



SH10 Network Horn Speaker

User Guide



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1. Preface

1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and manage the SH10 Network Horn Speaker. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the SH10 Network Horn Speaker.

1.2 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
1.0.0	1.0.0	Updated operating instructions for software version v1.0.0.	May,2024

2. Overview

2.1 Product Overview

SH10 Network Horn Speaker is a SIP enabled horn speaker which provides exceptionally clear and intelligible voice for SIP paging, notification/tone broadcasting and streamed background music. It is suitable to be deployed in outdoor and other open spaces for public notifications and public safety purposes.

2.2 Product Specifications

SH10 Network Horn Speaker Specifications	
Speaker Components	1.5" midrange driver unit
Sensitivity	105dB/1m/1W
Max Sound Pressure Level	115dB
Rated Power	DC: 8Ω 15W PoE: af-8Ω 8W /at-8Ω 15W
Frequency Range	400Hz – 8KHz
Coverage Pattern	50°H 50°V, effective distance 70m
Amplifier	Class D Amplifier



3. Login the Device

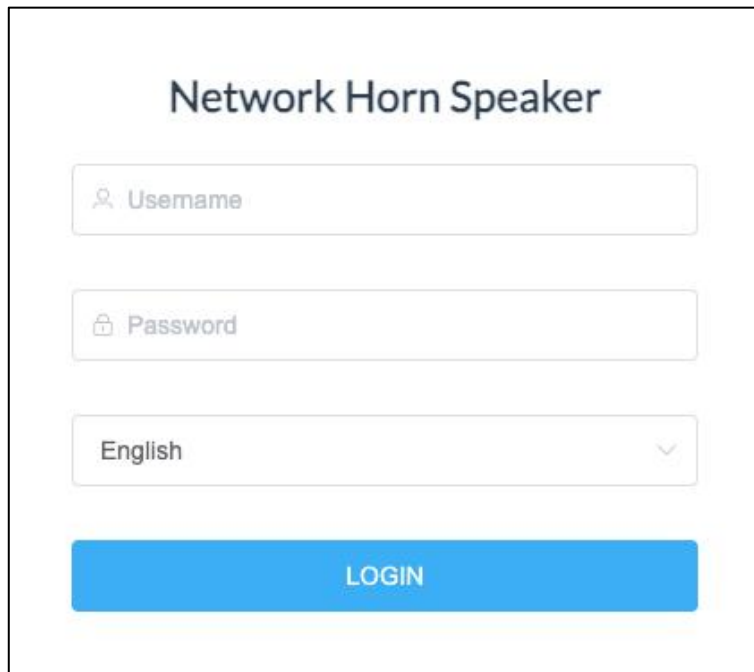
3.1 Accessing the Web GUI

SH10 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), 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.



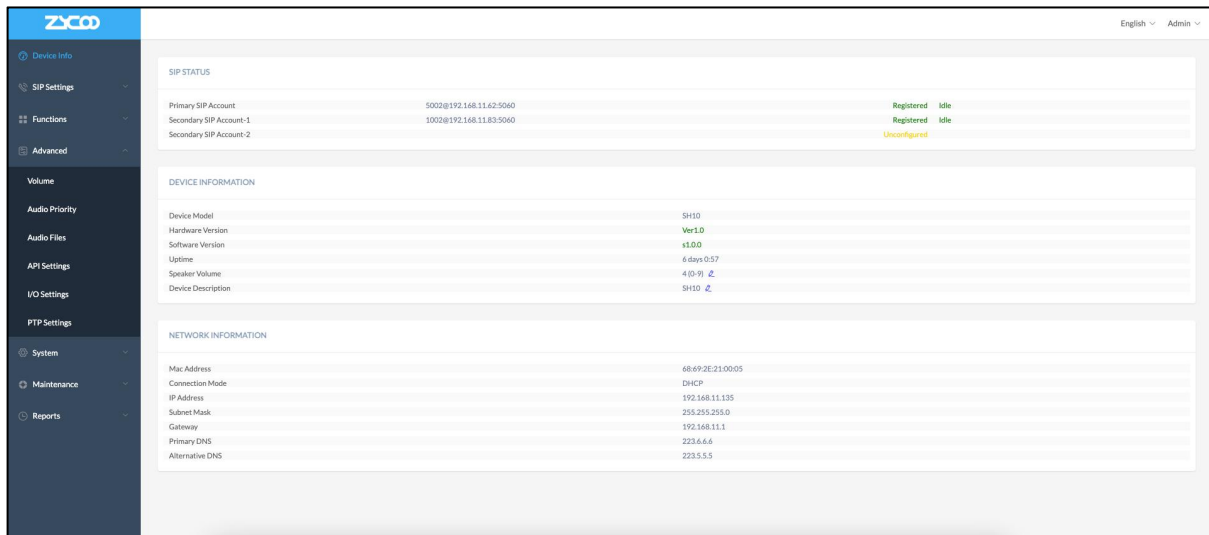
The screenshot shows the login page for the Network Horn Speaker. At the top, the title "Network Horn Speaker" is displayed. Below the title are three input fields: "Username" (with a magnifying glass icon), "Password" (with a lock icon), and a language dropdown menu (currently set to "English"). At the bottom of the form is a blue "LOGIN" button.

