

SIP Paging Gateway User Guide



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Overview

Product Overview

ZYCOO X10 SIP paging gateways are SIP enabled multifunctional IP audio devices dedicated for industry users. They can convert voice streams from a SIP paging system or IPPBX system to analogue sounds for background music, public address, intercom, etc. Except IP audio communications, the X10 SIP paging gateways can also be connected with sensors, sound-light alarm, door magnet and various peripherals for automated control and security applications.

Based on the compact hardware design, open standard SIP protocol support, rich functionality and high performance. Industry users can customize the X10 SIP paging gateway into any desired form of intercom device or paging speaker. These can then be widely used for smart and safe city applications, to increase the efficiency of communication and information sharing.

Product Specifications

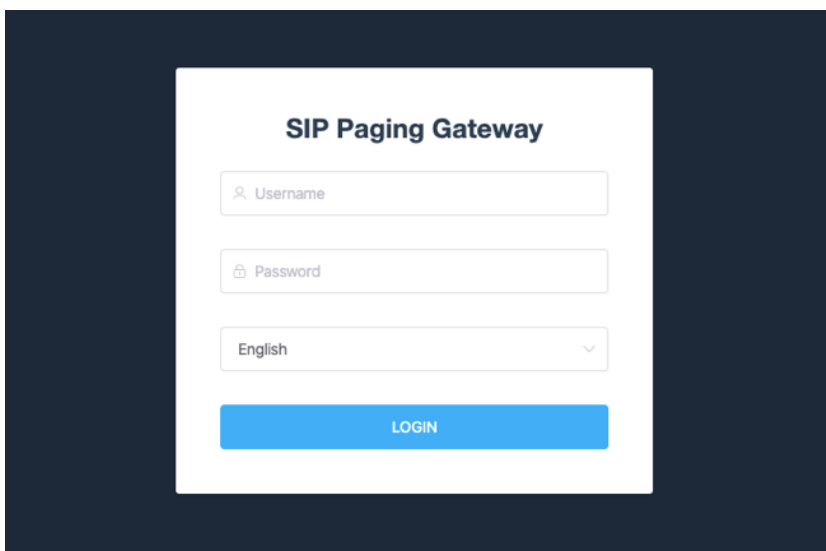
X10 Specifications:

- Audio Output: 2*10W, 8Ω SPK (4 pins) + 3.5mm audio jack
- Audio Input: 3.5mm audio jack
- Call Button: Support 2 switch buttons with LED indicators
- Protocols: SIP(RFC3261), HTTP, TCP/IP, SSL, DNS, SNTP, NTP, RTSP, RTP, RTCP, TCP, UDP, MQTT, ICMP, DHCP, ARP, SSH
- Audio Codecs: G.711(a, u), G.722, G.729
- Power Supply: PoE (IEEE802.3at) or DC 12V-3A
- Relay Switch: Max voltage AC 125V-1A/DC 60V-1A
- Network: ETH0+ETH1 10/100Mbps
- Working Temperature: -20°C ~ +50°C

Basic Settings

Web Interface Login

By default, the SIP paging gateway devices' IP assignment has been configured as DHCP. Please ensure there's DHCP server available in the LAN where the SIP paging gateway devices 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 VOL+ and VOL- at the same time for 5 seconds, the device will announce its IP address. Input the IP address in the browser address bar to open the web management interface of the SIP paging gateway. The login screen is shown as below image.

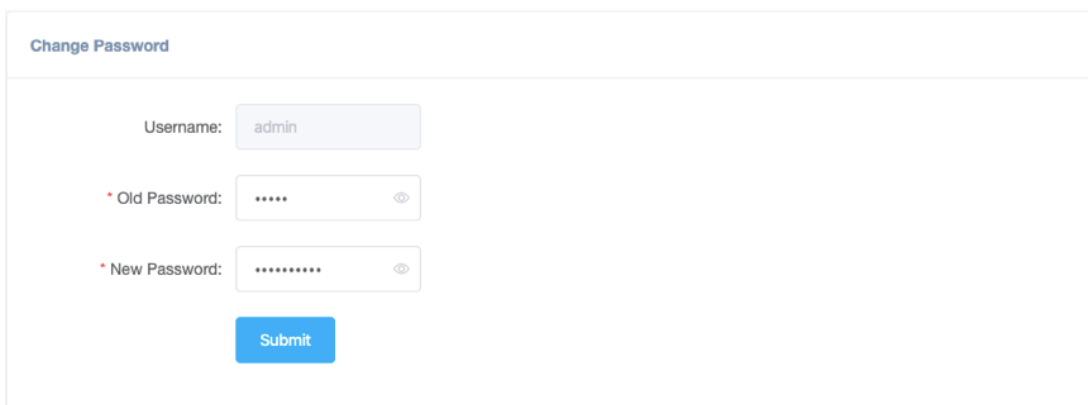


Use the default login credentials to login to the web management interface.

