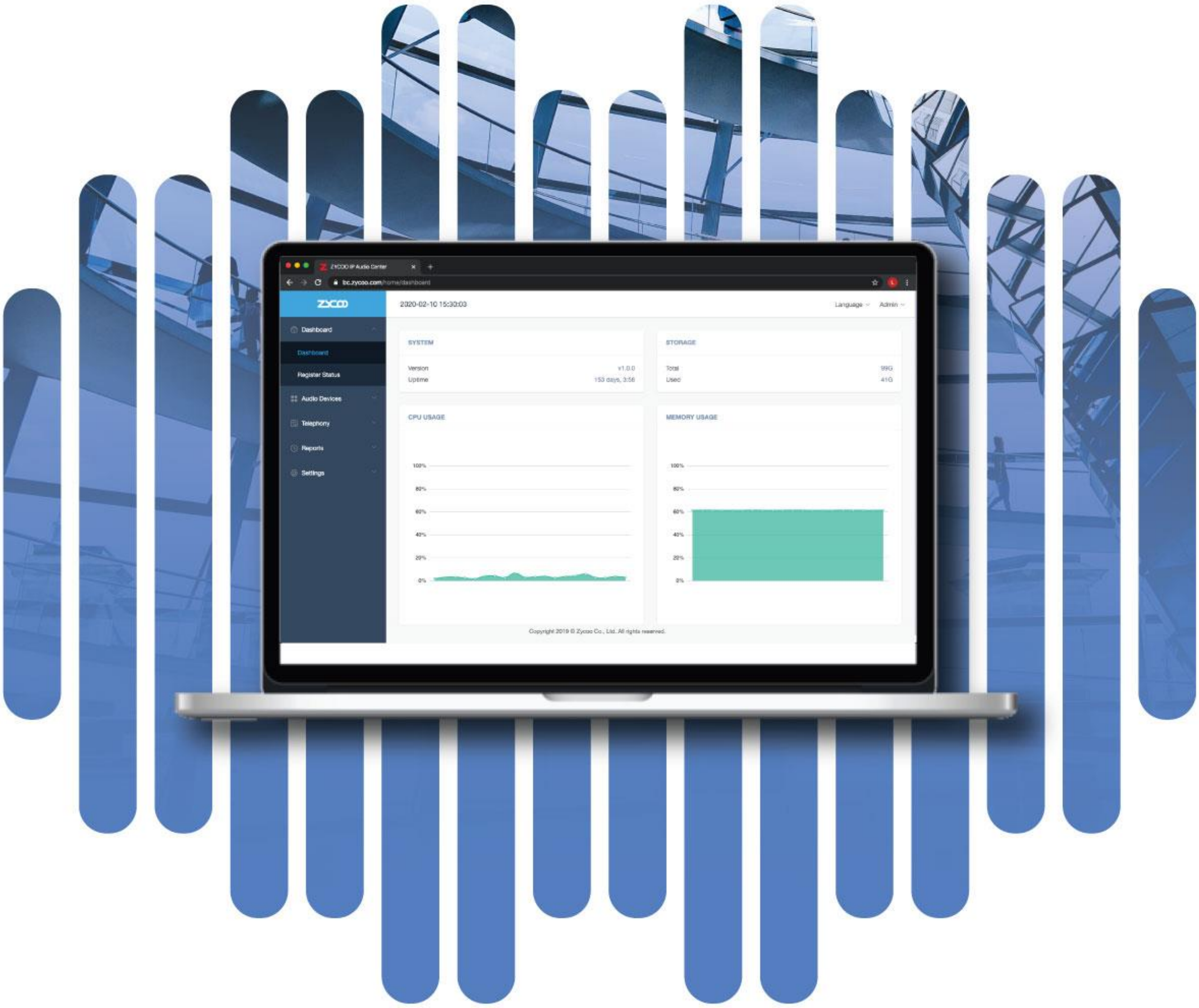
IP Audio Center User Guide



Contents

Overview *1*

About IP Audio Center 1

Specifications 1

Features 1

IP Audio Endpoints *3*

Speakers 3

Intercom Devices 3

IP Phones 4

Video Endpoints 4

SIP Paging Gateway 4

Informational *4*

Default Settings 4

Service Access Ports 5

Default Administrator Login Credentials 5

Default Feature Codes 5

Web Access 5

System Information 7

Device Register Information 8

Menu Introduction 8

Connect IP Audio Dispatch Console *9*

Create SIP Accounts 9

Setup IP Audio Devices 11

Setup Paging Groups 14

Create Dispatch User 16

Telephony *17*

Trunks 17

Outbound Rules 19

Inbound Rules 20

Digital Receptionist 20

Reports 23

Call Logs 23

Recordings 24

Phone Call Recordings 24

Live Announcements Recordings 25

Intercom Call Recordings 25

Meeting Recordings 25

Settings 26

System Settings 26

Default SIP Paging Volume 26

Default Alarm Volume 26

Indication Tone 27

Ring Duration 27

SIP Settings 27

SIP NAT 28

Modbus Settings 29

Feature Codes 29

Recording Settings 30

Audio Prompts 31

Labels 33

Security Center 33

Intrusion Prevention 34

IP Blacklist 35

IP Whitelist 35

License 1

Overview

About IP Audio Center

ZYCOO IP Audio Center is the engine of ZYCOO IP Audio Solution, it is a software solution which can be installed on hardware or cloud- based servers. It has IP public address system, SIP voice/video intercom and IP PBX system modules integrated, it provides an all-in- one unified IP audio solution.

The IP Audio Center provides centralized IP audio endpoints auto provisioning and management, public address, background music, intercom, emergency broadcasting, audio conferencing, IP phone call and more features. It is suitable for public safety, smart city, secure community, industry, transportation, health and more application scenarios.

Specifications

Number of IP Audio Endpoints	4000 (Max)
Number of Paging Groups (Zones)	Unlimited
Number of MP3 Audio Files	Unlimited
Number of Playlists	Unlimited
Number of Timetable Triggered Broadcast Tasks	Unlimited
Number of Number Triggered Broadcast Tasks	Unlimited
Number of SIP Concurrent Calls	500 (Max)
Number of Simultaneous Conference Attendees	500 (Max)
Number of SIP Paging Network Speakers	500 (Max)
Recording Time	Unlimited (Wav format, 1MB/min)

**Note: The Above specifications are software limitation of the ZYCOO IP Audio Center, the actual capacity depends on the server which deploys the IP Audio Center system.*

Features

- Scheduled Broadcasting
- Recording System
- Audio Conferencing
- Auto Provisioning
- Volume Control
- Barge Spy
- SIP Trunking
- Video Call
- Multiple Dispatch User
- Customized Group Number
- Customized IVR Voice Prompts
- Broadcast Begin Indication Tone
- Customized Device Label
- System Logs
- RTSP Address Association
- Call Logs
- Inbound/Outbound Call Control
- Call Forward
- Google TTS (Text-to-Speech)
- SIP Voice Intercom
- Intercom Recording
- Zone-based Paging
- Video Intercom
- Whisper Spy

- Do Not Disturb (DND)
- Wakeup Call
- Emergency Broadcasting
- Live SIP Paging
- Paging Group (Zone)
- Endpoints Status Monitoring
- Call Spy
- Call Split
- API Support
- Caller ID
- Streamed Media Music
- Device Fault Report
- Customized Device Number
- Device Location Mark
- Call Recording
- Feature Codes
- Call Transfer
- Auto Receptionist (IVR)

IP Audio Endpoints

The following IP audio endpoints can be integrated into ZYCOO IP Audio Center.

Speakers

- ZYCOO Network Speakers

ZYCOO SC15 Network Ceiling Speaker

ZYCOO SW15 Network Cabinet Speaker

ZYCOO SH30 Network Horn Speaker

ZYCOO SL50 Network Column Speaker

- Third-party SIP enabled speakers

Note: when the third-party SIP enabled speakers have been used with ZYCOO IP Audio Center, only SIP paging is supported, streamed music is not applicable.

Intercom Devices

- ZYCOO SIP Safety Intercom

ZYCOO SIP Safety Intercom IV03

ZYCOO SIP Safety Video Intercom IC03

- Third-party SIP enabled Intercom

IP Phones

- ZYCOO CooFone series IP phones

CooFone series H81

CooFone series H83

- Third-party SIP Phones

Video Endpoints

- SIP video phones
- IP cameras which support RTSP (Hikvision, Dahua)

SIP Paging Gateway

- ZYCOO SIP Paging Gateways

X10

X30

Informational

Default Settings

Service Access Ports

IP Audio Center web Port	TCP: 80,TCP: 443
SIP Port	UDP: 5060
RTP Ports	10001 - 12000
Stream Media Port	TCP: 8001
RTSP Port	UDP:554
API Port	TCP: 8000

Default Administrator Login Credentials

Username	admin
Password	admin

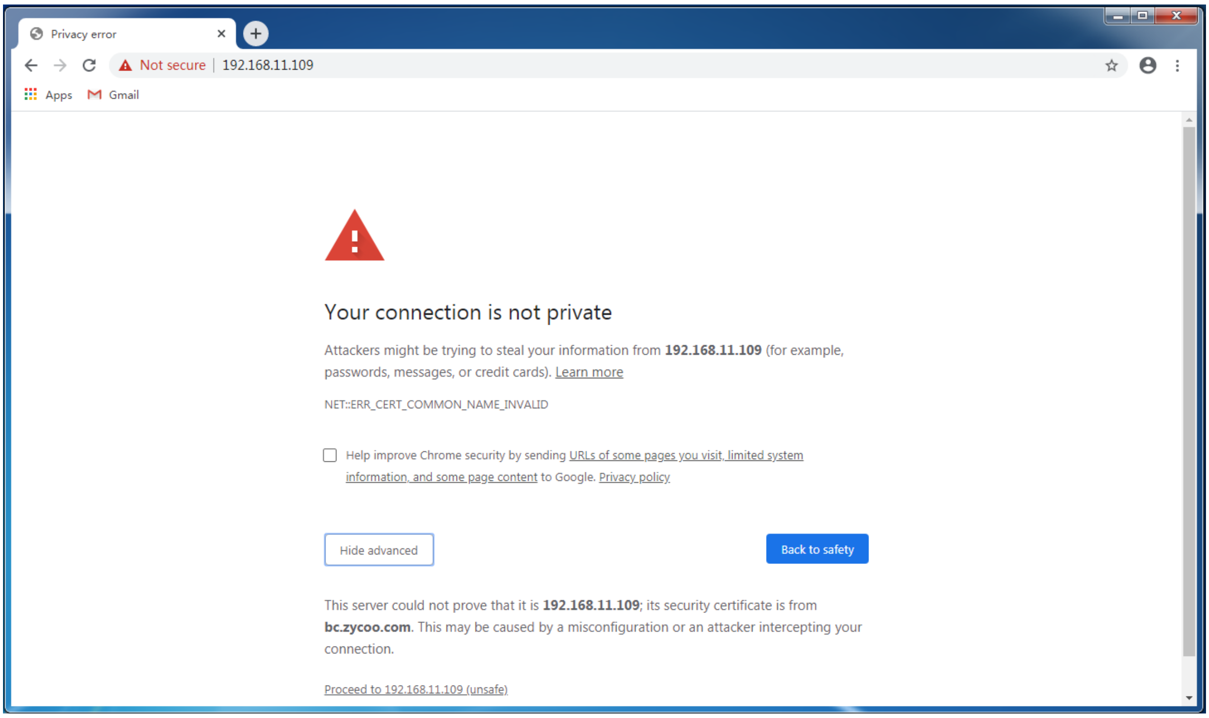
Default Feature Codes

Alarm Start	*11
Alarm Stop	*12
Barge Spy	*99
Call Split	*98
Call Spy	*97
Whisper Spy	*96
Wakeup Call	*95
Do Not Disturb	*94
Always Forward	*93
Forward on Busy	*92
Forward on No Answer	*91
Blind Transfer	#

Web Access

As the ZYCOO IP Audio center is deployed using Docker on your server, so, to access the web interface of the IP Audio Center you have to use the IP address or domain name of the server which you have deployed Docker.

Open your web browser, input the server address (e.g. your server IP is 192.168.11.109, input the URL http) and press enter, you'll see the below warning.

