

IP Audio Center User Guide



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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
- Customized Device Label
- System Logs
- RTSP Address Association
- Call Logs
- Inbound/Outbound Call Control
- Call Forward
- Google TTS (Text-to-Speech)
- SIP Voice Intercom

- Multiple Dispatch User
- Customized Group Number
- Customized IVR Voice Prompts
- Broadcast Begin Indication Tone
- Do Not Disturb (DND)
- Wakeup Call
- Emergency Broadcasting
- Live SIP Paging
- Paging Group (Zone)
- Endpoints Status Monitoring
- Call Spy
- Call Split
- API Support
- Intercom Recording
- Zone-based Paging
- Video Intercom
- Whisper Spy
- 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



- 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

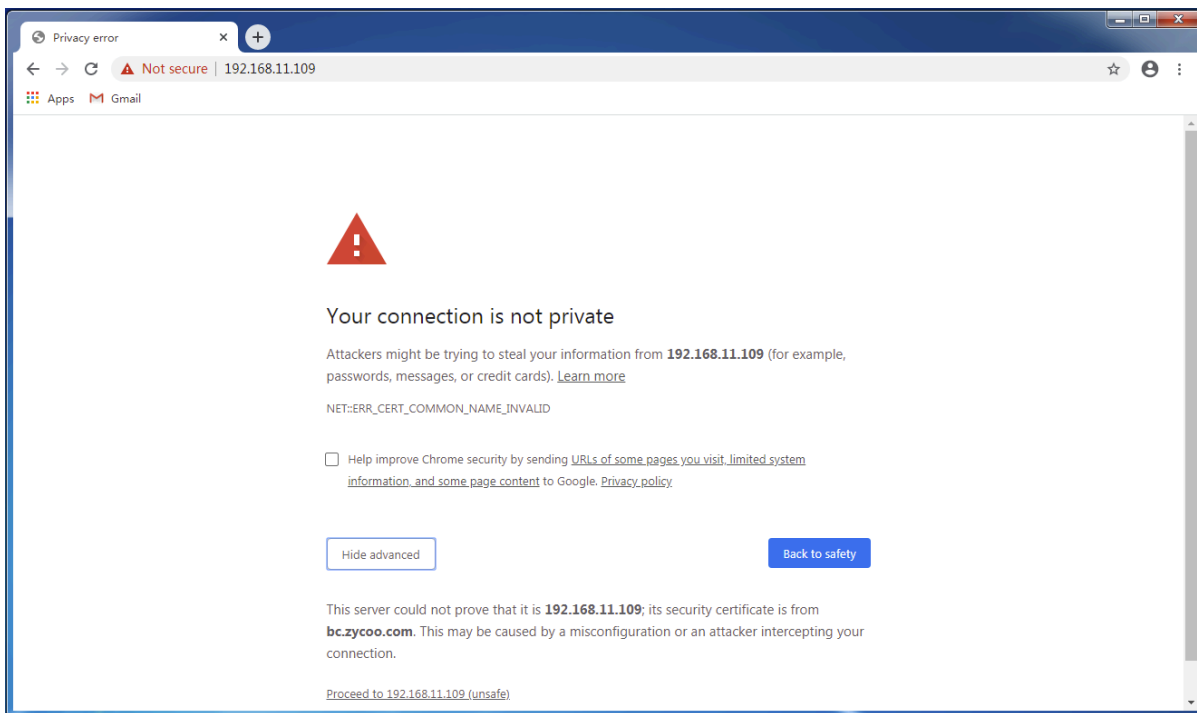
Default Feature Codes

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.

