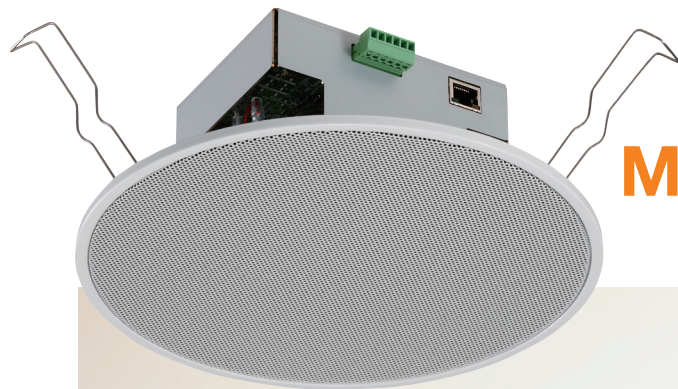# Sophisticated IP Audio Endpoints

IP-A1 Series

Beyond Audio Communications

More than meets the eye

All of these features are on board...

No, it's an **Audio file player.**

This is obviously a **Speaker.**

Isn't it a **Time Scheduler?**

It must be an **Audio matrix!**

IP-A1 series is a group of sophisticated IP audio endpoint devices which are designed in different forms. Although it looks like a simple speaker or an I/F box, it is capable of much more features than its appearance and performs as a minimal PA system even with a single device while multiple devices can also be managed as one controlled PA system.

01 What is IP-A1?

IP-A1 series consists of a variety of commercial-grade IP audio endpoints, which can be used as an independent audio system or a fully integrated audio communication system to be configured and operated in conjunction with external systems and platforms such as security video monitoring, access control, digital signage or fire alarm systems.

Common Key Features

Audio File Storage	Standard Protocols	Audio Management	Easy Configuration	Integration Friendly
MP3/WAV 80MB	SIP, Onvif Multicast	Priority & Volume	Browser UI & Software	HTTP API & Contact In/Out

Audio communication system design can be much simpler and more flexible with IP-A1 series
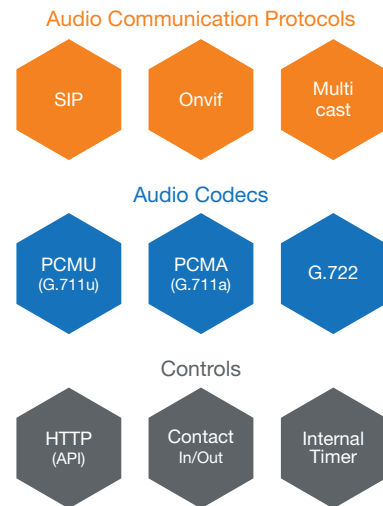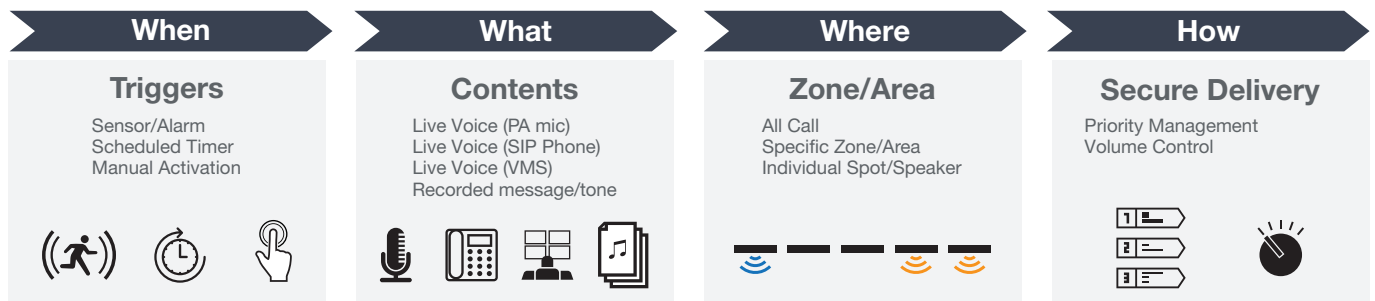
02 Why IP-A1?

Integration-friendly

IP-A1 series IP endpoints adopt common industrial standard protocols for its audio communications and controls, which helps to establish fully integrated systems by communicating not only between IP-A1 series devices but also with external devices and platforms such as SIP phone, security VMS (Video Management Software), Access Control or Sensing systems.

Adding an endpoint or group of endpoints into a commercial communication system brings it to the next level for being capable of flexible audio communications over the network.

Audio Communication Protocols

SIP · Onvif · Multi cast

Audio Codecs

PCMU (G.711u) · PCMA (G.711a) · G.722

Controls

HTTP (API) · Contact In/Out · Internal Timer

IP-A1 can be the easiest "**piece**" to be added for fulfilling **Audio System** requirements in your integration project designs.

Key Elements of Audio Communications

When	What	Where	How
Triggers	**Contents**	**Zone/Area**	**Secure Delivery**
Sensor/Alarm Scheduled Timer Manual Activation	Live Voice (PA mic) Live Voice (SIP Phone) Live Voice (VMS) Recorded message/tone	All Call Specific Zone/Area Individual Spot/Speaker	Priority Management Volume Control

IP-A1 series is designed to handle these elements flexibly for meeting evey single project requirements.

Server-less & Scalable

IP-A1 series does not require a dedicated server for its operation in standalone mode, so that the system budget can be minimized. The simplest PA system can be established by a single endpoint device such as IP Horn Speaker, while a building wide or even community wide audio communication system can also be configured with a large number of endpoints designed in different forms.
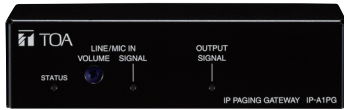
Server

Spot Accouncement

Large-scale Broadcast

03 Lineup

IP Paging Gateway
IP-A1PG

- Convert SIP audio, ONVIF Audio Backchannel, internal audio files or local audio source into Multicast streaming
- System mute function to mute all broadcasts made by every single IP-A1 series devices within the same network
- 1 local audio input (LINE/MIC selectable, phantom power On/Off)
- 4 control inputs and 1 control output
- HTTP commands (receive/send)
- Audio file storage (20 files, total 80MB, WAV/MP3)
- PoE powered
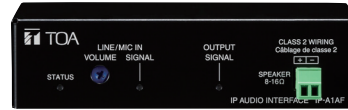
IP Audio Interface
IP-A1AF

- Receive SIP audio, ONVIF Audio Backchannel and Multicast
- Local broadcast using internal audio files or local audio source
- 1 audio input (LINE/MIC selectable, phantom power On/Off)
- 8W (PoE)/15W(PoE+) built-in amplifier, 1 LINE audio output
- 2 control inputs and 1 control output
- HTTP commands (receive)
- Audio file storage (20 files, total 80MB, WAV/MP3)
- Playback programs (Repeat, Weekly Timer)
- PoE/PoE+ powered

