IPS-W1 Network Wall Mount Speaker

User Guide

www.zycoo.com

zycoo@zycoo.com

© 2026 Zycoo Communications LLC All rights reserved

Contents

1. Preface 4
1.1 Audience 4
1.2 Revision History 4
2. Overview 5
2.1 Product Overview 5
2.2 Product Specifications 5
3. Login the Device 6
3.1 Accessing the Web GUI 6
3.2 Device Info 7
4. SIP Settings 10
4.1 SIP Account Settings 10
4.2 P2P Account Settings 12
4.3 Advance SIP Settings 13
4.3.1 SIP Parameter Settings 13
4.3.2 SIP Function Settings 15
4.3.3 Audio Codecs 16
5. Function Settings 17
5.1 ONVIF Settings 17
5.2 Multicast 18
6. Advanced Settings 20
6.1 Advanced Audio 20
6.2 Audio Priority Settings 24
6.3 Ringtones File 25
6.4 API Settings 25
6.5 I/O Settings 27
6.6 PTP Settings 30
7. System Settings 32
7.1 Network 32
7.2 Time 33
7.3 PoE Settings 33
7.4 Prompt Language 34
7.5 Account 34
7.6 Reboot & Reset 35

8. Maintenance 37
8.1 Upgrade 37
8.2 Import/Export 38
8.3 Auto Provisioning 38
8.4 Diagnostic 39
8.5 Ethernet Capture 39
8.6 Test 40
9. Reports 42
9.1 Call Logs 42
9.2 System Logs 42

1. Preface

1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and manage the IPS-W1 Network Wall Mount Speaker. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the IPS-W1 Network Wall Mount 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-W1 Network Wall Mount 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 clear, intelligible, and reliable audio performance for a wide range of indoor environments.

The IPS-W1 can operate as a standalone network speaker with up to 16W audio output, or be paired with a ZYCOO APS-W1 Extension Wall Mount Speaker to expand audio coverage. This deployment model helps reduce network cabling, PoE switch port usage, and overall installation costs.

The IPS-W1 is ideal for corridors, classrooms, offices, retail spaces, healthcare facilities, and other environments requiring clear and directional audio distribution. It provides dependable voice communication for both everyday announcements and critical emergency notifications.

2.2 Product Specifications

IPS-W1 Network Wall Mount Speaker Specifications		
Speaker Components	5.25" Woofer Driver Unit + 1.5" Tweeter Driver Unit	
Sensitivity	85dB/1m/1W	
Max Sound Pressure Level	95dB	
Frequency Range	70Hz – 20KHz	

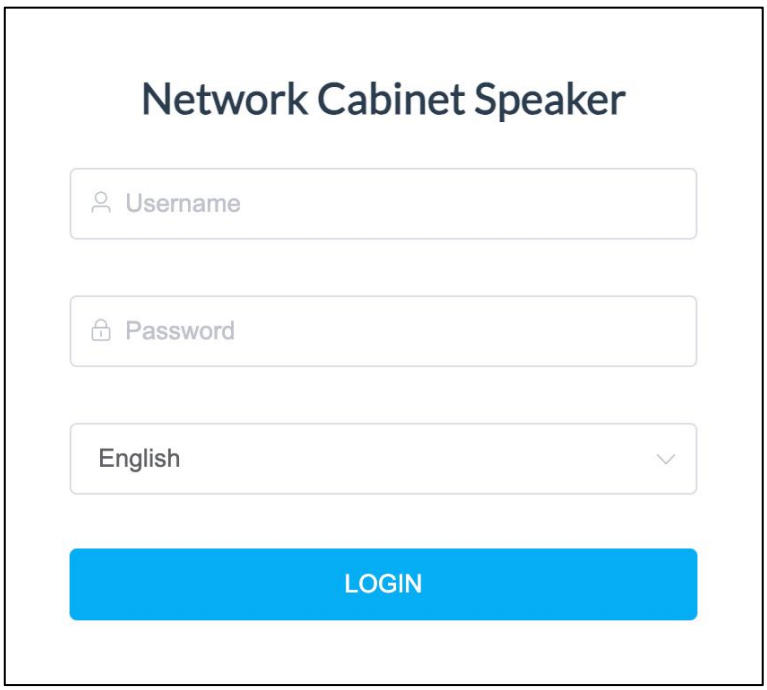
3. Login the Device

3.1 Accessing the Web GUI

IPS-W1 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.

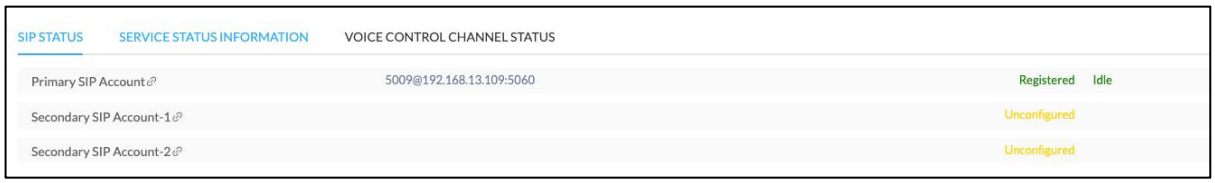


Login Interface

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

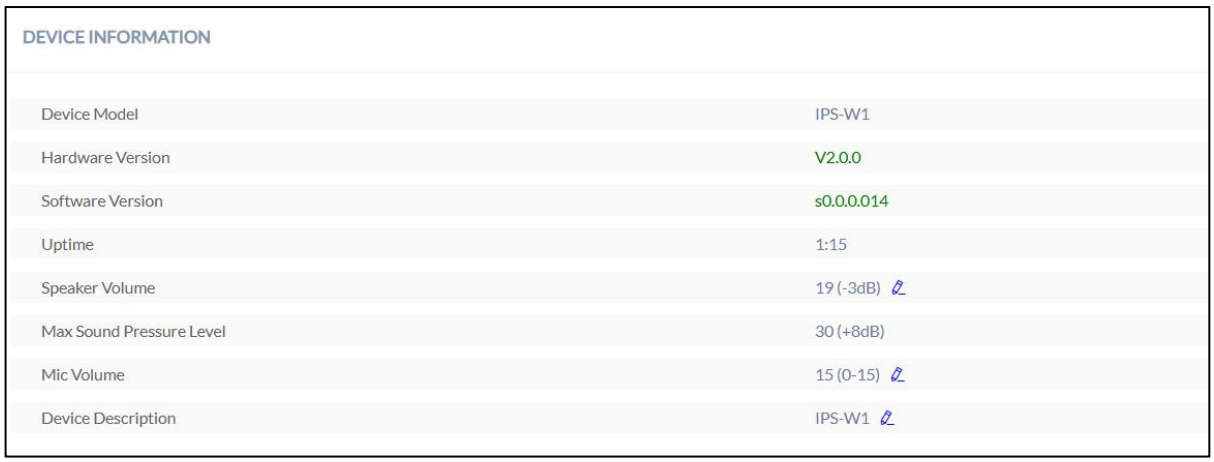
3.2 Device Info

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

- **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.

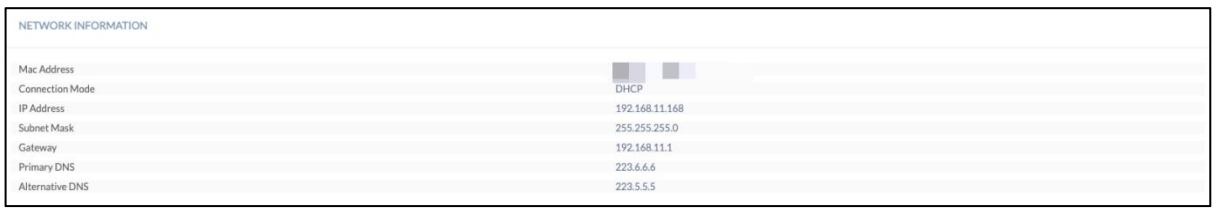


DEVICE INFORMATION

Device Model	IPS-W1
Hardware Version	V2.0.0
Software Version	s0.0.0.014
Uptime	1:15
Speaker Volume	19 (-3dB)
Max Sound Pressure Level	30 (+8dB)
Mic Volume	15 (0-15)
Device Description	IPS-W1

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.
- **Up Time:** 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.
- **Sub-net Mask:** The current sub-net mask of the device.

