Sitech Systems (NZ) Ltd

POE Public Address System

Contents

Overview

Sitech Systems (NZ) Ltd was established back in 1988 and evolved from Waikato Projector Services in response to local area needs for an audio and visual equipment supplier.

Sitech Systems have since established itself as one of the country's most innovative integration technology suppliers, offering complete Audio and Visual solutions at the forefront of technology to the corporate, tertiary, and education market sectors in New Zealand for over 30 years.

We know that one system does not fit all, each system is designed to fit exactly what our client's needs are. We design systems that include IP network attached paging devices and door access control. We ensure all systems work together and meet your specific requirements.

We provide systems that include:

- Emergency notification announcements.
- Network attached paging system.
- Network attached intercom systems.
- Network attached bell systems.

Consultation

When we provide an emergency paging sound system, we analyse and determine the sound distribution system required. We review the sound levels and define the speakers, amplifiers, and software required to meet your objectives.

Decision Assistance

Review Alternative Solution

Once we have completed the consultation phase, we work with you to review the pros and cons of several solutions, including a review against the budget available.

We make sure that you understand all the possible solutions, so you can make an educated decision on the best choice. For example, we review the various speaker systems that you can choose from as well as the software that will do exactly what you need.

We review your return on investment, so you can understand the benefits and value of all the solutions available.

Systems Integration

Make sure it all goes together.

When necessary, we integrate multiple systems and software. We provide integration that assures all parts of the system work together.

Getting the System Installed and Working

Once you have decided on the right solution, we help you determine the best time to install the system.

We will make sure that you have the best resources available. We install all over New Zealand.

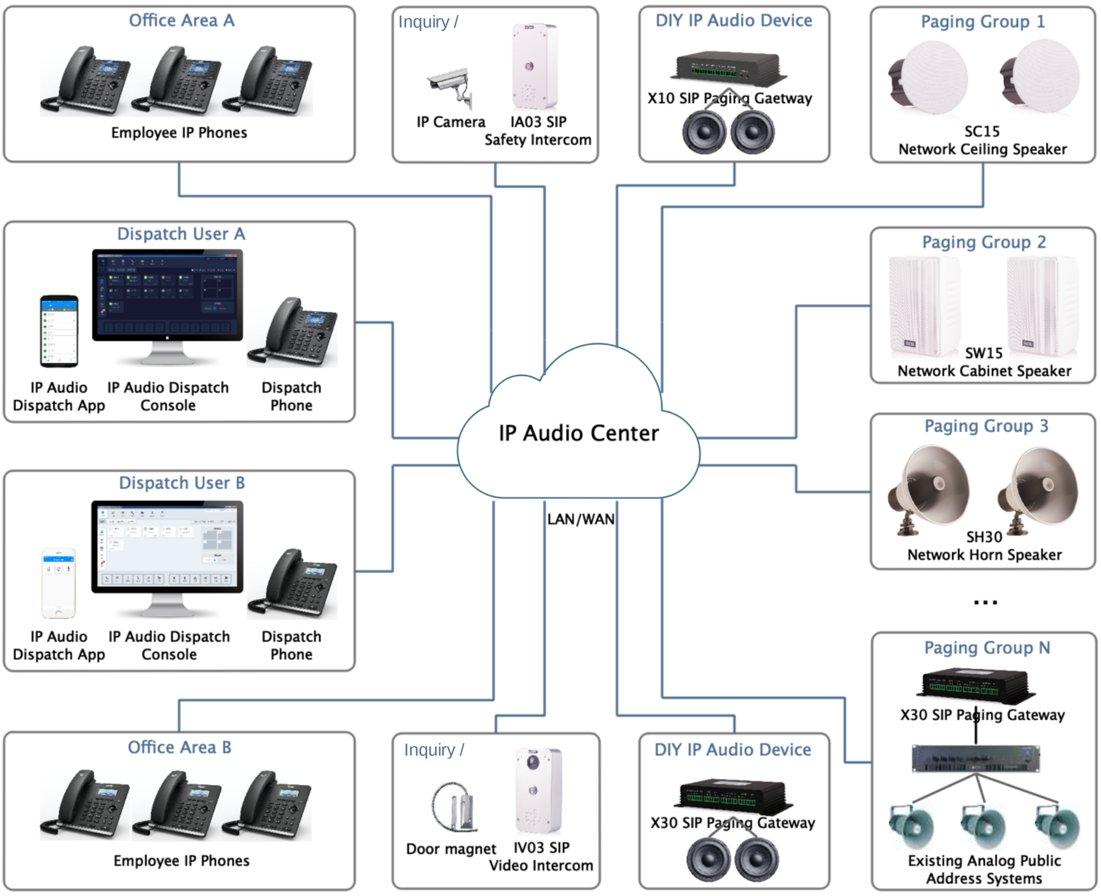
Technical Support

We assure you that the equipment you purchased operates as expected and that you get full use of all the components. We are the first people you call to get help.

