SD2140 Network Power Amplifier

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 SD2140 Network Power Amplifier. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the SD2140 Network Power Amplifier.

1.2 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
1.0.0	1.0.0	Added operating instructions for Firmware version v1.0.0.	June, 2025

2. Overview

2.1 Product Overview

The ZYCOO SD2140 Network Power Amplifier is designed for commercial audio applications, utilizing advanced Class-D technology to deliver crystal-clear sound and exceptional reliability.

The SD2140 provides 2 x 140W or 1 x 280W (bridged) outputs, supporting constant voltage and constant impedance configurations. It features balanced/unbalanced line, Bluetooth, and network inputs, with seamless compatibility with SIP, Dante, and ONVIF—all without the need for additional modules, providing a feature-rich solution.

The SD2140 is also equipped with a real-time adjustable mixer, an equalizer, and supports scheduled playback using USB storage devices.

2.2 Product Specifications

SD2140 Network Power Amplifier Specifications		
Amplifier Topology	Class-D	
THD+N(@ 1 kHz)	0.05%	
Frequency Response	20Hz～20KHz ± 0.25dB	
SNR	85dB	
Audio Output	SE: @ 100V 2 x 140W @ 25V 2 x 140W @ 4Ω(Min) 2 x 140W	

	BTL: @ 100VBridge 1 x 280W @ 70VBridge 1 x 280W @ 25VBridge 1 x 280W @ 8Ω(Min) 1 x 280W	
Audio Input	Balanced: 2 x 3-Pin Phoenix Connectors Unbalanced: 2 x RCA Jacks Bluetooth: 1 x IPX1 External 2.4G Antenna(BT 5.3 Compatible) Network: 1 x RJ45 Connector	

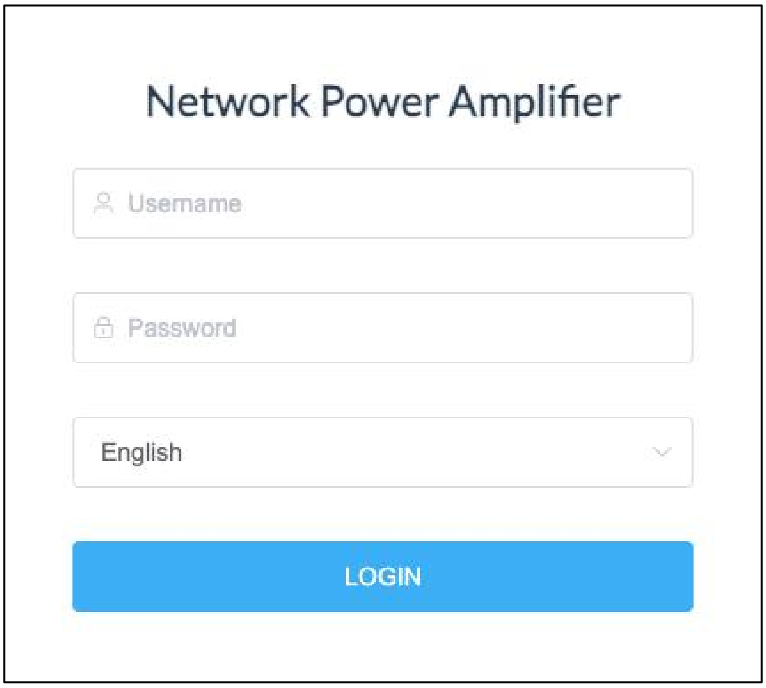
3. Login the Device

3.1 Accessing the Web GUI

SD2140 uses the static IP address: 192.168.1.101, please enter the IP address in the browser to access the device's Web management interface.

Default username: admin
Default password: admin

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

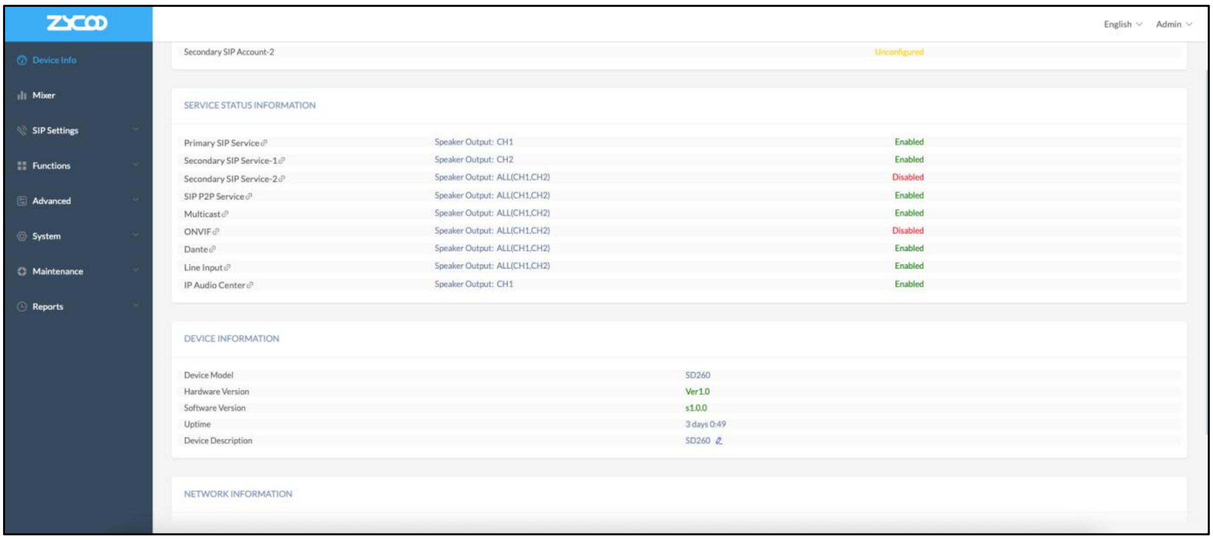


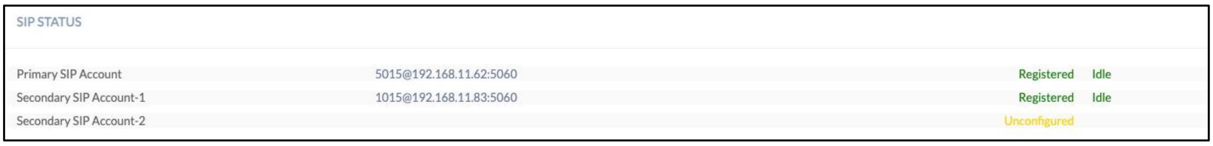
Login Interface

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

3.2 Device Info

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



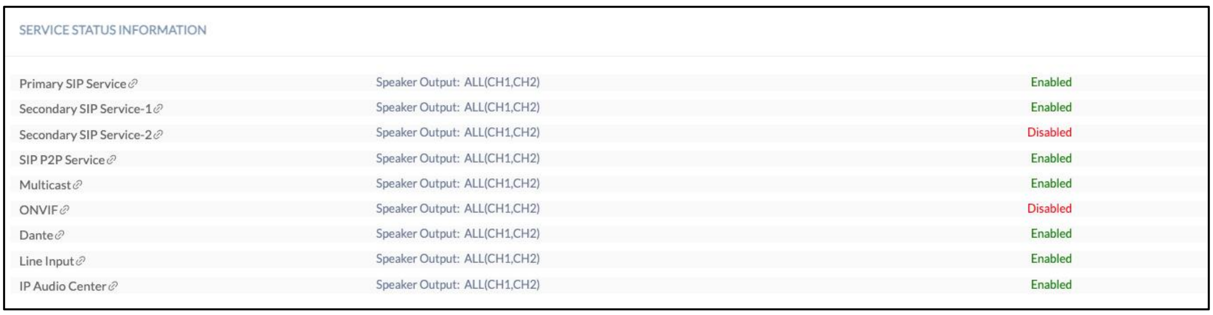


SIP STATUS

Primary SIP Account	5015@192.168.11.62:5060	Registered	Idle
Secondary SIP Account-1	1015@192.168.11.83:5060	Registered	Idle
Secondary SIP Account-2		Unconfigured	

SIP Status

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



SERVICE STATUS INFORMATION

Primary SIP Service	Speaker Output: ALL(CH1,CH2)	Enabled
Secondary SIP Service-1	Speaker Output: ALL(CH1,CH2)	Enabled
Secondary SIP Service-2	Speaker Output: ALL(CH1,CH2)	Disabled
SIP P2P Service	Speaker Output: ALL(CH1,CH2)	Enabled
Multicast	Speaker Output: ALL(CH1,CH2)	Enabled
ONVIF	Speaker Output: ALL(CH1,CH2)	Disabled
Dante	Speaker Output: ALL(CH1,CH2)	Enabled
Line Input	Speaker Output: ALL(CH1,CH2)	Enabled
IP Audio Center	Speaker Output: ALL(CH1,CH2)	Enabled

Service Status Information

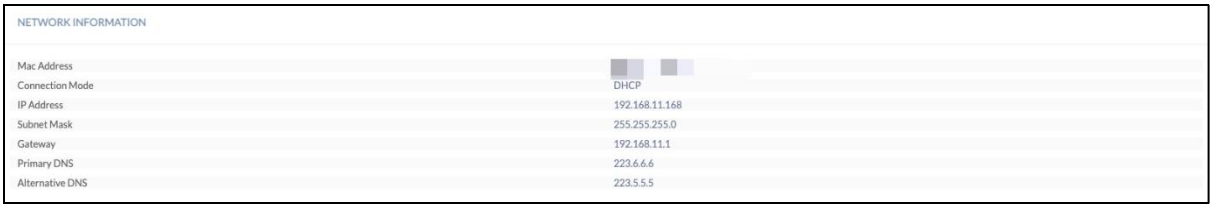
- **Primary SIP Service:** Display the service status of the main SIP account and the current audio output settings.
- **Secondary SIP Service-1:** Display the service status of the secondary SIP account 1 and the current audio output settings.
- **Secondary SIP Service-2:** Display the service status of the secondary SIP account 2 and the current audio output settings.
- **SIP P2P Service:** Display the service status of the P2P and the current audio output settings.
- **Multicast:** Display the service status of the multicast and the current audio output settings.
- **ONVIF:** Display the service status of the ONVIF and the current audio output settings.
- **Dante:** Display the service status of the Dante and the current audio output settings.
- **Line Input:** Display the service status of the line input and the current audio output settings.
- **IP Audio Center:** Display the service status of IP Audio Center service (such as ALARM, MUSIC features) and the current audio output settings.

DEVICE INFORMATION	
Device Model	SD260
Hardware Version	Ver1.0
Software Version	s1.0.0
Uptime	3 days 0:49
Device Description	SD260

Device Information

- **Device Model:** Displays the model of the device.
- **Hardware Version:** Displays the hardware version number of the device.
- **Software Version:** Display the system version number of the device.
- **UpTime:** Displays the device's operating time.

- **Device Description:** Remark the device information. The description will be displayed in a browser tab. After the Device Description is set, the description will be displayed in the browser tab, which is convenient for distinguishing different terminals when there are many terminal configuration pages.