IP Ceiling Speaker
IP-A1PC238

- 16cm (6") cone-type speaker for in-ceiling installations
- Receive SIP audio, ONVIF Audio Backchannel and Multicast
- Local broadcast using internal audio files or local audio source
- 1 audio input (LINE/MIC selectable, phantom power On/Off)
- 8W built-in amplifier, 1 LINE audio output
- 2 control inputs and 1 control output
- HTTP commands (receive)
- Audio file storage (20 files, total 80MB, WAV/MP3)
- Playback programs (Repeat, Weekly Timer)
- PoE powered

IP Horn Speaker
IP-A1SC15

- 124dB (PoE+ powered) with IP66 rating for outdoor installations
- Receive SIP audio, ONVIF Audio Backchannel and Multicast
- Local broadcast using internal audio files
- 8W (PoE)/15W(PoE+) built-in amplifier, 1 LINE audio output
- 2 control inputs and 1 control output
- HTTP commands (receive)
- Audio file storage (20 files, total 80MB, WAV/MP3)
- Playback programs (Repeat, Weekly Timer)
- PoE/PoE+ powered

04 Key Broadcast Functions

Internal Audio File

| PG | AF | PC238 | SC15 |

- Up to 20 audio files (Total 80MB)
- MP3, WAV
- Volume level, Number of times to repeat and Interval can be specified.

Playback

Audio File Formats

WAV: 8/16/44.1/48 kHz sampling frequency, 8/16bit, mono/stereo
MP3: 32/44.1/48 kHz sampling frequency, 64-320 kbps, CBR/VBR, mono/stereo

SIP Broadcast

| AF | PC238 | SC15 |

- Broadcasts can be made using SIP protocol via SIP server.

SIP-compatible devices and Apps → **SIP Broadcast**

Compatible Audio Codecs

G.722, PCMU (G.711u), PCMA (G.711a)

VMS (Onvif Audio Backchannel) Broadcast

| AF | PC238 | SC15 |

- Broadcasts can be made using Onvif Audio Backchannel protocol from VMS (Video Management System) software.

VMS Client → **VMS Broadcast**

Compatible Audio Codecs

PCMU (G.711u)

Multicast Streaming

| PG | AF | PC238 | SC15 |

- Up to 20 multicast addresses and ports can be managed and streamed by one IP-A1PG.
- Each receiving device is capable of 20 multicast ports.

IP-A1PG → **Multicast #1** / **Multicast #2** / ⋮ / **Multicast #20**

Compatible Audio Codecs

G.722, PCMU (G.711u), PCMA (G.711a) – Auto codec recognition

2-way Communication

| AF |

- Audio back stream can be made for audio monitoring and/or conversation applications.

SIP Phone/Intercom
VMS Client
SIP, VMS, Multicast
SIP, VMS
Microphone

Audio Input

Audio Input: LINE/MIC (LINE: 0dB, MIC: -60dB), PAD, Phantom Power On/Off

Priority Management

| PG | AF | PC238 | SC15 |

- Broadcast priorities can be changed between broadcast types and patterns on each device.

IP-A1PG
SIP
VMS
Pattern 1-20   **23**
LINE/MIC IN

SIP
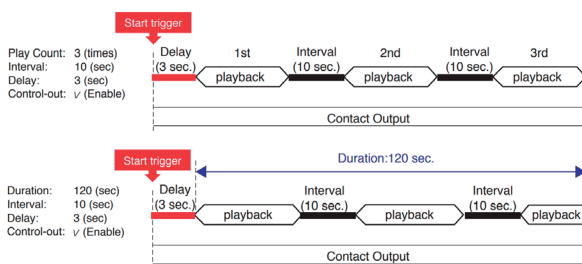VMS
Muticast 1-20
Pattern 1-20   **43**
Local

Default Priorities (High to Low)

IP-A1PG: SIP, VMS Pattern 1-20, LINE/MIC IN
Receivers: SIP, VMS Multicast 1-10, Pattern 1-20, Multicast 11-20, Local

Broadcast Patterns

| PG | AF | PC238 | SC15 |

- Up to 20 Broadcast Patterns can be registered by using internal audio files.
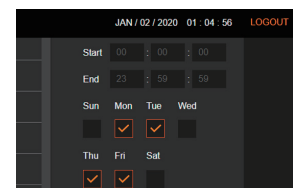- Play mode can be selected from the following;

Start trigger

Play Count: 3 (times)
Interval: 10 (sec)
Delay: 3 (sec)
Control-out: ∨ (Enable)

Delay (3 sec.) | 1st playback | Interval (10 sec.) | 2nd playback | Interval (10 sec.) | 3rd playback
Contact Output

Play Count

-Specify the number of times to repeat
-Specify Interval and Delay time.
-Enable/disable control-out.

Start trigger

Duration: 120 (sec)
Interval: 10 (sec)
Delay: 3 (sec)
Control-out: ∨ (Enable)

Duration:120 sec.
Delay (3 sec.) | playback | Interval (10 sec.) | playback | Interval (10 sec.) | playback
Contact Output

Duration

-Specify the total duration time to repeat.
-Specify Interval and Delay time.
-Enable/disable control-out.

