SIP Safety Intercom
User Guide

Contents

Overview

Product Overview

ZYCOO SIP Safety intercom IA03 and IV03 are SIP enabled intercom devices which can be used for consulting and SOS applications.

By default, IA03 can be used as a voice intercom and IV03 has a built-in HD camera which can support video intercom. If there is an existing IP camera, this can be linked with IA03 for video intercom. IP cameras which support RTSP, such as Dahua and Hikvision IP cameras can be linked with IA03.

Based on SIP and with PoE support, the installation of the IA03 and IV03 only requires a PoE enabled network cable, no power supply and extra wiring is required.

Each of the intercom devices has been equipped with a programmable dial button, any desired number can be preconfigured. By pressing the button users can talk with the target number in hands-free mode.

IA03 and IV03 have been equipped with digital level input for connection of sensors, and also with a relay switch which can be used to connect door magnet. The action (open/close the door) of the relay switch can be controlled by DTMF signal, intercom call status and the sensor input signal.

In addition to working with ZYCOO IP Audio Center, the IA03 and IV03 intercoms can be also integrated with third-party SIP servers and the features detailed above are still applicable.

Product Specifications

- Speaker Components: Ø 40 mm x 12.7 mm (1.6 x 0.5 in)
- Sensitivity: 90±3dB
- Distortion: <10%, 1000Hz 0.1w/1m
- Rated Power: 3W
- Max Power: 4W
- Frequency Range: 630Hz~4.5KHz
- Acoustics: Mono
- Microphone: -36±2dB RL=2.2K Vs=2.0V
- Relay Switch: Max voltage AC 125V-1A/DC 60V-1A
- Video Resolution (IV03):   1920*1080@20fps, 1280*720@25fps

Basic Settings

Web Interface Login

By default, the SIP safety intercom devices' IP assignment has been configured as DHCP. Please ensure there's DHCP server available in the LAN where the SIP safety intercom 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 call button for 5 seconds then release, 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 safety intercom. The login screen is shown as below image, here we take IA03 SIP safety intercom as an example.
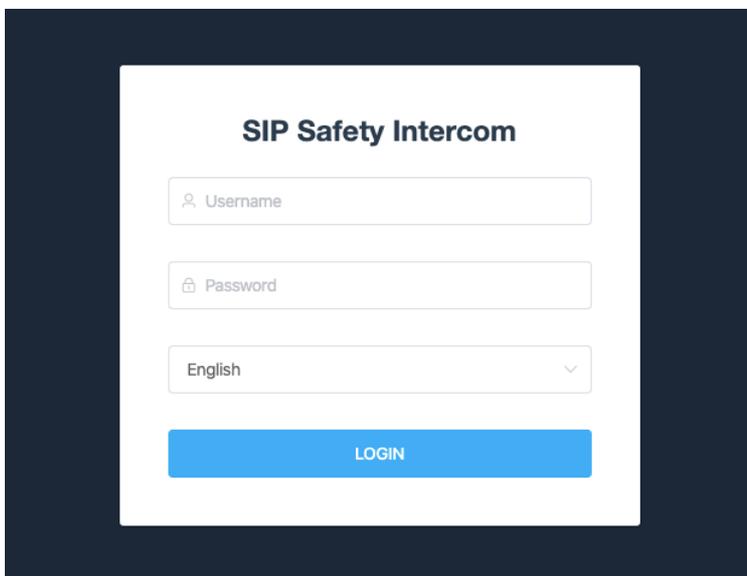
Use the default login credentials to login to the web management interface.

Default username: admin

Default password: admin

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

Change Password

Username: admin

* Old Password: •••••

* New Password: ••••••••••

Submit

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 device model, IA03 or IV03.
- 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.

