

ALGO

8305 Multi-Interface IP Paging Adapter

User Guide



MOMENTUM

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Information Notices

**Warning**

Warning indicates a potentially hazardous situation which, if not avoided, could result in death or serious injury

**Caution**

Caution indicates a potentially hazardous situation which, if not avoided, could result in minor or moderate injury and/or damage to the equipment or property

**Important**

Important indicates a key piece of updates, information, and instructions that need to be followed for correct and safe use of the device

**Note**

Note indicates useful updates, information, and instructions that should be followed

**Tips & Tricks**

Tips & Tricks indicate helpful instructions that could help you with your device

Disclaimer

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For additional information or vendor technical assistance in North America, please contact Algo's support team:

Algo Technical Support

1-604-454-3792

support@algosolutions.com

IMPORTANT WARNING AND SAFETY INFORMATION

**Important Notice**

This product is powered by a certified limited power source (LPS), Power over Ethernet (PoE); through CAT5 or CAT6 connection wiring to an IEEE 802.3af compliant network PoE switch. The product is intended for installation indoors. All wiring connections to the product must be in the same building. If the product is installed beyond the building perimeter or used in an inter-building application, the wiring connections must be protected against overvoltage/transient. Algo recommends that this product is installed by a qualified electrician.

If you are unable to understand the English language safety information, then please contact Algo by email for assistance before attempting an installation support@algosolutions.com.

**Consignes de Sécurité Importantes**

Ce produit est alimenté par une source d'alimentation limitée certifiée (alimentation par Ethernet); des câbles de catégorie 5 et 6 joignent un commutateur réseau à alimentation par Ethernet homologué IEEE 802.3af. Le produit est conçu pour être installé à l'intérieur. Tout le câblage rattaché au produit doit se trouver dans le même édifice. Si le produit est installé au-delà du périmètre de l'édifice ou utilisé pour plusieurs édifices, le câblage doit être protégé des surtensions transitoires. Algo recommande qu'un électricien qualifié se charge de l'installation de ce produit.

Si vous ne pouvez comprendre les consignes de sécurité en anglais, veuillez communiquer avec Algo par courriel avant d'entreprendre l'installation au support@algosolutions.com.

**Información de Seguridad Importante**

Este producto funciona con una fuente de alimentación limitada (Limited Power Source, LPS) certificada, Alimentación a través de Ethernet (Power over Ethernet, PoE); mediante un cable de conexión CAT5 o CAT6 a un conmutador de red con PoE en cumplimiento con IEEE 802.3af. El producto se debe instalar en lugares cerrados. Todas las conexiones cableadas al producto deben estar en el mismo edificio. Si el producto se instala fuera del perímetro del edificio o se utiliza en una aplicación en varios edificios, las conexiones cableadas se deben proteger contra sobretensión o corriente transitoria. Algo recomienda que la instalación de este producto la realice un electricista calificado.

Si usted no puede comprender la información de seguridad en inglés, comuníquese con Algo por correo electrónico para obtener asistencia antes de intentar instalarlo: support@algosolutions.com.

**Wichtige Sicherheitsinformationen**

Dieses Produkt wird durch eine zertifizierte Stromquelle mit begrenzter Leistung (LPS – Limited Power Source) betrieben. Die Stromversorgung erfolgt über Ethernet (PoE – Power over Ethernet). Dies geschieht durch eine Cat-5-Verbindung oder eine Cat-6-Verbindung zu einer IEEE 802.3af-konformen Ethernet-Netzwerkweiche. Das Produkt wurde konzipiert für die Installation innerhalb eines Gebäudes. Alle Kabelverbindungen zum Produkt müssen im selben Gebäude bestehen. Wenn das Produkt jenseits des Gebäudes oder für mehrere Gebäude genutzt wird, müssen die Kabelverbindungen vor Überspannung und Spannungssprüngen geschützt werden. Algo empfiehlt das Produkt von einem qualifizierten Elektriker installieren zu lassenv.

Sollten Sie die englischen Sicherheitsinformationen nicht verstehen, kontaktieren Sie bitte Algo per Email bevor Sie mit der Installation beginnen, um Unterstützung zu erhalten. Algo kann unter der folgenden E-Mail-Adresse erreicht werden: support@algosolutions.com.



安全须知

本产品由认证的受限电源(LPS)，以太网供电(PoE)，通过 CAT5 或 CAT6 线路联接至 IEEE 802.3af 兼容的 PoE 网络交换机供电。本产品适用于室内或建筑物周边安装。所有联接本产品的线路必须源自同一建筑物。本产品如需用于超出建筑物周边范围或跨建筑物的安装，线路联接部分必须有过压和瞬态保护。Algo 建议本产品由专业电工安装。

如果您对理解英文版安全须知有问题，安装前请通过电子邮件和 Algo 联系. support@algosolutions.com.



EMERGENCY COMMUNICATION

If used in an emergency communication application, the 8305 Multi-Interface IP Paging Adapter should be routinely tested. The Algo Device Management Platform (ADMP) or any third-party management tool supporting SNMP supervision is recommended for assurance of proper operation. Contact Algo for other methods of operational assurance.



DRY INDOOR LOCATION ONLY

The 8305 Multi-Interface IP Paging Adapter is intended for dry indoor locations only. For outdoor locations Algo offers weatherproof speakers and strobe lights.

CAT5 or CAT6 connection wiring to an IEEE 802.3af (PoE) compliant network PoE switch must not leave the building perimeter without adequate lightning protection.

No wiring connected to the 8305 Multi-Interface IP Paging Adapter may leave the building perimeter without adequate lightning protection.

1 PRODUCT OVERVIEW

1.1 Introduction

Algo's 8305 Multi-Interface IP Paging Adapter is a SIP-compliant, PoE device that enables you to integrate legacy communication systems and IP devices. Designed specifically to emulate an analog phone, the 8305 enables you to create a hybrid VoIP environment by continuing to use existing analog hardware connected to a telephone port, 8 Ω output, or line output.

The 8305 can act as a transmitter or receiver, giving you full control of your multicasting needs. Using wideband audio (G.722 voice codec), the 8305 allows you to deliver clear, crisp audio and high speech intelligibility for voice pages, tones, and alerts. The device also has calendaring functionality to set bell, announcement, or other notification schedules using audio files stored on the 8305. These can be single events or recurring events that play daily, weekly, monthly, or annually.

Compared to the Algo 8301 IP Paging Adapter & Scheduler which has the most versatility of Algo paging adapters and the Algo 8373 IP Zone Paging Adapter which was designed to eliminate the need for a legacy zone controller, the 8305 Multi-Interface IP Paging Adapter is best suited for those wanting to scale their use of a legacy communication system for paging.

1.2 Product Views

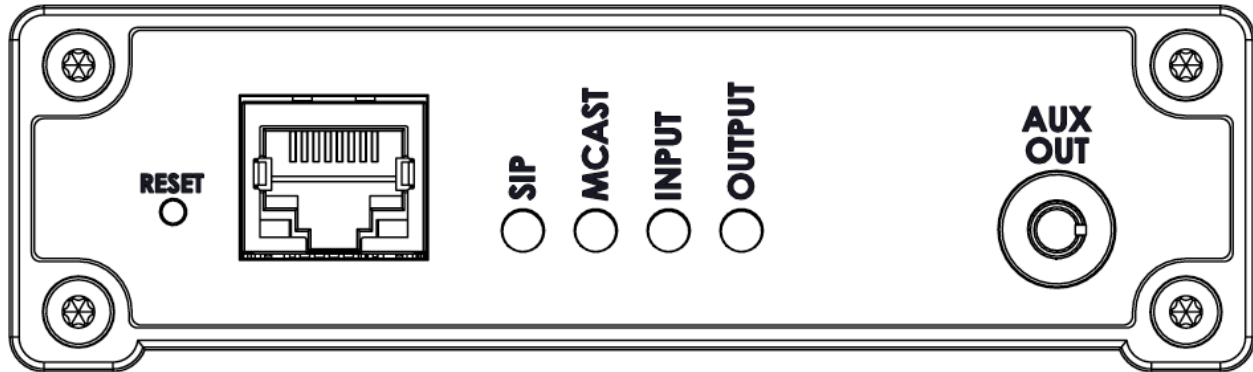


Figure 1: 8305 Multi-Interface IP Paging Adapter front faceplate.

8305 - Right Side	
RESET Button	A recessed reset button. This button is used to reset the device or play the device's IP address during setup.
RJ45 Ethernet Jack	For network connection. A cable run from the switch can be terminated to a modular jack with a connection by patch cord or terminated with an RJ45 plug. PoE (Power over Ethernet) must be 48 V 350 mA IEEE 802.3af compliant, whether provided by the network switch or injector.
RJ45 Ethernet Jack Light	There are two lights on the Ethernet jack: Amber: Turns on immediately after the Ethernet cable is first connected indicating that PoE power has been successfully applied. Once the device connects to the network, the light will turn off. Green: Turns on when Ethernet is working after a 100Mbps link has been established. Flickers to indicate activity on the port.
SIP Light (Blue)	A steady light will appear when a SIP extension is registered. The light will blink when the device is engaged in a SIP call.
MCAST Light (Blue)	A steady light will appear when the 8305 receives multicast audio as a Receiver. The light will blink when the 8305 sends multicast audio as a Transmitter.
INPUT Light (Blue)	Not currently used on this device.

OUTPUT Indicator (Blue)	Turns on when the analog output is enabled.
AUX OUT 3.5 mm Jack	Analog line level output for compatible PC speakers or headset. Non-isolated.

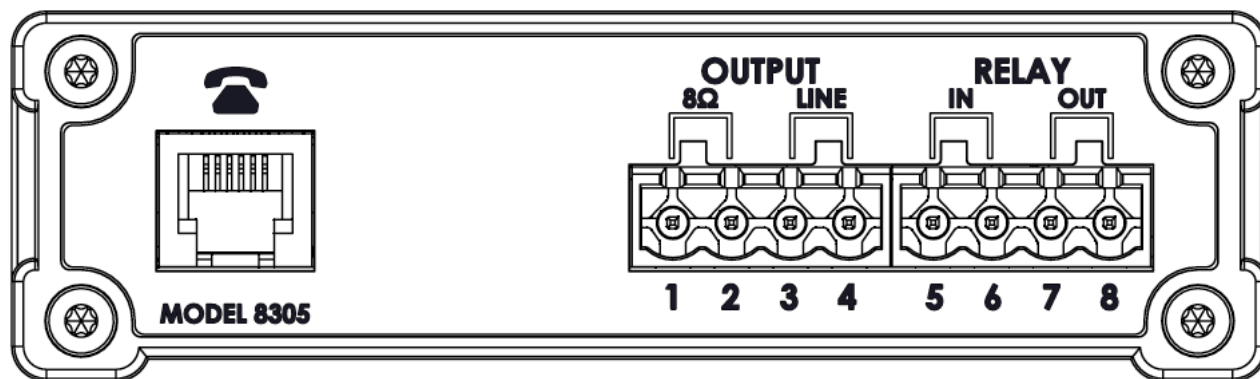


Figure 2: 8305 Multi-Interface IP Paging Adapter back faceplate.

8305 - Left Side	
Telephone Port	Emulates an analog telephone that can go on and off hook. Has built in ring detection with auto-answer. The telephone port is intended only for the connection to the FXS port on a legacy communication system and must not be connected to the PSTN.
Terminal Block 8 Ω Output (1/2)	Balanced and isolated wire pair output to connect one or many external self-amplified speakers connected in parallel with a total minimum resistance of 8 Ω. Intended use is for up to 100 nominal 2 kΩ or 1 kΩ self-amplified speakers. 8 Ω maximum output: +3dBm @ 8 Ω 2 kΩ maximum output: +1.5dB higher
Terminal Block Line Output (3/4)	Balanced and isolated wire pair. Output level defined using web interface.
Terminal Block Relay In (5/6)	Used to connect an external button or to detect a contact closure. Connection options include a normally closed switch, normally open switch, 1202 Call Button, 1203 Call Switch, 1204 Volume Control Switch, 1205 Audio Interface, or EOL resistor termination.

Terminal Block Relay Out (7/8)	By default, these terminals provide a contact closure when the 8305 IP Paging Adapter is active. Note this is a normally open relay only.
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2 SETUP AND INSTALLATION



Important

Read thoroughly to understand essential safety information before installing the product permanently.

What is Included

The following items are included with this device:

- 8305 Multi-Interface IP Paging Adapter
- Wall mount bracket and screws
- Network cable
- Two (2) pluggable terminal blocks
- Flat head screwdriver
- Quick Start Guide

2.1 Hardware Setup & Installation

Mounting Instructions

Use the supplied bracket to mount the 8305 horizontally. The following instructions can be used to install the 8305 on a 1/2" drywall:

1. Use appropriate drywall anchors for #8 screws and pre-drill per anchor manufacturer's instructions.
2. Insert 4 anchors into the wall, and then attach the bracket to the wall anchors using #8 screws.
3. Snap the 8305 into the bracket.



Figure 3: 8305 wall mount.

Wiring Connections

1. Connect the 8305 Multi-Interface IP Paging Adapter to an IEEE 802.3af compliant PoE network switch or PoE injector. Blue lights on the front will turn on.
2. Wait for the blue lights to turn off (about 60 seconds). Boot-up is complete when they turn off.
3. Press the recessed reset switch (RST) to play the IP address over the analog outputs. A headset can be connected to the green AUX output port. You can also find the IP address by downloading the Algo Network Device Locator or a third-party network scanner to find Algo devices on your network. Algo device MAC addresses start with 00:22:ee. You will need this IP address to configure the 8305 using the web interface.
4. Connect your desired devices to the telephone port, line output, or 8 Ω output.
 - a. **Telephone Port** – Connect to the telephone port on a legacy communication system. This port on the 8305 emulates an analog phone. The telephone port on the legacy device may be labeled as an FXS port.
 - b. **Line Output** – Connect directly to the telephone input on the legacy communication system with an input impedance between 600 Ohm and 10 kOhm. The output level can be adjusted to match the device's input volume and other audio specifications in the web interface under **Basic Settings** → **Features**. If required, the optional dry contact closure can be used to activate the legacy communication system.
 - c. **8 Ω Output** – Connect one or many self-amplified speakers. If many speakers are connected in parallel, the resulting effective impedance must not be less than 8 Ω . Intended use is for nominal 2 k Ω or 1 k Ω self-amplified speakers.

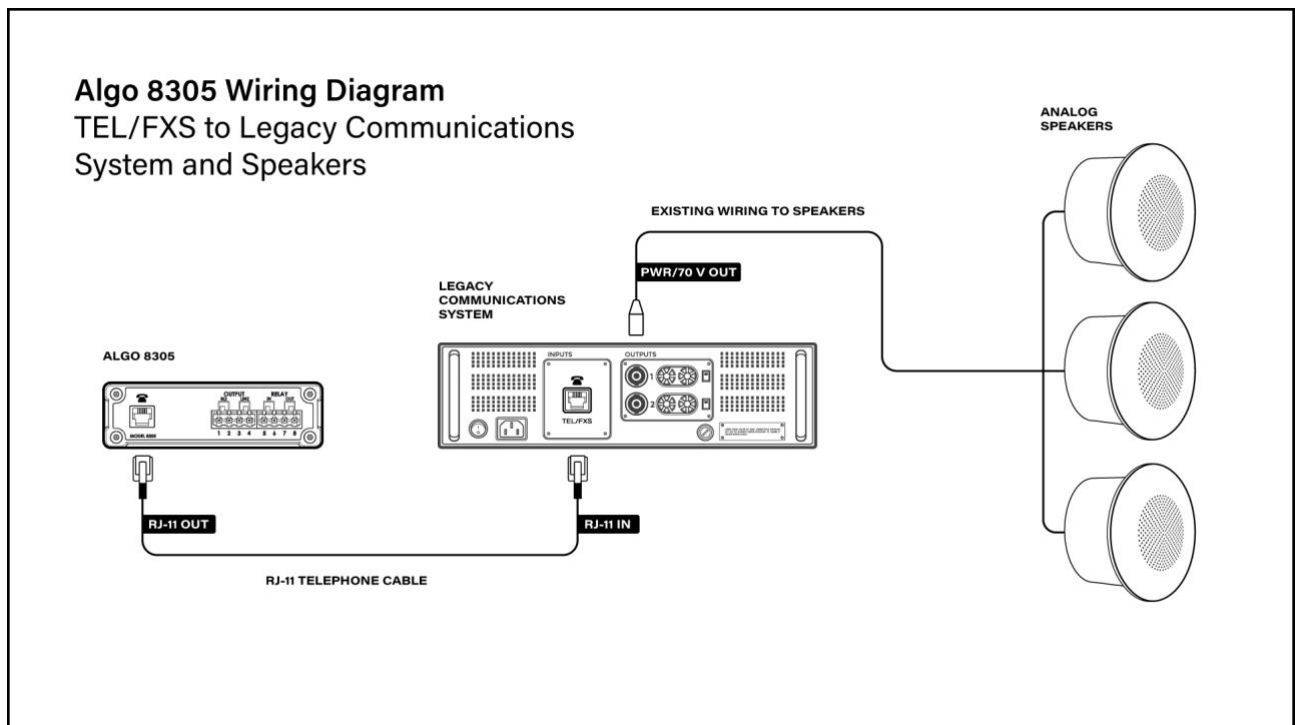


Figure 4. Wiring diagram example for the 8305.

2.2 Accessing the Web Interface

After you enter the IP address for your device into your browser, the web interface will appear.

You must log in to view device settings. The default password is *algo*. This password can be changed under **Advanced Settings** → **Admin** after logging in. Changing the default password is highly recommended if the device is directly connected to a public network.



Important

The **Save** button must be clicked to apply any changes made in the web interface.

The screenshot shows the web interface for the ALGO 8305 Multi-Interface IP Paging Adapter. The header features the ALGO logo and the product name. The main content area is titled 'Welcome to the Algo 8305 Multi-Interface IP Paging Adapter' and provides instructions for setting up the device. It includes four steps: 1. Configure your Multi-Interface IP Paging Adapter (Log in with default password and use Basic Settings), 2. Check network settings (Optional) (Use the Network page under Advanced Settings), 3. Secure your Multi-Interface IP Paging Adapter (Optional) (Use the Admin page under Advanced Settings to change the administrator password, with a warning about public network security), and 4. Register your Multi-Interface IP Paging Adapter (Optional) (Please register your product using the link below: <http://www.algosolutions.com/register>). Below the steps is a 'Login' section with a password field (default: algo) and a 'Login' button. At the bottom, there is a 'Status' section.

Figure 5: Welcome page for the 8305 web interface.

2.2.1 Web Interface Setup

1. Enter the IP address into a web browser to access the 8305 Multi-Interface IP Paging Adapter web interface.
2. Log in using the default password: *algo*.
3. Navigate to **Basic Settings** → **SIP** and enter the IP address or the domain name for the SIP server (provided by your IT team or hosted provider) into **SIP Domain (Proxy Server)**.
4. Enter the Page and/or Ring credentials **Extension**, **Authentication ID**, and **Authentication Password** (provided by your IT team or hosted provider). If you are not using an extension, leave the fields blank. Note that some SIP servers may say Username instead of Authentication ID.

5. Verify the extension is properly registered with the SIP server in the Status tab. Ensure the SIP registration says “Successful”.
6. Test the adapter by dialing the registered SIP extension from an IP phone connected to your network.

2.2.2 Check Device Status

By default, the **Status** page is available with and without a login. You may make the Status page only available to logged-in users via **Advanced Settings** → **Admin** → **General** → **Show Status Section on Status Page when Logged Out**

The **Status** page contains information such as:

- Device Name
- SIP Registration
- Call Status
- Proxy Status
- Provisioning Status
- MAC
- IP Address
- Date/Time
- Next Scheduled Event
- Multicast Mode
- Volume
- Relay Input Status
- InformaCast License
- ADMP Cloud Monitoring

Status	
Device Name	pagingadapter-00a11f
SIP Registration	Page No Account
Call Status	Idle
Proxy Status	Single proxy mode
Provisioning Status	None Found
MAC	00:22:ee:00:a1:1f
IPv4	10.30.254.244/8, Gateway: 10.0.0.1
Date / Time	Mon Jan 29 20:03:47 GMT 2024
Next Scheduled Event	No Events Scheduled
Next Scheduled Action	No Actions Scheduled
Current Action	None
Multicast Mode	Transmitter Mode. Idle
Volume	Page Volume: 10 (0dB)
Ambient Noise Volume Adjustment	0dB
Noise Level	45dB
Relay Input Status	Disabled

Figure 6: Device status tab.

2.3 Register Your Product

You may register your product at <https://www.algosolutions.com/product-registration/> to ensure access to the latest upgrades for the 8305 and receive important service notices.

2.4 Reset

The recessed reset button (RST) next to the Ethernet Jack can only reset the 8305 Multi-Interface IP Paging Adapter during power-up. A reset will set all configuration options to factory default, including the login password.

To return all the settings in the 8305 to the factory default,

1. Reboot or power cycle the 8305.
2. When the SIP LED flashes, press and hold the reset button until the SIP LED begins a double flash pattern.
3. Release the reset button and allow the unit to complete its boot process.
4. Once booting is complete, press the reset button to play the IP address via the analog output ports.

2.5 Security

Algo devices use TLS for provisioning and SIP signaling to mitigate cyberattacks by those trying to intercept, replicate, or alter Algo products. Algo devices also come pre-loaded with certificates from a list of trusted certificate authorities (CA) to ensure secure communication with reputable sources. Pre-installed trusted certificates are not visible to users and are separate from those in the 'certs' folder.

3 SIP CONFIGURATION

SIP signaling is the underlying protocol for transmitting SIP messages between different entities in a network. SIP signaling establishes the call but does not contain the audio.

The 8305 can function as an IP telephone in a system with a simple configuration when a connected SIP extension is called. Also called a page extension, this enables the 8305 Multi-Interface IP Paging Adapter to recognize the configured extension and auto-answer when called.

A SIP endpoint license associated with a UC platform may be required to register the 8305. One license will be required per extension registered. If one device has multiple extensions registered, each registered extension will require a license. On a hosted or cloud platform, the required endpoint extension or seat may be treated the same as any other extension on the system and incur a monthly cost or similar fee.

3.1 Basic Settings

ALGO 8305 Multi-Interface IP Paging Adapter

[Status](#)
[Basic Settings](#)
[Additional Features](#)
[Scheduler](#)
[Advanced Settings](#)
[System](#)
[Logout](#)

SIP [Features](#) [Multicast](#)

SIP Settings

SIP

This section allows the SIP server information & account credentials to be entered. This information should be obtained from your telephone system administrator or hosted account provider. After saving these settings, see the [Status](#) tab to confirm successful registration.

