

ZYCOO IP Audio Solution Whitepaper



www.zycoo.com

zycoo@zycoo.com

© 2023 Zycoo Communications LLC All rights reserved

Contents

1.	Preface	3
	1.1 Audience	3
	1.2 Revision History	3
2.	Overview	4
	2.1 Product Overview	4
	2.2 Software Product Overview	5
	2.2.1 IP Audio Center	5
	2.2.2 IP Audio Dispatch Console	5
	2.2.3 IP Audio Dispatch App	5
	2.3 Hardware Product Overview	6
	2.3.1 Hardware SIP Server	6
	2.3.2 Network Speaker	6
	2.3.2.1 SC10 Network Ceiling Speaker	7
	2.3.2.2 SC15 Network Ceiling Speaker	8
	2.3.2.3 SW15 Network Cabinet Speaker	8
	2.3.2.4 SH30 Network Horn Speaker	9
	2.3.2.5 SL30 Network Column Speaker	10
	2.3.2.6 SQ10 Network Square Speaker	11
	2.3.3 Network Safety Intercom	11
	2.3.4 Network Paging Gateway	12
	2.3.5 Network Microphone Console	13
	2.4 Feature Overview	14
3.	Advantages of ZYCOO IP Audio Solution	21
	3.1 Compare with Analog Public Address Systems	21
	3.2 Compare with other IP Public Address Systems	24
4.	Implementation Suggestions	26
	4.1 Server Deployment	26
	4.2 Network Requirement	26
	4.2.1 Protocol Classification	26
	4.2.2 Bandwidth Requirement Calculation	27
_	Conclusion	20

1. Preface

1.1 Audience

This manual is intended to provide clear operating instructions for those who will configure and use the IP Audio Dispatch Console. By carefully reading and consulting this guide, users could solve the setting and usage issues of the IP Audio Dispatch Console.

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

ZYCOO IP Audio Solution is an IP audio communication system based on SIP, streaming media and MQTT IoT technologies. Our solution provides functionalities such as broadcasting, intercom, telephony, paging, and many more. The entire solution architecture consists of IP audio endpoints, IP audio dispatch console and IP audio dispatch app, and an IP audio server. IP audio endpoints are hardware based, which include network speakers, IP intercoms, IP phones, and IP paging gateways, while IP audio dispatch console, which is installed on a computer, and IP audio dispatch app, which supports both Android and IOS, are software based. The IP audio server can be either a computer running our IP audio center software, or our IP audio server IAS-L100, and can be deployed on premise, or in the cloud. The solution architecture is designed to work within the standard intranet and or internet without the need for special equipment or wiring, as shown below.



2.2 Software Product Overview

2.2.1 IP Audio Center

IP Audio Center is the engine of ZYCOO IP Audio Solution, it is a software solution that 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.

2.2.2 IP Audio Dispatch Console

IP Audio Dispatch Console is a multi-platform (Windows, MacOS and Linux) application dedicated for ZYCOO IP Audio Solution dispatch users. With the IP Audio Dispatch Console, dispatch users can broadcast background music, make live SIP paging, create scheduled paging tasks, manage audio conference calls, initiate emergency broadcast, monitor IP audio endpoints status and more. It is suitable for public safety, smart city, secure community, industry, transportation, health and more application scenarios.

2.2.3 IP Audio Dispatch App

IP Audio Dispatch App is a simple and easy to use mobile application developed for ZYCOO IP Audio dispatch users. The app allows users of iPhone or Android based devices a mobile alternative to the PC based dispatch console.

With the IP Audio Dispatch App, background music, pre-recorded message paging, emergency paging and IP audio endpoint status monitoring can be configured and maintained. IP Audio Dispatch App is a useful application which is perfect for when the dispatch users have to leave their dispatch PC or full dispatch functionality is not critical.

2.3 Hardware Product Overview

2.3.1 Hardware SIP Server

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.

