ZYCOO
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SQ10 Network Square Speaker

Contents

Overview

1.1 Product Overview

ZYCOO SQ10 series of Network Square Speaker currently has a total of eight models, four for flush-mounted and four for surface-mounted, which can be selected according to different application scenarios. Each model is a full-featured and high-performance SIP-enabled speaker.

Integrated Microphone allows for half-duplex or full-duplex paging or intercom based on the software platform used. When used with the Push-to-talk Button, calls to a predetermined extension or trigger a task can be initiated from the room with the speaker.

The SQ10-C/CF and SQ10-T/TF models are integrated with two flashing LED lights, which support multiple combinations of flashing methods to correspond to different types of operation. Such as slow-flashing, fast-flashing, simultaneous flashing, and alternate flashing to alert room occupants of an incoming audio message or emergency notices. In addition, The LED displays time and eliminates the need for a separate clock system, perfect to use in scenarios like classrooms, libraries, offices. Viewable at 65ft.

The SQ10-C/CF and SQ10-V/VF models are integrated with the built-in camera, which can directly realize the video intercom and linkage functions. External IP cameras are also supported for all models of the SQ10 series as long as the IP cameras are RTSP-supported. Note: The video linkage feature needs to be used with the IP Audio Solution).

SQ10 is PoE supported, making it easy to connect into local area networks from your PoE switch with a CAT 5/6 cable. No external power supply or other additional wiring is required.

With the support of peripheral integration, the SQ10 can connect with strobes, call buttons, volume controller, external LED/LCD monitor, door sensor, etc.

Standard SIP protocol implemented, you can register the SQ10 into any third-party SIP server such as IPPBX system or PA system, and the features detailed above are partially applicable.

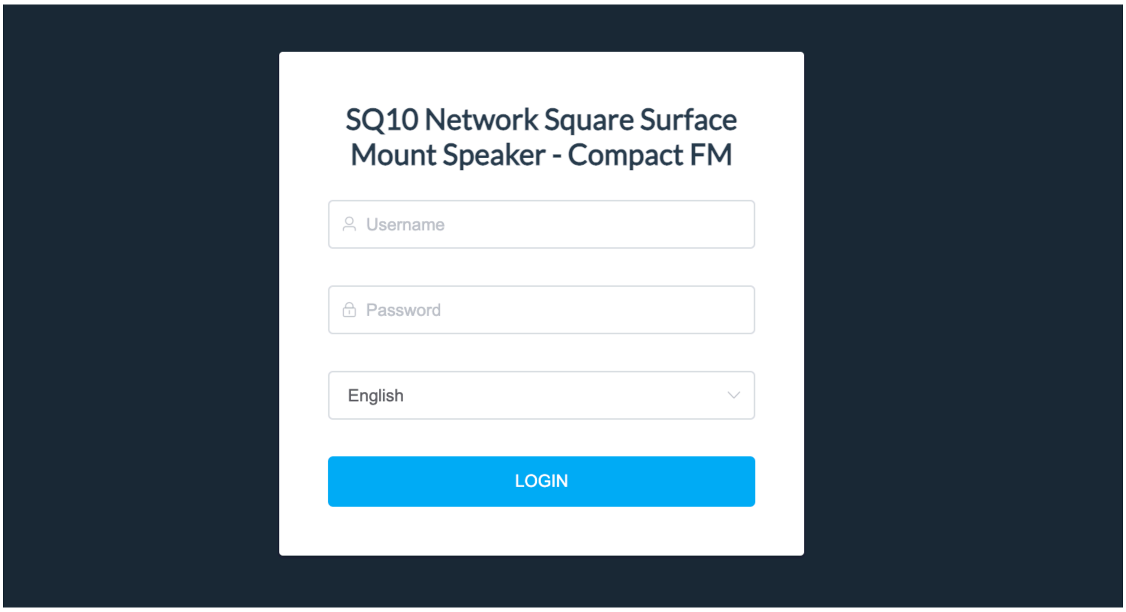
1.2 Product Specifications

Speaker Size:	4.5″
Average Sensitivity:	91dB/1m/1W
THD:	<0.1%
Frequency Range:	70Hz – 20KHz
Max SPL at 1m (Passive)	101dB SPL
Rated power:	8Ω 10W
Amplifier:	Single-Chanel Class D Topology
Coverage pattern::	90°H 50°V 30m2
SYS Light Status	On/Off/Flash
PWR Light Status	On/Off/Flash
RST Key	Press and hold for 3 seconds to announce IP address. Press and hold for 10 seconds to reset device to factory defaults.

Basic Settings

2.1 Web Interface Login

By default, the SQ10 speaker's IP assignment has been configured as DHCP. Please ensure there's a DHCP server available in the LAN where the SQ10 speakers are installed. If there's no DHCP server available or DHCP fails, you'll have to use the default static IP address 192.168.1.101 to access the web management interface. Press and hold the call button for 3 seconds then release on the device's front panel, the device will announce its IP address. Input the IP address in the browser address bar to open the web management interface of the SQ10 speaker. The login screen is shown as below image, here we take SQ10-CF as an example.

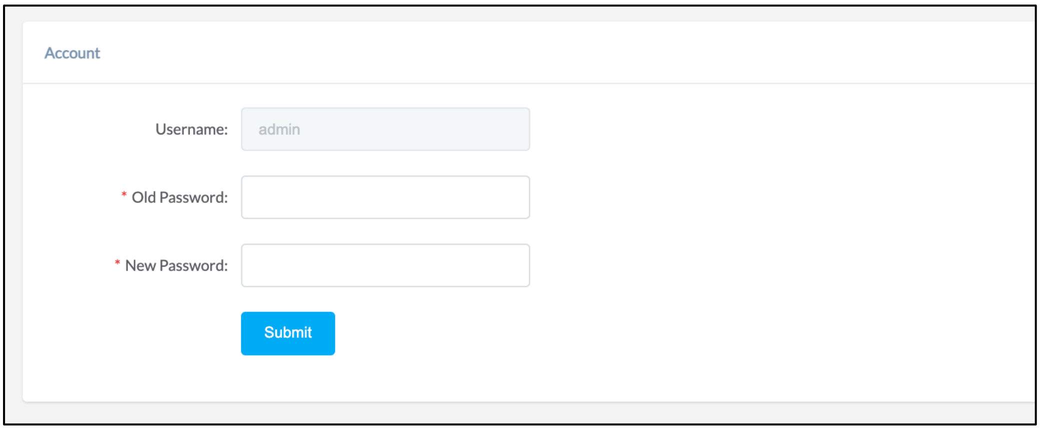


Use the default login credentials to login to the web management interface.