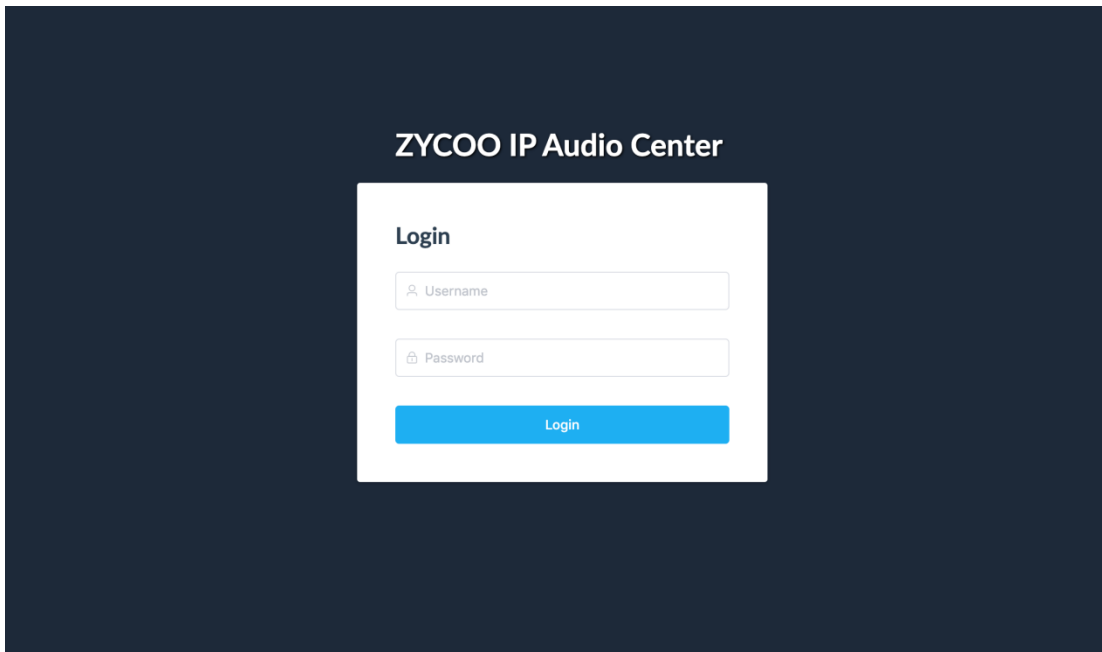


Please click on “Proceed to 192.168.11.109 (unsafe)” to continue accessing the web interface.

Notice:

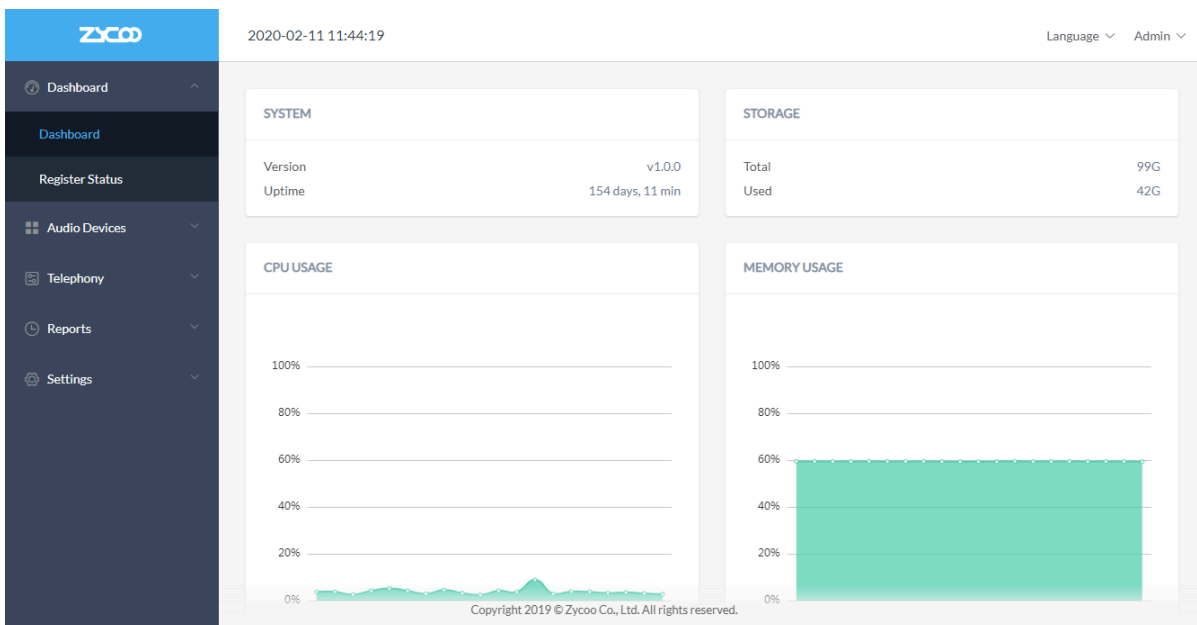
Your web browser must be Google Chrome 69 (or newer), Opera, IE10 (or newer), Firefox 65 (or newer).

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.

- 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.