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	5002@192.168.11.62:5060	Registered	Idle
Secondary SIP Account-1	1002@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.

DEVICE INFORMATION	
Device Model	SH10
Hardware Version	Ver1.0
Software Version	s1.0.0
Uptime	3 days, 19:39
Speaker Volume	7 (0-9) ↕
Device Description	SH10 ↕

Device Information

- Device Model:** Displays the model of the device.

- **Hardware Version:** Displays the hardware version number of the device.
- **Software Version:** Displays the system version number of the device.
- **Start Time:** Displays the last time the device was started up.
- **Speaker Volume:** Displays the current volume of the device.
- **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	
Connection Mode	DHCP
IP Address	192.168.11.168
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

* SIP Server: 192.168.11.62

* SIP Port: - 5060 +

* User ID: 5001

Auth User: 5001

Domain: 192.168.11.62

Password:

* Register Expiration(Sec): - 180 +

* Transport: UDP

Auto Answer: Yes

NAT Mode: Disabled

Enable Integration with

ZYCOO IP Audio Center:

Activate:

Submit

Primary SIP Account

Secondary SIP Account-1

* SIP Server: 192.168.11.83

* SIP Port: 5060

* User ID: 1007

Auth User: 1007

Domain: 192.168.11.83

Password: *****

* Register Expiration(Sec): 180

* Transport: UDP

Auto Answer: Yes

NAT Mode: Disabled

Activate:

Submit

Secondary SIP 1 Account

Secondary SIP Account-2

* SIP Server: eg: 192.168.1.100

* SIP Port: 5060

* User ID: eg: 100

Auth User: eg: 100

Domain: eg: pbx.com

Password:

* Register Expiration(Sec): 180

* Transport: UDP

Auto Answer: Yes

NAT Mode: Disabled

Activate:

Submit

Secondary SIP 2 Account

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

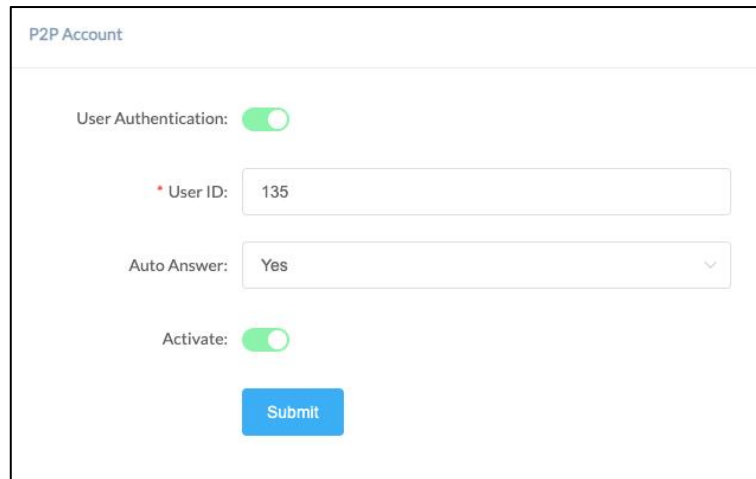
- **SIP Port:** Default SIP port is 5060. If the SIP server uses another port number as the SIP port, please modify this setting.
- **User ID:** The SIP account number provided by the SIP server.
- **Auth User:** Enter the authorized SIP account's username.
- **Domain:** Enter the SIP Domain.
- **Password:** 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 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 Authentication:

* User ID:

Auto Answer:

Activate:

P2P Account

- **User Authentication:** Enable/Disable the P2P authentication.
- **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 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:

* RTP Start Port:

* RTP End Port:

* RTP Timeout(Sec):

Jitter Buffer:

Acoustic Echo Cancellation:

Adaptive Noise Reduction:

Automatic Generation Control:

Comfort Noise Generator:

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:

Beep Sound File: Start Beep

Beep Volume: - 90 +

Answer Remote Beep:

Beep Sound File: Start Beep

Beep Volume: - 90 +

Hangup Beep:

Beep Sound File: End Beep

Beep Volume: - 90 +

SIP Function Settings

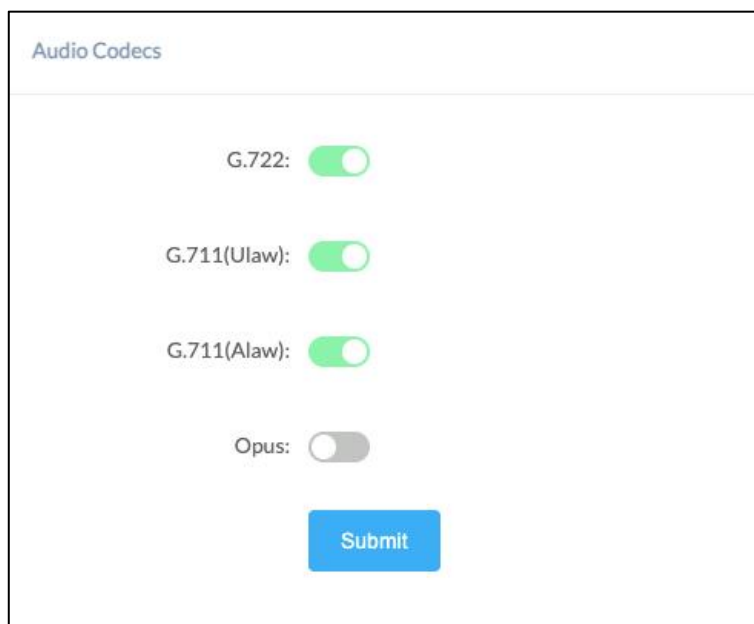
- **Answer Local Beep:** If this setting is enabled, the selected beep sound will be played first on the local device side after the SIP session is answered.
- **Beep Sound File:** Select a specific beep sound file. Click the Play button, you could listen to this audio file.
- **Beep Volume:** Set the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

4.3.3 Audio Codecs

SH10 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

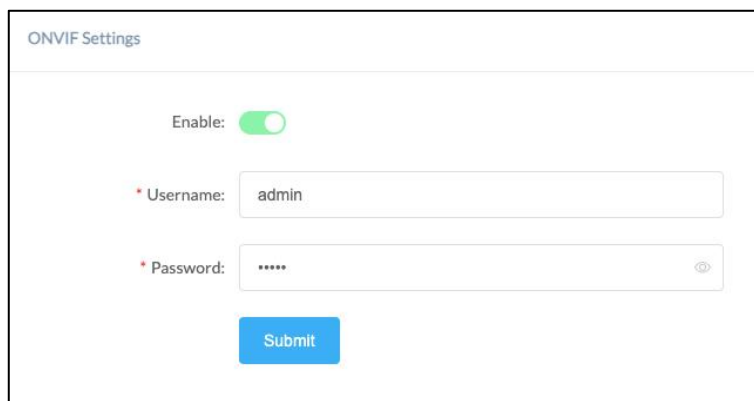
Please keep at least one codec enabled and supported by the SIP server, otherwise, SIP paging will not work.

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 **Advanced Settings** ---> **ONVIF Settings** to configure the ONVIF settings. For detailed multi-platform configuration guide, please refer to the ONVIF Feature Guide document on the ZYCOO official website.



ONVIF Settings

Enable:

* Username:

* Password:

ONVIF Settings

5.2 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

Enable Multicast:

Port range from 2000-65535

Priority from highest 9 to lowest 1

An audio stream with higher priority will supersede the lower one

Priority	Multicast Address	Multicast Port	Name	Relay Control	Volume
1	<input type="text"/>	<input type="text"/> 2000 <input type="text"/>	Background-Music	Disabled	<input type="text"/> 90 <input type="text"/>
2	239.168.15.206	<input type="text"/> 2200 <input type="text"/>	1	Slow Flashing	<input type="text"/> 90 <input type="text"/>
3	<input type="text"/>	<input type="text"/> 2000 <input type="text"/>	<input type="text"/>	Disabled	<input type="text"/> 90 <input type="text"/>
4	<input type="text"/>	<input type="text"/> 2000 <input type="text"/>	<input type="text"/>	Disabled	<input type="text"/> 90 <input type="text"/>
5	<input type="text"/>	<input type="text"/> 2000 <input type="text"/>	<input type="text"/>	Disabled	<input type="text"/> 90 <input type="text"/>
6	<input type="text"/>	<input type="text"/> 2000 <input type="text"/>	<input type="text"/>	Disabled	<input type="text"/> 90 <input type="text"/>
7	<input type="text"/>	<input type="text"/> 2000 <input type="text"/>	<input type="text"/>	Disabled	<input type="text"/> 90 <input type="text"/>
8	<input type="text"/>	<input type="text"/> 2000 <input type="text"/>	<input type="text"/>	Disabled	<input type="text"/> 90 <input type="text"/>
9	<input type="text"/>	<input type="text"/> 2000 <input type="text"/>	<input type="text"/>	Disabled	<input type="text"/> 90 <input type="text"/>

