SC15 Network Ceiling Speaker

User Guide

www.zycoo.com

zycoo@zycoo.com

Contents

1. Preface

1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and manage the SC15 Network Ceiling Speaker. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the SC15 Network Ceiling Speaker.

1.2 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
1.2.8	1.2.8 and 2.0.4 (hardware version 2.0 or above ONLY)	Updated operating instructions for software version v1.2.8/2.0.4.	Nov,2023

2. Overview

2.1 Product Overview

SC15 Network Ceiling Speaker is a compact high-performance SIP enabled ceiling speaker which can be used for SIP paging, notification/tone broadcasting and streamed high definition music playback. The high efficient, full-range drive units will provide you with a uniquely advanced listening experience.

2.2 Product Specifications

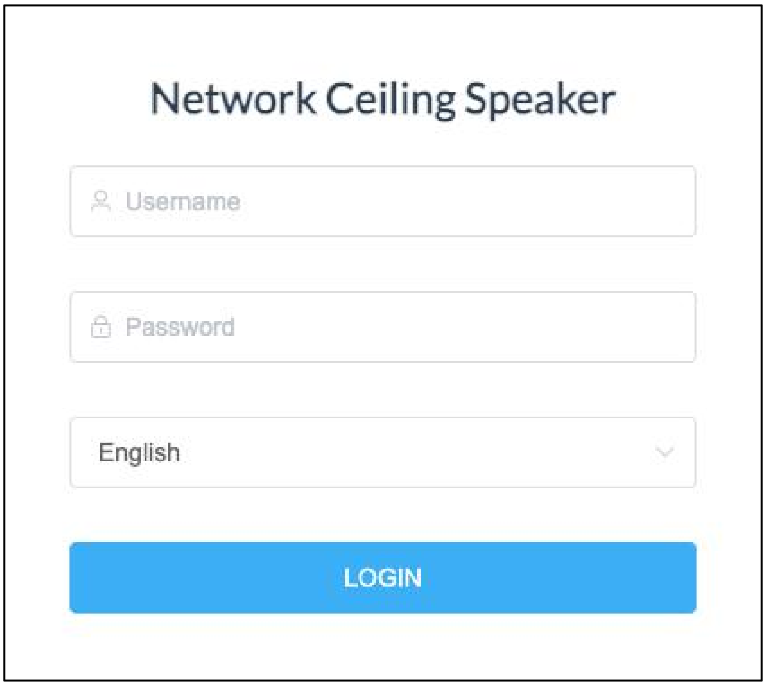
SC15 Network Ceiling Speaker Specifications		
Speaker Components	5.25" woofer unit + 1" tweeter unit	
Sensitivity	85dB / 1W / 1m	
Max Sound Pressure Level	100dB	
Rated Power	8Ω 15W	
Frequency Range	70Hz – 20KHz	
Coverage Pattern	90°H 90°V 30m²	
Amplifier	Built-in Class D Amplifier	

3. Login the Device

3.1 Accessing the Web GUI

SC15 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), or press and hold the reset button for 5 seconds and then release it to listen to the device's IP address broadcast, and 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.

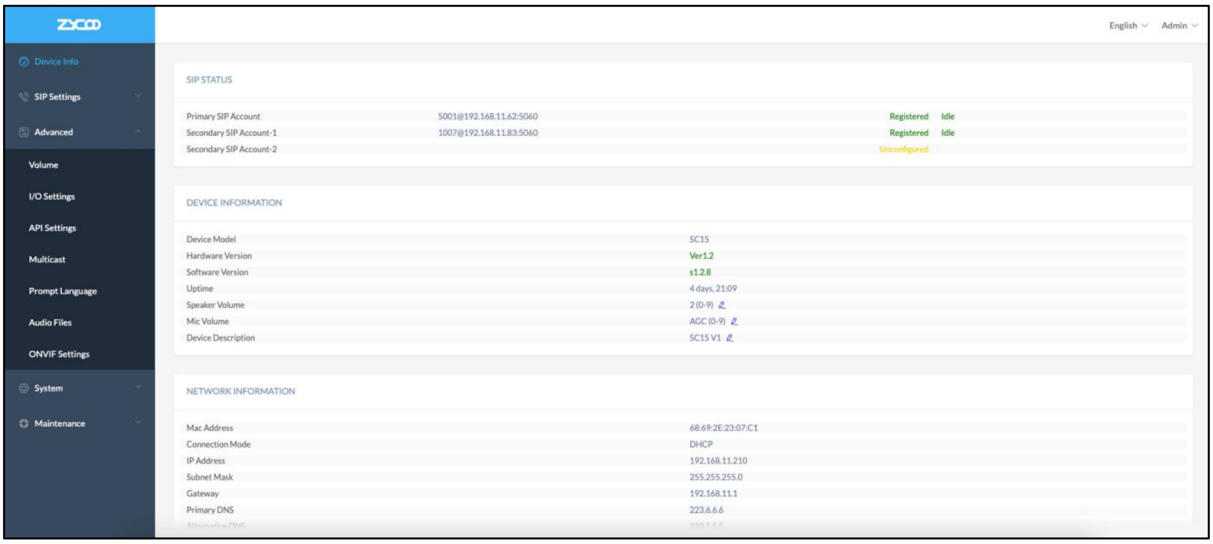


Login Interface

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

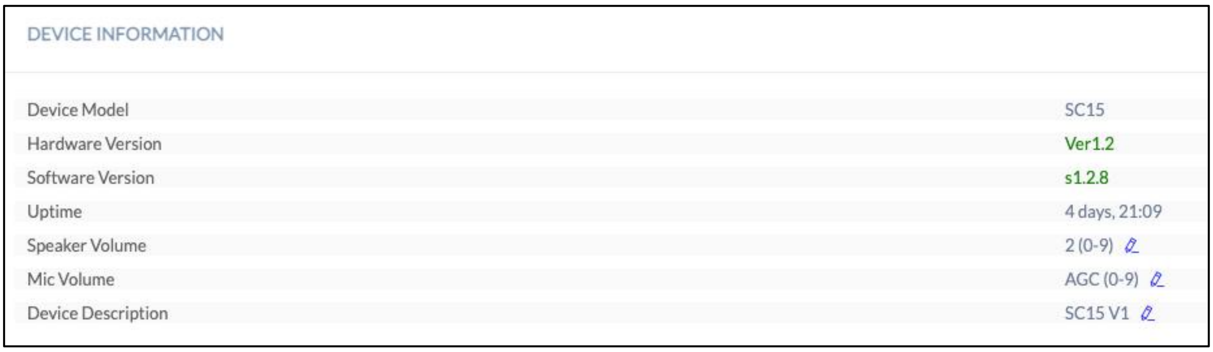
3.2 Device Info

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



SIP Status

- **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.



