



Ei Series Safety Intercom User Guide



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1. Preface

1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and manage the Ei Series Safety Intercom. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the Ei Series Safety Intercom.

1.2 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
1.0.4	1.0.4	Updated operating instructions for software version v1.0.4	Dec,2023

2. Overview

2.1 Product Overview

Ei Series Safety Intercom is available in four models designed to address the needs and requirements of various applications and installation types.

All models deliver full-duplex communication and secure access control for your VoIP system. Ideal for any setting such as commercial facilities, schools and universities, medical facilities, warehouses, and retail establishments.

2.2 Product Specifications

Ei Series Safety Intercom Specifications	
Drive Unit	Φ40mm full frequency drive unit
Sensitivity	96 ± 3dB/1M/1W
Distortion	<1%
Rated Power	8Ω 5W
Frequency Range	100Hz – 20KHz
Coverage Pattern	90°H 90°V 30m ²
Amplifier	5W Single Channel D-class



3. Login the Device

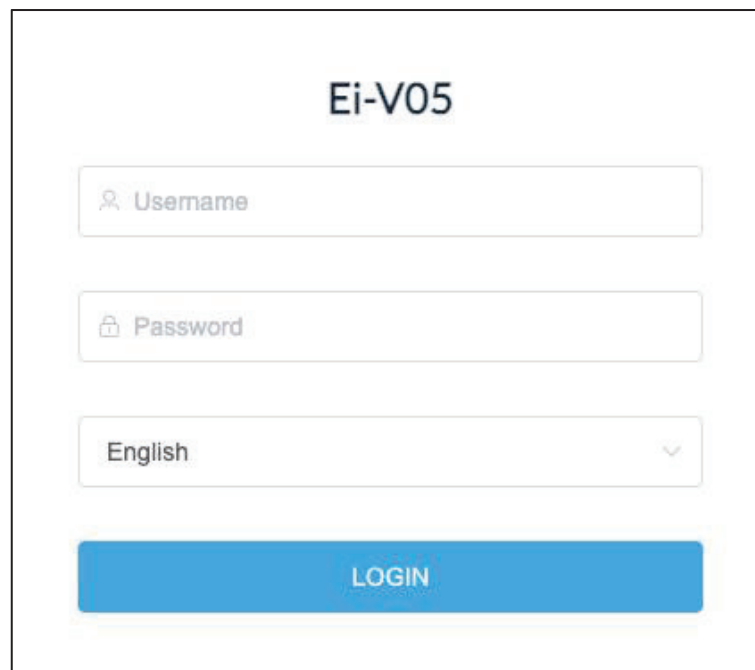
3.1 Accessing the Web GUI

Ei intercom 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), or press and hold the press button for 5 seconds and then release it to listen to the device's IP address broadcast, 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.

