



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
1.0.5	1.0.5	Updated operating instructions for software version v1.0.5.	Jul, 2025
1.0.3	1.0.3	Updated operating instructions for software version v1.0.3.	Nov, 2024
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

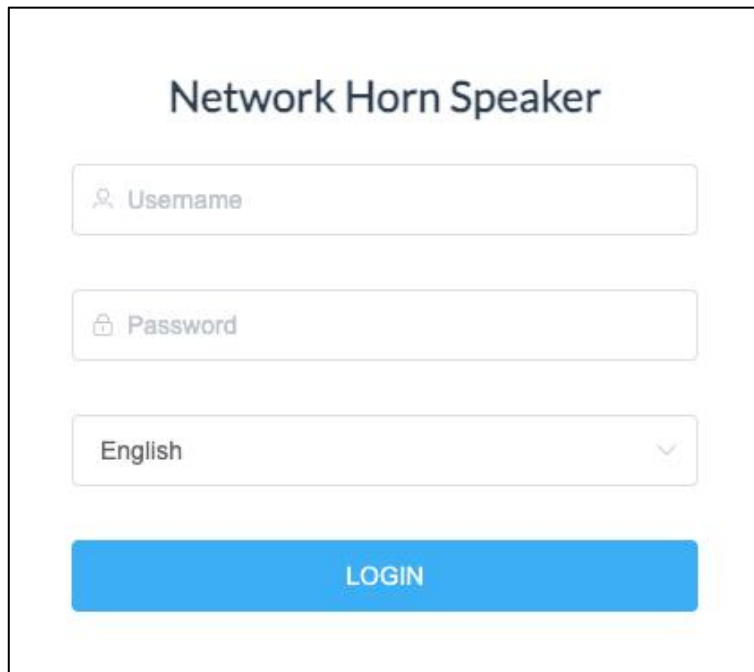
3.1 Accessing the Web GUI

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 image shows a web login interface for a device titled "Network Horn Speaker". It features three input fields: "Username" with a magnifying glass icon, "Password" with a lock icon, and a language dropdown menu currently set to "English". Below these fields is a prominent blue button labeled "LOGIN".

Network Horn Speaker

English

▼

LOGIN

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.

The screenshot shows the ZYCOO web interface. On the left is a sidebar with navigation options: Device Info, SIP Settings, Functions, Advanced, System, Maintenance, and Reports. The main content area is divided into three sections:

- SIP STATUS:** A table showing SIP account configurations.

Account	Address	Status	Register
Primary SIP Account	1008@192.168.11.109:5060	Registered	Idle
Secondary SIP Account-1		Unconfigured	
Secondary SIP Account-2		Unconfigured	
- DEVICE INFORMATION:** A table showing device details.

Device Model	SH10
Hardware Version	Ver1.0
Software Version	s1.0.3
Uptime	21 days 5:03
Speaker Volume	7 (0-9) 🔊
Device Description	SH10 🔗
- NETWORK INFORMATION:** A table showing network settings.

Mac Address	68:49:2E:2E:01:DE
Connection Mode	STATIC
IP Address	192.168.11.95
Subnet Mask	255.255.255.0
Gateway	192.168.11.1
Primary DNS	114.114.114.114
Alternative DNS	8.8.8.8

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	s1.0.3
Uptime	21 days 5:03
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:** Display the system version number of the device.
- **UpTime:** Displays the device's operating time.
- **Speaker Volume:** Displays the current volume of the device.
- **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.

Basic Configuration

Line Status: **Registered**

* SIP Server: 192.168.16.109

* SIP Port: 5060

* User ID: 1004

Password: *****

Auto Answer: Yes

Enable Integration with
ZYCOO IP Audio Center: ☒

Activate: ☒

SIP Account - Basic Configuration

- **Line Status:** Display the current registration status of the SIP account.
- **SIP Server:** Enter the IP address or domain name of the SIP server.

- **SIP Port:** The default SIP port is 5060. If your SIP server uses a different port, update this setting accordingly.
- **User ID:** Enter the SIP account number provided by your SIP server.
- **Password:** Enter the password for authorizing the SIP account.
- **Auto Answer:** Options include Yes, No, or Answer Delay. The default setting is 'Yes.'
- **Enable Integration with ZYCOO IP Audio Center:** Disabled by default. Enable this option when connecting to the ZYCOO IP Audio Center. This option is available only for the primary SIP account.
- **Activate:** Once enabled, the account will be activated and registered with the SIP server.

The screenshot shows a web interface titled "Advanced Configuration" for SIP account settings. It includes the following fields and controls:

- Auth User:** A text input field with a placeholder "eg: 100".
- Domain:** A text input field with a placeholder "eg: pbx.com".
- * Register Expiration(Sec):** A numeric input field with a value of 180, flanked by minus and plus buttons.
- * Transport:** A dropdown menu with "UDP" selected.
- NAT Mode:** A dropdown menu with "Disabled" selected.
- Keepalive:** A toggle switch that is currently turned on (green).
- * Keepalive Interval(Sec):** A numeric input field with a value of 30, flanked by minus and plus buttons.
- Submit:** A blue button at the bottom of the form.

