PB-S11 Network Panic Button

User Guide

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Contents

1. Preface

1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and manage the PB-S11 Network Panic Button. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the PB-S11 Network Panic Button.

1.2 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
1.0.0	1.0.0	Updated operating instructions for software version v1.0.0	May, 2025

2. Overview

2.1 Product Overview

The ZYCOO PB-S11 Network Panic Button is a reliable emergency device designed for quick response in critical. In the event of an emergency, users can press the button to initiate a silent emergency call to preset destinations. The PB-S11 supports outbound calls and allows up to 5 call destinations to be set. It automatically polls destinations when the device is busy, unanswered, or offline, ensuring call accessibility. Additionally, it is also suitable for daily scenarios requiring consultation or assistance.

The PB-S11 is characterized by its simplicity, quick response, and high reliability. It is powered via a PoE network cable and features flexible installation options, supporting both wall-mounted and flush-mounted (compatible with standard 1-Gang electrical boxes). It can be installed under desks, counters, or on walls. The PB-S11 also allows users to upload customized tones (such as device location information), which the recipient will hear the tones first. With these features, the PB-S11 is ideal for use in classrooms, banks, courtrooms, medical institutions, 24-hour restaurants, etc.

2.2 Product Specifications

PB-S11 Network Panic Button Specifications		
Power Input	PoE (IEEE 802.3af/at)	
Microphone Sensitivity	-36±3dB	
Mounting	Flush-mounted or Wall-mounted Installation	
Ethernet	10/100Mbps Adaptive	

SIP Audio Stream	MP3 Sampling Rate 8-48KHz, Bit Rate 64-320kbps, Mono or Stereo	
Dimension	114*70*46(mm)	
Weight	208g	

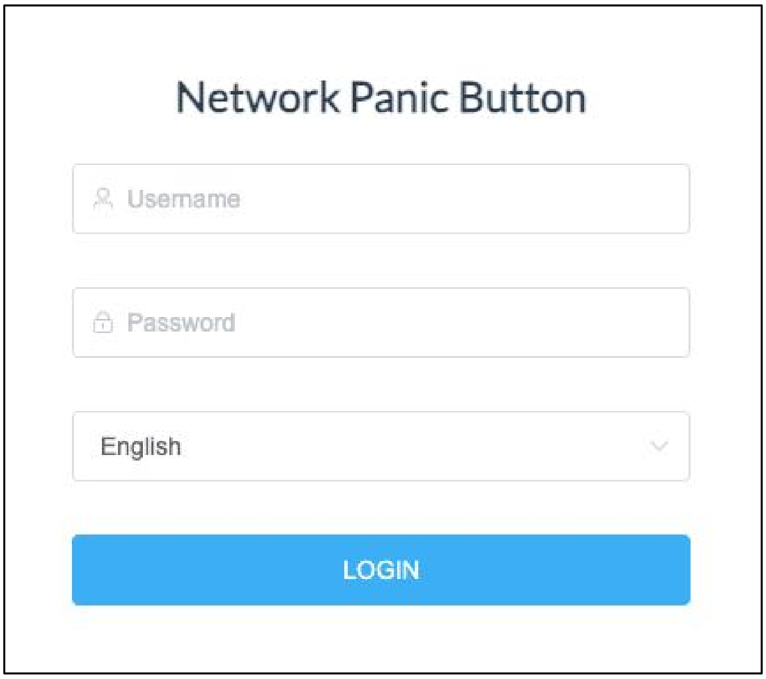
3. Login the Device

3.1 Accessing the Web GUI

PB-S11 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). You can also use the SAAD tool to scan for the device's IP address.. Enter the IP address in the browser to access the device's Web management interface.

Default username: admin
Default password: admin

For the safety purpose, it is recommended to change the default password on the first login, please go to **System --> Password Settings** page to change the password.

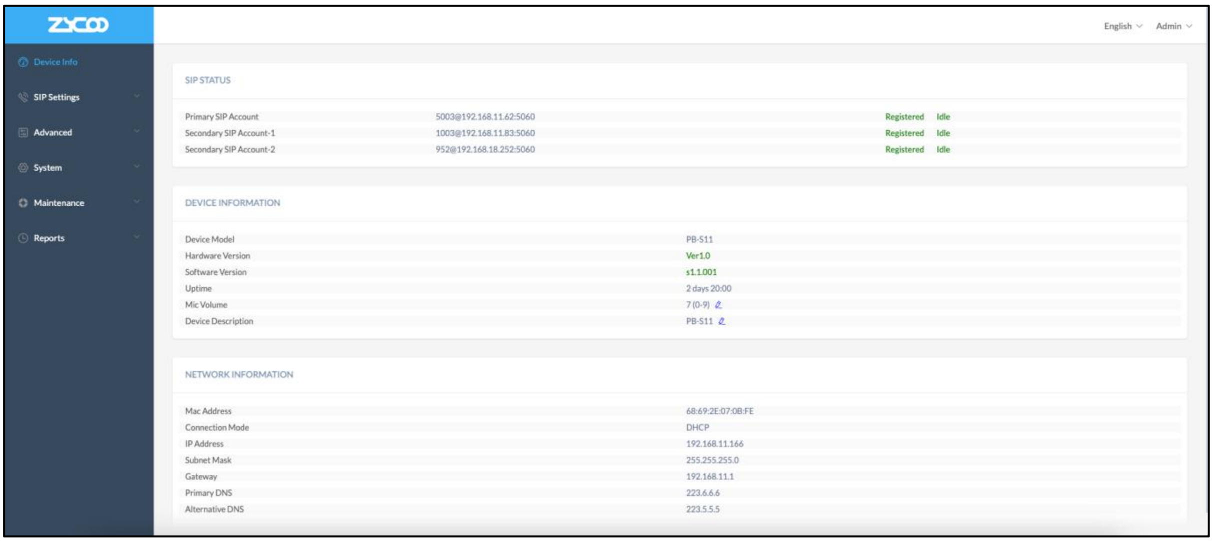


Login Interface

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

3.2 Device Info

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



SIP STATUS

Primary SIP Account	5003@192.168.11.62:5060	Registered	Idle
Secondary SIP Account-1	1003@192.168.11.83:5060	Registered	Idle
Secondary SIP Account-2	952@192.168.18.252:5060	Registered	Idle

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	PB-S11
Hardware Version	Ver1.0
Software Version	s1.1.001
Uptime	2 days 20:00
Mic Volume	7 (0-9)
Device Description	PB-S11

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 last time the device was started up.
- **Mic Volume:** Displays the current device microphone input volume.
- **Device Description:** Remark the device information. The description will be displayed in a browser tab. After the Device Description is set, the description will be displayed in the browser tab, which is convenient for distinguishing different terminals when there are many terminal configuration pages.