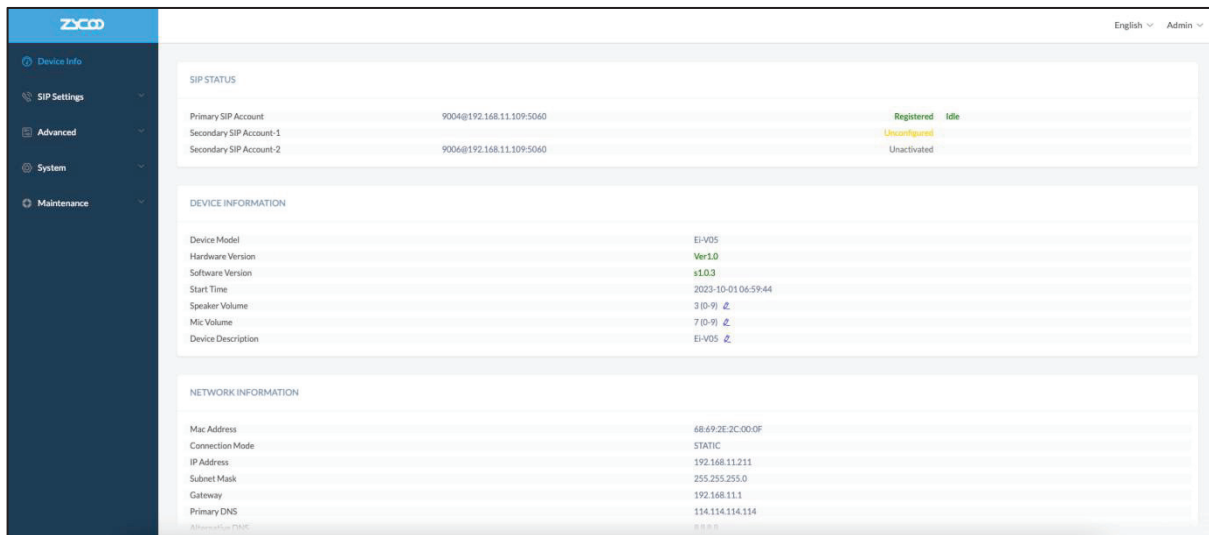


Login Interface

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

3.2 Device Info

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	9004@192.168.11.109:5060	Registered	Idle
Secondary SIP Account-1		Unconfigured	
Secondary SIP Account-2	9006@192.168.11.109:5060	Unactivated	

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	Ei-V05
Hardware Version	Ver1.0
Software Version	s1.0.3
Start Time	2023-10-01 06:59:44
Speaker Volume	3 (0-9) ↕
Mic Volume	7 (0-9) ↕
Device Description	Ei-V05 ↕

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.
- **Start Time:** Displays the last time the device was started up.
- **Speaker Volume:** Displays the current volume of the device.
- **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:2C:00:0F
Connection Mode	STATIC
IP Address	192.168.11.211
Subnet Mask	255.255.255.0
Gateway	192.168.11.1
Primary DNS	114.114.114.114
Alternative DNS	8.8.8.8

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.

Primary SIP Account

* SIP Server: 192.168.11.62

* SIP Port: - 5060 +

* User ID: 5001

Auth User: 5001

Domain: 192.168.11.62

Password: *****

* Register Expiration(Sec): - 180 +

* Transport: UDP

Auto Answer: Yes

NAT Mode: Disabled

Enable Integration with
ZYCOO IP Audio Center:

Activate:

Submit

Primary SIP Account

The screenshot shows a web form titled "Secondary SIP Account-1". It contains the following fields and controls:

- * SIP Server:** Text input field containing "192.168.11.83".
- * SIP Port:** Spinners with a minus sign on the left, a text input field containing "5060", and a plus sign on the right.
- * User ID:** Text input field containing "1007".
- Auth User:** Text input field containing "1007".
- Domain:** Text input field containing "192.168.11.83".
- Password:** Password input field with masked characters "*****" and a visibility toggle icon.
- * Register Expiration(Sec):** Spinners with a minus sign on the left, a text input field containing "180", and a plus sign on the right.
- * Transport:** Dropdown menu with "UDP" selected.
- Auto Answer:** Dropdown menu with "Yes" selected.
- NAT Mode:** Dropdown menu with "Disabled" selected.
- Activate:** Toggle switch that is currently turned on (green).
- Submit:** A blue button at the bottom.

Secondary SIP 1 Account

The screenshot shows a web form titled "Secondary SIP Account-2". It contains the following fields and controls:

- * SIP Server:** Text input field containing "eg: 192.168.1.100".
- * SIP Port:** Spinners with a minus sign on the left, a text input field containing "5060", and a plus sign on the right.
- * User ID:** Text input field containing "eg: 100".
- Auth User:** Text input field containing "eg: 100".
- Domain:** Text input field containing "eg: pbx.com".
- Password:** Empty password input field.
- * Register Expiration(Sec):** Spinners with a minus sign on the left, a text input field containing "180", and a plus sign on the right.
- * Transport:** Dropdown menu with "UDP" selected.
- Auto Answer:** Dropdown menu with "Yes" selected.
- NAT Mode:** Dropdown menu with "Disabled" selected.
- Activate:** Toggle switch that is currently turned off (grey).
- Submit:** A blue button at the bottom.

Secondary SIP 2 Account

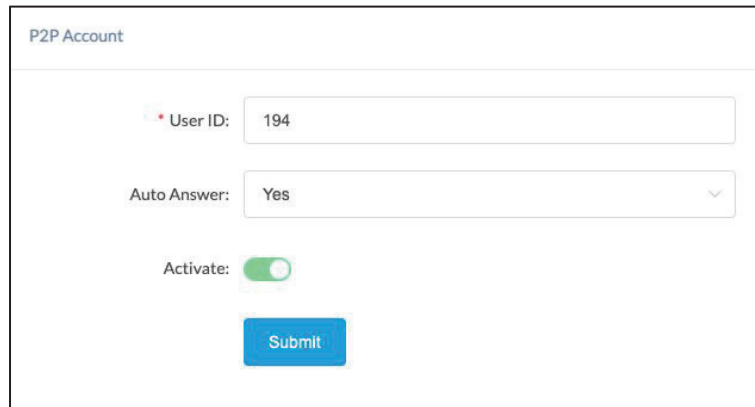
- **SIP Server:** Enter the IP address or domain name of the SIP server.