SIP Account - Advanced Configuration

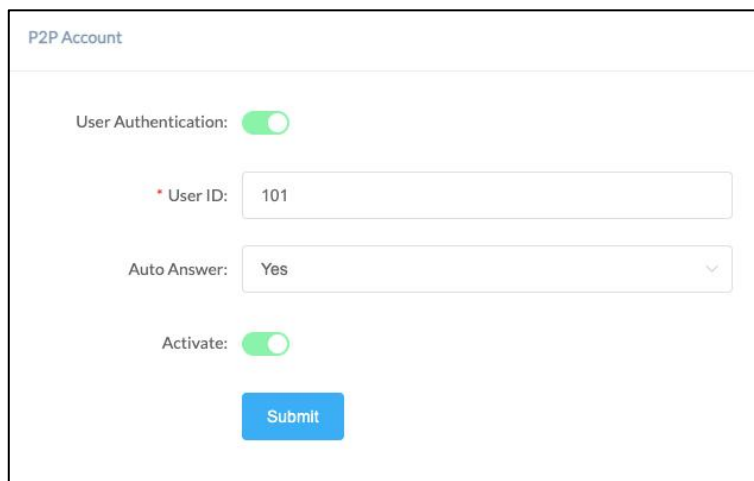
- **Auth User:** Enter the authorized username for the SIP account.

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

4.2 P2P Account Settings

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

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

The image shows a web form titled "P2P Account". It contains the following elements: a "User Authentication" toggle switch that is turned on (green); a "User ID" input field with a red asterisk icon and the value "101"; an "Auto Answer" dropdown menu with "Yes" selected; an "Activate" toggle switch that is turned on (green); and a blue "Submit" button at the bottom.

P2P Account

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

4.3 Advance SIP Settings

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

4.3.1 SIP Parameter Settings

SIP Parameter Settings

Local Port:

* 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

The screenshot shows the 'SIP Function Settings' page. It contains the following controls:

- Answer Local Beep:** A toggle switch that is currently turned off.
- Answer Remote Beep:** A toggle switch that is currently turned on.
- Beep Sound File:** A dropdown menu showing 'Start Beep' and a 'Play' button to the right.
- Beep Volume:** A volume slider set to 90, with minus and plus buttons on either side.
- Hangup Beep:** A toggle switch that is currently turned off.
- Second Call Handling:** A dropdown menu showing 'Hangup' and a help icon.
- Submit:** A blue button at the bottom center.

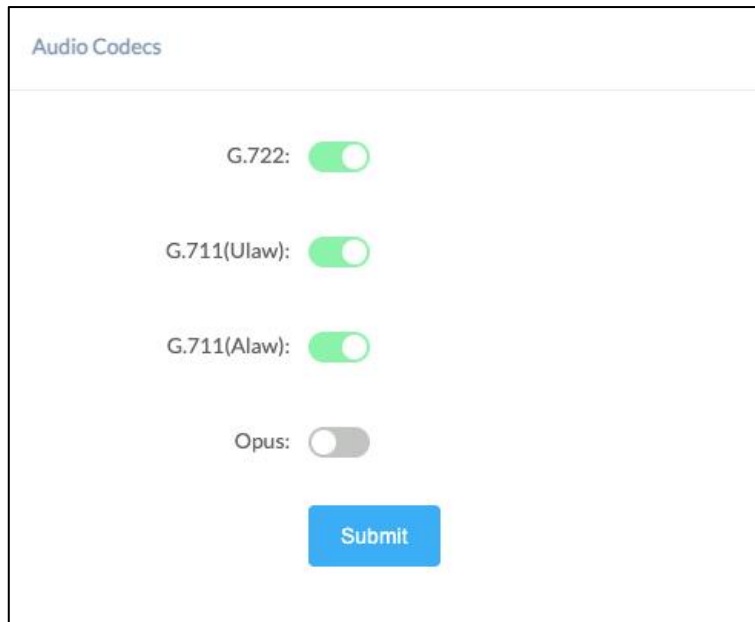
SIP Function Settings

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

- **Hangup Beep:** If this setting is enabled, the selected beep sound will be played on the local device side before the SIP session is completely hung up.
- **Second Call Hanging:** Set the volume of the beep. Options for handling the second call: Hangup: Directly hang up the second call. Hold: Hold the first call and automatically resume it after ending the second call. Merge: Join the second call into the first call, allowing all parties to speak simultaneously.

4.3.3 Audio Codecs

SH10 supports 4 audio codecs: G.722 (wideband codec), G.711(Ulaw), G.711(Alaw), and Opus.



The screenshot displays a web interface titled "Audio Codecs". It contains four rows, each with a codec name and a toggle switch. The first three rows (G.722, G.711(Ulaw), and G.711(Alaw)) have their toggle switches turned on (green). The fourth row (Opus) has its toggle switch turned off (grey). A blue "Submit" button is located at the bottom center of the form.