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:** 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	68:69:2E:23:07:C1
Connection Mode	DHCP
IP Address	192.168.11.210
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

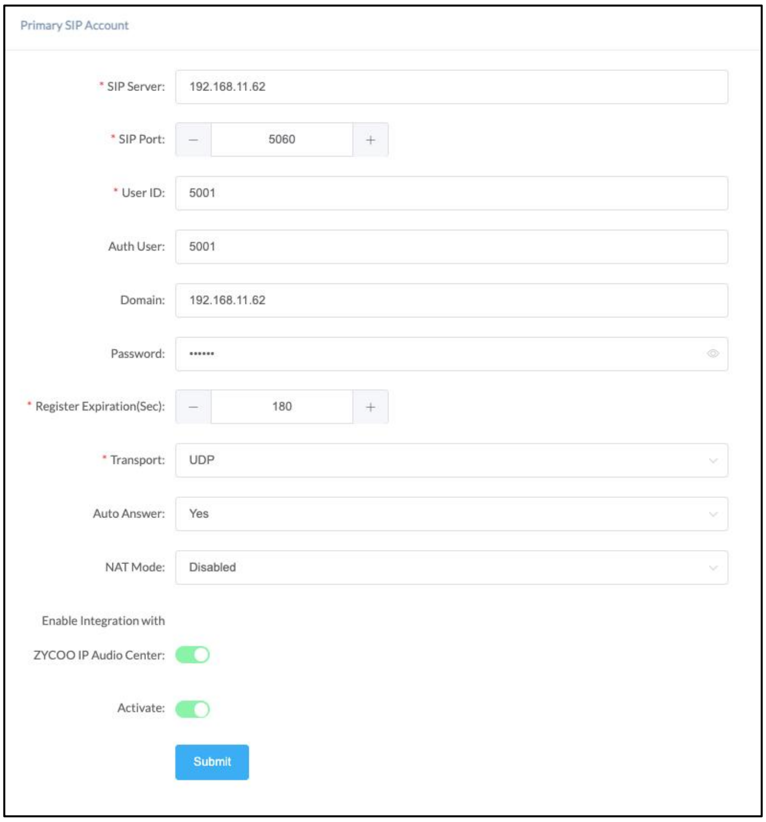
- **Mac Address:** Display the MAC address of the current device.
- **Connection Mode:** Display the network acquisition method of the device, DHCP (dynamic acquisition) or STATIC (static configuration).
- **IP Address:** The current IP address of the device.
- **Subnet Mask:** The current subnet mask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

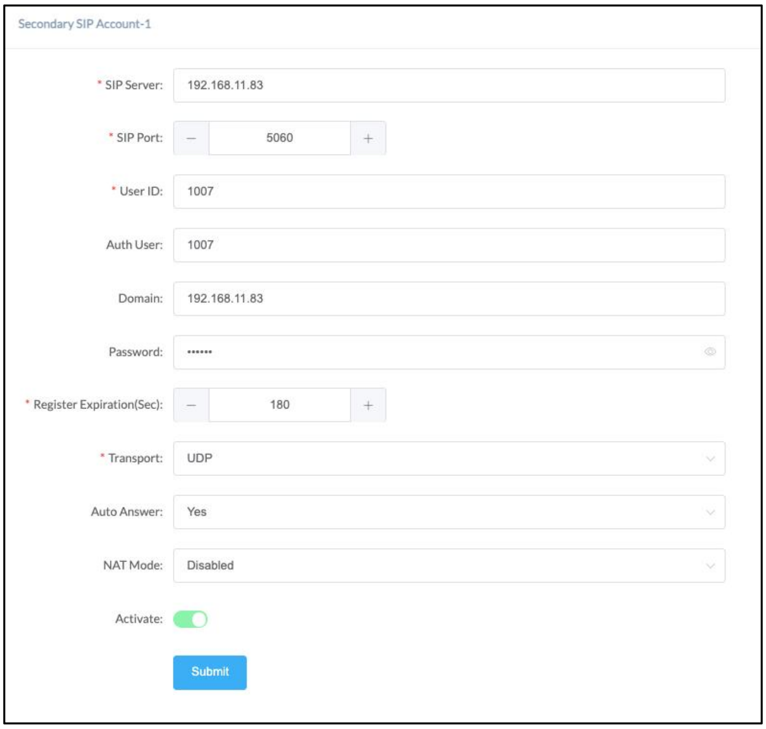
4.1 SIP Account Settings

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.

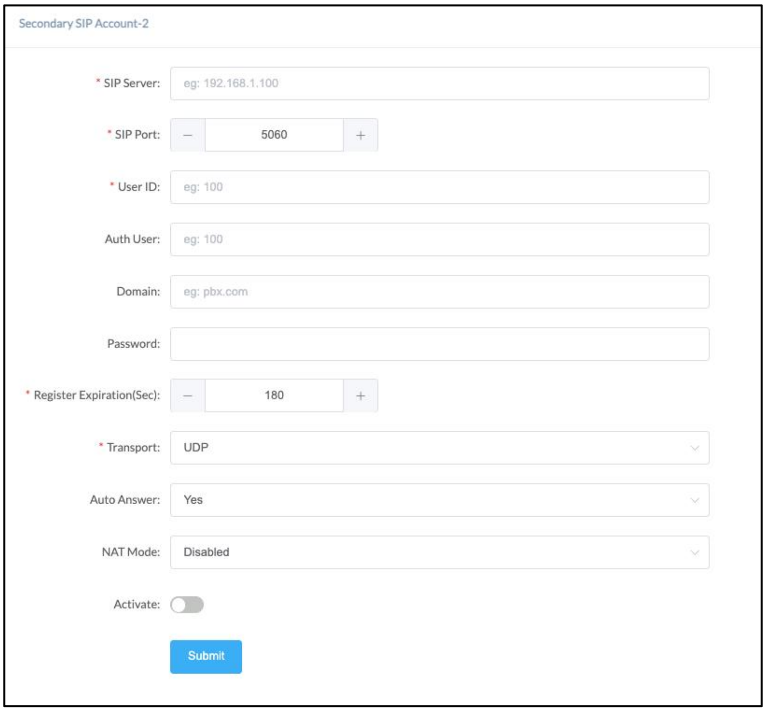
Please go to **SIP Settings --> Primary SIP Account / Secondary SIP Account-1 / Secondary SIP Account-2** page.



Primary SIP Account



Secondary SIP 1 Account



Secondary SIP 2 Account

- **SIP Server:** Enter the IP address or domain name of the SIP server.

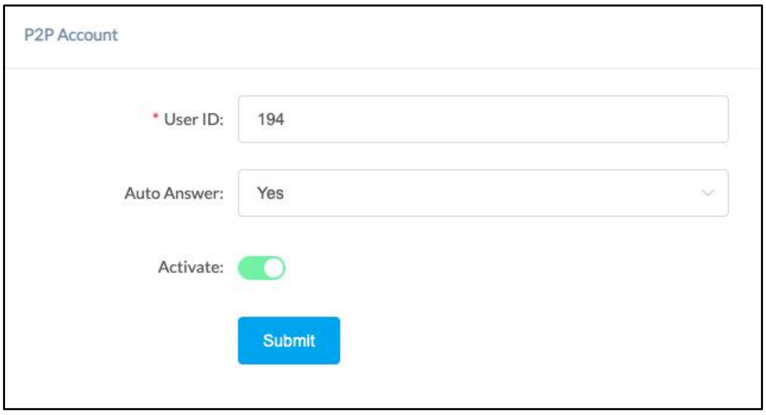
- **SIP Port:** Default SIP port is 5060. If the SIP server uses another port number as the SIP port, please modify this setting.
- **User ID:** The SIP account number provided by the SIP server.
- **Auth User:** Enter the authorized SIP account's username.
- **Domain:** Enter the SIP Domain.
- **Password:** Authorized SIP account password.
- **Register Expiration (sec):** SIP register expiration time, the default expiration time is 180 seconds.
- **Transport:** Set up the transport protocol, there are UDP, TCP, TLS options to choose.
- **Auto Answer:** Yes/No/Answer Delay, default in the Yes option.
- **NAT Mode:** Select the NAT mode and fill out the corresponding data. STUN, TURN, and ICE modes are supported.
- **Enable Integration with ZYCOO IP Audio Center:** This option is disabled by default. If you need to connect and use it with ZYCOO IP Audio Center, please enable this option. Only the main SIP account has this option.
- **Activate:** Once enabled, the account will be activated and registered to the SIP server.

