IPS-M1-BW

Network Multi-Info Speaker

User Guide

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1. Preface

1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and manage the IPS-M1-BW Network Multi-Info Speaker. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the IPS-M1-BW Network Multi-Info Speaker.

1.2 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
1.0.0	1.0.0	Updated operating instructions for software version v1.0.0.	2026.4

2. Overview

2.1 Product Overview

The ZYCOO IPS-M1-BW Network Multi-Info Speaker is an IP-based communication endpoint that integrates audio broadcasting and strobe light alerting into a single device. Built on SIP and multicast technologies, it is designed for environments that require reliable voice communication and enhanced alert notification capabilities.

2.2 Product Specifications

IPS-M1-BW Network Multi-Info Speaker Specifications		
Speaker	4.5" Full Range Driver Units	
Sensitivity	91dB / 1W / 1m	
Max Sound Pressure Level	101dB	
Frequency Range	70Hz~20KHz	
Strobe Light Color	Red, Green, Blue	
Amplifier	Built-in Class D Amplifier	

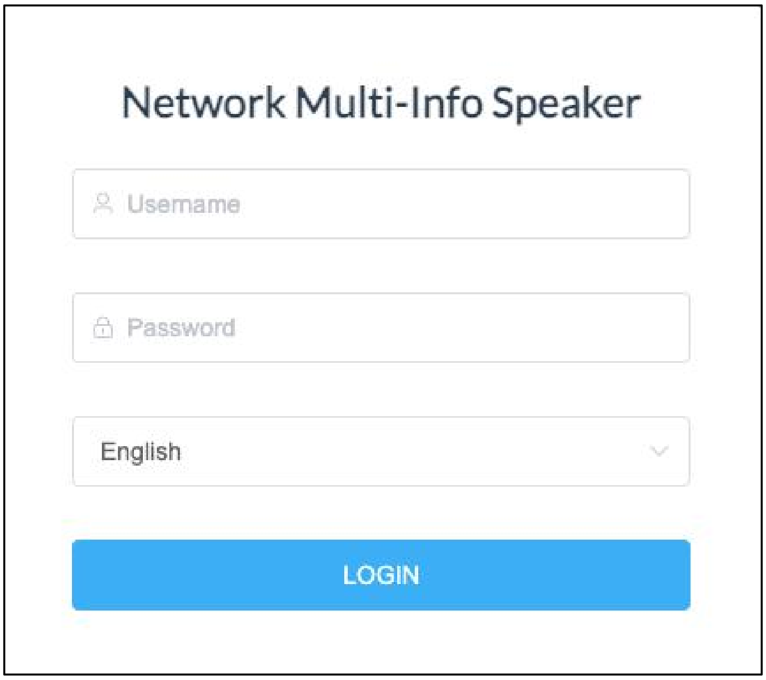
3. Login the Device

3.1 Accessing the Web GUI

IPS-M1-BW 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 wired static IP address: **192.168.32.101**). You can also visit the ZYCOO website to download the SAAD tool or the AudioConfig app to find ZYCOO devices under the LAN and get the IP addresses.

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.

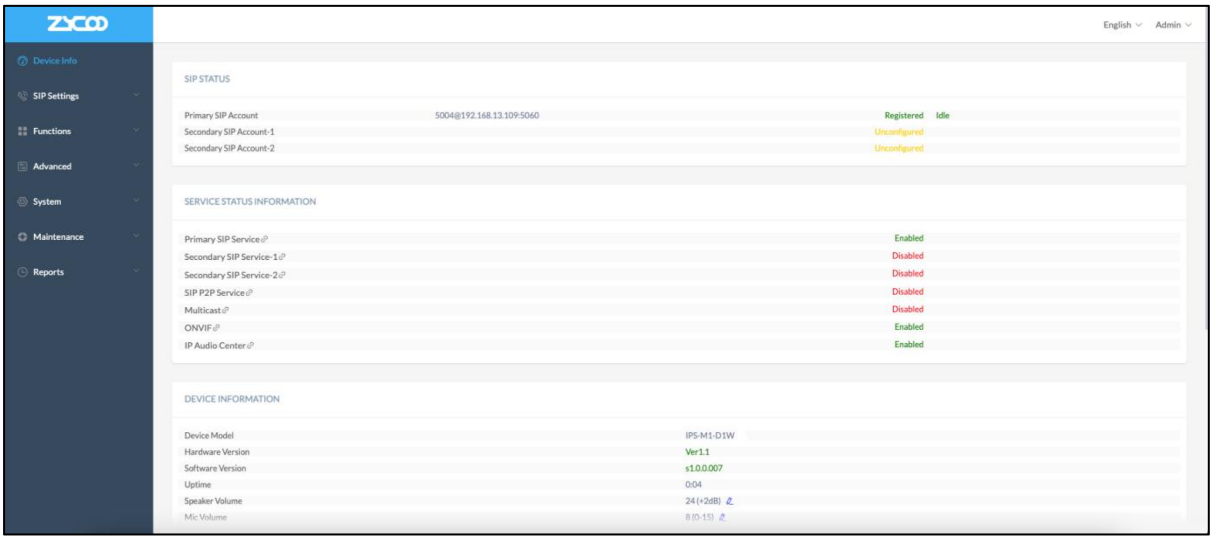


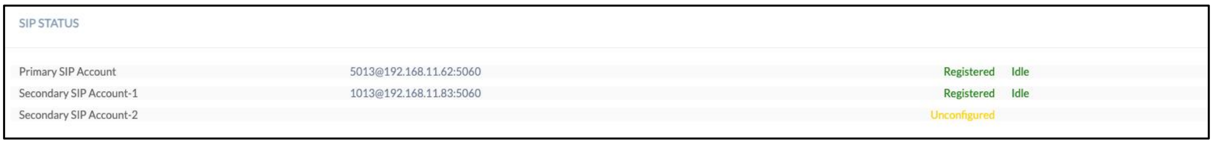
Login Interface

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

3.2 Device Info

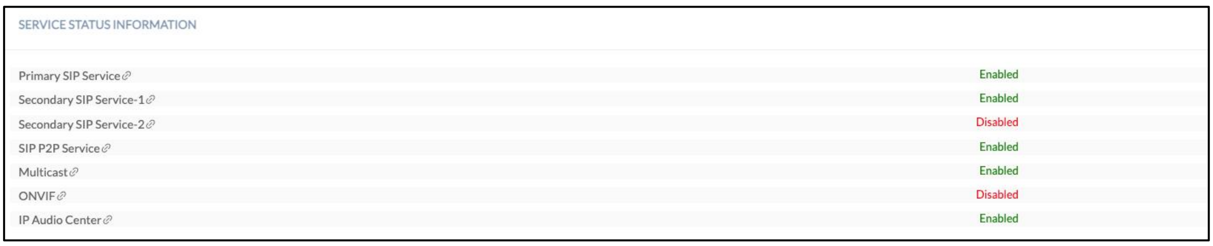
After successful login, you will see the information interface of the device, and you can view the basic information of the device.





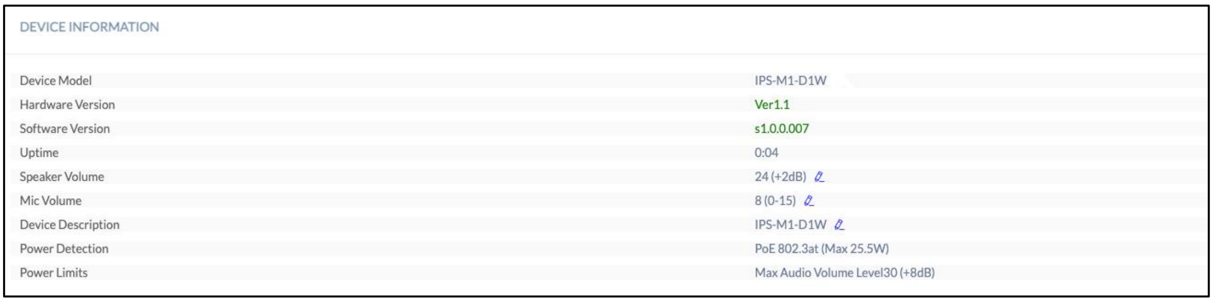
SIP Status

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



Service Status

- **Primary SIP Service:** Displays the status of the primary SIP service.
- **Secondary SIP Service-1:** Displays the status of the secondary SIP service.
- **Secondary SIP Service-2:** Displays the status of the secondary SIP service.
- **SIP P2P Service:** Displays the status of peer-to-peer SIP communication.
- **Multicast:** Displays the multicast service status.
- **ONVIF:** Displays the ONVIF service status.
- **IP Audio Center:** Displays the connection status with ZYCOO IP Audio Center.



DEVICE INFORMATION

Device Model	IPS-M1-D1W
Hardware Version	Ver1.1
Software Version	s1.0.0.007
Uptime	0:04
Speaker Volume	24 (+2dB)
Mic Volume	8 (0-15)
Device Description	IPS-M1-D1W
Power Detection	PoE 802.3at (Max 25.5W)
Power Limits	Max Audio Volume Level30 (+8dB)

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 how long the device has been running.
- **Speaker Volume:** Displays the current volume of the device.
- **Mic Volume:** Displays the current device microphone input volume.
- **Device Description:** Remarks 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.
- **Power Detection:** Displays the current power source (PoE 802.3at or DC input).
- **Power Limits:** Displays the maximum available audio output level under the current power condition (e.g., Max Audio Volume Level 30 (+8 dB)).

NETWORK INFORMATION

Wireless Network Status	Connected (Zycoo-Hall) Signal Strength: -46dBm
Wired Network Status	Connected
Mac Address	[illegible]
Connection Mode	DHCP
LAN IP Address	192.168.13.244
LAN Subnet Mask	255.255.255.0
Wi-Fi IP Address	192.168.12.221
Wi-Fi Subnet Mask	255.255.255.0
Gateway	192.168.12.1 192.168.13.1
Primary DNS	223.6.6.6
Alternative DNS	223.5.5.5

Network Information

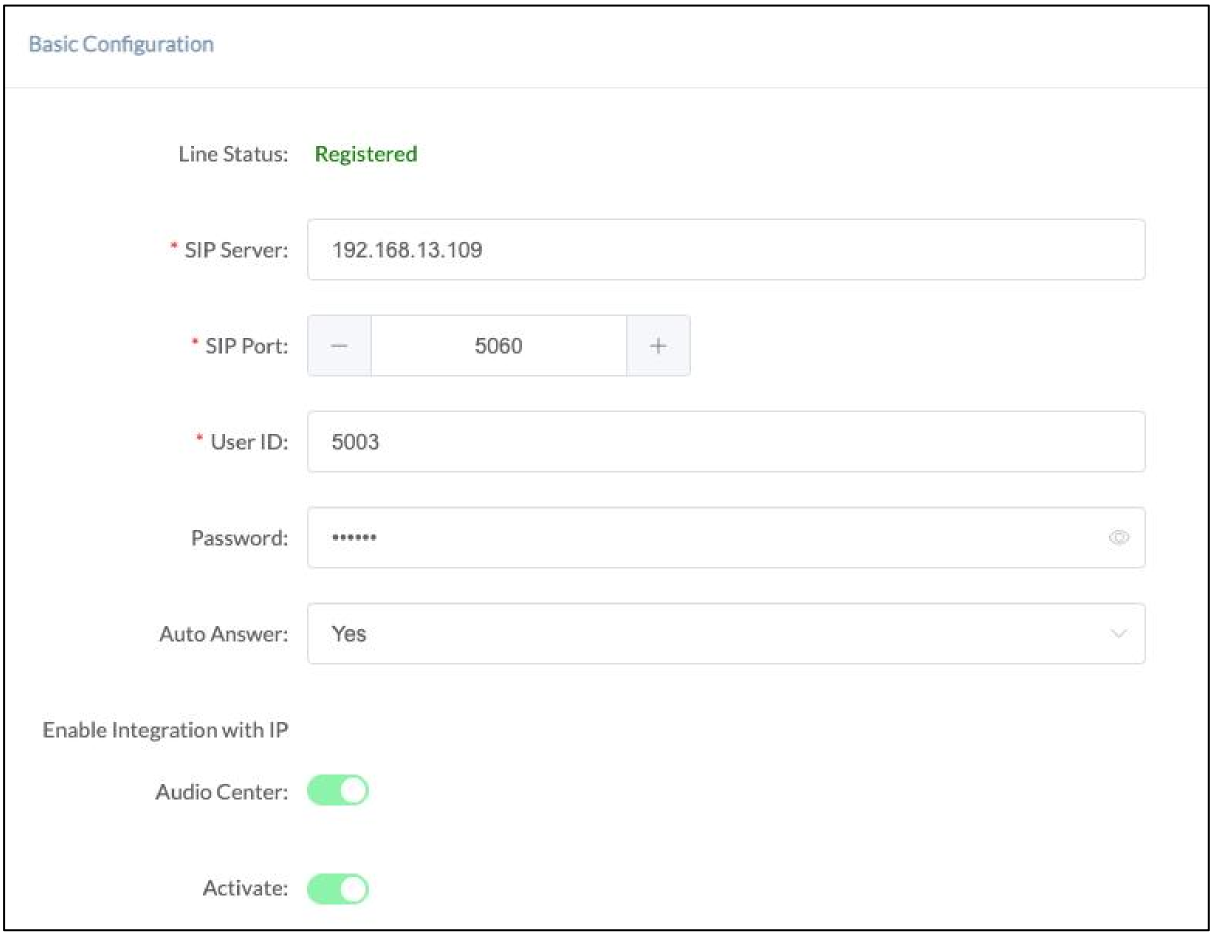
- **Wireless Network Status:** Displays the wireless connection status and related information.
- **Wired Network Status:** Displays the wired connection status and related information.
- **Mac Address:** Displays the MAC address of the current device.
- **Connection Mode:** Displays the network acquisition method of the device, DHCP (dynamic acquisition) or STATIC (static configuration).
- **LAN IP Address:** Displays the current IP address of the device.
- **LAN Subnet Mask:** Displays the current subnet mask of the device.
- **Wi-Fi IP Address:** Displays the IP address of the wireless network interface.
- **Wi-Fi Subnet Mask:** Displays the subnet mask of the wireless network.
- **Gateway:** Displays the gateway address currently used by the device.
- **Primary DNS:** Displays the primary domain name server address used by the device.
- **Alternative DNS:** Displays 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 IP Audio Center’ option.

