SM Series

- SM10 SIP Paging Interface
- SM10-SPK IP Box Speaker
- SM10-SPk-CL IP Clock Speaker
- SM10-HSPK IP Horn

User Guide

1550-00069/RevA 5-27-25

Contents

NOTE: The four products share a common user interface. Therefore, many of the images shown will display “SIP Paging Interface”.

Introduction

The SM Family of products has been designed to work seamlessly with Symphony, FrontRow's 3rd Generation Bell, Paging and Intercom system.

SM10 - SIP Paging Interface

A SIP and Multicast audio device that converts IP audio to analog. Used with bells, public address, intercom and background music.

- IP to Analog Audio interface use with 70-volt amplifier and speaker systems
- SIP / Multicast interface with FrontRow EzRoom systems.

SM10-SPK – IP Box Speaker

A SIP and Multicast IP Speaker. Used with bells, public address, intercom and background music. An integrated microphone allows two-way intercom communications.

Add an optional call button to allow intercom call to be initiated from the SM10-SPK.

Primarily used indoor, anywhere Bell, Paging, Intercom and Alert audio is required.

The SM10-SPK-CL – IP Clock Speaker

A SIP and Multicast IP Speaker with a digital clock. Used with bells, public address, intercom and background music. An integrated microphone allows two-way intercom communications. Add an optional call button to allow intercom call to be initiated from the SM10-SPK-CL.

Primarily used indoor, anywhere a Clock, Bell, Paging, Intercom and Alert audio is required.

The SM10-HSPK – IP Horn

An outdoor rated SIP and Multicast IP Horn with an integrated microphone. Used with bells, public address, intercom and background music.

Primarily used in large spaces or outdoors, where Bell, Paging, Intercom and Alert audio is required.

SIP Paging Interface

IP Box Speaker

IP Box Speaker

IP Box Speaker

Getting Started

FrontRow recommends configuring your SM Series devices prior to installing in its intended building.

Recommended tools

- SOHO Router
- PoE Network Switch AT capable recommended r
- Computer with wired network port
- 3 (minimum) network cables
- AC power cords as required by your devices

Starting the SM Series Device

Connect SM Series Device to PoE network port.

Allow the system to boot (roughly 60 seconds). When the Power LED is solid and the System LED is blinking, the SM Series Device may be accessed through its web browser interface.

By default, the SM Series Device is set to DHCP. If the SM Series Device fails to receive a DHCP address, the SM Series Devices will fall back to its default IP address 192.168.1.101.

Accessing the SM Series Device User Interface

SM10

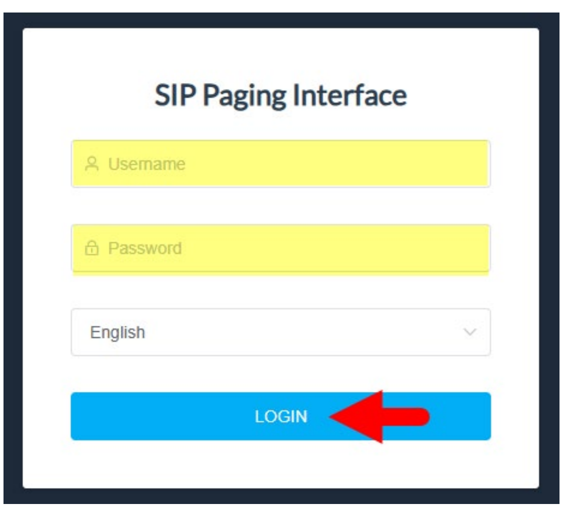
- Connect a speaker to the SM10 using the Phoenix connector or Connect headphones to the 3.5 Audio Out port.
- Once the SM10 has booted, press the volume+ button and volume- button simultaneously for 4 seconds, then release the buttons. The SM10 will play, (spoken words), the IP address.

Alternatively - Use IP Scanning software, for example Advanced IP Scanner.

- Scan your network to locate the MAC address of the SM Series Device and locate the IP address. .

Logging into the SM Series Device

1. Using your favorite web browser, type the IP address for SM Series Device into the URL field.
2. The default username and password for the SM Series Device is:
   Username = **admin**
   Password = **admin**

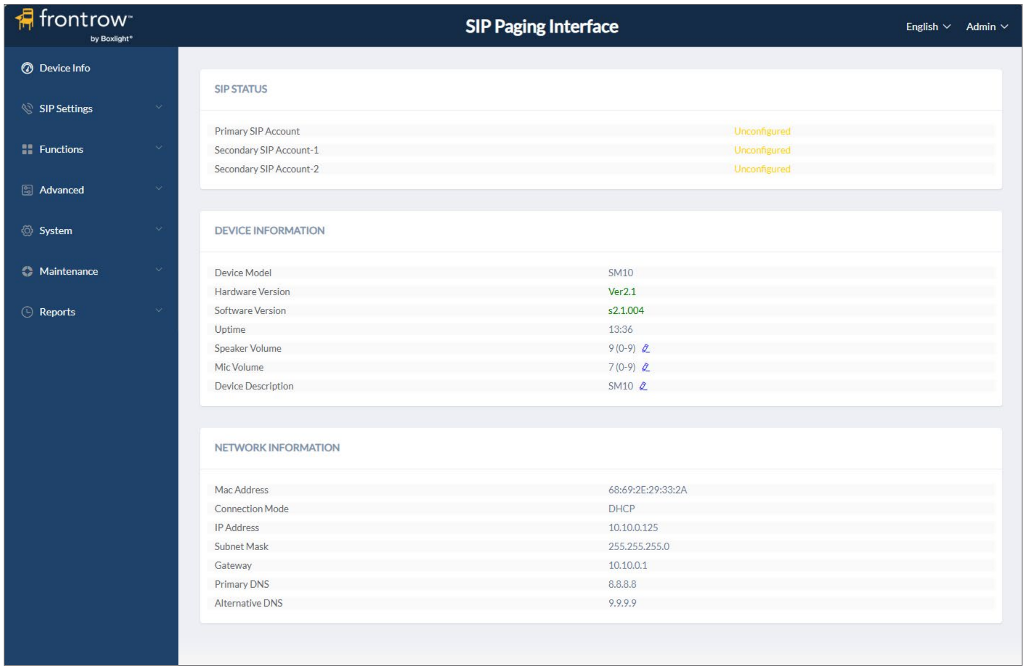


Device Info

A successful login will display the **Device Info** page.

The **Device Info** page contains three sections:

- SIP Status
- Device Information
- Network Information



SIP Status

- SIP Account: Displays the SIP Extension, SIP Server, Registration and Call Status.
  - Primary SIP Account
  - Secondary SIP Account-1
  - Secondary SIP Account-2
- Columns
  - Account
  - SIP Extension @ SIP Server address
  - Registration Status & Call Status

Device Information

Device Model: Displays the model of the device (Example: SM10).

Hardware Version: Displays the hardware version number of the device.

Software Version: Displays the system version number of the device.

Start Time: Displays the last time the device was started up.

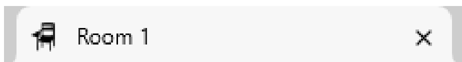
Speaker Volume: Displays the current volume of the device.
NOTE: This field may be edited.

Mic Volume: Displays the current device microphone input volume.
NOTE: This field may be edited.

Device Description: Displays the device information.
NOTE: This field may be edited.

Recommended – Location, example, "Room 1".

The description will be displayed in a browser tab after the Device Description is set.



Network 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).

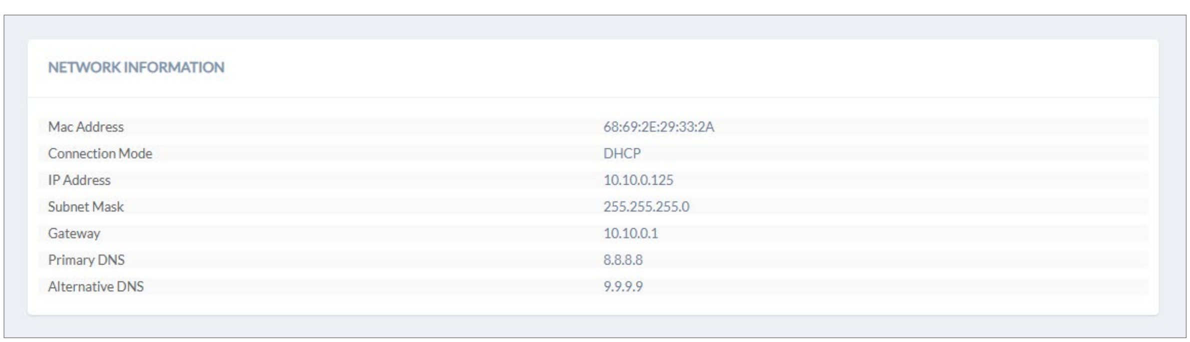
IP Address: The current IP address of the device.

Subnet Mask: The current subnet mask of the device.

Gateway: The current gateway address of the device.

Primary DNS: The current primary domain name server of the device.

Alternative DNS: The current secondary domain name server of the device.



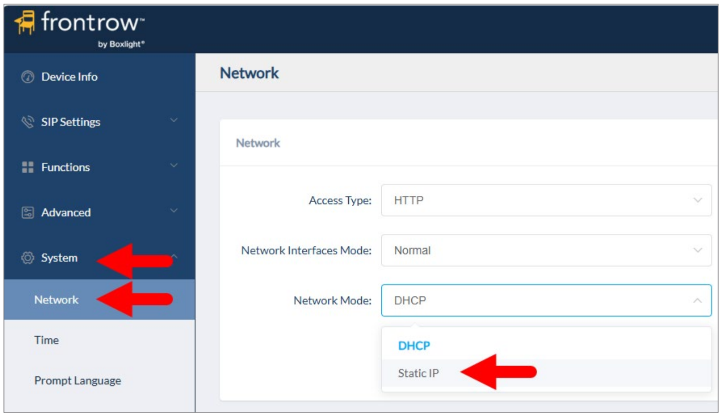
System Settings

Network

SM Series Device uses DHCP to dynamically obtain IP addresses by default.

To change the network IP from DHCP to Static, select **System** in the left menu. Then select **Network**.

On the Network page, set Network Mode to **Static**.



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.

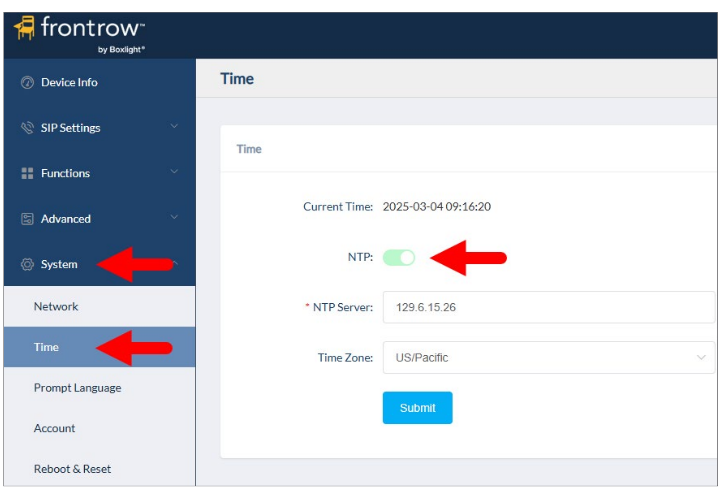
Gateway: Enter the default gateway of your LAN.

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.

Time

SM Series Device obtains the time from the network time servers using NTP.



Current Time: Display the current system time of the device.

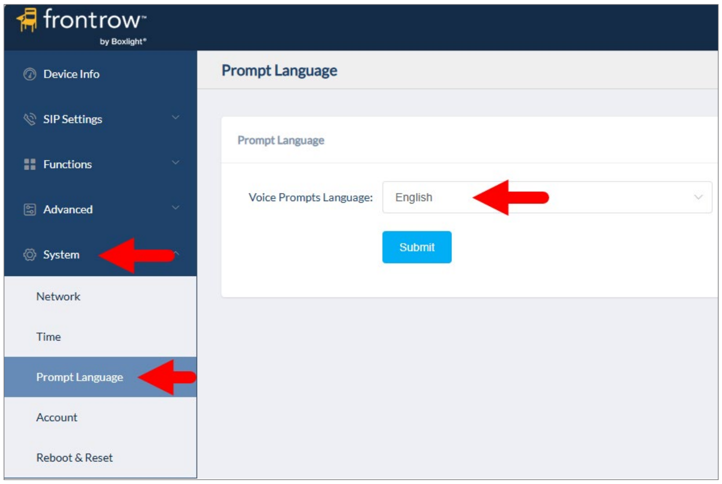
NTP: Enable/Disable using NTP to obtain the time.

NTP Server: The network time server used to obtain the time.

Time Zone: Set the time zone used by the device.

Prompt Language

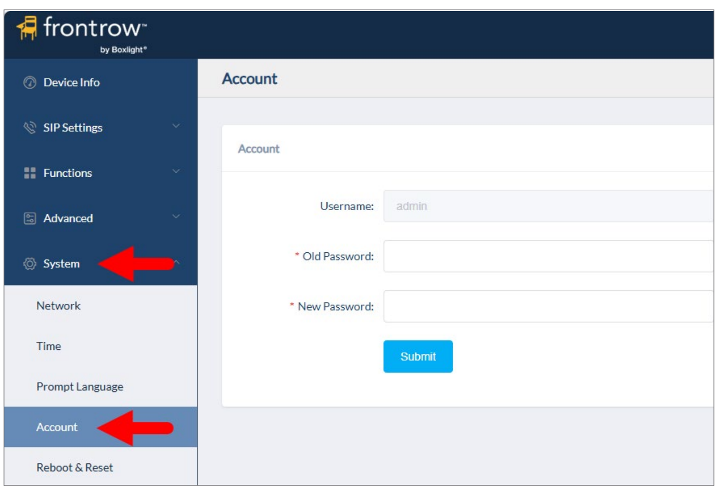
The language of local voice prompts, example IP address announcements. Currently, only Chinese and English are provided.



Account

To change the device's current password, select **System** in the left menu. Then select **Account**.

1. In the **Old Password** field, type the old password.
2. In the **New Password** field, type the new password.
3. Click the **Submit** button.



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

Reboot & Reset

The SM Series Device can be restarted from this page. The restart process takes about 10 seconds. After the restart is complete, refresh the page to log in again.

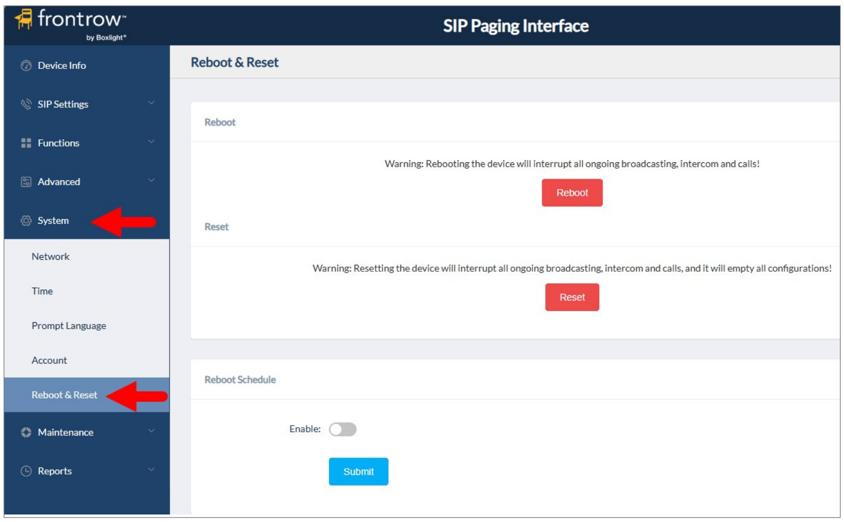
If you need to restore the factory settings of the SM Series Device, you can reset it through this page. After hearing the broadcast voice, the device will enter the state of restoration. After restarting, the pop-up window disappears, and the device is restored successfully.