JAN / 02 / 2020   01 : 04 : 56   LOGOUT

Start   00 : 00 : 00
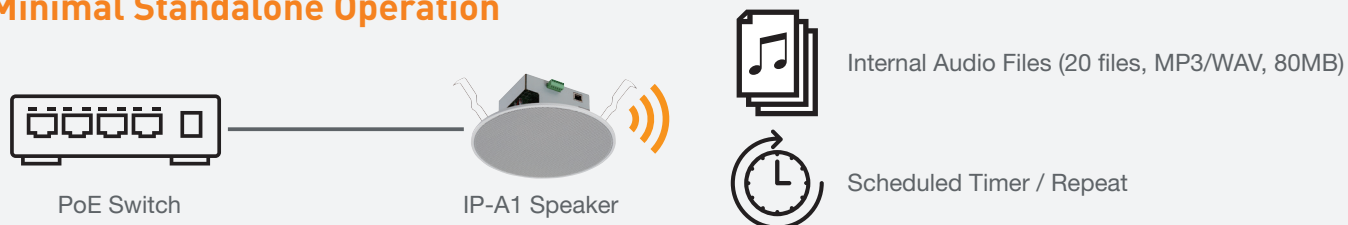End   23 : 59 : 59

Sun   Mon   Tue   Wed
Thu   Fri   Sat

Weekly Timer
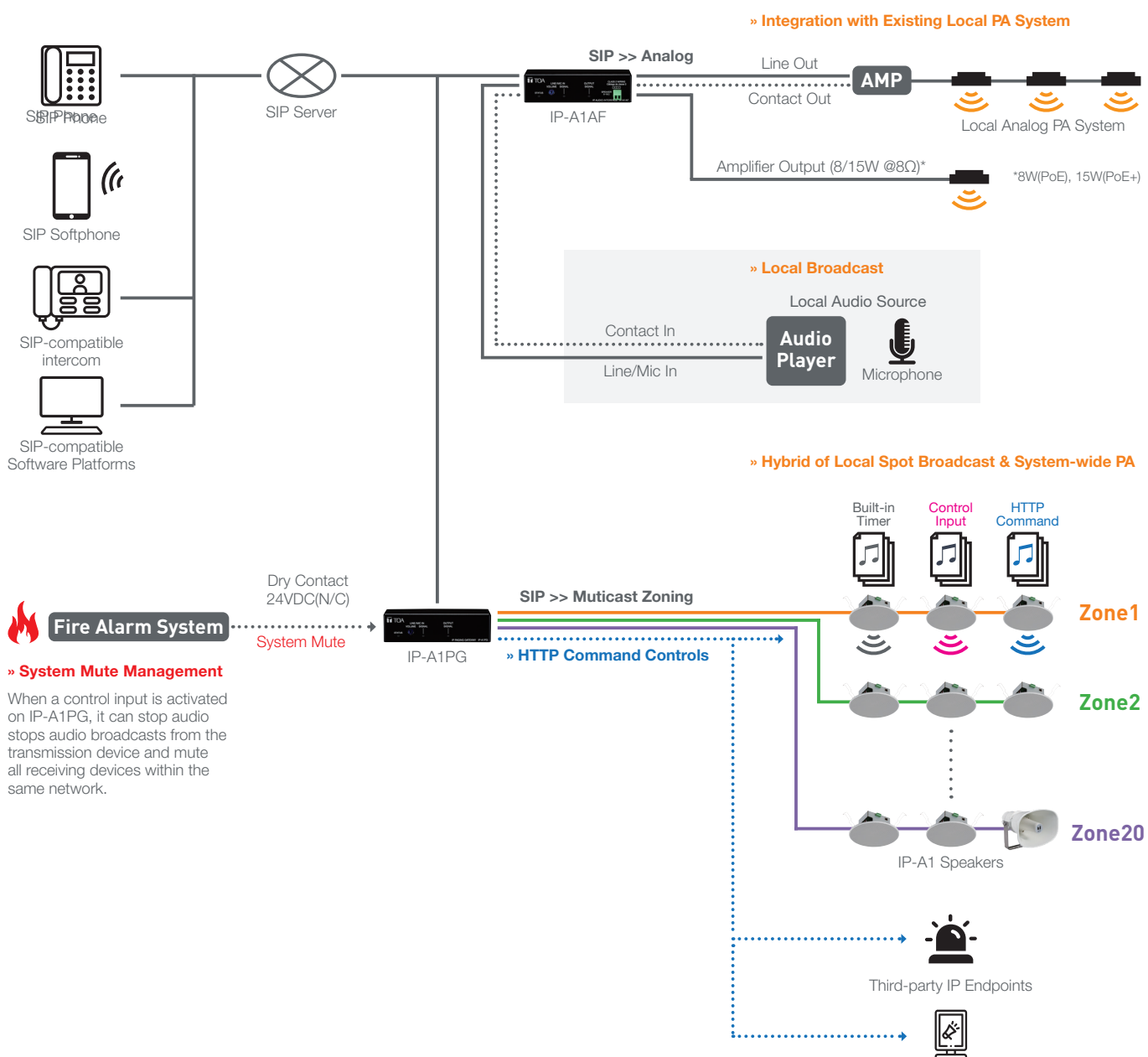
-Specify Start and End time to repeat.
-Select applicable days.
*This functions is available only on IP-A1AF, IP-A1PC238.

05 Applications

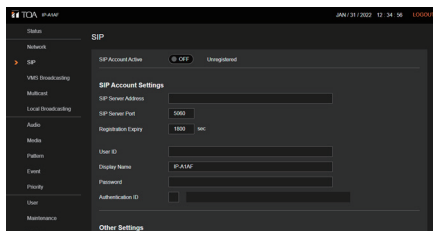## Minimal Standalone Operation

PoE Switch — IP-A1 Speaker

Internal Audio Files (20 files, MP3/WAV, 80MB)

Scheduled Timer / Repeat

Sophisticated PA System Integrations

» Integration with Existing Local PA System

SIP >> Analog

IP-A1AF

Line Out — AMP — Local Analog PA System

Contact Out

Amplifier Output (8/15W @8Ω)* *8W(PoE), 15W(PoE+)

SIP Phone

SIP Softphone

SIP-compatible intercom

SIP-compatible Software Platforms

SIP Server

» Local Broadcast

Local Audio Source

Contact In

Line/Mic In

Audio Player — Microphone

» Hybrid of Local Spot Broadcast & System-wide PA

Built-in Timer | Control Input | HTTP Command

Dry Contact 24VDC(N/C)

Fire Alarm System

SIP >> Muticast Zoning

System Mute

IP-A1PG

» HTTP Command Controls

Zone1

Zone2

Zone20

IP-A1 Speakers

» System Mute Management

When a control input is activated on IP-A1PG, it can stop audio stops audio broadcasts from the transmission device and mute all receiving devices within the same network.
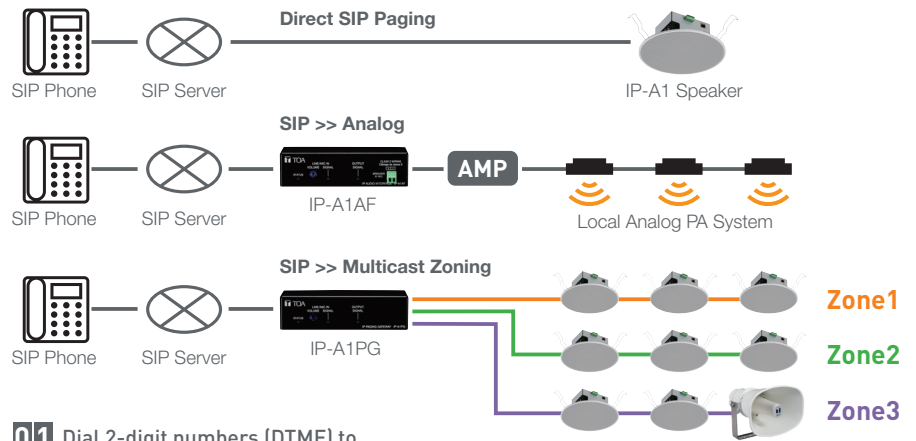
Third-party IP Endpoints

SIP Phone System Integrations

IP-A1 Browser Interface
(SIP Account Setting Menu)

» As simple as adding a "**Phone**"

