



User Guide

IP-Speaker

www.grundig-security.com

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1. Preface

1.1 Introduction

This manual is intended to provide clear operating instructions for those who will configure and manage the Grundig Network Speaker. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the Network Speaker. The appearance of the products, functions and firmware or software upgrade may differ from this manual. GRUNDIG reserves the right to perform needed changes without prior notice.

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You may obtain the complete corresponding open-source part of a specific product from us for a period of three years after our last shipment of this product by sending an email to:

info@grundig-security.com

1.5 Safety Instructions

Make sure that you only use the power adapter that is specified in the specifications sheet of the product. If you use any other adapter or connect the power supply incorrectly, this may cause explosion, fire, electric shocks or damage the product. Do not connect several devices to one power adapter as this may cause an adapter overload and can lead to overheating and fire. Make sure that the plug of the power adapter is firmly connected to the

power socket. Do not place containers with liquids on the product. Do not place conducting items like tools, screws, coins or other metal items on the product. These may fall from the product or can cause fire or electric shocks or other physical injuries. Do not push or insert any sharp items or any objects into the device as this may cause damage to the product, fire, electric shocks and/or physical injuries. Do not block any ventilation openings, if there are any. Ensure that the product is well ventilated to prevent any over-heating. Do not subject the device to physical shock or drop the product. If the product uses batteries, please use a battery type that is recommended by the manufacturer. Improper use or replacement of the battery may result in the hazard of explosion. Do not use any accessories that are not recommended by **GRUNDIG**. Do not modify the product in any way. If the product starts to smell or smoke comes out of the device, immediately stop using the product and disconnect it from the power supply to prevent fire or electric shocks. Then contact your dealer or the nearest service centre. If the product does not work correctly, contact your dealer or nearest service centre. Never open, disassemble or alter the product yourself. **GRUNDIG** cannot accept any liability or responsibility for problems caused by attempted and unauthorized repair and maintenance.

1.6 Installation Instructions

- It is necessary to fix the device firmly if the product is installed on a wall or ceiling.
- Do not install the product on surfaces or in places that are vibrating.
- Do not install the product near radiation sources.
- Do not install the product near heat sources, like radiators or other equipment that produces heat.
- If the product is not classified by any IP class, do not install the product in very cold or hot temperatures (please refer to the working temperature specified in the spec sheet of the product), dusty, dirty or damp environment. If the product is classified by any IP class, never touch the product cover directly with your fingers, to avoid damage on the surface coating of the product cover. To clean the inside and outside of the product cover, use a soft and dry cloth. In any case, do not use alkaline detergents. The correct configuration of all passwords and other security settings is the sole responsibility of the installer and/or end-user.

1.7 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
1.1	S1.0.9.002	Initial Issue	June, 2026

2. Overview

2.1 Product Overview

Thank you for purchasing a Grundig product. Further information about the product like Data Sheets, CE Documents, etc. can also be found on the Grundig Website <https://grundig-security.com> This User Guide is a manual for Network Speakers. Please see in the list of Model Overview the applicable models.

Please read this User Guide carefully and retain it for future use.

Model Overview

This User Guide is for the following products:

GU-SI-AC0015H	Active Network Horn Speaker 15 W
GU-SI-AC0030H	Active Network Horn Speaker 30 W
GU-SI-AP0015H	Active Network Horn Speaker 15 W
GU-SI-AP0030H	Active Network Horn Speaker 30 W
GU-SI-AC0030C	Active Network Column Speaker 30 W
GU-SI-AC0060C	Active Network Column Speaker 60 W
GU-SI-AC0015B	Active Network Cabinet Speaker 15 W
GU-SI-AC0015F	Active Network Ceiling Speaker 15 W

3. Login the Device

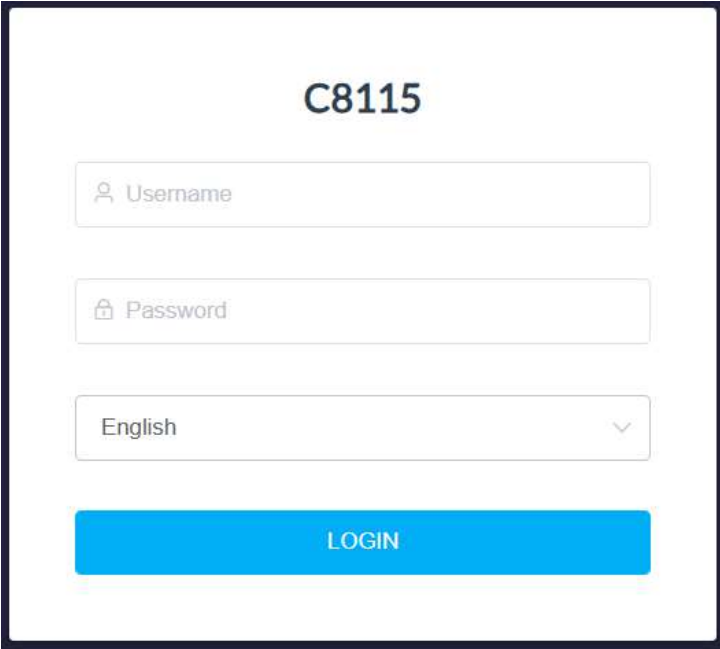
3.1 Accessing the Web GUI

The device 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 default static IP address: 192.168.1.101), you will listen to the device's IP address broadcast at first start up, and enter the IP address in the browser to access the device's Web management interface.