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

SM Series Device can be rebooted and reset from the web management interface.

If you need to reboot or reset the device, select **System** in the left menu.

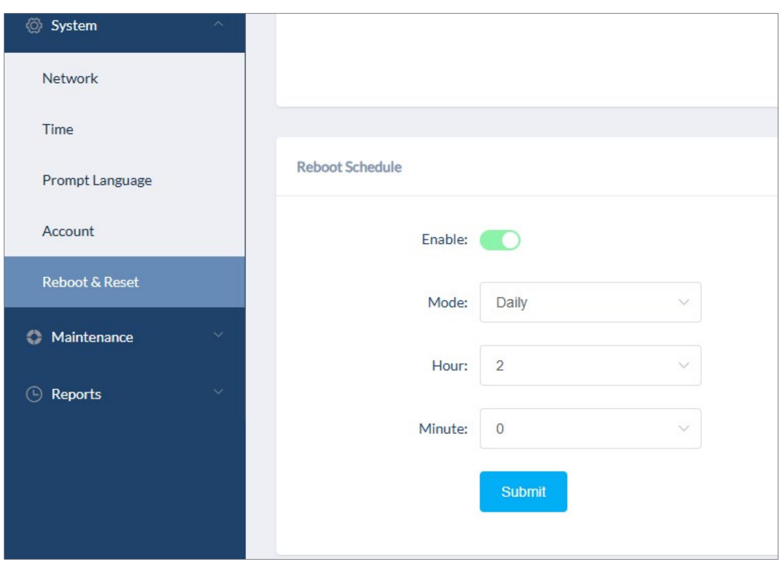
- To reboot the SM Series Device, click the **Reboot** button.
- To reset the SM Series Device, click the **Rest** button.



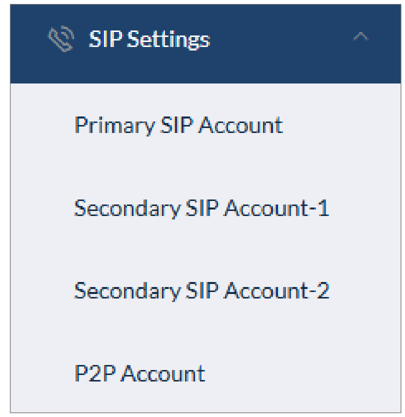
Reboot Schedule

When the Reboot Schedule feature is Enabled, the SM Series Device will automatically reboot daily, weekly, or monthly at a specified time.

NOTE: When selecting the time to reboot, the hour field is based on a 24-hour clock. As an example, if you wanted to reboot the device at 10:00 PM, you would select 22 with zero in the minutes field.



SIP Settings



The SM Series Device allows three SIP extensions to be registered:

- Primary SIP Account (Typically registered to the Symphony Server)
- Secondary SIP Account-1 (*Optional – Register to the building VoIP phone system)
- Secondary SIP Account-2 (*Optional – Register to the building VoIP phone system/Backup Symphony system)

NOTE: The SM Series Device also allows for Peer-to-Peer. This feature is not typically used in a Symphony system deployment. However, setup instructions are provided in the Appendix.

Each SIP account contains two sections: **Basic** and **Advanced**.

NOTE: When the SM Series Device is used with Symphony, only "Basic Configuration" is required. The pre-configured Advance Configurations will be sufficient.

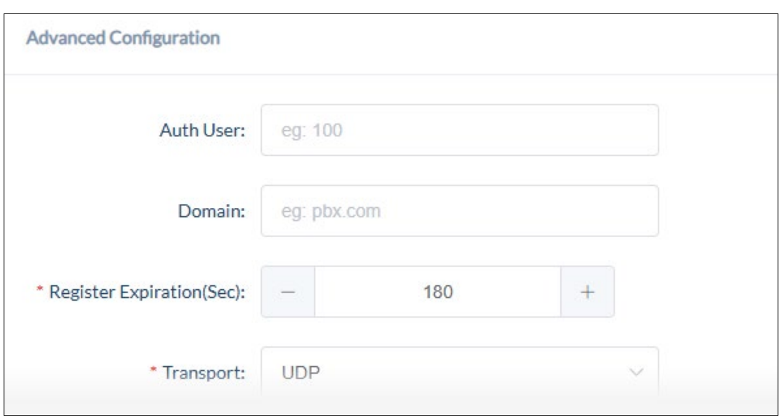
Basic Configuration

SIP Server:	Enter the IP address 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**.'
Activate:	Once enabled, the account will be activated and registered with the SIP server.

Advanced Configuration

NOTE: Non-Symphony systems may require the use of these settings

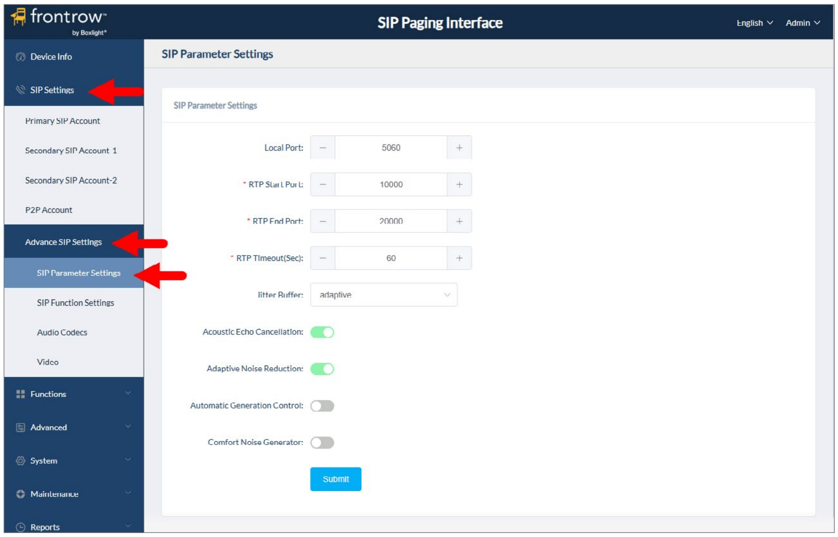
Auth User:	Enter the authorized username for the SIP account.
Domain:	Enter the SIP Domain.
Register Expiration (sec):	Set the SIP registration expiration time, with a default of 180 seconds.
Transport:	Choose the transport protocol: UDP, TCP, or TLS.
NAT Mode:	Select the NAT mode and provide the necessary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



SIP Parameter Settings

NOTE: Typically, Symphony does not require these settings to be modified from their default configuration.

To configure the Advance SIP Settings, select System in the left menu, select Advance SIP Settings.
Then select **Advance SIP Settings**.



Local Port:	This setting represents the port used to receive SIP packets.
RTP Start Port:	This setting represents the starting RTP port that the system 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 within a specific time range, if the system does not receive the RTP stream, the call will end.
Jitter 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. **NOTE:** The SM10-SPK, SM10-SPK-CL, and SM10-HSPK include integrated microphones. This option is turned on by default.
Comfort Noise Generator:	After enabling this feature, comfortable white noise can be added during calls. **NOTE:** The SM10-SPK, SM10-SPK-CL, and SM10-HSPK include integrated microphones. This option is turned on by default.

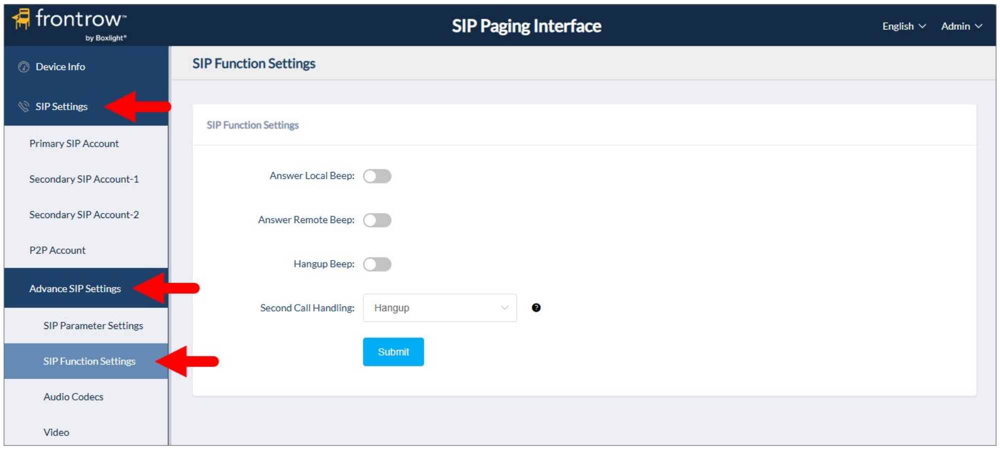
SIP Functions Settings

NOTE: Typically, Symphony does not require these settings to be modified from their default configuration.

In the event your municipality requires a “pre-announce” tone when an Intercom call has started, activate the **Answer Local Beep** slider.

To configure the SIP Function Settings, select **System** in the left menu, select **Advance SIP Settings**. Then select **SIP Function Settings**.

1. Modify the settings you wish to change.
2. Click the **Submit** button.



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 may listen to this audio file from the web user interface.
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.

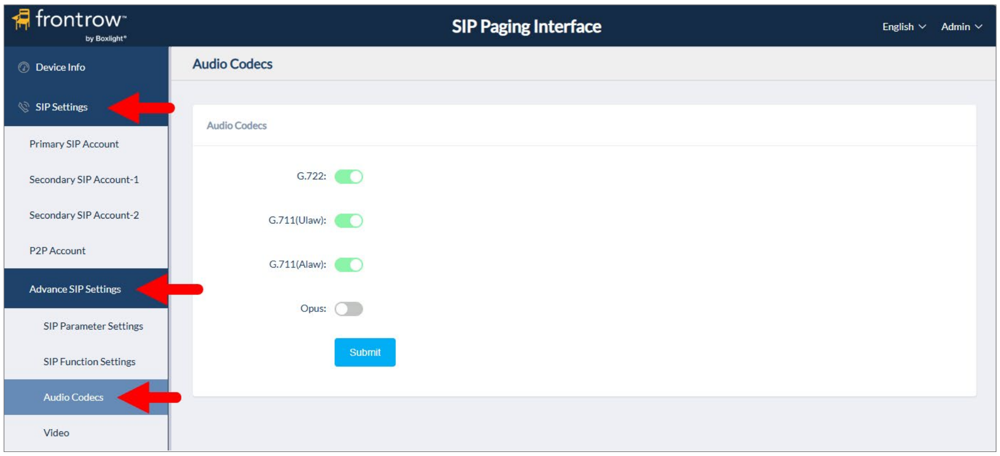
Audio Codec

SM10 supports four audio codecs:

- G.722 (wideband codec)
- G.711(Ulaw)
- G.711(Alaw)
- Opus

To modify the SM Series Device audio codecs, select **System** in the left menu, select **Advance SIP Settings**, then select **Audio Codecs**.

NOTE: Keep at least one codec enabled and supported by the SIP server.

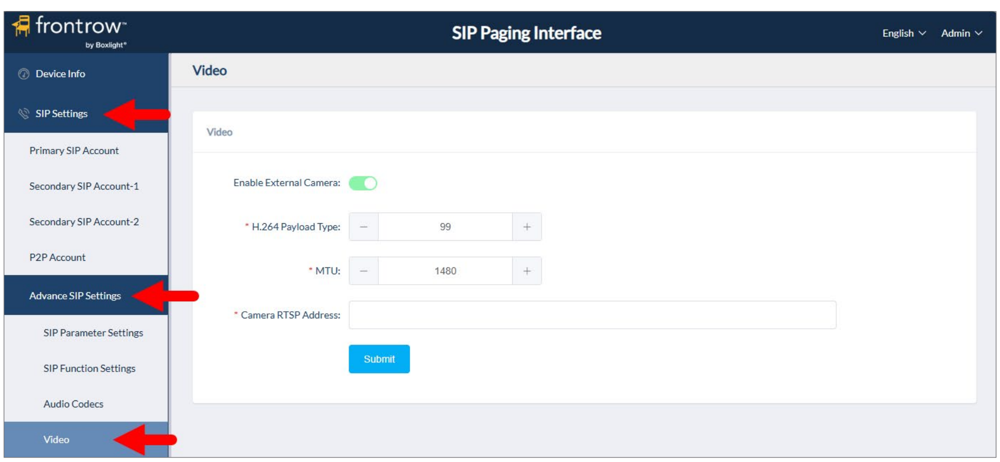


Video (SM10 Only)

NOTE: Typically, this feature will not be used with the SM Series Device.

By binding the RTSP address of the IP camera, real-time video can be provided when using the SM Series Device intercom function.

To set the SM10 Video options, select **System** in the left menu, select **Advance SIP Settings**, then select **Video**.



Enable External Camera:	Enable/Disable the video camera feature.
H.264 Payload Type:	Set the H.264 payload value.
MTU:	Set the Maximum Transmission Unit.
RTSP Address:	Set the RTSP address of the camera.

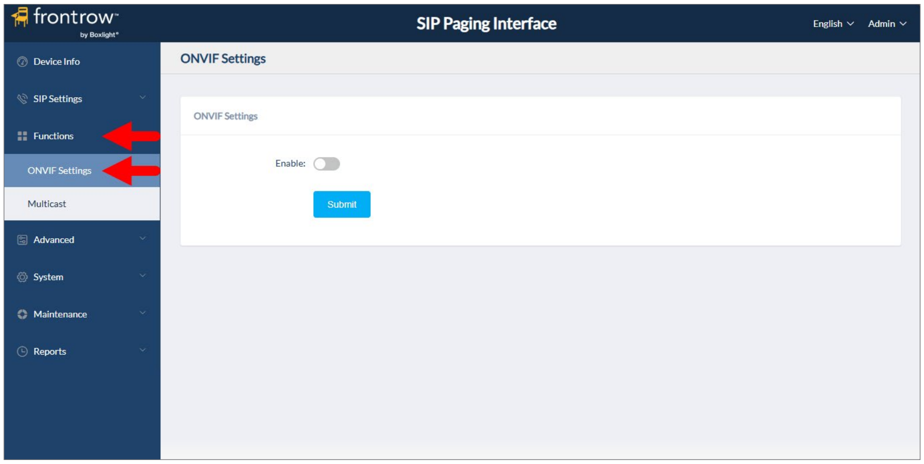
Functions

ONVIF Settings

NOTE: Typically, this feature will not be used with the SM Series Device.