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	271000 P Audio Contra
Number of Number Triggered Broadcast Tasks	Unlimited	
Number of SIP Concurrent Calls	100 (Max)	IP Audio Center
Number of Simultaneous Conference Attendees	100 (Max)	
Number of SIP Paging Network Speakers	100 (Max)	

2.3.2 Network Speaker

ZYCOO SC10&SC15 Network Ceiling Speaker and SW15 Network Cabinet Speaker are high efficient, full-range drive units SIP-enabled speakers. Specially designed for indoor purposes, to provide premium sound quality, enhance the audience experience. SH30

Network Horn Speaker is equipped with a midrange drive unit powered by a 30W class-D amplifier, extending audio communication in large areas and high noise outdoor environments, where voice coverage is of primary concern. SL30 Network Column Speaker is PoE+ supported, it could cover a wide frequency band to provide high sensitivity, beautiful sound, and a unique listening experience for listeners. As well as the IP65-Enclosure makes the SL30 perfect to work in any outdoor environment.

2.3.2.1 SC10 Network Ceiling Speaker

SC10 Network Ceiling Speaker is a compact high-performance SIP enabled ceiling speaker which can be used for SIP paging, notification/tone broadcasting and streamed high definition music playback. The high efficient, Coaxial speaker driver units will provide you with a uniquely advanced listening experience.

SC10 Network Ceiling Speaker Specifications		
Speaker Components	4" woofer unit + 1" tweeter unit	
Sensitivity	90dB / 1W / 1m	
Max Sound Pressure Level	100dB	
Rated Power	8Ω 10W	
Frequency Range	70Hz – 20KHz	
Coverage Pattern	90°H 90°V 30m²	
Amplifier	Built-in Class D Amplifier	

2.3.2.2 SC15 Network Ceiling Speaker

SC15 Network Ceiling Speaker is a compact high-performance SIP enabled ceiling speaker which can be used for SIP paging, notification/tone broadcasting and streamed high definition music playback. The high efficient, full-range drive units will provide you with a uniquely advanced listening experience.

SC15 Network Ceili	ng Speaker Specifications	
Speaker Components	5.25" woofer unit + 1" tweeter unit	
Sensitivity	85dB / 1W / 1m	
Max Sound Pressure Level	100dB	
Rated Power	8Ω 15W	
Frequency Range	70Hz – 20KHz	
Coverage Pattern	90°H 90°V 30m²	
Amplifier	Built-in Class D Amplifier	

2.3.2.3 SW15 Network Cabinet Speaker

SW15 Network Cabinet Speaker is a high-performance SIP enabled cabinet speaker which can be used for SIP paging, notification/tone broadcasting and streamed high definition music playback. The high efficient, full-range drive units will provide you with a uniquely advanced listening experience. And it's designed for the applications where voice coverage is of primary concern.

SW15 Network Ca	binet Speaker Specifications
Speaker Components	5.25" woofer unit + 1" tweeter unit
Sensitivity	85dB / 1W / 1m
Max Sound Pressure Level	100dB
Rated Power	8Ω 15W
Frequency Range	70Hz – 20KHz
Coverage Pattern	90°H 50°V 30m²
Amplifier	Built-in Class D Amplifier

2.3.2.4 SH30 Network Horn Speaker

SH30 Network Horn Speaker is a SIP enabled horn speaker which provides exceptionally clear and intelligible voice for SIP paging, notification/tone broadcasting and streamed background music. It is suitable to be deployed in outdoor and other open spaces for public notifications and public safety purposes.

SH30 Network Horn	n Speaker Specifications	
Speaker Components	2"midrange driver unit	
Sensitivity	105dB / 1W / 1m	
Max Sound Pressure Level	117dB	

Rated Power	DC: 8Ω 30W / PoE: 8Ω 18W	
Frequency Range	400Hz – 8KHz	
Coverage Pattern	50° H 50° V. effective distance 70m	
Amplifier	Built-in Class D Amplifier	

2.3.2.5 SL30 Network Column Speaker

SL30 Network Column Speaker is a high-performance SIP-enabled column speaker for SIP paging, notification broadcasting, and streamed high-definition music playback. SL30 has high sensitivity and capacity, providing clear and rich saturation, whether SIP-based propaganda broadcasting or background music playing. SL30 also supports dry contact signal output and switch signal input for various sensors integration. Meanwhile, the support of PoE+ makes installation much easier and more effort-less.

