IAS-L100 IP Audio Server

User Guide

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Contents

1. Preface

1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and manage the IAS-L100 IP Audio Server. By carefully reading and consulting this guide, users could solve the setting and deployment issues of the IAS-L100 IP Audio Server.

1.2 Revision History

Document Version	Applicable Firmware Version	Update Content	Update Date
1.0.11	1.0.11	Updated operating instructions for software version v1.0.11	Nov,2023

2. Overview

2.1 Product Overview

IAS-L100 is a hardware server pre-installed with ZYCOO IP Audio Center, which can provide users with the out-of-the-box feature to be ready to use. The IP Audio Center is a comprehensive public address system that provides centralized IP audio endpoints auto provisioning and management features. To achieve features like background music, group paging, audio conferencing, video linkage, IP phone call, and etc. It is suitable for public safety, smart city, secure community, industry, healthcare, and much more application scenarios.

2.2 Product Specifications

IAS-L100 IP Audio Server Specifications	
Number of IP Audio Endpoints	100 (Max)
Number of Paging Groups (Zones)	Unlimited
Number of MP3 Audio Files	Unlimited
Number of Playlists	24
Number of Timetable Triggered Broadcast Tasks	Unlimited
Number of Number Triggered Broadcast Tasks	Unlimited
Number of SIP Concurrent Calls	100 (Max)
Number of Simultaneous Conference Attendees	100 (Max)
Number of SIP Paging Network Speakers	100 (Max)

2.3Product 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
- Do Not Disturb (DND)
- Wakeup Call
- Emergency Broadcasting
- Live SIP Paging
- Paging Group (Zone)
- Endpoints Status Monitoring
- Call Spy
- Call Split
- API Support
- 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
- Caller ID
- Streamed Media Music
- Device Fault Report
- Customized Device Number
- Device Location Mark
- Call Recording
- Feature Codes
- Call Transfer
- Auto Receptionist (IVR)

2.4 Product Integration

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

Speakers

- ZYCOO Network Speakers
- 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.

Intercoms

- ZYCOO SIP Safety Intercom
- Third-party SIP enabled Intercom

IP Phones

- ZYCOO CooFone series IP phones
- Third-party SIP Phones

Video Devices

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

SIP Paging Gateway

- ZYCOO SIP Paging Gateways

3. Dashboard

3.1 Accessing the Web GUI

Open your web browser, input the default server address and press enter. (Please select the port address according to the actual situation)

WAN port: 192.168.1.100

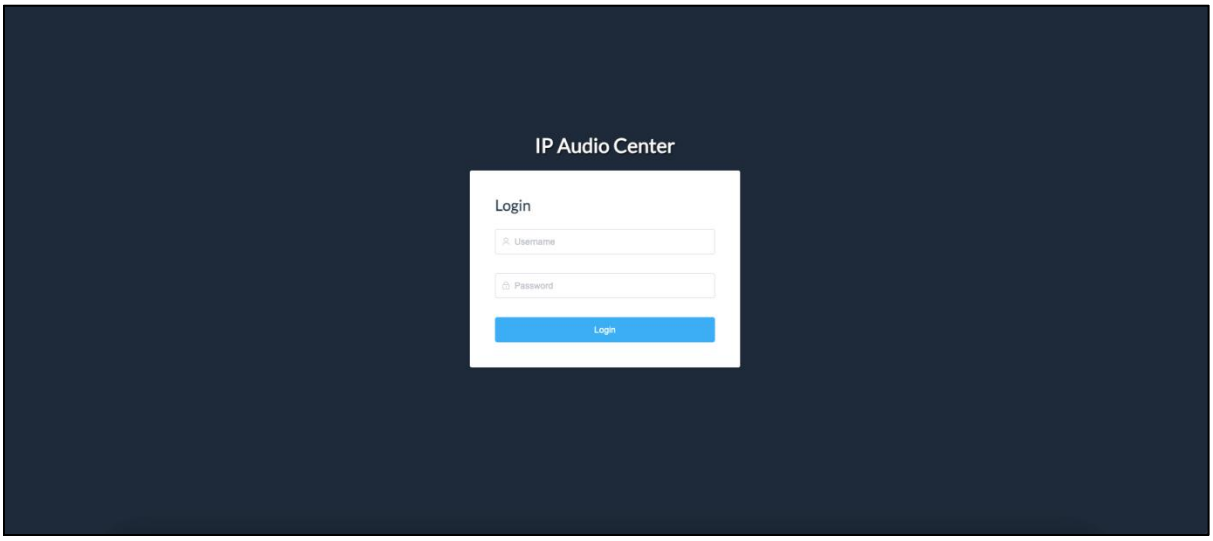
LAN port: 192.168.10.100

Please enter the default administrator login credentials to log in. After login, you'll first see the Dashboard page as shown in the figure below.

Default username: admin

Default password: admin

For the safety purpose, it is recommended to change the default password on the first login, please go to **Admin --> Change Password** on the top right corner of this page to change the password.

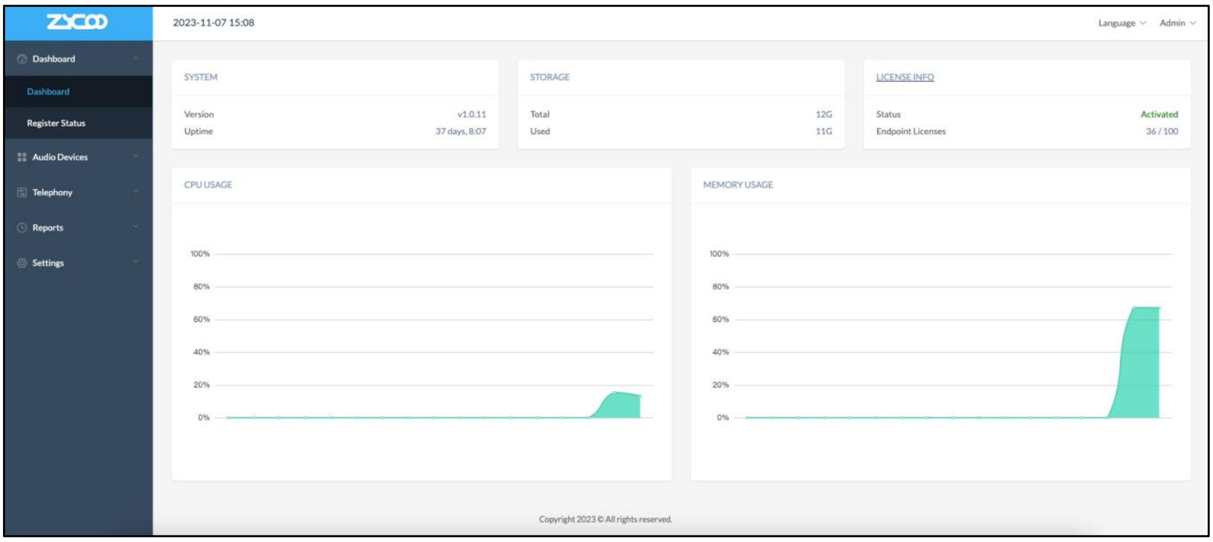


Login Interface

3.2 System Dashboard

In this page, you will see the information interface which can provide system information for convenient maintenance.

Please go to **Dashboard --> Dashboard** page to check details.



Dashboard

System

- **Version:** Display the software version of the server.
- **Uptime:** Indicates how long the server has been continuously working.

Storage

- **Total:** Display the total server storage space.
- **Used:** Display the storage space have been used.