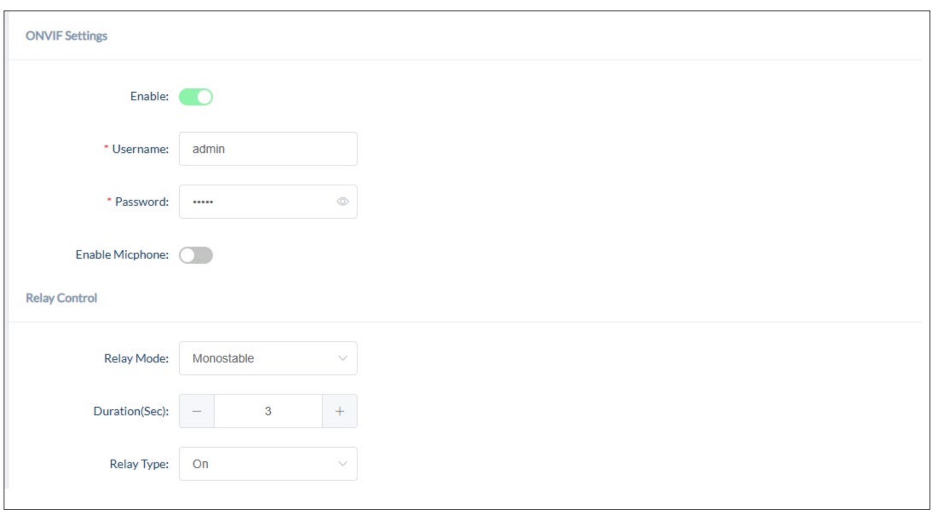
ONVIF provides and promotes standardize interfaces for effective interoperability of IPbased physical security products. If the user has installed a VMS that supports ONVIF, they can register SM Series network devices that support ONVIF on it for operation.

To configure the ONVIF Settings, select **Functions** in the left menu, select **ONVIF Settings**.



ONVIF & Relay Control Settings

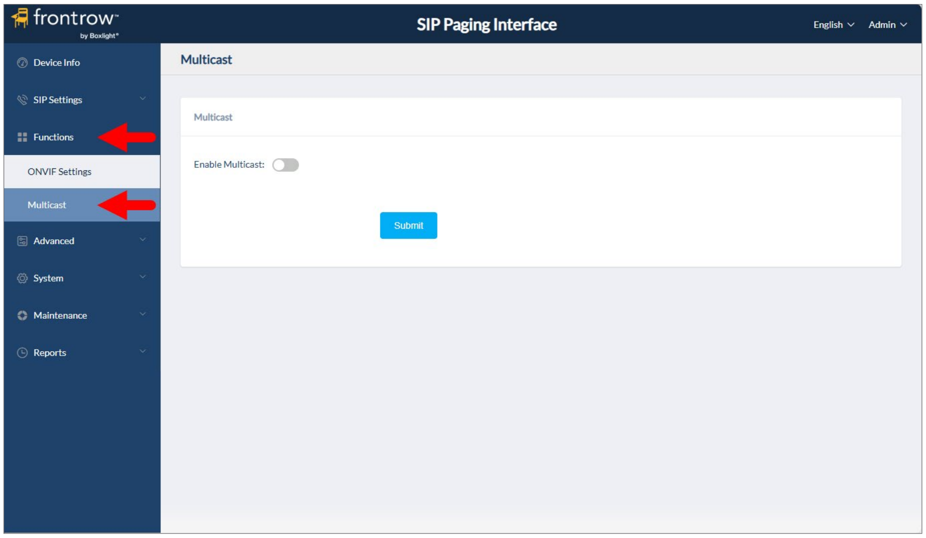
Enable:	Enable/Disable ONVIF integration for compatibility with ONVIF-supported VMS platforms.
Username:	Enter an account username with matching credentials for adding devices to the VMS platform.
Password:	Enter a matching password for the account to add devices to the VMS platform.
Enable Microphone: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



Multicast

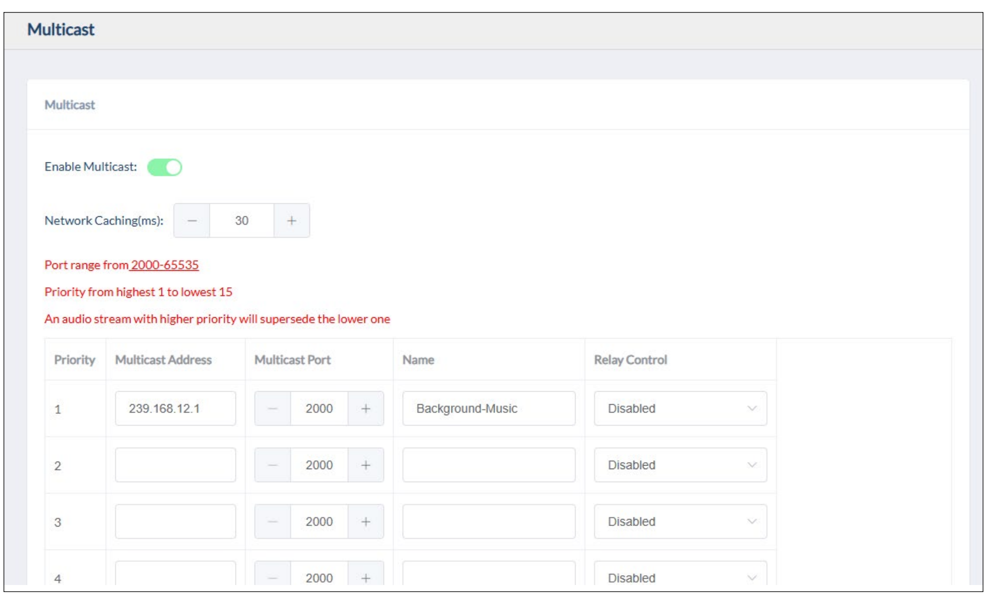
The multicast settings are used to configure Symphonies, Bell and Paging zones. The SM Series Device can be configured to monitor up to 15 different levels of multicast addresses.

To configure the Multicast Settings, select **Functions** in the left menu, select **Multicast**.



Multicast

Priority:	Priority from highest 1 to lowest 15.
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’.

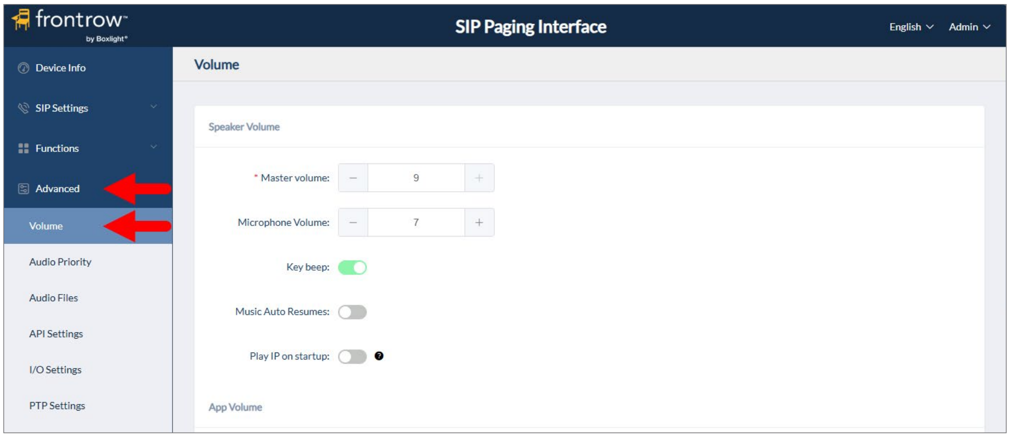


Advanced Settings

Volume Settings

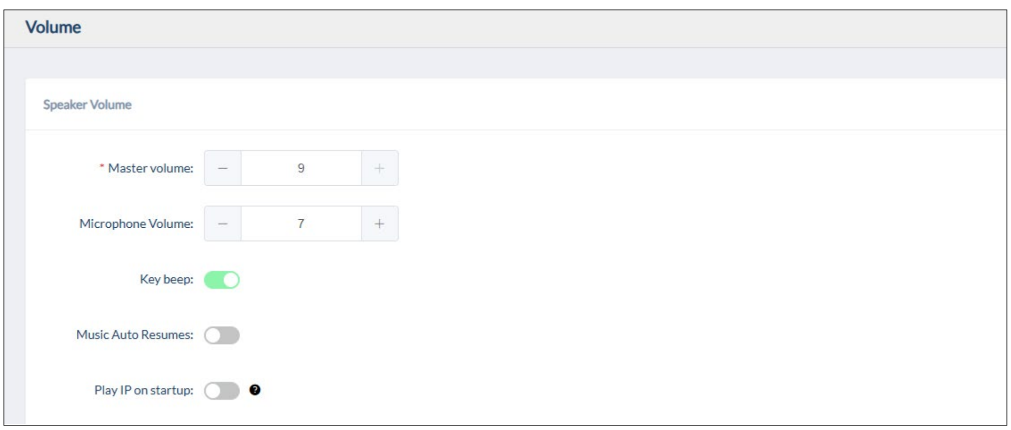
The SM Series Device allows you to adjust the main volume as well as individual volume adjustments for each application.

To modify Volume, select **Advance** in the left menu. Then select **Volume**.



Speaker Volume Settings

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

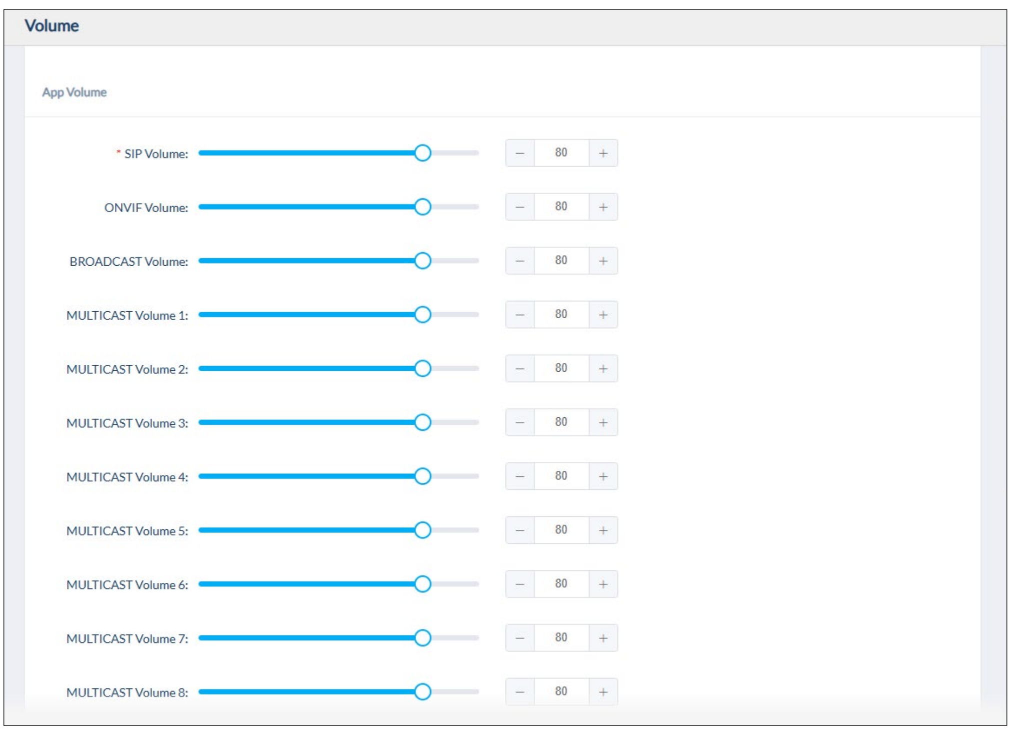


App Volume Settings

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.
Multicast Volume 10:	Set the specific volume for multicast 10 application.
Multicast Volume 11:	Set the specific volume for multicast 11 application.
Multicast Volume 12:	Set the specific volume for multicast 12 application.
Multicast Volume 13:	Set the specific volume for multicast 13 application.
Multicast Volume 14:	Set the specific volume for multicast 14 application.
Multicast Volume 15:	Set the specific volume for multicast 15 application.

To modify App Volumes:

- Select the App you wish to modify.
- Drag the slider to the left to lower the volume, to the right to raise the volume.
  Or
- Use the (-) sign to lower the volume, the (+) sign to raise the volume or type the number in the field to change.
- Click the **Submit** button at the bottom of the page.

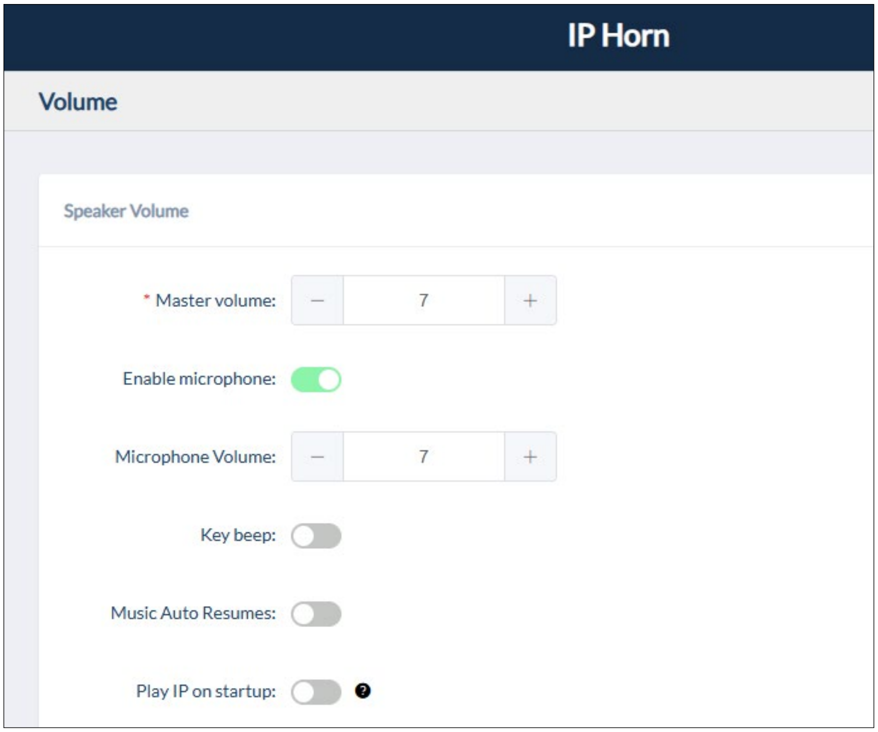


Volume – Enable Microphone (IP Horn)

The SM-HSPK contains an integrated microphone which is turned off by default. To enable the microphone, move the slider to the right.

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.



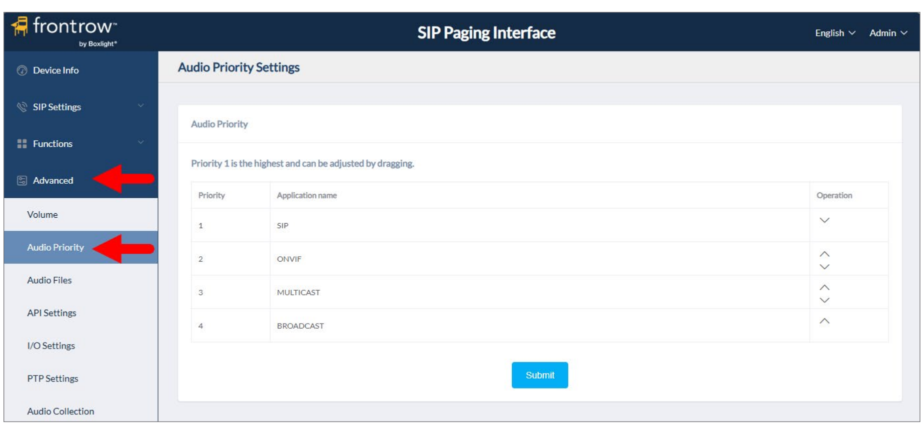
Audio Priority Settings

NOTE: Typically, Symphony does not require these settings to be modified from their default configuration.