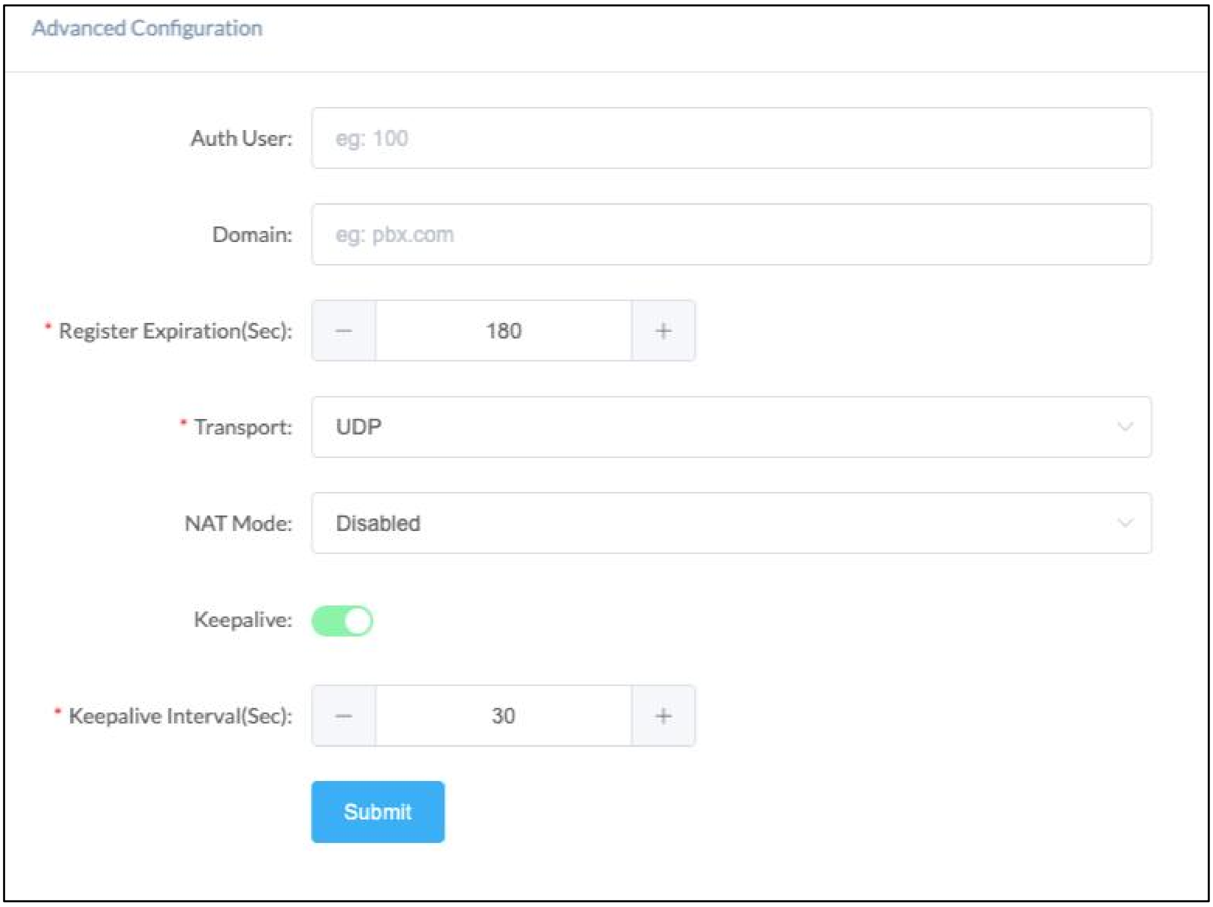
Please go to **SIP Settings --> Primary SIP Account / Secondary SIP Account-1 / Secondary SIP Account-2** page.



SIP Account - Basic Configuration

- **Line Status:** Display the current registration status of the SIP account.
- **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 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

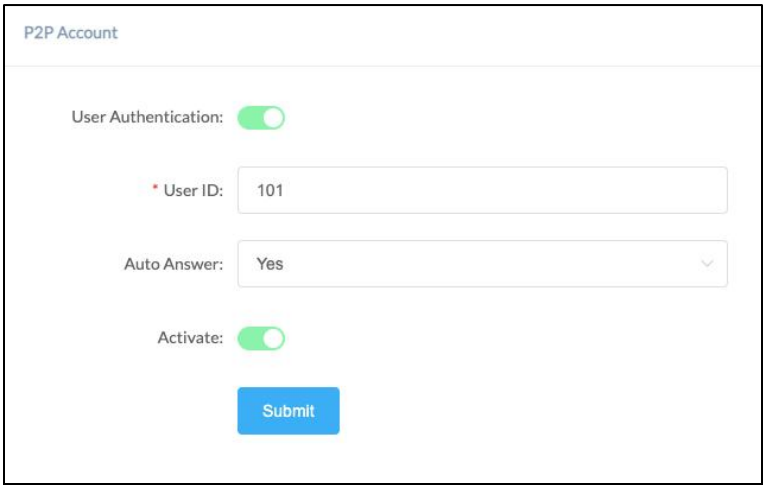
- **Auth User:** Enter the authorized username for the SIP account.

- **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 **Advanced Settings --> I/O Settings**, or use the Outgoing API in **Advanced Settings ---> API Settings** to make a P2P call.



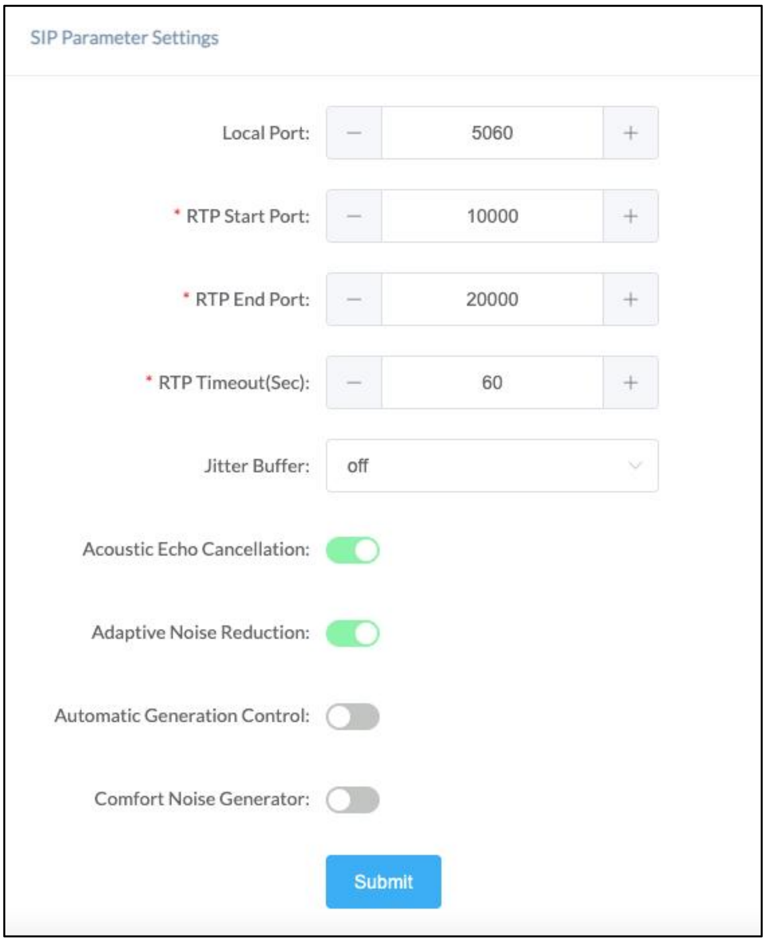
P2P Account

- **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

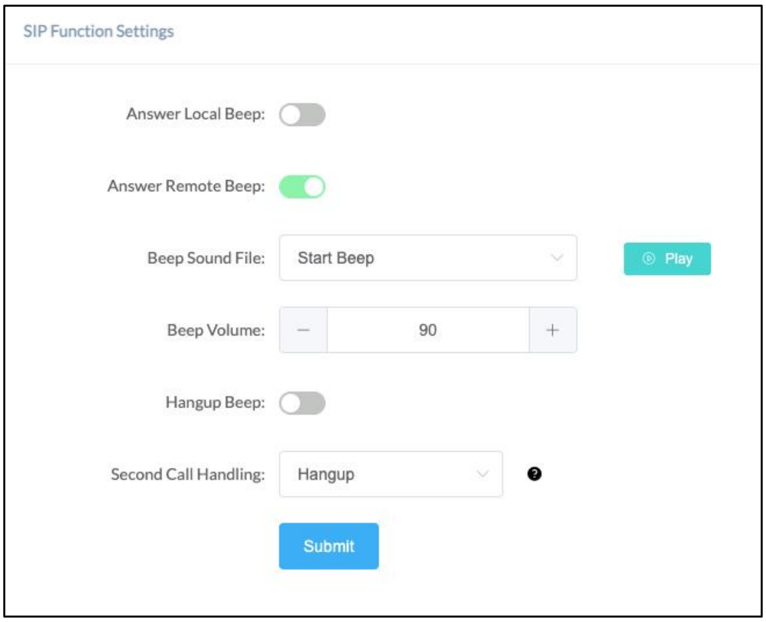


SIP Parameter Settings

- **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 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.

4.3.2 SIP Function Settings



SIP Function Settings

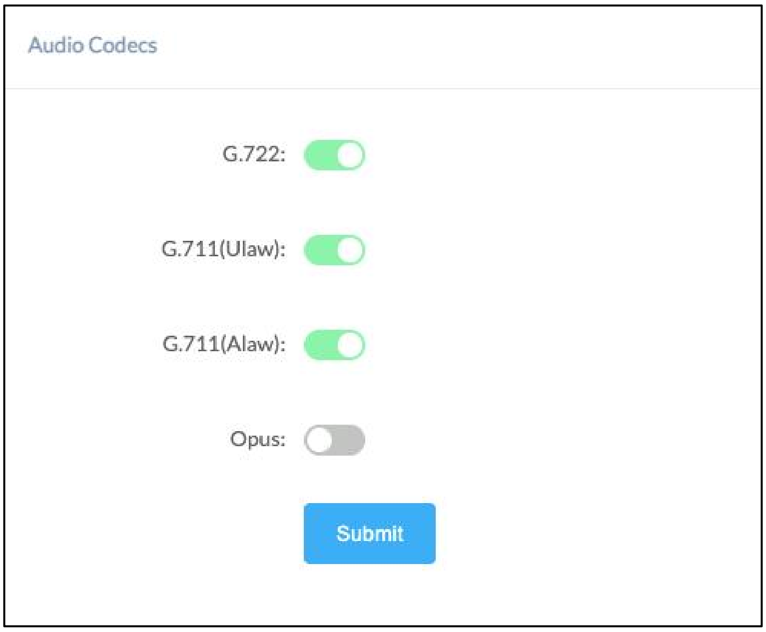
- **Answer Local Beep:** If this setting is enabled, the selected beep sound will be played first on the local device side after the SIP session is answered.
- **Beep Sound File:** Select a specific beep sound file. Click the Play button, you could listen to this audio file.
- **Beep Volume:** Set the 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

IPS-M1-BW supports 4 audio codecs: G.722 (wideband codec), G.711(Ulaw), G.711(Alaw), and Opus.



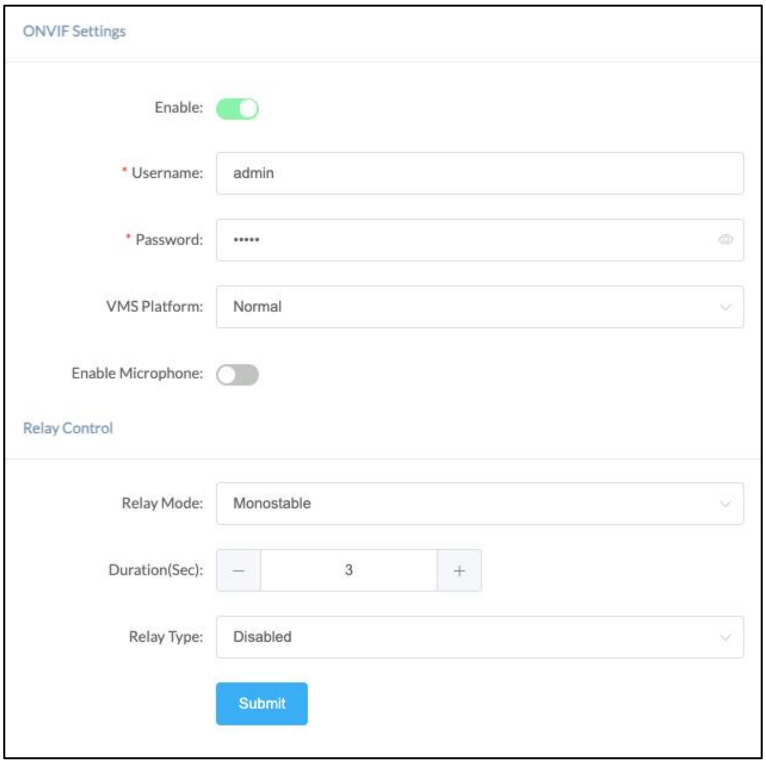
Audio Codecs

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 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

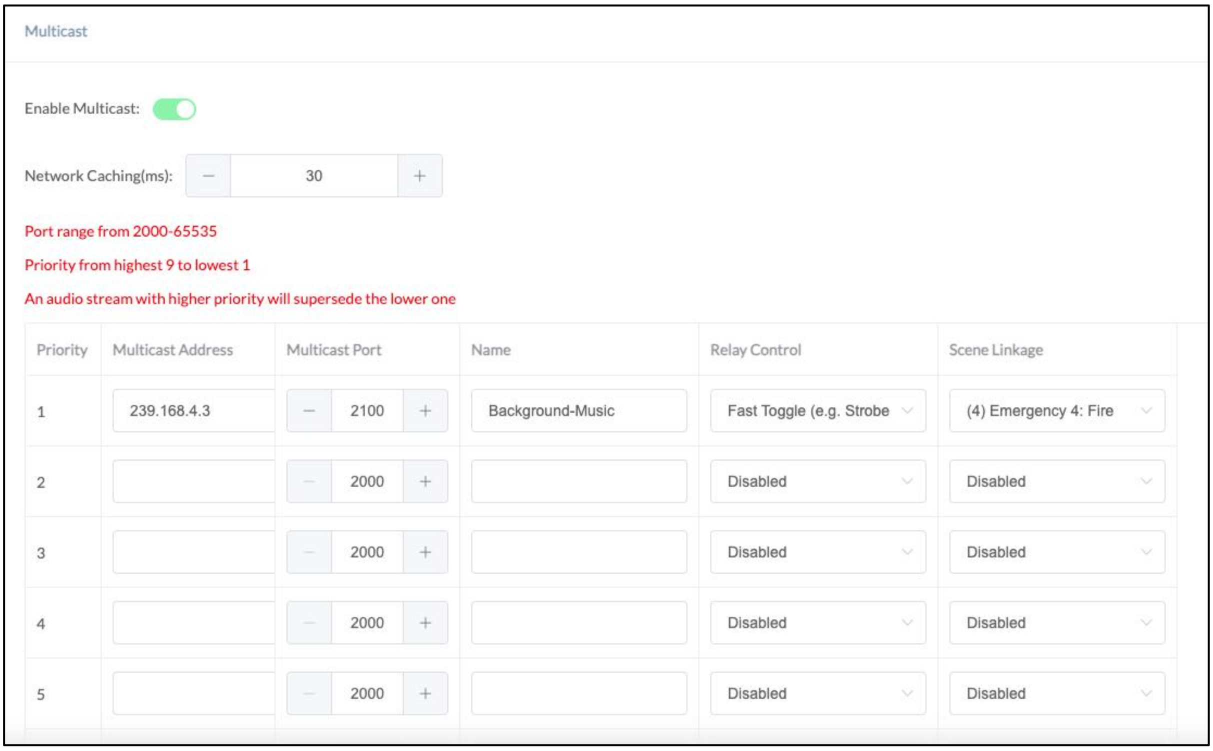
- **Enable:** Enable/Disable ONVIF integration for compatibility with ONVIF-supported VMS platforms.
- **Username:** Enter an account username with matching credentials for adding devices to the VMS platform.
- **Password:** Enter a matching password for the account to add devices to the VMS platform.

