SQ10 Series Network Square Speaker

User Guide

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Contents

1. Preface

1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and manage the SQ10 Series Network Square Speaker. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the SQ10 Series Network Square Speaker.

1.2 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
2.1.2	2.1.2 (hardware version 2.0 or above ONLY)	Updated operating instructions for software version v2.1.2.	Jul, 2025
2.1.0	2.1.0 (hardware version 2.0 or above ONLY)	Updated operating instructions for software version v2.1.0.	Nov, 2024
1.0.6	1.0.6	Updated operating instructions for software version v1.0.6.	Dec, 2023

2. Overview

2.1 Product Overview

SQ10 Network Square Speaker is an indoor wall-mounted IP speaker, available in two models: SQ10-B and SQ10-T. Both models are equipped with a built-in microphone, allowing operators to engage in two-way communication. Additionally, SQ10-T model features a programmable LCD screen that displays the date and time when in standby mode, eliminating the need for a separate clock system. SQ10-T also supports customizable text display for daily notifications, announcements, and alerts in emergencies. Its display-flashing feature enhances visibility, ensuring that important information is noticed promptly. This makes it ideal for use in business, education, healthcare, and other scenarios.

2.2 Product Specifications

SQ10 Network Square Speaker Specifications	
Speaker Size	4.5 inch
Sensitivity	91dB / 1W / 1m
Max Sound Pressure Level	101dB
Rated Power	8Ω 10W
Frequency Range	70Hz~20KHz
Coverage pattern	90°H 50°V 30 ㎡
Amplifier	Built-in Class D Amplifier

3. Login the Device

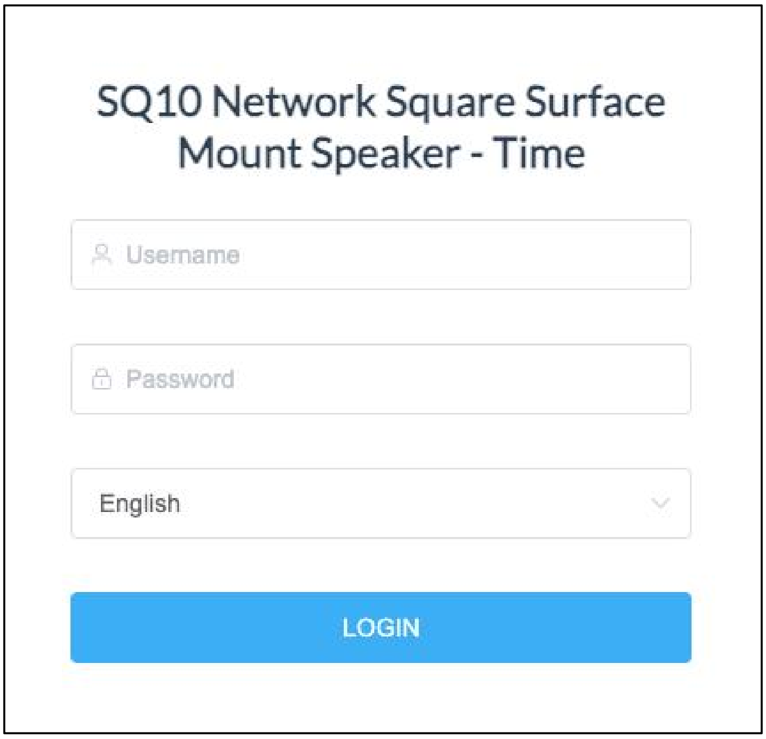
3.1 Accessing the Web GUI

SQ10 obtains the IP address through DHCP by default, please ensure that there is an available DHCP server in your LAN (If DHCP fails to obtain an address, it will use a static IP address: 192.168.1.101). 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.

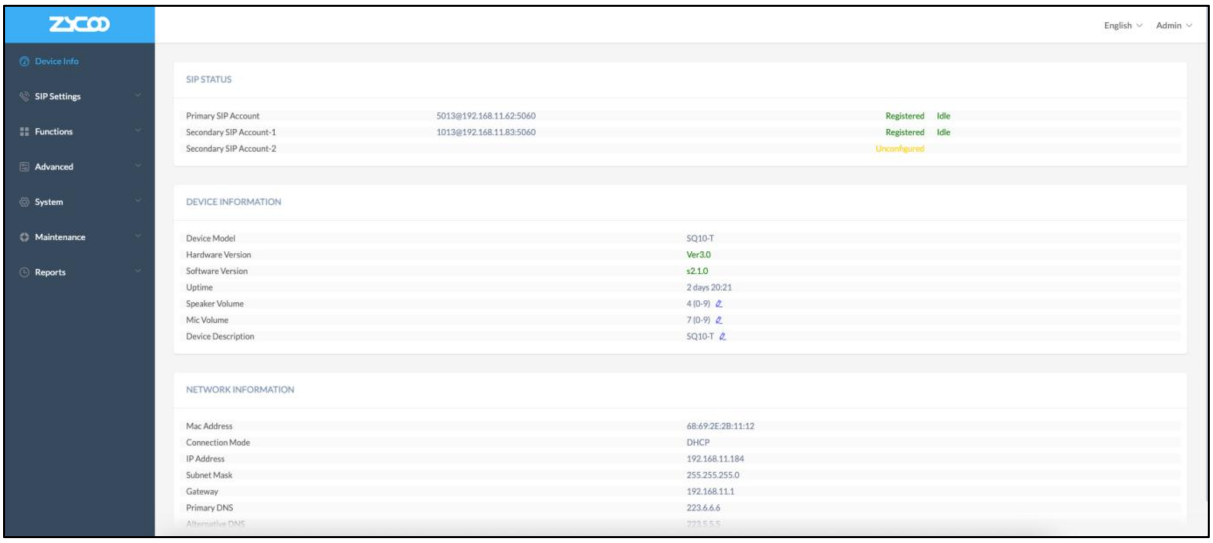


Login Interface

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.

DEVICE INFORMATION	
Device Model	SQ10-T
Hardware Version	Ver3.0
Software Version	s2.1.0
Uptime	2 days 20:21
Speaker Volume	4 (0-9)
Mic Volume	7 (0-9)
Device Description	SQ10-T

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

NETWORK INFORMATION	
Mac Address	68:69:2E:2B:11:12
Connection Mode	DHCP
IP Address	192.168.11.184
Subnet Mask	255.255.255.0
Gateway	192.168.11.1
Primary DNS	223.6.6.6
Alternative DNS	223.5.5.5

Network Information

- **Mac Address:** 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:** Displays the current IP address of the device.
- **Subnet Mask:** Displays the current subnet mask of the device.
- **Gateway:** Displays the gateway address currently used by the device.

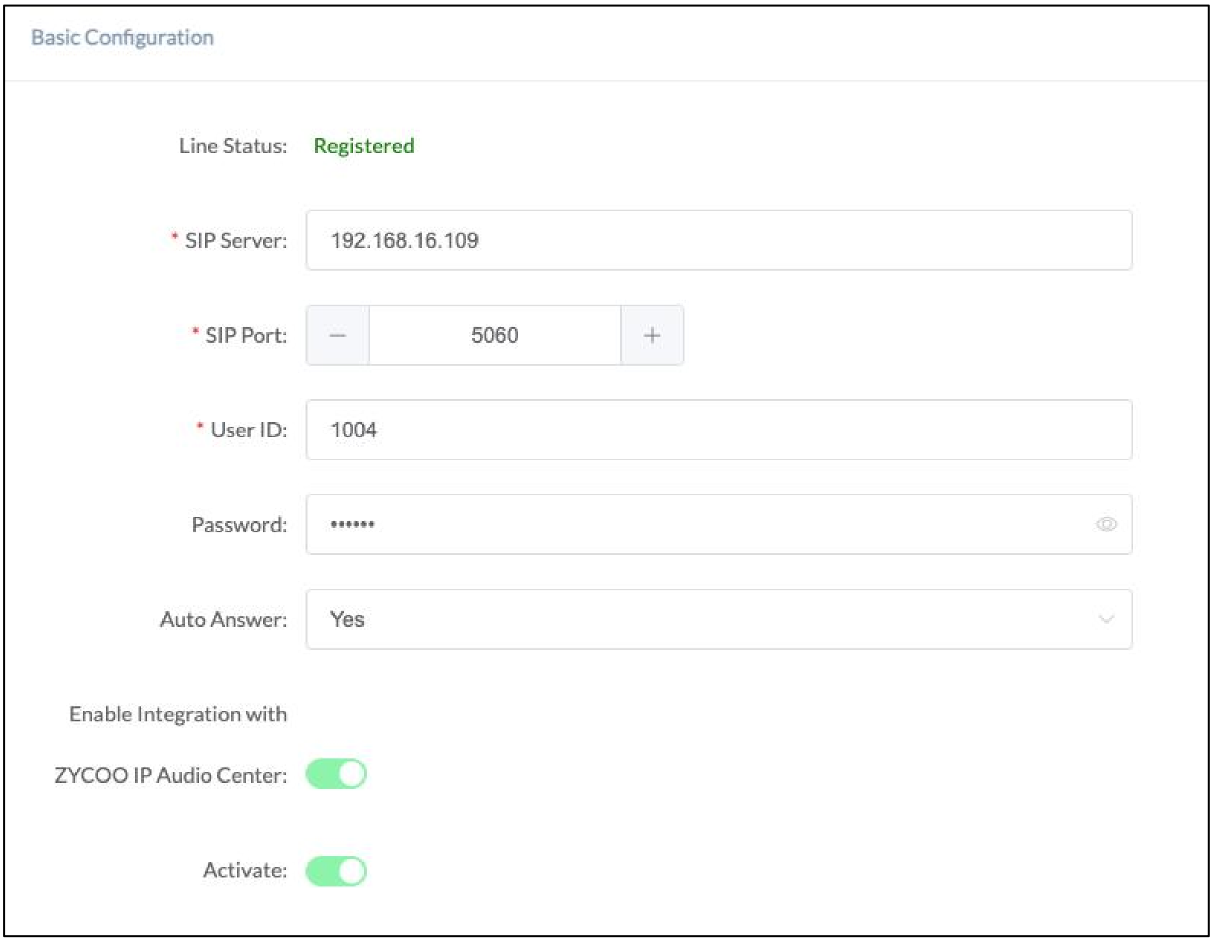
- **Primary DNS:** Displays the primary domain name server address used by the device.
- **Alternative DNS:** Displays the secondary domain name server address used by the device.

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 ZYCOO IP Audio Center' option.

