MONITOREAL

USER MANUAL

Monitoreal IP Speaker

ONVIF network horn speaker with microphone



See what matters.

monitoreal.com

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BEFORE YOU START

The speaker obtains its IP address over DHCP. The DHCP client runs for 165 seconds; if no address is obtained in that time, the device falls back to the static IP address 192.168.1.101. Switching the device off and on again restarts the DHCP process.

1. Overview

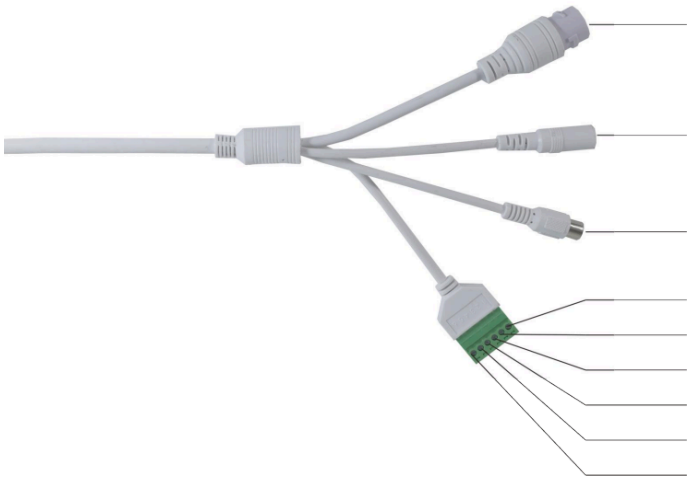
1.1 Product Overview

This series of ONVIF Network Horn Speaker with built-in microphone is ideal for indoor and outdoor use. Built for durability and exceptional audio quality, they are ideal for public address, emergency alerts, and broadcasting in schools, hospitals, malls, and transportation hubs. SIP-enabled with PoE and DC power, they integrate seamlessly with IP systems.

1.2 Product Specifications

Product Type	Horn Speaker with Microphone
Rated Power	30W
Network Protocol	SIP, ONVIF
Audio Decoding	MP3, WMA, WAV
Network Interface	10/100Base-TX, RJ45
SPL (1M/1W)	105dB
Max. SPL	120dB
Frequency Response	350Hz–8kHz
IP Rating	IP66
Power Supply	PoE / DC12V
Installation	Bracket Mount
Material	ABS
Product Dimension	285x205x290mm
Net Weight	2.3kg

1.3 Wiring



RJ45 Network Interface

DC12V Power Input

Audio Input

1. COM
2. I/O
3. Alarm Input
4. Alarm Output
5. NC
6. NC

2. Login the Device

2.1 Accessing the Web GUI

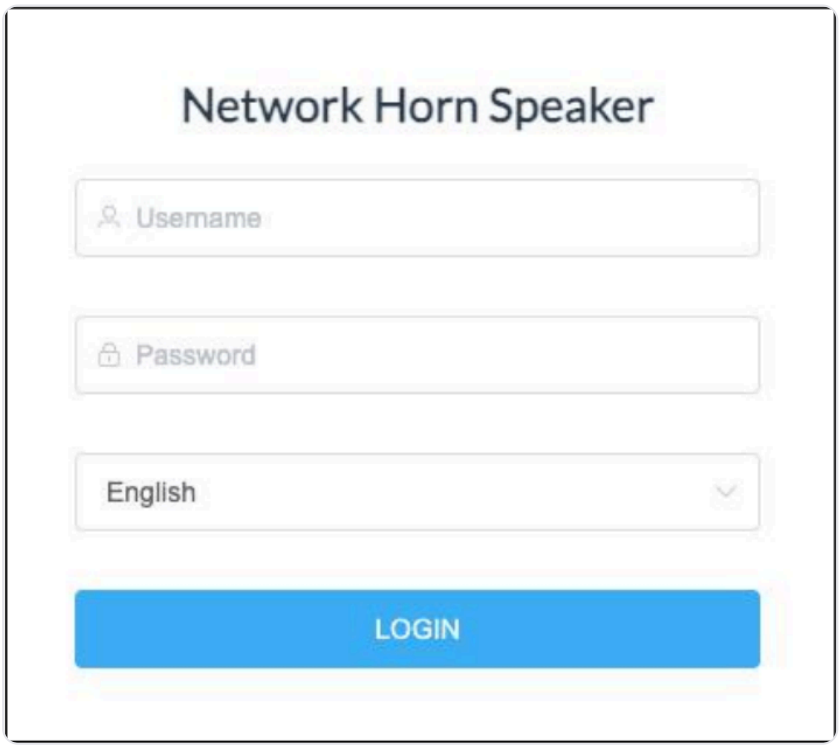
This speaker obtains the IP address through DHCP by default, please ensure that there is an available DHCP server in your LAN (if DHCP fails to obtain an address, it will use a static IP address: 192.168.1.101), and enter the IP address in the browser to access the device's Web management interface.

DHCP TIMING

The DHCP client runs for 165 seconds. If no address is obtained within this time, the device switches to the static IP address 192.168.1.101. Switching the device off and on again restarts the DHCP process from the beginning.

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.

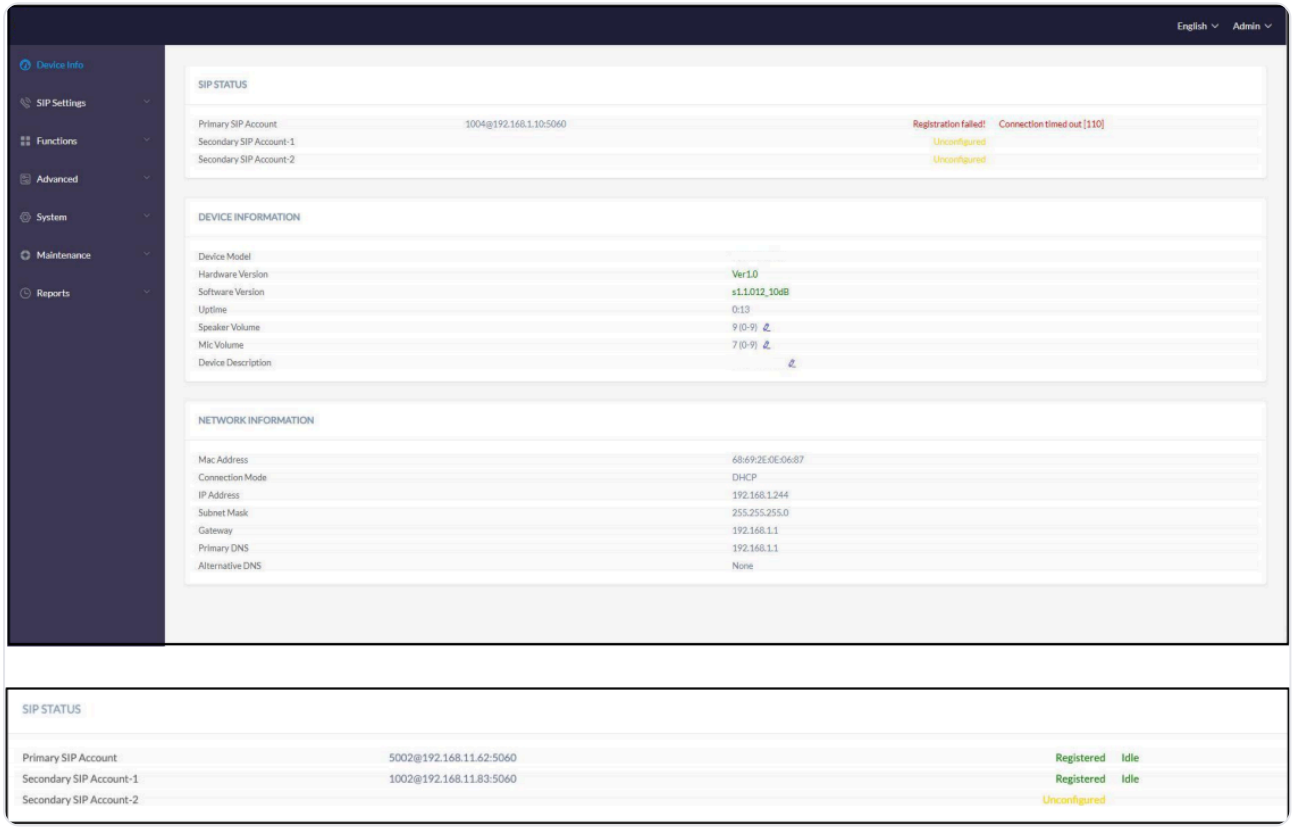


Login Interface

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

2.2 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.
- **Device Description:** Remark the device information. 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

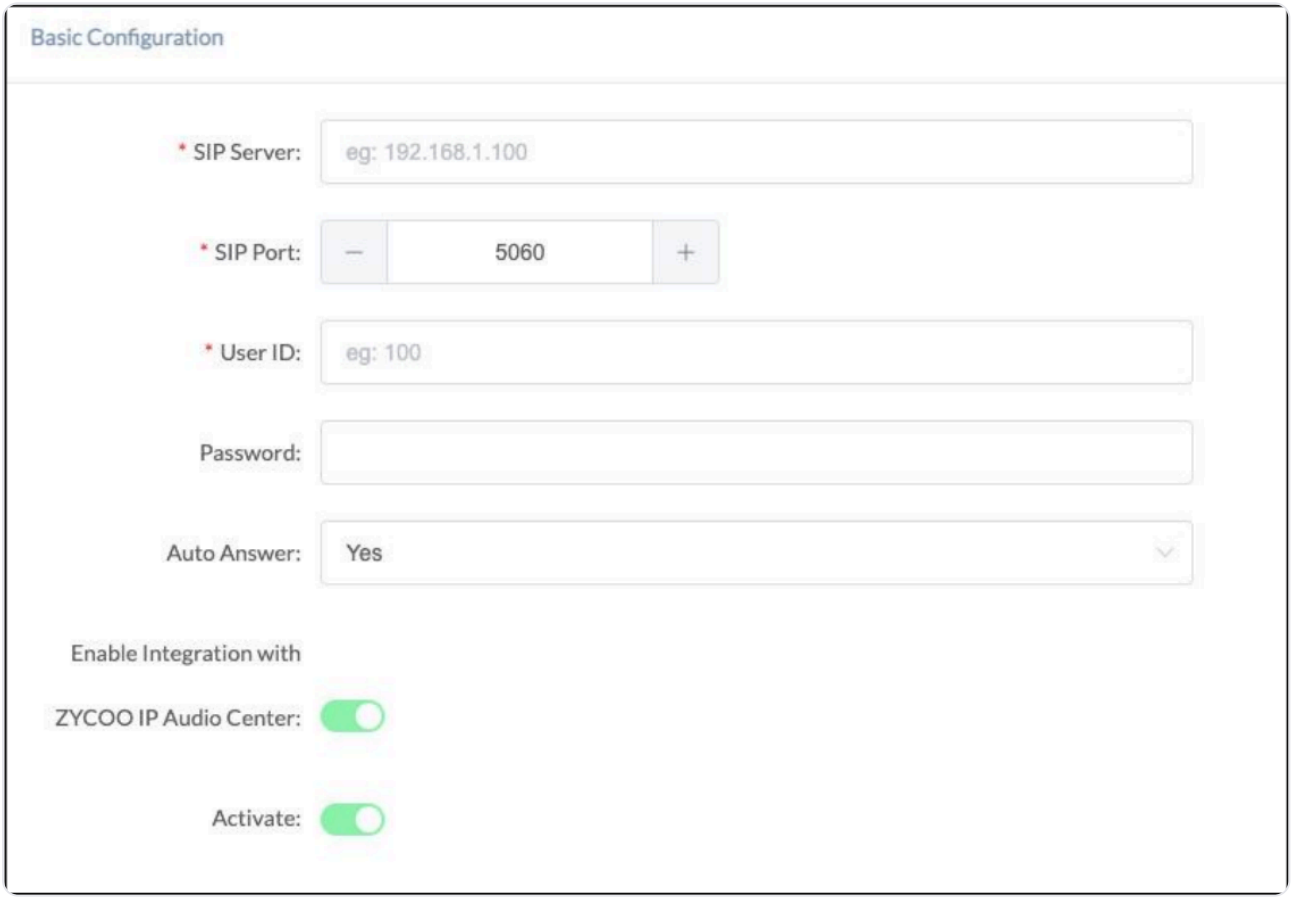
- **Mac Address:** Display the MAC address of the current device.
- **Connection Mode:** Display the network acquisition method of the device, DHCP (dynamic acquisition) or STATIC (static configuration).
- **IP Address:** The current IP address of the device.
- **Subnet Mask:** The current subnet mask 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.