NETWORK INFORMATION

Mac Address	68:69:2E:07:0B:FE
Connection Mode	DHCP
IP Address	192.168.11.166
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

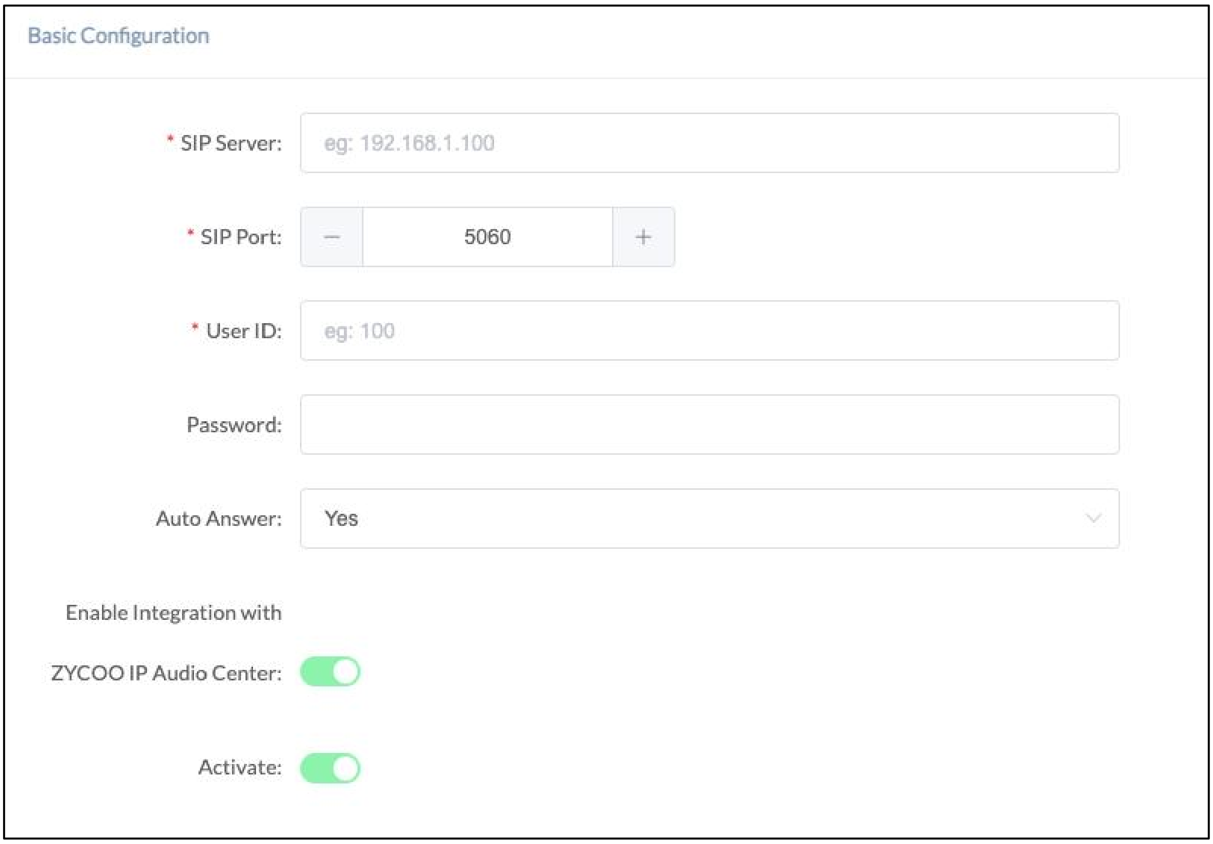
- **Mac Address:** Display the MAC address of the current device.
- **Connection Mode:** Display the network acquisition method of the device, DHCP (dynamic acquisition) or STATIC (static configuration).
- **IP Address:** The current IP address of the device.
- **Subnet Mask:** The current subnet mask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

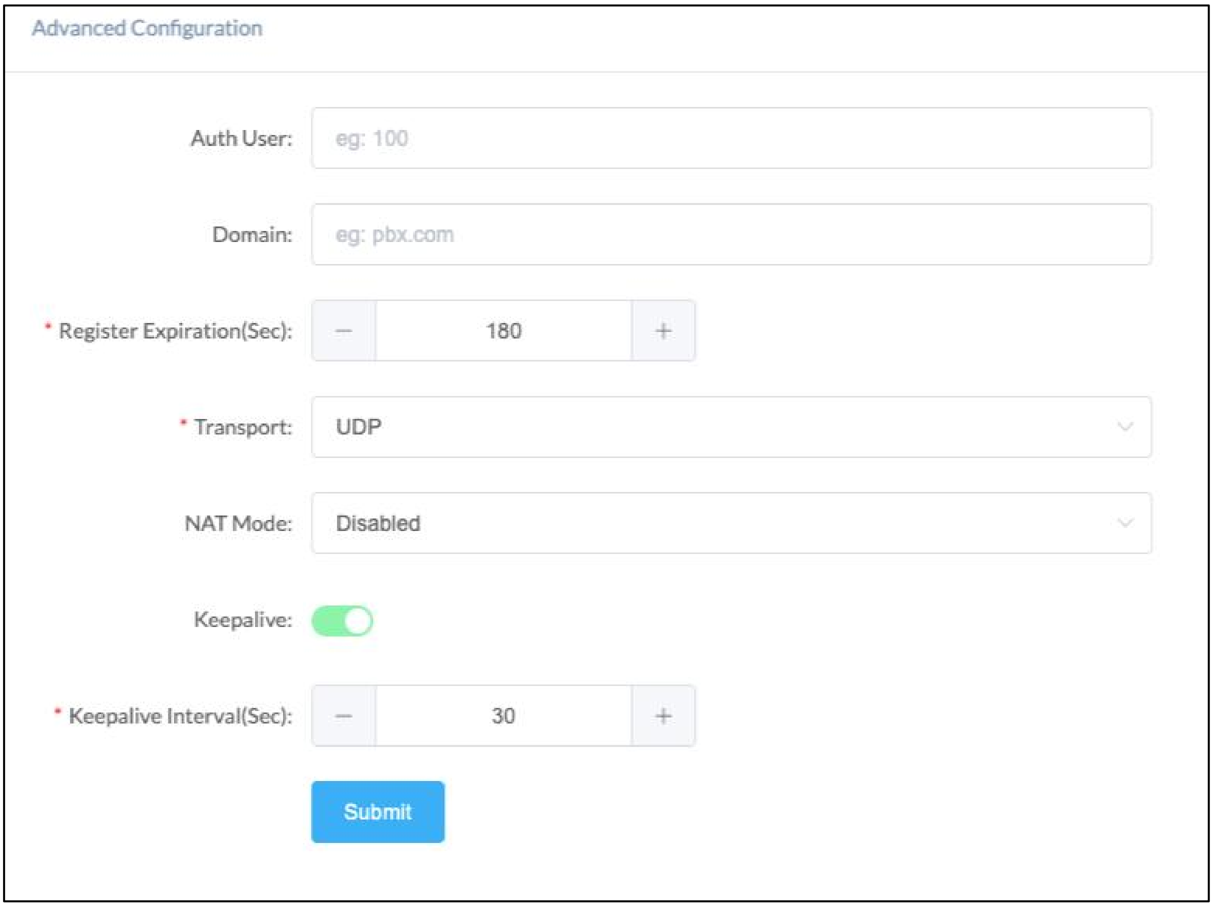
Please go to **SIP Settings --> Primary SIP Account / Secondary SIP Account-1 / Secondary SIP Account-2** page.