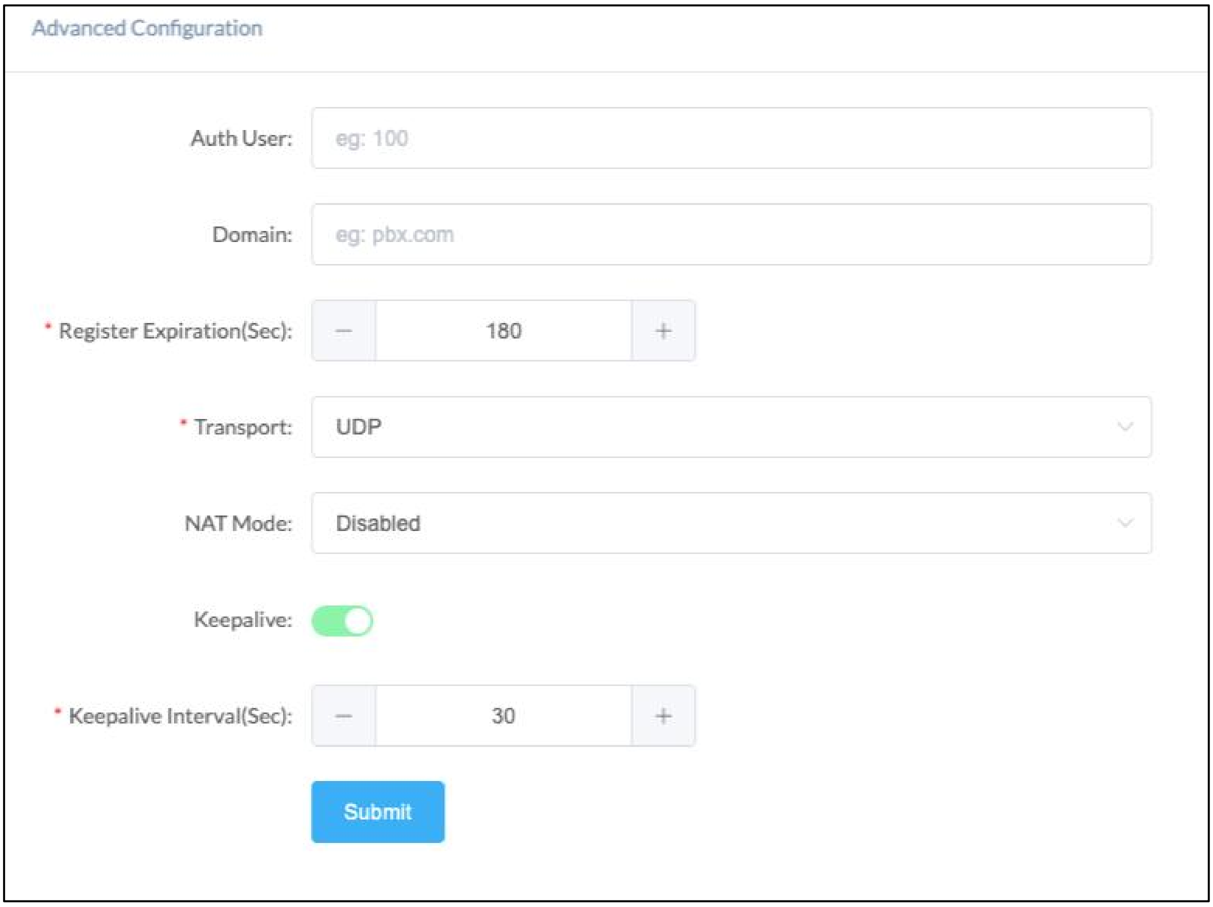
Please go to **SIP Settings --> Primary SIP Account / Secondary SIP Account-1 / Secondary SIP Account-2** page.



SIP Account - Basic Configuration

- **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 ZYCOO IP Audio Center:** Disabled by default. Enable this option when connecting to the ZYCOO IP Audio Center. This option is available only for the primary SIP account.
- **Activate:** Once enabled, the account will be activated and registered with the SIP server.



SIP Account - Advanced Configuration

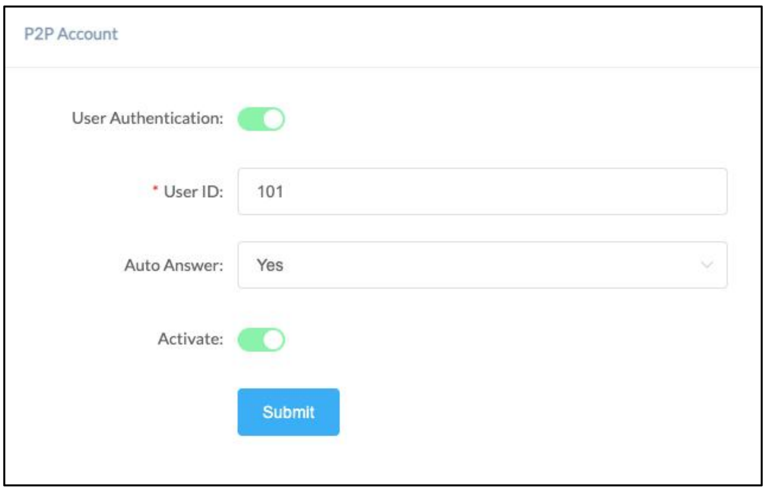
- **Auth User:** Enter the authorized username for the SIP account.

- **Domain:** Enter the SIP Domain.
- **Register Expiration (sec):** Set the SIP registration expiration time, with a default of 180 seconds.
- **Transport:** Choose the transport protocol: UDP, TCP, or TLS.
- **NAT Mode:** Select the NAT mode and provide the necessary details. Supports STUN, TURN, and ICE modes.
- **Keepalive:** Enable the SIP keepalive function to maintain an active connection.
- **Keepalive Interval(Sec):** Set the interval for SIP keepalive messages.

4.2 P2P Account Settings

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

Please go to **SIP Settings --> P2P Account Settings** page to configure the P2P settings first. After configuring the P2P account, it can be used with the Outgoing Call feature in **Basic Settings --> I/O Settings**, or use the Outgoing API in **Basic Settings ---> API Settings** to make a P2P call.



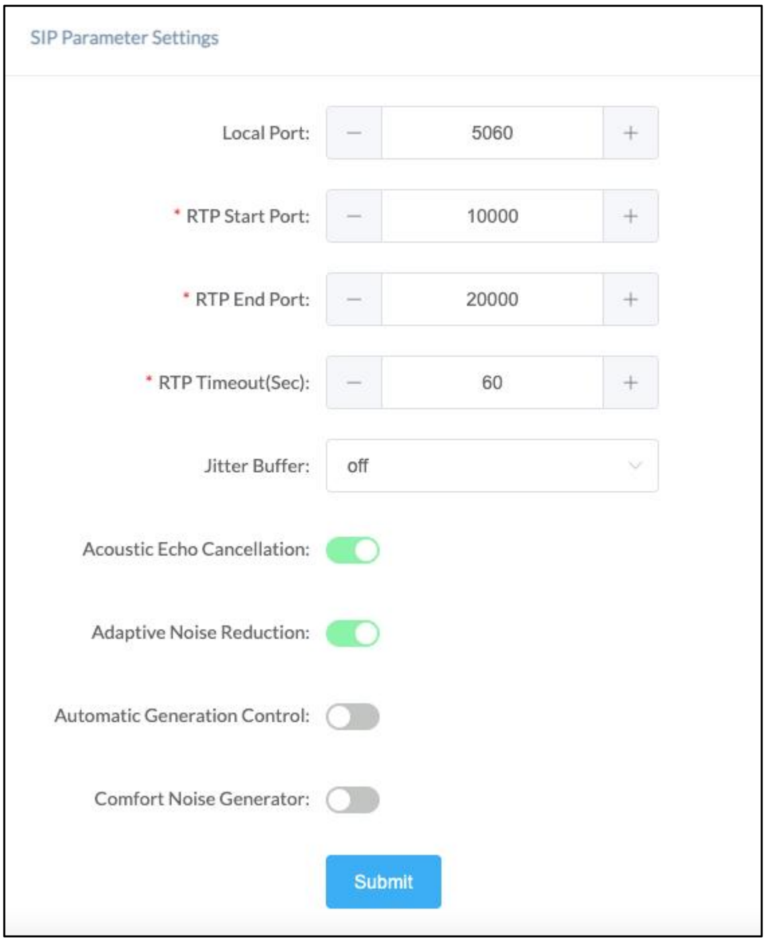
P2P Account

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

4.3 Advance SIP Settings

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

4.3.1 SIP Parameter Settings

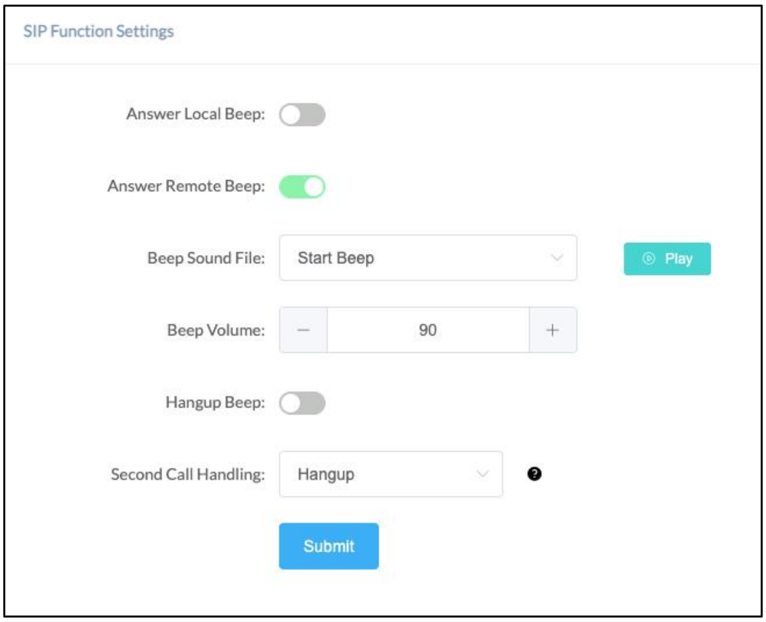


SIP Parameter Settings

- **Local Port:** This setting represents the port used to receive SIP packets.
- **RTP Start Port:** This setting represents the starting RTP port that will use for media sessions.
- **RTP End Port:** This setting represents the end RTP port that the system will use for media sessions.
- **RTP Timeout (sec):** This setting means that within a specific time range, if the system does not receive the RTP stream, the call will end.
- **Jitt Buffer:** This setting represents the Jitter buffer where voice packets can be collected, stored, and sent to the voice processor in even intervals. Three options are provided, off/adaptive/fixed. A fixed jitter buffer adds a fixed delay to voice packets. An adaptive jitter buffer can adjust based on the delays in the network.

- **Acoustic Echo Cancellation**: After enabling this feature, echo noise can be suppressed through algorithms.
- **Adaptive Noise Reduction:** After enabling this feature, algorithms can suppress environmental noise collected by microphones.
- **Automatic 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

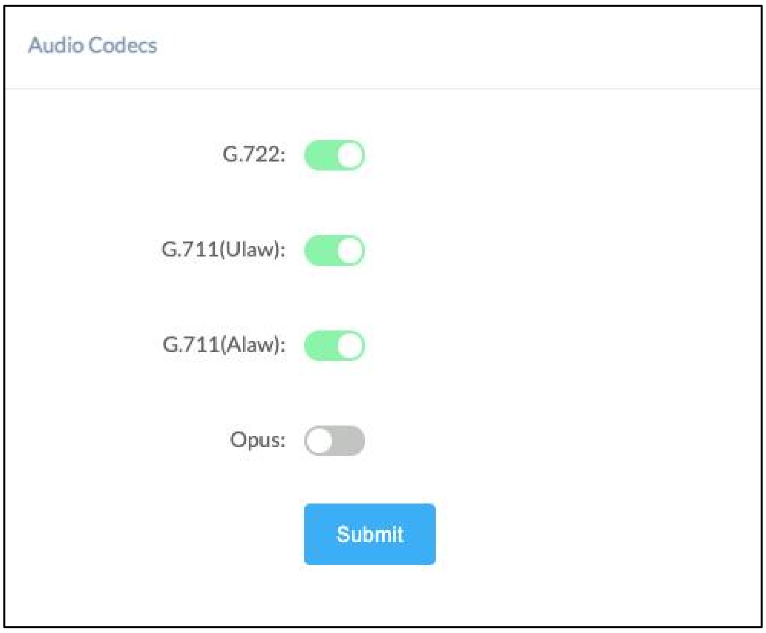
- **Answer Local Beep:** If this setting is enabled, the selected beep sound will be played first on the local device side after the SIP session is answered.
- **Beep Sound File:** Select a specific beep sound file. Click the Play button, you could listen to this audio file.
- **Beep Volume:** Set the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

SQ10 supports 4 audio codecs: G.722 (wideband codec), G.711(Ulaw), G.711(Alaw), and Opus.



