Network Speaker User Guide

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Overview

Product Overview

ZYCOO network speakers are SIP enabled high-performance speaker products which can be used for SIP paging, notification/tone broadcast and streaming media music playback.

The SC15 network ceiling speaker and SW15 network cabinet speaker are equipped with dual speaker drive units, the high-efficient, full-range speaker drive units can provide uniquely advanced listening experience, which makes the SC15 and SW15 suitable for high quality music, notification/tone broadcasting in the indoor environment.

The SH30 network horn speaker is equipped with a midrange drive unit powered by 30W class D amplifier, which makes it suitable for paging and notification/tone broadcasting to noisy large spaces and outdoor environments. On the other hand, the SL50 Network Column Speaker is 4G wireless compatible, while the SL30 Network Column Speaker is PoE+ supported. SL50 and SL30 cover a wide frequency band to provide high sensitivity, beautiful sound, and a unique listening experience for listeners. As well as the IP65-Enclosure makes the SL50 and SL30 perfect to work in any outdoor environment.

Product Specifications

SC15 Specifications

Speaker Components:	5.25" woofer unit + 1" tweeter unit
Sensitivity	85dB/1m/1W
Max Sound Pressure Level	100dB
Amplifier	Built-in Class D Amplifier
Rated Power	8Ω 15W
Frequency Range	70Hz~20KHz
Coverage Pattern	90°H 90°V 30 m²
Acoustics	Mono

SW15 Specifications

Speaker Components:	5.25" woofer unit + 1.5" tweeter unit
Sensitivity:	85dB/1m/1W
Max Sound Pressure Level	100dB
Amplifier	Built-in Class D Amplifier
Rated Power	8Ω 15W
Frequency Range	70Hz~20KHz
Coverage Pattern	90°H 90°V 30 m²
Acoustics	Mono

SH30 Specifications

Speaker Components	2" midrange driver unit
Sensitivity	105dB/1m/1W
Max Sound Pressure Level	117dB
Amplifier	Built-in Class D Amplifier
Rated Power:	8Ω 30W
Frequency Range:	400Hz~8KHz
Coverage Pattern:	50°H 50°V 70m effective distance
Acoustics:	Mono
IP Rating	IP65

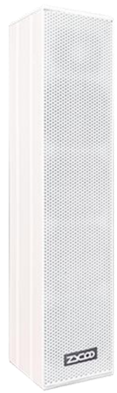
SL30 Specifications

Speaker Components	Two 3.25" woofer units + one 1" tweeter unit
Sensitivity	82dB/1m/1W
Max Sound Pressure Level	97dB
Amplifier	Built-in Class D Amplifier
Rated Power:	4Ω 30W
Frequency Range:	100Hz~20KHz

Coverage Pattern:	135°, best effective distance 50m
Acoustics:	Mono

SL50 Specifications

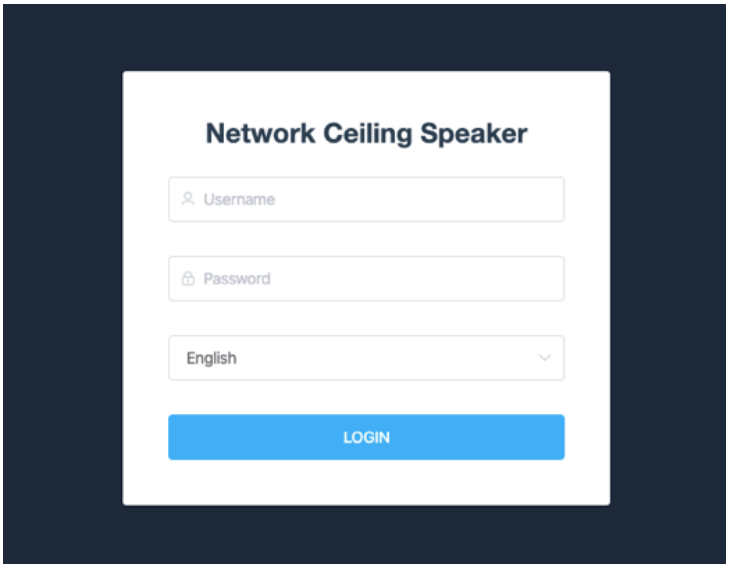
Speaker Components	4 3.25'' woofer units + 1 25ø tweet unit
Sensitivity	85dB/1m/1W
Max Sound Pressure Level	105dB
Amplifier	Built-in Class D Amplifier
Rated Power:	8Ω 50W
Frequency Range:	100Hz~20KHz
Coverage Pattern:	135° 50m, max effective distance 70m
Acoustics:	Mono
IP Rating	IP65



Basic Settings

Web Interface Login

By default, the network speaker's IP assignment has been configured as DHCP. Please ensure there's DHCP server available in the LAN where the network speakers are installed. If there's no DHCP server available or DHCP fails, you'll have to use the failover IP address 192.168.1.101 to access the web management interface. Press and hold the RST button for 5 seconds (10 seconds for reset) and release, the speaker will announce its IP address. Input the IP address in the browser address bar to open the web management interface of the speaker. The login screen is shown as below image, here we take SC15 network ceiling speaker as an example.