SL30 Network Column Speaker Specifications		
Speaker Components	Two 3.25" woofer units + one 1" tweeter unit	
Sensitivity	82dB / 1W / 1m	
Max Sound Pressure Level	97dB	
Rated Power	4Ω 30W	
Frequency Range	100Hz – 20KHz	
Coverage pattern	135°, Optimal distance 50m	200

Amplifier	Built-in Class D Amplifier	
-----------	----------------------------	--

2.3.2.6 SQ10 Network Square Speaker

SQ10 Network Square Speaker has SQ10-B and SQ10-T two models. Two LED flashers are located on the SQ10-T's top surface and can be used to alert room occupants of an incoming audio message or emergency notices. The LED display can show real-time time and eliminates the need for a separate clock system. Meanwhile, both of them have built-in microphones which can support two-way communication. They are perfect to use in scenarios like classrooms, libraries, and offices.

SQ10 Network Square Speaker Specifications		
Speaker Size	4.5 inch	
Sensitivity	91dB / 1W / 1m	* + +
Max Sound Pressure Level	101dB	
Rated Power	8Ω 10W	
Frequency Range	70Hz~20KHz	* O + + O *
Coverage pattern	90°H 50°V 30 m²	, 8888
Amplifier	Built-in Class D Amplifier	

2.3.3 Network Safety Intercom

Ei Series Safety Intercom is available in four models designed to address the needs and requirements of various applications and installation types.

All models deliver full-duplex communication and secure access control for your VoIP system. Ideal for any setting such as commercial facilities, schools and universities, medical facilities, warehouses, and retail establishments.

Ei Series Safety Intercom Specifications		
Drive Unit	Φ40mm full frequency drive unit	
Sensitivity	96 ± 3dB/1M/1W	
Distortion	<1%	
Rated Power	8Ω 5W	
Frequency Range	100Hz – 20KHz	
Coverage Pattern	90°H 90°V 30m²	
Amplifier	5W Single Channel D-class	

2.3.4 Network Paging Gateway

X10 SIP Paging Gateway is SIP enabled multifunctional IP audio device dedicated for industry users. It can convert voice streams from a SIP paging system or IP PBX system to analogue sounds for background music, public address, intercom, etc.Based on the compact hardware design, open standard SIP protocol support, rich functionality and high performance.

X10 SIP Paging Gateway Specifications

Amplifier Output	2x10W, 8Ω (4 pins)	
Headset Output	3.5mm Jack	
Microphone	3.5mm Jack	
Power Input	DC 12V-3A Jack	AGAGGGGGGGGGGGGGGGGGGGGGGGGGGGGGGGGGGG
Call button	Support 2 call buttons (switch button)	
Call button LED	Support 2 call button LED indicators	
Dry Contact	NO/NC contacts, max AC 125V-1A, DC 60V-1A	

2.3.5 Network Microphone Console

The M100 Dispatch Microphone Console is a versatile and high-performance SIP-enabled device for seamless communication. It features 30 programmable fast keys that can be used for paging, intercom, music playback, outbound phone calls, and emergency alarm activation. With multicast and peer-to-peer technology, it serves as a standalone serverless console for individual and group paging, internal calls, and more. The console also allows users to save music and prerecorded messages in the local storage TF card, ensuring reliable and efficient communication capabilities. The M100 Dispatch Microphone Console is ideal for a wide range of industries, including emergency services, schools, transportation, and hospitality, thanks to its advanced features and reliable performance.