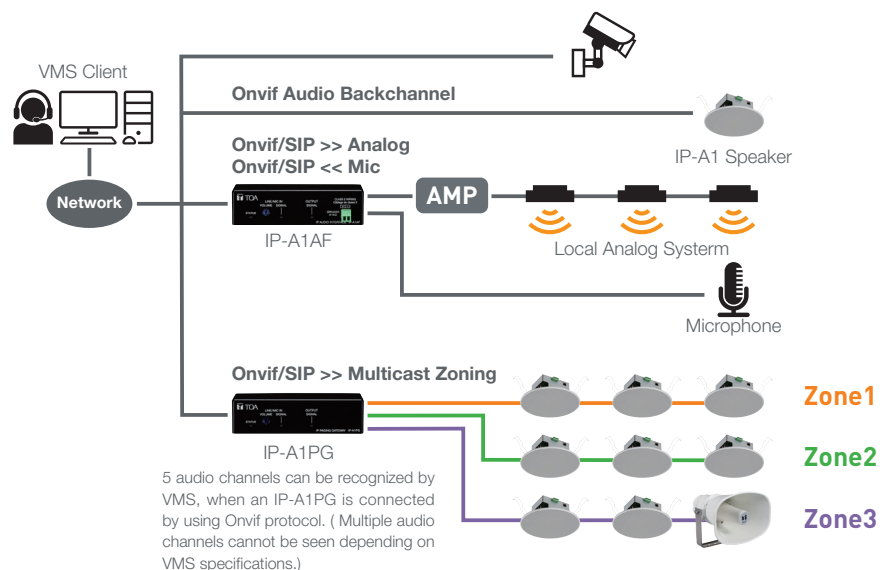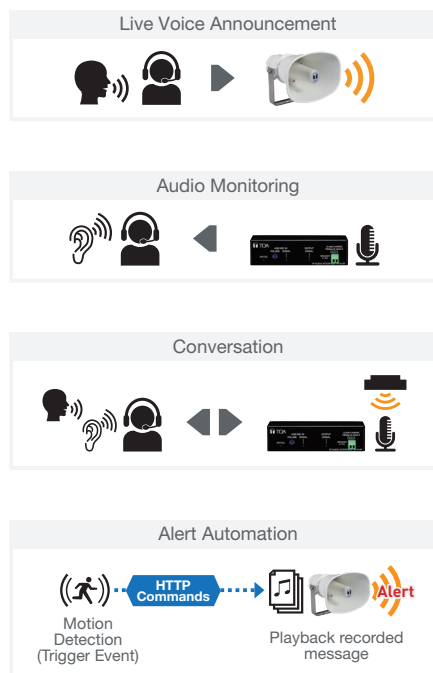**Direct SIP Paging**

SIP Phone   SIP Server   IP-A1 Speaker

SIP >> Analog

SIP Phone   SIP Server   IP-A1AF   **AMP**   Local Analog PA System

SIP >> Multicast Zoning

SIP Phone   SIP Server   IP-A1PG   **Zone1**
                                                    Zone2
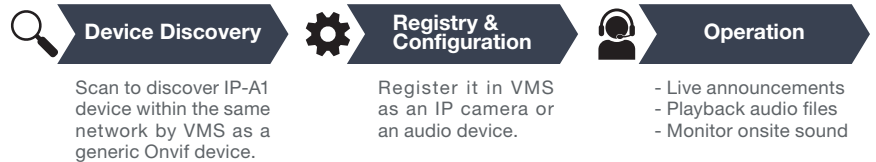   **Zone3**

01 Dial 2-digit numbers (DTMF) to
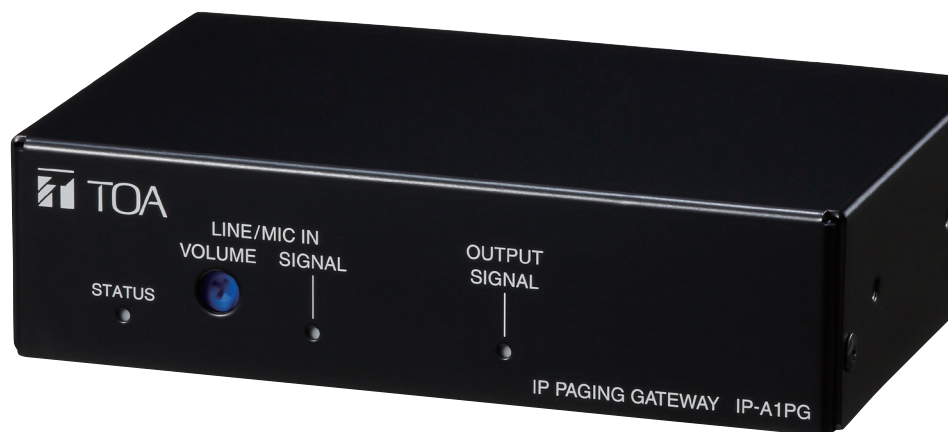select Multicast channel/zone

Security VMS Integrations

» As simple as adding a "**Camera**"

Device Discovery

Scan to discover IP-A1 device within the same network by VMS as a generic Onvif device.

Registry & Configuration

Register it in VMS as an IP camera or an audio device.

Operation

- Live announcements
- Playback audio files
- Monitor onsite sound

Live Voice Announcement

Audio Monitoring

Conversation

Alert Automation

Motion Detection (Trigger Event)   **HTTP Commands**   Playback recorded message

VMS Client   **Onvif Audio Backchannel**

Onvif/SIP >> Analog
Onvif/SIP << Mic

Network   IP-A1AF   **AMP**   Local Analog Systerm

IP-A1 Speaker

Microphone

Onvif/SIP >> Multicast Zoning

IP-A1PG   **Zone1**
                         Zone2
                         Zone3

5 audio channels can be recognized by VMS, when an IP-A1PG is connected by using Onvif protocol. (Multiple audio channels cannot be seen depending on VMS specifications.)

06 Features

IP-A1PG
the Intelligence of IP-A1 Series

IP-A1PG is designed to manage a variety of functions to make IP-A1 series a powerful communication system, while being integrated with external systems and platforms for receiving and sending signals to each other.


Audio Protocol Conversion

A variety of audio sources sent via different protocols can be converted into multicast format.

Analog MIC/LINE

Internal Audio File (MP3/WAV)

Onvif Backchannel

SIP
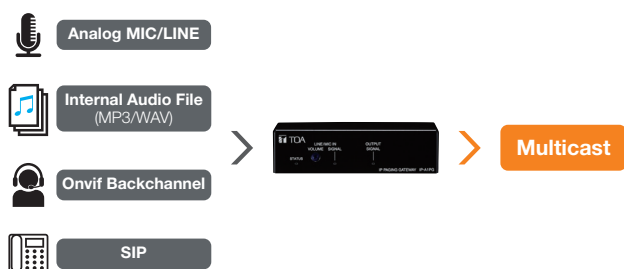
Multicast

Multicast Zoning

Up to 20 multicast addresses and ports can be managed by one IP-A1PG for zoning broadcast applications.