3. SIP Settings

3.1 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 IP Audio Center, please turn on the 'Enable Integration with IP Audio Center' option.

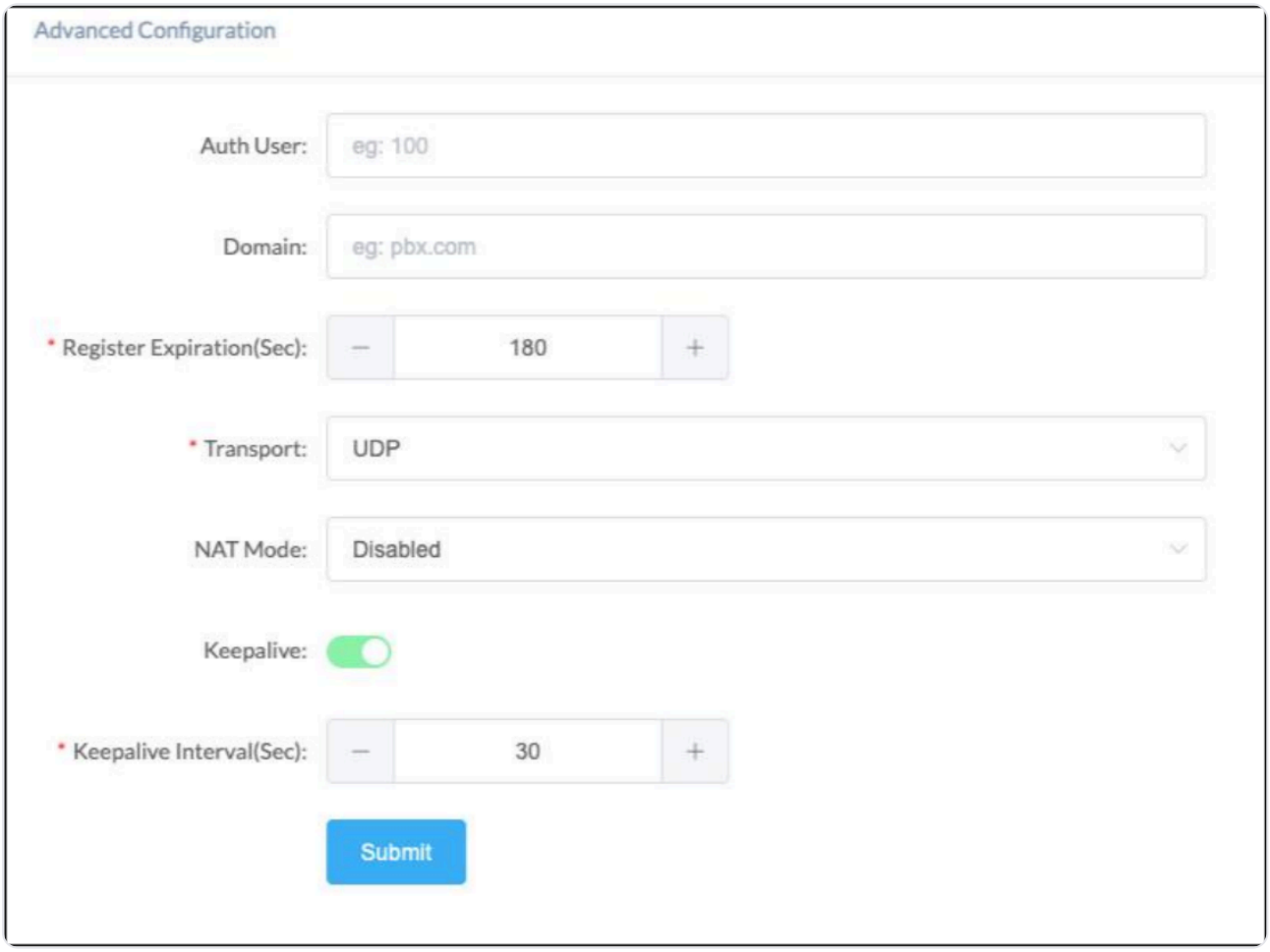
Please go to SIP Settings → Primary SIP Account / Secondary SIP Account-1 / Secondary SIP Account-2 page.



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

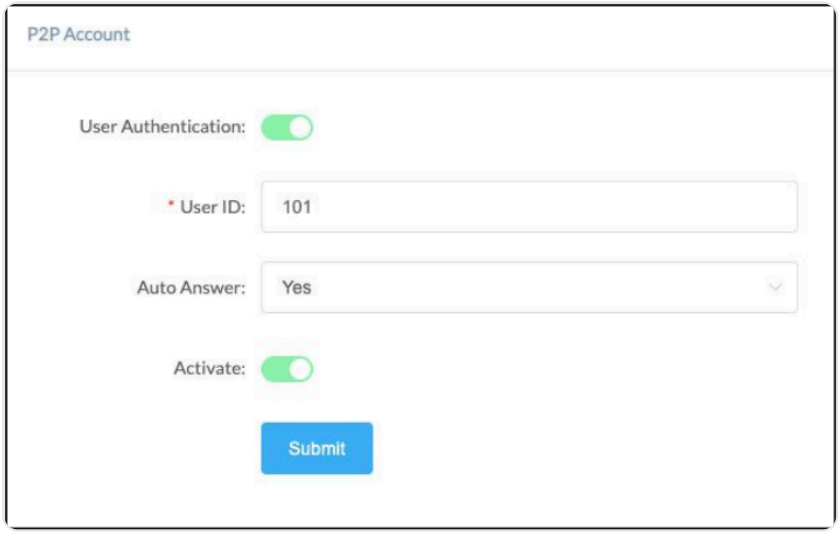
- **Auth User:** Enter the authorized username for the SIP account.
- **Domain:** Enter the SIP Domain.
- **Register Expiration (sec):** Set the SIP registration expiration time, with a default of 180 seconds.
- **Transport:** Choose the transport protocol: UDP, TCP, or TLS.
- **NAT Mode:** Select the 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.

3.2 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

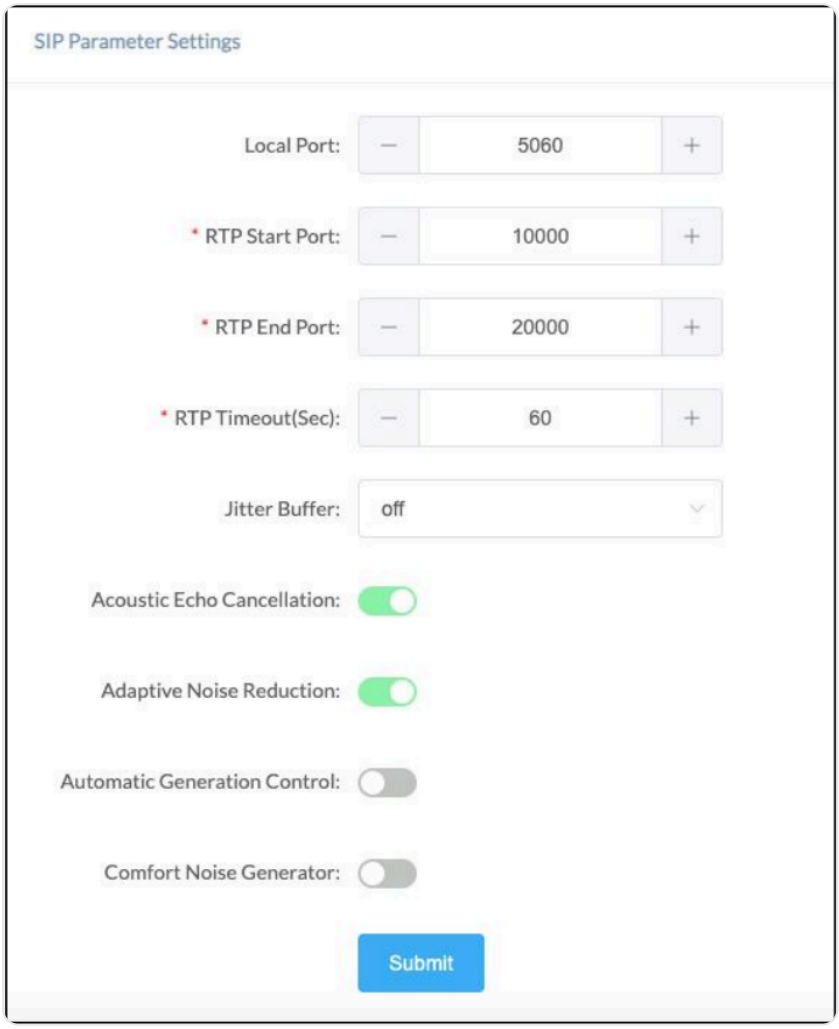
- **User Authentication:** Enable/Disable P2P authentication. If disabled, you can directly enter this device's IP address in the target field of the peer device. If enabled, you must use the following format in the target field of the peer device: this device's P2P User ID + IP address (e.g., 101@192.168.1.101).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