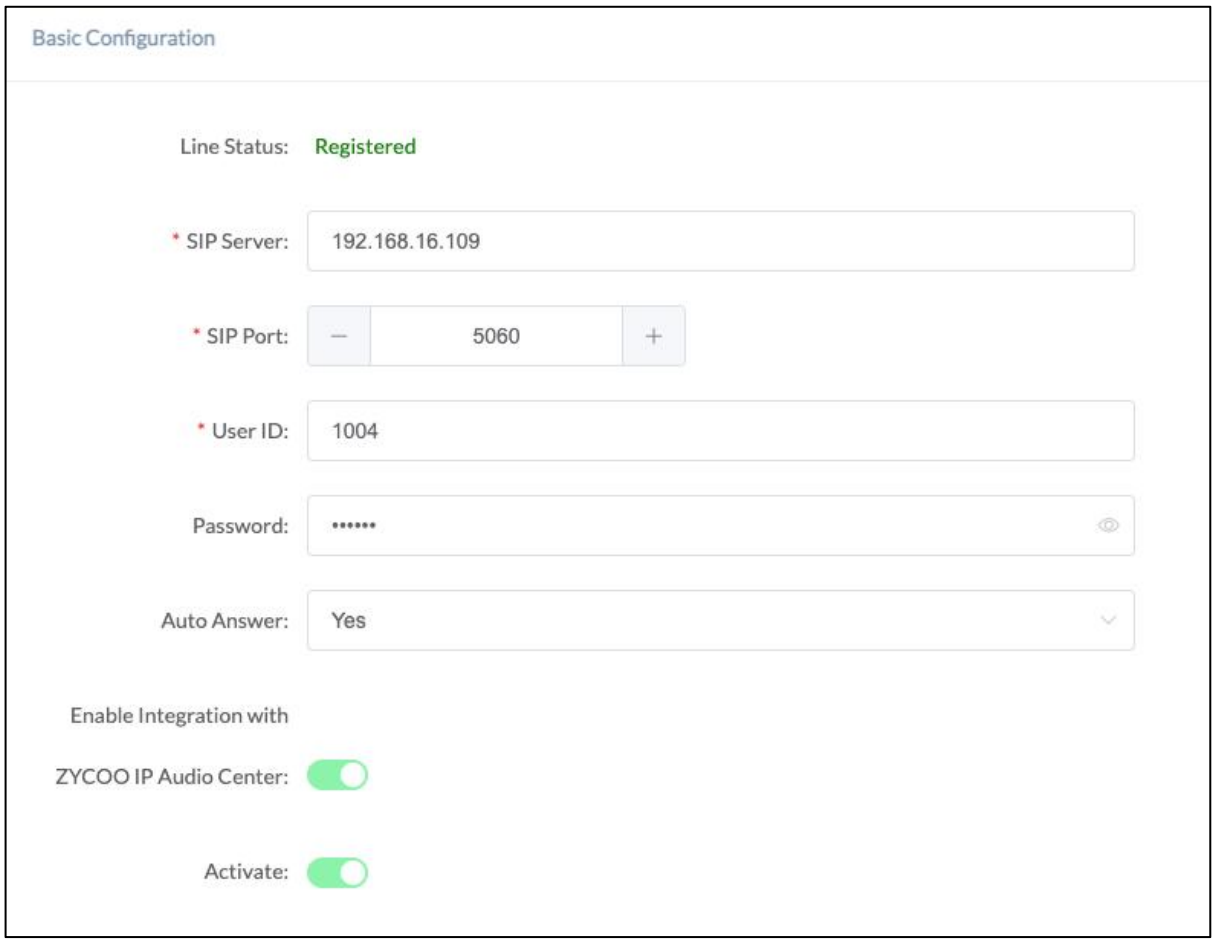
- **Gateway:** The gateway address currently used by the device.
- **Primary DNS:** The primary domain name server address used by the device.
- **Alternative DNS:** The secondary domain name server address used by the device.

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.

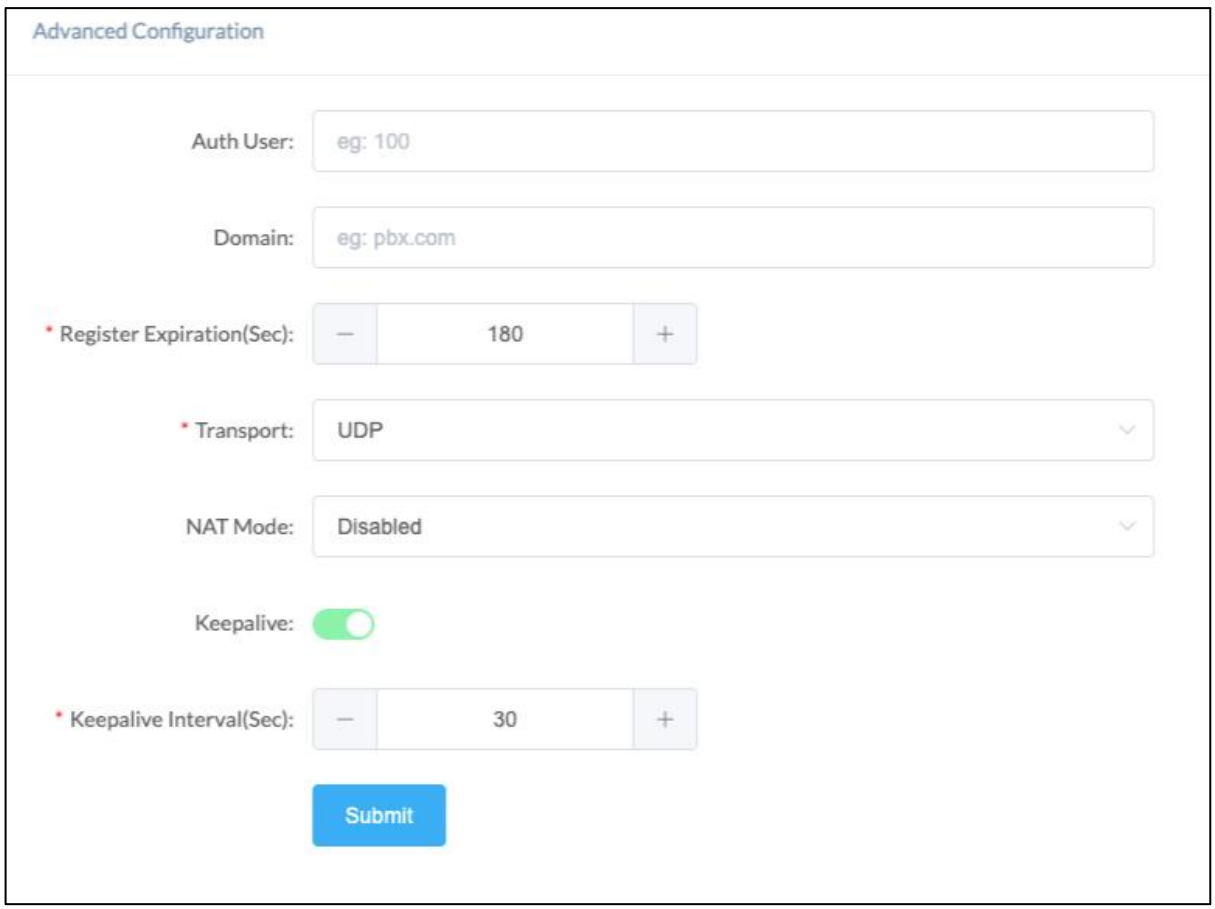
Please go to **SIP Settings --> Primary SIP Account / Secondary SIP Account-1 / Secondary SIP Account-2** page.



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.



SIP Account - Advanced Configuration

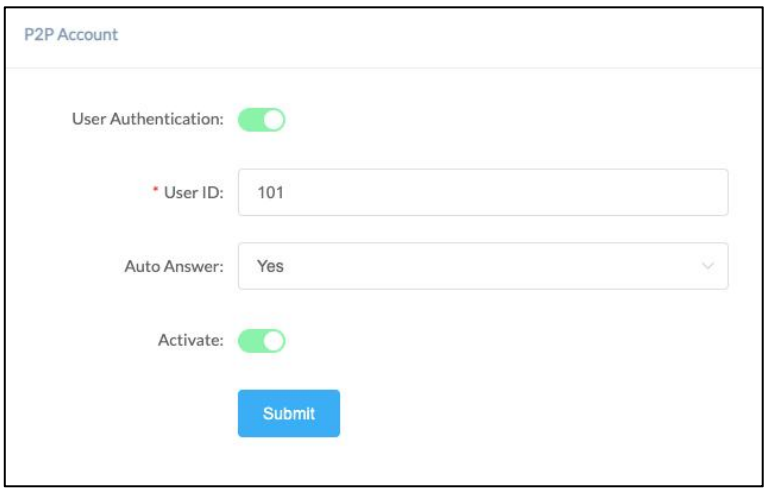
- **Auth User:** Enter the authorized username for the SIP account.

- **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.