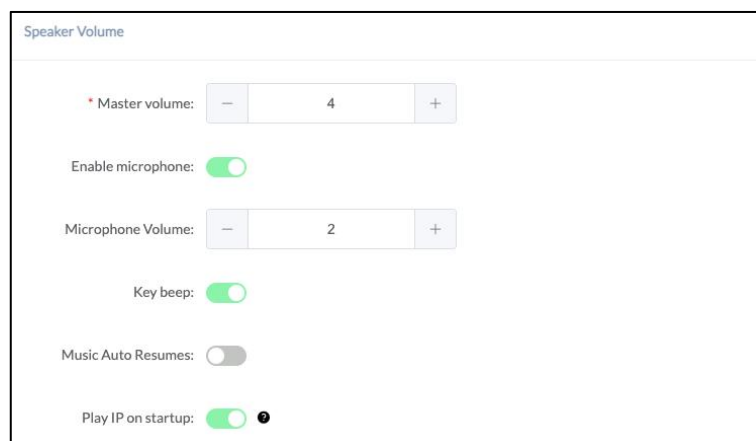
Multicast

- **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’.

6. Advanced Settings

6.1 Volume Settings

To set the volume of the SH10, please go to **Advanced Settings** --> **Volume** page to configure. You can adjust the main volume as well as individual volume adjustments for each application.



Speaker Volume Settings

- **Master Volume:** Set the speaker master volume. The default volume is 7 and the adjustable range is 0~9.
- **Enable Microphone:** Enable/Disable the built-in microphone.
- **Microphone Volume:** Set the microphone volume. The default volume is 7 and the adjustable range is 0~9.
- **Key Beep:** Enable/Disable the beep sound from the key button.
- **Music Auto Resumes:** When the device restarts or reconnects to the network, the previous music tasks will be automatically restored.
- **Play IP on Startup:** When the device starts, it automatically broadcasts its IP address once.

App Volume

SIP Volume: 80

ONVIF Volume: 80

BROADCAST Volume: 48

MULTICAST Volume 1: 80

MULTICAST Volume 2: 50

MULTICAST Volume 3: 80

MULTICAST Volume 4: 80

MULTICAST Volume 5: 80

MULTICAST Volume 6: 80

MULTICAST Volume 7: 80

MULTICAST Volume 8: 80

MULTICAST Volume 9: 80

Submit

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.

6.2 Audio Priority Settings

The audio priority can be set according to different applications(SIP, ONVIF, MULTICAST, BROADCAST). Priority 1 is the highest. You can drag the arrow on the right side to adjust the priority.

Audio Priority

Priority 1 is the highest and can be adjusted by dragging.

Priority	Application name	Operation
1	SIP	▼
2	ONVIF	▲ ▼
3	MULTICAST	▲ ▼
4	BROADCAST	▲

[Submit](#)






Audio Priority Settings

6.3 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	Select audio file 	Upload	Play	Delete
Custom audio file 3	Currently set to default	Select audio file 	Upload	Play	Delete
Custom audio file 4	Currently set to default	Select audio file 	Upload	Play	Delete
Custom audio file 5	Currently set to default	Select audio file 	Upload	Play	Delete

Audio Files

6.4 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.

The screenshot displays two sections of the configuration interface:

- Call Event URL Callback** (with a help icon):
 - Incoming Enable:
 - Outgoing Enable:
 - Answered Enable:
 - Hangup Enable:
 - Register Failed Enable:
- Relay Event URL Callback** (with a help icon):
 - On Enable:
 - Off Enable:

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:

<code>\${ip}</code> :	The current IP address of the device
<code>\${mac}</code> :	The current MAC address of the device
<code>\${ua}</code> :	The account of the current call
<code>\${number}</code> :	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:

<code>\${ip}</code> :	The current IP address of the device
<code>\${mac}</code> :	The current MAC address of the device

Call API Enable:

Outgoing API: <http://192.168.17.54/api/sipphone?action=call&number=101&line=auto>

Answer API: <http://192.168.17.54/api/sipphone?action=answer>

Hangup API: <http://192.168.17.54/api/sipphone?action=hangup>

Relay API Enable:

On API: <http://192.168.17.54/api/relay?action=on>

Off API: <http://192.168.17.54/api/relay?action=off>

Delay API: <http://192.168.17.54/api/relay?action=on&duration=5>

Play API Enable:

Start Play API: <http://192.168.17.54/api/player?action=start&id=1&repeat=0&volume=7>

Stop Play API: <http://192.168.17.54/api/player?action=stop>

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.