SIP Domain (Proxy Server)

Default port is 5060. To specify a different port, enter PROXY:PORT, e.g. my_proxy.com:5070, or 192.168.1.10:5080.

Ring/Alert Mode

☒ Monitor "Ring" event on registered SIP extension
☐ None

Ring Extension

Authentication ID

Authentication Password

Display Name (Optional)

The device will detect inbound ring events on this extension and play the alerting tone (and multicast if configured) until the inbound call stops ringing. It will not answer the call on this extension.

Page Extension

Authentication ID

Authentication Password

Display Name (Optional)

The device will auto-answer any inbound call received on this extension and provide a voice paging path (and multicast if configured).

Figure 7: Configure basic SIP settings.

Use these SIP settings to enter SIP server information and account credentials. You can ask your telephone system administrator or hosted account provider for more details. After entering the information and saving the settings, check the **Status** tab to confirm the successful registration.

SIP	
SIP Domain (Proxy Server)	The SIP Server's IP address (e.g., 192.168.1.111) or domain name (e.g., myserver.com).

Ring/Alert Mode	Ring extensions do not answer incoming calls but play a customizable, pre-recorded announcement, such as a loud ringer (night bell). Announcements are customizable and can be pre-recorded. Use this setting to add a second SIP extension for a Ring event. If Monitor "Ring" event on registered SIP extension is selected, you will see additional settings for Ring extension parameters. None is set by default. If set, the device will detect inbound ring events on this extension and play the alerting tone (and multicast if configured) until the inbound call stops ringing. The 8305 will not answer the call on this extension. Often, the 8305 will be a member of a hunt group or ring group to ring in conjunction with a telephone. You may change the alert tone via Basic Settings → Features.
Ring Extension	Enter the SIP extension for the ring parameter of the 8305. The device will detect inbound ring events on this extension and play the alerting tone (and multicast if configured) until the inbound call stops ringing. It will not answer the call on this extension.
Page Extension	Page extensions auto-answer and open a voice path, enabling live announcements. Enter the SIP page extension for the 8305 so the device will auto-answer any inbound call received on this extension and provide a voice paging path (and multicast if configured).
Authentication ID	The Authentication ID is a name that represent the page extension. It is also referred to as 'Username' for some SIP servers. This may be the same as the Ring or Page extension in some cases.
Authentication Password	This is the SIP password for the registered SIP account. Up to eight (8) characters can be used. The password can be used to authenticate SIP users. Contact your System Administrator for the password to obtain access.
Display Name (Optional)	Enter the name you want displayed when an SIP call is made. For the display name to be shown, the PBX and phone(s) must be configured to display this message as the Caller ID.

3.2 More Page Extensions

ALGO 8305 Multi-Interface IP Paging Adapter

Status Basic Settings **Additional Features** Scheduler Advanced Settings System Logout

Input/Output Emergency Alerts **More Page Extensions** More Ring Extensions

More Page Extensions

This section allows dedicated extensions to be registered for each paging zone. This provides an alternative to the "DTMF Selectable Zone" option, thus allowing any zone to be called directly without the need to enter DTMF. Depending on the features available on your SIP phone system, this can provide benefits in allowing speed-dial keys to be programmed on user phones for paging a particular zone more easily, or dialing restrictions could potentially be used to allow only selected phones to access certain zones. This feature requires several SIP extensions to be registered with the SIP phone system.

i The 8305 will auto-answer any inbound calls received on these numbers and provide a voice paging path and multicast if configured. Note that only a single call can be active at a time.

i Note: Some SIP phone systems may not support this feature if they limit the number of extensions that can be registered on a single device.

i Multicast Zone Definitions can be found in "Advanced Settings > [Advanced Multicast](#)".

Basic Extensions	
Priority Call Page Extension	☐ Enabled ☒ Disabled <i>i</i> A call to the Priority Extension will override all other events on the device.
All Call Page Extension	☐ Enabled ☒ Disabled
Zone 1 Page Extension	☐ Enabled ☒ Disabled
Zone 2 Page Extension	☐ Enabled ☒ Disabled
Zone 3 Page Extension	☐ Enabled ☒ Disabled
Zone 4 Page Extension	☐ Enabled ☒ Disabled
Zone 5 Page Extension	☐ Enabled ☒ Disabled
Zone 6 Page Extension	☐ Enabled ☒ Disabled
Music Page Extension	☐ Enabled ☒ Disabled

Save

Figure 11: More page extensions.

Additional SIP paging extensions can be registered for each multicast zone. This enables you to dial a zone directly without entering DTMF Codes; however, this may require additional SIP licenses depending on the SIP provider. Some SIP telephone systems may not support this capability altogether if there is a limit on the number of extensions registered on a single device.

To configure additional page extensions (up to 50):

1. Select 'Enable' beside the extension of interest.
2. Enter the **Extension**, **Authentication ID**, and **Authentication Password**. You may enter a Display Name if you'd like.

The 8305 will auto-answer any inbound calls received on these numbers and provide a voice paging path and multicast if configured. Only a single call can be active at a time.

3.3 More Ring Extensions

ALGO 8305 Multi-Interface IP Paging Adapter

Status Basic Settings **Additional Features** Scheduler Advanced Settings System Logout

Input/Output Emergency Alerts More Page Extensions **More Ring Extensions**

More Ring Extensions

i This section allows additional extensions to be registered for the purpose of providing loud ringing alerts for more than one line. Unique ring tones can be selected for each line to allow them to be easily distinguished - for example a "Sales" line could have a different ring tone from a personal line. Appropriate call routing must be configured on your SIP phone system of course in order to trigger it to send calls to these different numbers.

i The 8305 will detect inbound ring events on these numbers and play the alerting tone until the inbound call stops ringing. It will not answer the calls in this mode.

i Note: Some SIP phone systems may not support this feature if they limit the number of extensions that can be registered on a single device.

Ring Extension 2	☐ Enabled ☒ Disabled
Ring Extension 3	☐ Enabled ☒ Disabled
Ring Extension 4	☐ Enabled ☒ Disabled
Ring Extension 5	☐ Enabled ☒ Disabled
Ring Extension 6	☐ Enabled ☒ Disabled
Ring Extension 7	☐ Enabled ☒ Disabled
Ring Extension 8	☐ Enabled ☒ Disabled
Ring Extension 9	☐ Enabled ☒ Disabled
Ring Extension 10	☐ Enabled ☒ Disabled

Rule-based Ring Tones

Allows the device to play a custom ring tone based on the identity of the caller. When enabled, the device will play the selected ring tone for callers with a display name or extension that matches the rule.

#1 Custom Tone	☐ Enabled ☒ Disabled
#2 Custom Tone	☐ Enabled ☒ Disabled
#3 Custom Tone	☐ Enabled ☒ Disabled
#4 Custom Tone	☐ Enabled ☒ Disabled

Custom Ring Tone

Allows the device to play a custom ringtone when a call is received with the "Alert-Info" SIP header.

☐ Enabled ☒ Disabled

Figure 17: More ring extensions.

Up to 10 SIP Ring extensions can be registered. To configure additional ring extensions, select **Enabled** beside an extension and enter the Extension, Authentication ID, and Authentication Password. If desired, a unique ringtone and multicast zone can be assigned to each extension.

Set a rule-based ringtone so the device plays a custom ringtone based on the caller's identity. When enabled, the device will play the selected ringtone for callers with a display name or extension that matches the rule.

Enable a custom ring to allow the device to play a custom ringtone when receiving a call with the "Alert-Info" SIP header.

3.4 Emergency Alerts

The screenshot displays the web interface for the ALGO 8305 Multi-Interface IP Paging Adapter. The top navigation bar includes tabs for Status, Basic Settings, Additional Features (selected), Scheduler, Advanced Settings, System, and Logout. Below this, a sub-navigation bar shows Input/Output, Emergency Alerts (selected), More Page Extensions, and More Ring Extensions.

Emergency Alerts

Information: This section allows pre-recorded announcements to be triggered & latched by calling an extension and hanging up. The announcement will continue to play until a different "Cancel" extension is called to clear the announcement (or a pre-defined timeout is reached). This can be useful for emergency notifications (e.g. "Evacuation Alert"), allowing staff to quickly dial a pre-configured number and then exit the building. Audio files can be easily uploaded to create custom announcements.

Information: Up to 10 extensions can be registered allowing up to 10 different announcements. A single "Cancel" extension also needs to be registered; calling this number will cancel the currently active announcement.

Note: Some SIP phone systems may not support this feature if they limit the number of extensions that can be registered on a single device.

Settings

Default Announcement Duration	☐ Play Once ☒ Play Until Cancelled
Default Maximum Announcement Time	10 minutes
Announcement Selection Mode	☐ Direct Extensions ☒ DTMF Selectable Use "Direct Extensions" to register a separate extension for each announcement. Use "DTMF Selectable" to register a single extension that accepts DTMF input to select which announcement to play.
Passcode Protected Announcement Extensions	☐ Enabled ☒ Disabled

DTMF Selection

Extension	
Authentication ID	
Authentication Password	
Display Name (Optional)	
Prompt Tone	<Default>

Call-to-Cancel

Call-to-Cancel Selection Mode	☒ Direct Extension ☐ DTMF 0 If using "DTMF 0", dial the main DTMF Selection extension and select 0 to cancel the announcement.
Extension	
Authentication ID	
Authentication Password	
Display Name (Optional)	
Confirmation Tone	<None>

Announcements

Announcement 1	☐ Enabled ☒ Disabled
Announcement 2	☐ Enabled ☒ Disabled
Announcement 3	☐ Enabled ☒ Disabled
Announcement 4	☐ Enabled ☒ Disabled
Announcement 5	☐ Enabled ☒ Disabled
Announcement 6	☐ Enabled ☒ Disabled

Figure 15: Emergency Alerts.

The 8305 is often used for emergency (e.g., lockdown, evacuation, reverse evacuation), safety (e.g., medical, workplace accident), and security events (e.g., OSHA or similar workplace regulations) alerting.

Emergency alerts notify others of an emergency quickly and efficiently. Users can trigger and latch an emergency alert or announcement by dialing a pre-configured extension (of which you may have many) or dial a single SIP extension and use DTMF to select an announcement. The announcement will continue to play on a loop until a different “Call-to-Cancel” extension is called to clear the announcement or a pre-defined timeout is reached.

Up to 10 extensions can be registered allowing up to 10 different announcements. A single “Call-to-Cancel” extension also needs to be registered. Calling this number will cancel an active announcement.

Note: Some SIP telephone systems may not support this feature if they limit the number of extensions that can be registered on a single device.

Settings StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout Input/OutputEmergency AlertsMore Page ExtensionsMore Ring Extensions Emergency Alerts <i>ⓘ</i> This section allows pre-recorded announcements to be triggered & latched by calling an extension and hanging up. The announcement will continue to play until a different "Cancel" extension is called to clear the announcement (or a pre-defined timeout is reached). This can be useful for emergency notifications (e.g. "Evacuation Alert"), allowing staff to quickly dial a pre-configured number and then exit the building. Audio files can be easily uploaded to create custom announcements. <i>ⓘ</i> Up to 10 extensions can be registered allowing up to 10 different announcements. A single "Cancel" extension also needs to be registered; calling this number will cancel the currently active announcement. <i>ⓘ</i> Note: Some SIP phone systems may not support this feature if they limit the number of extensions that can be registered on a single device. Settings Default Announcement Duration ☐ Play Once ☒ Play Until Cancelled Default Maximum Announcement Time 10 minutes Announcement Selection Mode ☒ Direct Extensions ☐ DTMF Selectable <i>ⓘ</i> Use "Direct Extensions" to register a separate extension for each announcement. Use "DTMF Selectable" to register a single extension that accepts DTMF input to select which announcement to play. Answer Inbound Call ☐ Enabled ☒ Disabled <i>ⓘ</i> This option selects how the Announcement calls are handled. In both cases, the Emergency Announcement is started when the appropriate extension is called and continues until the Cancel Extension is called. <i>ⓘ</i> Select "Enabled" to answer the inbound call and provide the option to play a confirmation tone before starting the alert, then automatically release the call. <i>ⓘ</i> Select "Disabled" to detect just the inbound Ring signal, but not actually answer the call	
Default Announcement Duration	An announcement can be played once or continuously until canceled. Select Play Once to play a single cycle of the chosen tone file. If Play Until Cancelled is selected, the announcement will continue to play until the "Call-to-Cancel" extension is called to clear the announcement or a defined timeout is reached.
Default Maximum Announcement Time	Select the maximum time an announcement can be played.

Announcement Selection Mode	Select Direct Extensions to register a separate extension for each announcement. Select DTMF Selectable to register a single extension that accepts DTMF input to select which announcement to play.
Answer Inbound Call	This setting indicates how Announcement calls are handled. In both cases, the Emergency Announcement is started when the appropriate extension is called and continues until the “Call-to-Cancel” extension is called. Select Enabled to answer the inbound call and provide the option to play a Confirmation Tone before starting the alert, then automatically release the call or request a passcode before playing the announcement. Select Disabled to detect the inbound Ring signal but not answer the call. Select Disabled to only detect the inbound Ring signal, but not answer the call. In both instances, the announcement will play until the time limit is reached or the “Call-to-Cancel” extension is called. Enabling Answer Inbound Call can be useful when the caller cannot hear the announcement from their location. However, if the call might go to a group or multiple extension(s) (including this device), the auto-answer may intercept that call and prevent it from ringing on other devices.
Passcode Protected Announcement Extensions	Select Enabled to require the caller to enter a passcode after dialing an announcement or “Call-to-Cancel” extension. Setting a passcode helps prevent unintentional announcements.
Announcement Passcode	Enter a passcode that a caller must enter to play or cancel an announcement. When prompted, the caller must enter the passcode followed by the # sign before the announcement will be played or canceled. The passcode prompt will be played before any other action. If the passcode is not correctly entered within 15 seconds, the call will be ended.
Passcode Prompt Tone	Select a tone to play when the passcode is ready to be entered.

DTMF Selection

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Logout

Input/Output

Emergency Alerts

More Page Extensions

More Ring Extensions

Emergency Alerts

DTMF Selection

Extension

Authentication ID

Authentication Password

Display Name (Optional)

Prompt Tone

Call-*o-Cancel!

Extension	Enter the SIP extension for the DTMF Selection parameter of the 8305.
Authentication ID	Enter the Authentication ID. It may also be called Username for some SIP servers or may be the same as the extension.
Authentication Password	Enter the SIP password provided by the system administrator for the SIP account.
Display Name (Optional)	Enter a 'Display Name' that will be sent when the SIP call is made. The PBX and phone(s) must be configured to display this message as the Caller ID.
Prompt Tone	Select a tone to play when the passcode is ready to be entered.

Call-to-Cancel

Status

Basic Settings

Additional Features

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System

Logout

Input/Output

Emergency Alerts

More Page Extensions

More Ring Extensions

Emergency Alerts

Call-to-Cancel

Call-to-Cancel Selection Mode

☒ Direct Extension
 ☐ DTMF 0

ⓘ If using "DTMF 0", dial the main DTMF Selection extension and select 0 to cancel the announcement.

Extension

Authentication ID

Authentication Password

Display Name (Optional)

Confirmation Tone

<None>

Announcements

Call-to-Cancel Selection Mode	If using "DTMF 0", the user should dial the main DTMF Selection extension and select '0' to cancel the announcement.
Extension	Enter the SIP extension for the Call-to-Cancel Selection parameter of the 8305.
Authentication ID	Enter the Authentication ID provided by the System Administrator. It may also be called Username for some SIP servers or may be the same as the extension.
Display Name (Optional)	Enter a 'Display Name' that will be sent when the SIP call is made. The PBX and phone(s) must be configured to display this message as the Caller ID.
Prompt Tone	Select a tone to play when the passcode is ready to be entered.

Announcements <a>Status <a>Basic Settings <a>Additional Features <a>Scheduler <a>Advanced Settings <a>System <a>Logout <a>Input/Output <a>Emergency Alerts <a>More Page Extensions <a>More Ring Extensions Emergency Alerts Announcements Announcement 1 ☐ Enabled ☒ Disabled Announcement 2 ☐ Enabled ☒ Disabled Announcement 3 ☐ Enabled ☒ Disabled Announcement 4 ☐ Enabled ☒ Disabled Announcement 5 ☐ Enabled ☒ Disabled Announcement 6 ☐ Enabled ☒ Disabled	
Announcement #	To configure an Emergency Alert extension, select Enabled for an announcement number. Up to 10 extensions can be registered allowing up to 10 different announcements. Audio files can be easily uploaded to create custom announcements. Only one 'Call-to-Cancel' extension is needed. Some SIP telephone systems may not support multiple announcements if they limit the number of extensions that can be registered on a single device.
Announcement Duration	Choose the duration of an announcement. The Default option follows the behavior configured in Default Announcement Duration .
Maximum Announcement Time	Select the maximum announcement time.
Tone/Pre-recorded Announcement	Select a file to use as a ringtone or announcement.
Confirmation Tone	Select a file to use as a confirmation tone.

3.5 Advanced SIP

ALGO 8305 Multi-Interface IP Paging Adapter

StatusBasic SettingsAdditional FeaturesScheduler**Advanced Settings**SystemLogout

NetworkAdminUsersTimeProvisioningAdvanced Audio**Advanced SIP**Advanced Multicast

Advanced SIP Settings

General

SIP Transportation

Auto

Select Auto to check DNS NAPTR record, then try UDP/TCP.

In TLS mode, if the SIP Server requires endpoints to be authenticated, a PEM file containing both a device certificate and a private key needs to be installed on the Algo device. Use the "System > File Manager" tab to upload a certificate file renamed to 'sipclient.pem' in the 'certs' folder.

SIPS Scheme

☐ Enabled ☒ Disabled

Validate Server Certificate

☐ Enabled ☒ Disabled

Validate the SIP server against common certificate authorities. To validate against additional certificates, use the "System > File Manager" tab to upload a Base64 encoded X.509 certificate file in .pem, .cer, or .crt format to the 'certs/trusted' folder.

SIP Outbound Support (RFC 5626)

☐ Enabled ☒ Disabled

Only enable this option if the SIP server supports RFC 5626.

Outbound Proxy

Register Period (seconds)

3600

SRTP

SDP SRTP Offer

Disabled

NAT

Media NAT

☒ None ☐ ICE ☐ STUN

Server Redundancy

Server Redundancy Feature (Multiple SIP Server Support)

☒ Enabled ☐ Disabled

Backup Server #1

Backup Server #2

Polling Interval (seconds)

120 seconds (2 minutes)

Time to wait between sending monitoring packets to each server. Inactive servers are always polled and the active server may optionally be polled (see below).

Poll Active Server

☐ Enabled ☒ Disabled

Explicitly poll the current server to monitor its availability. Polling may also be handled automatically by other regular events, so this can be disabled to reduce network traffic.

Automatic Failback

☒ Enabled ☐ Disabled

Reconnect with a higher priority server once available, even if the backup connection is still working.

Polling Method

☒ SIP NOTIFY ☐ SIP OPTIONS

SIP message used to poll servers in order to monitor their availability.

Interoperability

Keep-Alive Method

☒ None ☐ Double CRLF

This setting will enable sending periodic CRLF messages for both UDP and TCP connections.

Use Outgoing TLS port in SIP headers

☒ Enabled ☐ Disabled

Use ephemeral port number from outgoing SIP TLS connection instead of listening port number in SIP Contact and Via headers. This is useful to connect the device to some local SIP servers, like Asterisk or FreeSWITCH.

Do Not Reuse Authorization Headers

☐ Enabled ☒ Disabled

When enabled, all SIP authorization information from the last successful request will not be reused in the next request.

Allow Missing Subscription-State Headers

☐ Enabled ☒ Disabled

When enabled, allow SIP NOTIFY messages that do not contain a "Subscription-State" header.

Save

Figure 8: Configure Advanced SIP settings.

Figure 8: Configure Advanced SIP settings.

General

Status Basic Settings Additional Features Scheduler **Advanced Settings** System Logout

Network Admin Users Time Provisioning Advanced Audio **Advanced SIP** Advanced Multicast

Advanced SIP Settings

General

SIP Transportation: Auto
Select Auto to check DNS NAPTR record, then try UDP/TCP.
In TLS mode, if the SIP Server requires endpoints to be authenticated, a PEM file containing both a device certificate and a private key needs to be installed on the Algo device. Use the "System > File Manager" tab to upload a certificate file renamed to 'sipclient.pem' in the 'certs' folder.

SIPS Scheme: ☐ Enabled ☒ Disabled

Validate Server Certificate: ☐ Enabled ☒ Disabled
Validate the SIP server against common certificate authorities. To validate against additional certificates, use the "System > File Manager" tab to upload a Base64 encoded X.509 certificate file in .pem, .cer, or .crt format to the 'certs/trusted' folder.

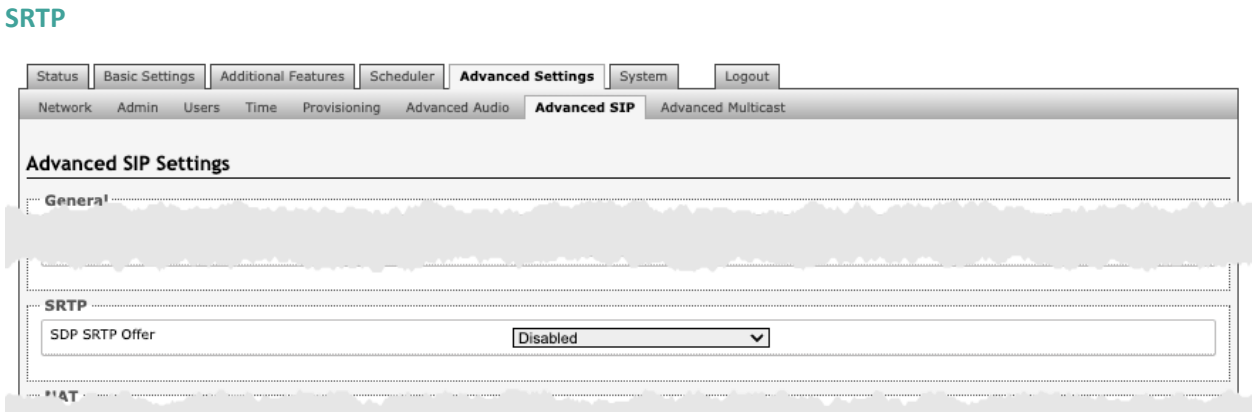
SIP Outbound Support (RFC 5626): ☐ Enabled ☒ Disabled
Only enable this option if the SIP server supports RFC 5626.