SIP Account - Basic Configuration

- **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.



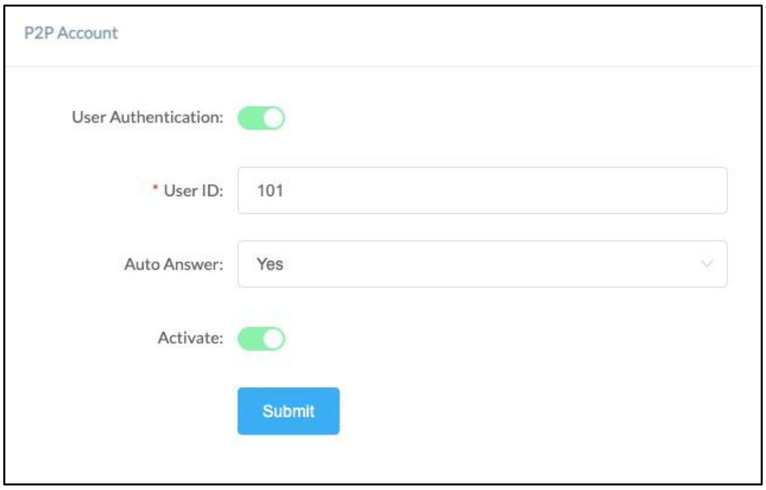
SIP Account - Advanced Configuration

- **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

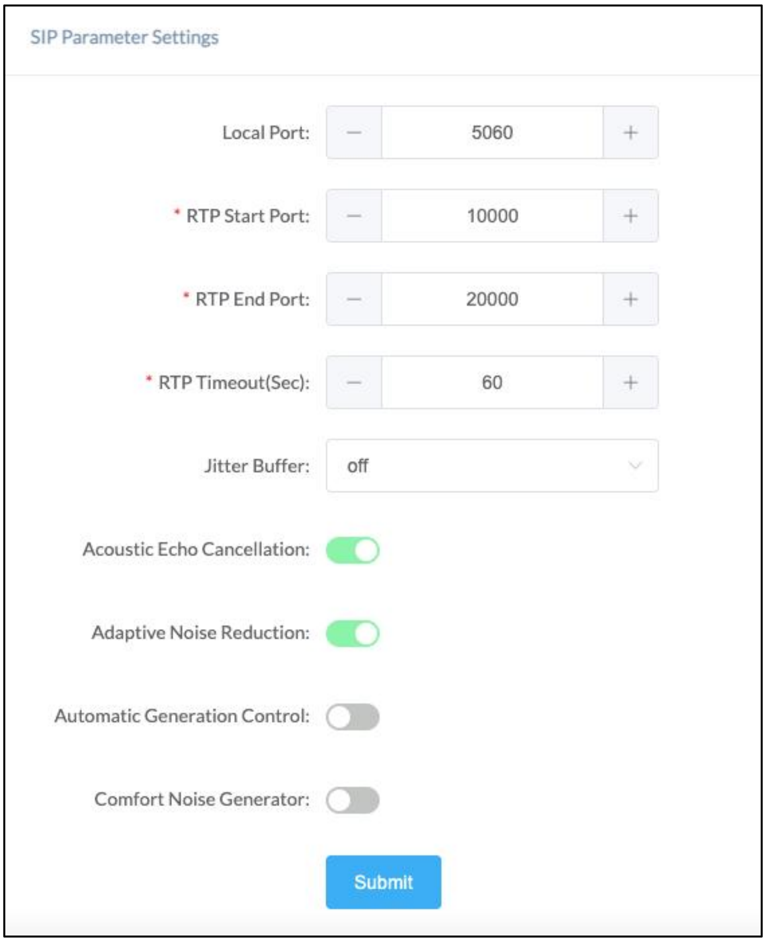
- **User Authentication:** Enable/Disable P2P authentication. If disabled, you can directly enter this device’s IP address in the target field of the peer device. If enabled, you must use the following format in the target field of the peer device: This device’s P2P User ID + IP address (e.g., 101@192.168.1.101).

- **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

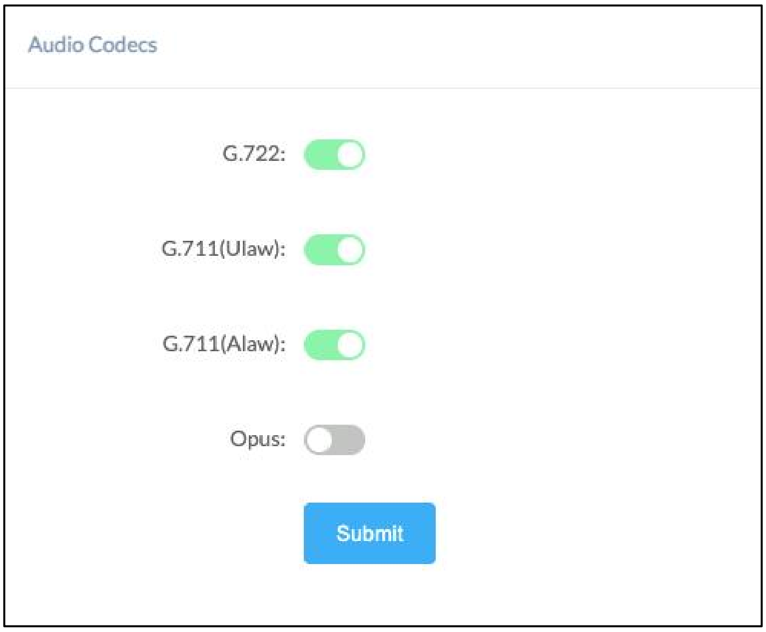


SIP Parameter Settings

- **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.
- **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 Audio Codecs

PB-S11 supports 4 audio codecs: G.722 (wideband csodec), 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. Advanced Settings