6.5 I/O Settings

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

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

Input Settings

Input Type: Key Signal

Key Event Action: Outgoing Call

Destination: 5000

Line: Auto

Press Again to End Call:

Input Settings

Input Type: Key Signal

Key Event Action: HTTP Request

* HTTP URL: http://api.com/test

Input Settings

Input Type: Key Signal

Key Event Action: Play Audio

Audio File: Toy

Repeat: - 1 +

Key Signal Input Settings

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

outgoing calls.

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

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

Input Settings

Input Type:

Closed Event Action:

Destination: Line:

Open Event Action:

Input Settings

Input Type:

Closed Event Action:

Open Event Action:

Input Settings

Input Type:

Closed Event Action:

* HTTP URL:

Open Event Action:

Input Settings

Input Type:

Closed Event Action:

Audio File: Repeat:

Open Event Action:

Input Settings

Input Type: Switch Signal ▼

Closed Event Action: Stop Audio ▼

Open Event Action: Disabled ▼

Switch Signal Input Settings

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

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

- **Hangup:** Hangup the Outgoing Call.
- **HTTP URL:** Configure the API URL address triggered by linkage.
- **Play Audio:** Play the pre-configured audio file.
- **Audio File:** Configure the audio triggered by linkage.
- **Repeat:** Configure the times of audio repetitions triggered by linkage.
- **Stop Audio:** Stop playing the pre-configured audio file.

Trigger Event

Broadcast music trigger:

Broadcast alarm trigger:

Input Signal Trigger:

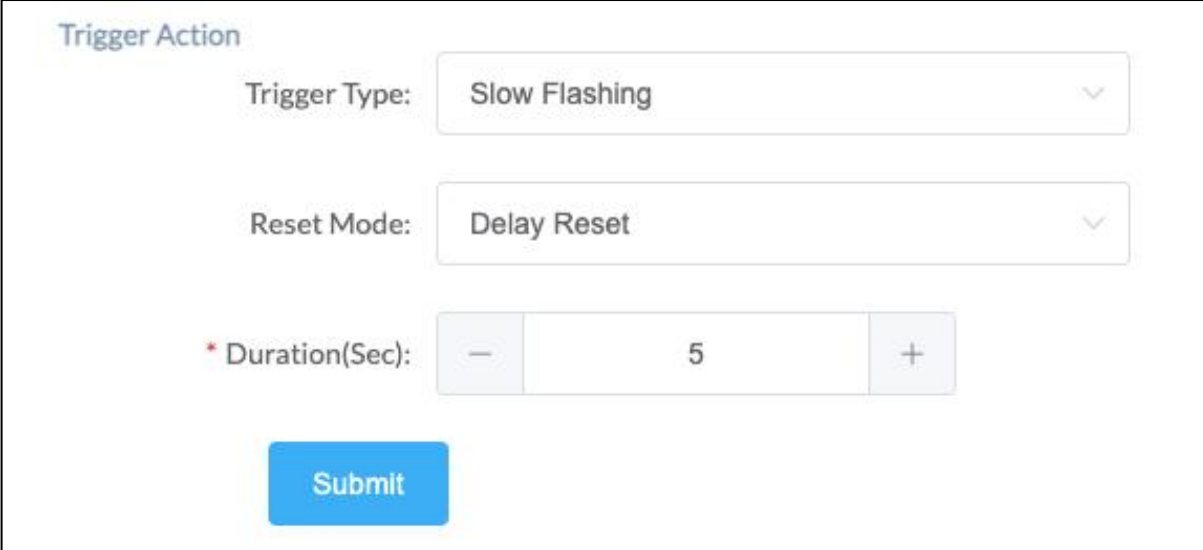
DTMF Signal Trigger:

Trigger by Call Status:

Event:

Output Trigger Event 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.
- **Input Signal Trigger:** Enable/Disable, enable this option if need to use the input signal to trigger.
- **DTMF Signal Trigger:** Enable/Disable, enable this option when need to use DTMF signal to trigger (only RF2833 supported).
- **DTMF:** This setting represents the number to dial to trigger DTMF.
- **Trigger by Call Status:** Enable/Disable, 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】** .



Trigger Action

Trigger Type: Slow Flashing

Reset Mode: Delay Reset

* Duration(Sec): 5

Submit

Output Trigger Action Settings

- **Trigger Type:** This setting represents the responses by the triggers, there are 'On', 'Fast Flashing', and 'Slow Flashing' options to choose from.
- **Reset Mode:** This setting represents the reset mode after the trigger is responded, there are 'Delay Reset' and 'Hang-up Reset' options to choose from.
- **Duration (Sec):** This setting is only available if the reply control mode is on Delay Reset, it represents the time duration when the configure interface status changed.

