



IPS-H1 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 IPS-H1 Network Horn Speaker. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the IPS-H1 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.	Sept., 2026

2. Overview

2.1 Product Overview

The ZYCOO IPS-H1 Network Horn Speaker is an IP-based audio communication endpoint designed for live paging, public announcements, background music, and emergency notifications. Built on SIP and multicast technologies, it delivers powerful and intelligible audio performance for both indoor and outdoor environments.

The IPS-H1 can operate as a standalone network horn speaker with up to 20W audio output, or be paired with a ZYCOO APS-H1 Extension Horn Speaker for expanded audio coverage. This deployment model helps reduce network cabling, PoE switch port usage, and overall installation costs.

Featuring a high-efficiency horn design, the IPS-H1 provides enhanced sound projection and speech intelligibility over longer distances, making it particularly suitable for open and noisy environments.

The IPS-H1 is ideal for campuses, factories, warehouses, transportation facilities, parking areas, industrial sites, and other public spaces, providing clear and reliable audio for daily announcements and emergency communications.

2.2 Product Specifications

IPS-H1 Network Horn Speaker Specifications		
Speaker Components	1.5"midrange driver unit	
Sensitivity	105dB/1m/1W	
Max Sound Pressure Level	108dB	
Frequency Range	400Hz – 8KHz	

3. Login the Device

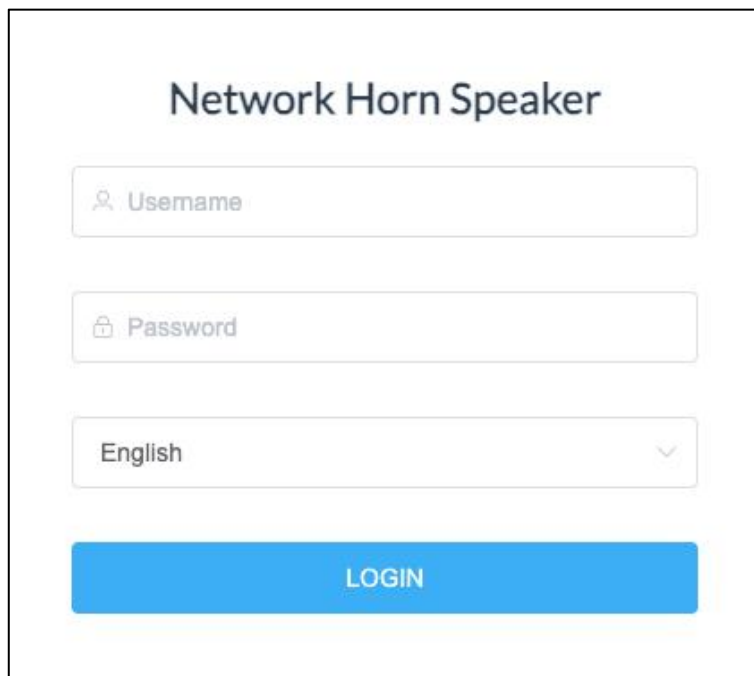
3.1 Accessing the Web GUI

IPS-H1 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 the login interface for the Network Horn Speaker. It features a white background with a black border. At the top, the text "Network Horn Speaker" is centered. Below it, there 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, there is a prominent blue button labeled "LOGIN".

Network Horn Speaker

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

Upon successful login, you will enter the device information page, where you can view the device's basic details and operating status. Icons at the top of the page display the connection status between the main and extension speakers, the current power source, the status of the noise compensation function, and the current volume level, enabling users to quickly check the status and troubleshoot issues.

SIP STATUS		SERVICE STATUS INFORMATION	VOICE CONTROL CHANNEL STATUS
Primary SIP Account	5009@192.168.13.109:5060	Registered	Idle
Secondary SIP Account-1		Unconfigured	
Secondary SIP Account-2		Unconfigured	

DEVICE INFORMATION	
Device Model	IPS-H1
Hardware Version	V1.0.0
Software Version	x0.0.0.014
Uptime	0:30
Speaker Volume	19 (-46dB)

SIP STATUS	SERVICE STATUS INFORMATION	VOICE CONTROL CHANNEL STATUS
Primary SIP Account	5009@192.168.13.109:5060	Registered Idle
Secondary SIP Account-1		Unconfigured
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	IPS-H1
Hardware Version	V1.0.0
Software Version	s0.0.0.014
Uptime	0:30
Speaker Volume	19 (-4dB) ↗
Max Sound Pressure Level	30 (+7dB)
Mic Volume	8 (0-15) ↗
Device Description	IPS-H1 ↗