5.1 Volume Settings

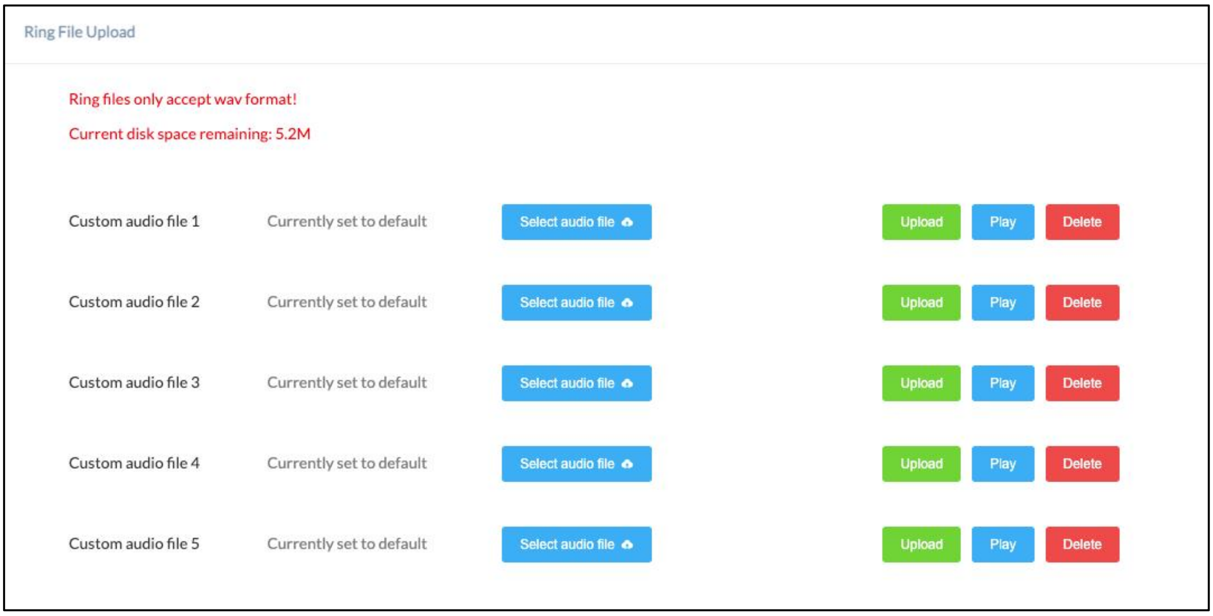
To set the microphone volume of the PB-S11, please go to **Advanced --> Volume** page to configure.

Volume Settings

- **Microphone Volume:** The default microphone volume is 7, adjustable range is 0 ~ 9.

5.2 Ring File

The Ring 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.

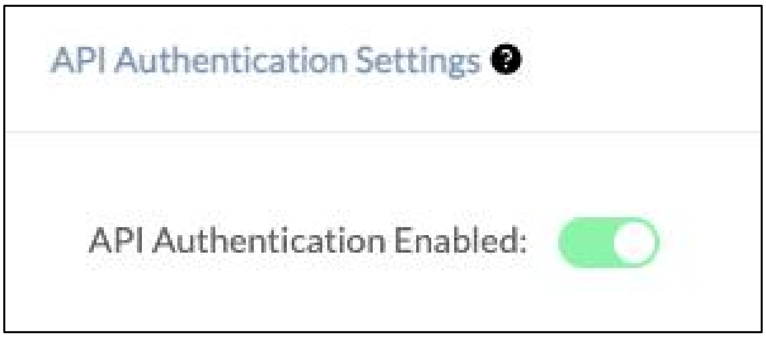


Audio Files

5.3 API Settings

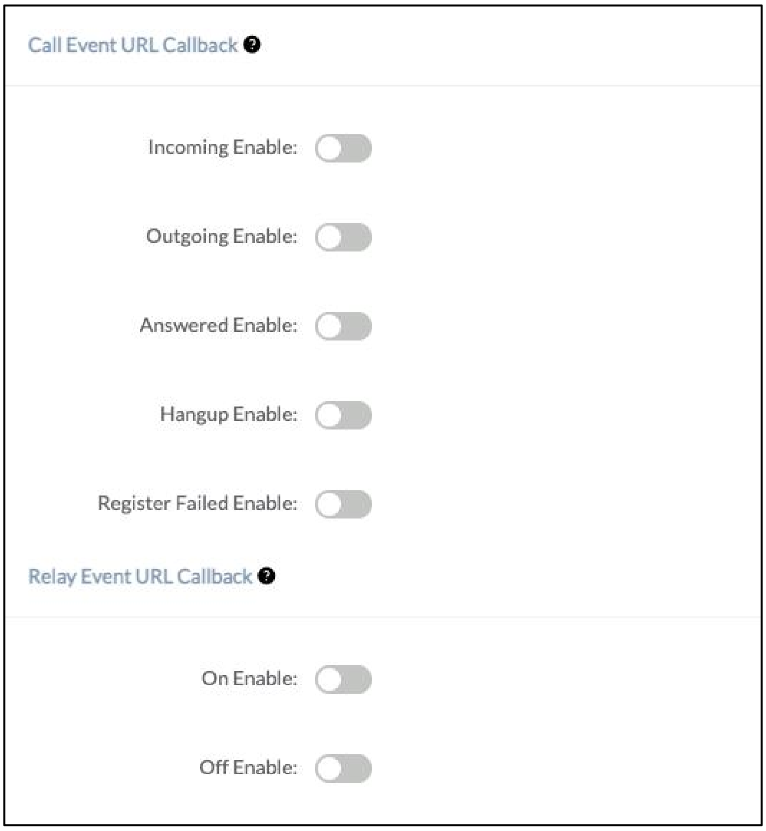
This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the changing status of the call and/or relay.

Please go to **Advanced --> 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.

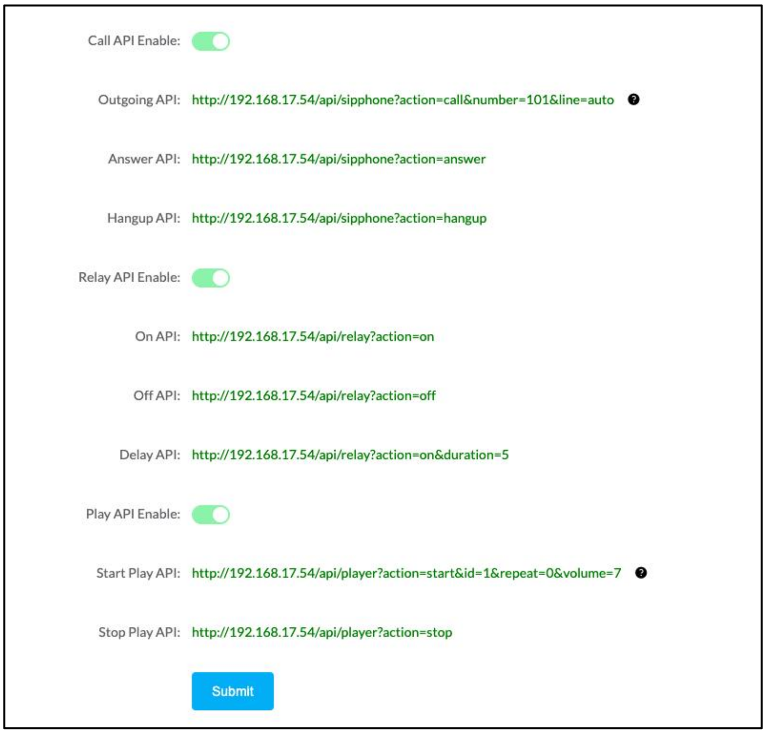
For example:

${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

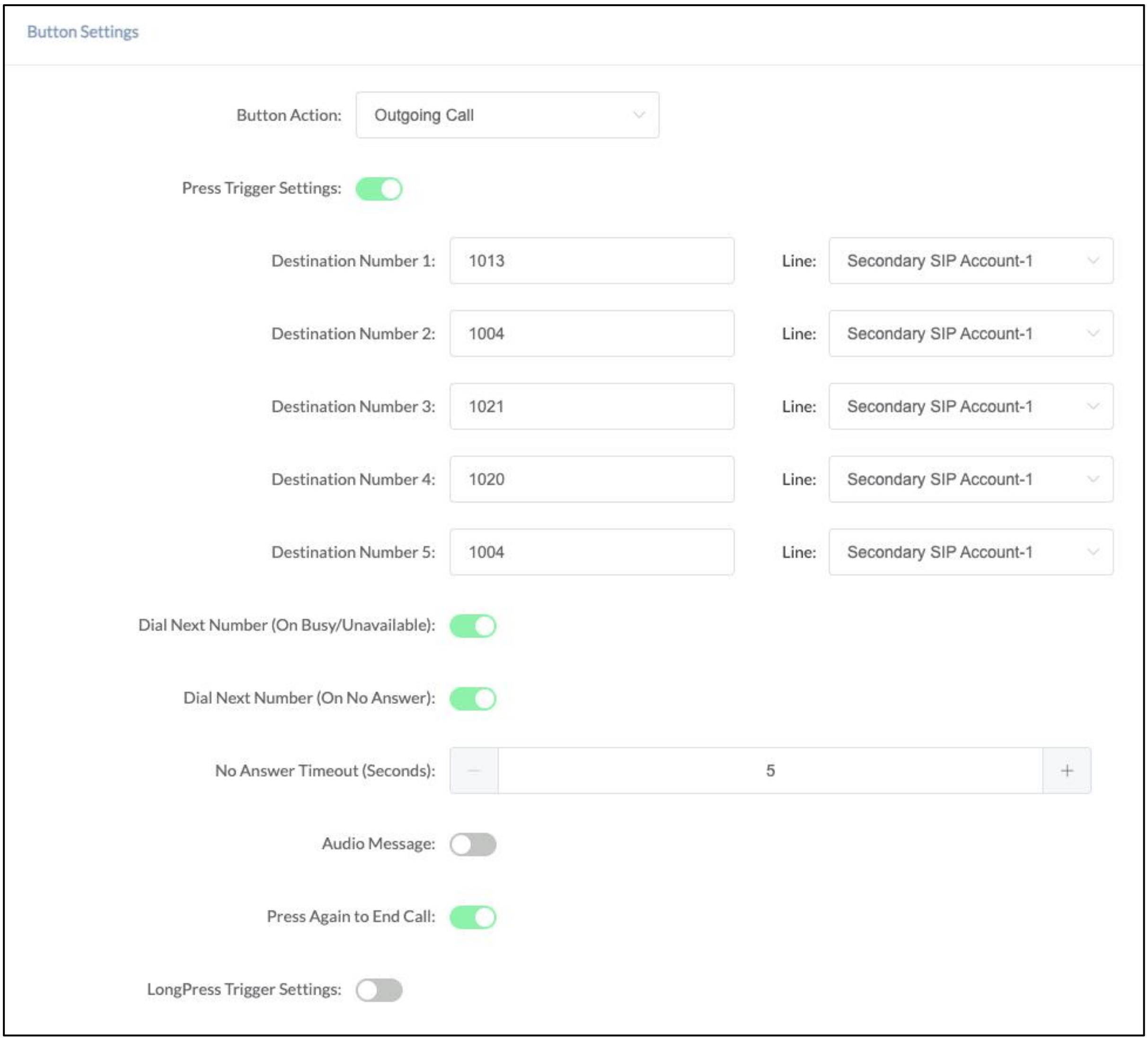
Using the API interface to realize features such as device linkage, call control, relay control, and play sound by the systems.

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.

5.4 I/O Settings

This page is used to configure the button trigger behavior and relay linkage settings of the Panic Button. It supports press-triggered and long-press-triggered actions, allowing users to configure it based on real-world scenarios. For instance, if you enable both, you can configure a short press can be used to report top-level emergencies (e.g., active shooter alerts), while a long press can be used for secondary incidents (e.g., medical emergencies). The system also supports failover logic, and voice prompts to ensure efficient and flexible emergency response.

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



Outgoing Call Settings

- **Button Action:** Defines the action to be performed when the button is pressed. You can choose between Outgoing Call (SIP call) or HTTP Request.
- **Press Trigger Settings:** Configure actions triggered by pressing the button.
- **Long Press Trigger Settings:** Configure actions triggered by long pressing the button.
- **Duration(Sec):** Configure the long pressing time in seconds.
- **Destination Number:** Configures the destination numbers to be dialed when the button is triggered. 1~5 number(s) can be defined. The system will call them in order. When multiple numbers are configured, enabling the Dial Next Number option allows the system to automatically try the next number in case of busy, no answer, or unreachable conditions, ensuring a failover mechanism.

- **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.

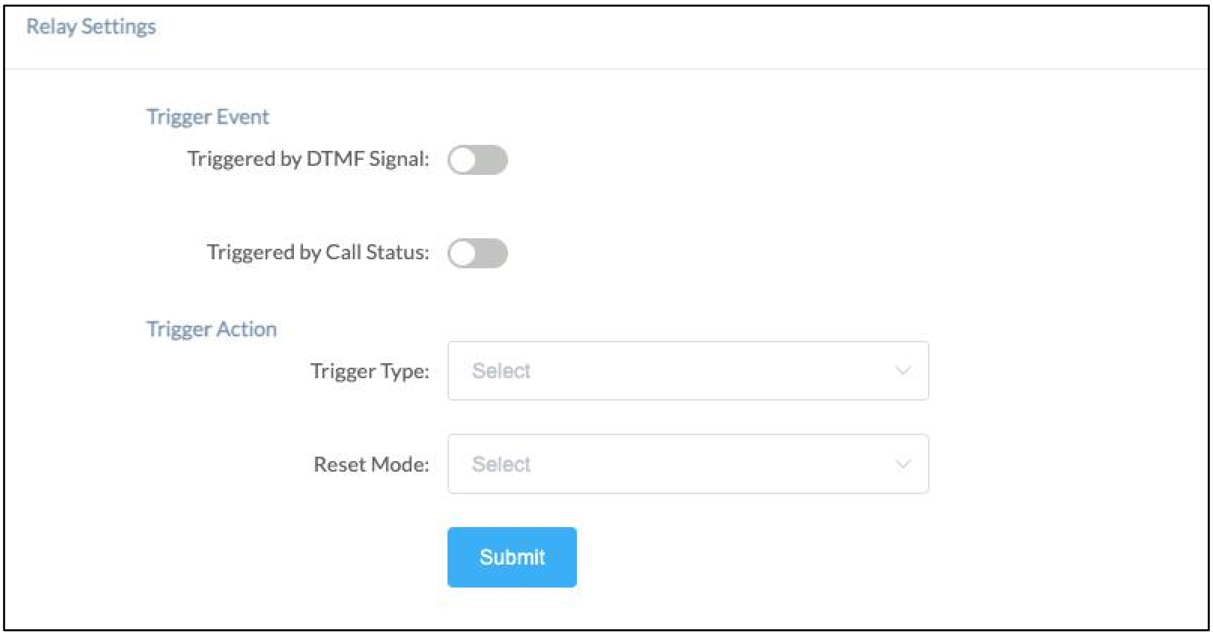
- **Dial Next Number (On Busy/Unavailable):** When the current number is busy or unreachable, proceed to the next number if enabled.
- **Dial Next Number (On No Answer):** If the current number doesn't answer within the timeout, proceed to the next number if enabled.
- **No Answer Timeout (Seconds):** Defines how long (in seconds) to wait for an answer before moving to the next number when the 'Dial Next Number (On No Answer)' option is enabled.
- **Audio Message:** A voice prompt plays before the remote side answers if enabled. Users can select a system tone, upload custom audio, or custom TTS to deliver context such as location or incident type — to help police or operator quickly understand the situation.
- **Press Again to End Call:** After the call is connected, users can end the call or conversation by pressing the button again.

