

8410 IP Display Speaker / 8420 IP Dual-Sided Display Speaker

User Guide





AL061-UG-GP258410-001-RA Firmware Version 5.2 <u>support@algosolutions.com</u> March 09, 2023 Algo Communication Products Ltd. 4500 Beedie Street, Burnaby V5J 5L2, BC, Canada 1-604-454-3790 www.algosolutions.com



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

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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 802.3at / 802.3bt 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 802.3at / 802.3bt 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 802.3at / 802.3bt. 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 802.3at / 802.3bt-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.



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 802.3at / 802.3bt 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.



An improperly installed 8410 IP Display Speaker or 8420 IP Dual-Sided Display Speaker could fall from a wall or ceiling and cause serious injury or death.

Local building code may require one or more additional safety measure(s), particularly in earthquake prone regions.

MEMERGENCY COMMUNICATION

If used in an emergency communication application, the 8410 IP Display Speakers and 8420 IP Dual-Sided Display Speaker should be routinely tested. SNMP supervision is recommended for assurance of proper operation. Contact Algo for other methods of operational assurance.

DRY INDOOR LOCATION ONLY

The 8410 IP Display Speaker and 8420 IP Dual-Sided Display Speaker 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 802.3at (PoE+) or IEEE 802.3 bt (PoE++) compliant network PoE switch must not leave the building perimeter without adequate lightning protection.

No wiring connected to the 8410 IP Display Speaker and 8420 IP Dual-Sided Display Speaker may leave the building perimeter without adequate lightning protection.



1 GENERAL

1.1 Introduction

Algo's 8410 IP Display Speaker is a SIP-compliant notification device that comprises of three core components – LCD screen, Wideband Speakers, and LED flashers – to create highly flexible and effective visual and audible communication. The combined functionality of the LCD display screen, speakers, and flashing LEDs enable voice paging, visual or audible alerting, and informational visual content, such as scrolling text messages, visual paging, images, or wayfinding. Two-way communication can also be established through the embedded microphone so real-time updates can easily be provided to admin staff. When idle, the Display Speakers can display an image, a clock face (shown in analog and/or digital format and synchronized to network time protocol (NTP)), rotating through a variety of announcements, or go to sleep.

The 8420 IP Dual-Sided IP Display Speaker has all the core components of the 8410 but double. The two-sided functionality of the 8420 is ideal for installation in corridors, large spaces, or hallways for easy opposite-facing visual and audible communication.

As 3rd-party, SIP-compliant devices, the 8410/8420 is designed to seamlessly integrate into most leading IP-based UC and Mass Notification platforms. The 8410/8420 is easily configured using central provisioning features or by accessing the web interface using your choice of browser.

1.1.1 Key Features

LCD Screen

The primary element of the 8410/8420 is the LCD screen(s). With nearly unlimited visual possibilities, the screen can produce scrolling text, flashing announcements, full-color images, or times and dates. Combined with LED backlighting and wide-angle viewing, the visual display can be seen from anywhere in a room or along a hallway. It can be customized to complement an audio announcement or to bring communication to an audio-less environment.

Speakers

For audible alerting and voice paging, the 8410/8420 offers similar functionality and voice quality to other Algo IP speakers, delivering wideband HD voice for clear voice communication and attention-grabbing audible alerting. An embedded microphone is included to enable ambient noise response and talkback capabilities.

LED Flashers

The LED flashers provide additional awareness for important messages. This can help draw attention to the screen, helping hearing-impaired audiences to notice the screen or bring awareness to audio-less environments.



2 SETUP AND INSTALLATION FOR 8410 AND 8420



Important

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

See the appropriate section below for mounting either the 8410 or 8420.

2.1 8410 Mounting & Installation

Algo's 8410 IP Display Speaker is a one-sided product intended for flush mounting on a wall. For the ceiling-mount option more suited to hallways, see the 8420 mounting and installation instructions below.

What is Included

The following items are included with the purchase of this device:

- 8410 IP Display Speaker
- Wall-mount bracket
- Bracket attachment screws (4x 6-32 Flat Head Hex Drive Machine Screws)
- Terminal Block Plug
- 5/64" Drive Size Allen Key
- Getting Started Sheet

2.1.1 Getting Started – Quick Install and Setup

- 1. Connect the 8410 to an IEEE 802.3at / 802.3bt compliant PoE+ or PoE++ network switch or power injector. See section 2.4 Power Requirements.
 - The green LED flashers will activate and a small blue light on the bottom below the flasher will remain on until boot up is completed about 30 seconds.
 - After a few moments, the screen will show the Algo logo.
- 2. Once the unit finishes booting, a configuration message with the IP address will appear on the screen if it is in factory default state. *If there is no DHCP server, the 8410 will default to the static IP address 192.168.1.111.*



Note

If the IP address does not appear on the screen, press the reset (RST) button to hear the IP address recited over the speakers. The IP address will be repeated over the speakers until the reset button is pressed again to turn off.

3. Access the 8410 IP Display Speaker web interface by entering the IP address into a browser and login using the default password *algo*.

2.1.2 Installation

The 8410 IP Display Speaker is designed to fit on a wall bracket. Concealed wiring may enter from the wall into the wiring cavity. Alternatively, surface wiring may enter through a channel from the bottom edge.

What is Not Included

The following items are not included with the purchase of this device:

- Ethernet Cable
- PoE++ Switch
- Wall bracket screws (#8 screws required as appropriate for your wall)



Mounting the wall bracket securely:

1. Use four #8 wall bracket screws to hang the wall bracket. The wall-mount bracket should be secured to a stud to prevent the 8410 from falling.



Attaching the 8410 to the wall-mount bracket:

- 1. Set the 8410 into the wall-mount bracket.
- 2. Align the holes on the side of the 8410 to the holes in the wall-mount bracket and tighten the four 6-32 mounting screws securely using the Allen key.





2.2 8420 Mounting & Installation

Algo's 8420 IP Dual-Sided Display Speaker is a two-sided SIP-compliant notification device that can be mounted on the ceiling or wall for two-sided visual communication. This is ideally suited for hallways, corridors, or open spaces.

What is Included

The following items are included with the purchase of this device:

- 8420 IP Display Speaker
- Mounting Bracket
- Mounting screws to attach device to mounting bracket (6x 10-32 screws, 2x 10-32 flat head screws Hex Drive Machine Screws)
- Terminal Block Plug
- 1/8" Drive Size Allen Key
- Getting Started Sheet
- Snap-On Ferrite for Ethernet Cable

2.2.1 Getting Started – Quick Install & Test

What is Not Included

The following items are not included with the purchase of this device:

- Ethernet Cable
- PoE++ Switch
- Mounting screws to attach mounting bracket to wall or ceiling (recommended mounting hardware outlined in this guide)

- 1. Connect the 8420 to an IEEE 802.3at / 802.3bt compliant PoE+ or PoE++ network switch or power Injector. See section 2.4 Power Requirements.
 - The green LED flashers will activate and a small blue light on the bottom below the flasher will remain on until boot up is completed about 30 seconds.
 - After a few moments, the screens on either side will show the Algo logo.
- 2. Attach the snap-on ferrite to the ethernet cable close to the connector to meet FCC requirements.
- 3. Once the unit finishes booting, a configuration message with the IP address will appear on each screen if it is in factory default state. If there is no DHCP server, the 8420 will default to the static IP address 192.168.1.111.



Note

If the IP address does not appear on the screen, press the reset (RST) button to hear the IP address recited over the speakers. The IP address will be repeated over the speakers until the reset button is turned off.

4. Access the 8420 IP Dual-Sided Display Speaker web interface by entering the IP address into a browser and login using the default password *algo*.

2.2.2 Installation

The 8420 IP Dual-Sided Display Speaker is designed to be both ceiling and wall mounted. Wiring enters through an opening in the hollow bracket for this product (see info in the Note after bullet 1 below).

Mounting the bracket securely to the wall/ceiling:

 Mount the bracket securely to a wall or ceiling. Because the material you mount the device into may differ, see below for specific wall or ceiling mount instructions by material type.
 Note: Wires can be routed into the hollow tubes of the bracket from the wall/ceiling or through the wire feed hole outside the tube. Wire must be routed on the same side as the speakers to match the connection area.



- Set the 8420 into the mounting bracket by partially threading the screws into the holes next to the connector opening and mount the screws through the larger hole of the keyhole slot. Slide the 8420 into the smaller holes to lock the device in place.
- 2. Slide the ceiling- or wall-mount connector cover into place.
- 3. Fasten all six mounting screws on the bracket and the two flat mounting screws on the connector cover.



8420 Wall Mount

- For concrete, brick, or block install, use 6 x 5/16" x 1 5/8" female threaded anchors rated for 200 lbs each or greater pull and install bracket using 6 x 1 ½" grade 5 threaded bolts with flat washer.
- For plaster or drywall install, use 6 x 5/16" toggle bolts and washers rated for 60 lbs each or greater. The bolts with a washer should be fed through the bracket prior to attaching toggles. If a wood stud is available and the position is known, use 5/16" x 2 ½" lag bolts in place of toggle bolts on the side of the bracket aligned to the stud.

8420 Ceiling Mount

- For concrete install, use 6 x 5/16" x 1 5/8" female threaded anchors rated for 200 lbs each or greater pull and install the bracket using 6 x 1 ½" grade 5 threaded bolts with flat washer.
- For plaster or drywall install, use 6 x 5/16" toggle bolts and washers rated for 60 lbs each or greater. The bolts with a washer should be fed through the bracket prior to attaching toggles. If a wood joist is available and the position is known, use 5/16" x 2 ½" lag bolts in place of toggle bolts on the side of the bracket aligned to the joist.
- For acoustic tile, cut and lay 3/4" plywood equal in size dimension to the tile and install the bracket using 6 x 5/16" x 2 ½" lag bolt with a washer.

2.3 Making Your First Configuration

Enabling a clock:

1. Navigate to *Display* \rightarrow *Slides* to create a screen display.



- 2. Enter a Name and select Digital Clock. Customize your clock preferences. Press Save.
- 3. Go to Display \rightarrow Screens to apply a screen display.
- 4. Under Default Screen, select the name of the display created in the previous step from the Slides dropdown menu.
- 5. Press **Save** at the bottom right of the page.

2.4 Power Requirements

For optimal performance, Algo recommends using an 802.3bt Type 3 (PoE++) switch for the 8410 and 802.3bt Type 4 (PoE++) switch for the 8420. The minimum power required for the 8410 is 802.3at (PoE+) power and the minimum power required for the 8420 is 802.3bt Type 3 power. If optimal switches are not available, a power injector may be used. If using a power injector that does not automatically negotiate its power capabilities, go to Advanced Settings \rightarrow Admin to manually configure Automatic Power Detection versus Forced PoE++ mode. off.

2.5 Reset

A recessed reset button (RST) is located at the bottom of the device below the speaker and can only be used to reset the 8410 IP Display Speaker and 8420 IP Dual-Sided Display Speaker at time of power up. To return all the settings to the factory default for the 8410, reboot or power cycle the 8410/8420. Wait until the SIP LED flashes and then press and hold the reset button until the single blue LED on the bottom of the device begins a double flash pattern. Release the reset button and allow the unit to complete its boot process.



Important

Do not press the reset button until the blue SIP LED begins flashing. A reset will set all configuration options to factory default including the login password.

Once booting has completed, pressing the reset button will cause the device to speak its IP address over the speakers.



3 APPLICATIONS

3.1 Voice Paging

The 8410 IP Display Speaker and 8420 IP Dual-Sided Display Speaker can be used for direct paging in addition to multicast. This allows direct communication with a single location.

3.2 Notification

The 8410 and 8420 provide effective notification alerting for emergency (e.g., lockdown, evacuation, reverse evacuation), safety (e.g., medical, workplace accident), and security (e.g., OSHA or similar workplace regulations) events.

For emergency alerting, the LCD screen, speaker, and LED flashers can be used together for a highly visible, attentiongrabbing communication.

3.3 Visual Communication

When not used to display an active alert, the LCD screen can be configured to display a chosen image or layout for when in idle mode. This screen display can be a digital or analog clock synchronized to NTP, or an uploaded image. It can also display a series of selected images in a slideshow.

3.4 Multicast Receiver

The 8410 and 8420 are ideal for use as multicast receivers. Each alert can be mapped to multicast zones which can be associated with a display on the LCD screens to complement the audio announcements.

3.5 Scheduling

The 8410 IP Display Speaker can be paired with the 8301 IP Paging Adapter (sold separately) for scheduling tones and bells for schools, automated announcements for retail and healthcare, and workplace shift changes and breaks in warehouses.

3.6 InformaCast Compatible

The Algo 8410 IP Display Speaker is fully compliant with Singlewire InformaCast for telephone, security, and emergency alerting. The 8410 can display scrolling text and audible notifications for safety and security communication.



4 FEATURES

4.1 Setting Up Visual Display Content

The display on the 8410 is completely customizable. To set up the display screen:

- 1. Go to Basic Settings \rightarrow Screen.
- 2. Under **Default** Screen, choose the following screen settings:
 - a. Number of Images
 - b. Images
 - c. Text Behavior
 - d. Text
 - e. Text Font
 - f. Text Position
 - g. Text Color
 - h. Text Size
 - i. Strobe Flash Pattern

To set the idle screen:

- 1. Go to Basic Settings \rightarrow Features.
- 2. Under the Screen heading, select an Idle Pattern and choose the Screen Brightness.

4.2 SIP Paging: One 8410

The 8410 IP Display Speaker can be registered as a third-party SIP extension with a hosted or enterprise Communications Server supporting 3rd-party SIP endpoints.

To register the 8410 with the SIP server, use the *Basic Settings* \rightarrow *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 \rightarrow Network tab to set VLAN options.



Important

Once the 8410 IP Display Speaker is using a VLAN, you will need to be on the same VLAN to access the web interface.

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

There are several configurable adapter options, such as:

- Increase or Decrease Speaker Volume
- Enable AGC (automatic gain control)
- Enable Ambient Noise Monitoring (speaker volume adapts to background noise)
- Enable Talkback
- Customize pre-announce tone 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.

4.3 Multicast Overview

The 8410 IP Display Speaker is a multicast receiver and is able to receive IP audio multicast messages over the network to support larger deployment for both paging and notification. This provides a scalable and efficient method of building large scale notification solutions.

An Algo 8301 IP Paging Adapter & Scheduler can be configured with a SIP Page Extension to multicast to the 8410 IP Display Speaker. When called from a phone, the SIP registered 8301 will auto-answer and play the page audio over the 8410/8420 speaker.

Simultaneously, the registered 8301 endpoint broadcasts the audio over the network using RTP multicast to any number/combination of Algo IP display speakers, speakers, visual alerters, and paging adapters as required.