Outbound Proxy:

Register Period (seconds):

SIP Transportation	Select a transport layer protocol to use for SIP messages from the dropdown. These options include: Auto: Will check the DNS NAPTR record, then try UDP/TCP. UDP TCP TLS: Ensures the encryption of SIP traffic. In this mode, if the SIP Server requires endpoints to be authenticated, a PEM file containing both a device certificate and a private key must be installed on the 8305. Upload a certificate via System → File Manager and rename it to 'sipclient.pem' in the 'certs' folder.
SIPS Scheme	Only visible when SIP Transportation is set to TLS . Enable to require the SIP connection from endpoint to endpoint to be secure.
Validate Server Certificate	Enable to validate the SIP server against common certificate authorities. To validate additional certificates, navigate to System → File Manager to upload a Base64 encoded X.509 certificate file in .pem, .cer, or .crt format to the certs folder.
SIP Outbound Support (RFC 5626)	Enable this option to support best networking practices according to RFC 5626. This option should be enabled if the 8305 is registered with a hosted server or TLS is used for SIP Transportation.

Outbound Proxy	Enter the IP address for an outbound proxy.
Register Period (seconds)	Enter the maximum requested period where the 8305 will re-register with the SIP server. The default setting is 3600 seconds (1 hour). Note that if an Expires header is provided by the SIP response 200 (OK), this time will take precedence over the Register Period defined time here. Only change if instructed to do so.

	
SDP SRTP Offer	Select an option from the dropdown menu: • Disabled: Only unencrypted calls will be accepted. • Standard: Only encrypted calls will be accepted. This option secures the audio data (SRTP) between parties by ensuring that it cannot be reconstructed and listened to later. • Optional (Non-standard AVP Profile): Both unencrypted and encrypted SIP calls will be accepted, no matter if the other party supports SRTP. The SIP call's RTP data will be unencrypted if the other party does not support SRTP.

NAT

Advanced SIP Settings

General

NAT

Media NAT ☒ None ☐ ICE ☐ STUN

ICE - TURN Server

ICE - TURN User

ICE - TURN Password

STUN - Server

Media NAT	IP address for STUN server if present or IP address/credentials for a TURN server.
ICE – TURN Server	Enter the IP address or domain of the ICE server.
ICE – TURN User	Enter the username.
ICE – TURN Password	Enter the password.
STUN - Server	Enter the IP address or domain of the STUN server.

Server Redundancy

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Advanced SIP Settings

Server Redundancy

Server Redundancy Feature (Multiple SIP Server Support) ☒ Enabled ☐ Disabled

Backup Server #1

Backup Server #2

Polling Interval (seconds) ⓘ Time to wait between sending monitoring packets to each server. Inactive servers are always polled and the active server may optionally be polled (see below).

Poll Active Server ☐ Enabled ☒ Disabled ⓘ Explicitly poll the current server to monitor its availability. Polling may also be handled automatically by other regular events, so this can be disabled to reduce network traffic.

Automatic Fallback ☒ Enabled ☐ Disabled ⓘ Reconnect with a higher priority server once available, even if the backup connection is still working.

Polling Method ☒ SIP NOTIFY ☐ SIP OPTIONS ⓘ SIP message used to poll servers in order to monitor their availability.

Server Redundancy Feature	Enable to configure up to two secondary backup servers. When enabled, the 8305 will attempt to register with the primary server but switch to a secondary server when necessary. The configuration allows re-registration to the primary server upon availability or to stay with a server until unresponsive.
Backup Server #1, #2	Provided by your SIP provider or IT team.
Polling Intervals (seconds)	Select the time interval for sending monitoring packets to each server from the dropdown menu. Inactive servers are always polled and the active server may optionally be polled.
Poll Active Server	Enable to explicitly poll the current server to monitor availability. Other regular events may also handle this automatically and can be disabled to reduce network traffic.
Automatic Fallback	Enable to allow the 8305 to reconnect with a higher priority server once available, even if the backup connection is still working.
Polling Method	Select a polling method based on what your SIP provider supports.

Interoperability Status Basic Settings Additional Features Scheduler Advanced Settings System Logout Network Admin Users Time Provisioning Advanced Audio Advanced SIP Advanced Multicast Advanced SIP Settings General Interoperability Keep-Alive Method ☒ None ☐ Double CRLF <i> ⓘ This setting will enable sending periodic CRLF messages for both UDP and TCP connections. </i> Use Outgoing TLS port in SIP headers ☒ Enabled ☐ Disabled <i> ⓘ Use ephemeral port number from outgoing SIP TLS connection instead of listening port number in SIP Contact and Via headers. This is useful to connect the device to some local SIP servers, like Asterisk or FreeSWITCH. </i> Do Not Reuse Authorization Headers ☐ Enabled ☒ Disabled <i> ⓘ When enabled, all SIP authorization information from the last successful request will not be reused in the next request. </i> Allow Missing Subscription-State Headers ☐ Enabled ☒ Disabled <i> ⓘ When enabled, allow SIP NOTIFY messages that do not contain a "Subscription-State" header. </i> Save	
Keep-Alive Method	Select a keep-alive method: None Double CRLF: The 8305 will send a packet regularly to maintain connection with the SIP Server if behind NAT.
Keep-Alive Interval	Set the interval in seconds that the CRLF message should be sent. 30 seconds is recommended.
Use Outgoing TLS port in SIP Headers	Enable to use the ephemeral port number from an outgoing SIP TLS connection instead of the listening port number in SIP Contact and Via headers. This is useful for connecting the device to some local SIP servers, like Asterisk or FreeSWITCH.
Do Not Reuse Authorization Headers	Enable so all SIP authorization information from the last successful request will not be reused in the next request.
Allow Missing Subscription-State Headers	Enable to allow SIP NOTIFY messages that do not contain a 'Subscription-State' header.

4 MULTICAST CONFIGURATION

The 8305 Multi-Interface IP Paging Adapter can be programmed as a multicast transmitter or receiver. As a paging adapter designed to emulate an analog telephone connection, the ability to multicast allows you to scale communications you are already making in a simple and effective way. IP endpoints connected to the 8305 can be grouped into up to 50 multicast zones and paged via DTMF Selectable Mode or multiple SIP extensions.

Dual-tone multi-frequency (DTMF) refers to the sounds or tones a telephone generates when the numbers are pressed. To page with DTMF Selectable Mode, a user can dial the SIP extension of the transmitter device and dial the desired DTMF page zone (e.g., 1, 2, etc.) on the keypad.

Another way to page multiple zones with the 8305 is through multiple registered SIP extensions on the transmitter device. Each extension is mapped to a unique zone, allowing zones to be called directly.

4.1 Multicast IP Addresses

Each 8305 has a unique IP address and shares a common multicast IP and port number (multicast zone) for multicast packets. The Transmitter units send to a configurable multicast zone, and the Receiver units listen to assigned multicast zones.

The network switches and router see the packet and deliver it to all the group members. The multicast IP and port number must be the same on all one group's Transmitter and Receiver units. The user may define multiple zones by picking different multicast IP addresses and/or port numbers.

1. Multicast IP addresses range: 224.0.0.0/4 (from 224.0.0.0 to 239.255.255.255)
2. Port numbers range: from 1 to 65535
3. By default, the 8305 is set to use the multicast IP address 224.0.2.60 and the port numbers 50000-50008

Ensure the multicast IP address and port number do not conflict with other services and devices on the same network.

4.2 Enable Multicast Streaming

The 8305 Multi-Interface IP Paging Adapter multicast features only require the first endpoint be registered as a SIP extension. If only one audio stream is active at any given time, additional Algo IP Endpoints, including any combination of paging adapters, speakers, and visual alerters, may be added as multicast receivers. If multiple unique audio streams are needed simultaneously, more than one transmitter will be required.

The Algo IP endpoint configured as the transmitter will stream audio to the receivers simultaneously. Receiver endpoints do not require SIP extensions and do not need to register with the SIP Communication Server.

To enable multicast streaming from the transmitter adapter, open the web interface and go to the **Basic Settings → Multicast** tab. For Multicast Mode, select **Transmitter (Sender)**. For Transmitter Single Zone, select **All Call**.

To enable multicast monitoring of the receiver endpoints, go to the web interface for each endpoint and navigate to the **Basic Settings → Multicast** tab. For Multicast Mode, select **Receiver (Listener)**. There is no need to select a Transmitter Single Zone. The endpoint will monitor the **All Call** zone IP address by default.

The page pre-announce tone is generated from the transmitter. The speaker volume can be increased or decreased for each multicast receiver individually.

4.3 Multicast: Transmitter (Sender)

ALGO
8305 Multi-Interface IP Paging Adapter

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Features
Multicast

Multicast Settings

Multicast Mode

Multicast Mode

☐ None
☒ Transmitter (Sender)
☐ Receiver (Listener)

ℹ Multicast Zone Definitions can be found in "Advanced Settings > [Advanced Multicast](#)".

Multicast Type

Multicast Type

☐ Regular (RTP)
☐ Polycom Group Page
☐ Polycom Push-to-Talk
☒ Regular RTP + Polycom Group Page
☐ Regular RTP + Polycom Push-to-Talk

ℹ Regular mode uses RTP audio packets compatible with all Algo SIP endpoints, and most multicast-enabled phones.

Number of Zones

Number of Zones

☒ Basic Zones Only
☐ Basic and Expanded Zones

Polycom Group Paging/Push-to-Talk

Polycom Zone

224.0.1.116:5001

ℹ Enter the same Multicast IP Address & Port number as configured on the Polycom phones.

Polycom Group Selection Mode

☐ DTMF Selectable Group
☒ Single Group

Polycom Default Channel

Group 1

Speaker Playback Groups

☒ Group 1
☒ Group 2
☒ Group 3
☒ Group 4
☒ Group 5
☒ Group 6
☒ Group 7
☒ Group 8
☒ Group 9
☒ Group 10
☒ Group 11
☒ Group 12
☒ Group 13
☒ Group 14
☒ Group 15
☒ Group 16
☒ Group 17
☒ Group 18
☒ Group 19
☒ Group 20
☒ Group 21
☒ Group 22
☒ Group 23
☒ Group 24
☒ Group 25

Select All
 Clear All

ℹ Allows Multicast Transmitter device to play audio for selected groups only. This is useful if using DTMF Selectable Zone mode (or [More Page Extensions](#) per zone) and wishing to make the Transmitter a member of only certain groups.

Transmitter (Sender) Zone Settings

Zone Selection Mode

☒ DTMF Selectable Zone
☐ Single Zone

ℹ For additional capabilities allowing unique SIP extensions per zone, see "Additional Features > [More Page Extensions](#)".

Transmitter Single Zone

Zone 1

ℹ If "DTMF Selectable Zone" is selected above, then this single zone setting will not apply to Paging (since the zone can now be dynamically selected per call using DTMF), but it will still apply to the Ring Extension and Relay triggered events.

Speaker Playback Zones

☒ Priority Call
☒ All Call
☒ Music
☒ Zone 1
☒ Zone 2
☒ Zone 3
☒ Zone 4
☒ Zone 5
☒ Zone 6

ℹ Allows Multicast Transmitter device to play audio for selected zones only. This is useful if using DTMF Selectable Zone mode (or [More Page Extensions](#) per zone) and wishing to make the Transmitter a member of only certain zones.

DTMF Settings

Zone Selection Tone

<Default>

Two Digit Selection

☐ Enabled
☒ Disabled

ℹ If enabled, all DTMF Selectable Zones will require two digits. As a result, Basic Zones must be prefixed with "0" (ie. 01, 02, etc) and Expanded Zones no longer need to be prefixed with "**".

Save

Figure 9: Multicast transmitter mode settings.

GoMomentum.com

32

888.538.3960

Multicast Mode

Always ensure that the multicast settings on all Receiver devices match those of the Transmitter.

Multicast Mode	If Transmitter (Sender) is selected, the 8305 will broadcast an IP stream when activated in addition to playing audio through the audio output. The 8305 cannot be both a multicast Transmitter and Receiver simultaneously.
Multicast Type	The 8305 may broadcast multicast paging compatible with Poly “on-premise group paging” protocol and most multicast-enabled phones that use RTP audio packets. Select Regular (RTP) if you are only multicasting to Algo IP endpoints or multicast-enabled phones. To multicast page announcements to Poly phones, select Poly Group Page or Poly Push-to-Talk. Select Regular RTP + Poly Group Page or Regular RTP + Push-to-Talk to multicast page audio to Poly phones, Algo IP endpoints, and multicast-enabled phones.
Number of Zones	Select Basic Zones Only if configuring nine or fewer multicast zones. Select Basic and Expanded Zones to configure up to 50 zones. The expanded zones have the same behavior as the basic Receiver zones but are hidden by default to simplify the interface.

Poly Group Paging/Push-to-Talk

This section is used if the Multicast Type includes Poly Group Page or Poly Push-to-Talk.

Multicast Settings

Polycom Group Paging/Push-to-Talk

Polycom Zone: 224.0.1.116:5001
Enter the same Multicast IP Address & Port number as configured on the Polycom phones.

Polycom Group Selection Mode: ☐ DTMF Selectable Group ☒ Single Group

Polycom Default Channel: Group 1

Speaker Playback Groups:

☒ Group 1	☒ Group 2	☒ Group 3	☒ Group 4	☒ Group 5
☒ Group 6	☒ Group 7	☒ Group 8	☒ Group 9	☒ Group 10
☒ Group 11	☒ Group 12	☒ Group 13	☒ Group 14	☒ Group 15
☒ Group 16	☒ Group 17	☒ Group 18	☒ Group 19	☒ Group 20
☒ Group 21	☒ Group 22	☒ Group 23	☒ Group 24	☒ Group 25

[Select All](#) [Clear All](#)

Allows Multicast Transmitter device to play audio for selected groups only. This is useful if using DTMF Selectable Zone mode (or [More Page Extensions](#) per zone) and wishing to make the Transmitter a member of only certain groups.

Poly Zone	Enter the same Multicast IP Address and Port number configured on the Poly phones.
Poly Group Selection Mode	Select Single Group to broadcast on one pre-configured group. Multiple SIP extensions can be registered on the Transmitter device. Each extension is mapped to a unique group, allowing groups to be called directly (e.g., from speed-dial keys). See Additional Features → More Page Extensions for additional configuration settings. If DTMF Selectable Group is selected, the group is determined by the DTMF selection between 0 – 25. To page using DTMF Selectable Zone: - Dial the SIP extension of the Transmitter device - Dial the desired DTMF page group number on the keypad when prompted. Groups 10 and higher start with “*”. DTMF group definitions include: - DTMF Extension 1 for Zone 1 - DTMF Extension 2 for Zone 2

	... • DTMF Extension *10 for Zone 10 • DTMF Extension *11 for Zone 11 All DTMF codes and respective zones are available in Advanced Settings → Advanced Multicast.
Poly Default Channel	Select the default group for the multicast stream to be sent to. If DTMF Selectable Group is chosen, this single group setting will not apply to paging since the group will be dynamically selected per call using DTMF. The Single Group setting will still apply to the ring extension and relay triggered events. The Poly Default Channel is the default channel used for multicast actions unless an option is available for a custom channel with specific parameters.
Speaker Playback Groups	Select Speaker Playback Groups to control which specific groups can play audio from the device. This is useful if using the DTMF Selectable Group mode or additional page extensions (Additional Features → More Page Extensions) per group to make 8305 a member of only certain zones. In this case, the Transmitter does not participate in the Zone but transmits certain traffic.

Transmitter (Sender) Zone Settings

This section is used if the Multicast Type includes Regular (RTP).

Multicast Settings

Transmitter (Sender) Zone Settings

Zone Selection Mode: ☒ DTMF Selectable Zone ☐ Single Zone
For additional capabilities allowing unique SIP extensions per zone, see "Additional Features > More Page Extensions".

Transmitter Single Zone: Zone 1
If "DTMF Selectable Zone" is selected above, then this single zone setting will not apply to Paging (since the zone can now be dynamically selected per call using DTMF), but it will still apply to the Ring Extension and Relay triggered events.

Speaker Playback Zones:

☒ Priority Call	☒ All Call	☒ Music
☒ Zone 1	☒ Zone 2	☒ Zone 3
☒ Zone 4	☒ Zone 5	☒ Zone 6

Allows Multicast Transmitter device to play audio for selected zones only. This is useful if using DTMF Selectable Zone mode (or [More Page Extensions](#) per zone) and wishing to make the Transmitter a member of only certain zones.

Zone Selection Mode

Select **Single Zone** to broadcast on one pre-configured zone. Multiple SIP extensions can be registered on the Transmitter device. Each extension is mapped to a unique zone, allowing zones to be called directly (e.g., from speed-dial keys). See **Additional Features → More Page Extensions** for additional configuration settings.

If **DTMF Selectable Zone** is selected, the zone is determined by the DTMF selection between 0 – 50. Once multicast Transmitter mode is enabled, navigate to **Advanced Settings → Advanced Multicast** to find the DTMF codes corresponding to each zone.

To page using **DTMF Selectable Zone**:

1. Dial the SIP extension of the Transmitter device
2. Dial the desired DTMF page zone number on the keypad when prompted. Zones 10 and higher start with "*".

DTMF zone definitions include:

- DTMF Extension 9 for Priority Call
- DTMF Extension 0 or 8 for All Call
- DTMF Extension 1 for Zone 1
- DTMF Extension *10 for Zone 10
- DTMF Extension *11 for Zone 11

	All DTMF codes and respective zones are available in Advanced Settings → Advanced Multicast .
Transmitter Single Zone	Select the default zone for the multicast stream to be sent to. If DTMF Selectable Zone is chosen, this single zone setting will not apply to Paging since the zone will be dynamically selected per call using DTMF. However, this single zone setting will still apply to the ring extension and relay triggered events. The Transmitter Single Zone is the default zone used for multicast actions unless an option is available for a custom zone with specific parameters.
Speaker Playback Zones	Select Speaker Playback Zones to control which specific zones the 8305 can play audio. This is useful if using the DTMF Selectable Zone mode or additional page extensions (Additional Features → More Page Extensions) per zone to make 8305 a member of only certain zones. In this case, the Transmitter does not participate in the Zone but transmits certain traffic.

DTMF Settings

[Status](#)
[Basic Settings](#)
[Additional Features](#)
[Scheduler](#)
[Advanced Settings](#)
[System](#)
[Logout](#)

[SIP](#)
[Features](#)
[Multicast](#)

Multicast Settings

DTMF Settings

Zone Selection Tone

<Default>

Two Digit Selection

☐ Enabled
 ☒ Disabled

ⓘ If enabled, all DTMF Selectable Zones will require two digits. As a result, Basic Zones must be prefixed with "0" (ie. 01, 02, etc) and Expanded Zones no longer need to be prefixed with "**".

Save

Zone Selection Tone	Select a tone to be played to prompt a user to select a zone to multicast to. This may be used as an interactive voice response (IVR) menu by uploading a custom audio file in the tones folder through System → File Manager. Each zone may use a different tone. This can be configured in Advanced Settings → Advanced Multicast.
Two-Digit Selection	When enabled, all DTMF Selectable Zones will require two digits. As a result, Basic Zones must be prefixed with 0, and Expanded Zones will no longer need to be prefixed with *.

4.4 Multicast: Receiver (Listener)

ALGO 8305 Multi-Interface IP Paging Adapter

Status Basic Settings Additional Features Scheduler Advanced Settings System Logout

SIP Features **Multicast**

Multicast Settings

Multicast Mode

Multicast Mode: ☐ None ☐ Transmitter (Sender) ☒ Receiver (Listener)
† Multicast Zone Definitions can be found in "Advanced Settings > [Advanced Multicast](#)".

Multicast Type: ☒ Regular (RTP) ☐ Polycom Group Page ☐ Polycom Push-to-Talk
† Regular mode uses RTP audio packets compatible with all Algo SIP endpoints, and most multicast-enabled phones.

Number of Zones: ☒ Basic Zones Only ☐ Basic and Expanded Zones

Receiver (Listener) Zone Settings

Basic Receiver Zones: ☒ Priority Call ☒ All Call ☐ Music
☒ Zone 1 ☐ Zone 2 ☐ Zone 3
☐ Zone 4 ☐ Zone 5 ☐ Zone 6
† A multicast to the Priority Call zone will override all other events on the device, except for a direct call to a Priority Page Extension in the More Page Extensions tab.

☒ Save

Figure 10: Multicast receiver mode settings.

Multicast Mode

Always ensure that the multicast settings on all Receiver devices match those of the Transmitter.

Multicast Mode	If Receiver (Listener) mode is selected, the 8305 will activate when receiving a multicast message. It will mimic the audio stream of the transmitter but use local volume settings. This can be set via Basic Settings → Features → Page Speaker Volume .
----------------	--

Multicast Type	Select Regular if receiving multicast from other Algo IP endpoint(s) and/or multicast-enabled phone(s) that use RTP audio packets. Select Poly Group Page or Poly Push-to-Talk if receiving multicast paging compatible with Poly “on-premise group paging” protocol.
Number of Zones	Select Basic Zones Only if configuring nine or fewer multicast zones. Select Basic and Expanded Zones to configure up to 50 zones. The expanded zones have the same behavior as the basic Receiver zones but are hidden by default to simplify the interface.

Receiver (Listener) Zone Settings Status Basic Settings Additional Features Scheduler Advanced Settings System Logout SIP Features Multicast Multicast Settings Receiver (Listener) Zone Settings Basic Receiver Zones ☒ Priority Call ☒ All Call ☐ Music ☒ Zone 1 ☐ Zone 2 ☐ Zone 3 ☐ Zone 4 ☐ Zone 5 ☐ Zone 6 <i>ⓘ</i> A multicast to the Priority Call zone will override all other events on the device, except for a direct call to a Priority Page Extension in the More Page Extensions tab. Save	
Basic Receiver Zones	Select one or more multicast zones for the 8305 to listen to. Multicast zone priority will be based on the zone definition list order defined in Advanced Settings → Advanced Multicast.
Expanded Receiver Zones	Select additional zones (up to 50) for the device to listen to. This is only possible when Basic and Expanded Zones is selected.

Poly Group Paging/Push-to-Talk

Status

Basic Settings

Additional Features

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SIP

Features

Multicast

Multicast Settings

Polycom Group Paging/Push-to-Talk

Polycom Zone

224.0.1.116:5001

Enter the same Multicast IP Address & Port number as configured on the Polycom phones.