6.6 PTP Settings

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

PTP Settings

Enable:

* PTP Server:

PTP Setting

7. System Settings

7.1 Network

SH10 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

Access Type: HTTP

DHCP:

* IP Address: 192.168.1.101

* Subnet Mask: 255.255.255.0

* Gateway: 192.168.1.1

* Primary DNS: 114.114.114.114

* Alternative DNS: 8.8.8.8

Submit

Network Configuration

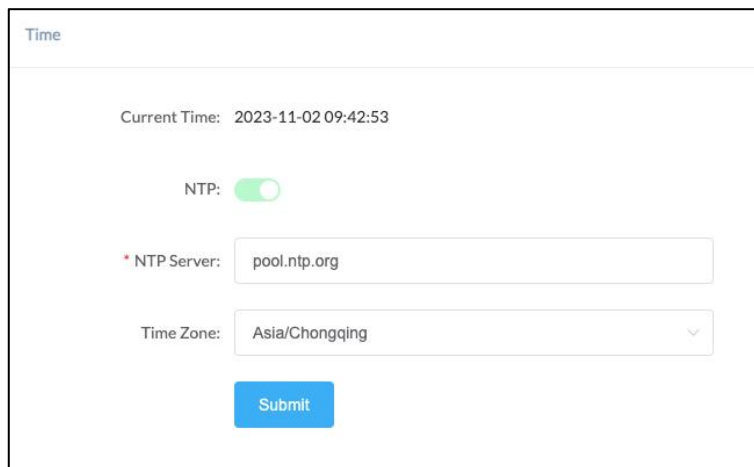
- **Access Type:** Specify the access method of the website, which currently supports HTTP and HTTPS.
- **IP Address:** Enter a vacant IP address within your LAN.
- **Subnet Mask:** Enter the subnet mask of your LAN.
- **Gateway:** Enter the default gateway of your LAN, this is essential for the device 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

SH10 obtains the time from the network time servers using NTP.

To change the NTP settings, please go to **System --> Time** page.



Time

Current Time: 2023-11-02 09:42:53

NTP:

* NTP Server:

Time Zone:

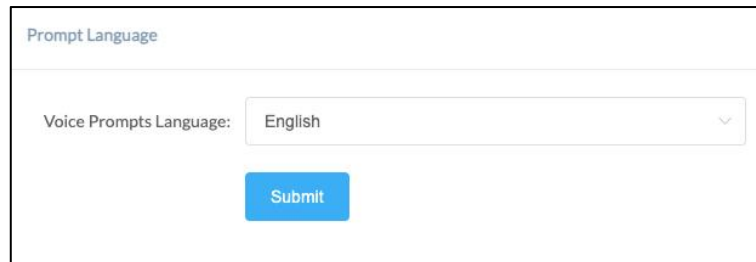
Time Settings

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

7.3 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

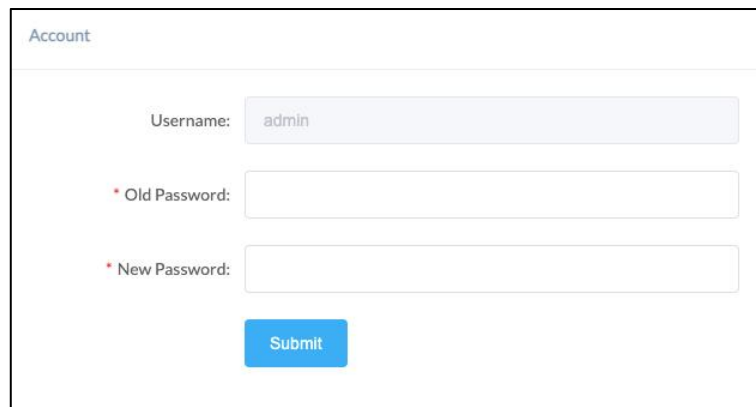
Voice Prompts Language: English

Submit

Prompt Language

7.4 Account

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



Account

Username: admin

* Old Password:

* New Password:

Submit

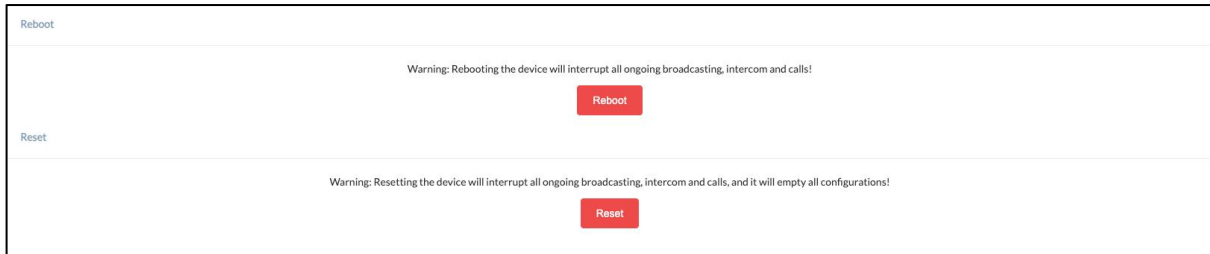
Web Password Settings

- **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

SH10 can be rebooted and reset from the web management interface.

