

8301 Paging Adapter FW Version 1.2.1

Installation & Configuration



Order Codes

8301 Paging Adapter

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Important Safety Information



EMERGENCY COMMUNICATION

If used in an emergency communication application, the 8301 Paging Adapter should be routinely tested. SNMP supervision is recommended for assurance of proper operation.



DRY INDOOR LOCATION ONLY

The 8301 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 compliant network PoE switch must not leave the building perimeter without adequate lightning protection.

No wiring connected to the 8301 Paging Adapter may leave the building perimeter without adequate lightning protection.

About the Algo SIP 8301 Paging Adapter

The 8301 Paging Adapter is a PoE SIP compliant and multicast capable IP adapter designed for integrating consumer, commercial, and professional audio amplifiers into an IP based Unified Communications environment for voice paging and notification.

Dual SIP extensions provide both voice paging and notification (ring) capability. One or both extensions can be registered with any Communication Server (hosted or enterprise) that supports 3rd party SIP Endpoints.

Connection to the amplifier is made using a balanced and isolated line level output provided as both a XLR-mini connection and pluggable terminal block for twisted pair wiring. Audio level can be adjusted manually or set to a defined level independent of input.

Multiple adapters in a SIP environment require only one adapter to register as a SIP extension. Multicasting capabilities allow the SIP registered adapter to page and simultaneously stream multicast audio to the other adapters.

The 8301 Paging Adapter is configured using central provisioning features or by accessing a web interface using browsers such as Google Chrome, Firefox, or Internet Explorer.

What is Included

- 8301 Paging Adapter
- Network Cable
- Wall Mount Bracket

What is not Included

- Optional Wall Switch (Algo 1202 or 1203)
- Optional 2504 Output XLR-Mini Female to XLR Male
- Optional 2505 Input XLR-Mini Male to XLR Female

Typical Application

The 8301 Paging Adapter is typically used to connect an existing paging amplifier to a UC environment either as a SIP extension or multicast endpoint.

The Line Output of the 8301 is connected directly to the dry audio input on an amplifier with an input impedance between 600 Ohm and 10 kOhm.

For amplifiers connected directly to the dry page port of an existing telephone system, the 8301 will provide a very similar interface providing both dry page audio and dry contact closure to activate the amplifier (if required).

For amplifiers connected to a FXS port or ATA through a "telephone answering device" the 8301 will replace the answering device and eliminate the need for a FXS port or ATA.

Getting Started - Quick Install & Test



This guide provides important safety information which should be read thoroughly before permanently installing the adapter.

1. Connect the 8301 Paging Adapter to an IEEE 802.3af compliant PoE network switch. The blue lights on the front will remain on until boot up is completed – about 30 seconds.
2. After the blue lights turn off, the IP address may be discovered by downloading the Algo locator tool to find Algo devices on your network: www.algosolutions.com/locator
3. Connect the adapter LINE OUT to an amplifier using the mini-XLR connector or pluggable terminal block.
4. Access the 8301 Paging Adapter web page by entering the IP address into a browser (Chrome, IE, Firefox etc) and login using the default password **algo**.
5. Enter the IP address for the SIP server into the SIP Domain field under the **BASIC SETTINGS > SIP** tab.
6. Enter the page SIP extension and password. (Note the adapter supports two SIP extensions. The page extension auto-answers for paging. The ring extension plays a WAV file without answering.)
7. Make a call to the adapter by dialling the page SIP extension of the adapter from a telephone.

Installation

The 8301 is wall mountable in a horizontal orientation using the supplied bracket.



Example installation on 1/2" drywall:

Use appropriate drywall anchors for #8 screws, and pre-drill per anchor manufacturer's instructions. Insert 4 anchors into the wall, and then attach bracket to wall anchors using #8 screws. Snap the 8301 into the bracket.

Connect the 8301 to a PoE network switch.

Connect the audio output of the 8301 to an amplifier using either the mini-XLR output (male pins) or pluggable terminal block. The 8301 provides a dry audio output and dry contact closure.

An optional XLR output audio cable (Algo 2504) may be ordered for audio amplifiers using standard XLR input connectors.

Web Interface

The 8301 Paging Adapter is configurable using the web interface or provisioning features.

After boot up the blue lights on the front will turn off and the adapter will have obtained an IP address. If there is no DHCP server the 8301 Paging Adapter will default to the static IP address **192.168.1.111**.

The IP address may be discovered by downloading the Algo locator tool to find Algo devices on your network:

www.algosolutions.com/locator

Enter the IP address (eg 192.168.1.111) into a browser such as Google Chrome, Firefox, or Internet Explorer (other than IE9). The web interface should be visible and the default password will be **algo** in lower case letters.

SIP Paging: One Adapter

The 8301 Paging Adapter can be registered as a third party SIP extension with a hosted or enterprise Communications Server supporting 3rd party SIP endpoints.

To register the adapter with the SIP server, use the **Basic Settings** → **SIP** tab in the web interface to enter the Communication Server IP address, extension, username, and password. This information will be available from the IT Administrator.

If VLAN is used, navigate to the **Advanced Settings** → **Network** tab to set VLAN options.

(Note, once the adapter is using VLAN you will need to be on the same VLAN to access the web interface.)

The adapter may now be accessed by dialling its assigned extension from a telephone, device, or client. The adapter will auto-answer, play the default WAV pre-announce tone, and allow voice paging until disconnected.

There are a number of configurable adapter options:

- Increase or Decrease Speaker Volume
- Enable AGC (automatic gain control)
- Customize pre-announce tone WAV file

The best voice paging quality and intelligibility will be obtained using the G.722 wideband audio codec. Most current IP telephones support G.722 which is sometimes referred to as "HD" voice or audio.

SIP Paging: Multiple Algo SIP Endpoints (Using Multicast)

Multicast features in the 8301 Paging Adapter require that only the first adapter be registered as a SIP extension. Additional Algo SIP endpoints may be added as multicast slaves receiving a stream from the SIP registered adapter, provided that only a single audio stream will be active at any given time across any or all of the devices. If multiple unique audio streams are needed simultaneously more than one Master device will be required.

The SIP registered adapter will simultaneously stream audio to the slave adapters. The slave adapters do not require SIP extensions and do not need to register with the SIP Communication Server.

To enable multicast streaming from the Master adapter, go to the web interface and navigate to the **Basic Settings → Multicast** tab. Choose multicast mode "Master/Sender" and pick "All Call" for the master single zone. The multicast addresses pre-populated in the table will work in most cases.

To enable multicast monitoring in the Algo SIP endpoint slaves, go to the web interface for each endpoint and navigate to the **Basic Settings → Multicast** tab. This time though, choose multicast mode "**Slave/Receiver**". There is no need to select a zone as the endpoint will automatically monitor the "**All Call**" zone IP address.

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

SIP Paging: Multiple SIP Endpoints (Using Individual SIP extensions)

In some cases it may be desirable for every adapter to have a SIP extension. Multicast may still be used to page multiple Algo SIP endpoints but each endpoint can also be called individually or generate a call when appropriately configured.

An Algo SIP endpoint configured as a SIP Multicast Slave will give its highest priority to a page using its SIP extension.

Communication Servers with the capability of dialling many SIP extensions simultaneously for paging may be able to create zones by call hunt groups and also page telephone speakers in conjunction with overhead endpoints.

Multicast Page Zones

The 8301 Paging Adapter supports nine "basic" multicast zones. These "zones" are defined by the multicast IP addresses.

Somewhat arbitrarily, these zones are defined below but may be used in other ways. The important consideration is that there is a priority hierarchy – streaming activity on a zone higher on the list will be treated as a higher priority than a zone lower on the list – with music being the lowest priority.

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

In a scenario involving only one adapter registered as a SIP extension, the multicast page zone can be selected by keypad input using "DTMF Selectable Mode".