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

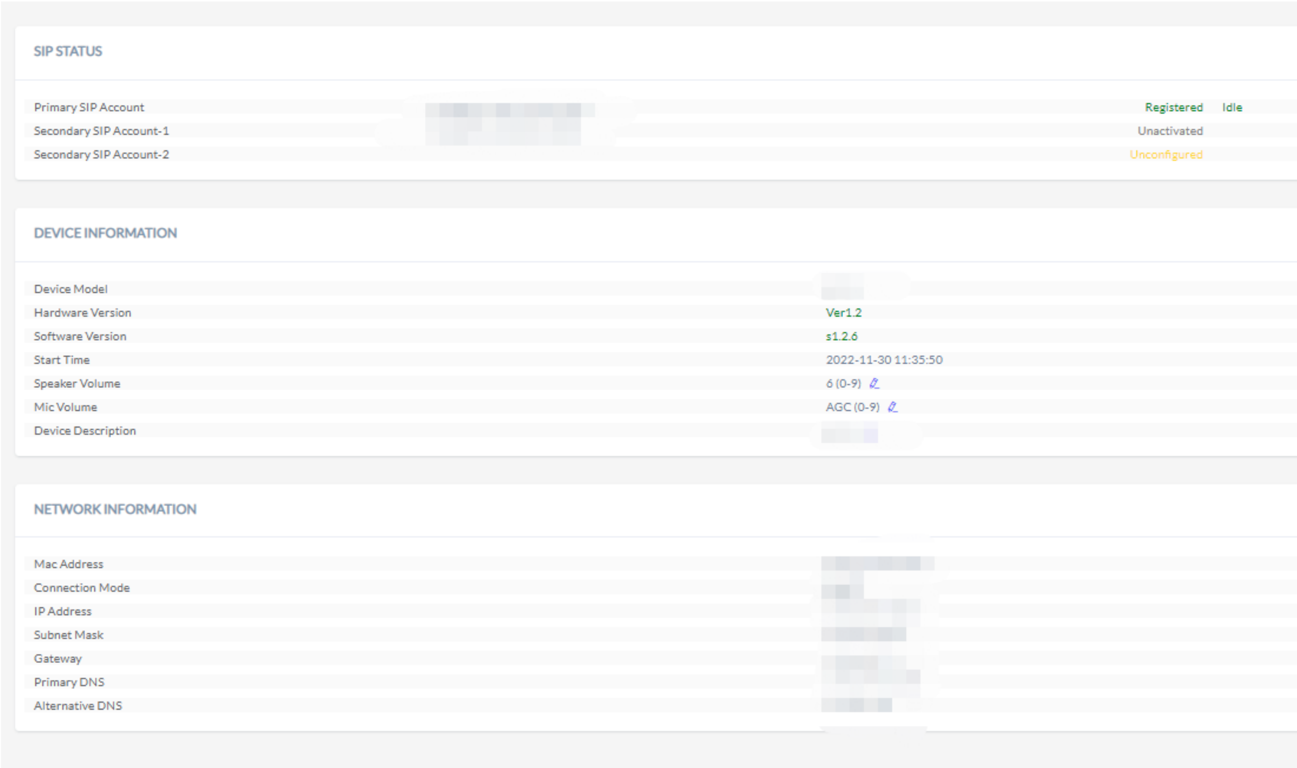
Default username: admin

Default password: admin

For the safety of the network speakers, it is recommended to change the default password on first login, please go to Settings -> Change Password page to change the password.

Device Info

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

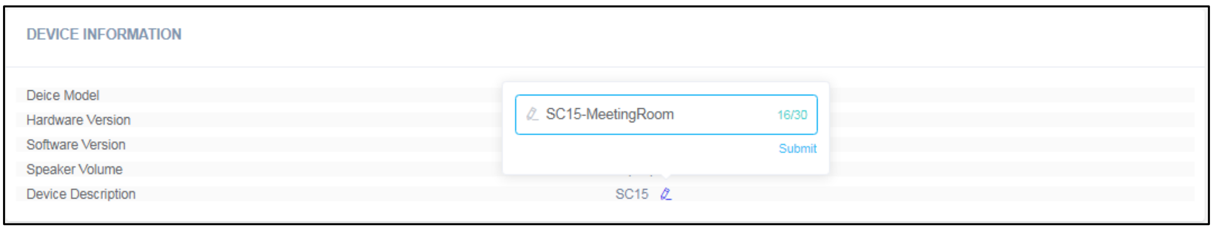


SIP STATUS

- SIP Account: The SIP number configured on this device.
- SIP Server: The SIP server (ZYCOO IP Audio Center or other IP PBX) address.
- Register Status: The SIP number registration status.

DEVICE INFORMATION

- Device Model: The speaker model, SC15, SW15 etc.
- Hardware Version: Speaker hardware version.
- Software Version: Speaker software version, can be upgraded.
- Uptime: Last startup time of the device.
- Speaker Volume: The current volume level of the speaker device.
- Mic Volume: The current volume level of the microphone.
- Device Description: The device description will be used to display as the tab name of the web browser. This is useful when configuring multiple speaker devices using the same web browser. Click on the button to edit.



After modification, the tab name will change.