Polycom Receiver Channels

☒Group 1
 ☐Group 2
 ☐Group 3
 ☐Group 4
 ☐Group 5

☐Group 6
 ☐Group 7
 ☐Group 8
 ☐Group 9
 ☐Group 10

☐Group 11
 ☐Group 12
 ☐Group 13
 ☐Group 14
 ☐Group 15

☐Group 16
 ☐Group 17
 ☐Group 18
 ☐Group 19
 ☐Group 20

☐Group 21
 ☐Group 22
 ☐Group 23
 ☒Group 24
 ☒Group 25

Select All

Clear All

A multicast to Groups 24 or 25 will override all other events on the device, except for a direct call to a Priority Page Extension in the More Page Extensions tab.

Save

Poly Zone	Enter the Poly Zone (IP Address and Port) that matches the configuration of the Poly phones and Channels.
Poly Receiver Channels	If using a Poly telephone as a Multicast Transmitter, a tone may be set for any of the 25 Poly Groups configured on the 8305. Poly Group Tones can be set in Advanced Settings → Advanced Multicast. The Poly telephone used as page audio source for the 8305 must be configured to use either the G.711 or G.722 audio codec. Note that Poly phone(s) must be configured with the “Compatibility” setting (“ptt.compatibilityMode”) disabled for this codec setting to be applied.

4.5 Using Multicast Page Zones

The 8305 Multi-Interface IP Paging Adapter can listen to nine basic multicast zones; however, up to 50 are available (See **Additional Features → More Page Extensions** for more details). The multicast IP addresses define these zones.

By default these zones have the names below but can be used however you prefer. When set as a multicast receiver, zones have a priority hierarchy where zones higher on the list will be treated with higher priority, with **Music** being the lowest priority. When set as a multicast transmitter, event priority is based on the event type that initiated the multicast rather than the output multicast channel that will be active.

- Priority
- All Call
- Zone 1
- Zone 2
- Zone 3
- Zone 4
- Zone 5
- Zone 6
- Music

There are two options for paging to multiple zones:

1. **DTMF Selectable Mode:** Has a dynamic page zone selection and requires only the transmitting device to have a registered SIP extension. To page, dial the SIP extension of the transmitter and dial the desired DTMF page zone (e.g., 1, 2, etc.) on the keypad. DTMF digits and their corresponding zone numbers can be found in the **Advanced Settings → Advanced Multicast** tab of the 8305 web interface.
2. **Multiple page extensions:** Multiple SIP extensions can be registered on the transmitter. Each extension is mapped to a unique zone, allowing zones to be called directly. See **Additional Features → More Page Extensions** tab of the 8305 web interface for more details.

4.6 Advanced Multicast

These settings are only visible when in Transmitter or Receiver multicast mode. This can be set in **Basic Settings** → **Multicast**. The default pre-populated multicast zone IP addresses and ports will work in most cases and should only be altered for rare cases.

ALGO 8305 Multi-Interface IP Paging Adapter

Status Basic Settings Additional Features Scheduler **Advanced Settings** System Logout

Network Admin Users Time Provisioning Advanced Audio Advanced SIP **Advanced Multicast**

Advanced Multicast Settings

Current multicast mode: Transmitter
Multicast mode can be set in "Basic Settings > Multicast".

Transmitter Settings

Transmitter Output Codec: **G.722**
When using Two-Way Paging mode, only G.711 and G.722 are supported.

Output Packetization Time (milliseconds): **20**

Multicast TTL: **1**
Only change this setting if custom routing is configured on the network that specifically routes multicast packets between subnets, and a longer TTL count is required. Regular multicast routing does not require a change to this setting.

RTP Control Protocol (RTCP)

RTCP Port Selection: ☒ Disabled ☐ Next Higher Port ☐ Multiplexed on Same Port
Select the port on which packets will be sent or received. If using the 'Next Higher Port' option, ensure that the default multicast zone definitions are modified such that zones are only assigned to even-numbered ports, leaving the next higher odd-numbered ports free for RTCP packets.

Basic Zone Definition

Zone	IP Address and Port	Page Tone
Priority Call (DTMF:9)	224.0.2.60:50000	<Use Default Page Tr>
All Call (DTMF:0/8)	224.0.2.60:50001	<Use Default Page Tr>
Zone 1 (DTMF:1)	224.0.2.60:50002	<Use Default Page Tr>
Zone 2 (DTMF:2)	224.0.2.60:50003	<Use Default Page Tr>
Zone 3 (DTMF:3)	224.0.2.60:50004	<Use Default Page Tr>
Zone 4 (DTMF:4)	224.0.2.60:50005	<Use Default Page Tr>
Zone 5 (DTMF:5)	224.0.2.60:50006	<Use Default Page Tr>
Zone 6 (DTMF:6)	224.0.2.60:50007	<Use Default Page Tr>
Music (DTMF:7)	224.0.2.60:50008	<Use Default Page Tr>

Expanded Zone Definition

Zone	IP Address and Port	Page Tone
Zone 10 (DTMF: *10)	224.0.2.110:50000	<Use Default Page Tr>
Zone 11 (DTMF: *11)	224.0.2.111:50000	<Use Default Page Tr>
Zone 12 (DTMF: *12)	224.0.2.112:50000	<Use Default Page Tr>

Figure 12: Advanced multicast - transmitter settings.

Transmitter Settings Status Basic Settings Additional Features Scheduler Advanced Settings System Logout Network Admin Users Time Provisioning Advanced Audio Advanced SIP Advanced Multicast Advanced Multicast Settings <i>Current multicast mode: Transmitter</i> Multicast mode can be set in "Basic Settings > Multicast". Transmitter Settings Transmitter Output Codec G.722 <i>When using Two-Way Paging mode, only G.711 and G.722 are supported.</i> Output Packetization Time (milliseconds) 20 Multicast TTL 1 <i>Only change this setting if custom routing is configured on the network that specifically routes multicast packets between subnets, and a longer TTL count is required. Regular multicast routing does not require a change to this setting.</i>	
Transmitter Output Codec	Select an audio encoding format for the Transmitter device to use when sending output to the Receivers. Supported formats include: - G.711 ulaw - G.722 - Opus - L16 Only G.711 and G.722 are supported when using Two-Way Paging mode.
Output Packetization Time (milliseconds)	Select the size of the audio packets the Transmitter sends to the Receivers from the dropdown menu. The default of 20 milliseconds is recommended unless a different value is specifically required for compatibility with other devices.
Multicast TTL	Only change the multicast time to live (TTL) setting if custom routing is configured on the network that specifically routes multicast packets between subnets and a longer TTL count is required. This ensures packets are not bounced back and forth in a network identity. When the TTL is reached, the router drops the packet.

RTP Control Protocol (RTCP)

Advanced Multicast Settings

RTP Control Protocol (RTCP)

RTCP Port Selection

☒ Disabled
 ☐ Next Higher Port
 ☐ Multiplexed on Same Port

ⓘ Select the port on which packets will be sent or received.
 If using the 'Next Higher Port' option, ensure that the default multicast zone definitions are modified such that zones are only assigned to even-numbered ports, leaving the next higher odd-numbered ports free for RTCP packets.

RTCP Port Selection

Select how a port will be chosen to send or receive RTCP packets.

Note: If **Next Higher Port** is selected, ensure that the default multicast zone definitions are modified so that zones are only assigned to even-numbered ports, leaving the next higher odd-numbered ports free for RTCP packets.

Receiver Settings

Advanced Multicast Settings

ⓘ Current multicast mode: Receiver
 Multicast mode can be set in "Basic Settings > [Multicast](#)".

Receiver Settings

Audio Sync (milliseconds, 0 ~ 1000)

0

ⓘ When using multicast with other third-party devices that have a delay in their audio path, the audio on the 8305 may be heard slightly earlier than on these other devices. Use this feature to add a small delay to the audio output on the 8305 in order to synchronize with these other devices. Applies to Multicast Receiver mode only.

Audio Sync

Available if under **Basic Settings** → **Multicast** the **Multicast Mode** is set to **Receiver (Listener)** and **Multicast Type** is set to **Poly Group Page** or **Poly Push-to-Talk**. When using multicast with other third-party devices that have a delay in their audio path, the audio on the 8305 may be heard slightly earlier than on these other devices. Use this feature to add a small delay to the audio output on the 8305 to synchronize with these other devices.

Polycom Receiver Tones

Status Basic Settings Additional Features Scheduler **Advanced Settings** System Logout

Network Admin Users Time Provisioning Advanced Audio Advanced SIP **Advanced Multicast**

Advanced Multicast Settings

Current multicast mode: Receiver
Multicast mode can be set in 'Basic Settings > Multicast'.

Polycom Receiver Tones

If using an Algo device as a Multicast Transmitter, it is recommended to set the Multicast Receiver tones to "None" to avoid conflicts, as the Algo devices already multicast a tone by default.

Group	Page Tone	Page Volume
Group 1	<None>	<Use Default Page Volume>
Group 2	<None>	<Use Default Page Volume>
Group 3	<None>	<Use Default Page Volume>
Group 4	<None>	<Use Default Page Volume>

Poly Receiver Tones	Available if under Basic Settings → Multicast the Multicast Mode is set to Receiver (Listener) and Multicast Type is set to Poly Group Page or Poly Push-to-Talk . A tone may be set for any of the 25 Poly Groups. If using an Algo device as a Multicast Transmitter, it is recommended to set the Receiver tones to None to avoid conflicts, as the Algo devices already multicast a tone by default.
---------------------	--

5 AUDIO CONFIGURATION

In addition to voice paging, the 8305 Multi-Interface IP Paging Adapter can play audio files for emergency, safety, security announcements, customer service, shift changes, etc. Audio files can be stored on the adapter and played over a speaker in response to an event such as a ring, relay input, or automated schedule.

An emergency notification system is essential for delivering critical messages in seconds to those within your facility. The 8305 can also be connected to a visual alerter, such as a strobe light, to accompany audio emergency notifications.

5.1 Basic Audio Settings

ALGO

8305 Multi-Interface IP Paging Adapter

Status

Basic Settings

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Features

Inbound Ring Settings

These settings apply to events triggered by the Ring Extension(s) & Emergency Alerts sections. The Play/Loop/Stop buttons can also be used to test the device and set the appropriate volume level.

Ring/Alert Tone

warble2-med.wav

Play

Loop

Stop

Ring/Alert Volume

10

Apply

Ring Limit

No limit

1 ring = 6 seconds.

Inbound Page Settings

Page Volume

10

Apply

When in Receiver mode, note that this is the default volume control for all audio received via multicast.

Page Mode

One-way

Two-way

Delayed

Delayed mode stores the page audio temporarily, and then broadcasts it after the call is hung-up. This can help avoid feedback.

Page Timeout

5 minutes

Maximum page timeout in Delayed mode is 5 minutes.

Page Tone

<Default>

Use only Default, or custom uploaded file. The other pre-installed tone files all contain silence at the end in order to generate ring "cadence" of 6 seconds. This silence will block the voice path for several seconds at the start of a page.

G.722 Support

Enabled

Disabled

Applies to codec used during SIP negotiation only. Multicast codec is configured separately.

Passcode Protected Page Extensions

Enabled

Disabled

Set all page extensions to require the caller to enter a passcode. Setting a passcode helps prevent unintentional pages. When prompted, the caller must enter the passcode followed by the # sign before the page can be accepted. The passcode prompt will be played before any other action.

DTMF Detection Type

Auto

RTP Telephony Event (RFC 4733)

RTP In-band

SIP INFO

Audio

Ambient Noise Compensation

Enabled

Disabled

Automatically adjust speaker level in response to ambient noise level detected prior to the start of each call. Note: This feature requires an external microphone and can cause issues if the ambient noise at the microphone is low and the ambient noise at the speaker location is high. Additional settings can be found in "Advanced Settings > Advanced Audio"

Automatic Gain Control (AGC)

Enabled

Disabled

Automatically maximize level of voice received from calling phone in order to make page volume more consistent.

Analog Output Port/Level

Phone or 8Ω Port

This setting controls the maximum voltage level available on the 'Line Out' analog audio port, phone port, or 8Ω port. To achieve this maximum level, also set the volume control to level 10, and enable the AGC feature (optional) to increase the level received from the far-end phone when paging.

Audio Input Settings

Audio Input Port

Phone Port

Save

Figure 14: Basic Settings → Features.

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888.538.3960

Inbound Ring Settings

Ring settings apply to events triggered by Ring Extensions and Emergency Alerts. Emergency Alert tones are configured under **Additional Features → Emergency Alerts**.

Status

Basic Settings

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Features

Inbound Ring Settings

These settings apply to events triggered by the Ring Extension(s) & Emergency Alerts sections. The Play/Loop/Stop buttons can also be used to test the device and set the appropriate volume level.

Ring/Alert Tone

warble2-med.wav

Play Loop Stop

Ring/Alert Volume

10

Apply

Ring Limit

No limit

1 ring = 6 seconds.

Inbound Page Settings

Ring/Alert Tone	Select an audio file to play when a ring event is detected on the SIP Ring Extension. Test the audio file immediately using the Play, Loop, and Stop buttons if the 8305 is connected to a speaker. During multicast, the device will broadcast an audio stream using the Transmitter's selected ringtone. This is the default tone that will be played if selected in the settings Multicast → Additional Ring Extension.
Ring/Alert Volume	Set the volume for a SIP Ring event using the dropdown. This setting is for gain control and the output level depends on the levels recorded into the source audio file played from memory. This setting is only used for local tones, not multicast. See Page Speaker Volume below for multicast settings.
Ring Limit	Typically set to no limit. Ring Limit will limit how long the speaker will ring before timing out. A new ring event must occur for the speaker to play the audio file again.

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Inbound Page Settings

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Features

Inbound Page Settings

Page Volume
10
Apply

When in Receiver mode, note that this is the default volume control for all audio received via multicast.

Page Mode
☒ One-way
☐ Two-way
☐ Delayed

"Delayed" mode stores the page audio temporarily, and then broadcasts it after the call is hung-up. This can help avoid feedback.

Page Timeout
5 minutes

Maximum page timeout in Delayed mode is 5 minutes.

Page Tone
<Default>

Use only Default, or custom uploaded file. The other pre-installed tone files all contain silence at the end in order to generate ring "cadence" of 6 seconds. This silence will block the voice path for several seconds at the start of a page.

G.722 Support
☒ Enabled
☐ Disabled

Applies to codec used during SIP negotiation only. Multicast codec is configured separately.

Passcode Protected Page Extensions
☐ Enabled
☒ Disabled

Set all page extensions to require the caller to enter a passcode. Setting a passcode helps prevent unintentional pages. When prompted, the caller must enter the passcode followed by the # sign before the page can be accepted. The passcode prompt will be played before any other action.

DTMF Detection Type
☐ Auto
☒ RTP Telephony Event (RFC 4733)
☐ RTP In-band
☐ SIP INFO

Page Volume	This setting is for gain control for SIP or multicast paging. The output level will depend on the streaming level. Page Speaker Volume will apply to all inbound multicast streams (for Receiver mode only) regardless of audio source or type.
Page Mode	Set calls to the SIP page extension as one-way, two-way (using an external microphone), or delayed. In delayed mode, the speaker will record a message to be played after disconnecting. The device will buffer an announcement up to 5 minutes long. To cancel a page while in delay mode, press "*" while recording to prevent it from being sent after hanging up.
Page Timeout	Set the maximum duration for a page. The page will end when the timeout limit has been reached. This is useful to ensure the paging system is not stuck in an active state in cases where someone accidentally forgets to hang up.

Page Tone	Select a pre-page tone to be played when a page is starting. Use only the Default or custom uploaded files. Other pre-installed tone files contain silence at the end to generate a ring "cadence" of 6 seconds. This silence will block the voice path for several seconds at the start of a page. The "Default" tone is set to page-notif.wav. The Default Page Tone in Advanced Multicast will play the tone set here.
G.722 Support	Enable or disable the G.722 codec. G.722 enables wideband audio for optimum speech intelligibility.
Passcode Protected Page Extensions	When enabled, the caller must enter the set passcode followed by the # sign before the page can be made. Setting a passcode helps prevent unintentional pages.
Apply to All Page Extensions	Only visible when Passcode Protected Page Extensions is set to Enabled. Enable or disable a passcode for all page extensions.
Passcode	Only visible when Passcode Protected Page Extensions is set to Enabled. Passcodes can be up to 15 digits and must be numbers only.
Passcode Prompt Tone	Only visible when Passcode Protected Page Extensions is set to Enabled. Select the tone to be played to prompt the user to enter the passcode before paging.
DTMF Detection Type	Select the preferred dual-tone multi-frequency (DTMF) detection method. DTMF is a technology used with touch tone phones (the sound made when pressing a number key). The 8305 uses this for multi-zone selection, passcode, etc.

Audio Status Basic Settings Additional Features Scheduler Advanced Settings System Logout SIP Features Multicast Features Ambient Noise Compensation ☒ Enabled ☐ Disabled <i>ⓘ</i> Automatically adjust speaker level in response to ambient noise level detected prior to the start of each call. Note: This feature requires an external microphone and can cause issues if the ambient noise at the microphone is low and the ambient noise at the speaker location is high. Additional settings can be found in "Advanced Settings > Advanced Audio" Automatic Gain Control (AGC) ☒ Enabled ☐ Disabled <i>ⓘ</i> Automatically maximize level of voice received from calling phone in order to make page volume more consistent. Analog Output Port/Level Phone or 8Ω Port ⌵ <i>ⓘ</i> This setting controls the maximum voltage level available on the 'Line Out' analog audio port, phone port, or 8Ω port. To achieve this maximum level, also set the volume control to level 10, and enable the AGC feature (optional) to increase the level received from the far-end phone when paging.	
Ambient Noise Compensation	When enabled, Ambient Noise Compensation will allow the speaker level to adjust automatically in response to ambient noise levels detected at the device before the start of each call. The 8305 requires an external microphone to be connected to do this.
Automatic Gain Control (AGC)	Enable or disable AGC to normalize the audio level. Enabling ensures the speaker is always played at a consistent volume.
Analog Output Port/Level	Set the maximum voltage level available on the 'Line Out' analog audio port, telephone port, or 8Ω port. To achieve the maximum level, also set the volume control to level 10 and enable the Automatic Gain Control (AGC) to increase the level received from the far-end telephone when paging.

Audio Input Settings

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Features

Audio Input Settings

Audio Input Port

Phone Port

Save

Audio Input Port	Select telephone Port or Aux Out to choose where audio will be played to.
------------------	---

5.2 Tones

The 8305 includes several pre-loaded audio files that can be selected to play for various events. The web interface allows you to select a file and play it immediately over the speaker for testing, available in **Basic Settings** → **Features**. Files may also be added, deleted, or renamed. For more information see section **X.Y.Z (system file manager)**.

ALGO 8305 Multi-Interface IP Paging Adapter

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File Manager

Tones

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Tones

Use the "System > [File Manager](#)" tab to upload tone files to "tones" subdirectory.

Files

Download and Install Ring Tones from the Algo Server

Download and Install

Tone files can be downloaded manually from [the Algo website](#).

Cache

Rebuild Tone Cache Files

Rebuild All

Only needed when the tone cache is out of sync. The operation might take a long time depending on the types and sizes of the tone files.

Test Tones

warble2-med.wav

Play Loop Stop

Figure 16: Tones settings.

Files

StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout

MaintenanceFirmwareFile ManagerTonesSystem LogCreditsAbout

Tones

Use the "System > [File Manager](#)" tab to upload tone files to "tones" subdirectory.

Files

Download and Install Ring Tones from the Algo Server
Download and Install
Tone files can be downloaded manually from [the Algo website](#).

Download and Install Ring Tones from the Algo Server	Tone files can be downloaded manually from the Algo website.
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Cache

StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout

MaintenanceFirmwareFile ManagerTonesSystem LogCreditsAbout

Tones

Use the "System > [File Manager](#)" tab to upload tone files to "tones" subdirectory.

Cache

Rebuild Tone Cache Files
Rebuild All
Only needed when the tone cache is out of sync. The operation might take a long time depending on the types and sizes of the tone files.

Test Tones
warble2-med.wav
Play Loop Stop

Rebuild Tone Cache Files	Only needed when the tone cache is out of sync. The operation might take a long time depending on the types and sizes of the tone files.
Test Tones	Listen to uploaded audio files before selecting them for your system.

5.3 Advanced Audio

ALGO 8305 Multi-Interface IP Paging Adapter

StatusBasic SettingsAdditional FeaturesScheduler**Advanced Settings**SystemLogout

NetworkAdminUsersTimeProvisioning**Advanced Audio**Advanced SIPAdvanced Multicast

Advanced Audio Functions

Functions

Dynamic Range Compression (DRC)

☐ Enabled ☒ Disabled

Compress the dynamic range of page audio to increase loudness.

Jitter Buffer Range (milliseconds, 10 ~ 500)

100

Adds more buffering if necessary to correct for inconsistent delays on the network. Use of the lowest value generally is recommended.

Generate In-Band DTMF Tones

☐ Enabled ☒ Disabled

Play DTMF tones during a SIP Call to allow interoperability with DTMF-controlled multi-zone amplifiers

Always Send RTP Media

☒ Enabled ☐ Disabled

Audio Filters

Speaker Filter

None

Bandwidth also limited by audio codecs.

Speaker Noise Filter

☐ Enabled ☒ Disabled

Aggressive 8th order Elliptical Filter (fc = 145Hz)

Microphone Filter

None

Microphone Noise Filter

☐ Enabled ☒ Disabled

Aggressive 8th order Elliptical Filter (fc = 145Hz)

Ambient Noise Compensation

Ambient Noise Compensation No Loss

☐ Enabled ☒ Disabled

Configure the Ambient Noise Compensation algorithm to only use levels at or above the current volume.

Ambient Noise Compensation Max Volume

10

Set maximum speaker level in response to ambient noise.

Exclude Polycom Groups

☐ Group 1☐ Group 2☐ Group 3☐ Group 4☐ Group 5

☐ Group 6☐ Group 7☐ Group 8☐ Group 9☐ Group 10

☐ Group 11☐ Group 12☐ Group 13☐ Group 14☐ Group 15

☐ Group 16☐ Group 17☐ Group 18☐ Group 19☐ Group 20

☐ Group 21☐ Group 22☐ Group 23☐ Group 24☐ Group 25

Select AllClear All

Save

Figure 18: Advanced audio settings.