Licence Info

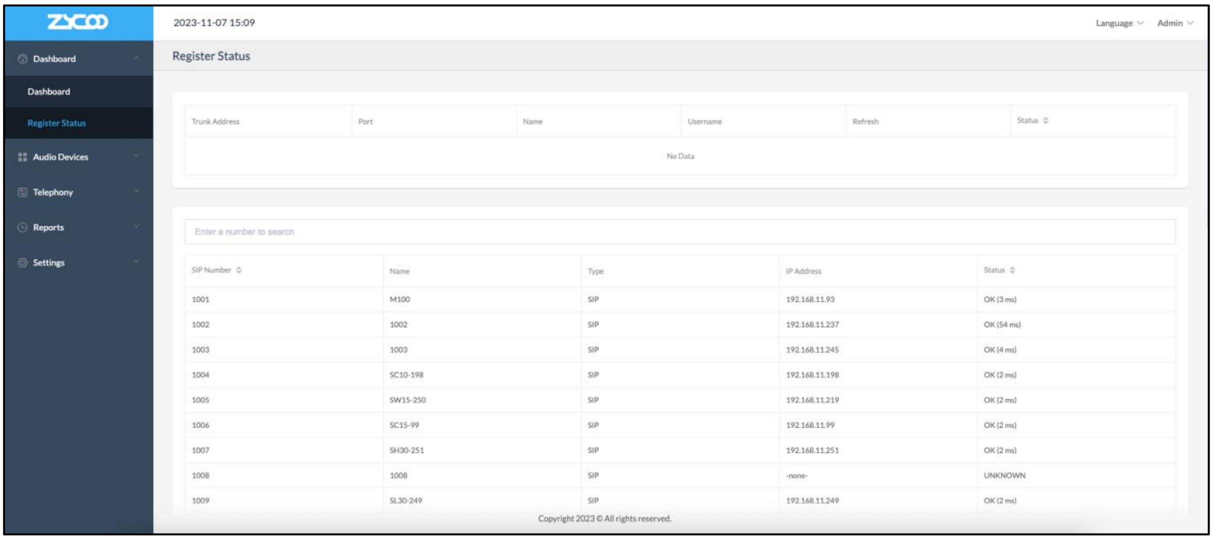
- **Status:** Display total server storage space.
- **Endpoint Licenses:** When the license is activated, it shows the number of endpoint licenses authorized.

CPU USAGE & MEMORY USAGE

- **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.

3.3 Register Status

In this page, it shows both the SIP trunk information, and all created SIP account information. Please go to **Dashboard --> Register Status** page to check details.



Register Status

4. Audio Devices

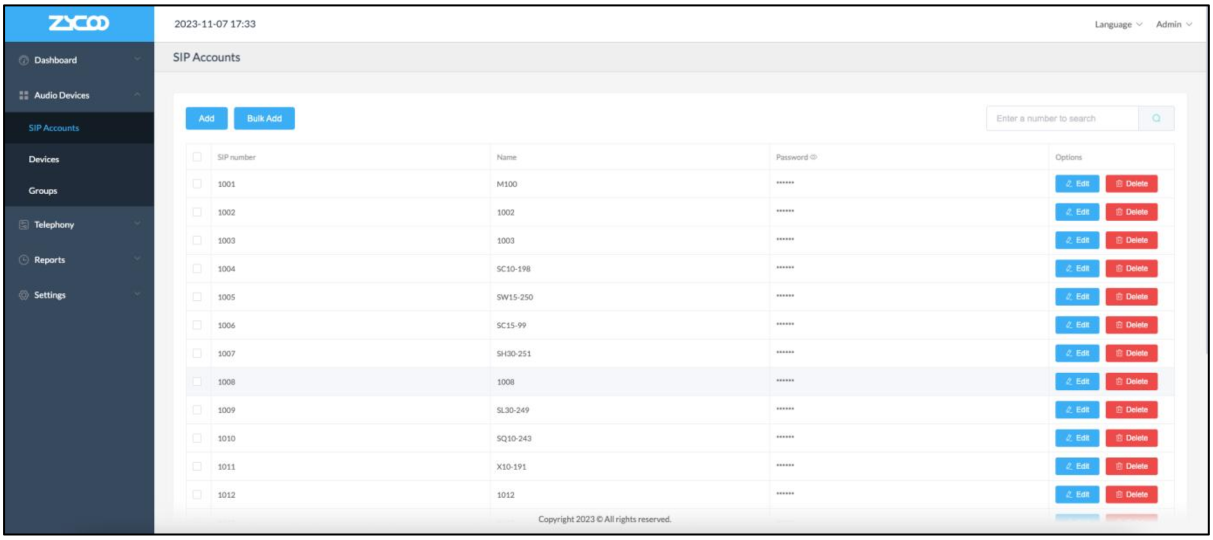
4.1 SIP Accounts Settings

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

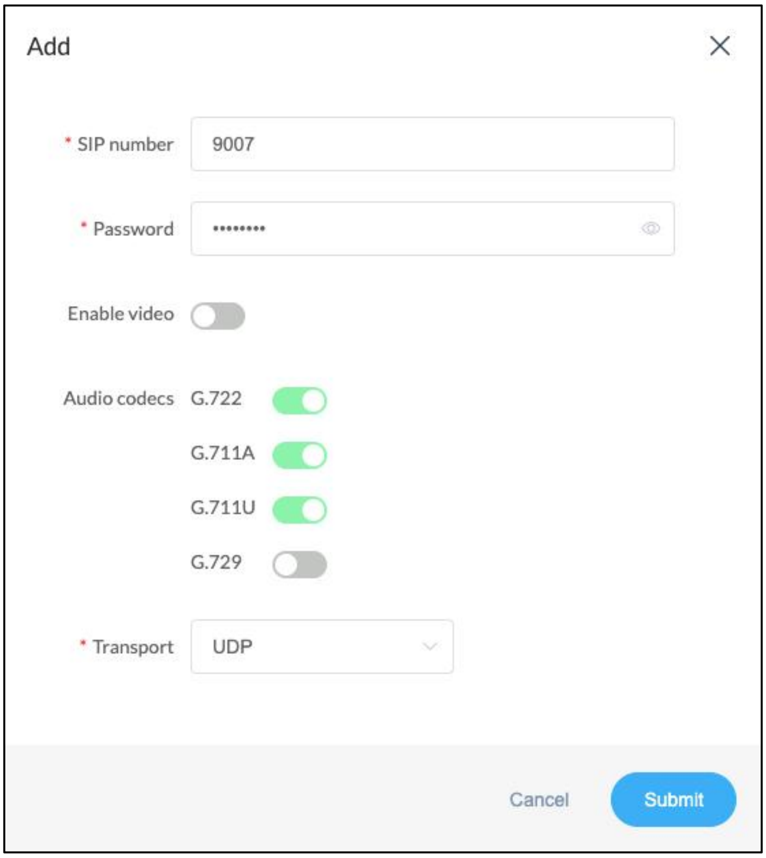
Please go to **Audio Devices --> SIP Accounts** page, you can create one SIP account at a time or you can bulk create SIP accounts.

To create a single SIP account, please click on the Add button. To Create bulk SIP numbers, please click on Bulk the Add button.

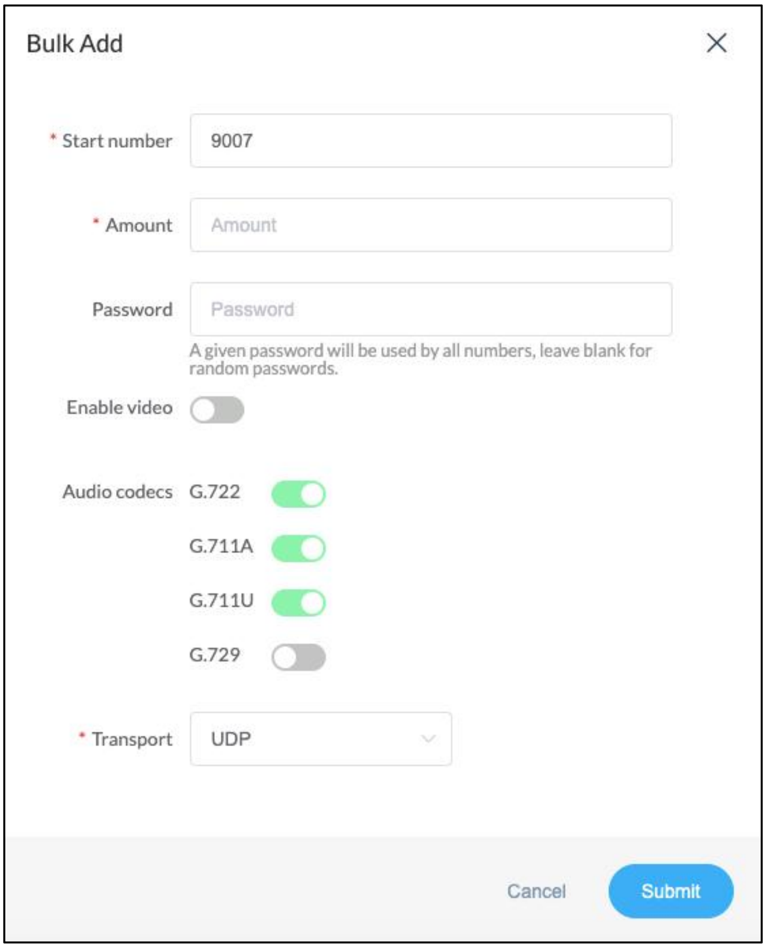
If you want to register the endpoint devices onto the IP Audio Center, you need to login to the endpoint devices' web GUI. Configure the profile on the SIP Accounts page and input server address and SIP account credentials to register the device to IP Audio Center. For detailed instructions please refer to the endpoint devices' User Guide.