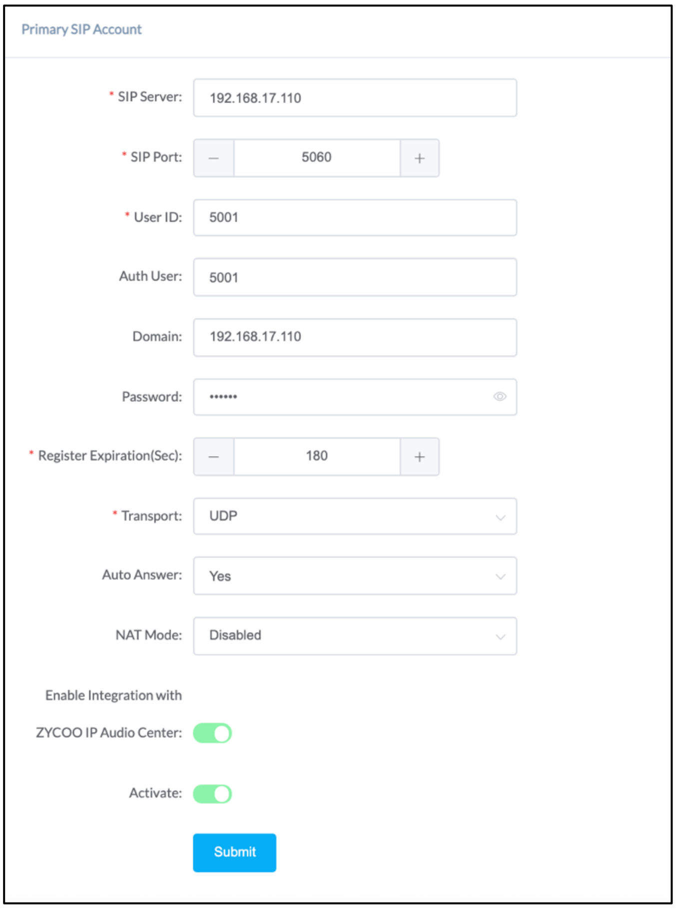
NETWORK INFORMATION

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

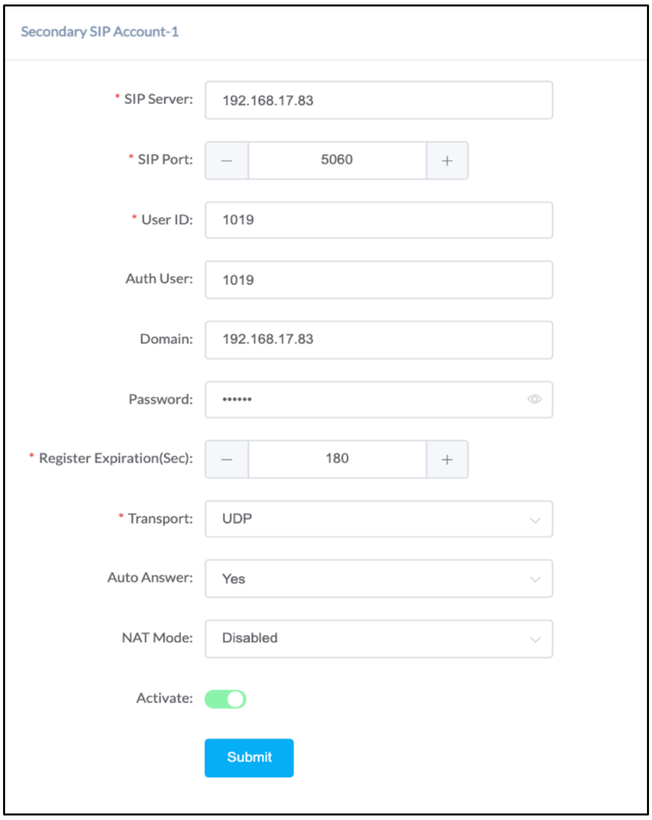
SIP Account

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

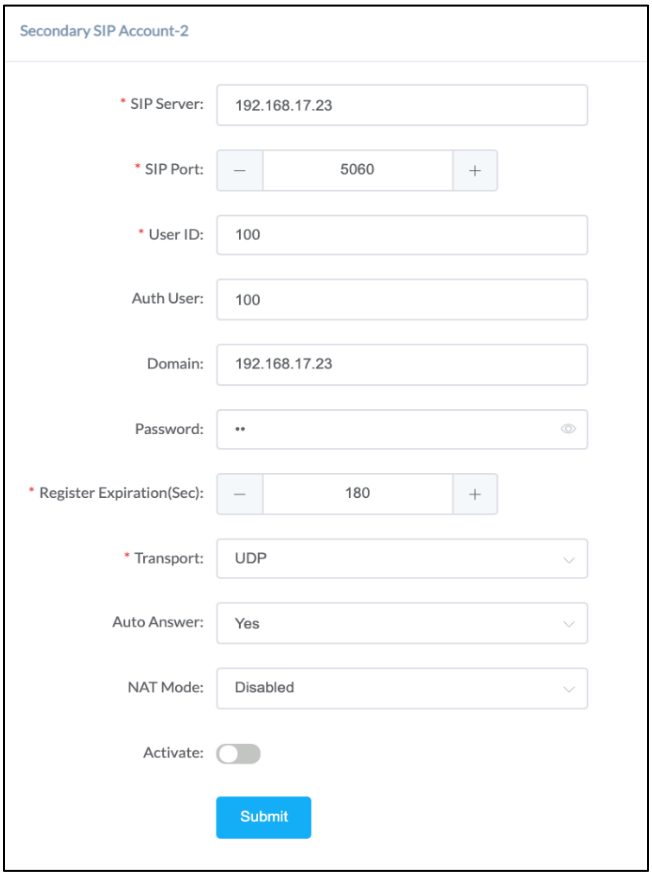
Primary SIP Account



Secondary SIP 1 Account



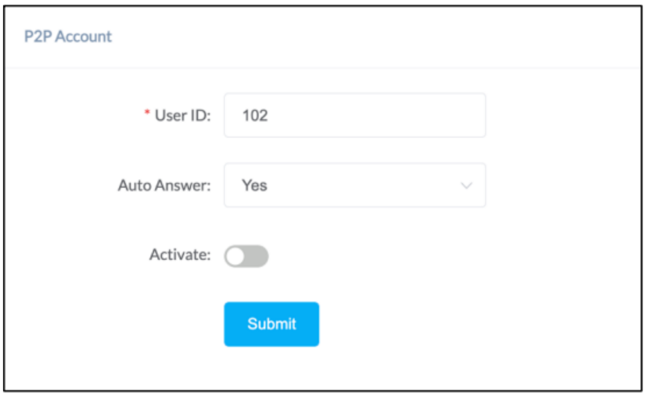
Secondary SIP 2 Account



- 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 Ye 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.
- SIP Auto answer: When the Auto Answer is in No or Answer Delay, you may still turn on the auto answer option through the SIP header detection.
- 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.

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.

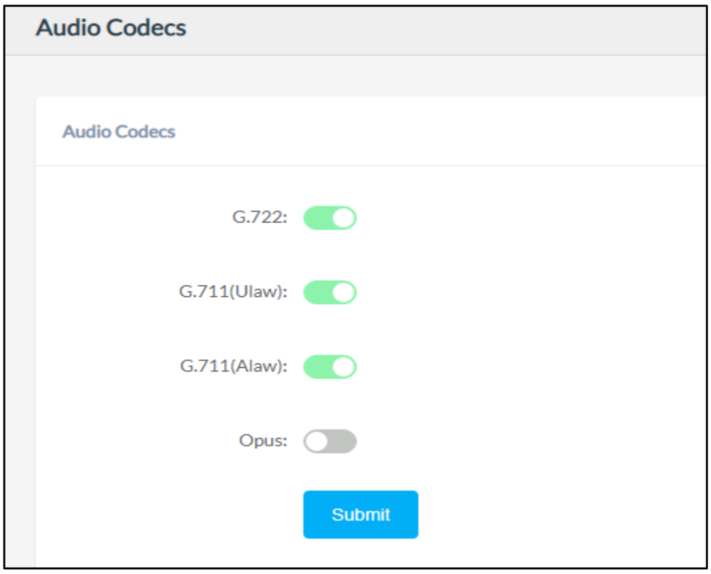


- User ID: The User ID will be displayed as the outgoing number when call out, or the number that other device needs to dial.
- Auto Answer: Yes/No/Answer Delay, default in Ye option.
- Ring Tone: When the Auto Answer is in No, you may choose the ring tone to play before the call is answered from this 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.

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.

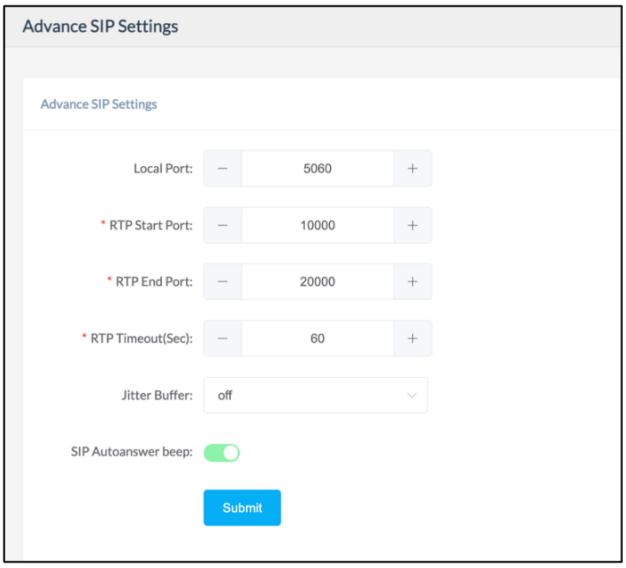


Please keep at least one codec enabled and is supported by the SIP server, otherwise SIP paging will not work.

Advance SIP Settings

Configuration on some more advance SIP protocol settings include the Loud Ringers feature.