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, dispatch user management, endpoint device label management 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

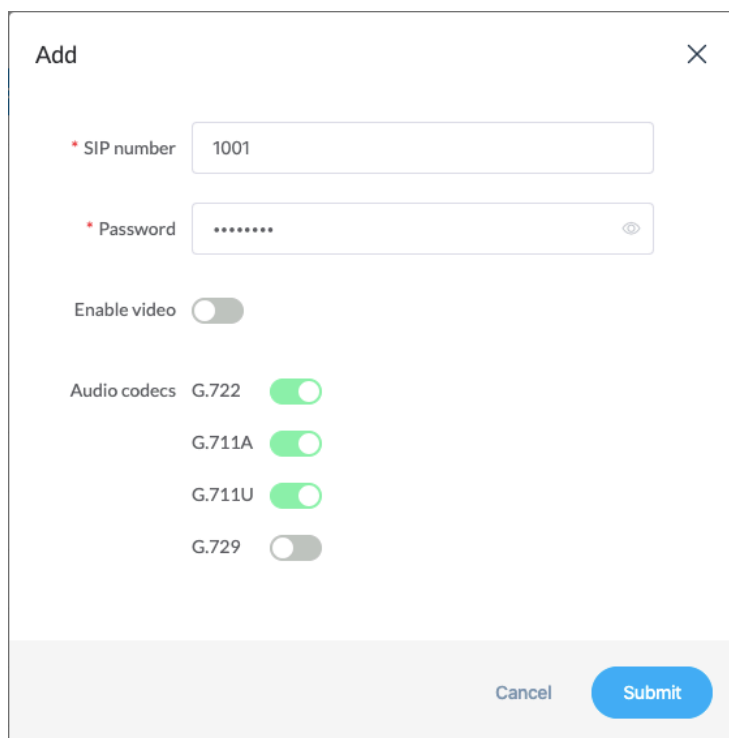
Before you can connect ZYCOO IP Audio Dispatch Console to the IP Audio Center, the following settings need to be done.

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.



The screenshot shows a modal dialog box titled "Add" with a close button (X) in the top right corner. It contains the following fields and controls:

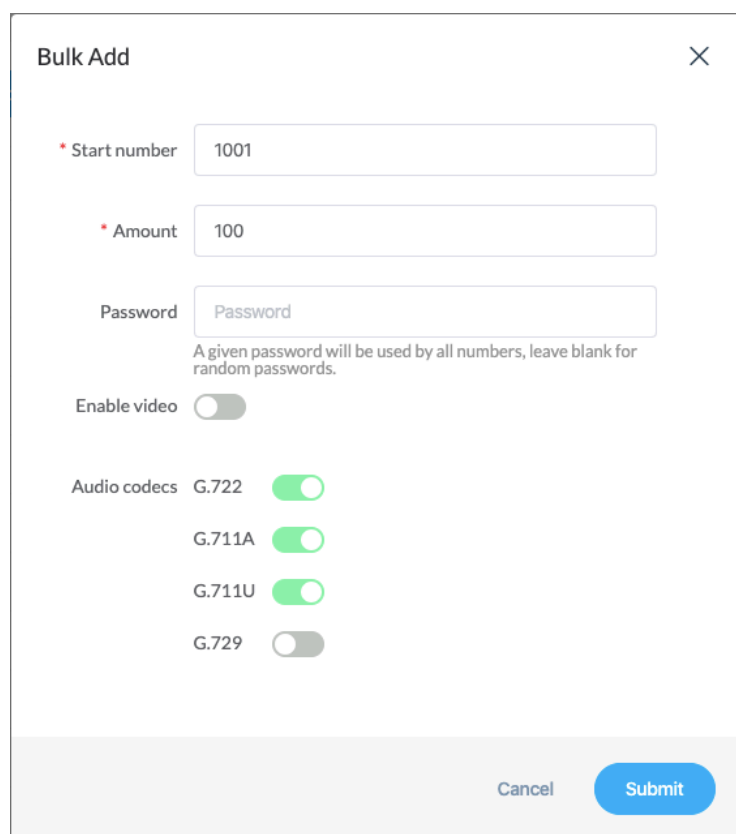
- * SIP number:** A text input field containing "1001".
- * Password:** A password input field with masked characters "*****" and an eye icon to toggle visibility.
- Enable video:** A toggle switch currently turned off.
- Audio codecs:** A list of four codecs with corresponding toggle switches:
 - G.722: Switched on (green).
 - G.711A: Switched on (green).
 - G.711U: Switched on (green).
 - G.729: Switched off (grey).

At the bottom right, there are two buttons: "Cancel" and "Submit".

- **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.