Default username: admin

Default password: admin

For the safety of the SIP paging gateway, 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		
Primary SIP Account	5010@192.168.12.205:5060	Registered
Secondary SIP Account-1	1011@192.168.12.83:5060	Registered
Secondary SIP Account-2	1012@192.168.12.83:5060	Registered

DEVICE INFORMATION	
Device Model	X10
Hardware Version	Ver1.0
Software Version	s1.2.0beta2
Speaker Volume	4 (0-9) ↗
Device Description	X10 ↗

NETWORK INFORMATION	
Mac Address	68:69:2E:29:00:8F
IP Assignment	DHCP
IP Address	192.168.12.157
Subnet Mask	255.255.255.0
Default Gateway	192.168.12.1
Primary DNS	114.114.114.114
Alternative DNS	8.8.8.8

SIP STATUS

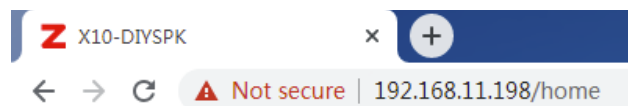
- **SIP Account:** The SIP number configured on this device.
- **SIP Server:** The SIP server (ZYCOO IP Audio Center or other IP PBX) address.
- **Register Status:** The SIP number registration status.

DEVICE INFORMATION

- **Device Model:** The device model, X10 or X30.

- **Hardware Version:** Device hardware version.
- **Software Version:** Device software version, can be upgraded.
- **Speaker Volume:** The current volume level of the device.
- **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.
- **IP Assignment:** 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.
- **Default 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.

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

Primary SIP Account

* SIP Server:

* SIP Port:

* User ID:

Auth User:

Password:

* Register Expiration(Sec):

* Transport:

Enable Integration with
ZYCOO IP Audio Center:

Activate:

Secondary SIP 1 Account

Secondary SIP Account-1

* SIP Server:

* SIP Port:

* User ID:

Auth User:

Password:

* Register Expiration(Sec):

* Transport:

Activate:

Secondary SIP 2 Account

Secondary SIP Account-2

* SIP Server:

* SIP Port:

* User ID:

Auth User:

Password:

* Register Expiration(Sec):

* Transport:

Activate:

- 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.
- 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.
- 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.

The screenshot displays a web interface for configuring a P2P account. At the top, there is a header 'P2P Account'. Below it, a form contains the following elements:

- A label '* User ID:' followed by a text input field containing the placeholder text 'eg: 100'.
- A label 'Activate:' followed by a green toggle switch that is currently turned on.
- A blue rectangular button labeled 'Submit' positioned below the toggle switch.

- User ID: The User ID will be displayed as the outgoing number when call out, or the number that other device need to dial.
- 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.

Please go to 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 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.

Advanced Settings

Volume Control

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

The screenshot shows a web interface for 'Volume Control'. It features three adjustable settings: 'Speaker Volume' (set to 3), 'MIC AGC' (a toggle switch currently turned off), and 'Microphone Volume' (set to 7). Each setting has minus and plus buttons for adjustment. A blue 'Submit' button is located at the bottom of the form.

- Speaker Volume: The default speaker volume is 7, adjustable range is 0 ~ 9.
- Microphone AGC: AGC stands for Automatic Gain Control, it is a unique circuit that listen to the incoming audio level and adjusts the recording level when sounds re too loud, or too soft.
- Microphone Volume: The default microphone volume is 7, adjustable range is 0 ~ 9.

I/O Settings

I/O settings are used to configure the press-to-talk, digital level input and trigger settings, relay control options. Please go to the Advanced -> I/O Settings page.

For press-to-talk (or push-to talk), any desired number can be setup here. As a result, when the call button has been pressed, the SIP safety intercom will try to call this number.

I/O Settings

Press-to-Talk

Press-to-Talk Number1: Line: ▾

Press-to-Talk Number2: Line: ▾

Press Again to End Call:

Input Settings

Input Detection:

Trigger Mode: ▾

Dial Number:

* Number: Line: ▾

Press-to-Talk

- **Press-to-Talk Number-1:** This setting represents the number will be dialed by using Key 1.
- **Press-to-Talk Number-2:** This setting represents the number will be dialed by using Key 2.
- **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.