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:

* SIP Port:

* User ID:

Password:

Auto Answer:

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

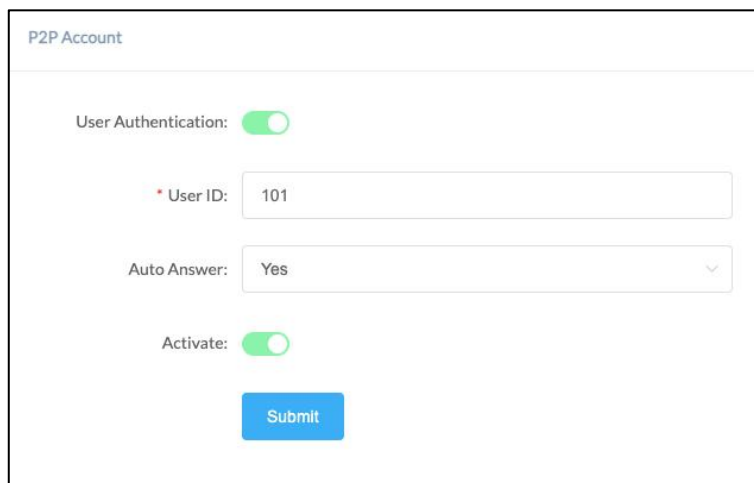
- **Auth User:** Enter the authorized username for the SIP account.

- **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 screenshot shows the 'P2P Account' configuration page. It includes a title 'P2P Account' at the top left. Below it, there are four settings: 'User Authentication' with a green toggle switch, 'User ID' with a text input field containing '101', 'Auto Answer' with a dropdown menu showing 'Yes', and 'Activate' with a green toggle switch. At the bottom center is a blue 'Submit' button.

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:

—

5060

+

* RTP Start Port:

—

10000

+

* RTP End Port:

—

20000

+

* RTP Timeout(Sec):

—

60

+

Jitter Buffer:

adaptive

▼

Acoustic Echo Cancellation:

☒

Adaptive Noise Reduction:

☒

Automatic Gain Control:

☒

* AGC Max Gain(5~32dB):

—

30

+

* AGC gate threshold(-87~-42dB):

—

-85

+

Comfort Noise Generator:

☐

De-reverb:

☐

Time Synchronization Over SIP:

☐

Submit

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.
- **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.
- **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 Gain 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.
- **AGC Max Gain:** This setting represents the maximum level compensation amplitude that the AGC circuit allows .
- **AGC gate threshold:** This setting defines the signal strength threshold at which AGC begins to operate,any signal with a strength exceeding this set value will be processed.
- **Comfort Noise Generator:** After enabling this feature, comfortable white noise can be added during calls.
- **De-reverb:** After enabling this feature,voice clarity in calls can be enhanced.
- **Time Synchronization Over SIP:** After enabling this feature,the device automatically retrieves and synchronizes the current time from the SIP server after registration, ensuring accurate call records and log timestamps.

4.3.2 SIP Function Settings

SIP Function Settings

Answer Local Beep: ☐

Answer Remote Beep: ☒

Beep Sound File: Start Beep

Beep Volume:

Hangup Beep: ☐

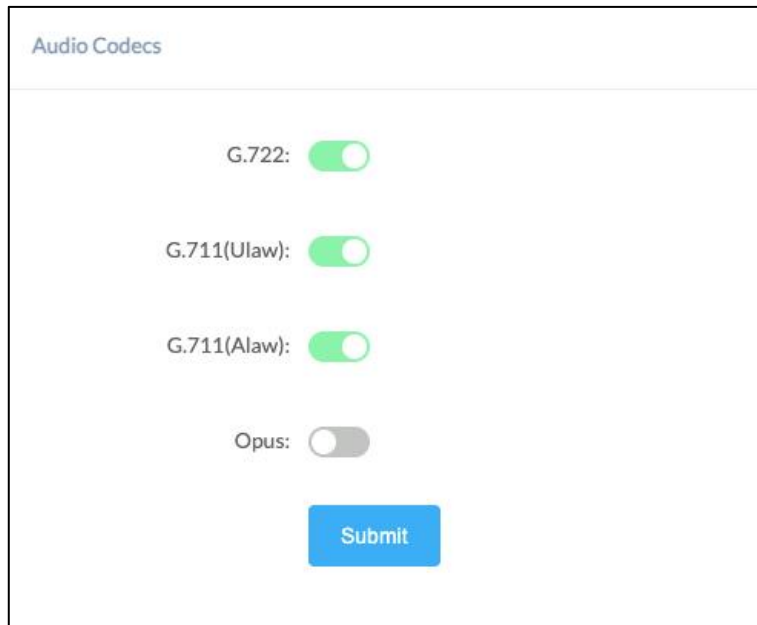
Second Call Handling: Hangup

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 Handling:** 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

IPS-H1 supports 4 audio codecs: G.722 (wideband codec), G.711(Ulaw), G.711(Alaw), and Opus.

A screenshot of a web interface titled "Audio Codecs". It contains four toggle switches for different audio codecs. The first three are turned on (green), and the last one is turned off (grey). A blue "Submit" button is at the bottom.

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

Submit

Audio Codecs

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

5. Function Settings

5.1 ONVIF Settings

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

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

ONVIF & Relay Control Settings

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

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

5.2 Multicast

The multicast settings are used to configure the parameter settings of the multicast function on the Network Speaker. 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	239.168.12.102	2000	Background-Music	Disabled
2		2000		Disabled
3		2000		Disabled
4		2000		Disabled
5		2000		Disabled
6		2000		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 Advanced Audio

To set the volume, please go to **Advanced --> Advanced Audio** page to configure. You can set the master 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.

The screenshot displays the 'Advanced Audio Settings' page. It contains two main sections. The top section has the following settings: 'Ambient Noise Compensation' set to 'Dynamic' with a help icon; 'Ambient Noise Compensation Value(dB)' set to 5 with minus and plus adjustment buttons; 'Minimum Speaker Volume' set to 12 (-6dB) with a dropdown arrow; 'Maximum Speaker Volume' set to 30 (+12dB) with a dropdown arrow; 'Microphone Volume' set to 8 with minus and plus adjustment buttons; 'Key beep' enabled with a green toggle switch; 'Music Auto Resumes' disabled with a grey toggle switch; and 'Play IP on startup' enabled with a green toggle switch and a help icon. The bottom section shows 'Ambient Noise Compensation' set to 'Disabled' with a help icon, and 'Speaker Output Volume' set to 19 (+1dB) with a dropdown arrow.

Advanced Audio Settings

- Ambient Noise Compensation:** Enable or disable automatic volume adjustment based on ambient noise. Automatically adjusts playback volume based on ambient noise levels to improve the intelligibility of broadcast content. Two modes are supported: Static Compensation (calculates the compensation gain once before playback) and Dynamic Compensation (continuously adjusts the compensation gain during playback). *Note: This feature typically provides good performance for voice broadcast applications. For music*

playback, the compensation performance may vary depending on the content type. For music with a wide dynamic range or frequent rhythm changes (e.g., rock, electronic music), the compensation gain may fluctuate more noticeably, which can affect the perceived volume adjustment behavior. It is recommended to evaluate the feature in the actual application scenario and select the appropriate compensation mode accordingly.

- **Ambient Noise Compensation Value(dB):** Set the compensation level applied when ambient noise is detected. A higher value results in greater volume adjustment.
- **Minimum Speaker Volume:** Set the minimum allowed speaker output level. The range is 0–30, representing a linear gain scale, where 0 = mute, 22 $\approx$ 0 dB (unity gain), and 30 $\approx$ +8 dB.
- **Maximum Speaker Volume:** Set the maximum allowed speaker output level. The range is 0–30, representing a linear gain scale, where 0 = mute, 22 $\approx$ 0 dB (unity gain), and 30 $\approx$ +8 dB.
- **Speaker output volume:** When ambient noise compensation is disabled, this setting determines the default speaker output volume. The range is 0–30, using a linear gain scale, where 0 = silent, 22 $\approx$ 0 dB (unit gain), and 30 $\approx$ +8 dB.
- **Microphone Volume:** Set the microphone volume level. The adjustable range is 0–15.
- **Key Beep:** Enable/Disable the beep sound from the key button.
- **Music Auto Resumes:** When enabled, the device will automatically resume previous music playback tasks after a restart or network reconnection.
- **Play IP on Startup:** When enabled, the device will announce its IP address once upon startup.