The audio priority can be set according to different applications (such as SIP, ONVIF, MULTICAST, BROADCAST...).

To modify Audio Priority, select **Advance** in the left menu. Then select **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.



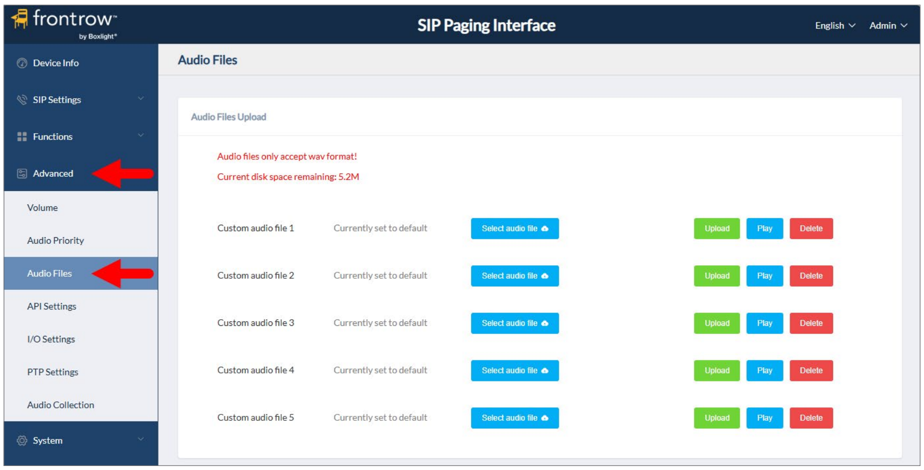
Audio/Ring Files

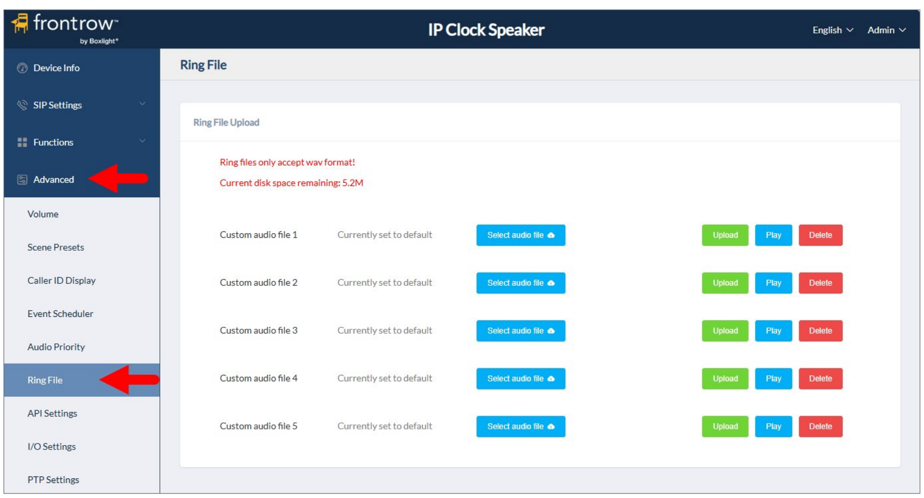
NOTE: Typically, Symphony does not require these settings to be modified from their default configuration.

The Audio files section allows users to self-upload up to 5Mb 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** button to test and play the audio file and the **Delete** button for deleting the audio file.

To play or modify Audio Files for the SM10 SIP Paging Interface, or SM10-HSPK IP Horn, select **Advance** in the left menu. Then select **Audio Files**.

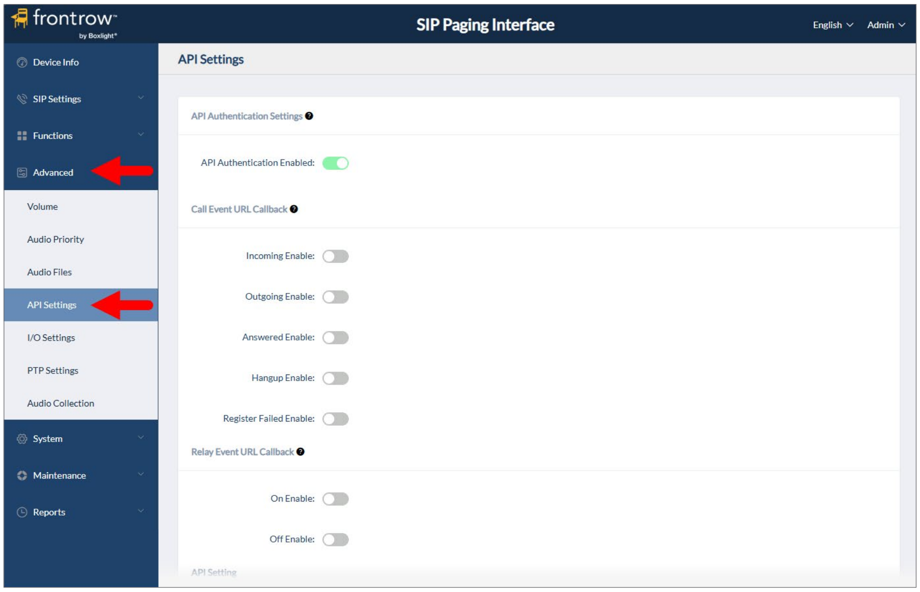
To play or modify Audio Files for the IP Box Speaker SM10-SPK-CL or IP Clock Speaker SM10-SPK, select **Advance** in the left menu, then select **Ring Files**.





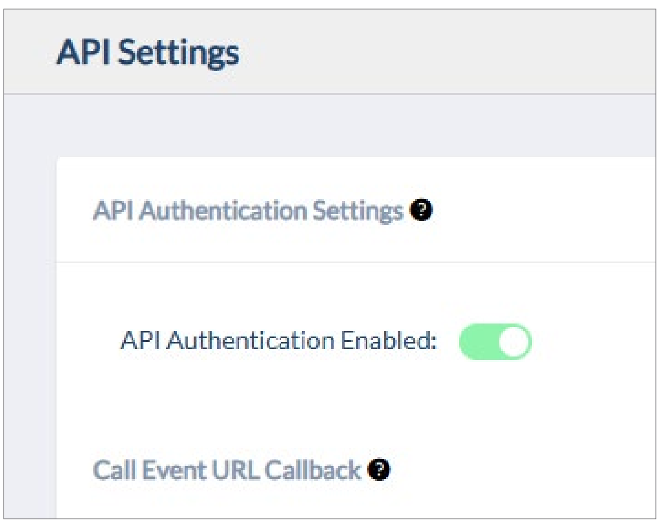
API Settings

To modify API Settings, select **Advance** in the left menu. Then select **API Settings**.



API Authentication Settings

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



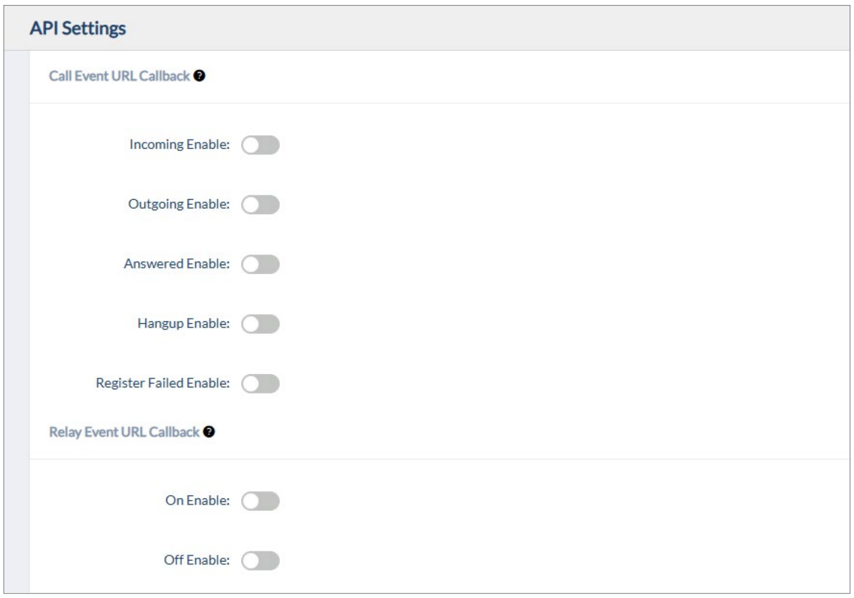
Call Event URL Callback & Relay Event URL Callback

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:

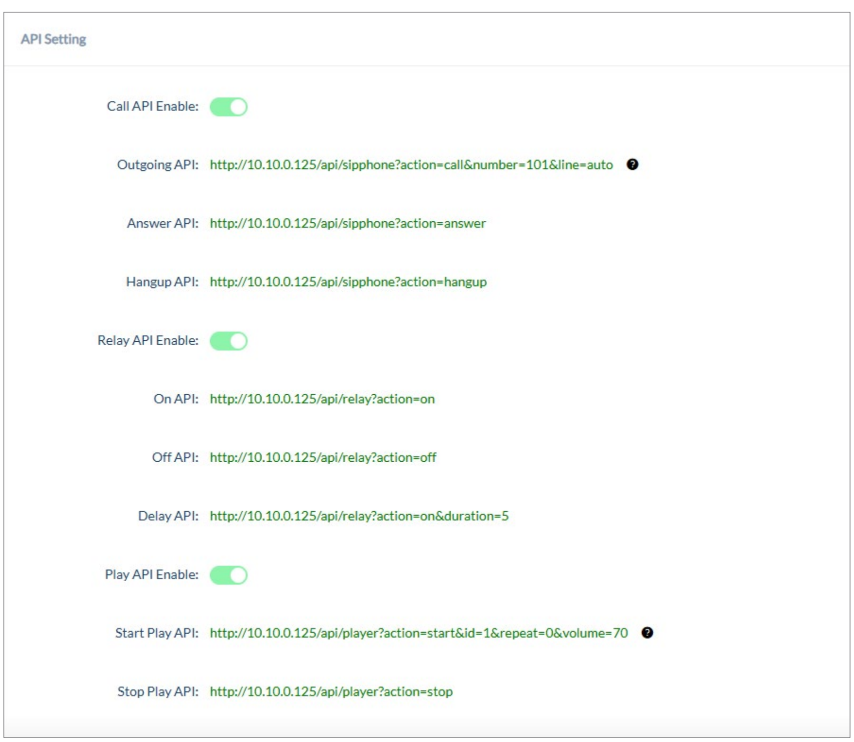
${ip}:	The current IP address of the device
${mac}:	The current MAC address of the device
${ua}:	The account of the current call
${number}:	The number of the current call



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.



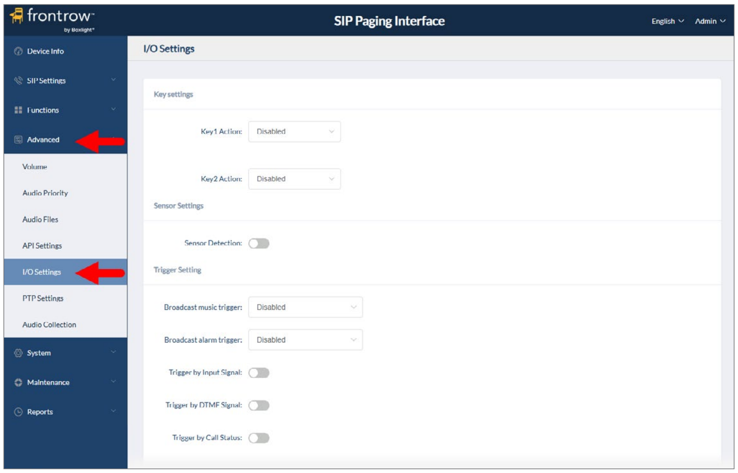
API Settings Display – (IP Clock Speaker)

The IP Clock Speaker SM10-SPK includes a show and cancel display API, allowing text to be shown on the devices panel.

I/O Settings

The I/O allows configuration parameters related to security linkage, such as: relay settings and other related configurations to be set.

To modify I/O Settings, select **Advance** in the left menu. Then select **I/O Settings**.

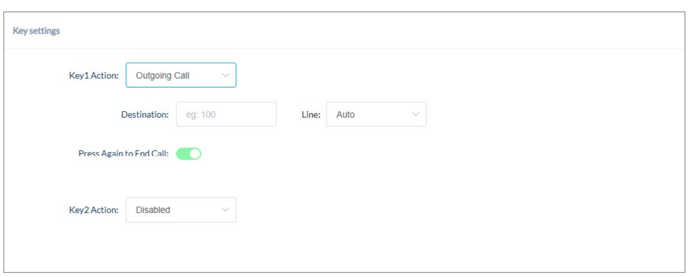


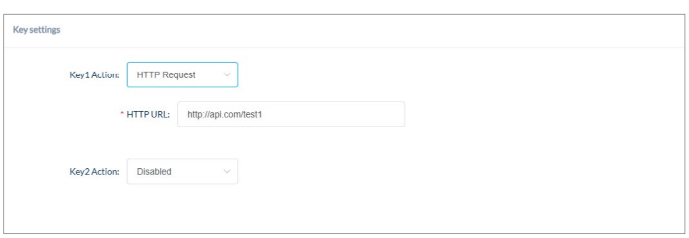
Key Settings

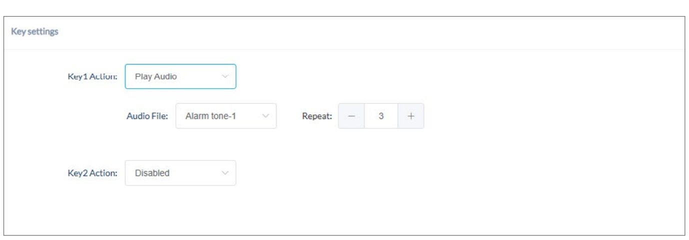
Key Action:	Choose different event linkage including Outgoing Call, HTTP Request and Play Audio.
Destination:	This setting represents the response device’s number when the external button is pressed.
Line:	This setting represents the corresponding line for making outgoing calls.

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.
Audio File:	Configure the audio triggered by linkage.
Repeat:	Configure the times of audio repetitions triggered by linkage.