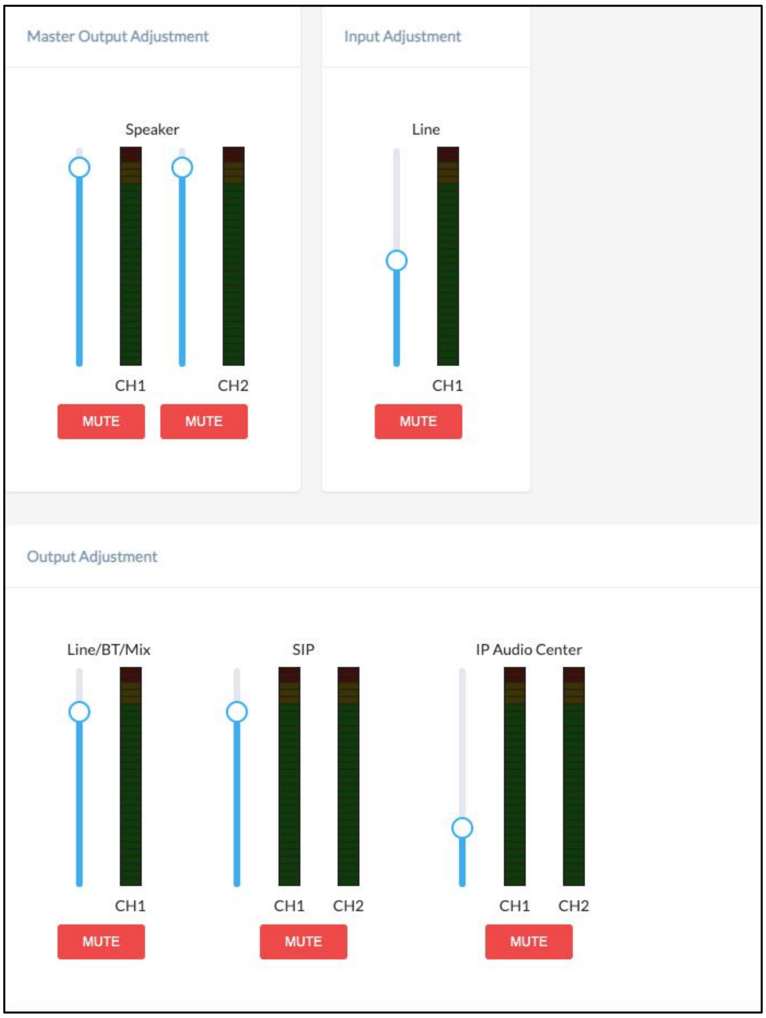


Network Information

- **Mac Address:** Display the MAC address of the current device.
- **Connection Mode:** Display the network acquisition method of the device, DHCP (dynamic acquisition) or STATIC (static configuration).
- **IP Address:** The current IP address of the device.
- **Subnet Mask:** The current subnet mask of the device.
- **Gateway:** The gateway address currently used by the device.
- **Primary DNS:** The primary domain name server address used by the device.
- **Alternative DNS:** The secondary domain name server address used by the device.

4. Mixer

The Mixer feature allows users to make real-time adjustments on input source volume, output source volume, and master output volume. Please go to the **Mixer** page for detailed adjustments. To adjust the volume, simply drag the volume slider. To mute a specific audio source, click the MUTE button.



Mixer

- **Input Adjustment:** Displays settings based on **Advanced Settings > Input & Output Settings**. Allows adjustments for line input and Bluetooth input.
- **Output Adjustment:** Displays settings dynamically according to the enabled features in **Input & Output Settings, SIP Settings, ONVIF Settings, Dante Settings, and**

Event Scheduler. Disabled features will not appear. Supports audio adjustments for line input, Bluetooth, line and Bluetooth mixing, SIP, Dante, ONVIF, multicast, IP Audio Center, and Event Scheduler.

- **Master Output Adjustment:** Provides overall volume control for speaker output CH1 and CH2 channels.

Note: The audio output priority for local audio sources (including Bluetooth, line input, and Bluetooth mixing) is set to the lowest by default. When they are playing, their output will be temporarily interrupted if another audio (such as a Event Scheduler task or SIP paging) takes precedence. Similarly, other audio also have priorities, which can be manually adjusted on the ***Advanced --> Audio Priority*** *page. By default, SIP audio has the highest priority.*

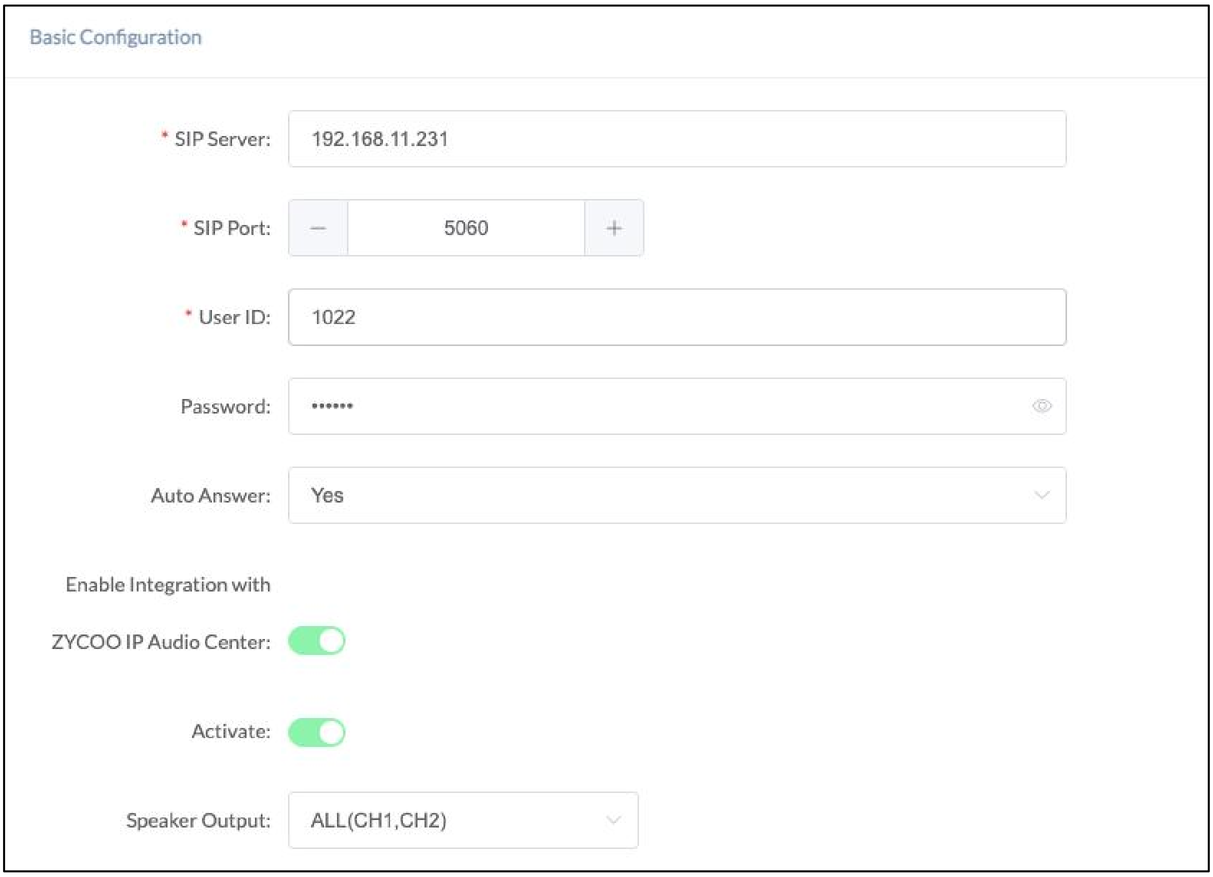
Note 2: The SIP output adjustment controls the volume for SIP calling, such as SIP paging and SIP intercom, while the IP Audio Center output adjustment controls the volume for emergency alarms, background music initiated from the Dispatch Console.

5. SIP Settings

5.1 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.

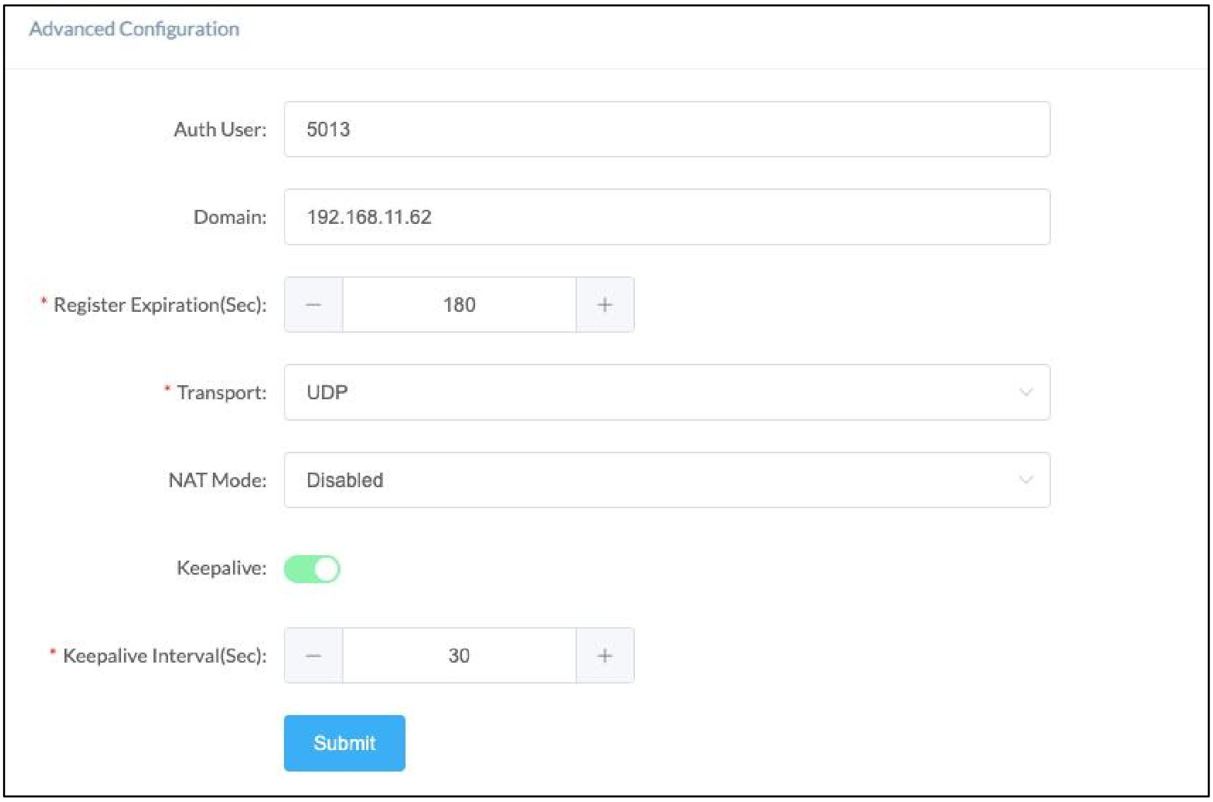
Please go to **SIP Settings --> Primary SIP Account / Secondary SIP Account-1 / Secondary SIP Account-2** page.



SIP Account - Basic Configuration

- **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.
- **Speaker Output:** Configure the output channel for SIP calling when using the corresponding SIP account. It can be set as CH1, CH2, and both.



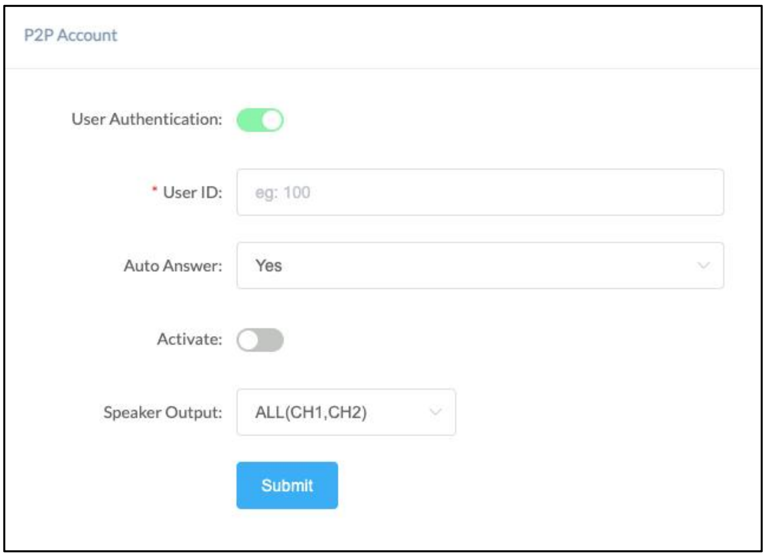
SIP Account - Advanced Configuration

- **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.