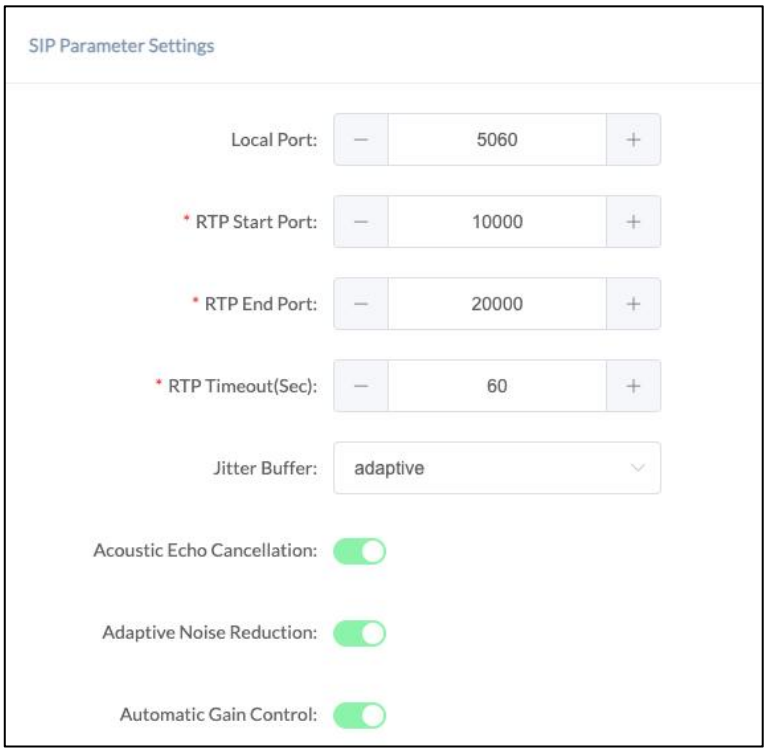
P2P Account

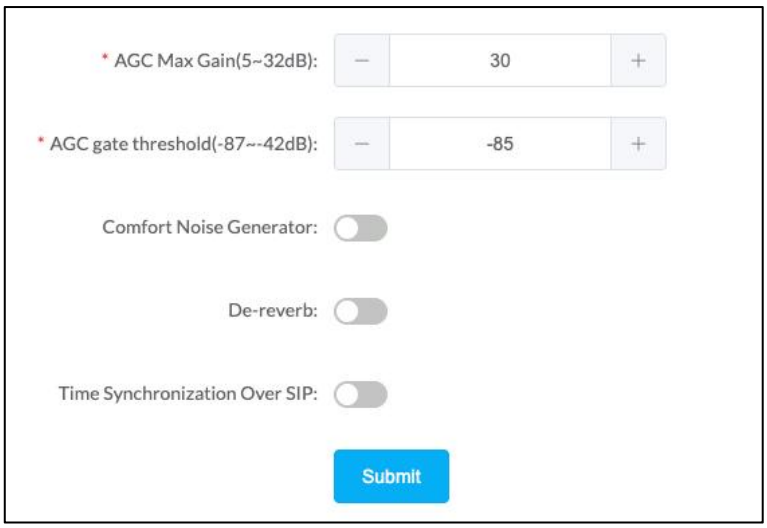
- **User Authentication:** Enable/Disable P2P authentication. If disabled, you can directly enter this device's IP address in the target field of the peer device. If enabled, you must use the following format in the target field of the peer device: This device's P2P User ID + IP address (e.g., 101@192.168.1.101).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



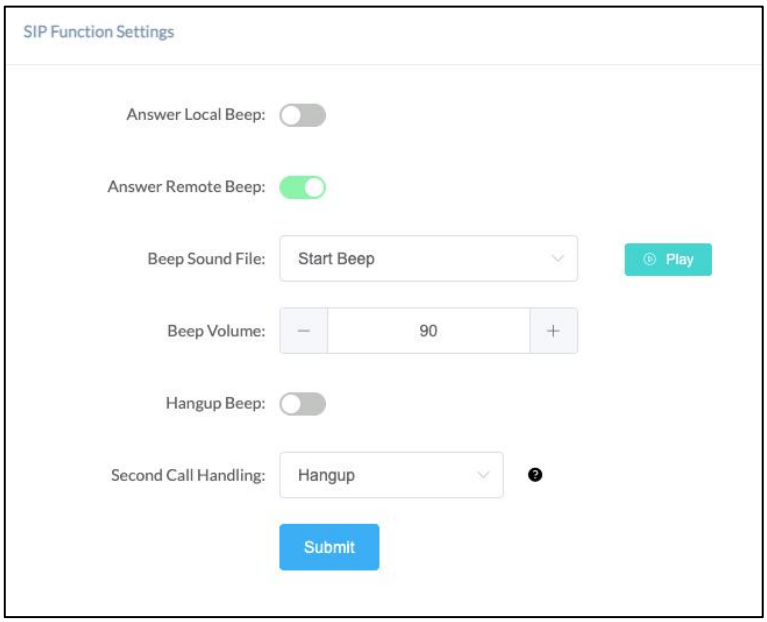


SIP Parameter Settings

- **Local Port:** 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:** 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:

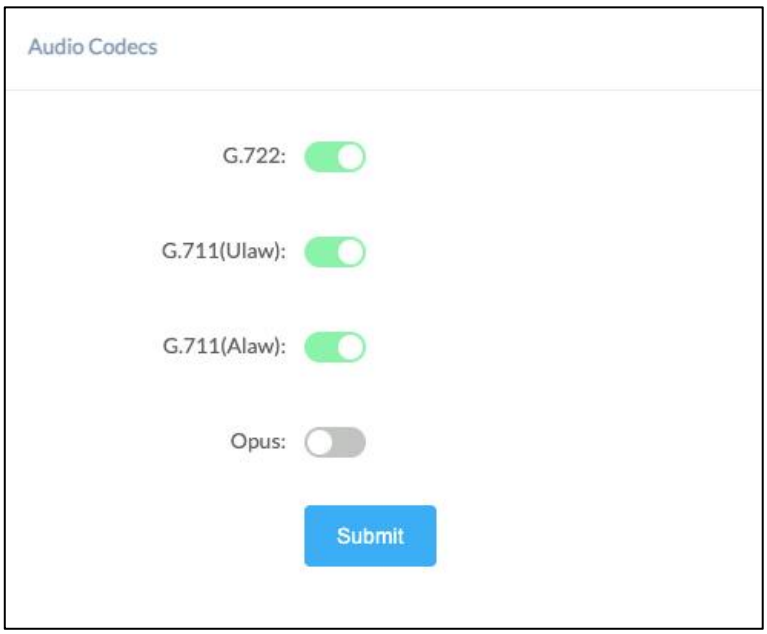
Hangup: Directly hang up the second call.

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-W1 supports 4 audio codecs: G.722 (wideband codec), G.711(Ulaw), G.711(Alaw), and Opus.



Audio Codecs

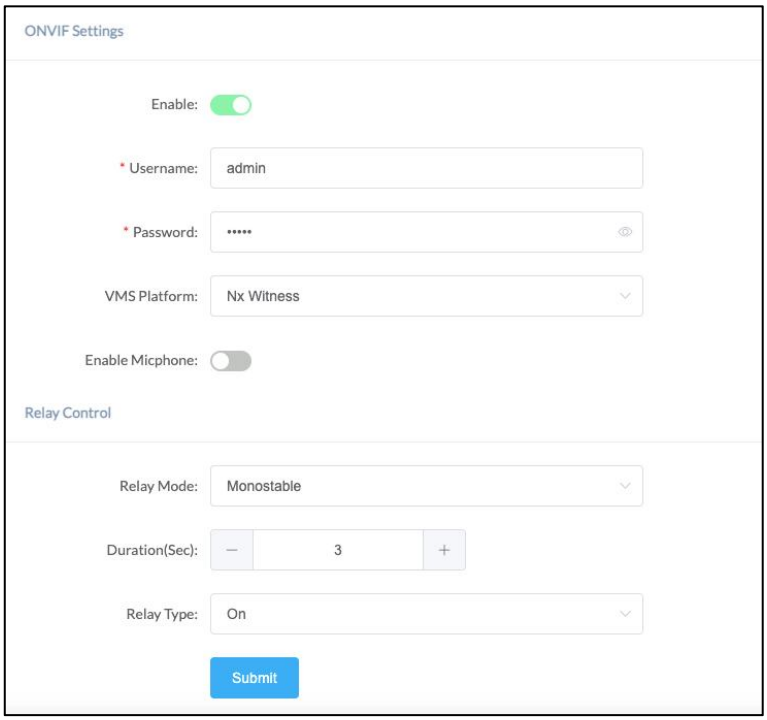
Please keep at least one codec enabled and supported by the SIP server, otherwise, SIP paging will not work.