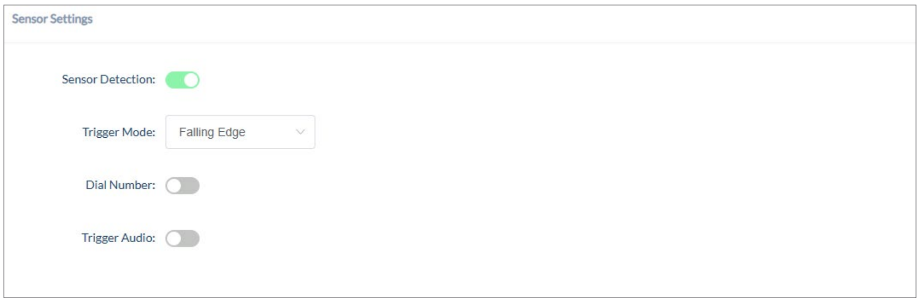






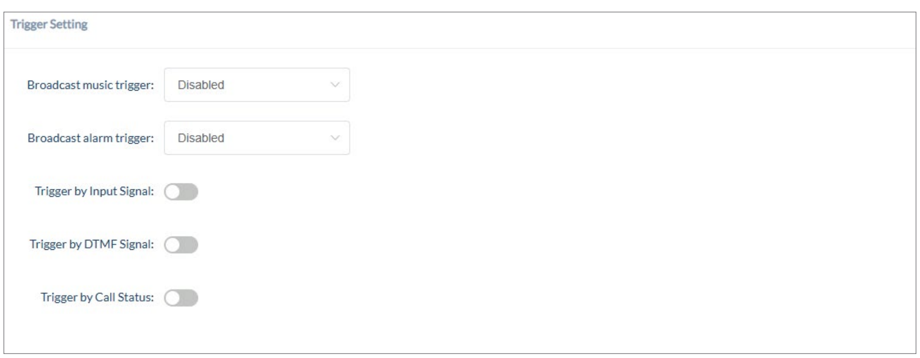
Sensor Settings

Sensor Detection:	Enable/Disable, enable this option to detect any digital level signals.
Trigger Mode:	You can choose 'Falling Edge' or 'Rising Edge' mode to be triggered.
Dial Number:	Enable/Disable, enable this option if you need to dial a number when the corresponding digital signal is received.
Number:	The number that needs to be dialed.
Line:	Line represents the line that will be used to dial calls.
Trigger Audio:	Enable/Disable, enable this option if you need to trigger an audio when the digital signal is received. Please note that you can only select either 'Dial Number' or 'Trigger Audio', not both.
Audio File:	Select the audio file to be played when triggered by the sensor input.
Repeat:	Configure the times of audio repetitions triggered by the sensor input.



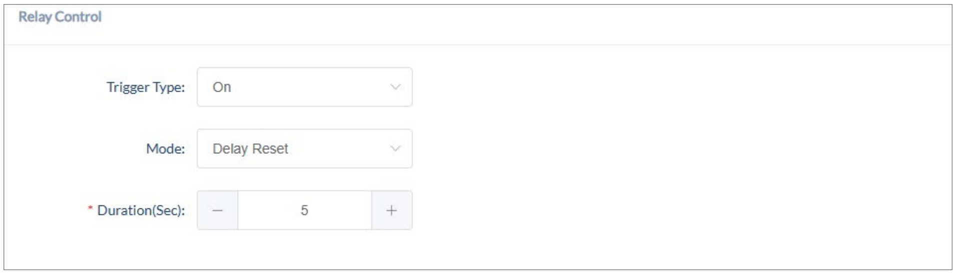
Trigger Settings

Broadcast Music Trigger:	Disabled/On/Fast Flashing/Slow Flashing, enabling this option will trigger the relay when there is broadcast music on.
Broadcast Alarm Trigger:	Disabled/On/Fast Flashing/Slow Flashing, enabling this option will trigger the relay when there is a broadcast alarm on.
Trigger by DTMF Signal:	Enable/Disable, enable this option when needed 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.
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.

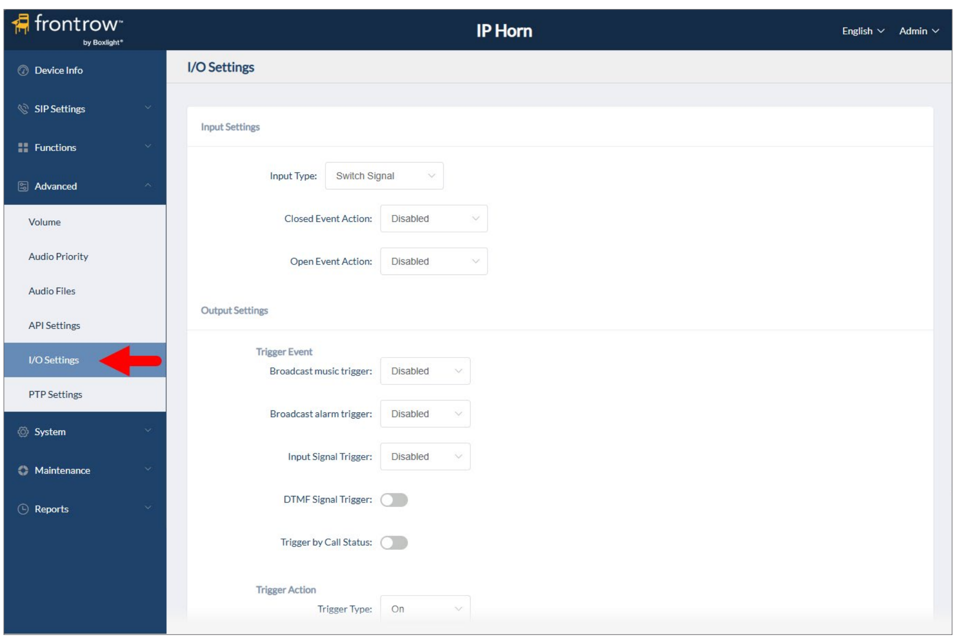


I/O Settings – (IP Horn)

The I/O settings are slightly different for the IP Speake Horn. If you intend to use these features, please review this section first.

The I/O allows configuration parameters related to security linkage, such as: relay settings and other related configurations to be set.

To modify I/O Settings, select **Advance** in the left menu. Then select **I/O Settings**.

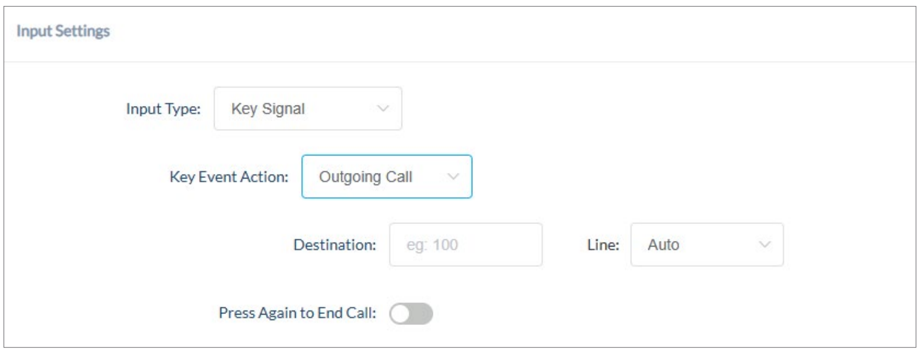


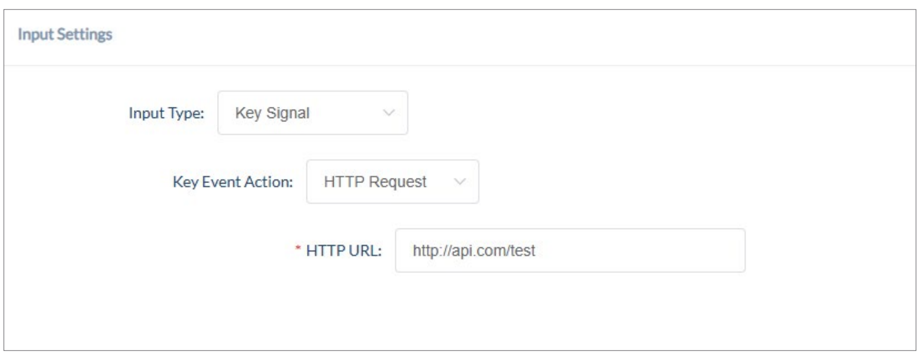
Key Signal Input Settings

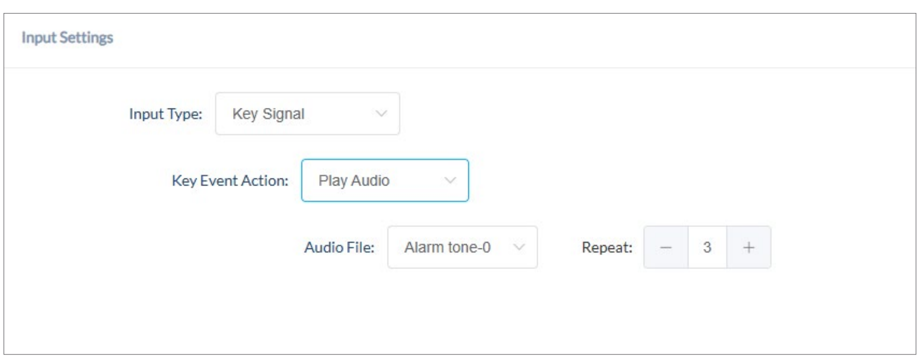
Input Type:	Choose the Key Signal or Switch Signal to trigger events.
Key Event Action:	Choose different event linkage including Outgoing Call, HTTP Request and Play Audio.
Outgoing Call:	Make a call to the destination.
Destination:	This setting represents the response device's number when the external 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.





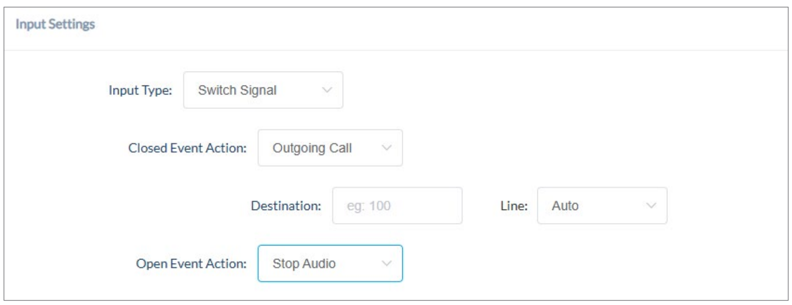


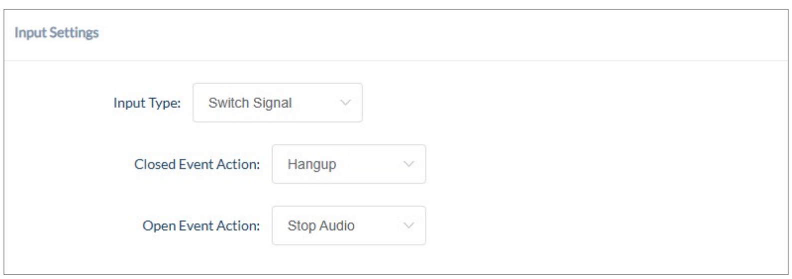
Switch Signal Input Settings

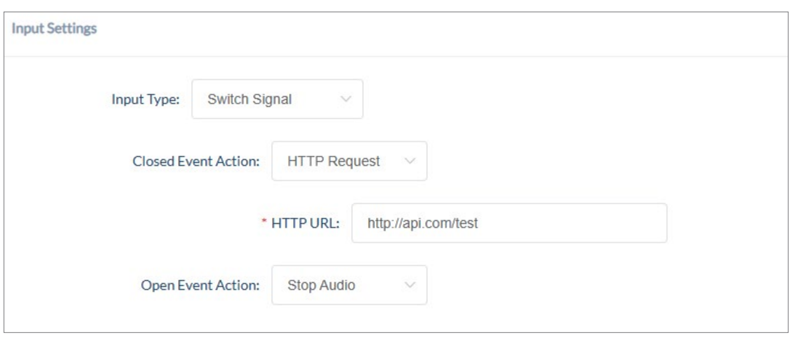
Input Type:	Choose the Key Signal or Switch Signal to trigger events.
Close/Open Event Action:	Choose different event linkage for the close/open status, including Outgoing Call, Hangout, HTTP Request, Play Audio and Stop Audio.
Outgoing Call:	Make a call to the destination.
Destination:	This setting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

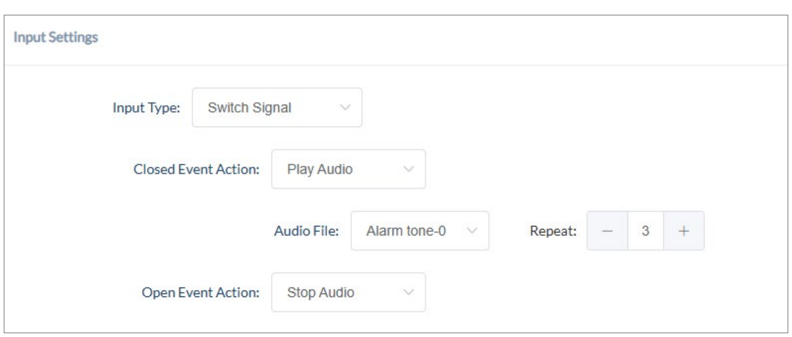
NOTE: When using the P2P line to call, please specify the device's number + IP address, such as 101@192.168.11.123.

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.



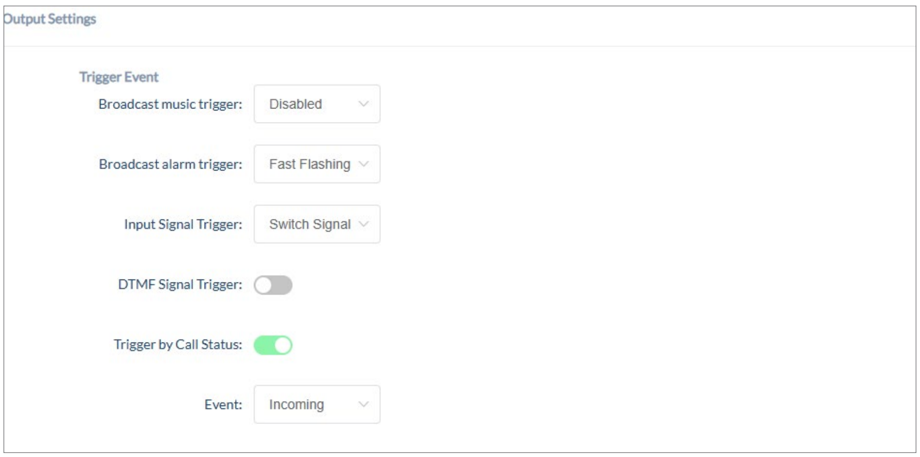






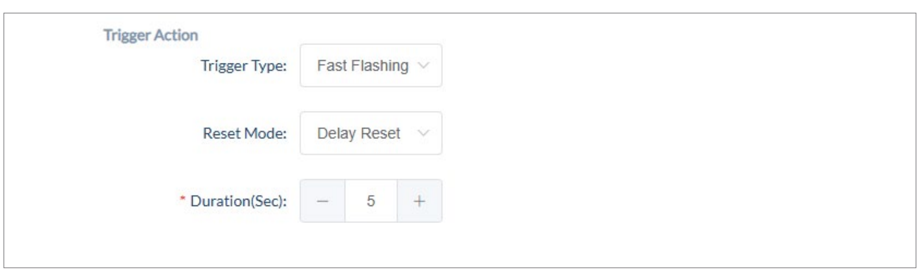
Output Trigger Event Settings

Broadcast Music Trigger:	Disabled/On/Fast Flashing/Slow Flashing, enabling this option will trigger the relay when there is broadcast music on.
Broadcast Alarm Trigger:	Disabled/On/Fast Flashing/Slow Flashing, enabling this option will trigger the relay when there is a broadcast alarm on.
Input Signal Trigger:	Enable/Disable, enable this option if needed 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, enabling 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.