Input Settings

Per the specification if the sensor, select from “Falling Edge” or “Rising Edge” as the trigger, and then setup the number to dial, so, when the sensor sends the corresponding digital level signal to the SIP paging gateway device, it will dial the given number.

- **Input Detection:** Enable/Disable, enable this option to detect any digital level signals.
- **Trigger Mode:** There are “Falling Edge” and “Rising Edge” options for the trigger.
- **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 need to dial
- **Line:** Line represents the line that will be using to dial the call.

Trigger Settings

To use digital level input signal as the trigger of dry contact relay output, please enable “Trigger by Input Signal”.

To use DTMF signal as the trigger of dry contact relay output, then please specify the DTMF key press.

The screenshot displays the 'I/O Settings' configuration page. It is divided into two main sections: 'I/O Settings' and 'Output Settings'.
In the 'I/O Settings' section, there are three toggle switches, all of which are turned on (green):
1. 'Trigger by Input Signal': Enabled.
2. 'Trigger by DTMF Signal': Enabled.
3. 'Trigger by Call Status': Enabled.
Below the 'Trigger by DTMF Signal' toggle, there is a text input field labeled '* DTMF:' containing the value '1*'.
Below the 'Trigger by Call Status' toggle, there is a dropdown menu labeled 'Event:' with 'Incoming' selected.
The 'Output Settings' section is located below a horizontal line and contains three controls:
1. 'Trigger Type': A dropdown menu with 'On' selected.
2. 'Mode': A dropdown menu with 'Delay Reset' selected.
3. '* Duration:': A numeric input field with '5' in the center, flanked by minus and plus buttons.

Trigger Settings

- Trigger by Input Signal: Enable/Disable, enable this option when need to use Input signal to trigger
- 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:
 - 1) If “Outgoing”, the dry contact relay output will be triggered when someone presses the call button of the SIP paging gateway to establish a paging/intercom call.
 - If “Incoming”, the dry contact relay output will be triggered when the SIP paging gateway gets an incoming SIP paging/intercom call.
 - If “Hangup”, 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 is only available if the 'Trigger by Input Signal' is disable and it represents the reset mode after the trigger is responded, there are 'Delay Reset', 'Answered 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.

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.

Multicast Settings

Enable Multicast

Port range can be from 2000-65535

Priority 9 is the highest and 1 is the lowest

A higher priority audio stream will always supersede a lower one

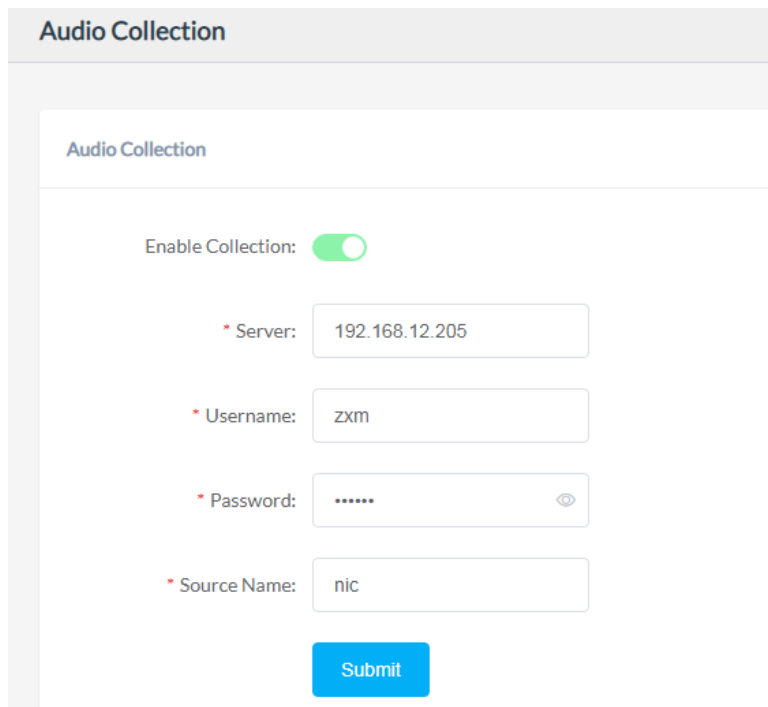
Priority	Multicast Address	Multicast Port	Name	GPIO Trigger
1	<input type="text" value="239.168.12.1"/>	<input type="text" value="2000"/>	<input type="text" value="Background-Music"/>	<input type="text" value="Disabled"/>
2	<input type="text" value="239.168.21.2"/>	<input type="text" value="2500"/>	<input type="text" value="2"/>	<input type="text" value="On"/>
3	<input type="text" value="239.168.21.3"/>	<input type="text" value="3000"/>	<input type="text" value="3"/>	<input type="text" value="Fast Flashing"/>
4	<input type="text" value="239.168.21.4"/>	<input type="text" value="3400"/>	<input type="text" value="4"/>	<input type="text" value="Slow Flashing"/>
5	<input type="text" value="239.168.21.5"/>	<input type="text" value="3500"/>	<input type="text" value="5"/>	<input type="text" value="Disabled"/>