Functions StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout NetworkAdminUsersTimeProvisioningAdvanced AudioAdvanced SIPAdvanced Multicast Advanced Audio Functions Functions Dynamic Range Compression (DRC) ☐ Enabled ☒ Disabled Compress the dynamic range of page audio to increase loudness. Jitter Buffer Range (milliseconds, 10 ~ 500) 100 Adds more buffering if necessary to correct for inconsistent delays on the network. Use of the lowest value generally is recommended. Generate In-Band DTMF Tones ☐ Enabled ☒ Disabled Play DTMF tones during a SIP Call to allow interoperability with DTMF-controlled multi-zone amplifiers Always Send RTP Media ☒ Enabled ☐ Disabled	
Dynamic Range Compression (DRC)	Enable to compress the dynamic range of page audio to increase loudness.
Dynamic Range Compression Gain	Select the amount of compression gain from the dropdown menu. More gain increases distortion.
Jitter Buffer Range	Enter a value between 10-500 to add more buffering if necessary to correct for inconsistent delays on the network. It is recommended to use the lowest value.
Generate In-Band DTMF Tones	Enable to play DTMF tones during an SIP call to allow interoperability with DTMF-controlled multi-zone legacy communication systems.
Always Send RTP Media	Enable to send audio packets at all times, even during one-way paging mode. This option is needed when the server expects to always see audio packets.

Audio Filters StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout NetworkAdminUsersTimeProvisioningAdvanced AudioAdvanced SIPAdvanced Multicast Advanced Audio Functions Speaker Filter None Bandwidth also limited by audio codecs. Speaker Noise Filter ☐ Enabled ☒ Disabled Aggressive 8th order Elliptical Filter (fc = 145Hz) Microphone Filter None Microphone Noise Filter ☐ Enabled ☒ Disabled Aggressive 8th order Elliptical Filter (fc = 145Hz) Ambient Noise Compensation	
Speaker Filter	Select a frequency from the dropdown to apply a high-pass filter to the speaker output. This setting reduces audio artifacts like humming or buzzing by filtering out unwanted frequencies.
Speaker Noise Filter	Enable to filter below 145 Hz to reduce mains-induced noise like fans.
Microphone Filter	Select a frequency from the dropdown to apply a high-pass filter to the microphone input. This setting reduces audio artifacts like humming or buzzing by filtering out unwanted frequencies.
Microphone Noise Filter	Enable to filter below 145 Hz to reduce mains-induced noise like fans.

Ambient Noise Compensation

Only available if **Ambient Noise Compensation** is **Enabled** in **Basic Settings** → **Features**.

Ambient Noise Compensation

Ambient Noise Compensation No Loss ☐ Enabled ☒ Disabled
 ⓘ Configure the Ambient Noise Compensation algorithm to only use levels at or above the current volume.

Ambient Noise Compensation Max Volume
 ⓘ Set maximum speaker level in response to ambient noise.

Exclude Polycom Groups

☐ Group 1 ☐ Group 2 ☐ Group 3 ☐ Group 4 ☐ Group 5
☐ Group 6 ☐ Group 7 ☐ Group 8 ☐ Group 9 ☐ Group 10
☐ Group 11 ☐ Group 12 ☐ Group 13 ☐ Group 14 ☐ Group 15
☐ Group 16 ☐ Group 17 ☐ Group 18 ☐ Group 19 ☐ Group 20
☐ Group 21 ☐ Group 22 ☐ Group 23 ☐ Group 24 ☐ Group 25

Ambient Noise Compensation No Loss

Configure the Ambient Noise Compensation algorithm to only use levels at or above the current volume. The current volume is the minimum volume when this setting is enabled.

Ambient Noise Compensation Max Volume

Based on ambient noise levels, a maximum volume can be set.

6 SCHEDULE CONFIGURATION

The 8305 includes a calendaring functionality synchronized to Network Time Protocol (NTP). It can schedule school bells, play automated announcements for retail and healthcare, and notify workplace shift changes and breaks.

Calendar events can be scheduled to play in specific zones, allowing the delivery of announcements to part or all of a facility. This feature is most applicable in education or manufacturing environments where specific building areas (e.g., classrooms or production floors) may need regular bell schedules and announcements, while other areas may not require ongoing messages.

6.1 Calendar

The Calendar displays scheduled events like bells and announcements. These can include events deployed with Algo IP speakers, paging adapters, and visual alerters.

The screenshot shows the ALGO 8305 Multi-Interface IP Paging Adapter web interface. The top navigation bar includes tabs for Status, Basic Settings, Additional Features, Scheduler, Advanced Settings, System, and Logout. The Scheduler tab is active, and the Calendar sub-tab is selected. The calendar displays a grid for January 2024. A dropdown menu at the top right of the calendar is set to 'Lunch Bell', and the 'Recurring' option is selected. The calendar shows 'Lunch Bell' events for every day from Monday to Friday. At the bottom, the recurrence settings are configured for 'Weekly' with days of the week M through F selected, a frequency of '1 weeks', and an end date of '2024-06-30'. A 'Save' button is located at the bottom right.

Figure 19. Calendar on the Schedules tab of the 8305 web interface.

After an event is created in the **Schedules** tab (see next section), you can add it to the calendar view by clicking the dropdown menu at the top right of the calendar.

To set a single date for your event, click on **Single** and then the date you would like the event to be played. Click **Save** to save your schedule.

To set a recurring schedule, click on **Recurring** then set the parameters at the bottom of the calendar. Once these are set, click on the date on the calendar that you want your recurring schedule to start. Click **Save** to save your schedule.

If you would like to remove a schedule, click **None (clear)** in the dropdown menu, then click on the schedule in the calendar you want to remove. Click **Save** to save your changes.

6.2 Schedules

The screenshot displays the ALGO 8305 Multi-Interface IP Paging Adapter web interface. The top navigation bar includes tabs for Status, Basic Settings, Additional Features, **Scheduler**, Advanced Settings, System, and Logout. Below this, a sub-navigation bar shows Calendar, **Schedules**, and Data. The main content area is titled 'Current Schedule: Lunch Bell' and features a table with columns for Colour, Schedule Name, Event Description, Time, Audio, and Page Group. A 'Create Schedule' button is located at the bottom left, while 'Add Event' and 'Save' buttons are at the bottom right. The system time 'Mon 29 Jan, 2024 22:48:37 (System Time)' is displayed in the top right corner.

Figure 20. Setting up a schedule for the 8305 in the web interface.

The Schedules tab is used to set and configure events such as preset pages, announcements, or notifications.

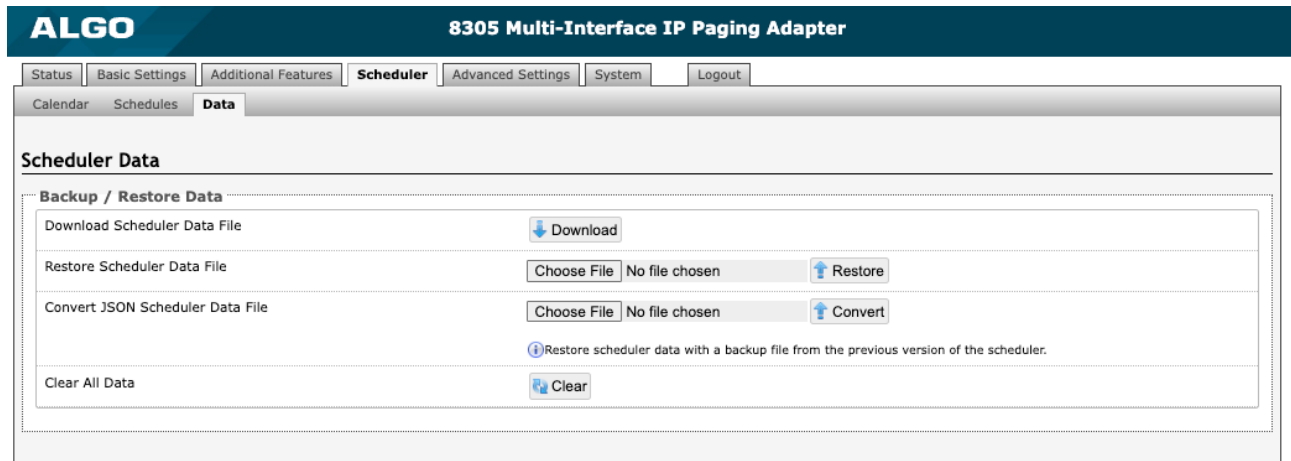
To create a schedule, click **Create Schedule**. Enter a **Schedule Name** and select a **Colour** to represent the schedule in the calendar.

To add an event to a schedule, select the schedule you want to modify and click **Add Event** at the bottom of the interface. Add an **Event Description**, **Time**, and **Audio**. If your device is set to Multicast Transmitter, you should also set the **Page Zone**. The selected audio file will be played locally over the network via multicast to Algo endpoints or RTP multicast compatible third-party equipment configured as Receivers in this zone.

Once a schedule has been configured, it can be added to the desired dates on the **Calendar**. Up to 30 different schedules can be created. For example, Fridays might have a different schedule than the other weekdays. Each schedule may contain up to 500 events.

	Copy event button		Delete the schedule or event
--	-------------------	--	------------------------------

6.3 Data



ALGO 8305 Multi-Interface IP Paging Adapter

Status Basic Settings Additional Features **Scheduler** Advanced Settings System Logout

Calendar Schedules **Data**

Scheduler Data

Backup / Restore Data

Download Scheduler Data File [Download](#)

Restore Scheduler Data File [Choose File](#) No file chosen [Restore](#)

Convert JSON Scheduler Data File [Choose File](#) No file chosen [Convert](#)

Restore scheduler data with a backup file from the previous version of the scheduler.

Clear All Data [Clear](#)

Figure 21. Configuring the Scheduler → Data tab.

Backup/Restore Data	
Download Scheduler Data File	Download a backup database file of schedules with events, times, and calendar dates. Note that this backup is independent from the rest of the configuration backup on the device.
Restore Scheduler Data File	Upload and restore a saved Scheduler data file.
Convert JSON Scheduler Data File	If you are migrating from old firmware and would like to restore your data to a newer device, you may upload the JSON file from your old system here.
Clear All Data	Clear all the Scheduler data including saved schedules and set calendar dates.

7 INTEGRATION

7.1 Input/Output

ALGO **8305 Multi-Interface IP Paging Adapter**

StatusBasic Settings**Additional Features**SchedulerAdvanced SettingsSystemLogout

Input/OutputEmergency AlertsMore Page ExtensionsMore Ring Extensions

Input/Output

General

Relay Input Mode

- ☐ Disabled
- ☐ Relay Normally Open
- ☒ Relay Normally Open with Supervision (e.g. Algo 1203 Call Switch)
- ☐ Relay Normally Closed
- ☐ Relay Normally Closed with Supervision
- ☐ Mute Switch
- ☐ Mute Switch with Supervision
- ☐ Algo 1202 Call Button
- ☐ Algo 1204 Volume Control Switch (Local or Remote)
- ☐ Algo 1204 Volume Control Switch with Supervision (Local or Remote)
- ☐ Algo 2507 Ring Detector

Action When Input Triggered

Action

- ☒ Play Tone
- ☐ Make Two-Way SIP Voice Call
- ☐ Make SIP Call with Tone

i "Play Tone" and "Stream Audio" will play sound on the Line Out and Aux Out ports as well as multicast if configured.

Tone/Pre-recorded Announcementchime.wav

Tone Duration

- ☒ Play Once
- ☐ Play While Held
- ☐ Play Until Completion

Action When Tamper Detected

Wiring Fault Supervision Mode

- ☒ Detect Open Circuit Fault Only
- ☐ Detect Both Open Circuit & Short Circuit Faults

i Open Circuit detection will trigger when the current draw is <4mA.
Short Circuit detection will trigger when the current draw is >36mA.
The nominal source voltage on the Relay Input circuit is 13V, with a 40mA current limit.

Action

- ☒ Play Tone
- ☐ Make Two-Way SIP Voice Call
- ☐ Make SIP Call with Tone

i "Play Tone" and "Stream Audio" will play sound on the Line Out and Aux Out ports as well as multicast if configured. Note that this action will occur 5 seconds after a wiring fault is detected. If the fault is resolved within 5 seconds, this action will not occur.

Tone/Pre-recorded Announcementbuzzer.wav

Tone Duration

- ☒ Play Once
- ☐ Play While Held
- ☐ Play Until Completion

Tone Multicast Settings

Use Separate Multicast

- ☐ Enabled
- ☒ Disabled

i This will allow the tone to be played via multicast even if the device is configured as a receiver.

Output

Output Relay

- ☒ Enabled
- ☐ Disabled

Save

Figure 13: Input settings.

General Status Basic Settings Additional Features Scheduler Advanced Settings System Logout Input/Output Emergency Alerts More Page Extensions More Ring Extensions Input/Output General Relay Input Mode ☐ Disabled ☐ Relay Normally Open ☒ Relay Normally Open with Supervision (e.g. Algo 1203 Call Switch) ☐ Relay Normally Closed ☐ Relay Normally Closed with Supervision ☐ Mute Switch ☐ Mute Switch with Supervision ☐ Algo 1202 Call Button ☐ Algo 1204 Volume Control Switch (Local or Remote) ☐ Algo 1204 Volume Control Switch with Supervision (Local or Remote) ☐ Algo 2507 Ring Detector	
Relay Input Mode	The 8305 has dry contact input terminals to connect external accessories, including Algo and third-party accessories. Options for Relay Input Mode include: • Disabled • Relay Normally Open • Relay Normally Open with Supervision (e.g. Algo 1203 Call Switch) • Relay Normally Closed • Relay Normally Closed with Supervision • Mute Switch • Mute Switch with Supervision • Algo 1202 Call Button • Algo 1204 Volume Control Switch (Local or Remote) • Algo 1204 Volume Control Switch with Supervision (Local or Remote) • Algo 1205 Audio Interface Switch • Algo 1205 Audio Interface Switch with Supervision • Algo 2507 Ring Detector Notification actions can be triggered via supervision settings if the input switch is disconnected.

Action When Input Triggered

Action

Play Tone

When the 8305 input is triggered, a tone or a pre-recorded audio file will play over the speaker or multicast. This function can be used to request support or assistance in service or retail environments, notify about an emergency at a specific location in medical or educational facilities, or sound an alarm during an intrusion.

Action When Input Triggered

Action	☒ Play Tone ☐ Make Two-Way SIP Voice Call ☐ Make SIP Call with Tone <i>Play Tone and "Stream Audio" will play sound on the Line Out and Aux Out ports as well as multicast if configured.</i>
Tone/Pre-recorded Announcement	chime.wav
Tone Duration	☒ Play Once ☐ Play While Held ☐ Play Until Completion

Make Two-Way SIP Voice Call

When the 8305 input is triggered, a voice path will open for an intercom-like call via an external microphone connected to a pre-configured telephone extension. This option can be used when a call needs to be made from a public place where a telephone would not be practical to use.

Action When Input Triggered

Action	☐ Play Tone ☒ Make Two-Way SIP Voice Call ☐ Make SIP Call with Tone <i>Play Tone and "Stream Audio" will play sound on the Line Out and Aux Out ports as well as multicast if configured.</i>
Extension to Dial	 <i>SIP account required in Page Extension fields in order to make a call.</i>
Call Mode	☒ Regular Two-Way Call ☐ Silent Microphone Monitoring ☐ Silent Microphone Monitoring with Tone
Allow 2nd Button Press	☒ Disabled ☐ End and Restart Call ☐ End Call

Make SIP Call with Tone

When the 8305 input is triggered, a private call can be made to a pre-configured telephone extension with a pre-recorded message. For instance, a call to a supervisor's telephone notifying about an emergency or intrusion at some location.

Action When Input Triggered

Action	☐ Play Tone ☐ Make Two-Way SIP Voice Call ☒ Make SIP Call with Tone <i>Play Tone and "Stream Audio" will play sound on the Line Out and Aux Out ports as well as multicast if configured.</i>
Extension to Dial	 <i>SIP account required in Page Extension fields in order to make a call.</i>
Allow 2nd Button Press	☒ Disabled ☐ End and Restart Call ☐ End Call
Tone/Pre-recorded Announcement	chime.wav
Interval Between Tones (seconds)	0
Maximum Tone Duration	None

Tone/Pre-recorded Announcement	Available when Action is set to Play Tone or Make SIP Call with Tone. Select a recording or tone to use. Custom audio files may be used and uploaded through System → File Manager.
Tone Duration	Available when Action is set to Play Tone .
Extension to Dial	Available when Action is set to Make Two-Way SIP Voice Call or Make SIP Call with Tone. A SIP account is required in Page Extension fields to make a call.
Call Mode	Available when Action is set to Make Two-Way SIP Voice Call .
Allow 2nd Button Press	Available when Action is set to Make Two-Way SIP Voice Call or Make SIP Call with Tone. If enabled, the 2nd button press will End Call or End and Restart Call. Therefore, if an input is triggered a second time, the SIP call will be terminated and, in some cases, immediately called again.
Interval Between Tones	Available when Action is set to Make SIP Call with Tone. Specify the time delay (seconds) between tones.
Maximum Tone Duration	Available when Action is set to Make SIP Call with Tone. Select the maximum tone duration. The tone will be terminated once the maximum time is reached.

Action When Tamper Detected

8305 can be configured with supervision to execute one of the above three actions (**Play Tone**, **Make Two-Way SIP Voice Call**, **Make SIP Call with Tone**) if the device goes offline due to wiring failure or after being tampered with.

For example, a tone could sound over the speaker(s) or a private pre-recorded message could be sent to a specified telephone extension. The supervision configuration options will appear if a Relay Input Mode with supervision is selected.

See “Action When Input Triggered” above for information on additional settings.

The screenshot shows the web interface of the 8305 Multi-Interface IP Paging Adapter. The top navigation bar includes tabs for Status, Basic Settings, Additional Features, Scheduler, Advanced Settings, System, and Logout. The 'Additional Features' tab is selected, and the 'Input/Output' sub-tab is active. The main content area is titled 'Input/Output' and contains a section for 'Action When Tamper Detected'. This section includes a 'Wiring Fault Supervision Mode' with three radio button options: 'Detect Open Circuit Fault Only' (selected), 'Detect Both Open Circuit & Short Circuit Faults', and 'Open Circuit detection will trigger when the current draw is <4mA. Short Circuit detection will trigger when the current draw is >36mA. The nominal source voltage on the Relay Input circuit is 13V, with a 40mA current limit.' Below this is the 'Action' section with three radio button options: 'Play Tone' (selected), 'Make Two-Way SIP Voice Call', and 'Make SIP Call with Tone'. A note states: '*Play Tone' and 'Stream Audio' will play sound on the Line Out and Aux Out ports as well as multicast if configured. Note that this action will occur 5 seconds after a wiring fault is detected. If the fault is resolved within 5 seconds, this action will not occur.' The 'Tone/Pre-recorded Announcement' section has a dropdown menu set to 'buzzer.wav'. The 'Tone Duration' section has three radio button options: 'Play Once' (selected), 'Play While Held', and 'Play Until Completion'.

Wiring Fault
Supervision Mode

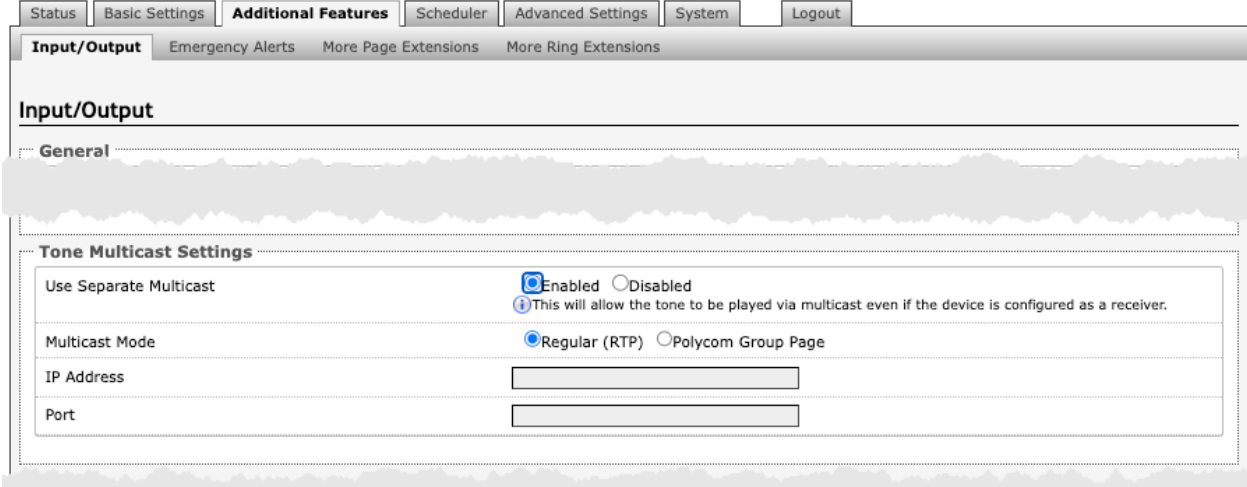
Short circuit detection will be triggered when the current draw is <4 mA. Short circuit detection will trigger when the current draw is >36 mA. The nominal source voltage on the Relay Input circuit is 13 V with a 40 mA current limit.

Action

Play Tone

When the 8305 input is triggered, a tone or a pre-recorded audio file will play over the speaker or multicast. This function can be used to request support or assistance in service or retail environments, notify about an emergency at a specific location in medical or educational facilities, or sound an alarm during an intrusion.

	Make Two-Way SIP Voice Call When the 8305 input is triggered, a voice path will open for an intercom-like call via an external microphone connected to a pre-configured telephone extension. This option can be used when a call needs to be made from a public place where a telephone would not be practical to use. Make SIP Call with Tone When the 8305 input is triggered, a private call can be made to a pre-configured telephone extension with a pre-recorded message. For instance, a call to a supervisor's telephone notifying about an emergency or intrusion at some location.
Tone/Pre-recorded Announcement	Available when Action is set to Play Tone or Make SIP Call with Tone. Select a recording or tone to use. Custom audio files may be used and uploaded through System → File Manager.
Tone Duration	Available when Action is set to Play Tone .

Tone Multicast Settings 	
Use Separate Multicast	When enabled, the set tone will be played via multicast even if the 8305 is configured as a receiver. See additional options when enabled.
Multicast Mode	Use the same details as the receiver zone that is being listened to.
IP Address	Use the same details as the receiver zone that is being listened to.

Port	Use the same details as the receiver zone that is being listened to.
------	--

Outbound SIP Call Settings

Status

Basic Settings

Additional Features

Scheduler

Advanced Settings

System

Logout

Input/Output

Emergency Alerts

More Page Extensions

More Ring Extensions

Input/Output

Outbound SIP Call Settings

Outbound Ring Limit

1 ring

1 ring = 6 seconds

Ringback Tone

<Default>

Maximum Call Duration

5 minutes

Outbound Ring Limit	Available when Action is set to Make Two-Way SIP Voice Call or Make SIP Call with Tone. Select the number of rings that will occur before the call reaches voicemail. One ring is six seconds.
Ringback Tone	Available when Action is set to Make Two-Way SIP Voice Call. Select a ringback tone to play during an outbound SIP call while waiting for the far-end party to answer.
Maximum Call Duration	Available when Action is set to Make Two-Way SIP Voice Call.