The screenshot displays the 'App Volume' configuration page. It contains the following settings:

Application	Volume Value
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

A 'Submit' button is located at the bottom center of the form.

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** → **Audio** --> **Advanced Audio** → **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		— 20 +
Custom Channel 12		— 20 +
Custom Channel 13		— 20 +
Custom Channel 14		— 20 +
Custom Channel 15		— 20 +

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	▲

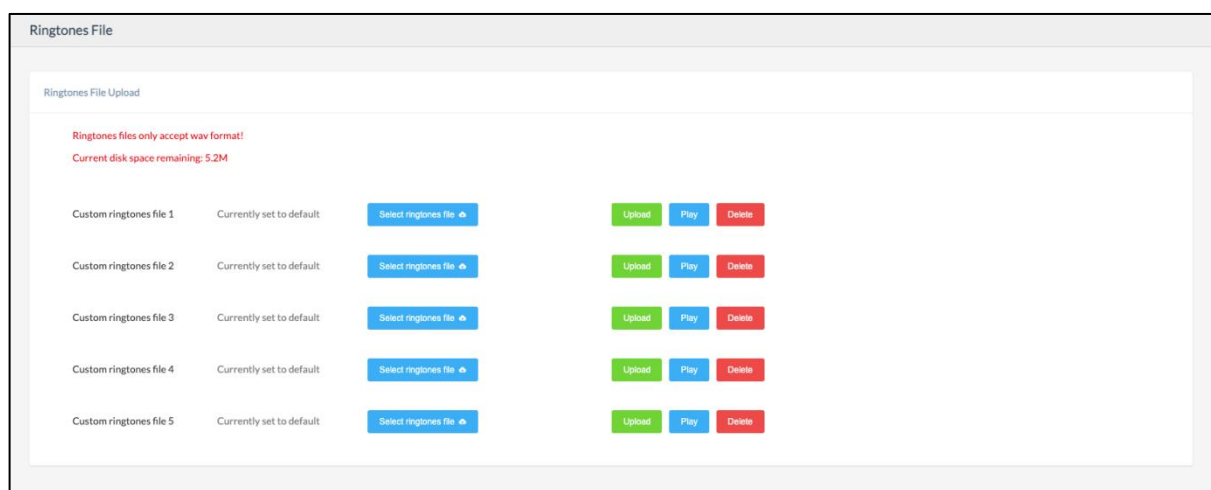
Submit

Audio Priority Settings

6.3 Ringtones File

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

Please go to **Advanced** ---> **Ringtones File** to manage the audio files.

