

SIP Paging Gateway User Guide



Contents

Overview.....	1
Product Overview	1
Product Specifications	1
Basic Settings.....	2
Web Interface Login.....	2
Device Info	3
SIP Account	5
P2P Account.....	7
Audio Codecs	8
Advance SIP Settings.....	8
Advanced Settings.....	9
Volume Control	9
I/O Settings.....	10
API Settings	12
Multicast Settings.....	14
Audio Collection	14
Language Settings	16
Settings.....	17
Network Setting	17
Time Settings	17
Upgrade.....	18
Reboot & Reset.....	19
Maintenance	20
Diagnostic.....	20
Ethernet Capture	20

Import/Export 21
Auto Provisioning 21

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

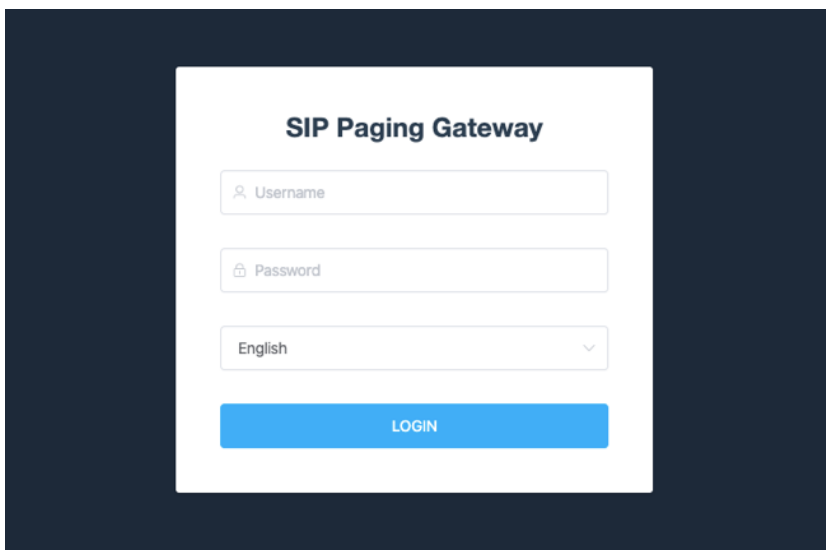
X10 Specifications:

- 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.

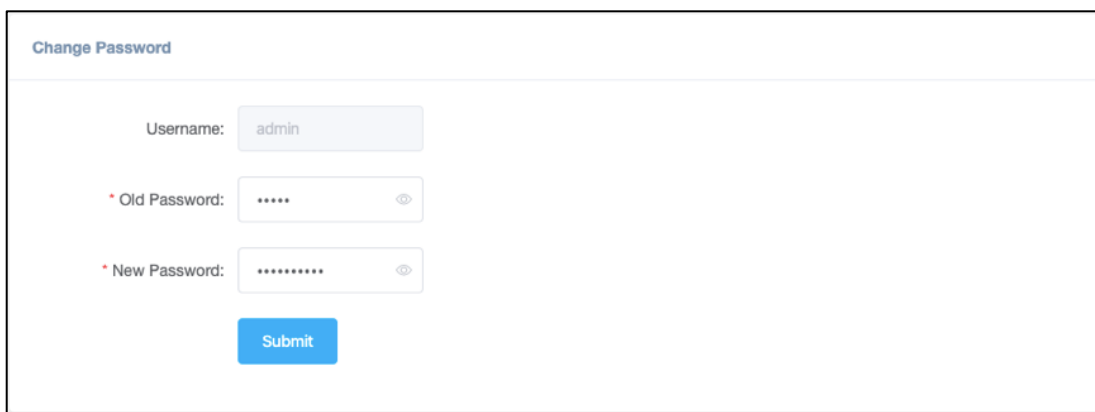


Use the default login credentials to login to the web management interface.