Channnel #1
#2
#3
#20

Multicast Transfer Channel Settings
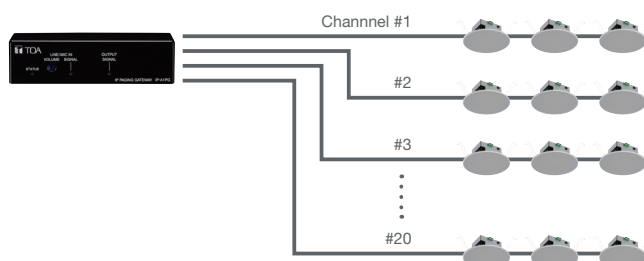
	Group Name	Multicast Address	Port
1	Multicast 1	239 . 239 . 14 . 1	48000
2	Multicast 2	239 . 239 . 14 . 2	48002
3	Multicast 3	239 . 239 . 14 . 3	48004

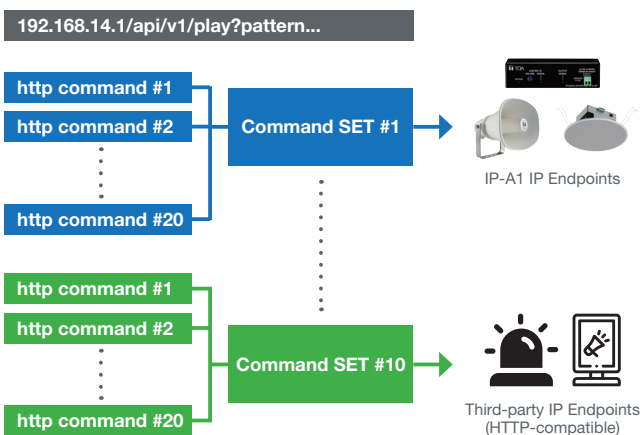
System Mute

All broadcasts made by IP-A1 series endpoints within the same network can be muted at once by triggering the control input.
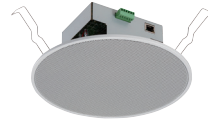

Dry Contact 24VDC(N/C)

MUTE

MUTED
MUTED
MUTED
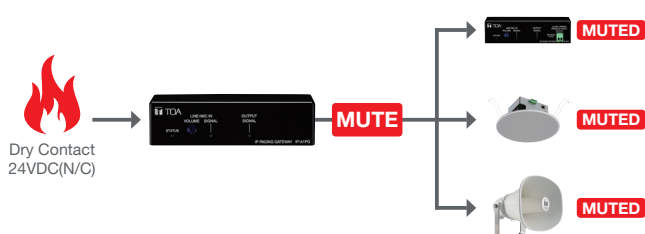
HTTP Command Distribution

General HTTP commands can be registered and sent out as a set to any HTTP command-compatible devices.

192.168.14.1/api/v1/play?pattern...

http command #1
http command #2
http command #20

Command SET #1

IP-A1 IP Endpoints

http command #1
http command #2
http command #20

Command SET #10

Third-party IP Endpoints (HTTP-compatible)

	IP-A1AF	IP-A1PC238	IP-A1SC15
Powersorce	PoE+ / PoE	PoE	PoE+ / PoE
Audio Protocols: SIP, Onvif, Multicast	✓	✓	✓
Audio Protocols: PCMU G.711u, PCMA G.711a, G.722	✓	✓	✓
Two-way Communication (MIC Input)	✓	–	–
Audio Output	✓	–	–
Audio Strage: Up to 20, 80MB, MP3 WAV	✓	✓	✓
Weekly Timer Triggering Broadcast Patterns	✓	✓	✓
Controls: HTTP (API), Contact In/out	✓	✓	✓
Environmental Ratings	– (-30 to +55°C / -22 to 131°F)	– (0 to +50°C / 32 to 122°F)	IP66 (-30 to +55°C / -22 to 131°F)

» Priority Management

Broadcast priority can be flexibly configured on each endpoint device independently.

High	
1	SIP
2	VMS Broadcasting
3	Multicast 1
	⋮
12	Multicast 10
13	Pattern 1(None)
	⋮
32	Pattern 20(None)
Low	

» Individual Volume Adjustment

Individual (Master and each Input) volume level can be flexibly adjusted to uniform the output level or set specific broadcasts at higher level intentionally.

Speaker Volume
Speaker Mute ☐
Master Volume 31: -30 dB
Offset Volume 0: 0 dB

Input Volume
SIP 0: 0 dB
VMS Broadcasting 0: 0 dB
Multicast 1 0: 0 dB
Multicast 2 0: 0 dB

» Weekly Timer

Weekly Timer function is available to play broadcast patterns by specifying "Start" time, "End" time and effective Day of Week.

JAN / 02 / 2020   01 : 04 : 56   LOGOUT
Start 00 : 00 : 00
End 23 : 59 : 59
Sun Mon ✓ Tue ✓ Wed
Thu ✓ Fri ✓ Sat

07 What can be achieved by HTTP commands?

http

Remote API
HTTP

IP-A1 Series

Play and Stop Internal Audio Files

| PG | AF | PC238 | SC15 |

> Internal audio files can be played back and stopped.

HTTP

MP3/WAV

Playback & Stop

Number of times to repeat / Interval time / Volume Level and others

Initiating a SIP Call

| AF | PC238 | SC15 |

> A SIP call can be initiated and cancelled from an IP-A1 device to a pre-registered SIP phone.

HTTP

SIP Call

SIP Phone

Volume Setting

| PG | AF | PC238 | SC15 |

> Master volume, Master offset volume and Internal audio source volume can be adjusted and configured.

HTTP

"Command Set" Distribution

| PG |

> Up to 10 pre-registered HTTP Command Set can be distributed from IP-A1PG.

HTTP

Command

IP-A1 IP Endpoints

Third-party IP Endpoints
(HTTP-compatible)

Get Device Status and Setting Values

| PG | AF | PC238 | SC15 |

> Device status and setting values can be obtained.

HTTP

- List of valid audio files
- Volume level
- Broadcasting status
- Current time
  and more

Responded

Device Maintenance

| PG | AF | PC238 | SC15 |

> Key device maintenance operations can be performed.

HTTP

- Reboot
- Firmware update
- Config file upload/download
- Time Zone setting
- Current Time setting
  and more
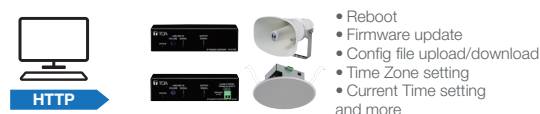
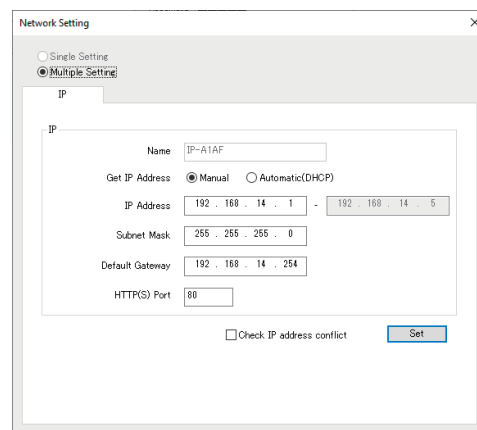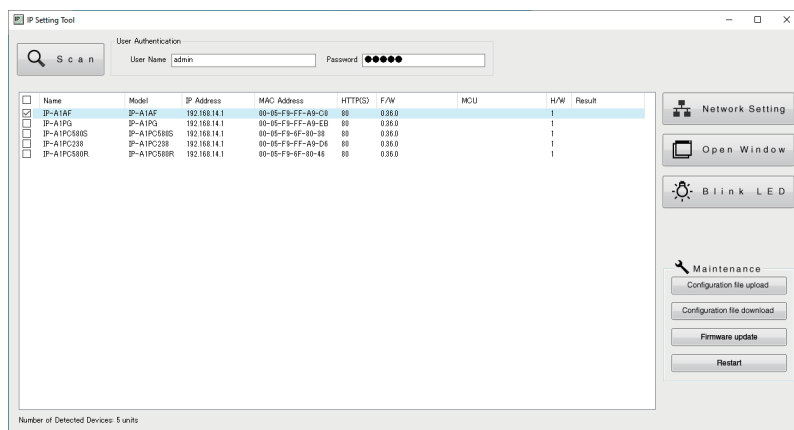Please contact us to get the full command list.