Audio Input Settings

Audio Input Port	Select the input port your device is connected to.
Audio Input Volume	Set the desired input volume.

Output

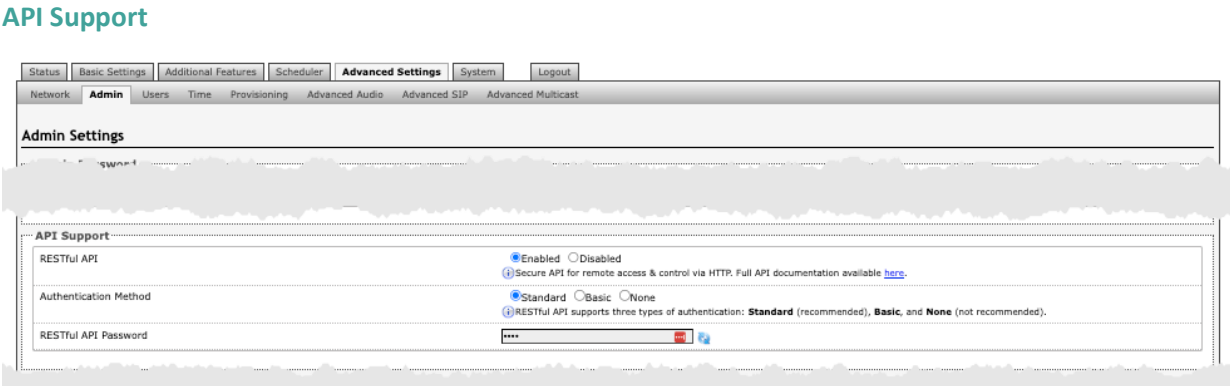
Output Relay	Enable to close the circuit when you attempt to trigger events elsewhere. This setting controls whether the output relay activates or not. Note that when enabled, the output relay will activate whenever the 8305 is activated (paging, alerting, etc.) This is a normally open relay only.
--------------	--

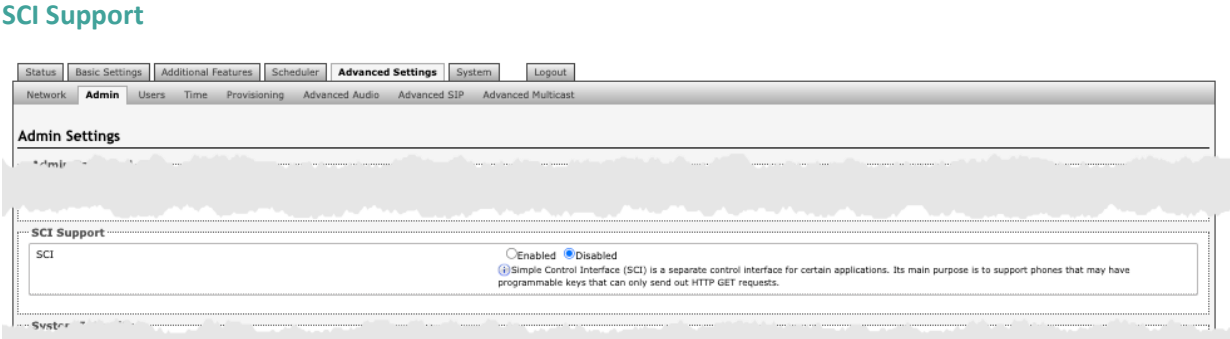
7.2 API

Algo RESTful API can be used to access, manipulate, and trigger Algo endpoints on your network through HTTP/HTTPS requests.

Requesting systems can interact with Algo devices through a uniform and predefined set of stateless operations.

To configure API settings on your 8305 Multi-Interface IP Paging Adapter, use the web interface and navigate to **Advanced Settings → Admin → API Support**.

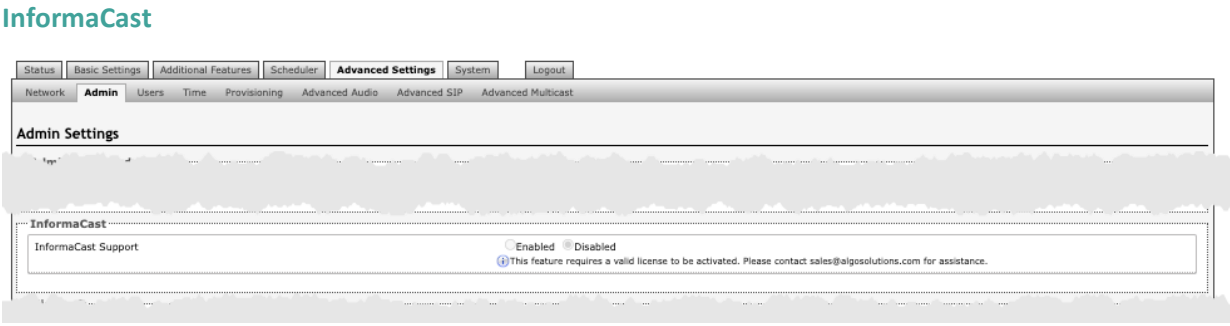
	
RESTful API	Enable a secure API for remote access and device control via HTTP.
Authentication Method	Speak to your IT Administrator for more information.
RESTful API Password	Speak to your IT Administrator for more information.

SCI Support 	
SCI	Simple Control Interface (SCI) is a separate control interface for certain applications. Its primary purpose is to support phones that may have programmable keys that can only send out HTTP GET requests.

7.3 InformaCast

As a Singlewire Solutions Partner, Algo products have been certified for compatibility and interoperability.

To set up your 8305 Multi-Interface IP Paging Adapter with Informacast, use the web interface and navigate to **Advanced Settings → Admin → InformaCast**.

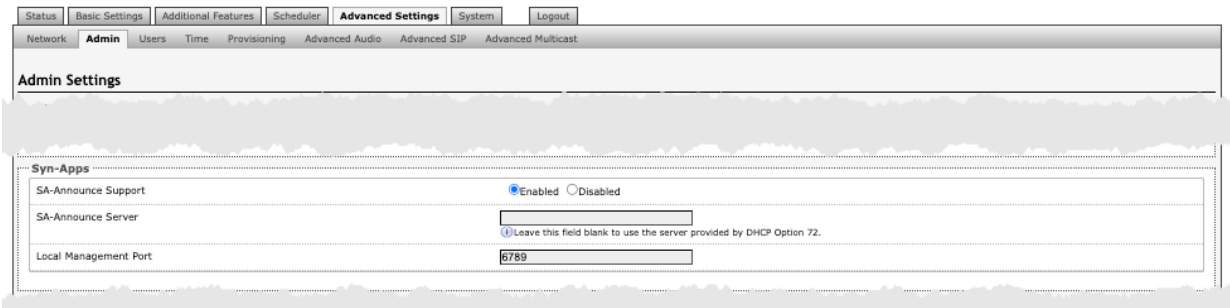
InformaCast 	
InformaCast Support	This feature requires a valid InformaCast license to be activated.

7.4 Syn-Apps

As a Syn-Apps Partner, Algo products have been Syn-Apps Certified for compatibility and interoperability.

Syn-Apps

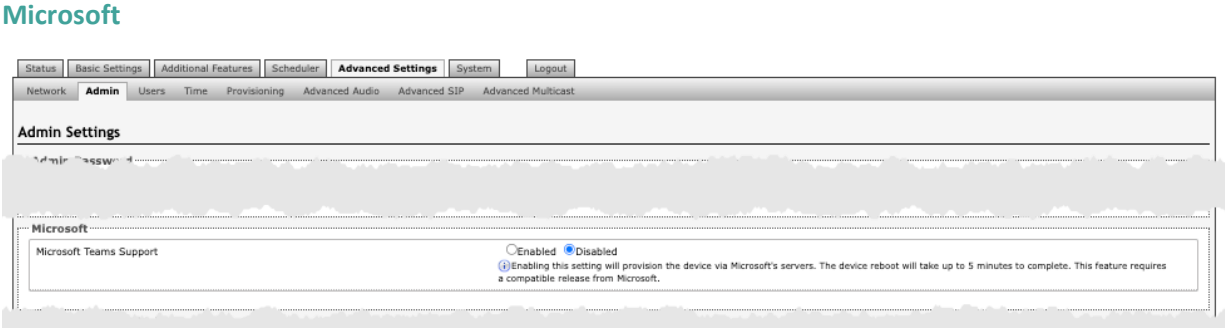
The SA-Announce feature cannot be used when Multicast Transmitter mode or Poly mode is enabled. To enable SA-Announce mode, set **Multicast Mode** to **None** in **Basic Settings** → **Multicast**.



SA-Announce Support	Enable to convert unicast streams to multicast and deliver them to the target endpoints.
SA-Announce Server	Enter the SA-Announce Server to use the Syn-Apps paging feature. Leave the field blank to use the server provided by the DHCP Option 72.
Local Management Port	Enter the local management port for the SA-Announce Server.

7.5 Microsoft Teams

Algo devices are certified by and compatible with Microsoft Teams. When registered in the Microsoft Teams SIP Gateway, the 8305 can be configured to accommodate dozens of applications or deliver Teams-based communication throughout facilities.

	
Microsoft Teams Support	Enable to provision the device via Microsoft's servers. The device reboot will take up to 5 minutes to complete. This feature requires a compatible release from Microsoft.

8 DEVICE MANAGEMENT

8.1 ADMP

The Algo Device Management Platform (ADMP) is a cloud-based device management solution to manage, monitor, and configure Algo IP endpoints from any location. Devices can be easily grouped via a tagging functionality, allowing devices to be coded by district, department, or function to easily oversee many devices. Devices can be supervised for connectivity and email-based notifications can be sent should devices go offline, allowing for a real-time overview of device status.

To connect your 8305 to your ADMP account, use the web interface and navigate to **Advanced Settings → Admin → ADMP Cloud Monitoring**.

Note that if you choose to use ADMP to manage your devices, the Algo 8300 IP Controller cannot be used at the same time.

ADMP Cloud Monitoring StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout NetworkAdminUsersTimeProvisioningAdvanced AudioAdvanced SIPAdvanced Multicast Admin Settings Admin Password ADMP Cloud Monitoring Enable ADMP Cloud Monitoring ☒ Enabled ☐ Disabled <i>(i)</i> This feature requires a valid Account ID. Please contact support@algosolutions.com for assistance. Account ID Allow Configuration File Sync ☐ Enabled ☒ Disabled <i>(i)</i> This feature allows ADMP to query and display settings stored on the device. Heartbeat Interval 30 seconds	
Enable ADMP Cloud Monitoring	The Algo Device Management Platform (ADMP) simplifies the process of managing, monitoring, and maintaining Algo devices from any location. This feature requires a valid Account ID.

8.2 Algo 8300 IP Controller

The Algo 8300 IP Controller is designed for centralized Algo endpoint monitoring and supervision. Any Algo SIP endpoint device, including the 8305, can be monitored on the network via the 8300 dashboard.

Note that if you choose to use the Algo 8300 IP Controller to manage your devices, ADMP cannot be used at the same time.

8.3 SNMP

Simple Network Management Protocol (SNMP) can be used to monitor and manage the 8305.

To configure your SNMP settings, use the web interface and navigate to **Advanced Settings** → **Admin** → **Simple Network Management Protocol**.

Simple Network Management Protocol

Status

Basic Settings

Additional Features

Scheduler

Advanced Settings

System

Logout

Network

Admin

Users

Time

Provisioning

Advanced Audio

Advanced SIP

Advanced Multicast

Admin Settings

Admin Password

Simple Network Management Protocol

SNMP Support

☒ Enabled
 ☐ Disabled
 [Download MIB file here.](#)

SNMP Community String

If left blank, the default string "public" will be used.

SNMPv3 Security

☐ Enabled
 ☒ Disabled

APT Support

SNMP Support	The existing setting will respond to a simple status query for automated supervision.
SNMP Community String	Speak to your IT Administrator for more information.
SNMPv3 Security	Speak to your IT Administrator for more information.

8.4 RTCP

Real-Time Transport Control Protocol (RTCP) can be used to monitor data delivery on the 8305.

To configure your RTCP settings, use the web interface and navigate to **Advanced Settings** → **Admin** → **RTP Control Protocol (RTCP)**.

RTP Control Protocol (RTCP)

StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout

NetworkAdminUsersTimeProvisioningAdvanced AudioAdvanced SIPAdvanced Multicast

Advanced Multicast Settings

RTP Control Protocol (RTCP)

RTCP Port Selection

☒ Disabled
 ☐ Next Higher Port
 ☐ Multiplexed on Same Port

Select the port on which packets will be sent or received.

If using the 'Next Higher Port' option, ensure that the default multicast zone definitions are modified such that zones are only assigned to even-numbered ports, leaving the next higher odd-numbered ports free for RTCP packets.

Basic Zone Definitions

RTCP Port Selection	Select how a port will be chosen to send or receive RTCP packets. Note: If Next Higher Port is selected, ensure that the default multicast zone definitions are modified so that zones are only assigned to even-numbered ports, leaving the next higher odd-numbered ports free for RTCP packets.
---------------------	---

9 SYSTEM CONFIGURATION

9.1 Network Settings

ALGO

8305 Multi-Interface IP Paging Adapter

Status

Basic Settings

Additional Features

Scheduler

Advanced Settings

System

Logout

Network

Admin

Users

Time

Provisioning

Advanced Audio

Advanced SIP

Advanced Multicast

Network Settings

Common

Internet Protocol

IPv4 only

DNS Servers

Use space, comma, or semicolon to separate multiple DNS servers, e.g. 192.168.1.10, 192.168.1.11

IPv4

IPv4 Method

☒ Static
 ☐ DHCP

IPv4 Address/Netmask

Address (dot delimited)/Netmask (CIDR), e.g. 192.168.1.23/24

IPv4 Gateway

802.1Q Virtual LAN

VLAN Mode

☐ None
 ☒ Manual
 ☐ Auto

VLAN ID

0

Value range: 0 to 4094

VLAN Priority

0

Value range: 0 to 7

802.1X Port-based Network Access Control

802.1X Authentication

☒ Enabled
 ☐ Disabled

Authentication Mode

EAP-PEAP/MSCHAPv2

In EAP-TLS mode, if the authentication server requires devices to be authenticated, a PEM file containing both a device certificate and a private key can be installed on the Algo device. Use the "System > File Manager" tab to upload a Base64 encoded X.509 certificate file renamed to 'client8021x.pem' in the 'certs' folder.

Anonymous ID

ID

Password

Validate Server Certificate

☐ Enabled
 ☒ Disabled

Validate the authentication server against common authorities. To validate against additional certificates, use the "System > File Manager" tab to upload a Base64 encoded X.509 certificate file in .pem, .cer, or .crt format to the 'certs/trusted' folder.

Differentiated Services

SIP (6-bit DSCP value)

0

Valid values range from 0 to 63

RTP (6-bit DSCP value)

0

Valid values range from 0 to 63

RTCP (6-bit DSCP value)

0

Valid values range from 0 to 63

DNS

DNS Caching Mode

☒ Disabled
 ☐ SIP
 ☐ All

In "SIP" mode, only the results of DNS queries for SIP requests will be cached. In "All" mode, the results of all DNS queries will be cached.

Save

Figure 22: Network settings

Common

Internet Protocol: IPv4 only

Supersede DNS provided by DHCP: ☒ Enabled ☐ Disabled

DNS Servers: Use space, comma, or semicolon to separate multiple DNS servers, e.g. 192.168.1.10, 192.168.1.11

Internet Protocol	Use the dropdown to select IPv4 Only or IPv4 and IPv6 . If IPv6 is also configured, it will have to be set up via DHCP or statically, similarly to the IPv4.
Supersede DNS provided by DHCP	Use this option to enter a custom DNS server address and supersede the one sent via DHCP.
DNS Servers	Add one or multiple DNS servers when Supersede DNS provided by DHCP is enabled. Separate each server by a space, comma, or semicolon.

IPv4

IPv4 Method: ☒ Static ☐ DHCP

IPv4 Address/Netmask: Address (dot delimited)/Netmask (CIDR), e.g. 192.168.1.23/24

IPv4 Gateway:

IPv4 Method	The 8305 can be set to a static or DHCP IP address. DHCP is an IP standard designed to simplify the administration of IP addresses. When selected, DHCP will automatically configure IP addresses for each 8305 on the network. DHCP is selected by default. When Static is selected, the IP address entered in the fields below will be used by the device.
-------------	---

IPv4 Address/Netmask	Enter the static IP address and netmask (CIDR format) for the 8305 (e.g., 192.168.1.23/24).
IPv4 Gateway	Enter the gateway address.

IPv6

Status

Basic Settings

Additional Features

Scheduler

Advanced Settings

System

Logout

Network

Admin

Users

Time

Provisioning

Advanced Audio

Advanced SIP

Advanced Multicast

Network Settings

Common

Internet Protocol

IPv4 and IPv6

Supersede DNS provided by DHCP

☒ Enabled
 ☐ Disabled

DNS Servers

Use space, comma, or semicolon to separate multiple DNS servers, e.g. 192.168.1.10, 192.168.1.11

IPv4

IPv4 Method

☐ Static
 ☒ DHCP

IPv6

IPv6 Method

☒ Static
 ☐ DHCP

IPv6 Address/Netmask

Address (colon delimited)/Netmask (CIDR), e.g. 2001:123::abcd:1234/64

IPv6 Gateway

IPv6 Method	The 8305 can be set to a static or DHCP IP address. DHCP is an IP standard designed to simplify the administration of IP addresses. When selected, DHCP will automatically configure IP addresses for each 8305 on the network. When DHCP is selected, the DHCP will automatically configure IP addresses for the 8305 on the network.
IPv6 Address/Netmask	Enter the static IP address and netmask (CIDR format) for the 8305 (e.g., 2001:123::abcd:1234/64).
IPv6 Gateway	Enter the gateway address.

ICMPv6 Options

Status

Basic Settings

Additional Features

Scheduler

Advanced Settings

System

Logout

Network

Admin

Users

Time

Provisioning

Advanced Audio

Advanced SIP

Advanced Multicast

Network Settings

Common

Internet Protocol

IPv4 and IPv6

Supersede DNS provided by DHCP

☒ Enabled
 ☐ Disabled

DNS Servers

Use space, comma, or semicolon to separate multiple DNS servers, e.g. 192.168.1.10, 192.168.1.11

IPv4

IPv4 Method

☐ Static
 ☒ DHCP

IPv6

IPv6 Method

☒ Static
 ☐ DHCP

IPv6 Address/Netmask

Address (colon delimited)/Netmask (CIDR), e.g. 2001:123::abcd:1234/64

IPv6 Gateway

ICMPv6 Options

Destination Unreachable messages

☒ Enabled
 ☐ Disabled

Neighbour Discovery Redirect messages

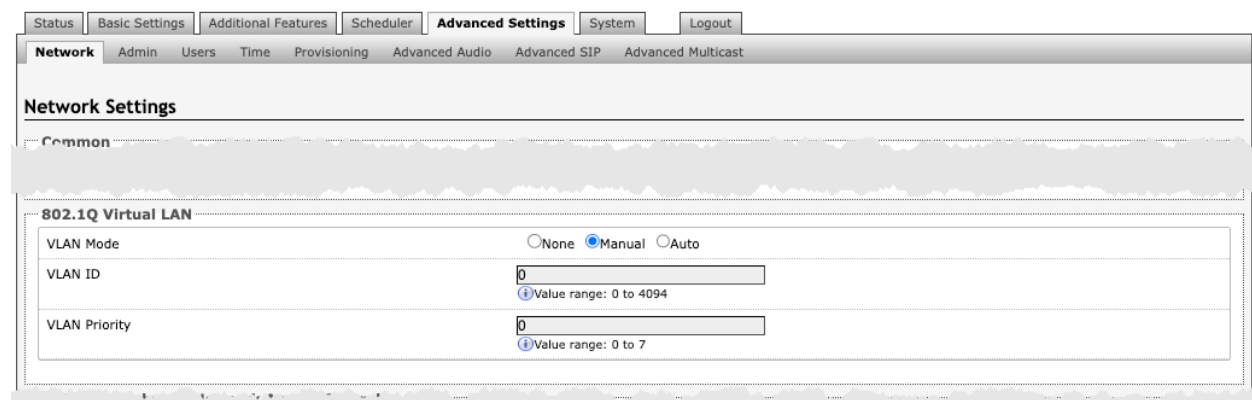
☐ Enabled
 ☒ Disabled

Anycast Echo Replies

☒ Enabled
 ☐ Disabled

Destination Unreachable messages	Enable to restrict traffic by filtering ICMPv6 packets.
Neighbour Discovery Redirect messages	Enable to restrict traffic by filtering ICMPv6 packets.
Anycast Echo Replies	Enable to restrict traffic by filtering ICMPv6 packets.

802.1Q Virtual LAN



Status Basic Settings Additional Features Scheduler **Advanced Settings** System Logout

Network Admin Users Time Provisioning Advanced Audio Advanced SIP Advanced Multicast

Network Settings

Common

802.1Q Virtual LAN

VLAN Mode ☐ None ☒ Manual ☐ Auto

VLAN ID
Value range: 0 to 4094

VLAN Priority
Value range: 0 to 7

VLAN Mode	VLAN tagging is the networking standard that supports Virtual LANs (VLANs) on an Ethernet network. The standard defines a system of VLAN tagging for Ethernet frames and the accompanying procedures to be used by bridges and switches in handling such frames. The standard also provides provisions for a quality-of-service prioritization scheme known as IEEE 802.1p and defines the Generic Attribute Registration Protocol.
VLAN ID	Specify the VLAN that the Ethernet frame belongs to. The hexadecimal values 0x000 and 0xFFFF are reserved. All other values may be used as VLAN identifiers, allowing up to 4094 VLANs. The reserved value 0x000 indicates that the frame does not belong to any VLAN. In this case, the 802.1Q tag specifies only a priority and is referred to as a priority tag.
VLAN Priority	Set the frame priority level. Otherwise known as Priority Code Point (PCP), VLAN Priority is a 3-bit field that refers to the IEEE 802.1p priority or frame priority level. Values are from 0 (lowest) to 7 (highest).

802.1X Port-based Network Access Control

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[Advanced SIP](#)
[Advanced Multicast](#)

Network Settings

802.1X Port-based Network Access Control

802.1X Authentication ☒ Enabled ☐ Disabled

Authentication Mode EAP-PEAP/MSCHAPv2

④ In EAP-TLS mode, if the authentication server requires devices to be authenticated, a PEM file containing both a device certificate and a private key can be installed on the Algo device. Use the "System > File Manager" tab to upload a Base64 encoded X.509 certificate file renamed to 'client8021x.pem' in the 'certs' folder.