5.2 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.
After configuring the P2P account, it can be used with the Outgoing Call feature in the Outgoing API in **Basic Settings ---> API Settings** to make a P2P call.



P2P Account

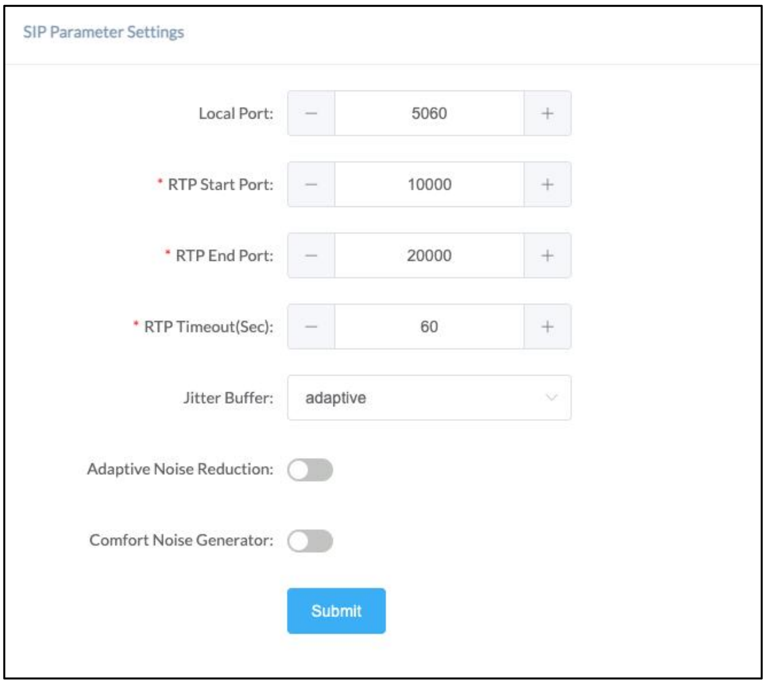
- **User Authentication:** Enable/Disable P2P authentication. If disabled, you can directly enter this device’s IP address in the target field of the peer device. If enabled, you must use the following format in the target field of the peer device: This device's P2P User ID + IP address (e.g., 101@192.168.1.101).

- **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.
- **Speaker Output:** Configure the output channel for SIP P2P calling when using the corresponding SIP account. It can be set as CH1, CH2, and both.

5.3 Advance SIP Settings

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

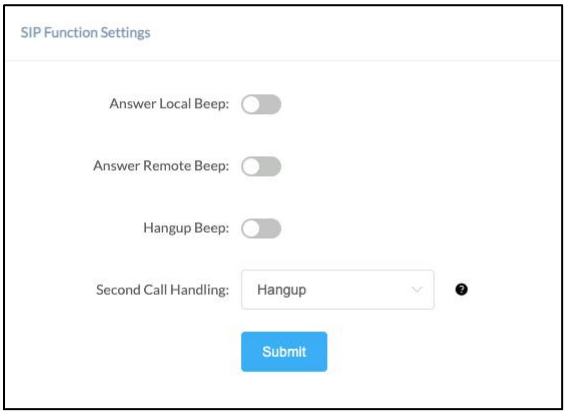
5.3.1 SIP Parameter Settings



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.
- **Adaptive Noise Reduction:** After enabling this feature, algorithms can suppress environmental noise.
- **Comfort Noise Generator:** After enabling this feature, comfortable white noise can be added during calls.

5.3.2 SIP Function Settings



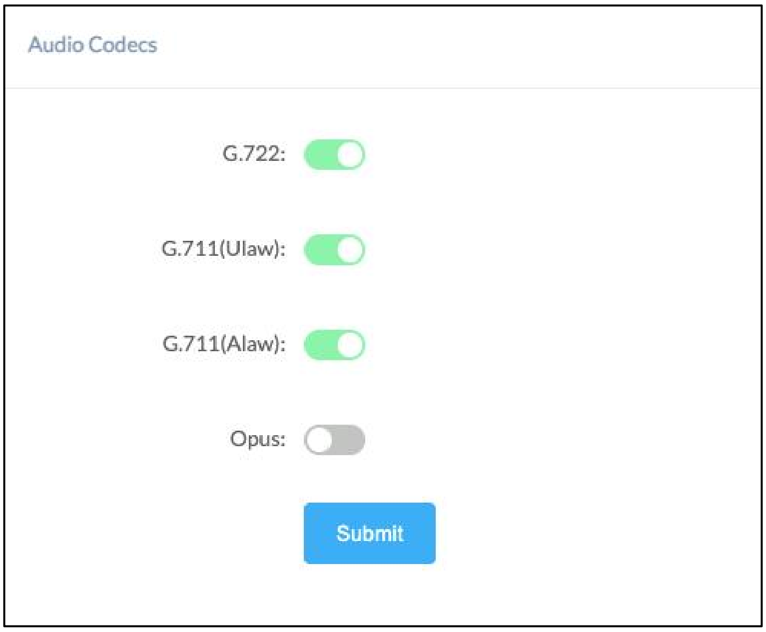
SIP Function Settings

- **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.
- **Remote Answer 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.
- **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.
- **Hangup Beep:** If this setting is enabled, the selected beep sound will be played before the SIP session is completely hung up.
- **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.
- **Second Call Handling:** Set how to hand 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.

5.3.3 Audio Codecs

SD2140 supports 4 audio codecs: G.722 (wideband codec), G.711(Ulaw), G.711(Alaw), and Opus.



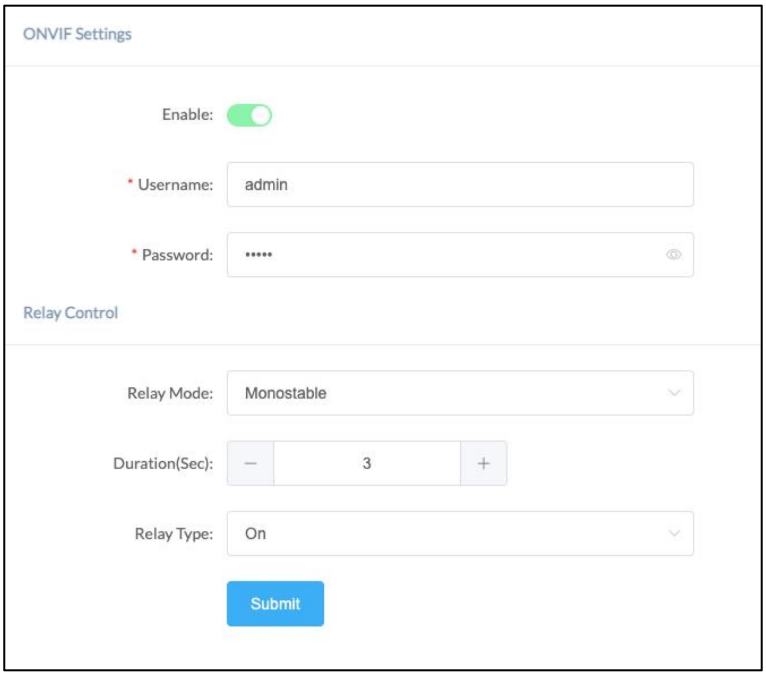
Audio Codecs

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

6. Function Settings

6.1 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.



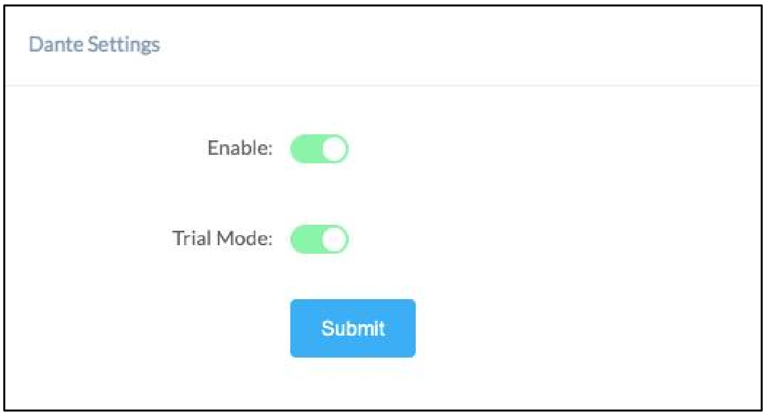
ONVIF & Relay Control Settings

- **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.
- **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’.

6.2 Dante Settings

Dante is the product name for a combination of software, hardware, and network protocols that delivers uncompressed, multi-channel, low-latency digital audio over a standard Ethernet network using Layer 3 IP packets.
If you are using Dante-related products and want to integrate the SD2140 into a Dante network, please go to **Functions ---> Dante Settings** to configure the Dante mode.



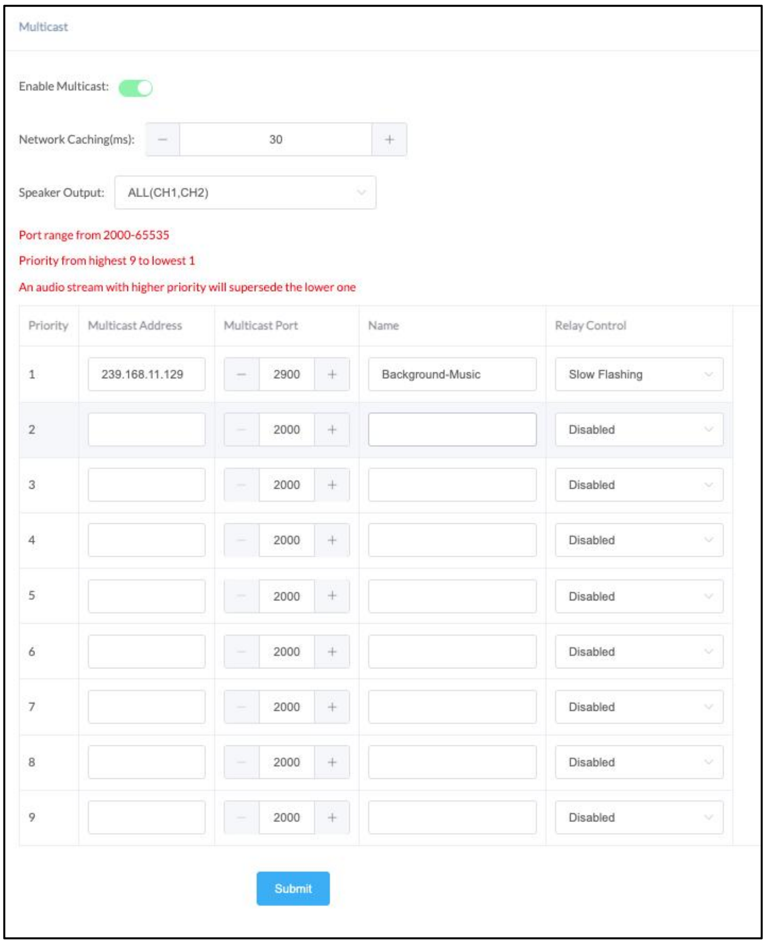
Dante Settings

- **Enable:** Enable/Disable the Dante feature.
- **Trial Mode:** Enable/Disable the device’s trial mode.