# 08 IP Setting Tool Software
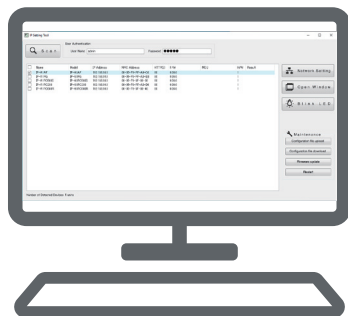
(Free to download)

- All IP-A1 series endpoint devices within the same network can be discovered and displayed as a list.
- Configuration file can be downloaded/uploaded.
- Firmware can be updated.
- Basic network settings can be configured on single device or multiple devices.
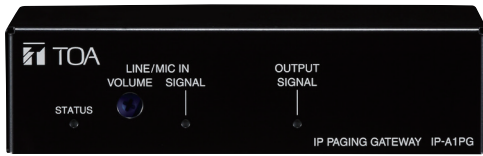

PC Requirements

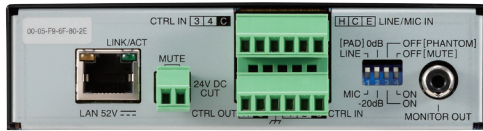| OS	Windows 10 Pro (64bit) / 10 Home (64bit) / 11 Pro (*) / 11 Home (*)
Browser	Google Chrome, Microsoft Edge
Display	Resolution: 1366 x 768 or more

*excluding IP-A1SC15
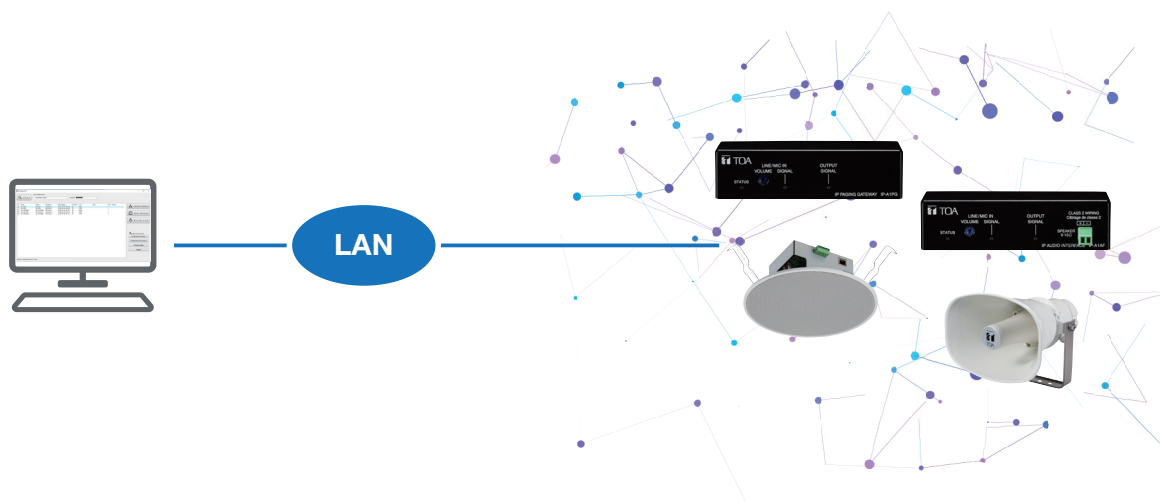
IP-A1PG   IP Paging Gateway

IP-A1PG front

IP-A1PG rear

> Convert SIP audio, ONVIF Audio Backchannel, internal audio files or local audio source into Multicast streaming

> System mute function to mute all broadcasts made by every single IP-A1 series devices within the same network

> 1 local audio input (LINE/MIC selectable, phantom power On/Off)

> 4 control inputs and 1 control output

> HTTP commands (receive/send)

> Audio file storage (20 files, total 80MB, WAV/MP3)

> PoE powered

Specifications

	IP-A1PG
Power Source	PoE(IEEE802.3af Class 3)
Power Consumption	2.5 W
Audio Transmition Method	Multicast Audio Streaming
Audio Codec	PCMU(G.711u), PCMA(G.711a), G.722
Audio Delay Time	Min. 100 ms(*1)
Network I/F	100BASE-TX, Auto MDI/MDI-X, RJ45 connector
Network Protocol	TCP/IP, UDP, HTTP, RTP, RTSP, RTCP, ARP, ICMP, IGMPv3, NTP, SIP(RFC3261)
Audio Input	1 channel, electronically-balanced, 10 kΩ LINE/MIC selectable (Rated input: LINE: 0 dB (*2), MIC: -60 dB (*2)) PAD function (-20 dB (*2)), Phantom power ON/OFF (12 V DC), volume adjustable removable terminal block (6 pins)
Monitor Output	1 channel, electronically-balanced, 600 Ω or less Rated output: 0 dB (*2), RCA pin jack
Control Input	4 channels, no-voltage make contact inputs, open voltage: 5 V DC, short-circuit current: 2 mA or less, removable terminal block (6 pins)
Mute Control Input	1 channel, 24 V DC cut signal, control current 5 mA or less, removable terminal block (2 pins)
Control Output	1 channel, open collector output, withstand voltage: 30 V DC, control current: 10 mA or less, removable terminal block (6 pins)
Indicator	STATUS (green/blue/orange/red), LINE/MIC IN (green/red), OUTPUT (green),LINK/ACT (green)
Broadcasting	Audio transmission Transmit internal messages by multicast audio streaming Transmit audio from audio input connected devices by multicast audio streaming Audio conversion Convert SIP voice to multicast audio stream and transmit Convert ONVIF Audio Backchannel audio to multicast audio stream and transmit
Event	Execute event triggered by control input Configurable actions: Internal message broadcast, audio input broadcast, command set transmission, broadcast disable, system mute
Internal Message	Max. 20 messages (Max. recording capacity: 80 MB) Supported fie format: WAV file: 8/16/44.1/48 kHz sampling frequency, 8/16 bit, monaural/stereo MP3 file: 32/44.1/48 kHz sampling frequency, 64 - 320 kbps, CBR/VBR, monaural/stereo Repeat playback: Playcount (1 - 10 times) or Duration (5 - 3600 sec) Interval time: 0 - 60 sec, Delay time: 0 - 30 sec
Command Set	20 commands can be registered in each of 10 command sets
Clock Accuracy	±13 seconds per month
Time Adjustment	Manual time setting, Time adjustment by NTP server
Power Outage Protection Period	24 hours (RTC time retention, at 40 °C (104 °F))
Operating Temperature	0 °C to +40 °C (32 °F to 104 °F)
Operating Humidity	90 %RH or less (no condensation)
Finish	Front case: Surface-treated steel plate, black, paint Rear chassis: Surface-treated steel plate
Dimensions	126 (W) x 33 (H) x 80 (D) mm (4.96" x 1.3" x 3.15") (excluding projection)
Weight	390 g (0.86 lb)
Accessory	Removable terminal plug (6 pins, preinstalled on the unit) ...2, Removable terminal plug (2 pins, preinstalled on the unit), Rubber feet ...4, Mounting screw (M3 x 6) ...4

(*1) When using Monitor output, assume an audio delay time.
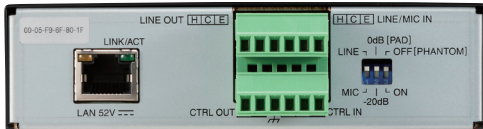(*2) 0 dB = 1 V
*ONVIF is a registered trademark of ONVIF Inc.