1. Solution Overview

Sitech Systems IP Audio Solution is an all-in-one IP audio communication system primarily based on SIP, streaming media, and MQTT IoT technologies. Our solution provides public address, intercom, IP telephony plus IP audio communication features. SIP is the protocol used for IP audio endpoint registration, SIP paging, intercom calls, IP phone calls, etc., while streaming media is used for background music broadcasting, scheduled broadcasting, emergency alarm broadcasting, automatic broadcasting, etc. Finally, MQTT technology is used for centralised management of all IP audio endpoints.

The entire solution architecture is based on a combination of software and hardware components. Software components include the IP Audio Center, IP Audio Dispatch Console. While network speakers, SIP safety intercom, SIP paging gateways, and IP phones are all hardware-based. Below is a diagram detailing the basic architecture of the IP Audio Solution.



As detailed in the above architecture, the IP Audio Solution can be deployed across both intranet and internet. The IP Audio Center is a software solution that can be deployed on hardware-based servers or virtual private servers in the cloud. **The IP Audio Dispatch Console is a software application to be installed on personal computers.**

The network speakers, intercom devices, and IP phones can all be installed on your existing IP network. In conclusion, the whole infrastructure is designed to work on a standard IP network without the need to add specialist equipment or wiring.

2. *Products Overview*

2.1 I P Audio Center

The IP Audio Center is the engine of the IP Audio Solution. It is a software solution that can be installed on dedicated hardware or cloud-based servers.

The IP Audio Center manages all IP audio endpoints, including network speakers, intercoms, IP phones, SIP paging gateways, etc. Paging groups (zones), paging schedules, intercom calls, music/audio files are also managed by the IP Audio Center.

For simplified control and distributed management, the IP Audio Center provides integration with the IP Audio Dispatch Console for users to manage different paging groups (zones) with different IP audio applications.

Key Features and Specifications	
SIP Accounts	4000 (Max)
Simultaneous Calls	500 (Max)
Simultaneous Conference Attendees:	500 (Max)
SIP Paging:	500 (Max)
MP3 Files:	Unlimited
Playlists:	200 (Max)
Triggered Paging Tasks:	Unlimited
Paging Groups (zones):	Unlimited
Recording Time:	Unlimited

2.2 IP Audio Dispatch Console

The IP Audio Dispatch Console is a multi-platform (Windows, macOS, and Linux) application dedicated to the 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, start, and initiate emergency broadcast, monitor IP audio endpoints status, and more.

Multiple dispatch user login is supported, and they can work separately without interfering with each other's paging zones. This feature makes the distributed multi-site IP audio system possible.

Features and Specifications	
Device Status	Idle, Busy, Error, Offline
Device Display Number per Group:	Unlimited
Group Display Number:	Unlimited
A playlist can be Created:	200 (Max)
Number of Triggered Paging Tasks:	Unlimited
Music Playing Tasks:	200 (Max)
Windows	Windows7 and newer
macOS	macOS10.10 and newer
Linux	Ubuntu12.04 and newer, Fedora 21, Debian 8

2.3 Network Speakers

Our network speakers are SIP-enabled high-performance speaker products that can be used for SIP paging, notification/tone broadcast, and streaming media music playback.

The SC15 network ceiling speaker and SW15 network cabinet speaker are equipped with dual speaker drive units, the high-efficient, full-range speaker drive units can provide a uniquely advanced listening experience, which makes the SC15 and SW15 suitable for high-quality music, notification/tone broadcasting in an indoor environment.

The SH30 network horn speaker is equipped with a midrange drive unit powered by a 30W class D amplifier, this makes it suitable for paging and notification/tone broadcasting to noisy large spaces and outdoor environments.

SC15 Specifications (Speaker Ceiling Mount)

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

SW15 Specifications (Speaker Wall mount)

Speaker Components:	5.25" woofer unit + 1.5" tweeter unit
Sensitivity:	85dB/1m/1W
Max Sound Pressure Level	100dB
Amplifier	Built-in Class D Amplifier
Rated Power	**8Ω 15W**
Frequency Range	70Hz~20KHz
Coverage Pattern	90°H 90°V 30 m²

SH30 Specifications

Speaker Components	2" midrange driver unit
Sensitivity	105dB/1m/1W
Max Sound Pressure Level	117dB
Amplifier	Built-in Class D Amplifier
Rated Power:	**8Ω 30W**
Frequency Range:	400Hz~8KHz
Coverage Pattern:	50°H 50°V 70m effective distance
Acoustics:	Mono
IP Rating	IP65

2.4 SIP Safety Intercom

The SIP Safety intercom IA03 and IV03 are SIP-enabled intercom devices that can be used for consulting and SOS applications.