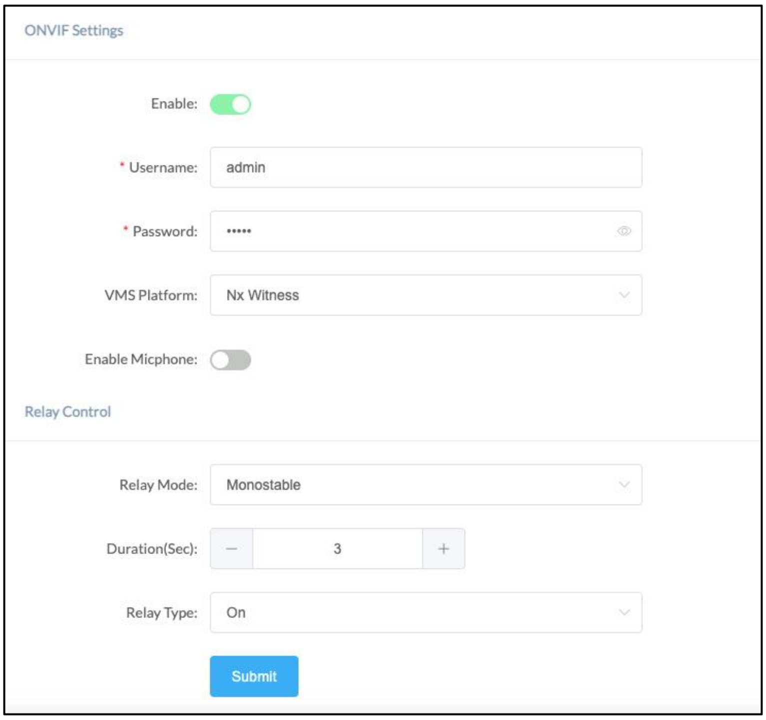
Audio Codecs

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.



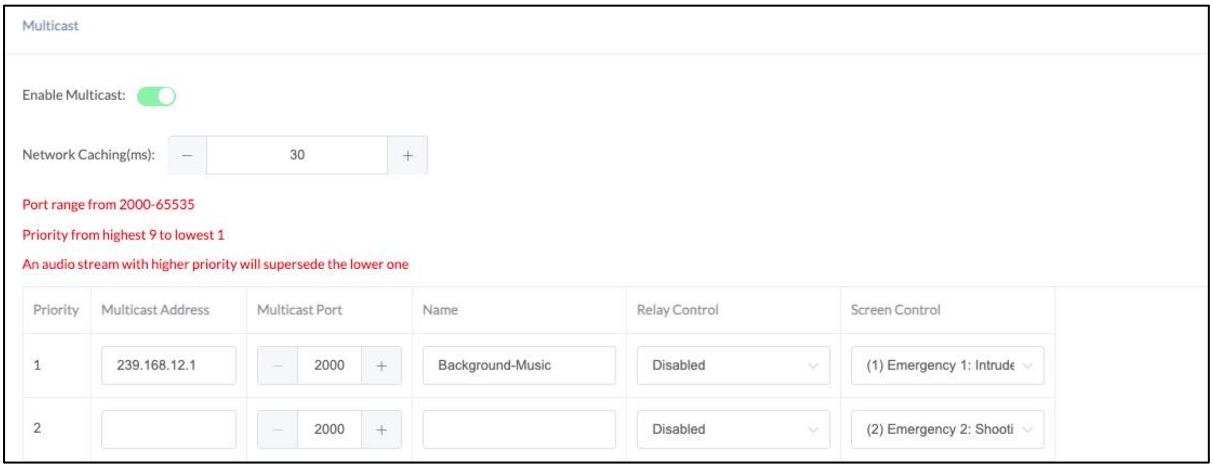
ONVIF & Relay Control Settings

- **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:** Choose a relay response to triggers: 'On', 'Fast Flashing', or 'Slow Flashing'.

5.2 Multicast

The multicast settings are used to configure the parameter settings of the multicast function. It can be configured to monitor up to 9 different levels of multicast addresses, the audio streams with a higher priority will interrupt the playback of the lower priority audio streams. Please go to **Functions ---> Multicast** page to enable the multicast feature.



Multicast

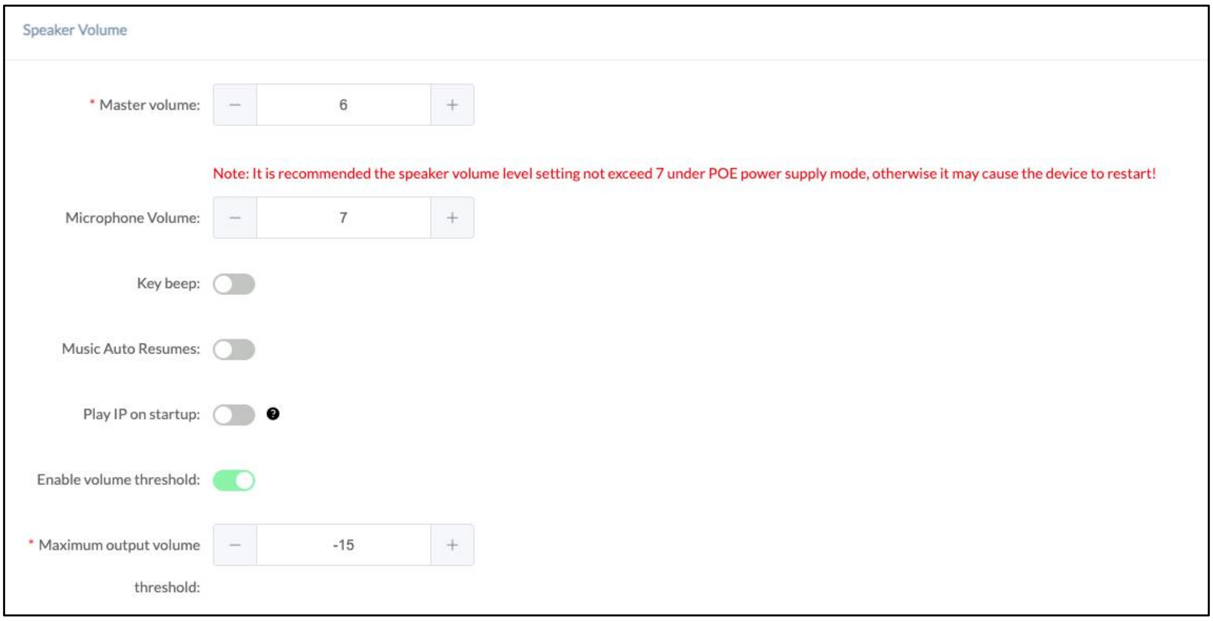
- **Priority:** Priority from highest 9 to lowest 1.
- **Multicast Address:** The multicast address range is 224.0.0.0 – 239.255.255.255.
- **Multicast Port:** The multicast port range is 2000 – 65535.
- **Name:** Customize the name of the multicast address.

- **Relay Control:** Choose a relay response to trigger: 'Disabled', 'On', 'Fast Flashing', or 'Slow Flashing'.
- **Screen Control:** Choose a scene preset to trigger.

6. Advanced Settings

6.1 Volume Settings

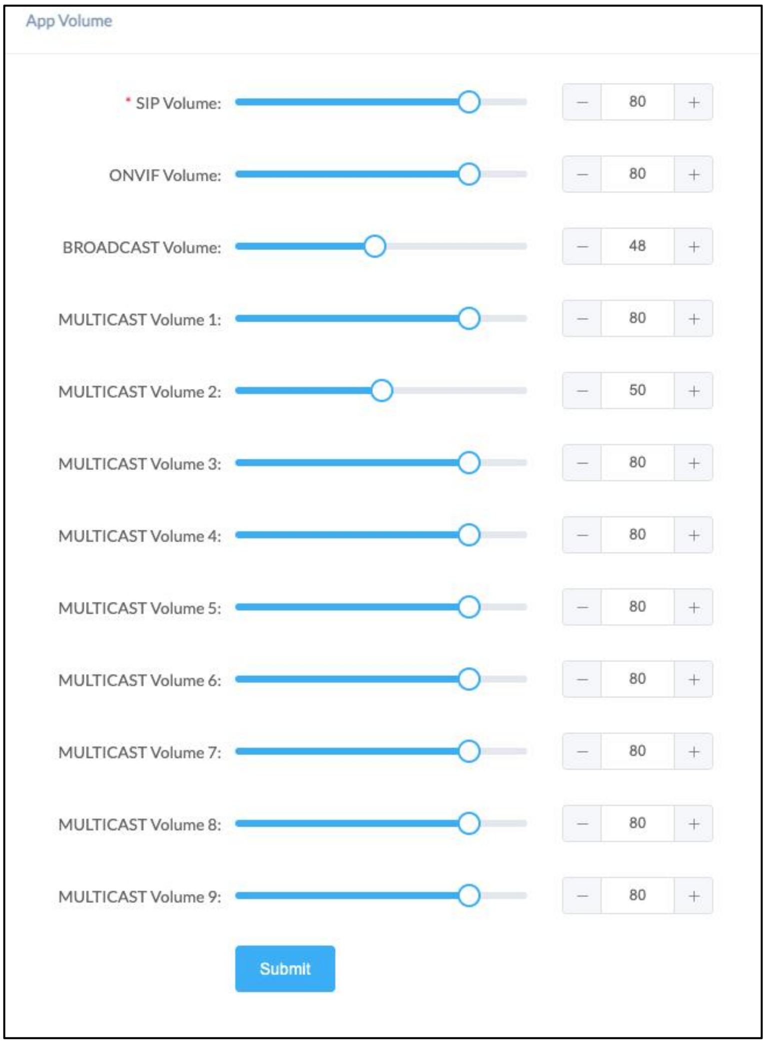
To set the volume, please go to **Advanced --> Volume --> Local Settings** page to configure. You can set the Main Volume as well as the individual volume levels for each application (e.g. SIP, multicast, ONVIF...), and you can also configure Remote Volume Control settings on this page.



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.