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.

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.



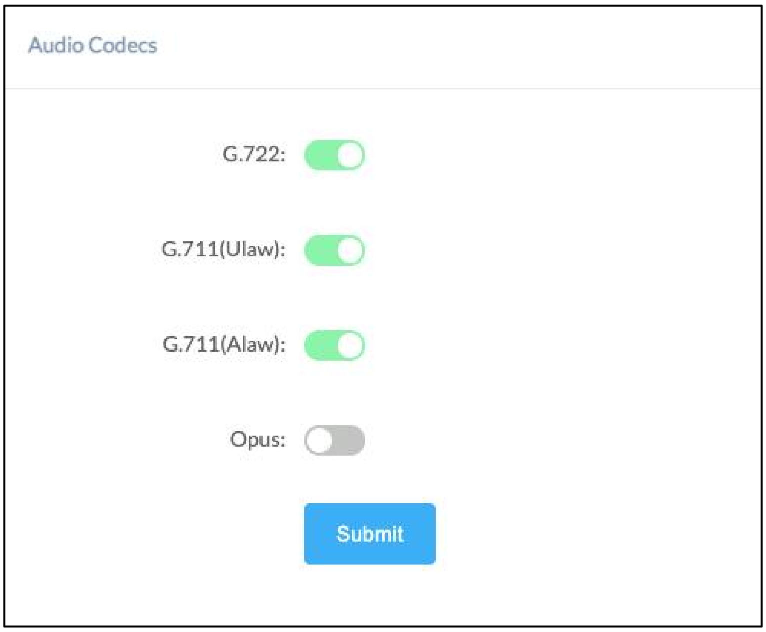
P2P Account

- **User ID:** The User ID will be displayed as the outgoing number when calling out, or the number that another device needs to dial.
- **Auto Answer:** Yes/No/Answer Delay, default in the Yes option.
- **Activate:** Enable/Disable the P2P feature.

4.3 Audio Codecs

SC15 supports 4 audio codecs: G.722 (wideband codec), G.711(Ulaw), G.711(Alaw), and Opus.

To enable or disable an audio codec/codecs, please go to **SIP Settings --> Audio Codecs** page.

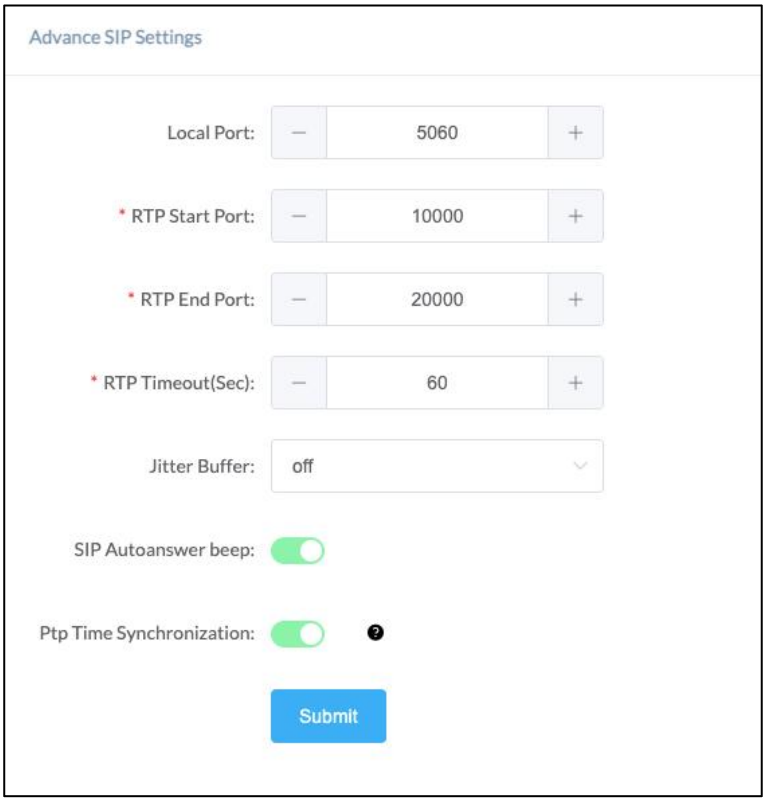


Audio Codecs

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

4.4 Advance SIP Settings

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



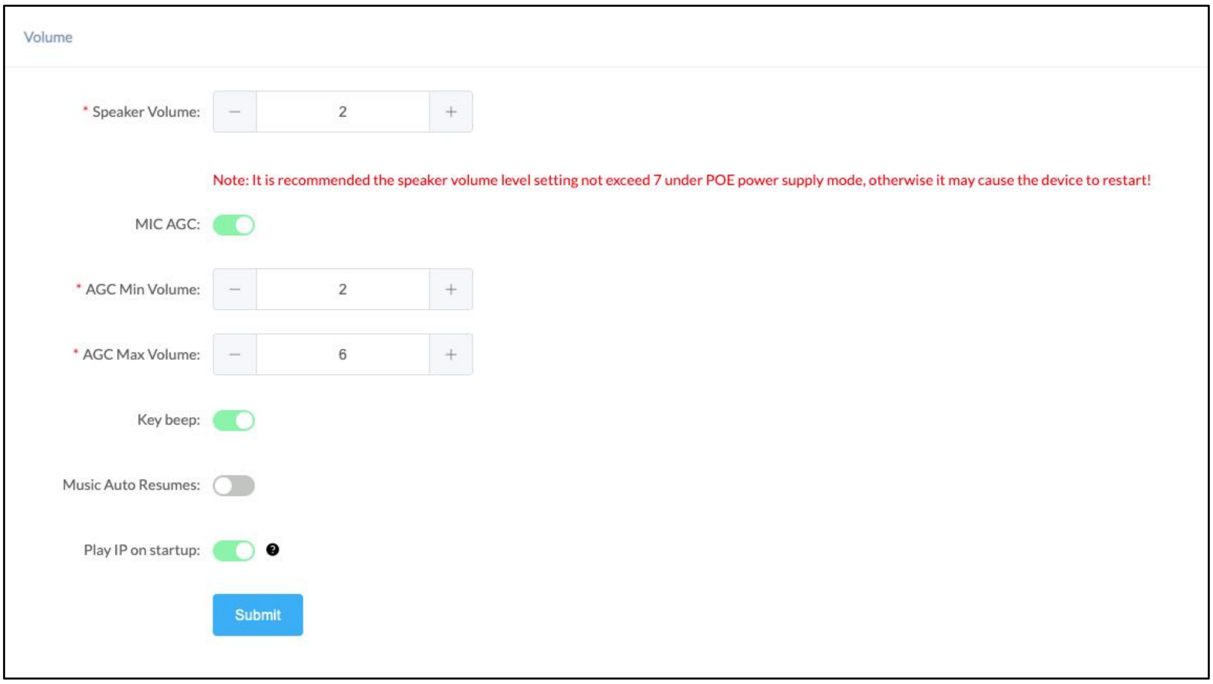
Advance SIP Settings

- **Local Port:** This setting represents the port used to receive SIP packets.
- **RTP Start Port:** This setting represents the starting RTP port that 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.
- **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.
- **SIP Autoanswer Beep:** Enable/Disable. This setting represents the ringtone beep when a call comes and only applies when the SIP Autoanswer feature is enabled.
- **PTP Time Synchronization:** Enable the PTP time synchronization with the SIP server, which requires the support of the SIP server.

5. Advanced Settings

5.1 Volume Settings

To set the volume of the SC15, please go to **Advanced Settings --> Volume** page to configure.