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

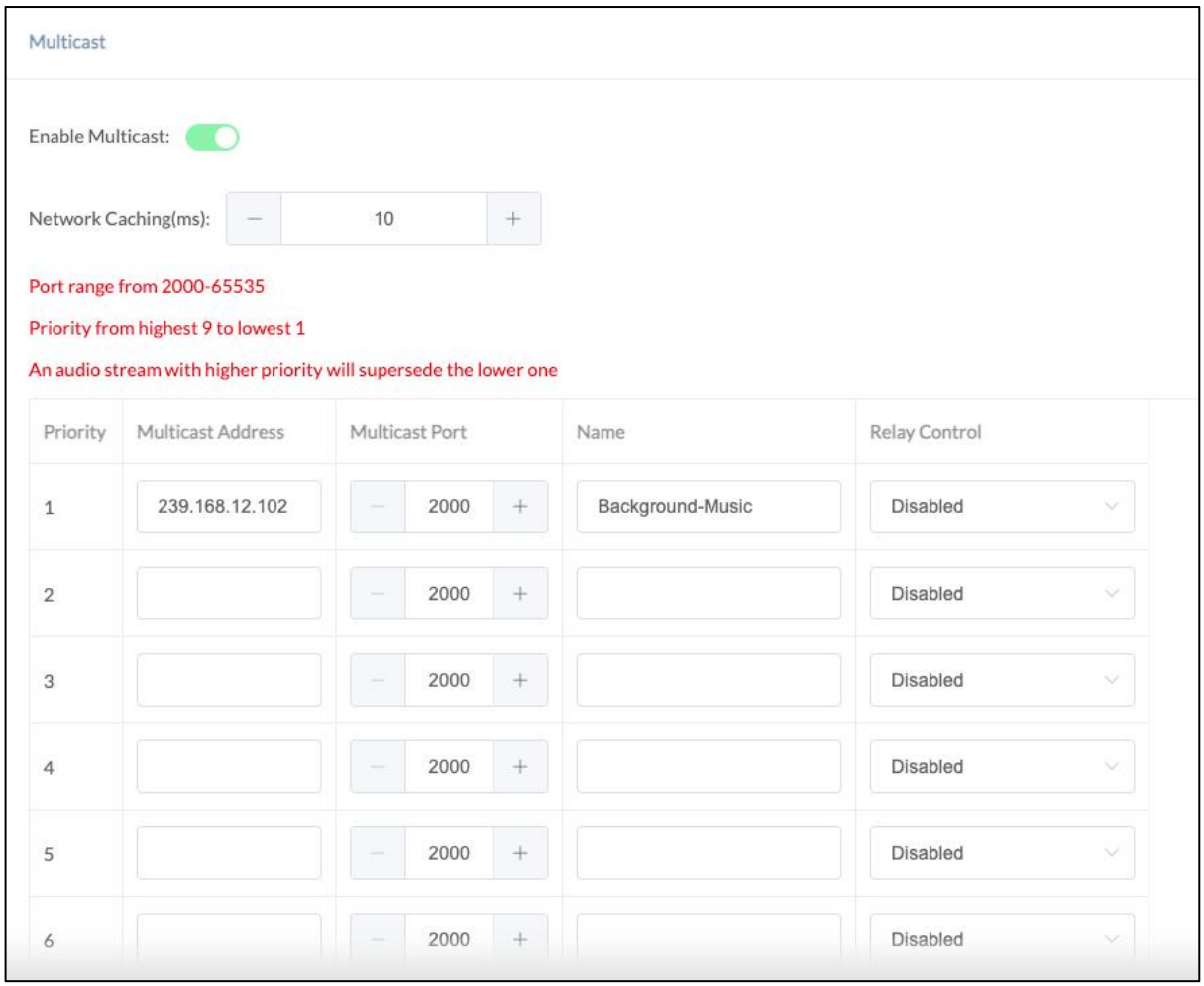
- **Enable:** Enable/Disable ONVIF integration for compatibility with ONVIF-supported VMS platforms.
- **Username:** Enter an account username with matching credentials for adding devices to the VMS platform.
- **Password:** Enter a matching password for the account to add devices to the VMS platform.

- **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.



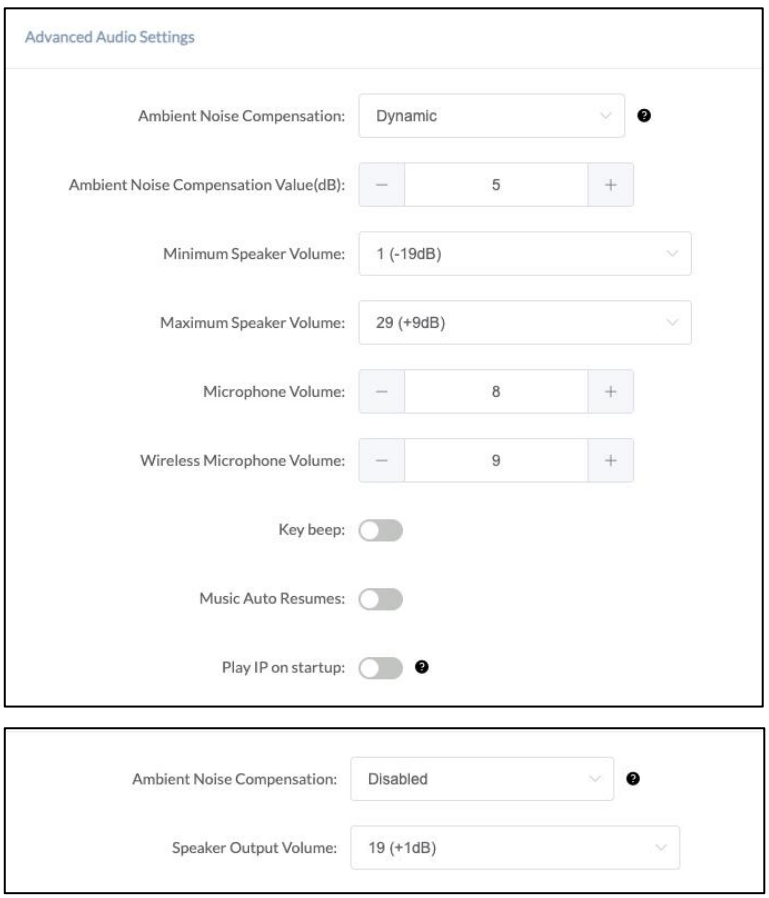
Multicast

- **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.

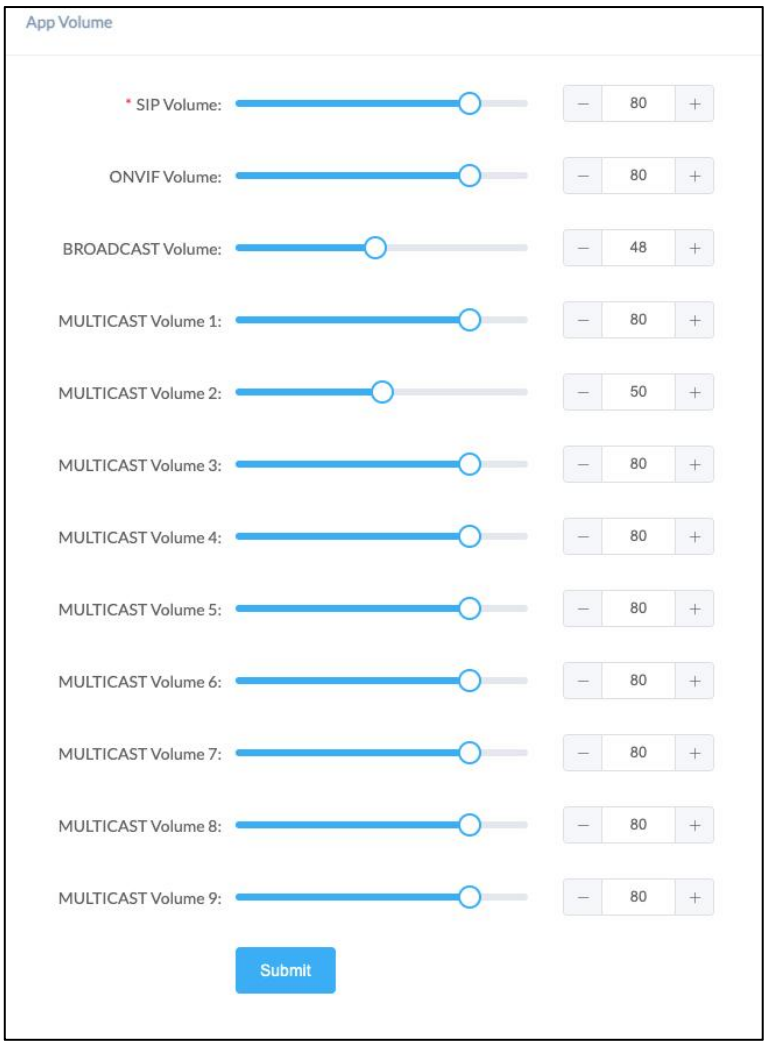


Advanced Audio Settings

- **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 ≈ 0 dB (unity gain), and 30 ≈ +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 ≈ 0 dB (unity gain), and 30 ≈ +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 ≈ 0 dB (unit gain), and 30 ≈ +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 re-connection.
- **Play IP on Startup:** When enabled, the device will announce its IP address once upon startup.