Alternatively, several SIP extensions can be registered on the device to allow direct-dial access to each zone. No keypad zone selection in

that case is necessary – the zones are defined by the SIP extension called.

“Expanded” zones can also be enabled, allowing up to 50 zones in total. These have the same behaviours as the basic zones, but are hidden by default to simplify the interface.

Polycom™ Group Paging

The 8301 Paging Adapter has been designed to support Polycom Group Paging.

The 8301 Paging Adapter can be added to a Polycom group page so that voice paging is heard over Polycom telephone speakers and overhead paging simultaneously.

Polycom Group Paging can be configured on the **Basic Settings** → **Multicast** tab.



The 8301 Paging Adapter may be accessed remotely via SIP and may generate a multicast page within the LAN sending voice page to both Algo paging endpoints and Polycom telephones. Audio delay may be added to the 8301 Paging Adapter to synchronize with voice page over the Polycom telephone speakers.

SIP Activated Notification

In addition to voice paging, the 8301 Paging Adapter can play audio files for emergency, safety, or security announcements.

Audio WAV files can be stored in adaptor memory and played over a speaker in response to an event such as a ring and multicast to other Algo SIP endpoints on the network.

SIP Ring Event

Set Monitoring Mode to "Monitor Ring". When a call is made to the SIP extension the 8301 Paging Adapter will play the selected WAV file from memory. Often, the 8301 Paging Adapter will be part of a hunt group or ring group to ring in conjunction with a telephone.

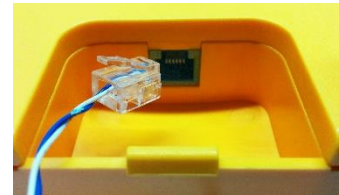
Wiring Connections

Connecting Input Devices to 8301

The input relay to the Algo 8301 Paging Adapter can be prompted by any normally open, normally closed switch, Algo 1202 Call Button, or Algo 1203 Call Switch. The input switches can be connected to the back of the 8301 via a Terminal Block on the "Relay SW Input" pair.



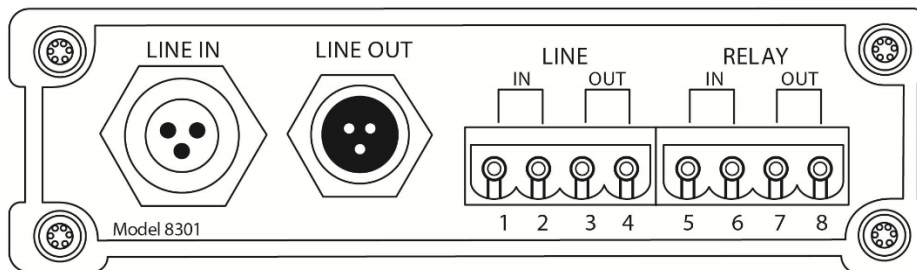
1202 SIP Call Button: A pair of wires from the terminal block on the back of the 8301 Paging Adapter can connect to the **centre pair** of the modular connector at the back of the Call Button.



1203 Call Switch: A pair of wires can be run from the back of the device via a screw output connector to the 8301 Paging Adapter via the screw input connector.



8301 Paging Adapter: Front View



8301 Paging Adapter: Back View

Network Connection

The adapter provides a RJ45 jack for network connection. A cable run from the switch can be terminated to a modular jack with connection by patch cord, or terminated with a RJ45 plug.

PoE (Power over Ethernet) must be 48V 350 mA IEEE 802.3af compliant whether provided by the network switch or injector.

AUX IN (Front)

Analog line level input from iPod or similar device for music input. Non-isolated.

AUX OUT (Front)

Analog line level output for compatible PC speakers or headset. Non-isolated.

LINE IN XLR-MINI (Back)

Balanced and isolated audio (page or music) input can be configured for pass-through to Line Out (when paging is idle), or for broadcast via multicast.

LINE OUT XLR-MINI (Back)

Balanced and isolated audio output to external amplifier. Locking mini-XLR female to standard XLR male cable available. Output level defined using web interface.

Terminal Block Line In

Wire pair input parallel to XLR-MINI LINE IN

Terminal Block Line Out

Wire pair output parallel to XLR-MINI LINE OUT

Terminal Block Relay In

By default, these terminals are inactive. Connection options are a normally closed switch, normally open switch, 1202 Call Button, 1203 Call Switch, or EOL resistor termination.

Terminal Block Relay Out

By default these terminals provide a contact closure when the 8301 Paging Adapter is active.

Blue LED Indicators

All 4 blue lights will be on during power up and boot process.

SIP

Steady light will appear when the SIP extension is registered. The light will blink when the device is engaged in a SIP page/ring.

Multicast

Steady light will appear when 8301 receives multicast messages as a slave. The light will blink when 8301 sends output to the slaves as a master.

Input

Input light is on when receiving audio input.

Output

Output light is on when analog output is enabled.

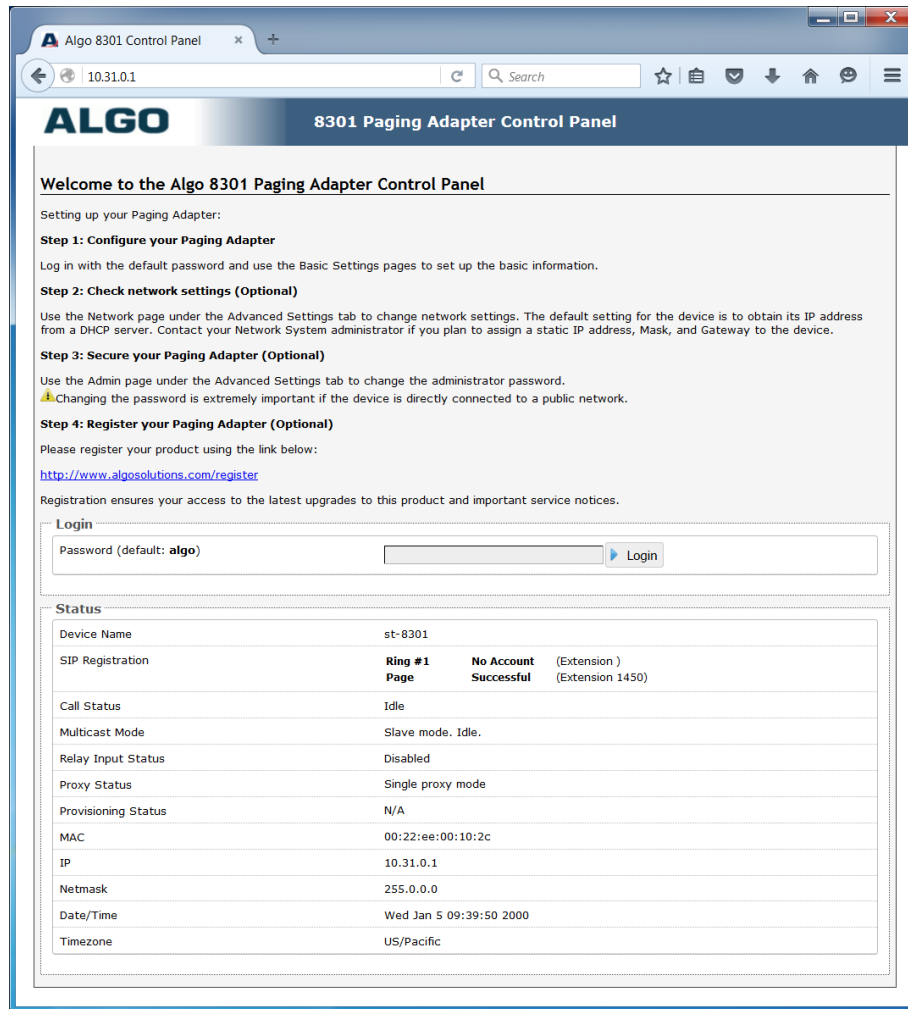
Reset

A recessed reset button (RST) next to the Ethernet Jack can only be used to reset the 8301 Paging Adapter at time of power up. To reset, reboot or power cycle the 8301 Paging Adapter. Wait until the power LED flashes and then press and hold the reset button until the blue LED begins a double flash pattern. Release the reset button and allow the unit to complete its boot process. **Do not press the reset button until the blue power LED begins flashing.**

A reset will set all configuration options to factory default including the password.