6.3 Multicast

The multicast settings are used to configure the parameter settings of the multicast function on the SIP Safety Intercom. It can be configured to monitor up to 9 different levels of multicast addresses, the audio streams with a higher priority will interrupt the playback of the lower priority audio streams.

Please go to **Functions ---> Multicast** page to enable the multicast feature.



Multicast

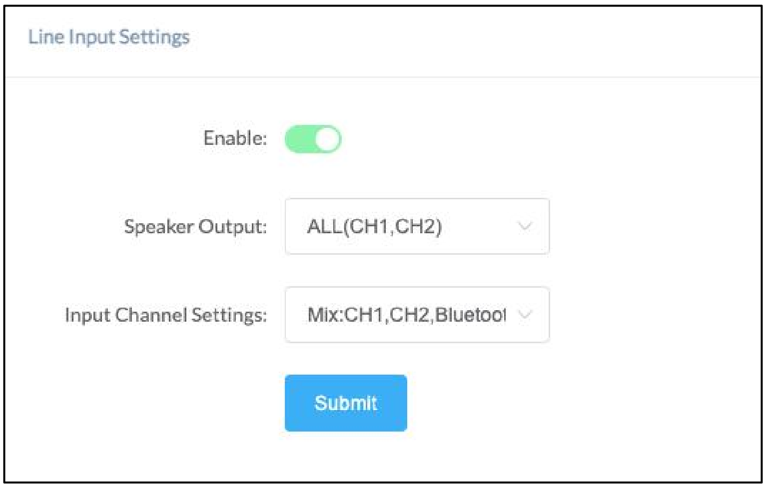
- **Enable Multicast:** Enable/Disable the Multicast feature.
- **Network Caching(ms):** Represents the buffer value, a smaller buffer value minimizes latency but may result in stuttering in poor network conditions. A larger buffer value increases latency but ensures smooth playback even in unstable network environments.
- **Speaker Output:** Configure the output channel for multicast source. It can be set as CH1, CH2, and both.
- **Priority:** Priority from highest 9 to lowest 1.
- **Multicast Address:** The multicast address range is 224.0.0.0 – 239.255.255.255.
- **Multicast Port:** The multicast port range is 2000 – 65535.

- **Name:** Customize the name of the multicast address.
- **Relay Control:** Options to choose from are ‘Disabled’, ‘On’, ‘Fast Flashing’, ‘Slow Flashing’.

7. Advanced Settings

7.1 Line Input Settings

To configure the local line input and output for the SD2140, please go to **Advanced --> Line Input Settings**. On this page, you can set up both the input sources and output destinations.



Line Input Settings

- **Enable:** Enable/Disable the local audio source input and output feature.
- **Speaker Output:** Configure the output channel for the local line-in source. It can be set as CH1, CH2, and both. When using BTL single channel mode output, please set this option to CH1.
- **Input Channel Settings:** Configure the input mode for local audio sources. It can be set to the following input modes:
  - ➢ Line: CH1

    The input channel is CH1 only.
  - ➢ Line: CH2

    The input channel is CH2 only.
  - ➢ Line: CH1, CH2

    The input channels are CH1 and CH2 as independent single-channel inputs, each routed to their respective output ports (CH1 and CH2). At the same time, the output option should also be set to ALL (CH1, CH2).

➢ Bluetooth

The input channel is a Bluetooth audio source.

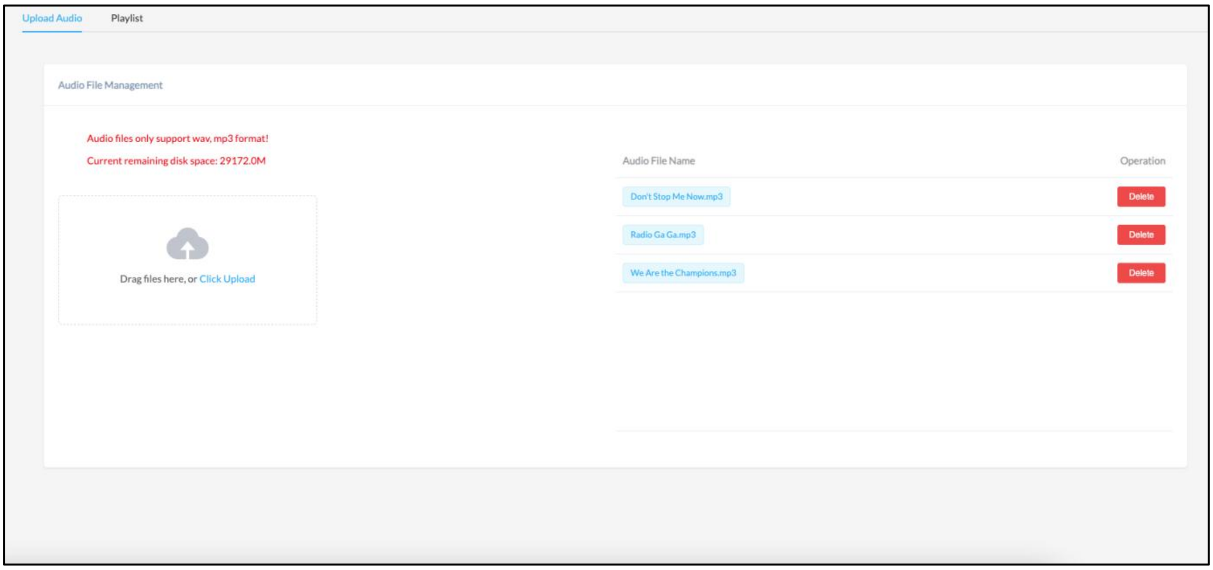
➢ Mix: CH1, CH2, Bluetooth

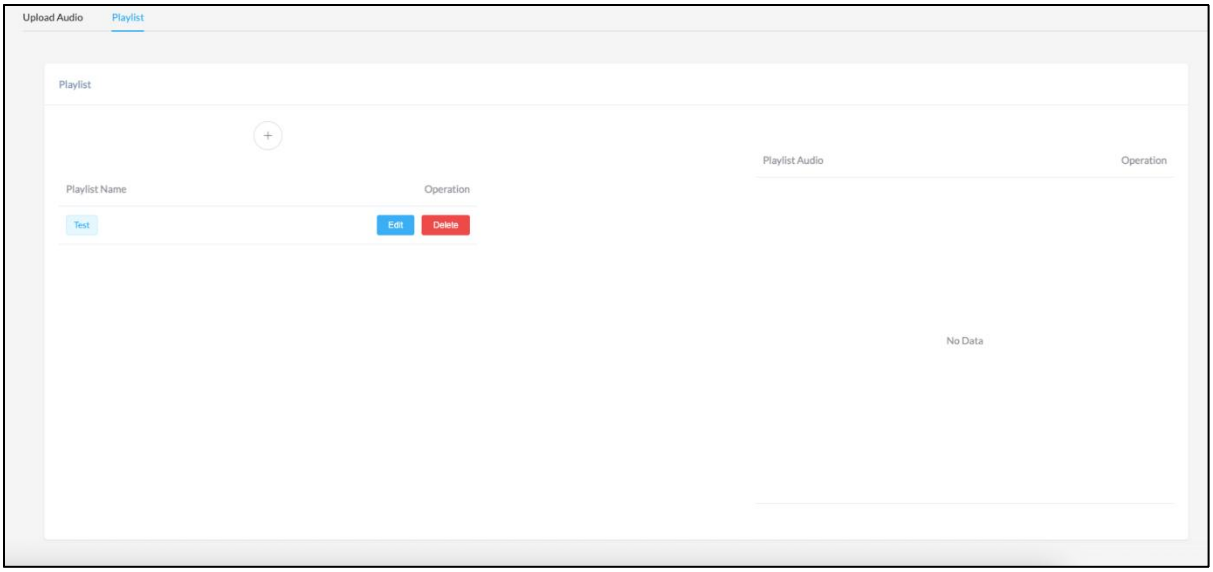
The input channels CH1, CH2, and Bluetooth audio sources are mixed into a single audio source for output.

7.2 Audio Management

When an external storage device, such as a portable hard drive, is connected to the SD2140's USB port, the SD2140 will read the audio files stored on the device. Users can also upload audio files, create playlists, and use the event scheduler feature to set offline scheduled playback tasks.