IP-A1AF   IP Audio Interface
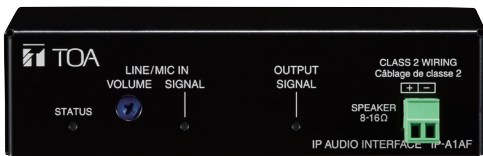
IP-A1AF front

IP-A1AF rear

- Receive SIP audio, ONVIF Audio Backchannel and Multicast
- Local broadcast using internal audio files or local audio source
- 1 audio input (LINE/MIC selectable, phantom power On/Off)
- 8W (PoE)/15W(PoE+) built-in amplifier, 1 LINE audio output
- 2 control inputs and 1 control output
- HTTP commands (receive)
- Audio file storage (20 files, total 80MB, WAV/MP3)
- Playback programs (Repeat, Weekly Timer)
- PoE/PoE+ powered

Specifications

|	IP-A1AF
Power Source	PoE+ (IEEE802.3at Class 4), PoE (IEEE802.3af Class 3)
Power Consumption	22 W (at PoE+ powered, rated output) 12.95 W (at PoE powered, rated output) 5 W (IEC62368-1)
Amplifier Rated Output	15 W (at PoE+, powered, 8 Ω) 8 W (at PoE, powered, 8 Ω) Applicable impedance: 8 - 16 Ω
Frequency Response	50 Hz - 20 kHz
Audio Codec	PCMU (G.711u), PCMA (G.711a), G.722
Audio Delay Time	Min. 100 ms (*1)
Broadcasting Mode	SIP Broadcasting/SIP calling Mode: PCMU/PCMA/G.722, P2P/SIP Server Connection Multicast Broadcasting Mode: PCMU/PCMA/G.722 Auto codec recognition, 20 ports VMS Broadcasting Mode: ONVIF Audio Backchannel, PCMU Internal Message Broadcasting Mode Local Broadcasting Mode: Output from LINE/MIC IN to SPEAKER OUT Note: Each broadcast mode can be assigned an order of priority using the Priority Setting function.
Internal Messages	Max. 20 messages (Max. recording capacity: 80 MB) Supported file formats WAV file: 8/16/44.1/48 kHz sampling frequency, 8/16 bit, monaural/stereo MP3 file: 32/44.1/48 kHz sampling frequency, 64 - 320 kbps, CBR/VBR, monaural/stereo Repeat playback: Playcount (1 - 10 times), Duration (5 - 3600 sec) or Timer (from Start time to End time) Interval time: 0 - 60 sec, Delay time: 0 - 30 sec Trigger: Control Input or Remote API (HTTP)
Network I/F	100BASE-TX, Auto MDI/MDI-X, RJ45 connector
Network Protocol	TCP/IP, UDP, HTTP, RTP, RTSP, RTCP, ARP, ICMP, IGMPv3, NTP, SIP (RFC3261)
Audio Input	1 channel, electronically-balanced, 10 kΩ LINE/MIC selectable (Rated input: LINE: 0 dB (*2), MIC: -60 dB (*2)) PAD function (-20 dB (*2)), Phantom power ON/OFF (12 V DC), volume adjustable removable terminal block (6 pins)
Audio Output	1 channel, electronically-balanced, 600 Ω or less Rated input: 0 dB (*2), removable terminal block (6 pins)
Control Input	2 channels, no-voltage make contact inputs, open voltage: 5 V DC, short-circuit current: 2 mA or less, removable terminal block (6 pins)
Control Output	1 channel, open collector output, withstand voltage: 30 V DC, control current: 10 mA or less, removable terminal block (6 pins)
Indicator	STATUS (green/blue/orange/red), LINE/MIC IN (green/red), OUTPUT (green),LINK/ACT (green)
Clock Accuracy	±13 seconds per month
Time Adjustment	Manual time setting, Time adjustment by NTP server
Power Outage Protection Period	24 hours (RTC time retention, at 40 °C (104 °F))
Operating Temperature	-30 °C to +55 °C (-22 °F to 131 °F)
Operating Humidity	90 %RH or less (no condensation)
Finish	Front case: Surface-treated steel plate, black, paint Rear chassis: Surface-treated steel plate
Dimensions	126 (W) x 33 (H) x 80 (D) mm (4.96" x 1.3" x 3.15") (excluding projection)
Weight	390 g (0.86 lb)
Accessory	Removable terminal plug (6 pins, preinstalled on the unit) ...2, Removable terminal plug (2 pins, preinstalled on the unit) ...1, Rubber feet ...4, Mounting screw (M3 x 6) ...4

(*1) When using in Local Input Broadcasting Mode, assume Audio Delay Time.
(*2) 0 dB = 1 V
*ONVIF is a registered trademark of ONVIF Inc.

IP-A1PC238　IP Ceiling Speaker
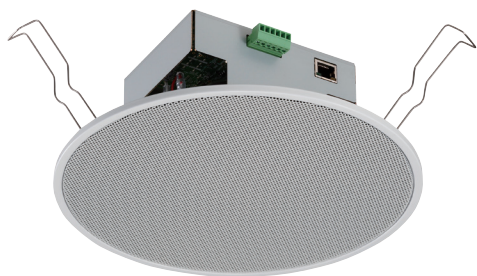
- 16cm (6") cone-type speaker for in-ceiling installations
- Receive SIP audio, ONVIF Audio Backchannel and Multicast
- Local broadcast using internal audio files or local audio source
- 1 audio input (LINE/MIC selectable, phantom power On/Off)
- 8W built-in amplifier, 1 LINE audio output
- 2 control inputs and 1 control output
- HTTP commands (receive)
- Audio file storage (20 files, total 80MB, WAV/MP3)
- Playback programs (Repeat, Weekly Timer)
- PoE powered