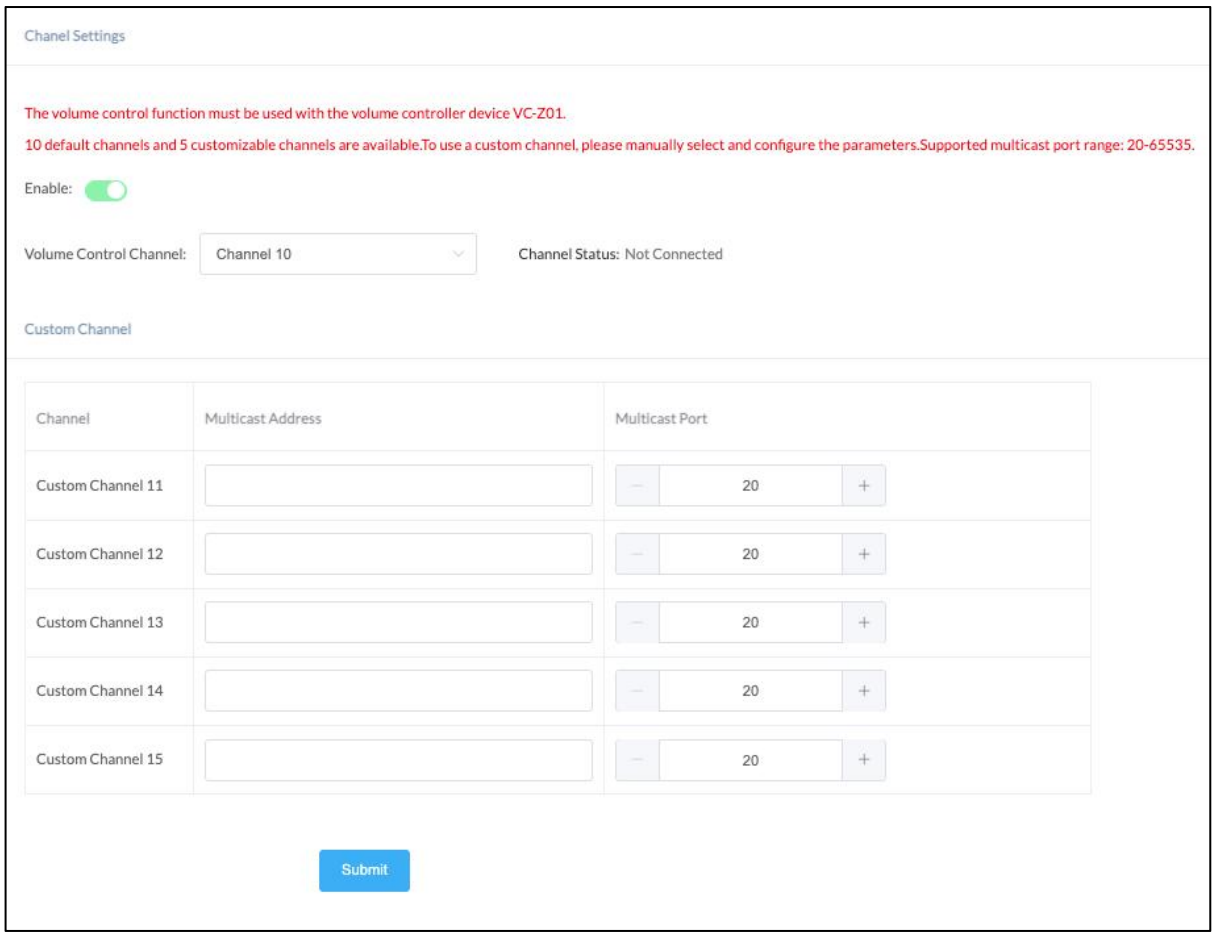


App Volume Settings

- **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.



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.

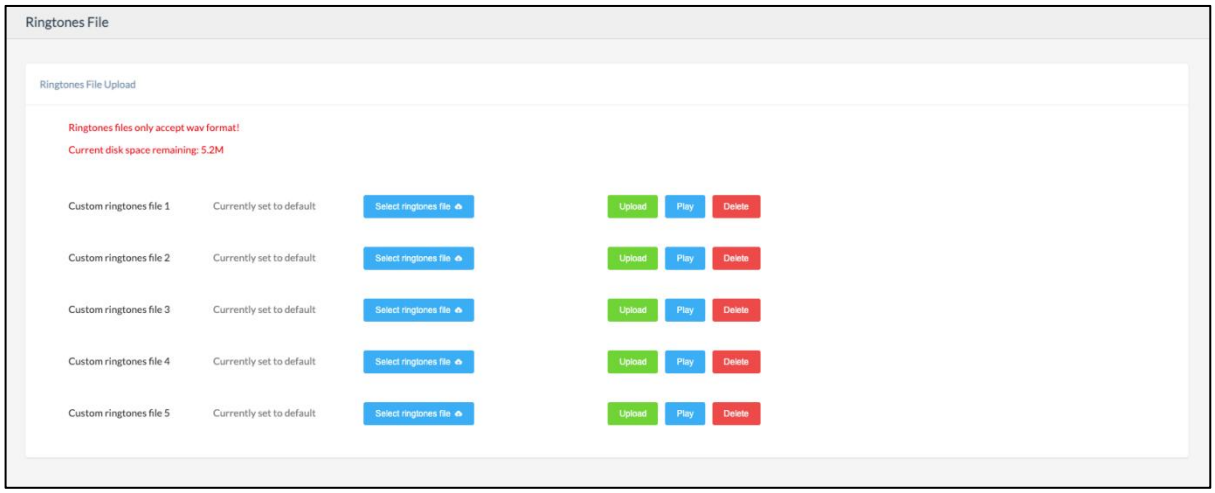
Priority 1 is the highest. You can drag the arrow on the right side to adjust the priority. The execution of a high-priority audio application will interrupt the current low-priority audio application.

Audio Priority Settings

6.3 Ringtones File

Ringtones File allows users to self-upload up to 5ßM 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.

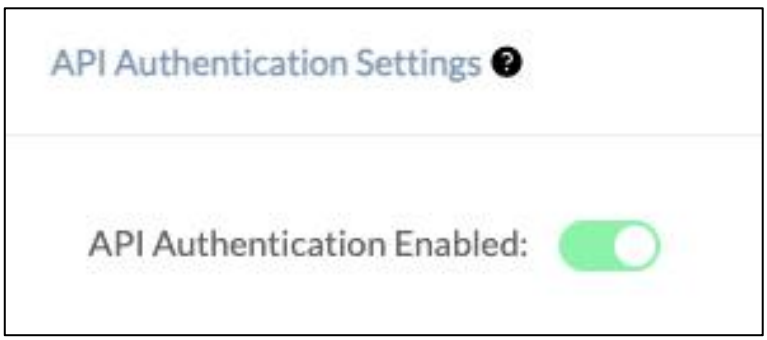


Ringtones File

6.4 API Settings

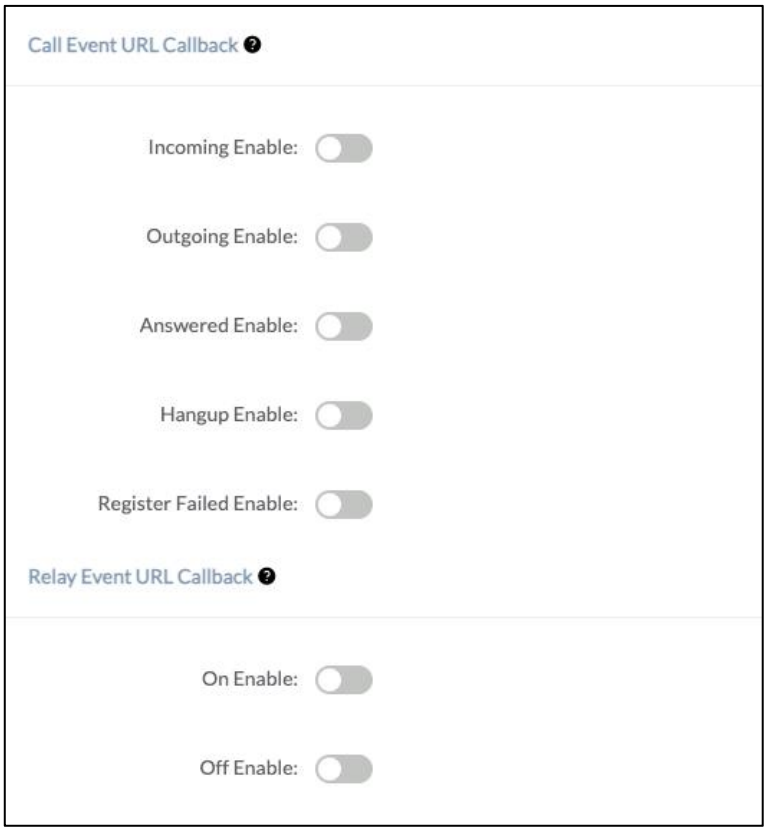
This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the changing status of the call and/or relay.

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.