3.3 Advance SIP Settings

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

3.3.1 SIP Parameter Settings

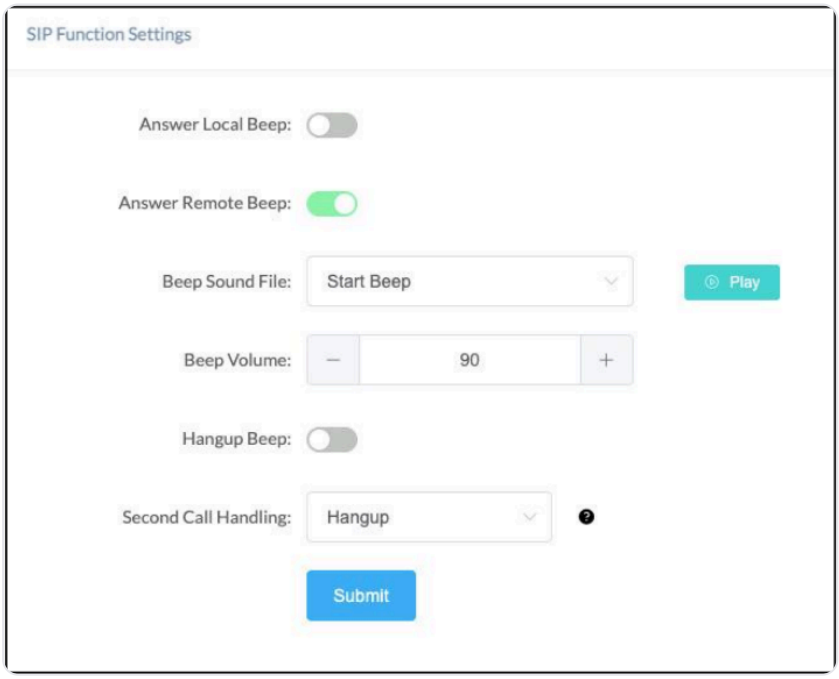


SIP Parameter Settings

- **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 Generation 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.
- **Comfort Noise Generator:** After enabling this feature, comfortable white noise can be added during calls.

3.3.2 SIP Function Settings

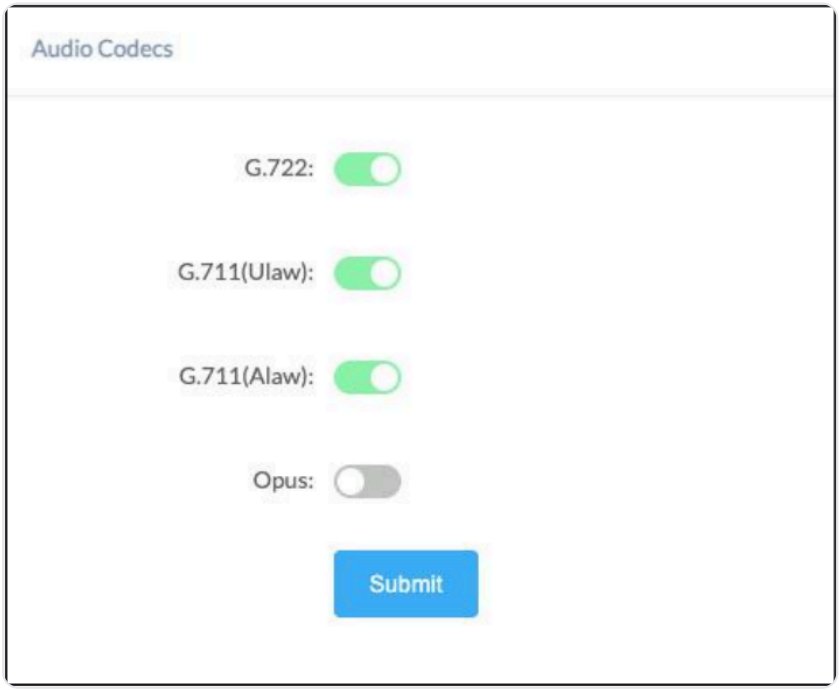


SIP Function Settings

- **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:** 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.

3.3.3 Audio Codecs

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



Audio Codecs

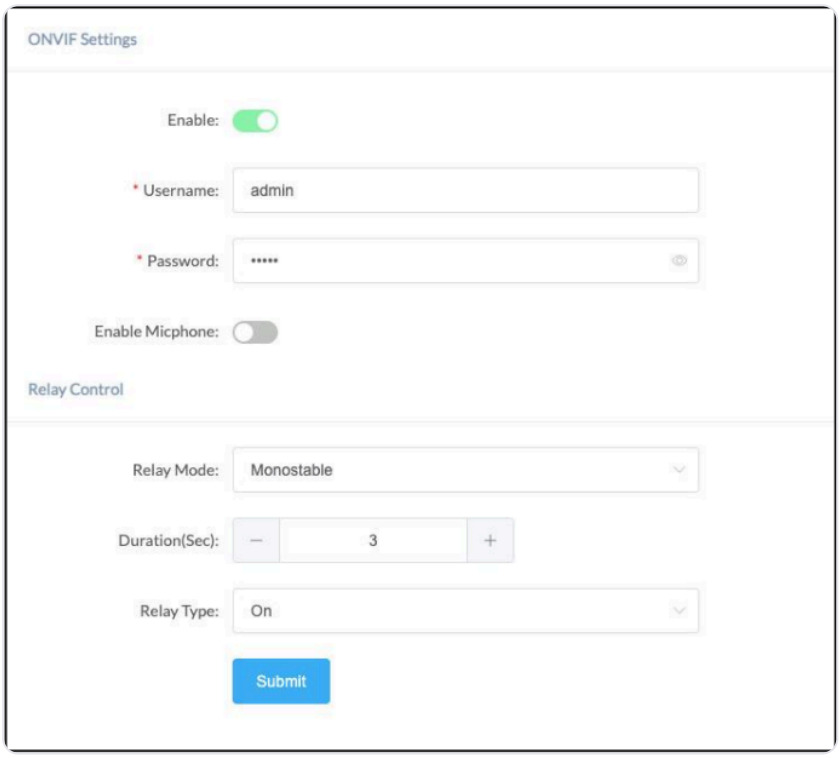
Please keep at least one codec enabled and supported by the SIP server, otherwise, SIP paging will not work.

4. Function Settings

4.1 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 network devices that support ONVIF on it for operation.

Please go to Functions → ONVIF Settings to configure the ONVIF settings.



ONVIF & Relay Control Settings

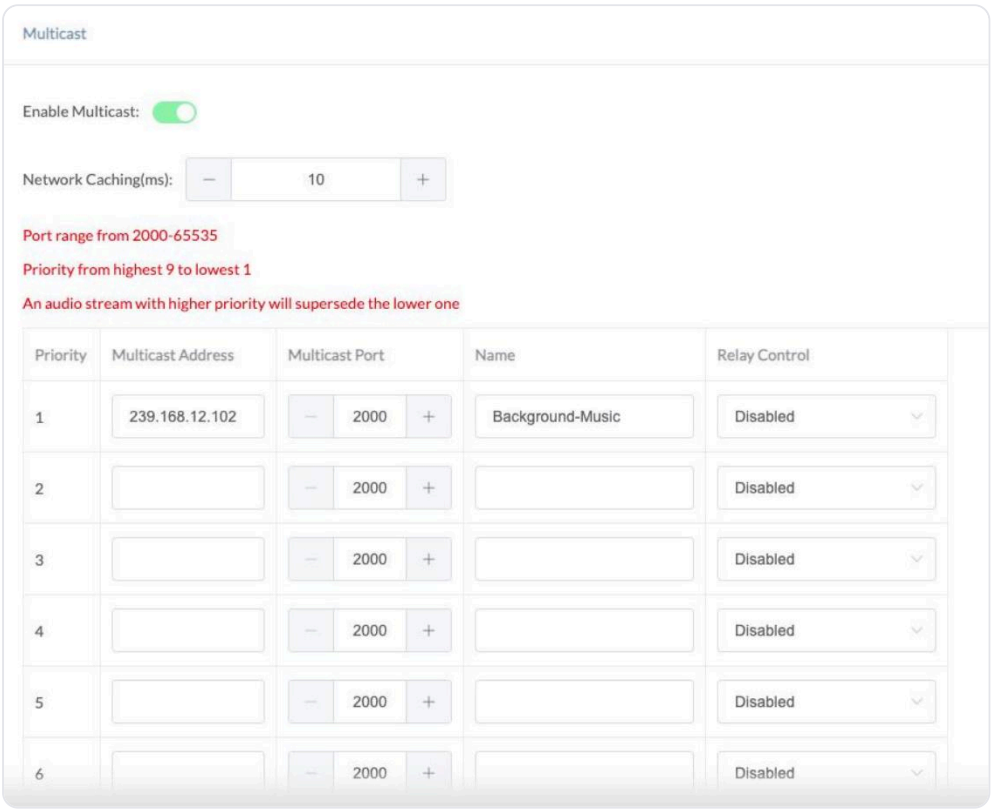
- **Enable:** Enable/Disable ONVIF integration for compatibility with ONVIF-supported VMS platforms.
- **Username:** Enter an account username with matching credentials for adding devices to the VMS platform.
- **Password:** Enter a matching password for the account to add devices to the VMS platform.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

4.2 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.



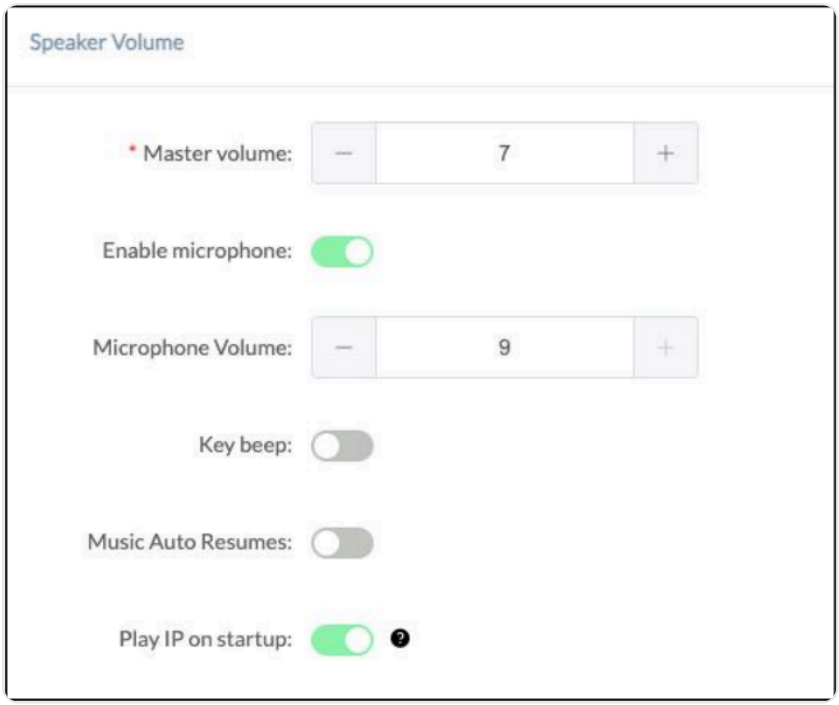
Multicast

- **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.