Anonymous ID

ID

Password

Validate Server Certificate ☐ Enabled ☒ Disabled

④ Validate the authentication server against common authorities. To validate against additional certificates, use the "System > File Manager" tab to upload a Base64 encoded X.509 certificate file in .pem, .cer, or .crt format to the 'certs/trusted' folder.

802.1x Authentication	Enable to add credentials to access LAN or WLAN that have 802.1X network access control (NAC). You can ask your IT Administrator for this information
Authentication Mode	Select the desired authentication mode.
Anonymous ID	If configured, the 8305 will send the anonymous ID to the authenticator instead of the 802.1X client username.
ID	The ID should contain a string identifying the IEEE 802.1X authenticator originating the request. Ask your IT administrator for details.
Password	Ask your IT administrator for details.
Validate Server Certificate	Enable to validate the authentication server against common authorities. To validate additional certificates, go to the System → File Manager to upload a Base64 encoded X.509 certificate file in .pem, .cer, or .crt format to the 'certs' folder.

Differentiated Services

Differentiated Services provide quality of service if the DSCP protocol is supported on your network. Differentiated Services can be specified independently for SIP control packets and RTP and RTCP audio packets.

The screenshot shows the 'Advanced Settings' tab selected in the top navigation bar. Below it, the 'Network' sub-tab is active. The 'Network Settings' section is expanded, showing the 'Differentiated Services' configuration. It contains three input fields, each with a value of '0' and a help icon. The help text for each field states: 'Valid values range from 0 to 63'.

Field	Value	Help Text
SIP (6-bit DSCP value)	0	Valid values range from 0 to 63
RTP (6-bit DSCP value)	0	Valid values range from 0 to 63
RTCP (6-bit DSCP value)	0	Valid values range from 0 to 63

SIP (6-bit DSCP value)	Enter the DSCP value for SIP packets.
RTP (6-bit DSCP value)	Enter the DSCP value for RTP packets.
RTCP (6-bit DSCP value)	Enter the DSCP value for RTCP packets.

DNS

The screenshot shows the 'Advanced Settings' tab selected in the top navigation bar. Below it, the 'Network' sub-tab is active. The 'Network Settings' section is expanded, showing the 'DNS' configuration. It contains a 'DNS Caching Mode' section with three radio button options: 'Disabled' (selected), 'SIP', and 'All'. A help icon is present next to the 'SIP' option. The help text states: 'In "SIP" mode, only the results of DNS queries for SIP requests will be cached. In "All" mode, the results of all DNS queries will be cached.' A 'Save' button is located at the bottom right of the form.

DNS Caching Mode	There are three mode options: Disabled: No DNS queries will be cached. SIP: Only the results of DNS queries for SIP requests will be cached. All: The results of all DNS queries will be cached.
------------------	--

9.2 Admin

ALGO 8305 Multi-Interface IP Paging Adapter

Status Basic Settings Additional Features Scheduler **Advanced Settings** System Logout

Network **Admin** Users Time Provisioning Advanced Audio Advanced SIP Advanced Multicast

Admin Settings

Admin Password

Old Password

Password

Confirmation

General

Device Name (Hostname)

Introduction Section on Status Page ☒ On ☐ Off

Show Status Section on Status Page when Logged Out ☒ On ☐ Off

Display Switch Port ID on Status Page ☐ On ☒ Off
(i) Requires the device to be connected to a switch that supports LLDP or CDP.

Web Interface Session Timeout
(i) Automatically log out web interface after period of inactivity.

Play Tone at Startup ☐ Enabled ☒ Disabled
(i) A tone can be played at startup to confirm that the device has booted. This can be useful when testing or configuring a device, but might not be desirable if the device is connected to an external amplifier and paging system.

Log Settings

Log Level ☐ Error (Lowest) ☐ Notice ("Event") ☒ Info ("SIP") ☐ Debug (Highest)

Log Method ☒ Local ☐ Network ☐ Both

Management

Web Interface Protocol ☒ Both HTTP and HTTPS ☐ HTTPS Only

Force Strong Password ☐ Enabled ☒ Disabled

Allow Secure SIP Passwords ☐ Enabled ☒ Disabled
(i) After enabling this option, it is recommended to re-enter SIP passwords and their corresponding realm to store the passwords securely.

Simple Network Management Protocol

SNMP Support ☒ Enabled ☐ Disabled
(i) Download MIB file [here](#).

SNMP Community String
(i) If left blank, the default string "public" will be used.

SNMPv3 Security ☐ Enabled ☒ Disabled

API Support

RESTful API ☐ Enabled ☒ Disabled
(i) Secure API for remote access & control via HTTP. Full API documentation available [here](#).

SCI Support

SCI ☐ Enabled ☒ Disabled
(i) Simple Control Interface (SCI) is a separate control interface for certain applications. Its main purpose is to support phones that may have programmable keys that can only send out HTTP GET requests.

System Integrity

System Integrity Checking ☐ Enabled ☒ Disabled
(i) This feature verifies installed system packages to ensure they have not been tampered with. Enabling this feature may cause reboots and upgrades to take 30 seconds longer. Verification results can be found on the Status page.

Syn-Apps

SA-Announce Support
The SA-Announce feature cannot be used when Multicast Transmitter mode or Polycom mode is enabled. To enable SA-Announce mode, set Multicast Mode to None in "Basic Settings > Multicast".

InformaCast

InformaCast Support ☐ Enabled ☒ Disabled
(i) This feature requires a valid license to be activated. Please contact sales@algosolutions.com for assistance.

Microsoft

Microsoft Teams Support ☐ Enabled ☒ Disabled
(i) Enabling this setting will provision the device via Microsoft's servers. The device reboot will take up to 5 minutes to complete. This feature requires a compatible release from Microsoft.

ADMP Cloud Monitoring

Enable ADMP Cloud Monitoring ☒ Enabled ☐ Disabled
(i) This feature requires a valid Account ID. Please contact support@algosolutions.com for assistance.

Account ID

Allow Configuration File Sync ☐ Enabled ☒ Disabled
(i) This feature allows ADMP to query and display settings stored on the device.

Heartbeat Interval

☒ Save

Figure 23: Admin settings.

Admin Password

Use this section to change the admin password for logging into your 8305 web interface. It's recommended that you change the admin password from the default to secure the device on your network.

Old Password	Enter the old admin password. The default password when you first get the device is <i>algo</i> .
Password	Enter a new admin password to log into the 8305 web interface. Make sure the new password is stored safely. If the password is forgotten, you must reset the device entirely with the Reset Button to restore the default password. All other settings will be reset to the original default settings as well. For additional password security, see the setting: Force Strong Password.
Confirmation	Re-enter your new admin password.

General

Status

Basic Settings

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Scheduler

Advanced Settings

System

Logout

Network

Admin

Users

Time

Provisioning

Advanced Audio

Advanced SIP

Advanced Multicast

Admin Settings

General

Device Name (Hostname)

pagingadapter-\$MAC\$

Introduction Section on Status Page

☒ On ☐ Off

Show Status Section on Status Page when Logged Out

☒ On ☐ Off

Display Switch Port ID on Status Page

☐ On ☒ Off

Requires the device to be connected to a switch that supports LLDP or CDP.

Web Interface Session Timeout

1 hour

Automatically log out web interface after period of inactivity.

Play Tone at Startup

☐ Enabled ☒ Disabled

A tone can be played at startup to confirm that the device has booted. This can be useful when testing or configuring a device, but might not be desirable if the device is connected to an external amplifier and paging system.

Device Name (Hostname)	Add a name to identify the device in the Algo Network Device Locator Tool .
Introduction Section on Status Page	Turn On to show the introduction text on the login screen.
Show Status Section on Status Page when Logged Out	Turn On to allow others to view the status page without logging in. If turned Off , the settings and configurations on the status page will be hidden entirely unless a user is logged in to ensure only trusted users can view device information.
Display Switch Port ID on Status Page	Turn On to display the Switch Port ID on the Status Page. This option is only possible if the 8305 is connected to a switch that supports LLDP or CDP.
Web Interface Session Timeout	Set the maximum duration of inactivity to log a user out of the web interface automatically.
Play Tone at Startup	Enable to play a tone at start-up to confirm that the device has booted. This can be useful when testing or configuring a device but might not be desirable if the device is connected to an external legacy communication system and paging system.

Log Settings StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout NetworkAdminUsersTimeProvisioningAdvanced AudioAdvanced SIPAdvanced Multicast Admin Settings Admin Password Log Settings Log Level ☐ Error (Lowest) ☐ Notice ("Event") ☒ Info ("SIP") ☐ Debug (Highest) Log Method ☒ Local ☐ Network ☐ Both	
Log Level	This setting should only be used after consulting with the Algo support team.
Log Method	Select a Log Method: - Local: The log file is saved on the device in RAM. - Method: Send the log file to a server repeatedly so settings are not lost if the device is rebooted. - Both: Use both methods.
Log Server	Enter the Syslog server address provided by your IT administrator.

Management StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout NetworkAdminUsersTimeProvisioningAdvanced AudioAdvanced SIPAdvanced Multicast Admin Settings Admin Password Management Web Interface Protocol ☒ Both HTTP and HTTPS ☐ HTTPS Only Force Strong Password ☐ Enabled ☒ Disabled Allow Secure SIP Passwords ☐ Enabled ☒ Disabled After enabling this option, it is recommended to re-enter SIP passwords and their corresponding realm to store the passwords securely.	
Web Interface Protocol	HTTPS is always enabled on the device. HTTP is enabled by default but may be disabled. To do so, select HTTPS Only mode so requests are automatically redirected to HTTPS.

	Note that no security certificate exists since the device can have any address on the local network. Therefore, most browsers will provide a warning when using HTTPS.
Force Strong Password	When Enabled, you can enforce a secure password for the 8305 web interface for additional protection. The password requirements for a strong password are: • Must contain at least 10 characters • Must contain at least 1 uppercase character • Must contain at least 1 digit (0 – 9) • Must contain at least 1 special character
Allow Secure SIP Password	When Enabled, SIP passwords are stored in the configuration file in an encrypted format to prevent viewing and recovery. If enabled, navigate to Basic Settings → SIP and fill out the field Realm. To obtain your SIP Realm information, contact your SIP Server administrator or check the SIP log file for a registration attempt. The Realms may be the same or different for all the extensions used. All the configured Authentication Password(s) must be re-entered here as well as any other locations where SIP extensions have been configured to save the encrypted password(s). If the Realm is changed later, all passwords must be re-entered to save the passwords with the new encryption.
Display Switch Port ID on Status Page	Turn On to display the Switch Port ID on the Status Page. This option is only possible if the 8305 is connected to a switch that supports LLDP or CDP.

Simple Network Management Protocol

Simple Network Management Protocol

SNMP Support ☒ Enabled ☐ Disabled
 ⓘ Download MIB file [here](#).

SNMP Community String

ⓘ If left blank, the default string "public" will be used.

SNMPv3 Security ☐ Enabled ☒ Disabled

SNMP Support	The existing setting will respond to a simple status query for automated supervision.
SNMP Community String	Speak to your IT Administrator for more information.
SNMPv3 Security	Speak to your IT Administrator for more information.

API Support

API Support

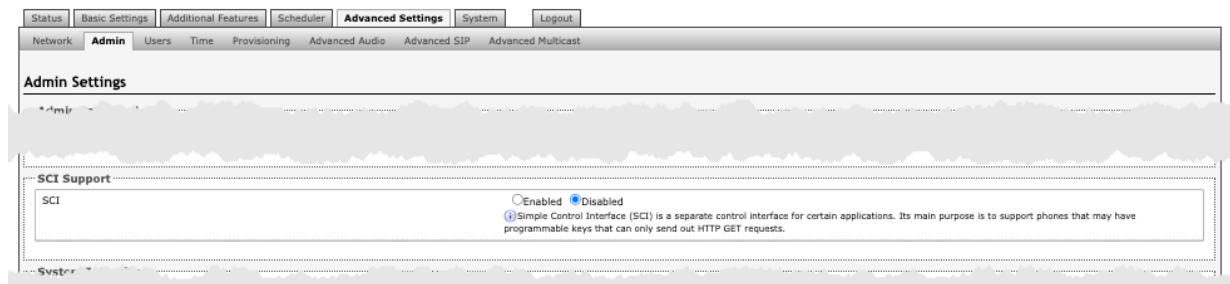
RESTful API ☒ Enabled ☐ Disabled
 ⓘ Secure API for remote access & control via HTTP. Full API documentation available [here](#).

Authentication Method ☒ Standard ☐ Basic ☐ None
 ⓘ RESTful API supports three types of authentication: **Standard** (recommended), **Basic**, and **None** (not recommended).

RESTful API Password

RESTful API	Enable a secure API for remote access and device control via HTTP.
Authentication Method	Speak to your IT Administrator for more information.
RESTful API Password	Speak to your IT Administrator for more information.

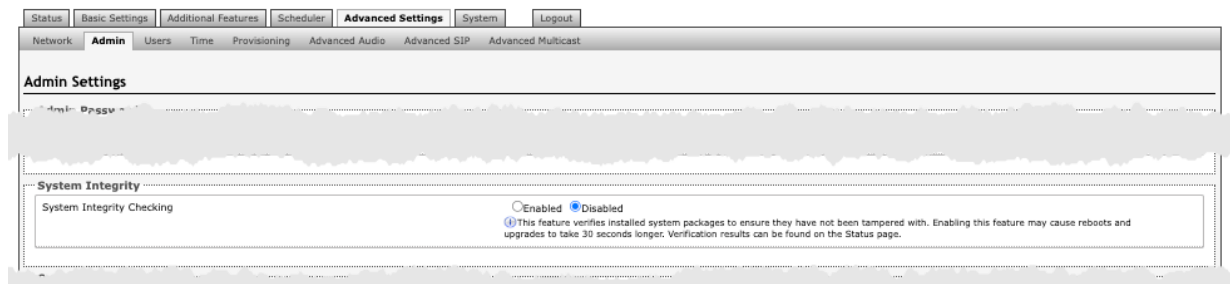
SCI Support



SCI

Simple Control Interface (SCI) is a separate control interface for certain applications. Its primary purpose is to support phones that may have programmable keys that can only send out HTTP GET requests.

System Integrity



System Integrity
Checking

Enable this feature to verify that installed system packages have not been tampered with by running a check. Enabling this feature may cause reboots and upgrades to take 30 seconds longer. Verification results can be found on the **Status** tab.

Syn-Apps

The SA-Announce feature cannot be used when Multicast Transmitter mode or Poly mode is enabled. To enable SA-Announce mode, set **Multicast Mode** to **None** in **Basic Settings** → **Multicast**.

SA-Announce Support	Enable to convert unicast streams to multicast and deliver them to the target endpoints.
SA-Announce Server	Enter the SA-Announce Server to use the Syn-Apps paging feature. Leave the field blank to use the server provided by the DHCP Option 72.
Local Management Port	Enter the local management port for the SA-Announce Server.

InformaCast

InformaCast Support	This feature requires a valid InformaCast license to be activated.
---------------------	--

Microsoft

The screenshot shows the 'Advanced Settings' tab in the web interface. Under the 'Microsoft' section, there is a toggle for 'Microsoft Teams Support' which is currently set to 'Disabled'. A help icon and text explain that enabling this setting will provision the device via Microsoft's servers, requiring a device reboot (up to 5 minutes) and a compatible release from Microsoft.

Microsoft Teams Support

Enable to provision the device via Microsoft's servers. The device reboot will take up to 5 minutes to complete. This feature requires a compatible release from Microsoft.

ADMP Cloud Monitoring

The screenshot shows the 'Advanced Settings' tab in the web interface. Under the 'ADMP Cloud Monitoring' section, there are three settings: 'Enable ADMP Cloud Monitoring' (set to 'Enabled'), 'Account ID' (a text input field), and 'Allow Configuration File Sync' (set to 'Disabled'). Below these is a 'Heartbeat Interval' dropdown menu set to '30 seconds'. A note indicates that enabling ADMP requires a valid Account ID and contact with support@algosolutions.com for assistance.

Enable ADMP Cloud Monitoring

The Algo Device Management Platform (ADMP) simplifies the process of managing, monitoring, and maintaining Algo devices from any location. This feature requires a valid Account ID.

9.3 Users

Use these settings to create a separate user who only has access to the scheduler on the 8305. This may be useful, for example, in a school where you may want someone to have access to bell schedule modification, but don't want them to have access to all 8305 configuration settings.

ALGO 8305 Multi-Interface IP Paging Adapter

Navigation: Status | Basic Settings | Additional Features | Scheduler | **Advanced Settings** | System | Logout

Sub-navigation: Network | Admin | **Users** | Time | Provisioning | Advanced Audio | Advanced SIP | Advanced Multicast

User Management

Scheduler

These settings enable a separate login account with limited access that allows the user to only modify the device scheduler

User Login: ☒ Enabled ☐ Disabled

Username:

Password:

Figure 24: Users settings.

Scheduler	
User Login	Enable to create a separate user who can only access the scheduler for the 8305.
Username	Enter a username
Password	Enter a password

9.4 Time

Time and date are used for logging purposes and the scheduler feature.

ALGO 8305 Multi-Interface IP Paging Adapter

Status Basic Settings Additional Features Scheduler **Advanced Settings** System Logout

Network Admin Users **Time** Provisioning Advanced Audio Advanced SIP Advanced Multicast

Time Settings

General

Timezone: GMT

NTP Time Server 1: *.debian.pool.ntp.org

NTP Time Server 2: *.debian.pool.ntp.org

NTP Time Server 3: *.debian.pool.ntp.org

NTP Time Server 4: *.debian.pool.ntp.org

Supersede NTP provided by DHCP: ☐ Enabled ☒ Disabled

By default, if an NTP Server address is provided via DHCP Option 42, it will be used instead of the NTP servers listed above. Enable this option to ignore DHCP Option 42.

Device Date/Time: Mon Jan 29 23:26:22 2024 [Sync with browser](#)

Manually Override Time: 23:26:17 [Manually Set Time](#)

Manual time and date are intended for testing purpose only. Time will be lost upon power down if NTP server is reachable.

[Save](#)

Figure 25: Time settings.

General	
Timezone	Select a time zone for 8305 settings.
NTP Time Servers 1/2/3/4	The device will attempt to use Timer Server 1 and work down the list if one or more of the time servers become unresponsive. These settings are pre-populated with public NTP servers hosted on the internet. To use these, the device requires internet connection. Alternatively, this can be customized to point the device to any other NTP server hosted or premise-based.
Supersede NTP provided by DHCP	By default, if an NTP Server address is provided via DHCP Option 42, it will be used instead of the NTP servers listed above. Enable this option to ignore DHCP Option 42.
Device Date/Time	This field shows the current time and date set on the device. If you are testing the device on a lab network that does not have access to an external NTP server, click Sync with browser to temporarily set the time on the device.

	This time value will be lost at power down or overwritten if connection to the NTP server is available. Time and date are used for logging purposes and the scheduler feature.
Manually Override Time	Manual time and date are intended for testing purposes only. Time will be lost upon power down if the NTP server is reachable.

9.5 Provisioning

ALGO

8305 Multi-Interface IP Paging Adapter

StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout

NetworkAdminUsersTimeProvisioningAdvanced AudioAdvanced SIPAdvanced Multicast

Provisioning Settings

Mode

Provisioning Mode

☒ Enabled ☐ Disabled

Settings

Server Method

☐ Auto (DHCP Option 66/160/150)
☐ DHCP Option 66 only
☐ DHCP Option 160 only
☐ DHCP Option 150 only
☒ Static

Auto mode automatically checks all 3 DHCP options for an active provisioning server, in the order listed.

Static Server

Download Method

☐ TFTP ☐ FTP ☐ HTTP ☒ HTTPS

Validate Server Certificate

☐ Enabled ☒ Disabled

Validate the server against common certificate authorities. To validate against additional certificates, use the "System > File Manager" tab to upload a Base64 encoded X.509 certificate file in .pem, .cer, or .crt format to the 'certs/trusted' folder.

Auth User Name

Auth Password

Config Download Path

Firmware Download Path

Partial Provisioning

☐ Enabled ☒ Disabled

Allow support for "-i" incremental provisioning files. Disable for enhanced security if not using this feature.

Check-sync Behavior

☒ Always Reboot ☐ Conditional Reboot

If 'Conditional Reboot' is selected, the device will check with the provisioning server and only reboot if new config is found (unless 'reboot=true' is provided as a parameter in the check-sync event).

Sync Start Time

Schedule a time (HH:mm:ss) for the device to perform a sync according to the 'Check-sync Behavior' option above. Leave blank to disable the feature.

Sync End Time

If set, the device will sync at a random time in the window between Start Time and End Time. Setting an End Time earlier than Start Time indicates an overnight period. Leave blank to sync at Start Time exactly.

Sync Frequency

☒ Daily ☐ Selected Days Only

Zero Touch Provisioning

Turn Off ZTP

ZTP is disabled and can only be re-enabled with a factory reset.

Save

Figure 26: Provisioning settings.

GoMomentum.com

94

888.538.3960

Mode StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout NetworkAdminUsersTimeProvisioningAdvanced AudioAdvanced SIPAdvanced Multicast Provisioning Settings Mode Provisioning Mode ☒Enabled ☐Disabled	
Provisioning Mode	Enabling provisioning allows installers to pre-configure 8305 on a network before installation. This is typically done for large deployments to save time and ensure consistent setups. It is recommended that Provisioning Mode be set to Disabled if this feature is not in use. This will prevent unauthorized re-configuration of the device if DHCP is used.

Settings

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Provisioning
Advanced Audio
Advanced SIP
Advanced Multicast

Provisioning Settings

Settings

Server Method

☐ Auto (DHCP Option 66/160/150)
☐ DHCP Option 66 only
☐ DHCP Option 160 only
☐ DHCP Option 150 only
☒ Static
Auto mode automatically checks all 3 DHCP options for an active provisioning server, in the order listed.

Static Server

Download Method

☐ TFTP
☐ FTP
☐ HTTP
☒ HTTPS

Validate Server Certificate

☐ Enabled
☒ Disabled
Validate the server against common certificate authorities. To validate against additional certificates, use the "System > File Manager" tab to upload a Base64 encoded X.509 certificate file in .pem, .cer, or .crt format to the 'certs/trusted' folder.

Auth User Name

Auth Password

Config Download Path

Firmware Download Path

Partial Provisioning

☐ Enabled
☒ Disabled
Allow support for "-" incremental provisioning files. Disable for enhanced security if not using this feature.