SIP Accounts



Add SIP Account

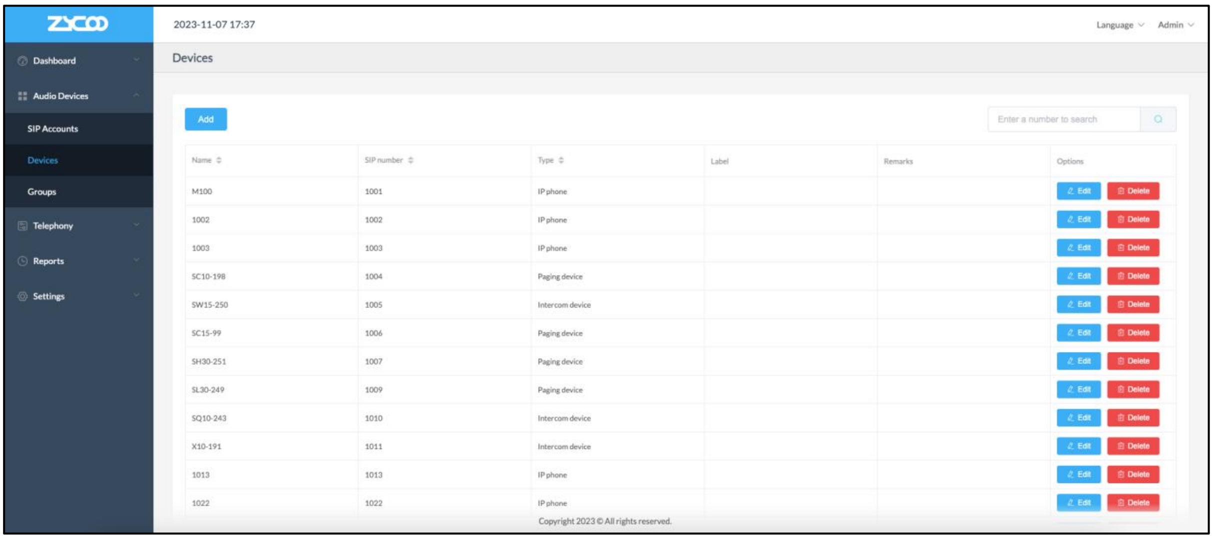


Bulk Add SIP Accounts

- **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 the 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 --> Networks --- SIP Settings**)

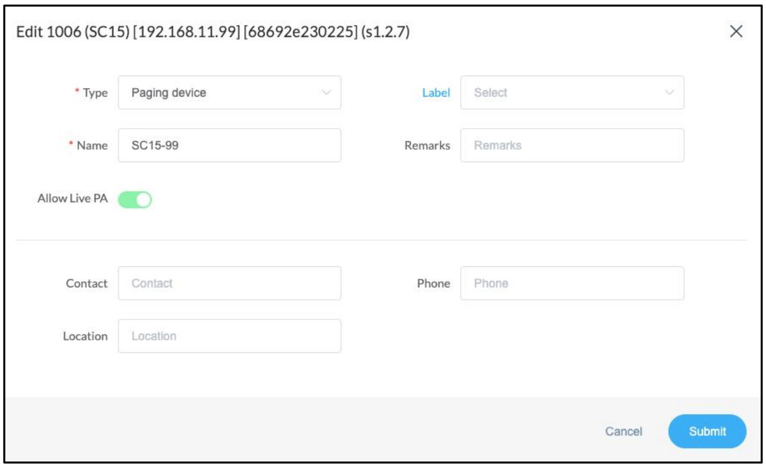
4.2 Devices Settings

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



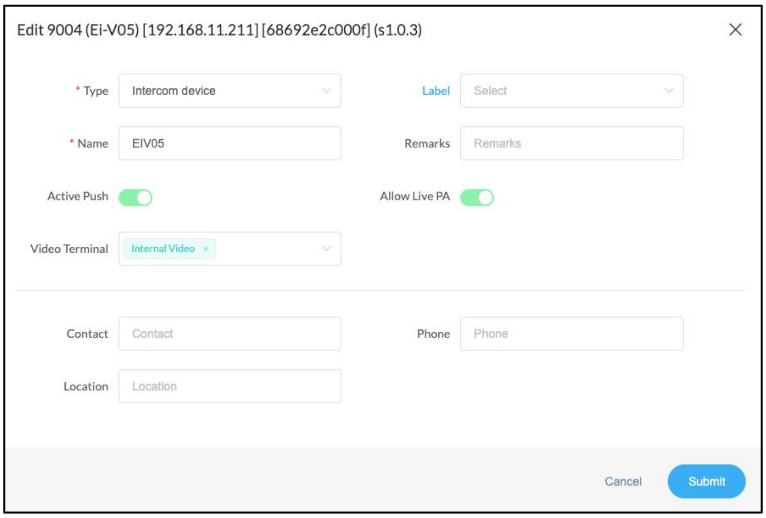
Devices Settings

Then you need to configure some advanced settings for the registered devices. Please click on the Edit button and complete the below settings.



Paging Devices 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 for maintaining this device.
- **Phone:** The contact person's phone number.
- **Location:** The location that shows this device's whereabouts.



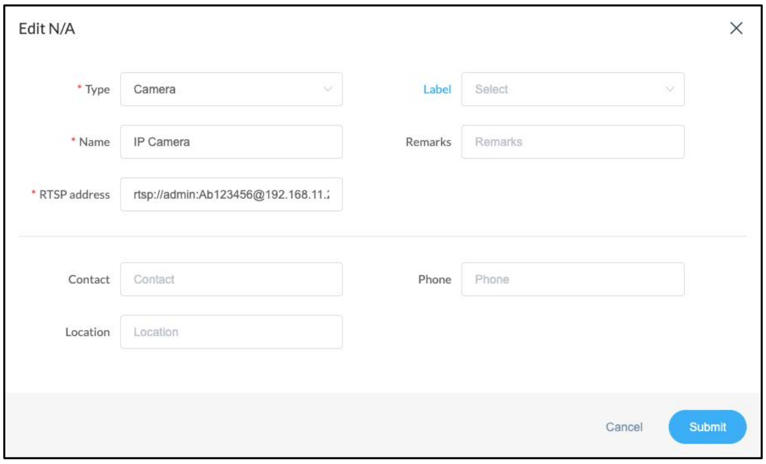
Intercom Devices Info

- **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 on the internal network (behind NAT), then the intercom device can push 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.

- **Remarks:** Additional information can be added.
- **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 for maintaining this device.
- **Phone:** The contact person's phone number.
- **Location:** The location that shows this device's whereabouts.



Camera Devices Info

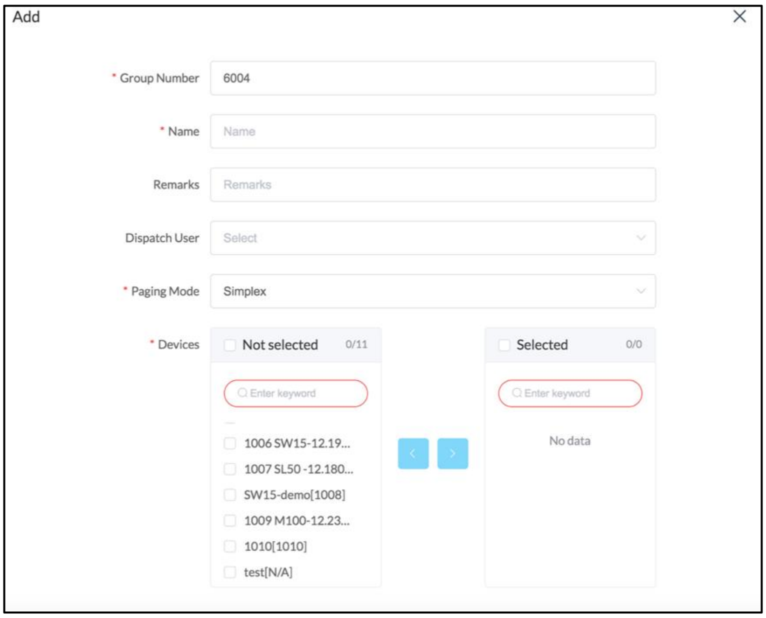
- **Type:** The device type, can be modified.
- **Label:** Label is an additional property of the IP audio devices. It will be used by the IP Audio Dispatch Console to filter IP audio devices in a group with the same label.
- **Name:** A user defined name to identify this device.
- **Remarks:** Additional 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 for maintaining this device.
- **Phone:** The contact person's phone number.
- **Location:** The location that shows this device's whereabouts.