5. Advanced Settings

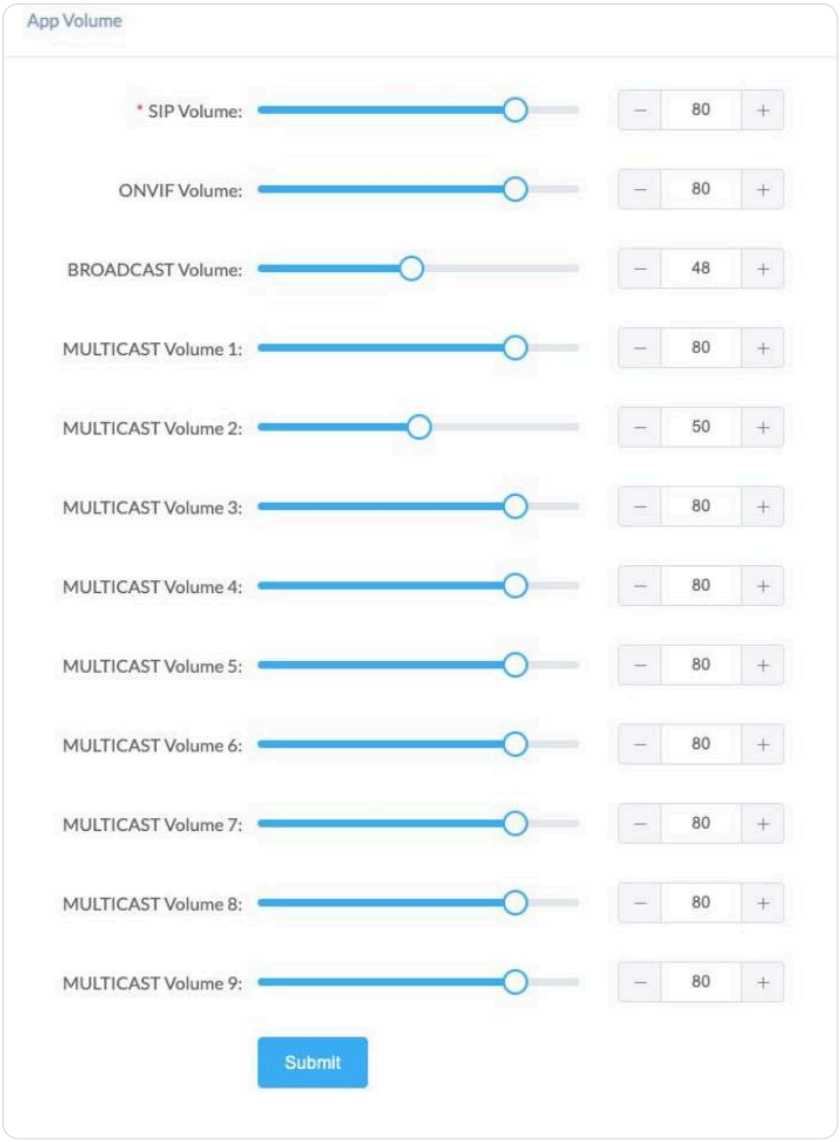
5.1 Volume Settings

To set the volume of the speaker, please go to Advanced Settings → Volume page to configure. You can adjust the main volume as well as individual volume adjustments for each application.



Speaker Volume Settings

- **Master Volume:** Set the speaker master volume. The default volume is 7 and the adjustable range is 0–9.
- **Enable Microphone:** Enable/Disable the built-in microphone.
- **Microphone Volume:** Set the microphone volume. The default volume is 7 and the adjustable range is 0–9.
- **Key Beep:** Enable/Disable the beep sound from the 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.



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.

5.2 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.

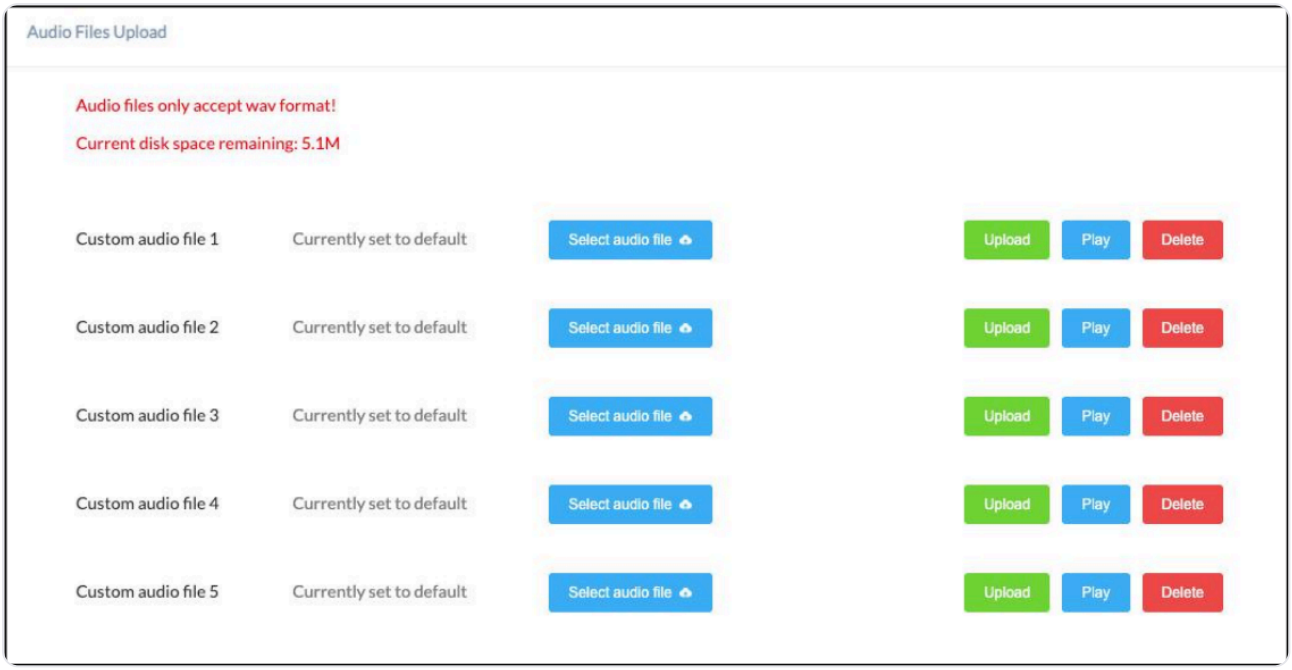


Audio Priority Settings

5.3 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.

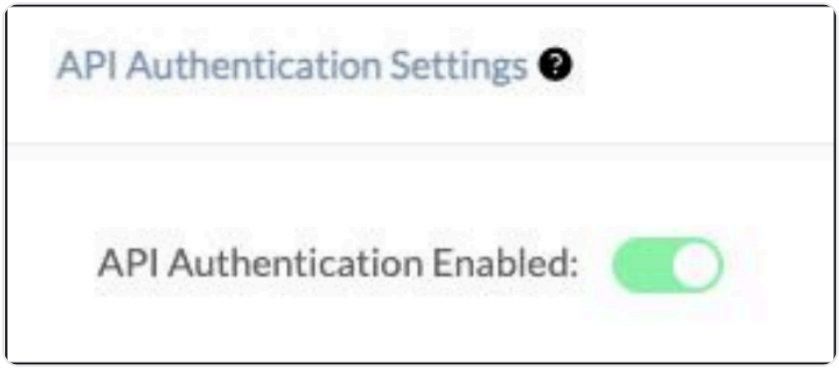


Audio Files

5.4 API Settings

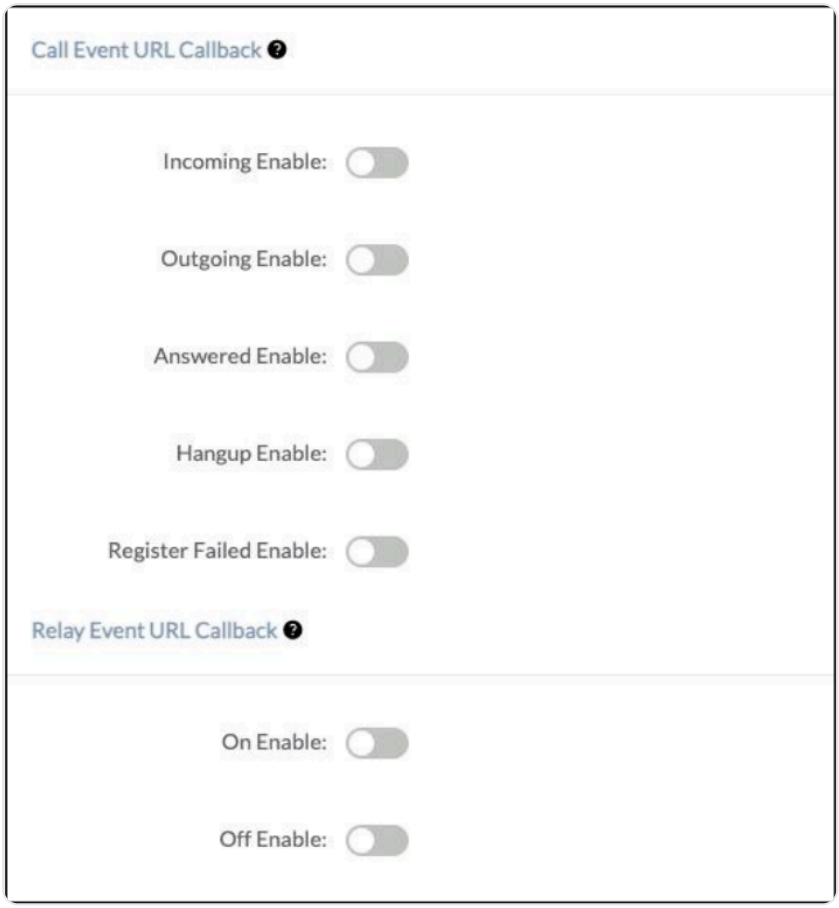
This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the changing status of the call and/or relay.

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.