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

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

A screenshot of a 'Bulk Add' dialog box. The dialog has a title bar with 'Bulk Add' and a close button (X). It contains several input fields and toggle switches. The 'Start number' field is labeled with a red asterisk and contains '1001'. The 'Amount' field is also labeled with a red asterisk and contains '100'. The 'Password' field contains 'Password' and has a note below it: 'A given password will be used by all numbers, leave blank for random passwords.' Below the password field is an 'Enable video' toggle switch, which is currently turned off. Under the 'Audio codecs' section, there are four toggle switches: 'G.722' (on), 'G.711A' (on), 'G.711U' (on), and 'G.729' (off). At the bottom right of the dialog are 'Cancel' and 'Submit' buttons.

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.

Device Info

SIP Settings

SIP Account

Audio Codescs

SIP Call Settings

Settings

SIP Account

SIP Account

* SIP Server: 192.168.11.109

* User ID: 1003

Password:

* Register Expiration(S): 200

* Starting RTP Port: 7078

Enable Integration with ☒

ZYCOO IP Audio Center:

Activate: ☒

Submit

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.

Devices

Add

Enter a number to search

SIP number	Name	Type	Label	Remarks	Options
1001	CeilingSPK	Paging device			Edit Delete
1002	HornSPK	Paging device			Edit Delete
1003	CabinetSPK	Paging device			Edit Delete

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

The screenshot shows a web form titled "Edit 1003" with a close button (X) in the top right corner. The form is organized into two columns. The left column contains fields for "Type" (a dropdown menu showing "Paging device"), "MAC" (a text box with "68692e220013"), "Contact" (a text box with "Contact"), "Location" (a text box with "Location"), and "RTSP address" (a text box with "RTSP address"). The right column contains fields for "Label" (a dropdown menu showing "Select"), "Name" (a text box with "CabinetSPK"), "Phone" (a text box with "Phone"), and "Remarks" (a text box with "Remarks"). Below these fields is a toggle switch for "Allow Live PA" which is currently turned on. At the bottom right of the form are two buttons: "Cancel" and "Submit".

- **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.
- **MAC:** The MAC address of the IP audio device.
- **Name:** A user defined name to identify this device.
- **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.
- **Remarks:** Additional information can be added.
- **RTSP Address:** RTSP stream address of the external IP camera. E.g. rtsp://user:pass@192.168.11.61/0 or rtsp://user:pass@192.168.11.61. If you are not sure about this please ask your IP camera vendor.
- **Allow Live PA:** Allow this device to be called for live SIP paging.

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.

Add [Close]

* Group Number: 201

* Name: 1stFloor

Remarks: Remarks

Dispatch User: demo [x]

* Devices

☒ Not selected 1/11

Enter keyword

- ☒ CeilingSPK[1006]
- ☐ SIPIntercom[1007]
- ☐ CabinetSPK[1008]
- ☐ HornSPK[1009]

☐ Selected 0/5

Enter keyword

- ☐ CeilingSPK[1001]
- ☐ HornSPK[1002]
- ☐ CabinetSPK[1003]
- ☐ SIPIntercom[1004]

Cancel Submit

- **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.
- **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.

Add

* Username

Username

* Password

Password

Address

Address

Phone Number

Phone Number

* Permissions

☒ SIP Paging
☒ One-click Alarm
☒ Intercom
☒ Schedule Tasks
☒ Background Music
☒ External Calls

* Access Level

1

Group

Select

Management

IP Phone

Select

Number

Ring Strategy

☒ Ring All
☐ Linear

Cancel

Submit

- **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.
- **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.

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.

The 'Add' dialog box for creating a new SIP trunk includes the following fields and controls:

- Trunk name:** A text input field with a red asterisk indicating it is required.
- Enable:** A toggle switch currently turned on (green).
- Server:** A text input field with a red asterisk indicating it is required.
- Port:** A text input field containing the value '5060'.
- Username:** A text input field.
- Password:** A text input field.
- Authentication user:** A text input field.
- DID:** A text input field.
- From domain:** A text input field.
- From user:** A text input field.
- DTMF Mode:** A dropdown menu with 'RFC2833' selected and a red asterisk indicating it is required.
- Outbound CID:** A text input field.
- Video Codecs:** A section with 'H264' selected and a toggle switch.
- Audio Codecs:** A section with four toggle switches: 'G.711(Alaw)' (on), 'G.711(Ulaw)' (on), 'G.722' (off), and 'G.729' (off).

At the bottom right, there are 'Cancel' and 'Submit' buttons.

- **Trunk Name:** A name to identify this trunk.
- **Enable:** Activate this trunk after configurations done.
- **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.

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.