- **VMS Platform:** Allows user to select a specific VMS platform from the drop-down list to enable compatibility with different VMS systems.
- **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:** Select the relay behavior: Disabled, Single Toggle (e.g. Door Strike, Audible Visual Alarm), Fast Toggle (e.g. Strobe Fast Flashing), Slow Toggle (e.g. Strobe Slow Flashing).

5.2 Multicast

The Multicast feature allows the device to receive and play audio streams from multicast addresses. It also enables synchronized actions across multiple devices within the same network, such as triggering scene presets for coordinated visual and audio alerts.

The device supports monitoring up to 9 multicast channels with configurable priorities. Higher-priority streams will automatically interrupt lower-priority audio playback. 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:** Select the relay behavior: Disabled, Single Toggle (e.g. Door Strike, Audible Visual Alarm), Fast Toggle (e.g. Strobe Fast Flashing), Slow Toggle (e.g. Strobe Slow Flashing).
- **Scene Linkage:** Select a scene preset to trigger alert actions.

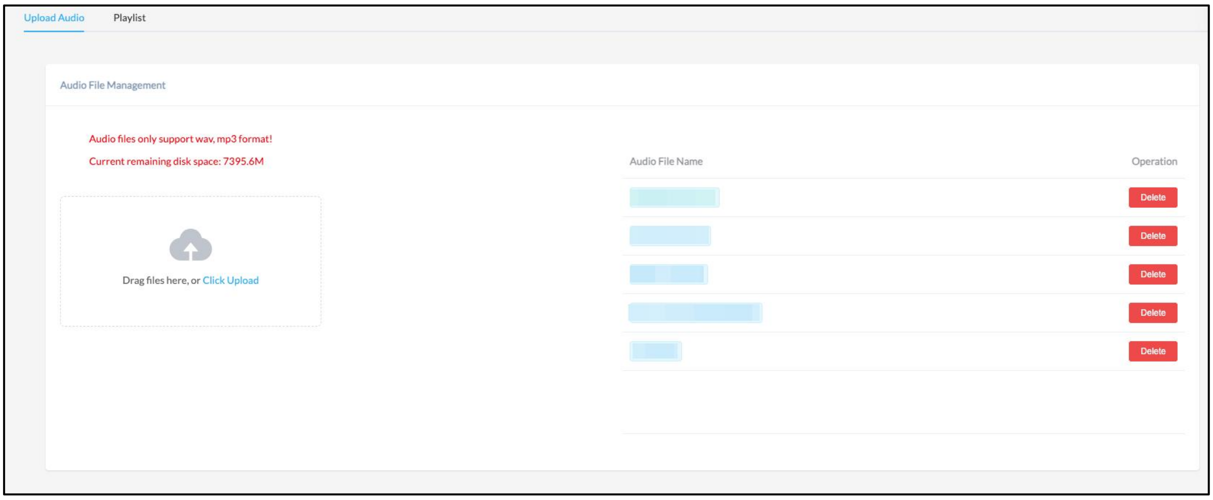
6. Advanced Settings

6.1 Audio

6.1.1 Audio Files

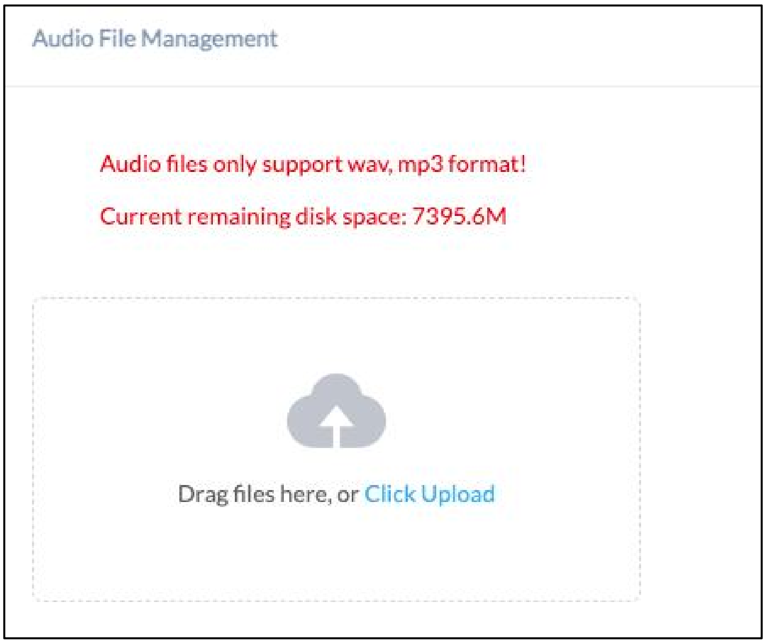
The Audio Files page is used to manage audio files and playlists stored in the device's onboard storage. These audio files can be used across multiple system functions, such as scheduled events. Supported audio formats include WAV and MP3.

Click **Advanced ---> Audio ---> Audio Files** to enter the audio file management page.



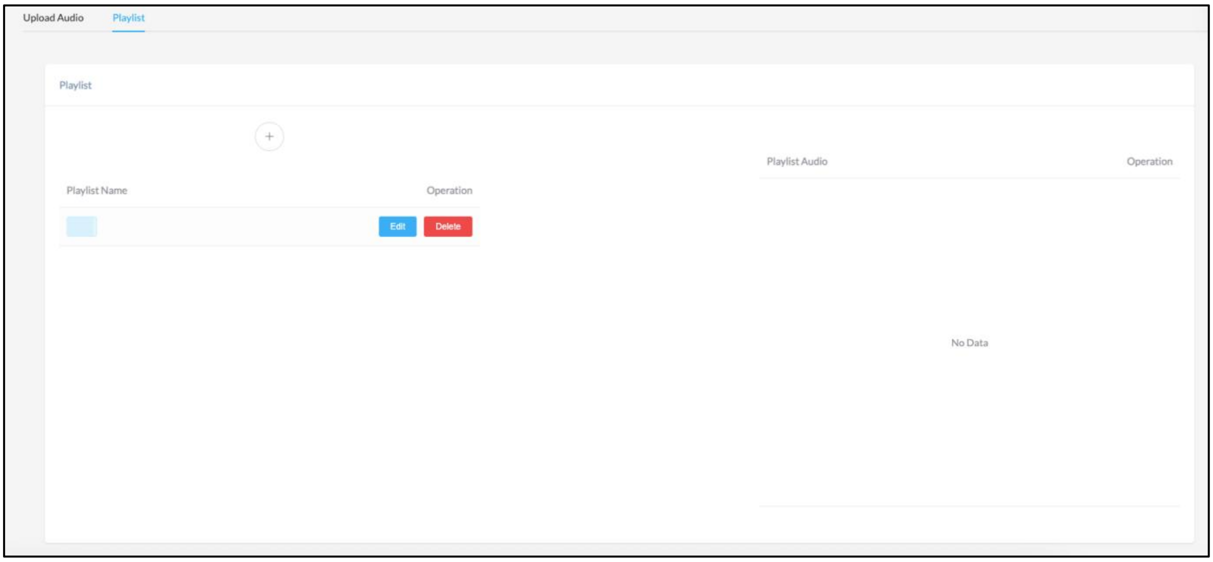
Audio File Management

Click the "Click Upload" button to select the local audio file to be uploaded. Click the "Delete" button to delete the audio file.



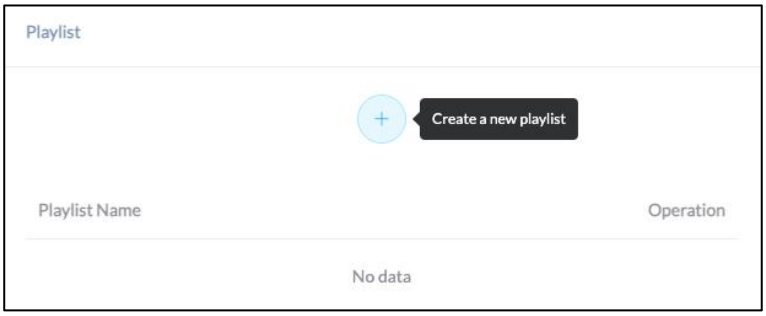
Upload Audio File

Click **Advanced ---> Audio Management** and click Playlist to create, check and edit the playlists.



Playlist

In the playlist interface, click the "+" button to select the required audio files to add, and click "Edit" and "Delete" to edit and delete the created playlist.

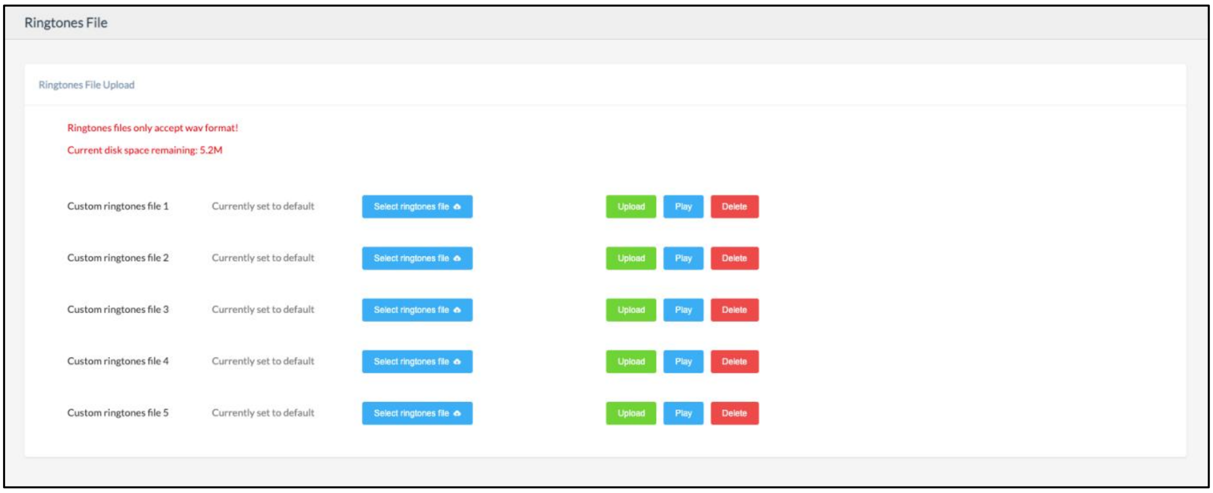


Create a New Playlist

6.1.2 Ringtones File

Ringtones File 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 to delete the audio file.

Please go to **Advanced ---> Audio ---> Ringtones File** to manage the audio files.



Ringtones File

6.1.3 Audio Priority

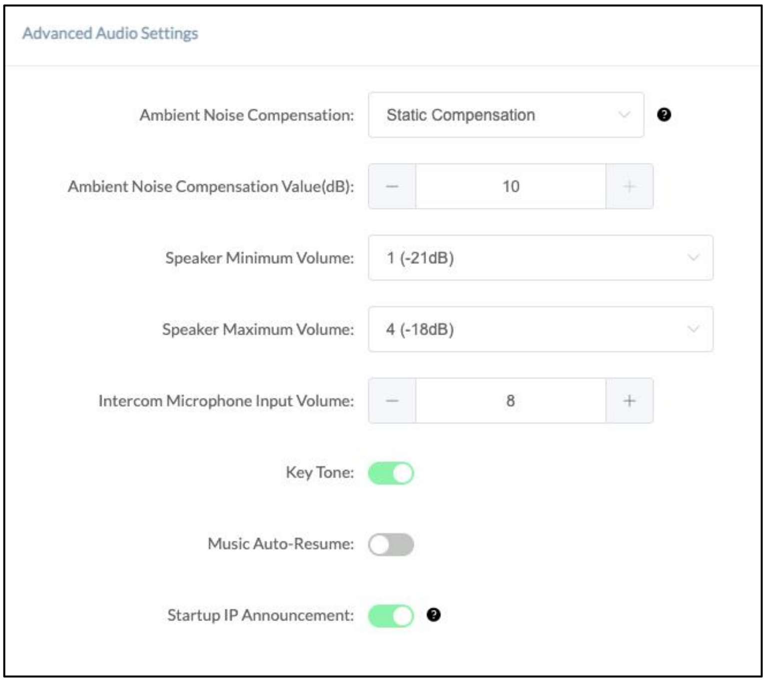
The audio priority can be set according to different applications(such as SIP, ONVIF, MULTICAST, BROADCAST...). Please go to **Advanced ---> Audio ---> Audio Priority** to set the priority.

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

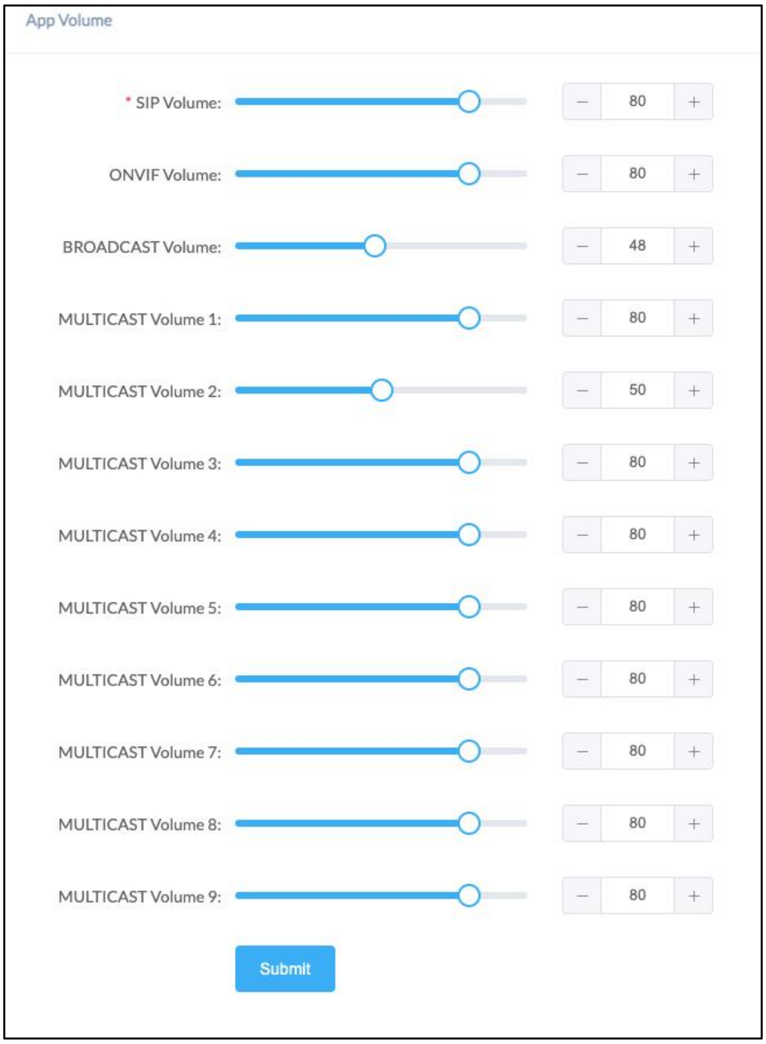
6.1.4 Advanced Audio

To set the volume, please go to **Advanced --> Audio --> Advanced Audio** page to configure. You can set the master volume as well as the individual volume levels for each application (e.g. SIP, multicast, ONVIF...), and you can also configure Remote Volume Control settings on this page.