Receiver endpoints require a PoE network connection but do not require registration to the communication server.

Multicasting can also be used to distribute loud ring or other alerting (e.g., safety, security, or emergency events) over multiple Algo endpoints (e.g., 8190, 8196, 8188, 8186, 8128, 8201, 8301, and 8373).

4.3.1 SIP Paging: Multiple 8410s (Using Multicast)

To use the multicast feature, set up an 8410 IP Display Speaker as the Multicast Transmitter.

The Sender device will page normally while simultaneously streaming audio to the Receiver speakers (such as the 8410/8420). The Receiver speakers do not require SIP extensions and do not need to register with the SIP Communication Server.

To enable multicast streaming from the 8301 IP Paging Adapter & Scheduler, go to the web interface of the device and navigate to the *Basic Settings* \rightarrow *Multicast* tab. Choose multicast mode 'Sender' and zone 'All Call'. The multicast addresses pre-populated in the table, under *Advanced Settings* \rightarrow *Advanced Multicast*, will work in most cases and should only be altered for rare cases.

To enable multicast monitoring in the 8410 IP Display Speaker, go to the web interface for reach device and navigate to *Basic Settings* \rightarrow *Multicast*. Choose multicast mode 'Receiver'. There is no need to select a zone as the speaker will automatically monitor the 'All Call' zone IP address.

The page pre-announce tone is generated from the Sender. The following options are valid for each multicast Receiver:

- Increase or Decrease Speaker Volume
- Enable Ambient Noise Monitoring (speaker volume adapts to background noise)

4.3.2 SIP Paging: Multiple Display Speakers (Using Individual SIP extensions)

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



A speaker configured as a SIP Multicast Slave will give its highest priority to the 'Priority Call' zone. Other than the 'Priority Call' zone, a page using its SIP extension, has priority over all other multicast zones.

Communication Servers with the capability of dialing many SIP extensions simultaneously for paging may be able to create zones by calling "page groups" and page telephone speakers in conjunction with overhead speakers.

4.4 SIP-Activated Visual Notification Alerts

In addition to audible notifications, the 8410/8420 can multicast visual alerting with the LED flashers. When a call is made to the SIP extension, the 8410/8420 will flash the selected light pattern. Often, the flashers for the 8410 will be part of a hunt group or ring group to flash in conjunction with an audio notification.

There are several configurable strobe options:

- Flash Pattern
- Brightness

4.5 Color TLS for SIP Signaling and Provisioning

Algo devices support Transport Layer Security (TLS). This feature adds security by ensuring that Algo products can trust the hosted SIP server. This is useful for when third-party devices or attackers may try to intercept, replicate, or alter Algo products, and try to connect to the server. TLS protocol will ensure that third parties cannot read/modify any actual data. Previously security was less of a concern because phone systems were on isolated networks, but hosted services are becoming increasingly more common. Using a hosted SIP service requires traffic to be sent over the public internet and thus much more susceptible to attacks. Signed certificates are an important piece in the Algo device's operation, to ensure the security, integrity, and privacy of its communication. Algo components that use TLS are **Provisioning** and **SIP Signaling**.

These Algo devices each come pre-loaded with certificates from a list of trusted certificate authorities (CA), which are installed in the hardware at the time of manufacture. Note these pre-installed trusted certificates are not visible to users and are separate from the 'certs' folder.

The TLS handshake happens to make sure that the client and server can trust each other, and once that trust is established, the two parties can freely send encrypted data and decrypt any data that they receive. After the TLS handshake process is complete, a TLS session is established, and the server and client can then exchange messages that are symmetrically encrypted with shared (pre-Sender) secret key.

For further details, reference Algo's guide for Securing Algo IP Endpoints: TLS and Mutual Authentication.

Uploading Public CA Certificates to Algo SIP Endpoints

To install the public CA certificate on the 8410/8420, follow the steps below:

- 3. Obtain a public certificate from your Certificate Authority.
- 4. Rename the public certificate 'siptrusted.pem' (only .pem format is supported).
- 5. In the web interface of the Algo device, navigate to the Advanced Settings \rightarrow File Manager.
- 6. Upload the certificate files into the 'certs' directory. Click the Upload button in the top left corner of the file manager and browse to the certificate.



For **SIP** TLS, no default public CA certificates are used; only the above .pem file is supported, so this certificate file must be uploaded in order for SIP TLS authentication to occur.

For **Provisioning** TLS, only the default pre-installed public CA certificates are supported; No .pem file can be uploaded in this case.

HTTPS Provisioning

Provisioning can be secured by setting the 'Download Method' to 'HTTPS' (under the *Advanced Settings* \rightarrow *Provisioning*). This prevents configuration files from being read by an unwanted third-party. This resolves the potential risk of having sensitive data stolen, such as admin passwords and SIP credentials.



Status	Basic Setti	ngs	Display	Additional Features	Advanced Se	ttings	System		Logout			
Network	Admin	Users	s Time	Provisioning	Advanced Audio	Advan	ced SIP	Advanc	ed Multicast	_	_	_
Provisioning Settings												
	oning Set	tting	S									
Mode					~							
Provisi	oning Mode				Enabled	ODisa	bled					
Settin	as											
	Method				OAuto (D	HCP Opt	ion 66/16	0/150)				
ODHCP Option 66 only ODHCP Option 160 only ODHCP Option 150 only Static () Auto mode automatically checks all 3 DHCP options for an active provisioning server, in the												
Static	Server				order listed.			±	7			
									_			
Downle	oad Method				Oteta C	Oftp C	HTTP OH	HTTPS				
Validat	te Server Ce	ertificat	te		certificates, u	ne server use the "S	against con System > F	ile Mana	rtificate authoriti <u>ger</u> " tab to upload ts/trusted' folder	a Base64 en	-	
Auth U	lser Name]			
Auth P	assword							ą	3			
Config	Download F	Path]			
Firmwa	are Downloa	ad Path	I									
Partial	Provisioning		Enabled (i)Allow supp using this fea	port for "		ntal prov	isioning files. Dis	ble for enhar	nced security if	not		
Check-sync Behavior Check-sync Behavior Check-sync Behavior Check-sync Behavior Conditional Reboot Conditional Reboot Conditional Reboot Conditional Reboot 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												
Sync E	nd Time				-	an End	Time earlie] time in the wind art Time indicate:			
Sync F	requency				Daily	Selecte	ed Days Or	nly				
Zero Te	ouch Provisi	oning			Turn Off Z		l can only b	e re-ena	bled with a factor	y reset.		
												✔ Sa



Important

To verify the server, enable the 'Validate Server Certificate' option. This then checks if the certificate that is provided by the server is signed by any of the CAs included in the list of trusted CAs (used by the Debian infrastructure and Mozilla browsers). If we receive a certificate signed by any of these CAs, then that server will be trusted.



The 'Validate Server Certificate' parameter can also be enabled through provisioning:

prov.download.cert = 1

Encrypting SIP and RTP Communication

SIP signalling is secured by setting 'SIP Transportation' to 'TLS' (under the Advanced Settings \rightarrow Advanced SIP tab). Setting it to 'TLS' ensures that the SIP traffic will be encrypted. The SIP signalling is responsible for establishing the call (the control signals to start and end the call with the other party), but it does not contain the audio.

For the audio (voice) path, use the setting 'SDP SRTP Offer'. Setting this to 'Optional', means the SIP call's RTP audio data will be encrypted (using SRTP) if the other party also supports audio encryption. If the other party does not support SRTP, then the call will still proceed, but with unencrypted audio. To make audio encryption mandatory for all calls, set 'SDP SRTP Offer' to 'Standard'. In this case, if the other party does not support audio encryption, then the call attempt will be rejected.



vanced SIP Settings						
General						
SIP Transportation	TLS -					
	 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 "Advanced Settings > File Manager" tab to upload a certificate file renamed to 'sipclient.pem' in the 'certs' folder. To force the Algo device to authenticate the SIP server, a certificate obtained from the SIP server needs to be installed. Use the "Advanced Settings > File Manager" tab to upload a certificate file renamed to 'siptrusted.pem' in the 'certs' folder. 					
SIPS Scheme	C Enabled O Disabled					
SDP SRTP Offer	Standard 🔹					
SIP Outbound Support (RFC 5626)	 Enabled Oisabled Enable this option to support best networking practices according to RFC 5626. This option should generally be enabled if the Algo device is being registered with a hosted server or if TLS is being used for SIP Transportation. 					
Outbound Proxy						
Register Period (seconds)	3600					
NAT Media NAT	None ICE STUN					
Server Redundancy						
Server Redundancy Feature (Multiple SIP Server	Support) Carabled Obisabled					
Interoperability						
Keep-Alive Method	• None Ouble CRLF (i)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. 					



Important

For a SIP server to validate the Algo device, an additional certificate has to be manually installed on the 8410/8420. To add this user certificate file use a '.pem' filetype extension and have the file named 'sipclient'. This is done by manually adding a file named 'sipclient.pem', which contains a device certificate and private key, to the 'certs' folder (under the 'Advanced Settings' tab File Manager'). In the future, '.crt', '.cer', and '.der' certificate extensions will also be supported and you will not be restricted to naming the file 'sipclient.pem'.



5 WEB INTERFACE

5.1 Stats

5.1.1 Device Status

1

Web Interface Login

ting up your IP Display Speaker:	
ep 1: Configure your IP Display Speake	er
g in with the default password and use the	Basic Settings pages to set up the basic information.
ep 2: Check network settings (Optional	1)
	ettings tab to change network settings. The default setting for the device is to obtain its IP address from a dministrator if you plan to assign a static IP address, Mask, and Gateway to the device.
ep 3: Secure your IP Display Speaker (Optional)
	ings tab to change the administrator password. tant if the device is directly connected to a public network.
ep 4: Register your IP Display Speaker	(Optional)
ase register your product using the link bel	low:
p://www.algosolutions.com/register	
gistration ensures your access to the latest	upgrades to this product and important service notices.
Login	
Password (default: algo)	Login
Device Name	displayspk-18000f
SIP Registration	Page No Account
SIP Registration Call Status	Page No Account Idle
-	
Call Status	Idle
Call Status Proxy Status	Idle Single proxy mode
Call Status Proxy Status Provisioning Status	Idle Single proxy mode None Found
Call Status Proxy Status Provisioning Status MAC	Idle Single proxy mode None Found 00:22:ee:18:00:0f
Call Status Proxy Status Provisioning Status MAC IPv4	Idle Single proxy mode None Found 00:22:ee:18:00:0f 10.30.250.48/8, Gateway: 10.0.1.1
Call Status Proxy Status Provisioning Status MAC IPv4 Date / Time	Idle Single proxy mode None Found 00:22:ee:18:00:0f 10.30.250.48/8, Gateway: 10.0.1.1 Thu Feb 9 16:36:12 GMT 2023
Call Status Proxy Status Provisioning Status MAC IPv4 Date / Time Multicast Mode	Idle Single proxy mode None Found 00:22:ee:18:00:0f 10.30.250.48/8, Gateway: 10.0.1.1 Thu Feb 9 16:36:12 GMT 2023 Disabled
Call Status Proxy Status Provisioning Status MAC IPv4 Date / Time Multicast Mode Volume	Idle Single proxy mode None Found 00:22:ee:18:00:0f 10.30.250.48/8, Gateway: 10.0.1.1 Thu Feb 9 16:36:12 GMT 2023 Disabled Page Volume: 4 (-18dB)
Call Status Proxy Status Provisioning Status MAC IPv4 Date / Time Multicast Mode Volume PoE Detection	Idle Single proxy mode None Found 00:22:ee:18:00:0f 10.30.250.48/8, Gateway: 10.0.1.1 Thu Feb 9 16:36:12 GMT 2023 Disabled Page Volume: 4 (-18dB) PoE++ 802.3bt Type 3 (Max 51W)

Figure 1: 8410 Status

The web interface requires a password to login to see the device settings. The default password is **'algo'**. This password can be changed in the **Admin** tab after logging in the first time.

ALGO

8410 IP Display Speaker & 8420 IP Dual-Sided Display Speaker



Note

Web Interface is accessed by entering 8410/8420 IP Address into the web browser.

Important

It is highly recommended to change the default password if the device is directly connected to a public network.

Status

The Status page of the device will be available before and after logging on. This section can be used to check the status of the 8410 for the following:

- SIP Registration
- Call Status
- Proxy Status
- Provisioning Status
- MAC
- IPv4
- Date/Time
- Multicast Mode
- Volume
- PoE Detection
- Relay Input Status



Note

For security purposes, the Status page can be hidden when logged out through the settings under the Advanced Settings > Admin tab.

These options may change depending on how the device is configured.

5.2 Basic Settings

5.2.1 SIP

The *SIP* tab allows for the SIP server information and account credentials to be entered. This information can be obtained from your telephone system administrator or hosted account provider. After entering the information and saving the settings, go to the *Status* tab to confirm the registration was successful.



atus	Basic Settings	Display	Additional Features	Advanced Settings	System	Logout
IP Fe	eatures Multicas	st			_	
9 Sett	ings					
SIP						
This s dministi	section allows the S rator or hosted acc	SIP server in ount provid	nformation & account cr er. After saving these se	edentials to be entered attings, see the <u>Status</u>	d. This information tab to confirm se	on should be obtained from your telephone system uccessful registration.
SIP Do	main (Proxy Serve	er)				a
				Default port is 506 my_proxy.com:5070,		ifferent port, enter PROXY:PORT, e.g.
				my_proxy.com:5070,	OF 192.168.1.10	:5080.
Ring/Al	ert Mode			Monitor "Ring"	event on registe	ered SIP extension
-				ONone	-	
Ring Ex	tension					
Authen	tication ID					
Authen	tication Password					
Display	Name (Optional)					
The ringing	device will detect . It will not answe	inbound rir r the call o	ng events on this exter n this extension.	nsion and play the ale	rting tone (and	multicast if configured) until the inbound call stops
Page Ex	xtension					
Authen	tication ID					
Authen	tication Password					0
Display	Name (Optional)					
The	device will auto-ar	nswer any	inbound call received of	on this extension and	provide a voice	paging path (and multicast if configured).
						🖌 S

Figure 2: 8410 Basic Settings → SIP



Important

Anytime changes are made to settings in the web interface the **'Save'** button must be clicked to save the changes.

SIP Domain (Proxy Server)

The IP address (e.g., 192.168.1.111) or domain name (e.g., myserver.com) of the SIP Server.

Ring / Alert Mode

This is the option for adding a second SIP extension for a Ring event. If activated ("Monitor" is selected), the screen expands to show blocks for SIP extension parameters for a Ring/Alert Extension to be entered.