Default username: admin

Default password: admin

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

The image shows a login interface for a device labeled 'C8115'. It features three input fields: 'Username' with a person icon, 'Password' with a lock icon, and a language dropdown menu currently set to 'English'. Below these fields is a prominent blue 'LOGIN' button. The entire interface is enclosed in a dark rectangular border.

Login Interface

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

3.2 Device Info

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

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

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

Account	Address	Status	Mode
Primary SIP Account	1018@192.168.16.100:5060	Registered	Idle
Secondary SIP Account-1		Unconfigured	
Secondary SIP Account-2		Unconfigured	
- DEVICE INFORMATION:** A table showing device details.

Device Model	CB115
Hardware Version	Ver1.0
Software Version	s1.0.002
Uptime	0:00
Speaker Volume	7 (0-9)
Mic Volume	7 (0-9)
Device Description	CB115
- NETWORK INFORMATION:** A table showing network settings.

Mac Address	68:69:2E:0F:00:D1
Connection Mode	DHCP
IP Address	192.168.16.124
Subnet Mask	255.255.255.0
Gateway	192.168.16.1
Primary DNS	223.5.5.5
Alternative DNS	223.6.6.6

SIP STATUS

Primary SIP Account	1018@192.168.16.100:5060	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 IP Audio Center or IP PBX) address.
- **Register Status:** Display the SIP number registration status.

DEVICE INFORMATION	
Device Model	C8115
Hardware Version	Ver1.0
Software Version	s1.0.002
Uptime	0:00
Speaker Volume	7 (0-9) ↗
Mic Volume	7 (0-9) ↗
Device Description	C8115 ↗

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.
- **Microphone Volume:** Displays the current input volume of the device's microphone.
- **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:0F:00:D1
Connection Mode	DHCP
IP Address	192.168.16.124
Subnet Mask	255.255.255.0
Gateway	192.168.16.1
Primary DNS	223.5.5.5
Alternative DNS	223.6.6.6

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).

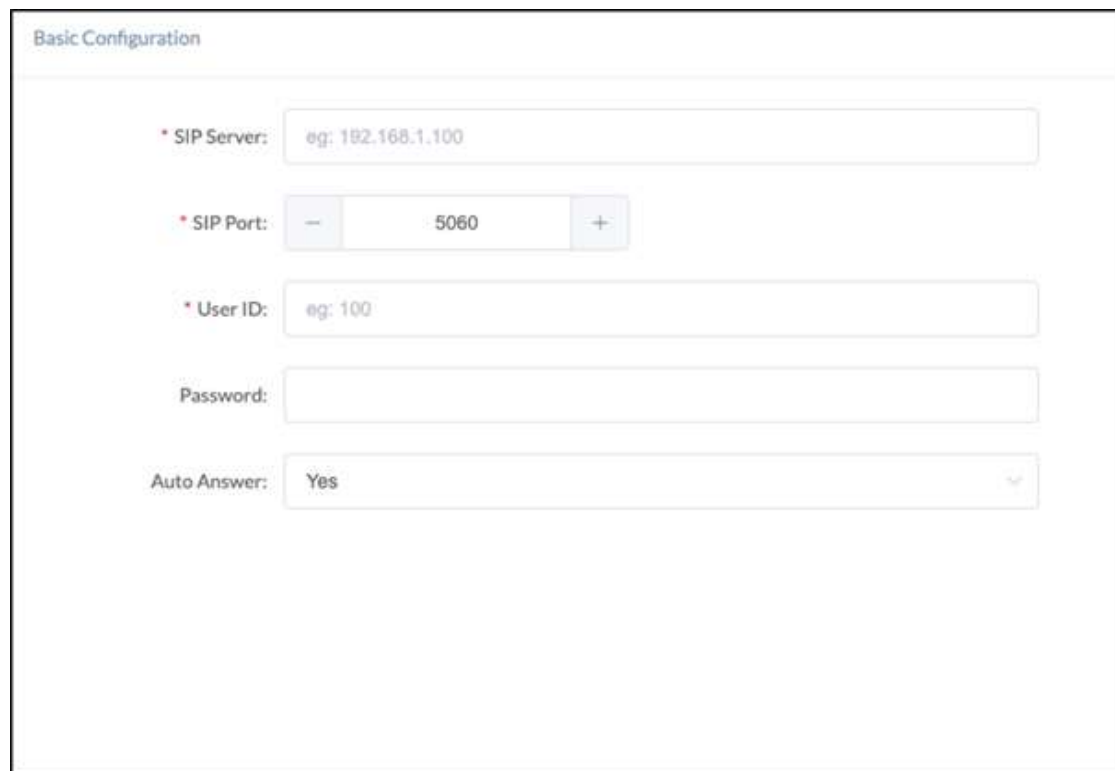
- **IP Address:** The current IP address of the device.
- **Subnet Mask:** The current subnet mask of the device.
- **Gateway:** The gateway address currently used by the device.
- **Primary DNS:** The primary domain name server address used by the device.
- **Alternative DNS:** The secondary domain 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.