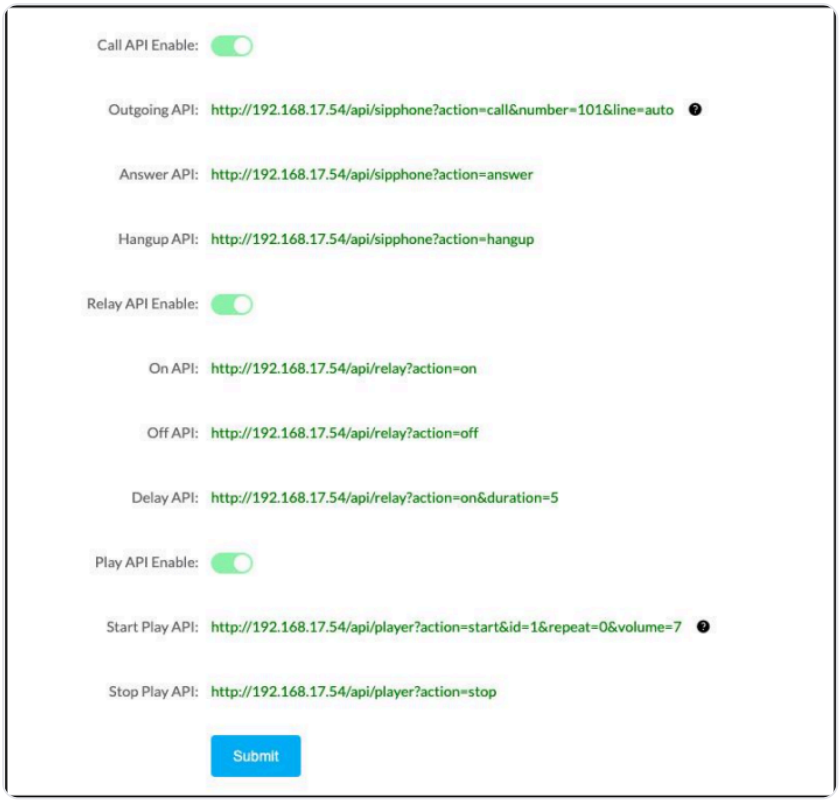
Call Event URL Callback & Relay Event URL Callback

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

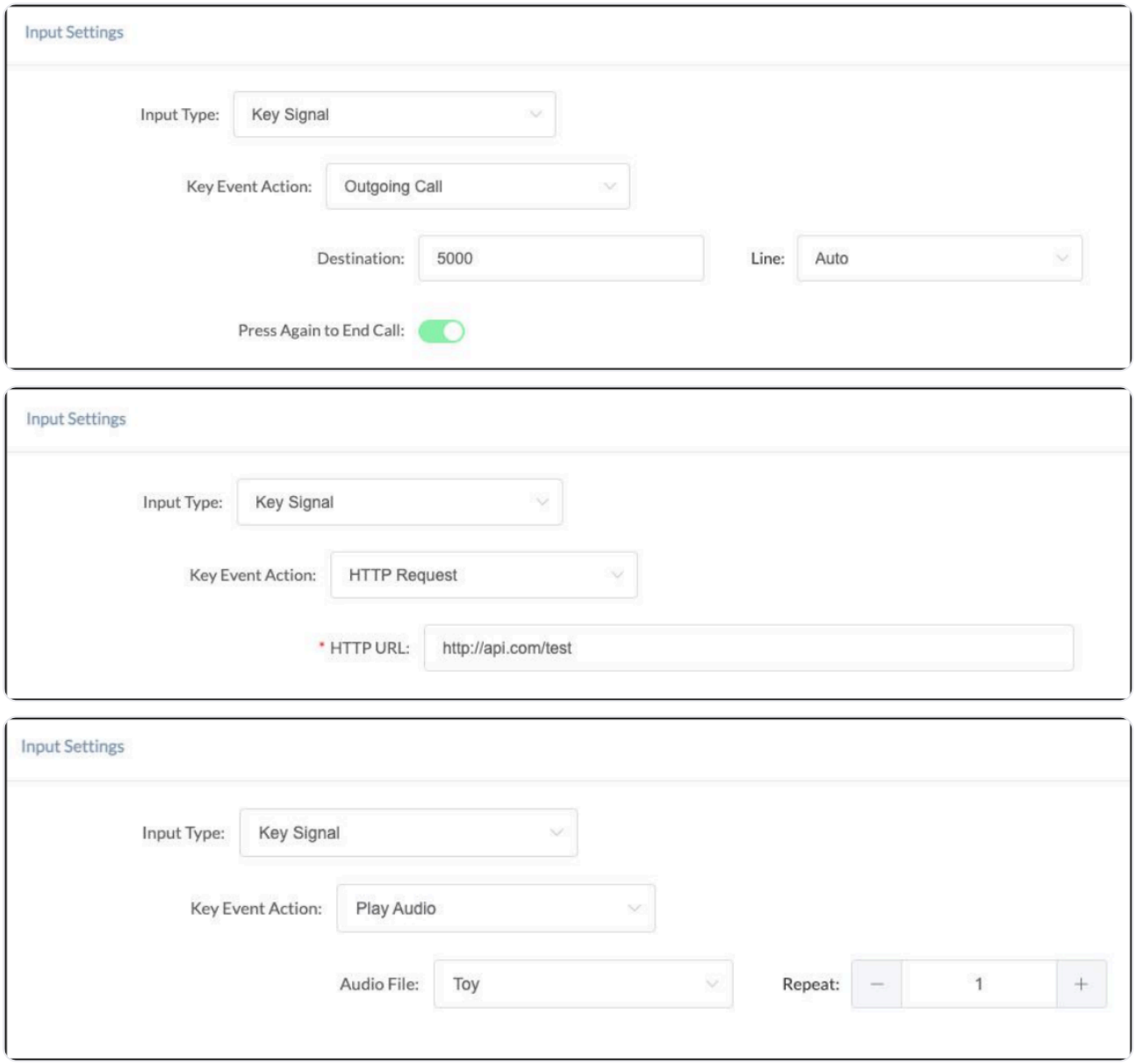
Using the API interface to realize features such as device linkage, call control, relay control, and play sound by the systems.

> 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.

5.5 I/O Settings

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

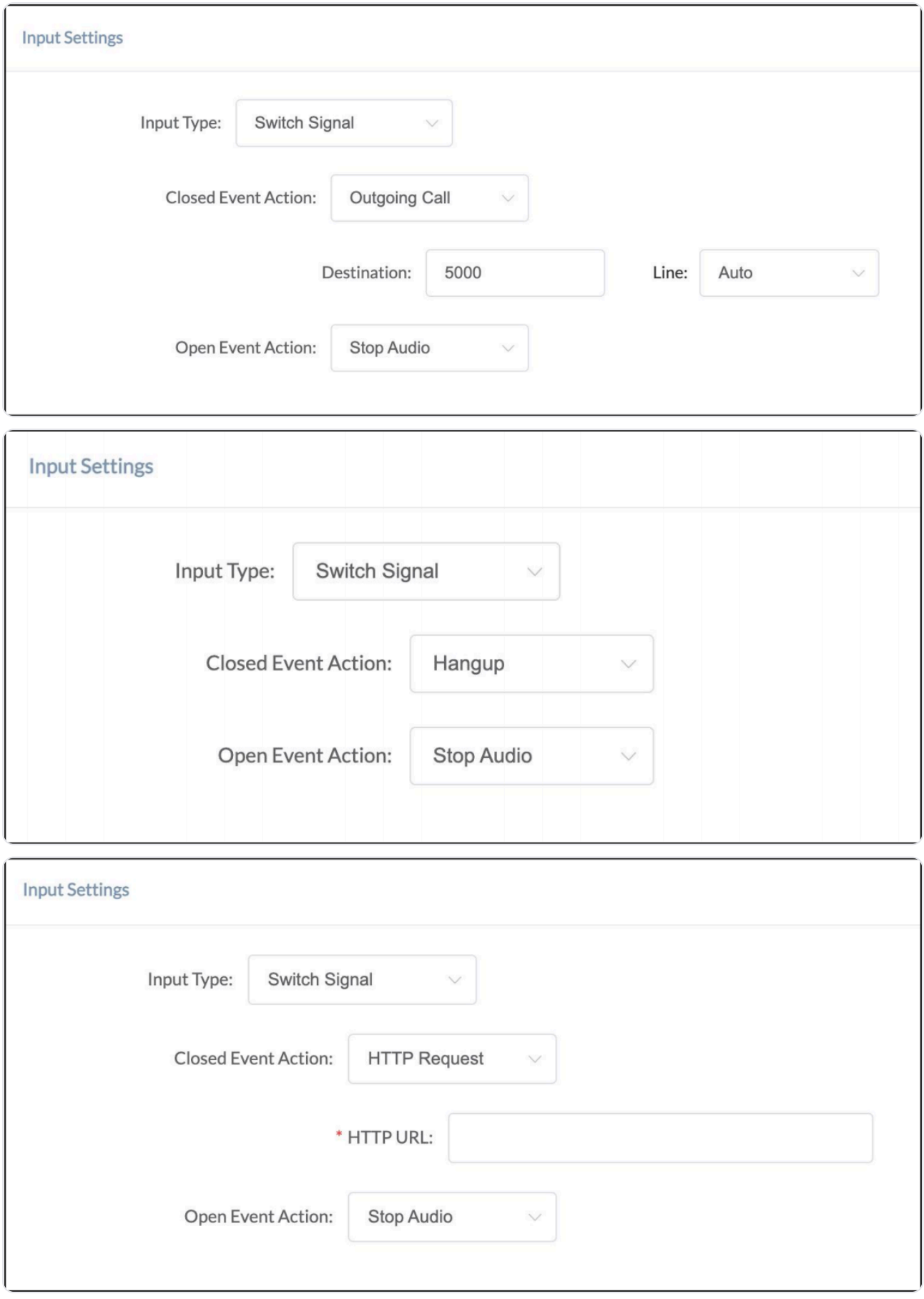
Please go to Advanced Settings → I/O Settings page to set the specific settings.