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





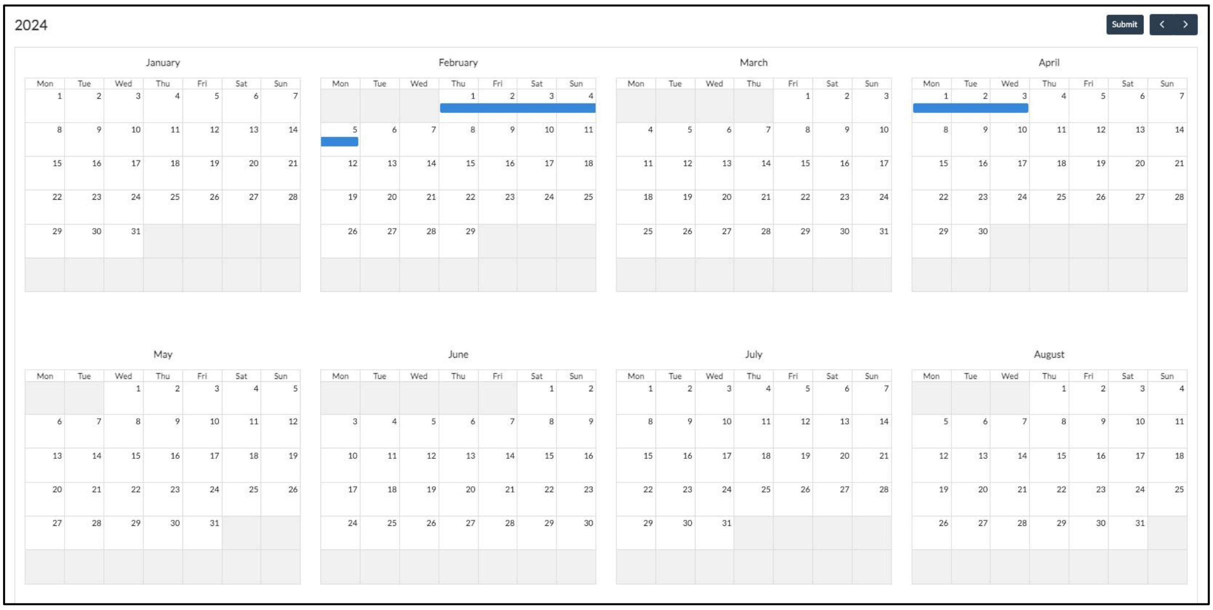
Audio Files

7.3 Event Scheduler

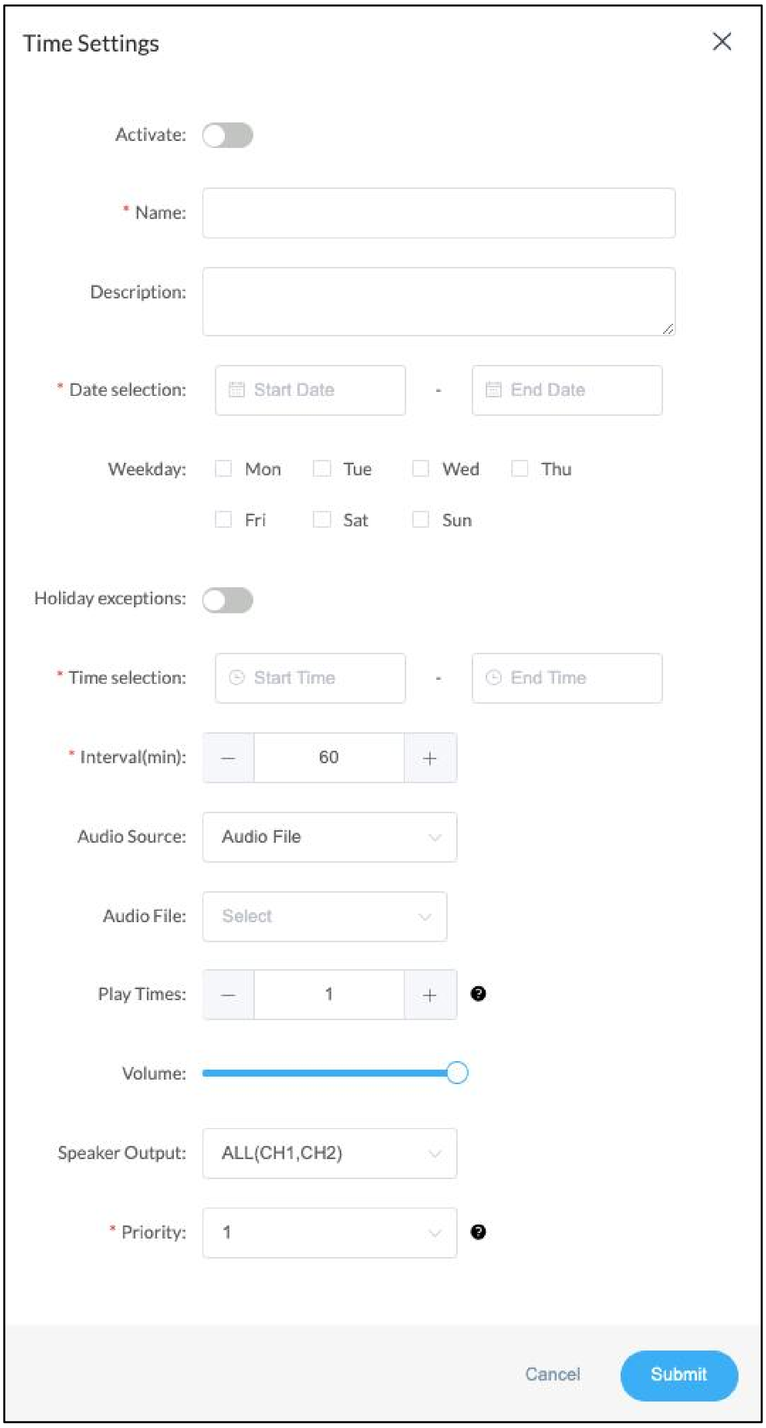
The event scheduler can edit up to 30 time plans, you can click the corresponding option to edit or delete them. Before you edit/create an event scheduler. The schedules in the holiday setting will not be executed.

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 Source:** Select an audio source for playback, choosing from audio files or playlists managed through the Audio Management feature.
- **Audio File:** Select an audio file to play.
- **Playlist:** Select an playlist 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.
- **Speaker Output:** Configure the output channel for event scheduler playback. It can be set as CH1, CH2, and both.
- **Priority:** Assign priority to the scheduler system; higher-priority schedules will always execute first.

7.4 Audio Priority Settings

The audio priority can be set according to different applications(SIP, ONVIF, MULTICAST, BROADCAST, SCHEDULER-AUDIO). 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.

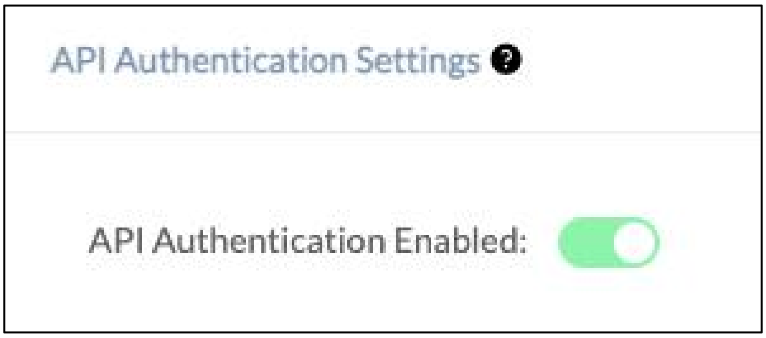
Please go to **Advanced --> Audio Priority** page to enable API settings.

Audio Priority Settings

7.5 API Settings

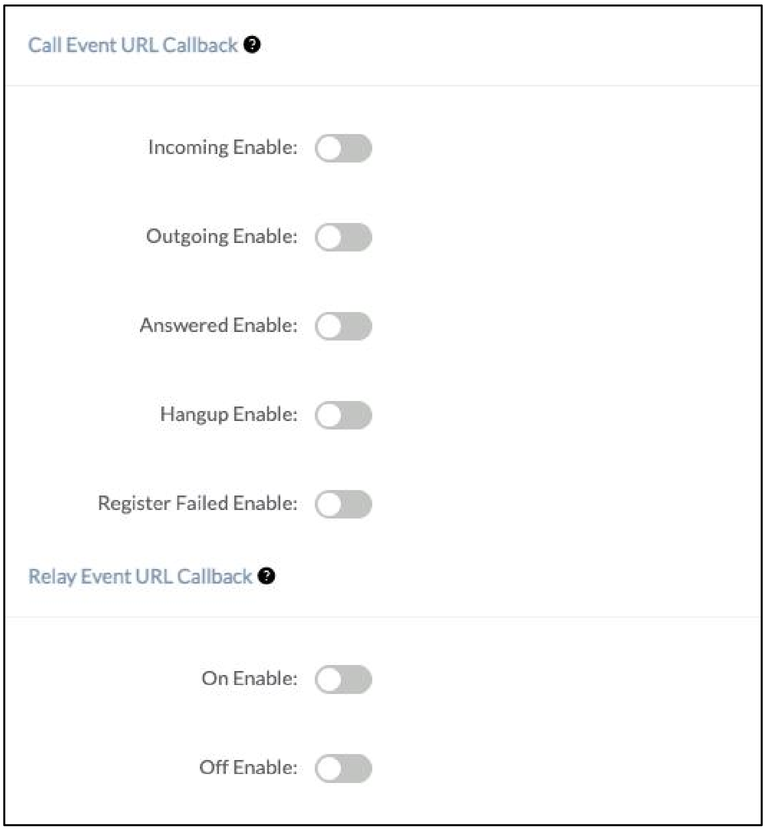
This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the changing status of the call and/or relay.

Please go to **Advanced --> API Settings** page to enable API settings.



API Authentication Settings

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



Call Event URL Callback & Relay Event URL Callback

When the call status changes, it will trigger an HTTP GET request to call a URL address. Within the URL address, you may use variables to identify some current information.

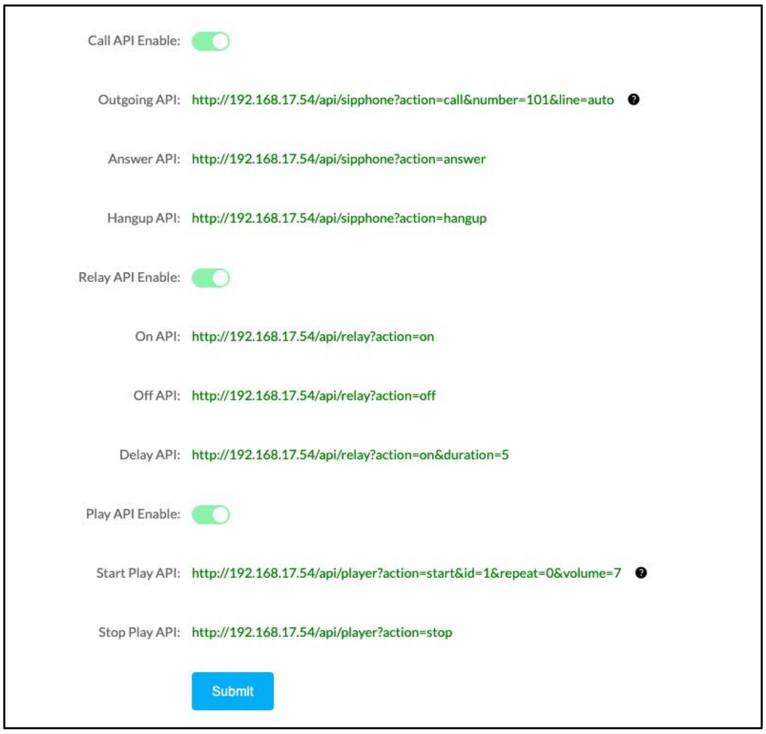
For example:

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

When the relay status changes, it will trigger an HTTP GET request to call a URL address. Within the URL address, you may use variables to identify some current information.

For example:

${ip}:	The current IP address of the device
${mac}:	The current MAC address of the device



API Settings

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

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

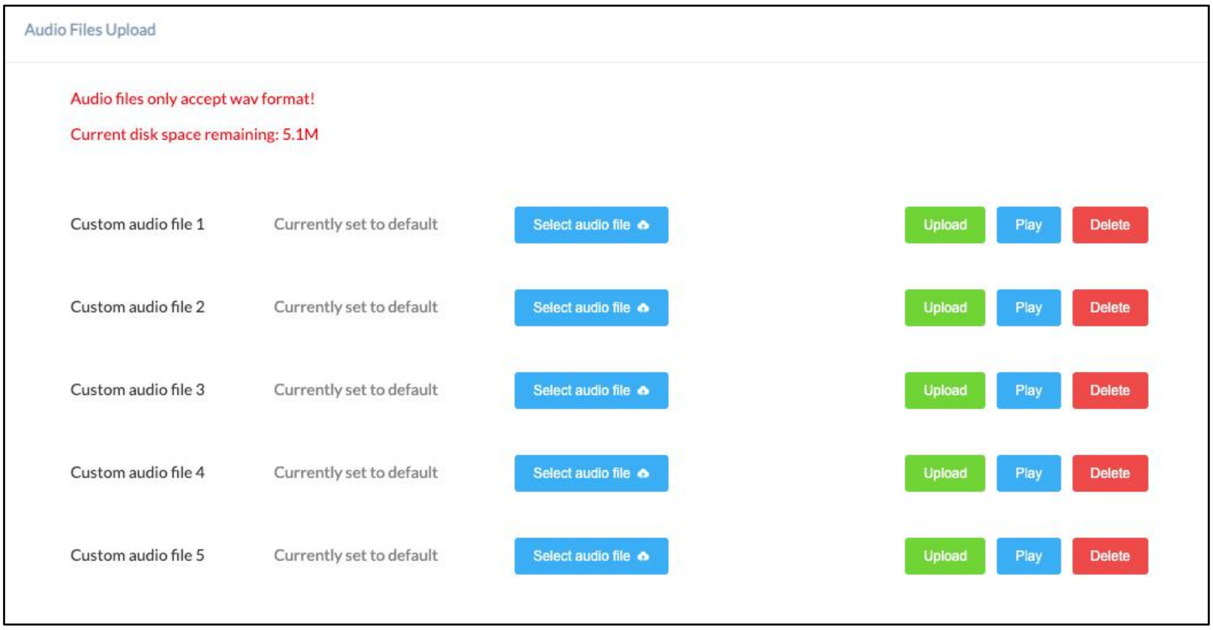
7.6 PTP Settings

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

PTP Setting

7.7 Ring File

The Audio files section allows users to self-upload up to 5M of audio files to the endpoint and use it as a ringtone or Play API audio file. Please click on the ‘Select audio file’ button to select and upload the local audio file, then click on the ‘upload’ button to upload it. Click on the ‘play’ to test and play the audio file and the ‘delete’ button for deleting the audio file. Please go to **Advanced ---> Ring File** to manage the audio files.