Check-sync Behavior

☒ Always Reboot
☐ Conditional Reboot
If 'Conditional Reboot' is selected, the device will check with the provisioning server and only reboot if new config is found (unless 'reboot=true' is provided as a parameter in the check-sync event).

Sync Start Time
Schedule a time (HH:mm:ss) for the device to perform a sync according to the 'Check-sync Behavior' option above. Leave blank to disable the feature.

Sync End Time
If set, the device will sync at a random time in the window between Start Time and End Time. Setting an End Time earlier than Start Time indicates an overnight period. Leave blank to sync at Start Time exactly.

Sync Frequency

☒ Daily
☐ Selected Days Only

Zero Touch Provisioning

Turn Off ZTP
ZTP is disabled and can only be re-enabled with a factory reset.

Save

Server Method

Select a Server Method.

- Auto:** All three DHCP options (66, 160, 150) will be automatically checked for an active provisioning server

	• DHCP Option 66 Only: Only DHCP Option 66 will be checked for a provisioning server • DHCP Option 160 Only: Only DHCP Option 160 will be checked for a provisioning server • DHCP Option 150 Only: Only DHCP Option 150 will be checked for a provisioning server • Static: Only the specified static server will be checked for a provisioning server For provisioning to work with a DHCP option, DHCP must be enabled under Advanced Settings → Network → IPv4.
Static Server	Enter the server address or domain.
Download Method	Select your preferred method for downloading provisioning files. The options are: • TFTP (Trivial File Transfer Protocol) — See MD5 Checksum below for more details. • FTP • HTTP • HTTPS — This may help prevent configuration files from being read by an unwanted third party and having sensitive data stolen. The 8305 configuration files can be automatically downloaded from a provisioning server using DHCP Option 66. This option code (when set) supplies a TFTP boot server address to the DHCP client to boot from. One of two files can be uploaded on the Provisioning Server (for access via TFTP, FTP, HTTP, or HTTPS): • Generic (for all Algo 8305 Multi-Interface IP Paging Adapter) algop8305.conf • Specific (for a specific MAC address) algom[MAC].conf Both protocol and path are supported for Option 66, allowing for http://myserver.com/config-path to be used.
Validate Server Certificate	Enable to verify the server. This checks if the certificate provided by the server is signed by any CAs included in the list of trusted CAs (used by the Debian infrastructure and Mozilla browsers). If a certificate signed by any of these CAs is received, that server will be trusted. This parameter can also be enabled through provisioning: Prov.download.cert = 1

(FTP) Auth User Name	Speak to your IT Administrator for more information.
(FTP) Auth Password	Speak to your IT Administrator for more information.
(HTTP) Auth User Name	Speak to your IT Administrator for more information.
(HTTP) Auth Password	Speak to your IT Administrator for more information.
(HTTPS) Validate Server Certificate	Speak to your IT Administrator for more information.
(HTTPS) Auth User Name	Speak to your IT Administrator for more information.
(HTTP) Auth Password	Speak to your IT Administrator for more information.
Config Download Path	Enter the path where the configuration file is located within the provisioning server (e.g., algo/config/8305).
Firmware Download Path	Enter the path where the firmware file is located within the provisioning server (e.g., algo/firmware/8305).
Partial Provisioning	Enable to allow support for “-i” incremental provisioning files. Disable for enhanced security if this is not required.
Check-sync Behavior	Select Always Reboot to set the 8305 to always reboot despite other settings. Select Conditional Reboot to set the 8305 to check the provisioning server. Only reboot if a new config is found (unless “reboot=true” is provided as a parameter in the check-sync event).
Sync Start Time	Set a time (HH:mm:ss) for the device to perform a sync according to the Check-sync Behavior setting. Leave this blank if not needed.
Sync End Time	If set, the device will sync randomly in the window between Sync Start Time and Sync End Time. Setting an End Time earlier than the Start Time indicates an overnight period. Leave blank to sync exactly at the set start time.
Sync Frequency	Select the sync frequency. Frequency can be set to Daily or Selected Days Only .

Sync Days	Select the days of the week to for syncs to occur.
-----------	--

MD5 Checksum

If using TFTP as a download mode, a **.md5** checksum file must be uploaded to the provisioning server In addition to the **.conf** file. This checksum file is used to verify that the **.conf** file is transferred correctly without error.

To generate a .md5 file, you can use tools such as <http://www.fourmilab.ch/md5>. To use this tool, simply download and unzip the .md5 program in a command prompt. The correct .md5 file will be generated in the same directory. To generate lowercase letters, use the “-l” parameter.

Generating a generic configuration file

This configuration file is device-generic in terms of MAC address and will be used by all connected 8305 devices.

If using a generic configuration file, extensions, and credentials must be entered manually once the 8305 has automatically downloaded the configuration file.

Generating a specific configuration file

The specific configuration file will only be downloaded by the 8305 with the MAC address specified in the configuration file name.

Since all necessary settings can be included in this file, the 8305 will be ready to work immediately after downloading the configuration file. The MAC address of each 8305 can be found on the back label of the unit.

9.6 Maintenance

Figure 27: Maintenance settings.

Backup/Restore Configuration	
Download Configuration File	Save 8305 configuration settings to a text file for backup or to set up a provisioning configuration file.
Restore Configuration File	Restore settings by uploading a backup file.

Restore Configuration to Defaults	Reset all 8305 Multi-Interface IP Paging Adapter settings to factory default values.
-----------------------------------	--

Backup/Restore All User Files

Status

Basic Settings

Additional Features

Scheduler

Advanced Settings

System

Logout

Maintenance

Firmware

File Manager

Tones

System Log

Credits

About

System Maintenance

Backup / Restore All User Files

Backup in zip format includes configuration file and all uploaded files.

Download Backup Zip File

Download

Restore from Backup Zip File

Choose File

No file chosen

Restore

Restore All Settings and Files to Defaults

Restore Defaults and Delete Files

All preloaded and uploaded files, including tone files, will be deleted.

Download Backup Zip File	Download the 8305 configuration settings and uploaded files in File Manager (ex., certificates, licenses, and tones) to a backup ZIP file.
Restore from Backup Zip File	Restore the 8305 configuration settings and files in File Manager (ex., certificates, licenses, and tones) by uploading a backup zip file.
Restore All Settings and Files to Defaults	Reset the 8305 configuration settings. All preloaded and uploaded files, including tone files, will be deleted

Reboot	
Status Basic Settings Additional Features Scheduler Advanced Settings System Logout Maintenance Firmware File Manager Tones System Log Credits About	
System Maintenance	
Reboot Reboot the device Reboot	
Reboot the Device	Reboots the device.

9.7 Firmware

ALGO 8305 Multi-Interface IP Paging Adapter	
Status Basic Settings Additional Features Scheduler Advanced Settings System Logout Maintenance Firmware File Manager Tones System Log Credits About	
Firmware	
Installed Firmware	
Product Firmware	algo-8305-5.5_alpha4
Online Upgrade	
Check for Firmware Updates	Check
Custom Upgrade	
Method	☒ From Local Files ☐ From URL
Signed Firmware File	Choose File No file chosen
Allow Downgrade	☐ Enabled ☒ Disabled Allow product or base firmware to be downgraded to an older patch version. Enabling this option could cause upgrade issues. Please contact support if necessary.
Upgrade	

Figure 28: Firmware settings.

Installed Firmware StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout MaintenanceFirmwareFile ManagerTonesSystem LogCreditsAbout Firmware Installed Firmware Product Firmware algo-8305-5.5_alpha4	
Product Firmware	Displays the current firmware on the device.

Online Upgrade StatusBasic SettingsAdditional FeaturesSchedulerAdvanced SettingsSystemLogout MaintenanceFirmwareFile ManagerTonesSystem LogCreditsAbout Firmware Online Upgrade Check for Firmware Updates Check	
Check for Firmware Updates	Click Check to check for the latest firmware. If the firmware is up to date, Latest Firmware will state Firmware up to date . If your firmware is outdated, the new firmware availability will be listed. Internet connection is required.

Custom Upgrade Status Basic Settings Additional Features Scheduler Advanced Settings System Logout Maintenance Firmware File Manager Tones System Log Credits About Firmware Custom Upgrade Method ☒ From Local Files ☐ From URL Signed Firmware File Choose File No file chosen Allow Downgrade ☐ Enabled ☒ Disabled ⓘ Allow product or base firmware to be downgraded to an older patch version. ⚠ Enabling this option could cause upgrade issues. Please contact support if necessary. Upgrade	
Method	Select a method for firmware upgrades to occur. This can be done From Local Files or From URL .
Signed Firmware File	Use to upgrade firmware from a local file. To do this: To do this, download the firmware file from https://www.algosolutions.com/firmware-downloads/ then upload the file by clicking on Choose File and selecting the firmware file. Click Upgrade at the bottom of the interface. Once the upgrade is complete, you can confirm the firmware version is changed by looking at the top right of the web interface.
Upgrade URL	Instead of downloading the firmware file https://www.algosolutions.com/firmware-downloads/, you may add the download link here instead. Click Upgrade at the bottom of the interface. Once the upgrade is complete, you can confirm the firmware version is changed by looking at the top right of the web interface.
Allow Downgrade	Enable to allow product or base firmware to be downgraded to an older patch version. Enabling this option could cause future upgrade issues.

9.8 File Manager

The 8305 has 1GB of storage space for additional files.

Name	Date	Type	Size
certs	12/31/1969 04:03 PM	Folder	
debug	03/24/2020 10:26 AM	Folder	
license	11/03/2016 10:16 AM	Folder	
tones	12/31/1969 04:04 PM	Folder	
scheduler.db	01/29/2024 02:47 PM	File	32KB
user.conf	01/29/2024 03:22 PM	Text File	14.368KB

Used: 331MB Available: 1.3GB

Figure 29: File manager settings.

certs Folder

If you have enabled **Validate Server Certificate** under **Advanced Settings** → **Advanced SIP** or **Advanced Settings** → **Provisioning** and would like to validate against additional certificates, you can upload them here.

To install a public CA certificate on the Algo 8305, follow the steps below:

1. Obtain a public certificate from your Certificate Authority (Base64 encoded X.509 .pem, .cer, or .crt).
2. Open the **certs** folder in the web interface by going to **System** → **File Manager**.
3. Upload the certificate files into the **certs** folder by clicking **Upload** in the top left corner of the file manager and select the certificate.

debug Folder

If you have any challenges with the device and work with the Algo support team to overcome or fix them, the debug folder will be used. The device will generate files containing information about the device and put them in the debug folder. You do not need to use this folder unless directed to by the Algo support team.

license Folder

If you would like to use Informacast on a device that hasn't been bundled with an Informacast license, you will need to purchase a license and put it into the license folder in the file manager.

tones Folder

Custom audio files may be uploaded to play notifications. Audio files should be stored in the **tones** directory.

Existing files may be modified by downloading the original file, making the desired changes, then uploading the updated file with a different name. To download, right-click the tone and click **Download**.

Audio files must be in the following format:

- WAV or MP3 format
- Smaller than 200 MB

File names must be limited to 32 characters, with no spaces.

9.9 System Log

System log files are automatically created and can assist with troubleshooting if the 8305 does not behave as expected.

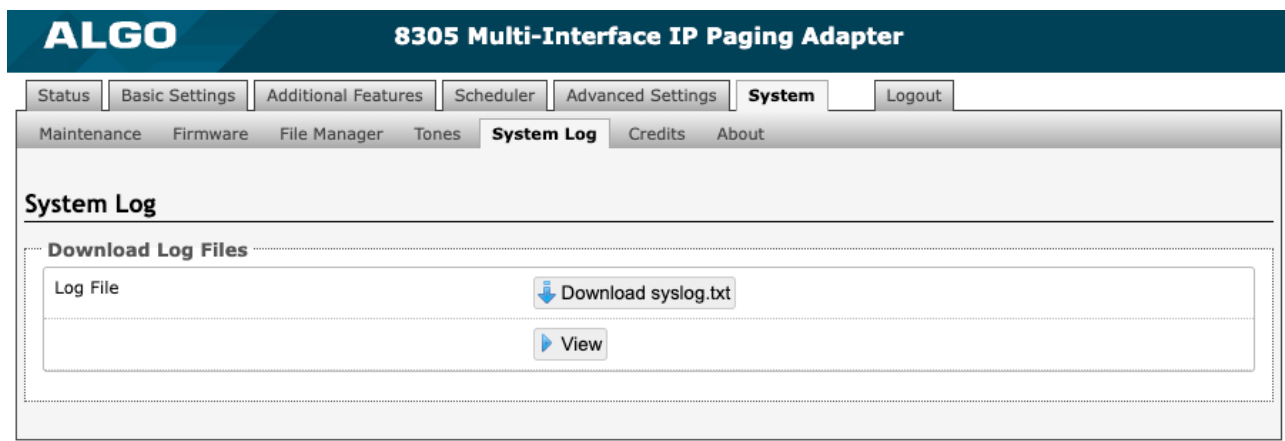


Figure 30: System log settings.

9.10 Logout

Log out of the web interface.

10 FCC COMPLIANCE STATEMENT

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy. If it is not installed and used in accordance with the instruction manual, it may cause harmful interference to radio communications. Operations of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at their own expense.

11 APPENDIX

11.1 Specifications Table

Power	
Power Source	PoE (IEEE 802.3af Class 0) 48V, 15.4W
Max Power (at Device)	4 W
Idle Power	2W
SIP	
SIP Extensions	50 Page 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 Compatibility	
RTP Multicast	Send and Receive 50 Zones
Audio Codecs	G.711 u-law, G.711 A-law, G.722 Wideband, Opus Fullband
Third-Party Compatibility	
	RESTful API
	Poly Group Page
	Singlewire InformaCast (additional license required)
	Syn-Apps Revolution
	Microsoft Teams
Configuration & Provisioning	

Configuration	Web interface or provisioning server		
Web interface	HTTP, HTTPS		
Provisioning	TFTP, FTP, HTTP, HTTPS		
	DHCP Options 66, 150, 160		
	Reboot via SIP 'check-sync'		
ZTP	Supported		
Supervision	Compatible with Algo Device Management Platform (ADMP), Algo 8300 IP Controller, any third-party SNMP monitoring software, and RTCP		
Network			
Network	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			
Audio memory & format	1 GB 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			
Telephone Port	Emulates an analog phone. Capable of ring detection and on-hook/off-hook		
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 30V 50mA (normally open)		
Terminal Block 8 Ω Out	Balanced and isolated wire pair output to external self-amplified speakers. Load impedance of 2 kR down to 8 R*. Max output of +3 dBm @ 8 R, +1.5 dB higher at 2 kR. *Intended use is nominal 2 kR or 1 kR self-amplified speakers connected in parallel to a total minimum resistance of 8 R		
Terminal Block Line Out	Balanced and isolated wire pair output to external legacy communication system. Output level defined using web interface. Polarity independent.		
Aux Out	3.5 mm jack for analog line level input for compatible PC speakers or headset. Non isolated.		
Relay Input Current Draw	Active	Idle	Tamper
Detection Threshold			
Normally Open	>4mA	<4mA	N/A
Normally Open with Supervision	>20mA	4-20mA	<4mA

Normally Closed	<4mA	>4mA	N/A
Normally Closed with Supervision	4-20mA	>20mA	<4mA
Nominal 12V Source, Current Limited to 40mA	Typical supervision resistor value = 1k Ohm		
Environmental & Mechanical			
Environmental	0 to +40° degree C (32 to 104° F), 10-95% RH, non-condensing. Dry indoor locations only.		
Dimensions (Product)	6.75" x 4.3" x 1.18" (17.2cm x 10.9cm x 3.0cm)		
Weight (Product)	0.9lbs (0.4kg)		
Weight (Shipping)	1.5lbs (0.7kg)		
Mounting	Snap mounting bracket included.		
Compliance			
RoHS, CE, FCC Class A, CISPR 22 Class A, CISPR 24, CSA/UL (USA & Canada), EN60950			
Firmware			

These specifications refer to the Algo 8305 running on firmware 5.2 and above.

11.2 Algo Compatible Accessories

The relay input of the Algo 8305 Multi-Interface IP Paging Adapter can be activated by any normally open or normally closed switch, such as Algo input buttons or interfaces. The input switches can be connected to the back of the 8305 via the Terminal Block Relay Input. You can configure the Relay Input Mode on the web interface under the tabs **Additional Features** → **Input/Output**.

11.2.1 1202 Call Button

The 1202 Call Button is a one-touch button for event notification and response. It can be used with the 8305 for improved customer service, emergency notification, and non-emergency alerting. The Call Button's one-touch button can trigger a single or continuous action, which can be halted via the small cancel/reset button located below the main call button.

Configuration

A pair of wires from the terminal block Relay Input on the back of the 8305 can connect to the **center pair** of the modular connector at the back of the Call Button.



Figure 31: 1202 Call Button – the insert card is interchangeable.

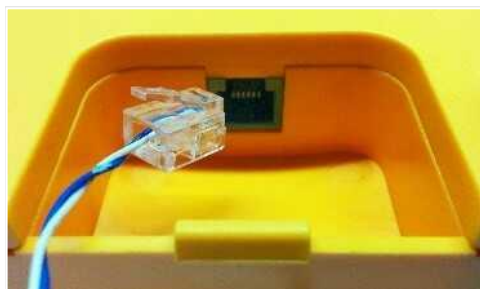


Figure 32: 1202 Call Button wiring.

11.2.2 1203 Call Switch

The 1203 Call Switch is a simple contact closure switch with an illuminated button and supervision capabilities. When used with the 8305, the 1203 can prompt a single action with one touch, or a continuous action if the button is held.

Configuration

A pair of wires can be run from the back of the device via a screw output connector to the 8305 via the Relay Input.



Figure 33: 1203 Call Switch



Figure 34: 1203 Call Switch wiring

11.2.3 Mute Switch

The relay input on the 8305 is also compatible with any third-party switch or button that provides a contact closure (short-circuit) such as a mute switch.

Configuration

Apply an external switch (short-circuit) across the 8305 Relay Input terminals to enable a "disable" switch to control the device. This can be helpful in situations where you only want audio on or off, such as a boardroom to block paging during important meetings.

When using a mute switch, leave the Relay Input terminals open (no-connect) for regular full-volume operation.

After saving the Relay Input Mode to Mute Switch in the web interface, you'll be able to settings for **Multicast Override**. Select one or many zones to override the mute switch settings in these zones.

The screenshot shows the ALGO 8305 web interface with the 'Additional Features' tab selected. The 'Input/Output' section is active, showing the 'General' tab. Under 'Relay Input Mode', the 'Mute Switch' option is selected. Below this, the 'Mute Switch' section explains that an external switch can be used to mute the speaker. The 'Multicast Override' section allows selecting zones to override the mute switch. Under 'Basic Zones Override', 'Priority Call' is checked. Under 'Expanded Zones Override', a grid of checkboxes for zones *10 through *50 is shown, with 'Select All' and 'Clear All' buttons at the bottom. The 'Output' section shows 'Output Relay' is enabled. A 'Save' button is at the bottom right.

Figure 36. Configuring the 8305 when using a mute switch.

11.2.4 1204 Volume Control Switch

The 1204 Volume Control Switch allows a person to control the paging volume.

Configuration

To install the 1204, connect a single twisted pair wire to the Terminal Block Relay Input on the 8305.

Once connected, position 10 on the 1204 will match the maximum volume set in the web interface. Volumes set to lower levels will attenuate from the maximum volume.

The web interface and additional configurations for the 1204 are listed below.

The screenshot displays the 'Input/Output' configuration page of the ALGO 8305 web interface. The page is divided into several sections:

- General:** The 'Relay Input Mode' is set to 'Algo 1204 Volume Control Switch (Local or Remote)'.
- Volume Control Switch Settings:** 'Mute On Lowest Setting' is 'Enabled', and 'Wire Length' is 'Short Cable (0-100ft)'.
- Multicast Override:** 'Basic Zones Override' is 'Priority Call', and 'Expanded Zones Override' includes 'Zone *10' through 'Zone *50'.
- Remote Volume Settings:** 'Remote Mode' is 'Disabled'.
- Output:** 'Output Relay' is 'Enabled'.

A 'Save' button is located at the bottom right of the page.

Figure 37. Configuring the 8305 when using a volume control switch.

Volume Control Switch Settings	
Mute On Lowest Setting	Enable to mute audio when the volume control switch is turned to the lowest setting (1)
Wire Length	Set to calibrate impedance for 24 AWG.
Multicast Override	Multicast Override allows selected multicast zones to override the 1204 Volume Control settings for the selected zones.
Remote Volume Settings	
Remote Mode	Configure the device to subscribe to a remote 1204 volume input or to notify remote devices of 1204 volume input. Note that if Notify (Local 1204) or Subscribe (Remote 1204) are selected that a RESTful API must be enabled under Advanced Settings → Admin.



Figure 38: 1204 Volume Control

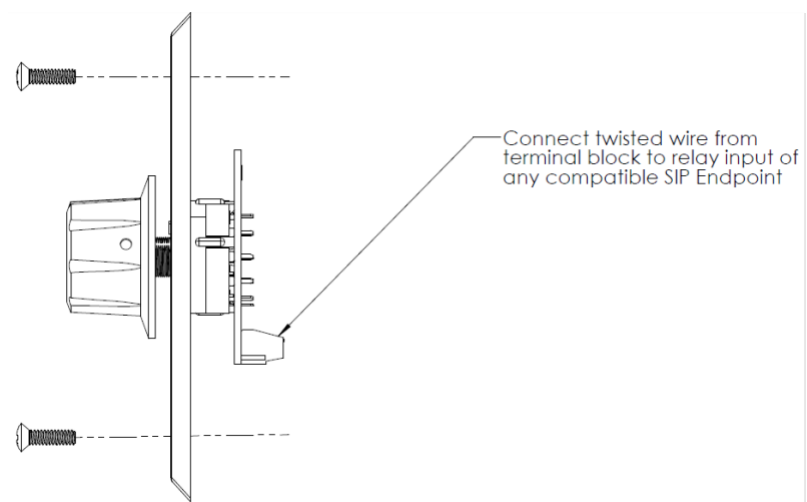


Figure 39: 1204 Volume Control Switch Wiring

11.2.5 2507 Ring Detector

The 2507 Ring Detector can detect when a telephone is ringing and activate the 8305 to play a tone or pre-recorded announcement.

Configuration

Plug an analog telephone into the headset jack on one side of the 2507 Ring Detector and use the other side to connect the to the 8305 using the provided cable.

Once connected, you can use the 8305 web interface to set a Tone/Pre-Recorded Message for when an action is triggered under **Additional Features** → **Input/Output**.



Figure 40. The Algo 2507 Ring Detector.