Network Settings

To change the IP assignment from DHCP to Static IP, please go to Settings -> Network Settings page.
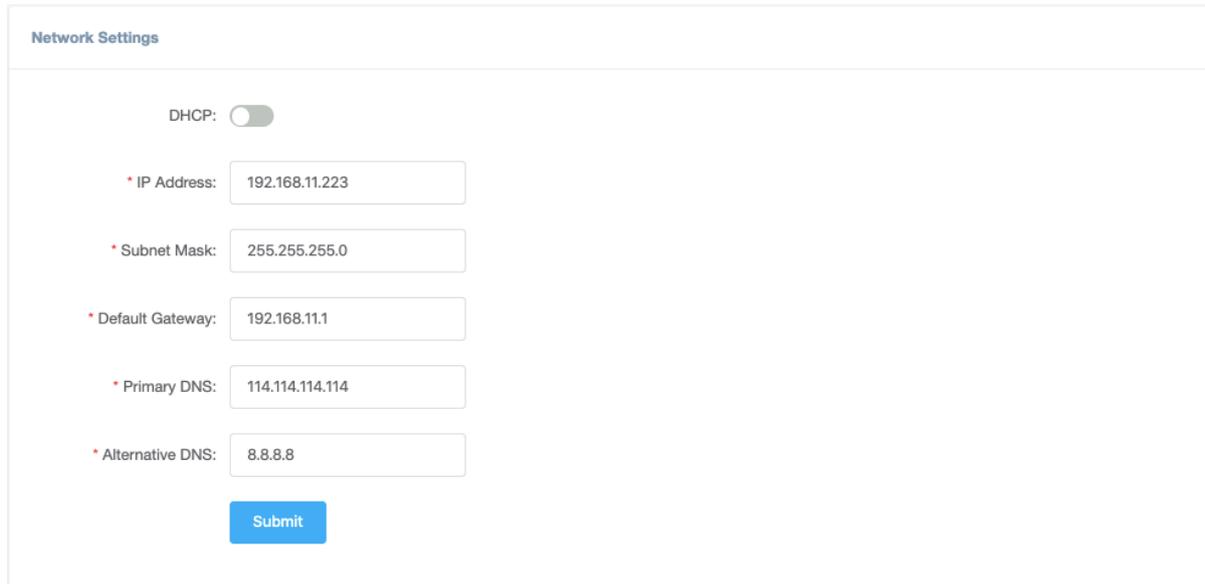
Turn the DHCP switch button off to show the network parameter settings.

Network Configuration Parameters

- 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 SIP safety intercom device when the IP Audio Center or other SIP server is installed outside the LAN.
- Primary DNS: Enter an effective primary DNS server address.
- Alternative DNS: Enter an alternative DNS server address, when the primary DNS fails, alternative DNS will be used.

SIP Account

For SIP account settings, go to SIP Settings -> SIP Account page.

SIP Account Configuration Parameters

- SIP Server: Enter the IP address or domain name of the SIP server. Default SIP port is 5060, if the SIP server uses other port number as SIP port, please enter the server address in the format of "ip_address:port" or "domain_name:port".
- User ID: The SIP account number provided by SIP server.
- Password: The corresponding password of the SIP account.
- Register Expiration(S): SIP register expiration time.
- Starting RTP Port: The starting local RTP port number.
- Enable Integration with ZYCOO IP Audio Center: If the SIP safety intercom device is registering to ZYCOO IP Audio Center, this parameter needs to be enabled, otherwise do not enable it.
- Activate: Used to enable or disable the SIP register.

Advanced SIP Settings

Audio Codecs

The SIP safety intercom device supports 4 audio codecs: G.722 (wideband codec), G.711(Ulaw), G.711(Alaw) and G.729.

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

Please keep at least one codec enabled which is also supported by the SIP server, otherwise SIP calls will not work.