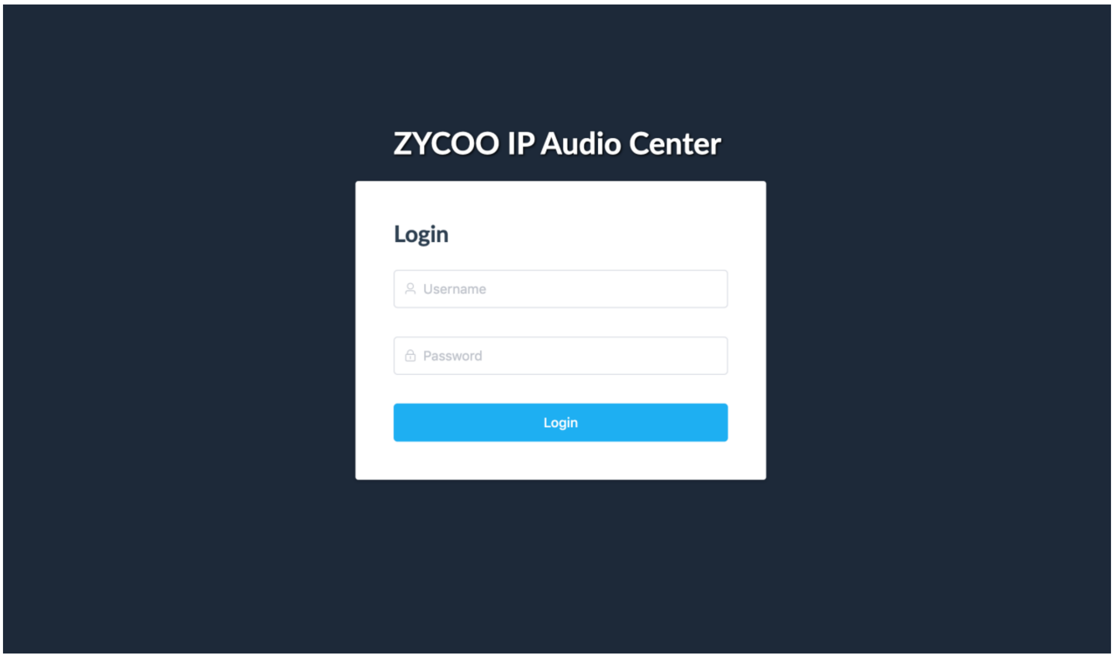


Please click on “Proceed to 192.168.11.109 (unsafe)” to continue accessing the web interface.

Notice:

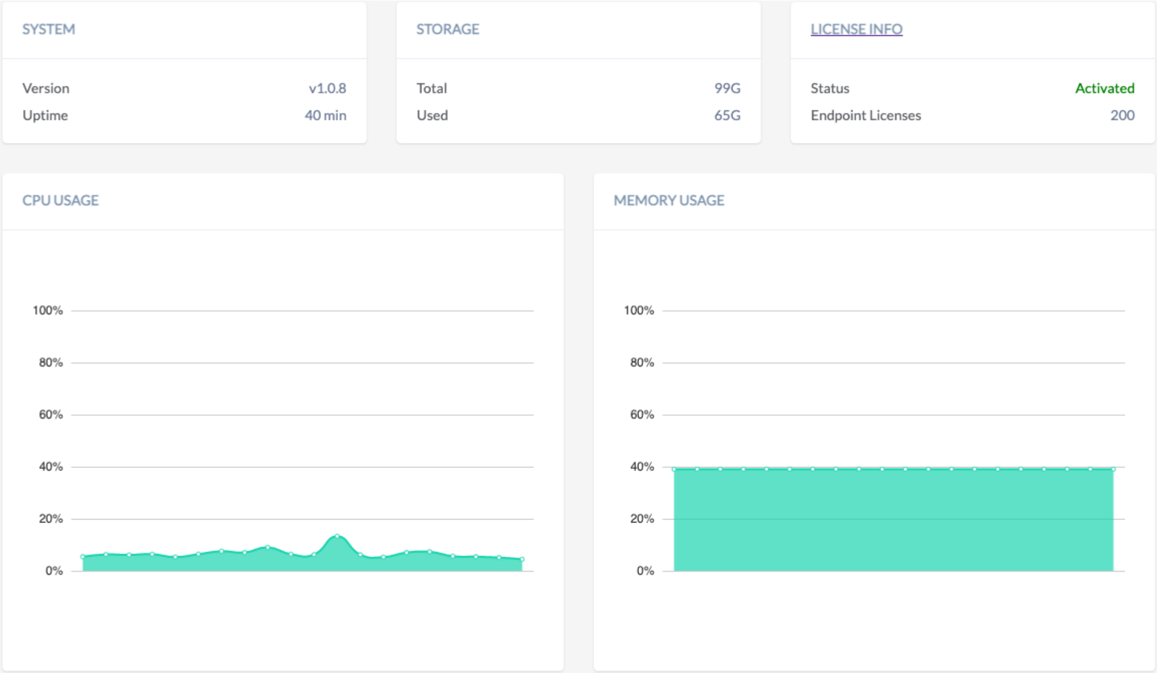
Your web browser must be Google Chrome 69 (or newer), Opera, IE10 (or newer), Firefox 65 (or newer).

The login screen as shown in the figure below.



Please enter the default administrator login credentials to log in.

After login, you'll first see the Dashboard screen as shown in the figure below.



For system security, please change admin password from Admin (top right corner) -> Change Password menu.

System Information

- SYSTEM

Version	The software version of the server.
Uptime	Indicates how long the server has been continuously working.

- STORAGE

Total	Total server storage space.
Used	Storage space have been used.

- LICENSE INFO

Status	Total server storage space.
Free Trial Expiration	Each system is given a free 45-days free trial period before activating the system.
Endpoint Licenses	When the license is activated, it shows the number of endpoint licenses authorized.

- CPU USAGE

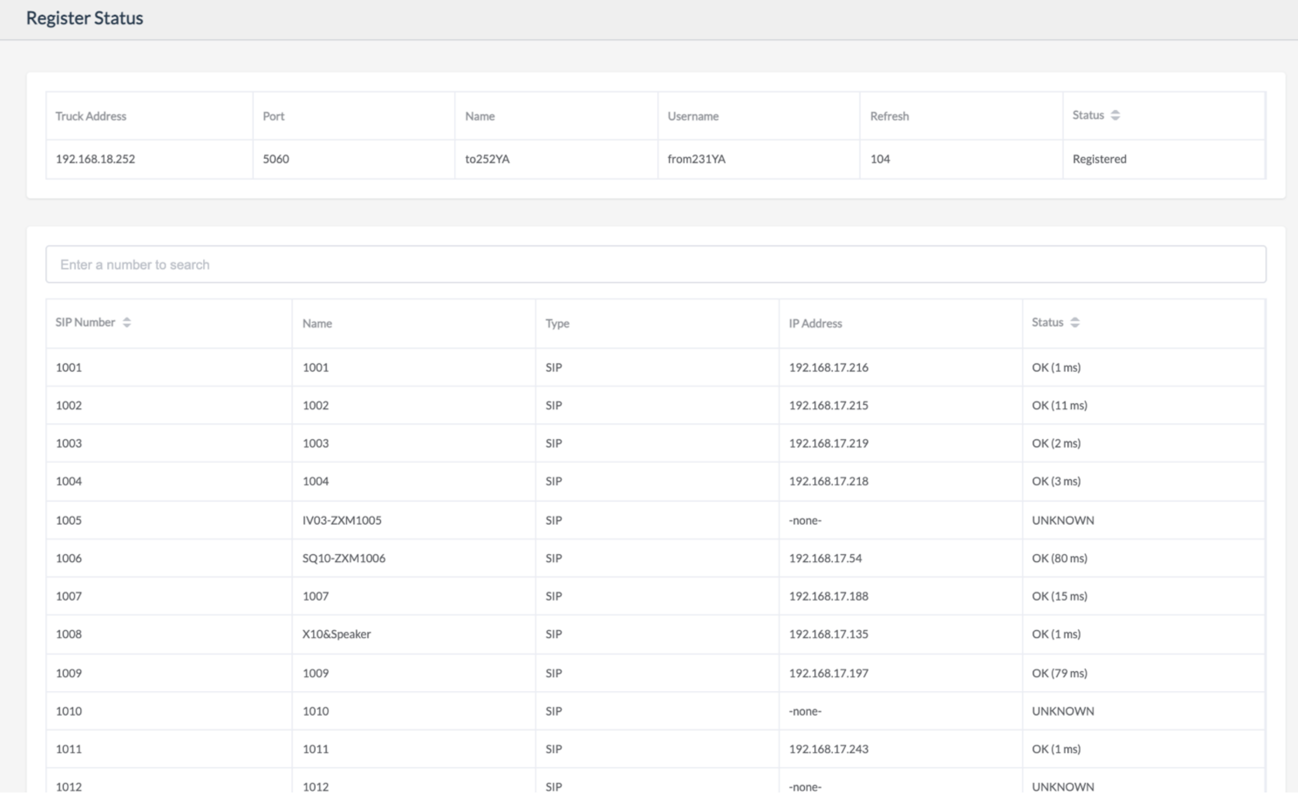
CPU usage graph, indicates the real-time CPU usage in percentage form.

- MEMORY USAGE

Memory usage graph, indicates the real-time memory usage in percentage form.

Device Register Information

On the **Dashboard/Register Status** page, it shows both the SIP truck information, and all created SIP account information.



Register Status

Truck Address	Port	Name	Username	Refresh	Status
192.168.18.252	5060	to252YA	from231YA	104	Registered

Enter a number to search

SIP Number	Name	Type	IP Address	Status
1001	1001	SIP	192.168.17.216	OK (1 ms)
1002	1002	SIP	192.168.17.215	OK (11 ms)
1003	1003	SIP	192.168.17.219	OK (2 ms)
1004	1004	SIP	192.168.17.218	OK (3 ms)
1005	IV03-ZXM1005	SIP	-none-	UNKNOWN
1006	SQ10-ZXM1006	SIP	192.168.17.54	OK (80 ms)
1007	1007	SIP	192.168.17.188	OK (15 ms)
1008	X10&Speaker	SIP	192.168.17.135	OK (1 ms)
1009	1009	SIP	192.168.17.197	OK (79 ms)
1010	1010	SIP	-none-	UNKNOWN
1011	1011	SIP	192.168.17.243	OK (1 ms)
1012	1012	SIP	-none-	UNKNOWN

Menu Introduction

Here's a list and brief introduction of menus available in the IP Audio Center admin web interface. The detailed explanation of the sub menu and settings will be explained in the following chapters.

- Dashboard

Provides system information and detailed endpoints register status.

- Audio Devices

Manages all SIP accounts, endpoint devices and paging groups (zones).

- Telephony

Configures all telephony features, including SIP trunking, inbound and outbound rules, digital receptionist and voice prompts for digital receptionist.

- Reports

Provides all call logs of the entire system and the recordings which have been generated by phone calls, live announcements, intercom calls and meetings.

- Settings

Provides system advanced settings, recording settings, audio prompts upload and management, dispatch user management, endpoint device label management, system security and license control.

- Language

Change web interface language from here.

- Admin

Change admin user password and logout admin web interface from here.

- Activate

This button only appears when you had submitted changes settings, don't forget to click on it to make all changes effective.

Connect IP Audio Dispatch Console

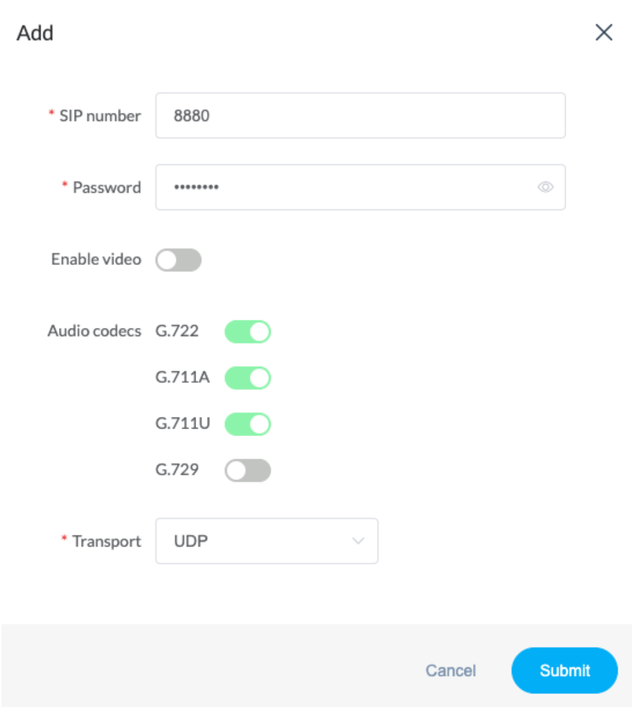
Before you can connect ZYCOO IP Audio Dispatch Console to the IP Audio Center, the following settings need to be done.