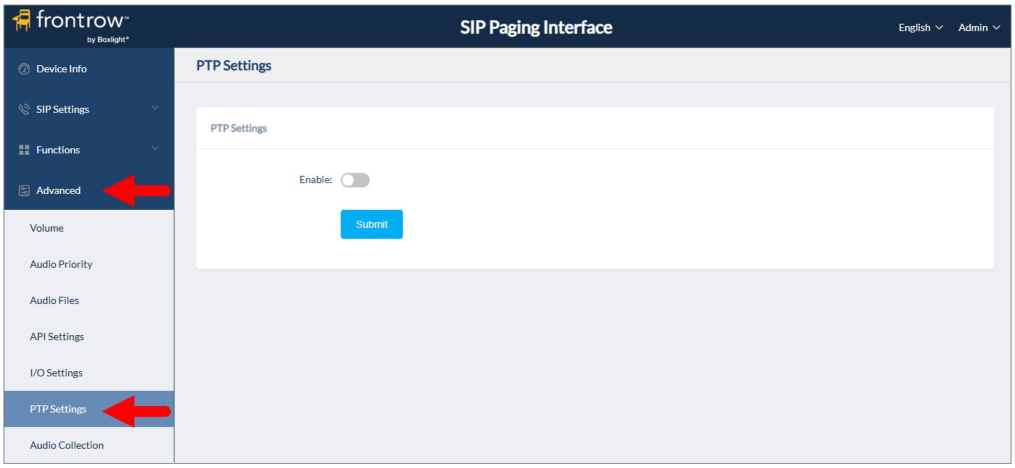
PTP Settings

NOTE: Typically, Symphony does not require these settings to be modified from their default configuration.

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 (Precision Time Protocol) is a network time protocol used to provide high-precision time synchronization.

To modify PTP Settings, select **Advance** in the left menu. Then select **PTP Settings**.

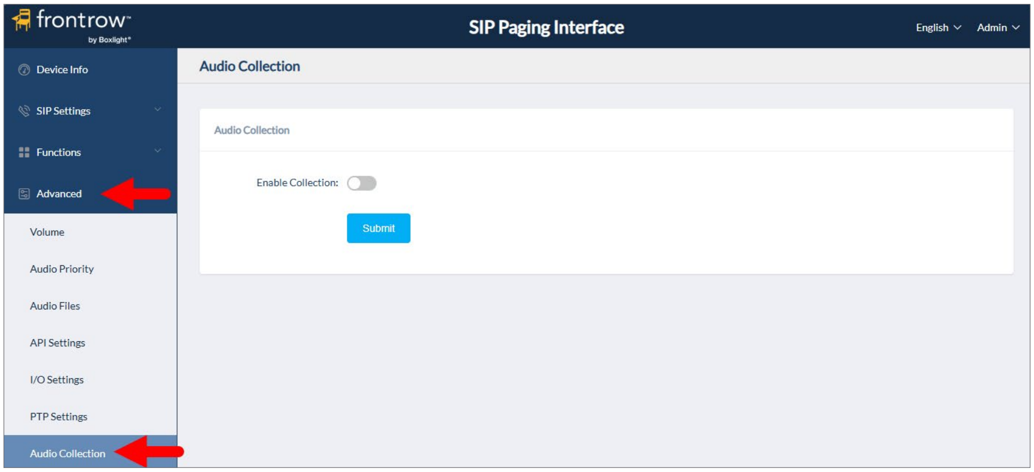


Audio Collection (SIP Paging Interface only)

NOTE: Typically, Symphony does not require these settings to be modified from their default configuration.

The Audio Collection is used to configure the parameter settings of the audio source collection function on the SIP Paging Gateway. The Audio Collection can collect external audio sources through the 3.5 Audio in the interface of the model SM Series Device and play the audio source through the paging endpoints.

To modify Audio Collections, select **Advance** in the left menu. Then select **Audio Collection**.



Audio Collection

Enable Collection: Enable the audio collection feature.

Server: IP Audio Center’s server address.

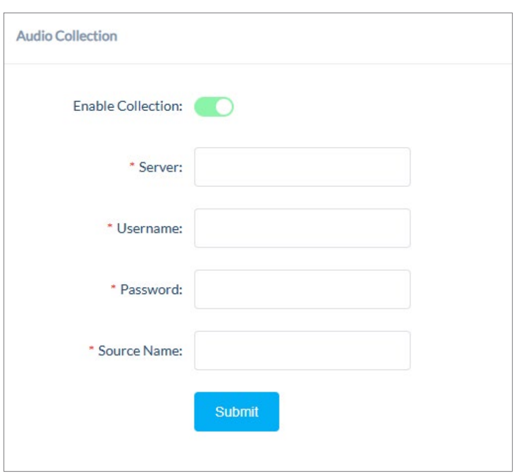
Username: The account username.

Password: The account password.

Source Name: Customize a name for the audio source (alphabet and number only).

After the relevant parameters are configured correctly, the audio source name will appear in the Playlist on the IP Audio Dispatch Console.

Name format: x10—xxx

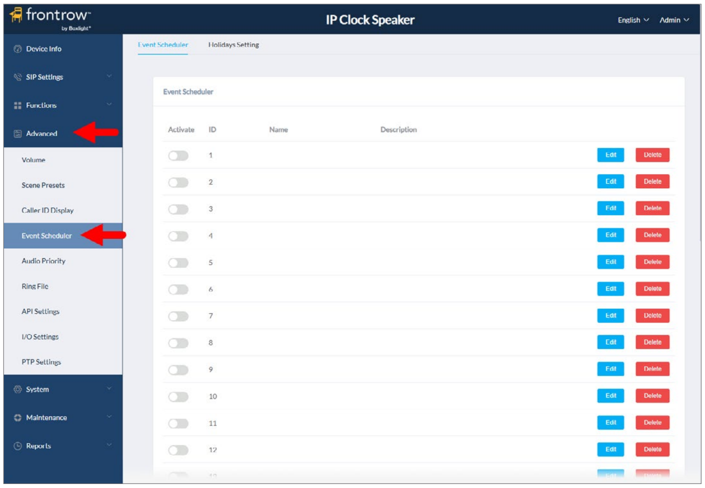


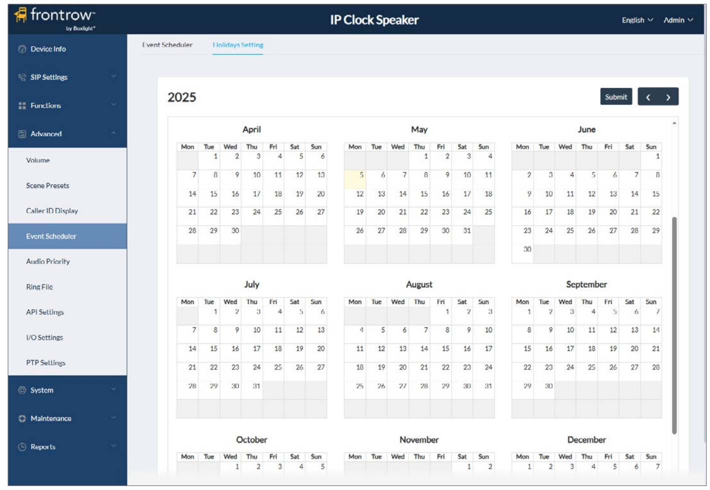
Event Scheduler – (IP Box Speaker & IP Clock Speaker)

NOTE: Typically, Symphony does not require these settings to be modified from their default configuration.

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.

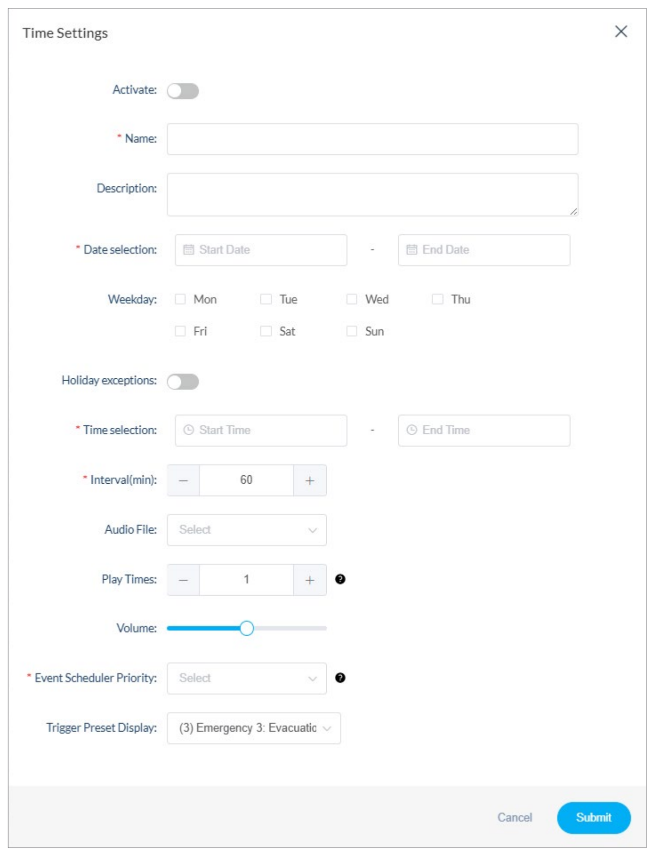
To modify Event Scheduler, select **Advance** in the left menu. Then select **Event Scheduler**.





Time Settings

Activate:	Activate/Deactivate the schedule.
Name:	Set the name of the schedule.
Description:	Description of the time schedule.
Date Selection:	Set the date range for the time schedule.
Weekday:	Set the execution weekday 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.
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 be executed first.



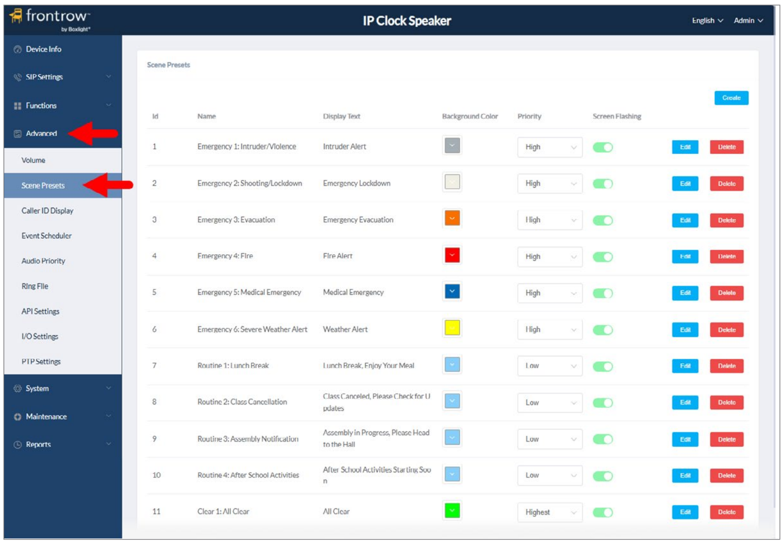
Scene Presets – (IP Clock Speaker)

NOTE: Typically, Symphony does not require these settings to be modified from their default configuration.

This page is only available on IP Speaker Clock. Users can create multiple scene presets and configure the corresponding display text. Based on different scenario needs, the relevant preset can be triggered, such as routine notifications or emergency alerts.

Scene Presets to manage the presets. The system provides several default presets for both emergency and routine scenarios, and users can customize them as needed. Presets can be triggered by buttons, incoming calls, DTMF, multicast, or event scheduler.

To modify Scene Presets, select **Advance** in the left menu. Then select **Scene Preset**.

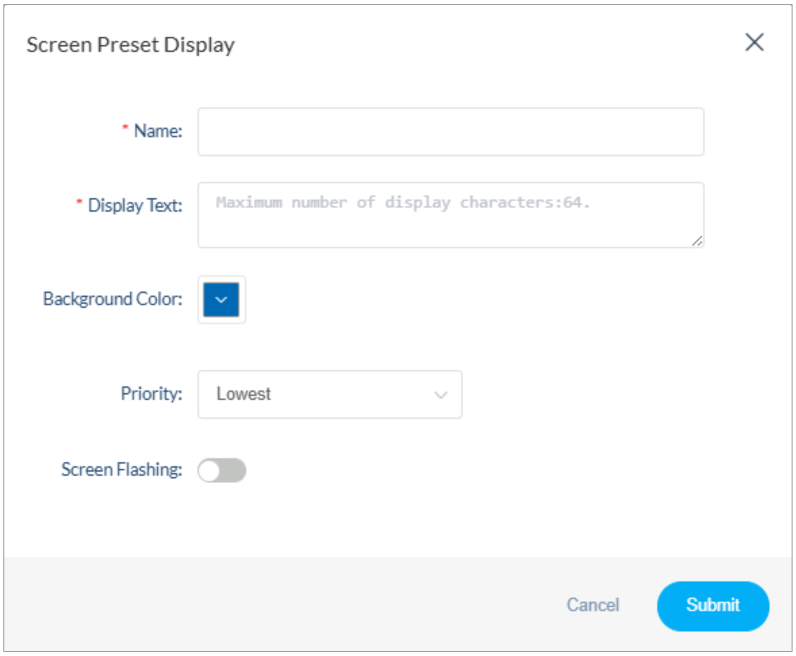


Scene Presets Settings

Create:	Create a new scene preset.
ID:	The ID of the preset.
Name:	The name of the preset.
Display Text:	The content that will display on the screen when the preset is triggered.
Background Color:	The background color of the screen when the preset is triggered.
Priority:	The priority level of the preset. There are five priority levels; When a higher-priority preset is triggered, it will override any lower-priority content currently being displayed. For presets with the same priority, the most recently triggered one will override the previous display. In the default presets, the Emergency preset is set to high priority, the Routine preset to low priority, and the Clear preset to the highest priority.
Screen Flashing:	Whether the screen will flash when the preset is triggered.
Edit:	Edit the preset.
Delete:	Delete the preset.

Create/Edit Presets

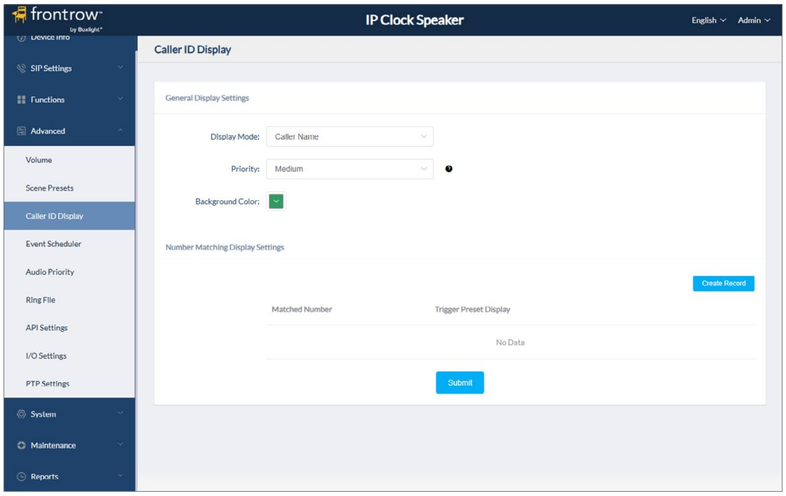
Name:	Set the name of the preset.
Display Text:	Set the content that will display on the screen when the preset is triggered.
Background Color:	Set the background color of the screen when the preset is triggered.
Priority:	Set the priority level of the preset. There are five priority levels; When a higher-priority preset is triggered, it will override any lower-priority content currently being displayed. For presets with the same priority, the most recently triggered one will override the previous display.
Screen Flashing:	Set whether the screen will flash when the preset is triggered.