Volume Settings

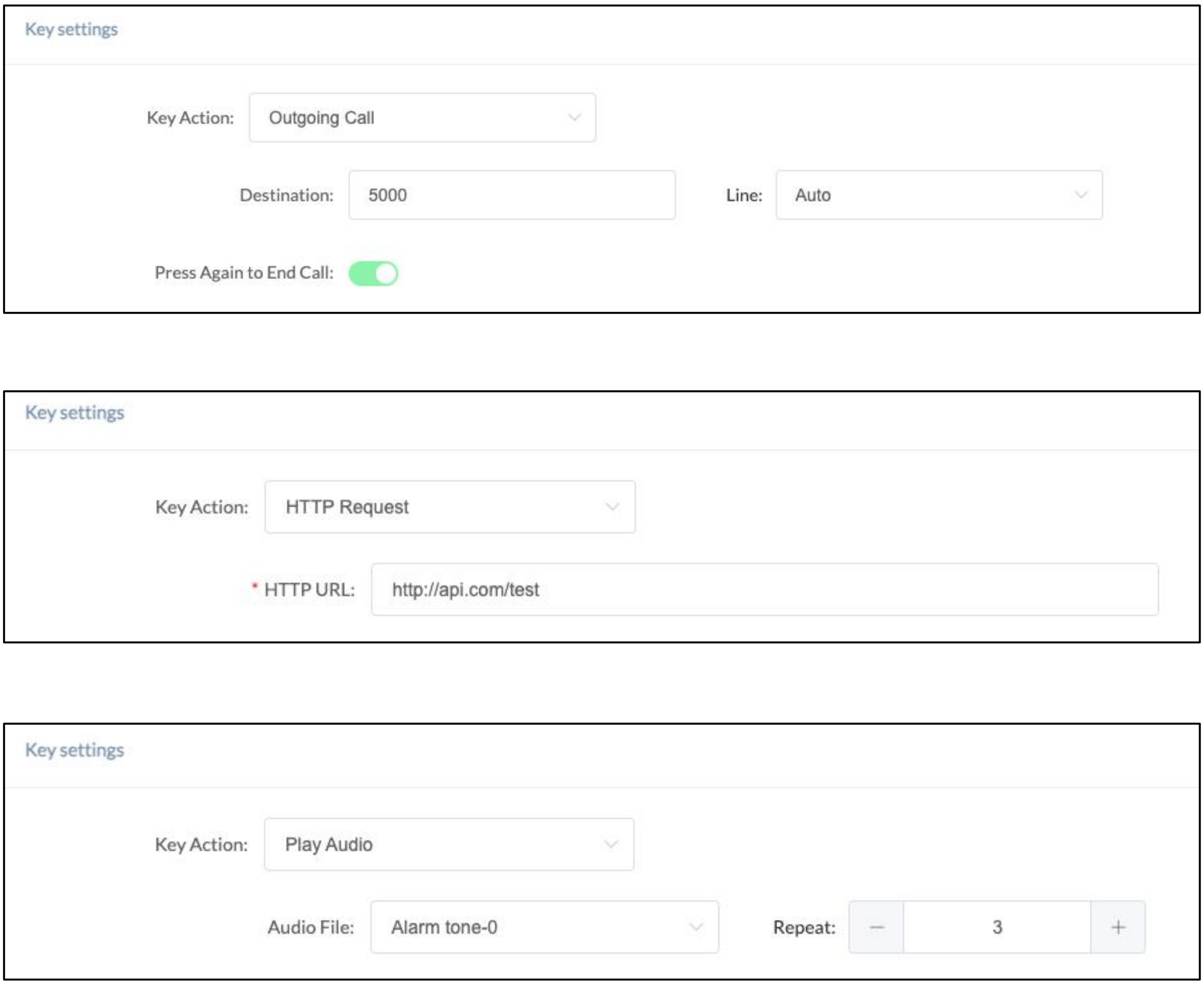
- **Speaker Volume:** The default speaker volume is 7, adjustable range is 0 ~ 9.
- **MIC AGC:** When this setting is enabled, the system will automatically adjust the microphone volume according to the environment. Users are able to adjust the microphone volume manually when this setting is disabled.
- **AGC Min Volume:** This setting represents the minimum value of the automatic gain control.
- **AGC Max Volume:** This setting represents the maximum value of the automatic gain control.
- **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.

5.2 I/O Settings

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

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

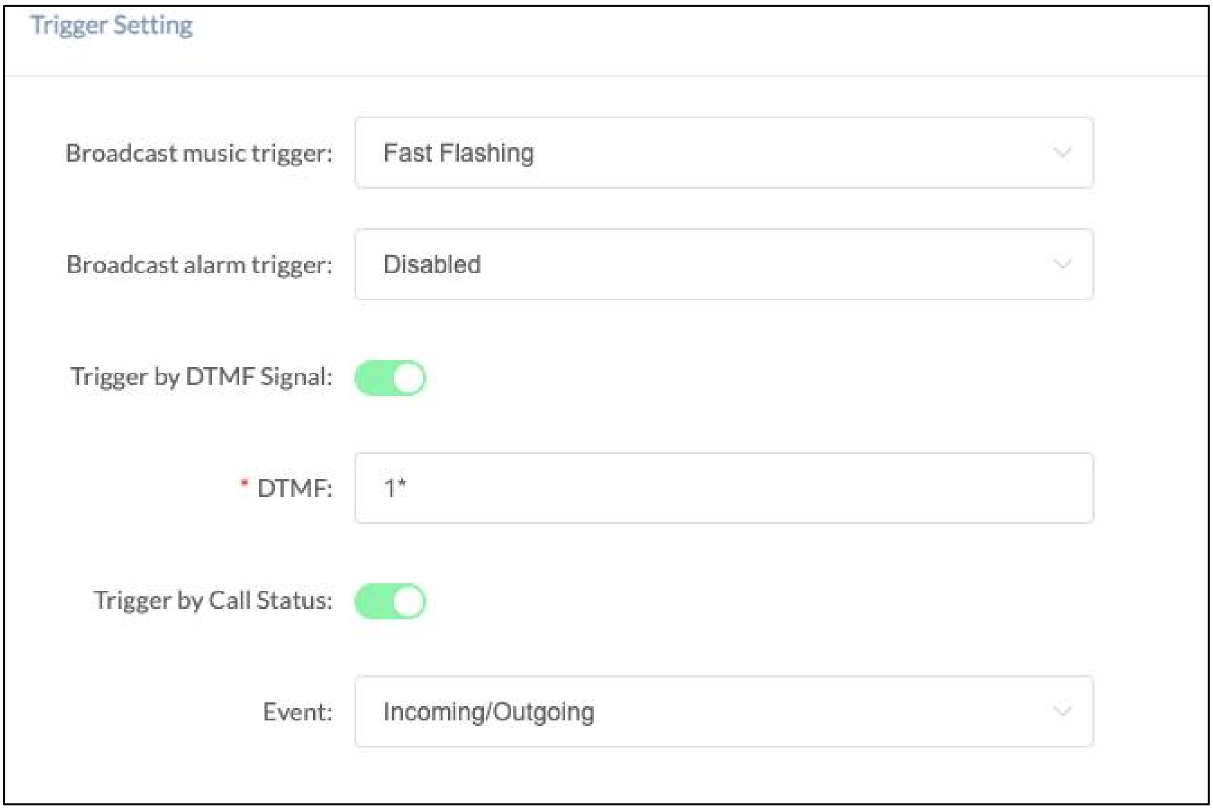


Key Settings

- **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 plus IP address, such as 101@192.168.11.123.