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



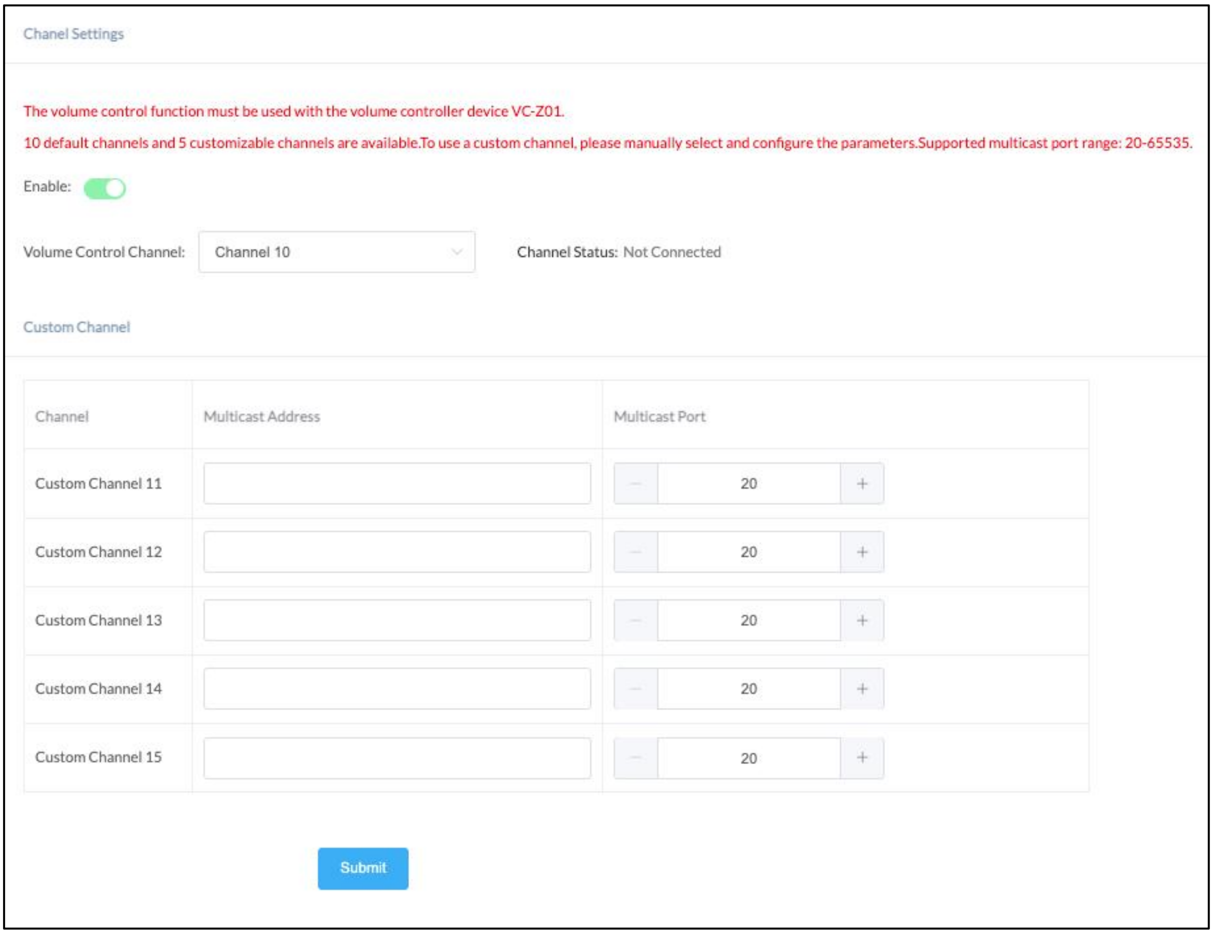
App Volume Settings

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

- **Multicast Volume 4:** Set the specific volume for multicast 4 application.
- **Multicast Volume 5:** Set the specific volume for multicast 5 application.
- **Multicast Volume 6:** Set the specific volume for multicast 6 application.
- **Multicast Volume 7:** Set the specific volume for multicast 7 application.
- **Multicast Volume 8:** Set the specific volume for multicast 8 application.
- **Multicast Volume 9:** Set the specific volume for multicast 9 application.

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 → Volume → 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.

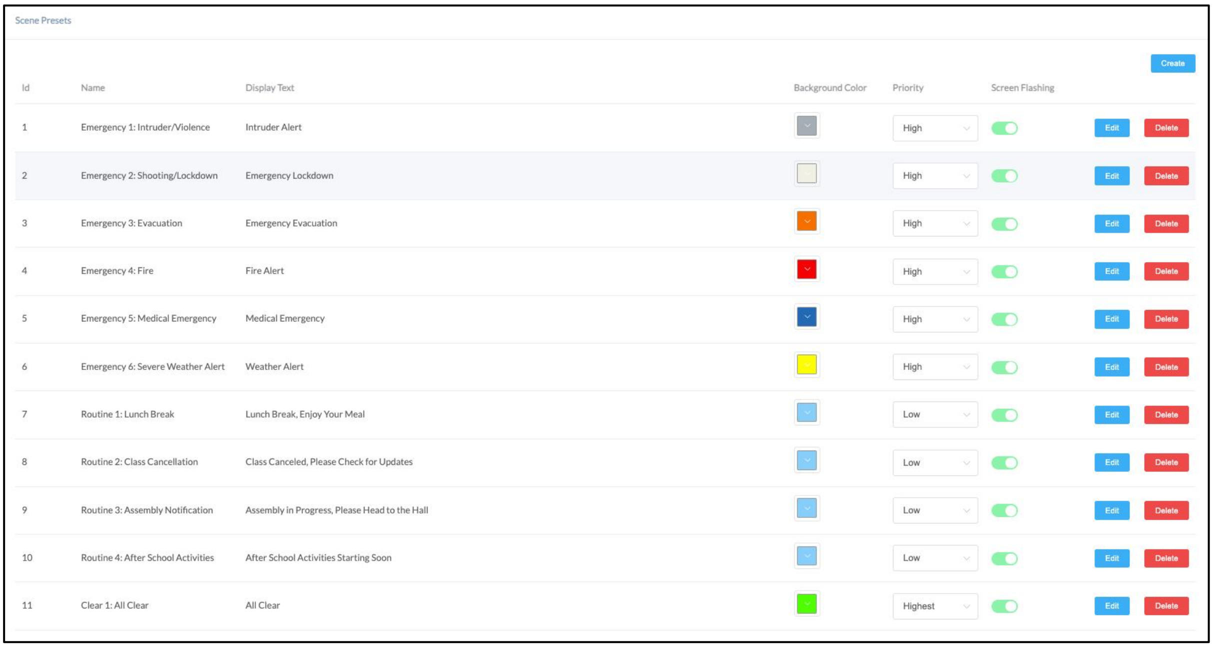
- **Volume Control Channel:** Provides 10 default channels (Channel 1 – 10), and 5 customizable channels (Channel 11 – 15).
- **Channel Status:** Displays the connection status between the VC-Z01 and the endpoint device. Connected: The devices are successfully paired and remote control is available. 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 (SQ10-T Only)

This page is only available on SQ10-T. 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. Please go to **Advanced ---> Scene Presets** to manage the presets.

The system provides few 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, event scheduler, or through the background music/alarm function of the ZYCOO IP Audio Dispatch Console**.



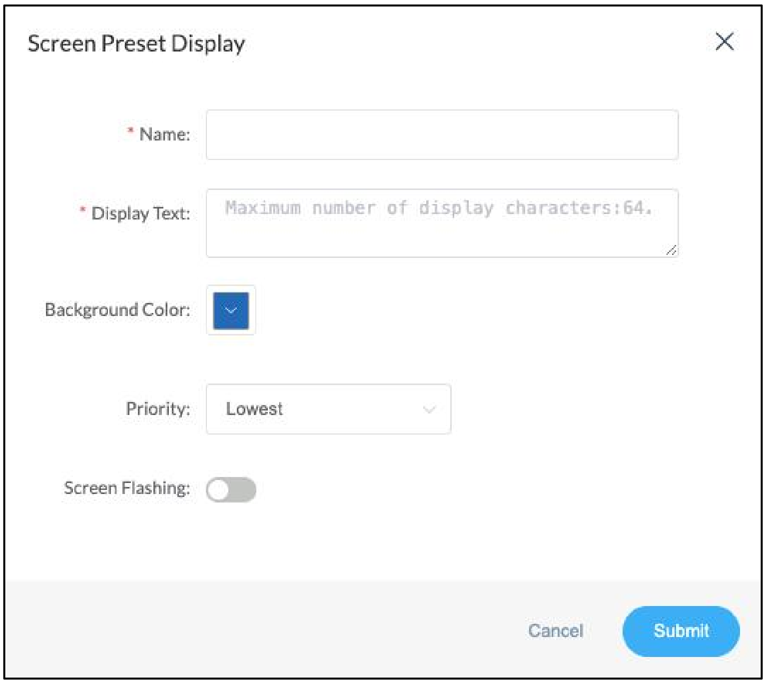
Scene Presets

Create

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

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.

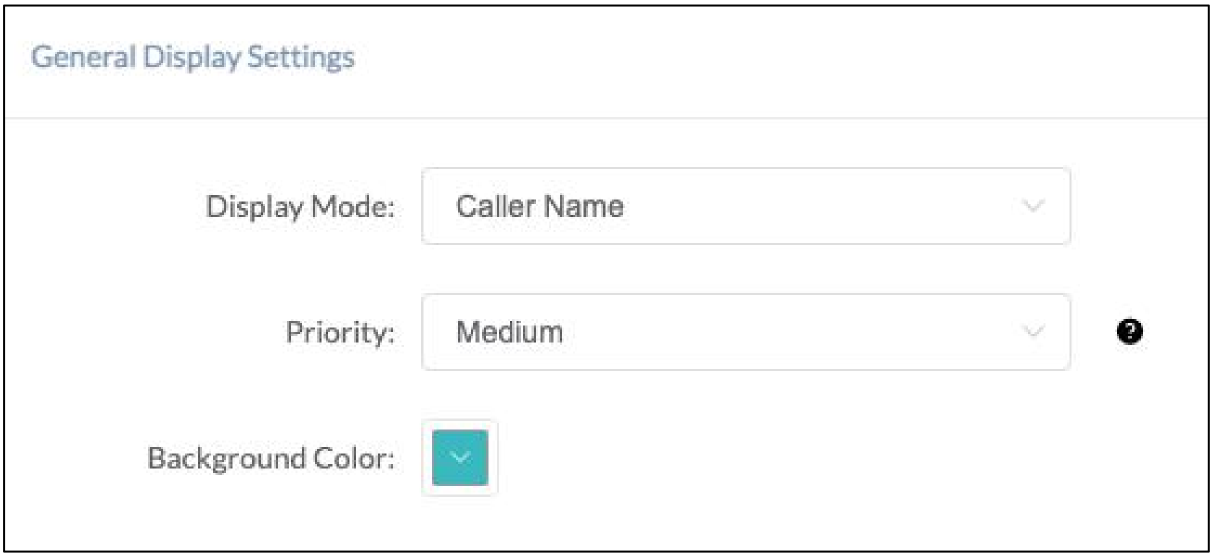
6.3 Caller ID Display (SQ10-T Only)

This page is only available on SQ10-T.

Users can configure the screen to display caller information when there is an incoming call.

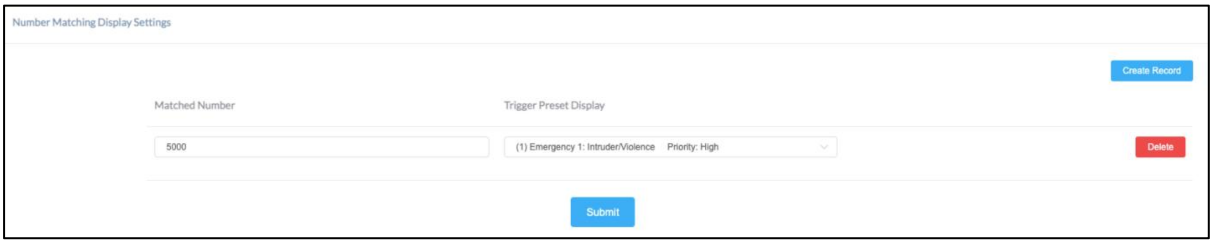
They can also set up a strict number-matching list to trigger preset displays.

Please go to **Advanced ---> Caller ID Display** to configure.



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.