4.3 Groups Settings

After you have registered all IP audio endpoints to the IP Audio Center, next you need to group them. A group can be the IP audio devices of a specific area or the 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** page. Click on the Add button to create a group.



Groups Settings

- **User ID:** The User ID will be displayed as the outgoing number when calling out, or the number that another device needs to dial.

- **Group Number:** The 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.

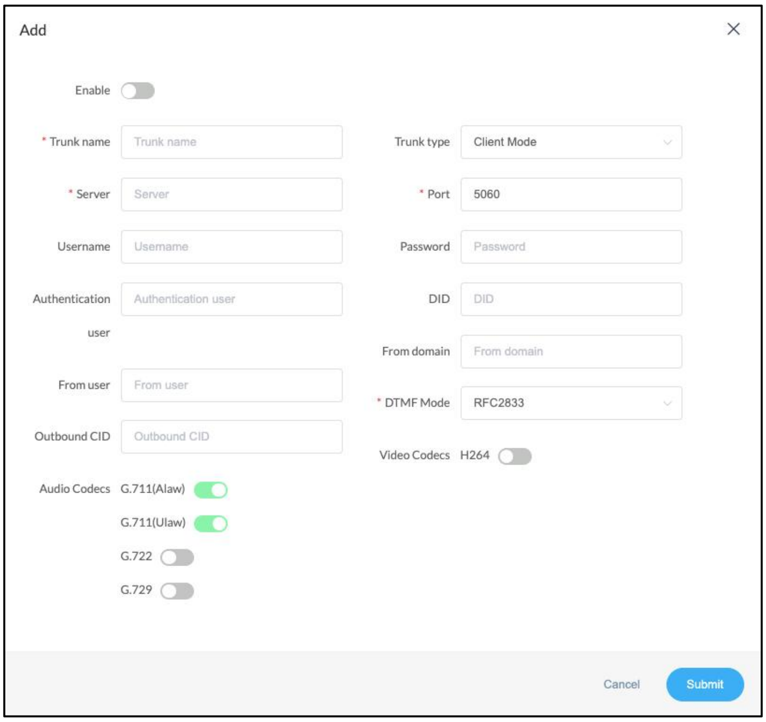
Please keep at least one codec enabled and supported by the SIP server, otherwise, SIP paging will not work.

5. Telephony

5.1 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** page. Click on the add button to add a new trunk.



SIP Trunk

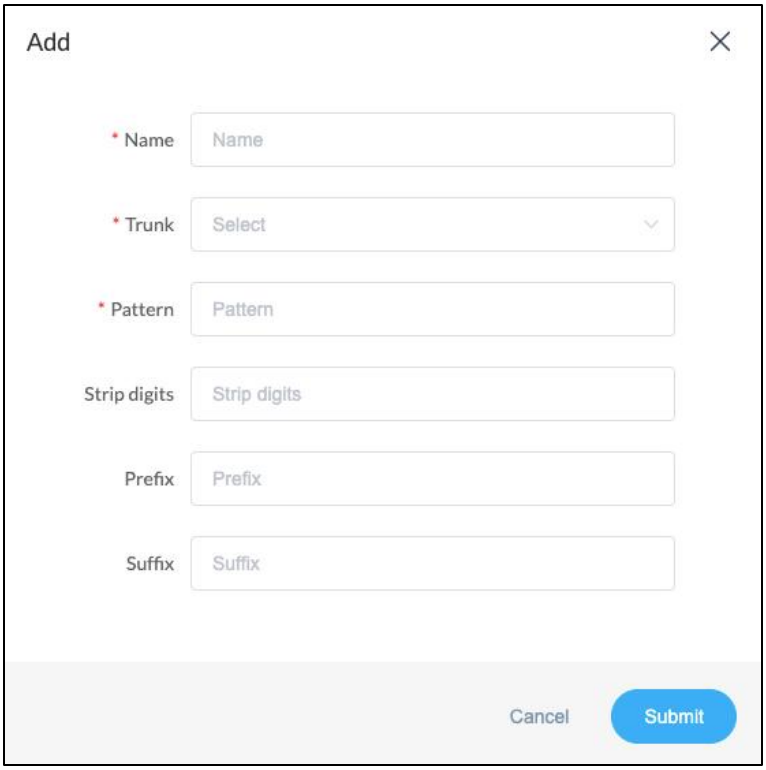
- **Enable:** Activate this SIP trunk setting.
- **Trunk Name:** The name to identify this SIP trunk.
- **Type:** The connection modes include client mode and server mode.
- **Server:** The ITSP server address or third-party SIP server address.
- **Port:** The SIP port number of the remote side.
- **Username:** 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 purposes.
- **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:** Select the audio codecs G.711(Alaw) G.711(Ulaw) G.722 G.729.
- **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**.

5.2 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 page, click on the Add button to create an outbound rule.



Outbound Rules

- **Enable:** Activate this SIP trunk setting.
- **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

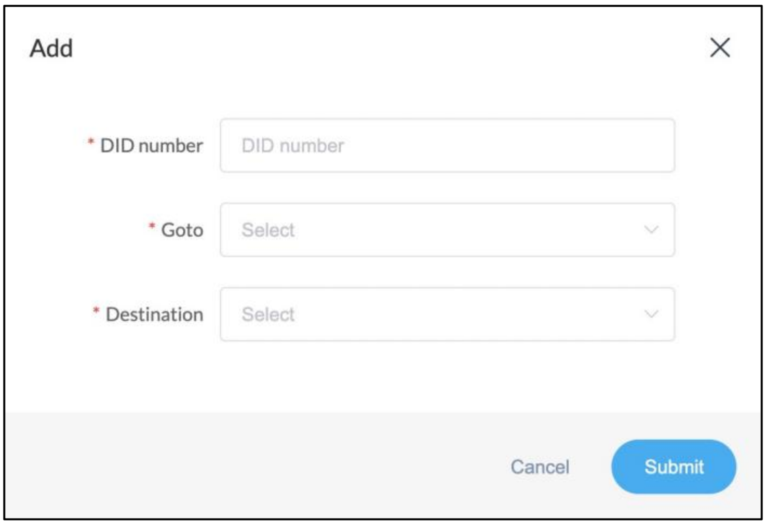
Z — Any digit that is not zero. (E.g.1to9)

. — 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.

5.3 Inbound Rules

The inbound phone calls from the SIP/IMS trunks to the IP Audio Center can be routed to a specific IP phone or a digital receptionist. It is controlled by the inbound rules.
Go to **Telephony --> Inbound Rules** page, click the Add button to add an inbound rule.

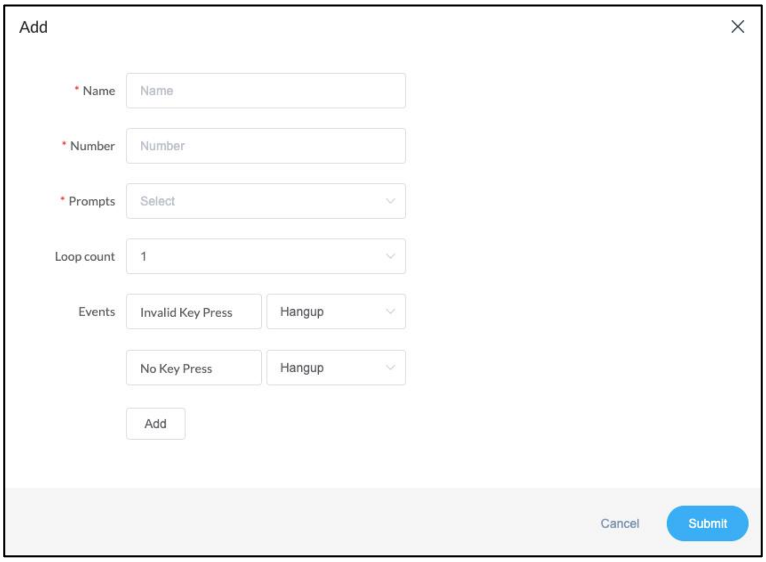


Inbound Rules

- **DID number:** Use the DID number corresponding to the trunk to configure the incoming call rules.
- **Goto:** You can choose to send the inbound calls to IP phone, Paging Group, digital receptionist by selecting or triggering 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.