- **SIP Port:** Default SIP port is 5060. If the SIP server uses another port number as the SIP port, please modify this setting.
- **User ID:** The SIP account number provided by the SIP server.
- **Auth User:** Enter the authorized SIP account's username.
- **Domain:** Enter the SIP Domain.
- **Password:** Authorized SIP account password.
- **Register Expiration (sec):** SIP register expiration time, the default expiration time is 180 seconds.
- **Transport:** Set up the transport protocol, there are UDP, TCP, TLS options to choose.
- **Auto Answer:** Yes/No/Answer Delay, default in the Yes option.
- **NAT Mode:** Select the NAT mode and fill out the corresponding data. STUN, TURN, and ICE modes are supported.
- **Enable Integration with ZYCOO IP Audio Center:** This option is disabled by default. If you need to connect and use it with ZYCOO IP Audio Center, please enable this option. Only the main SIP account has this option.
- **Activate:** Once enabled, the account will be activated and registered to the SIP server.

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. After configuring the P2P account, it can be used with the Outgoing Call feature in **Basic Settings --> I/O Settings**, or use the Outgoing API in **Basic Settings ---> API Settings** to make a P2P call.



The screenshot shows a web form titled "P2P Account". It contains three input fields: "User ID" with the value "194", "Auto Answer" with a dropdown menu set to "Yes", and "Activate" with a green toggle switch. A blue "Submit" button is located at the bottom of the form.

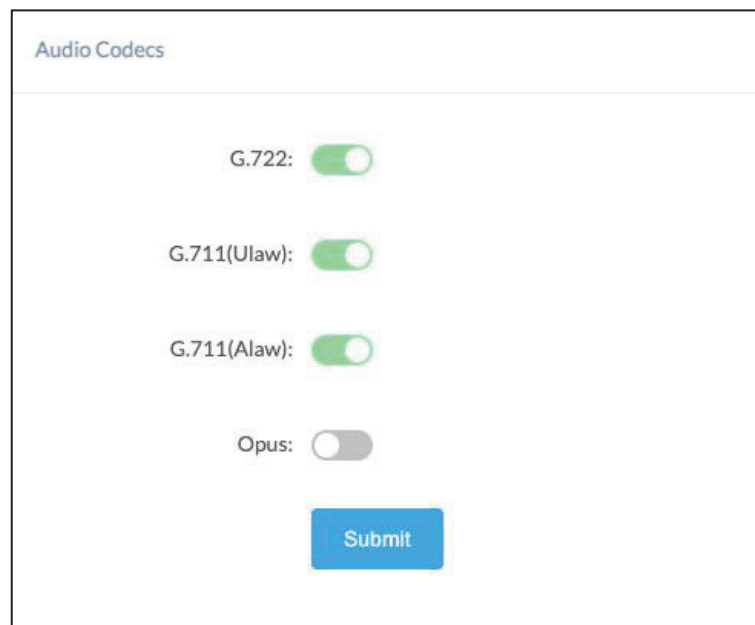
P2P Account

- **User ID:** The User ID will be displayed as the outgoing number when calling out, or the number that another device needs to dial.
- **Auto Answer:** Yes/No/Answer Delay, default in the Yes option.
- **Activate:** Enable/Disable the P2P feature.

4.3 Audio Codecs

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

To enable or disable an audio codec/codecs, please go to **SIP Settings --> Audio Codecs** page.



The screenshot shows a web interface for configuring audio codecs. The title is "Audio Codecs". There are four rows, each with a label and a toggle switch:

- G.722:
- G.711(Ulaw):
- G.711(Alaw):
- Opus:

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

Audio Codecs

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

4.4 Advance SIP Settings

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

Advance SIP Settings

Local Port:

* RTP Start Port:

* RTP End Port:

* RTP Timeout(Sec):

Jitter Buffer:

SIP Autoanswer beep:

Ptp Time Synchronization:

Advance SIP 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.
- **SIP Autoanswer Beep:** Enable/Disable. This setting represents the ringtone beep when a call comes and only applies when the SIP Autoanswer feature is enabled.
- **PTP Time Synchronization:** Enable the PTP time synchronization with the SIP server, which requires the support of the SIP server.