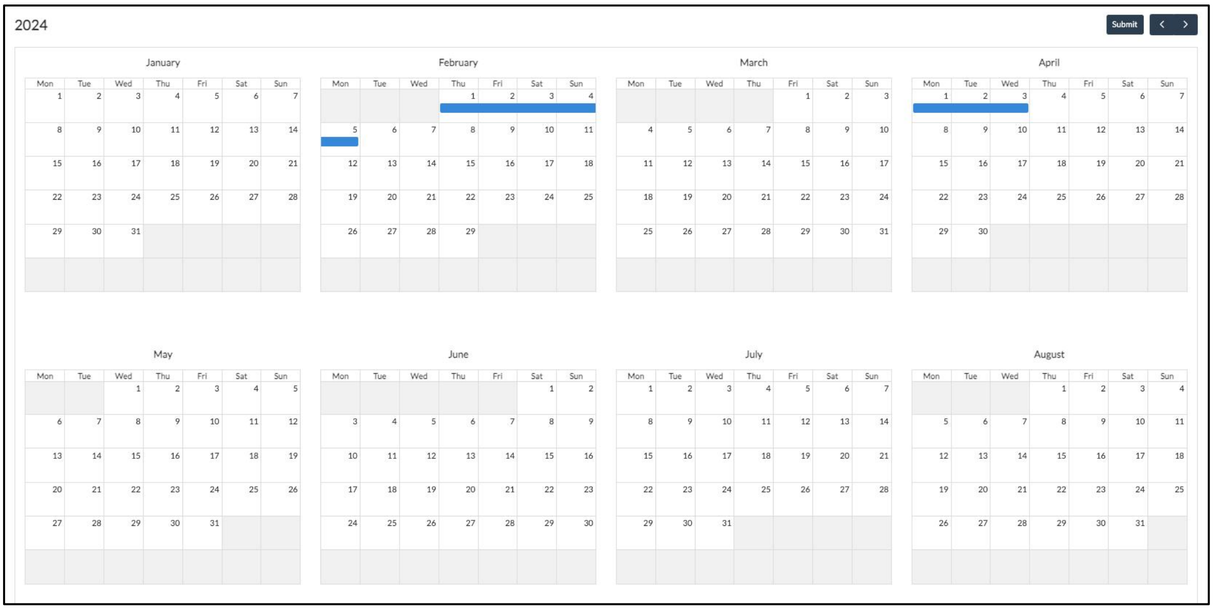
- **Trigger Preset Display:** Select the preset to be displayed when precise matching is activated.

6.4 Event Scheduler

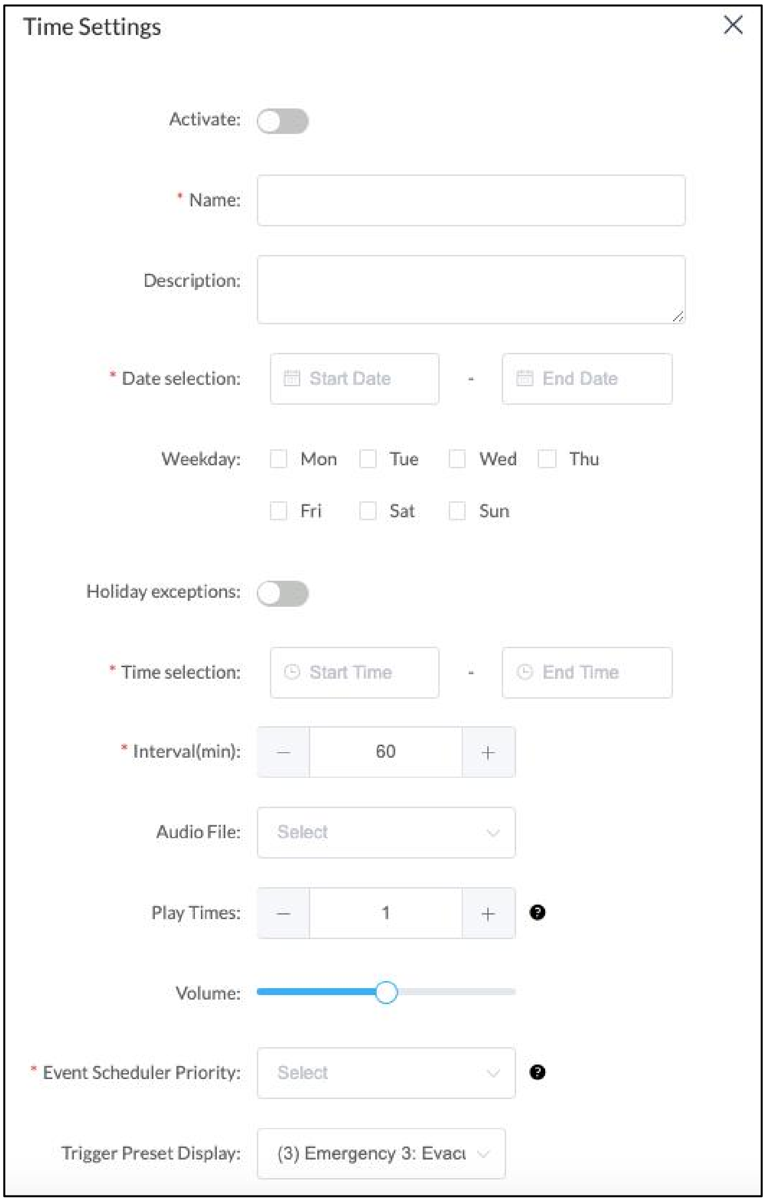
The event scheduler can edit up to 30 time plans, you can click the corresponding option to edit or delete them. Before you edit/create an event scheduler. The schedules in the holiday setting will not be executed.

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.
- **Trigger Preset Display (SQ10-T Only):** Select the preset to activate upon trigger.

6.5 Audio Priority Settings

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

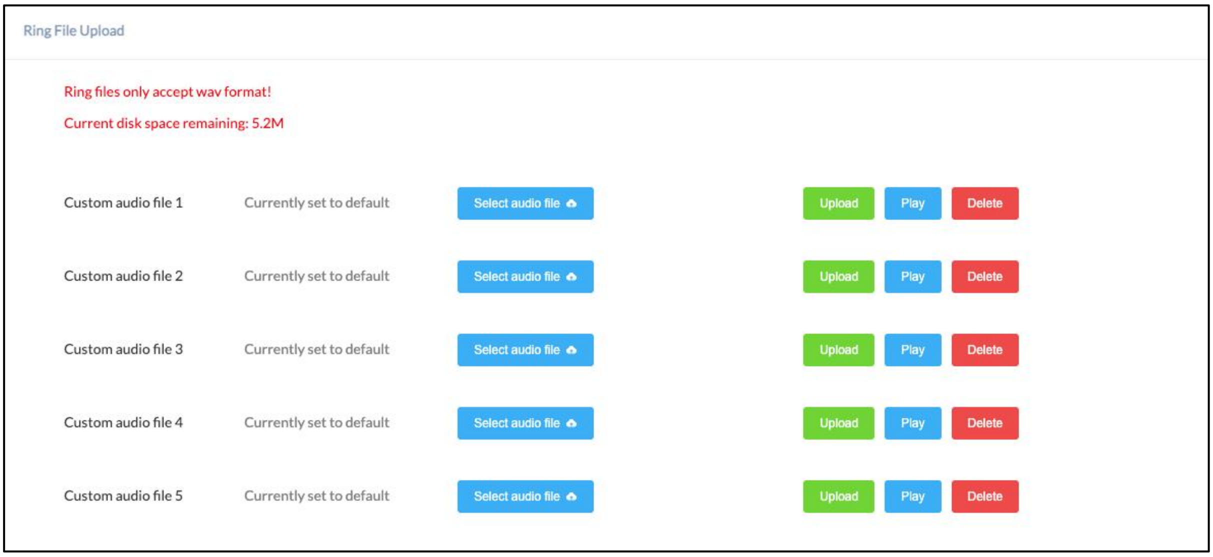
Priority 1 is the highest. You can drag the arrow on the right side to adjust the priority. The execution of a high-priority audio application will interrupt the current low-priority audio application.

Audio Priority Settings

6.6 Ring File

Ring 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 ---> Ring File** to manage the audio files.

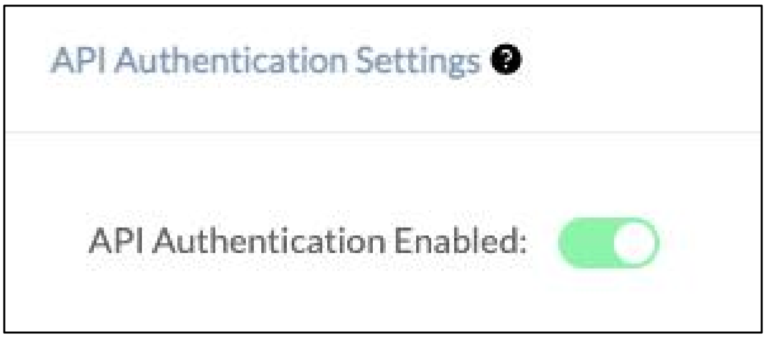


Ring Files

6.7 API Settings

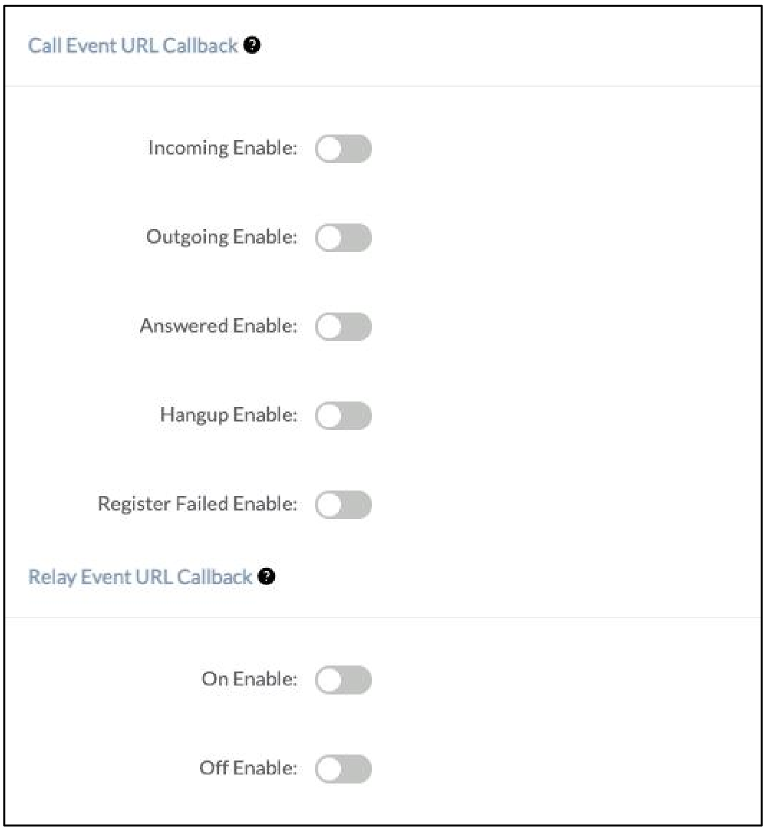
This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the changing status of the call and/or relay.

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.

