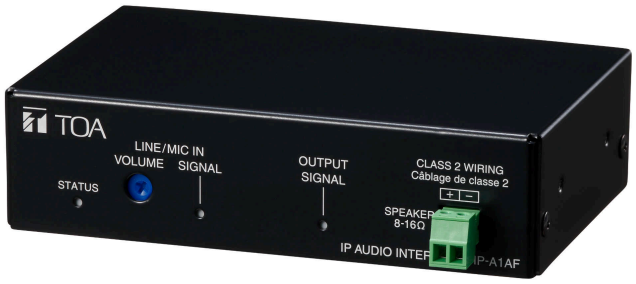
IP-A1AF

IP AUDIO INTERFACE



The IP-A1AF is a PoE-powered Audio Interface that decodes IP audio streams into analog audio signals. It equipped itself with a built-in 15 W amplifier to drive low-impedance speaker(s) and internal storage, onto which can store up to 20 audio files. Analog audio and control outputs can also be connected to an external amplifier. It supports SIP protocol for integrations with VoIP devices, and its 20 multicast ports are available for group paging operations. It can also receive broadcasts from Video Management Systems using ONVIF. Device control and further system integration is accessed through remote API (HTTP/HTTPS) or local control inputs.

Key features

- Receive SIP audio, ONVIF 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/HTTPS commands (receive)
- Audio file storage (20 files, total 80MB, WAV/MP3)
- Playback programs (Repeat, Weekly Timer)
- PoE/PoE+ powered

Specifications

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	Opus, PCMU (G.711u), PCMA (G.711a), G.722
Audio Delay Time	Min. 100 ms (*1)
Broadcasting Mode	SIP Broadcasting/SIP calling Mode: Opus/PCMU/PCMA/G.722, P2P/SIP Server Connection Multicast Broadcasting Mode: Opus/PCMU/PCMA/G.722 Auto codec recognition, 20 ports VMS Broadcasting Mode: 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.
Sound Source Files	Max. 20 files (File storage capacity: 80 MB total) 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/HTTPS)
Chime	Pre and post chime tones (applied for Incoming SIP Call and internal audio file broadcast) Preset chime tone x 5, editable tone x 2
Network I/F	100BASE-TX, Auto MDI/MDI-X, RJ45 connector
Network Protocol	IP, TCP, UDP, TLS, HTTP, HTTPS, RTP, SRTP, RTSP, RTCP, SRTCP, ARP, ICMP, IGMPv3, NTP, SIP (RFC3261), SIPS, SIP-TLS, DHCP
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: 150 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)
Feature Expansion	Device Connectivity Cloud Service (*3)
Operating Temperature	-30 °C to +55 °C (-22 °F to 131 °F)
Operating Humidity	90 %RH or less (no condensation)

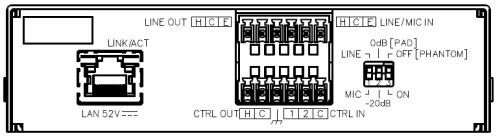
Finish	Front case: Surface-treated steel plate, black, paint Rear chassis: Surface-treated steel plate
Dimensions	126 (W) x 33 (H) x 80 (D) mm (4.96" x 1.3" x 3.15") (excluding projection)
Weight	390 g (0.86 lb)
Included Accessories	Removable terminal plug (6 pins, preinstalled on the unit) x 2, Removable terminal plug (2 pins, preinstalled on the unit) x 1, Rubber feet x 4, Mounting screw (M3 x 6) x 4

(*1) When the model of local broadcasting is set to Normal, assume audio delay time. Audio input can be output with no delay when the mode of local broadcasting is set to No Delay.
(*2) 0 dB = 1 V
*ONVIF is a registered trademark of ONVIF Inc.
*These specifications apply to the firmware version 3.6.0 or later. For the latest specifications and firmware, please refer to the TOA DATA Library (https://www.toa-products.com/international/).
(*3) Available only in Japan. A separate service contract is required.