4.5 Video Settings (Ei-V05 Only)

To configure the camera related parameters of Ei-V05, please go to **SIP Settings --> Video** page.

The screenshot shows the 'Video' settings page with the following configuration:

- RTSP Access:
- Mainstream: `rtsp://192.168.11.211/stream1`
- Substream: `rtsp://192.168.11.211/stream2`
- * H.264 Payload Type:
- * MTU:
- Effect Controls:
- Video Codec: `H.264`
- Resolution: `1280 x 720`
- Frame Rate: `25fps`
- Bitrate: `1536Kbps`

Submit

Video

- **RTSP Access:** Enable/Disable the RTSP access setting.
- **H.264 Payload Type:** Configure the payload used for the H.264 encoding.
- **MTU:** Set the maximum transmission unit of video stream packets in the network.
- **Effect Controls:** Standard: Resolution 1280*720, Frame rate 25fps, Bit rate 1536Kbps. Normal: Resolution 704*576, Frame rate 25fps, Bit rate 768Kbps.

5. Advanced Settings

5.1 Volume Settings

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

Volume

* Speaker Volume:

MIC AGC:

* Microphone Volume:

* Incoming Audio Output:

* Audio Output:

Key beep:

Music Auto Resumes:

Play IP on startup:

Volume Settings

- **Speaker Volume:** The default speaker volume is 7, adjustable range is 0 ~ 9.
- **MIC AGC:** When this setting is enabled, the system will automatically adjust the microphone volume according to the environment. Users are able to adjust the microphone volume manually when this setting is disabled.
- **Microphone Volume:** The default microphone volume is 7, adjustable range is 0 ~ 9.
- **Incoming Audio Output:** When a device is calling in, you can set 'Line out' option, such as using an external speaker for output to achieve the purpose of louder paging.
- **Audio Output:** When using the intercom feature to make outgoing calls, you can set 'Speaker' option to use the local speaker as the output sound for communication.
- **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.

5.2 I/O Settings

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

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

Key settings

Key Action:

Destination: Line:

Press Again to End Call:

Key settings

Key Action:

* HTTP URL:

Key settings

Key Action:

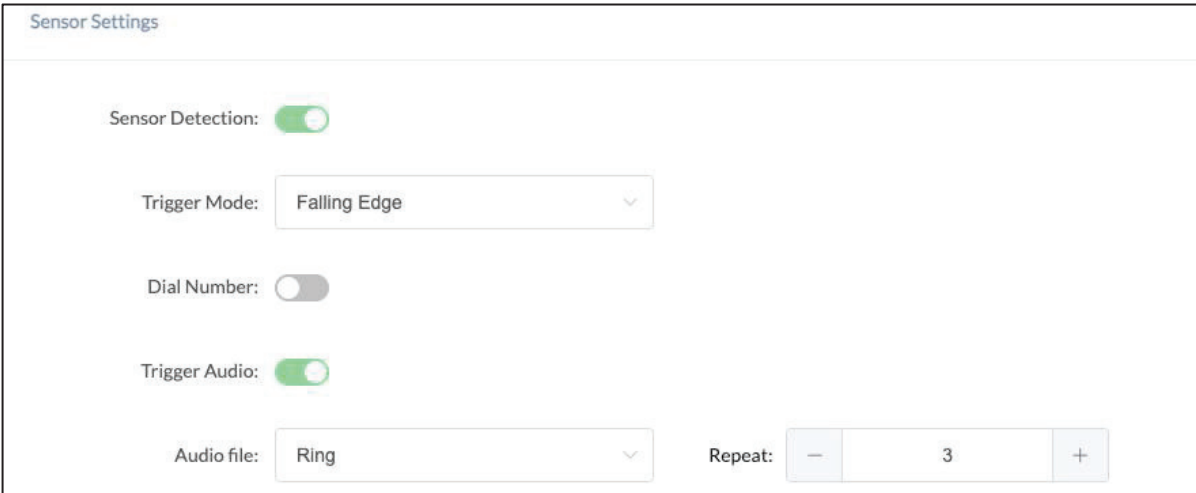
Audio File: Repeat:

Key Settings

- **Key Action / Wave to Call Action:** Choose different event linkage including Outgoing Call, HTTP Request and Play Audio.
- **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 plus 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.
- **Audio File:** Configure the audio triggered by linkage.
- **Repeat:** Configure the times of audio repetitions triggered by linkage.