Please go to SIP Settings -> Advance SIP Settings



Advance SIP Settings

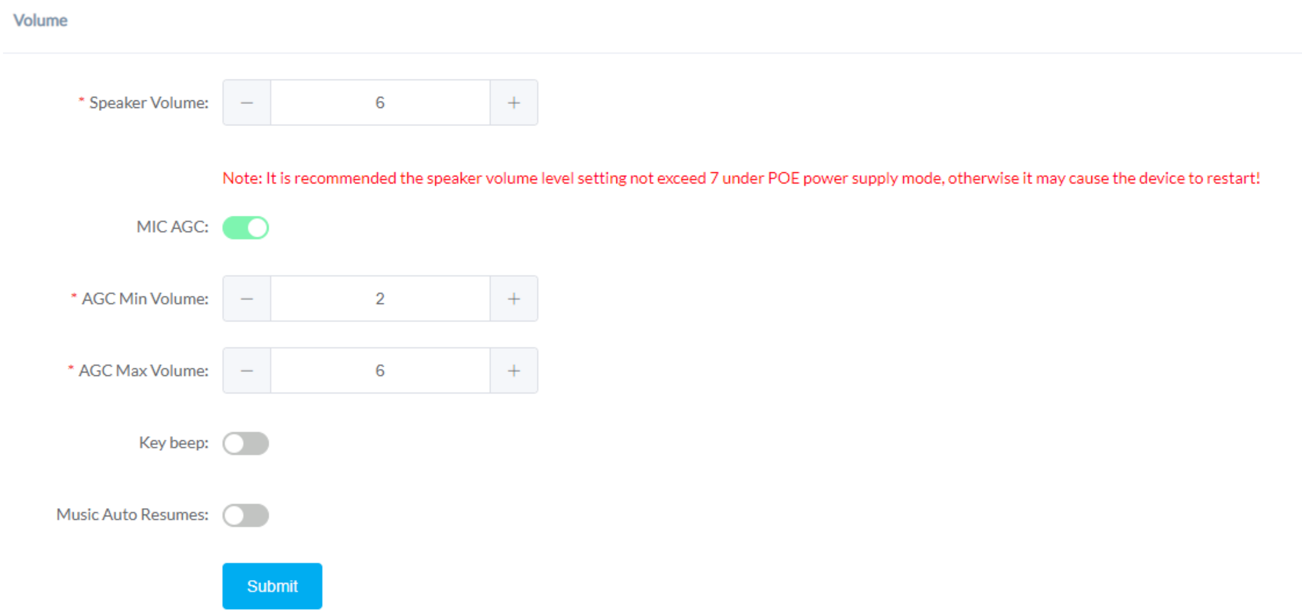
- Local Port: This setting represents the port that used to received SIP packets.
- RTP Start Port: This setting represents the starting RTP port that will use for media sessions.
- RTP End Port: This setting represents the end RTP port that the system will use for media sessions.

- RTP Timeout (sec): This setting represents in a specific time range, if the system doesn't receive the RTP stream, then the call will end.
- Jitter Buffer: This setting represents the Jitter buffer that 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.

Advanced Settings

Volume

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

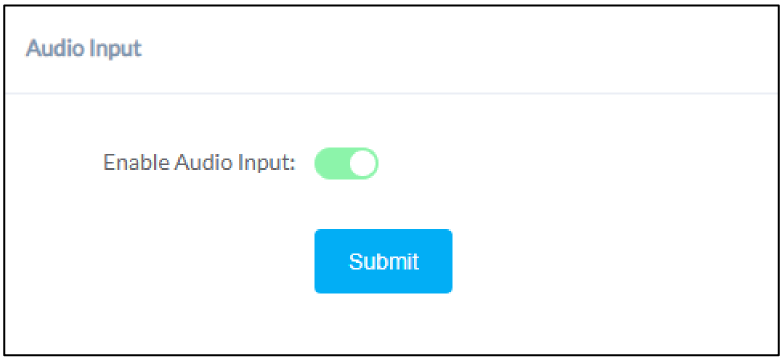


- Speaker Volume: The default speaker volume is 7, adjustable range is 0 ~ 9.

  (The below setting parameters are only available for hardware version 1.1 or above of SC15 & SW15.)

- 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.
- 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.
- Audio Input Volume: The default volume for input audio source is 7, adjustable range is 0 ~ 9.
- Key Beep: Enable/Disable the beep sound from key button.
- Music Auto Resumes: The previous music tasks will automatically resume when the device is restarted or reconnected to the network.

(Note: It is recommended the volume level setting not exceed 7 under POE power supply mode, otherwise it may cause the device to restart).
(For SL30 only)



- Audio Input: Enable/Disable Audio Input. When this opinion is enabled, the speaker will taking the external audio input source as well.

I/O Settings

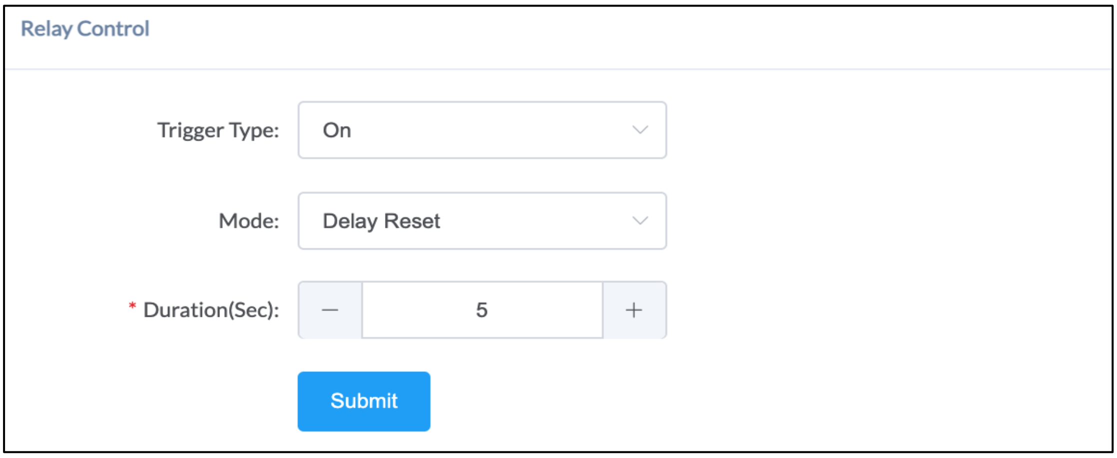
I/O settings (available for SC15, SW15 & SL30) are used to configure the dry contact relay control options. Please go to the Settings -> I/O Settings page.

Key Linkage

- Event Linkage Action: Select the event linkage action to select [Outgoing Call], [HTTP Request], [Play Audio].
- Destination: This setting represents the response device's number when the external button is press.
- 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, user can end the call or conversation by pressing the button again.
- HTTP URL: Configure the API URL address triggered by linkage.
- Audio File: Configure audio triggered by linkage.
- Repeat: Configure the number of audio repetitions triggered by linkage.

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 broadcast alarm on.
- 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 three option for the trigger call event:
  - If "Incoming", the dry contact relay output will be triggered when the SIP paging gateway gets an incoming SIP paging/intercom call.
  - If "Hang-up", the dry contact relay output will be triggered when a SIP call ends on the SIP paging gateway.

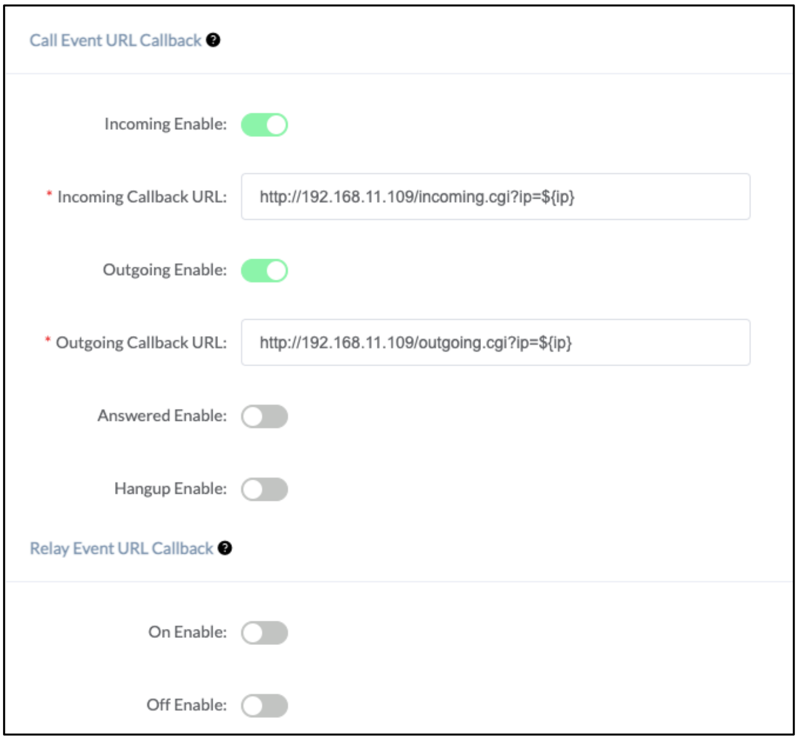


Relay Control

- **Trigger Type:** This setting represents the responds 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 are 'Delay Reset' and 'Hang-up Reset' options to choose from.
- **Duration (sec):** This setting is only available if reply control mode is on Delay Reset, it represents the time duration when the configure interface status changed.