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.

Ring Extension

This is the SIP extension for the Ring parameter of the 8410.



Page Extension

This is the SIP extension for the Page parameter of the 8410. The device will auto-answer any inbound call received on this extension and provide a voice paging path (and multicast if configured).

Authentication ID

Also referred to as 'Username' for some SIP servers. This, in some cases, may be the same as the Ring and/or Page extension. The authentication is a name you choose to represent the page extension.

Authentication Password

This is the SIP password provided by the system administrator for the registered SIP account. Up to eight (8) characters can be implemented. The password can be used to authenticate SIP users.

Display Name

The Display Name is shown on a receiving phone to which a SIP call is made. For the display name to be shown, the PBX and phone(s) need to be configured to display this message as the Caller ID.



5.2.2 Features

atures		
nbound Ring Settings		
These settings apply to events triggered by the Ring E propriate volume level.	extension(s) & Emergency Alerts sections.	The Play/Loop/Stop buttons can also be used to test the device and set the
Ring/Alert Tone	warble2-med.wav	✓ Play Loop Stop
Ring/Alert Screen Pattern	<none></none>	×
Ring/Alert Volume	4	✓ Apply
Ring Limit	No limit () 1 ring = 6 seconds.	v
nbound Page Settings		
Page Speaker Volume	4	✓ Apply
	When in Receiver mode, n	ote that this is the default volume control for all audio received via multicast.
Page Mode		Delayed page audio temporarily, and then broadcasts it after the call is hung-up. This the Opus transmitter codec is not supported with Two-way paging.
Delayed Page	Play once OPlay twice	3
Page Timeout	5 minutes () Maximum page timeout in	▶ ✓ Delayed mode is 5 minutes.
Page Tone		n uploaded file. The other pre-installed tone files all contain silence at the end ence" of 6 seconds. This silence will block the voice path for several seconds
Page Screen Pattern	<none></none>	×
G.722 Support	Enabled Obisabled ()Applies to codec used duri	ng SIP negotiation only. Multicast codec is configured separately.
Passcode Protected Page Extensions	unintentional pages. When pr	require the caller to enter a passcode. Setting a passcode helps prevent ompted, the caller must enter the passcode followed by the # sign before the sscode prompt will be played before any other action.
Apply To All Page Extensions	Enabled Obisabled	
Passcode	i)Maximum length = 15 digi	ts
Passcode Prompt Tone	<default></default>	▶
DTMF Detection Type	Auto ORTP Telephony	Event (RFC 4733) ORTP In-band OSIP INFO
Audio Processing		
Ambient Noise Compensation	Enabled Obisabled (1) Automatically adjust speak start of each call.	er level in response to ambient noise level detected at the device prior to
Ambient Noise Compensation No Loss	Enabled Oisabled Oisabled Oisabled	se Compensation algorithm to only use levels at or above the current volume.
Ambient Noise Compensation Max Volume	10 (i) Set maximum speaker lev	► I in response to ambient noise.
Automatic Gain Control (AGC)	Enabled Obisabled (i) Automatically maximize le consistent.	vel of voice received from calling phone in order to make page volume more
Screen		
Idle Pattern	<use default="" patt<="" screen="" td=""><td>ern> 🗸</td></use>	ern> 🗸
Screen Brightness	5	



Figure 3: 8410 Basic Settings → Features

Inbound Ring Settings

Ring settings apply to events triggered by Ring Extensions and Emergency Alerts.

Ring/Alert Tone

Select an audio file to play when a ring event is detected on the SIP Ring Extension. The audio file may be played immediately to an associated speaker from the web interface for test purposes using the Play, Loop, and Stop buttons. During multicast, the device will broadcast an audio stream using the Sender's selected ring tone.



This is the "Default" tone that will be played if selected for Multicast, Additional Ring Extension settings.

Ring/Alert Screen Pattern

Note

Select a screen pattern to display on the LCD screen on inbound ring events. Screen patterns are configured under Display \rightarrow Screens.

Ring/Alert Volume

Set the speaker volume for a SIP Ring event. This setting is an amplifier gain control, and the output level will depend on the levels recorded into the source audio file played from memory. This setting is only used for local tones, and not when receiving multicast (see Page Speaker Volume below).

Ring Limit

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

Inbound Page Settings

Page Speaker Volume

This is the Page Speaker Volume control for SIP or multicast paging. This is an amplifier gain control and the output level will depend on streaming level. This setting will apply to all inbound multicast streams (for Receiver mode only), regardless of content.

Page Mode

A call to the SIP page extension can be one-way, two-way (using an external microphone), or delayed. In delay mode, the speaker will store the page into memory and then play after disconnecting.

Delayed Page

Delayed Page allows for a user to record a message before it is played over the speakers. To cancel a page while in delay mode, press "*" to while the recording state is in process to prevent it from being sent after hanging up.



Page Timeout

Page Timeout is the maximum duration for a page. The call will be terminated when the timeout occurs whether anyone is speaking or not. This is useful for situations when someone accidentally forgets to hang up, preventing the paging system from getting stuck in the active state. A time limit may be set for an active page.

Page Tone

Select a pre-announce tone for paging. This tone will play to announce a Page is starting. Use only the Default or custom uploaded files. Other pre-installed tone files contain silence at the end to generate ring "cadence" of 6 seconds. This silence will block the voice path for several seconds at the start of a page. The "Default" tone will play the page-notif.wav file.



Note

The "Default Page Tone", in Advanced Multicast, will play the tone set here.

Page Screen Pattern

Select a screen pattern to display on the LCD screen on inbound page events. Screen patterns are configured under Display \rightarrow Screens.

G.722 Support

G.722 enables wideband audio for optimum speech intelligibility. Enable or disable the G.722 codec.

Passcode Protected Page Extensions

When enabled, the caller must enter the passcode followed by the # sign before the page can be accepted. Setting a passcode helps prevent unintentional pages. Passcodes can be up to 15 digits and must be numbers only.

Apply to All Page Extensions

Choose to apply a passcode to all page extensions.

Passcode

Only visible when 'Passcode Protected Page Extensions' is set to 'Enabled'. Enter the desired numerical passcode (maximum length of 15 digits).

Passcode Prompt Tone

Only visible when 'Passcode Protected Page Extensions' is set to 'Enabled'. Select the tone to be played to notify 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, best known to users as the sound made when pressing a number key. In the 8410 or 8420, this is used for multi-zone selection, passcode, etc.

Audio Processing

Ambient Noise Compensation

Ambient Noise compensation will allow the speaker level to adjust automatically in response to ambient noise levels detected at the device prior to the start of each call.

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 speaker volume when this setting is enabled.

Ambient Noise Compensation Max Volume

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

Automatic Gain Control (AGC)

AGC normalizes the audio level. This ensures the audio level heard near the speaker is always at a consistent level, independent of the phone that is used to answer the call.

Screen

Idle Pattern

Select a pattern to be displayed on the screen for when the 8410 or 8420 is idle. Patterns can include up to several screen displays and can vary to include a single image, a notification template, a clock display, and other configurations.

Screen Brightness

The screen brightness level can be adjusted according to the placement of the screen within a building to optimize visibility of content. Select a brightness level for the display screen on the 8410 or 8420.

5.2.3 Multicast

Multicast IP Addresses

Each 8410 or 8420 has its own IP address and shares a common multicast IP and port number (multicast zone) for multicast packets. The Sender transmits to a configurable multicast zone, and the Receiver 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 Transmitter and Receiver 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 8410 or 8420 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.

Multicast Page Zones

The 8410 or 8420 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 down on the list – with music being the lowest priority.

- 1. Priority
- 2. All Call
- 3. Zone 1
- 4. Zone 2
- 5. Zone 3
- 6. Zone 4
- 7. Zone 5
- 8. Zone 6
- 9. Music

"Expanded" zones can also be enabled, in *Basic Settings* \rightarrow *Multicast*, allowing up to 50 zones in total. These have the same behaviors as the basic zones but are hidden by default to simplify the interface.



5.3 Multicast (Transmitter/Sender Settings)

Status	Basic Se	ettings	Display	Additional Features	Advanced Settings System Logout
SIP	Features	Multica	st		
Multic	ast Setti	ings			
Multi	icast Mod	le			
Multi	cast Mode				ONone ●Transmitter (Sender) OReceiver (Listener) ④Multicast Zone Definitions can be found in "Advanced Settings > <u>Advanced Multicast</u> ".
Multi	cast 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.
Numi	ber of Zone	25			Basic Zones Only OBasic and Expanded Zones
Tran	smitter (S	Sender)	Zone Se	ettings	
Zone	Selection M	Mode			OTMF Selectable Zone Osingle Zone For additional capabilities allowing unique SIP extensions per zone, see "Additional Features > More Page Extensions".
Trans	smitter Sing	gle 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.
Spea	ker Playbac	ck Zones			 Priority Call Call 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 <u>More Page Extensions</u> per zone) and wishing to make the Transmitter a member of only certain zones.
DTMI	F Setting	s			
Zone	Selection 1	Tone			<default></default>
Two	Digit Select	tion			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

Figure 4: Multicast transmitter mode settings



Note

See (Advanced Settings > Advanced Multicast) section for more information on populated IP values.

Multicast Mode

Multicast Mode (Transmitter/Sender Selected)

If Transmitter mode is enabled the 8410 or 8420 will broadcast an IP stream when activated in addition to playing the audio. (Note that the 8410 or 8420 cannot be both a multicast Transmitter and Receiver simultaneously).

Multicast Type

The 8410 or 8420 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 8410 or 8420 with the 'Polycom Zone' (IP Address and Port) and 'Polycom Default Channel'.



Note

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

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

Number of Zones

Select 'Basic Zones Only' if configuring nine or fewer multicast zones (shown beside 'Speaker Playback Zones') or 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.

Transmitter (Sender) Zone Settings

Polycom Group Selection Mode

'Single Zone' always broadcasts on one pre-configured Polycom Group. In 'DTMF Selectable Zone' mode, the group is determined by the DTMF selection between 1 – 50.

Zone Selection Mode

"Single Zone" mode always broadcasts on one IP address.



Note

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 \rightarrow More Page Extensions.

'DTMF Selectable Zone' mode, offers dynamic zone selection and requires only the Transmitter device to have a registered SIP Extension. The zone definitions can be found in the Advanced Settings \rightarrow Advanced Multicast tab.

In 'DTMF Selectable Mode', to page, dial the SIP extension of the Transmitter device: ####, then dial the desired DTMF page zone (e.g., 1, 2, etc.) on the keypad when prompted.

- 1. Press DTMF Extension 9 for Priority Call
- 2. Press DTMF Extension 0 (or 8) for All Call
- 3. Press DTMF Extension 1 for Zone 1
- 4. Press DTMF Extension *10 for Zone 10
- 5. Press DTMF Extension *11 for Zone 11



Note

DTMF codes for zones 10 and higher start with an "*" All DTMF codes and respective zones are available in Advanced Settings \rightarrow Advanced Multicast.



Zone Selection Tone

Only visible when "Zone Selection Mode" is set to 'DTMF Selectable Zone'. The tone played over the phone to prompt the user to select a zone to multicast to.

Transmitter Single Zone

The zone that multicast stream will be sent to by default. If 'DTMF Selectable Zone' is chosen above, this setting will not apply to Paging, since the zone now must be dynamically selected per call via DTMF. However, the specified Transmitter 'Single Zone' setting is still used for any multicast events triggered by the Ring, analog input, or the relay input.



Note

The Transmitter Single Zone is the default zone used for any multicast actions unless an option is created for a custom zone with specific parameters.

Speaker Playback Zones

The Speaker Playback Zones allows the Transmitter device to play audio for selected zones only. This is useful if using the DTMF Selectable Zone mode (or More Page Extensions per zone) with the intention of making the Transmitter unit a member of only certain zones. In this case, the Transmitter does not participate in the Zone, but it transmits certain traffic.

Expanded Speaker Playback Zones

Up to 50 zones can be shown and are only visible when 'Basic and Expanded Zones' is selected.

DTMF Settings

Zone Selection Tone

This is the prompt to select a zone. This may be used as an interactive voice response (IVR) menu by uploading a custom audio file through *System* \rightarrow *File Manager* in the Tones folder. Each zone may use a different tone. This can be configured in *Advanced Settings* \rightarrow *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 '*'.



5.4 Multicast (Receiver Settings)

Status Basic Settings Display Additiona	al Features Advanced Settings System Logout
SIP Features Multicast	
ulticast Settings	
Multicast Mode	
Multicast Mode	○None ○Transmitter (Sender) ●Receiver (Listener) ④Multicast Zone Definitions can be found in "Advanced Settings > <u>Advanced Multicast</u> ".
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	OBasic Zones Only Basic and Expanded Zones
Receiver (Listener) Zone Settings	
Basic Receiver Zones	Priority Call Vall Call Music Zone 1 Zone 2 Zone 3 Call Call Zone 3 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.
Expanded Receiver 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 *32 Zone *33 Zone *34 Zone *30 Zone *36 Zone *37 Zone *38 Zone *39 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 Zone *48 Zone *49
	✓ Save



Multicast Mode

Multicast Mode (Receiver Selected)

If Receiver mode is enabled the 8410 or 8420 will activate when receiving a multicast message. It will mimic the audio stream but use local volume settings ('Page Speaker Volume' in *Basic Settings* \rightarrow *Features*).

Multicast Type - Regular

Select "Regular" if receiving multicast from other Algo SIP endpoint(s) and/or multicast- enabled phone(s) that use RTP audio packets.

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 Receiver zones but are hidden by default to simplify the interface.



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

The 8410 or 8420 may receive multicast paging compatible with Polycom "on premise group paging" protocol.

To configure the 8410 or 8420 as a Receiver to play Polycom page announcements, select 'Group Page' or 'Push-to-Talk'. Then enter the Polycom Zone (IP Address and Port) that matches the configuration of the Polycom phones and Channels. The 'Default Channel' is the target group in a Polycom paging environment.

Status Basic Settings	Additional Features Scheduler Advanced Settings System Logout
SIP Features Multicas	t i i i i i i i i i i i i i i i i i i i
ulticast Settings	
Multicast Mode	
Multicast Mode	ONone OTransmitter (Sender) <a>Receiver (Listener) (i)Multicast Zone Definitions can be found in "Advanced Settings > <u>Advanced Multicast</u> ".
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.
Polycom Group Paging Polycom Zone	g/Push-to-Talk 224.0.1.116:5001 (i)Enter the same Multicast IP Address & Port number as configured on the Polycom phones.
Polycom Receiver Channels	Is Image: 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 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.
	Sav

The Polycom phone used as page audio source for the 8410 or 8420 must be configured to use either the G.711 or G.722 audio codec. **The Polycom phone(s) must also be configured with the "Compatibility" setting** ("ptt.compatibilityMode") disabled in order for this codec setting to be applied.