Codec	Status
G.722:	Enabled
G.711(Ulaw):	Enabled
G.711(Alaw):	Enabled
Opus:	Disabled

Audio Codecs

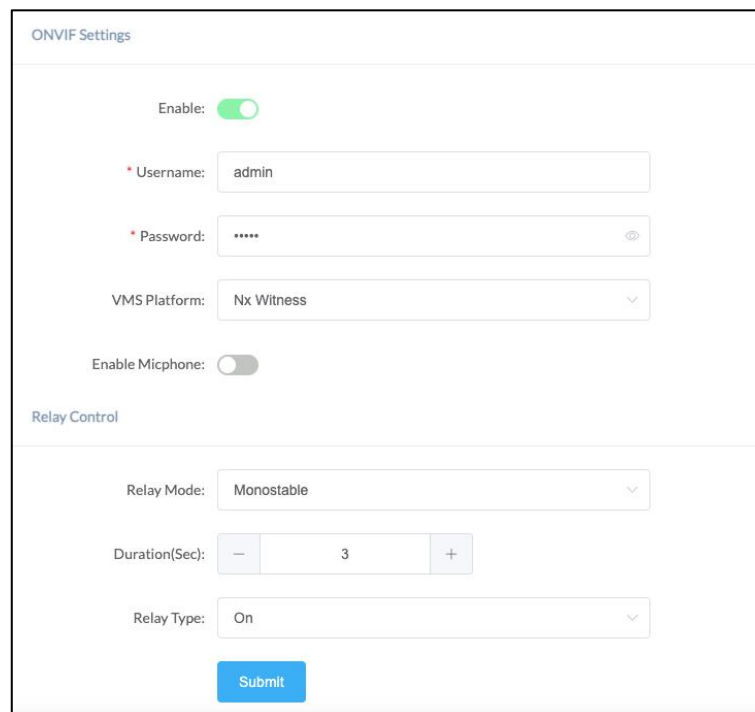
Please keep at least one codec enabled and supported by the SIP server, otherwise, SIP paging will not work.

5. Function Settings

5.1 ONVIF Settings

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

Please go to **Functions ---> ONVIF Settings** to configure the ONVIF settings.



The screenshot displays the 'ONVIF Settings' and 'Relay Control' configuration page. The 'ONVIF Settings' section includes a toggle for 'Enable' (currently on), a text field for 'Username' (set to 'admin'), a password field for 'Password' (masked with asterisks), a dropdown for 'VMS Platform' (set to 'Nx Witness'), and a toggle for 'Enable Micphone' (currently off). The 'Relay Control' section features a dropdown for 'Relay Mode' (set to 'Monostable'), a numeric input for 'Duration(Sec)' (set to 3), and a dropdown for 'Relay Type' (set to 'On'). A blue 'Submit' button is located at the bottom of the form.

ONVIF & Relay Control Settings

- **Enable:** Enable/Disable ONVIF integration for compatibility with ONVIF-supported VMS platforms.
- **Username:** Enter an account username with matching credentials for adding devices to the VMS platform.
- **Password:** Enter a matching password for the account to add devices to the VMS platform.

- **VMS Platform:** Allows user to select a specific VMS platform from the drop-down list to enable compatibility with different VMS systems.
- **Enable Microphone:** Enable/Disable the microphone function.
- **Relay Mode:** Set relay control to monostable or bistable. In monostable mode, you can specify the activation duration.
- **Duration(Sec):** Set the activation duration in monostable mode.
- **Relay Type:** Choose a relay response to triggers: 'On', 'Fast Flashing', or 'Slow Flashing'.

5.2 Multicast

The multicast settings are used to configure the parameter settings of the multicast function 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: ☒

Network Caching(ms):

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
1	<input type="text" value="239.168.12.102"/>	<input type="text" value="2000"/>	<input type="text" value="Background-Music"/>	<input type="text" value="Disabled"/>
2	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	<input type="text" value="Disabled"/>
3	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	<input type="text" value="Disabled"/>
4	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	<input type="text" value="Disabled"/>
5	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	<input type="text" value="Disabled"/>
6	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	<input type="text" value="Disabled"/>