Default username: admin

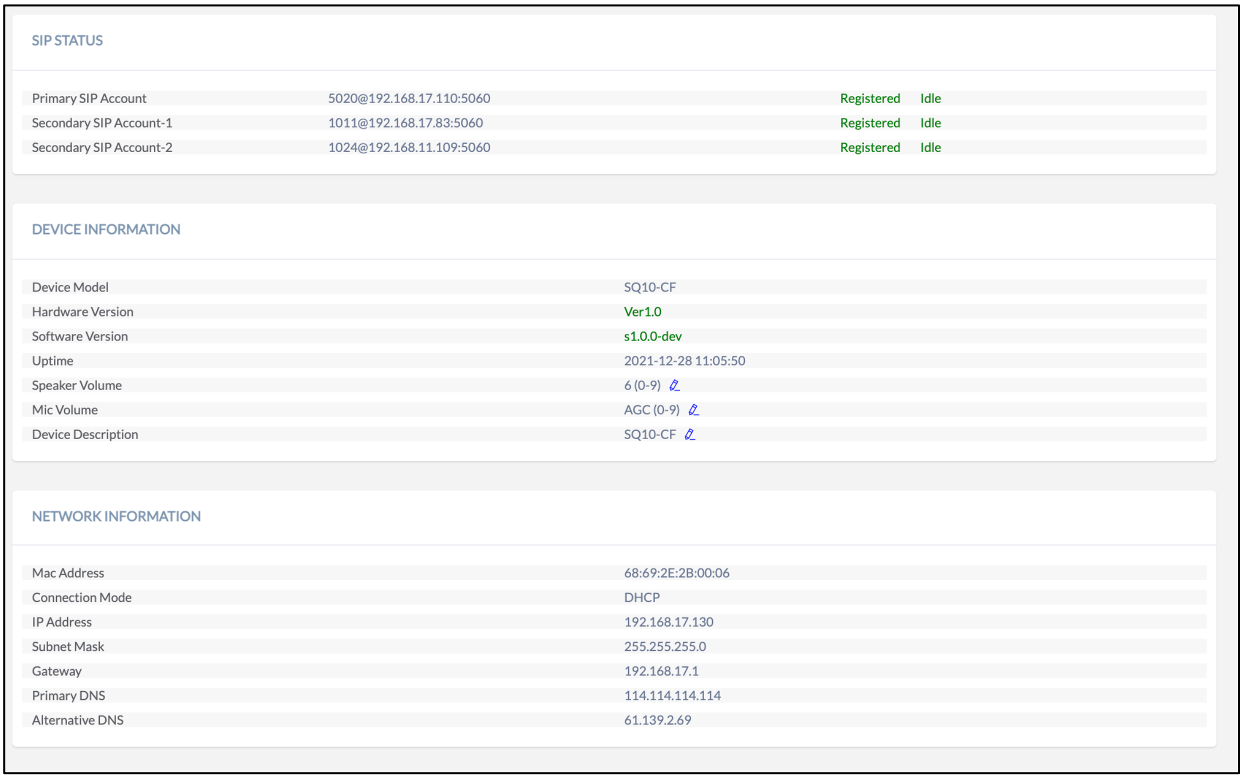
Default password: admin

For the safety of the SQ10 speaker, it is recommended to change the default password after the first login, please go to System -> Account page to change the password.



2.2 Device Info

After login, you'll first see the Device Info screen.



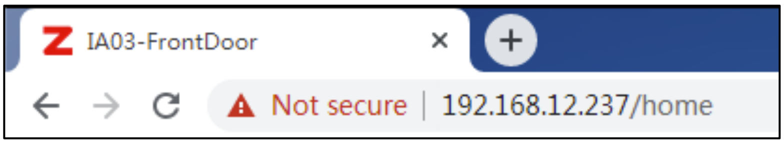
SIP STATUS

- SIP Account: The SIP number configured on this device.
- SIP Server: The SIP server address.
- Register Status: The SIP account registration status.

DEVICE INFORMATION

- Device Model: The device model.
- Hardware Version: Device hardware version.
- Software Version: Device software version, can be upgraded.
- Uptime: Last startup time of the device.
- Speaker Volume: The current volume level of the device.
- Mic Volume: The built-in microphone volume level.
- Device Description: The device description will be used to display as the tab name of the web browser. This is useful when configuring multiple devices using the same web browser. Click on the button to edit.

- After modification, the tab name will change.



NETWORK INFORMATION

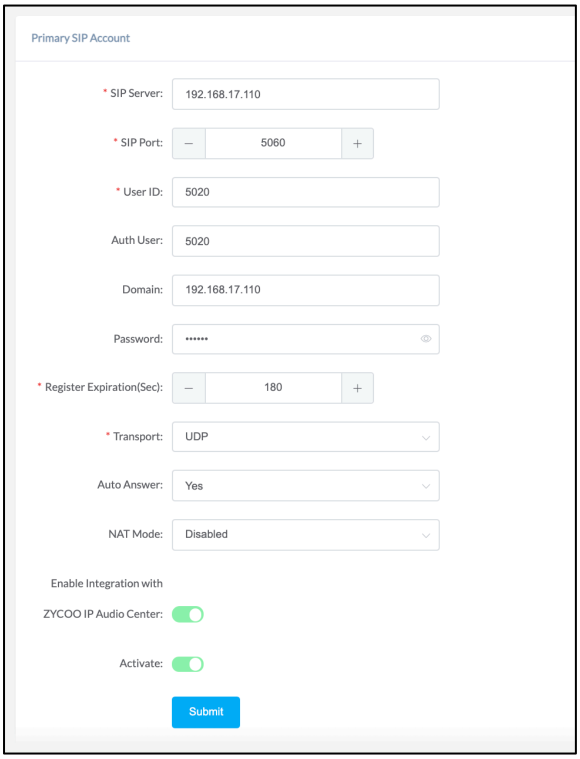
- Mac Address: Shows the device Mac address.
- Connection Mode: Shows the network mode of the device, either STATIC or DHCP.
- IP Address: Shows the current IP address of the device.
- Subnet Mask: Shows the current subnet mask of the device.
- Gateway: Shows the current default gateway of the device.
- Primary DNS: Shows the current primary DNS of the device.
- Alternative DNS: Shows the current alternative DNS of the device.

2.3 SIP Account

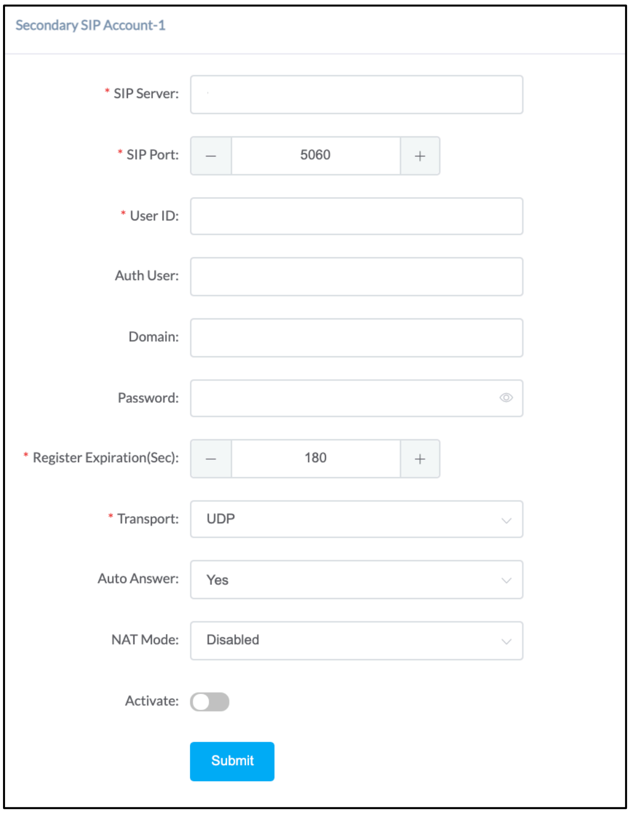
There are three (3) SIP accounts under the SIP Settings, one (1) primary and two (2) secondary for the use of different SIP accounts to proceed various tasks. If the current device needs to cooperate with the ZYCOO IP Audio Center, please turn on the 'ZYCOO IP Audio Center' option.