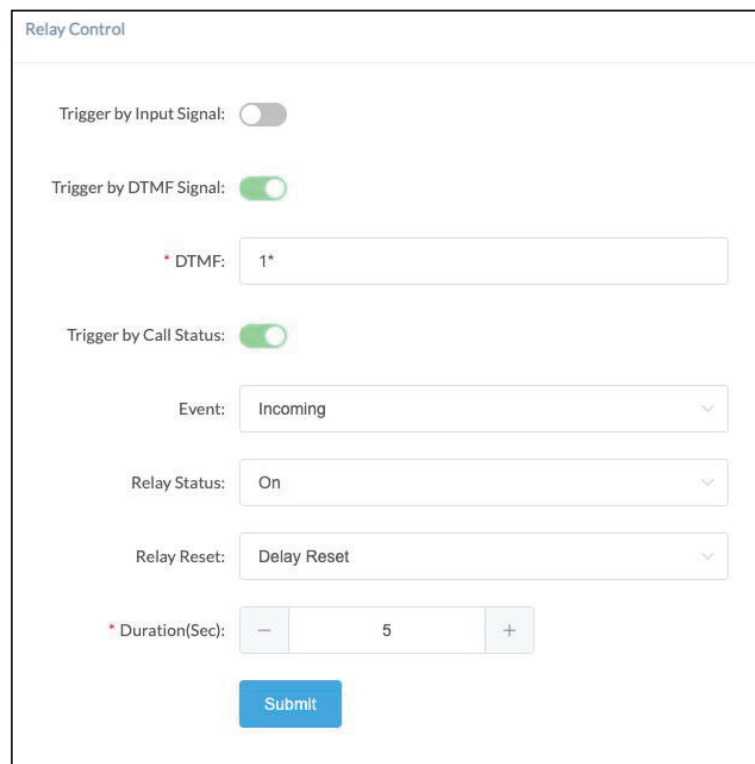


The screenshot shows the 'Sensor Settings' configuration page. It features several controls: a 'Sensor Detection' toggle switch that is turned on; a 'Trigger Mode' dropdown menu set to 'Falling Edge'; a 'Dial Number' toggle switch that is turned off; a 'Trigger Audio' toggle switch that is turned on; an 'Audio file' dropdown menu set to 'Ring'; and a 'Repeat' control consisting of a minus button, a text box containing the number '3', and a plus button.

Sensor Settings

- **Sensor Detection:** Enable/Disable, enable this option to detect any digital level signals.
- **Trigger Mode:** You can choose 'Falling Edge' or 'Rising Edge' mode to be triggered.
- **Dial Number:** Enable/Disable, enable this option if you need to dial a number when the corresponding digital signal is received.
- **Number:** The number that needs to be dialed.

- **Line:** Line represents the line that will be used to dial calls.
- **Trigger Audio:** Enable/Disable, enable this option if you need to trigger an audio when the digital signal is received. Please note that you can only select either ‘Dial Number’ or ‘Trigger Audio’, not both.
- **Audio File:** Select the audio file to be played when triggered by the sensor input.
- **Repeat:** Configure the times of audio repetitions triggered by the sensor input.



The screenshot shows a web interface titled "Relay Control". It contains several configuration options:

- Trigger by Input Signal:** A toggle switch that is currently turned off (grey).
- Trigger by DTMF Signal:** A toggle switch that is currently turned on (green).
- * DTMF:** A text input field containing the value "1*".
- Trigger by Call Status:** A toggle switch that is currently turned on (green).
- Event:** A dropdown menu with "Incoming" selected.
- Relay Status:** A dropdown menu with "On" selected.
- Relay Reset:** A dropdown menu with "Delay Reset" selected.
- * Duration(Sec):** A numeric input field with a value of "5", flanked by minus and plus buttons.

A blue "Submit" button is located at the bottom of the form.

Relay Control

- **Trigger by Input Signal:** Enable/Disable, enable this option if you need to use Input signal to trigger.
- **Trigger by DTMF Signal:** Enable/Disable, enable this option if you need to use DTMF signal to trigger (only RF2833 supported).
- **DTMF:** This setting represents the number to dial when DTMF is triggered.