5.4 Digital Receptionist

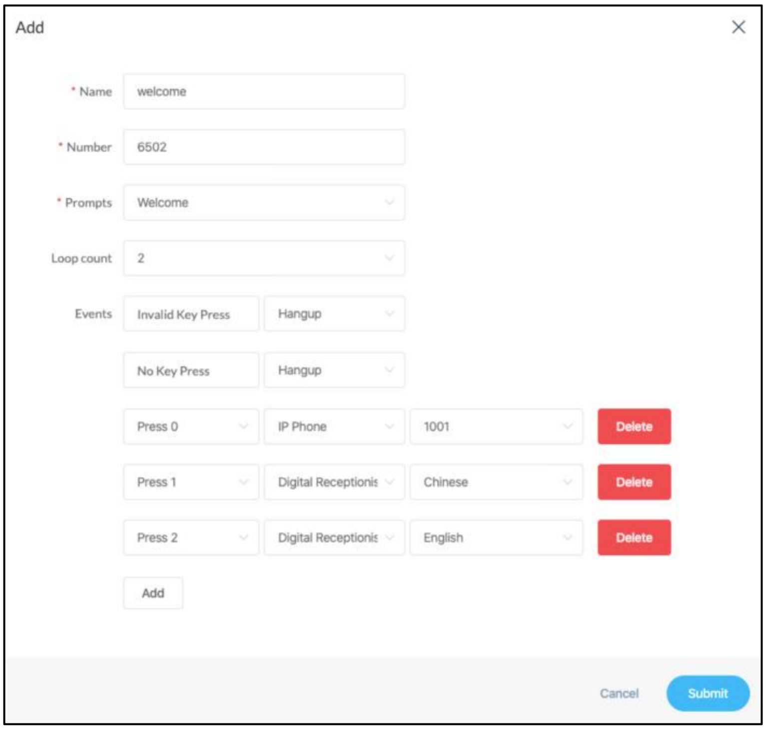
Digital receptionist is also known as IVR (Interactive Voice Response), it is one of the most important auto call distribution features in an IP phone system. ZYCOO IP Audio Center supports multi-level digital receptionists for handling the inbound phone calls.
Go to **Telephony --> Digital Receptionist** page, click the Add button to create a digital receptionist.

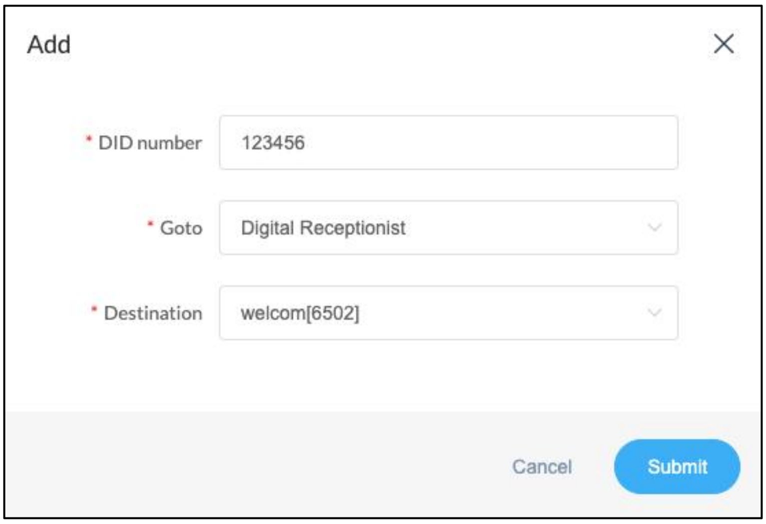


Add Rule

- **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 page.
- **Loop count:** Select the number of times to playback this IVR prompt before callers press 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 is not specified and it will be handled by the "Invalid Key Press" option. 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 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 callers to press different keys to enter the next level of their desired language of digital receptionists.





Rule Example

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

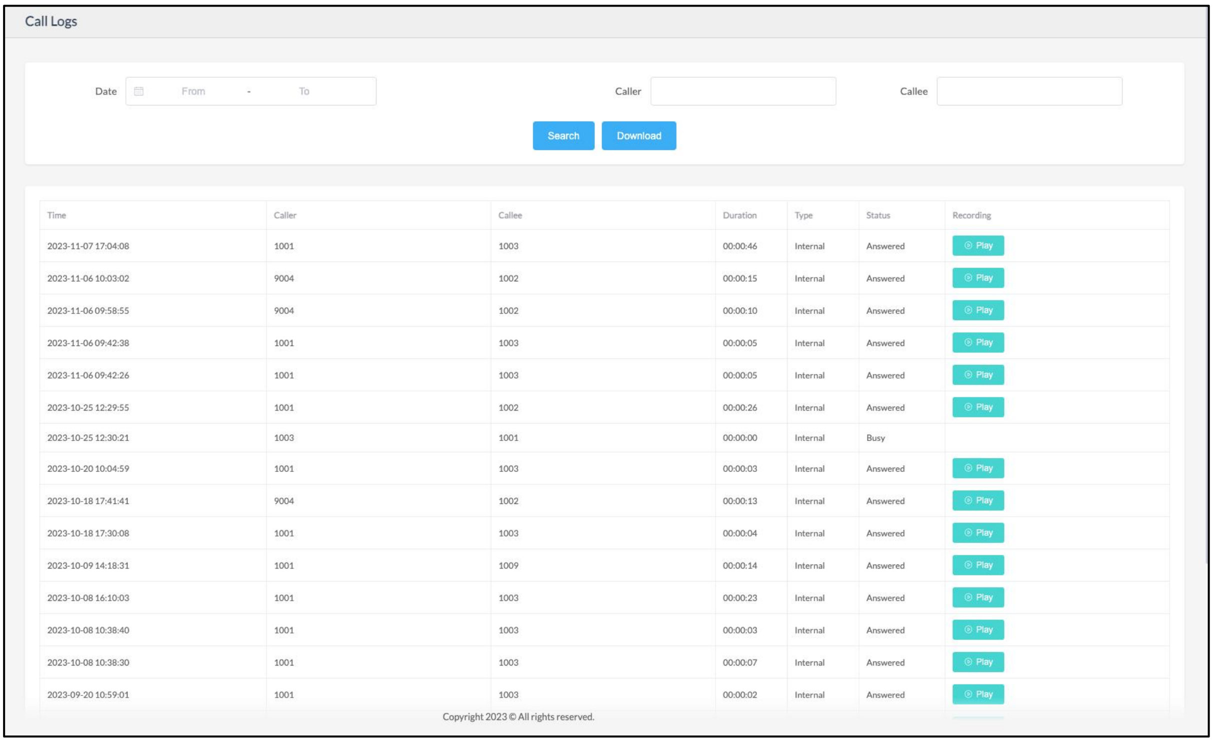
To realize the above feature, you need to go to the **Telephony -> Inbound Rules** page, the inbound rules should select the destination type as Digital Receptionist, and set the destination as the "welcome" digital receptionist. As a result, all inbound calls from the trunk with DID number 123456 will all land on the welcome digital receptionist.

6. Reports

6.1 Call Logs

IP Audio Center admin users can have a list of the full call logs of the system to help them check. And you can filter the call logs by date, caller and callee.

Please go to **Reports --> Call Logs** page to check details.