Create SIP Accounts

All IP audio endpoints to be connected to ZYCOO IP Audio Center support SIP protocol, and each of them needs to be configured with a SIP account.

Please go to **Audio Devices** -> **SIP Accounts** screen, here on this page you can create one SIP account at a time or you can bulk create SIP accounts.

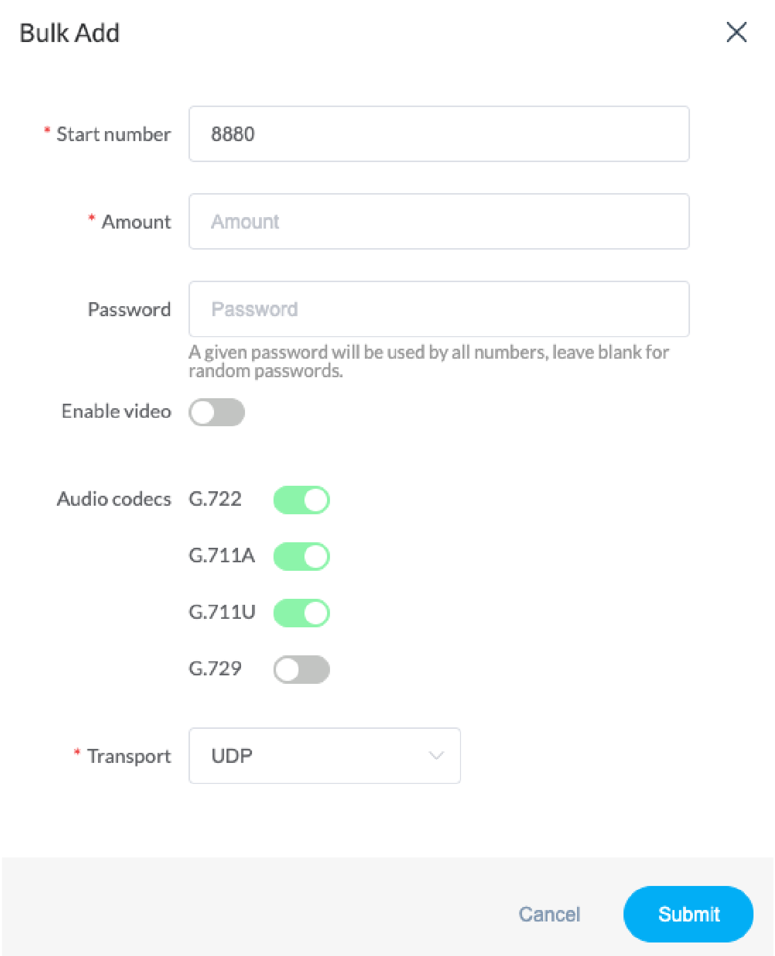
To create a single SIP account, please click on the **Add** button.



- SIP Number: The SIP account number to be used by an IP audio device, for SIP register and calling (phone call, intercom call, SIP paging call). Number length 2 to 15 digits.
- Password: The password to be used for SIP register. A default 8-character password will be generated, you may use a custom password but please ensure it is secure.
- Enable Video: Enable video calling, supported video codec H.264.
- Audio Codecs: Audio codecs to be used for SIP calling. Supported audio codecs G.722, G.711A, G.711U and G.729.
- Transport: The system currently supports UDP, TCP, and TLS protocols. (Note: If you use TLS/TCP, you need to enable related services in *System Settings/Network/SIP Settings*)

Once settings done, click on Submit button to save the settings.

To Create bulk SIP numbers, please click on Bulk Add button.

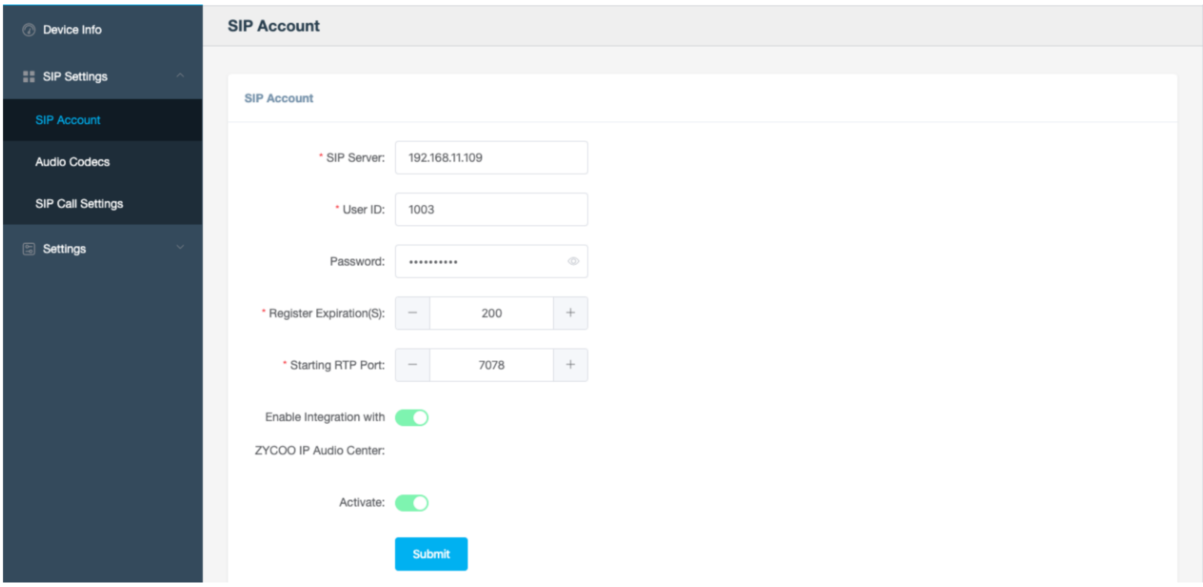


In the popup dialog, please define a start number and then input the amount of numbers to be created. In the password field, you may specify a STRONG password or you may leave it blank for system generated random passwords.

Setup IP Audio Devices

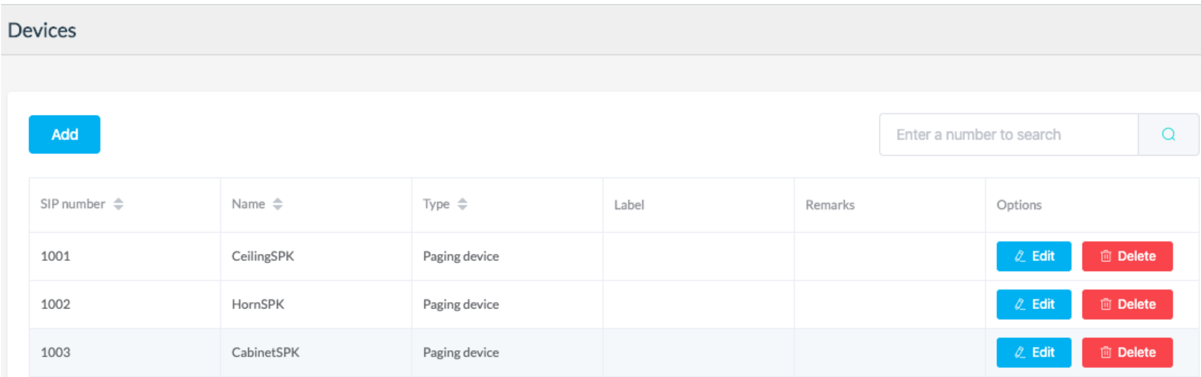
There are 2 steps to register an IP Audio device to the IP Audio Center. Here we take SW15 Network Speaker as an example.

Open the SW15 web interface, go to **SIP Settings** -> **SIP Account** screen.



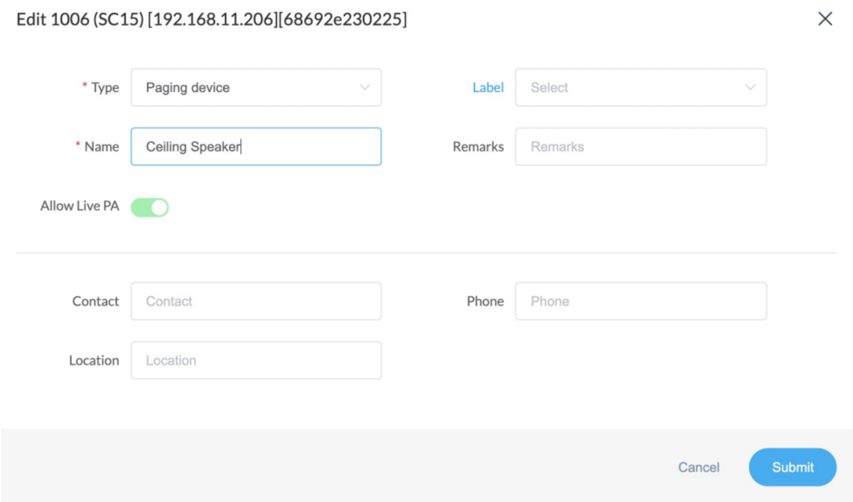
Input server address and SIP account credentials to register the device to IP Audio Center. For detailed instructions please refer to the SW15 Network Speaker User Guide.

When each IP Audio device sends register requests to the IP Audio Center, the device number, device type, device model and MAC address will be sent to IP Audio Center as well. Once registered, this device will be found on the IP Audio Center web interface on the **Audio Devices** -> **Devices** screen.



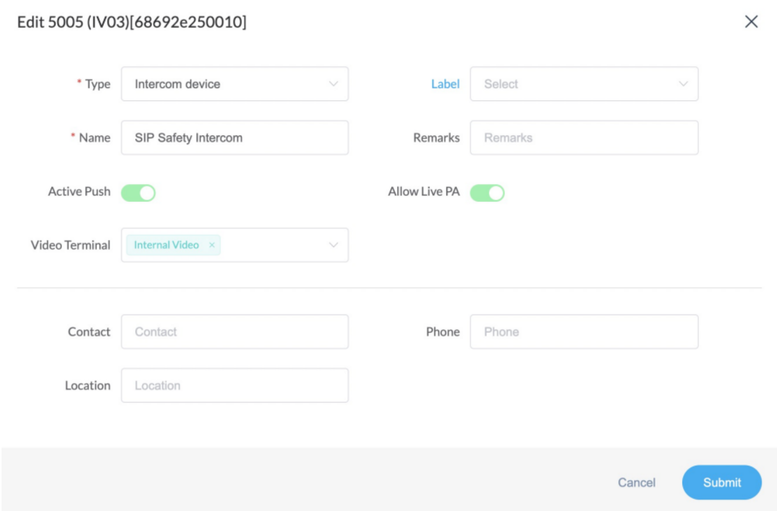
You now need to configure some advanced settings for the registered devices. Please click on **Edit** button and complete the below settings.

Paging device Info



- Type: The device type, can be modified.
- Label: Label is an additional property of the IP audio devices. It will be used by the IP Audio Dispatch Console to filter IP audio devices in a group with the same label.
- Name: A user defined name to identify this device.
- Remarks: Additional information can be added.
- Allow Live PA: Allow this device to be called for live SIP paging.
- Contact: The contact person who is responsible to maintain this device.
- Phone: The phone number if the contact person.
- Location: The location that shows this device's whereabouts.

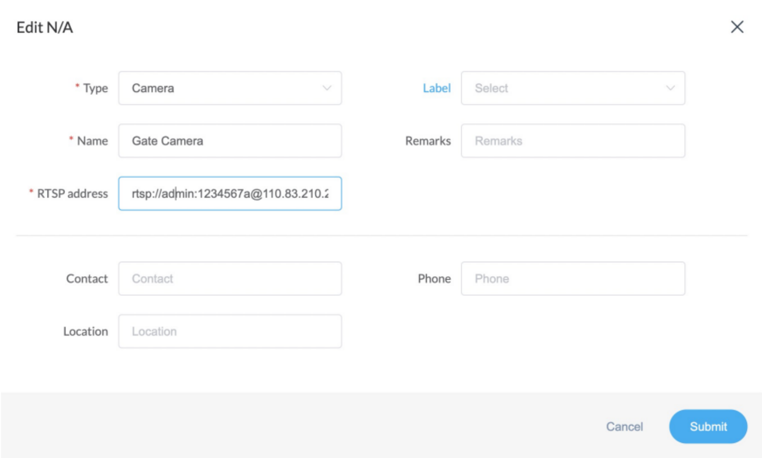
Intercom device Info



- Type: The device type, can be modified.
- Label: Label is an additional property of the IP audio devices. It will be used by the IP Audio Dispatch Console to filter IP audio devices in a group with the same label.
- Name: A user defined name to identify this device.
- Remarks: Additional information can be added.