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

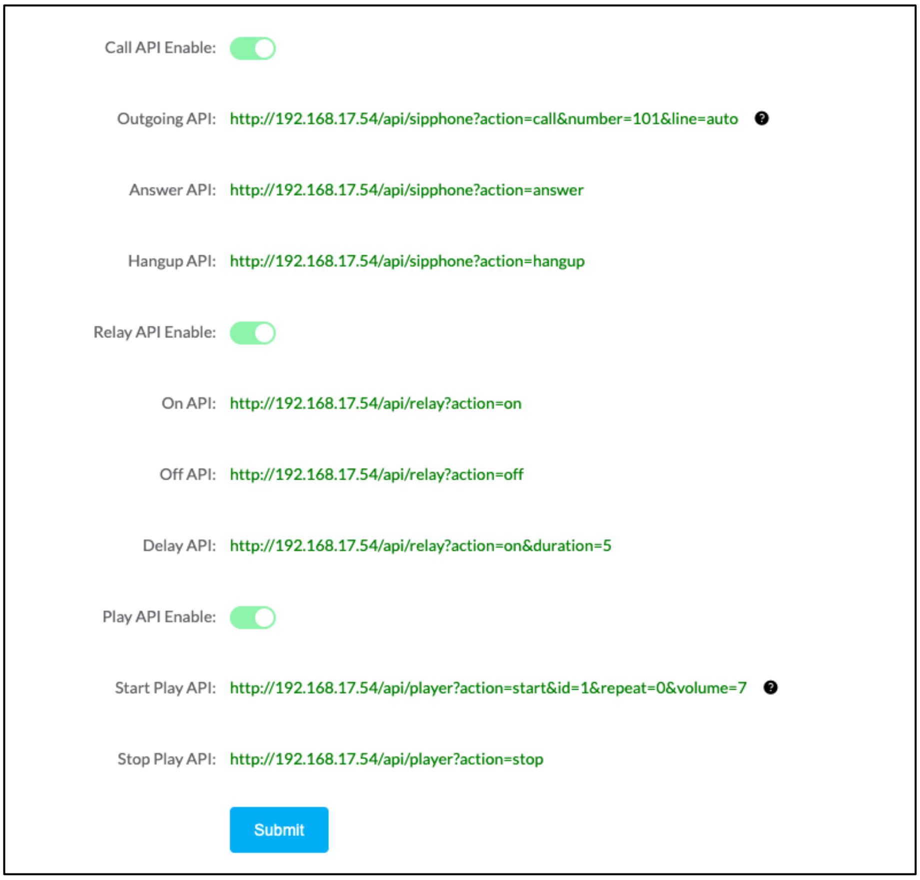
- **Relay Event URL Callback**

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, and play sound by the systems.

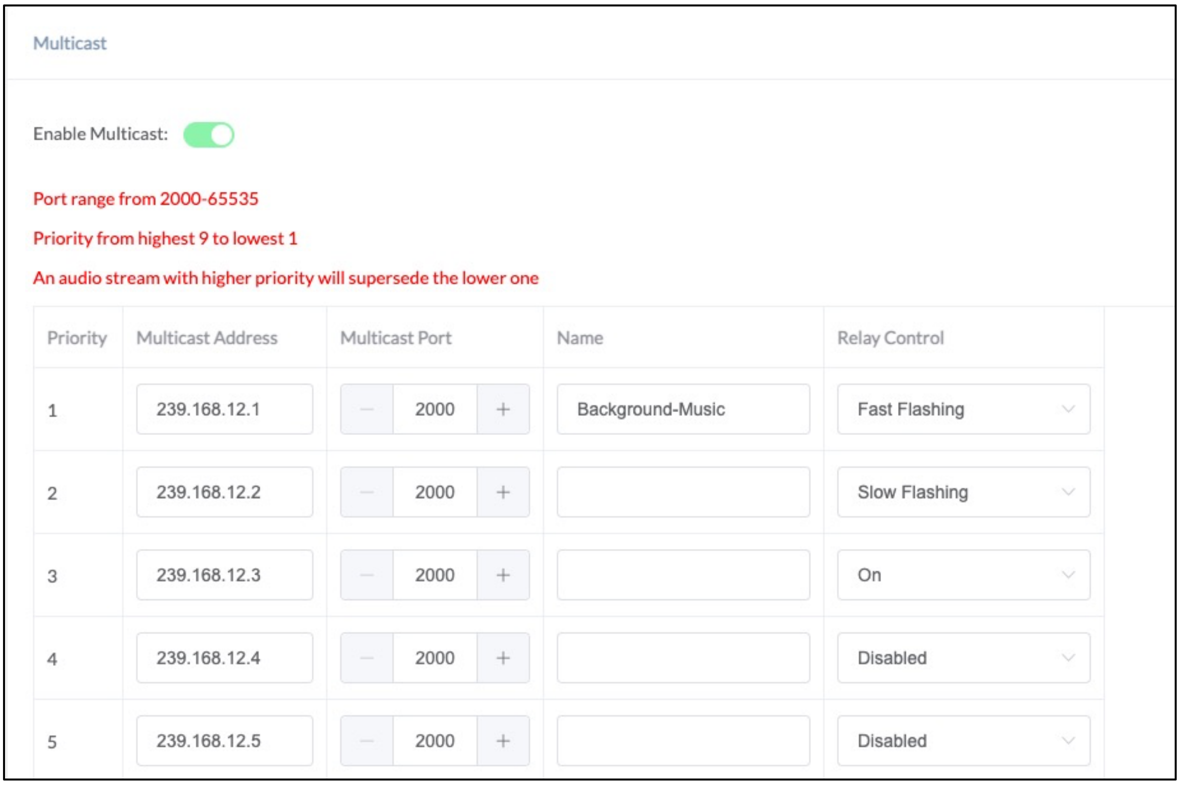


(Note: Authentication and encryption are not used in the API interface, so please pay attention the security of the network environment when opening and using these API interfaces).

Multicast

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.