M100 Dispatch Microphone Console Specifications		
Speaker Components	φ45mm full frequency	
Sensitivity	95±3dB / 1W / 1m	
Max Sound Pressure	100dB	

Level		
Rated Power:	8Ω 3W	
Microphone Sensitivity	36±2dB	
Microphone Max Sound Pressure Level	110dB	
Microphone Impedance	680Ω	
LCD Display Size	4.3 inches	
LCD Display Resolution	480p*272p	

2.4 Feature Overview

1. Paging Groups (Zones)

Network speakers can be grouped (zoned) based on their location or usage. For example, zone-based live announcements, scheduled broadcast, emergency broadcast plus further zone-based features are all possible. In addition to zone-based public address features, grouping can be also used for distributed operation and management. Allocate different groups to different dispatch users of the IP Audio Center system, each paging zones (or several zones) can work and be managed separately. This is useful for enterprises with multibranch offices to build a unified IP audio system which is maintained as one while operated as many

2. Background Music

IP Audio Center supports up to 44.1KHz sampling rate, 320Kbps bit rate MP3 music files. The music files are streamed to the network speakers using streaming media and the network speakers then decode and play out using on board decoder and amplifier. This ensures superior background music quality. Background music is played by the dispatch users from the IP Audio Dispatch Console. Random speakers or speaker group(s) can be selected to play a customized playlist.

3. Triggered Paging

Triggered paging are preconfigured paging tasks that will be automatically executed when the trigger has been invoked. The trigger types include immediate triggered, timetable triggered and dial number triggered. The timetable triggered paging will be executed according to the time rules configured by the dispatch users. It can be configured as a one-off paging task on a specific time point or routine paging tasks on specific time points of specific days. So, it can be used to schedule school bells or some other public address applications at pre-configured time intervals. Dial number triggered paging can be used with paging applications where time is uncertain but requires fast execution under particular situations. Dispatch users can simply dial a code from the dispatch phone to start/stop the dial number triggered paging. Immediate triggered paging is usually used to create interim paging tasks, dispatch users can take advantage of a pre-recorded message, text-to-speech as the audio source for immediate triggered paging, music and alarm sounds can be used as well.

4. SIP Paging

SIP paging is used to make live announcements from a SIP IP phone or SIP microphone to one or a group of network speakers. Dispatch users can either click to make live announcements by selecting the network speakers from IP Audio Dispatch Console or they can dial the network speaker number or group number from their IP phone or SIP microphone key pad to make live announcements when required.

5. Emergency Alarm

There are certain pre-configured emergency events in the IP Audio Center system, such as fire, earthquake, tornado, etc. Dispatch users can click to sound an alarm from the IP Audio Dispatch Console by using the built-in or customized emergency events as per their needs. Emergency alarm can be achieved by dial number triggered paging, so the dispatch users can also sound alarm by dialing a code from the IP phone or SIP microphone.

6. Prerecorded Message

Prerecorded messages can be used as the audio source of the triggered paging tasks to be created from the IP Audio Dispatch Console. Dispatch users can record audio messages by using the microphone attached to the dispatch PC, and they can review and re-record the

audio message to ensure the audio message is qualified for paging. Once dispatch user confirms the message is ready for paging, it will be uploaded from the IP Audio Dispatch Console to the IP Audio Center, and when the trigger has been invoked, IP Audio Center will page the message to the selected network speakers or paging groups.

7. Text-to-Speech

ZYCOO IP Audio Center can be integrated with Google Text-to-Speech services to convert multi-language of text to voice speech (audio files). The converted voice speech can be used to create triggered paging tasks from the IP Audio Dispatch Console. Dispatch users only have to insert the text content which needs to be paged, then click to convert the text content to voice speech. Once confirmed the voice speech is qualified for paging, the audio file will be uploaded to the IP Audio Center, and when the trigger has been invoked, IP Audio Center will page the voice speech to the selected network speakers or paging groups.

8. Volume Control

The volume of each network speakers can be controlled remotely from each of the speakers' Web interface or centrally from the IP Audio Dispatch Console. Volume control actions can be performed either when the network speakers are idle or busy. If the network speakers are broadcasting, the volume adjustment will take effect immediately. For background music, dispatch users can preset a volume level then to start playing music from the IP Audio Dispatch Console. For SIP paging and emergency alarm, forced volume level can be set in the IP Audio Center system. Therefore, in certain emergency situations, the SIP paging and emergency alarm can cover enough areas for the audience to be alerted.