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



The screenshot shows a web form titled "Basic Configuration" for SIP account settings. It contains the following fields:

- * SIP Server:** A text input field with a placeholder "eg: 192.168.1.100".
- * SIP Port:** A numeric input field with a default value of "5060" and minus/plus buttons on either side.
- * User ID:** A text input field with a placeholder "eg: 100".
- Password:** A text input field.
- Auto Answer:** A dropdown menu currently set to "Yes".

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.'

Advanced Configuration

Auth User:

Domain:

* Register Expiration(Sec):

* Transport:

NAT Mode:

Keepalive: ☒

* Keepalive Interval(Sec):

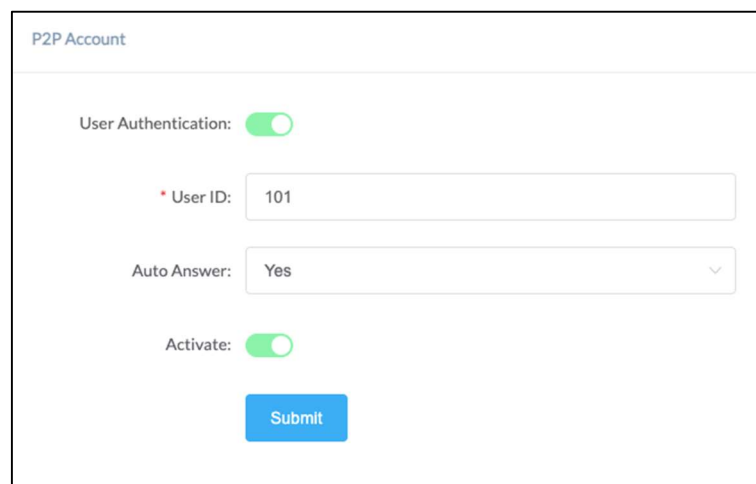
SIP Account - Advanced Configuration

- **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.



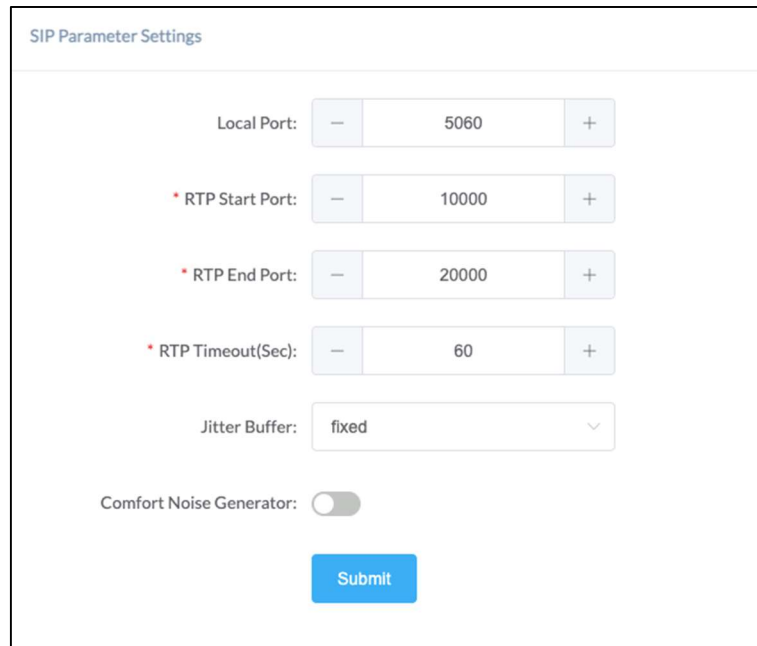
P2P Account

- **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



The screenshot shows the 'SIP Parameter Settings' page. It contains several input fields and a toggle switch. The 'Local Port' is set to 5060. The 'RTP Start Port' is set to 10000, and the 'RTP End Port' is set to 20000. The 'RTP Timeout(Sec)' is set to 60. The 'Jitter Buffer' is set to 'fixed'. The 'Comfort Noise Generator' is a toggle switch that is currently turned off. A blue 'Submit' button is at the bottom.

Parameter	Value
Local Port	5060
* RTP Start Port	10000
* RTP End Port	20000
* RTP Timeout(Sec)	60
Jitter Buffer	fixed
Comfort Noise Generator	Off

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.
- **Comfort Noise Generator:** After enabling this feature, comfortable white noise can be added during calls.