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	5009@192.168.17.110:5060	Registered	Idle
Secondary SIP Account-1	1012@192.168.17.83:5060	Registered	Idle
Secondary SIP Account-2		Unconfigured	

DEVICE INFORMATION	
Device Model	X10
Hardware Version	Ver1.1
Software Version	s1.2.4-dev7
Uptime	2021-12-29 13:44:03
Speaker Volume	6 (0-9) ↗
Mic Volume	AGC (0-9) ↗
Device Description	X10 ↗


NETWORK INFORMATION	
Mac Address	68:69:2E:29:0E:1F
Connection Mode	DHCP
IP Address	192.168.17.72
Subnet Mask	255.255.255.0
Gateway	192.168.17.1
Primary DNS	114.114.114.114
Alternative DNS	61.139.2.69


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.
- **Uptime:** Last startup time of the device.
- **Speaker Volume:** The current volume level of the 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 devices using the same web browser. Click on the  button to edit.

DEVICE INFORMATION	
Deice Model	X10-DIYSPK 10/30
Hardware Version	
Software Version	
Speaker Volume	
Device Description	X10 

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:

Domain:

Password:

* Register Expiration(Sec):

* Transport:

Auto Answer:

NAT Mode:

Enable Integration with
ZYCOO IP Audio Center:

Activate:

Secondary SIP 1 Account

Secondary SIP Account-1

* SIP Server:

* SIP Port:

* User ID:

Auth User:

Domain:

Password:

* Register Expiration(Sec):

* Transport:

Auto Answer:

NAT Mode:

Activate:

Secondary SIP 2 Account

Secondary SIP Account-2

* SIP Server:

* SIP Port:

* User ID:

Auth User:

Domain:

Password:

* Register Expiration(Sec):

* Transport:

Auto Answer:

NAT Mode:

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.
- 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.
- 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.

The screenshot shows a web form titled "P2P Account". It contains the following elements:

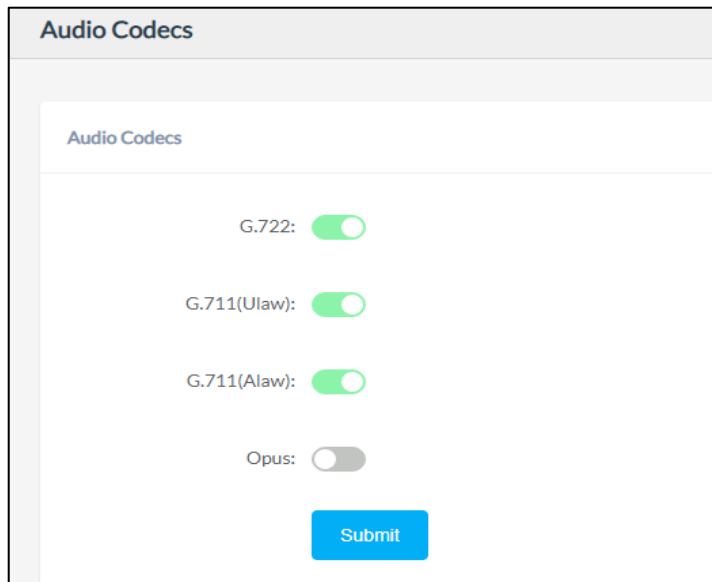
- A text input field for "User ID" with a red asterisk indicating it is required and a placeholder "eg: 100".
- A toggle switch for "Allow Anonymous Call" which is currently turned off.
- A dropdown menu for "Auto Answer" with "Yes" selected.
- A toggle switch for "Activate" which is currently turned off.
- A blue "Submit" button at the bottom center.

- 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 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.