HTTP Request Settings

- **HTTP URL:** Configure the API URL address triggered by linkage.

LED Settings

- **LED:** The PB-S11 offers multiple LED display modes to suit different use cases. Stealth Mode means the LED stays off at all times—whether idle, during inbound, or outbound calls—ideal for high-risk areas where discreet presence is essential. Emergency Mode - Outbound means the LED is off when idle and turns blue during outbound calls, suitable for one-way emergency reporting. Emergency Mode - Inbound & Outbound means the LED is off when idle and turns blue during both inbound and outbound calls. Assistance Mode keeps the LED blue in all function states—idle, inbound call, and outbound call—designed for non-emergency assistance situations.



Relay Control

- **Triggered by DTMF Signal:** Enable or disable relay triggering via DTMF signal (RFC2833 only).

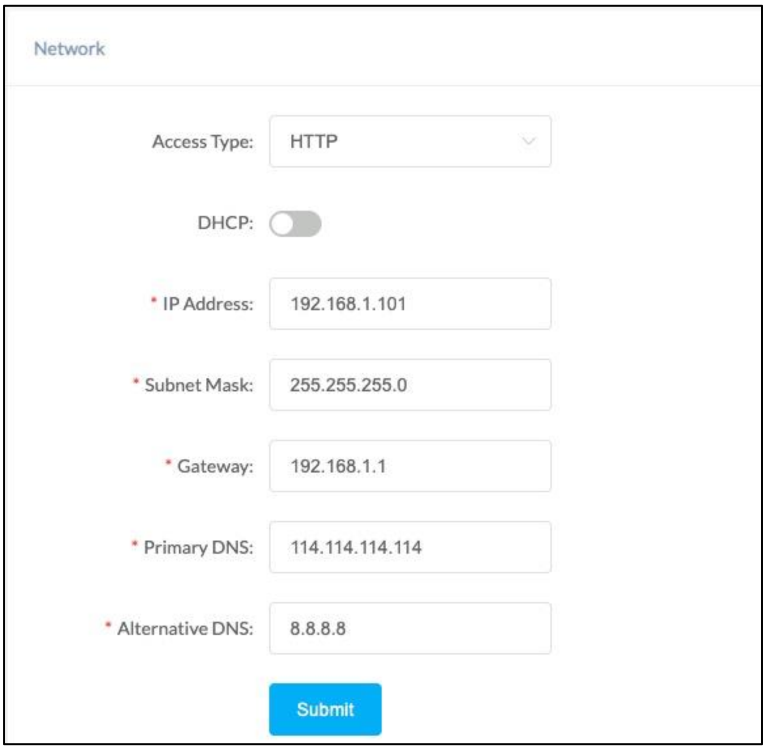
- **DTMF Code:** Specify the DTMF number to be dialed to trigger the relay.
- **Triggered by Call Status:** Enable or disable relay triggering based on call status changes.
- **Call Status:** Select the call condition that will trigger the relay: Outgoing, Incoming, Incoming/Outgoing, Answered, Hangup.
- **Trigger Type:** Defines the relay response mode when triggered by DTMF signal or call status. Available options: Door Strike Mode: Unlock / Lock (single trigger), Strobe Light Mode: Fast Flashing (continuous fast trigger), Strobe Light Mode: Slow Flashing (continuous slow trigger).
- **Reset Mode:** Configure how the relay resets after activation: Delay Reset: The relay resets after a specified duration, Hang-up Reset: The relay resets when the call ends, Answered Hang-up Reset: The relay resets after the call is answered and then hung up.
- **Duration (Sec):** This setting is only available when Delay Reset is selected. Defines how long (in seconds) the relay remains active after being triggered.

6. System Settings

6.1 Network

PB-S11 uses DHCP to dynamically obtain IP addresses by default.

To change the IP assignment from DHCP to Static IP, please go to **System--> Network** page. Turn the DHCP switch button off to show the network parameter settings.



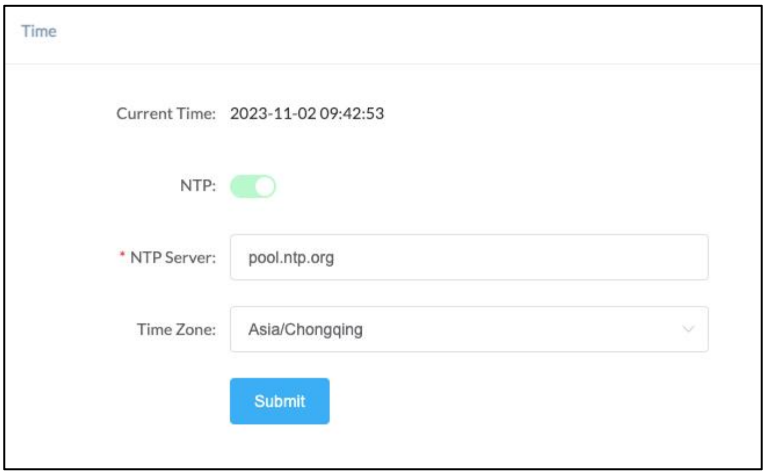
Network Configuration

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

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

6.2 Time

PB-S11 obtains the time from the network time servers using NTP.
To change the NTP settings, please go to **System --> Time** page.