Call Event URL Callback & Relay Event URL Callback

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

For example:

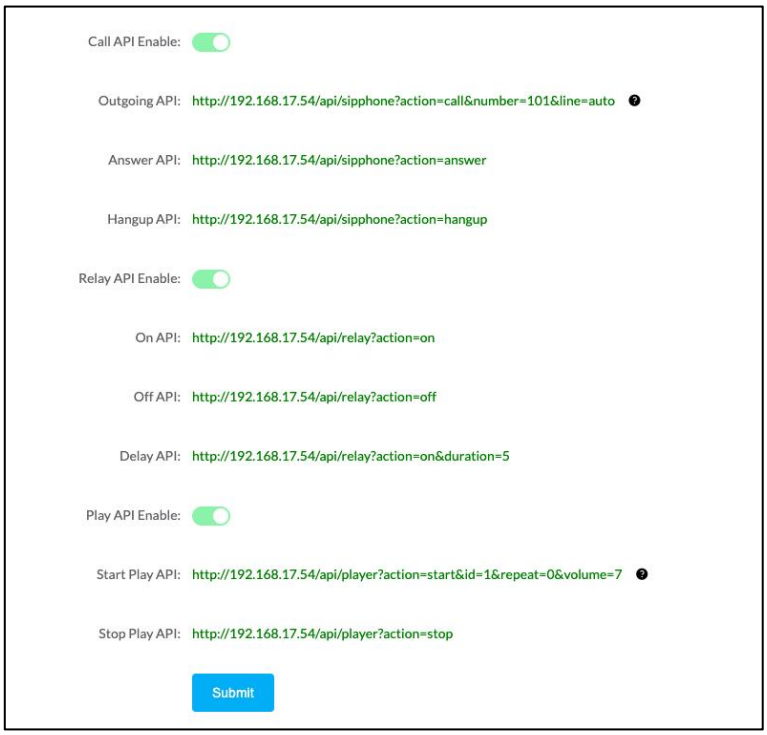
${ip}:	The current IP address of the device
${mac}:	The current MAC address of the device
${ua}:	The account of the current call
${number}:	The number of the current call

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

For example:

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

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



API Settings

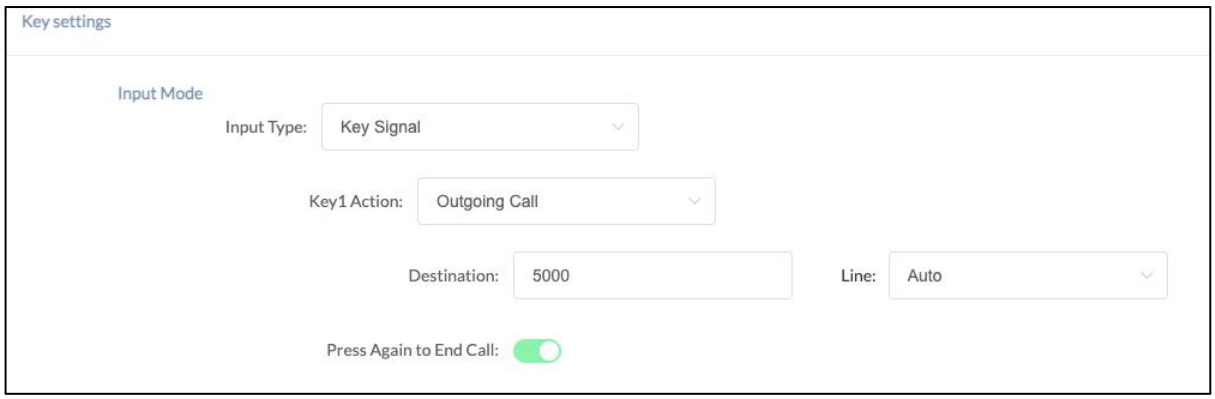
Using the API interface to realize features such as device linkage, call control, relay control, and play sound by the systems.

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

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.