4.3.2 SIP Function Settings

SIP Function Settings

Answer Local Beep: ☐

Answer Remote Beep: ☒

Beep Sound File: Start Beep Play

Beep Volume: 90

Hangup Beep: ☐

Second Call Handling: Hangup ⓘ

Submit

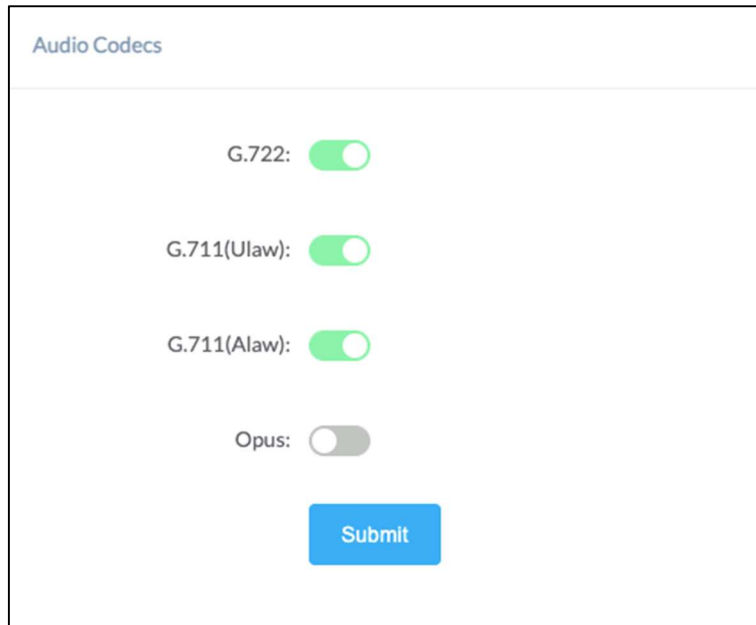
SIP Function Settings

- **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:** Set the Option 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

SC10 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: G.722, G.711(Ulaw), G.711(Alaw), and Opus. The first three toggles are turned on (green), and the Opus toggle is turned off (grey). A blue "Submit" button is located at the bottom right of the form.

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 OPA 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.
- **Enable Microphone:** Enable/Disable the microphone function.
- **Relay Mode:** Set relay control to monostable or bistable. In monostable mode, you can specify the activation duration.

- **Duration(Sec):** Set the activation duration in monostable mode.
- **Relay Type:** Choose a relay response to triggers: 'On', 'Fast Flashing', or 'Slow Flashing'.

5.2 Multicast

The multicast settings are used to configure the parameter settings of the multicast function on the SIP Safety Intercom. It can be configured to monitor up to 9 different levels of multicast addresses, the audio streams with a higher priority will interrupt the playback of the lower priority audio streams.

Please go to **Functions** ---> **Multicast** page to enable the multicast feature.

Multicast

Enable Multicast: ☒

Network Caching(ms):

Port range from 2000-65535
Priority from highest 9 to lowest 1
An audio stream with higher priority will supersede the lower one

Priority	Multicast Address	Multicast Port	Name	Relay Control
1	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.
- **Relay Control:** Options to choose from are 'Disabled', 'On', 'Fast Flashing', 'Slow Flashing'.

6. Advanced Settings

6.1 Volume Settings

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

You can adjust the main volume as well as individual volume adjustments for each application.

Speaker Volume

* Master volume: — 7 +

Microphone Volume: — 7 +

Key beep: ☒

Music Auto Resumes: ☒

Play IP on startup: ☒ ⓘ

Speaker Volume Settings

- **Master Volume:** Set the speaker master volume. The default volume is 7 and the adjustable range is 0~9.
Note: It is recommended the speaker volume level setting not exceed 7 under POE power supply mode, otherwise it may cause the device to restart!
- **Microphone Volume:** Set the microphone volume. The default volume is 7 and the adjustable range is 0~9, applicable to C8115 and W8115.
- **Key Beep:** Enable/Disable the beep sound from the key button.
- **Music Auto Resumes:** When the device restarts or reconnects to the network, the previous music tasks will be automatically restored.
- **Play IP on Startup:** When the device starts, it automatically broadcasts its IP address once.

App Volume

* SIP Volume:

80

+

ONVIF Volume:

80

+

BROADCAST Volume:

48

+

MULTICAST Volume 1:

80

+

MULTICAST Volume 2:

50

+

MULTICAST Volume 3:

80

+

MULTICAST Volume 4:

80

+

MULTICAST Volume 5:

80

+

MULTICAST Volume 6:

80

+

MULTICAST Volume 7:

80

+

MULTICAST Volume 8:

80

+

MULTICAST Volume 9:

80

+

Submit

App Volume Settings

- **SIP Volume:** Set the specific volume for SIP application.
- **ONVIF Volume:** Set the specific volume for ONVIF application.
- **Broadcast Volume:** Set the specific volume for broadcast application.
- **Multicast Volume 1:** Set the specific volume for multicast 1 application.
- **Multicast Volume 2:** Set the specific volume for multicast 2 application.
- **Multicast Volume 3:** Set the specific volume for multicast 3 application.
- **Multicast Volume 4:** Set the specific volume for multicast 4 application.
- **Multicast Volume 5:** Set the specific volume for multicast 5 application.
- **Multicast Volume 6:** Set the specific volume for multicast 6 application.
- **Multicast Volume 7:** Set the specific volume for multicast 7 application.
- **Multicast Volume 8:** Set the specific volume for multicast 8 application.
- **Multicast Volume 9:** Set the specific volume for multicast 9 application.

6.2 Audio Priority Settings

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

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

Audio Priority

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

Priority	Application name	Operation
1	SIP	⌵
2	ONVIF	⌴
3	MULTICAST	⌴
4	BROADCAST	⌴

Submit

Audio Priority Settings

6.3 Audio Files

The Audio files section allows users to self-upload up to 5M of audio files to the endpoint and use it as a ringtone or Play API audio file. Please click on the 'Select audio file' button to select and upload the local audio file, then click on the 'upload' button to upload it. Click on the 'play' to test and play the audio file and the 'delete' button for deleting the audio file.

Please go to **Advanced** ---> **Audio Files** to manage the audio files.

Audio Files Upload

Audio files only accept wav format!