- Active Push: When the camera and the intercom device are in the internal network (behind NAT), then the intercom device can be pushed the video stream from the video camera to the server. (Note: no need to turn on this option if the server and the video camera are in the same network).
- Video Terminal: Select the video terminal linked with the intercom device (Note: When all devices are in the same network and the Active Push option is ON, it can be only linked with one video terminal. Otherwise, it can link with multiple video terminals).
- Allow Live PA: Allow this device to be called for live SIP paging.
- Contact: The contact person who is responsible to maintain this device.
- Phone: The phone number if the contact person.
- Location: The location that shows this device's whereabouts.

Camera Info



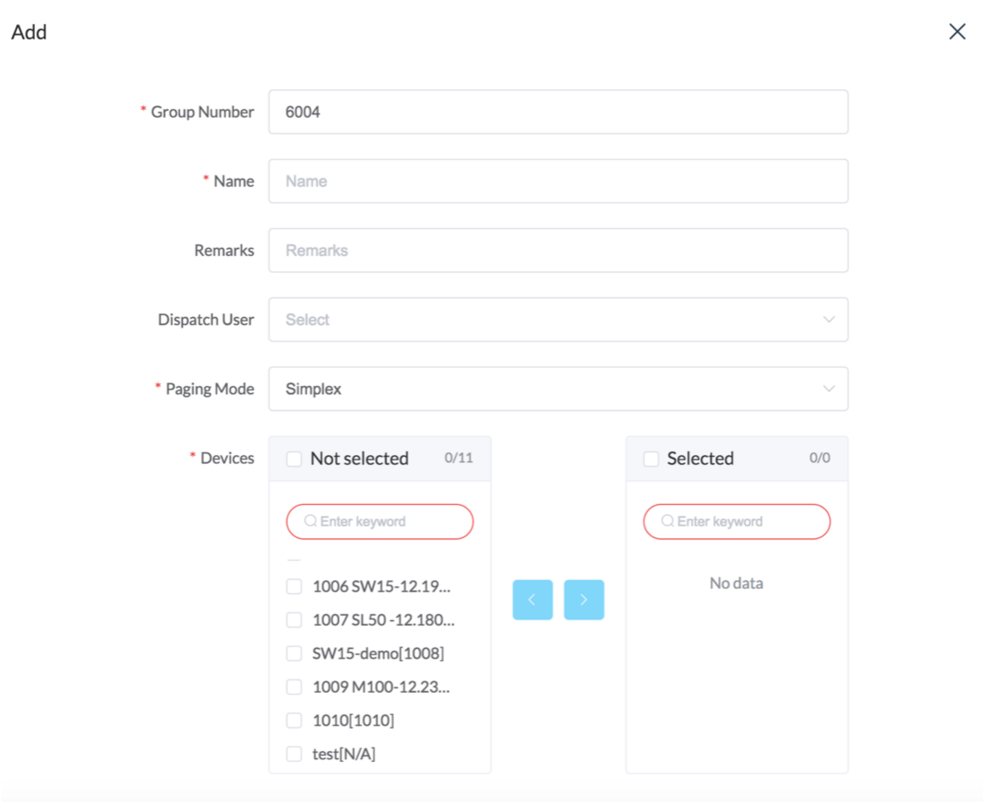
- Type: The device type, can be modified.
- Label: Label is an additional property of the IP audio devices. It will be used by the IP Audio Dispatch Console to filter IP audio devices in a group with the same label.
- Name: A user defined name to identify this device.
- Remarks: Additional information can be added.
- RTSP address: the RTSP address that the video terminal will be used.
- Allow Live PA: Allow this device to be called for live SIP paging.
- Contact: The contact person who is responsible to maintain this device.
- Phone: The phone number if the contact person.
- Location: The location that shows this device's whereabouts.

Setup Paging Groups

After you have registered all IP audio endpoints to the IP Audio Center, next step you have to group them. A group can be the IP audio devices of a specific area or a same type of IP audio devices.

By grouping the IP audio devices, you can achieve group (zone) -based SIP paging, music, notification, etc.

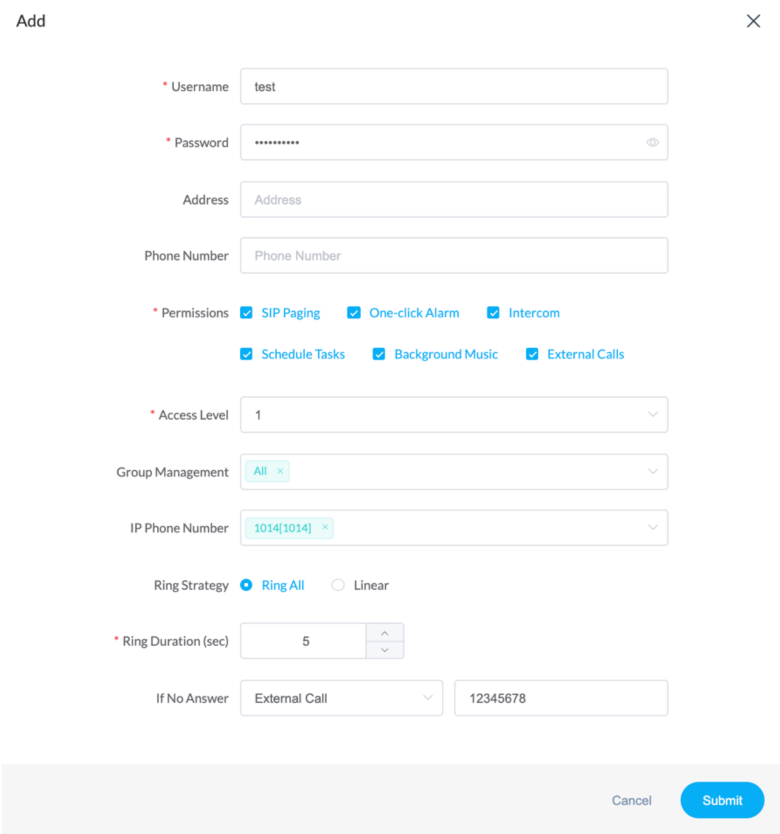
To create paging groups, please go to **Audio Devices** -> **Groups** screen. Click on **Add** button to create a group.



- Group Number: Group number can be dialed from an IP phone or other SIP IP audio device to make live announcements to this paging group.
- Name: A name to identify this group.
- Remarks: Additional information for this group.
- Dispatch User: Dispatch user(s) who has permission to manage this group from the IP Audio Dispatch Console.
- Paging Mode: Simplex mode does not allow member devices to use the talk-back feature. Duplex mode allows all the member devices to use the talk-back feature.
- Devices: Select the member devices of this group. Except the network speakers and intercom devices, you have to add the dispatch user's phone(s) to this group and the other phones too, so the dispatch user can manage paging, intercom, phone calls and conference calls from the dispatch console.

Create Dispatch User

Dispatch user needs to be created from **Settings** -> **Dispatch Users** screen. Click on **Add** button to create a dispatch user.



- User Name: The user name of the dispatch user, it will be used by the user to sign in the IP Audio Dispatch Console.
- Password: Password to be used along with the dispatch user name for authentication.
- Address: The contact address of the dispatch user.
- Phone Number: The contact phone number of the dispatch user.
- Permissions: The operations and features the dispatch user can access on the IP Audio Dispatch Console.
- Access Level: There are 12 access levels for the dispatch users, from the highest access level 1 to the lowest level 12, the higher access level user can override the lower access level user's operations includes telephony features and tasks.
- Group Management: The paging groups that this user can see and manage from the IP Audio Dispatch console.
- IP Phone Number: The dispatch user's IP phone number(s). The phones must be used by the dispatch user.
- Ring Strategy: If the dispatch user has two or more phones, calling any of the phones will cause all of the dispatch user's phone to ring, but you can set to ring them all or ring them one by one using the Ring Strategy parameter.
- Ring Duration: The time duration of the ringtone go on in second.
- If No Answer: When the Ring Duration is not 0, admin can set up the call transfer destination for this operator using the If No Answer option. Destination can be Hang-up, an Extension number, another

Dispatch User, or External Call. (Note: when choosing the External Call option, ensure you have the SIP trunks registered, and Outbound Rules created for dialing outbound calls. Path: *Telephony/Trunks, Telephony/Outbound Rules*).

Once you have created the dispatch user account, you may give the login credentials to the dispatch users. By using the login credentials, the dispatch users will be able to sign in to the IP Audio Dispatch Console. For more details of how to login and use the IP Audio Dispatch Console, please refer to the ZYCOO IP Audio Dispatch Console User Guide.

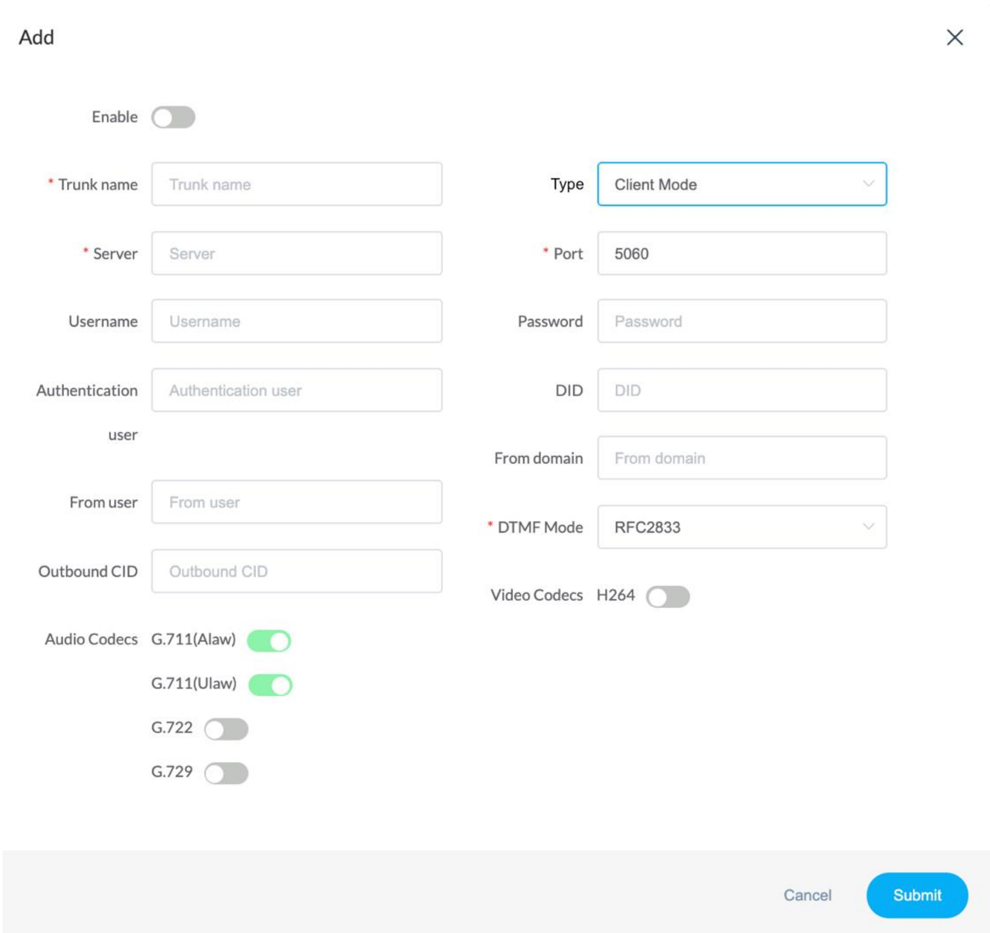
Telephony

ZYCOO IP Audio Center provides SIP trunking, inbound/outbound calling, digital receptionist and more IP telephony features. If you are going to make inbound and outbound phone calls through the IP Audio Center, please read the instructions of this chapter, otherwise you may skip this part.