Web Interface Login

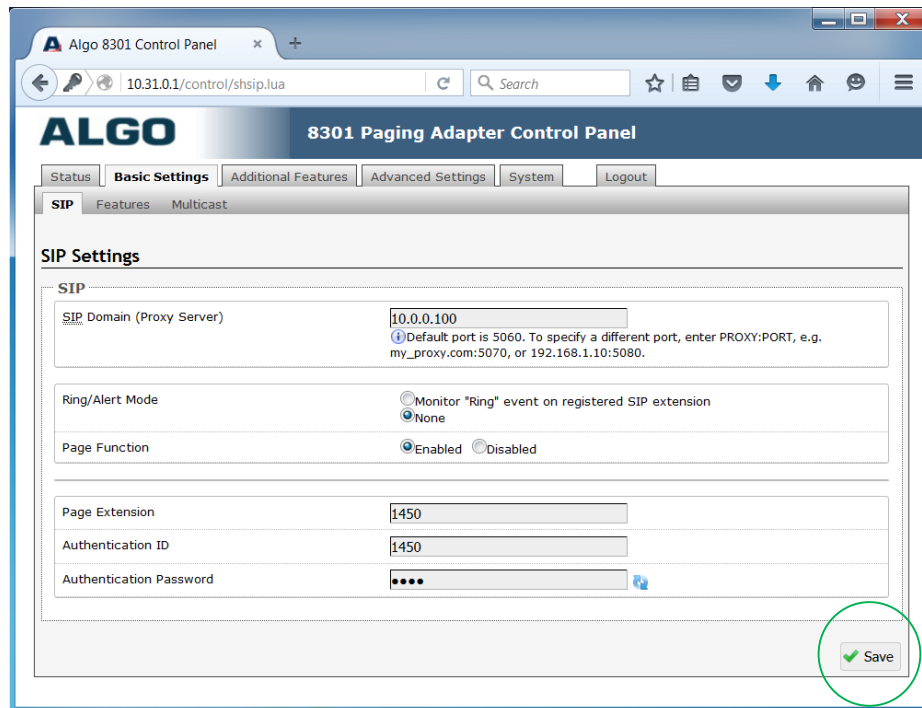
The web interface requires a password which is "**algo**" by default. This password can be changed using the *Admin* tab after logging in the first time.



Status

The device's Status page will be available before and after log on. The section can be used to check 8301's SIP Registration status of the Ring/Page extensions, Call Status, Multicast Mode (Slave/Master), Relay Input Status, Proxy Status, and general MAC, IP, Netmask, Date/Time, and Timezone information.

Basic Settings Tab – SIP



Note: Any time changes are made to settings in the Web Interface the "Save" key must be clicked to save the changes

SIP Domain (Proxy Server)

SIP Server Name or IP Address

Ring/Alert Mode

Option for adding a second SIP extension for ring detection and playing WAV file. If activated, screen expands to enter second SIP extension parameters.

Page Function

Enable or Disable SIP page extension

Page Extension

This is the SIP extension for the 8301 Paging Adapter.

Authentication ID

May also be called Username for some SIP servers and in some cases may be the same as the SIP extension.

Authentication Password

SIP password provided by the system administrator for the SIP account.

Basic Settings Tab – Features

The screenshot shows the 'Basic Settings' tab with the 'Features' sub-tab selected. The 'Features' section is divided into three main areas:

- Inbound Ring Settings:** Ring Tone is set to 'warble2-med.wav' with Play, Loop, and Stop buttons. Ring Speaker Volume is set to 10 with an Apply button. Ring Limit is set to 'No limit' with a note: '1 ring = 6 seconds.'
- Inbound Page Settings:** Page Speaker Volume is set to 10 with an Apply button. Page Mode is set to 'One-way'. Page Timeout is set to '5 minutes' with a note: 'Maximum page timeout in Delayed mode is 5 minutes.' Page Tone is set to 'page-notif.wav' with a note: 'Use only "page-notif.wav", 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 is set to 'Enabled'.
- Audio:** Automatic Gain Control (AGC) is set to 'Enabled' with a note: 'Automatically maximize level of voice received from calling phone in order to make page volume more consistent.' 'Line Out' Analog Output Level is set to '+4dBu 10k (1.23 Vrms)'. A 'Save' button is located at the bottom right.

Ring Tone (Test Tone)

Select WAV file to play when a ring event is detected on the ring SIP extension. The WAV file may be played immediately to an associated speaker from the web interface for test purposes using the Play, Loop, and Stop buttons.

The "Ring Tone" field will be available only when the "Ring/Alert Mode" is enabled in the SIP tab. The "Test Tone" field will appear when the Ring Mode is not enabled, to test the WAV tone over the associated speaker.

Ring Speaker Volume (Test Speaker Volume)

Set speaker volume for SIP ring event. This setting is an amplifier gain control and the output level will also depend on the levels recorded into the source WAV file played from memory.

The "Ring Speaker Volume" field will be available only when the "Ring/Alert Mode" is enabled in the SIP tab. The "Test Speaker Volume" field will be available when the Ring Mode is not enabled, to test the volume over the associated speaker.

Ring Limit

Typically set to no limit, this feature can be used to set a limit on how long the associated speaker will ring before timing out. A new ring event is required before the speaker will play the WAV file again.

Page Speaker Volume

Speaker page volume control for SIP or multicast paging. This setting is an amplifier gain control and output level will depend on streaming level.

Page Mode

A call to the SIP page extension can be one-way or delayed. In delay mode, the adapter will store the page into memory and then play after disconnect.

Page Timeout

A time limit may be set for an active page.

Page Tone

Select pre-announce tone for paging.

G.722 Support

Enable or disable the G.722 codec.

Automatic Gain Control (AGC)

Normalizes the audio level.

'Line Out' Analog Output Level

Select an audio output level:

+4dBu 10k (1.23 Vrms)
0dBu 10k (0.775 Vrms)
0dBV 10k (1.0 Vrms)
-10dBV 10k (0.316 Vrms)
0dBm 600 ohm (0.755 Vrms)
-10dBm 600 ohm (0.245 Vrms)

Multicast IP Addresses

Each 8301 Paging Adapter has its own IP address, and shares a common multicast IP and port number (multicast zone) for multicast packets. The master transmits to a configurable multicast zone, and the slave units listen to all the multicast zones assigned to them.

The network switches and router see the packet and deliver it to all the members of the group. The multicast IP and port number must be the same on all the master and slave units of one group. 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: 1 to 65535
3. By default, the 8301 Paging Adapter is set to use the multicast IP address 224.0.2.60 and the port numbers 50000-50008

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

Basic Settings Tab - Multicast (Master Settings)

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

SIP | Features | **Multicast**

Multicast Settings

Multicast Mode

Multicast Mode: None Master/Sender Slave/Receiver
ⓘ Multicast Zone Definitions can be found in "Advanced Settings > [Advanced Multicast](#)".

Number of Zones: Basic Zones Only Basic and Expanded Zones

Master Output Codec:

Master Output Packetization Time (milliseconds):

Polycom Group Paging/Push-to-Talk

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

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

Polycom Default Channel:

Master/Sender Zone Settings

Zone Selection Mode: DTMF Selectable Zone Single Zone

Master Single Zone:
ⓘ 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 Relay triggered events, including the analog audio input.