Time Settings

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

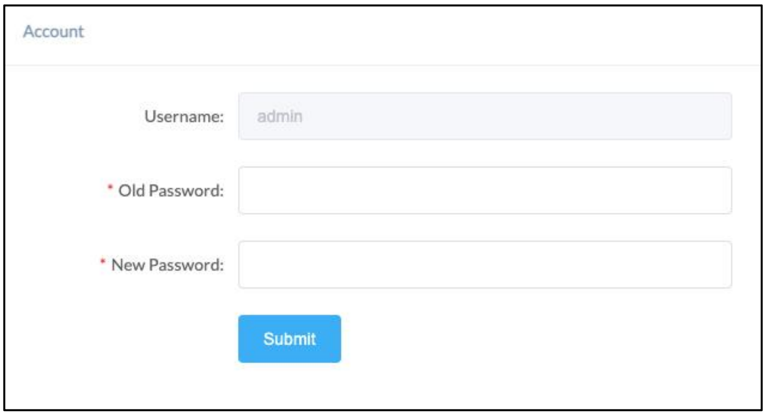
6.3 Prompt Language

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

6.4 Account

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



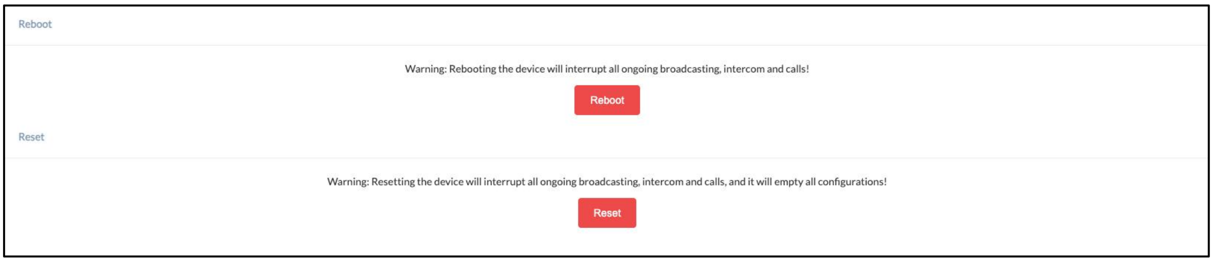
Web Password Settings

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

6.5 Reboot & Reset

PB-S11 can be rebooted and reset from the web management interface.

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



Reboot & Reset Settings

Users can restart the device without power failure on this page. The restart process takes about 10 seconds. After the restart is complete, refresh the page to log in again.

If you need to restore the factory settings of the intercom, you can reset it through this page or you can press and hold the RST button for more than 10 seconds and release it. After hearing the broadcast voice, the device will enter the state of restoration. The key will flash once. After restarting, the pop-up window disappears, and the device is restored successfully.

Note: Restoring factory settings will erase all user settings, please operate with caution!

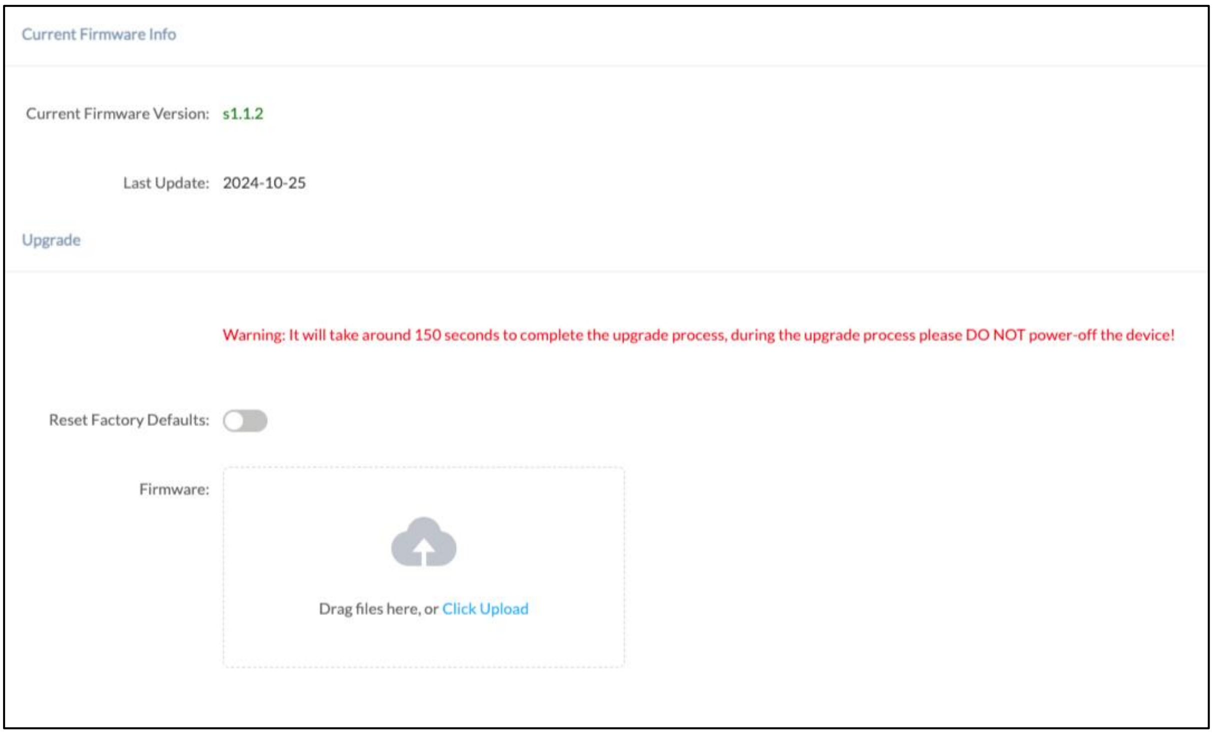
Reboot Schedule

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

7. Maintenance

7.1 Upgrade

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



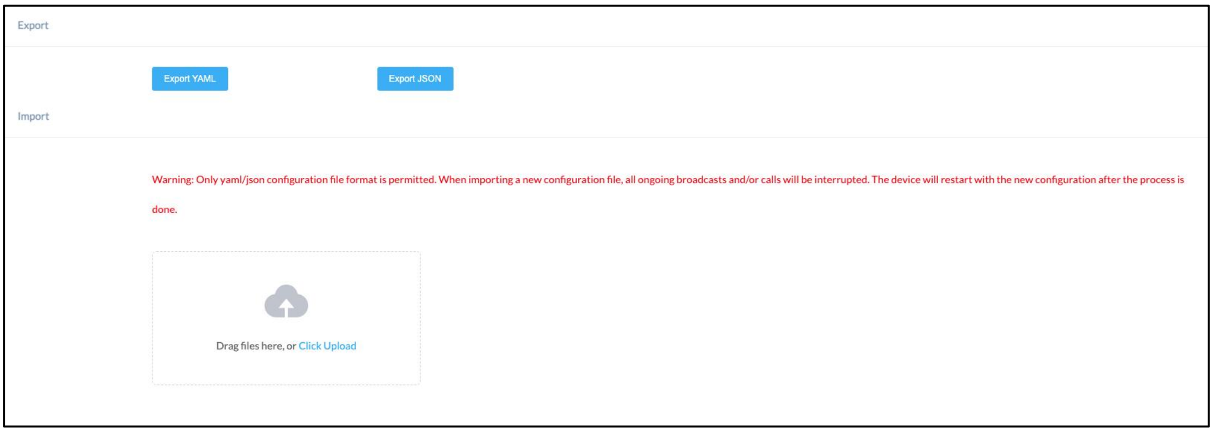
Upgrade Settings

- **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.