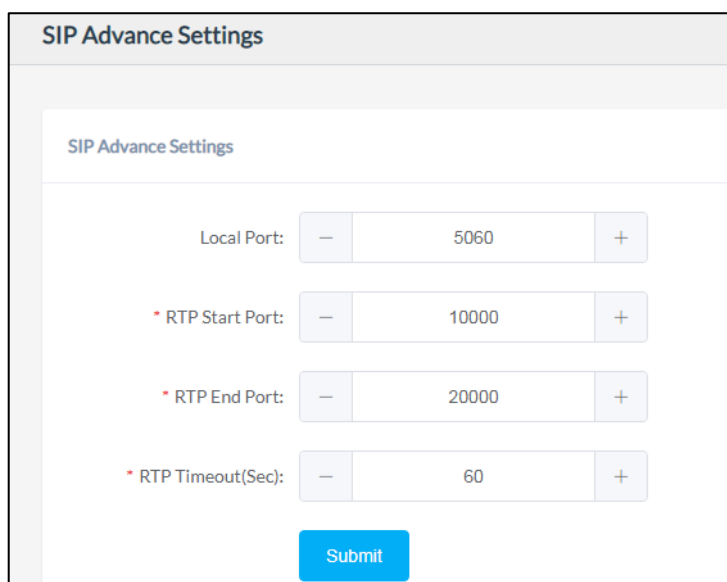
The screenshot shows the 'Audio Codecs' configuration page. It features a title bar 'Audio Codecs' and a sub-header 'Audio Codecs'. Below the header, there are four rows, each with a label and a toggle switch: 'G.722:' with a green toggle, 'G.711(Ulaw):' with a green toggle, 'G.711(Alaw):' with a green toggle, and 'Opus:' with a grey toggle. At the bottom of the form is a blue 'Submit' button.

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



The screenshot shows the 'SIP Advance Settings' configuration page. It features a title bar 'SIP Advance Settings' and a sub-header 'SIP Advance Settings'. Below the header, there are four rows, each with a label and a numeric input field with minus and plus buttons: 'Local Port:' with '5060', '* RTP Start Port:' with '10000', '* RTP End Port:' with '20000', and '* RTP Timeout(Sec):' with '60'. At the bottom of the form is a blue 'Submit' button.

- 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 adjusting volume settings. It includes the following elements:

- Speaker Volume:** A numeric input field with a value of 7, flanked by minus and plus buttons.
- MIC AGC:** A green toggle switch indicating it is turned on.
- AGC Min Volume:** A numeric input field with a value of 2, flanked by minus and plus buttons.
- AGC Max Volume:** A numeric input field with a value of 6, flanked by minus and plus buttons.
- Key beep:** A grey toggle switch indicating it is turned off.
- Submit:** A blue button at the bottom of the form.

- **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.

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.

Press-to-Talk

Press-to-Talk Number1: Line: ▾

Press-to-Talk Number2: Line: ▾

Press Again to End Call:

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.

Sensor Settings

Sensor Detection:

Trigger Mode: ▾

Dial Number:

Sensor 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 Setting

Trigger by Input Signal:

Trigger by DTMF Signal:

* DTMF:

Trigger by Call Status:

Event:

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.

To use change of call status signal as the trigger of dry contact relay output, then please select the call event.

- 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:
 - 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:

Mode:

* Duration(Sec):

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.

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

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**

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.

The screenshot displays a configuration page for API settings. It includes several sections, each with a toggle switch and a list of API endpoints:

- Call API Enable:** (toggled on)
- 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:** (toggled on)
- 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:** (toggled on)
- 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>

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

(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 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

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
1	<input type="text" value="239.168.12.1"/>	<input type="text" value="2000"/>	<input type="text" value="Background-Music"/>	<input type="text" value="Fast Flashing"/>
2	<input type="text" value="239.168.12.2"/>	<input type="text" value="2000"/>	<input type="text"/>	<input type="text" value="Slow Flashing"/>
3	<input type="text" value="239.168.12.3"/>	<input type="text" value="2000"/>	<input type="text"/>	<input type="text" value="On"/>
4	<input type="text" value="239.168.12.4"/>	<input type="text" value="2000"/>	<input type="text"/>	<input type="text" value="Disabled"/>
5	<input type="text" value="239.168.12.5"/>	<input type="text" value="2000"/>	<input type="text"/>	<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
- Relay Control: 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

- **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.