Current disk space remaining: 5.1M

Custom audio file 1	Currently set to default	Select audio file 	Upload	Play	Delete
Custom audio file 2	Currently set to default	Select audio file 	Upload	Play	Delete
Custom audio file 3	Currently set to default	Select audio file 	Upload	Play	Delete
Custom audio file 4	Currently set to default	Select audio file 	Upload	Play	Delete
Custom audio file 5	Currently set to default	Select audio file 	Upload	Play	Delete

Audio Files

6.4 API Settings

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

Please go to **Advanced --> API Settings** page to enable API settings.

API Authentication Settings 

API Authentication Enabled: 

API Authentication Settings

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

The screenshot shows a configuration window with two main sections. 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

- **Call 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

- **Relay Event URL Callback**

When the relay status changes, it will trigger an HTTP GET request to call a URL address.

Within the URL address, you may use variables to identify some current information.

For example:

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

Call API Enable: ☒

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

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

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

Relay API Enable: ☒

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

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

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

Play API Enable: ☒

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

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

[Submit](#)

API Settings

- **API Settings**

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

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

6.5 I/O Settings

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

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

Input Settings

Input Type: Key Signal

Key Event Action: Outgoing Call

Destination: 5000 Line: Auto

Press Again to End Call: ☒

Input Settings

Input Type: Key Signal

Key Event Action: HTTP Request

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

Input Settings

Input Type: Key Signal

Key Event Action: Play Audio

Audio File: Toy Repeat: 1

Key Signal Input Settings

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

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

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

Input Settings

Input Type: Switch Signal

Closed Event Action: Outgoing Call

Destination: 5000
Line: Auto

Open Event Action: Stop Audio

Input Settings

Input Type: Switch Signal

Closed Event Action: Hangup

Open Event Action: Stop Audio

Input Settings

Input Type: Switch Signal

Closed Event Action: HTTP Request

* HTTP URL:

Open Event Action: Stop Audio

Input Settings

Input Type: Switch Signal

Closed Event Action: Play Audio

Audio File: Sweet

Repeat: - 0 +

Open Event Action: Stop Audio

Input Settings

Input Type: Switch Signal

Closed Event Action: Stop Audio

Open Event Action: Disabled

Switch Signal Input Settings

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

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

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

Trigger Setting

Broadcast music trigger: Disabled

Broadcast alarm trigger: Fast Flashing

Trigger by Input Signal:

Trigger by DTMF Signal:

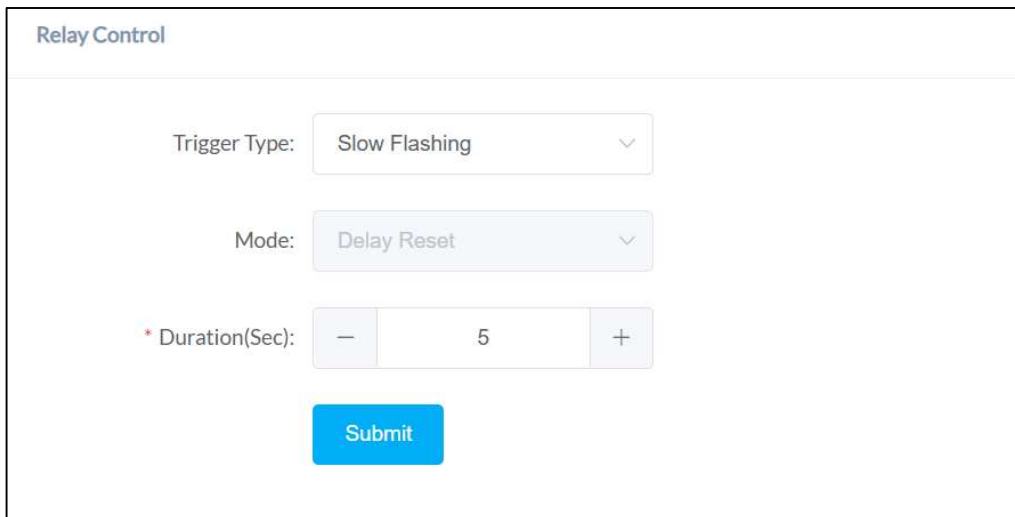
* DTMF: 1*

Trigger by Call Status:

Event: Incoming

Trigger Settings

- **Broadcast Music Trigger:** Disabled/On/Fast Flashing/Slow Flashing, enable this option will trigger the relay when there is broadcast music on.
- **Broadcast Alarm Trigger:** Disabled/On/Fast Flashing/Slow Flashing, enable this option will trigger the relay when there is a broadcast alarm on.
- **Trigger by Input Signal:** Enable/Disable, enable this option if need to use the input signal to trigger.
- **Trigger by DTMF Signal:** Enable/Disable, enable this option when need to use DTMF signal to trigger (only RFC2833 supported).
- **DTMF:** This setting represents the number to dial to trigger DTMF.
- **Trigger by Call Status:** Enable/Disable, enable this option will change the call status when triggered.
- **Event:** Set the corresponding call state, you can choose **【Outgoing】** , **【Incoming】** , **【Incoming/Outgoing】** , **【Answered】** and **【Hangup】** .



The screenshot shows a web interface titled "Relay Control". It contains three main settings:

- Trigger Type:** A dropdown menu currently set to "Slow Flashing".
- Mode:** A dropdown menu currently set to "Delay Reset".
- * Duration(Sec):** A numeric input field with a value of "5", flanked by minus and plus buttons for adjustment.