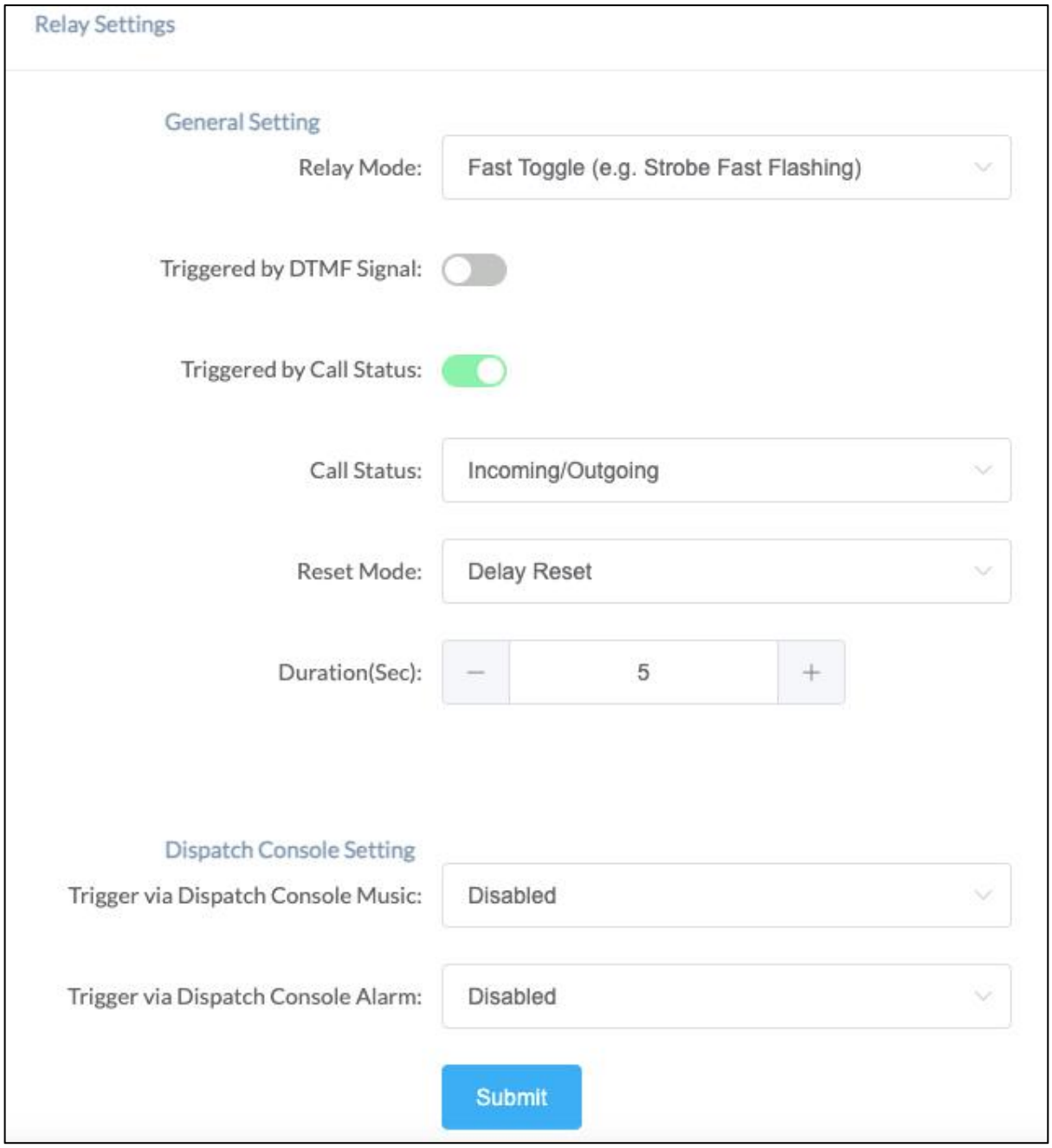
Input Settings - Button Signal

Input Settings - Switch Signal

- **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

- **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 Setting

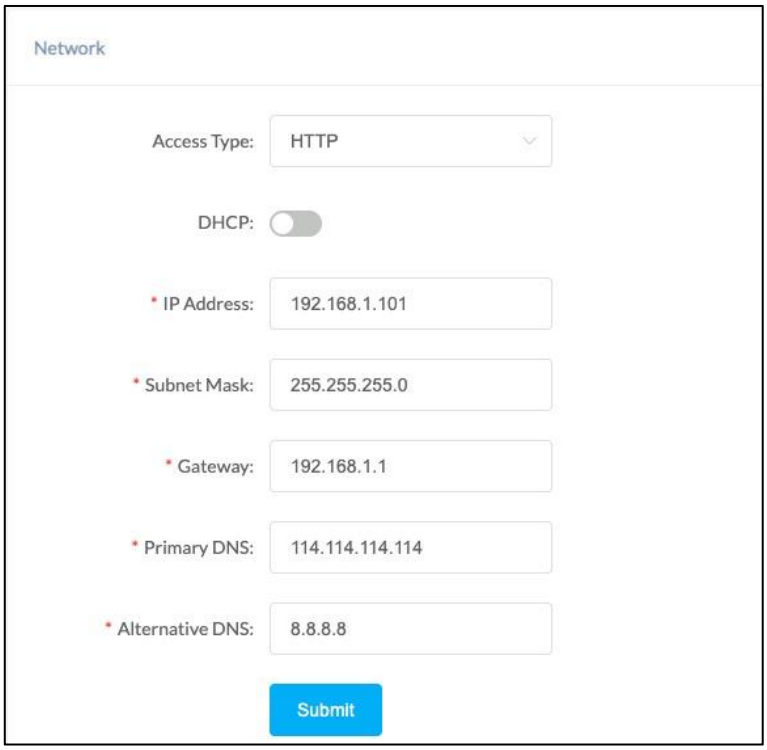
7. System Settings

7.1 Network

IPS-W1 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

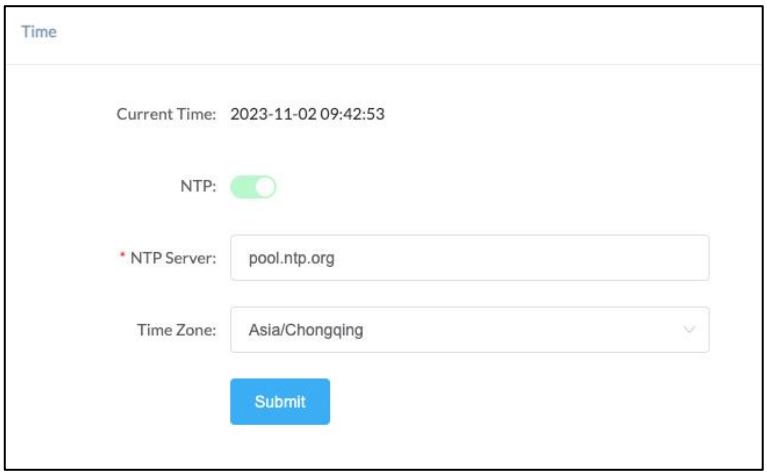
- **Access Type:** Specify the access method of the website, which currently supports HTTP and HTTPS.
- **IP Address:** Enter a vacant IP address within your LAN.
- **Subnet Mask:** Enter the subnet mask of your LAN.
- **Gateway:** Enter the default gateway of your LAN, this is essential for the device 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-W1 obtains the time from the network time servers using NTP.

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



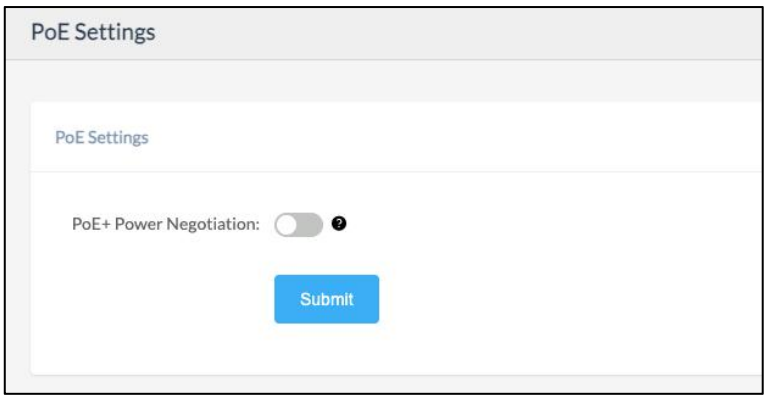
Time Settings

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

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.



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

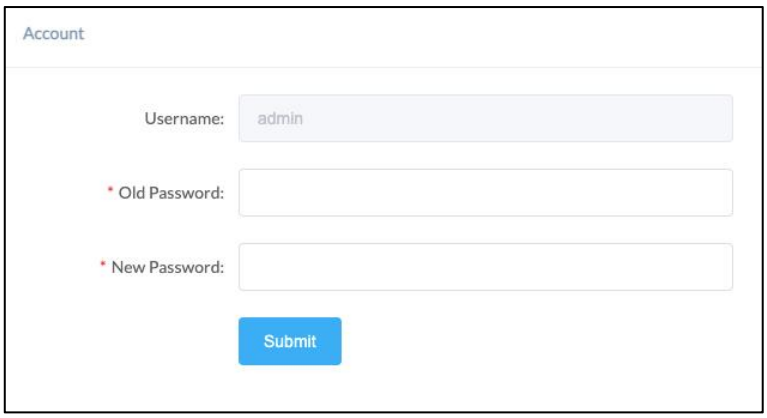
The language of local voice prompts, like IP address announcements. Currently, only Chinese and English are provided.

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

Prompt Language

7.5 Account

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



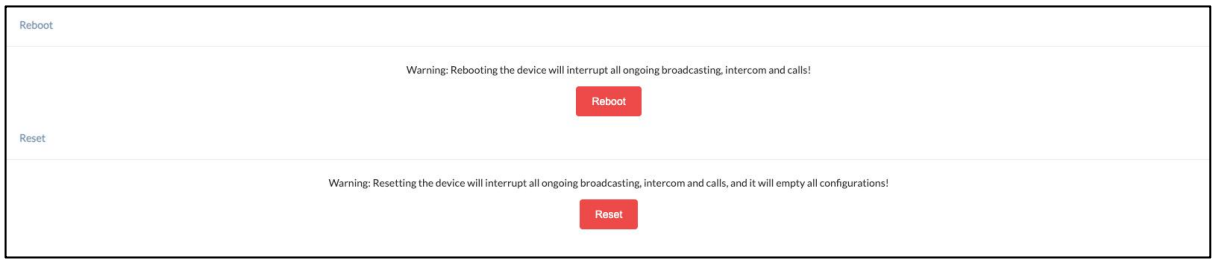
Web Password Settings

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

7.6 Reboot & Reset

IPS-W1 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!

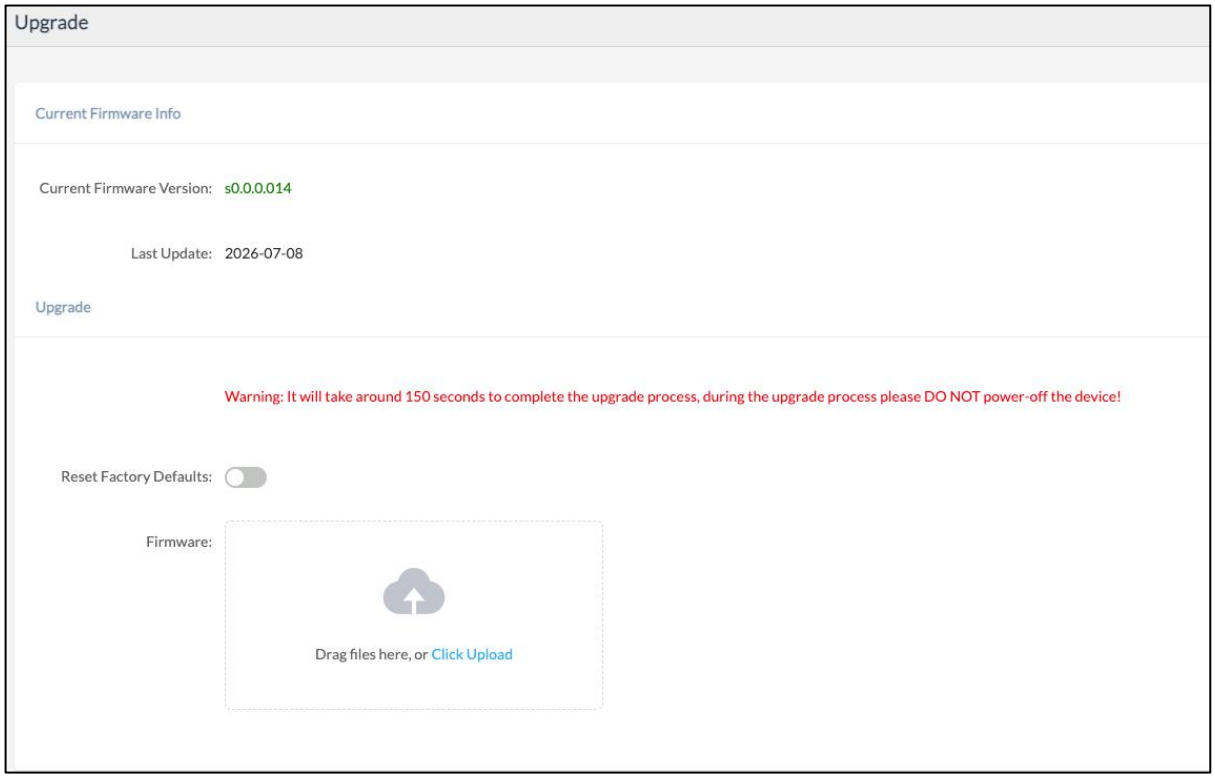
Reboot Schedule

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.