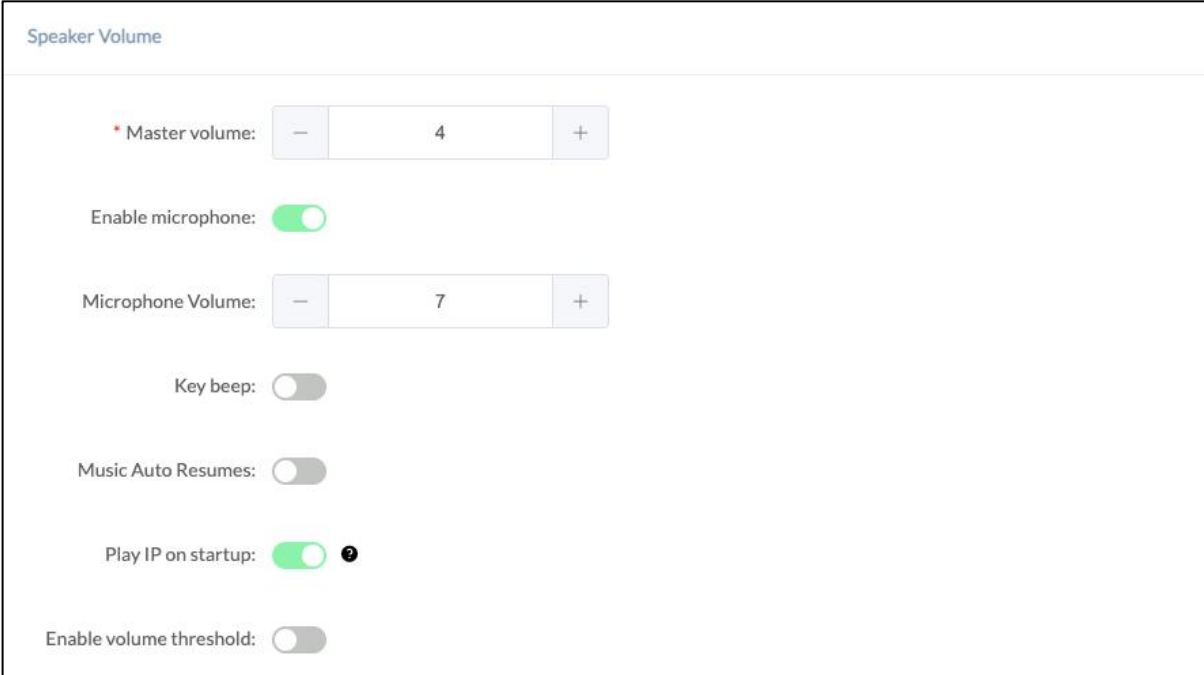
Multicast

- **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.

6. Advanced Settings

6.1 Volume Settings

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



Speaker Volume

* Master volume:

Enable microphone: ☒

Microphone Volume:

Key beep: ☐

Music Auto Resumes: ☐

Play IP on startup: ☒ ?

Enable volume threshold: ☐

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

App Volume

* SIP Volume:	<input type="range"/>	<input type="button" value="-"/> 80 <input type="button" value="+"/>
ONVIF Volume:	<input type="range"/>	<input type="button" value="-"/> 80 <input type="button" value="+"/>
BROADCAST Volume:	<input type="range"/>	<input type="button" value="-"/> 48 <input type="button" value="+"/>
MULTICAST Volume 1:	<input type="range"/>	<input type="button" value="-"/> 80 <input type="button" value="+"/>
MULTICAST Volume 2:	<input type="range"/>	<input type="button" value="-"/> 50 <input type="button" value="+"/>
MULTICAST Volume 3:	<input type="range"/>	<input type="button" value="-"/> 80 <input type="button" value="+"/>
MULTICAST Volume 4:	<input type="range"/>	<input type="button" value="-"/> 80 <input type="button" value="+"/>
MULTICAST Volume 5:	<input type="range"/>	<input type="button" value="-"/> 80 <input type="button" value="+"/>
MULTICAST Volume 6:	<input type="range"/>	<input type="button" value="-"/> 80 <input type="button" value="+"/>
MULTICAST Volume 7:	<input type="range"/>	<input type="button" value="-"/> 80 <input type="button" value="+"/>
MULTICAST Volume 8:	<input type="range"/>	<input type="button" value="-"/> 80 <input type="button" value="+"/>
MULTICAST Volume 9:	<input type="range"/>	<input type="button" value="-"/> 80 <input type="button" value="+"/>

App Volume Settings

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

- **Multicast Volume 2:** Set the specific volume for multicast 2 application.
- **Multicast Volume 3:** Set the specific volume for multicast 3 application.
- **Multicast Volume 4:** Set the specific volume for multicast 4 application.
- **Multicast Volume 5:** Set the specific volume for multicast 5 application.
- **Multicast Volume 6:** Set the specific volume for multicast 6 application.
- **Multicast Volume 7:** Set the specific volume for multicast 7 application.
- **Multicast Volume 8:** Set the specific volume for multicast 8 application.
- **Multicast Volume 9:** Set the specific volume for multicast 9 application.

This page is used to configure remote volume control using the VC-Z01 volume controller.

By matching multicast channel settings between the VC-Z01 and the endpoint speaker device, remote volume adjustments can be performed over the network.

Please go to **Advanced** → **Volume** → **Remote Control** page to configure.

Channel Settings

The volume control function must be used with the volume controller device VC-Z01.

10 default channels and 5 customizable channels are available. To use a custom channel, please manually select and configure the parameters. Supported multicast port range: 20-65535.