Add [X]

* Name

* Trunk

* Pattern

Strip digits

Prefix

Suffix

[Cancel] [Submit]

- 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.

The image shows a modal dialog box titled "Add" with a close button (X) in the top right corner. Inside the dialog, there are three labeled input fields, each preceded by a red asterisk indicating a required field:

- DID number:** A text input field with the placeholder text "DID number".
- Destination type:** A dropdown menu with the text "Select" and a downward arrow.
- Destination:** A dropdown menu with the text "Select" and a downward arrow.

At the bottom of the dialog, there are two buttons: a "Cancel" button and a blue "Submit" button.

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

Destination type: You can choose to send the inbound calls to IP phone or digital receptionist by selecting from the 2 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.

The screenshot shows a web form titled "Add" with a close button (X) in the top right corner. The form contains the following fields and controls:

- Name:** A text input field with a red asterisk indicating it is required.
- Number:** A text input field with a red asterisk indicating it is required.
- Prompts:** A dropdown menu with "Select" as the current selection and a downward arrow.
- Loop count:** A dropdown menu with "1" as the current selection and a downward arrow.
- Events:** Two rows of controls. Each row has a button and a dropdown menu.
 - Row 1: "Invalid Key Press" button and a dropdown menu with "Hangup" selected.
 - Row 2: "No Key Press" button and a dropdown menu with "Hangup" selected.
- Buttons:** An "Add" button is located below the Events section. At the bottom right of the form are "Cancel" and "Submit" buttons.

- **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.

Add
×

* Name
welcome

* Number
6502

* Prompts
Welcome

Loop count
2

Events
Invalid Key Press
Hangup

No Key Press
Hangup

Press 0
IP Phone
1001
Delete

Press 1
Digital Receptionist
Chinese
Delete

Press 2
Digital Receptionist
English
Delete

Add

Cancel
Submit

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.

Add
×

* DID number
02885337096

* Destination type
Digital Receptionist

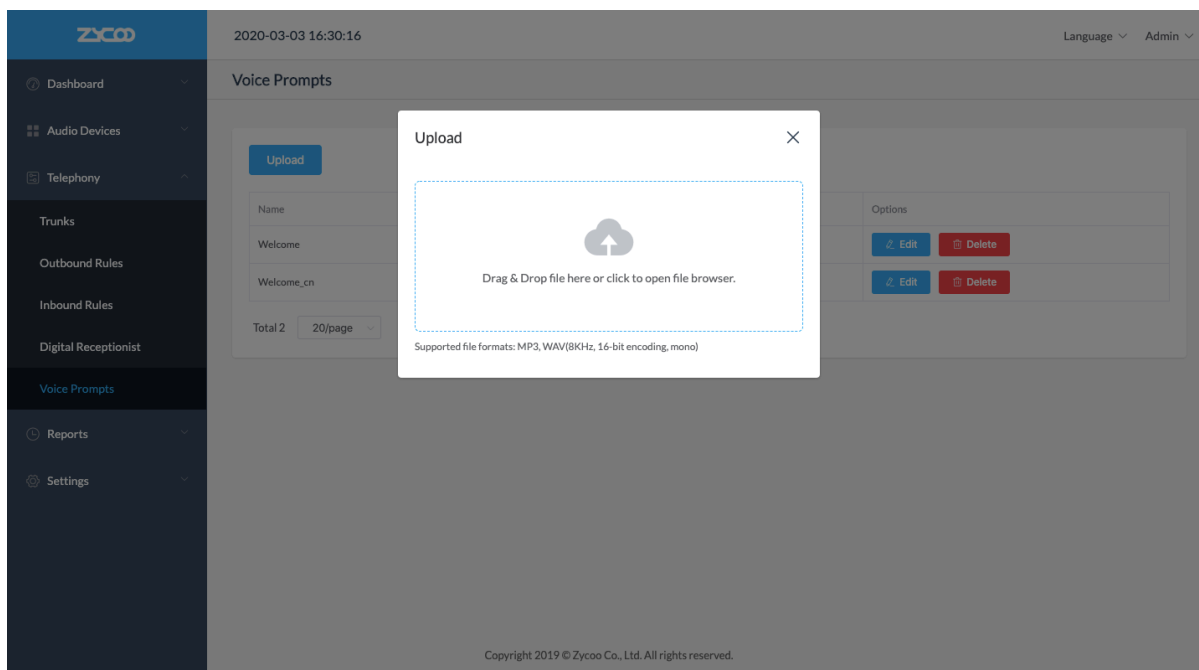
* Destination
welcome[6502]