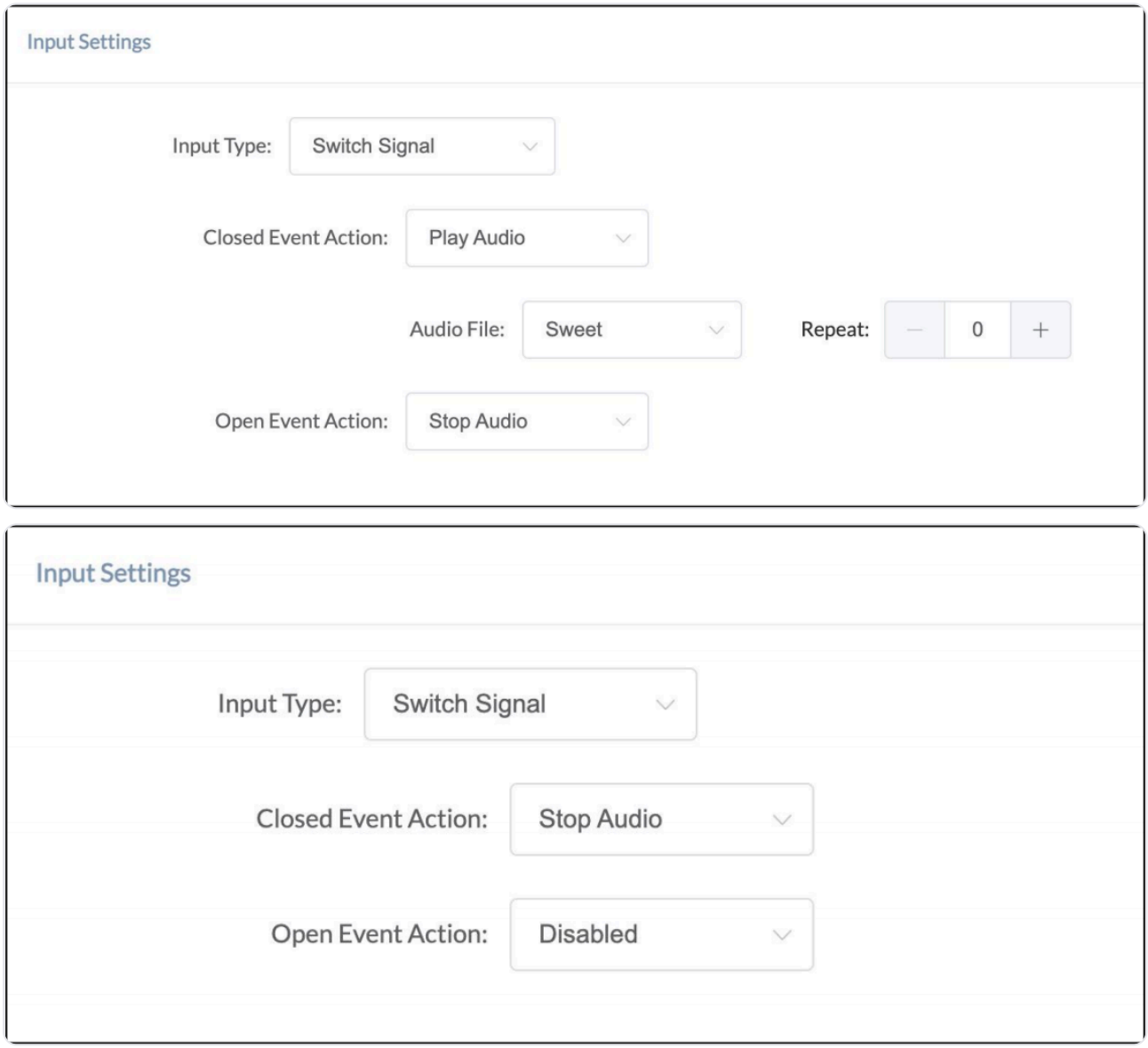
Key Signal Input Settings

- **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.



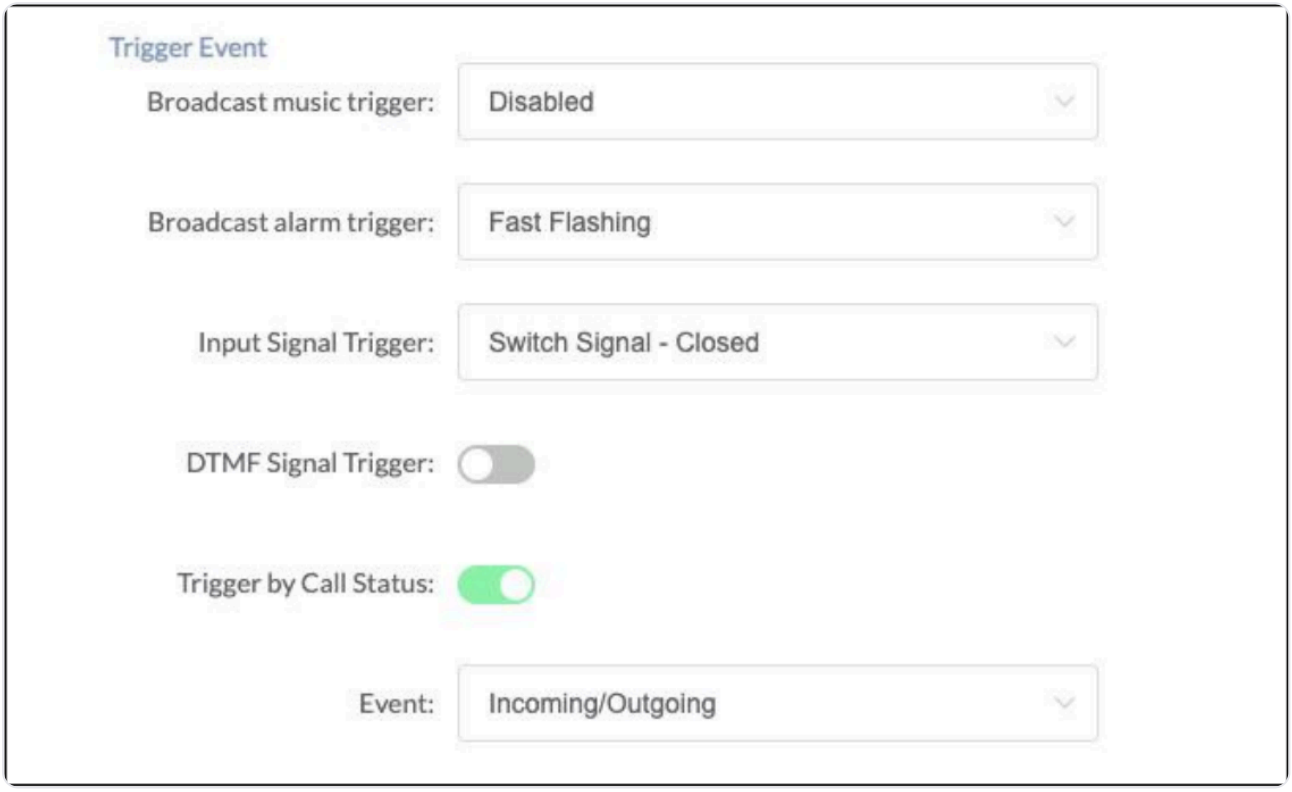
Switch Signal Input Settings



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.

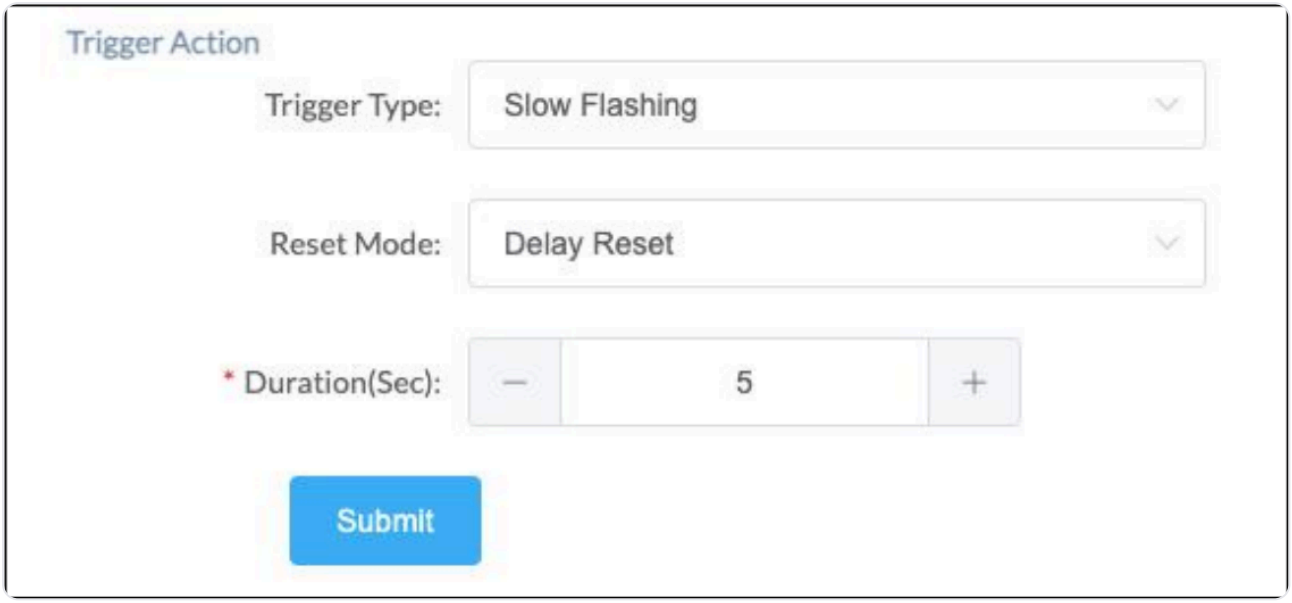
- **HTTP URL:** Configure the API URL address triggered by linkage.
- **Play Audio:** Play the pre-configured audio file.
- **Audio File:** Configure the audio triggered by linkage.
- **Repeat:** Configure the times of audio repetitions triggered by linkage.
- **Stop Audio:** Stop playing the pre-configured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



Output Trigger Event 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.
- **Input Signal Trigger:** Enable/Disable, enable this option if need to use the input signal to trigger.
- **DTMF Signal Trigger:** Enable/Disable, enable this option when need to use DTMF signal to trigger (only RF2833 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.



Output Trigger Action Settings

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

5.6 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

6. System Settings

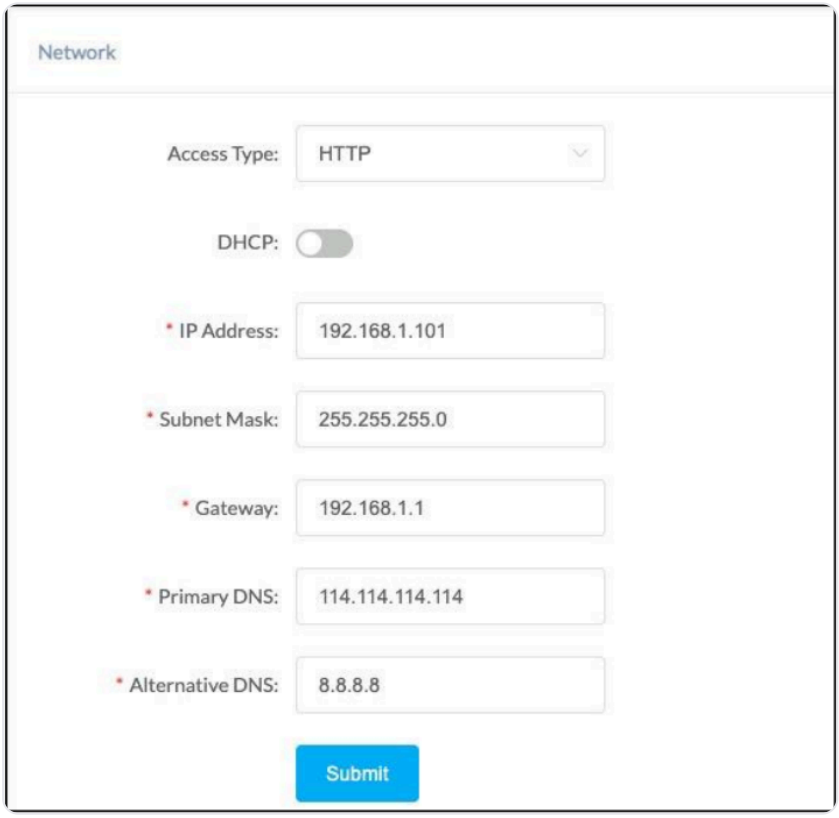
6.1 Network

This speaker uses DHCP to dynamically obtain IP addresses by default.