- **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】** .
- **Relay Status:** This setting represents the responses by the triggers, there are 'On', 'Fast Flashing', and 'Slow Flashing' options to choose from.
- **Relay Reset:** This setting represents the reset mode after the trigger is responded, there are 'Delay Reset' and 'Hang-up Reset' options to choose from.
- **Duration (Sec):** This setting is only available if the reply control mode is on Delay Reset, it represents the time duration when the configure interface status changed.

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.

Call Event URL Callback ⓘ

Incoming Enable:

* Incoming Callback URL:

Outgoing Enable:

* Outgoing Callback URL:

Answered Enable:

Hangup Enable:

Relay Event URL Callback ⓘ

On Enable:

Off Enable:

Call Event URL Callback & Relay Event URL Callback

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

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

For example:

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

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

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

For example:

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

Call API Enable:

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

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

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

Relay API Enable:

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

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

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

Play API Enable:

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

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

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 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 **Advanced** ---> **Multicast** page to enable the multicast feature.

Multicast

Enable Multicast:

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	Volume
1	<input type="text"/>	<input type="text" value="2000"/>	Background-Music	Disabled	<input type="text" value="90"/>
2	239.168.15.206	<input type="text" value="2200"/>	1	Slow Flashing	<input type="text" value="90"/>
3	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	Disabled	<input type="text" value="90"/>
4	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	Disabled	<input type="text" value="90"/>
5	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	Disabled	<input type="text" value="90"/>
6	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	Disabled	<input type="text" value="90"/>
7	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	Disabled	<input type="text" value="90"/>
8	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	Disabled	<input type="text" value="90"/>
9	<input type="text"/>	<input type="text" value="2000"/>	<input type="text"/>	Disabled	<input type="text" value="90"/>

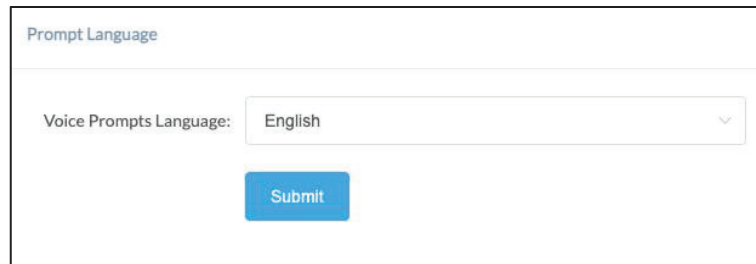
Multicast

- **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’.

5.5 Prompt Language

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

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



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

Prompt Language

5.6 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. System Settings

6.1 Network

Ei intercom uses DHCP to dynamically obtain IP addresses by default.

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

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

Network

Access Type: HTTP

DHCP:

* IP Address: 192.168.1.101

* Subnet Mask: 255.255.255.0

* Gateway: 192.168.1.1

* Primary DNS: 114.114.114.114

* Alternative DNS: 8.8.8.8

Submit

Network Configuration

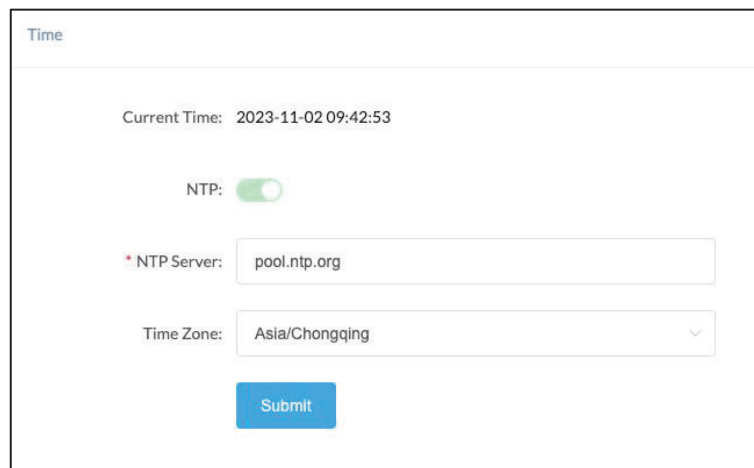
- **Access Type:** Specify the access method of the website, which currently supports HTTP and HTTPS.
- **IP Address:** Enter a vacant IP address within your LAN.
- **Subnet Mask:** Enter the subnet mask of your LAN.
- **Gateway:** Enter the default gateway of your LAN, this is essential for the device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

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

6.2 Time

Ei intercom obtains the time from the network time servers using NTP.