Trunks

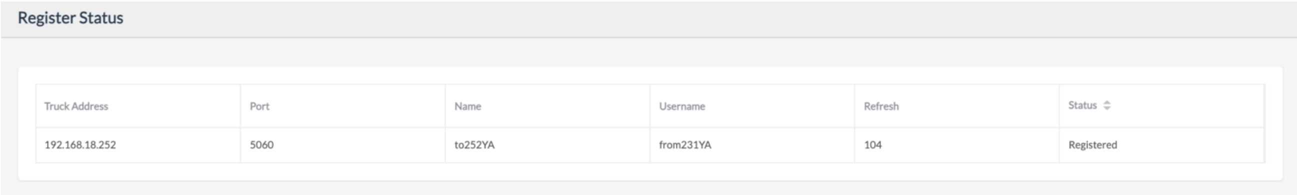
ZYCOO IP Audio Center supports connecting to ITSP and third-party SIP server or IP PBX system using SIP/IMS trunks for inbound and outbound phone calls.

To add a SIP trunk, please go to **Telephony** -> **Trunks** screen. Click on the add button to add a new trunk.



- Enable: Activate this trunk after configurations done.
- Trunk Name: A name to identify this trunk.
- Type: The connection modes include client, and server modes.
- Server: The ITSP server address or third-party SIP server address.
- Port: The SIP port number of the other party.
- Username: The user name of trunk given by the ITSP or the third-party SIP server.
- Password: The password of trunk given by the ITSP or the third-party SIP server.
- Authentication user: Usually it's the same as username, but when the ITSP provides a particular authentication user ID, you have to use this ID here.
- DID: A custom DID number can be assigned to this trunk for inbound call routing purpose.
- From user: Username to be used in "From" header for sending outbound call requests to this trunk.
- From domain: The service provider's domain name.
- Outbound CID: The number you want to display to the called party while dialing out through this trunk. It depends on the service provider whether it works or not.
- DTMF Mode: Used to inform the system how to detect the DTMF key press.
- Audio Codecs:
- Video Codecs: If the ITSP supports video calls then you can enable this parameter, the supported video codec is H.264.

Once the trunk has been created, you may check its register status from **Dashboard** -> **Register Status** -> **Trunk Status** screen.

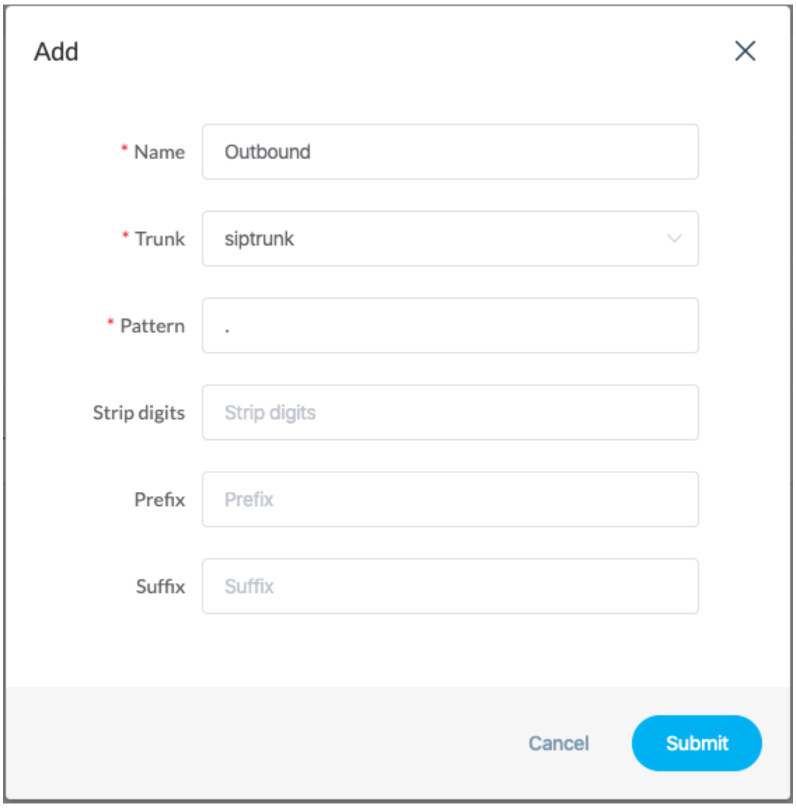

Register Status

Truck Address	Port	Name	Username	Refresh	Status
192.168.18.252	5060	to252YA	from231YA	104	Registered

Outbound Rules

Except trunks, outbound rules need to be configured before you can make outbound phone calls through the IP Audio Center. Each trunk needs an outbound rule to be created, so users can use these outbound rules to call external phone numbers using different trunks.

Go to **Telephony** -> **Outbound Rules** screen, click on Add button to create an outbound rule.



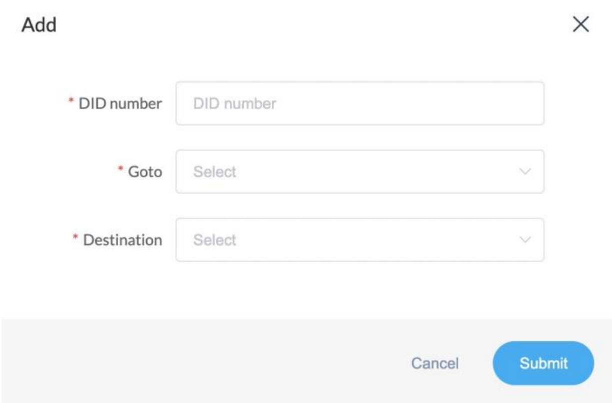
- Name: A name to identify this outbound rule.
- Trunk: Select a trunk to be associated with this outbound rule.
- Pattern: Dial Patterns act like a filter for matching numbers dialed with trunks. The various patterns you can enter are as below:
  X — Refers to any digit between 0 and 9
  N — Refers to any digit between 2 and 9
  Z — Any digit that is not zero. (E.g.1to9)
  . — Wildcard. Match any number of anything. Must match *something*.
- Strip Digits: Number of digits to be deleted in the front of a dialed number before sending it to the trunk.
- Prefix: Add a given prefix in front of the number before sending it to the trunk.

- Suffix: Add a given suffix in the end of the number before sending it to the trunk.

Inbound Rules

The inbound phone calls from the SIP/IMS trunks to the IP Audio Center can be routed to specific IP phone or to digital receptionist. It is controlled by the inbound rules.

Go to **Telephony** -> **Inbound Rules** screen, click **Add** button to add an inbound rule.



DID number: Use the corresponding DID number of a trunk here to configure inbound rule for that trunk.

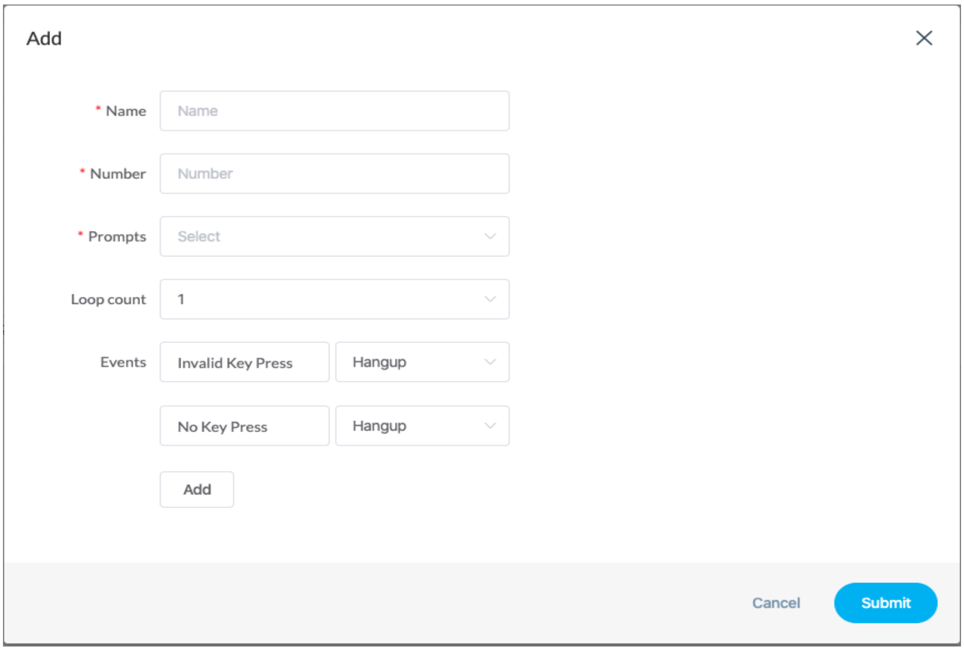
Goto: You can choose to send the inbound calls to IP phone, Paging Group, digital receptionist by selecting or trigger tasks from the 4 types here.

Destination: According to the destination type you have selected, you can select the actual IP phone or digital receptionist as the calling destination here.

Digital Receptionist

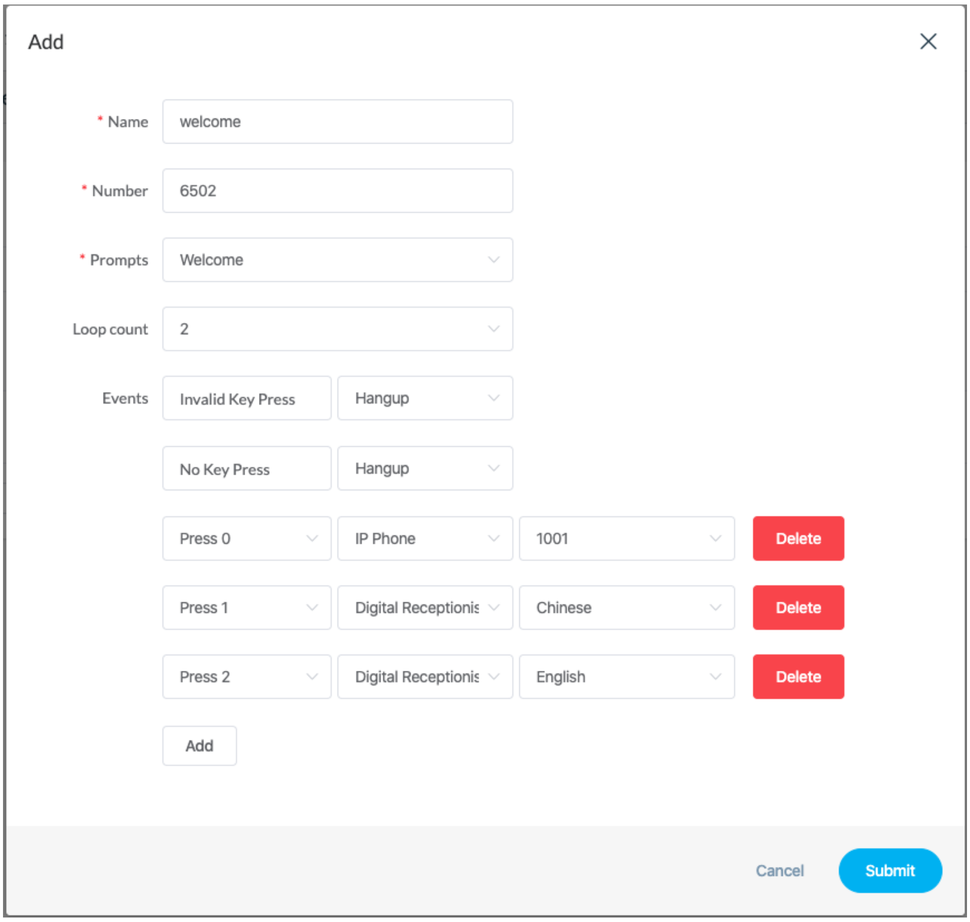
Digital receptionist is also as known as IVR (Interactive Voice Response), it is one of the most important auto call distribution features in an IP phone system. ZYCOO IP Audio Center supports multi-level digital receptionist for handling the inbound phone calls.

Go to **Telephony** -> **Digital Receptionist** screen, click **Add** button to create a digital receptionist.