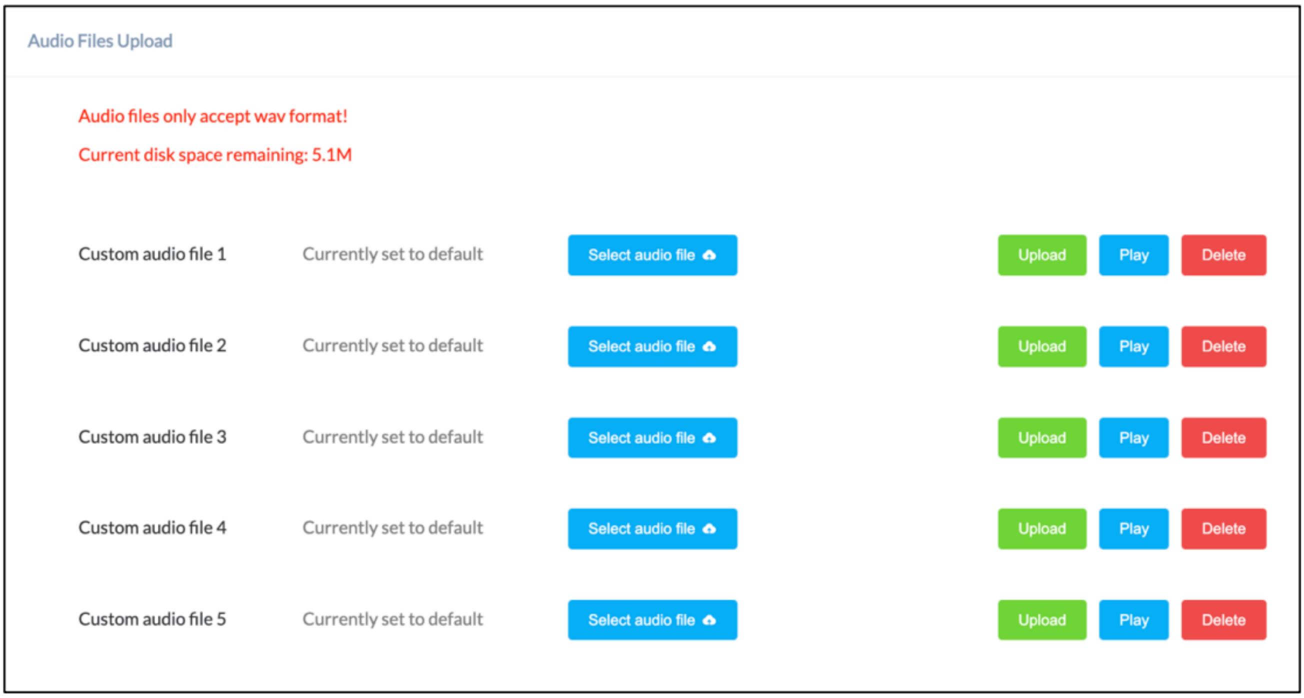
Prompt Language

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

Currently, only Chinese and English are provided.

Audio Files

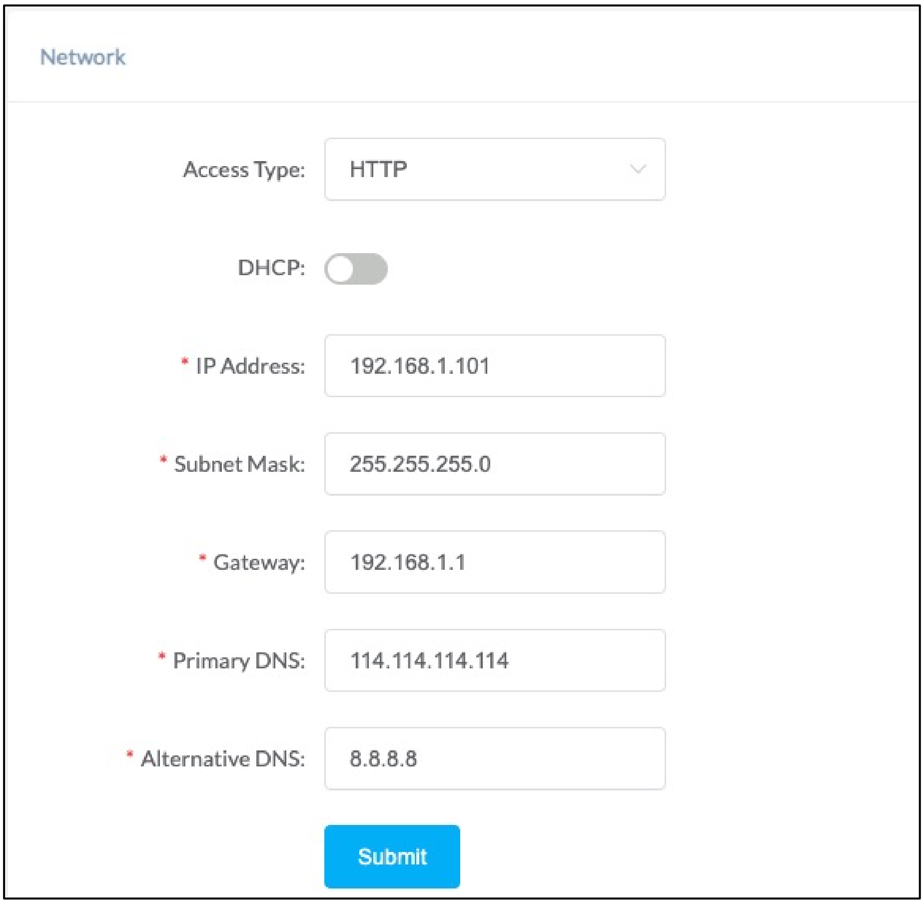
The Audio files section allows users to self-upload up to 5M of audio files to the endpoint and use it as ringtone or Play API audio file. Please click on the 'Select audio file' button to select and upload the local audio file, then click on the 'upload' button to upload it. Click on the 'play' to test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



System Settings

Network

The ZYCOO network speakers use DHCP to dynamically obtain IP addresses by default. To change the IP assignment from DHCP to Static IP, please go to Settings -> Network Settings page.



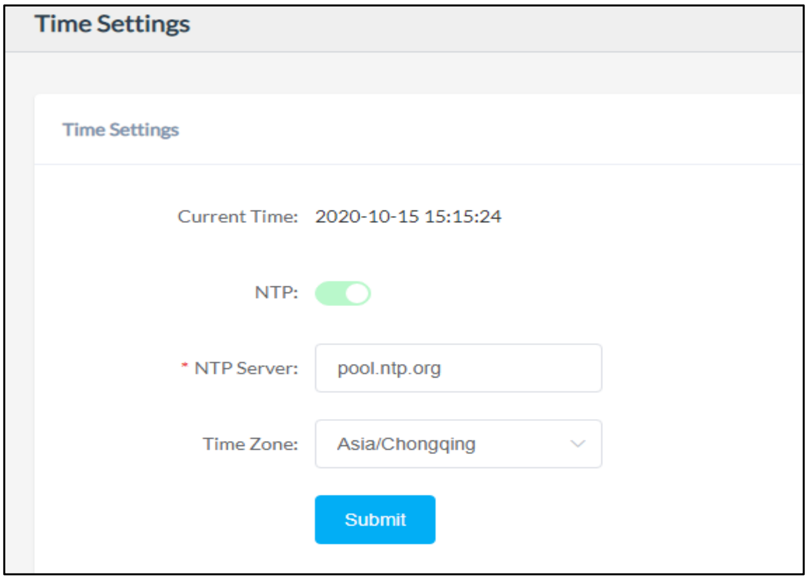
Turn the DHCP switch button off to show the network parameter settings.