By default, **IA03 can be used as a voice intercom** and IV03 has a built-in HD camera that can support video intercom. If there is an existing IP camera, this can be linked with IA03 for video intercom. IP cameras that support RTSP, such as Dahua and Hikvision IP cameras can be linked with IA03.

Based on SIP and with PoE support, the installation of the IA03 and IV03 only requires a PoE enabled network cable, no power supply, and extra wiring is required.

Each of the intercom devices has been equipped with a programmable dial button, any desired number can be preconfigured. By pressing the button users can talk with the target number in hands-free mode.

IA03 and IV03 have been equipped with digital level input for connection of sensors, and with a relay switch which can be used to connect door magnet. The action (open/close the door) of the relay switch can be controlled by the DTMF signal, intercom call status, and the sensor input signal.

In addition to working with the IP Audio Center, the IA03 and IV03 intercoms can be also integrated with third-party SIP servers and the features detailed above are still applicable.

Specifications	
Speaker Components	Ø 40 mm x 12.7 mm (1.6 x 0.5 in)
Sensitivity:	90±3dB
Distortion	<10%, 1000Hz 0.1w/1m
Rated Power	3W
Max Power:	4W
Frequency Range:	630Hz ~ 4.5KHz
Acoustics:	Mono
Microphone:	-36±2dB RL=2.2K Vs=2.0V
Relay Switch:	Max voltage AC 125V-1A/DC 60V-1A
Video Resolution (IV03):	1920*1080@20fps, 1280*720@25fps

2.5 SIP Paging Gateways

The X series SIP paging gateways are SIP-enabled multifunctional IP audio devices dedicated to industry users. They can convert voice streams from a SIP paging system or IPPBX 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. Industry users can customise the X series SIP paging gateway into any desired form of intercom device or paging speaker. These can then be widely used for smart and safe city applications, to increase the efficiency of communication and information sharing.

X10 Specifications (IP Audio Converter used with outdoor horn speakers)	
Audio Output	2*10W, 8Ω SPK (4 pins) + 3.5mm audio jack
Audio Input:	3.5mm audio jack
Call Button	Support 2 switch buttons with LED indicators
Protocols	SIP(RFC3261), HTTP, TCP/IP, SSL, DNS, SNTP, NTP, RTSP, RTP, RTCP, TCP, UDP, MQTT, ICMP, DHCP, ARP, SSH
Audio Codecs	G.711(a, u), G.722, G.729
Power Supply	PoE (IEEE802.3at) or DC 12V-3A
Relay Switch	Max voltage AC 125V-1A/DC 60V-1A
Network	ETH0+ETH1 10/100Mbps
Working Temperature:	-20°C ~ +50°C
	-40°C ~ +75°C

2.6 SIP Phones

Sitech Systems SIP phones including an entry-level SIP phone with a digital display screen.

Both phones support high definition (HD) voice quality and enabled with basic call features like Call Hold, Call Park, Call Transfer, etc., and advanced features like Voicemail, Call Logs, Phonebook, Busy Lamp Field, Do Not Disturb, etc., for users to communicate in a simple and flexible way.

Specifications

- 4 SIP Lines
- HD Voice
- POE Enabled
-Two Network interfaces (10M/100M)
- 2.8 Inch Colour Display
- 2 LCDs (Main + DSS)
- Handset(HS) / Hands-free(HF) / Headphone(HP) mode
- EHS support for Plantronics headsets
- Desktop / Wall-mount installation

3. *Features 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 multi-branch offices to build a unified IP audio system that 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 an onboard 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 is 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 dialed number triggered paging. Immediately 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. IP Paging

SIP paging is used to make live announcements from a SIP IP phone 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.

4. Emergency Alarm

There are certain pre-configured emergency events in the IP Audio Center system, such as fire, earthquake, Lockdown, etc. Dispatch users can click to sound an alarm from the IP Audio Dispatch Console by using the built-in or customised emergency events as per their needs.

An emergency alarm can be achieved by dial number triggered paging, so the dispatch users can also sound an alarm by dialing a code from the IP phone or SIP microphone.

5. 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 the 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.

6. 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 they can start playing music from the IP Audio Dispatch Console.