DHCP TIMING

The DHCP client runs for 165 seconds. If no address is obtained within this time, the device switches to the static IP address 192.168.1.101. Switching the device off and on again restarts the DHCP process from the beginning.

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.



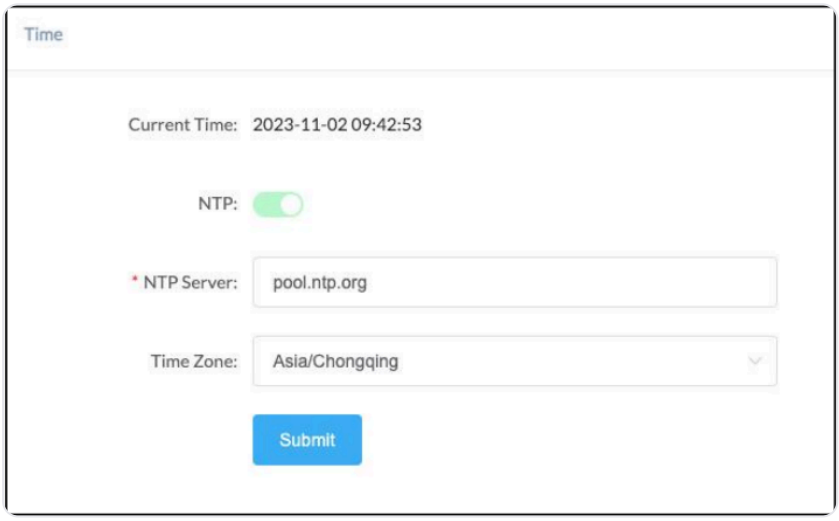
Network Configuration

- **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.

6.2 Time

This speaker 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.

6.3 Prompt Language

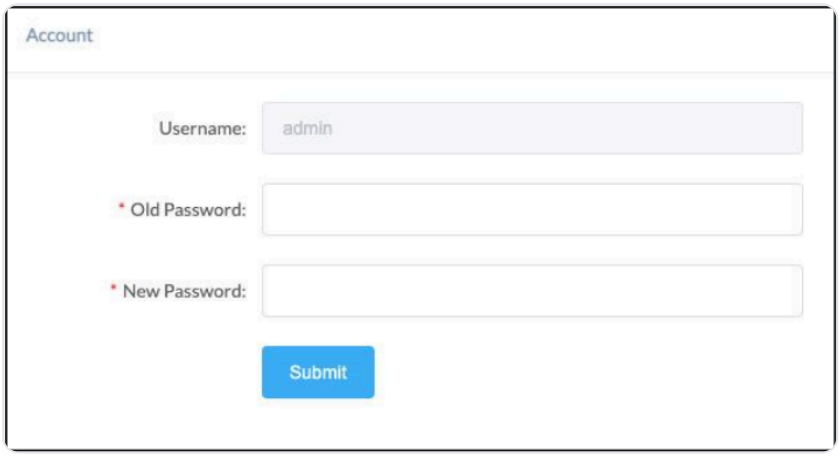
The language of local voice prompts, like IP address announcements. Currently, only Chinese and English are provided. Please go to System → Prompt Language page to set a voice prompt language.



Prompt Language

6.4 Account

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

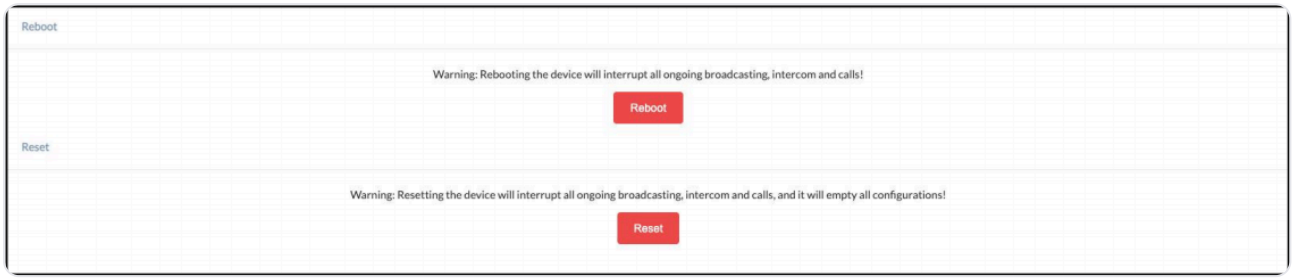


Web Password Settings

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

6.5 Reboot & Reset

This speaker 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.

NOTE

Restoring factory settings will erase all user settings, please operate with caution.



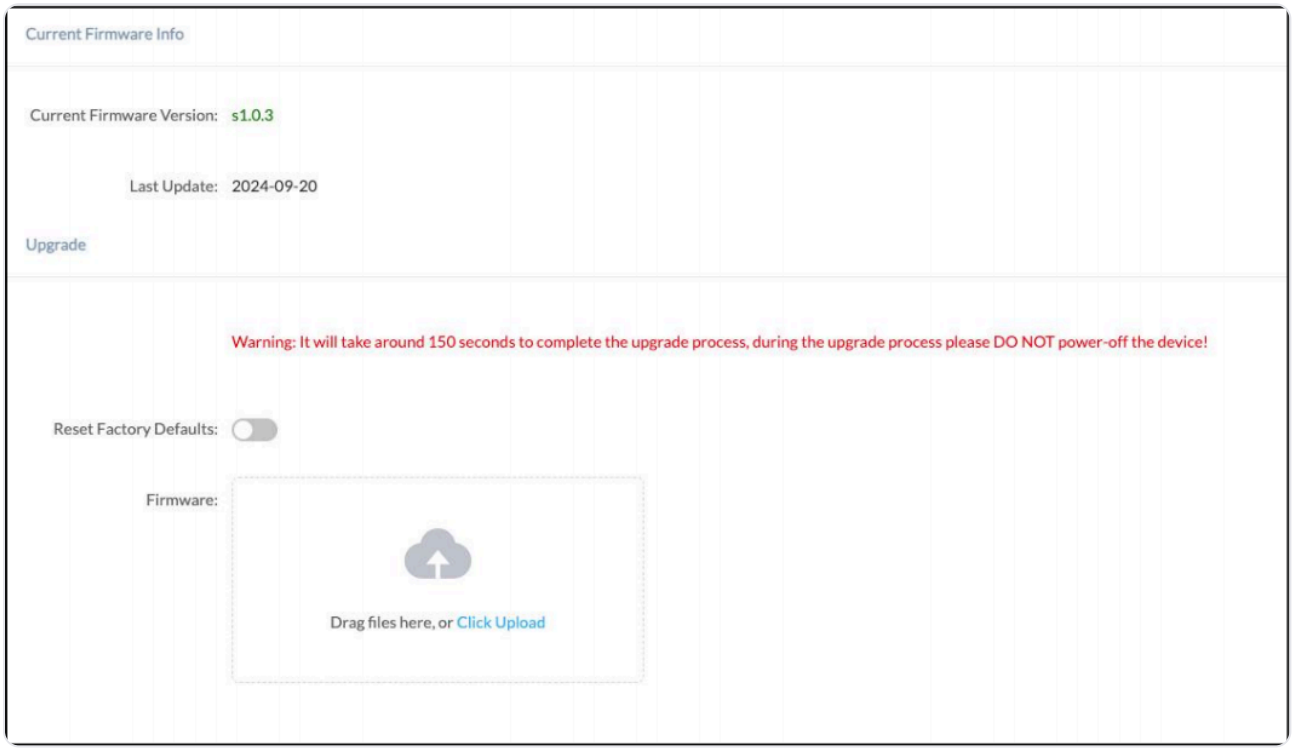
Reboot Schedule

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

7. Maintenance

7.1 Upgrade

To upgrade the device's firmware, please go to System → 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.