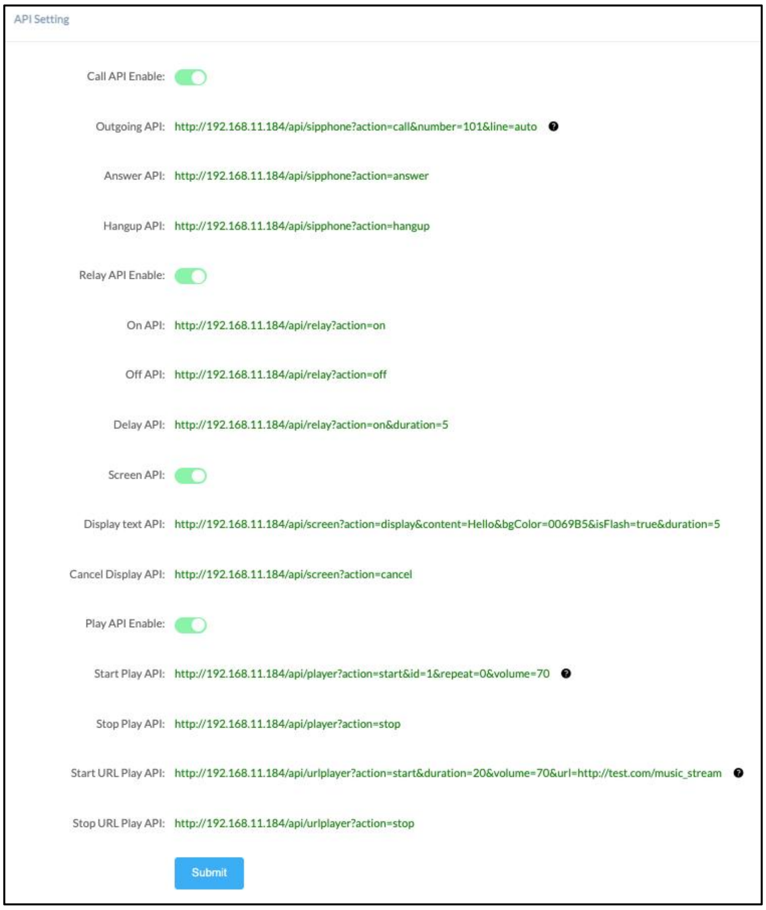
For example:

${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

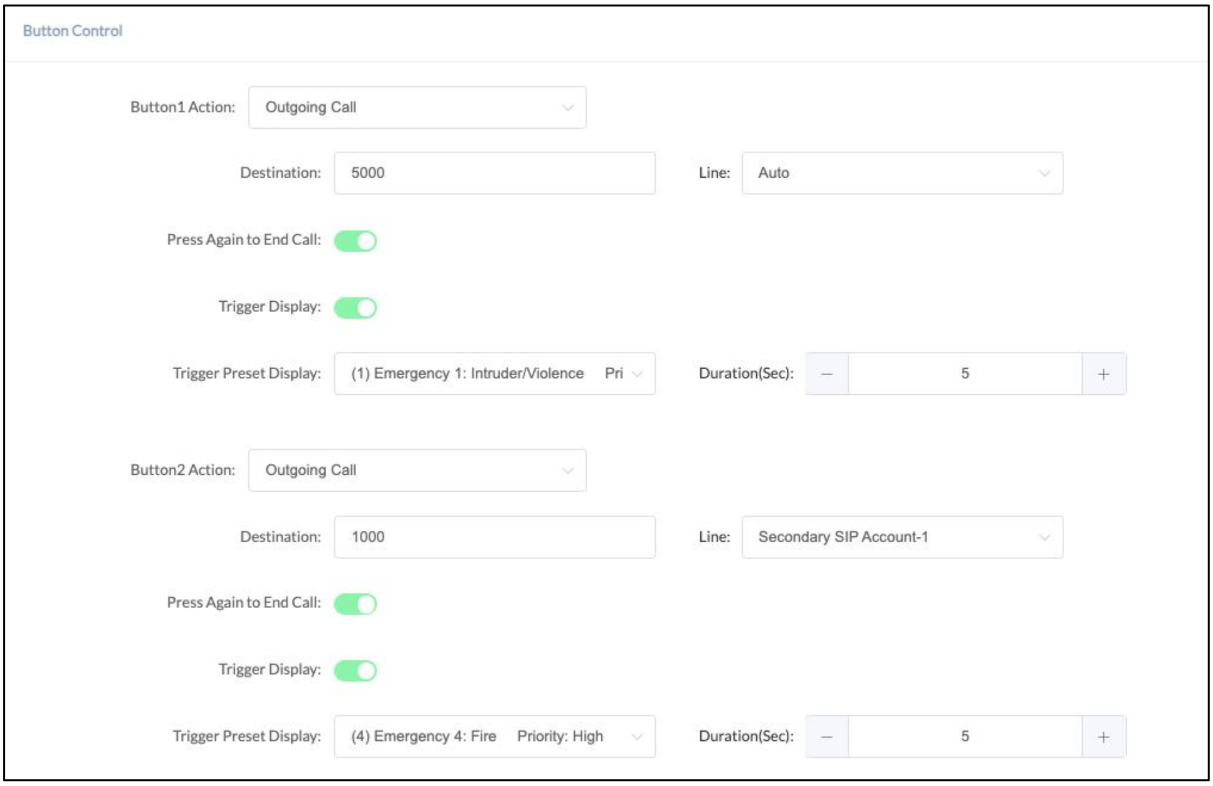
Using the API interface to realize features such as device linkage, call control, relay control, display control (SQ10-T Only), and play sound by the systems.

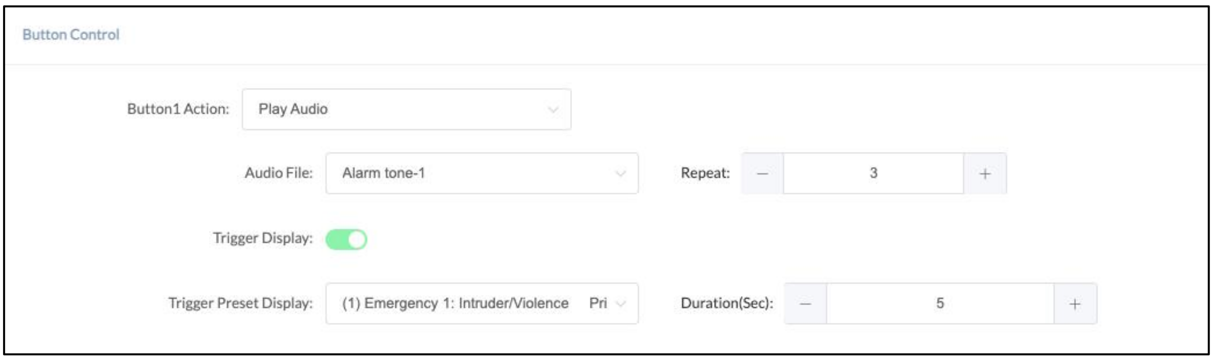
Note: Authentication and encryption are not used in the API interface, so please pay attention to the security of the network environment when opening and using these API interfaces.

6.8 I/O Settings

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

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





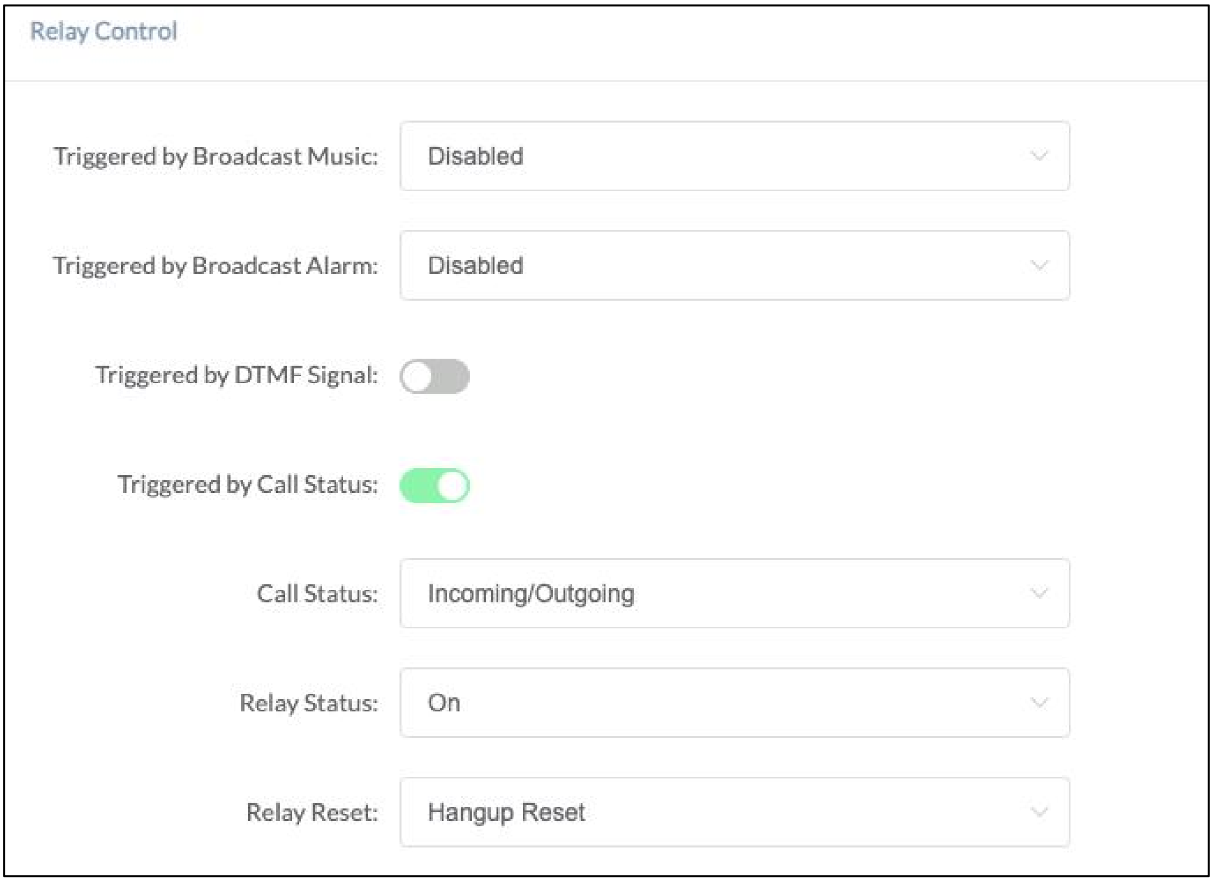
Button Control

- **Action:** Choose the event to trigger, such as Outgoing Call, HTTP Request, or Play Audio.

- **Destination:** Enter the response device's number that will be used when the external button is pressed.
- **Line:** Select the line for making outgoing calls.

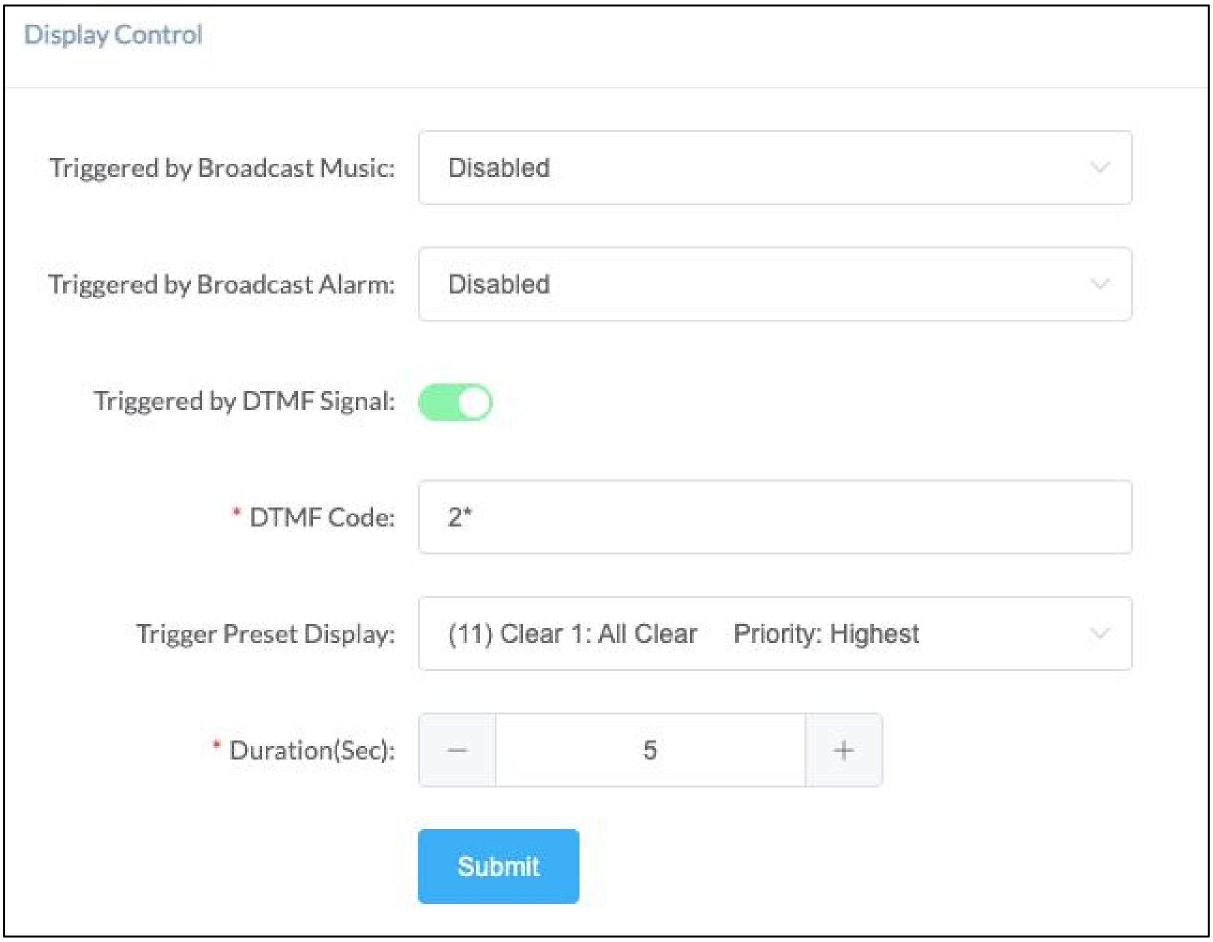
Note: when using the P2P line to call, please specify the device's number plus IP address, such as 101@192.168.11.123.