At the bottom of the form is a blue "Submit" button.

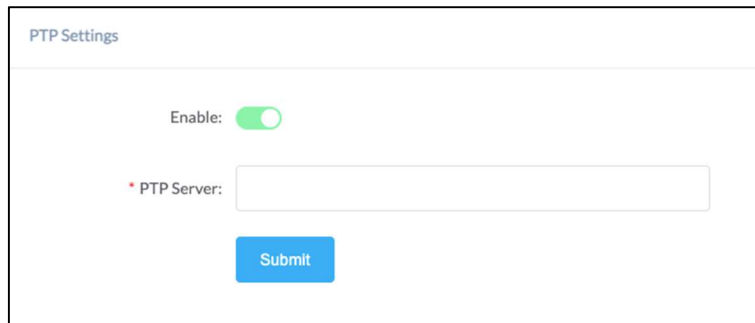
Relay Control

- **Trigger Type:** This setting represents the responses by the triggers, there are 'On', 'Fast Flashing', and 'Slow Flashing' options to choose from.
- **Mode:** This setting represents the reset mode after the trigger is responded, there is only 'Delay Reset' options to choose from.

- **Duration (Sec):** This setting is only available if the reply control mode is on Delay Reset, it represents the time duration when the configure interface status changed.

6.6 PTP Settings

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



PTP Settings

Enable: ☒

* PTP Server:

Submit

PTP Setting

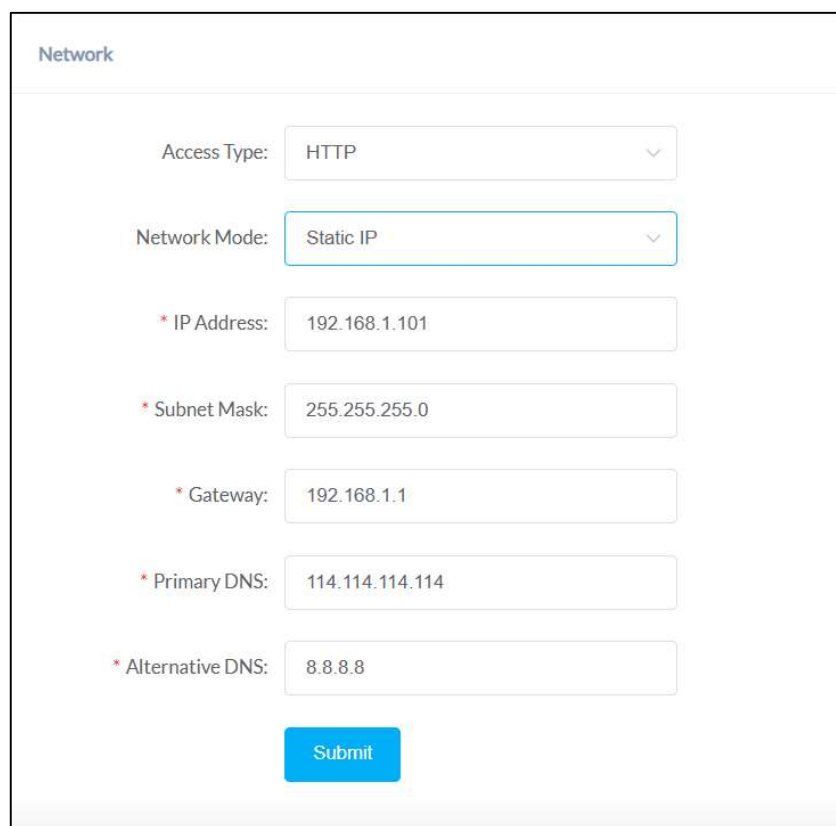
7. System Settings

7.1 Network

The device 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.

Select Static IP to show the network parameter settings.



Network

Access Type: HTTP

Network Mode: Static IP

* IP Address: 192.168.1.101

* Subnet Mask: 255.255.255.0

* Gateway: 192.168.1.1

* Primary DNS: 114.114.114.114

* Alternative DNS: 8.8.8.8

Submit

Network Configuration

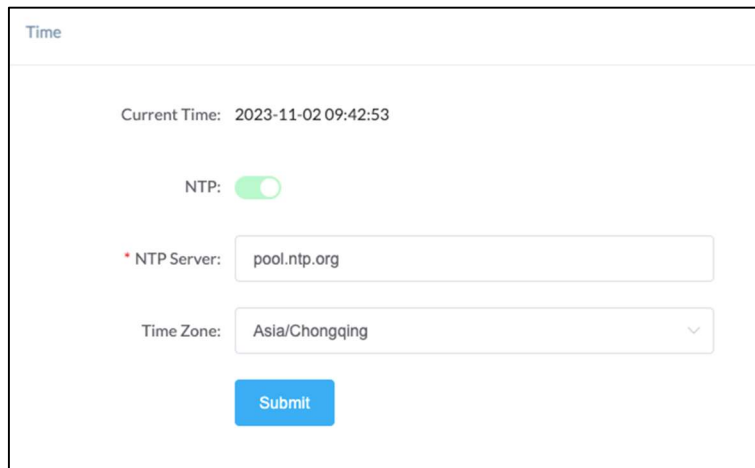
- **Access Type:** Specify the access method of the website, which currently supports HTTP、HTTPS and HTTP& HTTPS.
- **Network Mode:** Select DHCP or Static IP
- **IP Address:** Enter a vacant IP address within your LAN.
- **Subnet Mask:** Enter the subnet mask of your LAN.
- **Gateway:** Enter the default 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