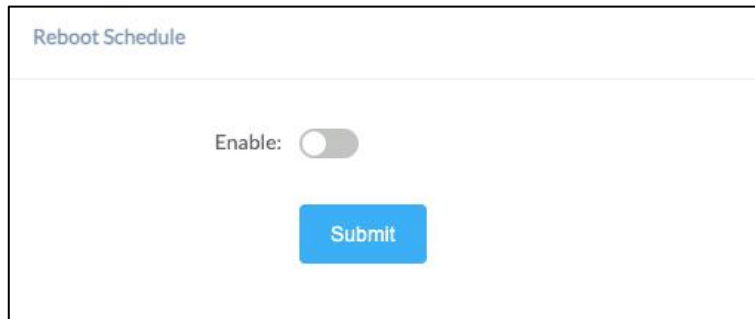
If you need to reboot or reset the device, please go to **System --> Reboot & Reset** page.



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!



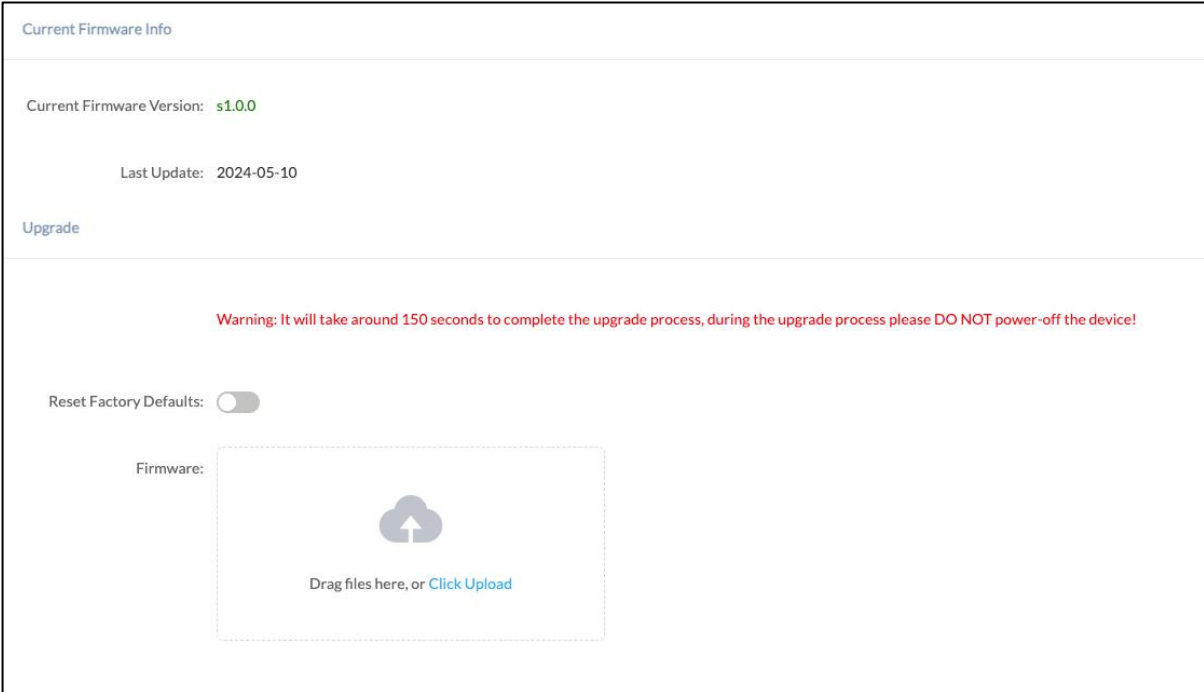
Reboot Schedule

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.



The screenshot shows the 'Upgrade' page with the following elements:

- Current Firmware Info** section:
 - Current Firmware Version: **s1.0.0**
 - Last Update: 2024-05-10
- Upgrade** section:
 - A red warning message: "Warning: It will take around 150 seconds to complete the upgrade process, during the upgrade process please DO NOT power-off the device!"
 - A toggle switch for "Reset Factory Defaults" which is currently turned off.
 - A "Firmware:" label next to a dashed box containing an upload icon and the text "Drag files here, or Click Upload".