SIP Call Settings

The SIP Call Settings are used to configure the SIP safety intercom of how to handle the SIP calls. Please go to SIP Settings -> SIP Call Settings page.
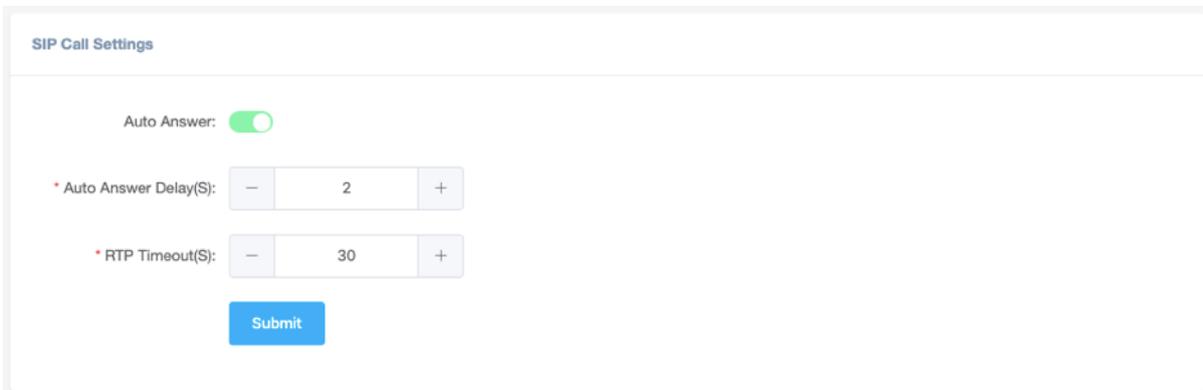
SIP Call Settings Configuration Parameters

- Auto Answer: If enabled, all SIP calls will be automatically answered without delay.
- Auto Answer Delay(S): When Auto Answer is disabled, the SIP calls will be answered with the given number of seconds of delay.
- RTP Timeout(S): If no RTP stream has been received within the given number of seconds, the SIP safety intercom will terminate the SIP call.

Advanced System Settings

Volume Control

The SIP safety intercom device's speaker and microphone volume level can be adjusted from its web management interface, on the Settings -> Volume Control page.
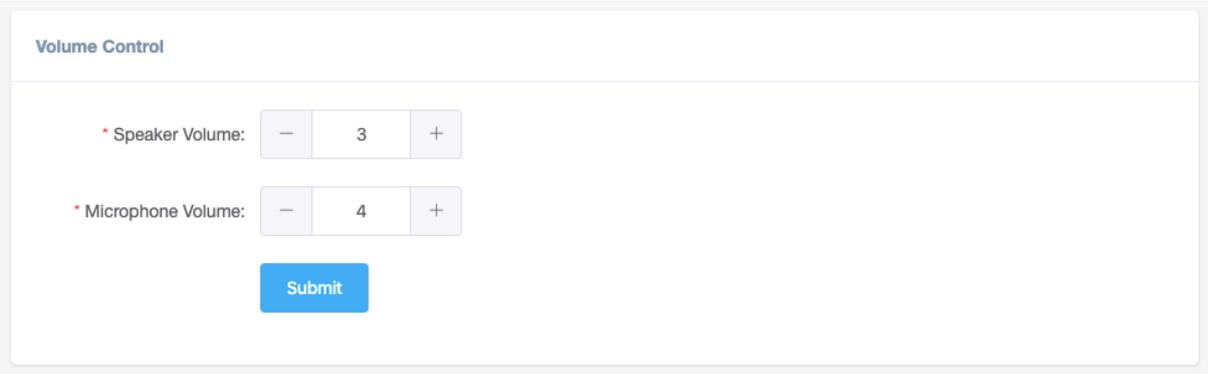
There are 10 (from the lowest 0 to the highest 9) speaker output and microphone input volume levels that can be set.

I/O Settings

I/O settings are used to configure the press-to-talk, digital level input and dry contact relay control options. Please go to the Settings -> 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.
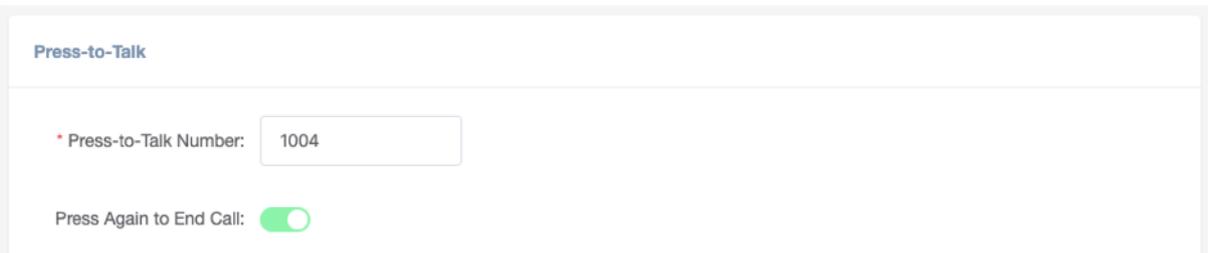
The "Press Again to End Call" option can be used for the caller to end or cancel an intercom call.