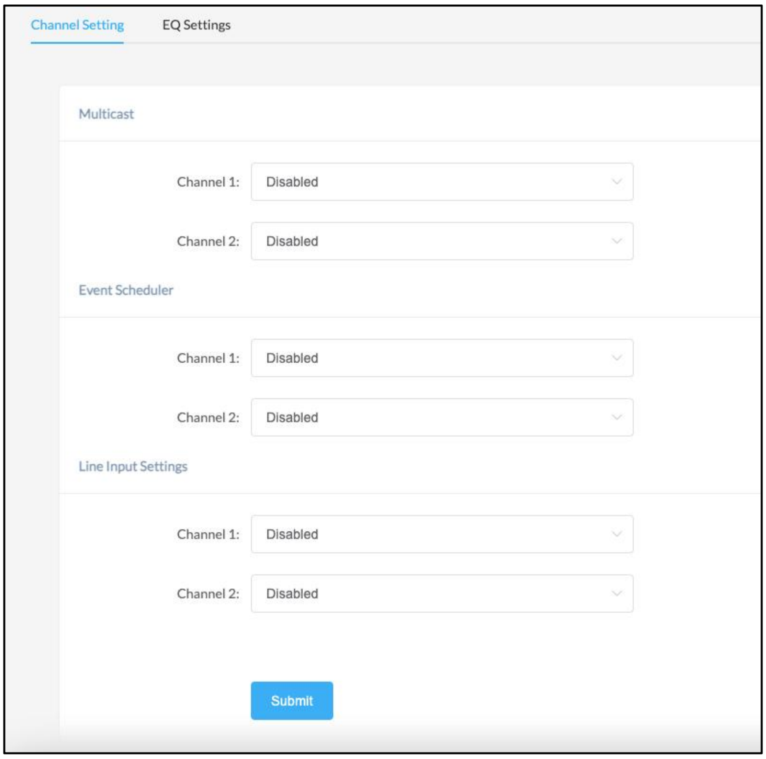
Audio Files

7.8 EQ Settings

The EQ features a 10-band frequency adjustment, allowing precise optimization of different frequency ranges for audio sources such as multicast, event scheduler playback, and local line inputs.

Users can create and manage multiple equalizer profiles, easily applying them to various audio sources and channels. This enables effortless sound quality adjustments tailored to different audio styles, including electronic, jazz, and rock, delivering professional-grade audio solutions for diverse scenarios.

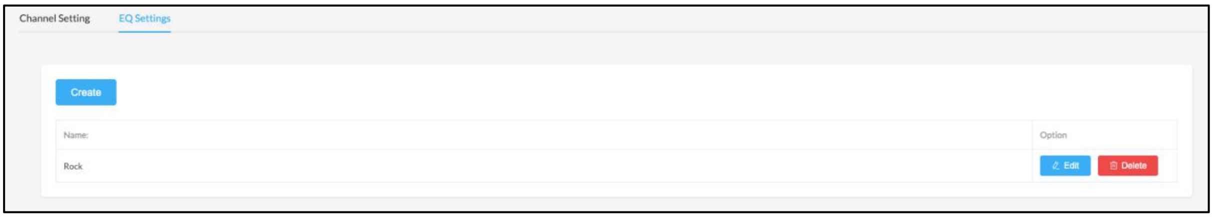
Please go to **Advanced --> EQ Settings** page to set the specific settings.



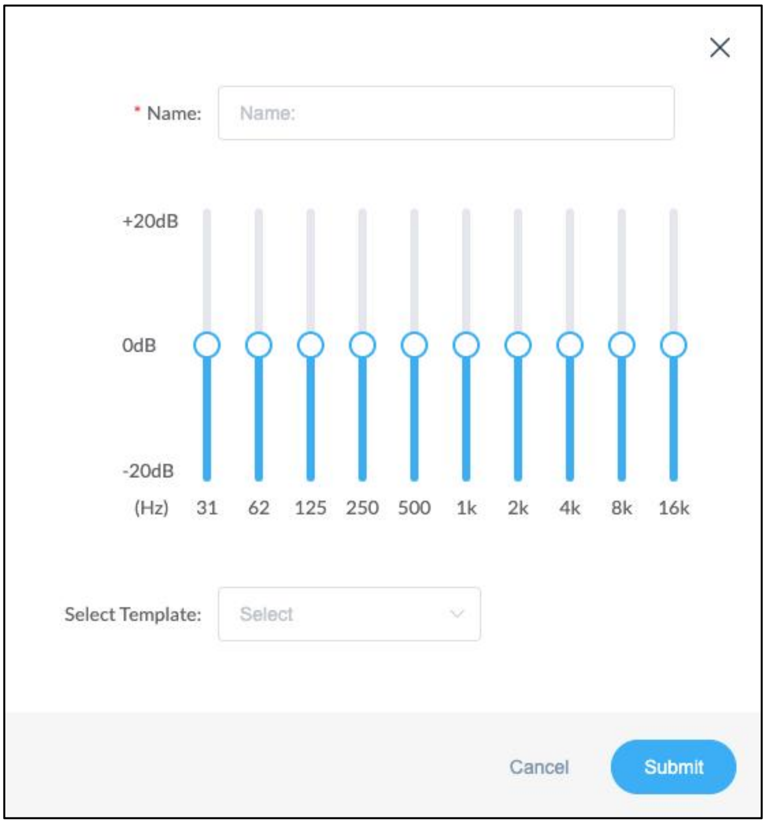
Channel Settings

- **Channel 1:** Configure the EQ profile applied to Channel 1 for audio sources such as multicast, event scheduler playback, or local line inputs. The profile must be created first in the EQ Settings section.

- **Channel 2:** Configure the EQ profile applied to Channel 2 for audio sources such as multicast, event scheduler playback, or local line inputs. The profile must be created first in the EQ Settings section.



EQ Settings



Create a New Profile

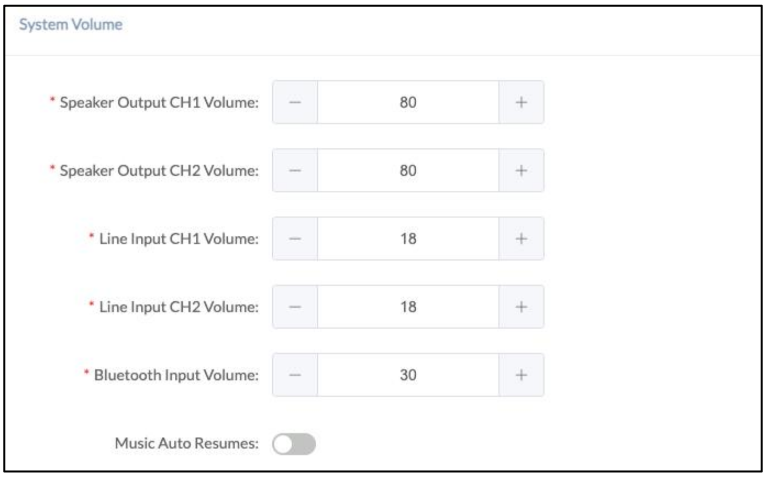
- **Name:** Add a custom name for the newly created equalizer profile.
- **Frequency (Hz):** Provides a 10-band equalizer frequency adjustment, allowing precise control of the gain (in decibels) for each band to fine-tune specific frequency ranges.

- **Select Template:** Choose from system-provided equalizer templates tailored to different audio needs, such as electronic, jazz, rock, and other styles, to quickly achieve the optimal sound effect.

8. System Settings

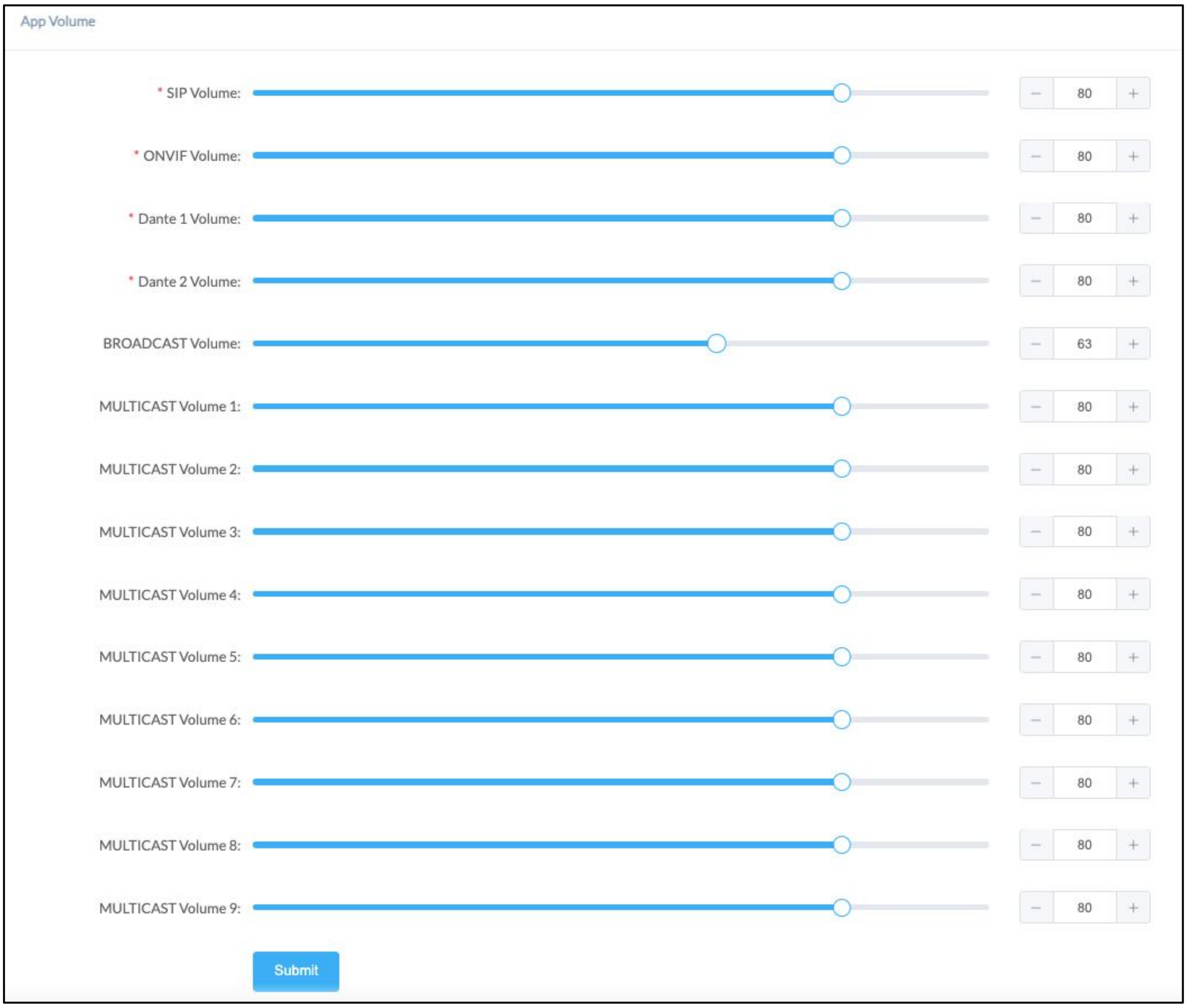
8.1 Volume Settings

To set the volume of the SD2140, please go to **System --> Volume** page to configure. You can adjust the main volume as well as individual volume adjustments for each application.