Caller ID Display – (IP Clock Speaker Only)

Users can configure the screen to display caller information when there is an incoming call. They can also set up a strict number-matching list to trigger preset displays.

To modify Caller ID Display, select **Advance** in the left menu. Then select **Caller ID Display**.

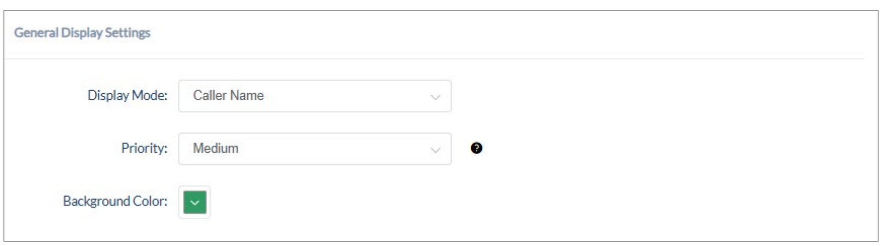


General Display Settings

Display Mode: Set the screen display mode triggered by incoming calls. Options include displaying the caller name, the caller number, or both.

Priority: This priority is used to control the execution order of the display content and follows the same priority rules as 'Scenario Presets'. For example, a SIP incoming call display with Medium priority will override the current preset display with Low priority.

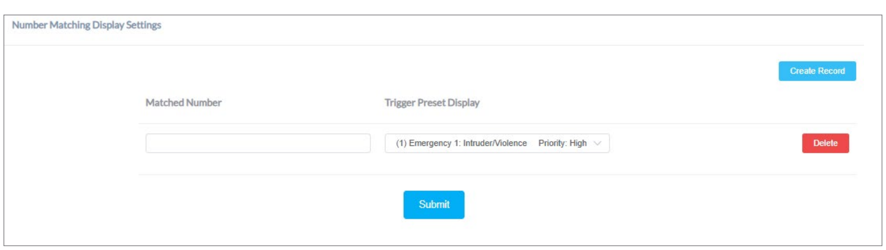
Background Color: Set the background color of the screen when an incoming call triggers the display.



Number Matching Display Settings

Matched Number: Configure the incoming call number to trigger specific display content through precise matching.

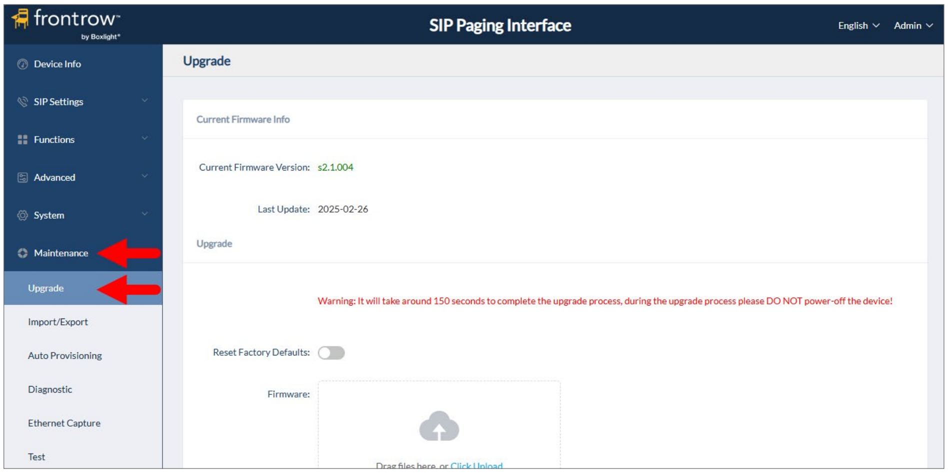
Trigger Preset Display: Select the preset to be displayed when precise matching is activated.



Maintenance

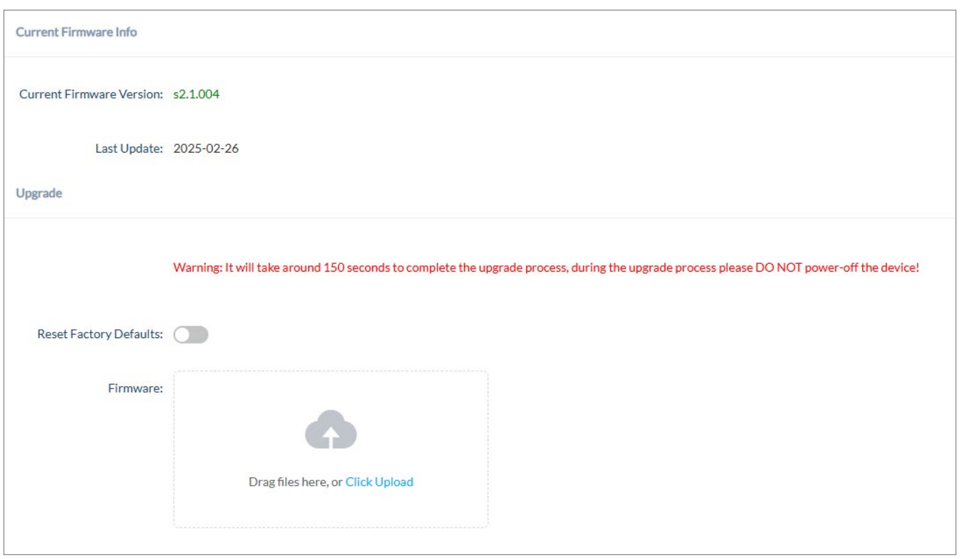
Upgrade

To update the SM Series Device firmware, select **Maintenance** in the left menu. Then select **Upgrade**.



Upgrade Settings

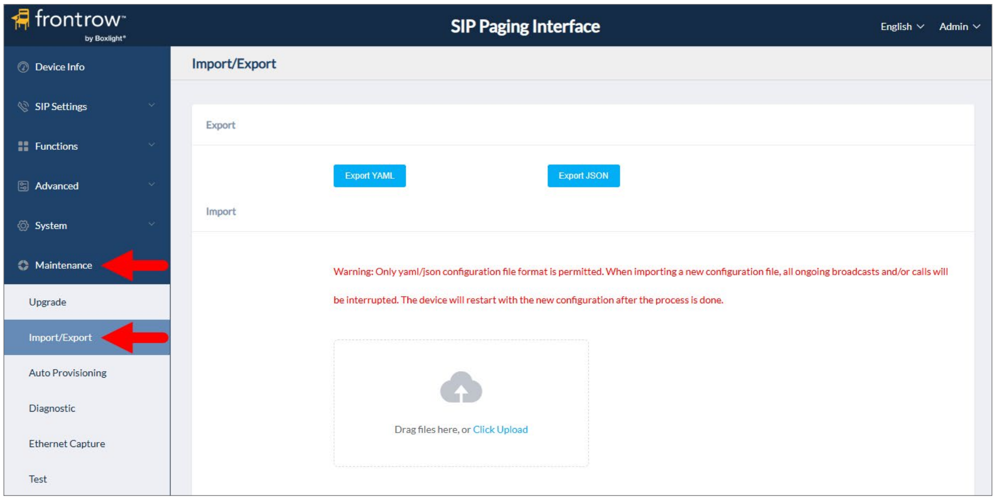
Current Firmware Version:	Displays the version currently used by the system.
Last Update:	Displays the last system updating time.
Reset Factory Defaults:	Specify whether to restore factory settings when upgrading.
Firmware:	Click to select the firmware that needs to be used to upgrade the current device.

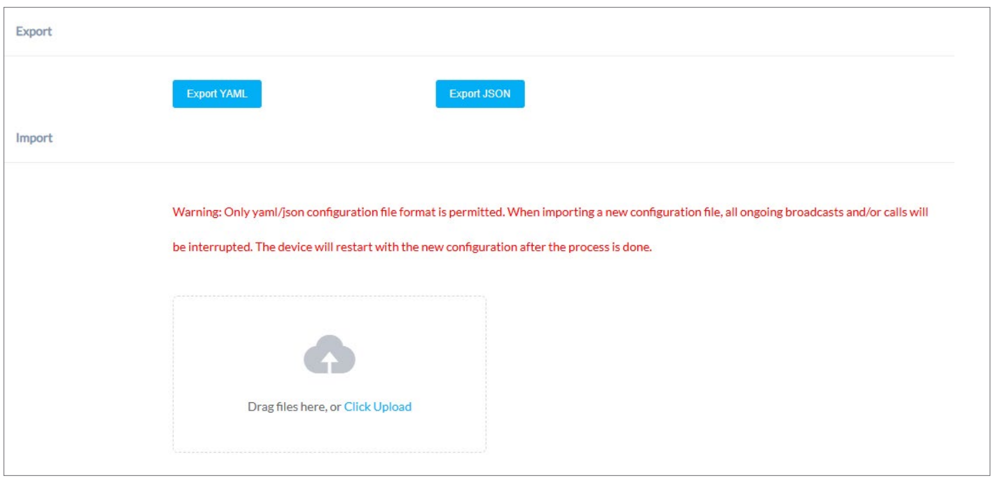


Import/Export

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.

To export or import the SM Series Device configuration, select **Maintenance** in the left menu. Then select **Import/Export**.





Auto Provisioning

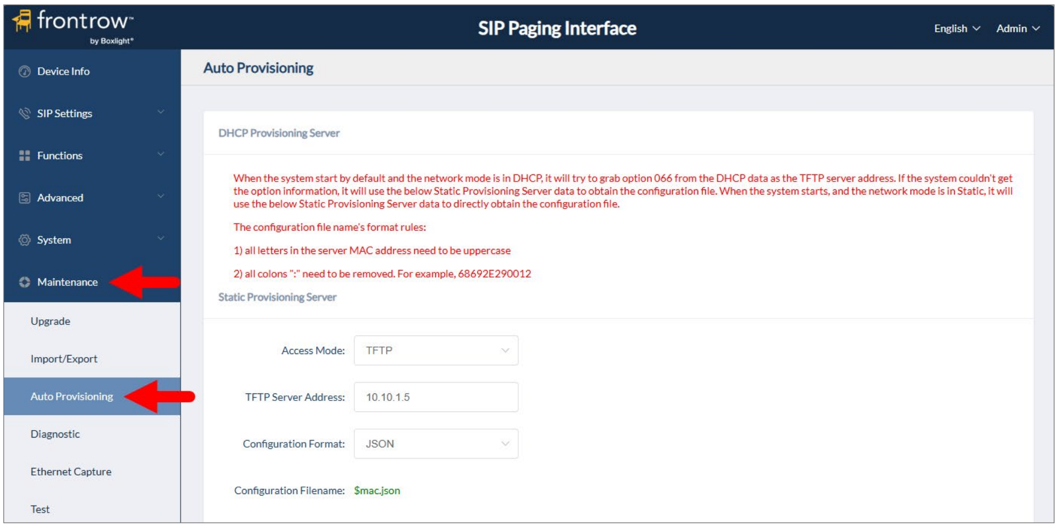
The SM Series Device supports DHCP Option 66 and static TFTP/HTTP auto provisioning methods.

When the system starts by default and the network mode is in DHCP, it will try to grab option 66 from the DHCP data as the TFTP server address. If the system does not receive the option information, it will use the 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

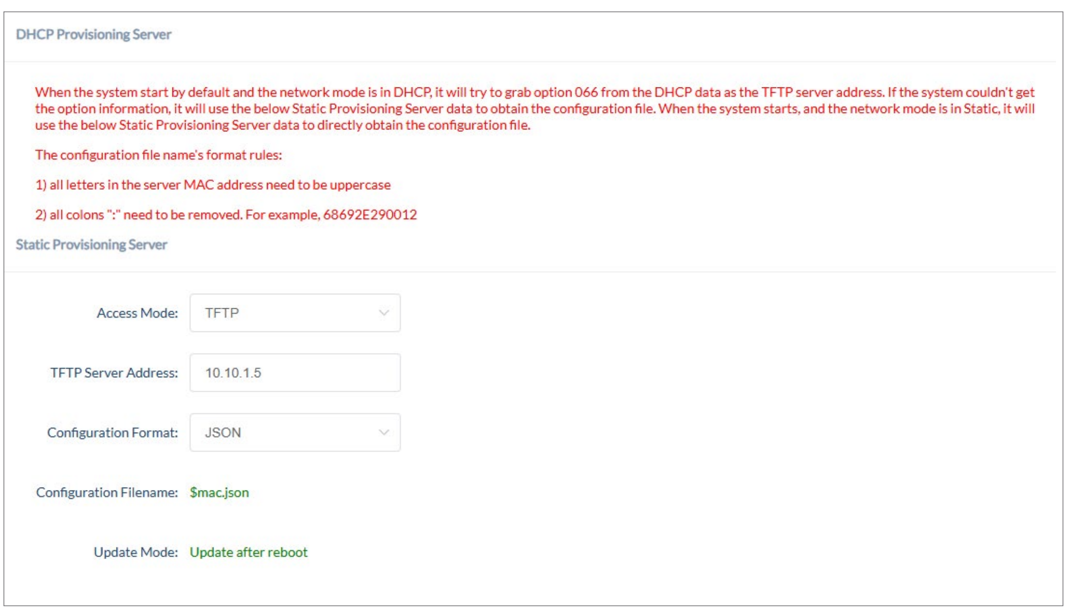
Provisioning Server data to directly obtain the configuration file.

To configure Auto Provisioning, select **Maintenance** in the left menu. Then select **Auto Provisioning**.



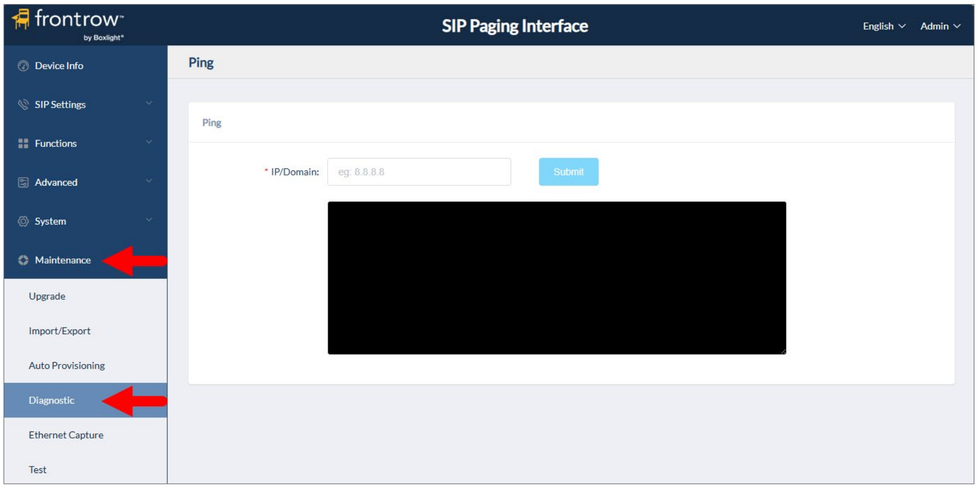
The configuration file name's format rules:

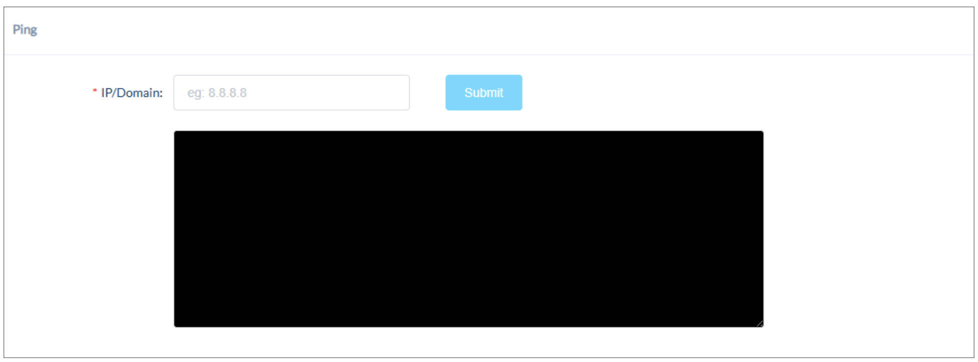
1. all letters in the server MAC address need to be **UPPERCASE**.
2. all colons ":" need to be removed. For example, 68692E290012.