Advanced Audio Settings

- **Ambient Noise Compensation:** Enable or disable automatic volume adjustment based on ambient noise. Automatically adjusts playback volume based on ambient noise levels to improve the intelligibility of broadcast content. Two modes are supported: Static Compensation (calculates the compensation gain once before playback) and Dynamic Compensation (continuously adjusts the compensation gain during playback). *Note: This feature typically provides good performance for voice broadcast applications. For music playback, the compensation performance may vary depending on the content type. For music with a wide dynamic range or frequent rhythm changes (e.g., rock, electronic music), the compensation gain may fluctuate more noticeably, which can affect the perceived volume adjustment behavior. It is recommended to evaluate the feature in the actual application scenario and select the appropriate compensation mode accordingly.*
- **Ambient Noise Compensation Value(dB):** Set the compensation level applied when ambient noise is detected. A higher value results in greater volume adjustment.
- **Speaker Minimum Volume:** Set the msinimum allowed speaker output level. The range is 0–30, representing a linear gain scale, where 0 = mute, 22 ≈ 0 dB (unity gain), and 30 ≈ +8 dB.
- **Speaker Maximum Volume:** Set the maximum allowed speaker output level. The range is 0–30, representing a linear gain scale, where 0 = mute, 22 ≈ 0 dB (unity gain), and 30 ≈ +8 dB.
- **Speaker Output Volume:** Set the default speaker output volume when Ambient Noise Compensation is disabled. The range is 0–30, representing a linear gain scale, where 0 = mute, 22 ≈ 0 dB (unity gain), and 30 ≈ +8 dB.
- **Intercom Microphone Input Volume:** Set the microphone input volume level. The adjustable range is 0–15.
- **Key Tone:** Enable/Disable the beep sound from the key button.
- **Music Auto-Resume:** When enabled, the device will automatically resume previous music playback tasks after a restart or network reconnection.
- **Startup IP Announcement:** When enabled, the device will announce its IP address once upon startup.



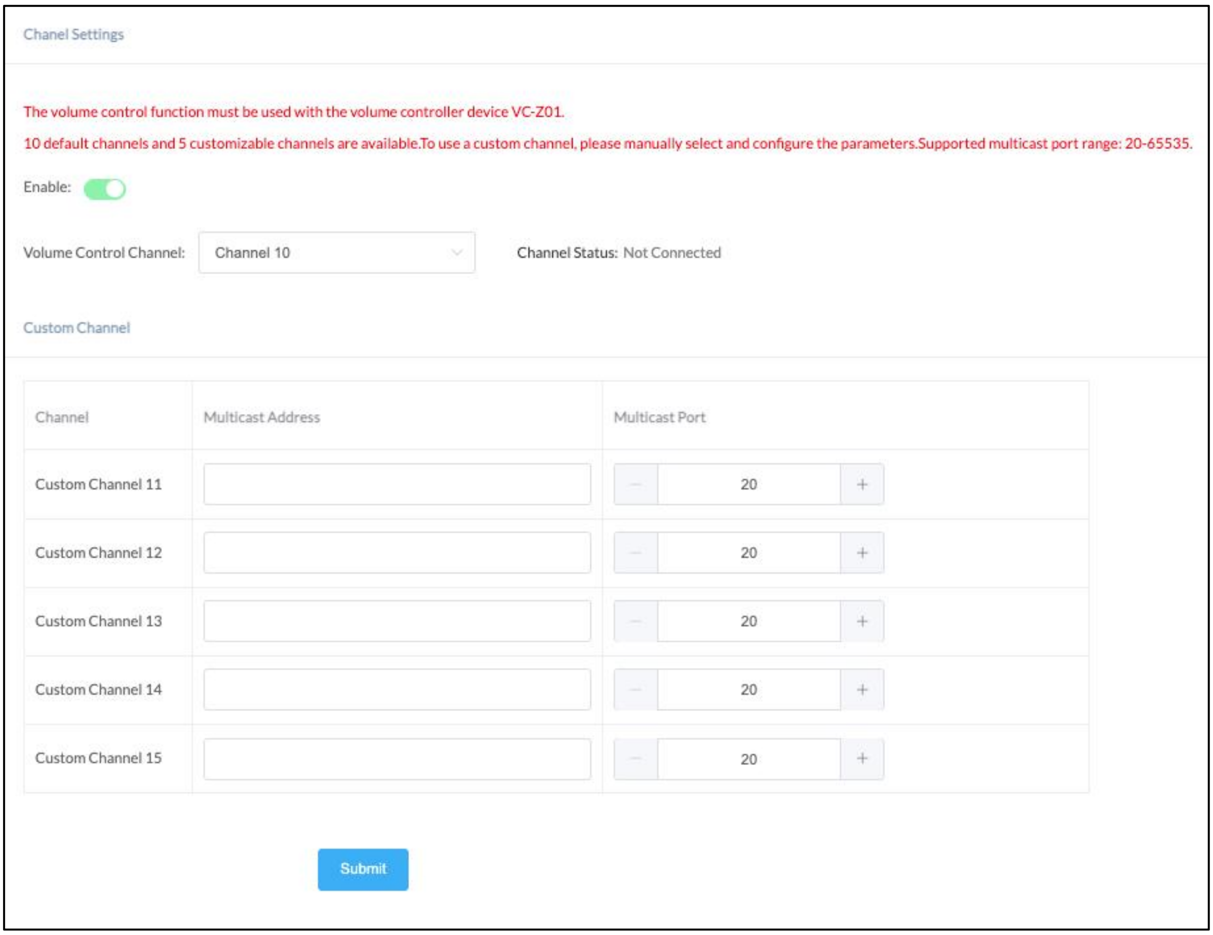
App Volume Settings

- **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.

This page is used to configure remote volume control using the VC-Z01 volume controller. By matching multicast channel settings between the VC-Z01 and the endpoint speaker device, remote volume adjustments can be performed over the network.

Please go to **Advanced** → **Audio --> Advanced Audio** → **Remote Control** page to configure.



Remote Control Settings

- **Enable:** To enable remote volume control, VC-Z01 must be configured together with the endpoint devices. This requires settings to be completed on both the VC-Z01 web GUI and the endpoint web GUI. Once the channels match between VC-Z01 and the endpoints within the same local network segment, the endpoint device will display a Connected status.

- **Volume Control Channel:** Provides 10 default channels (Channel 1 – 10), and 5 customizable channels (Channel 11 – 15).
- **Channel Status:** Displays the connection status between the VC-Z01 and the endpoint device. Connected: The devices are successfully paired and remote control is available. Not Connected: The devices are not matched.
- **Custom Channel:** For Channel 11 – 15, you can define a custom multicast address and port. Multicast Address Range: 224.0.0.0 – 239.255.255.255 Supported Port Range: 20 – 65535.

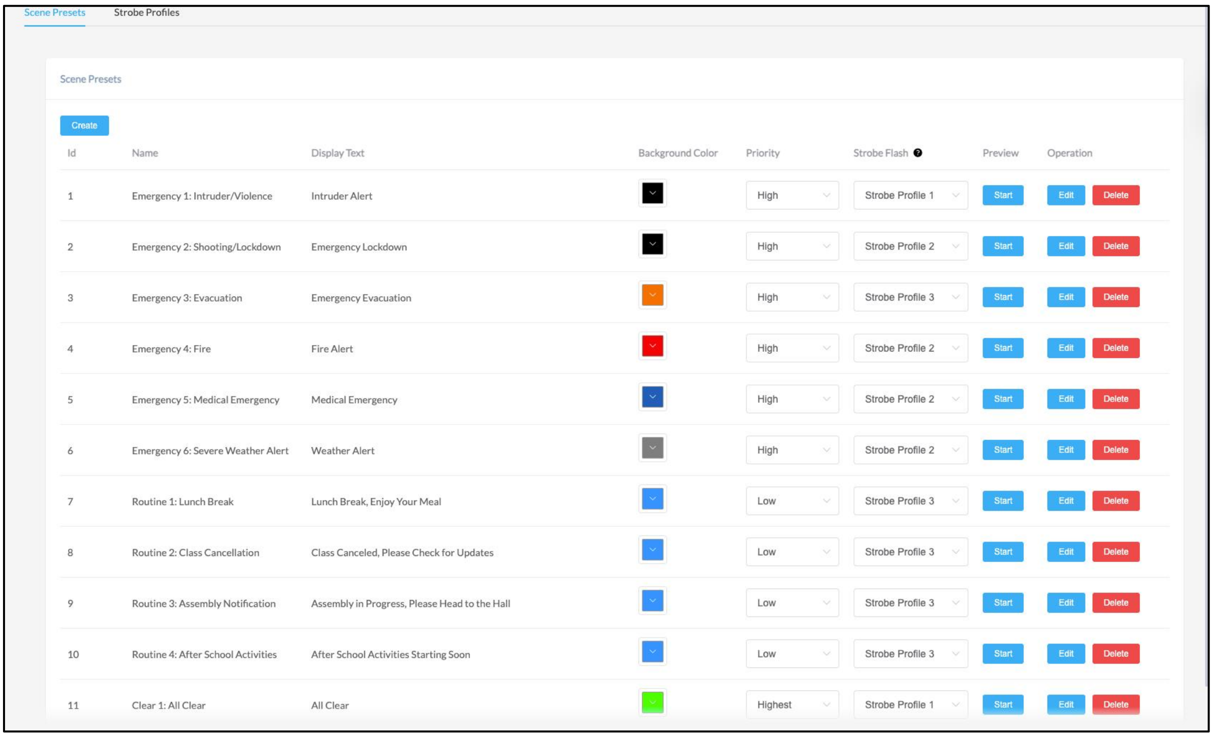
For example, if both the VC-Z01 and the endpoint device are set to Channel 1, and are within the same network segment, clicking Submit on both GUIs will initiate connection. Once connected, the VC-Z01 can remotely adjust the volume of any endpoint device configured on the same channel.

6.2 Scene Presets

Users can create multiple scene presets. When a preset is triggered, the system will optionally activate the strobe lights for visual notification. Based on different scenario requirements, different presets can be used for routine notifications or emergency alerts. Please go to **Advanced --> Scene Presets** to manage the presets.

The system provides several default presets for both emergency and routine scenarios. Users can also customize presets as needed. Scene Presets can be triggered by buttons, incoming calls, DTMF, multicast, event scheduler, or through the background music/alarm function of the ZYCOO IP Audio Dispatch Console.

Users can also create up to three different strobe profiles based on usage preferences. These profiles can then be assigned and used in Scene Presets to provide visual alerts in different scenarios. The system also provides default strobe settings for common use cases. For example, Emergency uses fast red flashing, Routine uses slow blue flashing, and Clear uses steady green light. Strobe light settings can be configured under the Strobe Profiles sub-page of this section.

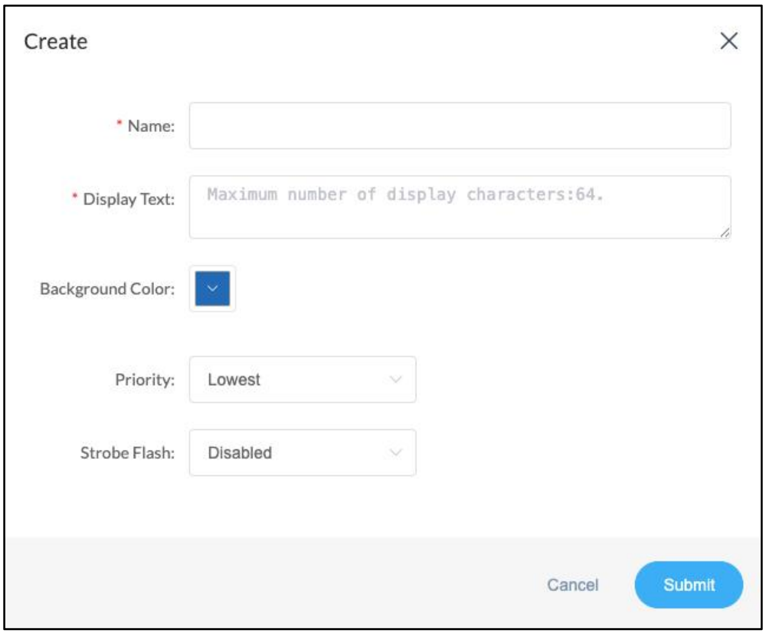


Id	Name	Display Text	Background Color	Priority	Strobe Flash	Preview	Operation
1	Emergency 1: Intruder/Violence	Intruder Alert		High	Strobe Profile 1	Start	Edit Delete
2	Emergency 2: Shooting/Lockdown	Emergency Lockdown		High	Strobe Profile 2	Start	Edit Delete
3	Emergency 3: Evacuation	Emergency Evacuation		High	Strobe Profile 3	Start	Edit Delete
4	Emergency 4: Fire	Fire Alert		High	Strobe Profile 2	Start	Edit Delete
5	Emergency 5: Medical Emergency	Medical Emergency		High	Strobe Profile 2	Start	Edit Delete
6	Emergency 6: Severe Weather Alert	Weather Alert		High	Strobe Profile 2	Start	Edit Delete
7	Routine 1: Lunch Break	Lunch Break, Enjoy Your Meal		Low	Strobe Profile 3	Start	Edit Delete
8	Routine 2: Class Cancellation	Class Canceled, Please Check for Updates		Low	Strobe Profile 3	Start	Edit Delete
9	Routine 3: Assembly Notification	Assembly in Progress, Please Head to the Hall		Low	Strobe Profile 3	Start	Edit Delete
10	Routine 4: After School Activities	After School Activities Starting Soon		Low	Strobe Profile 3	Start	Edit Delete
11	Clear 1: All Clear	All Clear		Highest	Strobe Profile 1	Start	Edit Delete

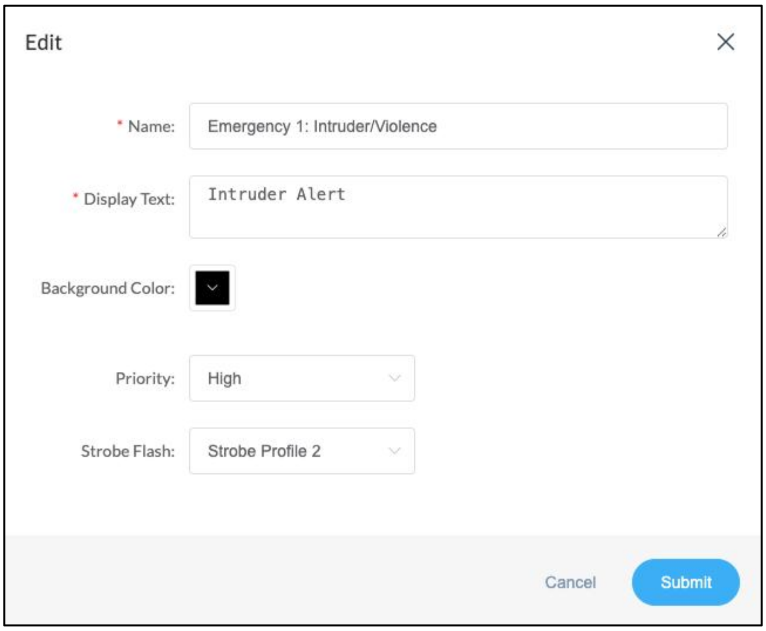
Scene Presets Settings

- **Create:** Create a new scene preset.
- **ID:** The unique identifier of the preset.
- **Name:** The name of the preset.
- **Priority:** The priority level of the preset. Five priority levels are supported. A higher-priority preset will override any lower-priority. For presets with the same priority, the most recently triggered preset will take effect. In the default presets, Emergency is set as high priority, Routine is set as low priority, and Clear is set as the highest priority.
- **Strobe Flash:** Select the strobe profile to be used when this preset is triggered. Available options include Profile 1, Profile 2, Profile 3, or disabled. When a profile is selected, the corresponding strobe behavior defined in the Strobe Profiles section will be applied.