- **Press Again to End Call:** After the call is connected, press the button again to end the call or conversation.
- **HTTP URL:** Set the API URL to be triggered by the linkage.
- **Audio File:** Set the audio file to be triggered by the linkage.
- **Repeat:** Configure how many times the audio will repeat during the linkage.
- **Trigger Display:** Enable/Disable the display triggered by the linkage.
- **Trigger Preset Display:** Select the preset to display when triggered.
- **Duration(Sec):** Set the display duration.



Relay Control

- **Triggered by Broadcast Music:** Disabled/On/Fast Flashing/Slow Flashing. Enabling this option will trigger the relay when broadcast music is playing.
- **Triggered by Broadcast Alarm:** Disabled/On/Fast Flashing/Slow Flashing. Enabling this option will trigger the relay when a broadcast alarm is active.
- **Triggered by DTMF Signal:** Enable/Disable. Enable this option to trigger the relay using DTMF signals (only supported by RF2833).
- **DTMF Value:** Set the number to dial in order to trigger the DTMF signal.
- **Trigger by Call Status:** Enable/Disable. The relay will be triggered when there is a change in the call status.
- **Call Status:** Select the call state for triggering, including: Outgoing, Incoming, Incoming/Outgoing, Answered, and Hangup.
- **Relay Status:** Select the response type for the relay: On, Fast Flashing, or Slow Flashing.
- **Relay Reset:** Choose the reset mode after the trigger response: Delay Reset or Hang-up Reset.
- **Duration (sec):** Available only if Delay Reset is selected. This setting determines the time duration for the change in status after the trigger.



Display Control

- **Triggered by Broadcast Music:** Select Disabled, On, Fast Flashing, or Slow Flashing. Enabling this option will trigger the relay when broadcast music is playing.
- **Triggered by Broadcast Alarm:** Select Disabled, On, Fast Flashing, or Slow Flashing. Enabling this option will trigger the relay when a broadcast alarm is active.
- **Triggered by DTMF Signal:** Enable/Disable. Enable this option if you want to use a DTMF signal to trigger the relay (supported only by RF2833).
- **DTMF Code:** Set the number to dial in order to trigger the DTMF signal.
- **Trigger Display:** Enable/Disable the display triggered by Broadcast Music, Broadcast Alarm, or DTMF signals.
- **Trigger Preset Display:** Select the preset to be displayed when triggered.
- **Duration(Sec):** Set the duration (in seconds) for which the display will be shown.

6.9 PTP Settings

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

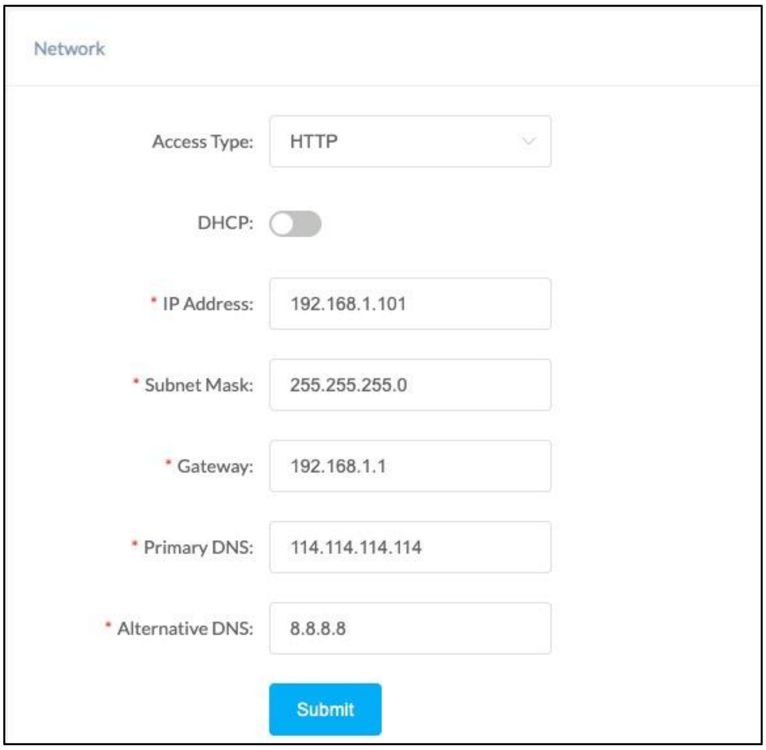
PTP Setting

7. System Settings

7.1 Network

SQ10 uses DHCP to dynamically obtain IP addresses by default.

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



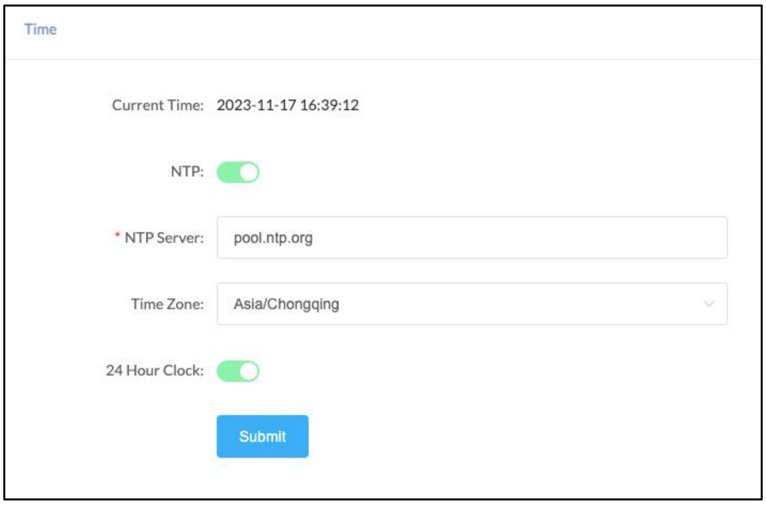
Network Configuration

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

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

7.2 Time

SQ10 obtains the time from the network time servers using NTP.
To change the NTP settings, please go to **System --> Time** page.



Time Settings

- **Current Time:** Display the current system time of the device.
- **NTP:** Enable/Disable using NTP to obtain the time.
- **NTP Server:** The network time server used to obtain the time.
- **Time Zone:** Set the time zone used by the device.
- **24 Hour Clock:** Using 24-hour format for time display.

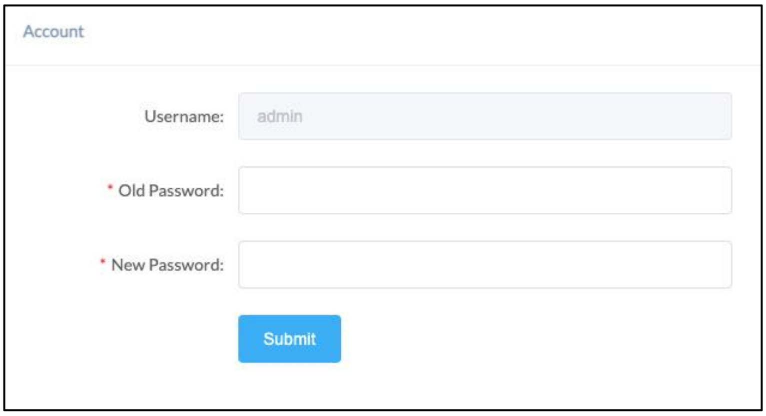
7.3 Prompt Language

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

Prompt Language

7.4 Account

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



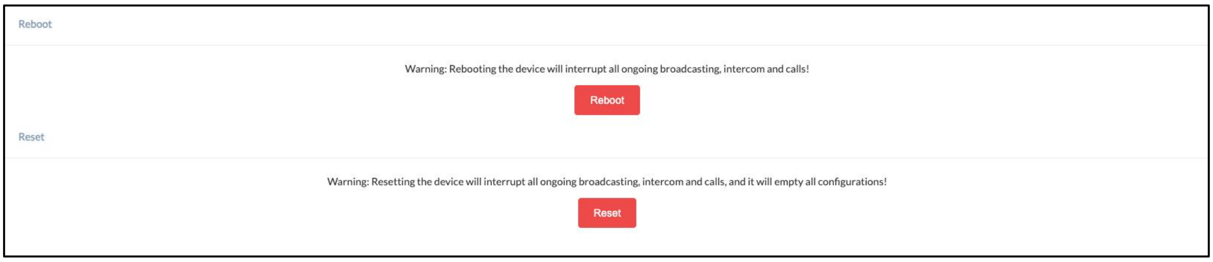
Web Password Settings

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

7.5 Reboot & Reset

SQ10 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!

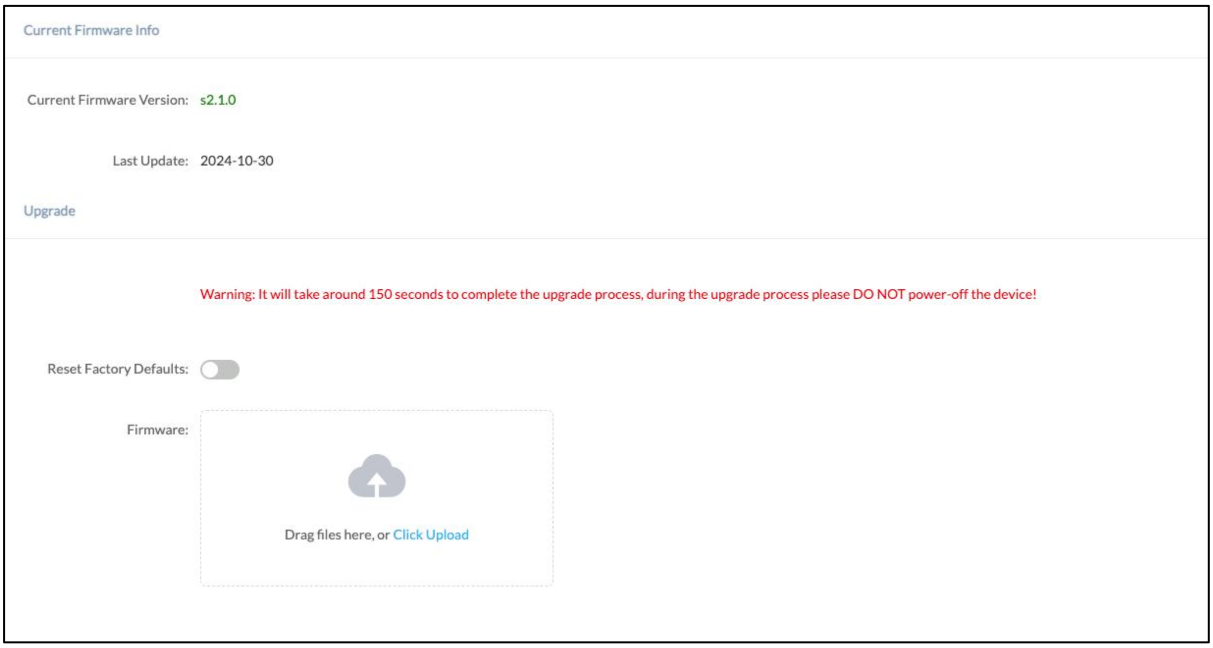
Reboot Schedule

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 **System --> Upgrade** page.