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

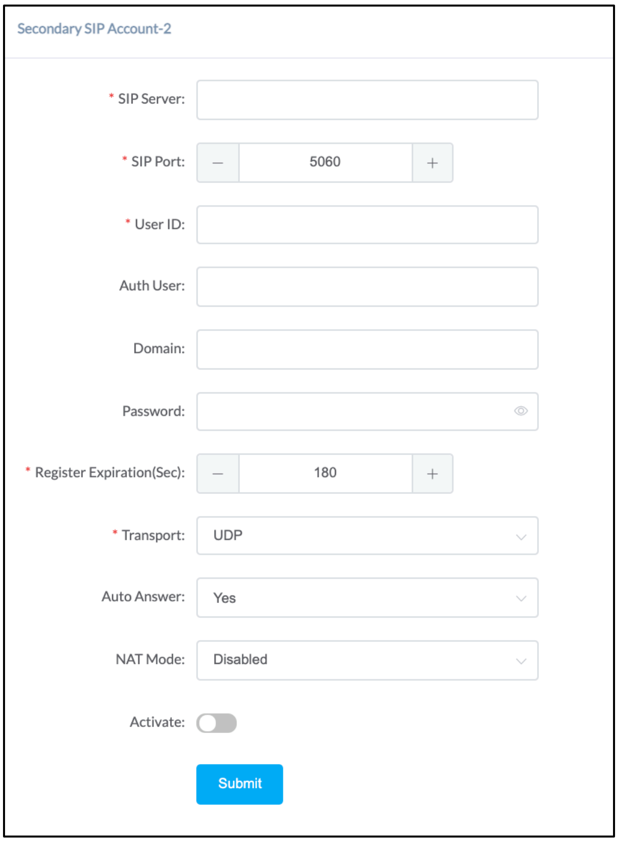
Primary SIP Account



Secondary SIP 1 Account



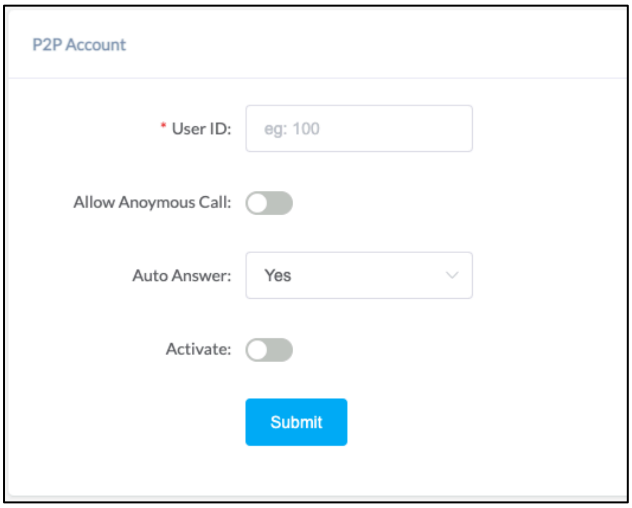
Secondary SIP 2 Account



- 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 other port number as SIP port, please modify in this setting.
- User ID: The SIP account number provided by SIP server.
- Auth User: Authorized SIP account's username.
- Domain: SIP domain.
- Password: Authorized SIP account's 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 Yes option.
- Ring Tone: When the Auto Answer is in No, you may choose the ring tone to play before the call is answered from this option.
- Answer Delay: When the auto Answer is in Answer Delay, you may set up the time of ring tone to play before the call is answered.
- NAT Mode: Select the NAT mode and fill out the corresponding data. STUN, TURN, and ICE modes are supported.
- Activate: Enable/Disable the SIP register feature.

2.4 P2P Account

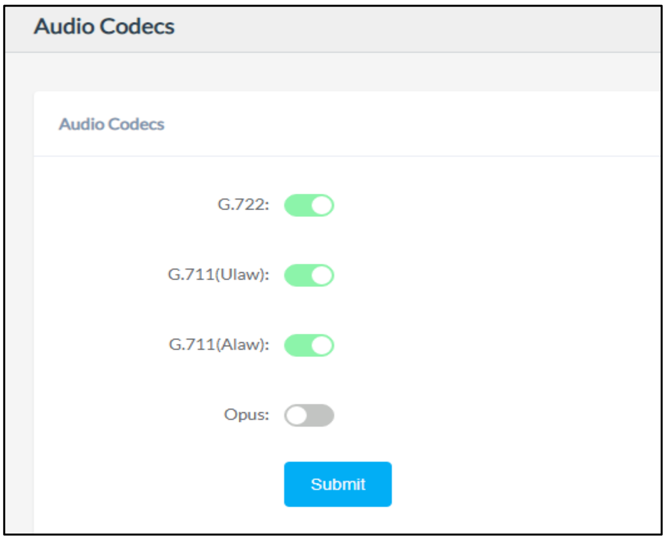
P2P stands for Peer to Peer. In a P2P network, the peers are connected to each other via the Internet, files can share, or peers can call each other directly between systems on the network without the need of a central server.



- User ID: The User ID will be displayed as the outgoing number when call out, or the number that other device need to dial.
- Allow Anonymous Call: When this option is enabled, the device is allowed to be called without user ID, default in Disable.
- Auto Answer: Yes/No/Answer Delay, default in Yes option.
- Ring Tone: When the Auto Answer is in No, you may choose the ring tone to play before the call is answered from this option.
- Answer Delay: When the auto Answer is in Answer Delay, you may set up the time of ring tone to play before the call is answered.
- Activate: Enable/Disable the P2P feature.

2.5 Audio Codecs

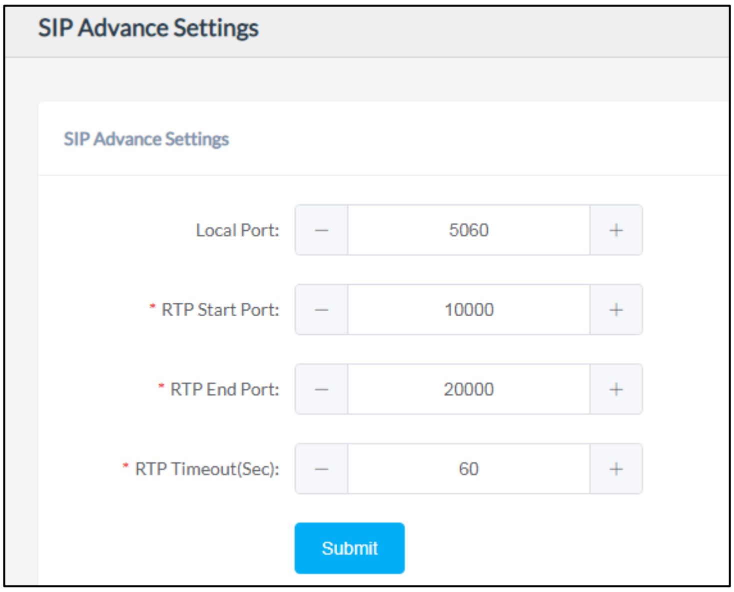
The network speakers support 4 audio codecs: G.722 (Wideband codec), G.711(Ulaw), G.711(Alaw) and Opus
To enabled or disable an audio codec/codecs, please go to *SIP Settings -> Audio Codecs page.*



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

2.6 Advance SIP Settings

Configuration on some more advance SIP protocol settings.
Please go to SIP Settings -> Advance SIP Settings

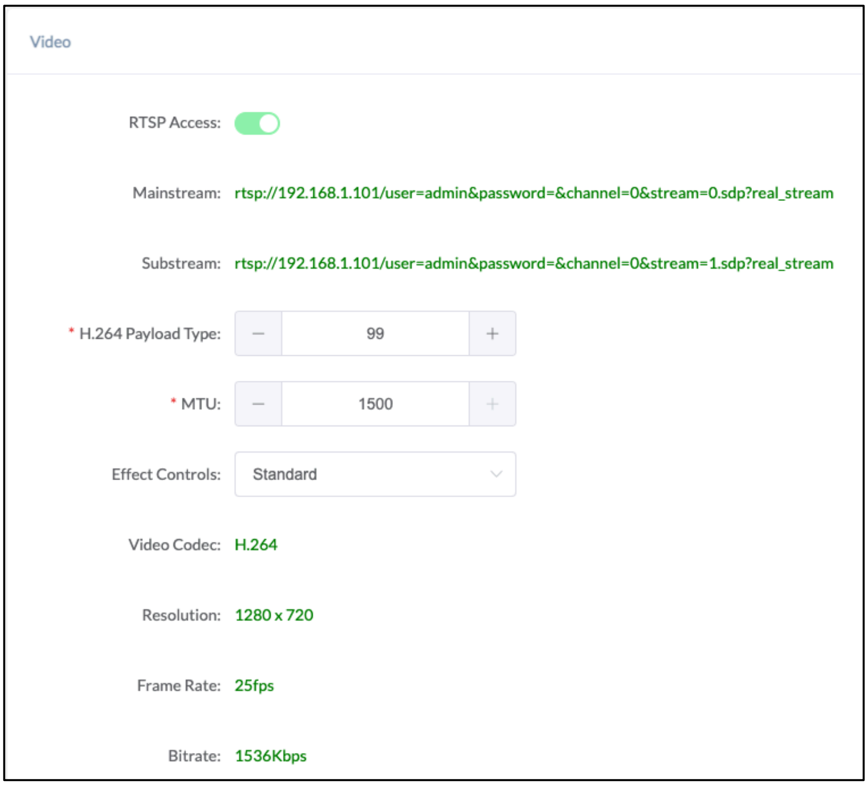


- Local Port: This setting represents the port that used to receive SIP packets.
- RTP Start Port: This setting represents the starting RTP port that the system will use for media sessions.
- RTP End Port: This setting represents the end RTP port that the system will use for media sessions.
- RTP Timeout (sec): This setting represents in a specific time range, if the system doesn't receive the RTP stream, then the call will end.