9. Intercom Calling

ZYCOO SIP Safety Intercom devices can be deployed as door phone intercom, inquiry or SOS terminals. The built-in press-to-talk (PTT) key can be programed with any desired number, by simply pressing the PTT key, the caller can then have two-way communication with the target number in handsfree mode.

10. Video Intercom

ZYCOO SIP Safety Video Intercom Ei-V05 has built-in HD camera, with which the dispatch user can see the real-time image from the other side of the intercom call. By default, Ei-A05 can perform voice intercom only, but if there is an existing IP camera which supports RTSP protocol then it can be linked with the Ei-A05 for video intercom the same way as Ei-V05. When someone initiates a call from the intercom device side to the dispatch user's phone number, or the dispatch user calls the intercom device. The video stream from the built-in camera of Ei-V05 or the IP camera linked with Ei-A05 will be forwarded to the IP Audio Dispatch Console and displayed in the "Video Intercom" window. Dispatch user will be able to see what's happening on the other side in real time. IP cameras which support RTSP protocol, such as Dahua and Hikvision, can be linked with Ei-A05 for video intercom. IP Audio Center administrators only have to use the RTSP URL of the IP camera to link with the Ei-A05, no further programing is required.

11. Call Monitor

Call monitor or call spy feature can be used by the dispatch user to monitor the IP phone call or intercom call conversation of two calling parties. It can be accomplished by using the IP Audio Dispatch Console to perform the actions and monitor from the dispatch phone, or by dialing the call monitor feature code directly from the IP phone to monitor the calls. During the call monitor process, the dispatch user can hear the other two calling parties, but the other two parties cannot hear the dispatch user.

12. Whisper Spy

Whisper spy is similar to call monitor and can be used with IP phone calls and intercom calls. It is used by the dispatch user to monitor the call conversations of two parties and with the ability to talk to one of the calling parties without being heard by the other calling party. This can be accomplished by using IP Audio Dispatch Console or by dialing feature code directly from the dispatch phone.

13. Barge Spy

Barge spy can be used by the dispatch user to establish a 3-way calling based on the on-going IP phone calls or intercom calls. This feature can be performed by using IP Audio Dispatch Console or by dialing feature code directly from the dispatch phone.

14. Call Split

Call split feature can be used with IP phone calls or intercom calls. It can force a selected calling party to establish a new call with the dispatch user and hang-up the other calling party. This feature can be performed by using IP Audio Dispatch Console or by dialing feature code directly from the dispatch phone.

15. Meeting

IP audio dispatch users can organize and manage meetings (or conferences) with the IP Audio Dispatch Console. Normally the participants should be IP phones. In some particular applications, intercom devices and network speakers can be also added as participants to the meetings. During a meeting, dispatch users have the options to add more participants, mute/unmute participants, kick out participant and more management options.

16. Recording

All SIP based live communications can be recorded by the ZYCOO IP Audio Center. For example, IP phone calls, SIP intercom calls, SIP paging, meetings, etc. The recordings will be saved in the server local storage as wav audio files. Each SIP session (including meetings) will generate around 1MB recording per minute. Both the IP Audio Center system administrator and the IP audio dispatch users can have access to the recordings from the IP Audio Center web interface or from the IP Audio Dispatch Console. The IP Audio Center administrator can have access to all recordings, while the IP audio dispatch users can only have access to the recordings related to the IP audio endpoints which belongs to the paging group(s) managed by that dispatch user. The recordings can be reviewed online both within the web interface and the IP Audio Dispatch Console application.

17. Centralized Management

The IP Audio Center system utilizes MQTT IoT protocol for IP audio endpoints management and control. Server Controls: music playback, alarm broadcasting, volume control, etc. Endpoints Reporting: status (idle, busy, error, offline) reporting, current volume level reporting, etc.