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.
- Default 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.

Time

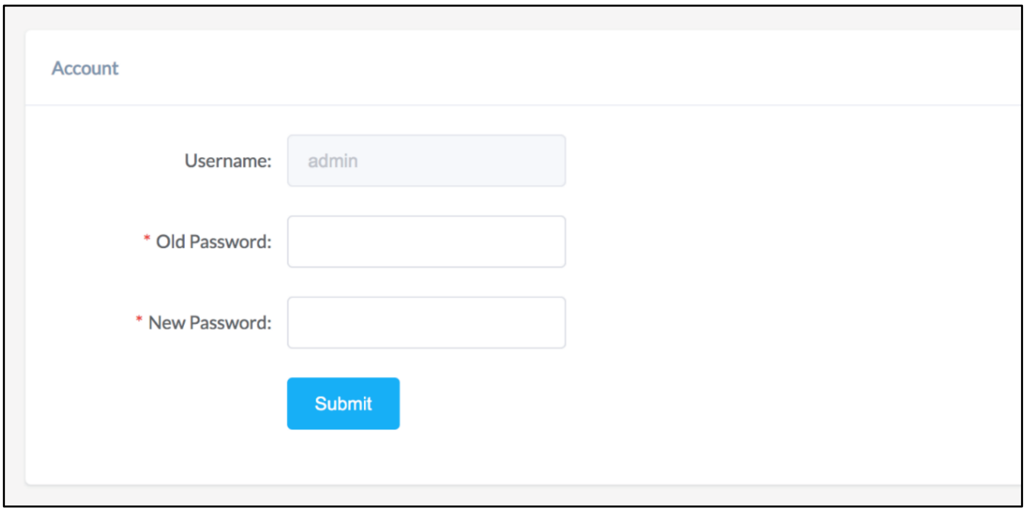
The network speakers obtain time from the network time servers using NTP, to change the NTP settings please go to Settings -> Time Settings page.



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.

Account

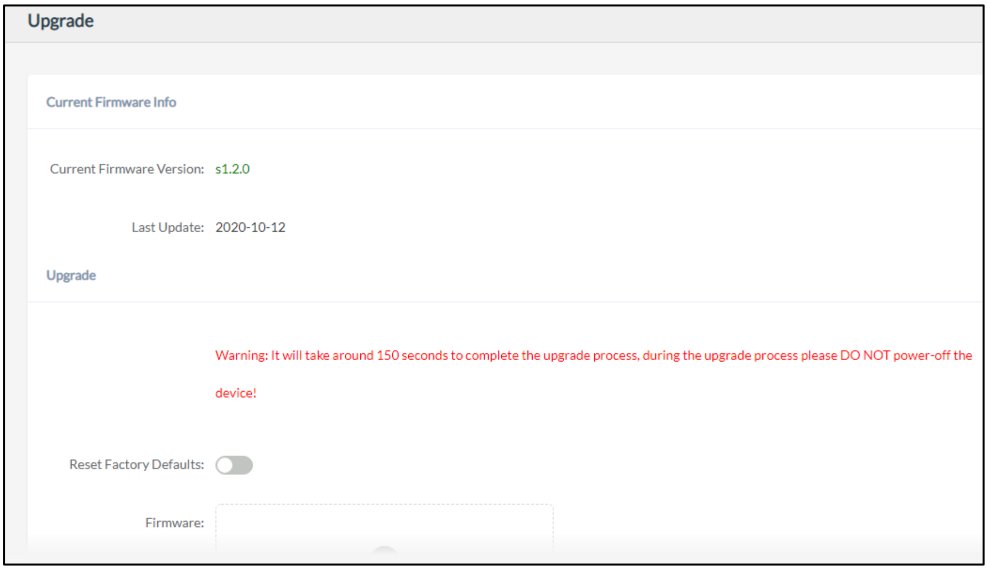
For resetting the current device's password, please go to Settings -> Account



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

Upgrade

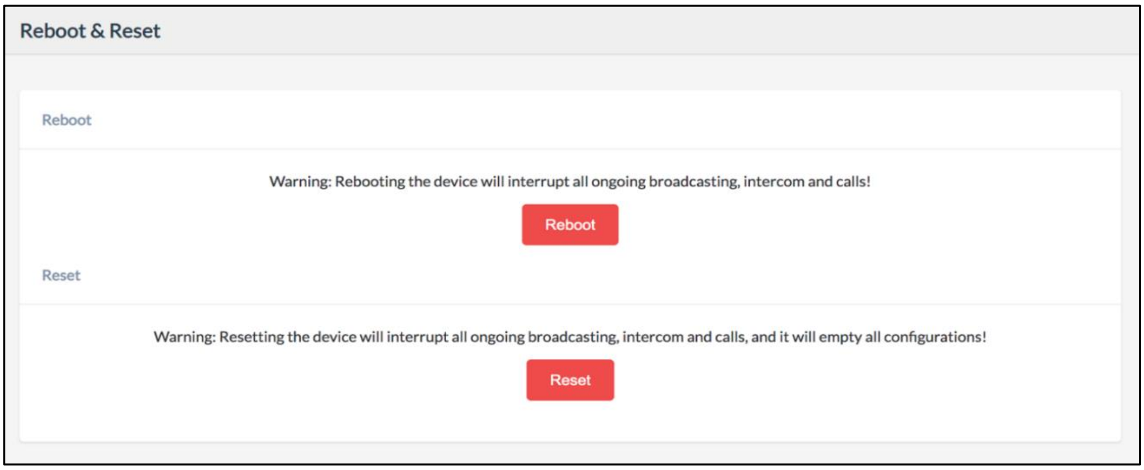
To upgrade the network speaker's firmware, please go to Settings -> Upgrade page.



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.

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.