Specifications

	IP-A1PC238
Power Source	PoE (IEEE802.3af Class 3)
Power Consumption	12.95 W (rated output) 5 W (IEC62368-1)
Amplifier Rated Output	8 W
Sensitivity	94 dB (1 W, 1 m) (500 Hz - 5 kHz, pink noise)
Maximum Sound Pressure Level	103 dB (8 W, 1 m)
Frequency Response	60 Hz - 20 kHz (peak - 20 dB)
Speaker Component	16 cm (6") cone-type
Audio Codec	PCMU (G.711u), PCMA (G.711a), G.722
Broadcasting Mode	SIP Broadcasting Mode: PCMU/PCMA/G.722, P2P/SIP Server Connection Multicast Broadcasting Mode: PCMU/PCMA/G.722 Auto codec recognition, 20 ports VMS Broadcasting Mode: ONVIF Audio Backchannel, PCMU Internal Message Broadcasting Mode Note: Each broadcast mode can be assigned an order of priority using the Priority Setting function.
Internal Messages	Max. 20 messages (Max. recording capacity: 80 MB) Supported file formats WAV file: 8/16/44.1/48 kHz sampling frequency, 8/16 bit, monaural/stereo MP3 file: 32/44.1/48 kHz sampling frequency, 64 - 320 kbps, CBR/VBR, monaural/stereo Repeat playback: Playcount (1-10 times), Duration (5-3600 sec) or Timer (from Start time to End time) Interval time: 0 - 60 sec, Delay time: 0 - 30 sec Trigger: Control Input or Remote API (HTTP)
Network I/F	100BASE-TX, Auto MDI/MDI-X, RJ45 connector
Network Protocol	TCP/IP, UDP, HTTP, RTP, RTSP, RTCP, ARP, ICMP, IGMPv3, NTP, SIP (RFC3261)
Control Input	2 channels, no-voltage make contact inputs, open voltage: 5 V DC, short-circuit current: 2 mA or less, removable terminal block (6 pins)
Control Output	1 channel, open collector output, withstand voltage: 30 V DC, control current: 10 mA or less, removable terminal block (6 pins)
Indicator	STATUS (orange), LINK/ACT (green)
Clock Accuracy	±13 seconds per month
Time Adjustment	Manual time setting, Time adjustment by NTP server
Power Outage Protection Period	24 hours (RTC time retention, at 40 °C (104 °F))
Dimensions for Fixing Hole	Mounting hole: φ200 ±2 mm (7.87" ±0.08") Ceiling thickness: 5 - 25 mm (0.2" - 0.98")
Speaker Mounting Method	Spring clamp
Operating Temperature	0 °C to +50 °C (32 °F to 122 °F)
Operating Humidity	90 %RH or less (no condensation)
Finish	Frame: Steel plate, white (RAL 9016 equivalent), paint Grill: Steel net, white (RAL 9016 equivalent), paint
Dimensions	φ230 x 89 (D) mm (9.06" x 3.5")
Weight	880 g (1.94 lb)
Accessory	Pattern paper ...1, Removable terminal plug (6 pins, preinstalled on the unit) ...1

NOTE: Please do not install the product near heat insulation material, or cover the product with heat insulation or acoustic absorbing materials to prevent fire risk. Please do not install the product in damp or wet locations or areas with high humidity (condensing) as it may cause damage to the product.
*ONVIF is a registered trademark of ONVIF Inc.

IP-A1SC15   IP Horn Speaker

- 124dB (PoE+ powered) with IP66 rating for outdoor installations
- Receive SIP audio, ONVIF Audio Backchannel and Multicast
- Local broadcast using internal audio files
- 8W (PoE)/15W(PoE+) built-in amplifier, 1 LINE audio output
- 2 control inputs and 1 control output
- HTTP commands (receive)
- Audio file storage (20 files, total 80MB, WAV/MP3)
- Playback programs (Repeat, Weekly Timer)
- PoE/PoE+ powered

Specifications

Specifications	IP-A1SC15
Power Source	PoE+ (IEEE802.3at Class 4), PoE (IEEE802.3af Class 3)
Power Consumption	22 W (at PoE+ powered, rated output), 13 W (at PoE powered, rated output), 5 W (IEC62368-1)
Amplifier Rated Output	15 W (at PoE+ powered), 8 W (at PoE powered)
Sensitivity	112 dB (1 W, 1 m) (500 Hz - 2.5 kHz, peak level)
Maximum Sound Pressure Level	124 dB (at PoE+ powered, 15 W, 1 m) (500 Hz - 2.5 kHz, peak level) 121 dB (at PoE powered, 8 W, 1 m) (500 Hz - 2.5 kHz, peak level)
Frequency Response	280 Hz - 12.5 kHz
Audio Codec	PCMU (G.711u), PCMA (G.711a), G.722
Broadcasting Mode	SIP Broadcasting Mode: PCMU/PCMA/G.722 Multicast Broadcasting Mode: PCMU/PCMA/G.722, Max. 20 ports VMS Broadcasting Mode: Audio Backchannel, PCMU Internal Message Broadcasting Mode Note: Each broadcast mode can be assigned an order of priority using the Priority Setting function.
Internal Messages	Max. 20 messages (Max. recording capacity: 80 MB) Supported file formats WAV file: 8/16/44.1/48 kHz sampling frequency, 8/16 bit, monaural/stereo MP3 file: 32/44.1/48 kHz sampling frequency, 64 - 320 kbps, CBR/VBR, monaural/stereo Repeat playback: Playcount(1-10 times), Duration (5-3600 sec) or Timer (from Start time to End time) Interval time: 0 - 60 sec, Delay time: 0 - 30 sec Trigger: Control Input or Remote API (HTTP)
Network I/F	100BASE-TX, MDI/MDI-X, RJ-45
Network Protocol	TCP/IP, UDP, HTTP, RTP, RTSP, ARP, ICMP, IGMPv3, NTP, SIP (RFC3261)
Control Input	2 channels, no-voltage make contact inputs, open voltage: 5 V DC, short-circuit current: 2 mA or less, removable terminal block (3 pins)
Control Output	1 channel, open collector output, withstand voltage: 30 V DC, control current: 10 mA or less, removable terminal block (3 pins)
Indicator	LAN LINK/ACT (green), STATUS (orange)
Dust/Water Protection	IP66
Operating Temperature	-30 °C to +55 °C (-22 °F to +131 °F)
Operating Humidity	90 %RH or less (no condensation)
Finish	Horn flare and body: Aluminum, off-white (RAL 9010 equivalent), paint Reflector horn: ABS resin, off-white (RAL 9010 equivalent) Rear cover: PC resin, off-white (RAL 9010 equivalent), paint Bracket, screws and bolts: Stainless steel
Dimensions	222 (W) x 211 (H) x 276 (D) mm (8.74" x 8.31" x 10.87")
Weight	1.4 kg (3.09 lb)
Accessory	Rear cover…1, Removable terminal plug (3 pins)…2
Option	Speaker mount bracket: SP-131, SP-201, SP-301 Pole band: YS-60B

Note: Take special care to avoid mounting this speaker directly to structures (such as ski lift towers) that generate large amounts of vibration. Also, do not use this speaker in environments where it may be exposed to oil or other chemicals, as mounting parts could rapidly deteriorate, possibly resulting in personal in jury or other accidents due to the speaker falling.