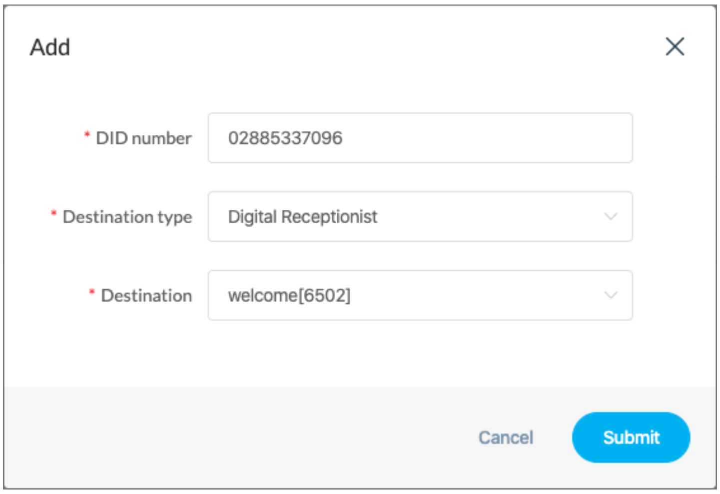
- Name: A name to identify this digital receptionist.
- Number: A number that can be called from IP phones or other SIP endpoints to test the digital receptionist menu options.
- Prompts: The voice prompts for this digital receptionist menu which will guide the callers to reach their desired call destinations in the IP Audio Center system. The voice prompts need to be uploaded from the **Telephony** -> **Voice Prompts** screen.
- Loop count: Select the number of times to playback this IVR prompts before callers pressing a key.
- Events: Events are the digital receptionist options to be configured according to the instructions you have specified in the selected voice prompts. Available key presses could be set from 0 to 9, * and #. If the caller presses the key which are not specified and it will be handled by the "Invalid Key Press" option. And if the caller didn't press any key during the whole digital receptionist process, the call will be handled by the "No Key Press" option.

To create multi-layer digital receptionist menus, you may need to create the sub-layers at first. And then create the top layer. For example, if you want to create a digital receptionist for language selection, create the sub-layer digital receptionist with different languages first, then create the general welcome digital receptionist with statements to guide caller press different keys to enter next level of their desired language of digital receptionists. Like to the figure shown below.



In the above example, when caller landed on the welcome digital receptionist, they will hear the welcome message and tell them to press 1 for Chinese and press 2 for English and press 0 to speaker with the operator.

To make this happen, in the Telephony -> Inbound Rules screen, the inbound rules should select destination type as Digital receptionist, and the destination as the “welcome” digital receptionist.

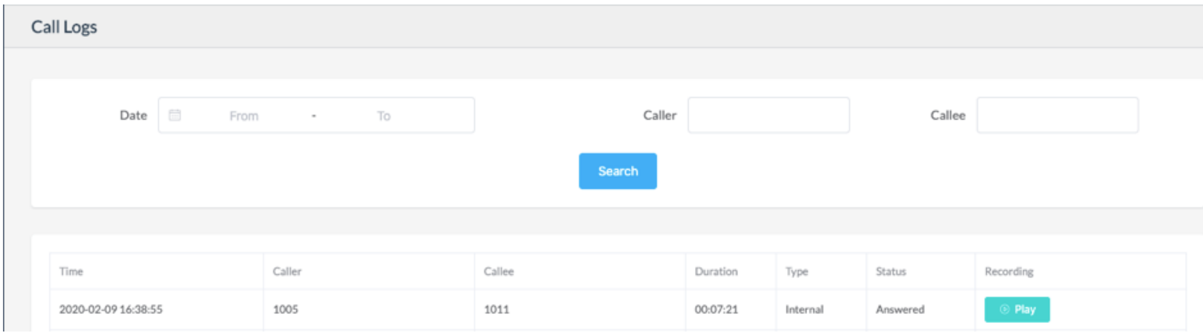


As a result, all inbound calls from the trunk with DID number 02885337096 will all land on the welcome digital receptionist.

Reports

Call Logs

IP Audio Center admin user can have a list of the full call logs of the system from **Reports** -> **Call Logs** screen. And you can filter the call logs by date, caller and callee.



The properties of the call logs are as below.

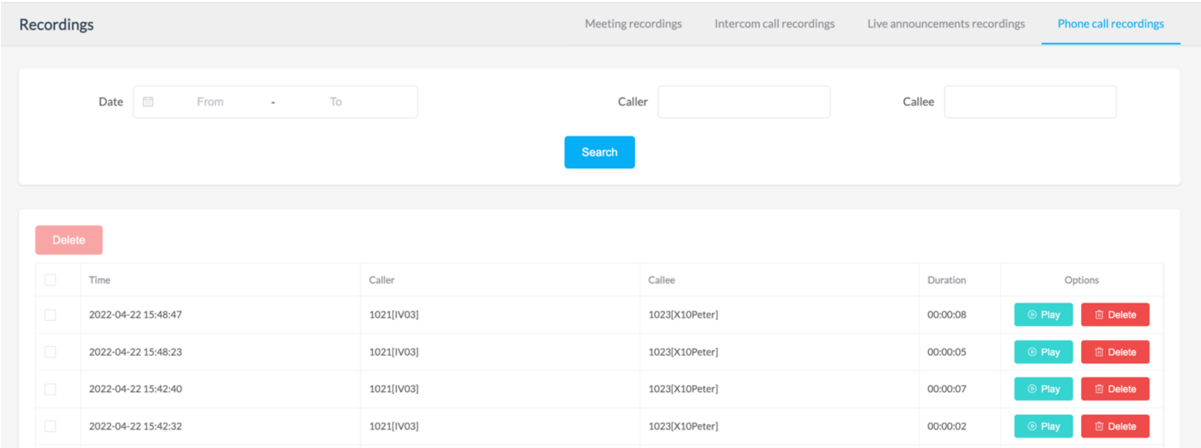
- Time: The time when the call started.
- Caller: The number of the party had made the call.
- Callee: The number of the party had been called.
- Duration: The talk time of the call.
- Type: The type of the call, internal, inbound or outbound.
- Status: The status of the call.
  Answered: The call had been connected successfully.
  Canceled: The caller canceled the call before the called party answering it.
  No Answer: The called party missed the call.
  Busy: The called party is on the phone.
  Failed: The call failed to be connected to the called party.
- Recording: If the call has been recorded, there will be a Play button available, you can click on the button to review the call recordings.

Recordings

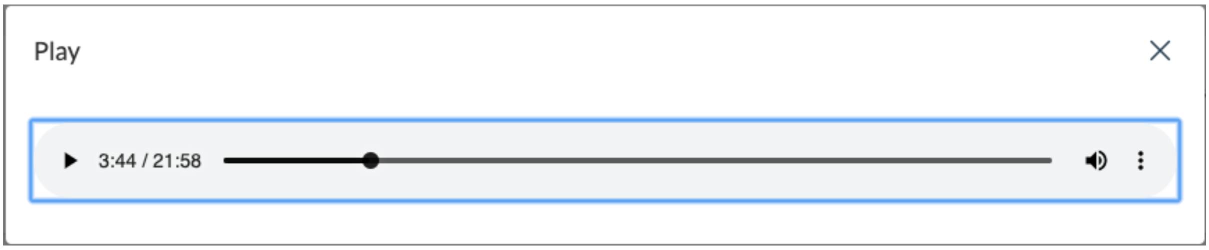
Phone calls, live announcements, intercom calls and meetings in the IP Audio Center will be automatically recorded. You can search the recording list, playback and download the recordings from the **Reports** -> **Recordings** screen.

Phone Call Recordings

All the phone call recordings can be found under the **Phone call recordings** menu.

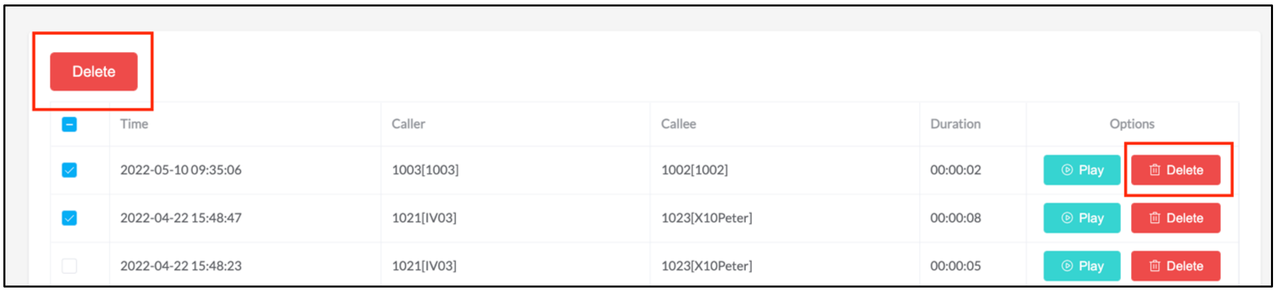


You can search call recordings by date, caller or the callee. And you can playback the recording online by simply click on the **Play** button.



And by click on the ⋮ button you can download this recording to your PC.

Or simply by click on the **Delete** button to delete a specific recording, bulk delete is also support by selecting multiple recordings.



IP phone call recordings can only be viewed and managed by IP Audio Center admin user, the IP Audio Dispatch users have no access to the IP phone call recordings.

Live Announcements Recordings

All live announcements made through the IP Audio Center will be recorded, you can find the recordings under the **Live announcement recordings** menu.

The IP Audio Dispatch users can view the live announcement recordings too, but only the live announcements made by themselves.

Intercom Call Recordings

The recordings of all intercom calls can be found under the **Intercom call recordings** menu. These calls are either initiated from the intercom device or calls to the intercom devices. If it's a video intercom call, then only the voice communication will be recorded.

Just like the IP phone calls, you can filter the recordings by date, caller and callee. And the recordings can be playback online and can be downloaded.

The IP Audio Dispatch users can view the intercom call recordings too, but only the intercom calls made to their dispatch phone or from their dispatch phone to the intercom devices.

Meeting Recordings

Meetings are organized by dispatch users from the IP Audio Dispatch Console. There's a unique meeting ID for each of the meetings. These IDs can be shared among IP Audio Center admin and the dispatch users for identifying a specific meeting.

Settings

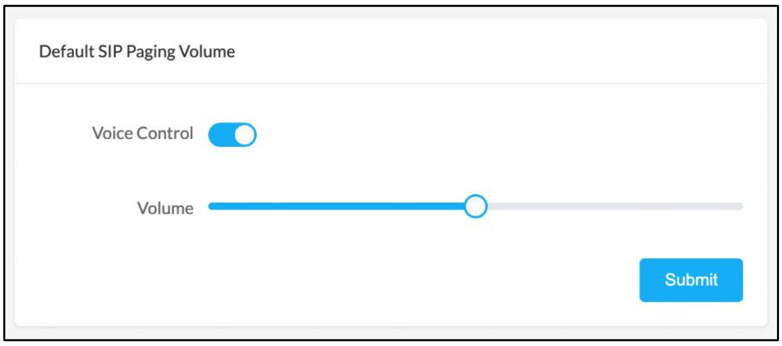
System Settings

On the Settings -> System Settings screen, there are some global settings including default volume, indication tone and feature codes.

Default SIP Paging Volume

During a live SIP paging, volume of the speakers will be forced to use the default SIP paging volume configured here. This is because the speakers might be installed on different locations and configured with different volume levels natively. To ensure the audiences can hear the live announcements, you can setup default SIP paging volume to override the volume settings on the speakers.

After SIP paging, the volume level will be restored.



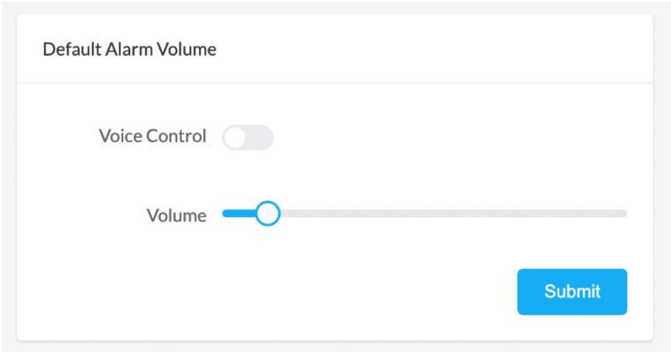
- Voice Control: Enable/Disable the option of forcing all devices to be set to the preset volume level when making SIP paging.
- Volume: Default Paging volume level for SIP paging, range 1-9.

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

Default Alarm Volume

The default alarm volume is the same as default SIP paging volume, to ensure the alarm sound is audible to all targeted areas it's better setup a higher default volume level.

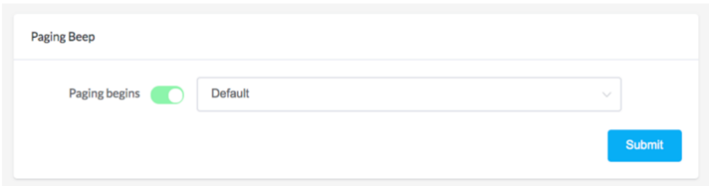
After alarm is stopped, the volume level will be restored.