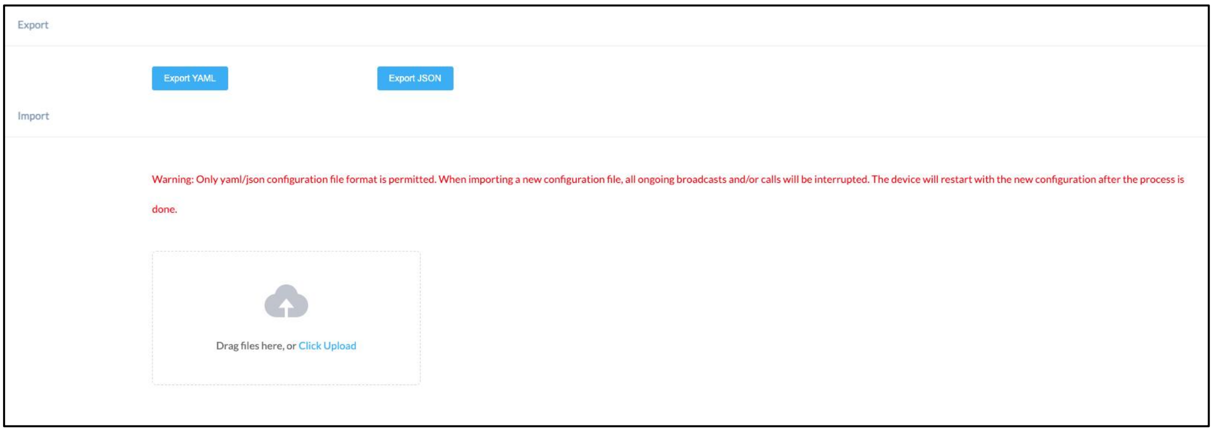
Upgrade Settings

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

8.2 Import/Export

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

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



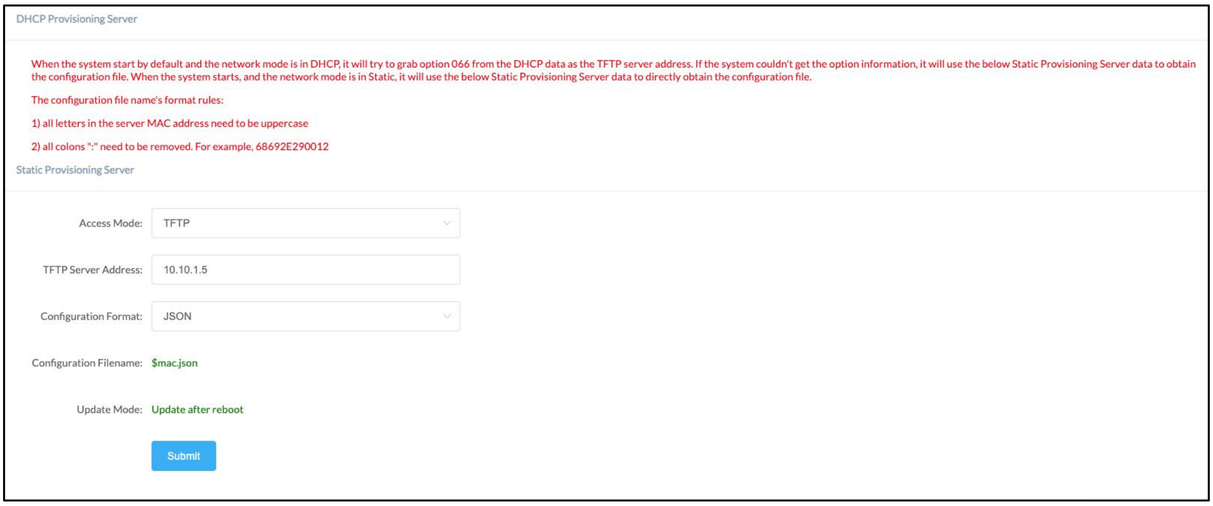
Import/Export

8.3 Auto Provisioning

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

The configuration file name's format rules:
1) all letters in the server MAC address need to be uppercase.
2) all colons ":" need to be removed. For example, 68692E290012.

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



Auto Provisioning

8.4 Diagnostic

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

Ping

8.5 Ethernet Capture

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

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

Ethernet Capture

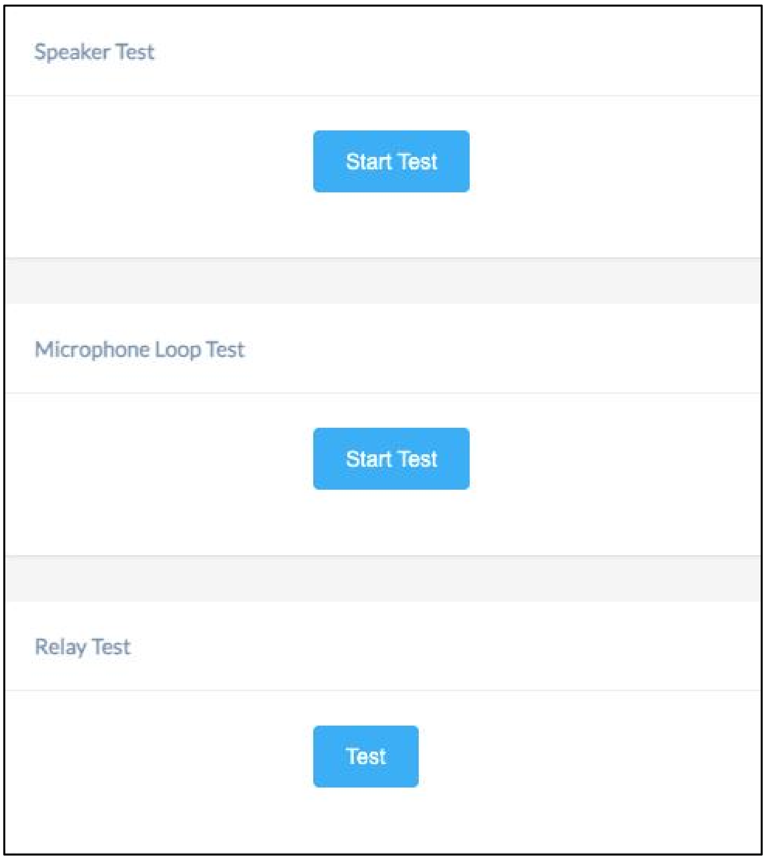
8.6 Test

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

For the SQ10-T, you can configure screen settings such as text, screen color, and flashing effects to preview the actual display outcome.

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

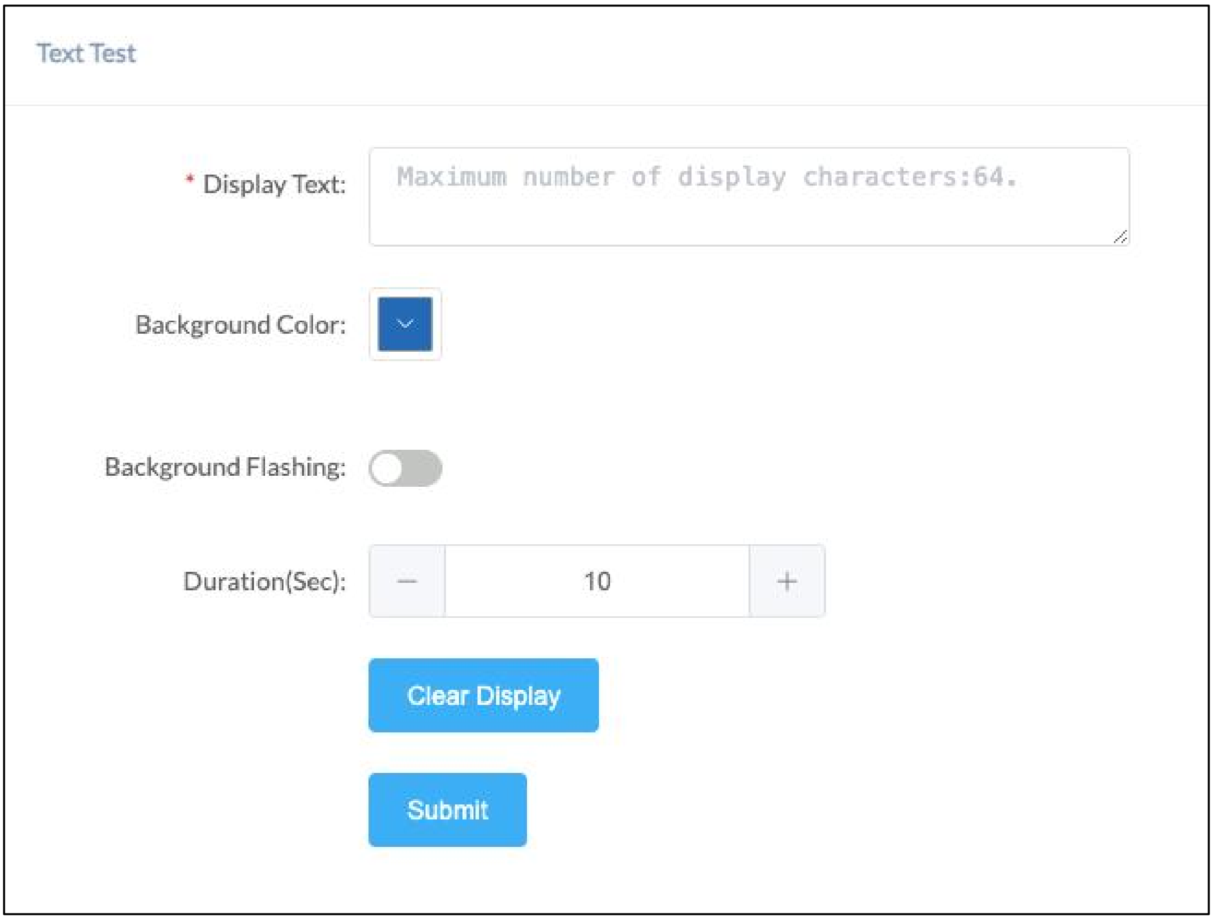
8.6.1 Audio Test



Audio Test Settings

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

8.6.2 Text Test



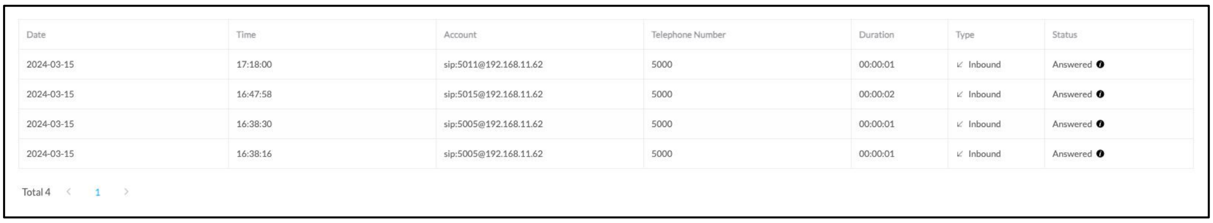
Text Test Settings

- **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.

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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