Enable: ☒

Volume Control Channel: Channel 10
Channel Status: Not Connected

Custom Channel

Channel	Multicast Address	Multicast Port
Custom Channel 11	<input type="text"/>	— 20 +
Custom Channel 12	<input type="text"/>	— 20 +
Custom Channel 13	<input type="text"/>	— 20 +
Custom Channel 14	<input type="text"/>	— 20 +
Custom Channel 15	<input type="text"/>	— 20 +

Submit

Remote Control Settings

- **Enable:** To enable remote volume control, VC-Z01 must be configured together with the endpoint devices. This requires settings to be completed on both the VC-Z01 web GUI and the endpoint web GUI. Once the channels match between VC-Z01 and the endpoints within the same local network segment, the endpoint device will display a Connected status.
- **Volume Control Channel:** Provides 10 default channels (Channel 1 – 10), and 5 customizable channels (Channel 11 – 15).
- **Channel Status:** Displays the connection status between the VC-Z01 and the endpoint device. Connected: The devices are successfully paired and remote control is available. Not Connected: The devices are not matched.
- **Custom Channel:** For Channel 11 – 15, you can define a custom multicast address and port. Multicast Address Range: 224.0.0.0 – 239.255.255.255 Supported Port Range: 20 – 65535.

For example, if both the VC-Z01 and the endpoint device are set to Channel 1, and are within the same network segment, clicking Submit on both GUIs will initiate connection. Once connected, the VC-Z01 can remotely adjust the volume of any endpoint device configured on the same channel.

6.2 Audio Priority Settings

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

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

Audio Priority

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** ---> **Audio Files** to manage the audio files.

Audio Files Upload

Audio files only accept wav format!

Current disk space remaining: 5.1M

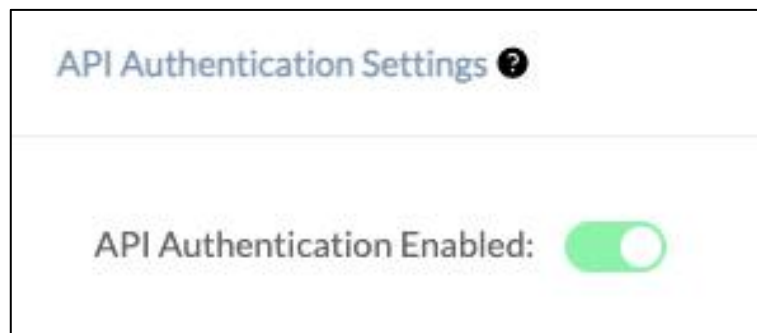
Custom audio file 1	Currently set to default	Select audio file	Upload	Play	Delete
Custom audio file 2	Currently set to default	Select audio file	Upload	Play	Delete
Custom audio file 3	Currently set to default	Select audio file	Upload	Play	Delete
Custom audio file 4	Currently set to default	Select audio 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 --> API Settings** page to enable API settings.



API Authentication Settings

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

The screenshot displays a configuration window with two main sections. The first section, titled 'Call Event URL Callback' with a help icon, contains five toggle switches, all of which are currently turned off: 'Incoming Enable', 'Outgoing Enable', 'Answered Enable', 'Hangup Enable', and 'Register Failed Enable'. The second section, titled 'Relay Event URL Callback' with a help icon, contains two toggle switches, also both turned off: 'On Enable' and '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>

[Submit](#)

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: Switch Signal

Closed Event Action: Outgoing Call

Destination: 5000

Line: Auto

Open Event Action: Stop Audio

Input Settings

Input Type: Switch Signal

Closed Event Action: Hangup

Open Event Action: Stop Audio

Input Settings

Input Type: Switch Signal ▼

Closed Event Action: HTTP Request ▼

* HTTP URL:

Open Event Action: Stop Audio ▼

Input Settings

Input Type: Switch Signal ▼

Closed Event Action: Play Audio ▼

Audio File: Sweet ▼ Repeat: — 0 +

Open Event Action: Stop Audio ▼

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: Disabled

Broadcast alarm trigger: Fast Flashing

Input Signal Trigger: Switch Signal - Closed

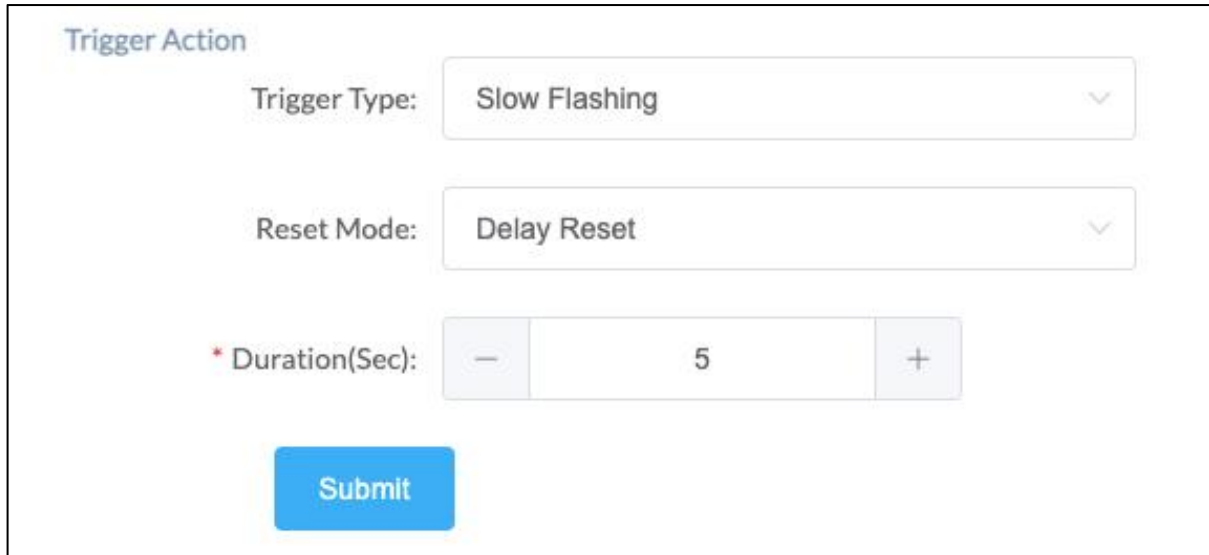
DTMF Signal Trigger: ☐

Trigger by Call Status: ☒

Event: Incoming/Outgoing

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 ---> 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

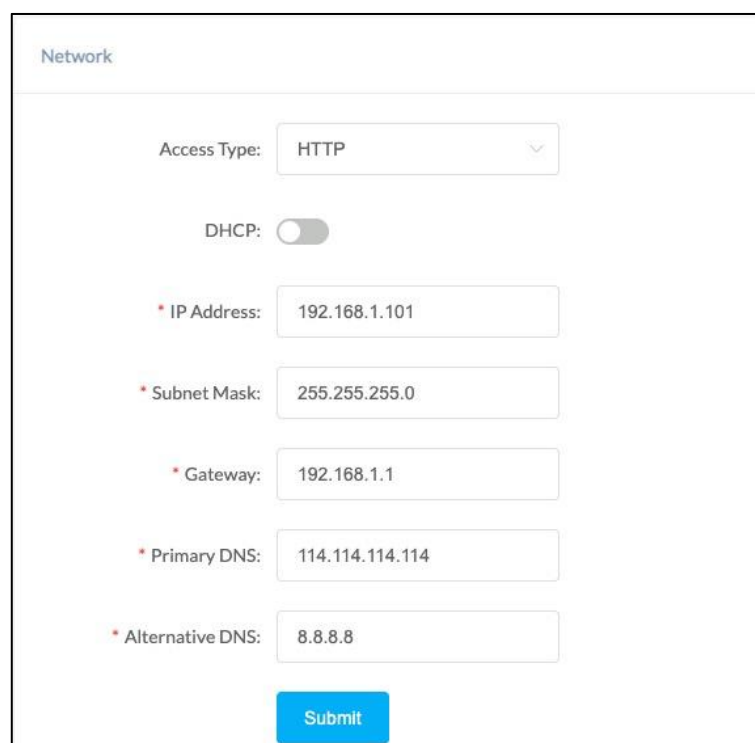
7. System Settings

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

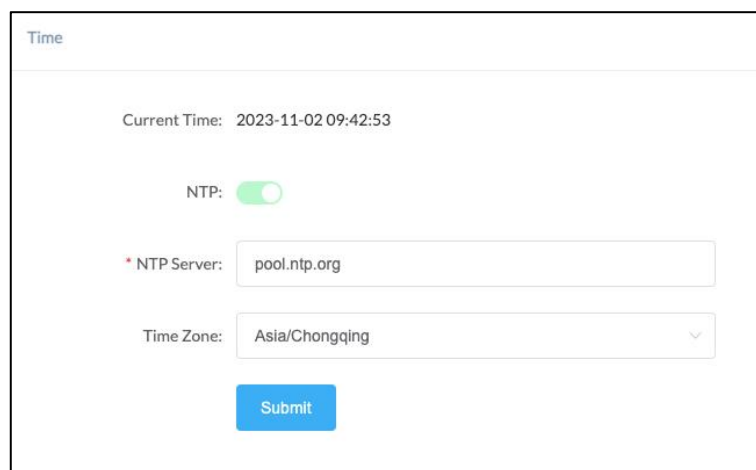
- **Access Type:** Specify the access method of the website, which currently supports HTTP and HTTPS.
- **IP Address:** Enter a vacant IP address within your LAN.
- **Subnet Mask:** Enter the subnet mask of your LAN.
- **Gateway:** Enter the default gateway of your LAN, this is essential for the device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



The screenshot shows the 'Time' configuration page. At the top, it displays 'Current Time: 2023-11-02 09:42:53'. Below this is a toggle switch for 'NTP' which is currently turned on (green). Underneath the toggle is a text input field for 'NTP Server' containing 'pool.ntp.org'. Below that is a dropdown menu for 'Time Zone' with 'Asia/Chongqing' selected. At the bottom of the form is a blue 'Submit' button.

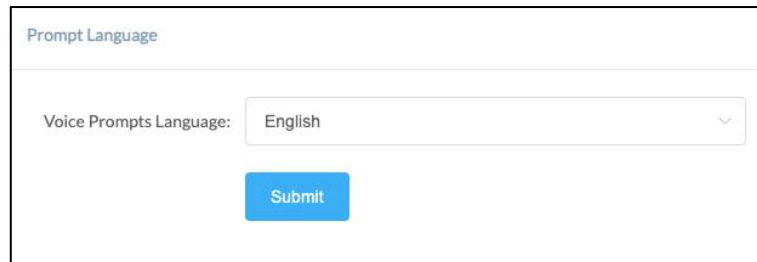
Time Settings

- **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.