Input settings are used to setup the reactions to the digital level input signal from sensors.

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 safety intercom device, it will dial the given number.

The dry contact relay output is used for electric door magnet control or other devices which can be controlled by dry contact relay.

The switch signal output duration can be configured from 1 to 600 seconds.

The trigger of the output action can be digital level input signal, DTMF signal or call state.

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.
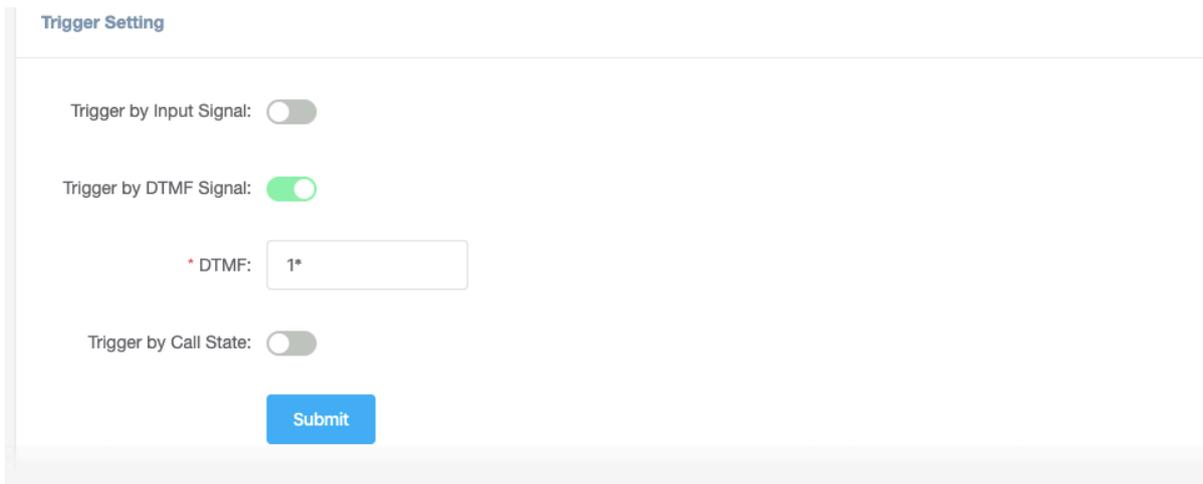
In this example, DTMF key press is *1, so during a SIP call, the caller press *1 from the phone key pad, it will trigger the dry contact relay output on the SIP safety intercom.

To use call state as the trigger, then please select from "Outgoing", "Incoming" and "Hangup" states.
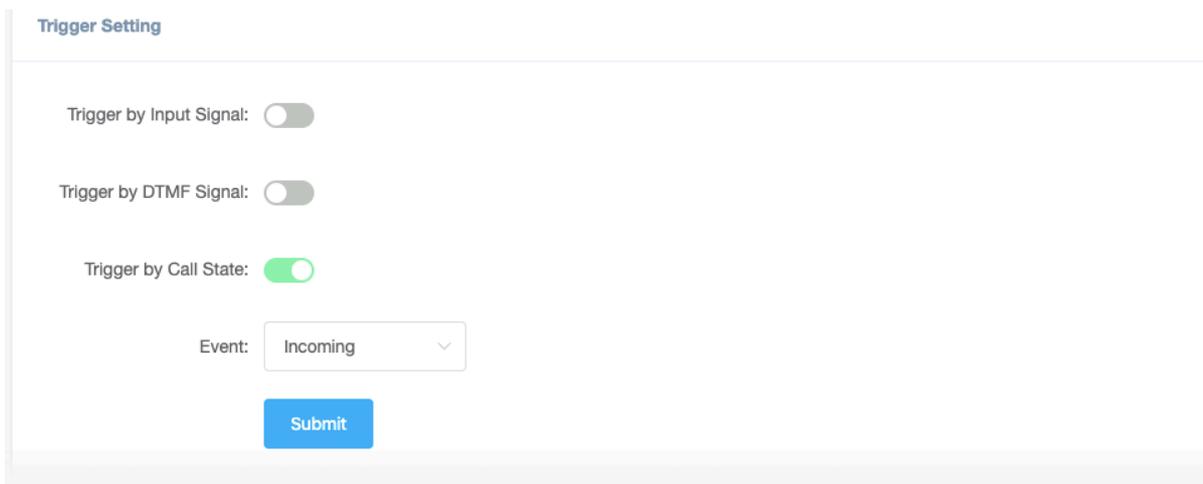
If "Outgoing", the dry contact relay output will be triggered when someone presses the call button of the SIP safety intercom to establish an intercom call.