Speaker Volume Settings

- **Speaker Output CH1 Volume:** Set the speaker output CH1's system volume, and the adjustable range is 0~99.
- **Speaker Output CH2 Volume:** Set the speaker output CH2's system volume, and the adjustable range is 0~99.
- **Line Input CH1 Volume:** Set the line input CH1's system volume, and the adjustable range is 0~31.
- **Line Input CH2 Volume:** Set the line input CH2's system volume, and the adjustable range is 0~31.
- **Bluetooth Input Volume:** Set the bluetooth input's system volume, and the adjustable range is 0~31.
- **Music Auto Resumes:** When the device restarts or reconnects to the network, the previous music tasks will be automatically restored.



App Volume Settings

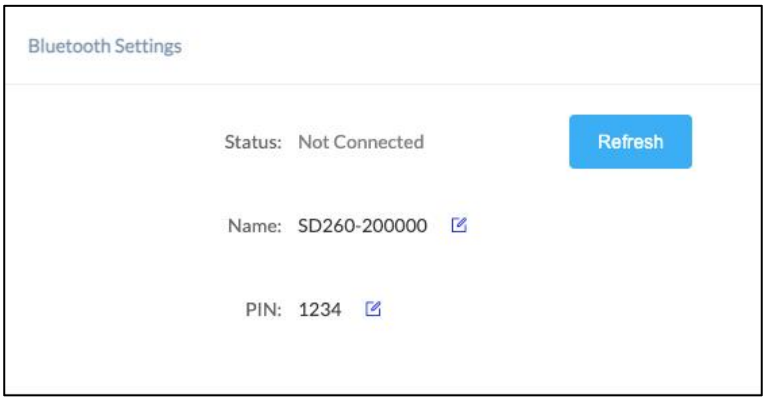
- **SIP Volume:** Set the specific volume for SIP application.
- **ONVIF Volume:** Set the specific volume for ONVIF application.
- **Dante 1 Volume:** Set the specific volume for Dante CH1 application.
- **Dante 2 Volume:** Set the specific volume for Dante CH2 application.
- **Broadcast Volume:** Set 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.

8.2 Bluetooth Settings

After connecting the Bluetooth antenna, the SD2140 can receive Bluetooth audio signals. Once paired with a mobile phone, you can use Spotify online music, local music, and other sources from the phone as audio input.

Please go to **System --> Bluetooth Settings** page to configure.



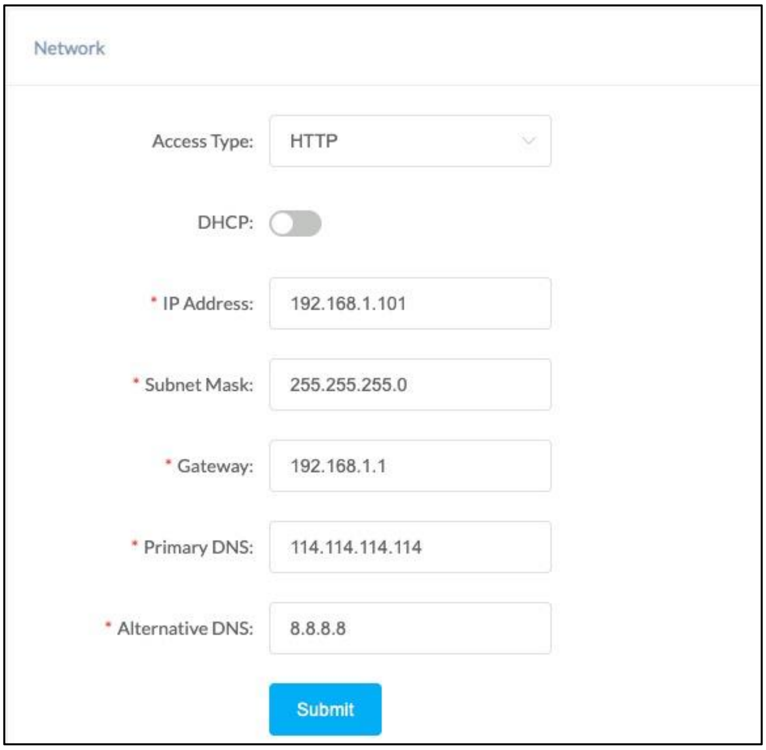
Bluetooth Settings

- **Status:** Display the current pairing status. When a device is successfully connected, it shows "Connected."
- **Name:** Display the name of the SD2140 as it appears in the Bluetooth pairing list of other devices. Click the edit button to customize the name.
- **PIN:** Display the PIN code used during SD2140's Bluetooth pairing. Click the edit button to customize the PIN code.

8.3 Network

SD2140 uses the static IP address: 192.168.1.101 by default.

To change the IP assignment mode, please go to **System--> Network** page.



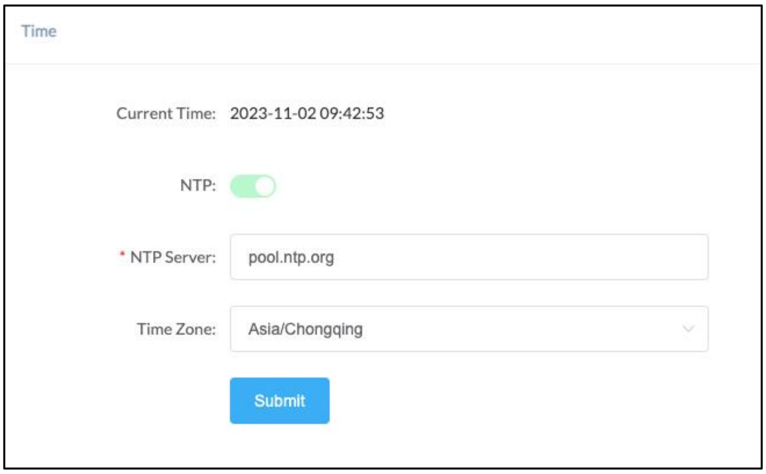
Network Configuration

- **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.

8.4 Time

SD2140 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.

8.5 Prompt Language

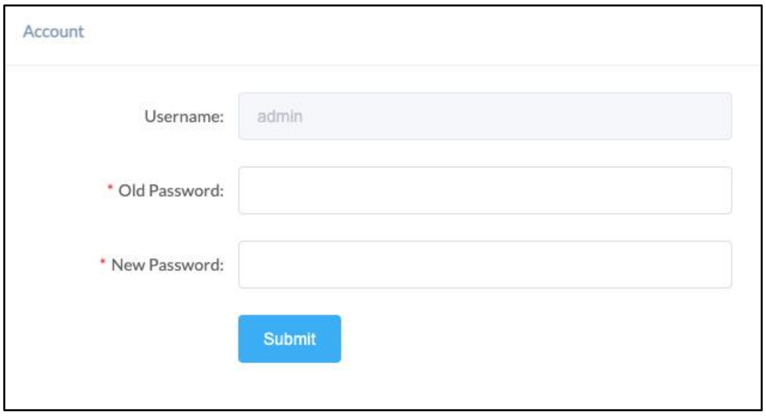
The language of local voice prompts, like IP address announcements. Currently, only Chinese and English are provided.

Please go to **System --> Prompt Language** page to set a voice prompt language.

Prompt Language

8.6 Account

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



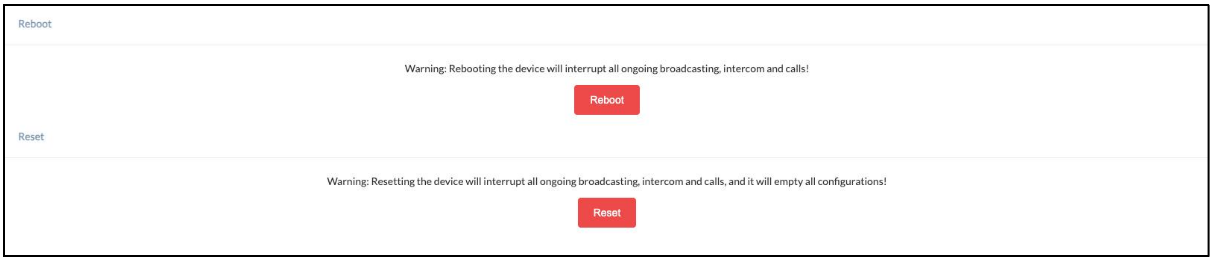
Web Password Settings

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

8.7 Reboot & Reset

SD2140 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!

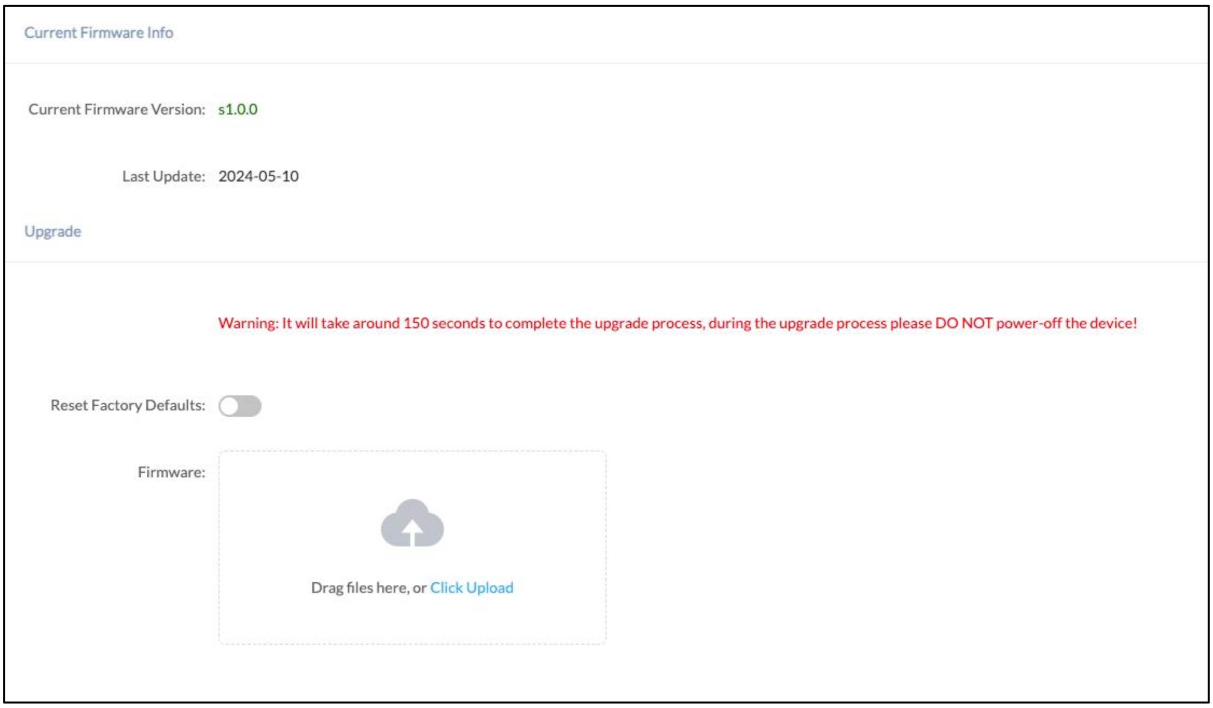
Reboot Schedule

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

9. Maintenance

9.1 Upgrade

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



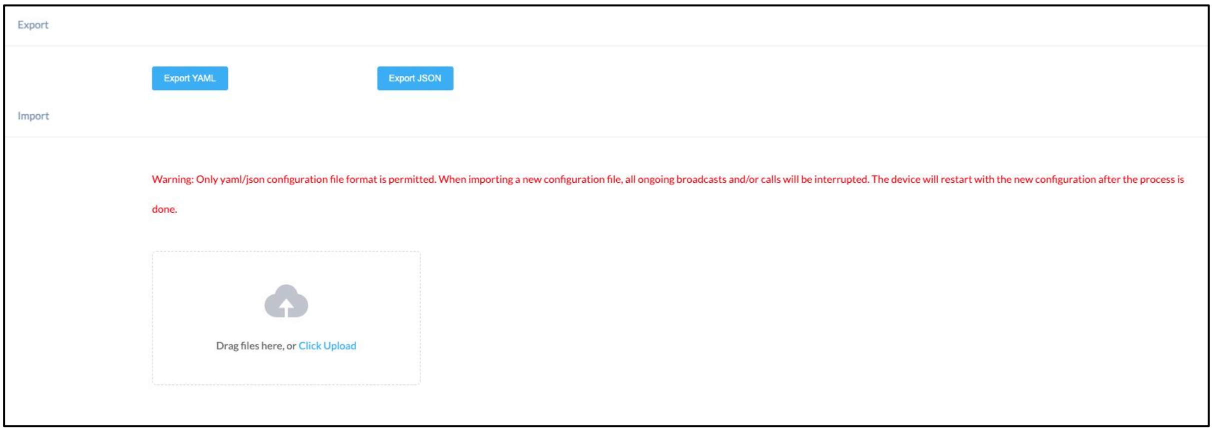
Upgrade Settings

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

9.2 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.



