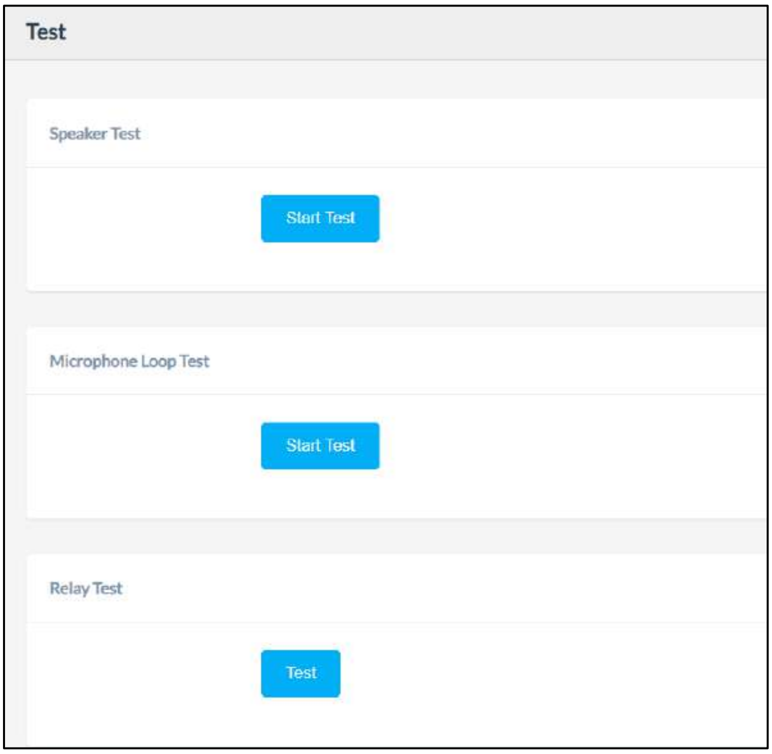
The purpose of the Ethernet capture tool is to capture Ethernet network packets and store them in a standard Wireshark-compatible packet capture '.pacp' file for immediate viewing and data analysis.

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, insert microphone device, the sound detection is normal（some models do not have this option）.

- **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.

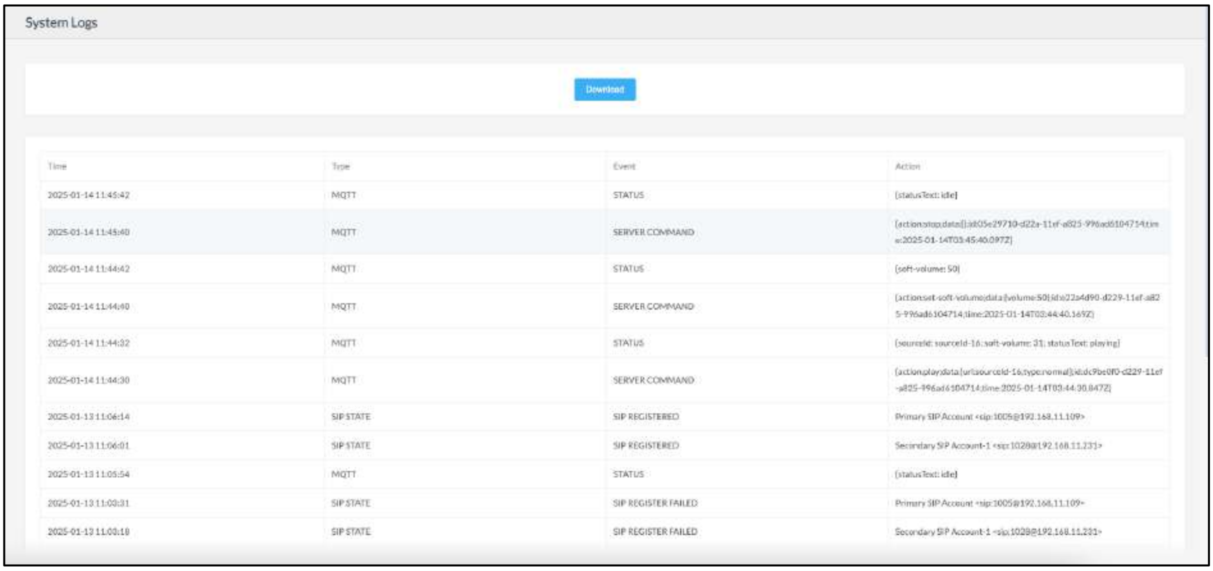
Date	Time	Account	Telephone Number	Duration	Type	Status
2024-03-15	17:18:00	sip:5011@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:47:58	sip:5015@192.168.11.62	5000	00:00:02	Inbound	Answered
2024-03-15	16:38:30	sip:5005@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:38:16	sip:5005@192.168.11.62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



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