- **Preview:** Preview only simulates display and strobe behavior locally, including display content and strobe effect. Click Start to run the preview. Click Stop to end the preview.
- **Operation:** Edit or delete the preset.

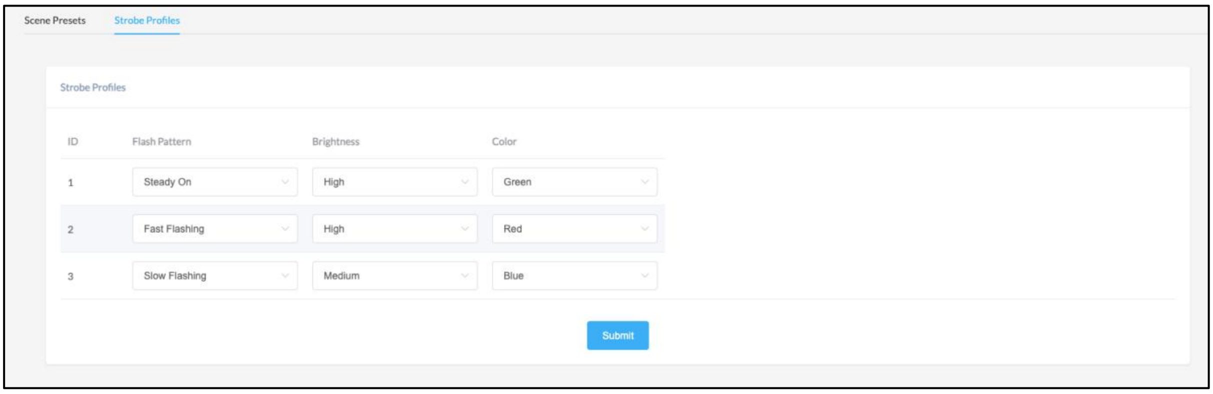


Create Presets



Edit Presets

- **Name:** Set the name of the preset.
- **Priority:** Set the priority level of the preset. Five priority levels are supported. A higher-priority preset will override any lower-priority content currently being displayed. For presets with the same priority, the most recently triggered preset will take effect.
- **Strobe Flash:** Select the strobe profile to be used when this preset is triggered. Available options include Profile 1, Profile 2, Profile 3, or disabled. When a profile is selected, the corresponding strobe behavior defined in the Strobe Profiles section will be applied.



Strobe Profiles

- **ID:** The unique identifier of the strobe profile.
- **Flash Pattern:** Defines the flashing behavior of the strobe light. The following options are supported: Steady On: Constant light without flashing. Fast Flashing: High-frequency flashing. Slow Flashing: Low-frequency flashing.
- **Brightness:** Sets the light intensity level. The following levels are available: High, Medium, Low.
- **Color:** Defines the strobe light color. RGB color options are supported.

6.3 Incoming Call Settings

Users can set up a strict number-matching list to trigger preset.

Please go to **Advanced ---> Incoming Call Settings** to configure.

Number Matching Display Settings

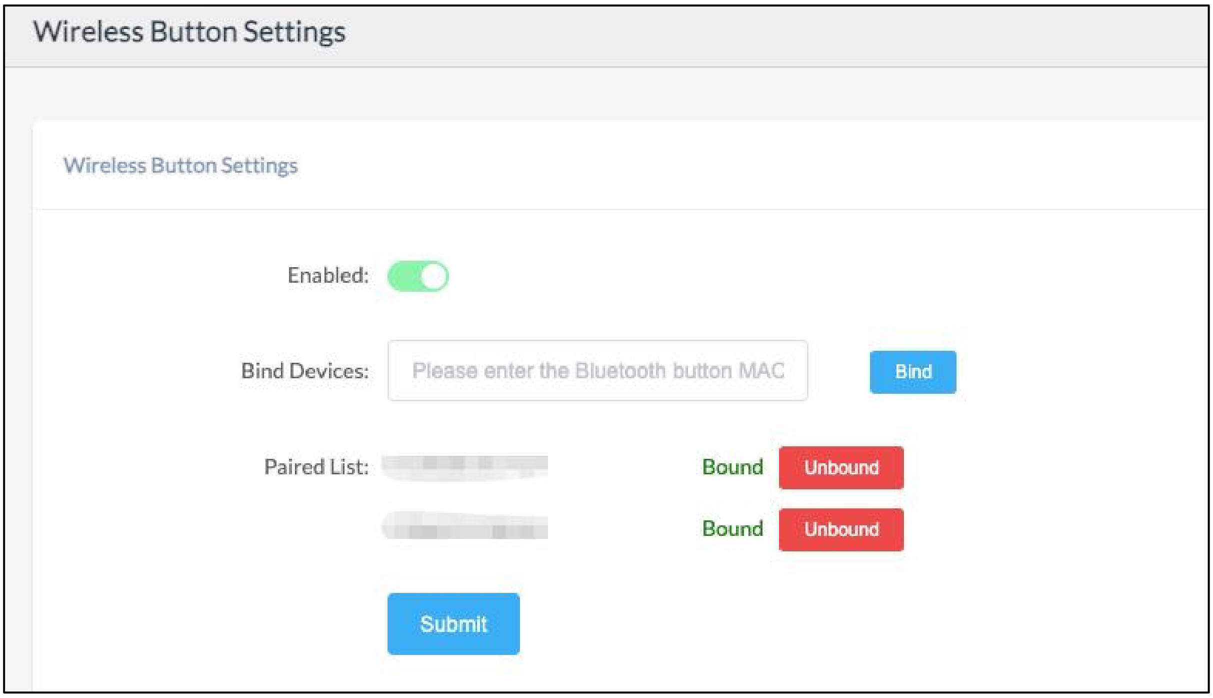
- **Matched Number:** Configure the incoming call number to trigger specific display content through precise matching.
- **Trigger Preset Display:** Select the preset to be displayed when precise matching is activated.

6.4 Wireless Button Settings

This page allows users to pair the device with BLE-based wearable smart buttons and configure trigger actions. Once paired, the wearable button can trigger predefined local actions on the device, such as initiating an outgoing call, activating scene linkage, or controlling relay outputs (e.g., door locks). In emergency situations, users can press the button within Bluetooth range to remotely trigger alerts and predefined responses.

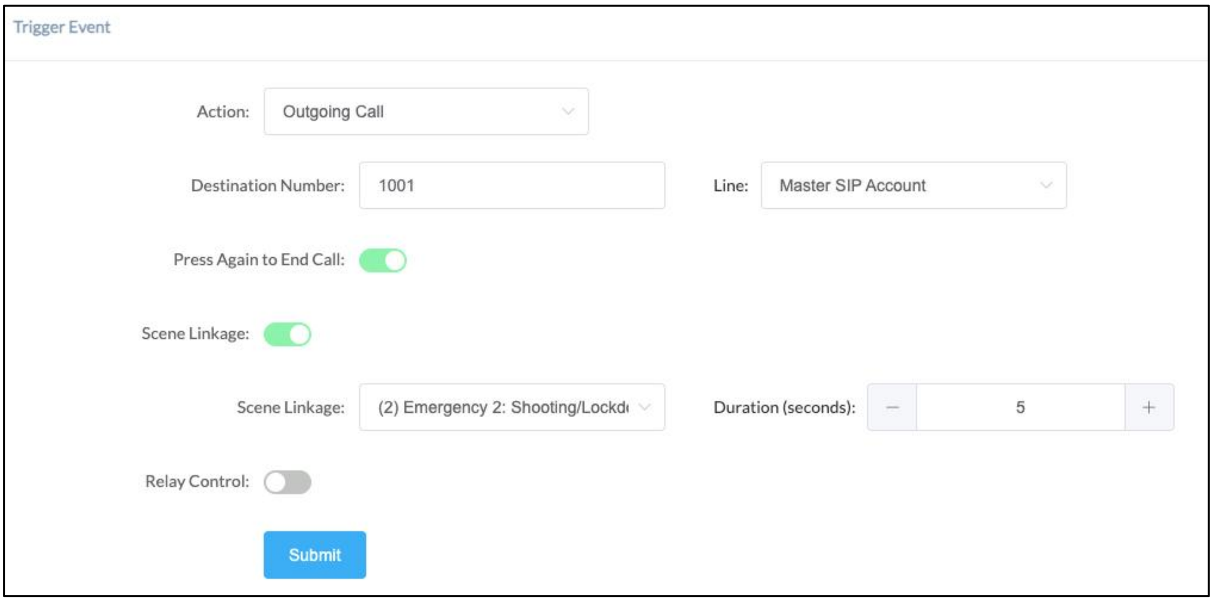
When multiple IPS-M1-BW devices are paired with the same wearable button, all devices within Bluetooth coverage range that successfully receive the BLE trigger signal will respond simultaneously and execute the configured actions.

To avoid unintended repeated triggering, the wearable button may apply an internal debounce and anti-retrigger mechanism. Each single press is treated as a single valid event, and repeated triggers within a short time window will be ignored. Please go to **Advanced ---> Wireless Button** to configure.



Bluetooth Settings

- **Enabled:** Enable or disable the Bluetooth function.
- **Bind Devices:** Enter the MAC address of the wearable smart button to bind with the device.
- **Paired List:** Displays the list of currently paired devices. Users can view or unbound paired devices.



Trigger Event Settings

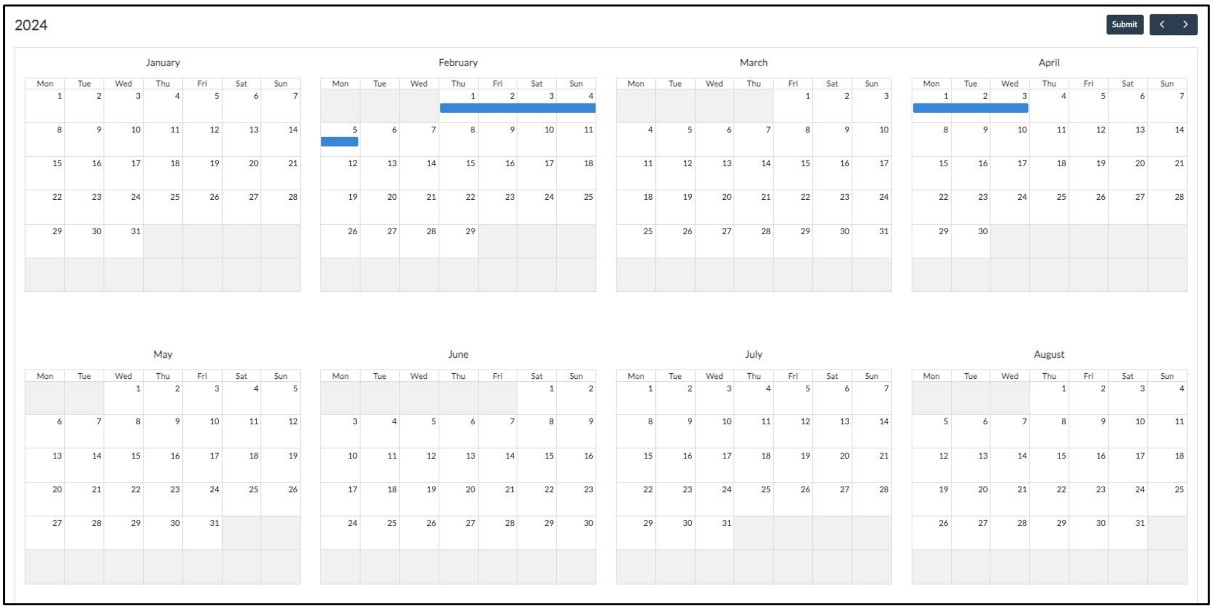
- **Action:** Select the action to be triggered when the button is pressed. Available options include: Outgoing Call, HTTP Request, Play Audio
- **Outgoing Call:** Configure the destination number or SIP account for the outgoing call.
- **HTTP Request:** Configure the target URL and parameters for the HTTP request.
- **Play Audio:** Select the audio source (ringtone, audio file, or playlist) for playback.
- **Scene Linkage:** Enable and select a predefined scene preset to trigger visual and strobe actions.
- **Relay Control:** Trigger relay output to control external devices, such as door locks or alarm systems.

6.5 Event Scheduler

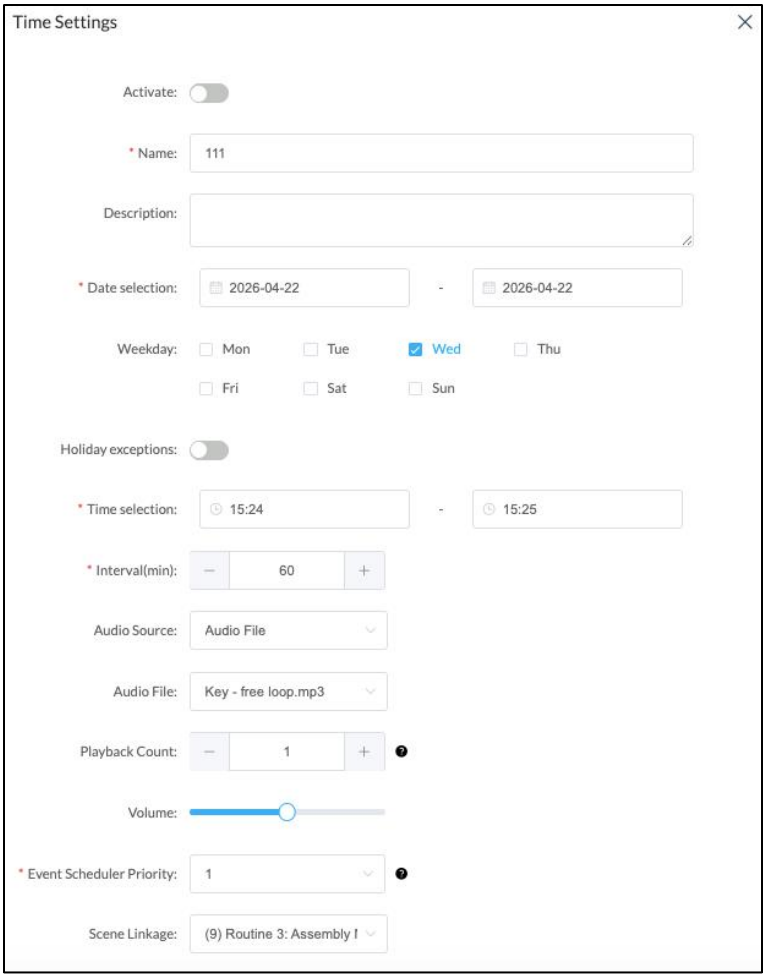
The event scheduler can edit up to 30 time plans, you can click the corresponding option to edit or delete them. Before you edit/create an event scheduler. The schedules in the holiday setting will not be executed.