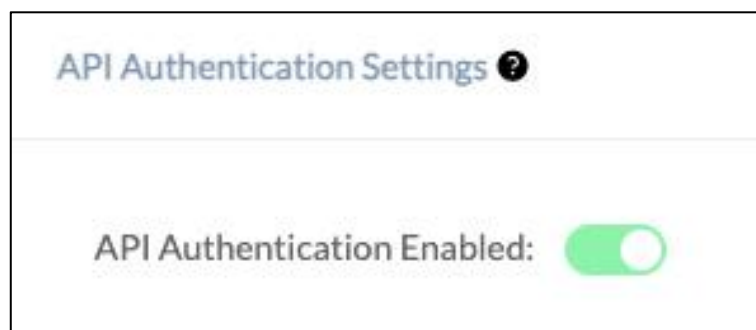


Ringtones File

6.4 API Settings

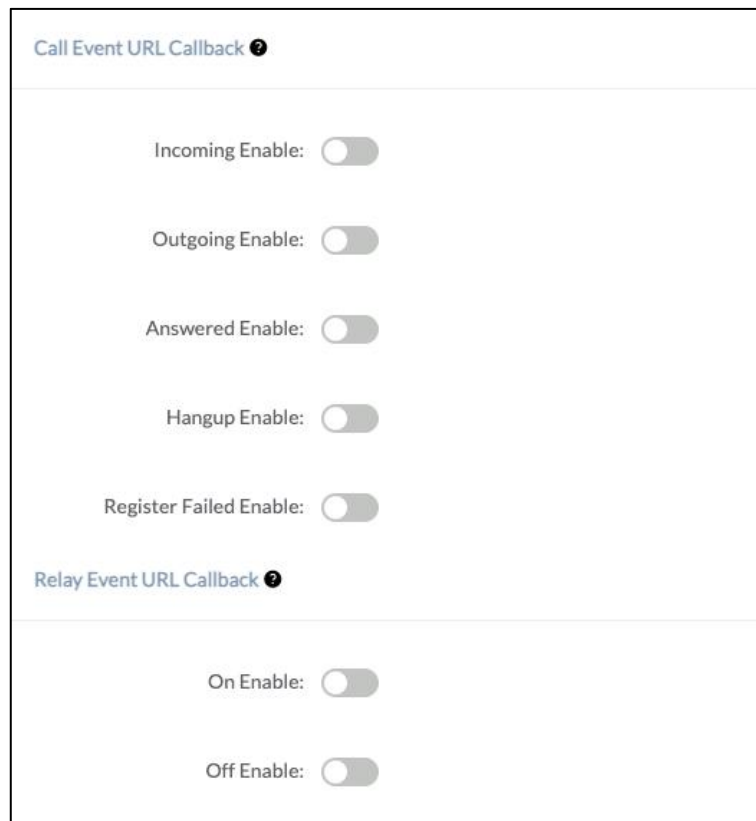
This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the changing status of the call and/or relay.

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 shows a settings page with two main sections, each with a title and a help icon (question mark in a circle). The first section, 'Call Event URL Callback', contains five toggle switches: 'Incoming Enable', 'Outgoing Enable', 'Answered Enable', 'Hangup Enable', and 'Register Failed Enable'. The second section, 'Relay Event URL Callback', contains two toggle switches: 'On Enable' and 'Off Enable'. All toggle switches are currently in the 'off' position.

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:

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

$\{\text{mac}\}$: 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 allows users to configure input and output behaviors for device integration and security linkage, including trigger conditions, input actions, relay control.

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

Key settings

Input Mode

Input Type: Key Signal

Key1 Action: Outgoing Call

Destination: 5000 Line: Auto

Press Again to End Call: ☒

Input Settings - Button Signal

Key settings

Input Mode

Input Type: Switch Signal

Closed Event Action: Disabled

Open Event Action: Disabled

Input Settings - Switch Signal

- **Input Mode:** Input Mode: Select the input type: Button Signal and Switch Signal.
- **Action:** Select the action to be triggered when the input is activated. Available options include: Outgoing Call, HTTP Request, Play Audio.
- **Destination:** Enter the target number or device to call when triggered.
- **Line:** Select the SIP line used for the call.

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:** Enable to allow pressing the button again to terminate the call.
- **HTTP URL:** Enter the API URL to be triggered.
- **Audio File:** Select the audio to be played (ringtone, audio file, or playlist).

- **Repeat:** Set how many times the audio will be played.

Relay Settings

General Setting

Relay Mode: Fast Toggle (e.g. Strobe Fast Flashing) ▼

Triggered by DTMF Signal: ☐

Triggered by Call Status: ☒

Call Status: Incoming/Outgoing ▼

Reset Mode: Delay Reset ▼

Duration(Sec): 5

Dispatch Console Setting

Trigger via Dispatch Console Music: Disabled ▼

Trigger via Dispatch Console Alarm: Disabled ▼

Relay Settings

- **Relay Mode:** Select the relay behavior: Disabled, Single Toggle (e.g. Door Strike, Audible Visual Alarm), Fast Toggle (e.g. Strobe Fast Flashing), Slow Toggle (e.g. Strobe Slow Flashing).

- **Triggered by DTMF Signal:** Enable or disable relay triggering via DTMF (RFC2833 supported).
- **DTMF Code:** Specify the DTMF code used to trigger the relay.
- **Trigger by Call Status:** Enable or disable relay triggering based on call status changes.
- **Call Status:** Select the call state that triggers the relay, including: Outgoing, Incoming, Incoming/Outgoing, Answered, and Hangup.
- **Reset Mode:** Select how the relay returns to its original state.
- **Duration (sec):** Available when Delay Reset is selected. Defines how long the relay remains active before resetting.
- **Triggered via Dispatch Console Music:** Select the relay behavior when the Dispatch Console broadcast music is active.
- **Triggered via Dispatch Console Alarm:** Select the relay behavior when the Dispatch Console broadcast alarm is active.