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

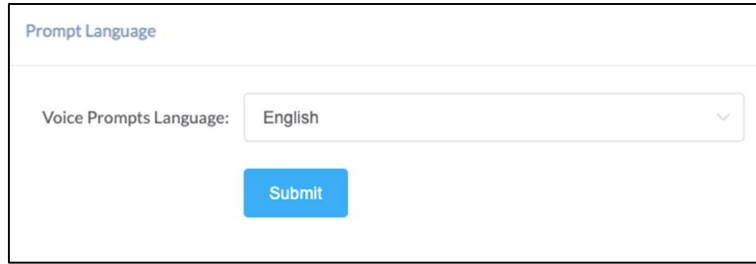
Time Settings

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

7.3 Prompt Language

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

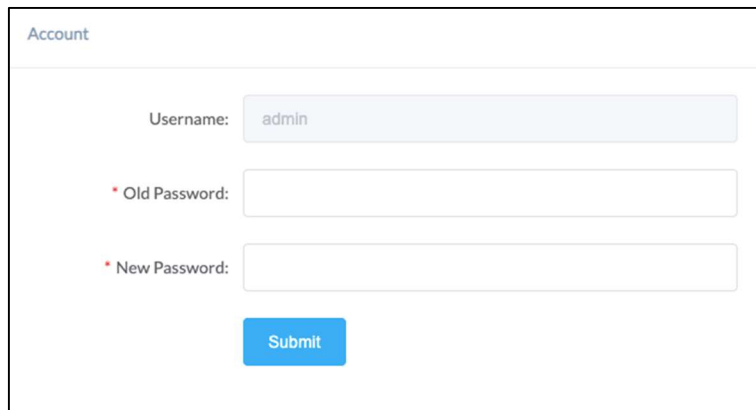
Please go to **System --> Prompt Language** page to set a voice prompt language.



Prompt Language

7.4 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.

7.5 Reboot & Reset

The device can be rebooted and reset from the web management interface.

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

Reboot

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

Reboot

Reset

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

Reset

Reboot & Reset Settings

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

If you need to restore the factory settings of the device, you can reset it through this page, after restarting, the current IP address of the device will be automatically announced by voice prompts. Users can log in to the device's Web management interface with this IP address.

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

Reboot Schedule

Enable: ☐

Submit

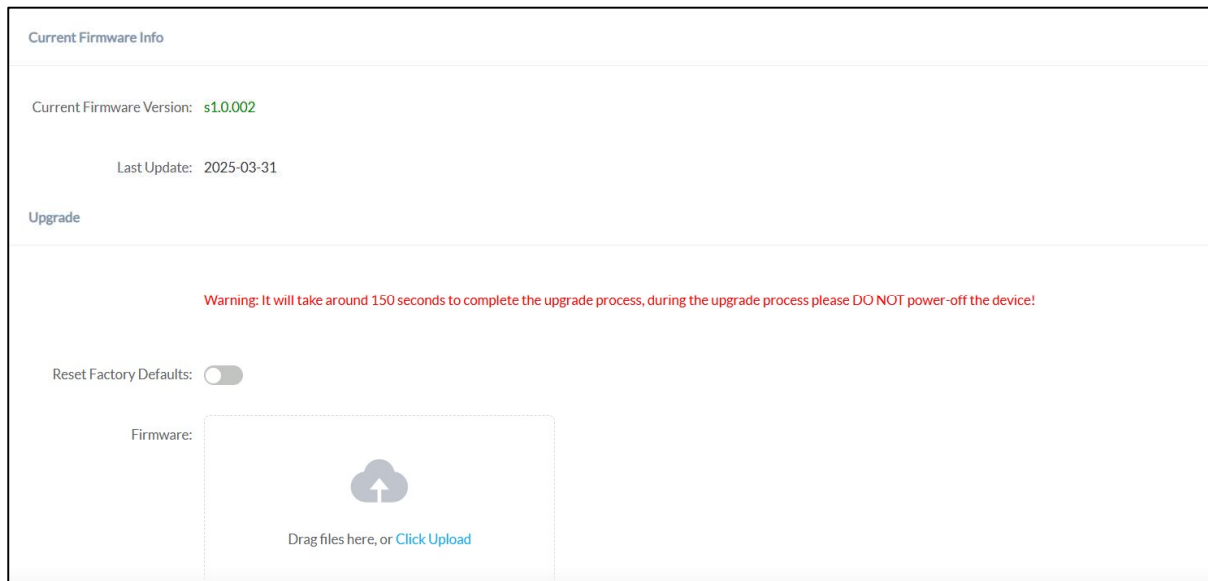
Reboot Schedule

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 **Maintenance --> Upgrade** page.



The screenshot shows the 'Upgrade' page of a Grundig device's maintenance interface. At the top, under 'Current Firmware Info', it displays 'Current Firmware Version: s1.0.002' and 'Last Update: 2025-03-31'. Below this is a red warning message: 'Warning: It will take around 150 seconds to complete the upgrade process, during the upgrade process please DO NOT power-off the device!'. There is a 'Reset Factory Defaults' toggle switch which is currently turned off. At the bottom, there is a 'Firmware:' label next to a large dashed box containing a cloud upload icon and the text 'Drag files here, or Click Upload'.