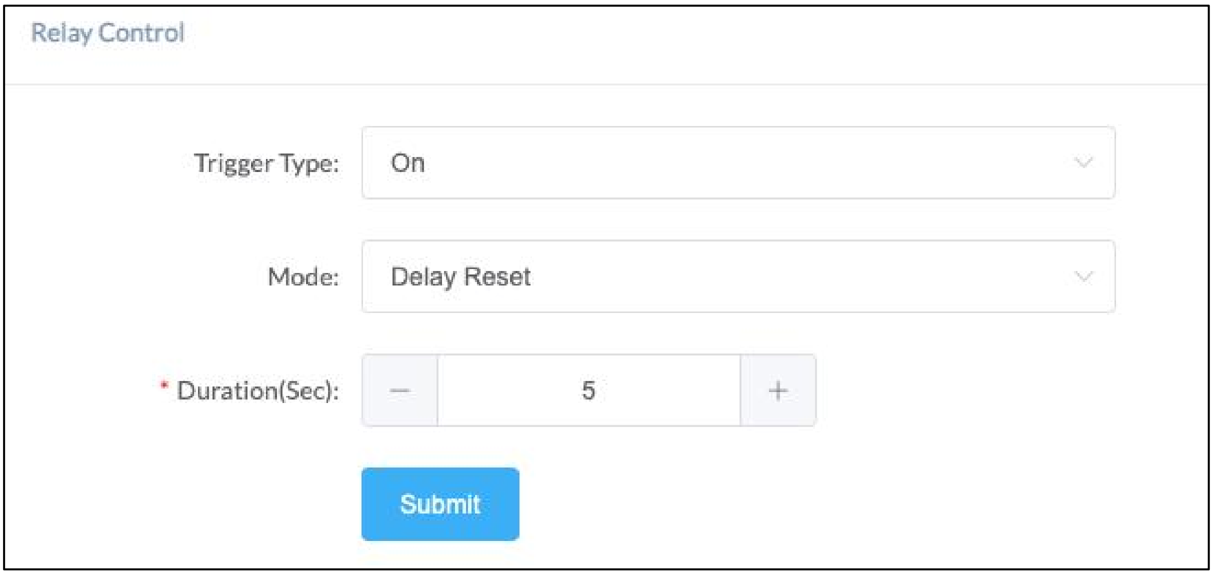
- **Press Again to End Call:** After the call is connected, users can end the call or conversation by pressing the button again.
- **HTTP URL:** Configure the API URL address triggered by linkage.
- **Audio File:** Configure the audio triggered by linkage.
- **Repeat:** Configure the times of audio repetitions triggered by linkage.



Trigger Settings

- **Broadcast Music Trigger:** Disabled/On/Fast Flashing/Slow Flashing, enable this option will trigger the relay when there is broadcast music on.

- **Broadcast Alarm Trigger:** Disabled/On/Fast Flashing/Slow Flashing, enable this option will trigger the relay when there is a broadcast alarm on.
- **Trigger by DTMF Signal:** Enable/Disable, enable this option when need to use DTMF signal to trigger (only RF2833 supported).
- **DTMF:** This setting represents the number to dial to trigger DTMF.
- **Trigger by Call Status:** Enable/Disable, enable this option will change the call status when triggered.
- **Event:** Set the corresponding call state, you can choose 【Outgoing】,【Incoming】,【Incoming/Outgoing】,【Answered】 and 【Hangup】.



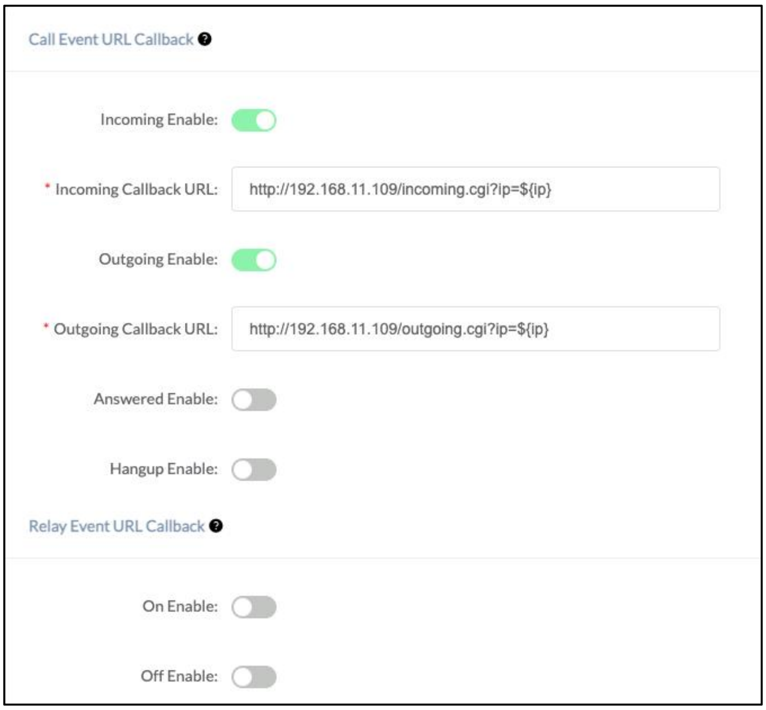
Relay Control

- **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.

5.3 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 Settings --> API Settings** page to enable API settings.



Call Event URL Callback & Relay Event URL Callback

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

For example:

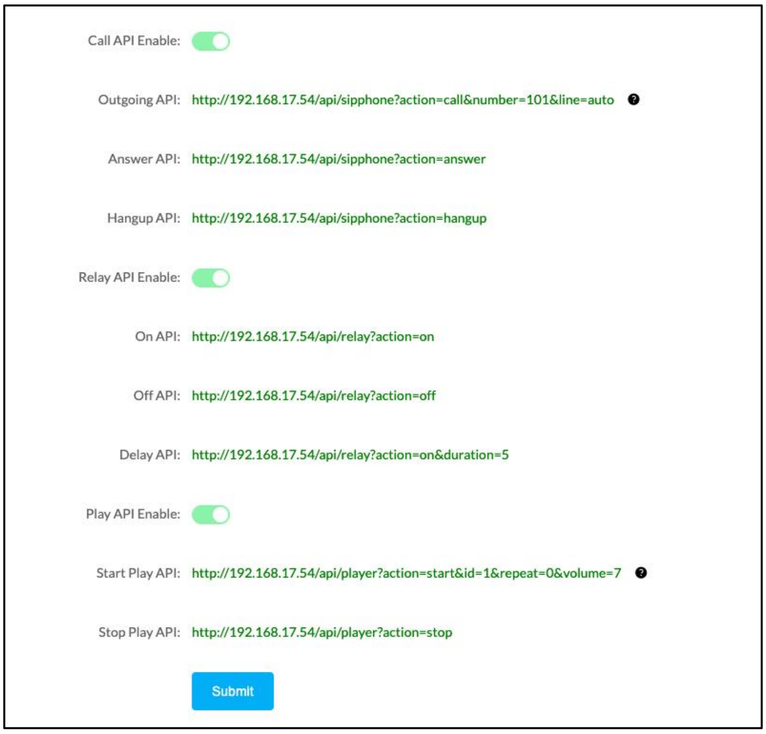
${ip}:	The current IP address of the device
${mac}:	The current MAC address of the device
${ua}:	The account of the current call
${number}:	The number of the current call

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

For example:

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

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



API Settings

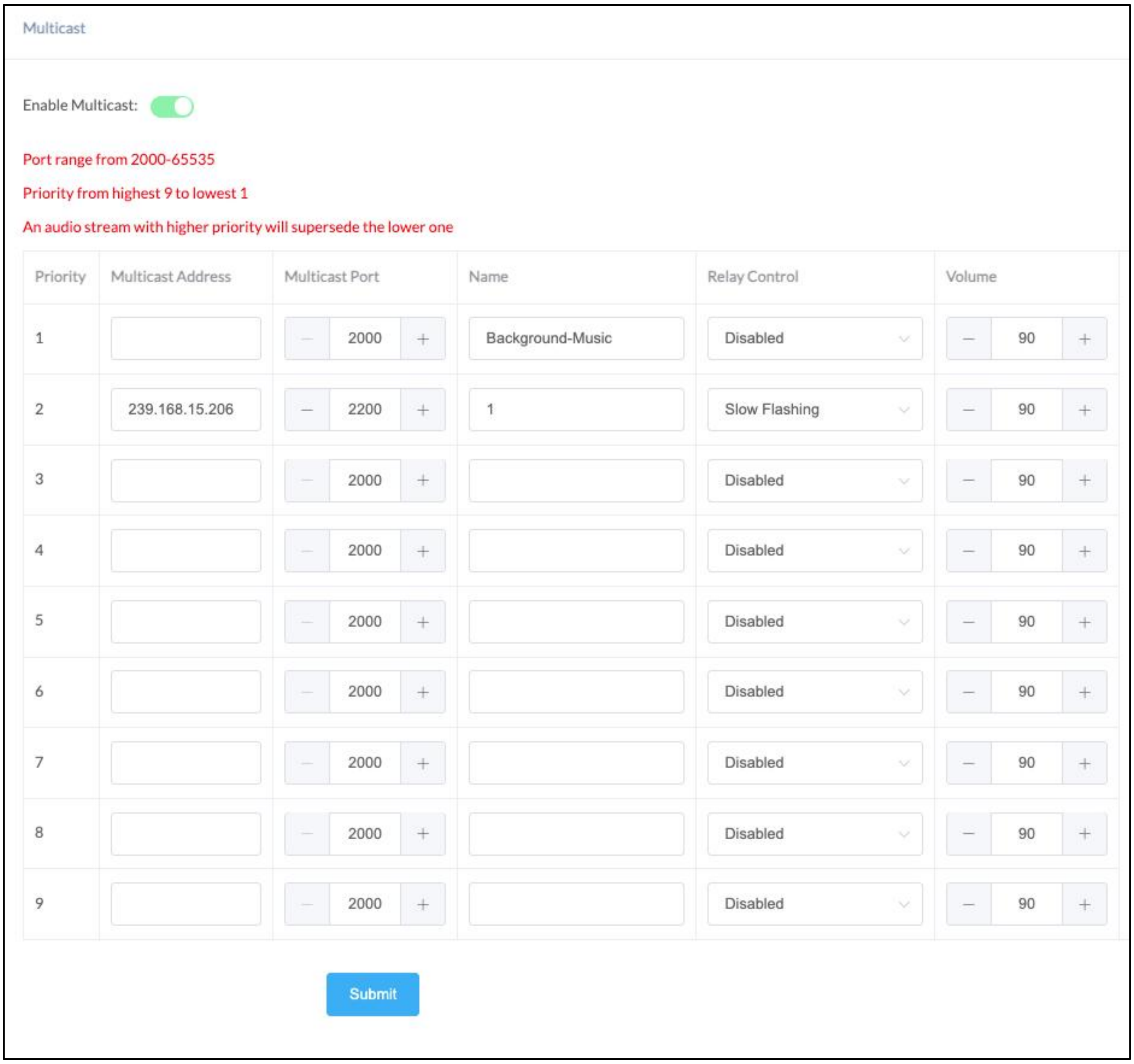
Using the API interface to realize features such as device linkage, call control, relay control, and play sound by the systems.

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

5.4 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 **Advanced Settings ---> 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:** Options to choose from are ‘Disabled’, ‘On’, ‘Fast Flashing’, ‘Slow Flashing’.

5.5 Prompt Language

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

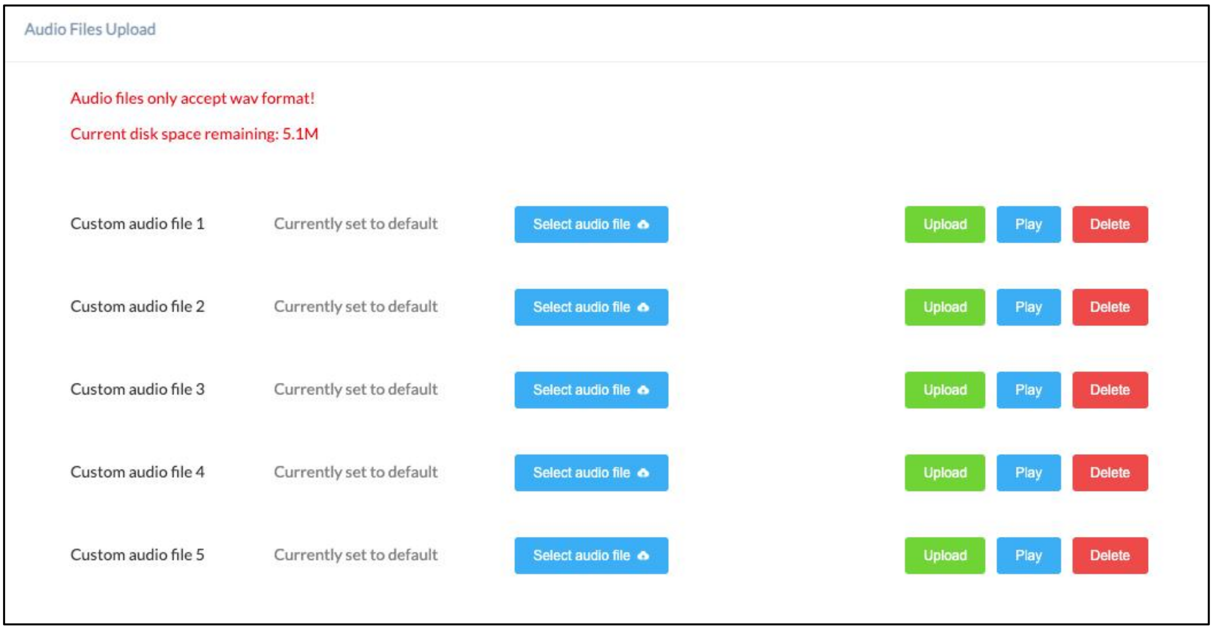
Please go to **Advanced Settings ---> Prompt Language** page to set a voice prompt language.