18. Multi-user Privileges

Administrator can create multiple IP audio dispatch users in the IP Audio Center system, and each of the dispatch users can be configured with different privilege levels. Dispatch user privileges include:

- SIP Paging
- One-click Alarm
- Intercom

- Schedule Paging Tasks
- Background Music
- External Calls

There are also 12 privilege levels which can be allocated to each of the dispatch users, from the highest 1 to the lowest 12. If two or more dispatch users manages a same paging group, then the dispatch user with high privilege level can override the operations performed by the dispatch user with lower privilege level, or in other words, the dispatch user with lower privilege level cannot perform certain operations to the higher privilege level dispatch user's operations. For example, dispatch user with higher privilege level can monitor the phone call of the lower privilege level user, but not vice versa.

19. Service Priorities

In the IP Audio Center system, IP audio services' priorities have been preconfigured with the sequences as below: SIP Paging > Alarm > Intercom > Triggered Paging (except alarm) > Background Music The services with higher priorities will interrupt the services with lower priorities, therefore, important and emergency broadcasts will reach the audiences with priority.

20. API

ZYCOO IP Audio Center provides rich API allowing integratation with other systems such as security systems, fire systems, etc. Third-party systems can interact with ZYCOO IP Audio Center via the API to invoke public address, calling and other IP audio features.

21. IP Telephony

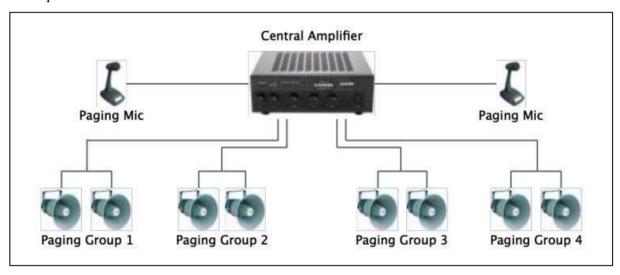
ZYCOO IP Audio Center supports SIP/IMS trunking with third-party SIP servers or ITSPs, with these trunks users will be able to make inbound and outbound phone calls through the IP Audio Center system. Users can configure flexible outbound calling rules for outbound call control. As for the inbound calls, users can create customized digital receptionist and other automatic call distribution features to handle the inbound calls. Besides the external calling features, ZYCOO IP Audio Center also supports rich internal calling features, for example, call transfer, call forward, video calling, wakeup calls, DND, and more.

3. Advantages of ZYCOO IP Audio Solution

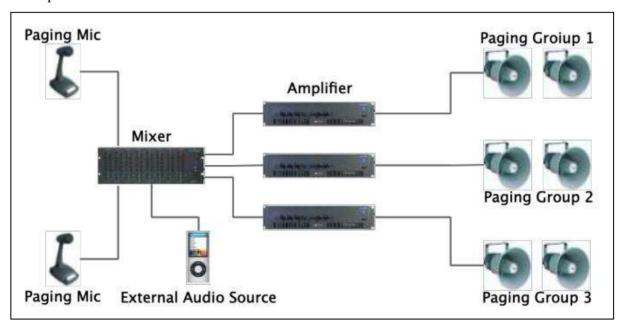
3.1 Compare with Analog Public Address Systems

First the Dispatch Users need to download the IP Audio Dispatch Console to your OS. Two examples of the analog public address system working principle.

Example A:



Example B:



Problems with the Analog Public Address Systems

1. Hard to install and manage

The analog public address system uses a very large central amplifier to connect all speakers using dedicated wiring. The status of the speakers cannot be monitored in real time.

2. Limited features and less flexibility

Due to the workzing principle of the analog public address system, only a few paging zones can be supported. To modify the paging zone will require modifications to the wiring of the system.

3. Poor Audio Quality

The analog public address system uses analog signals and transmission which will be affected by the distance of the wiring and also environmental conditions; therefore, the resulting interferences and distortions can impact audio quality.

4. Hard to use and maintain

The analog public address system uses old fashioned audio sources for paging, such as tapes, gramophone records and CDs. An on-site person will physically have to undertake a page where the analog public system is installed, there is no way to use modern technologies for remote control and management.