2.7 Video

Configuration the video intercom calling settings.

Please go to SIP Settings -> Video

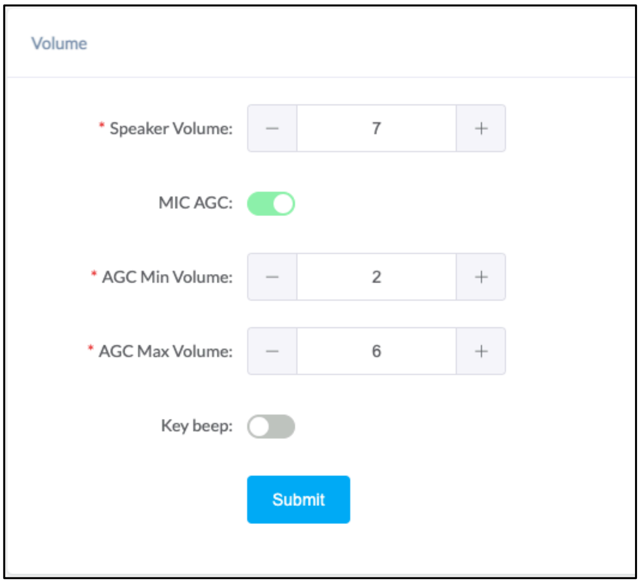


- RTSP Access: Enable/Disable the RTSP access setting.
- H.264 Payload Type: Configure the payload used for the H.264 encoding.
- MTU: Set the maximum transmission unit of video stream packets in the network.
- Effect Controls: Standard: Resolution 1280*720, Frame rate 25fps, Bit rate 1536Kbps.
  Normal: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

Advanced Settings

3.1 Volume Control

The network speaker's volume level can be adjusted from its web management interface, on the *Settings -> Volume Control* page.

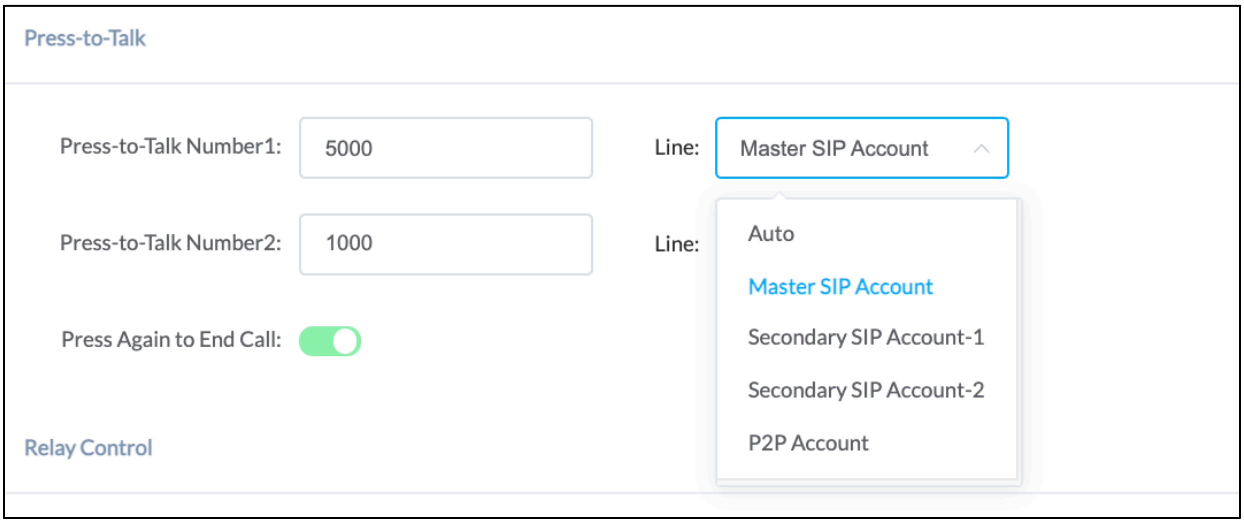


- Speaker Volume: The default speaker volume is 7, adjustable range is 0 ~ 9.
- MIC AGC: When this setting is enabled, the system will automatically adjust the microphone volume according to the environment. User able to adjust the microphone volume manually when this setting is disabled.
- Microphone Volume: The default microphone volume is 7, adjustable range is 0 ~ 9.
- AGC Min Volume: This setting represents the minimum value of the automatic gain control.
- AGC Max Volume: This setting represents the maximum value of the automatic gain control.
- Key Beep: Enable/Disable the beep sound from key button.

3.2 I/O Settings

I/O settings are used to configure the press-to-talk, and dry contact relay control options. Please go to the *Settings -> I/O Settings* page.

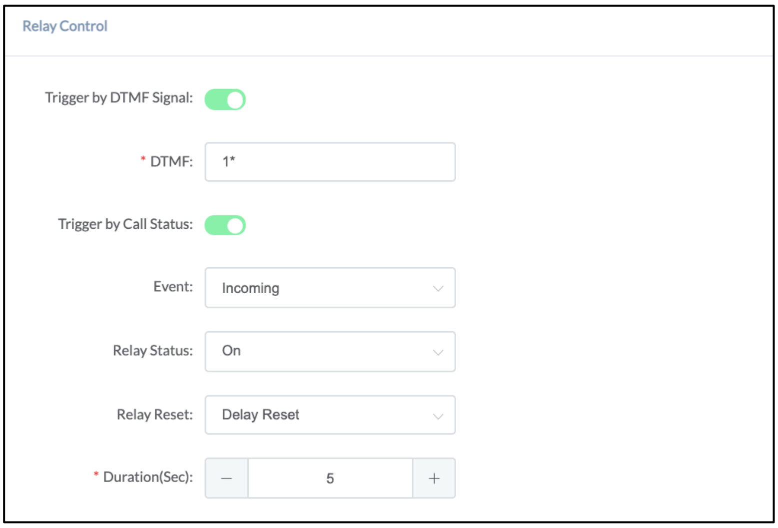
For Press-to-Talk, when the speaker is connected with external press-button(s), calls to a predetermined extension or trigger a task can be initiated from the room with the speaker.



