

NS300YA Speaker

The iVSEC NS300YA is a compact, ceiling-mounted SIP speaker designed for indoor paging and broadcast applications. Built for commercial environments such as offices, schools, and retail spaces, it can operate as a standalone audio device or integrate with broader communication and security systems.
With a 10W speaker and up to 105dB sound pressure, the iVSEC NS300YA delivers clear and consistent audio coverage across indoor areas. It supports HD codecs , along with MP3 and WAV playback, while a built-in microphone enables two-way intercom and monitoring functionality.
The iVSEC NS300YA supports SIP protocol, multicast paging, and flexible system integration via APIs and I/O interfaces. Features such as scheduled broadcasting, emergency alerts, and remote management make it a versatile solution, while PoE support and simple ceiling installation ensure efficient deployment in modern IP-based audio systems.



Specifications

Colour	White
Mounting	Ceiling
SIP Accounts	2
Multicast Partitions	20
Customised Audio Support	Yes
Auto Answer	Yes
HTTP/Webhook Support	Yes
DTMF Support	Yes
Web GUI	Yes
Storage Capacity (Mb)	500
ONVIF	Yes
Protocols	DHCP, FTP/TFTP, HTTP/HTTPS, IPv6, L2TP, OpenVPN, PPPoE, RTP/RTCP/SRTP, SIP2.0, SNTP, STUN, TR-069
Encoding	G.722, Opus
Language	English
Auto-Provisioning	Yes
Outdoor Use	No
Accessories	n/a
Frequency Response	25Hz-20KHz
Speaker Sensitivity	90± 3dB
Impedance	8Ω
Max SPL	100dB
Speaker Power Max	6W (PoE), 10W (PoE+), 10W (12-24V DC/2A)
Additional Speaker Support	No
Microphone	Yes
Microphone Sensitivity	-38± 3dB
Self Testing Feature	Yes
Ethernet	Yes (10/100 Mbps)
Power Over Ethernet	Yes
Power Supply	12-24V DC/2A, PoE, PoE+
Power Consumption	10W max
Connectors	2.1mm DC input, 1 x Relay input, 1 x Relay output, 1 x RJ45 port
Operating Environment	-25℃ - 55℃ (10-95% RH)
Storage Environment	-30℃ - 60℃
In The Box	1 x SIP Speaker, 1 x Quick Guide
Product Dimensions (H x W x L) mm	90 x 200 x 200
Packaging Dimensions (H x W x L) mm	105 x 200 x 200
Weight (Net) kg	14.00
Weight (Gross) kg	15.44
Warranty	2 Years Parts

https://ivsecurity.com.au/product/ns300ya/

Updated 06/11/2026

ALGO

8301 IP PAGING ADAPTER

IP Paging Adapter for Traditional Analog Amplifiers

Connect the 8301 IP Paging Adapter to legacy amplifiers to bridge analog paging systems to IP environments. The 8301 offers multicast and scheduling capabilities, making it a valuable resource in hybrid and IP environments.



The 8301 IP Paging Adapter is a SIP endpoint that integrates existing traditional paging infrastructure into a VoIP environment. Algo's paging adapter emulates a page port similar to what is found on a legacy PBX to connect directly to an analog amplifier, offering a simple and easy interface to a VoIP phone system as a SIP extension and multicast endpoint.

As a SIP-compliant device, the 8301 is compatible with most leading UC systems. While often used to bridge IP to traditional amplifiers, the 8301 also scales IP environments, acting as a multicast sender to other Algo IP endpoints grouped into zones. With wideband audio and an isolated balanced line output, the 8301 can deliver clear, crisp audio to play over both IP and analog speakers.

A common use for the 8301 is as a scheduler, which can function in both traditional and IP environments. Users can configure the 8301 to set timers for bells, automated announcements, or any other tone or usage. With 1 GB of internal memory, any selection of sound can be played by uploading the desired audio file.

The 8301 is easily managed directly via the web interface. Paired with an Algo device management tool, the 8301 can be monitored and managed in an Algo ecosystem along with any number or types of devices over a LAN or on the cloud.

FEATURES

- Isolated and balanced line output to eliminate hum / noise to amplifier
- Wideband HD audio (G.722 voice codec) for high intelligibility speech
- Secure SIP using TLS and SRTP
- Configurable via user-friendly web interface or auto-provisioning
- Customizable audio files for emergency notification and alerting
- Configure up to 50 page zones
- PoE powered (IEEE 802.3af)
- Dedicated audio/music input (3.5 mm jack)

APPLICATIONS

- Ensure seamless connectivity between IP and legacy infrastructure.
- Deliver clear, intelligible voice paging solutions to any size building or enterprise.
- Easily connect the 8301 to an emergency communication system to deliver highly noticeable notifications and alerts.
- Leverage multicast to extend communication reach to IP speakers while broadcasting to analog speakers via the amplifier connected to the 8301.
- Schedule bells, tones, and automated announcements for schools, retail, and healthcare facilities.