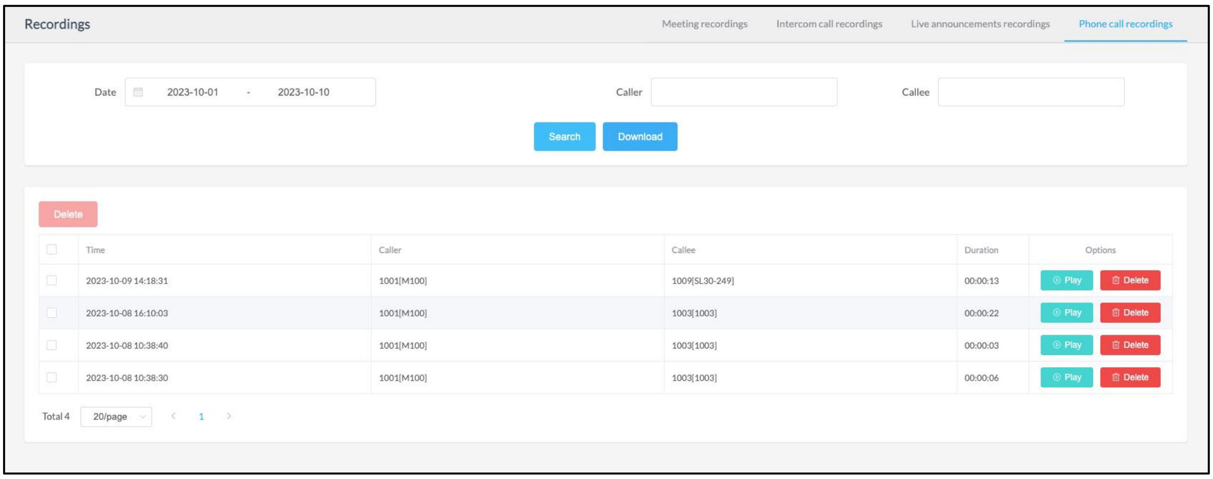
Time	Caller	Callee	Duration	Type	Status	Recording
2023-11-07 17:04:08	1001	1003	00:00:46	Internal	Answered	Play
2023-11-06 10:03:02	9004	1002	00:00:15	Internal	Answered	Play
2023-11-06 09:58:55	9004	1002	00:00:10	Internal	Answered	Play
2023-11-06 09:42:38	1001	1003	00:00:05	Internal	Answered	Play
2023-11-06 09:42:26	1001	1003	00:00:05	Internal	Answered	Play
2023-10-25 12:29:55	1001	1002	00:00:26	Internal	Answered	Play
2023-10-25 12:30:21	1003	1001	00:00:00	Internal	Busy	
2023-10-20 10:04:59	1001	1003	00:00:03	Internal	Answered	Play
2023-10-18 17:41:41	9004	1002	00:00:13	Internal	Answered	Play
2023-10-18 17:30:08	1001	1003	00:00:04	Internal	Answered	Play
2023-10-09 14:18:31	1001	1009	00:00:14	Internal	Answered	Play
2023-10-08 16:10:03	1001	1003	00:00:23	Internal	Answered	Play
2023-10-08 10:38:40	1001	1003	00:00:03	Internal	Answered	Play
2023-10-08 10:38:30	1001	1003	00:00:07	Internal	Answered	Play
2023-09-20 10:59:01	1001	1003	00:00:02	Internal	Answered	Play

Call Logs

- **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 duration of the call.
- **Type:** The type of the call: internal, inbound or outbound.
- **Status:** The status of the call.
- **Recording:** If the call has been recorded, you can click the Play button to review the call recordings.

6.2 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** page.

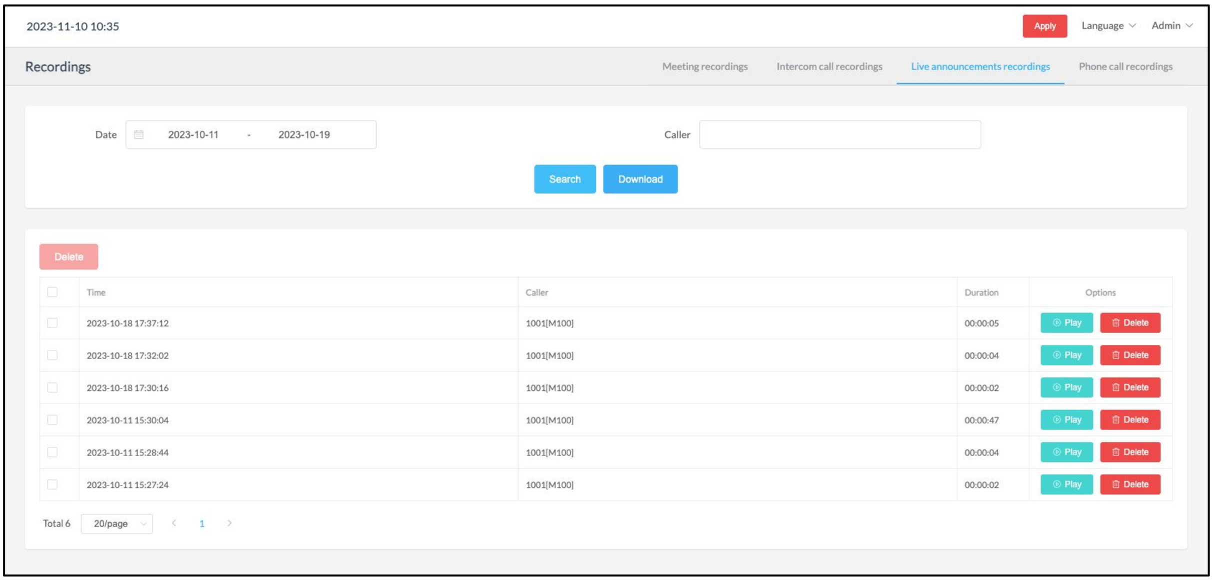


Phone Call Recordings

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. You can playback the recording online by simply clicking on the Play button. And you can also download this recording to your PC. Or simply by clicking on the Delete button to delete a specific recording, bulk delete is also supported by selecting multiple recordings.

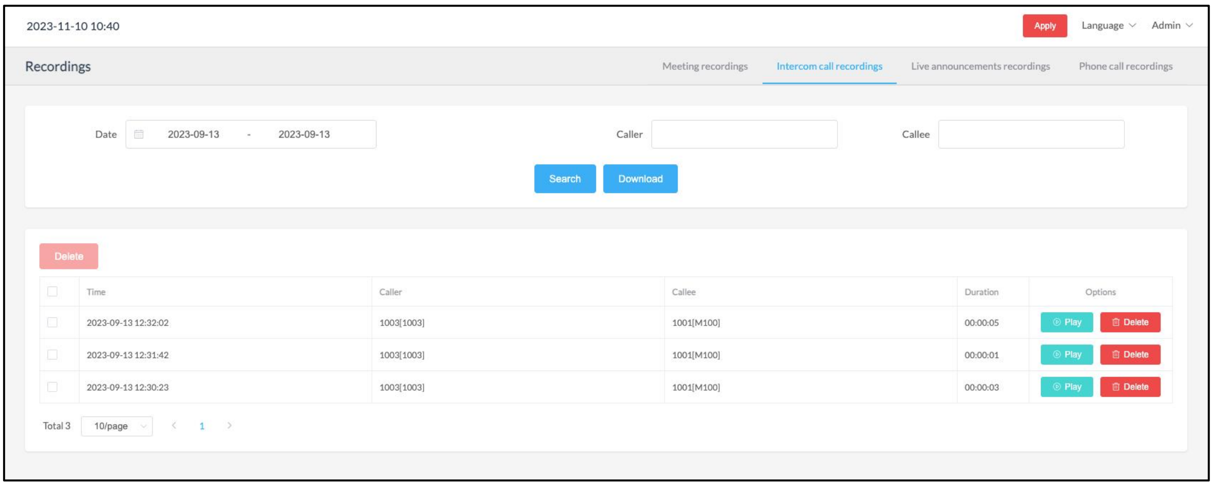
Note: IP phone call recordings can only be viewed and managed by IP Audio Center admin user, the IP Audio Dispatch users have no access to the IP phone call recordings.



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.