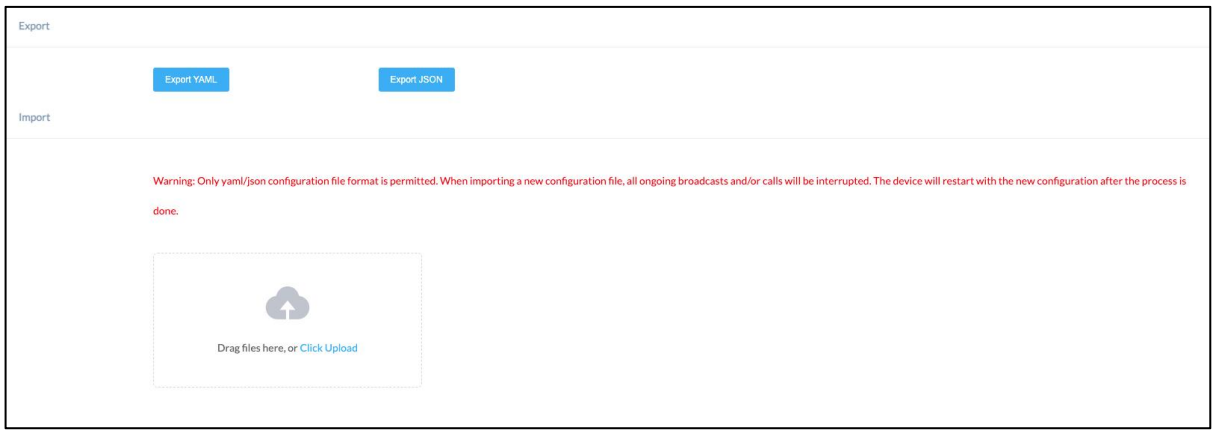
Upgrade Settings

- **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.

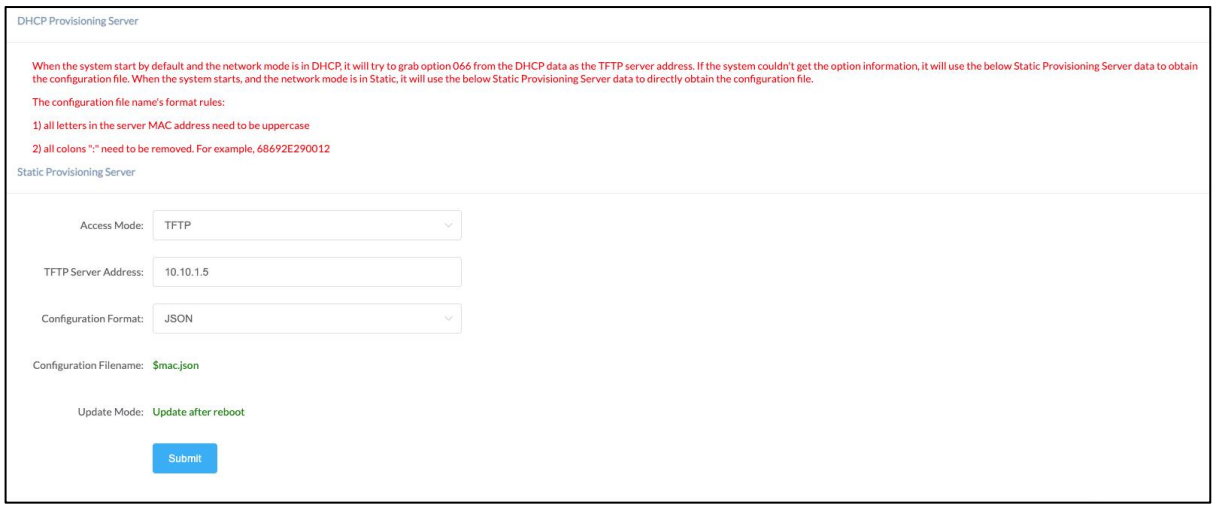
When the system starts by default and the network mode is in DHCP, it will try to grab option 066 from the DHCP data as the TFTP server address. If the system couldn't get the option information, it will use the below Static Provisioning Server data to obtain the configuration file. When the system starts,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



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

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.

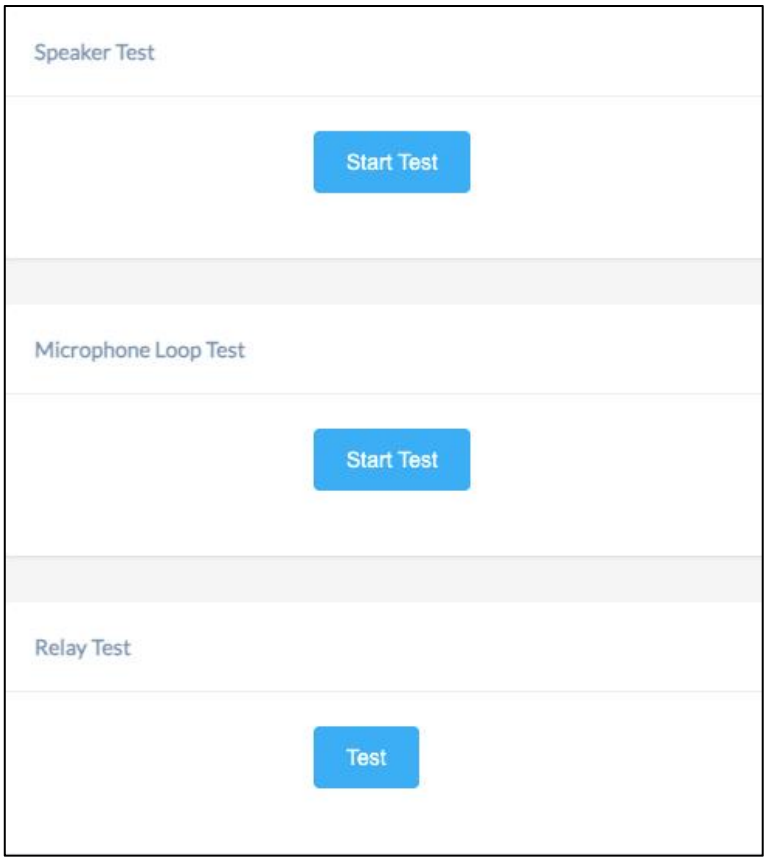
Please go to **Maintenance --> Ethernet Capture** page to operate.

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.

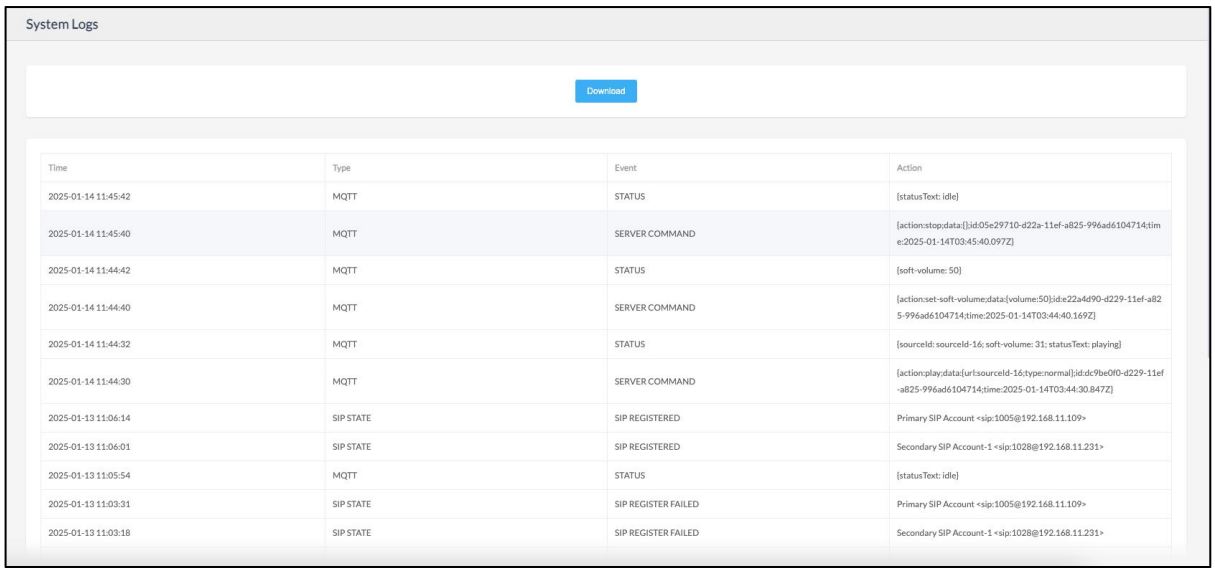
Date	Time	Account	Telephone Number	Duration	Type	Status
2024-03-15	17:18:00	sip:5011@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:47:58	sip:5015@192.168.11.62	5000	00:00:02	Inbound	Answered
2024-03-15	16:38:30	sip:5005@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:38:16	sip:5005@192.168.11.62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



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

www.zycoo.com

zycoo@zycoo.com

© 2023 Zycoo Communications LLC All rights reserved