If using a Polycom phone as the Multicast Transmitter, a tone may be set for any of the 25 Polycom Groups configured on the Algo device. If an Algo device is used as a Multicast Transmitter, a tone does not have to be set as the Algo Transmitter will provide its own tone. Polycom Group Tones can be set in *Advanced Settings* \rightarrow *Advanced Multicast*.

Receiver (Listener) Zone Settings

Basic Receiver Zones

Select one or more multicast zones for the 8410 or 8420 to subscribe to.



Note

Multicast zone priority is based on the zone definition list order (from top to bottom) in Advanced Settings \rightarrow Advanced Multicast.



Expanded Receiver Zones

Up to 50 zones can be shown, however, they are only visible when 'Basic and Expanded Zones' is selected.

5.5 Display

The 8410 and 8420 allow flexible visual communication via the display. The displays can be configured to deliver a variety of notification messages, such as clocks, scrolling texts, slideshow-style images, flashing announcements or general messages. The power of the device comes down to optimizing the screens for your application. In the next section, we will go through how to configure different slide options.

Within the Display tab, there are three (3) tab options: Screens, Slides, Data. The Screens tab allows you to select a slide to display on the device, the Slides tab allows you to create new or existing slides to use, the Data tab allows you to download images from the Algo server to use in the slides.

5.5.1 Screens

In this tab, you can dictate what will be displayed on the screen. Slides are created in the Slides tab. The Screens tab allows you to assign content to display. You can choose the screens to display to SIP or multicast events as well as the default display for when an event is not in progress.


en		
efault Screen		
umber of Slides	4	
lides		School date and time Analog clock
lide Duration (seconds)	5	1
Strobe Flash Pattern	Off	 Image: A set of the set of the
Test	Start Stop	
creens		
Name	Screen 1	
Number of Slides	1	2
Slides	<none></none>	2
Strobe Flash Pattern	Off	2
Test	Start Stop	
Name	Screen 2]
Number of Slides	1	2
Slides	<none></none>	2
Strobe Flash Pattern	Off	2
Test	Start Stop	
Name	Screen 9	
Number of Slides	1	2
Slides	<none></none>	2
Strobe Flash Pattern	Off	2
Test	Start Stop	
Name	Screen 10]
Number of Slides	1	2
Slides	<none></none>	•
Strobe Flash Pattern	Off	2
Test	Start Stop	

Default Screen

Number of Slides

Select the number of slides to cycle through on the display screen.

Slides

In order from left to right, top to bottom, select each slide to display. Slides can be created under Display \rightarrow Slides.



Slide Duration (seconds)

Choose how long each slide will show on the display screen in seconds.

Strobe Flash Pattern

Choose a flash pattern for the LEDs. If enabled, this will be displayed while the default screen is on.

Strobe Brightness

When strobe flash pattern is selected, the option for the strobe brightness can be selected.

Strobe Color

Select a color to display.

Screens

Name

Choose a name that coordinates with the intention of your screen's usage.

Number of Slides

Select the number of slides you wish to display for the screen display.

Slides

In order from left to right, top to bottom, select each slide to display. Slides can be created under Display \rightarrow Slides.

Strobe Flash Pattern

Choose a flash pattern for the LEDs. If enabled, this will be displayed while the default screen is on.

5.5.2 Slides

Slides can be created or edited in this tab based on event. Preconfigured templates are provided to customize content.



Status	Basic Settings	Display	Additional Features	Advanced Settings	System	Lo	Logout	
Screens	Slides Dat	а						
Slides								
Confi	gured Slides							
Name	2							
Analo	g clock			Start Test	Stop Test	Edit	lit Delete	
				Start rest	btop lest	Luit		
Digita	l clock			Start Test	Stop Test	Edit	lit Delete	
Schoo	l date and time			Charle Track	Charle Track		The state of the s	
				Start Test	Stop Test	Edit	lit Delete	
Today	is			Start Test	Stop Test	Edit	lit Delete	
-								_
lime	and date backgro	una		Start Test	Stop Test	Edit	lit Delete	
				Add Slide				
				Add Silde				
<u> </u>								

Edit preconfigured slides or add new slides to customize your experience using the 8410 with your company's branding.

Add Slides

Press **Add Slide** to create a new slide based on preconfigured templates. Slide types may include an analog or digital clock display, a new template for notifications or alerts.

Clock

The digital clock setting allows you to change background and font colors.

Template

Choose a template for your Slide based on the purpose. This can include emergency alerts, notifications, weather warnings, or event updates.



5.5.3 Data

	Logout	System	Advanced Settings	Additional Features	Display	Settings	Basic	Status
						s Data	s Slid	Screens
								Data
								Files
				- Comun	m the Ale	an films for	land im	
			- Downlo	o Server	m the Alg	ge files fro	ioad im	Downi
om <u>the Algo website</u> .	whiloaded manual	is can be do	(1) Image file					
								L
					ta	store Da	up / R	Backu
		ad	🕹 Downlo		2	e Data File	load Sli	Downl
						Data File	ra Slida	Pector
TRestore	chosen	le No file	Choose Fi			Jata File	re silde	Restor
			🍓 Clear				All Data	Clear
	*****							l
								L
Restore	chosen	ad ile No file	Choose Fi				load Sli re Slide	Downl Restor

Download Image Files from Algo Server

Downloads the factory images from the Algo Server (internet connection is required).

Download Slide Data File

Allows an image to be downloaded and stored for backup purposes. Note that this backup is independent from the rest of the configuration backup on the device.

Restore Slide Data File

Upload and restore a saved slide data file.

Clear All Data

This clears all the slide data, including saved slides.



5.6 Additional Features

5.6.1 Input/Output

The *Input* tab allows external accessories to be connected to the 8410 or 8420. This is a dry contact input which can be configured as 'normally open' or 'normally closed' mode. Algo offers accessories, such as the 1202, 1203, 1204, and 2507. Third-party accessories/systems may also be used provided they have a dry contact output.

put	
eneral	
Relay Input Mode	ODisabled
telay Input Houe	
	Relay Normally Open with Supervision (e.g. Algo 1203 Call Switch)
	ORelay Normally Closed
	ORelay Normally Closed with Supervision OMute Switch
	OMute Switch with Supervision
	Algo 1202 Call Button
	Algo 1204 Volume Control Switch (Local or Remote)
	OAlgo 1204 Volume Control Switch with Supervision (Local or Remote) OAlgo 2507 Ring Detector
ction When Input Triggered	
Action	Play Tone OMake SIP Voice Call OMake SIP Call with Tone OStream Mic Audio ()"Play Tone" will play a recorded audio file to a local speaker and multicast if configured. "Stream Mic
	Audio" will stream microphone audio to multicast only, so it requires Multicast "Transmitter" mode to be
	enabled in "Basic Settings > <u>Multicast</u> ".
Tone/Pre-recorded Announcement	chime.wav 🗸
Tone Duration	Play Once OPlay While Held OPlay Until Completion
Action When Tamper Detected	
Wiring Fault Supervision Mode	Detect Open Circuit Fault Only
	Opetect Both Open Circuit & Short Circuit Faults
	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 OMake SIP Voice Call OMake SIP Call with Tone
	IPlay Tone" will play a recorded audio file to a local speaker and multicast if configured. "Stream Mic Audio" will stream microphone audio to multicast only, so it requires Multicast "Transmitter" mode to be
	enabled in "Basic Settings > Multicast". 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 Announcement	buzzer.wav 🗸
Tone Duration	Play Once OPlay While Held OPlay Until Completion
one Multicast Settings	
Use Separate Multicast	
	This will allow the tone to be played via multicast even if the device is configured as a receiver.

Figure 6: Input settings



General

Relay Input Mode

The input relay to the 8410 and 8420 can be activated by any normally open or normally closed switch. Algo offers the 1202 Call Button, the 1203 Call Switch, or 1204 Volume Control Switch. Via supervision settings, notification actions can also be triggered if the input switch is disconnected.

1203 Call Switch

The 1203 Call Switch is a simple contact closure switch with an illuminated button and supervision capabilities. When used in conjunction with the 8410, the 1203 can prompt a single action with one-touch, or a continuous action if the button is held.



Figure 7: 1203 Call Switch

Mute Switch

Apply an external switch (short-circuit) across the Relay Input terminals in order to mute the 8410 or 8420. This allows a temporary "disable" switch to control the device if desired, for example in a boardroom to block paging during important meetings.

Leave the Relay Input terminals open (no-connect) for regular full-volume operation when in this mode.

Multicast Override

Allow selected multicast zones to override the Mute Switch settings for the selected zones.

1202 Call Button

The 1202 Call Button is a one-touch button for event notification and response. It can be used with the 8410 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.



Figure 8: 1202 Call Button – the insert card is interchangeable

While the 8410 and 8420 can be configured to play the audio file only once, it can also be enabled to play it continuously with just a press of the 1202 Call Button. The action can then be stopped via the smaller oval cancel button located below the main call button on the 1202 Call Button.



1204 Volume Control

The 1204 Volume Control Switch is a simple two terminal potentiometer that will allow attenuation below the maximum volume level (configured under *Basic Settings* \rightarrow *Features*).



Figure 9: 1204 Volume Control

Mute On Lowest Setting

Enabling 'Mute On Lowest Setting' allows audio to be completely muted when volume control switch is turned all the way down.

Wire Length

This allows you 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

Configure the device to subscribe to remote 1204 volume input or to notify remote devices of 1204 volume input.



Note

RESTful API must be enabled in the Advanced Settings \rightarrow Admin tab.

Notify (Local 1204) → remote device RESTful API password

Subscribe (Remote 1204)

- IP address
- Remote device RESTful API password

Action When Input Triggered

Action

Play Tone

When the 8410/8420 input is triggered, a tone or a pre-recorded audio file will play over the speaker or multicast if enabled. This function can be used to request support/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:



- Tone/Pre-recorded Announcement
- Tone Duration

Make SIP Voice Call

When the 8410/8420 input is triggered, a voice path will open for an intercom-like call via an external microphone 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.

- Action When Input Triggered:
 - Extension to Dial
 - o Call Mode
 - Allow 2nd Button Press
- Outbound SIP Call Settings:
 - o Outbound Ring Limit
 - Ringback Tone
 - Maximum Call Duration

Make SIP Call with Tone

When the 8410/8420 input is triggered, a private call can be generated 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.

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

Allow 2nd Button Press

If enabled, the 2nd button press will either simply End Call or End and Restart Call. Therefore, if an input is triggered for the second time the SIP call will either simply be terminated or terminated and immediately called again.

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 ('Play Tone', 'Make Two-Way SIP Voice Call', 'Make SIP Call with Tone') in case 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 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.



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 40 mA current limit.

Action

"Play Tone" will play a recorded audio file to a local speaker and multicast if configured. "Stream Mic Audio" will stream microphone audio to multicast only, so it requires multicast "Transmitter" mode to be enabled in *Basic Settings* \rightarrow *Multicast*.



Note

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.

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

Tone/Pre-recorded Announcement (Action – Play Tone / Make SIP call with Tone)

Select a recording or tone to use. Custom audio files may be used and uploaded though System \rightarrow File Manager.

Extension to Dial (Action – Make SIP Voice Call)

SIP account required in Page Extension fields in order to make a call. Can be configured if 'Make SIP Voice Call' or 'Make SIP Call with Tone' actions are enabled under 'Call Button Settings'.

Interval Between Tones (Action – Make SIP call with Tone)

Specify the time delay (seconds) between tones. Can be configured if 'Play Tone' or 'Make SIP Call with Tone' actions are enabled under 'Call Button Settings'.

Maximum Tone Duration (Action – Play Tone / Make SIP call with Tone)

Select the maximum tone duration. The tone will be terminated once the maximum time is reached. Can be configured if 'Play Tone' or 'Make SIP Call with Tone' actions are enabled.

Tone Multicast Settings

Use Separate Multicast

This allows the tone to be played via multicast even if the 8410/8420 is configured as receiver. See additional options when enabled.

- Multicast Mode
- IP Address
- Port



Outbound SIP Call Settings

Outbound Ring Limit

Typically set to ensure that a call will not reach voicemail. This feature, under 'Outbound SIP Call Settings', can be used to set a limit on how long the speaker will ring before timing out.