Speaker Playback Zones: Priority Call All Call Music
 Zone 1 Zone 2 Zone 3
 Zone 4 Zone 5 Zone 6
ⓘ Allows master 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 Master unit a member of only certain zones.

Expanded Speaker Playback Zones: Zone *10 Zone *11 Zone *12 Zone *13 Zone *14
 Zone *15 Zone *16 Zone *17 Zone *18 Zone *19
 Zone *20 Zone *21 Zone *22 Zone *23 Zone *24
 Zone *25 Zone *26 Zone *27 Zone *28 Zone *29
 Zone *30 Zone *31 Zone *32 Zone *33 Zone *34
 Zone *35 Zone *36 Zone *37 Zone *38 Zone *39
 Zone *40 Zone *41 Zone *42 Zone *43 Zone *44
 Zone *45 Zone *46 Zone *47 Zone *48 Zone *49
 Zone *50

Multicast Mode (Master/Sender Selected)

If master is enabled the 8301 Paging Adapter will broadcast an IP stream when activated in addition to playing the audio. (Note that the 8301 Paging Adapter cannot be both a multicast master and slave simultaneously).

Number of Zones

Select “basic” zones if configuring nine or fewer multicast zones or “expanded” to configure up to 50 zones. The expanded zones have the same behaviour as the basic speaker playback zones, but are hidden by default to simplify the interface.

Master Output Codec

Audio encoding format used by the master device when sending output to the slaves.

Master Output Packetization Time (milliseconds)

The size of the audio packets sent by the master to the slaves. 20ms recommended, unless a different value is specifically required for compatibility with other devices.

Multicast Type

The 8301 Paging Adapter may broadcast multicast paging, compatible with Polycom “**on premise group paging**” protocol and most multicast-enabled phones that use RTP audio packets.

Select “Regular” if solely multicasting to Algo SIP endpoints and/or multicast-enabled phones.

To multicast page announcements solely to Polycom phones, select “Polycom Group Page” or “Push-to-Talk”. Then, configure the 8301 with “Polycom Zone” (IP Address and Port) and “Polycom Default Channel”. *Always ensure that the multicast settings on all Slaves match those of the Master.*

Select “Regular RTP + Polycom Group Page/Push-to-Talk” to multicast page audio to both Polycom phones, Algo SIP endpoints, and multicast-enabled phones.

Zone Selection Mode

Select “DTMF Selectable Zone” to simultaneously ring/page multiple multicast zones, including the specified master single zone (below). Select “Single Zone” to ring/page solely the master single zone.

Master Single Zone

Select the multicast zone to broadcast on. If DTMF Selectable Zone is chosen, then SIP calls will use the zone selected by DTMF on each call, but the specified "Master Single Zone" setting is still used for any multicast events triggered by the analog input or the relay input.

Speaker Playback Zones

Allows master 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 Master unit a member of only certain zones.

Basic Settings Tab - Multicast (Slave Settings)

Status Basic Settings Additional Features Advanced Settings System Logout

SIP Features **Multicast**

Multicast Settings

Multicast Mode

Multicast Mode None Master/Sender Slave/Receiver
① Multicast Zone Definitions can be found in "Advanced Settings > [Advanced Multicast](#)".

Number of Zones Basic Zones Only Basic and Expanded Zones

Audio Delay (milliseconds, 0 ~ 1000)
① Use this feature to synchronize audio output if using multicast with other third-party devices that have a delay in their audio path. Applies to Multicast Slave mode only.

Polycom Group Paging/Push-to-Talk

Multicast Type Regular (RTP) Polycom Group Page Polycom Push-to-Talk
① Regular mode uses RTP audio packets compatible with all Algo SIP endpoints, and most multicast-enabled phones.

Slave/Receiver Zone Settings

Basic Slave Zones Priority Call All Call Music
 Zone 1 Zone 2 Zone 3
 Zone 4 Zone 5 Zone 6

Expanded Slave Zones Zone *10 Zone *11 Zone *12 Zone *13 Zone *14
 Zone *15 Zone *16 Zone *17 Zone *18 Zone *19
 Zone *20 Zone *21 Zone *22 Zone *23 Zone *24
 Zone *25 Zone *26 Zone *27 Zone *28 Zone *29
 Zone *30 Zone *31 Zone *32 Zone *33 Zone *34
 Zone *35 Zone *36 Zone *37 Zone *38 Zone *39
 Zone *40 Zone *41 Zone *42 Zone *43 Zone *44
 Zone *45 Zone *46 Zone *47 Zone *48 Zone *49
 Zone *50

Select All Clear All

Save

Multicast Mode (Slave Selected)

If slave mode is enabled the 8301 Paging Adapter will activate when receiving a multicast message.

Number of Zones

Select "basic" zones if configuring nine or fewer multicast zones or "expanded" to configure up to 50 zones. The expanded zones have the same behaviour as the basic slave zones, but are hidden by default to simplify the interface.

Audio Delay

When paging to the 8301 Paging Adapter as well as other third party devices, the low latency of the 8301 Paging Adapter may cause the audio to lead other devices. By adding audio delay up to one second, the 8301 Paging Adapter may be synchronized with other endpoints or telephones that have greater latency.

Multicast Type - Regular

Select "Regular" if solely multicasting to Algo SIP endpoint(s) and/or multicast-enabled phone(s) that use RTP audio packets.

Slave Zones

Select multicast slave zones for the 8301 Paging Adapter to monitor. Note that multicast zone priority is based on the zone definition list order (top to bottom).

Status Basic Settings Additional Features Advanced Settings System Logout

SIP Features **Multicast**

Multicast Settings

Multicast Mode

Multicast Mode None Master/Sender Slave/Receiver
Use this feature to synchronize audio output if using multicast with other third-party devices that have a delay in their audio path. Applies to Multicast Slave mode only.

Audio Delay (milliseconds, 0 ~ 1000)
Use this feature to synchronize audio output if using multicast with other third-party devices that have a delay in their audio path. Applies to Multicast Slave mode only.

Polycm Group Paging/ Push-to-Talk

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

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

Polycm Default Channel

Polycm Priority Channel

Polycm Emergency Channel

Multicast Type – Polycm Group Paging/ Push-to-Talk

The 8301 Paging Adapter may receive multicast paging compatible with Polycm **“on premise group paging”** protocol.

To configure the 8301 as a slave to play Polycm page announcements, select “Group Page” or “Push-to-Talk”. Then enter the Polycm Zone (IP Address and Port) that matches the configuration of the Polycm phones and Channels. The “Default Channel” is the target group in a Polycm paging environment.

The Polycm phone used as page audio source for the 8301 Paging Adapter(s) must be configured to use either the G.711 or G.722 audio codec. **The Polycm phone(s) must also be configured with the “Compatibility” setting (“ptt.compatibilityMode”) disabled** in order for this codec setting to be applied.

Additional Features Tab – Input/Output

When triggered by an input relay, Algo 8301 Paging Adapter can perform actions such as playing a pre-recorded announcement over the speaker(s), sending the announcement as a private message to a phone, or initiating a two-way conversation between the speaker and a phone.

The input relay to the Algo 8301 Paging Adapter can be prompted by any normally open or normally closed switch. Algo offers the 1202 Call Button or the 1203 Call Switch with supervision. Via supervision settings, notification actions can also be triggered if the input switch is disconnected.

The 8301 Paging Adapter can execute the following actions when triggered by an input relay:

- Play Tone
- Make SIP Voice Call
- Make SIP Call with Tone
- Stream Audio

Algo 1202 Call Button



The Algo 1202 Call Button is a one-touch button for event notification and response. It can be used with the 8301 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 button located below the main call button.

Algo 1203 Call Switch



The Algo 1203 Call Switch is a simple contact closure switch with an illuminated button and supervision capabilities. When used in conjunction with the 8301, the 1203 can prompt a single action with one-touch, or a continuous action if the button is held.