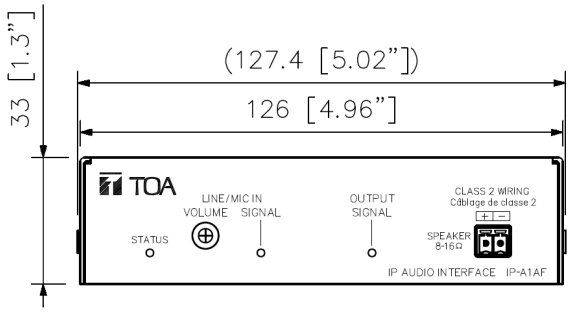
PC requirements

PC Requirements	
Display	1920 x 1080 resolution or greater
OS	Windows 10 Pro (64 bit edition) Windows 10 Home (64 bit editon) Windows 11 Pro Windows 11 Home
Web Browser	Google Chrome Microsoft Edge

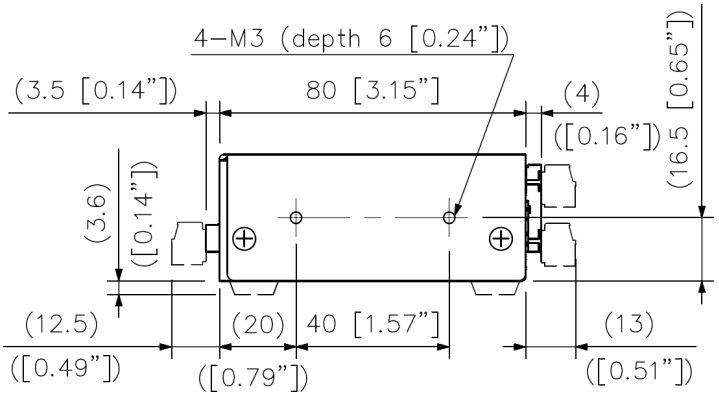
Dimensions



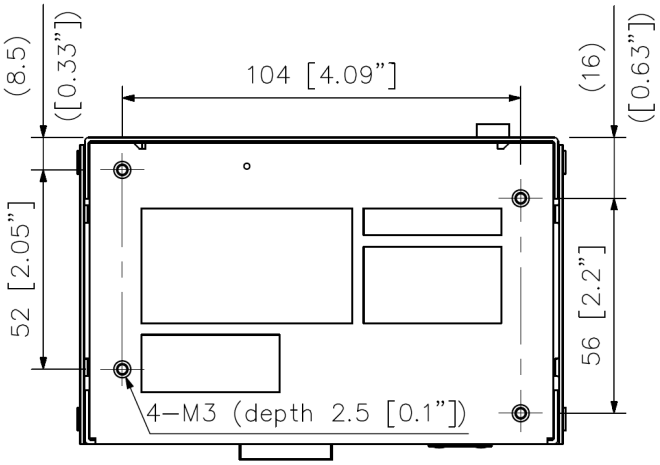
Rear View



Front View



Side View
(with accessories)



Top View

UNIT: mm

Note: Numerical values in parentheses are for reference only.

A&E specifications

The IP audio interface shall be able to convert analogue and digital audio streams from SIP, ONVIF or Multicast for example, into analogue signals for the purpose of broadcasting public announcements. Broadcast shall be able to be controlled using remote HTTP control API or control inputs from connected external equipment and local inputs.The IP audio interface shall have an internal 15 watt Class D type amplifier for the purpose of driving external 8 Ω low impedance loudspeakers and the system shall be able to be expanded by utilizing audio and control outputs to an external amplifier and loudspeakers.The IP audio interface shall offer priority status among the various broadcast modes that will override and interrupt a lower priority broadcast. The IP audio interface shall have an electronically balanced 10k Ω audio input with LINE/MIC selection and a 600 Ω electronically balanced output. The audio input shall have an independent volume control on the front panel that can only be adjusted by using a screwdriver. Twenty multi-cast ports shall be available for group paging and shall have the facility of storing up to twenty audio files with the code of WAV or MP3 format internally. The device shall have the flexibility of broadcasting these files at predetermined times by setting the start and end time along with the day(s) of the week, or repeated when necessary, triggered from a control input or from remote API (HTTP). The IP audio interface shall be capable of being muted by TOA IP-A1PG Paging Gateway as the highest priority operation. Remote settings shall be achieved by connection to a Windows® based personal computer and by using the PC's browser, operational settings, functions and system status can be established.Power source shall be supplied from a PoE+ switching hub.
The IP audio interface shall be housed in black treated steel plate and its dimension shall be 126 (W) x 33 (H) x 80 (D) mm (4.96" x 1.3" x 3.15") weighing 390 g (0.86 lbs).
Manufacturer: TOA Corporation
Model: IP-A1AF