Press-to-Talk

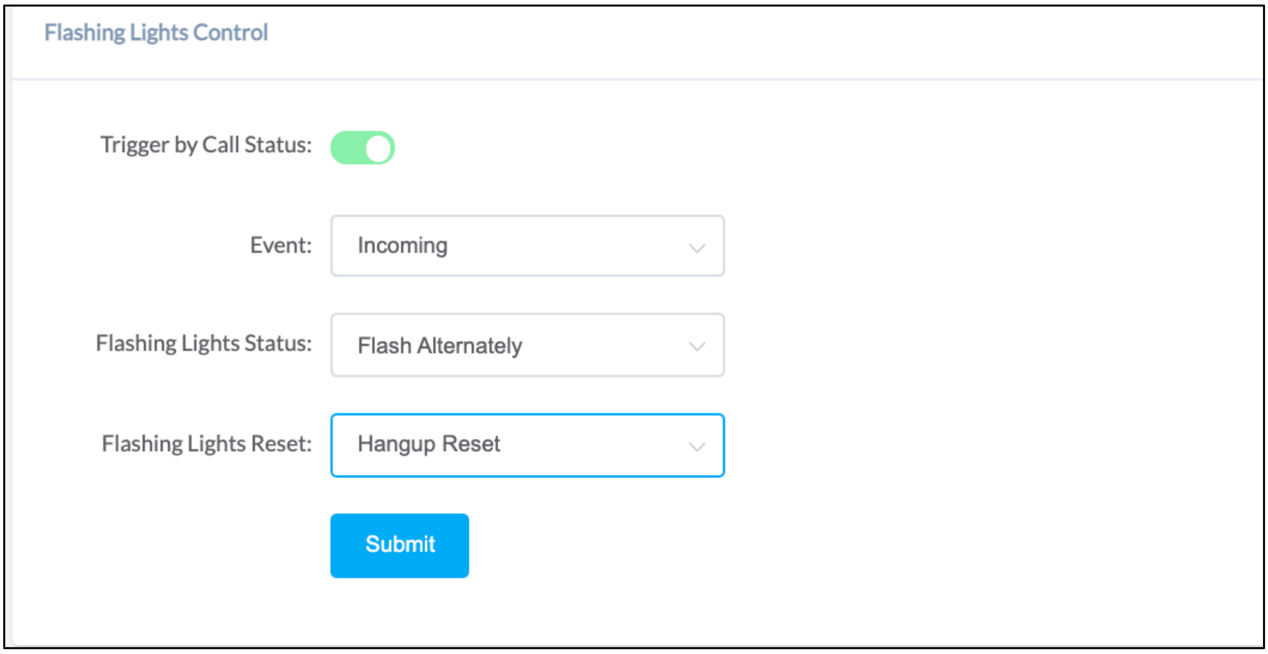
- Press-to-Talk Number: This setting represents the number will be dialed by using Key.
- Line: Line represents the line that will be using to dial the call.
- Press Again to End Call: Enable/Disable. If this setting is enabled, then press the key twice will end the call when it's in process.

Relay Control



- To use DTMF signal as the trigger of dry contact relay output, then please specify the DTMF key press.
- To use Call status signal as the trigger of dry contact relay output, then please specify the Call Event type.
- Trigger by DTMF Signal: Enable/Disable, enable this option when need to use DTMF signal to trigger (only RF2833 supported).
- DTMF: This setting represents the number to dial when DTMF triggered.
- Trigger by Call Status: Enable/Disable, enable this option will change the call status when triggered.
- Event: There are five options of call status, respectively are Outgoing/Incoming/Outgoing&Incoming/Answered/Hang-up.
- Relay Status: Choose the relay output as On/Fast Flashing/ Slow Flashing.
- Relay Reset: Choose the relay to be reset when the call is Delay/Answered/Hang-up.

Flashing Lights Control

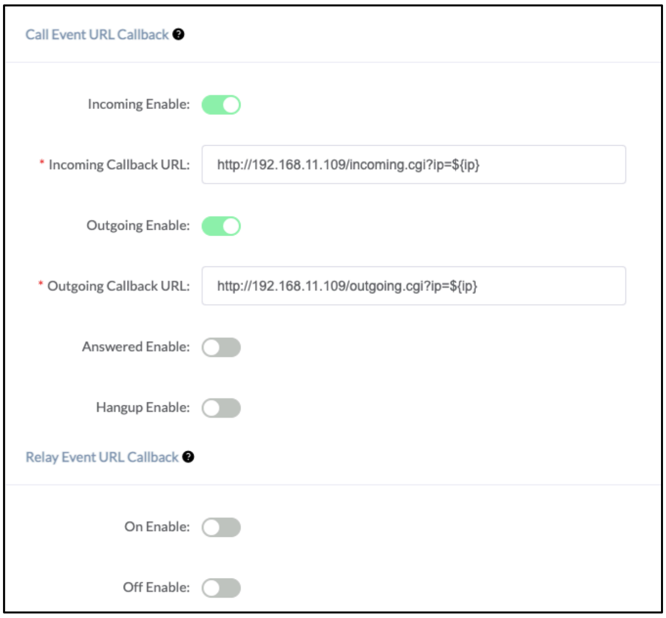


- Trigger by Call Status: Enable/Disable. When this option is enabled, light status change regarding the call status.
- Event: There are five options of call status, respectively are Outgoing/Incoming/Outgoing & Incoming/Answered/Hang-up.
- Flashing Lights Status: Choose the flasher output as On/ Flash Simultaneously/Flash Alternately.
- Flashing Lights Reset: Choose the flasher to be reset when the call is Delay/Answered/Hang-up.

3.3 API Settings

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

Path: *Advanced/API Settings*



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,

${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

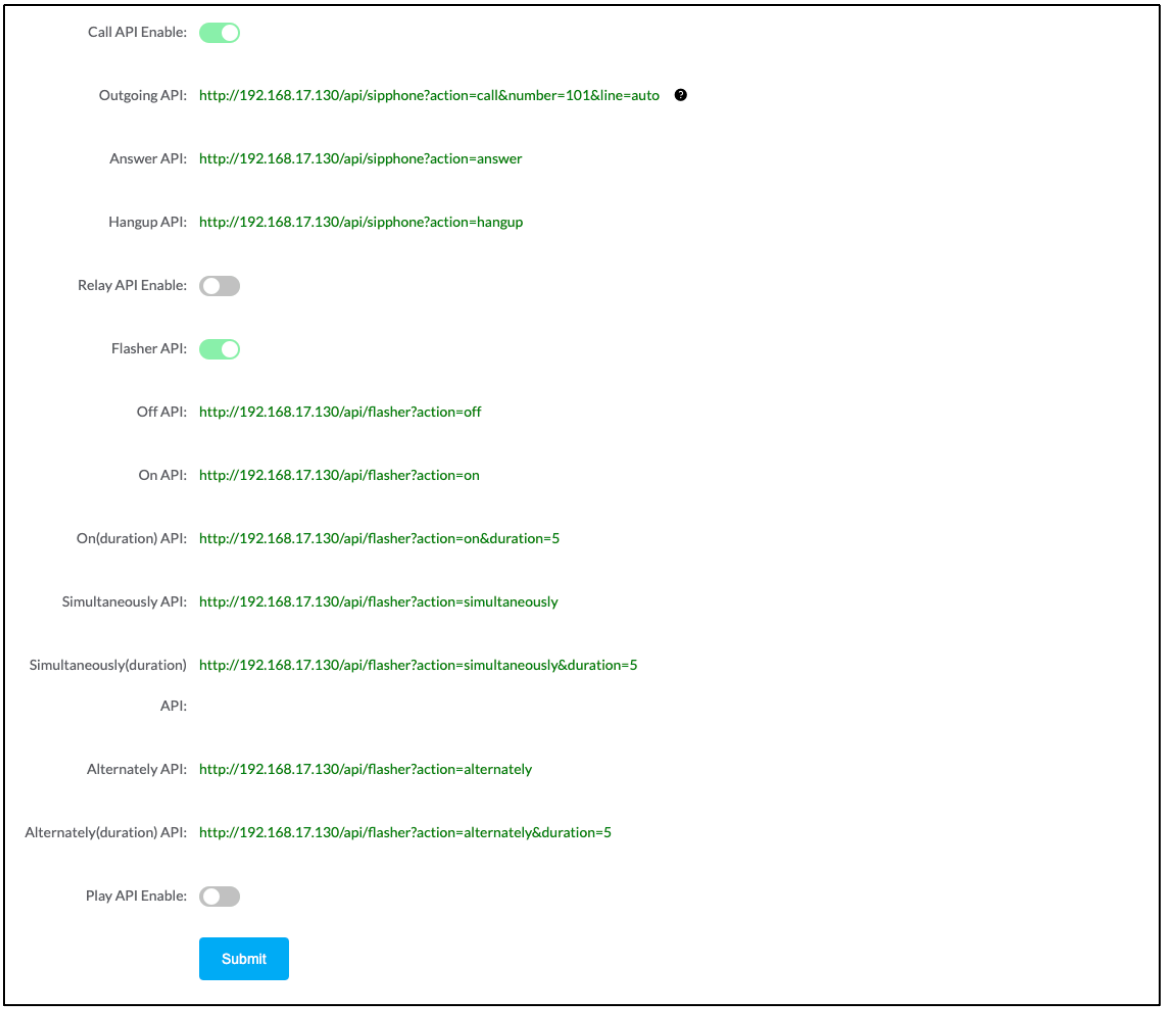
Relay Event URL Callback

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,

${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, devices linkage, call control, relay control, flasher control and play sound by the systems.