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

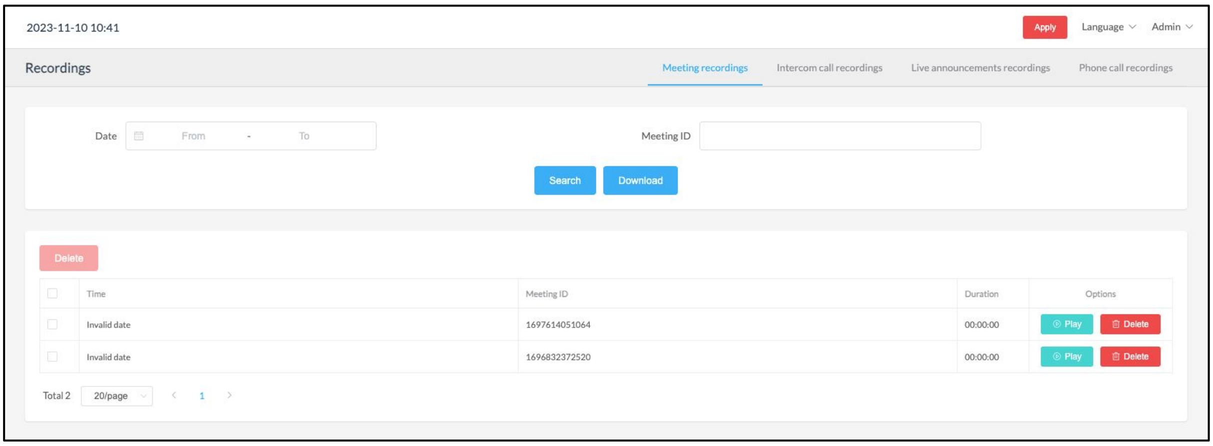


Intercom Call Recordings

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 played online and can be downloaded.

Note: 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.



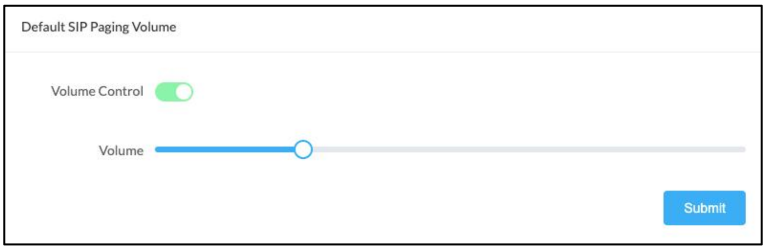
Meeting Recordings

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.

7. System Settings

7.1 System Settings

Please go to **Settings --> System Settings** page, there are some global settings including default volume, indication tone and feature codes.

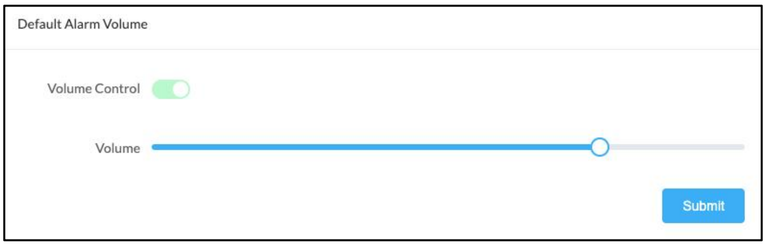


Default SIP Paging Volume

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

- **Volume 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 to setup a higher default volume level. After the alarm is stopped, the volume level will be restored.

- **Volume Control:** This option is mandatory enabled and it's used 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.

Paging Beep

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**

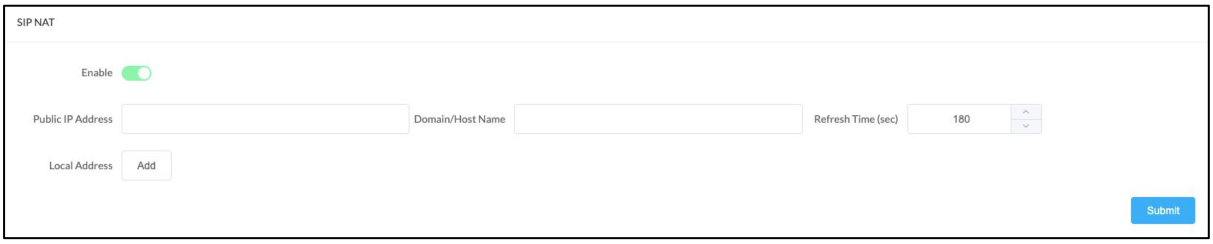
Ring Duration

Set the duration of the prompt tone. The default duration is 30 seconds.

SIP Settings

In the SIP Settings page, users 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 users need to set up remote extensions and need to specify the local network address after setting the public IP address or the domain name/host name. Please specify the local address here. For an example: 192.168.1.0/255.255.255.0

ModeBus

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

- **Enable:** Enable/Disable Modbus services.
- **Address:** Modbus server address.
- **Port:** Modbus server port number.
- **Connection Status:** Connection status with the Modbus server.

Feature Codes

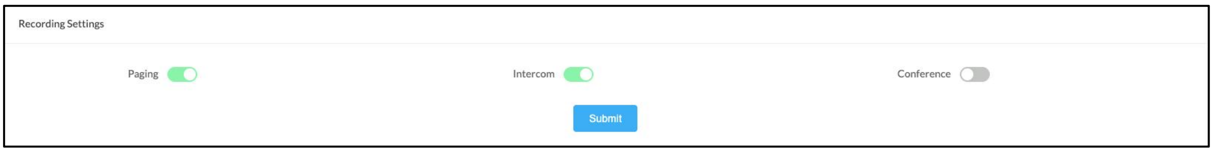
Feature codes are a series of codes that can be dialed from the IP phones or other IP audio endpoints with a keypad to achieve some specific calling or broadcasting features.

- **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 followed by the task number.
- **Do Not Disturb:** Dial this code to Set the IP phone to Do Not Disturb (DND) mode and dial the code again to disable DND.
- **Call Spy:** When an IP phone or intercom device is on a call, the dispatch user can dial the Call Spy feature code 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 on a call, the dispatch user can dial the call split feature code 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 users enter a 4-digit number of the time, for example 0930 to setup a wakeup call at 9:30 every day in the morning. To cancel the wakeup call users can simply dial the feature code and press # after the beep sound to cancel it directly.
- **Barge Spy:** When an IP phone or intercom device is on a call, the dispatch user can dial the barge spy feature code 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 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 on a call, the dispatch user can dial the whisper spy feature code 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 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 phone's ringing timeout (30 seconds). IP phone users can dial the forward on no answer feature code 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.

7.2 Recording Settings

On the recording settings page, you may configure the recording related parameters. Please go to **Settings --> Recording** to set details.



Recording Settings

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

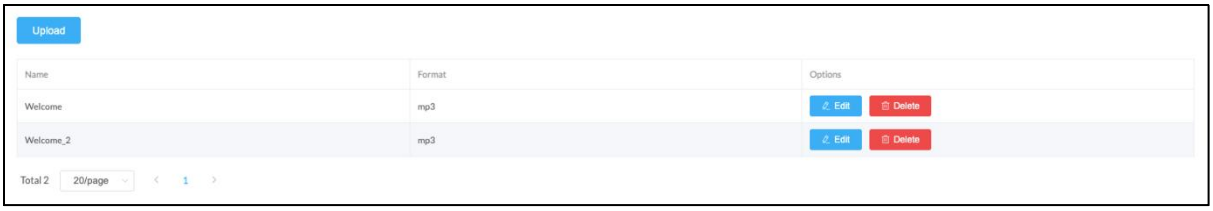
IP Phone Settings

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

7.3 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.

For resetting the current device's password, please go to **Settings --> Audio Prompts** page.



Audio Prompts Settings

Note:

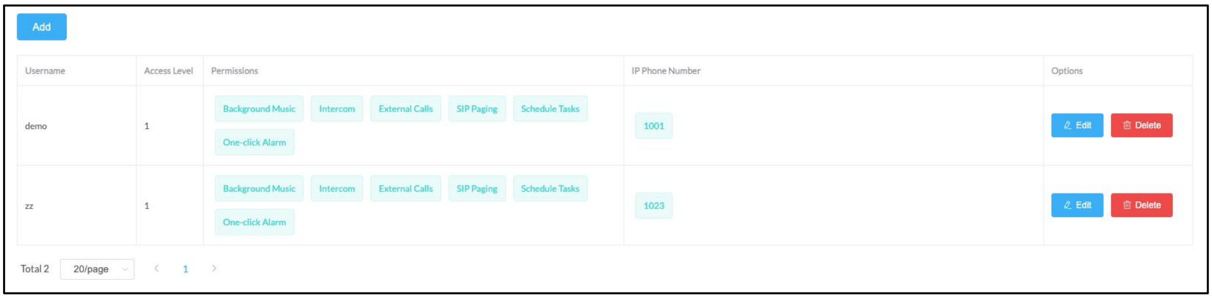
1) The system supports custom files in mp3, wav, gsm file format and less than 15MB.

