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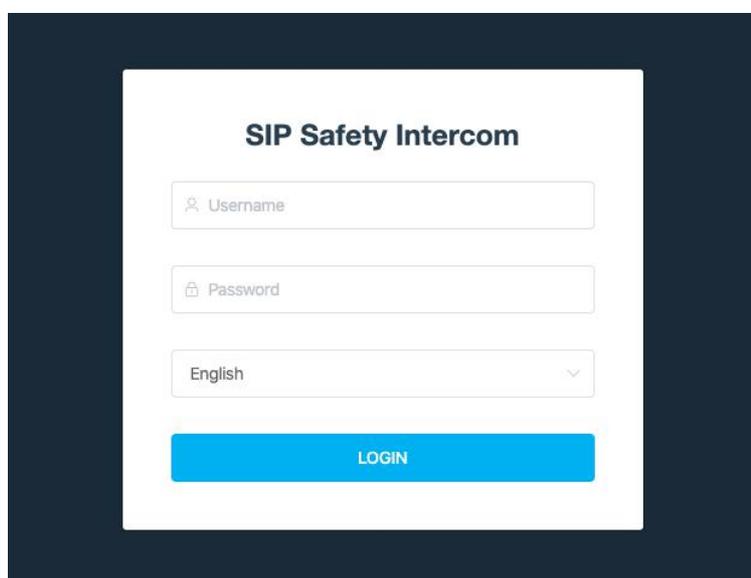
Use the default login credentials to login to the web management interface.

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

Primary SIP Account	5008@192.168.12.205:5060	Registered
Secondary SIP Account-1	3030@120.27.216.154:5060	Registration failed!
Secondary SIP Account-2		Unconfigured

DEVICE INFORMATION

Device Model	IA03
Hardware Version	Ver1.0
Software Version	s1.2.0beta2
Speaker Volume	1 (0-9)
Device Description	IA03

NETWORK INFORMATION

Mac Address	68:69:2E:24:08:08
IP Assignment	DHCP
IP Address	192.168.12.244
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, 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.

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.
- 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.

- 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.
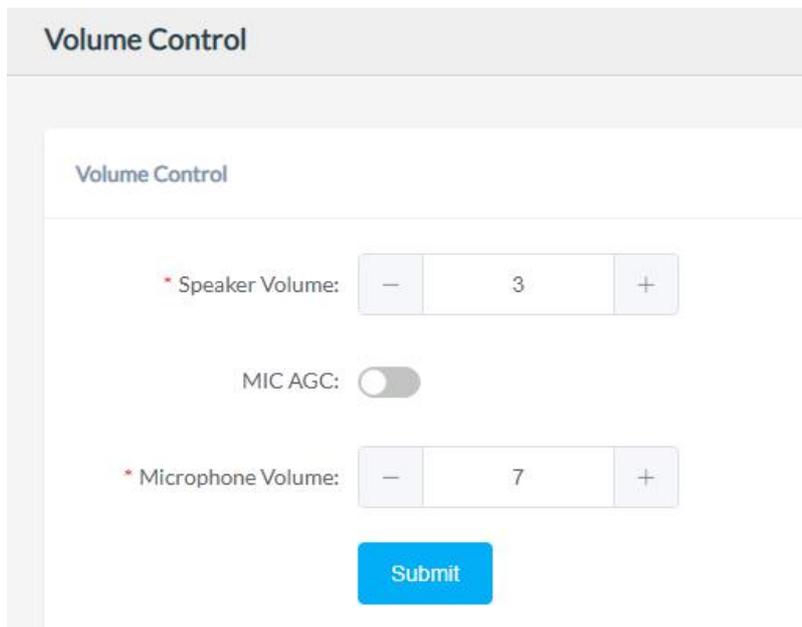

- Speaker Volume: The default speaker volume is 7, adjustable range is 0 ~ 9.