Cancel
Submit

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

Voice Prompts

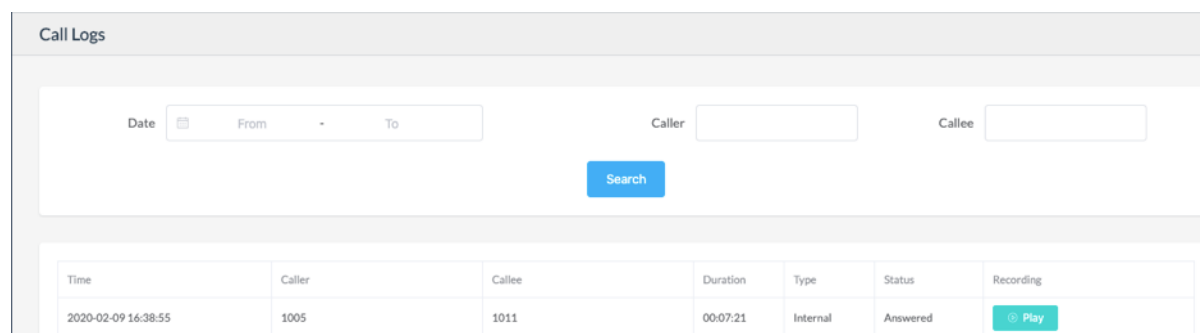
The voice prompts for setting up the digital receptionists are managed on the Telephony -> Voice Prompts screen. Users could upload prerecorded voice prompts files in MP3 or WAV (8KHz, 16-bit encoding, mono) format.



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 screenshot shows the 'Call Logs' interface. At the top, there is a search filter section with a 'Date' dropdown, 'From' and 'To' input fields, a 'Caller' input field, and a 'Callee' input field. A blue 'Search' button is located below these fields. Below the search section is a table with the following columns: Time, Caller, Callee, Duration, Type, Status, and Recording. The table contains one row of data: Time: 2020-02-09 16:38:55, Caller: 1005, Callee: 1011, Duration: 00:07:21, Type: Internal, Status: Answered, and Recording: a green 'Play' button.

Time	Caller	Callee	Duration	Type	Status	Recording
2020-02-09 16:38:55	1005	1011	00:07:21	Internal	Answered	Play

The properties of the call logs are as below.

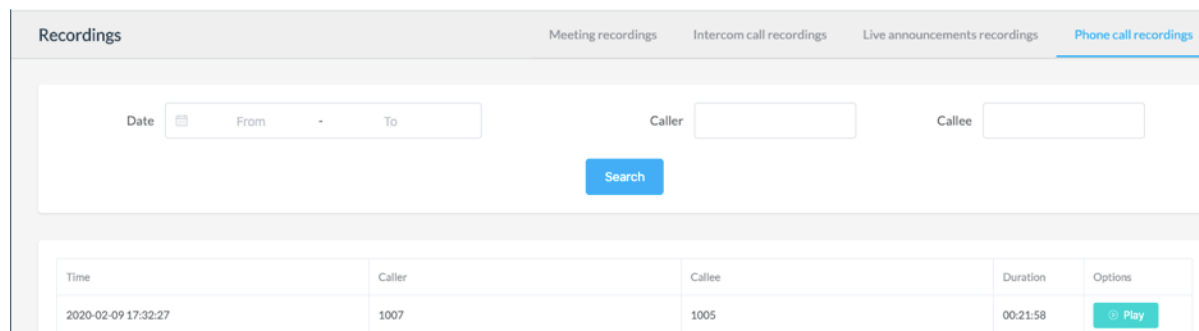
- **Time:** The time when the call started.
- **Caller:** The number of the party had made the call.
- **Callee:** The number of the party had been called.
- **Duration:** The talk time of the call.
- **Type:** The type of the call, internal, inbound or outbound.
- **Status:** The status of the call.
 - 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

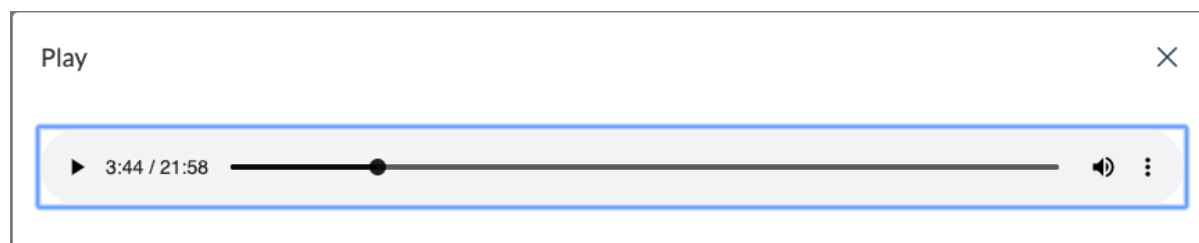
All the phone call recordings can be found under the **Phone call recordings** menu.



