

AXIS C1310-E Network Horn Speaker

Outdoor speaker for clear long-range speech

AXIS C1310-E Network Horn Speaker is perfect for outdoor environments in most climates. It allows users to remotely warn off intruders before they commit a crime, to deliver instructions during an emergency or to make general voice messages. Built-in memory supports pre-recorded messages, or security personal can respond to notifications with live speak. Digital signal processing (DSP) ensures clear sound. Open standards support easy integration with network video, access control, analytics, and VoIP (supporting SIP). AXIS C1310-E is a standalone unit that can be placed almost anywhere, which supports a flexible, scalable and cost-effective approach to system design.

- > All-in-one speaker system
- > Connects to standard network
- > Simple installation with PoE
- > Remote health testing
- > Two input/outputs (GPIO)



T10139377/EN/M2.2/2002 www.axis.com

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332	
Audio Audio streaming	One-way/two-way ^a (mono)
	<i>n</i>
Audio compression	AAC LC 8/16/32/48 kHz, G.711 PCM 8 kHz, G.726 ADPCM 8 kHz, Axis μ -law 16 kHz, WAV,
	MP3 in mono/stereo from 64 kbps to 320 kbps.
	Constant and variable bit rate. Sampling rate from 8 kHz up to 48 kHz.
Audio	Built-in microphone (can be disabled mechanically)
input/output	
Built-in	50 Hz - 12 kHz
microphone specification	
Speaker	
Max sound	>121 dB
pressure level	
Frequency	280 Hz - 12.5 kHz
response	70° hazizantal by 100° vartical (at 2 kHz)
	70° horizontal by 100° vertical (at 2 kHz)
Amplifier Amplifier	Built-in 7 W Class D amplifier
description	built-iii / W Class D ampinici
Network	
Security	Password protection, IP address filtering, HTTPSb encryption,
	IEEE 802.1X ^b network access control, Digest authentication, User access log
Supported	IPv4/v6, HTTP, HTTPS ^b , SIP, SSL/TLS ^b , QoS Layer 3 DiffServ, FTP,
protocols	CIFS/SMB, SMTP, Bonjour, UPnPTM, SNMP v1/v2c/v3 (MIB-II),
C	DNS, DynDNS, NTP, TCP, UDP, IGMP, ICMP, DHCP, ARP, SOCKS, SSH
System integra	
Application Programming	Open API for software integration, including VAPIX®, AXIS Video Hosting System (AVHS) with One-click Connection,
Interface	AXIS Camera Application Platform (ACAP).
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General	
Casing	Impact-resistant aluminum, IP66-, IP67-, NEMA 4X-rated, and MIL-STD-810G 509.5-rated.
Memory	256 MB RAM, 512 MB Flash
Power	Power over Ethernet (PoE) IEEE 802.3af/802.3at Type 1 Class 3 (max. 12.95 W)
Connectors	RJ45 10BASE-T/100BASE-TX PoE I/O: 4-pin 2.5 mm terminal block for one input and one output
Operating conditions	-40°C to 60 °C (-40 °F to 140 °F) Humidity 10-100% RH (condensing)
Approvals	EMC EN 55032 Class B, EN 50121-4, IEC 62236-4, EN 55024, EN 61000-6-1, EN 61000-6-2, FCC Part 15 Subpart B Class B, ICES-3(B)/NMB-3(B), VCCI Class B, RCM AS/NZS CISPR 32 Class B, KC KN32 Class B, KC KN35 Safety IEC/EN/UL 62368-1, IEC/EN/UL 60950-22 Environment IEC/EN 60529 IP67, IEC 60068-2-1, IEC 60068-2-2, IEC 60068-2-14, IEC 60068-2-27, IEC 60068-2-78, IEC/EN 60529 IP66, NEMA 250 Type 4X, MIL-STD-810G 509.5
Dimensions	Without bracket: $164 \times 225 \times 250$ mm (6 $1/2 \times 8$ $7/8 \times 9$ $7/8$ in.) With bracket: $164 \times 225 \times 305$ mm (6 $1/2 \times 8$ $7/8 \times 12$ in.)
Weight	1.3 kg (2.9 lb.)
Included accessories	Installation Guide, AVHS Authentication Key, AXIS Camera Station license key, AXIS Connector Guard A, Cable shoe
Optional accessories	AXIS T91B47 Pole Mount, AXIS T91F67 Pole Mount, Cable Gland M20x1.5, RJ45, Cable Gland A M20, AXIS Power over Ethernet Midspans, T94R01B Corner Bracket, T94P01B Corner Bracket, T94S01P Conduit Back Box
Video management software	AXIS Camera Station, Video management software from Axis' Application Development Partners available on axis.com/techsup/software
Languages	English, German, French, Spanish, Italian
Warranty	Axis 3-year warranty and AXIS Extended Warranty option, see axis.com/warranty

a. This product supports two-way audio for sending audio to the speaker and receiving audio from the microphone. The product does not support two-way communication for conversations with speaker operators.
b. This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit. (www.openssl.org), and cryptographic software written by Eric Young (eay@cryptsoft.com).
c. Audio synchronization with IPv4 only.

Environmental responsibility:

axis.com/environmental-responsibility