Language Settings

Voice Prompts Language:

Settings

Network Setting

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

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

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 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.

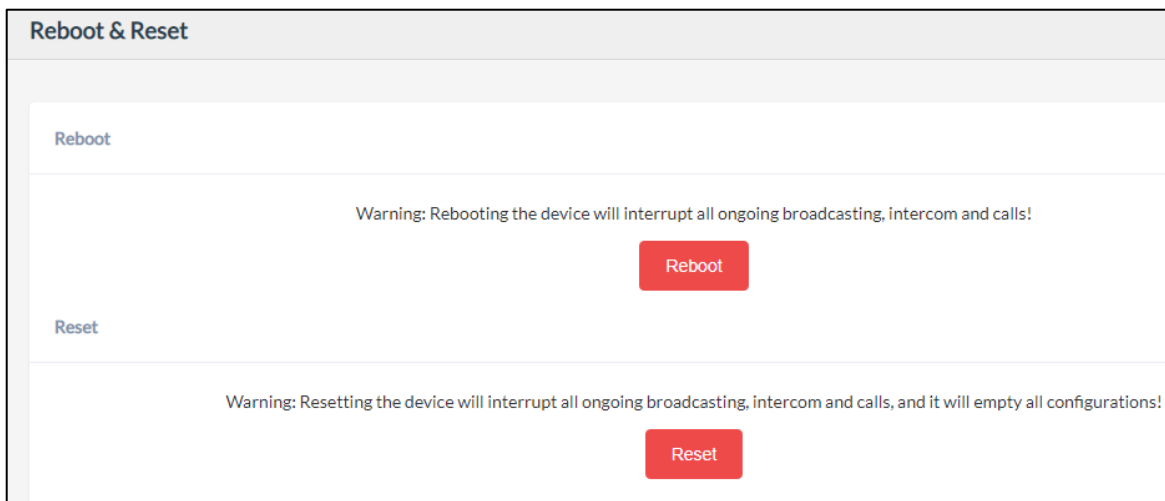
It will take around 2 minutes to complete the firmware upgrade, during upgrading process please DO NOT power off the network speaker.

Reboot & Reset

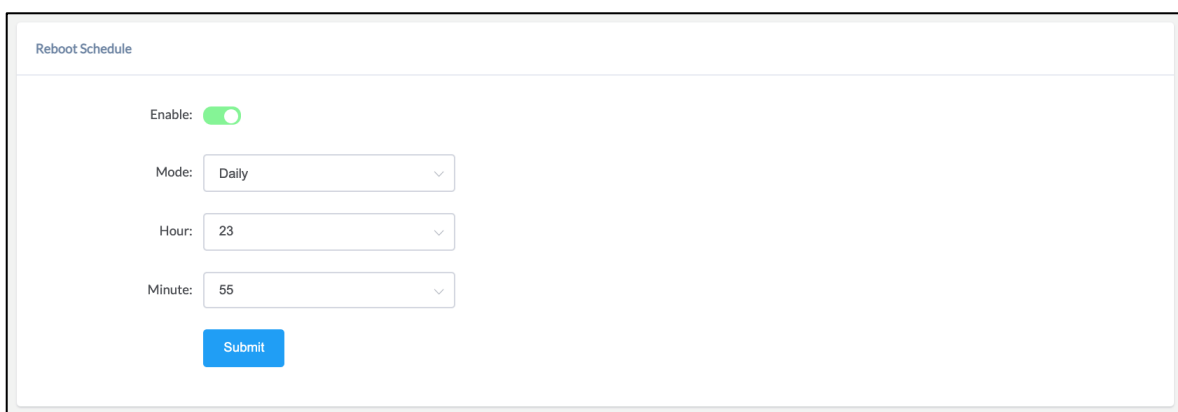
The network speakers can be rebooted and reset from the web management interface on the Settings -> Reboot & Reset page.