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8301 IP PAGING ADAPTER

POWER	
POWER SOURCE	PoE (IEEE 802.3af Class 0) 48 V, 15.4 W
MAX POWER (AT DEVICE)	4 W
IDLE POWER	2 W
SIP	
SIP EXTENSIONS	• 50 Page (Capable of talkback when appropriate speaker and microphone connected) • 10 Emergency Alert • 10 Ring
SIP SIGNALLING/ TRANSPORT PROTOCOLS	UDP, TCP, RTP
SECURITY	TLS, SRTP, Mutual Authentication
AUDIO CODECS	G.711 u-law, G.711 A-law, G.722 Wideband
MULTICAST	
RTP MULTICAST	Send and Receive 50 Zones
AUDIO CODECS	G.711 u-law, G.711 A-law, G.722 Wideband, Opus Fullband
COMPATIBILITY	
THIRD PARTIES	• RESTful API • Poly Group Page™ • Singlewire InformaCast® (additional license required) • Intrado OneAlert® • Microsoft Teams®
CONFIGURATION & PROVISIONING	
CONFIGURATION	Web interface or provisioning server
WEB INTERFACE	HTTP, HTTPS
PROVISIONING	TFTP, FTP, HTTP, HTTPS DHCP Options 66, 160, 160 Reboot via SIP 'check-sync'
ZTP	Supported
SUPERVISION	Compatible with Algo Device Mangement Platform (ADMP), Algo 8300 IP Controller, any third-party SNMP monitoring software, and RTCP
NETWORKING	
NETWORKING	IPv4, IPv6, DHCP, VLAN, MDNS
LINK LAYER	LLDP, CDP
SECURITY	IEEE 802.1X
QOS	DSCP
NAT	STUN, TURN, CRLF Keep Alive, SIP Outbound
ADDRESS RESOLUTION	DNS, SRV Record
REDUNDANCY	Secondary and tertiary SIP server
TIME	NTP Server (up to four)
AUDIO	
MICROPHONE	Connection available for external microphone
AUDIO MEMORY & FORMAT	1 GByte audio storage for WAV or MP3 files
AUDIO CONTROLS	Volume, AGC, Latency, LF Cut
ANTI-FEEDBACK DELAY	Cache to memory and release
AUDIO DELAY	Programmable 1-1000 ms synchronization delay
INPUT/OUTPUT	
RELAY INPUT	Normally open or normally closed dry contact with supervision. Compatible with Algo 1202 Call Button, 1203 Call Switch, 1204 Volume Control, 1205 Audio Interface, or EOL resistor termination.
RELAY OUTPUT	Max 30 V 50 mA (normally open)
AUX IN	3.5 mm jack for analog line level input for music input. Non-isolated.
AUX OUT	3.5 mm jack for analog line level output for compatible PC speakers or headset (with optional mic input from headset). Non isolated.
LINE IN MINI-XLR	Balanced and isolated audio (page or music) input can be configured for pass-through to Line Out (when paging is idle), or for broadcast via multicast.
LINE OUT MINI-XLR	Balanced and isolated audio output to external amplifier. Locking Mini-XLR female to standard XLR male cable available. Output level defined using web interface.
TERMINAL BLOCK LINE IN	Balanced and isolated wire pair input parallel to Mini-XLR line in (polarity independent).
TERMINAL BLOCK LINE OUT	Balanced and isolated wire pair output to external amplifier. Parallel to Mini-XLR line out (polarity independent).

RELAY INPUT CURRENT DRAW DETECTION			
THRESHOLD	ACTIVE	IDLE	TAMPER
NORMALLY OPEN	> 4 mA	< 4 mA	N/A
NORMALLY OPEN WITH SUPERVISION	> 20 mA	4-20 mA	< 4 mA
NORMALLY CLOSED	< 4 mA	> 4 mA	N/A
NORMALLY CLOSED WITH SUPERVISION	4-20 mA	> 20 mA	< 4 mA
NOMINAL 12 V SOURCE, CURRENT LIMITED TO 40 MA	Typical supervision resistor value = 1 k Ohm		

ENVIRONMENTAL & MECHANICAL	
ENVIRONMENTAL	0 to +40 deg C (32 to 104 deg F); 10-95% RH non-condensing. Dry indoor locations only.
DIMENSIONS (PRODUCT)	7.1" x 4.3" x 1.3" (18.0 cm x 10.9 cm x 3.2 cm)
WEIGHT (PRODUCT)	0.9 lbs (0.4 kg)
WEIGHT (SHIPPING)	1.5 lbs (0.7 kg)
MOUNTING	Snap mounting bracket included
COMPLIANCE	