Prompt Language

5.6 Audio Files

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

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


Audio Files Upload
Audio files only accept wav format!
Current disk space remaining: 5.1M
Custom audio file 1
Currently set to default
Select audio file
Upload
Play
Delete
Custom audio file 2
Currently set to default
Custom audio file 3
Currently set to default
Custom audio file 4
Currently set to default
Custom audio file 5
Currently set to default

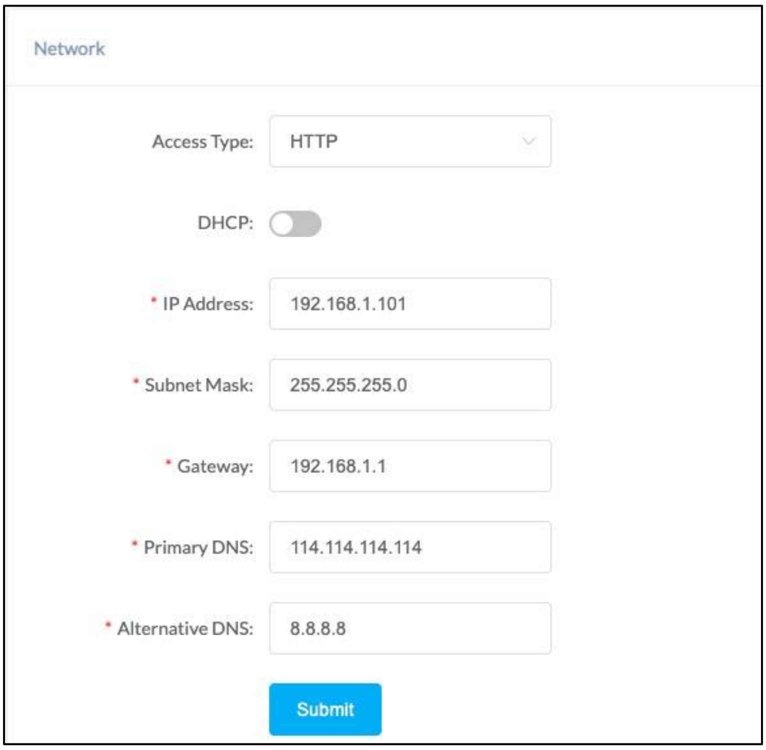
Audio Files

6. System Settings

6.1 Network

SC15 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.



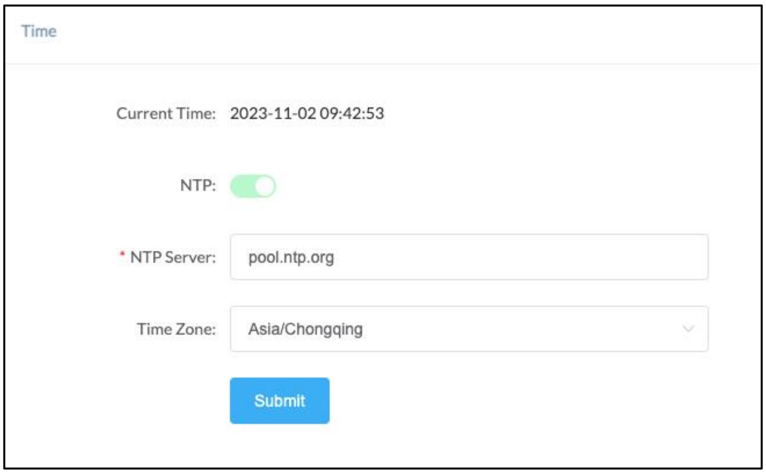
Network Configuration

- **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.

6.2 Time

SC15 obtains the time from the network time servers using NTP.
To change the NTP settings, please go to **System --> Time** page.

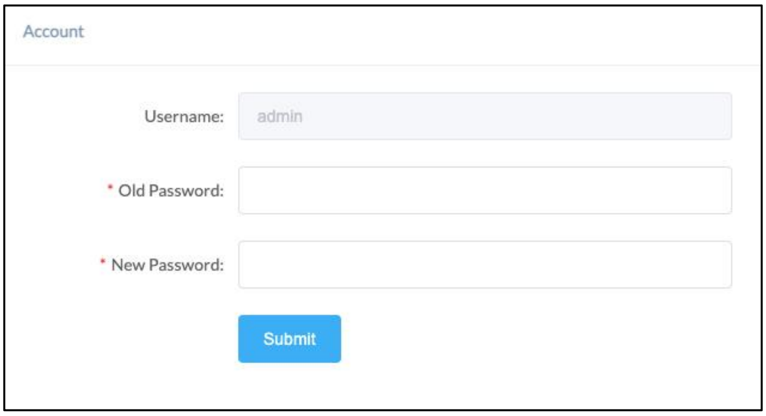


Time Settings

- **Current Time:** Display the current system time of the device.
- **NTP:** Enable/Disable using NTP to obtain the time.
- **NTP Server:** The network time server used to obtain the time.
- **Time Zone:** Set the time zone used by the device.

6.3 Account

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

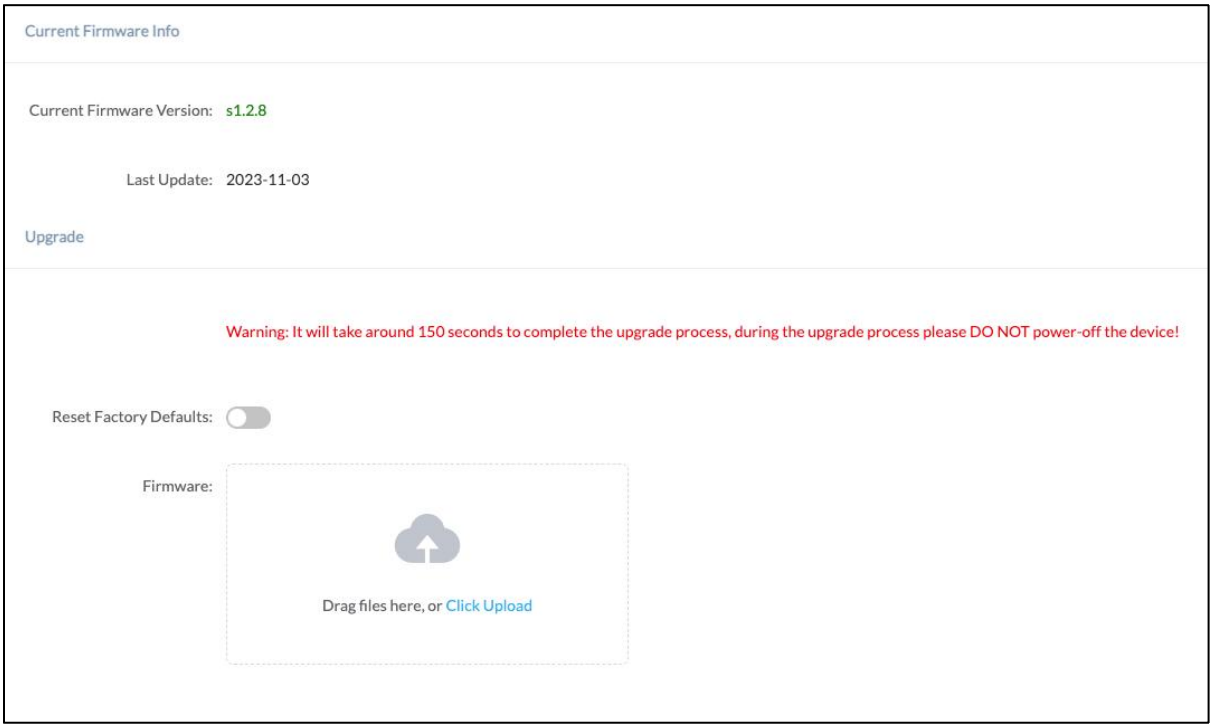


Web Password Settings

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

6.4 Upgrade

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

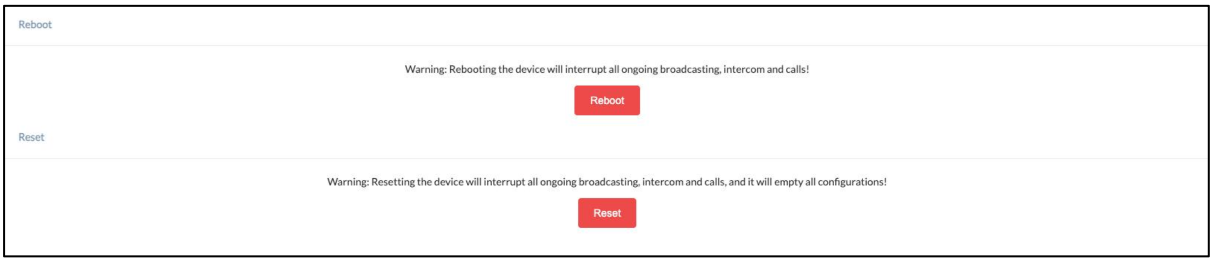


Upgrade Settings

- **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.