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

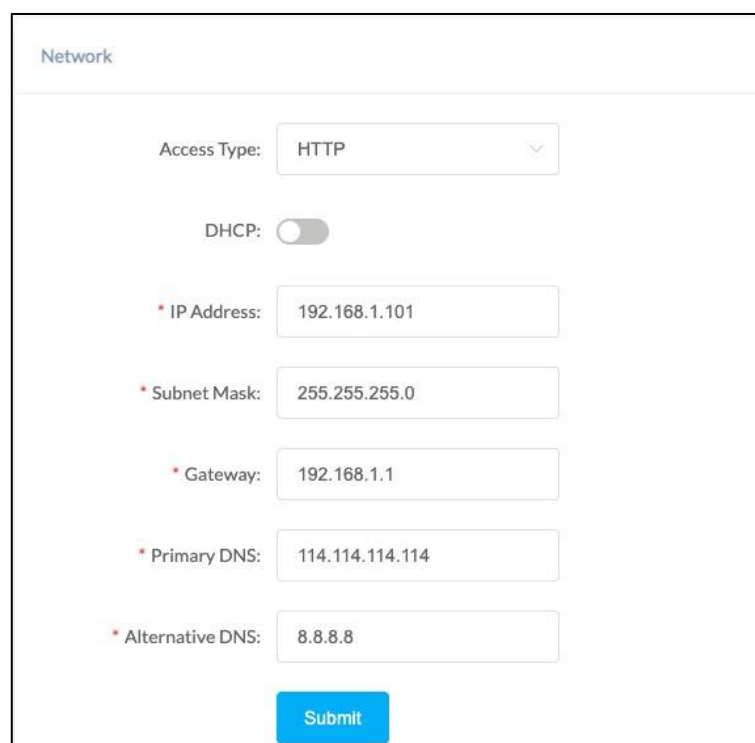
7. System Settings

7.1 Network

IPS-H1 uses DHCP to dynamically obtain IP addresses by default.

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

Turn the DHCP switch button off to show the network parameter settings.



Network Configuration

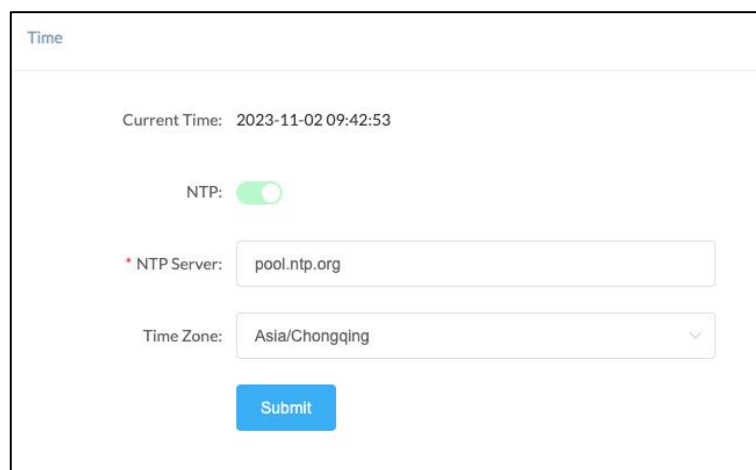
- **Access Type:** Specify the access method of the website, which currently supports HTTP and HTTPS.
- **IP Address:** Enter a vacant IP address within your LAN.
- **Subnet Mask:** Enter the subnet mask of your LAN.
- **Gateway:** Enter the default gateway of your LAN, this is essential for the device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

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

7.2 Time

IPS-H1 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' set to 'Asia/Chongqing'. At the bottom of the form is a blue 'Submit' button.

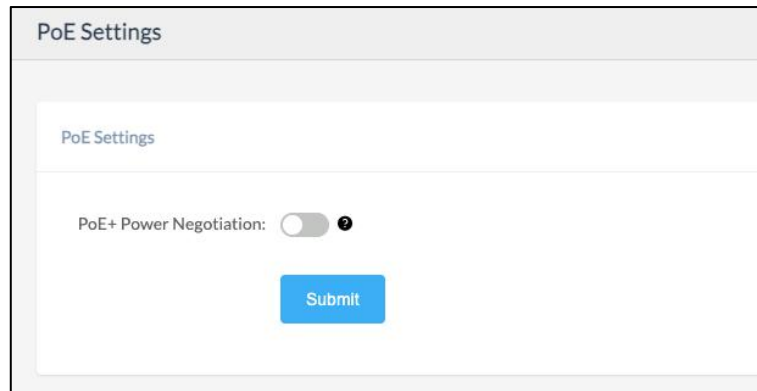
Time Settings

- **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 PoE Settings

This page allows users to configure PoE power negotiation settings. When the connected PoE injector or switch supports negotiation but does not provide PoE+ (IEEE 802.3at) power by default, this option can be enabled to request PoE+ power from the power source.