- **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
- **GPIO Trigger:** Options to choose from are 'Disabled', 'On', 'Fast Flashing', 'Slow Flashing'

Audio Collection

The Audio Collection is used to configure the parameter settings of the audio source collection function on the SIP Paging Gateway. The Audio Collection can collect external audio sources through the 3.5 Audio in the interface of the model X10 and play the audio source through the paging endpoints. This feature is required to use with the ZYCOO IP Audio Center.

Please go to Advance Settings -> Audio Collection



Audio Collection

Audio Collection

Enable Collection:

* Server:

* Username:

* Password:

* Source Name:

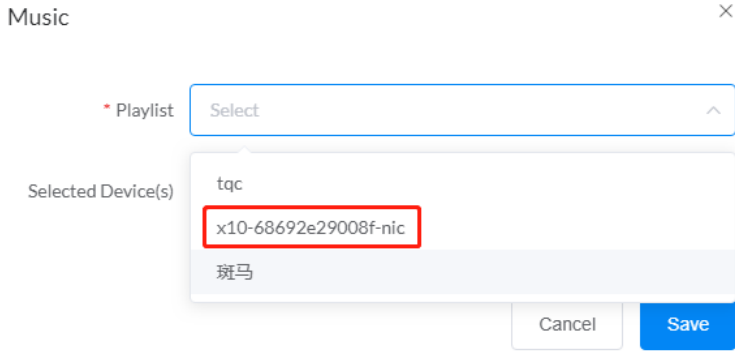
Submit

- Enable Collection: Enable the audio collection feature
- Server: ZYCOO IP Audio Center's server address
- Username: The account username used in the ZYCOO IP Audio Center
- Password: The account password used in the ZYCOO IP Audio Center
- Source Name: Customize a name for the audio source (alphabet and number only)

After the relevant parameters are configured correctly, the audio source name will appear in the Playlist on the IP Audio Dispatch Console.

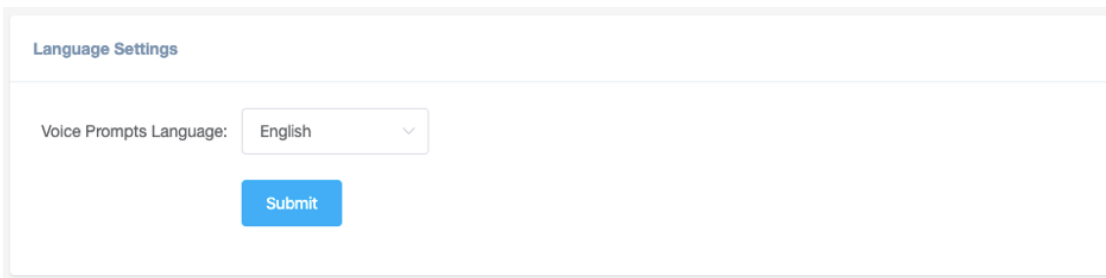
Name format: x10-<MAC>-<Source Name>

For example:



Language Settings

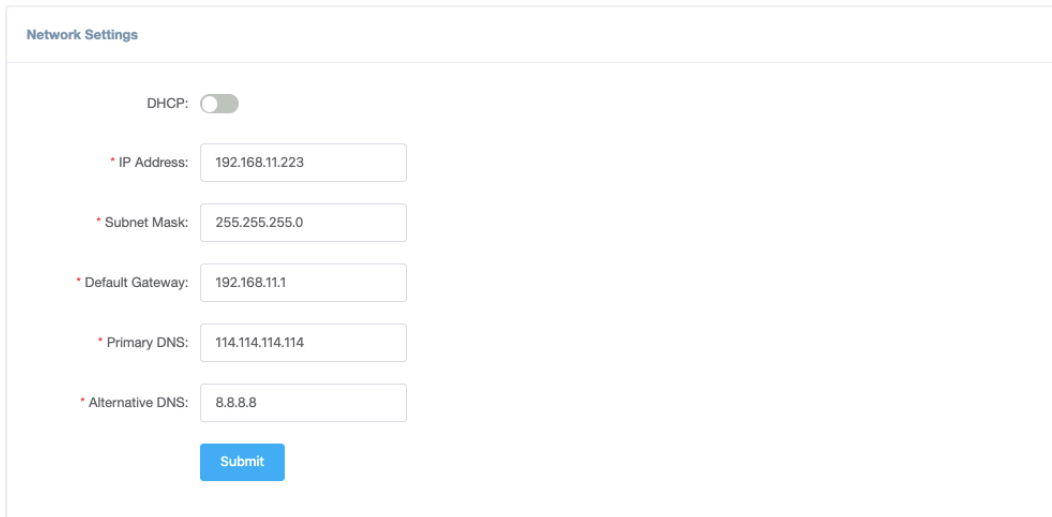
The language of local voice prompts, like IP address announcements, can be set on Settings -> Language Settings page.