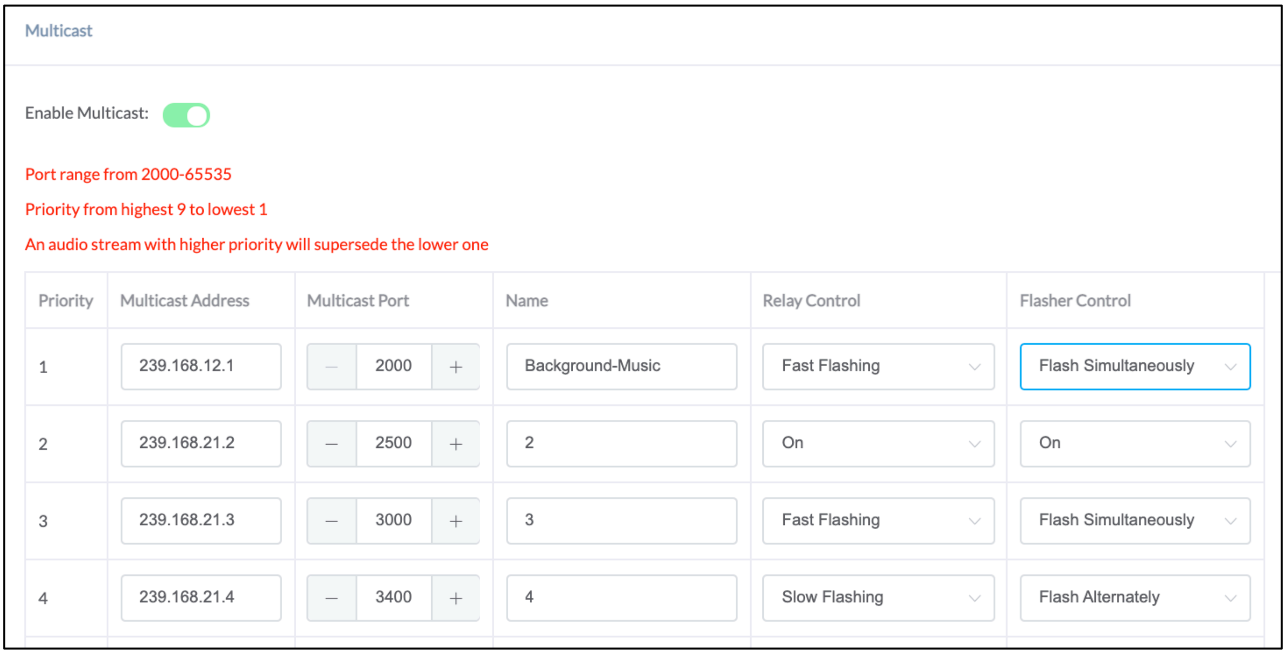


(Note: Authentication and encryption are not used in the API interface, so please pay attention the security of the network environment when opening and using these API interfaces).

3.4 Multicast Settings

The multicast settings are used to configure the parameter settings of the multicast function on the SIP Safety Intercom. It can configure 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 *Advance Settings -> Multicast Settings*.



- 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 a name of the multicast address.
- Relay Control: Options to choose from are Disabled/On/Fast Flashing/Slow Flashing.
- Flasher Control Options to choose from are Disable/On/Flash Simultaneously/Flash Alternately.

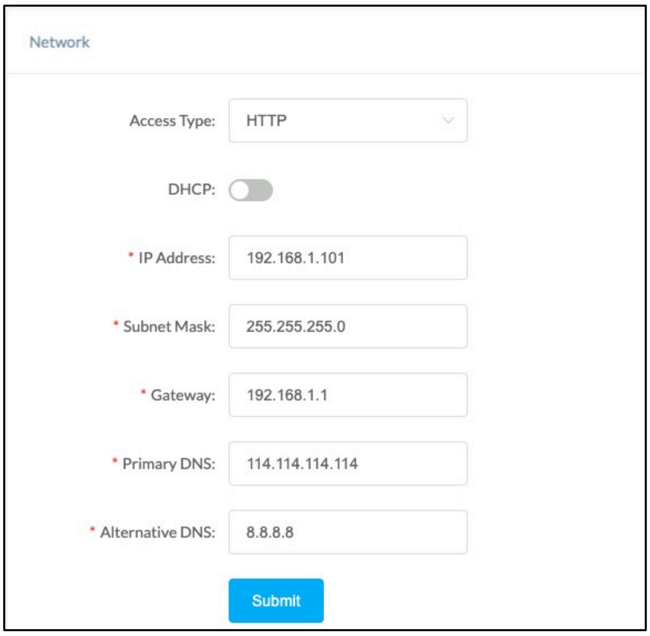
3.5 Language Settings

The language of local voice prompts, like IP address announcements, can be set on Settings -> Prompt Language page.

Settings

4.1 Network Setting

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



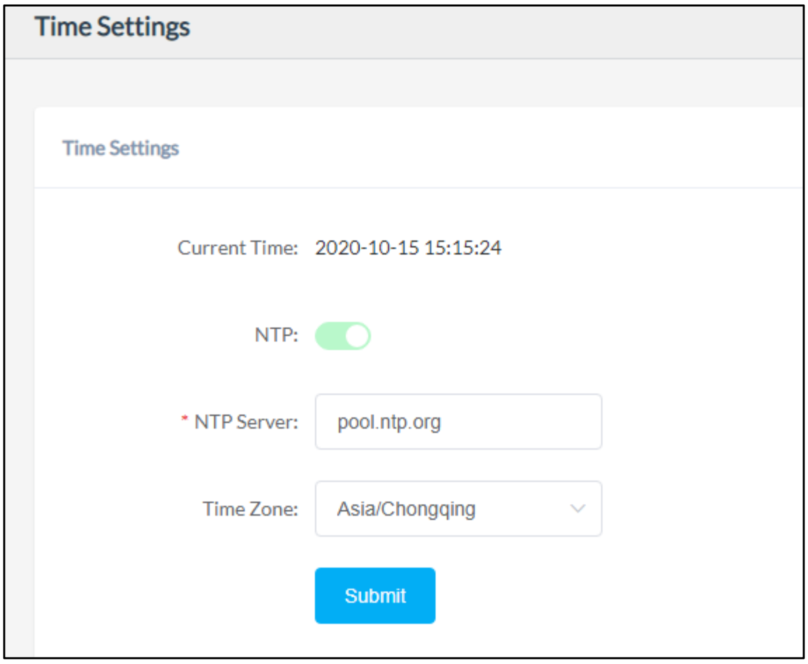
Turn the DHCP switch button off to show the static network parameter settings.

Network Configuration Parameters

- Access Type: Specify the access method of the website, 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 network speakers 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.