Ringback Tone

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



5.6.2 Emergency Alerts

atus Basic Settings Display Add		Advanced Settings System	Logout	
to go back, hold to see history ore Page	Extensions More	e Ring Extensions	_	
nergency Alerts				
erent "Cancel" extension is called to clear rt"), allowing staff to quickly dial a pre-	ar the announceme configured number	nt (or a pre-defined timeout is and then exit the building. Audi	eached). This ca o files can be ea:	ging up. The announcement will continue to play until a in be useful for emergency notifications (e.g. "Evacuatio sily uploaded to create custom announcements.
Up to 10 extensions can be registered a cel the currently active announcement.	Illowing up to 10 dif	fferent announcements. A single	e "Cancel" extens	sion also needs to be registered; calling this number wi
Note: Some SIP phone systems may no	t support this featu	re if they limit the number of e	xtensions that ca	n be registered on a single device.
Settings				
Default Announcement Duration		OPlay Once OPlay U		
Default Maximum Announcement Time Announcement Selection Mode			to register a separ	e rate extension for each announcement. Use "DTMF t accepts DTMF input to select which announcement to play.
Answer Inbound Call		Carabled Disabled This option selects how Announcement is started w Extension is called. Galect "Enabled" to ans starting the alert, then aut	the Announcement when the appropriat wer the inbound ca omatically release	t calls are handled. In both cases, the Emergency te extension is called and continues until the Cancel all and provide the option to play a confirmation tone before
Passcode Protected Announcement Exte	ensions	OEnabled OEnabled		
Announcement Passcode		Setting a passcode helps p the passcode followed by t	revent unintention he # sign before th	er dialing an announcement or call-to-cancel extension. al announcements. When prompted, the caller must enter he announcement will be played or canceled. The passcode . If the passcode is not correctly entered within 15 seconds,
		the call will be ended.		
Passcode Prompt Tone			~	
Passcode Prompt Tone		the call will be ended.	~	
Passcode Prompt Tone DTMF Selection		the call will be ended.		
DTMF Selection		the call will be ended.	✓	
DTMF Selection Extension		the call will be ended.		
DTMF Selection Extension Authentication ID		the call will be ended.	✓ 	2
DTMF Selection Extension Authentication ID Authentication Password		the call will be ended.		2
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional)		the call will be ended.		
Extension Authentication ID Authentication Password Display Name (Optional) Realm		the call will be ended.		
-		the call will be ended. CDefault> CDefault> CDefault> CDefault> CDefault> CDefault> CDefault> CDef	• • • • • • • • • • • • • • • • • • •	
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm Prompt Tone Call-to-Cancel		the call will be ended. CDefault> COEfault> COEfault> COEfault> COEfault> COEfault> COEfau	() () () () () () () () () () () () () (
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm Prompt Tone Call-to-Cancel		the call will be ended. Control of the call will be ended. Control of the call will be ended. Control of the call be end	() () () () () () () () () () () () () (ection extension and select 0 to cancel the announcement.
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm Prompt Tone Call-to-Cancel Call-to-Cancel Selection Mode		the call will be ended. Control of the call will be ended. Control of the call will be ended. Control of the call be end	() () () () () () () () () () () () () (
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm Prompt Tone Call-to-Cancel Call-to-Cancel Selection Mode Extension		the call will be ended. Control of the call will be ended. Control of the call will be ended. Control of the call be end	() () () () () () () () () () () () () (
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm Prompt Tone Call-to-Cancel Call-to-Cancel Selection Mode Extension Authentication ID		the call will be ended. Control of the call will be ended. Control of the call will be ended. Control of the call be end	() () () () () () () () () () () () () (ection extension and select 0 to cancel the announcement.
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm Prompt Tone Call-to-Cancel Call-to-Cancel Selection Mode Extension Authentication ID Authentication Password		the call will be ended. Control of the call will be ended. Control of the call will be ended. Control of the call be end	DTMF 0 he main DTMF Sel	ection extension and select 0 to cancel the announcement.
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm		the call will be ended. Control of the call will be ended. Control of the call will be ended. Control of the call be end	DTMF 0 he main DTMF Sel	ection extension and select 0 to cancel the announcement.
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm Prompt Tone Call-to-Cancel Call-to-Cancel Selection Mode Extension Authentication ID Authentication Password Display Name (Optional)		the call will be ended. Control of the call will be ended. Control of the call will be ended. Control of the call be end	DTMF 0 he main DTMF Sel	ection extension and select 0 to cancel the announcement.
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm Prompt Tone Call-to-Cancel Call-to-Cancel Selection Mode Extension Authentication ID Authentication Password Display Name (Optional) Realm Confirmation Tone		Contraction of the second	DTMF 0 he main DTMF Sel	ection extension and select 0 to cancel the announcement.
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm Prompt Tone Call-to-Cancel Call-to-Cancel Selection Mode Extension Authentication ID Authentication Password Display Name (Optional) Realm		e a constraint of the call will be ended.	DTMF 0 he main DTMF Sel	ection extension and select 0 to cancel the announcement.
DTMF Selection Extension Authentication ID Authentication Password Display Name (Optional) Realm Prompt Tone Call-to-Cancel Call-to-Cancel Selection Mode Extension Authentication ID Authentication Password Display Name (Optional) Realm Confirmation Tone		Contraction of the second	DTMF 0 he main DTMF Sel	ection extension and select 0 to cancel the announcement.

Figure 10: Emergency Alerts



Emergency Alerts allow for an announcement to be triggered and latched by calling a pre-configured Emergency extension and hanging up. Emergency Alerts are useful for emergency notifications (e.g., evacuation, lock down, medical emergency, etc.), allowing staff to quickly dial a pre-configured number under such circumstances.

Settings

Default Announcement Duration

The announcement can be chosen to play once or to play until cancel. 'Play Once' mode will play a single cycle of the chosen tone file, despite of its duration. 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

This represents the duration for how long the announcement plays for.

Announcement Selection Mode

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

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. 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. Select 'Disabled' to detect just the inbound Ring signal, but not actually answer the call.

If the 'Answer Inbound Call' option is 'Enabled' the call is auto answered and a configurable confirmation tone is played before starting the alert. If 'Disabled', the alert is triggered just by the inbound ring, without answering the call. (In both instances, the announcement will play until the time limit is reached or the 'Cancel Extension' is called). The auto-answering option can be useful when the caller cannot hear announcement from their location. However, in instances where the call might go to a group/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

When enabled, this setting requires the caller to enter a passcode after dialing an announcement or call-to-cancel extension. Setting a passcode helps prevent unintentional announcements.

Announcement Passcode

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

Extension

This is the SIP extension for the DTMF Selection parameter of the 8410/8420. 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 may also be called Username for some SIP servers and in some cases may be the same as the Ring and/or Page extension.

Authentication Password

This is 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) will have to 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

Call-to-Cancel Selection Mode

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

Extension

This is the SIP extension for the Call-to-Cancel Selection parameter of the 8410. 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 may also be called Username for some SIP servers and in some cases may be the same as the Ring and/or Page extension.

Authentication Password

The SIP password is 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) will have to 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

Announcement 1

To configure an emergency alert extension, select 'Enable' beside the target announcement and enter the Extension, Authentication ID, and Authentication password.

Up to 10 extensions can be registered allowing up to 10 different announcements. Audio files can also be easily uploaded to create custom announcements. Only one 'Call-to-Cancel' extension is needed, despite the number of the alert extensions.



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.

Announcement Duration

Choose how long to allow for the announcement duration. An announcement can:

- Play Once
- Play Until Cancelled
- Default

Extension

This is the SIP extension for the Call-to-Cancel Selection parameter of the 8410. 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 may also be called Username for some SIP servers and in some cases may be the same as the Ring and/or Page extension.

Authentication Password

The SIP password is 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) will have to be configured to display this message as the Caller ID.

5.6.3 More Page Extensions

Sta	atus B	Basic Settings	Display	Additional Feature	ures	Advanced Settings	System	Logout		
Inp	out Er	mergency Alerts	More	Page Extensions	Мо	re Ring Extensions				
Mo	More Page Extensions									
be c prog This	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 orogrammed 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. The 8410 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									
activ	 In our own auto-answer any induction cars received on these numbers and provide a voice paging path and mutucast in compared. Note that only a single can can be active at a time. 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. 									
_	Where some support on systems may not support on reacting in the number of extensions that can be registered on a single device. (i) Multicast Zone Definitions can be found in "Advanced Settings > <u>Advanced Multicast</u> ".									
		tensions			, Dett					
Ē		ension 2				Enabled				
E	Extension	n]	
4	Authentio	cation ID								
4	Authentio	cation Password	d						<u>0</u>	
C	Display N	lame (Optional)							
5	Screen Pa	attern				<use defau<="" td=""><td>t Screen Patte</td><td>rn> 🗸</td><td></td></use>	t Screen Patte	rn> 🗸		
F	age Ext	ension 3				OEnabled	Disabled			
F	age Ext	ension 4				OEnabled	Disabled			
F	age Ext	ension 5					Disabled			
F	age Ext	ension 6				OEnabled	Disabled			
F	age Ext	ension 7				OEnabled	Disabled			
F	age Ext	ension 8				OEnabled	Disabled			
F	age Ext	ension 9				OEnabled	Oisabled			
F	age Ext	ension 10				OEnabled	Disabled			
									✓ Save	

Figure 11: More page extensions

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

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



The 8410/8420 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.



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.

5.6.4 More Ring Extensions

Status Basic Settings Display Additional Features Ad	dvanced Settings System Logout								
Input Emergency Alerts More Page Extensions More Ring	g Extensions								
Aore Ring Extensions									
This section allows additional extensions to be registered for the purpose of providing loud ringing alerts for more than one line. Unique ring tones and screen patterns can e 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 nust be configured on your SIP phone system of course in order to trigger it to send calls to these different numbers.									
(i) The 8410 will detect inbound ring events on these numbers and	🕖 The 8410 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	f they limit the number of extensions that can be registered on a single device.								
Ring Extensions Disabled To edit ring extensions, enable ring extensions in "Basic Settings >	CTD ^a								
	<u></u>								
Custom Ring Tone Allows the device to play a custom ringtone when a call is received Enabled Disabled	with the "Alert-Info" SIP header. Tones can be set to display a certain strobe pattern.								
Screen-Tone Mapping									
Select which tone will cause the pattern to display. Note: For duplication									
Default Screen	(?)								
Screen 1	(?) •								
Screen 2	(?) 🗸								
Screen 3	(?)								
Screen 4	(?)								
Screen 5	(?) ✓								
Screen 6	(?) 🗸								
Screen 7	(?) 🗸								
Screen 8	(?) 🗸								
Screen 9	(?) 🗸								
Screen 10	(?) 🗸								
	✓ Save								

Figure 12: More ring extensions - screen options

Up to 10 SIP Ring extensions can be registered. To configure additional ring extensions, select 'Enable' beside the target extension and enter the Extension, Authentication ID, and Authentication Password. A unique Ring Tone and Multicast Zone can be assigned to each extension if desired.

Default Screen

Select a default screen to display during a ring.



5.7 Advanced Settings

5.7.1 Network

Status	Basic Settings	Display	Additional Features	Advanced Settings	System		Logout		
Network	Admin I	Users Time	e Provisioning A	dvanced Audio Advan	ced SIP	Advanc	ed Multica:	st	
letworl	k Settings								
Comm	on								
Interne	t Protocol			IPv4 only			~		
DNS Servers									
0113 30	UNS Servers								
L									
IPv4									
IPv4 M	ethod			Static	DHCP				
IPv4 Ac	ddress/Netmas	sk.							
				(i)Address (d	lot delimited	i)/Netma	sk (CIDR),	, e.g. 192.168.1.23/24	
IPv4 Ga	ateway								
802.10	Q Virtual LAI	N							
VLAN M	1ode			ONone 🤇	Manual	Auto			
VLAN I	D			0					
				(i) Value rang	e: 0 to 4094	4			
VLAN P	riority			0					
				🕕 Value rang	e: 0 to 7				
802.13	(Port-based	Network	Access Control						
802.1X	Authentication	n		Enabled	ODisable	ed			
Authen	tication Mode			EAP-PEAP			~		
					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				(.509 certificate file renamed to 'client8021x.pem' in the 'certs'	
				folder.					
Anonyn	nous ID								
ID									
Passwo	rd						(P)		
Validate	e Server Certif	icate		OEnabled	Oisable	ed			
				(i) Validate th	e authentica	ation ser		t common authorities. To validate against additional certificates,	
				format to the				oad a Base64 encoded X.509 certificate file in .pem, .cer, or .crt	
L									
Differe	entiated Ser	vices							
SIP (6-	bit DSCP value	e)		0					
				(i) Valid value	es range fror	m 0 to 6	3		
RTP (6-	bit DSCP value	e)		0	-				
				(i) Valid value	es range fror	m 0 to 6	3		
RTCP (6	6-bit DSCP val	ue)		0	e ranco fro	m 0 to 6	2		
				(i) Valid value	s range rror	n u to 6	2		
DNC									
DNS	ahing Mada			<u></u>	0.0000	2411			
DNS Ca	aching Mode				i OSIP (lode, only th		s of DNS qu	ueries for SIP requests will be cached. In "All" mode, the results	
				of all DNS qu					
								✓ Sav	

Figure 13: Network settings



<u>Common</u>

Internet Protocol

DHCP is an IP standard designed to make administration of IP addresses simpler. When selected, DHCP will automatically configure IP addresses for each 8410 or 8420 on the network. Alternatively, the 8410/8420 can be set to a static IP address.

DNS Servers

Add one or multiple DNS servers. Separate each server by a space, comma, or semicolon.

IPv4

IPv4 Method

The 8410/8420 can be set to a DHCP or a static IP address. When DHCP is selected, the DHCP will automatically configure IP addresses for each 8410 IP Display Speaker or 8420 IP Dual-Sided Display Speaker on the network.

IPv4 Address/Netmask

Enter the static IP address and netmask (CIDR format) for the 8410/8420 (e.g., 192.168.1.23/24).

IPv4 Gateway

Enter the gateway address.

802.1Q Virtual LAN

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).

802.1X Port-based Network Access Control

802.1x Authentication

Credentials to access LAN or WLAN that have 802.1X network access control (NAC) enabled. This information will be available from the IT Administrator.

Authentication Mode

Select the desired authentication mode.

Anonymous ID

If configured, the 8410/8420 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.

Password

Enter the password.

Validate Server Certificate

Validate the authentication server against common authorities. To validate against additional certificates, go to the *System* \rightarrow *File Manger* to upload a Base64 encoded X.509 certificate file in .pem, .cer, or .crt format to the 'certs/trusted' folder.

Differentiated Services

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

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

DNS Caching Mode

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.



5.7.2 Admin

Status Basic S	Settings	Display	Additional Features	Advanced	Settings	System		Logout	
Network Adn	nin L	lsers Time	Provisioning A	dvanced Audio	Advance	d SIP	Advand	ed Multica	st
Admin Settin	gs								
Admin Pass	word								
Old Password									
Password								8	
Confirmation									
General									
Device Name	(Hostna	me)		dist	playspk-180	00f]	
Introduction S	-	-	16		On Off				
			ge when Logged Out		On Ooff				
Display Switch									
Dispidy Switch	Port IL	7 on Status P	aye			device to l	oe conn	ected to a	switch that supports LLDP or CDP.
Web Interface	Sessio	n Timeout			our			~	
				(i)A	utomatically	log out w	eb inte	rface after	period of inactivity.
Play Tone at S	tartup				Enabled C			n to confirm	n that the device has booted.
L				<u>.</u>					
Log Settings									
Log Level				0	Error (Low	est) ON	otice ("Event")	Info ("SIP") ODebug (Highest)
Log Method				0	Local ON	etwork	Both		
L									
Managemen	t								
Web Interface	Protoco	bl		۲	Both HTTP	and HTT	os Or	HTTPS Onl	y
Force Strong	asswor	d		0	Enabled	Disabled	I		
Allow Secure S	SIP Pas	swords			Enabled				
					fter enabling tore the pase			recommen	ded to re-enter SIP passwords and their corresponding realm
L					-		-		
Simple Netw	ork M	anagemen	t Protocol						
SNMP Support	t				Enabled				
				0	ownload MI	5 file <u>here</u>	•		
API Support									
RESTful API				0	Enabled	Disabler	1		
RESTIULATI								& control vi	a HTTP. Contact Algo Support for more information
									······································
System Inte	grity								
System Integr	ity Che	cking			Enabled			untam as -	ages to oncure they have not been tomograd with Each!!
				this	feature may	cause reb			ages to ensure they have not been tampered with. Enabling s to take 30 seconds longer. Verification results can be found
				on t	he Status pa	ge.			