Diagnostic

To access SM Series Device Diagnostics, select **Maintenance** in the left menu. Then select **Diagnostic**.

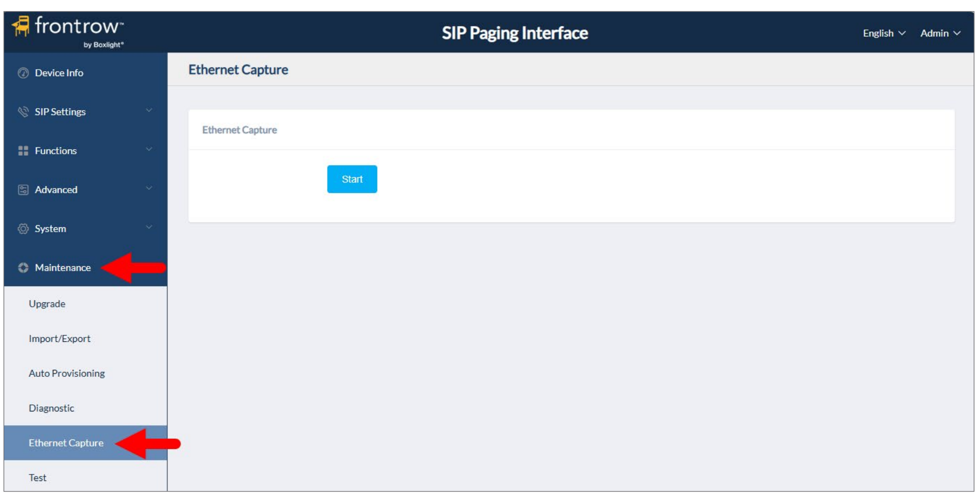


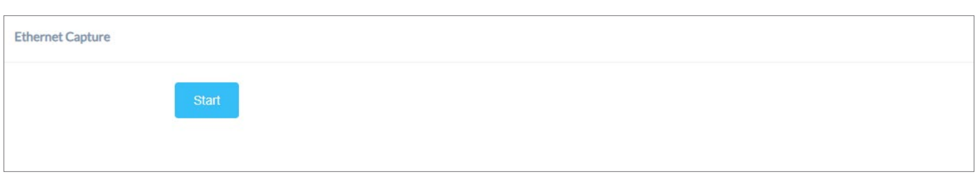


Ethernet Capture

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.

To access Ethernet Capture, select **Maintenance** in the left menu. Then select **Ethernet Capture**.

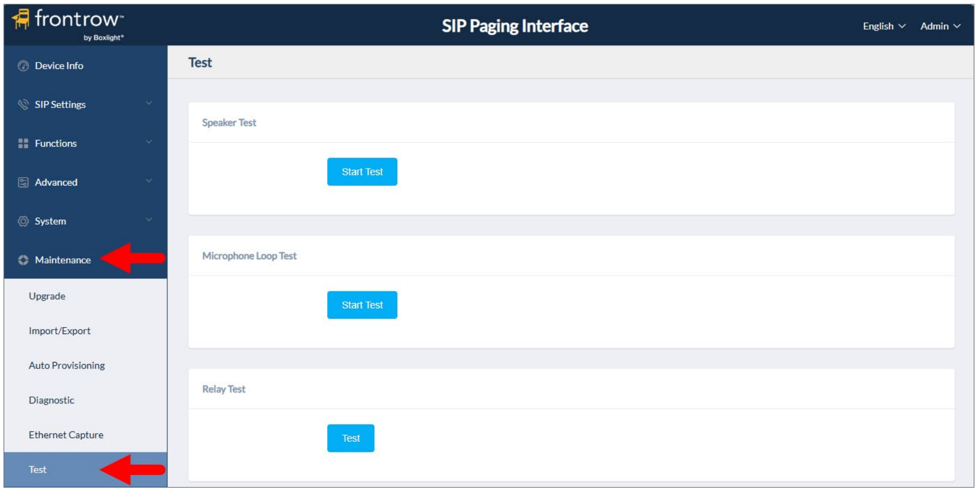




Test

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.

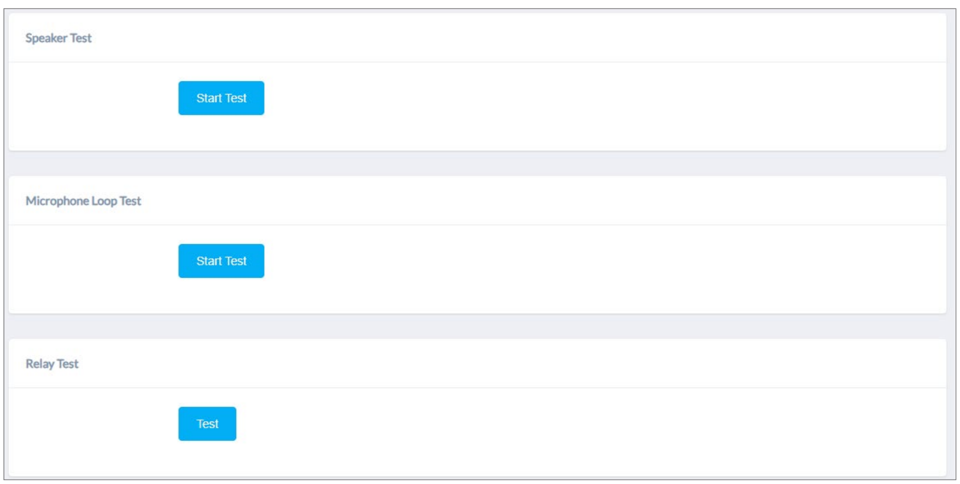
To access SM Series Device Test features, select **Maintenance** in the left menu. Then select **Test**.



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.

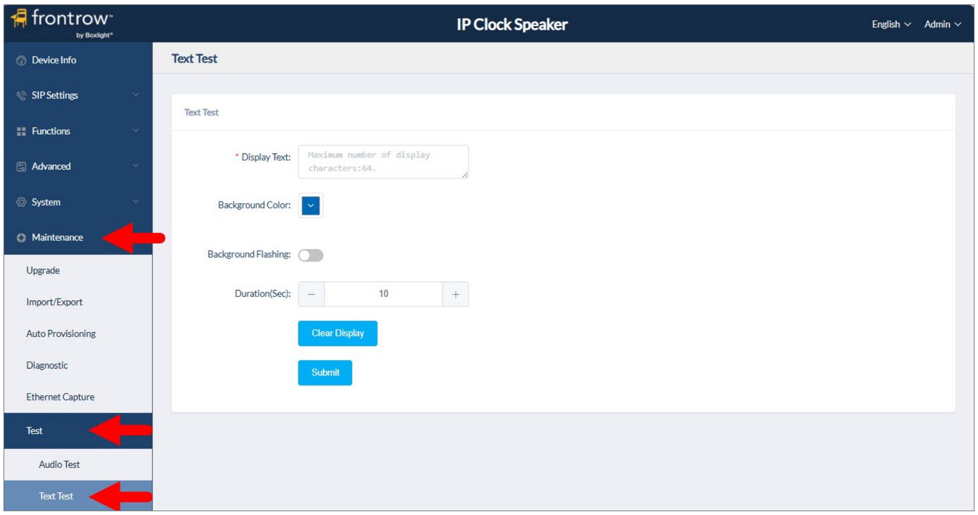
Relay Test: Click on the **Test** button and the device will output signals to the relay for testing.



Text Test – Audio Test (IP Clock Speaker)

Text Test allows the reader to visually test display text, background color, flashing and duration on the screen.

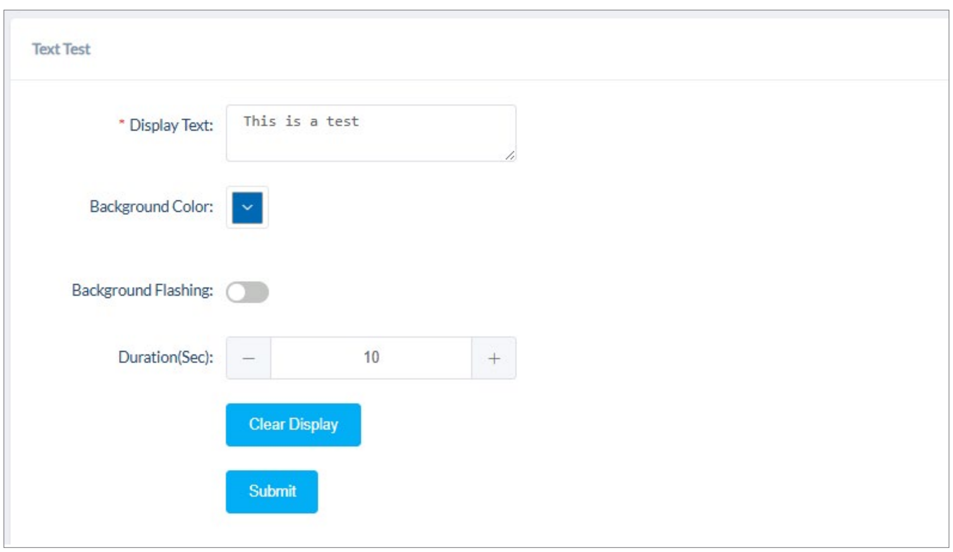
To access The IP Speaker Clock Text Test features, select **Maintenance** in the left menu. Then select **Test** and finally **Text Test**.



Text Test Settings

Display Text:	Set the screen display text for preview.
Background Color:	Set the screen display background color for preview.
Background Flashing:	Enable or disable flashing.
Duration (Sec):	Set the preview duration

IP Speaker Clock Text Test Screen

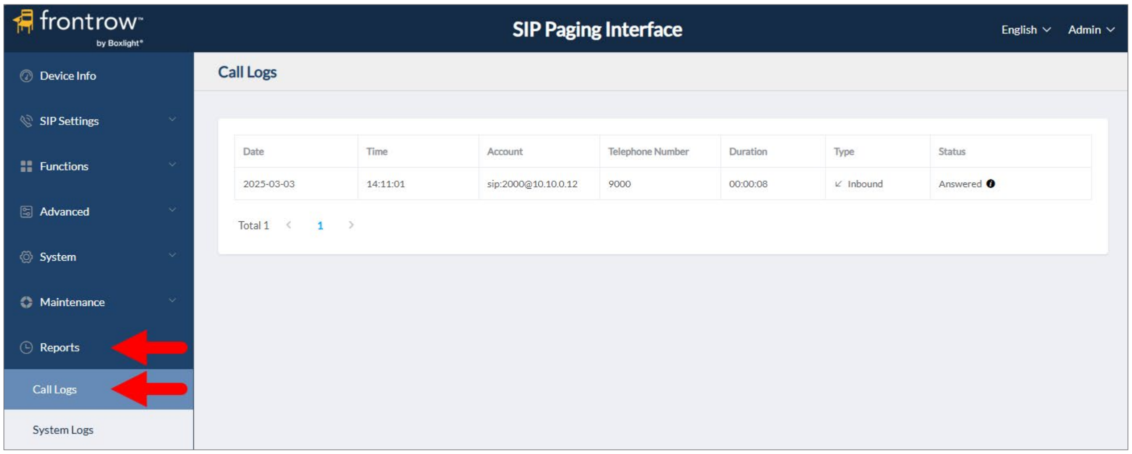


Reports

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.

To view the Call Log, select **System** in the left menu. Then select **Call Logs**.

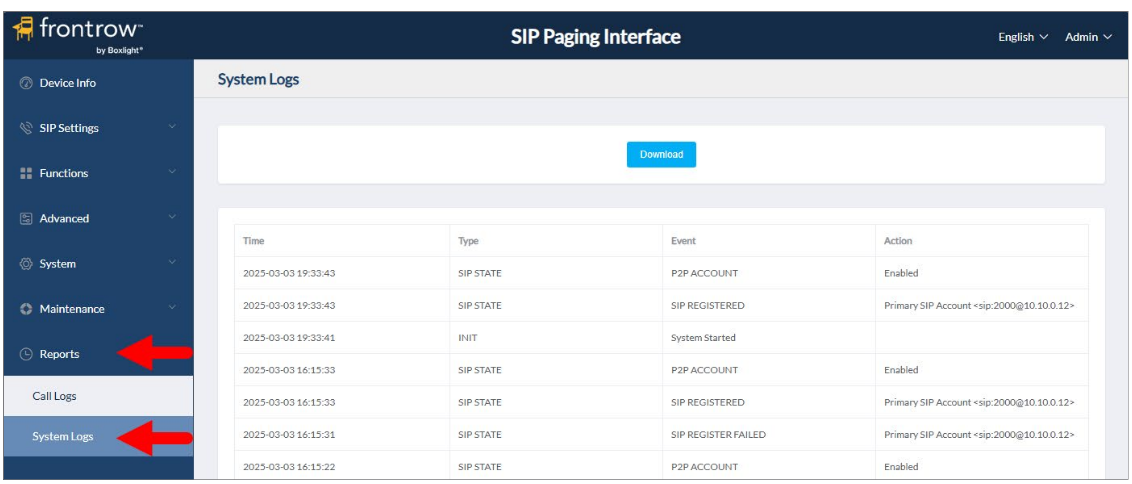


System Logs

System Logs allows you to check the event related information such as Operating Time, Operating Type (MQTT, Function, SIP, Multicast...), Event and Action details.

To view or download the Call Log, select **System** in the left menu. Then select **System Logs**.

Click the **Download** button and the .csv log file will be saved on your computer.



Appendix

P2P Account Settings

P2P stands for Peer to Peer. In a P2P network, peers are connected to each other via the network, peers can call each other directly between systems on the network without the need for a SIP server.

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.

Configuring P2P

1. In the left menu, select **SIP Settings**.
2. Select **P2P Account**.

P2P Account page

3. Optional – Set the User Authentication field to on (Slide to the right.).
4. Optional – Set Auto Answer to **Yes**.
5. Set the Activate field to on. (Slide to the right)
6. Click the **Submit** button.