- Voice Control: This option is mandatory enabled for forcing all devices to be set to the preset volume level when making SIP paging.
- Volume: Default Alarm volume level for alarm, range 1-99.

Indication Tone

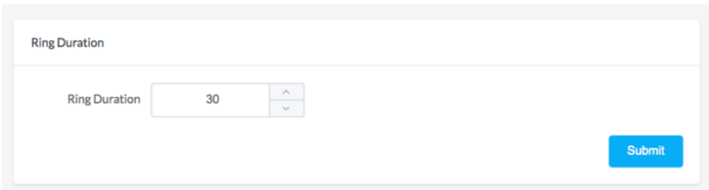
To draw attention of the audience before you making a live announcement, an indication tone can be played before you making the announcements.



If this feature is enabled, both the spokesman and the network speakers will hear the chosen prompt tone, once the tone is finish, the spokesman can now begin the speech. You may manage the prompt tones under -> *Settings/Audio Prompts*

Ring Duration

To set up the time duration of how long the prompt tone should ring.



SIP Settings

In the SIP Settings page, user can modify the relevant parameters of the SIP protocol, such as registration port, transport protocol , RTP port range.

UDP: SIP protocol uses UDP by default, you can modify the default UDP port number here.

TCP: Enable/Disable TCP protocol and specify the port number.

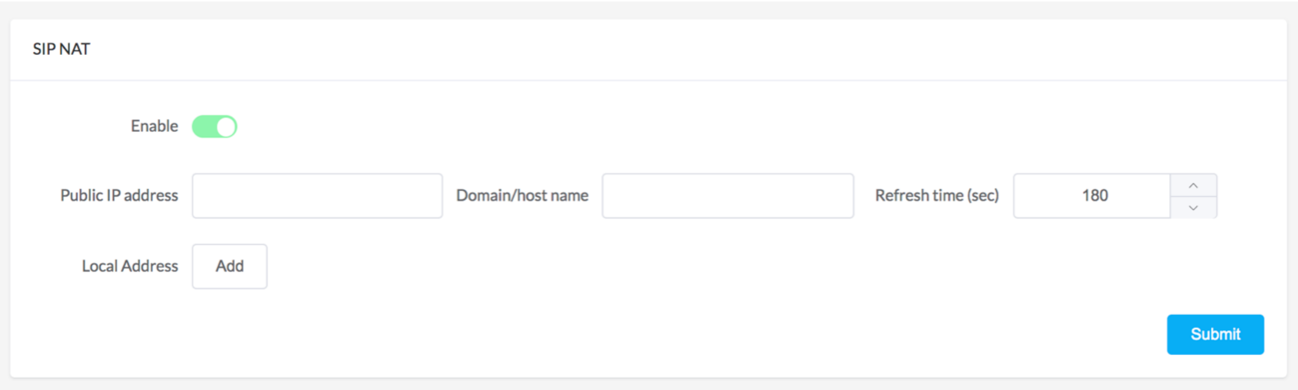
TLS: Enable/Disable TLS protocol and specify the port number.

RTP Start Port: Specify the start port of the RTP stream port range.

RTP End port: Specify the end port of the RTP stream port range.

SIP NAT

If the IP Audio Center is running behind NAT and users need to register end-point devices remotely, then they need to configure the SIP NAT setting parameters in this section.



- Enable: Enable/Disable the SIP NAT feature.
- Public IP Address: If the system is behind NAT, then it needs to set the public IP address to the Zycoo outgoing SIP package. Please input the public IP address of where the system is located here.
- Domain/Host name: If you are using a dynamic IP, you can specify a domain/host name, and the system will perform the address queries periodically.
- Refresh time: Set the time for the system to periodically perform address query.
- Local Address: If the user need to set up remote extensions and need to specify the local network address after setting the public IP address or domain name/host name. Please specify the local address here. For example: 192.168.1.0/255.255.255.0

Modbus Settings

The IP Audio Solution can connect third party Modbus gateway and support Modbus protocol fire systems through Modbus protocol. After Modbus feature is on, user can use Dispatch Console to configure related services.

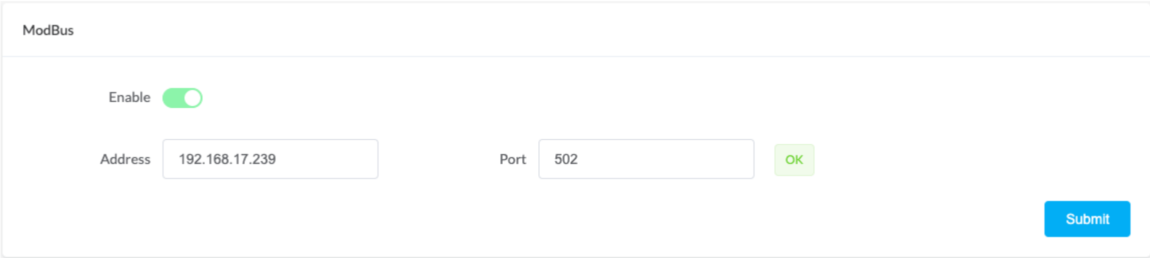


Table 5.1.7-1：Modbus Settings

Enable: Enable/Disable Modbus services

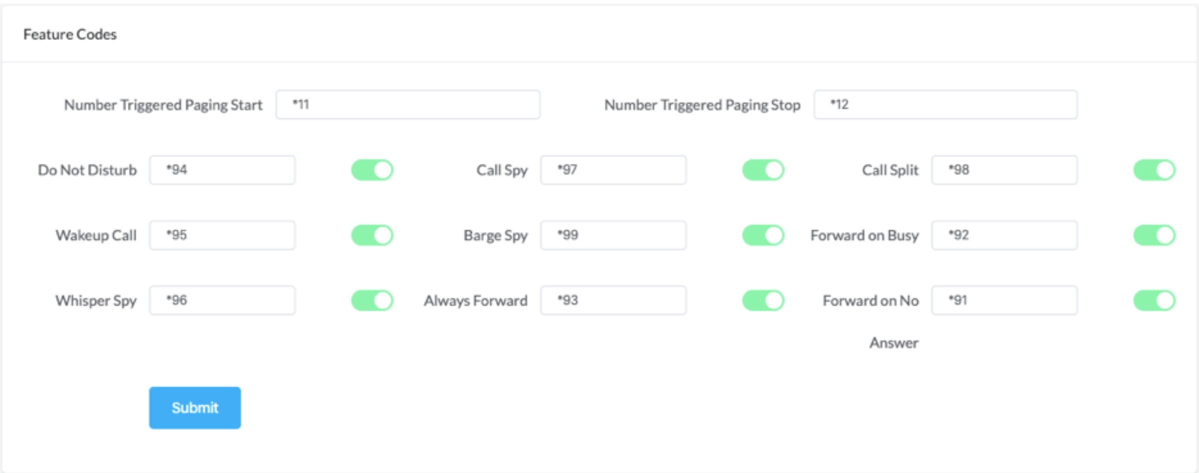
Address: Modbus server address

Port: Modbus server port number

Connection Status: Connection status with the Modbus server

Feature Codes

Feature codes are a series of codes that can be dialed from the IP phones or other IP audio endpoints with a keypad to achieve some specific calling or broadcasting features.



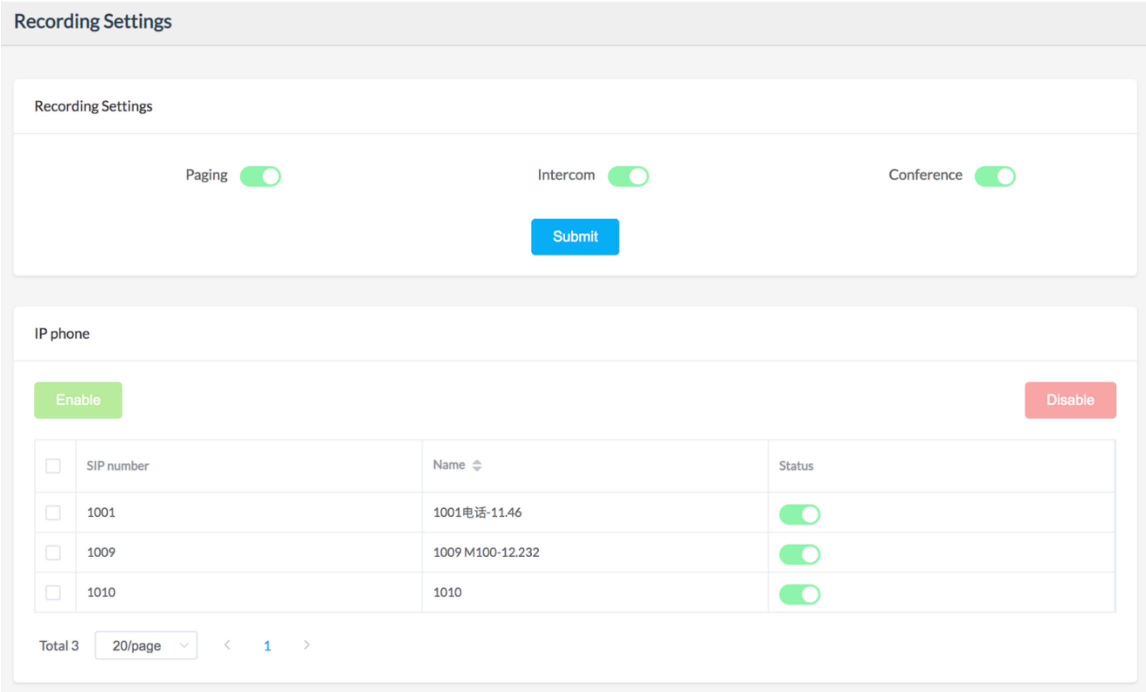
- Number Triggered Paging Start: The number triggered paging tasks created from the IP audio dispatch console are triggered by dialing the alarm start feature code followed by the task number.
- Number Triggered Paging Stop: To stop the number triggered paging tasks, dial the alarm stop feature code and followed by the task number.
- Do Not Disturb: Dial this code to Set the IP phone to Do Not Disturb (DND) mode, dial the code again to disable DND.

- Call Spy: When an IP phone or intercom device is in call, the dispatch user can dial the Call Spy feature code and followed by the IP phone or intercom device's number to spy on the call.
- Call Split: When an IP phone or intercom device is in call, the dispatch user can dial the call split feature code and followed by the IP phone or intercom device's number to split the call with the other party and establish a new call to the dispatch user.
- Wakeup Call: The IP phone users can schedule a wakeup call by dialing the wakeup call feature code, after the beep sound user enter a 4-digit number of time, for example 0930 to setup wakeup call at 9:30 every day in the morning. To cancel the wakeup call users can simply dial the feature code and after the beep sound then press # to cancel it directly.
- Barge Spy: When an IP phone or intercom device is in call, the dispatch user can dial the barge spy feature code and followed by the IP phone or intercom device's number to barge in to the call to establish a 3-way calling.
- Forward on Busy: Forward on busy will cause all incoming calls to a busy IP phone to be forwarded to another number. IP phone users can dial the forward on busy feature code and followed by the target number to setup forward on busy. To cancel forward on busy, just simply dial the feature code directly without any target number.
- Whisper Spy: When an IP phone or intercom device is in call, the dispatch user can dial the whisper spy feature code and followed by the IP phone or intercom device's number to whisper spy this call. The dispatch user can hear the conversation and talk to the whisper spied user but the other party cannot hear the dispatch user's voice.
- Always Forward: Always forward will cause all incoming calls to an IP phone to be always forwarded to another number. IP phone users can dial the always forward feature code and followed by the target number to setup always forward. To cancel always forward, just simply dial the feature code directly without any target number.
- Forward on No Answer: Forward on no answer will cause the incoming calls that no one answers after the phones ringing timeout (30 seconds). IP phone users can dial the forward on no answer feature code and followed by the target number to setup forward on no answer. To cancel forward on no answer, just simply dial the feature code directly without any target number.

Recording Settings

On the recording settings page, you may configure the recording related parameters.