To enable the settings, please go to **System --> PoE Settings** page.

The image shows a web interface for PoE Settings. It has a title bar 'PoE Settings' and a main content area with the same title. Below the title, there is a label 'PoE+ Power Negotiation:' followed by a toggle switch that is currently turned off. To the right of the toggle is a small circular icon with a question mark. At the bottom of the form is a blue 'Submit' button.

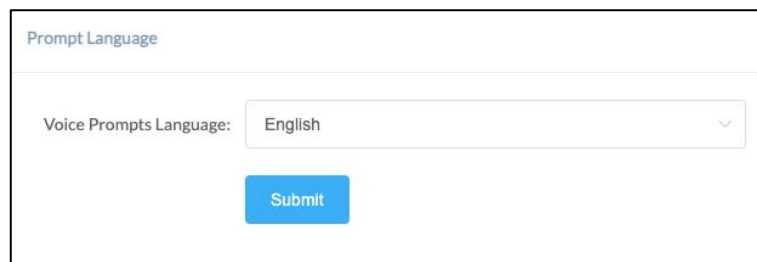
PoE Settings

- **PoE+ Power Negotiation:** Enable/Disable PoE+ (IEEE 802.3at) power negotiation. When enabled, the device will request higher power from the connected PoE power source if supported.

7.4 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 image shows a web interface for Prompt Language. It has a title bar 'Prompt Language' and a main content area with the same title. Below the title, there is a label 'Voice Prompts Language:' followed by a dropdown menu showing 'English'. At the bottom of the form is a blue 'Submit' button.

Prompt Language

7.5 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" (marked with a red asterisk), and "New Password" (marked with a red asterisk). 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.

7.6 Reboot & Reset

IPS-H1 can be rebooted and reset from the web management interface.

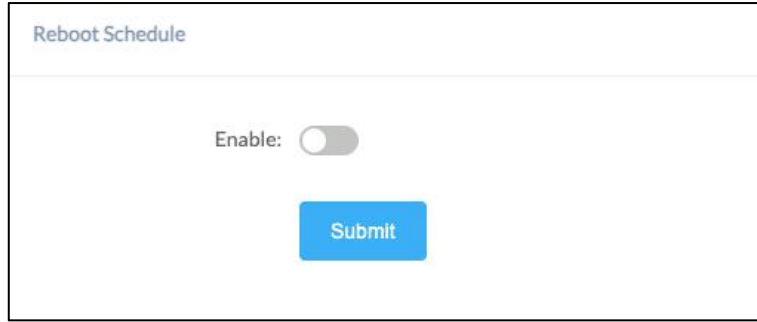
If you need to reboot or reset the device, please go to **System --> Reboot & Reset** page.

The screenshot shows a web interface with two sections. The top section is titled "Reboot" and contains a warning: "Warning: Rebooting the device will interrupt all ongoing broadcasting, intercom and calls!". Below the warning is a red "Reboot" button. The bottom section is titled "Reset" and contains a warning: "Warning: Resetting the device will interrupt all ongoing broadcasting, intercom and calls, and it will empty all configurations!". Below the warning is a red "Reset" button.

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.

Upgrade

Current Firmware Info

Current Firmware Version: s0.0.0.014

Last Update: 2026-07-08

Upgrade

Warning: It will take around 150 seconds to complete the upgrade process, during the upgrade process please DO NOT power-off the device!