7.2 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

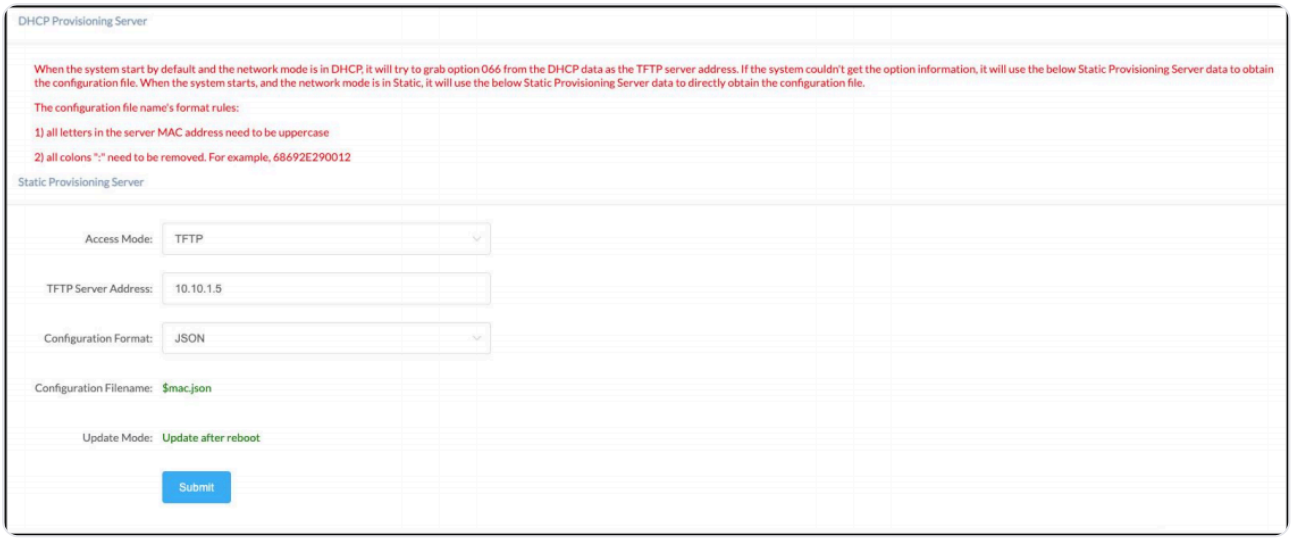
7.3 Auto Provisioning

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

7.4 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

7.5 Ethernet Capture

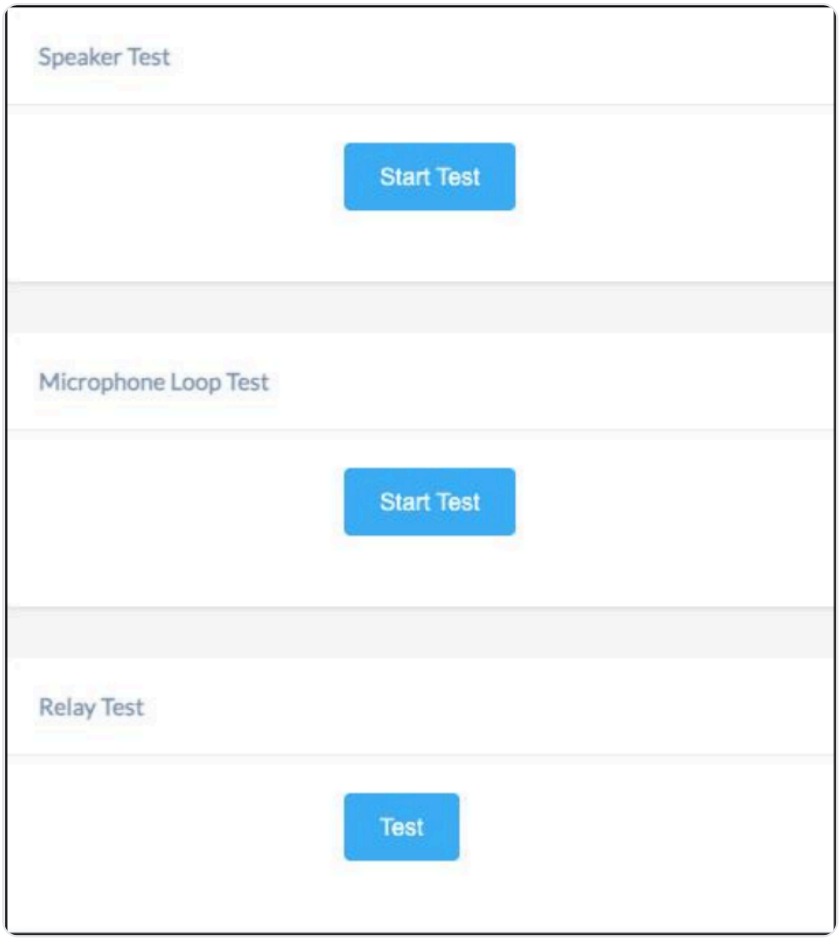
The purpose of the Ethernet capture tool is to capture Ethernet network packets and store them in a standard Wireshark-compatible packet capture '.pcap' file for immediate viewing and data analysis. Please go to Maintenance → Ethernet Capture page to operate.



Ethernet Capture

7.6 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.



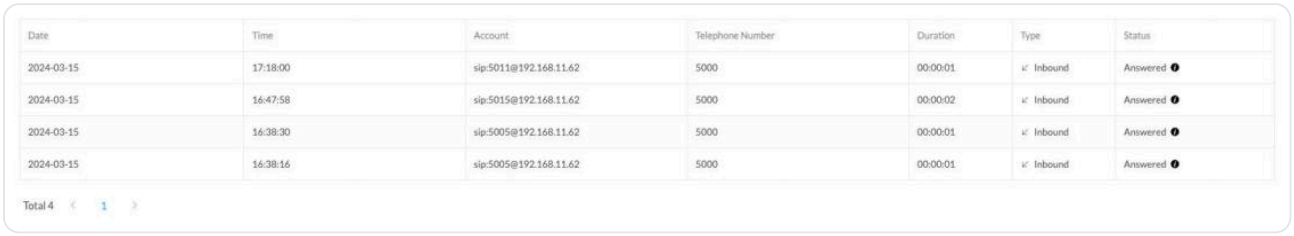
Test Settings

- **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.

8. Reports

8.1 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.



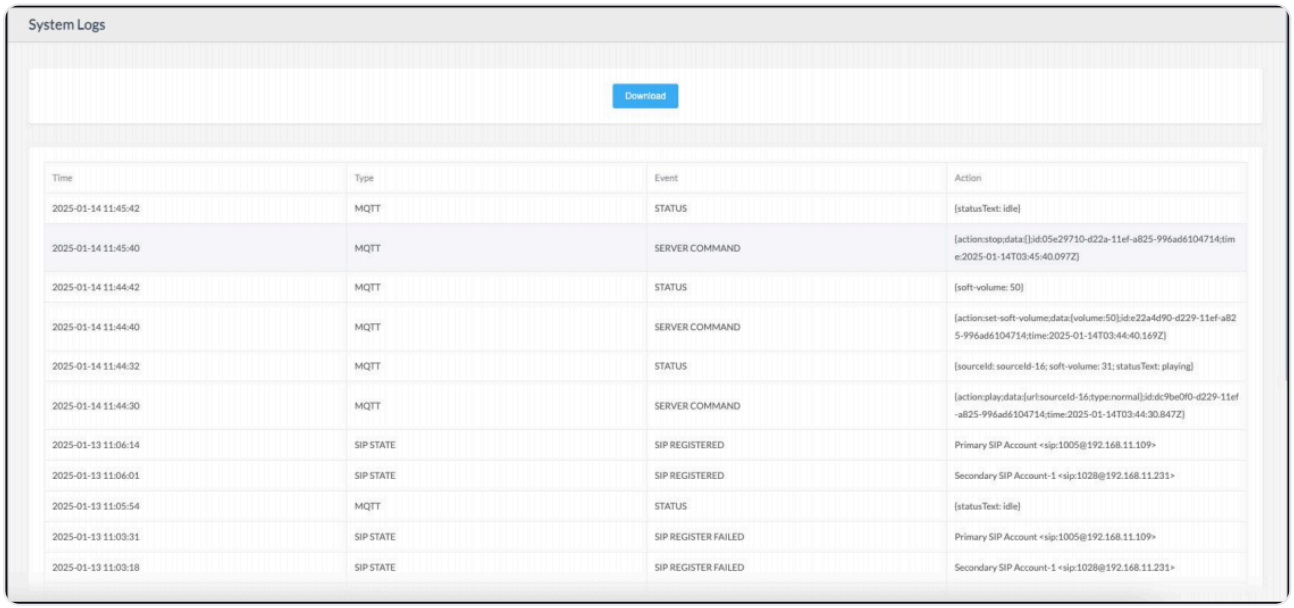
Date	Time	Account	Telephone Number	Duration	Type	Status
2024-03-15	17:18:00	sip:5011@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:47:58	sip:5015@192.168.11.62	5000	00:00:02	Inbound	Answered
2024-03-15	16:38:30	sip:5005@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:38:16	sip:5005@192.168.11.62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

Total 4 1

Call Logs

8.2 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>

9. Add Speaker to Monitoreal

9.1 Speaker preparation

1. Go to the IP Speaker's settings page.
2. Enable SIP communication.
    a. Within the SIP communication settings, enable the following codecs to ensure stable communication between the appliance and the IP Speaker:
        - G.711 PCMU/mulaw
        - G.711 PCMA/alaw
3. Specify the SIP port (When not specified, the Monitoreal IP Speaker uses port 5060 by default).

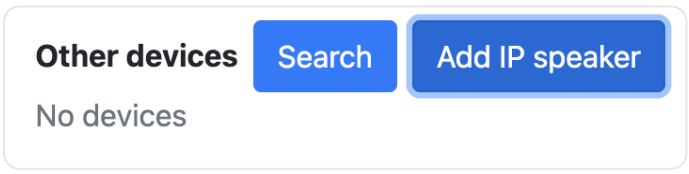
9.2 Add Speaker in Settings → Audio

Once Step 3 has been completed, go to the Web User Interface (WUI) of the Monitoreal Appliance (MR) and proceed to: Settings → Audio.

From there, proceed to add your IP Horn Speaker using one of the following options (points 4&5 below):

4. **Automatic Search.** (Will become available in one of the upcoming firmware versions of Monitoreal Appliance)

    This option will require you to specify the Speaker's login credentials. Optionally, you can specify the IP address of your desired speaker. Otherwise, the MR will find all IP Speakers in your range and add them.



Settings → Audio: Other devices



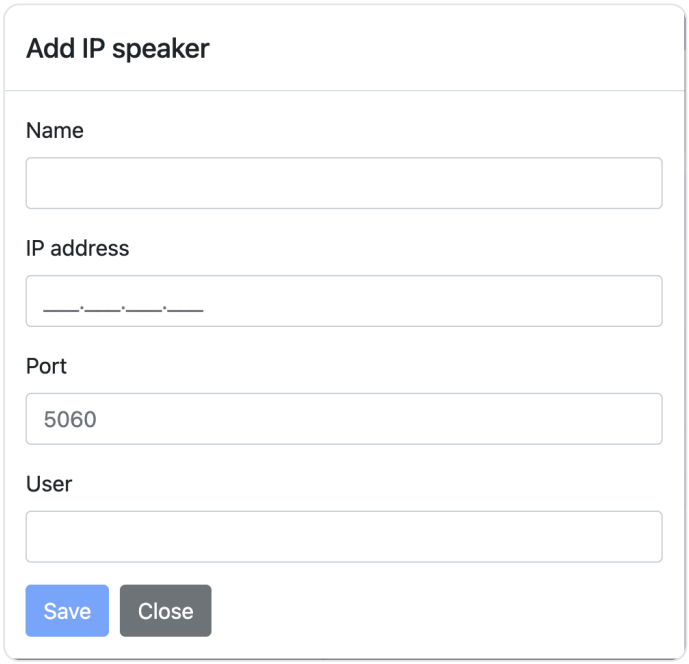
Automatic Search: login credentials

or

5. **Manual addition.**

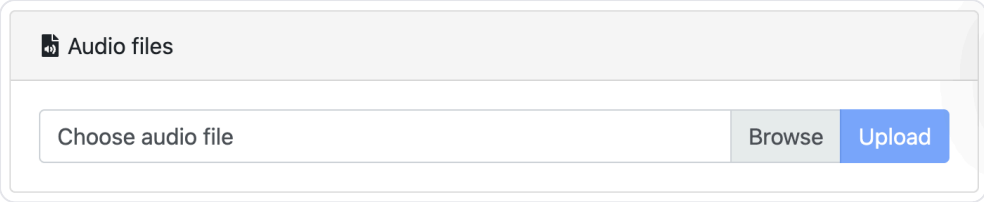
   This option will require you to specify the following:

   - **Name** - custom name;
   - **IP address** - the IP address of your speaker;
   - **SIP port** - the SIP port you have set in the settings of your IP Speaker (refer to point 3);
   - **User** - the User profile name to look for in the SIP server (optional field);
   - **Login credentials** - the Username & Password to be used when logging on to the SIP server (Optional fields. Will become available in one of the upcoming firmware versions of Monitoreal Appliance).



Add IP speaker

6. Once you have successfully added your IP Speaker, you can proceed to upload an audio file to your MR on the same “Audio” page in the relevant field.



Audio files

7. Once both the IP Speaker and the desired audio file have been added to the system, you can start creating audio actions in your rules. (Please refer to the User Manual: Add Audio Actions for creating the detection-related audio actions.)
8. In order to fully utilise the Live Talkdown feature, please use the Monitoreal Global Management system.

IMPORTANT NOTE

In the case where the MR and the Speaker are connected to the same network (e.g. 192.168.50.xx), the device will be able to communicate with the Speaker seamlessly.

In the case of the MR and the Speaker being in different networks, the Speaker’s IP address will need to be made publicly available (outside the connected network) for the MR to be able to communicate with it.

MONITOREAL

Support and contacts

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See what matters.

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