RoHS, CE, FCC Class A, CISPR 22 Class A, CISPR 24, CSA/UL (USA & Canada)

FRONT



BACK



ORDERING CODES:

8301 IP Paging Adapter and Scheduler
8301-IC IP Paging Adapter and Scheduler InformaCast

COMPATIBLE ACCESSORIES:

1202 Customer/Emergency Assistance Button
1203 Wall Switch Single Gang Blue LED



1202



1203



Balanced Audio Cables:
2504 Output XLR-Mini Female to XLR Male
2505 Input XLR-Mini Male to XLR Female
Other cables available on request

INFO@ALGOSOLUTIONS.COM
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GRANDSTREAM
CONNECTING THE WORLD



Unified Communication & Collaboration Solution

UCM6300 series

The UCM6300 series allows businesses to build powerful and scalable unified communication and collaboration solutions. This series of IP PBXs provide a platform that unifies all business communication on one centralized network, including voice, video calling, video conferencing, video surveillance, web meetings, data, analytics, mobility, facility access, intercoms and more. The UCM6300 series supports up to 3000 users and includes a built-in web meetings and video conferencing solution that allows employees to connect from the desktop, mobile, GVC series devices and IP phones. It can be paired with the UCM6300 ecosystem to offer a hybrid platform that combines the control of an on-premise IP PBX with the remote access of a cloud solution. The UCM6300 ecosystem consists of the Wave app for web and mobile, which provides a hub for collaborting remotely, and UCM RemoteConnect, a cloud NAT traversal service for ensuring secure remote connections. The UCM6300 series also offers cloud setup and management through GDMS and an API for integration with third-party platforms. By offering a high-end unified communications and collaboration solution packed with a suite of mobility, security, meeting and collaboration tools, the UCM6300 series provides a powerful platform for any organization.

3000 users

Supports up to 3000 users and up to 450 concurrent calls

zero CONFIG

Zero configuration provisioning of Grandstream SIP endpoints

Built-in conferencing & meetings platform; supports desktop, Wave app, and SIP endpoints

Wave App allows communication with all UCM6300 users & solutions

API

API available for third-party integrations, including CRM and PMS platforms

Advanced security protection with secure boot, unique certificate and random default password to protect calls and accounts

Gigabit

Three Gigabit auto-sensing RJ45 network ports with integrated PoE+ and support NAT router

UCM REMOTECONNECT

Automated NAT firewall traversal service facilitates secure remote connections

Enhanced reliability with support for Hot Standby High-Availability (pending)

HD

Supports Full-Band Opus voice codec and H.264/H.263/ H.263+/H.265/VP8 video codec, jitter resilience up to 50% packet loss