If "Incoming", the dry contact relay output will be triggered when the SIP safety intercom gets an incoming SIP intercom call.

If "Hangup", the dry contact relay output will be triggered when a SIP call ends on the SIP safety intercom.

Time Settings

The SIP safety intercom devices obtain time from the network time servers using NTP, to change the NTP settings please go to Settings -> Time Settings page.
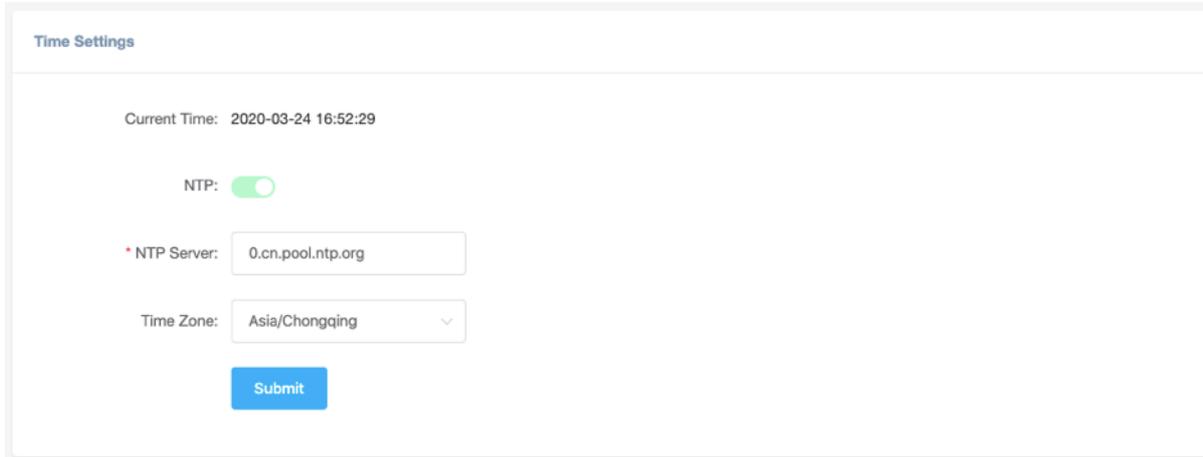
Here you can change an NTP server by modify the NTP server address and you can select the time zone of your location, so the SIP safety intercom will synchronize time of your time zone from the NTP server you have configured.

Language Settings

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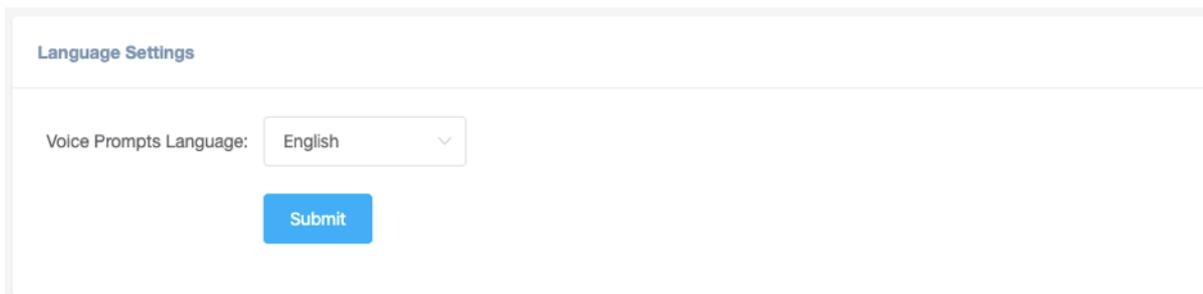
Upgrade