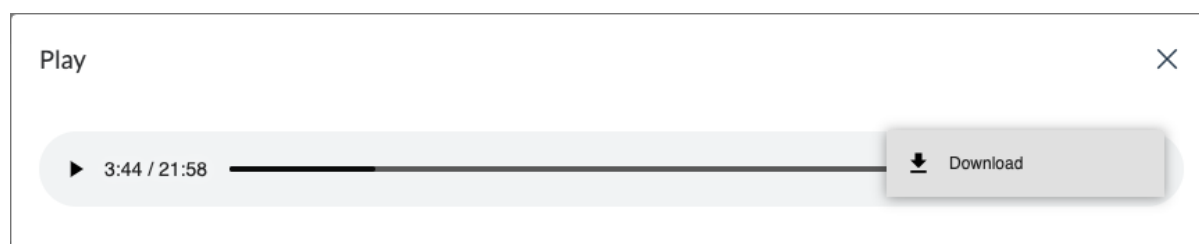
The screenshot shows the 'Recordings' interface with tabs for 'Meeting recordings', 'Intercom call recordings', 'Live announcements recordings', and 'Phone call recordings'. The 'Phone call recordings' tab is active. Below the tabs is a search section with fields for 'Date' (with a calendar icon), 'From', 'To', 'Caller', and 'Callee', and a 'Search' button. Below the search section is a table of recordings.

Time	Caller	Callee	Duration	Options
2020-02-09 17:32:27	1007	1005	00:21:58	Play

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 **Download** button you can download this recording to your PC.



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

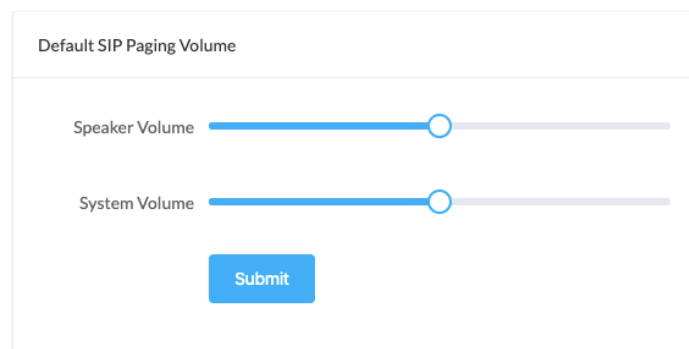
System Settings

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.



Default SIP Paging Volume

Speaker Volume

System Volume

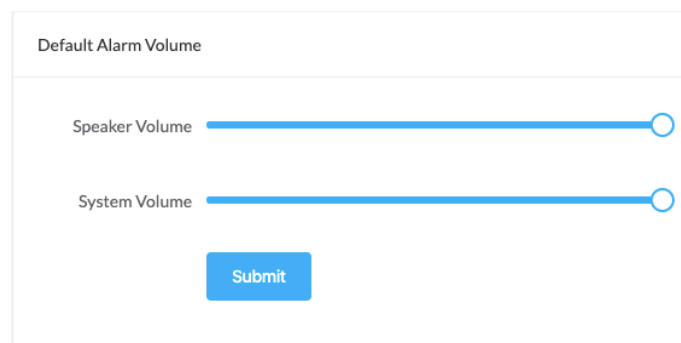
Submit

- Speaker Volume: Default hardware volume level for SIP paging, range 1-9.
- System Volume: Default software volume level for SIP paging, range 1-99.

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.



Default Alarm Volume

Speaker Volume

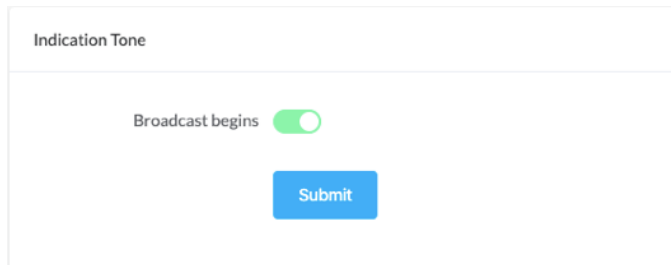
System Volume

Submit

- Speaker Volume: Default hardware volume level for alarm, range 1-9.
- System Volume: Default software 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.