Settings

Network Setting

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



The screenshot shows the 'Network Settings' interface. At the top, there is a 'DHCP' toggle switch that is turned off. Below this, there are five input fields, each with a red asterisk indicating a required field. The fields are: 'IP Address' with the value '192.168.11.223', 'Subnet Mask' with '255.255.255.0', 'Default Gateway' with '192.168.11.1', 'Primary DNS' with '114.114.114.114', and 'Alternative DNS' with '8.8.8.8'. At the bottom of the form is a blue 'Submit' button.

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

Network Configuration Parameters

- **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 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.

Time Settings

Time Settings

Current Time: 2020-10-23 14:45:23

NTP:

* NTP Server:

Time Zone:

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.

Upgrade

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

Upgrade

Current Firmware Info

Current Firmware Version: s1.2.0

Last Update: 2020-10-23

Upgrade

Warning: It will take around 150 seconds to complete the upgrade process, during the upgrade process please DO NOT power-off the device!

Reset Factory Defaults:

Firmware:

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.

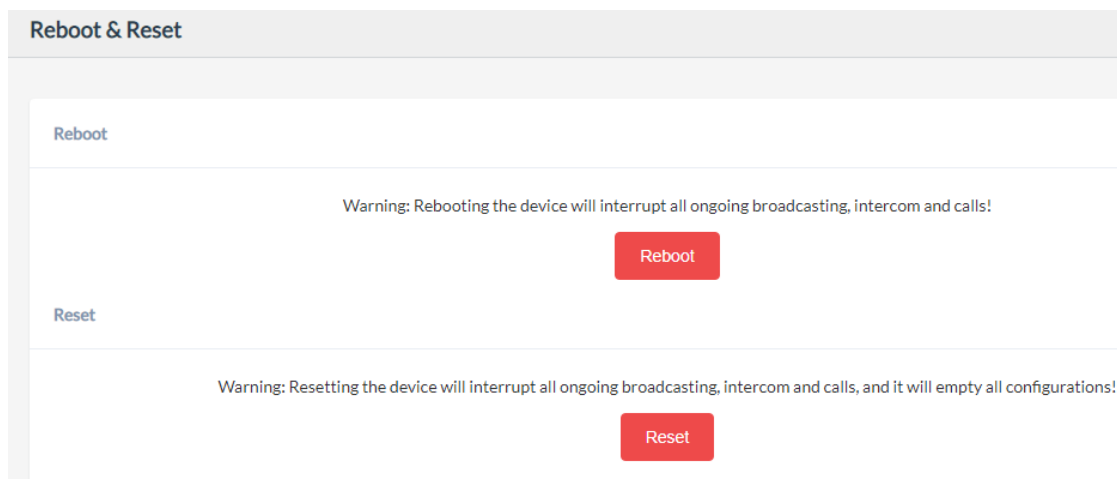
It will take around 2 minutes to complete the firmware upgrade, during upgrading process please DO NOT power off the network speaker.

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.



Maintenance

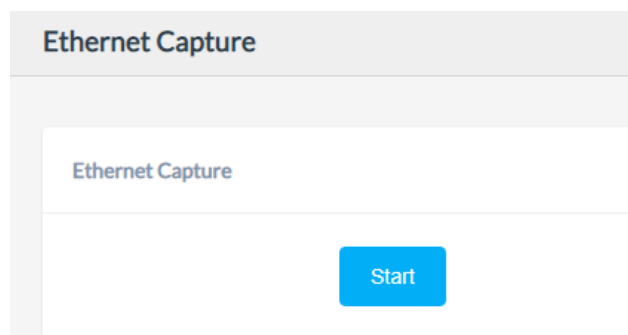
Diagnostic

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 them in a standard Wireshark compatible packet capture ".pacp" file for immediate viewing and data analyzing.



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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.