Power over Ethernet	
PoE Power Detection	 Automatic (Recommended) OForce PoE+ OForce PoE++ (Type 3) OForce PoE++ (Type 4) Use one of the Forced PoE options only when connected to a power injector that does not automatically negotiate its power capabilities. The power injector must be capable of providing 600mA for PoE+, 600mA per pair for PoE++ Type 3, or 960mA per pair for PoE++ Type 4. Incorrect use of this setting may cause the device to reboot if the power source is not capable of delivering the selected power.
Syn-Apps	
SA-Announce Support	Enabled Obisabled
SA-Announce Server	Leave this field blank to use the server provided by DHCP Option 72.
Local Management Port	6789
InformaCast	
InformaCast Support	Enabled Disabled This feature requires a valid license to be activated. Please contact sales@algosolutions.com for assistance.
Microsoft	
Microsoft Teams Support	○Enabled ●Disabled ④Enabling this setting will provision the device via Microsoft's servers. The device reboot will take up to 5 minutes to complete.
ADMP Cloud Monitoring	
Enable ADMP Cloud Monitoring	○Enabled ●Disabled ④This feature requries a valid Account ID. Please contact support@algosolutions.com for assistance.
	✓ Save

Figure 14: Admin settings

Admin Password

Old Password

Enter the old password.

Password

Password to log into the 8410/8420 web interface. You should change the default password *algo* to secure the device on the network. If you have forgotten your password, you will need to perform a reset using the **Reset Button** to restore the password (as well as all other settings) back to the original factory default conditions.

For additional password security see "Force Strong Password" below.

Confirmation

Re-enter network admin password.

General

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.

Show Status Section on Status Page when Logged Out

Use this option if you wish to block access to the status page when logged out. The settings and configurations, on the status page, will be hidden entirely unless you're logged in – this feature is useful when you want only trusted users to view possible sensitive device information.

Display Switch Port ID on Status Page

Enable this option to display the Switch Port ID. This option requires the 8410/8420 to be connected to a switch that supports LLDP or CDP.

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 start up to confirm that the device has booted.

Log Settings

Log Level

The Log Level is to be used on the advice of Algo technical support only.

Log Method

Allows the 8410/8420 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.

Management

Web Interface Protocol

HTTPS is always enabled on the device. Use 'HTTPS Only' mode to disable HTTP, then 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.

Force Strong Password

When enabled, ensures that a secure password is provided for the device's web interface for additional protection. The password requirements 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

Allows SIP passwords to be stored in the configuration file in an encrypted format, to prevent viewing and recovery. Once enabled, the SIP 'Realm' field should be entered and all the configured Authentication Password(s) must be reentered in *Basic Settings* \rightarrow *SIP*, and any other locations where SIP extensions have been configured, to save the encrypted password(s).

If the Realm is changed at a later time, all the passwords will also need to be re-entered again to save the passwords with the new encryption.

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.

Simple Network Management Protocol

SNMP Support

Additional SNMP support is anticipated for future. The current setting of the 8410/8420 will respond to a simple status query for automated supervision. Contact Algo technical support for more information.

API Support

RESTful API

Secure API for remote access and control via HTTP.

System Integrity

System Integrity Checking

This feature verifies installed system packages to ensure they have not been tampered with by running 'Perform Check'. Enabling this feature may cause reboots and upgrades to take 30 seconds longer. Verification results can be found on the *Status* tab.

Power over Ethernet

PoE Power Detection

Use one of the Forced PoE options only when connected to a power injector that does not automatically negotiate its power capabilities. The power injector must be capable of providing 600 mA for PoE+, 600 mA per pair for PoE++ Type



3, or 960 mA per pair for PoE++ Type 4. Incorrect use of this setting may cause the device to reboot if the power source is not capable of delivering the selected power.

Syn-Apps

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 8410/8420 when Multicast Sender Mode is disabled (set to 'None') in the *Basic Settings* \rightarrow *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 for the SA-Announce Server.

InformaCast

InformaCast Support

This feature requires a valid InformaCast license to be activated. Please contact <u>sales@algosolutions.com</u> for assistance.

Microsoft

Microsoft Teams Support

Enabling this setting will provision the device via Microsoft's servers. The device reboot will take up to 5 minutes to complete.

ADMP Cloud Monitoring

Enable ADMP Cloud Monitoring

This feature requires a valid Account ID. Please contact <u>sales@algosolutions.com</u> for assistance.



5.7.3 Users

Status Basic Settings	Display Additional Features Advanced Settings System Logout
Network Admin Use	rs Time Provisioning Advanced Audio Advanced SIP Advanced Multicast
User Management Display (i) These settings enable i	a separate login account with limited access that allows the user to only modify the display
User Login	OEnabled
	Save

Figure 15: Users settings

A separate login account with limited access can be set up. The user will only be able to modify the device scheduler.

5.7.4 Time

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

Status Basic Settings Display Additional	Features Advanced Settings System Logout ning Advanced Audio Advanced SIP Advanced Multicast
ime Settings	
General	
Timezone	GMT 🗸
NTP Time Server 1	0.debian.pool.ntp.org
NTP Time Server 2	1.debian.pool.ntp.org
NTP Time Server 3	2.debian.pool.ntp.org
NTP Time Server 4	3.debian.pool.ntp.org
Supersede NTP provided by DHCP	OEnabled
Device Date/Time	Thu Feb 9 20:56:11 2023 Sync with browser
Manually Override Time	20:55:58 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.
	✓ S

Figure 16: Time settings

Timezone

Select a time zone to be used.

NTP Time Servers 1/2/3/4

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



Supersede NTP provided by DHCP

When 'Use DHCP Option 42' is chosen, if an NTP Server address is provided via the DHCP Option 42, that NTP Server will be used instead of the four (4) mentioned above. Alternatively, 'Ignore DHCP Option 42' can be chosen to only use servers mentioned above.

Device Date/Time

This field shows the current time and date as set on the device. If testing the device on a lab network that may not have access to an external NTP server, the "Sync with browser" button can be used to temporarily set the time on the device.



Note

This time value will be lost at power down or overwritten if NTP is currently active. Time and date are used for logging purposes and for the scheduler feature.



5.7.5 Provisioning

S	tatus	Basic Settings	Display	Additional Features	Advanced Settings	System	Lo	ogout	
N	letwork	Admin Us	ers Time	Provisioning	Advanced Audio Advan	ced SIP A	Advanced	d Multicast	
Pr	ovisio	oning Settin	gs						
	Mode	-	-						
ſ		Marda			— ———————————————————————————————————				
	Provisi	oning Mode							
L									
ſ	Settin								
	Server	Method				ICP Option 6 tion 66 only	56/160/1 /	150)	
					ODHCP OP	tion 160 onl	ly		
					Static	tion 150 onl	iy		
					(i)Auto mode	automatically	y checks	all 3 DHCP o	options for an active provisioning server, in the order listed.
	Static :	Server						<u>+</u>	
	Downlo	oad Method			©tftp ○	FTP Онттр	р Онтт	TPS	
	Config	Download Path							
	Firmwa	are Download Pa	ith						
	Partial	Provisioning				Oisabled			
					feature.	ort for "-i" inc	rementa	I provisioning	g files. Disable for enhanced security if not using this
	Check-	sync Behavior				eboot OCo			
									will check with the provisioning server and only reboot if vided as a parameter in the check-sync event).
					-	•			
	Sync S	tart Time							
					Schedule a option above.				to perform a sync according to the 'Check-sync Behavior'
-					option above.	Leave blank t			h
	Sync E	nd Time			() If set, the	device will svr	nc at a ra	andom time i	in the window between Start Time and End Time. Setting
					an End Time e				n overnight period. Leave blank to sync at Start Time
					exactly.				
	Sync F	requency			ODaily 🦲	Selected Da	iys Only		
	Sync D	ays			Monday	Tuesday	✓Wedr	nesday 🔽	Thursday 🗹 Friday 🗹 Saturday 🗹 Sunday
ſ									
	Zero To	ouch Provisionin	g		Turn Off ZT				
					(i) ZTP is disa	bled and can	only be r	re-enabled w	vith a factory reset.
L									
									✓ Save
									▼ Save

Figure 17: Provisioning settings



Note

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.

Mode

Provisioning Mode

Provisioning allows installers to pre-configure the 8410/8420 units prior to installation on a network. It is typically used for large deployments to save time and ensure consistent setups.



Settings

Server Method

The device can be provisioned via the Auto mode (where all three DHCP options (Option 66/160/150) will be automatically checked for an active provisioning server), just one of the three specified DHCP options, or a Static Server. In addition, there are four different ways to download provisioning files from a "Provisioning Server": TFTP (Trivial File Transfer Protocol), FTP, HTTP, or HTTPS.

Static Server

Enter the server address or domain.

Download Method

The 8410/8420 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.



Important

DHCP must be enabled if using DHCP Option 66/160/150, in order for Provisioning to work.

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

- Generic (for all Algo 8410 IP Display Speaker) algop8410.conf
- Specific (for a specific MAC address) algom[MAC].conf

Both protocol and path are supported for Option 66, allowing for <u>http://myserver.com/config-path</u> to be used.

Config Download Path

Enter the path where the configuration file Is located within the provisioning server (e.g. algo/config/8410).

Firmware Download Path

Enter the path where the firmware file Is located within the provisioning server (e.g. algo/firmware/8410).

Partial Provisioning

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

Check-sync Behavior

If 'Conditional Reboot' is selected, the device will check with the provisioning server and only reboot if a 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 the Start Time indicates an overnight period. Leave blank to sync at Start Time exactly.

Sync Frequency

Choose the frequency for which this setting should occur. Select between daily or go to 'Sync Days' to choose specific days of the week.

Sync Days

Select the days of the week to apply this setting for.

MD5 Checksum

In addition to the **.conf** file, an **.md5** checksum file must also be uploaded to the Provisioning server (for TFTP mode only). 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 and 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

If using a generic configuration file, extensions and credentials have to be entered manually once the 8410/8420 has automatically downloaded the configuration file.

To see Algo's SIP endpoint provisioning guide, visit: www.algosolutions.com/provision

Generating a Specific Configuration File

The specific configuration file will only be downloaded by the 8410/8420 with the MAC address specified in the configuration file name. Since all the necessary settings can be included in this file, the 8410/8420 will be ready to work immediately after the configuration file is downloaded. The MAC address of each 8410/8420 can be found on the back label of the unit.

To see Algo's SIP endpoint provisioning guide, visit: <u>www.algosolutions.com/provision</u>



5.7.6 Advanced Audio

Status	Basic Setting	s Display	Additional Featu	res Advanced Se	ettings	System		Logout	
Networ	k Admin	Jsers Time	Provisioning	Advanced Audio	Advan	ced SIP	Advanc	ed Multi	cast
Advan	ced Audio	unctions							
Func	tions								
Dyna	mic Range Con	pression (DR	(C)	0	Enabled	Opisable	d		
-,			-,					f page a	udio to increase loudness.
Jitter	Buffer Range (milliseconds,	10 ~ 500)	100					
						buffering if ly is recomm		ry to cor	rect for inconsistent delays on the network. Use of the lowest
٨١٠٠٠	ys Send RTP M	- in-			2				
Aiwa	ys Send KTP Me	sula			Enabled		a		
- Audi	o Filters								
	ker Filter			No				~	
Spea	Kei Fillei					also limited	by audi		
Spea	ker Noise Filter			0	Enabled	Olisable	d		
				(i) A	ggressive	8th order E	lliptical I	ilter (fc	= 145Hz)
Micro	phone Filter			Nor	ne			~	
Micro	phone Noise Fi	ter				 Disable 			
				(i) A	ggressive	8th order E	lliptical I	Filter (fc	= 145Hz)
	ophone					<u></u>]
Globa	al Microphone N	lute				Disable Dis will disab		icrophon	e entirely.
Micro	phone Volume			Hig	ıh			~	

1									
									✓ Save

Figure 18: Advanced audio settings

Functions

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.

Always Send RTP Media

If enabled, audio packets will be sent at all times, even during one-way paging mode. This option is needed in cases when the server expects to see audio packets at all times.



Audio Filters

Speaker Filter

Applies a high-pass filter to the speaker output. Used to reduce audio artifacts like humming or buzzing by filtering out unwanted frequencies.

Speaker Noise Filter

Enables heavy filtering below 145 Hz to reduce mains induced noise (fans).

Microphone Filter

Applies a high-pass filter to the microphone input. Used to reduce audio artifacts like humming or buzzing by filtering out unwanted frequencies.

Microphone Noise Filter

Enables heavy filtering below 145 Hz to reduce mains induced noise (fans).

Microphone

Global Microphone Mute

Enabling this will disable the microphone entirely.

Microphone Volume

Select a volume for the microphone.



5.7.7 Advanced SIP

etwork Admin Users Time Provisioning Ad	dvanced Audio Advanced SIP Advanced Multicast
vanced SIP Settings	
ieneral	
SIP Transportation	Auto
SIPS Scheme	
Validate Server Certificate	OEnabled Disabled If 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)	OEnabled Obsabled Only enable this option if the SIP server supports RFC 5626.
Outbound Proxy	
Register Period (seconds)	3600
SRTP SDP SRTP Offer	Disabled V
NAT	
Media NAT	ONOR ICE OSTUN
TURN Server	
TURN User	
TURN Password	
Server Redundancy	
Server Redundancy Server Redundancy Feature (Multiple SIP Server Suppo	
Server Redundancy Server Redundancy Feature (Multiple SIP Server Suppo Backup Server #1	
Server Redundancy Server Redundancy Feature (Multiple SIP Server Suppo Backup Server #1 Backup Server #2	ort) Enabled Disabled 120 seconds (2 minutes) 17 me to wait between sending monitoring packets to each server. Inactive servers are always polled and
Server Redundancy Server Redundancy Feature (Multiple SIP Server Suppo Backup Server #1 Backup Server #2 Polling Interval (seconds)	Image: Seconds (2 minutes) Im
Server Redundancy Server Redundancy Feature (Multiple SIP Server Suppo Backup Server #1 Backup Server #2 Polling Interval (seconds) Poll Active Server	Image: Seconds (2 minutes) Im
Server Redundancy Server Redundancy Feature (Multiple SIP Server Suppo Backup Server #1 Backup Server #2 Polling Interval (seconds) Poll Active Server Automatic Fallback	Image: Sign point Image: Sign point <t< td=""></t<>
Server Redundancy Server Redundancy Feature (Multiple SIP Server Suppo Backup Server #1 Backup Server #2 Polling Interval (seconds) Poll Active Server Automatic Failback Polling Method	Image: Sign point Image: Sign point <t< td=""></t<>
Server Redundancy Server Redundancy Feature (Multiple SIP Server Support Backup Server #1 Backup Server #2 Polling Interval (seconds) Poll Active Server Automatic Failback Polling Method Interoperability	Image: Stress of the state of the stress
Server Redundancy Server Redundancy Feature (Multiple SIP Server Support Backup Server #1 Backup Server #2 Polling Interval (seconds) Poll Active Server Automatic Fallback Polling Method Interoperability Keep-Alive Method	ert) Enabled ODisabled I20 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). Cenabled ©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. Enabled ODisabled Reconnect with a higher priority server once available, even if the backup connection is still working. SIP NOTIFY OSIP OPTIONS SIP message used to poll servers in order to monitor their availability. None Opuble CRLF This setting will enable sending periodic CRLF messages for both UDP and TCP connections. Cenabled ODisabled Use phemeral 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

Figure 19: Advanced SIP Setting



General

SIP Transportation

Which transport layer protocol to use for SIP messages. Setting 'SIP Transportation' to 'TLS', ensures the encryption of SIP traffic.