Please go to **Advanced ---> Event Scheduler** to manage the events.

Event Scheduler



Holidays Setting



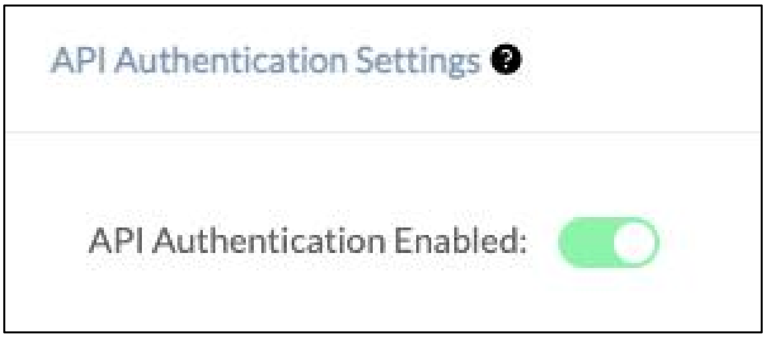
Time Settings

- **Activate:** Activate/Deactivate the schedule.
- **Name:** Set the name of the schedule.
- **Description:** Comment information for the time schedule.
- **Date Selection:** Set the date range for the time schedule.
- **Weekday:** Set the execution week day in the date range.
- **Holiday Exceptions:** Enable the holiday feature or not.
- **Time Selection:** Set the specific time period for executing the action.
- **Interval(min):** Set the interval time for performing actions.
- **Audio File:** Select an audio file to play.
- **Play Times:** Set the number of playbacks. When set to "0", it is loop playback.
- **Volume:** Set the playback volume for the scheduler.
- **Event Scheduler Priority:** Assign priority to the scheduler system; higher-priority schedules will always execute first.
- **Scene Linkage:** Enable and select a predefined scene preset to trigger strobe actions.

6.6 API Settings

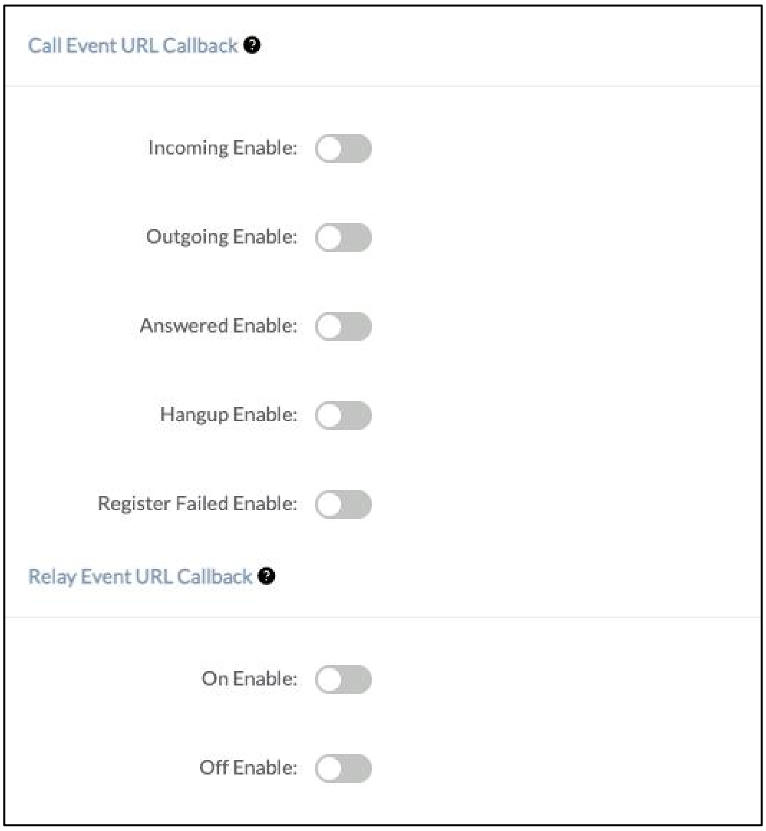
This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the changing status of the call and/or relay.

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:

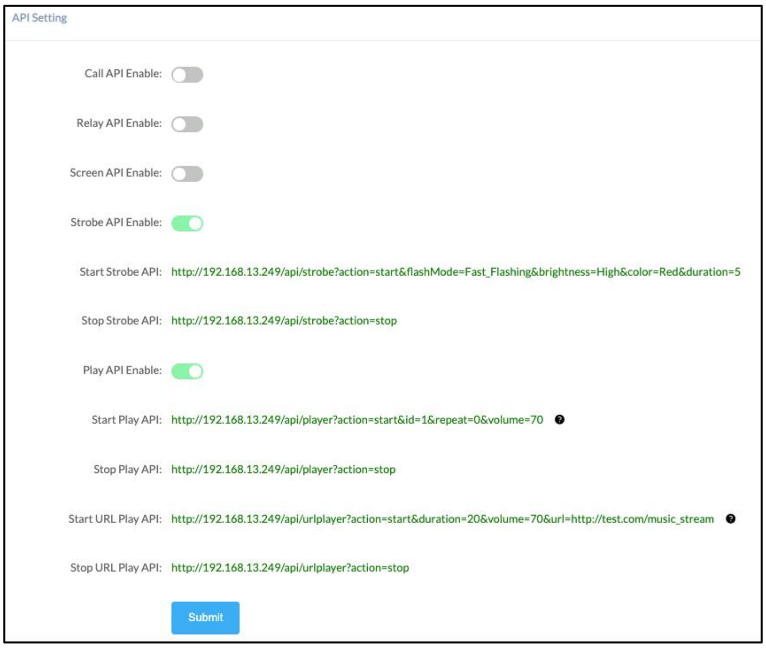
${ip}:	The current IP address of the device
${mac}:	The current MAC address of the device
${ua}:	The account of the current call
${number}:	The number of the current call

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



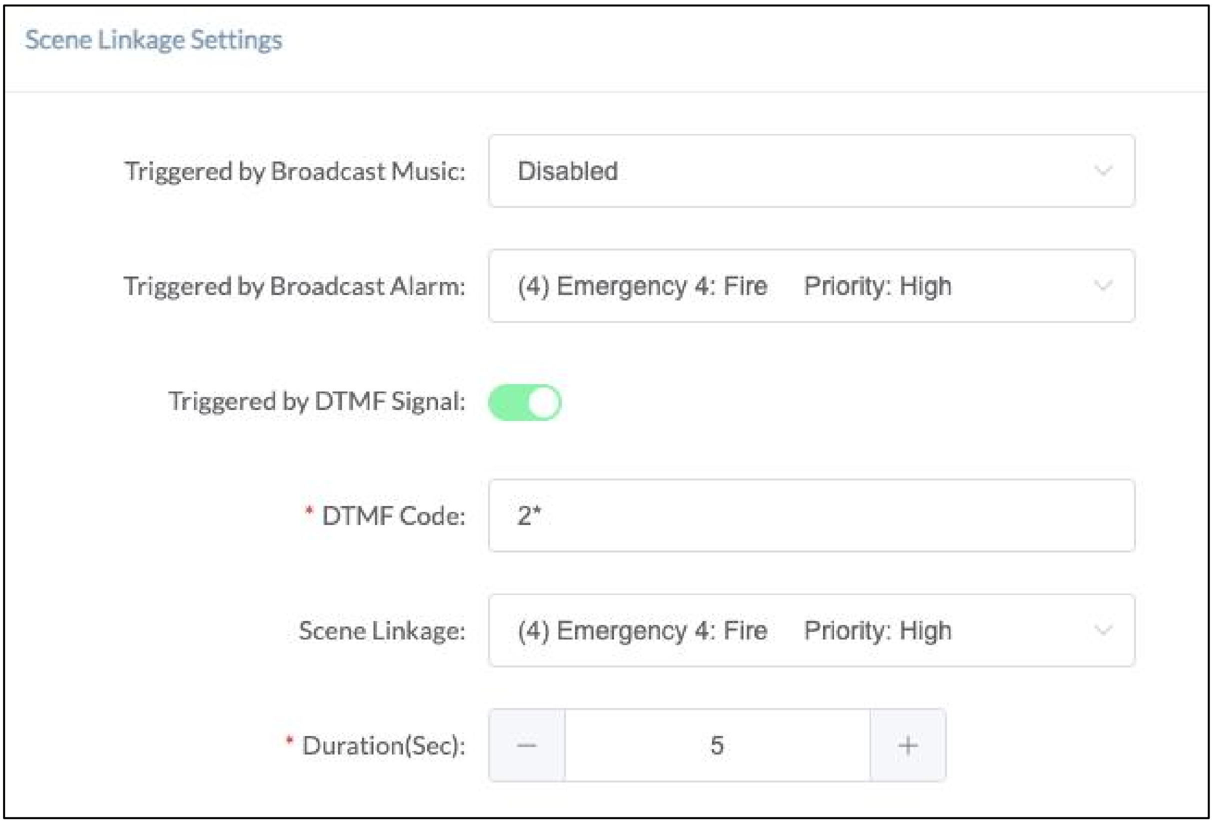
API Settings

Using the API interface to realize features such as device linkage, call control, relay control, strobe 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.

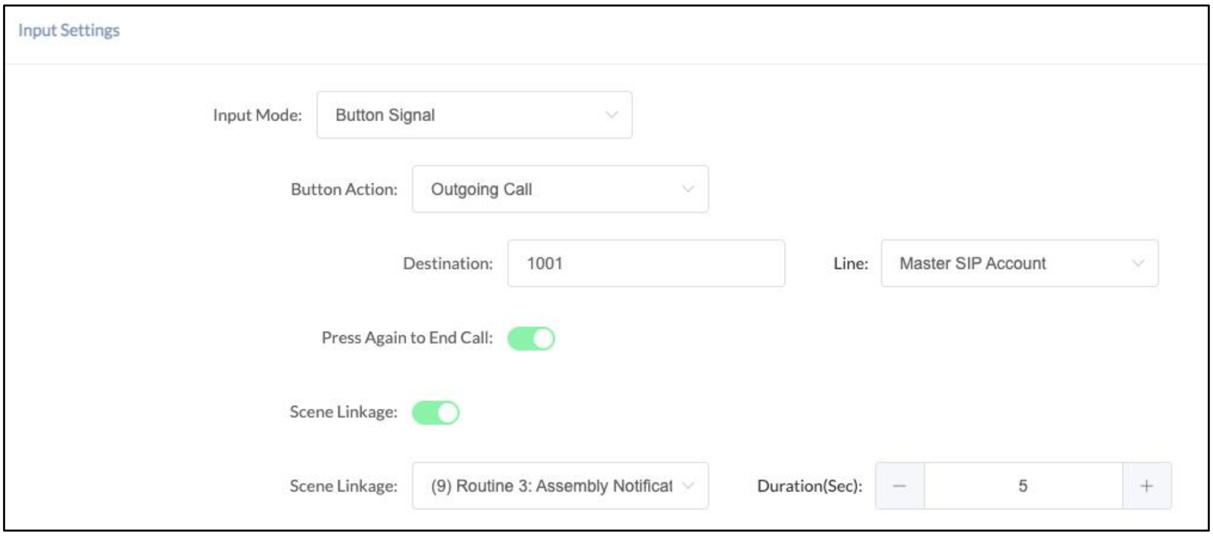
6.7 I/O Settings

This page allows users to configure input and output behaviors for device integration and security linkage, including trigger conditions, input actions, relay control, and scene linkage. Please go to **Advanced --> I/O Settings** page to set the specific settings.



Scene Linkage Settings

- **Triggered by Broadcast Music:** Enable or disable scene linkage when broadcast music is playing.
- **Triggered by Broadcast Alarm:** Enable or disable scene linkage when a broadcast alarm is active.
- **Triggered by DTMF Signal:** Enable or disable scene linkage triggered by DTMF signals.
- **DTMF Code:** Specify the DTMF code used to trigger the linkage.
- **Scene Linkage:** Select the scene preset to be activated when triggered.
- **Duration(Sec):** Set how long the scene will be perfumed.



Input Settings - Button Signal

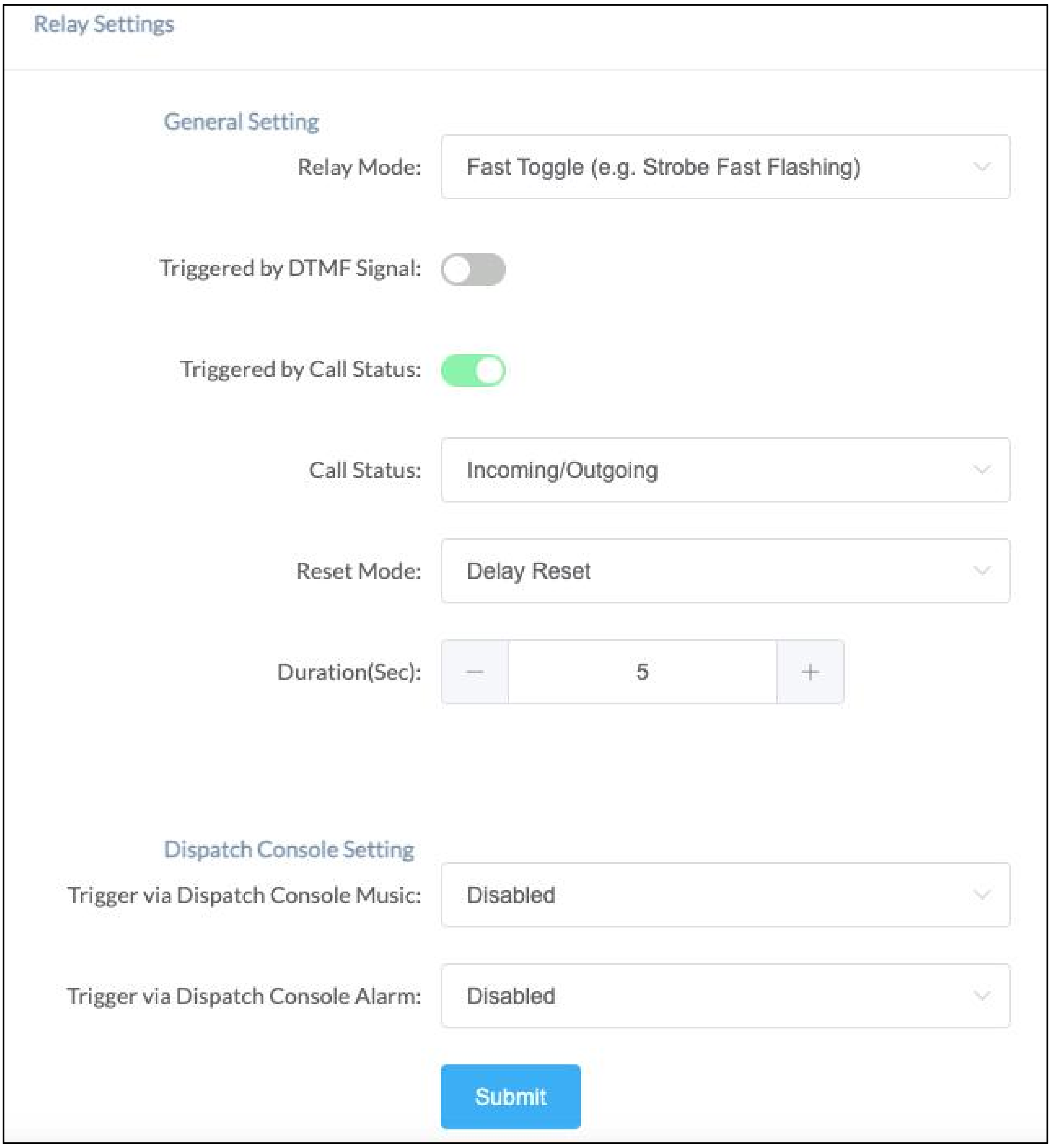
Input Settings - Switch Signal

- **Input Mode:** Input Mode: Select the input type: Button Signal and Switch Signal.
- **Action:** Select the action to be triggered when the input is activated. Available options include: Outgoing Call, HTTP Request, Play Audio.
- **Destination:** Enter the target number or device to call when triggered.
- **Line:** Select the SIP line used for the call.