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.



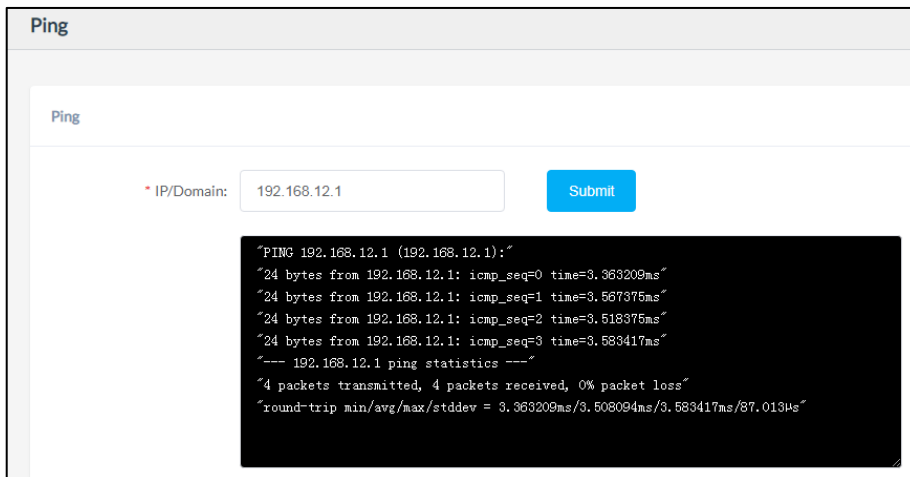
When the Reboot Schedule feature is Enabled, you can set up the automatic reboot daily, weekly, or monthly at a specify time.



Maintenance

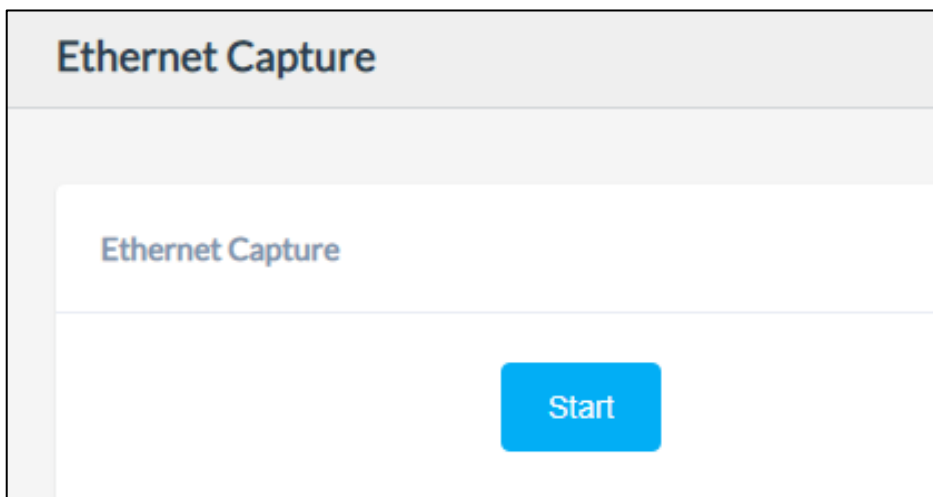
Diagnostic

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 ".pcap" 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.

Export

[Export YAML](#) [Export JSON](#)

Import

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



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

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

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, 68692E290012

Static Provisioning Server

Access Mode:

TFTP Server Address:

Configuration Format:

Configuration Filename: \$mac.json

Update Mode:

Audio Test

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

Speaker Test

Microphone Loop Test

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.

@ 2020 Zycoo Co., Ltd. All rights reserved.

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.