Compatible with GDMS for cloud setup, management and monitoring

Based on Asterisk* version 16 open source telephony operating system

www.grandstream.com

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	UCM6301	UCM6302	UCM6304	UCM6308
Analog Telephone FXS Ports	1 RJ11 Port	2 RJ11 Ports	4 RJ11 Ports	8 RJ11 Ports
	All ports have lifeline capability in case of power outage			
PSTN Line FXO Ports	1 RJ11 Port	2 RJ11 Ports	4 RJ11 Ports	8 RJ11 Ports
	All ports have lifeline capability in case of power outage			
Network Interfaces	Three self-adaptive Gigabit ports (switched, routed or dual mode) with PoE+			
NAT Router	Yes (supports router mode and switch mode)			
Peripheral Ports	1*USB 3.0, 1*SD card interface	1*USB 2.0, 1*USB 3.0, 1*SD card interface	2*USB 3.0, 1*SD card interface	
LED Indicators	None		Power 1/2, FXS, FXO, LAN, WAN, Heartbeat	
LCD Display	320x240 color LCD with touch screen for Shortcut Keys and Scroll Bar		128x32 dot matrix graphic LCD with DOWN and OK buttons	
Reset Switch	Yes, long press for factory reset and short press for reboot			
Voice-over-Packet Capabilities	LEC with NLP Packetized Voice Protocol Unit, 128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711, NetEQ, FEC 2.0, jitter resilience up to 50% audio packet loss			
Voice and Fax Codecs	Opus, G.711 A-law/U-law, G.722, G722.1 G722.1C, G.723.1 5.3K/6.3K, G.726-32, G.729A/B, iLBC, GSM; T.38			
Video Codecs	H.264, H.263, H263+, H.265, VP8			
QoS	Layer 2 QoS (802.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS			
API	Full API available for third-party platform and application integration			
Telephony Operating System	Based on Asterisk version 16			
DTMF Methods	In-band audio, RFC2833, and SIP INFO			
Provisioning Protocol & Plug-and-Play	Mass provisioning using AES encrypted XML configuration file, auto-discovery & auto-provisioning of Grandstream IP endpoints via ZeroConfig (DHCP Option 66 multicast SIP SUBSCRIBE mDNS), eventlist between local and remote trunk			
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, STUN, SRTP, TLS, LDAP, HDLC, HDLC-ETH, PPP, Frame Relay (pending), IPv6, OpenVPN®			
Disconnect Methods	Busy/Congestion/Howl Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect			
Media Encryption	SRTP, TLS, HTTPS, SSH, 802.1X			
Universal Power Supply	Input: 100 ~ 240VAC, 50/60Hz; Output: DC 12V, 1.5A		2x DC 12V Power Jack Input: 100~240VAC, 50/60Hz;Output: DC 12V, 2A	
Dimensions	270mm(L) x 175mm(W) x 36mm(H)		485mm(L) x 187.2mm(W) x 46.2mm(H)	
Weight	Unit Weight: 715g; Package Weight: 1211g	Unit Weight: 725g; Package Weight: 1221g	Unit Weight: 2490g; Package Weight: 3260g	Unit Weight: 2550g; Package Weight: 3320g
Temperature & Humidity	Operating: 32 - 113°F / 0 ~ 45°C, Humidity 10 - 90% (non-condensing) Storage: 14 - 140°F / -10 ~ 60°C, Humidity 10 - 90% (non-condensing)			
Mounting	Wall mount & Desktop		Rack mount & Desktop	
Multi-Language Support	-Web UI: English, Simplified Chinese, Traditional Chinese, Spanish, French, Portuguese, German, Russian, Italian, Polish, Czech, Turkish -Customizable IVR/voice prompts: English, Chinese, British English, German, Spanish, Greek, French, Italian, Dutch, Polish, Portuguese, Russian, Swedish, Turkish, Hebrew, Arabic, Nederlands -Customizable language pack to support any other languages			
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT, NTT			
Polarity Reversal/Wink	Yes, with enable/disable option upon call establishment and termination			
Call Center	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/ work-load, in-queue announcement			
Customizable Auto Attendant	Up to 5 layers of IVR (Interactive Voice Response) in multiple languages			
Maximum Call Capacity	Users: 500 Concurrent calls (G.711): 75 Max concurrent SRTP calls (G.711): 50	Users: 1000 Concurrent calls (G.711): 150 Max concurrent SRTP calls (G.711): 100	Users: 2000 Concurrent calls (G.711): 300 Max concurrent SRTP calls (G.711): 200	Users: 3000 Concurrent calls (G.711): 450 Max concurrent SRTP calls (G.711): 300
Maximum Attendees of Conference Bridges	2 Video Conference rooms and up to 12 parties with 1080p, assuming 4 video feeds + 1 screen sharing (H.264 & G.711) Voice Conference: Up to 75 parties (G.711)	3 Video Conference rooms and up to 20 parties with 1080p, assuming 4 video feeds + 1 screen sharing (H.264 & G.711) Voice Conference: Up to 150 parties (G.711)	4 Video Conference rooms and up to 40 parties with 1080p, assuming 4 video feeds + 1 screen sharing (H.264 & G.711) Voice Conference: Up to 200 parties (G.711)	8 Video Conference rooms and up to 60 parties with 1080p, assuming 4 video feeds + 1 screen sharing (H.264 & G.711) Voice Conference: Up to 300 parties (G.711)
Wave Mobile App	Allows Android & iOS users to join UCM-hosted meetings & communicate with other users/solutions registered to the UCM6300			
Call Features	Call park, call forward, call transfer, call waiting, caller ID, call record, call history, ringtone, IVR, music on hold, call routes, DID, DOD, DND, DISA, ring group, ring simultaneously, time schedule, PIN groups, call queue, pickup group, paging/intercom, voicemail, call wakeup, SCA, BLF, voicemail to email, fax to email, speed dial, call back, dial by name, emergency call, call follow me, blacklist/whitelist, voice conference, video conference, eventlist, feature codes, busy camp-on/ call completion, voice control			
Firmware Upgrade	Supported by Grandstream Device Management System (GDMS), a zero-touch cloud provisioning and management system, It provides a centralized interface to provision, manage, monitor and troubleshoot Grandstream products			
Compliance	FCC: Part 15 (CFR 47) Class B, Part 68 CE: EN 55032, EN 55035, EN 61000-3-2, EN 61000-3-3, EN 62368-1, ETSI ES 203 021, ITU-T K.21 IC: ICES-003, CS-03 Part I Issue 9 RCM: AS/NZS CISPR 32, AS/NZS 62368.1, AS/CA S002, AS/CA S003.1/.2 Power adapter: UL 60950-1 or UL 62368-1			

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