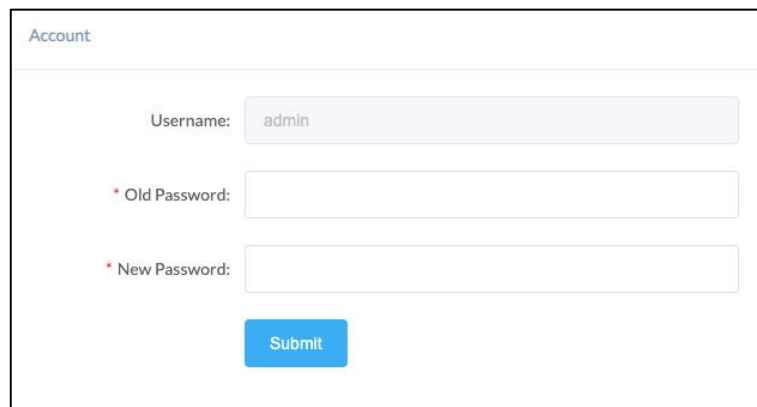
Please go to **System --> Prompt Language** page to set a voice prompt language.

The screenshot shows a web interface titled "Prompt Language". Below the title, there is a label "Voice Prompts Language:" followed by a dropdown menu currently set to "English". Below the dropdown is a blue "Submit" button.

Prompt Language

7.4 Account

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

The screenshot shows a web interface titled "Account". It contains three input fields: "Username:" with the value "admin", "* Old Password:", and "* New Password:". Below these fields is a blue "Submit" button.

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

Warning: Rebooting the device will interrupt all ongoing broadcasting, intercom and calls!

Reboot

Reset

Warning: Resetting the device will interrupt all ongoing broadcasting, intercom and calls, and it will empty all configurations!

Reset

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

Enable: ☐

Submit

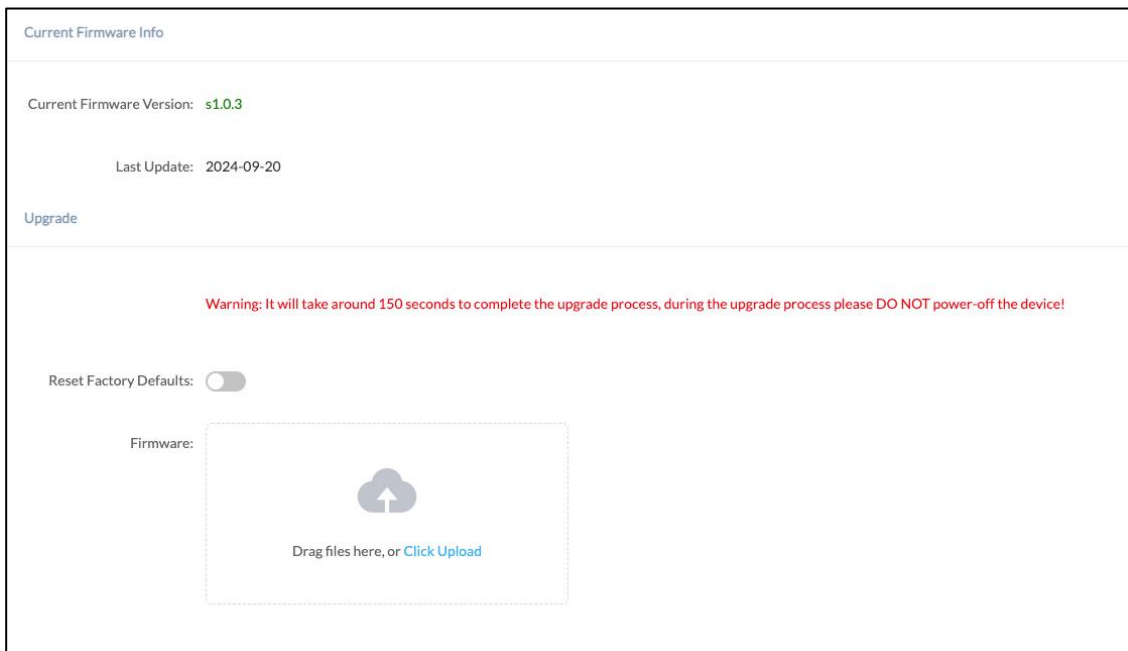
Reboot Schedule

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 in the SH10 Network Horn Speaker web interface. It features a 'Current Firmware Info' section with 'Current Firmware Version: s1.0.3' and 'Last Update: 2024-09-20'. Below this is a 'Warning: It will take around 150 seconds to complete the upgrade process, during the upgrade process please DO NOT power-off the device!'. There is a 'Reset Factory Defaults' toggle switch. The 'Firmware' section includes a dashed box with a cloud upload icon and the text 'Drag files here, or Click Upload'.