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

Input/Output More Page 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 Speaker Mute Call Button

Audio Streaming

Audio Always On Enabled Disabled
! "Audio Always On" will play sound on local speaker as well as multicast if configured.

Audio Input Port

Action When Input Triggered

Action Play Tone Make SIP Voice Call Make SIP Call with Tone Stream Audio
! "Play Tone" and "Stream Audio" will play sound on local speaker as well as multicast if configured.

Dialing Extension
! SIP account required in Page Extension fields in order to make a call.

Allow 2nd Button Press to End & Restart Call Enabled Disabled

Tone/Pre-recorded Announcement

Interval Between Tones (seconds)

Maximum Tone Duration

Action When Tamper Detected

Action Play Tone Make SIP Voice Call Make SIP Call with Tone
! "Play Tone" and "Stream Audio" will play sound on local speaker as well as multicast if configured.

Dialing Extension
! SIP account required in Page Extension fields in order to make a call.

Outbound SIP Call Settings

Outbound Ring Limit
! 1 ring = 6 seconds

Ringback Tone Enabled Disabled

Maximum Call Duration

Action – Play Tone

When the 8301 receives input, a tone or a pre-recorded WAV file will play over speaker, or multicast if enabled. This function can be used to call support in service environments, notify about an emergency at a specific location in medical/educational facilities, or sound an alarm during an intrusion. Play Tone has the following options:

- Action When Input Triggered:
 - Tone/Pre-recorded Announcement
 - Tone Duration
 - Interval Between Tones (seconds)
 - Maximum Tone Duration

Action - Make SIP Voice Call

Upon receiving input, a voice path will open for an intercom-like call via a microphone (connected to AUX IN) to a pre-configured phone extension. This option can be used when a call needs to be made from a public place where a phone would not be practical to use. Make SIP Voice Call has the following options:

- Action When Input Triggered:
 - Dialing Extension
 - Allow 2nd Button Press to End & Restart Call
 - Audio Input Port
- Outbound SIP Call Settings:
 - Outbound Ring Limit
 - Ringback Tone
 - Maximum Call Duration

Action - Make SIP Call with Tone

An input can also generate a private call to a pre-configured phone extension with a pre-recorded message. For instance, a call to a supervisor's phone notifying about an emergency or intrusion at some location. Make SIP Call with Tone has the following options:

- Action When Input Triggered:
 - Dialing Extension
 - Allow 2nd Button Press to End & Restart Call
 - Tone/Pre-recorded Announcement
 - Interval Between Tone (seconds)
 - Maximum Tone Duration
- Outbound SIP Call Settings:
 - Outbound Ring Limit
 - Ringback Tone
 - Maximum Call Duration

Action – Stream Audio

Will play sound from audio input on a local speaker as well as multicast if configured.

Action When Tamper Detected (Supervision)

In addition to the main events, the device can be configured with supervision to also execute one of the above three actions, except Stream Audio, in case the input switch is disconnected 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 phone extension. The supervision configuration options

will appear once a relay option with supervision is selected. See the Electrical Specification section for details on supervision detection circuit.

Speaker Mute

Allows contact closure to temporary disable the speaker for events like meetings or presentations.

Dialing Extension

SIP account required in Page Extension fields in order to make a call.

Interval Between Tones

Specify the time delay (seconds) between tones.

Maximum Tone Duration

Select the maximum tone duration. The tone will be terminated once the maximum time is reached.

Outbound Ring Limit

Typically set to ensure that a call will not reach voicemail. This feature can be used to set a limit on how long the speaker will ring before timing out.

Ringback Tone

If enabled, a ringback tone will play over the speaker during an outbound SIP call, while waiting for the far-end party to answer.

Maximum Call Duration

Select the maximum call length. The call will be terminated once the maximum time is reached. In the event that a call inadvertently reaches voicemail or gets accidentally left on hold, this setting ensures that the 8301 returns on-hook.

Audio Always On

Enable or disable the local speaker to always play a sound on the local speaker and multicast if configured.

Note: Audio Streaming Always On cannot be used when the relay trigger action is set to "Stream Audio" or "Make SIP Voice Call". To enable Audio Streaming Always On, set Relay Input Mode to "Disabled", or set Action When Input Triggered to "Play Tone" or "Make SIP Call with Tone".

Audio Input Port

Select an audio input port and choose if it is normally open/closed or a multicast master. "Aux Input" is the front blue 3.5mm jack for headsets (iPods/iPhones). "Aux Output" is the front green 3.5 headset with microphone (headsets or PC speakers). "Line Input" is the back terminal black and XLR.

Additional Features Tab – More Page Extensions

The screenshot shows the 'More Page Extensions' configuration page. At the top, there are navigation tabs: Status, Basic Settings, **Additional Features**, Advanced Settings, System, and Logout. Below these are sub-tabs: Input/Output and **More Page Extensions**. The main content is titled 'More Page Extensions' and is divided into two sections: 'Basic Extensions' and 'Expanded Extensions'. Each extension entry includes a radio button for 'Enabled' or 'Disabled', and some have input fields for 'Extension', 'Authentication ID', and 'Authentication Password'. A 'Save' button is located at the bottom right of the form.

Extension Name	Enabled	Disabled	Extension	Authentication ID	Authentication Password
Priority Call Page Extension	<input checked="" type="radio"/>	<input type="radio"/>			
All Call Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 1 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 2 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 3 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 4 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 5 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 6 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Music Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 10 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 11 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 48 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 49 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			
Zone 50 Page Extension	<input type="radio"/>	<input checked="" type="radio"/>			

Additional SIP extensions can be registered for each zone that will be used. This allows the advantage of dialing directly to a zone without needing to enter DMTF (e.g. speed-dial keys can be used), but may require additional SIP licenses depending on the SIP provider.

To configure additional page extensions (up to 50) click "Enable" beside the target extension and enter the Extension, Authentication ID, and Authentication password.

Advanced Settings Tab - Network

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

Network Admin Time Provisioning Tones Advanced Audio Advanced SIP Advanced Multicast

Network Settings

Network Interface

Protocol Static IP DHCP

IP Address

Netmask

Gateway

DNS Server 1

DNS Server 2

802.1Q Virtual LAN

VLAN Mode Enabled Disabled

VLAN ID
Value range: 0 to 4094

VLAN Priority
Value range: 0 to 7

Differentiated Services

SIP (6-bit DSCP value)

RTP (6-bit DSCP value)

Protocol

DHCP is an IP standard designed to make administration of IP addresses simpler. When selected, DHCP will automatically configure IP addresses for each 8301 Paging Adapter on the network. Alternatively the 8301 Paging Adapter can be set to a static IP address.

VLAN Mode

Enables or Disables VLAN Tagging. 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 commonly known as IEEE 802.1p and defines the Generic Attribute Registration Protocol.

VLAN ID

Specifies the VLAN to which the Ethernet frame belongs. A 12-bit field specifying the VLAN to which the Ethernet frame belongs. The hexadecimal values of 0x000 and 0xFFF 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. On bridges, VLAN 1 (the default VLAN ID) is often reserved for a management VLAN; this is vendor specific.

VLAN Priority

Sets the frame priority level. Otherwise known as Priority Code Point (PCP), VLAN Priority is a 3-bit field which refers to the IEEE 802.1p priority. It indicates the frame priority level. Values are from 0 (lowest) to 7 (highest).

Differentiated Services (6-bit DSCP value)

Provides quality of service if the DSCP protocol is supported on your network. Can be specified independently for SIP control packets versus RTP audio packets.