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

- **Press Again to End Call:** Enable to allow pressing the button again to terminate the call.

- **HTTP URL:** Enter the API URL to be triggered.
- **Audio File:** Select the audio to be played (ringtone, audio file, or playlist).
- **Repeat:** Set how many times the audio will be played.
- **Scene Linkage:** Enable scene preset to trigger alert actions.
- **Scene Linkage:** Select a scene preset to trigger display and alert actions.
- **Duration(Sec):** Set the display duration for the scene.



Relay Settings

- **Relay Mode:** Select the relay behavior: Disabled, Single Toggle (e.g. Door Strike, Audible Visual Alarm), Fast Toggle (e.g. Strobe Fast Flashing), Slow Toggle (e.g. Strobe Slow Flashing).
- **Triggered by DTMF Signal:** Enable or disable relay triggering via DTMF (RFC2833 supported).
- **DTMF Code:** Specify the DTMF code used to trigger the relay.
- **Trigger by Call Status:** Enable or disable relay triggering based on call status changes.
- **Call Status:** Select the call state that triggers the relay, including: Outgoing, Incoming, Incoming/Outgoing, Answered, and Hangup.
- **Reset Mode:** Select how the relay returns to its original state.
- **Duration (sec):** Available when Delay Reset is selected. Defines how long the relay remains active before resetting.
- **Triggered via Dispatch Console Music:** Select the relay behavior when the Dispatch Console broadcast music is active.
- **Triggered via Dispatch Console Alarm:** Select the relay behavior when the Dispatch Console broadcast alarm is active.

6.8 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 Settings

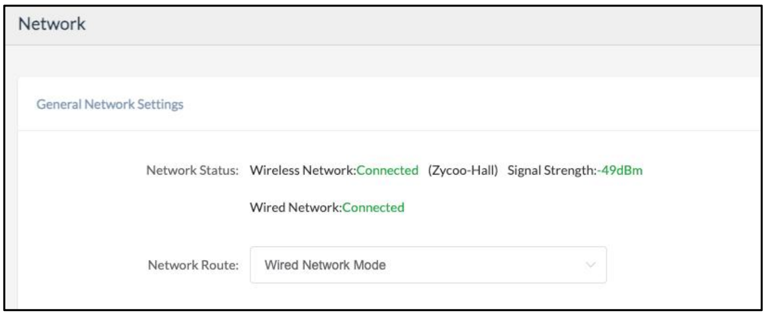
- **Enable:** Enable or disable PTP time synchronization. When enabled, the device will use the configured PTP server for high-precision time alignment instead of relying only on NTP.
- **PTP Server:** Specify the IP address or domain name of the PTP master clock server used for synchronization.

7. System Settings

7.1 Network

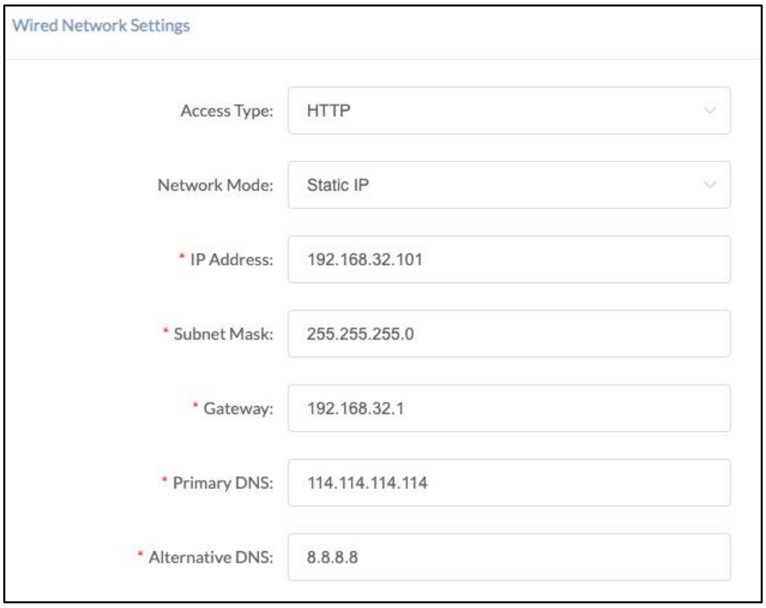
The IPS-M1-BW supports both wired and wireless network connections and uses DHCP to obtain an IP address automatically by default. When both wired and wireless are available, the system will follow Network Route configuration for traffic selection.

To configure a static IP address, please go to **System--> Network**, disable the DHCP option, and manually enter the required network parameters.



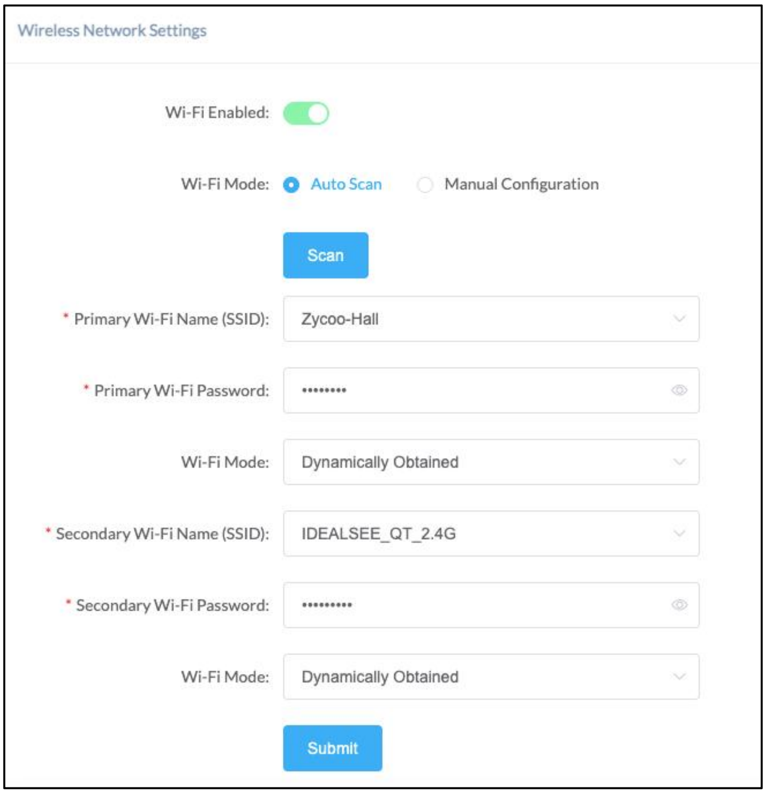
General Network Setting

- **Network Status:** Displays the current network connection status, including: Wireless network connection status and signal strength, Wired network connection status
- **Network Route:** Defines the preferred network interface for data transmission. Users can specify whether traffic is routed through the wired network or wireless network when both are available.



Wired Network Setting

- **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.

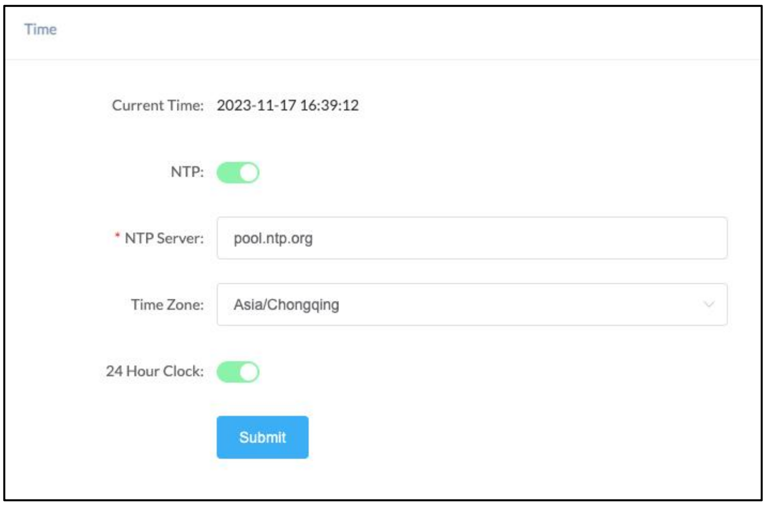


Wireless Network Setting

- **Wi-Fi Enabled:** Enable or disable the wireless network function.
- **Wi-Fi Mode:** Select the configuration mode for Wi-Fi connection. The following options are available: Auto Scan: Automatically scan and connect to available Wi-Fi networks. Manual Configuration: Manually configure Wi-Fi network parameters
- **Scan:** Click to scan and display available Wi-Fi networks.
- **Primary Wi-Fi Name (SSID):** Enter the SSID of the primary Wi-Fi network.
- **Primary Wi-Fi Password:** Enter the password of the primary Wi-Fi network.
- **Secondary Wi-Fi Name (SSID):** Enter the SSID of the backup Wi-Fi network.
- **Secondary Wi-Fi Password:** Enter the password of the backup Wi-Fi network.
- **Wi-Fi Mode:** Select how the device obtains an IP address: Dynamically Obtained (DHCP): Automatically obtain IP address from the network. Static IP: Manually configure IP address.

7.2 Time

The device synchronizes its system time with network time servers using NTP. The synchronized time is also displayed on the screen in idle mode.
To change the NTP settings, please go to **System --> Time** page.

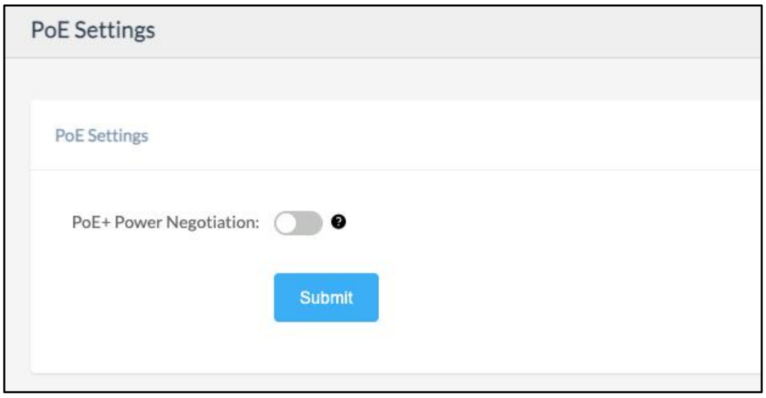


Time Settings

- **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 PoE Settings

This page allows users to configure PoE power negotiation settings. When the connected PoE injector or switch supports negotiation but does not provide PoE+ (IEEE 802.3at) power by default, this option can be enabled to request PoE+ power from the power source.
To enable the settings, please go to **System --> PoE Settings** page.



PoE Settings

- **PoE+ Power Negotiation:** Enable/Disable PoE+ (IEEE 802.3at) power negotiation. When enabled, the device will request higher power from the connected PoE power source if supported.

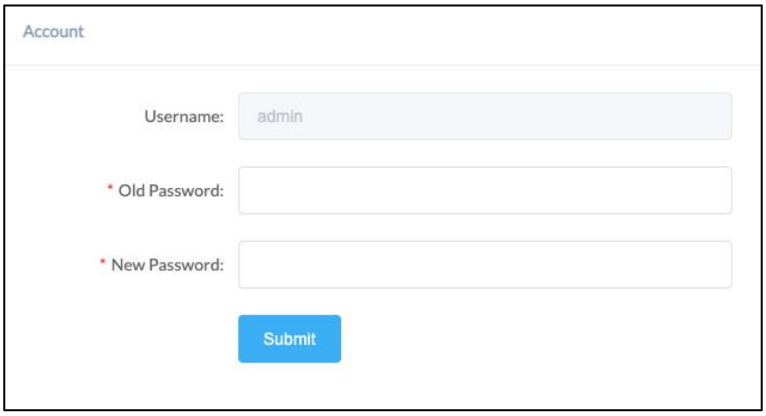
7.4 Prompt Language

The language of local voice prompts, like IP address announcements.
Please go to **System --> Prompt Language** page to set a voice prompt language.

Prompt Language

7.5 Account

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



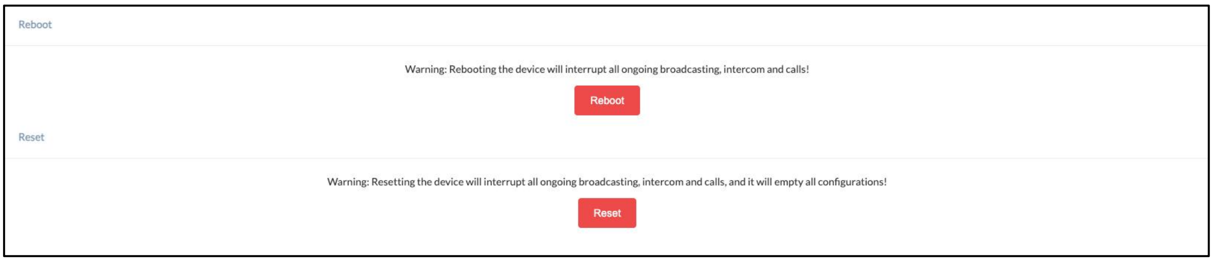
Web Password Settings

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

7.6 Reboot & Reset

IPS-M1-BW 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!