Path: *Settings -> Recording Settings*

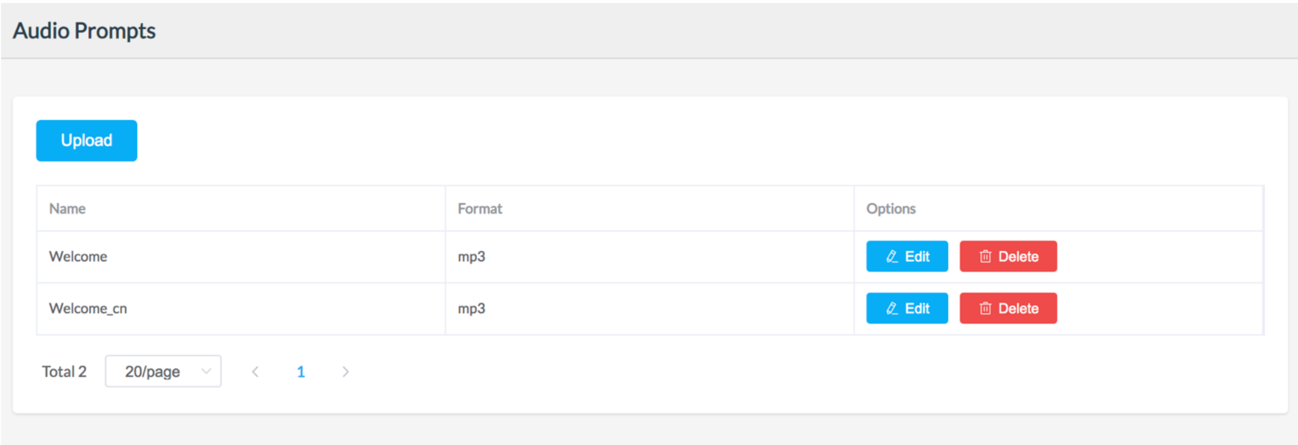


Paging: Enable/Disable paging recording.
Intercom: Enable/Disable intercom recording.
Conference: Enable/Disable conference recording.

Status: Enable/Disable extension recording feature.
Enable: Bulk enable extensions recording feature.
Disable: Bulk disable extensions recording feature

Audio Prompts

The Audio Prompts page is used to upload and manage the audio files used by the IVR and ring tones. Through this page, you can upload, delete, rename the audio files. Click on the 'Upload' button to upload the specific audio file.
Path: Settings/Audio Prompts



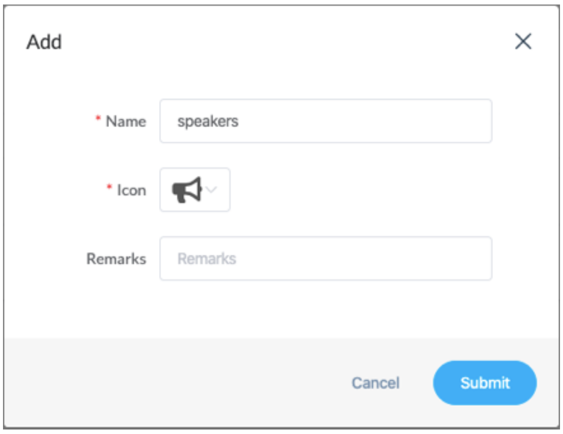
Note:
1) The system supports custom files in mp3, wav, gsm file format and less than 15MB.

2) The file requirement of wav format is: 16 bits, 8000 Hz, mono.

Labels

Label is an additional property of the IP audio devices. It will be used by the IP Audio Dispatch Console to filter IP audio devices in a group with the same label.

The labels are managed on the **Settings** -> **Labels** screen. To create a label please click on the **Add** button.



- Name: A name to be identify the label, and it's better be concise.
- Icon: Select an icon for this label.
- Remarks: Additional description for this label.

After you have created the labels, make sure on the **Audio Devices** -> **Devices** screen, assign a label to each of the IP audio endpoints.

Then on the IP Audio Dispatch Console, you can filter the endpoints with the label you have created. Like the figure below.

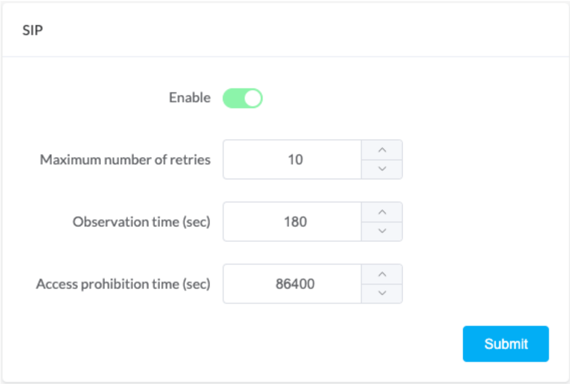
Click on the Speaker icon, it will filter all endpoints that you have assigned with the Speaker label.

Security Center

The security center is used to configure and maintain the intrusion of SIP and SSH protocols on the network. When a certain IP address illegally requests the server within the specified time range for the specified number of times, the system will automatically add that IP address to the Blacklist and block that IP address.

Intrusion Prevention

Configure the relevant parameters of the intrusion prevention services.

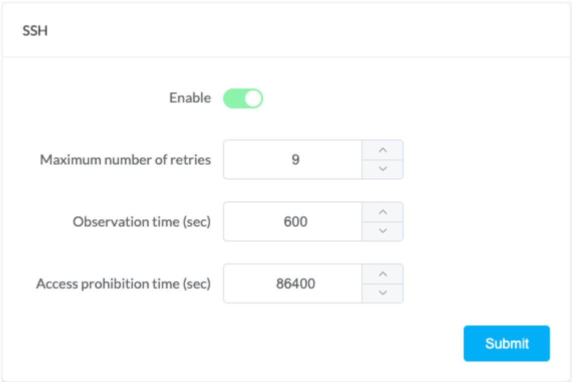


Enable: Enable/Disable the SIP intrusion prevention service.

Max number of retries: The maximum number of retries allowed for the same IP address to access during one observation time period.

Observation time: Specify the observation time in seconds.

Access prohibition time: Specify the time of the banned IP address to stay in the blacklist.



Enable: Enable/Disable the SSH intrusion prevention service.

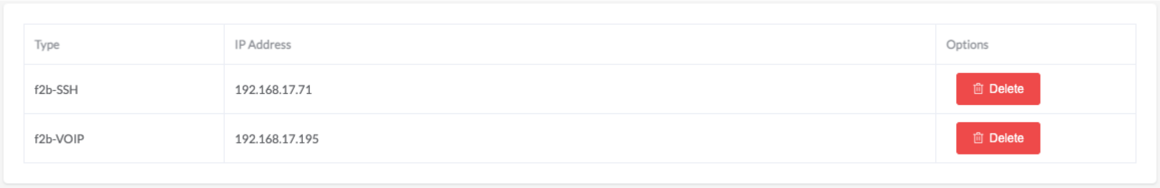
Max number of retries: The maximum number of retries allowed for the same IP address to access during one observation time period.

Observation time: Specify the observation time in seconds.

Access prohibition time: Specify the time of the banned IP address to stay in the blacklist.

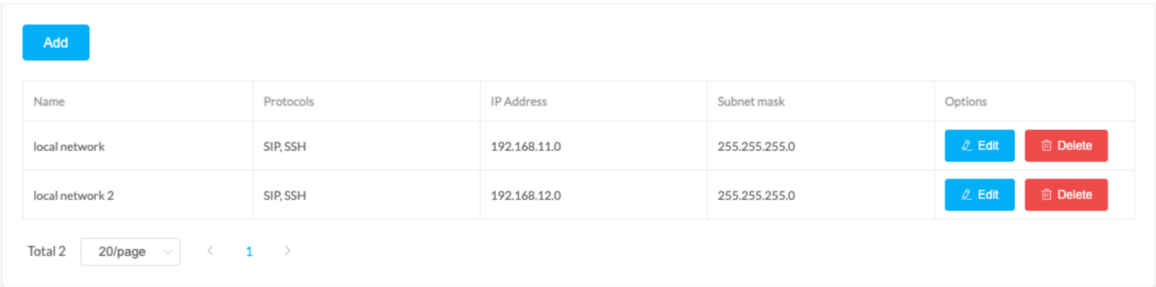
IP Blacklist

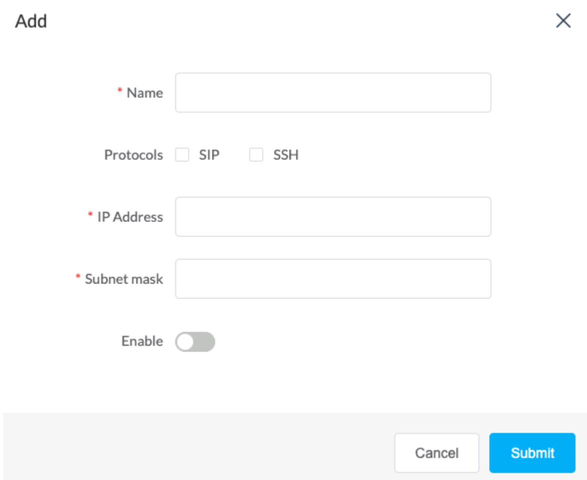
IP addresses in the Blacklist is forbidden to access the system. User can view and/or delete the IP address in the list to cancel the access restriction to that IP address.



IP Whitelist

User can add trusted IP addresses or network segments to this system, all trusted address and segments will not be subject to verification restrictions in the intrusion prevention.





Name: The name of whitelist rule.

Protocols: Specify SIP or SSH protocol to take effect on this rule.

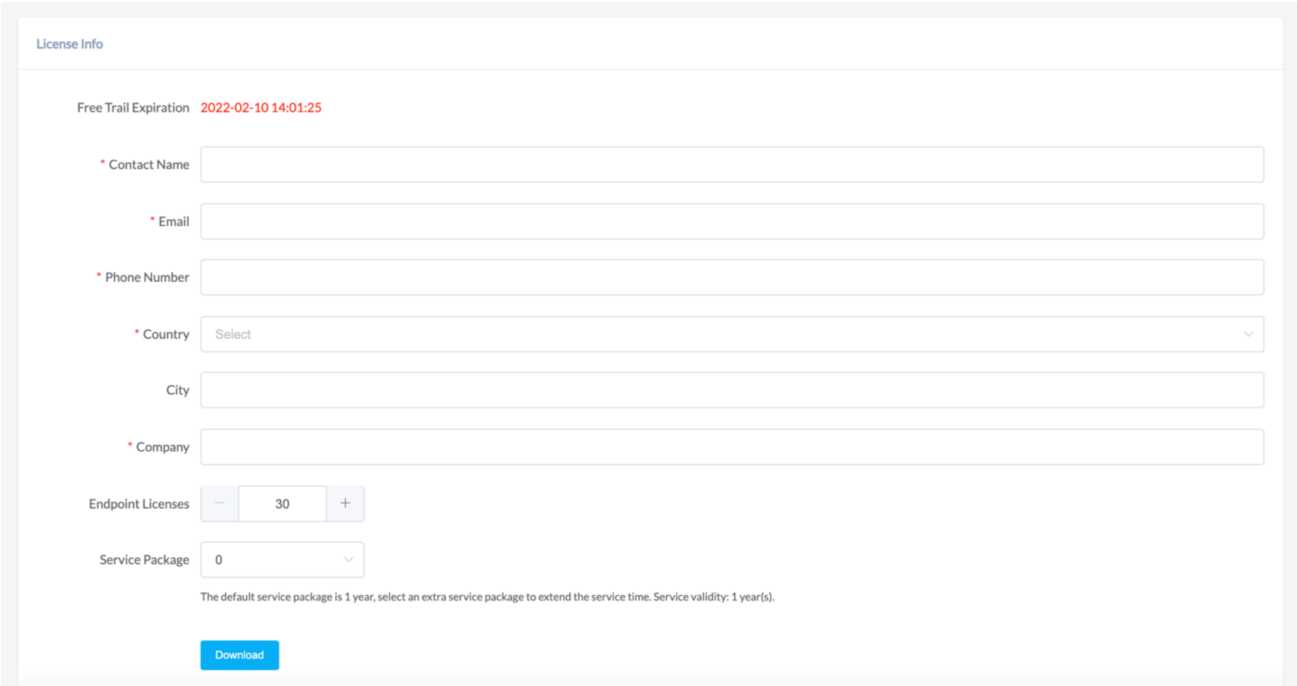
IP Address: Trusted IP address or network segment.

Subnet mask: Set the subnet mask of the address.

Enable: Enable/Disable this whitelist rule.

License

Each system is given a free 45-day trial period, and after the trial period, you need to upload the license key for activating the IP Audio Center. Please fill in the require information section within the License Info such as name, email, etc. Then, click on the Download button to download the license file and please send it to distributor or sales. Please click on the Upload button to upload license key file provided by the distributor or sales.



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.