Advanced Settings Tab – Admin

The screenshot shows the 'Advanced Settings' tab with the 'Admin' sub-tab selected. The interface includes the following sections:

- Admin Password:** Two input fields for 'Password' and 'Confirmation', both masked with dots and having a reset icon.
- General:** Includes 'Device Name (Hostname)' (text input), 'Introduction Section on Status Page' (radio buttons for On/Off), 'Web Interface Session Timeout' (dropdown menu set to '1 hour' with a help icon and note: 'Automatically log out web interface after period of inactivity.'), and 'Play Tone at Startup' (radio buttons for Enabled/Disabled with a help icon and note: '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:** Includes 'Log Level' (radio buttons for Error (Lowest), Notice ("Event"), Info ("SIP"), Debug (Highest)) and 'Log Method' (radio buttons for Local, Network, Both).
- Management:** Includes 'Web Interface Protocol' (radio buttons for Both HTTP and HTTPS, HTTPS Only) and 'SNMP Support (v1 get only)' (radio buttons for Enabled, Disabled).
- Syn-Apps:** Includes 'SA-Announce Support' (radio buttons for Enabled, Disabled), 'SA-Announce Server' (text input with a help icon and note: 'Leave this field blank to use the server provided by DHCP Option 72.'), and 'Local Management Port' (text input with value '6789').

A 'Save' button with a green checkmark is located at the bottom right of the form.

Password

Password to log into the 8301 Paging Adapter web interface. You should change the default password **algo** in order to secure the device on the network. If you have forgotten your password, you will need to perform a reset using the Reset Button in order to restore the password (as well as all other settings) back to the original factory default conditions.

Confirmation

Re-enter network admin password

Device Name (Hostname)

Name to identify the device in the Algo Network Device Locator Tool.

Introduction Section on Status Page

Allows the introduction text to be hidden from the login screen.

Web Interface Session Timeout

Set the maximum period of inactivity after which the web interface will log out automatically.

Play Tone at Startup

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 Level

Use on the advice of Algo technical support only.

Log Method

Allows the 8301 Paging Adapter to write to external Syslog server if the option for external (or both) is selected.

Log Server

If "Network" or "Both" is selected this is the address of the Syslog server on the network.

Web Interface Protocol

HTTPS is always enabled on the device. Use this setting to disable HTTP. When HTTP is disabled, requests will be automatically redirected to HTTPS. Also note that since the device can have any address on the local network, no security certificate exists, and thus most browsers will provide a warning when using HTTPS.

SNMP Support (v1 get only)

Additional SNMP support is anticipated for future, but the 8301 Paging Adapter will respond to a simple status query for automated supervision. Contact Algo technical support for more information.

SA-Announce Support

Syn-Apps' SA-Announce paging application converts unicast streams to multicast and delivers them to the target endpoints. The feature can only be used on the 8301 when Multicast Master Mode is disabled (set to 'None') in the Basic Settings > Multicast tab.

SA-Announce Server

Enter the SA-Announce Server to use the Syn-Apps paging feature. To use the server provided by the DHCP Option 72, leave the field blank.

Local Management Port

Enter the local management port.

Advanced Settings Tab – Time

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

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

Time Settings

General

Timezone

NTP Time Server 1

NTP Time Server 2

NTP Time Server 3

NTP Time Server 4

Set Time and Date Manually Automatic Manual

Manually set time and date settings are temporary. Upon rebooting the device, the current time will be fetched from the NTP servers above.

Manual Settings

Day

Month

Year

Time
 AM PM

Network time is used for logging events into memory for troubleshooting.

Timezone

Select timezone.

NTP Time Servers

The adapter will attempt to use Timer Server 1 and work down the list if one or more of the time servers become unresponsive.

Set Time and Date Manually

Date and Time can be set manually. These settings are temporary and the current time will be fetched from the NTP servers when the device is rebooted.

Advanced Settings Tab – Provisioning

The screenshot displays the 'Provisioning Settings' configuration page. At the top, there are tabs for 'Status', 'Basic Settings', 'Additional Features', 'Advanced Settings' (selected), 'System', and 'Logout'. Below these are sub-tabs for 'Network', 'Admin', 'Time', 'Provisioning' (selected), 'Tones', 'Advanced Audio', 'Advanced SIP', and 'Advanced Multicast'. The main content area is titled 'Provisioning Settings' and is divided into two sections: 'Mode' and 'Settings'. In the 'Mode' section, 'Provisioning Mode' is set to 'Enabled'. The 'Settings' section contains several fields: 'Server Method' is set to 'Static'; 'Static Server' is an empty text box; 'Download Method' is set to 'FTP'; 'Auth User Name' and 'Auth Password' are empty text boxes with a refresh icon next to the password field; 'Config Download Path' and 'Firmware Download Path' are empty text boxes. A 'Save' button with a green checkmark is located at the bottom right of the form.

Provisioning allows installers to pre-configure 8301 Paging Adapter units prior to installation on a network. It is typically used for large deployments to save time and ensure consistent setups.

There are two different Provisioning methods that can be used: via DHCP Option 66 or via a Static Server. In addition, there are three different ways to download provisioning files from a "Provisioning Server": TFTP (Trivial File Transfer Protocol), FTP, or HTTP.

For example, 8301 Paging Adapter configuration files can be automatically downloaded from a TFTP server using DHCP Option 66. This option code (when set) supplies a TFTP boot server address to the DHCP client to boot from.

DHCP must be enabled if using DHCP Option 66, in order for Provisioning to work.

One of two files can be uploaded on the Provisioning Server (for access via TFTP, FTP, or HTTP):

Generic (for all 8301 Paging Adaptors)
Specific (for a specific MAC address)

algot8301.conf
algot[MAC].conf

MD5 Checksum

In addition to the .conf file, an .md5 checksum file must also be uploaded to the Provisioning server. This checksum file is used to verify that the .conf file is transferred correctly without error.

A tool such as can be found at the website address below may be used to generate this file:

<http://www.fourmilab.ch/md5>

The application doesn't need an installation. To use the tool, simply unzip and run the application (md5) from a command prompt. The proper .md5 file will be generated in the same directory.

If using the above tool, be sure to use the "-l" parameter to generate lower case letters.

Generating a generic configuration file

1. Connect 8301 to the network
2. Access the 8301 Web Interface Control Panel
3. Configure the 8301 with desired options
4. Click on the System tab and then Maintenance.
5. Click "Backup" to download the current configuration file
6. Save the file settings.txt
7. Rename file settings.txt to algop8301.conf
8. File algop8301.conf can now be uploaded onto the Provisioning server

If using a generic configuration file, extensions and credentials have to be entered manually once the 8301 Paging Adapter has automatically downloaded the configuration file.

Generating a specific configuration file

1. Follow steps 1 to 6 as listed in the section "Generating a generic configuration file".
2. Rename file settings.txt to algom[MAC address].conf (e.g. algom0022EE020009.conf)
3. File algom[MAC address].conf can now be uploaded on the Provisioning server.

The specific configuration file will only be downloaded by the 8301 Paging Adapter with the MAC address specified in the configuration file name. Since all the necessary settings can be included in this file, the 8301 will be ready to work immediately after the configuration file is downloaded. The MAC address of each 8301 Paging Adapter can be found on the back label of the unit.

Advanced Settings Tab – Tones

The screenshot shows the 'Advanced Settings' tab for 'Tones'. The interface includes a navigation menu with options like 'Status', 'Basic Settings', 'Additional Features', 'Advanced Settings', 'System', and 'Logout'. Below the navigation, there are sub-tabs for 'Network', 'Admin', 'Time', 'Provisioning', 'Tones', 'Advanced Audio', 'Advanced SIP', and 'Advanced Multicast'. The main content area is titled 'Tone Management' and contains an 'Upload' section with a 'New Tone File' input, a 'Browse...' button, and an 'Upload' button. Below this is a 'Tone Files' section with a 'Functions' bar containing 'Delete', 'Rename', 'Play', and 'Auto Sound Level' buttons. A table lists various WAV files with columns for 'Name', 'Size', and 'Modification Date/Time'. At the bottom, there is a summary of the files: 'Number of Files: 12' and 'Total Size: 2075 kB'.