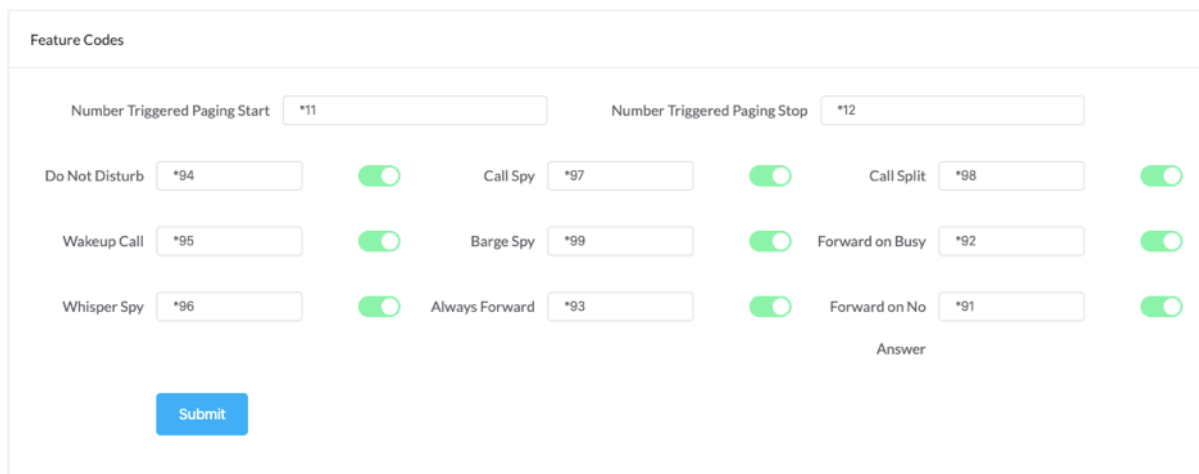


The 'Indication Tone' configuration interface shows a toggle switch for 'Broadcast begins' which is currently turned on (green). Below the toggle is a blue 'Submit' button.

If this feature is enabled, both the spokesman and the network speakers will hear the tone, one the tone is finish, the spokesman can now begin the speech.

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.



The 'Feature Codes' configuration interface displays various settings with input fields and toggle switches. At the top, 'Number Triggered Paging Start' is set to *11 and 'Number Triggered Paging Stop' is set to *12. Below these are three rows of features, each with a code input field and a toggle switch:

Feature	Code	Status
Do Not Disturb	*94	On
Wakeup Call	*95	On
Whisper Spy	*96	On
Call Spy	*97	On
Barge Spy	*99	On
Always Forward	*93	On
Call Split	*98	On
Forward on Busy	*92	On
Forward on No Answer	*91	On

A blue 'Submit' button is located at the bottom left of the interface.

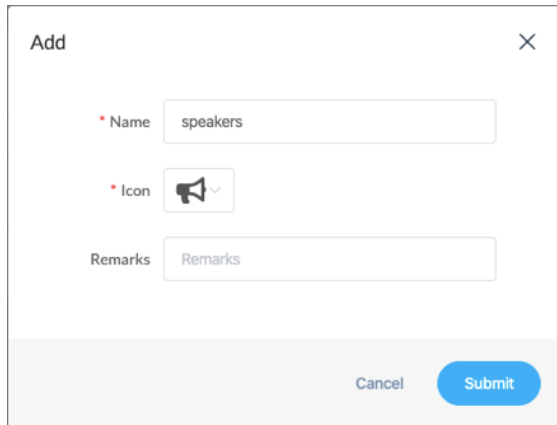
- Number Triggered Paging Start: The number triggered paging tasks created from the IP audio dispatch console are triggered by dialing the alarm start feature code followed by the task number.
- Number Triggered Paging Stop: To stop the number triggered paging tasks, dial the alarm stop feature code 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.

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.

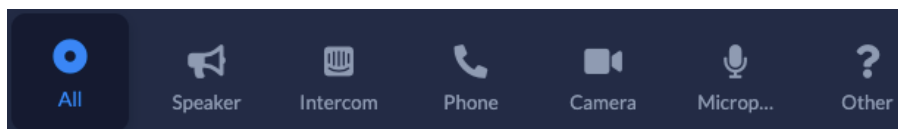
The labels are managed on the **Settings -> Labels** screen. To create a label please click on the **Add** button.



- 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.

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