- 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 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.
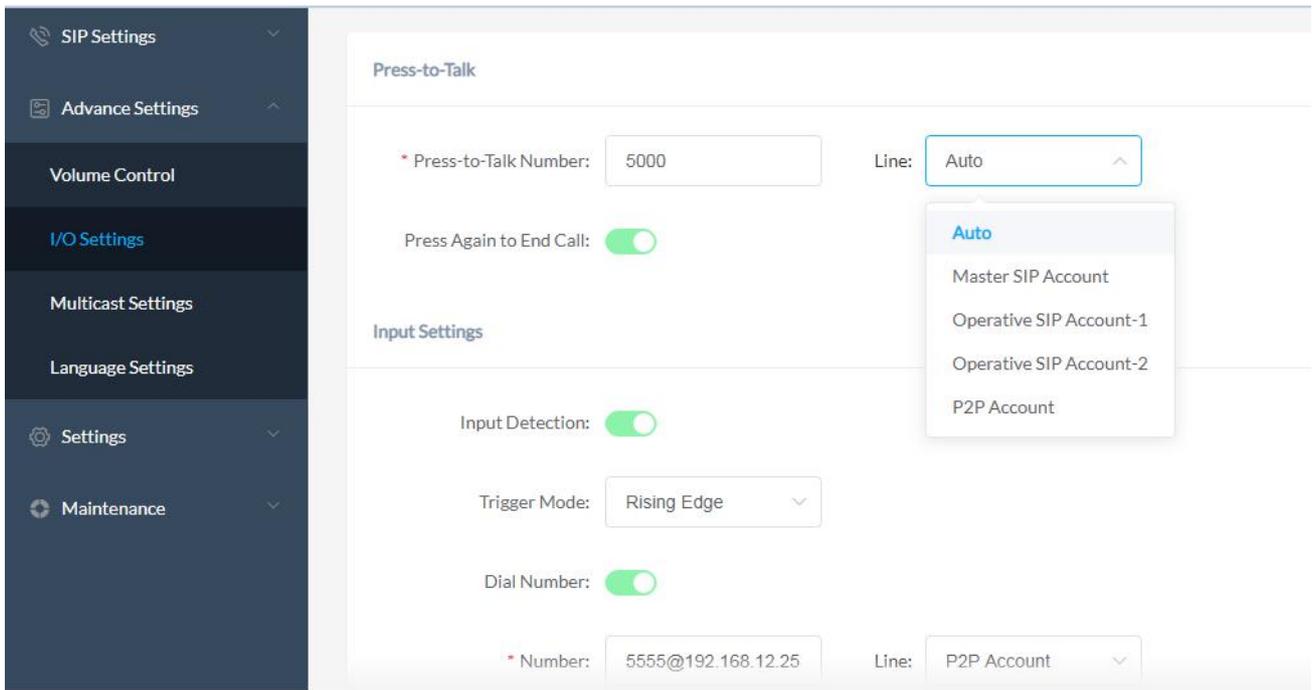
Press-to-Talk

- 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.

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.

- o Input Detection: Enable/Disable, enable this option to detect any digital level signals.
- o Trigger Mode: There are "Falling Edge" and "Rising Edge" options for the trigger.
- o Dial Number: Enable/Disable, enable this option if you need to dial a number when the corresponding digital signal is received.
- o Number: The number that need to dial
- o 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.

- 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.

- 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'
-

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.

Network Settings

DHCP: ⬤

* IP Address: 192.168.11.223

* Subnet Mask: 255.255.255.0

* Default Gateway: 192.168.11.1

* 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

- 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.

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.