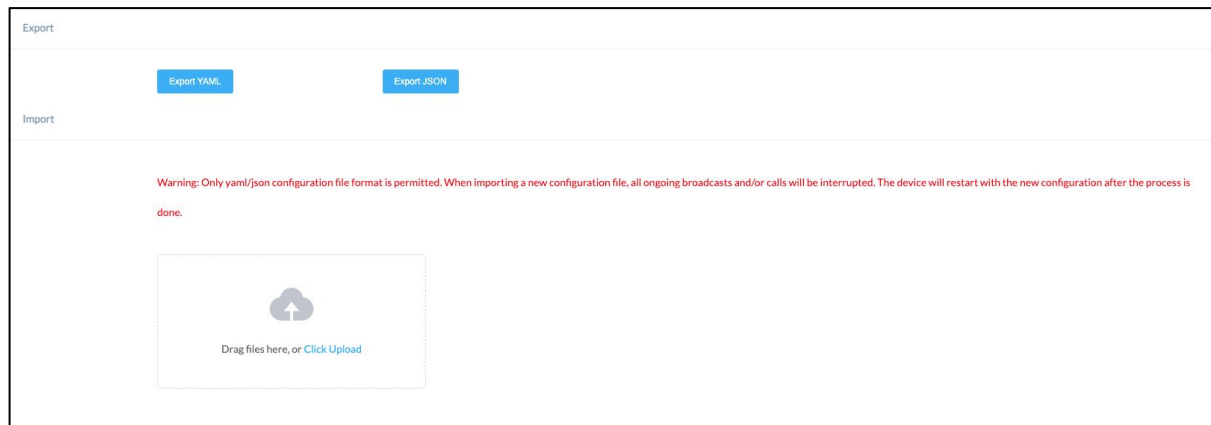
Upgrade Settings

- **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, 68:69:2E:29:00:12

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

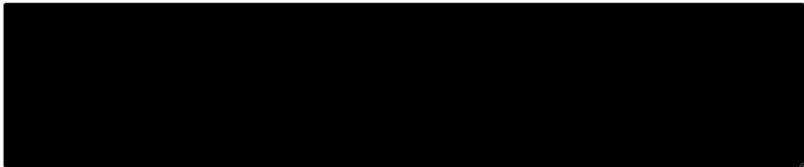
8.4 Diagnostic

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

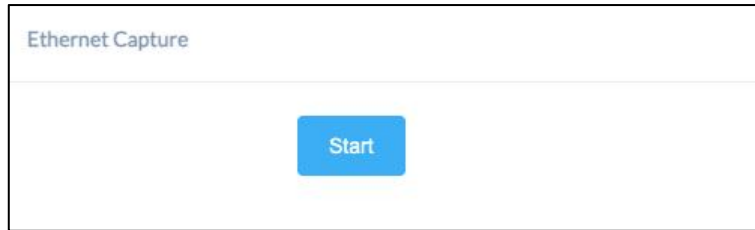


Ping

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.

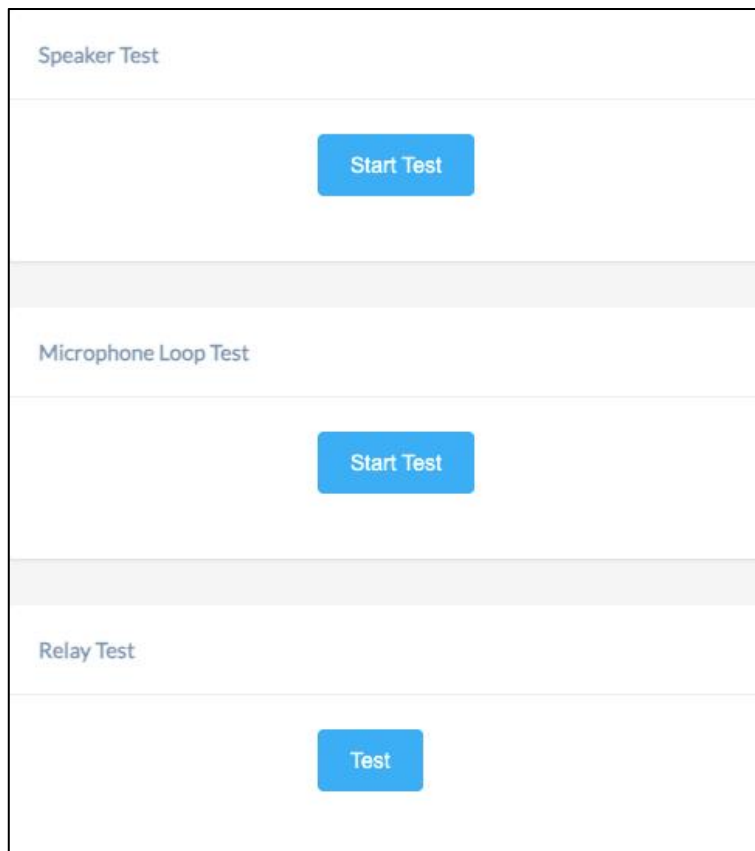
Please go to **Maintenance --> Ethernet Capture** page to operate.



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 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			
<div>Download</div>			
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