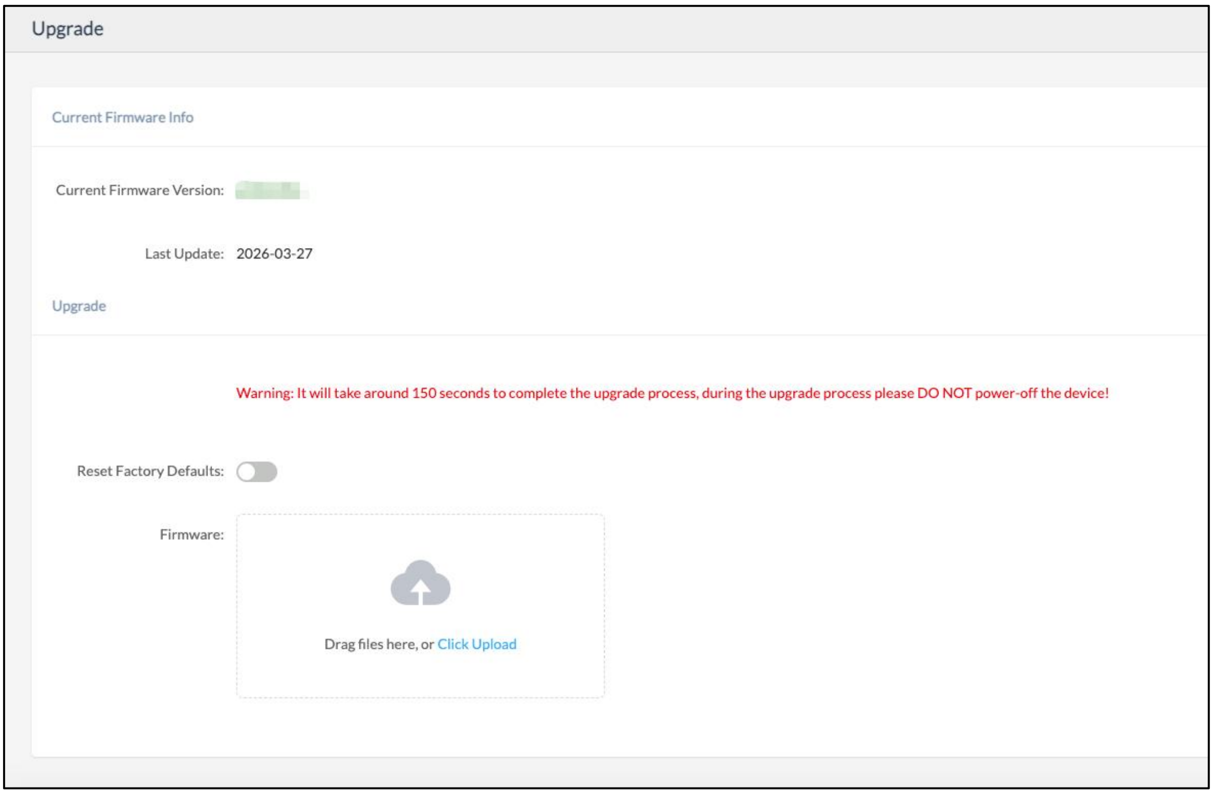
Reboot Schedule

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

8. Maintenance

8.1 Upgrade

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

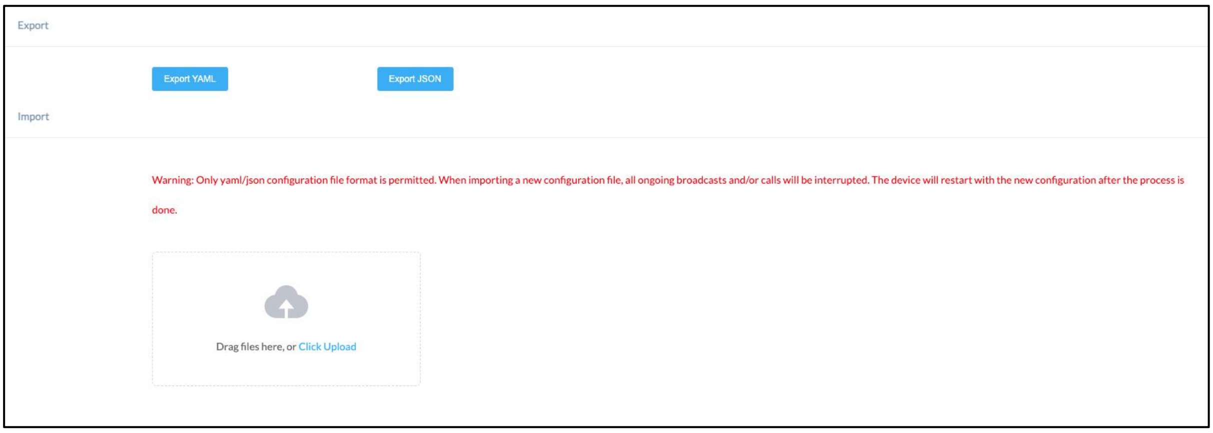


Upgrade Settings

- **Current Firmware Version:** Displays the version currently used by the system.
- **Last Update:** Displays the last system update 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.



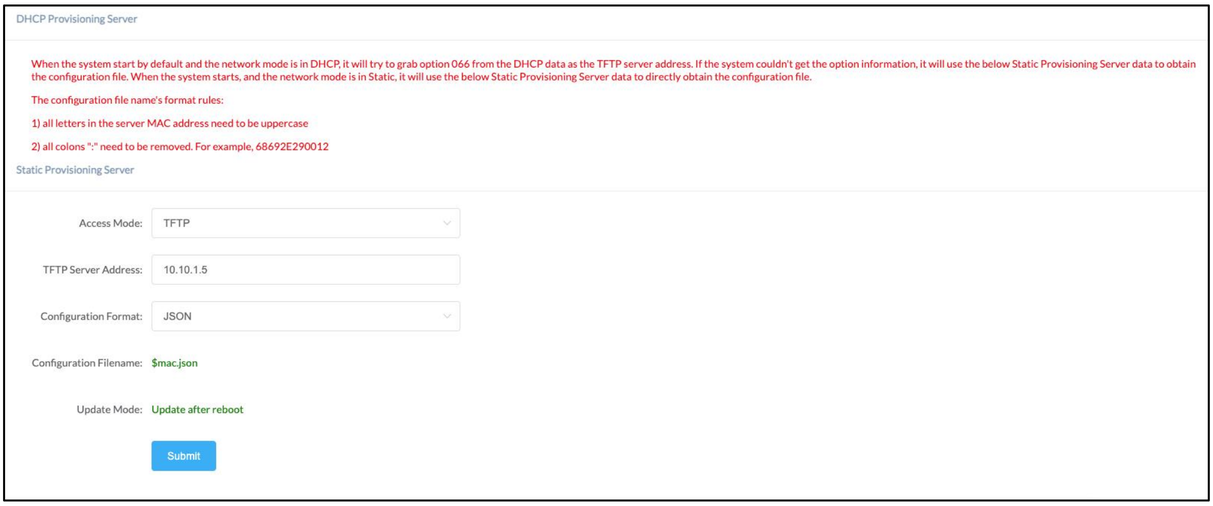
Import/Export

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

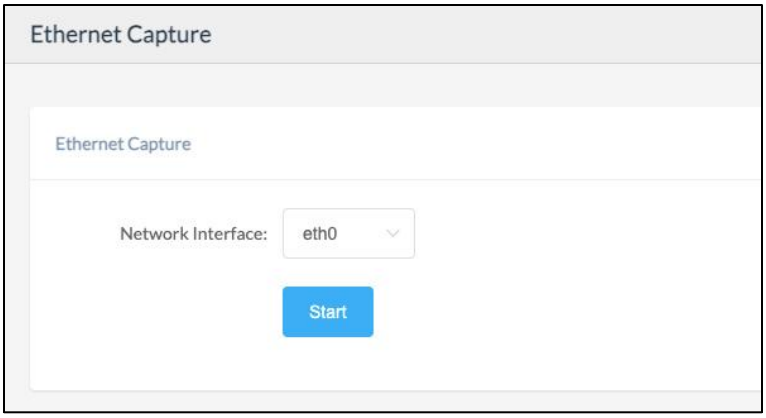
8.4 Diagnostic

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 Ethernet Capture tool allows users to capture network packets from the device and save them as standard Wireshark-compatible .pcap files for analysis.
Users can capture traffic from the following network interfaces: eth0 (Ethernet interface) and wlan0 (Wireless interface).
Please go to **Maintenance --> Ethernet Capture** page to operate.



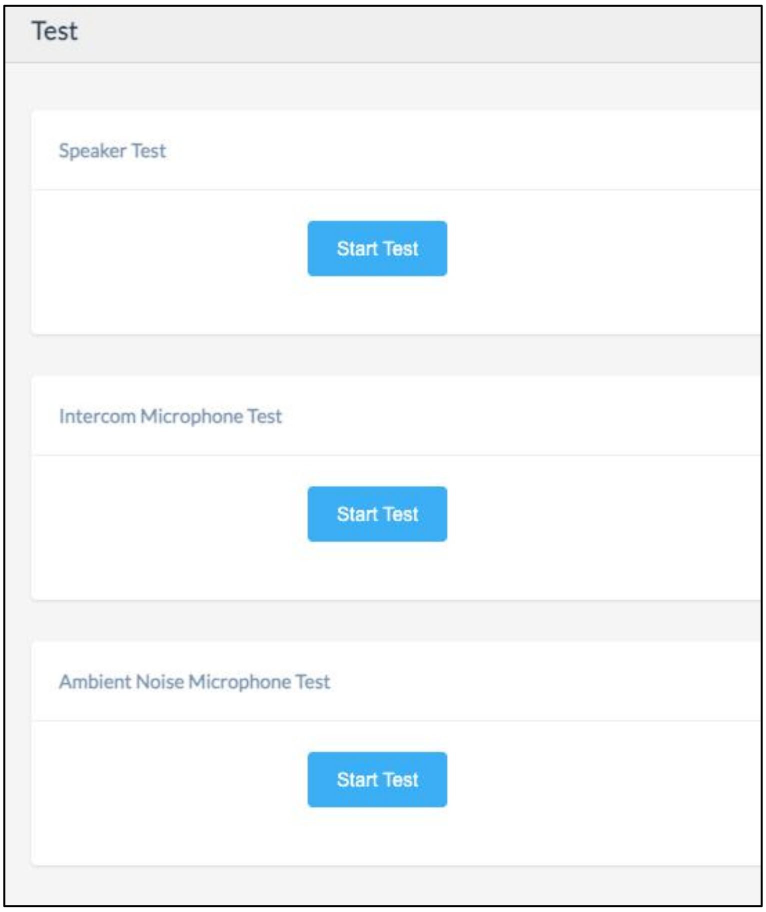
Ethernet Capture

8.6 Test

The Test feature allows users to verify whether key hardware components (such as speaker, microphones, strobe, and relay) are functioning properly before deployment.

Please go to **Maintenance --> Test** page to perform diagnostic tests.

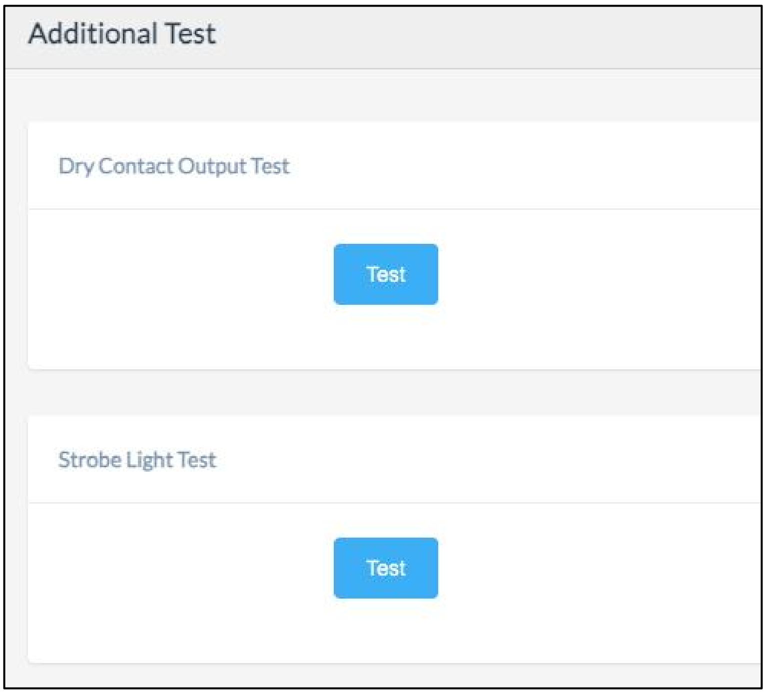
8.6.1 Audio Test



Audio Test Settings

- **Speaker Test:** Click Start Test, and the device will play a ringtone to verify that the speaker is functioning properly. If the test is successful, audio should be heard from the speaker.
- **Intercom Microphone Test:** Click Start Test, then speak toward the device. The system will perform a microphone-to-speaker playback test, and your voice should be played back through the speaker.
- **Ambient Noise Microphone Test:** Click Start Test to begin detection. A waveform will be displayed on the screen, allowing users to visually confirm whether the microphone is capturing ambient sound properly. This test is ONLY effective when the Ambient Noise Compensation feature is enabled.

8.6.2 Additional Test



Additional Test Settings

- **Dry Contact Output Test:** Click Test to trigger the relay output and verify that the dry contact interface is functioning properly.
- **Strobe Light Test:** Click Test to activate the strobe light and verify its behavior.

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.

Download

Time	Type	Event	Action
2024-03-15 17:27:12	MQTT	STATUS	{statusText: idle}
2024-03-15 17:27:10	MQTT	SERVER COMMAND	{action:stop;data:{};id:332b32c0-e2ae-11ee-8d39-5f2d8908e54b;time:2024-03-15T09:27:10.572Z}
2024-03-15 17:27:06	MQTT	STATUS	{soft-volume: 48}
2024-03-15 17:27:05	FUNCTION	RELAY Control	OFF
2024-03-15 17:27:05	MULTICAST	STOP PLAY	
2024-03-15 17:25:51	MQTT	STATUS	{soft-volume: 0}
2024-03-15 17:25:50	MULTICAST	START PLAY	Priority:2 <239.168.11.142:2200>
2024-03-15 17:25:50	FUNCTION	RELAY Control	SlowFlashing
2024-03-15 17:18:08	MQTT	STATUS	{soft-volume: 48}
2024-03-15 17:18:06	MULTICAST	STOP PLAY	
2024-03-15 17:18:05	FUNCTION	RELAY Control	OFF
2024-03-15 17:18:02	MQTT	SERVER COMMAND	{action:stop-pull-rtsp;data:{};id:ec25b310-e2ac-11ee-8d39-5f2d8908e54b;time:2024-03-15T09:18:01.921Z}
2024-03-15 17:18:02	MQTT	SERVER COMMAND	{action:resume;data:{value:play};id:ec205be0-e2ac-11ee-8d39-5f2d8908e54b;time:2024-03-15T09:18:01.886Z}
2024-03-15 17:18:02	SIP STATE	CALL CLOSED	Connection reset by peer [104]
2024-03-15 17:18:01	MQTT	SERVER COMMAND	{action:set_service_settings;data:{hard-volume:4;hard-volume-control:on};id:ec1f9890-e2ac-11ee-8d39-5f2d8908e54b;time:2024-03-15T09:18:01.881Z}
2024-03-15 17:18:00	SIP STATE	CALL ESTABLISHED	incoming

System Logs

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