7.2 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

7.3 Auto Provisioning

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

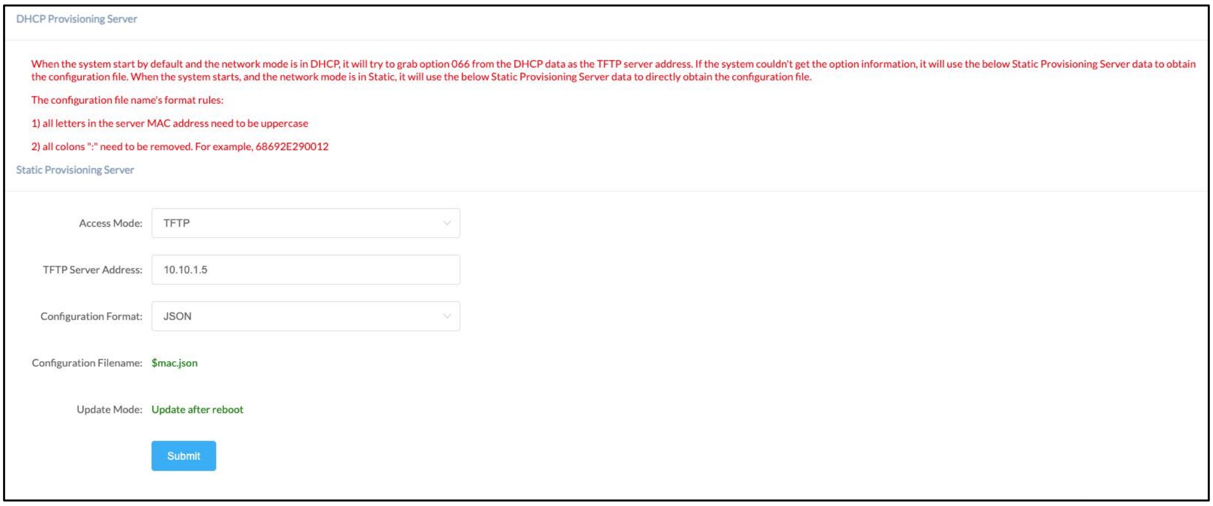
When the system starts by default and the network mode is in DHCP, it will try to grab option 066 from the DHCP data as the TFTP server address. If the system couldn't get the option information, it will use the below Static Provisioning Server data to obtain the configuration file. When the system starts,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

7.4 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

7.5 Ethernet Capture

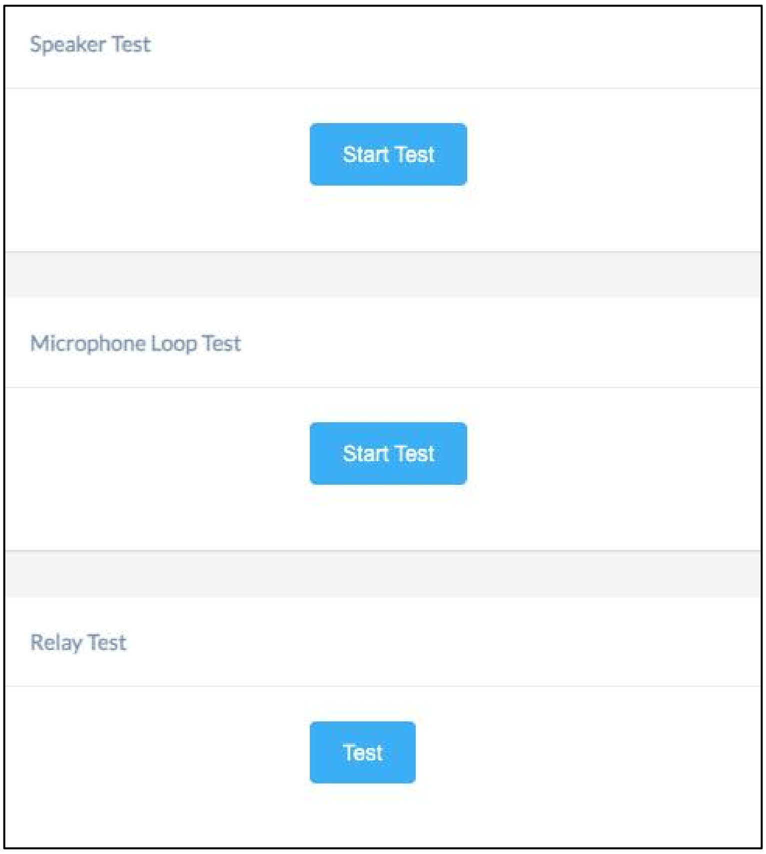
The purpose of the Ethernet capture tool is to capture Ethernet network packets and store them in a standard Wireshark-compatible packet capture '.pacp' file for immediate viewing and data analysis.

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

Ethernet Capture

7.6 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

- **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.

8. Reports

8.1 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.

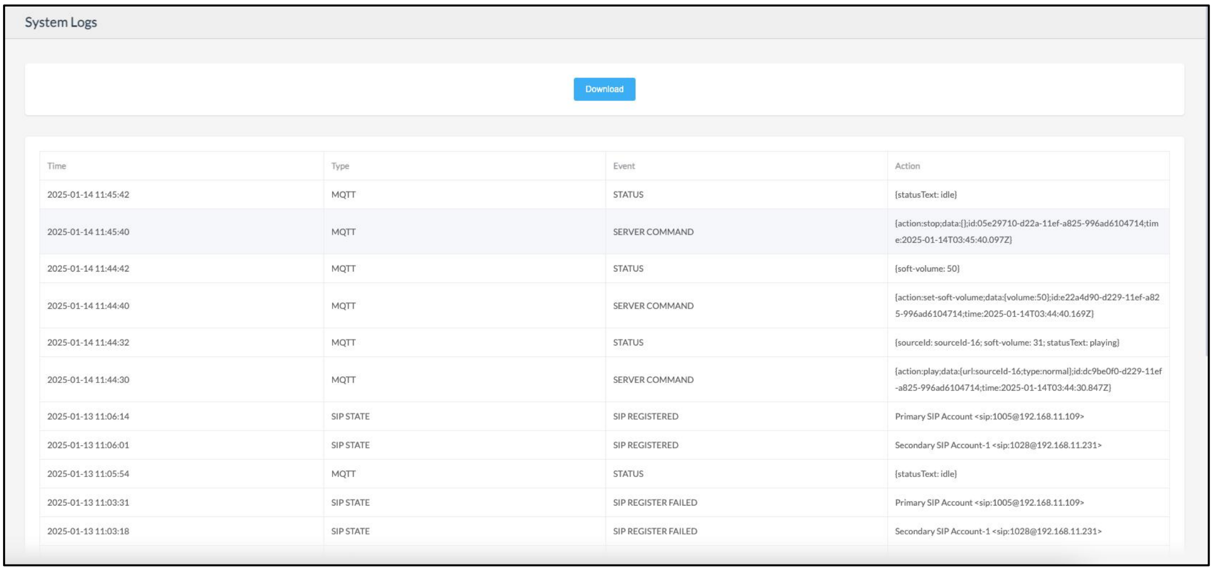
Date	Time	Account	Telephone Number	Duration	Type	Status
2024-03-15	17:18:00	sip:5011@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:47:58	sip:5015@192.168.11.62	5000	00:00:02	Inbound	Answered
2024-03-15	16:38:30	sip:5005@192.168.11.62	5000	00:00:01	Inbound	Answered
2024-03-15	16:38:16	sip:5005@192.168.11.62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

Total 4 < 1 >

Call Logs

8.2 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

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