Name	Size	Modification Date/Time
<input checked="" type="radio"/> bell-na.wav	187 kB	Jul 17, 2013 14:44
<input type="radio"/> bell-uk.wav	100 kB	Jul 17, 2013 14:44
<input type="radio"/> buzzer.wav	187 kB	Jul 17, 2013 14:44
<input type="radio"/> chime.wav	187 kB	Jul 17, 2013 14:44
<input type="radio"/> dogs.wav	357 kB	Jul 17, 2013 14:44
<input type="radio"/> gong.wav	187 kB	Jul 17, 2013 14:44
<input type="radio"/> page-notif.wav	23 kB	Jul 17, 2013 14:44
<input type="radio"/> test-tone-1kHz.wav	93 kB	Nov 26, 2015 14:16
<input type="radio"/> warble1-low.wav	187 kB	Jul 17, 2013 14:44
<input type="radio"/> warble2-med.wav	187 kB	Jul 17, 2013 14:44
<input type="radio"/> warble3-high.wav	187 kB	Jul 17, 2013 14:44
<input type="radio"/> warble4-trill.wav	187 kB	Jul 17, 2013 14:44

Uploading custom Ring Tones (WAV Files)

Custom WAV files may be uploaded into memory to play on a Ring event or for other notification applications. An option is provided to normalize the uploaded file using "Auto Sound Level" and/or using "Compression (u-law)".

An existing file may also be modified by downloading the original via the links in the web interface, making the desired changes, and then uploading the new version with a different name.

Ring Tone Files Included in Memory

The 8301 Paging Adapter includes several pre-loaded WAV files that can be selected to play for ring events. The web interface allows selection of the WAV file and also the ability to play the WAV file immediately over a speaker for testing. Files may also be deleted or renamed.

Advanced Settings Tab – Advanced Audio

The screenshot shows a web interface for configuring advanced audio settings. The top navigation bar includes tabs for Status, Basic Settings, Additional Features, **Advanced Settings**, System, and Logout. Below this, there are sub-tabs for Network, Admin, Time, Provisioning, Tones, **Advanced Audio**, Advanced SIP, and Advanced Multicast. The main content area is titled 'Advanced Audio Functions' and is divided into two sections: 'Functions' and 'Audio Filters'.
Functions Section:
- **Dynamic Range Compression (DRC):** Radio buttons for 'Enabled' (selected) and 'Disabled'. Description: 'Compress the dynamic range of page audio to increase loudness.'
- **Dynamic Range Compression Gain:** A dropdown menu set to '6'. Description: 'Specify the amount of compression gain. More gain increases distortion.'
- **Jitter Buffer Range (milliseconds, 10 ~ 500):** A text input field set to '100'. Description: '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:** Radio buttons for 'Enabled' and 'Disabled' (selected). Description: 'Play DTMF tones during a SIP Call to allow interoperability with DTMF-controlled multi-zone amplifiers'.
Audio Filters Section:
- **Speaker Filter G.711:** A dropdown menu set to '300Hz High-Pass'. Description: 'Bandwidth also limited by audio codecs.'
- **Microphone Filter G.711:** A dropdown menu set to '300Hz High-Pass'.
- **Speaker Filter G.722:** A dropdown menu set to '150Hz High-Pass'.
- **Microphone Filter G.722:** A dropdown menu set to '150Hz High-Pass'.
- **Speaker Noise Filter:** Radio buttons for 'Enabled' and 'Disabled' (selected). Description: 'Heavy filtering below 150Hz to reduce mains induced noise (fans).'
- **Microphone Noise Filter:** Radio buttons for 'Enabled' and 'Disabled' (selected). Description: 'Heavy filtering below 150Hz to reduce mains induced noise (fans).'
A 'Save' button with a green checkmark is located at the bottom right of the form.

Dynamic Range Compression (DRC)

If enabled, compresses the dynamic range of page audio to increase loudness.

Dynamic Range Compression Gain

Higher compression gain increases distortion.

Jitter Buffer Range

The jitter buffer removes the jitter in arriving network packets by temporarily storing them. This process corrects the inconsistent delays on the network. It is recommended to use the lowest value.

Generate In-Band DTMF Tones

If enabled, plays DTMF tones during a SIP call to allow interoperability with DTMF-controlled multi-zone amplifiers.

Microphone Filter G.711

G.711 microphone filter.

Speaker Filter G.722

G.722 speaker filter.

Microphone Filter G.722

G.722 microphone filter.

Speaker Noise Filter

Enables heavy filtering below 150Hz to reduce mains induced noise (fans).

Microphone Noise Filter

Enables heavy filtering below 150Hz to reduce mains induced noise (fans).

Advanced Settings Tab – Advanced SIP

The screenshot displays the 'Advanced SIP Settings' configuration page. At the top, there are navigation tabs: Status, Basic Settings, Additional Features, **Advanced Settings**, System, and Logout. Below these are sub-tabs: Network, Admin, Time, Provisioning, Tones, Advanced Audio, **Advanced SIP**, and Advanced Multicast. The main content area is titled 'Advanced SIP Settings' and is divided into two sections:

- SIP**:
 - Outbound Proxy: [Empty text box]
 - STUN Server: [Empty text box]
 - Register Period (seconds): 3600
 - Keep-alive Method: None Double CRLF
 - Keep-alive Period (seconds): 30
 - Different Ports for Extensions: Enabled Disabled
Turn this option on for certain proxies, e.g. Cisco Communication Manager 7, to send ring and page SIP requests through different port numbers.
- Server Redundancy**:
 - Server Redundancy Feature (Multiple SIP Server Support): Enabled Disabled
 - Backup Server #1: [Empty text box]
 - Backup Server #2: [Empty text box]
 - Polling Interval (seconds): 120 seconds (2 minutes) [Dropdown menu]
 - Time period between sending monitoring packets to each server. Non-active servers are always polled, and active server may optionally be polled (see below).
 - Poll Active Server: Enabled Disabled
Explicitly poll current server to monitor availability. May also be handled automatically by other regular events, so can be disabled to reduce network traffic.
 - Automatic Failback: Enabled Disabled
Reconnect with higher priority server once available, even if backup connection still fine.
 - Polling Method: SIP NOTIFY SIP OPTIONS
SIP message used to poll servers to monitor availability.

A green 'Save' button is located at the bottom right of the configuration area.

Outbound Proxy

IP address for outbound proxy. A proxy (server) stands between a private network and the internet.

STUN Server

IP address for STUN server if present.

Register Period (seconds)

Maximum requested period of time where the 8301 Paging Adapter will re-register with the SIP server. Default setting is 3600 seconds (1 hour). Only change if instructed otherwise.

Keep-alive Method

If Double CRLF is selected the 8301 Paging Adapter will send a packet every 30 seconds (unless changed) to maintain connection with the SIP Server if behind NAT.

Different Ports for Extensions

Enable different ports for extensions for certain proxies such as Cisco Communication Manager 7, to send and page SIP requests through different port numbers.

Server Redundancy Feature

Two secondary SIP servers may be configured. The 8301 Paging Adapter 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.

If Server Redundancy is selected the web page will expand as shown below.

Backup Server #1

If primary server is unreachable the 8301 Paging Adapter will attempt to register with the backup servers. If enabled, the 8301 Paging Adapter will always attempt to register with the highest priority server.

Backup Server #2

If backup server #1 is unreachable the 8301 Paging Adapter will attempt to register with the 2nd backup server. If enabled, the 8301 Paging Adapter will always attempt to register with the highest priority server.

Polling Intervals (seconds)

Time period between sending monitoring packets to each server. Non-active servers are always polled, and active server may optionally be polled (see below).

Poll Active Server

Explicitly poll current server to monitor availability. May also be handled automatically by other regular events, so can be disabled to reduce network traffic.

Automatic Failback

Reconnect with higher priority server once available, even if backup connection is still fine.