6.5 Reboot & Reset

SC15 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.
If you need to restore the factory settings of the SC15, you can reset it through this page or you can press and hold the RST button for more than 10 seconds and release it. After hearing the broadcast voice, the device will enter the state of restoration. The key will flash once. 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!

Reboot Schedule

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

7. Maintenance

7.1 Diagnostic

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

Ping

7.2 Ethernet Capture

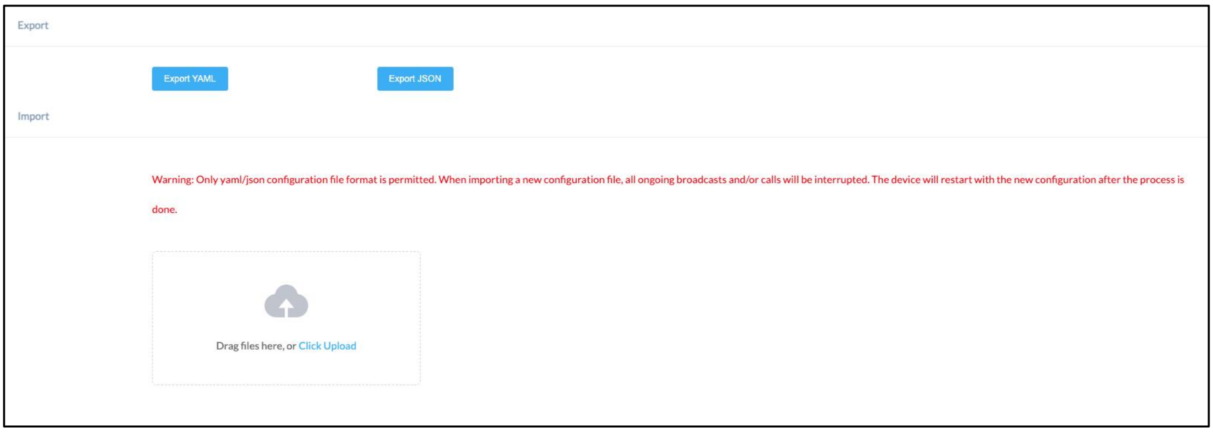
The purpose of the Ethernet capture tool is to capture Ethernet network packets and store them in a standard Wireshark-compatible packet capture '.pacp' file for immediate viewing and data analysis.
Please go to **Maintenance --> Ethernet Capture** page to operate.

Ethernet Capture

7.3 Import/Export

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

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



Import/Export

7.4 Auto Provisioning

The system is supporting DHCP Option 066 and static TFTP/HTTP two auto provisioning methods.

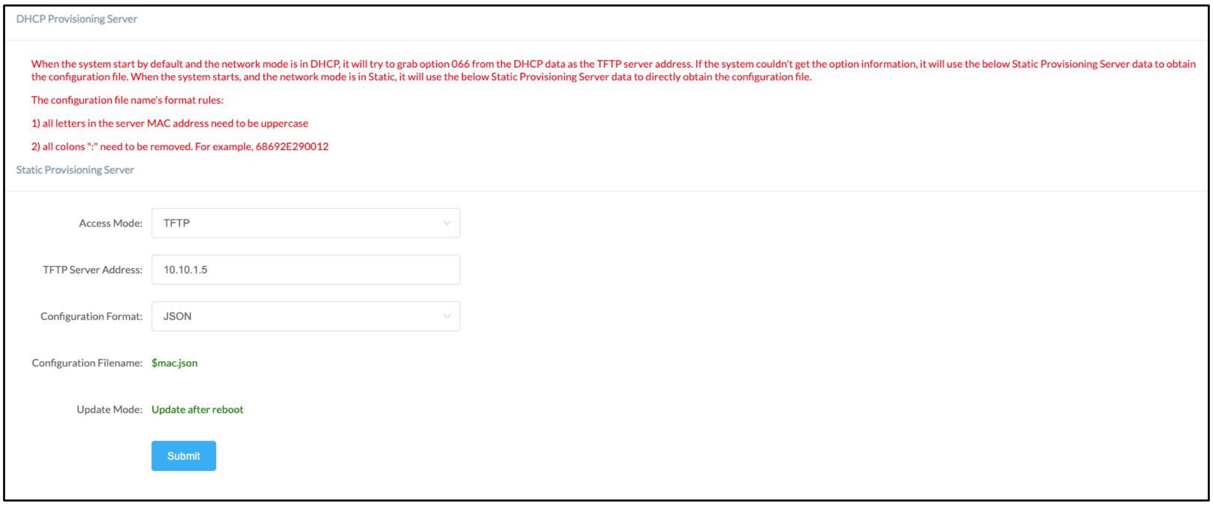
When the system starts by default and the network mode is in DHCP, it will try to grab option 066 from the DHCP data as the TFTP server address. If the system couldn't get the option information, it will use the below Static Provisioning Server data to obtain the configuration file. When the system starts,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

The configuration file name's format rules:

1) all letters in the server MAC address need to be uppercase.

2) all colons ":" need to be removed. For example, 68692E290012.

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

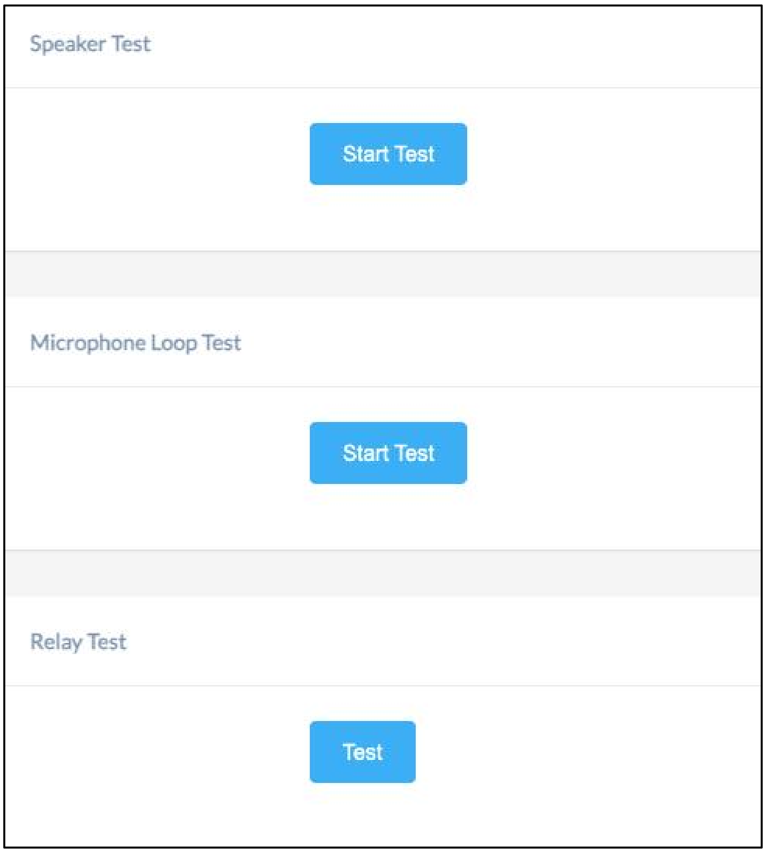


Auto Provisioning

7.5 Test

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

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



Test Settings

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

www.zycoo.com
zycoo@zycoo.com