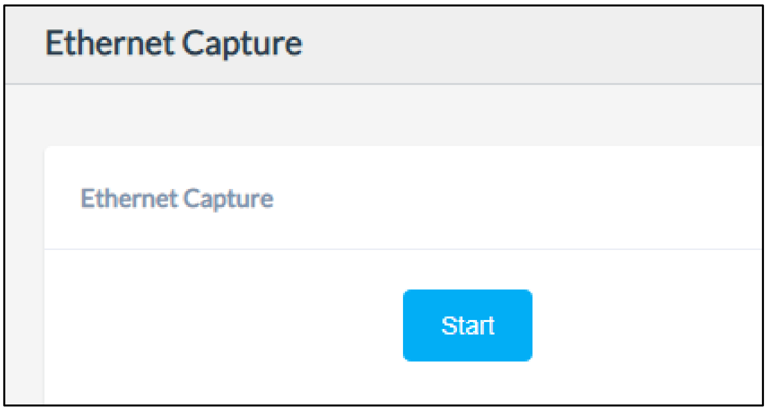
Maintenance Settings

Diagnostic

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

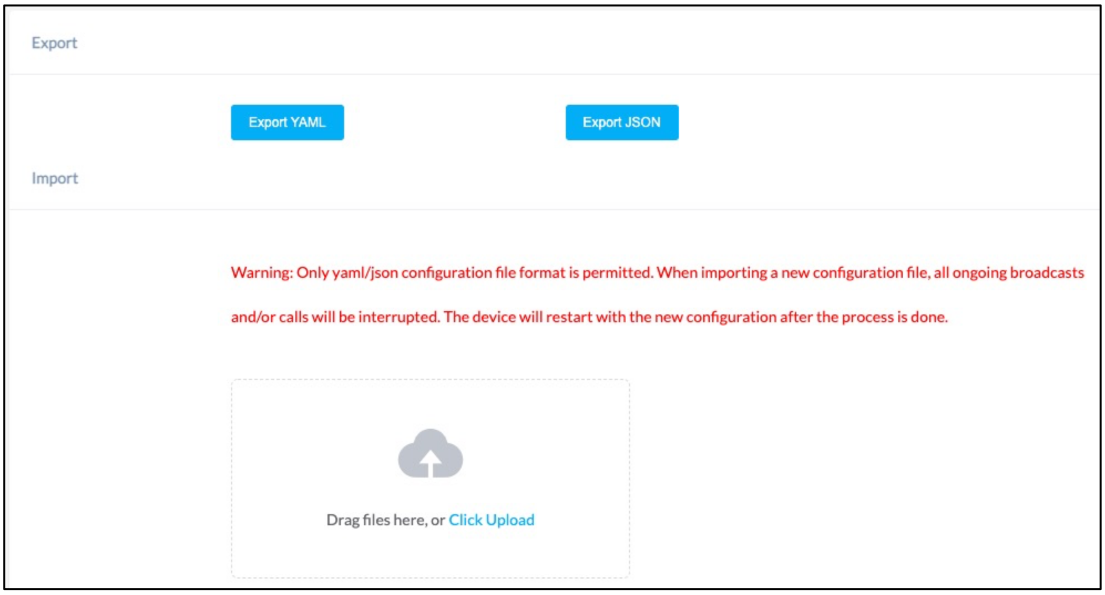
Ethernet Capture

The purpose of the Ethernet capture tool is to capture Ethernet network packets and store then in a standard Wireshark compatible packet capture ".pacp" file for immediate viewing and data analyzing.



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.



Auto Provisioning

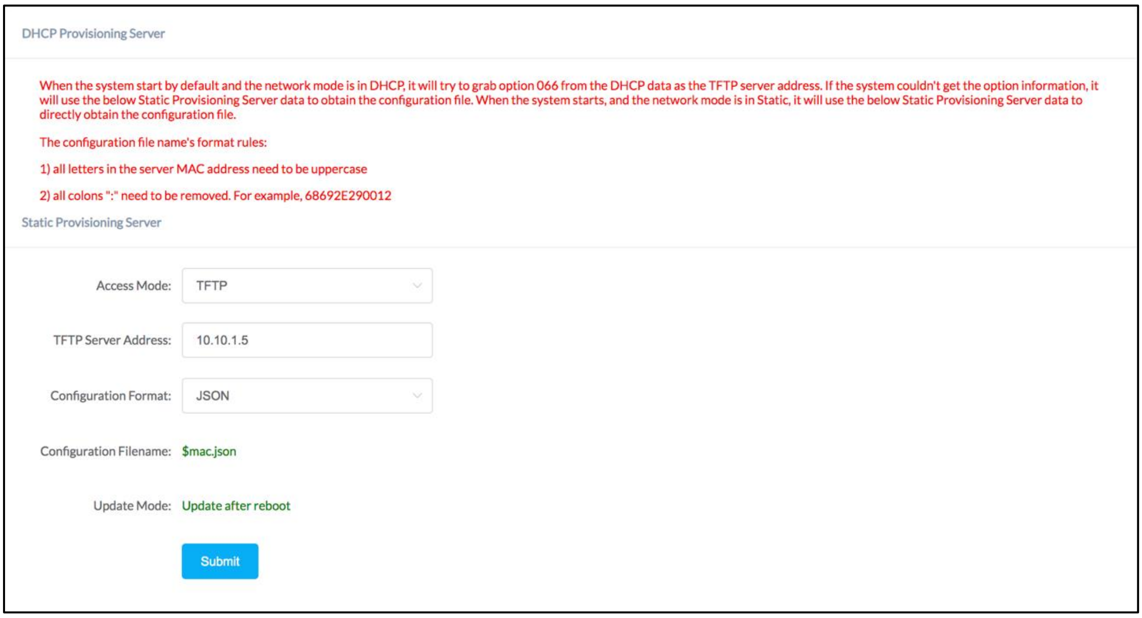
The system is supporting DHCP Option 066 and static TFTP/HTTP two auto provisioning methods.

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:

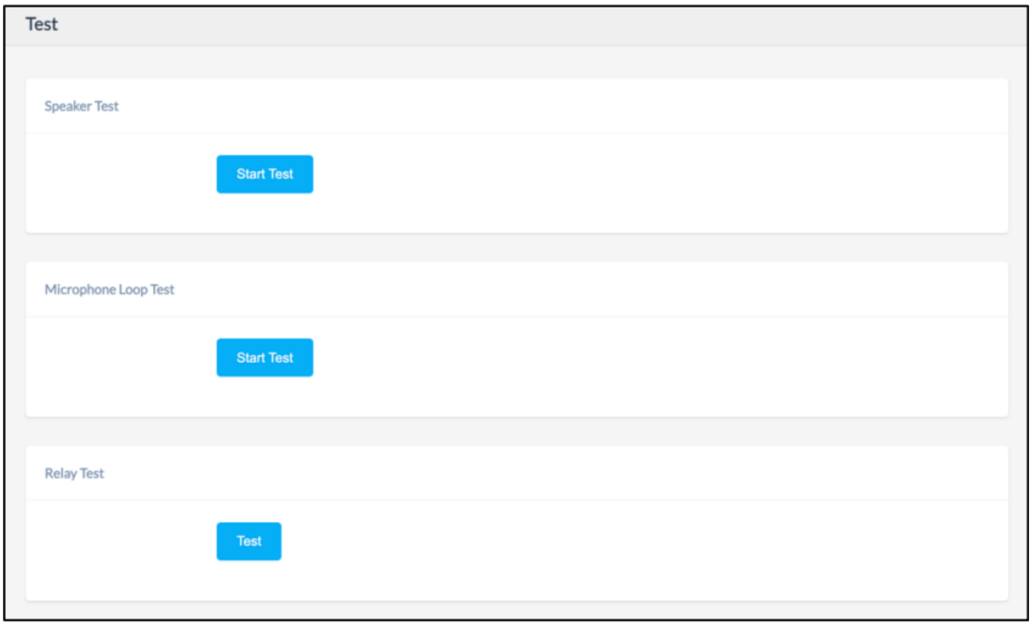
1) all letters in the server MAC address need to be uppercase.

2) all colons ":" need to be removed. For example, 68692E290012.



Test

The detection feature provides an option for user to check whether the speaker, microphone, and relay 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.

Relay Test: Click: On the Test button, then start using the relay device to test whether the device is working

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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.