Polling Method

SIP message used to poll servers to monitor availability.

Advanced Settings Tab – Advanced Multicast

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

Network | Admin | Time | Provisioning | Tones | Advanced Audio | Advanced SIP | **Advanced Multicast**

Advanced Multicast Settings

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

Basic Zone Definition

When an Algo device is the multicast master, a page tone will play on the slave device, so it is recommended to set the slave tone to "None".

Zone	IP Address and Port	Page Tone	Page Volume
Priority Call (DTMF:9)	224.0.2.60:50000	<None>	<Use Default Page Volume>
All Call (DTMF:0)	224.0.2.60:50001	<None>	<Use Default Page Volume>
Zone 1 (DTMF:1)	224.0.2.60:50002	<None>	<Use Default Page Volume>
Zone 2 (DTMF:2)	224.0.2.60:50003	<None>	<Use Default Page Volume>
Zone 3 (DTMF:3)	224.0.2.60:50004	<None>	<Use Default Page Volume>
Zone 4 (DTMF:4)	224.0.2.60:50005	<None>	<Use Default Page Volume>
Zone 5 (DTMF:5)	224.0.2.60:50006	<None>	<Use Default Page Volume>
Zone 6 (DTMF:6)	224.0.2.60:50007	<None>	<Use Default Page Volume>
Music (DTMF:7)	224.0.2.60:50008	<None>	<Use Default Page Volume>

Expanded Zone Definition

Zone	IP Address and Port	Page Tone	Page Volume
Zone 10 (DTMF: *10)	224.0.2.110:50000	<None>	<Use Default Page Volume>
Zone 11 (DTMF: *11)	224.0.2.111:50000	<None>	<Use Default Page Volume>

↓

Zone 48 (DTMF: *48)	224.0.2.148:50000	<None>	<Use Default Page Volume>
Zone 49 (DTMF: *49)	224.0.2.149:50000	<None>	<Use Default Page Volume>
Zone 50 (DTMF: *50)	224.0.2.150:50000	<None>	<Use Default Page Volume>

Save

Zone Definition

The "Expanded" Slave or Master zones can be enabled/disabled in Basic Settings > Multicast. Default IP addresses and ports may be revised for any given zone in the table.

Ensure that the Address and Port settings are the same for all master and slave devices.

Page Tone and Page Volume

Master Mode: By default, the same tone can be set for all slave zones in the Basic Settings > Features tab. Unique paging tones may be revised for any given slave zone in the table above.

Slave Mode: When an Algo device is the multicast master, a page tone will play on the slave device, so it is recommended to set the slave tone to "None". If a page is received from a non-Algo device that doesn't send a tone, a tone can be inserted on the slaves (above) each time they detect page audio starting, allowing them to play a tone.

By default, the same page volume can be set for all slave zones in the Basic Settings > Features tab. Unique page volumes may be revised on a per-zone basis in the table above. For instance, emergency pages can be louder on certain slave speakers.

System Tab - Maintenance

The screenshot shows a web interface for system maintenance. At the top, there are navigation tabs: Status, Basic Settings, Additional Features, Advanced Settings, System (selected), and Logout. Below these, there are sub-tabs: Maintenance (selected), System Log, and About. The main content area is titled "System Maintenance" and is divided into three sections:

- Backup / Restore Configuration:** This section contains three rows. The first row is "Download Configuration File" with a "Download" button. The second row is "Restore Configuration File" with a "Browse..." button, the text "No file selected.", and a "Restore" button. The third row is "Restore Configuration to Defaults" with a "Restore Defaults" button.
- Reboot:** This section contains one row: "Reboot the device" with a "Reboot" button.
- Upgrade to New Firmware:** This section contains three rows. The first row is "Method" with two radio buttons: "From Local Files" (selected) and "From URL". The second row is "Firmware Image" with a "Browse..." button, the text "No file selected.", and a greyed-out button. The third row is "MDS Checksum" with a "Browse..." button, the text "No file selected.", and a greyed-out button. At the bottom of this section is an "Upgrade" button.

Download Configuration File

Save the device settings to a text file for backup or to setup a provisioning configuration file.

Restore Configuration File

Restore settings from a backup file.

Restore Configuration to Defaults

Resets all 8301 Paging Adapter device settings to factory default values.

Reboot the Device

Reboots the device.

Method

Specify whether the firmware files will be downloaded from the local computer or a remote URL.

Firmware Image

Point to the firmware image provided by Algo

MD5 Checksum

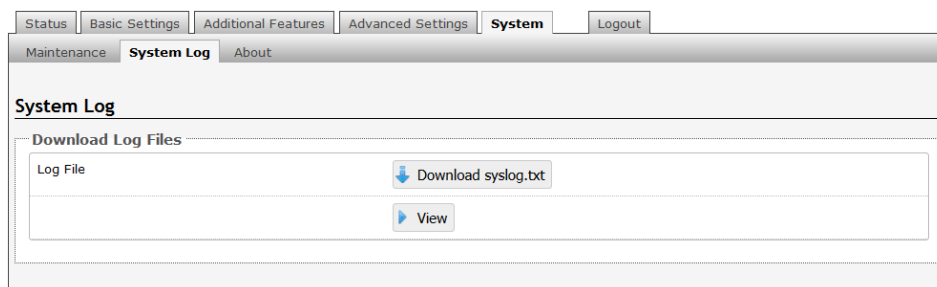
Point to the checksum file provided by Algo

Upgrade 8301 Paging Adapter Firmware

1. From the top menu, click on System, then Maintenance.
2. In the Maintenance section, click Reboot, and wait 30-60 seconds for the device to reboot and the web page to automatically reload.
3. Login to the device again, and click on System.
4. In the Upgrade section, click on Choose File and select the 8301 Paging Adapter firmware file to upload. Note that both the FW firmware and MD5 checksum files must be loaded.
5. Click Upgrade
6. After the upgrade is complete, confirm that the firmware version has changed (refer to top right of Control Panel).

System - Network Logging

System log files are automatically created and assist with troubleshooting in the event the 8301 Paging Adapter does not behave as expected.



Specifications

Power Input:	48 V PoE IEEE 802.3af Class 1 (Max 3.84W - Idle nominal 2.0W)
SIP:	Dual extensions for Page or Alerting
Multicast:	Receive or transmit
Codec Support:	G.711 A-law, G.711 u-law, G.722, Polycom Group Page
Processor:	Linux OS ARM Cortex-A8 32-Bit RISC Processor
AUX Input:	3.5mm jack for iPod/iPhone
AUX Output:	3.5mm jack for headset or PC speakers
Line Input:	Female mini-XLR 10 kOhm balanced maximum level +4 dBu. Transformer isolated internally
Line Output:	Low impedance balanced output. Line level -10 dBm/0 dBm/+4 dBu. Transformer isolated internally. Male mini-XLR connector and pluggable terminal block
Audio Memory:	1 GByte
Speech Processing:	ALC, filtering, compression
Audio Delay:	Programmable 1-1000 ms synchronization delay
Page Mode:	Live or cache and release
Relay Output:	Normally open or normally closed. Max rating 30 V 50 mA.
Relay Input:	Normally open or normally closed dry contact supervision. Algo 1202 Call Button, Algo 1203 Call Switch, EOL resistor termination.

Relay Input Current Draw Detection Thresholds:

	Active	Idle	Tamper
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.

Configuration: Web interface or auto-provisioning server.

Provisioning: TFTP, FTP, or HTTP

Supervision: SNMP

NAT: STUN, CRLF Keep Alive

Environmental: +32 to +122 deg F (0 to +50 deg C);
Suitable for dry indoor environments only.

Dimensions: 6.5" x 4.27" x 2.3" (cm x cm x cm)

Mounting: Wall mountable or tabletop

Weight: 2.2 lb (1.0 Kg)

FCC Compliance Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.