4.2 Time Settings

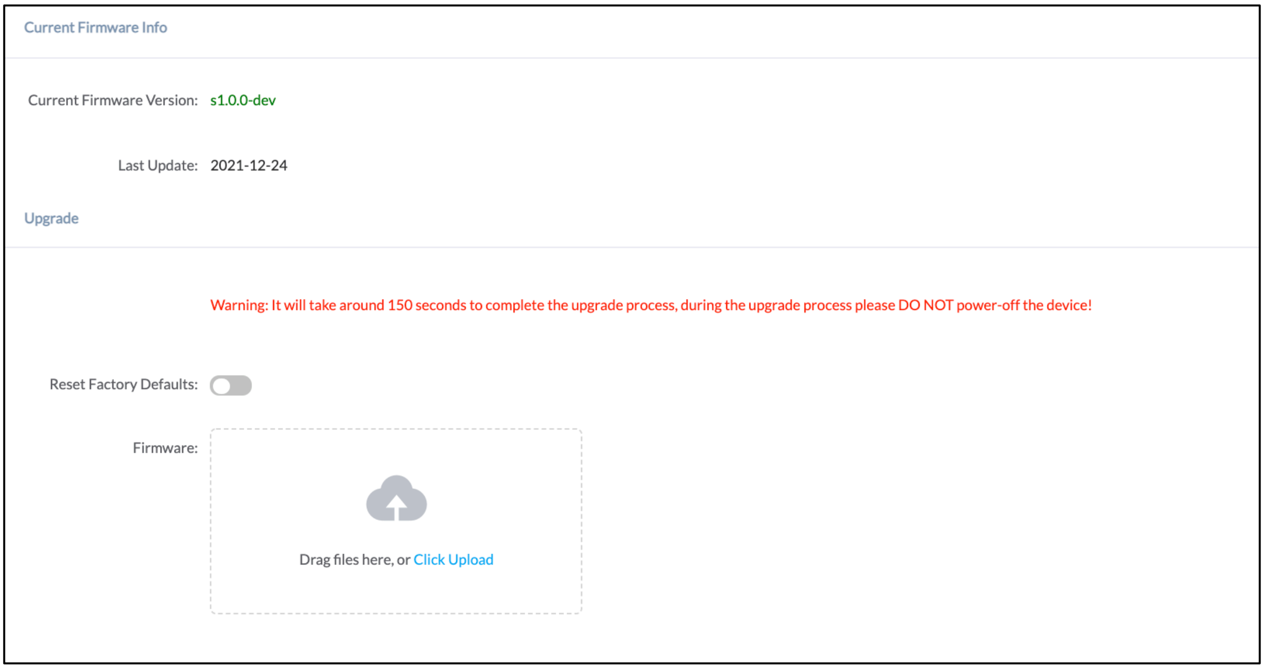
The network speakers obtain time from the network time servers using NTP, to change the NTP settings please go to Settings -> Time Settings page.



Here you can change a NTP server by modify the NTP server address and you can select the time zone of your location, so the network speaker will synchronize time of your time zone from the NTP server you have configured.

4.3 Upgrade

To upgrade the network speaker's firmware, please go to Settings -> Upgrade page.



You'll first see the current firmware version of the network speaker and the last upgrade time.

Upload the .img file provided by ZYCOO to perform the upgrade action. If you wish to reset the network speaker to factory defaults after upgrading, please enable the "Reset Factory Defaults" parameter.

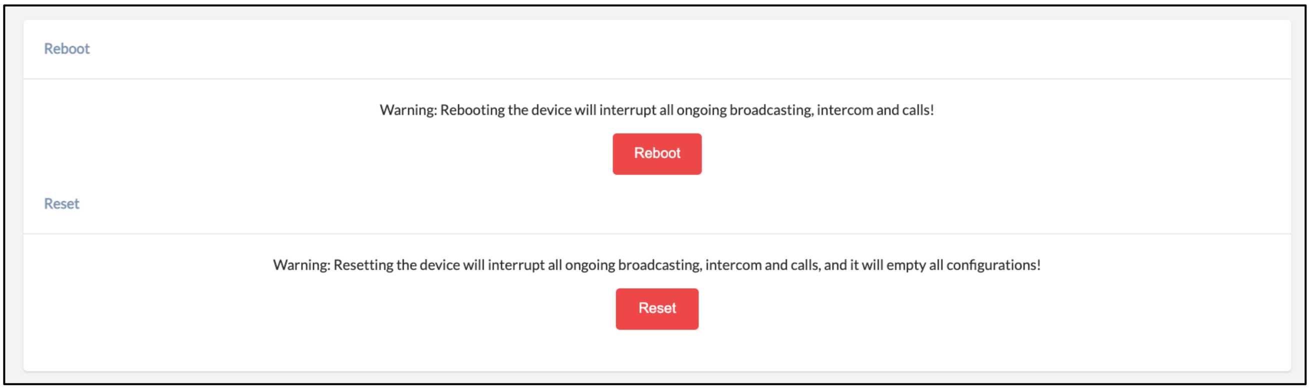
It will take around 2 minutes to complete the firmware upgrade, during upgrading process please DO NOT power off the network speaker.

4.4 Reboot & Reset

The network speakers can be rebooted and reset from the web management interface on the Settings -> Reboot & Reset page.

Both reboot and reset action will terminate all broadcasting and SIP calls (paging). And the reset action will erase all configurations of the network speakers. Please reboot or reset the devices when they are not in use.

Except resetting from web management interface, the network speakers can be also reset by the RST button on the rear panel of the speakers. Press and hold the RST button for 10 seconds (5 seconds for IP address announcements) and release, now you should hear voice prompts "Resetting factory defaults, rebooting…", it means the speaker will now reset.

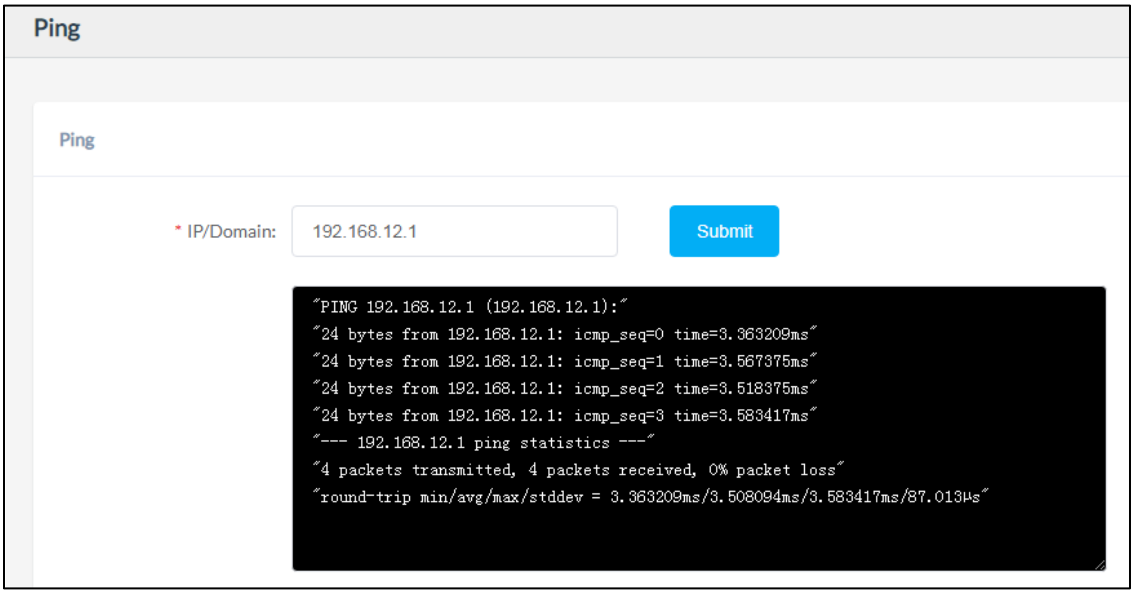


When the Reboot Schedule feature is Enabled, you can set up the automatic reboot daily, weekly, or monthly at a specify time.