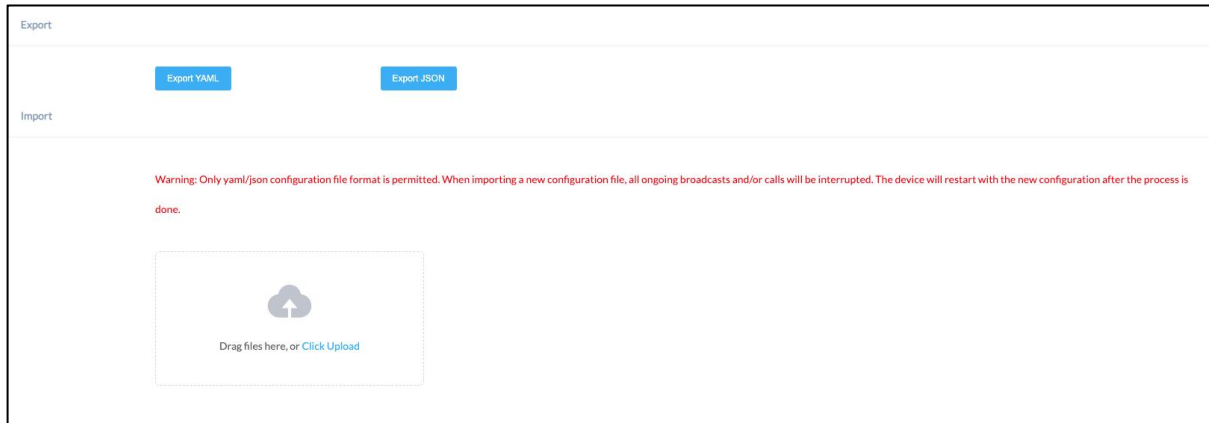
Upgrade Settings

- **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.

DHCP Provisioning Server

When the system start 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

Static Provisioning Server

Access Mode: TFTP

TFTP Server Address: 10.10.1.5

Configuration Format: JSON

Configuration Filename: \$mac.json

Update Mode: Update after reboot

Submit

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

* IP/Domain: eg: 8.8.8.8

Submit



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.

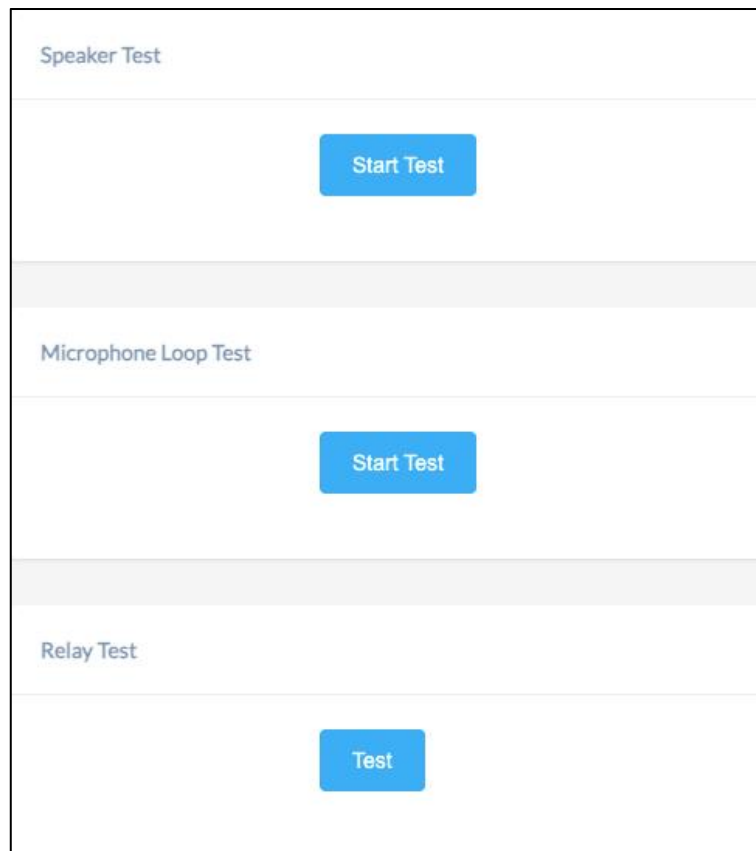


Ethernet Capture

8.6 Test

The detection feature provides an option for the user to check whether the speaker, microphone 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.

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 Operation Logs

Operation 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 --> Call 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-rtspdata[{}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:0;{}id:ec1f9890-e2ac-11ee-8d39-5f2d8908e54b;time:2024-03-15T09:18:01.881Z]
2024-03-15 17:18:00	SIP STATE	CALL ESTABLISHED	Incoming

Operation Logs



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