To upgrade the SIP safety intercom device's firmware, please go to Settings -> Upgrade page.
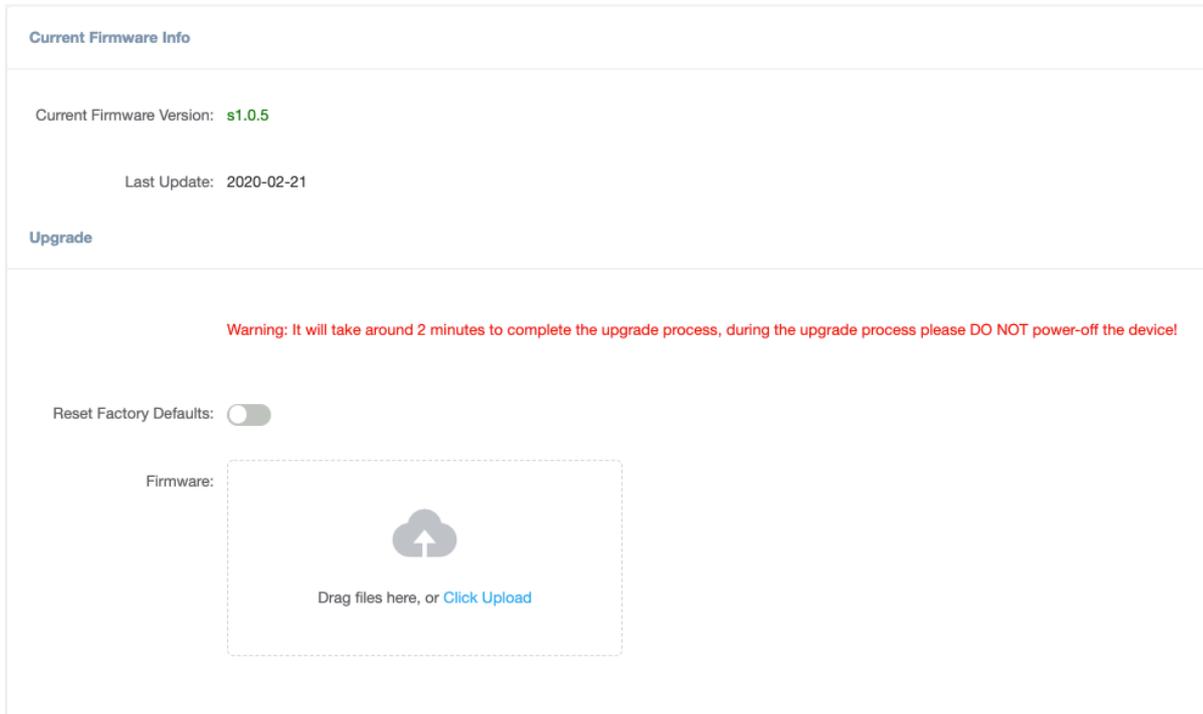
Current Firmware Info

Current Firmware Version: s1.0.5

Last Update: 2020-02-21

Upgrade

Warning: It will take around 2 minutes to complete the upgrade process, during the upgrade process please DO NOT power-off the device!

Reset Factory Defaults:

Firmware:

Drag files here, or Click Upload

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

Upload the .img file provided by ZYCOO to perform the upgrade action. If you wish to reset the device 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 device.

Reboot & Reset

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

Both reboot and reset action will terminate all SIP intercom calls. And the reset action will erase all configurations of the device. Please reboot or reset the devices when they are not in use.

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.