Maintenance

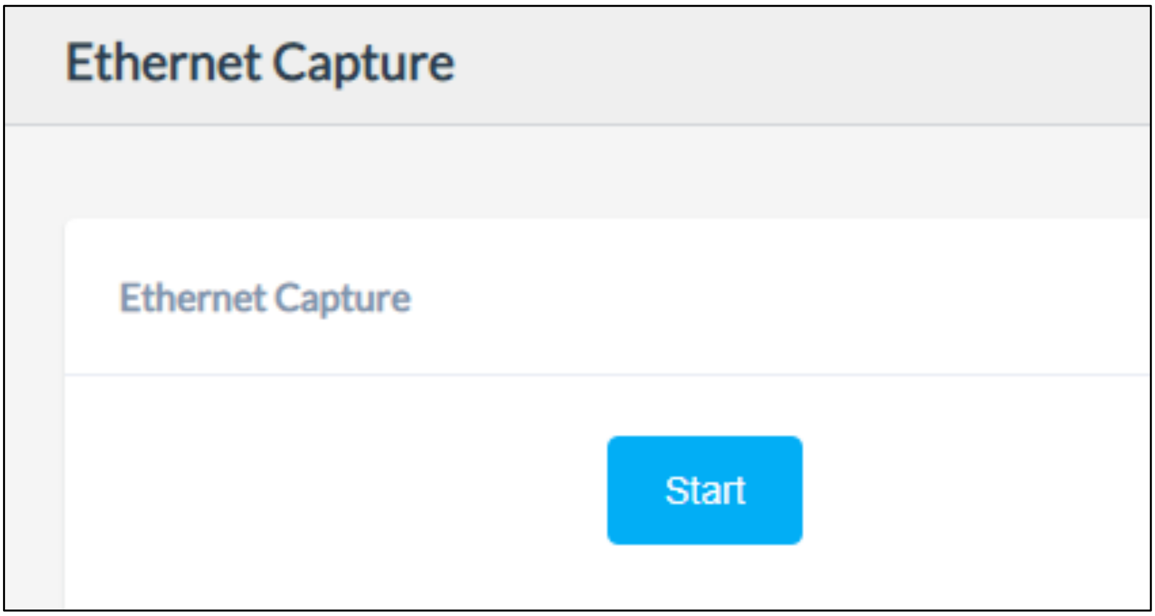
5.1 Diagnostic

Ping is a network administration utility or tool used to test connectivity on an IP network. Input other device's IP address and click on the submit button to trace network route.



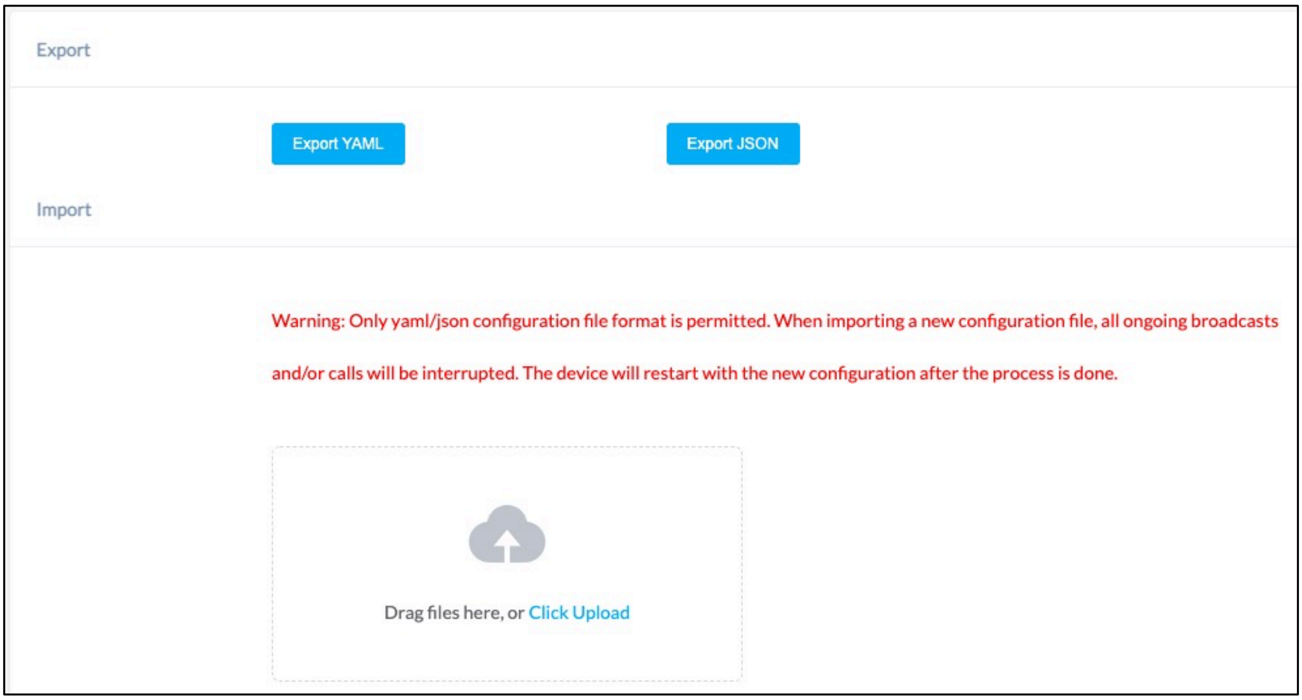
5.2 Ethernet Capture

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 analyzing.



5.3 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 recovery. Both YAML and JSON formats are supported.



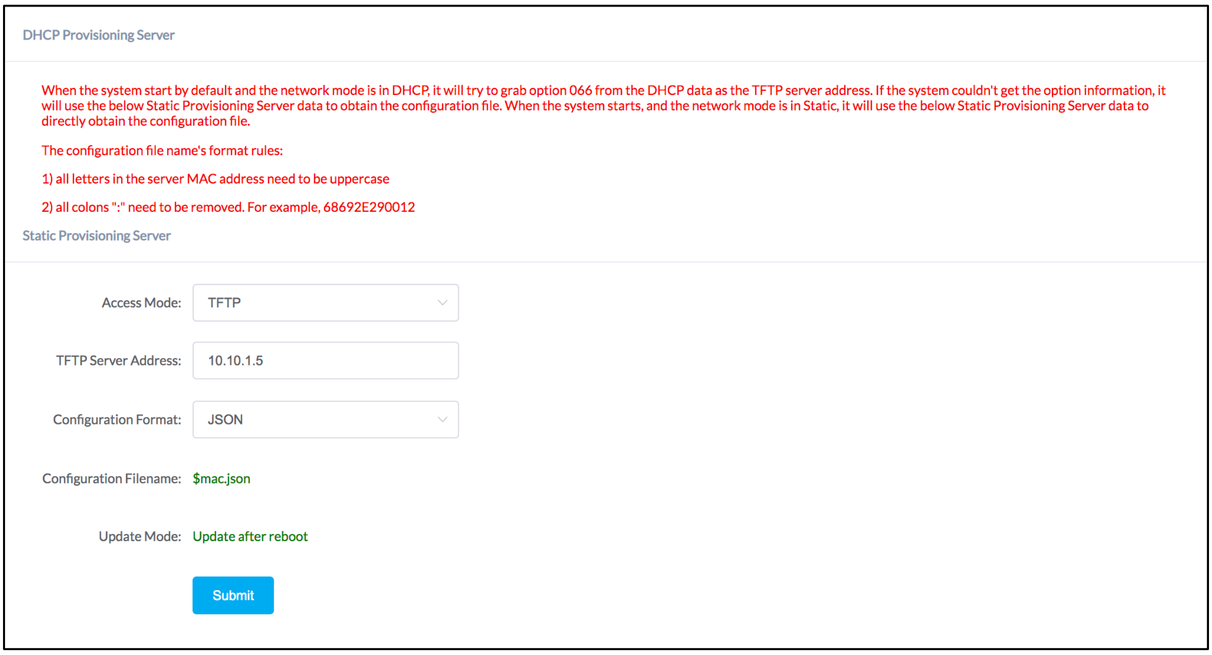
5.4 Auto Provisioning

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

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, 68692E290012.



5.5 Audio Detection

The audio detection feature provides an option for user to check whether the speaker is working functionally before registering it to the server.

Speaker Detection: Click on the Start button, the speaker will play a ringtone to test whether the speaker is working.

Microphone Loop Detection: Click on the Start button, then start speaking to the device. If the speaker is working functionally, you should hear the voice back.

www.zycoo.com

ZYCOO, NEEQ Code 839487, is the leading developer and manufacturer of IP telephony devices and related systems, devoting itself over the past years to R&D powerful and scalable voice over IP (VoIP) solutions that mainly serve the SMEs market and industry filed.