Advantages of ZYCOO IP Audio Solution

1. Ease of deployment and maintain

- IP Audio Center: ZYCOO IP Audio Center can be installed on either hardware servers or virtual private servers (VPS) in the cloud and can be installed on both Linux and Windows based servers.
- IP Audio Dispatch Console: IP Audio Dispatch Console can be installed on Windows, MacOS and Linux desktops the same way as the users installing any other regular applications, no expertise required at all.
- IP Audio Dispatch App: IP Audio Dispatch App can be installed directly from Apple's App Store for iPhone users, and from Google Play store for the Android phone users.

• IP Audio Endpoints: No dedicated wiring is required for the installation of ZYCOO IP audio endpoints, they use your existing IP network. All IP audio endpoints support PoE, so you don't require any external power supply to power those devices. Additionally, the IP Audio Center is capable of centralized provisioning and management of the IP audio endpoints, resulting in significant man power, time and costs saving when installing devices.

2. Powerful features

In addition to all of public address features that an analog public address system supports, ZYCOO IP Audio Solution also provides many more innovative features far beyond the capabilities of an analog public address system.

- Centralized Management: ZYCOO IP Audio Solution can perform centralized management and real-time status monitoring of all IP audio endpoints connected to the IP Audio Center. Any device error or failure will be reported by the IP Audio Center to the IP Audio Dispatch Console; therefore, dispatch users will be informed of problems immediately and are able to perform necessary actions to correct problems. This will help ensure high availability for the whole infrastructure.
- Services and Features: In addition to a complete set of IP public address features, ZYCOO IP Audio Solution also provides SIP voice intercom, video intercom, IP telephony and more advanced IP audio features. ZYCOO IP Audio Solution also provides unlimited grouping (zoning), scheduled paging, automatic emergency paging/alarm, streaming media background music, multisite support, remote paging and more advanced services and features.
- Usability: Compared to an analog public address system, ZYCOO IP Audio Solution brings in-demand usability features, for example: o Various audio source options, including superior quality MP3, prerecorded audio, text-to-speech audio, unlike a analog public address system which uses old fashioned audio source and needs to be paged from a fixed place. o Page can be instigated from PC and mobile phone-based applications, live announcement can be made through local or remote IP phones, SIP microphones and other SIP endpoints. o Graphic user interface for click to page, click to intercom, click to sound alarm, click to call and more.

3. Superior Audio Quality

For SIP based communication services such as SIP paging, the wideband audio codec G.722 can be used. The 16KHz sampling rate will provide far better audio quality and can maximize the reproduction ability of the original sound effect. Background music and other paging services which utilize streaming media, a 44.1KHz sampling rate and 320Kbps bit rate MP3 music can be streamed to the network speakers for decoding and playback. Also, with dual speaker driver units and the wide frequency range support, background music and other streaming media-based paging will provide a superb listening experience for the intended audience.

3.2 Compare with other IP Public Address Systems

Problems of other IP Public Address Systems

1. Hardware based servers

Most of the IP public address servers in the market now are hardware-based systems. The cost is usually high and it's impossible to expand the system capacity.

2. Private protocols

Private protocols are used by most of the IP public address system manufacturers to realize their services and features, existing third-party systems and endpoints cannot be integrated in to their solutions.

3. Dedicated endpoints

As private protocols have been used, choosing a server means you have to choose its dedicated endpoints. Including speakers, intercoms and phones. Your existing IP phone system and IP phones cannot be integrated with the IP public address system.

Advantages of ZYCOO IP Audio Solution

1. Software Based Server

ZYCOO IP Audio Center is a software solution, users can choose to install it on existing hardware servers or virtualization platforms in the data center or in the cloud. For upgrading of future system capacity, users only have to upgrade their generic hardware server or allocate more resources to IP Audio Center server from the virtualization platform.

2. Open Standard SIP Support

SIP is a powerful and efficient communications protocol due to its flexibility, ease of implementation, and it is easy to expand characteristics. More and more service providers and communication equipment manufacturers are providing their services and products with SIP support. ZYCOO IP Audio Solution supports open standard SIP, which means users can easily integrate the IP Audio Solution with service providers and third-party SIP enabled IP audio endpoints to enjoy more IP audio communication features.