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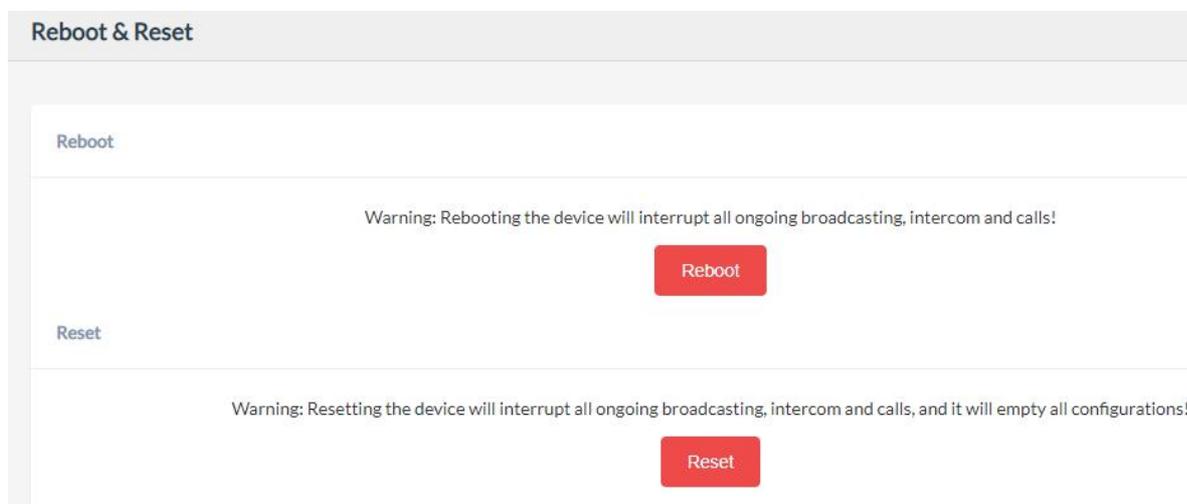
It will take around 2 minutes to complete the firmware upgrade, during upgrading process please DO NOT power off the network speaker.

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.

Reboot & Reset

Reboot

Warning: Rebooting the device will interrupt all ongoing broadcasting, intercom and calls!

Reboot

Reset

Warning: Resetting the device will interrupt all ongoing broadcasting, intercom and calls, and it will empty all configurations!

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.
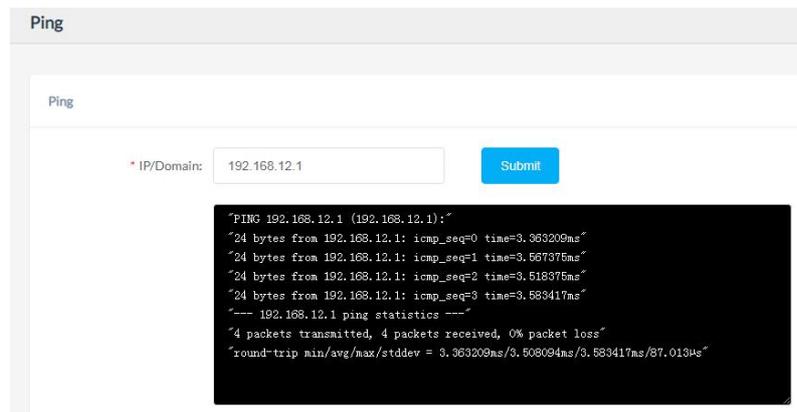
Ethernet Capture

The purpose of the Ethernet capture tool is to capture ethernet network packets and store then in a standard Wireshard compatible packet capture '.pacp' file for immediate viewing and data analyzing.
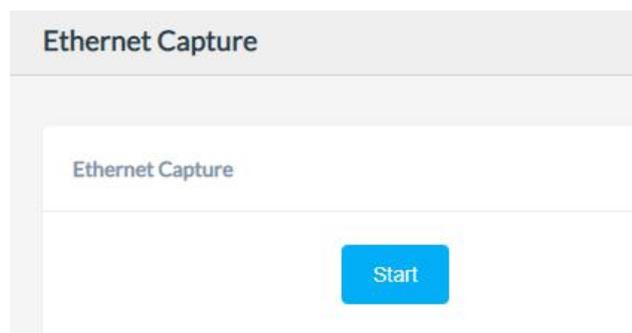
www.zycoo.com

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.