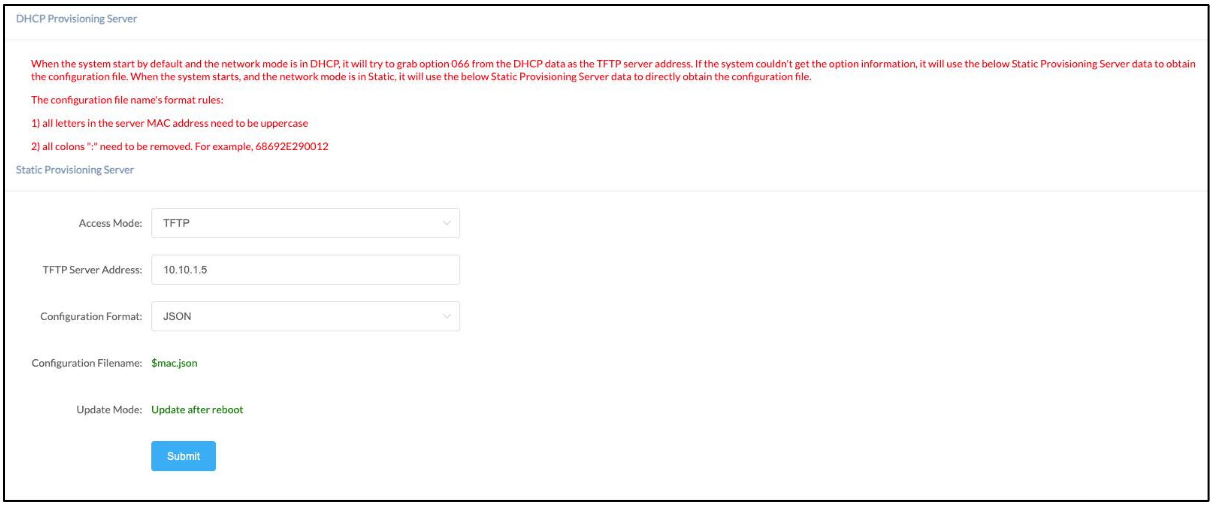
Import/Export

9.3 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

9.4 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

9.5 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.

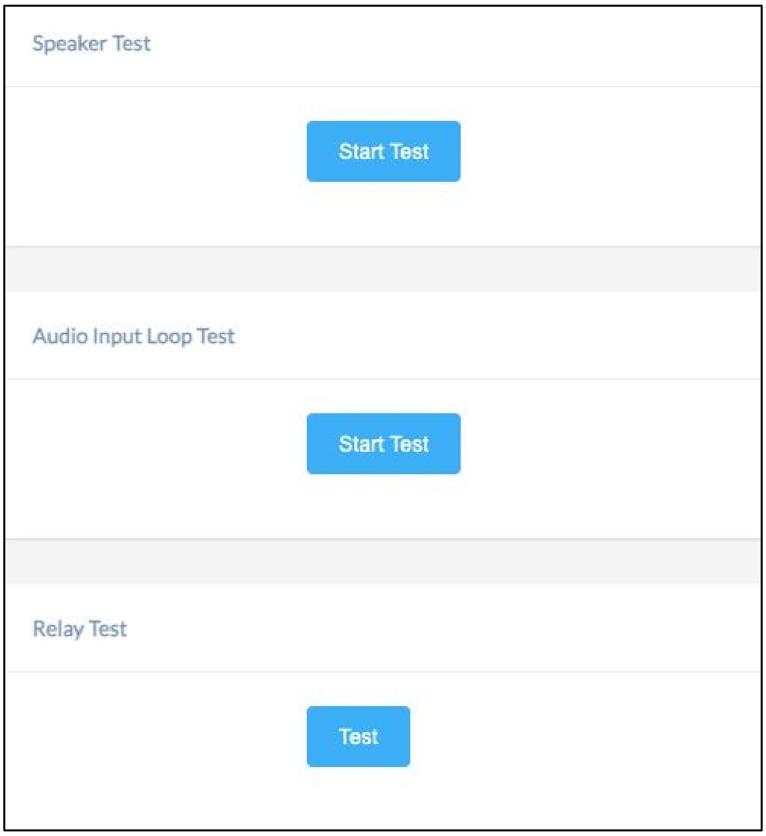
Please go to **Maintenance --> Ethernet Capture** page to operate.

Ethernet Capture

9.6 Test

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

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



Test Settings

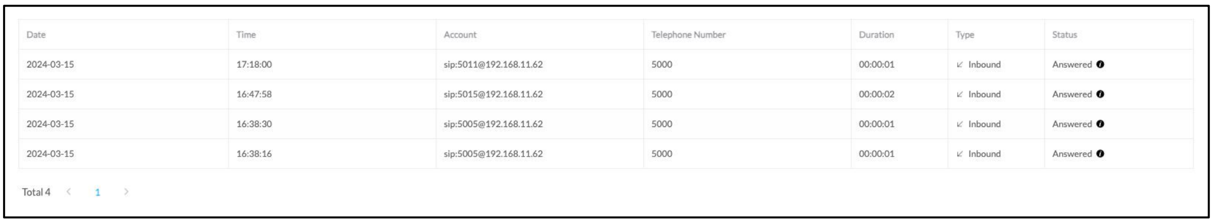
- **Speaker Test:** Click on the Start Test button, and the connected analog speaker will play a ringtone to test whether the speaker is working. If the speaker is working functionally, you should hear the voice back.

- **Audio Input Loop Test:** Click on the Start Test button, and the SD2140 routes the audio input signal back to the output, allowing users to verify the performance.
- **Relay Test:** Click on the Test button and the device will output signals to the relay for testing.

10. Reports

10.1 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.



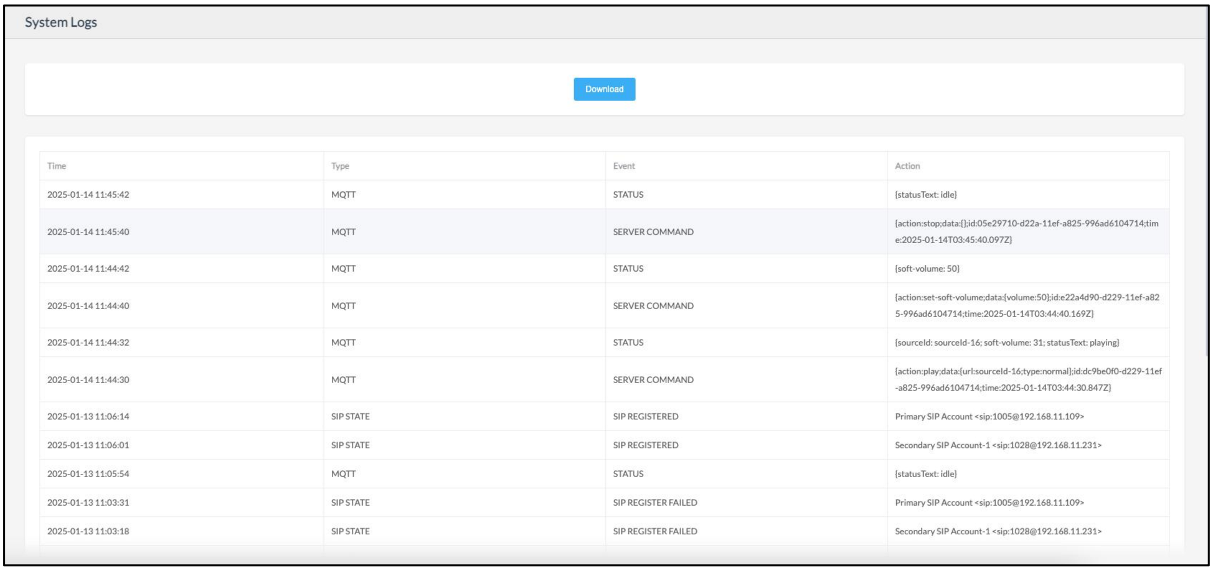
Date	Time	Account	Telephone Number	Duration	Type	Status
2024-03-15	17:18:00	sip:5011@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:47:58	sip:5015@192.168.11.62	5000	00:00:02	Inbound	Answered
2024-03-15	16:38:30	sip:5005@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:38:16	sip:5005@192.168.11.62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

Total 4 1

Call Logs

10.2 System Logs

System Logs allows you to check the event related information such as Operating Time, Operating Type (MQTT, Function, SIP, Multicast...), Event and Action details. Please go to **Reports --> System Logs** page to view the logs. Click the Download button and the .csv log file will be saved on your computer.



System Logs

Download

Time	Type	Event	Action
2025-01-14 11:45:42	MQTT	STATUS	{statusText: idle}
2025-01-14 11:45:40	MQTT	SERVER COMMAND	{action:stop;data:{};id:05e29710-d22a-11ef-a825-996ad6104714;time:2025-01-14T03:45:40.097Z}
2025-01-14 11:44:42	MQTT	STATUS	{soft-volume: 50}
2025-01-14 11:44:40	MQTT	SERVER COMMAND	{action:set-soft-volume;data:{volume:50};id:e22a4d90-d229-11ef-a825-996ad6104714;time:2025-01-14T03:44:40.169Z}
2025-01-14 11:44:32	MQTT	STATUS	{sourceId: sourceId-16; soft-volume: 31; statusText: playing}
2025-01-14 11:44:30	MQTT	SERVER COMMAND	{action:play;data:{url:sourceId-16;type:normal};id:dc9be0f0-d229-11ef-a825-996ad6104714;time:2025-01-14T03:44:30.847Z}
2025-01-13 11:06:14	SIP STATE	SIP REGISTERED	Primary SIP Account <sip:1005@192.168.11.109>
2025-01-13 11:06:01	SIP STATE	SIP REGISTERED	Secondary SIP Account-1 <sip:1028@192.168.11.231>
2025-01-13 11:05:54	MQTT	STATUS	{statusText: idle}
2025-01-13 11:03:31	SIP STATE	SIP REGISTER FAILED	Primary SIP Account <sip:1005@192.168.11.109>
2025-01-13 11:03:18	SIP STATE	SIP REGISTER FAILED	Secondary SIP Account-1 <sip:1028@192.168.11.231>

System Logs

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