		For SIP paging and emergency alarm, the forced volume level can be set in the IP Audio Center system. Therefore, in certain emergencies, the SIP paging and emergency alarm can cover enough areas for the audience to be alerted.
7.	**Intercom Calling**	The 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 programmed 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.
8.	**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.
9.	**Whisper Spy**	Whisper spy is similar to a 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.

10.	**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.
11.	**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.
12.	**Recording**	All SIP based live communications can be recorded by the 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 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 belong 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.
13.	**Centralised Management**	The IP Audio Center system utilizes the 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.

14.	**Multi-user Privileges**	Administrators 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 The administrator can enable or disable these privileges for the dispatch users as required. 12 privilege levels can be allocated to each of the dispatch users, from the highest 1 to the lowest 12. If two or more dispatch users manage the same paging group, then the dispatch user with a high privilege level can override the operations performed by the dispatch user with a lower privilege level, or in other words, the dispatch user with a lower privilege level cannot perform certain operations to the higher privilege level dispatch user's operations. For example, dispatch users with higher privilege levels can monitor the phone call of the lower privilege level user, but not vice versa.
15.	**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.
16.	**API**	The IP Audio Center provides a rich API allowing integration with other systems such as security systems, fire systems, etc. Third-party systems can interact with IP Audio Center via the API to invoke public address, calling, and other IP audio features.

Advantages of 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 Google Play store for Android phone users.
- **IP Audio Endpoints:** No dedicated wiring is required for the installation of these IP audio endpoints; they use your existing IP network. All IP audio endpoints support PoE, so you do not 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 manpower, time, and costs saving when installing devices.

2. Powerful features

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

- **Centralised Management：**
  IP Audio Solution can perform centralised 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 can 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, This IP Audio Solution also provides SIP voice intercom, IP telephony, and more advanced IP audio features. this 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.

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.

5. Implementation Suggestions

5.1 Server Deployment

Sitech IP Audio Center is a software-based solution and therefore can be integrated into a company's preferred infrastructure architecture. Therefore, the IP Audio Center can be installed either on dedicated hardware servers or a virtualized platform. The installation will be undertaken by using a stand-alone ISO image for offline installation or using an existing host machine for the 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 us to make a USB boot disk and install it in the same way as installing CentOS Linux servers.

Online Installation

The online installation of the docker containers is suitable for installing this IP Audio Center on existing virtualisation 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.

System capacity and recommended server configurations

	Capacity		Server Configurations			
			CPU	RAM	NIC	Storage
A	IP Audio Endpoints	100	2 cores	4GB	100Mbps	120GB
	Streaming Media Paging Capacity	100				
	SIP Session Capacity	50				
B	IP Audio Endpoints	300	4 cores	4GB	1000Mbps	500GB
	Streaming Media Paging Capacity	300				
	SIP Session Capacity	100				
C	IP Audio Endpoints	500	4 cores	8GB	1000Mbps	1TB
	Streaming Media Paging Capacity	500				
	SIP Session Capacity	200				

D	IP Audio Endpoints	1000	6 cores	8GB	1000Mbps	2TB
	Streaming Media Paging Capacity	1000				
	SIP Session Capacity	300				
E	IP Audio Endpoints	2000	8 cores	16GB	1000Mbps	RAID recommended
	Streaming Media Paging Capacity	2000				
	SIP Session Capacity	500				
F	IP Audio Endpoints	4000	8 cores	32GB	1000Mbps	RAID recommended
	Streaming Media Paging Capacity	4000				
	SIP Session Capacity	500				

5.2 Network

Protocol Classification

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

Technology	Features	
Streaming Media	• Background Music • Timetable Triggered Paging • Immediate Triggered Paging • Dial Number Triggered Paging	• Pre-recorded Message Paging • Text-to-Speech Paging • Emergency Alarm
SIP	• SIP Paging (Live Announcements) • Meeting (Audio Conference) • Phone Calls • Intercom Calls • Wakeup Calls	• 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 simultaneous paging/call 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.

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. The 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.

6. Conclusion

Sitech Systems 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 organisations, This IP Audio Solution can also support multisite application scenarios. Our flexible and e a s y to use dispatch applications allow users to access all the IP audio features available without the need for specialist expertise or training.

Sitech Systems IP Audio Solution is rich in features, flexible in how it is deployed and configured, and is amazingly simple and easy to use. A perfect solution for use in public safety, smart city, the security community, industry, transportation, health, and many other areas.

An End to End Integrated Solutions Company

We know that one system does not fit all, each system is designed to fit exactly what our client's needs are.

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