SIPS Scheme

Only visible when 'SIP Transportation' set to 'TLS'. Enabling SIPS Scheme requires the SIP connection from endpoint to endpoint to be secure.

Validate Server Certificate

Validate the SIP server against common certificate authorities.

SIP Outbound Support (RFC 5626)

Enable this option to support best networking practices according to RFC 5626. This option should generally be enabled if the Algo device is being registered with a hosted server or if TLS is being used for SIP Transportation.

Outbound Proxy

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

Register Period (seconds)

Maximum requested period of time where the 8410/8420 will re-register with the SIP server. Default setting is 3600 seconds (1 hour). Only change if instructed otherwise.

<u>SRTP</u>

SDP SRTP Offer

Setting 'SDP SRTP Offer' to 'Optional', means the SIP call's RTP data will be left unencrypted if the other party does not support SRTP. Setting 'SDP SRTP Offer' to 'Standard', encrypts RTP voice data, meaning the normal audio RTP packets will now be secure (SRTP). This means SIP calls will be rejected if other party does not support SRTP. The 'Standard' option secures the audio data between parties, by making sure that it's not left out in the open for third parties to later reconstruct and listen to.

<u>NAT</u>

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

Server Redundancy Feature

Two secondary SIP servers may be configured. The 8410/8420 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 the primary server is unreachable, the 8410/8420will attempt to register with the backup servers. If enabled, the 8410/8420 will always attempt to register with the highest priority server.

Backup Server #2

If backup server #1 is unreachable, the 8410/8420 will attempt to register with the 2nd backup server. If enabled, the 8410/8420 will always attempt to register with the highest priority server.

Polling Intervals (seconds)

The 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. This may also be handled automatically by other regular events and can be disabled to reduce network traffic.

Automatic Fallback

This enables device to reconnect with a higher priority server once available, even if the backup connection is still fine.



Polling Method

A SIP message used to poll servers to monitor availability.

Interoperability

Keep-Alive Method

If Double CRLF is selected, the 8410/8420 will send a packet every 30 seconds (recommended value) to maintain connection with the SIP Server if behind NAT.

Keep-Alive Interval

This is the interval in seconds that the CRLF message should be sent.

Use Outgoing TLS port in SIP Headers

Use ephemeral port number from outgoing SIP TLS connection instead of the 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

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

Allow Missing Subscription-State Headers

When enabled, this allows SIP NOTIFY messages that do not contain a 'Subscription-State' header.


5.7.8 Advanced Multicast

Iticast mode can be set in "B Transmitter Settings	asic Settings > <u>Multicast</u> ".	
rightamiller setunds		
Transmitter Output Codec		G.711 ulaw
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 (R RTCP Port Selection	RTCP)	●Disabled ○Next Higher Port ○Multiplexed on Same Port
		(1) 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	IP Address and	
	IP Address and Port	Page Tone
Zone		Page Tone
	Port	
Zone Priority Call (DTMF:9)	Port 224.0.2.60:50000	<use default="" page="" td="" ton="" ✔<=""></use>
Zone Priority Call (DTMF:9) All Call (DTMF:0/8)	Port 224.0.2.60:50000 224.0.2.60:50001	<use <ul="" default="" page="" ton="" ✓=""> <use default="" li="" page="" ton="" ✓<=""> </use></use>
Zone Priority Call (DTMF:9) All Call (DTMF:0/8) Zone 1 (DTMF:1)	Port 224.0.2.60:50000 224.0.2.60:50001 224.0.2.60:50002	<use default="" page="" td="" ton="" ∨<=""> <use default="" page="" td="" ton="" ∨<=""> <use default="" page="" td="" ton="" ∨<=""></use></use></use>
Zone Priority Call (DTMF:9) All Call (DTMF:0/8) Zone 1 (DTMF:1) Zone 2 (DTMF:2)	Port 224.0.2.60:50000 224.0.2.60:50001 224.0.2.60:50002 224.0.2.60:50003	<use default="" page="" td="" ton="" ∨<=""> <use default="" page="" td="" ton="" ∨<=""> <use default="" page="" td="" ton="" ∨<=""> <use default="" page="" td="" ton="" ∨<=""></use></use></use></use>
Zone Priority Call (DTMF:9) All Call (DTMF:0/8) Zone 1 (DTMF:1) Zone 2 (DTMF:2) Zone 3 (DTMF:3)	Port 224.0.2.60:50000 224.0.2.60:50001 224.0.2.60:50002 224.0.2.60:50003 224.0.2.60:50004	<use default="" page="" td="" ton="" ∨<=""> <use default="" page="" td="" ton="" ∨<=""></use></use></use></use></use>
Zone Priority Call (DTMF:9) All Call (DTMF:0/8) Zone 1 (DTMF:1) Zone 2 (DTMF:2) Zone 3 (DTMF:3) Zone 4 (DTMF:4)	Port 224.0.2.60:50000 224.0.2.60:50001 224.0.2.60:50002 224.0.2.60:50003 224.0.2.60:50004 224.0.2.60:50005	<use default="" page="" td="" ton="" ∨<=""> <use default="" page="" td="" ton="" ∨<=""></use></use></use></use></use></use></use>

Figure 20: Advanced multicast - transmitter settings



Note

The settings on this tab are only visible when in Sender or Receiver multicast mode.

The default pre-populated multicast addresses above will work in most cases and should only be altered for rare cases.

Transmitter Settings

Transmitter Output Codec

This is the audio encoding format used by the Transmitter device when sending output to the Receivers.



Output Packetization Time (milliseconds)

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

Multicast TTL

The multicast time to live (TTL) setting should only be changed 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.

Receiver Settings

RTCP Port Selection

Select the port on which RTCP 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.

Polycom Receiver Tones

Available if Multicast Receiver and 'Polycom Group Page' or 'Polycom Push-to-Talk' are selected in the *Basic Settings* \rightarrow *Multicast* tab. A tone may be set for any of the 25 Polycom Groups. If using an Algo device as a Multicast Sender, it is recommended to set the Receiver tones to 'None' to avoid conflicts, as the Algo devices already multicast a tone by default.

Basic Zone Definitions

Zones

The 'Expanded' Receiver zones can be enabled/disabled in *Basic Settings* \rightarrow *Multicast*. Default IP addresses and ports may be revised for any given zone in the table.



Important

Ensure that the Address and Port settings are the same for all Sender and Receiver devices.

Page Tone and Page Volume

Sender Mode: By default, the same tone can be set for all Receiver zones in the *Basic Settings* \rightarrow *Features* tab. Unique paging tones may be revised for any given zone in the table above.

Receiver Mode: When an Algo device is the multicast Sender, a page tone will play on the Receiver device, so it is recommended to set the Receiver 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 Receivers (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 Receiver zones in the *Basic Settings* \rightarrow *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 Receiver speakers.

5.8 System

5.8.1 Maintenance

Status Basic Settings Display Additional Features	Advanced Settings System Logout
Maintenance Firmware File Manager Tones Sy	rstem Log Credits About
System Maintenance	
Backup / Restore Configuration	
Download Configuration File	
Restore Configuration File	Choose File No file chosen
Restore Configuration to Defaults	Restore Defaults
Backup / Restore All User Files Backup in zip format includes configuration file and all uploa Download Backup Zip File	ded files.
Restore from Backup Zip File	Choose File No file chosen
Restore All Settings and Files to Defaults	Restore Defaults and Delete Files () All preloaded and uploaded files, including tone files, will be deleted.
Reboot	
Reboot the device	Reboot

Figure 21: Maintenance settings

Backup / Restore Configuration

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 8410 IP Display Speaker settings to factory default values.



Backup / Restore All User Files

Download Backup Zip File

Saves the device settings (configuration) and all the files in File Manager: certificates, licenses, and tones to a backup zip file.

Restore from Backup Zip File

Restores the device settings (configuration) and all the files in *File Manager*: certificates, licenses, and tones from a backup zip file.

Restore All Settings and Files to Defaults

Resets the device settings (configuration) and all the files in *File Manager*: certificates, licenses, and tones to factory default values.

Reboot

Reboot the Device

Reboots the device.

5.8.2 Firmware

Status	Basic Settings	Display	Addition	al Features	Advance	d Settings	System	Logou	ut					
Mainten	ance Firmwar	e File Ma	anager	Tones Sy	stem Log	Credits	About							
Firmwa	are													
Insta	lled Firmware													
	ct Firmware					algo-8410	-5.3_beta3.	1]
Froud	ct rinnware					alg0-0410	-5.5_06(85.	-						
Onlin	e Upgrade													
Check	for Firmware Up	dates				Check	i i							

Custo	m Upgrade													
Metho	d					From	Local Files	OFrom URL						
Signe	d Firmware File					Choose I	File No file	chosen						
Allow	Downgrade					Enable	ed ODisable	ed						
						Allow pr	oduct or base	firmware to b	be downgrade	ed to an olde	r patch vers	ion.		
						Enabling	this option c	ould cause up	grade issues.	Please cont	act support	if necessar	у.	
						👚 Upgra	de							
<u>.</u>														

Figure 22: Firmware settings



Installed Firmware

Product Firmware

Shows the current firmware on the device.

Online Upgrade

Check for Firmware Updates

Check for the latest firmware. If firmware is current, **Latest Firmware** will show as 'Firmware up to date'. If firmware needs to be upgraded, the new firmware availability will be listed.

Custom Upgrade

Method

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

Signed Firmware File

How to upgrade Firmware

- 1. From the top menu, go to System \rightarrow Firmware.
- 2. In the Upgrade section, press **Choose File** and select the 8410 firmware file to upload. Note that both FW firmware and MD5 checksum files must be loaded.
- 3. Press Upgrade.
- 4. After the upgrade is complete, confirm that the firmware version has changed (refer to top right of Control Panel).

Allow Downgrade

Allow product or base firmware to be downgraded to an older patch version. Enabling this option could cause upgrade issues.



5.8.3 File Manager

$\begin{array}{ c c c c } \hline & & \\ \hline \\ \hline$							
	≝″≣ [*]	Name	Date	Туре	Size		
🕞 Files		🗀 certs	02/08/2023 03:37 PM	Folder			
> 🗀 certs		🗀 debug	03/24/2020 10:26 AM	Folder			
🗀 debug		🗅 icons	02/08/2023 03:42 PM	Folder			
icons		🗅 images	02/08/2023 03:45 PM	Folder			
images		🗅 license	11/03/2016 10:16 AM	Folder			
🗅 tones		🗅 tones	02/08/2023 03:37 PM	Folder			
		🗋 display.db	02/08/2023 03:46 PM	File	12KE		
		🖹 user.conf	02/09/2023 01:26 PM	Text File	20.079KE		

Figure 23: File manager settings

Uploading Custom Audio Files

Custom audio files may be uploaded into memory (1 GB) to play for notification applications. Place your audio files into the **tones** directory.

An existing file may also be modified. Download the original file and right clicking the tone and selecting 'Download', making the desired changes, and then uploading the new version with a different name. Audio files must be in the following format:

- WAV format
- 8 kHz or 16 kHz sampling rate
- 16-bit PCM, or u-law
- Mono
- Smaller than 200 MB

File names must be limited to 32 characters, with no spaces. For further instructions reference the Custom Tone Conversion and Upload Guide.



5.8.4 Tones

Status Basic Settings Display Additional Features Advanced Settings System Logout	
Maintenance Firmware File Manager Tones System Log Credits About	
Tones Use the "System > File Manager" tab to upload tone files to "tones" subdirectory.	
Files	
Download and Install Ring Tones from the Algo Server Jownload and Install Image: Im	
Cache	
Rebuild Tone Cache Files Rebuild All Image: 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 V Play Loop Stop	

Figure 24: Tones settings

Tone Files Included in Memory

The 8410/8420 includes several pre-loaded audio files that can be selected to play for various events. The web interface allows selection of the file and the ability to play it immediately over the speaker for testing (available in *Basic Settings* \rightarrow *Features*). Files may also be deleted or renamed.

5.8.5 System Log

System log files are automatically created and assist with troubleshooting in the event the 8410/8420 does not behave as expected.

	Status Ba	ic Settings	Display Addi	tional Features	Advance	ed Settings	System	Logout				
	Maintenance	Firmware	File Manager	Tones Sy	stem Log	Credits	About					
S	ystem Lo	g										
	Download Log Files											
	Log File Jownload syslog.txt											
						View						ĺ
	l							 	 			

Figure 25: System log settings

5.9 Logout

Log out of the 8410/8420 web interface.