Reset Factory Defaults: ☐

Firmware:



Drag files here, or [Click Upload](#)

Upgrade Settings

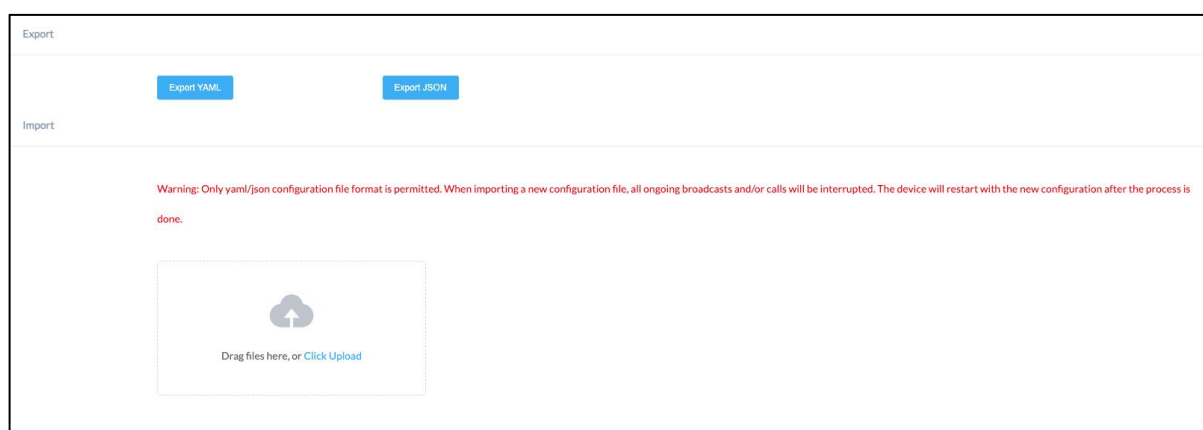
- **Current Firmware Version:** Displays the version currently used by the system.
- **Last Update:** Displays the last system updating time.
- **Reset Factory Defaults:** Specify whether to restore factory settings when upgrading.
- **Firmware:** Click to select the firmware that needs to be used to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

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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

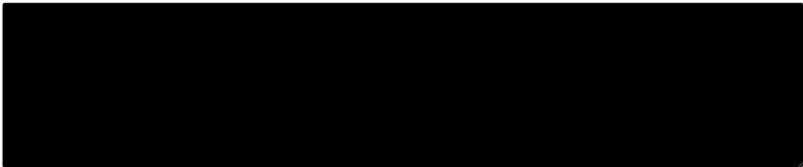
8.4 Diagnostic

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 '.pcap' file for immediate viewing and data analysis.

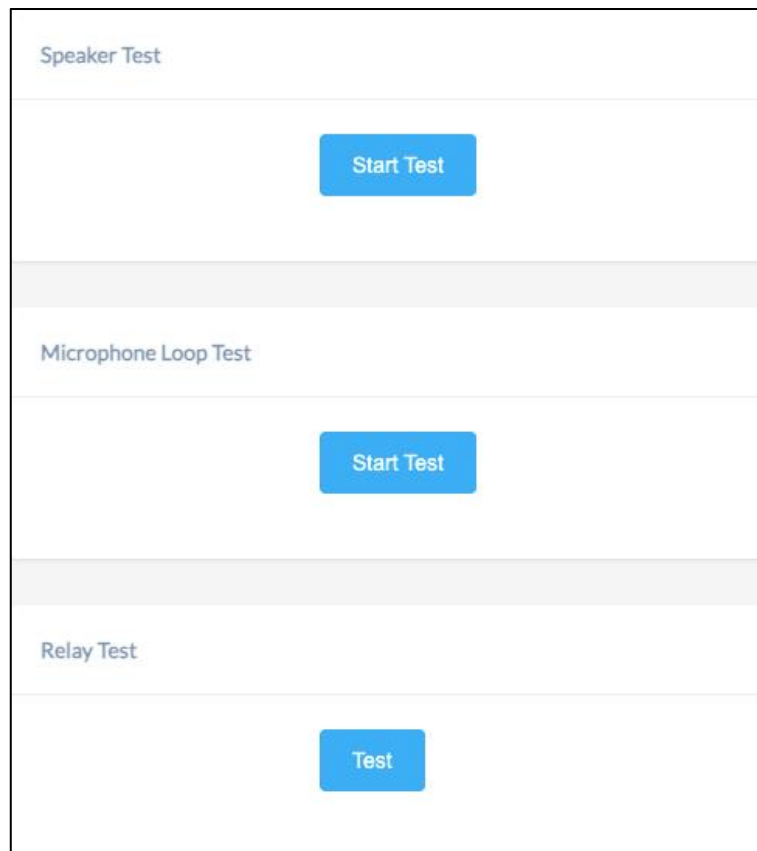
Please go to **Maintenance --> Ethernet Capture** page to operate.



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

System Logs