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



Time

Current Time: 2023-11-02 09:42:53

NTP:

* NTP Server:

Time Zone:

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 Account

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

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

Web Password Settings

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

6.4 Upgrade

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

The screenshot shows the "Upgrade" page. At the top, it says "Current Firmware Info". Below that, it displays "Current Firmware Version: s1.0.3" and "Last Update: 2023-08-24". A red warning message states: "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 and a large dashed box containing a cloud upload icon and the text "Drag files here, or Click Upload".

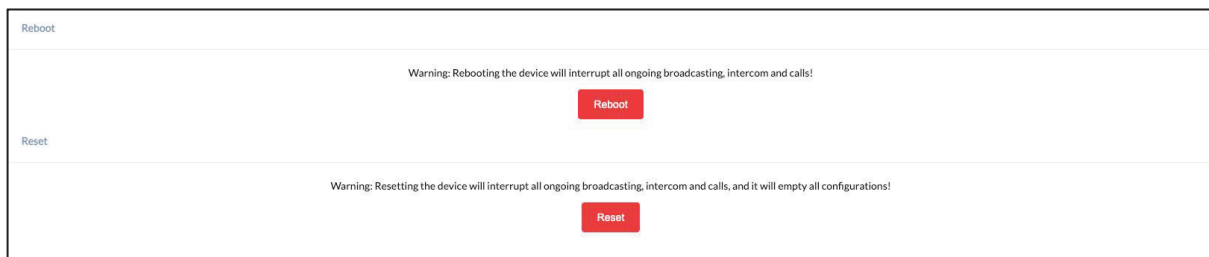
Upgrade Settings

- **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.

6.5 Reboot & Reset

Ei intercom 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.

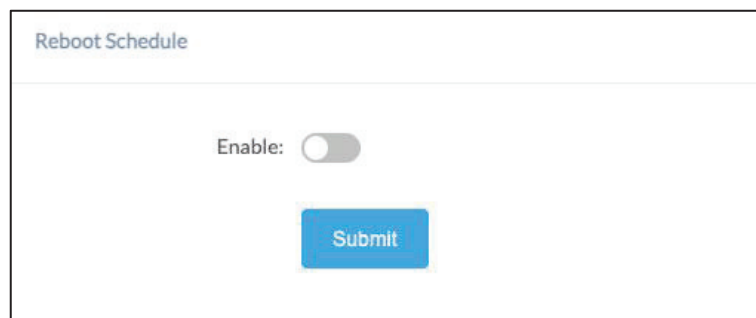


Reboot & Reset Settings

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!



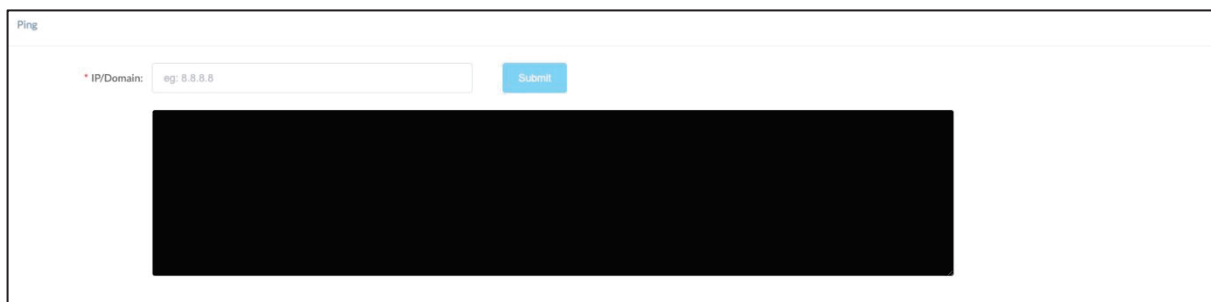
Reboot Schedule

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 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.

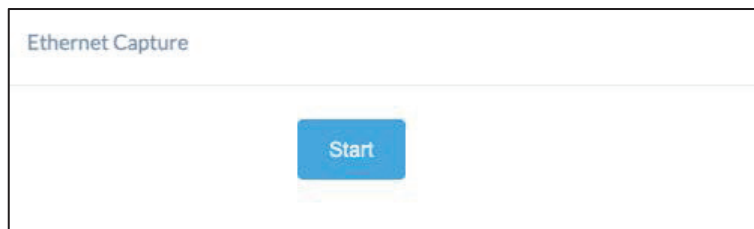


Ping

7.2 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.