3. Compatible with third-party SIP endpoints

In addition to ZYCOO network speakers, SIP safety intercoms, SIP paging gateways and IP phones, ZYCOO IP Audio Solution can also support thirdparty SIP enabled endpoints. For example, the existing SIP phones, SIP microphones, SIP video phones, desktop SIP softphones, mobile softphones plus other SIP endpoints. By using SIP trunking, IP Audio Solution can be integrated with service provides and existing IP phone systems for unified IP audio communications.

4. Implementation Suggestions

4.1 Server Deployment

ZYCOO IP Audio Center is a software-based solution and therefore can be integrated into a company preferred infrastructure architecture. Therefore, ZYCOO IP Audio Center can be installed either on dedicated hardware servers or a virtualized platform. The installation will be undertaken by using stand-alone ISO image for offline installation or using existing host machine for online installation of the docker container.

Offline Installation

The ISO image of the IP Audio Center is based on CentOS 7 and is suitable to be installed on a local hardware server. Users can use the ISO image provided by ZYCOO to make a USB boot disk and install it in the same way as installing CentOS Linux servers.

Online Installation

Online installation of the docker containers is suitable for installing ZYCOO IP Audio Center on existing virtualization platforms where there is a host system already installed. Users can pre-install CentOS 7 (or newer), Ubuntu 14 (or newer) and Debian 9 (or newer) Linux operating systems. Always ensure the Linux system has a good internet connection. Next just follow the instructions provided and enter several commands to complete the installation.

4.2 Network Requirement

4.2.1 Protocol Classification

The IP audio communications components of the ZYCOO IP Audio Solution are based on SIP and streaming media. The below table details the classification of the IP audio features and the technologies these features are based.

Technology	Features
	Background Music
	Timetable Triggered Paging
	Immediate Triggered Paging
Streaming media	• Dial Number Triggered Paging
	Pre-recorded Message Paging
	Text-to-Speech Paging
	Emergency Alarm
	SIP Paging (Live Announcements)
	Meeting (Audio Conference)
	• Phone Calls • Intercom Calls
SIP	Wakeup Calls
SIF	Call Monitor (Call Spy)
	• Barge Spy
	• Whisper Spy
	• Call Split

As per the previously recommended server configurations, IP Audio Center can perform full paging to all speakers using streaming media. While paging and telephony features will use SIP. The IP Audio Center can support a maximum of 500 of simultaneous paging/calls sessions. In addition to the server configuration which will limit the system capacity of the IP Audio Center, network bandwidth is also a key factor that must be considered, please read the following notes.

4.2.2 Bandwidth Requirement Calculation

Media	Channel	Bandwidth
Streaming media	1	≈33Kbps

HD codec G.722	1	≈100Kbps
Codec G.711	1	≈80Kbps

An example of actual bandwidth calculation. Customer wishes to deploy an IP Audio Solution with 100 network speakers which will be used for live SIP paging and background music. With background music paged to all network speakers. Required bandwidth: 100 * 33Kbps = 3.3Mbps With live SIP paging to all network speakers using wideband codec G.722. Required bandwidth: 100 * 100Kbps = 10Mbps In the above example, the network bandwidth needs to cover the max bandwidth taken by SIP paging using HD audio codecs. As a result, the minimum network bandwidth must be at least 10Mbps.

5. Conclusion

ZYCOO IP Audio Solution is an all-in-one solution that provides you with a complete IP public address system, an IP intercom system, plus IP phone system and more. You benefit from a full set of music, tone notification, public address, voice and video intercom calls, IP telephony plus other extremely useful IP audio communication features.

Our solution can be deployed straight into your existing IP network and for large enterprises and organizations, ZYCOO IP Audio Solution can also support multisite application scenarios. Our flexible and easy-to-use dispatch applications allow users to access all of the IP audio features available without the need for specialist expertise or training.

ZYCOO IP Audio Solution is rich in features, flexible in how it is deployed and configured, and is very simple and easy to use. It's a perfect solution for use in public safety, smart city, security community, industry, transportation, health and many other areas.

www.zycoo.com

zycoo@zycoo.com

© 2023 Zycoo Communications LLC All rights reserved