6 SPECIFICATIONS

Power PoE-Powered Recommended: PoE++ (IEEE 802.3bt Type 3 Max 51W) Optional: PoE+ (IEEE 802.3bt Type 2 Max 25.5W) SIP SIP SIP Extensions S0 Page & 10 Alerting/Ring extensions with multicast scalability Transport Protocols UDP, RTP, TCP Security TLS, MTLS, SRTP Multicast Compatibility TLS, MTLS, SRTP Multicast RTP Multicast (Send and Receive 50 Zones) Third-Party Multicast PolyTM Group Page, SinglewireTM InformaCast, Syn-AppsTM Revolution Digital IO Normally open or normally closed dry contact supervision. Compatible with Algo 1202, 1203, and 1204 Accessories. API REST ful Audio Codes G.711 A-law, G.711 u-law, G.722, Opus 48 kHz. Speakers 3 x Neodymlum magnet with polypropylene cone Frequery Response 150 – 19,000 Hz (10 dB) Driver Sensitivity 92 dBA 1m/1W (1 kHz sine wave) SPI Peak 106 dBA/1m Max Audio Power 8Watts Audio Controls Volume, AGC, Latency, LF Cut Audio Memory + Format 1 GB user space for WAV or MP3 files Microphone Ominidirectional – talkback and ambitent noise monitoring		Table 1: 8410 Specification Table
Optional: PoE+ (IEEE 802. 3at Type 2 Max 25.SW)SIPSIPSIP ExtensionsS0 Page & 10 Alerting/Ring extensions with multicast scalabilityTransport ProtocolsUDP, RTP, TCPSecurityTIS, MTLS, SRTPMulticastRTP Multicast (Send and Receive 50 Zones)Third-Party MulticastPolyTM Group Page, SinglewireTM InformaCast, Syn-AppsTM RevolutionDigital IONormally open or normally closed dry contact supervision. Compatible with Algo 1202, 1203, and 1204 Accessories.APIRESTfulAudio CodesG.711 A-law, G.711 u-law, G.722, Opus 48 kHz.Speakers3 x Neodymium magnet with polypropylene coneFrequency Response150 - 19,000 Hz (- 10 dB)Oriver Sensitivity92 dBA 1m/1W (1 kHz sine wave)Spl0 condinal - talkback and ambient noise monitoringMulticationNondilycectional - talkback and ambient noise monitoringAudio Chrons0 volume, AGC, tatency, LF CutMicrophone0 moldirectional - talkback and ambient noise monitoringAudio Chrons16 Suer space for WAV or MP3 filesMicrophone0 nondirectional - talkback and ambient noise monitoringAudio Chrons1920 x 1080 @ fipsKierophone1920 x 1080 @ fopsKierophone1920 x	Power	
Transport ProtocolsUDP, RTP, TCPSecurityTLS, MTLS, SRTPMulticast CompatibilityMulticast (Send and Receive 50 Zones)Third-Party MulticastPolyTM Group Page, SinglewireTM InformaCast, Syn-AppsTM RevolutionDigital IOMulticast (Send and Receive 50 Zones)Relay Configured as InputNormally open or normally closed dry contact supervision. Compatible with Algo 1202, 1203, and 1204 Accessories.APIRESTfulAudio CodesG. 711 A-law, G. 711 u-law, G. 722, Opus 48 kHz.Speakers3 x Neodymium magnet with polypropylene coneFrequency Response150 – 19,000 Hz (- 10 dB)Driver Sensitivity92 dBA 1m/1W (1 kHz sine wave)SPLPeak 106 dBA/1mAudio ControlsKoutisch and ambient noise monitoringAudio ControlsVolume, AGC, Latency, LF CutAudio ControlsVolume, AGC, Latency, LF CutAudio ControlsVolume, AGC, Latency, LF CutAudio Controls13.5" x 7.6" (34.4 cm x 19.4 cm)Kesolution1920 x 1080 @ 60 fpsVideo1920 x 1080 @ 60 fpsVideo88 deg (horizontal and vertical)Color16.7 M		
SecurityTLS, STPMulticast CompatibilityMulticastRTP Multicast (send and Receive 50 Zones)Third-Party MulticastPolyTM Group Page, SinglewireTM InformaCast, Syn-AppsTM RevolutionDigital IOGlaigen as InputNormally open or normally closed dry contact supervision. Compatible with Algo 1202, 1203, and 1204 Accessories.APIReSTfulAudio CodesG.711 A-law, G.711 u-law, G.722, Opus 48 kHz.Speakers3 x Neodymium magnet with polypropylene coneFrequency Response150 – 19,000 Hz (- 10 dB)Driver Sensitivity92 dBA 1m/1W (1 kHz sine wave)SPLPeak 106 dBA/1mMax Audio Power8 WattsAudio ControlsOmnidirectional – takback and ambient noise monitoringAudio ControlsOutput chart and precisesGuite ControlsCache to memory and releaseVideoICD Viewing Area13.5" x 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	SIP Extensions	50 Page & 10 Alerting/Ring extensions with multicast scalability
MulticastCompatibilityMulticastRTP Multicast (Send and Receive 50 Zones)Third-Party MulticastPolyTM Group Page, SinglewireTM InformaCast, Syn-AppsTM RevolutionDigital IORelay Configured as InputNormally open or normally closed dry contact supervision. Compatible with Algo 1202, 1203, and 1204 Accessories.APIRESTfulAudio CodesG.711 A-law, G.711 u-law, G.722, Opus 48 kHz.Speakers3 x Neodymium magnet with polypropylene coneFrequency Response150 – 19,000 Hz (- 10 dB)Driver Sensitivity92 dBA 1m/1W (1 kHz sine wave)SPLPeak 106 dBA/1mMax Audio Power8 WattsAudio ControlsOmnidirectional – talkback and ambient noise monitoringAudio ControlsVolume, AGC, Latency, LF CutAnti-feedback DelayCache to memory and releaseVideo150 × 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	Transport Protocols	UDP, RTP, TCP
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Relay Configured as InputNormally open or normally closed dry contact supervision. Compatible with Algo 1202, and 1204 Accessories.APIRESTfulAudioRESTfulAudio CodesG.711 A-law, G.711 u-law, G.722, Opus 48 kHz.Speakers3 x Neodymium magnet with polypropylene coneFrequency Response150 – 19,000 Hz (- 10 dB)Driver Sensitivity92 dBA 1m/1W (1 kHz sine wave)SPLPeak 106 dBA/1mMax Audio Power8 WattsAudio ControlsOmnidirectional – talkback and ambient noise monitoringMicrophoneCache to memory and releaseVideo13.5" x 7.6" (34.4 cm x 19.4 cm)Cito Viewing Area1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Citoor16.7 M	Third-Party Multicast	PolyTM Group Page, SinglewireTM InformaCast, Syn-AppsTM Revolution
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Audio CodesG.711 A-law, G.711 u-law, G.722, Opus 48 kHz.Speakers3 x Neodymium magnet with polypropylene coneFrequency Response150 - 19,000 Hz (- 10 dB)Driver Sensitivity92 dBA 1m/1W (1 kHz sine wave)SPLPeak 106 dBA/1mMax Audio Power8 WattsAudio Memory + Format1 GB user space for WAV or MP3 filesMicrophoneOmnidirectional - talkback and ambient noise monitoringAudio ControlsVolume, AGC, Latency, LF CutAnti-feedback DelayCache to memory and releaseVideo1920 x 1080 @ 60 fpsKesolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	ΑΡΙ	RESTful
Speakers3 x Neodymium magnet with polypropylene coneFrequency Response150 – 19,000 Hz (- 10 dB)Driver Sensitivity92 dBA 1m/1W (1 kHz sine wave)SPLPeak 106 dBA/1mMax Audio Power8 WattsAudio Memory + Format1 GB user space for WAV or MP3 filesMicrophoneOmnidirectional – talkback and ambient noise monitoringAudio ControlsVolume, AGC, Latency, LF CutAnti-feedback DelayCache to memory and releaseVideoILCD Viewing Area13.5" x 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	Audio	
Frequency Response150 – 19,000 Hz (- 10 dB)Driver Sensitivity92 dBA 1m/1W (1 kHz sine wave)SPLPeak 106 dBA/1mMax Audio Power8 WattsAudio Memory + Format1 GB user space for WAV or MP3 filesMicrophoneOmnidirectional – talkback and ambient noise monitoringAudio ControlsVolume, AGC, Latency, LF CutAnti-feedback DelayCache to memory and releaseVideo13.5" x 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	Audio Codes	G.711 A-law, G.711 u-law, G.722, Opus 48 kHz.
Driver Sensitivity92 dBA 1m/1W (1 kHz sine wave)SPLPeak 106 dBA/1mMax Audio Power8 WattsAudio Memory + Format1 GB user space for WAV or MP3 filesMicrophoneOmnidirectional – talkback and ambient noise monitoringAudio ControlsVolume, AGC, Latency, LF CutAnti-feedback DelayCache to memory and releaseVideoILCD Viewing Area13.5" x 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	Speakers	3 x Neodymium magnet with polypropylene cone
SPLPeak 106 dBA/1mMax Audio Power8 WattsAudio Memory + Format1 GB user space for WAV or MP3 filesMicrophoneOmnidirectional – talkback and ambient noise monitoringAudio ControlsVolume, AGC, Latency, LF CutAnti-feedback DelayCache to memory and releaseVideoIS.5" x 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	Frequency Response	150 – 19,000 Hz (- 10 dB)
Max Audio Power8 WattsAudio Memory + Format1 GB user space for WAV or MP3 filesMicrophoneOmnidirectional – talkback and ambient noise monitoringAudio ControlsVolume, AGC, Latency, LF CutAnti-feedback DelayCache to memory and releaseVideoI3.5" x 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	Driver Sensitivity	92 dBA 1m/1W (1 kHz sine wave)
Audio Memory + Format1 GB user space for WAV or MP3 filesMicrophoneOmnidirectional – talkback and ambient noise monitoringAudio ControlsVolume, AGC, Latency, LF CutAnti-feedback DelayCache to memory and releaseVideoILCD Viewing Area13.5" x 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	SPL	Peak 106 dBA/1m
MicrophoneOmnidirectional – talkback and ambient noise monitoringAudio ControlsVolume, AGC, Latency, LF CutAnti-feedback DelayCache to memory and releaseVideoICD Viewing Area13.5" x 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	Max Audio Power	8 Watts
Audio ControlsVolume, AGC, Latency, LF CutAnti-feedback DelayCache to memory and releaseVideoILCD Viewing Area13.5" x 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	Audio Memory + Format	1 GB user space for WAV or MP3 files
Anti-feedback DelayCache to memory and releaseVideoLCD Viewing Area13.5" x 7.6" (34.4 cm x 19.4 cm)Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	Microphone	Omnidirectional – talkback and ambient noise monitoring
Video LCD Viewing Area 13.5" x 7.6" (34.4 cm x 19.4 cm) Resolution 1920 x 1080 @ 60 fps Viewing Angle 88 deg (horizontal and vertical) Color 16.7 M	Audio Controls	Volume, AGC, Latency, LF Cut
LCD Viewing Area 13.5" x 7.6" (34.4 cm x 19.4 cm) Resolution 1920 x 1080 @ 60 fps Viewing Angle 88 deg (horizontal and vertical) Color 16.7 M	Anti-feedback Delay	Cache to memory and release
Resolution1920 x 1080 @ 60 fpsViewing Angle88 deg (horizontal and vertical)Color16.7 M	Video	
Viewing Angle 88 deg (horizontal and vertical) Color 16.7 M	LCD Viewing Area	13.5" x 7.6" (34.4 cm x 19.4 cm)
Color 16.7 M	Resolution	1920 x 1080 @ 60 fps
	Viewing Angle	88 deg (horizontal and vertical)
Backlight LED Auto-dimmable	Color	16.7 M
	Backlight	LED Auto-dimmable
Contrast 1000:1	Contrast	1000:1



Network	
Network	IPv4, IPv6, DHCP, VLAN, MDNS
Link Layer	LLDP, CDP
QOS	DSCP (SIP, RTP, RTCP)
Web Interface	HTTP, HTTPS
Provisioning	TFTP, FTTP, HTTP, HTTPS, DHCP Options 66, 150, 160 Reboot via SIP Check-sync
NAT	STUN, TURN, CRLF Keep Alive, SIP Outbound
Address Resolution	DNS, SRV Record
Supervision	SNMP V1.3, RTCP, Algo 8300, ADMP
Redundancy	Secondary and tertiary SIP server
Environmental & Mechanical	
Environmental	32 to 104 deg F (0 to + 40 degC); 10-95% RH non-condensing. Dry indoor locations only.
Dimensions (Product)	19.5″ x 9.5″ x 1.75″
Compliance	
IEC 62368-1, IEEE 802.3-2018, RoHS	, CE, FCC Class A, CISPR 32 Class A, CISPR 24, CSA/UL (USA & Canada),

Firmware

These specifications refer to the Algo 8410 running on firmware 5.2 and above.

Table 2: 8420 Specification Table

Power	
PoE-Powered	Recommended: PoE++ (IEEE 802.3bt Type 4 Max 71.3)
	Minimum: PoE+ (IEEE 802.3bt Type 3 Max 51W)
SIP	
SIP Extensions	50 Page & 10 Alerting/Ring extensions with multicast scalability
Transport Protocols	UDP, RTP, TCP
Security	TLS, MTLS, SRTP
Multicast Compatibility	
Multicast	RTP Multicast (Send and Receive 50 Zones)
Third-Party Multicast	PolyTM Group Page, SinglewireTM InformaCast, Syn-AppsTM Revolution
Digital IO	
Relay Configured as Input	Normally open or normally closed dry contact supervision. Compatible with Algo 1202, 1203, and 1204 Accessories.
ΑΡΙ	
ΑΡΙ	RESTful
Audio	
Audio Codes	G.711 A-law, G.711 u-law, G.722, Opus 48 kHz.
Speakers	6 x Neodymium magnet with polypropylene cone
Frequency Response	150 – 19,000 Hz (- 10 dB)
Driver Sensitivity	92 dBA 1m/1W (1 kHz sine wave)
Maximum SPL	100 dBA/1m
Max Audio Power	12 Watts
Audio Memory + Format	1 GB user space for WAV or MP3 files
Microphone	Omnidirectional – talkback and ambient noise monitoring
Audio Controls	Volume, AGC, Latency, LF Cut
Anti-feedback Delay	Cache to memory and release
Video	
LCD Viewing Area	Two LCD screens of 13.5" x 7.6" (34.4 cm x 19.4 cm)
Resolution	1920 x 1080 @ 60 fps
Viewing Angle	88 deg (horizontal and vertical)
Color	16.7 M
Backlight	LED Auto-dimmable
Contrast	1000:1



Network			
Network	IPv4, IPv6, DHCP, VLAN, MDNS		
Link Layer	LLDP, CDP		
QOS	DSCP (SIP, RTP, RTCP)		
Web Interface	HTTP, HTTPS		
Provisioning	TFTP, FTTP, HTTP, HTTPS, DHCP Options 66, 150, 160 Reboot via SIP Check-sync		
NAT	STUN, TURN, CRLF Keep Alive, SIP Outbound		
Address Resolution	DNS, SRV Record		
Supervision	SNMP V1.3, RTCP, Algo 8300, ADMP		
Redundancy	Secondary and tertiary SIP server		
Environmental & Mechanical			
Environmental	32 to 104 deg F (0 to + 40 degC); 10-95% RH non-condensing. Dry indoor locations only.		
Dimensions (Product)	19.5″ x 9.5″ x 3″		
Compliance			
IEC 62368-1, IEEE 802.3-2018, RoHS, CE, FCC Class A, CISPR 32 Class A, CISPR 24, CSA/UL (USA & Canada),			

Firmware

These specifications refer to the Algo 8410 running on firmware 5.2 and above.



7 FCC COMPLIANCE STATEMENT

his 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, and 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 his or her own expense.