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

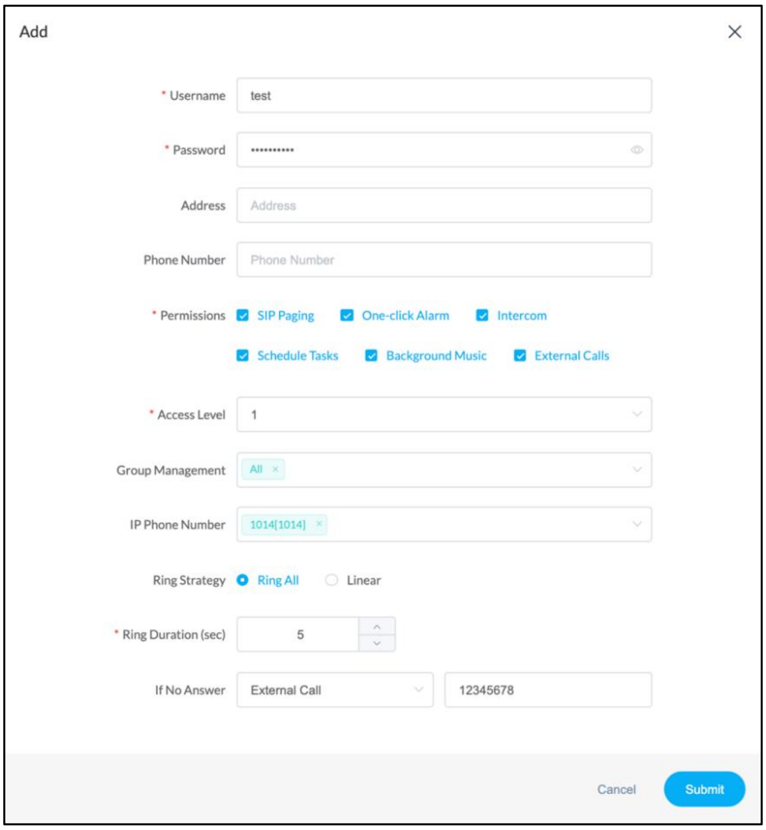
7.4 Dispatch Users

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

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 on how to login and use the IP Audio Dispatch Console, please refer to the ZYCOO IP Audio Dispatch Console User Guide.



Dispatch Console Settings



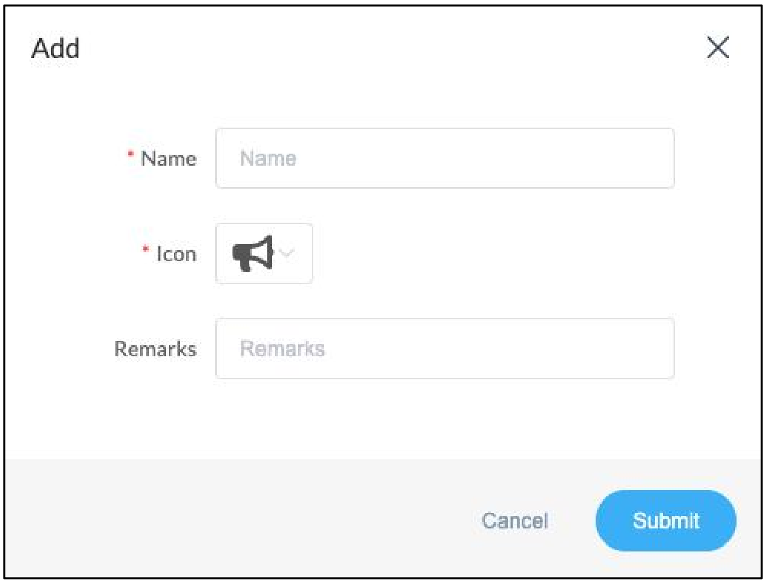
Add Dispatch Console Users

- **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).

7.5 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.
Please go to the **Settings --> Labels** page to manage all the labels. To create a label please click on the Add button.



Lable Settings

- **Last Update:** Displays the last system updating time.
- **Name:** A name to identify the label, and it's better to 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** interface, assign a label to each of the IP audio endpoints. Then you can filter the endpoints with the label you have created on the IP Audio Dispatch Console.

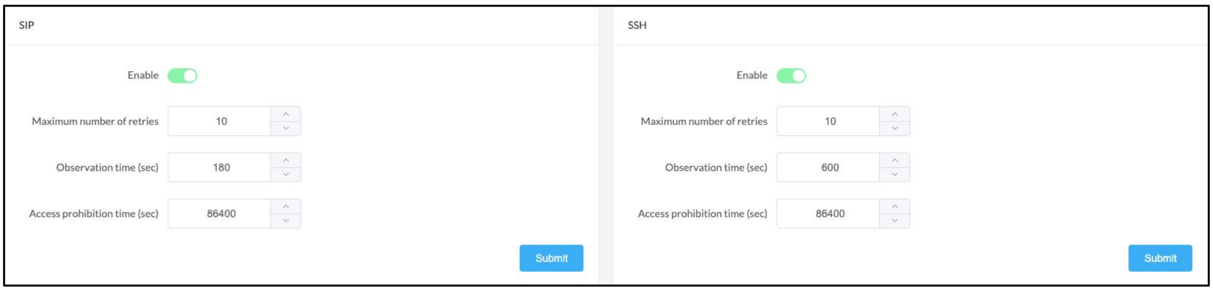
Lable Interface

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

7.6 Security

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 a specified number of times, the system will automatically add that IP address to the Blacklist and block that IP address.

Please go to **System --> Security** page to change security configuration.



Intrusion Prevention

SIP Secure Settings

- **Icon:** Select an icon for this label.
- **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 period.
- **Observation Time:** Specify the observation time in seconds.
- **Access Prohibition Time:** Specify the time of the banned IP address to stay the blacklist.

SSH Secure Settings

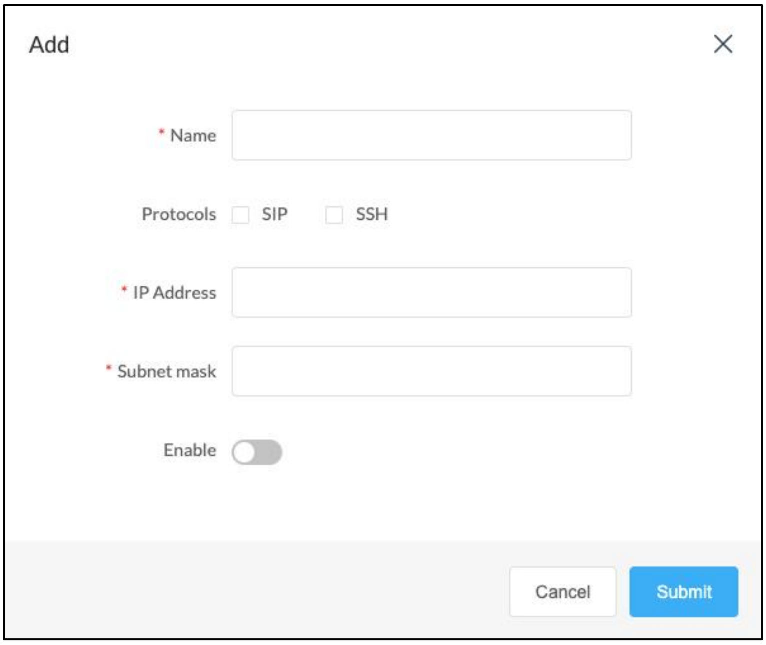
- **Icon:** Select an icon for this label.
- **Enable:** Enable/Disable the SSH intrusion prevention service.
- **Max number of retries:** The maximum number of retries allowed for the same IP address to access during one observation time period.
- **Observation time:** Specify the observation time in seconds.
- **Access prohibition time:** Specify the time of the banned IP address to stay in the blacklist.

IP Blacklist

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

IP Whitelist

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



Add IP Whitelist

- **Icon:** Select an icon for this label.
- **Name:** The name of the 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.

7.7 License

After the using period, you need to upload the license key for activating the IP Audio Center. Please fill in the required 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 the distributors or sales. Please click on the Upload button to upload the license key file provided by the distributor or sales.

Please go to **System --> Licence** page to upload the license file.


License Info
Free Trail Expiration 2022-02-10 14:01:25
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