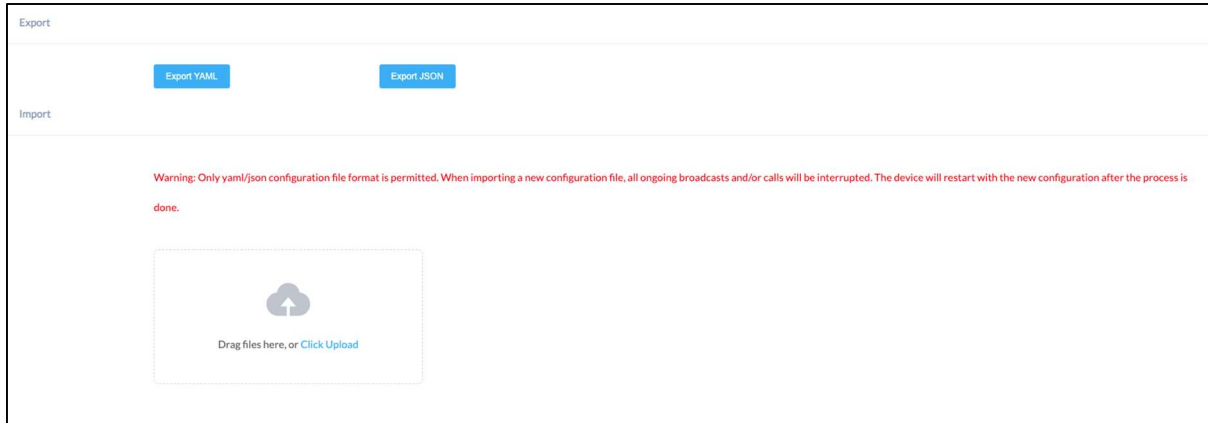
Upgrade Settings

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

8.2 Import/Export

This page is used to import and export the current configuration of the device, and you may use this configuration file to backup and/or recover. Both YAML and JSON formats are supported.

Please go to **Maintenance --> Import/Export** page to backup or recover.



Export

Export YAML Export JSON

Import

Warning: Only yaml/json configuration file format is permitted. When importing a new configuration file, all ongoing broadcasts and/or calls will be interrupted. The device will restart with the new configuration after the process is done.

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

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 Server Address:

Configuration Format:

Configuration Filename: `$mac.json`

Update Mode: `Update after reboot`

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

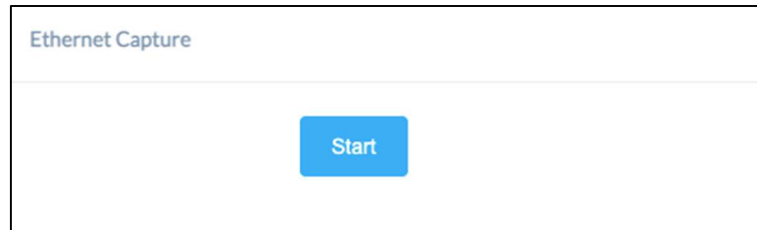
* IP/Domain:

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.

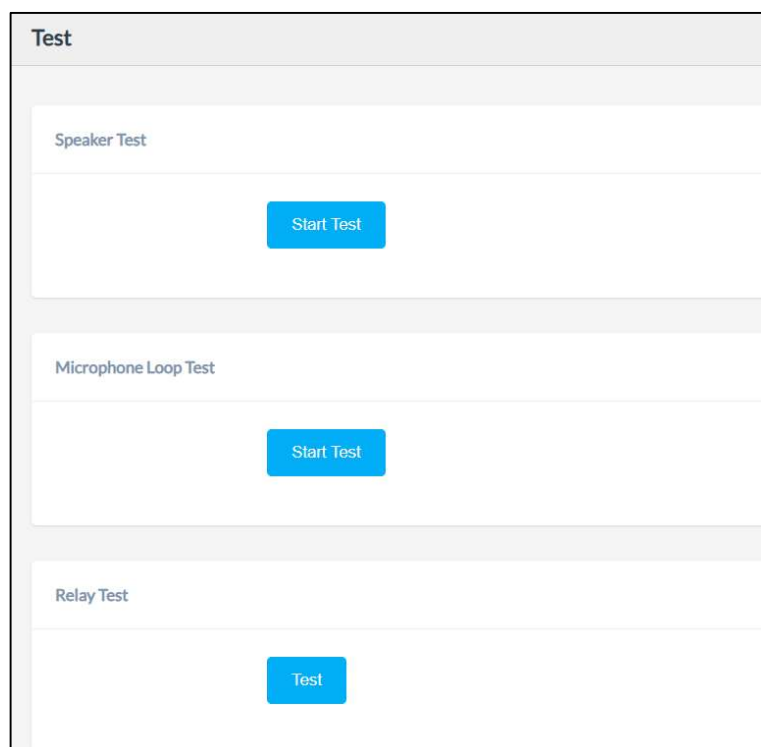


Ethernet Capture

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, insert microphone device, the sound detection is normal (some models do not have this option) .

- **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:e22a4690-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:[uri:sourceId-16;type:normal];id:dc9be090-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>

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