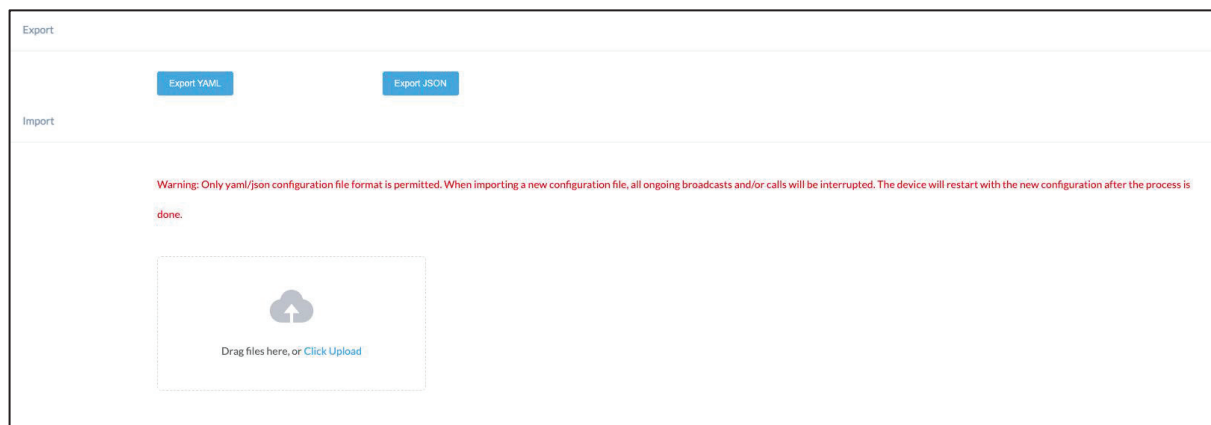


Ethernet Capture

7.3 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

7.4 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, 6869E290012

Static Provisioning Server

Access Mode: TFTP

TFTP Server Address: 10.10.1.5

Configuration Format: JSON

Configuration Filename: \$mac.json

Update Mode: Update after reboot

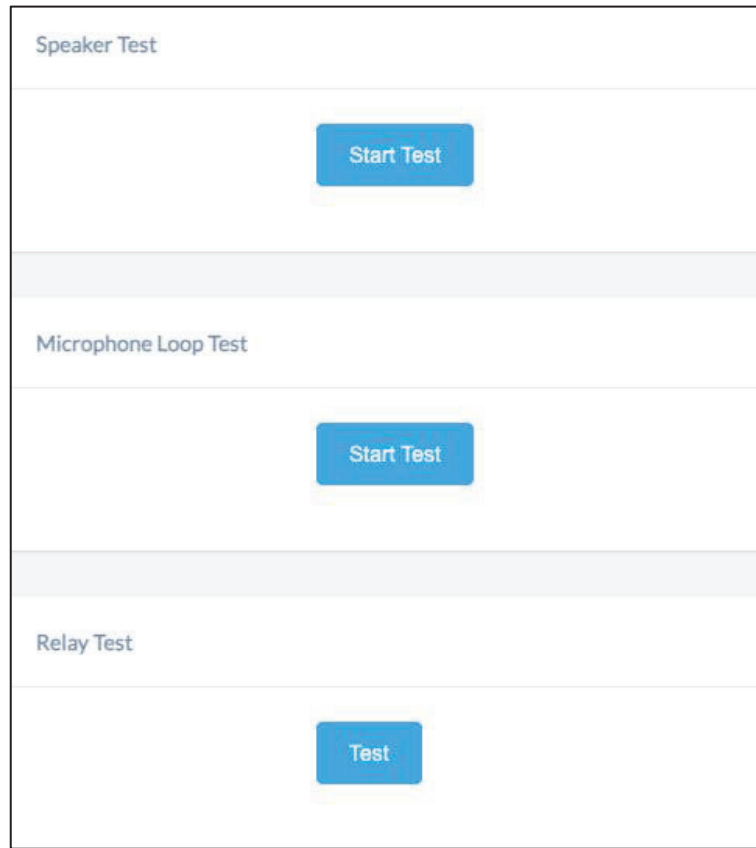
Submit

Auto Provisioning

7.5 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.